

## Configuration Guide

# Configuring Voice Quality Monitoring in AOS

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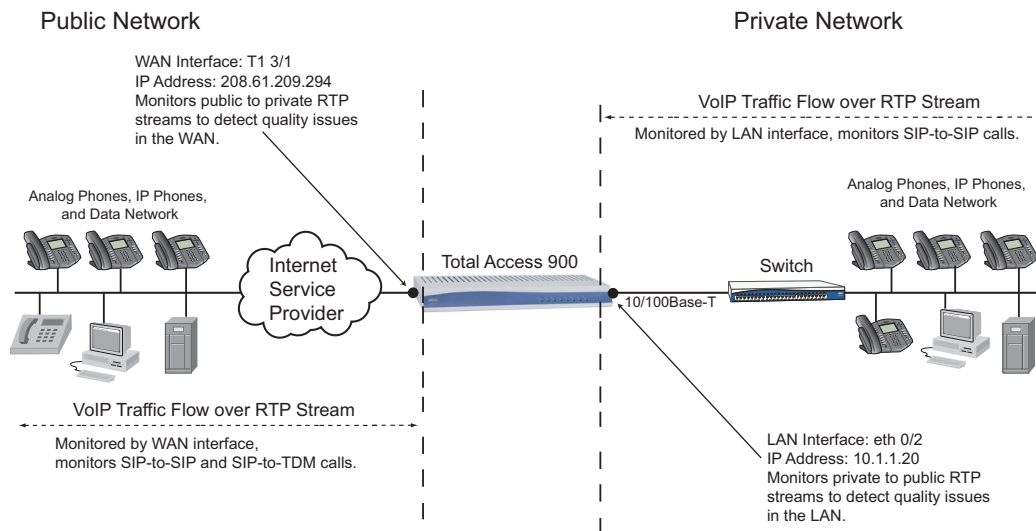
This configuration guide describes the configuration and use of the voice quality monitoring (VQM) feature in ADTRAN Operating System (AOS) voice and data products. An overview of the feature, including its various configurable components, detailed Web-based graphical user interface (GUI) and command line interface (CLI) configuration instructions, and instructions on viewing and using VQM statistics provide step-by-step assistance for implementing VQM on your Voice over Internet Protocol (VoIP) network. A basic understanding of VoIP, VoIP quality of service (QoS), and Session Initiation Protocol (SIP) are prerequisites to fully understanding the information covered in this document.

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## Introduction to VQM

VQM is an AOS feature that gathers real time passive VoIP quality measurements from Realtime Transport Protocol (RTP) voice streams transmitted through either an AOS voice or data device. VQM gathers quality statistics on SIP-signaled RTP VoIP calls passed through network interfaces on the AOS unit, which allows the voice quality of calls on both the public and private sides of the network to be monitored. Monitoring the RTP streams through the interface provides a method for determining if quality issues, such as packet loss, jitter, and echo, are present in RTP VoIP calls and if these issues are being detected on the private side local area network (LAN), or the public side wide area network (WAN). The statistics gathered reveal if a quality issue is on your side of the connection or on the service provider side of the connection. The following illustration displays the place of VQM in a typical network setting:



**Figure 1. VQM in a Typical Network Configuration**



*SIP-to-analog calls can only be monitored on the WAN interface, and can only be monitored in AOS voice products. Refer to [Hardware and Software Requirements and Limitations](#) on page 9 for more information.*

VQM is a complex feature, with many configurable components, that uses the RTP stream statistics gathered on WAN and LAN interfaces to create reports for network management. The statistics measured by VQM include packet loss, delay, jitter, and discarded and out-of-order packets. In addition, VQM monitors the quality mean opinion score (MOS) of transmitted packets, recording perceived, listening, and

conversation quality statistics. These statistics are gathered for both real-time reports and historical reports, and can be viewed using the GUI, the CLI, or sent to a server using the VQM reporter. The following sections provide more detail about the statistics gathered by VQM, the various components of VQM, and the necessary configuration steps to use VQM.

A small icon consisting of the word "NOTE" in a bold, sans-serif font, with a diagonal line striking through it from the top-left to the bottom-right.

*This document does not cover the configuration or use of the VQM reporter, but focuses instead on viewing VQM statistics using the GUI and CLI. For more information about the VQM reporter, refer to the **VQM Reporter** section of the **AOS Command Reference Guide** available online at <http://kb.adtran.com> (article number 2219). If you want to report VQM statistics to a n-Command<sup>®</sup> managed service provider (MSP) server, refer to the quick configuration guide **Configuring VQM Reporter for AOS and n-Command MSP**, also available online at <http://kb.adtran.com> (article number 3099).*

## Configurable Components of VQM

VQM operates in the AOS unit using a number of user-configurable components. These components include a simulated jitter buffer, call quality thresholds, a call history store, and monitoring filters. A thorough understanding of these components and how they contribute to VQM functionality is important to understanding the necessary configuration steps for VQM. Details about each of these components are provided in the following sections.

### The VQM Simulated Jitter Buffer

Delay, the time it takes information to travel from source to destination, can vary when VoIP networks are congested, resulting in longer delays when there is more congestion and shorter delays when there is less congestion. When the space between the packets differs in the RTP stream, jitter occurs and affects audio quality. IP phones, private branch exchanges (PBXs), and integrated access devices (IADs) incorporate jitter buffers to counteract jitter by adding a small amount of delay between receiving a VoIP packet and playing it back. The VQM jitter buffer is a simulated buffer that captures quality data of RTP streams as if it was a real jitter buffer without affecting the packet playback. In other words, the VQM simulated jitter buffer passively collects data about the RTP stream without influencing packet transmission. The simulated jitter buffer is used to monitor the RTP stream and collect data about lost and discarded packets, and the MOS of the perceived, listening, and conversation qualities of packets.

### VQM RTP Stream Monitoring

VQM monitors RTP streams for lost packets by checking the timestamp of RTP packets within the stream. Lost packets are dropped RTP packets that were never received for processing by the voice endpoint. When an inbound RTP packet is sent to VQM's simulated jitter buffer, the timestamp in the RTP header is checked. If the next packet's timestamp shows a variance in the packetization period between the packets, the simulated jitter buffer considers this a sign of lost packets and a lost packet is logged. However, if the packet is delayed and arrives beyond the window of the jitter buffer, an early or late arrival is logged. Early and late packets are counted as discarded packets, and are subtracted from the total number of lost packets reported by the simulated jitter buffer.

The simulated jitter buffer also monitors the RTP streams for MOS qualities, which are tracked and tabulated to determine a score for each of the MOS qualities (perceived, listening, and conversation quality). MOSs are determined by tests on the network specified by ITU-T recommendation P.800 and are commonly on a scale of 1 (bad) to 5 (excellent). However, ADTRAN's MOS scale range is 0 (poor) to 4.4 (excellent). ADTRAN's MOS scale is for typical coder-decoders (CODECs) and uses the quality ranges described in [Table 1](#).

**Table 1. ADTRAN MOS Voice Quality Ranges**

Quality	MOS Range
Excellent	4.40 to 4.00
Good	3.99 to 3.60
Fair	3.59 to 2.60
Poor	2.59 to 0.00



*Wideband CODECs can scale differently and a call with an MOS of 3.8 may have excellent quality. Therefore, the R factor measurements should also be used with wideband CODECs.*

The statistics gathered by VQM are dependent upon the quality thresholds configured in VQM, and the configuration of the VQM call history. Quality statistics, such as MOS, lost packets, and jitter are collected and reported in a searchable output, on both the GUI and the CLI. In addition, MOS gathered by VQM are reported differently based on your location. The scoring adjustments are based on the region from which you are monitoring, and can be set to the standard method or to the Japanese method.

## VQM Thresholds

Thresholds are set in the VQM simulated jitter buffer to determine when events are recorded for low-quality calls and when calls are added to the call history. When a call is completed and the policy sessions for the RTP flows are deleted, the call is compared to a set of thresholds. If the call quality is lower than the specified threshold, then an event is sent to the event manager, which then sends the event to any devices listening for events, such as the event-history log file monitor, Simple Mail Transfer Protocol (SMTP) email monitors, CLI sessions, and SYSLOG. Thresholds are configured for listening quality, perceived quality, conversation quality MOSs, number of out-of-order and lost packets, and the amount of jitter in the monitored call. Thresholds are configured to trigger specific event types (info, notice, warning, or error) depending on the severity of the quality statistic issue monitored by VQM.

## VQM Call History

Statistics gathered by VQM about network interfaces, voice endpoints, and active calls are stored in the VQM call history. The call history holds 100 past calls by default, but its size is user configurable. The call history can be limited to a maximum number of stored statistics for completed calls, and the stored statistics can be filtered so that only statistics crossing thresholds for MOSs, lost and out-of-order packets, and jitter are stored. Call history statistics can also be viewed by specific calls, based on known information about the call, such as any substring of the To or From URI and the Call-ID information, and these calls can be sorted based on MOS, lost and out-of-order packets, and jitter.

## VQM Statistic Filters

The statistics gathered and stored by VQM can be filtered based on a sample rate, a specific extension, or by using an access control list (ACL). These filters can be configured at the same time as VQM is configured (sample rate or extension monitoring), or they can be configured prior to VQM and applied through the VQM configuration (ACLs). This guide does not go into detail about how to configure ACLs, but rather focuses on how to apply ACLs to VQM statistic gathering. For more information about ACL configuration, refer to the configuration guide *Configuring IP ACLs in AOS* (article number 3087) available online at <http://kb.adtran.com>.

## VQM Configuration Overview

Because VQM uses so many components and can be used to monitor many different quality characteristics of RTP streams, the configuration of VQM can be complicated. VQM can use a large amount of the AOS device's CPU, and careful consideration should be given to each part of the VQM configuration in order to maintain optimum device performance. The following sections give an overview of each part of the VQM configuration. The overall configuration parameters will be the same whether you are configuring VQM using the GUI or the CLI.

### Step One: Configure the Firewall and SIP Settings

For VQM to function properly, the firewall on the AOS device must be enabled. When using VQM with Media Gateway Control Protocol (MGCP), the SIP back-to-back user agent (B2BUA), or SIP proxy, VQM only works on the default virtual router and forwarding (VRF) interface because MGCP, the SIP B2BUA, and SIP proxy only use the default VRF. When using the firewall SIP application layer gateway (ALG), VQM works over multiple VRFs, but in this configuration multiple IP addresses cannot be reused.

When configuring VQM, you must keep in mind your firewall and SIP settings. If non-default access control policies (ACPs) are used with **allow** policy entries, you must make sure the SIP and RTP traffic do not match stateless **allow** policy entries. If you have to enable the firewall specifically for VQM functionality, performance of the AOS device can be diminished.

This guide does not go into detail about the necessary firewall and SIP configurations for your VoIP network, but rather focuses on the necessary settings for VQM functionality. For more information on these subjects, refer to the configuration guides *Configuring the IP Firewall in AOS* (article number 1543) and *Configuring the SIP Proxy in AOS* (article number 2183). These guides are available online at <http://kb.adtran.com>.

### Step Two: Configure the VQM Monitoring Settings

The VQM monitoring settings are broken down into three major categories: basic VQM settings, simulated jitter buffer settings, and RTP packet round-trip delay settings. Configuring these settings are detailed in the following sections.

#### *Configuring the Basic VQM Settings*

The basic settings for VQM include enabling the VQM feature globally on the AOS device, and specifying the type of traffic that will be monitored, the quality scoring method, the call history size, and the monitoring sampling rate. By default, VQM is disabled globally on the AOS device.

VQM can monitor both SIP and MGCP or User Datagram Protocol (UDP) traffic. When configuring VQM using the CLI, you can specify whether SIP and MGCP or UDP is monitored. When configuring VQM using the GUI, you can specify whether to monitor **any** RTP or SIP and MGCP RTP traffic. Specifying **any** RTP monitoring forces the AOS unit to inspect every UDP packet to determine if it is an RTP packet, placing significant load on the AOS unit.



*Monitoring **any** RTP traffic (UDP traffic) should only be done if IP phones are being used and they do not pass through the SIP ALG, SIP proxy, or the SIP B2BUA.*

You can specify MOS using either the standard method or the Japanese method. In the GUI, you specify the locale as standard or Japan, and in the CLI, you specify that VQM is enabled with a scoring adjustment for Japan (if needed).

Another basic VQM configuration is specifying the call history size, which defines the maximum number of RTP streams held in the call history. By default, the call history is set to hold **100** calls, but it can hold between **0** and **2000** calls. The size of the call history directly affects the performance of the AOS unit.



*If the call history size is set to a large value, the AOS unit could run out of memory.*

The last part of basic VQM configuration is specifying the monitor sampling size. Sampling provides a method of filtering VQM statistics, and allows VQM to monitor only a fraction of the total RTP packets out of a stream, thus limiting the pressure on the AOS unit's CPU resources. A sampling rate of **1** indicates all calls are monitored, and a sampling rate of **100** indicates that only 1 percent of calls are monitored (1 out of 100 calls). By default, the VQM sampling rate is set to **1**.

### *Configuring the Simulated Jitter Buffer Settings*

The VQM simulated jitter buffer must be configured to specify the buffer type and the early or late threshold for determining lost or discarded packets. Buffer types include fixed or adaptive. An adaptive simulated buffer automatically increases or decreases playout delay to adjust for varying jitter. When configuring the simulated jitter buffer as adaptive, you must also specify the smallest allowed delay (adaptive minimum), the beginning delay detected by the buffer (adaptive nominal), and the maximum allowed delay (adaptive maximum). The simulated jitter buffer can also be configured as a fixed buffer that uses a fixed playout delay for all packets. By default, the VQM simulated jitter buffer is set to be **adaptive** with a nominal setting of **50** ms. If the simulated jitter buffer is configured as adaptive, and the buffer continually detects packets outside of the jitter window, the buffer increases the window size in **5** ms increments.

Packets that reach the simulated jitter buffer outside the configured jitter window are termed either early or late. The early threshold can be set between **0** and **1000** ms, and packets that arrive earlier than the specified threshold are considered discarded packets. The late threshold can be set between **0** and **1000** ms, and packets that arrive later than the threshold are also considered discarded packets.

### Configuring Round-Trip Delay Settings

VQM gathers statistics on VoIP quality that can be used with RTP Control Protocol extended reports (RTCP XRs). These statistics are reflected in the per-call statistics gathered by VQM, and help with gathering round-trip delay, end system delay, and analog metrics for MOS calculations. When RTCP XR reports are not available, Internet Control Message Protocol (ICMP) ping or timestamps can be used to gather a best guess round-trip delay metric for the voice endpoints. The metric is based on ICMP requests sent to both endpoints, which are added to the round-trip delay for each endpoint. If RTCP XRs are available, they will supersede any data gathered from ICMP pings, and it is not necessary to configure VQM round-trip delay settings.

If no RTCP XRs are available, you can enable the round-trip delay calculations for VQM to get an estimate of the delay between the voice endpoints. By default, VQM ICMP ping capability is enabled. This means that by default, VQM sends out five ping requests, spaced five seconds apart, immediately after a call is started. The data received from the ping responses is averaged to determine the round-trip delay. If the ping is not received by the voice endpoint, then the round-trip delay statistics are unavailable.

Round-trip delay calculations can be based on ICMP ping or timestamp echo requests. Using timestamp echo requests provides more accurate results than ping echo requests; however, when using timestamp requests both the endpoints and the local AOS units must have synchronized time and date information. For either method to work, any firewalls between the endpoints must be configured to allow ICMP traffic to pass.

### Step Three: Specify the VQM Thresholds

VQM requires thresholds to determine when calls are logged into the call history (the Past Calls record) and when events are generated about calls. Past call thresholds define the lowest call quality permissible for listening, conversation, and perceived quality MOS, the highest number of lost and out-of-order packets permissible, and the amount of jitter that is permissible. Any calls that have quality equal to or lower than the defined thresholds are recorded in the past calls record. By default, all past call MOS thresholds are set to **4.40**. See [Table 1 on page 4](#) for a listing of MOS quality scored in AOS products.

Thresholds are also used to determine when call quality events are generated. Event logging thresholds include listening and perceived quality MOS, the number of lost and out-of-order packets, and the amount of jitter detected. Event thresholds determine if an information, notice, warning, or error message is generated for the call. When call quality is equal to or less than the configured event threshold, the user-specified message type is generated.

### Step Four: Set the VQM Filters (optional)

The statistics VQM gathers can be filtered for more specific result. Filtering is accomplished using the VQM sample rate (configured in the basic VQM settings), applied ACLs, or user extensions. Filters should be configured to minimize the effect of VQM on the AOS unit's resources.

An ACL can be used to limit the calls measured by VQM. When an ACL is applied, only calls where one of the RTP streams matches the ACL are measured. By default, no ACL is configured or applied. ACLs must be configured before attempting to apply them to VQM. For more information about configuring ACLs, refer to the configuration guide *Configuring IP ACLs in AOS* available online at <http://kb.adtran.com> (article number 3087).

The statistics VQM gathers can also be limited by only measuring the calls made by specific users. When calls are placed by users whose To or From header matches the specified header, the call is monitored by VQM. Multiple headers can be specified. By default, no headers are specified.

### Step Five: Enable VQM on the Network Interface (optional)

Once VQM has been configured, it must be enabled on the network interface that you want to monitor. VQM must be enabled on both the inbound and outbound network interface; however, only the inbound interface is monitored. Each WAN or LAN interface that is assigned an IP address, or is configured with a method for getting an IP address assigned, should have VQM enabled. By default, VQM is enabled on all interfaces. However, if you have changed the configuration of the interfaces, or disabled VQM on some interfaces, you must make sure it is enabled on the interfaces you want to monitor. This can be done using either the GUI or the CLI, however, not all platforms support GUI configuration of VQM on particular interfaces.



*Enabling VQM will degrade performance. In addition, because RTP packets are sent through an ALG for processing, RTP streams will bypass RapidRoute.*

### Viewing VQM Statistics

VQM statistics are gathered into reports that can be organized and searched based on your particular interest. Results can focus on poor quality calls detected on interfaces, user extensions, source IP addresses, active calls, or past calls. In addition, results for particular interfaces, extensions, IP addresses, and calls can be displayed by MOS statistics, call statistics, jitter statistics, and delay statistics. These results can be viewed in the GUI, CLI, and exported in a comma separated value (CSV) file.



## Hardware and Software Requirements and Limitations

VQM is supported by various types of AOS platforms running the enhanced feature pack firmware. Both AOS data and voice products support VQM. AOS data products began to support VQM in AOS firmware 17.1.1, and AOS voice products began to support VQM in AOS firmware A1. For a complete listing of AOS products that support VQM, refer to the *Product Feature Matrix* (article number 2272) available online at <http://kb.adtran.com>.

### RTP Streams and VQM

VQM monitors RTP streams passing through the AOS unit. Each AOS platform can monitor differing numbers of RTP streams simultaneously. *Table 2* outlines the number of supported RTP streams by platform.

**Table 2. Supported Number of RTP Streams by AOS Platform**

AOS Platform	Number of Supported Streams
NetVanta 1335	25
NetVanta 3120	10
NetVanta 3130	10
NetVanta 3200 (3rd Generation)	10
NetVanta 3205 (3rd Generation)	10
NetVanta 3430 (1st and 2nd Generations)	50
NetVanta 3448	50
NetVanta 3450	50
NetVanta 3458	50
NetVanta 4305 (2nd Generation)	75
NetVanta 4430	75
NetVanta 5305	100
NetVanta 6310/6330	unlimited (large volume affects performance)
NetVanta 6355	unlimited (large volume affects performance)
NetVanta 7100	unlimited (large volume affects performance)
Total Access 900 (1st and 2nd Generations)	44
Total Access 900e (1st and 2nd Generations)	44

## General VQM Limitations and Interactions

The firewall must be enabled when using VQM. If an **allow** statement is used in IP ACP, it cannot be stateless. For more information regarding stateful firewall configuration, refer to the quick configuration guide *Configuring Stateful Firewall in AOS* available online at <http://kb.adtran.com> (article number 2150).



*Enabling the firewall only for the purpose of using VQM in your application can affect your network performance.*

By default, VQM is globally disabled on the AOS product, but it is enabled on each interface. Before VQM can gather statistics from an interface, it must be enabled globally.

The MOS statistics can be scored differently depending on your location. Japan uses a different scoring adjustment than other parts of the world. If you need to change your scoring adjustment, you will need to disable VQM (globally) on the AOS unit and then re-enable it with the proper scoring adjustment. If you have to disable VQM, all active call statistics, new call statistics, and interface statistics are lost. However, past call statistics and voice endpoint statistics are retained and can only be removed by clearing the VQM statistics.

In order for VQM subsystems to gather statistics on RTP streams, these streams must be routed through an AOS product running VQM. Call routing and physical interfaces can be used to filter which traffic is monitored by using separate interfaces to route specific calls and then gathering statistics on those specific interfaces and disabling VQM on other interfaces. This could help to reduce network performance issues caused by VQM consuming CPU resources.

VQM will not work on switched RTP traffic, RTP traffic over a virtual private network (VPN) or a generic routing encapsulation (GRE) tunnel that is **not** terminated by the AOS device, layer 3 switched interfaces, bridged traffic, or secure RTP packets.

VQM disables RapidRoute for RTP traffic. Since the RTP stream bypasses RapidRoute and is sent through an ALG for processing, the performance load is greater and can degrade the AOS product's performance.

VQM will not gather information on RTP traffic received on unsupported CODECs. VQM works with the list of supported CODECs outlined in [Appendix B: ADTRAN-Supported CODECs on page 73](#).

## Configuring VQM Using the GUI

VQM can be configured using the GUI. Refer to the [VQM Configuration Overview on page 5](#) for an overview of the steps involved in VQM configuration. To begin configuring VQM using the GUI, follow the steps outlined in this section.

### Accessing the GUI

To begin configuring VQM using the GUI, connect to the GUI following these steps:

1. Open a new Web page in your Internet browser.
2. Type your AOS product's IP address in the Internet browser's address field in the following form:  
**http://<ip address>**. For example:  
**http://208.61.209.1**
3. At the prompt, enter your user name and password and select **OK**.



