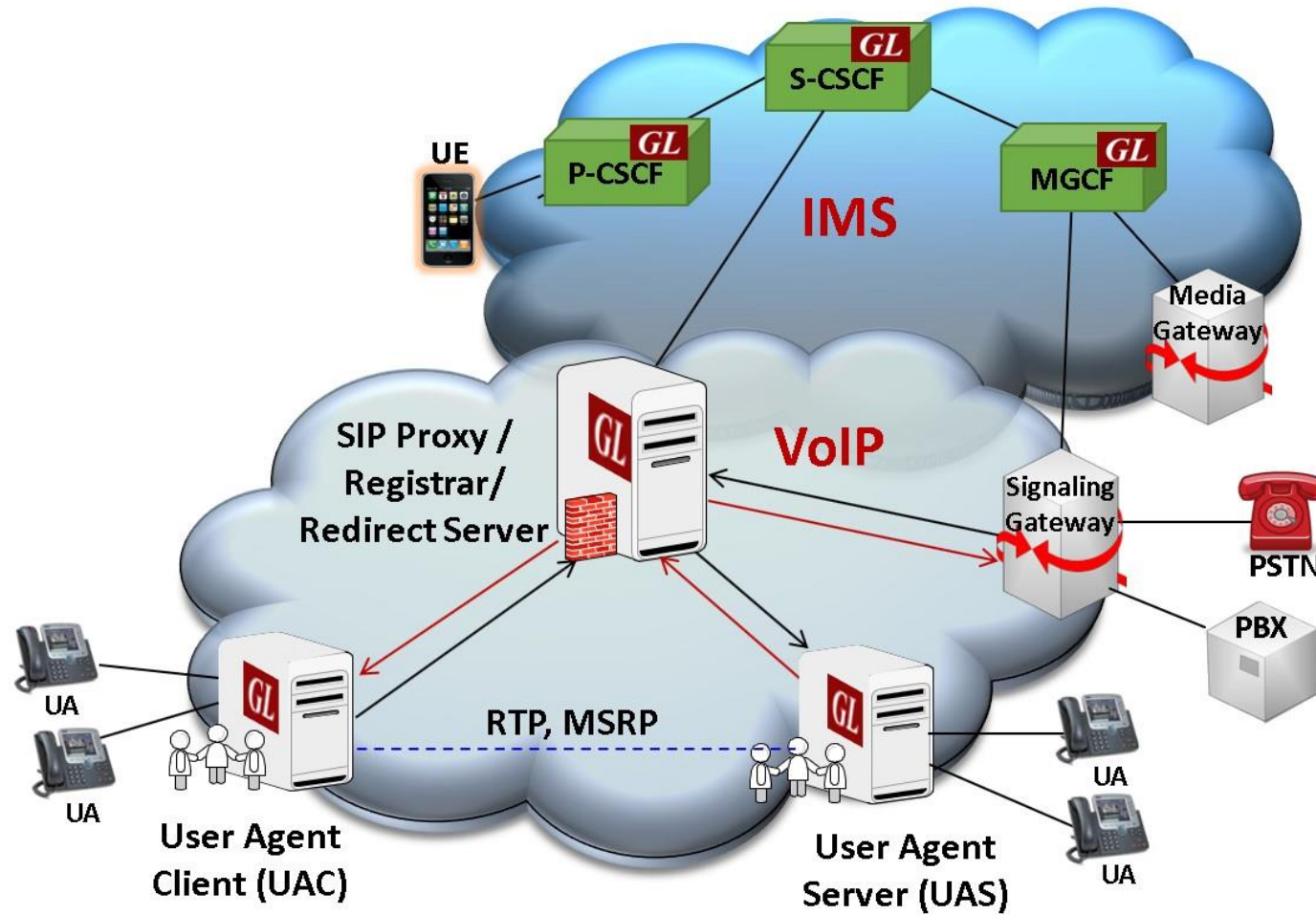

MAPS™ SIP

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

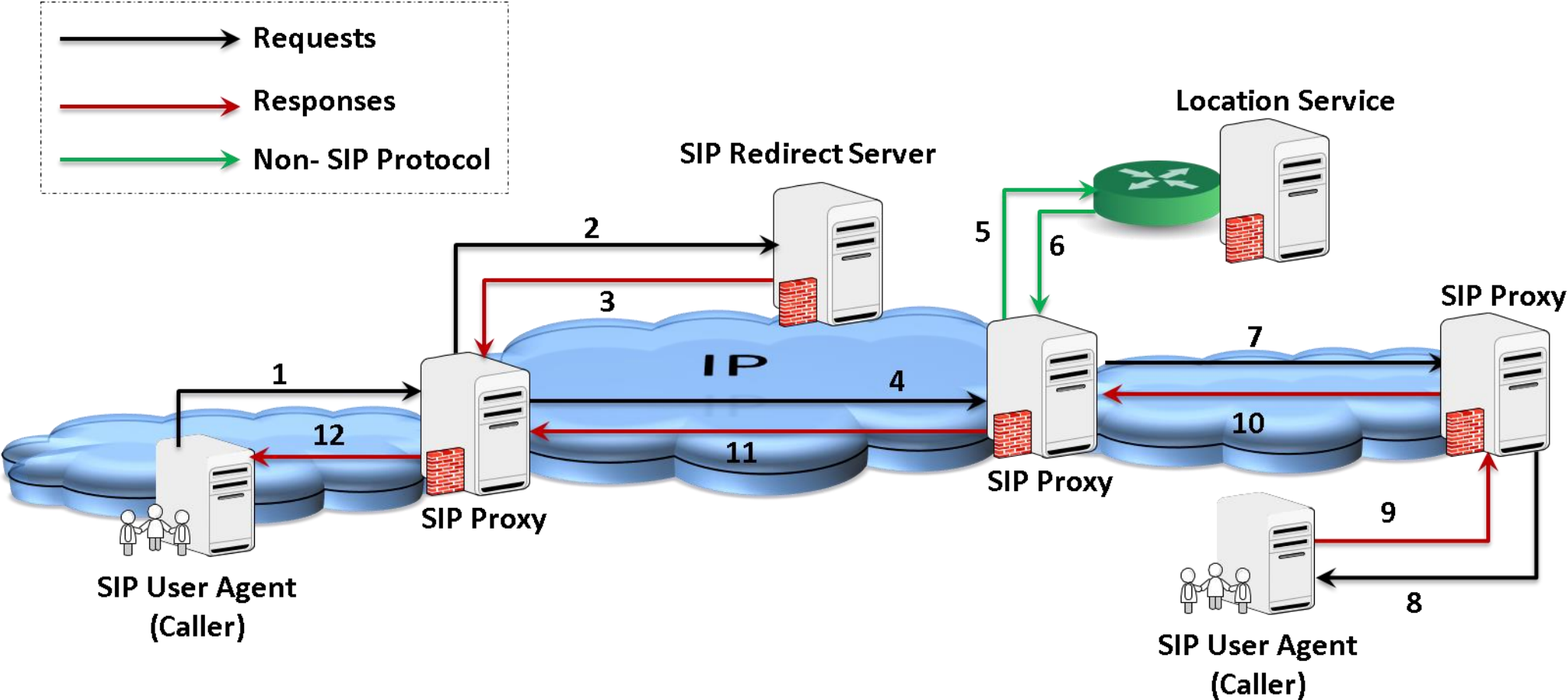


MAPS™ SIP
Normal RTP Traffic Generation
(2000 simultaneous calls)

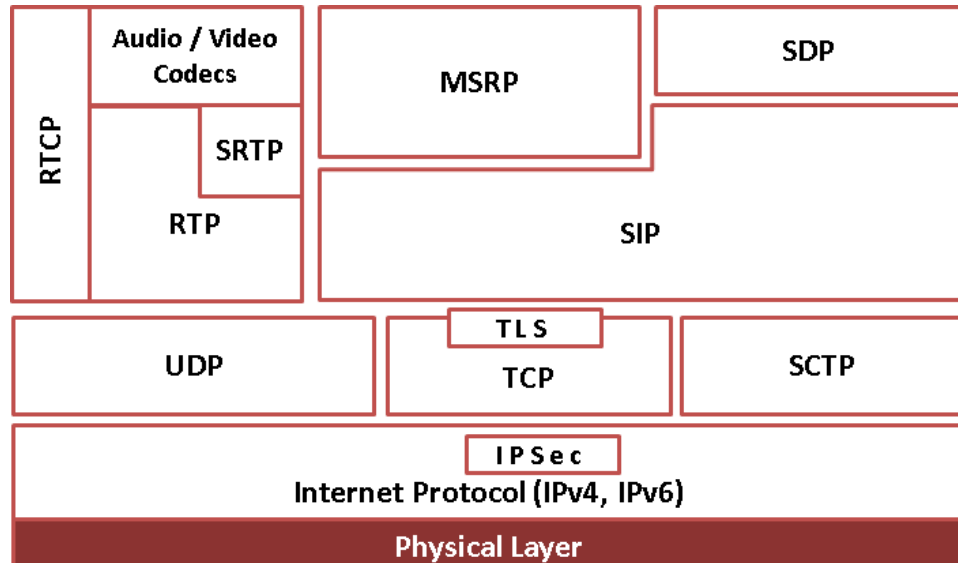


MAPS™ SIP (w/ 4 x 1G cards)
HD RTP Traffic Generator
32,000 Simultaneous Calls (with RTP Traffic)

SIP Architecture and Entities

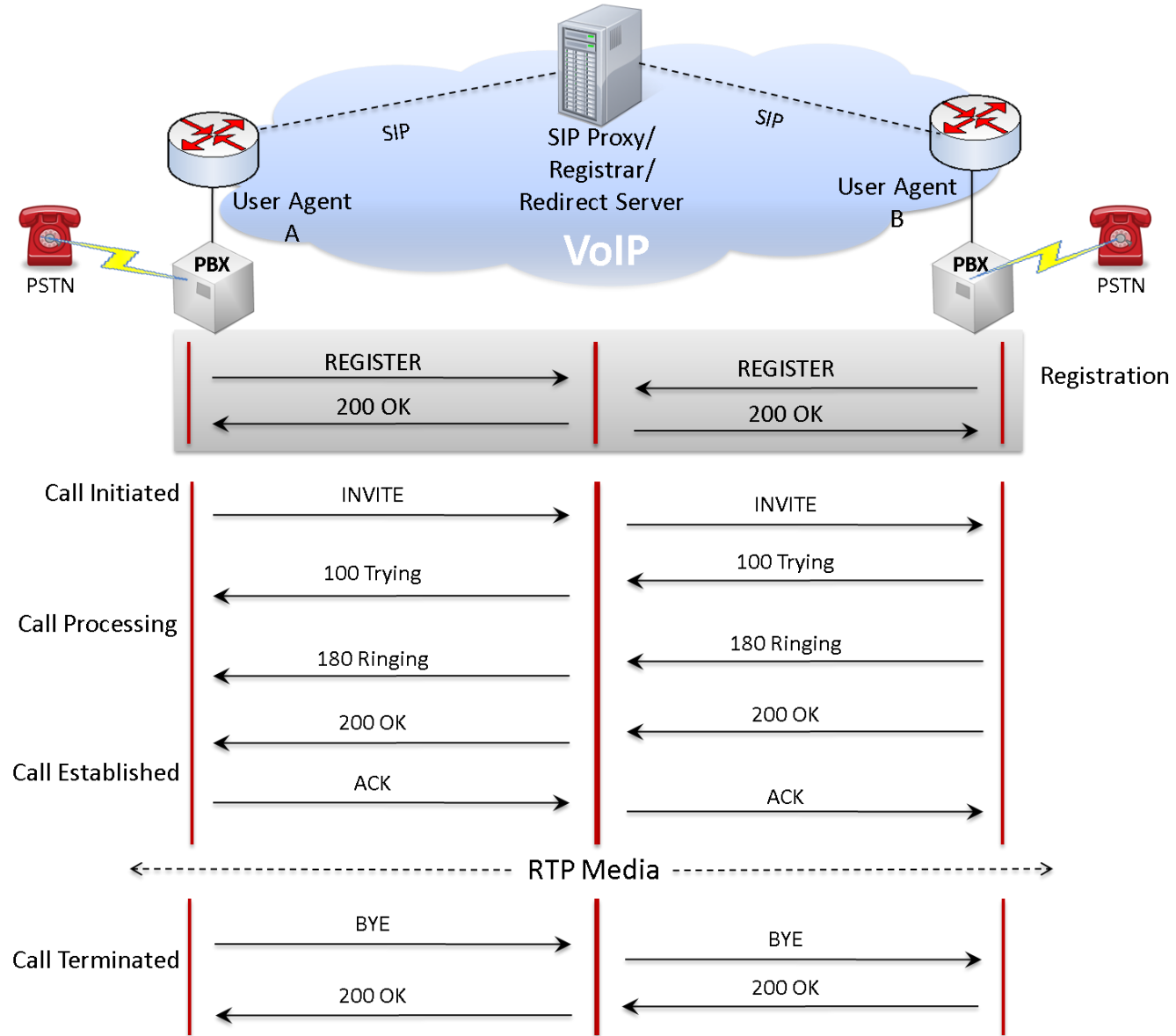


SIP Protocol Stack



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



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 32,000 SP Endpoints.



