Call Preservation for H.323 VoIP Calls

Revised: December 12, 2005

History for Call Preservation for H.323 VoIP Calls Feature

<table>
<thead>
<tr>
<th>Release</th>
<th>Cisco CallManager Express (Cisco CME) Version</th>
<th>Modification</th>
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<tr>
<td>12.4(4)XC</td>
<td>4.0</td>
<td>This feature was introduced.</td>
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</table>

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click Cancel at the login dialog box and follow the instructions that appear.

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- Restrictions for Call Preservation for H.323 VoIP Calls, page 2
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- How to Configure Call Preservation for H.323 VoIP Calls, page 4
- Configuration Examples for Call Preservation for H.323 VoIP Calls, page 10
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- Command Reference, page 13
- Open Source License Acknowledgements, page 18
- Open Source License Acknowledgements, page 18
Prerequisites for Call Preservation for H.323 VoIP Calls

- For bidirectional silence detection, Cisco IOS gateways with 5510 digital signal processors (DSPs) are needed.
- It is recommended that media inactivity detection be configured so that preserved calls are torn down after conversations are over. Two available media inactivity detection features are discussed in the “How to Configure Call Preservation for H.323 VoIP Calls” section on page 4. They are Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) inactivity detection and bidirectional silence detection. For more information about media inactivity detection, see the “Configuring Media Inactive Call Detection” chapter in the Cisco IOS Tcl IVR and VoiceXML Application Guide —12.3(14)T and Later.

Restrictions for Call Preservation for H.323 VoIP Calls

Call preservation for H.323 VoIP calls does not support the following:

- Calls in transient call states.
- Calls in for which a H.225.0 connection has not occurred,
- Calls on which supplementary services are in progress. For example, when one of the parties is on hold.
- Calls that involve a media resource located across a WAN, such as conference resources.
- Calls where the two parties are registered to different CallManager clusters.

Information About Call Preservation for H.323 VoIP Calls

This feature expands Cisco Survivable Remote Site Telephony’s (Cisco SRST’s) call preservation functionality to include the following types of failures and connections:

Failure Types

- WAN failures that include WAN links flapping or degraded WAN links
- Cisco CallManager software failure, such as when the ccm.exe service crashes on a Cisco CallManager server.
- LAN connectivity failure, except when a failure occurs at the local branch

Connection Types

- Calls between two Cisco CallManager controlled endpoints
  - During Cisco CallManager reloads
  - When a Transmission Control Protocol (TCP) connection between one or both endpoints and Cisco CallManager used for signaling H.225.0 or H.245 messages is lost or flapping
  - Between endpoints that are registered to different Cisco CallManagers in a cluster and the TCP connection between the two Cisco CallManagers is lost
  - Between IP phones and the PSTN at the same site
Call Preservation for H.323 VoIP Calls

- Calls between Cisco IOS gateway and an endpoint controlled by a softswitch where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint.
  - When the softswitch reloads.
  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint.
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the soft-switch does not clear the call on the gateway.
- Call flows that involve a Cisco IP in IP (IPIP) gateway running in media flow-around mode that reload or lose connection with the rest of the network.

Note that after the media is preserved, the call is torn down later when either one of the parties hangs up or media inactivity is detected. In cases where there is a machine-generated media stream, such as music streaming from a media server, the media inactivity detection will not work and the call may hang. Cisco CallManager addresses such conditions by indicating to the gateway that such calls should not be preserved, but third-party devices or IPIP gateways would not do this.

Flapping is defined for this feature as the repeated and temporary loss of IP connectivity that can be caused by WAN or LAN failures. H.323 VoIP calls between a Cisco IOS gateway and Cisco CallManager may be torn down when flapping occurs. When Cisco CallManager detects that the TCP connection is lost, it clears the call and closes the TCP sockets used for the call by sending a TCP FIN, without sending an “H.225.0 Release Complete” or “H.245 End Session” message. This is called quiet clearing. The TCP FIN sent from the Cisco CallManager could reach the gateway if the network comes up for a short duration, and the gateway will tear the call down. Even if the TCP FIN does not reach the gateway, the TCP keepalives sent from the gateway could reach Cisco CallManager when the network comes up. Cisco CallManager will send TCP RST messages in response to the keepalives as it has already closed the TCP connection. The gateway will tear down H.323 calls if it receives the RST message.

