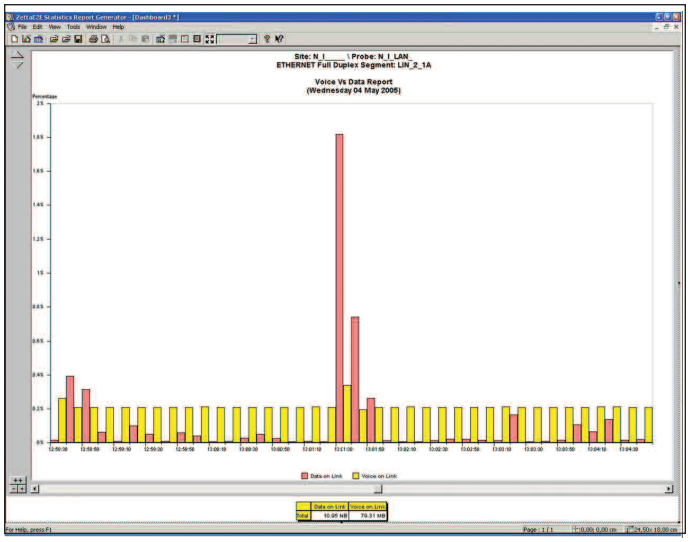
ZettaVoIP

Voice over IP Performance Monitoring



KEY BENEFITS

- Monitor Real Time Protocol (RTP stream)
- Monitor key Mean Opinion Score (MOS) parameters including Availability, Latency, Throughput, Jitter and Packet Loss with a single tool
- Over 300 pre-defined reports
- Monitor voice and data with one system

VoIP is poised for a breakthrough. Many corporate networks and telcos have already implemented VoIP and even more are convinced of its attraction as a more flexible and cost-effective replacement of conventional PSTN services.
A positive user perception of voice quality and related services is critical to the success of VoIP and this will require effective, end-to-end network testing and monitoring.

Monitoring and maintaining a VoIP network is no simple task. Behind the perceived simplicity of VoIP appear a number of complex issues such as security, quality and interoperability in carrier grade solutions. The only way to really gain a detailed knowledge about the correlation between voice service quality and transient network behavior is to track (ideally network wide) both network behavior and service behavior with a high degree of granularity. Analysis of this combined set of data enables identification of the service degrading parts of the network. Because of the fluctuating behavior over time of IP networks, nothing less than continuous monitoring and analysis of voice service metrics and network performance metrics can ensure a full understanding of the network's ability to deliver a toll level of voice service quality.

The ZettaView Solution

Introducing ZettaVoIP, the performance monitoring solution for VoIP networks. As part of the world class ZettaView family, ZettaVoIP features non-intrusive hardware for seamless integration with existing networks. Well established ZettaView applications such as protocol analysis, alarm management and customized reporting are also supported for VoIP. In fact, all core ZettaView solution components for VoIP re-use the exact same architecture as the ZettaView systems already deployed world-wide, giving users the advantage of well-proven, mature applications.

ZettaVoIP
Monitor Key Quality of Service (QoS) and Service Level Agreement (SLA) Parameters:

Availability: Time in which the network has enough bandwidth available to transmit the packets.

Jitter: While network latency affects the length of time a voice packet spends in the network, jitter controls the regularity in which the packets arrive.

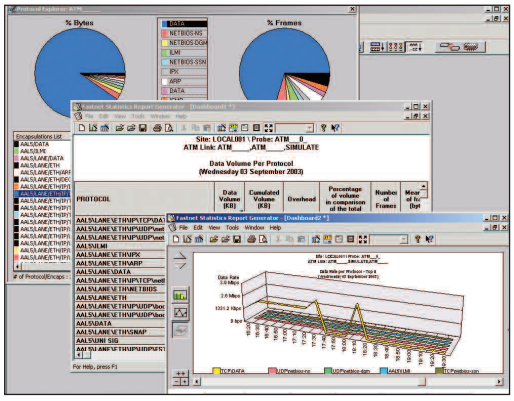
Latency: The time from the source to the destination is known as latency, or delay.

