PESQ LQ/LQO (ITU P.862/ P.862.1)

PAMS LE/LQ (ITU P.800)

PSQM (ITU P.861) and PSQM plus

VQT over VOIP, PSTN, ATM, Frame Relay, Wireless Networks

Measure Effects of Noise, Delay and Echo in Networks

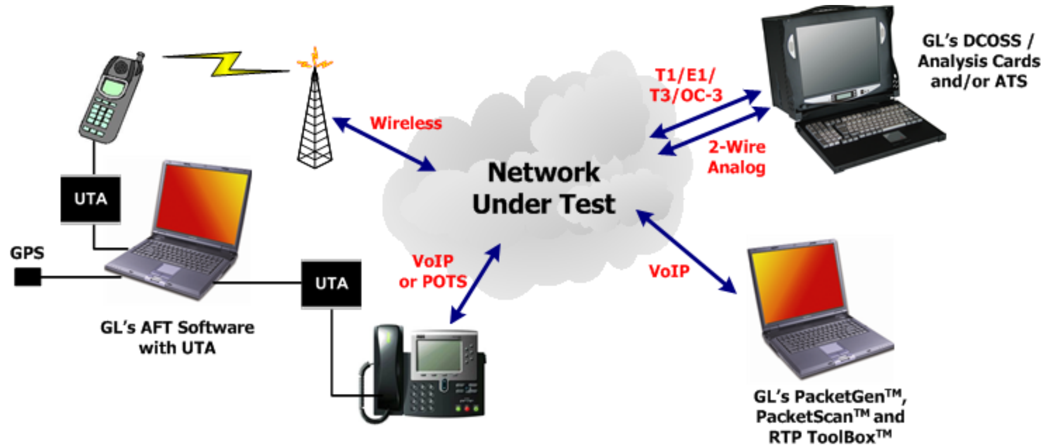
Measure Effects of Packet Jitter in VOIP Networks

Compatible with AFT, ATS, DCOSS, T1/E1 Analysis, and PacketGen/PacketScan

Manual or Automatic Operation with automatic result logging

Voice Quality Testing Solutions

(Wireless, VoIP, T1/E1, Landline)



Providing clear, uninterrupted voice is critical in Network and Echo Cancellation development. GL's Voice Quality Testing (VQT), accessed through an easy to use GUI interface, provides the voice quality measurement and analysis tools for all types of networks carrying voice traffic. Typical network applications include VoIP systems, PSTN, ATM networks, Frame Relay, and Wireless Networks.

The GL VQT utilizes three widely accepted algorithms to perform the voice comparisons, the Perceptual Evaluation of Speech Quality (PESQ LQ/LQO) per Rec. P.862/P.862.1, the Perceptual Analysis / Measurement System (PAMS) per Rec. P.800, and the Perceptual Speech Quality Measurement (PSQM) per Rec. P.861. PESQ provides an objective measurement of subjective listening tests on telephony systems. PAMS predicts overall subjective listening quality (a human's perception of quality) without requiring actual subjective testing (a very expensive and time-consuming process). PSQM predicts subjective quality of speech codecs without requiring subjective testing. The GL VQT performs PESQ LQ/LQO, PAMS, and PSQM (+) simultaneously, using two voice files (Reference File and Degraded File) and provides the algorithm results in both a graphical and tabular format.

