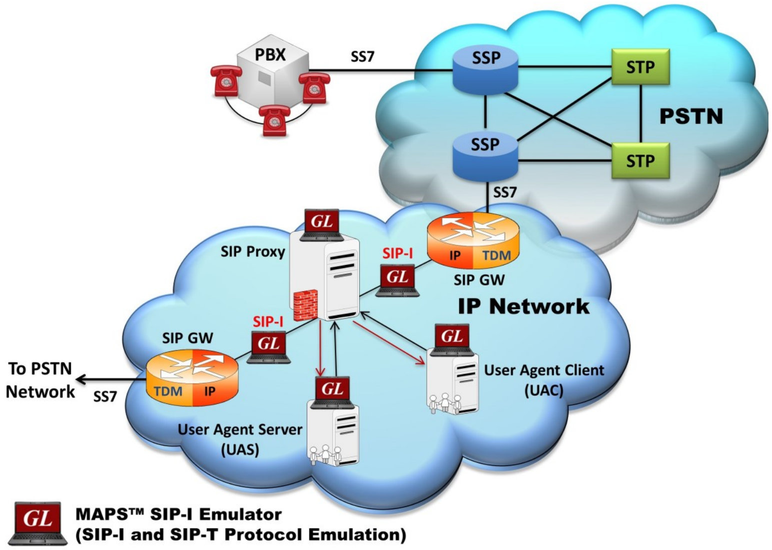
MAPS™ SIP - I

(Simulate SIP signaling with encapsulated ITU/ANSI/ETSI ISUP messages)



Overview

VoIP networks predominantly use SIP to setup and tear down voice calls and increasingly for video and multimedia calls. PSTN networks predominantly use SS7 to do the same. PSTN SS7 signaling is quite different from SIP signaling and in many cases PSTN SS7 signaling may be richer than SIP. There may be no one-to-one correspondence between SIP signaling messages and SS7 signaling messages. Also, it may not be possible to enhance SIP to accommodate the additional features of SS7, and vice-a-versa.

When a SIP-I is used to bridge the SS7 endpoints, the ISUP messages are carried (encapsulated) along with SIP signaling messages.

GL's Message Automation & Protocol Simulation (MAPS™) is a powerful Protocol Test platform-supporting a wide range protocols. MAPS™ SIP-I can simulate SIP-ISUP signaling specification as defined by the ITU / IETF standards ITU-T Q.1912.5.

MAPS™ SIP-I is a test tool/traffic generator can simulate Signaling Gateway / Softswitch as a User Agent Client (UAC) sending SIP requests with ISUP messages and as a User Agent Server (UAS) receiving requests and returning SIP responses with proper ISUP messages attached.

Test cases include general messaging and call flow scenarios for multimedia call session setup and control over IP networks. The application is available as MAPS™ SIP-I (Item # PKS126).

MAPS™ can support transmission and detection of various RTP traffic such as, digits, voice file, tones, fax, and IVR over IP networks, with additional RTP traffic licensing. It also supports auto traffic impairment with packet loss, latency, and packet effect, over the duration of the call. For more details, refer to RTP Traffic Generator.

MAPS™ SIP-I supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) Transport / SSL (OpenSSL) with a Certificate and Key. To Secure traffic communication between the Client-server, applications use the TLS protocol across a network, adding SRTP (secure RTP). SRTP encrypts the actual media portion of the calls preventing eavesdropping and tampering

MAPS™ SIPI provides global statistics for RTP audio traffic. Voice quality metrics includes Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter.

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

MAPS™ supports Command Line Interface (CLI) allowing remote controlling of the application through multiple command-line based clients. MAPS™ can be configured as server-side application which can be controlled using commands from the client environment. Supported clients include TCL, Python, Java and C#.

For more information, please visit MAPS™ SIP-I 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

- Supports both UDP and TCP (Ipv4 and Ipv6).
- Generates and processes SIP-I valid and invalid messages.
- Each SIP-I message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts.
- Handles Retransmissions and Remote Retransmissions.
- Scripted call generation and call reception.