*The default user name is **admin** and the default password is **password**.*

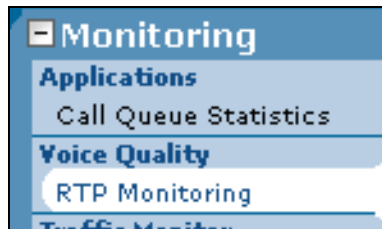
### Configure the Firewall and SIP Settings

To configure the firewall for VQM, make sure that the firewall is enabled. If non-default ACPs are used with **allow** policy entries, you must make sure the SIP and RTP traffic do not match stateless **allow** policy entries. For more information about configuring the AOS firewall and SIP settings, refer to the configuration guides *Configuring the SIP Proxy in AOS* (article number 2183) and *Configuring the IP Firewall in AOS* (article number 1543) available online at <http://kb.adtran.com>.

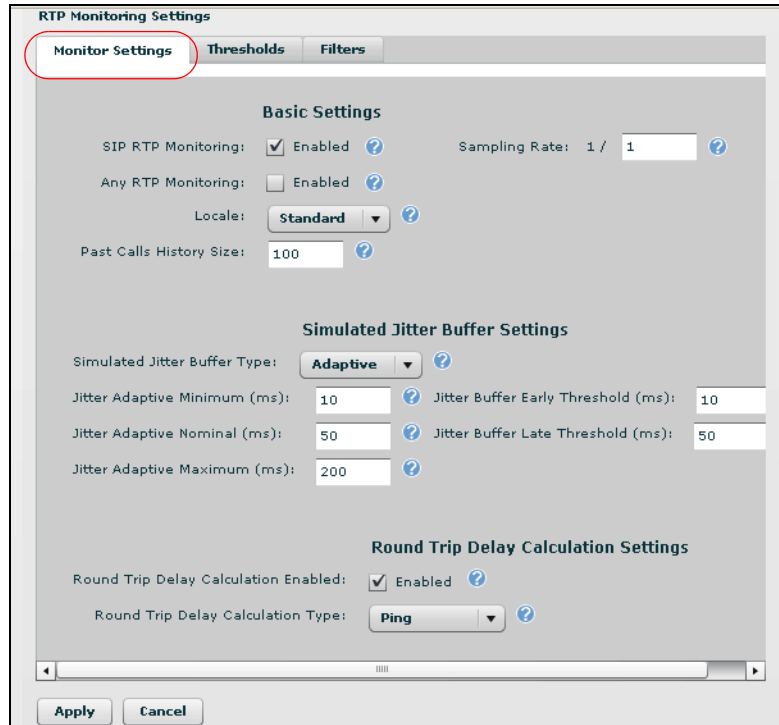
## Enabling VQM and Configuring the VQM Monitoring Settings

To enable VQM and configure the monitoring settings, follow these steps:

1. Navigate to **Monitoring** > **Voice Quality** > **RTP Monitoring** in the menu on the left of the GUI.



2. Scroll down in the RTP monitoring menu to the **RTP Monitoring Settings** menu and select the **Monitor Settings** tab. Here you will configure the VQM basic settings, simulated jitter buffer settings, and the round-trip delay calculation settings.

A screenshot of the 'RTP Monitoring Settings' dialog box. The 'Monitor Settings' tab is selected and circled in red. The dialog is divided into three sections: 'Basic Settings', 'Simulated Jitter Buffer Settings', and 'Round Trip Delay Calculation Settings'.  
**Basic Settings:**  
SIP RTP Monitoring:  Enabled  
Any RTP Monitoring:  Enabled  
Locale: Standard  
Past Calls History Size: 100  
Sampling Rate: 1 / 1  
**Simulated Jitter Buffer Settings:**  
Simulated Jitter Buffer Type: Adaptive  
Jitter Adaptive Minimum (ms): 10  
Jitter Adaptive Nominal (ms): 50  
Jitter Adaptive Maximum (ms): 200  
Jitter Buffer Early Threshold (ms): 10  
Jitter Buffer Late Threshold (ms): 50  
**Round Trip Delay Calculation Settings:**  
Round Trip Delay Calculation Enabled:  Enabled  
Round Trip Delay Calculation Type: Ping  
Buttons: Apply, Cancel

3. In the **Basic Settings** menu, enable VQM by selecting the check box next to **SIP RTP Monitoring**, **Any RTP Monitoring**, or both. **SIP RTP Monitoring** monitors streams detected by the SIP ALG, SIP proxy, SIP B2BUA, or MGCP. **Any RTP Monitoring** monitors all RTP streams, except those that are affected by local SIP and MGCP services or streams that are being handled by the SIP ALG or SIP proxy.

The screenshot shows the 'Basic Settings' configuration page. It includes the following fields and controls:

- SIP RTP Monitoring:** A checked checkbox followed by the text 'Enabled' and a help icon.
- Any RTP Monitoring:** An unchecked checkbox followed by the text 'Enabled' and a help icon.
- Locale:** A dropdown menu currently set to 'Standard' with a help icon.
- Past Calls History Size:** A text input field containing the number '100' and a help icon.
- Sampling Rate:** A text input field showing '1 / 1' with a help icon.



*Enabling **Any RTP Monitoring** forces the AOS unit to inspect every UDP packet to determine if it is an RTP packet, placing a significant load on the AOS unit. **Any RTP Monitoring** should only be enabled if IP phones are being used and they do not pass through the SIP ALG, SIP proxy, or SIP B2BUA.*

Specify the MOS scoring adjustment needed by selecting either **Standard** or **Japan** from the **Locale** drop-down menu.

Specify the size of the call history by entering the number of calls that will be stored in the **Past Calls History Size** field. By default, the call history is set to hold **100** calls. The valid range is **0** to **2000** calls.



*Setting the **Past Calls History Size** to a large value can result in the AOS unit running out of memory.*

Next, specify the VQM sampling rate by entering a value in the **Sampling Rate** field. This value specifies that 1 out of the entered number of calls are monitored by VQM. For example, entering **1** indicates that all calls are monitored (1 out of 1 calls are monitored), and entering **100** indicates that only 1 percent of calls are monitored (1 out of every 100 calls are monitored). Using a sampling rate can help to reduce the amount of resources used by the VQM feature and allow a larger part of the incoming calls to proceed as normal, including through RapidRoute (if enabled). Valid range is **1** to **100**. By default, the sampling rate is **1**.

4. After you have enabled VQM, and specified the MOS scoring adjustment locale, call history size, and sampling rate, begin configuring the **Simulated Jitter Buffer Settings**.

The screenshot shows the 'Simulated Jitter Buffer Settings' window with the 'Adaptive' type selected. The settings are as follows:

Setting	Value
Simulated Jitter Buffer Type	Adaptive
Jitter Adaptive Minimum (ms)	10
Jitter Adaptive Nominal (ms)	50
Jitter Adaptive Maximum (ms)	200
Jitter Buffer Early Threshold (ms)	10
Jitter Buffer Late Threshold (ms)	50

From this menu, specify if the simulated jitter buffer is **Adaptive** or **Fixed** by selecting the appropriate option from the **Simulated Jitter Buffer Type** drop-down menu. Selecting **Adaptive** indicates the simulated jitter buffer automatically increases or decreases playout delay to adjust for varying jitter. If you select **Adaptive**, you will also need to specify the adaptive minimum, nominal, and maximum settings. By default, the simulated jitter buffer type is **Adaptive**.

The **Jitter Adaptive Minimum (ms)** value sets the smallest allowed delay for the adaptive simulated jitter buffer. The range is **10** to **240** ms. By default, the adaptive minimum is set to **10** ms.

The **Jitter Adaptive Nominal (ms)** value sets the beginning delay of the adaptive simulated jitter buffer. The range is **40** to **320** ms. By default, the adaptive nominal value is set to **50** ms.

The **Jitter Adaptive Maximum (ms)** value sets the maximum delay of the adaptive simulated jitter buffer. Any RTP packets received after this delay value are not analyzed. The range is **40** to **320** ms. By default, the adaptive maximum is set to **200** ms.

Selecting **Fixed** from the **Simulated Jitter Buffer Type** drop-down menu indicates that the simulated jitter buffer uses a fixed playout delay for all packets. If you select **Fixed**, you will also need to specify the fixed nominal setting and the fixed jitter buffer size.

The screenshot shows the 'Simulated Jitter Buffer Settings' window with the 'Fixed' type selected. The settings are as follows:

Setting	Value
Simulated Jitter Buffer Type	Fixed
Jitter Fixed Nominal (ms)	
Jitter Fixed Size (packets)	
Jitter Buffer Early Threshold (ms)	
Jitter Buffer Late Threshold (ms)	

The **Jitter Fixed Nominal (ms)** value sets the beginning delay of the fixed simulated jitter buffer. The valid delay range is **4** to **250** ms. There is no fixed nominal value set by default.

The **Jitter Fixed Size (packets)** value specifies the size of the simulated jitter buffer by the number of packets it can hold (if it was a non-emulated jitter buffer). Valid range is **10** to **500** packets. There is no simulated jitter buffer size set by default.



*VQM uses a simulated jitter buffer for quality measurements. For the most accurate results, select settings that are consistent with the jitter buffer settings on the VoIP endpoints in your network. In addition, because the VQM jitter buffer is a simulated buffer, it does not affect the playback of RTP streams.*

### Simulated Jitter Buffer Settings

Simulated Jitter Buffer Type: Adaptive ?

Jitter Adaptive Minimum (ms):  ? Jitter Buffer Early Threshold (ms):

Jitter Adaptive Nominal (ms):  ? Jitter Buffer Late Threshold (ms):

Jitter Adaptive Maximum (ms):  ?

Next, specify the early and late thresholds for the simulated jitter buffer. These settings will determine when packets are marked as discarded because they arrive either earlier or later than the simulated jitter buffer window.

Specify the simulated jitter buffer early threshold by entering the appropriate value in the **Jitter Buffer Early Threshold (ms)** field. Range is **0** to **1000** ms. By default, the early threshold is **10** ms.

Specify the simulated jitter buffer late threshold by entering the appropriate value in the **Jitter Buffer Late Threshold (ms)** field. Range is **0** to **1000** ms. By default, the late threshold is **50** ms.

5. After you have configured the simulated jitter buffer type and early or late thresholds, configure the **Round Trip Delay Calculation Settings**. These settings specify how the round-trip delay for RTP packets is calculated. These calculations appear in the VQM statistics; however, if you also have the ability to generate RTCP XR reports, the data in the received RTCP XR report will supersede the round-trip delay settings. RTCP is monitored by VQM automatically, and requires no additional configuration.

### Round Trip Delay Calculation Settings

Round Trip Delay Calculation Enabled:  Enabled ?

Round Trip Delay Calculation Type: Ping ?

First, verify that round-trip delay calculation is enabled by selecting the check box next to **Round Trip Delay Calculation Enabled**. Next, specify the round-trip delay calculation type by selecting either **Ping** or **Timestamp** from the **Round Trip Delay Calculation Type** drop-down menu. By default, round-trip delay calculation is enabled and uses ICMP ping requests to calculate delay.

Selecting **Ping** specifies the round-trip delay is calculated by using ICMP ping requests. VQM sends out five ping requests, spaced five seconds apart, immediately after a call is started. The data received from the ping responses is averaged to determine the round-trip delay.

Selecting **Timestamp** specifies that ICMP timestamp requests are used to calculate round-trip delay. Using ICMP timestamp requests can provide more accurate results than ICMP ping requests; however, when using timestamp requests, both the voice endpoints and any AOS units must have synchronized time and date information.



*When using either method for calculating round-trip delay, any firewalls between the voice endpoints must be configured to allow ICMP traffic to pass.*

- After configuring the VQM monitoring settings, select **Apply** from the **RTP Monitoring Settings** menu.

## Configuring the VQM Thresholds

To configure the VQM event and past calls thresholds, follow these steps:

- Navigate to **Monitoring > Voice Quality > RTP Monitoring**. Scroll to the **RTP Monitoring Settings** section and select the **Thresholds** tab.

RTP Monitoring Settings				
Monitor Settings <b>Thresholds</b> Filters				
<b>Past Calls Thresholds</b> ?				
MOS-LQ Past Calls Threshold:	4.40	Lost Packet Past Calls Threshold:	0	
MOS-CQ Past Calls Threshold:	4.40	Out of Order Packet Past Calls Threshold:	0	
MOS-PQ Past Calls Threshold:	4.40	Jitter Past Calls Threshold:	0	
<b>Event Logging Thresholds</b> ?				
	Info	Notice	Warning	Error
MOS-LQ Thresholds:	<input checked="" type="checkbox"/> 4.4	<input checked="" type="checkbox"/> 4.0	<input checked="" type="checkbox"/> 3.6	<input checked="" type="checkbox"/> 2.6
MOS-PQ Thresholds:	<input checked="" type="checkbox"/> 4.4	<input checked="" type="checkbox"/> 4.0	<input checked="" type="checkbox"/> 3.6	<input checked="" type="checkbox"/> 2.6
Out-of-Order Packets Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 25	<input checked="" type="checkbox"/> 50	<input checked="" type="checkbox"/> 100
Lost Packets Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 25	<input checked="" type="checkbox"/> 50	<input checked="" type="checkbox"/> 100
Jitter Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 250	<input checked="" type="checkbox"/> 350	<input checked="" type="checkbox"/> 450
<input type="button" value="Apply"/> <input type="button" value="Cancel"/>				



- In the **Past Calls Thresholds** menu, specify the listening, call, and perceived quality MOS thresholds, the lost packet threshold, the out-of-order packet threshold, and the jitter threshold. These thresholds are used to determine when calls are logged into the call history (Past Calls record). The past calls thresholds define the lowest call quality permissible for MOS quality and packet quality. If call or packet quality is less than the defined threshold, the calls are stored in the call history.

Past Calls Thresholds ?	
MOS-LQ Past Calls Threshold:	4.40
MOS-CQ Past Calls Threshold:	4.40
MOS-PQ Past Calls Threshold:	4.40
Lost Packet Past Calls Threshold:	0
Out of Order Packet Past Calls Threshold:	0
Jitter Past Calls Threshold:	0

The **MOS-LQ Past Calls Threshold** defines the lowest tolerable listening quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. The MOS thresholds have ranges based on the MOS voice quality ranges outlined in [Table 1 on page 4](#). By default, all past call MOS thresholds are **4.40**.

The **MOS-CQ Past Calls Threshold** defines the lowest tolerable conversation quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. The MOS thresholds have ranges based on the MOS voice quality ranges outlined in [Table 1 on page 4](#). By default, all past call MOS thresholds are **4.40**.

The **MOS-PQ Past Calls Threshold** defines the lowest tolerable perceived quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. The MOS thresholds have ranges based on the MOS voice quality ranges outlined in [Table 1 on page 4](#). By default, all past call MOS thresholds are **4.40**.

The **Lost Packet Past Calls Threshold** defines the highest number of lost packets permissible in the RTP stream. If more packets are lost than the specified number, the call is logged in the call history. Valid range is **0** to **30000** packets. By default, the past calls lost packet threshold is **0**.

The **Out of Order Packet Past Calls Threshold** defines the highest number of out-of-order packets permissible within the RTP stream. If more packets are out-of-order than the specified number, the call is logged in the call history. Valid range is **0** to **30000** packets. By default, the past calls out-of-order packet threshold is **0**.

The **Jitter Past Calls Threshold** defines the highest amount of jitter permissible within the RTP stream. If more jitter than the specified amount is detected, the call is logged in the call history. Valid range is **0** to **30000** ms. By default, the past calls jitter threshold is **0**.

- After specifying the past calls thresholds, specify the thresholds that determine when an event is logged from the **Event Logging Thresholds** menu. When call quality is less than the specified event threshold, an information (lowest severity), notice, warning, or error (greatest severity) message is generated for the call. Event thresholds are configured by selecting the check box under the event type for each of the

MOS qualities, out-of-order and lost packets, and jitter statistics and entering the threshold value for the event type.

**RTP Monitoring Settings**

Monitor Settings | **Thresholds** | Filters

**Past Calls Thresholds** ?

MOS-LQ Past Calls Threshold: 4.40      Lost Packet Past Calls Threshold: 0

MOS-CQ Past Calls Threshold: 4.40      Out of Order Packet Past Calls Threshold: 0

MOS-PQ Past Calls Threshold: 4.40      Jitter Past Calls Threshold: 0

**Event Logging Thresholds** ?

	Info	Notice	Warning	Error
MOS-LQ Thresholds:	<input checked="" type="checkbox"/> 4.4	<input checked="" type="checkbox"/> 4.0	<input checked="" type="checkbox"/> 3.6	<input checked="" type="checkbox"/> 2.6
MOS-PQ Thresholds:	<input checked="" type="checkbox"/> 4.4	<input checked="" type="checkbox"/> 4.0	<input checked="" type="checkbox"/> 3.6	<input checked="" type="checkbox"/> 2.6
Out-of-Order Packets Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 25	<input checked="" type="checkbox"/> 50	<input checked="" type="checkbox"/> 100
Lost Packets Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 25	<input checked="" type="checkbox"/> 50	<input checked="" type="checkbox"/> 100
Jitter Thresholds:	<input checked="" type="checkbox"/> 0	<input checked="" type="checkbox"/> 250	<input checked="" type="checkbox"/> 350	<input checked="" type="checkbox"/> 450

Apply    Cancel



*No more than 2 events are sent per call.*

To specify the event threshold for the listening quality MOS, select the check box under the event message type and enter a threshold value in the **MOS-LQ Thresholds** fields. By default, **Info** messages have a threshold value of **4.4**, **Notice** messages have a threshold value of **4.0**, **Warning** messages have a threshold value of **3.6**, and **Error** messages have a threshold value of **2.6**.

To specify the event threshold for the perceived quality MOS, select the check box under the event message type and enter a threshold value in the **MOS-PQ Thresholds** fields. By default, **Info** messages have a threshold value of **4.4**, **Notice** messages have a threshold value of **4.0**, **Warning** messages have a threshold value of **3.6**, and **Error** messages have a threshold value of **2.6**.

To specify the event threshold for the number of out-of-order packets in the RTP stream, select the check box under the event message type and enter a threshold value in the **Out-of-Order Packets Thresholds** fields. By default, **Info** messages for out-of-order packets are enabled with a threshold value of **0** packets. By default, **Notice** messages for out-of-order packets are enabled with a threshold value of **25** packets. By default, **Warning** messages for out-of-order packets are enabled with a threshold value of **50** packets. By default, **Error** messages for out-of-order packets are enabled with a threshold value of **100** packets.

To specify the event threshold for the number of lost packets in the RTP stream, select the check box under the event message type and enter a threshold value in the **Lost Packets Thresholds** fields. By default, **Info** messages for lost packets are enabled with a threshold value of **0** packets. By default, **Notice** messages for lost packets are enabled with a threshold value of **25** packets. By default, **Warning** messages for lost packets are enabled with a threshold value of **50** packets. By default, **Error** messages for lost packets are enabled with a threshold value of **100** packets.

To specify the event threshold for the amount of jitter in the RTP stream, select the check box under the event message type and enter a threshold value in the **Jitter Thresholds** fields. By default, **Info** messages for jitter are enabled with a threshold value of **0** ms. By default, **Notice** messages for jitter are enabled with a threshold value of **250** ms. By default, **Warning** messages for jitter are enabled with a threshold value of **350** ms. By default, **Error** messages for jitter are enabled with a threshold value of **450** ms.

4. When you have configured the past calls and events thresholds, select **Apply** at the bottom of the **Thresholds** configuration tab to apply these settings.

## Specifying VQM Monitoring Filters (optional)

To configure the VQM monitoring filters, follow these steps:

1. Navigate to **Monitoring > Voice Quality > RTP Monitoring**. Scroll to the **RTP Monitoring Settings** section and select the **Filters** tab.

The screenshot shows the 'RTP Monitoring Settings' window with the 'Filters' tab selected. The window has three sub-tabs: 'Monitor Settings', 'Thresholds', and 'Filters'. The 'Filters' tab is active and contains the following elements:

- User Filters** section with a help icon (i):
  - A 'User' dropdown menu.
  - A 'Total Selected: 0' indicator.
  - 'Select All' and 'Delete Selected' buttons.
  - 'Add New' button and an empty text input field.
- Access Control List (ACL) Filter** section with a help icon (i):
  - An 'ACL:' dropdown menu.

At the bottom of the window are 'Apply' and 'Cancel' buttons.

2. Specify the **User Filters** by selecting **Add New** and entering either the SIP To or From header of the user you want to monitor in the adjacent field. Enter user information in the format **user@host**. By default, no user filters exist. When new user filters are created, they appear in the **User** list. To delete a user filter, select the filter from the list and select **Delete Selected**. If no user filter is specified, all calls are monitored unless the sample rate is specified or an ACL filter is used.
3. Specify the **Access Control List (ACL) Filter** by selecting the appropriate previously configured ACL from the **ACL** drop-down menu. Using these filters causes VQM to only monitor calls where one of the RTP streams matches the ACL. By default, no ACL filters are applied to VQM. If no ACL filter is specified, all calls are monitored unless the sample rate is specified or a user filter is specified. For more information about configuring ACLs, refer to the configuration guide *Configuring IP ACLs in AOS* available online at <http://kb.adtran.com> (article number 3087).
4. When you have finished adding filters to VQM, select **Apply** at the bottom of the **Filters** menu to apply the settings.

### Enabling VQM on the Network Interface (optional)

For VQM to function properly, it must be enabled on the interfaces through which the RTP streams flow. VQM is enabled by default on all interfaces. If you need to change these settings, for example to monitor fewer interfaces (by disabling VQM on the interfaces you do not want to monitor) or to monitor only a specific interface, use the CLI as described in *Enabling VQM on a Network Interface (optional)* on page 27. Enabling or disabling VQM on an interface can only be accomplished using the GUI on some AOS products.

VQM is now configured and you can begin monitoring the voice quality of your network and use the gathered VQM statistics for troubleshooting voice quality issues.

## Configuring VQM Using the CLI

VQM can also be configured using the CLI. Refer to *VQM Configuration Overview on page 5* for an overview of the steps involved in VQM configuration. To begin configuring VQM using the CLI, follow the steps outlined in this section.

