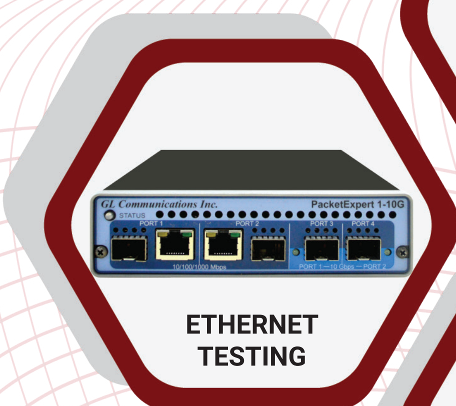
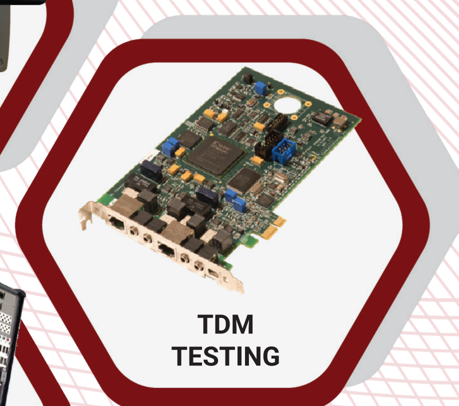
www.gl.com

GL Communications Inc.

TEST & MEASUREMENT SOLUTIONS FOR TELECOM NETWORKS





TEST
ANALYZE
MONITOR

Ethernet Test Solutions

Voice and Data Testing

TDM Test Solutions

Protocol Emulation & Analysis

Consulting Services

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Test & Measurement Solutions for Telecom Networks

Product Catalog

1st Edition

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Company profile

GL Communications Inc. has over 35 years of experience in the telecommunications industry providing test and measurement equipment, custom hardware and software development and consulting services. GL offers testing solutions to verify network performance in various telecom networks including Ethernet, wireless, fiber optic and analog networks. GL's customers include service providers, equipment manufacturers, government agencies and contractors.

GL offers support and services from its headquarters in Gaithersburg, MD USA, Bangalore India and Shanghai China.

GL's test solutions cover a wide array of networks:

- Wireless (5G, 4G LTE, 3G, 2G, Land Mobile Radio)
- Ethernet and IP
- T1 E1, T3 E3, PSTN
- OC-3/STM-1, OC-12/STM-4

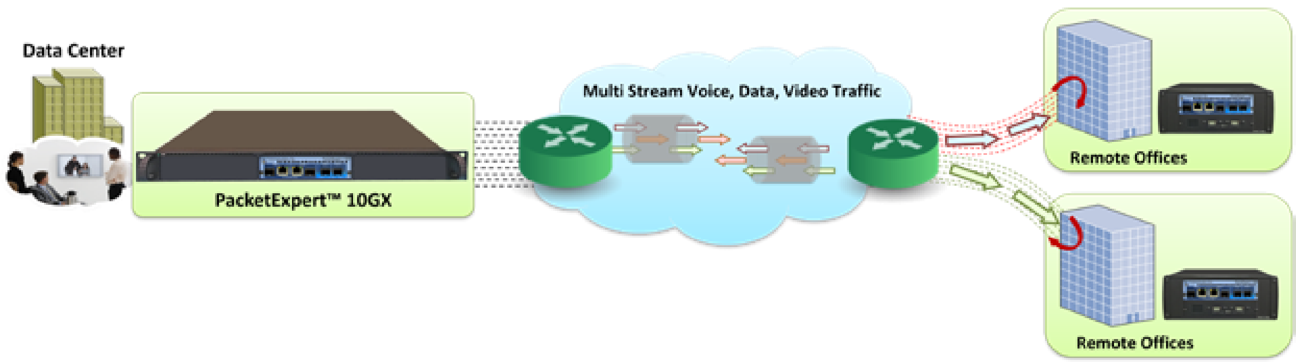
SOLUTIONS

Testing Solutions using GL Tools

GL has a comprehensive suite of telecom testing solutions to verify and ensure quality and reliability of telecom networks including Air Traffic Management (ATM), High Speed Ethernet Capture, Legacy Testing (T1 E1 and 2-Wire Analog), and Voice Quality Testing over any Network or Device.

Ethernet Testing for Wide Area Networks, Data Centers and Infrastructure

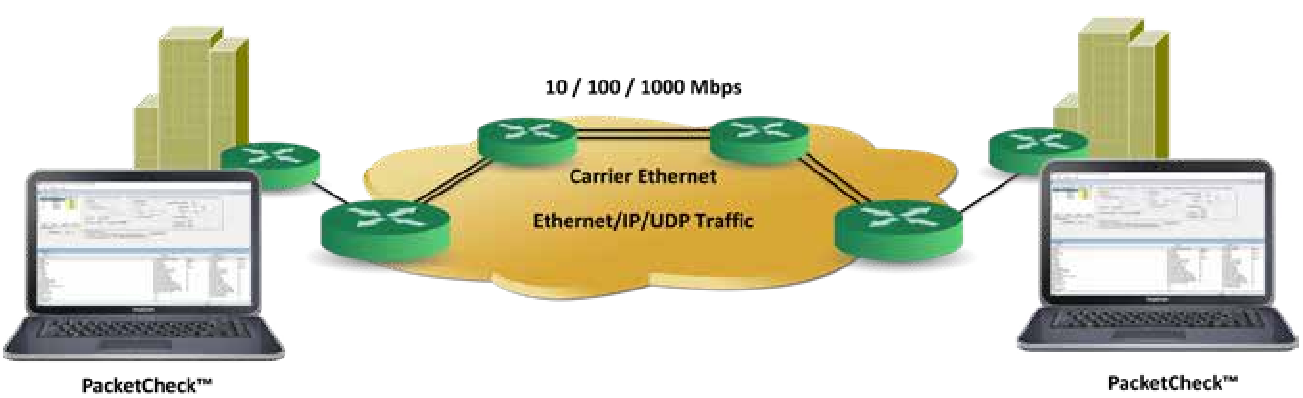
GL's PacketExpert™-10GX provides comprehensive testing of 10/2.5/1 Gbps Ethernet/IP networks. It can generate and receive traffic with customizable protocol headers. Measurements include bit error rate testing, throughput, packet loss, latency, jitter and other fundamental packet statistics. PacketExpert™ 10GX can test a wide range of networks - from testing individual links/switches, testing local Ethernet/IP networks (LAN), end to end testing of Wide Area Networks (WAN), testing Core/MPLS networks, and more. [See more on page 15]



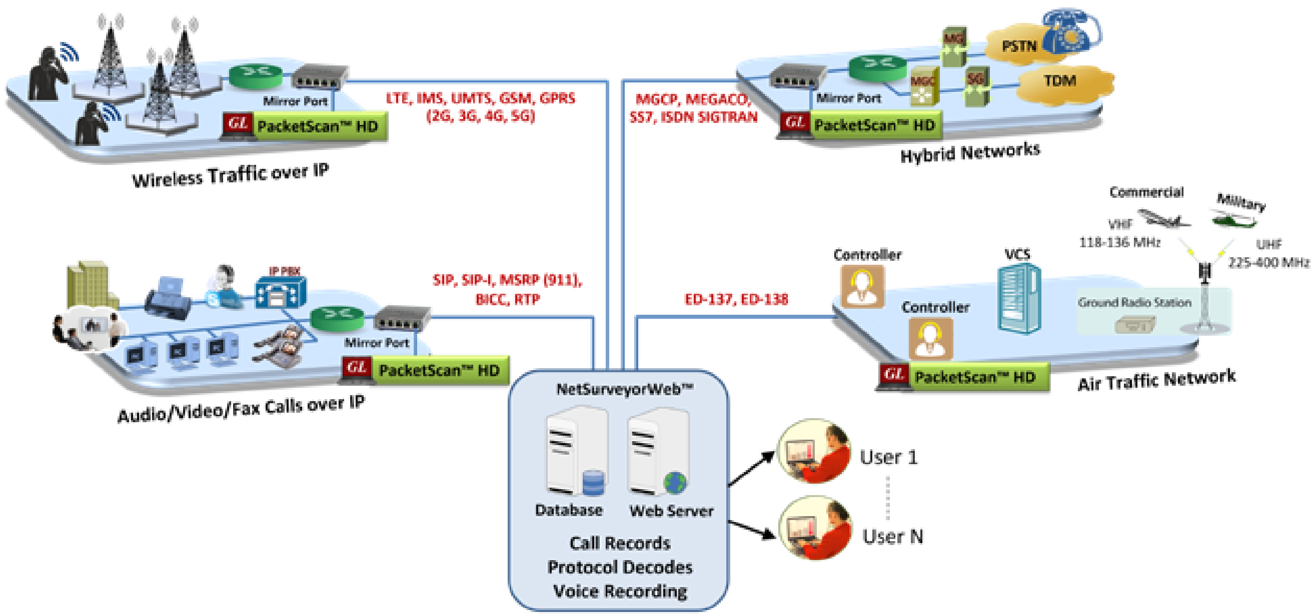
PacketCheck™ - Software Ethernet/IP Tester

GL's enhanced PacketCheck™ is a comprehensive PC based Ethernet / IP test tool with BERT, Throughput and Delay, Impairment (up to 500 Mbps) testing features. It is a general purpose network performance analysis tool for 10Mbps, 100Mbps and 1Gbps LANs and WANs. Throughput up to 500 Mbps can be easily tested.

PacketCheck™ makes use of PC's network interface card (NIC) to transmit and receive Ethernet or IP packets over the network. [See more on page 23]



PacketScan™ HD - All IP Network Monitoring



PacketScan™ HD is a high density Ethernet monitoring appliance with specialized network interface cards, large storage capacity and protocol analysis software. Customers can choose the specific Ethernet data rate for the network interface cards including 4 x 1 GigE (PKV120), 2 x 10 GigE (PKV122), 2 x 40 GigE (PKV123) and 2 x 40 / 2 x 100 GigE (PKV124) variations. Almost all VoIP and Wireless protocols over IP transport layer can be captured and decoded for troubleshooting network problems.

PacketScan™ HD includes the PacketScan™ software which can provide in-depth real-time and post-process data investigation with its Packet Data Analysis (PDA) feature. The PDA view assists in comparing multiple RTP sessions. The PDA view displays call information in graphical format and tabular format.

PDA feature is available for SIP, H323, MEGACO, MGCP, GSMA, Skinny, and IuCS based calls. PacketScan™ HD can work with NetSurveyorWeb™, a central monitoring system for a comprehensive view of network performance. It features rich graphics, ladder diagrams and CDRs (Call Detail Records). A web server is deployed in a central location along with an Oracle® database. Multiple PacketScan™ HD probes can be deployed in remote locations across the network to passively monitor VoIP traffic and send data to NetSurveyorWeb™. [See more on page 25]

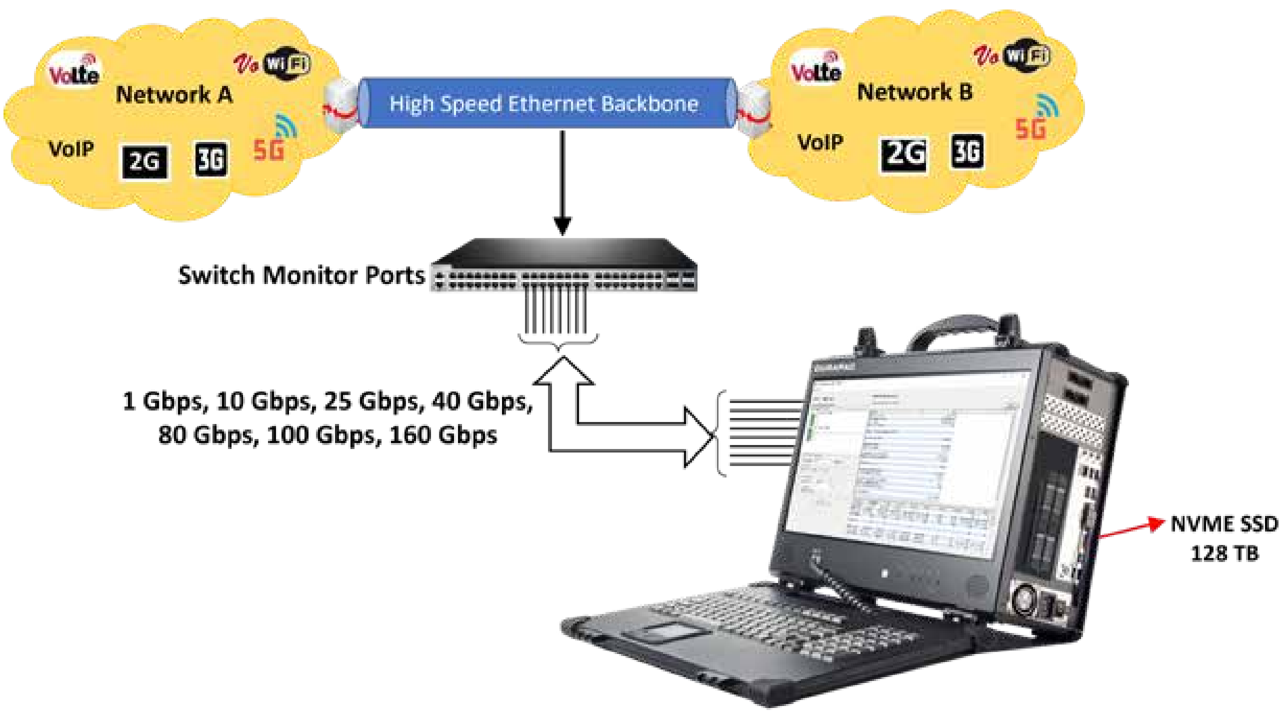
High Speed Ethernet and IP Capture

Today's networks are built with switches, routers, and gateways interconnected with 1, 10, 25, 40, and 100 Gbps full duplex fiber optic lines. These fiber connections increasingly carry packets (vs. circuits) with data, voice, and video, interleaved, aggregated, and burst out at ever faster, bigger, and longer distances.

Diagnosing issues on these networks requires error free, non-intrusive interception at full line-rate (wirespeed) and then storing the captures for post analysis with a protocol analyzer. At such high rates, capturing and storing requires a unique architecture, large RAM, and hundreds of terabytes of storage.

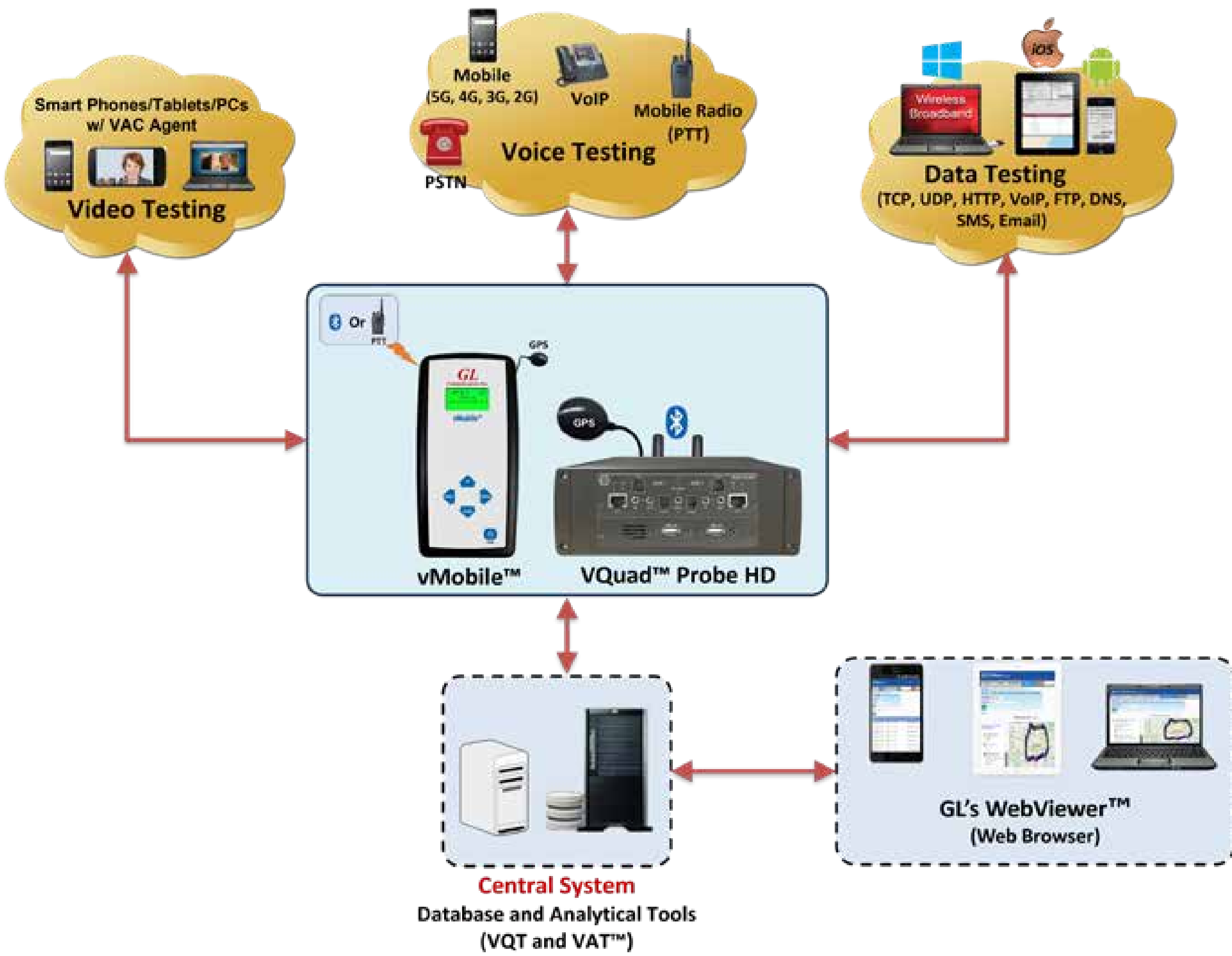
Additional features, like wirespeed filtering, packet slicing, storing and later extracting intended application streams are also vital.

GL offers the portable or rackmount versions of FastRecorder™ and PacketExtractor™, providing the ultimate packet capture and analysis solutions for managing networks of all sizes. These tools ensure lossless capture of high-speed IP traffic. The FastRecorder™ and PacketExtractor™ applications are compatible with GL's network appliance, PacketScan™ HD, and can also be used with Wireshark® protocol analyzers. [See more on page 27]



FastRecorder™ and PacketExtractor™

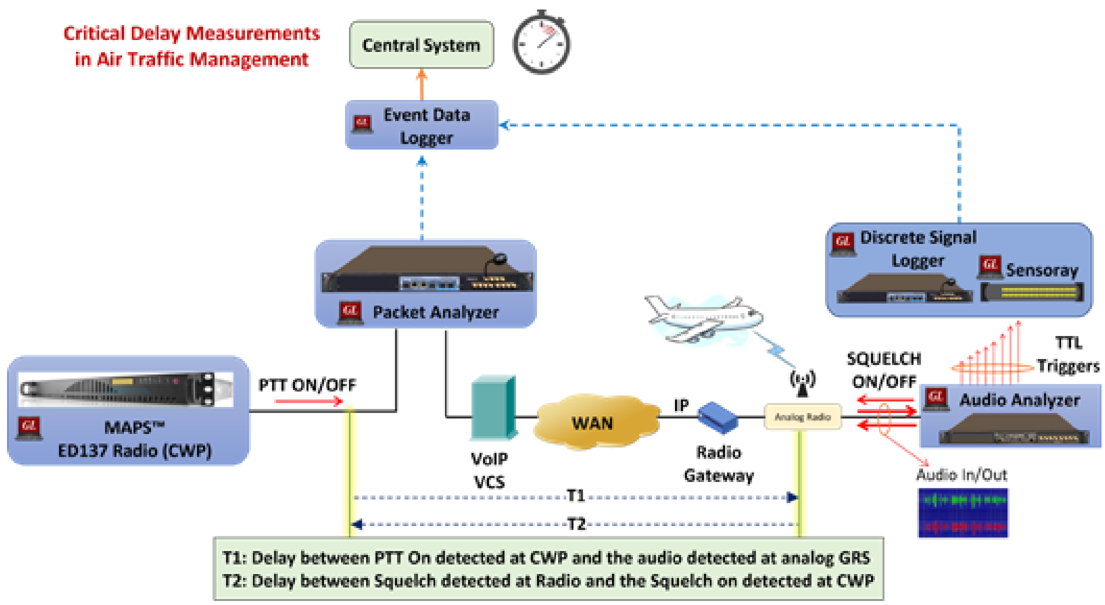
Voice Quality Testing over any Network or Device



GL's Voice Quality Testing (VQT) software supports next-generation voice quality testing standards for fixed, mobile, and IP-based networks, utilizing POLQA version 2.4 and optional version 3 (ITU-T P.863), along with PESQ ITU-T P.862, PESQ LQ/LQO (P.862.1), and PESQ WB (P.862.2). VQT analyzes degraded files, comparing them with reference files through ITU-standard algorithms PESQ LQ/LQO/WB and POLQA (Narrowband, Wideband, Super Wideband) to produce Mean Opinion Scores.

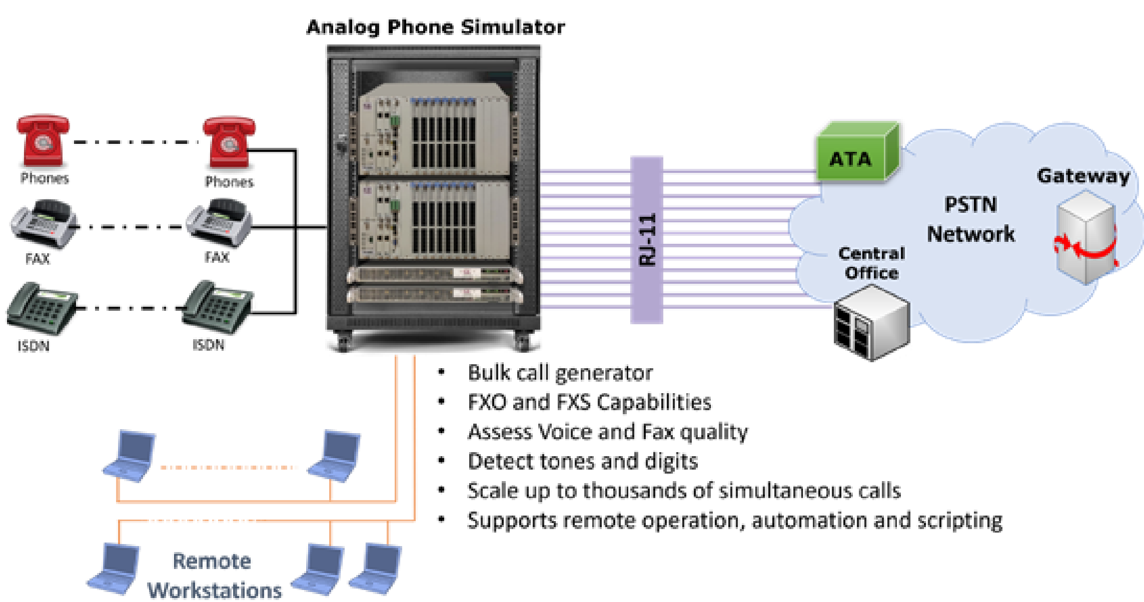
With VQT, VQuad™, Voice Analysis Tool (VAT™), and vMobile™ handheld device, this solution offers end-to-end assessment with automated call control and traffic simulation for tests on Wireless, IP, PSTN, and TDM networks. All the events/results from this solution are sent to Central Database, accessed through GL WebViewer™ (web browser). [See more on page 33]

Air Traffic Management



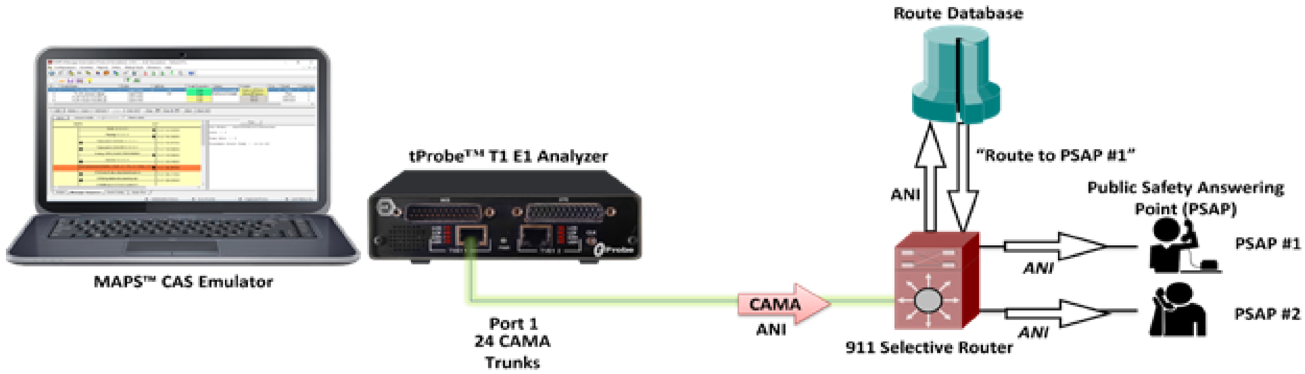
GL's Timing Measurements in Air Traffic Management (TM-ATM) test suite accurately emulates end points in ATM networks and provides critical timing measurements for various types of delay occurrences in signaling and voice transmission through the network. It includes all necessary hardware and software to identify, capture, timestamp, and correlate events at Analog, TDM and IP interfaces. End-to-end Voice Quality measurement using POLQA/ PESQ ITU-T standards. [See more on page 59]

Testing (T1 E1 and 2-Wire Analog)



GL's high-capacity Analog 2-Wire FXO/FXS or (4-Wire E&M Bulk Call Generator) used to test a Central Office (CO), PBX, ATAs, Optical Network Terminal (ONT) / Optical Line Terminal (OLT), Gateway or other telecommunications equipment. [See more on page 63]

Testing Emergency Call Services: 911, Enhanced 911 (E-911) & NG-911



Test Legacy 911 and E911 Emergency Services over PSTN (Using CAMA, Analog, CAS, and FGD-OS)

Carriers are swiftly adopting IP and Wireless infrastructure over Legacy, but the conversion of PSAPs (Public Safety Access Points) from Legacy to NG lags behind. In legacy circuits, a CAMA trunk links a carrier switch to the Selective Router (SR), which connects to multiple PSAPs. PSAPs serve as call centers for emergency response. GL's T1 E1 solutions can emulate and monitor legacy 911 calls, supporting both analog and digital CAMA simulations using MAPS™ CAS Emulator and MAPS™ FXO FXS Emulator. They also monitor analog CAMA trunks for 911 calls, generate CDRs, and analyze protocol exchanges.

- Generate and receive 911 Emergency CAMA calls over T1
- Emulate the 911 Selective Router or the PSAP or both simultaneously
- Generate 911 Emergency CAMA calls over analog FXO or FXS
- Follow the CAMA protocol precisely - MF signaling for "calling #"
- Analyze/monitor T1 CAMA trunks for 911 calls, generate CDRs, get precise protocol exchange

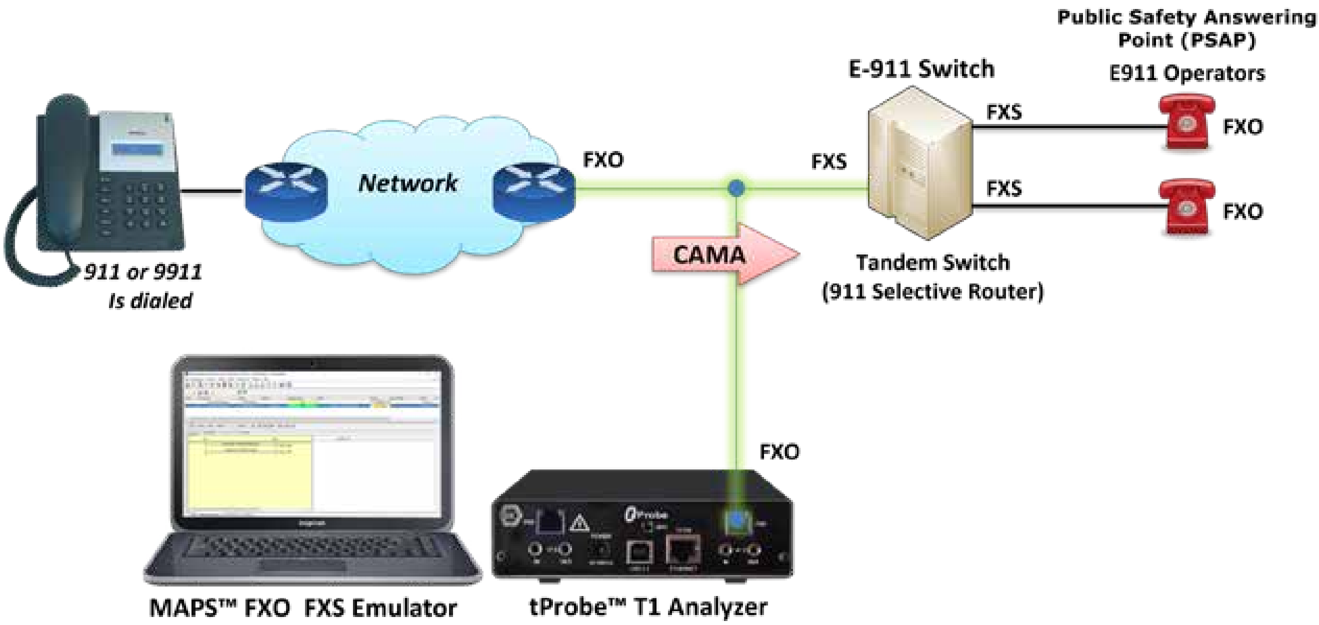
NG9-1-1 Call Simulation using SIP/RTP and Instant Messaging using SIP/MSRP

NG9-1-1 technology enables users to access emergency services not only through voice calls but also through text, images, video, and data sent to PSAPs. The MAPS™ SIP emulator for Emergency Services Internet Protocol Networks (ESInets) facilitates call delivery via Session Internet Protocol (SIP) and supports Instant Messaging (IM) delivery in accordance with RFC 4975/4976 - Message Session Relay Protocol (MSRP) protocol.

- Test NG-PSAP(s) for Voice calls, SMS and Instant Messaging
- Emulate SIP+MSRP endpoints, establish connected sessions and record transport statistics on MSRP text flows
- Test call performance for both narrowband and wideband voice codecs
- Perform advanced tests using SIP methods like SUBSCRIBE/NOTIFY, REFER and INFO for testing NG 911 interfaces
- Python and JAVA APIs for scripting and automation
- Test advanced voice features such as interactive voice response (IVR), conferencing

For more information, please visit Testing Emergency Call Services Webpage.

