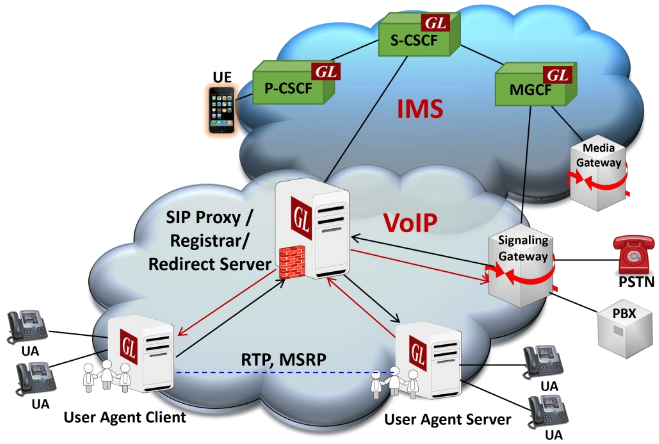
SIP Protocol Simulation - MAPS™ SIP



MAPS™ SIP with RTP Traffic Generation
(2000 simultaneous calls)
MAPS™ SIP Conformance

MAPS™ SIP (w/ 8 x 1Gbps Ethernet Ports)
HD RTP Traffic Generator
64,000 Simultaneous Calls (with RTP Traffic)

Overview

Message Automation & Protocol Simulation (MAPS™) designed for **SIP testing** can simulate SIP entities such as User Agents (User Agent Client- UAC, User Agent Server-UAS), Redirect and Registrant servers. This test tool/traffic generator can be used to simulate any interface in a SIP network and perform protocol conformance testing (SIP protocol implementation).

The application is available as **MAPS™ SIP (PKS120)** and **MAPS™ SIP Conformance (PKS121)**.

The MAPS™ SIP Conformance Suite (PKS121) is designed with 400+ test cases, as per SIP specification of ETSI TS 102 027-2 V4.1.1 (2006-07) standard.

MAPS™ can support transmission and detection of various RTP audio traffic such as, digits, voice file, single tone, dual tones, FAX, Dynamic VF, and Video Quality testing over IP networks, with additional RTP traffic licensing. For more details, refer to RTP Traffic Simulation.

The **RTP Video Traffic Generation** capability is now added to GL's MAPS™ SIP emulator. During bulk Video call simulation, pre-recorded video traces (*.HDL GL's Proprietary format) supporting video codecs like H.264, H.263 and VP8 are transmitted over established sessions. It has the capability of generating more than 500 simultaneous video calls on a core i7 systems.

MAPS™ SIP supports **Message Session Relay Protocol (MSRP) for instant messaging** over SIP sessions (PKS112), conforming to RFC 4975 MSRP protocol specifications. It can be used to simulate SIP/MSRP User Agents in an NG9-1-1 network and send and receive communications over ESInets. The supported call types include IM Only Calls, Audio and IM Calls, and Video and IM Calls between multiple UAs.

MAPS™ SIP supports **FAX over IP (FoIP)** simulation and monitoring. With Additional licensing, both **RTP G.711 Pass Through Fax Simulation (PKS200)** and **T.38 Fax Simulation over UDPTL (PKS211)** simulation are supported.

MAPS™ SIP supports Interactive Voice Response (IVR) testing that recognizes and responds to voice prompts using DTMF digits or voice, allowing automated IVR traversal and testing.

MAPS™ supports Command Line Interface (CLI) allowing remote controlling of the application through multiple command-line based clients.

GL's MAPS™ SIP is also available in High Density version (requires a special purpose 1U network appliance and PKS109 RTP HD licenses). This is capable of high call intensity (hundreds of calls/sec) and high volume of sustained calls (100K – 200K simultaneous calls with scaling).

MAPS™ supports stress and load testing with massive UA generation using CSV based profiles configured with 'n' number of User Agent (UA) unique parameters.

For more information, refer to MAPS™ SIP Protocol Emulator webpage.

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

Main Features

Signaling

- Generates and processes SIP valid and invalid messages
- Supports complete customization of scripts, user agent parameters, SIP headers, SDP headers, call flow, and messages
- Supports IPv4 /IPv6 and transport over UDP, TCP, SSL and TLS (Version 1.0 and 1.2)
- Handles Retransmissions of messages with specific interval
- Scripted call generation and call reception
- Supports joining a conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, and silence packets generation
- Ability to send "reliable provisional responses" and start early media actions
- Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER and INFO SIP methods
- Ability to implement IP Spoofing for any network like Class C, Class B etc.

Traffic

- Supports transmission and detection of various RTP traffic such as, digits, voice file, single tone, dual tones, FAX, Dynamic VF, IVR, and Video Quality over IP networks
- Supports different traffic options across simultaneous calls
- Supports almost all industry standard codec types - G.711 (mu-Law and A-Law), G.722, G.729, G.726, GSM, AMR, EVRC, SMV, iLBC, SPEEX, EVS, OPUS and more. Click here for comprehensive information on supported codecs. *AMR and EVRC variants require additional licenses
- Supports both RTP G.711 Pass Through Fax Simulation (PKS200) and T.38 Fax Simulation over UDPTL (PKS211) simulation over IP
- Impairments can be applied to RTP traffic simulating error conditions that occur in real-time networks
- Bulk Video call generation supported with H.264, H.263, and VP8 video codecs
- Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL)
- Supports simulation of MSRP sessions - IM Only Calls, Audio and IM Calls, and Video and IM Call types
- Supports simulation of short message services (SMS) over IP/IMS network via IP-SM-GW IMS entity

Reports

- Reports call control statistics - number of outgoing and incoming calls, number of SIP registrations, SIP options, SIP message statistics, and MSRP statistics
- Detailed test result reports generation in PDF file format
- Option to send complete test report (traffic information and call events) to a central database, such as Oracle
- Provides voice quality metrics such as Listening MOS, Conversational MOS, Packet loss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter
- Option to calculate and update RTP statistics per call periodically during run-time to a csv file

Bulk Call Capability

- Capable of high call intensity (hundreds of calls/sec) and high volume of sustained calls (100K - 200K simultaneous calls with scaling)
- Enhanced with CSV based profiles feature supporting massive UA simulation (up to 32,000 users)
- **Performance:**
  - Only Signaling Simulation - up to 100,000 active calls, 500 cps **
  - With RTP Traffic - up to 2000 active calls (Voice), 250 cps **
  - With RTP HD - up to 32000 active calls (Voice), 250 CPS (Requires PKS109 + GL's HD NIC)

** Based on System Requirements – Windows® Server, 64 GB RAM, Intel(R) Xeon(R) Silver 4210 CPU @2.20GHz 2.19GHz

Other Features

- Supports Interactive Voice Response (IVR), automate the IVR testing process
- Supported on Windows® 10 , Windows® Server operating systems
- Supports 64-bit version to enhance signaling performance

Main Features *(Contd.)*

CLI

- CLI interface (PKS170) based on a client-server model allows users to control all features of MAPS™ through APIs
- Supported clients are TCL, Java, Python and C# (Linux and Windows)

