Configuring Voice over IP for Cisco MC3810 Series Concentrators

Feature Summary

Voice over IP (VoIP) enables a Cisco MC3810 concentrator to carry voice traffic (for example, telephone calls and faxes) over an IP network. Voice over IP is primarily a software feature; however, to support this feature, a Cisco MC3810 must be equipped with a digital voice module (DVM) or an analog voice module (AVM). The Cisco MC3810's LAN/WAN multiservice routing capabilities provide analog and digital (T1/E1) VoIP gateway capabilities for packetized voice traffic.

In Voice over IP, the DSP segments the voice signal into frames, which are then coupled in groups of two and stored in voice packets. These voice packets are transported using IP in compliance with ITU-T specification H.323. Because it is a delay-sensitive application, you need to have a well-engineered network end-to-end to successfully use Voice over IP. Fine-tuning your network to adequately support Voice over IP involves a series of protocols and features geared toward quality of service (QoS). Traffic shaping considerations must be taken into account to ensure the reliability of the voice connection.

Benefits

Voice over IP offers the following benefits:

- Toll bypass
- Remote PBX presence over WANs
- Unified voice/data trunking
- POTS-Internet telephony gateways
- Interoperability with third-party H.323 applications and devices
- Integration as a VoIP gateway for Cisco AVVID solutions

Related Documents

- *Cisco MC3810 Series Multiservice Access Concentrators Hardware Installation Guide*
- *Cisco IOS 12.0 Voice, Video, and Home Applications Configuration Guide*
- *Voice Port Enhancements in Cisco 2600, 3600, MC3810 Routers and Concentrators*, Cisco IOS Release 12.0(7)XK online document
- *QSIG Protocol Support on Cisco 3810, 7200, 2600, and 3600 Series Routers*, Cisco IOS Release 12.0(7)XK online document

- *Transparent CCS and Frame Forwarding Enhancments on the Cisco MC3810*, Cisco IOS Release 12.0(7)XK online document
- *Voice Port Enhancements on Cisco 2600 and 3600 Series Routers and MC3810 Concentrators*, Cisco IOS Release 12.0(7)XK online document

Supported Platform

- Cisco MC3810 series concentrators

Supported Standards, MIBs, and RFCs

This feature supports the following standards and RFCs:

- ITU-T H.323v2—*Packet-Based Multimedia Communications Systems*, February 1998
- ITU-T Q.400-490 series—*Signalling System R2*, 1988 to 1993
- RFC 1889—*RTP: A Transport Protocol for Real-Time Applications*, January 1996; H. Schulzrinne, GMD Fokus; S. Casner, Precept Software, Inc; R. Frederick, Xerox Palo Alto Research Centre; V. Jacobson, Lawrence Berkeley National Laboratory
- RFC 1890—*RTP Profile for Audio and Video Conferences with Minimal Control*, January 1996; H. Schulzrinne, GMD Fokus
- RFC 2127—*ISDN Management Information Base using SMIv2*, March 1997; G. Roeck, Editor; Cisco Systems
- RFC 2128—*Dial Control Management Information Base using SMIv2*, March 1997; G. Roeck, Editor; Cisco Systems

Prerequisites

The voice enhancements described in this document require the use of Cisco IOS Release 12.0(7)XK or newer.

Configuration Tasks

To configure Voice over IP on the Cisco MC3810 concentrator, you need to complete the following tasks:

1. Preparing to Configure VoIP
2. Configuring IP Networks for Real-Time Voice Traffic

   Configure your IP network to support real-time voice traffic. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward quality of service (QoS). To configure your IP network for real-time voice traffic, you need to take into consideration the entire scope of your network, then select and configure the appropriate QoS tool or tools:

   (a) Configuring Multilink PPP with Interleaving

   (b) Configuring RTP Header Compression

(c) Configuring IP RTP Priority

Refer to the "Configuring IP Networks for Real-Time Voice Traffic" section for information about how to select and configure the appropriate QoS tools to optimize voice traffic on your network.

3 Configuring Number Expansion

Use the **num-exp** command to configure number expansion if your telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Refer to the "Configuring Number Expansion" section for information about number expansion.

4 Configuring Dial Peers

Use the **dial-peer voice** command to define dial peers and switch to the dial-peer configuration mode. Each dial peer defines the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. An end-to-end call is comprised of four call legs, two from the perspective of the source access server, and two from the perspective of the destination access server. Dial peers are used to apply attributes to call legs and to identify call origin and destination. There are two different kinds of dial peers:

(a) POTS—Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device. To minimally configure a POTS dial peer, you need to configure the following two characteristics: associated telephone number and logical interface. Use the **destination-pattern** command to associate a telephone number with a POTS peer. Use the **port** command to associate a specific logical interface with a POTS peer. In addition, you can specify direct inward dialing for a POTS peer by using the **direct-inward-dial** command.

(b) VoIP—Dial peer describing the characteristics of a packet network connection; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices. To minimally configure a VoIP peer, you need to configure the following two characteristics: associated destination telephone number and a destination IP address. Use the **destination-pattern** command to define the destination telephone number associated with a VoIP peer. Use the **session target** command to specify a destination IP address for a VoIP peer.

Refer to the "Configuring Dial Peers" section for additional information about configuring dial peers and dial-peer characteristics.

5 Optimizing Dial Peer and Network Interface Configurations

You can use VoIP peers to define characteristics such as IP precedence, CODEC, and VAD. Use the **ip precedence** command to define IP precedence. Use the **codec** command to configure specific voice coder rates. Use the **vad** command to disable voice activation detection and the transmission of silence packets. Refer to the "Optimizing Dial Peer and Network Interface Configurations" section for additional information about optimizing dial-peer characteristics.

6 Configuring Voice Ports

You need to configure your Cisco MC3810 concentrator to support voice ports. In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. Voice ports on the Cisco MC3810 concentrator support three basic voice signaling types:

(a) FXO—Foreign Exchange Office interface

(b) FXS—The Foreign Exchange Station interface

(c) E&M—The "Ear and Mouth" interface (or "RecEive and TransMit" interface)

Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network.

7 Configuring the H.323 Gateway

The gateway capability allows a Cisco MC3810 to function as an H.323 endpoint. Therefore, the gateway provides admission control, and address lookup and translation.

Preparing to Configure VoIP

Before you can configure your Cisco MC3810 concentrator to use Voice over IP, you must first:

- Establish a working IP network. For more information about configuring IP, refer to the "IP Overview," "Configuring IP Addressing," and "Configuring IP Services" chapters in the *Cisco IOS 12.0 Network Protocols Configuration Guide, Part 1*.
- Install a digital voice module (DVM) or an analog voice module (AVM) into the appropriate bays of your Cisco MC3810 concentrator. For more information about the physical characteristics of the voice modules, or how to install them, refer to the *Cisco MC3810 Series Multiservice Access Concentrators Hardware Installation Guide* which came with your Cisco MC3810 concentrator.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.
- Integrate your dial plan and telephony network into your existing IP network topology. Merging your IP and telephony networks depends on your particular IP and telephony network topology. In general, Cisco recommends the following suggestions:
    - — Use canonical numbers wherever possible. It is important to avoid situations where numbering systems are significantly different on different routers or access servers in your network.
    - — Make routing and/or dialing transparent to the user—for example, avoid secondary dial tones from secondary switches, where possible.
    - — Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

After you have analyzed your dial plan and decided how to integrate it into your existing IP network, you are ready to configure your network devices to support Voice over IP.

Configuring IP Networks for Real-Time Voice Traffic

You need to have a well-engineered network end-to-end when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward quality of service (QoS). It is beyond the scope of this document to explain the specific details relating to wide-scale QoS deployment. Cisco IOS software provides many tools for enabling QoS on your backbone, such as Random Early Detection (RED), Weighted Random Early Detection (WRED), Fancy queuing (meaning custom, priority, or weighted fair queuing), and IP Precedence. To configure your IP network for real-time voice traffic, you need to take into consideration the entire scope of your network, then select the appropriate QoS tool or tools.

The important thing to remember is that QoS must be configured throughout your network—not just on the Cisco MC3810 concentrator devices running VoIP—to improve voice network performance. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations; the QoS tasks they perform might differ as well. To configure your IP network for real-time voice traffic, you need to take into consideration the functions of both edge and backbone routers in your network, then select the appropriate QoS tool or tools.

In general, edge routers perform the following QoS functions:

- Packet classification
- Admission control
- Bandwidth management
- Queuing

In general, backbone routers perform the following QoS functions:

- High-speed switching and transport
- Congestion management
- Queue management

Scalable QoS solutions require cooperative edge and backbone functions.

Although not mandatory, some QoS tools have been identified as being valuable in fine-tuning your network to support real-time voice traffic. To configure your IP network for QoS using these tools, perform one or more of the following tasks:

- Configuring Multilink PPP with Interleaving
- Configuring RTP Header Compression
- Configuring IP RTP Priority

Each of these components is discussed in the following sections.

Configuring Multilink PPP with Interleaving

Multiclass Multilink PPP Interleaving allows large packets to be multilink-encapsulated and fragmented into smaller packets to satisfy the delay requirements of real-time voice traffic; small real-time packets, which are not multilink-encapsulated, are transmitted between fragments of the large packets. The interleaving feature also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be transmitted earlier than other flows. Interleaving provides the delay bounds for delay-sensitive voice packets on a slow link that is used for other best-effort traffic.

Note Interleaving applies only to interfaces that can configure a multilink bundle interface. These include virtual templates, dialer interfaces, and Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) or Primary Rate Interface (PRI) interfaces.

In general, Multilink PPP with interleaving is used in conjunction with weighted fair queuing or IP Precedence to ensure voice packet delivery. Use Multilink PPP with interleaving and weighted fair queuing to define how data will be managed; use IP Precedence to give priority to voice packets.

You should configure Multilink PPP if the following conditions exist in your network:

- Point-to-point connection using PPP Encapsulation
- Slow links

Note Multilink PPP should not be used on links greater than 2 Mbps.

Multilink PPP support for interleaving can be configured on virtual templates, dialer interfaces, and ISDN BRI or PRI interfaces. To configure interleaving, you need to complete the following tasks:

- Configure the dialer interface or virtual template, as defined in the relevant chapters of the *Cisco IOS 12.0 Dial Solutions Configuration Guide*.
- Configure Multilink PPP and interleaving on the interface or template.

To configure Multilink PPP and interleaving on a configured and operational interface or virtual interface template, use the following commands in interface mode:

Step	Command	Purpose
1	`router(config-if)#` **`ppp multilink`**	Enable Multilink PPP.
2	`router(config-if)#` **`ppp multilink interleave`**	Enable real-time packet interleaving.
3	`router(config-if)#` **`ppp multilink fragment-delay`** *`milliseconds`*	Optionally, configure a maximum fragment delay.
4	`router(config-if)#` **`ip rtp priority`** *`starting-rtp-port-number port-number-range bandwidth`*	Reserve a strict priority queue for a set of RTP packet flows belonging to a range of UDP destination ports

For more information about Multilink PPP, refer to the “Configuring Media-Independent PPP and Multilink PPP” chapter in the *Dial Solutions Configuration Guide*.

Multilink PPP Configuration Example

The following example defines a virtual interface template that enables Multilink PPP with interleaving and a maximum real-time traffic delay of 20 milliseconds, and then applies that virtual template to the Multilink PPP bundle:

```
interface virtual-template 1
 ppp multilink
 encapsulated ppp
 ppp multilink interleave
 ppp multilink fragment-delay 20
 ip rtp priority 16384 16383 25

multilink virtual-template 1
```

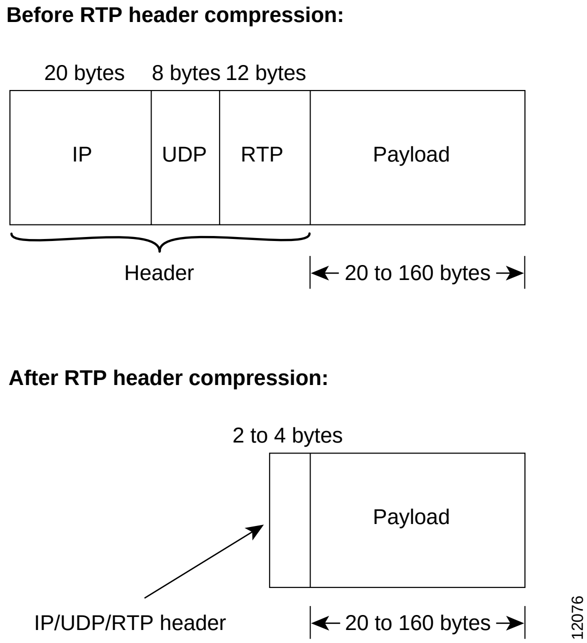
Configuring RTP Header Compression

Real-Time Transport Protocol (RTP) is used for carrying packetized audio traffic over an IP network. RTP header compression compresses the IP/UDP/RTP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes (most of the time), as shown in Figure 1.

