



Avaya™ VoIP Monitoring Manager

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User Guide

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Avaya™ VoIP Monitoring Manager User Guide

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Preface

Welcome to Avaya VoIP Monitoring Manager. This chapter provides an introduction to the structure and assumptions of this manual. It includes the following sections:

- **The Purpose of this Manual** - A description of the goals of this manual.
- **Who Should Use this Manual** - The intended audience of this manual.
- **Organization of this Manual** - A brief description of the subjects contained in the various sections of this manual.

The Purpose of this Manual

This manual contains information needed to use Avaya VoIP Monitoring Manager efficiently and effectively.

Who Should Use this Manual

This manual is intended for network managers familiar with network management and its fundamental concepts.

Organization of this Manual

This manual is structured to reflect the following conceptual divisions:

- **Preface** - A description of the manual's purpose, intended audience, and organization.
- **What is Avaya VoIP Monitoring Manager** - Includes an overview and system requirements.
- **Installing the Software** - Provides installation instructions.
- **How to Use Avaya VoIP Monitoring Manager** - Explains how to use the software.
- **Interpreting Reports** - Explains how to interpret the reports.
- **Glossary** - Provides a glossary of commonly used terms.

1 What is Avaya VoIP Monitoring Manager?

This chapter provides a brief explanation about Avaya™ VoIP Monitoring Manager, what you can do with this tool, its components and minimum requirements.

- **What is Avaya VoIP Monitoring Manager?**
- **What you can do with Avaya VoIP Monitoring Manager**
- **Components of Avaya VoIP Monitoring Manager**
- **What you need to run Avaya VoIP Monitoring Manager**

What is Avaya VoIP Monitoring Manager?

Avaya VoIP Monitoring Manager is a Voice over IP (VoIP) Quality of Service (QoS) monitoring tool. It enables you to monitor and review the quality of a call on an AVAYA™ VoIP Network.

Using the Avaya VoIP Monitoring Manager you can view the QoS data i.e. the Jitter, Round Trip Time (RTT) and Packet Loss experienced at the endpoints and during a session. This data displays in real-time or for previously active endpoints. With this information, you can begin to troubleshoot and isolate problems.

What You Can Do With Avaya VoIP Monitoring Manager

Search Endpoints You can search endpoints active from a specified time in the past or between a date range. Advanced search options enable you to narrow your search to match phone numbers, network addresses, or QoS levels.

View Reports Once you have a list of endpoints, you can select an endpoint or endpoints in a session and view its report. The reports display the QoS data i.e. Jitter, Round Trip Time (RTT) and Packet Loss. This is particularly useful for monitoring gateways or locating problems at a particular endpoint. As you can view reports for endpoints involved in a session, this will assist you with determining problems that occur between two endpoints or in an isolated area of the network.

Export Reports You can export the report data to a comma separated value (csv) file. You can open this file in most database and spreadsheet programs such as Microsoft Excel. Exporting the data to a spreadsheet enables you to manipulate the data so you can create your own reports.

Generate Automatic Alarms You can generate SNMP Traps/Alarms, which allows the Monitoring Manager to alert you when the Jitter, RTT or Packet Loss reaches certain levels. You can routinely monitor the network, and troubleshoot problems.

Components of Avaya VoIP Monitoring Manager

The VoIP Monitoring Manager incorporates the Avaya VoIP Monitoring Manager RTCP Monitor and the Avaya VoIP Monitoring Manager Server, which accepts connections from the Avaya VoIP Monitoring Manager Client. The Server software needs to be installed onto the network to work correctly. You will also need to have a Windows SNMP Agent installed on the Server. The components and their relationship are described in more detail as follows:

Avaya VoIP Monitoring Manager Server

The Avaya VoIP Monitoring Manager Server acts as a proxy between the Client and the RTCP Monitor. The main purpose of the Server is to reduce the amount of traffic to the Client by performing large data downloads and extensive processing of the MIB data stored on the RTCP Monitor. The Server resides on the same PC as the RTCP Monitor.

Avaya VoIP Monitoring Manager RTCP Monitor

The RTCP Monitor collects the RTCP packets sent from the AVAYA™ endpoints and stores the information in a proprietary database. The RTCP Monitor also runs as a sub-agent of the Windows SNMP agent. All the information contained in the database can be queried through SNMP.

The specifications for querying the database are found in:

- The RTP MIB. The reference is located at:
<http://www.ietf.org/rfc/rfc2959.txt>
- The proprietary AVAYA-VMON-MIB (The ASN.1 definitions of this MIB and associated traps are included as text files in the installation.)

Avaya VoIP Monitoring Manager Client

The Avaya VoIP Monitoring Manager Client provides the graphical user interface (GUI). The Client does not communicate with the RTCP Monitor and does not use the Windows SNMP service. The data that is displayed is gathered from the Server. The Client may be installed on the same machine as the Server, or it may be installed on another machine on the network. It is possible for the Server and the Client to communicate over a dial-up connection.

Avaya VoIP Monitoring Manager Web Client

The VoIP Monitoring Manager Client can run as a web application in a browser. This is useful if you only have the Server installed.

The Server needs to be running a web server. The machine running the Web Client needs to have the Sun Java Plug-in installed to run the Web Client. If you purchased the VisAbility Management Suite, the Apache web server will already be installed and running. If you choose to run the Apache web server, the installation will assist with configuration.