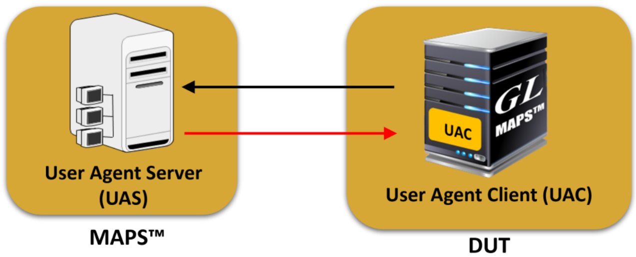
Applications

- Fully integrated, complete test environment for SIP
- Supports testing UAC, UAS, Proxy, Registrars, Registrants, Redirect Servers, Gateways, and other SIP entities
- Handles strict routing & loose routing, when requests are routed through proxies
- Testing NG9-1-1 emergency services and components within the ESInet

Configuration Scenarios

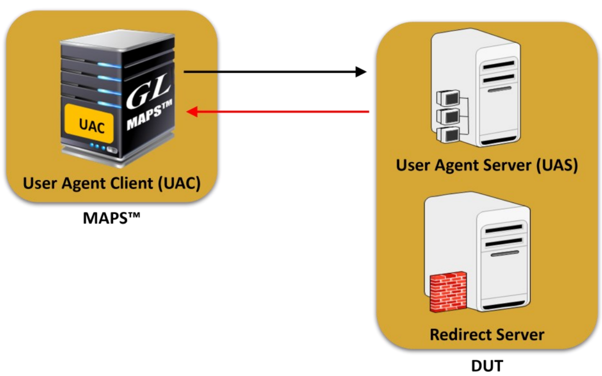
Scenario 1: MAPS™ acting as UAS and testing UAC

MAPS™ acting as UAS receives messages from UAC (DUT) that generates SIP messages.



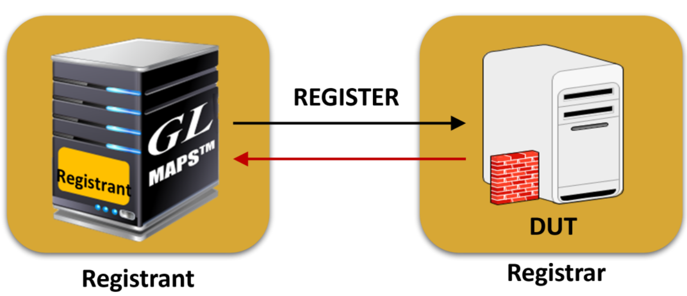
Scenario 2: MAPS™ acting as UAC & testing Redirect Server / UAS

MAPS™ can be configured to act as UAC and to test Redirect Server and/or UAS. This allows the redirection call scenarios to be automated and test DUTs.



Scenario 3: MAPS™ acting as Registrant to test Registrar

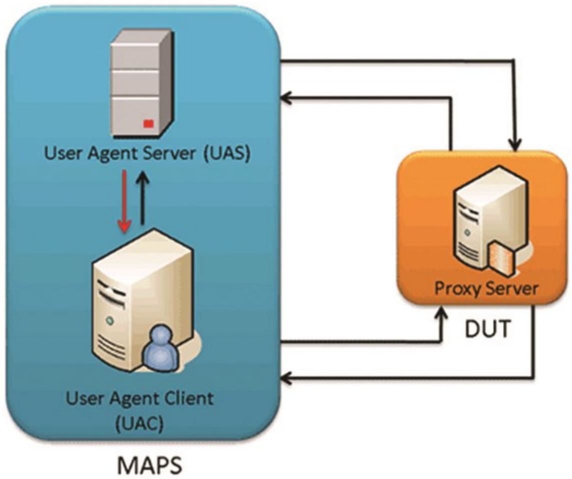
MAPS™ can be configured to act as Registrant and to generate registration request messages to automate the entire Registrar (DUT) testing.



Configuration Scenarios *(Contd.)*

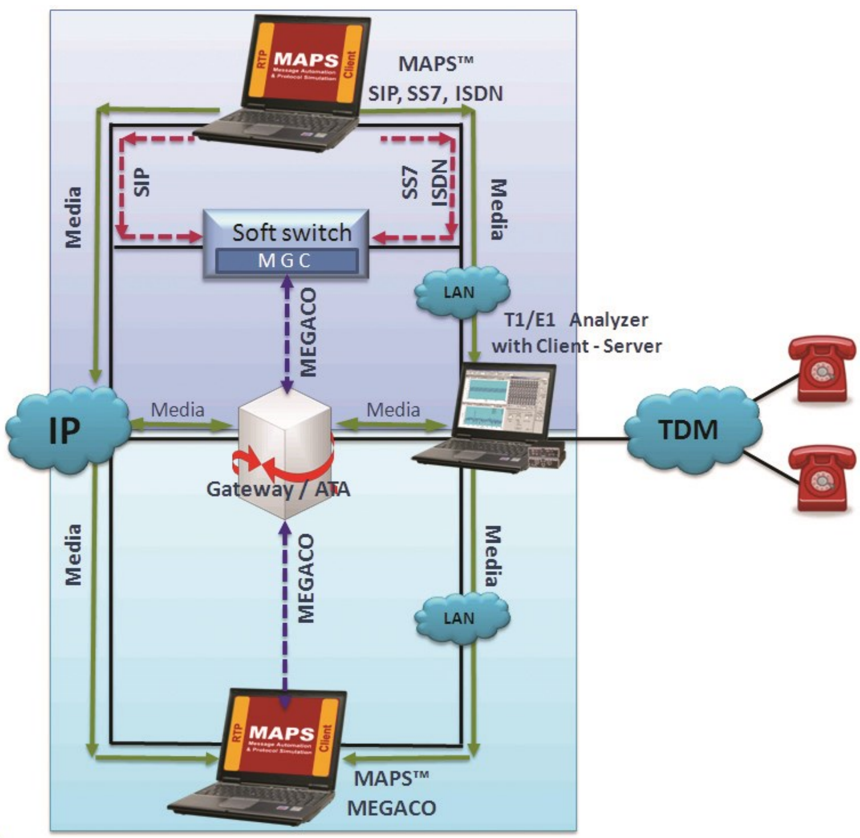
Scenario 4: MAPS™ acting as UAS and UAC to test Proxy Server

MAPS™ can be configured to act as UAC and UAS simultaneously so that entire Proxy testing can be automated.



Scenario 5: End-to-End Gateway Testing

MAPS™ can be used as a tool to evaluate Gateway / ATA product features such as call connectivity, call signaling, traffic generation, voice quality testing, codec, and hundreds of other features. The below image shows End to End Gateway Testing.

