

Call Trace & Recording



Fraud Detection / Location



Remote Analysis and Troubleshooting



Real-time Monitoring



Traffic Optimization / Engineering / Statistics



Revenue Billing Verification



Alarm Monitoring



Quality of Service Measurements



SIP, RTP, MEGACO, CAS, ISDN, SS7, GSM, GPRS, and more



Network Wide Testing Solutions

TDM, VoIP, & Wireless Networks



In today's networks whether TDM, VoIP, or Wireless, network wide testing and monitoring is becoming more important than ever before. Performance monitoring, security, fraud detection, alarm monitoring, billing verification, remote protocol analysis, failure prediction, and traffic engineering are some network aspects that need to be monitored continuously. A network operator or service provider must have the means to perform the above surveillance tasks cost-effectively, remotely, automatically, and non-intrusively. Fortunately, the network backbone contains a wealth of information that can be monitored and collected to support these activities.

GL provides a variety of solutions for network wide monitoring and surveillance.

The solutions consist of:

- Intrusive and non-intrusive 'PC Probes' for TDM, VoIP, and Wireless networks
- Probes deployed at strategic locations in a network transmit and collect voice, data, protocol, statistics, and performance information, and relay this information to a central / distributed network management system (NMS)
- NMS may be client-server based or WEB based system and consists of a database and applications for controlling, collecting, and analyzing the information provided by the various probes

GL's current VoIP NMS solutions are:

- Wireless, Wireline, and VoIP Voice Quality Monitoring System (intrusive)
- Packet and VoIP Monitoring and Surveillance System (non-intrusive)

For more details, please visit our web page <http://www.gl.com/networkmonitoring.html>.



GL Communications Inc.

818 West Diamond Avenue - Third Floor. Gaithersburg, MD 20878 • (V) 301-670-4784 (F) 301-670-9187

Web Page Address: <http://www.gl.com/> • E-Mail Address: gl-info@gl.com

Active / Intrusive Network VQT System for VoIP

This system provides real-time voice quality measurement across a diverse set of networks. Voice calls are automatically placed between end points; quality is measured and provided for display at an NMS. Voice measurements include MOS (Mean Opinion Score), round trip delay (RTD), jitter, clipping, voice levels, etc. Visit <http://www.gi.com/netvoicequality.html> for more details.

The essential elements of this system are:

- Intrusive VoIP Probes** Use GL's **VQuad™**, or **PacketGen™** to establish calls and send / receive voice files in real-time in an end-to-end manner. Received voice (degraded) files are transmitted to VQT software for analysis. Specific nodal hardware is available for connection to wireless phones with automatic call control.
- Voice Quality Testing (VQT)** software compares the two files ('reference' and 'degraded') and provides an ITU-standard score (PESQ, PESQ WB, PAMS, & PSQM).
- The **Regional Command Center (RCC)** controls the "Nodes". It controls, schedules, and analyzes the degraded voice traffic received by the nodes. Advance features include real-time monitoring, scripting of voice calls, scheduling of tests, voice quality testing, round trip delay (RTD), post dial delay (PDD) calculations, and much more. The RCC includes a database for storage and retrieval.
- The **Remote Client VQT NetViewer™** application remotely controls the RCC, individual VQuad™ node sites and the VQT Measurement process. The system facilitates result display using a simple web browser application **VQT WebViewer™**.

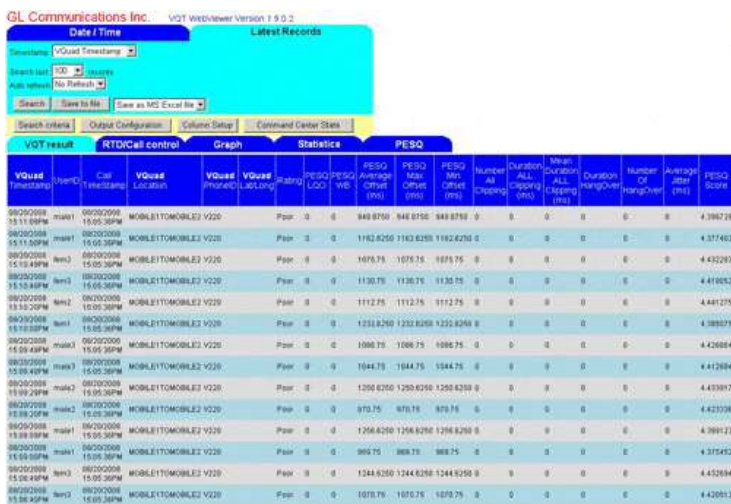
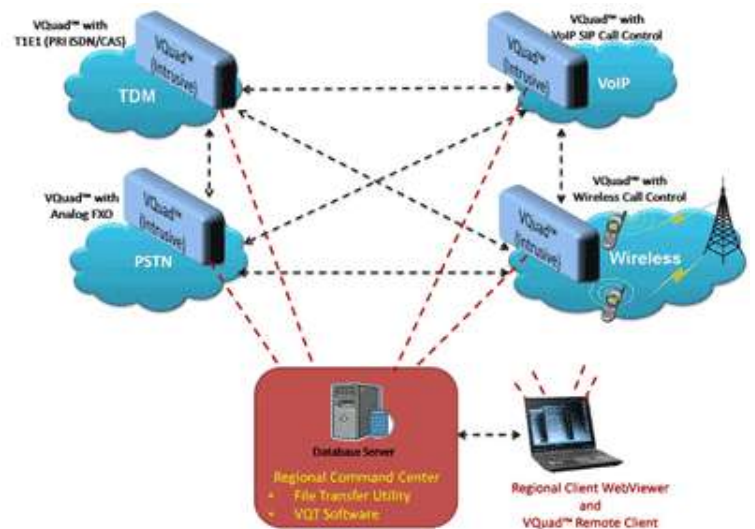


