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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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 Recommendation H.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.320 gateway > 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
Integrated Data, Voice, and Video Services for ISDN Interfaces

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

DETAILED STEPS

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

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 shows 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> dial-peer voice tag pots</td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.</td>
</tr>
<tr>
<td>Example: Router(config)# dial-peer voice 12 pots</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> incoming called-number string</td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td>Example: Router(config-dial-peer)# incoming called-number 408</td>
<td></td>
</tr>
</tbody>
</table>
Integrated Data, Voice, and Video Services for ISDN Interfaces

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

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]

Example:

Router(config-dial-peer)# direct-inward-dial
Enables the direct inward dialing (DID) call treatment for an incoming called number.

Step 6

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.
## Detailed Steps

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
- Enter your password if prompted. |
| **Step 2** configure terminal | Enters global configuration mode. |
| **Step 3** voice class codec tag | 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] | 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] | 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.

```plaintext
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>Enters dial-peer configuration mode for a specific dial peer.</strong></td>
</tr>
<tr>
<td><code>dial-peer voice tag voip</code></td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td><code>Router(config)# dial-peer voice 12 voip</code></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>** Specifies a digit string that can be matched by an incoming call to associate the call with an incoming dial peer.**</td>
</tr>
<tr>
<td><code>incoming called number string</code></td>
<td></td>
</tr>
<tr>
<td>or</td>
<td>`destination pattern [+</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td><code>Router(config-dial-peer)# incoming called number 408</code></td>
</tr>
<tr>
<td>or</td>
<td><code>Router(config-dial-peer)# destination-pattern 4085550100</code></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>(Optional) Assigns a previously defined voice class codec to this VoIP dial peer.</strong></td>
</tr>
<tr>
<td><code>voice class codec tag</code></td>
<td></td>
</tr>
<tr>
<td>or</td>
<td>`video codec [h261</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td><code>Router(config-dial-peer)# voice class codec 10</code></td>
</tr>
<tr>
<td>or</td>
<td><code>Router(config-dial-peer)# video codec h261</code></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>(Optional. Only available if H.263+ or H.264 video codecs are configured.) Defines the RTP payload type for this dial peer.</strong></td>
</tr>
<tr>
<td>`rtp payload-type [cisco-codec-video-h263+</td>
<td>cisco-codec-video-h264] [number]`</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td><code>Router(config-dial-peer)# rtp payload-type cisco-codec-video-h264</code></td>
</tr>
</tbody>
</table>
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 non-H.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 (non-H.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, page 13 (optional)
- Configuring Static H.320 Secondary Call Dial Plans, page 20 (optional)
- Configuring a Combined Static and Dynamic H.320 Secondary Call Dial Plan, page 27 (optional)

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 (non-H.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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice class called number pool tag</td>
<td>Defines a dynamic voice class called number pool, which can be allocated by the application to match the incoming H.320 secondary calls.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice class called number pool 100</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• tag—Identifier for the voice class called number pool. The range is 1 to 10000.</td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td>Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> dial-peer voice tag pots</td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# dial-peer voice 12 pots</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> destination-pattern [+ string [T]]</td>
<td>Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# destination-pattern 4085550100</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> information-type [fax</td>
<td>video</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# information-type video</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> bandwidth maximum value [minimum value]</td>
<td>Specifies the maximum and minimum bandwidth for an H.320 call.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# bandwidth maximum 256 minimum 64</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> no digit-strip</td>
<td>(Optional) Disables digit stripping on a POTS dial-peer call leg.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# no digit-strip</td>
<td></td>
</tr>
</tbody>
</table>

### Note
To return to the default value, use the `default information-type` command in dial-peer configuration mode.
Integrated Data, Voice, and Video Services for ISDN Interfaces

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

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
</tbody>
</table>
Integrated Data, Voice, and Video Services for ISDN Interfaces

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

### 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`

---

**Command or Action** | **Purpose**
--- | ---
**Step 3** | **voice-port slot/port:D-channel-number**
Example: | Enters voice-port configuration mode.
Router(config)# voice-port 1/0:23
- **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: | Assigns a previously defined voice class called number pool to the voice port.
Router(config-voiceport)# voice-class called-number-pool 100
- **tag**—Identifier for the voice class called number pool.
## DETAILED STEPS

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

<table>
<thead>
<tr>
<th>Step 7</th>
<th>port slot/port:D-channel-number</th>
</tr>
</thead>
</table>

**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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
  • Enter your password if prompted. |
| Example:  
  Router> enable | |
| **Step 2** configure terminal | Enters global configuration mode. |
| Example:  
  Router# configure terminal | |
| **Step 3** voice class called number inbound *tag* | 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. |
| Example:  
  Router(config)# voice class called number inbound 200 | |
| **Step 4** index *number* *called-number* | 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:  
  Router(config-class)# index 1 6505550111  
  index 2 6505550112  
  index 3 6505550113  
  index 4 6505550114 | |

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

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

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

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> dial-peer voice tag pots</td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# dial-peer voice 7001 pots</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• <strong>tag</strong>—Identifier for the dial peer. The range is 1 to 2147483647.</td>
</tr>
<tr>
<td></td>
<td>• <strong>pots</strong>—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.</td>
</tr>
</tbody>
</table>
### How to Configure Static and Dynamic H.320 Secondary Call Dial Plans

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 4</strong> <em>incoming called-number</em> <em>string</em></td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# <em>incoming called-number</em> 408</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> <em>direct-inward-dial</em></td>
<td>Enables the direct inward dialing (DID) call treatment for an incoming called number.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# <em>direct-inward-dial</em></td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> <em>information-type</em> {fax</td>
<td>video</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# <em>information-type</em> video</td>
<td></td>
</tr>
<tr>
<td><strong>Note</strong> To return to the default value, use the <em>default information-type</em> command in dial-peer configuration mode.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> <em>voice class called number</em> <em>inbound</em> <em>tag</em></td>
<td>Defines one or more static voice class called numbers for H.320 calls.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# <em>voice class called number</em> inbound 200</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> <em>bandwidth maximum</em> <em>value</em> {minimum <em>value</em>}</td>
<td>Specifies the maximum and minimum bandwidth for an H.320 call.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# <em>bandwidth maximum</em> 256 minimum 64</td>
<td></td>
</tr>
</tbody>
</table>

### 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 dial-peer voice tag pots</td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td>Example: Router(config)# dial-peer voice 12 pots</td>
<td></td>
</tr>
<tr>
<td>Step 4 destination-pattern ([+) string ([T])</td>
<td>Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.</td>
</tr>
<tr>
<td>Example: Router(config-dial-peer)# destination-pattern 4085550100</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

**Step 5**  
information-type [fax | video | voice]

**Example:**
Router(config-dial-peer)# information-type video

**Purpose:**  
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

**Purpose:**  
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

**Purpose:**  
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

**Purpose:**  
(Optional) Disables digit stripping on a POTS dial-peer call leg.

**Step 9**  
trunkgroup name preference-num

**Example:**
Router(config-dial-peer)# trunkgroup isdn1

**Purpose:**  
(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 isdn1
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, page 27 (required)
- Configuring Combined Static and Dynamic Dial Plan Inbound POTS Dial Peer for Originating Gateway, page 31 (required)
- Configuring Dynamic Outbound POTS Dial Peers for Terminating Gateway, page 32 (required)
- Configuring Static Outbound POTS Dial Peers for Terminating Gateway, page 35 (required)

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

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

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice class called number outbound tag</td>
<td>Defines one or more static voice class called numbers for H.320 calls.</td>
</tr>
<tr>
<td>Example: Router(config)# voice class called number outbound 300</td>
<td>- outbound—Outbound voice class called number.</td>
</tr>
<tr>
<td></td>
<td>- tag—Identifier for the outbound voice class called number.</td>
</tr>
<tr>
<td><strong>Step 4</strong> index number called-number</td>
<td>Defines an index for a voice class called number. You can define multiple indexes.</td>
</tr>
<tr>
<td>Example: Router(config-class)# index 1 4085550100</td>
<td>- number—Identifier for the index. The range is 1 to 2147483647.</td>
</tr>
<tr>
<td></td>
<td>- called-number—Specifies a called number, in E.164 format.</td>
</tr>
</tbody>
</table>

Example

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

