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Reference Guide

Understanding Call Quality Statistics in AOS

This reference guide explains the call quality statistics available in ADTRAN Operating System (AOS) products. The guide includes an overview of call quality statistics and explains the results displayed using the web-based graphical user interface (GUI) or the command line interface (CLI). Additional resources are provided at the end of this document to further assist in understanding call quality statistics.

This guide consists of the following sections:

Call Quality Statistics Overview

AOS provides a number of call quality and media gateway statistics displayed using the CLI **show** commands or GUI menus. These statistics measure the quality of Realtime Transport Protocol (RTP) voice streams as they are received by the AOS device. Understanding these statistics is useful in diagnosing voice quality problems. The following sections provide an overview of these statistics by explaining what they mean and how they are recorded in AOS. This guide does not provide diagnostic advice or troubleshooting tips. For additional information, refer to the ADTRAN Support Forums at https://supportforums.adtran.com.

The **show** commands and GUI menus are divided into two groups which can be easily referenced.

- *Media Gateway Statistics on page 3*
- *Show Media-Gateway Channel on page 13*

How the Statistics are Derived

Many of the statistics presented through the CLI **show** commands and GUI menus refer to the packetized voice traffic received by the AOS unit and are presented in cumulative totals for the device or separated into smaller groupings, such as per channel or session. Regardless of the reporting structure, understanding the terminology used to describe the statistics is useful in comprehending these statistics and how their values are derived by AOS.

Average Delay and Maximum Delay

The average and maximum delay indicate the size of the jitter buffer in milliseconds (ms). By default, the jitter buffer is set to adaptive mode with a nominal depth of 50 ms. If the unit continues to log packets outside the 50 ms window, the jitter buffer depth will automatically increase up to a max of 100 ms. The nominal and maximum jitter buffer size can be configured per foreign exchange station (FXS) user or E&M/foreign exchange office (FXO)/primary rate interface (PRI) trunk. The jitter buffer size can also be set to a fixed value by changing the delay mode to fixed and configuring the nominal packet delay to the desired window size.

Out-of-Order Packets

Out-of-order packets are defined as the number of packets received that do not match the expected sequence number. When an inbound RTP packet is sent to the jitter buffer, the sequence number in the RTP header is checked. The sequence number on each successive packet should be one greater than the sequence number of the previous packet. If the receive sequence number does not match the expected sequence number, an out-of-order packet is logged.

Discarded Packets

If a packet arrives before or after the window of the jitter buffer (based on the RTP timestamp), it is considered an early or a late arrival. All early and late arrivals are logged as discarded packets and are not played out. If the jitter buffer is set to adaptive mode and the unit continues to log packets outside of the jitter buffer window, the digital signal processor (DSP) will increase the window size in 10 ms increments.

80 timestamp units represent 10 ms of audio. If the packetization period of an RTP stream is 20 ms, each sequential RTP timestamp should increment by 160. If the unit receives an RTP packet with a timestamp of 1084455193, the next expected timestamp would be 1084455353 (160 units difference).

Hardware and Software Requirements and Limitations

Call quality statistics are available on the Total Access 900(e) Series, NetVanta 600 Series, NetVanta 6000 Series, and NetVanta 7000 Series products.

Media Gateway Statistics

The media gateway statistics are displayed from the CLI by entering the **show media-gateway** command, or through the GUI by navigating to **Voice** > **Reports** > **RTP Session Stats**, or **Voice** > **Reports** > **RTP Channel Stats**. The information presented varies from the CLI to the GUI and is explained in detail in the following sections:

- *Show Media-Gateway [info | summary | summary active] on page 3*
- *Show Media-Gateway Session on page 8*
- *Show Media-Gateway Channel on page 13*

NOTE

*The output of all **show** commands can be limited by appending the following modifiers to the end of the command: | **begin** <text>, | **exclude** <text>, and | **include** <text>. The **include** modifier limits output to lines that contain the specified text, the **exclude** modifier excludes any lines with the specified text, and the **begin** modifier displays the first line of output with the specified text and all lines thereafter.*

Show Media-Gateway [info | summary | summary active]

Cumulative totals for all RTP channels are displayed through the CLI using the **show media-gateway** command. Using the **info** parameter displays general DSP information. Using the **summary** parameter displays a status summary of all DSP channels. Using the **summary active** parameter shows the summary for currently active DSP channels.

The statistics displayed in the CLI using the **show media-gateway** command display combined statistics for all DSP channels and are explained in *Table 1 on page 4*. The following is a sample of the CLI output for the **show media-gateway** command.

```
>enable
#show media-gateway
Media Gateway
 1 slots, 2 DSPs, 60 channels
 6 total sessions, 1 active sessions,
 00:00:11 total session duration
 Last clearing of counters: never
Receive
 601 total rx packets, 96160 total rx bytes
 Jitter Buffer Totals:
 0 out of order packets
 0 early arrival discards
 0 late arrival discards
 0 buffer full discards
 0 unknown packets
 13 flushed packets
```


Transmit
647 total tx packets, 103520 total tx bytes

Table 1. Media Gateway Statistics

Statistic	Description
Slots, DSPs, Channels	Indicates the total number of slots, DSPs, and channels available in the unit.
Total Sessions	The **Total** sessions count is incremented each time an RTP channel is started, which could occur multiple time per call.
Active Sessions	The **Active** sessions count is the number of active sessions or DSP channels in use.
Total Session Duration	The total amount of time all DSP channels have been active.
Last clearing of counters	When the counters were last cleared.
Total Rx Packets	The number of packets received.
Total Rx Bytes	The number of bytes received.
Out-of-Order Packets	Out-of-order packets is the number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded by the DSP because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Unknown Packets	Packets received by the DSP that do not contain valid RTP frames.
Flushed Packets	Packets discarded by the DSP due to an operation that required the jitter buffer to be flushed.
Total Tx Packets	The number of packets transmitted.
Total Tx Bytes	The number of bytes transmitted.