Packet loss: The lost packets must be retransmitted and this adds to the total transmission time.

Throughput: This is the effective data transfer rate measured in bits per second.

Protocols Supported: Protocol Analyzer Decodes	
Signaling	
H.323	H.323: Packet-based multimedia communications (VoIP) architecture H.225: Call Signaling and RAS in H.323 VoIP Architecture H.235: Security for H.323 based systems and communications H.245: Control Protocol for Multimedia Communication
MGCP	MGCP: Media Gateway Control Protocol
SIP	SIP: Session Initiation Protocol SDP: Session Description Protocol SAP: Session Announcement Protocol
Cisco Skinny	SCCP: Skinny Client Control Protocol
Media	RTP: Real Time Transport Protocol RTCP: RTP Control Protocol
Codec	G.7xx: Voice Compression Protocols (G.711, G.721, G.722, G.723, G.726, G.727. G.728, G.729) H.261: Video Coding and Decoding (CODEC) H.263: Video Coding and Decoding (CODEC)
Others	COPS: Common Open Policy Service RTSP: Real Time Streaming Protocol SCTP: Stream Control Transmission Protocol TRIP: Telephony Routing Over IP

VoIP Results



Once the information is captured, who has time to compile it? With ZettaVoIP, http based reports can be customized and generated with a few mouse clicks.

Historical Statistics

Historical Data Reports

Total RTP data volume
- Volume In
- Volume Out
- Cumulative

Total Signaling data-volume by signaling protocol
- Volume In
- Volume Out
- Cumulative

Max RTP packet loss
- Packets In
- Packets Out
- Cumulative

Maximum and average Jitter
- Average Jitter In
- Average Jitter Out
- Average Cumulative Jitter
- Max Jitter In
- Max Jitter Out
- Cumulative Max Jitter

Average & Maximum amount of RTP streams per 10s, 10 min/day
- Average Number of RTP Streams In
- Average Number of RTP Streams Out
- Average Cumulative RTP Streams
- Max Number of RTP Streams In
- Max Number of RTP Streams Out
- Cumulative Max Number of RTP Streams

Max RTP packet loss
- Packets In
- Packets Out
- Cumulative

Total amount of RTP streams/day
- Total Number of RTP Streams In
- Total Number of RTP Streams Out
- Cumulative Number of RTP Streams

Real Time Statistics

VoIP Explorer

- Source IP Address
- Destination IP Address
- Protocol
- Throughput
- Avg Round Trip Time (Latency) - From Source (1) to Destination (2)
- Avg Round Trip Time (Latency) - From Destination (2) to Source (1)
- Max Round Trip Time (Latency) - Total
- Bytes from Source (1) to Destination (2)
- Bytes from Destination (2) to Source (1)
- Jitter
- Packet Loss
- Volume 1 -> 2
- UDP Source
- UDP Destination
- Codec ID
- Stream Duration

Reports: (Histograms)

- Average Latency over 10 minute time range
- Network Unavailable (10s)
- Throughput
- Voice vs. Data on Link Report Comparison
- Voice vs. Data on Link Utilization Report
- Average RTP data-rate of all streams, 10s and 10 min
- Average Signaling data-rate, 10s and 10 min
- Average and Max RTP packet loss, 10s and 10 min.
- Average and Max Jitter, 10s and 10 min.
- Average Number of RTP streams per 10s and 10 min

Alarm/Events

- Network Unavailable
- Packet Loss
- Throughput Unavailable
- RTP per stream packet loss threshold
- RTP per stream max jitter threshold exceeded
- RTP Stream duration (defined by SSRC)
- RTP active streams > x (defined by SSRC)

ZettaView
"Know Your Network"

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