This compression feature is beneficial if you are running Voice over IP over slow links. Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if there is a lot of RTP traffic on that slow link.

Typically, an RTP packet has a payload of approximately 20 to 160 bytes for audio applications that use compressed payloads. RTP header compression is especially beneficial when the RTP payload size is small (for example, compressed audio payloads between 20 and 50 bytes).

Figure 1 RTP Header Compression



You should configure RTP header compression if the following conditions exist in your network:

- Slow links
- Need to save bandwidth

Note RTP header compression should not be used on links greater than 2 Mbps.

Perform the following tasks to configure RTP header compression for Voice over IP. The first task is required; the second task is optional.

- Enable RTP Header Compression on a Serial Interface
- Change the Number of Header Compression Connections

Enable RTP Header Compression on a Serial Interface

To use RTP header compression, you need to enable compression on both ends of a serial connection. To enable RTP header compression, use the following command in interface configuration mode:

Command	Purpose
`router(config-if)#` **`ip rtp header-compression`** [**`passive`**]	Enable RTP header compression.

If you include the **passive** keyword, the software compresses outgoing RTP packets only if incoming RTP packets on the same interface are compressed. If you use the command without the **passive** keyword, the software compresses all RTP traffic.

Change the Number of Header Compression Connections

By default, the software supports a total of 32 RTP header compression connections on an interface. To specify a different number of RTP header compression connections, use the following command in interface configuration mode:

Command	Purpose
`router(config-if)#` **`ip rtp compression connections`** *`number`*	Specify the total number of RTP header compression connections supported on an interface.

RTP Header Compression Configuration Example

The following example enables RTP header compression for a serial interface:

```
interface 0
 ip rtp header-compression
 encapsulation ppp
 ip rtp compression-connections 25
```

For more information about RTP header compression, see the “Configuring IP Multicast Routing” chapter of the *Network Protocols Configuration Guide, Part 1*.

Configuring IP RTP Priority

IP RTP Priority provides a strict priority queueing scheme for delay-sensitive data such as voice. Voice traffic can be identified by its Real-Time Transport Protocol (RTP) port numbers and classified into a priority queue configured by the **ip rtp priority** command. The result is that voice is serviced as strict priority in preference to other nonvoice traffic.

This feature allows you to specify a range of User Datagram Protocol (UDP)/RTP ports whose voice traffic is guaranteed strict priority service over any other queues or classes using the same output interface. Strict priority means that if packets exist in the priority queue, they are dequeued and sent first—that is, before packets in other queues are dequeued.

The IP RTP Priority feature does not require that you know the port of a voice call. Rather, the feature gives you the ability to identify a range of ports whose traffic is put into the priority queue. Moreover, you can specify the entire voice port range—16384 to 32767—to ensure that all voice traffic is given strict priority service. IP RTP Priority is especially useful on slow-speed links whose speed is less than 1.544 Mbps.

This feature can be used in conjunction with Weighted Fair Queueing (WFQ) on the same outgoing interface.Traffic matching the range of ports specified for the priority queue is guaranteed strict priority over other WFQ flows; voice packets in the priority queue are always serviced first.

When used in conjunction with WFQ, the **ip rtp priority** command provides strict priority to voice, and WFQ scheduling is applied to the remaining queues.

Because voice packets are small in size and the interface also can have large packets going out, the Link Fragmentation and Interleaving (LFI) feature should also be configured on lower speed interfaces. When you enable LFI, the large data packets are broken up so that the small voice packets can be interleaved between the data fragments that make up a large data packet. LFI prevents a voice packet from needing to wait until a large packet is sent. Instead, the voice packet can be sent in a shorter amount of time.

For more information about the IP RTP Priority feature, see the *IP RTP Priority* Cisco IOS Release 12.0(5)T online document.

To reserve a strict priority queue for a set of RTP packet flows belonging to a range of UDP destination ports, use the following command in interface configuration mode:

Command	Purpose
`router(config-if)#` **`ip rtp priority`** *`starting-rtp-port-number port-number-range bandwidth`*	Reserves a strict priority queue for a set of RTP packet flows belonging to a range of UDP destination ports.

Configuring Number Expansion

This section describes how to use the **num-exp** command to expand a set of dialed digits, such as an extension number, into a destination pattern representing a complete telephone number for Voice over IP on Cisco MC3810 concentrators.

Enter the following command in global configuration mode for each extension number to be expanded into a destination pattern.

Command	Purpose
`router(config)#` **`num-exp`** *`extension-number extension-string`*	(Optional) If using the number expansion feature, define a destination pattern for an extension number. Repeat for each extension to be expanded.

Configuring Dial Peers

This section describes how to use new commands defining dial-peer operation for Voice over IP on Cisco MC3810 series concentrators.

Configure POTS Dial Peers

POTS dial peers enable incoming calls to be received by a particular telephony device. To configure a POTS peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its telephone number(s), and associate it with a voice port through which calls will be established. Under most circumstances, the default values for the remaining dial-peer configuration commands will be sufficient to establish connections.

To enter dial-peer configuration mode (and select POTS as the method of voice-related encapsulation), use the following command in global configuration mode:

Command	Purpose
`router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enter the dial-peer configuration mode to configure a POTS peer.

The *number* value of the **dial-peer voice pots** command is a tag that uniquely identifies the dial peer. (This number has local significance only.) The tag value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.

To configure the identified POTS peer, use the following commands in dial-peer configuration mode:

Step	Command	Purpose
1	`router(config-dialpeer)#` **`destination-pattern`** *`string`*	Define the telephone number associated with this POTS dial peer. **Note**

Step	Command	Purpose
2	router(config-dialpeer)# **port** *slot*/*port*	Associate this POTS dial peer with a specific voice port.

To configure direct inward dial (DID) for a particular POTS dial peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	router(config)# **dial-peer voice** *number* **pots**	Enter dial-peer configuration mode to configure a POTS peer.
2	router(config-dialpeer)#**direct-inward-dial**	Specify direct inward dial for this POTS peer.

Note Direct inward dial is configured for the calling POTS dial peer.

Note Direct inward dial is only configured on the POTS dial peer if it corresponds to a BRI or PRI/QSIG interface. It should not be configured to correspond to an analog or T1/E1 CAS interface.

For additional POTS dial-peer configuration options, refer to the "Voice-Related Commands" section of the *Cisco IOS 12.0 Voice, Video, and Home Applications Command Reference*.

Configure VoIP Peers

VoIP peers enable outgoing calls to be made from a particular telephony device. To configure a VoIP peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its destination telephone number and destination IP address. As with POTS peers, under most circumstances, the default values for the remaining dial-peer configuration commands will be adequate to establish connections.

To enter the dial-peer configuration mode (and select VoIP as the method of voice-related encapsulation), use the following command in global configuration mode:

Command	Purpose
router(config)#**dial-peer voice** *number* **voip**	Enter the dial-peer configuration mode to configure a VoIP peer.

The *number* value of the **dial-peer voice voip** command is a tag that uniquely identifies the dial peer.

To configure the identified VoIP peer, use the following commands in dial-peer configuration mode:

Step	Command	Purpose
1	router(config-dialpeer)#**destination-pattern** *string*	Define the destination telephone number associated with this VoIP dial peer.
2	router(config-dialpeer)#**session target** {**ipv4:***destination-address* | **dns:***host-name* | **ras**}	Specify a destination IP address for this dial peer.

Step	Command	Purpose
3	`router(config-dialpeer)#` **`dtmf-relay`** `[`**`cisco-rtp`**`] [`**`h245-signal`**`] [`**`h245-alphanumeric`**`]`	(Optional) Specify how an H.323 gateway relays DTMF tones through an IP network. Options allow the gateway to forward tones "out-of-band", or separate from the voice stream. **Note** This command is only supported if your Cisco MC3810 has version 549 or newer DSPs.

For additional VoIP dial-peer configuration options, refer to the "Voice-Related Commands" section of the *Cisco IOS 12.0 Voice, Video, and Home Applications Command Reference.* For examples of how to configure dial peers, refer to the section, "Voice over IP Configuration Examples."

Validation Tips

You can check the validity of your dial-peer configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show dialplan number** command to show the dial peer to which a particular number (destination pattern) resolves.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with dial-peer configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS 12.0 Network Protocols Configuration Guide, Part 1*.
- Use the **show dial-peer voice** command to verify that the operational status of the dial peer is up.
- Use the **show dialplan number** command on the local and remote routers to verify that the data is configured correctly on both.
- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.
- If you have configured a codec value, there can be a problem if both VoIP dial peers on either side of the connection have incompatible codec values. Make sure that both VoIP peers have been configured with the same codec value.
- Use the **debug vpm spi** command to verify the output string the router dials is correct.
- Use the **debug cch323 rtp** command to check RTP packet transport.
- Use the **debug cch323 h225** command to check the call setup.

Configuring Dial Peer Hunting

After you have configured dial peers, you can configure how the router or concentrator performs dial-peer hunting functions. To configure dial-peer hunting behavior, perform the following steps beginning in global configuration mode.

Step	Command	Purpose
1	router(config)# **dial-peer hunt**	(Optional) Specify the hunting selection order for dial peers.
2	router(config)# **dial-peer terminator** *character*	(Optional) Designate a terminating character for variable length dialed numbers. The default character is # (pound sign).

If using dial peer hunting, there may be situations in which you want to disable dial-peer hunting on a specific dial peer. To disable dial-peer hunting on a dial peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	router(config)# **dial-peer voice** *tag* {**pots** | **voip**}	Enter dial-peer configuration mode for the specified dial peer.
2	router(config-dial-peer)# **huntstop**	Disable dial-peer hunting on the dial peer. Once you enter this command, no further hunting will be allowed if a call fails on the specified dial peer.

To reenable dial-peer hunting on a dial peer, enter the **no huntstop** command.

Configuring Dial Peer Digit Manipulation

After you have configured dial peers, you can configure the dial-peer digit manipulation. To configure dial-peer digit manipulation, perform one or more of the following steps beginning in dial-peer configuration mode.

Step	Command	Purpose
1	router(config-dialpeer)# **forward-digits** {*num-digit* | **all** | **extra**} or router(config-dialpeer)# **default forward-digits** or router(config-dialpeer)# **no forward-digits**	(Optional) If using the forward-digits feature, configure the digit-forwarding method. The range for the number of digits forwarded (*num-digit*) is 0 to 32. Refer to the command reference section for an explanation of the command options. In the default condition, dialed digits not matching the destination pattern are forwarded. **Note** The **no** state is not the **default** state.
2	router(config-dialpeer)# **prefix** *string*	(Optional) If the forward-digits feature was not configured in the last step, assign the dialed digits prefix for the dial peer.
3	router(config-dialpeer)# **preference** *value*	(Optional) Configure a preference for the POTS dial peer. The value is a number from 0 (highest preference) to 10 (lowest preference). If POTS and voice-network (VoFR, VoATM, VoIP) dial peers are mixed in the same hunt group, POTS dial peers will be searched first, even if a voice-network peer has a higher preference number.

Optimizing Dial Peer and Network Interface Configurations

Depending on how you have configured your network interfaces, you might need to configure additional VoIP dial-peer parameters. This section describes the following topics:

- Configuring IP Precedence for Dial Peers
- Configuring Codec and VAD for Dial Peers
- Configuring Codec Selection Order

Configuring IP Precedence for Dial Peers

If you want to give real-time voice traffic a higher priority than other network traffic, you can weight the voice data traffic associated with a particular VoIP dial peer by using IP Precedence. IP Precedence provides no admission control.

To give real-time voice traffic precedence over other IP network traffic, use the following commands, beginning in global configuration mode:

Step	Command	Purpose
1	`router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enter the dial-peer configuration mode to configure a VoIP peer.
2	`router(config-dialpeer)#` **`ip precedence`** *`number`*	Select a precedence level for the voice traffic associated with that dial peer.

In IP Precedence, the numbers 1 through 5 identify classes for IP flows; the numbers 6 through 7 are used for network and backbone routing and updates.

For example, to ensure that voice traffic associated with VoIP dial peer 103 is given a higher priority than other IP network traffic, enter the following:

```
dial-peer voice 103 voip
 ip precedence 5
```

In this example, when an IP call leg is associated with VoIP dial peer 103, all packets transmitted to the IP network via this dial peer will have their precedence bits set to 5. If the networks receiving these packets have been configured to recognize precedence bits, the packets will be given priority over packets with a lower configured precedence value.

Configuring Codec and VAD for Dial Peers

Coder-decoder (codec) and voice activity detection (VAD) for a dial peer determine how much bandwidth the voice session uses. Codec typically is used to transform analog signals into a digital bit stream and digital signals back into analog signals—in this case, it specifies the voice coder rate of speech for a dial peer. VAD is used to disable the transmission of silence packets.