### Accessing the CLI

To access the CLI in your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet <ip address>**). For example,  
**telnet 208.61.209.1**
3. Enter your user name and password at the prompt.



*The AOS default user name is **admin** and the default password is **password**. If your product no longer has the default user name and password, contact your administrator for the appropriate user name and password.*

4. Enter the Enable mode by entering **enable** at the prompt as follows:  
**>enable**
5. Enter your Enable mode password at the prompt.
6. Enter the unit's Global Configuration mode as follows:  
**#configure terminal**  
**(config)#**

### Configuring the Firewall and SIP Settings

To configure the firewall for VQM, make sure that the firewall is enabled. If non-default ACPs are used with **allow** policy entries, you must make sure the SIP and RTP traffic do not match stateless **allow** policy entries. For more information, about configuring the AOS firewall and SIP settings, refer to the configuration guides *Configuring the SIP Proxy in AOS* (article number 2183) and *Configuring the IP Firewall in AOS* (article number 1543) available online at <http://kb.adtran.com>.

## Enabling VQM and Configuring the VQM Monitoring Settings

Use the following steps to enable VQM and configure the monitoring settings.

1. Enable VQM using the **ip rtp quality-monitoring [scoring-adjustment japan]** command. This command enables VQM globally and optionally sets the MOS adjustment to Japanese standards using the **scoring-adjustment japan** parameter. By default, VQM is disabled and, when enabled, uses the standard MOS scoring adjustment. Use the **no** form of this command to disable VQM globally.



*Disabling VQM erases all active call, new call, and interface statistics. Call history and endpoint statistics are not affected. These statistics must be cleared manually using the **clear ip rtp quality-monitoring** command.*

To enable VQM, enter the command from the Global Configuration mode prompt as follows:

```
(config)#ip rtp quality-monitoring
```

2. Specify the signaling type of the RTP stream to monitor using the **ip rtp quality-monitoring [sip | udp]**. The **sip** parameter specifies that VQM monitors streams detected by the SIP ALG, SIP proxy, SIP B2BUA, or MGCP. If this command is issued and VQM has not been globally enabled using the **ip rtp quality-monitoring** command, VQM will be enabled. By default, VQM is disabled and no RTP stream types are defined. Using the **no** form of this command disables VQM and its packet inspection.



*Enabling UDP packet inspection forces the AOS unit to inspect every UDP packet to determine if it is an RTP packet, placing a significant load on the AOS unit. UDP packet inspection should only be enabled if IP phones are being used and they do not pass through the SIP ALG, SIP proxy, or SIP B2BUA.*

To specify the types of packets that VQM inspects, enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring sip
```

3. Specify the size of the call history using the **ip rtp quality-monitoring history max-stream <number>** command. The **<number>** parameter is the number of calls stored in the call history. Valid range is **0** to **2000** calls. By default, the call history holds **100** calls. Using the **no** form of this command returns the call history size to the default value. Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring history max-stream 200
```



*Setting the size of the call history to a large value can result in the AOS unit running out of memory.*

- Specify the VQM simulated jitter buffer type as adaptive using the **ip rtp quality-monitoring jitter-buffer adaptive [min <value> | nominal <value> | max <value>]** command or as fixed using the **ip rtp quality-monitoring jitter-buffer fixed [nominal <value> | jitter-buffer-size <size>]** command. By default, the VQM simulated jitter buffer is set to **adaptive**. Using the **no** form of these commands returns the simulated jitter buffer type to the default setting.

The **ip rtp quality-monitoring jitter-buffer adaptive** command specifies that the simulated jitter buffer automatically increases or decreases playout delay to adjust for varying jitter. If the simulated jitter buffer is **adaptive**, you must also specify the minimum delay (smallest allowed delay), the maximum delay (largest allowed delay), and the nominal value (beginning delay) of the simulated jitter buffer. The minimum delay is set using the **min <value>** parameter. Valid range is **10** to **240** ms and the default is **10** ms. The maximum delay is set using the **max <value>** parameter. Valid range is **40** to **320** ms and the default is **200** ms. The nominal delay is set using the **nominal <value>** parameter. Valid range is **40** to **320** ms and the default is **50** ms.

The **ip rtp quality-monitoring jitter-buffer fixed** parameter specifies that the simulated jitter buffer uses a fixed playout delay for all packets. If the simulated jitter buffer is **fixed**, you must also specify the nominal delay setting and the simulated jitter buffer size (number of packets the buffer can hold, as if it were a non-emulated jitter buffer). The nominal delay setting for the fixed jitter buffer is set using the **nominal <value>** parameter. The valid range is **4** to **250** ms. By default, no fixed nominal delay is specified. The fixed jitter buffer size is set using the **jitter-buffer-size <size>** parameter. Valid range is **10** to **500** packets. By default, no fixed jitter buffer size is specified.



*VQM uses a simulated jitter buffer for quality measurements. For the most accurate results, select settings that are consistent with the jitter buffer settings on the VoIP endpoints in your network. In addition, because the VQM jitter buffer is a simulated buffer, it does not affect the playback of RTP streams.*

To specify the simulated jitter buffer type, enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring jitter-buffer adaptive min 50 nominal 50 max 100
```

- Specify the early and late thresholds for the simulated jitter buffer using the **ip rtp quality-monitoring jitter-threshold [early <ms> | late <ms>]** command. These thresholds determine when packets are either too early or too late for the jitter buffer window, and are thus logged as discarded packets. Specify the early packet threshold using the **early <ms>** parameter. Valid range is **0** to **1000** ms and the default is **10** ms. Specify the late packet threshold using the **late <ms>** parameter. Valid range is **0** to **1000** ms and the default is **50** ms. Using the **no** form of this command returns the thresholds to the default values. To change the early or late packet threshold, enter the commands from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring jitter-threshold early 20
```

```
(config)#ip rtp quality-monitoring jitter-threshold late 40
```

- Specify the call sampling rate using the **ip rtp quality-monitoring sample one-out-of <number>**. This number specifies that 1 out of the entered number of calls are monitored by VQM. For example, entering **1** indicates that all calls are monitored (1 out of 1 calls are monitored), and entering **100** indicates that only 1 percent of calls are monitored (1 out of 100 calls are monitored). Using a sampling rate can help to reduce the amount of resources used by the VQM feature and allow a larger part of the incoming calls to proceed as normal, including through RapidRoute (if enabled). Valid sampling range is **1** to **100**. By default, the sampling rate is **1**. Using the **no** form of this command returns the sample rate to the default

value. To change the call sampling rate, enter the command from the Global Configuration mode prompt as follows:

```
(config)#ip rtp quality-monitoring sample one-out-of 10
```

- Specify the round-trip delay calculation settings using the **ip rtp quality-monitoring round-trip-delay [icmp-ping | icmp-timestamp]** command. The round-trip delay settings specify how the round-trip delay for RTP packets is calculated. These calculations appear in the VQM statistics; however, if you also have the ability to generate RTCP XR reports, the data from the RTCP XR reports will supersede the round-trip delay settings. RTCP is monitored by VQM automatically, and requires no additional configuration. Using the **ip rtp quality-monitoring round-trip-delay** command enables round-trip delay calculation and specifies the method by which these calculations are made. The **icmp-ping** parameter specifies that the round-trip delay is calculated by using ICMP ping requests. VQM sends out five ping requests, spaced about five seconds apart, immediately after a call is started. The data received from the ping responses is averaged to determine the round-trip delay. The **icmp-timestamp** parameter specifies that ICMP timestamp requests are used to calculate round-trip delay. Using ICMP timestamp requests can provide more accurate results than ICMP ping requests; however, when using timestamp requests, the voice endpoints and the AOS unit must have synchronized time and date information. By default, round-trip delay is enabled and uses ICMP ping requests for calculation. Using the **no** form of this command disables round-trip delay calculations. To change the round-trip delay calculation setting, enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring round-trip-delay icmp-timestamp
```



*When using round-trip delay calculations, any firewalls between the voice endpoints must be configured to allow ICMP traffic to pass.*

- After configuring the monitoring settings for VQM, configure the VQM thresholds.

## Configuring the VQM Thresholds

To configure the VQM event and past calls thresholds, follow these steps:

- Specify the past calls thresholds using the **ip rtp quality-monitoring history [cq-mos <value> | jitter <value> | loss <number> | lq-mos <value> | out-of-order <number> | pq-mos <value>]** command. This command sets the quality threshold for determining when calls are logged into the call history. These thresholds define the lowest call quality permissible for MOS and packet quality. If call or packet quality is less than the defined threshold, the calls are stored in the call history. Using the **no** form of this command returns the thresholds to the default values. Each of the parameters of this command are described below.

The **cq-mos <value>** parameter defines the lowest tolerable conversation quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. By default, all MOS call history thresholds are **4.4**. MOS past call thresholds have a range based on the MOS voice quality range as defined in [Table 1 on page 4](#).

The **jitter <value>** parameter of this command defines the highest amount of jitter permissible within the RTP stream. If more jitter than the specified amount is detected, the call is logged into the call history. Valid range is **0** to **30000** ms. By default, the jitter threshold is **0**.



The **loss** *<number>* parameter of this command defines the highest number of lost packets permissible in the RTP stream. If more packets are lost than the specified number, the call is logged in the call history. Valid range is **0** to **30000** packets. By default, the lost packet threshold is **0**.

The **lq-mos** *<value>* of this command defines lowest permissible listening quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. By default, all MOS call history thresholds are **4.4**. MOS past call thresholds have a range based on the MOS voice quality range as defined in [Table 1 on page 4](#).

The **out-of-order** *<number>* parameter of this command defines the highest number of out-of-order packets permissible within the RTP stream. If more packets are out-of-order than the specified number, the call is logged in the call history. Valid range is **0** to **30000** packets. By default, the out-of-order threshold is **0**.

The **pq-mos** *<value>* parameter of this command defines the lowest permissible perceived quality MOS. If a call displays quality lower than this threshold, the call is logged in the call history. By default, all MOS call history thresholds are **4.4**. MOS past call thresholds have a range based on the MOS voice quality range as defined in [Table 1 on page 4](#).

To change the past calls threshold values, enter the commands in the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring history lq-mos 4.0
(config)#ip rtp quality-monitoring history cq-mos 4.0
(config)#ip rtp quality-monitoring history pq-mos 4.0
(config)#ip rtp quality-monitoring history loss 50
(config)#ip rtp quality-monitoring history out-of-order 20
(config)#ip rtp quality-monitoring history jitter 10
```

2. Specify the event thresholds using the **ip rtp quality-monitoring threshold** commands. These commands specify the thresholds used to determine when an event is logged and what severity the event has. When call quality is less than the specified event threshold, an information (lowest severity), notice, warning, or error (greatest severity) message is generated for the call. Using the **no** form of these commands disables the event reporting for that message type and threshold. Each threshold command is described below.

To specify the event reporting threshold for listening quality MOS, use the **ip rtp quality-monitoring threshold lq-mos [info | notice | warning | error] <value>** command. The **info** parameter specifies an information event is logged. The **notice** parameter specifies a notice event is logged. The **warning** parameter specifies a warning event is logged. The **error** parameter specifies an error event is logged. The *<value>* parameter specifies the threshold value. By default, all **info** messages have a threshold value of **4.4**, all **notice** messages have a threshold value of **4.0**, all **warning** messages have a threshold value of **3.6**, and all **error** messages have a threshold value of **2.6**. Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring threshold lq-mos info 4.3
```

To specify the event reporting threshold for perceived quality MOS, use the **ip rtp quality-monitoring threshold pq-mos [info | notice | warning | error] <value>** command. The **info** parameter specifies an information event is logged. The **notice** parameter specifies a notice event is logged. The **warning** parameter specifies a warning event is logged. The **error** parameter specifies an error event is logged.

The *<value>* parameter specifies the threshold value. By default, all **info** messages have a threshold value of **4.4**, all **notice** messages have a threshold value of **4.0**, all **warning** messages have a threshold value of **3.6**, and all **error** messages have a threshold value of **2.6**. Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring threshold pq-mos error 3.0
```

To specify the event reporting threshold for out-of-order packets, use the **ip rtp quality-monitoring threshold out-of-order [info | notice | warning | error] <value>** command. The **info** parameter specifies an information event is logged, and has a default threshold of **0**. The **notice** parameter specifies a notice event is logged, and has a default threshold of **25**. The **warning** parameter specifies a warning event is logged, and has a default threshold of **50**. The **error** parameter specifies an error event is logged, and has a default threshold of **100**. The *<value>* parameter specifies the threshold value (in packets). Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring threshold out-of-order notice 20
```

To specify the event reporting threshold for lost packets, use the **ip rtp quality-monitoring threshold loss [info | notice | warning | error] <value>** command. The **info** parameter specifies an information event is logged, and has a default threshold of **0**. The **notice** parameter specifies a notice event is logged, and has a default threshold of **25**. The **warning** parameter specifies a warning event is logged, and has a default threshold of **50**. The **error** parameter specifies an error event is logged, and has a default threshold of **100**. The *<value>* parameter specifies the threshold value (in packets). Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring threshold loss info 10
```

To specify the event reporting threshold for jitter, use the **ip rtp quality-monitoring threshold jitter [info | notice | warning | error] <value>** command. The **info** parameter specifies an information event is logged, and has a default threshold of **0**. The **notice** parameter specifies a notice event is logged, and has a default threshold of **250**. The **warning** parameter specifies a warning event is logged, and has a default threshold of **350**. The **error** parameter specifies an error event is logged, and has a default threshold of **450**. The *<value>* parameter specifies the threshold value (in ms). Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring threshold jitter warning 300
```

3. When you have configured the event and past calls thresholds, configure the VQM monitoring filters.

## Specifying VQM Monitoring Filters (optional)

Using filters causes VQM to only monitor calls where either a SIP To or From header matches a list of users to monitor or where one of the RTP streams matches a preconfigured ACL. If no filter type is applied, VQM monitors all calls, unless the call sample rate is specified. To configure the VQM monitoring filters, follow these steps:

1. Specify that VQM only measure calls from certain users using the **ip rtp quality-monitoring filter user <user>** command. The *<user>* parameter of this command specifies the SIP To or From header for a specific user. Enter the *<user>* parameter in the **user@host** format. By default, no user filters exist. Using the **no** form of this command removes the user from the filter. Multiple users can be monitored simultaneously. Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring filter user stephaniea@companya
```

2. Specify that VQM only measure RTP streams that match a specific previously configured ACL using the **ip rtp quality-monitoring filter access-class** *<name>*. The *<name>* parameter is the name of the ACL you want to use. By default, no ACLs are defined or applied to VQM. Using the **no** form of this command removes the association between VQM and the ACL. Only one ACL can be applied to VQM at a time. Enter the command from the Global Configuration mode as follows:

```
(config)#ip rtp quality-monitoring filter access-class MYACL
```

3. After configuring any VQM filters, enable VQM on the appropriate network interfaces.

### Enabling VQM on a Network Interface (optional)

For VQM to function properly, it must be enabled on the network interfaces to be monitored. By default, VQM is enabled on all interfaces. Each WAN or LAN interface that is assigned an IP address, or has a method of getting an IP address assigned, must have VQM enabled. To enable VQM on an interface, use the **rtp quality-monitoring** command from the interface's configuration mode. Using the **no** form of this command disables VQM on the interface. Although VQM statistics are only gathered on the inbound interface, VQM must be enabled on both inbound and outbound interfaces that will have RTP streams. To enable VQM on an interface, enter the command as follows:

```
(config)#interface eth 0/1  
(config-eth 0/1)#rtp quality-monitoring
```

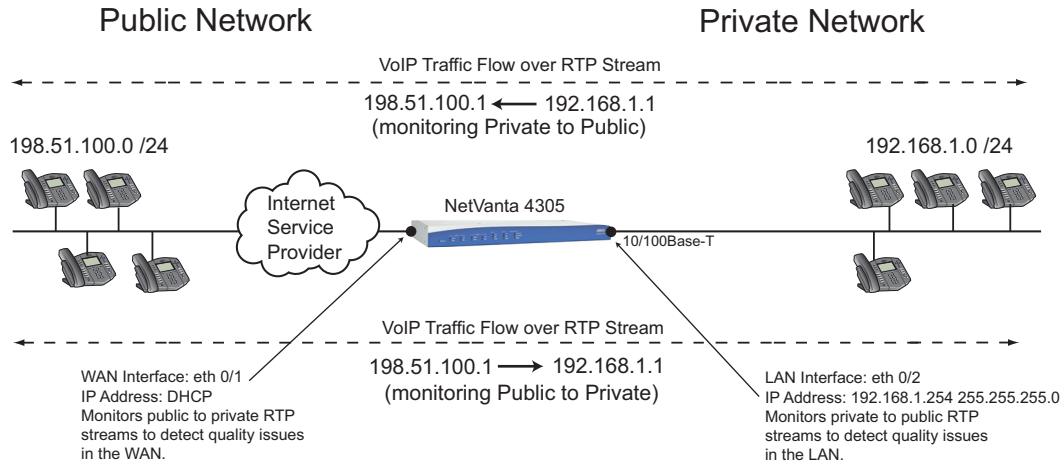
VQM is now configured and you can begin monitoring the voice quality of your network, as well as use the gathered VQM statistics for troubleshooting voice quality issues.

## VQM Configuration Examples

The following are sample configurations of VQM on an AOS product. These scenarios are provided for example purposes only and only include the information necessary for configuring VQM on the unit. Example configurations should be modified to fit your specific configuration needs.

## Basic VQM Configuration for Router Monitoring with SNMP Traps

In this scenario, a router is acting as the gateway to the Internet for the private network **192.168.1.0 /24**. The private side of this network has IP phones that register and operate through a public VoIP gateway. This configuration monitors the voice quality statistics for RTP streams for all private to public calls.



**Figure 2. Basic VQM in an AOS Router Monitoring with SNMP Traps**

This scenario can be configured a number of ways, depending upon the firewall and SIP settings used on your network. There are two configurations shown here: one in which the firewall SIP ALG is used with VQM, and one in which the SIP proxy is used with VQM. In each configuration, the firewall is enabled, the SIP ALG or SIP proxy is configured, ACPs are created and applied to the interfaces, and VQM is configured.

### Basic VQM Configuration with Firewall SIP ALG

```
!
ip firewall
ip firewall alg sip
!
interface eth 0/1
    ip address dhcp
    no shutdown
!
interface eth 0/2
    ip address 192.168.1.254 255.255.255.0
    access-policy Private
    no shutdown
!
ip access-list standard MATCHALL
    permit any
!
ip policy-class Private
    allow list MATCHALL self
    nat source list MATCHALL interface eth 0/1 overload
```

```
!  
snmp-server enable traps snmp  
snmp-server community public RO  
snmp-server host 192.168.1.253 traps version 1 public snmp  
!  
ip rtp quality-monitoring  
ip rtp quality-monitoring sip  
ip rtp-quality-monitoring snmp trap priority-level notice  
!
```

### Basic VQM Configuration with SIP Proxy

```
!  
ip firewall  
!  
interface eth 0/1  
    ip address dhcp  
    media-gateway ip primary  
    access-policy Public  
    no shutdown  
!  
interface eth 0/2  
    ip address 192.168.1.254 255.255.255.0  
    media-gateway ip primary  
    access-policy Private  
    no shutdown  
!  
ip access-list standard MATCHALL  
    permit any  
!  
ip access-list extended SIP  
    permit udp any any eq 5060  
!  
ip policy-class Private  
    allow list MATCHALL self  
    nat source list MATCHALL interface eth 0/1 overload policy Public  
!  
ip policy-class Public  
    allow list SIP self  
!  
snmp-server enable traps snmp  
snmp-server community public RO  
snmp-server host 192.168.1.253 traps version 1 public snmp  
!
```

```

ip sip
ip sip proxy
ip sip proxy transparent
!
ip rtp quality-monitoring
ip rtp quality-monitoring sip
ip rtp quality-monitoring snmp trap priority-level notice
!

```

## VQM Configuration Command Summary

The following tables summarize the VQM configuration commands.

**Table 3. Basic VQM Configuration Command Summary**

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring [scoring-adjustment japan]</b>	Enables VQM globally and optionally changes the MOS scoring adjustment to the Japanese standard (using the <b>scoring-adjustment-japan</b> parameter). By default, VQM is disabled and, when enabled, uses the standard MOS scoring adjustment. Use the <b>no</b> form of this command to disable VQM globally. Note that disabling VQM erases all active call, new call, and interface statistics. Call history and endpoint statistics are not affected. These statistics must be cleared manually using the <b>clear ip rtp quality-monitoring</b> command.
(config)#	<b>[no] ip rtp quality-monitoring [sip   udp]</b>	Specifies the signaling type of the RTP stream to monitor. VQM inspects the specified packet type to determine if it is an RTP stream that should be monitored. The <b>sip</b> parameter indicates that SIP and MGCP packets are inspected. The <b>udp</b> parameter indicates that UDP packets are inspected. If this command is issued and VQM has not been globally enabled, VQM will be enabled globally. By default, VQM is disabled and no packet inspection type is defined. Using the <b>no</b> form of this command disables VQM and its packet inspection. Note that enabling UDP packet inspection forces the AOS unit to inspect every UDP packet to determine if it is an RTP packet, placing a significant load on the AOS unit. UDP packet inspection should only be enabled if IP phones are being used and they do not pass through the SIP ALG, SIP proxy, or SIP B2BUA.