Traffic

- Supports transmission and detection of various RTP auto traffic such as, digits, voice file, tones, fax, and IVR in IP networks. It also supports user-defined traffic over a call.
- Supports auto traffic impairments with packet loss, latency, and packet effect, over the duration of the call.
- Supports various codecs in the Session Description Protocol (SDP).
- Supported codec types includes G.711, G.729, G.726, GSM, AMR, EVRC, SMV, iLBC, SPEEX, G.722, and more.
  Click here for comprehensive information on supported codecs. *AMR and EVRC variants require additional licenses.
- Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL).

Other Features

- User defined statistics for voice quality metrics including Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter.
- Automation, Remote access, and Schedulers to run tests 24/7
- Supported on Windows® 8 or higher version operating systems.
- Supports 64-bit version to enhance signaling performance.
- Enhanced with CSV based profiles feature supporting massive UA simulation (up to 20,000 users).

CLI

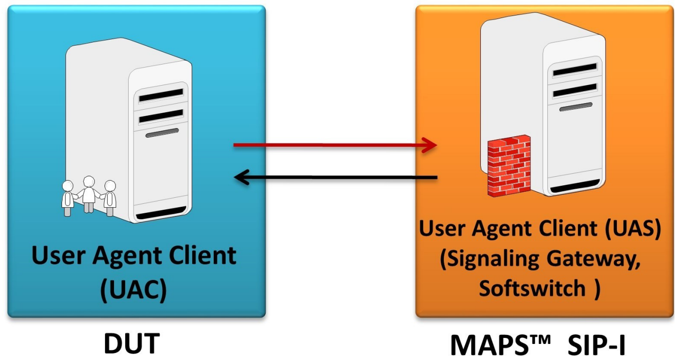
- Supports Client-Server functionality requires additional license; clients supported are TCL, Python, VBScript, Java and .Net.

Applications

- Simulates Signaling Gateway, Softswitch with SIP-I (Profile C) support to test interworking of PSTN services over IP networks.
- Fully integrated, complete test environment for SIP-I or SIP-T.

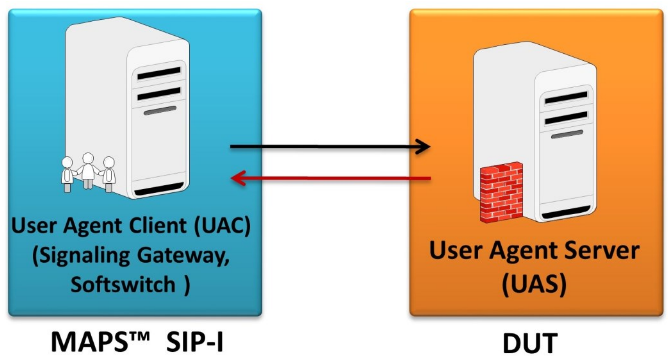
Scenario 1: MAPS™ SIP-I acting as UAS & testing UAC

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



Scenario 2: MAPS™ SIP-I acting as UAC & testing UAS

MAPS™ SIP-I can be configured to act as UAC and to test UAS. This allows the call scenarios to be automated and test DUTs.



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 - DUT (UAS), and User Agent Server (MAPS™)-to DUT (DUT - SoftPhone, IPPhone). Note that the SoftPhone, IPPhone used as DUT should support SIP-I messages. Default profile is used to configure MAPS™ SIP-I end-users.

