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.
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:

<table>
<thead>
<tr>
<th>NanoCube IOS VQM, Voice/Audio Quality Metric</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>GwReceivedCalledNumber</td>
<td>The directory number portion of To URI from the Session Initiation Protocol (SIP) signaling or the extension receiving the call.</td>
</tr>
<tr>
<td>GwReceivedCallingNumber</td>
<td>The directory number portion of From URI from the SIP signaling or the extension originating the call.</td>
</tr>
<tr>
<td>SetupTime</td>
<td>The time at which monitoring began on this RTP stream.</td>
</tr>
<tr>
<td>CallDuration</td>
<td>The time (in milliseconds) from when monitoring began on this RTP stream until the reception of the last RTP packet on this stream.</td>
</tr>
<tr>
<td>Metric</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>DspRXBadPkt</td>
<td>The total number of packets determined by the simulated jitter buffer to be having bad protocols and that need to be dropped.</td>
</tr>
<tr>
<td>DspRXOutSeq</td>
<td>The total number of packets that arrive at the jitter buffer out of sequence.</td>
</tr>
<tr>
<td>DspConf CodecID</td>
<td>The last voice coder-decoder (CODEC) detected in this RTP stream. Note that an endpoint may change the voice CODEC mid-stream.</td>
</tr>
<tr>
<td>DspPlayDelay Cur</td>
<td>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).</td>
</tr>
<tr>
<td>DspPlayDelayMin</td>
<td>The minimum jitter buffer delay in milliseconds (does not apply to a fixed jitter buffer configuration).</td>
</tr>
<tr>
<td>DspPlayDelayMax</td>
<td>The maximum jitter buffer delay in milliseconds (does not apply to a fixed jitter buffer configuration).</td>
</tr>
<tr>
<td>rfc3550JitterMeanMilliseonds</td>
<td>The packet-to-packet delay variation (jitter) in milliseconds, as defined in RFC 3550.</td>
</tr>
<tr>
<td>ProtocolCallId</td>
<td>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.</td>
</tr>
<tr>
<td>GlobalCallId</td>
<td>Internally generated ID identifying this call.</td>
</tr>
<tr>
<td>DspDely RT</td>
<td>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.</td>
</tr>
<tr>
<td>DspDelyED</td>
<td>The instantaneous one-way delay, including any delay that can be introduced by the jitter buffer and codec processing.</td>
</tr>
<tr>
<td>DspRFactrR1</td>
<td>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.</td>
</tr>
<tr>
<td>Metric</td>
<td>Description</td>
</tr>
<tr>
<td>-------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>DspRFactrR2</td>
<td>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.</td>
</tr>
<tr>
<td>DspRFactrMosConv</td>
<td>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.</td>
</tr>
<tr>
<td>DspRFactrMosLisn</td>
<td>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.</td>
</tr>
<tr>
<td>DspCealRatioAV</td>
<td>Average of Concealment Ratio reports since the start of a call.</td>
</tr>
<tr>
<td>DspConfJtrTyp</td>
<td>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.</td>
</tr>
<tr>
<td>DspConfJtrMin</td>
<td>The minimum delay that will be applied to packets received when using an adaptive jitter buffer.</td>
</tr>
<tr>
<td>DspConfJtrInit</td>
<td>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.</td>
</tr>
<tr>
<td>DspConfJtrMax</td>
<td>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.)</td>
</tr>
<tr>
<td>DspRXEarlPkt</td>
<td>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.</td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td><code>Device&gt; enable</code></td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2</td>
<td><code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td><code>Device# configure terminal</code></td>
<td></td>
</tr>
</tbody>
</table>
### Purpose