Table 3. Basic VQM Configuration Command Summary (Continued)

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring history max-stream &lt;number&gt;</b>	Specifies the size of the call history. The <number> parameter is the maximum number of calls that can be stored in the call history. Valid range is <b>0</b> to <b>2000</b> calls. By default, the call history holds <b>100</b> calls. Using the <b>no</b> form of this command returns the call history size to the default value. Note that setting the size of the call history to a large number can result in the AOS unit running out of memory.
(config)#	<b>[no] ip rtp quality-monitoring jitter-buffer adaptive [min &lt;value&gt;   nominal &lt;value&gt;   max &lt;value&gt;]</b>	Specifies that the VQM simulated jitter buffer is an adaptive buffer and configures the adaptive buffer settings. The <b>min &lt;value&gt;</b> parameter specifies the minimum buffer delay and has a range of <b>10</b> to <b>240</b> ms. By default, the minimum delay is <b>10</b> ms. The <b>nominal &lt;value&gt;</b> parameter specifies the beginning buffer delay and has a range of <b>40</b> to <b>320</b> ms. By default, the nominal delay is <b>50</b> ms. The <b>max &lt;value&gt;</b> parameter specifies the maximum buffer delay and has a range of <b>40</b> to <b>320</b> ms. By default, the maximum delay is <b>200</b> ms. Using the <b>no</b> form of this command returns the simulated jitter buffer type to the default setting. Note that this jitter buffer is simulated and does not affect the playback of RTP streams.
(config)#	<b>[no] ip rtp quality-monitoring jitter-buffer fixed [nominal &lt;value&gt;   jitter-buffer-size &lt;size&gt;]</b>	Specifies that the VQM simulated jitter buffer is a fixed buffer and configures the fixed buffer settings. The <b>nominal &lt;value&gt;</b> parameter specifies the buffer delay and has a range of <b>4</b> to <b>250</b> ms. By default, no fixed nominal delay is specified. The <b>jitter-buffer-size &lt;size&gt;</b> parameter specifies that you are configuring how many packets the simulated jitter buffer can hold. The <size> parameter is the number of packets that can be held. The range is <b>10</b> to <b>500</b> packets. By default, no fixed jitter buffer size is specified. Using the <b>no</b> form of this command returns the simulated jitter buffer type to the default settings. Note that this jitter buffer is simulated and does not affect the playback of RTP streams.
(config)#	<b>[no] ip rtp quality-monitoring jitter-threshold [early &lt;ms&gt;   late &lt;ms&gt;]</b>	Specifies the early and late thresholds for the simulated jitter buffer. These thresholds determine when packets are considered either too early or too late for the jitter buffer window, and are then marked as discarded packets. The <b>early &lt;ms&gt;</b> parameter sets the early threshold. Valid range is <b>0</b> to <b>1000</b> ms. By default, the early threshold is <b>10</b> ms. The <b>late &lt;ms&gt;</b> parameter sets the late threshold. Valid range is <b>0</b> to <b>1000</b> ms. By default, the late threshold is <b>50</b> ms. Using the <b>no</b> form of this command returns the jitter buffer thresholds to the default values.

Table 3. Basic VQM Configuration Command Summary (Continued)

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring sample one-out-of &lt;number&gt;</b>	Specifies the call sampling rate, which causes VQM to only monitor 1 out of the specified number of RTP streams. Sampling range is <b>1</b> to <b>100</b> . By default, the sampling rate is <b>1</b> , which means that all RTP streams are monitored. Using the <b>no</b> form of this command disables sampling.
(config)#	<b>[no] ip rtp quality-monitoring round-trip-delay [icmp-ping   icmp-timestamp]</b>	Enables round-trip calculations and specifies on what these calculations are based. Round-trip delay settings appear in the VQM statistics; however, if RTCP XR reports are also available, the received RTCP XR reports supersede the round-trip delay settings. The <b>icmp-ping</b> parameter specifies round-trip delay calculations are based on ICMP ping requests. The <b>icmp-timestamp</b> parameter specifies round-trip delay calculations are based on ICMP timestamp requests. By default, round-trip delay calculations are enabled and based on ICMP ping requests. Using the <b>no</b> form of this command disables round-trip delay calculations. Note that to use round-trip delay calculations, any firewalls between the voice endpoints must be configured to allow ICMP traffic to pass. In addition, when using ICMP timestamp requests, both voice endpoints and any AOS devices must have synchronized date and time information.

Table 4. Configuring VQM Thresholds Command Summary

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring history cq-mos &lt;value&gt;</b>	Specifies the lowest permissible conversation quality MOS for the RTP stream. If a call displays quality lower than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>4.4</b> . By default, the conversation quality MOS threshold is <b>4.4</b> . Using the <b>no</b> form of this command returns the threshold to the default value.
(config)#	<b>[no] ip rtp quality-monitoring history jitter &lt;value&gt;</b>	Specifies the highest amount of jitter permissible for the RTP stream. If a call displays jitter higher than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>30000</b> ms. By default, the jitter threshold is <b>0</b> . Using the <b>no</b> form of this command returns the threshold to the default value.



Table 4. Configuring VQM Thresholds Command Summary (Continued)

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring history loss &lt;number&gt;</b>	Specifies the highest number of lost packets permissible for the RTP stream. If a call displays more lost packets than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>30000</b> packets. By default, the loss threshold is <b>0</b> packets. Using the <b>no</b> form of this command returns the threshold to the default value.
(config)#	<b>[no] ip rtp quality-monitoring history lq-mos &lt;value&gt;</b>	Specifies the lowest permissible listening quality MOS for the RTP stream. If a call displays quality lower than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>4.4</b> . By default, the listening quality MOS threshold is <b>4.4</b> . Using the <b>no</b> form of this command returns the threshold to the default setting.
(config)#	<b>[no] ip rtp quality-monitoring history out-of-order &lt;number&gt;</b>	Specifies the highest number of out-of-order packets permissible for the RTP stream. If a call displays more out-of-order packets than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>30000</b> packets. By default, the out-of-order threshold is <b>0</b> packets. Using the <b>no</b> form of this command returns the threshold to the default value.
(config)#	<b>[no] ip rtp quality-monitoring history pq-mos &lt;value&gt;</b>	Specifies the lowest permissible perceived quality MOS for the RTP stream. If a call displays quality lower than this threshold, the call is logged in the call history. Valid range is <b>0</b> to <b>4.4</b> . By default, the perceived quality MOS threshold is <b>4.4</b> . Using the <b>no</b> form of this command returns the threshold to the default setting.
(config)#	<b>[no] ip rtp quality-monitoring threshold lq-mos [info   notice   warning   error] &lt;value&gt;</b>	Specifies the event threshold for listening quality MOS. When call quality is less than the specified event threshold, an event message is generated. The <b>info</b> , <b>notice</b> , <b>warning</b> , and <b>error</b> keywords specify the type of event logged, and the <b>&lt;value&gt;</b> parameter specifies the threshold value for the message to be generated. By default, all <b>info</b> messages have a threshold of <b>4.4</b> , all <b>notice</b> messages have a threshold of <b>4.0</b> , all <b>warning</b> messages have a threshold of <b>3.6</b> , and all <b>error</b> messages have a threshold of <b>2.6</b> . Using the <b>no</b> form of this command disables the event reporting for that message type and threshold.

Table 4. Configuring VQM Thresholds Command Summary (Continued)

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring threshold pq-mos [info   notice   warning   error] &lt;value&gt;</b>	Specifies the event threshold for perceived quality MOS. When call quality is less than the specified event threshold, an event message is generated. The <b>info</b> , <b>notice</b> , <b>warning</b> , and <b>error</b> keywords specify the type of event logged, and the <value> parameter specifies the threshold value for the message to be generated. By default, all <b>info</b> messages have a threshold of <b>4.4</b> , all <b>notice</b> messages have a threshold of <b>4.0</b> , all <b>warning</b> messages have a threshold of <b>3.6</b> , and all <b>error</b> messages have a threshold of <b>2.6</b> . Using the <b>no</b> form of this command disables the event reporting for that message type and threshold.
(config)#	<b>[no] ip rtp quality-monitoring threshold out-of-order [info   notice   warning   error] &lt;value&gt;</b>	Specifies the event threshold for out-of-order packets. When more out-of-order packets are detected than the specified event threshold, an event message is generated. The <b>info</b> , <b>notice</b> , <b>warning</b> , and <b>error</b> keywords specify the type of event logged, and the <value> parameter specifies the threshold value for the message to be generated. By default, <b>info</b> messages have a threshold of <b>0</b> , <b>notice</b> messages have a threshold of <b>25</b> , <b>warning</b> messages have a threshold of <b>50</b> , and <b>error</b> messages have a threshold of <b>100</b> packets. Using the <b>no</b> form of this command disables the event reporting for that message type and threshold.
(config)#	<b>[no] ip rtp quality-monitoring threshold loss [info   notice   warning   error] &lt;value&gt;</b>	Specifies the event threshold for lost packets. When more lost packets are detected than the specified event threshold, an event message is generated. The <b>info</b> , <b>notice</b> , <b>warning</b> , and <b>error</b> keywords specify the type of event logged, and the <value> parameter specifies the threshold value for the message to be generated. By default, <b>info</b> messages have a threshold of <b>0</b> , <b>notice</b> messages have a threshold of <b>25</b> , <b>warning</b> messages have a threshold of <b>50</b> , and <b>error</b> messages have a threshold of <b>100</b> packets. Using the <b>no</b> form of this command disables the event reporting for that message type and threshold.

Table 4. Configuring VQM Thresholds Command Summary (Continued)

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring threshold jitter [info   notice   warning   error] &lt;value&gt;</b>	Specifies the event threshold for jitter. When more jitter is detected than the specified event threshold, an event message is generated. The <b>info</b> , <b>notice</b> , <b>warning</b> , and <b>error</b> keywords specify the type of event logged, and the <value> parameter specifies the threshold value for the message to be generated. By default, <b>info</b> messages have a threshold of <b>0</b> , <b>notice</b> messages have a threshold of <b>250</b> , <b>warning</b> messages have a threshold of <b>350</b> , and <b>error</b> messages have a threshold of <b>450</b> . Using the <b>no</b> form of this command disables the event reporting for that message type and threshold.

Table 5. VQM Monitoring Filter Configuration Command Summary

Prompt	Command	Description
(config)#	<b>[no] ip rtp quality-monitoring filter user &lt;user&gt;</b>	Specifies that only calls from certain users are measured. The <user> parameter specifies the SIP To or From header for a specific user. Users are specified in <b>user@host</b> format. By default, no user filters exist. Multiple users can be monitored simultaneously. Using the <b>no</b> form of this command removes the user from the filter.
(config)#	<b>[no] ip rtp quality-monitoring filter access-class &lt;name&gt;</b>	Specifies that only RTP streams that match the previously configured ACL are measured. The <name> parameter is the ACL to be used. By default, no ACLs are defined or applied to VQM. Using the <b>no</b> form of this command removes the association between VQM and the ACL. Only one ACL can be applied to VQM at a time.

Table 6. VQM Interface Configuration Command Summary

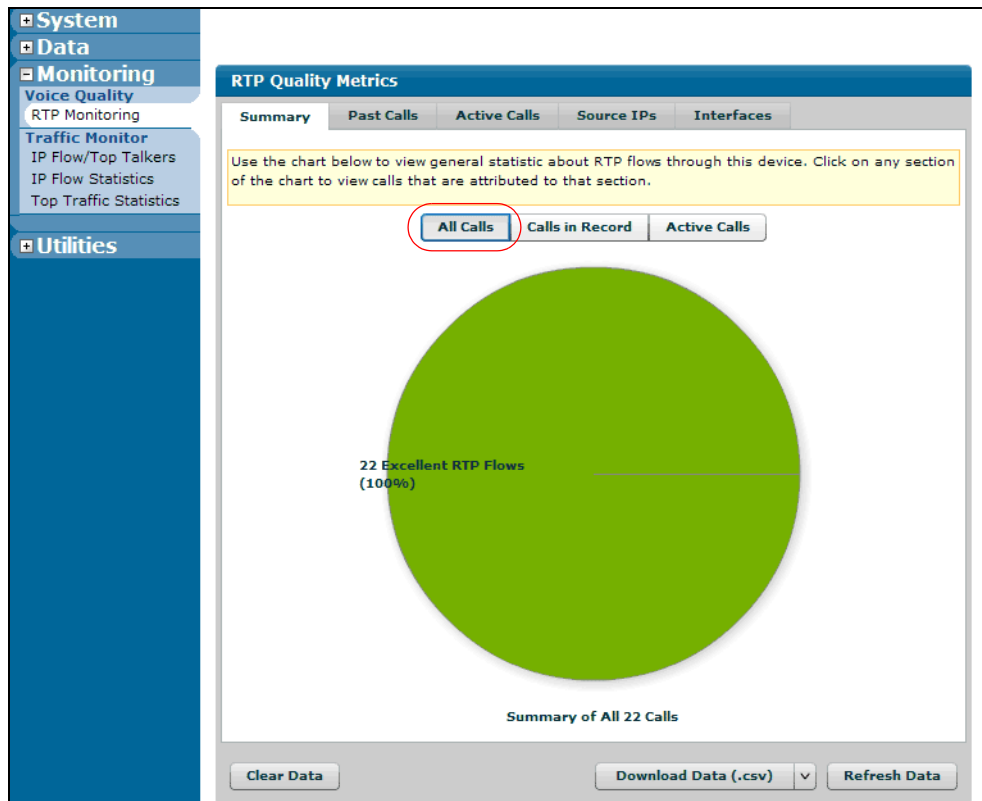
Prompt	Command	Description
(config-<interface>)#	<b>[no] rtp quality-monitoring</b>	Enables VQM on the specific interface. By default, VQM is enabled on all interfaces. Using the <b>no</b> form of this command disables VQM on the interface.

## Viewing VQM Statistics Using the GUI

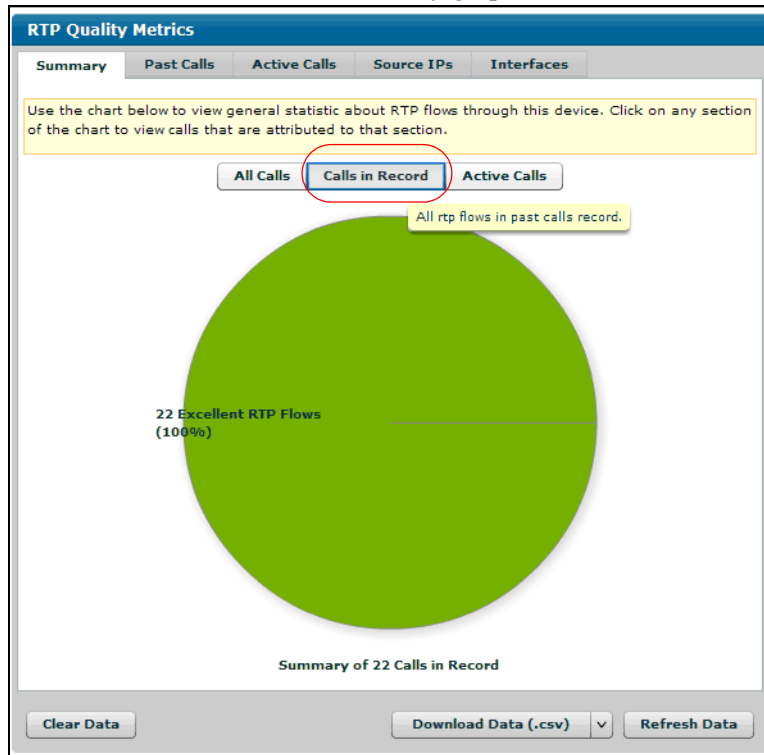
VQM statistics provide a method for determining the cause and network location of quality issues on VoIP networks. VQM statistics can be viewed in a summary format of all calls, past calls (calls logged in the call history), active calls, source IP addresses, and interfaces. Each of these statistical views can then be sorted by MOS, jitter, out-of-order packets, lost packets, or delayed packets. Summary views can be sorted by all calls, calls in the record, and active calls. The following sections describe how to access these statistics from the GUI and gives examples of the output you can expect from these statistics.

To view the various VQM statistics using the GUI, follow these steps:

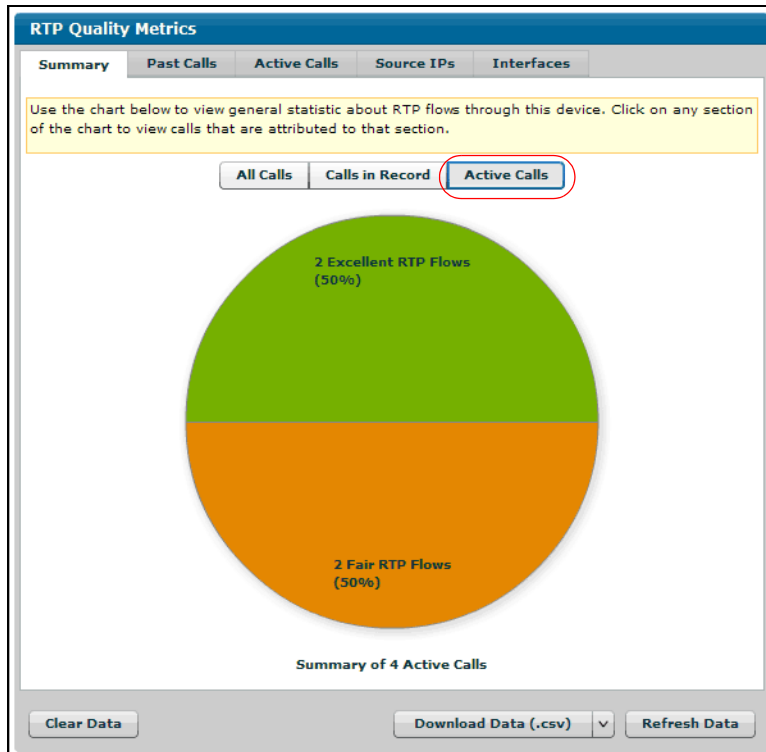
1. Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Summary** tab to view a summary of the collected voice traffic statistics. The summary view can be sorted by **All Calls**, **Calls in Record**, and **Active Calls**. The **All Calls** button displays a graph containing all RTP flows since the last reload or since the VQM statistics were cleared.



Select the **Calls in Record** button to view a summary graph of all the RTP flows in the call history.



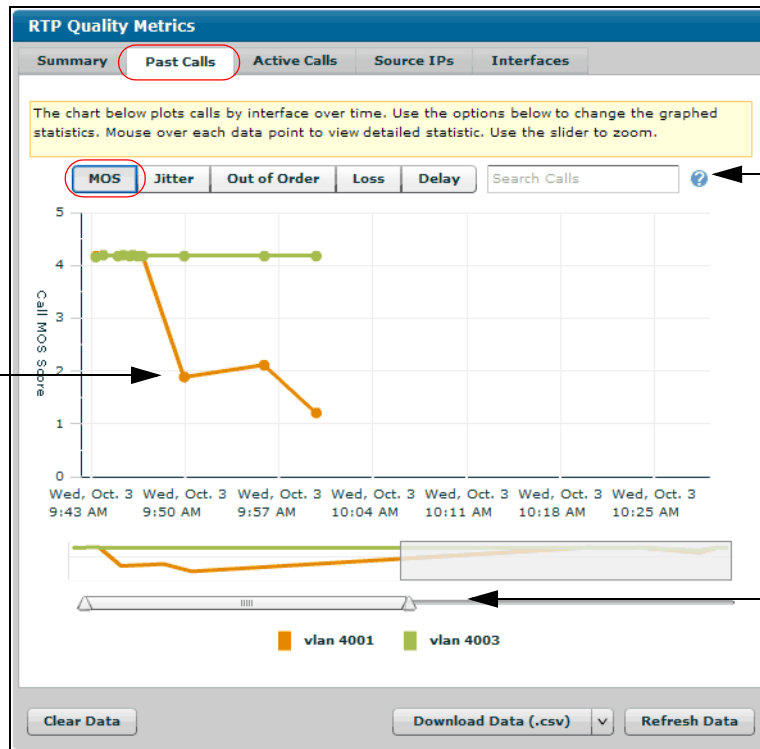
Select the **Active Calls** button to view a summary graph of the currently active RTP flows on the unit.



- Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Past Calls** tab to view the calls stored in the call history. **Past Calls** statistics can be sorted by MOS, jitter, out-of-order packets, lost packets, and delay. You can view the past calls of a specific interface by selecting the color-coded interface below the graph. Each data point on the graph provides detailed statistics of the stored call, and the slider bar can be used to zoom in and out on the data points plotted on the chart.

Select the **MOS** button on the **Past Calls** tab to view the statistics for MOS scores on the stored calls.

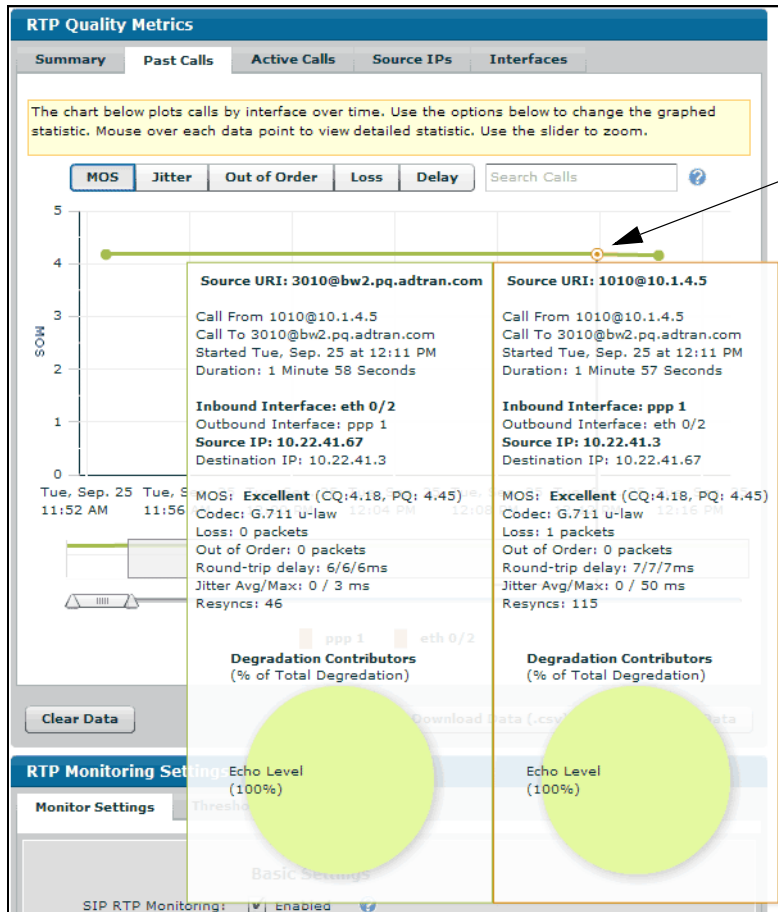
Use your mouse to hover over any data point throughout the **RTP Quality Metrics** menu tabs to see detailed information pertaining to that specific call.