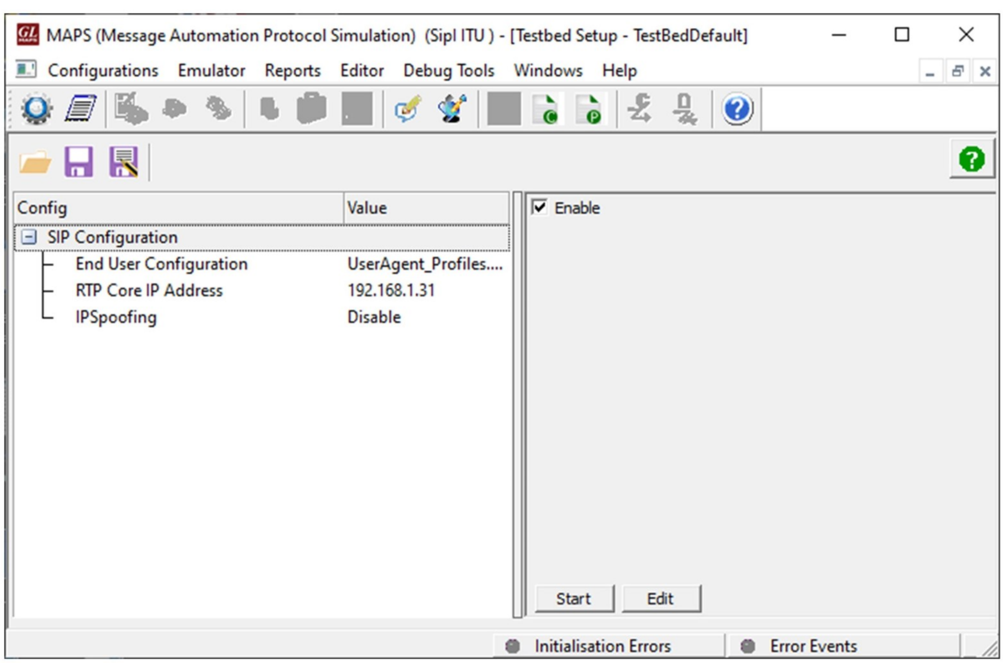


Figure: Testbed Configuration

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.

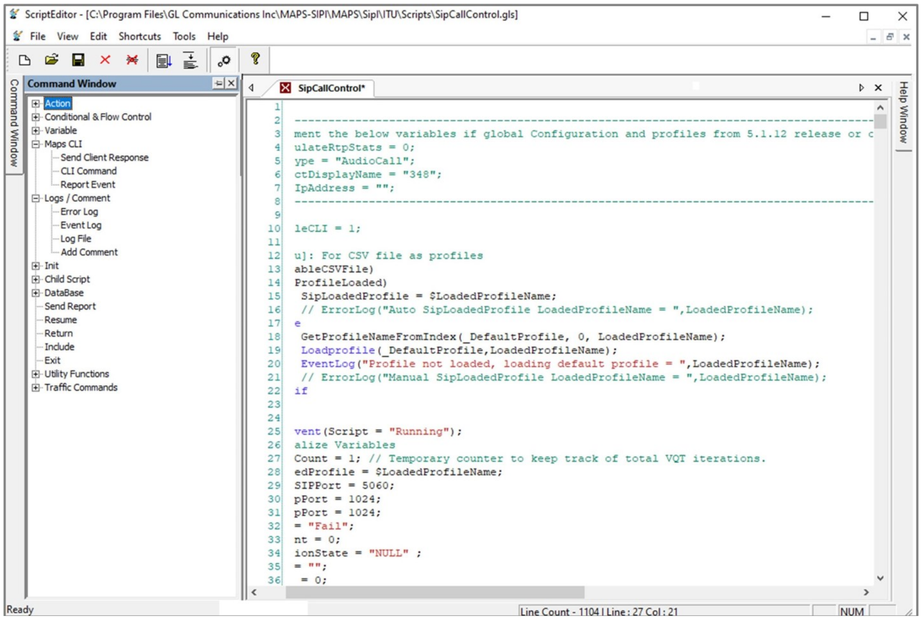


Figure: Script Editor

Profile Editor

This feature allows loading profile to edit the values of the variables using GUI, replacing the original value of the variables in the message template. An XML file defines a set of multiple profiles with varying parameter values that allow users to configure call instances in call generation and to receive calls.

Traffic profiles are available supporting RTP traffic types - Auto Traffic Digits, Auto Traffic File, Auto Traffic Tones, Auto Traffic Fax, IVR, and User-defined traffic.

