Voice Quality Monitoring

The Voice Quality Monitoring (VQM) feature uses Flexible NetFlow to export 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.

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Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and 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.

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 for Voice Quality Monitoring

The **aqm-register-fnf** command must be configured before you use the **media monitoring** command to configure voice quality metrics.

Information About Voice Quality Monitoring

The VQM (Voice Quality Monitor) uses Flexible NetFlow to export voice quality metrics measured by the **media monitoring** command. To help the NetFlow collector to process the flow record, VQM also reports call-related information such as calling number, called number, call setup time, and so on. The Voice Quality Metrics enables statistics gathering on packet arrival (late/lost/early). 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 are the five metrics added to Call Detail Record (CDR) and Management Information Base (MIB) in NanoCUBE, indicating voice quality:

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

<voice-quality-out-of-order>0.0000 %</ voice-quality-out-of-order>

VQM Metrics

The following are the metrics exported by VQM:

NanoCube IOS VQM, Voice/Audio Quality Metric	Description
GwReceivedCalledNumber	The directory number portion of To URI from the Session Initiation Protocol (SIP) signaling or the extension receiving the call.
GwReceivedCallingNumber	The directory number portion of From URI from the SIP signaling or the extension originating the call.
SetupTime	The time at which monitoring began on this RTP stream.
CallDuration	The time (in milliseconds) from when monitoring began on this RTP stream until the reception of the last RTP packet on this stream.

DspRXBadPkt	The total number of packets determined by the simulated jitter buffer to be having bad protocols and that need to be dropped.
DspRXOutSeq	The total number of packets that arrive at the jitter buffer out of sequence.
DspConf CodecID	The last voice coder-decoder (CODEC) detected in this RTP stream. Note that an endpoint may change the voice CODEC mid-stream.
DspPlayDelay Cur	The current jitter buffer delay in milliseconds. In the case of an RTP stream in the call history, the last jitter buffer delay (does not apply to a fixed jitter buffer configuration).
DspPlayDelayMin	The minimum jitter buffer delay in milliseconds (does not apply to a fixed jitter buffer configuration)
DspPlayDelayMax	The maximum jitter buffer delay in milliseconds (does not apply to a fixed jitter buffer configuration).
rfc3550JitterMeanMilliseconds	The packet-to-packet delay variation (jitter) in milliseconds, as defined in RFC 3550.
ProtocolCallId	The SIP call ID read-only by the SIP proxy. This value may be unknown if using the B2BUA or if call signaling is not being monitored.
GlobalCallId	Internally generated ID identifying this call.
DspDely RT	The instantaneous round-trip delay. This may be obtained from RTCP XR or SR reports; or if no reports are available, from an average of ICMP echo or timestamp requests sent to both endpoints. If no report information is available and round-trip delay cannot be determined from ICMP (example, a firewall in the path did not allow the traffic), this statistic will be reported as unavailable.
DspDelyED	The instantaneous one-way delay, including any delay that can be introduced by the jitter buffer and codec processing.
DspRFactrR1	The listening quality R factor. Listening quality indicates the perceived quality of the transmission for a user not actively involved in the conversation, but passively listening. Listening quality does not consider delay or recency. Some users may prefer R-factor measurements to MOS scores, because MOS scales may differ based on the CODEC type and region of deployment, whereas R factor measurements are consistent across CODECs and regions.

DspRFactrR2	The conversational quality R factor. 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. For example, for a link with a large delay, participants in a conversation might frequently find themselves interrupting each other and talking over each other, since one party will be unable to perceive when the other party has started talking. Some users may prefer R factor measurements to MOS, since MOS scales may differ based on the CODEC type and region of deployment, whereas R factor measurements are consistent across CODECs and regions.
DspRFactrMosConv	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. For example, for a link with a large delay, participants in a conversation might frequently find themselves interrupting each other and talking over each other, since one party will be unable to perceive when the other party has started talking.
DspRFactrMosLisn	The listening quality MOS. Listening quality indicates the perceived quality of the transmission for a user not actively involved in the conversation, but passively listening. Listening quality does not consider delay or recency.
DspCealRatioAV	Average of Concealment Ratio reports since the start of a call.
DspConfJtrTyp	The configured jitter buffer type for this RTP stream, either adaptive or fixed. An adaptive jitter buffer dynamically varies the delay from packet reception to packet playback; a fixed jitter buffer uses the same delay for each packet. This is a jitter buffer; no packets are actually being discarded.
DspConfJtrMin	The minimum delay that will be applied to packets received when using an adaptive jitter buffer.
DspConfJtrInit	The value that represents the initial delay that will be applied to received packets when using an adaptive jitter buffer. When using a fixed jitter buffer, this represents the delay that will be applied to each packet when it is received.
DspConfJtrMax	The value that represents an upper bound on the delay that will be applied to received packets when using an adaptive jitter buffer. When using a fixed jitter buffer, this metric represents the maximum number of packets that can be inserted into the buffer. (Subsequently, inserted packets will be discarded.)
DspRXEarlPkt	The total number of packets for this RTP stream arriving early (prior to the anticipated packet arrival). Each packet is classified as either late or early with the exception of the first packet that is treated as a reference packet.

DspRXLatPkt	The total number of packets for this RTP stream arriving late (after the anticipated packet arrival). Each packet is classified as either late or early with the exception of the first packet that is treated as a reference packet.
DspPktBfrOvr	The total number of packets discarded by the jitter buffer due to jitter buffer overrun.
DspPktCealCount	The total number of packets discarded by the jitter buffer due to jitter buffer underrun.

