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™ SIP also supports VoIP implementation as per ED-137B of EUROCAE (European Organization for Civil Aviation Equipment) standard.

MAPS™ can support transmission and detection of various RTP 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

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) are used with the supporting codecs like H.264, H.263 etc. It has the capability of generating more than 500 simultaneous video calls on a core i7 systems.

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, etc). MAPS™ TCL Client application includes a MapsClientIfc interface, a packaged library that enables communication with the MAPS™ Server. TCL (Tool Command Language) Client is a command-line interface (TClsh85.exe) which is distributed along with MAPS™ Server application, using which any real-time scenarios can be simulated.

GL’s MAPS™ is enhanced to a High Density version and a special purpose 1U network appliance that is capable of high call intensity (hundreds of calls/sec) and high volume of sustained calls (tens of thousands of simultaneous calls/1U platform).

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.

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Main Features

Signaling

- Generates and processes SIP valid and invalid messages.
- 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 both UDP and TCP (IPv4 and IPv6).
- Handles Retransmissions of messages with specific interval.
- Scripted call generation and call reception.
- MAPS™ HD network appliance can easily achieve 4 to 20,000 endpoints per 1U server
- Supports conference, attended call transfer, blind call transfer, and call forwarding.
- 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 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, and more. Click here for comprehensive information on supported codecs. *AMR and EVRC variants require additional licenses.
- Test IVR, and Instant messaging features.
- Bulk Video Call Generation supported with H.264 and H.263 video codecs
- Capability to generate more than 500 simultaneous video calls on a core I7 systems
- User-defined statistics for RTP Voice and Video quality calls

Other Features

- Automation, Remote access, and Schedulers to run tests 24/7
- Supported on Windows® 7, 8 or higher version operating systems.
- Supports 64-bit version to enhance signaling performance.

CLI

- Supports Client-Server functionality clients supported are TCL, and Python.

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.

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**Figure: End-to-end Gateway Testing**

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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).

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.

The call type parameter can be set to Video or Audio type to support RTP Video & Voice Traffic Generation in MAPS™ SIP.

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.
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.

SIP Call Control script
- UAC initiates the call by sending INVITE request message to UAS
- UAS replies with 100 Trying/180 Ringing messages while call initiation and processing
- UAS on successful media negotiation, sends a Success 200 OK
- and waits for the ACK message from UAC
- With the call establishment RTP Media flow in both the ways allowing conversation to carry out between the entities
- Either of the entity can terminate the Call, by sending BYE
- Call is terminated after receiving 200 OK from the other end

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 etc. 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.

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
Command Line Interface

The MAPS™ TCL Client application includes a MapsClientIfc interface, a packaged library that enables communication with the MAPS™ Server from a TCL environment. The advantage of such communication enables user to control MAPS™ using send and receive commands.

TCL (Tool Command Language) Client is a command-line interface (TCIsh85.exe) which is distributed along with MAPS™ Server application.

Using TCL client, any real-time scenarios can be simulated by sending instructions to the MAPS™ server. MAPS™ Server processes the commands and takes necessary actions. MAPS™ Client can get the server status by exporting the variables.

TestShell or scripting languages such as TCL with library of functional capabilities can easily create compliance tests for simple to complex Next Generation Networks (NGN) voice features.

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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

**PKS103** - RTP IuUP Softcore

**PKS107** - RTP EUROCAE ED137

**PKS106** - RTP Video Traffic Generation

**PKS190** - RTP Pass Through Fax Emulation

**PKS109** - 2 Fax Ports, RO

**PKS109** - 8 Fax Ports, RO

**PKS104** - 30 Fax Ports, RO

**PKS105** - 60 Fax Ports, RO

**PKS106** - 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™

Related Software

**PKS122** – MAPS™ MEGACO

**PKS123** - MAPS MEGACO Conformance Test Suite (Test Scripts)

**PKS124** - MAPS MGCP w/ Conformance Test Suite

**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

**PKS126** - MAPS™ SIP-I for SIP ISUP Interface Emulation

**XX651** - MAPS™ CAS Emulator (requires T1/E1 Analyzer, includes CLI)

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Supported Protocol Standards

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