MAPSTM SIP
SIP + RTP + MSRP Simulation
MAPS™ SIP

Normal RTP Traffic Generation
(2000 simultaneous calls)

MAPS™ SIP (w/ 4 x 1G cards)
HD RTP Traffic Generator
32,000 Simultaneous Calls (with RTP Traffic)
SIP Architecture and Entities

- Requests
- Responses
- Non-SIP Protocol

1. SIP User Agent (Caller) → SIP Proxy
2. SIP Proxy → SIP Redirect Server
3. SIP Redirect Server → SIP Proxy
4. SIP Proxy → SIP Proxy
5. SIP Proxy → Location Service
6. Location Service → SIP Redirect Server
7. SIP Redirect Server → SIP Proxy
8. SIP Proxy → SIP User Agent (Caller)
9. SIP Proxy → SIP Proxy
10. SIP Proxy → SIP Proxy
11. SIP Proxy → SIP Proxy
12. SIP Proxy → SIP User Agent (Caller)
SIP Protocol Stack

<table>
<thead>
<tr>
<th>Supported Protocols</th>
<th>Standard / Specification Used</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>RFC 3261</td>
</tr>
<tr>
<td>SIP Conformance</td>
<td>ETSI TS 102-027-2 v4.1.1</td>
</tr>
<tr>
<td>SIP Extensions</td>
<td>RFC 3262 - Reliability of Provisional Responses in the Session Initiation Protocol (SIP)</td>
</tr>
<tr>
<td></td>
<td>RFC 3311 - The Session Initiation Protocol (SIP) UPDATE Method</td>
</tr>
<tr>
<td></td>
<td>RFC 3455 - Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)</td>
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<td>RFC 3515 - The Session Initiation Protocol (SIP) Refer Method</td>
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<td></td>
<td>RFC 3310 - HTTP/SIP Digest Authentication Using Authentication and Key Agreement (AKA)</td>
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<tr>
<td></td>
<td>RFC 3263 - Session Initiation Protocol (SIP): Locating SIP Servers</td>
</tr>
<tr>
<td>Secure Real-time Transport Protocol (SRTP)</td>
<td>RFC 3711 - Secure Real-time Transport Protocol (SRTP)</td>
</tr>
<tr>
<td></td>
<td>RFC 3551 - Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control</td>
</tr>
<tr>
<td>Message session Relay Protocol (MSRP)</td>
<td>RFC 4975 - Message Session Relay Protocol (MSRP)</td>
</tr>
</tbody>
</table>
Generic SIP Call Flow

Registration

- REGISTER
- 200 OK

Call Initiated

- INVITE
- 100 Trying
- 180 Ringing
- 200 OK

Call Processing

- ACK

Call Established

RTP Media

- INVITE
- 100 Trying
- 180 Ringing
- 200 OK

Call Terminated

- BYE
- 200 OK

- BYE
- 200 OK
MAPS™ SIP Protocol Test Tool (Item # PKS120):
- RFC 3261 - Primary SIP standard
- RFC 3262 - PRACK
- RFC 3515 – REFER

MAPS™ SIP Conformance Suite (Item # PKS121):
- ETSI TS 102-027-2 v4.1.1 (2006-07) - 300+ scripts designed to test SIP UAs for conformance to RFC 3261.

MAPS™ SIP HD (Item # PKS109):
- Purpose built 1U appliance capable of emulating up to 32,000 SIP Endpoints.
### MAPSTM SIP Highlights

