Integrated Data, Voice, and Video Services for ISDN Interfaces

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The Integrated Data, Voice, and Video Services for ISDN Interfaces feature allows multimedia communications between H.320 endpoints and H.323 or Skinny Client Control Protocol (SCCP) endpoints.

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the "Feature Information for Integrating Data, Voice, and Video for ISDN Interfaces" section on page 97.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

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Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Prerequisites for Configuring Integrated Data, Voice, and Video Services for ISDN Interfaces

Before you configure integrated services using H.320 protocol, you must do the following:

- Ensure that you have a Cisco IOS image that supports this feature. Access Cisco Feature Navigator.
- Establish a working H.323 network for voice calls or a network using Cisco Unified CallManager Express with SCCP endpoints.
- Perform basic ISDN voice configuration. For more information, see *Configuring ISDN PRI Voice-Interface Support*.
- Ensure that the ISDN layer is up. Use the **show isdn status** command to display the current status of each ISDN layer.
- Set T1/E1 clocking. Use the **network-clock-select** command to name a source to provide timing for the network clock and to specify the selection priority for this clock source.

Supported Routers, Hardware Modules, Codecs, Endpoints, and Topologies

- This feature supports the following routers:
  - Cisco 2600XM
  - Cisco 2800 series
  - Cisco 3700 series
  - Cisco 3800 series
- This feature supports the following hardware modules:
  - NM-HDV2
  - NM-HD-xx
  - Onboard DSP module
  - VIC2-2BRI
  - VWIC-xMFT-x
  - VWIC2-xMFT-x
- This feature supports the following video codecs:
  - ITU-T Recommendation H.261
  - ITU-T Recommendation H.263
  - ITU-T Recommendation H.263+
  - ITU-T Recommendation H.264 (only Annex A packetization is supported)
- This feature supports the following ITU-T RecommendationH.320 endpoints:
  - Polycom
  - Tandberg

Supported Topologies

Integrated services for ISDN BRI and PRI interfaces allows multimedia communications between H.320 endpoints and ITU-T Recommendation H.323 or SCCP endpoints, including the following topologies:

- Bridge an H.320 endpoint (terminal) and an H.323 endpoint (terminal)

H.323 endpoint > H.320 gateway > BRI or PRI interface > H.320 endpoint

- Bridge an SCCP endpoint and an H.320 endpoint

  SCCP endpoint > H.320 gateway > BRI or PRI interface > H.320 endpoint

- Cisco CME video survivability

  SCCP endpoint > H.320 gateway > BRI or PRI interface > H.320 gateway > SCCP endpoint

- H.320 endpoint > IP network > H.320 endpoint
- H.320 endpoint > SCCP endpoint > H.320 endpoint
- Videoconferencing offload to the ISDN network

  H.323 endpoint > H.320 gateway > BRI or PRI interface > H.320gateway > H.323 endpoint

Restrictions for Configuring Integrated Data, Voice, and Video Services for ISDN Interfaces

Restrictions for configuring integrated services for ISDN interfaces are as follows:

- If the minimum bandwidth is not available for a video call, the call falls back to audio-only.
- This feature is supported only for C5510 DSP-based platforms.
- H.320 calls are limited to 16 B-channels.
- ISO-13871 bonding is **not** supported for H.320 calls with the initial release of the H.320 feature. When connected to third party H.320 devices that require ISO-13871 bonding, only 128k (2B) calls are supported. Support for ISO-13871 bonding is available starting with Release 12.4(20)T.

Information About Integrated Data, Voice, and Video Services for ISDN Interfaces

Integrated data, voice, and video services through a single ISDN interface allows multimedia communications between H.320 endpoints and H.323 or SCCP endpoints. Before you configure integrated services for ISDN interfaces, you should be familiar with the following concepts:

- Integrated Services Mode, page 4
- Primary and Secondary Incoming H.320 Calls, page 4
- Dynamic and Static H.320 Secondary Called Numbers, page 5
- Video Information Type, page 5
- Bandwidth for H.320 Calls, page 6

Integrated Services Mode

An ISDN interface must be configured for integrated services mode to enable H.320 primary and secondary call type checking. Enabling integrated services allows data, voice, and video call traffic to occur from a single ISDN BRI or PRI interface. When an interface is in integrated service mode:

- ISDN performs call type checking for the incoming call. The call is rejected by ISDN if no voice or data dial peer is matched for an incoming call.
- The **voice** option for the **isdn incoming-voice** command, which causes all calls to bypass the modem and be handled as voice, is not available.

By default, the integrated services option is disabled from the supported interfaces.

Primary and Secondary Incoming H.320 Calls

An H.320 call consists of 1 to 16 ISDN B-channels. The first B-channel in an H.320 session is the primary B-channel and all additional B-channels are handled as secondary B-channels. Secondary B-channels are distinguished from primary B-channels by the call number received in the Q.931 ISDN setup message. The secondary called numbers for H.320 B-channels can be exchanged between the terminals using H.242 format (dynamic method), or can be configured statically (static method).

An H.320 primary B-channel is different from the secondary B-channels in the following ways:

- A primary B-channel is the first ISDN call made in an H.320 call.
- The primary B-channel always carries voice. Depending on the audio codec selected, the remaining available bandwidth is used for video.
- The primary B-channel carries the H.221 in-band-signaling. The secondary B-channels also contain bit-rate allocation signal (BAS), and only the appropriate values for a secondary leg. For more information on values for secondary B-channels, see *ITU H.221 Annex A, Table A-5*.
- Only the primary call with each H.320 session is passed to the session application. Secondary B-channels are handled by the H.320 B-channel aggregator.
- Secondary B-channels only provide more B-channels for additional video bandwidth.

During inbound dial peer matching, the list of H.320 sessions is searched before the incoming voice dial peer lookup. If the new called number matches a called number associated with an existing H.320 call session (dynamic or static), the leg is added to the existing H.320 call session as a secondary B-channel.

The B-channel aggregator is responsible for handling call setup of additional B-channels for H.320 calls. It also allocates dynamic called numbers from the voice class called number pool to the gateway and frees them back up again.

The B-channel aggregator creates a video conferencing session for individual incoming H.320 primary calls. The setup and teardown of each B-channel is handled as one independent call on the ISDN side, which means that each H.320 call can have multiple B-channels. On the H.323 side, only one call is presented to the endpoint. For this reason, multiple ISDN calls are grouped together to form one logical H.320 to H.323 call. The H.320 B-channel aggregator provides this function.

Dynamic and Static H.320 Secondary Called Numbers

A called number is a digit string that can be matched by an incoming or outgoing call to associate the call with a dial peer. From the originating gateway, a set of unique incoming called numbers can be allocated for an incoming H.320 primary call to the originating H.320 terminal. The allocated incoming called numbers are associated with one active H.320 session and used by the originating H.320 terminal as dialing numbers to initiate the H.320 secondary calls.

To connect secondary B-channels into an H.320 call, additional called numbers might be needed if each leg has a called number different from the primary. This is accomplished using either dynamic or static secondary dial plans.

- With a dynamic dial plan, which uses H.242, additional numbers are allocated from the called number pool referenced from the voice port.
- With a static dial plan, the called numbers are defined on the gateway.

Dynamic Called Numbers

A called number pool is a group of dynamic called numbers to be referenced by the gateway for handling primary and secondary calls. If the originating H.320 terminal supports receiving dynamic secondary called numbers (H.242), the H.320 leg aggregator module allocates the idle called numbers from a pool referenced by the voice interface on the originating gateway for the H.320 primary call. The number of dynamic called numbers to be allocated is based on the bandwidth requirement of the incoming H.320 session.

Static Called Numbers

Static called numbers are configured for H.320 endpoints that are not capable of receiving dynamic secondary calling numbers (non-H.242). The static called numbers are referenced by the incoming and outgoing POTS dial peers. Up to 15 called numbers (in E.164 format) can be configured as static called numbers to match the incoming H.320 secondary calls.

Video Information Type

When a dial peer is created, the default information type is voice. To enable H.320 call support, you must configure a video information type on the POTS dial peer for inbound dial peer matching.

A POTS dial peer configured with a video information-type is marked as a specific type of voice dial peer. During the ISDN call type checking for an incoming H.320 call, the matching of voice dial peers with video information-type takes precedence over the matching of voice dial peers with other information type settings. Outgoing H.320 primary calls are initiated by the default application by matching an outbound POTS dial peer with a video information type.

An incoming POTS dial peer with a video information type provisions for incoming H.320 primary calls using the incoming called-number.

Bandwidth for H.320 Calls

Each c5510 digital signal processor (DSP) channel supports 64 kilobits of bandwidth. Each c5510 DSP has 16 channels available. One of those channels can support a bandwidth of 1024 kbps, allowing the DSP to support one H.320 call with a maximum of 16 B-channels. For each dial peer configured for information-type video, an optional **bandwidth** command can be added that specifies the minimum acceptable and maximum allowed bandwidth for the H.320 call, in 64-kbit increments. If the number of call legs connected falls between the minimum and maximum configured, then video is allowed. If the minimum bandwidth cannot be met for the call, the call drops back to an audio-only H.320 call.

How to Configure Integrated Data, Voice, and Video Services for ISDN Interfaces

This section describes how to configure integrated data, voice, and video services for ISDN BRI or PRI interfaces, and includes the following tasks:

- Enabling Integrated Services on the Interface, page 6 (required)
- Configuring ISDN Inbound POTS Dial Peers, page 8 (required)
- Configuring the Voice Class Codec, page 9 (required)
- Configuring the VoIP Dial Peer, page 11 (required)

Enabling Integrated Services on the Interface

Enabling integrated services allows video and voice call traffic to occur from ISDN BRI or PRI interfaces simultaneously.

When an interface is in integrated service mode:

- ISDN performs call type checking for the incoming call. The call is rejected by ISDN if no voice or data dial peer is matched for an incoming call.
- The **voice** option for the **isdn incoming-voice** command, which handles all incoming calls as if they are voice calls, is not available.

By default, the integrated service option is disabled from the supported interfaces. Use the following procedure to enable integrated mode on a serial interface.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **interface serial** *slot***/***port***:***timeslot*
4. **shutdown**
5. **isdn integrate calltype all**
6. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**interface serial** *slot***/***port***:***timeslot* **Example:** `Router(config)# interface serial4/1:15`	Specifies a serial interface for ISDN PRI common-channel signaling and enters interface configuration mode.
Step 4	**shutdown** **Example:** `Router(config-if)# shutdown`	Shuts down the interface.
Step 5	**isdn integrate calltype all** **Example:** `Router(config-if)# isdn integrate calltype all`	Enables the serial interface for integrated mode, which allows data and voice call traffic to occur simultaneously. **Note** This configuration disables the *voice* option for the **isdn incoming-voice** command on the interface.
Step 6	**no shutdown** **Example:** `Router(config-if)# no shutdown`	Returns the interface to the active state.

Examples

In the following example, the interface is shut down.

```
Router(config)# interface Serial4/1:15
Router(config-if)# shutdown
```

This example shows that integrated mode is enabled.

```
Router(config)# interface Serial4/1:15
Router(config-if)# isdn integrate calltype all
% This command line will enable the Serial Interface to "integrated service" mode.
% The "isdn incoming-voice voice" setting will be removed from the interface.
% Continue? [confirm]
```

When you confirm, the default incoming-voice configuration is removed from the interface, and the interface is now in integrated service mode. The interface does not reset back to voice mode if an incoming call is originated from the interface.

This example show the interface being set to active again.

```
Router(config)# interface Serial4/1:15
Router(config-if)# no shutdown
```

Configuring ISDN Inbound POTS Dial Peers

Use the following procedure to configure the inbound POTS dial peer for an ISDN interface.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **incoming called-number** *string*
5. **direct-inward-dial**
6. **information-type** [**fax** | **video** | **voice**]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 12 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode. • *tag*—Identifier for the dial peer. The range is 1 to 2147483647. • **pots**—Indicates a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**`incoming called-number`** *`string`* **Example:** `Router(config-dial-peer)# incoming called-number 408`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer. • *string*—Incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number. The default is the calling number pattern.

	Command or Action	Purpose
Step 5	**`direct-inward-dial`** **Example:** `Router(config-dial-peer)# direct-inward-dial`	Enables the direct inward dialing (DID) call treatment for an incoming called number.
Step 6	**`information-type`** `[`**`fax`** `|` **`video`** `|` **`voice`**`]` **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial peer configuration mode.

Examples

```
dial-peer voice 12 pots
 information-type video
 incoming called-number 408
 direct-inward-dial
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

What to Do Next

To configure a voice class codec, continue with the "Configuring the Voice Class Codec" section on page 9. If a voice class codec is already configured, or if you plan reference a video codec on the dial peer, proceed to the "Configuring the VoIP Dial Peer" section on page 11.

Configuring the Voice Class Codec

Use this procedure to configure a voice class codec, to be referenced by the VoIP dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class codec** *tag*
4. **codec preference** *value codec-type* [**bytes** *payload-size*]
5. **video codec** [**h261** | **h263** | **h263+** | **h264**]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class codec`** *`tag`* **Example:** `Router(config)# voice class codec 10`	Enters voice-class configuration mode and assigns an identification tag number for a voice class codec. • *tag*—Identifier for the voice class. Range is 1 to 10000. There is no default.
Step 4	**`codec preference`** *`value codec-type`* `[`**`bytes`** *`payload-size`*`]` **Example:** `Router(config-class)# codec preference 1 g722`	Specifies a list of preferred audio codecs to use on a dial peer. • *value*—Order of preference. The range is 1 (most preferred) to 14 (least preferred). • *codec-type*—Preferred codec. **Note** You can configure multiple codec types with different preferences for a voice class. **Note** We recommend codec G.722 for filtering H.320 calls. See the CLI help for the complete list of codec types. • **bytes** *payload-size*—(Optional) Size of the voice frame in bytes and the number of bytes in the voice payload of each frame. Values depend on the codec type and the packet voice protocol.
Step 5	**`video codec`** `[`**`h261`** `|` **`h263`** `|` **`h263+`** `|` **`h264`**`]` **Example:** `Router(config-class)# video codec h263`	Specifies a list of preferred video codecs. **Note** You can configure multiple video codecs for a voice class. • **h261**—Video codec H.261 • **h263**—Video codec H.263 • **h263+**—Video codec H.263+ • **h264**—Video codec H.264

Example

Multiple video codecs can be defined to a voice class codec, as shown in the following example.