Configuring Codec for a VoIP Dial Peer

To specify a voice coder rate for a selected VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enter the dial-peer configuration mode to configure a VoIP peer.

Step	Command	Purpose
2	router(config-dialpeer)# **codec** {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8**}[**bytes** *payload-size*]	Specify the desired voice coder rate of speech.Optionally specify the voice payload (in bytes) of each frame.

The default for the **codec** command is **g729r8**; normally the default configuration for this command is the most desirable. If, however, you are operating on a high bandwidth network and voice quality is of the highest importance, you should configure the **codec** command for **g711alaw** or **ulaw**. Using this value will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to specify a codec rate of G.711a-law for VoIP dial peer 108, enter the following:

```
dial-peer voice 108 voip
 destination-pattern +14085551234
 codec g711alaw
 session target ipv4:10.0.0.8
```

Configuring VAD for a VoIP Dial Peer

To disable the transmission of silence packets for a selected VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	router(config)# **dial-peer voice** *number* **voip**	Enter dial-peer configuration mode to configure a VoIP peer.
2	router(config)# **vad**	Disable the transmission of silence packets (enabling VAD).

The default for the **vad** command is enabled; normally the default configuration for this command is the most desirable. If you are operating on a high bandwidth network and voice quality is of the highest importance, you should disable **vad**. Using this value will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to enable VAD for VoIP dial peer 108, enter the following:

```
dial-peer voice 108 voip
 destination-pattern +14085551234
 vad
 session target ipv4:10.0.0.8
```

Configuring Codec Selection Order

To configure codec selection order, perform the following tasks:

- Configuring a Voice Class to Define Codec Selection Order
- Applying a Voice Class for Codec Selection to a VoIP Dial Peer

Configuring a Voice Class to Define Codec Selection Order

You can define a voice class in which you configure a selection order for codecs, and then map the voice class to a VoIP dial peer.

To configure a voice class in which you can define the order of preference in which a router selects a codec when it negotiates with a far-end router, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`router(config)#` **`voice class codec`** *`tag`*	Create a voice class for a codec preference list. The range for the *tag* number is 1 to 10000. The *tag* number must be unique on the router.
2	`router(config-voice-class)#` **`codec preference`** *`priority codec`* [**`bytes`** *`payload-size`*]	Configure the selection order of preference for a codec. Repeat this command to specify selection orders of preference for additional codecs, if required.
3	`router(config-voice-class) #`**`exit`**	Exit from voice-class configuration mode.

Applying a Voice Class for Codec Selection to a VoIP Dial Peer

After you have created the voice class, assign it to a VoIP dial peer. You cannot assign voice-class codec attributes to POTS dial peers.

To apply voice-class signaling attributes to a VoIP dial peer, complete the following steps beginning in global configuration mode:

Step	Command	Purpose
1	`router(config)#` **`dial-peer voice`** *`tag`* **`voip`**	Define a VoIP dial peer and enter dial-peer configuration mode. All subsequent commands that you enter in dial-peer voice mode before you exit will apply to this dial peer. The *tag* is a number that identifies the dial peer and must be unique on the router. Do not assign duplicate tag numbers.
2	`router(config-dialpeer)#` **`voice-class codec`** *`tag`*	Assign to the dial peer the voice class that you created in the “Configuring a Voice Class to Define Codec Selection Order” section. **Note** The **voice-class** command in dial-peer configuration mode is entered with a hyphen. The **voice class** command in global configuration mode is entered without the hyphen.

Verifying Codec Settings of Dial Peers

To display the codec voice-classes assigned to VoIP dial peers, enter the **show running-config** command.

The following example shows exerpts from the **show running-config** command output, where three codec voice classes (10, 20 and 30) have been applied to three VoIP dial peers (101, 102 and 102):