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>signaling</strong></td>
<td>• Generates and processes SIP valid and invalid messages</td>
</tr>
<tr>
<td></td>
<td>• Supports complete customization of SIP headers, call flow, and messages</td>
</tr>
<tr>
<td></td>
<td>• Supports complete customization of scripts and parameters in the profiles</td>
</tr>
<tr>
<td></td>
<td>• Each SIP message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts</td>
</tr>
<tr>
<td></td>
<td>• Supports IPv4 /IPv6 and transport over UDP and TCP, and TLS for secure transport</td>
</tr>
<tr>
<td></td>
<td>• Handles Retransmissions of messages with specific interval</td>
</tr>
<tr>
<td></td>
<td>• Scripted call generation and call reception</td>
</tr>
<tr>
<td></td>
<td>• Supports 64-bit version to enhance signalling performance</td>
</tr>
<tr>
<td></td>
<td>• Supports joining conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, and silence packets generation</td>
</tr>
<tr>
<td></td>
<td>• Ability to send &quot;reliable provisional responses&quot; and start early media actions</td>
</tr>
<tr>
<td></td>
<td>• Ability to implement IP Spoofing for any network like Class C, Class B etc</td>
</tr>
<tr>
<td></td>
<td>• Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER and INFO SIP methods</td>
</tr>
<tr>
<td><strong>Automation</strong></td>
<td>• Automation, Remote access, and Schedulers to run tests 24/7</td>
</tr>
<tr>
<td></td>
<td>• Client-server model allows users to control all features of MAPSTM through APIs</td>
</tr>
<tr>
<td></td>
<td>• Supported clients include TCL, Python, VB, Java, and .Net</td>
</tr>
</tbody>
</table>
| Traffic | • Supports various RTP traffic (PKS102) such as, digits, voice file, tones, IVR, FAX, and Video in IP networks  
• Supports almost all industry standard voice codec types - G.722, G.729, G.726, GSM, AMR, EVRC, EVS, OPUS, SMV, iLBC, SPEEX, and more. *AMR and EVRC variants require additional licenses. 
• Supports 64-bit RTP core to enhance performance - handles increased call rate of up to 3000 calls with high volume traffic. 
• Supports both G.711 Pass Through Fax Simulation (PKS200) and T.38 Fax Simulation over UDPTL (PKS211) 
• Transmit and receive pre-recorded video traces supporting video codecs like H.264, H.263, and VP8. 
• Study packet effects through impairment generation –  
  • Latency (Uniform distributed & Normal distributed)  
  • Packet loss (Periodic & Random)  
  • Packet effects (Duplicate & Out of order)  
• 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)  
• User-defined voice quality statistics for received RTP Traffic can be calculated and updated periodically during run-time to a csv file. 
• Supports simulation of SIP/MSRP User Agents end-points in an NG9-1-1 network and send and receive communications over IP networks. MSRP sessions supports simulation of IM Only Calls, Audio and IM Calls, and Video and IM Call types. |
SIP Call Types

- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX) and Global Error (6xx)
Testing UAC

Scenario: MAPS™ acting as UAS and testing UAC

- MAPS™ acting as UAS receives messages from UAC (DUT)
- DUT (UAC) generates SIP messages
Testing Proxy Server / B2B UA

Scenario: MAPS™ acting as UAS and UAC and testing Proxy.

- MAPS™ can be configured to act as UAC and UAS simultaneously so that entire Proxy testing can be automated.
MAPS™ SIP Configured as Registrant

Testing Registrar

Scenario: MAPS™ acting as Registrant and testing Registrar.

- MAPS™ can be configured to act as Registrant and to generate registration request messages to automate the entire Registrar (DUT) testing.
Testing UAS & Redirect Server

Scenario: MAPS™ testing Redirect Server and / or UAS

- MAPS™ can be configured to act as UAC & generate SIP messages
- Tests Redirect Server and /or UAS; Allows redirection of call scenarios to be automated
Testing Registrant
Scenario: MAPS™ acting as Registrar and testing Registrant

- MAPS™ acts as Registrar and processes received registration request messages from Registrant (DUT) while conforming Registrant.
- DUT (Registrant) generates REGISTRATION SIP messages.
SIP Redirect Server

- Returns the next address to originator instead of forwarding
- Originator retries with the new address
Call Generation (UAC)

