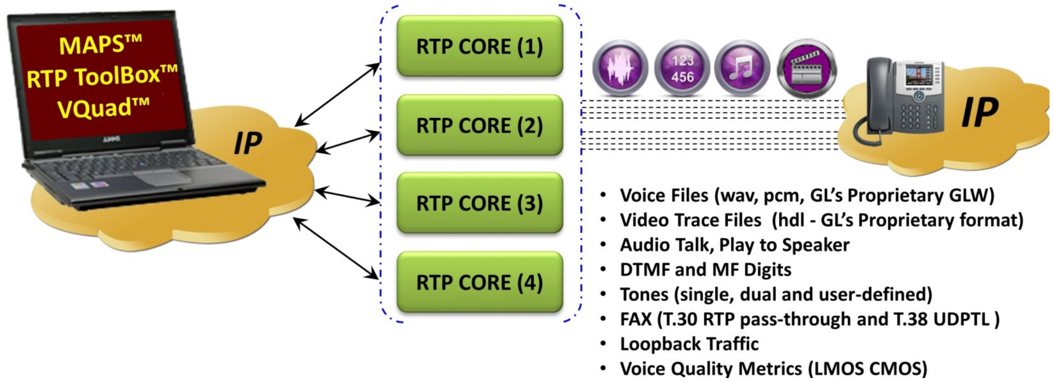
RTP Traffic Simulation



Overview

GL's **RTP Core** allows to create, manage and delete RTP sessions. RTP Core is a stand alone application that acts as an interface between RTP module and the integrating applications such as GL's MAPS™, VQuad™, and RTP ToolBox™. RTP module carries the media streams including real-time Audio, Voice Files, Video, DTMF/MF Digits, Tones, IVR, FAX, Impairments, and Loopback traffic over created sessions. RTCP is used to monitor transmission statistics and Quality of Service (QoS). Different RTCP packets are also generated at random intervals carrying all control information about the session.

GL's RTP Core provides option to run pre-defined RTP Action Scripts automating the traffic actions to be performed on the sessions. One good example of these scripts is IVR applications.

Customize RTP packet transmission over traffic sessions with the **codec type**, sampling rate, voice payload type, RFC 2833 payload type, comfort noise payload type, packetization time (ptime), SSRC, timestamp, and sequence number. On receiving session, users can specify the **jitter buffer** size and the amount of delay. On Tx session, users can also assign **Quality Of Service (QoS)** control values, such as precedence, delay, throughput, and reliability to the stream.

For complete details, refer to RTP Traffic Generator webpage.

Features

- Create, manage RTP sessions and generate and receive RTP traffic over the sessions with complete automation capability
- Transmit and receive pre-recorded video traces with video codecs like H.264, H.263 etc.**
- Transmit and receive pre-recorded voice files, and live voice
- RTP based Voice Quality (MOS and R-Factor) measurement for the received streams
- Customize codec options (payload type, ptime) over Tx/Rx sessions. All Voice Codecs are supported (Visit Voice Codecs webpage for more comprehensive information)
- Talk using Microphone - allows the user to generate live voice
- "Play to Speaker" streams voice to a speaker
- Transmit and receive FAX files in T.38 pass-through mode**
- Loopback real-time voice traffic (all received traffic is retransmitted as sent traffic)
- Generation and Detection of RTP Events per RFC-2833 & RFC-4733 such as Answering Tone, Calling Tone, Special Dial Tone and other Call Progress Tones
- Generation of user-configurable impairments (Latency, Packet Loss & Effects) over established RTP calls
- Supports RTP traffic implementation over Iu-UP (Iu User Plane Interface) layer of the UMTS IuCs Network **
- Supports RTP traffic as per ED-137B of EUROCAE standards used for voice communication in Air Traffic Control networks**
- Provides some vital statistics like total packets received and transmitted, Jitter, delay, lost packets, duplicate packets and out of order packets on each session
- Detailed statistical information of RTP and RTCP packets
- Jitter Buffer implementation for the received traffic to give near real time affect

** Some of these features requires additional licenses – contact GL for more information

GL Communications Inc.