Use the search field to sort a large number of data points. Use your mouse to hover over the question mark symbol for examples of multiple term searches and valid filters.

Use the slider bar with color-coded interfaces to filter RTP streams (by time).

Use your mouse to hover over any data point to view specific statistics on the graph. Use your mouse to select the data point for more detailed statistical information.



Hover over any data point to view statistics on inbound and outbound RTP streams. Click on the data point for more details.

The more detailed information provides source URI information, basic VQM statistics, and external delay sources for the call, as well as call quality. Use the **Back to Chart** button to return to the **Past Calls** menu and use the **Extended** button to view additional statistics on the data point.

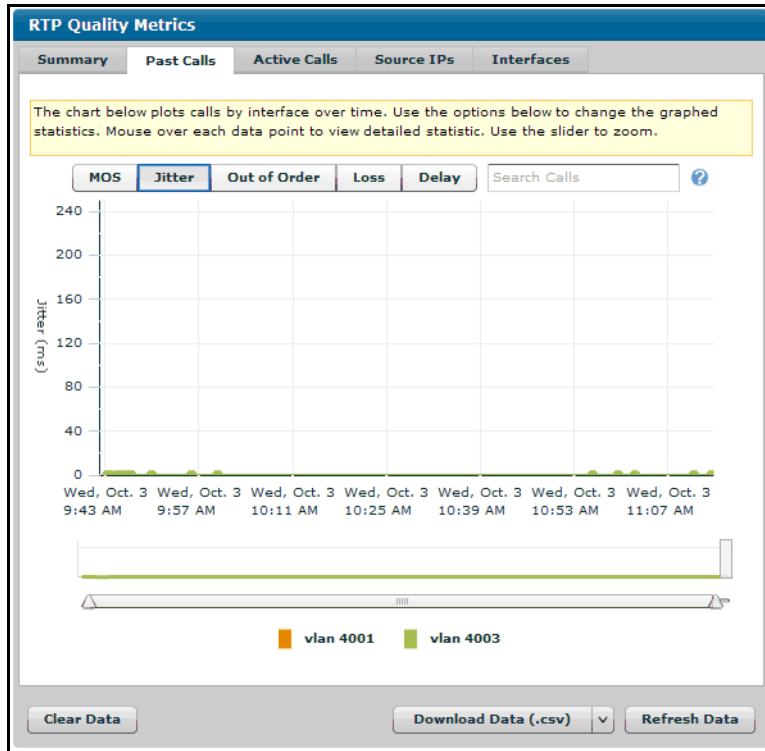
The screenshot displays the 'RTP Quality Metrics' window with the 'Active Calls' tab selected. The call identifier is '1015@10.1.4.5 to 3015@bw2.pq.adtran.com'. The interface is split into two columns, each showing detailed metrics for a specific RTP data point.

Selected RTP Data Point	Corresponding RTP Data Point
<b>Source URI: 1015@10.1.4.5</b>	<b>Source URI: 3015@bw2.pq.adtran.com</b>
Inbound Interface: ppp 1 Outbound Interface: eth 0/2	Inbound Interface: eth 0/2 Outbound Interface: ppp 1
Source IP: 10.22.41.2:2224 Destination IP: 10.22.41.68:2226	Source IP: 10.22.41.68:2226 Destination IP: 10.22.41.2:2224
<b>Basic Statistics:</b> Jitter (PDV) (avg/max): 0 / 360 ms Local Delay (min/avg/max): -4 / 486 / 999 ms Codec: G.711 u-law DSCP: 46	<b>Basic Statistics:</b> Jitter (PDV) (avg/max): 0 / 361 ms Local Delay (min/avg/max): 0 / 2 / 57 ms Codec: G.711 u-law DSCP: 46
<b>External delay sources (inst/avg/max):</b> Round-trip delay: 11 / 11 / 11 ms One-way delay: 50 / 57 / 108 ms Origination delay: NA / NA / NA ms Termination delay: 30 / 43 / 145 ms	<b>External delay sources (inst/avg/max):</b> Round-trip delay: 9 / 9 / 9 ms One-way delay: 49 / 59 / 107 ms Origination delay: NA / NA / NA ms Termination delay: 30 / 49 / 145 ms
<b>Quality:</b> R (LQ/CQ/Nominal): 65 / 64 / 93 MOS (LQ/CQ/Nominal): 3.21 / 3.17 / 4.20	<b>Quality:</b> R (LQ/CQ/Nominal): 93 / 92 / 93 MOS (LQ/CQ/Nominal): 4.2 / 4.18 / 4.20

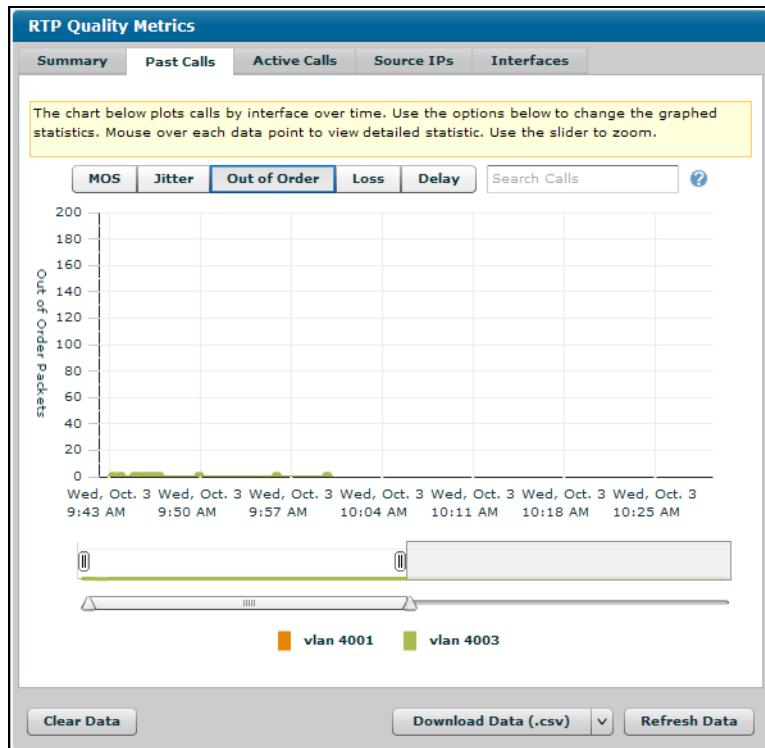
At the bottom of the window, there are three buttons: 'Clear Data', 'Download Data (.csv)', and 'Refresh Data'.



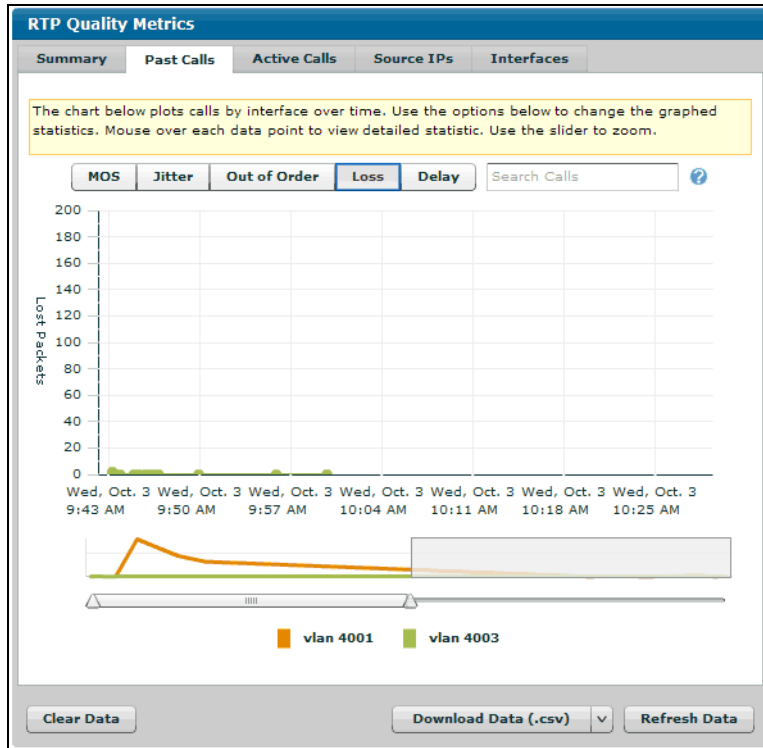
Select the **Jitter** button to view a chart that displays the jitter experienced on the RTP streams of the past calls.



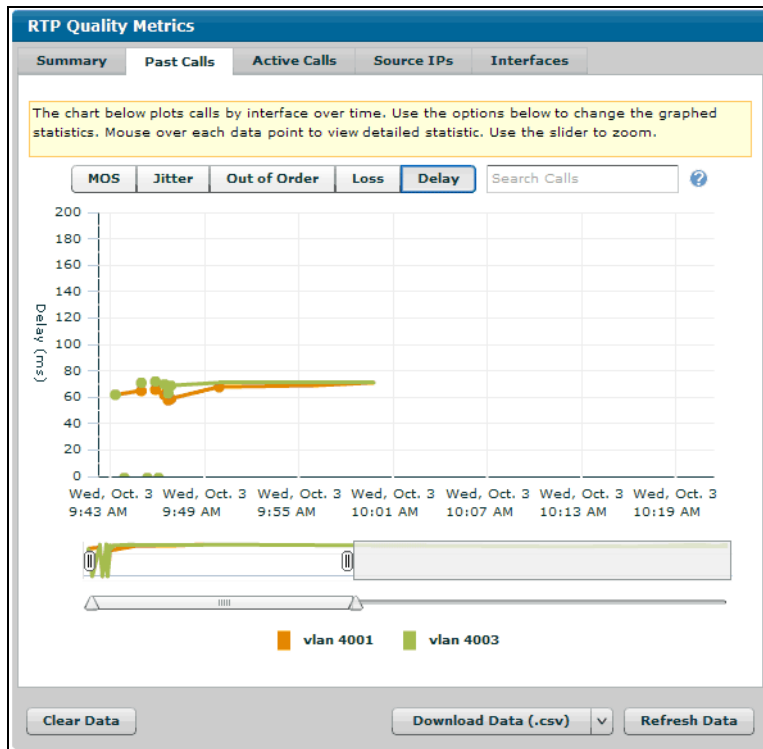
Select the **Out of Order** button to view a chart that displays the number of packets received out of order on the RTP streams of the past calls.



Select the **Loss** button to display a chart of the lost packets recorded on each past call.

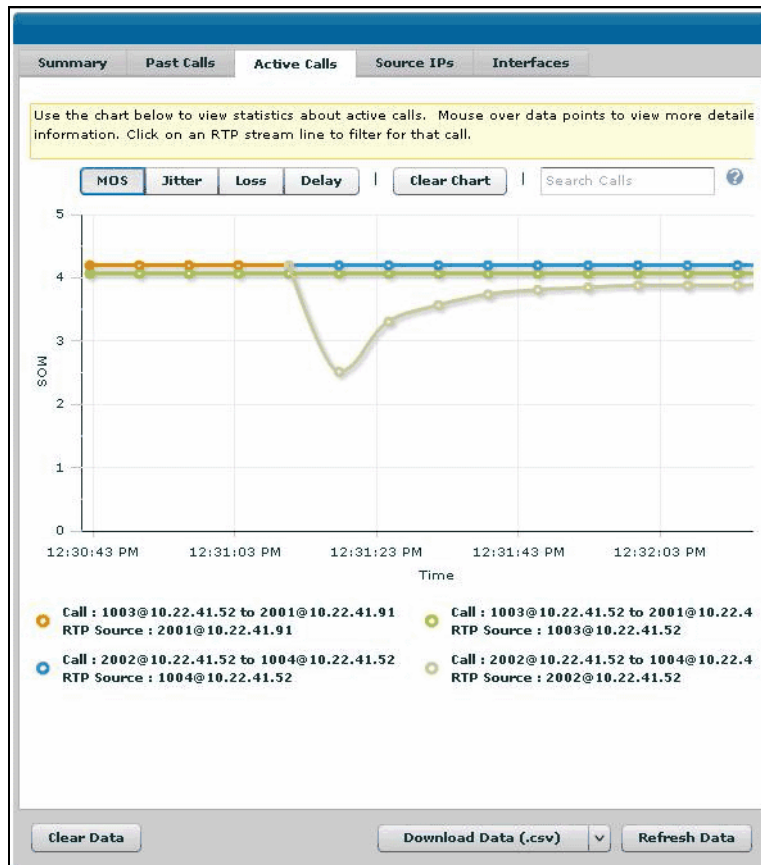


Select the **Delay** button to view a chart of the delay that occurred on each past call.

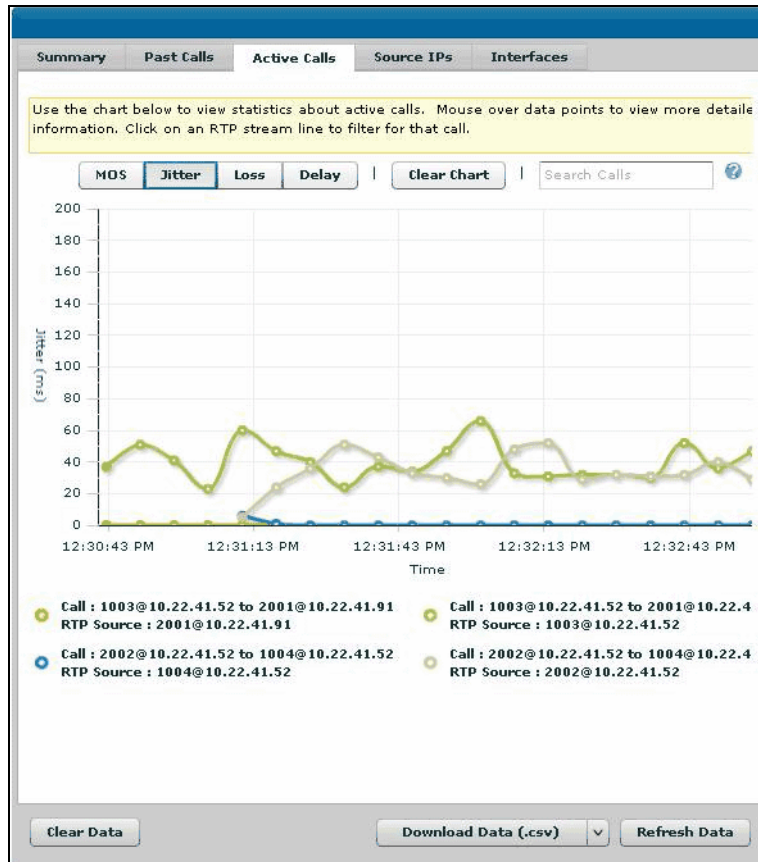


3. Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Active Calls** tab to view the live RTP streams passing through the AOS unit. **Active Calls** statistics can be sorted by MOS, jitter, lost packets, and delay. Use your mouse to hover over the data points to view more detailed information. All the active call metrics are the same as past call metrics, but they are for live RTP streams and are updated every 7 seconds. Once the call ends, it will appear in the past calls log, depending on the configured past calls threshold.

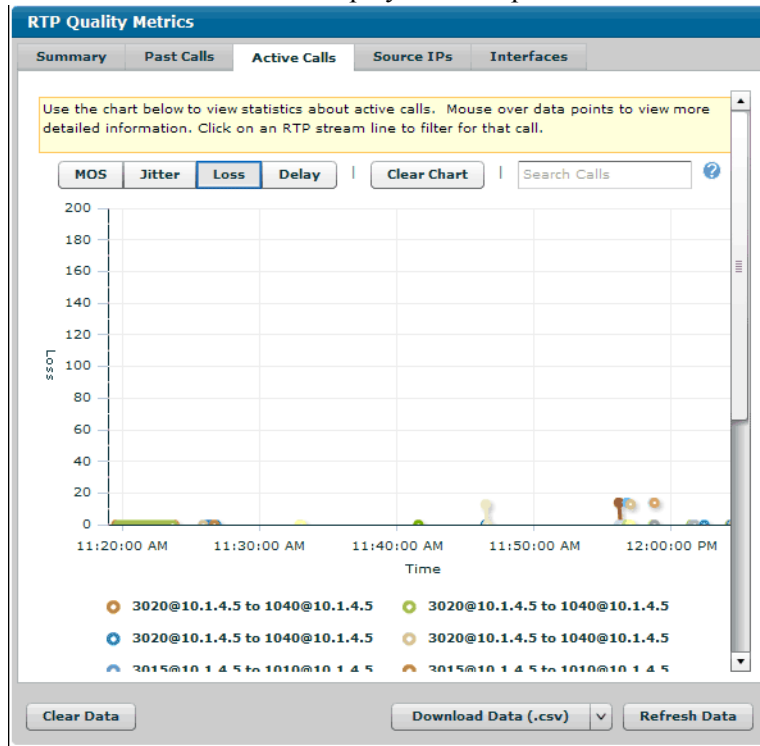
Select the **MOS** button to view a chart of MOS statistics on all live RTP streams.



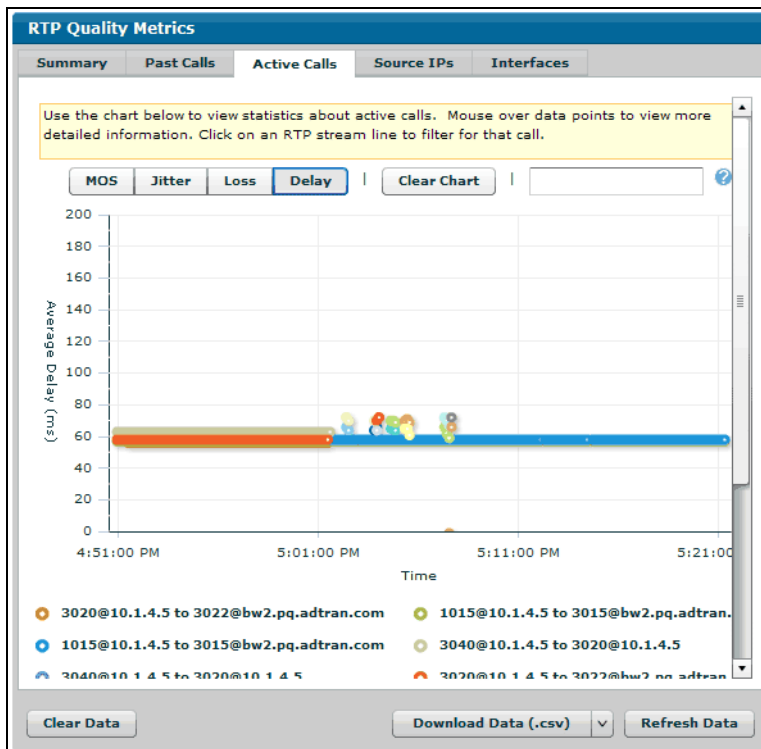
Select the **Jitter** button to view a chart that displays the jitter experienced on the live RTP streams.



Select the **Loss** button to view a chart that displays the lost packets detected on the live RTP stream.

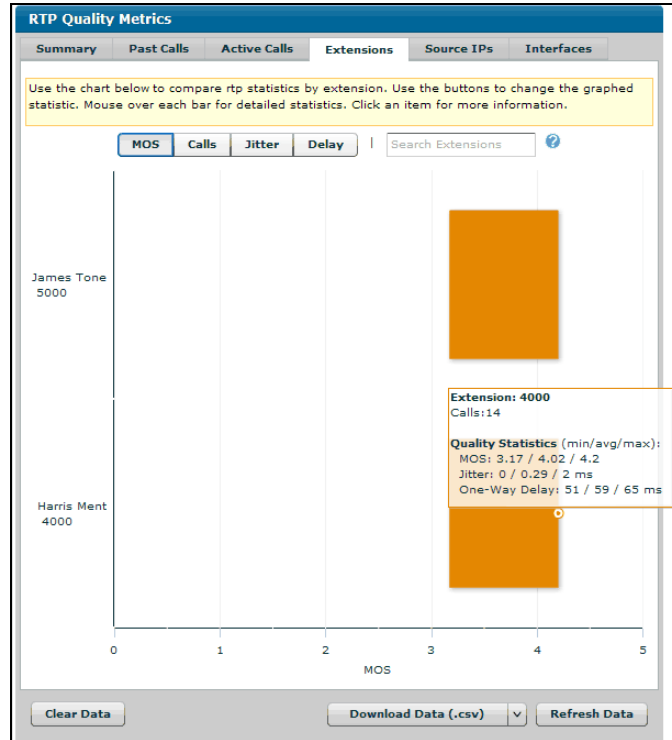


Select the **Delay** button to view a chart that displays the delay occurring on live RTP streams.

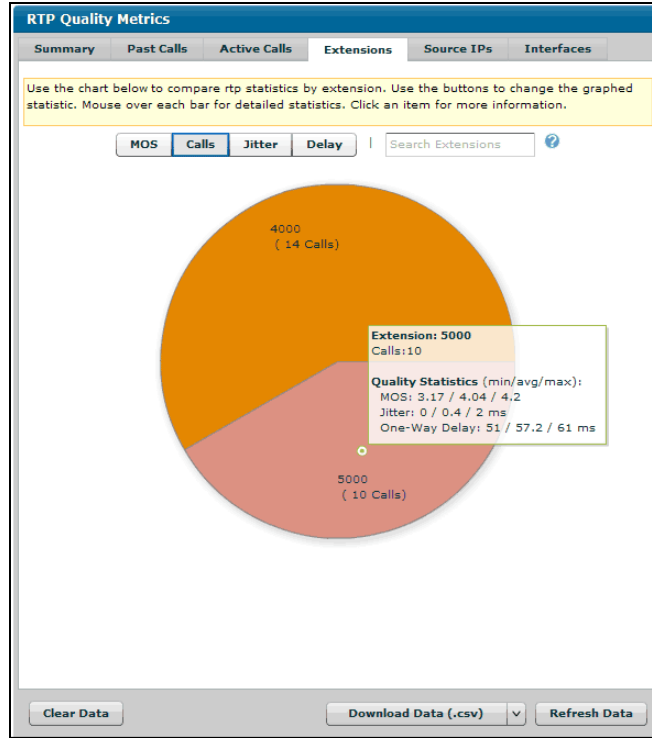


4. Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Extensions** tab to view the RTP streams for various user extensions. This option is only available in AOS voice products. **Extensions** statistics can be sorted by MOS, calls, jitter, and delay. Use your mouse to hover over the data points to view more detailed information.