- Registrant – Registers with Registrar
- Call with Auto Traffic of RTP Action
- Traffic Impairments
- Simulates IVR (Interactive Voice Response) for RTP traffic
- Call through Proxy
- Sequential and Random Generation of Calls
- Simultaneous Generation of Calls
- Load Generation (Stress Testing)
Call Reception (UAS)

- Registrar – Accepts the registration from registrant
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX), and Global Error (6xx)
End-to-End Gateway Testing

- Evaluates Gateway / ATA product features such as call connectivity, call signaling, traffic generation, voice quality testing, codec, and hundreds of other features
End-to-End Gateway Testing Call Scenario
**Test Bed Configuration**

**End User Configuration**: xml file containing one or more endpoint configurations.

**RTP Core IP Address**: IP Address of the system on which the RTP Core should be invoked.

**IP Spoofing**: permits user to assign one or more virtual IP addresses to NIC
Global Configuration

- A list of variables/values that are automatically declared and assigned at the start of any script execution

- A script may locally override the values assigned here

- A script may also ignore these variables entirely. For example, Call Duration is not a hard limit on the length of a call, it is just a variable the script may use
Each Profile Group contains one or several sub-profiles

Each sub-profile is a set of variables which together define a single SIP Endpoint

Not every field in a profile is relevant to every script execution

Profile Editor has a “Quick Config” tool to help users create multiple different sub-profiles in one shot
IP Traffic Simulation Capabilities and Performance

2000+ Simultaneous Calls (SIP + RTP Voice)
500 Simultaneous Calls (SIP + RTP Video)
500 Simultaneous Calls (SIP + IM MSRP)

MAPS™ SIP HD
Up to 100K to 200K Sustained Calls

Remote Scripting and Client Access

MAPS™ SIP

SIP / RTP / MSRP

Video (Pre-recorded)
Voice (Pre-recorded, Live)
Fax Files
Packet Impairments
Digits and Tone Generation
IM Sessions
Bulk/Manual Call Generation

PacketScan™ Analysis and Monitoring

VoIP Client 1
VoIP Client 2
VoIP Client 3
# SIP Capabilities and Performance

<table>
<thead>
<tr>
<th>Product Version</th>
<th>Max Simultaneous Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Only Signaling</td>
</tr>
<tr>
<td>MAPSTM SIP 64-bit (Core i7 with 12GB RAM)</td>
<td>30,000 Calls @ 250 CPS</td>
</tr>
<tr>
<td>MAPSTM SIP HD 64-bit (Zeon Server with 16 Processors and 64GB RAM)</td>
<td>100,000 Calls @350 CPS</td>
</tr>
</tbody>
</table>
Call Generation with Voice Traffic
Call Generation with IVR Traffic
RTP Voice Quality Measurements

- RTP based Voice Quality (MOS and R-Factor) measurement are calculated and updated periodically for the received streams
- Call quality metrics includes Listening MOS, Conversational MOS, Packet Loss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter
## Event Log

### Events

<table>
<thead>
<tr>
<th>Date/Time</th>
<th>Captured Events</th>
<th>Call Trace Id</th>
<th>Script Name</th>
<th>Script Id</th>
</tr>
</thead>
<tbody>
<tr>
<td>2015-11-15 15:11:57.064000</td>
<td>UDPR Port = 5000</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.064000</td>
<td>INVITE Sent</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.074000</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.074000</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.097000</td>
<td>ACK Sent</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.197000</td>
<td>Call Connected</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:11:57.203000</td>
<td>Sending RTP Digits</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:12:00.032000</td>
<td>RTP Digits Sent</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:12:01.363000</td>
<td>Detected Digits=1234567890aBCD</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:12:08.832000</td>
<td>BYE Sent</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
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<tr>
<td>2015-11-15 15:12:08.940000</td>
<td>200 OK to BYE Received</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
<tr>
<td>2015-11-15 15:12:08.940000</td>
<td>Call Terminated</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
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<tr>
<td>2015-11-15 15:12:08.940000</td>
<td>Inter Call Duration = 1000</td>
<td>GL-MAPS_1_186765655-4436-8172@132.168.1.203</td>
<td>SipCallControl.ghi</td>
<td>CGeProtScriptId_10_186735684-4432-4442</td>
</tr>
</tbody>
</table>
Fax Simulation over IP