How to Configure Voice Quality Monitoring

Configuring Voice Quality Metrics

Before You Begin

The **aqm-register-fnf** command must be configured at the global configuration mode to export the audio and video call quality metrics to flow record using Flexible NetFlow collector.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media monitoring** *max-calls*
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **media monitoring** *max-calls*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters voice service configuration mode and specified Voice over IP as the voice-encapsulation type.
Step 4	**media monitoring** *max-calls* **Example:** `Device(conf-voi-serv)# media monitoring 300`	Enables media monitoring and specifies the maximum number of calls to be monitored. **Note** You can monitor only up to 302 channels for NANOCUBE, that is, about 151 calls.
Step 5	**exit** **Example:** `Device(conf-voi-serv)# exit`	Exits voice service configuration mode and returns to global configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Device(conf-voi-serv)# dial-peer voice 5 voip`	Enters dial-peer configuration mode, defines a particular dial peer, and specifies the method of voice encapsulation as VoIP.
Step 7	**media monitoring** *max-calls* **Example:** `Device(config-dial-peer)# media monitoring 300`	Enables media monitoring for calls landing on the dial peer specified in Step 6.

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	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters voice service VoIP configuration mode.
Step 4	**media statistics** **Example:** `Device(conf-voi-serv)# media statistics`	Enables media statistics to estimate the values of packet loss, jitter, and Round Trip Time (RTT) statistics. • The statistics are displayed using the **show voice history** and **show call active voice** commands. • If the **media statistics** command is disabled, the values will be zero.
Step 5	**end** **Example:** `Device(conf-voi-serv)# end`	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 voip rtp connections**
3. **show sccp connections**
4. **show voice monitoring-channels**
5. **show call active voice**
6. **show call active voice stats**

DETAILED STEPS

Step 1 **enable**
Enables privileged EXEC mode.

Example:
`Device>` **enable**

Step 2 **show voip rtp connections**
Displays Real-Time Transport Protocol (RTP) named event packets.

Example:
`Device#` **show voip rtp connections**

```
No. CallId   dstCallId    LocalRTP   RmtRTP    LocalIP    RemoteIP   MPSS
1      37      38          16582     18236    10.1.1.2   10.1.1.7   NO
2      38      37          16524     19542    10.1.1.2   10.1.1.1   NO
3      39      40          17644     2000     10.1.1.2   10.1.1.2   NO
4      41      40          16622     2000     10.1.1.2   10.1.1.2   NO
```

Step 3 **show sccp connections**
Displays information about the connections controlled by the Skinny Client Control Protocol (SCCP) transcoding and conferencing applications.

Example:
`Device#` **show sccp connections**

```
sess_id      conn_id    stype    mode       codec     ripaddr     rport    sport
3            4          xcode    sendrecv   g711u     100.1.1.2   2000     16622
3            3          xcode    sendrecv   g711u     100.1.1.2   2000     17644
Total number of active session(s) 1, and connection(s) 2
```

Step 4 **show voice monitoring-channels**
Displays voice monitoring statistics.

Example:
`Device#` **show voice monitoring-channels**

```
max vq mon channels = 10 vq mon channels in use = 2 vq mon channels left =8
```

Step 5 **show call active voice**
Displays statistics on the CUBE if the Voice Quality Metrics feature is configured.

Example:

```
Device# show call active voice

RxPakNumber=5496
RxSignalPak=0
RxComfortNoisePak=0
RxDuration=109900
RxVoiceDuration=109920
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
RxBadProtocol=0
LevelRxPowerMean=0
ErrRxDrop=0
ErrRxControl=0
```

Step 6 **show call active voice stats**

Displays Concealment Statistics and R-Factor Statistics (G.107 MOS) on the Cisco UBE 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 statslsec MC

DSP/CS: CR=0.0527, AV=0.0502, MX=0.0527, CT=1220, TT=24270, OK=50, CS=44, SC=0, TS=50, DC=0
SP/RF: ML=3.9855, MC=0.0000, R1=79, R2=0, IF=15, ID=0, IE=0, BL=25, R0=94, VR=1.1
```

In the sample output, the following can be noted:

- The average conceal ratio (AV) is about 5 percent.
- The ratio of total conceal time and total speech time is about 5 percent (1220/24270).
- BL for codec G.711 is 25 (based on G.113).
- IE for codec G.711 is 0 (G.113).
- IE for codec G.711 is 0.
- R0 is 94 (G.107).

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

Table 1: Router Output Definitions for the show call active voice stats command

Type	Abbreviation	Definition

DSP/CS: Concealment Statistics	CR	concealRatioCurrent
	AV	ConcealRatioAverage
	MX	ConcealRatioMaximum
	CT	ConcealDuration
	TT	SpeechDuration
	OK	OkSeconds
	CS	ConcealSeconds
	SC	SevereConcealSeconds
	TS	SevereConcealThreshold
DSP/RF: R-Factor Statistics (G.107 MOS)	ML	MOSLQE
	R1	RFactorProfile1
	IF	IeEff
	BL	CodecBaselineBPL
	R0	R0Default
	VR	R-Factor version

Troubleshooting Tips

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

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

Configuration Examples for Voice Quality Monitoring

Example: Configuring Voice Quality Metrics

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# media monitoring 300
Device(conf-serv-sip)# exit
Device(config)# dial-peer voice 5 voip
Device(config-dial-peer)# media monitoring 300
```

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
```

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 2: Feature Information for Voice Quality Monitoring

Feature Name	Releases	Feature Information
Voice Quality Monitoring	15.3(3)M	The Voice Quality Monitoring (VQM) feature uses Flexible NetFlow to export 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.