Configure the web server to publish the file to the following default install path:

C:\Program Files\Avaya\VoIP Monitoring Manager\jars\ClientApplet.htm

If you run the web client, you will not have access to all the functionality available in the application. This includes features such as copy; connect to a new server, and export.

What You Need to Run Avaya VoIP Monitoring Manager

The minimum system requirements to install and operate Avaya VoIP Monitoring Manager are as follows:

Operating System	Windows 2000
Software	The Simple Network Management Protocol Agent (SNMP Agent) is the Windows Service that runs on your computer. It is provided with the Windows 2000 CD but is not installed by default. When installing the VoIP Monitor Manager, you will be prompted to install it if it is not installed.
Processor	400 MHz Pentium II or higher compatible Pentium
RAM	128 MB (256 MB preferred)
Video	SVGA 800 x 600 display
Free Disk Space	500 MB
Port	<p>The Client and Server software communicate using Java Remote Method Invocation (RMI), and uses the port 1099 on the machine on which the Server is running.</p> <p>If this port is not available, the Server will attempt to use the following ports: 49177, 51173, or 63006. Although it is unlikely that all of these ports will be in use on a single machine, please ensure that at least one of these ports is available.</p>

2 Installing the Software

This chapter explains how to install Avaya VoIP Monitoring Manager and includes the following sections:

- **Installing the Server Software**
- **Installing the Client Software**
- **Solving Installation Problems**

Installing the Server Software

The Avaya VoIP Monitoring Manager Server needs to be installed on the VoIP network. If you are downloading the program from a web site, select to **Run this program from its current location**. The installation program will start automatically.

Alternatively, you can select to **save the file to disk** which may be the faster option. Once saved to your hard drive, double-click on the saved program to start the install. If you are installing the program from a CD-Rom, insert the CD into your drive and follow the instructions.

Ensuring Windows SNMP Agent is installed

The installation will check to see if the Windows SNMP Agent is installed. The Windows SNMP Agent must be installed for the Avaya VoIP Monitoring Manager Server to function. If the Windows SNMP Agent is not installed, the **Add/Remove Windows Components** will automatically start and you will be prompted for the Windows 2000 CD location so that you can install the Windows SNMP Agent.

To see if the Windows SNMP Agent is installed:

1. Click **Start > Settings > Control Panel > Administrative Tools > Services**
2. Scroll down until you see the SNMP Service status as **Started** and Startup Type as **Automatic**. If it is not included in the list you will need to install it from the Windows 2000 CD.

If it is listed but not set to run automatically, you will need to set it to start automatically as explained in Solving Installation Problems.

Installing the Client Software

The Avaya VoIP Monitoring Manager Client can be installed on the same machine as the Avaya VoIP Monitoring Manager Server, or it may be installed on another machine on the network. You install the Client software using one of the options as described above for Installing the Server software.

Solving Installation Problems

Avaya VoIP Monitoring Manager Server needs to be installed on the network. The Server software and the Windows SNMP Agent must be running before you can start the Client.

Changing RTCP Monitor Properties

If you need to change the RTCP Monitor properties, open the RTCP Monitor properties dialog and change the SNMP Agent Community ID (default: public) and the RTCP Listen Port as follows:

To Change RTCP Monitor Properties

1. From the Server dialog, click **Edit > RTCP Monitor Properties**.

The RTCP Monitor properties displays.

2. Type in a value in either the **SNMP Agent Community ID** or the **RTCP Listen Port** field. The Community ID must match the ID defined in the Windows SNMP Service Properties dialog.
3. Click **OK** to save the changes or **Cancel** to close without saving.

Avaya VoIP Monitoring Manager Server will reset the properties and attempt to re-connect to the Windows SNMP Agent based on the new properties.

Changing the RTCP port will result in a warning that it must match the port configured on the Avaya Call Processing. See <http://www.iana.org/assignments/port-numbers> and your Avaya Call Processing documentation. Also when entering a Windows SNMP Agent Community ID ensure it has write access (default:private). It is unusual to change the listen port from the default of 5005 as the default should work in most situations.

Set Windows SNMP Agent to Start Automatically

You need to have the Windows SNMP Agent installed and running on the Avaya VoIP Monitoring Manager Server before you start the Client. It enables the RTCP Monitor to publish the data.

To check to see if the Windows SNMP Agent is Installed and Set to Start Automatically

1. Click **Start > Settings > Control Panel > Administrative Tools > Services**
2. Scroll down until you see the SNMP Service status as **Started** and Startup Type as **Automatic**. If it is not included in the list, you will need to install it from the Windows 2000 CD. If it is included but not set to run automatically, you will need to change its properties.

To Set Windows SNMP Agent to Start Automatically

1. From Windows Service dialog as explained above, right-click on **SNMP** and select **Properties** from the context menu. The SNMP Service Properties dialog opens.
2. Select **Automatic** from the **Startup Type** drop down list and click **OK**

Check for a Valid Community ID

The Community ID assigned for your Windows SNMP Agent must match the Community ID defined in the RTCP Monitor Properties dialog. By default it is public but it may have been changed.

To Check for a Valid Community ID

1. Click **Start > Settings > Control Panel > Administrative Tools > Services**.
2. Scroll down and select the **SNMP Service**.
3. Right-click on SNMP Service and select **Properties** from the context menu.
4. Select the **Security** tab.

The RTCP Monitor dialog must have a Community ID from the list of Community Names.