Figure: VQT WebViewer™ - Graphical View

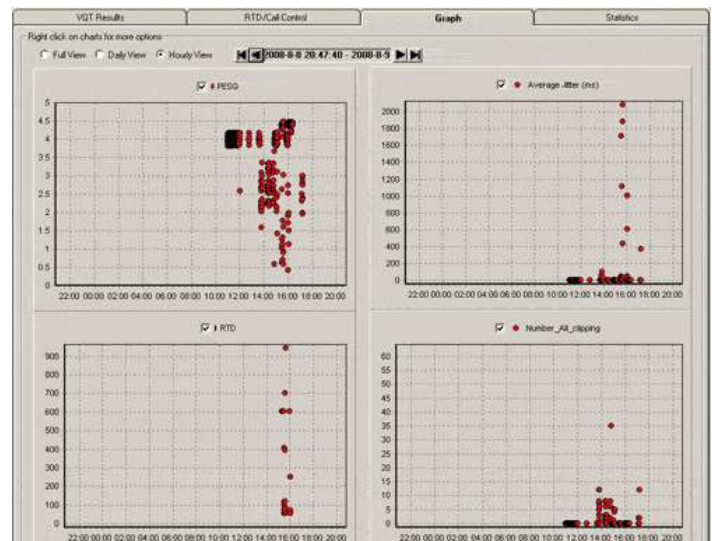


Figure: VQT NetViewer™ - Graphical View

VoIP NMS Solution 2

Passive / Non-Intrusive Voice-band Monitoring System for VoIP

GL's PacketScan™ with Voice Band analyzer (VBA) performs detailed analysis of voice band streams gathering QOS statistics such as E-model based MOS (Mean Opinion Score), R-factor, total packet count, reordered, duplicate and missing packet counts, gap, jitter, and delay. Visit <http://www.gl.com/netvoip.html> for more details.

- All major VoIP protocols (SIP, H.323, Megaco, and MGCP) and codec, including video RTP streams are supported.
- Host of graphs - Gap, Jitter, Gap Distribution, Jitter MOS, Quality, Wave and Spectral Display for media stream analysis of jitter, delay, packet-loss, sequencing, and more.
- A central database stores the real-time and historic data. A web-server access the data and allows clients across WAN to view results.
- GL's Voice Band analyzer (VBA) works in conjunction with GL's PacketScan™ (VoIP Analysis Tool) to monitor the quality of voice band traffic over VoIP. With appropriate modules, one can perform P.56 Active Voice Level analysis, Noise analysis, Hybrid, Line, and Acoustic echo analysis. Other analysis modules such as ITU-T P.561, P.562, P.563, fax and modem analysis, and many others can be hosted as plug-ins and will be available soon.
- The system also facilitates result display using a web interface, called as PacketScanWeb™. With this, one can view real-time data, navigate through records, filter the collected VoIP traffic summary, and graphically analyze the call volume, MOS, call completion, failed calls, completed calls, PDD, and so on through a simple web browser.

