



# Voice Quality Monitoring

The Voice Quality Monitoring (VQM) feature gives information on the voice quality metrics related to media (voice) quality, such as conversational mean opinion score (MOS), packet loss rate, and so on. VQM enables you to monitor the quality of calls traversing your VoIP network, and you can diagnose the cause of voice quality issues and troubleshoot them.

The Voice Quality Statistics feature provides information about the quality of the Time-Division Multiplexing Internet Protocol (TDM-IP) voice call.

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## Feature Information for Voice Quality Monitoring

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 <https://cfng.cisco.com/>. An account on Cisco.com is not required.

**Table 1: Feature Information for Voice Quality Monitoring and Voice Quality Statistics**

Feature Name	Releases	Feature Information
Voice Quality Statistics	Cisco IOS XE Everest 16.5.1b	Voice quality statistics provides information about the quality of the voice TDM-IP call. This feature is already implemented on ISR-G2, and the feature gap is filled in ISR 4000 series.

Feature Name	Releases	Feature Information
Voice Quality Monitoring	Cisco IOS XE Denali 16.3.1	The Voice Quality Monitoring (VQM) feature provides information on the voice quality metrics related to media (voice) quality, such as conversational mean opinion score (MOS), packet loss rate, and so on. VQM enables you to monitor the quality of calls traversing your VoIP network, and you can diagnose the cause of voice quality issues and troubleshoot them.

## Prerequisites for Voice Quality Monitoring

The following commands must be executed to configure the voice quality metrics:

- **callmonitor**
- **rtcp all-pass-through**
- **media statistics**
- **media bulk-stats**
- **call-quality**
  - **max-dropout** *number-of-packets*—Configures the acceptable out of sequence future packets to drop. The range is 2–2000 packets. The default value is 100.
  - **max-reorder** *number-of-packets*—Configures the acceptable out of sequence late packets. The range is 2–2000 packets. The default value is 100.




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**Note** The values of **max-dropout** and **max-reorder** must be configured based on the network loss and network latency. In a lossy or high latency network, it's recommended to configure higher values. In a loss less or low latency network, lower values are fine. Packet loss and packet reorder are calculated based on RFC3550.

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### Sample Configuration

The following is a sample for the recommended voice quality monitoring configuration:

```
callmonitor
rtcpall-pass-through
media statistics
media bulk-stats
call-quality
  max-dropout 100
  max-reorder 100
```

# Restrictions for Voice Quality Monitoring and Voice Quality Statistics

- 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.
- VQM MOS values are not calculated for DSP based calls.
- MOS value shows 0 if endpoint does not send RTCP packets.
- The voice quality statistics covers only the TDM-IP call. The implementation focuses on filling the following statistics based on query response from DSP for TDM-SIP and TDM-H323 call:
  - RoundTripDelay
  - GapFillWithSilence
  - GapFillWithPrediction
  - GapFillWithInterpolation
  - GapFillWithRedundancy
  - HiWaterPlayoutDelay
  - LoWaterPlayoutDelay
  - PlayDelayJitter

## Information About Voice Quality Monitoring

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 exist 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>
```

## VQM Metrics

The following are the metrics exported by VQM:

IOS VQM, Voice/Audio Description Quality Metric	Description
MOS-Con	The conversational quality MOS. Conversational quality indicates the impact of the quality of the transmission on the dynamics of conversational exchanges between two parties; such metrics take into account delay, echo, and recency.
round-trip-delay	The instantaneous round-trip delay. This may be obtained from the RTCP SR reports.
receive-delay	The minimum delay that will be applied to the packets received when using an adaptive jitter buffer.
voice-quality-total-packet-loss	The total number of packets lost by the jitter buffer in the RTP stream.

# How to Configure Voice Quality Monitoring

## Enabling Media Statistics Globally

Perform this task to globally enable media statistics in voice-service configuration mode to estimate the values for packet loss, jitter, and round-trip time.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media statistics**
5. **end**

## DETAILED STEPS

	Command or Action	Purpose
Step 1	<b>enable</b> <b>Example:</b> <pre>Device&gt; enable</pre>	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b> <b>Example:</b> <pre>Device# configure terminal</pre>	Enters global configuration mode.
Step 3	<b>voice service voip</b> <b>Example:</b> <pre>Device(config)# voice service voip</pre>	Enters voice service VoIP configuration mode.
Step 4	<b>media statistics</b> <b>Example:</b> <pre>Device(conf-voi-serv)# media statistics</pre>	Enables media statistics to estimate the values of packet loss, jitter, and Round Trip Time (RTT) statistics. <ul style="list-style-type: none"> <li>• The statistics are displayed using the <b>show voice history</b> and <b>show call active voice</b> commands.</li> <li>• If the <b>media statistics</b> command is disabled, the values will be zero.</li> </ul>
Step 5	<b>end</b> <b>Example:</b> <pre>Device(conf-voi-serv)# end</pre>	Returns to privileged EXEC mode.

## Verifying Voice Quality Monitoring

Perform this task to verify the configuration for voice quality monitoring. The **show** commands can be entered in any order.

## SUMMARY STEPS

1. **enable**
2. **show call active voice | include LostPackets**
3. **show call active voice | include ReceiveDelay**
4. **show call active voice brief | sec RTT**
5. **show call active voice stats | sec MC**

## DETAILED STEPS

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### Step 1 enable

Enables privileged EXEC mode.

#### Example:

```
Device> enable
```

### Step 2 show call active voice | include LostPackets

Displays statistics on the CUBE if the Voice Quality Metrics feature is configured.

#### Example:

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

LostPackets=0
LostPackets=0
```

### Step 3 show call active voice | include ReceiveDelay

Displays statistics on the CUBE if the Voice Quality Metrics feature is configured.

#### Example:

```
Device# show call active voice | include ReceiveDelay

ReceiveDelay=0
ReceiveDelay=0
```

### Step 4 show call active voice brief | sec RTT

Displays a truncated version of call information for voice calls in CUBE if the Voice Quality Metrics feature is configured.

**Note** This command is not applicable for TDM-IP voice calls.

#### Example:

```
Device# show call active voice brief | sec RTT

IP 173.39.65.81:7078      SRTP: off  rtt:12ms pl:0/0ms  lost:0/0/0  delay:0/0/0ms  g711ulaw
TextRelay: off  Transcoded: No  ICE: Off
IP 10.127.17.141:18920  SRTP: off  rtt:12ms pl:0/0ms  lost:0/0/0  delay:0/0/0ms  g711ulaw
TextRelay: off  Transcoded: No  ICE: Off
```

### Step 5 show call active voice stats | sec MC

Displays R-Factor Statistics (G.107 MOS) on the CUBE if the Voice Quality Metrics feature is configured. A sample output is provided below for a voice call using G.711ulaw, VAD on, and at 5 percent packet loss rate.

#### Example:

```
Device# show call active voice stats | sec MC

DSP/RF: ML=, MC=, R1=, R2=, IF=, ID=, IE=, BL=, R0=, VR=
DSP/RF: ML=4.2346, MC=4.2346, R1=92, R2=92, IF=0, ID=0, IE=0, BL=0, R0=93, VR=2.0
DSP/RF: ML=4.2346, MC=4.2346, R1=92, R2=92, IF=0, ID=0, IE=0, BL=0, R0=93, VR=2.0
```

The following is an example output for the SNMP MIB:  
**cmqVoIPCallActiveRxPred107RMosConv.8520964.1 = 423 (MC)**

For more information on the SNMP MIB "`cmqVoIPCallActiveRxPred107RMosConv`", see [SNMP Object Navigator](#).

In the sample output, the following can be noted:

- ML for codec G.711ulaw is 4.2346.
- MC for codec G.711ulaw is 4.2346.
- IE for codec G.711 is 0.
- R0 is 93.

The following table defines the abbreviations used in the sample output.

**Table 2: Router Output Definitions for the show call active voice stats command**

Type	Abbreviation	Definition
DSP/RF: R-Factor Statistics (G.107 MOS)	ML	R-factor MOS listening quality objective
	MC	R-factor MOS-CQE
	R1	R-factor for LQ profile1
	R2	R-factor for LQ
	IF	Effective codec impairment (IeEff)
	ID	Delay factors
	IE	Codec baseline score (Ie)
	BL	Codec baseline (Bpl)
	R0	Nominal value for R0 (default)
	VR	R-Factor version

## Troubleshooting Tips

Use the following debug commands to troubleshoot the Voice Quality Monitoring feature:

- **debug voip rtp packets**
- **debug performance monitor**
- **debug radius accounting**
- **debug aaa accounting**

# Configuration Examples for Voice Quality Monitoring

## Example: Configuring Media Statistics Globally

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# media statistics
Device(conf-voi-serv)# end
```

## Example: CDR Enabled MOS Output

At the end of a call, the MOSQe output is displayed in CDR only if the **debug radius accounting** is enabled.

The **show log | sec MOS-Con** command displays the MOS-Con value as shown below:

```
Device# show log | sec MOS-Con

*Jan 21 22:31:42.313: RADIUS: Cisco AVpair [1] 16 "MOS-Con=4.2312"
*Jan 21 22:31:42.313: RADIUS: Cisco AVpair [1] 16 "MOS-Con=4.2312"
```