3 How to Use Avaya VoIP Monitoring Manager

This chapter explains how to use Avaya VoIP Monitoring Manager for searching endpoints and viewing reports. It includes the following sections:

- **Before You Start Using the VoIP Monitoring Manager**
- **How to Start the VoIP Monitoring Manager**
- **Run a Search**
- **View Reports**
- **Export Reports**

Before You Start Using Avaya VoIP Monitoring Manager

Before you start the Avaya VoIP Monitoring Manager Client:

- Ensure that the Avaya VoIP Monitoring Manager Server and the Windows SNMP Agent are installed on the network.
- Configure the Switch Administration Forms (SAT)

Configure Switch Administration Forms (SAT)

There are two Switch Administration Forms (SAT) that need to be configured to send RTCP reports to the RTCP Monitor.

These forms are called:

- System-Parameters IP-Options Form
- IP-Network-Region Form

Configure IP- Network-Region Form

- Set the **RTCP Enabled?** field to **y** (yes).
- Set **Use Default Server Parameters?** field to **y** (yes). This indicates that this network region uses the default values specified previously on the system-parameters ip-options form as well.

Figure 3-1. IP Networks Region Form

```

1 | 2 |
IP Network Region

Region: 1
Name: Master RegionMaster

Audio Parameters
Codec Set: 1
Location:
UDP Port Range
Min: 2050
Max: 3027

Direct IP-IP Audio Connections? y
IP Audio Hairpinning? y

RTCP Enabled? y
RTCP Monitor Server Parameters
Use Default Server Parameters? y

DiffServ/TOS Parameters
Call Control PHB Value: 34
VoIP Media PHB Value: 46
BBE PHB Value: 43

802.1p/Q Enabled? n
Call Control 802.1p Priority: 7
VoIP Media 802.1p Priority: 6
802.1Q VLAN: 200

Resource Reservation Parameters
RSVP Enabled? y
RSUP Refresh Rate(secs): 15
Retry upon RSUP Failure Enabled? y
RSUP Profile: guaranteed-service
  
```

Do Not Want to Set Default Parameters to Yes

In some cases, you may not want to set the Default Parameters to yes, set the field to **n** (no) and specify the IP address of the Windows 2000 PC running the Server for that network region.

This situation might be where you have multiple Servers installed on a large system in order to reduce the network traffic between a set of endpoints and the RTCP Monitor (e.g. low bandwidth link between endpoints in one network region and a remote RTCP Monitor). The network traffic due to RTCP reports being sent from the endpoints to the RTCP Monitor is usually low, less than 40 bytes per second per currently active VoIP call (RTP session). Therefore, it is usually unnecessary to have multiple RTCP Monitors.

If multiple Servers are installed on the system then the endpoints in each network region can be configured to send their RTCP reports to different RTCP Monitors.

Configure System - Parameters IP-Options Form

- Set the **RTCP MONITOR SERVER, Default Server IP Address**, to the address of the Windows 2000 PC running the VoIP Monitoring Manager Server.

Figure 3-2. System-Parameters IP-Options Form

```

change system-parameters ip-options                                page 1 of 1

                                IP-OPTIONS SYSTEM PARAMETERS

IP MEDIA PACKET PERFORMANCE THRESHOLDS
Roundtrip Propagation Delay (ms)      High: 900      Low: 400
      Packet Loss (%)                 High: 40       Low: 15
      Ping Test Interval (sec): 20
Number of Pings Per Measurement Interval: 10

RTCP MONITOR SERVER
      Default Server IP Address: 123.123.123.123
      Default Server Port: 5005
      Default RTCP Report Period(secs): 5

```

How to Start Avaya VoIP Monitoring Manager

To Start Avaya VoIP Monitoring Manager

- From the machine where the VoIP Monitoring Server software is installed, select **Start > Programs > Avaya > VoIP Monitoring Manager > VoIP Monitoring Manager Server**.

Avaya VoIP Monitoring Manager Server starts.

- From the machine where the VoIP Monitoring Manager Client software is installed, select **Start > Programs > Avaya > VoIP Monitoring Manager > VoIP Monitoring Manager Client**.

Avaya VoIP Monitoring Manager Client starts. Now you can search for endpoints and then view the QoS data in a report format.

Search for Endpoints

The first action required when using the VoIP Monitoring Manager Client is to search for endpoints. The search dialog enables you to search for endpoints active in the past or between a date range. You can also use the advanced search options to narrow your search to match a specific phone number, network address or QoS value. Once you have completed your search, the Results lists updates with a list of endpoint(s) where you can select an endpoint from to view its Quality of Service (QoS) data in a report format.

To Run a Search

1. From the Search dialog, click the **In the last** drop down arrow to select a time period to search for active endpoints. The default is 1 minute but you can select hours, days, weeks or months.

If you want to select a date range, click **From** and then click the calendar(s) drop down arrow to select the start (**From**) and end date (**To**) of the range. You can select the day, months, hours, minutes, seconds and AM/PM.

2. Click **Search**. The Results list updates with a list of endpoints. Now, you can select an endpoint and view its report.

Once you run the search, you can view reports on selected endpoints and endpoints involved in a session.

View Reports

There are two types of reports, Summary Reports and Detailed Reports. Summary Reports display the QoS data as a reading on a gauge. The needle on the gauge shows the average values measured and the black inner arc shows the minimum and maximum values measured.

Detailed Reports show how the QoS values changes during the call and when this occurred. The upper values on the Y-axis indicate unacceptable limits. Each point on the line graph represents the maximum value since the last point displayed.

View Reports contd.

