Land Mobile Radio over IP Enhancement

The Land Mobile Radio over IP Enhancement feature allows Cisco multiservice routers to transport Land Mobile Radio (LMR) traffic over IP networks by modifying voice gateway functionality. LMR over IP enables LMR systems to extend beyond their traditional geographic limitations created by transmitter signal strength and enables interoperability, allowing public safety personnel in different agencies or jurisdictions to communicate with each other by radio on demand, in real time.

Note Some support restrictions apply to use of the Cisco Land Mobile Radio (LMR) over IP feature. See the "DISCLAIMER" section on page 164 for important information regarding Cisco support for this feature.

Throughout this document, references to LMR radios apply to all types of radios, including LMR, military, amateur, and others.

Feature History for Land Mobile Radio over IP Enhancement

Release	Modification
12.3(4)XD	This feature was introduced.
12.3(7)T	This feature was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This feature was integrated into Cisco IOS Release 12.3(14)T and support was provided for the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This feature was integrated into Cisco IOS Release 12.4(2)T. The **auto-control** keyword and *auto-dbm* argument were added to the **input gain** and **output attenuation** commands to enable automatic gain control.
12.4(2)T1	This feature was integrated into Cisco IOS Release 12.4(2)T1. The **e&m-lmr** keyword was added to the **ds0-group (E1)** command.
12.4(15)XY	This feature was integrated into Cisco IOS Release 12.5(1)T. The **lmr-tone** and **nte-tone** keywords were added to the **rtp payload-type** command.

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 Land Mobile Radio over IP Enhancement

- Install the appropriate Cisco IOS images on each router. Table 1 lists the images that support the Land Mobile Radio over IP Enhancement feature. Land Mobile Radio over IP Enhancement features require the spservicesk9 image at a minimum.
- Make sure you have the required amount of memory on each router. Table 2 lists the memory requirements for various platform images.
- Configure a working VoIP network.

Table 1 Cisco IOS Images for the Land Mobile Radio over IP Enhancement Feature

Platform	Software Image
Cisco 2600XM	• c2600-adventerprisek9-mz • c2600-advipvoicek9-mz • c2600-entservicesk9-mz • c2600-spservicesk9-mz
Cisco 2691	• c2691-adventerprisek9-mz • c2691-advipvoicek9-mz • c2691-entservicesk9-mz • c2691-spservicesk9-mz
Cisco 2800 (except Cisco 2801)	• c2800nm-adventerprisek9-mz • c2800nm-advipvoicek9-mz • c2800nm-entservicesk9-mz • c2800nm-spservicesk9-mz
Cisco 3660	• c3660-jk9o3s-mz

Table 1 Cisco IOS Images for the Land Mobile Radio over IP Enhancement Feature

Platform	Software Image
Cisco 3700	• c3700-adventerprisek9-mz • c3700-advipvoicek9-mz • c3700-entservicesk9-mz • c3700-spservicesk9-mz
Cisco 3800	• c3800-adventerprisek9-mz • c3800-advipvoicek9-mz • c3800-entservicesk9-mz • c3800-spservicesk9-mz

Table 2 Memory Requirements for the Land Mobile Radio over IP Enhancement Feature

Platform	Memory Required for spservices	Memory Required for adventerprisek9
Cisco 2651XM	128 MB	128 MB (I/O memory < 5%)
Cisco 2691	128 MB	160 MB (128 + 32)
Cisco 3725	128 MB	160 MB (128 + 32)
Cisco 3745	128 MB	160 MB (128 + 32)
Cisco 2800 Series	128 MB	160 MB (128 + 32)
Cisco 3825	128 MB	160 MB (128 + 32)
Cisco 3845	128 MB (I/O memory< 5%)	160 MB (128 + 32)

Restrictions for Land Mobile Radio over IP Enhancement

- VIC-2E/M voice interface cards (VICs) work with the NM-2V network modules only.
- VIC2-2E/M VICs work with the NM-HD-2V and NM-HD-2VE network modules only.
- The NM-2Vnetwork module is not supported on the Cisco 2800 series and Cisco 3800 series platforms.
- When performing the following configuration tasks remember that even after you issue the correct configuration commands, some configurations require a new call to be set up or a **shutdown** command then a **no shutdown** command to be issued before taking effect. Table 3 shows the configuration items, their associated commands, and the events that must occur for some of the configuration items to take effect. In Table 3, the values in the New Call Set Up and Shut, No Shut columns have the following meanings:
  - A Yes in the New Call Set Up column indicates that the configuration task does not take effect until a new call is set up.

- A No in the New Call Set Up column indicates that the configuration task takes effect without a new call being set up.
- A Yes in the Shut, No Shut column indicates that the configuration task does not take effect until a **shtudown** command then a **no shutdown** command is issued on the port.
- A No in the Shut, No Shut column indicates that the configuration task takes effect without a **shtudown** command then a **no shutdown** command being issued on the port.

Table 3 Actions Required to Enable Configuration Items

	Action Required	
Configuration Item (Command Name)	**New Call Set Up**	**Shut, No Shut**
Duplex mode (lmr duplex)	Yes	Yes
Input gain (input gain)	No	No
Output attenuation (output attenuation)	No	No
VOX noise (threshold noise)	Yes	No
COR polarity (define rx-bits)	No	No
PTT polarity (define tx-bits)	No	No
PTT tone (inject tone)	No	No
PTT tone sequence (inject tone, inject pause)	No	No
PTT timeout (timeout ptt)	No	No
COR timeout (timeout ptt)	No	No
Transmit buffering 0–1.5 seconds (timing delay-voice tdm)	Yes	No
PTT hangover (timing hangover)	Yes	No
COR delay (timing hookflash-input)	No	No
Duplex hold-off (timing ignore m-lead	No	No
Filter out guard tone 1950 or 2175 (digital-filter)	Yes	No
Automatic gain control (input gain auto-control)	Yes	No
COR idle 1950 or 2175 detect (inject guard-tone idle)	Yes	No
PTT idle 1950 or 2175 encode (inject guard-tone idle)	Yes	No
Voice class tone (voice class tone-signal)	Yes	No
E&M interface type (type)	Yes	Yes
2- or 4-wire operation (operation)	Yes	No
Echo cancellation (echo-cancel enable)	Yes	No
Comfort noise (comfort-noise)	Yes	No
Any dial peer change	Yes	No

Information About Land Mobile Radio over IP Enhancement

To configure Land Mobile Radio over IP Enhancement, you need to understand the following concepts:

- LMR Feature Enhancements in Cisco IOS Release 12.4(15)XY, page 5

LMR Feature Enhancements in Cisco IOS Release 12.4(15)XY

This section describes the new features in Cisco IOS Release 12.4(15)XY.

Bootup Without Radio Keying

In this release, the router can boot and reboot without keying the attached radio. This capability is present only with a specific combination of VIC and Cisco IOS software release. The behavior of the various VIC and Cisco IOS software release combinations is described in Table 4.

Table 4 Bootup Behavior for VIC and Cisco IOS Software Release Combinations

VIC with Hardware Version 5.1 or Later	Cisco IOS Software Image with Change	Behavior
Yes	Yes	Will not key the radio during reboot
Yes	No	VIC may not function as expected
No	No	Will key the radio during reboot
No	Yes	No impact and will key up radio during reboot

Transmit Delay

LMR gateways are prone to front-end clipping when they are connecting to a trunked radio system because of the time required to acquire a channel. This feature provides a configurable delay before the voice packet is played out to compensate for the channel acquisition time. The maximum delay is 1.5 seconds. The transmit delay is available for LMR ports only.

Tone Injection

Many conventional radio systems use in-band tone signaling to indicate activity, key the transmitter, and control channel selection. There are three phases of tone signaling:

- Wakeup tone—A tone of a specific duration and frequency that acts as preamble to base stations to indicate that additional signaling is coming.
- Frequency selection (or control) tone—One of a range of tones used to select a frequency (channel) for the audio.
- Guard tone— A tone of a specific frequency that is maintained as long as there is activity on the channel. This tone indicates that the channel has been seized.

To eliminate the need for tones to be passed across the WAN, this feature provides the capability to inject tones at the gateway. Static tone injection is one fixed sequence of single tones, no more than ten tones or pauses in a given sequence, used on all transmissions from that voice port to the attached LMR system. Static tone injection begins with E-lead activity and ends when the hangover time expires on voice playout. The tone sequence comprises some combination of the following:

- Single tone—Of fixed frequency, duration, and amplitude.
- Pause—Of fixed duration.
- Guard tone—Of fixed frequency and amplitude. To be played out with the voice packet, for the duration of the voice packet.
- Idle tone—To be played in the absence of voice packets. Idle tone and guard tone are mutually exclusive.

If you configure injected tones, be sure to use the command to configure a delay before the voice packet is played out. Configuring a delay prevents the voice packet from being overwritten by the injected tones. The delay must be equal to the sum of the durations of the injected tones and pauses in the tone-signal voice class.

Digital Filter

The digital filter improves voice quality by preventing transmission of the guard tone from the LMR system to the VoIP network. The digital filter can be configured to filter out either 2175 Hz or 1950 Hz through the command. Only one of these frequencies can be filtered out at a time. Filtering is performed by the digital signal processor (DSP). Digital filtering is disabled by default. The digital filter is available for LMR ports only.

Improved Debugging Capabilities

The command enhances debugging capability by providing LMR-related dynamic and static information along with detailed voice port and active call information, eliminating the need to use several different debugging commands. Information displayed by the **show voice lmr** command allows for improved troubleshooting of the interface between the LMR gateways and LMR radio systems.

LED Troubleshooting Enhancement

The default behavior of the E&M LED is to indicate voice activity only. You can now configure the LED to provide industry standard behavior using the command. The LED indicates transmission and reception as follows:

- Red—Voice from the network toward the radio is active (E-lead)
- Green—Voice from the radio towards the network is active (M-lead or voice activity detection (VAD))
- Yellow—Voice in both directions is active

Configurable PTT Timeout

To limit extended radio transmission, a configurable Push To Talk (PTT) timeout feature has been added. The PTT timeout can be configured for different durations, up to 30 minutes, on a port-by-port basis using the command. This timeout can be configured on LMR ports only.

Automatic Gain Control

Because of radio network loss and other environmental factors, the speech level arriving at a router from an LMR system can be very low. Automatic gain control, which is performed by the DSP, adjusts speech to a comfortable volume when it becomes too loud or too soft. You can use automatic gain control to ensure that the speech is played back at a more comfortable level. Because the gain is inserted digitally, the background noise can also be amplified. You can configure input gain and output attenuation in decibels per milliwatt (dBm). Automatic gain control can be configured on LMR ports only and is mutually exclusive with set input-gain function.

Connection Types

The Land Mobile Radio over IP Enhancement feature works with the connection types discussed in the following sections:

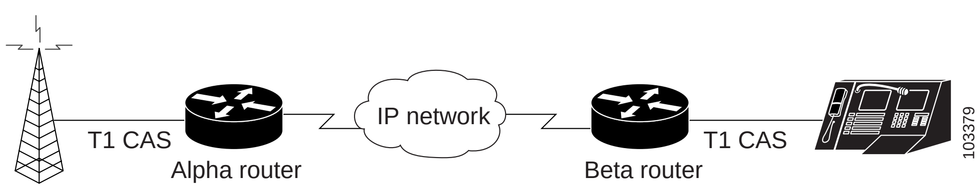
- Connection Trunk, page 7
- PLAR, page 7
- VoIP Multicast, page 8

Connection Trunk

LMR features can be integrated into traditional point-to-point trunk connections. VoIP simulates two types of trunk connections—*switched* and *permanent*—that can be configured for both analog and digital systems. Switched connections are discussed in the "PLAR" section. The connections are created with the **connection** command. Refer to the *Cisco IOS Voice Command Reference*, Release 12.3 T, for a description of the **connection** command.

The **connection trunk** command creates a permanent call that is connected as soon as the E&M voice ports on the routers on each end are brought up (see Figure 1). Permanent calls pass limited telephony signaling and operate without collecting digits or requiring changes to the overall dial plan.

Figure 1 Connection Trunk Configuration

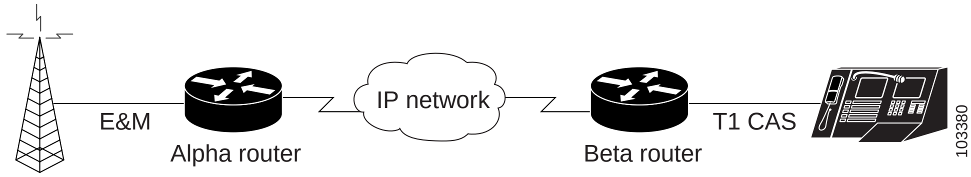


The calls simulate a permanent tie-line between a radio system and its dispatch console. Both ends must be configured for E&M voice port signaling.

PLAR

Private line, automatic ringdown (PLAR) is a switched simulated connection that can be configured for both analog and digital systems. LMR features can be integrated into the traditional point-to-point PLAR connections. When a switched call is configured (see Figure 2), the user can make a call without dialing any digits. The router uses the digits configured with the **connection plar** command internally to send the call to a dial peer.

Figure 2 Connection PLAR Configuration



The switched call configuration works with any type of voice port (ear and mouth (E&M), Foreign Exchange Office (FXO), or Foreign Exchange Station (FXS)) and can be used without any effect on an existing dial plan. Switched call configuration is commonly used to connect PBXs in which the remote devices appear to be physical extensions.

VoIP Multicast

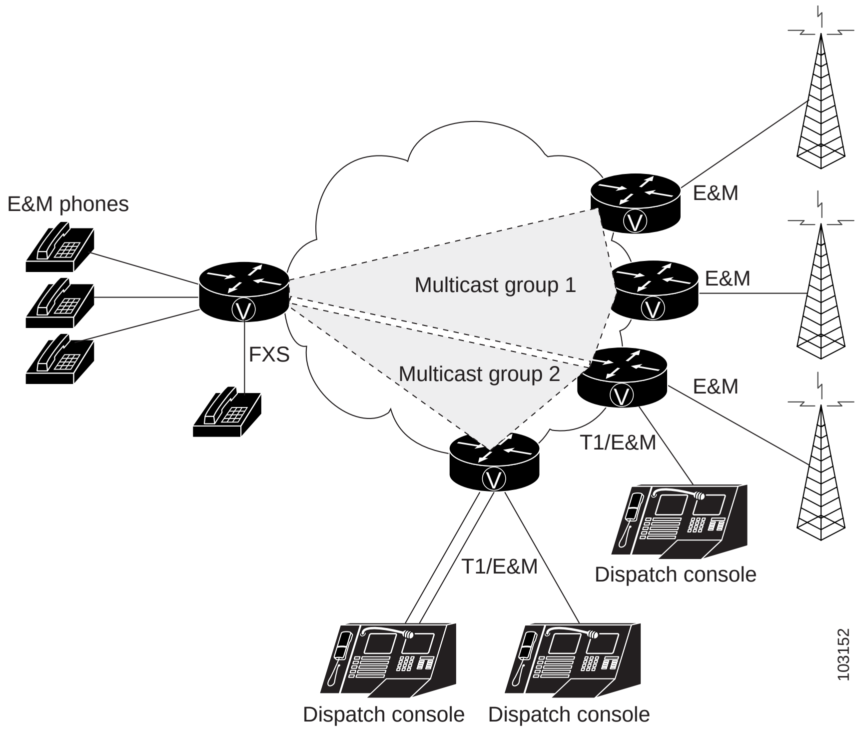
VoIP multicast (VoIPmc) networks provide "always on" multiuser conferences without requiring that users dial in to a conference. By using the inherent point-to-multipoint connectivity of IP multicast (IPmc), the routers can take several inbound voice streams and forward the packetized voice over the IP network to all parties within a defined VoIPmc group. In LMR systems, VoIPmc can connect more than two radios and is required if an IP-based dispatcher application is used to mix and manage different radio channels.

Cisco's VoIP technology, which was initially focused on traditional PBX toll-bypass applications, can be used to combine VoIPmc networks with data networks. VoIP's characteristic dynamic sharing of bandwidth is even more compelling with VoIPmc than with a toll-bypass application because in an LMR environment the relatively short, infrequent bursts of voice activity leave ample bandwidth available for data applications during the long periods of inactivity.

Figure 3 shows a diagram of the Cisco VoIPmc solution connecting legacy equipment over an IP network.

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

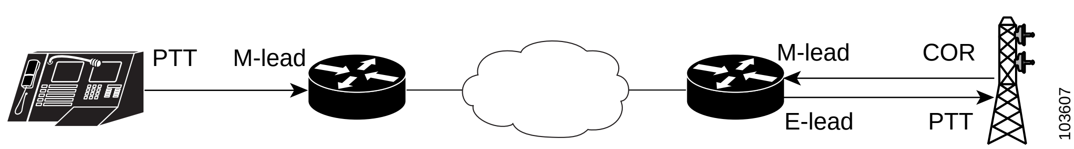
Figure 3 VoIPmc Using Cisco 3725 and Cisco 3745 Routers



Use of E-Lead and M-Lead Signaling

The Land Mobile Radio over IP Enhancement feature allows you to define the use of the E-lead and M-lead in signaling between the E&M voice port on the router and the attached LMR device. The E-lead connects to the Push To Talk (PTT) of the LMR system as shown in Figure 4. The M-lead corresponds to the Carrier Operated Relay (COR) of the LMR system, which indicates receive activity on the LMR system. You can change how the E-lead and M-lead signals are used to suit the needs of your LMR system with the and commands.

Figure 4 E-Lead and M-Lead Connections



The **lmr e-lead** command has the following options:

- Inactive—The router never sends a seize signal on the E-lead to the LMR device. The router sends voice packets to LMR devices. Use this option if you are connecting a tone-controlled radio to the router.
- Seize—The router sends a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection. This is the default. Use this option if your radio requires PTT operation.

- Voice—The router sends a seize signal on the E-lead only when it receives voice packets from the network. When no packets are detected on the network, the seize signal is removed from the E-lead. This option is the same as voice operated transmit (VOX).

The command has the following options:

- Inactive—The router ignores signals sent by voice on the M-lead. The flow of voice packets is determined by VAD. The router sends voice received from the LMR device. This is the default. This option is the same as tone control or VOX.
- Audio-gate-in—The router generates VoIP packets when a seize signal is detected on the M-lead. The router stops generating VoIP packets when the seize signal is removed from the M-lead. An LMR voice port configured for audio-gate-in cannot initiate a PLAR connection.
- Dialin—When the LMR device is not involved in a VoIP connection, the first seize signal detected on the M-lead triggers the router to set up a VoIP connection. This behavior gives the ability for activity on the radio COR to trigger a VoIP call to another VoIP endpoint. Once the connection is made, the router behaves as described in the audio-gate-in option, which is the same as tone control or VOX. The VoIP connection can then remain active indefinitely, or it can time out because of inactivity based on the timer set with the command. Use this option with PLAR connections only.

Polarity

The Land Mobile Radio over IP Enhancement feature allows you to configure the voice port to match E&M bit patterns with the attached LMR device. E&M interfaces use two-state signaling, in which the interface is in either seize or idle state for both transmit and receive. E-lead signal polarity is independent from M-lead signal polarity. In normal polarity, the idle bit pattern is 0000, and the seize bit pattern is 1111. In reverse polarity, the idle bit pattern is 1111, and the seize bit pattern is 0000.

An LMR device with PTT functionality usually looks for an open relay contact, which is normal polarity. However, some LMR devices look for a closed relay pattern, which is reverse polarity.

You can customize the seize and idle patterns with the **define** command. For analog voice ports, bit patterns are not usually customized. Customizing the bit patterns for reverse polarity is a special circumstance reserved for LMR signaling.

In LMR systems that use connection trunk connections, the M-lead signal is sent to the far-end router as a keepalive signal, which the far-end router plays out. If you do not want the M-lead signal played out, define the seize bit pattern to be the same as the idle bit pattern to make sure that only the idle signal will be sent to the far-end router and that the M-lead signal is ignored.

Virtual Interface

In all Cisco VoIPmc 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 VoIPmc 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, using a 30-bit subnet mask. The virtual interface uses two of the subnet addresses. The virtual interface subnet should be included in the routing protocol updates (Routing Information Protocol [RIP], Open Shortest Path First [OSPF], and so on). The virtual interface should not be used as the source address for protocols. Loopback addresses should be used instead.

In this example of the interface vif command, the resulting multicast source address is the address above the interface address. In this example, the VoIPmc source would be 192.168.5.2.

```
interface Vif1
```

```
ip address 192.168.5.1 255.255.2555.252
ip pim sparse-dense-mode
```

E&M Signaling Types

Note This section describes only E&M signaling Type II, Type III, and Type V. Cisco routers do not support Type IV, and Type I is not conducive to LMR.

Type II is preferred for use with LMR because the absence of DC connectivity between radios and the router ensures that no ground loops are created.

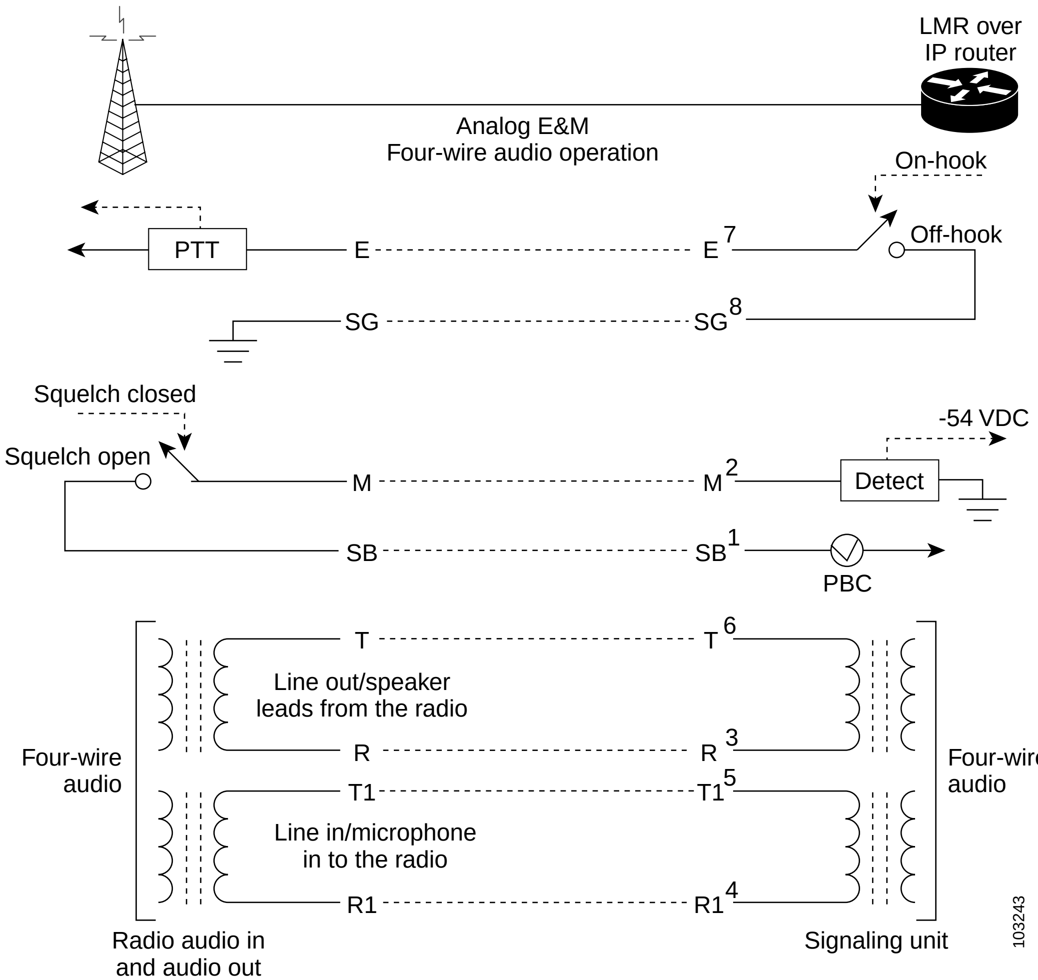
Cisco LMR routers support E&M signaling Type II, Type III, and Type V. With each signaling type, the router supplies one signal, known as the M signal (for Mouth), and accepts one signal, known as the E signal (for Ear). Conversely, the LMR equipment accepts the M signal from the router and provides the E signal to the router. The M signal accepted by the LMR equipment at one end of a circuit becomes the E signal output by the remote LMR interface.

Figures 5 through 7 show the interface models for the different E&M signaling types supported for LMR. Table 5 explains terms used in the figures.

Table 5 E&M Interface Supervision Signal Terms

Term	Description
E (Ear or Earth)	Signal wire from trunking (Central Office (CO)) side to signaling side.
M (Mouth or Magnet)	Signal wire from signaling side to trunking (CO) side.
SG (Signal Ground)	Used on E&M Types II, III, IV (Type IV is not supported on Cisco routers and gateways).
SB (Signal Battery)	Used on E&M Types II, III, IV (Type IV is not supported on Cisco routers and gateways).
Two-Wire Mode	
T1/R1 (Tip-1/Ring-1)	In two-wire operation, the T1/R1 leads carry the full-duplex audio path
Four-Wire Mode	
T/R (Tip/Ring)	In a four-wire operation configuration, this pair of leads carries the audio in from the radio to the router and would typically be connected to the line out or speaker of the radio.
T1/R1 (Tip-1/Ring-1)	In a four-wire operation configuration, this pair of leads carries the audio out from the router to the radio and would normally be connected to the line in or microphone on the radio.

Figure 5 E&M Type II Interface Model



The interface model shown in Figure 5 is correct for a dry relay contact closure for COR functionality. For open collector or open drain outputs, you would have to wire through a user-supplied applique before connecting to the LMR over IP router.

Caution Failure to add an applique may result in damage to radio equipment by the –48 VDC present on the SB lead.