**Command or Action**

**Step 3**

voice service voip

*Example:*

Device(config)# voice service voip

**Purpose**

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

**Purpose**

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

**Purpose**

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

**Purpose**

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

**Purpose**

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> media statistics</td>
<td>Enables media statistics to estimate the values of packet loss, jitter, and Round Trip Time (RTT) statistics.</td>
</tr>
<tr>
<td>Example:</td>
<td>• The statistics are displayed using the <strong>show voice history</strong> and <strong>show call active voice</strong> commands.</td>
</tr>
<tr>
<td>Device(conf-voi-serv)# media statistics</td>
<td>• If the media statistics command is disabled, the values will be zero.</td>
</tr>
<tr>
<td><strong>Step 5</strong> end</td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(conf-voi-serv)# end</td>
<td></td>
</tr>
</tbody>
</table>

### 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  
<table>
<thead>
<tr>
<th>No.</th>
<th>CallId</th>
<th>dstCallId</th>
<th>LocalRTP</th>
<th>RmtRTP</th>
<th>LocalIP</th>
<th>RemoteIP</th>
<th>MFSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>37</td>
<td>38</td>
<td>16582</td>
<td>18236</td>
<td>10.1.1.2</td>
<td>10.1.1.7</td>
<td>NO</td>
</tr>
<tr>
<td>2</td>
<td>38</td>
<td>37</td>
<td>16524</td>
<td>19542</td>
<td>10.1.1.2</td>
<td>10.1.1.1</td>
<td>NO</td>
</tr>
<tr>
<td>3</td>
<td>39</td>
<td>40</td>
<td>17644</td>
<td>2000</td>
<td>10.1.1.2</td>
<td>10.1.1.2</td>
<td>NO</td>
</tr>
<tr>
<td>4</td>
<td>41</td>
<td>40</td>
<td>16622</td>
<td>2000</td>
<td>10.1.1.2</td>
<td>10.1.1.2</td>
<td>NO</td>
</tr>
</tbody>
</table>

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  
<table>
<thead>
<tr>
<th>sess_id</th>
<th>conn_id</th>
<th>stype</th>
<th>mode</th>
<th>codec</th>
<th>ripaddr</th>
<th>rport</th>
<th>sport</th>
</tr>
</thead>
<tbody>
<tr>
<td>3</td>
<td>4</td>
<td>xcode</td>
<td>sendrecv</td>
<td>g711u</td>
<td>100.1.1.2</td>
<td>2000</td>
<td>16622</td>
</tr>
<tr>
<td>3</td>
<td>3</td>
<td>xcode</td>
<td>sendrecv</td>
<td>g711u</td>
<td>100.1.1.2</td>
<td>2000</td>
<td>17644</td>
</tr>
</tbody>
</table>

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, SC=44, 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

<table>
<thead>
<tr>
<th>Type</th>
<th>Abbreviation</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
DSP/CS: Concealment Statistics

<table>
<thead>
<tr>
<th></th>
<th>Conceal Ratio Current CR</th>
<th>AV</th>
<th>Conceal Ratio Average AV</th>
<th>MAX</th>
<th>Conceal Ratio Maximum MX</th>
</tr>
</thead>
<tbody>
<tr>
<td>CT</td>
<td>Conceal Duration CT</td>
<td>TT</td>
<td>Speech Duration TT</td>
<td>OK</td>
<td>Ok Seconds OK</td>
</tr>
<tr>
<td>CS</td>
<td>Conceal Seconds CS</td>
<td>SC</td>
<td>Severe Conceal Seconds SC</td>
<td>TS</td>
<td>Severe Conceal Threshold TS</td>
</tr>
</tbody>
</table>

DSP/RF: R-Factor Statistics (G.107 MOS)

<table>
<thead>
<tr>
<th></th>
<th>MOSLQE</th>
<th>ML</th>
<th>R-Factor Profile 1 R1</th>
<th>IF</th>
<th>IeEff</th>
</tr>
</thead>
<tbody>
<tr>
<td>BL</td>
<td>CodecBaselineBPL</td>
<td>R0</td>
<td>R0Default</td>
<td>VR</td>
<td>R-Factor version</td>
</tr>
</tbody>
</table>

**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

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

Example: Configuring Media Statistics Globally

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

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Quality Monitoring</td>
<td>15.3(3)M</td>
<td>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.</td>
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</tbody>
</table>