Simulates UAC, UAS, Registrants, & Redirect Servers

SIP Conformance Testing based on ETSI TS 102027-2

Supports both UDP and TCP / IP (IPv4 & IPv6)

Supports RTP Traffic (Voice, Digits, Tones, Fax, IVR) Simulation

Secured RTP (SRTP) over TLS Transport

Bulk Video Traffic Generation (500 Simultaneous Calls)

Massive UA Simulation (up to 20,000 Users) with CSV based Profiles

FAX over IP (T.38, T.30 Pass Through) Simulation

Message Session Relay Protocol (MSRP) Emulation Supporting Instant Messaging

Scripted Call Generation and Automated Call Reception

Load Generation Feature for Stress/Load Testing

TCL, Python, Java and C# Clients for Remote Operations

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), Registrant and Redirect 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 http://www.gl.com/rtp-traffic-generator.html

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 & 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™ 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#.

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.

## 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 conference call, blind call transfer, hold, auto call rejection, and silence packets generation.
- Ability to send "reliable provisional responses" and start early media actions.
- Supports VoIP implementation as per ED-137B of EUROCAE standard. *RTP EUROCAE ED137 option require additional licenses.
- 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 service (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, PacketLoss, 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 20,000 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 20000 active calls (Voice), 250 CPS (Requires PKS109 + GL’s HD NIC)

**Based on System Requirements – Windows® 8.1, 16 GB RAM, Core i7 @3.5 GHz

### Other Features
- 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.

### 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.

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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.

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.

Figure: End-to-End Gateway Testing
MAPSTM SIP Call Flow Scenarios

SIP Registration and Call Control Procedure
MAPSTM 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.

Call Generation and Reception
In call generation, MAPSTM is configured for the outgoing 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.

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.

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: Script Editor

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Figure: Call Generation and Call Reception

Figure: Profile Editor
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.

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.

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 figure 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.
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.

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.
MAPS™ SIP APIs

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. Client provides a simple scripting language, with programming facilities such as looping, procedures, and variables. The Client application includes a dll file, a packaged library that enables communication with the Server from a Client environment.

This client application is distributed along with MAPS™ Server application. Multiple MAPS™ CLI servers can be controlled remotely from single client application.

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.

Figure: Sample UserAgentClient TCL Script

Figure: Sample CSV File

Figure: Bulk Calls Simulation using CSV File

Figure: Sample Sip Basic Call using Python Script
## Supported Protocol Standards

<table>
<thead>
<tr>
<th>Supported Protocols</th>
<th>Standard / Specification Used</th>
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<tr>
<td>SIP</td>
<td>RFC 3261</td>
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| 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

- **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
- **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
- **PKS109** - High Density RTP Traffic Generation
- **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™