Figure 6 E&M Type III Interface Model

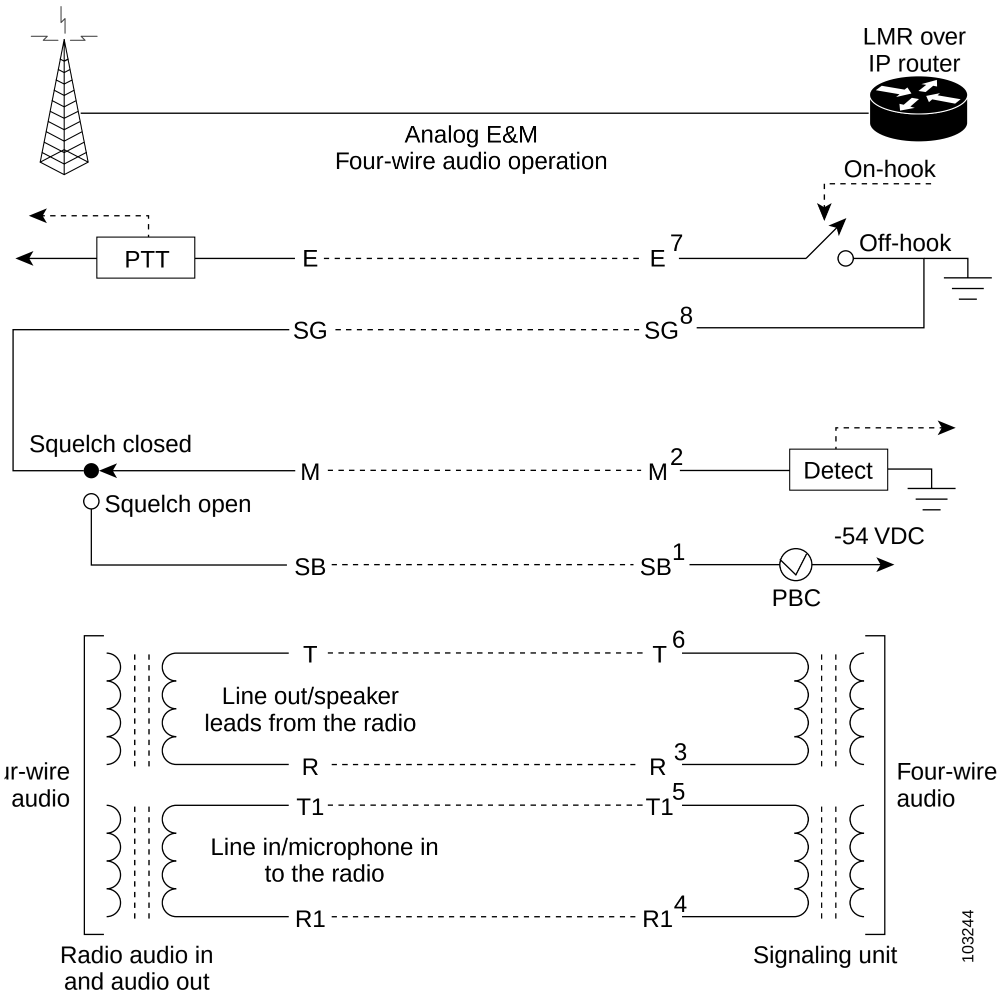
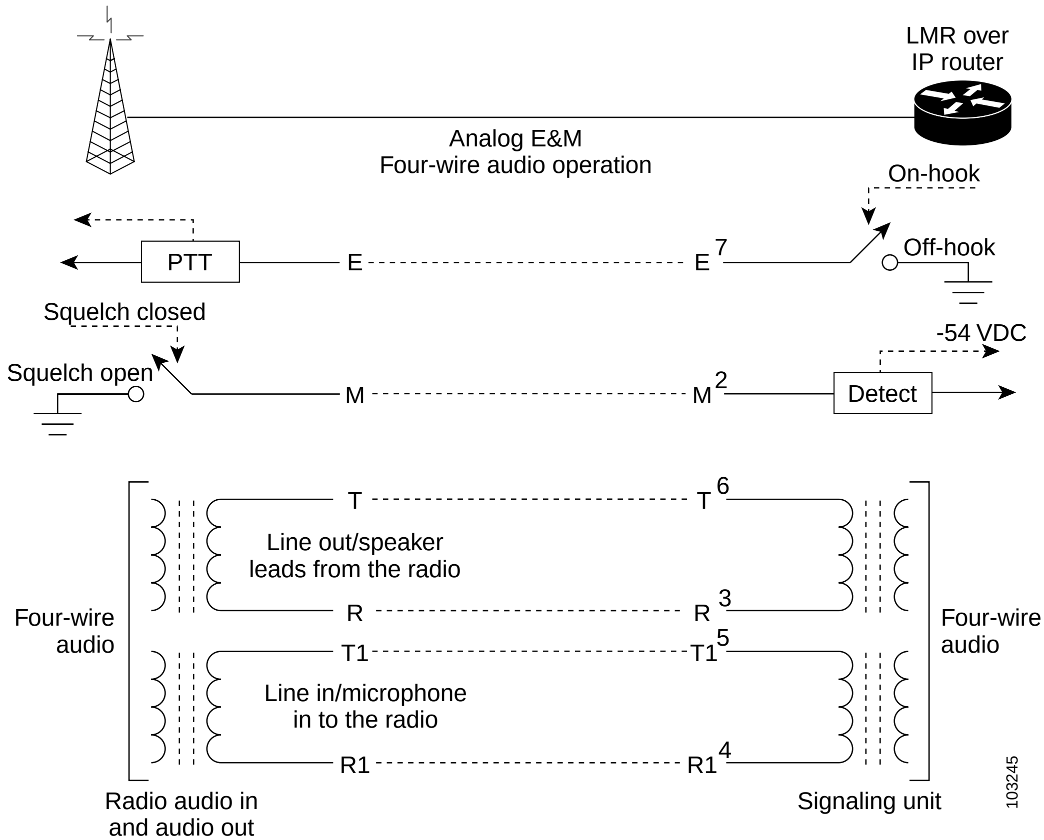


Figure 7 E&M Type V Interface Model



Note Using Type V will create a ground loop between router and radio, which may impact performance.

Codecs

Cisco VoIP gateways use coder-decoders (codecs), which are DSP software algorithms used to compress and decompress speech or audio signals.

Some codec compression techniques require more processing power than others. Codec complexity is broken into two categories, medium and high complexity. The difference between medium and high complexity codecs is the amount of CPU utilization necessary to process the codec algorithm, and therefore, the number of voice channels that can be supported by a single DSP. Medium complexity codecs support four channels per DSP. High complexity codecs support two channels per DSP. For this reason, all the medium complexity codecs can also be run in high complexity mode, but fewer (usually half) of the channels are available per DSP.

Refer to *Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation* for a list of network modules and the codecs they support.

VAD Tuning

Cisco voice activity detection (VAD) has two layers: application programming interface (API) layer and processing layer. There are three states into which the processing layer classifies incoming signals: speech, unknown, and silence. The state of the incoming signals is determined by the noise threshold, which can be configured with the **threshold noise** command.

If the voice level is below the noise threshold, then the signal is classified as silence. If the incoming signal cannot be classified, the variable thresholds that are computed with the statistics of speech and noise that VAD gathers are used to make a determination. If the signal still cannot be classified, then it is marked as unknown. The final decision is made by the API. In some applications, you could have the noise create unwanted spurious packets (for example, a voice stream) taking up bandwidth.

Speech and unknown packets are sent over the network; silence packets are discarded. The sound quality of the connection is slightly degraded with VAD, but the connection monopolizes much less bandwidth. If VAD is disabled, voice data is continuously sent to the IP backbone.

When the **aggressive** keyword is used with the **vad** command in dial peer configuration mode, the VAD noise threshold is reduced from –78 to –62 dBm. Noise that falls below the –62 dBm threshold is considered to be silence and is not sent over the network. Additionally, unknown packets are considered to be silence and are discarded.

The **music-threshold** command specifies the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. The music threshold affects only the operation of VAD when the voice port is receiving voice.

For more information on VAD tuning, refer to *Troubleshooting Hissing and Static*.

How to Configure Land Mobile Radio over IP Enhancement

This section contains the following procedures:

- Configuring an LMR Voice Port, page 15 (required)
- Configuring Polarity and Additional Restrictions on the LMR Voice Port, page 21 (optional)
- Configuring Tone Signaling, page 46 (optional)
- Configuring Connections Between LMR Routers, page 49 (required)
- Adjusting the Voice Quality on the LMR Voice Port, page 69 (optional)
- Verifying Land Mobile Radio over IP Enhancement, page 72 (optional)

Configuring an LMR Voice Port

An LMR voice port is similar to an E&M voice port with Immediate Start signaling and auto-cut-through enabled. Perform one of the following tasks to create an LMR voice port:

- Configuring a Digital LMR Voice Port, page 16 (optional)
- Configuring an Analog LMR Voice Port, page 18 (optional)

LMR Basics

When the LMR system sends voice to the LMR router, Cisco IOS software detects either that the LMR port M-lead is on or that the VAD status has changed. When an LMR voice port receives voice, either the Cisco IOS software turns on the LMR voice port E-lead or a third-party application sends tone on the voice path.

An LMR voice port on a PLAR connection cannot initiate a call unless the **dialin** option of the command was used to configure dial-in capability. LMR PLAR connections are torn down manually with the **test lmr clear-call** command or upon expiration of a teardown timer set with the command.

Duplex Mode

You can configure the LMR voice port to operate in half-duplex or full-duplex mode. Configuring half-duplex mode helps avoid noise being fed back into the network. The duplex mode is on a per port basis. Cisco IOS software does not prevent full-duplex ports from talking to half-duplex ports.

Configuring a Digital LMR Voice Port

Perform this task to configure a digital LMR voice port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **controller t1** *slot/port*
4. **ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type e&m-lmr**
5. **exit**
6. **voice-port** *slot/port***:***ds0-group-number*
7. **shutdown**
8. **lmr duplex half**
9. **lmr led-on**
10. **timing ignore m-lead** *milliseconds*
11. **timing delay-voice tdm** *milliseconds*
12. **timeout ptt** {**rcv** | **xmt**} *minutes*
13. **no comfort-noise**
14. **no echo-cancel enable**
15. **no shutdown**
16. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**controller t1** *slot*/*port* **Example:** `Router(config)# controller t1 1/0`	Enters controller configuration mode and configures a T1 controller.
Step 4	**ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type e&m-lmr** **Example:** `Router(config-controller)# ds0-group 0 timeslots 1-24 type e&m-lmr`	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller, specifies the signaling type by which the router communicates with the PBX or public switched telephone network (PSTN), and defines T1or E1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN.
Step 5	**exit** **Example:** `Router(config-controller)# exit`	Exits controller configuration mode.
Step 6	**voice-port** *slot*/*port*:*ds0-group-number* **Example:** `Router(config)# voice-port 1/0:1`	Enters voice-port configuration mode and specifies a voice card.
Step 7	**shutdown** **Example:** `Router(config-voiceport)# shutdown`	Takes the voice ports for a specific voice interface card offline.
Step 8	**lmr duplex half** **Example:** `Router(config-voiceport)# lmr duplex half`	(Optional) Specifies having the voice path for a voice port operate in half duplex mode.
Step 9	**lmr led-on** **Example:** `Router(config-voiceport)# lmr led-on`	(Optional) Use the ear and mouth (E&M) LED to indicate the E-lead and M-lead status.

	Command or Action	Purpose
Step 10	**timing ignore m-lead** *milliseconds* **Example:** `Router(config-voiceport)# timing ignore m-lead 500`	(Optional) For connection trunk connections, specifies that the router ignore M-lead or VAD changes for a specified amount of time after sending the E-lead off signal, which reduces echo feedback. • This command has an effect only if the voice port is configured for half duplex mode.
Step 11	**timing delay-voice tdm** *milliseconds* **Example:** `Router(config-voiceport)# timing delay-voice tdm 470`	(Optional) Specifies the delay before a voice packet is played out.
Step 12	**timeout ptt** {**rcv** | **xmt**} *minutes* **Example:** `Router(config-voiceport)# timeout ptt xmt 10`	(Optional) Specifies a maximum time for transmitting or receiving a voice packet. • Range is integers from 1 to 30.
Step 13	**no comfort-noise** **Example:** `Router(config-voiceport)# no comfort-noise`	Provides silence when the remote party is not speaking and VAD is enabled at the remote end of the connection.
Step 14	**no echo-cancel enable** **Example:** `Router(config-voiceport)# no echo-cancel enable`	Disables the cancellation of voice that is sent out the interface and received back on the same interface.
Step 15	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Puts the voice ports for a specific voice interface card back in service.
Step 16	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring an Analog LMR Voice Port

Perform this task to configure an analog LMR voice port.

Prerequisites

For analog LMR voice ports, the voice interface card must be an E&M card.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot-number/subunit-number/port*
4. **shutdown**

5. **type** {**1** | **2** | **3** | **5**}
6. **operation** {**2-wire** | **4-wire**}
7. **signal lmr**
8. **lmr duplex half**
9. **lmr led-on**
10. **timing ignore m-lead** *milliseconds*
11. **timing delay-voice tdm** *milliseconds*
12. **timeout ptt** {**rcv** | **xmt**} *minutes*
13. **bootup e-lead off**
14. **no comfort-noise**
15. **no echo-cancel enable**
16. **no shutdown**
17. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot-number`*`/`*`subunit-number`*`/`*`port`* **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`shutdown`** **Example:** `Router(config-voiceport)# shutdown`	Takes the voice ports for a specific voice interface card offline.
Step 5	**`type`** `{`**`1`** `|` **`2`** `|` **`3`** `|` **`5`**`}` **Example:** `Router(config-voiceport)# type 2`	Specifies the E&M interface type. • The default is Type 1.
Step 6	**`operation`** `{`**`2-wire`** `|` **`4-wire`**`}` **Example:** `Router(config-voiceport)# operation 2-wire`	Selects a specific cabling scheme for E&M ports. • The default is 2-wire E&M cabling scheme.

	Command or Action	Purpose
Step 7	**signal lmr** **Example:** `Router(config-voiceport)# signal lmr`	Specifies the type of signaling for a voice port.
Step 8	**lmr duplex half** **Example:** `Router(config-voiceport)# lmr duplex half`	(Optional) Specifies having the voice path for a voice port operate in half duplex mode.
Step 9	**lmr led-on** **Example:** `Router(config-voiceport)# lmr led-on`	(Optional) Use the ear and mouth (E&M) LED to indicate the E-lead and M-lead status.
Step 10	**timing ignore m-lead** *milliseconds* **Example:** `Router(config-voiceport)# timing ignore m-lead 500`	(Optional) For connection trunk connections, specifies that the router ignore M-lead or VAD changes for a specified amount of time after sending the E-lead off signal, which reduces echo feedback. • This command has an effect only if the voice port is configured for half duplex mode.
Step 11	**timing delay-voice tdm** *milliseconds* **Example:** `Router(config-voiceport)# timing delay-voice tdm 470`	(Optional) Specifies the delay after which voice packets are played out.
Step 12	**timeout ptt** {**rcv** | **xmt**} *milliseconds* **Example:** `Router(config-voiceport)# timeout ptt xmt 10`	(Optional) Specifies a maximum time for transmitting or receiving a voice packet. • Range is integers from 1 to 30. • The purpose of this command is to limit extended radio transmission. When the time limit configured with this command expires, the radio transmitter unkeys, so that listeners on the channel cannot hear the speaker, even if the speaker continues to talk. When the speaker unkeys the radio, the timer is reactivated.
Step 13	**bootup e-lead off** **Example:** `Router(config-voiceport)# bootup e-lead off`	(Optional) Prevents an analog E&M voice port from keying the attached radio on router boot up.
Step 14	**no comfort-noise** **Example:** `Router(config-voiceport)# no comfort-noise`	Provides silence when the remote party is not speaking and VAD is enabled at the remote end of the connection.
Step 15	**no echo-cancel enable** **Example:** `Router(config-voiceport)# no echo-cancel enable`	Disables the cancellation of voice that is sent out the interface and received back on the same interface.

	Command or Action	Purpose
Step 16	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Puts the voice ports for a specific voice interface card back in service.
Step 17	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring Polarity and Additional Restrictions on the LMR Voice Port

Perform the following tasks to configure polarity and additional restrictions on the LMR voice port:

- Configuring Polarity and Additional Restrictions on the M-Lead, page 21 (optional)
- Configuring Polarity and Additional Restrictions on the E-Lead, page 33 (optional)

Configuring Polarity and Additional Restrictions on the M-Lead

Note Some of the tasks involved in configuring polarity and additional restrictions on the M-lead require different steps for PLAR connections versus connection trunk connections. Be sure to choose the right set of steps for your connection type.

Choose one of the following optional tasks to configure polarity and additional restrictions on the M-lead of an LMR voice port:

- Configuring the LMR Voice Port to Ignore the M-Lead Signal, Normal Polarity, page 21 (optional)
- Configuring the LMR Voice Port to Ignore the M-Lead Signal, Reverse Polarity, page 24 (optional)
- Configuring the LMR Voice Port to Gate the M-Lead Audio, Normal Polarity, page 26 (optional)
- Configuring the LMR Voice Port to Gate the M-Lead Audio, Reverse Polarity, page 28 (optional)
- Configuring the LMR Voice Port to Trigger a Call on First Activity, Normal Polarity, page 30 (optional)
- Configuring the LMR Voice Port to Trigger a Call on First Activity, Reverse Polarity, page 31 (optional)

Configuring the LMR Voice Port to Ignore the M-Lead Signal, Normal Polarity

Perform this task to configure normal polarity on the M-lead and to configure the router to ignore signals sent by voice on the M-lead. The flow of voice packets is determined by VAD. The router sends voice received from the LMR device.

Note This task applies to PLAR and connection trunk connections. Step 6 uses a different keyword for PLAR and connection trunk. Be sure to choose the correct keyword for your connection type.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead inactive**
5. **define rx-bits idle 0000**
6. **define rx-bits seize 1111**
   or
   define rx-bits seize 0000
7. **exit**
8. **dial-peer voice** *tag* **voip**
9. **vad aggressive**
10. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr m-lead inactive** **Example:** `Router(config-voiceport)# lmr m-lead inactive`	(Optional) Configures the voice port to ignore signals sent by voice on the M-lead. • The flow of voice packets is determined by VAD. The router sends voice received from the LMR device. • This is the default and will not appear in the output of the **show running-config** command.

	Command or Action	Purpose
Step 5	**`define rx-bits idle 0000`** **Example:** `Router(config-voiceport)# define rx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.

Step 6	**`define rx-bits seize 1111`**
	or
	`define rx-bits seize 0000`
	
	Example:
	`Router(config-voiceport)# define rx-bits seize 1111`
	or
	`Router(config-voiceport)# define rx-bits seize 0000`
		Defines the transmit and receive bits for North American E&M, E&M Mercury Exchange Limited Channel-Associated Signaling (MELCAS), and LMR voice signaling.
		• (Optional for PLAR) The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.
		• In connection trunk connections, the M-lead signal might be sent to the far-end router as a keepalive signal, which the far-end router might play out. Define the seize bit pattern to be the same as the idle bit pattern (0000) to make sure that only the idle signal will be sent to the far-end router and that the M-lead signal is ignored.

Step 7	**`exit`** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 8	**`dial-peer voice`** *`tag`* **`voip`** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 9	**`vad aggressive`** **Example:** `Router(config-dialpeer)# vad aggressive`	Enables VAD for the calls using a particular dial peer. • The **aggressive** option reduces the noise threshold from –78 to –62 dBm. • VAD status change messages replace the function of the M-lead signal. **Note** The incoming VoIP dial peer on the terminating gateway must also have the **vad aggressive** command configured.

	Command or Action	Purpose

Step 10	`codec {clear channel | g711alaw | g711ulaw | g723ar53 | g723ar63 | g723r53 | g723r63 | g726r16 | g726r24 | g726r32 | g726r53 | g726r63 | g728 | g729abr8 | g729ar8 | g729br8 | g729r8 | gsmefr | gsmfr} [bytes payload-size]`
	
	Example:
	`Router(config-dialpeer)# codec g711ulaw`
		(Optional) Configures the codec.
		• For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used.
		• You must configure the same codec on all dial peers in a session.

Step 11	`end`
	
	Example:
	`Router(config-dialpeer)# end`
		Exits to privileged EXEC mode.

Configuring the LMR Voice Port to Ignore the M-Lead Signal, Reverse Polarity

Perform this task to configure reverse polarity on the M-lead and to configure the router to ignore signals sent by voice on the M-lead. The flow of voice packets is determined by VAD. The router sends voice received from the LMR device.

Note This task applies to PLAR and connection trunk connections. Step 6 uses a different keyword for PLAR and connection trunk. Be sure to choose the correct keyword for your connection type.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead inactive**
5. **define rx-bits idle 1111**
6. **define rx-bits seize 0000**
   or
   define rx-bits seize 1111
7. **exit**
8. **dial-peer voice** *tag* **voip**
9. **vad aggressive**
10. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr m-lead inactive`** **Example:** `Router(config-voiceport)# lmr m-lead inactive`	(Optional) Configures the voice port to ignore signals sent by voice on the M-lead. • The flow of voice packets is determined by VAD. The router sends voice received from the LMR device. • This is the default and will not appear in the output of the **show running-config** command.
Step 5	**`define rx-bits idle 1111`** **Example:** `Router(config-voiceport)# define rx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.

Step 6	**`define rx-bits seize 0000`**
	or
	`define rx-bits seize 1111`
	
	Example:
	`Router(config-voiceport)# define rx-bits seize 0000`
	or
	`Router(config-voiceport)# define rx-bits seize 1111`
		Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling.
		• For PLAR connections, setting the seize bit pattern to 0000 reverses polarity.
		• (Optional for connection trunk) The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.
		• In connection trunk connections, the M-lead signal might be sent to the far-end router as a keepalive signal, which the far-end router might play out. Define the seize bit pattern to be the same as the idle bit pattern (1111) to make sure that only the idle signal will be sent to the far-end router and that the M-lead signal is ignored.

	Command or Action	Purpose
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 8	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 9	**vad aggressive** **Example:** `Router(config-dialpeer)# vad aggressive`	Enables VAD for the calls using a particular dial peer. • The **aggressive** option reduces the noise threshold from –78 to –62 dBm. **Note** The incoming VoIP dial peer on the terminating gateway must also have the **vad aggressive** command configured.
Step 10	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.
Step 11	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring the LMR Voice Port to Gate the M-Lead Audio, Normal Polarity

Perform this task to configure normal polarity on the M-lead and to configure the router to generate VoIP packets when a seize signal is detected on the M-lead. The router stops generating VoIP packets when the seize signal is removed from the M-lead.

Note This task applies to PLAR and connection trunk connections.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead audio-gate-in**
5. **define rx-bits idle 0000**
6. **define rx-bits seize 1111**

7. **timing hookflash-input** *milliseconds*
8. **exit**
9. **dial-peer voice** *tag* **voip**
10. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr m-lead audio-gate-in`** **Example:** `Router(config-voiceport)# lmr m-lead audio-gate-in`	Configures the voice port to generate VoIP packets when a seize signal is detected on the M-lead. • The router stops generating VoIP packets when the seize signal is removed from the M-lead.
Step 5	**`define rx-bits idle 0000`** **Example:** `Router(config-voiceport)# define rx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.
Step 6	**`define rx-bits seize 1111`** **Example:** `Router(config-voiceport)# define rx-bits seize 1111`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.

	Command or Action	Purpose
Step 7	**timing hookflash-input** *milliseconds* **Example:** `Router(config-voiceport)# timing hookflash-input 0`	Configures the delay between when the M-lead is raised and voice is sent. • Range is 0 to 1550 milliseconds. Default is 480 milliseconds. Setting the value to 0 specifies no delay in the audio input.
Step 8	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 9	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 10	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.
Step 11	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring the LMR Voice Port to Gate the M-Lead Audio, Reverse Polarity

Perform this task to configure reverse polarity on the M-lead and to configure the router to generate VoIP packets when a seize signal is detected on the M-lead. The router stops generating VoIP packets when the seize signal is removed from the M-lead.

Note This task applies to PLAR and connection trunk connections.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead audio-gate-in**
5. **define rx-bits idle 1111**
6. **define rx-bits seize 0000**
7. **timing hookflash-input** *milliseconds*

8. **exit**
9. **dial-peer voice** *tag* **voip**
10. **no vad aggressive**
11. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
12. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr m-lead audio-gate-in`** **Example:** `Router(config-voiceport)# lmr m-lead audio-gate-in`	Configures the voice port to generate VoIP packets when a seize signal is detected on the M-lead. • The router stops generating VoIP packets when the seize signal is removed from the M-lead.
Step 5	**`define rx-bits idle 1111`** **Example:** `Router(config-voiceport)# define rx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.
Step 6	**`define rx-bits seize 0000`** **Example:** `Router(config-voiceport)# define rx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the seize bit pattern to 0000 reverses polarity.
Step 7	**`timing hookflash-input`** *milliseconds* **Example:** `Router(config-voiceport)# timing hookflash-input 0`	Configures the delay between when the M-lead is raised and voice is sent. • Range is 0 to 1550 milliseconds. Default is 480 milliseconds. Setting the value to 0 specifies no delay in the audio input.

	Command or Action	Purpose

Step 8	`exit`
	
	Example:
	`Router(config-voiceport)# exit`
		Exits voice-port configuration mode.

Step 9	**`dial-peer voice`** *`tag`* **`voip`** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 10	**`no vad aggressive`** **Example:** `Router(config-dialpeer)# no vad aggressive`	Disables VAD for the calls using a particular dial peer. • The **aggressive** option reduces the noise threshold from –78 to –62 dBm. **Note** The incoming VoIP dial peer on the terminating gateway must also have the **no vad aggressive** command configured.
Step 11	**`codec`** {**`clear channel`** | **`g711alaw`** | **`g711ulaw`** | **`g723ar53`** | **`g723ar63`** | **`g723r53`** | **`g723r63`** | **`g726r16`** | **`g726r24`** | **`g726r32`** | **`g726r53`** | **`g726r63`** | **`g728`** | **`g729abr8`** | **`g729ar8`** | **`g729br8`** | **`g729r8`** | **`gsmefr`** | **`gsmfr`**} [**`bytes`** *`payload-size`*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.

Step 12	`end`
	
	Example:
	`Router(config-dialpeer)# end`
		Exits to privileged EXEC mode.

Configuring the LMR Voice Port to Trigger a Call on First Activity, Normal Polarity

Perform this task to configure normal polarity on the M-lead and to configure the router to set up a VoIP connection when the LMR device is not involved in a VoIP connection and the first seize signal is detected on the M-lead. Once the connection is made, the router behaves as in the audio-gate-in option.

Note This task applies to PLAR connections only.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead dialin**
5. **define rx-bits idle 0000**
6. **define rx-bits seize 1111**

7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr m-lead dialin** **Example:** `Router(config-voiceport)# lmr m-lead dialin`	Configures the voice port to set up a VoIP connection when the LMR device is not involved in a VoIP connection and the first seize signal is detected on the M-lead. • Once the connection is made, the router behaves as in the audio-gate-in option.
Step 5	**define rx-bits idle 0000** **Example:** `Router(config-voiceport)# define rx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.
Step 6	**define rx-bits seize 1111** **Example:** `Router(config-voiceport)# define rx-bits seize 1111`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the LMR Voice Port to Trigger a Call on First Activity, Reverse Polarity

Perform this task to configure reverse polarity on the M-lead and to configure the router to set up a VoIP connection when the LMR device is not involved in a VoIP connection and the first seize signal is detected on the M-lead. Once the connection is made, the router behaves as in the audio-gate-in option.