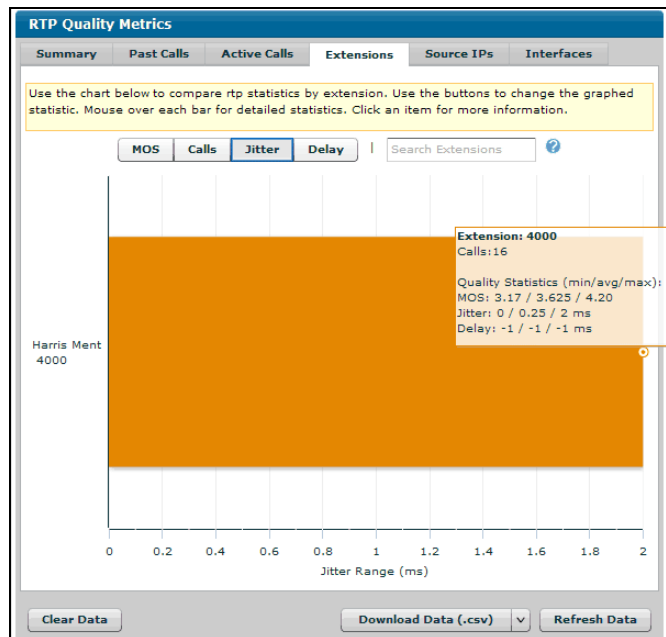
Select the **MOS** button to view a chart of MOS statistics on all monitored extensions.



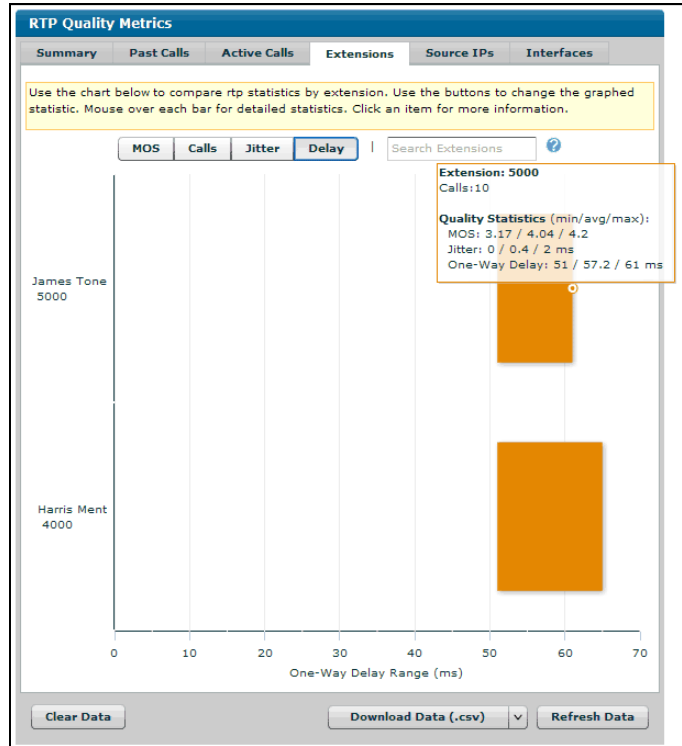
Select the **Calls** button to view a chart of calls sorted by each extension being monitored.



Select the **Jitter** button to view a chart that illustrates jitter experienced on the RTP streams of the extensions being monitored.



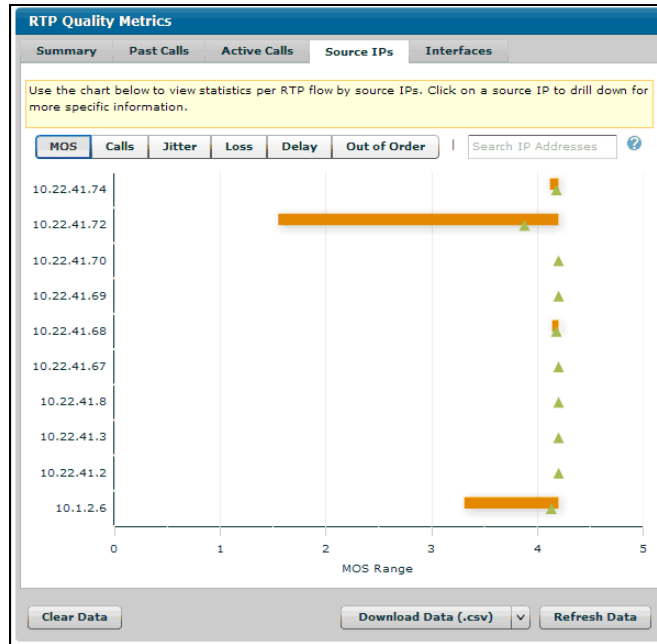
Select the **Delay** button to view a chart of the delay that is occurring on the extensions being monitored.



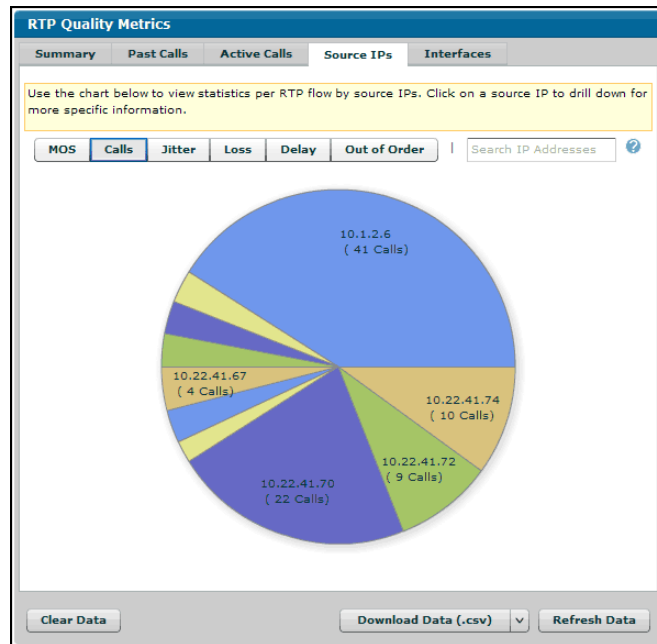


- Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Source IPs** tab to view VQM statistical data sorted by source IP address. **Source IPs** statistics can be sorted by MOS, calls, jitter, lost packets, delay, and out-of-order packets. Use your mouse to hover over the data points to view more detailed information.

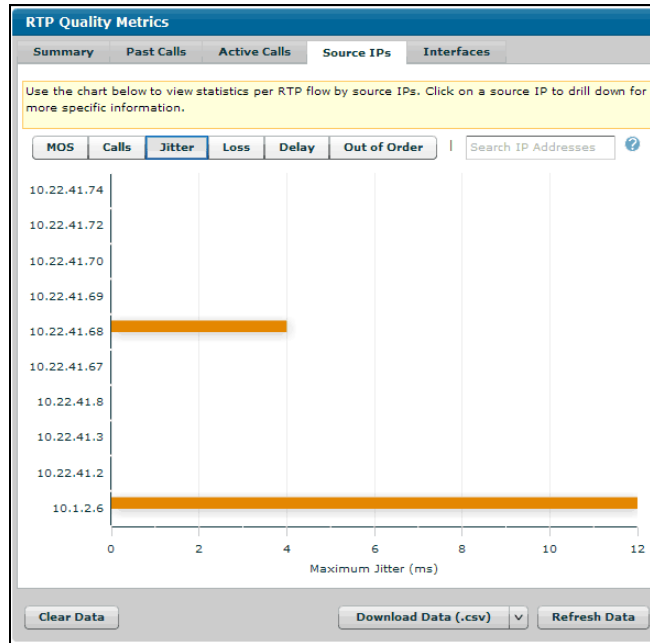
Select the **MOS** button to view a chart of MOS statistics on all source IP RTP streams.



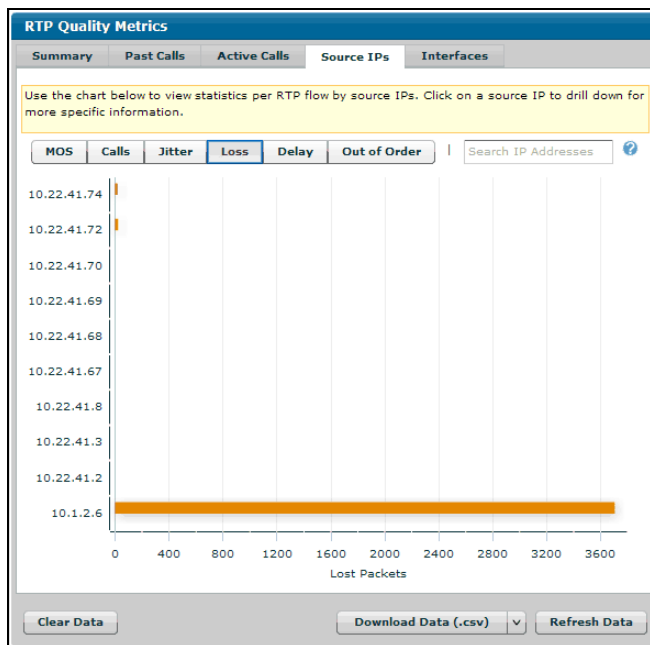
Select the **Calls** button to view a pie chart of the number of calls per source IP address.



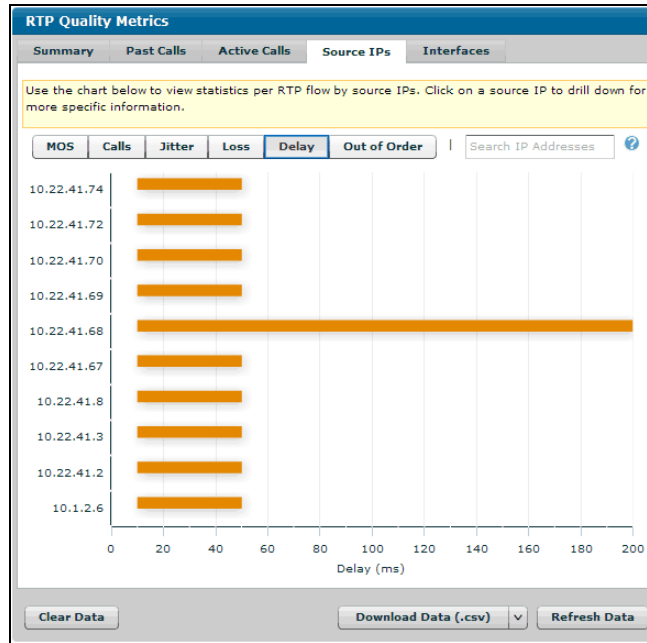
Select the **Jitter** button to view a chart that displays the jitter on RTP streams sourced from the given IP addresses.



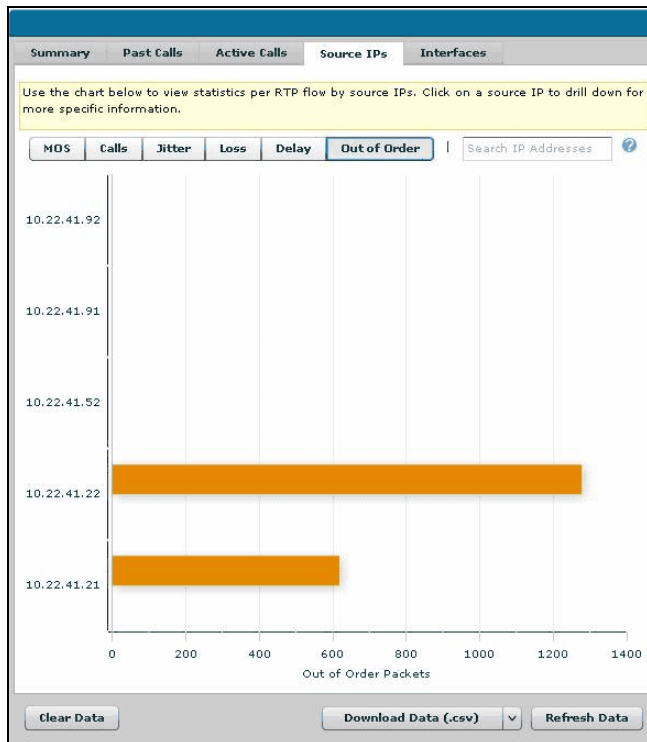
Select the **Loss** button to view a chart that displays any lost packets as they occur during voice calls on the specific source IP addresses.



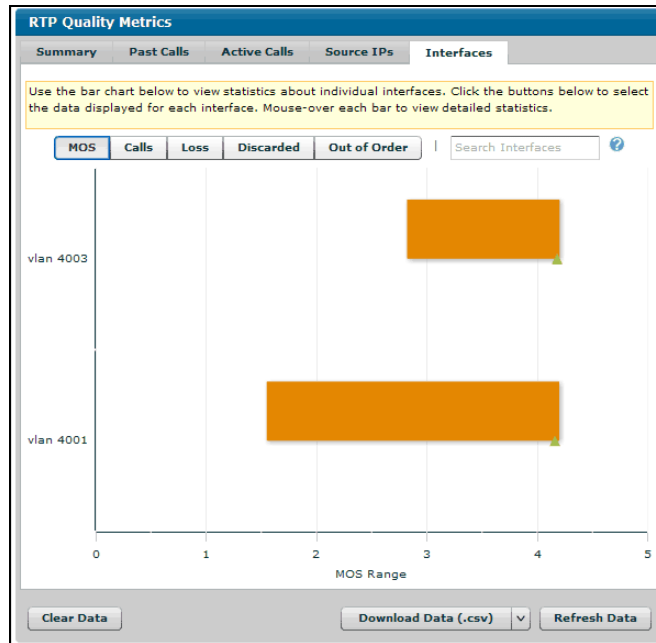
Select the **Delay** button to view a chart of delay experienced on the RTP streams sourced from the given IP addresses.



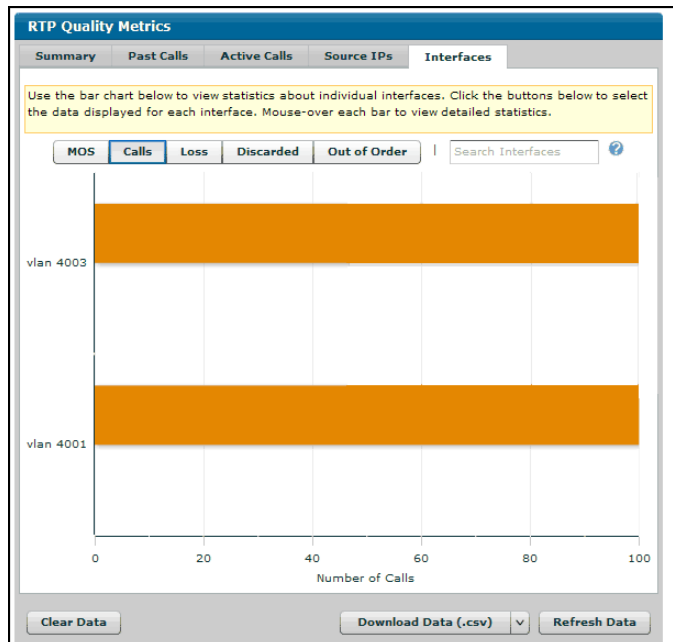
Select the **Out of Order** button to view a chart that displays packets received out of order on the RTP streams of the source IP addresses.



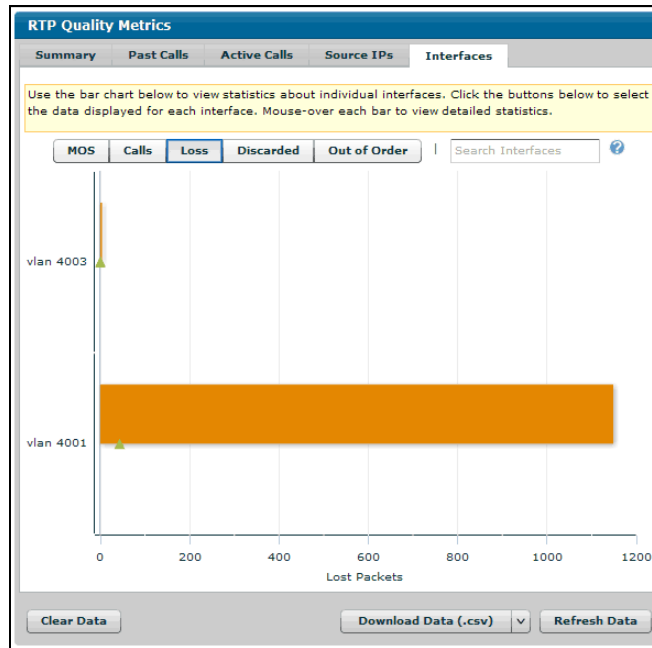
- Navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Interfaces** tab to view VQM statistical data sorted by interface. **Interfaces** statistics can be sorted by MOS, calls, lost packets, discarded packets, and out-of-order packets. Use your mouse to hover over the data points to view more detailed information. Select a bar on any graph to view additional statistics. Select the **MOS** button to view a chart of MOS statistics on specific interfaces.



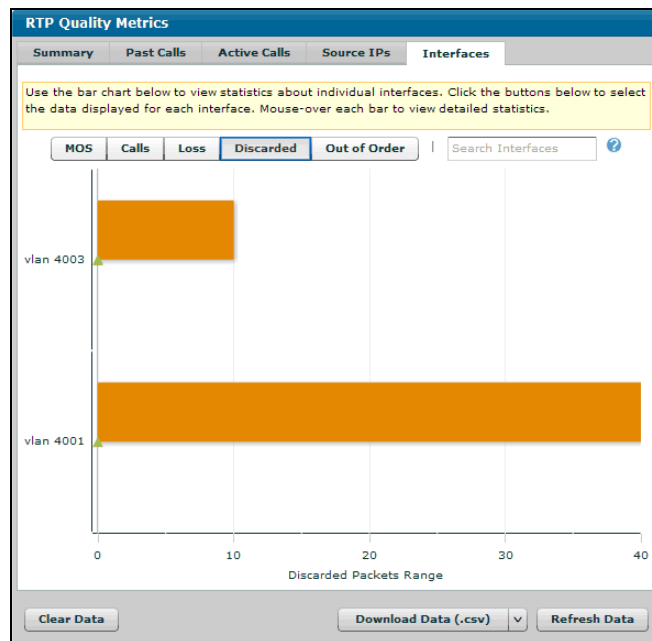
Select the **Calls** button to view bar graphs of the calls for specific interfaces.



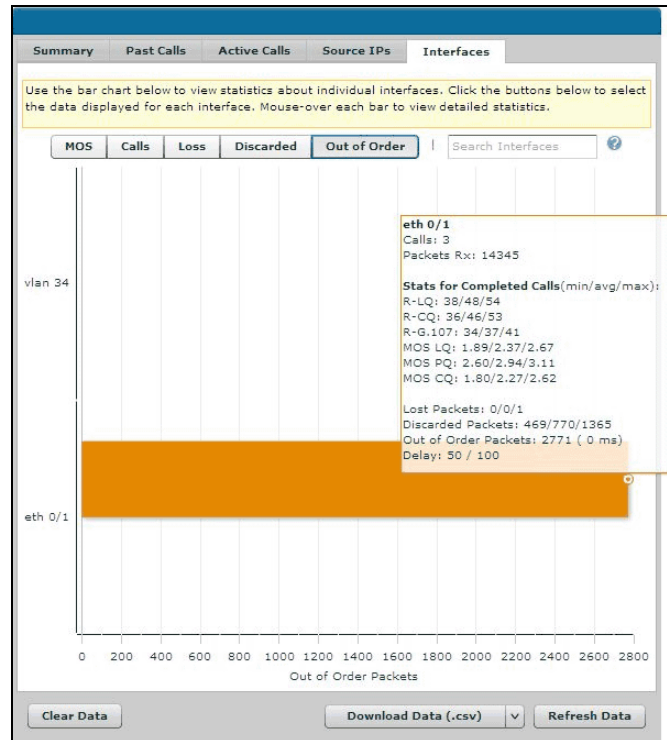
Select the **Loss** button to view the bar graphs of lost packets that occurred on specific interfaces.



Select the **Discarded** button to view bar graphs of discarded packets on specific interfaces.



Select the **Out of Order** button to view a chart that displays the packets received out of order on the RTP streams of specific interfaces.



## Viewing VQM Statistics Using the CLI

VQM statistics provide a method for determining the cause and network location of quality issues on VoIP networks. VQM statistics can be viewed in a summary format of all calls, past calls (calls logged in the call history), active calls, source IP addresses, and interfaces. Each of these statistical views can then be sorted by MOS, jitter, out-of-order packets, lost packets, or delayed packets. Summary views can be sorted by all calls, calls in the record, and active calls. The following section describes how to access these statistics from the CLI and gives an example of the type of output you can expect from the commands.

To view the VQM statistics using the CLI, follow these steps:

1. To view a summary of VQM statistics, enter the **show rtp quality-monitoring** command from the Enable mode as follows:

**#show rtp quality-monitoring**

Voice Quality Monitoring is ENABLED

Jitter Buffer: adaptive 10/50/100 ms (min/nominal/max)

Quality	Active Streams	Call History	All Streams	MOS range
Excellent	60	50	13462	4.40 - 4.00
Good	20	37	3456	3.99 - 3.60
Fair	3	1	45	3.59 - 2.60
Poor	1	7	7	2.59 - 0.00
Totals:	84	95	16970	

(Note: Statistics for All Streams are updated at call completion and do not include currently active streams. Call history statistics are available for up to 100 streams.)

