SOLARWINDS WHITEPAPER

Fundamentals of VoIP Call Quality Monitoring & Troubleshooting

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Introduction

Voice over IP, or VoIP, refers to the delivery of voice and multimedia over IP networks. The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream.[1]

Since the transmission of the packet occurs through an IP network, the quality of communications is inherently less reliable than a circuit switched public telephone network. There is no network mechanism to ensure that data packets are not lost and are delivered in sequential order. Essentially, it is a best effort without Quality of Service (QoS) guarantees.

Several metrics are available for measuring VoIP call quality. This paper will provide a basic understanding of VoIP traffic and of the quality metrics used to monitor VoIP calls.

VoIP Metrics

Maintaining high quality VoIP calls can be difficult as VoIP is more sensitive to network delays and packet loss compared to any other network applications. VoIP quality is measured based on the following metrics:

- Network Jitter and Delay
- Packet data Loss
- Latency
- Mean Opinion Score (MOS)

Network Jitter and Delay

Real-time voice communications over the network are sensitive to delay in packet arrival time or packets arriving out of sequence. Excess jitter results in calls breaking up. Jitter can be reduced to a certain extent by using jitter buffers. Jitter buffers are small buffers that cache packets and provide them to the receiver in sequence and evenly spaced for proper playback. Buffer lengths can be modified, however, if jitter buffer is increased too much then the call will experience an unacceptable delay. Consequently, a reduction in buffer turns results in less delay but more packet loss. Jitter is measured in milliseconds (ms).

Packet data Loss

Packet loss occurs when one or more packets of data fail to reach their destination. A single packet loss is referred to as “packet gap”, and series of packet loss is known as” burst”. Packet loss can occur for a variety of reasons including link failure, high congestion levels, misrouted packets, buffer overflows and a number of other factors. Packet loss causes interrupted playback and degradation in voice quality. Packet loss can be controlled using packet loss concealment techniques within the playback codec.

Latency

Latency, or lag, is the time delay caused in the transmission of a voice packet. Excess latency results in delay and finally to echo. Latency is measured in milliseconds (ms)

Mean Opinion Score (MOS)

Mean Opinion Score is a numerical value to indicate the perceived quality of the call from the user’s perspective of the received call after compression, transmission, and decompression. MOS is a calculation based on the performance of the IP network and is defined in the ITU-T PESQ P.862 standard and is expressed as a single number in the range of 1 to 5, where 1 is lowest perceived quality and 5 is the highest.

Desirability Scale	R-Factor	MOS Value
Desirable	94 - 80	4.4 - 4.0
Acceptable	80 - 70	4.0 - 3.6
Reach Connection	70 - 50	3.6 - 2.6
Not Recommended	50 - 0	2.6 - 0

R-Factor

The R-factor is a score that takes into account both user perceptions (User R-factor) and the effect of equipment impairments (Network R-factor) to arrive at a numeric expression of voice quality. These metrics are calculated by a formula that balances all equipment impairments and perception factors and is reported as a single number on a per-call basis, typically in the range of 15 to 94.The MOS and R-Factor helps in determining the QoS. The R-factor is calculated using a basic formula called E-Model as shown below.

R-Factor = Ro – Is – Id – Ie + A

R-Factor: Overall network quality rating (ranges between 0 and 100)

Ro: Signal to noise ratio

Is: Impairments simultaneous to voice signal transmission

Id: Impairments delayed after voice signal transmission

Ie: Effects of Equipment (MOS is converted to Ie)

A: Advantage factor (attempts to account for caller expectations)

Call Detail Records (CDRs)

A CDR is a data record that documents the details of a phone call and fall into one of two classifications: the basic CDR, and the diagnostic CDR or CMR (Call Monitor Records). The basic CDR includes information about the call such as originating phone number, receiving phone number, call starting time, call duration, billing details, and other call details. The CMR includes metric information needed for monitoring VoIP calls including addresses, status (successful, failed, transferred), number of packets, packet bytes, packet loss, start time, initial setup duration, duration, current jitter, maximum jitter, MOS, R-factor, QoS for each call, number of packets that arrive out of order, detailed analysis for packet loss and delay, gap and burst measurements, and long-term Call Detail Records trending.