Note This task applies to PLAR connections only.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr m-lead dialin**
5. **define rx-bits idle 1111**
6. **define rx-bits seize 0000**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*`slot-number`*/*`subunit-number`*/*`port`* | *`slot`*/*`port`*:*`ds0-group-number`*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr m-lead dialin`** **Example:** `Router(config-voiceport)# lmr m-lead dialin`	Configures the voice port to set up a VoIP connection when the LMR device is not involved in a VoIP connection and the first seize signal is detected on the M-lead. • Once the connection is made, the router behaves as in the audio-gate-in option.
Step 5	**`define rx-bits idle 1111`** **Example:** `Router(config-voiceport)# define rx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.

	Command or Action	Purpose
Step 6	**define rx-bits seize 0000** **Example:** `Router(config-voiceport)# define rx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the seize bit pattern to 0000 reverses polarity.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring Polarity and Additional Restrictions on the E-Lead

Note Some of the tasks involved in configuring polarity and additional restrictions on the M-lead require different steps for PLAR connections versus connection trunk connections. Be sure to choose the right task for your type of connection.

Choose one of the following optional tasks to configure polarity and additional restrictions on the E-lead of an LMR voice port for PLAR connections:

- Configuring the E-Lead to Be Always Inactive, Normal Polarity, page 33
- Configuring the E-Lead to Be Always Inactive, Reverse Polarity, page 37
- Configuring the E-Lead for Active Call, Normal Polarity, page 40
- Configuring the E-Lead for Active Call, Reverse Polarity, page 42
- Configuring the E-Lead for Voice Packet, Normal Polarity, page 43
- Configuring the E-Lead for Voice Packet, Reverse Polarity, page 45

Configuring the E-Lead to Be Always Inactive, Normal Polarity

Perform this task to configure normal polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device. The router sends voice packets to LMR devices.

Note This task has different steps for PLAR connections and connection trunk connections. Be sure to choose the correct set of steps for your connection type.

Configuring the E-Lead to Be Always Inactive, Normal Polarity, PLAR

Perform this task to configure normal polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device for PLAR connections. The router sends voice packets to LMR devices.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number/subunit-number/port* | *slot/port***:***ds0-group-number*}

4. **lmr e-lead inactive**
5. **define rx-bits idle 0000**
6. **define rx-bits seize 1111**
7. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**voice-port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.

Step 4	`lmr e-lead inactive`
	
	Example:
	`Router(config-voiceport)# lmr e-lead inactive`
		(Optional) Configures the router to never send a seize signal on the E-lead to the LMR device.
		• The router sends voice packets to LMR devices.
		• This is the default and will not appear in the output of the **show running-config** command.

Step 5	`define rx-bits idle 0000`
	
	Example:
	`Router(config-voiceport)# define rx-bits idle 0000`
		(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling.
		• The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.

Step 6	`define rx-bits seize 1111`
	
	Example:
	`Router(config-voiceport)# define rx-bits seize 1111`
		(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling.
		• The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.

Step 7	`end`
	
	Example:
	`Router(config-voiceport)# end`
		Exits to privileged EXEC mode.

Configuring the E-Lead to Be Always Inactive, Normal Polarity, Connection Trunk

Perform this task to configure normal polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device for connection trunk connections. The router sends voice packets to LMR devices.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port***:***ds0-group-number*}
4. **lmr e-lead inactive**
5. **define tx-bits idle 0000**
6. **define tx-bits seize 0000**
7. **exit**
8. **dial-peer voice** *tag* **voip**
9. **vad aggressive**
10. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr e-lead inactive** **Example:** `Router(config-voiceport)# lmr e-lead inactive`	Configures the router to never send a seize signal on the E-lead to the LMR device. • The router sends voice packets to LMR devices.

	Command or Action	Purpose
Step 5	**define tx-bits idle 0000** **Example:** `Router(config-voiceport)# define rx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.
Step 6	**define tx-bits seize 0000** **Example:** `Router(config-voiceport)# define rx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • In connection trunk connections, the E-lead signal might be sent to the far-end router as a keepalive signal, which the far-end router might play out. To make sure the E-lead signal is ignored, define the seize bit pattern to be the same as the idle bit pattern to make sure that only the idle signal will be sent to the far-end router.
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 8	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 9	**vad aggressive** **Example:** `Router(config-dialpeer)# vad aggressive`	Enables VAD for the calls using a particular dial peer. • The **aggressive** option reduces the noise threshold from –78 to –62 dBm. **Note** The incoming VoIP dial peer on the terminating gateway must also have the **vad aggressive** command configured.
Step 10	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.
Step 11	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead to Be Always Inactive, Reverse Polarity

Perform this task to configure reverse polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device. The router sends voice packets to LMR devices.

Note This task has different steps for PLAR connections and connection trunk connections. Be sure to choose the correct set of steps for your connection type.

Configuring the E-Lead to Be Always Inactive, Reverse Polarity, PLAR

Perform this task to configure reverse polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device for PLAR connections. The router sends voice packets to LMR devices.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr e-lead inactive**
5. **define tx-bits idle 1111**
6. **define tx-bits seize 0000**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr e-lead inactive** **Example:** `Router(config-voiceport)# lmr e-lead inactive`	(Optional) Configures the router to never send a seize signal on the E-lead to the LMR device. • The router sends voice packets to LMR devices. • This is the default and will not appear in the output of the **show running-config** command.

	Command or Action	Purpose
Step 5	**define tx-bits idle 1111** **Example:** `Router(config-voiceport)# define tx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.
Step 6	**define tx-bits seize 0000** **Example:** `Router(config-voiceport)# define tx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the seize bit pattern to 0000 reverses polarity.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead to Be Always Inactive, Reverse Polarity, Connection Trunk

Perform this task to configure reverse polarity on the E-lead and to configure the router to never send a seize signal on the E-lead to the LMR device for connection trunk connections. The router sends voice packets to LMR devices.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr e-lead inactive**
5. **define tx-bits idle 1111**
6. **define tx-bits seize 1111**
7. **exit**
8. **dial-peer voice** *tag* **voip**
9. **vad aggressive**
10. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr e-lead inactive** **Example:** `Router(config-voiceport)# lmr e-lead inactive`	Configures the router to never send a seize signal on the E-lead to the LMR device. • The router sends voice packets to LMR devices.
Step 5	**define tx-bits idle 1111** **Example:** `Router(config-voiceport)# define tx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.
Step 6	**define tx-bits seize 1111** **Example:** `Router(config-voiceport)# define tx-bits seize 1111`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command. • In connection trunk connections, the E-lead signal might be sent to the far-end router as a keepalive signal, which the far-end router might play out. To make sure the E-lead signal is ignored, define the seize bit pattern to be the same as the idle bit pattern to make sure that only the idle signal will be sent to the far-end router.
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.

	Command or Action	Purpose
Step 8	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 9	**vad aggressive** **Example:** `Router(config-dialpeer)# vad aggressive`	Enables VAD for the calls using a particular dial peer. • The **aggressive** option reduces the noise threshold from –78 to –62 dBm. **Note** The incoming VoIP dial peer on the terminating gateway must also have the **vad aggressive** command configured.
Step 10	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.
Step 11	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead for Active Call, Normal Polarity

Perform this task to configure normal polarity on the E-lead and to configure the router to send a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection.

Note This task is optional for PLAR and connection trunk connections. E-lead active call and normal polarity is the default.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr e-lead seize**
5. **define rx-bits idle 0000**
6. **define rx-bits seize 1111**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr e-lead seize`** **Example:** `Router(config-voiceport)# lmr e-lead seize`	(Optional) For PLAR and multicast connections, configures the router to send a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection. • For connection trunk connections, the router does not send a seize signal when the LMR port is connected. Instead, if the trunk connection is up, the M-lead signal from the far-end router is passed through as the E-lead on the near-end router. When the M-lead is dropped on the far-end router and the trunk connection is still up, the E-lead is dropped on the near-end router. • This is the default and will not appear in the output of the **show running-config** command.
Step 5	**`define rx-bits idle 0000`** **Example:** `Router(config-voiceport)# define rx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.

	Command or Action	Purpose
Step 6	**define rx-bits seize 1111** **Example:** `Router(config-voiceport)# define rx-bits seize 1111`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead for Active Call, Reverse Polarity

Perform this task to configure reverse polarity on the E-lead and to configure the router to send a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection.

Note This task applies to PLAR and connection trunk connections.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port***:***ds0-group-number*}
4. **lmr e-lead seize**
5. **define tx-bits idle 1111**
6. **define tx-bits seize 0000**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr e-lead seize** **Example:** `Router(config-voiceport)# lmr e-lead seize`	(Optional) For PLAR and multicast connections, configures the router to send a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection. • For connection trunk connections, the router does not send a seize signal when the LMR port is connected. Instead, if the trunk connection is up, the M-lead signal from the far-end router is passed through as the E-lead on the near-end router. When the M-lead is dropped on the far-end router and the trunk connection is still up, the E-lead is dropped on the near-end router. • This is the default and will not appear in the output of the **show running-config** command.
Step 5	**define tx-bits idle 1111** **Example:** `Router(config-voiceport)# define tx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.
Step 6	**define tx-bits seize 0000** **Example:** `Router(config-voiceport)# define tx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the seize bit pattern to 0000 reverses polarity.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead for Voice Packet, Normal Polarity

Perform this task to configure normal polarity on the E-lead and to configure the router to send a seize signal on the E-lead only when it receives voice packets from the network. When no packets are detected on the network, the seize signal is removed from the E-lead.

Note This task applies to PLAR and connection trunk connections.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr e-lead voice**
5. **define tx-bits idle 0000**
6. **define tx-bits seize 1111**
7. **timing hangover** *milliseconds*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** {*`slot-number`*/*`subunit-number`*/*`port`* | *`slot`*/*`port`*:*`ds0-group-number`*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**`lmr e-lead voice`** **Example:** `Router(config-voiceport)# lmr e-lead voice`	Configures the router to send a seize signal on the E-lead only when it receives voice packets from the network. • When no packets are detected on the network, the seize signal is removed from the E-lead.
Step 5	**`define tx-bits idle 0000`** **Example:** `Router(config-voiceport)# define tx-bits idle 0000`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The idle bit pattern 0000 is the default and will not appear in the output of the **show running-config** command.
Step 6	**`define tx-bits seize 1111`** **Example:** `Router(config-voiceport)# define tx-bits seize 1111`	(Optional) Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • The seize bit pattern 1111 is the default and will not appear in the output of the **show running-config** command.

	Command or Action	Purpose
Step 7	**timing hangover** *milliseconds* **Example:** `Router(config-voiceport)# timing hangover 300`	Specifies the number of milliseconds for which the E-lead will stay active after voice detection determines that the voice stream has stopped. • Valid values are 0 to 10000. The default it 250. • Use this command to adjust the timing if the E-lead is being turned on and off too frequently.
Step 8	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the E-Lead for Voice Packet, Reverse Polarity

Perform this task to configure reverse polarity on the E-lead and to configure the router to send a seize signal on the E-lead only when it receives voice packets from the network. When no packets are detected on the network, the seize signal is removed from the E-lead.

Note This task applies to PLAR and connection trunk connections.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **lmr e-lead voice**
5. **define tx-bits idle 1111**
6. **define tx-bits seize 0000**
7. **timing hangover** *milliseconds*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**lmr e-lead voice** **Example:** `Router(config-voiceport)# lmr e-lead voice`	Configures the router to send a seize signal on the E-lead only when it receives voice packets from the network. • When no packets are detected on the network, the seize signal is removed from the E-lead.
Step 5	**define tx-bits idle 1111** **Example:** `Router(config-voiceport)# define tx-bits idle 1111`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the idle bit pattern to 1111 reverses polarity.
Step 6	**define tx-bits seize 0000** **Example:** `Router(config-voiceport)# define tx-bits seize 0000`	Defines the transmit and receive bits for North American E&M, E&M MELCAS, and LMR voice signaling. • Setting the seize bit pattern to 0000 reverses polarity.
Step 7	**timing hangover** *milliseconds* **Example:** `Router(config-voiceport)# timing hangover 300`	Specifies the number of milliseconds for which the E-lead will stay active after voice detection determines that the voice stream has stopped. • Valid values are from 0 to 10000. The default it 250. • Use this command to adjust the timing if the E-lead is being turned on and off too frequently.
Step 8	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Troubleshooting Tips

If the E-lead is configured for voice packet and the far-end dial peer uses VAD, the E-lead will turn on and off too frequently causing clipping. Disable VAD on the far-end dial peer to reduce clipping.

Configuring Tone Signaling

To configure a wakeup tone, frequency selection tone, or guard tone to be played out before or with a voice packet, you need to:

- Create a tone-signal voice class.
- Configure the desired tones and pauses.
- Assign the voice class to the LMR voice port.

Note To avoid voice loss at the receiving end of an LMR system, use the command to configure a delay for the voice packet equal to the sum of the durations of all the injected tones and pauses configured with the command and the **inject pause** command in this task.

Perform this task to configure tone signaling.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class tone-signal** *tag*
4. **inject tone** *index frequency amplitude duration*
5. **inject pause** *index milliseconds*
6. **inject guard-tone** *frequency amplitude* [**idle**]
7. **exit**
8. **voice-port** {*slot-number/subunit-number/port* | *slot/port***:***ds0-group-number*}
9. **voice-class tone-signal** *tag*
10. **exit**
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class tone-signal`** *`tag`* **Example:** `Router(config)# voice class tone-signal mytones`	Enters voice-class configuration mode and creates a tone-signal voice class. • Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice-class tone-signal**, which is used in voice-port configuration mode.

	Command or Action	Purpose
Step 4	**inject tone** *index frequency amplitude duration* **Example:** `Router(config-class)# inject tone 1 2175 -10 120`	Specifies a wakeup or frequency selection tone to be played out before the voice packet. • *index*—Use the this argument in conjunction with the *index* argument of the **inject pause** command to specify the order of the tones and pauses. Range is integers from 1 to 10. • *frequency*—In Hz. Range is integers from 1 to 4000. • *amplitude*—In dBm. Range is integers from –30 to 3. • *duration*—In milliseconds. Range is integers from 10 to 500.
Step 5	**inject pause** *index milliseconds* **Example:** `Router(config-class)# inject pause 2 100`	Specifies a pause between injected tones. • *index*—Use the this argument in conjunction with the *index* argument of the **inject tone** command to specify the order of the tones and pauses. Range is integers from 1 to 10. • *duration*—In milliseconds. Range is integers from 10 to 500.
Step 6	**inject guard-tone** *frequency amplitude* [**idle**] **Example:** `Router(config-class)# inject guard-tone 2175 -30`	Plays out a guard tone with the voice packet to keep the radio channel up. • *frequency*—In Hz. Range is integers from 1 to 4000. • *amplitude*—In dBm. Range is integers from –50 to –3. • Use the **idle** keyword to play out the guard tone when there are no voice packets.
Step 7	**exit** **Example:** `Router(config-class)# exit`	Exits voice-class configuration mode.
Step 8	**voice-port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 9	**voice-class tone-signal** *tag* **Example:** `Router(config-voiceport)# voice-class tone-signal mytones`	Assigns a previously configured tone-signal voice class to a voice port. • Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice class tone-signal**, which is used in global configuration mode.

	Command or Action	Purpose
Step 10	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 11	**end** **Example:** `Router(config)# end`	Exits to privileged EXEC mode.

Configuring Connections Between LMR Routers

Choose one of the following optional tasks to configure connections between the LMR routers on your IP network. Connection trunk and PLAR are usually used when there are two LMR routers on the network. VoIPmc is used when there are more than two LMR routers on the network.

- Configuring Connection Trunk, page 49 (optional)
- Configuring PLAR, page 53 (optional)
- Configuring VoIPmc, page 54 (optional)

Configuring Connection Trunk

Perform this task to configure connection trunk connections.

Dial Peers

For more information on configuring and troubleshooting dial peers refer to *Understanding Inbound and Outbound Dial Peers Matching on IOS Platforms*.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number/subunit-number/port* | *slot/port*:*ds0-group-number*}
4. **shutdown**
5. **connection trunk** *digits* [**answer-mode**]
6. **no shutdown**
7. **exit**
8. **dial-peer voice** *tag* **pots**
9. **destination-pattern** [+]*string* [**T**]
10. **port** {*slot-number/subunit-number/port* | *slot/port*:*ds0-group-number*}
11. **exit**
12. **dial-peer voice** *tag* **voip**
13. **destination-pattern** [+]*string* [**T**]

14. **session target** {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **e.** | **u.**]*host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed** | **ras**}
15. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
16. **dtmf-relay** [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**] [**sip-notify**]
17. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**voice-port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.

Step 4	`shutdown`
	
	Example:
	`Router(config-voiceport)# shutdown`
		Takes the voice ports for a specific voice interface card offline.

Step 5	**connection trunk** *digits* [**answer-mode**] **Example:** `Router(config-voiceport)# connection trunk 123456`	Specifies a connection mode for a voice port. • When Cisco-trunk permanent calls are configured, one side must be the call initiator (master) and the other side must be the call answerer (slave). By default, the voice port operates in master mode. Enter the **answer-mode** keyword to specify that the voice port should operate in slave mode.

Step 6	`no shutdown`
	
	Example:
	`Router(config-voiceport)# no shutdown`
		Puts the voice ports for a specific voice interface card back in service.

Step 7	`exit`
	
	Example:
	`Router(config-voiceport)# exit`
		Exits voice-port configuration mode.

	Command or Action	Purpose
Step 8	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 100 pots`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 9	**destination-pattern** [+]*string* [**T**] **Example:** `Router(config-dialpeer)# destination-pattern 654321`	Defines the destination telephone number associated with this VoIP dial peer.
Step 10	**port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config-dialpeer)# port 1/0/0`	Associates a dial peer with a specific voice port.
Step 11	**exit** **Example:** `Router(config-dialpeer)# exit`	Exits dial peer configuration mode.
Step 12	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 101 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode. • The *tag* argument uniquely identifies the dial peer.
Step 13	**destination-pattern** [+]*string* [**T**] **Example:** `Router(config-dialpeer)# destination-pattern 654321`	Defines the destination telephone number associated with this VoIP dial peer.

	Command or Action	Purpose

Step 14	**session target** {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **e.** | **u.**]*host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed** | **ras**}
	
	Example:
	`Router(config-dialpeer)# session-target ipv4:239.192.7.8:19888`
		Identifies the IP address of the appropriate port on the destination end router.
		• **ipv4:***destination-address*—IP address of the dial peer.
		• **dns:***host-name*—Valid entries are characters representing the name of the host device.
		  – **s.**—Source destination pattern is part of the domain name.
		  – **d.**—Destination number is part of the domain name.
		  – **e.**—Called number digits are reversed; periods are added between each digit of the called number. The string is part of the domain name.
		  – **u.**—Unmatched portion of the destination pattern (such as a defined extension number) is part of the domain name.
		• **loopback:rtp**—Specifies that all voice data is looped back to the originating source. Applicable for VoIP peers.
		• **loopback:compressed**—Specifies that all voice data is looped back in compressed mode to the originating source. Applicable for plain old telephone service (POTS) peers.
		• **loopback:uncompressed**—Specifies that all voice data is looped back in an uncompressed mode to the originating source. Applicable for POTS peers.
		• **ras**—Indicates that the Registration, Admission, and Status (RAS) Protocol signaling function protocol is used. A gatekeeper will translate the E.164 address into an IP address.

Step 15	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.

	Command or Action	Purpose
Step 16	**dtmf-relay** [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**] [**sip-notify**] **Example:** `Router(config-dialpeer)# dtmf-relay h245-alphanumeric`	Specifies how an H.323 or Session Initiation Protocol (SIP) gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network. • By default, DTMF tones are disabled and sent in-band. That is, they are left in the audio stream. • Use the **h245-alphanumeric** option for non multicast networks. If you configure multicasting, the DTMF relay method defaults to the **cisco-rtp** option.
Step 17	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring PLAR

Perform this task to configure PLAR connections. PLAR connections are activated based on the M-lead, so PLAR can be used only with LMR systems that can raise the M-lead. PLAR connections can also be initiated by the FXS port to dial in to the radio connected to an E&M voice port for point-to-point deployments.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
4. **connection plar** *digits*
5. **no shutdown**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 4	**connection plar** *digits* **Example:** `Router(config-voiceport)# connection plar 123456`	Specifies a connection mode for a voice port.
Step 5	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Puts the voice ports for a specific voice interface card back in service.
Step 6	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring VoIPmc

Perform the tasks in the following sections to configure VoIPmc connections. VoIPmc can be used with connection trunk or PLAR connections. Connection trunk is recommended for E&M voice ports. Connection trunk has a retry mechanism, whereas PLAR does not attempt to retry in case of failure. All VoIPmc configurations require multicast routing and a virtual interface (vif) configured on the router.

- Configuring Multicast Routing (VoIPmc), page 54 (required)
- Configuring the Virtual Interface (VoIPmc), page 55 (required)
- Configuring VoIP Dial Peers, VoIPmc, page 56 (required)
- Configuring RTP Payload Type, page 59
- Configuring E&M Voice Ports, VoIPmc, page 60 (required)
- Configuring the Relevant Interface (VoIPmc), page 64 (required)
- Configuring Voice Ports in High-Density Voice Network Modules, VoIPmc, page 65 (required, if using T1/E1)

Configuring Multicast Routing (VoIPmc)

Perform this task to enable multicast routing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip multicast-routing**
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ip multicast-routing** **Example:** `Router(config)# ip multicast-routing`	Enables multicast routing.
Step 4	**end** **Example:** `Router(config)# end`	Exits to privileged EXEC mode.

Configuring the Virtual Interface (VoIPmc)

Perform this task to configure the virtual interface for multicast fast switching.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** *type number* [*name-tag*]
4. **ip address** *ip-address mask*
5. **ip pim sparse-dense-mode**
6. **end**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command	Purpose
Step 3	**interface** *type number* [*name-tag*] **Example:** `Router(config)# interface Vif 1`	Defines a virtual interface for multicast fast switching and enters interface configuration mode. • Routers joining the same session must have their virtual interfaces on different subnets. Otherwise, packets are not switched to the IP network.
Step 4	**ip address** *ip-address mask* **Example:** `Router(config-if)# ip address 10.2.92.1 255.255.255.0`	Assigns the IP address and subnet mask for the virtual interface.
Step 5	**ip pim sparse-dense-mode** **Example:** `Router(config-if)# ip pim 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 6	**end** **Example:** `Router(config-if)# end`	Exits to privileged EXEC mode.

Configuring VoIP Dial Peers, VoIPmc

Perform this task to configure the VoIP dial peers on the router.

Dial Peers

For more information on configuring and troubleshooting dial peers refer to *Understanding Inbound and Outbound Dial Peers Matching on IOS Platforms*.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **destination-pattern** [+]*string*[***T***]
5. **session protocol multicast**
6. **session target ipv4:***destination-address*
7. **ip precedence** *number*
8. **codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
9. **end**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 101 voip`	Assigns a variable number (*tag*) to the VoIP dial peer and enters dial peer voice configuration mode.
Step 4	**destination-pattern** [+]*string*[***T***] **Example:** `Router(config-dialpeer)# destination-pattern 54321`	Specifies the E.164 address associated with this dial peer. • The destination pattern for the VoIP dial peer must match the value of the *multicast-session-number* string for the corresponding voice port. • The keywords and arguments are as follows: – **+**—(Optional) Specifies a character indicating an E.164 standard number. The plus sign (+) is not supported on the Cisco MC3810. – *string*—Indicates a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: The asterisk (*) and pound sign (#)—Indicate the keys that appear on standard touch-tone dial pads. Comma (,)—Inserts a pause between digits. Period (.)—Matches any entered digit (this character is used as a wildcard). Percent sign (%)—Indicates that the previous digit occurred zero or multiple times, similar to the wildcard usage in the regular expression.

	Command	Purpose
		Plus sign (+)—Matches a sequence of one or more matches of the digit. **Note** The plus sign used as part of the digit string is different from the plus sign that can be used in front of the digit string to indicate that the string is an E.164 standard number. Circumflex (^)—Indicates a match to the beginning of the string. Dollar sign ($)—Matches the null string at the end of the input string. Backslash symbol (\)—Is followed by a single character matching that character or used with a single character having no other significance (matching that character). Question mark (?)—Indicates that the previous digit occurred zero or one time. Brackets ([])—Indicates a range of digits. A range is a sequence of characters enclosed in the brackets, and only numeric characters from 0 to 9 are allowed in the range. This is similar to a regular expression rule. Parentheses (())—Indicates a pattern and is the same as the regular expression rule—for example, 408(555). Parentheses are used with symbols ?, %, or +. – **T**—(Optional) Control character indicating that the **destination-pattern** value is a variable length dial string.

Step 5	`session protocol multicast`
	
	Example:
	`Router(config-dialpeer)# session protocol multicast`
		Sets the session protocol as multicast, which allows more than two ports to join the same session simultaneously.
		Note This step is mandatory for voice multicasting and is the command introduced specifically for the Cisco VoIPmc application.

	Command	Purpose
Step 6	**session target ipv4:***destination-address* **Example:** `Router(config-dialpeer)# session target ipv4:239.192.7.8:19888`	Assigns the session target for voice multicasting dial peers. • The session target is a multicast address in the range from 224.0.1.0 to 239.255.255.255 and must be the same for all ports in a session. • The audio Real-Time Transport Protocol (RTP) port is an even number in the range from 16384 to 32767 and must also be the same for all ports in a session. An odd-numbered port (User Datagram Protocol (UDP) port number + 1) is used for the RTP Control Protocol (RTCP) traffic for that session.
Step 7	**ip precedence** *number* **Example:** `Router(config-dialpeer)# ip precedence 5`	(Optional) Specifies the IP precedence.
Step 8	**codec** {**clear channel** | **g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711ulaw`	(Optional) Configures the codec. • For VoIP, the default is **g729r8**, 20-byte payload, which does not appear in the configuration when the **show running-config** command is used. • You must configure the same codec on all dial peers in a session.
Step 9	**end** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring RTP Payload Type

Perform this task to configure RTP payload type on the router.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **rtp payload-type lmr-tone** *number*
5. **rtp payload-type nte-tone** *number*
6. **end**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`voip`** **Example:** `Router(config)# dial-peer voice 1000 voip`	Assigns a variable number (*tag*) to the VoIP dial peer and enters dial peer voice configuration mode.
Step 4	**`rtp payload-type lmr-tone`** *`number`* **Example:** `Router(config-dialpeer)# rtp payload-type lmr-tone 96-127`	Configures payload type for the packet carrier's LMR tone from DSP to playout.
Step 5	**`rtp payload-type nte-tone`** *`number`* **Example:** `Router(config-dialpeer)# rtp payload-type nte-tone 96-127`	Configures a payload type for the packet carrier's RFC2833 tone from DSP to playout
Step 6	**`end`** **Example:** `Router(config-dialpeer)# end`	Exits to privileged EXEC mode.