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

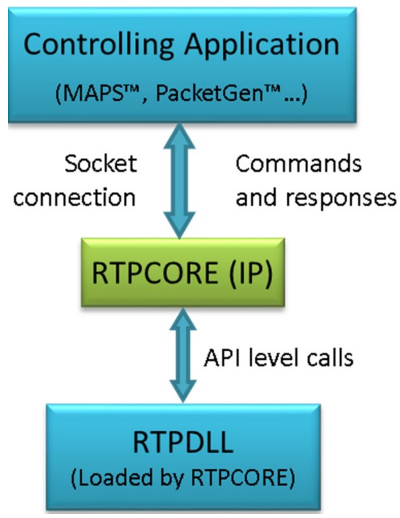
(Web) www.gl.com - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) info@gl.com

Working Principle

RTP Core is a stand alone application that acts as an interface between RTPDll and the application. RTPDll is the RTP server that allows to create, manage, and delete RTP sessions. GL's protocol simulation applications such as VQuad™, RTPToolBox™, MAPS™, and others support RTP Core. Basically applications and RTP Core act as client and server respectively and communicate using socket connection.

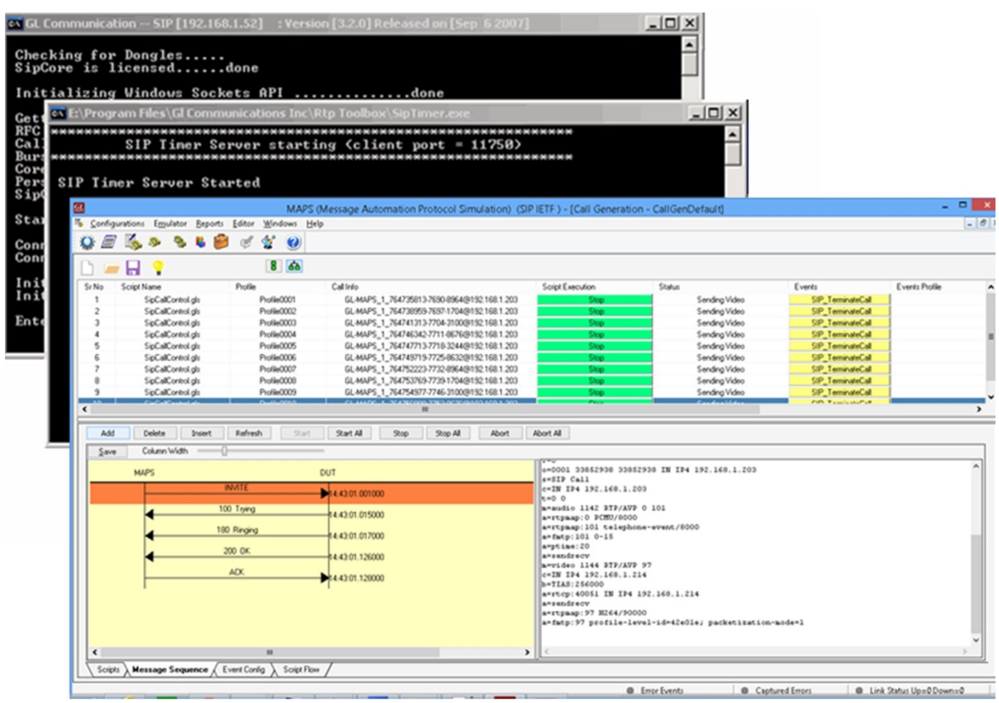
RTP Core interacts between applications (client) and the loaded RTP dll API sending and receiving commands and responses. The diagram depicts operations and usage of RTP Core.

The advantage of the socket communication is that it allows traffic load distribution across different systems running RTP Core and the applications. Each application can connect to multiple RTP Core running on same system or different systems. However, multiple applications cannot connect to a single RTP Core. RTP Core listens on TCP port 30102 for connection from application.



Bulk Video Call Simulation (PKS106)

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.

