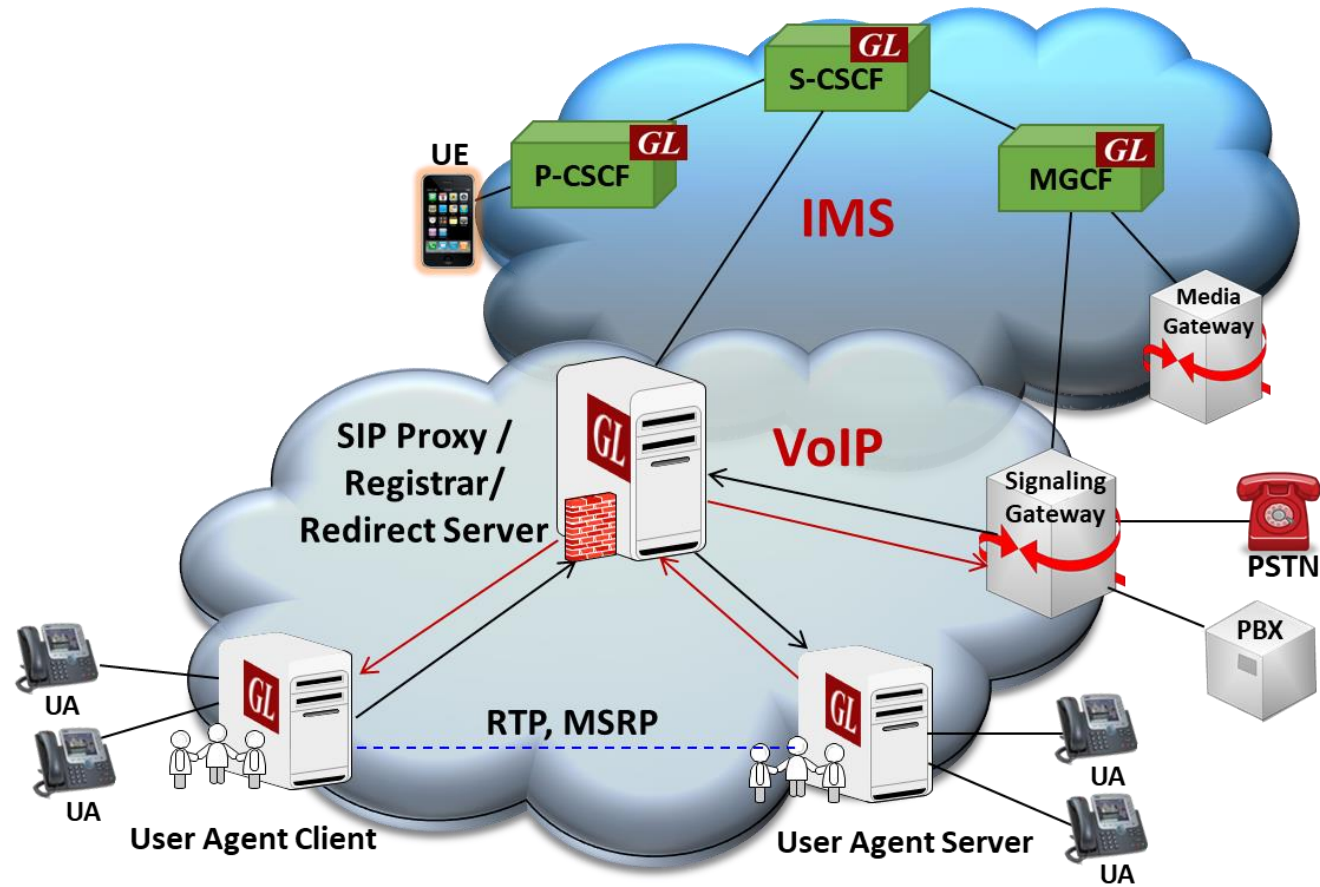

MAPS™ SIP

SIP + RTP + MSRP Simulation



818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878
Phone: (301) 670-4784 Fax: (301) 670-9187 Email: info@gl.com
Website: <http://www.gl.com>

MAPS™ SIP

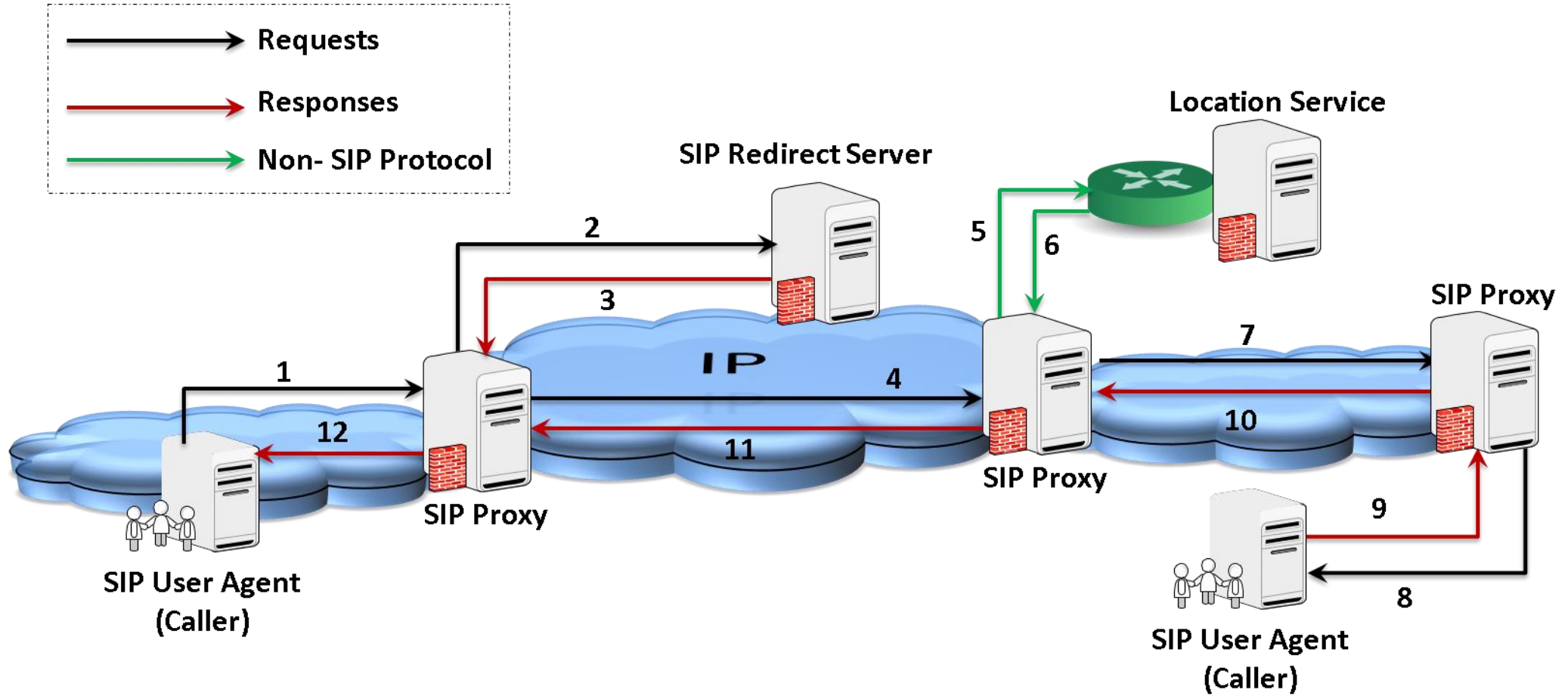


MAPS™ SIP with RTP Traffic Generation
(2000 simultaneous calls)
MAPS™ SIP Conformance

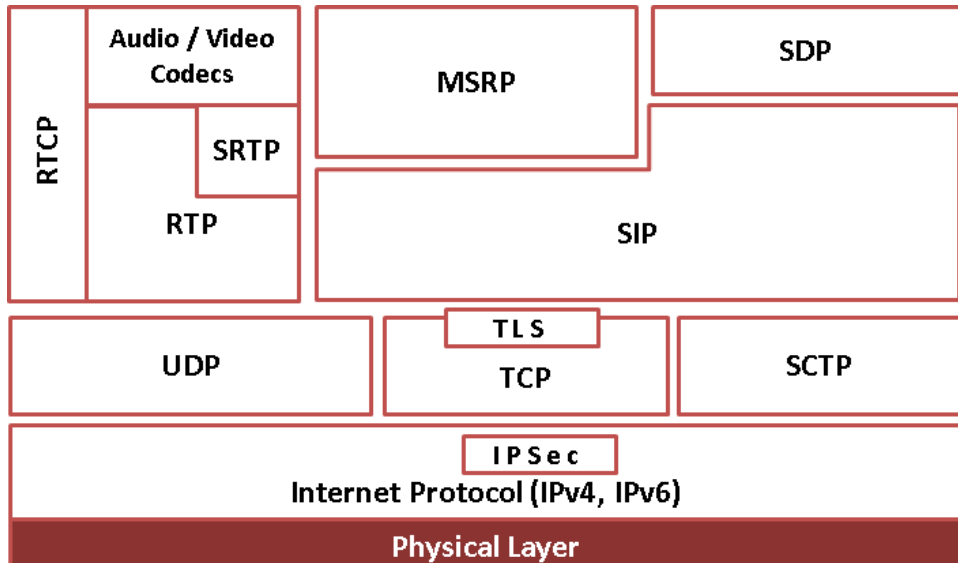


MAPS™ SIP (w/ 8 x 1Gbps Ethernet Ports)
HD RTP Traffic Generator
64,000 Simultaneous Calls (with RTP Traffic)

SIP Architecture and Entities

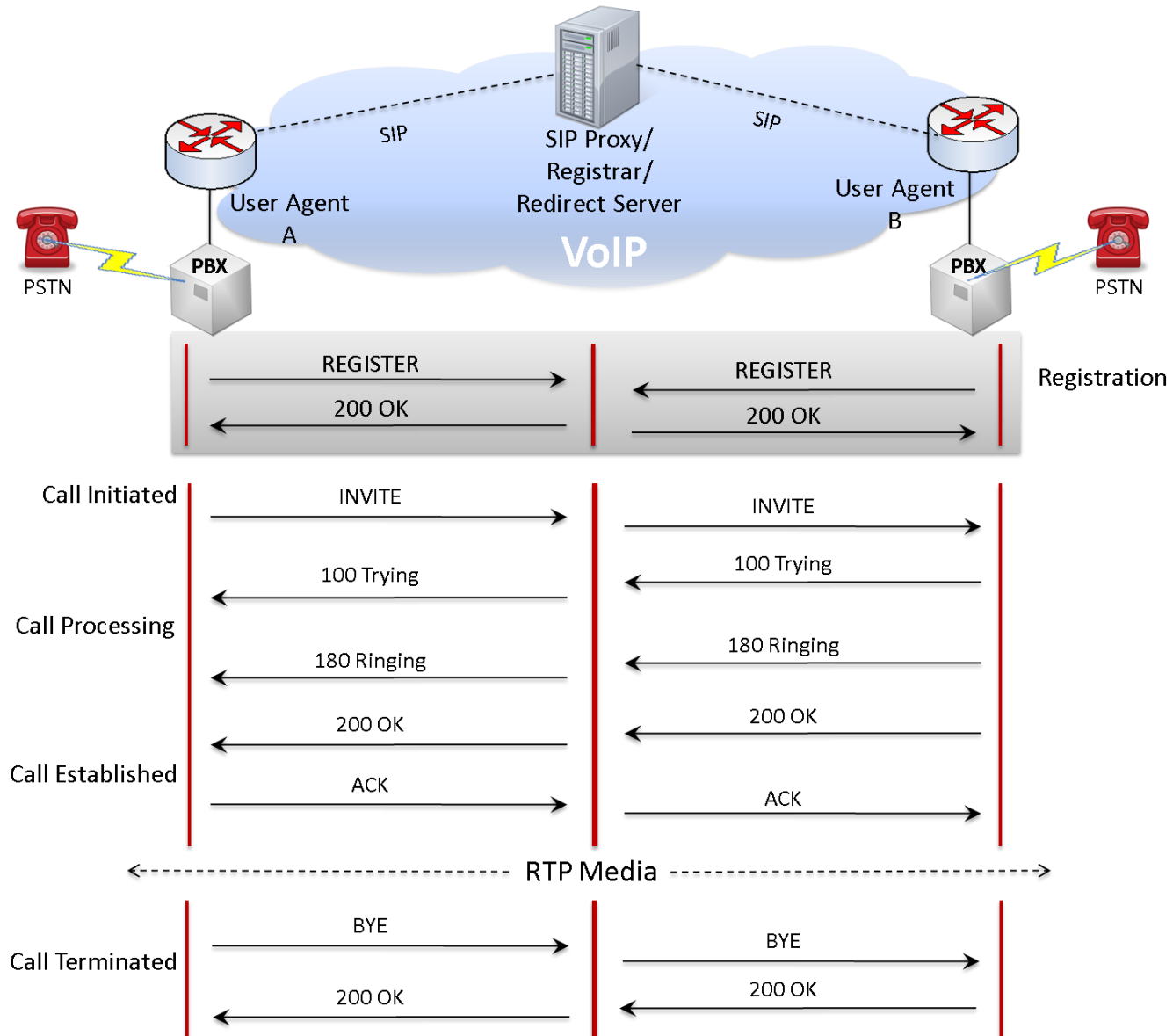


SIP Protocol Stack



Supported Protocols	Standard / Specification Used
SIP SIP Conformance	RFC 3261 ETSI TS 102-027-2 v4.1.1
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 - The 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)
Message session Relay Protocol (MSRP)	RFC 4975 - Message Session Relay Protocol (MSRP)

Generic SIP Call Flow



MAPS™ SIP Variants

MAPS™ SIP Software with Notebook PC

MAPS™ SIP Protocol Test Tool (Item # PKS120):

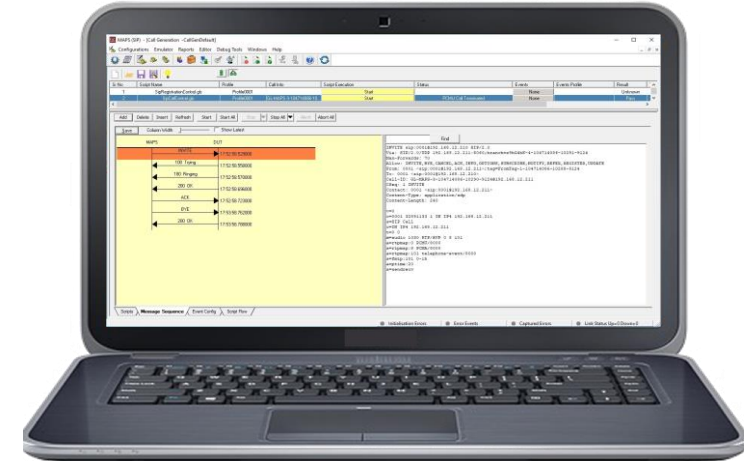
- RFC 3261 - Primary SIP standard
- RFC 3262 - PRACK
- RFC 3515 - REFER

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):

- MAPS™ SIP HD emulates up to 64,000 simultaneous calls using 8 Gigabit Ethernet ports
- Available in Portable or Rack-mount form factors



MAPS™ SIP HD



8x1GigE

High Performance Smart NIC

MAPS™ SIP Highlights

signaling	<ul style="list-style-type: none">• Generates and processes SIP valid and invalid messages• Supports complete customization of SIP headers, call flow, and messages• Supports complete customization of scripts and parameters in the profiles• 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 64-bit version to enhance signalling performance• Supports joining conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, and silence packets generation• Ability to send "reliable provisional responses" and start early media actions• Ability to implement IP Spoofing for any network like Class C, Class B etc• Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER and INFO SIP methods
Automation	<ul style="list-style-type: none">• Automation, Remote access, and Schedulers to run tests 24/7• Client-server model allows users to control all features of MAPS™ through APIs• Supported clients include Python and Java

MAPS™ SIP Highlights (Contd.)

Traffic	<ul style="list-style-type: none">• 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 –<ul style="list-style-type: none">• 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

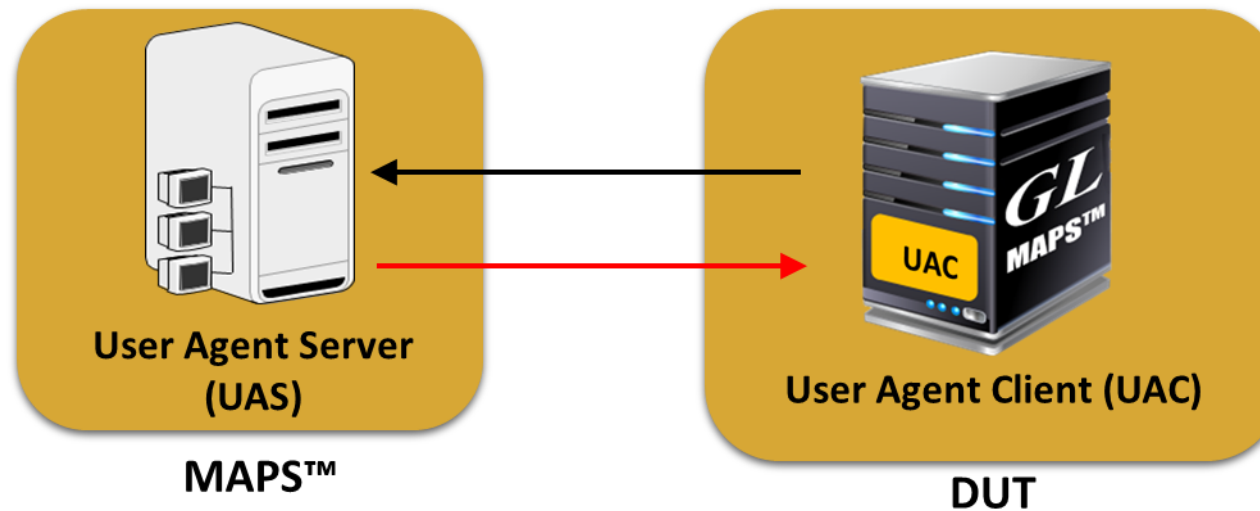
- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX)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