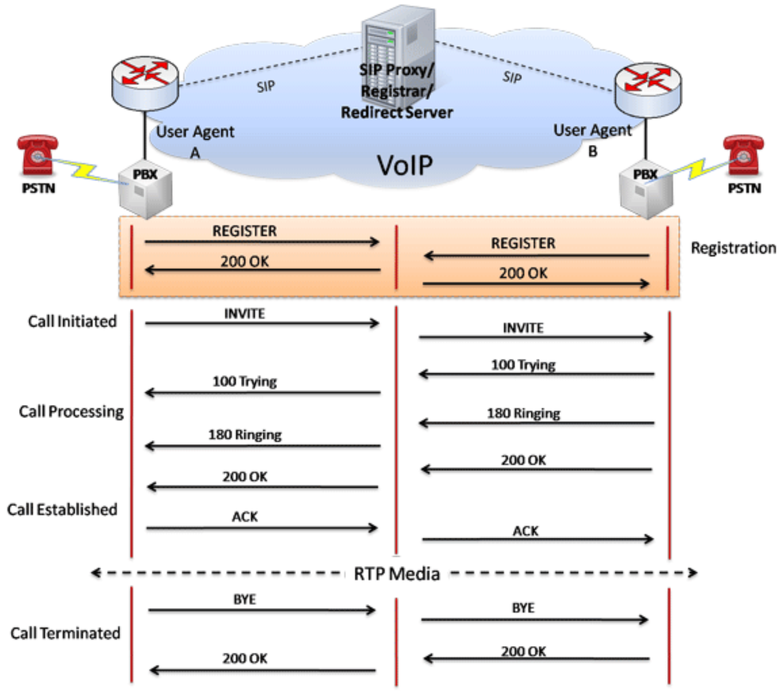


MAPS™ SIP Call Flow Scenarios

SIP Registration and Call Control Procedure

MAPS™ SIP configured as Client (Caller) registers with the Server by sending initial **REGISTER** request message. Registration procedure is completed on receiving **200 OK** reply message.

Once registration process is completed, call control script proceeds with the call establishment and RTP Media flow in both the ways allowing conversation to carry out between the entities.



Pre-processing Tools

Script Editor - The script editor allows the user to create / edit scripts and access protocol fields as variables for the message template parameters. The script uses pre-defined message templates to perform send and receive actions.

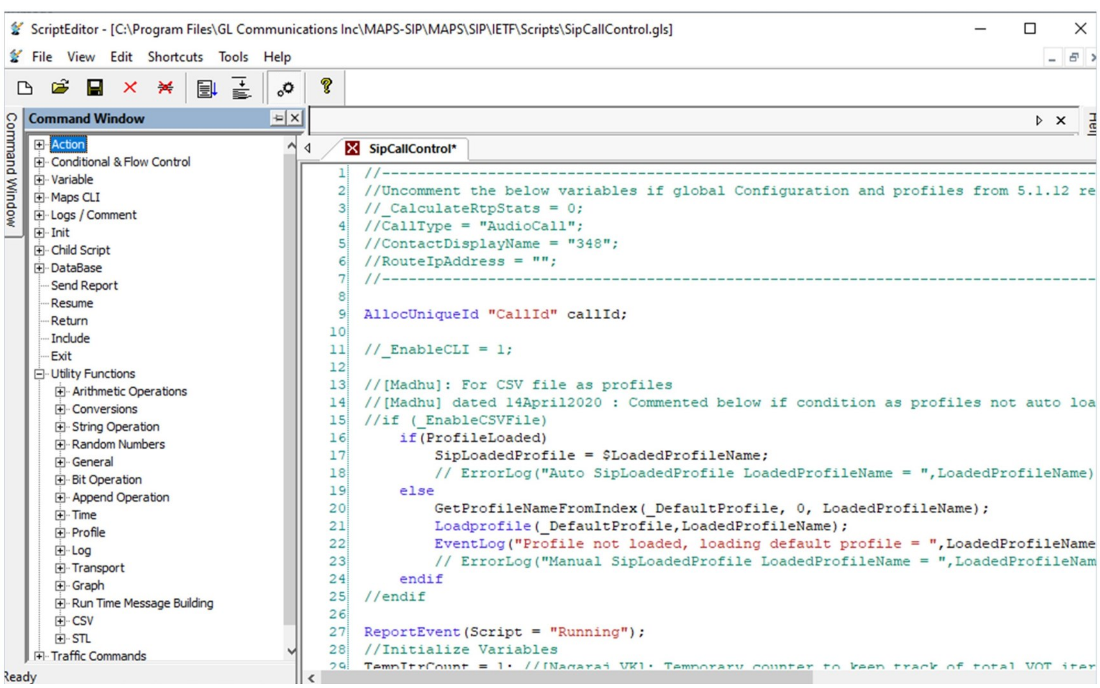


Figure: Script Editor

Pre-processing Tools *(Contd.)*

Profile Editor - This feature allows to edit the values of the variables used in the message templates. During call simulation these values replace the original values of the variables in the message template. Each profile is an XML based file that defines multiple configuration parameter values necessary for simulating calls. Profile Editor allows users to create their own profiles to suit their custom scripts.

For example, traffic profiles includes configurations for defining RTP traffic options- Digits, Voice File, Tones, Fax, IVR, and User-defined traffic. The call type parameter can be set to Video or Audio type to support RTP Video & Voice Traffic Generation. The transport types parameter can be set to UDP, TCP and TLS. Additionally, it also includes MSRP parameters for supporting Instant Messaging (IM).

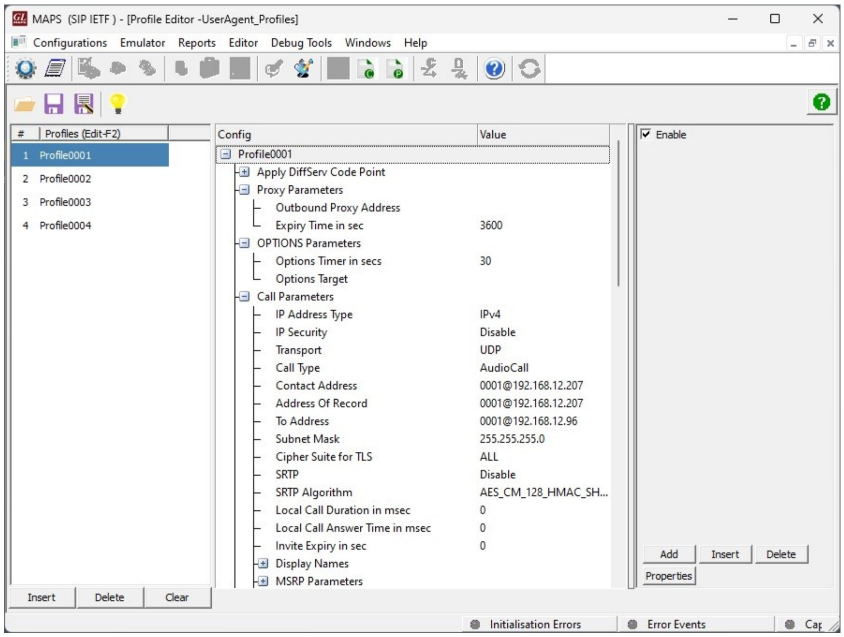


Figure: Profile Editor

Call Generation and Reception

In call generation, MAPS™ is configured for the out going messages, while in call receive mode, it is configured to respond to incoming messages. Tests can be configured to run once, multiple iterations and continuously. Also, allows users to create multiple entries using quick configuration feature.

The editor allows to run the added scripts sequentially (order in which the scripts are added in the window) or randomly (any script from the list of added script as per the call flow requirements).