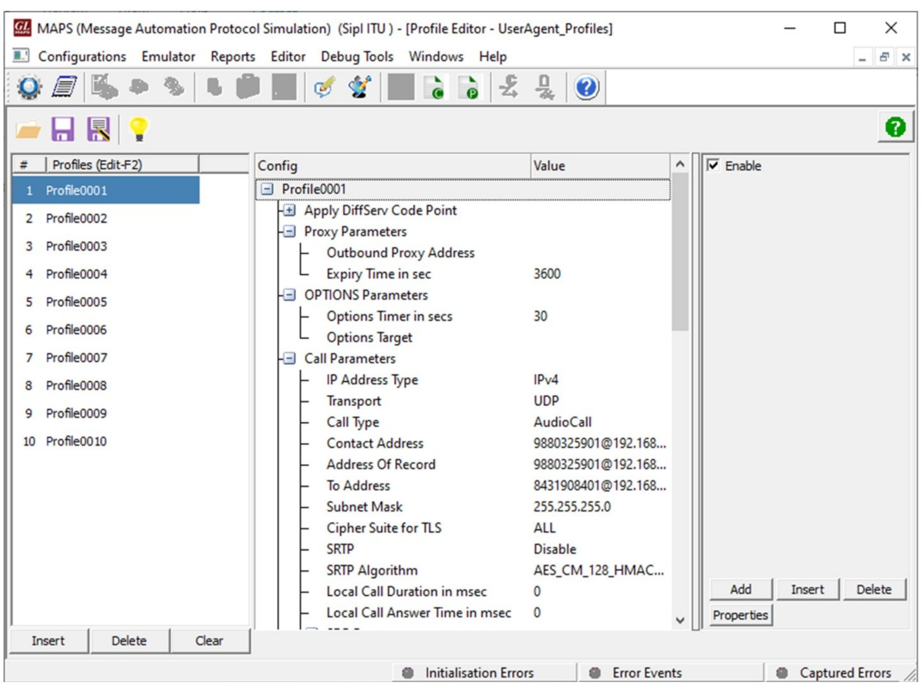
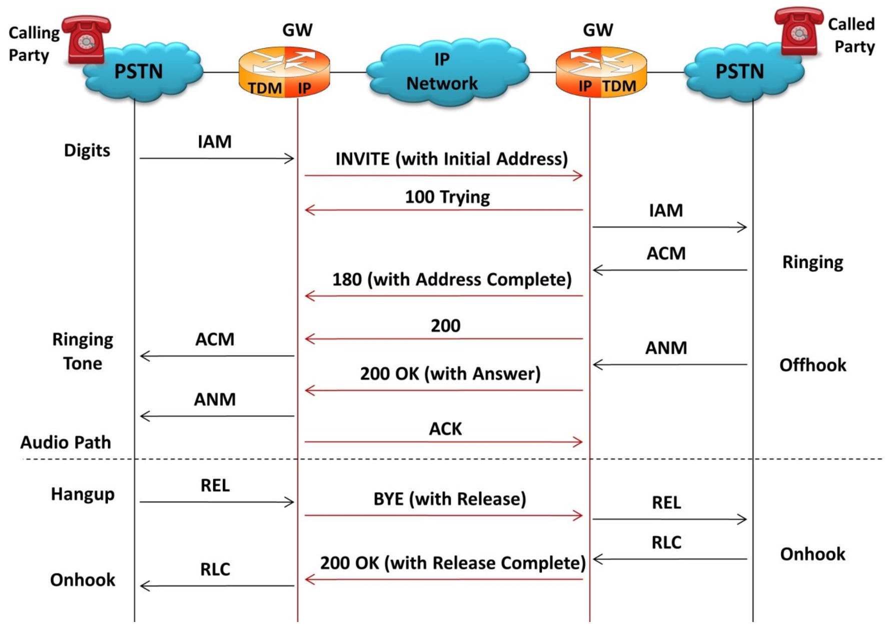


Figure: Profile Editor

GL Communications Inc.

MAPS™ SIP-I Call Flow Scenarios

MAPS™ SIP-I is configured as a User Agent Client (UAC) in ISUP-IP network. It can generate calls to a Device under Test (DUT) and the DUT can be any IP Phone, Soft phone, Proxies, Registrar, or any SIP Server that supports ISUP-IP interworking.



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 may be started manually or they can be automatically triggered by incoming messages.