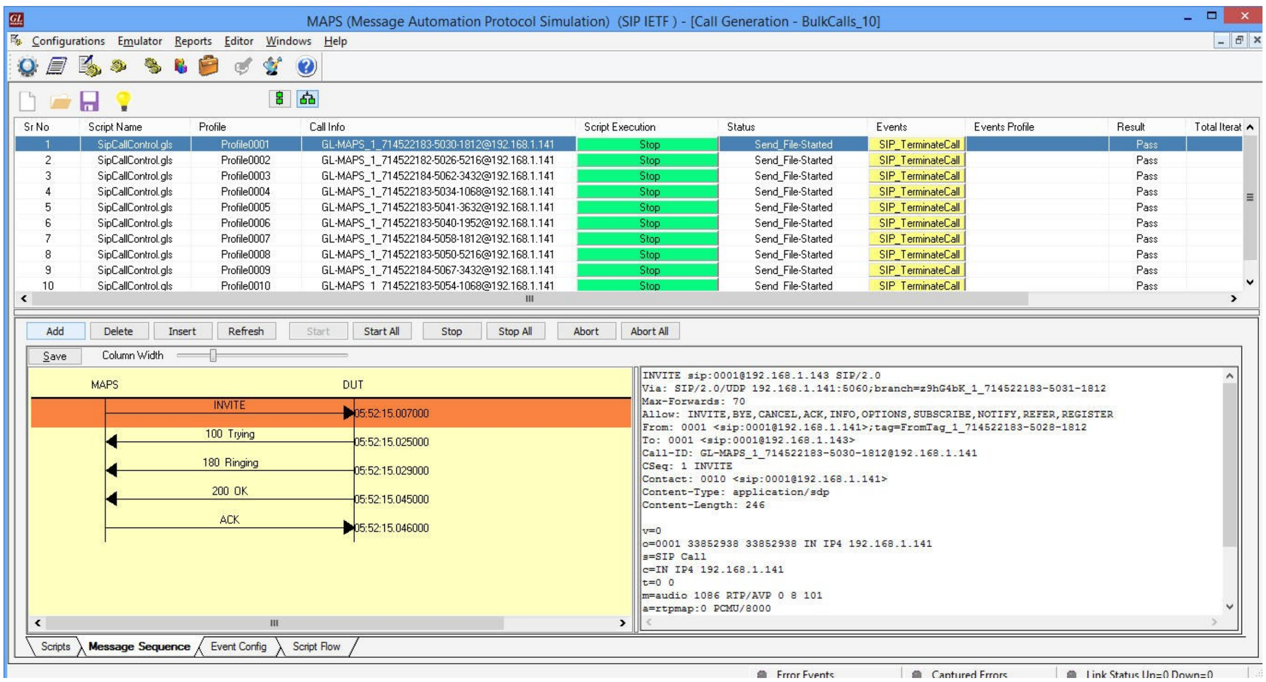


Bulk Video Call Simulation in MAPS™ SIP

Bulk Voice Traffic 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



Bulk voice traffic simulation

Digits, and Tones, and Impairments Options in RTP Traffic

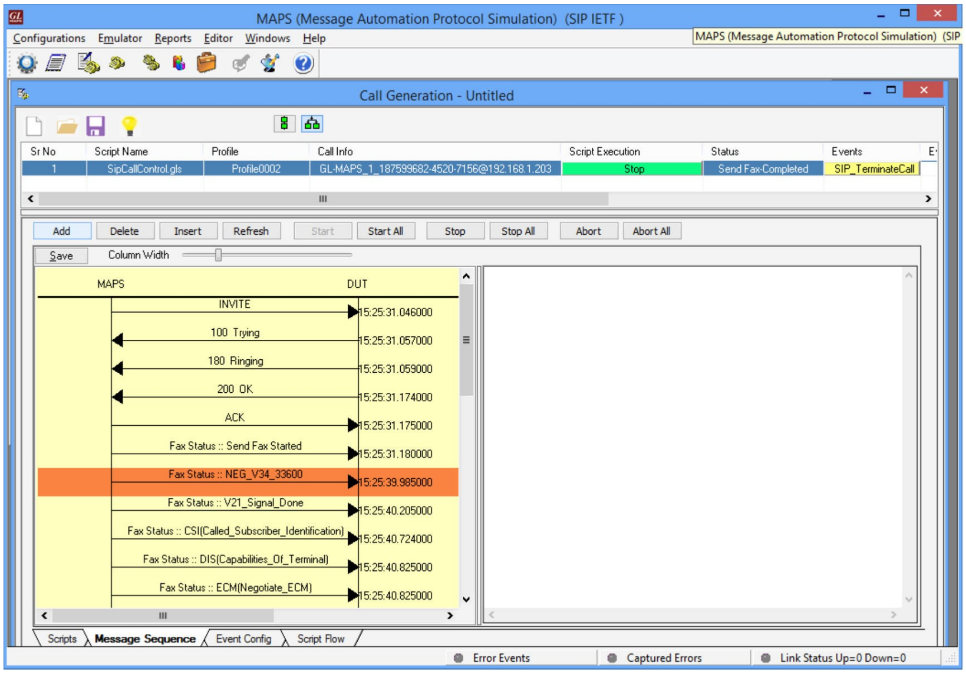
- RTP allows to generate and to receive RTP traffic over created sessions (RFC3550). Different RTCP packets are also generated at random intervals carrying control information about the session
- In-band Digit transmission & detection (DTMF and MF digits)
- Out-band Digit transmission & detection (all 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.