The test scripts are started manually at call generation; and at the call reception, the script is automatically triggered by incoming messages**.**

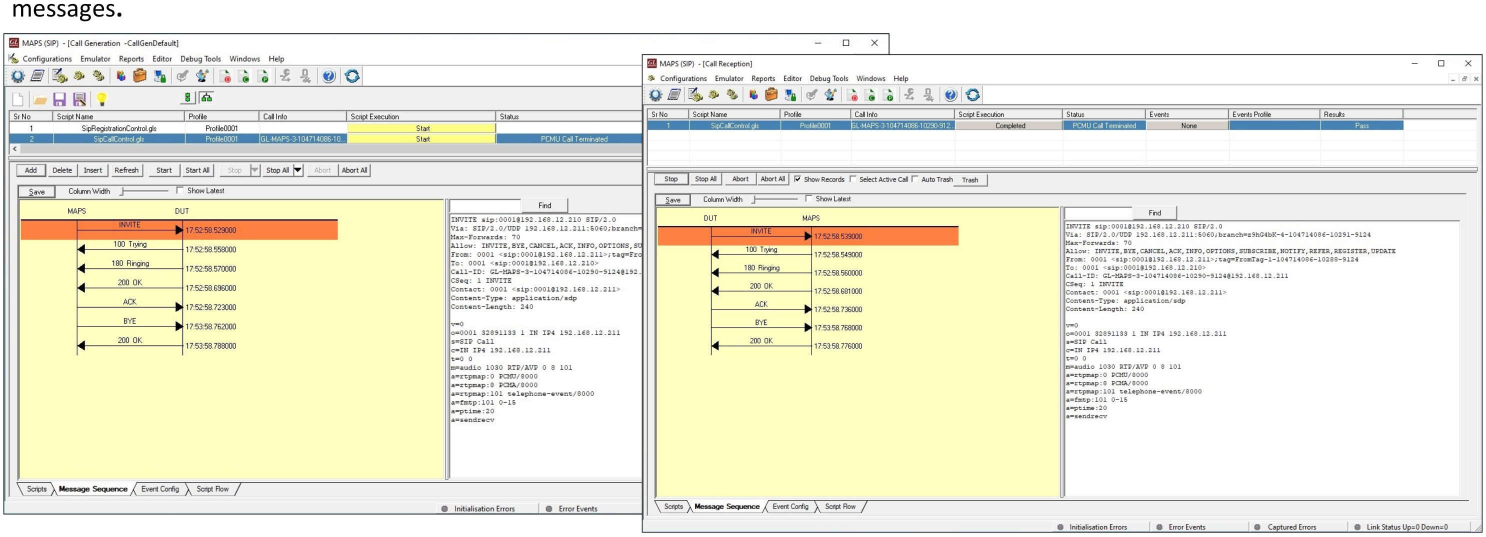
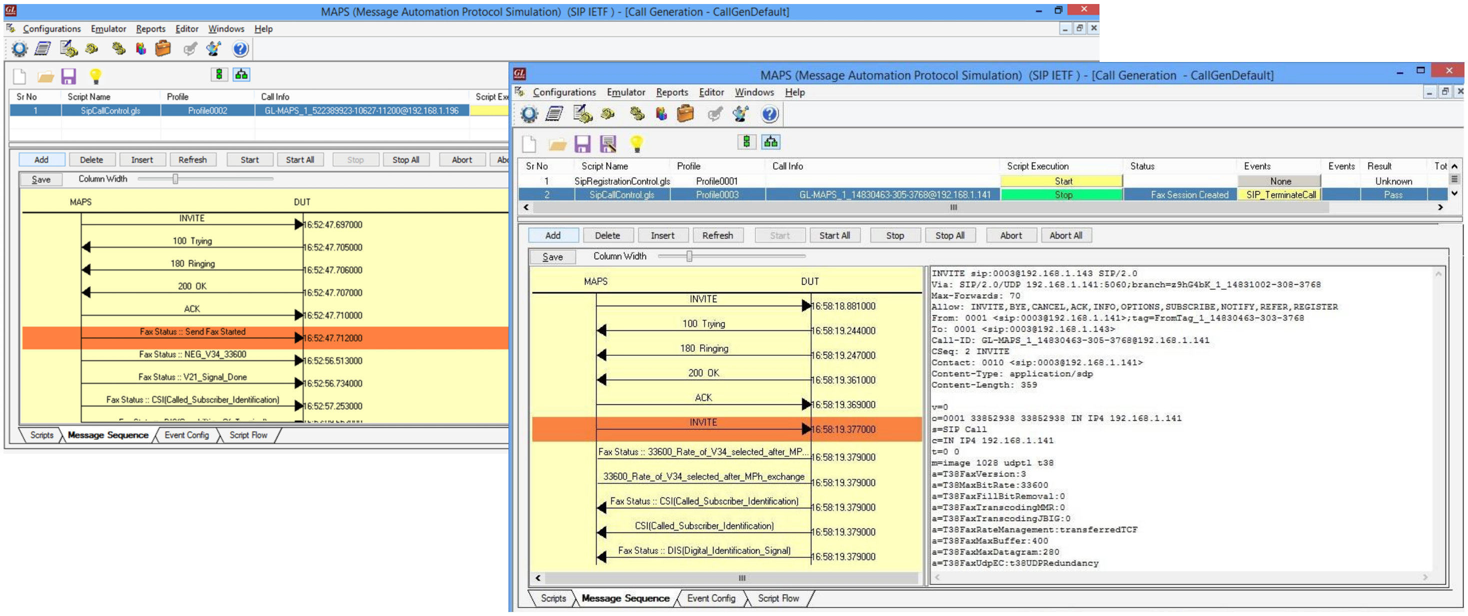


Figure: Call Generation and Reception

Generation of Fax Calls (T.30 and T.38)

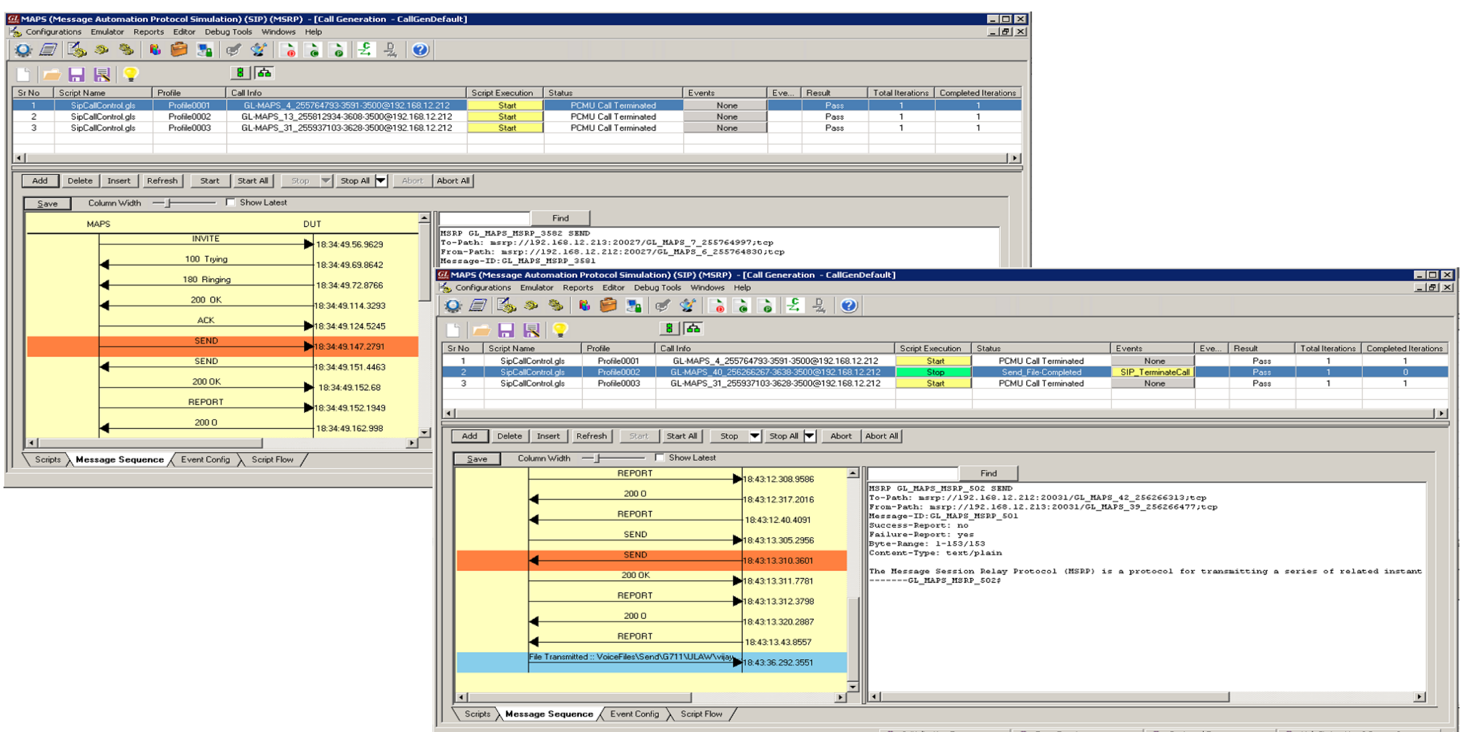
MAPS™ SIP can initiate a typical SIP call to the ATA which is configured in Pass through fax mode. Now, the ATA will initiate the call to the connected real time fax machine. Once the call is established **MAPS™** can transmit pre-recorded tiff image in pass-through mode to the fax machine at the other end. Similarly, fax generated from real fax machine can be recorded in the tiff format, and the fax call flow can be analysed in-detail for further troubleshooting.