The statistics displayed in the CLI using the **show media-gateway info** command provide detailed information specific to each DSP and are explained in *Table 2 on page 5*. A sample of the CLI output for the **show media-gateway info** command is shown below.

>enable
#show media-gateway info
slot 0, DSP 1
DSP software version: G2.R10.5.0.0
DSP hardware version: Freescale MSC7119
DSP utilization: 49%
maximum DSP utilization: 52%
free packet buffers: 5998
total channels: 30

```
  active channels: 0
  DSP uptime: 2d 23:42:10

slot 0, DSP 2
  DSP software version: G2.R10.5.0.0
  DSP hardware version: Freescale MSC7119
  DSP utilization: 49%
  maximum DSP utilization: 51%
  free packet buffers: 5998
  total channels: 30
  active channels: 0
  DSP uptime: 2d 23:42:04

   system uptime: 2d 23:42:18
```

Table 2. Media Gateway Info Statistics

Statistic	Description
Slot / DSP	Specifies the slot and port of the DSP.
DSP Software Version	The DSP software version currently running.
DSP Hardware Version	The DSP hardware version.
DSP Utilization	The current DSP utilization percentage. Idle utilization of approximately 50 percent is normal.
Maximum DSP Utilization	The maximum DSP utilization percentage.
Free Packet Buffers	A debug statistic used to check for packet leaks in the DSP. This value should remain fairly constant when idle. If it begins to decrease over time, a packet leak may be indicated.
Total Channels	The number of channels supported by the listed DSP.
Active Channels	The number of channels currently in use on the listed DSP.
DSP Uptime	The total uptime for the listed DSP.
System Uptime	The total system uptime of the unit.

The statistics displayed in the CLI using the **show media-gateway summary** command provide a summary of the DSP channels and are explained in *Table 3 on page 6*. Using the **active** parameter will display a summary of the active DSP channels only. A sample (showing only the first 30 channels) of the CLI output for the **show media-gateway summary** command is shown below.

>enable
#show media-gateway summary

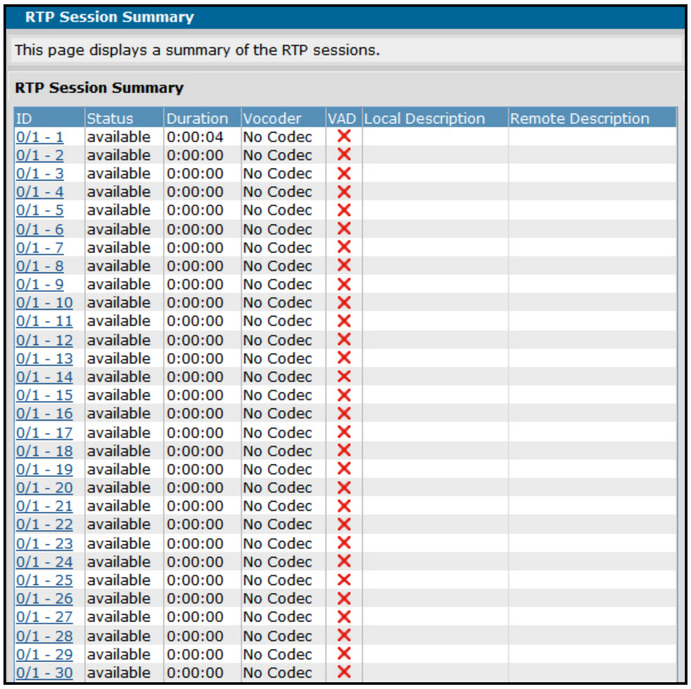
```
Slot/                          Local               Remote
DSP  Chan   Codec     VAD  Description         Description      Status    Duration
----------------------------------------------------------------------------------------------------------
0/1   1      No Codec                                           Down      00:00:00
0/1   2      No Codec                                           Down      00:00:00
0/1   3      No Codec                                           Down      00:00:00
0/1   4      No Codec                                           Down      00:00:00
```


0/1	5	No Codec	Down	00:00:00
0/1	6	No Codec	Down	00:00:00
0/1	7	No Codec	Down	00:00:00
0/1	8	No Codec	Down	00:00:00
0/1	9	No Codec	Down	00:00:00
0/1	10	No Codec	Down	00:00:00
0/1	11	No Codec	Down	00:00:00
0/1	12	No Codec	Down	00:00:00
0/1	13	No Codec	Down	00:00:00
0/1	14	No Codec	Down	00:00:00
0/1	15	No Codec	Down	00:00:00
0/1	16	No Codec	Down	00:00:00
0/1	17	No Codec	Down	00:00:00
0/1	18	No Codec	Down	00:00:00
0/1	19	No Codec	Down	00:00:00
0/1	20	No Codec	Down	00:00:00
0/1	21	No Codec	Down	00:00:00
0/1	22	No Codec	Down	00:00:00
0/1	23	No Codec	Down	00:00:00
0/1	24	No Codec	Down	00:00:00
0/1	25	No Codec	Down	00:00:00
0/1	26	No Codec	Down	00:00:00
0/1	27	No Codec	Down	00:00:00
0/1	28	No Codec	Down	00:00:00
0/1	29	No Codec	Down	00:00:00
0/1	30	No Codec	Down	00:00:00

Table 3. Media Gateway Summary Statistics

Statistic	Description
Slot / DSP	Specifies the location of the DSP in the format *slot/dsp.*
Channel	Specifies the channel number on the DSP.
Codec	The voice CODEC selected for the channel.
VAD	Indicates whether voice activity detection (VAD) is enabled or disabled.
Local Description	A description of the local party.
Remote Description	A description of the remote party.
Status	Indicates the current channel status.
Duration	The length of time the session was active.