```plaintext
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**  
   - Example: `Router> enable`  
   - Enables privileged EXEC mode.  
   - Enter your password if prompted.

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

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.

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.

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1 enable</strong></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2 configure terminal</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3 dial-peer voice tag pots</strong></td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td>Example: Router(config)# dial-peer voice 12 pots</td>
<td>• <code>tag</code>—Identifier for the dial peer. The range is 1 to 2147483647.</td>
</tr>
<tr>
<td></td>
<td>• <code>pots</code>—Indicates that this is a POTS peer that uses VoIP encapsulation on the IP backbone.</td>
</tr>
<tr>
<td><strong>Step 4 incoming called-number string</strong></td>
<td>Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.</td>
</tr>
<tr>
<td>Example: Router(config-dial-peer)# incoming called-number 408</td>
<td>• <code>string</code>—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.</td>
</tr>
</tbody>
</table>
### How to Configure Static and Dynamic H.320 Secondary Call Dial Plans

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 5</strong> direct-inward-dial</td>
<td>Enables the direct inward dialing (DID) call treatment for an incoming called number.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# direct-inward-dial</td>
<td></td>
</tr>
</tbody>
</table>

**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 non-H.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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Step 3 dial-peer voice tag pots</td>
<td>Defines a specific dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Step 4 destination-pattern [+] string [T]</td>
<td>Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
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<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 5</td>
<td>information-type [fax</td>
<td>video</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# information-type video</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Note</td>
<td>To return to the default value, use the default information-type command in dial-peer configuration mode.</td>
</tr>
<tr>
<td>Step 6</td>
<td>bandwidth maximum value [minimum value]</td>
<td>Specifies the maximum and minimum bandwidth for an H.320 call.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# bandwidth maximum 512</td>
<td></td>
</tr>
<tr>
<td>Step 7</td>
<td>no digit-strip</td>
<td>(Optional) Disables digit stripping on a POTS dial-peer call leg.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# no digit-strip</td>
<td></td>
</tr>
<tr>
<td>Step 8</td>
<td>trunkgroup name preference-num</td>
<td>(Optional) Assigns a dial peer to a trunk group for trunk group label routing.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# trunkgroup isdntg</td>
<td></td>
</tr>
</tbody>
</table>

### 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

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

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| 5    | `information-type [fax | video | voice]` | 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. |
| 6    | `voice-class called-number [inbound | outbound] tag` | 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. |
| 7    | `bandwidth maximum value [minimum value]` | 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`. |
| 8    | `no digit-strip` | (Optional) Disables digit stripping on a POTS dial-peer call leg. |
| 9    | `trunkgroup name preference-num` | (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
```
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.