MAPS™ SIP Configured as UAS

Testing UAC

Scenario: MAPS™ acting as UAS and testing UAC

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

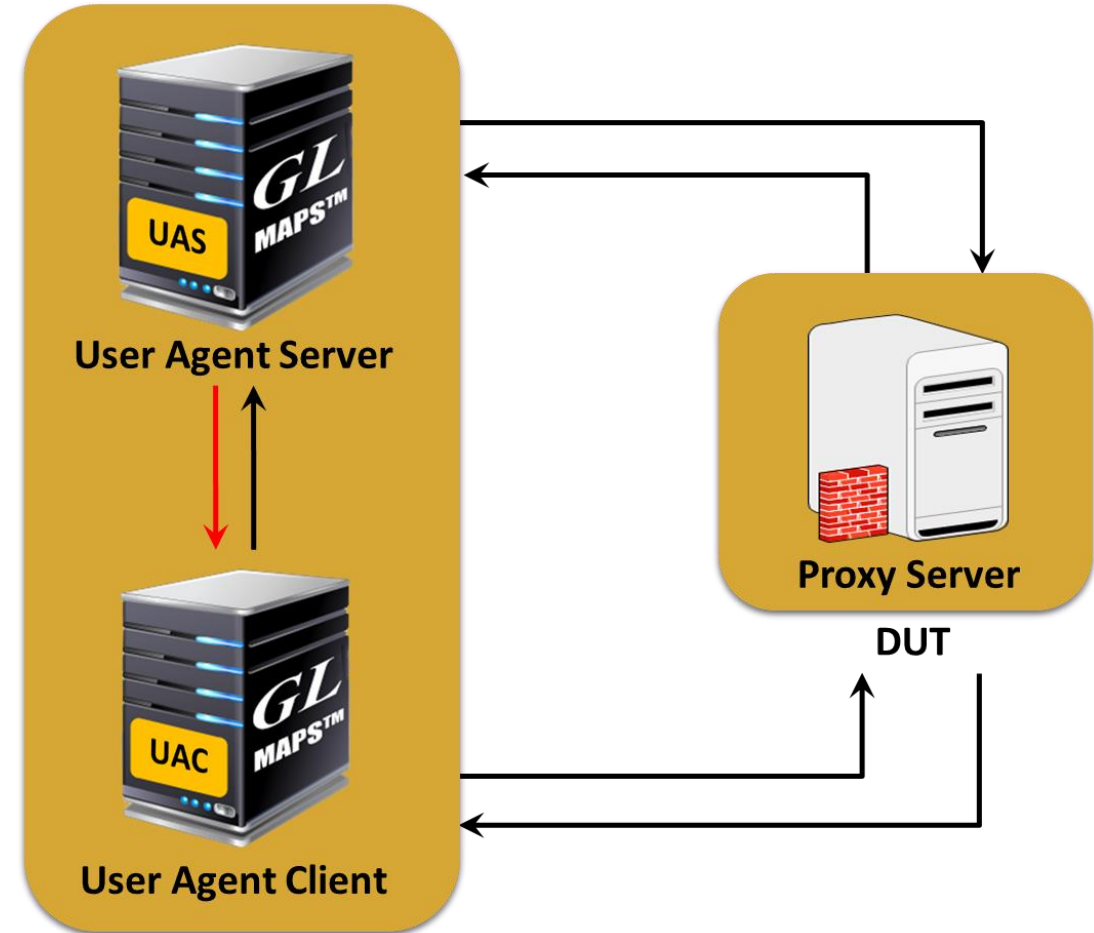


MAPS™ SIP Configured as UAC / UAS

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

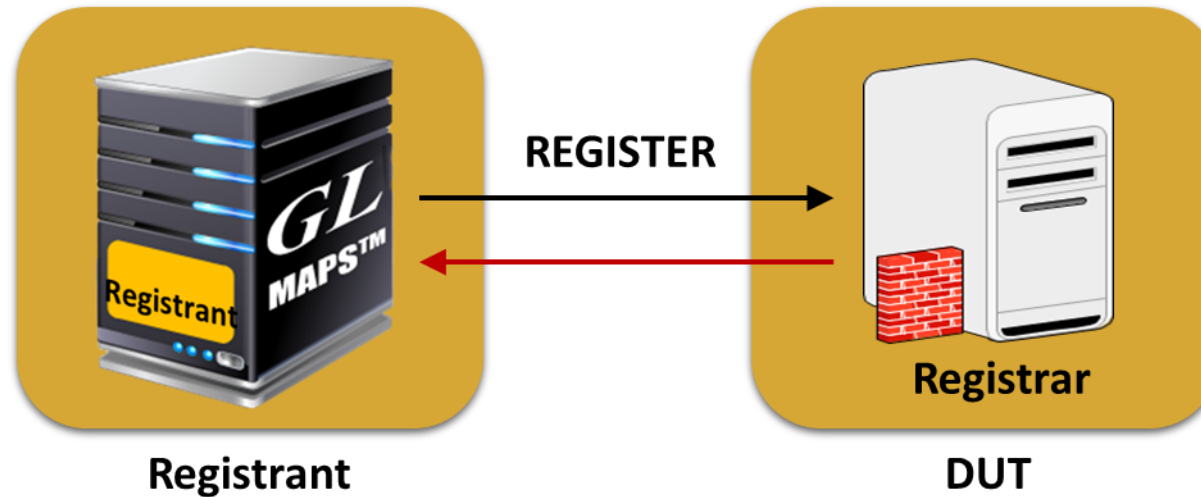


MAPS™ SIP Configured as Registrant

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

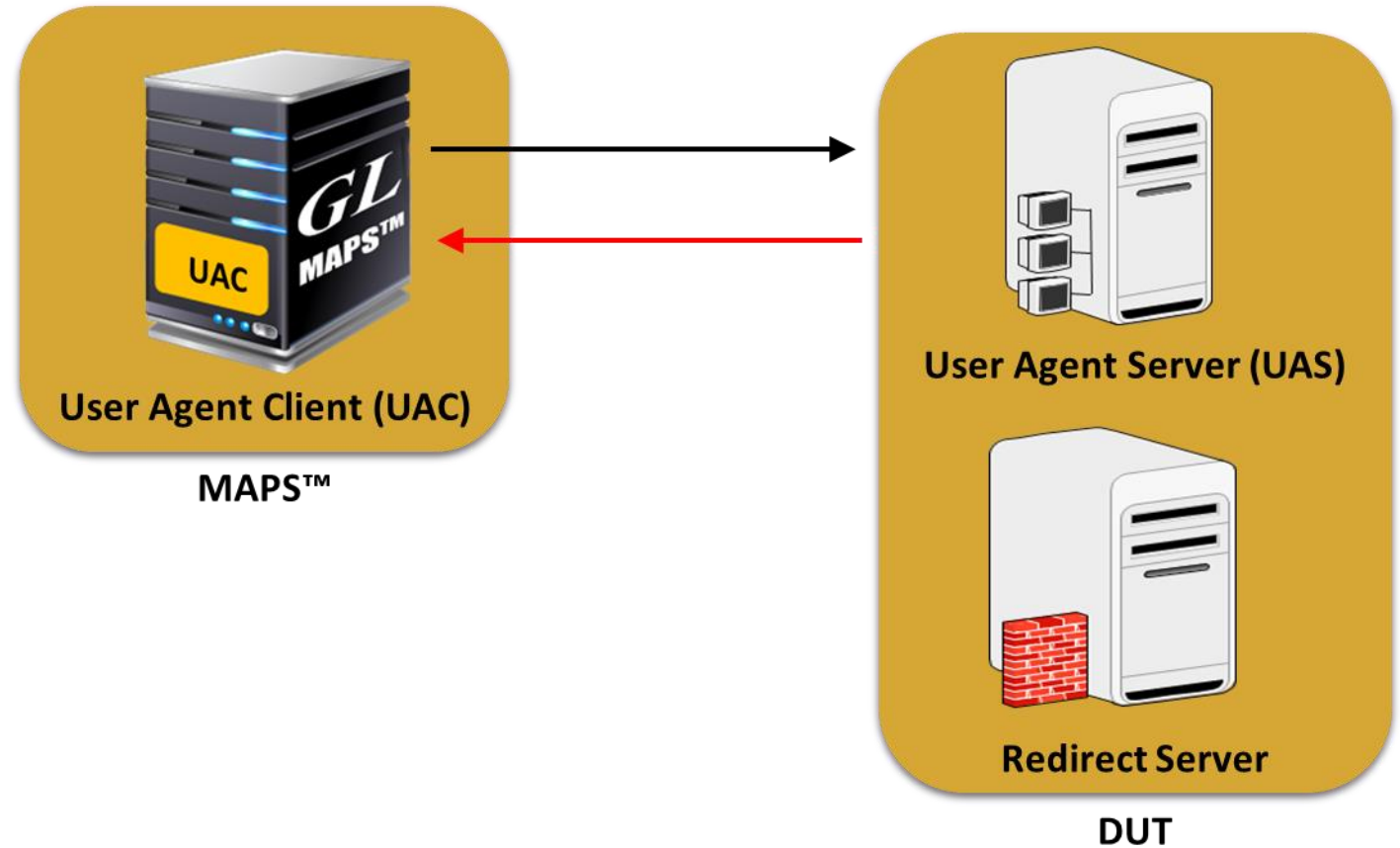


MAPS™ SIP Configured as UAC

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

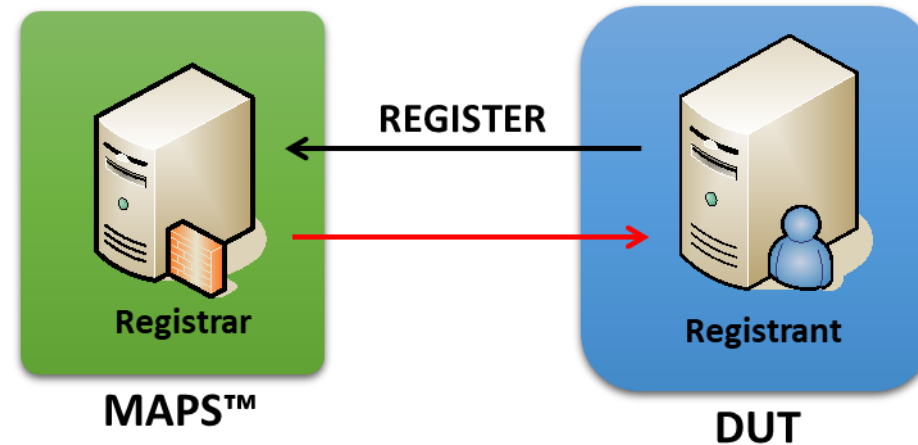


MAPS™ SIP Configured as Registrar

Testing Registrant

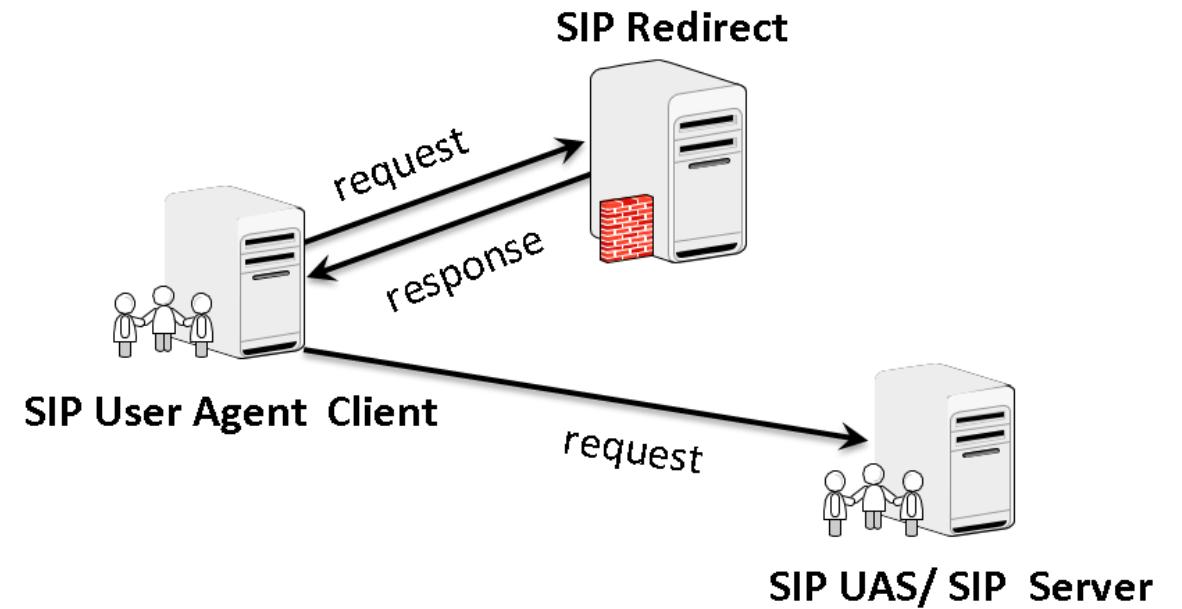
Scenario: MAPS™ acting as Registrar and testing Registrant

- 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)

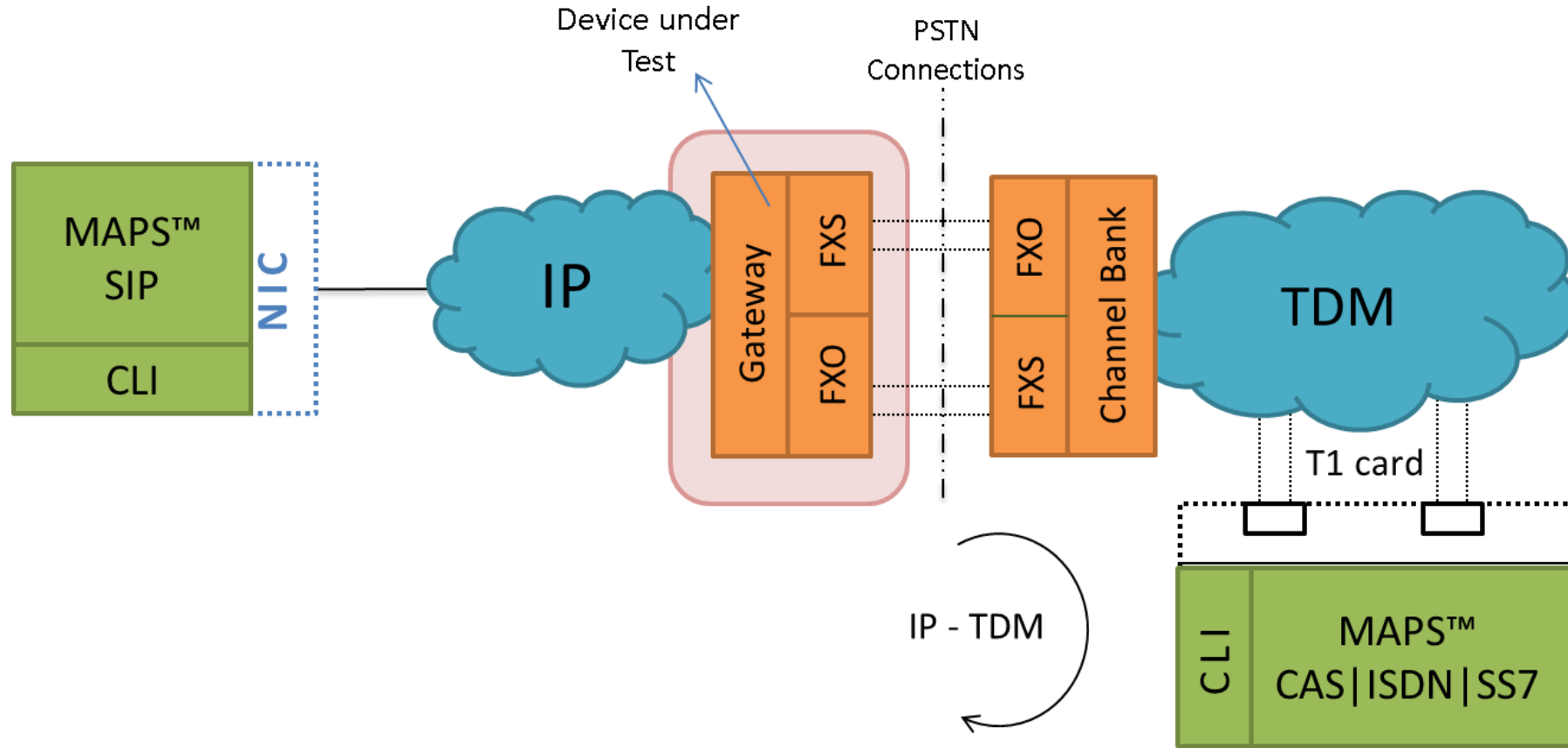
- Registrant – Registers with Registrar
- Call with Auto Traffic of RTP Action
- Traffic Impairments
- Simulates IVR (Interactive Voice Response) for RTP traffic
- Call through Proxy
- Sequential and Random Generation of Calls
- Simultaneous Generation of Calls
- Load Generation (Stress Testing)

Call Reception (UAS)

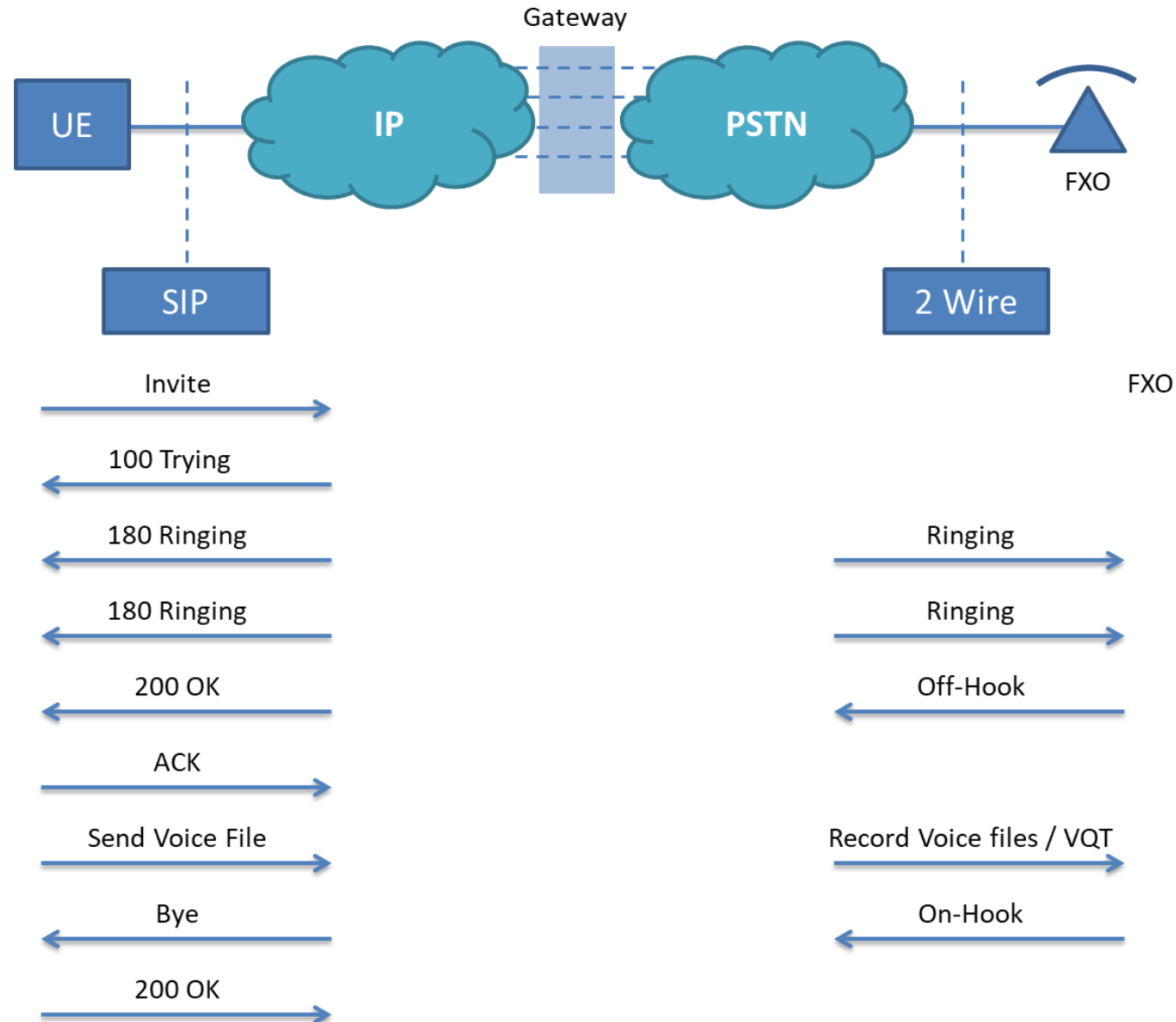
- Registrar – Accepts the registration from registrant
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (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

