MAPS™ SIP
SIP + RTP + MSRP Simulation
MAPS™ SIP

- **P-CSCF** - Proxy - Call Session Control Function
- **S-CSCF** - Serving - Call Session Control Function
- **MGCF** – Media Gateway Control Function

**MAPS™ SIP**

- Normal RTP Traffic Generation (2000 simultaneous calls per NIC card)
- HD RTP Traffic Generator (w/ 4 x 1G cards, w/ 2x 10G cards)
  - 20,000 Simultaneous Calls (with RTP Traffic)
SIP Architecture and Entities

1. SIP User Agent (Caller) sends a request to SIP Proxy.
2. SIP Proxy redirects the request to SIP Redirect Server.
3. SIP Redirect Server forwards the request to SIP Proxy.
4. SIP Proxy relays the request to SIP User Agent (Called).
5. SIP User Agent (Called) responds to SIP Proxy.
6. SIP Proxy redirects the response to SIP Redirect Server.
7. SIP Redirect Server forwards the response to SIP Proxy.
8. SIP Proxy relays the response to SIP User Agent (Caller).
9. SIP User Agent (Caller) receives the response.

Legend:
- Requests
- Responses
- Non-SIP Protocol
SIP Protocol Stack

<table>
<thead>
<tr>
<th>RTCP</th>
<th>Audio / Video Codecs</th>
<th>MSRP</th>
<th>SDP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>RTP</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SRTP</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>UDP</td>
<td>TLS</td>
<td>SCTP</td>
</tr>
<tr>
<td></td>
<td>TCP</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Supported Protocols**

<table>
<thead>
<tr>
<th>Protocol</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>
</tr>
<tr>
<td></td>
<td>RFC 3515 - The Session Initiation Protocol (SIP) Refer Method</td>
</tr>
<tr>
<td></td>
<td>RFC 3310 - HTTP/SIP Digest Authentication Using Authentication and Key Agreement (AKA)</td>
</tr>
<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

1. **Registration**
   - User Agent A REGISTER SIP Proxy/Registrar/Redirect Server
   - SIP Proxy/Registrar/Redirect Server 200 OK
   - User Agent B REGISTER SIP Proxy/Registrar/Redirect Server
   - SIP Proxy/Registrar/Redirect Server 200 OK

2. **Call Initiated**
   - User Agent A INVITE User Agent B
   - User Agent B 100 Trying
   - User Agent A 180 Ringing
   - User Agent B 200 OK
   - User Agent A ACK
   - User Agent B ACK

3. **Call Processing**
   - User Agent A INVITE User Agent B
   - User Agent B 100 Trying
   - User Agent A 180 Ringing
   - User Agent B 200 OK
   - User Agent A ACK
   - User Agent B ACK

4. **Call Established**
   - User Agent A BYE
   - User Agent B 200 OK
   - User Agent A 200 OK
   - User Agent B BYE

5. **RTP Media**

---

GL Communications Inc.
About MAPS™ SIP

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 20,000 SIP Endpoints.
<table>
<thead>
<tr>
<th>Signaling</th>
<th>Automation</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Generates and processes SIP valid and invalid messages.</td>
<td>• Automation, Remote access, and Schedulers to run tests 24/7</td>
</tr>
<tr>
<td>• Supports complete customization of SIP headers, call flow, and messages.</td>
<td>• Client-server model allows users to control all features of MAPS™ through APIs</td>
</tr>
<tr>
<td>• Supports complete customization of scripts and parameters in the profiles</td>
<td>• Supported clients include TCL, Python, VB, Java, and .Net.</td>
</tr>
<tr>
<td>• Each SIP message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts.</td>
<td></td>
</tr>
<tr>
<td>• Supports IPv4 /IPv6 and transport over UDP and TCP, and TLS for secure transport.</td>
<td></td>
</tr>
<tr>
<td>• Handles Retransmissions of messages with specific interval.</td>
<td></td>
</tr>
<tr>
<td>• Scripted call generation and call reception.</td>
<td></td>
</tr>
<tr>
<td>• Supports 64-bit version to enhance signalling performance.</td>
<td></td>
</tr>
<tr>
<td>• Supports conference call, blind call transfer hold, auto call rejection, and silence packets generation.</td>
<td></td>
</tr>
<tr>
<td>• Ability to send &quot;reliable provisional responses&quot; and start early media actions.</td>
<td></td>
</tr>
<tr>
<td>• Supports VoIP implementation as per ED-137B of EUROCAE standard.</td>
<td></td>
</tr>
<tr>
<td>• Ability to implement IP Spoofing for any network like Class C, Class B etc.</td>
<td></td>
</tr>
</tbody>
</table>
### MAPS™ SIP Highlights