```plaintext
no digit-strip
trunkgroup isdntg

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
!
o network-clock-participate slot 1
ip subnet-zero
ip cef
!
o ip dhcp use vrf connected
!
o 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

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

Standards

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

MIBs

<table>
<thead>
<tr>
<th>MIB</th>
<th>MIBs Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>CISCO-VOICE-COMMON-DIAL-CONTROL-MIB</td>
<td>To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: <a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></td>
</tr>
<tr>
<td>CISCO-VOICE-DIAL-CONTROL-MIB</td>
<td></td>
</tr>
<tr>
<td>CISCO-H320-DIAL-CONTROL-MIB</td>
<td></td>
</tr>
</tbody>
</table>
### RFCs

<table>
<thead>
<tr>
<th>RFC</th>
<th>Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>RFC 2190</td>
<td>RTP Payload Format for H.263 Video Streams</td>
</tr>
<tr>
<td>RFC 2198</td>
<td>RTP Payload for Redundant Audio Data</td>
</tr>
</tbody>
</table>

### Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</td>
<td><a href="http://www.cisco.com/techsupport">http://www.cisco.com/techsupport</a></td>
</tr>
<tr>
<td>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.</td>
<td></td>
</tr>
<tr>
<td>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</td>
<td></td>
</tr>
</tbody>
</table>

### 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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>maximum value</td>
<td>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.</td>
</tr>
<tr>
<td>minimum value</td>
<td>(Optional) Sets the minimum bandwidth. Acceptable values are 64 kbps or minimum value = maximum value.</td>
</tr>
</tbody>
</table>

**Command Default**

No maximum bandwidth is set.

**Command Modes**

Dial-peer configuration

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bandwidth</td>
<td>Specifies the maximum aggregate bandwidth for H.323 traffic and verifies the available bandwidth of the destination gatekeeper.</td>
</tr>
</tbody>
</table>
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.

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

no debug voice h221
```

**Syntax Description**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>all</code></td>
<td>(Optional) Enables all H.221 debugging, except the <code>raw</code> option.</td>
</tr>
<tr>
<td><code>default</code></td>
<td>(Optional) Activates function, inout, error call, and software debugging.</td>
</tr>
<tr>
<td><code>error</code></td>
<td>(Optional) Enables H.221 call error and software error debugging.</td>
</tr>
<tr>
<td><code>error [call]</code></td>
<td>(Optional) Enables H.221 major call processing error debugging.</td>
</tr>
</tbody>
</table>
| `error [call
informational]` | (Optional) Enables H.221 major and informational call processing error debugging. |
| `error [software]`  | (Optional) Enables H.221 major software error debugging.                    |
| `error [software
informational]` | (Optional) Enables H.221 major and informational software error debugging |
| `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**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

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

**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 1 0 0: AUDIO CAPS=g722 48k
BAS=85:1 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 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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>debug voip ccapi</strong></td>
<td>Enables debugging for the call control application programming interface (CCAPI) contents.</td>
</tr>
<tr>
<td><strong>debug voip rtp</strong></td>
<td>Enables debugging for Real-Time Transport Protocol (RTP) named event packets.</td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>all</td>
<td>(Optional) Enables all H.221 debugging, except the raw option.</td>
</tr>
<tr>
<td>default</td>
<td>(Optional) Activates function, inout, error call, and software debugging.</td>
</tr>
<tr>
<td>error</td>
<td>(Optional) Enables H.221 call error and software error debugging.</td>
</tr>
<tr>
<td>error [call]</td>
<td>(Optional) Enables H.221 major call processing error debugs related to the H.221 subsystem.</td>
</tr>
<tr>
<td>error [call informational]</td>
<td>(Optional) Enables H.221 major and informational call processing error debugs related to the H.221 subsystem.</td>
</tr>
<tr>
<td>error [software]</td>
<td>(Optional) Enables H.221 major software error debugs related to the H.221 subsystem.</td>
</tr>
<tr>
<td>error [software informational]</td>
<td>(Optional) Enables H.221 major and informational software error debugs related to the H.221 subsystem.</td>
</tr>
<tr>
<td>function</td>
<td>(Optional) Enables procedure tracing.</td>
</tr>
<tr>
<td>individual</td>
<td>(Optional) Activates individual H.221 debugging.</td>
</tr>
<tr>
<td>inout</td>
<td>(Optional) Enables subsystem inout debugging.</td>
</tr>
<tr>
<td>raw</td>
<td>(Optional) Displays raw BAS messages.</td>
</tr>
<tr>
<td>raw [decode]</td>
<td>(Optional) Decodes raw BAS data.</td>
</tr>
</tbody>
</table>

### Command Modes

Privileged EXEC

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### 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

<table>
<thead>
<tr>
<th>Index Number</th>
<th>Debug Category</th>
</tr>
</thead>
<tbody>
<tr>
<td>1, 2, 30, 31, 32</td>
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</tr>
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<td>5, 6, 14, 15, 16, 22</td>
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<tr>
<td>34, 35, 36, 37, 38, 39, 40, 41</td>
<td>BAS squelch</td>
</tr>
</tbody>
</table>

### 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 1 0 0 0: AUDIO CAPS=g711 u-law
BAS=84:1 0 0 0 1 0 0 0: AUDIO CAPS=g722 48k
BAS=85:1 0 0 0 1 0 1 0: AUDIO CAPS=g723
BAS=F9:1 1 1 1 1 0 0 1: H.242 MBE start indication
BAS=02:0 0 0 0 0 1 0 0: H.242 MBE length=2
BAS=0A:0 0 0 1 0 0 0 1: H.242 MBE type=H.263 caps
BAS=8A:1 - - - - - - - -: Always 1
BAS=8A:- 0 0 0 0 - - -: H.263 MPI=1
BAS=8A:- - - - - 0 1 -: H.263 FORMAT=h.263_cif
BAS=8A:- - - - - - 0: No additional options
```