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

Current configuration:
!
version 12.0
.
.
.
voice class codec 10
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g726r16 bytes 120
!
voice class codec 20
 codec preference 1 g726r24 bytes 90
 codec preference 2 g726r32 bytes 120
!
voice class codec 30
 codec preference 1 g729ar8
 codec preference 2 g726r16
 codec preference 3 g726r32
!
.
.
.
dial-peer voice 101 voip
 voice-class codec 10
!
dial-peer voice 102 voip
 voice-class codec 20
!
dial-peer voice 103 voip
 voice-class codec 30
!
line con 0
 transport input none
line aux 0
line 2 3
line vty 0 4
 password #1writer
 login
!
end
```

Configuring Voice Ports

This section describes how to configure voice ports for Voice over IP (VoIP) on Cisco MC3810 series concentrators.

Perform the following tasks, as applicable, to configure voice ports:

- Configuring FXO or FXS Voice Ports
- Fine-Tuning FXO and FXS Voice Ports

- Configuring E&M Voice Ports
- Fine-Tuning E&M Voice Ports
- Activating the Voice Port

Configuring FXO or FXS Voice Ports

Under most circumstances the default values are adequate for FXO and FXS voice ports.

Task List

If you need to change the default configuration for these voice ports, perform the following tasks:

1. Configure the applicable parameters for the voice port.
2. Verify the configuration.
3. Troubleshoot and correct any configuration errors.

Configuration Procedure

To configure FXO and FXS voice ports, enter the following commands, beginning in global configuration mode. Commands apply to both analog and digital voice ports unless otherwise indicated.

Step	Command	Purpose
1	`router(config)#` **`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voice-port)#`**`connection`** {**`plar`** | **`tie-line`** | **`trunk`** | **`plar-opx`**} *`string`*	Specify the voice-port connection type and the destination telephone number. • **plar** for private line auto ringdown • **tie-line** for a tie-line connection to a PBX • **plar-opx** for PLAR off-premises extension (the local voice port provides a local response before the remote voice port receives an answer) • *string* specifies the destination telephone number.
3	`router(config-voice-port)#`**`voice confirmation-tone`**	If **connection plar** or **connection plar-opx** is configured, enable the two-beep confirmation tone that a caller hears when picking up the handset.
4	`router(config-voice-port)#`**`dial-type`** {**`dtmf`** | **`pulse`**}	If you are configuring for rotary dialing, select **pulse** as the out-dialing type. The default is touch-tone (**dtmf**). Out-dialing type is not applicable on FXS voice ports.
5	`router(config-voice-port)#`**`signal`** {**`loop-start`** | **`ground-start`**}	(Analog only) Select the appropriate signaling type.
6	`router(config-voice-port)#`**`cptone`** *`country`*	Select the appropriate call progress tone for your country location. The default is **northamerica**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*.

Step	Command	Purpose
7	router(config-voice-port)#**compand-type** {**u-law** | **a-law**}	Configure the companding standard used to convert between analog and digital signals in PCM systems. Defaults are: **u-law** for T1; **a-law** for E1.
8	router(config-voice-port)#**vad**	(Optional) Enable voice activity detection (VAD).
9	router(config-voice-port)#**comfort-noise**	(Optional) Enable background noise if VAD is enabled.
10	router(config-voice-port)#**music-threshold** *number*	(Optional) Specify the maximum volume (in dBm) for on-hold music. Valid entries are –70 to –30.
11	router(config-voice-port)#**description** *string*	(Optional) Describe the location, connected equipment, or other information about the voice port. The description is displayed when a **show** command is entered.
12	router(config-voice-port)#**exit**	Exit from voice-port configuration mode.
13	router(config)# **voice-card 0**	Enter voice-card configuration mode and specify voice card 0. Voice card 0 provides the configuration mode for setting the codec complexity on a Cisco MC3810.
14	router(config-voicecard)# **codec complexity** {**high** | **medium**}	Specify the codec complexity for this Cisco MC3810 according to the bandwidth requirements and the number of voice channels to be supported per DSP. The default is medium complexity, which provides four voice channels per DSP. **Note** You cannot change codec complexity while DS0 groups are defined. If they are already set up, use the **no ds0-group** command before resetting the codec complexity.
15	router(config-voice-ca)#**exit**	Exit from voice-card configuration mode.
16	router(config-voice-port)#**exit**	Exit from voice-port configuration mode.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, the voice port is most likely configured properly.
- Use the **show voice port** or **show voice port summary** command to view the voice-port configuration.
- Use the **show voice dsp** command to view the current status of all DSP voice channels.
- Use the **show voice call summary** command to view the call status for all voice ports.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, enter the **no shutdown** command.
- Check to see if the analog personality module is correctly installed. For more information, refer to the hardware installation guide for your router or concentrator.

Fine-Tuning FXO and FXS Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation. The commands for these parameters are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

Task List

To fine tune FXO and FXS voice ports, perform the following tasks:

1. Perform the voice-port tuning procedure for the voice port.
2. Verify the configuration.
3. Troubleshoot and correct any configuration errors.

Voice-Port Tuning Procedure

To fine-tune FXO and FXS voice ports, perform the following optional steps, beginning in global configuration mode. Commands apply to both analog and digital voice ports unless otherwise indicated.

Note After you change voice-port parameters, Cisco recommends that you cycle the port by entering the **shutdown** and **no shutdown** commands.

Step	Command	Purpose
1	`router(config)#`**`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#`**`input gain`** *`value`*	Specify the receive gain (in dB) for the voice port. Value range is –6 to 14.
3	`router(config-voiceport)#`**`output attenuation`** *`value`*	Specify the transmit attenuation (in dB) for the voice port. Value range is 0 to 14.
4	`router(config-voiceport)#`**`echo-cancel enable`**	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
5	`router(config-voiceport)#`**`echo-cancel coverage`** {**16** | **24** | **32**}	Set the duration (in milliseconds) of echo cancellation. Values are 16, 24, and 32.
6	`router(config-voiceport)#`**`non-linear`**	Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
7	`router(config-voiceport)#`**`playout-delay`**	Tune the playout buffer to accommodate packet jitter caused by switches in the WAN.

Step	Command	Purpose
8	router(config-voiceport)# **condition** {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}	(For T1/E1 digital voice ports only.) Configure the voice port to manipulate the transmit and/or receive bit patterns to match the bit patterns required by a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate both an on-hook and off-hook state. **Note** The **show voice port** command reports at the protocol level, while the **show controller** command reports at the driver level. The driver is not notified of any bit manipulation using the **condition** command. As a result, the **show controller** command output will not account for the bit conditioning.
9	router(config-voiceport)# **timeouts initial** *seconds*	Specify the number of seconds the system waits for a caller to dial the first digit. The range is 10 to 120. The default is 10.
10	router(config-voiceport)# **timeouts interdigit** *seconds*	Specify the number of seconds the system waits, after a caller has dialed the initial digit, for the caller to dial each subsequent digit. The range is 0 to 120. The default is 10.
11	router(config-voiceport)# **timeouts ringing** {*seconds* | **infinity**}	Specify the maximum number of seconds that a voice port allows ringing to continue if a call is not answered. The range is 5 to 60000. The default is 180.
12	router(config-voiceport)# **timeouts wait-release** {*seconds* | **infinity**}	Specify the maximum number of seconds that a voice port can remain in the call failure state while the router or concentrator sends a busy tone, reorder tone, or out-of-service tone to the port. The *value* range is 5 to 3600. The default is 30.
13	router(config-voiceport)# **timing digit** *milliseconds*	If the dial type is DTMF, configure the DTMF digit signal duration in milliseconds. The range is 50 to 100. The default is 100.
14	router(config-voiceport)# **timing inter-digit** *milliseconds*	If the dial type is DTMF, configure the DTMF inter-digit signal duration in milliseconds. The range is 50 to 500. The default is 100.
15	router(config-voiceport)# **timing pulse-digit** *milliseconds*	If the dial type is pulse, configure the pulse digit signal duration in milliseconds. The range is 10 to 20. The default is 20.
16	router(config-voiceport)# **timing pulse-inter-digit** *milliseconds*	If the dial type is pulse, configure the pulse inter-digit signal duration in milliseconds. The range is 100 to 1000. The default is 500.
17	router(config-voiceport)# **timing percentbreak** *percent*	(FXO only) Specify the percentage of the break period for dialing pulses. The range is 20 to 80. The default is 50.
18	router(config-voiceport)# **timing guard-out** *milliseconds*	(FXO only) Specify the duration in milliseconds of the guard-out period to prevent this port from seizing a remote FXS port before the remote port detects a disconnect signal. The range is 300 to 3000. The default is 2000.
19	router(config-voiceport)# **impedance** {**600r** | **600c** | **900r** | **900c**}	(FXO only) Configure the impedance. The default is **600r** (600 ohms real).
20	router(config-voiceport)# **ring number** *number*	(Analog FXO only) Configure the number of rings detected before a call is answered on the FXO port. The range is 1 to 10. The default is 1.
21	router(config-voiceport)# **ring frequency** *number*	(FXS only) Specify the local ring frequency (Hertz) for the FXS voice port. Valid entries are 20 and 30. The default is 20.

Step	Command	Purpose
22	router(config-voiceport)# **disconnect-ack**	(FXS only) Configure the voice port to return an acknowledgment upon receipt of a disconnect signal.
23	router(config-voiceport)# **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse-interval*]}	(FXS only) Specify the on and off times for the ringing pulses. See the command reference section for details on the ring cadence options.
24	router(config-voiceport)#**exit**	Exit from voice-port configuration mode.

Configuring E&M Voice Ports

The default E&M voice-port parameters will probably not be sufficient to enable voice transmission over your network. Configuration parameters depend on the PBX to which the voice port is connected.

Note E&M voice-port values must match those of the PBX to which the voice port is connected. Refer to the documentation that came with your PBX to determine the E&M voice-port configuration values.

Task List

To configure E&M voice ports, perform the following tasks:

1 Configure the applicable parameters for the voice port.

2 Verify the configuration.

3 Troubleshoot and correct any configuration errors.

Configuration Procedure

To configure E&M voice ports, enter the following commands beginning in global configuration mode. Commands apply to both analog and digital voice ports unless otherwise indicated.

Step	Command	Purpose
1	router(config)# **voice-port** *slot/port*	Identify the voice port you want to configure and enter voice-port configuration mode.

Step	Command	Purpose
2	`router(config-voiceport)#` **`connection`** {**`plar`** | **`tie-line`** | **`trunk`** | **`plar-opx`**} *`destination-string`* [**`answer-mode`**]	Specify the voice-port connection type and the destination telephone number. • **plar** specifies a private line automatic ring down (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off hook a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX. • **tie-line** specifies a connection that emulates a temporary tie-line trunk to a private branch exchange (PBX). A tie-line connection is automatically set up for each call and torn down when the call ends. • **trunk** specifies a connection that emulates a permanent trunk connection to a private branch exchange (PBX). A trunk connection remains “nailed up” in the absence of any active calls. • **plar-opx** specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice-port provides a local response before the remote voice-port receives an answer. On FXO interfaces, the voice-port will not answer until the remote side answers. • *destination-string* specifies the destination telephone number. When configuring Cisco-trunk permanent calls, one side must be the call initiator (master) and the other side is normally 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.
3	`router(config-voiceport)#` **`voice confirmation-tone`**	If **connection plar-opx** is configured, enable the two-beep confirmation tone that a caller hears when picking up the handset.
4	`router(config-voiceport)#` **`dial-type`** {**`dtmf`** | **`pulse`** | **`mf`** }	Select the dial type for dialing out. • **dtmf** for touch-tone (the default) • **pulse** for rotary dial • **mf** for multifrequency tone dialing
5	`router(config-voiceport)#` **`operation`** {**`2-wire`** | **`4-wire`**}	Select the appropriate cabling scheme for this voice port.

Step	Command	Purpose
6	router(config-voiceport)# **type** {**1** | **2** | **3** | **5**}	Select the appropriate E&M interface type. Type 1 lead configuration: E—output, relay to ground M—input, referenced to ground Type 2 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 lead configuration: E—output, relay to ground M—input, referenced to ground SB—connected to –48V SG—connected to ground Type 5 lead configuration: E—output, relay to ground M—input, referenced to –48V.
7	router(config-voiceport)# **signal** {**wink-start** | **immediate** | **delay-dial**}	Configure the E&M signaling type. The default is **wink-start**.
8	router(config-voiceport)# **cptone** *country*	Select the appropriate call progress tone for your country location. The default is **northamerica**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*.
9	router(config-voiceport)# **compand-type** {**u-law** | **a-law**}	Configure the companding standard used to convert between analog and digital signals in PCM systems. Defaults are: **u-law** for T1; **a-law** for E1.
10	router(config-voiceport)# **no vad**	(Optional) Disable voice activity detection (VAD). VAD is enabled by default.
11	router(config-voiceport)# **comfort-noise**	(Optional) Enable background noise if VAD is enabled.
12	router(config-voiceport)# **music-threshold** *number*	(Optional) Specify the maximum volume (in dBm) for on-hold music. Valid entries are –70 to –30. The default is –38.
13	router(config-voiceport)# **voice confirmation-tone**	(Optional) If the voice port is configured for **connection plar-opx** for Off-Premises eXtension, disable the two-beep confirmation tone that a caller hears when picking up the handset.
14	router(config-voiceport)# **description** *string*	(Optional) Describe the location, connected equipment, or other information about the voice port. The description is displayed when a **show** command is entered.
15	router(config-voice-port)#**exit**	Exit from voice-port configuration mode.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, the voice port is most likely configured properly.
- Use the **show voice port** command to view the voice-port configuration.
- Use the **show voice dsp** command to view the current status of all DSP voice channels.
- Use the **show voice call summary** command to view the call status for all voice ports.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS 12.0 Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, enter the **no shutdown** command.
- Make sure that the values pertaining to your PBX setup, such as timing and type, are correct.
- Check to see if the analog personality module is correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*.

Fine-Tuning E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation. The commands for these parameters are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

Task List

To fine tune E&M voice ports, perform the following tasks:

1. Perform the voice-port tuning procedure for the voice port.
2. Verify the configuration.
3. Troubleshoot and correct any configuration errors.

Voice-Port Tuning Procedure

To fine-tune E&M voice ports, perform the following steps, beginning in privileged EXEC mode. Commands apply to both analog and digital voice ports unless otherwise indicated.

Note After you change voice-port parameters, Cisco recommends that you cycle the port by entering the **shutdown** and **no shutdown** commands.

Step	Command	Purpose
1	`router# `**`configure terminal`**	Enter global configuration mode.
2	`router(config)# `**`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
3	`router(config-voiceport)# `**`input gain`** *`value`*	Specify the receive gain (in dB) for the voice port. Value range is –6 to 14.
4	`router(config-voiceport)# `**`output attenuation`** *`value`*	Specify the transmit attenuation (in dB) for the voice port. Value range is 0 to 14.
5	`router(config-voiceport)# `**`echo-cancel enable`**	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	`router(config-voiceport)# `**`echo-cancel coverage`** *`milliseconds`*	Set the duration (in milliseconds) of echo cancellation. Values are 16, 24, and 32.
7	`router(config-voiceport)# `**`non-linear`**	Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
8	`router(config-voiceport)# `**`playout-delay`**	Tune the playout buffer to accommodate packet jitter caused by switches in the WAN.
9	`router(config-voiceport)# `**`condition`** {**`tx-a-bit`** | **`tx-b-bit`** | **`tx-c-bit`** | **`tx-d-bit`**} {**`rx-a-bit`** | **`rx-b-bit`** | **`rx-c-bit`** | **`rx-d-bit`**} {**`on`** | **`off`** | **`invert`**}	(For T1/E1 digital voice ports only.) Configure the voice port to manipulate the transmit and/or receive bit patterns to match the bit patterns required by a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate both an on-hook and off-hook state. **Note** The **show voice port** command reports at the protocol level, while the **show controller** command reports at the driver level. The driver is not notified of any bit manipulation using the **condition** command. As a result, the **show controller** command output will not account for the bit conditioning.
10	`router(config-voiceport)# `**`define`** {**`Tx-bits`** | **`Rx-bits`**} {**`seize`** | **`idle`**} {**`0000`** | **`0001`** | **`0010`** | **`0011`** | **`0100`** | **`0101`** | **`0110`** | **`0111`** | **`1000`** | **`1001`** | **`1010`** | **`1011`** | **`1100`** | | **`1101`** | **`1110`** | **`1111`**}	(For T1/E1 digital voice ports only.) Define specific transmit and/or receive signaling bits to match the bit patterns required by a connected device.
11	`router(config-voiceport)# `**`ignore`** {**`rx-a-bit`** | **`rx-b-bit`** | **`rx-c-bit`** | **`rx-d-bit`**}	(For T1/E1 digital voice ports only.) Configure the voice port to ignore specified transmit and/or receive bits.
12	`router(config-voiceport)# `**`timeouts initial`** *`seconds`*	Specify the number of seconds the system waits for a caller to dial the first digit. The range is 0 to 120. The default is 10.
13	`router(config-voiceport)# `**`timeouts interdigit`** *`seconds`*	Specify the number of seconds the system waits (after a caller has dialed the initial digit) for the caller to dial each subsequent digit. The range is 0 to 120. The default is 10.
14	`router(config-voiceport)#`**`timeouts ringing`** {*`seconds`* | **`infinity`**}	Specify the maximum number of seconds that a voice port allows ringing to continue if a call is not answered. The range is 5 to 60000. The default is 180.
15	`router(config-voiceport)# `**`timeouts wait-release`** {*`seconds`* | **`infinity`**}	Specify the maximum number of seconds that a voice port can remain in the call failure state while the router or concentrator sends a busy tone, reorder tone or out-of-service tone to the port. The *value* range is 5 to 3600. The default is 30.

Step	Command	Purpose
16	`router(config-voiceport)#` **`timing clear-wait`** *`milliseconds`*	Specify the number of milliseconds between the inactive seizure signal and the call being cleared. The range is 100 to 2000. The default is 400.
17	`router(config-voiceport)#` **`timing delay-duration`** *`milliseconds`*	Specify the delay signal duration in milliseconds for delay dial signaling. This command applies only if the **signal** command is set to **delay-dial**. The range is 100 to 5000. The default is 140.
18	`router(config-voiceport)#` **`timing delay-start`** *`milliseconds`*	Specify the number of milliseconds of delay from the outgoing seizure to the outdial address. This value applies only if the **signal** command is set to **delay-dial**. The range is 100 to 290. The default is 150.
19	`router(config-voiceport)#` **`timing dialout-delay`** *`milliseconds`*	Configure the delay interval before sending a dialed digit or cut-through. This value applies only if the **signal** command is set to **immediate**. The range is 100 to 5000. The default is 300.
20	`router(config-voiceport)#` **`timing delay-with-integrity`** *`milliseconds`*	Specify the number of milliseconds duration of the wink pulse for delay dials. The range is 0 to 5000. The default is 0.
21	`router(config-voiceport)#` **`timing dial-pulse min-delay`** *`milliseconds`*	If the dial type is **pulse**, specify the number of milliseconds between generation of wink-like pulses. The range is 140 to 5000. The default is 140.
22	`router(config-voiceport)#` **`timing wink-duration`** *`milliseconds`*	Specify the length in milliseconds of the wink-start signal. This command applies only if the **signal** command is set to **wink-start**. The range is from 100 to 400 milliseconds and the default is 200.
23	`router(config-voiceport)#` **`timing wink-wait`** *`milliseconds`*	Specify the wink-wait duration in milliseconds for a wink-start signal. This command applies only if the **signal** command is set to **wink-start**. The range is 100 to 5000. The default is 200.
24	`router(config-voiceport)#` **`timing percentbreak`** *`percent`*	Specify the percentage of the break period for dialing pulses. The range is 20 to 80. The default is 50.
25	`router(config-voiceport)#` **`timing digit`** *`milliseconds`*	If the dial type is DTMF, configure the DTMF digit signal duration in milliseconds. The range is 50 to 100. The default is 100.
26	`router(config-voiceport)#` **`timing inter-digit`** *`milliseconds`*	If the dial type is DTMF, configure the DTMF inter-digit signal duration in milliseconds. The range is 50 to 500. The default is 100.
27	`router(config-voiceport)#` **`timing pulse`** *`pulses-per-second`*	If the dial type is pulse, specify the pulse dialing rate in pulses per second. The range is 10 to 20. The default is 10.
28	`router(config-voiceport)#` **`timing pulse-digit`** *`milliseconds`*	If the dial type is pulse, specify the pulse digit duration in milliseconds. The range is 10 to 20. The default is 20.
29	`router(config-voiceport)#` **`timing pulse-inter-digit`** *`milliseconds`*	If the dial type is pulse, configure the pulse inter-digit duration in milliseconds. The range is 100 to 1000. The default is 500.
30	`router(config-voice-port)#`**`exit`**	Exit from voice-port configuration mode.

Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. Cisco recommends that you cycle the port—shut the port down and then bring it online again.

To activate a voice port, enter the following command in voice-port configuration mode:

Command	Purpose
`router(config-voiceport)#` **`no shutdown`**	Activate the voice port.

To cycle a voice port, enter the following commands in voice-port configuration mode:

Step	Command	Purpose
1	`router(config-voiceport)#` **`shutdown`**	Deactivate the voice port.
2	`router(config-voiceport)#` **`voice-port`** *`slot/port`*	Identify the voice port you want to activate and enter the voice-port configuration mode.
3	`router(config-voiceport)#` **`no shutdown`**	Activate the voice port.
4	`router(config-voice-port)#`**`exit`**	Exit from voice-port configuration mode.

Note If you are not going to use a voice port, shut it down.

Configuring the H.323 Gateway

In this release, basic gateway Registration, Admission, and Status (RAS) protocol capability is extended to the Cisco MC3810. Other features, such as authentication, authorization, and accounting (AAA) enhancements for security and accounting services, interactive voice response (IVR), Integrated Services Digital Network (ISDN) redirect number support, and rotary call pattern support, will be offered in future Cisco IOS releases.

To configure the H.323 Gateway, you need to perform the following tasks

- Configuring POTS and VoIP Dial Peers
- Enabling VoIP Gateway Functionality
- Configuring Gateway Interface Parameters

Configuring POTS and VoIP Dial Peers

The first step in configuring the H.323 gateway is to define the applicable POTS and VoIP dial peers. The POTS dial peer informs the system which voice port to direct incoming VoIP calls. The VoIP dial peer defines how to direct calls that originate from a local voice port into the VoIP cloud to the session target. The **session target** command indicates the address of the remote gateway where the call is terminated. There are several different ways to define the destination gateway address: by statically configuring the IP address of the gateway, by defining the DNS of the gateway, or by using RAS. If you use RAS, that gateway determines the destination target by querying the RAS gatekeeper. See the “Configuring Dial Peers” section on page 9 to define dial peers for VoIP.

Enabling VoIP Gateway Functionality

Enable VoIP gateway functionality by using the **gateway** command.

To enable gateway functionality, use the following commands:

Step	Command	Purpose
1	`router#` **`configure terminal`**	Enter global configuration mode.
2	`router(config)#` **`gateway`**	Enable the VoIP gateway.

Configuring Gateway Interface Parameters

The next step in configuring an H.323 gateway is to configure the gateway interface parameters. First define which interface will be presented to the VoIP network as this gateway's H.323 interface. Only one interface is allowed to be the gateway interface. You can select either the interface that is connected to the gatekeeper or a loopback interface. The interface that is connected to the gatekeeper is usually a LAN interface (for example, Fast Ethernet, Ethernet, FDDI, or Token Ring).

After you define the gateway interface, configure the gateway to discover the gatekeeper either through multicasting or by directing it to a specific host. Then configure the gateway's H.323 identification number and any technology prefixes that this gateway should register with the gatekeeper.

To define the interface to be used as the H.323 gateway interface and configure the H.323 gateway interface parameters, use the following commands, beginning in global configuration mode:

Step	Command	Purpose
1	router(config)# **interface** *type slot/port*	Enter interface configuration mode to configure parameters for the specified interface.
2	router(config-if)# **ip address** *ip-address subnet-mask*	Specify the IP address for this interface.
3	router(config-if)#**h323-gateway voip interface**	Designate this interface as the H.323 gateway interface.
4	router(config-if)#**h323-gateway voip h323-id** *interface-id*	Specify an H.323 name (ID) for the gateway associated with this interface. This ID is used by this gateway when this gateway communicates with the gatekeeper. Usually, this H.323 ID is the name given to the gateway with the gatekeeper domain name appended to the end.
5	router(config-if)#**h323-gateway voip id** *gatekeeper* {**ipaddr** *ip-address* [*port*] | **multicast**}	Specify the name (ID) of the gatekeeper associated with this gateway and how the gateway finds it. The gatekeeper ID configured here must exactly match the gatekeeper ID in the gatekeeper configuration. The gateway determines the location of the gateway in one of two ways: either by a defined IP address or through multicast.
6	router(config-if)#**h323-gateway voip tech-prefix** *prefix*	Specify a technology prefix. A technology prefix is used to identify a type of service that this gateway is capable of providing. **Note** If a gateway is capable of handling multiple services, specify each service with a tech-prefix command.
7	router(config-if)#**exit**	Exit interface configuration mode.
8	router(config)#**exit**	Exit global configuration mode.

Configuration Example

The actual Voice over IP configuration procedure you complete depends on the actual topology of your voice network. The following configuration examples should give you a starting point. Of course, these configuration examples would need to be customized to reflect your network topology.

Configuration examples are supplied for the following scenarios:

- Linking PBX Users with E&M Trunk Lines
- PSTN Gateway Access Using FXO Connection
- PSTN Gateway Access Using FXO Connection (PLAR Mode)
- Codec Preference Configuration

These examples are described in the following sections. The following examples use the term "router" to generically describe Cisco routers and concentrators.

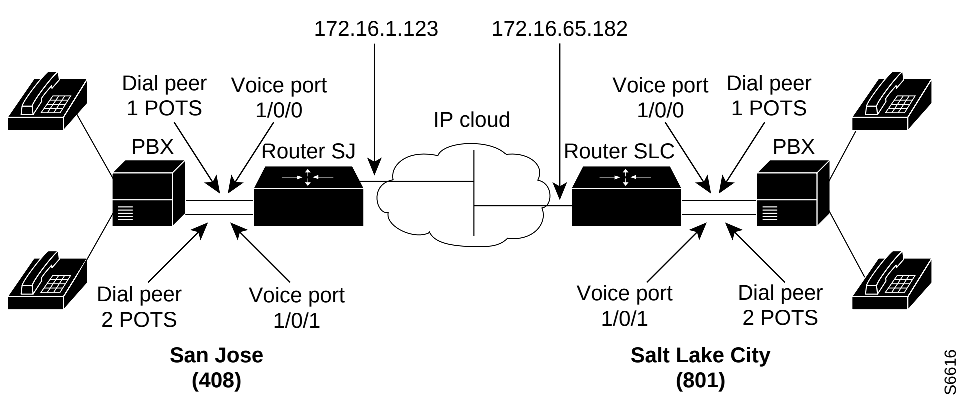
Linking PBX Users with E&M Trunk Lines

The following example shows how to configure Voice over IP to link PBX users with E&M trunk lines.

In this example, a company wants to connect two offices: one in San Jose, California and the other in Salt Lake City, Utah. Each office has an internal telephone network using PBX, connected to the voice network by an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and ImmediateStart signaling. Each E&M interface connects to the router using two voice interface connections. Users in San Jose dial "8-569" and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial "4-527" and then the extension number to reach a destination in San Jose.

Figure 2 illustrates the topology of this connection example.

Figure 2 Linking PBX Users with E&M Trunk Lines Example



Note This example assumes that the company already has established a working IP connection between its two remote offices.

Configuration for Router SJ