FXO Monitoring of CAMA Type Trunks for 911 Circuits



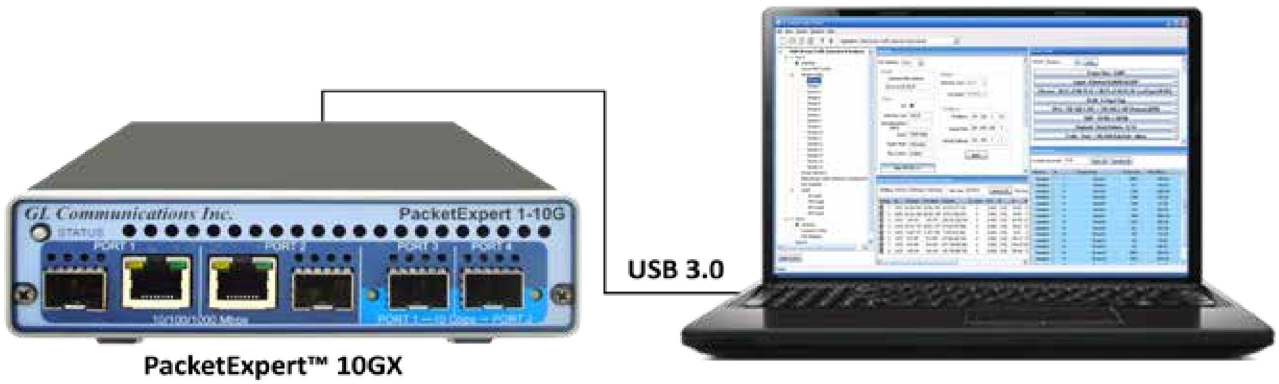
The tProbe™ FXO port can be tapped onto CAMA-type circuits for non-intrusive monitoring of 911 service. Monitoring capabilities include seizure and wink start detection, onhook and offhook detection and MF digit (calling party ANI) detection. A normal analog call is routed based on the destination (called party) phone number. However, 911 calls are routed based on the calling party number.

MAPS™ FXO FXS Emulator displays a real-time ladder diagram of the CAMA type trunk signaling sequence as captured by the FXO port. Typically, there are 5 CAMA signaling types based on the number of digits in ANI, these include 7-digit transmission (kp-0-nxx-xxxx-st), 8-digit transmission (KP-npd-nxx-xxxx-st), 10-digit transmission (kp-0-npa-nxx-xxxx-st), 20-digit transmission (kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-st), and kp–2–st (indicates a failure to receive ANI).

For more information, please visit FXO Monitoring of CAMA Type Trunks for 911 Circuits Webpage.

PRODUCTS

PacketExpert™ - 10G, 2.5G, 1G Carrier Grade Ethernet Networks



Overview

Testing the performance of high-speed ethernet links is challenging and requires specialized hardware. PacketExpert™ 10GX is a multi-functional ethernet tester that supports both Electrical and Optical interfaces. It can conduct a wide variety of testing including Bit Error Rate Testing (BERT), RFC 2544, Wide Area Network Emulation, Packet Recording and Playback, Multi-stream UDP/TCP traffic generation and ITU-T Y.1564 testing for verifying service level agreements. It supports testing up to 10 Gbps. GL can also provide supporting SFPs and cables for a ready to use test solution.

The PacketExpert™ 10GX includes two 1/2.5/10 Gbps Ethernet ports and two 10/100/1000 Mbps Ethernet ports. The ports support Copper and Optical (single-mode and multi-mode) Small Form-Factor Pluggable (SFPs).

The device supports multiple functionalities - bit error rate testing, loopback, RFC 2544 testing, Network tap capability, Record and Playback, ITU-T Y.1564 testing, Multi-Stream UDP traffic generation, and RFC 6349 for TCP throughput testing.

PacketExpert™ 10GX is controlled by a Windows® PC via a USB 3.0 cable. The software features an easy-to-use Graphical User Interface where users can configure test parameters, start and stop tests, view real-time results and graphs, and export reports.

GL can supply the controlling computer in the same chassis as the PacketExpert™ 10GX. The chassis can come in a rackmount or probe form factor. The rackmount enclosure can include up to 6 PacketExpert™ 10GX devices. With additional licensing, PacketExpert™ supports Python and C# APIs for scripting and automation.

Key Features

Ethernet / IP Testing

- Generate and receive full-duplex Ethernet traffic at wirespeed
- Generate traffic up to the UDP layer with configurable frame length, and frame size with varying traffic rates
- User selectable Electrical and/or Optical interfaces for ports allows mixed technology testing
- Wirespeed BERT, Loopback, RFC 2544, Record and Playback, ITU-T Y.1564 testing, IP Wide Area Network (WAN) Emulation capability, Network Tap Capability, Multi-stream Traffic Generation and Analysis, and RFC 6349 (TCP throughput testing)
- Layer-wise Testing - Generate traffic from Layer 1, Layer 2 (Ethernet), Layer 2.5 (Stacked MPLS), Layer 3 (IP) and Layer 4 (UDP/TCP)
- Customizable protocol headers

Automation and Remote Testing

- PacketExpert™ 10GX can be configured as server-side application based and controlled via standard C#, Python clients to automate execution of test scripts, read responses etc.
- Remotely control multiple PacketExpert™ 10GX from single client application

Wirespeed BERT

- BERT is applicable for Layers 1, Ethernet (Layer 2), up to 3 Stacked VLAN (Q-in-Q), up to 3 Stacked MPLS (Layer 2.5), IP (Layer 3) and
- UDP (Layer 4)
- Capable of handling full wirespeed BERT, in both directions Electrical/Optical ports
- Single as well as constant rate Bit Error and FCS Error Insertion
- User-defined header parameters for MAC, VLAN, MPLS, IPv4/IPv6 and UDP layers
- Multi-device support for wire-speed BERT and simultaneous BERT/Loopback applications

RFC 2544

- RFC 2544 is applicable for Layers Ethernet, MPLS, IPv4/IPv6
- Supports Throughput, Latency, Frame Loss, and Back-to-Back performance tests
- Uni-directional and bi-directional traffic can be generated and transmitted on single or dual Electrical/Optical ports
- User-defined configuration parameters such as frame size, trial duration, number of trials, etc.
- Multi-device support for single and dual ports RFC 2544 application

Loopback

- Loopback is applicable for Layers Ethernet, MPLS, IPv4/IPv6, and UDP
- Supports both smart loopback (auto layer detection) and user-defined layer-wise loopback capabilities for incoming traffic
- Multi-device support for all port loopback application

PacketExpert™ 10GX Probe

The PacketExpert™ 10GX can be placed in a Probe unit which includes a Single Board Computer. This solution retains portability and is ideal for field testing. Users do not need to carry a separate laptop. The Probe is lightweight and comes with all software and licenses pre-installed.

PacketExpert™ 10GX Rackmount

The PacketExpert™ 10GX can be placed in a 1U or 2U rackmount enclosure. Up to six devices can be housed in the 2U rackmount enclosure. This solution is ideal for long term testing from a single location such as a network room or lab environment.

PacketExpert™ 100G - Ethernet/IP Network Test Solutions

(Next-Generation 100G Carrier-Grade Ethernet Networks)

Overview

GL's PacketExpert™ 100G is a cutting-edge hardware platform designed for extensive testing of wire-speed Ethernet and IP networks, supporting speeds of up to 100 Gbps. The PacketExpert™ 100G is a high performance appliance with specialized network interface cards, GL's proprietary PacketExpert™ software, large RAM and storage, with optimized processing, and cooling capability. It is available in rackmount and portable platforms.

This versatile device comes with a web-based user interface. All functionalities can be easily accessed through any standard web browser, allowing convenient control from multiple locations and various access devices such as PCs, laptops, and tablets.

PacketExpert™ 100G can perform Bit Error Rate Testing (BERT), Loopback Testing and RFC 2544 Testing (throughput, packet loss and latency measurements) methodologies. Each 100G port provides independent Ethernet/VLAN/MPLS/IP/UDP layer-wise testing at wirespeed. BERT, RFC 2544, and Loopback applications are implemented on all transport Layers including Layer 2 (Ethernet), Layer 2.5 (VLAN / MPLS), Layer 3 (IPv4 / IPv6), and Layer 4 (UDP).

Key Features

PacketExpert™ 100G Hardware - Portable/Rackmount

- Portable PCIe based hardware supports 2 x 100G ports
- Upgradeable to 4 x 100G ports or 10 x 100G ports with Portable Lunchbox PC
- Supports QSFP28 form factor
- Supports 1G, 10G, 25G, 40G, 50G and 100G speed on the same ports

Web based User Interface

- Includes web-based interface, accessible through all standard web browsers across different operating systems
- The web interface allows multiple users to connect to a single web server and independently run tests on different hardware units
- Control multiple devices from a single GUI, multiplying the number of ports available per system

Wirespeed Ethernet/IP Testing

- Simultaneously generate and receive Ethernet traffic at 100% wire-speed (bidirectional 100 Gbps rate)
- User-configurable frame size and rate
- Wirespeed BERT, Smart Loopback and RFC 2544 applications
- Test at Ethernet (Layer 2), VLAN / Stacked MPLS (Layer 2.5), IP (Layer 3 including IPv4 and IPv6) and UDP (Layer 4)
- Multi-board support for all the applications for high density testing
- Bit Error Rate Testing (BERT) supports industry standard PRBS patterns – 2^9-1, 2^{11}-1, 2^{15}-1, 2^{20}-1, 2^{23}-1 and 2^{31}-1, as well as user defined static patterns
- Python Application Programming Interfaces to allow scripting and automation (coming soon)
- Real-time results are displayed in both tabular and graphical representations
- Test result reports available in PDF and CSV file formats
- Detailed frame statistics presented in tabular format for all the ports

Wirespeed BERT Across all layers

- BERT is applicable for Layers 1, Ethernet (Layer2), up to 3 Stacked VLAN (Q-in-Q), up to 3 Stacked MPLS (Layer 2.5), IP (Layer3) and UDP (Layer4)
- Intentionally introduce bit errors individually or at a desired rate
- User-defined header parameters for MAC, VLAN, MPLS, IPv4/IPv6 and UDP layers
- Multi-device support for wire-speed BERT and simultaneous BERT/Loopback applications to increase the number of parallel BERT tests
- Real-time graphical representation of the combined Throughput and Bit Error rate can be plotted over time for BERT testing

RFC 2544 Network Testing

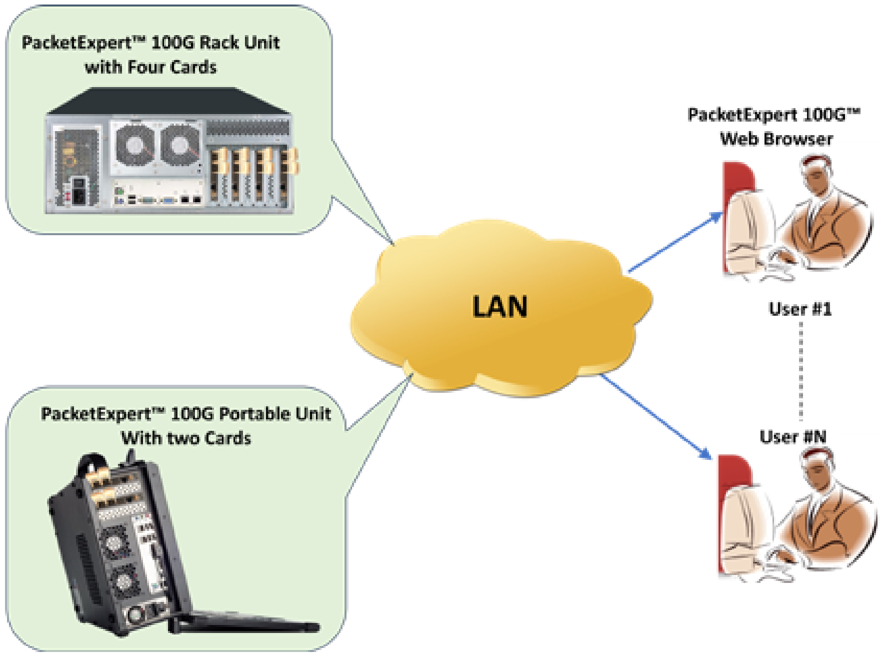
- RFC 2544 is applicable for Layers Ethernet, VLAN, MPLS, IPv4/IPv6
- Supports Throughput, Latency, Frame Loss, and Back-to-Back performance tests
- Uni-directional and bi-directional RFC 2544 testing supported
- User-defined configuration parameters such as frame size, trial duration, number of trials, etc.
- User selectable single or dual ports RFC 2544 testing
- Multi-device support for multiple parallel RFC 2544 tests
- Graphs and Statistics for all the RFC 2544 tests

Smart Loopback Testing

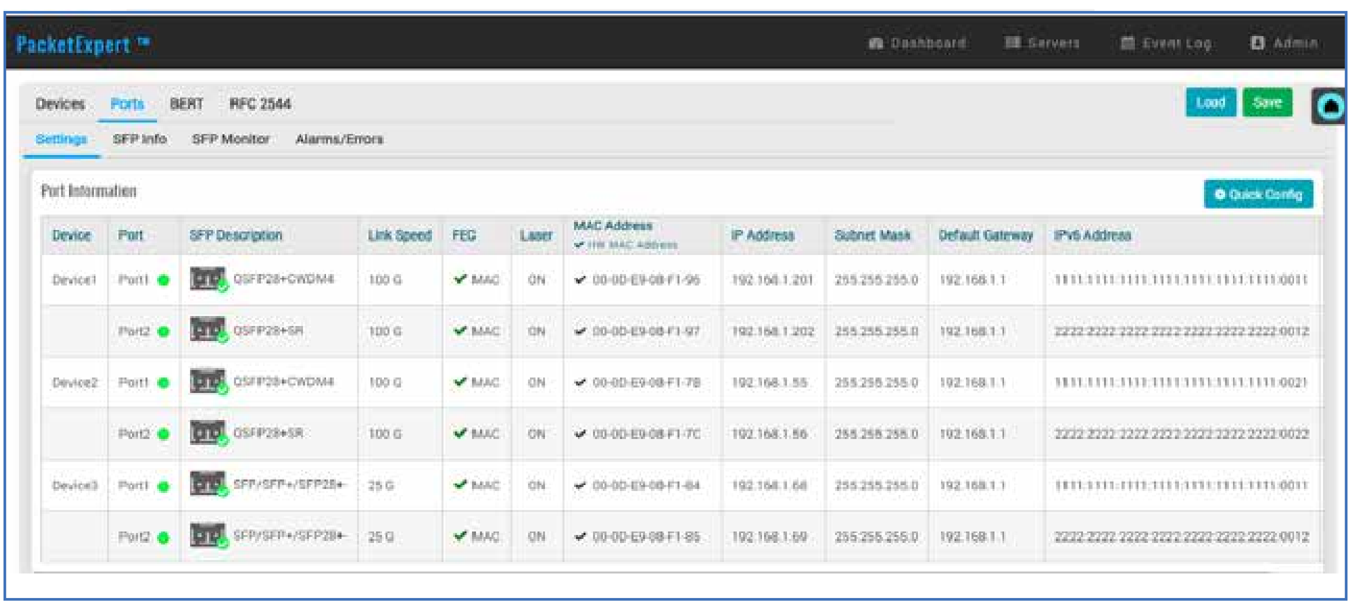
- Supports smart loopback (auto layer detection)
- Multi-device support for all port loopback application to increase the number of simultaneous Loopback ports

Multiple Users with Multiple Servers and Boards

The PacketExpert™ 100G Web interface allows users to access multiple servers located in different areas within the same Local Area Network. This allows for seamless connectivity and management of multiple PacketExpert™ 100G devices from a single server, enhancing efficiency and control.



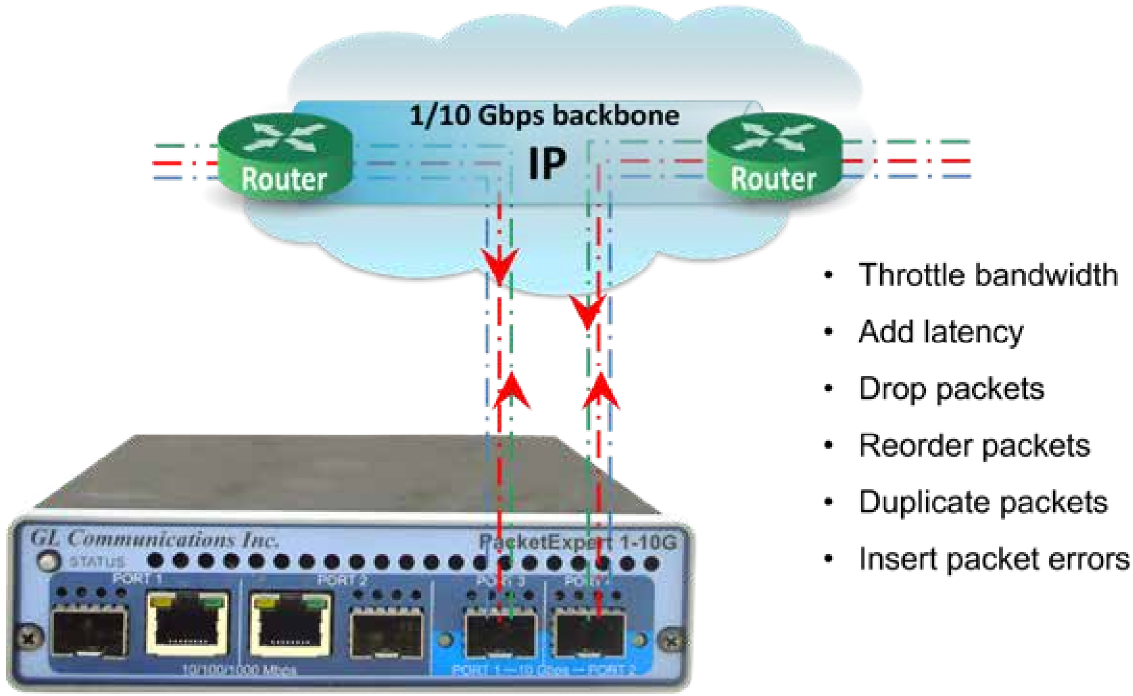
PacketExpert™ 100G - Multiple Users with Multiple Servers and Devices



Device	Port	SFP Description	Link Speed	FEC	Laser	MAC Address (HW MAC Address)	IP Address	Subnet Mask	Default Gateway	IPv6 Address
Device1	Port1	QSFP28+CWDM4	100 G	MAC	ON	00-0D-E9-08-F1-96	192.168.1.201	255.255.255.0	192.168.1.1	1111:1111:1111:1111:1111:1111:1111:0011
	Port2	QSFP28+SR	100 G	MAC	ON	00-0D-E9-08-F1-97	192.168.1.202	255.255.255.0	192.168.1.1	2222:2222:2222:2222:2222:2222:2222:0012
Device2	Port1	QSFP28+CWDM4	100 G	MAC	ON	00-0D-E9-08-F1-7B	192.168.1.55	255.255.255.0	192.168.1.1	1111:1111:1111:1111:1111:1111:1111:0021
	Port2	QSFP28+SR	100 G	MAC	ON	00-0D-E9-08-F1-7C	192.168.1.56	255.255.255.0	192.168.1.1	2222:2222:2222:2222:2222:2222:2222:0022
Device3	Port1	SFP/SFP+/SFP28+	25 G	MAC	ON	00-0D-E9-08-F1-84	192.168.1.68	255.255.255.0	192.168.1.1	1111:1111:1111:1111:1111:1111:1111:0011
	Port2	SFP/SFP+/SFP28+	25 G	MAC	ON	00-0D-E9-08-F1-85	192.168.1.69	255.255.255.0	192.168.1.1	2222:2222:2222:2222:2222:2222:2222:0012

PacketExpert™ 100G: Link status for multiple devices

Wide Area Network Emulation



Overview

GL's PacketExpert™ 10GX can function as a [Wide Area Network Emulator] by introducing impairments on existing traffic streams. Impairments include bandwidth throttling, latency, packet loss, packet reordering, packet duplication and bit error insertion. These impairments are common on backhaul networks, Satellite based networks and other Wireless networks. By emulating these conditions in a lab environment, users can test their applications in a realistic environment

The emulator is connected to the two end points of a WAN link. It can be configured to act either as a transparent bidirectional Ethernet link or a simple Ethernet bridge between two end points. The links are emulated between Port 1 (P1) and Port 2 (P2). The application emulates an IP network of 10 Gbps full duplex link or a 10/100/1000 Mbps full duplex link.

In IPNetSim™, the incoming traffic can be classified into separate streams (up to 16 streams for 1 Gbps pipe and up to 4 streams for 10 Gbps pipe). These user defined streams can be modified to simulate network impairments like bandwidth control, delay, jitter, packet loss, packet duplication, dropped packets, packet corruption, error insertion, etc.

IPLinkSim™ option supports a single stream 10 Gbps or a 10/100/1000 Mbps full duplex WAN IP Link emulation, where all the incoming traffic is streamed as a single link which can be modified to simulate network conditions.

Key Features

Network Interfaces

- Supported on 1G Electrical/Optical ports and 10G optical only ports
- Supports 2.5 Gbps ports with appropriate SFP
- IPNetSim™ and IPLinkSim™ applications are supported on both Portable and Rackmount PacketExpert™ 10GX units

WAN Emulation

- IPNetSim™ operates in multi-stream mode. Emulates unique bi-directional multi-streams (up to 16 streams on 1G ports and 4 streams on 10G ports)
- IPLinkSim™ operates in a single stream mode and emulates only 1 bidirectional WAN Link per unit
- Various WAN parameters can be configured on each bidirectional streams separately and independently
- Acts as a transparent bi-directional link or a simple Ethernet Bridge for easy integration with any test setup
- Check the stability or performance of the network with various real world impairments such as Bandwidth throttling, Latency, Packet-Loss, Error Insertion, Reordering, and Duplication
- Supports both periodic and random impairments such as packet loss, packet reordering, packet duplication, and error insertion
- Burst Loss for Packet Loss to emulate real-world impairment conditions
- Manual Packet Drop, Reorder, Duplication and Error Insertion impairments at run-time
- Bandwidth control features to mimic slower WAN links like RS232/DSL/Modem/T1 E1/T3 E3 etc.
- Introduce bi-directional delay in milliseconds increments
- Scheduler to automate the stream impairment using pre-defined csv file

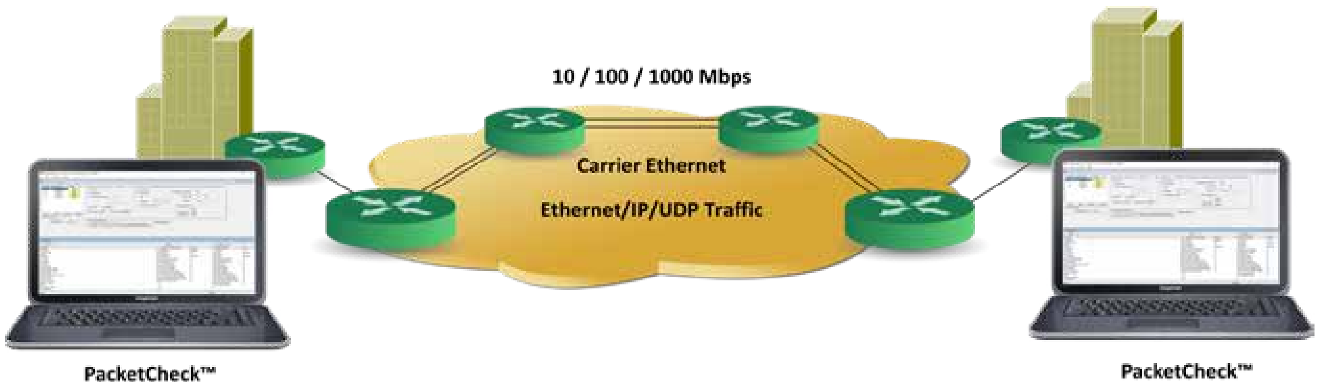
Stream Definition feature in IPNetSim™

- Traffic on each port can be classified into separate user defined streams (16 streams on 1G ports and 4 streams on 10G ports)
- Packet and Raw modes of stream configurations
- Streams can be defined based on various fields like Source/Destination MAC Address, VLAN Id, MPLS Label, Source/Destination IPv4 Address, Source/Destination UDP ports
- Stream definition feature flexibility to define mask at bit level, so that each bit can either be compared or ignored
- Up to 120 bytes wide stream definitions that covers almost entire packet header up to UDP
- User defined offset configuration to compare and identify the stream from anywhere within the frame (starting from MAC Destination Address field till the end of Payload)

Statistics

- Easily monitor the bandwidth performance using live throughput graphs for each stream
- Provides real-time statistics for unique multi-streams (16 bidirectional unique streams on 1G ports and 4 streams on 10G ports)
- Provides port level statistics like Total Frames/Bytes Received, Rx Frame Rate, Rx Data Rate, etc.

PacketCheck™ - Software Ethernet/IP Tester



Overview

GL's PacketCheck™ is a PC based Ethernet / IP test tool that provides multi stream capabilities with BERT, Throughput and Delay, and Impairment testing features with on-demand bandwidth (up to 500 Mbps). It is a general purpose network performance analysis tool for 10 Mbps, 100 Mbps and 1 Gbps LANs and WANs. Throughput up to 500 Mbps can be tested.

PacketCheck™ can generate traffic at multiple layers, from raw Ethernet frames to Stacked VLAN, Stacked MPLS, and IP packets (with UDP payloads). PacketCheck™ uses the host PC's Network Interface Card to transmit and receive Ethernet frames over the network.

The application measures end-to-end performance such as Bit Error Rate, Bit Error Count, Total Packets, Packet Loss, Out-of-Sequence Packets, Errored Packets, Round Trip Delay, and One Way Delay. Additional features include transmission of pre-recorded file traffic, recording per stream traffic to file, GTP traffic simulation, traffic generation with IFG (Inter Frame Gap) of up to 5 msec accuracy, impairment generation, and BER testing capability with provision to generate PRBS patterns or user-defined test patterns. Supports recording of the user defined stream traffic to a PCAP (PCAPNG/NTAR) or HDL (GL Proprietary) file format and playback the pre-recorded traffic from a PCAP (PCAPNG/NTAR) or HDL (GL Proprietary) file format.

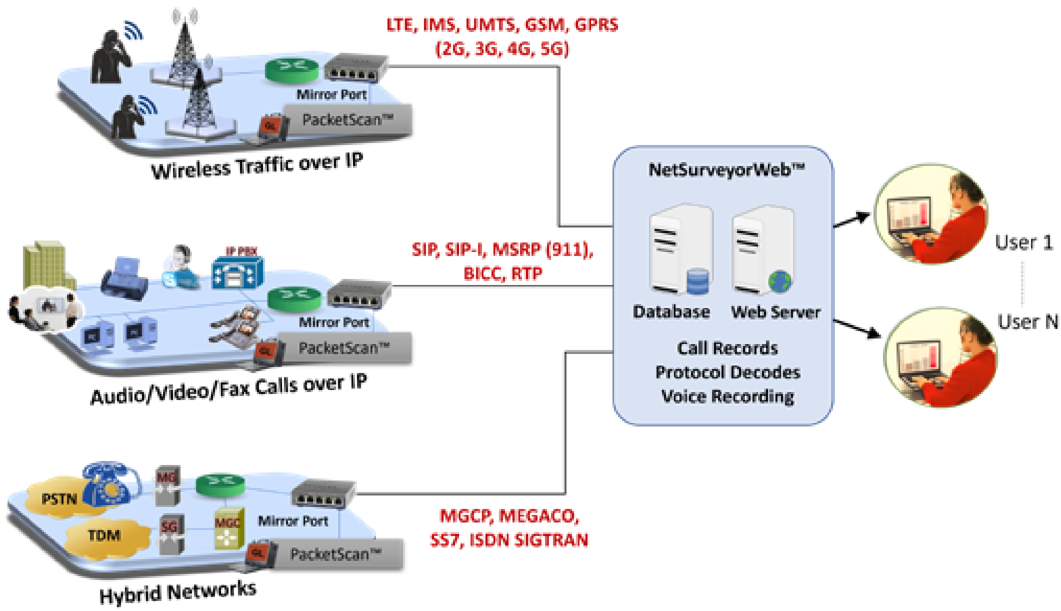
Applications

- Create multiple streams of traffic for network testing from layer 2, 3, or 4
- Bit Error Rate Testing for checking networks for dropped packets, out-of-order, non-test frames, and so on. Write packet errors to an error log
- Determine Round Trip Delay between two IP addresses or two Ethernet MAC addresses with microsecond accuracy
- Determine One Way Delay between two Network Interface Cards on the same test PC with microsecond accuracy
- Record test traffic in binary and/or PCAPNG or NTAR file format
- Playback PCAPNG files for test traffic generation. Either recorded from test BERT traffic or recorded traffic of interest
- Record non-test packets to a PCAPNG file. i.e. Non-BERT traffic related packets

Key Features

- Test Ethernet traffic of up to 500 Mbps bandwidth. Supports minimum line rate of 64 Bps
- Generate full duplex traffic at any of the four layers (Layer1, Layer2 (Ethernet) with stacked VLAN/MPLS, Layer3 (IPv4), Layer4 (UDP)) with on-demand bandwidth
- Capture stream traffic in PCAP (PCAPNG/NTAR) or HDL file format
- Playback pre-recorded traffic from PCAP (PCAPNG/NTAR) or HDL file format
- Provides options to record unidentified network traffic which does not belongs to any user defined stream into a PCAP (PCAPNG/NTAR) or HDL file format and analyze the recorded traffic in Wireshark or PacketScan™ application
- Supports stacked VLAN (up to 3 stacks) and customizable stacked MPLS (up to 3 stacks)
- Measures throughput, round trip delay, one-way delay, total packets, packet loss, out of sequence frames, error frames, correct pattern frames
- BER Testing - Bit Error Rate, Sync Loss Count, Bit Error Count, PRBS Pattern Generation/Verification of various patterns like QRSS, 2^6-1, 2^9-1, 2^{11}-1, 2^{15}-1, 2^{20}-1, and 2^{23}-1
- Run-time impairments generation of various types including Insert/Delete Bytes, and Byte Level Impairments (AND, OR, XOR)
- Jumbo frames supported
- Display statistics for each stream independently as well as aggregate statistics
- Independently define each stream to operate as Layer 2 (Ethernet) or Layer 3 (IP) or Layer 4 (UDP)
- For Layer 3 or Layer 4 streams, analyzes the received payload based on the IP or UDP length and ignore any MAC padded bytes added in transit
- Define the frame size/rate to be generated for each stream independently
- Up to 500 Mbps total combined rate (all streams combined) is possible
- The transmission rate can be configured to operate in 2 modes – Burst mode or Inter Frame Gap (IFG) mode
  - **In Burst mode**, each stream's rate can be set in Mbps, Kbps, etc.
  - Burst mode tries to generate traffic with the configured rate, but also as smoothly and evenly distributed so that the Device Under Test (DUT) node buffers do not overflow due to a temporary spike in the peak traffic
  - **In IFG mode**, the Inter Frame gap in milliseconds can be configured. The estimated rate achievable based on the IFG and the frame size is displayed for user convenience
- Can be used as a Loopback device for sending traffic back to the source
- Capability to generate/respond to ARP requests, making it easy to work with Routers
- Measure One-Way Delay or Round Trip Delay
- Generate reports in XML or PDF formats
- Support to configure IP Protocol Type from 0 to 255
- Run multiple instances on a single PC to utilize all available Network Interface Cards

PacketScan™ - IP Network Monitoring & Protocol Analysis



Overview

GL's PacketScan™ - is a Protocol Analysis software for capturing Ethernet and IP traffic. The application captures packets and intelligently groups them into Call Data Records. It can decode all VoIP and Wireless protocols and provides Quality of Service statistics on the voice calls. PacketScan™ is available both as a software solution and as a hardware appliance known as PacketScan™ HD. This network monitoring appliance captures and analyzes high-speed Ethernet traffic over a range of network speeds including 1 Gbps, 10 Gbps, 25 Gbps, 40 Gbps, and 100 Gbps. It supports the FastRecorder™ and PacketExtractor™ application for wirespeed IP traffic filtering and recording capabilities of up to 320 Gbps directly onto disk for offline filtering, extraction, and analysis. These applications are equipped to support various Ethernet interfaces such as 4 x 1 Gbps or 2 or 4 x 1/10 Gbps or 2 x 25/40/100 or 8 x 10 Gbps Ethernet interfaces.