Monitoring VoIP Performance

In order to properly troubleshoot VoIP call quality, you should have a number of tools at your disposal that monitor and measure the critical call quality components. VoIP monitoring typically falls into one of the following categories:

- Network monitoring tools that monitor VoIP performance based on network performance statistics
- Protocol analyzers
- Dedicated VoIP tools
- Synthetic VoIP traffic generators

Network Monitoring Tools

Network monitoring tools are non-intrusive, or passive, and examine each stream of voice traffic across the network to estimate the MOS score and calculate the R-Factor. The use of network monitoring tools has the advantage that all calls in the network can be monitored without any additional network overhead. For the IT professional, monitoring VoIP using network monitoring tools is essentially just another network task and, therefore, typically the most practical approach.

Protocol Analyzers

Protocol analyzers are hardware or software tools that capture and analyze VoIP traffic packets and calculate jitter, and latency directly from the packet stream.

Dedicated VoIP Tools

Most dedicated VoIP tools were originally developed for the telecomm industry and are great for testing IP phone and gateway designs but not as good at solving deployment problems within the network.

Synthetic VoIP Traffic Generators

Synthetic VoIP traffic generators reduce time to the readiness of VoIP services, while assuring they meet quality requirements as perceived by users.

Troubleshooting VoIP Quality Issues

Quality VoIP calls require a quality IP network that can deliver voice packets within the minimum requirements around jitter, packet loss, and latency. Network monitoring systems and specifically network traffic monitoring systems can help the IT professional better understand the impact of network performance on VoIP call quality. At a minimum, you need to analyze at the end points (both call origination point and call destination point), however, measuring at additional points within the call path will help further narrow the potential problem areas.

Step 1: Check the error/status code to indicated failed calls–see error description. (In case of a fail call go to step 4)

Step 2: Check MOS score. If it’s bad, try to find calls placed at the same time and verify if they MOS was similar

Step 3: Analyze packet loss, latency or jitter in order to see what could be a potential problem (slow network, unreliable network, overloaded routers or switches)

Step 4: If possible do comparison of the calls that were placed between the same locations at the same time to confirm symptoms

Step 5: Look for the related gateways and call managers and their performance status (usually via call connection topology)

Step 6: Try to see call path and all related network element like routers and switches and see their performance metrics.

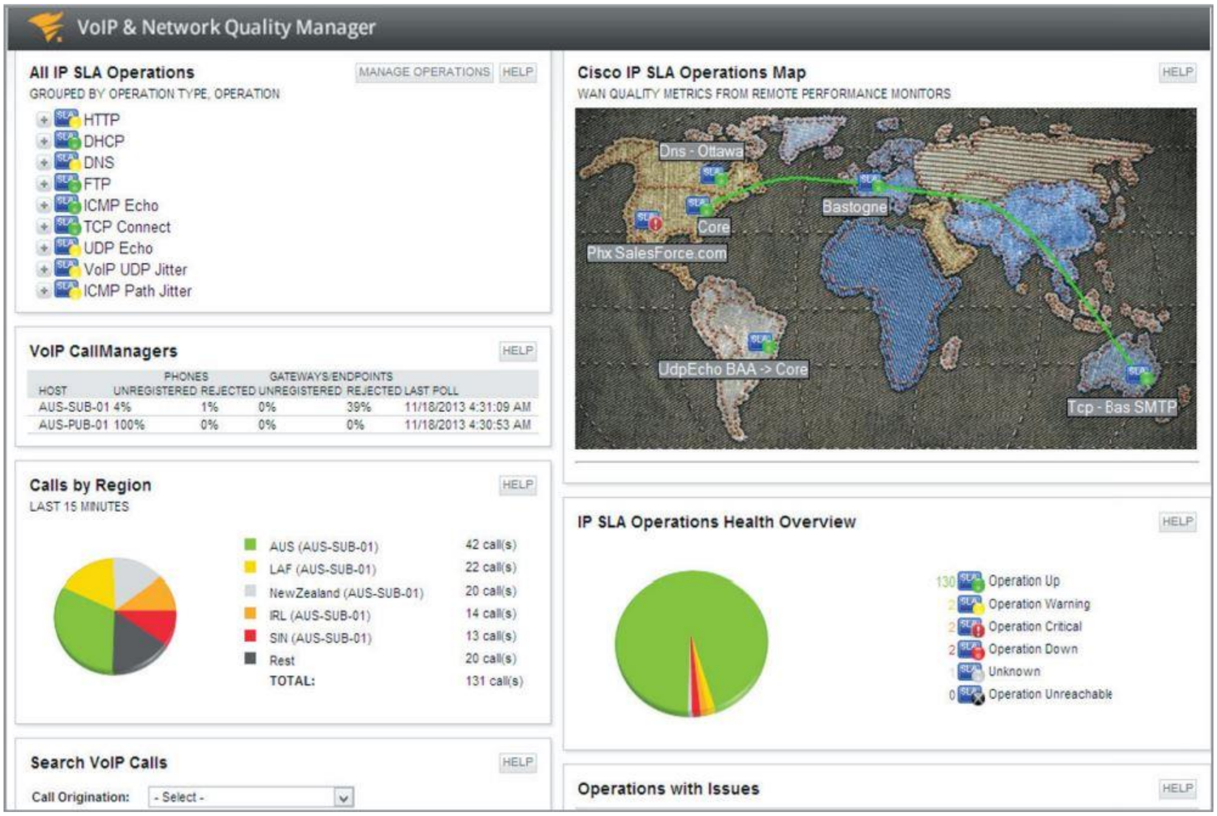
Step 7: Verify channel utilization at the time of the call.