```
hostname sanjose
```

```
!Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern 555....
 port 1/0/0

!Configure pots dial peer 2
dial-peer voice 2 pots
 destination-pattern 555....
 port 1/0/1

!Configure voip dial peer 3
dial-peer voice 3 voip
 destination-pattern 119....
 session target ipv4:172.16.65.182

!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2

voice-port 1/0/1
 signal immediate
 operation 4-wire
 type 2

!Configure the serial interface
interface serial 0/0
 description serial interface type dce (provides clock)
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Configuration for Router SLC

```
hostname saltlake

!Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern 119....
 port 1/0/0

!Configure pots dial peer 2
dial-peer voice 2 pots
 destination-pattern 119....
 port 1/0/1

!Configure voip dial peer 3
dial-peer voice 3 voip
 destination-pattern 555....
 session target ipv4:172.16.1.123

!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2

voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2

!Configure the serial interface
```

```
interface serial 0/0
 description serial interface type dte
 ip address 172.16.65.182
 no shutdown
```

Note PBXs should be configured to pass all DTMF signals to the Cisco voice router. Cisco recommends that you do not configure store and forward tone.

Note If you change the gain or the telephony port, make sure that the telephony port still accepts DTMF signals.

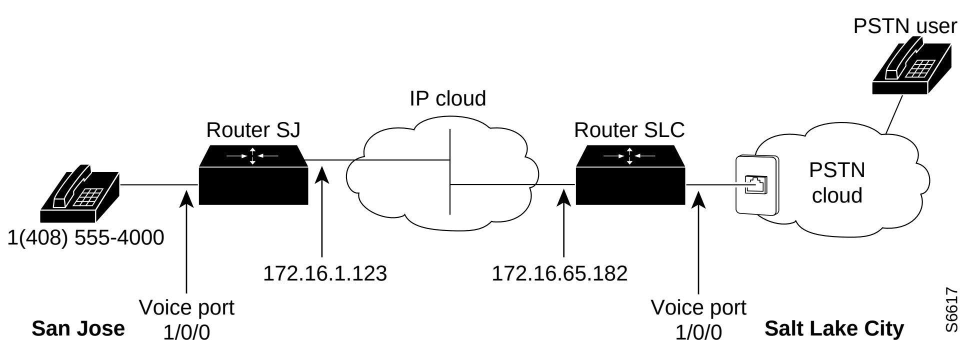
PSTN Gateway Access Using FXO Connection

The following example shows how to configure Voice over IP to link users with the PSTN gateway using an FXO connection.

In this example, users connected to Router SJ in San Jose, California can reach PSTN users in Salt Lake City, Utah via Router SLC. Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure 3 illustrates the topology of this connection example.

Figure 3 PSTN Gateway Access Using FXO Connection Example



Note This example assumes that the company already has established a working IP connection between its two remote offices.

Configuration for Router SJ

```
! Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern +14085554000
 port 1/0/0

! Configure voip dial peer 2
dial-peer voice 2 voip
 destination-pattern 9...........
 session target ipv4:172.16.65.182
```

```
! Configure the serial interface
interface serial 0/0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Configuration for Router SLC

```
! Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern 9...........
 port 1/0/0

! Configure voip dial peer 2
dial-peer voice 2 voip
 destination-pattern +14085554000
 session target ipv4:172.16.1.123

! Configure serial interface
interface serial 0/0
 ip address 172.16.65.182
 no shutdown
```

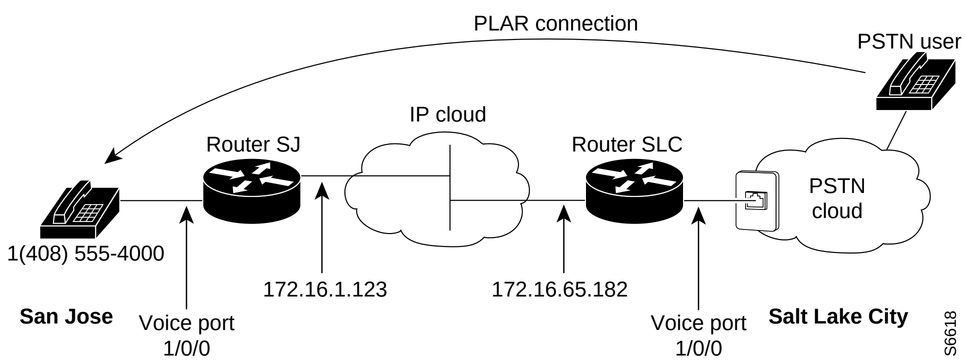
PSTN Gateway Access Using FXO Connection (PLAR Mode)

The following example shows how to configure Voice over IP to link users with the PSTN gateway using an FXO connection (PLAR mode).

In this example, PSTN users in Salt Lake City, Utah, can dial a local number and establish a private line connection in a remote location. As in the previous example, Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure 4 illustrates the topology of this connection example.

Figure 4 PSTN Gateway Access Using FXO Connection (PLAR Mode)



Note This example assumes that the company already has established a working IP connection between its two remote offices.

Configuration for Router SJ

```
! Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern +14085554000
 port 1/0/0

! Configure voip dial peer 2
dial-peer voice 2 voip
 destination-pattern 9...........
 session target ipv4:172.16.65.182

! Configure the serial interface
interface serial 0/0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Configuration for Router SLC

```
! Configure pots dial peer 1
dial-peer voice 1 pots
 destination-pattern 9...........
 port 1/0/0

! Configure voip dial peer 2
dial-peer voice 2 voip
 destination-pattern +14085554000
 session target ipv4:172.16.1.123

! Configure the voice-port
voice-port 1/0/0
connection plar 14085554000

! Configure the serial interface
interface serial 0/0
 ip address 172.16.65.182
 no shutdown
```

Codec Preference Configuration

The following example enters voice class codec configuration mode, creates voice class 10, and defines a preference list of 12 codecs:

```
router(config)# voice class codec 10
router(config-class)# codec preference 1 g711alaw
router(config-class)# codec preference 2 g711ulaw bytes 80
router(config-class)# codec preference 3 g723ar53
router(config-class)# codec preference 4 g723ar63 bytes 144
router(config-class)# codec preference 5 g723r53
router(config-class)# codec preference 6 g723r63 bytes 120
router(config-class)# codec preference 7 g726r16
router(config-class)# codec preference 8 g726r24
router(config-class)# codec preference 9 g726r32 bytes 80
router(config-class)# codec preference 10 g728
router(config-class)# codec preference 11 g729br8
router(config-class)# codec preference 12 g729r8 bytes 50
router(config-class)# exit
router(config-class)# exit
router(config)#
```

The following example assigns a voice class 10 to a VoIP dial peer:

```
router(config)# dial-peer voice 25 voip
router(config-dial-peer)# voice-class codec 10
```

Command Reference

This section documents new or modified commands. Modified commands are indicated by an asterisk (*). All other commands used on these platforms are documented in the Cisco IOS Release 12.0 command reference publications.