**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)
Scenario: MAPS™ acting as UAS and testing UAC

- MAPS™ acting as UAS receives messages from UAC (DUT)
- DUT (UAC) generates SIP messages
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
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.
MAPS™ SIP Configured as UAC

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
MAPS™ SIP Configured as Registrar

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

- UE
- IP
- PSTN
- Gateway
- SIP
- 2 Wire
- FXO

- Invite
- 100 Trying
- 180 Ringing
- 180 Ringing
- 200 OK
- ACK
- Send Voice File
- Bye
- 200 OK

- Ringing
- Off-Hook
- Record Voice files / VQT
- On-Hook

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MAPS™ SIP HD Network Configurations

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Test Bed Setup

**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

**Enable CSV Profile**: permits user to provide CSV file name from which profile values to be taken.
Test Bed Setup – HD RTP Media Configuration

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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.
User Agents Configuration

- 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

MAPSTM SIP

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

MAPSTM SIP HD

Up to 100K to 200K Sustained Calls

Remote Scripting and Client Access

SIP / RTP/ MSRP

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

PacketScanSTM Analysis and Monitoring

GL Communications Inc.
## 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>MAPS™ SIP 64-bit (Core i7 with 12GB RAM)</td>
<td>30,000 Calls @ 250 CPS</td>
</tr>
<tr>
<td>MAPS™ 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 IVR Traffic

[Image of a software interface showing a network diagram with labels such as INVITE, 100 Ringing, 200 OK, and an example INVITE message with SIP URI and headers.]
• 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.
<table>
<thead>
<tr>
<th>Event Log</th>
<th>Error Events</th>
<th>Captured Errors</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Date/Time</strong></td>
<td><strong>Captured Events</strong></td>
<td><strong>Call Trace Id</strong></td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:072000</td>
<td>PROGRESS Received</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:072000</td>
<td>PROGRESS Received</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:072000</td>
<td>PROGRESS Received</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:192000</td>
<td>Call Connected</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:202000</td>
<td>Sending RTP Digits</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:302000</td>
<td>200 OK to BYE Received</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:302000</td>
<td>Call Terminated</td>
</tr>
<tr>
<td>2015-1-15</td>
<td>15:15:57:302000</td>
<td>Inter Call Duration = 1000</td>
</tr>
</tbody>
</table>
Fax Simulation over IP

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

GL’s Fax Call Simulators & Analyzers (TDM, IP, Wireless)
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)

Control and Capabilities Exchange
Phase B

- Sends Confirmation to Receive (CFR)

Phase C

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

Page Transfer

End of Page & Multipage Signaling

Phase D

- End of Procedure (EOP)

End of Call Release

Phase E

- Sends Message Confirmation (MCF)
- Sends Disconnect (DCN)
T.38 Fax Emulation over IP using MAPS™

GL Communications Inc.
T.38 Fax Call in Progress and Related Events
Call Generation with FAX Traffic
## FAX Traffic Events

### Table of 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>2015-1-15 15:27:08.544000</td>
<td>UDP Port = 5060</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.544000</td>
<td>INVITE Sent</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
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<tr>
<td>2015-1-15 15:27:08.556000</td>
<td>PROGRESS Received</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.673000</td>
<td>ACK Sent</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:08.673000</td>
<td>Call Connected</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:09.679000</td>
<td>Sending RTP Fax</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:32.397000</td>
<td>RTP Fax Sent</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:32.397000</td>
<td>BYE Sent</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:32.397000</td>
<td>200 OK to BYE Received</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
<tr>
<td>2015-1-15 15:27:39.394000</td>
<td>Call Terminated</td>
<td>GL-MAP3_1_18765716945285320@192.168.1.203</td>
<td>SIP:Protocol.css</td>
<td>CBPotScriptId_14_18769716845246432</td>
</tr>
</tbody>
</table>

### Event Log

- **Clear**
- **Capture Events to file**: [ ]