### Related Commands

<table>
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<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>debug voip ccapi</code></td>
<td>Enables debugging for the call control application programming interface (CCAPI) contents.</td>
</tr>
<tr>
<td><code>debug voip rtp</code></td>
<td>Enables debugging for Real-Time Transport Protocol (RTP) named event packets.</td>
</tr>
</tbody>
</table>
**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
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>number</th>
<th>Digits that identify this index. Range is 1 to 2147483647.</th>
</tr>
</thead>
<tbody>
<tr>
<td>called-number</td>
<td></td>
<td>Specifies a called number, or a range of called numbers, in E.164 format.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command Default</th>
<th>No index is configured.</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Command Modes</th>
<th>Voice class configuration</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Command History</th>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>voice class called</td>
<td>One or more called numbers configured for a voice class.</td>
</tr>
<tr>
<td></td>
<td>number</td>
<td></td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fax</td>
<td>The information type is set to store-and-forward fax.</td>
<td></td>
</tr>
<tr>
<td>voice</td>
<td>The information type is set to voice. This is the default.</td>
<td></td>
</tr>
<tr>
<td>video</td>
<td>The information type is set to video.</td>
<td></td>
</tr>
</tbody>
</table>

### Command Default

Voice

### Command Modes

Dial peer configuration

### Command History

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

### 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
```
<table>
<thead>
<tr>
<th>Related Commands</th>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><strong>isdn integrate caltype all</strong></td>
<td>Enables integrated mode (for data, voice, and video) on ISDN BRI or PRI interfaces.</td>
</tr>
</tbody>
</table>
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\}]
```

**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
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
```
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**

<table>
<thead>
<tr>
<th>Argument</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>brief</td>
<td>(Optional) Displays a truncated version of active video call information.</td>
</tr>
<tr>
<td>compact</td>
<td>(Optional) Displays a compact version of active video call information.</td>
</tr>
<tr>
<td>echo-canceller call-id</td>
<td>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 <code>show call active video brief</code> command. Range is 0 to FFFFFFFF.</td>
</tr>
<tr>
<td>id identifier</td>
<td>(Optional) Displays only the video call with the specified identifier. Range is a hexadecimal value from 1 to FFFF.</td>
</tr>
</tbody>
</table>

**Command Default**

No default behavior or values.

**Command Modes**

User EXEC
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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
```
Integrated Data, Voice, and Video Services for ISDN Interfaces

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
show call active video

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
show call active video

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
show call active video

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

TEL:
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
TransactedCallingNumber=555556
TransactedCallingOctet=0x0
TransactedCalledNumber=7001
TransactedCalledOctet=0x80
TransactedRedirectCalledNumber=
TransactedRedirectCalledOctet=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

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>VideoCap_Codec</td>
<td>Codec for the active video call.</td>
</tr>
<tr>
<td>VideoCap_Format</td>
<td>Video format for the active video call.</td>
</tr>
<tr>
<td>VideoEarlyPackets</td>
<td>Number of early packets for a video call.</td>
</tr>
<tr>
<td>VideoLatePackets</td>
<td>Number of late packets in a video call.</td>
</tr>
<tr>
<td>VideoLostPackets</td>
<td>Number of lost packets in a video call.</td>
</tr>
<tr>
<td>VideoNumberOfChannels</td>
<td>Number of channels used for a video call.</td>
</tr>
<tr>
<td>VideoUsedBandwidth</td>
<td>Bandwidth, in kbps, used for a video call.</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show call history video</td>
<td>Displays call history information for SCCP video calls.</td>
</tr>
</tbody>
</table>
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**

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

**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.
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',
```

---

**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',
```

---

**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, Icmpf = 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,
show dial-peer voice

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>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accepted Calls</td>
<td>Number of calls accepted from this peer since system startup.</td>
</tr>
<tr>
<td>acc-qos</td>
<td>Lowest acceptable quality of service configured for calls for this peer.</td>
</tr>
<tr>
<td>Admin state</td>
<td>Administrative state of this peer.</td>
</tr>
<tr>
<td>answer-address</td>
<td>Answer address configured for this dial peer.</td>
</tr>
<tr>
<td>bandwidth maximum/minimum</td>
<td>The maximum and minimum bandwidth.</td>
</tr>
<tr>
<td>Charged Units</td>
<td>Total number of charging units that have applied to this peer since system startup, in hundredths of a second.</td>
</tr>
<tr>
<td>CLID Restriction</td>
<td>Indicates if CLID restriction is enabled.</td>
</tr>
<tr>
<td>CLID Network Number</td>
<td>Displays the network number sent as CLID, if configured.</td>
</tr>
<tr>
<td>CLID Second Number sent</td>
<td>Displays whether a second calling number is stripped from the call setup.</td>
</tr>
<tr>
<td>CLID Override RDNIS</td>
<td>Indicates whether the CLID is overridden by the redirecting number.</td>
</tr>
<tr>
<td>codec</td>
<td>Default voice codec rate of speech.</td>
</tr>
<tr>
<td>Connect Time</td>
<td>Accumulated connect time to the peer since system startup for both incoming and outgoing calls, in hundredths of a second.</td>
</tr>
<tr>
<td>connections/maximum</td>
<td>Indicates maximum call connections per peer</td>
</tr>
<tr>
<td>Destination</td>
<td>Indicates the voice class which is used to match destination url</td>
</tr>
<tr>
<td>destination-pattern</td>
<td>Destination pattern (telephone number) for this peer.</td>
</tr>
<tr>
<td>digit_strip</td>
<td>Indicates if digit stripping is enabled.</td>
</tr>
<tr>
<td>direct-inward-dial</td>
<td>Indicates if direct-inward-dial is enabled.</td>
</tr>
<tr>
<td>disconnect-cause</td>
<td>Indicates the disconnect cause code to be used when an incoming call is blocked</td>
</tr>
<tr>
<td>dnis-map</td>
<td>Name of the dialed-number identification service (DNIS) map.</td>
</tr>
<tr>
<td>DTMF Relay</td>
<td>Indicates if dual-tone multifrequency (DTMF) relay is enabled.</td>
</tr>
<tr>
<td>Expect factor</td>
<td>User-requested expectation factor of voice quality for calls through this peer.</td>
</tr>
<tr>
<td>Failed Calls</td>
<td>Number of failed call attempts to this peer since system startup.</td>
</tr>
<tr>
<td>fax rate</td>
<td>Fax transmission rate configured for this peer.</td>
</tr>
</tbody>
</table>
### Table 4  show dial-peer voice Field Descriptions (continued)