Packet Data Analysis (PDA) is part of the PacketScan™ software program and allows users to monitor live IP networks including capture, analysis, and reporting of every call in detail. Supported protocols include SIP, MSRP, MEGACO, MGCP, H.323, SCCP, RANAP (UMTS IuCS), GSM A, CAMEL, BICC, ISUP, MAP, Gb, and GTP. It can capture IP packets over different transmission lines, including IP, T1 E1, T3 E3, and OC-3 STM-1/ OC-12 STM-4. PDA then processes the captured packets, identifies, and segregates calls based on signaling and traffic parameters.

PacketScan™ 5G protocol analyzer efficiently captures, segregates, monitors, and gathers statistics for all calls conducted across N1, N2, N4, N8, N12, and N13 interfaces within the 5G network.

TCP Analytics application analyzes TCP connections between both internal Local Area Network (LAN) and external Wide Area Network (WAN) computers including servers and clients. The application helps troubleshoot large bandwidth consumption, failed TCP sessions, packet loss, poor TCP throughput and more. TCP Analytics is an optional application with PacketScan™.

In addition, PacketScan™ can work with **NetSurveyorWeb™** a web-based dashboard for centralized network monitoring. It features rich graphics, ladder diagrams and call data records. NetSurveyorWeb™ collects data and provides comprehensive analysis of network health, detailed protocol monitoring with historical data retention.

Key Features

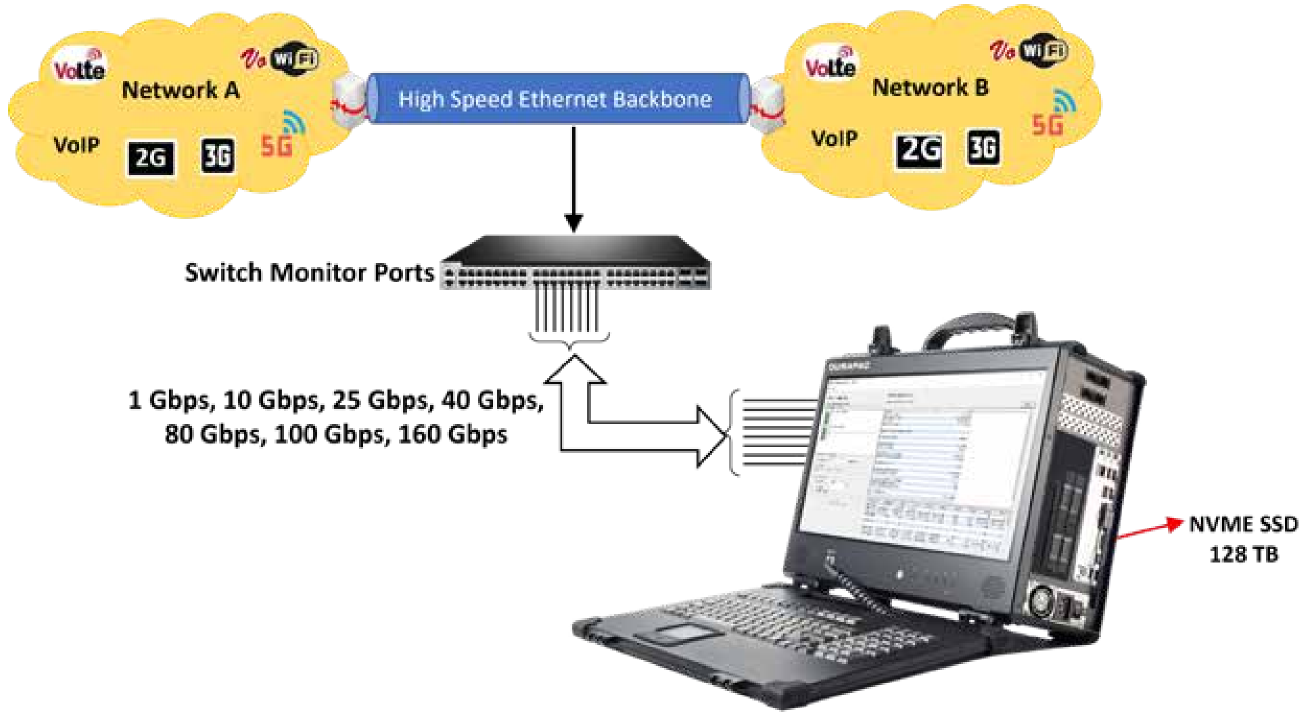
- Capture real-time calls over packet network for infinite time
- Enhanced to support Non Access Stratum (NAS), Next Generation Application Protocol (NGAP), Packet Forwarding Control Protocol (PFCP), Xn Application Protocol (XnAP) protocols
- PDA feature in Packetscan™ provide a complete call flow of a 5G session
- Analyze with rich graphics, ladder diagrams, call trace
- Flexibility to add any protocol field to the summary view, filtering, and search features
- Complex filtering and search capabilities to record all or filtered traffic into a trace file
- Option to create multiple aggregate column groups and prioritize the groups as per the requirement to display the summary results efficiently
- Allows the user to automatically create search/filter criteria from the current screen selection
- Consolidated interface allows access to all the important settings and auto-startup actions
- Permits analysis of adherence to protocol standards for the system under test or observation
- Graphical representation of statistics including ladder diagrams of VoIP calls
- Analyze recorded trace files offline
- Decrypt and analyze Voice over Long-Term Evolution (VoLTE) calls secured over Internet Protocol Security (IPsec) connection
- Decode support for multi-layer tunnelled traffic - GTP, GRE, VXLAN
- Supports BFD protocol decode
- Enhanced to support export frame summary for tunnelled traffic
- Supports decoding of eCPRI protocol
- Supports Encapsulating Security Payload (ESP) protocol to decrypt ESP packets on both IPv4 and IPv6 by providing ESP SAs value
- PacketScan™ can work with GL's Voice Band Analyzer (VBA) and Call Data Records (CDR) applications to generate CDRs as (*.CSV files) along with voice files for each direction
- The call data records are used for further analysis using built-in Excel® tools
- Supports decoding of almost all industry standard signaling protocols
- SIP ED-137 / ED-138 for Air Traffic Monitoring (Air-to-Ground and Ground-to-Ground Calls)
- Live monitoring Ipv4 and IPv6 (version 4 and version 6) networks; users can listen / record a session in real-time and extracts Fax images into TIFF format
- Monitors QOS on voice and video calls; perform power, frequency, spectral, tone and digit analysis, and video analysis with ease and precision; get an exact picture of QoS

PacketScan™ HD

PacketScan™ HD is a high density Ethernet monitoring appliance with specialized network interface cards, large storage capacity and protocol analysis software. Customers can choose the specific Ethernet data rate for the network interface cards including 4 x 1 GigE, 2 x 10 GigE, 2 x 40 GigE and 2 x 40 / 2 x 100 GigE variations. Almost all VoIP and Wireless protocols over IP transport layer can be captured and decoded for troubleshooting network problems.

- PacketScan™ HD can monitor 20000 simultaneous calls with bidirectional RTP traffic from 1 Gbps to 100 Gbps link rates. Up to 50000 calls can be achieved by scaling with higher configurations
- Users can deploy multiple PacketScan™ HD systems throughout a network and send data to a centralized database for complete network visibility
- PacketScan™ HD can work with GL's Voiceband Analyzer (VBA) and Call Data Records (CDR) applications to generate Call Detail Records as (*.CSV files) along with voice files for each direction
- PacketScan™ HD can send protocol fields, and call detail records, along with traffic summary of captured calls to a central database and NetSurveyorWeb™ displays the data from the database in a simple web-based browser, featuring rich graphics, custom search, report and filter configurations
- Supports three stages of filtering:
  - Hardware Filter - high speed, discards unwanted packets at the hardware level
  - Capture Filter - slower discards unwanted packets at the application level
  - View Filter and Search (Post Capture Filter) - performs filtering on the captured trace only for viewing purposes; filtered trace can be exported to PCAP or GL's HDL file format

High Speed Ethernet and IP Capture



FastRecorder™ and PacketExtractor™

Overview

GL offers the portable or rackmount versions of [FastRecorder™ and PacketExtractor™](), providing the ultimate packet capture and analysis solutions for managing networks of all sizes. These tools ensure lossless capture of high-speed IP traffic. The FastRecorder™ and PacketExtractor™ applications are compatible with GL's network appliance, PacketScan™ HD, and can also be used with Wireshark® packet analyzers. They support a wide range of Ethernet interface configurations, including 2 x 100 GigE, 2 x 40 GigE, 4 x 10/25 GigE, 2 x 10/25 GigE, 2 x 1/10 GigE, 8 x 10 GigE, and 4 x 1/10/25 GigE.

The application includes four modules - FastRecorder™, PacketExtractor™, PacketRecorder™, and PacketReplay™.

FastRecorder™ is a dedicated application designed for seamless interconnection with multiple interfaces, rapid configuration, and continuous, error-free capture to large NVMe SSDs for extended durations. Users have the flexibility to define filters to capture only packets of interest and set triggers to record incoming traffic based on user-defined conditions.

PacketExtractor™ allows users to extract packets of interest by defining complex filters, specifying streams, setting time periods, controlling storage size, and even selecting specific portions of packets, such as headers, among other customizable parameters for diagnosing network issues. The extracted data can be saved in PCAP, PCAPNG, or HDL (GL's proprietary) formats for in-depth analysis. Additionally, PacketExtractor™ supports monitoring and analysis of the eCPRI protocol. For more details, refer to eCPRI Protocol Analysis webpage.

FastRecorder™ and PacketExtractor™ applications are compatible with GL's PacketScan™ HD Packet Analyzers, as well as Wireshark®. PacketScan™ HD represents a comprehensive IP traffic analysis solution for its enhanced capabilities compared to Wireshark®. For instance, it offers real-time voice quality assessment, fax quality analysis, call and session separation, and powerful ladder diagrams.

The PacketRecorder™ and PacketReplay™ provide record and replay of IP traffic up to 10 Gbps.

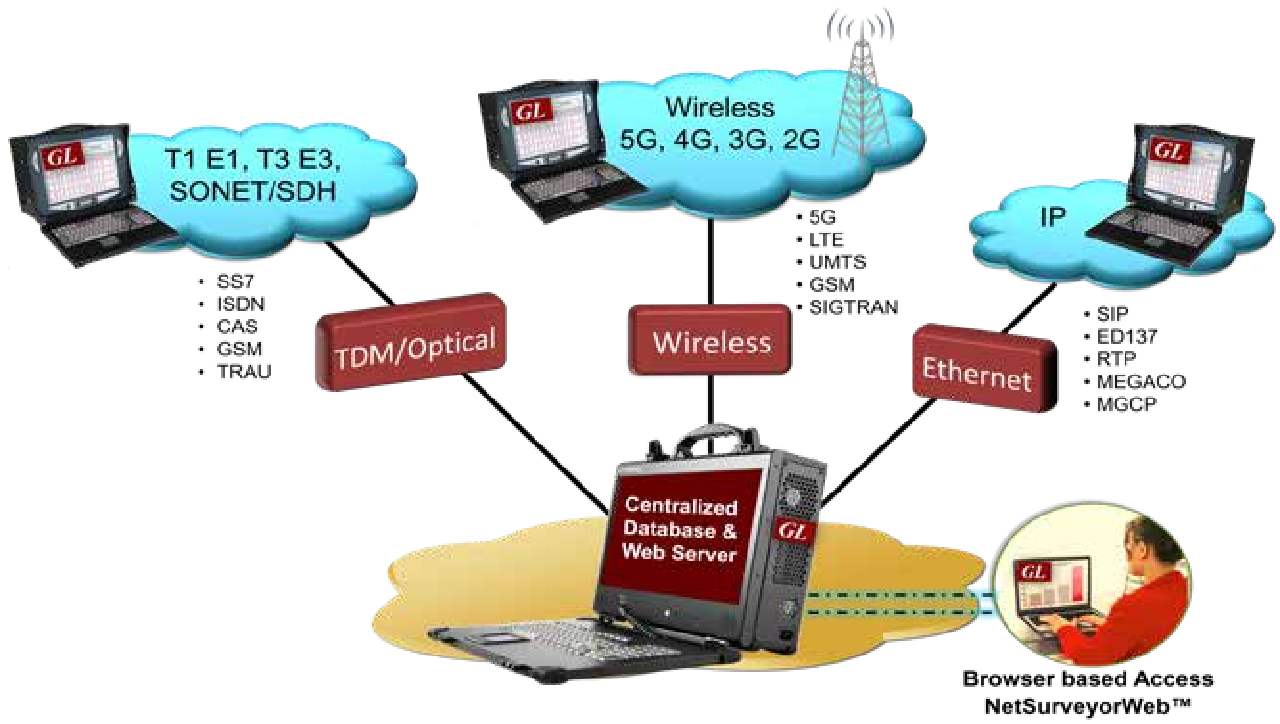
Key Features - FastRecorder™

- Lossless wirespeed capture of IP traffic across high-speed (1, 10, 25, 40, and 100 GigE) links
- Non-intrusive capture and record over Ethernet (Electrical and Optical) interfaces with nanosecond precision
- Recording on multiple ports by merging traffic with high-precision timestamps
- Up to 120 TB of total storage (NVMe SSD) in the portable platform
- Record only traffic of interest by applying efficient hardware filters based on MAC, 802.1Q (VLANs), IPv4/IPv6, Tunnel Traffic (Tunnel 1 and Tunnel 2), TCP, UDP, SCTP, SIP, and RTP parameters
- Filter on inner layers of GTP, GRE, and VXLAN tunnel traffic, such as inner IPv4/IPv6 addresses and Transport Protocol (UDP, TCP, and SCTP) port numbers
- Create custom filters using the custom filter option, providing flexibility to check fields and use logical conditions more efficiently
- Slice packets to limited lengths to store only selected packet content
- Optimized distributed disk operation to achieve wirespeed recording to disk
- Supports recording of eCPRI traffic based on eCPRI message types and UDP port numbers
- Option to record traffic continuously by retaining the latest traffic with a user-defined record size
- Statistics, such as captured, filtered/unfiltered, dropped frame percentage, and error counts per Ethernet interface or aggregated
- Create custom filters based on added fields using the custom filter option, providing flexibility in checking fields and using logical conditions efficiently
- Start recording without specifying the recording name; the current time is taken as the recording name in the format "YYYY-MM-DD_HH-Min-Sec"
- Option to view graphical representations of history, including overall rate, frames/second, per-port rate, per-port frames/second, and port link status, with Zoom In and Zoom Out options
- Configure trigger-based conditions based on capture rate, filter rate, per-port capture rate, and per-port filter rate
- Supports email alerts for specified trigger conditions
- Provides the option to schedule recording start/stop by setting triggering conditions based on datetime/time format

Key Features - PacketExtractor™

- Extract the intended traffic from previous recordings into PCAP, PCAPNG (Wireshark® format), or HDL (GL Proprietary format) output traces
- Analyze the extracted trace in PacketScan™ HD or Wireshark®
- Choose to extract the packets into single or multiple output traces
- The extraction filter provides options for IP, TCP, UDP, Inner IP, Inner UDP, and other protocols
- Extract traces with file size, time period, or packet count as the limit criteria
- Slice packets to a limited length to optimize output trace size
- Option to compress extracted trace files using 7-Zip for storage optimization
- Supports eCPRI analysis to monitor eCPRI traffic for packet impairments such as Missed Packets, Out of Order, Duplicate Packets, One-Way Delay, etc.
- Display recorded aggregated and per-port statistics, including captured, filtered/unfiltered, dropped frame percentage, and counts
- Graph option to view selected recording statistics and history of overall rate, frames/sec, per-port rate, per-port frames/sec, and port link status from the record start time to end time, along with Zoom In and Zoom Out options
- View applied hardware filters
- Supports Encapsulating Security Payload (ESP) protocol to decrypt ESP packets on both IPv4 and IPv6 by providing ESP SAs value
- Extraction can be performed from user-specified start and end times
- Supports renaming of recorded filenames
- Provides Recording Status options as Complete or Partial

Network Monitoring and Surveillance Solutions



Overview

GL's Network Surveillance System is based on a scalable and flexible architecture and is used in conjunction with GL's Protocol Analyzer probes to non-intrusively monitor from one or many testing locations.

GL's NetSurveyorWeb™ is a web-based dashboard that displays call data records and key performance indicators collected from GL's other products including PacketScan™ and tProbe™. Supported protocols include 5G, 4G (LTE, IMS, Diameter), 3G UMTS (IuCS, IuPS), 2G (GSM, TRAU, MAP, CAMEL), VoIP (SIP, ED-137, SIGTRAN, H.323, MEGACO, MGCP), TDM (SS7, ISDN, CAS, GSM, TRAU) and Analog systems.

The central system comprises of a database engine and NetSurveyorWeb™ to facilitate data storage and retrieval through web browser clients. The NetSurveyorWeb™ client application remotely or locally facilitates to view database using a simple web browser application. It includes database to store real-time and/or historic data.

This surveillance system requires TDM and IP protocol analysis probes deployed at various physical locations. These probes can capture the data at high speed, store locally decode, and segregate the traffic to calls and forwards the CDRs, signaling frame details, and other statistics to the centralized database server.

GL also offers NetSurveyorWeb™ Lite which is installed on the same PC as the PacketScan™ software. This allows the user to access the dashboard in the field directly from the Probe system. However, information from other field Probes is not displayed.

Applications

- Comprehensive analysis from overall network health to detailed protocol monitoring
- Call Data Records, fraud detection and location, remote protocol analysis and troubleshooting, real-time signaling monitor, traffic optimization engineering, and statistics
- Determine actual call signaling routes to verify network functionality under all situations including congestion and loss of SS7 nodes
- Revenue and billing verification, alarm monitoring, intrusive testing
- Quality of service measurements, call trace and recording

Key Features

Web Based UI

- Access real-time and historic data remotely via browser based clients
- Interfaces with Oracle database
- Web administration features to monitor the connected probe status, database loader status, alarms, and perform database maintenance
- Multi-user support
- Modular and distributed architecture is capable of theoretically 'infinite capacity'

Call Data Records

- Ability to customize column views with sorting capabilities for call data records
- Provides End-to-End Call Flow analysis
- Easy navigation of records to display Previous or Next Hour, Day, Month, Year through navigation tool
- Ability to export the call data records displayed based on time filter or record index as PDF and CSV formats
- Provides option to send the exported call flow or reports to the specified email address
- Ability to play voice files for the recorded calls
- Download the selected Call Trace in *.hdl and *.pcap formats
- Decode SMS in different languages for GSM CDRs
- Provides options to view CDR, Ladder Diagram, and Protocol Decodes of a selected frame in a single view

Filter and Search Calls of Interest

- Drill-down to calls of interest with filter and/or search options
- Customize Filters (Date, Time, and other call control parameters
- Apply single or multiple filters for data analysis; use logical operators between filters

Key Performance Indicators (KPI's)

- Voice Quality (MOS, R-Factor)
- Voice Analysis (VBA)
- Signal level, Nosie Level, and Echo
- Delay Measurements (RTD, OWD)
- Signaling Messages and Traffic Types
- Call Duration and Call Volume
- Call Status (Completed, Busy, Success, Failure)

Physical Layer Monitoring

- Physical Layer Alarms (Link Status, Carries Loss, Sync Loss, and so on)
- Automatically alert users when "Calls of Interest" occur
- Set alarm conditions and generate alerts of different types such as email alert, visual alert, audible alert, or even log into tables for future analysis
- Provides database query methods to gather status, statistics, events, and results

Alerts and Indicators

- Automatically alert users when "Calls of Interest" occur
- Set alarm conditions and generate alerts of different types like email alert, visual alert, audible alert, or even log into tables for future analysis
- Provides database query methods to gather status, statistics, events, and results

Call Data Records (CDR) View

The real-time data view provides visibility into each individual call. Each call can be investigated based on call control, signaling and traffic parameters. Flexible filtering can help you organize and identify “Calls of Interest”. The CDR view includes -

Frame Summary

Frame summary view provides summary of signaling data along with the decodes in the form of Hexdump.

Traffic Summary

Each call can be expanded to reveal per stream RTP statistics. The RTP/audio parameters such as payload type, total packet count, missing / duplicate / reordered / discarded packet count or %, MOS/R-Factor, cumulative packet loss, delay, and jitter values are displayed.

Graph View for each call

- This call flow graph allows easy verification of the messages exchanged and the status of the call
- Users can also select any messages and observe the corresponding decode message details in the decode view

Merge View

This feature displays Ladder diagram and Decodes of the selected message in a single view. Hide/Show any of these views to easily view the information properly.

Navigation and Search Tools

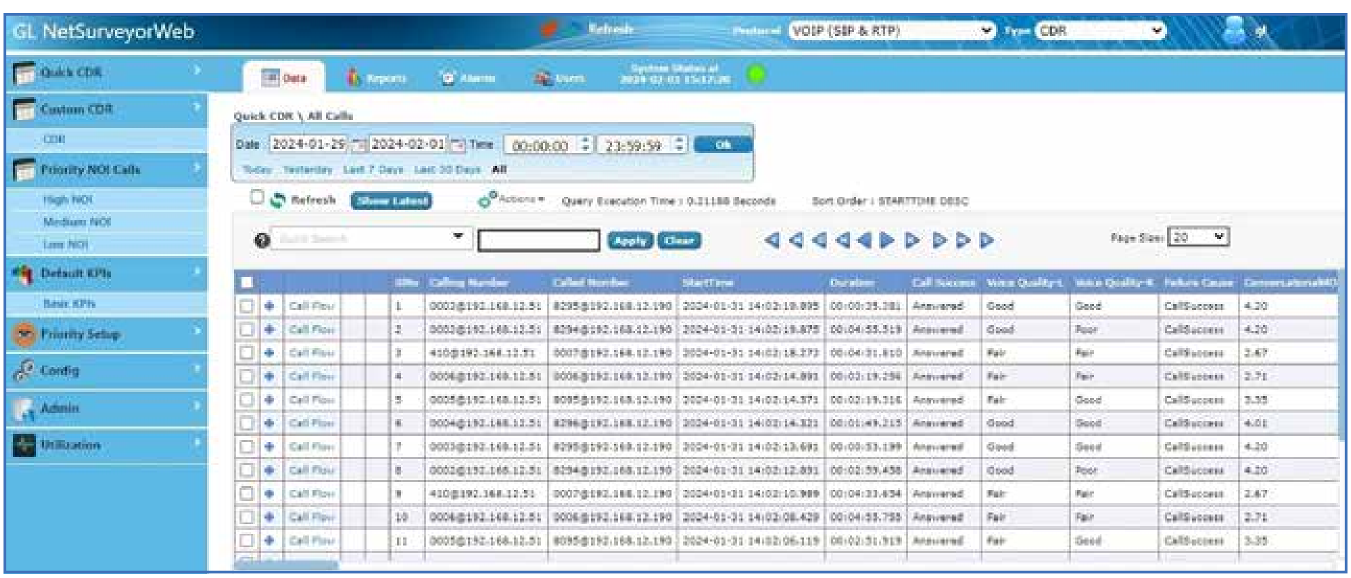
Navigate through records easily using Previous and Next Hour, Day, Month, and Year options as required. A particular call of interest can be searched using the Quick Search option.

Quick View CDR

Quick CDR View is a combination of Custom Filters and Column View, user can create their own Quick View groups and add the required columns in the created group to be displayed on the Data View. Default Quick CDR View is provided for all the protocols such as All Calls, Failed Calls, Passed Calls, VoLTE Enabled Calls, CS Fallback, Poor LMOS, Good LMOS, Longer Duration Calls, and more.

Multi-protocol call flow

This feature is useful in testing the inter-operability of different types of networks, say for example SIP-to-SS7. The Multi-protocol Call Flow provides the flow of messages exchanged between different nodes in the form of a ladder diagram along with the ability to display respective signaling decodes, thus providing visibility into complete end-to-end call flow.



Alarm Settings

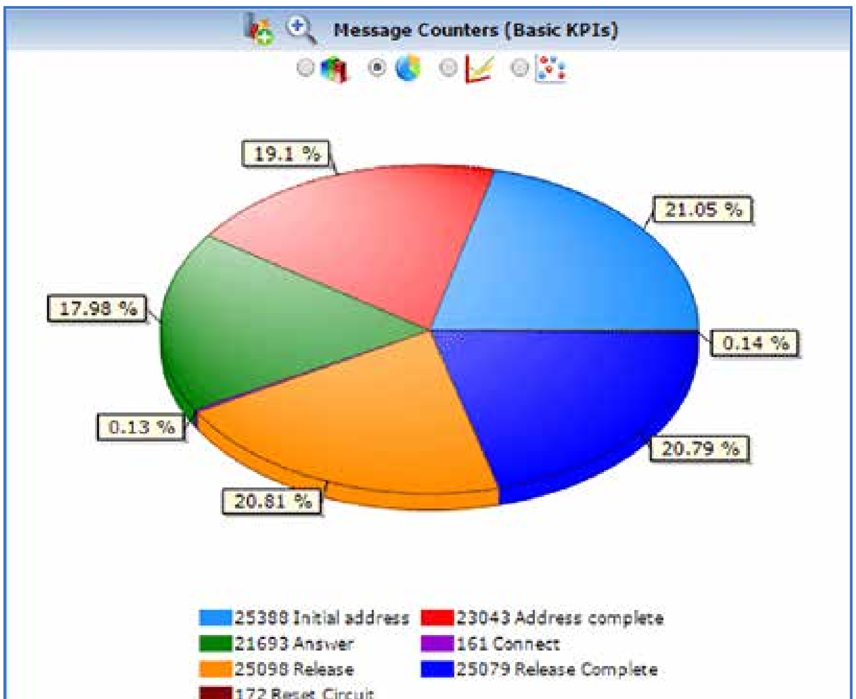
Trigger alarms and alerts whenever calls-of-interest occur, a network link failure is detected, or regularly at scheduled intervals. Directly access the pre-configured filter profiles or the KPI profiles to trigger alarms and alerts either when the custom filters conditions are passed, or send the pre-defined KPI report hourly, daily, monthly, or yearly. Alert actions can be defined based on the output of the alarm conditions such as email alert, visual alert, audible alert, SMS alerts, exporting data, setting alarm severity, or even log into tables for future analysis. Alarm Severity type can be set as Minor, Major, or Critical.

Flexible options are provided to save alarm filters as profiles, add, edit, or delete the existing alarms, selection of user KPIs, and selection of Custom filters. Schedule alarms and alerts for hourly, daily, monthly, or yearly.

Graphs and Reports

The report provides an overall summary of the captured signaling and traffic over the entire network with the help of useful graphs. Graphs are available in the form of Bar Graph, Pie Chart, Dot Graph, along with the data in tabular format for each of the plotted graphs. Reports can be generated for all calls or filtered records only.

NetSurveyorWeb™ empowers users to enhance reports by adding new Key Performance Indicators (KPIs) and customizing them through SQL Queries using the Report Configuration feature. The Add/Import KPI functionality allows users to effortlessly integrate required KPIs into existing groups, eliminating the need for redundant creation.



Time	Initial address	Address complete	Answer	Connect	Release	Release Complete
09-06	0	0	0	0	0	0
09-07	0	0	0	0	0	0
09-08	0	0	0	0	0	0
09-09	2277	2092	2051	0	2248	2248
09-10	713	667	638	0	723	723
09-11	183	151	83	2	178	182
09-12	0	0	0	0	0	0
09-13	0	0	0	0	0	0
09-14	22215	20133	18921	159	21949	21926
09-15	0	0	0	0	0	0
09-16	0	0	0	0	0	0

Voice Quality & Call Feature Testing

Overview

The VQuad™ with Dual Universal Telephony Adapter (UTA) HD, offers a comprehensive solution for sending, recording, and testing voice, video, and data across diverse network interfaces. This one-box solution supports a wide array of interfaces, including Wireless (wired, Bluetooth®, Mobile ACC, Wi-fi, Broadband - 5G, 4G, LTE, 3G), VoIP, TDM, and Analog. The flexibility of VQuad™ extends to centralized and automated testing of Voice, Video, Data, and Fax quality, allowing seamless connection to any network, service, or interface.

The Dual UTA HD supports various interfaces and telephony devices. Noteworthy features include support for FXO Wide Band, hardware loopback controlled through the VQuad™ software (including self-test mechanisms), and compatibility with the Wired headset method. The latter facilitates the connection of mobile phones to the Dual UTA HD Push-to-Talk (PTT) interface using the GL Smartphone ACC cable. Additional enhancements include loopback functionality, increased flexibility in the VQuad™ Script, and comprehensive IPv6 support.

This solution analyzes the audio content within any Narrowband (NB), Wideband (WB), or Super Wideband (SWB) PCM audio file and generates a variety of audio metrics including Frequency Bandwidth, Speech Activity, Active Speech Level, Noise Level, DC Offset, and RMS Power. When both the Reference file (pre-defined file) and Recorded files are available, the solution can generate additional metrics such as Round Trip and One Way Delay measurement, Audio Dropout analysis, Double-Talk measurements, and Voice Quality Analysis (when also coupled with the GL VQT POLQA solution). Additional metrics of the captured audio includes Speech to Text analysis (IVR Testing) with pass/fail when coupled with the GL Speech to Text Analysis solution.