How To View a Report

1. From the Results List, select an **endpoint** or click on the expanding icon and select a child endpoint that was in a session with the parent endpoint. The Report button enables.
2. Click **Report**. The Report opens, displaying a summary of the QoS data.

To view a Detailed Report or the Session Properties, click its tab on top of the dialog.

View the Session Data

Reports can display both endpoints involved in a session. The reports display the parent endpoint involved in the session in the top part of the report with the child endpoint below.

The terms parent and child endpoints are purely for describing the way endpoints are displayed in the Results List. A parent is like the branch in a tree view. A child is like a leaf in a tree view. You will see the same endpoint can be shown as both a parent and a child. A parent endpoint is any endpoint listed as a result of a Search.

To View Sessions in a Report

1. Click on the expanding icon positioned in the far left column of the Results List. A sub list displays.
2. Select a **child endpoint** from the sub list.
3. Click the Report button. The reports display.

Export the Data

You can export the data in the results list and/ or a from a single report to a comma separated value (csv) file. You can open this file in most database and spreadsheet programs. Exporting the data to a spreadsheet enables you to manipulate the data so you can create your own reports. The data exported is divided into 3 tables:

- Session Table
- TimeStamped Data
- Trace Route Table

To Export Data

1. **File > Export Result List** or click the **Report button** located at the bottom of the **Result List**. A Save dialog opens.
2. Navigate to a folder.
3. In the **File name:** field, type a name for the file.
4. Click **Save as**. The file saves with the CSV extension.
5. From Microsoft Excel, open the file. From here you can build your own report.

If you want to export a single report, open your report and click the Report button located on the Report dialog.

- * **Note:** Microsoft Excel can only handle 65,536 rows of data. If you need to export more data, you will need to write a script that splits the data into smaller files before you import the data.

4 Interpreting Reports

This chapter provides a description on how to interpret the reports. It includes the following sections:

- **About Reports**
- **Interpreting Summary Reports**
- **Interpreting Detailed Reports**

About Reports

As explained in the previous section, there are two types of reports, Summary Reports and Detailed Reports.

Summary Reports

Summary Reports display the QoS data as a reading on a gauge. The needle on the gauge shows the average values measured and the black inner arc shows the minimum and maximum values measured. Each of the QoS parameters is displayed on a separate gauge, one for each of the three QoS parameters. You can alter these values using the Report Properties dialog accessed from the Edit menu.

Summary Report Features:

- Displays an average, minimum and maximum value for each QoS parameter
- Date range
- Type of endpoint
- Phone number and IP Address
- RSVP status
- Codec

Detailed Reports

Detailed Reports show how the QoS values change during the call and when this occurred. This is displayed on a line graph. The X-axis shows the time range and the Y-axis shows the value for each of the QoS parameters. The upper values on the Y-axis indicate unacceptable limits. Each point on the line graph represents the maximum value since the last point displayed.

Each of the QoS parameters is represented on the graph by a different color. This makes it easier for you to see the data on the same line graph. You can uncheck the display of one or more of the QoS parameters on the active line graph.

- Jitter is shown in red.
- Round Trip Time is shown in blue.
- Packet Loss is shown in brown

Detailed Report Features:

- Displays the QoS data as it changes during the call and shows when this occurred.
- QoS data is color coded.
- Ability to uncheck the display of one or more of the QoS data.
- A tool tip enables you to point your mouse at the data on the line graph to see the exact data measured.
- Alter the date range to show more or less detail.

Difference Between the QoS Data for an Endpoint and a Session

The QoS data that displays for an endpoint on the report is an average of all the sessions active at this endpoint.

As an endpoint can participate in multiple concurrent sessions, a high value on the report indicates that one or more of the sessions is experiencing degradation of quality. It does not indicate which session.

In contrast, session reports displays the QoS data as experienced by both endpoints for that session only. To assist with isolating your analysis, use the advanced search features to narrow down the search for a specific QoS value or alter the date range of the reports.

Interpreting the Values Using Summary Reports

You interpret the Summary Reports by noting where the needle on the gauge is positioned for each of the QoS gauges. When the needle is positioned in either the yellow or red ranges, it is indicating degradation in the QoS. The needle on the gauge shows the average values measured and the black inner arc shows the minimum and maximum values measured.

Figure 4-1. Summary Report

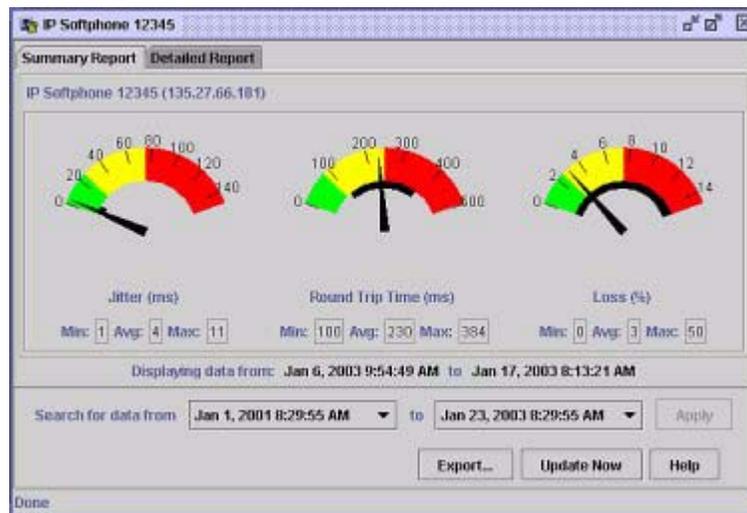


Table 4-1. The Values for the Summary Reports

Gauges	Acceptable (Green)	Warning (Yellow)	Not Acceptable (Red)
Jitter (ms)	0 to 50ms Conversation was smooth.	50 to 150ms Crackling, static or intermittent delay could be reported.	> 150ms
Round Trip Time (ms)	0 to 180ms No delay between each endpoint.	180 to 500ms Slight pause in the conversation if at the lower end of the range to more lengthy delays at the top end of the range could be reported.	> 500ms
Loss (%)	0 to 10% No drop out in conversation.	10 to 30% Drop out and missing parts of the conversation could be reported.	> 30%

Interpreting the Values Using Detailed Report

You interpret the Detailed Report by noting where the sampled points for each QoS value displays on the line graph and when this may have occurred. The upper values on the Y-axis indicate unacceptable limits.