```
voice class codec 10
 codec preference 1 g722
 codec preference 2 g711alaw
```

```
video codec h261
video codec h263
video codec h264
```

Configuring the VoIP Dial Peer

Use the following procedure to configure the inbound or outbound VoIP dial peer.

Restrictions

Restrictions for configuring the VoIP dial peer are as follows:

- You can assign a previously defined voice class codec or a video codec to a VoIP dial peer. When adding a codec to the VoIP dial peer configuration, this does not mean that the specific codec is selected. It only means that the gateway filters the video codec capabilities passing through the gateway, in both directions.

  Note Audio codec commands, configured in the voice class codec, can also be used for filtering audio codecs.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **incoming called number** *string* (incoming dial peer)

   or

   destination pattern [+] *string* [T] (outgoing dial peer)
5. **voice-class codec** *tag*

   or

   video codec [**h261** | **h263** | **h263+** | **h264**]
6. **rtp payload-type** [cisco-codec-video-h264+ | cisco-codec-video-h264] [*number*]

DETAILED STEPS

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

	Command or Action	Purpose
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 12 voip`	Enters dial-peer configuration mode for a specific dial peer.

Step 4	**incoming called number** *string*
	
	or
	
	destination pattern [**+**] *string* [**T**]
	
	Example:
	`Router(config-dial-peer)# incoming called number 408`
	
	or
	
	Example:
	`Router(config-dial-peer)# destination-pattern 4085550100`
		Specifies a digit string that can be matched by an incoming call to associate the call with an incoming dial peer.
		• *string*—Incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number. The default is the calling number pattern.
		or
		Specifies either the prefix or the full E.164 telephone number to be used for an outgoing dial peer.

Step 5	**voice class codec** *tag*
	
	or
	
	video codec [**h261** | **h263** | **h263+** | **h264**]
	
	Example:
	`Router(config-dial-peer)# voice class codec 10`
	
	or
	
	Example:
	`Router(config-dial-peer)# video codec h261`
		(Optional) Assigns a previously defined voice class codec to this VoIP dial peer.
		• *tag*—Identifier for the voice class codec.
		or
		Defines a video codec for the VoIP dial peer to be used for H.320 call setup.
		• **h261**—Video codec H.261
		• **h263**—Video codec H.263
		• **h263+**—Video codec H.263+
		• **h264**—Video codec H.264
		Note Assign either a voice class codec or a video codec to a dial peer, and not both.

Step 6	**rtp payload-type** [**cisco-codec-video-h263+** | **cisco-codec-video-h264**] [*number*] **Example:** `Router(config-dial-peer)# rtp payload-type cisco-codec-video-h264`	(Optional. Only available if H.263+ or H.264 video codecs are configured.) Defines the RTP payload type for this dial peer. • **cisco-codec-video-h263+**—RTP video codec H.263+ payload type. • **cisco-codec-video-h264**—RTP video codec H.264 payload type. • *number*—Value for the RTP payload type. The dynamic range is 96 to 127. – Default payload-type for H.263+ video codec is 118. – Default payload-type for H.264 video codec is 119.

Examples

```
dial-peer voice 12 voip
 destination-pattern 4085550100
 video codec h263+
 rtp payload-type 118

dial-peer voice 12 voip
 shutdown
 incoming called-number 408
 voice-class codec 10
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

What to Do Next

Configure a secondary call dial plan, for both H.242 (dynamic) and nonH.242 endpoints (static) using one or more of the following sections.

- For a dynamic dial plan, proceed with the "Configuring Dynamic H.320 Secondary Call Dial Plans" section on page 13.
- For a static dial plan, proceed with the "Configuring Static H.320 Secondary Call Dial Plans" section on page 20.
- For a combined static and dynamic dial plan, proceed with the "Configuring a Combined Static and Dynamic H.320 Secondary Call Dial Plan" section on page 27.

How to Configure Static and Dynamic H.320 Secondary Call Dial Plans

If your endpoint is capable of dynamic receipt of secondary calling numbers (using H.242), configure a dynamic H.320 secondary call dial plan. To configure the secondary call number statically (nonH.242 endpoints), configure a static H.320 secondary call dial plan.

This section describes how to configure static and dynamic H.320 secondary call dial plans and includes the following tasks:

Configuring Dynamic H.320 Secondary Call Dial Plans

Use a dynamic secondary call dial plan when a gateway is connected to a H.320 endpoint that supports dynamic allocation of secondary call numbers (using H.242).

Note Use a static secondary call dial plan when a gateway is connected to an H.320 endpoint that does not support dynamic allocation of secondary call numbers (nonH.242). See the "Configuring Static H.320 Secondary Call Dial Plans" section on page 20 for more information.

Use the following tasks to configure a dynamic H.320 secondary call dial plan.

- Defining Voice Class Called Number Pool for Dynamic Dial Plan, page 14 (required)
- Configuring Dynamic Dial Plan Inbound POTS Dial Peer for Terminating Gateway, page 15 (required)
- Configuring Called Number Pool on Voice Port, page 17 (required)
- Configuring Dynamic Dial Plan Outbound POTS Dial Peer for Originating Gateway, page 18 (required)

Defining Voice Class Called Number Pool for Dynamic Dial Plan

In a dynamic dial plan, you define a pool of dynamic called numbers to be referenced by the gateway for handling primary and secondary calls. Use the following procedure to configure a voice class called number pool for the dynamic H.320 secondary call dial plan.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class called number pool** *tag*
4. **index** *number called-number*

DETAILED STEPS

	Command or Action	Purpose

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

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

Step 3	`voice class called number pool` *tag* **Example:** `Router(config)# voice class called number pool 100`	Defines a dynamic voice class called number pool, which can be allocated by the application to match the incoming H.320 secondary calls. • *tag*—Identifier for the voice class called number pool. The range is 1 to 10000.

	Command or Action	Purpose
Step 4	**index** *number called-number* **Example:** `Router(config-class)# index 1 6505550100 - 6505550111`	Defines an index for a voice class called number pool. **Note** You can define multiple indexes. • *number*—Identifier for the index. The range is 1 to 2147483647. • *called-number*—Specifies a range of called numbers, in E.164 format.

Examples

```
voice class called number pool 100
 index 1 6505550100 - 6505550111

voice class called number pool 200
 index 1 6505550100 - 6505550111 (Range of called numbers are 6505550100 up to 6505550111)
 index 2 6505550112 - 6505550121 (Range of called numbers are 6505550112 up to 6505550121)
```

Configuring Dynamic Dial Plan Inbound POTS Dial Peer for Terminating Gateway

The dynamic inbound POTS dial peer on the terminating gateway handles outgoing H.320 primary and secondary calls. Define the POTS dial peer with ISDN trunk group as the routing interface. The called number for the outgoing H.320 secondary calls are retrieved from the remote H.320 endpoint.

Note The dynamic called number for H.320 secondary calls is propagated across the H.323 network.

Use the following steps to configure an inbound POTS dial peer for a terminating gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **destination pattern** [**+**] *string* [**T**]
5. **information-type** [**fax** | **video** | **voice**]
6. **bandwidth maximum** *value* [**minimum** *value*]
7. **no digit-strip** (optional)
8. **trunkgroup** *name preference-num* (optional)

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 12 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for a specific dial peer. The range is 1 to 2147483647. • **pots**—Indicates a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**destination-pattern [+]** *string* **[T]** **Example:** `Router(config-dial-peer)# destination-pattern 4085550100`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
Step 5	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 6	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 256 minimum 64`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.
Step 7	**no digit-strip** **Example:** `Router(config-dial-peer)# no digit-strip`	(Optional) Disables digit stripping on a POTS dial-peer call leg.

	Command or Action	Purpose
Step 8	**`trunkgroup`** *`name preference-num`* **Example:** `Router(config-dial-peer)# trunkgroup isdntg`	(Optional) Assigns a dial peer to a previously defined trunk group for trunk group label routing. • *name*—Label of the trunk group to use for the call. Valid trunk group names contain a maximum of 63 alphanumeric characters. • *preference-num*—Preference or priority of the trunk group. Range is 1 (highest priority) to 64 (lowest priority).

Examples

```
dial-peer voice 12 pots
 information-type video
 destination-pattern 4085550100
 bandwidth maximum 256 minimum 64
 no digit-strip
 trunkgroup isdntg
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

Configuring Called Number Pool on Voice Port

Dynamic called number support for ISDN calls occurs at the voice port level. Multiple ISDN interfaces can reference the same called number pool if the range of dynamic called numbers are valid routing dialed numbers from the H.320 endpoint to the originating gateway. Use the following steps to assign the voice class called number pool to the ISDN voice port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port:D-channel-number*
4. **voice-class called-number-pool** *tag*

DETAILED STEPS

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

	Command or Action	Purpose
Step 3	**voice-port** *slot/port:D-channel-number* **Example:** `Router(config)# voice-port 1/0:23`	Enters voice-port configuration mode. • *slot*—Router location in which the voice port adapter is installed. Valid entries are 0 to 3. • *port:*—Voice interface card location. Valid entries are 0 and 3. • *D-channel-number*—D-channel number. 23 for T1, 15 for E1.
Step 4	**voice-class called-number-pool** *tag* **Example:** `Router(config-voiceport)# voice-class called-number-pool 100`	Assigns a previously defined voice class called number pool to the voice port. • *tag*—Identifier for the voice class called number pool.

Examples

```
voice class called number pool 100
index 1050 - 1075

dial-peer voice 1000 pots
destination-pattern 1000
information-type video
bandwidth maximum 1024

voice-port 1/0:23
voice-class called-number-pool 100
```

Troubleshooting Tips

Use the **show voice port** command to verify voice port configuration.

Configuring Dynamic Dial Plan Outbound POTS Dial Peer for Originating Gateway

Use the following steps to configure an outbound POTS dial peer on the originating gateway, including the settings for maximum bandwidth and a video information-type.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **destination pattern** [**+**] *string* [**T**]
5. **information-type** [**fax** | **video** | **voice**]
6. **bandwidth maximum** *value* [**minimum** *value*]
7. **port** *slot/port:D-channel-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 1000 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. The range is 1 to 2147483647. • **pots**—Indicates a POTS dial peer that uses VoIP encapsulation on the IP backbone.
Step 4	**`destination-pattern [+]`** *`string`* **`[T]`** **Example:** `Router(config-dial-peer)# destination-pattern 1000`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
Step 5	**`information-type`** `[`**`fax`** `|` **`video`** `|` **`voice`**`]` **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 6	**`bandwidth maximum`** *`value`* `[`**`minimum`** *`value`*`]` **Example:** `Router(config-dial-peer)# bandwidth maximum 1024 minimum 64`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.

	Command or Action	Purpose
Step 7	**port** *slot/port:D-channel-number* **Example:** `Router(config-dial-peer)# port 1/0:23`	Associates a dial peer with a specific voice port. • *slot*—Router location in which the voice port adapter is installed. Valid entries are 0 to 3. • *port:*—Voice interface card location. Valid entries are 0 and 3. • *D-channel-number*—D-channel number. 23 for T1, 15 for E1.

Examples

```
dial-peer voice 1000 pots
 destination-pattern 1000
 information-type video
 bandwidth maximum 1024 minimum 64
 port 1/0:23
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

What to Do Next

To configure a static H.320 secondary dial plan, proceed to the "Configuring Static H.320 Secondary Call Dial Plans" section on page 20. To configure a combined static and dynamic H.320 secondary dial plan, proceed to the "Configuring a Combined Static and Dynamic H.320 Secondary Call Dial Plan" section on page 27.

Configuring Static H.320 Secondary Call Dial Plans

Use a static secondary call dial plan when a gateway is connected to a H.320 endpoint that does not support H.242. A static secondary call dial plan uses called number tables in E.164 format to use as called numbers for incoming and outgoing calls to H.320 endpoints.

Use the following tasks to configure a static H.320 secondary call dial plan:

- Defining Inbound Voice Class Called Numbers for Static Dial Plan, page 21 (required)
- Defining Outbound Voice Class Called Numbers for Static Dial Plan, page 22 (required)
- Configuring Static Dial Plan Outbound POTS Dial Peer for Originating Gateway, page 23 (required)
- Configuring Static Dial Plan Inbound POTS Dial Peer for Terminating Gateway, page 25 (required)

Note To configure a dynamic H.320 secondary call dial plan, see the "Configuring Dynamic H.320 Secondary Call Dial Plans" section on page 13.

Defining Inbound Voice Class Called Numbers for Static Dial Plan

Define a the inbound called number table to associate incoming H.320 secondary calls with H.320 primary calls. Use this procedure to define one or more voice class called numbers for the inbound POTS dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class called number inbound** *tag*
4. **index** *number called-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice class called number inbound** *tag* **Example:** `Router(config)# voice class called number inbound 200`	Defines one or more static voice class called numbers for H.320 calls. • **inbound**—Inbound voice class called number. • *tag*—Identifier for the inbound voice class called number.

Step 4	**index** *number called-number*
	
	Example:
	`Router(config-class)# index 1 6505550111`
	`index 2 6505550112`
	`index 3 6505550113`
	`index 4 6505550114`
		Defines an index for a voice class called number. You can define multiple indexes.
		• *number*—Identifier for the index. The range is 1 to 2147483647.
		• *called-number*—Specifies a called number, in E.164 format.

Examples

```
voice class called number inbound 200
 index 1 5550100
 index 2 5550101
 index 3 5550102
 index 4 5550103

voice class called number inbound 9001
 index 1 9001
!
```

```
voice class called number inbound 9999
 index 1 9997
 index 2 9998
 index 3 9999
!
```

Defining Outbound Voice Class Called Numbers for Static Dial Plan

Define an outbound called number table to associate outgoing H.320 secondary calls with H.320 primary calls. Use these steps to define one or more voice class called number for the outbound POTS dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class called number outbound** *tag*
4. **index** *number called-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice class called number outbound** *tag* **Example:** `Router(config)# voice class called number outbound 50`	Defines one or more static voice class called numbers for H.320 calls. • **outbound**—Outbound voice class called number. • *tag*—Identifier for the outbound voice class called number.
Step 4	**index** *number called-number* **Example:** `Router(config-class)# index 1 100A11 index 2 +7878*55`	Defines an index for a voice class called number. You can define multiple indexes. • *number*—Identifier for the index. The range is 1 to 2147483647. • *called-number*—Specifies a called number, in E.164 format.