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

```
Router# show dial-peer voice summary
```

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Redundancy</td>
<td>Packet redundancy (RFC 2198) for modem traffic.</td>
</tr>
<tr>
<td>Refused Calls</td>
<td>Number of calls from this peer refused since system startup.</td>
</tr>
<tr>
<td>register E.164 number with H.323 GK and/or SIP Registrar</td>
<td>Indicates &quot;register e.164&quot; option of this peer</td>
</tr>
<tr>
<td>req-qos</td>
<td>Configured requested quality of service for calls for this dial peer.</td>
</tr>
<tr>
<td>session-target</td>
<td>Session target of this peer.</td>
</tr>
<tr>
<td>sess,proto</td>
<td>Session protocol to be used for Internet calls between local and remote routers through the IP backbone.</td>
</tr>
<tr>
<td>source carrier-id</td>
<td>Indicates source carrier-id of this peer which will be used to match the source carrier-id of an incoming call</td>
</tr>
<tr>
<td>source trunk-group label</td>
<td>Indicates source trunk-group-label of this peer which can be used to match the source trunk-group-label of an incoming call</td>
</tr>
<tr>
<td>Successful Calls</td>
<td>Number of completed calls to this peer.</td>
</tr>
<tr>
<td>supported-language</td>
<td>Indicates list of supported languages of this peer</td>
</tr>
<tr>
<td>tag</td>
<td>Unique dial peer ID number.</td>
</tr>
<tr>
<td>target carrier-id</td>
<td>Indicates target carrier-id of this peer which will be used to match the target carrier-id for an outgoing call</td>
</tr>
<tr>
<td>target trunk-group label</td>
<td>Indicates target trunk-group-label of this peer which can be used to match the target trunk-group-label of an outgoing call</td>
</tr>
<tr>
<td>Time elapsed since last clearing of voice call statistics</td>
<td>Elapsed time between the current time and the time when the &quot;clear dial-peer voice&quot; command was executed</td>
</tr>
<tr>
<td>Translation profile (Incoming)</td>
<td>Indicate translation profile for incoming calls</td>
</tr>
<tr>
<td>Translation profile (Outgoing)</td>
<td>Indicate translation profile for outgoing calls</td>
</tr>
<tr>
<td>translation-profile</td>
<td>Indicate number translation profile of this peer</td>
</tr>
<tr>
<td>type</td>
<td>Indicate peer encapsulation type such as pots, voip, vofr, voatm or mmoip</td>
</tr>
<tr>
<td>VAD</td>
<td>Whether voice activation detection (VAD) is enabled for this dial peer.</td>
</tr>
<tr>
<td>voice class called-number inbound/outbound</td>
<td>Indicates voice-class called-number inbound or outbound setup of this peer</td>
</tr>
<tr>
<td>voice-port</td>
<td>Indicates the voice interface setting of this POTS peer</td>
</tr>
</tbody>
</table>
Table 5 describes significant fields shown in this output.

**Table 5  ** *show dial-peer voice summary Field Descriptions*

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

**Related Commands**

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

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>inbound</td>
<td>Displays the specified inbound voice class called-number.</td>
</tr>
<tr>
<td>outbound</td>
<td>Displays the specified outbound voice class called-number.</td>
</tr>
<tr>
<td>tag</td>
<td>Digits that identify this voice class called-number.</td>
</tr>
</tbody>
</table>

### Command Modes

<table>
<thead>
<tr>
<th>Command Modes</th>
<th>Privileged EXEC</th>
</tr>
</thead>
</table>

### Command History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

### 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

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Number Inbound/Outbound</td>
<td>The tag for the specified inbound or outbound voice class called-number.</td>
</tr>
<tr>
<td>index number</td>
<td>The number or range of numbers for this voice class called-number.</td>
</tr>
</tbody>
</table>

### Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice class called-number-pool</td>
<td>Displays voice class called number pool configuration information.</td>
</tr>
</tbody>
</table>

---

Integrating Data, Voice, and Video Services for ISDN Interfaces
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**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Digits that identify this voice class called-number-pool. Range is 1 to 10000.</td>
</tr>
<tr>
<td><code>detail</code></td>
<td>Displays idle called number and allocated called number information.</td>
</tr>
</tbody>
</table>

**Command Modes**

Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Number Pool</td>
<td>Tag that identifies the called number pool.</td>
</tr>
<tr>
<td>index</td>
<td>Number or range of numbers for this called number pool.</td>
</tr>
</tbody>
</table>
| 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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice class called-number</td>
<td>Displays a specific voice class called-number.</td>
</tr>
</tbody>
</table>
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