Bulk RTP FAX Simulation (PKS200)

GL's RTP Fax Simulator simulates multiple fax calls over IP in T.38 pass through mode (using G.711 PCMU and PCMA). It can transmit pre -recorded Tiff image to DUT (Real-time Fax machine), receive Pass-Through fax from DUT, and record complete fax call messages as log file along with a Tiff image. Typical applications of our Fax Emulation software are load testing of fax servers, qualification testing of T.38 Gateways, testing of ATAs (Analog Terminal Adapters), testing of fax machines, and many more.

Almost all MAPS™ IP products support fax simulation – MAPS™ SIP, MAPS™ SIP-I, MAPS™ MEGACO, MAPS™ BICC, MAPS™ GSM, and MAPS™ UMTS.

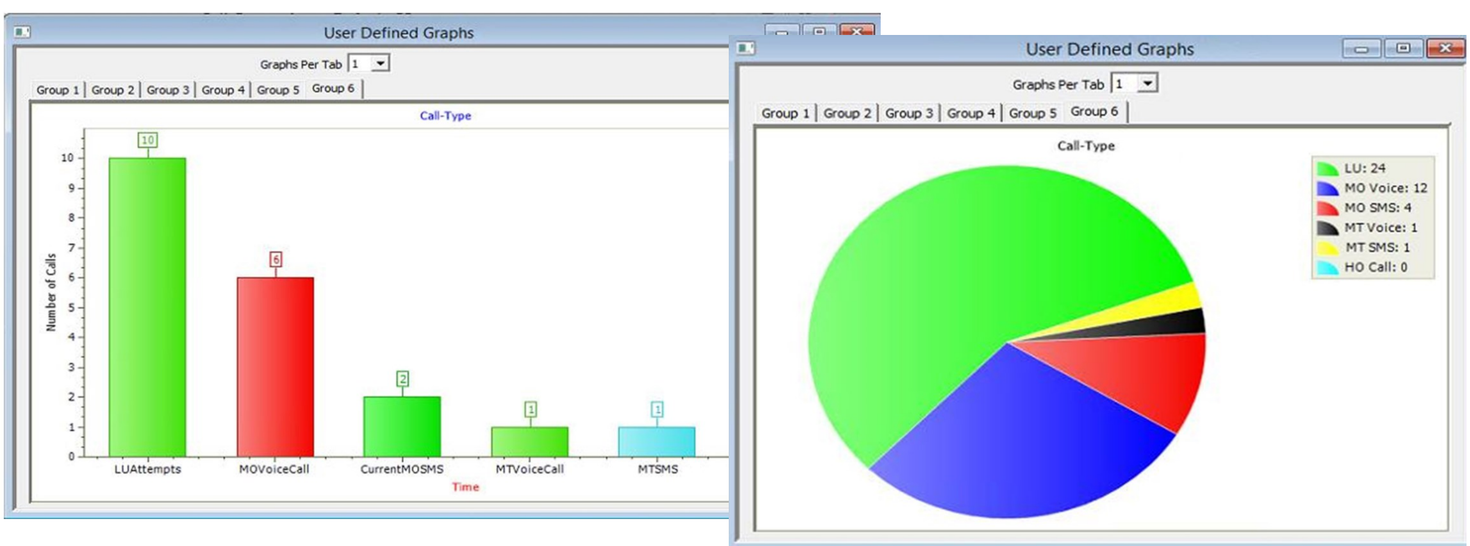


T.30 G.711 Pass through mode FAX simulation using MAPS™ SIP

RTP Voice Quality Measurements (PKS108)

With support of additional licensing (PKS108) RTP voice quality metrics for the received calls are calculated and are reported to MAPS™ application. Quality metrics include R-Factor, Listening and Conversational Quality MOS scores, PacketLoss, Discarded Packets, Out of Sequence Packets and Duplicate Packets.