Examples

```
voice class called number outbound 50
 index 1 100A11
 index 2 +7878*55
```

```
voice class called number outbound 1
 index 1 6001
!
voice class called number outbound 7101
 index 1 7101
!
voice class called number outbound 1111
 index 1 1111
 index 2 1112
 index 3 1113
 index 4 1114
!
```

Configuring Static Dial Plan Outbound POTS Dial Peer for Originating Gateway

The originating gateway handles outgoing H.320 primary and secondary calls. Use this procedure to configure the outbound POTS dial peer for the originating gateway for a static dial plan, including the outbound called number table.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **incoming called-number** *string*
5. **direct-inward-dial**
6. **information-type** [**fax** | **video** | **voice**]
7. **voice-class called-number** [**inbound**] *tag*
8. **bandwidth maximum** *value* [**minimum** *value*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 7001 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. The range is 1 to 2147483647. • **pots**—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.

	Command or Action	Purpose
Step 4	**incoming called-number** *string* **Example:** `Router(config-dial-peer)# incoming called-number 408`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer. • *string*—Incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number. The default is the calling number pattern.
Step 5	**direct-inward-dial** **Example:** `Router(config-dial-peer)# direct-inward-dial`	Enables the direct inward dialing (DID) call treatment for an incoming called number.
Step 6	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 7	**voice class called number inbound** *tag* **Example:** `Router(config)# voice class called number inbound 200`	Defines one or more static voice class called numbers for H.320 calls. • **inbound**—Inbound voice class called number. • *tag*—Identifier for the inbound voice class called number.
Step 8	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 256 minimum 64`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.

Examples

```
dial-peer voice 7001 pots
 information-type video
 voice-class called-number inbound 1
 incoming called-number 408
 bandwidth maximum 256 minimum 64
 direct-inward-dial
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

Configuring Static Dial Plan Inbound POTS Dial Peer for Terminating Gateway

The terminating gateway handles incoming H.320 primary and secondary calls. Use this procedure to configure the inbound POTS dial peer for the terminating gateway for a static dial plan, including the inbound called number table.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **destination pattern** [**+**] *string* [**T**]
5. **information-type** [**fax** | **video** | **voice**]
6. **voice-class called-number** [**inbound**] *tag*
7. **bandwidth maximum** *value* **minimum** *value*
8. **no digit-strip** (optional)
9. **trunkgroup** *name preference-num* (optional)

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 12 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. The range is 1 to 2147483647. • **pots**—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**destination-pattern [+]** *string* **[T]** **Example:** `Router(config-dial-peer)# destination-pattern 4085550100`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

	Command or Action	Purpose
Step 5	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 6	**voice-class called-number** [**inbound**] *tag* **Example:** `Router(config-dial-peer)# voice-class called-number inbound 50`	Assigns a previously defined voice class called number to an inbound or outbound POTS dial peer. • **inbound**—Assigns an inbound voice class called number to the dial peer. • *tag*—Identifier for the voice class called number.
Step 7	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 192`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.
Step 8	**no digit-strip** **Example:** `Router(config-dial-peer)# no digit-strip`	(Optional) Disables digit stripping on a POTS dial-peer call leg.
Step 9	**trunkgroup** *name preference-num* **Example:** `Router(config-dial-peer)# trunkgroup isdntg`	(Optional) Assigns a dial peer to a trunk group for trunk group label routing. • *name*—Label of the trunk group to use for the call. Valid trunk group names contain a maximum of 63 alphanumeric characters. • *preference-num*—Preference or priority of the trunk group. Range is 1 (highest priority) to 64 (lowest priority).

Examples

```
dial-peer voice 12 pots
 information-type video
 voice-class called-number inbound 50
 destination-pattern 4085550100
 bandwidth maximum 192
 no digit-strip
 trunkgroup isdntg
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

What to Do Next

To configure a combined static and dynamic H.320 secondary dial plan, proceed to the "Configuring a Combined Static and Dynamic H.320 Secondary Call Dial Plan" section on page 27. To configure a dynamic dial plan, proceed to the "Configuring Dynamic H.320 Secondary Call Dial Plans" section on page 13.

Configuring a Combined Static and Dynamic H.320 Secondary Call Dial Plan

Determining whether to use static or dynamic H.320 secondary dial plan depends on the capability of the remote H.320 endpoints. In some networks, the ISDN interface between an originating and terminating gateway might need to support both static and dynamic dial plans.

Use the following tasks to configure a combined static and dynamic H.320 secondary call dial plan:

Defining Inbound Static Called Numbers and Dynamic Called Number Pool for Combined Static and Dynamic Dial Plan

With a combined static and dynamic configuration, the secondary numbers match the static inbound voice-class called-number inbound for the incoming dial-peer first. If the voice-class called-number-pool is configured under voice-port for a specific T1 or E1 controller, dynamic secondary numbers are chosen. Static secondary numbers are chosen only if no dynamic secondary number pool is found under the voice port.

In a combined static and dynamic H.320 secondary call dial plan, the inbound called number table and dynamic called number pool are configured on the same gateway.

Note There is no call fallback for a dynamic dial plan. If a combined static and dynamic dial plan is configured, the static dial plan takes precedence.

Use the following procedure to define a static inbound called number table and a dynamic called number pool and to assign the number pool to the voice port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class called number pool** *tag*

4. **index** *number called-number*
5. **exit**
6. **voice-port** *slot/port:D-channel-number*
7. **voice-class called-number-pool** *tag*
8. **exit**
9. **voice-class called-number** [**inbound** | **outbound**] *tag*
10. **index** *number called-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class called number pool`** *`tag`* **Example:** `Router(config)# voice class called number pool 10`	Defines a dynamic voice class called number pool, which can be allocated by the application to match the incoming H.320 secondary calls. • *tag*—Identifier for the voice class called number pool. The range is 1 to 10000.
Step 4	**`index`** *`number called-number`* **Example:** `Router(config-class)# index 1 6505550100 - 6505550111`	Defines an index for a voice class called number pool. You can define multiple indexes. • *number*—Identifier for the index. The range is 1 to 2147483647. • *called-number*—Specifies a called number, or a range of called numbers, in E.164 format.
Step 5	**`exit`** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.
Step 6	**`voice-port`** *`slot/port:D-channel-number`* **Example:** `Router(config)# voice-port 2/0:15`	Enters voice-port configuration mode. • *slot*—Router location in which the voice port adapter is installed. Valid entries are 0 to 3. • *port:*—Voice interface card location. Valid entries are 0 and 3. • *D-channel-number*—D-channel number. 23 for T1, 15 for E1.

	Command or Action	Purpose
Step 7	**voice-class called-number-pool** *tag* **Example:** `Router(config-voiceport)# voice-class called-number-pool 10`	Assigns a previously defined voice class called number pool to the voice port. • *tag*—Identifier for the voice class called number. **Note** You can repeat Step 6 and this step for multiple voice ports.
Step 8	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice port configuration mode.
Step 9	**voice class called number inbound** *tag* **Example:** `Router(config)# voice class called number inbound 200`	Defines one or more static voice class called numbers for inbound H.320 calls. • **inbound**—Inbound voice class called number. • *tag*—Identifier for the inbound voice class called number.
Step 10	**index** *number called-number* **Example:** `Router(config-class)# index 1 40844420..&`	Defines an index for a voice class called number. You can define multiple indexes. • *number*—Identifier for the index. The range is 1 to 2147483647. • *called-number*—Specifies a called number, in E.164 format.

Examples

Multiple voice ports can be configured with the same called number pool as shown in the following example.

```
voice class called number pool 10
 index 1 4085550100 - 4085550111

voice-port 2/0:15
 voice-class called-number-pool 10

voice-port 1/0:23
 voice-class called-number-pool 10

voice class called number inbound 200
 index 1 40844420..
```

Troubleshooting Tips

Use the **show voice port** command to verify voice port configuration.

Configuring the Outbound Static Called Numbers for Combined Static and Dynamic Dial Plan

Define the outbound stated called number table on a separate gateway. Use the following steps to define the outbound called number table.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-class called-number** [**inbound** | **outbound**] *tag*
4. **index** *number called-number*

DETAILED STEPS

	Command or Action	Purpose

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

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

Step 3	**voice class called number outbound** *tag* **Example:** `Router(config)# voice class called number outbound 300`	Defines one or more static voice class called numbers for H.320 calls. • **outbound**—Outbound voice class called number. • *tag*—Identifier for the outbound voice class called number.

Step 4	**index** *number called-number*
	
	Example:
	`Router(config-class)# index 1 4085550100`
	`index 2 4085550102`
	`index 3 4085550103`
	`index 4 4085550104`
	`index 5 4085550105`
	`index 6 4085550106`
	`index 7 4085550107`
		Defines an index for a voice class called number. You can define multiple indexes.
		• *number*—Identifier for the index. The range is 1 to 2147483647.
		• *called-number*—Specifies a called number, in E.164 format.

Example

This example configuration shows multiple indexes defined for an outbound voice class called number.

```
voice class called number outbound 300
 index 1 4085550101
 index 2 4085550102
 index 3 4085550103
 index 4 4085550104
 index 5 4085550105
 index 6 4085550106
 index 7 4085550107
```

Configuring Combined Static and Dynamic Dial Plan Inbound POTS Dial Peer for Originating Gateway

The same inbound dial peer is used to support both dynamic and static incoming H.320 secondary calls. The static inbound called number table is used to select a primary call when dynamic called numbers are not allocated for a primary call. Use the following steps to configure the inbound POTS dial peer for the originating gateway in a combined static and dynamic H.320 secondary call dial plan.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **incoming called-number** *string*
5. **direct-inward-dial**
6. **information-type** [**fax** | **video** | **voice**]
7. **voice-class called-number** [**inbound**] *tag*
8. **bandwidth maximum** *value* [**minimum** *value*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 12 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. The range is 1 to 2147483647. • **pots**—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**`incoming called-number`** *`string`* **Example:** `Router(config-dial-peer)# incoming called-number 408`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer. • *string*—Incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number. The default is the calling number pattern.

	Command or Action	Purpose
Step 5	**direct-inward-dial** **Example:** `Router(config-dial-peer)# direct-inward-dial`	Enables the direct inward dialing (DID) call treatment for an incoming called number.
Step 6	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 7	**voice-class called-number** [**inbound**] *tag* **Example:** `Router(config-dial-peer)# voice-class called-number inbound 50`	Assigns a previously defined voice class called number to an inbound or outbound POTS dial peer. • **inbound**—Assigns an inbound voice class called number to the dial peer. • *tag*—Identifier for the voice class called number.
Step 8	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 256 minimum 64`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.

Examples

```
dial-peer voice 12 pots
 incoming called-number 408
 information-type video
 voice-class called-number inbound 200
 bandwidth maximum 256 minimum 64
 direct-inward-dial
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

Configuring Dynamic Outbound POTS Dial Peers for Terminating Gateway

The outbound POTS dial peers on the terminating gateway handle outgoing H.320 primary and secondary calls. Configure separate dial peers for H.242 and nonH.242 endpoints.

The dynamic H.320 outbound dial peer with routing dialed numbers terminates H.242 endpoints. On the dynamic outbound POTS dial peer, called numbers are allocated from the dynamic called number pool configured on the voice port.

Use the following steps to configure an outbound POTS dial peer for a terminating gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **destination pattern** [**+**] *string* [**T**]
5. **information-type** [**fax** | **video** | **voice**]
6. **bandwidth maximum** *value* [**minimum** *value*]
7. **no digit-strip** (optional)
8. **trunkgroup** *name preference-num* (optional)

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 22 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. The range is from 1 to 2147483647. • **pots**—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**`destination-pattern [+]`** *`string`* **`[T]`** **Example:** `Router(config-dial-peer)# destination-pattern 4085550100`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

	Command or Action	Purpose
Step 5	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 6	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 512`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.
Step 7	**no digit-strip** **Example:** `Router(config-dial-peer)# no digit-strip`	(Optional) Disables digit stripping on a POTS dial-peer call leg.
Step 8	**trunkgroup** *name preference-num* **Example:** `Router(config-dial-peer)# trunkgroup isdntg`	(Optional) Assigns a dial peer to a trunk group for trunk group label routing. • *name*—Label of the trunk group to use for the call. Valid trunk group names contain a maximum of 63 alphanumeric characters. • *preference-num*—Preference or priority of the trunk group. Range is 1 (highest priority) to 64 (lowest priority).

Example

```
dial-peer voice 22 pots
 destination-pattern 4085550100
 information-type video
 bandwidth maximum 512
 no digit-strip
 trunkgroup isdntg
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

Configuring Static Outbound POTS Dial Peers for Terminating Gateway

The outbound POTS dial peers on the terminating gateway handle outgoing H.320 primary and secondary calls. Configure separate dial peers for H.242 and nonH.242 endpoints.

The static outbound dial peer with routing dialed numbers terminates to nonH.242 endpoints. On the static outbound POTS dial peer, called numbers are allocated from the inbound called number table.