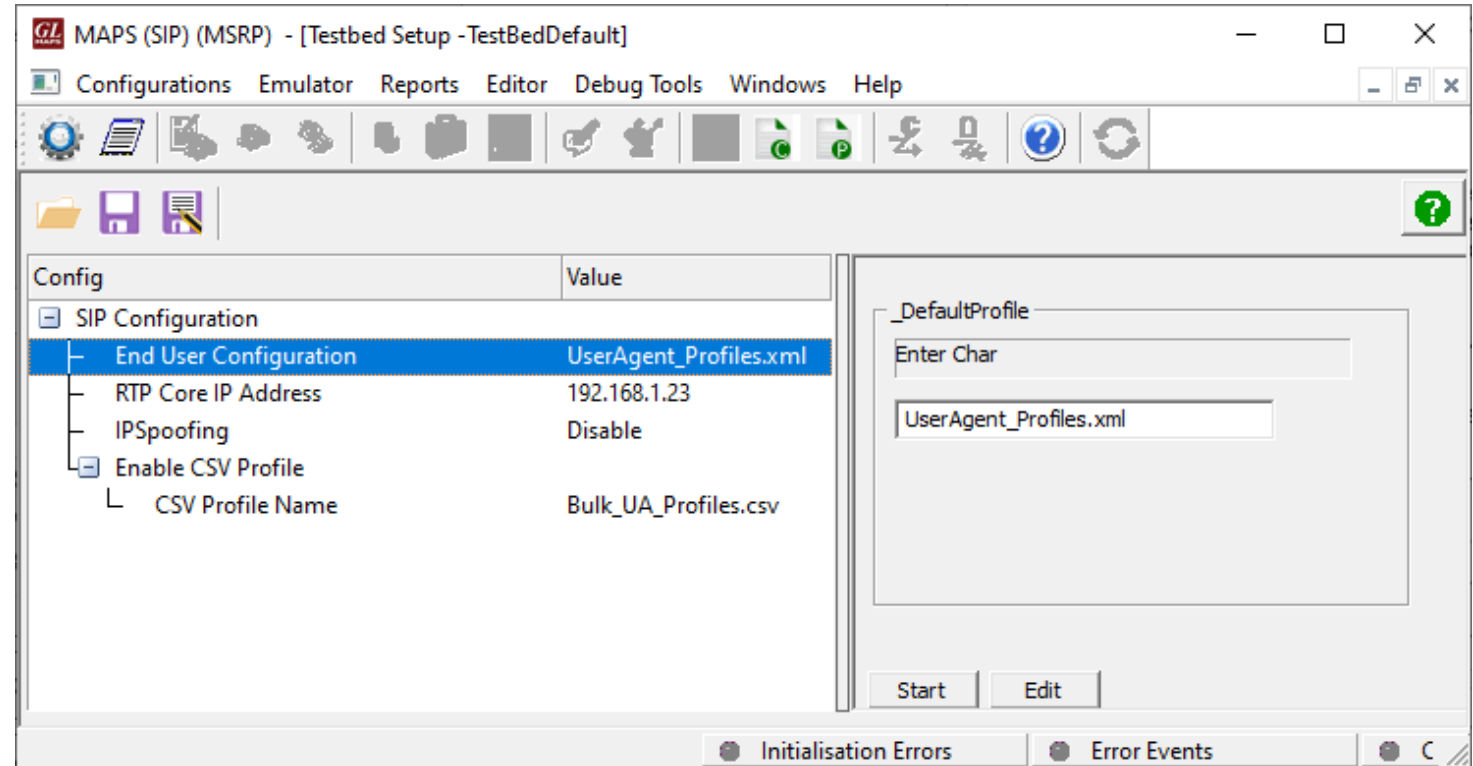


End-to-End Gateway Testing Call Scenario



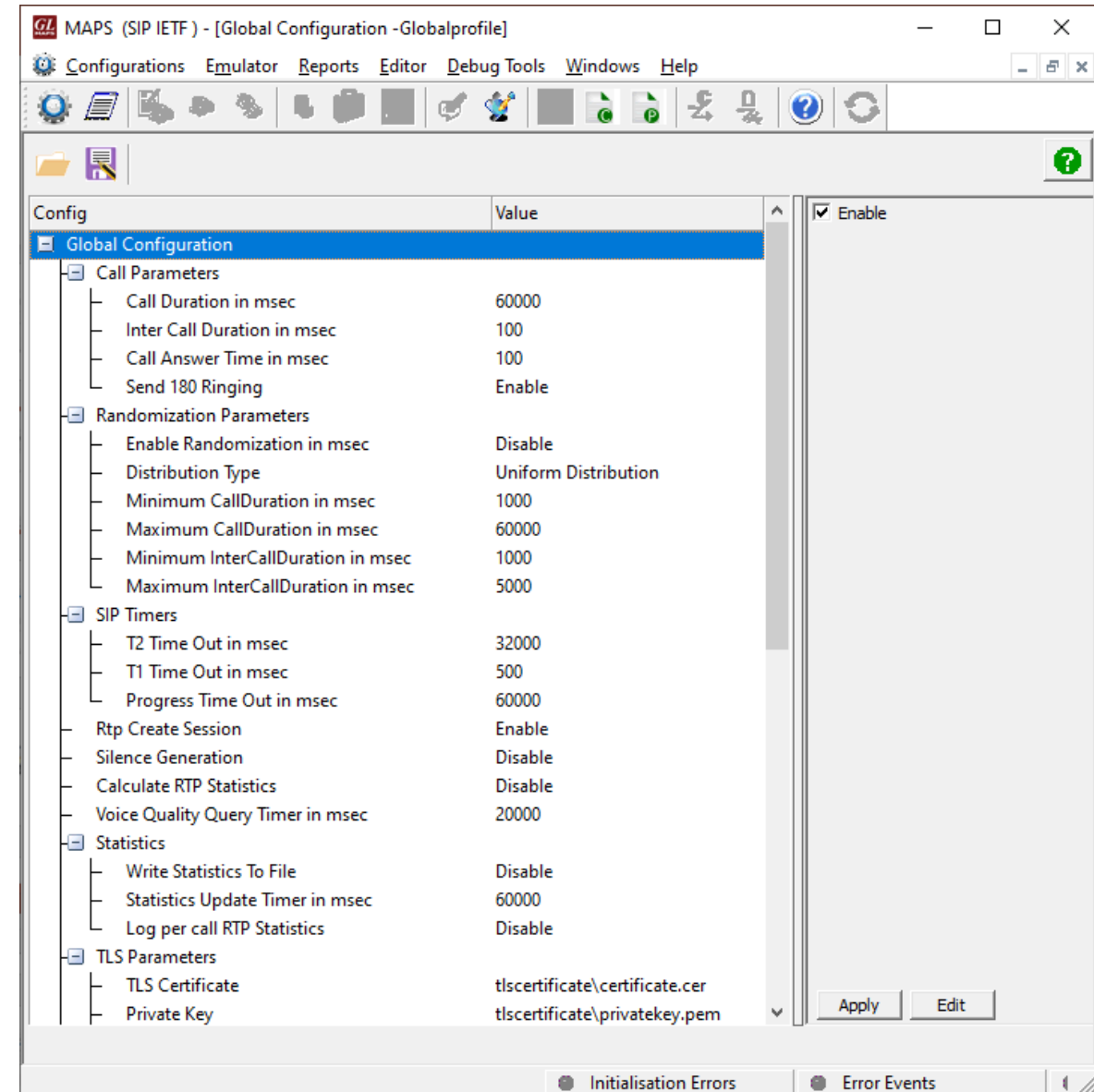
Test Bed Configuration

- **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



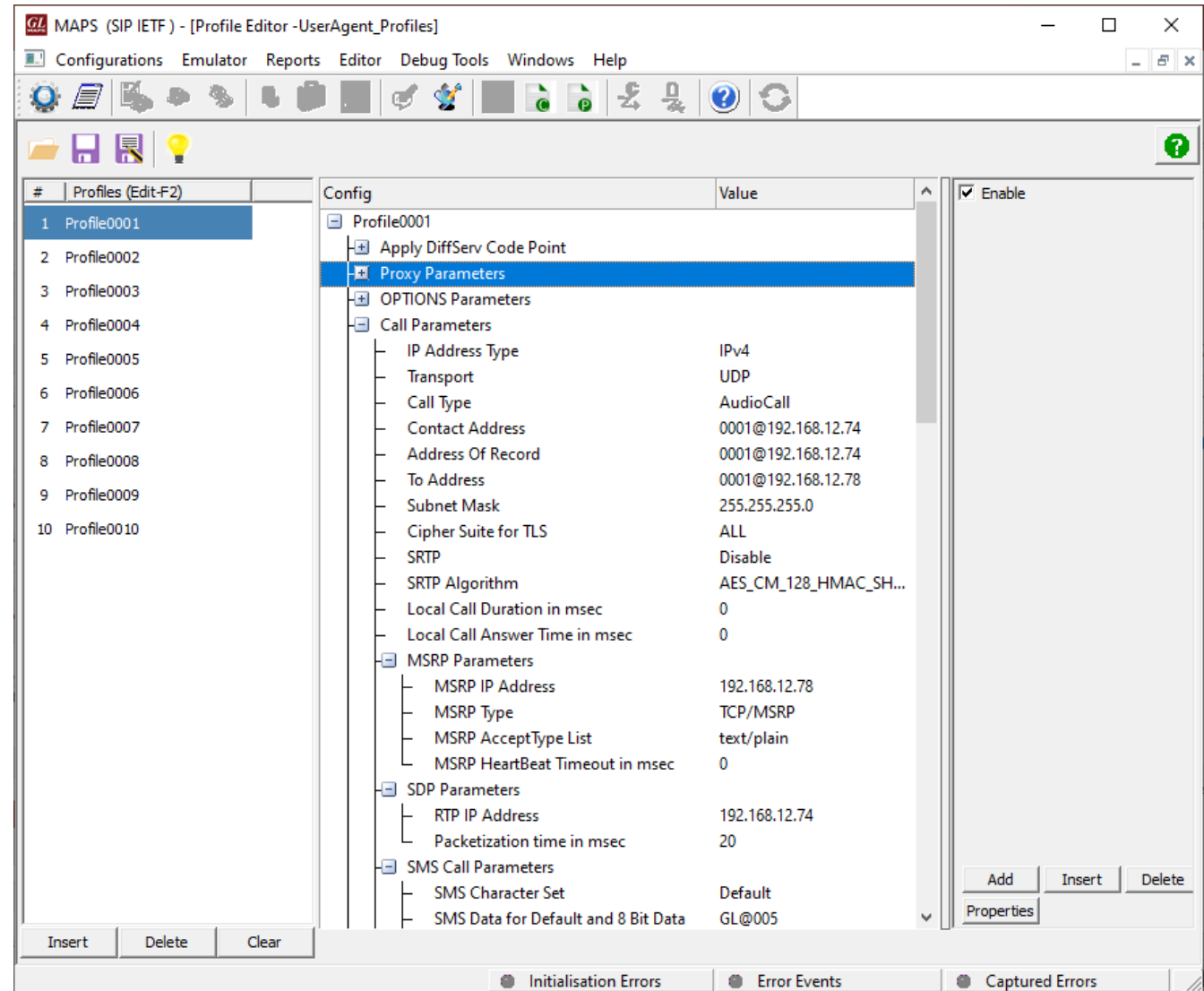
Global Configuration

- 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



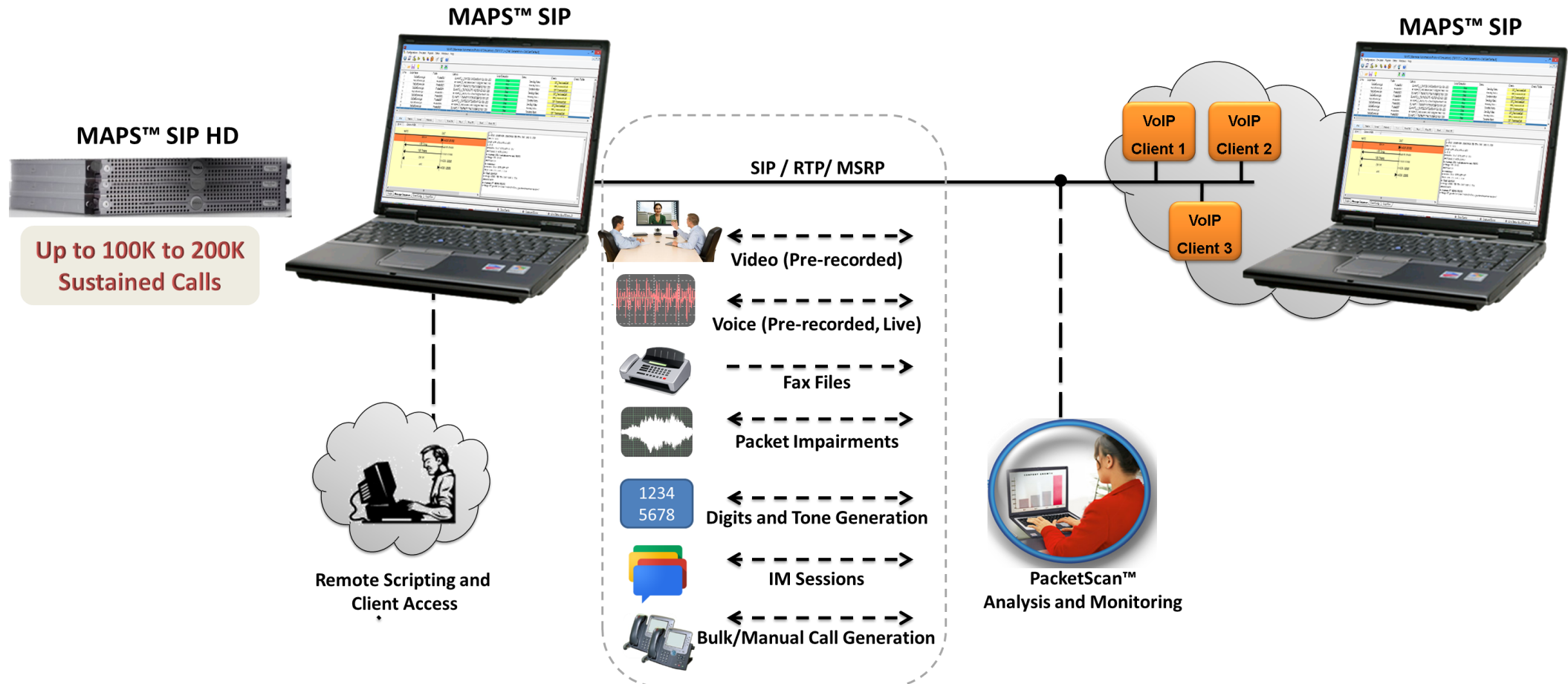
User Agents Configuration

- 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



IP Traffic Simulation Capabilities and Performance

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



SIP Capabilities and Performance

Product Version	Max Simultaneous Calls			
	Only Signaling	Signaling + RTP Voice Traffic	Signaling + RTP VideoTraffic	Signaling + MSRP (IM) Traffic
MAPS™ SIP 64-bit (Core i7 with 12GB RAM)	30,000 Calls @ 250 CPS	2000 @ 250 CPS	500	500
MAPS™ SIP HD 64-bit (Zeon Server with 16 Processors and 64GB RAM)	100,000 Calls @250 CPS	20000 @ 250 CPS	-	-

Call Generation with Voice Traffic

The screenshot displays the GL MAPS (SIP) - [Call Generation - CallGenDefault] application window. The interface includes a menu bar (Configurations, Emulator, Reports, Editor, Debug Tools, Windows, Help) and a toolbar with various icons. Below the toolbar is a table with columns: Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, Events Profile, and Result.

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown
2	SipCallControl.gls	Profile0001	GL-MAPS-3-104714086-10...	Start	PCMU Call Terminated	None		Pass

Below the table is a control bar with buttons: Add, Delete, Insert, Refresh, Start, Start All, Stop, Stop All, Abort, and Abort All. The main area is divided into two panes. The left pane, titled 'Message Sequence', shows a sequence of messages between MAPS and DUT:

MAPS	DUT
INVITE	17:52:58.529000
100 Trying	17:52:58.558000
180 Ringing	17:52:58.570000
200 OK	17:52:58.696000
ACK	17:52:58.723000
BYE	17:52:58.762000
200 OK	17:53:58.788000

The right pane, titled 'Find', displays the SIP message details for the selected message:

```
INVITE sip:0001@192.168.12.210 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.211:5060;branch=z9hG4bK-4-104714086-10291-9124
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, UPDATE
From: 0001 <sip:0001@192.168.12.211>;tag=FromTag-1-104714086-10288-9124
To: 0001 <sip:0001@192.168.12.210>
Call-ID: GL-MAPS-3-104714086-10290-9124@192.168.12.211
CSeq: 1 INVITE
Contact: 0001 <sip:0001@192.168.12.211>
Content-Type: application/sdp
Content-Length: 240

v=0
o=0001 32891133 1 IN IP4 192.168.12.211
s=SIP Call
c=IN IP4 192.168.12.211
t=0 0
m=audio 1030 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
aptime:20
a=sendrecv
```

At the bottom of the window, there is a status bar with tabs: Scripts, Message Sequence (selected), Event Config, and Script Flow. The status bar also displays: Initialisation Errors, Error Events, Captured Errors, and Link Status Up=0 Down=0.