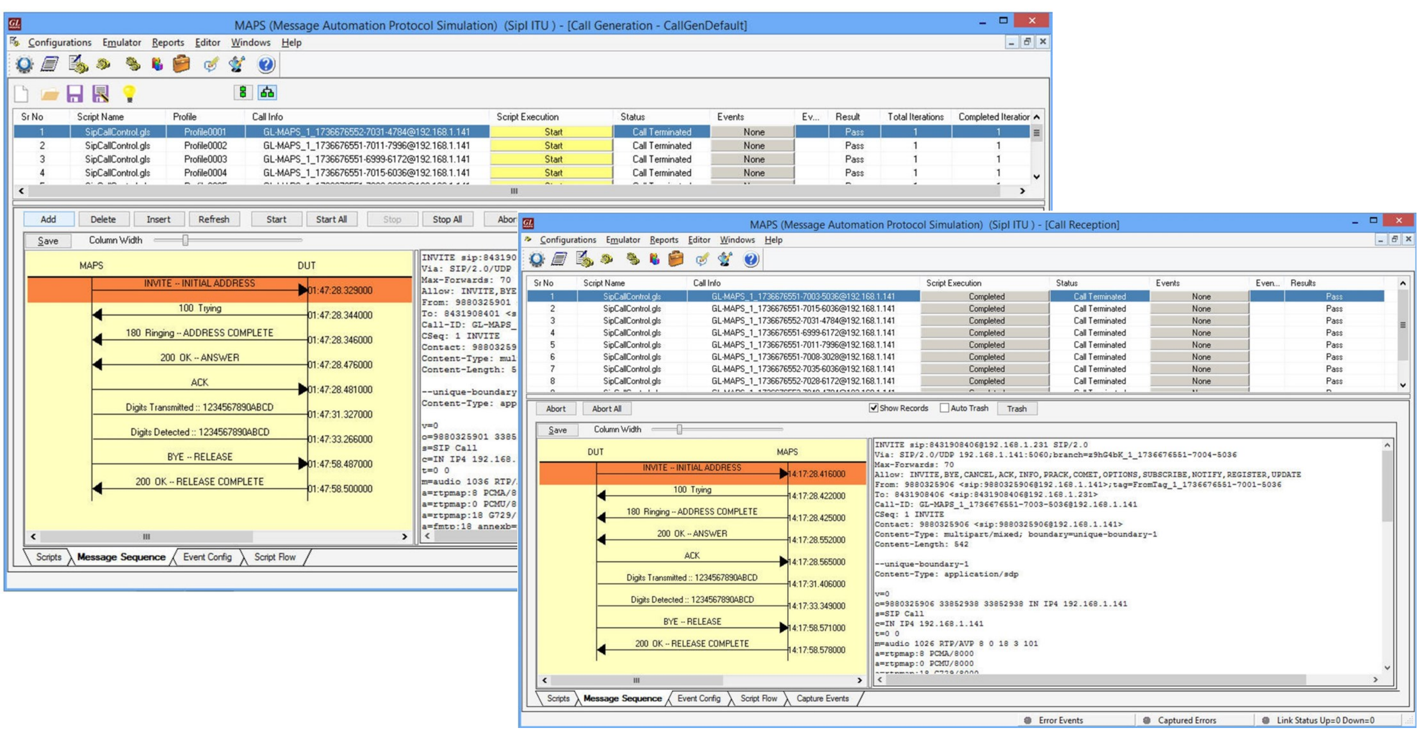


Figure: Call Generation and Reception

SIP-I Messages

'Messages' are created using pre-defined Message Templates, which are then internally called by the certain commands in the script, based on the scenario requirement. A message template is nothing but a text file containing a SIP message to which ISUP message are attached at the run time. ISUP Message Templates are created using Message Editor, in which user can specify values of certain fields to be supplied at run time.

Users may also create custom message templates and place it in these directories for later use with Script Editor.

SIP-I uses multipart MIME bodies to enable SIP messages to contain multiple payloads (SDP, ISUP, etc.).

The SIP headers and encapsulated ISUP bodies form the SIP requests. The SIP headers takes precedence over the ISUP as the contents of SIP headers may be updated in routing within the IP network.