Figure 4-2. Detailed Report

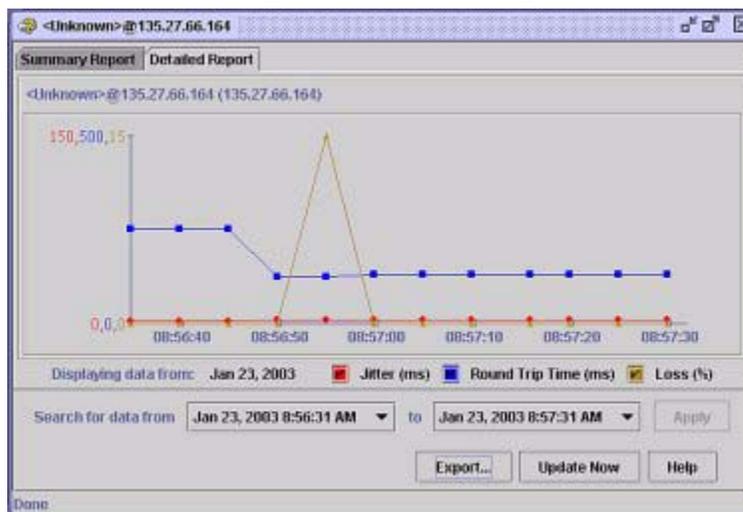


Table 4-3. The Values for Detailed Reports

QoS	Lower Acceptable	Middle Warning	Upper Not Acceptable
Jitter (ms) Displayed (Red)	0 to 50ms Conversation was smooth.	50 to 150ms Crackling, static or intermittent delay could be reported.	> 150ms
Round Trip Time (ms) Displayed (Blue)	0 to 180ms No delay reported.	180 to 500ms Slight pause in the conversation if at the lower end of the range to more lengthy delays at the top end of the range could be reported.	> 500ms
Packet Loss (%) Displayed (Brown)	0 to 10% No drop out in conversation.	10 to 30% Drop out and missing parts of the conversation could be reported.	> 30%

Glossary

This chapter provides a description of the key terms used in this document.

802.1D

The 802.1D metric represents the value of the 802.1p tag field and the 802.1Q of the incoming RTP packet. The 802.1D metric is a 16-bit unsigned integer. This metric is sent for the first RTP packet received, and resent when the 802.1D value has changed from the previously reported value.

Acoustic Echo Cancellation

The acoustic echo cancellation metric indicates if an endpoint is configured for full-duplex, half-duplex or acoustic echo cancellation mode. The acoustic echo cancellation metric is an enumerated type metric. The possible values are:

- 0 = Half-duplex
- 1 = Full-duplex
- 2 = AEC

AVAYA-VMON-MIB

The AVAYA-VMON-MIB is for the storage of RTCP data received from IP endpoints in the IP telephony system. (The ASN.1 definitions of this MIB and associated Traps are included as text files in the installation.)

Canonical Name (CNAME)

The canonical name or CNAME is the unique identifier for each participant within one RTP session, or set of related RTP sessions.

The format is user@host, or host if a user name is not available as on single-user systems. For both formats, host is either the fully qualified domain name or IP address of the host from which the real-time data originates. For Avaya VoIP systems CNAMEs are of the format:

- IP Telephone ext<extension>@<IP address>
- IP Softphone exs<extension>@<IP address>
- Gateway Board gwp@<IP address>
- Gateway Box gwt@<IP address>

Child Endpoint

The terms parent and child endpoints are purely for describing the way endpoints are displayed in the Results List. A parent is like the branch in a tree view. A child is like a leaf in a tree view. The same endpoint can be shown as both a parent and a child.

You click on the expanding icon positioned in the far left column of the Results List to expand the tree to display a sub list with the child endpoints. A child endpoint represents a session between itself and its parent. This is different from a parent endpoint that just represents a physical endpoint.

Codec

A Codec is an encoder/decoder. In the context of RTP, it is the type of encoding used for the payload of the RTP packets exchanged as part of a conversation. For example, some RTP Codecs are G.723, G.711 aLaw and G.729. The Codec used will be displayed just under the Title Bar on the reports.

DiffServ Code Point

The DiffServ Code Point (DSCP) metric represents the value of the IP DSCP field of the incoming RTP packets. The DSCP metric is a number in the range 0-63. This value is sent for the first RTP packet received, and resent when the DSCP value has changed from the previously reported value.

Echo Tail Length

The echo tail length metric represents the length of echo cancellation processing determined by the distance between the gateway and the endpoint. The echo tail length metric is represented in milliseconds and can have typical value ranging from 8ms to 32ms. The default value is 16ms.