Call Generation with IVR Traffic

GL MAPS (SIP) - [Call Reception]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Results
1	SipCallControl.gls	Profile0001	GL-MAPS-3-104714086-10290-912...	Completed	PCMU Call Terminated	None		Pass

Stop Stop All Abort Abort All ☒ Show Records ☐ Select Active Call ☐ Auto Trash Trash

Save Column Width ☐ Show Latest

DUT MAPS

INVITE 17:52:58.539000

100 Trying 17:52:58.549000

180 Ringing 17:52:58.560000

200 OK 17:52:58.681000

ACK 17:52:58.736000

BYE 17:53:58.768000

200 OK 17:53:58.776000

Find

```
INVITE sip:0001@192.168.12.210 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.211:5060;branch=z9hG4bK-4-104714086-10291-9124
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER,UPDATE
From: 0001 <sip:0001@192.168.12.211>;tag=FromTag-1-104714086-10288-9124
To: 0001 <sip:0001@192.168.12.210>
Call-ID: GL-MAPS-3-104714086-10290-9124@192.168.12.211
CSeq: 1 INVITE
Contact: 0001 <sip:0001@192.168.12.211>
Content-Type: application/sdp
Content-Length: 240

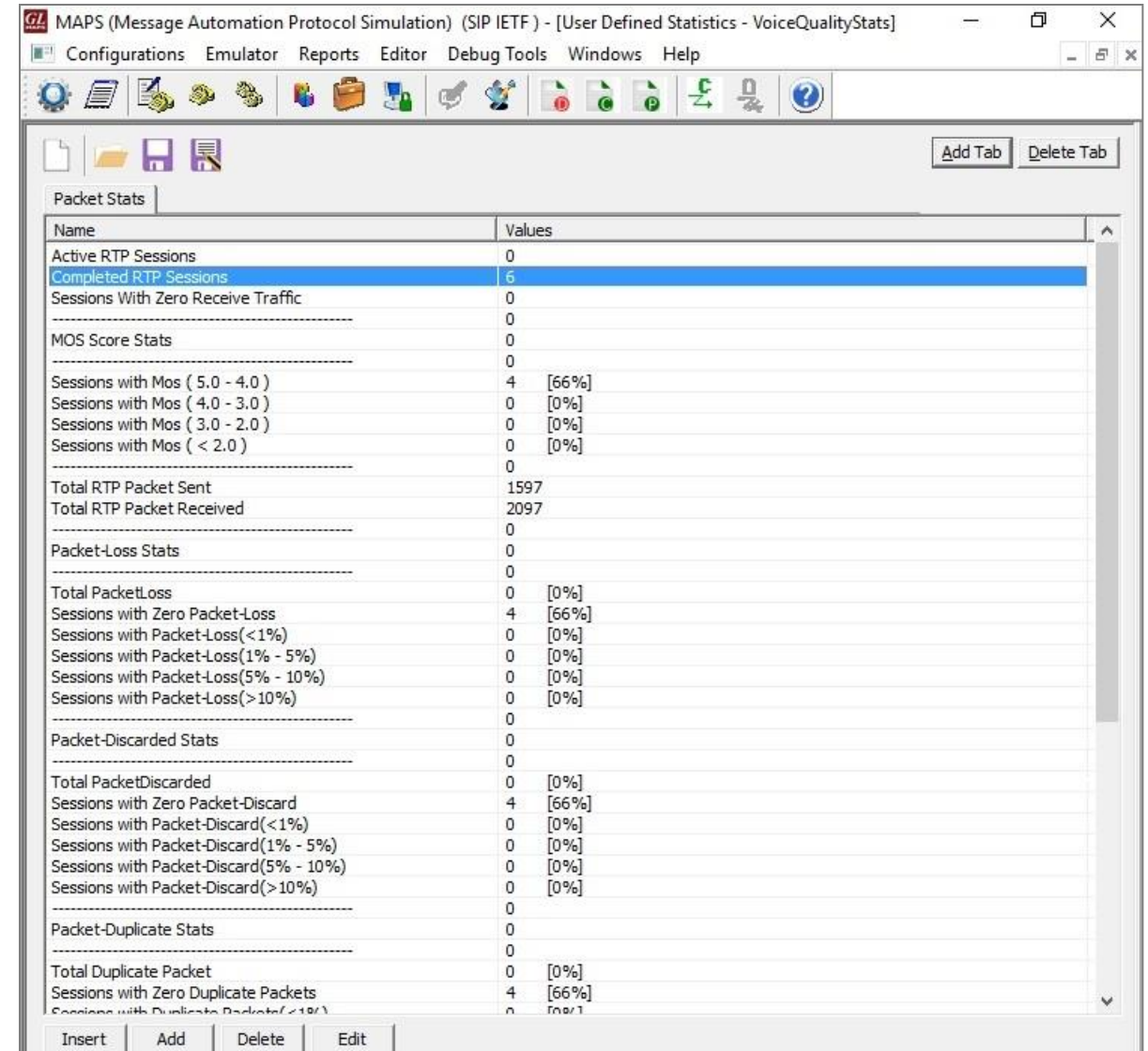
v=0
o=0001 32891133 1 IN IP4 192.168.12.211
s=SIP Call
c=IN IP4 192.168.12.211
t=0 0
m=audio 1030 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

Scripts Message Sequence Event Config Script Flow

● Initialisation Errors ● Error Events ● Captured Errors ● Link Status Up=0 Down=0

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



Name	Values
Active RTP Sessions	0
Completed RTP Sessions	6
Sessions With Zero Receive Traffic	0
MOS Score Stats	0
Sessions with Mos (5.0 - 4.0)	4 [66%]
Sessions with Mos (4.0 - 3.0)	0 [0%]
Sessions with Mos (3.0 - 2.0)	0 [0%]
Sessions with Mos (< 2.0)	0 [0%]
Total RTP Packet Sent	1597
Total RTP Packet Received	2097
Packet-Loss Stats	0
Total PacketLoss	0 [0%]
Sessions with Zero Packet-Loss	4 [66%]
Sessions with Packet-Loss(<1%)	0 [0%]
Sessions with Packet-Loss(1% - 5%)	0 [0%]
Sessions with Packet-Loss(5% - 10%)	0 [0%]
Sessions with Packet-Loss(>10%)	0 [0%]
Packet-Discarded Stats	0
Total PacketDiscarded	0 [0%]
Sessions with Zero Packet-Discard	4 [66%]
Sessions with Packet-Discard(<1%)	0 [0%]
Sessions with Packet-Discard(1% - 5%)	0 [0%]
Sessions with Packet-Discard(5% - 10%)	0 [0%]
Sessions with Packet-Discard(>10%)	0 [0%]
Packet-Duplicate Stats	0
Total Duplicate Packet	0 [0%]
Sessions with Zero Duplicate Packets	4 [66%]
Sessions with Duplicate Packets(>10%)	0 [0%]

Event Log, Error Events, Captured Errors

The image displays three overlapping screenshots of the 'Events' application window, illustrating different views of system events and errors.

Top Screenshot (Event Log Tab): Shows a list of events with columns: Date/Time, Captured Events, Call Trace Id, Script Name, and Script Id. The 'Event Log' tab is selected.

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 13:06:42.359000	Script Initialized	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	TransportId = 1	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	TransportStatus =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	MediaPAddress=192.168.12.216, AudioMediaPort=1038	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.395000	RtpCoreSrtipAlgorithm =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.395000	RtpCoreSrtipKey =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.405000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.420000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.420000	PROGRESS Received	GL-MAPS-18-1006011716-9553-7856@192.168.12.216	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.427000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080

Middle Screenshot (Error Events Tab): Shows a list of error events with columns: Date/Time, Captured Events, Call Trace Id, Script Name, and Script Id. The 'Error Events' tab is selected.

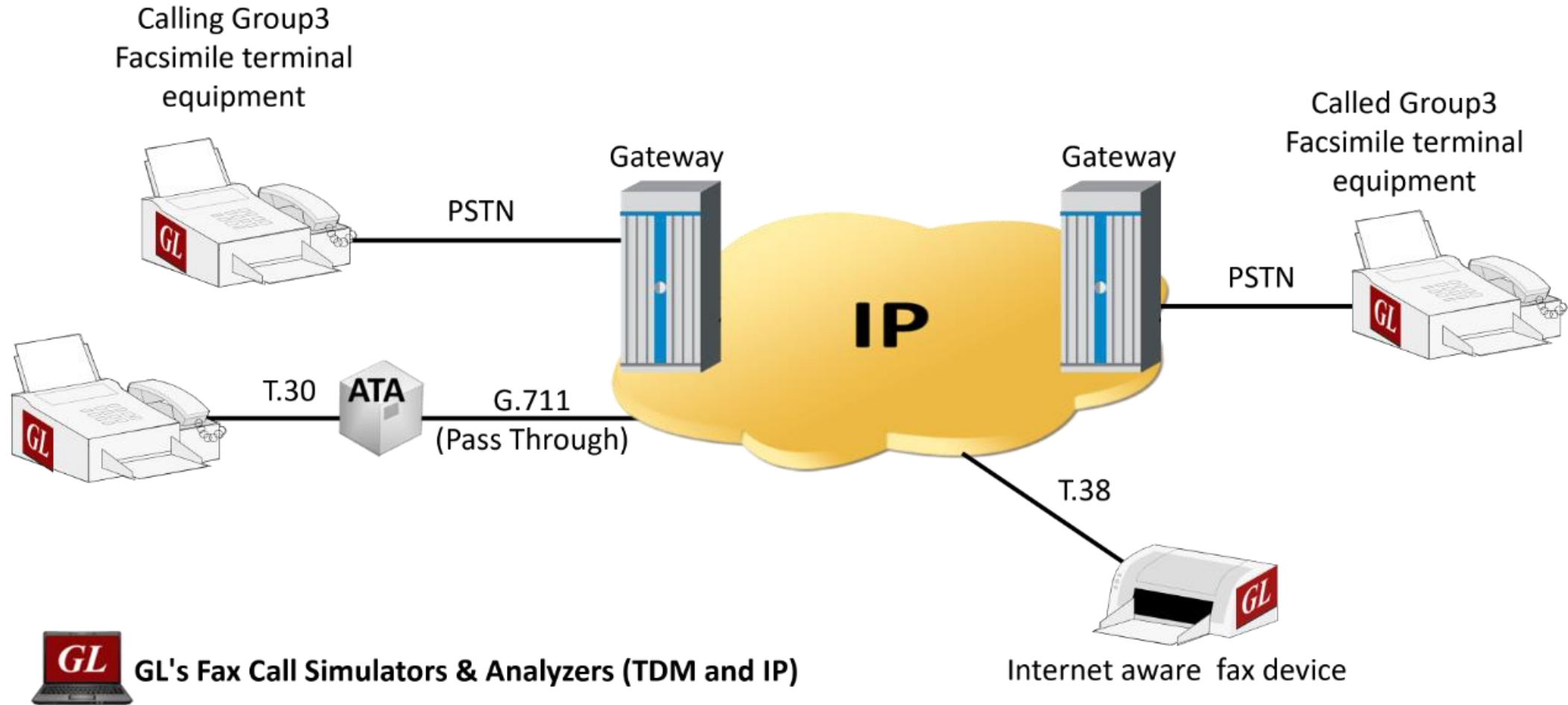
Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 12:58:37.078000	Session Creation Failed	CGProtScriptId-3-1005519694-9487-14080	SipCallControl.gls	CGProtScriptId-3-1005519694-9487-14080
2024-9-23 13:00:01.136000	Session Creation Failed	CGProtScriptId-4-1005603631-9496-14080	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:00:01.237000	4xx Failure Message=480 INVITE	GL-MAPS-10-1005605428-9501-14132@192.168.12.216	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:01:05.296000	Session Creation Failed	CGProtScriptId-5-1005669593-9505-14080	SipCallControl.gls	CGProtScriptId-5-1005669593-9505-14080
2024-9-23 13:03:37.277000	6xx Failure Message=606 INVITE	GL-MAPS-10-1005826538-9530-13992@192.168.12.216	SipCallControl.gls	CGProtScriptId-7-1005826534-9526-14080
2024-9-23 13:05:06.385000	Retransmission Time Out	GL-MAPS-9-1005884067-9537-23124@192.168.12.216	SipCallControl.gls	CGProtScriptId-8-1005882418-9532-14080
2024-9-23 13:05:06.385000	Retransmission Timeout for the Message = INVITE	GL-MAPS-9-1005884067-9537-23124@192.168.12.216	SipCallControl.gls	CGProtScriptId-8-1005882418-9532-14080

Bottom Screenshot (Captured Errors Tab): Shows a list of captured errors with columns: Date/Time, Captured Events, Call Trace Id, Script Name, and Script Id. The 'Captured Errors' tab is selected.

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 12:57:27.877000	Error::Creating UDP Socket Failed due to Errorcode=10049			
2024-9-23 12:57:27.878000	192.168.12.200 : 5060 : please verify IP/Port not in use			
2024-9-23 12:57:27.879000	192.168.12.200 : 5061 : please verify IP/Port not in use			
2024-9-23 12:58:32.060000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-3-1005519694-9487-14080	SipCallControl.gls	CGProtScriptId-3-1005519694-9487-14080
2024-9-23 12:59:56.116000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-4-1005603631-9496-14080	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:01:00.291000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-5-1005669593-9505-14080	SipCallControl.gls	CGProtScriptId-5-1005669593-9505-14080

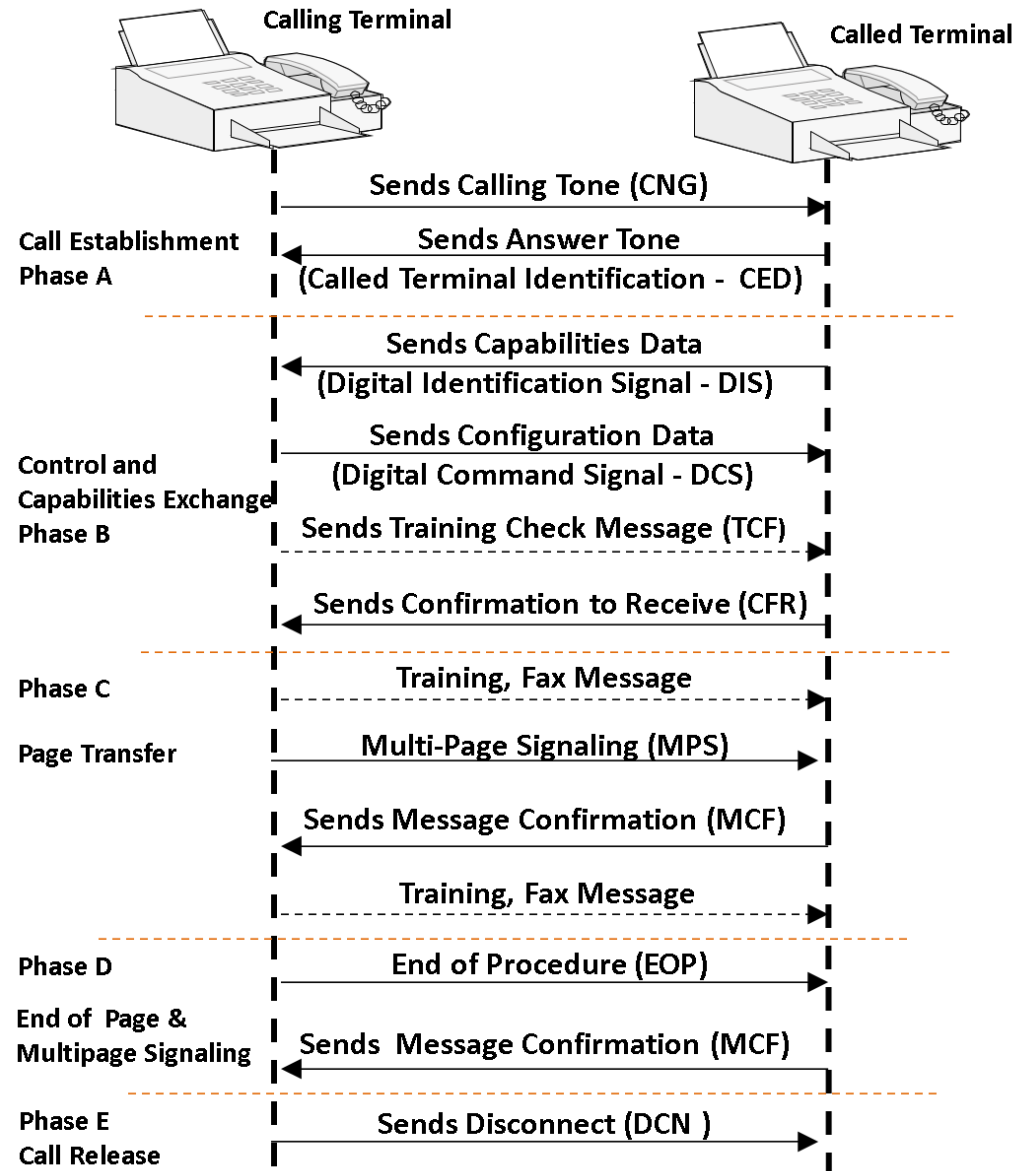
The bottom screenshot also includes a 'Save Events' section with a 'Clear' button and a checkbox for 'Capture Events to file'.

Fax Simulation over IP

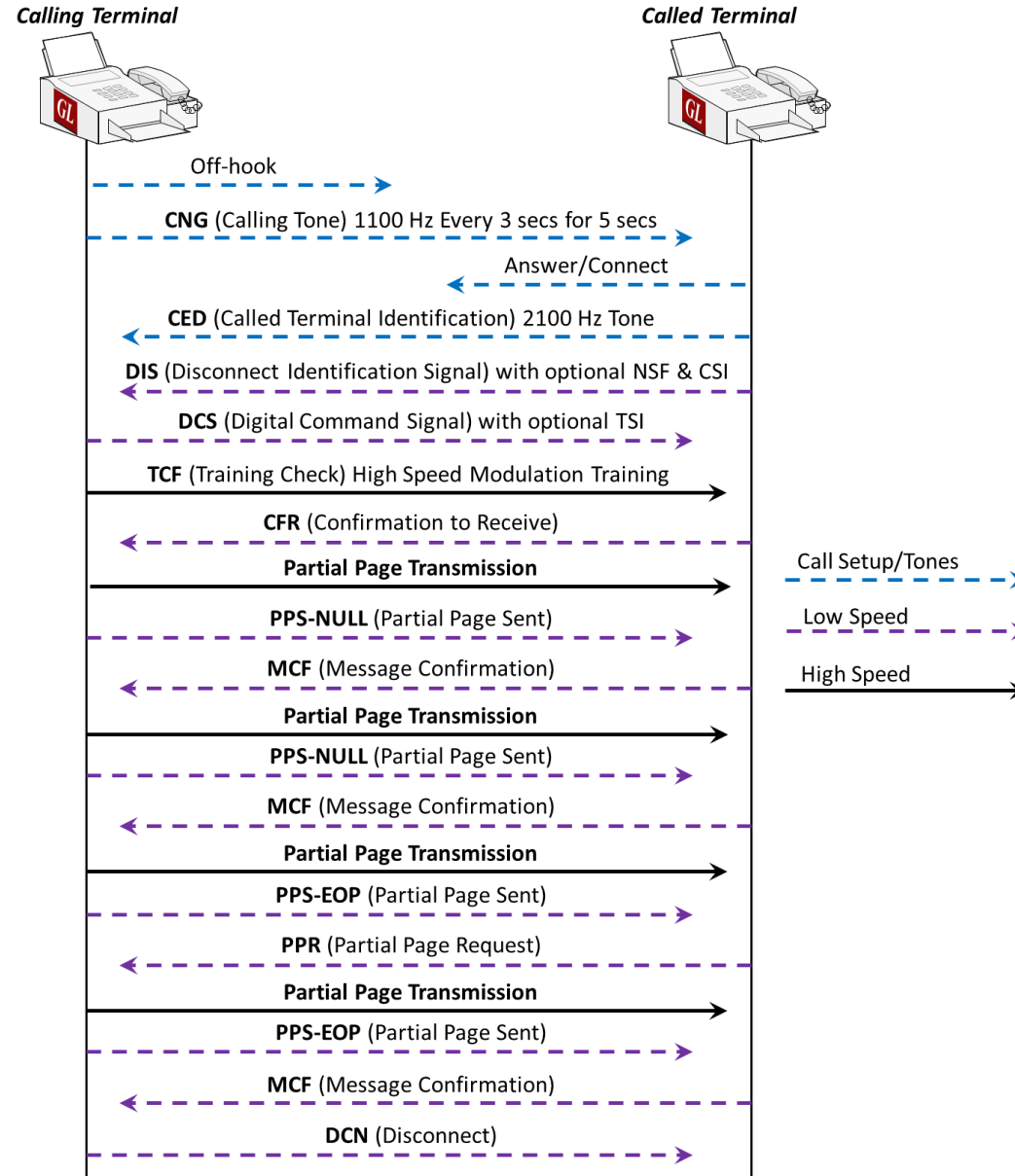


- RTP G.711 Pass Through Fax Simulation (PKS200)
- T.38 Fax Simulation over UDPTL (PKS211)

Call Scenarios - Fax T.30



T.38 Fax Emulation over IP using MAPS™



T.38 Fax Call in Progress and Related Events

GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation]

Configurations Emulator Reports Editor Debug Tools Windows Help

St No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile
1	SipCallControl.gls	Profile0001	GL-MAPS_3_775735732-4923-3768@192.168.12.212	Stop	Fax Session Successful	SIP_TerminateCall	

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

MAPS DUT

INVITE 11:12:41.097000

100 Trying 11:12:41.123000

180 Ringing 11:12:41.129000

200 OK 11:12:41.248000

ACK 11:12:41.260000

INVITE 11:12:41.299000

200 OK 11:12:41.306000

ACK 11:12:41.308000

33600 Rate of V34 selected after 11:13:15.904000

CSI(Called Subscriber Identification) 11:13:15.905000

DIS(Digital Identification Signal) 11:13:15.906000

ECM mode Selected in DCS 11:13:15.906000

MMR Encoding selected in DCS 11:13:15.906000

Scripts Message Sequence Event Config Script Flow

Initialisation Errors Error Events

Find

INVITE sip:0001@192.168.12.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.212:5060;branch=0
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTION
From: 0001 <sip:0001@192.168.12.212>;tag=1
To: 0001 <sip:0001@192.168.12.213>;tag=1
Call-ID: GL-MAPS_3_775735732-4923-3768@192.168.12.212
CSeq: 2 INVITE
Contact: 0001 <sip:0001@192.168.12.212>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 361

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.212
s=SIP Call
c=IN IP4 192.168.12.212
t=0 0
m=image 1030 udpt1 t38
a=T38FaxVersion:3
a=T38FaxMaxBitRate:33600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingJMR:0
a=T38FaxTranscodingJBR:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:400
a=T38FaxMaxDatagram:280
a=T38FaxUdpEC:t38UDPRedundancy

GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Events]

Configurations Emulator Reports Editor Debug Tools Windows Help

Event Log Error Events Captured Errors

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2017-10-26 11:12:41.132000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.132000	PROGRESS Received	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.260000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.261000	Call Connected	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.308000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.904000	Fax - Status: 33600 Rate of V34 selected after MP exchange	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.905000	Fax - Status: CSI(Called Subscriber Identification)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.906000	Fax - Status: DIS(Digital Identification Signal)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.906000	Fax - Status: ECM mode Selected in DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.907000	Fax - Status: MMR Encoding selected in DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.908000	Fax - Status: 200x200 Resolution selected in the DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.908000	Fax - Status: A4 pagesize selected in the DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.909000	Fax - Status: TSI(Transmitting Subscriber Identification)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.909000	Fax - Status: DCS(Digital Command Signal)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.910000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.911000	Fax - Status: Transmitter Started To Train	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.911000	Fax - Status: Transmitter Train Successful	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.912000	Fax - Status: CFR(Confirmation To Receive)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.912000	Fax - Status: Image Transmit Start	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.913000	Fax - Status: Image Transmit End	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.913000	Fax - Status: PPS NULL(Current Partial Page Block: Transmission Complete)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.914000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.914000	Fax - Status: MCF(Message Confirmation)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.915000	Fax - Status: Image Transmit Start	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.915000	Fax - Status: Image Transmit End	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.916000	Fax - Status: PPS EOP(All Pages Transmitted)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.916000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.917000	Fax - Status: MCF(Message Confirmation)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.917000	Fax - Status: DCN(Disconnect)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.918000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.919000	Fax Session Successful	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.919000	Fax - Status: FaxSessionDuration = 2898 msecInitialModem = "V34" InitialRate = "33600" FinalModem = "V34"	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.280000	BYE Sent		SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	200 OK to BYE Received		SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	Call Terminated	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	Inter Call Duration = 100	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...

Clear Save Events Capture Events to file

Initialisation Errors Error Events Captured

Call Generation with FAX Traffic

The screenshot displays the GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation] interface, showing two windows.

Left Window: MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation]

This window shows a table of call generation results:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile
1	SipCallControl.gls	Profile0001	GL-MAPS_3_776162744-4937-3896@192.168.12.212	Stop	Fax Session Successful	SIP_TerminateCall	

Below the table, a message sequence is displayed between MAPS and DUT:

MAPS	DUT
INVITE	11:19:48.114000
100 Trying	11:19:48.140000
180 Ringing	11:19:48.145000
200 OK	11:19:48.268000
ACK	11:19:48.280000
Fax Status :: Send Fax Started	11:19:48.343000
33600 Rate of V34 selected after ...	11:20:22.163000
V21 Signal Done	11:20:22.164000
CSI(Called Subscriber Identification)	11:20:22.164000
DIS(Digital Identification Signal)	11:20:22.165000
ECM mode Selected in DCS	11:20:22.166000
MMR Encoding selected in DCS	11:20:22.166000
200x200 Resolution selected in th...	11:20:22.167000

Right Window: MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation - CallGenDefault]

This window shows a detailed view of the message sequence, including SIP headers and fax-related parameters:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Tot
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown	
2	SipCallControl.gls	Profile0003	GL-MAPS_1_14830463-305-3768@192.168.1.141	Stop	Fax Session Created	SIP_TerminateCall		Pass	

Below the table, a message sequence is displayed between MAPS and DUT:

MAPS	DUT
INVITE	16:58:18.881000
100 Trying	16:58:19.244000
180 Ringing	16:58:19.247000
200 OK	16:58:19.361000
ACK	16:58:19.369000
INVITE	16:58:19.377000
Fax Status :: 33600_Rate_of_V34_selected_after_MP...	16:58:19.379000
33600_Rate_of_V34_selected_after_MPh_exchange	16:58:19.379000
Fax Status :: CSI(Called_Subscriber_Identification)	16:58:19.379000
CSI(Called_Subscriber_Identification)	16:58:19.379000
Fax Status :: DIS(Digital_Identification_Signal)	16:58:19.379000

The right window also displays SIP headers and fax-related parameters:

```
INVITE sip:0003@192.168.1.143 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.141:5060;branch=z9hG4bK_1_14831002-308-3768
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0003@192.168.1.141>;tag=FromTag_1_14830463-303-3768
To: 0001 <sip:0003@192.168.1.143>
Call-ID: GL-MAPS_1_14830463-305-3768@192.168.1.141
CSeq: 2 INVITE
Contact: 0010 <sip:0003@192.168.1.141>
Content-Type: application/sdp
Content-Length: 359

v=0
o=0001 33852938 33852938 IN IP4 192.168.1.141
s=SIP Call
c=IN IP4 192.168.1.141
t=0 0
m=image 1028 udpt1 t38
a=T38FaxVersion:3
a=T38MaxBitRate:33600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:400
a=T38FaxMaxDatagram:280
a=T38FaxUdpEC:t38UDPRedundancy
```

FAX Traffic Events

Events				
Event Log Error Events Captured Errors				
Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2015-1-15 15:27:08.544000	UDP Port = 5060		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.544000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	Call Connected	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.678000	Sending RTP Fax	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:32.397000	RTP Fax Sent	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.386000	BYE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	200 Ok to BYE Received		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Call Terminated	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Inter Call Duration = 1000	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432

Save Events

Clear

☐ Capture Events to file

File Traffic Events

Events			
Event Log Error Events Captured Errors			
Date/Time	Captured Events	Call Trace Id	Script M
2015-1-15 15:32:20.946000	UDP Port = 5060		SIP-Pro
2015-1-15 15:32:20.946000	INVITE Sent		SIP-Pro
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:20.958000	PROGRESS Received		SIP-Pro
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.073000	ACK Sent		SIP-Pro
2015-1-15 15:32:21.073000	Call Connected	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.074000	RxFilename = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_9.glw	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Receiving RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Sending RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Sent	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.083000	BYE Sent		SIP-Pro
2015-1-15 15:33:21.091000	200 Ok to BYE Received		SIP-Pro
2015-1-15 15:33:21.091000	Call Terminated	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.091000	Inter Call Duration = 1000	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC

Save Events

Clear

☐ Capture Events to file

...

Video Call Generation

GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation - CallGenDefault]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Total Iter
1	SipCallControl.gls	Profile0001	GL-MAPS_3_851042897-7265-11744@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
2	SipCallControl.gls	Profile0001	GL-MAPS_3_851045200-7276-5692@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
3	SipCallControl.gls	Profile0001	GL-MAPS_3_851046272-7287-7876@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
4	SipCallControl.gls	Profile0001	GL-MAPS_3_851047176-7298-2364@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
5	SipCallControl.gls	Profile0001	GL-MAPS_3_851048304-7309-11840@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
6	SipCallControl.gls	Profile0001	GL-MAPS_11_851048991-7320-9392@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
7	SipCallControl.gls	Profile0001	GL-MAPS_9_851049784-7327-11744@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
8	SipCallControl.gls	Profile0001	GL-MAPS_9_851050200-7334-5692@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
9	SipCallControl.gls	Profile0001	GL-MAPS_9_851050815-7341-7876@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
10	SipCallControl.gls	Profile0001	GL-MAPS_9_851052304-7348-2364@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

MAPS DUT

INVITE 10:55:08.130000

100 Trying 10:55:08.147000

180 Ringing 10:55:08.149000

200 OK 10:55:08.280000

ACK 10:55:08.280000

Find

Content-Type: application/sdp
Content-Length: 291

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.74
s=-
c=IN IP4 192.168.12.74
t=0 0
m=audio 1028 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv
m=video 1030 RTP/AVP 97
b=TIAS:256000
a=sendrecv
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42e01e; packetization-mode=1

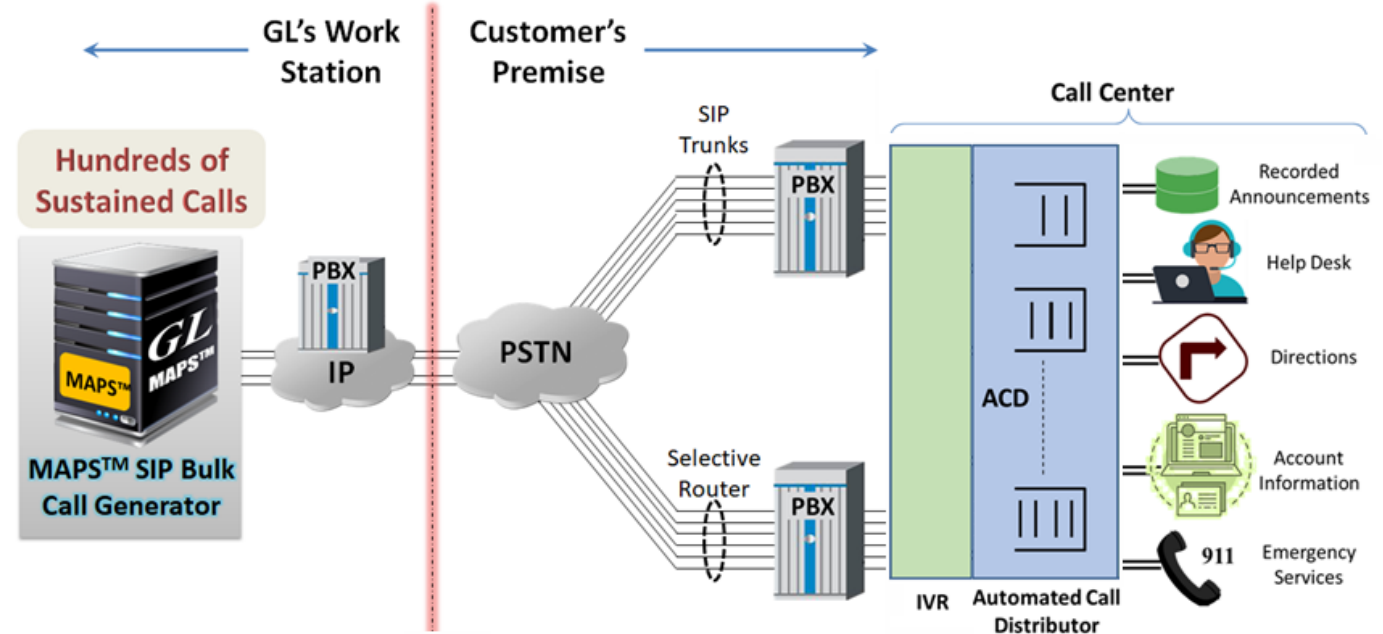
Scripts Message Sequence Event Config Script Flow

Initialisation Errors Error Events Captured Errors Link Status Up=0 Down=0

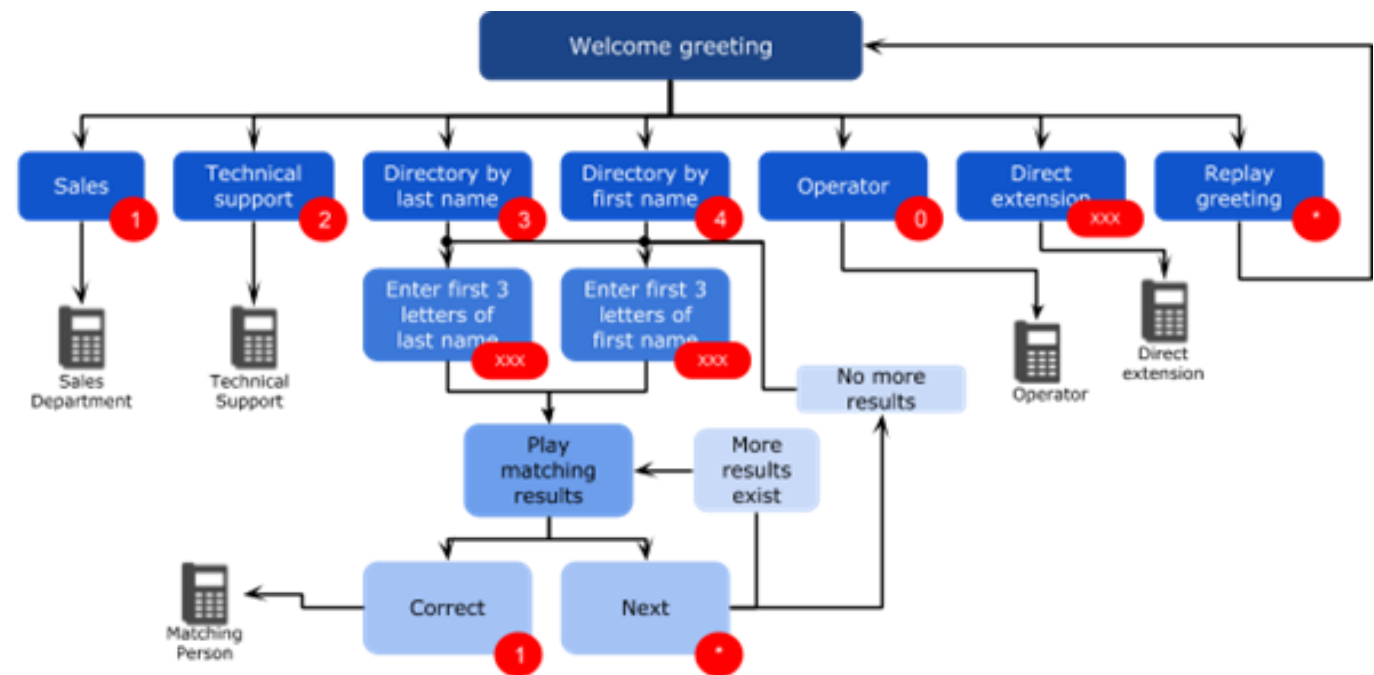
Transmit pre-recorded video traces with video codecs like H.264, and H.263

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



GL's Interactive Voice Response Scenario



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

	A	B	C	D	E	F	G
1	IVRIndex	IVRPromptLanguage	IVRExpectedTranscript	IVRResponseType	IVRResponseDTMF	IVRResponseSpeech	IVRNextPromptId
2	int	string	string	string	string	string	int
3	1	en-US	Welcome to GL Communications If you know your partys extension you can dial it at any time For sales press one for technical support press 2 for a directory by last name press 3	DTMF	3		2
4	2	en-US	Welcome to the directory. please enter the first 3 letters of your partys last name using your touch tone keypad Use the seven key for q and the nine key for z	DTMF	926		0

IVR Call Simulation

GL MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - CallGenDefault]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No Script Name Profile Call Info Script Execution Status Events Events Profile Result

3	SipCallControl.gls	Xbox	GL-MAPS-16-1305510-5309-6508@50.76.16.185	Start	PCMU Call Terminated	None	Pa
4	SipCallControl.gls	Profile0001	GL-MAPS-20-59922966-5347-4708@50.76.16.185	Start	PCMU Call Terminated	None	Pa

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

ACK 07:26:35.042000

Stage 1: Welcome to GL communications 07:26:41.452000

Stage 1: If you know your parties extension you can download at anytime 07:26:45.625000

Stage 1: For sales press 1 07:26:47.072000

Stage 1: For Technical Support Press 2 07:26:49.996000

Stage 1: Or directory by last name press 3 07:26:53.245000

Digits Transmitted :: 3 07:26:53.504000

Stage 2: Welcome to the directory Please enter the first 3 letters of your party's last name 07:27:06.247000

Stage 2: Using your touch tone keypad use the Seven key for Q and the nine key for Z I'm sorry 07:27:19.124000

Digits Transmitted :: 926 07:27:19.826000

BYE 07:27:20.786000

200 OK 07:27:20.949000

Find

```
INVITE sip:13016704784@104.219.163.73 SIP/2.0
Via: SIP/2.0/UDP 50.76.16.185:5060;branch=z9hG4bK-21-59922966-5348-4708
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER,UPDATE
From: 3016704784 <sip:3016704784@50.76.16.185>;tag=FromTag-18-59922966-5345-4708
To: 13016704784 <sip:13016704784@104.219.163.73>
Call-ID: GL-MAPS-20-59922966-5347-4708@50.76.16.185
CSeq: 1 INVITE
Contact: 3016704784 <sip:3016704784@50.76.16.185>
Content-Type: application/sdp
Content-Length: 242

v=0
o=3016704784 38929794 1 IN IP4 50.76.16.185
s=SIP Call
c=IN IP4 50.76.16.185
t=0 0
m=audio 1026 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

Scripts Message Sequence Event Config Script Flow

Initialisation Errors Error Events Captured Errors Link Status Up=0 Down=0

IVR Call Simulation Reports

SIP IVR Detailed Log

Maps_IVR_DetailedLog_2020-05-05_08-25-32_Profile0001.pdf - Adobe Acrobat Reader DC

File Edit View Window Help

Home Tools Maps_IVR_DetailedLog_2020-05-05_08-25-32_Profile0001.pdf x

GL Communications Inc Date: 05/05/2020

MAPS IVR Test Start Time: 08:25:32

Time	Type	Event	Certainty	Stage	Received Prompt	Expected Prompt	Similarity
2020-05-05 08:25:39.088000	Rx	Welcome to GL communications	0.8831	1			
2020-05-05 08:25:39.087000	Analysis			1	Welcome to GL communications	Welcome to GL Communications If you know your party's extension you can dial it at any time For Sales press one for technical support press 2 for a directory by last name press 3	15.819208
2020-05-05 08:25:42.775000	Rx	If you know your party's extension you can download at anytime	0.9424	1			
2020-05-05 08:25:42.775000	Analysis			1	Welcome to GL communications If you know your party's extension you can download at anytime	Welcome to GL Communications If you know your party's extension you can dial it at any time For Sales press one for technical support press 2 for a directory by last name press 3	45.762711
2020-05-05 08:25:44.458000	Rx	For sales press 1	0.8577	1			
2020-05-05 08:25:44.458000	Analysis			1			
2020-05-05 08:25:51.230000	Rx	For Technical Support Press 2 for directory by last name press 3	0.9056	1			
2020-05-05 08:25:51.230000	Analysis			1			
2020-05-05 08:25:51.231000	Tx	3					
2020-05-05 08:25:52.511000	Rx	For a directory by First Name Press	0.8785	1			
2020-05-05 08:26:21.479000		Failed to transcribe audio		2			