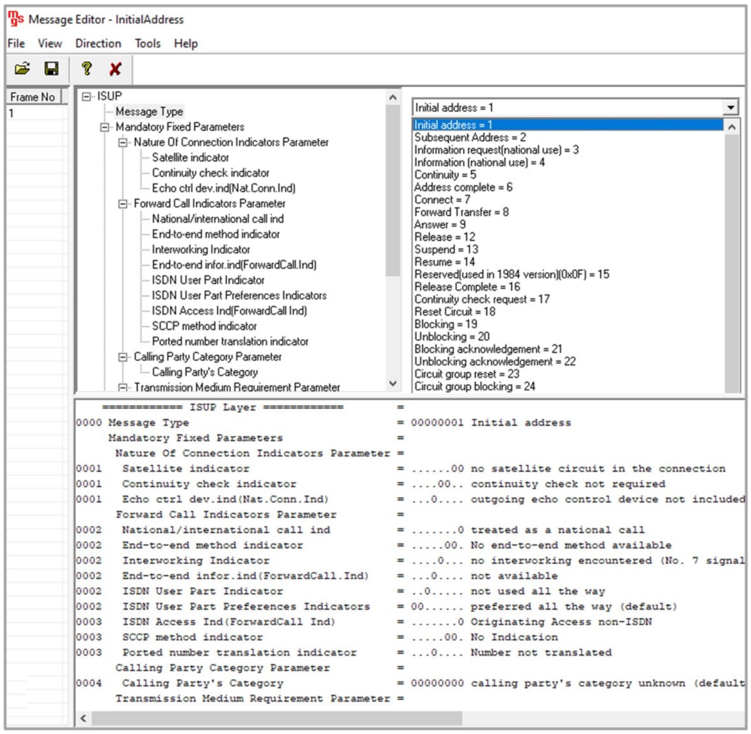
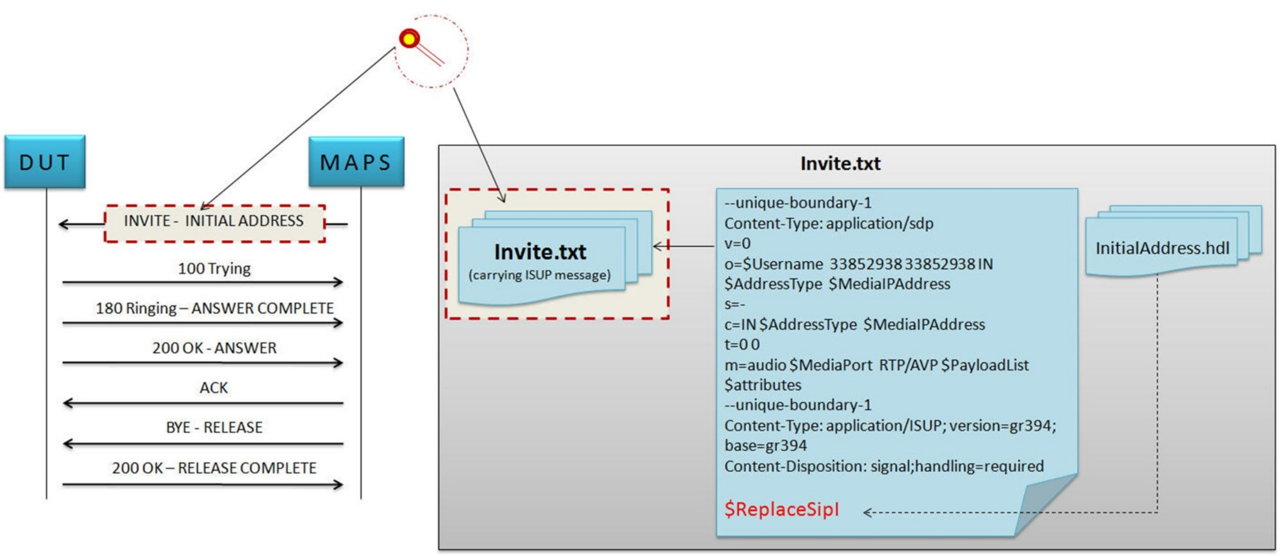


Figure: SIP-I Messages

The following illustrates the ISUP (IAM) message encapsulation in the SIP (INVITE.txt) message:

send *"Invite.txt" SendIp Port "IAMSIPI" "InitialAddressImport"*;



GL Communications Inc.