- **codec preference***
- **connection***
- **dial-peer hunt***
- **dial-peer terminator***
- **dial-peer voice***
- **ds0-group***
- **dtmf-relay**
- **forward-digits***
- **huntstop***
- **icpif**
- **incoming called-number**
- **num-exp***
- **session target***
- **show call active voice***
- **show call history voice***
- **show num-exp***
- **voice class codec***
- **voice-class codec (dial-peer)***
- **voice-group***

codec preference

To define the order of preference in which network dial peers select codecs, use the **codec preference** voice-class configuration command. Enter the **no** form of this command to restore the default order of preference.

codec preference *priority codec* **bytes** *payload-size*
no codec preference

Syntax Description

priority	The order of selection preference you assign to a codec. The valid range is 1 to 12, where 1 is the highest priority.
codec	Codec options. Note Codecs with asterisk (*) are not supported on Cisco MC3810 series equipped with a voice compression module (VCM); a high-performance compression module (HCM) is required to support these codecs. **g711alaw**—G.711 A Law 64000 bps **g711ulaw**—G.711 u Law 64000 bps **g723ar53**—*G.723.1 Annex A 5300 bps **g723ar63**—*G.723.1 Annex A 6300 bps **g723r53**— *G.723.1 5300 bps **g723r63**—*G.723.1 6300 bps **g726r16**—G.726 16000 bps **g726r24**— G.726 24000 bps **g726r32**—G.726 32000 bps **g728**—*G.728 16000 bps **g729abr8**—*G.729 Annex A and Annex B 8000 bps **g729ar8**—G.729 Annex A 8000 bps **g729br8**—*G.729 Annex B 8000 bps **g729r8**—G.729 8000 bps
bytes	(Optional) The voice payload for each frame.
payload-size	(Optional) Number of bytes you specify as the voice payload of each frame. Values depend on the codec type and the packet voice protocol. See Table 1 for valid entries and default values.

Defaults

If no codec is specified, dial peers are configured for **g729r8** and the voice payload is as shown in Table 1 for G.729r8.

If a codec is specified without the **bytes** keyword, the voice payload is as shown in Table 1.

Command Modes

Voice class configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was first supported on the Cisco 2600 and 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

The routers at opposite ends of the WAN may have to negotiate the codec selection for the network dial peers. The codec preference command specifies the order of preference for selecting a negotiated codec for the connection. Table 1 describes the voice payload options and default values for the codecs and packet voice protocols.

Table 1 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (bytes)	Default Voice Payload (bytes)
g711alaw **g711ulaw**	VoIP	80, 160	160
	VoFR	40 to 240 in multiples of 40	240
	VoATM	40 to 240 in multiples of 40	240
g723ar53 **g723r53**	VoIP	20 to 220 in multiples of 20	20
	VoFR	20 to 240 in multiples of 20	20
	VoATM	20 to 240 in multiples of 20	20
g723ar63 **g723r63**	VoIP	24 to 216 in multiples of 24	24
	VoFR	24 to 240 in multiples of 24	24
	VoATM	24 to 240 in multiples of 24	24
g726r16	VoIP	20 to 220 in multiples of 20	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60
g726r24	VoIP	30 to 210 in multiples of 30	60
	VoFR	15 to 240 in multiples of 15	90
	VoATM	30 to 240 in multiples of 15	90
g726r32	VoIP	40 to 200 in multiples of 40	80
	VoFR	20 to 240 in multiples of 20	120
	VoATM	40 to 240 in multiples of 20	120
g728	VoIP	10 to 230 in multiples of 10	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60

Table 1 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (bytes)	Default Voice Payload (bytes)
g729abr8	VoIP	10 to 230 in multiples of 10	20
g729ar8	VoFR	10 to 240 in multiples of 10	30
g729br8	VoATM	10 to 240 in multiples of 10	30
g729r8			

Examples

The following example shows how to create a voice class and specify a codec selection preference for the voice class starting from global configuration mode:

```
router(config)# voice class codec 10
router(config-class)# codec preference 1 g711alaw
router(config-class)# codec preference 2 g711ulaw bytes 80
router(config-class)# codec preference 3 g723ar53
router(config-class)# codec preference 4 g723ar63 bytes 144
router(config-class)# codec preference 5 g723r53
router(config-class)# codec preference 6 g723r63 bytes 120
router(config-class)# codec preference 7 g726r16
router(config-class)# codec preference 8 g726r24
router(config-class)# codec preference 9 g726r32 bytes 80
router(config-class)# codec preference 10 g728
router(config-class)# codec preference 11 g729br8
router(config-class)# codec preference 12 g729r8 bytes 50
router(config-class)# exit
router(config)# exit
router)#
```

Related Commands

Command	Description
voice class codec	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class.
voice-class codec (dial-peer)	Assigns a previously-configured codec selection preference list to a dial peer.

connection

To specify a connection mode for a voice port, use the **connection** voice-port configuration command. Use the **no** form of this command to disable the selected connection mode.

connection {**plar | tie-line | plar-opx**} *digits* | {**trunk** *digits* **[answer-mode**]}
no connection {**plar | tie-line | plar-opx**} *digits* | {**trunk** *digits* **[answer-mode**]}

Syntax Description

plar	Specifies a private line auto ring down (PLAR) connection. PLAR is handled by associating a peer directly with an interface; when an interface goes off-hook, the peer is used to set up the second call leg and conference them together without the caller having to dial any digits.
tie-line	Specifies a tie-line connection to a private branch exchange (PBX).
plar-opx	Specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice-port provides a local response before the remote voice-port receives an answer. On FXO interfaces, the voice-port will not answer until the remote side answers.
digits	The destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
trunk	Specifies a straight tie-line connection to a private branch exchange (PBX).
answer-mode	(Optional; used only with the **trunk** keyword.) Specifies that the router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk.

Defaults

No connection mode is specified.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was first introduced.
11.3(1)MA1	This command was first supported on the Cisco MC3810, and the **tie-line** keyword was first made available on the Cisco MC3810.
11.3(1)MA5 and 12.0(2)T	The **plar-opx** keyword was first made available on the Cisco MC3810 as the **plar-opx-ringrelay** keyword. The keyword was shortened in a subsequent release.
12.0(3)XG and 12.0(4)T	The **trunk** keyword was made available on the Cisco MC3810. The **trunk answer-mode** option was added.
12.0(7)XK	This command options were unified across the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

Use this command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all incoming calls over this connection. The destination peer is determined by the called number.

Use the **connection trunk** command to specify a straight tie-line connection to a PBX. You can use the **connection trunk** command for E&M-to-E&M trunks, FXO-to-FXS trunks, and FXS-to-FXS trunks. Signaling will be transported for E&M-to-E&M trunks and FXO-to-FXS trunks; signaling will not be transported for FXS-to-FXS trunks.

If you desire one of the devices in a static trunk connection to act as slave and receive calls only, use the **answer-mode** option with the **connection trunk** command when configuring that device.

Note When using the **connection trunk** command, you must perform a shutdown/no shutdown command sequence on the voice port.

The **connection tie-line** command is used on the Cisco router when a dial plan requires that additional digits be added in front of any digits dialed by the PBX, and that the combined set of digits be used to route the call via the dial-peers and into the network. The operation is similar to the **connection plar** command operation, but in this case the tie-line port also waits to collect digits from the PBX. The tie-line digits are also automatically stripped by a terminating port.

If the **connection** command is not configured, the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial-peer and complete the call.

Examples

The following example selects PLAR as the connection mode on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection plar 5559262
```

The following example selects tie-line as the connection mode on a Cisco MC3810, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/1
router(config-voiceport)# connection tie-line 5559262
```

The following example specifies a PLAR off-premises extension connection on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
 router(config-voiceport)# connection plar-opx 5559262
```

The following example configures a Cisco 3600 series router for a trunk connection and specifies that it will establish the trunk only when it receives an incoming call:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection trunk 5559262 answer-mode
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.
session-protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session-target	Configures a network-specific address for a dial peer.

dial-peer hunt

To specify a hunt selection order for dial-peers, use the **dial-peer hunt** dial-peer configuration command. Use the **no** form of this command to restore the default selection order.

dial-peer hunt *hunt-order-number*
no dial-peer hunt

Syntax Description

hunt-order-number	A number from 0 to 7 that selects a predefined hunting selection order: 0—Longest match in phone number, explicit preference, random selection. This is the default hunt order number. 1—Longest match in phone number, explicit preference, least recent use. 2—Explicit preference, longest match in phone number, random selection. 3—Explicit preference, longest match in phone number, least recent use. 4—Least recent use, longest match in phone number, explicit preference. 5—Least recent use, explicit preference, longest match in phone number. 6—Random selection. 7—Least recent use.

Defaults

The default is longest match in phone number, explicit preference, random selection (hunt order number 0).

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was first introduced and was first supported on the Cisco 2600 and 3600 Series routers and on the Cisco MC3810 multiservice access concentrator.

Usage Guidelines

Use the **dial-peer hunt** dial-peer configuration command if you have configured hunt groups. “Longest match in phone number” refers to the destination pattern that matches the greatest number of the dialed digits. “Explicit preference” refers to the **preference** setting in the dial-peer configuration. "Least recent use" refers to the destination pattern that has waited the longest since being selected. "Random selection" weights all of the destination patterns equally in a random selection mode.

Example

The following example configures the dial peers to hunt in the following order: (1) longest match in phone number, (2) explicit preference, (3) random selection.

```
configure terminal
 dial-peer hunt 0
```

Related Commands

Command	Description
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
preference	Specifies the preferred selection order of a dial peer within a hunt group.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer terminator

To change the character used as a terminator for variable length dialed numbers, use the **dial-peer terminator** global configuration command. Use the **no** form of this command to restore the default terminating character.

dial-peer terminator *character*
no dial-peer terminator

Syntax Description

character	Designates the terminating character for a variable-length dialed number. Valid numbers and characters are #, *, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, a, b, c, and d. The default is #.

Defaults

The default terminating character is #.

Command Mode

Global configuration

Command History

Release	Modification
12.0	This command was introduced.
12.0(7)XK	Usage was restricted to variable-length dialed numbers.

Usage Guidelines

There are certain areas in the world (for example, in certain European countries) where telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed-number, the system does not place a call until the configured value for the **timeouts interdigits** command has expired, or until the caller dials the terminating character. Use the **dial-peer terminator** global configuration command to change the terminating character.

Example

The following example specifies “9” as the terminating character for variable-length dialed numbers:

```
configure terminal
 dial-peer terminator 9#
```

Related Commands

Command	Description
answer-address	Specifies the preferred selection order of a dial peer within a hunt group.
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
timeouts interdigit	Specifies the interdigit timeout value for a voice port, in seconds.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer voice

To enter dial-peer configuration mode and specify the method of voice encapsulation, use the **dial-peer voice** global configuration command. Use the **no** form of this command to disable the selected encapsulation mode.

For the Cisco 2600 series:

dial-peer voice *tag* {**pots** | **voip** | **vofr**}
no dial-peer voice *tag*

For the Cisco 3600 series:

dial-peer voice *tag* {**pots** | **voip** | **voatm** | **vofr** }
no dial-peer voice *tag*

For the Cisco MC3810 series:

dial-peer voice *tag* {**pots** | **voip** | **voatm** | **vofr** }
no dial-peer voice *tag*

Syntax Description

tag	A number identifying a particular dial peer. Valid entries are 1 to 2147483647.
pots	POTS dial peer using basic telephone service.
voip	VoIP dial peer using voice encapsulation on the POTS network.
voatm	(Cisco 3600 and MC3810 only) Voice over ATM dial peer using real-time AAL5 voice encapsulation on the ATM backbone network.
vofr	Voice over Frame Relay dial peer using encapsulation on the Frame Relay backbone network.

Defaults

No default behavior or values.

Command Mode

Global configuration

Command History

Release	Modification
11.3(1)T	This command was first introduced.
11.3(1)MA	This command was first supported on the Cisco MC3810, with support for POTS, VoFR, and VoATM.
12.0(3)XG and 12.0(4)T	This command added VoFR to the Cisco 2600 and 3600 series routers.
12.0(4)T	This command added VoFR to the Cisco 7200 series platform.
12.0(7)XK	This command added VoIP to the Cisco MC3810 and VoATM to the Cisco 3600 series routers.

Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global configuration mode.

Example

The following example accesses dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
configure terminal
 dial-peer voice 10 pots
```

Related Commands

Command	Description
voice-port	Enters voice-port configuration mode.

ds0-group

To specify the DS0 timeslots that make up a logical voice port on a T1 or E1 controller, and to specify the signaling type, use the **ds0-group** controller configuration command. Use the no form of the command to remove the DS0 group and signaling setting.

ds0-group *ds0-group-no* **timeslots** *timeslot-list* **type** *signal-type*

no ds0-group *ds0-group-no*

Syntax Description

ds0-group-no	A value from 0 to 23 that identifies the DS0 group.
timeslot-list	*timeslot-list* is a single timeslot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1, allowable values are from 1 to 24. Examples are: • 2 • 1-15, 17-24 • 1-23 • 2, 4, 6-12
type	The signaling method selection for **type** 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 PBXs. The FXO interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations. The FXO interface is often used for off-premises extensions. The options are as follows: • **e&m-immediate-start—**no specific off-hook and on-hook signaling • **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-wink-start—**the originating endpoint sends an off-hook signal and waits for a wink signal from the destination • **fxs-ground-start—**Foreign Exchange Station ground-start signaling support • **fxs-loop-start —**Foreign Exchange Station loop-start signaling support • **fxo-ground-start—**Foreign Exchange Office ground-start signaling support • **fxo-loop-start—**Foreign Exchange Office loop-start signaling support

The following options are available only on E1 controllers on the Cisco MC3810:

- **e&m-melcas-immed**—E&M Mercury Exchange Limited Channel Associated Signaling (MELCAS) immediate start signaling support
- **e&m-melcas-wink**—E&M MELCAS wink start signaling support
- **e&m-melcas-delay**—E&M MELCAS delay start signaling support
- **fxo-melcas**—MELCAS Foreign Exchange Office signaling support
- **fxs-melcas**—MELCAS Foreign Exchange Station signaling support

The following options are available only when the **mode ccs** command is enabled on the Cisco MC3810 for transparent CCS support:

- **ext-sig-master—**For the specified channel(s), automatically generates the off-hook signal and stays in the off-hook state.
- **ext-sig-slave—**For the specified channel(s), automatically generates the answer signal when a call is terminated to that channel.