GL's MAPS™ SIP is a useful tool for simulation of T.38 fax call. It uses SIP signaling to establish the fax session. It generates Re-Invite to switch from audio mode to image (FAX) mode. Fax Call Generation using MAPS™ as shown in the below screenshots.



Instant Messaging using SIP/MSRP

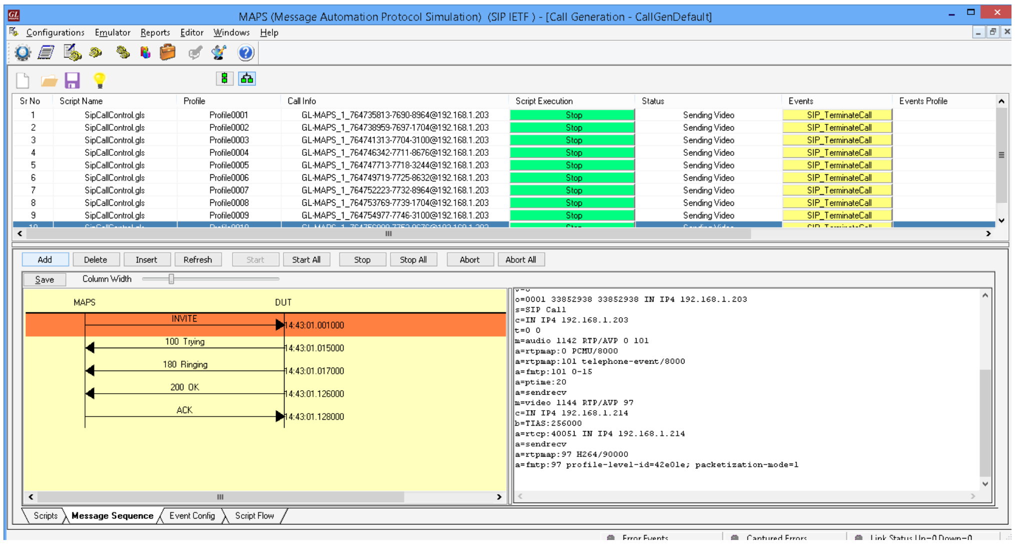
MAPS™ SIP supports **Message Session Relay Protocol (MSRP**) for instant messaging over SIP sessions (PKS112). MAPS™ SIP handles simulation of Audio and Video along with IM call, during which the pre-recorded audio/video is sent and received using RTP and text messages using MSRP during the same call. During this call type, there will be three media lines, one for Audio, one for Video and another for text messages using MSRP. The below screenshots depicts Audio, Video and IM Call types Simulation.



Bulk Video Call Generation

Video Call Simulation in MAPS™ SIP is done by using pre-recorded video traces (*.HDL GL's Proprietary format) with supporting codecs like H.264, H.263 and VP8. It has the capability of generating more than 500 simultaneous video calls on a core i7 systems. Below screenshot depicts the bulk video call simulation and RTP video transmission.

It also provides global statistics for RTP traffic such as Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter.



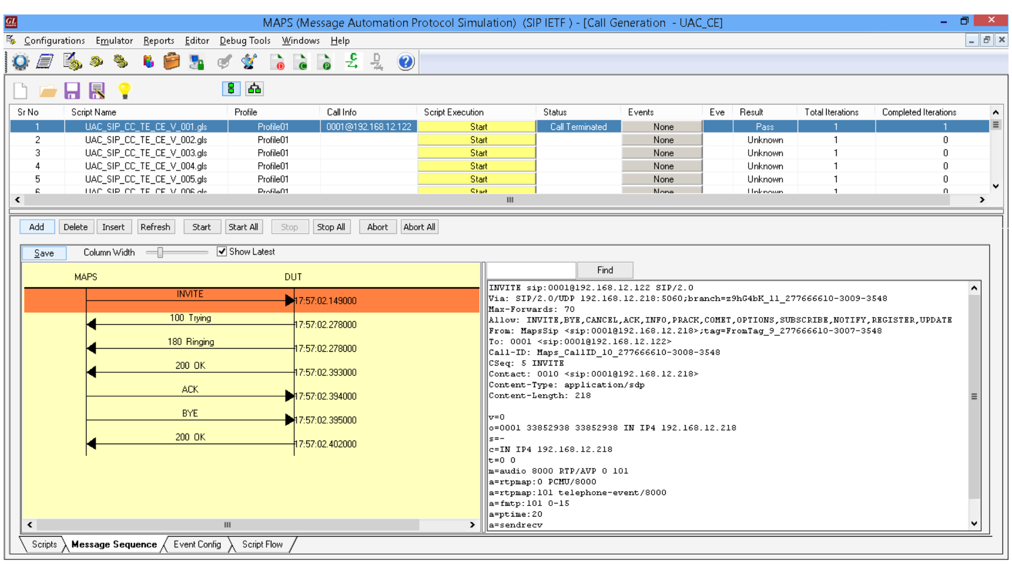
SIP Conformance Testing

MAPS™ include inbuilt scripts (*.gls) for Proxy conformance, Redirect Server conformance, Registrar conformance, UAC conformance, and UAS conformance to test the Proxy, Redirect Server, Registrar, UAC, and UAS as per ETSI standard.

Sequences Tested:

- Test Purposes For Registration
- Test Purposes For Call Control (UAC)
- Test Purposes For Call Control (UAS)
- Test Purposes For Proxy
- Test Purposes For Redirect Servers

Refer to [MAPS™ SIP Conformance Test Suite (PKS121)] brochure for more details. The below screenshot depicts UAS Conformance Script.