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 TCL, Python, VB, Java, and .Net

MAPS™ SIP Highlights

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

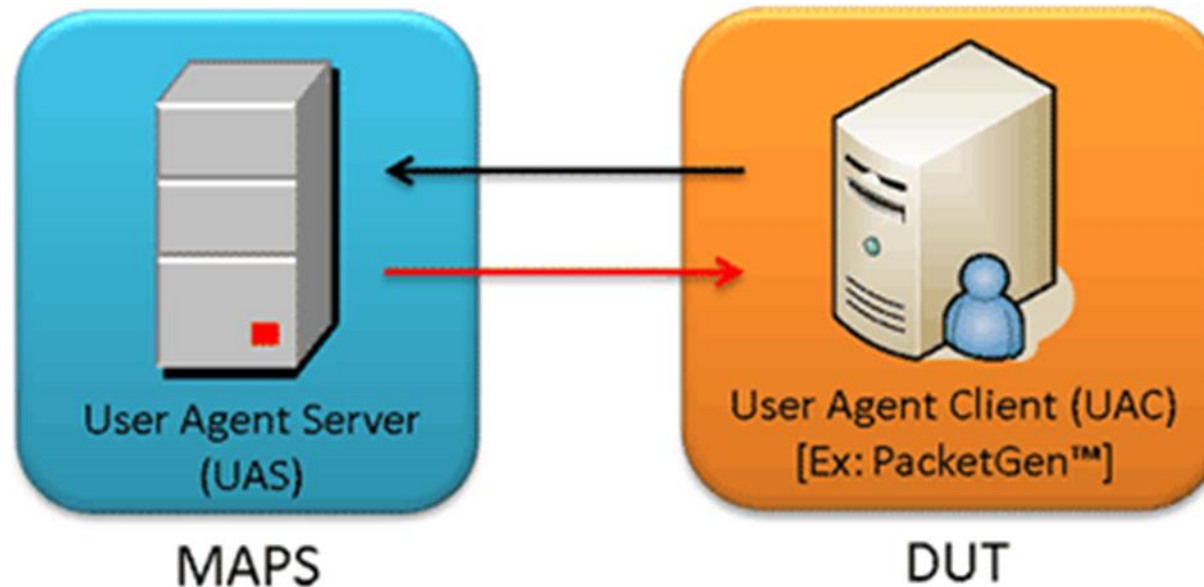
- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX)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

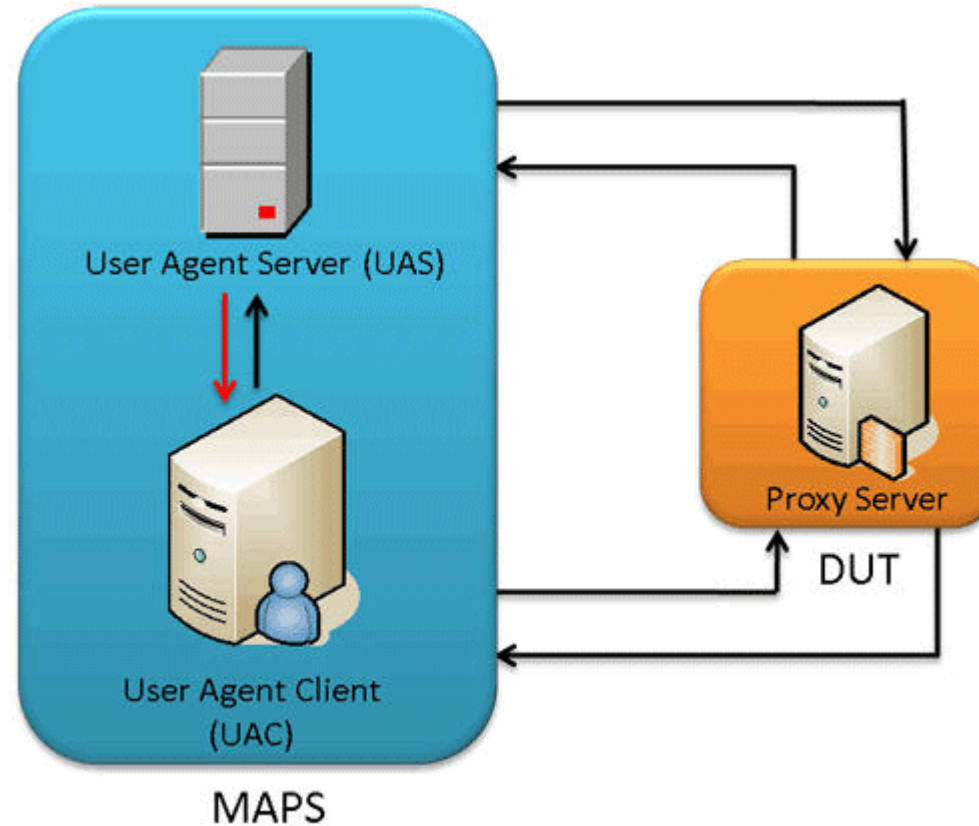


MAPS™ SIP Configured as UAC / UAS

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

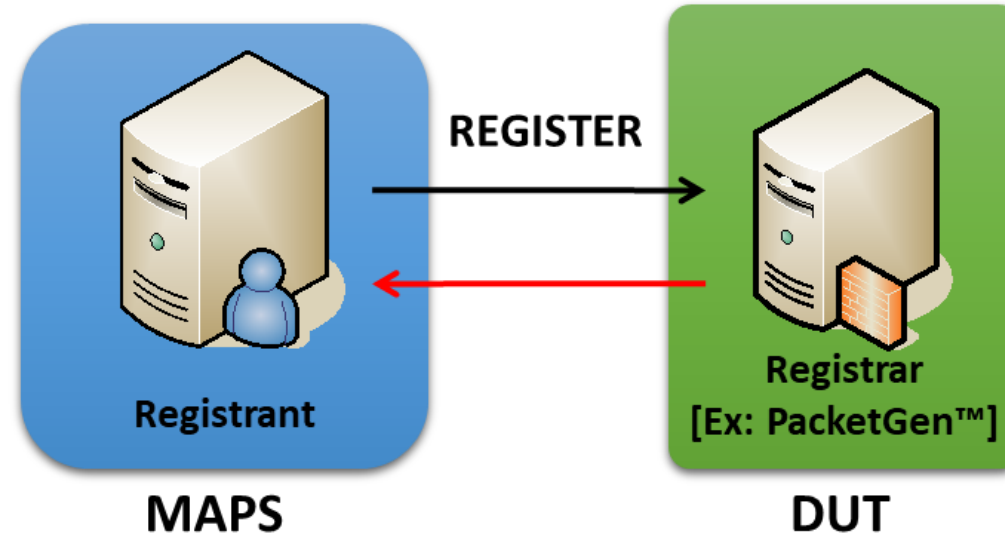


MAPS™ SIP Configured as Registrant

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

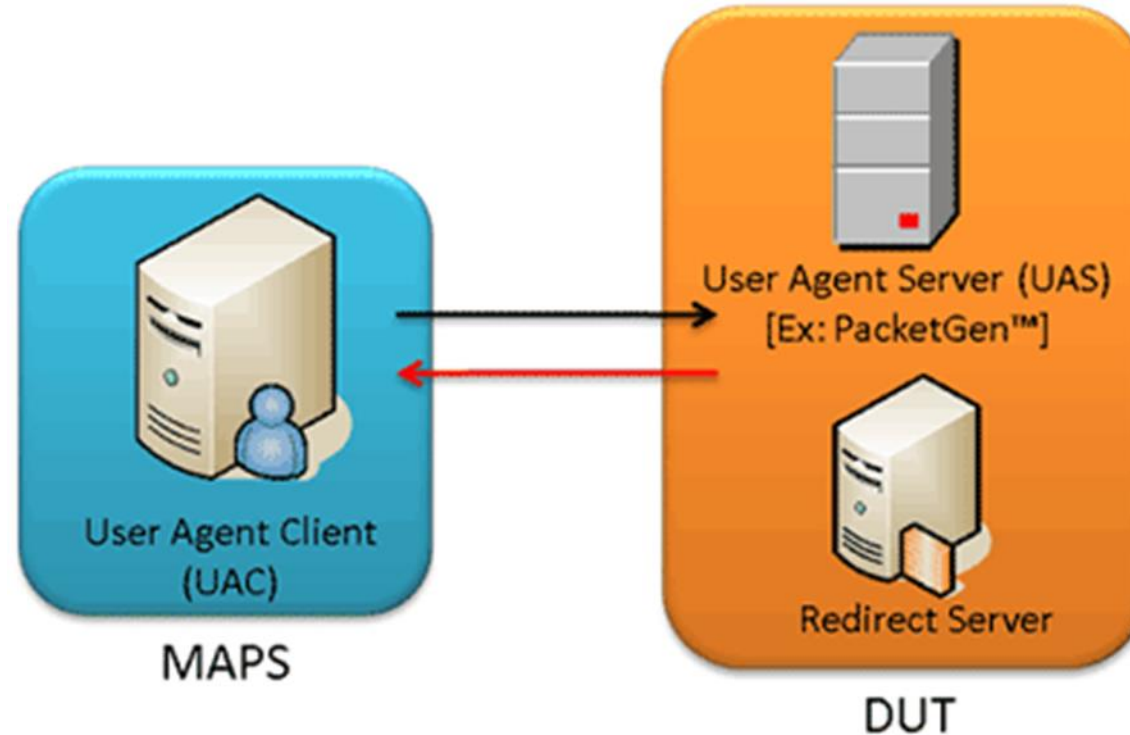


MAPS™ SIP Configured as UAC

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

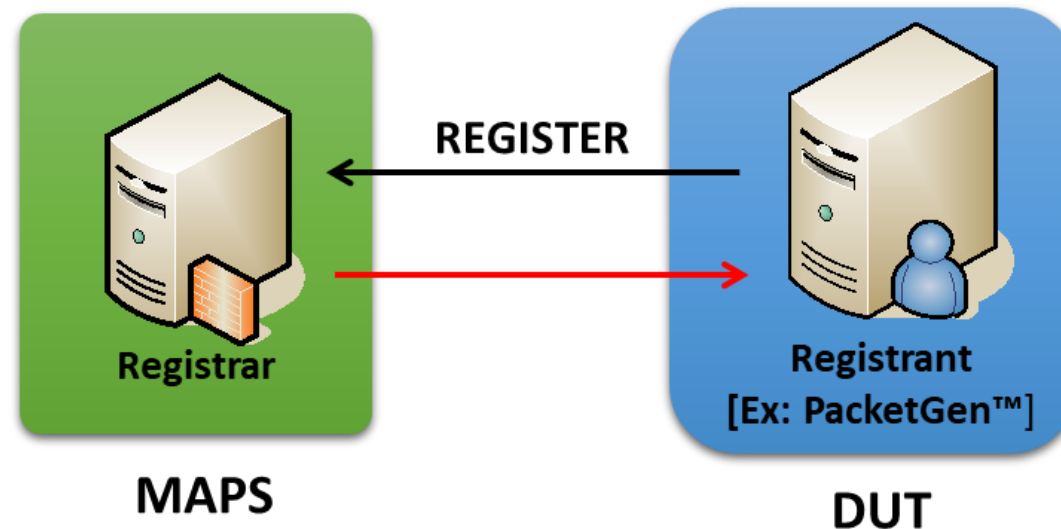


MAPS™ SIP Configured as Registrar

Testing Registrant

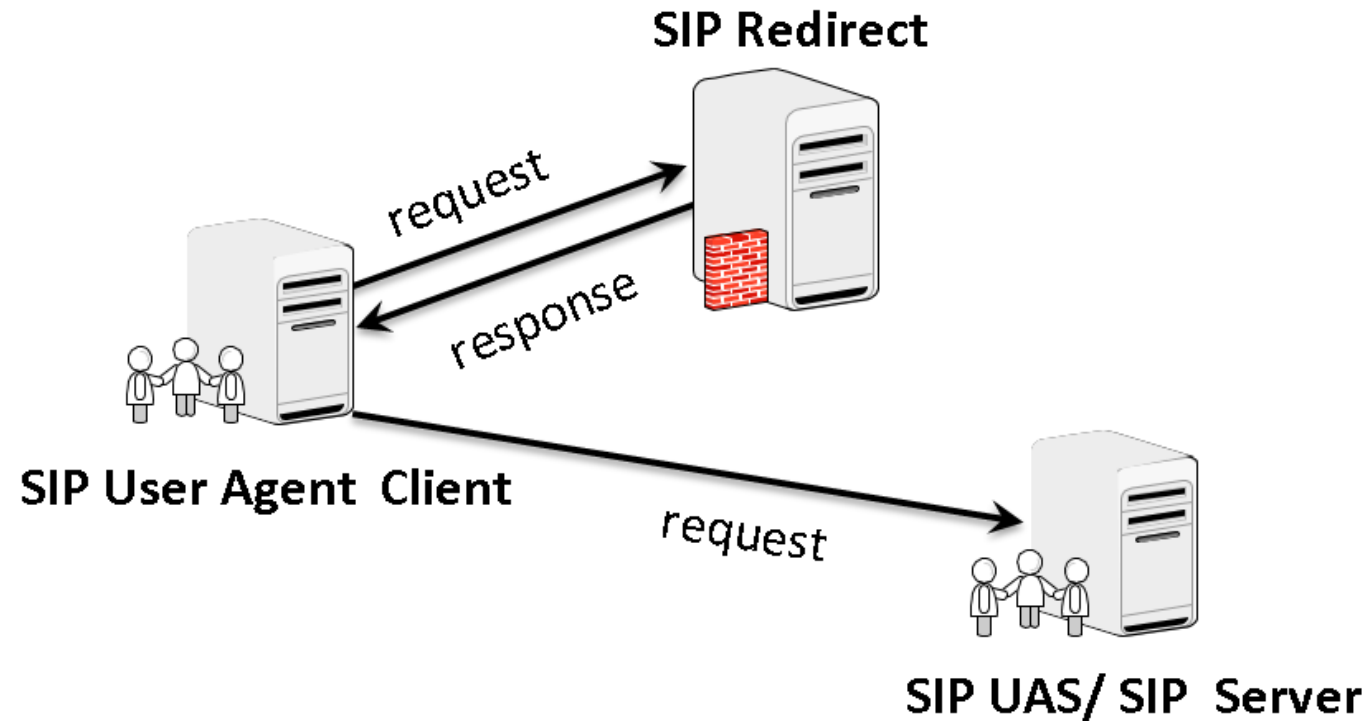
Scenario: MAPS™ acting as Registrar and testing Registrant