Directed configurations enable the automatic sending and recording of sample voice files between telephony nodes, covering Bluetooth®, Mobile Monitor phones, wired Headset Smartphone ACC cable, PTT radios, RJ-11 POTS lines, Handset Phones (POTS, Digital, VoIP), and Balanced I/O networks. These files undergo analysis in GL's Voice Quality Testing (VQT) software, aligning with International Telecommunications Union (ITU) voice comparison algorithms.

Both VQuad™ and the stand-alone VQT software support the latest voice quality testing standard for fixed, mobile, and IP-based networks through POLQA (ITU-T P.863). POLQA analysis yields comprehensive results, including POLQA MOS, E-Model, Signal Level, Noise Level, and Jitter. Furthermore, VQT extends its support to other international voice quality test methods, such as PESQ (ITU-T P.862), PESQ LQ / LQO (P.862.1), PESQ WB (P.862.2). All the events/ results from this solution are sent to Central Database, accessed through GL WebViewer™ (web browser).

Key Features

- Smaller (Compact) Hardware Design with PTT, FXO, GPS, In/Out Interfaces
- Interfaces to Mobile Phones, Smartphones, and Bluetooth® (NB and WB)
- Loopback Functionality with Cross-Loopback Support
- Interfaces to any Telephone Subscriber Instruments
- Supports WB FXO (HD Voice)
- Supports 2-Wire and 4-Wire Direct Loopback with or without Delay
- Supports 4-Wire Outward Loopback for Codec Self-Test
- Voice, Data, Video Testing, Fax Events, Round Trip Delay and One-Way Delay Measurements
- Echo Identification and Analysis
- Automate the IVR testing process - call establishment, menu traversal, and traffic generation detection process through scripts

Dual UTA HD - Hardware Interfaces

- **Mobile Phones:**
  - **Bluetooth®** : Works with all Bluetooth® phones for both call control and send/record audio functions. Bluetooth® also performs RSSI, Battery level functions, Network verification. Supports Bluetooth NB with Frequency Range 204Hz to 3404Hz, WB with Frequency Range: 204Hz to 7200Hz
  - **Mobile audio interface for Smartphones (iPhone, Android)** : Includes Audio Headset Jack - 2.5mm (typical) for mobile phones, 3.5mm terminations for Smartphones (iPhone, Android)
  - **Wired Headset Smartphone ACC connectivity** : Connects the mobile phone to the Dual UTA HD PTT interface using the GL Smartphone Automated Call Control (ACC) cable

- **Mobile Radios with Push-to-Talk functionality:** Provides radio keying and sends/records audio
- **RJ-11 POTS lines:** Detect dial tone, go off hook, CallerID detection, send digits (two stage dialing), answer calls, detect a variety of Special Information Tones (SIT), and much more as well as send/record audio for Voice Quality measurement
- **Handset Phones (POTS, Digital, VoIP):** Replaces handset of any telephone (POTS, Digital, VoIP) that contains a coiled cord and handset
- **2-Wire Analog (WB, NB - FXO)** supporting next generation gateways
- **Dual UTA HD 4-Wire** analog interfaces supporting Tx/Rx Headset including HATS, Mobile Phone Headset, and any Handset Phone (RJ22 connection)

VQuad™ Probe

GL's VQuad™ Probe solution is an all-in-one self-contained VQuad™ with Dual UTA HD test instrument designed for conveniently testing multi-interface telephony devices for Smartphone/Handset Benchmark drive testing. A single VQuad™ Probe includes dual independent sides connecting to any type of telephony devices across various interfaces including FXO, 4-Wire Analog, PTT, Handset Phones, Mobile audio, and Bluetooth. This solution retains portability and is ideal for field testing.

Front Panel

Back Panel

VQuad™ Rackmount

GL's VQuad™ rackmount solution is an all-in-one self-contained VQuad™ with Dual UTA HD test instrument designed for conveniently testing up to 12 independent telephony devices for Smartphone/Handset Benchmark Testing. A single VQuad™ along with up-to 6 Dual UTA HD units [Two-stacked 1U rackmount], supports connection to 12 independent telephony devices. It has the ability to generate Wireless as well as 2-Wire and 4-Wire analog calls using same hardware. Users can perform simultaneous Voice, Video, Data, Fax, and Time Delay Measurements from a single VQuad™ rackmount test solution - greatly reducing the licensing costs per device. This solution is ideal for long term testing from a single location such as a network room or lab environment.

vMobile™ - Hand-Portable Voice Quality Drive & Walk Testing

Overview

The GL vMobile™ makes drive and walk testing simple and convenient. During the test you can connect to two mobile phones using the vMobile™ internal Bluetooth interfaces or connect to one mobile radio using the vMobile™ PTT analog interface. Automated testing is achieved using the vMobile™ scripting for placing and receiving calls as well as sending/recording audio during the established calls. Audio analytical metrics include Voice Quality MOS using the POLQA algorithm (ITU-P.863) or using the PESQ algorithm (ITU-P.862), with automated DAQ conversion if required. In addition to audio MOS, other metrics include one way and round-trip delay, signal and noise levels, audio dropout, frequency, and power analysis. In addition, call metrics such as failed or dropped calls are also provided. The vMobile™ can work with the VQuad™ solution where one end of the call is vMobile™ whereas other end of the call is VQuad™ with Dual UTA HD.

The vMobile™ includes embedded WiFi for control/status as well as sending results and recorded audio to a centralized system for real-time analysis. If WiFi is not available control and status of the vMobile™ can be done using a Bluetooth connection or directly from the onboard vMobile™ hardware menu. During the test, all results and events can be stamped with GPS coordinates using the onboard GPS receiver which includes external antenna to be used during drive testing while inside a vehicle. If testing inside a building or where GPS is not available, the vMobile™ Indoor Tracking System (ITS) can be used for plotting results.

vMobile™ Control including configuration, operation and status can be done using the vMobile™ Console web browser or Console app (installed on Android and IOS devices). If WiFi is unavailable all control can be done via Bluetooth connection and all captured audio files can be pulled off the vMobile™ for analysis by connecting the vMobile™ to a PC via the USB-C interface.

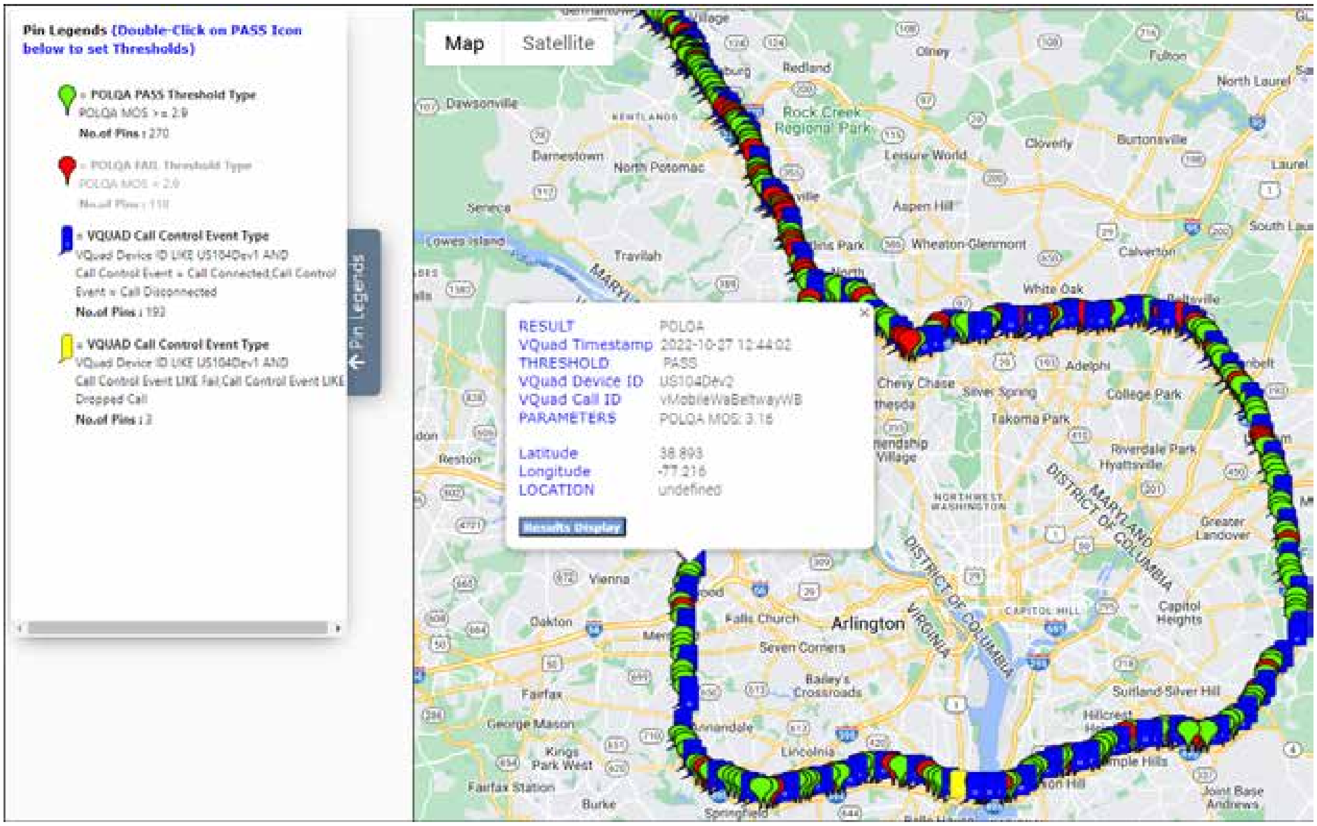
vMobile™ can also automate the GL NetTest (data testing) from any mobile device. NetTest includes an app (supports both Android and IOS devices) and can generate a variety of custom tests such as TCP (speed), UDP (capacity), HTTP, VoIP, FTP, DNS, and Video simulation. All results are sent to the same Central Database and can be plotted on Google Maps using the mobile device GPS receiver.

All results and events are sent to a Central database and accessed via the WebViewer™ (web browser). The WebViewer™ displays all measurements and call events and can generate Custom Reports which include line and bar graphs. Results can be plotted to Google Maps using custom pins depicting pass/fail and errors during the testing. From WebViewer™ users can schedule automatic reports to be emailed to any address.

Key Features

- Fully automated voice and data testing in any mobile network
- Automation includes remote operation of far-end vMobile™ or GL VQuad™ system
- Connect to any radio via wired headset
- Can operate either in Bluetooth mode or Analog mode (connect to any 4-Wire Analog device including Mobile Radio with PTT)
- Automated mobile Voice Quality Testing using embedded Wi-Fi for connecting to Central system and supporting full remote configuration and operation
- Onboard battery with availability of small portable external battery providing up to 12 hours operation
- Hand Portable including several remote options for operation and configuration
- Operation and Configuration supported via web-browser Console or Android/IOS Console app
- Drive and Walk Testing fully supported using any Mobile Phone (any carrier) or Mobile Radio
- Supports GPS along with GL's ITS for automated drive and walk testing
- Supports Voice Quality Testing using POLQA (ITU-P.863) and PESQ (ITU-P.862) algorithms
- Supports several audio metrics including Signal and Noise levels, power, frequency, and Audio Dropout analysis
- vMobile™ scripting supports all operations including conditional statements
- Bluetooth supports both NB (8000 sampling) and WB, (16000 sampling)
- Supports fully automated operation including voice and enabling PTT
- Analog PTT supports NB, WB, and SWB (48000 sampling)
- Measure One Way Delay, PTT Audio Connection Delay time, PTT Grant Tone Delay time on Radio networks
- Fully automated tests while sending events/ results to Central System for analysis and access (WebViewer™)
- Full Audio Analysis using GL VAT™ supports One Way and Round-Trip Delay measurements, Signal and Noise Levels, Speech Activity, Audio Dropout Analysis along with additional analytical functions
- Access all results via a web browser (WebViewer™) and view results on Google Maps and generates custom reports
- Network independent Drive/Walk Testing solution (supports any Network and any Carrier)
- Plot results using GPS coordinates or ITS (Indoor Tracking System) when GPS is not available
- The vMobile™ runs independent of Network connection and can be controlled directly from the onboard menu or via Bluetooth connection. All network drops (both data and voice) are recorded to the vMobile™ logs and can be retrieved through the vMobile™ Console
- Test measurements along with GPS and ITS information are sent to a central database. Results can be queried/filtered, plotted on Google Maps or ITS Viewer, and exported to a customizable report.
- Automate the IVR testing process - call establishment, menu traversal, and traffic generation detection process through scripts

Google Maps Plotting



Google Maps plotting option is provided to display the GPS coordinates of various results, vMobile™, VQuad™ Nodes, and Mobile Devices. Results include PESQ and/or POLQA® scores, Data tests, Call Connect Events, EMU, FAX, and VBA tests. These are differentiated using color codes and text call-outs. Plotting can be done in real-time while drive testing and also while testing at a location.

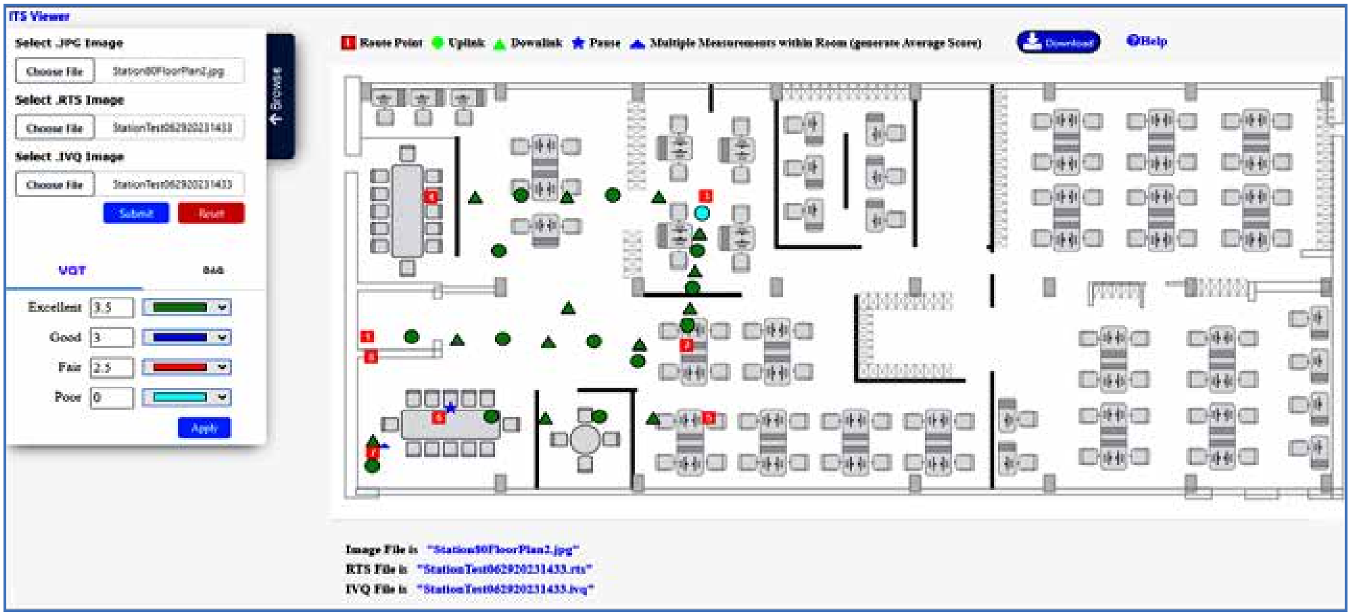
Google Maps plotting can be customized based on the specific map configuration. The customized map plotting can be further refined by choosing the test parameter such as VQuad™ location, VQuad™ Phone ID, Call Events, Threshold Settings; and other result types.

Within the map, any node result details such as Result type, Device PhoneID, Device Name, GPS Position, and Location of the test result can be obtained. Users can also get the individual device or application status or result statistics.

The figure above depicts a real drive test conducted on the I-495 Beltway around Washington D.C. Periodic tests of voice quality using the POLQA® algorithm were computed and plotted in real time.

Indoor Tracking System

Indoor Tracking System (ITS) functionality provides plotting of voice quality results to an indoor location in areas where GPS is not available. Indoor locations include underground train stations, inside buildings, tunnels, or any location where users wish to plot Voice Quality and GPS is not available. The ITS results include the voice quality measurements (based on user-defined ratings) plotted against the user-provided graphical location map. ITS is an optional application available within the VQuad™ and vMobile™ for both online and offline viewing of the results associated with the ITS.



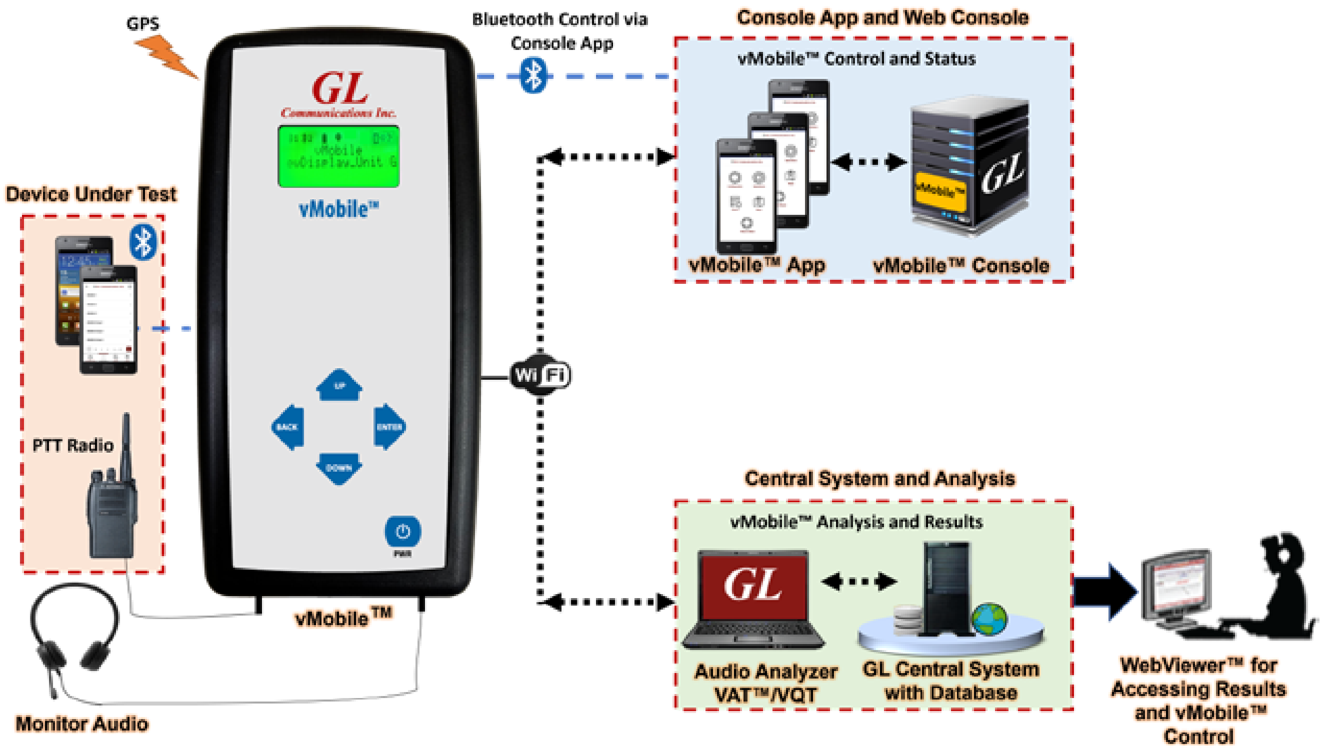
Radio and Mobile Phone Testing

The ultra-portable device brings true mobility to voice and data quality testing for wireless devices (any mobile phone or radio), changing the way automated drive and walk testing is performed. It is simple to set up and conduct simultaneous voice and data quality tests to benchmark the performance of any type of telephony device.

The vMobile™ can interface with mobile phones via wireless Bluetooth or mobile radios via wired headset for voice quality testing. When connected to a mobile radio, fully automated PTT operation is available within the vMobile™ automated scripting.

In addition, the vMobile™ 4-Wire analog interface replaces any analog headset for any device.

- Connect to any mobile radio as an analog headset or replace an analog headset to the device
- Connect to any mobile phone via wireless Bluetooth headset
- Fully automated testing including control of device and performing voice quality analysis



Voice Analysis Tool - VAT™

The fully automated VAT™ application analyzes the audio content within any NB, WB, or SWB PCM audio file and generates a variety of audio metrics including Frequency Bandwidth, Speech Activity, Active Speech Level, Noise Level, DC Offset, RMS Power, Round Trip and One Way Delay measurement, Audio Dropout analysis, Double-Talk measurements, and Voice Quality Analysis when coupled with the GL VQT POLQA solution.

Voice Quality Solution

GL's Voice Quality Testing (VQT) software utilizes various ITU standard algorithms, including Perceptual Objective Listening Quality Assessment (POLQA) version 2.4 and optional upgrade version 3 (ITU-T P.863), Perceptual Evaluation of Speech Quality (PESQ ITU-T P.862), PESQ LQ/LQO (P.862.1), and PESQ WB (P.862.2) to support nextgeneration voice quality testing standards for fixed, mobile, and IP-based networks. The software evaluates voice quality across multiple parameters by analyzing received (degraded) files and comparing them with sent (reference) files using both manual and automated methods.

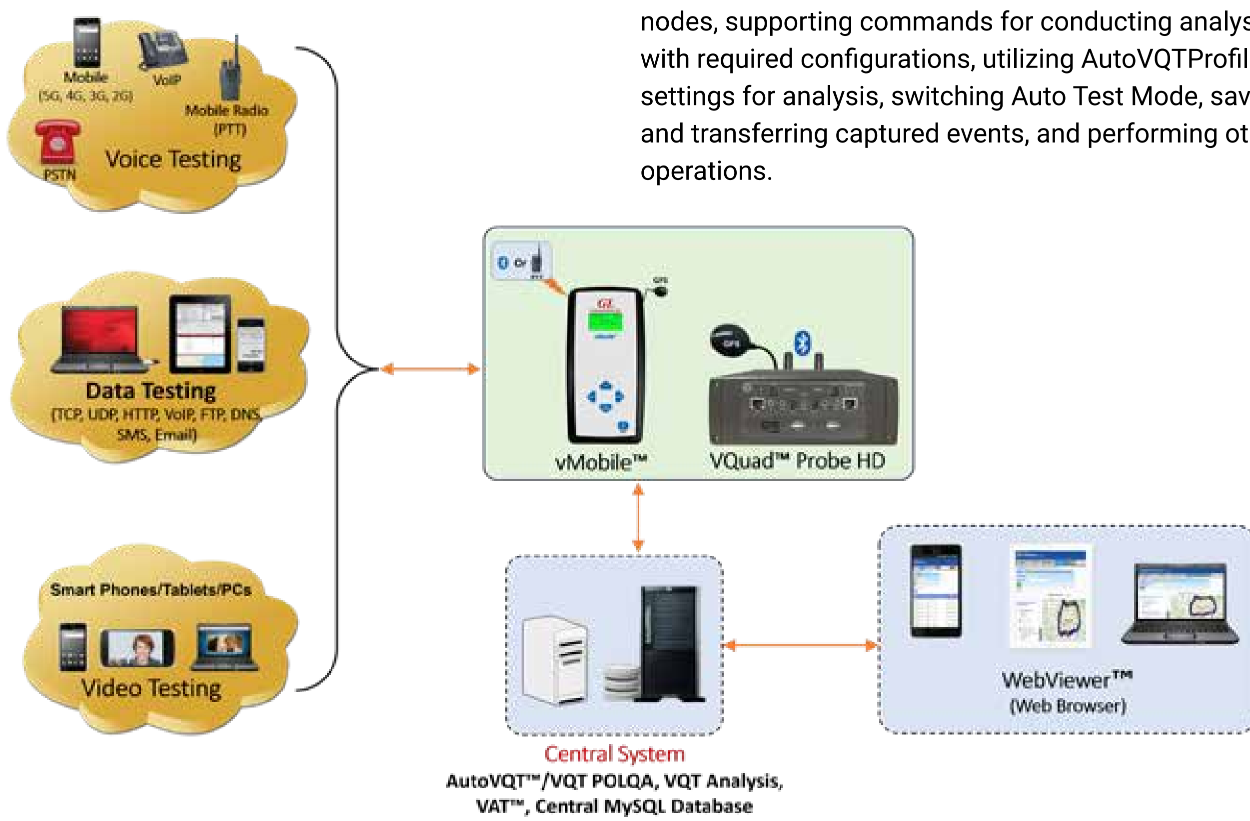
AutoVQT™ is an advanced, automated solution that analyzes thousands of voice files using POLQA algorithm in mere minutes, effectively evaluating the quality of voice communications across various networks, including VoIP, Mobile, and PSTN. This solution utilizes the Perceptual Objective Listening Quality Assessment (POLQA per ITU-T P.863 version 2.4) algorithm, which is widely acknowledged as the industry benchmark for assessing voice quality.

The AutoVQT™ application works in conjunction with GL's VQuad™, vMobile™, Voice Analysis Tool (VAT™), Message Automation and Protocol Simulation (MAPS™), or T1 E1 Analysis platforms reducing analysis time and increasing efficiency.

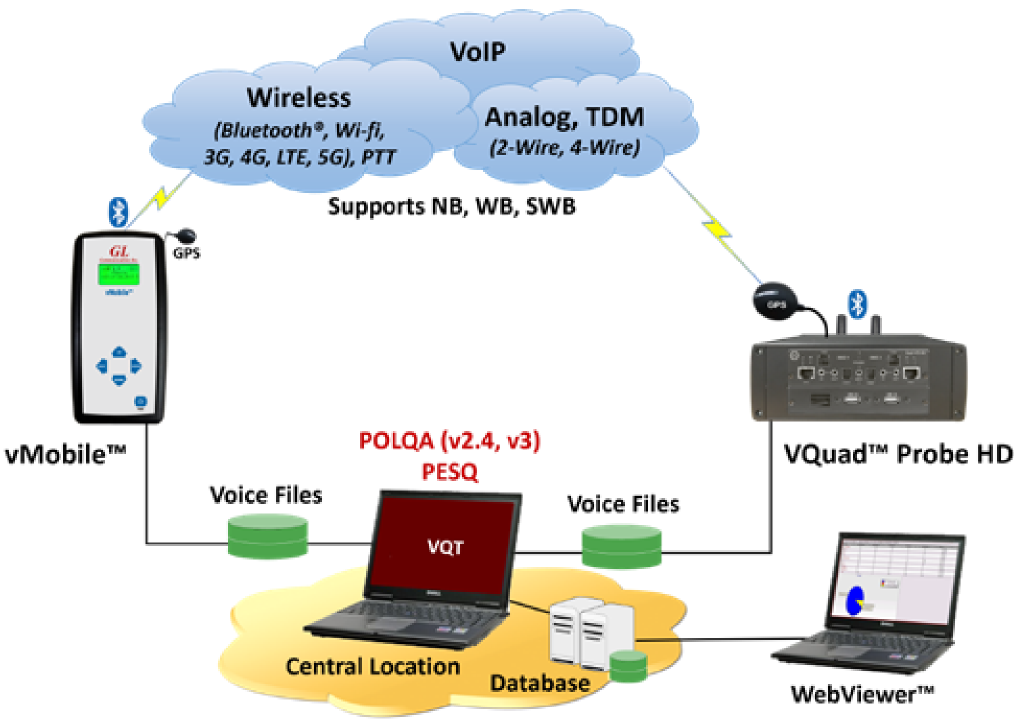
With the Voice Quality server software installed on Windows® PC and client software running on Windows®/Linux® platforms, users can analyze large quantities of PCM/WAV files obtained from any network as well as different sampling rates in minutes, significantly reducing the time required for analysis.

The AutoVQT™ application operates automatically by applying the POLQA algorithm to degraded (recorded) audio files located within a user-specified local directory. Once the application detects a PCM/WAV voice file in the configured directory, the application automatically applies the required algorithm against the reference and degraded voice files and generates the POLQA MOS. The analyzed POLQA results can be viewed through log files and on the centralized database system - Webviewer™ which can filter, query, and generate custom reports.

AutoVQT™ CLI remotely controls VQT Auto™ Server nodes, supporting commands for conducting analysis with required configurations, utilizing AutoVQTProfile.ini settings for analysis, switching Auto Test Mode, saving and transferring captured events, and performing other operations.



Voice Quality Testing (VQT)



Overview

GL's Voice Quality Testing (VQT) software supports the next-generation voice quality testing standard for fixed, mobile and IP-based networks using POLQA v2.4 and v3 (ITU-T P.863), PESQ (ITU-T P.862), PESQ LQ / LQO (P.862.1), and PESQ WB (P.862.2).

The VQT fully supports analysis using POLQA ITU version 2.4 algorithm for Narrowband (NB 8000 sampling), Wideband (WB 16000 sampling), and Super Wideband (SWB 48000 sampling) in both manual and automated testing. It also supports analysis using latest PESQ ITU release including ITU-T P.862, 862.1 and 862.2 (supports PESQ, PESQ LQ, PESQ LQO, PESQ WB).

The optional POLQA v3 (latest version of the POLQA algorithm) supports Full Band Audio analysis which provides improved scoring for mobile based VoLTE, 5G and OTT applications using EVS and OPUS codecs. This latest POLQA v3 includes analysis which is more sensitive to distortions across the entire audio spectrum. In addition, POLQA v3 supports less harsh analysis of micropauses within the speech, reacts with less sensitivity to linear frequency distortions, and includes a significantly improved and streamlined perceptual model.

The VQT software can work either independently, or with vMobile™, VQuad™ - Dual UTA HD, Voice Analysis Tool (VAT™), VQuad™ Probe HD, Message Automation and Protocol Simulation (MAPS™), and T1/E1 Analysis platforms. VQT performs PESQ LQ/LQO/WB, and POLQA (NB, WB, SWB) simultaneously, using two voice files (Reference File and Degraded File) and provides the algorithm results in both a graphical and tabular format. Additional analytical results are displayed as part of the assessment such as MOS, E-Model, Signal Level, SNR, jitter, clipping, noise level, and delay (end to end as well as per speech utterance).

All results can also be sent to a Central Database where GL's web-based dashboard, known as WebViewer™, is deployed. These results are saved to database for post-processing viewing, featuring sophisticated searching through WebViewer™ for both remote and local access.