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

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*

<table>
<thead>
<tr>
<th>DSP</th>
<th>DSP</th>
<th>DSPWARE CURR BOOT</th>
<th>RST</th>
<th>AI</th>
<th>VOICEPORT</th>
<th>TS</th>
<th>ABRT</th>
<th>PACK</th>
<th>COUNT</th>
</tr>
</thead>
<tbody>
<tr>
<td>TYPE</td>
<td>NUM</td>
<td>CH CODEC</td>
<td>STATE</td>
<td>STATE</td>
<td>RST</td>
<td>AI</td>
<td>VOICEPORT</td>
<td>TS</td>
<td>ABRT</td>
</tr>
<tr>
<td>=====</td>
<td>===</td>
<td>== ======</td>
<td>======</td>
<td>======</td>
<td>===</td>
<td>===</td>
<td>=========</td>
<td>===</td>
<td>======</td>
</tr>
<tr>
<td>C5510</td>
<td>001</td>
<td>01 modem-re</td>
<td>4.5.909</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/1/0</td>
<td>05</td>
</tr>
</tbody>
</table>

*DSP SIGNALING CHANNELS*

<table>
<thead>
<tr>
<th>DSP</th>
<th>DSP</th>
<th>DSPWARE CURR BOOT</th>
<th>PAK</th>
<th>TX/RX</th>
</tr>
</thead>
<tbody>
<tr>
<td>TYPE</td>
<td>NUM</td>
<td>CH CODEC</td>
<td>PK</td>
<td>TX/RX</td>
</tr>
<tr>
<td>=====</td>
<td>===</td>
<td>== ======</td>
<td>----</td>
<td>-------</td>
</tr>
<tr>
<td>C5510</td>
<td>001</td>
<td>05 (flex)</td>
<td>4.5.909</td>
<td>alloc</td>
</tr>
<tr>
<td>C5510</td>
<td>001</td>
<td>06 (flex)</td>
<td>4.5.909</td>
<td>alloc</td>
</tr>
<tr>
<td>C5510</td>
<td>001</td>
<td>07 (flex)</td>
<td>4.5.909</td>
<td>alloc</td>
</tr>
<tr>
<td>C5510</td>
<td>001</td>
<td>08 (flex)</td>
<td>4.5.909</td>
<td>alloc</td>
</tr>
</tbody>
</table>

----------------------------------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`

<table>
<thead>
<tr>
<th>DSP</th>
<th>DSP</th>
<th>DSPWARE</th>
<th>CURR</th>
<th>BOOT</th>
<th>RST</th>
<th>AI</th>
<th>VOICEPORT</th>
<th>TS</th>
<th>ABRPT</th>
<th>PACK</th>
<th>TX/RX</th>
</tr>
</thead>
<tbody>
<tr>
<td>C549 1 01 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>1 0</td>
<td>9357/9775</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C549 1 02 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>2 0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C549 2 01 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>3 0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C549 2 02 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>4 0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C549 3 01 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>5 0</td>
<td>0/13</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C549 3 02 (medium)</td>
<td>4.4.3</td>
<td>IDLE</td>
<td>idle</td>
<td>0 0</td>
<td>1/0:0</td>
<td>6 0</td>
<td>0/13</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

Router# `show voice dsp`

<table>
<thead>
<tr>
<th>DSP</th>
<th>DSP</th>
<th>DSPWARE</th>
<th>CURR</th>
<th>BOOT</th>
<th>RST</th>
<th>AI</th>
<th>VOICEPORT</th>
<th>TS</th>
<th>ABRPT</th>
<th>PACK</th>
<th>TX/RX</th>
</tr>
</thead>
<tbody>
<tr>
<td>C5510 001 05 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 06 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 07 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 08 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 09 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 10 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>C5510 001 11 None</td>
<td>9.0.105</td>
<td>idle</td>
<td>idle</td>
<td>0 0</td>
<td>0</td>
<td>0/0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

*DSP VOICE CHANNELS*
Table 8 describes significant fields shown in the output.

**Table 8 show voice dsp Field Descriptions**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSP</td>
<td>Number of the DSP.</td>
</tr>
<tr>
<td>channel</td>
<td>Number of the channel and its status.</td>
</tr>
<tr>
<td>DSP TYPE</td>
<td>TI-549 (C549) DSP.</td>
</tr>
<tr>
<td>DSP NUM</td>
<td>Number of the DSP.</td>
</tr>
<tr>
<td>CH</td>
<td>Channel number.</td>
</tr>
<tr>
<td>CODEC</td>
<td>Complexity setting.</td>
</tr>
<tr>
<td>DSPWARE VERSION</td>
<td>Version of DSPware.</td>
</tr>
<tr>
<td>CURR STATE</td>
<td>Current status of the channel, either IDLE or BUSY.</td>
</tr>
<tr>
<td>BOOT STATE</td>
<td>DSP readiness, either idle or in service.</td>
</tr>
<tr>
<td>RST</td>
<td>Number of times the DSP has been reset or restarted.</td>
</tr>
<tr>
<td>AI</td>
<td>Alarm indication count on the channel.</td>
</tr>
<tr>
<td>VOICEPORT</td>
<td>Voice card number and slot.</td>
</tr>
<tr>
<td>TS</td>
<td>Time slot.</td>
</tr>
</tbody>
</table>
Table 8  show voice dsp Field Descriptions (continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PAK ABORT</td>
<td>Number of dropped packets.</td>
</tr>
<tr>
<td>TX/RX PACKCOUNT</td>
<td>Number of transmitted and received packets</td>
</tr>
</tbody>
</table>