Configuring E&M Voice Ports, VoIPmc

Perform this task to configure E&M voice ports.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class permanent** *tag*
4. **signal timing oos timeout** [*seconds* | **disabled**]
5. **signal keepalive** {*seconds* | **disabled**}
6. **exit**
7. **voice-port** {*slot-number/subunit-number/port* | *slot/port***:***ds0-group-number*}
8. **voice-class permanent** *tag*
9. **connection trunk** *digits*

10. **music-threshold** *decibels*
11. **operation 4-wire**
12. **type** {**1** | **2** | **3** | **5**}
13. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
14. **voice-class permanent** *tag*
15. **connection trunk** *digits*
16. **music-threshold** *decibels*
17. **operation 4-wire**
18. **end**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice class permanent** *tag* **Example:** `Router(config)# voice class permanent 1`	Defines voice class for transmit-receive mode and enters voice class configuration mode.
Step 4	**signal timing oos timeout** [*seconds* | **disabled**] **Example:** `Router(config-class)# signal timing oos timeout disabled`	Disables signaling loss detection. • Use the **disabled** keyword in VoIPmc applications. The *seconds* argument is not used in these applications.
Step 5	**signal keepalive** {*seconds* | **disabled**} **Example:** `Router(config-class)# signal keepalive 65535`	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks. • The *seconds* argument specifies the keepalive signaling packet interval, in seconds. The valid range is from 1 to 65535. • We recommend that you use the **disabled** keyword when configuring this command for use in networks that use connection trunk connections and multicasting to avoid sending keepalive signals to a multicasting network with no specified destination.

	Command	Purpose
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.
Step 7	**voice-port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-number*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 8	**voice-class permanent** *tag* **Example:** `Router(config-voiceport)# voice-class permanent 1`	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port (for the port that is allowed to speak).
Step 9	**connection trunk** *digits* **Example:** `Router(config-voiceport)# connection trunk 123456`	Ties the voice port to a multicast-session number. • Use the **trunk** keyword to specify a connection that emulates a permanent trunk connection to a PBX. • The *digits* argument specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
Step 10	**music-threshold** *decibels* **Example:** `Router(config-voiceport)# music-threshold -70`	(Optional) Sets the music threshold to make VAD less sensitive. • The *decibels* argument is the on-hold music threshold in decibels (dB). Valid entries are any integer from –70 to –30.
Step 11	**operation 4-wire** **Example:** `Router(config-voiceport)# operation 4-wire`	Specifies the cabling scheme for E&M ports. • Use **4-wire** operation for the VoIPmc connections.

	Command	Purpose
Step 12	**type** {**1** | **2** | **3** | **5**} **Example:** `Router(config-voiceport)# type 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
Step 13	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(config)# voice-port 2/0/0`	Selects another voice card and enters voice-port configuration mode.
Step 14	**voice-class permanent** *tag* **Example:** `Router(config-voiceport)# voice-class permanent 1`	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port (for the port that is allowed to speak).
Step 15	**connection trunk** *digits* **Example:** `Router(config-voiceport)# connection trunk 123456`	Ties the voice port to the same multicast-session number as in Step 9.
Step 16	**music-threshold** *decibels* **Example:** `Router(config-voiceport)# music-threshold -70`	(Optional) Sets the music threshold to make VAD less sensitive.

	Command	Purpose
Step 17	**`operation 4-wire`** **Example:** `Router(config-voiceport)# operation 4-wire`	Specifies the calling scheme for E&M ports. • Specify **4-wire** operation for the VoIPmc application.
Step 18	**`end`** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Configuring the Relevant Interface (VoIPmc)

Perform this task to configure either the serial or the Ethernet interface.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface** *type slot/port*
4. **ip address** *ip-address mask* [**secondary**]
5. **ip pim** {**sparse-mode** | **dense-mode** | **sparse-dense-mode**}
6. **no shutdown**
7. **end**

DETAILED STEPS

.

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command	Purpose
Step 3	**interface** *type slot*/*port* **Example:** `Router(config)# interface Ethernet 0/0`	Configures the physical interface for transmitting multicast packets and enters interface configuration mode. The arguments are as follows: • *type*—Type of interface to be configured. The *type* may be either of the following: – **ethernet**—Ethernet IEEE 802.3 interface. – **serial**—Serial interface. • *slot*/*port*—Number of the slot and port being configured. Refer to the appropriate hardware manual for slot and port information.
Step 4	**ip address** *ip-address mask* [**secondary**] **Example:** `Router(config-if)# ip address 10.2.82.80 255.255.255.0`	Sets a primary or secondary IP address for an interface. • The arguments and keyword are as follows: – *ip-address*—IP address. – *mask*—Mask for the associated IP subnet. – **secondary**—(Optional) Specifies that the configured address is a secondary IP address. If this keyword is omitted, the configured address is the primary IP address.
Step 5	**ip pim** {**sparse-mode** | **dense-mode** | **sparse-dense-mode**} **Example:** `Router(config-if)# ip pim sparse-dense-mode`	Specifies PIM. • Whatever mode you choose should match all the interfaces in all the routers of your network.
Step 6	**no shutdown** **Example:** `Router(config-if)# no shutdown`	Enables the interface.
Step 7	**end** **Example:** `Router(config-if)# end`	Exits to privileged EXEC mode.

Configuring Voice Ports in High-Density Voice Network Modules, VoIPmc

A multiflex trunk interface card (NM-HDV) in a high-density voice network module requires special voice-port configuration when being connecting for T1/E1 operation. Perform this task to configure a multiflex trunk interface card in a high-density voice network module.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **voice class permanent** *tag*
4. **signal timing oos timeout** [*seconds* | **disabled**]
5. **signal keepalive** {*seconds* | **disabled**}
6. **exit**
7. **controller** {**t1** | **e1**} *slot/port*
8. **ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type e&m-lmr**
9. **exit**
10. **voice-port** *slot/port***:***ds0-group-number*
11. **connection trunk** *digits* [**answer-mode**]
12. **voice-class permanent** *tag*
13. **end**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice class permanent** *tag* **Example:** `Router(config)# voice class permanent 1`	Defines a voice class for receive-only mode and enters voice class configuration mode. • The *tag* argument is a unique number that you assign to the voice class. The tag number must be unique on the router. The valid range for this tag is from 1 to 10000.

	Command	Purpose
Step 4	**signal timing oos timeout** [*seconds* | **disabled**] **Example:** `Router(config-class)# signal timing oos timeout disabled`	Changes the delay time between the loss of signaling packets from the network and the start time for the out-of-service (OOS) state. • The keywords and arguments are as follows: – *seconds*—(Optional) Delay duration, in seconds, between the loss of signaling packets and the beginning of the OOS state. The default is 30 seconds. The range is from 1 to 65535. – **disabled**—(Optional) Deactivates the detection of packet loss. If no signaling packets are received from the network, the router does not send an OOS pattern to the PBX and it continues sending voice packets to the network. Use this option to disable busyout to the PBX.
Step 5	**signal keepalive** {*seconds* | **disabled**} **Example:** `Router(config-class)# signal keepalive 65535`	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks. • The *seconds* argument specifies the keepalive signaling packet interval, in seconds. The valid range is from 1 to 65535. • We recommend that you use the **disabled** keyword when configuring this command for use in networks that use connection trunk connections and multicasting to avoid sending keepalive signals to a multicasting network with no specified destination.
Step 6	**exit** **Example:** `Router(config-class)#` **`exit`**	Exits voice class configuration mode.
Step 7	**controller** {**t1** | **e1**} *slot*/*port* **Example:** `Router(config)# controller t1 6/0`	Selects the T1 or E1 controller and enters controller configuration mode. • The *slot*/*port* argument is the backplane slot number and port number on the interface. Refer to your hardware installation manual for the specific values and slot numbers.

	Command	Purpose
Step 8	**ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type e&m-lmr** **Example:** `Router(config-controller)# ds0-group 0 timeslots 1-24 type e&m-lmr`	Specifies the DS0 time slots that make up a logical voice port on a T1 or E1 controller and specifies the signaling type by which the router communicates with the PBX or PSTN. • The keywords and arguments are as follows: – *ds0-group-number*—A value from 0 to 23 that identifies the DS0 group. – **timeslots** *timeslot-list*—A single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1 or E1, allowable values are from 1 to 24. Examples are as follows: 2 1-15,17-24 1-23 2,4,6-12 – **type**—The signaling method selection for the **type** keyword depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie-lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBX. The FXO interface is for connecting the CO to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Type must be **lmr**.
Step 9	**exit** **Example:** `Router(config-controller)# exit`	Exits controller configuration mode.
Step 10	**voice-port** *slot*/*port*:*ds0-group-number* **Example:** `Router(config)# voice-port 1/0:0`	Enters voice-port configuration mode and configures a DS0 group that was created in Step 8.

	Command	Purpose
Step 11	**connection trunk** *digits* [**answer-mode**] **Example:** `Router(config-voiceport)# connection trunk 654321`	Specifies a connection mode for a voice port. • Ties the connection trunk to a multicast-session number. This command is repeated for each DS0 group. All groups use the same multicast address, if connecting to the same multicast session. • Use the **trunk** keyword to specify a connection that emulates a permanent trunk connection to a PBX. • The *digits* argument specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
Step 12	**voice-class permanent** *tag* **Example:** `Router(config-voiceport)# voice-class permanent 1`	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port (for the port that is allowed to speak).
Step 13	**end** **Example:** `Router(config-voiceport)# end`	Exits to privileged EXEC mode.

Adjusting the Voice Quality on the LMR Voice Port

Perform this task to adjust the voice quality on the LMR voice port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class tone-signal** *tag*
4. **digital-filter** {**1950hz** | **2175hz**}
5. **exit**
6. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
7. **shutdown**
8. **input gain** {*decibels* | **auto-control** [*auto-dbm*]}
9. **output attenuation** {*decibels* | **auto-control** [*auto-dbm*]}
10. **music-threshold** *decibels*
11. **threshold noise** *value*
12. **voice-class tone-signal** *tag*
13. **no shutdown**
14. **exit**

15. **voice vad-time** *milliseconds*
16. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class tone-signal`** *`tag`* **Example:** `Router(config)# voice class tone-signal mytones`	Enters voice-class configuration mode and creates a tone-signal voice class. • Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice-class tone-signal**, which is used in voice-port configuration mode.
Step 4	**`digital-filter`** {**`1950hz`** | **`2175hz`**} **Example:** `Router(config-class)# digital-filter 1950hz`	Specifies the digital filter to be used before the voice packet is sent from the DSP to the network. • Only one of the frequencies, 1950 Hz or 2175 Hz, can be filtered out at a time.
Step 5	**`exit`** **Example:** `Router(config-class)# exit`	Exits voice-class configuration mode.
Step 6	**`voice-port`** {*`slot-number`*/*`subunit-number`*/*`port`* | *`slot`*/*`port`*:*`ds0-group-number`*} **Example:** `Router(config)# voice-port 1/0/0`	Enters voice-port configuration mode and specifies a voice card.
Step 7	**`shutdown`** **Example:** `Router(config-voiceport)# shutdown`	(Optional) Takes the voice ports for a specific voice interface card offline. • If you are using connection trunk connections, you must shut down the voice port and bring it back up for configuration changes to take effect.
Step 8	**`input gain`** {*`decibels`* | **`auto-control`** [*`auto-dbm`*]} **Example:** `Router(config-voiceport)# input gain -1`	Configures a specific input gain value or enables automatic gain control. • A negative value decreases the input gain. • The *auto-dbm* argument is an absolute dBm value.

	Command or Action	Purpose
Step 9	**output attenuation** {*decibels* | **auto-control** [*auto-dbm*]} **Example:** `Router(config-voiceport)# output attenuation -2`	Configures a specific output attenuation value. • Contrary to the effect of values for the **input gain** command, a negative value increases the output attenuation, and a positive value decreases it. • The *auto-dbm* argument is an absolute dBm value.
Step 10	**music-threshold** *decibels* **Example:** `Router(config-voiceport)# music-threshold -70`	Sets the music threshold to make VAD less sensitive. • The *decibels* argument is the on-hold music threshold in decibels (dB). Valid entries are any integer from –70 to –30.
Step 11	**threshold noise** *value* **Example:** `Router(config-voiceport)# threshold noise -70`	Configures a noise threshold for incoming calls. • The *value* argument establishes a noise threshold that is used to categorize incoming signals as speech, unknown, or silence. Valid values are from –30 to –90 dB. The default is –62 dB.
Step 12	**voice-class tone-signal** *tag* **Example:** `Router(config-voiceport)# voice-class tone-signal mytones`	Assigns a previously configured tone-signal voice class to a voice port. • Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice class tone-signal**, which is used in global configuration mode.
Step 13	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	(Optional) Puts the voice ports for a specific voice interface card back in service. • If you are using connection trunk connections, you must shut down the voice port and bring it back up for configuration changes to take effect.
Step 14	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.

	Command or Action	Purpose
Step 15	`voice vad-time` *milliseconds* **Example:** `Router(config)# voice vad-time 250`	Changes the minimum silence detection time for VAD for all voice ports, but does not affect calls already in progress. • With a longer silence detection delay, VAD reacts to the silence of an idle voice channel, but not to pauses in conversation. Range is from 250 to 65536. The default is 250. • Minimum silence detection time is similar to drop-out time on VOX LMR systems, in which a pause longer than the drop-out time unkeys the system. If your LMR system has problems with the transmitter keying and unkeying, increase the value of the v**oice vad-time** command.

Step 16	`end`
	
	Example:
	`Router(config)# end`
		Exits to privileged EXEC mode.

Troubleshooting Tips

- Make sure that the hardware interface is configured correctly for 2-wire or 4-wire operation.
- Make sure that the hardware interface is configured correctly for E&M signaling Type II, III, or V.
- Make sure the hardware interface matches the configured transmit and receive signal protocol and signal polarity.
- Determine whether the problem is related to generic call handling or is LMR-specific.
- Collect information with **debug** and **show** commands.
- Adjust configuration parameters.

Verifying Land Mobile Radio over IP Enhancement

Perform this task to verify that the Land Mobile Radio over IP Enhancement feature is working.

SUMMARY STEPS

1. **enable**
2. **debug vpm signal**
3. **debug vpm trunk-sc**
4. **show call active voice** [[**brief**] [**called-number** *number* | **calling-number** *number*] | **compact** [**duration** {**less** *time* | **more** *time*}] | **echo-canceller** *call-id* | **id** *identifier* | **media-inactive** [**called-number** *number* | **calling-number** *number*] | **redirect** {**rtpvt** | **tbct**}]
5. **show interfaces vif 1**
6. **show ip mroute** [**vrf** *vrf-name*] [*group-address* | *group-name*] [*source-address* | *source-name*] [*interface-type interface-number*] [**summary**] [**count**] [**active** *kbps*]

7. **show ip pim** [**vrf** *vrf-name*] **neighbor** [interface-type *interface-number*]
8. **show voice call** [[*slot/port***:***ds0-group-number* | *slot/subunit/port*] | **status** *call-id* [**sample** *sample-period*] | **summary**]]
9. **show voice dsp**
10. **show voice lmr** [*slot/subunit/port* | *slot/port*:*ds0-group-number*] [**details**]
11. **show voice port** [[*slot/subunit/port* | *slot/port***:***ds0-group-number*] | **summary**]
12. **show voip rtp connections** [**detail**]
13. **show voice trunk-conditioning signaling** [**summary** | *voice-port*]
14. **show voice trunk-conditioning supervisory** [**summary** | *voice-port*]
15. **test voice port** {*slot/subunit/port* | *slot/port***:***ds0-group-number*} **detector m-lead** {**on** | **off** | **disable**}
16. **test voice port** {*slot/port***:***ds0-group-number* | *slot/subunit/port*} **relay e-lead** {**on** | **off** | **disable**}
17. **test voice port** {*slot/subunit/port* | *slot/port*:*ds0-group-number*} **inject-tone** {**local** | **network**} {**200hz** | **300hz** | **500hz** | **1000hz** | **2000hz** | **3000hz** | **3200hz** | **3400hz** | **quiet** | **disable**}

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**debug vpm signal** **Example:** `Router# debug vpm signal`	(Optional) Collects debug information only for signaling events.
Step 3	**debug vpm trunk-sc** **Example:** `Router# debug vpm trunk-sc`	(Optional) Enables the display of trunk conditioning supervisory component trace information.
Step 4	**show call active voice** [[**brief**] [**called-number** *number* | **calling-number** *number*] | **compact** [**duration** {**less** *time* | **more** *time*}] | **echo-canceller** *call-id* | **id** *identifier* | **media-inactive** [**called-number** *number* | **calling-number** *number*] | **redirect** {**rtpvt** | **tbct**}] **Example:** `Router# show call active voice brief`	(Optional) Displays call information for voice calls in progress.
Step 5	**show interfaces vif 1** **Example:** `Router# show interfaces vif 1`	(Optional) Displays statistics for all interfaces configured on the router.

	Command or Action	Purpose
Step 6	**show ip mroute** [**vrf** *vrf-name*] [*group-address* | *group-name*] [*source-address* | *source-name*] [*interface-type interface-number*] [**summary**] [**count**] [**active** *kbps*] **Example:** `Router# show ip mroute summary`	(Optional) Displays the contents of the IP multicast (mroute) routing table.
Step 7	**show ip pim** [**vrf** *vrf-name*] **neighbor** [*interface-type interface-number*] **Example:** `Router# show ip pim neighbor`	(Optional) Lists the PIM neighbors discovered by the Cisco IOS software.
Step 8	**show voice call** [[*slot*/*port*:*ds0-group-number* | *slot*/*subunit*/*port*] | **status** *call-id* [**sample** *sample-period*] | **summary**]] **Example:** `Router# show voice call summary`	(Optional) Displays the call status for voice ports on the Cisco router.
Step 9	**show voice dsp** **Example:** `Router# show voice dsp`	(Optional) Displays the current status of all DSP voice channels.
Step 10	**show voice lmr** [*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group-number*] [**details**] **Example:** `Router# show voice lmr 1/0/0 details`	(Optional) Displays LMR-related dynamic and static information for LMR ports or ds0 groups.
Step 11	**show voice port** [[*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group-number*] | **summary**] **Example:** `Router# show voice port summary`	(Optional) Displays configuration information about a specific voice port.
Step 12	**show voip rtp connections** [**detail**] **Example:** `Router# show voip rtp connections`	(Optional) Displays the local and remote calling ID and IP address and port information for active RTP connections.
Step 13	**show voice trunk-conditioning signaling** [**summary** | *voice-port*] **Example:** `Router# show voice trunk-conditioning signaling`	(Optional) Displays the status of trunk-conditioning signaling and timing parameters for a voice port.

	Command or Action	Purpose
Step 14	**show voice trunk-conditioning supervisory** [**summary** | *voice-port*] **Example:** `Router# show voice trunk-conditioning supervisory`	(Optional) Displays the status of trunk supervision and configuration parameters for a voice port.
Step 15	**test voice port** {*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group-number*} **detector m-lead** {**on** | **off** | **disable**} **Example:** `Router# test voice port 4/0:2 detector m-lead on`	(Optional) Tests detector-related functions on a voice port. • Forces a detector into specific states for testing. – **on** (seize)—Cisco IOS software believes that the COR from the radio is active or squelch is open. – **off** (release)—Simulates release. Cisco IOS software believes that the COR from the radio is released or squelch is closed. – **disable**—Returns the M-lead to operating state.
Step 16	**test voice port** {*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group-number*} **relay e-lead** {**on** | **off** | **disable**} **Example:** `Router# test voice port 4/0:2 relay e-lead on`	(Optional) Tests relay-related functions on a voice port. • Forces a relay into specific states for testing. Use this command to enable and disable the E-lead and see if it keys the transmitter. – **on** (seize)—Activates PTT on the radio. – **off** (release)—Releases PTT on the radio. – **disable**—Returns the M-lead to operating state.
Step 17	**test voice port** {*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group-number*} **inject-tone** {**local** | **network**} {**200hz** | **300hz** | **500hz** | **1000hz** | **2000hz** | **3000hz** | **3200hz** | **3400hz** | **quiet** | **disable**} **Example:** `Router# test voice port 4/0:2 inject-tone local 1000hz`	(Optional) Injects a test tone into a voice port. • Instructs Cisco IOS software to send a tone on the E&M interface of the specified frequency at –7.5 dBm0. • It is sometimes required to seize the M-lead in order to inject tone on the local interface.

Examples

Examples for commands not shown in this section can be found in the "Configuration Examples for Land Mobile Radio over IP Enhancement" section. The **test voice port detector** and **test voice port relay** commands do not produce any output and are not included in this section. This section provides the following output examples:

Sample Output for the debug vpm signal Command

In the following example, the E-lead and M-lead of the LMR voice port are configured as follows:

```
voice-port 4/0:1
 lmr m-lead dialin
 lmr e-lead voice
```

In the following sample output of the **debug vpm signal** command at the terminating side of the call, the output in bold indicates that the call connects:

```
TermRouter#

1w3d:htsp_timer_stop3 htsp_setup_req
1w3d:htsp_process_event:[4/0:1(1), LMR_ONHOOK,
E_HTSP_SETUP_REQ]lmr_onhook_setup
1w3d:htsp_timer_stop htsp_progress
1w3d:lmr_start_timer:2000 ms
1w3d:htsp_timer - 2000 msechtsp_call_bridged
1w3d:htsp_process_event:[4/0:1(1), LMR_WAIT_CUT_THRU,
E_HTSP_VOICE_CUT_THROUGH]lmr_cut_thru
1w3d:htsp_timer_stop
1w3d:lmr_pak_suppress_enable FALSE
1w3d:lmr_start_timer2:1800 second
1w3d:htsp_timer2 - 1800000 msec
1w3d:htsp_process_event:[4/0:1(1), LMR_CONNECT,
E_DSP_SIG_0000]lmr_conn_onhook
1w3d:htsp_timer_stop
1w3d:lmr_start_timer:480 ms
1w3d:htsp_timer - 480 msec
```

In the following sample output of the **debug vpm signal** command at the originating side of the call, the output in bold indicates that the call connects:

```
OrigRouter#

1w3d:htsp_process_event:[4/0:1(1), LMR_ONHOOK,
E_DSP_SIG_1100]lmr_onhook_offok
1w3d:htsp_timer_stop htsp_setup_ind
1w3d:[4/0:1(1)] get_local_station_id calling num= calling name= calling
time=/18 00:53  orig called=
1w3d:htsp_timer - 3000 msec
1w3d:htsp_process_event:[4/0:1(1), LMR_WAIT_SETUP_ACK,
E_HTSP_SETUP_ACK]lmr_it_setup_ack_get_ack
1w3d:htsp_timer_stop
1w3d:htsp_process_event:[4/0:1(1), LMR_OFFHOOK, E_HTSP_PROCEEDING]
1w3d:htsp_timer_stop3 htsp_setup_req
E_HTSP_VOICE_CUT_THROUGHxsls_waitoff_voice
1w3d:htsp_process_event:[4/0:1(1), LMR_OFFHOOK,
E_HTSP_VOICE_CUT_THROUGH]lmrffhook_voice_cut
1w3d:htsp_timer_stop
ST3745#
1w3d:htsp_process_event:[4/0:1(1), LMR_OFFHOOK,
E_HTSP_CONNECT]lmr_offhook_cnect
1w3d:htsp_timer_stop
1w3d:htsp_timer_stop2
```

Sample Output for the debug vpm trunk-sc Command

When tone is injected into port 1/0/0 locally, the loopback cable sends tone back into the multicast causing the other port 1/0/1 to receive the voice packets and the playout pattern 0xF to appear in the **debug vpm trunk-sc** command output. In the following output, the signal pattern 0xF shown in bold confirms that voice packet detection is working on the voice port.

```
*Jun 13 23:52:39.699: 1/0/1: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_RCVD
```

```
*Jun 13 23:52:39.699: 1/0/1: trunk_rtc_gen_pattern : sig pattern 0xF
Router# sh voice call 1/0/0
1/0/0
      vtsp level 0 state = S_CONNECTvpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
calling number , calling name unavailable, calling time 06/13 23:12
Router#   ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 12360, Tx Sig Pkts: 0, Tx Comfort Pkts: 7
Tx Dur(ms): 2423550, Tx Vox Dur(ms): 247180, Tx Fax Dur(ms): 0
```

Sample Output for the show call active voice Command

The following example shows information from a router in a multicast group for a connection trunk call in progress made in a VoIPmc network:

```
Telephony call-legs:1
SIP call-legs:0
H323 call-legs:0
MGCP call-legs:0
Multicast call-legs:1
Total call-legs:2
GENERIC:
SetupTime=565861590 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=23
ConnectTime=56586159
CallDuration=00:00:30 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=148
TransmitBytes=24864
ReceivePackets=50
ReceiveBytes=800
TELE:
! CoderTypeRate is the codec used in this call
! TranslatedCalledNumber is the number being called
ConnectionId=[0x467E1D6E 0x5BAD11D8 0x805CDA45 0x64E0FF68]
IncomingConnectionId=[0x467E1D6E 0x5BAD11D8 0x805CDA45 0x64E0FF68] CallID=80
TxDuration=30720 ms VoiceTxDuration=2000 ms FaxTxDuration=0 ms CoderTypeRate=g711ulaw
NoiseLevel=-72 ACOMLevel=57 OutSignalLevel=-15 InSignalLevel=-71 InfoActivity=2
ERLLevel=57 SessionTarget= ImgPages=0 CallerName= CallerIDBlocked=False
OriginalCallingNumber= OriginalCallingOctet=0x0 OriginalCalledNumber=
OriginalCalledOctet=0x80 OriginalRedirectCalledNumber= OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber= TranslatedCallingOctet=0x0 TranslatedCalledNumber=7667
TranslatedCalledOctet=0x80 TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GENERIC:
SetupTime=565861590 ms
Index=2
! PeerAddress is the number being called
PeerAddress=7667
PeerSubAddress=
PeerId=20
PeerIfIndex=36
LogicalIfIndex=0
ConnectTime=0
CallDuration=00:00:00 sec
CallState=4
```

```
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=0
TransmitBytes=0
ReceivePackets=148
ReceiveBytes=4294965520
VOIP:
! IP address of the called side and call parameters
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x467E1D6E 0x5BAD11D8 0x805CDA45 0x64E0FF68] CallID=81
RemoteIPAddress=0.0.0.0 RemoteUDPPort=19878 RemoteSignallingIPAddress=0.0.0.0
RemoteSignallingPort=0 RemoteMediaIPAddress=234.5.6.7 RemoteMediaPort=19878
RoundTripDelay=0 ms SelectedQoS=best-effort tx_DtmfRelay=inband-voice FastConnect=FALSE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=FALSE
! Session protocol and session target set in the selected dial peer
SessionProtocol=multicast
ProtocolCallId=
SessionTarget=ipv4:234.5.6.7:19878
OnTimeRvPlayout=0
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=0 ms
LoWaterPlayoutDelay=0 ms
TxPakNumber=100
TxSignalPak=0
TxComfortNoisePak=1
TxDuration=30720
TxVoiceDuration=2000
RxPakNumber=45
RxSignalPak=0
RxDuration=0
TxVoiceDuration=900
VoiceRxDuration=850
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=64
PlayDelayMin=64
PlayDelayMax=65
PlayDelayClockOffset=11533
PlayDelayJitter=67085464
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-15
InSignalLevel=-71
LevelTxPowerMean=0
LevelRxPowerMean=-715
LevelBgNoise=0
ERLLevel=57
ACOMLevel=57
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=45
ErrRxControl=66
PlayoutMode = undefined
```

```
PlayoutInitialDelay=0 ms
ReceiveDelay=0 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
!VAD status of the call
VAD = enabled
CoderTypeRate=g711ulaw
CodecBytes=0
Media Setting=flow-around
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber= TranslatedRedirectCalledOctet=0x0 MediaInactiveDetected=no
MediaInactiveTimestamp= MediaControlReceived= Username= Telephony call-legs:1 SIP
call-legs:0 H323 call-legs:0 MGCP call-legs:0 Multicast call-legs:1 Total call-legs:2
```