Key Features

- Voice quality testing using POLQA version 2.4 (ITU-T P.863), with an optional upgrade to POLQA version 3 (ITU-T P.863), and PESQ (ITU-T P.862)
- Updates associated with POLQA v3 include redesign perceptual model for Full Band Audio analysis which is validated for VoLTE, 5G and OTT apps (supporting EVS and OPUS codecs)
- Provides Active Speech and Noise Levels, Latency, Jitter, Clipping, and Power measurements
- Manual or Auto modes of operations with centralized data access
- Testing the voice quality over all types of telecom networks - Wireless, VoIP, TDM, and PSTN
- Automatic mode allows the GL's VQT to execute on a network system
- VQT Command Line Interface (CLI) or API is enhanced to support both Windows® and Linux® for remote operations
- Support for Central DB Primary and Secondary IP addresses configuration for backup and redundancy
- Remote monitoring with result query and real-time statistics using web based WebViewer™
- Real-time mapping of results with GPS option used in conjunction with VQuad™
- Full support for IPv6 as well as IPv4 (includes VQT, GL Listener, and VQTCLI)
- POLQA v3 SWB supports 14kHz to full audio bandwidth up to 24kHz
- Full band analysis improves accuracy in assessment of codecs such as EVS, OPUS, AAC and LC3, as these codecs are used in many OTT applications
- With Full band support the discriminative power of POLQA at the upper high quality range of the MOS scale is increased
- Current OTT voice services using VoLTE/5G include highly dynamic delay jitter which leads to variations of the duration of very short pauses during speech. POLQA v3 handles these variations with increased precision
- POLQA v3 reacts with less sensitivity to linear frequency distortions than POLQA v2.4. This makes measurements less dependent on the frequency characteristics of headsets
- Perceptual model of POLQA v3 is significantly improved and streamlined
- Enhanced to support Python scripting for automation and remote access of voice quality testing
- Playback and display of audio from within VQT software using Goldwave software
- The WebViewer™ directly plots results or events from Drive or Walk tests on Google Maps when GPS is available
- When GPS is unavailable, the VQuad™ and vMobile™ Indoor Tracking option actively plots results on user-provided JPG floor plans or location diagrams associated with the testing environment

Modes of Operation

Manual Measurement

The GL VQT software provides a user-friendly interface to perform manual voice quality assessments using Reference File and Degraded File. The results of the VQT algorithms, POLQA, PESQ LQ/LQO/WB are displayed both in tabular format as well as graphically. All results may be saved to file for post processing viewing along with sophisticated searching on the results within the VQT application.

Auto Measurement

VQT can be executed in Auto Mode, which is used when VQT resides 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, it includes an option to analyze 12-bit degraded files in comparison with 16-bit reference files (NB, WB, SWB POLQA). Along with the standard sampling rates, POLQA also supports user-specified Sampling Rate (between 8K to 48K).

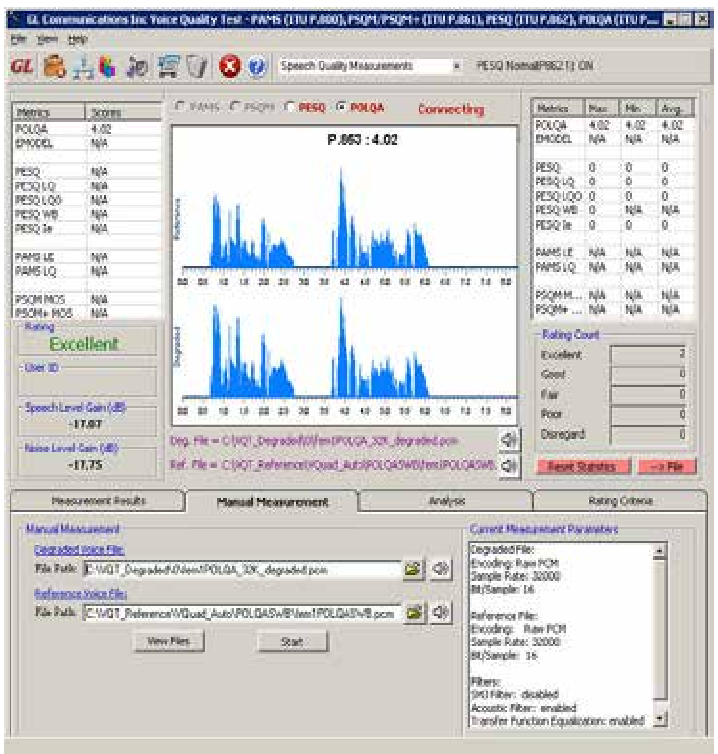
CLI/API

VQT Python libraries provide a range of Python functions which can be used to remotely or locally control the application. The VQT library can be used to run automatic and manual VQT tests with custom settings.

Voice Quality Testing with POLQA

Perceptual Objective Listening Quality Analysis (POLQA), the successor of PESQ (ITU-T P.862) analysis, is the next generation voice quality testing standard for fixed, mobile and IP-based networks. Based on ITU-T P.863 standard, POLQA supports the HD-quality speech coding and network transport technology, with higher accuracy for 3G, 4G/LTE and VoIP networks. Upgrading to 3rd edition of ITU-T P.863, POLQA extends its scope and applicability towards 5G telephony and OTT codecs. The below screenshot shows VQT POLQA Measurement Results.

- VQT POLQA supports analysis of 16-bit uncompressed PCM and WAV files, including NB (8000 sampling), WB (16000 sampling), SWB (48000 sampling)
- POLQA supports user-specified Sampling Rate (specify any rate between 8K to 48K)
- VQT POLQA supports analysis of 8-bit compressed A-Law and µ-Law files
- VQT POLQA supports 12-bit Raw PCM Degraded voice files (NB, WB, SWB)
- POLQA analysis results include POLQA MOS, E-Model R-Factor, Signal Level, Noise Level, Delay, and Jitter
- VQT optionally supports POLQA v3 for VoLTE, 5G and OTT analysis
- Playback and display of audio from within VQT software using Goldwave software

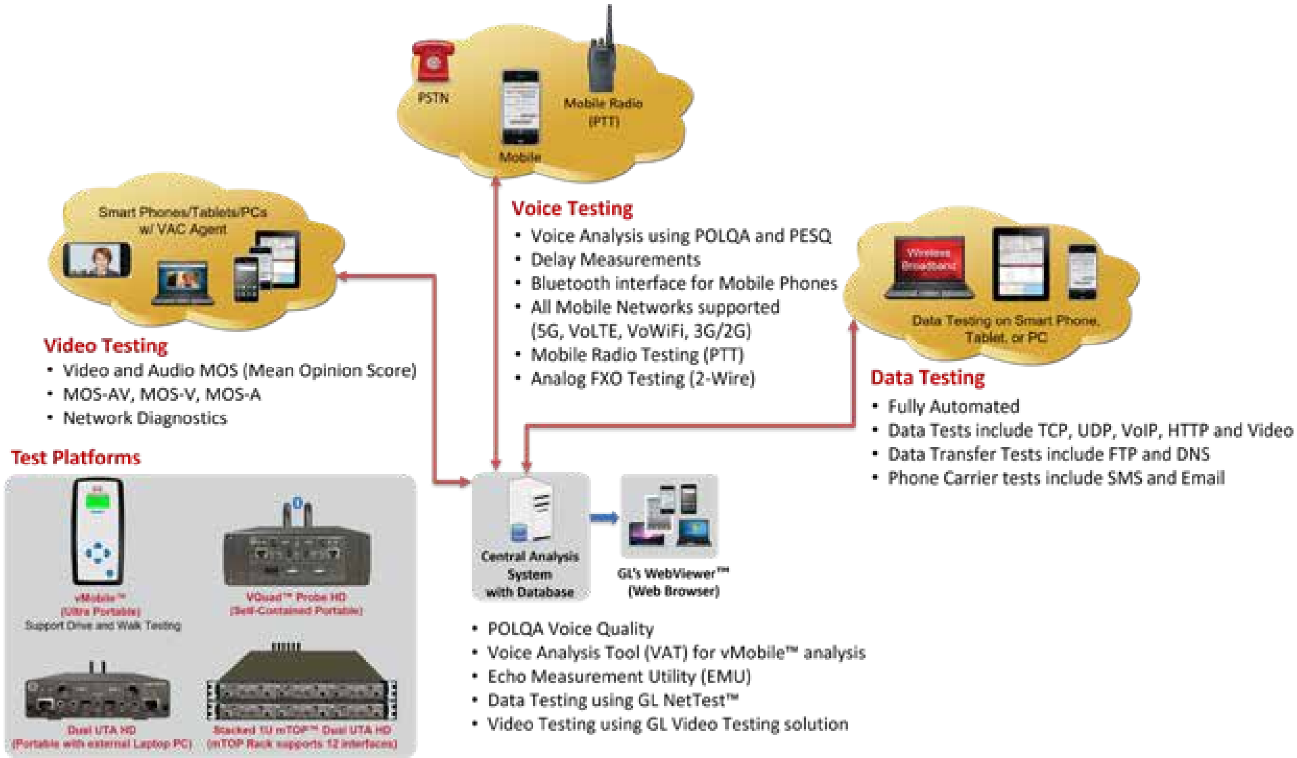


Voice Quality Testing (PESQ)

PESQ (ITU-T P.862) is an earlier algorithm which can be used for legacy testing as well as for Radio testing where DAQ results are required. Provides an objective measure that predicts the results of subjective listening tests.

- VQT PESQ supports analysis of 16-bit uncompressed PCM and WAV files, including NB (8000 sampling) and WB (16000 sampling)
- VQT PESQ supports analysis of 8-bit compressed A-Law and µ-Law files
- PESQ Results also include Signal Level, Noise Level, Delay, Delay per Utterance, and Jitter
- Playback and display of audio from within VQT software using Goldwave Software

Video & Data Testing



Overview

VQuad™ - Dual UTA HD, or the all-in-one VQuad™ Probe HD forms the central point of control for performing automated tests for voice, data, and video to and from Wireless devices (WiFi, Bluetooth®, 3G, 4G LTE, 5G), Broadband Internet (3G, 4G, 5G), VoIP devices, PSTN, and TDM circuits assessing performance, quality and reliability of practically any network. Various associated applications (Voice Quality, Video Quality, Data tests, Echo and Delay tests, Fax tests, VBA) works with VQuad™ to provide "end-to-end assessment" with additional test and measuring capabilities. All the applications work in conjunction with the VQuad™ for automatically and remotely analysing the captured data and sending the test results to the central database.

The vMobile™ handheld device supports fully automated audio testing for mobile phones (any phone, any carrier, any network) including Voice Quality using POLQA, Delay measurements, Audio Dropout analysis, and full call control with Call Fail and Call Drop metrics. The audio content within any PCM audio files can be analyzed using GL's Voice Analysis Tool (VAT™) application and generates a variety of audio metrics.

Data Testing Solutions

GL's VQuad™ solution is enhanced to support Data Testing using the NetTest application from PC and from mobile device (using MDC).

Using the VQuad™ product along with a standard PC internet connection, automated tests including TCP, UDP, HTTP, VoIP, Route, FTP, DNS, SMS, Email, PhoneInfo, SimInfo, and UEInfo are supported. The PC internet connection can be wired Ethernet, WiFi, or even Broadband card (both LTE and 3G fully supported).

Also available is the GL's NetTest package for Data Testing on any supported Mobile Device. This feature uses downloadable apps for both Apple and Android devices and is remotely controlled via the GL Mobile Device Controller (MDC) system. Using VQuad™ one can remotely and automatically perform these same Data Tests on Apple and Android devices when connected to 3G, 4G and 5G networks or over a WiFi network.

- Automated Data Testing (NetTest) for Apple/Android Mobile Devices and for PC Internet connection (Wired, Broadband 5G, 4G, 3G, WiFi) including TCP, UDP capacity, VoIP, Route, HTTP, FTP, DNS, SMS, Email, PhoneInfo, SimInfo, and UEInfo
- Data Testing supports remotely deployed GL Data Server target points for end-to-end testing
- Statistics and complete results are available through the WebViewer™ for both Mobile Device NetTests and PC based NetTests

Video Test Solutions

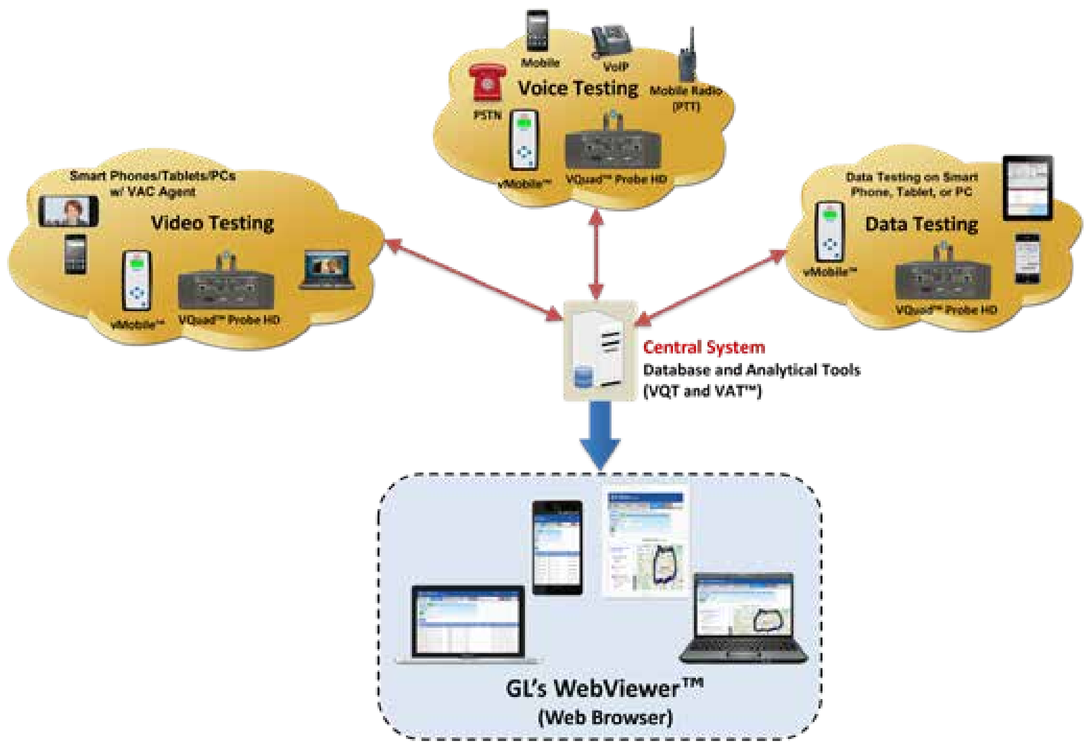
Video Quality Test Solution is used to determine the performance of a video call over a given network. It provides support for Android, PC and Linux based clients (end points) for active Video Quality Testing. Video Tests can be configured between any two active agents with flexibility on specifying the characteristics of the Video test. Tests can be run directly from the VQuad™ interface, or from the Android phones using the GLNetTest App configuring end points (Agents) and the test plan for the test.

The test plans are pre-configured on the VAC server and the list of available test plans is visible in both VQuad™/MDC, as well as GLNetTest App. A test can run between android to android, android to PC and PC to PC. Both manual and automated tests are supported using the flexible and versatile VQuad™ scripts.

Once the testing is complete, the results are sent to the central database, which can be viewed using WebViewer™.

- Single or multiple (consecutive and/or concurrent) test IP video calls between licensed Agents
- Supports Android, Windows, and Linux video end client devices
- Unlimited test plans configurations with Codec, Frame Rate, GoP (Group of Pictures) Structure and Video Image Size
- Video conferencing test results include end point details, Video Quality (Relative MOS-V), Audio Quality (Relative MOS-A), Audio Video Quality (Relative MOS-AV), IP Network condition parameters, Signaling Performance, and Call Config Info

WebViewer™ - Voice and Data Quality Testing with Centralized Monitoring



Overview

GL's **WebViewer™** is a robust web-based network quality and analysis tool to ensure optimal performance and reliability. WebViewer™ uses any web browser to display and access, in real time, all individual Voice, Data and Video Quality testing tools such as vMobile™ and VQuad™. The status of each operational test is displayed along with customized user statistics and graphical depiction of the testing results. In addition, using the WebViewer™ the user has full remote access to any node connected to the network.

GL's VQuad™ and vMobile™ offers automatic call control across multiple networks and supports real-time transmission/capture of voice files from various nodes. Recorded voice files are automatically transferred to the Central System where GL's Voice Quality software (VQT) analyzes and generates results using the POLQA (ITU-T P.863) or PESQ (ITU-T P.862) algorithms. The GL VQT algorithm delivers additional analytical metrics, such as jitter, clipping, and level measurements, providing comprehensive insights into audio quality.

GL's WebViewer™ empowers users to access Voice, Data, and Video Quality Measurements, Call Control Events, Errors, and Statistical results with ease, using a simple web browser. All results are retrieved from a central MySQL database, ensuring real-time and historical data availability.

With WebViewer™, users can easily query results from individual VQuad™/vMobile™ nodes, as well as Voice Quality measurements, One-Way and Round-Trip Delay measurements, Data Quality Tests, Video Quality Tests, Call Control events including Call Failed, Call Dropped, and FAX Events. Utilize user-specified search criteria and time filters to retrieve specific data.

Results are presented in both tabular and graphical formats, allowing for clear visual representation. Users can export all results to CSV, PDF, or Excel formats. Moreover, an user-customized consolidated view enables the user to customize how the results are displayed including line and bar graphs.

WebViewer™ - Centralized Monitoring

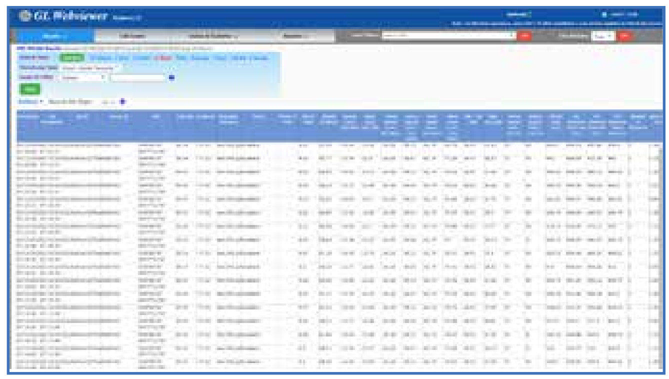
Using this feature results can be shown as Average/ Min/Max or as percentage. All results within the database are accessible for customized report viewing.

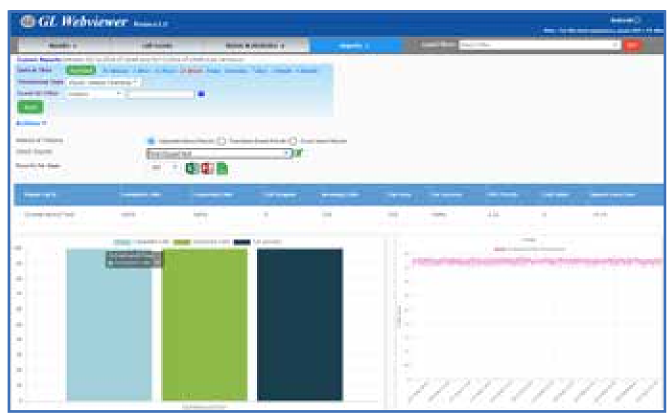
WebViewer™ can manually and automatically create custom reports with graphics and send the reports through email after the test is completed or during any interval (for instance every morning at 9am send a report).

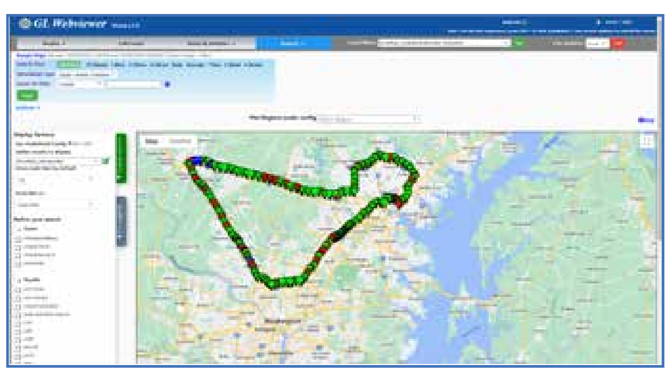
These WebViewer™ reports are based on the user-customizable reports which can also include both line and bar graphs. The emailed report can be in PDF, CSV, or Excel formats.

Key Features

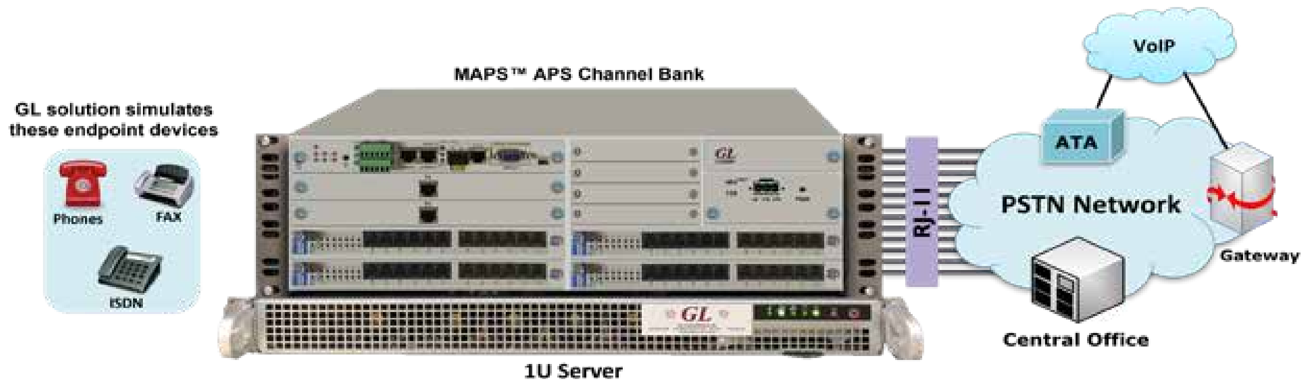
- Supports both Oracle and MySQL Databases on Windows® 10 and above Operating System
- Instantly view the real-time status of the entire network and remotely access any node within the network
- Remotely control both VQuad™ and vMobile™ nodes for starting/stopping scripts directly from the WebViewer™ web browser
- Pre-define vMobile™ and VQuad™ nodes and operations within WebViewer™, regardless of their network connection (public or private IP)
- Automatically sends all results from vMobile™ nodes, VQuad™ probes, Voice Quality software, NetTest (Data Testing), Video Conference testing, and FAX applications to the central database
- Seamlessly communicates with nodes and databases, supporting effortless backup, export, and import functionalities
- Easily segregates NetTest results (TCP, UDP, VoIP, Route, HTTP, FTP, DNS, SMS, and Email) and Mobile Device Information results (PhoneInfo, SimInfo, UEInfo) for clear and organized viewing
- Results include VQT (POLQA and PESQ), Call Control (Call Failure, Call Dropped, Call ID), Echo Measurements, Data Test (via Mobile device or PC Ethernet), Delay Measurements, Fax Tx Rx Events, and Video
- Analyze standard measurements and events (Call Control, Time Delay, VQT) through graphical and tabular views
- Export and schedule results, statistics, or custom output reports to *.csv, *.xls, or *.pdf file formats
- Plot results from vMobile™ and VQuad™ Nodes, as well as VQT software on Google Maps using available GPS coordinates during drive or walk testing
- Filter any results view based on specific measurements and call control conditions
- Supports Indoor Tracking (ITS) plotting







2-Wire Analog FXO/FXS Bulk Call Generator



Overview

MAPS™ APS is a high capacity analog 2-Wire FXO/FXS (or 4-Wire E&M) Bulk Call Generator that performs quality assurance tests for Central Office, PBX, Analog Telephone Adapter, Gateway or other telecommunications equipment, which provide local loop interfaces. It includes server hardware, GL MAPS™ software, and channel bank(s), along with optional modules (Fax Emulation and Voice Quality Testing Analysis) in a compact rackmount system. MAPS™ APS system supports up to 96 independent FXO ports or FXS ports per 1U MAPS™ APS/ALS server. More can be achieved by simply scaling the system with a 4U MAPS™ APS Server connected with 2 Octal T1 E1 Cards which can then support up to 384 analog ports.

MAPS™ APS supports Supplementary Service Testing and Interactive Voice Response (IVR) testing. Users can input DTMF digits or tones based on which the system presents a menu for automating various services. Provides high-density connection to any 2-Wire analog interface for fully automated custom testing.

MAPS™ platforms offers automated, scripted, multi-user, multi-protocol, and high-capacity Bulk Call Generation. This platform is the basis for all signaling protocols and for traffic generation – whether voice, tones, digits, fax, data, or video, depending on the network support. MAPS™ covers legacy PSTN, TDM, SONET SDH, next generation VoIP, and wireless protocols, interfaces, and equipment. MAPS™ can support any of the following protocols in TDM networks for establishing signaling links and generate or receive traffic - CAS, FXO FXS, ISDN, SS7, PPP, GSM, INAP, CAP, and MAP.

Supported Call Scenarios

- Caller ID
- Two-way Calling
- Three-way Conference Calling
- Three-way Calling with Calling Party Number ID
- VMWI – Voice Mail with MWI (message waiting indicator) and SDT (stutter dial tone)
- Call Waiting – Detect tone, Call ID, Flash to accept call
- Call Forwarding

Key Features

- Up to 192 independent ports per MAPS™ APS
- Test central office, PBX, Gateway, Analog/Digital/VoIP networks
- Call monitoring and call recording
- Concurrent users and tests per system
- Fully Automated with CLI and external control
- Full FXO and FXS Functionality via flexible scripts
- Supports IVR using GL's Speech Transcription Server
- API support (Python, Java) for integration with automation frameworks
- Supports E&M (Type I, II, III, IV, V) signaling – immediate start, wink start, delay start
- Voiceband Measurement Tests using T1 E1 Ports and VF ports

FXO Capabilities

- Support for up to 96 independent FXO ports per MAPS™ APS
- Full FXO functionality via flexible scripts
- Narrowband supported
- Supports Loop Start and Ground Start signaling
- Supported call scenarios:
  - Caller ID
  - 2-way Calling
  - 3-way Conference Calling
  - 3-way Calling with Calling Party Number
  - VMWI – Voice Mail with MWI (message waiting indicator) , SDT (stutter dial tone) and SIT (special information tone)
  - Call Waiting – Detect tone, Call ID, Flash to accept call
  - Call Forwarding

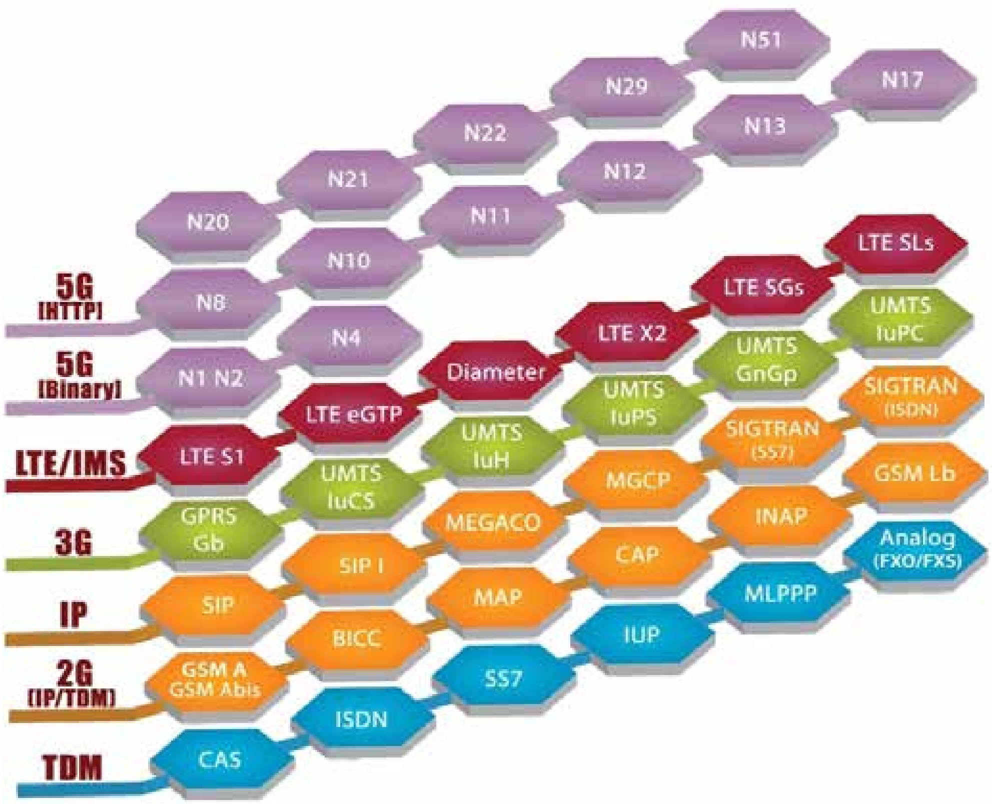
FXS Capabilities

- Support Up to 96 independent FXS ports per MAPS™ APS
- Central office emulation with two way calling
- Supports Loop Start and Ground Start signaling
- User-programmable call progress tone generation for different countries/regions:
  - Dial tone
  - Ringback tone
  - Busy tone
  - Reorder tone
  - Howler tone (extended off-hook signal)
  - Ring generation with programmable ring cadence

Reporting

- Multi-User, Multi-Test reporting in PDF and CSV file formats
- Reports Executed, Successful, and Failed test cases
- Call Failure, Completion, and Call Drop (sustain calls) events
- Voice Quality Test MOS Scores
- Delay Measurements (one-way and round-trip)
- Summarization with Failure Details sufficient to determine root cause
- Central DB of events/results/errors

Signaling and Traffic Simulator for Wireless, IP, & TDM Networks



Overview

Message Automation & Protocol Simulation (MAPS™) is a protocol emulation tool that supports various protocols across different technologies like 5G, 4G, 3G, 2G, VoIP and legacy TDM. The application gives users the unlimited ability to customize the control plane signaling messages and call flow sequences. Along with signaling, MAPS can also generate data and voice traffic (VoNR or VoLTE support) on established calls/ sessions.