The GUI displays the RTP session summary information through the **RTP Session Summary** menu, available by navigating to **Voice** > **Reports** > **RTP Session Stats**. An example of the output from the GUI is shown in *Figure 1 on page 7*. The statistics are explained in *Table 4 on page 7*.



RTP Session Summary

This page displays a summary of the RTP sessions.

RTP Session Summary

ID	Status	Duration	Vocoder	VAD	Local Description	Remote Description
0/1 - 1	available	0:00:04	No Codec			
0/1 - 2	available	0:00:00	No Codec			
0/1 - 3	available	0:00:00	No Codec			
0/1 - 4	available	0:00:00	No Codec			
0/1 - 5	available	0:00:00	No Codec			
0/1 - 6	available	0:00:00	No Codec			
0/1 - 7	available	0:00:00	No Codec			
0/1 - 8	available	0:00:00	No Codec			
0/1 - 9	available	0:00:00	No Codec			
0/1 - 10	available	0:00:00	No Codec			
0/1 - 11	available	0:00:00	No Codec			
0/1 - 12	available	0:00:00	No Codec			
0/1 - 13	available	0:00:00	No Codec			
0/1 - 14	available	0:00:00	No Codec			
0/1 - 15	available	0:00:00	No Codec			
0/1 - 16	available	0:00:00	No Codec			
0/1 - 17	available	0:00:00	No Codec			
0/1 - 18	available	0:00:00	No Codec			
0/1 - 19	available	0:00:00	No Codec			
0/1 - 20	available	0:00:00	No Codec			
0/1 - 21	available	0:00:00	No Codec			
0/1 - 22	available	0:00:00	No Codec			
0/1 - 23	available	0:00:00	No Codec			
0/1 - 24	available	0:00:00	No Codec			
0/1 - 25	available	0:00:00	No Codec			
0/1 - 26	available	0:00:00	No Codec			
0/1 - 27	available	0:00:00	No Codec			
0/1 - 28	available	0:00:00	No Codec			
0/1 - 29	available	0:00:00	No Codec			
0/1 - 30	available	0:00:00	No Codec			

Figure 1. RTP Session Summary Menu

Table 4. RTP Session Summary Displayed in the GUI

Statistic	Description
ID	Specifies the ID of the RTP channel in the format *slot/dsp - channel*.
Status	Indicates whether the session is active or inactive.
Duration	The length of time the session was active.
Vocoder	The voice CODEC selected for the session.
VAD	Indicates whether VAD is enabled or disabled.
Local Description	A description of the local party.
Remote Description	A description of the remote party.

Show Media-Gateway Session

Display detailed RTP session statistics through the CLI by entering the **show media-gateway session** command. Adding the <*slot/dsp.channel*> parameter to this command limits the displayed information to the specified channel only. The statistics are explained in *Table 5 on page 8*.

A sample of the CLI output for the s**how media-gateway session** <*slot/dsp.channel*> command is shown below.

>enable
#show media-gateway session 0/1.1


```
Session 0/1.1 (INACTIVE)
   slot 0, DSP 1, channel 1
   start time: 5:53 PM Thu, Nov 29, 2012, duration: 00:00:00
Local
   Description:
   IP: N/A, UDP port: 0
Remote
   Description:
   IP: 10.100.13.243, UDP port: 10760
Vocoder: No Codec, VAD disabled, 2 frames per packet
Echo-cancellation enabled
Receive
   30 packets expected
   30 rx packets, 4800 rx bytes
   0 packet difference between expected and received
   Jitter Buffer Stats:
   0 current depth (frames), 6 highest depth (frames)
   50 current delay (ms), 50 highest delay (ms)
   0 out of order packets
   0 early arrival discards
   0 late arrival discards
   0 buffer full discards
   0 unknown packets
   0 flushed packets
   0x32dd8e9e sync source
   0 SSRC changes
Transmit
   40 tx packets, 6400 tx bytes
   0x20fc1b17 sync source
```

Table 5. Detailed Media Gateway Session Statistics

Statistic	Description
Session	Specifies the ID of the DSP channel in the format *slot/dsp.channel,* followed by the status in parenthesis. The status indicates whether the session is active or inactive.
Slot, DSP, Channel	Specifies the location of the RTP channel by slot, DSP and channel.
Start Time	The starting time of the session based on the system clock.
Duration	The length of time the DSP channel has been active for this session.
Local Description	A description of the local party.
Local IP Address	The local IPv4 address from which RTP is being transmitted and received.
Local UDP Port	The local UDP port from which the RTP is being transmitted and received.
Remote Description	A description of the remote party.
Remote Address	The IPv4 address to which RTP is being transmitted and received.
Remote UDP Port	The UDP port to which RTP is being transmitted and received.