Sample Output for the show voice call Command

In the following example, the fields in bold show that the call is connected:

```
1/0/0
      vtsp level 0 state = S_CONNECTvpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
calling number , calling name unavailable, calling time 06/13 23:12
Router#   ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 5022, Tx Sig Pkts: 0, Tx Comfort Pkts: 4
Tx Dur(ms): 2182560, Tx Vox Dur(ms): 100430, Tx Fax Dur(ms): 0
```

Sample Output for the show voice dsp Command

When nothing is connected to the E&M voice port, there is a very slow increment of the PACK COUNT field due to RTP Control Protocol (RTCP). However, when the radio is transmitting or receiving, the PACK COUNT increments by a relatively large amount. Incrementing of the PACK COUNT field also indicates whether the packets are being received and sent by the router. The output is shown for the C542 and C5510 DSP.

```
Router# show voice dsp

!The following output is for the NM-HD-2V analog module

DSP  DSP             DSPWARE CURR  BOOT                           PAK     TX/RX
TYPE NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABORT  PACK COUNT
==== === == ======== ======= ===== ======= === == ========= == ===== ===========

C542 001 01 g711ulaw  4.3.10 busy  idle      0  0 1/0/0     NA     0        16/16
C542 002 01 g711ulaw  4.3.10 busy  idle      0  0 1/0/1     NA     0        16/16

!The following output is for the NM-HD-2V analog module

                           *DSP SIGNALING CHANNELS*
DSP   DSP             DSPWARE CURR  BOOT                           PAK   TX/RX
TYPE  NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ===========
```

```
C5510 001 01 {flex}    4.3.10 alloc idle      0  0 2/1/0     02    0           0/0
C5510 001 02 {flex}    4.3.10 alloc idle      0  0 2/1/1     02    0           0/0
```

Sample Output for the show voice lmr Command

Use the **show voice lmr** command to verify LMR-specific configuration, for example, E-lead and M-lead status, timeouts, and injected tones, shown in bold in the following example output.

```
Router# show voice lmr 2/0/0 details

2/0/0
=========
 Description:
 Connection type: n/a
 Out Attenuation = 0 db, In Gain = 0 dB
 Timing hangover: 500 ms
 E-lead capability is inactive, polarity = normal
 M-lead capability is inactive, polarity = normal
 Timing hookflash-in: 480
 Timing delay-voice: 470 ms
 Music On Hold Threshold: -38 dB, Noise Threshold: -62 dB
 E&M type: 1, Operation:  2-wire
 Impedance is set to 600r Ohm
 lmr tear down timeout is set to 1800 second
 lmr PTT transmit timeout is not set
 lmr PTT receive timeout is not set
 voice-class tone-signal test
         inject tone 1 1950 3 150
         inject tone 2 2000 0 60
         inject pause 3 60
         inject tone 4 2175 3 150
         inject tone 5 1000 0 50
         inject guard-tone 1950 -10
 state = LMR_CONNECT, e-lead = off, m-lead = off
 full duplex, voice path = rx
 Terminating side of the connection
 TransmitPackets=113, TransmitBytes=2241
 ReceivePackets=113, ReceiveBytes=2241
 CoderTypeRate=g729r8
 NoiseLevel=-66, ACOMLevel=22
 OutSignalLevel=-68, InSignalLevel=-79
 PeerAddress=37200
 PeerSubAddress=
 PeerId=200
 SessionTarget=

 RemoteIPAddress=10.5.25.40, RemoteUDPPort=17272
 Remote SignallingIPAddress=10.5.25.40, Port=15418
 Remote MediaIPAddress=10.5.25.40, Port=17272
 RoundTripDelay=0 ms
 SessionProtocol=cisco
 VAD =enabled
 SelectedQoS=best-effort
 ProtocolCallId=
 SessionTarget=
```

Sample Output for the test voice port inject-tone Command

When packets are sent to the multicast group using the **test voice port inject-tone network** command, the packets are also received on the ports. The same multicast IP:port is used for sending and receiving when configured for multicast. Only symmetric send and receive configuration is supported on dial peers.

```
Router# test voice port 1/0/1 inject-tone network 1000

*Jun 13 23:44:51.943: 1/0/0: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_RCVD
*Jun 13 23:44:51.943: 1/0/0: trunk_rtc_gen_pattern : sig pattern 0xF
*Jun 13 23:44:52.035: 1/0/1: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_RCVD
*Jun 13 23:44:52.035: 1/0/1: trunk_rtc_gen_pattern : sig pattern 0xF

Router# test voice port 1/0/1 inject-tone network quiet

*Jun 13 23:45:07.455: 1/0/0: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_STOPPED
*Jun 13 23:45:07.455: 1/0/0: trunk_rtc_gen_pattern : sig pattern 0x0
*Jun 13 23:45:07.543: 1/0/1: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_STOPPED
*Jun 13 23:45:07.543: 1/0/1: trunk_rtc_gen_pattern : sig pattern 0x0
```

Injecting the tone locally into the E&M port causes the high dB levels in TX and RX, which confirms that the audio leads of the E&M voice port are functional. The tone remains active for only 30 seconds.

```
Router# test voice port 1/0/0 inject-tone local 1000

*Jun 13 23:52:39.699: 1/0/1: TRUNK_SC state : TRUNK_SC_CONNECT, event TRUNK_VOICE_RCVD
*Jun 13 23:52:39.699: 1/0/1: trunk_rtc_gen_pattern : sig pattern 0xF
Router# sh voice call 1/0/0

1/0/0
      vtsp level 0 state = S_CONNECTvpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
calling number , calling name unavailable, calling time 06/13 23:12
Router#   ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 12360, Tx Sig Pkts: 0, Tx Comfort Pkts: 7
Tx Dur(ms): 2423550, Tx Vox Dur(ms): 247180, Tx Fax Dur(ms): 0
<<<<Removed for clarity >>>>>

***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -0.2 from PBX/Phone, Tx -0.5 to PBX/Phone
TDM ACOM Levels(dBm0): +20.0, TDM ERL Level(dBm0): +20.0
TDM Bgd Levels(dBm0): -75.2, with activity being voice
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
```

Configuration Examples for Land Mobile Radio over IP Enhancement

This section provides configuration examples to match the identified configuration tasks in the previous section. This section does not provide examples for every option under every configuration task.

- Configuring Connection Trunk on an Analog LMR Voice Port: Example, page 82
- Verifying Connection Trunk on an Analog LMR Voice Port: Example, page 83
- Configuring Connection Trunk on a Digital LMR Voice Port: Example, page 84
- Verifying Connection Trunk on a Digital LMR Voice Port: Example, page 85

Configuring Connection Trunk on an Analog LMR Voice Port: Example

The following output shows an analog LMR voice port and a connection trunk connection configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 2456 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
!
interface FastEthernet0/0
 ip address 10.2.82.81 255.255.255.0
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
!
!
voice-port 1/0/0
 lmr m-lead audio-gate-in
 lmr e-lead voice
 operation 4-wire
 type 5
 signal lmr
 timeouts call-disconnect 3
 connection trunk 7667
!
!
```

```
dial-peer voice 30 voip
 destination-pattern 7667
 session target ipv4:10.2.82.82
 codec g711ulaw
 vad aggressive
!
dial-peer voice 40 pots
 destination-pattern 8668
 port 1/0/0
!
```

Verifying Connection Trunk on an Analog LMR Voice Port: Example

In this example, the **show voice port summary** command displays configuration information about the LMR voice port. The fields in bold show that the voice port's signaling type is E&M LMR and that connection trunk is configured.

```
Router# show voice port summary

                                     IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0/0     --  e&m-lmr     up    up   trunked  trunked  y
```

In this example, the **show voice trunk-conditioning supervisory** command displays the status of trunk supervision and configuration parameters for the LMR voice port. The **voice packet detection enable** statement shows that the **voice** option of the command has been configured. If an option other than **voice** has been configured, the **voice packet detection disable** statement appears.

```
Router# show voice trunk-conditioning supervisory

SLOW SCAN, SCAN LMR
1/0/0 : state : TRUNK_SC_CONN_WO_CLASS, voice : off , signal : on ,master
        status: rcv IDLE, trunk connected
        sequence oos : no-action
        pattern :tx_idle = 0000
        timing : idle = 480, restart = 0, standby = 0, timeout = 30
        supp_all = 0, supp_voice = 0, keep_alive = 5
        timer: oos_ais_timer = 0, timer = 0
        voice packet detection enable
```

In this example, the **show voice trunk-conditioning signaling** command displays the status of trunk-conditioning signaling and timing parameters for the LMR voice port.

When the trunk is the ACTIVE state, the forced playout pattern field is set to **0xF**. In this example, the forced playout pattern field is set to **0x0** indicating that the trunk is idle.

```
Router# show voice trunk-conditioning signaling

1/0/0 :
hardware-state IDLE signal type is NorthamericanCAS
status : IDLE
forced playout pattern = 0x0
last-TX-ABCD=0000, last-RX-ABCD=0000
idle monitoring : tx
tx_idle = TRUE, rx_idle = FALSE, tx_oos = FALSE, lost_keepalive = FALSE
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0,tx_oos_timer = 0
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary
```

```
PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0/0          g711ulaw  y  S_CONNECT            S_TRUNKED
```

Configuring Connection Trunk on a Digital LMR Voice Port: Example

The following output shows a digital LMR voice port and a connection trunk connection configured on the router:

```
Router# show running-config
Building configuration...

Current configuration : 1615 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
voice-card 1
 dspfarm
!
!
controller T1 1/0
 framing esf
 crc-threshold 320
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-lmr
!
!
interface FastEthernet0/0
 ip address 10.2.82.82 255.255.255.0
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
!
!
!
voice-port 1/0:1
 lmr m-lead audio-gate-in
 timeouts call-disconnect 3
 connection trunk 8668
!
!
dial-peer voice 30 voip
 destination-pattern 8668
 session target ipv4:10.2.82.81
 codec g711ulaw
 vad aggressive
!
dial-peer voice 40 pots
 destination-pattern 7667
 port 1/0:1
!
```

Verifying Connection Trunk on a Digital LMR Voice Port: Example

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0:1.1         g711ulaw  y  S_CONNECT             S_TRUNKED
```

In this example, the **show voice trunk-conditioning supervisory** command displays the status of trunk supervision and configuration parameters for the LMR voice port:

```
Router# show voice trunk-conditioning supervisory

SLOW SCAN, SCAN LMR
1/0:1(1) : state : TRUNK_SC_CONN_WO_CLASS, voice : off , signal : on ,master
        status: rcv IDLE, trunk connected
        sequence oos : no-action
        pattern :tx_idle = 0000
        timing : idle = 480, restart = 0, standby = 0, timeout = 30
        supp_all = 0, supp_voice = 0, keep_alive = 5
        timer: oos_ais_timer = 0, timer = 0
        voice packet detection disable
```

In this example, the **show voice trunk-conditioning signaling** command displays the status of trunk-conditioning signaling and timing parameters for the LMR voice port. When the trunk is the ACTIVE state, the forced playout pattern field is set to **0xF**.

```
Router# show voice trunk-conditioning signaling

1/0:1(1) :
hardware-state IDLE signal type is NorthamericanCAS
status : IDLE
forced playout pattern = STOPPED
last-TX-ABCD=0000, last-RX-ABCD=0000
idle monitoring : tx
tx_idle = TRUE, rx_idle = FALSE, tx_oos = FALSE, lost_keepalive = FALSE
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0,tx_oos_timer = 0
```

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                     IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0:1     01  e&m-lmr     up    up   trunked  trunked  y
```

Configuring PLAR on an Analog LMR Voice Port: Example

The following output shows an analog LMR voice port and a PLAR connection configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 2427 bytes
!
version 12.3
```

```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
interface FastEthernet0/0
 ip address 10.2.82.81 255.255.255.0
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
!
!
voice-port 1/0/0
 lmr m-lead audio-gate-in
 lmr e-lead voice
 operation 4-wire
 type 5
 signal lmr
 connection plar 7667
!
!
!
dial-peer voice 30 voip
 destination-pattern 7667
 session target ipv4:10.2.82.82
 codec g711ulaw
 vad aggressive
!
dial-peer voice 40 pots
 destination-pattern 8668
 port 1/0/0
!
```

Verifying PLAR on an Analog LMR Voice Port: Example

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                     IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0/0     --  e&m-lmr     up    up   idle     idle     y
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0/0          g711ulaw  y  S_CONNECT            LMR_CONNECT
```

In this example, the **show voip rtp connections** command displays the local and remote calling ID and IP address and port information for active RTP connections:

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   67      68         16884    16572  10.2.82.81      10.2.82.82
Found 1 active RTP connections
```

Configuring PLAR on a Digital LMR Voice Port: Example

The following output shows a digital LMR voice port and a PLAR connection configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 1579 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
voice-card 1
 dspfarm
!
!
controller T1 1/0
 framing esf
 crc-threshold 320
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-lmr
!
!
interface FastEthernet0/0
 ip address 10.2.82.82 255.255.255.0
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
!
!
voice-port 1/0:1
 lmr m-lead dialin
 connection plar 8668
!
!
dial-peer voice 30 voip
 destination-pattern 8668
 session target ipv4:10.2.82.81
 codec g711ulaw
 vad aggressive
!
dial-peer voice 40 pots
 destination-pattern 7667
 port 1/0:1
```

Verifying PLAR on a Digital LMR Voice Port: Example

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE           VPM STATE
============== ======== === ==================== ======================
1/0:1.1         g711ulaw  y  S_CONNECT             LMR_CONNECT
```

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                     IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0:1     01  e&m-lmr     up    up   idle     seized   y
```

In this example, the **show voip rtp connections** command displays the local and remote calling ID and IP address and port information for active RTP connections:

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   80      79         16572    16884  10.2.82.82      10.2.82.81
Found 1 active RTP connections
```

Configuring VoIPmc with Connection Trunk on an Analog LMR Voice Port: Example

The following output shows an analog LMR voice port and a VoIPmc connection using connection trunk configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 3101 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
voice class permanent 1
 signal timing oos timeout disabled
 signal keepalive disabled
!
!
!
interface Loopback1
 ip address 10.0.0.1 255.0.0.0
 ip pim sparse-dense-mode
```

```
!
interface Vif1
 ip address 10.3.92.1 255.255.255.0
 ip accounting output-packets
 ip pim sparse-dense-mode
 load-interval 30
!
interface FastEthernet0/0
 ip address 10.2.82.81 255.255.255.0
 ip pim sparse-dense-mode
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.0.0.0 255.0.0.0 FastEthernet0/0
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
ip route 10.2.92.0 255.255.255.0 FastEthernet0/0
!
ip http server
no ip http secure-server
ip pim bidir-enable
ip pim rp-address 10.0.0.1 override bidir
ip pim send-rp-announce Loopback1 scope 16
ip pim send-rp-discovery Loopback1 scope 16
!
voice-port 1/0/0
 voice-class permanent 1
 lmr m-lead audio-gate-in
 lmr e-lead voice
 operation 4-wire
 type 5
 signal lmr
 timeouts call-disconnect 3
 connection trunk 7667
!
!
dial-peer voice 20 voip
 destination-pattern 7667
 session protocol multicast
 session target ipv4:239.5.6.7:19878
 codec g711ulaw
 vad aggressive
!
!
```

Verifying VoIPmc with Connection Trunk on an Analog LMR Voice Port: Example

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                      IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0/0     --  e&m-lmr     up    up   trunked  trunked  y
```

In this example, the **show voice trunk-conditioning supervisory** command displays the status of trunk supervision and configuration parameters for the LMR voice port. The **voice packet detection enable** statement shows that the **voice** option of the command has been configured. If an option other than **voice** has been configured, the **voice packet detection disable** statement appears.

```
Router# show voice trunk-conditioning supervisory

SLOW SCAN, SCAN LMR
1/0/0 : state : TRUNK_SC_CONNECT, voice : off , signal : on ,master
        status: rcv IDLE, trunk connected
        sequence oos : idle-only
        pattern :rx_idle = 0000 tx_idle = 0000
        timing : idle = 480, restart = 0, standby = 0, timeout = 0
        supp_all = 0, supp_voice = 0, keep_alive = 0
        timer: oos_ais_timer = 0, timer = 0
        voice packet detection enable
```

In this example, the **show voice trunk-conditioning signaling** command displays the status of trunk-conditioning signaling and timing parameters for the LMR voice port. When the trunk is the ACTIVE state, the forced playout pattern field is set to **0xF**.

```
Router# show voice trunk-conditioning signaling

1/0/0 :
hardware-state IDLE signal type is NorthamericanCAS
status : IDLE
forced playout pattern = 0x0
last-TX-ABCD=0000, last-RX-ABCD=0000
idle monitoring : tx
tx_idle = TRUE, rx_idle = FALSE, tx_oos = FALSE, lost_keepalive = FALSE
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0,tx_oos_timer = 0
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0/0         g711ulaw  y  S_CONNECT             S_TRUNKED
```

In this example, the **show ip pim neighbor** command lists the PIM neighbors discovered by the Cisco IOS software:

```
Router# show ip pim neighbor

PIM Neighbor Table
Neighbor          Interface                Uptime/Expires    Ver   DR
Address                                                            Prio/Mode
10.2.82.82        FastEthernet0/0          00:06:51/00:01:17 v2    1 / DR B S
```

In this example, the **show voip rtp connections** command displays the local and remote calling ID and IP address and port information for active RTP connections:

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   37      36         17090    19878 10.3.92.2       239.5.6.7
Found 1 active RTP connections
```

In this example, the **show interfaces vif 1** command displays statistics for the vif 1 interface:

```
Router# show interfaces vif 1
```

```
Vif1 is up, line protocol is up
  Hardware is PGMVIF
  Internet address is 92.3.92.1/24
  MTU 1514 bytes, BW 8000000 Kbit, DLY 5000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation LOOPBACK, loopback not set
  Last input 00:00:02, output never, output hang never
  Last clearing of "show interface" counters 00:10:57
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 0
  Queueing strategy: fifo
  Output queue: 0/0 (size/max)
  30 second input rate 0 bits/sec, 0 packets/sec
  30 second output rate 0 bits/sec, 0 packets/sec
     181 packets input, 29721 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     482 packets output, 80478 bytes, 0 underruns
     0 output errors, 0 collisions, 0 interface resets
     0 output buffer failures, 0 output buffers swapped out
```

Configuring VoIPmc with Connection Trunk on a Digital LMR Voice Port: Example

The following output shows a digital LMR voice port and a VoIPmc connection using connection trunk configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 2250 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
!
voice-card 1
 dspfarm
!
!
voice class permanent 1
 signal timing oos timeout disabled
 signal keepalive disabled
!
!
controller T1 1/0
 framing esf
 crc-threshold 320
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-lmr
!
!
interface Loopback1
 ip address 10.0.0.1 255.0.0.0
 ip pim sparse-dense-mode
!
```

```
interface Vif1
 ip address 10.2.92.1 255.255.255.0
 ip accounting output-packets
 ip pim sparse-dense-mode
 load-interval 30
!
interface FastEthernet0/0
 ip address 10.2.82.82 255.255.255.0
 ip pim sparse-dense-mode
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.0.0.0 255.0.0.0 FastEthernet0/0
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
ip route 10.3.92.0 255.255.255.0 FastEthernet0/0
!
ip http server
no ip http secure-server
ip pim bidir-enable
ip pim rp-address 50.0.0.1 override bidir
ip pim send-rp-announce Loopback1 scope 16
ip pim send-rp-discovery scope 16
!
!
voice-port 1/0:1
 voice-class permanent 1
 lmr m-lead audio-gate-in
 timeouts call-disconnect 3
 connection trunk 8668
!
dial-peer voice 20 voip
 destination-pattern 8668
 session protocol multicast
 session target ipv4:239.5.6.7:19878
 codec g711ulaw
 vad aggressive
!
```

Verifying VoIPmc with Connection Trunk on a Digital LMR Voice Port: Example

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                         IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0:1     01  e&m-lmr     up    up   trunked  trunked  y
```

In this example, the **show voice trunk-conditioning supervisory** command displays the status of trunk supervision and configuration parameters for the LMR voice port. The **voice packet detection disable** statement shows that an option other than **voice** is configured for the command. If the **voice** option of the **lmr e-lead** command was configured, the **voice packet detection enable** statement would appear.

```
Router# show voice trunk-conditioning supervisory

SLOW SCAN, SCAN LMR
```

```
1/0:1(1) : state : TRUNK_SC_CONNECT, voice : off , signal : on ,master
        status: rcv IDLE, trunk connected
        sequence oos : no-action
        pattern :tx_idle = 0000
        timing : idle = 480, restart = 0, standby = 0, timeout = 0
        supp_all = 0, supp_voice = 0, keep_alive = 0
        timer: oos_ais_timer = 0, timer = 0
        voice packet detction disable
```

In this example, the **show voice trunk-conditioning signaling** command displays the status of trunk-conditioning signaling and timing parameters for the LMR voice port. When the trunk is the ACTIVE state, the forced playout pattern field is set to 0xF.