- 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)

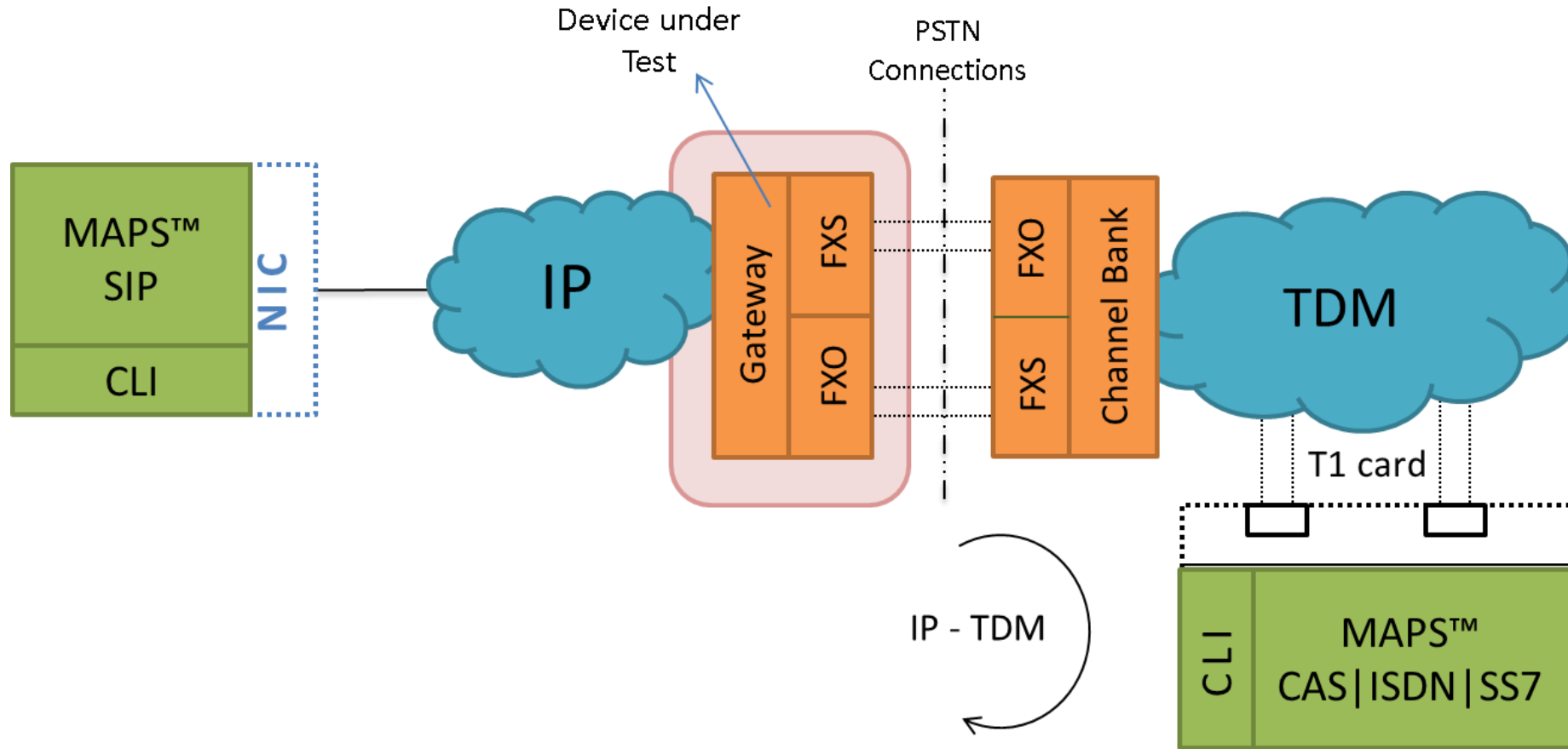
- Registrant – Registers with Registrar
- Call with Auto Traffic of RTP Action
- Traffic Impairments
- Simulates IVR (Interactive Voice Response) for RTP traffic
- Call through Proxy
- Sequential and Random Generation of Calls
- Simultaneous Generation of Calls
- Load Generation (Stress Testing)

Call Reception (UAS)

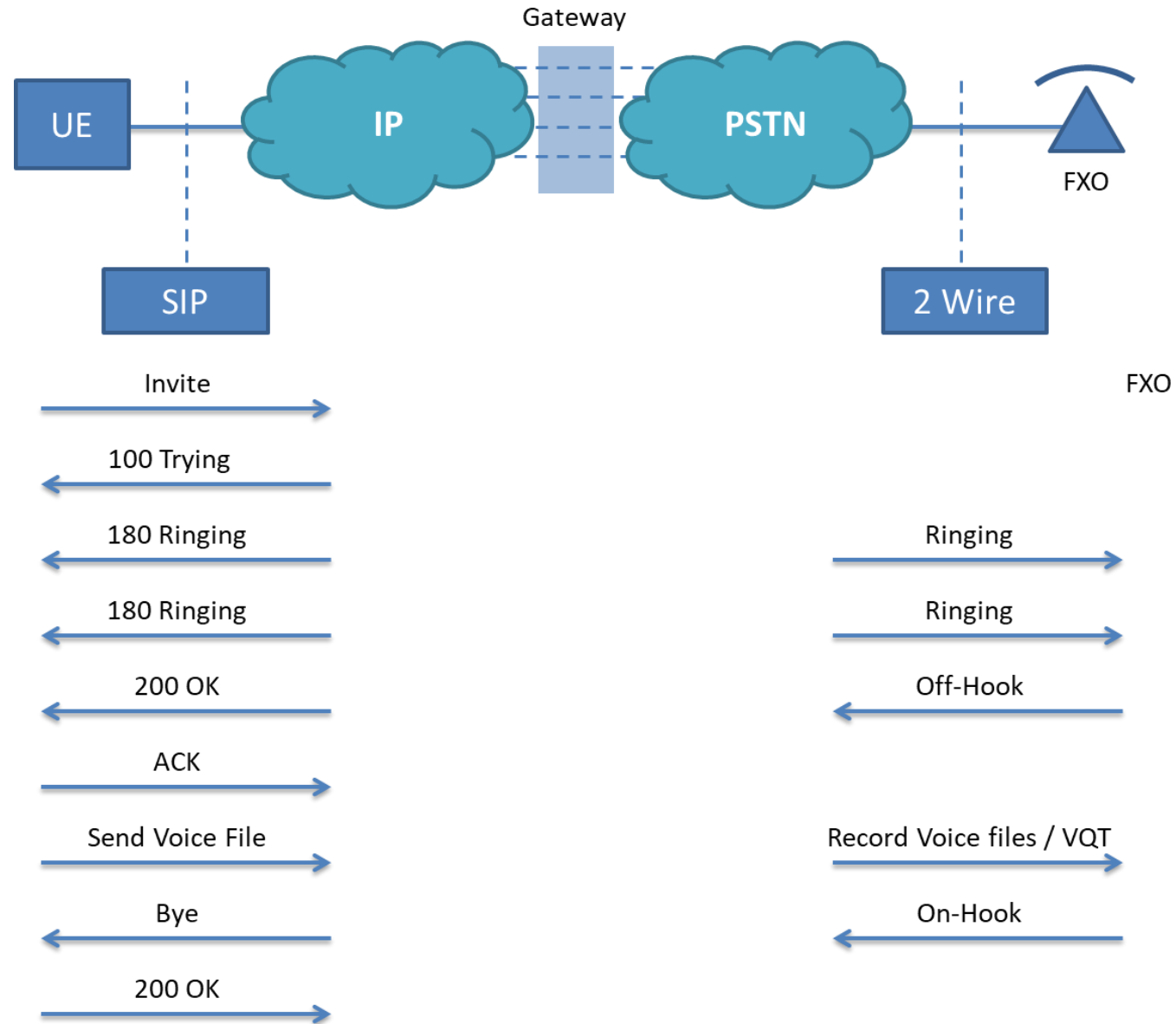
- Registrar – Accepts the registration from registrant
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error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



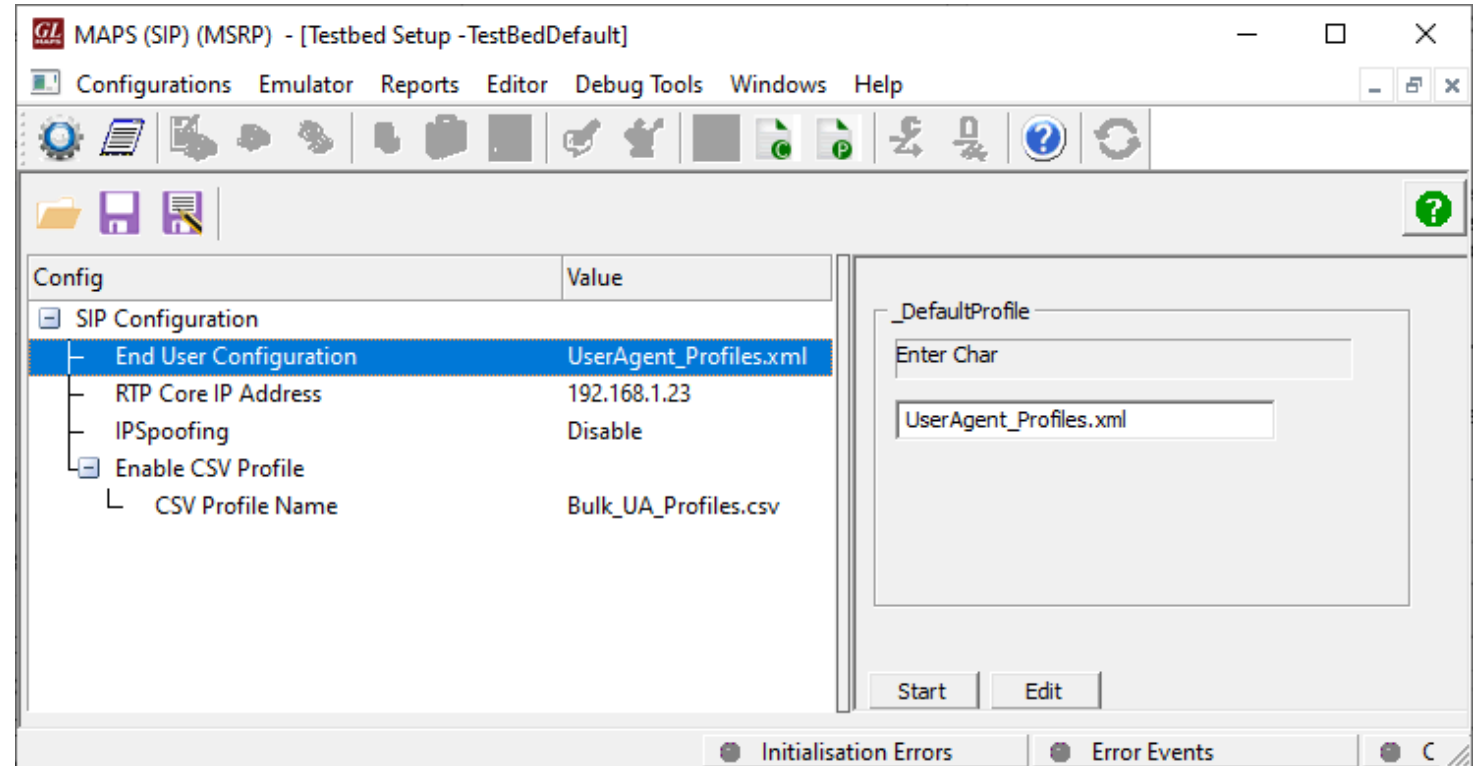
Test Bed Configuration

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

RTP Core IP Address:

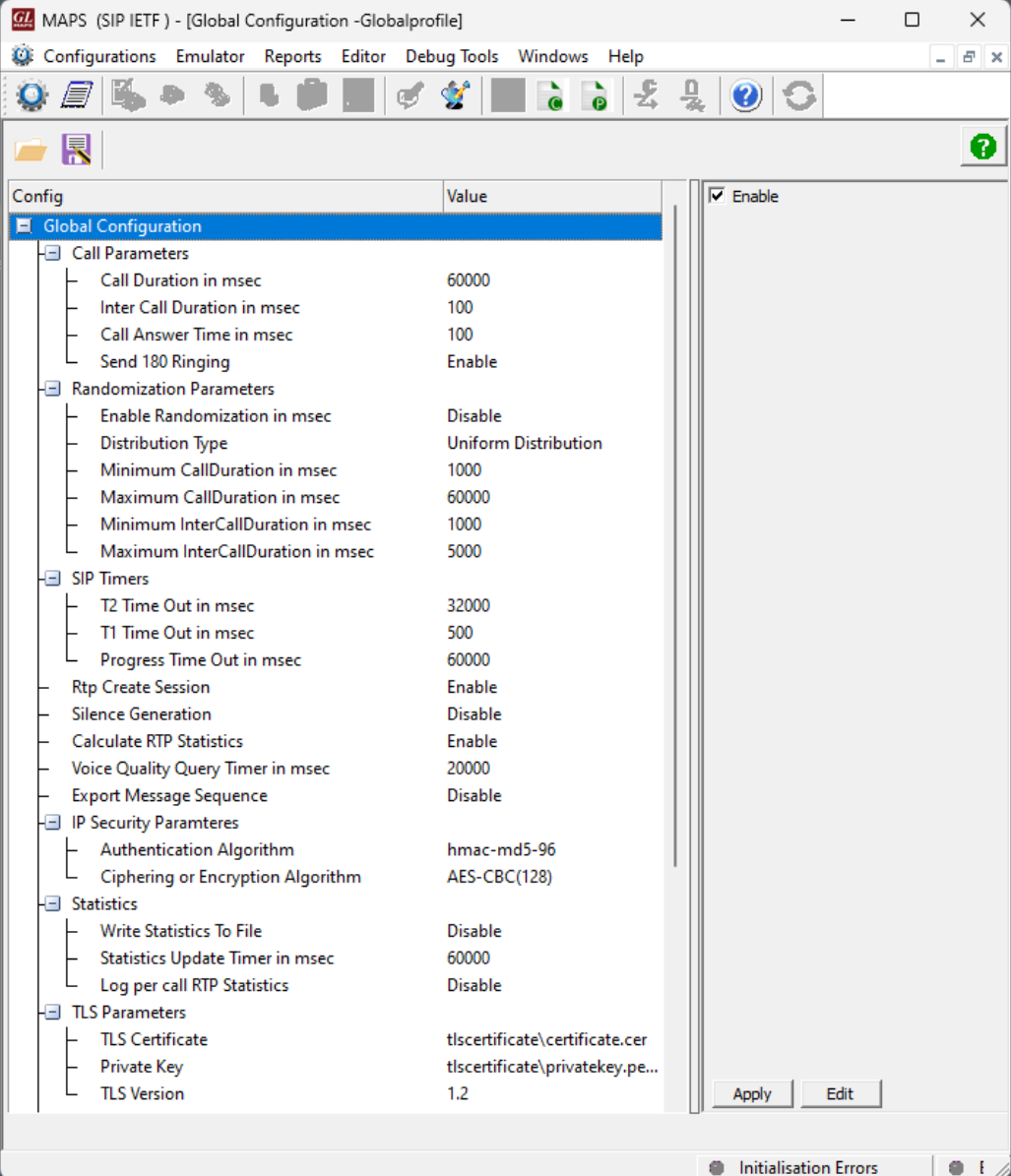
IP Address of the system on which the RTP Core should be invoked

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

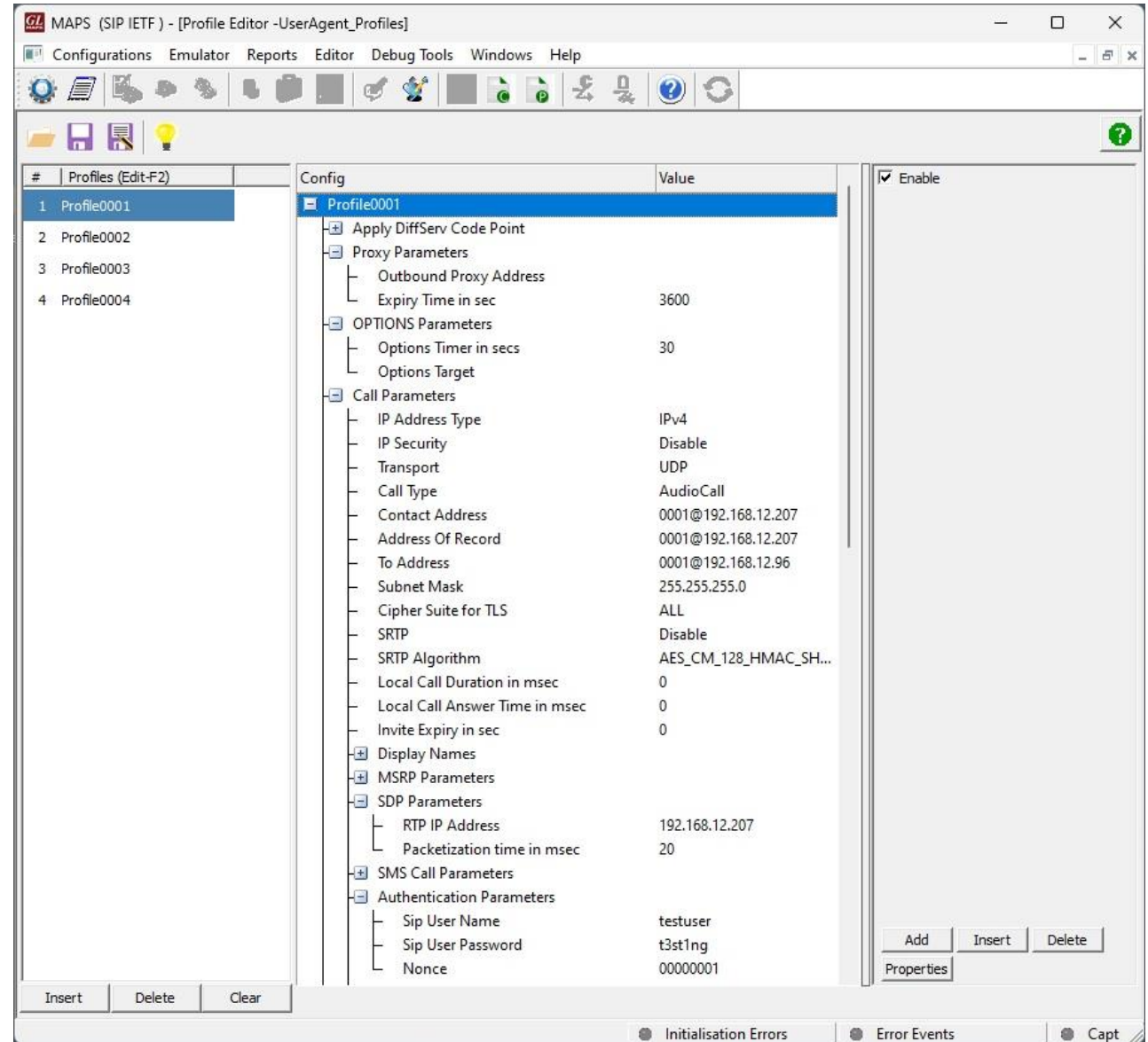


The screenshot shows the 'Global Configuration' window in the MAPS (SIP IETF) application. The window title is 'MAPS (SIP IETF) - [Global Configuration -Globalprofile]'. The interface includes a menu bar (Configurations, Emulator, Reports, Editor, Debug Tools, Windows, Help) and a toolbar with various icons. The main area is a tree view of configuration parameters, with 'Global Configuration' selected. The parameters are listed in a table with columns for 'Config' and 'Value'. A 'Global Configuration' summary box is visible on the right, showing 'Enable' checked. At the bottom right, there are 'Apply' and 'Edit' buttons. The status bar at the bottom indicates 'Initialisation Errors'.

Config	Value
Global Configuration	Enable
Call Parameters	
Call Duration in msec	60000
Inter Call Duration in msec	100
Call Answer Time in msec	100
Send 180 Ringing	Enable
Randomization Parameters	
Enable Randomization in msec	Disable
Distribution Type	Uniform Distribution
Minimum CallDuration in msec	1000
Maximum CallDuration in msec	60000
Minimum InterCallDuration in msec	1000
Maximum InterCallDuration in msec	5000
SIP Timers	
T2 Time Out in msec	32000
T1 Time Out in msec	500
Progress Time Out in msec	60000
Rtp Create Session	Enable
Silence Generation	Disable
Calculate RTP Statistics	Enable
Voice Quality Query Timer in msec	20000
Export Message Sequence	Disable
IP Security Paramteres	
Authentication Algorithm	hmac-md5-96
Ciphering or Encryption Algorithm	AES-CBC(128)
Statistics	
Write Statistics To File	Disable
Statistics Update Timer in msec	60000
Log per call RTP Statistics	Disable
TLS Parameters	
TLS Certificate	tlscertificate\certificate.cer
Private Key	tlscertificate\privatekey.pe...
TLS Version	1.2

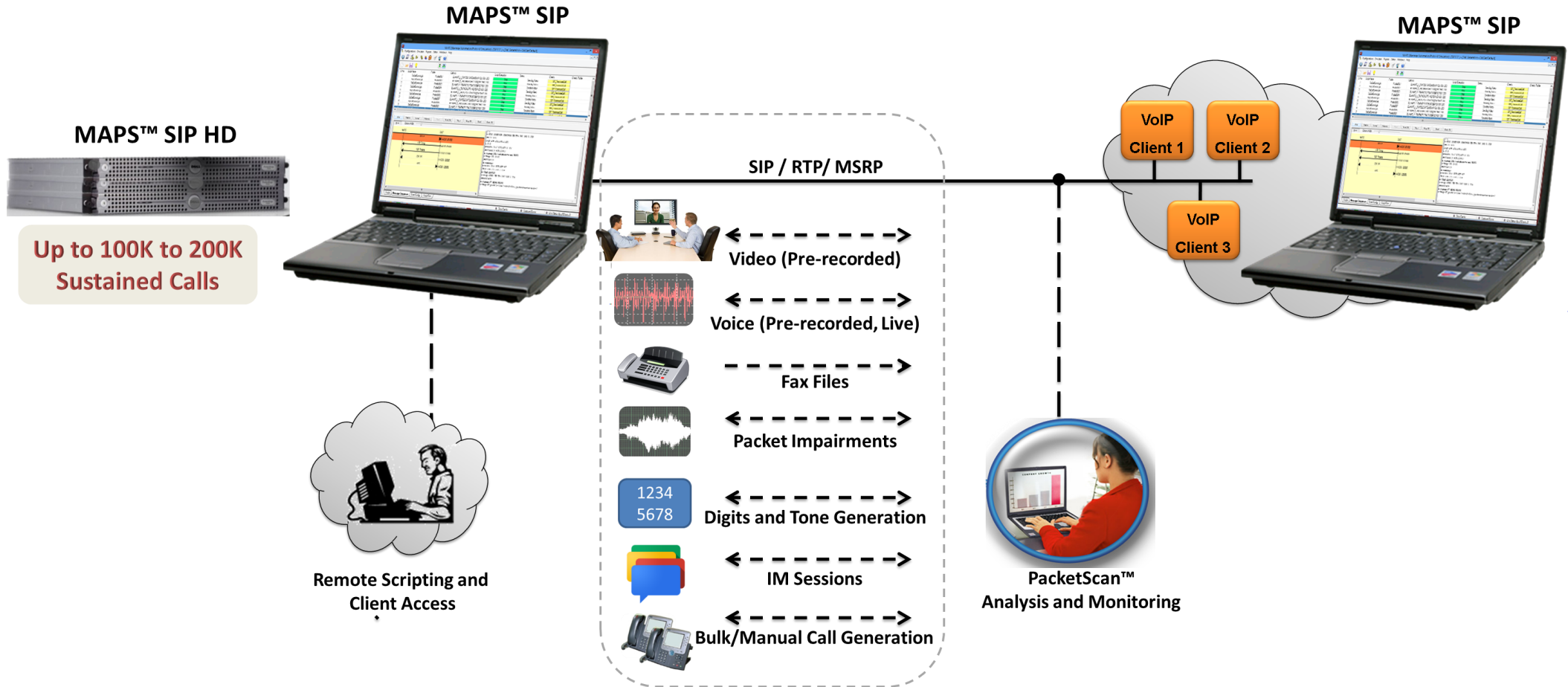
User Agents Configuration

- 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

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 Video Traffic	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 @350 CPS	20000 @ 350 CPS	-	-