Bulk Call Generation

The CSV database system used within MAPS™ SIPI is a simple Excel® file format, which can be used to create N number of UA entries each with unique UA parameters 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 figure depicts sample CSV File and Bulk Calls Simulation using CSV File.

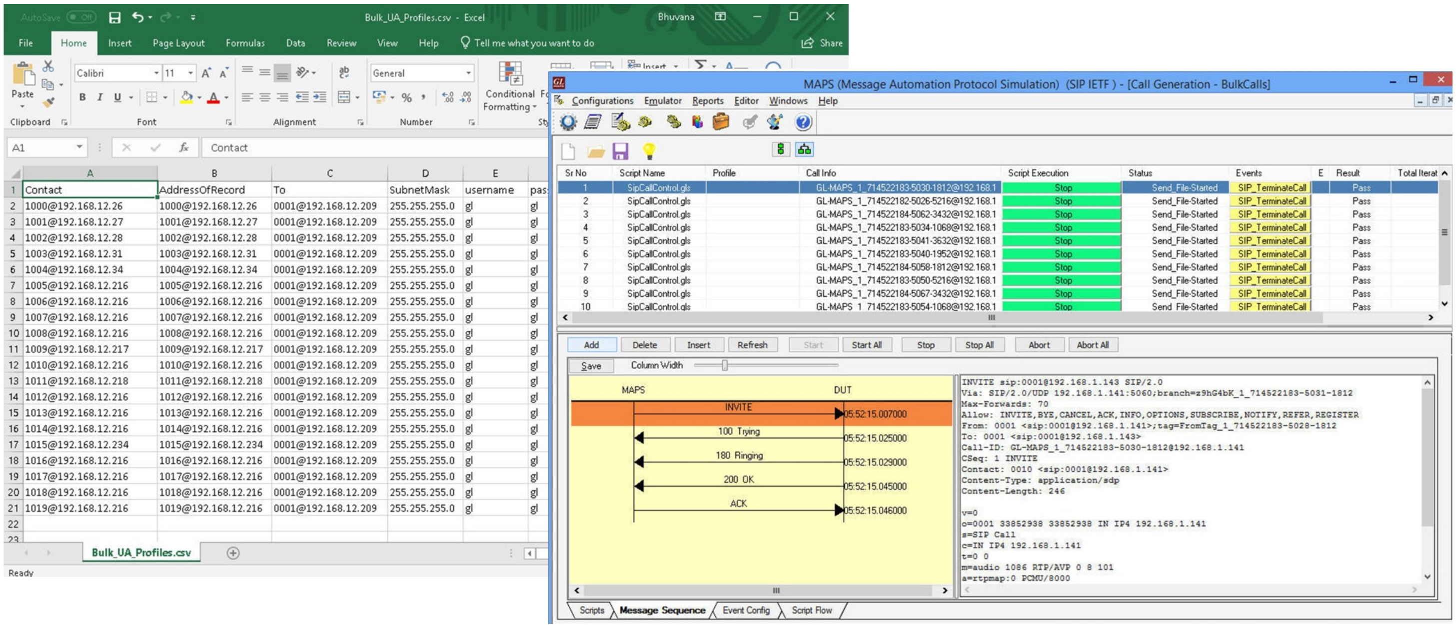


Figure: Sample CSV File and Bulk Call Generation

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 TCL, Python, VBScript, Java, and .Net.

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. This client application is distributed along with MAPS™ Server application. The below figure shows sample Python Client Script and MAPS CLI Server.