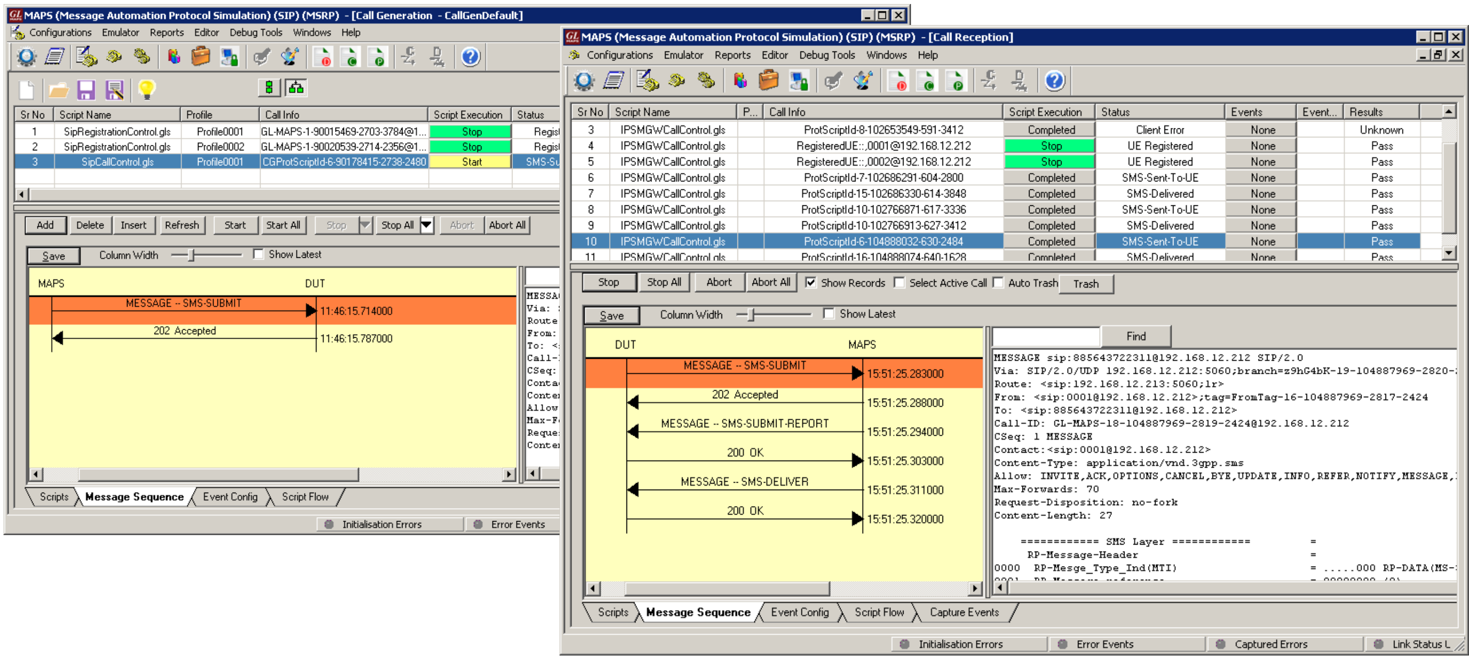
GL Communications Inc.

SMS Call Simulation over IP/IMS

GL's MAPS™ SIP facilitates Short Message Service (SMS) send/receive across SIP-IMS network interfaces. In testing end to end, a "smartphone app" is commanded to send or receive SMS messages and record results such as pass or fail.

As per the figure below, MAPS™ SIP application is configured as UEs initiating the SMS call and to accept and respond to request messages. Both the end terminals (UAs) are registered with the IP-SM-GW server.

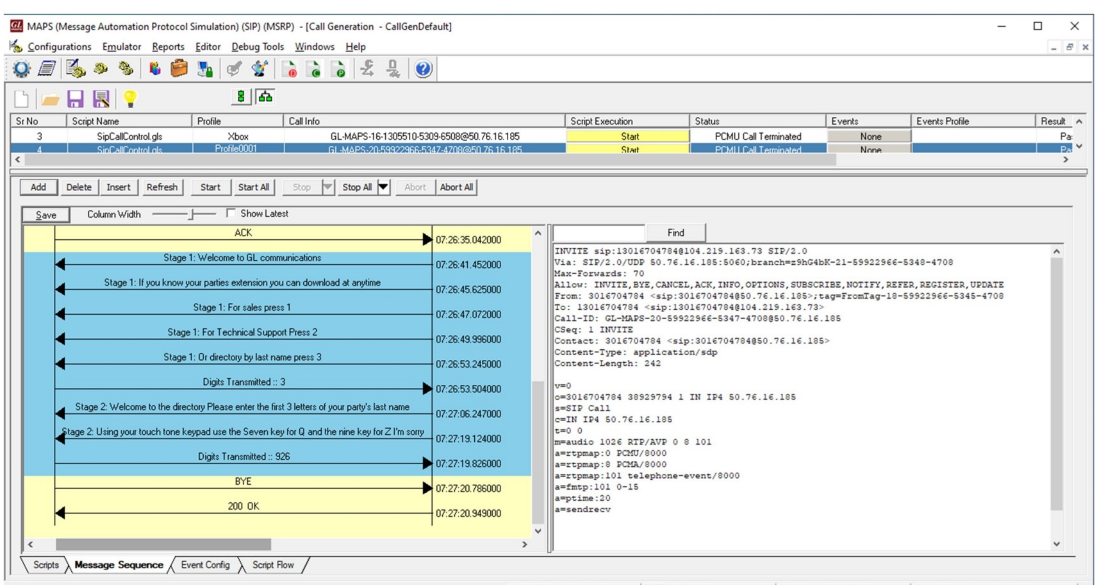
MAPS™ SIP is also configured to act as IP-SM-GW elements receiving and processing the SIP messages in IMS network. The below screenshots depicts SMS Call Simulation over IP/IMS using MAPS™.



SIP Speech to Text Interactive Voice Response (IVR)

MAPS™ SIP with GL's Speech Transcription Server provides automated IVR testing by using speech to text to navigate through an IVR tree. IVR prompts are recorded by MAPS™ SIP and transcribed by the Speech Transcription Server. Transcribed text is compared to an expected text at each IVR stage to confirm the prompt.

