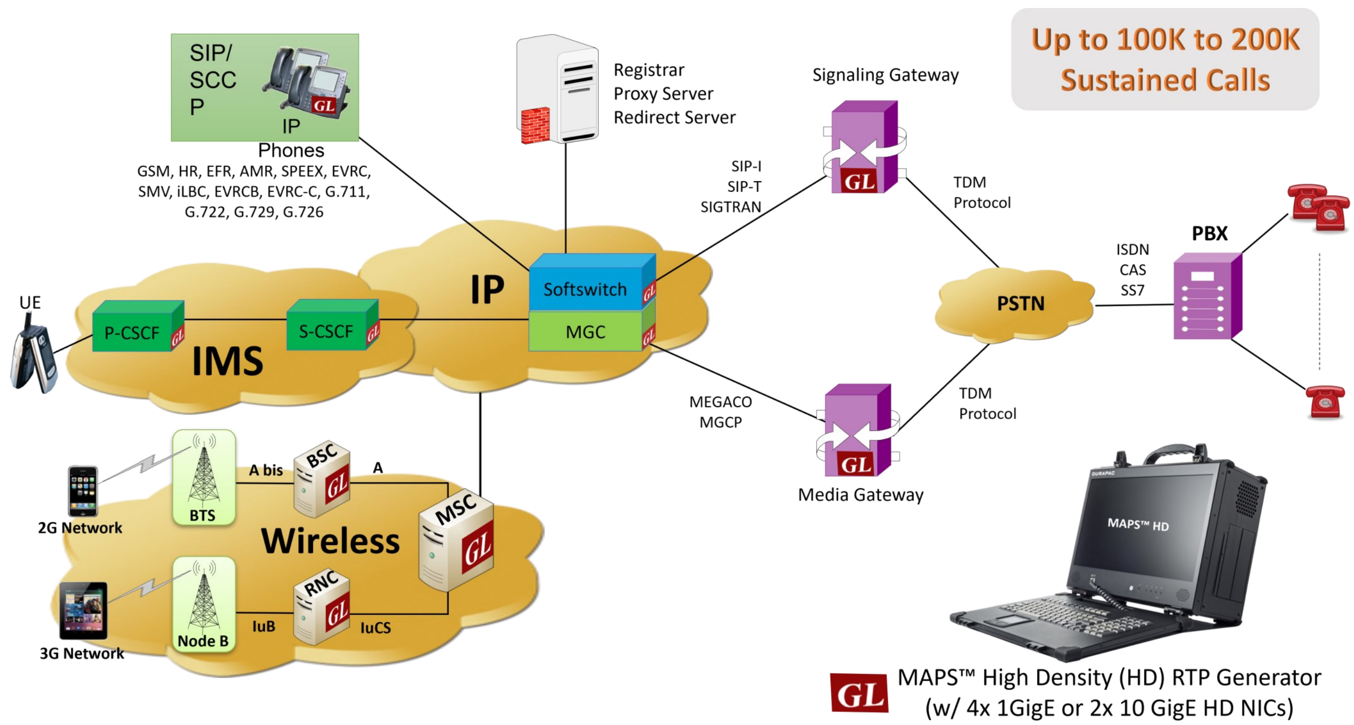
MAPS™ HD - High-Density Call Generator for IP & Wireless Networks



Overview

MAPS™ (Message Automation and Protocol Simulation) is GL's standard framework for emulation of IP, TDM, and Wireless protocols. GL's **MAPS™ HD**, is an advanced bulk call generator used to simulate high volume calls with traffic. It is available as special purpose 2U network appliance w/ 4x1GigE (PKS109) capable of high call intensity (hundreds of calls/sec) and high volume of sustained calls (tens of thousands of simultaneous calls/ platform).

The network appliance performs signaling and traffic generation for a vast array of communication protocols covering IP and Wireless networks. It supports simulation of SIP, IMS SIP UE, GSM A, BICC, MGCP, H.248/Megaco, and provides non-reference based voice quality using E-model (R-factor) and MOS with five mapping scales.

