Enhancement to Cisco H.323 to Support RSVP Slow Connect

Feature History

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.1(1)T</td>
<td>This feature was introduced.</td>
</tr>
<tr>
<td>12.1(5)XM2</td>
<td>Support was added for the Cisco AS5350 and Cisco AS5400 universal gateways.</td>
</tr>
</tbody>
</table>

This feature module describes the Enhancement to Cisco H.323 to Support RSVP Slow Connect feature that was introduced in Cisco IOS Release 12.1(1)T. This document contains the following sections:

- Feature Overview, page 1
- Supported Platforms, page 2
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Feature Overview

This feature modifies H.323 call flows so that when RSVP-based (Resource Reservation Protocol) QoS (Quality of Service) is enabled, the H.323 endpoint waits for the bandwidth reservation to be confirmed by RSVP before allowing call setup to be completed.

In previous releases of Cisco IOS software, an H.323 gateway requested an RSVP reservation and proceeded with call establishment without first waiting for a response to the RSVP request. As a result, calls were allowed to be completed even if only best effort service was available, which was unacceptable for many scenarios. For example, a call might proceed to the alerting phase (ringing), only for reservations to fail afterwards. In this case, although a certain desired quality of service of reservation had been requested, it was not honored.

This feature resolves this pitfall by ensuring that if calls are associated with dial peers that have been appropriately configured with QoS, the calls will not succeed without confirmation of QoS reservations.
This feature is not explicitly configurable. If the corresponding dial peer has been configured for RSVP (in other words, if the req-qos is set to a value other than the default of best-effort), then traditional slow connect procedures will be followed, and the endpoint will neither attempt to initiate Fast Connect nor respond to a Fast Connect request from its peer.

Benefits

QoS Reservation Confirmation
When RSVP-based QoS is enabled, the H.323 endpoint waits for the bandwidth reservation to be confirmed by RSVP before allowing the call setup to be completed.

Restrictions

If the Cisco terminating endpoint receives a set-up message that does not contain an H.245 address, and if the terminating endpoint is configured only for nonbest effort quality of service, then the endpoint will terminate the call with an appropriate release value so that an attempt can be made to route the call using a different dial peer. If the Cisco originating endpoint receives an alerting message from its peer before RSVP reservations are completed, then the call will proceed, but QoS is not guaranteed.

Related Documents

“Voice-Related Commands” chapter of the Cisco IOS Release 12.0 Voice, Video, and Home Applications Command Reference

Supported Platforms

- Cisco 2600 series routers
- Cisco 3600 series routers
- Cisco AS5300 access server
- Cisco AS5350 universal gateway
- Cisco AS5400 universal gateway

Supported Standards, MIBs, and RFCs

Standards
Appendix II of H.323 v.4.

MIBs
No new or modified MIBs are supported by this feature.

To obtain lists of MIBs supported by platform and Cisco IOS release and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (CCO) at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.
Prerequisites

The Cisco AS5350 and Cisco AS5400 do not support the Mica Modem Card, Microcom Modem Card, or VoIP Feature Card. Voice and modem functions are provided by the Universal Port Dial Feature card running SPE firmware. See the Cisco AS5350 Universal Gateway Card Installation Guide and the Cisco AS5400 Universal Gateway Card Installation Guide for more information. All references to the Cisco AS5300 in this document apply to the Cisco AS5350 and Cisco AS5400 platforms with the following exceptions:

- Use the Universal Port Dial Feature Card instead of the Mica or Microcom modem cards.
- Use SPE firmware instead of portware version 6.7.7.
- Run Cisco IOS Release 12.1(5)XM2 software for VoIP functionality.

Configuration Tasks

See the following sections for configuration tasks for the H.323 Support for RSVP feature. Each task in the list indicates if the task is optional or required.

- Configuring Active Interfaces (required)
- Configuring QoS (required)

Configuring Active Interfaces

To minimally configure RSVP for voice traffic, you must enable RSVP on each interface where priority needs to be set.

<table>
<thead>
<tr>
<th>Command</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Router(config)# int e 0/0</td>
</tr>
<tr>
<td></td>
<td>Enters interface configuration mode.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Router(config-interface)# ip rsvp bandwidth 100 32</td>
</tr>
<tr>
<td></td>
<td>Enables RSVP and sets the maximum bandwidth to 100 kbps and the maximum bandwidth per single request to 32 kbps.</td>
</tr>
<tr>
<td></td>
<td>The default maximum bandwidth is up to 75% of the bandwidth available on the interface. By default, the amount reservable by a flow can be up to the entire reservable bandwidth.</td>
</tr>
<tr>
<td></td>
<td>Router(config-interface)# fair-queue</td>
</tr>
<tr>
<td>Step 3</td>
<td>Router(config-interface)# exit</td>
</tr>
<tr>
<td></td>
<td>Exits interface configuration mode.</td>
</tr>
</tbody>
</table>

Configuring QoS

Configuring the appropriate level of QoS ensures that the desired and minimum level quality of service is honored when establishing H.323 VoIP calls.
The default setting for this command is best-effort, which indicates that RSVP makes no bandwidth reservation. One of the other two options should be selected to not allow the H.323 call to proceed and possibly fail.

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**Configuration Examples**

This section provides the following configuration examples:

- Enabling RSVP
- Requesting an RSVP Session

### Enabling RSVP

The following example enables RSVP and sets the maximum bandwidth to 100 kbps and the maximum bandwidth per single request to 32 kbps (the example presumes that both VoIP dial peers have been configured):

```plaintext
Router(config)# interface serial 0/0
Router(config-if)# ip rsvp bandwidth 100 32
Router(config-if)# fair-queue
Router(config-if)# end
```

### Requesting an RSVP Session

After enabling RSVP, you must also use the `req-qos` dial-peer configuration command to request an RSVP session on each VoIP dial peer. Otherwise, no bandwidth is reserved for voice traffic.

```plaintext
Router(config)# dial-peer voice 211 voip
Router(config-dial-peer)# req-qos controlled-load
Router(config)# dial-peer voice 212 voip
Router(config-dial-peer)# req-qos controlled-load
```
Command Reference

There are no new or changed commands for this feature.

Glossary

dial peer—An addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.

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endpoint—An H.323 terminal or gateway. An endpoint can call and be called. It generates and/or terminates the information stream.

gatekeeper—A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup, and request admission to a call from the gatekeeper.

The gatekeeper is an H.323 entity on the LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper may provide other services to the H.323 terminals and gateways, such as bandwidth management and locating gateways.

gateway—A gateway allows H.323 terminals to communicate with non-H.323 terminals by converting protocols. A gateway is the point at which a circuit-switched call is encoded and repackaged into IP packets.

An H.323 gateway is an endpoint on the LAN that provides real-time, two-way communications between H.323 terminals on the LAN and other ITU-T terminals in the WAN, or to another H.323 gateway.

H.323—An International Telecommunications Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. H.323 is an umbrella standard that describes the architecture of the conferencing system, and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol.

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POTS—Plain old telephone service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the PSTN.

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PSTN—Public Switched Telephone Network. PSTN refers to the local telephone company.

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QoS—Quality of Service. Refers to the measure of service quality provided to the user.

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VoIP—Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term which generally refers to Cisco’s standards-based (for example, H.323) approach to IP voice traffic.

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For a list of other internetworking terms, see Internetworking Terms and Acronyms, available on the Documentation CD-ROM and Cisco Connection Online (CCO) at the following URL: http://www.cisco.com/univercd/cc/td/doc/cisintwk/ita/index.htm.