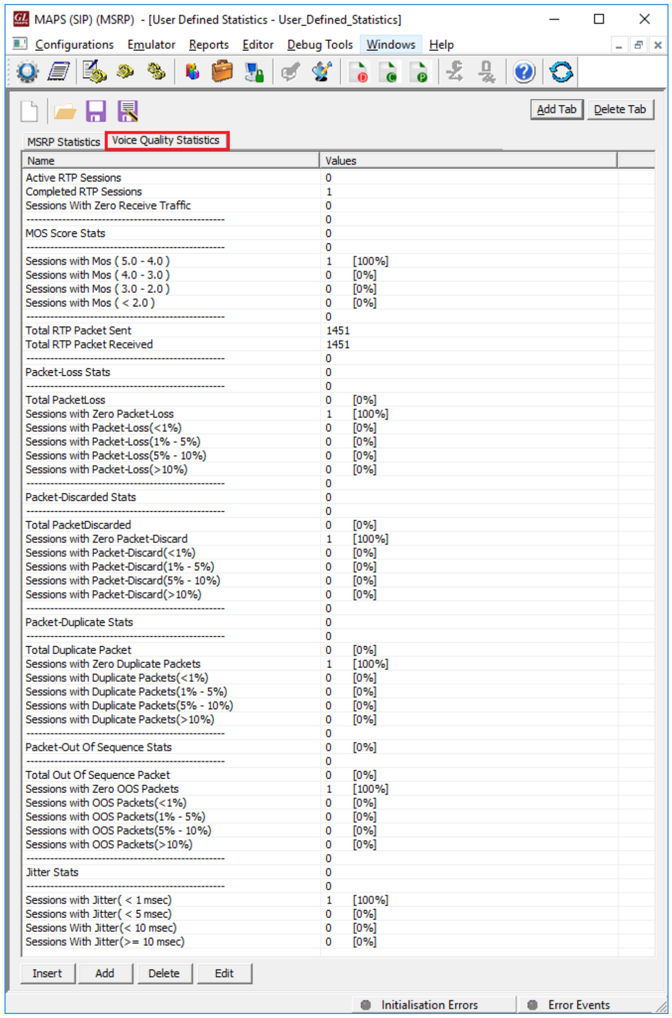
Once the IVR prompt is confirmed, MAPS™ sends DTMF or voice-based responses to move to the next stage. The expected IVR prompts and responses are defined by the customer to ensure completely customizable tests that are suitable for all IVR systems.



GL Communications Inc.

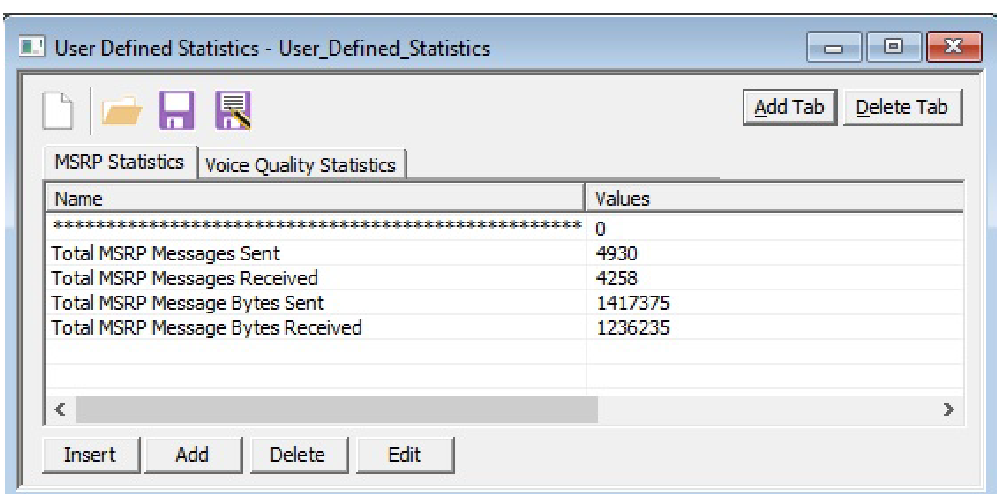
Voice Quality Statistics

MAPS™ SIP provides global voice quality statistics on RTP, which includes metrics such as Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter. These statistics are calculated and updated periodically on run time.



MSRP Statistics

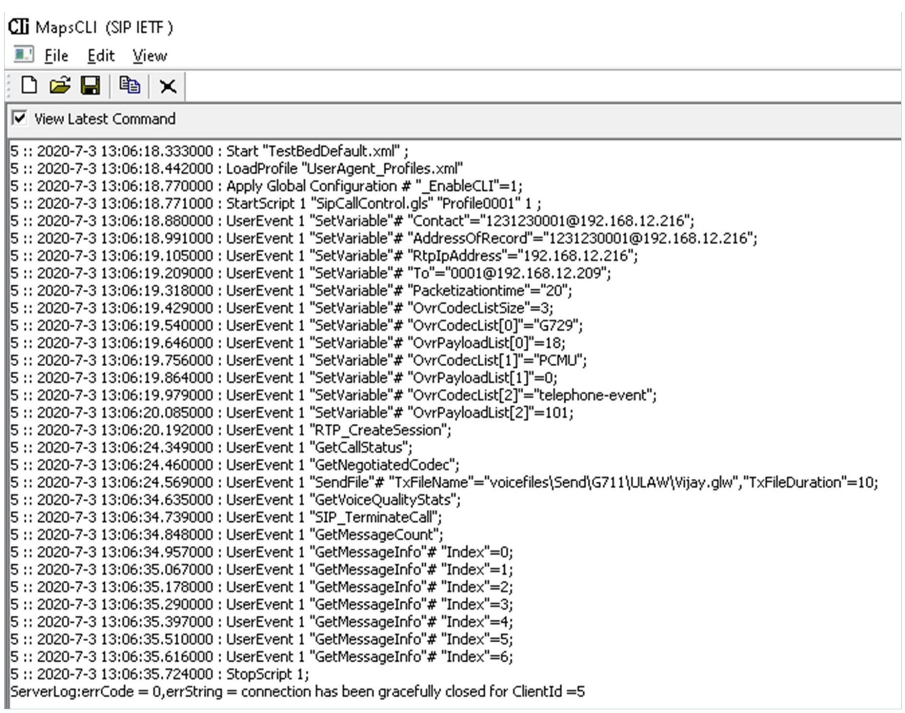
MAPS™ SIP also provides MSRP statistics including metrics such as Total MSRP Messages Sent, Received, Bytes Sent, and Bytes Received. These statistics are calculated and updated periodically on run time. The below screenshot shows User Defined MSRP Statistics.



GL Communications Inc.