Configuration of the Call Preservation for H.323 VoIP Calls feature involves configuring the call preserve command and enabling the “Allow Peer to Preserve H.323 Calls” parameter from Cisco CallManager’s Service Parameters window.

The call preserve command causes the gateway to ignore socket closure or socket errors on H.225.0 or H.245 connections for active calls, allowing the socket to be closed without tearing down calls using those connections.

The Cisco CallManager’s “Allow Peer to Preserve H.323 Calls” parameter preserves the following:
- Active H323 calls with quiet clear triggered by the other half of the call.
- Active H323 calls with TCP socket closed on the H.323 end before the H.225 or H.245 release signal is received.
- Active H323 calls with a signal distribution layer (SDL) link that is out of service and detected on the H323 end.

Management

Call preservation may be reported through Syslog, which optionally can be obtained through a simple network management protocol (SNMP) trap. New syslog messages are printed when call preservation is applied. An SNMP trap can be configured on this syslog message, so you can be notified when call preservation occurs on a gateway.

Preservation information is displayed through the show h323 calls preserved command. The following is an example of the command’s output:
CallID = 11EC , Calling Number = , Called Number = 3210000 ,
RemoteSignallingIPAddress=9.13.0.26 , RemoteSignallingPort=49760 ,
RemoteMediaIPAddress=9.13.0.11 , RemoteMediaPort=17910 , Preserved Duration = 262 , Total
Duration = 562 , H225 FD = -1 , H245 FD = -1

The previous example represents one preserved call. One such display is provided per preserved call.
The show h323 calls preserved displays active calls only. No history is output.
To obtain additional information about a call, you can also use the show call active voice command.
Calls can be cleared with the clear call voice causecode command.

How to Configure Call Preservation for H.323 VoIP Calls

The tasks for configuring Call Preservation for H.323 VoIP Calls include the following:

- Configure Cisco IOS Gateway, page 4
- Configure Cisco CallManager, page 9

Configure Cisco IOS Gateway

The call preserve command activates H.323 VoIP call preservation. RTP and RTCP inactivity detection
and bidirectional silence detection can be used with this feature. Note that voice activity detection (VAD)
must be set to off if you are using RTP and RTCP inactivity detection. VAD may be set to on, for
bidirectional silence detection. For configuration examples, see the “RTP and RTCP Inactivity Detection
Configuration Example” section on page 11 and “Bidirectional Silence Detection Enable Example”
section on page 11.

When bidirectional silence and RTP and RTCP inactivity detection are configured, they are enabled for
all calls by default. To enable them for H.323 VoIP preserved calls only, you must use the call preserve
command’s limit-media-detection keyword.

H.323 VoIP call preservation can be applied to all calls and to dial peers. The required steps are described
in the following sections:

- Configure H.323 VoIP Call Preservation for All Calls, page 4
- Configure H.323 VoIP Call Preservation for a Dial Peer, page 6

Configure H.323 VoIP Call Preservation for All Calls

The following describes how to configure H.323 VoIP call preservation for all calls.

SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
4. h323
5. call preserve [limit-media-detection]
### 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 voice service voip</td>
<td>Enters voice-service configuration mode.</td>
</tr>
<tr>
<td>Example: Router (config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>Step 4 h323</td>
<td>Enables the H.323 voice service configuration commands.</td>
</tr>
<tr>
<td>Example: Router (config-voi-serv)# h323</td>
<td></td>
</tr>
<tr>
<td>Step 5 call preserve [limit-media-detection]</td>
<td>Enables the preservation of H.323 VoIP calls.</td>
</tr>
<tr>
<td>Example: Router (config-voi-h323)# call preserve</td>
<td>limit-media-detection—Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.</td>
</tr>
<tr>
<td>Step 6 exit</td>
<td>Exits H.323 configuration mode.</td>
</tr>
<tr>
<td>Example: Router# exit</td>
<td></td>
</tr>
<tr>
<td>Step 7 exit</td>
<td>Exist voice service voip configuration mode.</td>
</tr>
<tr>
<td>Example: Router# exit</td>
<td></td>
</tr>
</tbody>
</table>

### Examples

The following configuration example enables H.323 VoIP call preservation for all calls.

```
voice service voip
h323
call preserve
```

The following configuration example enables H.323 VoIP call preservation and limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to preserved calls only:

```
voice service voip
h323
call preserve limit-media-detection
```
Configure H.323 VoIP Call Preservation for a Dial Peer

The following describes how to configure H.323 VoIP call preservation for a dial peer.