History thresholds:

MOS (LQ/CQ/PQ): 3.0/3.0/3.0

Loss: 20 pkts

Out-of-order packets: 300 pkts

Jitter: 300 ms

Notification thresholds:

Metric	Info	Notice	Warning	Error
PQ MOS	4.4	4.0	3.6	2.6
LQ MOS	4.4	4.0	3.6	2.6
Loss	N/A	25	50	100 pkts
Out-of-order	N/A	N/A	50	100 pkts
Jitter	N/A	20	100	300 ms

Traps: ENABLED at Notice

Traps sent: 14

2. To view VQM statistics about past calls (calls recorded in the call history), enter the **show rtp quality-monitoring call-history [call-id] [to-uri <string>] [from-uri <string>] [source-uri <string>] [sort-by [loss | out-of-order | jitter | pq-mos | lq-mos]] [degradation] [detail]** command from the Enable mode. These statistics can be sorted by MOS, jitter, or lost or out-of-order packets by entering the **sort-by** parameter and the sorting metric (**loss**, **out-of-order**, **jitter**, **pq-mos**, and **lq-mos** parameters). The **sort-by** parameter displays the calls with the lowest call quality statistics first. Use the **call-id** parameter to display call history statistics based on a substring of the Call-ID. Use the **to-uri** parameter to display call history based on a substring of the URI or extension from which this RTP stream is sourced. Use the **from-uri** parameter to display call history statistics based on a substring of the From URI. Use the **source-uri** parameter to display call history statistics based on a substring of the URI/extension from which the RTP stream is sourced. Use the **degradation** parameter to display possible sources of voice quality degradation. Using the **detail** parameter displays all available

statistics. By default, only the most commonly used statistics are shown. This command will still display call history statistics even if VQM is disabled. The statistics for the call history must be deleted manually. In the following example, the past call statistics are sorted by lost packets.

**#show rtp quality-monitoring call-history sort-by loss**

Displaying 30 estimated stream statistics from 15 completed calls

RTP stream: 3.3.3.3 : 50000, ppp 1 -> 1.1.1.1 : 3000, vlan 1

To: 5551234@voip.com

Call-start (duration): 11 Apr 2010 20:58:39 (93 s)

MOS LQ: 3.800

MOS PQ: 3.800

Loss: 413 pkts

Out-of-order: 30 pkts

CODEC: g711

RTP stream: 1.1.1.1 : 3000, vlan 1 -> 3.3.3.3: 50000, ppp 1

To: 5551234@voip.com

From: 5551235@voip.com

Call-start (duration): 11 Apr 2010 20:58:39 (93 s)

MOS LQ: 3.800

MOS PQ: 3.800

Loss: 0 pkts

Out-of-order: 0 pkts

CODEC: g711

--MORE--

3. To view VQM statistics for active calls, enter the **show rtp quality-monitoring active-calls [call-id] [to-uri <string>] [from-uri <string>] [source-uri <string>] [sort-by [loss | out-of-order | jitter | pq-mos | lq-mos] [degradation]] [detail]** command from the Enable mode. These statistics can be sorted by MOS, jitter, or lost or out-of-order packets by entering the **sort-by** parameter and the sorting metric (**loss**, **out-of-order**, **jitter**, **pq-mos**, and **lq-mos** parameters). The **sort-by** parameter displays the calls with the lowest call quality statistics first. Use the **call-id** parameter to display call history statistics based on a substring of the Call-ID. Use the **to-uri** parameter to display call history based on a substring of the URI or extension from which this RTP stream is sourced. Use the **from-uri** parameter to display call history statistics based on a substring of the From URI. Use the **source-uri** parameter to display call history statistics based on a substring of the URI/extension from which the RTP stream is sourced. Use the **degradation** parameter to display possible sources of voice quality degradation. Using the **detail** parameter displays all available statistics. By default, only the most commonly used statistics are shown. In the following example, active call statistics are sorted by loss.



**#show rtp quality-monitoring active-calls sort-by loss**

Displaying 30 estimated stream statistics from 15 completed calls

RTP stream: 3.3.3.3 : 50000, ppp 1 -> 1.1.1.1 : 3000, vlan 1

To: 5551234@voip.com

Call-start (duration): 11 Apr 2010 20:58:39 (93 s)

MOS LQ: 3.800

MOS PQ: 3.800

Loss: 413 pkts

Out-of-order: 30 pkts

Jitter (max): 20 ms

CODEC: g711

RTP stream: 1.1.1.1 : 3000, vlan 1 -> 3.3.3.3: 50000, ppp 1

To: 5551234@voip.com

From: 5551235@voip.com

Call-start (duration): 11 Apr 2010 20:58:39 (93 s)

MOS LQ: 3.800

MOS PQ: 3.800

Loss: 0 pkts

Out-of-order: 0 pkts

Jitter (max): 1 ms

CODEC: g711

--MORE--

4. To view VQM statistics for voice endpoints (the source IP address of the RTP stream), enter the **show rtp-quality monitoring endpoints [sort-by [loss | out-of-order | jitter | mos-pq | mos-lq]] [summary]** command from the Enable mode. These statistics can be sorted by MOS, jitter, or lost or out-of-order packets by entering the **sort-by** parameter and the sorting metric (**loss**, **out-of-order**, **jitter**, **mos-pq**, and **mos-lq** parameters). The **sort-by** parameter displays the calls with the lowest call quality statistics first. Using the **summary** parameter displays a summary of VQM voice endpoint statistics. By default, only the most commonly used statistics are shown. This command will still display endpoint VQM statistics (based on call history) even when VQM is disabled. In the following example, a summary of the endpoint statistics is shown.

**#show rtp-quality monitoring endpoints summary**

Displaying 2 estimated endpoint statistics from 50 completed calls

RTP source: 3.3.3.3, ppp 1

Quality	Completed Calls	MOS range
Excellent	0	4.40 - 4.00
Good	0	3.99 - 3.60
Fair	24	3.59 - 2.60
Poor	3	2.59 - 0.00
Totals:	27 (of the last 100 recorded calls)	

RTP source: 5.5.5.5, ppp 1

5. To view VQM statistics for a particular interface, enter the **show rtp quality-monitoring interface** [*<interface>*] [**sort-by** [**loss** | **out-of-order** | **jitter** | **mos-pq** | **mos-lq**]] [**summary**] command from the Enable mode. These statistics can be sorted by MOS, jitter, or lost or out-of-order packets by entering the **sort-by** parameter and the sorting metric (**loss**, **out-of-order**, **jitter**, **mos-pq**, and **mos-lq** parameters). The **sort-by** parameter displays the calls with the lowest call quality statistics first. Using the **summary** parameter displays a summary of VQM voice endpoint statistics. By default, only the most commonly used statistics are shown. Specifying an interface is optional; if you do not specify an interface, the statistics for all interfaces on which VQM is enabled are displayed. If you want to see the statistics for a particular interface, specify the interface in the format *<interface type [slot/port | slot/port.subinterface id | interface id | interface id.subinterface id]>*. For example, for an Ethernet subinterface, use **eth 0/1.1**; for a PPP interface, use **ppp 1**; and for an ATM subinterface, use **atm 1.1**. In the following example, a summary of all the interface VQM statistics is shown.

**#show rtp quality-monitoring interface summary**

ppp 1

Quality	Completed Calls	MOS range
-----	-----	-----
Excellent	0	4.40 - 4.00
Good	2	3.99 - 3.60
Fair	34	3.59 - 2.60
Poor	3	2.59 - 0.00
-----	-----	-----
Totals:	37 (of the last 100 recorded calls)	

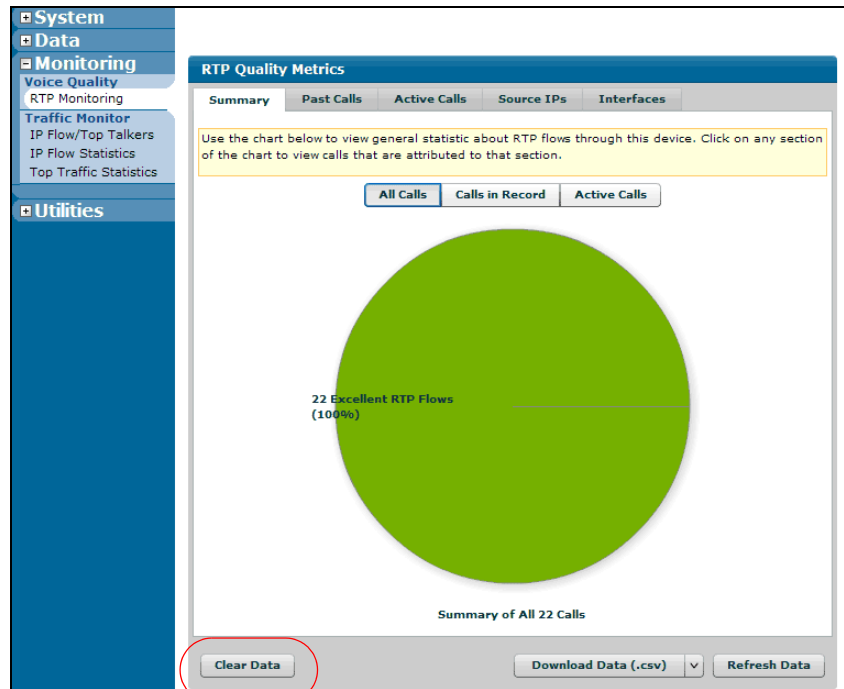
vlan 1

Quality	Completed Calls	MOS range
-----	-----	-----
Excellent	36	4.40 - 4.00
Good	1	3.99 - 3.60
Fair	0	3.59 - 2.60
Poor	0	2.59 - 0.00
-----	-----	-----
Totals:	37 (of the last 100 recorded calls)	

## Clearing VQM Statistics

VQM call history and endpoint statistics must be cleared manually, and VQM statistics can be cleared using either the GUI or the CLI. When using the GUI to clear statistics, *all* VQM statistics are cleared, not just the call history and endpoint statistics. To clear statistics, follow these steps:

1. To clear VQM statistics using the GUI, connect to the GUI and navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Clear Data** button to clear *all* VQM statistics on the system.



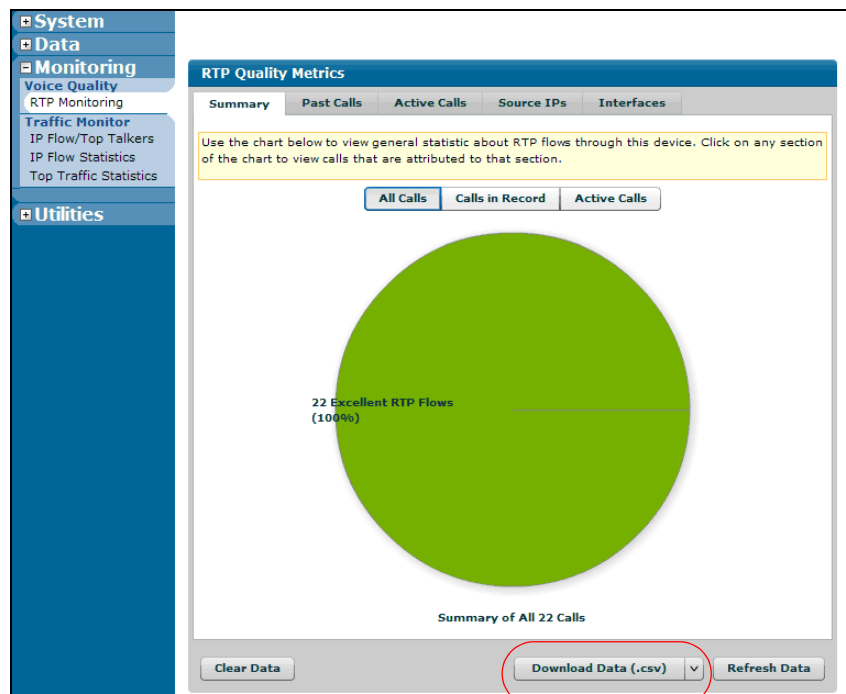
2. To clear VQM statistics using the CLI, enter the **clear ip rtp quality-monitoring [interface <interface> | call-history | reporter]** command from the Enable mode. You can specify that only the statistics for a single interface are cleared by using the **interface <interface>** command. Specify the interface in the format **<interface type [slot/port | slot/port.subinterface id | interface id | interface id.subinterface id]>**. For example, for an Ethernet subinterface, use **eth 0/1.1**; for a PPP interface, use **ppp 1**; and for an ATM subinterface, use **atm 1.1**. You can also specify that only the call history statistics are cleared using the **call-history** parameter. You can also specify that only the statistics used by a reporter are cleared using the **reporter** parameter. If no additional parameters are entered, the command clears all calls from the call history and all statistics for all interfaces. Enter the command as follows:

```
#clear ip rtp quality-monitoring interface ppp 1
```

## Exporting VQM Statistics

VQM statistics can be exported as CSV or XML files. This option is available only using the GUI. To export VQM statistics, follow these steps:

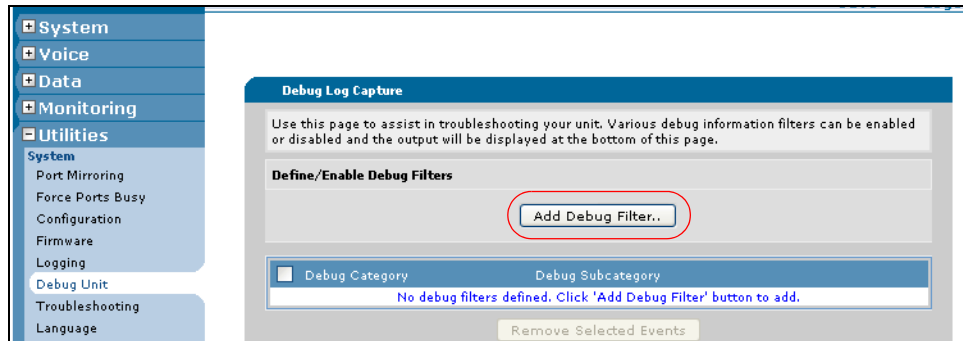
1. To export VQM statistics using the GUI, connect to the GUI and navigate to **Monitoring > Voice Quality > RTP Monitoring > RTP Quality Metrics**. Select the **Download Data (.csv)** button to export VQM statistics as a CSV file, or select **Download Data (.xml)** from the download drop-down menu to export VQM statistics as an XML file. Use the glossary included as [Appendix A: Glossary of VQM Statistics on page 65](#) in this document to help you read the exported statistics.



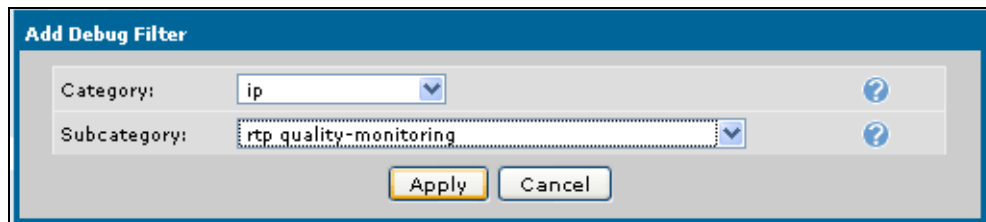
## Troubleshooting VQM Configuration

You can troubleshoot your VQM configuration using either the GUI or the CLI. Troubleshooting VQM relies on debug events and messages to verify your configuration. To troubleshoot VQM using the debug feature, follow these steps:

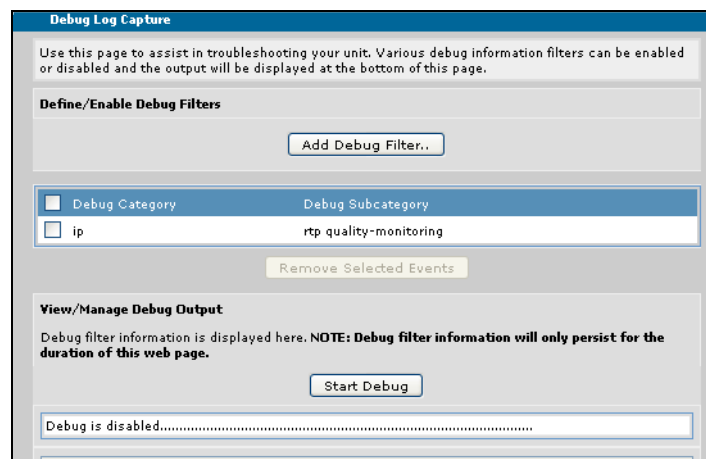
1. To debug VQM using the GUI, connect to the GUI and navigate to **Utilities > Debug Unit** and select **Add Debug Filter**.



In the **Add Debug Filter** menu that appears, select **ip** from the **Category** drop-down menu and **rtp quality-monitoring** from the **Subcategory** drop-down menu. Then select **Apply**.



The VQM debug filter appears in the debug filter list. You can start VQM debugging by selecting **Start Debug**. To end a debug session, select **Stop Debug**. To remove the VQM debug filter, select the check box next to the debug category, and select **Remove Selected Events**.



- To debug VQM using the CLI, enter the **debug ip rtp quality-monitoring [packets [rtp | rtcp]] [round-trip-delay]** command from the Enable mode. This command displays debug events for the VQM configuration, the RTP or RTCP packets that travel through the unit, or the round-trip-delay mechanism. To enable debug event messages for VQM, enter the command as follows:

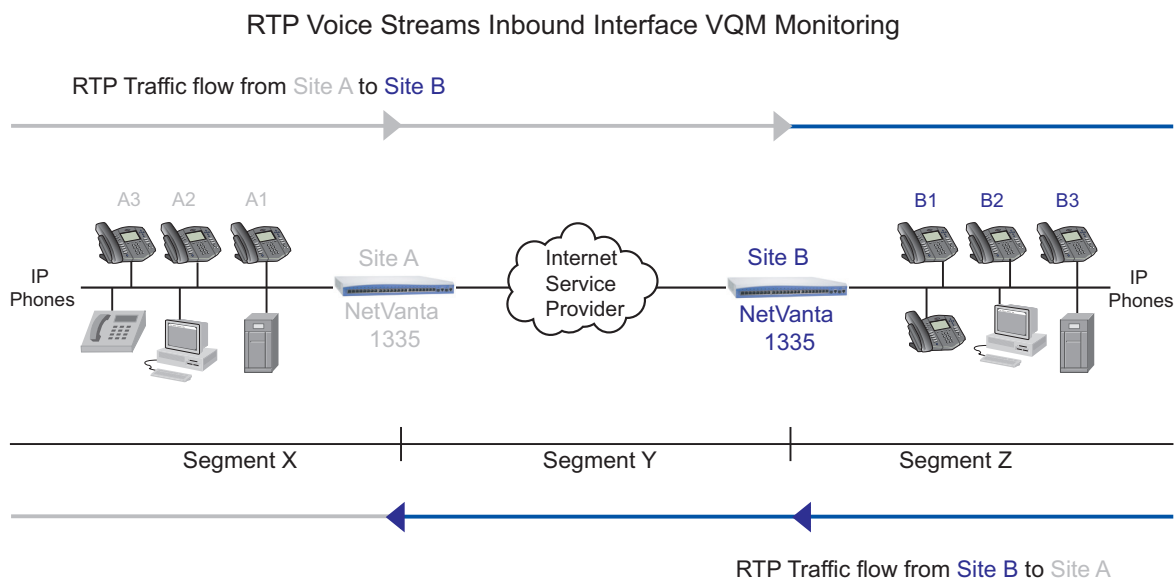
**#debug ip rtp quality-monitoring**



*Turning on a large amount of debug information can adversely affect the performance of your unit.*

## Using VQM to Troubleshoot VoIP Networks

VQM is primarily a troubleshooting feature. The example below illustrates using VQM to monitor network segment points and to help identify the cause of voice quality issues on the network based on the gathered reports (statistics). Three troubleshooting scenarios are provided based on this network diagram.



**Figure 3. RTP Voice Stream Monitoring Segment Points**

In [Figure 3](#), the RTP streams flow in both directions and both routers are using VQM. This figure illustrates the possible directions in the network to monitor in order to resolve detected voice quality issues or to narrow down the troubleshooting to a specific segment.

The router at Site A monitors the inbound stream along Segment Y plus Segment Z (blue) up to the WAN interface, and the RTP stream along Segment X (gray) up to the LAN interface. The router at Site B monitors the inbound RTP stream along Segment X plus Segment Y (gray) up to the WAN interface, and the RTP stream along Segment Z (blue) up to the LAN interface.

## Troubleshooting General VoIP Issues

If issues occur with the voice quality on calls flowing through your AOS unit(s), here are a few tips to help you find where the issue(s) is occurring:

1. Review the VQM statistics provided on the GUI, or issue RTP quality CLI **show** commands to identify the interface(s) and endpoint(s) with perceived voice quality issues.
2. If you discover poor RTP metrics on the WAN (public) interface packets, check the physical WAN interface for configuration and connectivity issues. If the LAN (private) interface does not show any errors, but the WAN interface is experiencing errors, contact your Internet Service Provider (ISP), report the errors, and request testing of their network to resolve the issues.
3. If you discover poor RTP metrics on the LAN (private) interface packets, check to see if the issues are specific to an endpoint. If so, test the endpoint for proper connectivity, functionality, and configuration. Also, check the interface configuration to make sure the interface negotiated the correct speed and duplex. If there are no errors, discards, or throttles and the configuration is correct, the AOS device is not causing the issues.

## Troubleshooting Lost Packets

The following is a troubleshooting scenario using VQM to troubleshoot packet loss. The scenario is based on [Figure 3 on page 62](#).

During a call between A1 and B1 in [Figure 3 on page 62](#), A1 experienced poor voice quality and reports it to the ISP. Further investigation by the ISP discovers lost and out-of-order packets on the public segment of the network (Segment Y). This leads the ISP technician to suspect either the ISP's uplink, the ISP's network, or a specific endpoint as the source of the problem. To check the WAN, the router at Site A is monitored, and to check the LAN, the router at Site B is monitored.

