SIP Protocol Simulation - MAPS™ SIP

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](http://www.gl.com/rtp-traffic-generator.html)

As a new enhancement, 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 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.

Users can remotely control MAPS™ using commands from the TCL environment. Multiple MAPS™ CLI servers can be controlled remotely from single client application (such as TCL, Python, VBScript, Java, and .Net).

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).

GL also provides PacketScan™ HD (PKV120) for High Density IP Traffic Analysis w/ 4x1GigE and Network Monitoring w/ 2x10GigE and PacketScan™ FB (PKV121), a File Based IP Traffic Analyzer Server for near real-time processing of traces.

### Main Features

| **Signaling** | Generates and processes SIP valid and invalid messages.  
|              | Supports complete customization of scripts and parameters in the profiles  
|              | Supports complete customization of SIP headers, call flow, and messages.  
|              | Each SIP message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts.  
|              | Supports IPv4/IPv6 and transport over UDP and TCP, and TLS for secure transport.  
|              | 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.  
|              | User-defined voice quality statistics for received RTP Traffic can be calculated and updated periodically during run-time to a csv file.  
|              | Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL) |
| **Bulk Call Capability** | With normal RTP (PKS102 licensing) Maximum Simultaneous Calls - 2500, and Calls per Second - 250 (in high end server machines).  
|              | Without RTP (only signaling) Maximum Simultaneous Calls - 70,000, and Calls per Second - 750 (in high end server machines).  
|              | With MAPS™ HD RTP network appliance up to 20,000 endpoints per unit can be easily achieved (requires PKS109 and specialized hardware).  
|              | Capability to generate more than 500 simultaneous video calls on a Core i7 systems. |
| **Other Features** | Automation, Remote access, and Schedulers to run tests 24/7.  
|              | Supported on Windows® 7, or higher version operating systems.  
|              | Supports 64-bit version to enhance signaling performance.  
|              | 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 call quality metrics such as Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter.  
| **CLI**      | Supports Client-Server functionality. Supported clients are TCL, Python, VBScript, Java, and .Net.  
| **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. |
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**
**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 - Proxy, Redirect Server), and User Agent Server (MAPS™) - to - DUT (DUT - SoftPhone, IP Phone). IPSpoofing for auto generation of virtual IP addresses used during bulk call generation.

End user configuration profile used to configure MAPS™ SIP with User Agent parameters.

**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 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, IVR, and User-defined traffic. Allow users to create their own profiles to suit their custom scripts. The call type parameter can be set to Video or Audio type to support RTP Video & Voice Traffic Generation in MAPS™ SIP. Profile Editor includes configurations for the supported transport types (UDP, TCP and TLS).

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**Figure: Testbed Setup**

**Call Generation and Reception**

In call generation, MAPS™ 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.

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**Figure: Call Generation and Call Reception**

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**Figure: Profile Editor**

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MAPS™ SIP Call Flow Scenarios

**SIP Registration Control script**

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.

**MAPS™ SIP Call Flow Scenarios**

**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.

**Bulk Video Call Generation**

Video Call Simulation in MAPS™ SIP is done by using pre-recorded video traces (*.HDLS 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.

**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.

**Generation of Fax Calls**

**Figure: Bulk Video Call Generation**

**Figure: Fax Call Generation using MAPS™**

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**Voice Quality Metrics**

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

![Voice Quality Statistics](image)

**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. 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.

![Sample UserAgentClient TCL Script](image)

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**Supported Protocol Standards**

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<td>RFC 3551, Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control</td>
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**Buyer’s Guide**

PKS120 - MAPS™ SIP
PKS121 - MAPS™ SIP Conformance Test Suite (Test Scripts)
PKS102 - RTP Soft Core for RTP Traffic Generation
PKS108 - RTP Voice Quality Measurements
PKS106 - RTP Video Traffic Generation
PKS211 - T.38 Fax Simulation over UDP/TPL
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™