MAPS™ HD network appliance is designed to easily achieve more than 20,000 endpoints per server. Using a stack of multiple servers, a larger test system with 100K-200K calls is achievable for enterprise to carrier grade testing.

With this platform, it is possible to generate Bulk Video Calls using pre-recorded video traces supporting codecs like H.264, H.263 etc. The network appliance provides a modular and flexible solution to generate real voice calls using industry standard voice codecs such as G.711 A/µ-law, G.722, G.722.2 (AMR-WB), G.722.1, G.726, G.729A/B, GSM (EFR, FR and HR), AMR (Narrowband and Wideband), EVRC, EVRCB, EVRC-C, iLBC, Speex, SpeexWB, RFC 2833, and user-defined codecs for voice and tones.

GL also provides PacketScan™ HD for High Density IP Traffic Analysis and Network Monitoring and PacketScan™ FB, a File Based IP Traffic Analyzer Server for near real-time processing of traces.

For more details, refer to High-Density Call Generator for IP & Wireless Networks.

GL Communications Inc.

818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878, U.S.A

(Web) www.gl.com - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) info@gl.com

Features

HD Characteristics

- Transport over UDP and TCP, IPv4 and IPv6, and TLS for secure transport
- Unique endpoint emulation using IP address, MAC address, and VLAN tagging
- With HD RTP (PKS109 licensing) Maximum Simultaneous Calls - 20000, and Calls per Second - 350 (in high end server machine).
- Without RTP (only signaling) Maximum Simultaneous Calls - 70,000, and Calls per Second - 750 (in high end server machine).
- Scales to around 100,000 to 200,000 endpoints with multiple servers.
- Manage 10+ MAPS™ systems from a Master Controller (single point of control)
- Powerful Load Generation, Scheduler, and Command Line Interface (CLI) for automation, remote control, and load testing
- Configurations, test scripts, and profiles can be saved and reused on a different systems
- Easy-to-use MAPS™ Scripting interface (UI) for test scripts to be created
- Real-time monitoring and reporting of registration and call statistics
- Export data to other applications for customized user report
- Create, manage RTP sessions and generate and receive RTP traffic over the sessions with complete automation capability

Traffic Types

- Capability to generate more than 500 simultaneous video calls on a Core i7 systems.
- Simulation of RTP Traffic such as voice file, single tone, dual tones, FAX, Dynamic VF, IVR, and Video Quality over IP networks, and Impairments
- Automate the IVR testing process (call establishment and traffic generation / detection) process through scripts
- All Voice Codecs supported including –
  - G711 µ-law and A-law with VAD
  - GSM-FR, HR, EFR
  - G729A, G729B, G722, G722.1
  - G726 (40K, 32K, 24K, 16K) with VAD
  - AMR (codec rates-4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2, 12.2) with VAD (requires additional licenses)
  - AMR-WB (Codec rates-6.60 kbps, 8.85 kbps, 12.65 kbps, 14.25 kbps, 15.85 kbps, 18.25 kbps, 19.85 kbps, 23.05 kbps, 23.85 kbps) with VAD (requires additional licenses)
  - EVRC, EVRCB, EVRC-C (requires additional licenses)
  - SMV, SPEEX, SPEEX-WB, ILBC
- Simulation of RTP Video Traffic (with H.264, H.263, and VP8 video codecs)
- RTP Voice Quality Measurements – MOS, R-Factor scores
- Simulation of RTP FAX Traffic - G.711 Pass-through and T.38 UDPTL

Test Bed Configuration

The configuration window allows users to setup the required test environment to simulate messaging from different SIP entities such as the User Agent Client (MAPS™) - to - User Agent Server (MAPS™).

MAPS™ HD testbed includes local IP addresses configured to Normal NIC for SIP Signaling, and 4 RTP cores for traffic management between MAPS™ SIP HD and RTP core. Each RTP core uses one port on the GL's HD Interface for sending and receiving RTP traffic.