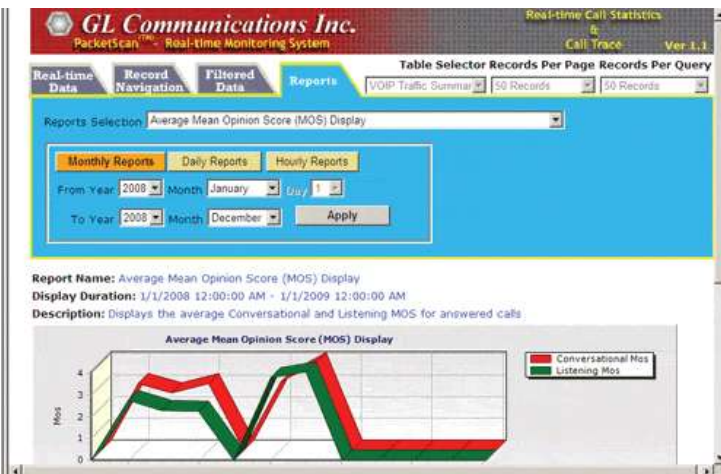
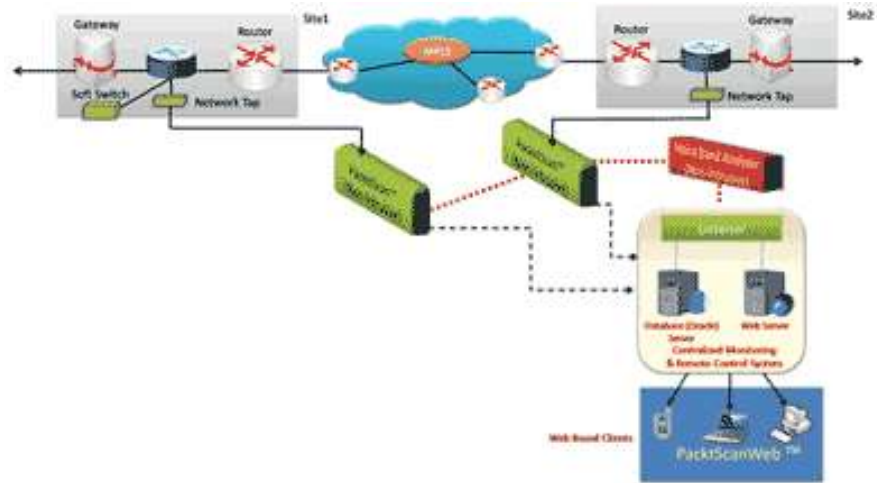


Figure: PacketScanWeb™

Call ID	QIR	Probe Name	Start Time	Classed Time	Speech Lvl./% Active	RMS/Noise Lvl	Chn	DC Offset	Line Echo FdB/Line	Acoustic Echo FdB/Line
137156		SanDiego_Line3	2008-08-19 05:26:01	23.67	-41.5/73	-42.9/-62.2	0	-1	-21.1/70	0/0
137158		SanDiego_Line3	2008-08-19 05:26:01	25	-21.5/72.5	-22.6/-43.5	0	-9	0/0	0/0
137157		SanDiego_Line3	2008-08-19 05:24:00	188	-26.2/38.8	-30.3/-91	0	-4	0/0	0/0
137157		SanDiego_Line3	2008-08-19 05:24:00	188	-44.6/37.3	-48.9/-98.8	0	0	-20/91	0/0
137156		SanDiego_Line3	2008-08-19 05:22:01	190	-36.6/68.5	-38.3/-101.3	0	0	-18.3/62	0/0
137156		SanDiego_Line3	2008-08-19 05:22:01	190	-22.7/69.9	-24.2/-59.7	0	-1	0/0	0/0
137155		SanDiego_Line3	2008-08-19 05:20:14	200	-23.7/32.3	-26.6/-64.1	0	0	0/0	0/0
137155		SanDiego_Line3	2008-08-19 05:20:14	200	-25.5/43.4	-29.1/-67.9	0	0	0/0	0/0
137154		SanDiego_Line3	2008-08-19 05:18:06	185	-25.3/54.5	-28/-57.5	0	11	0/0	0/0