SIP IVR Result Log

MAPS_SIP_IVR_Result_2020_05_05_08_25_25.pdf - Adobe Acrobat Reader DC

File Edit View Window Help

Home Tools MAPS_SIP_IVR_Result_2020_05_05_08_25_25.pdf x

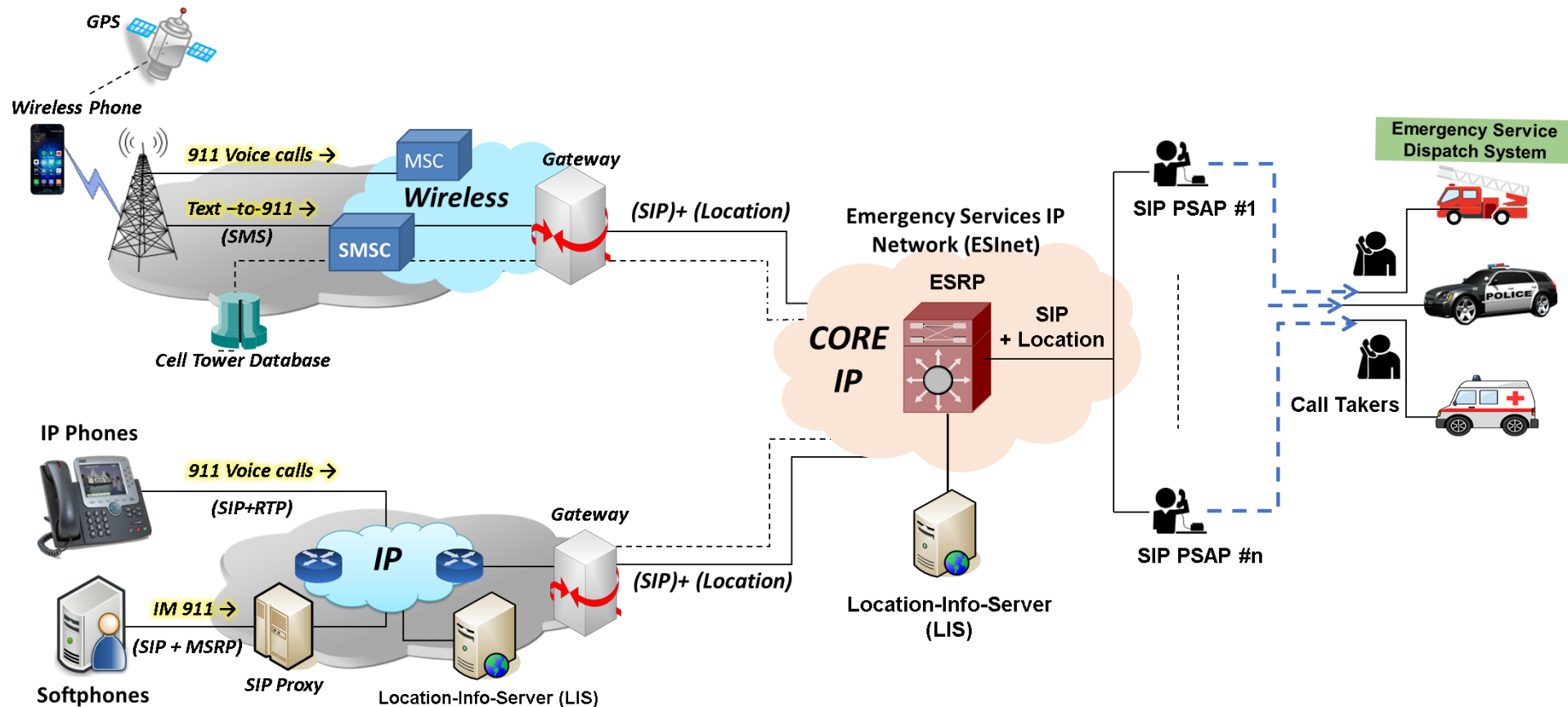
GL Communications Inc Date: 05/05/2020

MAPS SIP IVR Test Start Time: 08:25:25

SLNo	Time	Profile	Destination TN	IVR File	SIP Result	IVR Result	Detailed Report
1	2020-05-05 08:26:22.979000	Profile0001	13016704784	maps/sip/ivr/ivr_prompt_gl.csv	Pass	Pass	MAPS/SIP/IVR/Log/DetailedLog/Maps_IVR_DetailedLog_2020-05-05_08-25-32_Profile0001.pdf

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

MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Profile Editor - TrafficProfile]

Configurations Emulator Reports Editor Debug Tools Windows Help

Profiles (Edit-F2)

#	Profiles (Edit-F2)
1	Profile0001
2	Profile0002
3	Profile0003
4	Profile0004
5	Profile0005
6	Profile0006
7	Profile0007
8	Profile0008
9	Profile0009
10	Profile0010

Config Value

- Send Recv T38 Fax
 - Tx T38 Fax File Name C:\Program Files\GL Communications Inc\MAPS-SIP\Fa...
 - T38 Rx Fax Path C:\Program Files\GL Communications Inc\MAPS-SIP\Fa...
 - T38 Rx Fax File Prefix SIP
 - Rx File Creation Type Random Number
- TxVideo
 - RTP Transport Type UDP
 - Video Trace File Path videofiles\pcmu-h264.hdl
 - Mute Audio RTP Stream Disable
 - Mute Video RTP Stream Disable
- MSRP Text Message Configurations
 - Send IM
 - IM File Name imfiles\send\msrpinputmessage.txt
 - IM File Iterations 1
 - Inter IM Timeout in msec 1000
 - IM Chunking Size 0
 - IM Success Report no
 - IM Failure Report yes
 - Recv IM
 - Rx IM File Path C:\Program Files\GL Communications Inc\MAPS-SIP\I...
 - Rx IM File Creation Type Sequence Number
 - Rx IM File Prefix SIP-IM

MsrpFileName

Select File

imfiles\send\msrpinputmessage.txt

Open

MsrpInputMessage.txt - Notepad

File Edit Format View Help

Hi, Welcome
This is MAPS SIP MSRP Simulator.
Test Message 1.
Test Message 2.
Test Message 3.

Add Insert Delete

Properties

Initialisation Errors Error Events Captured Errors Link Status Up=0 Down=0

MSRP Call Generation

MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - BulkCalls_10]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Ev	Result	Total Iterations	Completed Iterations
1	SipCallControl.gls	Profile0001	GL-MAPS_457_86849705-8370-17280@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
2	SipCallControl.gls	Profile0002	GL-MAPS_458_86849705-8374-14176@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
3	SipCallControl.gls	Profile0003	GL-MAPS_458_86849705-8366-2664@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
4	SipCallControl.gls	Profile0004	GL-MAPS_468_86849705-8358-4812@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
5	SipCallControl.gls	Profile0005	GL-MAPS_470_86849705-8363-17328@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
6	SipCallControl.gls	Profile0006	GL-MAPS_467_86849704-8354-16532@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
7	SipCallControl.gls	Profile0007	GL-MAPS_462_86849706-8386-17280@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
8	SipCallControl.gls	Profile0008	GL-MAPS_463_86849707-8394-14176@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
9	SipCallControl.gls	Profile0009	GL-MAPS_463_86849706-8390-2664@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
10	SipCallControl.gls	Profile0010	GL-MAPS_473_86849706-8381-4812@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

MAPS DUT

INVITE 15:39:35.705000

100 Trying 15:39:35.727000

180 Ringing 15:39:35.737000

200 OK 15:39:35.859000

ACK 15:39:35.861000

SEND 15:39:35.909000

200 OK 15:39:35.949000

REPORT 15:39:35.991000

SEND 15:39:35.991000

200 OK 15:39:35.992000

REPORT 15:39:36.010000

SEND 15:39:36.943000

200 OK 15:39:36.998000

REPORT 15:39:37.030000

SEND 15:39:37.030000

200 OK 15:39:37.032000

REPORT 15:39:37.040000

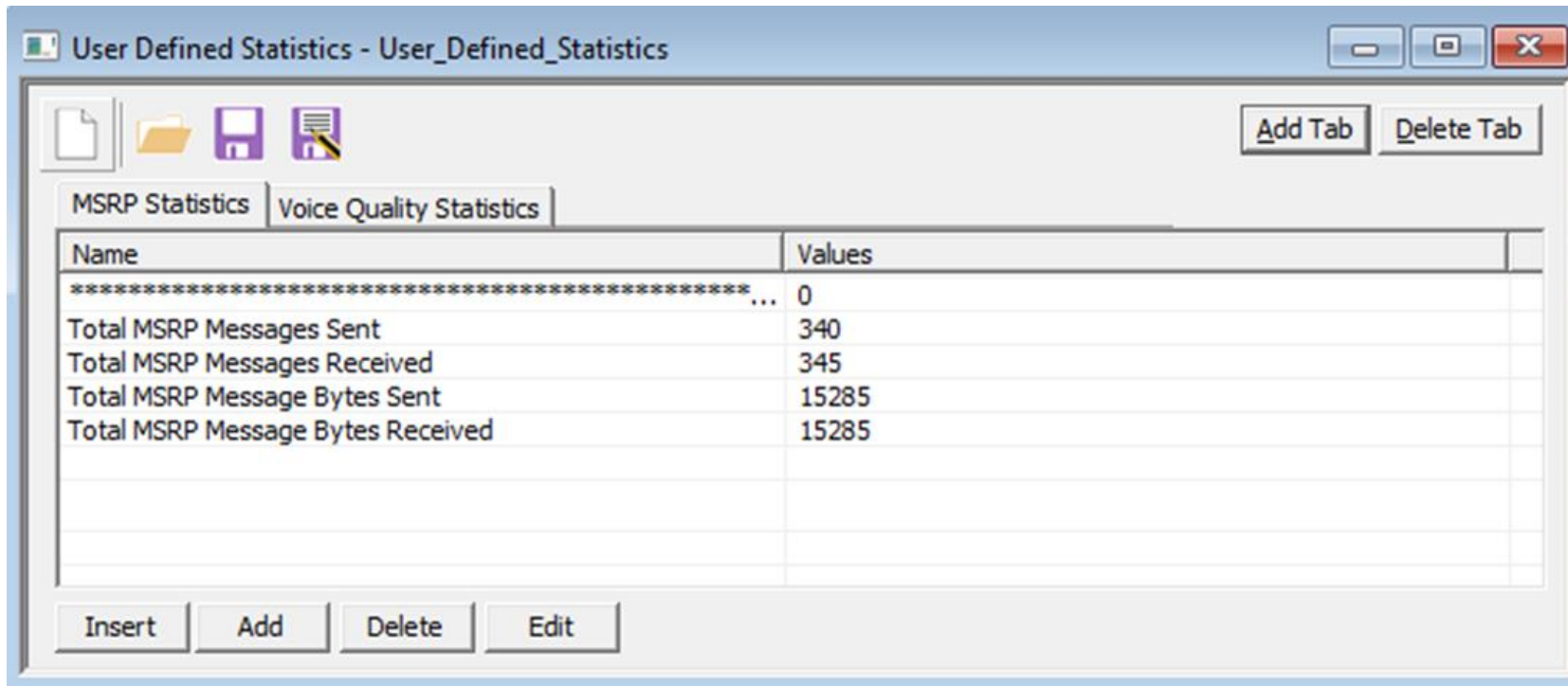
Find

MSRP glMapsMsrpBB9A66F9-153935908-6777 SEND
To-Path: msrp:///192.168.12.209:20148/GL_MAPS_302_86849888;tcp
From-Path: msrp:///192.168.12.216:20151/GL_MAPS_464_86849744;tcp
Message-ID: glMapsMsrpBB9A66F9-153935908-6776
Success-Report: no
Failure-Report: yes
Byte-Range: 1-270/270
Content-Type: text/plain

GL's Message Automation & Protocol Simulation (MAPS™) is a protocol simulation and conformance test tool that supports a variety of
-----glMapsMsrpBB9A66F9-153935908-6777\$

Scripts Message Sequence Event Config Script Flow

MSRP Statistics



User Defined Statistics - User_Defined_Statistics

MSRP Statistics | Voice Quality Statistics

Name	Values
*****...	0
Total MSRP Messages Sent	340
Total MSRP Messages Received	345
Total MSRP Message Bytes Sent	15285
Total MSRP Message Bytes Received	15285

Insert Add Delete Edit

Load Generation

Load Generation - LoadGendefault

Total Calls To Generate: * (* indicates no limit)
Max Active Calls: 2000 ☐ Unique Distributions Per Script

☒ Multi Distributions

Distributions	Description
Uniform	MinCR=40 , MaxCR=80 , Duration=10
Fixed	Call Rate=250 , Duration=10
Normal	MinCR=40 , MaxCR=80 , Duration=10

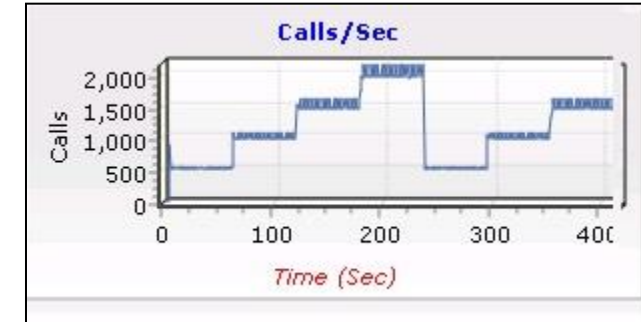
Scripts: ☒ Exclusive Profiles

Scripts	Profile
SipCallControl	Profile0003
Registration	Profile0005

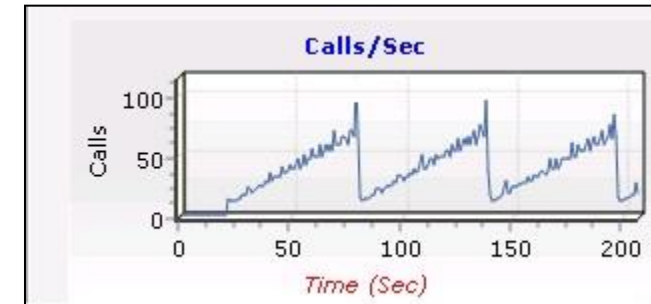
Buttons: Add, Delete, Add, Delete, Start, Pause

Stop Time: Days 0 Hours 0 Minutes 0
Start Time - 00:00:00.000
End Time - 00:00:00.000