**Inter Call Duration = 1000**
### File Traffic Events

#### Event Log

<table>
<thead>
<tr>
<th>Date/Time</th>
<th>Event Type</th>
<th>Description</th>
<th>Call Trace Id</th>
<th>Script Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>UDP Port = 5060</td>
<td>UDP Port = 5060</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>INVITE Sent</td>
<td>INVITE Sent</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>PROGRESS Received</td>
<td>PROGRESS Received</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>ACK Sent</td>
<td>ACK Sent</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>Call Connected</td>
<td>Call Connected</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>RxFilename = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_9.jpg</td>
<td>RxFilename = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_9.jpg</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>Receiving FTP File</td>
<td>Receiving FTP File</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>Sending FTP File</td>
<td>Sending FTP File</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>FTP File Received</td>
<td>FTP File Received</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>FTP File Sent</td>
<td>FTP File Sent</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>BYE Sent</td>
<td>BYE Sent</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
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<td>2015-1-15 03:22:30.948000</td>
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<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>Call Terminated</td>
<td>Call Terminated</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
<tr>
<td>2015-1-15 03:22:30.948000</td>
<td>Inter Call Duration = 1000</td>
<td>Inter Call Duration = 1000</td>
<td>GL-MAPS_1-108009580.4552-5476@152.158.1.203</td>
<td>SIP-PROF</td>
</tr>
</tbody>
</table>
Transmit pre-recorded video traces with video codecs like H.264, and H.263
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.
MSRP

• 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

<table>
<thead>
<tr>
<th>Sr No</th>
<th>Script Name</th>
<th>Protocol</th>
<th>CallID</th>
<th>Script Execution</th>
<th>Status</th>
<th>Events</th>
<th>Ex</th>
<th>Result</th>
<th>Total Failures</th>
<th>Completed Failures</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>ScriptCallControl</td>
<td>Prelink</td>
<td>Call ID</td>
<td>Step</td>
<td>Call Connected</td>
<td>SP_TerminateCall</td>
<td>Pass</td>
<td>1</td>
<td>0</td>
<td></td>
</tr>
<tr>
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<td>Call ID</td>
<td>Step</td>
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<td>SP_TerminateCall</td>
<td>Pass</td>
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<td></td>
</tr>
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<td>Step</td>
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<td>Call ID</td>
<td>Step</td>
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<td>SP_TerminateCall</td>
<td>Pass</td>
<td>1</td>
<td>0</td>
<td></td>
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<td>6</td>
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<td>Step</td>
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<td>Call ID</td>
<td>Step</td>
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<td>SP_TerminateCall</td>
<td>Pass</td>
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<td>0</td>
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<td>8</td>
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<td>SP_TerminateCall</td>
<td>Pass</td>
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<td>9</td>
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<td>Step</td>
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<td>SP_TerminateCall</td>
<td>Pass</td>
<td>1</td>
<td>0</td>
<td></td>
</tr>
</tbody>
</table>

GL’s Message Automation & Protocol Simulation (MASP®) is a protocol simulation and conformance test tool that supports a variety of...
### MSRP Statistics

![User Defined Statistics - User_DEFINED_Statistics](image)

<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>

**Note:** The table above shows the statistics for MSRP messages sent and received, including the total number of messages and bytes sent and received.
Load Generation

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

Call Graph

Call Stats

GL Communications Inc.
SIP RTP Analyzer - PacketScan™
PacketScan™ VoIP Traffic Analysis
SIP / H323 / MEGACO / MGCP / RTP / RTCP / Video Analysis

GL Communications Inc.
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

SIP Decode in PacketScan™

```
-------------- sip3261 Layer --------------

HDR
HDR
HDR
HDR
HDR
HDR
HDR
HDR
HDR

BODY
BODY
BODY
BODY
BODY
BODY
BODY
BODY
BODY
BODY

INVITE sip 0001:0192.168.1.103 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.200;branch=z9hG4xZ8511333536-3
Max-Forwards: 70
Allow: INVITE, CANCEL, ACK, INFO, PRACK, COME, OPTIONS, SUBSCRIBE
From: 0001 <sip:0001@192.168.1.200;tag=GLPC_3811333536-333>
To: 0001 <sip:0001@192.168.1.103>
Call-ID: GLPC-489633760331
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 349

v=0
o=0001 47706128 47706129 IN IP4 192.168.1.200
s=-
e=-
c=IN IP4 192.168.1.200
t=0 0
m=audio 1024 RTP/AVP 0 8 104 9 101
a=rtcpmap:0 PCMU/8000/1
a=rtcpmap:8 PCMA/8000/1
a=rtcpmap:18 annexb/0
a=rtcpmap:18 annexb/0
a=rtcpmap:104 G726-32/8000/1
a=rtcpmap:101 telephone-event/8000
a=ptime:20
a=xmsrrev
```
PacketScan™ PDA with SIP Call Summary
PacketScan™ Fax T.38 Analysis
Multiple PacketScan™ probes can be used for network monitoring, with call detail reports exported to an central database.

Results can be accessed remotely using NetSurveyorWeb™, a simple web browser based application.
Thank you!