Message Automation & Protocol Simulation (MAPS™)

MAPS™ SIP - I

Simulates SIP Server and Client entities

Scripted Call Generation and Automated Call Reception

Provides Fault Insertion, & Erroneous Call Flow Testing

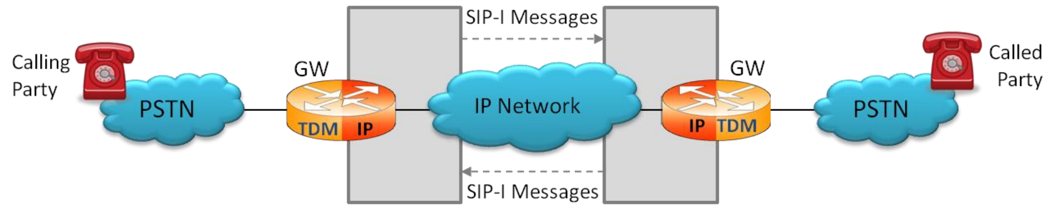
Supports a Multipart Payload and/or the ISUP MIME Type

Customization of Call Flow with Script and Profile Editors

Option to Change Event Parameters at Run-time

Load Generation Feature for Stress/Load Testing

Analysis and Simulation Capability on par with any Protocol Tester in the Market



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)

For more details, refer to http://www.gl.com/maps-sip-i-emulator.html

Main Features

- Simulates Signaling Gateway, Softswitch as UAC, UAS, in the network.
- Supports transmission and detection of RTP traffic - digits, voice file, single /dual tones
- Handles Retransmissions and remote Retransmissions.
- Supports both UDP and TCP.
- Generates and processes SIP-I valid and invalid messages.
- Fully integrated, complete test environment for SIP-I.
- Supports complete customization of call flow and messages.
- Supports scripted call generation and automated call reception.
- Supports message templates for each SIP-I message and customization of the field values.
- Facilitates defining variables for the various protocol fields of the selected SIP-I message type.
- Supported on Windows® XP or higher version operating systems

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Working Principle

- **Message Templates** - Forms the backbone of MAPS™ application that contains protocol fields with default values
- **Script Editor** -
  - Creates a script for scenario based testing (call flow)
  - Uses pre-defined message templates in the script
  - Access protocol fields as variables
- **Message Editor** - is used to edit /create ISUP message templates. But, SIP messages are manually edited / created using Notepad®.
- **Profile Editor** – Creates or edits profiles containing values assigned to the variables replacing the original values.
- **Event Profile Editor -** Allows you to create **Event Profiles** for user-defined events in a script. The values of the variables in the user-events can be changed during script execution.

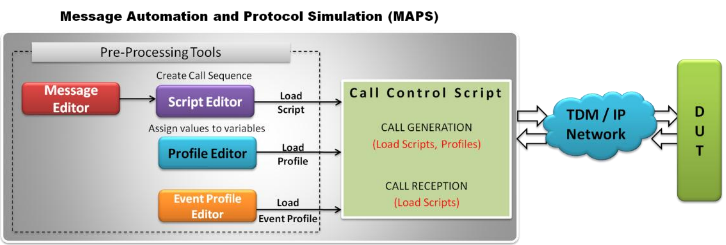
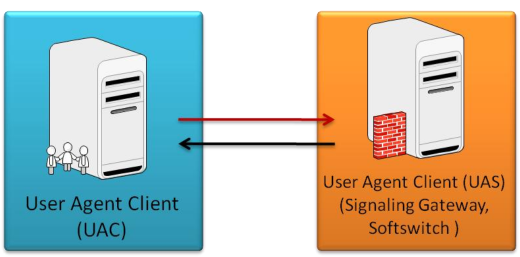


Figure: MAPS™ application Working Principle

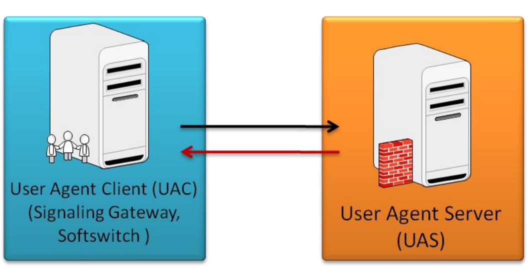
Scenario 1: MAPS™ SIP-I acting as UAS and testing UAC

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.

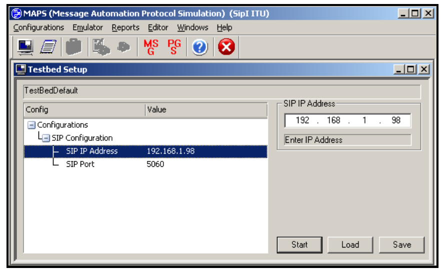


Figure: Testbed Setup

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.