Call Generation with Voice Traffic

GL MAPS (SIP) - [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
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown
2	SipCallControl.gls	Profile0001	GL-MAPS-3-104714086-10...	Start	PCMU Call Terminated	None		Pass

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

Save Column Width Show Latest

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 → 17:53:58.762000
← 200 OK 17:53:58.788000
            
```

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
c=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

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

Show Records
 Select Active Call
 Auto Trash

Column Width
 Show Latest

DUT MAPS

```

DUT-> INVITE 17:52:58.539000
MAPS-> 100 Trying 17:52:58.549000
MAPS-> 180 Ringing 17:52:58.560000
MAPS-> 200 OK 17:52:58.681000
DUT-> ACK 17:52:58.736000
DUT-> BYE 17:53:58.768000
MAPS-> 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
aptime:20
a=sendrecv
                    
```

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(<1%)	0 [0%]

Event Log

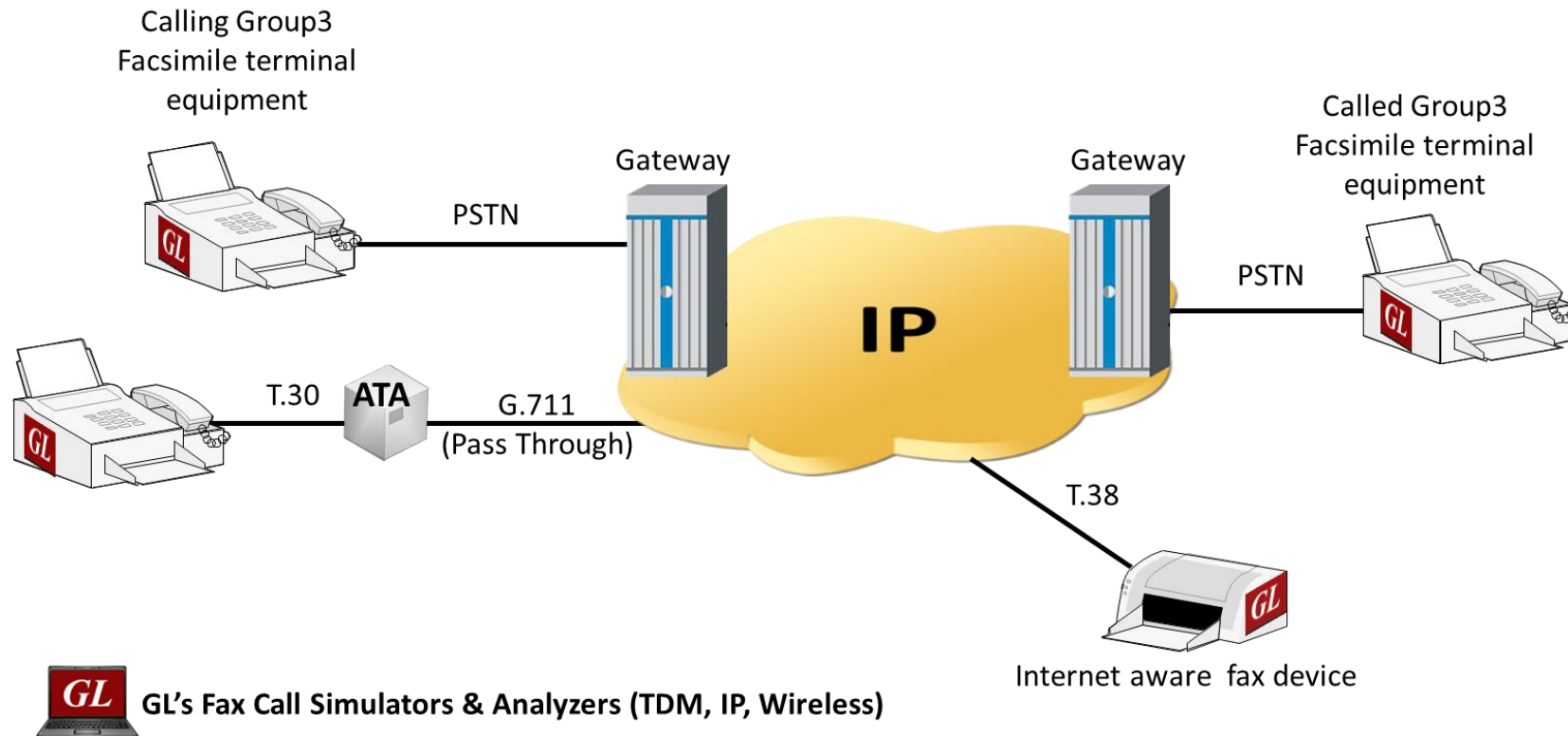
Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2015-1-15 15:11:57.064000	UDP Port = 5060		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.064000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.074000	PROGRESS Received	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.074000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.074000	PROGRESS Received	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.197000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.197000	Call Connected	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:11:57.203000	Sending RTP Digits	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:00.022000	RTP Digits Sent	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:01.963000	Detected Digits=1234567890ABCD	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:08.832000	BYE Sent		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:08.840000	200 Ok to BYE Received		SIP-Protocol.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:08.840000	Call Terminated	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432
2015-1-15 15:12:08.840000	Inter Call Duration = 1000	GL-MAPS_1_186785685-4496-8172@192.168.1.203	SipCallControl.gls	CGProtScriptId_10_186785684-4492-6432

Save Events

Capture Events to file

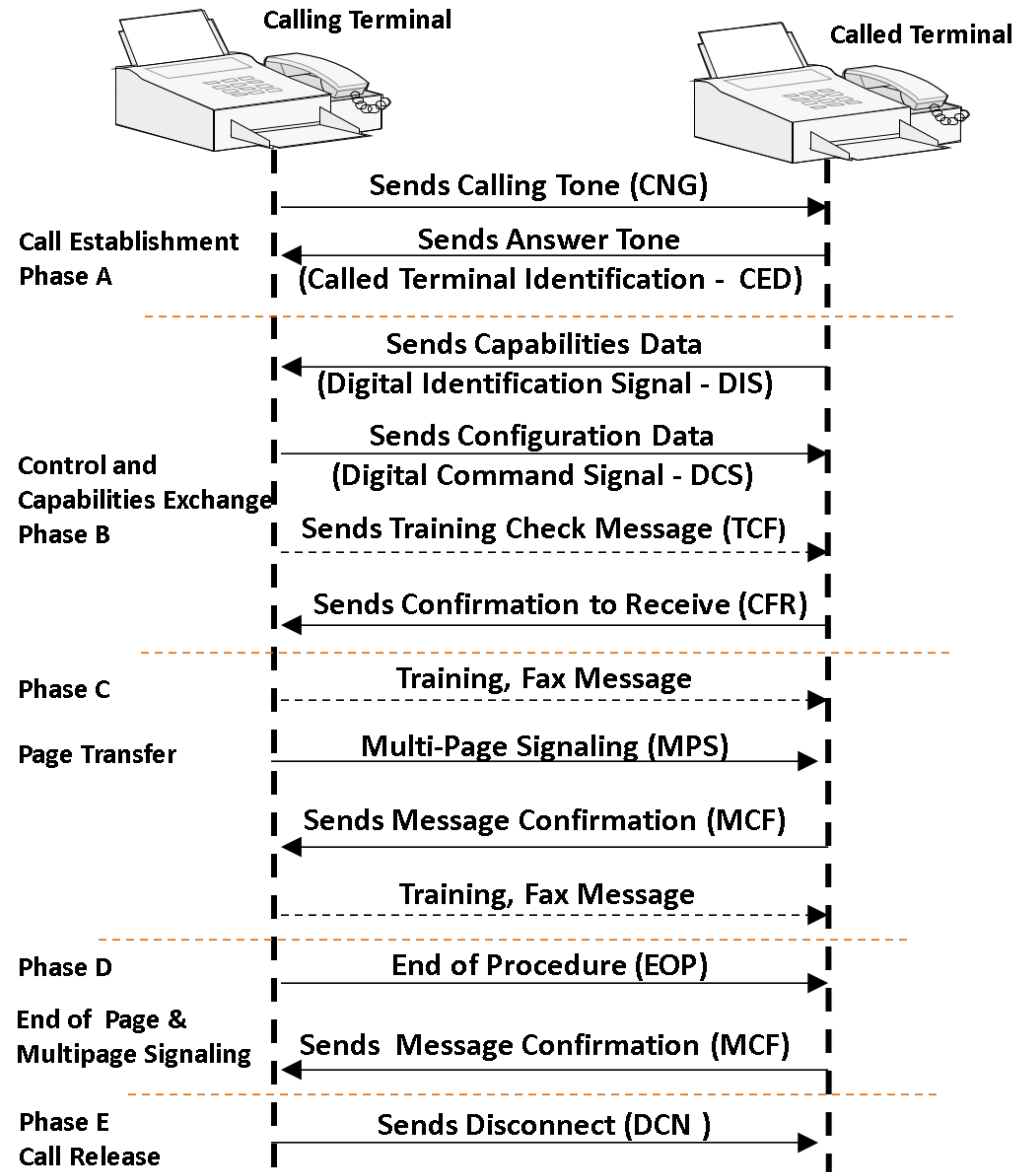
Clear

Fax Simulation over IP

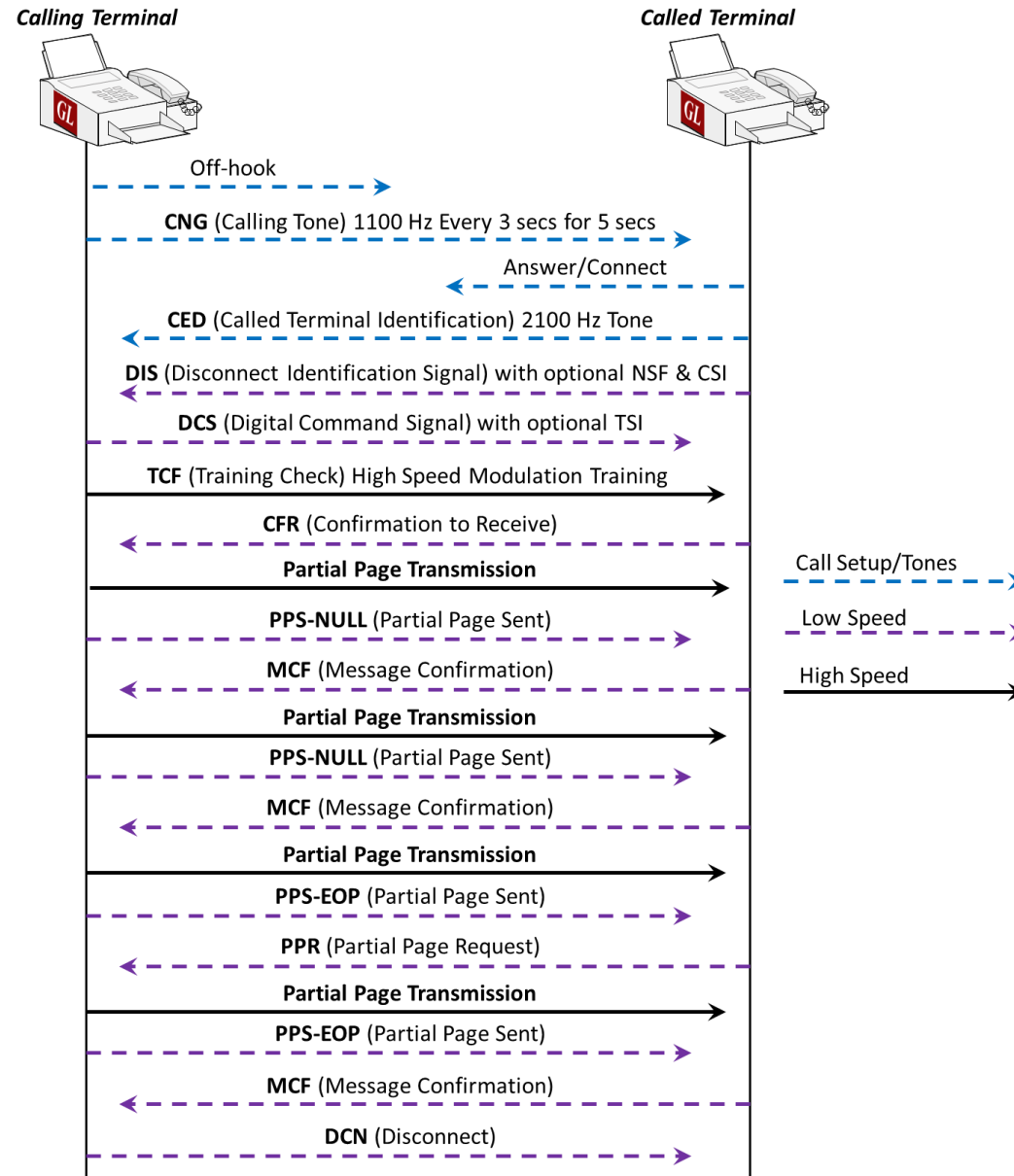


- RTP G.711 Pass Through Fax Simulation (PKS200)
- T.38 Fax Simulation over UDPTL (PKS211)

Call Scenarios - Fax T.30



T.38 Fax Emulation over IP using MAPS™



T.38 Fax Call in Progress and Related Events

The screenshot displays the GL MAPS (Message Automation Protocol Simulation) interface, showing a T.38 fax call in progress. The interface is divided into several sections:

- Call Log:** Shows a single call with the following details:

Sr 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	
- Message Sequence Diagram:** Shows the sequence of SIP messages between MAPS and DUT. Key events include:
 - 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) - Highlighted in orange
 - 200 OK (11:12:41.306000)
 - ACK (11:12:41.308000)
 - 33600 Rate of V34 selected after (11:13:15.904000) - Highlighted in blue
 - 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)
- Event Log:** Shows a detailed list of events captured during the call. Key events include:
 - PROGRESS Received (11:12:41.132000)
 - ACK Sent (11:12:41.260000)
 - Call Connected (11:12:41.261000)
 - ACK Sent (11:12:41.308000)
 - Fax - Status: 33600 Rate of V34 selected after MPH exchange (11:13:15.904000)
 - Fax - Status: CSI(Called Subscriber Identification) (11:13:15.905000)
 - Fax - Status: DIS(Digital Identification Signal) (11:13:15.906000)
 - Fax - Status: ECM mode Selected in DCS (11:13:15.906000)
 - Fax - Status: MMR Encoding selected in DCS (11:13:15.907000)
 - Fax - Status: 200x200 Resolution selected in the DCS (11:13:15.908000)
 - Fax - Status: A4 pagesize selected in the DCS (11:13:15.908000)
 - Fax - Status: TSII(Transmitting Subscriber Identification) (11:13:15.909000)
 - Fax - Status: DCS(Digital Command Signal) (11:13:15.909000)
 - Fax - Status: V21 Signal Done (11:13:15.910000)
 - Fax - Status: Transmitter Started To Train (11:13:15.911000)
 - Fax - Status: Transmitter Train Successful (11:13:15.911000)
 - Fax - Status: CFRJ(Confirmation To Receive) (11:13:15.912000)
 - Fax - Status: Image Transmit Start (11:13:15.912000)
 - Fax - Status: Image Transmit End (11:13:15.913000)
 - Fax - Status: PPS NULL(Current Partial Page Block Transmission Complete) (11:13:15.913000)
 - Fax - Status: V21 Signal Done (11:13:15.914000)
 - Fax - Status: MCF(Message Confirmation) (11:13:15.914000)
 - Fax - Status: Image Transmit Start (11:13:15.915000)
 - Fax - Status: Image Transmit End (11:13:15.915000)
 - Fax - Status: PPS EOP(All Pages Transmitted) (11:13:15.916000)
 - Fax - Status: V21 Signal Done (11:13:15.916000)
 - Fax - Status: MCF(Message Confirmation) (11:13:15.917000)
 - Fax - Status: DCN(Disconnect) (11:13:15.917000)
 - Fax - Status: V21 Signal Done (11:13:15.918000)
 - Fax Session Successful (11:13:15.919000)
 - Fax - Status: FaxSessionDuration = 2898 msecInitialModem = "V34" InitialRate = "33600" FinalMod. (11:13:15.919000)
 - BYE Sent (11:14:41.280000)
 - 200 Ok to BYE Received (11:14:41.290000)
 - Call Terminated (11:14:41.290000)
 - Inter Call Duration = 100 (11:14:41.290000)

Call Generation with FAX Traffic

The screenshot displays the GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation] interface. It consists of two main windows showing call logs and message sequences.

Top Window: Call Log

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	

Bottom Window: Call Log and Message Sequence

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

Message Sequence (Left Panel):

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

Message Sequence (Right Panel):

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

SIP Message Body (Right Panel):