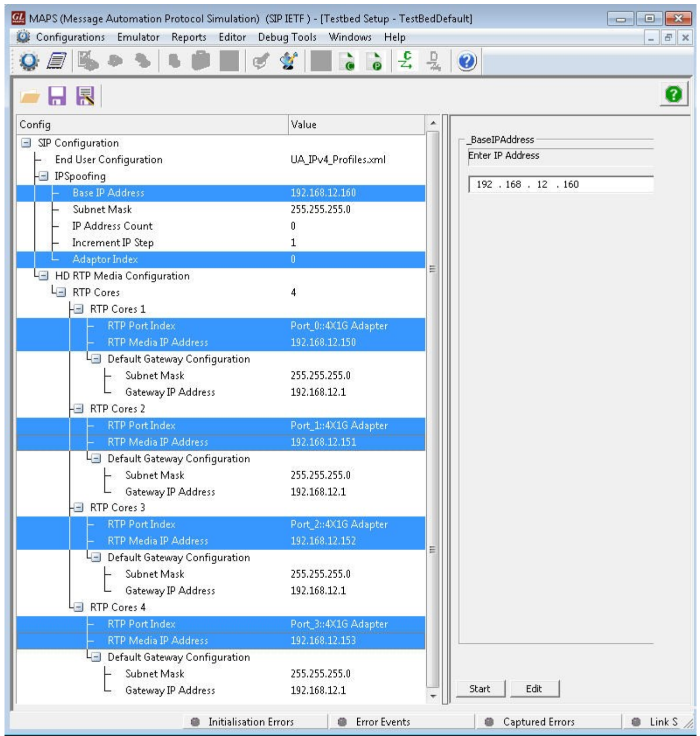


Figure: Testbed Setup Configuration

Bulk Video Call Simulation

[MAPS™ SIP](#) provides the Bulk Video Call Simulation capability using its pre-recorded video traces supporting codecs like H.264, H.263 etc. On a high-performance computing platform (core-i7), it is possible to generate more than 500 simultaneous video calls. With a [High Density (MAPS™ HD)](#) platform, it is possible to achieve much more capacity.

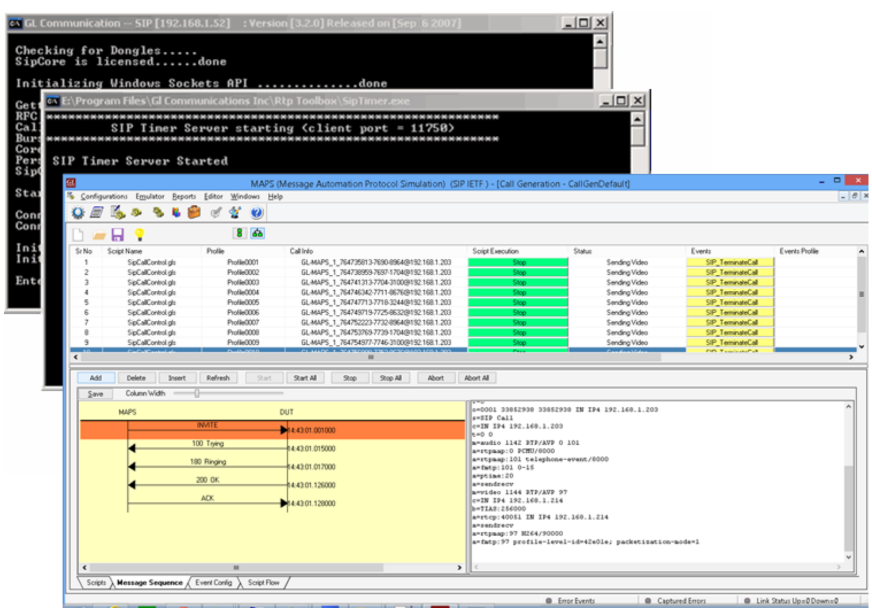


Figure: Bulk Video Call Simulation in MAPS™ SIP

Bulk RTP Voice Simulation

RTP allows to generate and to receive voice traffic over IP networks. Transmit and Receive pre-recorded Voice Files in wave, pcm, and GL's proprietary pre-compressed GLW files with a synchronous Tx/Rx option. You can also directly send live voice using Talk using Microphone feature, and play the recorded voice files directly on to PC speakers. Some additional features that help in the voice traffic simulation are listed below-