Table 5. Detailed Media Gateway Session Statistics *(Continued)*

Statistic	Description
Vocoder	The voice CODEC selected for the session.
VAD	Indicates whether VAD is enabled or disabled.
Frames Per Packet	The number of 10 ms RTP frames transmitted per packet.
Echo-Cancellation	Indicates whether echo cancellation is enabled or disabled.
Receive Packets Expected	The number of packets expected to be received.
Rx Packets	The number of packets received.
Rx Bytes	The number of bytes received.
Difference Between Expected and Received	The difference between the expected and the actual number of packets received.
Jitter Buffer Current Depth (in frames)	The current size of the jitter buffer in 10 ms RTP frames.
Jitter Buffer Highest Depth (in frames)	The maximum size of the jitter buffer in 10 ms RTP frames, for this session.
Jitter Buffer Current Delay (ms)	The current size of the jitter buffer provided in milliseconds.
Jitter Buffer Highest Delay (ms)	The maximum size of the jitter buffer provided in milliseconds, for this session.
Out-of-Order Packets	The number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded by the DSP because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Unknown Packets	Packets received by the DSP that do not contain valid RTP frames.
Flushed Packets	Packets discarded by the DSP due to an operation that required the jitter buffer to be flushed.
Receive Sync Source	The synchronization source ID (SSRC) value as defined by RFC 3550. Receive SSRC is the SSRC in the RTP packets being received by the DSP.
SSRC Changes	The number of times the SSRC of the received RTP packets changed during this session.
Tx Packets	The number of packets transmitted.
Tx Bytes	The number of bytes transmitted.
Transmit Sync Source	The SSRC value as defined by RFC 3550. Transmit SSRC is generated by the DSP and used in transmitted RTP packets.

To display specific RTP session details in the GUI, select the call ID hyperlink from the **RTP Session Summary** menu. An example of the output from the GUI is shown in *Figure 2*. The statistics are explained in *Table 6 on page 11*.

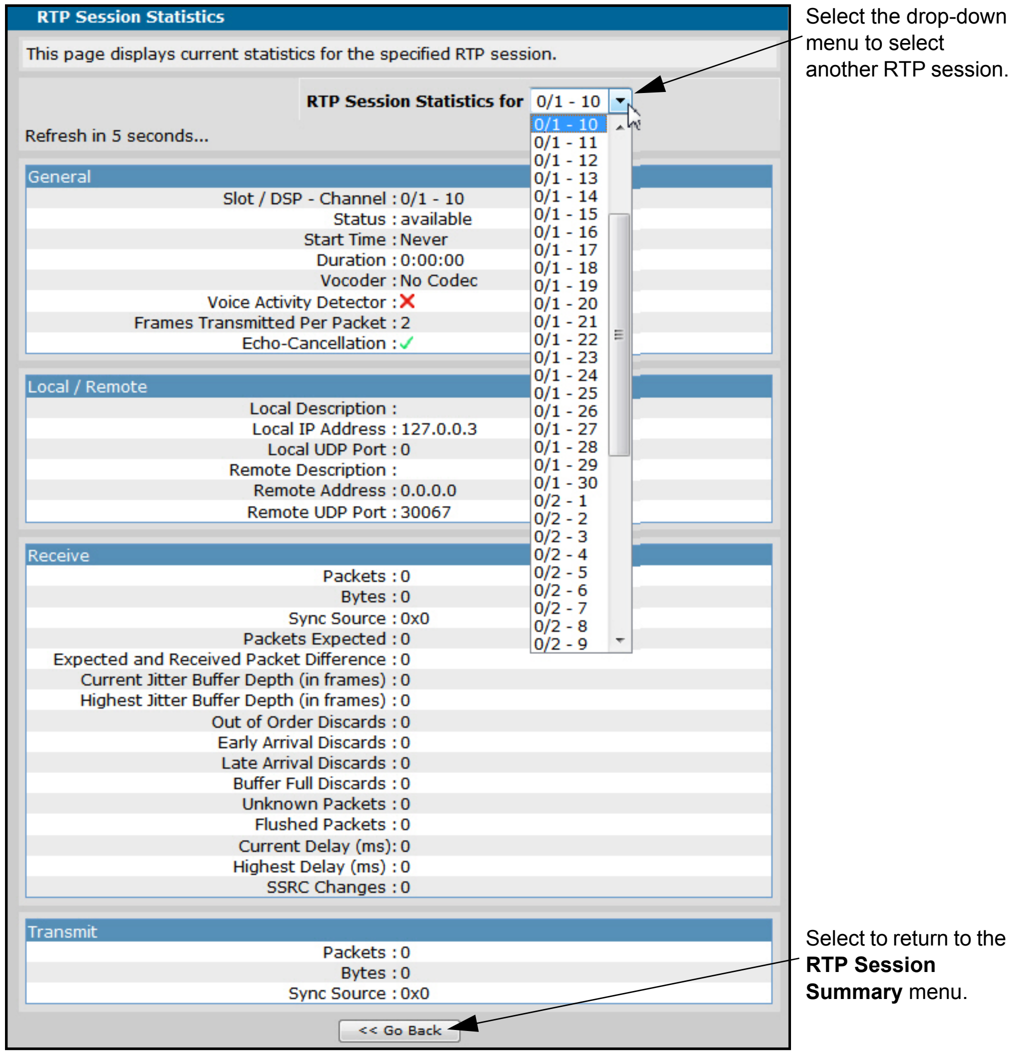


Figure 2. Detailed RTP Session Statistics Menu

Table 6. Detailed RTP Session Statistics Displayed in the GUI

Statistic	Description
General	
Slot / DSP - Channel	Specifies the ID of the DSP channel.
Status	Indicates the current status of the channel.
Start Time	The starting time of the session based on the system clock.
Duration	The length of time the DSP channel has been active for this session.