MAPS™ SIP Command Line Interface

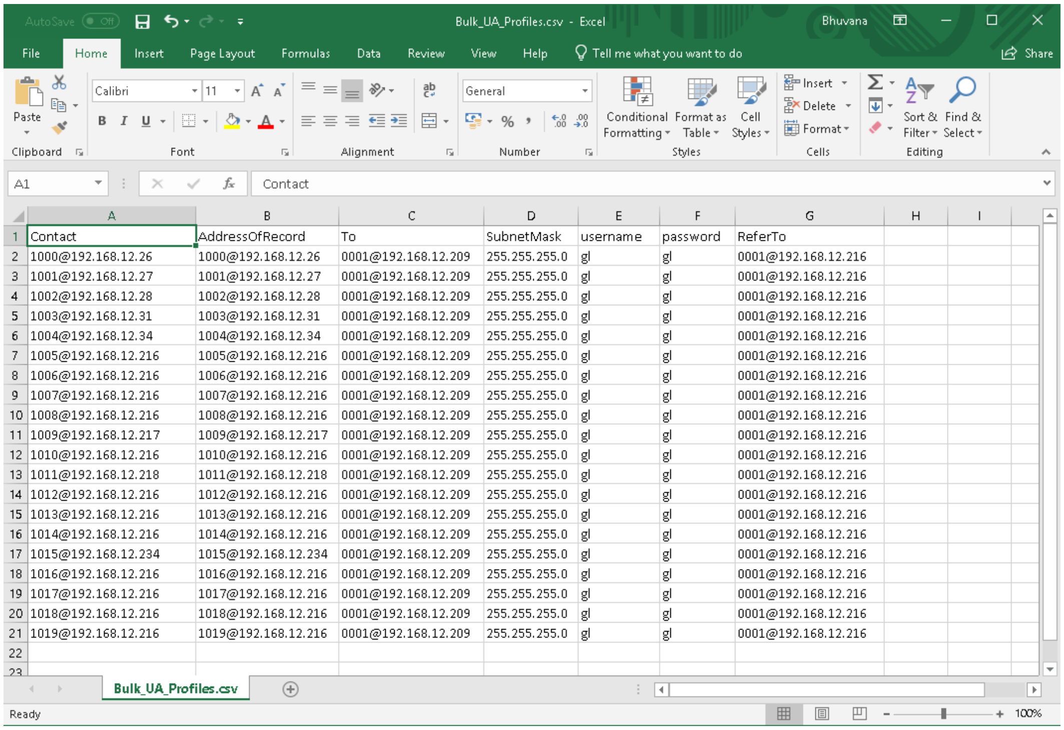
MAPS™ can be configured as server-side application, to enable remote controlling of the application through multiple command-line based clients. Supported clients include Python, Java, and TCL. Client provides a simple scripting language, with programming facilities Clients can remotely perform all functions such as start testbed setup, load scripts, and profiles, apply user events such as send digits/ file/tones, detect digits/file/tones, dial, originate call, terminate call, start and stop traffic and so on. User can also generate and receive calls through commands. The figure below depicts MAPS™ Python Client Interface used to place call and handle traffic between the end terminals. Also, observe the executed commands in the MAPS™ CLI Server window after completing the call.

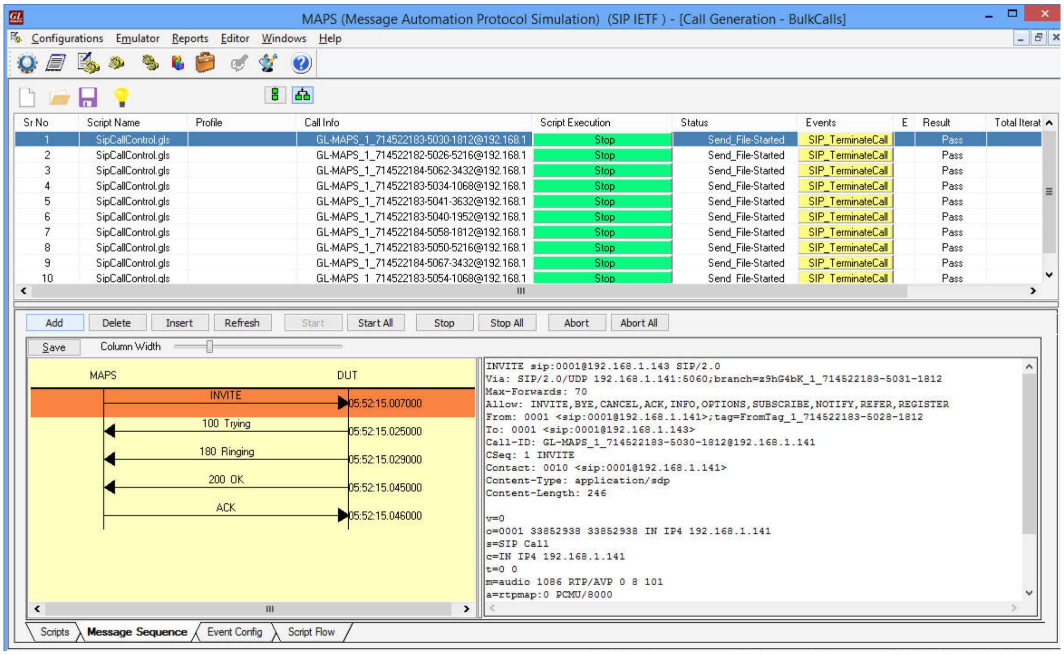


 GL Communications Inc.

Bulk Call Generation using CSV profiles

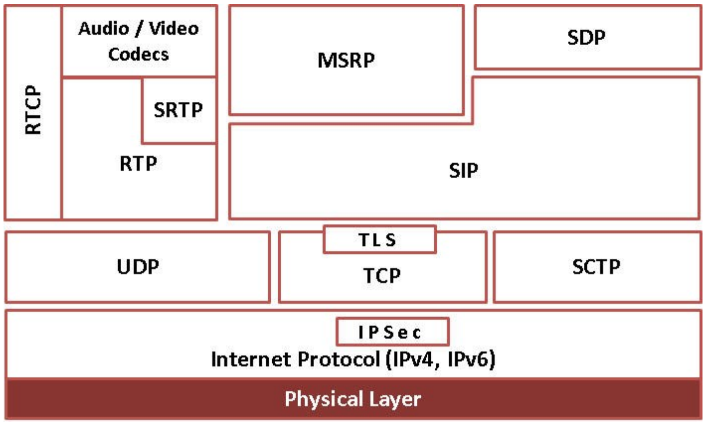
The CSV database system used within MAPS™ SIP is a simple Excel® file format, which can be used to create *N* number of UA entries each with unique UA parameters such as Contact, Address of Record, To Address, RTP Address as in real-time bulk call simulation. For MAPS™ to work with CSV profiles, it is required to enable CSV Profile. They get initialized when test bed is started. The records are accessed using the commands within the scripts. The below screenshot shows sample CSV file and Call Generation using CSV file.





GL Communications Inc.

Supported Protocol Standards



Supported Protocols	Standard / Specification Used
SIP	RFC 3261
SIP Extensions	RFC 3262 – Reliability of Provisional Responses in the Session Initiation Protocol (SIP) RFC 3311 – The Session Initiation Protocol (SIP) UPDATE Method RFC 3455 – Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd -Generation Partnership Project (3GPP) RFC 3515 – Session Initiation Protocol (SIP) Refer Method RFC 3310 – HTTP/SIP Digest Authentication Using Authentication and Key Agreement (AKA) RFC 3263 – Session Initiation Protocol (SIP): Locating SIP Servers
Secure Real-time Transport Protocol (SRTP)	RFC 3711, Secure Real-time Transport Protocol (SRTP) RFC 3551, Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control AES_CM_128_HMAC_SHA1_80 and AES_CM_128_HMAC_SHA1_32 – SRTP Algorithm
Fax (T.38)	V.34, V.21, V.27, V.29, V.8 and V.17 2400 bps to 33600 bps
Message Session Relay Protocol (MSRP)	RFC 4975 – Message Session Relay Protocol (MSRP)

Buyer's Guide

Item No	Product Description
PKS120	MAPS™ SIP
PKS121	MAPS™ SIP Conformance Test Suite (Test Scripts)
PKS170	MAPS™ CLI
PKS112	Message Session Relay Protocol for MAPS™ SIP
PKS102	RTP Soft Core for RTP Traffic Generation
PKS108	RTP Voice Quality Measurements
PKS106	RTP Video Traffic Generation
PKS109	MAPS™ High Density RTP Generator
PKS211	T.38 Fax Simulation over UDPTL
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™
PCD108	EVS codec for MAPS™
PCD109	OPUS codec for MAPS™

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

For more information, refer to MAPS™ SIP Protocol Emulator webpage.

For more information on MAPS™ products, refer to Signaling and Traffic Simulation Solutions for Telecom Networks webpage.

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