- RTP G.711 Pass Through Fax Simulation (PKS200)
- T.38 Fax Simulation over UDPTL (PKS211)
Call Scenarios - Fax T.30

Call Establishment
Phase A
- Sends Calling Tone (CNG)
- Sends Answer Tone
  (Called Terminal Identification - CED)
- Sends Capabilities Data
  (Digital Identification Signal - DIS)
- Sends Configuration Data
  (Digital Command Signal - DCS)
- Sends Training Check Message (TCF)
- Sends Confirmation to Receive (CFR)

Control and Capabilities Exchange
Phase B

Phase C
- Training, Fax Message
- Multi-Page Signaling (MPS)
- Sends Message Confirmation (MCF)
- Training, Fax Message

Page Transfer

Phase D
- End of Procedure (EOP)
- Sends Message Confirmation (MCF)

End of Page & Multipage Signaling

Phase E
- Sends Disconnect (DCN )

Call Release
T.38 Fax Emulation over IP using MAPS™

Calling Terminal

Off-hook
CNG (Calling Tone) 1100 Hz Every 3 secs for 5 secs
Answer/Connect
CED (Called Terminal Identification) 2100 Hz Tone
DIS (Disconnect Identification Signal) with optional NSF & CSI
DCS (Digital Command Signal) with optional TSI
TCF (Training Check) High Speed Modulation Training

CFR (Confirmation to Receive)
Partial Page Transmission
PPS-NUL (Partial Page Sent)
MCF (Message Confirmation)
Partial Page Transmission
PPS-NUL (Partial Page Sent)
MCF (Message Confirmation)
Partial Page Transmission
PPS-EOP (Partial Page Sent)
PPR (Partial Page Request)
Partial Page Transmission
PPS-EOP (Partial Page Sent)
MCF (Message Confirmation)
DCN (Disconnect)

Called Terminal

Call Setup/Tones
Low Speed
High Speed
T.38 Fax Call in Progress and Related Events
## Call Generation with FAX Traffic

### MAPS (Message Automation Protocol Simulation) GUI Interface

The MAPS GUI interface is used to simulate and analyze call generation with FAX traffic. The interface includes various elements such as script names, profiles, and call details for each entry. The interface is designed to provide real-time monitoring and testing of FAX call sequences.

### Call Sequence Example

- **INVITE** message:
  - Time: 11:19:48.140000
  - Details: Send FAX Status

- **Main Sequence:**
  - Call Progression:
    - 1st Segment: 11:19:48.140000
    - 2nd Segment: 11:19:48.140000
    - 3rd Segment: 11:19:48.140000
    - 4th Segment: 11:19:48.140000
    - 5th Segment: 11:19:48.140000
    - 6th Segment: 11:19:48.140000

- **Response Details:**
  - 1st Response: 11:19:48.140000
  - 2nd Response: 11:19:48.140000
  - 3rd Response: 11:19:48.140000
  - 4th Response: 11:19:48.140000
  - 5th Response: 11:19:48.140000
  - 6th Response: 11:19:48.140000

- **Event Handling:**
  - Fan Status: Send FAX Status
  - VSI Signal Done
  - T3C Advanced Features
  - ECH Code Submitted in DCS
  - VSI Signal Done
  - INVITE Response
  - ACK Response
  - 33630 Frame of VSI selected after AP
  - VSI Signal Done
  - INVITE Response