Voice Quality Assessment Main Features

- Manual or Automatic operation using GL's AFT, ATS, PacketGen/PacketScan DCOSS, or T1/E1 Analysis Cards.
- Testing the Voice Quality of all Telecom Networks.
- Measuring the affect of Packet Jitter in VoIP Network.
- Measuring Voice Performance Over Frame Relay Networks.
- Analyze the Effects of Codec Compression in Wireless Networks.
- Provides PESQ LQ/LQO results along with Active Speech and Noise Levels, Latency, Jitter, Clipping, and power Measurements.
- Provides PAMS Listening Effort (LE) and Listening Quality (LQ) results.
- Provides PSQM (+) Mean Opinion Score (MOS) results.
- Tabular as well as Graphical Results.
- Automatic Mode allows the GL VQT to execute on a network system and point to a user-defined network drive.
- Complete automatic logging of all results with the ability to import log back into VQT.
- Fully remote controllable

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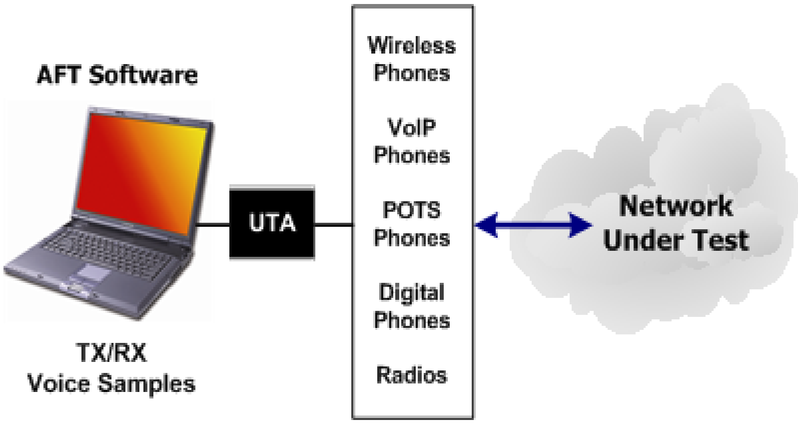
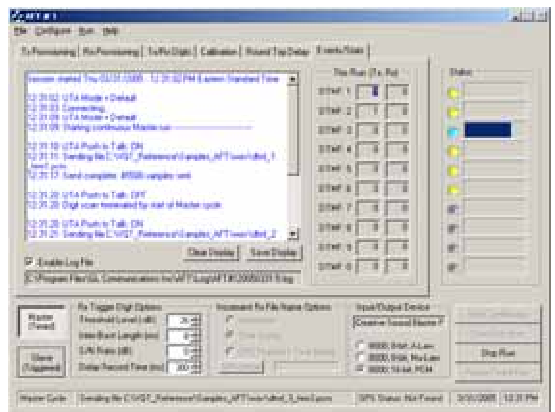
Supported Networks

Automated File Transceiver (AFT) with Universal Telephony Adapter (UTA)
Networks Supported: Wireless, VoIP (Phones/ATA's) and Landline (POTS)

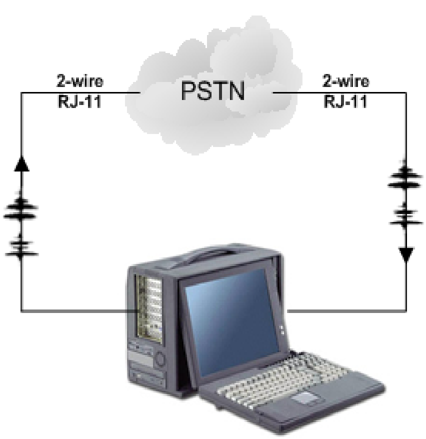


GL's **Automated File Transceiver (AFT)** with the **Universal Telephony Adapter (UTA)** solution is compact and portable; two notebook PC's utilizing AFT and VQT software packages and a couple hardware pieces. Independent locations connected to the network under test, via the endpoints (mobile, VoIP, or landline) can be easily configured for sending and recording, thus allowing end-to-end path analysis. The AFT GUI acts as the engine for synchronously transmitting and recording voice files. The VQT software provides the ITU-standard score and other detailed measurements for each recording. In addition to the ITU-standard scoring, a Round Trip Delay (RTD) measurement between any two endpoints is possible.