R Factor graph will display statistics such as, R-Listening, R-Conversational, R-G107 and R-Nom. MOS Factor graph will display statistics such as MOS CQ, MOS PQ and MOS Nom. Estimates are based on the ITU G.107 E Model.



RTP voice quality measurement (PKS108)

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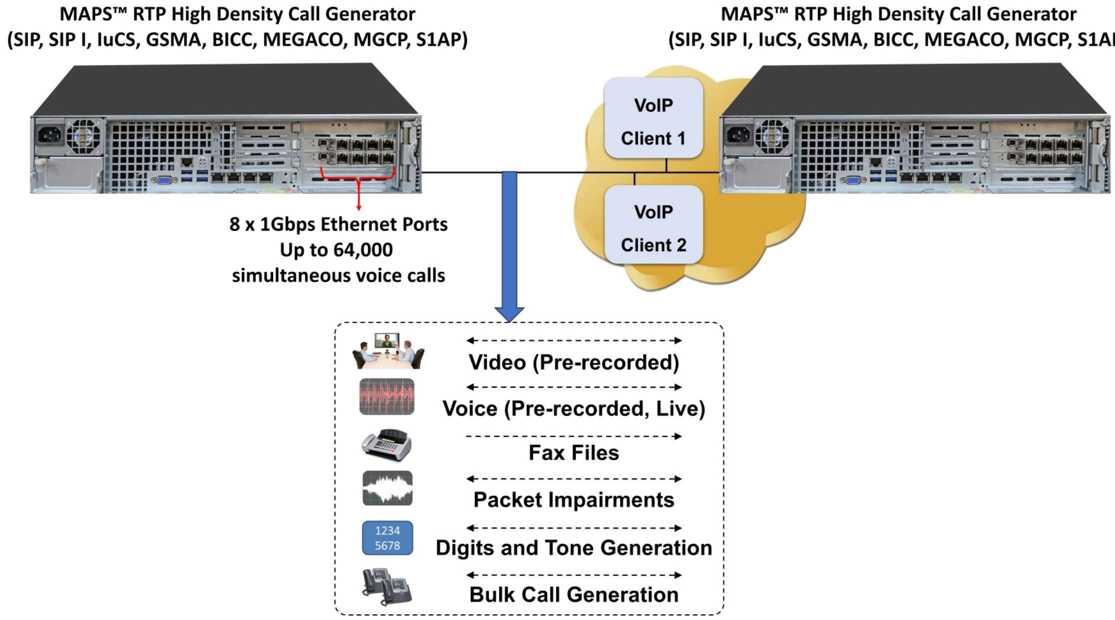
MAPS™ HD RTP (PKS109)

MAPS™ HD RTP is an advanced bulk call generator used to simulate high volume calls with traffic. It is available as special purpose rackmount network appliance with 4x1GigE NIC capable of high call intensity (hundreds of calls/sec) and high volume of sustained calls (tens of thousands of simultaneous calls/platform).

MAPS™ HD network appliance is designed to easily achieve up to 20,000 endpoints per appliance (5000 per port). 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 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. The below figure indicates MAPS™ HD Appliance



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Buyer's Guide

Item No	Product Description
PKS102	RTP Soft Core for RTP Traffic Generation
PKS103	RTP IuUP Softcore
PKS106	RTP Video Traffic Generation
PKS107	RTP EUROCAE ED137
PKS108	RTP Voice Quality Measurements
PKS200	RTP Pass Through Fax Emulation
PKS202	2 Fax ports licenses
PKS203	8 Fax ports licenses
PKS204	30 Fax ports licenses
PKS205	60 Fax ports licenses
PKS206	120 Fax ports licenses

Note: PCs which include GL hardware/software require Intel or AMD processors for compliance.

For more details, refer to RTP Traffic Generator webpage.