```
Router# show voice trunk-conditioning signaling

1/0:1(1) :
hardware-state IDLE signal type is NorthamericanCAS
status : IDLE
forced playout pattern = STOPPED
last-TX-ABCD=0000, last-RX-ABCD=0000
idle monitoring : tx
tx_idle = TRUE, rx_idle = FALSE, tx_oos = FALSE, lost_keepalive = FALSE
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0,tx_oos_timer = 0
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0:1.1         g711ulaw  y  S_CONNECT             S_TRUNKED
```

In this example, the **show ip pim neighbor** command lists the PIM neighbors discovered by the Cisco IOS software:

```
Router# show ip pim neighbor

PIM Neighbor Table
Neighbor         Interface              Uptime/Expires    Ver   DR
Address                                                         Prio/Mode
10.2.82.81       FastEthernet0/0        00:25:20/00:01:34 v2    1 / B S
```

Configuring VoIPmc with Connection PLAR on an Analog LMR Voice Port: Example

The following output shows an analog LMR voice port and a VoIPmc connection using PLAR configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 3065 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
```

```
!
!
voice class permanent 1
 signal timing oos timeout disabled
 signal keepalive disabled
!
!
interface Loopback1
 ip address 10.0.0.1 255.0.0.0
 ip pim sparse-dense-mode
!
interface Vif1
 ip address 10.3.92.1 255.255.255.0
 ip accounting output-packets
 ip pim sparse-dense-mode
 load-interval 30
!
interface FastEthernet0/0
 ip address 10.2.82.81 255.255.255.0
 ip pim sparse-dense-mode
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.0.0.0 255.0.0.0 FastEthernet0/0
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
ip route 10.2.92.0 255.255.255.0 FastEthernet0/0
!
ip http server
no ip http secure-server
ip pim bidir-enable
ip pim rp-address 10.0.0.1 override bidir
ip pim send-rp-announce Loopback1 scope 16
ip pim send-rp-discovery Loopback1 scope 16
!
!
voice-port 1/0/0
 voice-class permanent 1
 lmr m-lead dialin
 lmr e-lead voice
 operation 4-wire
 type 5
 signal lmr
 connection plar 7667
!
!
dial-peer voice 20 voip
 destination-pattern 7667
 session protocol multicast
 session target ipv4:239.5.6.7:19878
 codec g711ulaw
 vad aggressive
!
```

Verifying VoIPmc with Connection PLAR on an Analog LMR Voice Port: Example

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                       IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0/0     --  e&m-lmr     up    up   seized   seized   y
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE            VPM STATE
============== ======== === ==================== ======================
1/0/0          g711ulaw  y  S_CONNECT            LMR_CONNECT
```

In this example, the **show voip rtp connections** command displays the local and remote calling ID and IP address and port information for active RTP connections:

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   45      44         25470    19878 10.3.92.2       239.5.6.7
Found 1 active RTP connections
```

In this example, the **show ip pim neighbor** command lists the PIM neighbors discovered by the Cisco IOS software:

```
Router# show ip pim neighbor

PIM Neighbor Table
Neighbor          Interface                Uptime/Expires    Ver   DR
Address                                                            Prio/Mode
10.2.82.82        FastEthernet0/0          00:07:20/00:01:18 v2    1 / DR B S
```

Configuring VoIPmc with Connection PLAR on a Digital LMR Voice Port: Example

The following output shows a digital LMR voice port and a VoIPmc connection using PLAR configured on the router:

```
Router# show running-config

Building configuration...

Current configuration : 2214 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
```

```
!
voice-card 1
 dspfarm
!
!
voice class permanent 1
 signal timing oos timeout disabled
 signal keepalive disabled
!
!
controller T1 1/0
 framing esf
 crc-threshold 320
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-lmr
!
!
interface Loopback1
 ip address 10.0.0.1 255.0.0.0
 ip pim sparse-dense-mode
!
interface Vif1
 ip address 10.2.92.1 255.255.255.0
 ip accounting output-packets
 ip pim sparse-dense-mode
 load-interval 30
!
interface FastEthernet0/0
 ip address 10.2.82.82 255.255.255.0
 ip pim sparse-dense-mode
 load-interval 30
 duplex auto
 speed auto
!
!
ip classless
ip route 10.0.0.0 255.0.0.0 FastEthernet0/0
ip route 10.2.82.0 255.255.255.0 FastEthernet0/0
ip route 10.3.92.0 255.255.255.0 FastEthernet0/0
!
ip http server
no ip http secure-server
ip pim bidir-enable
ip pim rp-address 10.0.0.1 override bidir
ip pim send-rp-announce Loopback1 scope 16
ip pim send-rp-discovery scope 16
!
!
voice-port 1/0:1
 voice-class permanent 1
 lmr m-lead dialin
 connection plar 8668
!
!
dial-peer voice 20 voip
 destination-pattern 8668
 session protocol multicast
 session target ipv4:239.5.6.7:19878
 codec g711ulaw
 vad aggressive
!
```

Verifying VoIPmc with Connection PLAR on a Digital LMR Voice Port: Example

In this example, the **show voip rtp connections** command displays the local and remote calling ID and IP address and port information for active RTP connections:

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   58      57         29636    19878 10.2.92.2       239.5.6.7
Found 1 active RTP connections
```

In this example, the **show voice call summary** command displays the call status for the LMR voice port:

```
Router# show voice call summary

PORT           CODEC    VAD VTSP STATE           VPM STATE
============== ======== === ==================== ======================
1/0:1.1        g711ulaw  y  S_CONNECT            LMR_CONNECT
```

In this example, the **show voice port summary** command displays configuration information about the LMR voice port:

```
Router# show voice port summary

                                     IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
1/0:1     01  e&m-lmr     up    up   idle     seized   y
```

In this example, the **show ip pim neighbor** command lists the PIM neighbors discovered by the Cisco IOS software:

```
Router# show ip pim neighbor

PIM Neighbor Table
Neighbor          Interface                Uptime/Expires    Ver   DR
Address                                                            Prio/Mode
10.2.82.81        FastEthernet0/0          00:07:13/00:01:25 v2    1 / B S
```

In this example, the **show ip mroute summary** command displays the contents of the IP multicast (mroute) routing table:

```
Router# show ip mroute summary

IP Multicast Routing Table
Flags: D - Dense, S - Sparse, B - Bidir Group, s - SSM Group, C - Connected,
       L - Local, P - Pruned, R - RP-bit set, F - Register flag,
       T - SPT-bit set, J - Join SPT, M - MSDP created entry,
       X - Proxy Join Timer Running, A - Candidate for MSDP Advertisement,
       U - URD, I - Received Source Specific Host Report, Z - Multicast Tunnel
       Y - Joined MDT-data group, y - Sending to MDT-data group
Outgoing interface flags: H - Hardware switched
 Timers: Uptime/Expires
 Interface state: Interface, Next-Hop or VCD, State/Mode

(*, 239.5.6.7), 00:01:56/00:02:53, RP 50.0.0.1, OIF count: 2, flags: BC

(*, 239.0.1.39), 00:08:03/00:02:56, RP 50.0.0.1, OIF count: 3, flags: BCL

(*, 239.0.1.40), 00:08:08/00:02:55, RP 50.0.0.1, OIF count: 2, flags: BCL
```

Additional References

The following sections provide references related to the Land Mobile Radio over IP Enhancement feature:

Related Documents

Related Topic	Document Title
Cisco IOS commands	*Cisco IOS Voice Command Reference*, Release 12.3 T
Codecs	Tech Note, *Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation*, Document ID 14069
E&M interfaces	• Tech Note, *Voice - Analog E&M Signaling Overview*, Document ID 14003 • Tech Note, *Understanding and Troubleshooting Analog E & M Interface Types and Wiring Arrangements*, Document ID 8111
Land Mobile Radio over IP	*Cisco Land Mobile Radio over IP Solution Reference Network Design*
V^3PN	White Paper, *Voice and Video Enabled IPSec VPN (V3PN)*
VoIPmc	*Cisco Hoot and Holler over IP*

Standards

Standards	Title
No new or modified standards are supported by this feature, and support for existing standards 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
RFC 2833	*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*

Technical Assistance

Description	Link
Technical Support Website, 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/en/US/support/index.html

Command Reference

This section documents only new and modified commands.

New Commands

- **bootup e-lead off**
- **digital-filter**
- **inject guard-tone**
- **inject pause**
- **inject tone**
- **lmr duplex half**
- **lmr e-lead**
- **lmr led-on**
- **lmr m-lead**
- **show voice lmr**
- **rtp payload-type**
- **test lmr clear-call**
- **timeout ptt**
- **timeouts teardown lmr**
- **timing delay-voice tdm**
- **timing hangover**
- **timing ignore m-lead**
- **voice class tone-signal**
- **voice-class tone-signal**

Modified Commands

- **define**
- **ds0-group (E1)**
- **ds0-group (T1)**
- **input gain**
- **music-threshold**

- **output attenuation**
- **show voip rtp connections**
- **signal**
- **signal keepalive**
- **timing hookflash-input**

bootup e-lead off

To prevent an analog ear and mouth (E&M) voice port from keying the attached radio on router boot up, use the **bootup e-lead off** command in voice-port configuration mode. To allow the analog E&M voice port to key the attached radio on boot up, use the **no** form of this command.

bootup e-lead off

no bootup e-lead off

Syntax Description

This command has no arguments or keywords.

Defaults

The analog E&M voice port keys the attached radio on radio boot up.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This feature was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command configures the E-lead behavior on boot up for both voice ports on the voice interface card (VIC).

Examples

The following example configures the analog E&M voice port to not key the attached radio on router boot up:

```
voice-port 1/0/0
 bootup e-lead off
```

define

To define the transmit and receive bits for North American ear and mouth (E&M), E&M Mercury Exchange Limited Channel-Associated Signaling (MELCAS), and Land Mobile Radio (LMR) voice signaling, use the **define** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

define {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

no define {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

Syntax Description

tx-bits	The bit pattern applies to the transmit signaling bits.
rx-bits	The bit pattern applies to the receive signaling bits.
seize	The bit pattern defines the seized state.
idle	The bit pattern defines the idle state.
0000 through **1111**	Specifies the bit pattern.

Defaults

The default is to use the preset signaling patterns as defined in American National Standards Institute (ANSI) and European Conference of Postal and Telecommunications Administrations (CEPT) standards, as follows:

- For North American E&M:
  - tx-bits idle 0000 (0001 if on E1 trunk)
  - tx-bits seize 1111
  - rx-bits idle 0000
  - rx-bits seize 1111
- For E&M MELCAS:
  - tx-bits idle 1101
  - tx-bits seize 0101
  - rx-bits idle 1101
  - rx-bits seize 0101
- For LMR:
  - tx-bits idle 0000
  - tx-bits seize 1111
  - rx-bits idle 0000
  - rx-bits seize 1111

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA3	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.1(2)T	The command was integrated into Cisco IOS Release 12.1(2)T.
12.3(4)XD	The LMR signaling type was added to the signaling types to which this command applies.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **define** command applies to E&M digital voice ports associated with T1/E1 controllers.

Use the **define** command to match the E&M bit patterns with the attached telephony device. Be careful not to define invalid configurations, such as all 0000 on E1, or identical seized and idle states. Use this command with the **ignore** command.

In LMR signaling, the **define** command is used to define polarity on E&M analog and digital voice ports.

Examples

To configure a voice port on a Cisco 2600 or Cisco 3600 series router that is sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
voice-port 1/0/0
 define rx-bits idle 1101
 define rx-bits seize 0101
 define tx-bits idle 1101
 define tx-bits seize 0101
```

In this example, reverse polarity is configured on a voice port on a Cisco 3700 series router that is sending traffic in LMR signaling format:

```
voice-port 1/0/0
 define rx-bits idle 1111
 define rx-bits seize 0000
 define tx-bits idle 1111
 define tx-bits seize 0000
```

Related Commands

Command	Description
condition	Manipulates the signaling bit-pattern for all voice signaling types.
ignore	Configures a North American E&M or E&M MELCAS voice port to ignore specific receive bits.

digital-filter

To specify the digital filter to be used before the voice packet is sent from the digital signal processor (DSP) to the network, use the **digital-filter** command in voice-class configuration mode. To remove the digital filter, use the **no** form of this command.

digital-filter {**1950hz** | **2175hz**}

no digital-filter {**1950hz** | **2175hz**}

Syntax Description

1950hz	Filter out 1950 Hz frequency.
2175hz	Filter out 2175 Hz frequency.

Defaults

Digital filtering is disabled.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **digital-filter** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). The digital filter improves voice quality by preventing transmission of the guard tone with the voice packet from the LMR system to the VoIP network. The guard tone is configured with the **inject guard-tone** command. The digital filter can be configured to filter out either 2175 Hz or 1950 Hz. Only one of these frequencies can be filtered out at a time. Filtering is performed by the DSP.

Examples

The following example specifies that 1950 Hz guard tone be filtered out of the voice packet before it is sent from the DSP to the network:

```
voice class tone-signal mytones
 digital-filter 1950hz
```

Related Commands

Command	Description
inject guard-tone	Plays out a guard tone with the voice packet.

ds0-group (E1)

To specify the DS0 time slots that make up a logical voice port on an E1 controller, specify the signaling type by which the router communicates with the PBX or PSTN, and define E1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

Cisco 1750 and Cisco 1751

ds0-group *ds0-group-number* **timeslots** *timeslot-list* {[**service** *service-type*] | [**type** **e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fgd-eana** | **fgd-os** | **fxs-ground-start** | **fxs-loop-start** | **none** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start**]}

no ds0-group *ds0-group-number*

Cisco 2600 Series (Except Cisco 2691), Cisco 3600 Series (Except Cisco 3660)

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-melcas-delay** | **e&m-melcas-immed** | **e&m-melcas-wink** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **r2-analog** | **r2-digital** | **r2-pulse**}

no ds0-group *ds0-group-number*

Cisco 2691, Cisco 2600XM Series, Cisco 2800 Series (Except Cisco 2801), Cisco 3660, Cisco 3700 Series, Cisco 3800 Series

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-lmr** |**e&m-melcas-delay** | **e&m-melcas-immed** | **e&m-melcas-wink** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **r2-analog** | **r2-digital** | **r2-pulse**}

no ds0-group *ds0-group-number*

Cisco 7200 Series and Cisco 7500 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | e&m-fgd | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group-number*

Cisco 7700 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group-number*

Cisco AS5300 and the Cisco AS5400

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**none** | **p7** | **r2-analog** | **r2-digital** | **r2-lsv181-digital** | **r2-pulse**}

no ds0-group *ds0-group-number*

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5x00 series.

Syntax Description

ds0-group-number	A value that identifies the DS0 group. Range is from 0 to 14 and 16 to 30; 15 is reserved.
timeslots *timeslot-list*	Lists time slots in the DS0 group. The *timeslot-list* argument is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Range is from 1 through 31. Examples are as follows: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12

type	Specifies the type of signaling for the DS0 group. The signaling method selection for the **type** keyword depends on the connection that you are making. The ear and mouth (E&M) interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The Foreign Exchange Station (FXS) interface allows connection of basic telephone equipment and PBX. The Foreign Exchange Office (FXO) interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are as follows: • **e&m-delay-dial**—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination. • **e&m-fgb**—E&M Type II Feature Group B. • **e&m-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start—**E&M immediate start. • **e&m-lmr**—E&M Land Mobile Radio (LMR). • **e&m-melcas-delay**—E&M MELCAS delay-start signaling support. • **e&m-melcas-immed—**E&M MELCAS immediate-start signaling support. • **e&m-melcas-wink**—E&M MELCAS wink-start signaling support. • **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a wink-start from the destination. • **fgd-eana**—**Feature Group D exchange access North American.** • **fgd-os—**Feature Group D operator services. • **fxo-ground-start—FXO ground-start signaling.** • **fxo-loop-start—FXO loop-start signaling.** • **fxo-melcas—FXO MELCAS signaling.** • **fxs-ground-start—FXS ground-start signaling.** • **fxs-loop-start—FXS loop-start signaling.** • **fxs-melcas—FXS MELCAS signaling.** • **none**—Null signaling for external call control. • **p7—**Specifies the p7 switch type.

	• **r1-itu**—Line signaling based on international signaling standards. • **r1-modified**—An international signaling standard that is common to channelized T1/E1 networks. • **r1-turkey**—A signaling standard used in Turkey. • **r2-analog—R2 analog line signaling.** • **r2-digital**—**R2 digital line signaling.** • **r2-lsv181-digital**—Specifies a specific R2 digital line. • **r2-pulse—7-pulse line signaling, a transmitted pulse that indicates a change in the line state.** • **sas-ground-start—**Single attachment station (SAS) ground-start. • **sas-loop-start**—SAS loop-start.
service *service-type*	(Optional) Specifies the type of service. • **data—**data service • **fax—** store-and-forward fax service • **voice—**voice service (for FGD-OS service) • **mgcp**—Media Gateway Control Protocol service

Defaults

There is no DS0 group. Calls are allowed in both directions.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as the **cas-group** command.
11.3(1)MA	The command was introduced as the **voice-group** command for the Cisco MC3810.
12.0(1)T	This command was integrated into Cisco IOS Release 12.0(1)T, and the **cas-group** command was implemented on the Cisco 3600 series routers.
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300 and Cisco 2600 series and Cisco 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(7)XK	Support for this command was implemented on the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported. The **ext-sig** keyword replaced the **ext-sig-master** and **ext-sig-slave** keywords that were available with the **voice-group** command.
12.0(7)XR	The **mgcp** service type was added.
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.

Release	Modification
12.1(5)XM	The **sgcp** keyword was removed.
12.1(3)T	This command was modified for Cisco 7500 series routers. The **fgd-os** signaling type and the **voice** service type were added.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.
12.2(2)XN	Support for the **mgcp** keyword was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was supported with Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command is supported on the Cisco IAD2420 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850 in this release.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The Cisco 1750 and Cisco 1751 do not support T1 and E1 voice and data cards in Cisco IOS Release 12.2(13)T. The Cisco 17xx platforms can support only HC DSP firmware images in this release.
12.2(15)T	This command was implemented on the Cisco 2600XM, Cisco 3725, and Cisco 3745.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(8)T	Documentation of the **ds0-group** command was divided into the individual **ds0-group (E1)** and **ds0-group (T1)** commands.
12.4(2)T1	Support was added for the **e&m-lmr** signaling type on the Cisco 2691, Cisco 2600XM series, Cisco 2800 series (except Cisco 2801), Cisco 3660, Cisco 3700 series, and Cisco 3800 series.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows:

- Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745, and Cisco 7200 series:
  – *slot*/*port*:*ds0-group-number*

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Be sure you take the following into account when you are configuring DS0 groups:

- Channel groups, CAS voice groups, DS0 groups, and time-division multiplexing (TDM) groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, DS0 groups, and TDM groups must be unique on the local router. For example, you cannot use the same group number for a channel group and for a TDM group.

- The keywords available for the **ds0-group** command are dependent upon the Cisco IOS software release that you are using. For the most current information, go to the Cisco Feature Navigator home page at the following URL:
  http://www.cisco.com/go/fn
- When you are using command-line interface (CLI) help, the keywords for the **ds0-group** command are configuration specific. For example, if Media Gateway Control Protocol (MGCP) is configured, you see the **mgcp** keyword. If you are not using MGCP, you do not see the **mgcp** keyword.

Examples

The following example shows ranges of E1 controller time slots configured for FXS ground-start and FXO loop-start signaling:

```
E1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-10 type fxs-ground-start
 ds0-group 2 timeslots 11-24 type fxo-loop-start
```

The following example shows ranges of T1 controller time slots configured for FXS ground-start signaling:

```
controller E1 1/0
 ds0-group 1 timeslots 1-4 type fxs-ground-start
```

The following example illustrates setting the E1 channels for Signaling System 7 (SS7) service on any trunking gateway using the **mgcp** keyword:

```
Router(config-controller)# ds0-group 0 timeslots 1-24 type none service mgcp
```

In the following example, the time slot maximum is 12 and the time slot is 1, so two voice-ports are created successfully.

```
controller E1 0/0
 ds0-group 0 timeslots 1-4 type e&m-immediate-start
 ds0-group 1 timeslots 6-12 type e&m-immediate-start
```

If a third DS0 group is added, the voice-port is rejected even though the total number of voice channels is less than 16.

```
 ds0-group 2 timeslots 17-18 type e&m-immediate-start
```

In the following example, the signaling type is set to e&m-lmr:

```
ds0-group 0 timeslots 1-10 type e&m-lmr
```

Related Commands

Command	Description
cas-group	Configures channelized T1 time slots with robbed bit signaling.
codec	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard that you are using.

ds0-group (T1)

To specify the DS0 time slots that make up a logical voice port on a T1 controller, to specify the signaling type by which the router communicates with the PBX or PSTN, and to define T1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

Cisco 1750 and Cisco 1751

ds0-group *ds0-group-number* **timeslots** *timeslot-list* [**service** *service-type*] **type** {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fgd-eana** | **fgd-os** | **fxs-ground-start** | **fxs-loop-start** | **none** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start**}

no ds0-group *ds0-group-number*

Cisco 2600 Series (Except Cisco 2691), Cisco 3600 Series (Except Cisco 3660), and Cisco VG 200

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

no ds0-group *ds0-group-number*

Cisco 2691, Cisco 2600XM Series, Cisco 2800 Series (Except Cisco 2801), Cisco 3660, Cisco 3700 Series, Cisco 3800 Series

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-lmr** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

no ds0-group *ds0-group-number*

Cisco 7200 Series and Cisco 7500 Series

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

no ds0-group *ds0-group-number*

Cisco 7700 Series Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-wink-start** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

no ds0-group *ds0-group-number*

Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850

ds0-group *ds0-group-number* **timeslots** *timeslot-list* [**service** *service-type*] [**type** [**e&m-fgb** [**dtmf** | **mf**] | **e&m-fgd** [**dtmf** | **mf** [**dnis** | **ani-dnis** [**info-digits-no-strip**] | **service** *service-type*] | **e&m-immediate-start** | **fxs-ground-start** | **fxs-loop-start** | **fgd-eana** [**ani-dnis** | **mf**] | **fgd-os** [**dnis-ani** | **mf**] | **r1-itu** [**dnis**] | **sas-ground-start** | **sas-loop-start** | **none**]]

no ds0-group *ds0-group-number*

Syntax Description

ds0-group-number	A value that identifies the DS0 group. Range is from 0 to 23.
timeslots *timeslot-list*	Lists time slots in the DS0 group. The *timeslot-list* argument is a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. Range is from 1 to 24. Examples are as follows: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12

type	Specifies the type of signaling for the DS0 group. The signaling method selection for the **type** keyword depends on the connection that you are making. The ear and mouth (E&M) interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The Foreign Exchange Station (FXS) interface allows connection of basic telephone equipment and PBX. The Foreign Exchange Office (FXO) interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are as follows: • **e&m-delay-dial**—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination. • **e&m-fgb**—E&M Type II Feature Group B. • **e&m-fgd**—E&M Type II Feature Group D. • **e&m-immediate-start—**E&M immediate start. • **e&m-lmr**—E&M Land Mobile Radio (LMR). • **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a wink-start from the destination. • **ext-sig—**The external signaling interface specifies that the signaling traffic comes from an outside source. • **fgd-eana—Feature Group D exchange access North American.** • **fgd-os—**Feature Group D operator services. • **fxo-ground-start**—**FXO ground-start signaling.** • **fxo-loop-start—FXO loop-start signaling.** • **fxs-ground-start—FXS ground-start signaling.** • **fxs-loop-start—FXS loop-start signaling.** • **none**—Null signaling for external call control. • **r1-itu**—Line signaling based on international signaling standards. • **r1-modified**—An international signaling standard that is common to channelized T1/E1 networks. • **r1-turkey**—A signaling standard used in Turkey. • **sas-ground-start—**Single attachment station (SAS) ground-start. • **sas-loop-start**—SAS loop-start.
service *service-type*	(Optional) Specifies the type of service. • **data—**Data service. • **fax**— Store-and-forward fax service. • **mgcp**[1]—Media Gateway Control Protocol service. • **sccp**[1]—Simple Gateway Control Protocol service • **voice**—Voice service (for FGD-OS service).
dtmf	(Optional) Specifies dual tone multifrequency (DTMF) tone signaling.
mf	(Optional) Specifies multifrequency (MF) tone signaling

ani-dnis	(Optional) Specifies automatic number identification (ANI) and dialed number identification service (DNIS) address information provisioning for FGD OS.
dnis-ani	(Optional) Specifies ANI and DNIS address information provisioning for FGD EANA.
dnis	(Optional) Specifies DNIS address information provisioning.
info-digits-no-strip	(Optional) Retains info digits on the Cisco AS5x00 platforms.

1. Used only with the **type none** keywords on the Cisco AS5x00 platforms.

Defaults

There is no DS0 group. Calls are allowed in both directions.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as the **cas-group** command.
11.3(1)MA	The command was introduced as the **voice-group** command for the Cisco MC3810.
12.0(1)T	This command was integrated into Cisco IOS Release 12.0(1)T, and the **cas-group** command was implemented on the Cisco 3600 series routers.
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300 and Cisco 2600 series and Cisco 3600 series routers. Some keyword modifications were implemented.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(7)XK	Support for this command was implemented on the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported. The **ext-sig** keyword replaced the **ext-sig-master** and **ext-sig-slave** keywords that were available with the **voice-group** command.
12.0(7)XR	The **mgcp** service type was added.
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.
12.1(5)XM	The **sgcp** keyword was removed.
12.1(3)T	This command was modified for Cisco 7500 series routers. The **fgd-os** signaling type and the **voice** service type were added.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included in this release.

Release	Modification
12.2(2)XN	Support for the **mgcp** keyword was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was supported in Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command is supported on the Cisco IAD2420 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850 in this release.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The Cisco 1750 and Cisco 1751 do not support T1 and E1 voice and data cards in Cisco IOS Release 12.2(13)T. The Cisco 17xx platforms can support only HC DSP firmware images in this release.
12.2(15)T	This command was implemented on the Cisco 2600XM, Cisco 3725, and Cisco 3745.
12.3(4)XD	This command was modified for the Cisco 3725 and Cisco 3745. The **e&m-lmr** signaling type was added.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(8)T	Documentation of the **ds0-group** command was divided into the individual **ds0-group (E1)** and **ds0-group (T1)** commands.
12.3(10)	The **info-digits-no-strip** keyword was added for use on the Cisco AS5x00 platforms.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows:

- Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, Cisco 3745, and Cisco 7200 series:
  - *slot/port***:***ds0-group-number*
- Cisco AS5300 with a T1 controller:
  - *slot/port*
- Cisco AS5850 with a T1 controller:
  - *slot/port***:***ds0-group-number*

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Be sure that you take the following into account when you are configuring DS0 groups:

- Channel groups, CAS voice groups, DS0 groups, and time-division multiplexing (TDM) groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, DS0 groups, and TDM groups must be unique on the local router. For example, you cannot use the same group number for a channel group and for a TDM group.

- The keywords available for the **ds0-group** command are dependent upon the Cisco IOS software release that you are using. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

  http://www.cisco.com/go/fn

- When you are using command-line interface (CLI) help, the keywords for the **ds0-group** command are configuration specific. For example, if Media Gateway Control Protocol (MGCP) is configured, you see the **mgcp** keyword. If you are not using MGCP, you do not see the **mgcp** keyword.

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5x00 series.

Examples

The following example shows ranges of T1 controller time slots configured for FXS ground-start and FXO loop-start signaling:

```
controller T1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-10 type fxs-ground-start
 ds0-group 2 timeslots 11-24 type fxo-loop-start
```

The following example shows ranges of T1 controller time slots configured for FXS ground-start signaling:

```
controller T1 1/0
 ds0-group 1 timeslots 1-4 type fxs-ground-start
```

The following example illustrates setting the T1 channels for Signaling System 7 (SS7) service on any trunking gateway using the **mgcp** keyword:

```
ds0-group 0 timeslots 1-24 type none service mgcp
```

In the following example, the time slot maximum is 12 and the time slot is 1, so two voice-ports are created successfully.

```
controller T1 0/0
 ds0-group 0 timeslots 1-4 type e&m-immediate-start
 ds0-group 1 timeslots 6-12 type e&m-immediate-start
```

If a third DS0 group is added, the voice port is rejected even though the total number of voice channels is less than 16.

```
 ds0-group 2 timeslots 17-18 type e&m-immediate-start
```

In the following example, the signaling type is set to E&M LMR:

```
 ds0-group 0 timeslots 1-10 type e&m-lmr
```

You have the option to retain info digits when you are configuring E&M Type II Feature Group D with MF signaling and ANI/DNIS for calls being sent over IP. Info digits denote the subscriber type, and the info-digits keyword prepends info digits to the calling number.

On inbound calls from a T1 FGD voice-port with MF ANI-DNIS, when ANI information is obtained, it is passed unaltered to the next matching dial peer, either POTS or VoIP. The addition of the **info-digits-no-strip** keyword allows you to retain the info digits portion of the ANI information; the modified ANI is then passed to the next matching dial peer. Ordinarily, info digits are not valid for calls going over IP and are, therefore, stripped off. The ability to retain info digits is particularly useful for calls that are not leaving the PSTN network and are just being hairpinned back.

In the following example, the E&M Type II Feature Group D is configured with MF signaling and ANI/DNIS over IP while retaining info digits:

```
ds0-group 0 timeslots 1-24 type e&m-fgd mf ani-dnis info-digits-no-strip
```

Related Commands

Command	Description
cas-group	Configures channelized T1 time slots with robbed bit signaling.
codec	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard that you are using.

inject guard-tone

To play out a guard tone with the voice packet, use the **inject guard-tone** command in voice-class configuration mode. To remove the guard tone, use the **no** form of this command.

inject guard-tone *frequency amplitude* [**idle**]

no inject guard-tone *frequency amplitude* [**idle**]

Syntax Description

frequency	Frequency, in Hz, of the tone to be injected. Range is integers from 1 to 4000.
amplitude	Amplitude, in dBm, of the tone to be injected. Range is integers from –50 to –3.
idle	(Optional) Play out the inverse of the guard tone when there are no voice packets. Idle tone and guard tone are mutually exclusive.

Defaults

No guard tone is injected.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **inject guard-tone** command has an effect on an ear and mouth (E&M) analog or digital voice port only if the signal type for that port is Land Mobile Radio (LMR). The guard tone is played out with the voice packet to keep the radio channel up. Guard tones of 1950 Hz and 2175 Hz can be filtered out before the voice packet is sent from the digital signal processor (DSP) to the network using the **digital-filter** command.