```

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
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=T38FaxUpdEC:t38UDPRedundancy
    
```

FAX Traffic Events

Events				
Event Log				
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

Capture Events to file ...

File Traffic Events

Date/Time	Captured Events	Call Trace Id	Script Name
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 Capture Events to file

Clear

Video Call Generation

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface. The main window shows a table of call generation results with columns for Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, Events Profile, Result, and Total Iter. Below the table, there are control buttons for Add, Delete, Insert, Refresh, Start, Start All, Stop, Stop All, Abort, and Abort All. A detailed view of a SIP message sequence is shown, including a sequence diagram and a text-based representation of the message content.

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

Sequence Diagram:

```
sequenceDiagram
    participant MAPS
    participant DUT
    MAPS->>DUT: INVITE 10:55:08.130000
    DUT-->>MAPS: 100 Trying 10:55:08.147000
    DUT-->>MAPS: 180 Ringing 10:55:08.149000
    DUT-->>MAPS: 200 OK 10:55:08.280000
    MAPS->>DUT: ACK 10:55:08.286000
```

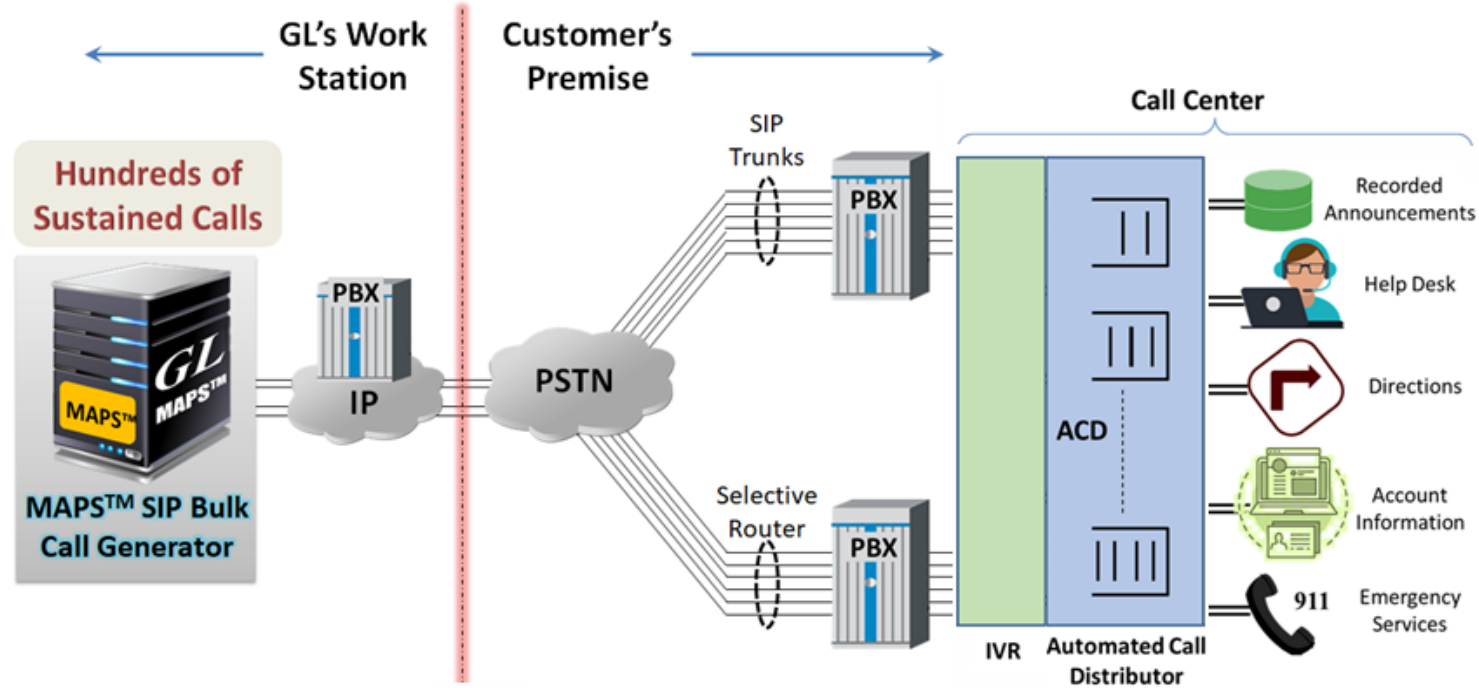
Message Content:

```
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
```

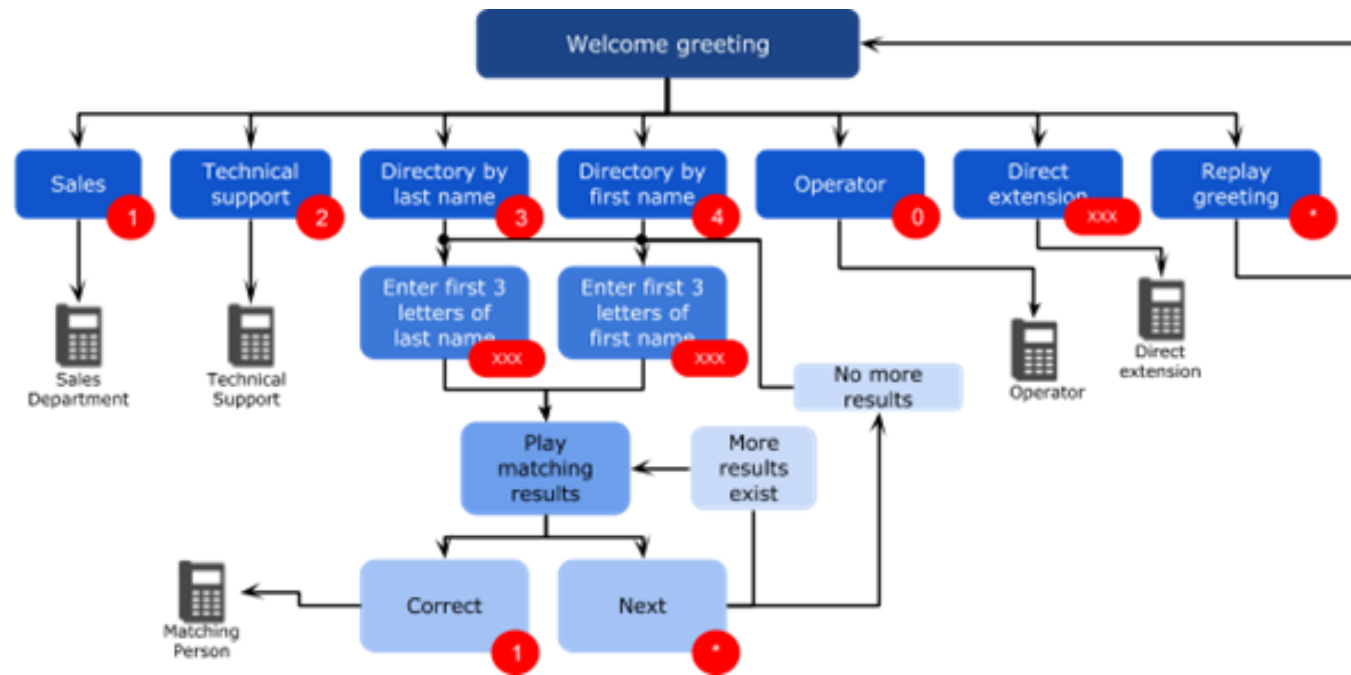
Transmit pre-recorded video traces with video codecs like H.264, and H.263

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

The screenshot displays the MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) software interface. The window title is "MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - CallGenDefault]". The interface includes a menu bar with "Configurations", "Emulator", "Reports", "Editor", "Debug Tools", "Windows", and "Help". Below the menu bar is a toolbar with various icons. The main area is divided into several sections:

- Table:** A table with columns: "Sr No", "Script Name", "Profile", "Call Info", "Script Execution", "Status", "Events", "Events Profile", and "Result". It contains two rows of data.
- Buttons:** A row of buttons including "Add", "Delete", "Insert", "Refresh", "Start", "Start All", "Stop", "Stop All", "Abort", and "Abort All".
- Message Sequence:** A section with a "Save" button and "Column Width" and "Show Latest" options. It displays a sequence of messages in a timeline format, including "ACK", "Stage 1: Welcome to GL communications", "Stage 1: If you know your parties extension you can download at anytime", "Stage 1: For sales press 1", "Stage 1: For Technical Support Press 2", "Stage 1: Or directory by last name press 3", "Digits Transmitted : 3", "Stage 2: Welcome to the directory Please enter the first 3 letters of your party's last name", "Stage 2: Using your touch tone keypad use the Seven key for Q and the nine key for Z I'm sorry", "Digits Transmitted : 925", "BYE", and "200 OK".
- SIP Message Details:** A section with a "Find" button and a scrollable area containing SIP message details, including headers like "INVITE sip:13016704784@104.219.163.78 SIP/2.0" and "Via: SIP/2.0/UDP 80.76.14.188:8060;branch=z9hG4kK-21-59922946-8346-4708".
- Bottom Bar:** A bar with tabs for "Scripts", "Message Sequence", "Event Config", and "Script Flow". It also includes status indicators for "Initialisation Errors", "Error Events", "Captured Errors", and "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_Detailed... x

59.1%

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 parties extension you can download at anytime	0.9424	1			
2020-05-05 08:25:42.775000	Analysis			1			
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_Res... x

59.1%

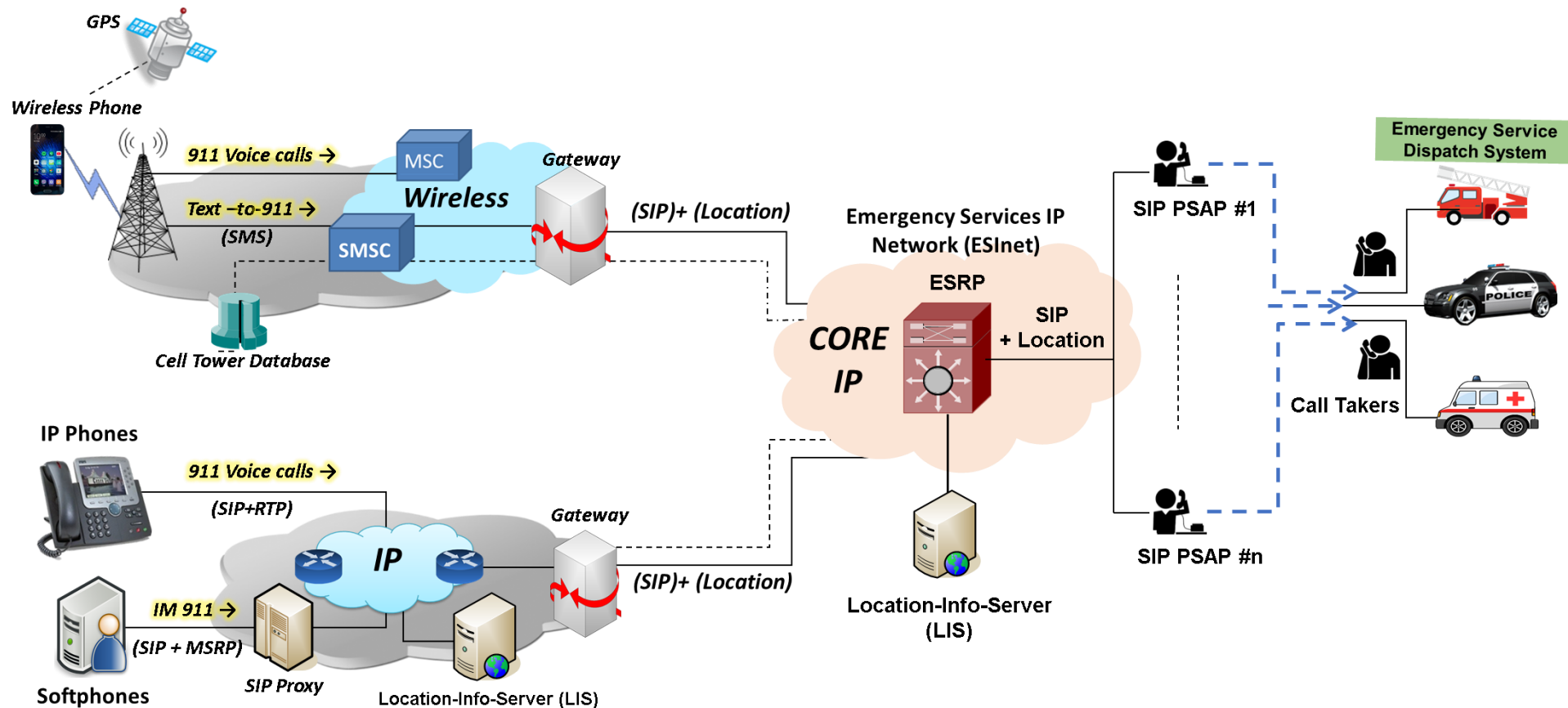
GL Communications Inc Date: 05/05/2020

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

SI.No	Time	Profile	Destination TN	IVR File	SIP Result	IVR Result	Detailed Report
1	2020-05-05 08:26:22.979000	Profile0001	13016704784	maps/sip/iet/ivr/prompt_gl.csv	Pass	Pass	MAPS\SIP\IET\IVR\Log\DetailedLog\Maps_IVR_DetailedLog_2020-05-05_08-26-32_Profile0001.pdf

MSRP

- 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

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface, specifically the Profile Editor for TrafficProfile. The main window is titled "GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Profile Editor - TrafficProfile]". The interface includes a menu bar (Configurations, Emulator, Reports, Editor, Debug Tools, Windows, Help) and a toolbar with various icons. On the left, a list of profiles is shown, with "Profile0001" selected. The central pane displays the configuration tree for "Send IM", which is expanded to show the following settings:

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

The "IM File Name" field is highlighted with a blue box. In the foreground, a Notepad window titled "MsrpInputMessage.txt - Notepad" is open, displaying the following text:

```
File Edit Format View Help
Hi, Welcome
This is MAPS SIP MSRP Simulator.
Test Message 1.
Test Message 2.
Test Message 3.
```

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 Find

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

```

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

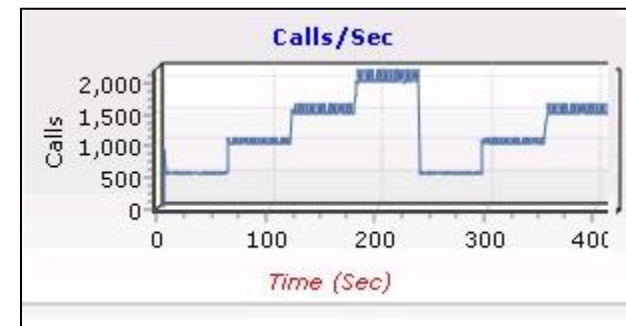
Distributions	Description	
Uniform	MinCR=40 , MaxCR=80 , Duration=10	Add
Fixed	Call Rate=250 , Duration=10	Remove
Normal	MinCR=40 , MaxCR=80 , Duration=10	Remove All
		Edit

Scripts Exclusive Profiles

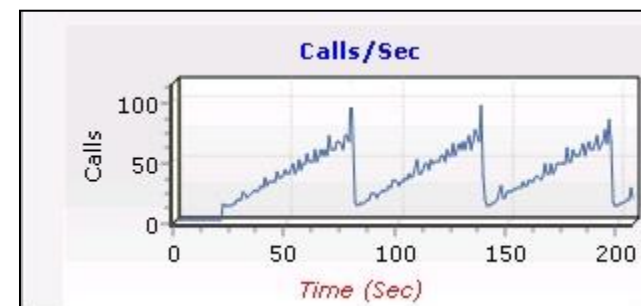
Scripts	Profile
SipCallControl	Profile0003
Registration	Profile0005