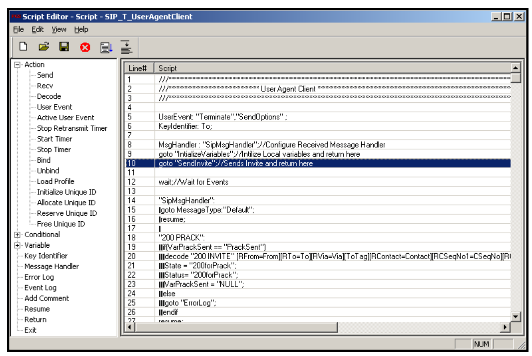


Figure: Script Editor

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.

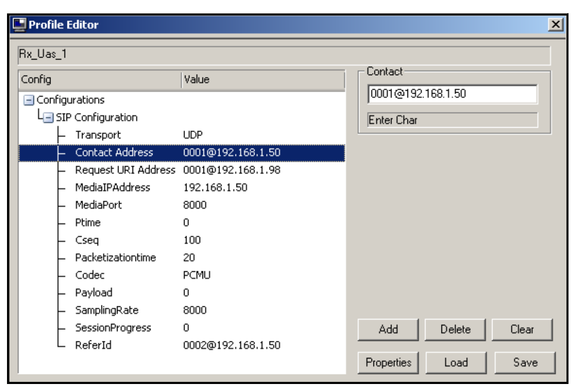


Figure: Profile Editor

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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.

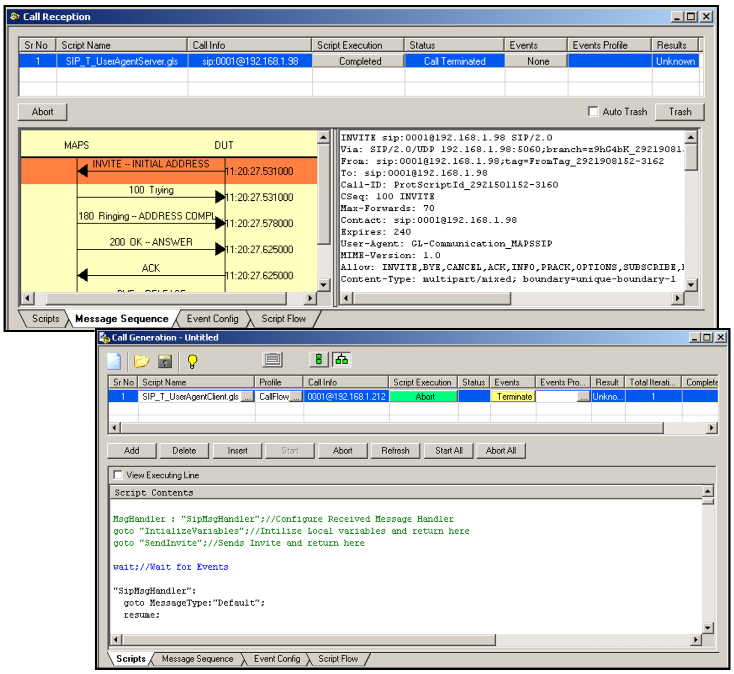


Figure: Call Generation and Reception

Incoming Call Handler Configuration

The script configuration option is used to preset the script required to handle all possible SIP-I signaling and call control messages against particular message expected to arrive.

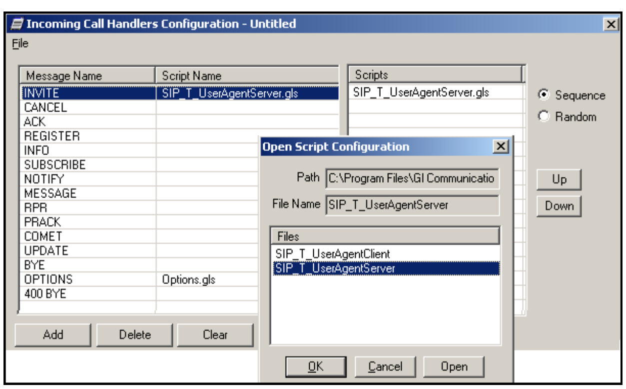
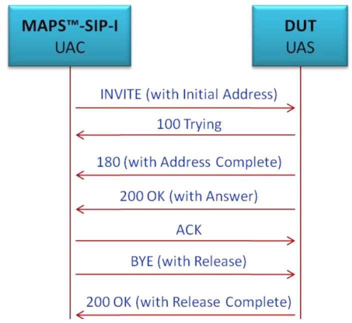


Figure: Incoming Call Handler

MAPS™ SIP-I Call Flow Scenarios

Scenario 1: MAPS™ SIP-I acting as UAC

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.

