



Configuring RTCP Report Generation

Last Updated: December 20, 2011

The assisted Real-time Transport Control Protocol (RTCP) feature adds the ability for Cisco Unified Border Element (Cisco UBE) to generate standard RTCP keepalive reports on behalf of endpoints. RTCP reports determine the liveliness of a media session during prolonged periods of silence, such as call hold or mute. Therefore, it is important for the Cisco UBE to generate RTCP reports irrespective of whether the endpoints send or receive media.

Cisco UBE generates RTCP report only when inbound and outbound call legs are SIP, or SIP to H.323, or H.323 to SIP.

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Finding Feature Information

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 Table at the end of this document.

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

Prerequisites

Cisco Unified Border Element

- Cisco IOS Release 15.1(2)T or a later release must be installed and running on your Cisco Unified Border Element.



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Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release <TBD> or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions

- RTCP report generation over IPv6 is not supported.
- RTCP report generation is not supported for Secure Real-time Transport Protocol (SRTP) or SRT Control Protocol (SRTCP) pass-through as Cisco UBE is not aware of the media encryption or decryption keys.
- RTCP report generation is not supported for loopback calls, T.38 fax, and modem relay calls.
- RTCP or SRTCP report generation is not supported when Cisco UBE inserts a Digital Signal Processor (DSP) for RTP-SRTP interworking on RTP and SRTP call legs.
- RTCP report generation is not supported when there is a call hold with an invalid media address such as 0.0.0.0 in Session Description Protocol (SDP) or Open Logical Channel (OLC).
- RTCP report generation is not supported for RTCP multiplexed with RTP on the same address and port.
- RTCP report generation is not supported on enterprise aggregation services routers (ASR) Cisco UBE.
- RTCP packet generation is not supported on the SIP leg when the H.323 leg puts the SIP leg on hold in a Slow Start to Delayed-Offer call.

Configuring RTCP Report Generation on Cisco UBE

RTCP keepalive packets indicate session liveliness. When configured on Cisco UBE, RTCP keepalive packets are sent on both inbound and outbound SIP or H.323 call legs.

Perform this task to configure RTCP report generation on Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type to to-type*
5. **rtcp keepalive**
6. **end**

DETAILED STEPS

Command or Action	Purpose
<p>Step 1 <code>enable</code></p> <p>Example:</p> <pre>Router> enable</pre>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> Enter your password if prompted.
<p>Step 2 <code>configure terminal</code></p> <p>Example:</p> <pre>Router# configure terminal</pre>	<p>Enters global configuration mode.</p>
<p>Step 3 <code>voice service voip</code></p> <p>Example:</p> <pre>Router(config)# voice service voip</pre>	<p>Enters voice service configuration mode.</p>
<p>Step 4 <code>allow-connections from-type to-type</code></p> <p>Example:</p> <pre>Router(conf-voi-serv)# allow-connections sip to sip</pre>	<p>Allows connections between SIP endpoints in a VoIP network.</p>
<p>Step 5 <code>rtcp keepalive</code></p> <p>Example:</p> <pre>Router(conf-voi-serv)# rtcp keepalive</pre>	<p>Configures RTCP keepalive report generation.</p>
<p>Step 6 <code>end</code></p> <p>Example:</p> <pre>Router(conf-voi-serv)# end</pre>	<p>Exits voice service configuration mode and returns to privileged EXEC mode.</p>

Troubleshooting Tips

Use the following debug commands for debugging related to RTCP keepalive packets:

- debug voip rtcp packet** --Shows details related to RTCP keepalive packets such as RTCP sending and receiving paths, Call ID, Globally Unique Identifier (GUID), packet header, and so on.

```
Router# debug voip rtcp packet
```

```

01:06:27.450: //6/xxxxxxxxxxxxx/RTP//Event/voip_rtp_send_rtcp_keepalive: Generate RTCP
Keepalive
*Mar 17 01:06:27.450: rtcp_send_report: Attributes
      (src ip=192.168.30.3, src port=17101, dst ip=192.168.30.4, dst port=18619
      bye=0, initial=1, ssrc=0x07111E02, keepalive=1)
*Mar 17 01:06:27.450: rtcp_construct_keepalive_report: Constructed Report
      (rtcp=0x2E5AF214, ssrc=0x07111E02, source->ssrc=0x000001E03, total_len=36)
2E5AF210:      80C90001 07111E02 81CA0006      .I.....J..
2E5AF220: 07111E02 010F302E 302E3040 392E3435      .....0.0.0@9.45
2E5AF230: 2E33302E 33000000 00      .30.3....

```

**Caution**

Under moderate traffic loads, the **debug voip rtp packet** command produces a high volume of output and the command should be enabled only when the call volume is very low.

- **debug voip rtp packet** --Shows details about VoIP RTP packet debugging trace.

```

Router# debug voip rtp packet
VOIP RTP All Packets debugging is on

```

- **debug voip rtp session** --Shows all RTP session debug information.

```

Router# debug voip rtp session
VOIP RTP All Events debugging is on

```

- **debug voip rtp error** --Shows details about debugging trace for RTP packet error cases.

```

Router# debug voip rtp error
VOIP RTP Errors debugging is on

```

- **debug ip rtp protocol** --Shows details about RTP protocol debugging trace.

```

Router# debug ip rtp protocol
RTP protocol debugging is on

```

- **debug voip rtcp session** --Shows all RTCP session debug information.

```

Router# debug voip rtcp session
VOIP RTCP Events debugging is on

```

- **debug voip rtcp error** -- Shows details about debugging trace for RTCP packet error cases.

```

Router# debug voip rtcp error
VOIP RTCP Errors debugging is on

```

Feature Information for Configuring RTCP Report Generation

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

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

Feature History Table entry for the Cisco Unified Border Element. .

Table 1 **Feature Information for Configuring RTCP Report Generation**

Feature Name	Releases	Feature Information
Assisted RTCP	15.1(2)T	<p>This feature adds the ability for Cisco UBE to generate standard RTCP keepalive reports on behalf of endpoints and ensures the liveliness of a media session during prolonged periods of silence, such as call hold.</p> <p>The following commands were introduced or modified in this release: rtcp keepalive, debug voip rtcp, debug voip rtp, debug ip rtp protocol, and ip rtcp report interval.</p>

Feature History Table entry for the Cisco Unified Border Element (Enterprise) .

Table 2 **Feature Information for Configuring RTCP Report Generation**

Feature Name	Releases	Feature Information
Assisted RTCP	TBD	<p>This feature adds the ability for Cisco UBE to generate standard RTCP keepalive reports on behalf of endpoints and ensures the liveliness of a media session during prolonged periods of silence, such as call hold.</p> <p>The following commands were introduced or modified in this release: rtcp keepalive, debug voip rtcp, debug voip rtp, debug ip rtp protocol, and ip rtcp report interval.</p>

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