EndTime

The EndTime column in the exported file displays the date and time the session ended. This column appears in the Session Table of the exported file.

Framesize

Frame size is the logical units into which data is partitioned for processing. In the case of a voice coder/decoder (codec) this is the time sliced blocks used by the codec algorithm. For example, the G.729 codec breaks the input audio signal into 10ms blocks for encoding purposes; therefore if the RTP packet payload is in 30ms blocks then there are 3 frames per packet.

VoIP Monitoring Manager displays the framesize in the Session Properties tab of the report dialog.

Gatekeeper

The Gatekeeper column in the Session Table displays the telephony switch that manage/administers the endpoint.

Gateway

A Gateway is generally used as a bridge between signaling protocols and bearer media. In this context, the Gateways allow IP endpoints to communicate with non-IP endpoints (e.g. the traditional circuit switched world of analogue and digital phones). AVAYA™ Gateways also perform the task of mixing the media channels in a conference call. A pair of Gateways can also be set up as an IP trunk.

VoIP Monitoring Manager

The Results List will display one or more phone numbers next to the Gateway endpoint type. These phone numbers are the phone numbers that the Gateway is acting as an intermediary for. Therefore, the phone number of the Gateway can change and can be multiple phone numbers. The Results List will separate endpoints involved in a session with a comma (,). Conferenced calls are separated by a colon (:).

For example, if the following phone number 8616,1111:1222, 8904 displays in the Results List then the Gateway has three active sessions as explained:

- Telephone 8616 is a Non-IP telephone which is in a session with a Softphone.
- Telephones 1111 and 1222 are conferenced (e.g. IP phone 8888 is in a Session with these two phones).
- Telephone 8904 is a Non-IP telephone in a session with an IP telephone.

HopAddress

The Hop Address column in the Trace Route Table displays the network node of each IP address in the trace route.

HopCount

The HopCount column in the Trace Route Table indicates the hop number, i.e. that is the position in the path of the trace route.

HopTime (ms)

The HopTime column in the Trace Route Table displays the results in milliseconds the round-trip-time of the trace route packet, from the source to each path in the trace route.

Jitter

Jitter is a measure of variance in the time it takes for communications to traverse from the sender (application) to the receiver, as seen from the application layer, or the difference between when a packet is supposed to be received and when it is actually received. We tend to think of jitter as the statistical average variance in delivery time between packets or datagrams. Jitter displays in milliseconds.

Jitter contd.**Removing Jitter**

Jitter can result from bad queuing strategies set-up on network equipment. Check your equipment manual for recommended settings. To remove jitter the endpoints need to collect packets and hold them long enough to allow the slowest packets to arrive, allowing them to be played at even intervals in the correct sequence, which causes additional delay.

Jitter Effects

Jitter can create audible voice-quality problems if the variation is greater than 60ms. Symptoms of excessive jitter could be reported as crackling or static. A faulty microphone or other hardware problems can be reported as a similar sound problem to jitter but they are not related. You need to rule out that this is not the cause of the problem.

Jitter Buffer Over Runs

The number of jitter buffer over-runs metric represents the number of times during a call the jitter buffer was too small. This metric is an 8-bit unsigned integer.

Jitter Buffer Under Runs

The number of jitter buffer under-runs metric represents the number of times during a call the jitter buffer became empty or starved. This metric is an 8-bit unsigned integer.

**Largest Sequence Fall
(LargestSeqFall)**

The Largest Sequence Fall metric represents the number of packets that are received from when an out-of-order packet was expected to be received.

* **Note:** A value of 0xFF implies that there were too many packets out of order to be able to calculate the correct value.

**Largest Sequence Jump
(LargestSeqJump)**

The Largest Sequence Jump metric represents the maximum number of consecutive packets lost in the last RTCP reporting interval.

For example, when the following packet sequence numbers 1,2,3,8, are received, the largest Sequence Jump is 4.

* **Note:** A value of 0xFF implies that there were too many packets lost to be able to calculate the correct value.

Maximum Jitter

The Maximum Jitter metric represents the maximum value of jitter seen in the RTCP reporting interval. This metric would be useful to identify transient spikes of jitter in a session. This metric is a 32-bit unsigned integer displayed in milliseconds. The unit is defined by the profile of the RTP session.

Media Encryption

The Media Encryption metric indicates whether media encryption is enabled or disabled for the RTP session. The Media encryption metric is an enumerated type metric. The possible values are:

- 0 = No encryption
- 1 = AEA1.2
- 2 = AES
- 3-255 = Reserved for future use

**Number Sequence Falls
(NumberSeqFalls)**

The number of sequence fall metric represents how many times during the RTP session there was at least 1 packet that was out of order.

**Number Sequence Jumps
(NumberSeqJumps)**

The number of sequence jump instances metric represents how many times during the reporting interval there was at least 1 packet which was lost.

Octet

The Octet column in the Session Table indicates the size of the packets.

Packet

A packet is the logical grouping of information that includes a header containing control information and (usually) the user data. The term *packet* is most often used to refer to the application layer data units.

Packet Loss

Packet Loss is the result of packets being lost in the transmission from one endpoint to another. When packet loss occurs there could be a drop out of words or partial words in the conversation. At low levels, poor voice quality would result. At high levels, the conversation becomes unintelligible. Packet Loss can result from line congestion.

Parent Endpoint

The terms parent and child endpoints are purely for describing the way endpoints are displayed in the Results List. A parent is like the branch in a tree view. A child is like a leaf in a tree view. The same endpoint can be shown as both a parent and a child.