Examples

The following example configures a guard tone of 1950 Hz and –10 dBm to be played out with voice packets:

```
voice class tone-signal tone1
 inject guard-tone 2175 -30
```

Related Commands

Command	Description
digital-filter	Specifies the digital filter to be used before the voice packet is sent from the DSP to the network.

inject pause

To specify a pause between injected tones, use the **inject pause** command in voice-class configuration mode. To remove the pause, use the **no** form of this command.

inject pause *index milliseconds*

no inject pause *index milliseconds*

Syntax Description

index	Order of pauses and tones. Range is integers from 1 to 10.
milliseconds	Duration, in milliseconds, of the pause between injected tones. Range is integers from 10 to 500.

Defaults

milliseconds: 0 milliseconds

Command Modes

Voice-class configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **inject pause** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command to specify the pause between injected tones specified with the **inject tone** command. Use the *index* argument of this command in conjunction with the *index* argument of the inject tone command to specify the order of the pauses and tones.

Examples

The following example configures a pause of 100 milliseconds after the injected tone:

```
voice class tone-signal 100
 inject tone 1 2000 0 200
 inject pause 2 100
```

Related Commands

Command	Description
inject tone	Specifies a wakeup or frequency selection tone to be played out before the voice packet.

inject tone

To specify a wakeup or frequency selection tone to be played out before the voice packet, use the **inject tone** command in voice-class configuration mode. To remove the tone, use the **no** form of this command.

inject tone *index frequency amplitude duration*

no inject tone *index frequency amplitude duration*

Syntax Description

index	Order of pauses and tones. Range is integers from 1 to 10.
frequency	Frequency, in Hz, of the tone to be injected. Range is integers from 1 to 4000.
amplitude	Amplitude, in dBm, of the tone to be injected. Range is integers from –30 to 3.
duration	Duration, in milliseconds, of the tone to be injected. Range is integers from 10 to 500.

Defaults

No tone is injected.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **inject tone** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command with the **inject pause** command to configure wakeup and frequency selection tones. Use the *index* argument of this command in conjunction with the *index* argument of the **inject pause** command to specify the order of the pauses and tones.

If you configure injected tones with this command, be sure to use the **timing delay-voice tdm** command to configure a delay before the voice packet is played out. The delay must be equal to the sum of the durations of the injected tones and pauses in the tone-signal voice class.

Examples

The following example configures a frequency selection tone to be played out before the voice packet:

```
voice class tone-signal 100
 inject tone 1 1950 3 150
 inject tone 2 2000 0 60
 inject pause 3 60
```

```
inject tone 4 2175 3 150
inject tone 5 1000 0 50
```

Related Commands

Command	Description
inject pause	Specifies a pause between injected tones.
timing delay-voice tdm	Specifies the delay before a voice packet is played out.

input gain

To configure a specific input gain value or enable automatic gain control, use the **input gain** command in voice-port configuration mode. To disable the selected amount of inserted gain, use the **no** form of this command.

input gain {*decibels* | **auto-control** [*auto-dbm*]}

no input gain {*decibels* | **auto-control** [*auto-dbm*]}

Syntax Description

decibels	Gain, in decibels (dB), to be inserted at the receiver side of the interface. Range is integers from –27 to 16. The default is 0.
auto-control	Enable automatic gain control.
auto-dbm	(Optional) Target speech level, in decibels per milliwatt (dBm), to be achieved at the receiver side of the interface. Range is integers from –30 to 3. The default is –9.

Defaults

decibels: 0 decibels
auto-dbm: –9 dBm

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.3(4)XD	The range of values for the *decibels* argument was increased.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	The **auto-control** keyword and *auto-dbm* argument were added.

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. You must consider other equipment (including PBXs) in the system when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there is typically a minimum attenuation of –6 dB between phones, especially if echo cancellers are present. Connections are implemented to provide 0 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

You cannot increase the gain of a signal to the public switched telephone network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

Typical Land Mobile Radio (LMR) signaling systems send 0 dB out and expect –10 dB in. Setting output attenuation to 10 dB is typical. Output attenuation should be adjusted to provide the voice level required by the radio to produce correct transmitter modulation.

The **auto-control** keyword and *auto-dbm* argument are available on an ear and mouth (E&M) voice port only if the signal type for that port is LMR. The **auto-control** keyword enables automatic gain control, which is performed by the digital signal processor (DSP). Automatic gain control adjusts speech to a comfortable volume when it becomes too loud or too soft. Because of radio network loss and other environmental factors, the speech level arriving at a router from an LMR system could be very low. You can use automatic gain control to ensure that the speech is played back at a more comfortable level. Because the gain is inserted digitally, the background noise can also be amplified. Automatic gain control is implemented as follows:

- Output level: –9 dB
- Gain range: –12 dB to 20 dB
- Attack time (low to high): 30 milliseconds
- Attack time (high to low): 8 seconds

Examples

The following example inserts a 3-dB gain at the receiver side of the interface in the Cisco 3600 series router:

```
port 1/0/0
 input gain 3
```

Related Commands

Command	Description
output attenuation	Configures a specific output attenuation value or enables automatic gain control for a voice port.

lmr duplex half

To have the voice path for a voice port operate in half duplex mode, use the **lmr duplex half** command in voice-port configuration mode. To return to the default, use the **no** form of this command.

lmr duplex half

no lmr duplex half

Syntax Description

This command has no arguments or keywords.

Defaults

Full duplex mode

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

When a radio system is receiving voice traffic from the radio, operating the voice path in half duplex mode prevents the speaker from being interrupted and prevents the voice stream from being fed back to itself.

Examples

In the following example, the voice path for voice port 1/0/0 on a Cisco 3700 series router is set to operate in half duplex mode:

```
voice-port 1/0/0
 lmr duplex half
```

lmr e-lead

To define the use of the E-lead in signaling between the ear and mouth (E&M) voice port on the router and the attached Land Mobile Radio (LMR) device, use the **lmr e-lead** command in voice-port configuration mode. To return to the default use of the E-lead, use the **no** form of this command.

lmr e-lead {**inactive** | **seize** | **voice**}

no lmr e-lead {**inactive** | **seize** | **voice**}

Syntax Description

inactive	Specifies that the router never sends a seize signal on the E-lead to the LMR device. The router sends voice packets to LMR devices.
seize	Specifies that for PLAR and multicast connections, the router sends a seize signal on the E-lead when the LMR port is connected and removes the seize signal from the E-lead when the LMR port is not involved in a VoIP connection. This is the default. Specifies that for connection trunk connections, the router does not send a seize signal when the LMR port is connected. Instead, if the trunk connection is up, the M-lead signal from the far-end router is passed through as the E-lead on the near-end router. When the M-lead is dropped on the far-end router and the trunk connection is still up, the E-lead is dropped on the near-end router.
voice	Specifies that the router sends a seize signal on the E-lead only when it receives voice packets from the network. When no packets are detected on the network, the seize signal is removed from the E-lead.

Defaults

seize

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **lmr e-lead** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is LMR. The **lmr e-lead** command is effective only if the attached LMR device operates under E-lead control. Use the **lmr e-lead** command to configure the voice port when using private line, automatic ringdown (PLAR) connections. The E-lead connects to the Push To Talk (PTT) of the LMR system.

Examples

In the following example, packet transmission from the E&M voice port on a Cisco 3745 to an attached LMR radio system is disabled:

```
lmr e-lead inactive
```

Related Commands

Command	Description
lmr m-lead	Defines the use of the M-lead in signaling between the E&M voice port on the router and the attached LMR device.

lmr led-on

To use the ear and mouth (E&M) LED to indicate the E-lead and M-lead status, use the **lmr led-on** command in voice-port configuration mode. To return to the default use of the E&M LED, use the **no** form of this command.

lmr led-on

no lmr led-on

Syntax Description

This command has no arguments or keywords.

Defaults

The E&M LED indicates voice port activity only.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **lmr e-lead** command is available on an E&M voice port only if the signal type for that port is Land Mobile Radio (LMR). This command enables the use of the E&M LED to indicate the E-lead and M-lead status as follows:

- Red—E-lead active
- Green—M-lead active
- Yellow—Both E-lead and M-lead active

The default behavior of the E&M LED is to light up when there is activity on the voice port and to turn off when there is no activity.

Examples

The following example specifies that the E&M LED is used to indicate the E-lead and M-lead status:

```
voice-port 1/0/0
 lmr led-on
```

lmr m-lead

To define the use of the M-lead in signaling between the ear and mouth (E&M) voice port on the router and the attached Land Mobile Radio (LMR) device, use the **lmr m-lead** command in voice-port configuration mode. To return to the default use of the M-lead, use the **no** form of this command.

lmr m-lead {**inactive** | **audio-gate-in** | **dialin**}

no lmr m-lead {**inactive** | **audio-gate-in** | **dialin**}

Syntax Description

inactive	The router ignores signals sent by voice on the M-lead. The flow of voice packets is determined by voice activity detection (VAD). The router sends voice received from the LMR device. This is the default.
audio-gate-in	The router generates VoIP packets when a seize signal is detected on the M-lead. The router stops generating VoIP packets when the seize signal is removed from the M-lead.
dialin	When the LMR device is not involved in a VoIP connection, the first seize signal detected on the M-lead triggers the router to set up a VoIP connection. Once the connection is made, the router behaves as in the **audio-gate-in** option.

Defaults

inactive

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **lmr m-lead** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is LMR. The **lmr e-lead** command is effective only if the attached LMR device operates under M-lead control. The M-lead corresponds to the Carrier Operated Relay (COR) of the LMR system, which indicates receive activity on the LMR system.

Examples

In the following example, an LMR radio system attached to the E&M voice port on a Cisco 3745 is allowed to transmit audio by first raising the E-lead, then transmitting:

```
lmr m-lead dialin
```

Related Commands

Command	Description
lmr e-lead	Defines the use of the E-lead in signaling between the E&M voice port on the router and the attached LMR device.

music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** command in voice-port configuration mode. To disable this feature, use the **no** form of this command.

music-threshold *decibels*

no music-threshold *decibels*

Syntax Description

decibels	On-hold music threshold, in decibels (dB). Range is from –70 to –10 (integers only). The default is –38 dB.

Defaults

–38 dB

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(4)T	This command was implemented on the Cisco MC3810.
12.3(4)XD	The range of values for the *decibels* argument was increased.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

Use this command to specify the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. It affects the operation of voice activity detection (VAD) only when the voice port is receiving voice.

If the value for this command is set too high, VAD interprets music-on-hold as silence, and the remote end does not hear the music. If the value for this command is set too low, VAD compresses and passes silence when the background is noisy, creating unnecessary voice traffic.

Examples

The following example sets the decibel threshold to –35 for the music played when calls are put on hold:

```
voice port 0:D
 music-threshold -35
```

The following example sets the decibel threshold to –35 for the music played when calls are put on hold on a Cisco 3600 series router:

```
voice-port 1/0/0
 music-threshold -35
```

output attenuation

To configure a specific output attenuation value or enable automatic gain control, use the **output attenuation** command in voice-port configuration mode. To disable the selected output attenuation value, use the **no** form of this command.

output attenuation {*decibels* | **auto-control** [*auto-dbm*]}

no output attenuation {*decibels* | **auto-control** [*auto-dbm*]}

Syntax Description

decibels	Attenuation, in decibels (dB), at the transmit side of the interface. Range is integers from –27 to 16. The default is 0.
auto-control	Enable automatic gain control.
auto-dbm	(Optional) Target speech level, in decibels per milliwatt (dBm), to be achieved at the transmit side of the interface. Range is integers from –30 to 3. The default is –9.

Defaults

For Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and ear and mouth (E&M) ports: *decibels*: 0 decibels
auto-dbm: –9 dBm

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.3(4)XD	The range of values for the *decibels* argument was increased.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	The **auto-control** keyword and *auto-dbm* argument were added.

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. You must consider other equipment (including PBXs) in the system when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

You cannot increase the gain of a signal to the public switched telephone network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

The **auto-control** keyword and *auto-dbm* argument are available on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). The **auto-control** keyword enables automatic gain control, which is performed by the digital signal processor (DSP). Automatic gain control adjusts speech to a comfortable volume when it becomes too loud or too soft. Because of radio network loss and other environmental factors, the speech level arriving at a router from an LMR system could be very low. You can use automatic gain control to ensure that the speech is played back at a more comfortable level. Because the gain is inserted digitally, the background noise can also be amplified. Automatic gain control is implemented as follows:

- Output level: –9 dB
- Gain range: –12 dB to 20 dB
- Attack time (low to high): 30 milliseconds
- Attack time (high to low): 8 seconds

Examples

On the Cisco 3600 series router, the following example configures a 3-dB loss to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

On the Cisco 3600 series router, the following example configures a 3-dB gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation -3
```

On the Cisco AS5300, the following example configures a 3-dB loss to be inserted at the transmit side of the interface:

```
voice-port 0:D
 output attenuation 3
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
echo-cancel enable	Enables the cancellation of voice that is sent out the interface and received back on the same interface.
input gain	Configures a specific input gain value or enables automatic gain control for a voice port.

show voice lmr

To display the Land Mobile Radio (LMR) related dynamic information and static information for LMR ports or a ds0 group, use the **show voice lmr** command in EXEC mode.

show voice lmr [*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group*] [**details**]

Syntax Description

slot/subunit/port	(Optional) Voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1. The slash marks are required.
slot/port:*ds0-group*	(Optional) Voice port that you specify with the *slot*/*port*:*ds0-group* designation. • *slot* specifies a router slot in which the packet voice NM is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. T1 range is from 0 to 23. E1 range is from 0 to 30. The colon is required.
details	(Optional) Displays more information. If this keyword is omitted, less information is displayed.

Command Modes

EXEC

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command displays information for LMR voice ports only. If no voice port is specified, the command displays information for all E&M LMR voice ports.

When the **details** keyword is used, this command displays information about timeouts, timers, and injected tones and pauses, in addition to detailed voice port and active call information found in the **show voice port** and **show call active voice** commands.

Examples

The following is sample output from the **show voice lmr** command for an E&M LMR analog voice port on a Cisco 3745 router:

```
Router# show voice lmr 2/0/0

2/0/0
=========
 Connection type: n/a
 Out Attenuation = 0 db, In Gain = 0 dB
 E-lead capability is inactive, polarity = normal
 M-lead capability is inactive, polarity = normal
 voice-class tone-signal test
 state = LMR_CONNECT, e-lead = off, m-lead = off
 full duplex, voice path = rx
 Terminating side of the connection
 TransmitPackets=113, TransmitBytes=2241
 ReceivePackets=113, ReceiveBytes=2241
 CoderTypeRate=g729r8
 NoiseLevel=-65, ACOMLevel=22
 OutSignalLevel=-68, InSignalLevel=-79
 RemoteIPAddress=10.5.25.40, RemoteUDPPort=17272
 Remote SignallingIPAddress=10.5.25.40, Port=15418
 Remote MediaIPAddress=10.5.25.40, Port=17272
 RoundTripDelay=2 ms
 SessionProtocol=cisco
 VAD =enabled
```

The following is sample output from the **show voice lmr details** command for an E&M LMR analog voice port on a Cisco 3745 router:

```
Router# show voice lmr 2/0/0 details

2/0/0
=========
 Description:
 Connection type: n/a
 Out Attenuation = 0 db, In Gain = 0 dB
 Timing hangover: 500 ms
 E-lead capability is inactive, polarity = normal
 M-lead capability is inactive, polarity = normal
 Timing hookflash-in: 480
 Timing delay-voice: 470 ms
 Music On Hold Threshold: -38 dB, Noise Threshold: -62 dB
 E&M type: 1, Operation:  2-wire
 Impedance is set to 600r Ohm
 lmr tear down timeout is set to 1800 second
 lmr PTT transmit timeout is not set
 lmr PTT receive timeout is not set
 voice-class tone-signal test
         inject tone 1 1950 3 150
         inject tone 2 2000 0 60
         inject pause 3 60
         inject tone 4 2175 3 150
         inject tone 5 1000 0 50
         inject guard-tone 6 1950 -10
 state = LMR_CONNECT, e-lead = off, m-lead = off
 full duplex, voice path = rx
 Terminating side of the connection
 TransmitPackets=113, TransmitBytes=2241
 ReceivePackets=113, ReceiveBytes=2241
 CoderTypeRate=g729r8
 NoiseLevel=-66, ACOMLevel=22
 OutSignalLevel=-68, InSignalLevel=-79
```

```
PeerAddress=37200
PeerSubAddress=
PeerId=200
SessionTarget=

RemoteIPAddress=10.5.25.40, RemoteUDPPort=17272
Remote SignallingIPAddress=10.5.25.40, Port=15418
Remote MediaIPAddress=10.5.25.40, Port=17272
RoundTripDelay=0 ms
SessionProtocol=cisco
VAD =enabled
SelectedQoS=best-effort
ProtocolCallId=
SessionTarget=
```

Table 6 describes the significant fields shown in the output, in the order in which they appear.

Table 6 show voice lmr Field Descriptions

Field	Description
Connection type	Type of connection between LMR routers: private line, automatic ringdown (PLAR), trunk, or n/a
Out Attenuation	Output attenuation.
In Gain	Input gain.
E-lead capability	Active or inactive.
polarity	Polarity of the E&M voice port: normal or reverse.
M-lead capability	Active or inactive.
voice class tone-signal	Name of the tone-signal voice class.
state=	Signaling state.
e-lead =	On or off.
m-lead =	On or off.
full duplex	Voice path for the voice port is operating in full duplex mode.
half duplex	Voice path for the voice port is operating in half duplex mode.
voice path	Transmit or receive.
TransmitPackets	Number of packets sent by this peer during this call.
TransmitBytes	Number of bytes sent by this peer during this call.
ReceivePackets	Number of packets received by this peer during this call.
ReceiveBytes	Number of bytes received by the peer during this call.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice or fax compression to its associated call leg for this call.
NoiseLevel	Active noise level for this call.

Table 6 show voice lmr Field Descriptions (continued)

Field	Description
ACOMLevel	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
OutSignalLevel	Active output signal level to the telephony interface used by this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are sent.
Remote SignallingIPAddress, Port	Call control server IP address and signaling port number.
Remote MediaIPAddress, Port	Remote side media server IP address and RTP port number.
RoundTripDelay	Voice packet round trip delay between the local and remote systems on the IP backbone for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote routers through the IP backbone.
VAD	Whether voice activation detection (VAD) is enabled.
Description	Description of what the port is connected to.
Timing hangover	Number of milliseconds of delay before the digital signal processor (DSP) tells Cisco IOS software to turn off the E-lead after the DSP detects that the voice stream has stopped.
Timing hookflash-in	Maximum duration of a hookflash for a Foreign Exchange Station (FXS) interface.
Timing delay-voice	Delay before a voice packet is played out.
Music On Hold Threshold	Decibel level of music played when calls are put on hold.
Noise Threshold	Noise threshold for incoming calls.
E&M type	E&M signaling type.
Operation	2-wire or 4-wire operation.
Impedance	Terminating impedance of the interface.
lmr tear down timeout	Time for which the voice port waits before tearing down an LMR connection after detecting no voice activity.
lmr PTT transmit timeout	Maximum time for transmitting a voice packet.
lmr PTT receive timeout	Maximum time for receiving a voice packet.
inject pause	Pause injected before the voice packet is played out.

Table 6 show voice lmr Field Descriptions (continued)

Field	Description
inject tone	Tone injected before the voice packet is played out.
inject guard-tone	Guard tone played out with the voice packet.
PeerAddress	Destination pattern or number associated with this peer.
PeerSubAddress	Subaddress when this call is connected.
PeerId	ID value of the peer table entry to which this call was made.
SessionTarget	Network-specific address to receive calls from the dial peer.
SelectedQoS	Selected RSVP quality of service (QoS) for this call.
ProtocolCallId	Voice signaling specific call ID.

Related Commands

Command	Description
show call active voice	Displays call information for voice calls in progress.
show voice port	Displays configuration information about a specific voice port.

rtp payload-type

To identify the payload type of a Real-Time Transport Protocol (RTP) packet, use the **rtp payload-type** command in dial peer voice configuration mode. To remove the RTP payload type, use the **no** form of this command.

rtp payload-type {**cisco-cas-payload** *number* | **cisco-clear-channel** *number* | **cisco-codec-fax-ack** *number* | **cisco-codec-fax-ind** *number* | **cisco-codec-gsmamrnb** *number* | **cisco-codec-ilbc** *number* | **cisco-codec-video-h263+** *number* | **cisco-codec-video-h264** *number* | **cisco-fax-relay** *number* | **cisco-pcm-switch-over-alaw** *number* | **cisco-pcm-switch-over-ulaw** *number* | **cisco-rtp-dtmf-relay** *number* | **lmr-tone** *number* | **nse** *number* | **nte** *number* | **nte-tone** *number*} [**comfort-noise** {**13** | **19**}]

no rtp payload-type {**cisco-cas-payload** *number* | **cisco-clear-channel** *number* | **cisco-codec-fax-ack** *number* | **cisco-codec-fax-ind** *number* | **cisco-codec-gsmamrnb** *number* | **cisco-codec-ilbc** *number* | **cisco-codec-video-h263+** *number* | **cisco-codec-video-h264** *number* | **cisco-fax-relay** *number* | **cisco-pcm-switch-over-alaw** *number* | **cisco-pcm-switch-over-ulaw** *number* | **cisco-rtp-dtmf-relay** *number* | **lmr-tone** *number* | **nse** *number* | **nte** *number* | **nte-tone** *number*} [**comfort-noise** {**13** | **19**}]

Syntax Description

cisco-cas-payload *number*	Cisco channel-associated signaling (CAS) RTP payload. Range: 96 to 127. Default: 123.
cisco-clear-channel *number*	Cisco clear-channel RTP payload. Range: 96 to 127. Default: 125.
cisco-codec-fax-ack *number*	Cisco codec fax acknowledge. Range: 96 to 127. Default: 97.
cisco-codec-fax-ind *number*	Cisco codec fax indication. Range: 96 to 127. Default: 96.
cisco-codec-gsmamrnb *number*	Cisco Global System for Mobile Adaptive Multi-Rate Narrow Band (GSMAMR-NB) codec. Range is from 96 to 127. Default is 117.
cisco-codec-ilbc *number*	Cisco Internet Low Bit Rate Codec (iLBC) codec. Range: 96 to 127. Default: 116.
cisco-codec-video-h263+ *number*	RTP video codec H.263+ payload type. Range: 96 to 127. Default: 118.
cisco-codec-video-h264 *number*	RTP video codec H.264 payload type. Range: 96 to 127. Default: 119.
cisco-fax-relay *number*	Cisco fax relay. Range: 96 to 127. Default: 122.
cisco-pcm-switch-over-alaw *number*	Cisco RTP pulse code modulation (PCM) codec switch over indication (a-law). Default: 8.
cisco-pcm-switch-over-ulaw *number*	Cisco RTP PCM codec switchover indication (mu-law). Default: 0.
cisco-rtp-dtmf-relay *number*	Cisco RTP dual-tone multifrequency (DTMF) relay. Range: 96 to 127. Default: 121.
lmr-tone *number*	LMR payload type. Range: 96 to 127. Default: 0. The default value is set by the **no rtp payload-type lmr-tone** command.
nse *number*	A named signaling event (NSE). Range: 96 to 117. Default: 100.
nte *number*	A named telephone event (NTE). Range: 96 to 127. Default: 101.

nte-tone *number*	RFC-2833 tone payload type. Range 96 to 127. Default: 101.
comfort-noise {**13** | **19**}	(Optional) RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group, designates 13 as the payload type for comfort noise. If you are connecting to a gateway that complies with the *RTP Payload for Comfort Noise* draft, use 13. Use 19 only if you are connecting to older Cisco gateways that use DSPware before version 3.4.32. **Note** This command option is not available on the Cisco AS5400 running NextPort digital signal processors (DSPs). This command option is available on the Cisco AS5400 only if the platform has a high-density packet voice/fax feature card (AS5X-FC) with one or more AS5X-PVDM2-64 DSP modules installed. This support was added in Cisco IOS Release 12.4(4)XC, and integrated into Release 12.4(9)T and later 12.4T releases.

Command Default

No RTP payload type is configured.

Command Modes

Dial peer voice configuration

Command History

Release	Modification
12.2(2)T	This command was introduced.
12.2(2)XB	The **nte** and **comfort-noise** keywords were added.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	The **cisco-codec-gsmamrnb** keyword was added.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)T	The **cisco-codec-ilbc**, **cisco-codec-video-h263+**, and **cisco-codec-video-h264** keywords were added.
12.4(15)XY	The **lmr-tone** and **nte-tone** keywords were added.

Usage Guidelines

Use this command to identify the payload type of an RTP. Use this command after the **dtmf-relay** command is used to choose the NTE method of DTMF relay for a Session Initiation Protocol (SIP) call.

Configured payload types of NSE and NTE exclude certain values that have been previously hard-coded with Cisco-proprietary meanings. Do not use the following numbers, which have preassigned values: 96, 97, 100, 117, 121 to 123, and 125 to 127.

Use of any of these values results in an error message when the command is entered. You must first reassign the value in use to a different unassigned number; for example:

```
rtp payload-type cisco-codec-ilbc 100
ERROR: value 100 in use!
```

```
rtp payload-type nse 105
rtp payload-type cisco-codec-ilbc 100
```

Examples

The following example identifies the RTP payload type as GSMAMR-NB124:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-gsmamrnb 124
```

The following example identifies the RTP payload type as NTE 99:

```
Router(config-dial-peer)# rtp payload-type nte 99
```

The following example identifies the RTP payload type for the iLBC as 100:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-ilbc 100
```

Related Commands

Command	Description
dtmf-relay	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.

show voip rtp connections

To display Real-Time Transport Protocol (RTP) named event packets, use the **show voip rtp connections** command in privileged EXEC mode.

show voip rtp connections [**detail**]

Syntax Description

detail	(Optional) Displays the called-party and calling-party numbers associated with a call.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0	This command was introduced.
12.3(7)T	The **detail** keyword was added.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command displays information about RTP named event packets, such as caller ID number, IP address, and port for both the local and remote endpoints. The output from this command provides an overview of all the connections in the system, and this information can be used to narrow the criteria for debugging. The **debug voip rtp** command floods the console with voice packet information. You can use the **show voip rtp connections** command to get caller ID, remote IP address, or remote port identifiers that you can use to limit the output from the **debug voip rtp** command.

The **detail** keyword allows you to identify the phone or phones that have connected two RTP call legs together to create VoIP-to-VoIP or VoIP-to-POTS hairpins. If the **detail** keyword is omitted, the output does not display calls that are connected by hairpin call routing.

