

Call Quality Statistics

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Overview

Call quality statistics in CUBE, such as packet loss, jitter, and round trip delay can be added to the call detail record (CDR), and these voice metrics can be calculated in IOS. For more information, refer to Voice Quality Enhancements on Cisco Unified Border Element.

The call quality statistics feature is enhanced to provide the following capabilities:

- Enable or disable Quality of Service (QoS) for CUBE.
- Enable or disable Real-time Transport Protocol (RTP) Control Protocol (RTCP) passthrough.
- Configure call quality criteria parameters.

Call quality configuration parameters include max_dropout, max_reorder, and clock_rate. A maximum of three codecs (codec_number, payload_type, clock_rate) per media flow is collected by the PI and sent to CPP, which uses these values in statistics calculation. Calculated statistics such as Jitter, Packet Loss, and Delay are then fetched from the CPP to the CDR. These statistics can be viewed in the command line interface.

The CDR has the following data per call leg of the call:

- Packet Loss-Calculated based on methods shown in RFC3550. The RTCP sender/receiver reports are recalculated, and not just copied from the inbound leg to the outbound leg.
- Delay-Calculated based on timestamp received or timestamp of packets sent.
- Jitter-Variation of delay.

For more information on how to calculate the voice quality metrics related to media(voice) quality, such as conversational mean opinion score (MOS), jitter, and so on, see http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voi-cube-call-monitoring.html.

The VQM (Voice Quality Monitor) gives information on the voice quality metrics. The VQM on Cisco IOS XE platforms enables statistics gathering based on the received RTCP packets. From these statistics, a voice

quality measurement is developed to show the quality of the call. The output is in a simple format, using a system of good, poor, and bad types of ratings.

The following metrics exists in Call Detail Record (CDR) and Management Information Base (MIB) in CUBE, indicating voice quality:

1. MOSQe (conversational quality MOS)
2. Round-trip-delay.
3. Receive-delay (current jitter buffer size).
4. Packet-Loss-Rate.

The CDR is sent at the end of a call if AAA accounting is configured.

A CDR example is as follows:

<MOS-Con>4.4072</MOS-Con>

<round-trip-delay>1 ms</round-trip-delay>

<receive-delay>64 ms</receive-delay>

<voice-quality-total-packet-loss>0.0000 %</ voice-quality-total-packet-loss>

Feature Information

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.

Table 1: Feature Information for Call Quality Statistics Enhancement

Feature Name	Releases	Feature Information
Call Quality Statistics Enhancement	Cisco IOS XE 3.14S	Call quality statistics in CUBE, such as packet loss, jitter, and round trip delay can be added to the call detail record (CDR), and these voice metrics can be calculated in IOS. For more information, refer to Voice Quality Enhancements on Cisco Unified Border Element. The call quality statistics feature is enhanced to provide the following capabilities: • Enable or disable Quality of Service (QoS) for CUBE. • Enable or disable Real-time Transport Protocol (RTP) Control Protocol (RTCP) passthrough. • Configure call quality criteria parameters.

Restrictions

- Only SIP-to-SIP call quality statistics calculation is supported.
- The RTCP field is not recalculated, as it is end-to-end statistics.
- The round trip delay is only retrieved by RTCP, which means the round trip delay is not calculated if there is no related RTCP.
- Only three codec types are supported for one media flow to calculate the jitter; considering the data path performance, these three codecs would be the maximum number in one cache line.
- Only one RTP synchronization source (SSRC) is supported concurrently per media flow, which is indicated in the m-line of the session description protocol (SDP).
- Round trip delay calculation for transcoding calls is not supported.

Configure Call Quality Parameters

Configure Call Quality Criteria Parameters

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **call-quality**
5. **max-dropout** *number-of-packets*
6. **max-reorder** *number-of-packets*
7. **clock-rate** *payload-type-number frequency*
8. **clock-rate dynamic-default** *frequency*
9. **exit**
10. **rtcp all-pass-through**
11. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters global VoIP configuration mode.
Step 4	**call-quality** **Example:** `Device(conf-voi-serv)# call-quality`	Enters call quality configuration mode; this is the global call quality of service setup.
Step 5	**max-dropout** *number-of-packets* **Example:** `Device(conf-serv-call-quality)# max-dropout 300`	Configures the acceptable out of sequence future packets to drop. The range is from 2 to 2000 packets. The default value is 100.
Step 6	**max-reorder** *number-of-packets* **Example:** `Device(conf-serv-call-quality)# max-reorder 500`	Configures the acceptable out of sequence late packets. The range is from 2 to 2000 packets. The default value is 100.
Step 7	**clock-rate** *payload-type-number frequency* **Example:** `Device(conf-serv-call-quality)# clock-rate 5 1500`	Sets the payload type number and frequency. Clock rate is the RTP timestamp field's sampling frequency.
Step 8	**clock-rate dynamic-default** *frequency* **Example:** `Device(conf-serv-call-quality)# clock-rate dynamic-default 10000`	(Optional) Changes the default clock rate for all the dynamic payload types. The frequency range (in Hz) is from 1000 to 192000. • You have several options to set the clock rate, such as for the different codecs.
Step 9	**exit** **Example:** `Device(conf-serv-call-quality)# exit`	Exits to global VoIP configuration mode.
Step 10	**rtcp all-pass-through** **Example:** `Device(conf-voi-serv)# rtcp all-pass-through`	(Optional) Passes through all RTCP in data path.
Step 11	**end** **Example:** `Device(conf-voi-serv)# end`	Returns to privileged EXEC mode.

Tips to Troubleshoot

Use the following **debug** and **show** commands to enable the logs, which helps in debugging:

- **debug ccsip verbose**
- **debug voip fpi all**
- **debug platform hardware qfp active feature sbc dbe datapath all**
- **debug platform hard qfp act feature sbc dbe client all**
- **debug ccsip message**
- **debug ccsip info**
- **show call active voice**
- **show platform hardware qfp active feature sbc data path call** *call-id*

The following are some show command outputs that would be useful in troubleshooting:

- **`Device# show call active voice | include LostPackets`**

LostPackets=0

LostPackets=36 ---->//*Lost packets detail present in* ***show call active voice*** *output. View the complete command output based on the filters such as call-id to check the packet loss for a particular call leg.*//

- **`Device# show call active voice | include PlayDelayJitter`**

PlayDelayJitter=0

PlayDelayJitter=38 ----->//*Jitter detail present in* ***show call active voice*** *output. View the complete command output based on the filters such as call-id to check the Jitter for a particular call leg.*//

Configuration Example for Call Quality Statistics

```
voice service voip
 no ip address trusted authenticate
 callmonitor
 rtcp all-pass-through
 media statistics
 media bulk-stats
 allow-connections sip to sip
 call-quality
  max-dropout 2
  max-reorder 2
 sip
  g729 annexb-all
  no call service stop
```