MAPS™ supports the following protocols:

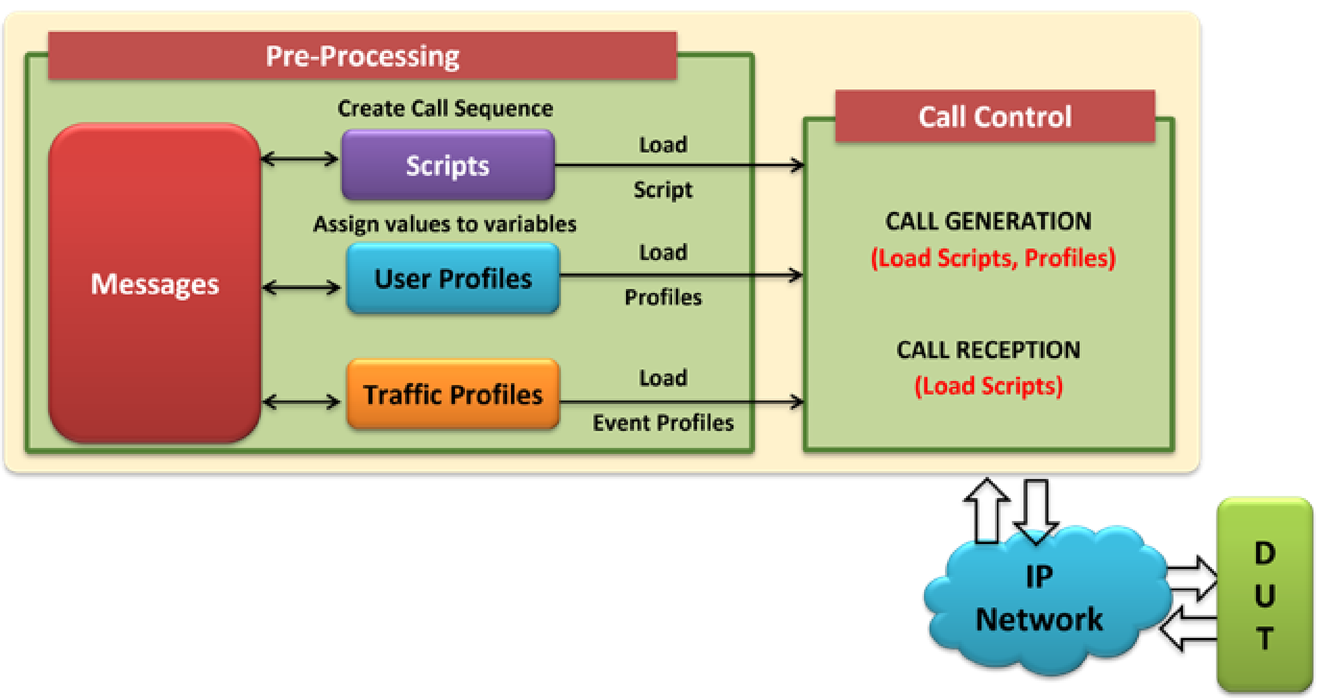
- **5G**: N1N2, N4, N8, N10, N11, N12, N13, N17, N20, N21, N22, N29, N51
- **4G**: S1AP, eGTP, Diameter, SLs, SGs
- **3G**: IuCS, IuPS, IuH, Gn GP
- **2G**: GSM A, GSM abis, GPRS GB
- **VoIP**: SIP, SIPI, MEGACO, MGCP, SIP Conformance suit
- **SIGTRAN**: ISUP, MAP, CAP, INAP, BICC,
- **TDM:** CAS, ISDN, SS7, IUP, FXO-FXS, Analog
- **Conformance Suits**: LTE S1, SIP, ISUP, SCTP, M3UA, M2UA, M2PA

MAPS™ transmits and detects various traffic types over IP (RTP, GTP), ATM, GSM (TRAU), and TDM such as digits, voice files, single tone, dual tones, fax, SMS, email, HTTP, FTP and video. MAPS™ also includes support for wide range of codecs. In GPRS, packet data traffic can be generated and validated with GTP traffic modules. Circuit switched traffic can be generated and recorded using RTP core module.

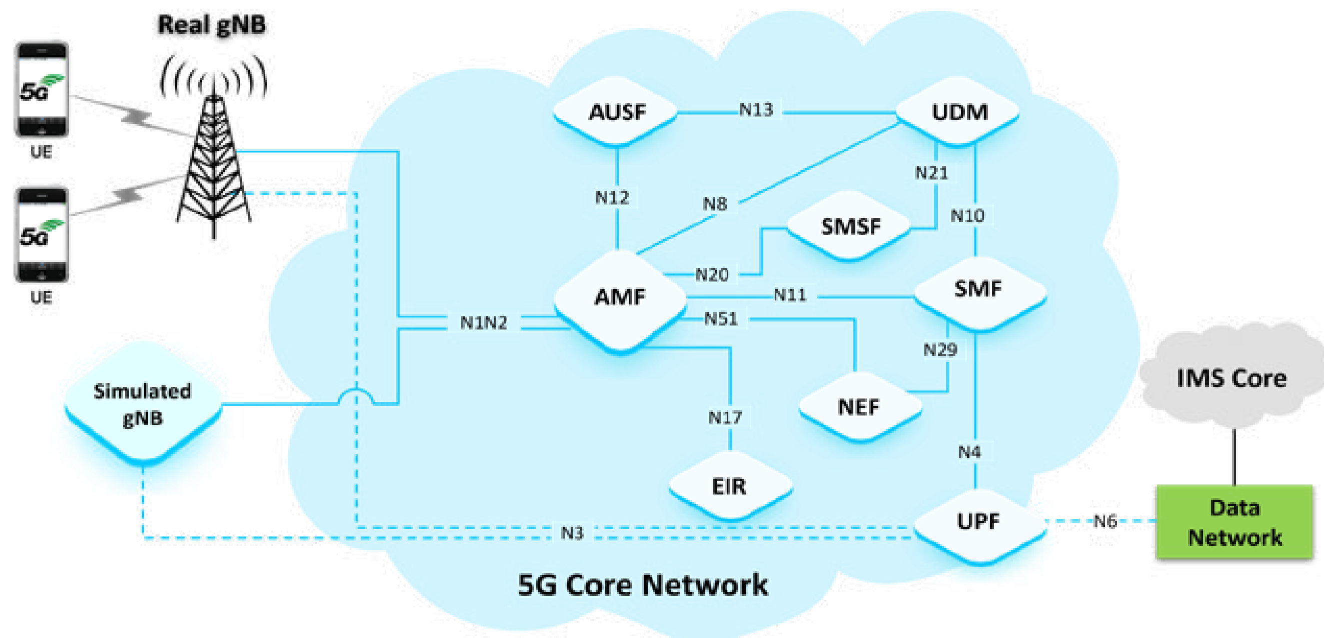
Key Features

- MAPS™ Wireless Lab Suite emulates the complete network allowing vendors to test applications and equipment
- Multi-protocol, Multi-interface emulation
- Flexible framework facilitates validation of any node functionality of any network element
- Provides complete access to user to customize any protocol IE's according to 3GPP standards
- Easily setup a test scenario within lab for educational purposes
- Powerful tool to test any element before deployment
- Supports both functionality and performance testing
- Easy to use GUI and allows integration with Python APIs
- Runs on Virtual Machines
- Emulate negative test scenarios by impairing messages and tweaking call flows
- Script based architecture allows customization of test scenarios
- High density user plane traffic generation in any network capability
- Centralized Control and Remote Access with CLI
- Conformance Test suites
- Customized call scenarios and test configurations
- Test Report and Statistics Generation
- Voice Quality measurements
- A single Remote Client GUI to remotely control/ monitor multiple MAPS™ Servers
- Client-server communication is facilitated through a Listener over TCP/IP
- Single Licensing Server option available for controlling number of MAPS™ Server and Clients (users)
- Auto generate massive number of subscriber profiles using internal Database, and CSV methods

Message Automation and Protocol Simulation (MAPS)



5G Core Network Emulation



Overview

5G network testing involves the emulation of control plane signaling with network traffic and data traffic ensure that the network's functionalities and performance meet the expected standards. GL's comprehensive Signaling and Traffic Emulation 5G test equipment suite, commonly referred to as MAPS™, provides a programmable framework for 5G device testing including emulation and [5G Core network testing].

In the End-to-End 5G network architecture, the network comprises various components, including the 5G Access Network (gNB), Access and Mobility Management Function (AMF), Authentication Server Function (AUSF), Network Slice Selection Function (NSSF), Unified Data Management (UDM), Session Management Function (SMF), Short Message Service Function (SMSF), Equipment Identity Register (EIR), and User Plane Function (UPF) connected to Data Server or Application Functions, and to EPC/IMS core for interoperability. All these underlying entities of the core network can be accurately tested for functionalities and performance with MAPS™ 5G test equipment suite.

With the capability of supporting enormous services and applications, massive connections, and new channel coding schemes at very high bandwidth, 5G network testing and troubleshooting are vital for ensuring the smooth operations of 5G networks.The use of a comprehensive 5G test equipment suite like MAPS™ is crucial for performing cross-domain testing, 5G Emulation,and 5G analysis using PacketScan™ application leading to the successful transition to new technology such as 5G.

The PacketScan™ 5G protocol analyzer supports monitoring of 5G networks. It captures, segregates, monitors, and collects statistics on all calls over N1N2, N4, N8, N10, N11, N12 and N13 interfaces of the 5G network. The 5G Protocol Analyzer is an optional module available within PacketScan™ on purchase of additional licensing. Monitoring Probes for 5G Wireless Networks capture CDRs, detect fraudulent activities, alert on critical parameters, measure KPIs, and performance statistics.

Key Features

Emulate Core Network Functions

- End-to-End 5G Network Emulation
- Emulates 5G - UE+gNB, AMF, SMF, UPF, AUSF, UDM, SMSF and EIR
- Feature and Functional Testing
- Performance Testing
- Inter-Operability Testing
- Migration Testing
- Advanced Voice Feature Testing - IVR, Voice Recognition, Speech-to-Text

Performance based on Massive UEs, GTP Traffic and Voice Quality Metrics

- Emulate Massive UEs (up to 64,000) with Voice Traffic
- Emulate User-plane GTP traffic at high line rates (up to 40 Gbps)
- Assess Voice Quality (eModel, PESQ, POLQA)

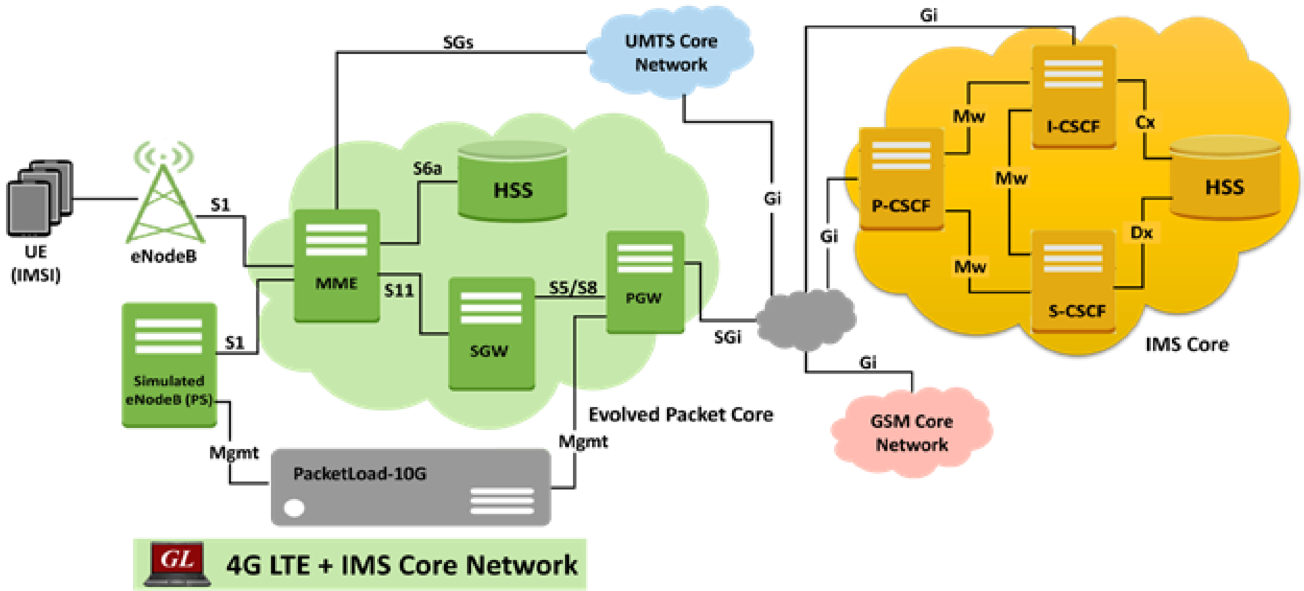
Monitoring Core Network

- Voice Quality, Data Retention, Lawful Interception, Fraud Detection
- Record thousands of Voice Calls - Filter and Record Only Calls-Of-Interest
- Capture up to 30,000 Simultaneous Voice Calls

End-to-End Voice, Video, and Data QoS Testing

- Speech metrics such as PESQ, and POLQA
- Automated data testing - TCP, UDP, VoIP, Route, HTTP, FTP, DNS, SMS, Email and more
- Measure delay, packet loss, drops, and more
- True performance of 5G and VoNR can be realized

LTE and IMS Emulation



Overview

GL offers solutions for monitoring, emulating and troubleshooting LTE, and IMS networks. The [LTE IMS Test Tool]() captures, decodes and conducts measurements across various interfaces in LTE and IMS wireless networks. MAPS™ can emulate nearly all elements in wireless LTE and IMS networks. When used in conjunction with the High-Density LTE Network Emulator, it can generate up to 100,000 UEs along with a high volume of mobile GTP and packet traffic to perform load testing on core LTE networks.

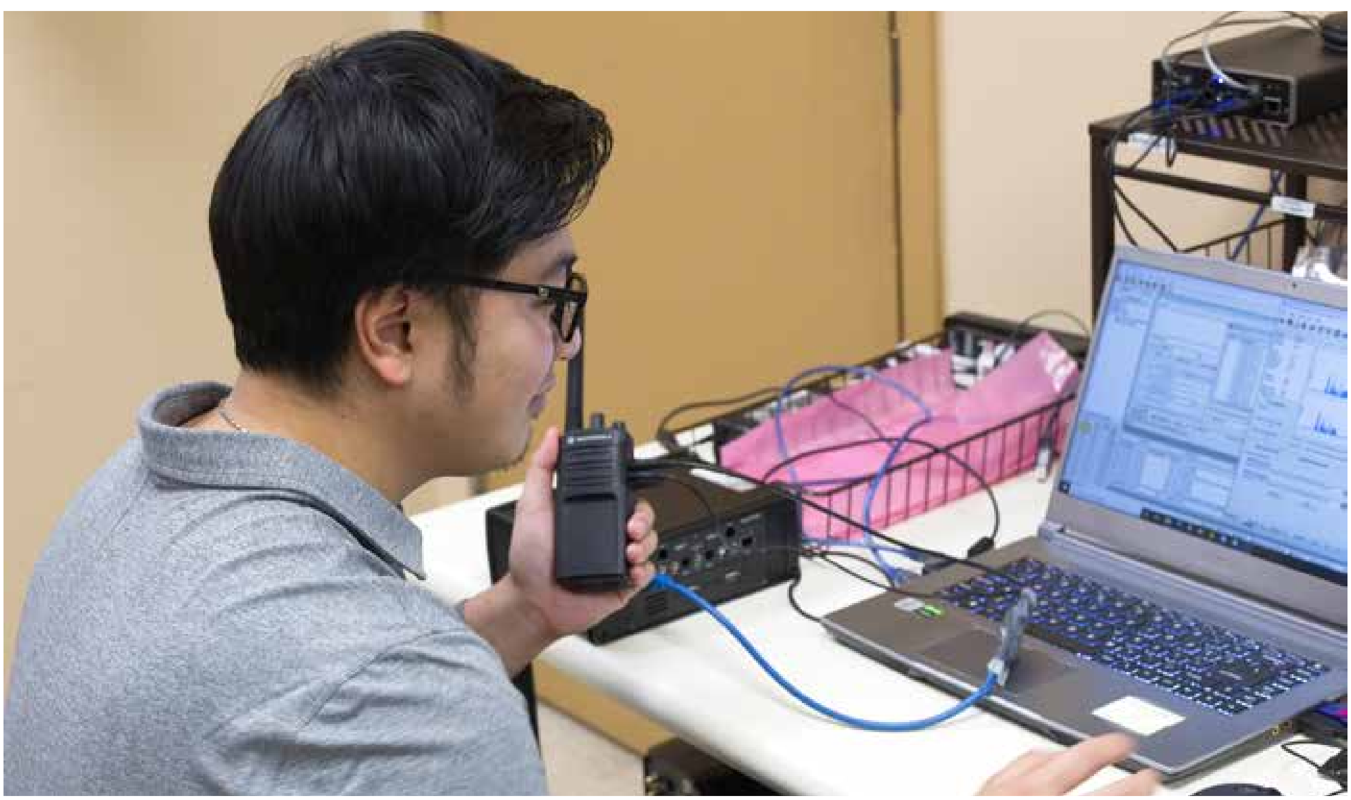
Users can establish a virtual real-time network with End-to-End 4G LTE-IMS Network Emulation Test Suite that emulates components such as Evolved NodeB (eNodeB), Mobility Management Entity (MME), Serving Gateway, PDN Gateway, Home Subscriber Server (HSS), Policy and Charging Rules Function (PCRF), Policy and Charging Enforcement Function (PCEF), Application Function (AF) and others to emulate Evolved Packet Core (EPC), allowing complete testing of the LTE network. All functionalities conform to industry standards. The End-to-End 4G LTE-IMS network emulation test suite offers reliable integrated solutions to vendors and service providers for emulating, monitoring and troubleshooting wireless networks.

LTE IMS Network Test Solutions Comprise:

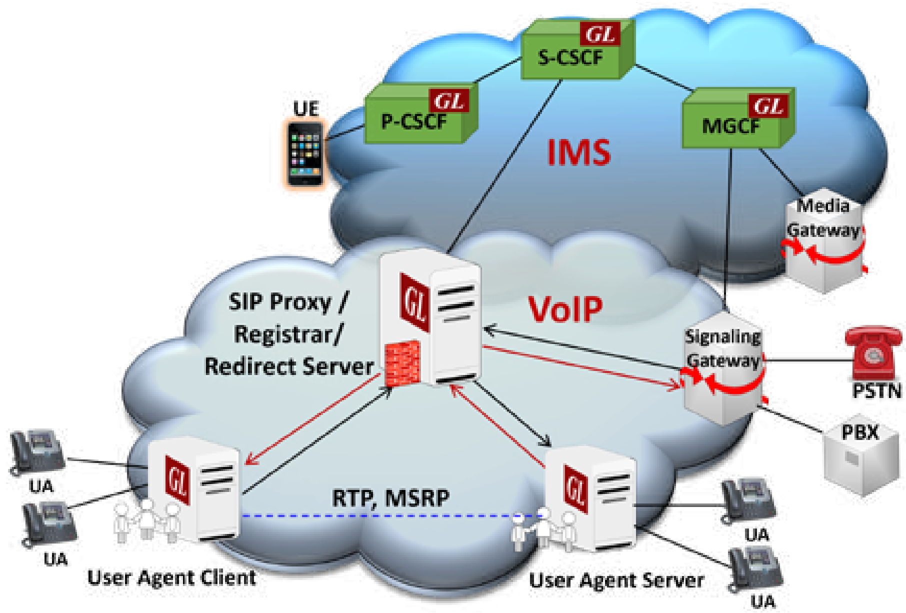
- 4G LTE Communications Network Lab and IMS Network
- Centralized Web-based LTE Monitoring System
- LTE/Quality of Service (QoS) Test Suite for Voice, Video, and Data Quality Testing
- Massive LTE UE and Traffic Emulation

Key Features

- MAPS™ IMS test suite emulates multiple UEs and IMS core elements such as P-CSCF, I-CSCF, S-CSCF, PCRF, MGCF which provides the IMS core network
- VoLTE Lab setup can be operated in real-time for making VoLTE calls and also for interworking with PSTN and VoIP networks
- Emulate several LTE interfaces such as S1, X2-AP, S3, S4, S5, S8, S10, S11 and S16, Diameter and IMS interfaces including Cx/Dx, Rx, Gx, Gm, SGi, Mw, Mi, Mj
- Generate and verify traffic over LTE, including VoLTE (Voice), FTP, Web (HTTP), Video, and more with additional licenses - Mobile traffic core - GTP and Mobile Traffic Core – Gateway
- Supports IMS-based technologies such as VoLTE
- Emulate up to 500 Smartphones (UEs) Powering Up and Down
- Authenticate and confirm security procedures
- QoS requests for greater or lesser bandwidth
- Temporary addressing management for mobility and security
- 4G IMS network setup for inter-networking with 2G/3G and UE roaming scenarios emulation
- Integrate IMS core network easily with any other networks (wired or wireless) to test any call scenario
  It can emulate various nodes across LTE network :
  - **S1-MME interface** : eNodeB (also called Evolved NodeB), and MME (Mobility Management Entity) nodes in
  - **eGTP-c interfaces** : MME (Mobility Management Entity), SGSN (Serving GPRS Support Node), SGW (Serving Gateway), and PGW (Packet Data Network Gateway)
  - **X2-AP interface** : Two eNodeB (also called Evolved NodeB) end-points
  - **Diameter interfaces** : MME (Mobility Management Entity), HSS (Home Subscriber Server), AF (Application Function), PCRF (Policy and Charging Rules Function), CSCF (Call Session Control Function), SGSN (Serving GPRS Support Node), PCEF (Policy and Charging Enforcement Function), EIR (Equipment Identity Register) and PDN GW (Packet Data Network Gateway)
- Authenticate and confirm security procedures
- Temporary addressing management for mobility and security
- 4G IMS network setup for inter-networking with 2G/3G and UE roaming scenarios emulation



SIP Protocol Emulation



Overview

GL's MAPS™ designed for SIP testing can emulate User Agents (User Agent Client- UAC, User Agent Server- UAS) Redirect and Registrant servers. MAPS™ test tool/ traffic generator can emulate any interface in a SIP network and perform protocol conformance testing (SIP protocol implementations).

The application is available as:

- MAPS™ SIP Protocol Test Tool
- MAPS™ SIP Conformance Test Suite
- MAPS™ HD Call Generator
- MAPS™ SIP Message Session Relay Protocol

A single MAPS™ instance can emulate thousands of SIP User Agents and can generate any SIP message on wire in the VoIP network, offering comprehensive testing capabilities that cover a wide range of SIP-based call procedures. MAPS™ SIP supports calls over UDP, TCP, and TLS transport types. Additionally, it supports Secure Real-time Transport Protocol (SRTP) for media traffic.

MAPS™ SIP supports transmission and detection of various RTP media traffic types, including digits, voice files, single tones, dual tones, IVR, FAX, and Video. It can handle up to 2000 concurrent voice calls at the rate of 250 calls per second. RTP traffic emulation is supported for almost all industry standard codecs. Capable of Interactive Voice Response (IVR) testing that recognizes and responds to voice prompts using DTMF digits or voice, allowing automated IVR traversal and testing.

MAPS™ SIP provides the Bulk Video Call emulation capability using pre-recorded video traces supporting codecs like H.264, H.263, and VP8. On a high-performance computing platform (core-i7), it is possible to generate more than 500 simultaneous video calls. MAPS™ SIP supports FAX over IP (FoIP) simulation for both RTP G.711 Pass Through Fax and T.38 Fax over UDPTL are supported along with monitoring and analysis.

MAPS™ SIP also supports generation of high volume of calls with traffic for load testing network using MAPS™ RTP HD network appliance.

MAPS™ RTP HD is a specialized 1U rackmounted network appliance designed to easily achieve up to 32,000 endpoints per appliance. Using a stack of multiple servers, a larger test system with 100K-200K calls (all controlled from a single Master Controller) is achievable for enterprise to carrier grade testing.

The MAPS™ SIP Conformance Scripts is designed with 400+ test cases, as per SIP specification of ETSI TS 102-027-2 v4.1.1 (2006-07) standard. Test cases include general messaging and call flow scenarios for multimedia call session setup and control over IP networks. Logging and pass/fail results are also reported. Test cases verify conformance of actions such as registration, call control, registrants, proxies and redirect servers.

MAPS™ SIP supports Message Session Relay Protocol for instant messaging over SIP sessions, simulating SIP/MSRP User Agents end-points in an NG9-1-1 network and send and receive communications over ESInets.

Key Features

Signaling

- Generates and processes SIP valid and invalid messages
- Supports IPv4 /IPv6 and transport over UDP and TCP, and TLS for secure transport
- Supports joining a conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, early media, and silence packets generation
- Implement IP Spoofing for any network like Class C, Class B, etc.
- Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER, and INFO SIP methods
- Generate custom SIP messages and call scenarios
- Feature with configurations to insert proprietary SIP headers in run time
- Automated the SUBSCRIBE transaction upon successful User Agent registration
- RTP Statistics log includes call detail record information for each call

Traffic

- Transmit and detect various RTP traffic such as digits, voice file, single tone, dual tones, IVR, FAX, and Video in IP networks
- Supports all industry-standard codec types - G.711 (μ-Law and A-Law), G.722, G.729, G.726, GSM, AMR, AMR -WB, EVRC, EVS, OPUS, SMV, iLBC, SPEEX, and more
- Supports Secure Real-time Transport Protocol (SRTP)
- Provides Voice Quality statistics such as Mean Opinion Score (MOS), Packet loss, and Jitter
- Supports both RTP G.711 Pass-Through Fax and T.38 emulation and analysis over IP
- Message Session Relay Protocol emulation (MSRP) supporting instant messaging
- Interactive Voice Response (IVR) testing that recognizes and responds to voice prompts using DTMF digits or voice, allowing automated IVR traversal and testing
- Supports Short Message Service (SMS) over IP/ IP Multimedia Subsystem (IMS) communication, SMS is encapsulated in a SIP message and carried over IMS core network

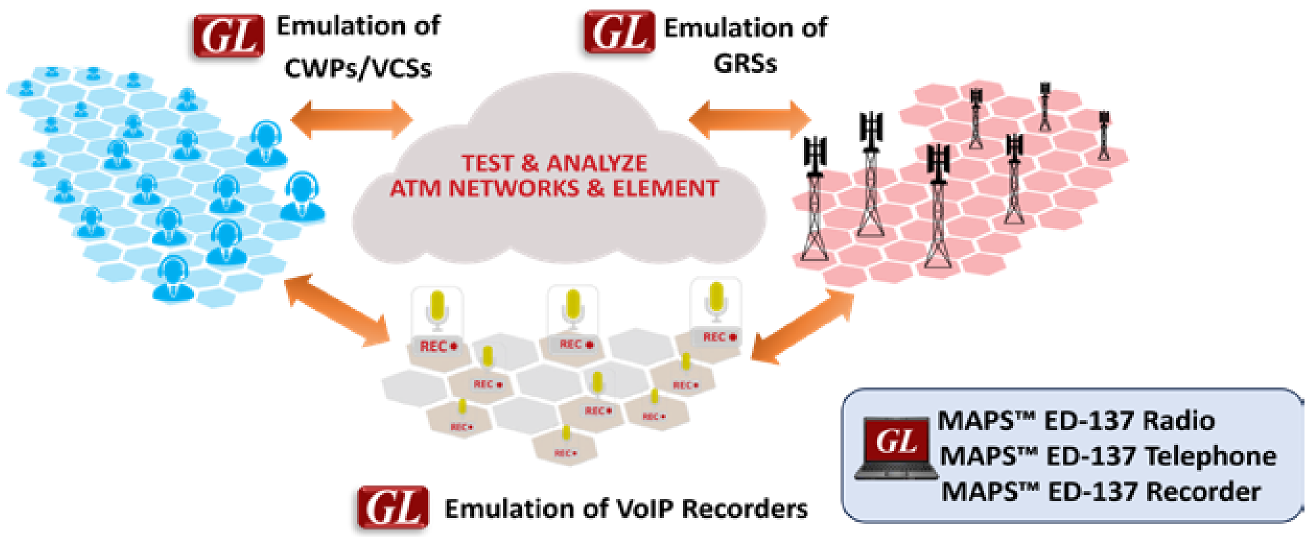
API / CLI

- MAPS™ CLI interface based on a client-server model allows users to control all features of MAPS™ through APIs
- Supported clients include Python and Java

Applications

- Complete SIP test environment
- Supports end-to-end gateway testing
- Supports conformance testing UAC, UAS, Proxy, Registrars, Registrants, Redirect Servers, and other SIP entities
- Handles strict routing and loose routing, when requests are routed through proxies
- Multi-protocol call trace for TDM / VoIP
- Testing NG9-1-1 emergency services and components within the ESInet

ATM Analysis and Simulation Solutions



Overview

GL's MAPS™ framework is used to simulate the Radio, Telephone and Recorder interfaces as per EUROCAE (European Organization for Civil Aviation Equipment) ED-137 standards. MAPS™ ED-137 Radio simulates air-to-ground calls by emulating either CWPs or Radios as per ED-137 volume 1 Radio. MAPS™ ED-137 Telephone simulates ground-to-ground calls by emulating CWPs as per ED-137 volume 2 Telephone. Optionally Telephone emulator supports Addendum 2: FAA Legacy Telephone Interworking, Addendum 4: Override Call and Addendum 5: Voice Call.

Similarly, MAPS™ ED-137 Recorder can emulate recorder interface at CWP, GRS, and Recorder Server as per ED-137 volume 4 Recorder. Hundreds of CWPs or Radios can be simulated and can generate hundreds of air-to-ground or ground-to-ground calls using a single instance/license.

Emulators support both B and C versions of ED-137 standards. All three emulators are enhanced to latest versions of standards (current release ED-137/C Change 1) and also support sessions over IPv6. Emulators are also validated against EUROCAE's VOTER tool. Emulators can be used for both functionality and load testing of CWP, VCS, GRS, Recorder and Radio Gateway interfaces. MAPS™ framework allows editing messages and creating custom call scenarios to simulate custom and negative test cases.