### Technical Specifications

- **Software Version:** GL-MAPS_3_776120744_4007_381692_10_18_12_012
- **Network Configuration:** Stop/Fax Session Successful
- **Event Monitoring:** SIP Terminate Call

The MAPS interface is a powerful tool for simulating and testing complex call sequences, particularly in scenarios involving FAX traffic.
<table>
<thead>
<tr>
<th>Date/Time</th>
<th>Captured Events</th>
<th>Call Trace Id</th>
<th>Script Name</th>
<th>Script Id</th>
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<tbody>
<tr>
<td>2015-1-15 15:27:08.544000</td>
<td>UDP Port = 5060</td>
<td></td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
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<td>2015-1-15 15:27:08.673000</td>
<td>Call Connected</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.678000</td>
<td>Sending RTP Fax</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
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<tr>
<td>2015-1-15 15:27:32.397000</td>
<td>RTP Fax Sent</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
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<tr>
<td>2015-1-15 15:27:39.394000</td>
<td>200 Ok to BYE Received</td>
<td></td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:39.394000</td>
<td>Call Terminated</td>
<td>GL-MAPS_1_187697169-4528-8320@192.168.1.203</td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
</tr>
<tr>
<td>2015-1-15 15:27:39.394000</td>
<td>Inter Call Duration = 1000</td>
<td></td>
<td>SIP-Protocol.gls</td>
<td>CGProtScriptId_14_187697168-4524-6432</td>
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### File Traffic Events

<table>
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<tr>
<th>Date/Time</th>
<th>Event Type</th>
<th>Captured Events</th>
<th>Call Trace Id</th>
<th>Script Name</th>
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<tr>
<td>2015-1-15 15:32:20.946000</td>
<td>UDP Port = 5060</td>
<td></td>
<td></td>
<td>SIP-Prollo</td>
</tr>
<tr>
<td>2015-1-15 15:32:20.946000</td>
<td>INVITE Sent</td>
<td></td>
<td></td>
<td>SIP-Prollo</td>
</tr>
<tr>
<td>2015-1-15 15:32:20.958000</td>
<td>PROGRESS Received</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:20.958000</td>
<td>PROGRESS Received</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:20.958000</td>
<td>PROGRESS Received</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:21.073000</td>
<td>Call Connected</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:21.074000</td>
<td>RxFile = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_3.glw</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:21.076000</td>
<td>Receiving RTP File</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:21.076000</td>
<td>Sending RTP File</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:41.093000</td>
<td>RTP File Received</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:32:41.093000</td>
<td>RTP File Sent</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:33:21.091000</td>
<td>200 Ok to BYE Received</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:33:21.091000</td>
<td>Call Terminated</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
<tr>
<td>2015-1-15 15:33:21.091000</td>
<td>Inter Call Duration = 1000</td>
<td></td>
<td>GL-MAPS_1_188009580:4552-5476@192.168.1.203</td>
<td>SipCallCl</td>
</tr>
</tbody>
</table>
Transmit pre-recorded video traces with video codecs like H.264, and H.263
Speech to Text Interactive Voice Response (IVR)

- MAPS™ SIP with GL’s Speech Transcription Server provides automated IVR testing by using speech to text to navigate through an IVR tree. IVR prompts are recorded by MAPS™ SIP and transcribed by the Speech Transcription Server.
- Transcribed text is compared to an expected text at each IVR stage to confirm the prompt. Once the IVR prompt is confirmed, MAPS™ sends DTMF or voice-based responses to move to the next stage.
- The expected IVR prompts and responses are defined by the customer to ensure completely customizable tests that are suitable for all IVR systems.

[Diagram of IVR testing process]
GL’s Interactive Voice Response Scenario

- The CSV file in the screenshot below shows a basic IVR traversal test of this IVR system.