Step Statistical Distribution



Ramp Statistical Distribution

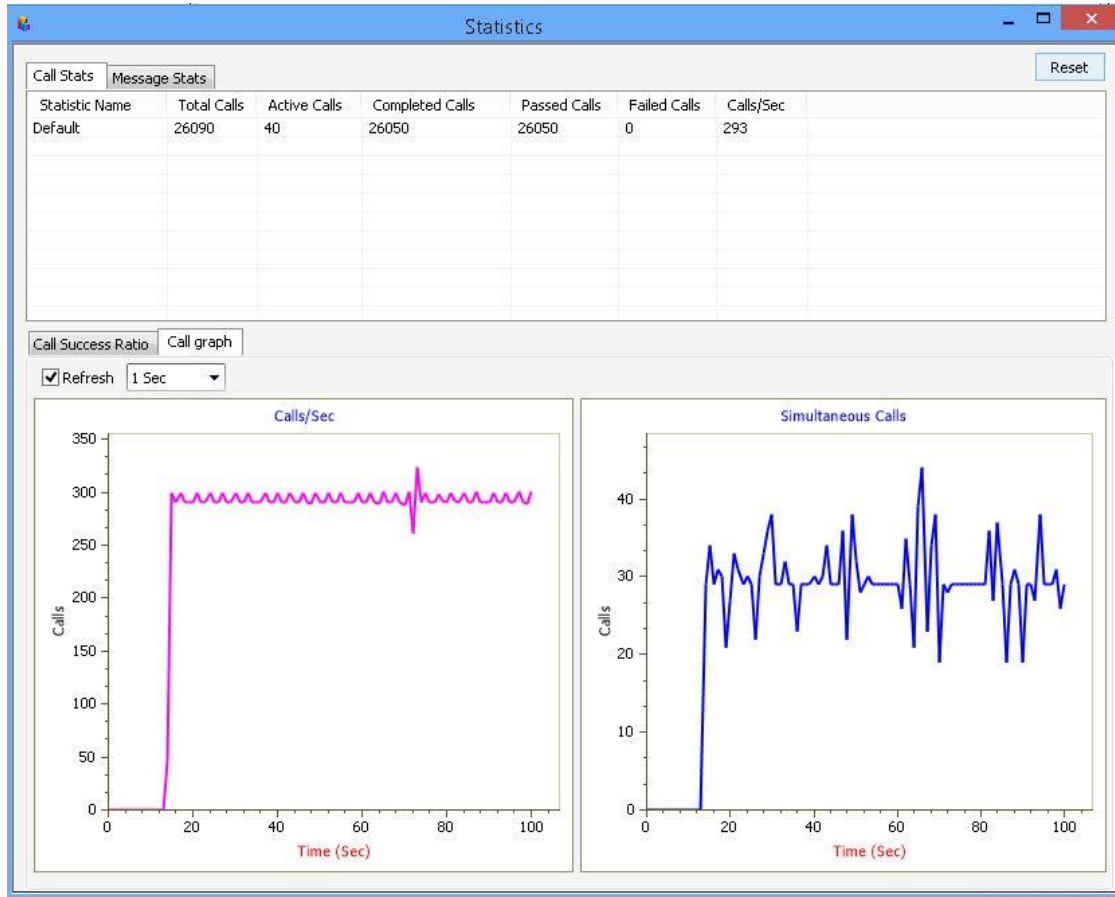


Saw-tooth Statistical Distribution

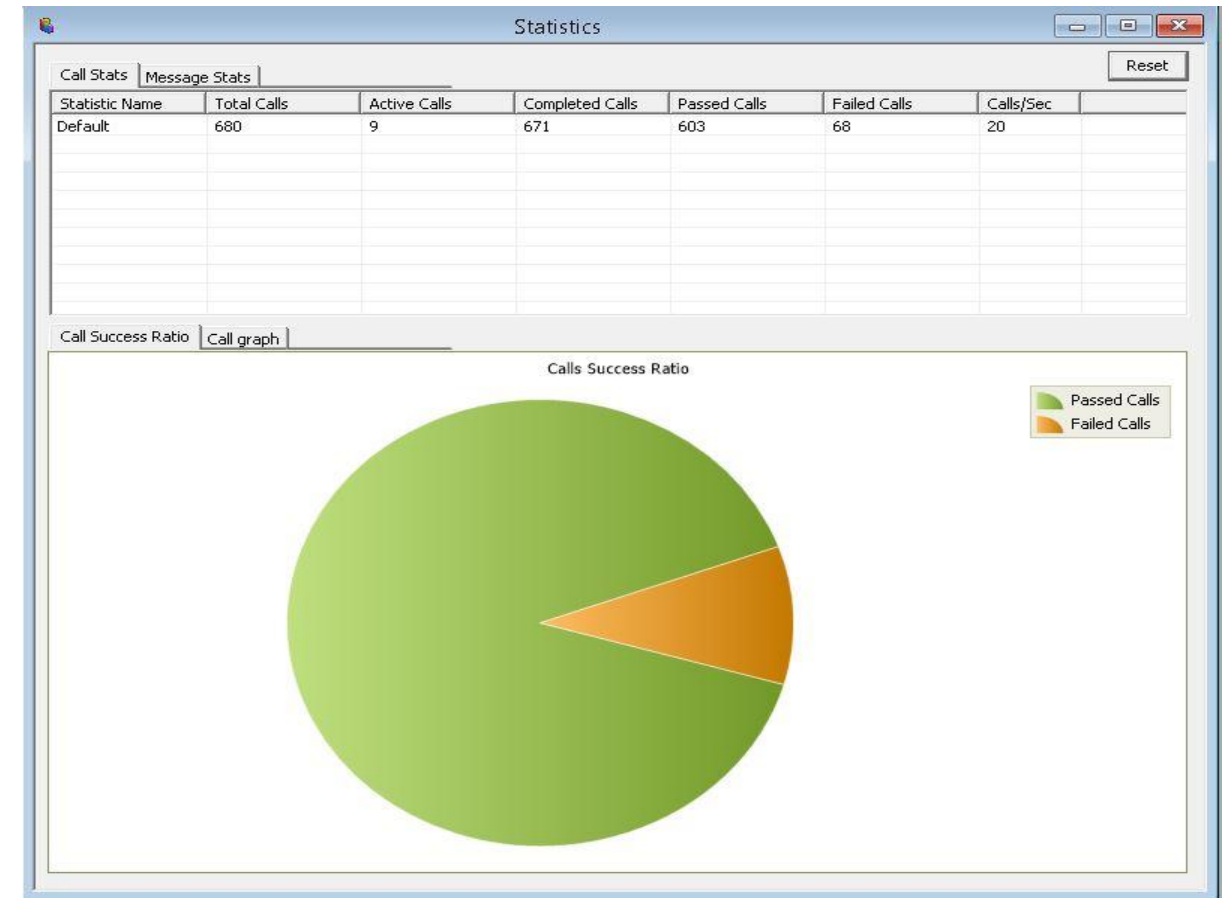


Success Call Ratio Statistics

Call Graph



Call Stats



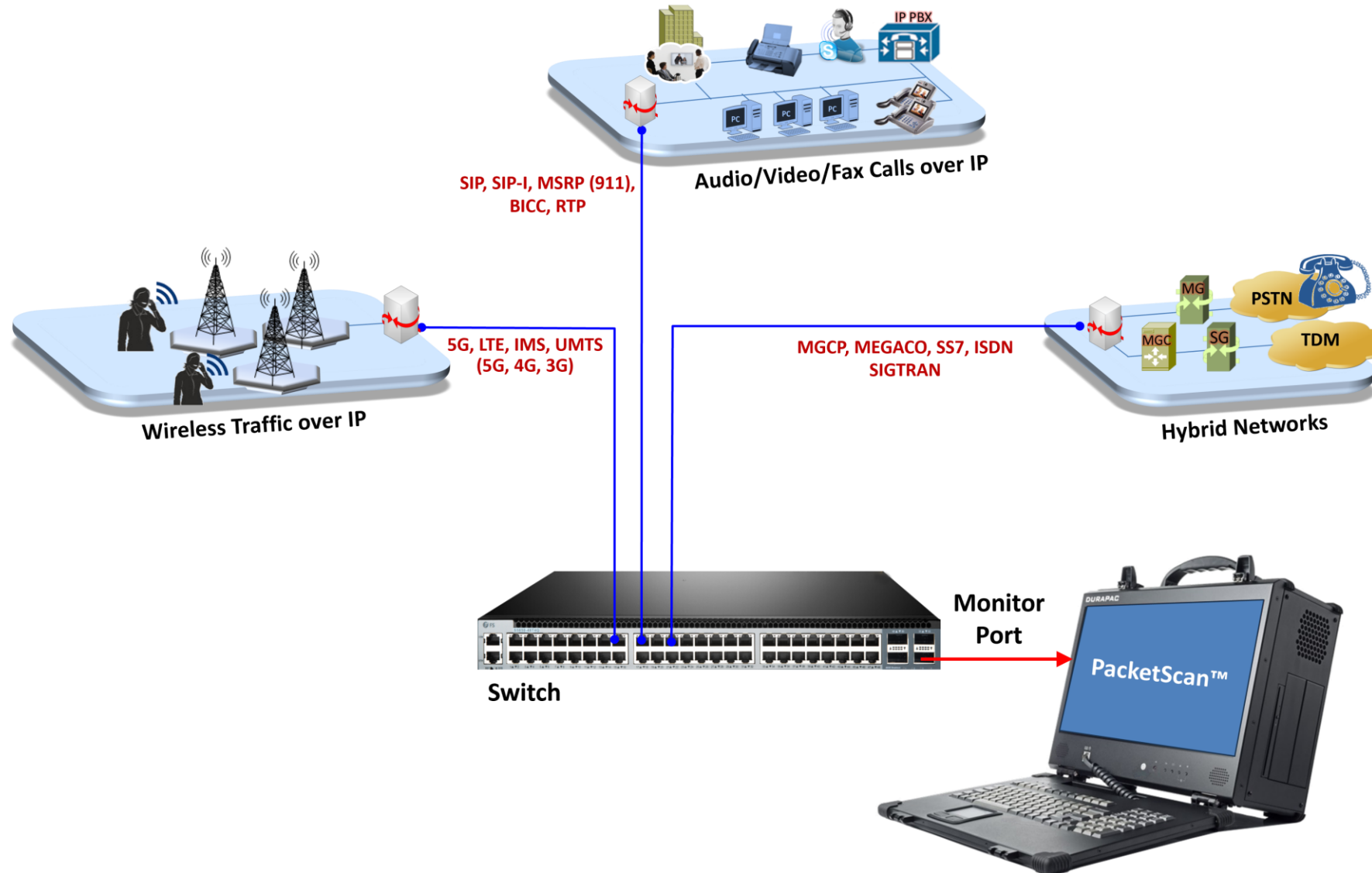
Message Statistics

Statistics				
Call Stats		Message Stats		Reset
Message Type	Tx Count	Rx Count	Retransmit Count	
100 INVITE	0	66040	0	
180 INVITE	0	66040	0	
200 BYE	0	46808	0	
200 INVITE	0	66040	0	
ACK	66040	0	0	
BYE	46808	0	0	
INVITE	66040	0	0	

SIP RTP Analyzer - PacketScan™

PacketScan™ VoIP Traffic Analysis

SIP / MSRP / H323 / MEGACO / MGCP / RTP / RTCP / Video Analysis



PacketScan™ Analyzer with SIP CDR

- 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

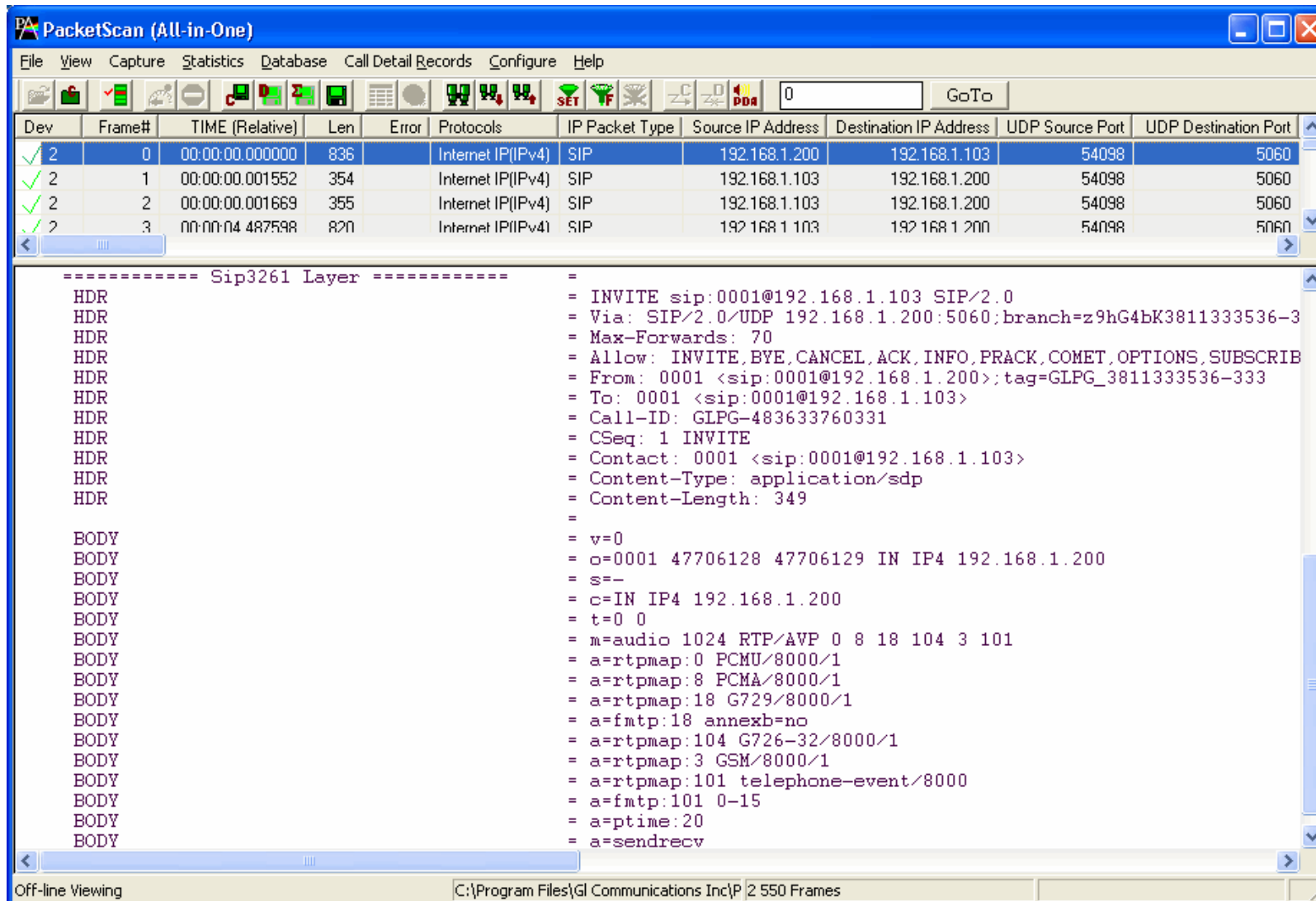
PacketScan 64-bit									
File View Capture Statistics Database Call Detail Records Configure Help									
Device	Frame#	TIME (Relative)	Length (Bytes)	Error	WireLen	Length/Protocol Type MAC	Packet Type MAC	SIP CSeq SIP	Destination IP Address IPv4.GTP
✓ 0	37	00:00:00.159853000	781			Internet IP(IPv4)	SIP	1 INVITE	
✓ 0	38	00:00:00.169817000	447			Internet IP(IPv4)	SIP	1 ACK	
✓ 0	39	00:00:00.170045000	447			Internet IP(IPv4)	SIP	1 ACK	
✓ 0	40	00:01:00.181815000	406			Internet IP(IPv4)	SIP	2 BYE	
✓ 0	41	00:01:00.182039000	406			Internet IP(IPv4)	SIP	2 BYE	
✓ 0	42	00:01:00.191764000	490			Internet IP(IPv4)	SIP	2 BYE	

0026 Length (Header + Data)	= 747 (x02EB)
0028 Checksum	= x9D21
===== SIP Layer =====	
HDR	= SIP/2.0 200 OK
HDR	= Via: SIP/2.0/UDP 192.168.12.117:5060;branch=z9hG4bK-177-30416881-12227-6588
HDR	= From: 0003 <sip:0003@192.168.12.117>;tag=FromTag-174-30416881-12221-6588
HDR	= To: 0003 <sip:0003@192.168.12.95>;tag=ToTag-156-30416929-8946-4008
HDR	= Call-ID: GL-MAPS-176-30416881-12225-6588@192.168.12.117
HDR	= CSeq: 1 INVITE
HDR	= Contact: 0003 <sip:0003@192.168.12.95>
HDR	= Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER,UPDATE
HDR	= Content-Type: application/sdp
HDR	= Content-Length: 255
BODY	= v=0
BODY	= o=0003 38820482 1 IN IP4 192.168.12.95
BODY	= s=SIP Call
BODY	= c=IN IP4 192.168.12.95
BODY	= t=0 0
BODY	= m=audio 1338 RTP/AVP 107 101
BODY	= a=rtpmap:107 AMR/8000
BODY	= a=fmtp:107 octet-align=0 ;mode-set=6
BODY	= a=rtpmap:101 telephone-event/8000
BODY	= a=fmtp:101 0-15
BODY	= a=ptime:20
BODY	= a=sendrecv

Call ID	Call Status	Protocol	Call Originating (Number / Address)	Call Destination (Number / Address)	Call Start Date & Time	Call Duration	Protocol Specific Info
2	Terminated	SIP	0001@192.168.12.117	0001@192.168.12.95	54400-28190-00 02:00:00.0...	01:40:21.264700	<SIPCallID> GL-MAPS-17...
3	Terminated	SIP	0010@192.168.12.117	0010@192.168.12.95	54400-28190-00 03:00:00.0...	01:40:19.158800	<SIPCallID> GL-MAPS-17...

Capture Rate : 0.01 Mbps C:\Program Files\GL Communications Inc\PacketScan\T\Fitr 60 of 2 396 frames
Missed Frames : 0

SIP Decode in PacketScan™



PacketScan™ PDA with SIP Call Summary

Traffic Analyzer - Summary View

File View Call Summary Settings Help

Sip Calls Show All Sessions

Call Summary Registraton Summary Alert Summary

Call #	SSRC	Payload	Packet Received	Conversational MOS/R-Factor	Listening MOS/R-Factor	Packets Discard...	Missing Packets...	Duplicate Packets...	Out Of Sequen...	Average Gap(ms)	Average Delay	Average Jitter	Average Inter Arri...	Cumulativ Packet ...	Max/Min Gap	Max/Min Delay	Max/M Jitter
Call#000001 Caller:0001@192.168.1.203 Callee:0001@192.168.1.213 CallId:GL-MAPS_1_185372727-4480-8320@192.168.1.203 Call StartTime:2015-01-15 14:48:24.106 Call Duration: 00:01:00.023																	
1	2217509121	PCMU/8000	1005	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	21.17 ...	1 / -1	0.45 /
1	2217326337	PCMU/8000	146	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.09 ...	0 / 0	0.07 /

TimeStamp

192.168.1.203

192.168.1.213

00.00.000	5060	INVITE	5060
00.00.007	5060	SIP/2.0 100 Trying	5060
00.00.009	5060	SIP/2.0 180 Ringing	5060
00.00.132	5060	SIP/2.0 200 OK	5060
00.00.137	5060	ACK	5060
00.00.141	1036	RTP (PCMU/8000)	1036
00.00.147	1036	RTP (PCMU/8000)	1036
01.00.156	5060	BYE	5060
01.00.160	5060	SIP/2.0 200 OK	5060

INVITE sip:0001@192.168.1.213 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_185372727-4481-8320

Max-Forwards: 70

Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UPDATE

From: "MapsSip" <sip:0001@192.168.1.203>;tag=FromTag_1_185372727-4478-8320

To: 0001 <sip:0001@192.168.1.213>

Call-ID: GL-MAPS_1_185372727-4480-8320@192.168.1.203

CSeq: 1 INVITE

Contact: 0010 <sip:0001@192.168.1.203>

Content-Type: application/sdp

Content-Length: 317

v=0

o=0001 33852938 33852938 IN IP4 192.168.1.203

s=-SIP Call

c=IN IP4 192.168.1.203

t=0 0

m=audio 1036 RTP/AVP 0 8 18 3 101

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

a=fmtp:18 annexb=no

a=rtpmap:3 GSM/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

aptime:20

Active Calls Graph Average Jitter Distribution E-Model RTP Packets Graph T.38 Analysis Call Graph Call Summary

PacketScan™ Fax T.38 Analysis

Traffic Analyzer - Summary View

File View Call Summary Settings Help

Sip Calls Show Fax Calls

Call Summary Registraton Summary Alert Summary

Call #	SSRC	Payload	Packet Received	Conversational MOS/R-Factor	Listening MOS/R-Fac...	Packets Discard...	Missing Packets...	Duplicate Packets...	Out Of Sequen...	Average Gap[ms]	Average Delay	Average Jitter	Average Inter Arri...	Cumulativ Packet ...	Max/Min Gap	Max/Min Delay	Max/Min Jitter	Max RTT
F Call#000001 Caller:4000@192.168.1.60 Callee:1000@192.168.1.60 CallId:1620788079-5060-3@192.168.1.244 Call StartTime:2011-09-02 12:35:48.113 Call Duration: 00:02:39.529																		
1	390089559	PCMU/8000	698	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.10 ...	0 / 0	0.08 / ...	0.0
1	1321168996	PCMU/8000	697	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.07 ...	0 / 0	0.08 / ...	0.0

00.30.538 5004 v21-preamble 5004

00.31.580 5004 NSF 5004

00.31.955 5004 CSI NUM:918040488401 5004

00.32.648 5004 DIS:DSR:ITU-T V.27 ter and V.29 5004

00.33.110 5004 no-signal 5004

00.34.559 5004 v21-preamble 5004

00.35.657 5004 TSI NUM:40488401 5004

00.36.402 5004 DCS:DSR:9600bps, ITU-T V.29 5004

00.36.622 5004 no-signal 5004

00.36.914 5004 v29-9600-training 5004

00.37.156 5004 t4-non-ecm-data:v29-9600: 0 pkts lost 5004

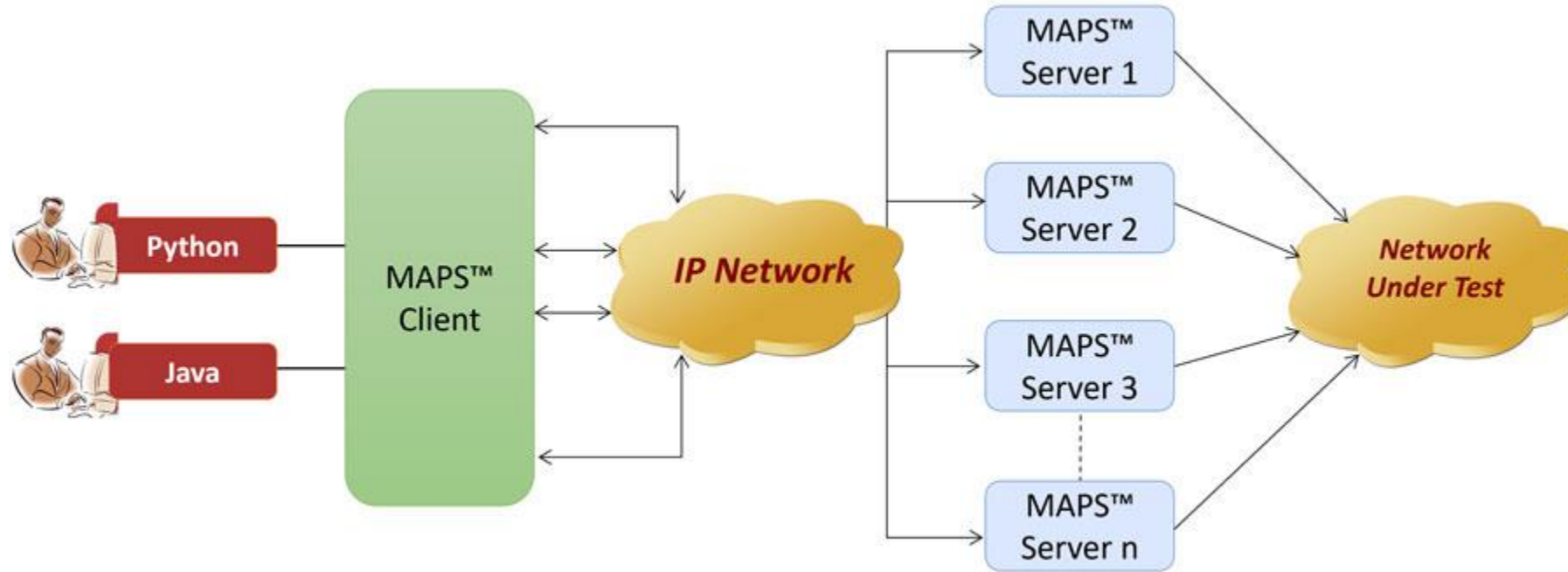
00.38.678 5004 no-signal 5004

===== T.38 Layer =====

```
UDPTLPacket = SEQUENCE
seq-number = INTEGER
Contents = 6
primary-ifp-packet = Open Type
Length = 12
IFPPacket = SEQUENCE
Preamble = 1
type-of-msg = CHOICE
Choice Index = 1
data = ENUMERATOR
Extensibility Marker = 0
Contents = 0 v21(0)
data-field = SEQUENCE OF
Iteration Count = 2
data-field = Instance 0
data-field = SEQUENCE
Preamble = 1
field-type = ENUMERATOR
Contents = 0 hdlc-data(0)
field-data = OCTET STRING
Length Determinant = 6
Contents = xFFC0042A20EB
data-field = Instance 1
data-field = SEQUENCE
Preamble = 0
field-type = ENUMERATOR
Contents = 0 hdlc-data(0)
```