The AFT application is used for sending and recording the voice files ("reference" and "degraded" files) across the network path. While in operational mode, the AFT application provides a detailed log of all activity, including timestamps, power levels, sent/recorded samples, and time durations. A status bar also provides a quick look at current AFT send/record activity. File recording options include, sequential or timestamp, along with GPS position information. AFT provides a Save/Load Profile feature for quick execution from test to test. Optional call control support for various mobile phones provides a truly automatic test from call setup to teardown.

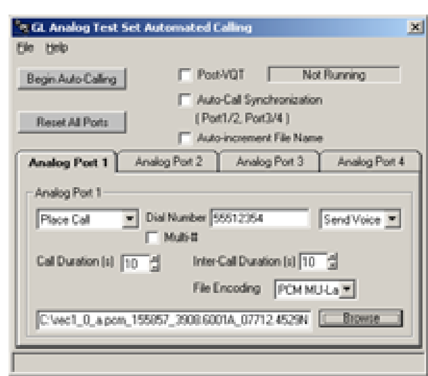


Analog Test Set (ATS)
Network Supported: Landline (POTS)



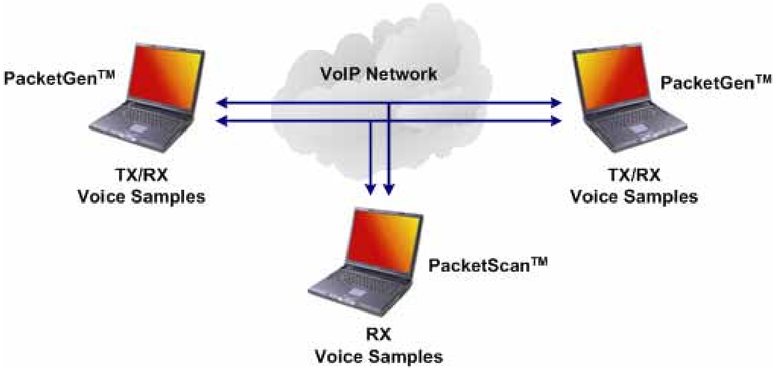
GL's **Analog Test Set (ATS)** supports four analog ports with RJ-11 interfaces and supports full call control and traffic generation. The ATS may be connected to any US/European PSTN or any VoIP ATA and, using the user-friendly GUI, the user can place calls to any desirable number as well as answer incoming calls. The Analog Test Set can be configured manually to send and record voice files. The ATS may also be automatically configured to send and record a variety of voice files to be used in the VQT algorithms.

Placing two ATS ports in synchronized mode, the automated call control and sending/recording voice file can provide a precise end-to-end test measurement. With four available analog ports, one can perform two synchronized tests simultaneously or allow each analog port to act independently, thus providing four simultaneous tests. Each port can be configured for generating or answering the call along with sending or recording the voice file. When combined with GL's Automatic File Transceiver (AFT) and wireless/VoIP adaptors, one can perform a variety of tests for a multitude of network conditions.



Supported Networks cont…

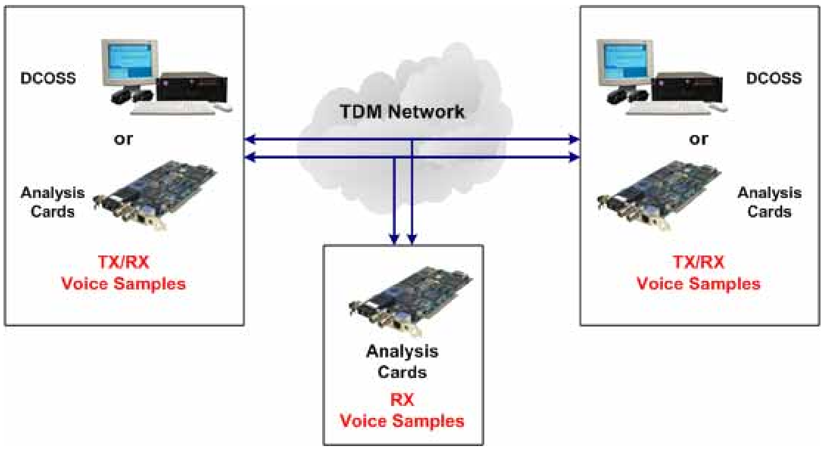
GL's Packet Series
Network Supported: VoIP