Key Features

Customized test solutions for VoIP Air Traffic Management networks

- Emulation Test Tools for ATM per ED-137
- ATM Network Quality Monitoring Tools per ED-138
- Critical Timing Measurement Tools for ATM
- Wide Area Network link emulation
- Inter-operability Test Tools

Emulation Test Tools for ATM per ED-137

- ED-137 B and C compliant and VOTER Validated
- MAPS™ ED-137 tools generate Air-to-Ground calls and Ground-to-Ground calls as per EUROCAE ED-137 (1B and 1C)
- MAPS™ ED-137 Recorder (4B and 4C) emulates call re- cording functionality at CWP, GRS, and Recorder interfaces
- Test the functions of Controller Working Position, Ground Radio Station, or Radio Media Gateway entities
- Emulate hundreds of CWPs/Radios with unique IP ad- dresses in a single instance
- Supports hundreds of simultaneous calls and complete
- automation of bulk call generation with traffic
- Fully integrated, complete test environment for Air Traffic
- Management
- Supports audio codecs G.711 (A-Law and μ-Law) and G729
- Supported traffic actions – send and record to file, send and detect digits/tones, Talk using microphone and play to speaker.
- Impairments (Packet Loss, Duplicate, Out of sequence and Latency) can be applied to RTP traffic
- Supports Python APIs to integrate into third party test platform.Depicts easy to understand call flow graphs of SIP mes- sage exchanges and message contents (SIP headers and SDP attributes)

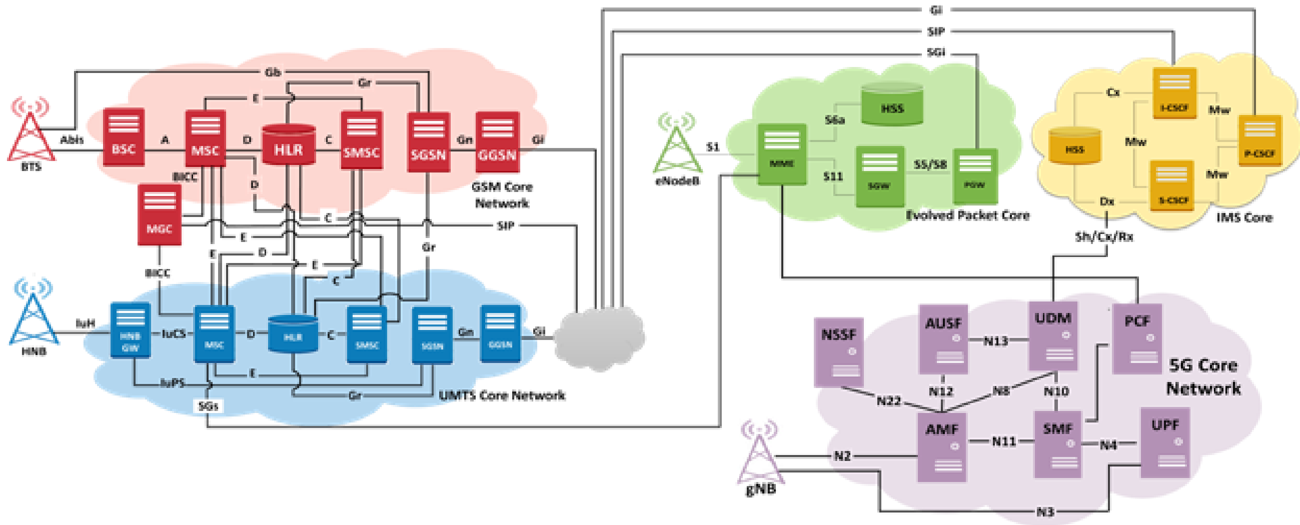
ATM Network Quality Monitoring Tools per ED-138

- PacketScan™ captures and monitors live signaling and traffic over Air Traffic Management network
- Analyze calls for voice quality (MOS), packet loss, jitter, latency, etc.
- Waveform viewer, Call-flow graphs, and QoS monitoring
- Real-time and/or historical data analysis using graphical reports
- Centralized monitoring of several probes deployed over the network with NetSurveyorWeb™
- Triggers and Actions allows to filter calls of interest and send e-mail notifications or log the event.

Precise Timing Measurement Tools for ATM

- MAPS™ TM-ATM (Timing Measurements in Air Traffic Management) test suite accurately emulates end points in ATM networks and provides critical timing measurements for various types of delay occurrences in signaling and voice transmission through the network
- Includes all necessary hardware and software to identify, capture, timestamp, and correlate events at Analog, TDM and IP interfaces
- Performs End-to-End voice quality measurement using POLQA/PESQ ITU-T standards. Capture, Filter and Fimestamp only packets of interest
- Tests can be automated and run for multiple iterations to performance stability and consistency
- Packet filtering can be based on all Layer 2 (Ethernet), Layer 3 (IP), Layer 4 (UDP/TCP) Headers and Data
- Uses GPS time to perform precise timing measurement during deployment and field testing

End-to-End Wireless Lab Solutions



Overview

GL's End-to-End Wireless Network Simulation Test Suite offers an advanced and comprehensive "Live Network" experience directly at your company premises, tailored to meet specific testing requirements in a customized package.

This complete test suite is developed on the standardized MAPS™ platform framework architecture. The application allows to simulate various elements within the wireless network infrastructure, utilizing straightforward, ready-to-use test bed setups.

Ideal for vendors and service providers, the test suite delivers reliable integrated solutions for simulating, monitoring, and troubleshooting wireless networks, covering 5G, 4G, 3G, and 2G technologies—all in adherence to industry standards.

Efficiently test, monitor, and troubleshoot core network elements and diverse traffic types within the wireless infrastructure to ensure deployment readiness, functionality, interoperability, optimal performance, and compatibility with the latest mobile features. Additionally, validate end-user applications, devices, and services within a simulated wireless environment before deploying them on a real-time network.

Key Features

Simple-to-setup and execute

- All operations and individual nodes in a network can be controlled remotely from a Remote MAPS™ system
- Simple test bed configurations to establish communication between different network elements, and the mobile phones
- Ability to add unlimited number of user profiles, and scripts
- The ready-to-use scripts makes testing procedure simpler which are used to quickly setup calls generating and verifying data traffic

Performance

- Higher volume Voice and SMS calls (hundreds of calls/sec and 64,000 simultaneous calls/platform) with MAPS™ RTP HD appliance
- Easily achieve massive simulation of UEs (up to 500000) with high density (up to 4 Gbps or 40 Gbps) mobile traffic (stateful HTTP/PCAP) generation per MAPS™ PacketLoad appliance

Scalable

- Simply scale up with a greater number of systems to achieve higher performances

Programmable and Scriptable

- MAPS™ based simulators provide a unique "programmable, scriptable" framework independent of any protocol or network
- Supports a variety of protocols under common architecture framework
- Fine control any parameter in the emulated network environment. The scripts are events-driven and provide fine control over call behaviour
- Unlimited ability to edit signaling messages, protocol fields, and call flows
- Precisely time communication between different core network entities

Use Cases

- Test and verify web services, and mobile applications over emulated wireless networks
- Pin-point and troubleshoot network issues with comprehensive logging and analysis tools
- Monitor End-to-End connectivity with automated call flows and data connectivity over multiple interfaces
- Test Inter-Operability issues and handovers with different network lab setups
- Perform Feature Tests, Conformance Tests, and Functional Tests on individual nodes - supports both single and Multi-interface testing
- Performance Testing
- Migration Testing
- Voice and Data Quality measurements with comprehensive voice and data quality tools
- Advanced Voice Feature Testing

tProbe™ - T1 E1 Analysis & Emulation

Overview

GL's tProbe™ is a test and measurement device for many legacy networks including T1, E1, Analog and Datacom. These networks are still used throughout the world due to their reliability and the prohibitive expense of removing such infrastructure. The tProbe™ can monitor and emulate common voice protocols including ISDN, SS7, CAS, etc. The tProbe™ also includes optional boards such as Datacom (DCE or DTE) and FXO FXS ports. The FXO port on the tProbe™ can simulate a 2-Wire FXO device such as a telephone or a fax machine.

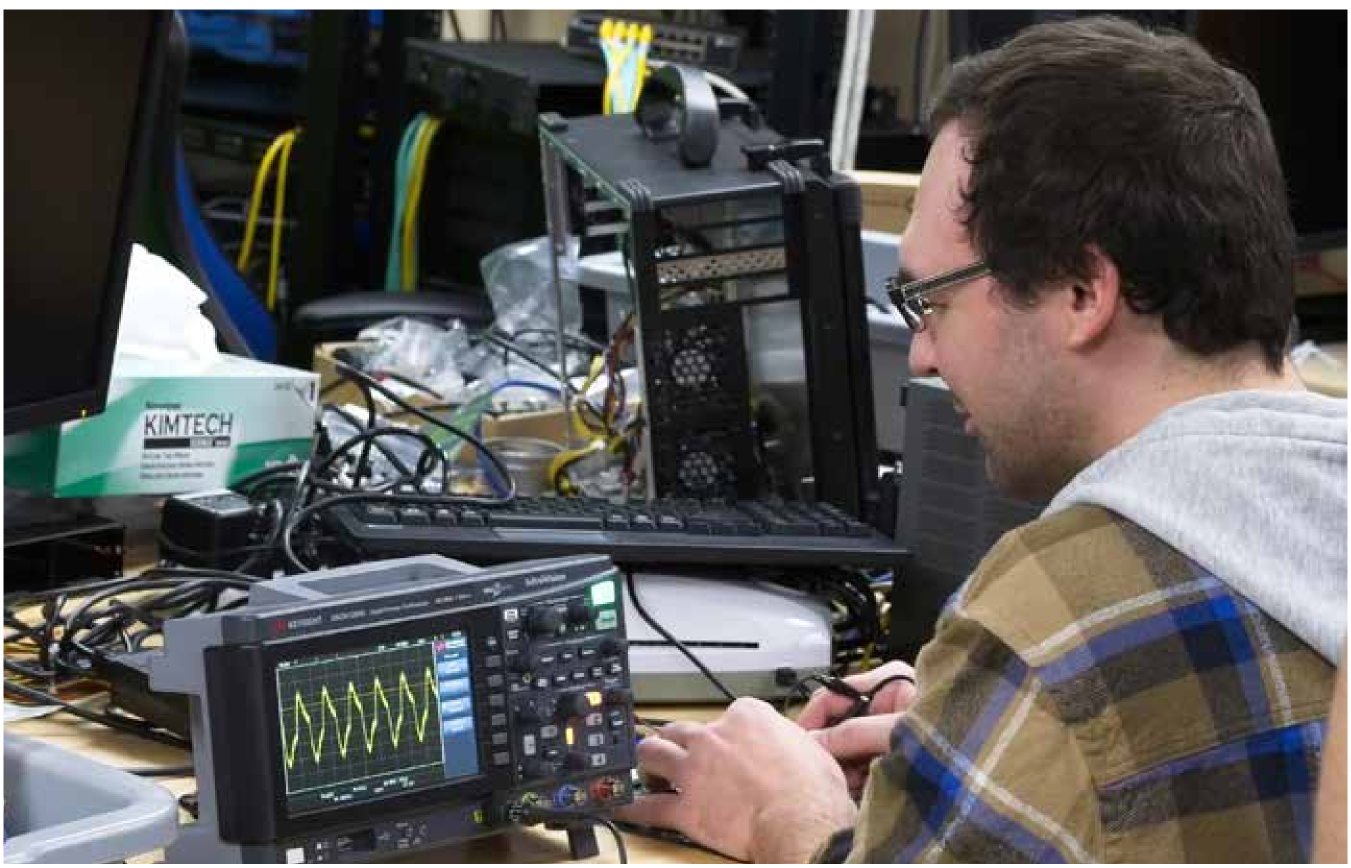
The tProbe™ Datacom Analyzer is designed for the installation, verification, and maintenance of data communication and telecom equipment. It offers a software-selectable interface for emulating DTE and DCE, as well as monitoring data communication lines for both Synchronous (Sync) and Asynchronous (Async) modes of operation.

The tProbe™ is controlled by a Windows® PC via a USB connection. The tProbe™ software includes an easy-to-use Graphical User Interface for configuring ports, test parameters, starting and stopping tests and exporting results. It also includes Python scripting capability.

In addition, GL also provides T1 E1 Octal/Quad Card PCIe based solution for higher density which supports multiple T1 E1 ports for analyzing and emulating TDM networks. Multiple PCIe cards can be placed in a single server grade PC for enhanced scalability.

Key Features

- Comprehensive analysis/emulation of Voice, Data, Fax, Protocols (such as SS7, ISDN, Frame Relay, GSM, HDLC, PPP, V5.X, MLPPP, ATM, CAS), Analog, and Digital signals, including Echo and Voice Quality testing
- Monitor T1 E1 line conditions such as frame errors, violations, alarms, frequency, power level and clock (or frame/bit) slips. Monitor all timeslots in real-time
- T1 E1 Pulse Shape, Jitter Measurement Analysis, and Jitter Generation
- Software selectable T1 or E1 interface along with Drop and Insert
- tProbe™ FXO and FXS board allows simulating FXO and FXS ports. The FXO port is used to simulate a 2-Wire FXO device such as a telephone or a fax machine. The FXS port is used to simulate a 2-Wire FXS service such as a telephone wall jack
- Datacom board supports V.24, V.35, V.36, RS-449, RS-485, EIA-530, and EIA-530A interfaces and can be configured as DTE or DCE to test Channel Service Unit (CSU) and Data Service Unit (DSU) entities
- Physical layer analysis includes the ability to send alarms and errors via SNMP Traps
- Enhanced VF Drop and VF Insert Capabilities using 3.5mm Balanced (Stereo), or Unbalanced (Mono) physical connections
- Python scripting support on both Windows® and Linux® operating systems
- Routing and Bridging emulation over Multi T1 E1 WAN interfaces using MLPPP (Multi Link PPP) and Multi Link Frame relay (MFR) protocols
- Call Recording, generation, and monitoring for hundreds to thousands of calls in one platform
- Capable of simulating as well as decoding and demodulating fax calls over T1 E1 lines using Fax Simulator and FaxScan™
- Cross-port Through and Cross-port Transmit modes configurations make cabling with Drop/Insert and Fail-Safe Inline monitoring easy
- Lightweight (1.24 lbs) and small footprint (6.05” x 5.55” x 1.60”)



tProbe™ Probe

The controlling computer can be placed into the same chassis as the tProbe™. The controlling computer comes pre-installed with all software and licenses. It contains an Ethernet port for remote accessibility (via Remote Desktop Protocol), HDMI and USB ports for monitor, mouse and keyboard. This solution retains portability and is ideal for field testing.

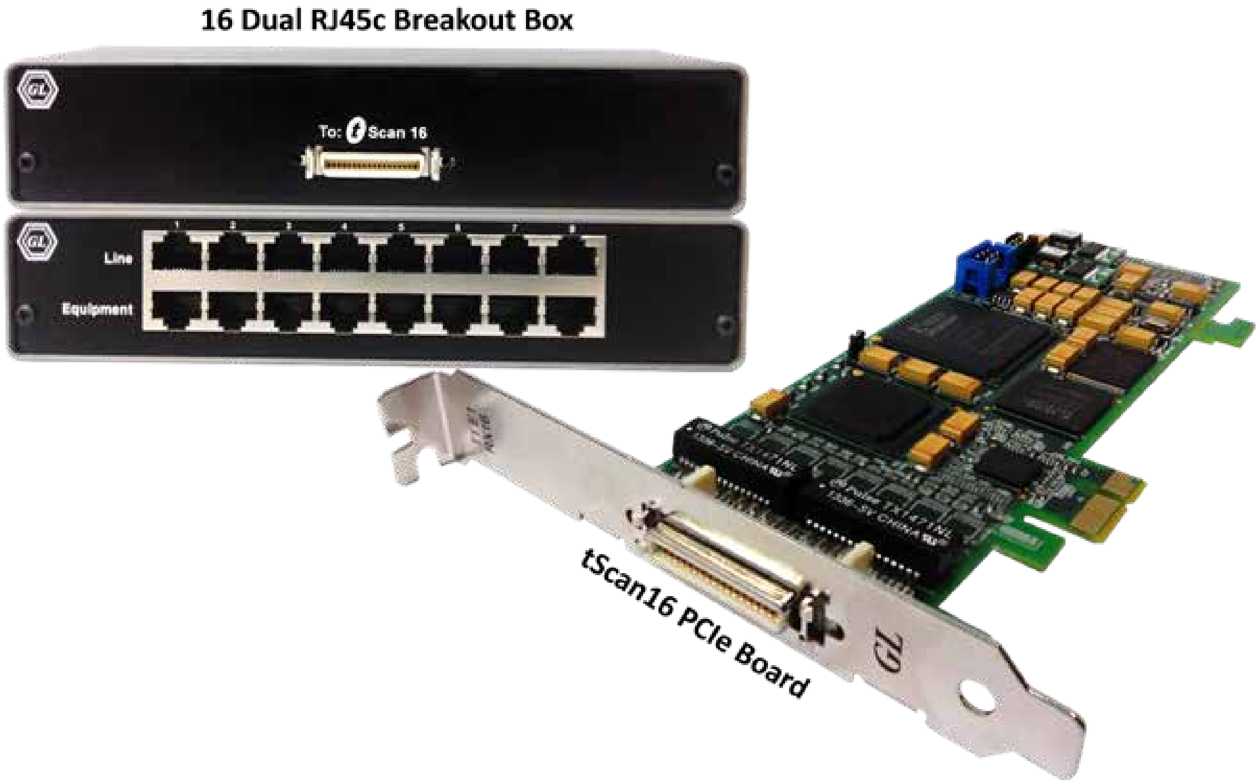
Front Panel

Back Panel

tProbe™ Rackmount

The tProbe™ and controlling computer can be placed into a 1U rackmount chassis. This solution is ideal for long term testing from a single location such as a network room or lab environment.

tScan16™ - T1 E1 Analysis Hardware



Overview

GL's tScan16™ provides greater density, increased ports, and reduced power as compared to other TDM analysis tools on the market. The boards (with Direct Memory Access) are significantly faster, more efficient, and well-suited for High-Density cabling.

The tScan16™ consists of a PCI Express board housed in a host Windows® computer. Customers can use their own computer or GL can provide the PC as a rackmount (1U) or portable (Lunchbox) PC. Also included is the
16 port Breakout box for connecting to the TDM network under test. The tScan16™ Breakout Box is used to receive the T1 E1 traffic on 16 ports for tScan16™ application. It consists of 8 pairs of Line and Equipment RJ45c ports. Users need to connect straight cables to Line side and Cross-over cables to Equipment side to receive the signals from both Line and Equipment.

Key Features

- Software selectable 16 Rx only T1 E1 interfaces
- Monitor T1 E1 line conditions such as frame errors, bipolar violations, alarms, frequency, power level, and clock (or frame/bit) slips
- Analysis of ISDN, SS7, Multilink Frame Relay, Multilink PPP, HDLC, GSM, GPRS, UMTS, and many more protocols
- Comprehensive analysis of Voice, Data, Fax, Modem, including Echo and Voice Quality testing
- Call Recording, Analysis, and Monitoring for hundreds to thousands of calls in one platform
- Reduces hardware costs and power consumption

T3 E3 Signal Analyzer for Channelized & Unchannelized Solutions

T3 E3 Analyzer

Overview

T3 and E3 network infrastructure remains highly used throughout the modern world. GL's T3 E3 Analyzer is a versatile tool that handles signaling, voice, and data in full T3 (DS3) or E3 data streams. It can drop and insert T1 (DS1) or E1 and analyze HDLC, ATM, Frame Relay, and PPP Protocols. This analyzer is equipped for both Unchannelized (Unstructured) and Channelized (Structured) T3 E3 Traffic.

Channelized: In this form, T3 (DS3) includes 7 T2s (DS2), and each T2 has 4 T1s (DS1), totaling 28 T1s or 672 voice channels. Similarly, an E3 includes 4 E2s, with each E2 having 4 E1s, totaling 16 E1s or 480 voice channels. Channelized T3 carries 28 T1 signals, each at 1.544 Mbps, while channelized E3 carries 16 E1 signals at 2.048 Mbps.

Unchannelized: Here, there are no T2s, T1s, E2s, or E1s within T3 or E3. Most of the capacity is used for data, with some overhead. T3 and E3 can transport data, packetized voice, and video, including protocols like ATM, PPP, HDLC, and Frame Relay.

A T3 line can transmit data at speeds up to 44.736 Mbps, while an E3 line operates at 34.368 Mbps within the Plesiochronous Digital Hierarchy (PDH). The channelized option in the T3 E3 Analyzer directly accesses all the channels within a T3 or E3 line, whether it is 2x28 T1s, 2x21 E1s, or 2x16 E1s, allowing for emulation, analysis, and monitoring on a single PC. This includes support for various T1 E1 framing formats, physical layer alarms, and payloads.

Similar to other GL products, the USB T3 E3 Analyzer is controlled by a Windows® PC via a USB connection. The software has an easy-to-use Graphical User Interface where users can configure test parameters, start and stop tests and view real time data and graphs.

GL also offers T3 E3 probe and rackmount variants, combining the USB-based T3 E3 Analyzer hardware unit with a Single Board Computer pre-installed with all software and licenses. This setup is ideal for field testing (probe version) or high-density lab testing (rackmount version).

The T3 E3 hardware platform with associated T1 E1 Send Receive Server and Channelized T3 analyzer software can transmit T1 E1 frames over T3 E3 lines and capture, record, and monitor multiple T1 or E1 channels over Channelized T3 or E3 links. It can perform analysis of various signal types including voice, digits, tones, fax, modem, and raw data.

Key Features

- Portable light-weight T3 (DS3)/E3 analysis platforms with dual data stream capture capability
- Multiple interfaces for analysis (T3 (DS3)/E3, T1 E1) to support a wide array of testing scenarios
- Controllable from a Windows® PC through a USB 2.0 control interface
- Software Selectable T3 (DS3) and E3 interface along with T1 and E1 Drop and Insert
- Dual T3 (DS3) /E3 Receivers and Transmitters for non-intrusive and intrusive testing of both eastbound and westbound signals at the same time
- Flexible clocking - internal, recovered (from T3 E3, T1 or E1) and external
- Scripting and Automation through GL's Windows Client Server (WCS) approach
- WCS clients are available for Windows® and Linux® operating system via console/terminal Command Line Interface and accessible remotely through SSH
- WCS commands can be issued in Python scripts running in Windows® or Linux®
- Includes HDL File Conversation utility to convert ethereal format file (*. pcap, *.cap, and *. pcapng) to GL's file format (*.hdl) and vice- versa

Unchannelized (Unstructured) Testing

- Analysis and simulation of ATM, PPP, HDLC, and Frame Relay protocols
- Transmit/Verify HDLC frames with user-defined headers
- Support for Scrambling and Subrate with DSU vendors' algorithms for T3 interface, including Digital Link, Larscom, Verilink, and Ad- tran

Channelized (Structured) Testing

- Drop and Insert Functionality
- Utilizes the Dual T1 E1 ports on the hardware
- Multiplexes and De-multiplexes T1 (DS1)/E1 signals (Drop and Insert)
- Receivers for bidirectional monitoring with Dual T1 (DS1)/E1 drop
- Transmits multiplexed externally inserted or internally generated T1 or E1 streams into T3 (DS3)/E3
- Conducts stress tests on M13 (E13) multiplexers and 3/1 Digital cross-connect systems
- Dual channel drop and insert of T1 or E1 signals from any one of the T3 (DS3)/E3 signals
- Offers options for broadcasting or loopback of individual T1s/E1s received from T3 (DS3)/E3
- Conducts stress tests on M13 (E13) multiplexers and 3/1 Digital cross-connect systems.

Bit Error Rate Test (BERT)

- Perform BERT on T3 E3 channels simultaneously
- Support BERT through WCS commands

T1 E1 Tx/Rx Server Software

- Transmit T1 E1 frames over T3 E3 lines using the T1 E1 Transmit Server
- Receive T3 E3 data with the T1 E1 Receive Server, which demultiplexes it into T1 or E1 channels
- Send/Receive 28 T1 Channels per port from T3 signal
- Receive 21 E1 Channels per port (G.747 Mapping) from T3 signal
- Receive 16 E1 Channels per port from E3 signal
- Supports monitoring of framed and unframed T1 E1 (Rx Only)
- Simultaneous analysis of all 56 T1s (1.544 Mbps each) or 32 E1s (2.048 Mbps each)
- Analysis of Fractional T1s and E1s, N x T1s or N x E1s
- Analysis of any combination of DS0s (64 kbps each) within the T1s or E1s, totaling 1,344 DS0s for T1 or 1,024 DS0s for E1
- Supports Protocol Analysis for all structured protocols including HDLC, ISDN, CAS, and more
- Monitors T1 E1 Alarms, Payload, and Framing structure

T3 E3 Probe

T3 E3 Probe unit includes GL's USB based T3 E3 Analyzer hardware unit combined with necessary PC interface, which makes it portable stand-alone unit suitable for field testing. A single portable USB-based unit supports 2x T3 E3 ports per unit. This solution retains portability and is ideal for field testing.

GL's T3 E3 Analyzer is capable of processing signaling, voice, and data full T3 (DS3) or E3 data streams, dropping and inserting T1 (DS1) or E1, and analysis of HDLC, ATM, Frame Relay, and PPP Protocols. It includes various signal testing capabilities for Unchannelized (Unstructured) and Channelized (Structured) T3 E3 Traffic. Shown below is a Probe based T3 E3 Analyzer unit.

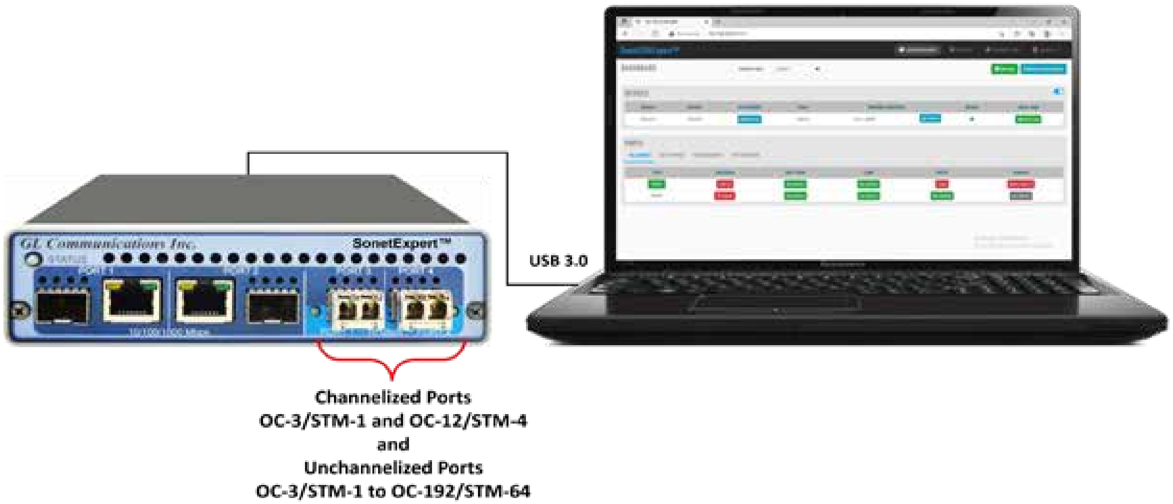
Front Panel

Back Panel

T3 E3 Rackmount System

The T3 E3 analyzer can be housed in a 1U rackmount chassis. Also included in the chassis is a Single Board Computer (SBC) for controlling the T3 E3 analyzer. The SBC has a Windows PRO Operating System and comes pre-installed with all software and licenses. It can be accessed locally using HDMI and USB ports. It can also be accessed remotely through Remote Desktop Protocol through its Ethernet port. This solution is ideal for long term testing from a single location such as a network room or lab environment.

SonetExpert™ - Channelized & Unchannelized Testing up to OC-192



Overview

Most of the backbone transport for voice, video and data applications continues to be SONET and SDH optical transmission networks. SONET and SDH transmission networks also carry channelized traffic such as T1 E1, T3 E3 pipes.

GL's SonetExpert™ application has the following variants of OC-3/STM-1 and OC-12/STM-4 Analyzers:

- SonetExpert™ Channelized Analyzer
- SonetExpert™ Unchannelized Analyzer

Channelized Analyzer comprises of hardware and software. The hardware receives and transmits data using SONET/SDH traffic which tranfers the traffic in to the GL's Soft T1 E1 analyzer application. The T1 E1 Analyzer application provides the same functionality as GL hardware based T1 E1 Analyzers with the difference that T1 E1 frames are multiplexed into SONET/SDH frames and transmitted over optical lines.

GL's SonetExpert™ Unchannelized Analyzer is capable of SONET/SDH testing over OC-3/STM-1, OC-12/STM-4, OC-48/STM-16 and OC-192/STM-64 transports. It is based on the PacketExpert™/SonetExpert™ hardware platform. Packetexpert™/SonetExpert™ is a versatile hardware platform that supports both Ethernet (up to 10G) and SONET/SDH (up to OC-192/STM-64) testing, two ports support SONET/SDH testing.