Table 6. Detailed RTP Session Statistics Displayed in the GUI *(Continued)*

Statistic	Description
Vocoder	The voice CODEC selected for the session.
Voice Activity Detector	Indicates whether VAD is enabled or disabled.
Frames Transmitted Per Packet	The number of 10 ms RTP frames transmitted per packet.
Echo-Cancellation	Indicates whether echo cancellation is enabled or disabled.
Local / Remote	
Local Description	A description of the local party.
Local IP Address	The local Pv4 address from which RTP is being transmitted and received.
Local UDP Port	The local UDP port from which RTP is being transmitted and received.
Remote Description	A description of the remote party.
Remote Address	The IPv4 address to which RTP is being transmitted and received.
Remote UDP Port	The UDP port to which RTP is being transmitted and received.
Receive	
Packets	The number of packets received.
Bytes	The number of bytes received.
Sync Source	The synchronization source ID (SSRC) value as defined by RFC 3550. Receive SSRC is the SSRC in the RTP packets being received by the DSP.
Packets Expected	The number of packets expected to be received.
Expected and Received Packet Difference	The difference between the expected and the actual number of packets received.
Current Jitter Buffer Depth	The current size of the jitter buffer in 10 ms RTP frames.
Highest Jitter Buffer Depth	The maximum size of the jitter buffer in 10 ms RTP frames, for this session.
Out-of-Order Discards	The number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded by the DSP because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Unknown Packets	Packets received by the DSP that do not contain valid RTP frames.

Table 6. Detailed RTP Session Statistics Displayed in the GUI *(Continued)*

Statistic	Description
Flushed Packets	Packets discarded by the DSP due to an operation that required the jitter buffer to be flushed.
Current Delay	The current size of the jitter buffer provided in milliseconds.
Highest Delay	The maximum size of the jitter buffer provided in milliseconds, for this session.
SSRC Changes	The number of times the SSRC of the received RTP packets changed during this session.
Transmit	
Packets	The number of packets transmitted.
Bytes	The number of bytes transmitted.
Sync Source	The SSRC value as defined by RFC 3550. Transmit SSRC is generated by the DSP and used in transmitted RTP packets.

Show Media-Gateway Channel

Cumulative statistics for individual RTP channels are displayed through the CLI by entering the **show media-gateway channel** command. Adding the <*slot/dsp.channel*> parameter to this command limits the displayed information to the specified channel only. The statistics are explained in *Table 7 on page 13*.

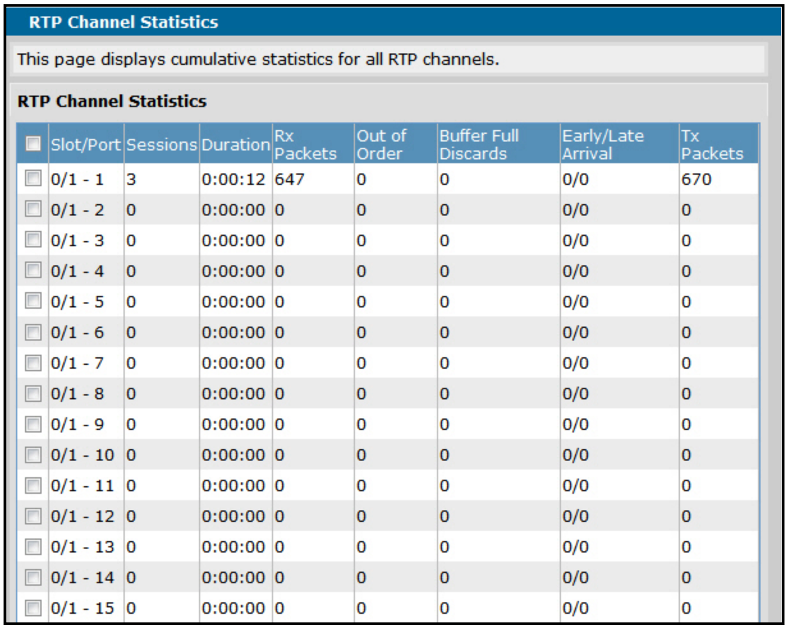
A sample of the CLI output for the s**how media-gateway channel** <*slot/dsp.channel*> command is shown below.

```
>enable
#show media-gateway channel 0/1.1
Channel 0/1.1
  slot 0, DSP 1, channel 1
  5 total sessions, 00:00:11 total session duration
  Last clearing of counters: never
Receive
  531 total rx packets, 84960 total rx bytes
  Jitter Buffer Totals:
  0 out of order packets
  0 early arrival discards
  0 late arrival discards
  0 buffer full discards
  0 unknown packets
  13 flushed packets
Transmit
  571 total tx packets, 91360 total tx bytes
```

Table 7. Media Gateway Channel Statistics

Statistic	Description
Channel	Specifies the ID of the DSP channel in the format *slot/dsp.channel*.
Slot, DSP, Port	Specifies the location of the RTP channel by slot, DSP and channel.
Total Sessions	The total number of sessions the channel has been used.
Total Session Duration	The length of time the all sessions have been active.
Last clearing of counters	When the counters were last cleared.
Total Rx Packets	The number of packets received.
Total Rx Bytes	The number of bytes received.
Out-of-Order Packets	The number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Unknown packets	Packets received by the DSP that did not contain valid RTP frames.
Flushed packets	Packets discarded by the DSP due to an operation that required the jitter buffer to be flushed.
Total Tx Packets	The number of packets transmitted.
Total Tx Bytes	The number of transmitted bytes.