PacketGen™ is a PC-based emulator of endless User Agents (VoIP Phones). It provides real-time VoIP bulk call generator for stress testing and precise analysis of the VoIP network equipment. PacketGen™ 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. After the calls are established the PacketGen™ can automatically send and record voice files across the network, thus providing VQT measurements.

PacketScan™ is a real-time VoIP analyzer that runs on a standard PC with a NIC card. PacketScan™ is an invaluable tool for testing IP phones, Gateways, IP Routers and Switches, and Proxies. Hundreds of calls can be monitored in real-time including detailed analysis of selected voiceband streams. All calls can be recorded and used to test the voice quality at different points in the VoIP network. Detailed call statistics, call trace, RTP performance statistics, and unparalleled voiceband statistics can be viewed simultaneously. Listen in real-time to VoIP calls; perform power, frequency, spectral, tone and digit analysis with ease and precision. QOS statistics are also gathered such as packet loss, gap, jitter, and delay. Sophisticated filters permit zooming and recording of specific calls of interest.

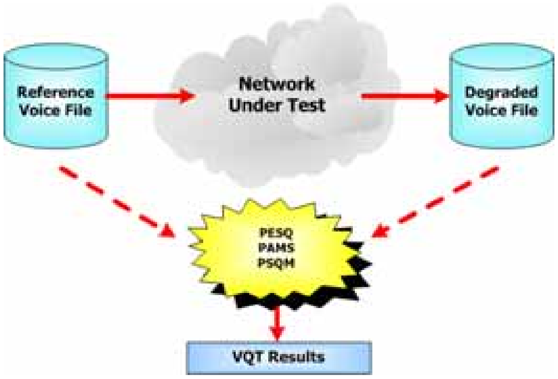
Time Division Multiplex (TDM)
Networks Supported: T1/E1/T3/OC3



The **Digital Central Office Switch Simulator (DCOSS)** converts a Pentium PC (portable, tower, rack-mount) into a digital central office switch simulator, PBX and switch, complete with T1, E1, and POTS Interfaces. A user-friendly graphical interface (GUI), through which complex switching, signaling, and digital transmission functions are easily controlled, provides the ease of operation as well as the flexibility required from telephony test equipment. DCOSS is ideal for simulating and testing advanced telecom networks and products, including switches, gateways, and transmission systems. The DCOSS can also be used for verifying T1/E1 signaling protocols of new systems. These protocols include R1, MFC-R2, PRI ISDN, SS7 and SS5. With each of these protocols, calls can be generated and received and voice samples can be sent and recorded for voice quality testing operations.