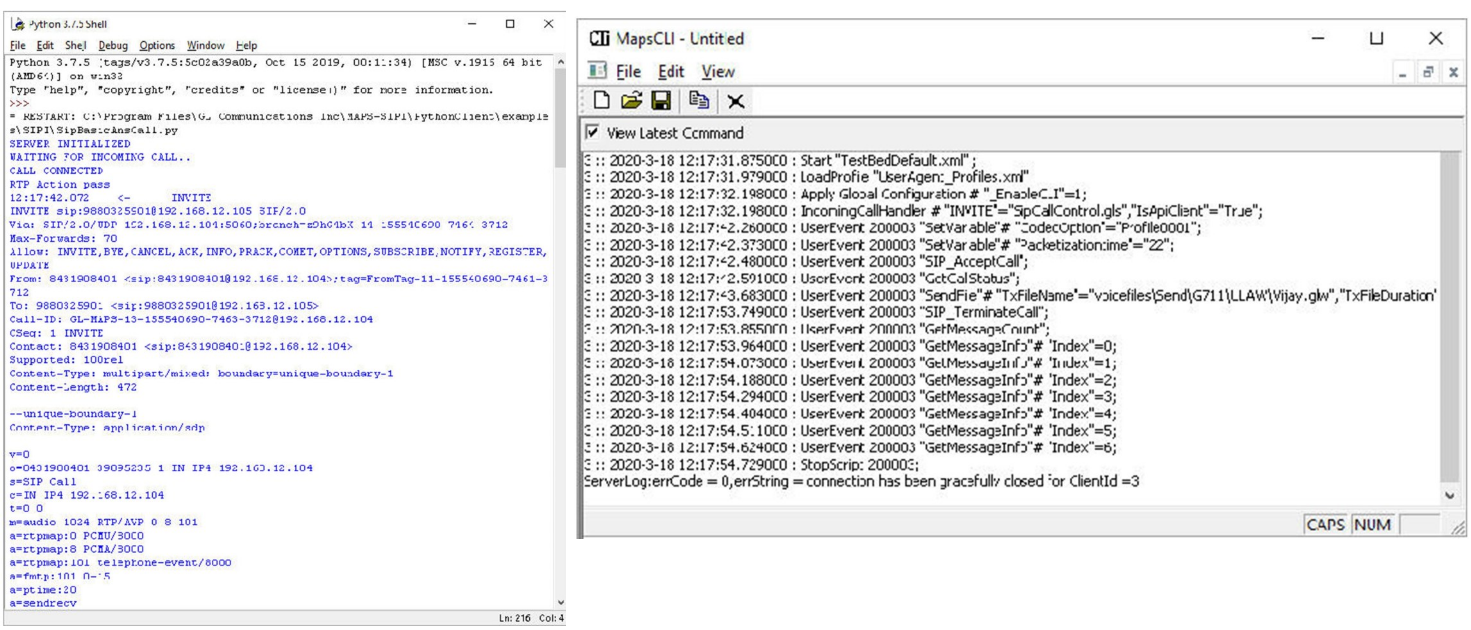


Figure: Sample Python Client and MAPS CLI Server

GL Communications Inc.

Supported Protocol Standards

Available Standards	Standard / Specification Used
SIP-I (Profile-C)	ITU Q.1912.5 - Interworking between Session Initiation Protocol (SIP) and ISDN User Part ND1007:2001/07, PNO-ISC/SPEC/007- Interworking between Session Initiation Protocol (SIP) and UK ISDN User Part (UK ISUP)
SIP-T	IETF RFC 3372

Buyer's Guide

Item No	Product Description
PKS126	MAPS™ SIP-I
PKS102	RTP Soft Core for RTP Traffic Generation
PKS103	RTP IuUP Softcore
PKS107	RTP EUROCAE ED137
PKS108	RTP Voice Quality Measurements
PKS200	RTP Pass Through Fax Emulation
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™

Item No	Related Software
PKS120	MAPS™ SIP
PKS122	MAPS™ MEGACO
PKS124	MAPS™ MGCP
PKS135	MAPS™ ISDN-SIGTRAN (ISDN over IP)
PKS130	MAPS™ SIGTRAN (SS7 over IP)
PKS140	MAPS™ LTE S1 Interface
PKS142	MAPS™ LTE eGTP (S11, S5/S8) Interfaces
PKS164	MAPS™ UMTS IuPS (over IP) Interface Emulation
PKS160	MAPS™ UMTS IuCS and IuH Interface Emulation

For more information, please visit Signaling and Traffic Simulator 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