A parent endpoint is any endpoint listed as a result of a search. You click on the expanding icon positioned in the far left column to expand the parent endpoint and show the child endpoints.

ParticipantID

The ParticipantID column assigns a unique identifier to each participant in the exported file. Each exported session has two participants. The exported data contains three sets of data. This data is listed in three separate tables that are separated by a blank row: Session Table, TimeStamped Data Table and the TraceRoute Table. For every session a participant was involved in there will be a unique pair: SessionID and ParticipantID, enabling you to associate the session data, timestamped data and the trace route data as belonging to that participant in a specific session. Use the ParticipantID to identify the participant in each table to analyze the data.

Payload

Payload refers to the contents of a packet. In RTP, it is encoded audio that is the user data of a packet.

Perceived Delay

Perceived delay is the total effect RTT and Jitter have on a phone user's conversation.

Quality of Service (QoS)

QoS is the measure of the level of quality that a service requires. The VoIP Monitoring Manager monitors and displays the 3 main factors that determine the quality of calls. These factors are Jitter, Round Trip Time, and Packet Loss. On the Summary Report each of the 3 factors display as a separate gauge. The Detailed Report displays the QoS as follows;

- Jitter is shown on the Detailed Report in red.
- Round Trip Time is shown on the Detailed Report in blue.
- Packet Loss is shown on the Detailed Report in brown.

RcvrIPAddr

The RcvrIPAddr column displays the IP address of this session participant (i.e. endpoint). This column appears in the Session Table of the exported file.

RcvrPHONE

The RcvrIPAddr column displays the phone number of the participant. This column appears in the Session Table of the exported file.

Real-Time Transport Control Protocol (RTCP)

A protocol providing support for applications with real-time properties, including timing reconstruction, loss detection, security, and content identification. It reports information about the RTP stream.

RTCP provides support for real-time conferencing for large groups within an Internet, including source identification and support for Gateways (like audio and video bridges) and multicast-to-unicast translators.

RTCP provides information about Round Trip Time, Jitter, Packet Loss and other data useful for analyzing voice quality.

Endpoints transmitting real time data send an RTP stream, which carries the actual data (e.g. audio, video). The endpoints also send a corresponding RTCP stream. For more information see RFC 1889 located at <http://www.ietf.org/rfc/rfc1889.txt>

Real-Time Transport Protocol (RTP)

Real-Time Transport Protocol is the protocol used for transmitting real-time data. For more information see IETF RFC 1889 located at <http://www.ietf.org/rfc/rfc1889.txt>

Resource ReSerVation Protocol (RSVP)

RSVP is a protocol for reserving network bandwidth on the routers and switches between two endpoints in a session (in some other protocol, such as RTP. There are two reservations per session, one for each direction the data has to travel. For further reference see the IETF RFCs 2205 and 2750 located at <http://www.ietf.org/rfc/rfc2205.txt>

Round Trip Time (RTT)

Round trip time is the length of time it takes a packet to traverse the network and return (thus being a round trip). It is the sum of the two one-way network delays between two endpoints. Callers can experience difficulties in carrying on a normal conversation when the one-way network delay exceeds 500 milliseconds (ms). However, some users may elect to tolerate this. It can comprise the following four components:

Propagation delay: The time it takes for a packet to travel across the network from sender to receiver. This variable is based on the speed of light and the distance the signal must travel. For example, the propagation delay between Singapore and Boston is much longer than the propagation delay between New York and Boston.

Transport delay: The time it takes to traverse the network devices along a transmission path. Networks containing many routers, firewalls, congestion and low-speed WAN services, for example, introduce more delay than an overprovisioned LAN on a single floor of a building.

Packetization delay: The time it takes for a compressor/decompressor (codec) to digitize an analog signal, build frames and then reverse the process at the other end. The G.729 codec has a higher packetization delay than the G.711 codec.

Jitter buffer delay: The delay introduced by the receiver while it holds one or more packets to reduce variations in packet arrival times.

RSVP Status

The RSVP status for an endpoint shows whether the RSVP is enabled on the endpoint, and if it is, whether a reservation was established for the received RTP data stream.

The RSVP status can change during a session. For example, if the RSVP status for a single endpoint in a session has changed between significant states (such as Failed and Success) then VoIP Monitoring Manager will use the label Various to represent this situation. However, if the status has only changed from Pending to Success, then VoIP Monitoring Manager will report Success. Also, the RSVP status can be different for each endpoint in the session. For example, RSVP may be disabled for one endpoint in the session, and enabled for the other.

The RSVP status can be:

- n **Unknown:** Information about the RSVP status was not available.
- n **Disabled:** The end-point has been configured to ignore RSVP signaling.
- n **Not in Use:** RSVP is enabled for use but there is no receiver RTP channel session active, or no attempt has been made by the sender to protect the receiver's RTP channel (i.e. no Path message has been received).
- n **Reservation Pending:** This state indicates that the receiver has responded to the first Path message it has received since the call started with a Resv message, and is waiting for a ResvConf to confirm the reservation is installed.
- n **Reservation Failed:** This state indicates that the receiver has had a reservation fail or timeout, or an existing reservation was torn down prematurely.
- n **Reservation Success:** This state shows that the receiver's receiving RTP channel is protected by an installed RSVP reservation. Ideally this reservation will need to be successfully refreshed until the RTP session ends.
- n **Various:** The RSVP status for a single endpoint in a session has changed between significant states (such as Failed and Success).