- Allows to specify a desired voice payload type to each codec for sending and receiving payload;
- Sampling rate of the codec is displayed for the selected codec.
- Comfort noise generation is supported for A-law, µ-law and G.726 codecs for sending and receiving payload.
- Allows to set the buffer used for delayed packets that arrive at receiving end (both static and dynamic jitter buffers are supported)
- Allows to set QoS (Type of Service) properties such as precedence, delay, throughput and reliability values to the outgoing stream
- Customize codec options (payload type, ptime) over Tx/Rx sessions. All Voice Codecs are supported (Visit Voice Codecs webpage for more comprehensive information)

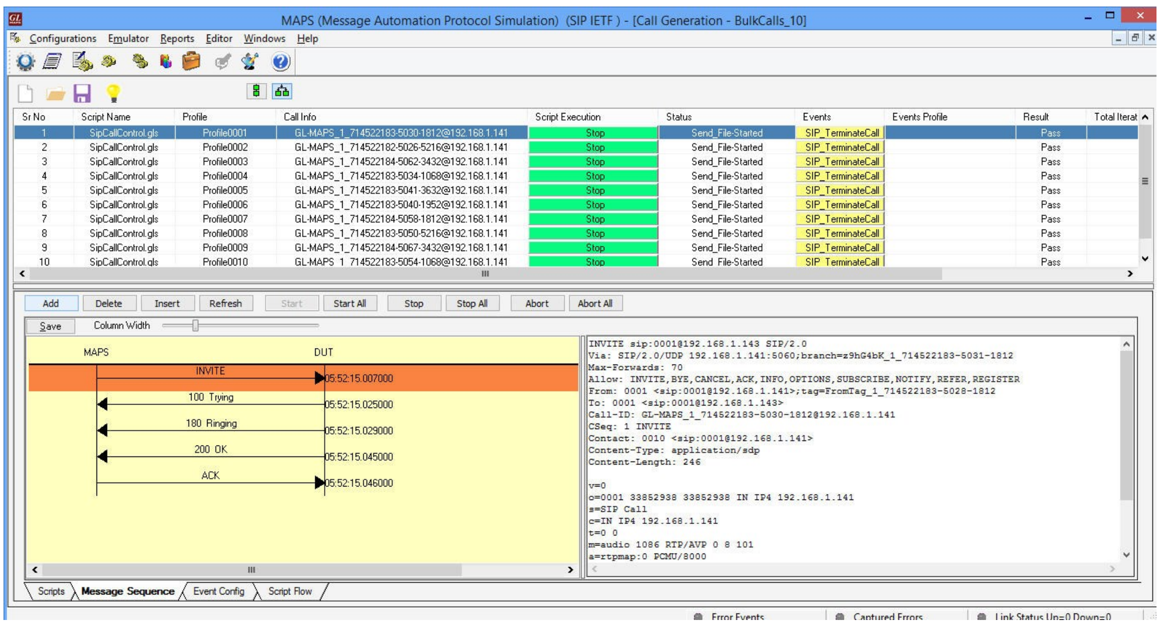


Figure: Bulk RTP Voice Simulation

Digits, and Tones, and Impairments Options in RTP Traffic

- RTP allows to generate and to receive RTP traffic over created sessions (RFC3550). Different RTCP packets are also generated at random intervals carrying control information about the session.
  In-band Digit transmission & detection (DTMF and MF digits)
- Out-band Digit transmission & detection (all events defined as per RFC2833 and RFC4733)
- Tone transmission & detection (single, dual and user defined tones)
- Loopback traffic

Users can introduce various impairments on outgoing RTP streams. The supported impairment types includes Latency, Packet Loss, and Duplicate Packets and Out of sequence Packets.