The GUI displays the RTP channel information through the **RTP Channel Statistics** menu, available by navigating to **Voice** > **Reports** > **RTP Channel Stats**. Due to the limited amount of space available on the GUI, only a portion of the information is displayed. Examples of the output for the GUI is shown in *Figure 3 on page 14*. The statistics are explained in *Table 8 on page 15*.



RTP Channel Statistics

This page displays cumulative statistics for all RTP channels.

RTP Channel Statistics

Slot/Port	Sessions	Duration	Rx Packets	Out of Order	Buffer Full Discards	Early/Late Arrival	Tx Packets
0/1 - 1	3	0:00:12	647	0	0	0/0	670
0/1 - 2	0	0:00:00	0	0	0	0/0	0
0/1 - 3	0	0:00:00	0	0	0	0/0	0
0/1 - 4	0	0:00:00	0	0	0	0/0	0
0/1 - 5	0	0:00:00	0	0	0	0/0	0
0/1 - 6	0	0:00:00	0	0	0	0/0	0
0/1 - 7	0	0:00:00	0	0	0	0/0	0
0/1 - 8	0	0:00:00	0	0	0	0/0	0
0/1 - 9	0	0:00:00	0	0	0	0/0	0
0/1 - 10	0	0:00:00	0	0	0	0/0	0
0/1 - 11	0	0:00:00	0	0	0	0/0	0
0/1 - 12	0	0:00:00	0	0	0	0/0	0
0/1 - 13	0	0:00:00	0	0	0	0/0	0
0/1 - 14	0	0:00:00	0	0	0	0/0	0
0/1 - 15	0	0:00:00	0	0	0	0/0	0

Figure 3. RTP Channel Statistics Menu

Table 8. RTP Channel Statistics Displayed in the GUI

Statistic	Description
Slot/Port	Specifies the ID of the RTP channel in the format *slot/dsp - channel*.
Sessions	The total number of sessions the channel has been used.
Duration	The length of time all session have been active.
Rx Packets	The number of packets received.
Out-of-Order	The number of packets received that did not match the expected sequence number.
Buffer Full Discards	Packets discarded because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Early/Late Arrival	Packets discarded due to arriving before (early) or after (late) the window of the jitter buffer.
Tx Packets	The number of packets transmitted.

Voice Quality Statistics

The voice quality statistics are displayed from the CLI by entering the **show voice quality-stats** command, or through the GUI by navigating to **Voice** > **Reports** > **Call Quality Stats**. The information presented varies from the CLI to the GUI and is explained in detail in the following sections:

- *Show Voice Quality-Stats on page 15*
- *Show Voice Quality-Stats <id> on page 17*

NOTE

*The output of all **show** commands can be limited by appending the following modifiers to the end of the command: | **begin** <text>, | **exclude** <text>, and | **include** <text>. The **include** modifier limits output to lines that contain the specified text, the **exclude** modifier excludes any lines with the specified text, and the **begin** modifier displays the first line of output with the specified text and all lines thereafter.*

NOTE

*Using the **realtime** argument for this command can adversely affect the performance of your unit.*

Show Voice Quality-Stats

A summary of the voice quality statistics is displayed from the CLI by using the **show voice quality-stats** command. By default, only the last 20 calls are displayed. This setting can be changed through the CLI by using the **voice quality-stats history max-streams** <*number*> command (valid range is **0** to **2000**). To display voice quality statistics for active calls only, use the **active** parameter. The information displayed is explained in *Table 9 on page 16*.

A sample of the CLI output for the **show voice quality-stats** command is shown below.

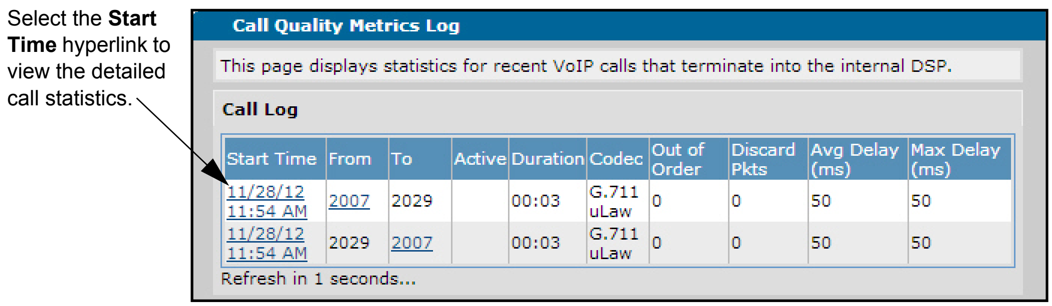
```
>enable
#show voice quality-stats
        Start                                          Out of  Discard     Delay
ID      Time        From    To          Length   Codec  Order     Pkts*    Avg   Max
-----------------------------------------------------------------------------------------
1       11:54 AM    2007    2029        0:03     G711u  0         0        50    50
2       11:54 AM    2029    2007        0:03     G711u  0         0        50    50

* - Use "show voice quality-stats <ID>" command for more detailed information on discarded packets.
```

Table 9. Voice Quality Statistics

Statistic	Description
ID	The internal call ID used by the switchboard to differentiate call legs.
Start Time	The starting time of the call based on the system clock.
From	The calling party number.
To	The called party number.
Length	The length of time the call is connected.
Codec	The voice CODEC selected for the call.
Out-of-Order	The number of packets received that did not match the expected sequence number.
Discard Pkts	The number of packets discarded.
Delay Avg / Delay Max	**Delay Avg** and **Delay Max** indicate the size of the jitter buffer in milliseconds.