<table>
<thead>
<tr>
<th>A</th>
<th>B</th>
<th>C</th>
<th>D</th>
<th>E</th>
<th>F</th>
<th>G</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>IVRIndex</td>
<td>IVRPromptLanguage</td>
<td>IVRExpectedTranscript</td>
<td>VRResponseType</td>
<td>VRResponseDTMF</td>
<td>VRResponseSpeech</td>
</tr>
<tr>
<td>1</td>
<td>int</td>
<td>string</td>
<td>string</td>
<td>string</td>
<td>string</td>
<td>string</td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
<td>Welcome to GL Communications. If you know your party's extension, you can dial it at any time. For sales, press 1; for technical support, press 2; for a directory, press 3.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>en-US</td>
<td>Welcome to GL Communications. If you know your party's extension, you can dial it at any time. For sales, press 1; for technical support, press 2; for a directory, press 3.</td>
<td>DTMF</td>
<td></td>
<td>string</td>
</tr>
<tr>
<td>4</td>
<td>2</td>
<td>en-US</td>
<td>Welcome to the directory. Please enter the first 3 letters of your party's last name using your touch tone keypad. Use the seven key for q and the nine key for z.</td>
<td>DTMF</td>
<td>926</td>
<td>string</td>
</tr>
</tbody>
</table>
IVR Call Simulation
IVR Call Simulation Reports

SIP IVR Detailed Log

SIP IVR Result Log
Message Session Relay Protocol

Message Session Relay Protocol is a text-based, connection-oriented protocol for transmitting a series of related instant messages in the context of a session. MSRP sessions are typically arranged using SIP the same way a session of audio or video media is set up.
Message Session Relay Protocol (Contd.)

- Reads text messages from a pre-defined text file (user-configurable) and transmits them on established IM session
- Received messages on every MSRP session can be recorded to a text file
- Text file can have multiple lines of message. The CRLF will be the de-limiter to treat each line as a new message
- Supports message chunking with user configured chunk size
- Configuration options allow to –
  - Record and report success and failure reports in MSRP SEND method
  - Define message generation interval to control the message frequency on the call
- Supports mixed media SIP sessions i.e., Audio with IM / Video with IM / Only IM
- Provides IM statistics per call and aggregated statistics of over-all calls (Number and size of messages received and sent)
- Flexibility to validate MSRP devices through negative tests with invalid MSRP URI's, validate success and failure reports
- Supports up to 500 simultaneous MSRP sessions
MSRP Traffic Configuration
### MSRP Call Generation

![MSRP Call Generation](image)

#### MSRP Call Generation Details

<table>
<thead>
<tr>
<th>Script Name</th>
<th>Call ID</th>
<th>Script Execution</th>
<th>Status</th>
<th>Events</th>
<th>Error</th>
<th>Result</th>
<th>Completed</th>
</tr>
</thead>
<tbody>
<tr>
<td>MyScript</td>
<td>1000000</td>
<td>1</td>
<td>Pass</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>MyScript2</td>
<td>1000001</td>
<td>2</td>
<td>Pass</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
</tbody>
</table>

---

**MSRP Protocol Simulation**

MSRP (Message Signal Part Protocol Simulation) is a protocol simulation and conformance test tool that supports a variety of protocols. It is designed to simulate the behavior of MSRP and other similar protocols, allowing developers to test and validate their implementations.

---

**Message Sequence and Event Flow**

The sequence diagram illustrates the flow of messages and events in a MSRP call. Each message type (e.g., INVITE, 100 Trying) is shown with its corresponding timestamp, indicating when it was sent or received. The diagram covers various error codes and status codes, demonstrating the protocol's robustness and reliability.