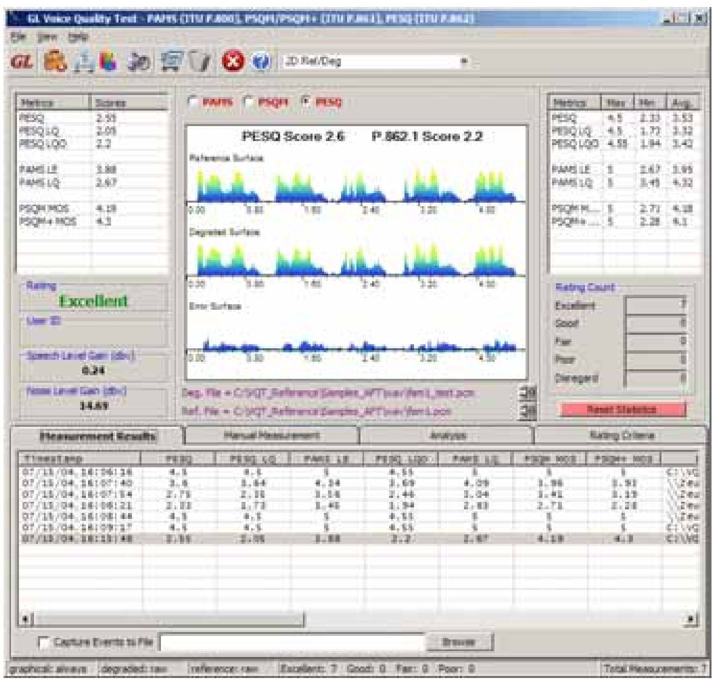
The **Ultra T1 and Ultra E1 Analysis Cards** plug into PC expansion slots, providing digital T1 and E1 input/output for analyzing, testing, simulating, and monitoring T1/E1 signals. A single (two for dual cards) analog input and output is provided to insert and receive analog signals into the digital stream. One basic operation using the Analysis cards is the ability to send and record voice samples across the network. Since this ability is all that is required to perform voice quality tests, the analysis cards work very nice along side the Voice Quality software.

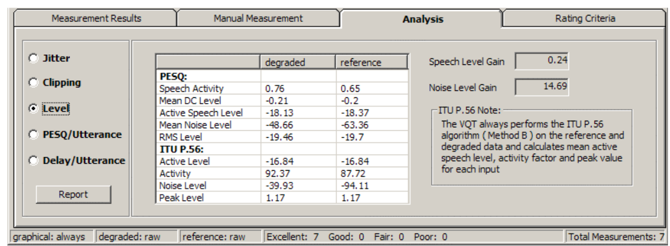
Voice Quality Testing (VQT) Software Details

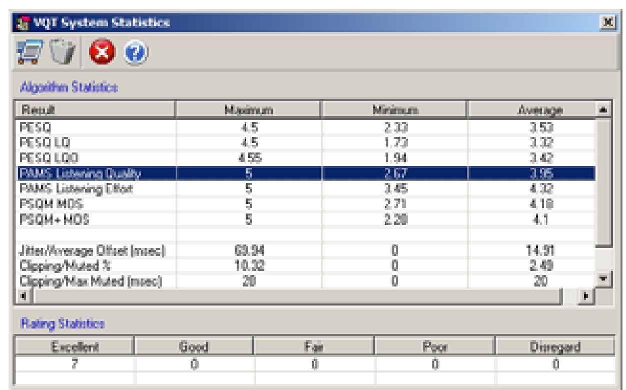


GL's **Voice Quality Testing (VQT)** software requires two files for operation. These files are typically referred to as the 'reference' and 'degraded' files. The software can act as a stand-alone product or can work along side other existing GL products. Other GL products assist with the interfacing to the network under test and the sending/recording of voice samples through the network under test. After this process has taken place the VQT software simply compares the two files ('reference' and 'degraded') and provides an ITU-standard score (PESQ, PAMS and PSQM).

The GL VQT provides a user-friendly interface, which allows the user to perform manual voice quality assessments by simply entering a Reference File and a Degraded File. The results of the VQT algorithms, PESQ LQ/LQO, PAMS, are displayed both in tabular format as well as graphically. Additional analytical results are displayed as part of the assessment such as jitter, clipping, noise level, and delay (end to end as well as per speech utterance). All results may be saved to file for post processing viewing along with sophisticated searching on the results within the VQT application.







The GL VQT may also be executed in Auto Mode. This allows the GL VQT to reside on a Network computer and point to a single or multiple user-specified network drives/directories. Voice files are recorded to this network drive/directory and GL VQT automatically performs the voice quality algorithms and displays the results. Multiple GL VQT Auto-Measurement sessions may be configured, each session with a unique set of requirements and a unique reference voice file. In addition, the user may specify voice files to be saved based on the rating criteria (i.e. if VQT is fair or poor, save the degraded voice file).