RTCP Listen Port

The RTCP Listen Port is the configurable port that is used to collect the AVAYA™ endpoints. The default port is 5005. You can edit the port that is used in the RTCP Monitor Properties dialog.

RTP MIB

The RTP MIB stores the information for the active RTP Sessions. The reference for the definition of the RTP MIB is located at <http://www.ietf.org/rfc/rfc2959.txt>

RTP Session

A session is a VoIP connection between two IP endpoints. For more information see RFC 1889 located at <http://www.ietf.org/rfc/rfc1889.txt?number=1889>

SessionID

The SessionID column assigns a unique identifier to each session in the exported file. Each exported session contains three sets of data. This data is listed in three separate tables that are separated by a blank row: Session Table, TimeStamped Data Table and the TraceRoute Table. Use the SessionID to identify the session in each table to analyze the data.

Silence Suppression

In Voice over IP (VoIP), silence suppression is a method of detecting the silence in audio and purposefully dropping silent packets at the sender to conserve network bandwidth. The receiver will generate comfort noise or conceal the loss of packets when packets are dropped. Because the receiver conceals loss and generates comfort noise, silence suppression is usually imperceptible to the listener. The Silence Suppression field will be reported as enabled, disabled or unknown.

Session Table

The Session Table is one of the exported tables containing data related to the Session. It will display in Microsoft Excel at the top of the same worksheet as the TimeStamped and Trace Route Table. It contains the following information:

- SessionID
- ParticipantID
- StartTime
- EndTime
- RcvrIPAddr
- CNAME
- RcvrPHONE
- TOOL
- Payload
- Gatekeeper
- DiffServ Code Point
- 802.1D
- Media Encryption
- Echo Tail Length
- Silence Suppression
- Acoustic Echo Cancellation

StartTime

The StartTime column in the exported file displays the date and time the session started. This column appears in the Session Table of the exported file.

TimeOffset

The TimeOffset column displays the number of seconds since the session started for this set of data. This column appears in the TimeStamped Data Table of the exported file.

TimeStamped Data Table

The TimeStamped Data Table is one of the exported tables containing data related to the call. The table displays all the data associated at a given point in time. The time is measured from the beginning of the session. It will display in Microsoft Excel below the Session Table on the same worksheet. To view the information more easily, you may want to copy the table and paste it to another worksheet.

The SessionID and the ParticipantID columns link the information in the TimeStamp Data Table to the Session Table. If you have a session of interest in the Session Table, use the Ids to analyze the data.

The TimeStamped Data contains the following information:

- SessionID
- ParticipantID
- TimeOffset
- Jitter
- RTT
- LostPackets
- Packets
- RSVP
- LargestSeqJump
- NumberSeqJumps
- LargestSeqFall
- NumberSeqFalls
- Time-To-Live
- MaxJitter
- JitterBufferOverRuns
- JitterBufferUnderRuns

Time-To-Live (TTL)

Time-to-live (TTL) is a value in an Internet Protocol (IP) packet that tells a network router if a packet has been forwarded towards its destination too many times and should be discarded. For a number of reasons, packets may not get delivered to their destination in a reasonable length of time. For example, a combination of incorrect routing tables could cause a packet to loop endlessly. A solution is to discard the packet after the packet has been forwarded a certain number of times and send a message to the originator, who decides whether to resend the packet.

The initial TTL value is set, usually by a system default, in a field of the IP packet header with a value in the range 0 to 255. The original idea of TTL was that it would specify a certain time span in seconds that, when exhausted, would cause the packet to be discarded.

Since each router is required to subtract at least one count from the TTL field, the count usually indicates the number of router hops the packet has remaining before it must be discarded. Each router that receives a packet subtracts one from the count in the TTL field. When the count reaches zero, the router detecting it discards the packet and sends an Internet Control Message Protocol (ICMP) message back to the originating host.

TOOL

The TOOL value is the name and version of the application generating the stream, e.g., Avaya VoIP Engine v.123. This information may be useful for diagnosis. The TOOL value should remain constant for the duration of the session.

Trace Route Table

The Trace Route Table contains data related to the route of the call. It will display in Microsoft Excel below the TimeStamped Data Table. It contains the following information:

- SessionID
- ParticipantID
- HopCount
- HopTime (ms)
- HopAddress

Trap or Alarm

A Trap or Alarm is a message sent by a Windows SNMP Agent to a Trap Manager, console, or terminal to indicate the occurrence of a significant event, such as a specifically defined condition or a threshold that was reached. It is also referred to as an Alarm. The Trap Manager is generally configured to be the Gateway Alarm Manager (GAM) or Network Alarm Manager (NAM) but any Trap Manager application can be used with the AVAYA(tm) VoIP Monitoring Manager.

VoIP or Voice over Internet Protocol

VoIP is an acronym for Voice over Internet Protocol. This is the technology standard that allows Internet telephony. It provides the capability for live voice communication over the Internet so that you can talk using the multimedia capabilities of your computer, in the same way you would talk using a telephone.

Windows SNMP Agent

The Simple Network Management Protocol (SNMP) Agent is the Windows SNMP service that runs on your computer. SNMP is a protocol for communications between remote network management stations and managed network elements (such as AVAYA(tm) devices).

The VoIP Monitoring Manager Server needs the Windows SNMP Agent installed as it enables the RTCP Monitor to collect and publish the data. The Windows SNMP service is provided with the Windows 2000 CD but is not installed by default. You will be prompted during the VoIP Monitoring Manager install to install it from the Windows 2000 CD.

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