The **Call Quality Metrics Log** GUI menu provides a summary of the voice quality statistics and is displayed in *Figure 4*. The information displayed is explained in *Table 10 on page 17*. More detailed statistics for a particular call is displayed by selecting the hyperlink for the call start time (see *Show Voice Quality-Stats <id> on page 17* for more information).



Start Time	From	To	Active	Duration	Codec	Out of Order	Discard Pkts	Avg Delay (ms)	Max Delay (ms)
11/28/12 11:54 AM	2007	2029		00:03	G.711 uLaw	0	0	50	50
11/28/12 11:54 AM	2029	2007		00:03	G.711 uLaw	0	0	50	50

Figure 4. Call Quality Metrics Menu

Table 10. Call Quality Statistics Displayed in the GUI

Statistic	Description
Start Time	The starting time of the call based on the system clock.
From	The calling party number.
To	The called party number.
Active	Indicates whether the call is currently active or not.
Duration	The length of time the call is connected.
Codec	The voice CODEC selected for the call.
Out-of-Order	The number of packets received that did not match the expected sequence number.
Discard Pkts	The number of packets discarded.
Avg Delay (ms)	Indicates the average size of the jitter buffer in milliseconds.
Max Delay (ms)	Indicates the maximum size of the jitter buffer in milliseconds.

Show Voice Quality-Stats *<id>*

More detailed statistics for a particular call are displayed by using the **show voice quality-stats** *<id>* command where *<id>* is the identity number for the call. The detailed voice quality statistics are explained in *Table 11 on page 18*.

A sample of the CLI output for the **show voice quality-stats** *<id>* command is shown below.

```
>enable
#show voice quality-stats 1
Call ID:  1    Active:  No
Start Time:                 Nov 28 2012 11:54 AM
End Time:                   Nov 28 2012 11:54 AM
Duration:                   0:03
```


Source: 2007 IP: 10.19.247.243 UDP: 10000 Name:
Destination: 2029 IP: 10.100.13.243 UDP: 16230 Name: T03
SSRC Changed: No
DSP Channel: dsp 0/1.1
Codec Used: G.711u
Jitter Buffer Mode: Adaptive
Voice Activity Detection: Disabled
Packet Loss Concealment: Disabled
Auto Level Control: Disabled
Non-Linear Suppression: Enabled
Comfort Noise Generation: Enabled
Non-Linear Processing: Enabled
Out of Order Packets: 0
Early Arrival Discards: 0
Late Arrival Discards: 0
Buffer Full Discards: 0
Jitter Delay Current: 50 ms, Max: 50 ms, Avg: 50 ms

Table 11. Detailed Voice Quality Statistics

Statistic	Description
Call ID	The internal call ID used by the switchboard to differentiate call legs.
Active	Indicates whether the call is currently active.
Start Time	The starting time of the call based on the system clock.
End Time	The ending time of the call based on the system clock.
Duration	The length of time the call is connected.
Source	The calling party number or source originating the call.
IP	The IPv4 address from which the calling party sources RTP.
UDP	The UDP port from which the calling party sources RTP.
Name	The calling party name, if one is assigned or received.
Destination	The called party number.
IP	The IPv4 address from which the called party sources RTP.
UDP	The UDP port from which the called party sources RTP.
Name	The called party name, if one is assigned or received.
SSRC Changed	Indicates whether the SSRC of received RTP has changed during the call.
DSP Channel	The DSP channel selected for the call.
Codec Used	The voice CODEC selected for the call.
Jitter Buffer Mode	The jitter buffer mode selected for the call (either **adaptive** or **fixed**). An **adaptive** jitter buffer dynamically adapts its size based on network conditions. A **fixed** jitter buffer does not change its size.

Table 11. Detailed Voice Quality Statistics *(Continued)*

Statistic	Description
Voice Activity Detection	Indicates whether VAD is enabled. VAD detects silence and transmits comfort noise packets during periods of silence, reducing bandwidth.
Packet Loss Concealment	Indicates whether packet loss concealment (PLC) is enabled. PLC is used to mitigate the effect of lost packets by replacing a lost packet with another voice packet in the data stream.
Auto Level Control	Indicates whether automatic level control (ALC) is enabled. ALC helps reduce the variance between loud and soft volume levels.
Non-Linear Suppression	Indicates whether non-linear suppression (NLS) is enabled.
Comfort Noise Generation	Indicates whether comfort noise generation (CNG) is enabled.
Non-Linear Processing	Indicates whether non-linear processing (NLP) is enabled.
Out-of-Order Packets	The number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Jitter Delay Current	The current size of the jitter buffer in ms.
Jitter Delay Max	The maximum size of the jitter buffer in ms.
Jitter Delay Avg	The average size of the jitter buffer in ms.

More detailed statistics for a particular call is displayed from the GUI by selecting the hyperlink for the start time of the call displayed in the **Call Quality Statistics** menu (shown in *Figure 4 on page 16*). An example of the **Call Details** menu is shown in *Figure 5 on page 19* for call ID 1. An explanation of the statistics is provided in *Table 12 on page 20*.