Load Generation

The Load Generation feature allows bulk call configuration, which includes Total Calls to generate, and Maximum Active Calls, Max/Min Call rate, Start Call Rate, and Maximum Call Rate parameters.

Sip Call Control script which can handle SIP signaling and traffic is loaded along with the profiles configured for each HD port.

MAPS™ SIP uses 4 ports on GL's HD card to generate and receive RTP packets. Maximum supported simultaneous calls on each port is 5000.

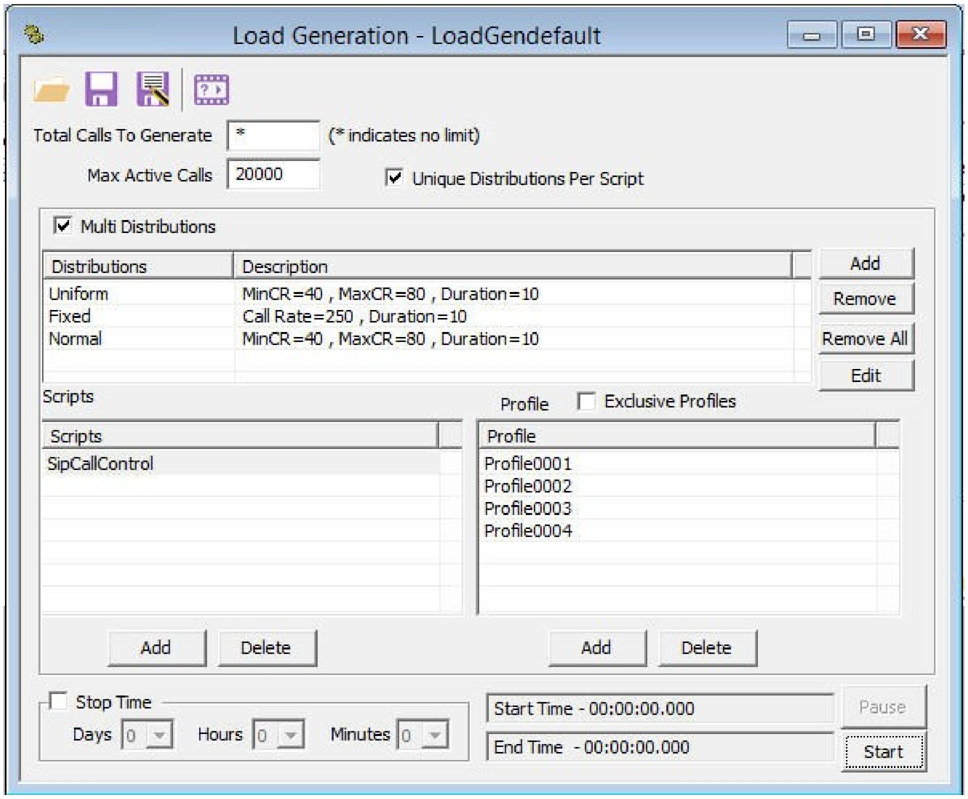
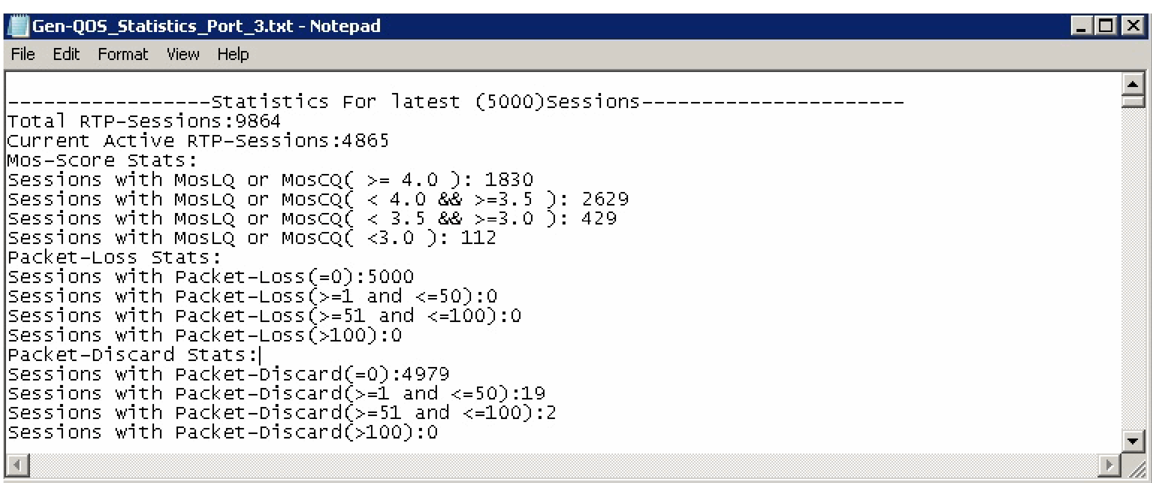