**SUMMARY STEPS**

1. enable
2. configure terminal
3. voice-class h323 tag
4. call preserve [limit-media-detection]
5. exit
6. dial-peer voice tag voip
7. voice-class h323 tag
8. exit

**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 h323 tag | Assigns an H.323 voice class to a VoIP dial peer.  
• tag—Unique number to identify the voice class. Range is from 1 to 10000. |
| **Example:**       |         |
| Router (config)# voice-class h323 4 |         |
| **Step 4** call preserve [limit-media-detection] | Enables the preservation of H.323 VoIP calls.  
• limit-media-detection—Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only. |
| **Example:**       |         |
| Router (config-class)# call preserve |         |
| **Step 5** exit | Exits H.323 voice class configuration mode. |
| **Example:**       |         |
| Router (config)# exit |         |
| **Step 6** dial-peer voice tag voip | Defines a particular dial peer. |
| **Example:**       |         |
| Router (config)# dial-peer voice 1 voip |         |
Call Preservation for H.323 VoIP Calls

How to Configure Call Preservation for H.323 VoIP Calls

Examples

The following configuration example enables H.323 VoIP call preservation for dial peer 1.

```plaintext
voice-class h323 4
call preserve
dial-peer voice 1 voip
voice-class h323 4
```

Troubleshooting Tips

- Enable the `voice iec syslog` command in global configuration mode to display the reason that a call has disconnected after call preservation. The following is an example of the `voice iec syslog` command output line that displays this information:


- Calls on hold are not preserved and a non-standard message with “callPreserveIE FALSE” is sent in the notify message. Use the `debug h225 asn` command for debug. The following is example output:

  ```plaintext
  Router# debug h225 asn
  H.225 ASN1 Messages debugging is on
  3725-GW1#
  *May 3 15:57:27.920: H225.0 INCOMING ENCODE BUFFER::=
  28501900060008914A00040000D2D6D6D87EB11D0200000090D194410A00100110140B50000120A80A480040001000100
  h323-uu-pdu
  {
  h323-message-body notify :
  {
  protocolIdentifier { 0 0 8 2250 0 4 }
  callIdentifier
  {
  guid '00D2D6D6D87EB11D0200000090D1944'H
  }
  h245Tunneling FALSE
  nonStandardControl
  {
  {
  nonStandardIdentifier h221NonStandard :
  {
  t35CountryCode 181
t35Extension 0
  ```
How to Configure Call Preservation for H.323 VoIP Calls

When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'\n
When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'\n
When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'\n
When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'\n
When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'\n
When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

Router# debug h225 asc
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::= 80A68004000101000100'}
How to Configure Call Preservation for H.323 VoIP Calls

Call Preservation for H.323 VoIP Calls

Cisco IOS Release 12.4(4)XC

{ 
callMgrParam 
{ 
terclusterVersion 1 
enterpriseID {} 
} 
callSignallingParam 
{ 
connectedNumber '4C058032303030'H 
} 
callPreserveParam 
{ 
callPreserveIE TRUE 
}
}

- Use the `debug cch323 all` command after call setup to see if call is going into preserved state. Note that this command generates verbose output, and a console message is printed for every preserved call. In the following output, the relevant information appears in boldface:

```
Router# debug cch323 all
(CCH323-6-CALL_PRESERVED).
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/cch323_ct_main: SOCK 3 Event 0x1
Nov 29 12:39:55.167: //31/A9E0FB268017/H323/cch323_h225_handle_conn_loss:
cch323_h225_handle_conn_loss Call not torn down despite H.225.0 socket error: socket
error status = 1, ccb status = 403760899, fd = 3, pre-V3 = 0
call preserved due to socket closure or error, Call Id = 4593, fd = 3
IEC=1.1.186.5.7.6 on callID 31 GUID=A9E0FB26600B11DA801700653455072
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/h323_set_release_source_for_peer:
ownCallId[31], src[6]
TcpFDTbl
```

- The following are additional debug commands can be used to troubleshoot the problems associated with H.323 VoIP call preservation:
  - `debug h225 asn1`
  - `debug h225 q931`
  - `debug h245 asn1`

**What to Do Next**

Configure Cisco CallManager.