---

**Sample Call Trace**

```
MSRP Call Trace:

```

---

**Further Information**

For more detailed information on MSRP and its applications, refer to the official documentation and resources available online. GL Communications offers comprehensive support and training for developers working with MSRP and related protocols.
### MSRP Statistics

<table>
<thead>
<tr>
<th>Name</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Total MSRP Messages Sent</td>
<td>340</td>
</tr>
<tr>
<td>Total MSRP Messages Received</td>
<td>345</td>
</tr>
<tr>
<td>Total MSRP Message Bytes Sent</td>
<td>15285</td>
</tr>
<tr>
<td>Total MSRP Message Bytes Received</td>
<td>15285</td>
</tr>
</tbody>
</table>
Load Generation

- Ramp Statistical Distribution
- Step Statistical Distribution
- Saw-tooth Statistical Distribution
Success Call Ratio Statistics

Call Graph

Call Stats
# Message Statistics

<table>
<thead>
<tr>
<th>Message Type</th>
<th>Tx Count</th>
<th>Rx Count</th>
<th>Retransmit Count</th>
</tr>
</thead>
<tbody>
<tr>
<td>100 INVITE</td>
<td>0</td>
<td>66040</td>
<td>0</td>
</tr>
<tr>
<td>180 INVITE</td>
<td>0</td>
<td>66040</td>
<td>0</td>
</tr>
<tr>
<td>200 BYE</td>
<td>0</td>
<td>46808</td>
<td>0</td>
</tr>
<tr>
<td>200 INVITE</td>
<td>0</td>
<td>66040</td>
<td>0</td>
</tr>
<tr>
<td>ACK</td>
<td>66040</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>BYE</td>
<td>46808</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>INVITE</td>
<td>66040</td>
<td>0</td>
<td>0</td>
</tr>
</tbody>
</table>
SIP RTP Analyzer - PacketScan™
PacketScan™ VoIP Traffic Analysis
SIP / MSRP / H323 / MEGACO / MGCP / RTP / RTCP / Video Analysis
What the software does?

- Captures, segregates, and monitors packets; perform voice quality testing in real-time over VoIP network

- Unlimited traffic and signaling capturing capability; captured VoIP calls with video can be played back using 3rd party applications

- Can be deployed as a Probe for a centralized monitoring system with Oracle database.

PacketScan™ Analyzer with SIP CDR
**SIP Decode in PacketScan™**

![PacketScan UI](image)

<table>
<thead>
<tr>
<th>Dev</th>
<th>Format</th>
<th>TIME (Relative)</th>
<th>Len</th>
<th>Error</th>
<th>Protocol</th>
<th>IPv4</th>
<th>Source IP Address</th>
<th>Destination IP Address</th>
<th>UDP Source Port</th>
<th>UDP Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>✓</td>
<td>✓</td>
<td>00:00:00:00:0000</td>
<td>6666</td>
<td></td>
<td>Internet (IPv4)</td>
<td>SIP</td>
<td>192.168.120.100</td>
<td>192.168.1103</td>
<td>5060</td>
<td>5060</td>
</tr>
<tr>
<td>✓</td>
<td>✓</td>
<td>00:00:00:00:0062</td>
<td>664</td>
<td></td>
<td>Internet (IPv4)</td>
<td>SIP</td>
<td>192.168.1103</td>
<td>192.168.120.200</td>
<td>5060</td>
<td>5060</td>
</tr>
<tr>
<td>✓</td>
<td>✓</td>
<td>00:00:00:00:0069</td>
<td>665</td>
<td></td>
<td>Internet (IPv4)</td>
<td>SIP</td>
<td>192.168.1103</td>
<td>192.168.120.200</td>
<td>5060</td>
<td>5060</td>
</tr>
<tr>
<td>✓</td>
<td>✓</td>
<td>00:00:00:00:0093</td>
<td>699</td>
<td></td>
<td>Internet (IPv4)</td>
<td>SIP</td>
<td>192.168.1103</td>
<td>192.168.120.200</td>
<td>5060</td>
<td>5060</td>
</tr>
</tbody>
</table>