Use the following steps to configure the static outbound POTS dial peer on the terminating gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **destination pattern** [**+**] *string* [**T**]
5. **information-type** [**fax** | **video** | **voice**]
6. **voice-class called-number** [**inbound** | **outbound**] *tag*
7. **bandwidth maximum** *value* **minimum** *value*
8. **no digit-strip** (optional)
9. **trunkgroup** *name preference-num* (optional)

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`pots`** **Example:** `Router(config)# dial-peer voice 2222 pots`	Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. • *tag*—Identifier for the dial peer. Range is 1 to 2147483647. • **pots**—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.
Step 4	**`destination-pattern [+]`** *`string`* **`[T]`** **Example:** `Router(config-dial-peer)# destination-pattern 4085550100`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

	Command or Action	Purpose
Step 5	**information-type** [**fax** | **video** | **voice**] **Example:** `Router(config-dial-peer)# information-type video`	Selects a specific information type for a VoIP or POTS dial peer. • **fax**—Sets information type to fax. • **video**—Sets information type to video. • **voice**—Sets information type to voice. This is the default. **Note** To return to the default value, use the **default information-type** command in dial-peer configuration mode.
Step 6	**voice-class called-number** [**inbound** | **outbound**] *tag* **Example:** `Router(config-dial-peer)# voice-class called-number outbound 50`	Assigns a previously defined voice class called number to an inbound or outbound POTS dial peer. • **inbound—**Assigns an inbound voice class called number to the dial peer. • **outbound**—Assigns an outbound voice class called number to the dial peer. • *tag*—Identifier for the voice class called number.
Step 7	**bandwidth maximum** *value* [**minimum** *value*] **Example:** `Router(config-dial-peer)# bandwidth maximum 256 minimum 64`	Specifies the maximum and minimum bandwidth for an H.320 call. • **maximum** *value*—Sets the maximum bandwidth. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64. • **minimum** *value*—(Optional) Sets the minimum bandwidth. Acceptable values are 64 or **minimum** *value*=**maximum** *value*.
Step 8	**no digit-strip** **Example:** `Router(config-dial-peer)# no digit-strip`	(Optional) Disables digit stripping on a POTS dial-peer call leg.
Step 9	**trunkgroup** *name preference-num* **Example:** `Router(config-dial-peer)# trunkgroup isdntg`	(Optional) Assigns a dial peer to a trunk group for trunk group label routing. • *name*—Label of the trunk group to use for the call. Valid trunk group names contain a maximum of 63 alphanumeric characters. • *preference-num*—Preference or priority of the trunk group. Range is 1 (highest priority) to 64 (lowest priority).

Examples

The following example configuration shows a static outbound POTS dial peer for a terminating gateway.

```
dial-peer voice 2222 pots
 destination-pattern 4085550100
 information-type video
 voice-class called-number outbound 50
 bandwidth maximum 256 minimum 64
```

```
 no digit-strip
 trunkgroup isdntg
```

Troubleshooting Tips

Use the **show dial-peer voice** command to verify the dial peer configuration.

Configuration Examples for Integrated Data, Voice, and Video Services for ISDN Interfaces

This section provides the following configuration examples:

- Integrated Services with Combined Static and Dynamic H.320 Secondary Call Dial Plan: Example, page 37
- Integrated Services with Static H.320 Secondary Call Dial Plan: Example, page 40

Integrated Services with Combined Static and Dynamic H.320 Secondary Call Dial Plan: Example

The following example shows a combined static and dynamic H.320 secondary call dial plan. The dynamic dial plan is configured on the voice ports and the static dial plan is configured on the dial peers.

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router1
!
boot-start-marker
boot-end-marker
!
logging buffered 4096000 debugging
no logging console
!
no aaa new-model
!
resource manager
!
no network-clock-participate slot 1
ip subnet-zero
ip cef
!
no ip dhcp use vrf connected
!
no ftp-server write-enable
isdn switch-type basic-net3
voice-card 1
 no dspfarm
!
voice service voip
 h323
  call start slow
  h245 caps mode restricted
```

```
!
voice class codec 1
 codec preference 1 g728
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class called number inbound 3
 index 1 5550100
!
voice class called number outbound 3
 index 1 5550120
 index 2 5550121
 index 3 5550122
 index 4 5550123
!
voice class called number pool 1
 index 1 5550130 - 5550133
!
interface FastEthernet0/0
 ip address 10.7.50.103 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface BRI1/0
 no ip address
 isdn switch-type basic-net3
 isdn protocol-emulate network
 isdn layer1-emulate network
 isdn calling-number 12345
 isdn supp-service name calling
 isdn skipsend-idverify
 isdn integrate calltype all
!
interface BRI1/1
 no ip address
 isdn switch-type basic-net3
 isdn protocol-emulate network
 isdn layer1-emulate network
 isdn calling-number 98765
 isdn skipsend-idverify
 isdn integrate calltype all
!
interface BRI1/2
 no ip address
 isdn switch-type basic-net3
 isdn protocol-emulate network
 isdn layer1-emulate network
 isdn calling-number 98765
 isdn skipsend-idverify
 isdn integrate calltype all
!
interface BRI1/3
 no ip address
 isdn switch-type basic-net3
 isdn protocol-emulate network
 isdn layer1-emulate network
 isdn calling-number 98765
 isdn skipsend-idverify
```

```
 isdn integrate calltype all
!
ip default-gateway 10.7.0.1
ip classless
ip route 172.16.254.254 255.255.255.255 FastEthernet0/0
!
ip http server
!
control-plane
!
voice-port 1/0/0
 voice-class called-number-pool 1
!
voice-port 1/0/1
 voice-class called-number-pool 1
!
voice-port 1/1/0
 voice-class called-number-pool 1
!
voice-port 1/1/1
 voice-class called-number-pool 1
!
dial-peer voice 1 pots
 information-type video
 voice-class called-number inbound 3
 incoming called-number 5550100
 bandwidth maximum 128
 direct-inward-dial
!
dial-peer voice 2 voip
 shutdown
 destination-pattern 5550100
 session target ipv4:10.7.50.201
 codec g711ulaw
!
dial-peer voice 3 voip
 shutdown
 destination-pattern 5550100
 voice-class codec 1
 session target ipv4:10.7.50.50
!
dial-peer voice 4 pots
 destination-pattern 5550120
 information-type video
 direct-inward-dial
 port 1/1/1
!
dial-peer voice 5 voip
 destination-pattern 5550120
 session target ipv4:10.7.50.50
!
dial-peer voice 6 voip
 destination-pattern 5550155
 session target ipv4:10.7.50.12
!
line con 0
line aux 0
line vty 0 4
 login
!
end
```

Integrated Services with Static H.320 Secondary Call Dial Plan: Example

The following example shows a static H.320 secondary call dial plan for calls between an SCCP endpoint and an H.320 endpoint:

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router2
!
boot-start-marker
boot-end-marker
!
logging buffered 1000000 debugging
no logging console
!
no aaa new-model
!
resource manager
!
no network-clock-participate slot 2
ip subnet-zero
ip cef
!
no ip dhcp use vrf connected
!
ip dhcp pool phone1
   host 10.7.50.114 255.255.0.0
   client-identifier 0100.ffff.ffff.ffff
   default-router 10.7.50.211
   option 150 ip 10.7.50.211
!
no ip domain lookup
no ftp-server write-enable
isdn switch-type primary-ni
voice-card 2
 no dspfarm
!
trunk group  1
!
voice service voip
 h323
  call start slow
!
voice class called number inbound 1
 index 1 7001
!
voice class called number inbound 201
 index 1 2001
!
voice class called number inbound 202
 index 1 2002
!
voice class called number inbound 203
 index 1 2003
!
voice class called number inbound 204
 index 1 2004
!
voice class called number inbound 205
 index 1 2005
```

```
!
voice class called number inbound 206
 index 1 2006
!
voice class called number inbound 207
 index 1 2007
!
voice class called number inbound 9001
 index 1 9001
!
voice class called number inbound 9999
 index 1 9997
 index 2 9998
 index 3 9999
!
voice class called number outbound 1
 index 1 6001
!
voice class called number outbound 7101
 index 1 7101
!
voice class called number outbound 1111
 index 1 1111
 index 2 1112
 index 3 1113
 index 4 1114
!
voice class called number pool 8888
 index 1 5550190
 index 2 5550191 - 5550199
!
controller T1 2/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 2/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-20,24
!
interface FastEthernet0/0
 ip address 10.7.50.211 255.255.0.0
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip id dralion_gk ipaddr 10.7.50.49 1719
 h323-gateway voip h323-id b2b_3725
 h323-gateway voip tech-prefix 86001
!
interface FastEthernet0/1
 ip address 10.0.0.7 255.255.255.0
 shutdown
 duplex auto
 speed auto
!
interface BRI2/0
 no ip address
 isdn switch-type basic-ni
 isdn point-to-point-setup
!
interface Serial2/0:23
 no ip address
 isdn switch-type primary-ni
```

```
 isdn integrate calltype all
 no cdp enable
!
interface BRI2/1
 no ip address
 isdn switch-type basic-ni
 isdn point-to-point-setup
!
interface Serial2/1:23
 no ip address
 isdn switch-type primary-ni
 isdn integrate calltype all
 no cdp enable
!
ip default-gateway 10.7.0.1
ip classless
ip route 172.16.254.254 255.255.255.255 10.5.0.1
ip route 172.16.254.254 255.255.255.255 FastEthernet0/0
!
ip http server
!
tftp-server flash:P00000000111.bin
tftp-server flash:P00000000222.bin
tftp-server flash:P00000000333.loads
tftp-server flash:P00000000444.sbn
tftp-server flash:P00000000555.sb2
!
control-plane
!
voice-port 2/0:23
!
voice-port 2/1/0
!
voice-port 2/1/1
!
voice-port 2/1:23
!
dial-peer voice 3201 pots
 destination-pattern 86001
 information-type video
 voice-class called-number outbound 1
 bandwidth maximum 384
 direct-inward-dial
 port 2/0:23
 forward-digits 4
!
dial-peer voice 348906 voip
 destination-pattern 348906
 video codec h263+
 session target ipv4:10.7.50.107
 req-qos controlled-load
!
dial-peer voice 7001 pots
 information-type video
 voice-class called-number inbound 1
 incoming called-number 7001
 bandwidth maximum 384
 direct-inward-dial
!
dial-peer voice 9001 voip
 destination-pattern 9001
 session target ipv4:10.7.50.107
 codec g711ulaw
!
```

```
dial-peer voice 2001 pots
 information-type video
 voice-class called-number inbound 201
 incoming called-number 2001
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 2002 pots
 information-type video
 voice-class called-number inbound 202
 incoming called-number 2002
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 2003 pots
 information-type video
 voice-class called-number inbound 203
 incoming called-number 2003
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 2004 pots
 information-type video
 voice-class called-number inbound 204
 incoming called-number 2004
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 2005 pots
 information-type video
 voice-class called-number inbound 205
 incoming called-number 2005
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 2006 pots
 information-type video
 voice-class called-number inbound 206
 incoming called-number 2006
 bandwidth maximum 192
 direct-inward-dial
!
dial-peer voice 7101 pots
 destination-pattern 7101
 information-type video
 voice-class called-number outbound 7101
 bandwidth maximum 384
 direct-inward-dial
 port 2/0:23
 forward-digits all
!
dial-peer voice 99001 pots
 information-type video
 voice-class called-number inbound 9001
 incoming called-number 9001
 bandwidth maximum 384
 direct-inward-dial
!
gateway
 timer receive-rtp 1200
!
telephony-service
 video
 load 7960-7940 P00000000111
```

```
 max-ephones 20
 max-dn 20
 ip source-address 10.7.50.211 port 2000
 service phone videoCapability 1
 create cnf-files version-stamp Jan 01 2002 00:00:00
 max-conferences 8 gain -6
 call-forward pattern .T
 transfer-system full-blind
 transfer-pattern 6..
 transfer-pattern 5..
 transfer-pattern 4..
 transfer-pattern 2..
 transfer-pattern .T
 transfer-pattern ....
!
ephone-dn  1  dual-line
 number 2001
 application default
!
ephone-dn  2  dual-line
 number 2002
!
ephone-dn  3  dual-line
 number 2003
!
ephone-dn  4  dual-line
 number 2004
!
ephone-dn  5  dual-line
 number 2005
!
ephone-dn  20
 number 7001
!
ephone  1
 video
 mac-address ffff.ffff.fff1
 type 7960
 button  1:1
!
ephone  2
 video
 mac-address ffff.ffff.fff2
 type 7960
 button  1:2
!
ephone  3
 video
 mac-address ffff.ffff.fff3
 type 7960
 button  1:3
!
ephone  4
 video
 mac-address ffff.ffff.fff4
 type 7960
 button  1:4
!
ephone  5
 video
 mac-address ffff.ffff.fff5
 type 7960
 button  1:5
!
```

```
ephone  20
 video
 mac-address ffff.ffff.fff6
 type 7960
 button  1:20
!
line con 0
line aux 0
line vty 0 4
 login
!
end
```

Additional References

The following sections provide references related to integrated data, voice, and video services for ISDN interfaces.

Related Documents

Related Topic	Document Title
Information on integrating data and voice	*Integrating Data and Voice Services for ISDN PRI Interfaces on Multiservice Access Routers*
ISDN configuration information	*Cisco IOS ISDN Voice Configuration Guide*
ISDN voice interface information	*Configuring ISDN PRI Voice-Interface Support*
Video command reference information	*Cisco IOS Voice Command Reference*
Video telephony	*Understanding Video Telephony*
Voice command reference information	*Cisco IOS Voice Command Reference*
Voice configuration information	*Cisco IOS Voice Configuration Library*

Standards

Standard	Title
ITU-T E.164	The international public telecommunication numbering plan.
ITU-T H.221	Frame structure for a 64 to 1920 kbps channel in audiovisual teleservices.
ITU-T H.242	System for establishing communication between audiovisual terminals using digital channels up to 2 MB per second (Mbps).
ITU-T H.242 Amendment 1	Support for 14 kHz audio bandwidth extension of G.722.1 Annex C in H.242.
ITU-T H.261	Video codec for audiovisual services where data rates are multiples of 64 kbps.
ITU-T H.263	Video coding for low bit rate communication.
ITU-T H.263+	Enhancements and improved performance for H.263 video codec.
ITU-T H.264	Advanced video coding for generic audiovisual services.
ITU-T H.320	Narrow-band visual telephone systems and terminal equipment.