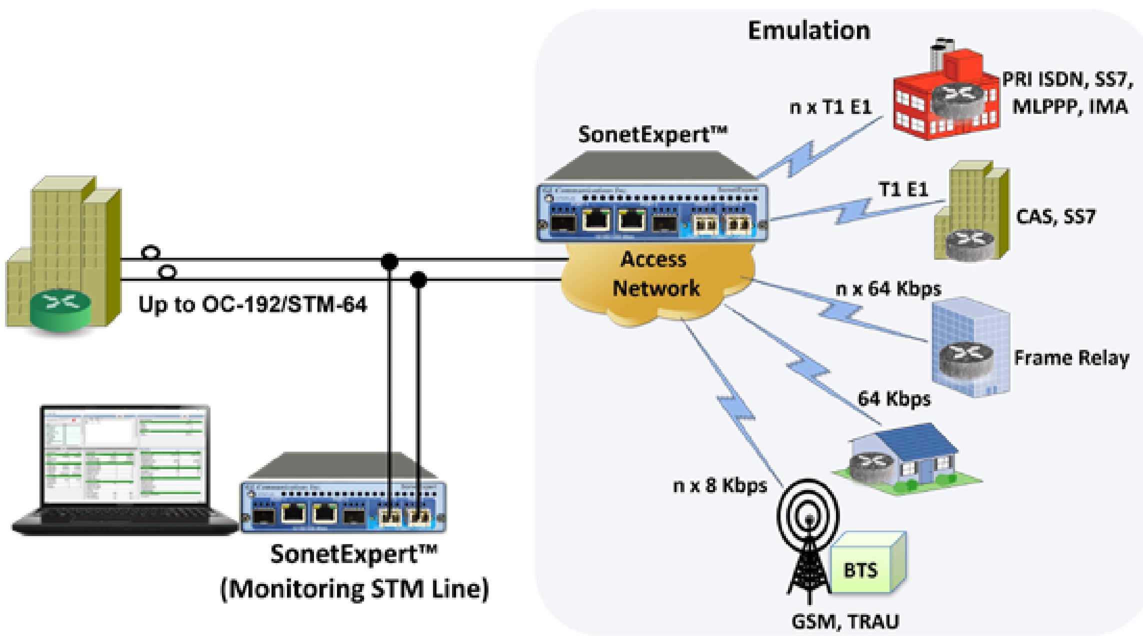
Channelized Key Features

- Allows direct access to everything on SONET / SDH – Framing and Payload, including structured traffic (T1, E1, STS-1 etc.)
- Analyze or emulate voice, data, fax, protocols, analog and digital signals, including echo and voice quality
- Pluggable SFPs allow Single-mode (SM), and Multi-mode (MM) fiber optic non-intrusive tap
- Comprehensive protocol analysis and emulation - HDLC, SS7, ISDN, CAS, PPP, Frame Relay, ATM, UMTS, and more
- Capture and transmit at wirespeed to/from hard disk on all interfaces. Also capture traffic for off-line analysis and playback the captured traffic

Unchannelized Key Features

- Software is provided as a web interface
- SonetExpert™ can perform Unchannelized BER Testing over OC-3/STM-1, OC-12/STM-4, OC-48/STM-16 and OC-192/STM-64 SONET/SDH rates
- PoS/ATM/Raw captured traffic can be analyzed in real time (for OC-3/STM-1 and OC-12/STM-4)
- SonetExpert™ supports capturing wirespeed traffic (for OC-3/STM-1 and OC-12/STM-4) traffic on 2 ports simultaneously to a file on hard disk, with hardware filtering and timestamping
- SCAN application supported on OC-3/STM-1, OC-12/STM-4, OC-48/STM-16 and OC-192/STM-64 rates

SonetExpert™ Channelized Application

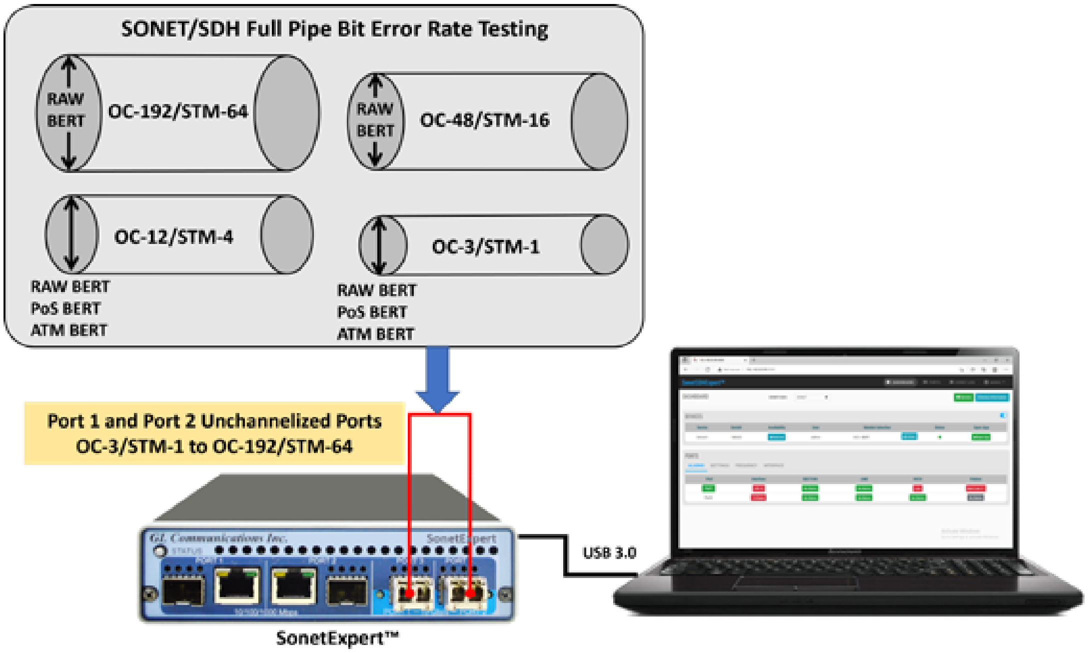


GL's SonetExpert™ Channelized Analyzer supports direct access to all or user-defined 2 x 336 T1s or 2 x 252 E1s per unit for analysis and simulation – all within a one PC. With this hardware, any combination of DS0/64 kbps, fractional T1 E1, and N x T1 E1 interface definitions (a total of 126 E1s or 168 T1s – each port supporting 84 T1s or 63 E1s) can be accessed.

Supports analysis and simulation of various TDM and wireless protocols in real-time / remote / offline. The following are the available protocol analyzers:

- HDLC, SS7, ISDN, CAS, GSM, TRAU, SS1
- Sa Bits HDLC, SSM, V5.x, DCME
- FDL (T1 Interface only)
- ML-PPP, ML-Frame Relay
- ATM IMA
- GPRS (Gb and IP Gx)
- UMTS

SonetExpert™ Unchannelized Application



GL's SonetExpert™ Unchannelized Analyzer supports Bit Error Rate (BER) testing, BER Traffic generation, verification of various PRBS and user defined test patterns over OC-3 / STM-1, OC-12 / STM-4, OC-48/ STM-16 and OC-192/STM-64. Various Error insertions like, Bit Error Insertion, B1/B2/B3 BIP Error insertion, Alarm generation etc. are supported along with BERT testing.

Three types of BER testing are supported - BERT over Raw SONET/SDH frames, BERT over ATM and BERT over PoS. SonetExpert™ Unchannelized Analyzer also includes Record to File and Playback from File features (for OC-3 / STM-1, OC-12 / STM-4 only). The Record and Playback application allows users to capture and transmit packets at wirespeed. This feature allows capturing real world traffic and simulating it by playing back. Users can Record or Playback Raw SONET/SDH frames, ATM cells or PoS packets.

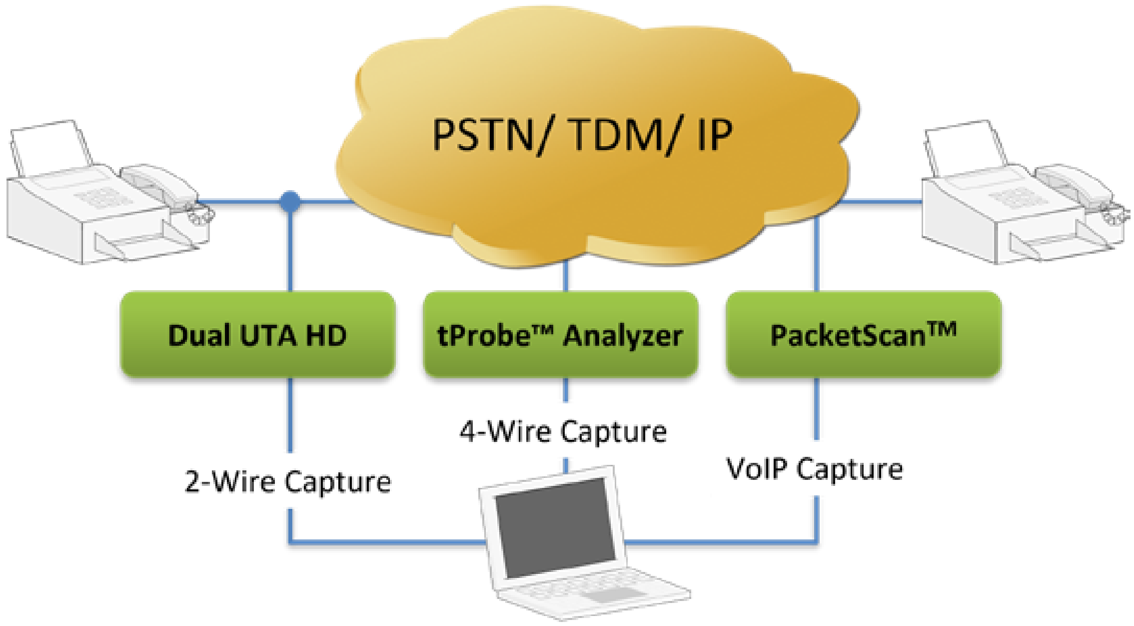
In addition to record and Playback on OC-3 / STM-1, OC-12 / STM-4 interfaces, the captured traffic can also be analyzed in real time. ATM Analyzer is used to analyze and decode different ATM protocols like ATM, AAL2 Protocols (CPS-SDU, SSSAR-SDU, and SSCS), AAL5 (CPCS), UNI and others across U plane and C plane of UNI and NNI interface. The analyzer can also decode ATM frames constituting Classical IP over ATM, or CIP based networks, and traditional SS7 Stack (ISUP, SCCP, MAP, CAMEL (CAP) etc) over ATM.

The PoS Analyzer captures a host of PoS protocols exchanged between the two nodes over SONET & provides useful analysis, which includes distribution of protocols, protocol fields, frame lengths, and frame status. PoS is a highly scalable layer 2 protocol that uses PPP encapsulation to carry IP packets directly in SONET/SDH networks. Currently it carries a majority of the Internet traffic because it can make efficient use of existing SONET infrastructure. Raw SONET/ SDH analyzer analyzes the RAW captured frames and decodes in real time.

GL's SonetExpert™ Unchannelized web application includes the Scan feature which scans the SONET/SDH interfaces starting with the highest speed interface down to the slowest one (OC-192/STM-64, OC-48/STM-16, OC-12/STM-4, OC-3/STM-1 and STS-1/STM-1e), automatically detects the traffic structure and reports the structure in an easy to view graphical format.

The software provides a web-based interface and can be accessed using any standard web browser.

FaxScan™ - Fax Analysis over IP, TDM, and PSTN



FaxScan™
Fax T30 and T38 Capture, Decode and Analysis

Overview

GL's FaxScan™ can generate fax sessions on both sides or only one side, supporting both T.30 and T.38 fax protocols. Additionally, GL offers the capability to analyze fax sessions, allowing users to monitor and assess the performance and quality of fax transmissions.

Call-center quality engineers, brokerages, government agencies, and other entities have a need to monitor fax transactions. Monitoring is done by recording the analog or IP traffic using suitable call capture applications. These files need to be decoded to investigate issues with protocol messages and Fax image quality.

FaxScan™ application is used to process 2-Wire and 4-Wire voice band capture files as well as Win PCAP captures to provide analysis of the T.38 packets, T.30 frames, a Fax TIF image decode, and general call-flow indicators for detail analysis. It is a valuable T.30 and T.38 debug and test tool, aiding significantly in system development. Fax sessions can contain standard G3 or V.34-based sessions.

FaxScan™ can work with popular packet capture applications such as GL's PacketScan™ or Wireshark®, as well as TDM / 2-Wire capture applications such as GL's tProbe™ T1 E1 FXO FXS Analyzer, and VQuad™ Dual UTA. FaxScan™ is also available as an integrated analysis module within Voiceband Analyzer.

- Generates Fax sessions (one side or both sides of the call) while also analyzing the fax sessions
- Process up to V.34 T.30 recordings in 2wire, 4 wire, μ-Law, A-Law, 16 bit, 14 bit, and 13 bit PCM captures (requires VBA038 License)
- Process V.34 T.38 IP captures and SIP/RTP PCAP captures. Win PCAP captures can be processed from T.38 packets alone or as part of a capture file with multiple SIP calls

Key Features

- 3 modes of operation depending on the type of input file: PCM, SIP, and T.38
- Supports 2-Wire or 4-Wire PCM captures that are sampled at 8-Khz only
  - G.711 A-Law, µ-Law encoded 8-bit PCM data formats
  - 13-bit linear PCM, and16-bit linear PCM data formats
- Supports Modems: V.8, V.17, V.21, V.27, V.29, V.33, and V.34
- Output contains page-by-page packet statistics and Fax image summary
- Fax image output in TIFF-F format (as specified in RFC 2301)
- Creates detailed SIP Ladder diagram files for SIP calls
- Reports Modem Rate, Resolution, Encoding, and Page Size
  - 2400, 4800, 7200, 9600, 12000, 14400, 16800, 19200, 21600, 24000, 26400, 31200, or 33600 bits per second
  - HIGH (204x196dpi), LOW (204x98dpi), or SUPER_HIGH (204x391dpi).
  - Modified Huffman (MH), Modified READ (MR), or Modified-Modified READ (MMR).
  - A4, B4, or A3
- Single and multi-page ECM and non-ECM fax sessions are supported
- Integrated with GL's Voice Band Analysis product for Automated Operation

Supported File Formats

- Analog Inputs
  - G.711 A-Law encoded 8-bit samples
  - G.711 µ-Law encoded 8-bit samples
  - 16-bit linear samples that utilize only the low 13 bits. The upper 3 bits are sign extended
  - 16-bit linear samples utilizing all 16-bits
- IP Inputs
  - PCAP files with SIP, RTP, T.38 packets captured on Windows® OS and the Ethernet interface
- Fax Image Output
  - Class-F TIFF format as specified in RFC 2301

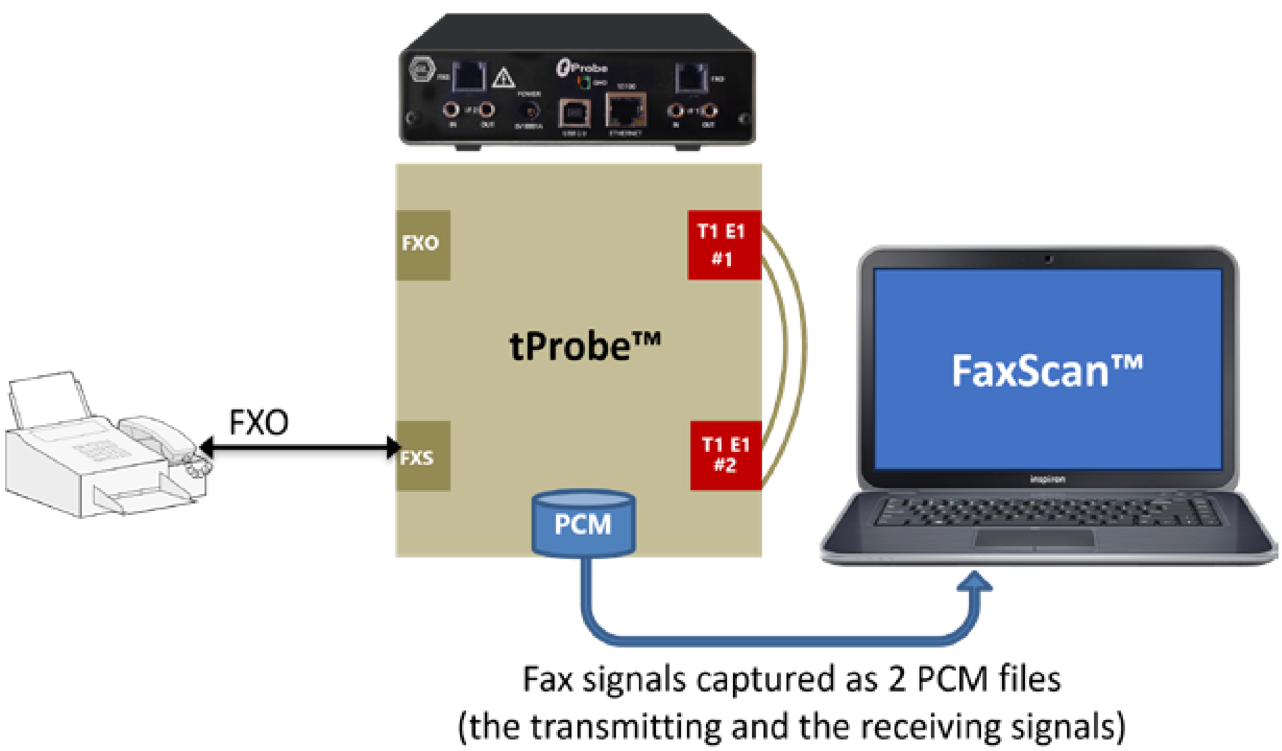
FaxScan™ for 2-Wire and 4-Wire Captures (PCM)

FaxScan™ processes two synchronized audio recordings captured using tools such as GL's Fax Simulator and MAPS™ FXO FXS applications. A fax machine can be connected to tProbe™ in the monitoring mode using RJ-11 splitter. In place of a fax machine a fax call can also be achieved using MAPS™ FXO FXS application or Fax Simulator.

The FaxScan™ reports for PCM files take the form of a ladder diagram. The ladder listing is used to print the events in the list in three time-ordered columns, calling terminal, neither, and called terminal respectively.

Some of the parameters summarized are:

Bad Lines : The number of bad lines received.
Total Lines : The total number of lines.
Pages : The number of pages processed.
Bytes : The number of bytes processed by the high-speed (non V.21) modem.
Trains : The number of training signals processed.
Sender ID : The sending fax machine identification number
Receiver ID : The receiving fax machine identification number

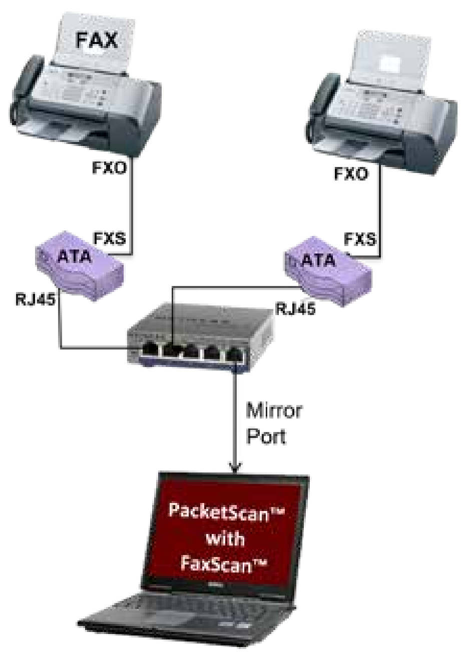


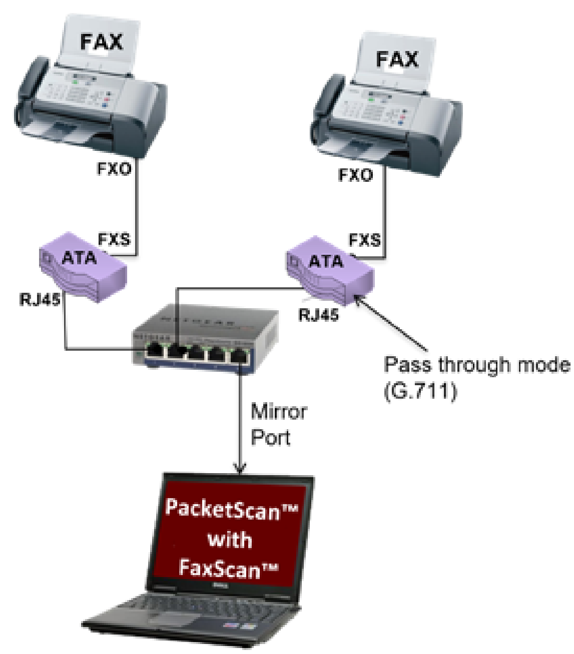
FaxScan™ for Fax over IP (SIP/RTP, SIP/T.38)

FaxScan™ supports T.38 and pass-through modes for capture and analysis of Fax over IP. Calls are captured using GL's PacketScan™ - All IP analyzer via port mirroring on a Ethernet switch. The PacketScan™ monitors, decodes, and records the captured sessions as a single PCAP file and then these captures are fed to FaxScan™ software for Fax decode and analysis.

The following features are supported

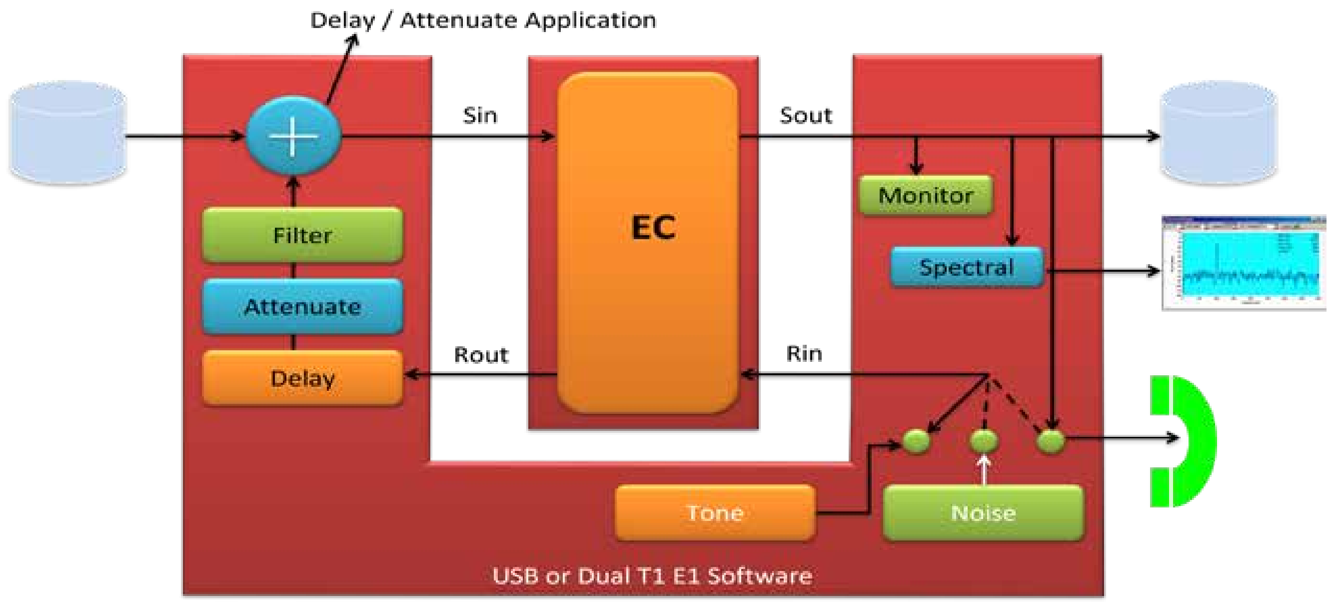
- Analysis of files captured by popular IP capture tools in Transparent (pass-through mode) and T.38 mode
- Single- and multi-page ECM and non-ECM fax sessions are supported
- Decoding of transmitter-only captures is supported for non-ECM faxes
- ECM faxes must have both transmitter and receiver packets present in the capture
- Output contains page-by-page packet statistics and fax image summary
- Fax image output in TIFF-F format
- Generates a SIP ladder file with a summary of the fax call flow





Echo Canceller Testing Solutions

Echo Canceller Test Solutions over TDM Network



Overview

GL's T1 E1 Echo Canceller Test Suite includes optional licensed applications including Echo Path Delay/ Loss Measurement, Echo Path Delay/Loss Simulation, and Echo Canceller Simulator with graphical echo path representation. The Measure Loop Delay/ERL application measures and displays loop delay and echo return loss (ERL) on one or more time slots.

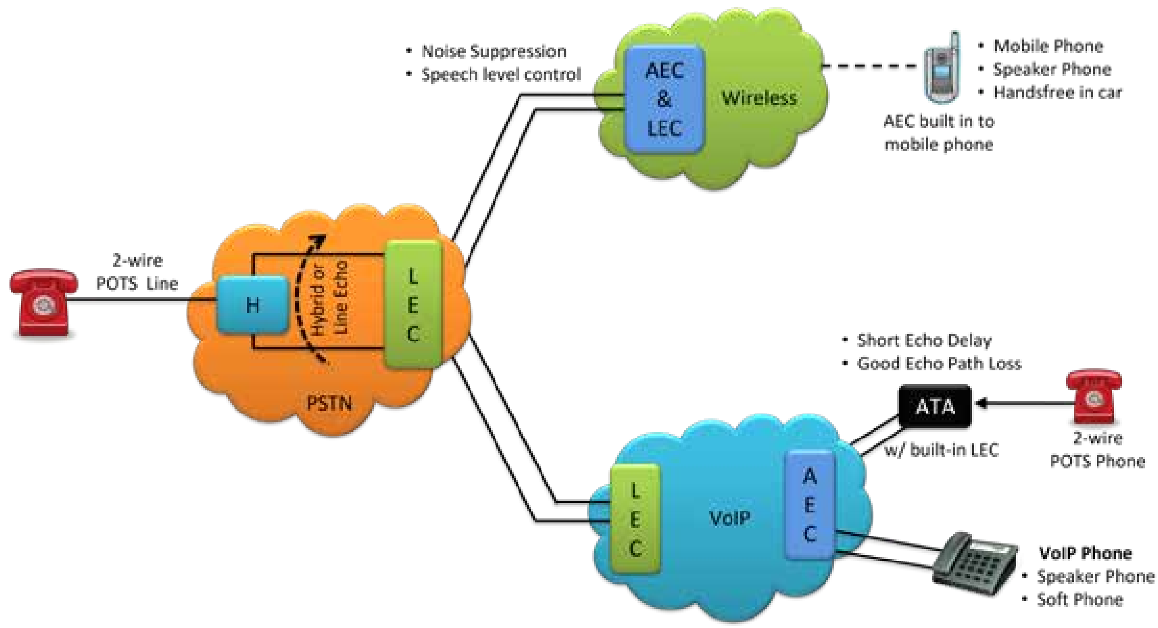
The Delay/Attenuate Timeslots application lets you delay, attenuate, or amplify, and/or apply a filter to a received signal on any number of timeslots. The Delay/ Attenuate Timeslots – Single Channel application supports short delay echo path modeling. This application also allows you to apply delay, attenuation, or gain, and/or digital filtration to a received signal on a single timeslot.

The Digital Echo Canceller (DEC) is a four-port device that supports bi-directional voice traffic between the two ends of a connection. The GLC View application is a waveform viewer application. It has been designed specifically as a companion component for GL's T1 E1 Echo Canceller Test Suite.

Key Features

- Multiple timeslots and multiple measurement strategies
- User-specified minimum and maximum delays expected in the echo path
- Inputs from T1 E1 timeslots, Gaussian noise generator, and A-Law/µ-Law files
- Single and Multi-channel versions available
- Single-channel module supports very short delays. Multiple instances may be run simultaneously
- Noise and double-talk may be injected from noise generator of signal files
- G.168 Echo Path models provided
- Supports real-time and offline processing
- Interfaces directly with A-Law or µ-Law encoded signals
- 16, 32, 64 or 128 ms tail length; programmable tail offset
- Comfort noise generator with adaptation to background noise level
- Continuous reporting of echo path delay, ERL, and dispersion
- Synchronized viewing of waveform and power graph
- Programmable power window
- Zoom-in and zoom-out capability

Echo Canceller Test Solutions over VoIP Network



Overview

In VoIP networks - gateways and ATAs usually contain echo cancellers (ECs) to cancel the echo generated by the landline 2-Wire/4-Wire hybrids. To effectively test ECs in such elements, access to 2-Wire, T1 E1, and IP sides of these elements are necessary. GL's test tools provide access to all these interfaces for performance testing and G.168 compliance testing of ECs in such VoIP network elements. Various solutions and configurations are described below.

GL's ITU-T Specification G.168 EC Compliance Test Suite is developed for testing Echo Cancellers (EC) that reside within a VoIP (Voice over Internet Protocol) and TDM (Time Division Multiplex) environments.

GL's RTP ToolBox™ is used to provide a VoIP test interface creating RTP streams to send and record test files. The application includes the ability to send different types of traffic including voice files, digits, tones, RTP events, and so on. For inter-working with TDM networks, RTP Toolbox™ WWWcan be used with GL's T1E1 analyzer. In addition, RTP Toolbox™ includes client-server command-line modules for automation and GLC View application for graphical analysis.
GL's Voice Band Analyzer (VBA) is an analysis tool developed for monitoring the quality of voice band Traffic for Voice Quality Analysis over VoIP, TDM and Wireless Networks.

Key Features

- Performance testing of ECs in ATAs and Gateways
- G.168 compliance testing of ECs in ATAs and Gateways
- Access to the IP interface with RTP Toolbox™
- Access to the T1 E1 interface with GL's T1 and E1 Cards
- Ability to simulate in real time - echo, delay, attenuation, dispersion, and more
- Ability to measure and verify compliance to G.168
- Manual, semi-automated, and fully automated test configurations
- Monitor voice band traffic for voice quality using VBA

Telecom & Information Technology Consulting Services

GL's expertise covers infrastructure design, installation and inspection as well as application configuration and management. Our team includes engineers, developers, scientists, and project managers. Our customer base includes large internet and wireless service providers, equipment manufacturers, government contractors, research laboratories and universities world-wide.

GL's vast skill set includes the following:

- Custom hardware development or board modification and validation
- Copper and Fiber cabling: Design, installation, maintenance, inspection, testing, validation
- Switches and Routers: Installation, configuration, design, and load testing
- Voice over IP: Installation, testing, monitoring, configuration
- Microsoft Windows®: OS installation, configuration, Windows Server and Active Directory management, file sharing, patching and updates, Windows Secure Host Baseline
- Linux®: Debian application development, installation, configuration
- Application Development: C++, JavaScript, Python, Visual Basic, Visual Studio, Android, iOS, TCL, Fortran, Database Management including Maximo
- PC hardware setup and assembly, printers, Keyboard Video Mouse systems, remote access management, uninterruptible power supplies, server racks
- Network Setup: Virtual Private Networks, network segmentation, Network Address Translation, IPv4 and IPv6, Virtual Local Area Networks, wireless networks, GigE and large infrastructure fiber-based Wide Area Networks with ruggedized networking equipment for harsh environments such as underground railroads, railyards and bus depots
- Wireless Testing: Signal strength, voice quality, drive testing, in-building testing as well as cellular coverage comparison and verification
- Digital Displays, Passenger Information Display System, Customer Information Systems using LED Signs
- Physical security: Access Control, Card readers, Video Management System, CCTV design, testing, and inspection
- Company specific hardware: CISCO®, Juniper®, Oracle®, Microsoft®, Polycom®, Digium®, Dell® and more

GL is a certified small business and a minority owned business and can help large prime contractors satisfy their SBE and MBE requirements in the USA.

GL's services can be utilized at a daily or even hourly rate, or on a project basis. GL can often incorporate its own products into its services (often free of charge).

GL's cost-effectiveness, flexibility, and unmatched expertise is why so many customers chose GL to solve their toughest telecom and IT challenges.

For more information, please visit [Consulting Services](#) Webpage.

Training and Support

Product Training

Customized Training Programs

GL Communications offers tailored training programs to address the unique requirements of our clients. Whether you are a new user seeking basic training or an experienced professional looking for advanced insights, our programs cater to all skill levels.

Hands-on Learning

Our training sessions emphasize practical, hands-on learning experiences. Participants work directly with our products, gaining valuable insights into their functionalities and applications.

Comprehensive Curriculum

The training curriculum covers a wide range of topics, including product features, configuration, troubleshooting, and best practices. Participants acquire a thorough understanding of the capabilities of our solutions, enabling them to make the most of our products in real-world scenarios.

Flexible Training Options

Recognizing the varied schedules and learning preferences of our clients, GL Communications offers flexible training options. Whether through on-site training sessions or web conferences, users can choose the format that best suits their needs.

Product Support

Dedicated Support Team

GL Communications boasts a highly skilled and responsive support team dedicated to assisting customers with any queries or issues they may encounter. Our support staff is well-versed in the intricacies of our products and strives to provide timely and effective solutions. GL offers support across multiple time zones, from our Headquarters in Maryland, USA to our development office in Bangalore, India.

Multi-tiered Support Structure

Our support services are organized into multiple tiers to ensure that inquiries are addressed with the appropriate level of expertise. This tiered structure enables us to efficiently resolve issues, whether they are routine queries or complex technical challenges.

Regular Updates and Resources

Customers benefit from regular product updates, documentation, and knowledge base resources. These resources keep users informed about the latest features and enhancements, empowering them to stay current with our evolving product landscape.

Thank You!

We sincerely appreciate your interest in our products and services. Thank you for taking the time to explore our catalog. Please contact us to discuss your requirements. We look forward to hearing from you!

Best regards,
GL Communications Inc.

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