Figure: Load Generation

QoS Statistics

The QoS Statistics for each port is calculated for the received traffic for every specified number of sessions in RTPConfig.ini configuration file.

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-----------------Statistics For latest (5000)Sessions----------------------
Total RTP-Sessions:9864
Current Active RTP-Sessions:4865
Mos-Score Stats:
Sessions with MosLQ or MosCQ( >= 4.0 ): 1830
Sessions with MosLQ or MosCQ( < 4.0 && >=3.5 ): 2629
Sessions with MosLQ or MosCQ( < 3.5 && >=3.0 ): 429
Sessions with MosLQ or MosCQ( <3.0 ): 112
Packet-Loss Stats:
Sessions with Packet-Loss(=0):5000
Sessions with Packet-Loss(>=1 and <=50):0
Sessions with Packet-Loss(>=51 and <=100):0
Sessions with Packet-Loss(>100):0
Packet-Discard Stats:
Sessions with Packet-Discard(=0):4979
Sessions with Packet-Discard(>=1 and <=50):19
Sessions with Packet-Discard(>=51 and <=100):2
Sessions with Packet-Discard(>100):0
```

Figure: QoS Statistics

Statistics

Statistics feature provides both the Call Statistics and Message Statistics with the pie graph indicating the Pass/Fail Call Ratio. By default, all call handling scripts are assessed by MAPS™ to provide statistical information.

The statistical distribution of number of calls per second (time) and total number of simultaneous calls per second is also plotted for the statistics log as per the pattern configured in Load Generation.

Figure: Call and Message Statistics

Buyer's Guide

Item No	Product Description
PKS109	MAPS™ HD (requires GL's high interface NIC)
PKS102	RTP Soft Core for RTP Traffic Generation
PKS120	MAPS™ SIP
PKV100	PacketScan™
PKV120	PacketScan™ HD (4 x 1GigE)
PKV122	PacketScan™ HD (2 x 10 GigE)
PKV121	PacketScan™ FB
PKS106	RTP Video Traffic Generation
PKS103	RTP IuUP Softcore
PKS107	RTP EUROCAE ED137
PKS108	RTP Voice Quality Measurements
PKS200	RTP Pass Through Fax Emulation, requires one of the licenses below, (w/dongle)
PKS202	2 Fax Ports, RO
PKS203	8 Fax Ports, RO
PKS204	30 Fax Ports, RO
PKS205	60 Fax Ports, RO
PKS206	120 Fax Ports, RO
PCD103	AMR codec for MAPS™
PCD104	EVRC codec for MAPS™
PCD105	EVR_B codec for MAPS™
PCD106	EVR_C codec for MAPS™

For more details, refer to Signaling and Traffic Simulator webpage.