Default

No DS0 group is defined.

Command Mode

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as **cas-group**.
12.0(1)T	The **cas-group** command was first supported on the Cisco 3600 series.
12.0(5)T	This command was renamed **ds0-group** on the Cisco AS5300 and on the Cisco 2600 and 3600 series (requires Digital T1 Packet Voice Trunk Network Modules).
12.0(7)XK	Support for this command was extended to the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and is no longer supported.

Usage Guidelines

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

Cisco 2600 and 3600 series:

slot/*port*:*ds0-group-no*.

Cisco MC3810:

slot:ds0-group-no

On the Cisco MC3810, the *slot* number is the controller number. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

On the Cisco MC3810 when configured for transparent CCS, the channel type configured as the **ext-sig-master** is considered the master side of the permanent virtual circuit (PVC) connection which is responsible for establishing the PVC connection. After the master channel is configured, a fixed timer of 30 seconds starts. The voice-signaling driver then generates an off-hook signal on the master voice channel after the timer expires. The call is treated as a regular call, and the master channel does not hang up after the connection is made. If the call does not go through, or if the T1/E1 trunk is down, the 30-second timer on the master channel side restarts. A new off-hook signal is then generated at the master channel side after the timer expires.

Examples

The following example configures ranges of T1 controller timeslots for FXS ground-start and FXO loop-start signaling on a Cisco 2600 or 3600 Series router:

```
router(config)# controller T1 1/0
router(config-controller)# framing esf
router(config-controller)# linecode b8zs
router(config-controller)# ds0-group 1 timeslot 1-10 type fxs-ground-start
router(config-controller)# ds0-group 2 timeslot 11-24 type fxo-loop-start
```

The following example configures DS0 groups 1 and 2 on controller T1 1 on the Cisco MC3810 to support transparent CCS:

```
router(config)# controller T1 1
router(config-controller)# mode ccs cross-connect
router(config-controller)# ds0-group 1 timeslot 1-10 type ext-sig-master
router(config-controller)# ds0-group 2 timeslot 11-24 type ext-sig-slave
```

Related Command

Command	Description
codec complexity	Matches the DSP complexity packaging to the codec(s) to be supported
mode ccs	Configures the T1/E1 controller to support CCS cross-connect or CCS frame-forwarding.

dtmf-relay

Use the **dtmf-relay** command to specify how an H.323 gateway relays DTMF tones through an IP network. Options allow the gateway to forward tones “out-of-band”, or separate from the voice stream. The **no** form of this command removes all signaling options and transmits the DTMF tones as part of the audio stream.

dtmf-relay [**cisco-rtp**] [**h245-signal**] [**h245-alphanumeric**]
no dtmf-relay

Syntax Description

cisco-rtp	(Optional) Forwards DTMF tones using RTP protocol with a Cisco proprietary payload type.
h245-signal	(Optional) Forwards DTMF tones using the H.245 “signal” User Input Indication method. Supports tones 0-9, *, #, and A-D.
h245-alphanumeric	(Optional) Forwards DTMF tones using the H.245 “alphanumeric” User Input Indication method. Supports tones 0-9, *, #, and A-D.

Default

DTMF tones are sent “inband”, or left in the audio stream, unless you use this command.

Command Mode

EXEC

Command History

Release	Modification
11.3(2) NA	This command was introduced.
12.0(5)T	This command was modified for H.323 V2, adding dtmf-relay and h245-signal.
12.0(7)XK	This command is supported on the Cisco MC3810

Usage Guidelines

The **dtmf-relay** command determines the outgoing format of relayed DTMF tones. The gateway automatically accepts all formats.

The gateway only sends DTMF tones in the format you specify if the remote device supports it. If the remote device supports multiple formats, the gateway chooses the format based on the following priority:

1. cisco-rtp (highest priority)
2. h245-signal
3. h245-alphanumeric
4. None – DTMF sent inband

Note The cisco-rtp version of dtmf-relay is a proprietary Cisco implementation and only interoperates between Cisco AS5300 universal access servers, Cisco 2600 or 3600 modular access routers, or Cisco MC3810 concentrators running Cisco IOS Release 12.0(7)XK, or later releases. Otherwise, the DTMF relay feature will not function and the gateway will send DTMF tones inband.

Note The **h245-alphanumeric** and **h245-signal** DTMF settings on an MC310 concentrator require a high-performance compression module (HCM) and are not supported on an MC3810 concentrator with a non-HCM voice compression module (VCM).

Example

The following are two examples of the **dtmf-relay** command:

- Configuring with dtmf-relay cisco-rtp or h245-signal when sending to dial-peer 103. Enter the configuration commands, one per line.

```
Router# configure terminal
Router(config)# dial-peer voice 103 voip
Router(config-dial-peer)# dtmf-relay cisco-rtp h245-signal
Router(config-dial-peer)# end
Router#
```

- Configuring the gateway to send DTMF inband (the default) when sending to dial-peer 103. Enter the configuration commands, one per line.

```
Router# configure terminal
Router(config)# dial-peer voice 103 voip
Router(config-dial-peer)# no dtmf-relay
Router(config-dial-peer)# end
```

Related Commands

Command	Description
dial-peer	Switch to the voice-port configuration mode form the global configuration mode.

forward-digits

To specify which digits to forward for voice calls, use the **forward-digits** dial-peer configuration command. If the **no** form of this command is entered, any digits not matching the destination-pattern are not forwarded. Use the **default** form of this command to restore the default state.

forward-digits {*num-digit* | **all** | **extra**}
no forward-digits
default forward-digits

Syntax Description

num-digit	The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. The valid range is 0 to 32. Setting the value to 0 is equivalent to entering **no forward-digits**.
all	Forward all digits. If **all** is entered, the full length of the destination pattern is used.
extra	If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length (ending with character T, for example: T, 123T, 123...T), extra digits are not forwarded.

Defaults

Dialed digits not matching the destination-pattern are forwarded.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1) MA	This command was first introduced on the Cisco MC3810.
12.0(2) T	The **implicit** option was added.
12.0(4) T	This command was modified to support ISDN PRI QSIG signaling calls.
12.0(7)XK	This command was first supported on the Cisco 2600 series and 3600 series platforms, the **implicit** keyword was removed, and the **extra** keyword was added.

Usage Guidelines

This command applies only to POTS dial peers.

Forwarded digits are always right-justified so that extra leading digits are stripped.

The destination pattern includes both explicit digits and wildcards, if present.

Use the **default** form of this command if a non-default digit-forwarding scheme was entered previously, and you wish to restore the default.

For QSIG ISDN connections, entering **forward-digits all** implies that all of the digits of the called party number are sent to the ISDN connection. When you enter **forward-digits** *num-digit* and enter a number from 1 to 32, the number of digits specified (right justified) of the called part number are sent to the ISDN connection.

Examples

The following example forwards all of the digits in the destination pattern of a POTS dial peer:

```
dial-peer voice 1 pots
  destination-pattern 8...
  forward-digits all
```

The following example forwards four of the digits in the destination pattern of a POTS dial peer:

```
dial-peer voice 1 pots
  destination-pattern 555....
  forward-digits 4
```

The following example forwards the extra right-justified digits that exceed the length of the destination pattern of a POTS dial peer:

```
dial-peer voice 1 pots
  destination-pattern 555....
  forward-digits extra
```

Related Commands

Command	Description
destination-pattern	Defines the prefix or the full E.164 telephone number to be used for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

huntstop

To disable all further dial-peer hunting if a call fails when using hunt groups, enter the **huntstop** dial-peer configuration command. To reenable dial-peer call hunting, enter the **no** form of this command.

huntstop
no huntstop

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	Support for this command was extended to the Cisco 2600 and 3600 series routers.

Usage Guidelines

After you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

This command can be used with all types of dial peers.

Examples

The following example shows how to disable dial-peer hunting on a specific dial peer:

```
router(config)# dial peer voice 100 vofr
router(config-dial-peer)# huntstop
```

The following example shows how to reenable dial-peer hunting on a specific dial peer:

```
router(config)# dial peer voice 100 vofr
router(config-dial-peer)# no huntstop
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

icpif

To specify the Impairment/Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** dial peer configuration command. Use the **no** form of this command to restore the default value for this command.

icpif *number*
no icpif *number*

Syntax Description

number	Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are from 0 to 55.

Default

The default value for this command is 30.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

This command is applicable only to VoIP peers.

Example

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

incoming called-number

To identify the service type for a call on a router handling both voice and modem calls, use the **incoming called-number** dial peer configuration command. To return to the default value, use the **no** form of this command.

incoming called-number *string*
no incoming called-number *string*

Syntax Description

string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Default

The default value for this command is no associated called number.

Command Mode

Dial peer configuration

Command History

Release	Modification
11.3NA	This command was introduced on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

When the Cisco MC3810 is handling both modem and voice calls, it needs to be able to identify the service type of the call—meaning whether the incoming call to the server is a modem or a voice call. When the access server handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the called number (DNIS). In a mixed environment, where the server receives both modem and voice calls, you need to identify the service type of a call by using the **incoming called-number** command.

If you do not use the **incoming called-number** command, the server attempts to resolve whether an incoming call is a modem or voice call based on the interface over which the call comes. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a dial peer, the call is assumed to be a voice call.

By default, there is no called number associated with the dial peer, which means that incoming calls will be associated with dial peers based on matching calling number with answer address, call number with destination pattern, or calling interface with configured interface.

This command applies to both VoIP and POTS dial peers.

Example

The following example configures calls coming in to the server with a called number of “3799262” as voice calls:

```
dial peer voice 10 pots
 incoming called-number 3799262
```

num-exp

To define a complete telephone number for an extension, use the **num-exp** global configuration command. Use the **no** form of this command to cancel a configured number expansion.

num-exp *extension-number expanded-number*
no num-exp *extension-number*

Syntax Description

extension-number	Digit(s) defining an extension number to be expanded.
expanded-number	Digit(s) defining the expanded telephone number or destination pattern.

Defaults

No number expansion is configured.

Command Mode

Global configuration

Command History

Release	Modification
11.3(1)T	This command was first introduced on the Cisco 3600 platform.
12.0(3)T	This command was first supported on the Cisco AS5300 platform.
12.0(4)XL	This command was first supported on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

Use the **num-exp** global configuration command to expand a set of numbers (for example, an extension number) into a destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing less than seven digits.

Use a period (.) as a variable or wild card, representing a single number. Use a separate period for each number you want to represent with a wildcard; if you want to replace four numbers in an extension with wildcards, type in four periods.

Example

The following example specifies that extension number 55541 be expanded to 14085555541:

```
num-exp 55541 14085555541
```

The following example specifies that all five-digit extensions beginning with 5 be expanded to 1408555 . . . .