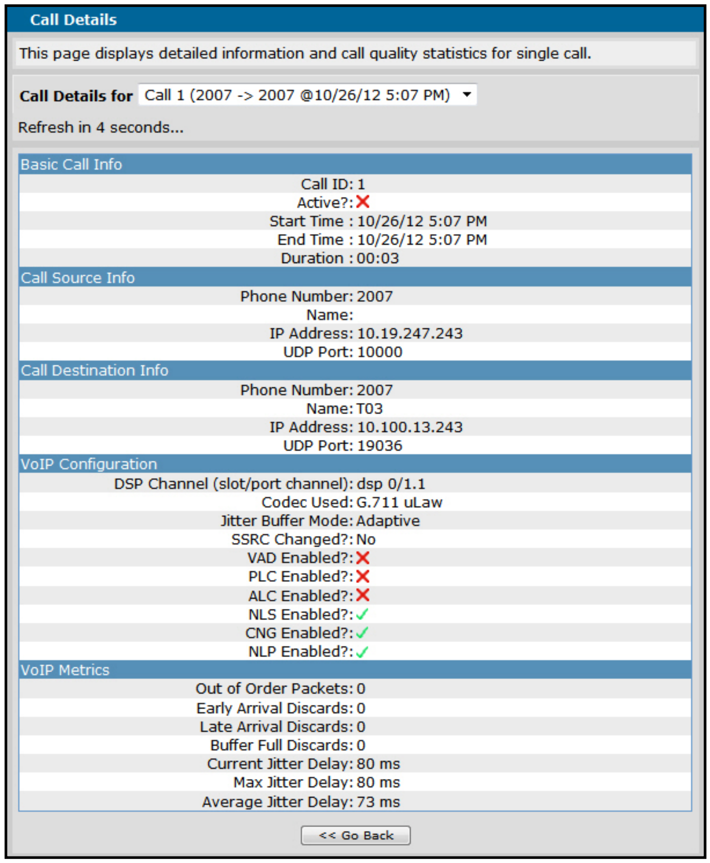


Figure 5. Call Details

Table 12. Detailed Call Statistics Displayed in the GUI

Statistic	Description
Basic Call Info	
Call ID	The internal call ID used by the switchboard to differentiate call legs.
Active	Indicates whether the call is currently active.
Start Time	The starting time of the call based on the system clock.
End Time	The ending time of the call based on the system clock.
Duration	The length of time the call is connected.
Call Source Info	
Phone Number	The calling party number or source originating the call.
Name	The calling party name, if one is assigned or received.
IP Address	The IPv4 address from which the calling party sources RTP.
UDP Port	The UDP port from which the calling party sources RTP.

Table 12. Detailed Call Statistics Displayed in the GUI *(Continued)*

Statistic	Description
Call Destination Info	
Phone Number	The called party number.
Name	The called party name, if one is assigned or received.
IP Address	The IPv4 address from which the called party sources RTP.
UDP Port	The UDP port from which the called party sources RTP.
VoIP Configuration	
DSP Channel	The DSP channel selected for the call, provided in the format *slot/port.channel*.
Codec Used	The voice CODEC selected for the call.
Jitter Buffer Mode	The jitter buffer mode selected for the call (either **adaptive** or **fixed**). An **adaptive** jitter buffer dynamically adapts its size based on network conditions. A **fixed** jitter buffer does not change its size.
SSRC Changed?	Indicates whether the synchronization source ID (SSRC) has changed.
VAD Enabled?	Indicates whether VAD is enabled. VAD detects silence and transmits comfort noise packets during periods of silence, reducing bandwidth.
PLC Enabled?	Indicates whether packet loss concealment (PLC) is enabled. PLC is used to mitigate the effects of lost packets by replacing a lost packet with another voice packet in the data stream.
ALC Enabled?	Indicates whether automatic level control (ALC) is enabled. ALC helps reduce the variance between loud and soft volume levels.
NLS Enabled?	Indicates whether non-linear suppression (NLS) is enabled.
CNG Enabled?	Indicates whether comfort noise generation (CNG) is enabled.
NLP Enabled?	Indicates whether non-linear processing (NLP) is enabled.
VoIP Metrics	
Out-of-Order Packets	The number of packets received that did not match the expected sequence number.
Early Arrival Discards	Packets discarded due to arriving before the window of the jitter buffer.
Late Arrival Discards	Packets discarded due to arriving after the window of the jitter buffer.
Buffer Full Discards	Packets discarded because the jitter buffer was full. Often caused by an RTP stream that was not completely torn down in the network.
Current Jitter Delay	The current size of the jitter buffer in ms.
Max Jitter Delay	The maximum size of the jitter buffer in ms.
Average Jitter Delay	The average size of the jitter buffer in ms.

Show Command Summary

The CLI commands used to view call quality statistics are summarized in *Table 13*.

Table 13. Call Quality Statistics Commands

Command	Description
show media-gateway	Displays cumulative totals for all RTP channels.
show media-gateway channel [<*slot/dsp.channel*>]	Displays cumulative totals for individual RTP channels. Specifying the ID of the DSP channel by entering the <*slot/dsp.channel*> parameter limits the output to display only for the specified channel.
show media-gateway info	Displays general DSP information.
show media-gateway session [<*slot/dsp.channel*>]	Displays current RTP session statistics. Specifying the ID of the DSP channel by entering the <*slot/dsp.channel*> parameter limits the output to display only for the specified channel.
show media-gateway summary [active]	Displays a summary of the state of the DSP channels. Using the **active** parameter shows the summary for currently active RTP sessions only.
show voice quality-stats **[active][realtime]**	Display voice quality statistics for all calls. Using the **active** parameter shows the statistics for active calls only.
show voice quality-stats <*id*> **[realtime]**	Display detailed voice quality statistics for a specific call.

Additional Resources

There are additional resources available to assist you in understanding call quality statistics in AOS. The following ADTRAN documents are available online at ADTRAN's Support Forum at https://supportforums.adtran.com.

- *AOS Command Reference Guide*