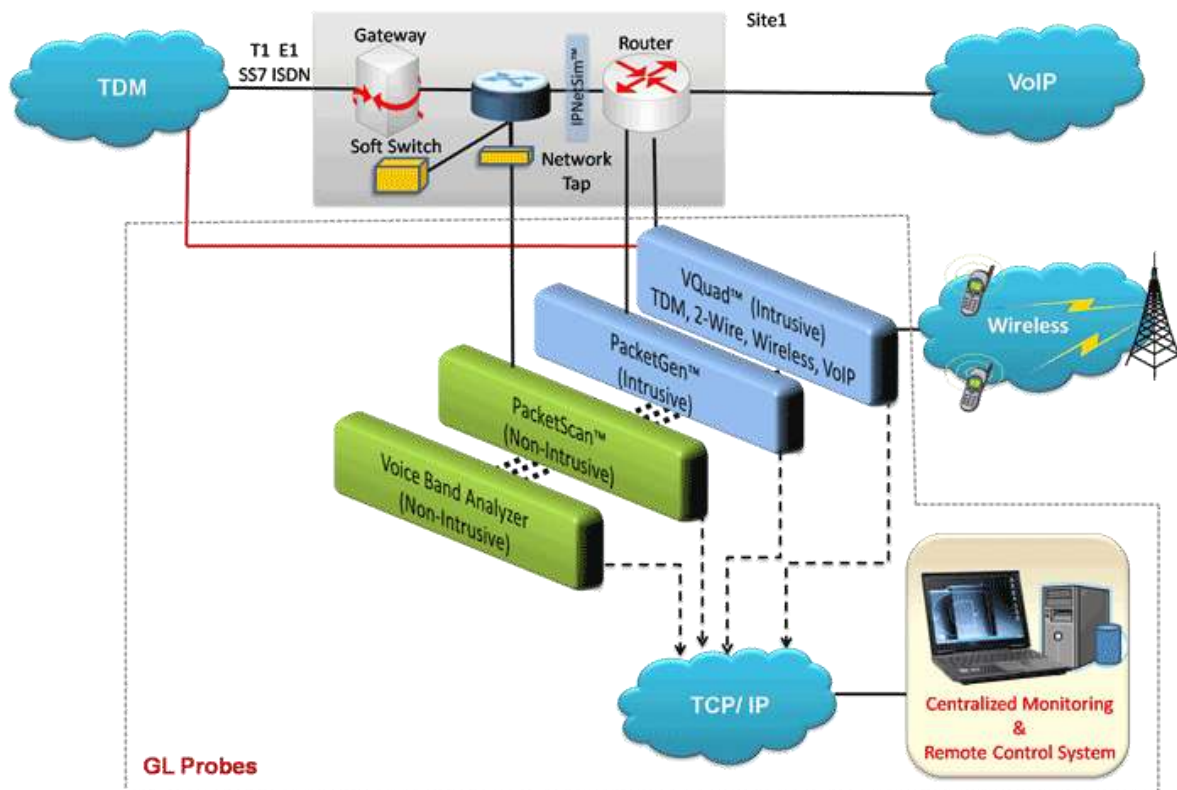
Figure: VBA WebViewer™

 **GL Communications Inc.**

818 West Diamond Avenue - Third Floor. Gaithersburg, MD 20878 • (V) 301-670-4784 (F) 301-670-9187

Web Page Address: <http://www.gl.com/> • E-Mail Address: gl-info@gl.com

VoIP Simulation Probes – An Overview



Optionally, GL's **PacketGen™** (Intrusive VoIP Probes) or **IPNetSim™** may also be used to simulate real-time IP network conditions.

- **PacketGen™** is a PC-based real-time VoIP bulk call generator (including both SIP signaling and RTP generation) for stress testing and precise analysis of the VoIP network equipment. It is based on a distributed architecture, wherein SIP and RTP software cores can be modularly stacked in one or many PCs to create a scalable high capacity test system. Visit <http://www.gl.com/packetgen.html> for more details.
- **IPNetSim™** can simulate an entire IP network in a single box. All the conditions encountered in a real-time IP network are simulated such as network latency, network delay variation (jitter), bandwidth, congestion, packet errors, bit errors and other link impairments independently in both directions at speeds of up to 100 Mbps (or 1 Gbps per link). The IPNetSim™ 1 Gbps PRO can emulate up to 4 separate individual links simultaneously to an aggregate throughput of 4 Gbps, making it ideal for both multi-link configurations and multi-user labs. Visit <http://www.gl.com/ipnetsim.html> for more details.
- **RTPToolbox™** is an RTP simulation tool designed not only to monitor RTP and RTCP packets, but also to allow users to manually create and terminate RTP sessions, independent of call-signaling protocols such as SIP, H323, MEGACO, or MGCP. Quality Metrics with R-Factor and MOS Factors, Jitter Buffer Statistics, Degradation Factor, Burst Metrics, and Delay Metrics are graphically represented. Visit <http://www.gl.com/rtptoolbox.html> for more details.
- **VQuad™** makes assessing voice quality between and within differing networks easy, portable, and convenient. The system provides all necessary capability for automatic measurement of voice quality and QoS for any desired network including Wireless, VoIP, Analog, and TDM. Up to 8 devices (interfaces) can be controlled simultaneously on a single portable system, which makes it suitable as a low-density low-cost system. Visit <http://www.gl.com/vquad.html> for more details.