MIBs

MIB	MIBs Link
• CISCO-VOICE-COMMON-DIAL-CONTROL-MIB • CISCO-VOICE-DIAL-CONTROL-MIB • CISCO-H320-DIAL-CONTROL-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 2190	RTP Payload Format for H.263 Video Streams
RFC 2198	RTP Payload for Redundant Audio Data
RFC 2429	RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Command Reference

This section documents the following new and modified commands:

New Commands

- **bandwidth (dial-peer)**
- **debug voice h221**
- **debug voip h221**
- **index (voice class)**
- **show voice class called-number**
- **show voice class called-number-pool**
- **video codec (dial-peer)**
- **video codec (voice-class)**
- **voice class called number**
- **voice-class called-number (dial peer)**
- **voice-class called-number-pool**

Modified Commands

- **information-type**

- **rtp payload-type**
- **show call active video**
- **show dial-peer voice**
- **show voice dsp**
- **show voice port**

bandwidth (dial-peer)

To set the maximum bandwidth on a POTS dial peer for an H.320 call, use the **bandwidth** command in dial-peer configuration mode. To remove the bandwidth setting, use the **no** form of this command.

bandwidth maximum *value* [**maximum** *value*]

no bandwidth

Syntax Description

maximum *value*	Sets the maximum bandwidth for an H.320 call on a POTS dial peer. The range is 64 to 1024, entered in increments of 64 kilobits per second (kbps). The default is 64.
minimum *value*	(Optional) Sets the minimum bandwidth. Acceptable values are 64 kbps or **minimum** *value*=**maximum** *value*.

Command Default

No maximum bandwidth is set.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to set the maximum and minimum bandwidth for an H.320 POTS dial-peer. Only the maximum bandwidth is required. The value must be entered in increments of 64 kbps. The minimum bandwidth setting is optional, and the value must be either 64 kbps or equal to the maximum value setting.

Examples

The following example shows configuration for POTS dial peer 200 with a maximum bandwidth of 1024 kbps:

```
dial-peer voice 200 pots
 bandwidth maximum 1024
```

The following example shows configuration for POTS dial peer 11 with a maximum bandwidth of 640 and a minimum of 64:

```
dial-peer voice 11 pots
 bandwidth maximum 640 minimum 64
```

Related Commands

Command	Description
bandwidth	Specifies the maximum aggregate bandwidth for H.323 traffic and verifies the available bandwidth of the destination gatekeeper.

debug voice h221

To debug telephony call control information, use the **debug voice h221** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug voice h221 [**all** | **default** | **error** [**call** [**informational**] | **software** [**informational**]] | **function** | **individual** | **inout** | **raw** [**decode**]]

no debug voice h221

Syntax Description

all	(Optional) Enables all H.221 debugging, except the **raw** option.
default	(Optional) Activates function, inout, error call, and software debugging.
error	(Optional) Enables H.221 call error and software error debugging.
error [call]	(Optional) Enables H.221 major call processing error debugs related to the H.221 subsystem.
error [call [informational]]	(Optional) Enables H.221 major and informational call processing error debugs related to the H.221 subsystem.
error [software]	(Optional) Enables H.221 major software error debugs related to the H.221 subsystem.
error [software [informational]]	(Optional) Enables H.221 major and informational software error debugs related to the H.221 subsystem.
function	(Optional) Enables procedure tracing.
individual	(Optional) Activates individual H.221 debugging.
inout	(Optional) Enables subsystem inout debugging.
raw	(Optional) Displays raw BAS messages.
raw [decode]	(Optional) Decodes raw BAS data.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

This command enables debugging for H.221 message events (voice telephony call control information).

Note This command provides the same results as the **debug voip h221** command.

Caution We recommend that you log the output from the **debug voice h221 all** command to a buffer, rather than sending the output to the console; otherwise, the size of the output could severely impact the performance of the gateway.

Use the **debug voice h221 individual** *x* command, (where *x* is an index number for a debug category), to activate a single debug, selected by index number instead of entering a group of debug commands. See Table 1 for a list of debug categories and corresponding index numbers.

Table 1 Indexes and Categories for the debug voice h221 individual command

Index Number	Debug Category
1, 2, 30, 31, 32	Secondary number exchange
5, 6, 14, 15, 16, 22	Audio mode/caps
7, 10, 12, 13, 17, 28	Video mode/caps
8, 9, 23	B-channel mode/caps
11, 24, 33	Miscellaneous command exchange
18	Bandwidth calculations
19, 20, 21	DSP configuration
3, 4, 25, 27, 42, 43	General caps/internal
26	Non-standard caps/command
29	Loop request
34, 35, 36, 37, 38, 39, 40, 41	BAS squelch

Examples

The **raw** keyword displays the raw BAS information coming from or to the DSP. It is displayed in a hexadecimal octet format. The **decode** option decodes the BAS information into a readable English format.

The following is sample output from the **debug voice h221 raw decode** command:

```
BAS=81:1 0 0 0 0 0 0 1: AUDIO CAPS=g711 a-law
BAS=82:1 0 0 0 0 0 1 0: AUDIO CAPS=g711 u-law
BAS=84:1 0 0 0 0 1 0 0: AUDIO CAPS=g722 48k
BAS=85:1 0 0 0 0 1 0 1: AUDIO CAPS=g728
BAS=F9:1 1 1 1 1 0 0 1: H.242 MBE start indication
BAS=02:0 0 0 0 0 0 1 0: H.242 MBE length=2
BAS=0A:0 0 0 0 1 0 1 0: H.242 MBE type=H.263 caps
BAS=8A:1 - - - - - - -: Always 1
BAS=8A:- 0 0 0 1 - - -: H.263 MPI=1
BAS=8A:- - - - - 0 1 -: H.263 FORMAT=h.263_cif
BAS=8A:- - - - - - - 0: No additional options
```

Related Commands

Command	Description
debug voip ccapi	Enables debugging for the call control application programming interface (CCAPI) contents.
debug voip rtp	Enables debugging for Real-Time Transport Protocol (RTP) named event packets.

debug voip h221

To debug telephony call control information, use the **debug voip h221** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug voip h221 [**all** | **default** | **error** [**call** [**informational**] | **software** [**informational**]] | **function** | **individual** | **inout** | **raw [decode**]]

no debug voip h221

Syntax Description

all	(Optional) Enables all H.221 debugging, except the **raw** option.
default	(Optional) Activates function, inout, error call, and software debugging.
error	(Optional) Enables H.221 call error and software error debugging.
error [call]	(Optional) Enables H.221 major call processing error debugs related to the H.221 subsystem.
error [call [informational]]	(Optional) Enables H.221 major and informational call processing error debugs related to the H.221 subsystem.
error [software]	(Optional) Enables H.221 major software error debugs related to the H.221 subsystem.
error [software [informational]]	(Optional) Enables H.221 major and informational software error debugs related to the H.221 subsystem.
function	(Optional) Enables procedure tracing.
individual	(Optional) Activates individual H.221 debugging.
inout	(Optional) Enables subsystem inout debugging.
raw	(Optional) Displays raw BAS messages.
raw [decode]	(Optional) Decodes raw BAS data.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

This command enables debugging for H.221 message events (voice telephony call control information).

Note This command provides the same results as the **debug voice h221** command.

Caution We recommend that you log the output from the **debug voip h221 all** command to a buffer, rather than sending the output to the console; otherwise, the size of the output could severely impact the performance of the gateway.

Use the **debug voip h221 individual** *x* command, (where *x* is an index number for a debug category), to activate a single debug, selected by index number instead of entering a group of debug commands. See Table 2 for a list of debug categories and corresponding index numbers.

Table 2 Indexes and Categories for the debug voip h221 individual command

Index Number	Debug Category
1, 2, 30, 31, 32	Secondary number exchange
5, 6, 14, 15, 16, 22	Audio mode/caps
7, 10, 12, 13, 17, 28	Video mode/caps
8, 9, 23	B-channel mode/caps
11, 24, 33	Miscellaneous command exchange
18	Bandwidth calculations
19, 20, 21	DSP configuration
3, 4, 25, 27, 42, 43	General caps/internal
26	Non-standard caps/command
29	Loop request
34, 35, 36, 37, 38, 39, 40, 41	BAS squelch

Examples

The **raw** keyword displays the raw BAS information coming from or to the DSP. It is displayed in a hexadecimal octet format. The **decode** option decodes the BAS information into a readable English format.

The following is sample output from the **debug voip h221 raw decode** command:

```
BAS=81:1 0 0 0 0 0 0 1: AUDIO CAPS=g711 a-law
BAS=82:1 0 0 0 0 0 1 0: AUDIO CAPS=g711 u-law
BAS=84:1 0 0 0 0 1 0 0: AUDIO CAPS=g722 48k
BAS=85:1 0 0 0 0 1 0 1: AUDIO CAPS=g728
BAS=F9:1 1 1 1 1 0 0 1: H.242 MBE start indication
BAS=02:0 0 0 0 0 0 1 0: H.242 MBE length=2
BAS=0A:0 0 0 0 1 0 1 0: H.242 MBE type=H.263 caps
BAS=8A:1 - - - - - - -: Always 1
BAS=8A:- 0 0 0 1 - - -: H.263 MPI=1
BAS=8A:- - - - - 0 1 -: H.263 FORMAT=h.263_cif
BAS=8A:- - - - - - - 0: No additional options
```

Related Commands

Command	Description
debug voip ccapi	Enables debugging for the call control application programming interface (CCAPI) contents.
debug voip rtp	Enables debugging for Real-Time Transport Protocol (RTP) named event packets.

index (voice class)

To define one or more numbers for a voice class called number, or a range of numbers for a voice class called number pool, use the **index** command in voice class configuration mode. To remove the number or range of numbers, use the **no** form of this command.

index *number called-number*

no index *number called-number*

Syntax Description

number	Digits that identify this index. Range is 1 to 2147483647.
called-number	Specifies a called number, or a range of called numbers, in E.164 format.

Command Default

No index is configured.

Command Modes

Voice class configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to define one or more numbers for a voice class called number, or a range of numbers for a voice class called number pool. You can define multiple indexes for any inbound or outbound voice class called number or voice class called number pool.

When defining a range of numbers for a called number pool:

- The range of numbers must be in E.164 format.
- The beginning number and ending number must be the same length.
- The last digit of each number must be 0 to 9.
- Leading '+' (if used) must be defined from in the range of called numbers.

Examples

The following example shows the configuration for indexes in voice class called number pool 100:

```
voice class called number pool 100
 index 1 4085550100 - 4085550111 (Range of called numbers are 4085550100 up to 4085550111)
 index 2 +3227045000
```

The following example shows configuration for indexes in voice class called number outbound 222:

```
voice class called number outbound 222
 index 1 4085550101
 index 2 4085550102
 index 2 4085550103
```

Related Commands

Command	Description
voice class called number	One or more called numbers configured for a voice class.

information-type

To select a specific information type for a Voice over IP (VoIP) or plain old telephone service (POTS) dial peer, use the **information-type** command in dial-peer configuration mode. To remove the current information type setting, use the **no** form of this command. To return to the default configuration, use the **default** form of this command.

information-type {**fax** | **voice** | **video**}

no information-type

default information-type

Syntax Description

fax	The information type is set to store-and-forward fax.
voice	The information type is set to voice. This is the default.
video	The information type is set to video.

Command Default

Voice

Command Modes

Dial peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.4(11)T	The **video** keyword was added.

Usage Guidelines

The **fax** keyword applies to both on-ramp and off-ramp store-and-forward fax functions.

Examples

The following example shows the configuration for information type fax for VoIP dial peer 10:

```
dial-peer voice 10 voip
 information-type fax
```

The following example shows the configuration for information type video for POTS dial peer 22:

```
dial-peer voice 22 pots
 information-type video
```

Related Commands

Command	Description
isdn integrate calltype all	Enables integrated mode (for data, voice, and video) on ISDN BRI or PRI interfaces.

rtp payload-type

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

rtp payload-type {**cisco-cas-payload** *number* | **cisco-clear-channel** *number* | **cisco-codec-fax-ack** *number* | **cisco-codec-fax-ind** *number* | **cisco-codec-video-263+** *number* | **cisco-codec-video-264** *number* | **cisco-fax-relay** *number* | **cisco-pcm-switch-over-alaw** *number* | **cisco-pcm-switch-over-ulaw** *number* | **cisco-rtp-dtmf-relay** *number* | **nte** *number* | **nse** *number*} [**comfort-noise {13 | 19**}]

no rtp payload-type {**cisco-cas-payload | cisco-clear-channel | cisco-codec-fax-ack | cisco-codec-fax-ind | cisco-codec-video-263+ | cisco-codec-video-264 | cisco-fax-relay | cisco-pcm-switch-over-alaw | cisco-pcm-switch-over-ulaw | cisco-rtp-dtmf-relay | nte | nse** }

Syntax Description

cisco-cas-payload *number*	Cisco CAS RTP payload.
cisco-clear-channel *number*	Cisco clear-channel RTP payload.
cisco-codec-fax-ack *number*	Cisco codec fax acknowledge.
cisco-codec-fax-ind *number*	Cisco codec fax indication.
cisco-codec-video-h263+	RTP video codec H.263+ payload type.
cisco-codec-video-h264	RTP video codec H.264 payload type.
cisco-fax-relay *number*	Cisco fax relay.
cisco-pcm-switch-over-alaw *number*	Cisco RTP PCM codec switch over indication (a-law).
cisco-pcm-switch-over-ulaw *number*	Cisco RTP PCM codec switch over indication (mu-law).
cisco-rtp-dtmf-relay *number*	Cisco RTP DTMF relay.
nte *number*	Named telephone event (NTE).
nse *number*	Named signaling event (NSE).
comfort-noise	(Optional) RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the IETF AVT working group, designates 13 as the payload type for comfort noise. Previous Cisco equipment uses 19 as the payload type for comfort noise. If you are connecting to a gateway that complies with the *RTP Payload for Comfort Noise* draft, use 13. Use 19 only if you are connecting to older Cisco gateways that use DSPware earlier than version 3.4.32.

Command Default

No RTP payload type is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)T	This command was introduced.
12.2(2)XB	The **nte** and **comfort-noise** keywords were introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(11)T	The **cisco-codec-video-h263+** and **cisco-codec-video-h264** keywords were added.

Usage Guidelines

Use this command to identify the payload type of an RTP packet. For all payload types, the *number* range is 96 to 127 and the default is 101, with the exception of the video codec payload types:

- For payload type cisco-codec-video-h263+, the default *number* is 119.
- For payload type cisco-codec-video-h264, the default *number* is 120.