---

**SIP Message Example**

**INVITE sip:0001@192.168.1.103 SIP/2.0**

**Via**: sip:0001@192.168.1.200;branch=z9hG4kX81133536-3

**Max-Forwards**: 70

**Allow**: INVITE, CANCEL, ACK, OPTIONS, NOTIFY, SUBSCRIBE

**From**: sip:0001@192.168.1.103;tag=GLFG_81133536-333

**To**: sip:0001@192.168.1.103

**Call-ID**: GLFG-813638760351

**CSeq**: 1 INVITE

**Contact**: sip:0001@192.168.1.103

**Content-Type**: application/sdp

**Content-Length**: 349

**s=0**

**sip:0001 47706129 IN IP4 192.168.1.200**

**t=0 0**

**a=mid IP4 192.168.1.200**

**a=control RTP/AVP 8 10 100 3 101**

**a=rtpmap:0 PCMU/8000**

**a=rtpmap:8 PCMA/8000**

**a=rtpmap:14 G722/8000**

**a=ptime:10**

**a=fmtp:18 anneexs=0**

**a=rtpmap:104 G728-32/8000**

**a=rtpmap:9 G723/8000**

**a=rtpmap:101 telephone-event/8000**

**a=ptime:10**

**a=sendrecv**

---

**Offline Viewing**: C:\Program Files\GL Communications Inc\P2.555 Frames
PacketScan™ PDA with SIP Call Summary
PacketScan™ Fax T.38 Analysis
MAPS™ Command Line Interface

- MAPS™ can be configured as server-side application, to enable remote controlling through multiple command-line based clients. Supported clients include Java, VBScripts, TCL, Python and others.

- The MAPS™ APIs allows for programmatic and automated control over all MAPS™ platforms. Each MAPS™ server can receive multiple client connections and offer independent execution to each client.

- Likewise, a single client can connect to multiple MAPS™ servers, including servers running different protocols, permitting complex cross-protocol test cases.
MAPS™ SIP CLI Test System

- As depicted in the figure above, MAPS™ SIP CLI test system consists of the following -
  - TCL user communicating over TCP/IP
  - MAPS™ Client IFC, and MAPS™ SIP CLI Server
MAPSTM CLI Server and Python Client

Python 3.7.3 (v3.7.3:af28fd1d, Mar 25 2019, 22:12:05) [MSC v.1916 64 bit (AMD64)] on win32
Type "help", "copyright", "credits" or "license()" for more information.
>>
RESTART: C:\Program Files\GL Communications Inc\MAPSTM\PythonClient\examples\3IP\3IPBasicCall.py
SERVICES_INITIALIZED
CONNECTED
Negotiated Codec = PCMU
0
<sdp> = 4.18531
LENGTH = 4.18531
/audio factor = 43
LG FACTOR = 93
TX PACKETS = 505
RX PACKETS = 792
LOST PACKETS = 0
DISCARDED PACKETS = 0
OUT OF SEQ PACKETS = 0
MEGACALL PACKETS = 0
APP JITTER = 0.385

1212401010 -> INVITE
INVITE sip:00048190.166.12.209 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.124:5060;branch=z9hG4bK-4-114832888-21704-17172
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, PUBLISH
From: sip:1234567890@192.168.12.210;tag=FromEng-1-304812930-12701-17872
To: sip:00048190.166.12.209 SIP/2.0
Call-ID: GL-MAPS-3-114832888-21704-17872@192.168.12.216
CSeq: 1 INVITE
Contact: sip:1234567890@192.168.12.210;transport=udp
Content-Type: application/sdp
Content-Length: 269

v=0
o=-0 3029132000 39377840 2 IN IP4 192.168.12.216
s=3IP Call
i=IP4 192.168.12.216
m=audio 1024 RTP/AVP 0 101
a=rtpmap:18 0725/8000
a=rtpmap:10 0725/8000

• Multiple PacketScan™ probes can be used for network monitoring, with call detail reports exported to a central database
• Results can be accessed remotely using NetSurveyorWeb™, a simple web browser based application
NetSurveyorWeb™ – Reports
Thank you