**Configure Cisco CallManager**

This section describes how to configure Cisco CallManager for the call preservation of H.323 VoIP calls.

**Procedure**

- **Step 1** Choose Service > Service Parameters.
- **Step 2** From the Service menu select Cisco CallManager.
Step 3  Click Advanced.

Step 4  Scroll to the Clusterwide Parameter (Device — H.323) section.

Step 5  Set the “Allow Peer to Preserve H.323 Calls” parameter to True.

Step 6  At the top of the screen click Update.

Configuration Examples for Call Preservation for H.323 VoIP Calls

The configuration examples in this section include the following:

- H.323 VoIP Call Preservation for All Calls Example
- H.323 VoIP Call Preservation for a Dial Peer Example
- H.323 Call Preservation for RTP and RTCP and Silence Detection Example
- RTP and RTCP Inactivity Detection Configuration Example
- Bidirectional Silence Detection Enable Example

H.323 VoIP Call Preservation for All Calls Example

The following configuration example enables H.323 VoIP call preservation for all calls:

```
voice service voip
h323
call preserve
```

H.323 VoIP Call Preservation for a Dial Peer Example

The following configuration example enables H.323 VoIP call preservation for one dial peer:

```
voice class h323 4
call preserve
dial-peer voice 1
voice class h323 4
```

H.323 Call Preservation for RTP and RTCP and Silence Detection Example

The following configuration example enables H.323 VoIP call preservation and limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only:

```
voice service voip
h323
call preserve limit-media-detection
```
RTP and RTCP Inactivity Detection Configuration Example

The following configuration example can be used to enable RTP and RTCP inactivity detection for dial peers. Note that for call preservation VAD must be set to off (no vad command):

dial-peer voice 10 voip
    no vad
gateway
    timer receive-rtcp 4
    ip rtcp report-interval 60

Bidirectional Silence Detection Enable Example

The following configuration example enables bidirectional silence detection:

gateway
    timer media-inactive 5
    ip rtcp report interval
Additional References

The following sections provide references related to Call Preservation for H.323 VoIP Calls.

Related Documents

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
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<tbody>
<tr>
<td>Cisco CallManager</td>
<td>Cisco CallManager Documentation</td>
</tr>
<tr>
<td>Cisco SRST</td>
<td>Cisco SRST Documentation</td>
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<tr>
<td>Syslog</td>
<td>Syslog Documentation</td>
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Standards

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<td>No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.</td>
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MIBs

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<tr>
<td>Cisco SRST SNMP 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>
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RFCs

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Technical Assistance

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<tr>
<td></td>
<td>The Cisco Technical Support &amp; Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.</td>
</tr>
</tbody>
</table>
Command Reference

This section documents new commands only.
**Call Preserves**

To enable the preservation of H.323 VoIP calls, use the `call preserve` command in `h323`, `voice-class h323`, and `voice service voip` configuration modes. To reset to the default, use the `no` form of this command.

```
call preserve [limit-media-detection]
```
```
no call preserve [limit-media-detection]
```

<table>
<thead>
<tr>
<th>Syntax Description</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>limit-media-detection</code></td>
<td>Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.</td>
</tr>
</tbody>
</table>

| Command Default | H.323 VoIP call preservation is disabled. |
| Command Modes | `h323`, `voice-class h323`, or `voice service voip` |

<table>
<thead>
<tr>
<th>Command History</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco IOS Release</strong></td>
<td><strong>Version</strong></td>
</tr>
<tr>
<td>12.4(4)XC</td>
<td>Cisco SRST 4.0</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `call preserve` command activates H.323 VoIP call preservation for following types of failures and connections:

**Failure Types**
- WAN failures that include WAN links flapping or degraded WAN links
- Cisco CallManager software failure, such as when the `ccm.exe` service crashes on a Cisco CallManager server.
- LAN connectivity failure, except when a failure occurs at the local branch

**Connection Types**
- Calls between two Cisco CallManager controlled endpoints
  - During Cisco CallManager reloads
  - When a Transmission Control Protocol (TCP) connection between one or both endpoints and Cisco CallManager used for signaling H.225.0 or H.245 messages is lost or flapping
  - Between endpoints that are registered to different Cisco CallManagers in a cluster and the TCP connection between the two Cisco CallManagers is lost
  - Between IP phones and the PSTN at the same site
Call Preservation for H.323 VoIP Calls

- Calls between Cisco IOS gateway and an endpoint controlled by a softswitch where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint.
  - When the softswitch reloads.
  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint.
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the soft-switch does not clear the call on the gateway.
- Call flows that involve a Cisco IP in IP (IPIP) gateway running in media flow-around mode that reload or lose connection with the rest of the network.