Related Commands

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>clear counters</td>
<td>Clears all the current interface counters from the interface.</td>
</tr>
<tr>
<td>show dial-peer voice</td>
<td>Displays configuration information for dial peers.</td>
</tr>
<tr>
<td>show voice call</td>
<td>Displays the call status for all voice ports.</td>
</tr>
<tr>
<td>show voice port</td>
<td>Displays configuration information about a specific voice port.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>slot</code></td>
<td>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.</td>
</tr>
<tr>
<td><code>port</code></td>
<td>Voice port. Valid entries are 0 and 1.</td>
</tr>
</tbody>
</table>

**Cisco 2600 and Cisco 3600 Series Router with Analog Voice Ports**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>slot/subunit/port</code></td>
<td>(Optional) Output displays information for the analog voice port that you specify using the <code>slot/subunit/port</code> designation.</td>
</tr>
<tr>
<td></td>
<td>• <code>slot</code>—Router slot in which a voice network module (VNM) is installed. Valid entries are router slot numbers for the specific platform.</td>
</tr>
<tr>
<td></td>
<td>• <code>subunit</code>—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.)</td>
</tr>
<tr>
<td></td>
<td>• <code>port</code>—Analog voice port number. Valid entries are 0 and 1.</td>
</tr>
<tr>
<td><code>summary</code></td>
<td>(Optional) Output displays a summary of all voice ports.</td>
</tr>
</tbody>
</table>
show voice port