Step 8: Verify call related end-point devices (soft IP phones, HW phones)

Step 9: Establish VoIP synthetic operation between the same locations to prevent the same issue and continuously monitor situation

SolarWinds VoIP & Network Quality Manager Can Help!

Stay a step ahead of VoIP Call Quality issues and end-user complaints within your Avaya® VoIP Environment. SolarWinds VoIP & Network Quality Manager monitors VoIP performance by analyzing Call Detail Records generated by Avaya Communication Manager and helps you proactively identify and eliminate distortion, latency, and noise. It also generates synthetic VoIP traffic using Cisco® IP SLA technology and facilitates capacity planning and measurement of voice quality in advance of new VoIP deployments. But that's not all! SolarWinds VoIP & Network Quality Manager tracks key edge router and switch statistics and helps you keep a close eye on site-to-site WAN performance.



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Feature Highlights

Monitor VoIP Call Performance

SolarWinds VoIP & Network Quality Manager monitors the performance of VoIP calls by analyzing call detail records (CDRs) generated by Avaya Communication Manager. Quickly view jitter, latency, packet loss, and MOS between Avaya IP Phones.

Troubleshoot VoIP Call Performance

SolarWinds VoIP and Network Quality Manager Enables advanced VoIP troubleshooting by correlating individual call performance with corresponding network performance metrics.

Search & Filter Call Detail Records

SolarWinds VoIP & Network Quality Manager provides the ability to search, filter, and display call detail records (CDRs) to aid in troubleshooting.

VoIP & WAN Monitoring Dashboards

SolarWinds VoIP & Network Quality Manager provides customizable at-a-glance insight into all aspects of your VoIP and WAN performance through an intuitive LUCID™ Web interface.

VoIP Gateway Monitoring

SolarWinds VoIP & Network Quality Manager provides Avaya VoIP gateway performance details including distribution of VoIP and data, and a list of the top 10 quality issues through a designated gateway so you can see exactly how your VoIP capacity is being used and identify potential issues before they impact users.

Call Signaling Chart View

SolarWinds VoIP & Network Quality Manager provides a pictorial representation of the packets exchanged from the call initiation, during progress and end of call, as well as displays the requests that took place from call start to call end.

Customizable Performance Reports

VoIP and Network Quality Manager makes it easy to generate VoIP and WAN performance reports using out-of-the-box templates that can be customized with a few mouse clicks, automating report creation and distribution.

Quick, Do-it-Yourself Deployment

Download, install and deploy VoIP & Network Quality Manager in less than an hour using three simple steps.

References

[1] Voice over IP. (2012, June 6). In Wikipedia, The Free Encyclopedia. Retrieved 15:35, June 12, 2012, from http://en.wikipedia.org/w/index.php?title=Voice_over_IP&oldid=4962430

About SolarWinds

SolarWinds (NYSE: SWI) provides powerful and affordable IT management software to customers worldwide from Fortune 500 enterprises to small businesses. In all of our market areas, our approach is consistent. We focus exclusively on IT Pros and strive to eliminate the complexity that they have been forced to accept from traditional enterprise software vendors. SolarWinds delivers on this commitment with unexpected simplicity through products that are easy to find, buy, use and maintain while providing the power to address any IT management problem on any scale. Our solutions are rooted in our deep connection to our user base, which interacts in our online community, [thwack®](), to solve problems, share technology and best practices, and directly participate in our product development process. Learn more today at http://www.solarwinds.com.