```
num-exp 5.... 1408555....
```

Related Commands

Command	Description
forward-digits	Specifies which digits to forward for voice calls.
prefix	Specifies a prefix for a dial peer.
dial-peer terminator	Change the character used as a terminator for variable length dialed numbers.

session target

To configure a network-specific address for a dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

Cisco MC3810 Voice over IP:

```
session target {ipv4:destination-address | dns:[$s$. | $d$. | $e$. | $u$.] host-name |
loopback:rtp | loopback:compressed | loopback:uncompressed}

no session target
```

Syntax Description

For the Cisco MC3810 Voice over IP:

ipv4:*destination-address*	IP address of the dial peer.
dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **e.**—Indicates that the digits in the called number will be reversed, periods will be added in-between each digit of the called number, and that this string will be used as part of the domain name. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped back to the originating source. This is applicable for VoIP peers.
loopback:compressed	Indicates that all voice data will be looped back in compressed mode to the originating source. This is applicable for POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers.

Defaults

Enabled with no IP address or domain name defined.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1) T	This command was first introduced.
11.3(1) MA	Support was added for VoFR,VoATM and VoHDLC dial peers on the Cisco MC38110.
12.0(3) XG and 12.0(4)T	The *cid* option was added. Support was added for VoFR dial peers on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	Support was added for VoATM dial peers on the Cisco 3600 series routers. Support was added for VoIP dial peers on the Cisco MC3810. Support for VoHDLC on the Cisco MC3810 was removed in this release.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

Use the **session target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial peer session targets you need to configure if you have groups of numbers associated with a particular router.

Examples

The following example configures a session target using DNS for a host, “voice_router,” in the domain “cisco.com”:

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using DNS, with the optional **u**. wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310222. The optional wildcard **u.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is “cisco.com.”

```
dial-peer voice 10 voip
 destination-pattern 1310222....
 session target dns:$u$.cisco.com
```

The following example configures a session target using dns, with the optional **d.** wildcard. In this example, the destination pattern has been configured for 13102221111. The optional wildcard **d**. indicates that the router will use the destination pattern to identify the dial peer in the “cisco.com” domain.

```
dial-peer voice 10 voip
 destination-pattern 13102221111
 session target dns:$d$.cisco.com
```

The following example configures a session target using DNS, with the optional **e.** wildcard. In this example, the destination pattern has been configured for 12345. The optional wildcard **e.** indicates that the router will reverse the digits in the destination pattern, add periods between the digits, and then use this reverse-exploded destination pattern to identify the dial peer in the "cisco.com" domain.

```
dial-peer voice 10 voip
 destination-pattern 12345
 session target dns:$e$.cisco.com
```

Related Commands

Command	Description
called-number	Enables an incoming VoFR call leg to be bridged to the correct POTS call leg.
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
cptone	Specifies a regional tone, ring, and cadence setting for an analog voice port.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
dtmf-relay	Enables the DSP to generate FRF.11 Annex A frames for a dial peer.
preference	Indicates the preferred selection order of a dial peer within a hunt group.
session protocol	Establishes a VoFR protocol for calls between the local and the remote routers via the packet network.

show call active voice

To show the active call table, use the **show call active voice** privileged EXEC command.

show call active voice

Syntax Description

This command has no arguments or keywords.

Command Mode

User EXEC and Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 series and 3600 series.
12.0(3)XG	Support for VoFR was added.
12.0(4)T	This command was first supported on the Cisco 7200 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

This command applies to Voice over IP, Voice over Frame Relay, and Voice over ATM on the Cisco 2600 series, 3600 series, and MC3810 series.

Use this command to display the contents of the active call table, which shows all of the calls currently connected through the router. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information.

See Table 2 for a listing of the information types associated with this command.

Example

The following is sample output from the **show call active voice** command:

```
router# show call active voice
GENERIC: SetupTime=21072 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 ConnectTime=0 CallState=3 CallOrigin=2 ChargedUnits=0
InfoType=0 TransmitPackets=375413 TransmitBytes=7508260 ReceivePackets=377734
ReceiveBytes=7554680

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
```

```
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=+14085271001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

TELE: ConnectionId=[0x19BDF910 0xAF500007 0x0 0x58ED0] TxDuration=16640
VoiceTxDuration=16640 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=4
OutSignalLevel=-440 InSignalLevel=-440 InfoActivity=2 ERLLevel=227
SessionTarget=
```

Table 2 provides an alphabetical listing of the fields in this output and a description of each field.

Table 2 Show Call Active Voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during the call.
ConnectionId	Global call identifier of a gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer transmitting this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during this call.
Index	Dial peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the call.

Table 2 Show Call Active Voice Field Descriptions (continued)

Field	Description
NoiseLevel	Active noise level for the call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer.
PeerSubaddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone during the call.
SelectedQoS	Selected quality of service (QoS) for the call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetupTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during the call.
TransmitPackets	Number of packets transmitted from this peer during the call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show call history voice

To display the call history table, use the **show call history voice** privileged EXEC command.

show call history voice [**last** *number* | **brief**]

Syntax Description

last *number*	(Optional) Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* Valid entries for the argument *number* are numbers from 1 to 2147483647.
brief	(Optional) Displays abbreviated call history information for each leg of a call.

Command Mode

User EXEC and Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(3)XG	Support for VoFR was added on the Cisco 2600 and 3600 series.
12.0(4)T	The **brief** keyword was added and the command was first supported on the Cisco 7200 series.
12.0(7)XK	Support for the **brief** keyword was added on the Cisco MC3810 platform.

Usage Guidelines

This command applies to all voice applications on the Cisco 2600 series, 3600 series, MC3810, and 7200 series platforms.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all voice calls connected through this router in descending time order. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number.* To display a shortened version of the call history table, use the keyword **brief**.

Example

The following is sample output from the **show call history voice** command for a VoFR call using the frf11-trunk session protocol:

```
router# show call history voice last 1
   GENERIC:
   SetupTime=8283963 ms
   Index=3149
   PeerAddress=3623110
   PeerSubAddress=
   PeerId=3400
   PeerIfIndex=18
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283963
   DisconectTime=8285463
   CallOrigin=1
   ChargedUnits=0
   InfoType=2
   TransmitPackets=94
   TransmitBytes=2751
   ReceivePackets=0
   ReceiveBytes=0

   VOFR:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   Subchannel=[Interface Serial0/0, DLCI 160, CID 10]
   SessionProtocol=frf11-trunk
   SessionTarget=Serial0/0 160 10
   CalledNumber=2603100
   VADEnable=ENABLED
   CoderTypeRate=g729r8
   CodecBytes=30
   SignalingType=cas
   DTMFRelay=DISABLED
   UseVoiceSequenceNumbers=DISABLED

   GENERIC:
   SetupTime=8283963 ms
   Index=3150
   PeerAddress=2601100
   PeerSubAddress=
   PeerId=1100
   PeerIfIndex=7
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283964
   DisconectTime=8285464
   CallOrigin=2
   ChargedUnits=0
   InfoType=2
   TransmitPackets=0
   TransmitBytes=-121
   ReceivePackets=94
   ReceiveBytes=2563
   TELE:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   TxDuration=15000 ms
   VoiceTxDuration=2010 ms
   FaxTxDuration=0 ms
   CoderTypeRate=g729r8
   NoiseLevel=-68
```

```
ACOMLevel=20
SessionTarget=
```

The following is sample output from the **show call history voice** command for a VoIP call:

```
router# show call history voice
   GENERIC:
   SetupTime=20405
   Index=0
   PeerAddress=
   PeerSubAddress=
   PeerId=0
   PeerIfIndex=0
   LogicalIfIndex=0
   DisconnectCause=NORMAL
   DisconnectText=
   ConnectTime=0
   DisconectTime=20595
   CallOrigin=2
   ChargedUnits=0
   InfoType=0
   TransmitPackets=0
   TransmitBytes=0
   ReceivePackets=0
   ReceiveBytes=0

   VOIP:
   ConnectionId[0x19BDF910 0xAF500006 0x0 0x56590]
   RemoteIPAddress=17635075
   RemoteUDPPort=16392
   RoundTripDelay=0
   SelectedQoS=0
   SessionProtocol=1
   SessionTarget=
   OnTimeRvPlayout=0
   GapFillWithSilence=0
   GapFillWithPrediction=0
   GapFillWithInterpolation=0
   GapFillWithRedundancy=0
   HiWaterPlayoutDelay=0
   LoWaterPlayoutDelay=0
   ReceiveDelay=0
   VADEnable=0
   CoderTypeRate=0

   TELE: ConnectionId=[0x19BDF910 0xAF500006 0x0 0x56590]
   TxDuration=3030
   VoiceTxDuration=2700
   FaxTxDuration=0
   CoderTypeRate=0
   NoiseLevel=0
   ACOMLevel=0
   SessionTarget=
```

Table 3 provides an alphabetical listing of the fields in this output and a description of each field.

Table 3 Show Call History Voice Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time the call was connected.
DisconnectCause	Description explaining why the call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time the call was disconnected.
FaxDuration	Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during the voice call.
Index	Index number identifying the voice-peer for this call.
InfoType	Information type for this call.
LogicalIfIndex	Index of the logical voice port for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number to which this call is connected.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call.
PeerSubAddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during the call.

Table 3 Show Call History Voice Field Descriptions (continued)

Field	Description
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected quality of service for the call.
SessionProtocol	Session protocol to be used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetUpTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted by this peer during the call.
TransmitPackets	Number of packets transmitted by this peer during the call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show num-exp

To show the number expansions configured, use the **show num-exp** privileged EXEC command.

show num-exp [*dialed-number*]

Syntax Description

dialed-number	(Optional) Dialed number.

Command Mode

User EXEC and Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was first introduced on the Cisco 3600 platform.
12.0(3)T	This command was first supported on the Cisco AS5300 platform.
12.0(4)XL	This command was first supported on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

This command applies to VoFR, VoATM, and Voice over IP on the Cisco 2600 series, 3600 series, and MC3810 platforms.

Use the **show num-exp** privileged EXEC command to display all of the number expansions configured for this router. To display number expansion for only one number, specify that number by using the *dialed-number* argument.

Example

The following is sample output from the **show num-exp** command:

```
router# show num-exp
Dest Digit Pattern = '0...'     Translation = '+14085270...'
Dest Digit Pattern = '1...'     Translation = '+14085271...'
Dest Digit Pattern = '3..'      Translation = '+140852703..'
Dest Digit Pattern = '4..'      Translation = '+140852804..'
Dest Digit Pattern = '5..'      Translation = '+140852805..'
Dest Digit Pattern = '6....'    Translation = '+1408526....'
Dest Digit Pattern = '7....'    Translation = '+1408527....'
Dest Digit Pattern = '8...'     Translation = '+14085288...'
```

Table 4 explains the fields in the sample output.

Table 4 Show Num-Exp Voice Field Descriptions

Field	Description
Dest Digit Pattern	Index number identifying the destination telephone number digit pattern.
Translation	Expanded destination telephone number digit pattern.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information for dial peers.
show voice port	Displays configuration information about a specific voice port.

voice class codec

To enter voice-class configuration mode and assign an identification tag number for a codec voice class, use the **voice class codec** global configuration command. Use the **no** form of this command to delete a codec voice class.

voice class codec *tag*
no voice class codec *tag*

Syntax Description

tag	The unique number you assign to the voice class. The valid range is 1 to 10000. Each tag number must be unique on the router.

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was first supported on the Cisco 2600 and 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

This command only creates the voice class for codec selection preference, and assigns an identification tag. Use the **codec preference** command to specify the parameters of the voice class, and use the **voice-class codec** dial-peer command to apply the voice class to a VoIP dial peer.

Note The **voice class codec** command in global configuration mode is entered without the hyphen. The **voice-class codec** command in dial-peer configuration mode is entered with the hyphen.

Example

The following example shows how to enter voice-class configuration mode and assign a voice class tag number starting from global configuration mode:

```
router(config)# voice class codec 10
router(config-class)#
```

After you enter voice-class configuration mode for codecs, use the **codec preference** command to specify the parameters of the voice class.

Related Commands

Command	Description
codec preference	Defines the order of preference in which network dial peers select codecs.
voice-class codec (dial-peer)	Assigns a previously-configured codec selection preference list to a dial peer.

voice-class codec (dial-peer)

To assign a previously-configured codec selection preference list (codec voice class) to a VoIP dial peer, enter the **voice-class codec** dial-peer configuration command. Enter the **no** form of this command to remove the codec preference assignment from the dial peer.

voice-class codec *tag*
no voice-class codec *tag*

Syntax Description

tag	The unique number assigned to the voice class. The valid range for this tag is 1 to 10000. The *tag* number maps to the tag number created using the **voice class codec** global configuration command.

Defaults

Dial peers have no codec voice class assigned.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was first supported on the Cisco 2600 and 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Note The **voice-class codec** command in dial-peer configuration mode is entered with a hyphen. The **voice class codec** command in global configuration mode is entered without the hyphen.

Examples

The following example shows how to assign a previously-configured codec voice class to a dial peer:

```
router(config)# dial-peer voice 100 voip
router(config-dial-peer)# voice-class codec 10
```

Related Commands

Command	Description
codec preference	Defines the order of preference in which network dial peers select codecs.
voice class codec	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class.
show dial-peer voice	Displays the configuration for all dial peers configured on the router.

voice-group

This command was added in Cisco IOS Release 11.3(1)MA on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.