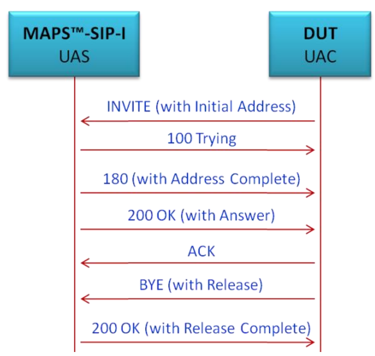


Sample UAC Script

```
send "Invite.txt" SendIp Port "IAMSipI"
"InitialAddressImport";
decode “100Trying.txt” SendIp Port;
decode “180ringing.txt” SendIp Port "ACMSipI"
"AddressCompleteExport";
decode "200oktoINVITE.txt" SendIp Port "ANMSipI"
"AnswerExport";
send “Ack.txt” SendIp Port;
send "Bye_UAS.txt" SendIp Port "RELSipI"
"ReleaseImport";
decode "200ok_to_bye.txt" SendIp Port "RLCSipI"
"ReleaseCompleteExport";
```

Scenario 2: MAPS™ SIP-I acting as UAS

MAPS™ SIP-I acts as the UAS automated with receive script to reply back to the incoming request messages from the client (DUT).



Sample UAS Script

```
decode "INVITE" # "InitialAddressExport";
send "100Trying.txt" SendIp Port;
send "180ringing.txt" SendIp Port "ACMSipI"
"AddressCompleteImport";
send "200oktoINVITE.txt" SendIp Port "ANMSipI"
"AnswerImport";
decode "Ack.txt" SendIp Port ;
send "Bye_UAS.txt" SendIp Port "RELSipI"
"ReleaseImport";
decode "200ok_to_bye.txt" SendIp Port "RLCSipI"
"ReleaseCompleteExport";
```

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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.

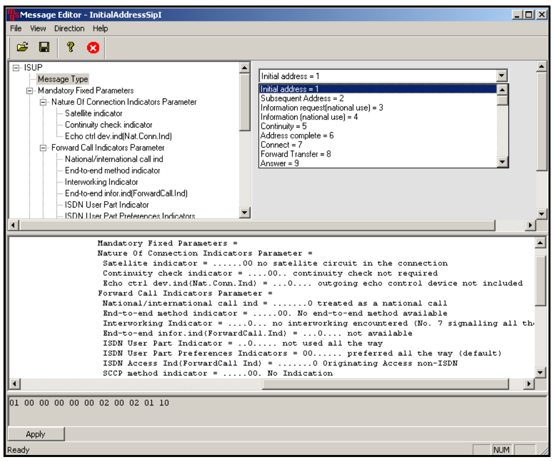

Figure: ISUP Message Editor

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

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

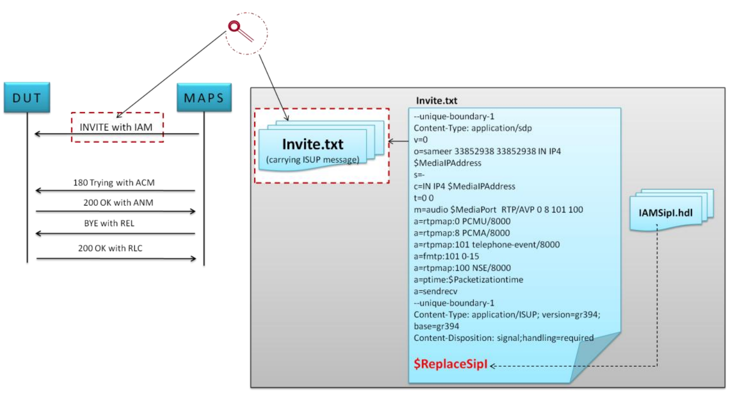

Figure: Generating SIP-I Messages

Supported Protocol Standards

Available Standards	Standard / Specification Used
SIP-I	ITU Q.1912.5
SIP-T	IETF RFC 3372

Buyer's Guide

PKS126 - MAPS™ SIP-I

PKS102 - RTP Traffic Option

Related Software

PKS120 - MAPS™ SIP

PKS121 - MAPS™ SIP Conformance Test Suite (Test Scripts)

PKS122 – MAPS™ MEGACO

PKS123 – MAPS™ MEGACO Conformance Test Suite (Test Scripts)

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 complete list of MAPS™ products, refer to http://www.gl.com/maps.html webpage.

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