Active Calls Graph Average Jitter Distribution E-Model RTP Packets Graph T.38 Analysis Call Graph Call Summary

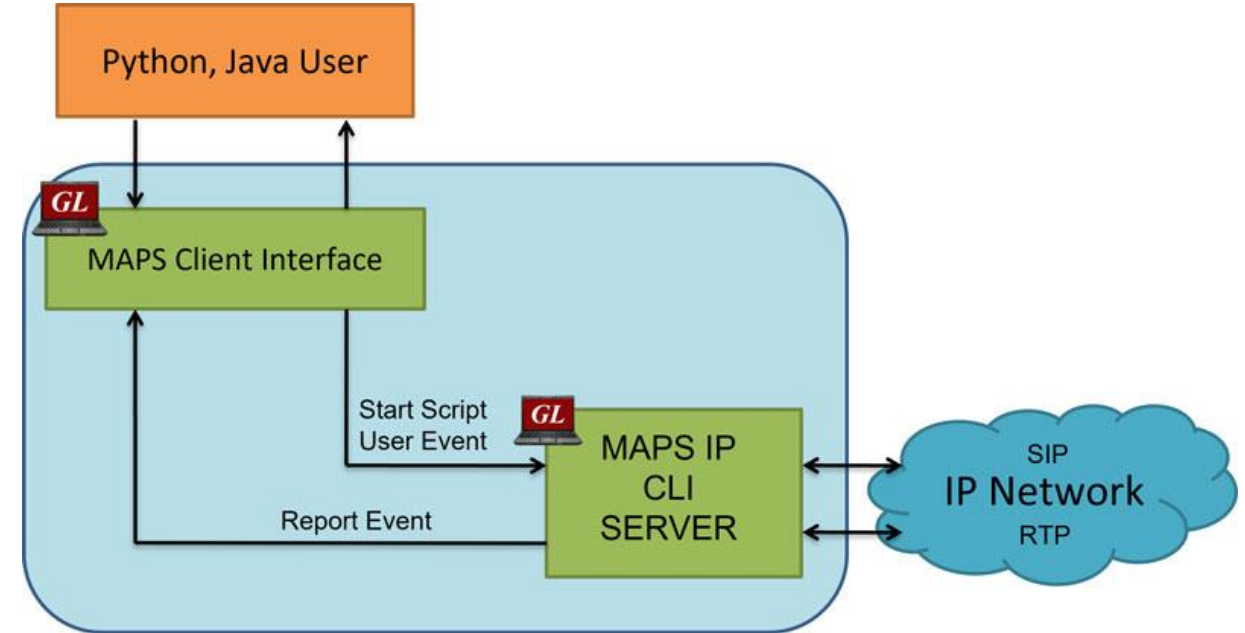
MAPS™ Command Line Interface



- MAPS™ can be configured as server-side application, to enable remote controlling through multiple command-line based clients. Supported clients include Python and Java
- 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, MAPS™ SIP CLI test system consists of the following -
 - Python, Java user communicating over TCP/IP
 - MAPS™ Client IFC, and MAPS™ SIP CLI Server



MAPS™ CLI Server and Python Client

```
CLI MapsCLI (SIP IETF)
File Edit View
View Latest Command

5 :: 2020-7-3 13:06:18.333000 : Start "TestBedDefault.xml" ;
5 :: 2020-7-3 13:06:18.442000 : LoadProfile "UserAgent_Profiles.xml"
5 :: 2020-7-3 13:06:18.770000 : Apply Global Configuration # "_EnableCLI"=1;
5 :: 2020-7-3 13:06:18.771000 : StartScript 1 "SipCallControl.gls" "Profile0001" 1 ;
5 :: 2020-7-3 13:06:18.880000 : UserEvent 1 "SetVariable"# "Contact"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:18.991000 : UserEvent 1 "SetVariable"# "AddressOfRecord"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:19.105000 : UserEvent 1 "SetVariable"# "RtpIpAddress"="192.168.12.216";
5 :: 2020-7-3 13:06:19.209000 : UserEvent 1 "SetVariable"# "To"="0001@192.168.12.209";
5 :: 2020-7-3 13:06:19.318000 : UserEvent 1 "SetVariable"# "Packetizationtime"="20";
5 :: 2020-7-3 13:06:19.429000 : UserEvent 1 "SetVariable"# "OvrCodecListSize"=3;
5 :: 2020-7-3 13:06:19.540000 : UserEvent 1 "SetVariable"# "OvrCodecList[0]"="G729";
5 :: 2020-7-3 13:06:19.646000 : UserEvent 1 "SetVariable"# "OvrPayloadList[0]"=18;
5 :: 2020-7-3 13:06:19.756000 : UserEvent 1 "SetVariable"# "OvrCodecList[1]"="PCMU";
5 :: 2020-7-3 13:06:19.864000 : UserEvent 1 "SetVariable"# "OvrPayloadList[1]"=0;
5 :: 2020-7-3 13:06:19.979000 : UserEvent 1 "SetVariable"# "OvrCodecList[2]"="telephone-event";
5 :: 2020-7-3 13:06:20.085000 : UserEvent 1 "SetVariable"# "OvrPayloadList[2]"=101;
5 :: 2020-7-3 13:06:20.192000 : UserEvent 1 "RTP_CreateSession";
5 :: 2020-7-3 13:06:24.349000 : UserEvent 1 "GetCallStatus";
5 :: 2020-7-3 13:06:24.460000 : UserEvent 1 "GetNegotiatedCodec";
5 :: 2020-7-3 13:06:24.569000 : UserEvent 1 "SendFile"# "TxFileName"="voicefiles\Send\G711\ULAW\Vijay.glw", "TxFileDuration"=10;
5 :: 2020-7-3 13:06:34.635000 : UserEvent 1 "GetVoiceQualityStats";
5 :: 2020-7-3 13:06:34.739000 : UserEvent 1 "SIP_TerminateCall";
5 :: 2020-7-3 13:06:34.848000 : UserEvent 1 "GetMessageCount";
5 :: 2020-7-3 13:06:34.957000 : UserEvent 1 "GetMessageInfo"# "Index"=0;
5 :: 2020-7-3 13:06:35.067000 : UserEvent 1 "GetMessageInfo"# "Index"=1;
5 :: 2020-7-3 13:06:35.178000 : UserEvent 1 "GetMessageInfo"# "Index"=2;
5 :: 2020-7-3 13:06:35.290000 : UserEvent 1 "GetMessageInfo"# "Index"=3;
5 :: 2020-7-3 13:06:35.397000 : UserEvent 1 "GetMessageInfo"# "Index"=4;
5 :: 2020-7-3 13:06:35.510000 : UserEvent 1 "GetMessageInfo"# "Index"=5;
5 :: 2020-7-3 13:06:35.616000 : UserEvent 1 "GetMessageInfo"# "Index"=6;
5 :: 2020-7-3 13:06:35.724000 : StopScript 1;
ServerLog:errCode = 0,errString = connection has been gracefully closed for ClientId =5
```

```
Python 3.7.3 Shell
File Edit Shell Debug Options Window Help

Python 3.7.3 (v3.7.3:ef4ec6ed12, Mar 25 2019, 22:22:05) [MSC v.1916 64 bit (AMD64)] on win32
Type "help", "copyright", "credits" or "license()" for more information.
>>>
RESTART: C:\Program Files\GL Communications Inc\MAPS-SIP\PythonClient\examples\SIP\SipBasicCall.py
SERVER INITIALIZED
CONNECTED
Negotiated Codec = PCMU
0
CMOS = 4.19531
LMOS = 4.19531
CR_FACTOR = 93
LR_FACTOR = 93
TX_PACKETS = 501
RX_PACKETS = 712
LOST_PACKETS = 0
DISCARDED_PACKETS = 0
OUT_OF_SEQ_PACKETS = 0
DUPLICATE_PACKETS = 0
AVG_JITTER = 0.125

12:24:01.120 -> INVITE
INVITE sip:0001@192.168.12.209 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.216:5060;branch=z9hG4bK-4-1348328288-22704-17372
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER,UPDATE
From: 1231230001 <sip:1231230001@192.168.12.216>;tag=FromTag-1-1348328288-22701-17372
To: 0001 <sip:0001@192.168.12.209>
Call-ID: GL-MAPS-3-1348328288-22703-17372@192.168.12.216
CSeq: 1 INVITE
Contact: 1231230001 <sip:1231230001@192.168.12.216>
Content-Type: application/sdp
Content-Length: 269

v=0
o=1231230001 39377840 1 IN IP4 192.168.12.216
s=SIP Call
c=IN IP4 192.168.12.216
t=0 0
m=audio 1024 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=sendrecv
```

NetSurveyorWeb™

- 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

GL Communications Inc.
Telecommunication Products and Consulting
NetSurveyorWeb 3.2.12 - Real Time Monitoring System

Protocol Type: VOIP

System Status as of 2013-02-12 15:33:19

Admin

Date Range: 2013-02-12 To 2013-02-12
Hour Range: 00:00:01 To 23:59:59

Enable Alarms

CDR Data

120 Secs

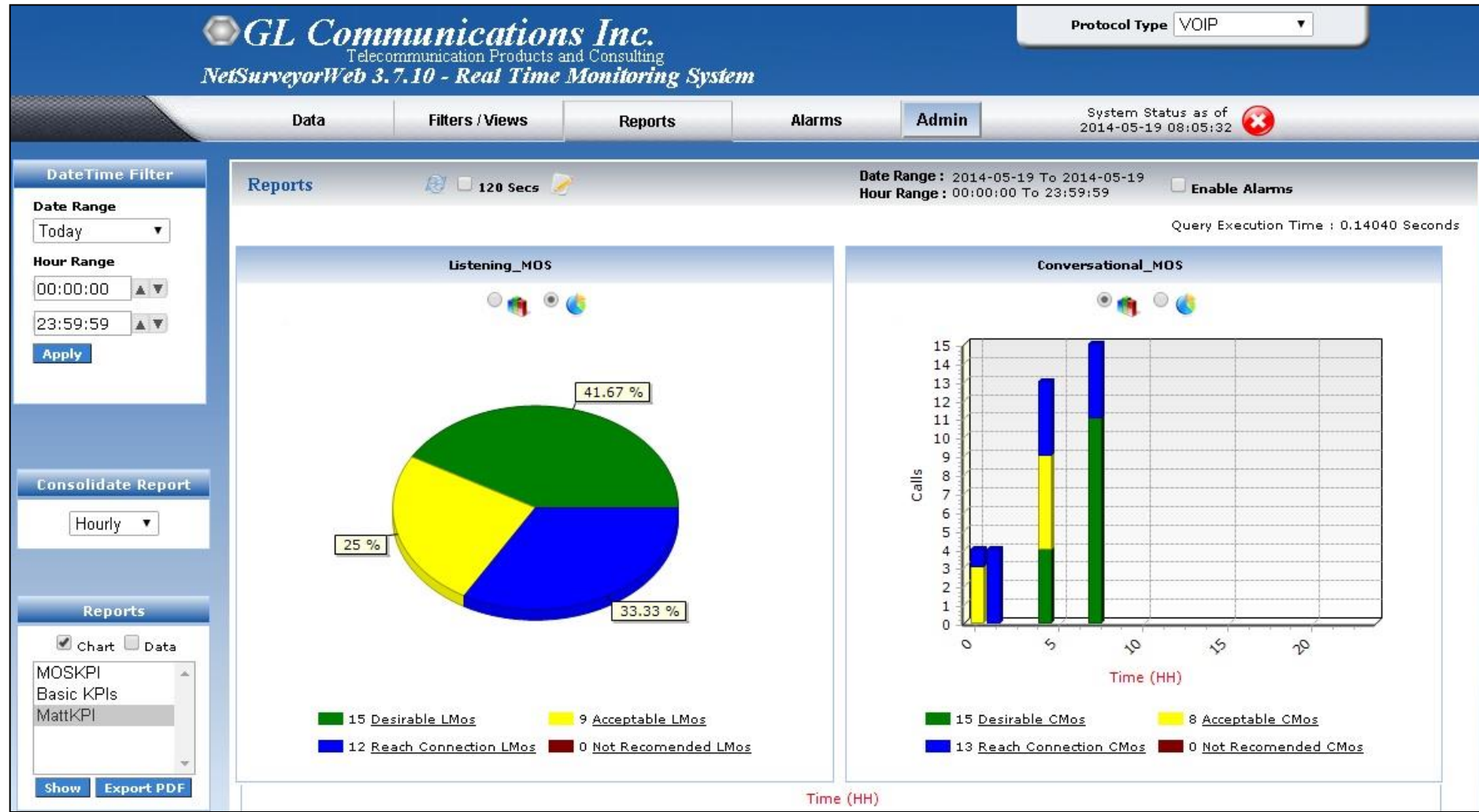
Export as PDF Export as CSV (Filter OFF / No Filters Added) Query Execution Time: 0.01000 Seconds

Quick Search: Trafficsumid GO

View Records Per Page: 20 Sort Expression: STARTTIME DE

	Trafficsumid	Probename	Calling Number	Called Number	Starttime	Duration	Payload1	Payload2	Cor																								
Call Flow	2062488	PacketProbe0	13016704784@px1.nexvortex.com	12027621401@px1.nexvortex.com	2013-02-12 15:30:59.000000	00:01:20.000977	PCMU/8000	PCMU/8000	4.2																								
Call Flow	2062487	PacketProbe0	103@192.168.20.45	912027621401@192.168.20.45;user=phone	2013-02-12 15:30:59.000000	00:01:20.000804	G722/16000	G722/16000	3.9																								
Call Flow	2062486	PacketScan	69.54.92.148::63022	192.168.20.45::17728	2013-02-12 15:30:57.214	00:00:00.000000	PCMU/8000	PCMU/8000	4.2																								
Call Flow	2062490	PacketScan	13016704784@px1.nexvortex.com	12027621401@px1.nexvortex.com	2013-02-12 15:30:53.572	00:01:20.000976	PCMU/8000	PCMU/8000	4.2																								
<table border="1"><thead><tr><th>SSRC#</th><th>Payload</th><th>Total Packet Count</th><th>Missing Packet Count/(%)</th><th>Dupl. Packet Count/(%)</th><th>Re-ordered Packet Count/(%)</th><th>Packets Discarded/(%)</th><th>Conversational MOS/R</th></tr></thead><tbody><tr><td>621711797</td><td>PCMU/8000</td><td>4054</td><td>0/0</td><td>0/0</td><td>0/0</td><td>0/0</td><td>4.2/93</td></tr><tr><td>2908065327</td><td>PCMU/8000</td><td>4055</td><td>0/0</td><td>0/0</td><td>0/0</td><td>0/0</td><td>4.2/93</td></tr></tbody></table>										SSRC#	Payload	Total Packet Count	Missing Packet Count/(%)	Dupl. Packet Count/(%)	Re-ordered Packet Count/(%)	Packets Discarded/(%)	Conversational MOS/R	621711797	PCMU/8000	4054	0/0	0/0	0/0	0/0	4.2/93	2908065327	PCMU/8000	4055	0/0	0/0	0/0	0/0	4.2/93
SSRC#	Payload	Total Packet Count	Missing Packet Count/(%)	Dupl. Packet Count/(%)	Re-ordered Packet Count/(%)	Packets Discarded/(%)	Conversational MOS/R																										
621711797	PCMU/8000	4054	0/0	0/0	0/0	0/0	4.2/93																										
2908065327	PCMU/8000	4055	0/0	0/0	0/0	0/0	4.2/93																										
Call Flow	2062489	PacketScan	103@192.168.20.45	912027621401@192.168.20.45;user=phone	2013-02-12 15:30:53.162	00:01:20.000803	G722/16000	G722/16000	3.9																								
Call Flow	2062478	PacketProbe0	115@192.168.20.45	114@192.168.20.140	2013-02-12 15:28:34.000000	00:09:30.000060	G722/16000	G722/16000	3.9																								
Call Flow	2062474	PacketProbe0	114@192.168.20.45	117@192.168.20.126	2013-02-12 15:28:12.000000	00:00:00.000000																											
Call Flow	2062475	PacketProbe0	114@192.168.20.45	117@192.168.20.45;user=phone	2013-02-12 15:28:11.000000	00:00:00.000000																											
Call Flow	2062476	PacketScan	114@192.168.20.45	117@192.168.20.126	2013-02-12 15:28:06.515	00:00:00.000000																											
Call Flow	2062477	PacketScan	114@192.168.20.45	117@192.168.20.45;user=phone	2013-02-12 15:28:05.316	00:00:00.000000																											
Call Flow	2062468	PacketProbe0	19134163019@192.168.20.45	102@192.168.20.129	2013-02-12 15:23:09.000000	00:00:00.000000																											
Call Flow	2062469	PacketScan	19134163019@192.168.20.45	102@192.168.20.129	2013-02-12 15:23:03.161	00:00:00.000000																											
Call Flow	2062471	PacketProbe0	19134163019@sip.skype.com	12407506065@sip.skype.com	2013-02-12 15:22:54.000000	00:01:03.000459	PCMU/8000	PCMU/8000	4.2																								
Call Flow	2062472	PacketScan	19134163019@sip.skype.com	12407506065@sip.skype.com	2013-02-12 15:22:48.621	00:01:03.000458	PCMU/8000	PCMU/8000	4.2																								
Call Flow	2062470	PacketProbe0	19134163019@sip.skype.com	12407506065@sip.skype.com	2013-02-12 15:22:48.000000	00:00:00.000000	PCMU/8000	PCMU/8000	0																								

NetSurveyorWeb™ – Reports



Thank you