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.

```
ds0-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**

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

receive 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
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 10 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:

<table>
<thead>
<tr>
<th>DS0 channel specific status info:</th>
</tr>
</thead>
<tbody>
<tr>
<td>PORT</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
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<td>2/0:23</td>
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</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
<tr>
<td>2/0:23</td>
</tr>
</tbody>
</table>

Table 9 describes significant fields shown in each these output.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Administrative State</td>
<td>Administrative state of the voice port.</td>
</tr>
<tr>
<td>Alias</td>
<td>User-supplied alias for the voice port.</td>
</tr>
<tr>
<td>Analog interface A-D gain offset</td>
<td>Gain offset for analog-to-digital conversion.</td>
</tr>
<tr>
<td>Analog interface D-A gain offset</td>
<td>Gain offset for digital-to-analog conversion.</td>
</tr>
<tr>
<td>Clear Wait Duration Timing</td>
<td>Time of inactive seizure signal to declare call cleared.</td>
</tr>
<tr>
<td>Coder Type</td>
<td>Voice compression mode used.</td>
</tr>
<tr>
<td>Companding Type</td>
<td>Companding standard used to convert between analog and digital signals in PCM systems.</td>
</tr>
<tr>
<td>Connection Mode</td>
<td>Connection mode of the interface.</td>
</tr>
<tr>
<td>Connection Number</td>
<td>Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.</td>
</tr>
<tr>
<td>Currently Processing</td>
<td>Type of call currently being processed: none, voice, or fax.</td>
</tr>
<tr>
<td>Delay Duration Timing</td>
<td>Maximum delay signal duration for delay dial signaling.</td>
</tr>
<tr>
<td>Delay Start Timing</td>
<td>Timing of generation of delayed start signal from detection of incoming seizure.</td>
</tr>
<tr>
<td>Description</td>
<td>Description of the voice port.</td>
</tr>
</tbody>
</table>
Table 9  show voice port Field Descriptions (continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial Type</td>
<td>Out-dialing type of the voice port.</td>
<td></td>
</tr>
<tr>
<td>Digit Duration Timing</td>
<td>DTMF digit duration, in milliseconds.</td>
<td></td>
</tr>
<tr>
<td>E&amp;M Type</td>
<td>Type of E&amp;M interface.</td>
<td></td>
</tr>
<tr>
<td>Echo Cancel Coverage</td>
<td>Echo cancel coverage for this port.</td>
<td></td>
</tr>
<tr>
<td>Echo Cancellation</td>
<td>Whether echo cancellation is enabled for this port.</td>
<td></td>
</tr>
<tr>
<td>Hook Flash Duration Timing</td>
<td>Maximum length of hookflash signal.</td>
<td></td>
</tr>
<tr>
<td>Hook Status</td>
<td>Hook status of the FXO/FXS interface.</td>
<td></td>
</tr>
<tr>
<td>Impedance</td>
<td>Configured terminating impedance for the E&amp;M interface.</td>
<td></td>
</tr>
<tr>
<td>In Gain</td>
<td>Amount of gain inserted at the receiver side of the interface.</td>
<td></td>
</tr>
<tr>
<td>In Seizure</td>
<td>Incoming seizure state of the E&amp;M interface.</td>
<td></td>
</tr>
<tr>
<td>Initial Time Out</td>
<td>Amount of time the system waits for an initial input digit from the caller.</td>
<td></td>
</tr>
<tr>
<td>InterDigit Duration Timing</td>
<td>DTMF interdigit duration, in milliseconds.</td>
<td></td>
</tr>
<tr>
<td>InterDigit Pulse Duration Timing</td>
<td>Pulse dialing interdigit timing, in milliseconds.</td>
<td></td>
</tr>
<tr>
<td>Interdigit Time Out</td>
<td>Amount of time the system waits for a subsequent input digit from the caller.</td>
<td></td>
</tr>
<tr>
<td>Maintenance Mode</td>
<td>Maintenance mode of the voice port.</td>
<td></td>
</tr>
<tr>
<td>Maximum Playout Delay</td>
<td>The amount of time before the digital signal processor (DSP) starts to discard voice packets from the digital DSP buffer.</td>
<td></td>
</tr>
<tr>
<td>Music On Hold Threshold</td>
<td>Configured music-on-hold threshold value for this interface.</td>
<td></td>
</tr>
<tr>
<td>Noise Regeneration</td>
<td>Whether background noise should be played to fill silent gaps if VAD is activated.</td>
<td></td>
</tr>
<tr>
<td>Nominal Playout Delay</td>
<td>The amount of time the DSP waits before starting to play out the voice packets from the DSP buffer.</td>
<td></td>
</tr>
<tr>
<td>Non Linear Processing</td>
<td>Whether nonlinear processing is enabled for this port.</td>
<td></td>
</tr>
<tr>
<td>Number of signaling protocol errors</td>
<td>Number of signaling protocol errors.</td>
<td></td>
</tr>
<tr>
<td>Operation State</td>
<td>Operational state of the voice port.</td>
<td></td>
</tr>
<tr>
<td>Operation Type</td>
<td>Operation type of the E&amp;M signal: two-wire or four-wire.</td>
<td></td>
</tr>
<tr>
<td>Out Attenuation</td>
<td>Amount of attenuation inserted at the transmit side of the interface.</td>
<td></td>
</tr>
<tr>
<td>Out Seizure</td>
<td>Outgoing seizure state of the E&amp;M interface.</td>
<td></td>
</tr>
<tr>
<td>Port</td>
<td>Port number for the interface associated with the voice interface card.</td>
<td></td>
</tr>
<tr>
<td>Pulse Rate Timing</td>
<td>Pulse dialing rate, in pulses per second (pps).</td>
<td></td>
</tr>
<tr>
<td>Region Tone</td>
<td>Configured regional tone for this interface.</td>
<td></td>
</tr>
<tr>
<td>Ring Active Status</td>
<td>Ring active indication.</td>
<td></td>
</tr>
<tr>
<td>Ring Cadence</td>
<td>Configured ring cadence for this interface.</td>
<td></td>
</tr>
</tbody>
</table>
### Table 9  *show voice port Field Descriptions (continued)*

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Frequency</td>
<td>Configured ring frequency for this interface.</td>
</tr>
<tr>
<td>Ring Ground Status</td>
<td>Ring ground indication.</td>
</tr>
<tr>
<td>Ringing Time Out</td>
<td>Ringing timeout duration.</td>
</tr>
<tr>
<td>Signal Type</td>
<td>Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.</td>
</tr>
<tr>
<td>Slot</td>
<td>Slot used in the voice interface card for this port.</td>
</tr>
<tr>
<td>Sub-unit</td>
<td>Subunit used in the voice interface card for this port.</td>
</tr>
<tr>
<td>Tip Ground Status</td>
<td>Tip ground indication.</td>
</tr>
<tr>
<td>Type of VoicePort</td>
<td>Type of voice port: FXO, FXS, or E&amp;M.</td>
</tr>
<tr>
<td>The Interface Down Failure Cause</td>
<td>Text string describing why the interface is down,</td>
</tr>
<tr>
<td>Voice Activity Detection</td>
<td>Whether voice activity detection is enabled or disabled.</td>
</tr>
<tr>
<td>Wait Release Time Out</td>
<td>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.</td>
</tr>
<tr>
<td>Wink Duration Timing</td>
<td>Maximum wink duration for wink start signaling.</td>
</tr>
<tr>
<td>Wink Wait Duration Timing</td>
<td>Maximum wink wait duration for wink start signaling.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>h261</td>
<td>Video codec H.261</td>
</tr>
<tr>
<td>h263</td>
<td>Video codec H.263</td>
</tr>
<tr>
<td>h263+</td>
<td>Video codec H.263+</td>
</tr>
<tr>
<td>h264</td>
<td>Video codec H.264</td>
</tr>
</tbody>
</table>

**Command Default**

No video codec is configured.

**Command Modes**

Dial-peer configuration

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>video codec (voice-class)</td>
<td>Specifies a video codec for a voice class.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>video codec (dial-peer)</td>
<td>Specifies a video codec for a VoIP dial peer.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>inbound</td>
<td>Inbound voice class called number.</td>
</tr>
<tr>
<td>outbound</td>
<td>Outbound voice class called number.</td>
</tr>
<tr>
<td>pool</td>
<td>Voice class called number pool.</td>
</tr>
<tr>
<td>tag</td>
<td>Digits that identify a specific inbound or outbound voice class called number or voice class called number pool.</td>
</tr>
</tbody>
</table>

**Command Default**

No voice class called number is configured.

**Command Modes**

Global configuration

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>show voice class called-number</td>
<td>Displays a specific voice class called number.</td>
</tr>
<tr>
<td>voice-class called-number</td>
<td>Assigns a previously defined voice class called number to an inbound or</td>
</tr>
<tr>
<td>(dial-peer)</td>
<td>outbound POTS dial peer.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>voice class called number</td>
<td>Defines a voice class called number or range of numbers for H.320 calls.</td>
</tr>
<tr>
<td>voice-class called-number-pool</td>
<td>Defines a pool of dynamic voice class called numbers for a voice port.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Syntax</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tag</code></td>
<td>Digits that identify a specific voice class called number pool.</td>
</tr>
</tbody>
</table>

**Command Default**

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

**Command Modes**

Voice class configuration

**Command History**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(11)T</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>voice class called number</code></td>
<td>Defines a voice class called number or range of numbers for H.320 calls.</td>
</tr>
<tr>
<td><code>voice-class called-number</code></td>
<td>Defines a called number or range of called numbers for a POTS dial peer.</td>
</tr>
<tr>
<td><code>(dial-peer)</code></td>
<td></td>
</tr>
</tbody>
</table>
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.

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

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
</table>
| 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 |

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