When bidirectional silence and RTP and RTCP inactivity detection are configured, they are enabled for all calls by default. To enable them for H.323 VoIP preserved calls only, you must use the `call preserve` command's `limit-media-detection` keyword.

H.323 VoIP call preservation can be applied globally to all calls and to a dial peer.

### Examples

The following example enables H.323 VoIP call preservation for all calls.

```plaintext
voice service voip
  h323
  call preserve
```

The following configuration example enables H.323 VoIP call preservation for dial peer 1.

```plaintext
voice-class h323 4
  call preserve
dial-peer voice 1 voip
  voice-class h323 4
```

The following example enables H.323 VoIP call preservation and enables RTP and RTCP inactivity detection and bidirectional silence detection for preserved calls only:

```plaintext
voice service voip
  h323
  call preserve limit-media-detection
```

The following example enables RTP and RTCP inactivity detection. Note that for H.323 VoIP call preservation VAD must be set to off (`no vad` command).

```plaintext
dial-peer voice 10 voip
  no vad
  gateway
timer receive-rtcp
  ip rtcp report-interval
```

The following configuration example enables bidirectional silence detection:

```plaintext
gateway
timer media-inactive
  ip rtcp report interval
```
<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>h323</td>
<td>Enables the H.323 voice service configuration commands.</td>
</tr>
<tr>
<td>show h323 calls</td>
<td>Displays data about active H.323 VoIP preserved calls.</td>
</tr>
<tr>
<td>preserved</td>
<td></td>
</tr>
<tr>
<td>voice-class h323</td>
<td>Assigns an H.323 voice class to a VoIP dial peer.</td>
</tr>
<tr>
<td>voice service voip</td>
<td>Enters voice-service configuration mode</td>
</tr>
</tbody>
</table>
**show h323 calls preserved**

To display data about active H.323 VoIP preserved calls, use the `show h323 calls preserved` command in user EXEC or privileged EXEC mode.

```
show h323 calls preserved
```

**Command Modes**

User EXEC
Privileged EXEC

**Command History**

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Version</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.4(4)XC</td>
<td>Cisco SRST 4.0</td>
<td>This command was introduced.</td>
</tr>
</tbody>
</table>

**Usage Guidelines**

The `show h323 calls preserved` command displays data per preserved call. Only active calls are displayed; preserved call history is not.

If translation rules are configured, the value displayed in the “Calling Number” field may have been translated by a gateway. Gateways handle called number values as the numbers to which calls are routed.

The “CallID” field displays the shorter form of the 16-octet, globally-unique connection ID that is allocated for each call leg. The show call active voice brief command also displays a shorter form of the CallID value (part of the third octet and the fourth octet). The longer form of the CallID value is output by the `show call active voice` command.

The CallID value can be used to refer to a call leg associated with the CallID when issuing other voice commands on the gateway, such as the `show voice call status` command and the `clear call voice` command.

An output value of -1 displayed in the “H225 FD” or “H245 FD” field denotes that the call was preserved due to an error detected on the H.225.0 connection. The actual H.225.0 socket file descriptor used for this call can be found from the syslog message that was output when this call was preserved.

To obtain more information about a call, you can also use the `show call active voice` command. Calls can be cleared with the `clear call voice causecode` command.

**Examples**

The following is sample output from the `show h323 calls preserved` command where one active call is preserved:

```
Router# show h323 calls preserved

CallID = 11EC , Calling Number = , Called Number = 3210000 ,
RemoteSignallingIPAddress=9.13.0.26 , RemoteSignallingPort=49760 ,
RemoteMediaIPAddress=9.13.0.11 , RemoteMediaPort=17910 , Preserved Duration = 262 , Total Duration = 562 , H225 FD = -1 , H245 FD = -1
```

Table 1 provides an alphabetical listing of the fields displayed in the output of the `show h323 calls preserved` command and a description of each field.
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The implementation was written so as to conform with Netscapes SSL.

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