Examples

Table 7 describes the significant fields shown in the examples. Each line of output under “VoIP RTP active connections” shows information for one call leg. A phone call normally consists of two call legs, one connected to the calling party and one connected to the called party. The router joins (or bridges) the two call legs together to make a call. The **show voip rtp connections** command shows the RTP information for H.323 and SIP calls only; it does not directly show the POTS call legs. The information for the IP phone can be seen using the **show ephone offhook** command.

The following sample output shows an incoming H.323 call that is being directed to an IP phone attached to a Cisco CME system.

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP          RemoteIP
1   21      22         16996    18174  10.4.204.37      10.4.204.24
Found 1 active RTP connections
```

The following sample output shows the same call as in the previous example, but using the **detail** keyword with the command. The sample output shows the called number (1509) and calling number (8108) on both call legs (21 and 22); the called and calling numbers are the same on both legs for a simple A-to-B call. Leg 21 is the H.323 segment of the and leg 22 is the POTS segment that goes to the IP phone.

```
Router# show voip rtp connections detail

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP          RemoteIP
1   21      22         16996    18174  10.4.204.37      10.4.204.24
  callId 21 (dir=1):called=1509 calling=8108 redirect=
     dest callId 22:called=1509 calling=8108 redirect=
   1 context 64FB3358 xmitFunc 6032E8B4
Found 1 active RTP connections
```

The following example shows the call from the previous example being transferred by extension 1509 to extension 1514. Notice that the dstCallId changed from 22 to 24, but the original call leg (21) for the transferred party is still present. This implies that H.450.2 capability was disabled for this particular call, because if H.450.2 was being used for the transfer, the transfer would have caused the incoming H.323 call leg to be replaced with a new call.

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   21      24         16996    18174  10.4.204.37      10.4.204.24
Found 1 active RTP connections
```

The following example shows the detailed output for the same transfer as shown in the previous example. The original incoming call leg is still present (21) and still has the original called and calling numbers. The transferred call leg (24) shows 1509 (the transferring party) as the calling party and 1514 (the transfer destination) as the called party.

```
Router# show voip rtp connections detail

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP RmtRTP LocalIP         RemoteIP
1   21      24         16996    18174  10.4.204.37      10.4.204.24
  callId 21 (dir=1):called=1509 calling=8108 redirect=
     dest callId 24:called=1514 calling=1509 redirect=
   1 context 6466E810 xmitFunc 6032E8B4
Found 1 active RTP connections
```

The following sample output shows a cross-linked call with two H.323 call legs. The first line of output shows that the CallID for the first call leg is 7 and that this call leg is associated with another call leg that has a destination CallId of 8. The next line shows that the CallID for the leg is 8 and that it is associated with another call leg that has a destination CallId of 7. This cross-linkage between CallIds 7 and 8 shows that the first call leg is related to the second call leg (and vice versa). From this you can infer that the two call legs are actually part of the same phone call.

In an active system you can expect many lines of output that you would have to sort through to see which ones have this cross-linkage relationship. The lines showing two related call legs are not necessarily listed in adjacent order.

```
Router# show voip rtp connections

VoIP RTP active connections :
No. CallId   dstCallId         LocalRTP     RmtRTP         LocalIP        RemoteIP
1        7           8             16586      22346      172.27.82.2      172.29.82.2
2        8           7             17010      16590      172.27.82.2      192.168.1.29
Found 2 active RTP connections
```

Table 7 show voip rtp connections Field Descriptions

Field	Description
No.	Identifier of an RTP connection in this output.
CallId	Internal call identifier of a telephony call leg (RTP connection).
dstCallId	Internal call identifier of a VoIP call leg.
LocalRTP	RTP port of the media stream for the local entity.
RmtRTP	RTP port of the media stream for the remote entity.
LocalIP	IP address of the media stream for the local entity.
RemoteIP	IP address of the media stream for the remote entity.
dir	0 indicates an outgoing call. 1 indicates an incoming call.
called	Extension that received the call.
calling	Extension that made the call.
redirect	Original called number if the incoming call was forwarded.
context	Internal memory address for the control block associated with the call.
xmitFunc	Internal memory address for the transmit function to which incoming RTP packets (on the H.323 and SIP side) are sent; the address for the function that delivers the packets to the ephone.

Related Commands

Command	Description
debug voip rtp	Enables debugging for RTP named event packets.
show ephone offhook	Displays information and packet counts for phones that are currently off hook.

signal

To specify the type of signaling for a voice port, use the **signal** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

Foreign Exchange Office (FXO) and Foreign Exchange Station (FXS) Voice Ports

signal {**loop-start** | **ground-start**}

no signal {**loop-start** | **ground-start**}

Ear and mouth (E&M) Voice Ports

signal {**wink-start** | **immediate** | **delay-dial** | **lmr**}

no signal {**wink-start** | **immediate** | **delay-dial** | **lmr**}

Centralized Automatic Message Accounting (CAMA) Ports

signal {**cama** {**kp-0-nxx-xxxx-st** | **kp-0-npa-nxx-xxxx-st** | **kp-2-st** | **kp-npd-nxx-xxxx-st**} | **groundstart** | **loopstart**}

no signal {**cama** {**kp-0-nxx-xxxx-st** | **kp-0-npa-nxx-xxxx-st** | **kp-2-st** | **kp-npd-nxx-xxxx-st**} | **groundstart** | **loopstart**}

Syntax Description

loop-start	Specifies the use of loop start signaling. Used for FXO and FXS interfaces. With loop-start signaling, only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports. **Note** The CAMA version of this keyword is “**loopstart**.” Both forms operate identically.
cama	Selects and configures the port for 911 calls.
delay-dial	The calling side seizes the line by going off-hook on its E-lead. After a timing interval, the calling side looks at the supervision from the called side. If the supervision is on-hook, the calling side starts sending information as dual tone multifrequency (DTMF) digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information. Used for E&M tie trunk interfaces.
ground-start	Specifies the use of ground-start signaling. Used for FXO and FXS interfaces. Ground-start signaling allows both sides of a connection to place a call and to hang up. **Note** The CAMA version of this keyword is “**groundstart**.” Both forms operate identically.
immediate	The calling side seizes the line by going off-hook on its E-lead and sends address information as DTMF digits. Used for E&M tie trunk interfaces.
kp-0-npa-nxx-xxxx-st	10-digit transmission. The E.164 number is fully transmitted.
kp-0-nxx-xxxx-st	7-digit automatic number identification (ANI) transmission. The Numbering Plan Area (NPA) or area code is implied by the trunk group and is not transmitted.

kp-2-st	Default transmission when the CAMA trunk cannot get a corresponding Numbering Plan Digit (NPD) digit in the lookup table, or when the calling number is fewer than ten digits in length. (NPA digits are not available.)
kp-npd-nxx-xxxx-st	8-digit ANI transmission, where the NPD is a single multifrequency (MF) digit that is expanded into the NPA. The NPD table is preprogrammed in the sending and receiving equipment (on each end of the MF trunk); for example: 0 = 415, 1 = 510, 2 = 650, 3 = 916 05550100 = (415) 555-0100, 15550100 = (510) 555-0100, and so on. NPD range is from 0 to 3.
lmr	Specifies the use of Land Mobile Radio signaling.
wink-start	The calling side seizes the line by going off-hook on its E-lead then waits for a short off-hook "wink" indication on its M-lead from the called side before sending address information as DTMF digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.

Defaults

FXO and FXS interfaces: **loop-start**
E&M interfaces: **wink-start**
CAMA interfaces: **loop-start**

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.2(11)T	This command was modified to support ANI transmission.
12.3(4)XD	The **lmr** keyword was added.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command applies to analog voice ports only.

Using the **signal** command for an FXO or FXS voice port changes the signal value for both voice ports on a voice port module (VPM) card.

Note If you change the signal type for an FXO voice port on Cisco 3600 series routers, you need to move the appropriate jumper in the voice interface card of the voice network module. For more information about the physical characteristics of the voice network module, refer to the installation documentation, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

Configuring this command for an E&M voice port changes only the signal value for the selected voice port. In either case, the voice port must be shut down and then activated before the configured values take effect.

Some PBXs miss initial digits if the E&M voice port is configured for immediate start signaling. Immediate start signaling should be used for dial pulse outpulsing only and only on circuits for which the far end is configured to accept digits within a few milliseconds of seizure. Delay dial signaling, which is intended for use on trunks and not lines, relies on the far end to return an off-hook indication on its M-lead as soon as the circuit is seized. When a receiver is attached, the far end removes the off-hook indication to indicate that it is ready to receive digits. Delay dial must be configured on both ends to work properly. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

To specify which VIC-2CAMA ports are designated as dedicated CAMA ports for emergency 911 calls, use the **signal cama** command. No two service areas in the existing North American telephony infrastructure supporting E911 calls have identical service implementations, and many of the factors that drive the design of emergency call handling are matters of local policy and therefore outside the scope of this document. Local policy determines which ANI format is appropriate for the specified Physical Service Access Point (PSAP) location.

The following four types of ANI transmittal schemes are based on the actual number of digits transmitted toward the E911 tandem. In each instance, the actual calling number is proceeded with a key pulse (KP) followed by an information (I) field or a NPD, which is then followed by the ANI calling number, and finally is followed by a start pulse (ST), STP, ST2P, or ST3P, depending on the trunk group type in the PSTN and the traffic mix carried.

The information field is one or two digits, depending on how the circuit was ordered originally. For one-digit information fields, a value of 0 indicates that the calling number is available. A value of 1 indicates that the calling number is not available. A value of 2 indicates an ANI failure. For a complete list of values for two-digit information fields, refer to *SR-2275: Telcordia Notes on the Networks* at www.telcordia.com.

- 7-digit transmission (**kp-0-nxx-xxxx-st**):

  The calling phone number is transmitted, and the NPA is implied by the trunk group and not transmitted.

- 8-digit transmission (**KP-npd-nxx-xxxx-st**):

  The I field consists of single-digit NPD-to-NPA mapping. When the calling party number of 415-555-0122 places a 911 call, and the Cisco 2600 series or Cisco 3600 series has an NPD (0)-to-NPA (415) mapping, the NPA signaling format is received by the selective router at the central office (CO).

  Note NPD values greater than 3 are reserved for signifying error conditions.

- 10-digit transmission (**kp-0-npa-nxx-xxxx-st)**:

  The E.164 number is fully transmitted.

- kp–2–st transmission (**kp–2–st**):

  kp–2–st transmission is used if the PBX is unable to out-pulse the ANI. If the ANI received by the Cisco router is not as per configured values, kp–2–st is transmitted. For example, if the voice port is configured for out-pulsing a ten-digit ANI and the 911 call it receives has a seven-digit calling party number, the router transmits kp–2–st.

Note Emergency 911 calls are not rejected for an ANI mismatch. The call establishes a voice path. The E911 network, however, does not receive the ANI.

Examples

The following example configures ground-start signaling on the Cisco 3600 series as the signaling type for a voice port, which means that both sides of a connection can place a call and hang up:

```
voice-port 1/1/1
 signal ground-start
```

The following example configures a ten-digit ANI transmission:

```
Router(config)# voice-port 1/0/0
Router(config-voiceport)# signal cama kp-0-npa-nxx-xxxx-st
```

Related Commands

Command	Description
ani mapping	Preprograms the NPA, or area code, into a single MF digit.

signal keepalive

To configure the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks, use the **signal keepalive** command in voice-class configuration mode. To reset to the default, use the **no** form of this command.

signal keepalive {*seconds* | **disabled**}

no signal keepalive {*seconds* | **disabled**}

Syntax Description

seconds	Keepalive signaling packet interval, in seconds. Range is from 1 to 65535. Default is 5 seconds.
disabled	Specifies that no keepalive signals are sent.

Defaults

seconds: 5 seconds

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.3(7)T	The **disabled** keyword was added.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

Before configuring the keepalive signaling interval, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer using the **voice-class permanent** (dial-peer) command.

To avoid sending keepalive signals to a multicasting network with no specified destination, we recommend that you use the **disabled** keyword when configuring this command for use in networks that use connection trunk connections and multicasting.

Examples

The following example shows the keepalive signaling interval set to 3 seconds for voice class 10:

```
voice class permanent 10
 signal keepalive 3
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial-peer type.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
voice-class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

test lmr clear-call

To tear down a Land Mobile Radio (LMR) connection, use the **test lmr clear-call** command in privileged EXEC mode.

test lmr {*slot*/*subunit*/*port* | *slot*/*port*:*ds0-group*} **clear-call**

Syntax Description

slot/subunit/port	Voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1. The slash marks are required.
slot/port:*ds0-group*	Voice port that you specify with the *slot*/*port*:*ds0-group* designation. • *slot* specifies a router slot in which the packet voice NM is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. T1 range is from 0 to 23. E1 range is from 0 to 30. The colon is required.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

Because the LMR signaling protocol cannot terminate a call, the **test lmr clear-call** command can be used to tear down the call manually. This command tears down all LMR connections on the specified voice port.

Examples

In this example, all existing LMR connections on voice port 1/0/0 are torn down:

```
test lmr 1/0/0 clear-call
```

test lmr clear-call

To tear down a Land Mobile Radio (LMR) connection, use the **test lmr clear-call** command in privileged EXEC mode.

test lmr {*slot*/*subunit*/*port* | *slot*/*port***:***ds0-group*} **clear-call**

Syntax Description

slot/subunit/port	Voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1. The slash marks are required.
*slot/port***:***ds0-group*	Voice port that you specify with the *slot*/*port***:***ds0-group* designation. • *slot* specifies a router slot in which the packet voice NM is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. T1 range is from 0 to 23. E1 range is from 0 to 30. The colon is required.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

Because the LMR signaling protocol cannot terminate a call, the **test lmr clear-call** command can be used to tear down the call manually. This command tears down all LMR connections on the specified voice port.

Examples

In this example, all existing LMR connections on voice port 1/0/0 are torn down:

```
test lmr 1/0/0 clear-call
```

timeout ptt

To specify a maximum time for transmitting or receiving a voice packet, use the **timeout ptt** command in voice-port configuration mode. To return to the default, use the **no** form of this command.

timeout ptt {**rcv** | **xmt**} *minutes*

no timeout ptt {**rcv** | **xmt**} *minutes*

Syntax Description

rcv	Applies the specified time limit to the reception of voice packets.
xmt	Applies the specified time limit to the transmission of voice packets.
minutes	Maximum time, in minutes, allowed for transmitting or receiving a voice packet. Range is integers from 1 to 30.

Defaults

minutes: 0 minutes

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **timeout ptt** command is available on an ear and mouth (E&M) analog or digital voice port only if the signal type for that port is Land Mobile Radio (LMR). The purpose of this command is to limit extended radio transmission. When the time limit configured with this command expires, the radio transmitter unkeys, so that listeners on the channel cannot hear the speaker, even if the speaker continues to talk. When the speaker unkeys the radio, the timer is reactivated.

Examples

The following example specifies a maximum time of 10 minutes for transmitting a voice packet:

```
voice-port 1/0/0
 timeout ptt xmt 10
```

timeouts teardown lmr

To configure the time for which a Land Mobile Radio (LMR) voice port waits before tearing down an LMR connection after detecting no voice activity, use the **timeouts teardown lmr** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

timeouts teardown lmr {*seconds* | **infinity**}

no timeouts teardown lmr {*seconds* | **infinity**}

Syntax Description

seconds	Duration in seconds for which an LMR voice port waits before tearing down an LMR connection after detecting no voice activity. Valid values are 5 to 60000. The default is 180 seconds.
infinity	Disables disconnect supervision. The voice port does not disconnect when no voice activity is detected.

Defaults

180 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **timeouts teardown lmr** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is LMR.

Examples

The following example configures voice port 1/0/1 on a Cisco 3745 to remain connected for 6 seconds after no voice activity is detected by the voice port:

```
voice-port 1/0/1
 timeouts teardown lmr 6
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

Command	Description
timeouts wait-release	Specifies the delay time for releasing the calling voice port after a disconnect tone is received from the called voice port.
timeouts delay-duration	Configures the delay dial signal duration for a specified voice port.

timing delay-voice tdm

To specify the delay after which voice packets are played out, use the **timing delay-voice tdm** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

timing delay-voice tdm *milliseconds*

no timing delay-voice tdm *milliseconds*

Syntax Description

milliseconds	Duration, in milliseconds, of the timing delay. Range is integers from 1 to 1500. Default is 0.

Defaults

milliseconds: 0 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **timing delay-voice tdm** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). To avoid voice loss at the receiving end of an LMR system, use this command to configure a delay for the voice packet equal to the sum of the durations of all the injected tones and pauses configured with the **inject tone** command and the **inject pause** command.

Examples

The following example configures a timing delay of 470 milliseconds before the voice packet is played out:

```
voice class tone-signal mytones
 inject tone 1 1950 3 150
 inject tone 2 2000 0 60
 inject pause 3 60
 inject tone 4 2175 3 150
 inject tone 5 1000 0 50
voice-port 1/0/0
 voice-class tone-signal mytones
 timing delay-voice tdm 470
```

Note that the delay of 470 milliseconds is equal to the sum of the durations of the injected tones and pauses in the tone-signal voice class.

Related Commands

Command	Description
inject pause	Specifies a pause between injected tones.
inject tone	Specifies a wakeup or frequency selection tone to be played out before the voice packet.

timing hangover

To specify the number of milliseconds of delay before the digital signal processor (DSP) tells Cisco IOS software to turn off the E-lead after the DSP detects that the voice stream has stopped, use the **timing hangover** command in voice-port configuration mode. To return to the default value, use the **no** form of this command.

timing hangover *milliseconds*

no timing hangover *milliseconds*

Syntax Description

milliseconds	The number of milliseconds for which the E-lead stays active after VAD determines that the voice stream has stopped. Valid values are 0 to 10000. The default is 250 milliseconds.

Defaults

milliseconds: 250 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **timing hangover** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). If the voice port has been configured with the **lmr e-lead voice** command, use the **timing hangover** command to adjust the timing if the E-lead is being turned on and off too frequently.

Examples

The following example configures E-lead on voice port 1/0/1 on a Cisco 3745 to stay active for 300 milliseconds after VAD determines that the voice stream has stopped:

```
voice-port 1/0/1
 timing hangover 300
```

timing hookflash-input

To specify the maximum duration of an on-hook condition that will be interpreted as a hookflash by the Cisco IOS software, use the **timing hookflash-input** command in voice-port configuration mode. To restore the default duration for hookflash timing, use the **no** form of this command.

timing hookflash-input *milliseconds*

no timing hookflash-input

Syntax Description

milliseconds	Upper limit of the hookflash duration range, in milliseconds. • E&M voice ports—Range is 0 to 1550 milliseconds. Default is 480 milliseconds. • FXS voice ports—Range is 50 to 1550 milliseconds. Default is 1000 milliseconds.

Defaults

milliseconds: 480 milliseconds for E&M voice ports, 1000 milliseconds for FXS voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 3600 series.
12.3(7)T	Lower limit of the range for E&M voice ports was extended to 0 milliseconds.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command is applied to E&M or Foreign Exchange Station (FXS) interfaces.

For Land Mobile Radio E&M voice ports, the **timing hookflash-input** command configures the delay between when the M-lead is raised and when voice is transmitted. Setting the hookflash duration to 0 milliseconds specifies no delay in the audio input and eliminates front-end clipping.

Analog phones connected to FXS ports use hookflash to access a second dial tone to initiate some phone features, such as transfer and conference. Hookflash is an on-hook condition of short duration that is usually generated when a phone user presses the Flash button on a phone. Cisco voice gateways measure the duration of detected on-hook conditions to determine whether they should be interpreted as hookflash or not. The duration for the on-hook conditions generated by Flash buttons on phones varies for different phone types and is interpreted by Cisco IOS software as follows:

- An on-hook condition that lasts for a time period that falls inside the hookflash duration range is considered a hookflash.
- An on-hook condition that lasts for a shorter period than the lower limit of the range is ignored.
- An on-hook condition that lasts for a longer period than the higher limit of the range is considered a disconnect.

The hookflash duration range for FXS voice ports is defined as follows:

- The lower limit of the range is set in software at 150 ms, although there is also a hardware-imposed lower limit that is typically about 20 ms, depending on platform type. An on-hook condition that lasts for a shorter time than this hardware-imposed lower limit is simply not reported to the Cisco IOS software.
- The upper limit of the range is set in software at 1000 ms by default, although this value can be changed using the **timing hookflash-input** command in voice-port configuration mode on the voice gateway. The upper limit can be set to any value from 50 to 1550 ms. For more information, see the explanations in the “Examples” section.

This command does *not* affect whether hookflash relay is enabled; hookflash relay is enabled only when the **dtmf-relay h245-signal** command is configured on the applicable VoIP dial peers. When the **dtmf-relay h245-signal** command is configured, the H.323 gateway relays hookflash by using an H.245 “signal” User Input Indication method. Hookflash is sent only when an H.245 signal is available.

Examples

The following example sets an upper limit of 200 milliseconds for the hookflash duration range:

```
voice-port 1/0/0
 timing hookflash-input 200
```

If the **timing hookflash-input** command is set to X, a value greater than 150, then any on-hook duration between 150 and X is interpreted as a hookflash. For example, if X is 1550, the hookflash duration range is 150 to 1550 ms. An on-hook signal that lasts for 1250 ms is interpreted as a hookflash, but an on-hook signal of 55 ms is ignored.

```
voice-port 1/0/0
 timing hookflash-input 1550
```

If the **timing hookflash-input** command is set to X, a value less than 150, then any on-hook duration between Y, the hardware lower limit, and X is interpreted as a hookflash. For example, if X is 65, the hookflash duration range is Y to 65 ms. An on-hook signal that lasts for 1250 ms is interpreted as a disconnect, but an on-hook signal of 55 ms is interpreted as a hookflash. (This example assumes that Y for the voice gateway is lower than 55 ms.)

```
voice-port 1/0/0
 timing hookflash-input 65
```

Related Commands

Command	Description
dtmf-relay (Voice over IP)	Specifies how an H.323 gateway relays DTMF tones between telephony interfaces and an IP network.

timing ignore m-lead

To ignore M-lead or voice activity detection (VAD) changes for a specified amount of time after sending the E-lead off signal, use the **timing ignore m-lead** command in voice-port configuration mode. To return to the default value, use the **no** form of this command.

timing ignore m-lead *milliseconds*

no timing ignore m-lead *milliseconds*

Syntax Description

milliseconds	The number of milliseconds following the sending of the E-lead off signal for which the M-lead and VAD changes are ignored. Valid values are 0 to 10000. The default is 0 milliseconds.

Defaults

milliseconds: 0 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

- The **timing ignore m-lead** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command to reduce echo feedback on an LMR voice port. This command has an effect only if the voice port is configured for half duplex mode.

Examples

The following example configures voice port 1/0/1 on a Cisco 3745 to ignore M-lead or VAD changes for 500 milliseconds after sending the E-lead off signal:

```
voice-port 1/0/1
 timing ignore m-lead 500
```

voice class tone-signal

To enter voice-class configuration mode and create a tone-signal voice class, use the **voice class tone-signal** command in global configuration mode. To delete a tone-signal voice class, use the **no** form of this command.

voice class tone-signal *tag*

no voice class tone-signal *tag*

Syntax Description

tag	Label that uniquely identifies the voice class. Can be up to 32 alphanumeric characters.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

Use the **voice class tone-signal** command to define wakeup, frequency selection, and guard tones to be played out before and during the voice packets for a specific voice port. Use the **inject guard-tone**, **inject pause**, and **inject tone** commands to define the tone signaling in this class. You can configure up to ten tones in a tone-signal voice class.

To avoid voice loss at the receiving end of an LMR system, the maximum of the sum of the durations of the injected tones and pauses in the voice class should not exceed 1500 milliseconds. You must also use the **timing delay-voice tdm** command to configure a delay for the voice packet equal to the sum of the durations of all the injected tones and pauses.

Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice-class tone-signal**, which is used in voice-port configuration mode.

Examples

The following example shows how to create a tone-signal voice class starting from global configuration mode:

```
voice class tone-signal mytones
 inject tone 1 1950 3 150
 inject tone 2 2000 0 60
 inject pause 3 60
 inject tone 4 2175 3 150
```

```
inject tone 5 1000 0 50
```

Related Commands

Command	Description
inject guard-tone	Plays out a guard tone with the voice packet.
inject pause	Specifies a pause between injected tones.
inject tone	Specifies a wakeup or frequency selection tone to be played out before the voice packet.
timing delay-voice tdm	Specifies the delay before a voice packet is played out.
voice-class tone-signal	Assigns a previously configured tone-signal voice class to a voice port.

voice-class tone-signal

To assign a previously configured tone-signal voice class to a voice port, use the **voice-class tone-signal** command in voice-port configuration mode. To delete a tone-signal voice class, use the **no** form of this command.

voice-class tone-signal *tag*

no voice-class tone-signal *tag*

Syntax Description

tag	Unique label assigned to the voice class. The *tag* label maps to the tag label created using the **voice class tone-signal** global configuration command. Can be up to 32 alphanumeric characters.

Defaults

Voice ports have no tone-signal voice class assigned.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **voice-class tone-signal** command is available on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice class tone-signal**, which is used in global configuration mode.

Examples

The following example assigns a previously configured voice class to voice port 1/1/0:

```
voice-port 1/0/0
 voice-class tone-signal mytones
```

Related Commands

Command	Description
voice class tone-signal	Enters voice-class configuration mode and assigns an identification tag number for a tone-signal voice class.

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Glossary

Applique—Any hardware unit that provides the external interface connections from a router to the network.

COR—Carrier Operated Relay. A signal from a receiver that indicates that the receiver is receiving a signal or carrier and that the receiver is not squelched.

PTT—Push-to-talk. A signal to a radio transmitter that causes the transmission of radio frequency energy.

RTP—Real-Time Transport Protocol. Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time-stamping, and delivery monitoring to real-time applications.

Squelch—An electric circuit that stops input to a radio receiver when the signal being received is too weak to be anything but noise.

Tone control—The process of sending an in-band tone (2175 Hz) with voice transmission to control receiving radios remotely. In-band tones can be used to control functions such as frequency selection and channel monitoring also. For example, a 1950 Hz tone can be used to select frequency 1, a 1850 Hz tone to select second frequency, or a 2050 Hz tone to activate channel monitor on radio. All of these tones are 40 msec in duration.

VAD—voice activity detection. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

VOX—Voice operated transmit. A keying relay that is actuated by sound or voice energy above a certain threshold sensed by a connected acoustoelectric transducer. VOX uses voice energy to key a transmitter, eliminating the need for push-to-talk operation.

Note Refer to *Internetworking Terms and Acronyms* for terms not included in this glossary.

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