Check the public interface for errors, discards, or throttles to eliminate the AOS device as the source of any network issues. It is possible for the AOS device to drop packets due to interface errors, network load issues, or incorrect speed/duplex negotiations. If there are no dropped packets, errors, or throttles, and the speed and duplex settings are correct, the AOS device can be cleared in the troubleshooting exercise.

Next, the ISP technician can view the call history (**Past Calls**) to search for the source of the lost packets. Using the sort-by (loss) option, the technician can quickly identify specific endpoints with errors. If only one endpoint has errors, that particular endpoint or connection is faulty. If multiple endpoints have errors, the problem is most likely a general issue with the uplink to the ISP (network congestion due to provisioning or other errors). A **traceroute** can be performed from the AOS device to the endpoint to determine on which portion of the network the problem occurs.

## Troubleshooting Jitter

The following is a troubleshooting scenario using VQM to troubleshoot jitter. The scenario is based on [Figure 3 on page 62](#).

In [Figure 3 on page 62](#), several phone users at Site B are complaining of choppy voice connections. After viewing the VQM statistics for the inbound RTP traffic at Site B, the site technician discovers jitter problems on the public interface only. The cause of the jitter problem is narrowed down to the ISP's uplink, the ISP network, or a specific endpoint on the far end.

To determine the source of the jitter, the technician sorts by jitter the RTP streams stored in the call history (**Past Calls**) to determine if the public network (in general) is the cause, or if it is a specific endpoint within the network. If a specific endpoint has a problem, the jitter errors will only appear on that particular RTP source (endpoint). If the network in general is the cause, jitter errors will be present on multiple RTP sources (endpoints). A potential source of jitter is a device with misconfigured quality of service (QoS).

## Troubleshooting with Loopback Accounts

The loopback account feature is a diagnostic tool that allows the system administrator to set up a voice account similar to a user account. The loopback account will automatically answer any call received based on the number of rings. Calls can also be originated and terminated from the loopback account using the CLI commands. Active voice loopbacks will return RTP streams back to the remote site for monitoring. The system allows up to five simultaneously active loopback calls for troubleshooting purposes. In conjunction with VQM, the loopback account RTP packets from a remote phone or device are quickly analyzed. This process allows the administrator to test round-trip network connectivity from a remote site and judge the quality of the connection with the provided VQM metrics. Once the loopback call is placed, the system administrator can use the VQM feature to diagnose voice quality issues or listen to the quality of the voice echoed back on a call into the loopback account.



*The loopback account feature is only available in AOS voice products.*

For more information about loopback account configuration, refer to the technical note *AOS Voice Loopback Account* (article number 2363) available online at <http://kb.adtran.com>.



## Appendix A: Glossary of VQM Statistics

The following appendix provides a glossary of VQM statistics and terms. These terms appear when VQM statistics are exported to a CSV or XML file.

### Basic Statistics

The following are basic statistic terms you might see in the exported VQM statistic files.

**VqmRtpSourceIp** is the source IP address of this Realtime Transport Protocol (RTP) stream. If network address translation (NAT) is involved, private side addresses and ports are reported. For source NAT, the address and port before NAT are reported. For destination NAT, the address and port after NAT are reported.

**VqmRtpSourcePort** is the source port of this RTP stream. If NAT is involved, private side addresses and ports are reported. For source NAT, the address and port before NAT are reported. For destination NAT, the address and port after NAT are reported.

**VqmSourceIntName** is the interface at which the RTP stream arrived inbound to the unit; RTP is monitored at this inbound interface.

**VqmRtpDestIp** is the destination IP address of this RTP stream. If NAT is involved, private side addresses and ports are reported. For source NAT, the address and port before NAT are reported. For destination NAT, the address and port after NAT are reported.

**VqmRtpDestPort** is the destination port of this RTP stream. If NAT is involved, private side addresses and ports are reported. For source NAT, the address and port before NAT are reported. For destination NAT, the address and port after NAT are reported.

**VqmDestIntName** is the destination interface for this RTP stream.

**VqmRtpSourceUri** is the URI of the sender of the RTP stream. This is always either the To URI or the From URI. It may be the case (such as for an RTP stream from a music on hold server) that the source URI is incorrect. If so, it will correspond to the direction in which the URI is located in the network. If call signaling is not being monitored, this value will be unavailable.

**VqmTo** is the To URI from the Session Initiation Protocol (SIP) signaling or the extension receiving the call. If call signaling is not being monitored, this value may be unknown.

**VqmFrom** is the From URI from the SIP signaling, or the extension originating the call. If call signaling is not being monitored, this value may be unknown.

**VqmCallStart** is the time that monitoring began on this RTP stream.

**VqmCallDurationMs** is the time in milliseconds from when monitoring began on this RTP stream until the reception of the last RTP packet on this stream.

**VqmMosLq** is 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.

**VqmMosPq** is the listening quality MOS normalized to the PESQ scale. Users interested in scoring based on the ITU-T recommendation P.862 (PESQ) should use this value.

**VqmPktsLostTotal** is the total number of packets determined by the simulated jitter buffer to be lost.

**VqmOutOfOrder** is the total number of packets that arrive at the jitter buffer out of sequence.

**VqmPdvAverageMs** is the maximum instantaneous packet delay variation (PDV) for packets in the RTP stream, reported in milliseconds.

**VqmCodec** is the last voice coder-decoder (CODEC) detected in this RTP stream. Note that an endpoint may change the voice CODEC mid-stream.

**VqmResyncCount** is the total number of jitter buffer resynchronizations caused by discontinuous transmission (DTX), voice activity detection (VAD), or silence suppression.

**VqmDelayCurrentMsec** is the current jitter buffer delay in milliseconds, or in the case of an RTP stream in the call history, the last jitter buffer delay. (This does not apply to a fixed jitter buffer configuration.)

**VqmDelayMinMsec** is the minimum jitter buffer delay in milliseconds. (This does not apply to a fixed jitter buffer configuration.)

**VqmDelayMaxMsec** is the maximum jitter buffer delay in milliseconds. (This does not apply to a fixed jitter buffer configuration.)

## Detailed Statistics

The following are detailed statistic terms you might see in the exported VQM statistic files.

**VqmCallid** is 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.

**VqmCcmid** is an internally generated ID identifying this call.

**VqmSsrcid** is the synchronization source ID (SSRC) for this stream. RFC 3550 defines this as a randomly generated 32-bit number that is globally unique within an RTP session.

**VqmJitterMaximum** is the maximum instantaneous packet delay variation (PDV) for packets in the RTP stream, reported in milliseconds.

**VqmBitrate** is the actual bitrate of the RTP stream, calculated using the size of each RTP packet in bits and the duration of audio represented in each packet. This value can give an idea of how much bandwidth is required for this RTP stream.

**VqmDscp** is the differentiated services code point (DSCP) value recorded in the IP header of RTP packets in this stream. The DSCP displayed is taken from the first RTP packet in this stream, and is refreshed whenever the voice CODEC changes. This value can be useful when troubleshooting QoS in the path of the RTP stream.

## Delay Statistics

The following are delay statistic terms you might see in the exported VQM statistic files.

**VqmRtDelayInst** is 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 (e.g., a firewall in the path did not allow the traffic), this statistic will be reported as unavailable.

**VqmRtDelayAverage** is the average round-trip delay. This is obtained from all available information, including RTCP XR and SR reports and 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 (e.g., a firewall in the path did not allow the traffic), this statistic will be reported as unavailable.

**VqmRtDelayMaximum** is the maximum round-trip delay. This is obtained from all available information, including RTCP XR and SR reports and 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 (e.g., a firewall in the path did not allow the traffic), this statistic will be reported as unavailable.

**VqmOnewayDelayInst** is the instantaneous one-way delay, including any delay that can be introduced by the jitter buffer. The calculation for this statistic assumes that round-trip delay is symmetric. If round-trip delay is unavailable, this statistic will also be unavailable.

**VqmOnewayDelayAverage** is the average one-way delay, including any delay that can be introduced by the jitter buffer. The calculation for this statistic assumes that round-trip delay is symmetric. If round-trip delay is unavailable, this statistic will also be unavailable.

**VqmOnewayDelayMaximum** is the maximum one-way delay, including any delay that can be introduced by the jitter buffer. The calculation for this statistic assumes that round-trip delay is symmetric. If round-trip delay is unavailable, this statistic will also be unavailable.

**VqmOrigDelayInst** is the instantaneous origination point end-system delay. This value is obtained from RTCP XR reports. If no reports are available, this statistic will be unavailable.

**VqmOrigDelayAverage** is the average origination point end-system delay. This value is obtained from RTCP XR reports. If no reports are available, this statistic will be unavailable.

**VqmOrigDelayMaximum** is the maximum origination point end-system delay. This value is obtained from RTCP XR reports. If no reports are available, this statistic will be unavailable.

**VqmTermDelayMinimum** is the instantaneous termination point end-system delay. Computation of this delay involves delay introduced by the jitter buffer and voice CODEC-specific delay related to sampling and encoding.

**VqmTermDelayAverage** is the average termination point end-system delay. Computation of this delay involves delay introduced by the jitter buffer and voice CODEC-specific delay related to sampling and encoding.

**VqmTermDelayMaximum** is the maximum termination point end-system delay. Computation of this delay involves delay introduced by the jitter buffer and voice CODEC-specific delay related to sampling and encoding.

## Quality Statistics

The following are quality statistic terms you might see in the exported VQM statistic files.

**VqmRLq** is 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, since MOS scales may differ based on the CODEC type and region of deployment, whereas R factor measurements are consistent across CODECs and regions.

**VqmRCq** is 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.

**VqmRG107** is the G.107 R factor. Users interested in the R factor score based exclusively on the ITU G.107 E model should use this value. 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.

**VqmRNominal** is the nominal or generally accepted maximum obtainable R factor for this RTP stream. The nominal value may change based on the CODEC used. 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.

**VqmMosCq** is 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.

**VqmMosNominal** is the nominal, or generally accepted maximum obtainable MOS for this RTP stream. The nominal value may change based on CODEC used.

## Degradation Statistics

The following are degradation statistic terms you might see in exported VQM statistic files.

**VqmDegLoss** reports an estimated percentage of the overall degradation in quality caused by network packet loss.

**VqmDegDiscard** reports an estimated percentage of the overall degradation in quality caused by discards by the jitter buffer.

**VqmDegVocoder** reports an estimated percentage of the overall degradation in quality caused by the voice CODEC selection.

**VqmDegRecency** reports an estimated percentage of the overall degradation in quality caused by loss or discard recency. Participants in a conversation are likely to be more forgiving of quality degradation occurring near the beginning of a call that later resolves than they are to be forgiving of quality degradation occurring near the end of a call.

**VqmDegDelay** reports an estimated percentage of the overall degradation in quality caused by delay.

**VqmDegSiglvl** reports an estimated percentage of the overall degradation in quality caused by low speech energy signal level.

**VqmDegNoiselvl** reports an estimated percentage of the overall degradation in quality caused by high noise levels.

**VqmDegEcholvl** reports an estimated percentage of the overall degradation in quality caused by high echo levels.

## **Burst, Gap, and Loss Statistics**

The following are packet burst, gap, and loss statistic terms you might see in exported VQM statistic files.

### **Burst, Gap, and Loss Metrics**

**VqmBurstRLq** is the average listening quality R factor while the stream is in a burst condition. A stream is in a burst condition if two or more successive voice encoder/decoder frames have been lost or discarded. (Note that a stream is always in either a burst or a gap condition.)

**VqmBurstCount** is the number of times the given stream was in a burst condition during the call. A stream is in a burst condition if two or more successive voice encoder/decoder frames have been lost or discarded. (Note that a stream is always in either a burst or a gap condition.)

**VqmBurstLossRateAvg** is the total average percentage of voice encoder/decoder frames lost or discarded while in burst conditions. A stream is in a burst condition if two or more successive voice encoder/decoder frames have been lost or discarded. (Note that a stream is always in either a burst or a gap condition.)

**VqmBurstLenAvgPkts** is the average burst length in packets. A stream is in a burst condition if two or more successive voice encoder/decoder frames have been lost or discarded. (Note that a stream is always in either a burst or a gap condition.)

**VqmBurstLenAvgMsec** is the average burst length in milliseconds. A stream is in a burst condition if two or more successive voice encoder/decoder frames have been lost or discarded. (Note that a stream is always in either a burst or a gap condition.)

**VqmExcessBurst** is the total length (in milliseconds) of speech lost during burst conditions not handled effectively by packet loss concealment.

**VqmGapR** is the average listening quality R factor while the stream is in a gap condition. A stream is in a gap condition if 16 voice encoder/decoder frames are received without any intervening lost or discarded frames. (Note that a stream is always in either a burst or a gap condition.)

**VqmGapCount** is the number of times the given stream was in a gap condition during the call. A stream is in a gap condition if 16 voice encoder/decoder frames are received without any intervening lost or discarded frames. (Note that a stream is always in either a burst or a gap condition.)

**VqmGapLossRateAvg** is the total average percentage of voice encoder/decoder frames lost or discarded while in gap conditions. A stream is in a gap condition if 16 voice encoder/decoder frames are received without any intervening lost or discarded frames. (Note that a stream is always in either a burst or a gap condition.)

**VqmGapLenPkts** is the average gap length in packets. A stream is in a gap condition if 16 voice encoder/decoder frames are received without any intervening lost or discarded frames. (Note that a stream is always in either a burst or a gap condition.)

**VqmGapLenMsec** is the average gap length in milliseconds. A stream is in a gap condition if 16 voice encoder/decoder frames are received without any intervening lost or discarded frames. (Note that a stream is always in either a burst or a gap condition.)

**VqmExcessGap** is the total length (in milliseconds) of speech lost during gap conditions not handled effectively by packet loss concealment.

**VqmLossRateAvg** is the total average percentage of voice encoder/decoder frames lost or discarded, regardless of burst or gap conditions. (This is a sum of the average network loss rate and the average discard rate.)

**VqmDiscardRateAvg** is the total average percentage of voice encoder/decoder frames as discarded by the jitter buffer.

**VqmNetworkLossAvg** is the total average percentage of voice encoder/decoder frames lost in the network. Note that this does not include frames that are discarded by the jitter buffer. If this percentage is very small when compared with the overall average loss rate, examine your jitter buffer configuration. If it is large, examine the network to determine the source of the loss.

## Simulated Jitter Buffer Statistics

The following are terms associated with VQM's simulated jitter buffer that can appear in exported VQM statistic files.

**VqmJitterBufferType** is 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.

**VqmJbCfgMin** is the minimum delay that will be applied to packets received when using an adaptive jitter buffer.

**VqmJbCfgNom** is 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.

**VqmJbCfgMax** is 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 represents the maximum number of packets that can be inserted into the buffer. (Subsequently, inserted packets will be discarded.)

**VqmPpdvMsec** is the packet-to-packet delay variation (jitter) in milliseconds, as defined in RFC 3550.

**VqmEarlyThreshMs** is the early threshold in milliseconds; this is a configured value. Early packets arriving under this threshold will not be discarded by the jitter buffer as early.

**VqmEarlyThreshPc** is the percentage of total packets (including both early and late packets) not judged as being under the early jitter threshold.

**VqmEarlyUnderThresh** is a count of early packets that arrived under the configured early jitter threshold (of the total count of early packets).

**VqmEarlyTotalCount** is the total count of early packets arriving before the expected delay. (This count is not necessarily the same as the count of early packets listed below, that is based on the first reference packet.)

**VqmEarlyPeakJitterMs** is the largest jitter encountered among packets counted as early (the total of which is given as the total early count under the jitter buffer statistics).

**VqmLateThresholdMs** is the late threshold in milliseconds; this is a configured value. Late packets arriving under this threshold will not be discarded by the jitter buffer as late.

**VqmLateThresholdPc** is the percentage of total packets (including both early and late packets) not judged as being under the late jitter threshold.

**VqmLateUnderThresh** is a total count of packets that arrived under the configured late jitter threshold (of the total count of late packets).

**VqmLateTotalCount** is the total count of late packets arriving after the expected delay. (This count is not necessarily the same as the count of late packets listed below, that is based on the first reference packet.)

**VqmLatePeakJitterMs** is the largest jitter encountered among packets counted as late (the total of which is given as the total late count under the jitter buffer statistics).

**VqmDelayIncreaseCount** is the total number of jitter buffer delay increases. (This applies to adaptive mode only.)

**VqmDelayDecreaseCount** is the total number of jitter buffer delay decreases. (This applies to adaptive mode only.)

**VqmPktsRcvdTotal** is the total number of packets received by the jitter buffer.

**VqmDuplicatePkts** is the total number of duplicate packets for this RTP stream discarded by the jitter buffer.

**VqmEarlyPkts** is 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, treated as a reference packet. (This count is not necessarily the same as the count shown above under the jitter buffer statistics.)

**VqmLatePkts** is 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, treated as a reference packet. (This count is not necessarily the same as the count shown above under the jitter buffer statistics.)

**VqmPktsDiscardedTotal** is the total number of packets discarded by the jitter buffer.

**VqmOverrunDiscardPkts** is the total number of packets discarded by the jitter buffer due to jitter buffer overrun.

**VqmUnderrunDiscardPkts** is the total number of packets discarded by the jitter buffer due to jitter buffer underrun.

## External Quality Statistics

The following are terms for quality statistics associated with RTCP XR reports.

**VqmExtRLqIn** is the external listening quality R factor (in), as determined from an RTCP XR report (RFC 3611). It represents the listening quality R factor incoming to the other side of this endpoint; the other side may be an external PCM or cellular network. For example, suppose VQM were deployed in the following network: Phone A <-- VQM --> Bridge <--> Phone B. For RTCP XR reports sent from Bridge to Phone A, the external LQ R factor (in) would represent quality for RTP flowing from Phone B to Bridge. If no applicable reports are received, this statistic will be unavailable.

**VqmExtRLqOut** is the external listening quality R factor (out), as determined from an RTCP XR report (RFC 3611). This measurement is copied from an RTCP XR message received from a remote endpoint on the other side of this endpoint. For example, suppose VQM were deployed in the following network: Phone A <-- VQM --> Bridge <--> Phone B. For RTCP XR reports sent from Bridge to Phone A, the external LQ R factor (out) would represent quality for RTP flowing from Bridge to Phone B. If no applicable reports are received, this statistic will be unavailable.

**VqmExtRCqIn** is the external conversational quality R factor (in), as determined from an RTCP XR report (RFC 3611). It represents the conversational quality R factor incoming to the other side of this endpoint; the other side may be an external PCM or cellular network. For example, suppose VQM were deployed in the following network: Phone A <-- VQM --> Bridge <--> Phone B. For RTCP XR reports sent from Bridge to Phone A, the external CQ R factor (in) would represent quality for RTP flowing from Phone B to Bridge. If no applicable reports are received, this statistic will be unavailable.

**VqmExtRCqOut** is the external conversational quality R factor (out), as determined from an RTCP XR report (RFC 3611). This measurement is copied from an RTCP XR message received from a remote endpoint on the other side of this endpoint. For example, suppose VQM were deployed in the following network: Phone A <-- VQM --> Bridge <--> Phone B. For RTCP XR reports sent from Bridge to Phone A, the external CQ R factor (out) would represent quality for RTP flowing from Bridge to Phone B. If no applicable reports are received, this statistic will be unavailable.

## Fax and Modem Statistics

The following are VQM statistics associated with fax and modem traffic.

**VqmThroughPutIndex** is the estimated throughput for a fax or data call; a bitrate ranging from **0** to **35000** bps. This value is calculated based on gap/burst conditions and loss/discard rates.

**VqmReliabilityIndex** is a reliability index for a fax or data call ranging from **0** (least reliable) to **100** (most reliable).



## Appendix B: ADTRAN-Supported CODECs

The following CODECs are supported by VQM in AOS:

- G.711 64k  $\mu$ -law voice encoder/decoder
- G.711 64k  $\mu$ -law with PLC voice encoder/decoder
- G.723.1 5.3k voice encoder/decoder
- G.723.1 6.3k voice encoder/decoder
- G.728 voice encoder/decoder
- G.729 voice encoder/decoder
- G.729A voice encoder/decoder
- GSM 6.10 (full-rate)
- Reserved for GSM 6.10 (half-rate)
- GSM 6.10 (enhanced full-rate)
- Lucent/elemedia SX7300/8300
- Lucent/elemedia SX9600
- G.711 64k A-law voice encoder/decoder
- G.711 64k A-law with PLC voice encoder/decoder
- G.726 ADPCM voice encoder/decoder (16, 24, 32, 40 kbit)
- GIPS Enhanced G.711  $\mu$ -law voice encoder/decoder
- GIPS Enhanced G.711 A-law voice encoder/decoder
- GIPS iLBC voice encoder/decoder
- GIPS iSAC voice encoder/decoder
- GIPS iPCM-wb voice encoder/decoder
- G.729E voice encoder/decoder (8.0, 11.8 kbit)
- Wideband Linear PCM voice encoder/decoder
- Wideband Linear PCM voice encoder/decoder with PLC
- G.722 voice encoder/decoder (64, 56, 48 kbit)
- G.722.1 voice encoder/decoder (32, 24 kbit)
- G.722.2 voice encoder/decoder (23.85 kbit, 23.05, 19.85, 18.25, 15.85, 14.25, 12.85, 8.85, 6.6 kbit)
- QCELP voice encoder/decoder (8, 13 kbit)
- EVRC voice encoder/decoder
- SMV voice encoder/decoder (8.12, 5.79, 4.44, 3.95 kbit)
- AMR Narrowband voice encoder/decoder (12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15, 4.75 kbit)
- iLBC voice encoder/decoder (13.3, 15.2 kbit)
- G.711 56k  $\mu$ -law voice encoder/decoder
- G.711 56k  $\mu$ -law with PLC voice encoder/decoder
- G.711 56k A-law voice encoder/decoder
- G.711 56k A-law with PLC voice encoder/decoder
- G.723.1 Annex C voice encoder/decoder
- Speex Narrowband voice encoder/decoder (2.15, 5.95, 8, 11, 15, 18.2, 24.6, 3.95)
- Speex Wideband voice encoder/decoder (12.8, 16.8, 20.6, 23.8, 27.8, 34.2, 42.2 kbit)