For Session Initiation Protocol (SIP) calls, use this command after using the **dtmf-relay** command to choose the NTE method of dual-tone multifrequency (DTMF) relay.

Examples

The following command configuration identifies the RTP payload type as NTE 99:

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

The following command configuration identifies the RTP payload type as cisco-codec-video-h264:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-video-h264
```

Related Commands

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

show call active video

To display call information for Signaling Connection Control Protocol (SCCP), Session Initiation Protocol, (SIP), and H.323 video calls in progress, use the **show call active video** command in user EXEC or privileged EXEC mode.

show call active video [**brief** | **compact** | **echo-canceller** *call-id* | **id** *identifier*]

Syntax Description

brief	(Optional) Displays a truncated version of active video call information.
compact	(Optional) Displays a compact version of active video call information.
echo-canceller *call-id*	(Optional) Displays information about the state of the extended echo canceller (EC). To query the echo state, you need to know the hexadecimal ID in advance. To find the hexadecimal ID, enter the **show call active video brief** command. Range is 0 to FFFFFFFF.
id *identifier*	(Optional) Displays only the video call with the specified identifier. Range is a hexadecimal value from 1 to FFFF.

Command Default

No default behavior or values.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to display the contents of the active video call table.

Examples

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

```
Router # show call active video

Telephony call-legs: 7
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 7

 GENERIC:
SetupTime=903690 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
```

```
PeerIfIndex=106
LogicalIfIndex=12
ConnectTime=906160 ms
CallDuration=00:21:33 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=64654
TransmitBytes=10861872
ReceivePackets=129336
ReceiveBytes=10346880
TELE:
ConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=10
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
DSPIdentifier=2/1:1
VIDEO:
H320CallType=Primary
VideoTransmitCodec=H263
VideoReceiveCodec=H263
VideoUsedBandwidth=384
H221 STATS (AUDIO):
 TxPackets=129236
 TxDuration=1292360 ms
 RxPackets=64604
 RxDuration=1291990 ms
 BadHeaders=0
 PacketsLate=0
 PacketsEarly=1
 ReceiveDelay=85 ms
 ConcealmentDuration=0 ms
 BufferOverflowDiscards=10
```

```
H221 STATS (VIDEO):
 TxPackets=7693
 TxBytes=8214946
 PSC=6324
 GBSC=8401
 TxVideoFormat=3
 RxPackets=9514
 RxBytes=8185670
 VideoBytesConsumed=8117148
 FillBytesConsumed=40898670
 PSCPacketDrops=0
 LatePacket=0
 OutOfSequence=0
 BadHeader=0
 BadSSRC=0
 BadPayloadType=0
 BufferOverflow=0
 ControlHeaderOverflow=0
 FilteredDelay=250 ms
 MinimumDelay=43 ms
 MaximumDelay=1858 ms
 RxVideoFormat=3


 GENERIC:
SetupTime=903700 ms
Index=1
PeerAddress=7001
PeerSubAddress=
PeerId=20006
PeerIfIndex=127
LogicalIfIndex=126
ConnectTime=906150 ms
CallDuration=00:21:35 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=0
TransmitBytes=0
ReceivePackets=64768
ReceiveBytes=10362880
TELE:
ConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=11
TxDuration=1294180 ms
VoiceTxDuration=1294180 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=2
ERLLevel=0
EchoCancellerMaxReflector=62709
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
AlertTimepoint=903700 ms
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
```

```
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
GwOutpulsedCallingNumber=555556
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x80
VIDEO:
H320CallType=None
VideoTransmitCodec=None
VideoReceiveCodec=None
VideoCap_Codec=H263
VideoCap_Format=CIF
VideoUsedBandwidth=3101


 GENERIC:
SetupTime=903910 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
PeerIfIndex=106
LogicalIfIndex=13
ConnectTime=906160 ms
CallDuration=00:21:36 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x918888CA 0x34D311DA 0x80090012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=12
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
```

```
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
VIDEO:
H320CallType=Secondary


 GENERIC:
SetupTime=904230 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
PeerIfIndex=106
LogicalIfIndex=14
ConnectTime=906160 ms
CallDuration=00:21:37 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x91B95C6E 0x34D311DA 0x800A0012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=13
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
```

```
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
VIDEO:
H320CallType=Secondary


 GENERIC:
SetupTime=904550 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
PeerIfIndex=106
LogicalIfIndex=15
ConnectTime=906160 ms
CallDuration=00:21:40 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x91EA317E 0x34D311DA 0x800B0012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=14
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
```

```
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
VIDEO:
H320CallType=Secondary


 GENERIC:
SetupTime=904870 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
PeerIfIndex=106
LogicalIfIndex=16
ConnectTime=906160 ms
CallDuration=00:21:41 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x921B0522 0x34D311DA 0x800C0012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=15
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
VIDEO:
H320CallType=Secondary


 GENERIC:
```

```
SetupTime=905190 ms
Index=1
PeerAddress=555556
PeerSubAddress=
PeerId=7001
PeerIfIndex=106
LogicalIfIndex=17
ConnectTime=906160 ms
CallDuration=00:21:42 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=video
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=0
TELE:
ConnectionId=[0x924BD82D 0x34D311DA 0x800D0012 0x803F3110]
IncomingConnectionId=[0x9166F770 0x34D311DA 0x80080012 0x803F3110]
CallID=16
TxDuration=0 ms
VoiceTxDuration=0 ms
FaxTxDuration=0 ms
CoderTypeRate=None
NoiseLevel=0
ACOMLevel=0
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=0
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=555556
OriginalCallingOctet=0x0
OriginalCalledNumber=7001
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0xFF
TranslatedCallingNumber=555556
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7001
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=7001
GwReceivedCalledOctet3=0x80
GwReceivedCallingNumber=555556
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
VIDEO:
H320CallType=Secondary

Telephony call-legs: 7
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 7
```

Table 3 describes significant fields shown in this output.

Table 3 show call active video Field Descriptions

Field	Description
VideoCap_Codec	Codec for the active video call.
VideoCap_Format	Video format for the active video call.
VideoEarlyPackets	Number of early packets for a video call.
VideoLatePackets	Number of late packets in a video call.
VideoLostPackets	Number of lost packets in a video call.
VideoNumberOfChannels	Number of channels used for a video call.
VideoUsedBandwidth	Bandwidth, in kbps, used for a video call.

Related Commands

Command	Description
show call history video	Displays call history information for SCCP video calls.

show dial-peer voice

To display information for voice dial peers, use the **show dial-peer voice** command in user EXEC or privileged EXEC mode.

show dial-peer voice [*number* | **summary**]

Syntax Description

number	(Optional) A specific voice dial peer. Output displays detailed information about that dial peer.
summary	(Optional) Output displays a short summary of each voice dial peer.

Command Default

If both the *name* argument and **summary** keyword are omitted, output displays detailed information about all voice dial peers.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	The **summary** keyword was added for Cisco MC3810.
12.0(3)XG	This command was implemented for Voice over Frame Relay (VoFR) on the Cisco 2600 series and Cisco 3600 series.
12.0(4)T	This command was implemented for VoFR on the Cisco 7200 series.
12.1(3)T	This command was implemented for Modem Passthrough over VoIP on the Cisco AS5300.
12.2(2)XB	This command was modified to support VoiceXML applications.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager 3.2 and implemented on the and Cisco IAD2420.
12.4(11)T	This command was enhanced to display configuration information for bandwidth, video codec, and rtp payload-type for H.263+ and H.264 video codec.

Usage Guidelines

Use this command to display the configuration for all VoIP and POTS dial peers configured for a gateway. To show configuration information for only one specific dial peer, use the *number* argument to identify the dial peer. To show summary information for all dial peers, use the **summary** keyword.

Examples

The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
Router# show dial-peer voice 100

VoiceEncapPeer3201
peer type = voice, information type = video,
description = `',
tag = 3201, destination-pattern = `86001',
answer-address = `', preference=0,
CLID Restriction = None
CLID Network Number = `'
CLID Second Number sent
CLID Override RDNIS = disabled,
source carrier-id = `',target carrier-id = `',
source trunk-group-label = `',target trunk-group-label = `',
numbering Type = `unknown'
group = 3201, Admin state is up, Operation state is up,
Outbound state is up,
incoming called-number = `', connections/maximum = 0/unlimited,
DTMF Relay = disabled,
URI classes:
        Destination =
huntstop = disabled,
in bound application associated: 'DEFAULT'
out bound application associated: ''
dnis-map =
permission :both
        incoming COR list:maximum capability
outgoing COR list:minimum requirement
Translation profile (Incoming):
Translation profile (Outgoing):
incoming call blocking:
translation-profile = `'
disconnect-cause = `no-service'
advertise 0x40 capacity_update_timer 25 addrFamily 4 oldAddrFamily 4
type = pots, prefix = `',
forward-digits 4
session-target = `', voice-port = `2/0:23',
direct-inward-dial = enabled,
digit_strip = enabled,
register E.164 number with H323 GK and/or SIP Registrar = TRUE
fax rate = system,   payload size =  20 bytes
supported-language = ''
preemption level = `routine'
bandwidth:
        maximum = 384 KBits/sec, minimum = 64 KBits/sec
voice class called-number:
        inbound = `', outbound = `1'
Time elapsed since last clearing of voice call statistics never
        Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
```

The following is sample output from this command for a VoIP dial peer:

```
Router# show dial-peer voice 101

VoiceOverIpPeer101
        peer type = voice, information type = voice,
        description = `',
        tag = 6001, destination-pattern = `6001',
```

```
answer-address = `', preference=0,
CLID Restriction = None
CLID Network Number = `'
CLID Second Number sent
CLID Override RDNIS = disabled,
source carrier-id = `', target carrier-id = `',
source trunk-group-label = `',  target trunk-group-label = `',
numbering Type = `unknown'
group = 6001, Admin state is up, Operation state is up,
incoming called-number = `', connections/maximum = 0/unlimited,
DTMF Relay = disabled,
modem transport = system,
URI classes:
    Incoming (Called) =
    Incoming (Calling) =
    Destination =
huntstop = disabled,
in bound application associated: 'DEFAULT'
out bound application associated: ''
dnis-map =
permission :both
incoming COR list:maximum capability
outgoing COR list:minimum requirement
Translation profile (Incoming):
Translation profile (Outgoing):
incoming call blocking:
translation-profile = `'
disconnect-cause = `no-service'
advertise 0x40 capacity_update_timer 25 addrFamily 4 oldAddrFamily 4
type = voip, session-target = `ipv4:1.7.50.50',
technology prefix:
settle-call = disabled
ip media DSCP = ef, ip signaling DSCP = af31,
ip video rsvp-none DSCP = af41,ip video rsvp-pass DSCP = af41
ip video rsvp-fail DSCP = af41,
UDP checksum = disabled,
session-protocol = cisco, session-transport = system,
req-qos = best-effort, acc-qos = best-effort,
req-qos video = best-effort, acc-qos video = best-effort,
req-qos audio def bandwidth = 64, req-qos audio max bandwidth = 0,
req-qos video def bandwidth = 384, req-qos video max bandwidth = 0,
RTP dynamic payload type values: NTE = 101
Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121, fax-relay=122
       CAS=123, ClearChan=125, PCM switch over u-law=0,A-law=8
       h263+=118, h264=119
RTP comfort noise payload type = 19
fax rate = fax,   payload size =  20 bytes
fax protocol = system
fax-relay ecm enable
fax NSF = 0xAD0051 (default)
codec = g711ulaw,   payload size =  160 bytes,
video codec = h263+
voice class codec = `'
Media Setting = flow-through (global)
Expect factor = 10, Icpif = 20,
Playout Mode is set to adaptive,
Initial 60 ms, Max 250 ms
Playout-delay Minimum mode is set to default, value 40 ms
Fax nominal 300 ms
Max Redirects = 1, signaling-type = cas,
VAD = enabled, Poor QOV Trap = disabled,
Source Interface = NONE
voice class sip url = system,
voice class sip rel1xx = system,
```

```
redirect ip2ip = disabled
probe disabled,
voice class perm tag = `'
Time elapsed since last clearing of voice call statistics never
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
```

Table 4 describes significant fields shown in this output.

Table 4 show dial-peer voice Field Descriptions

Field	Description
Accepted Calls	Number of calls accepted from this peer since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
answer-address	Answer address configured for this dial peer.
bandwidth maximum/minimum	The maximum and minimum bandwidth.
Charged Units	Total number of charging units that have applied to this peer since system startup, in hundredths of a second.
CLID Restriction	Indicates if CLID restriction is enabled.
CLID Network Number	Displays the network number sent as CLID, if configured.
CLID Second Number sent	Displays whether a second calling number is stripped from the call setup.
CLID Override RDNIS	Indicates whether the CLID is overridden by the redirecting number.
codec	Default voice codec rate of speech.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls, in hundredths of a second.
connections/maximum	Indicates maximum call connections per peer
Destination	Indicates the voice class which is used to match destination url
destination-pattern	Destination pattern (telephone number) for this peer.
digit_strip	Indicates if digit stripping is enabled.
direct-inward-dial	Indicates if direct-inward-dial is enabled.
disconnect-cause	Indicates the disconnect cause code to be used when an incoming call is blocked
dnis-map	Name of the dialed-number identification service (DNIS) map.
DTMF Relay	Indicates if dual-tone multifrequency (DTMF) relay is enabled.
Expect factor	User-requested expectation factor of voice quality for calls through this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
fax rate	Fax transmission rate configured for this peer.

Table 4 show dial-peer voice Field Descriptions (continued)

Field	Description
forward-digits	Indicates the destination digits to be forwarded of this peer
group	Group number associated with this peer.
huntstop	Indicates whether dial-peer hunting is turned on, by using the huntstop command, for this dial peer.
Icpif	Configured calculated planning impairment factor (ICPIF) value for calls sent by a dial peer.
in bound application associated	Interactive voice response (IVR) application that is configured to handle inbound calls to this dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
incoming call blocking	Indicates the incoming call blocking setup of this peer
incoming called-number	Indicates the incoming called number if it has been set.
incoming COR list	Indicates the level of Class of Restrictions for incoming calls of this peer
Incomplete calls	Indicates number of outgoing disconnected calls with user busy (17), no user response (18) or no answer (19) cause code
information type	Information type for this call (voice, fax, video)
Last Disconnect Cause	Encoded network cause associated with the last call. This value is updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the system uptime when the last call to this peer was started.
Modem passthrough	Modem pass-through signaling method is named signaling event (NSE).
numbering type	Indicates the numbering type for a peer call leg
Operation state	Operational state of this peer.
outgoing COR list	Indicates the level of Class of Restrictions for outgoing calls of this peer
outbound application associated	The voice application that is configured to handle outbound calls from this dial peer. Outbound calls are handed off to the named application.
Outbound state	Indicates the current outbound status of a POTS peer
payload size	Indicates the size of payload of fax rate or codec setup
Payload type	NSE payload type.
peer type	Dial peer type (voice, data).
permission	Configured permission level for this peer.
Poor QOV Trap	Indicates if poor quality of voice trap messages is enabled.
preemption level	Indicates the call preemption level of this peer
prefix	Indicates dialed digits prefix of this peer

Table 4 show dial-peer voice Field Descriptions (continued)

Field	Description
Redundancy	Packet redundancy (RFC 2198) for modem traffic.
Refused Calls	Number of calls from this peer refused since system startup.
register E.164 number with H.323 GK and/or SIP Registrar	Indicates "register e.164" option of this peer
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote routers through the IP backbone.
source carrier-id	Indicates source carrier-id of this peer which will be used to match the source carrier-id of an incoming call
source trunk-group label	Indicates source trunk-group-label of this peer which can be used to match the source trunk-group-label of an incoming call
Successful Calls	Number of completed calls to this peer.
supported-language	Indicates list of supported languages of this peer
tag	Unique dial peer ID number.
target carrier-id	Indicates target carrier-id of this peer which will be used to match the target carrier-id for an outgoing call
target trunkgroup label	Indicates target trunk-group-label of this peer which can be used to match the target trunk-group-label of an outgoing call
Time elapsed since last clearing of voice call statistics	Elapsed time between the current time and the time when the "clear dial-peer voice" command was executed
Translation profile (Incoming)	Indicate translation profile for incoming calls
Translation profile (Outgoing)	Indicate translation profile for outgoing calls
translation-profile	Indicate number translation profile of this peer
type	Indicate peer encapsulation type such as pots, voip, vofr, voatm or mmoip
VAD	Whether voice activation detection (VAD) is enabled for this dial peer.
voice class called-number inbound/outbound	Indicates voice-class called-number inbound or outbound setup of this peer
voice-port	Indicates the voice interface setting of this POTS peer

The following is sample output from this command with the **summary** keyword:

```
Router# show dial-peer voice summary

dial-peer hunt 0
                                                      PASS
  TAG TYPE   ADMIN OPER PREFIX   DEST-PATTERN     PREF THRU SESS-TARGET    PORT
  100 pots   up    up                              0
  101 voip   up    up            5550112           0   syst ipv4:10.10.1.1
  102 voip   up    up            5550134           0   syst ipv4:10.10.1.1
   99 voip   up    down                            0   syst
   33 pots   up    down                            0
```

Table 5 describes significant fields shown in this output.

Table 5 show dial-peer voice summary Field Descriptions

Field	Description
dial-peer hunt	Hunt group selection order that is defined for the dial peer by using the **dial-peer hunt** command.
TAG	Unique identifier assigned to the dial peer when it was created.
TYPE	Type of dial peer: POTS, VoIP, VoFR, VoATM, or MMoIP.
ADMIN	Whether the administrative state is up or down.
OPER	Whether the operational state is up or down.
PREFIX	Prefix that is configured in the dial peer by using the **prefix** command.
DEST-PATTERN	Destination pattern that is configured in the dial peer by using the **destination-pattern** command.
PREF	Hunt group preference that is configured in the dial peer by using the **preference** command.
PASS THRU	Modem pass-through method that is configured in the dial peer by using the **modem passthrough** command.
SESS-TARGET	Destination that is configured in the dial peer by using the **session target** command.
PORT	Router voice port that is configured for the dial peer. Valid only for POTS dial peers.

Related Commands

Command	Description
show call active voice	Displays the VoIP active call table.
show call history voice	Displays the VoIP call history table.
show dialplan incall number	Displays which POTS dial peer is matched for a specific calling number or voice port.
show dialplan number	Displays which dial peer is reached when a specific telephone number is dialed.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

show voice class called-number

To display a specific voice class called-number, use the **show voice class called-number** command in privileged EXEC mode.

show voice class called-number [**inbound** | **outbound**] *tag*

Syntax Description

inbound	Displays the specified inbound voice class called-number.
outbound	Displays the specified outbound voice class called-number.
tag	Digits that identify this voice class called-number.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to display a specific inbound or outbound voice class called-number.

Examples

The following is sample output from this command:

```
Router# show voice class called-number outbound 200
Called Number Outbound: 200
           index 1      4085550100
           index 2      4085550102
           index 3      4085550103
           index 4      4085550104
```

Table 6 describes significant fields shown in the display.

Table 6 show voice class called-number Field Descriptions

Field	Description
Called Number Inbound/Outbound	The tag for the specified inbound or outbound voice class called-number.
index *number*	The number or range of numbers for this voice class called number.

Related Commands

Command	Description
show voice class called-number-pool	Displays voice class called number pool configuration information.

show voice class called-number-pool

To display a voice class called-number pool, use the **show voice class called-number-pool** command in privileged EXEC mode.

show voice class called-number-pool *tag* [**detail**]

Syntax Description

tag	Digits that identify this voice class called-number-pool. Range is 1 to 10000.
detail	Displays idle called number and allocated called number information.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to display the voice class called number pool configuration information. The **detail** keyword displays up to 16 idle called numbers, and up to 4 allocated called numbers for each allocated request.

Examples

The following sample output displays configuration information for voice class called-number-pool 100, including idle called numbers and allocated called numbers:

```
Router(config)# show voice class called-number-pool 100 detail

Called Number Pool: 100
index 1 100A11 - 100A20
index 2 200#55 - 200#77
index 3 5551111 - 6662333
index 99 123C11 - 123C99
All called numbers are generated from table: FALSE
No of idle called numbers: 16
List of idle called numbers:
100A11 100A12 .. Display up to 16 idle called number from the pool
100A13 100A14
100A15 100A16
100A17 100A18
100A19 100A20
200#55 200#56
200#57 200#58
200#59 200#60
No of alloc requests : 1
Ref Id Alloc PC Size
2 41F84190 16
List of alloc called numbers: .. Display the first 4 allocated called number for RefId 2
200#61 200#62
200#63 200#64
```

Table 7 describes significant fields shown in the display.

Table 7 show voice class called-number-pool Field Descriptions

Field	Description
Called Number Pool	Tag that identifies the called number pool.
index	Number or range of numbers for this called number pool.
All called numbers are generated from table	• FALSE—Numbers are not generated from called number table. • TRUE—Numbers are generated from called number table.
No. of idle called numbers	Number of idle called numbers in the called number pool.
List of idle called numbers	List of idle numbers in the called number pool.
No. of alloc requests	Number of requests for numbers from the called number pool.
Ref Id Alloc PC Size	Reference ID for a specific list of allocated numbers.
List of alloc called numbers	List of first four allocated numbers from the called number pool.

Related Commands

Command	Description
show voice class called-number	Displays a specific voice class called-number.

show voice dsp

To show the current status of all digital signal processor (DSP) voice channels, use the **show voice dsp** command in privileged EXEC mode.

show voice dsp

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series, and the display format was modified.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.3(14)T	Command output was enhanced to display status information for NM-HDV network module TI-549 DSPs.
12.4(4)T	Command output was enhanced to display codec setting for modem relay operation.
12.4(11)T	Command output was enhanced to display information about DSP H.320 channels.

Usage Guidelines

Use this command if abnormal behavior occurs in the DSP voice channels.

Examples

The following sample output shows the current status of the codec, set for modem relay, on channel 1.

```
Router# show voice dsp

----------------------------FLEX VOICE CARD 1 ------------------------------
                           *DSP VOICE CHANNELS*
DSP   DSP              DSPWARE CURR  BOOT                         PAK   TX/RX
TYPE  NUM CH CODEC     VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 01 modem-re 4.5.909 busy  idle      0  0 1/1/0     05    0      298/353
                           *DSP SIGNALING CHANNELS*
DSP   DSP              DSPWARE CURR  BOOT                         PAK   TX/RX
TYPE  NUM CH CODEC     VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 05 {flex}   4.5.909 alloc idle      0  0 1/1/3     02    0         15/0
C5510 001 06 {flex}   4.5.909 alloc idle      0  0 1/1/2     02    0         17/0
C5510 001 07 {flex}   4.5.909 alloc idle      0  0 1/1/1     06    0         31/0
C5510 001 08 {flex}   4.5.909 alloc idle      0  0 1/1/0     06    0        321/0
------------------------END OF FLEX VOICE CARD 1 ----------------------------
```

The following sample output shows the current status of all DSP voice channels:

```
Router# show voice dsp

DSP# 0, channel# 0 G729A BUSY
DSP# 0, channel# 1 G729A BUSY
DSP# 1, channel# 2 FAX IDLE
DSP# 1, channel# 3 FAX IDLE
DSP# 2, channel# 4 NONE BAD
DSP# 2, channel# 5 NONE BAD
DSP# 3, channel# 6 NONE BAD
DSP# 3, channel# 7 NONE BAD
DSP# 4, channel# 8 NONE BAD
DSP# 4, channel# 9 NONE BAD
DSP# 5, channel# 10 NONE BAD
DSP# 5, channel# 11 NONE BAD
```

The following is sample output from this command on a Cisco 1750 router:

```
Router# show voice dsp

DSP#0: state IN SERVICE, 2 channels allocated
channel#0: voice port 1/0, codec G711 ulaw, state UP
channel#1: voice port 1/1, codec G711 ulaw, state UP
DSP#1: state IN SERVICE, 2 channels allocated
channel#0: voice port 2/0, codec G711 ulaw, state UP
channel#1: voice port 2/1, codec G711 ulaw, state UP
DSP#2: state RESET, 0 channels allocated
```

The following is sample output from this command on a secure Cisco Survivable Remote Site Telephony (Cisco SRST) router with the NM-HDV network module and the TI-549 (C549) DSP installed:

```
Router# show voice dsp

DSP  DSP    DSPWARE  CURR     BOOT                               PAK   TX/RX
TYPE NUM CH CODEC    VERSION  STATE STATE   RST AI VOICEPORT TS ABORT PACK COUNT
==== === == ======== ======= ===== ======= === == ======== === ==== ===========
C549  1  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0   1   0    9357/9775
C549  1  02 {medium} 4.4.3    IDLE  idle     0      1/0:0   2   0    0/0
C549  2  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0   3   0    0/0
C549  2  02 {medium} 4.4.3    IDLE  idle     0      1/0:0   4   0    0/0
C549  3  01 {medium} 4.4.3    IDLE  idle     0  0   1/0:0   5   0    0/13
C549  3  02 {medium} 4.4.3    IDLE  idle     0      1/0:0   6   0    0/13
```

The following is sample output from this command for an H.320 network configured for video support:

```
Router# show voice dsp

DSP  DSP             DSPWARE CURR  BOOT                              PAK     TX/RX
TYPE NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABORT  PACK COUNT
==== === == ======== ======= ===== ======= === == ========= == ===== ============ edsp 001
01 g711ulaw  0.1 IDLE  50/0/1.1 edsp 002 02 g711ulaw  0.1 IDLE  50/0/1.2 edsp 003 01
g729r8 p  0.1 IDLE  50/0/2.1 ----------------------------FLEX VOICE CARD 1
------------------------------
                           *DSP VOICE CHANNELS*
DSP   DSP             DSPWARE CURR  BOOT                              PAK   TX/RX
TYPE  NUM CH CODEC    VERSION STATE STATE   RST AI VOICEPORT TS ABRT PACK COUNT
===== === == ======== ======= ===== ======= === == ========= == ==== ============
C5510 001 05 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 06 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 07 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 08 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 09 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 10 None     9.0.105 idle  idle      0  0                 0          0/0
C5510 001 11 None     9.0.105 idle  idle      0  0                 0          0/0
```

```
C5510 001 12 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 13 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 14 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 15 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 001 16 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 01 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 02 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 03 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 04 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 05 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 06 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 07 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 08 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 09 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 10 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 11 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 12 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 13 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 14 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 15 None     9.0.105 idle  idle      0  0                  0          0/0
C5510 003 16 None     9.0.105 idle  idle      0  0                  0          0/0
                                   *DSP H.320 CHANNELS*
DSP   DSP      TX/RX       DSPWARE CURR                     PAK   TX/RX
TYPE  NUM CH   CODEC       VERSION STATE VOICEPORT TS ABRT PACK COUNT
===== === === =========== ======= ===== ========= == ==== ============
C5510 001 01   h320p(01)   9.0.105 busy  1/0/0:15  06
      001 02   h320s(02)   9.0.105 busy  1/0/0:15  07
      001 03   h320s(03)   9.0.105 busy  1/0/0:15  08
      001 04   h320s(04)   9.0.105 busy  1/0/0:15  09
      001 01a g711ulaw     9.0.105 busy                 0 1013663/5083
                                                                      00
      001 01v h263  /h263  9.0.105 busy                 0 104908/30911
                                                                      4
------------------------END OF FLEX VOICE CARD 1 ---------------------------
```

Table 8 describes significant fields shown in the output.

Table 8 show voice dsp Field Descriptions

Field	Description
DSP	Number of the DSP.
channel	Number of the channel and its status.
DSP TYPE	TI-549 (C549) DSP.
DSP NUM	Number of the DSP.
CH	Channel number.
CODEC	Complexity setting.
DSPWARE VERSION	Version of DSPware.
CURR STATE	Current status of the channel, either IDLE or BUSY.
BOOT STATE	DSP readiness, either idle or in service.
RST	Number of times the DSP has been reset or restarted.
AI	Alarm indication count on the channel.
VOICEPORT	Voice card number and slot.
TS	Time slot.

Table 8 show voice dsp Field Descriptions (continued)

Field	Description
PAK ABORT	Number of dropped packets.
TX/RX PACKCOUNT	Number of transmitted and received packets

Related Commands

Command	Description
clear counters	Clears all the current interface counters from the interface.
show dial-peer voice	Displays configuration information for dial peers.
show voice call	Displays the call status for all voice ports.
show voice port	Displays configuration information about a specific voice port.

show voice port

To display configuration information about a specific voice port, use the **show voice port** command in privileged EXEC mode.

Cisco 1750 Router

show voice port *slot/port*

Cisco 2600 and Cisco 3600 Series Router with Analog Voice Ports

show voice port [*slot/subunit/port* | **summary**]

Cisco 2600 and Cisco 3600 Series Router with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)

show voice port [*slot/port***:***ds0-group* | **summary**]

Cisco AS5300 Universal Access Server

show voice port *controller-number***:D**

Cisco 7200 Series Router

show voice port {*slot/port***:***ds0-group-no*} | {*slot/subunit/port*}

Syntax Description

Cisco 1750 Router

slot	Slot number in the router in which the voice interface card (VIC) is installed. Range is 0 to 2, depending on the slot in which it is installed.
port	Voice port. Valid entries are 0 and 1.

Cisco 2600 and Cisco 3600 Series Router with Analog Voice Ports

slot/subunit/port	(Optional) Output displays information for the analog voice port that you specify using the *slot/subunit/port* designation. • *slot*—Router slot in which a voice network module (VNM) is installed. Valid entries are router slot numbers for the specific platform. • *subunit*—Voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • *port*—Analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Output displays a summary of all voice ports.

Cisco 2600 and Cisco 3600 Series Router with Digital Voice Ports

*slot/port***:***ds0-group*	(Optional) Output displays information for the digital voice port that you specify using the *slot/port:ds0-group* designation. • *slot*—Router slot in which the packet voice trunk network module (NM) is installed. Valid entries are specific router slot numbers. • *port*—T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) • *ds0-group*— T1 or E1 logical port number. T1 range is 0 to 23. E1 range is 0 to 30.
summary	(Optional) Output displays a summary of all voice ports.

Cisco AS5300 Access Server

controller-number	T1 or E1 controller.
:D	D channel that is associated with ISDN PRI.

Cisco 7200 Series Router

slot	Router location where the voice port adapter is installed. Range is 0 to 3.
port	Voice interface card location. Valid entries are 0 and 1.
dso-group-no	Defined DS0 group number. Because each defined DS0 group number is represented on a separate voice port, you can define individual DS0s on the digital T1/E1 card.
slot	Slot number in the Cisco router where the voice interface card is installed. Range is 0 to 3, depending on the slot where it is installed.
subunit	Subunit on the voice interface card where the voice port is located. Valid entries are 0 and 1.
port	Voice port number. Valid entries are 0 and 1.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	Port-specific values for the Cisco MC3810 were added.
12.0(3)T	Port-specific values for the Cisco MC3810 were added.
12.0(5)XK	The *ds0-group* argument was added for the Cisco 2600 series and Cisco 3600 series.
12.0(5)XE	Additional syntax was created for digital voice to allow specification of the DS0 group. This command applies to VoIP on the Cisco 7200 series.
12.0(7)T	The additions were integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	The **summary** keyword was added for the Cisco 2600 series and Cisco 3600 series. The *ds0-group* argument was added for the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(8)T	This command was implemented for DID on the Cisco IAD2420 series.

Release	Modification
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco Gateway 200 (Cisco VG200).
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager 3.2. It was implemented on the Cisco IAD2420 series.
12.4(11)T	This command was enhanced to display voice class called-number-pool configuration information for the voice port.

Usage Guidelines

Use this command to display configuration and voice-interface-card-specific information about a specific port.

This command applies to Voice over IP, Voice over Frame Relay, and Voice over ATM.

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on Cisco 2600, Cisco 3600 series, and Cisco 7200 series routers: *slot*/*port***:***ds0-group-no*. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Examples

The following is sample output from the **show voice port** command for an E&M analog voice port:

```
Router# show voice port 1/0/0

E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0

 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
```

```
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for a foreign exchange station (FXS) analog voice port:

```
Router# show voice port 1/0/0

Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hook Flash Duration Timing is set to 600 ms
```

The following is sample output from the **show voice port** command for an E&M digital voice port:

```
Router# show voice port 1/0/1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
```

```
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

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

```
Router# show voice port 1/0/1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 DBMS
 In Gain is Set to 0 dBm
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

The following is sample output from the **show voice port** command for an ISDN voice port:

```
Router# show voice port

ISDN 2/0:23 Slot is 2, Sub-unit is 0, Port is 23
 Type of VoicePort is ISDN-VOICE
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 64 ms
 Echo Cancel worst case ERL is set to 6 dB
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 250 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number None
```

```
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Voice class called number pool:

DS0 channel specific status info:
                                      IN       OUT
    PORT       CH  SIG-TYPE    OPER  STATUS   STATUS    TIP     RING
    2/0:23     01  isdn-voice  up    none     none
    2/0:23     02  isdn-voice  up    none     none
    2/0:23     03  isdn-voice  up    none     none
    2/0:23     04  isdn-voice  up    none     none
    2/0:23     05  isdn-voice  up    none     none
    2/0:23     06  isdn-voice  up    none     none
    2/0:23     07  isdn-voice  dorm  none     none
    2/0:23     08  isdn-voice  dorm  none     none
    2/0:23     09  isdn-voice  dorm  none     none
    2/0:23     10  isdn-voice  dorm  none     none
    2/0:23     11  isdn-voice  dorm  none     none
    2/0:23     12  isdn-voice  dorm  none     none
    2/0:23     13  isdn-voice  dorm  none     none
    2/0:23     14  isdn-voice  dorm  none     none
    2/0:23     15  isdn-voice  dorm  none     none
    2/0:23     16  isdn-voice  dorm  none     none
    2/0:23     17  isdn-voice  dorm  none     none
    2/0:23     18  isdn-voice  dorm  none     none
    2/0:23     19  isdn-voice  dorm  none     none
    2/0:23     20  isdn-voice  dorm  none     none
    2/0:23     21  isdn-voice  dorm  none     none
    2/0:23     22  isdn-voice  dorm  none     none
    2/0:23     23  isdn-voice  dorm  none     none
```

Table 9 describes significant fields shown in each these output.

Table 9 show voice port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for the voice port.
Analog interface A-D gain offset	Gain offset for analog-to-digital conversion.
Analog interface D-A gain offset	Gain offset for digital-to-analog conversion.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Coder Type	Voice compression mode used.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
Description	Description of the voice port.

Table 9 show voice port Field Descriptions (continued)

Field	Description
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF digit duration, in milliseconds.
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo cancel coverage for this port.
Echo Cancellation	Whether echo cancellation is enabled for this port.
Hook Flash Duration Timing	Maximum length of hookflash signal.
Hook Status	Hook status of the FXO/FXS interface.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
InterDigit Duration Timing	DTMF interdigit duration, in milliseconds.
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing, in milliseconds.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Maintenance Mode	Maintenance mode of the voice port.
Maximum Playout Delay	The amount of time before the digital signal processor (DSP) starts to discard voice packets from the digital DSP buffer.
Music On Hold Threshold	Configured music-on-hold threshold value for this interface.
Noise Regeneration	Whether background noise should be played to fill silent gaps if VAD is activated.
Nominal Playout Delay	The amount of time the DSP waits before starting to play out the voice packets from the DSP buffer.
Non Linear Processing	Whether nonlinear processing is enabled for this port.
Number of signaling protocol errors	Number of signaling protocol errors.
Operation State	Operational state of the voice port.
Operation Type	Operation type of the E&M signal: two-wire or four-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for the interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate, in pulses per second (pps).
Region Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Cadence	Configured ring cadence for this interface.

Table 9 show voice port Field Descriptions (continued)

Field	Description
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing timeout duration.
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, or E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Voice Activity Detection	Whether voice activity detection is enabled or disabled.
Wait Release Time Out	Length of time that a voice port stays in call-failure state while a busy tone, reorder tone, or out-of-service tone is sent to the port.
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

video codec (dial-peer)

To assign a video codec to a VoIP dial peer, use the **video codec** command in dial-peer configuration mode. To remove a video codec, use the **no** form of this command.

video codec {h261 | h263 | h263+ | h264}

no video codec

Syntax Description

h261	Video codec H.261
h263	Video codec H.263
h263+	Video codec H.263+
h264	Video codec H.264

Command Default

No video codec is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to configure a video codec for a VoIP dial peer. If no video codec is configured, the default is transparent codec operation between the endpoints.

Examples

The following example shows configuration for video codec H.263+ on VoIP dial peer 30:

```
dial-peer voice 30 voip
 video codec h263+
```

Related Commands

Command	Description
video codec (voice-class)	Specifies a video codec for a voice class.

video codec (voice-class)

To specify a video codec for a voice class, use the **video codec** command in voice class configuration mode. To remove the video codec, use the **no** form of this command.

video codec {h261 | h263 | h263+ | h264}

no video codec {h261 | h263 | h263+ | h264}

Syntax Description

h261	Apply this preference to video codec H.261
h263	Apply this preference to video codec H.263
h263+	Apply this preference to video codec H.263+
h264	Apply this preference to video codec H.264

Command Default

No video codec is configured.

Command Modes

Voice class configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to specify one or more video codecs for a voice class.

Examples

The following example shows configuration for voice class codec 10 with two audio codec preferences and three video codec preferences:

```
voice class codec 10
 codec preference 1 g711alaw
 codec preference 2 g722
 video codec h261
 video codec h263
 video codec h264
```

Related Commands

Command	Description
video codec (dial-peer)	Specifies a video codec for a VoIP dial peer.

voice class called number

To define a voice class called number or range of numbers, use the **voice class called number** command in global configuration mode. To remove a voice class called number, use the **no** form of this command.

voice class called number {inbound | outbound | pool} *tag*

no voice class called number

Syntax Description

inbound	Inbound voice class called number.
outbound	Outbound voice class called number.
pool	Voice class called number pool.
tag	Digits that identify a specific inbound or outbound voice class called number or voice class called number pool.

Command Default

No voice class called number is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to define one or more static voice class called numbers for inbound and outbound POTS dial peers or a dynamic voice class called number pool. The indexes for a voice class called number are defined with the index (voice class) command.

To configure the gateway to use the same called number as both primary and secondary numbers for an H.320 call, configure an outbound called-number voice-class with no index defined and apply it to the outbound POTS dial-peer as follows:

```
voice class called-number outbound 1
dial-peer voice 1 pots
  voice-class called-number outbound 1
```

Note Enter the **voice class called number** command in global configuration mode without hyphens. Enter the **voice-class called-number** command in dial-peer configuration mode with hyphens.

Examples

The following example shows configuration for an outbound voice class called number:

```
voice class called number outbound 30
 index 1 5550100
 index 2 5550101
 index 3 5550102
 index 4 5550103
```

The following example shows configuration for a voice class called number pool:

```
voice class called number pool 1
 index 1 5550100 - 5550199
```

Related Commands

Command	Description
show voice class called-number	Displays a specific voice class called number.
voice-class called-number (dial-peer)	Assigns a previously defined voice class called number to an inbound or outbound POTS dial peer.

voice-class called-number (dial peer)

To assign a previously defined voice class called number to an inbound or outbound POTS dial peer, use the **voice-class called-number** command in dial peer configuration mode. To remove a voice class called number from the dial peer, use the **no** form of this command.

voice-class called-number [inbound | outbound] *tag*

no voice-class called-number

Syntax Description

inbound	Assigns an inbound voice class called number to the dial peer.
outbound	Assigns an outbound voice class called number to the dial peer.
tag	Digits that identify a specific voice class called number.

Command Default

No voice class called number is configured on the dial peer.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to assign a previously defined voice class called number to a dial peer for a static H.320 secondary call dial plan. Use the **inbound** keyword for inbound POTS dial peers, and the **outbound** keyword for outbound POTS dial peers.

Note The **voice class called number** command in global configuration mode is entered without hyphens. The **voice-class called-number** command in dial-peer configuration mode is entered with hyphens.

Examples

The following example shows configuration for an outbound voice class called number outbound on POTS dial peer 22:

```
dial-peer voice 22 pots
 voice-class called-number inbound 300
```

Related Commands

Command	Description
voice class called number	Defines a voice class called number or range of numbers for H.320 calls.
voice-class called-number-pool	Defines a pool of dynamic voice class called numbers for a voice port.

voice-class called-number-pool

To assign a previously defined voice class called number pool to a voice port, use the **voice-class called-number-pool** command in voice port configuration mode. To remove a voice class called number pool from the voice port, use the **no** form of this command.

voice-class called-number-pool *tag*

no voice-class called-number-pool

Syntax Description

tag	Digits that identify a specific voice class called number pool.

Command Default

No voice class called number pool is assigned to the voice port.

Command Modes

Voice class configuration

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

Use this command to assign a voice class called number pool to a voice port for a dynamic H.320 secondary call dial plan.

Examples

The following example shows configuration for voice class called number pool 100 on voice port 1/0/0:

```
voice-port 1/0/0
 voice-class called-number-pool 100
```

Related Commands

Command	Description
voice class called number	Defines a voice class called number or range of numbers for H.320 calls.
voice-class called-number (dial-peer)	Defines a called number or range of called numbers for a POTS dial peer.

Feature Information for Integrating Data, Voice, and Video for ISDN Interfaces

Table 10 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 10 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 10 Feature Information for Integrating Data, Voice, and Video Services for ISDN Interfaces

Feature Name	Releases	Feature Information
Cisco IOS H.320 Video Gateway	12.4(11)T	The Cisco IOS H.320 Video Gateway provides the capability to send H.320 encapsulated Audio/Video calls over TDM voice interfaces. The following sections provide information about this feature: • "Information About Integrated Data, Voice, and Video Services for ISDN Interfaces" section on page 3 • "How to Configure Integrated Data, Voice, and Video Services for ISDN Interfaces" section on page 6