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

Step Statistical Distribution



Ramp Statistical Distribution

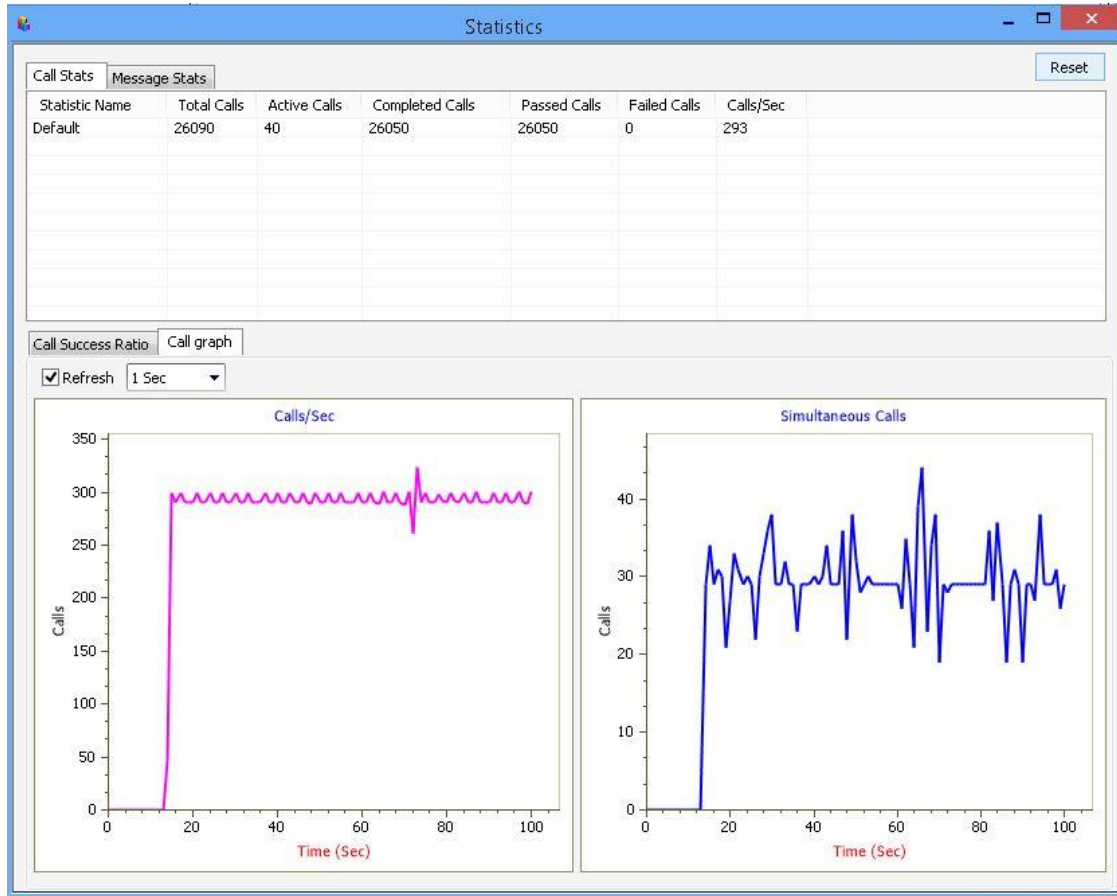


Saw-tooth Statistical Distribution

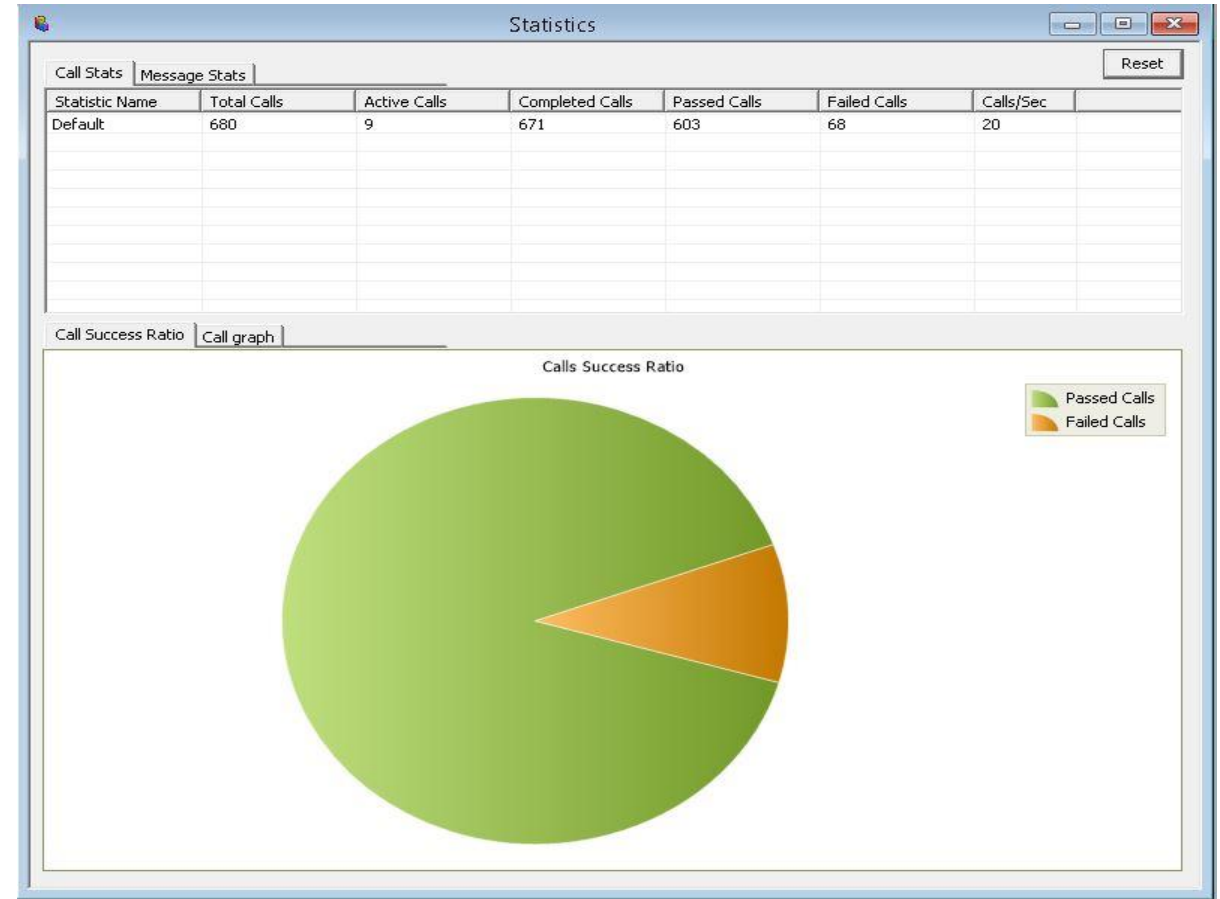


Success Call Ratio Statistics

Call Graph



Call Stats



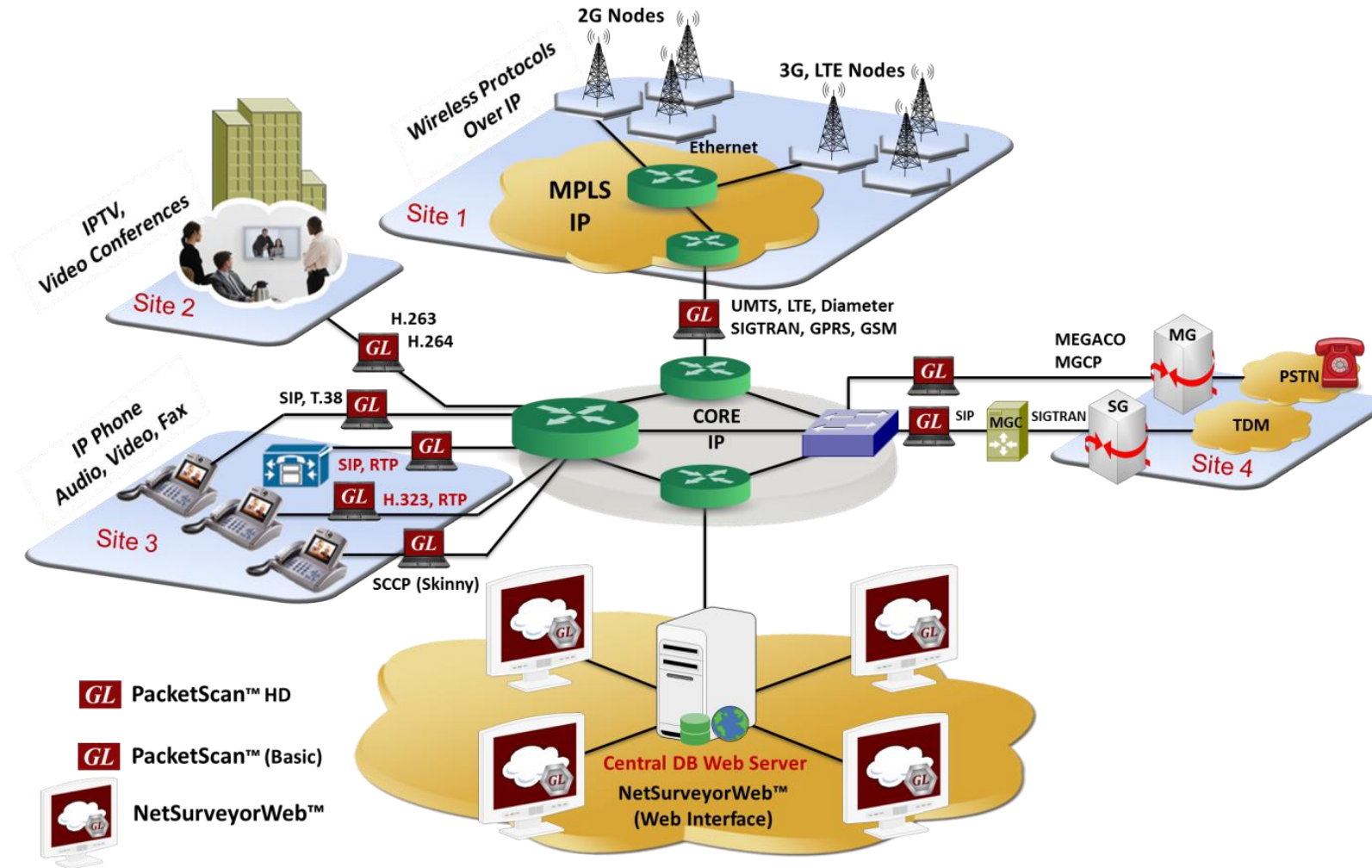
Message Statistics

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



What the software does?

- Captures, segregates, and monitors packets; perform voice quality testing in real-time over VoIP network
- Unlimited traffic and signaling capturing capability; captured VoIP calls with video can be played back using 3rd party applications
- Can be deployed as a Probe for a centralized monitoring system with Oracle database

For complete details, please visit <http://www.gl.com/packetscan-all-ip-packet-analyzer.html>

PacketScan™ Analyzer with SIP CDR

PacketScan (All-in-One)

File View Capture Statistics Database Call Detail Records Configure Help

0 GoTo

Type	Packet Type MAC	Source IP Address IP	Destination IP Address IP	Source Port UDP	Destination Port UDP	SIP Method Sip3261	SIP From Sip3261	SIP To Sip3261	SIP Call ID Sip3261	SIP CSeq Sip3261
SIP		192.168.1.203	192.168.1.213	5060	5060	INVITE	0001@192.168.1.203	0001@192.168.1.213	GL-MAPS_1_185372727-4480-8320@192.168.1.203	1 INVITE
SIP		192.168.1.213	192.168.1.203	5060	5060	SIP/2.0 100 Trying	0001@192.168.1.203	0001@192.168.1.213	GL-MAPS_1_185372727-4480-8320@192.168.1.203	1 INVITE
SIP		192.168.1.213	192.168.1.203	5060	5060	SIP/2.0 180 Ringing	0001@192.168.1.203	0001@192.168.1.213	GL-MAPS_1_185372727-4480-8320@192.168.1.203	1 INVITE
SIP		192.168.1.213	192.168.1.203	5060	5060	SIP/2.0 200 OK	0001@192.168.1.203	0001@192.168.1.213	GL-MAPS_1_185372727-4480-8320@192.168.1.203	1 INVITE
SIP		192.168.1.203	192.168.1.213	5060	5060	ACK	0001@192.168.1.203	0001@192.168.1.213	GL-MAPS_1_185372727-4480-8320@192.168.1.203	1 ACK

```

===== Sip3261 Layer =====
HDR      = INVITE sip:0001@192.168.1.213 SIP/2.0
HDR      = Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_185372727-4481-8320
HDR      = Max-Forwards: 70
HDR      = Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UPDATE
HDR      = From: "MapsSip" <sip:0001@192.168.1.203>;tag=FromTag_1_185372727-4478-8320
HDR      = To: 0001 <sip:0001@192.168.1.213>
HDR      = Call-ID: GL-MAPS_1_185372727-4480-8320@192.168.1.203
HDR      = CSeq: 1 INVITE
HDR      = Contact: 0010 <sip:0001@192.168.1.203>
HDR      = Content-Type: application/sdp
HDR      = Content-Length: 317
HDR      =
BODY     = v=0
BODY     = o=0001 33852938 33852938 IN IP4 192.168.1.203
BODY     = s=-SIP Call
BODY     = c=IN IP4 192.168.1.203
BODY     = t=0 0
BODY     = m=audio 1036 RTP/AVP 0 8 18 3 101
BODY     = a=rtpmap:0 PCMU/8000
BODY     = a=rtpmap:8 PCMA/8000
BODY     = a=rtpmap:18 G729/8000
BODY     = a=fmtp:18 annexb=no
BODY     = a=rtpmap:3 GSM/8000
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	Call Details
0	Terminated	SIP	0001@192.168.1.203	0001@192.168.1.213	2015-01-15 14:48:24.106754	00:01:00.160991	<SIPCallID> GL-MAPS_1_185372727-4480-8

Capture Rate : 0.05 Mbps C:\Temp.hdl Captured 1158 frames Missed Frames : 0

SIP Decode in PacketScan™

The screenshot displays the PacketScan (All-in-One) interface. The main window shows a list of captured packets. The selected packet (Frame 0) is a SIP INVITE from 192.168.1.200 to 192.168.1.103. The detailed view below shows the SIP message structure:

```
----- Sip3261 Layer -----
HDR = INVITE sip:0001@192.168.1.103 SIP/2.0
HDR = Via: SIP/2.0/UDP 192.168.1.200:5060;branch=z9hG4bK3811333536-3
HDR = Max-Forwards: 70
HDR = Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, COMET, OPTIONS, SUBSCRIB
HDR = From: 0001 <sip:0001@192.168.1.200>;tag=GLPG_3811333536-333
HDR = To: 0001 <sip:0001@192.168.1.103>
HDR = Call-ID: GLPG-483633760331
HDR = CSeq: 1 INVITE
HDR = Contact: 0001 <sip:0001@192.168.1.103>
HDR = Content-Type: application/sdp
HDR = Content-Length: 349
HDR =
BODY = v=0
BODY = o=0001 47706128 47706129 IN IP4 192.168.1.200
BODY = s=-
BODY = c=IN IP4 192.168.1.200
BODY = t=0 0
BODY = m=audio 1024 RTP/AVP 0 8 18 104 3 101
BODY = a=rtpmap:0 PCMU/8000/1
BODY = a=rtpmap:8 PCMA/8000/1
BODY = a=rtpmap:18 G729/8000/1
BODY = a=fmtp:18 annexb=no
BODY = a=rtpmap:104 G726-32/8000/1
BODY = a=rtpmap:3 GSM/8000/1
BODY = a=rtpmap:101 telephone-event/8000
BODY = a=fmtp:101 0-15
BODY = a=ptime:20
BODY = a=sendrecv
```

Off-line Viewing C:\Program Files\GL Communications Inc\P 2 550 Frames

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

```

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
a=ptime: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 | Registrator 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

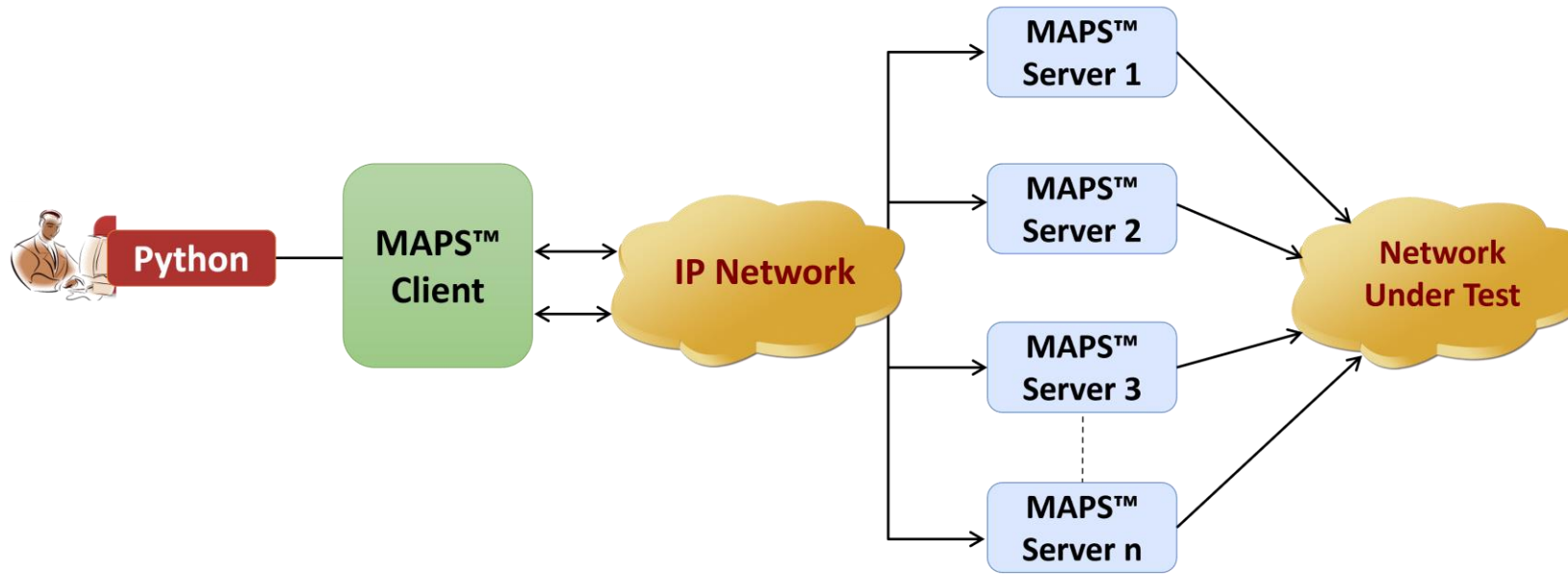
00.30.538	5004	←	v21-preamble	5004	↑
00.31.580	5004	←	NSF	5004	↑
00.31.955	5004	←	CSI NUM:918040488401	5004	↑
00.32.648	5004	←	DIS:DSR:ITU-T V.27 ter and V.29	5004	↑
00.33.110	5004	←	no-signal	5004	↑
00.34.559	5004	→	v21-preamble	5004	↓
00.35.657	5004	→	TSI NUM:40488401	5004	↓
00.36.402	5004	→	DCS:DSR:9600bps, 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)
  
```

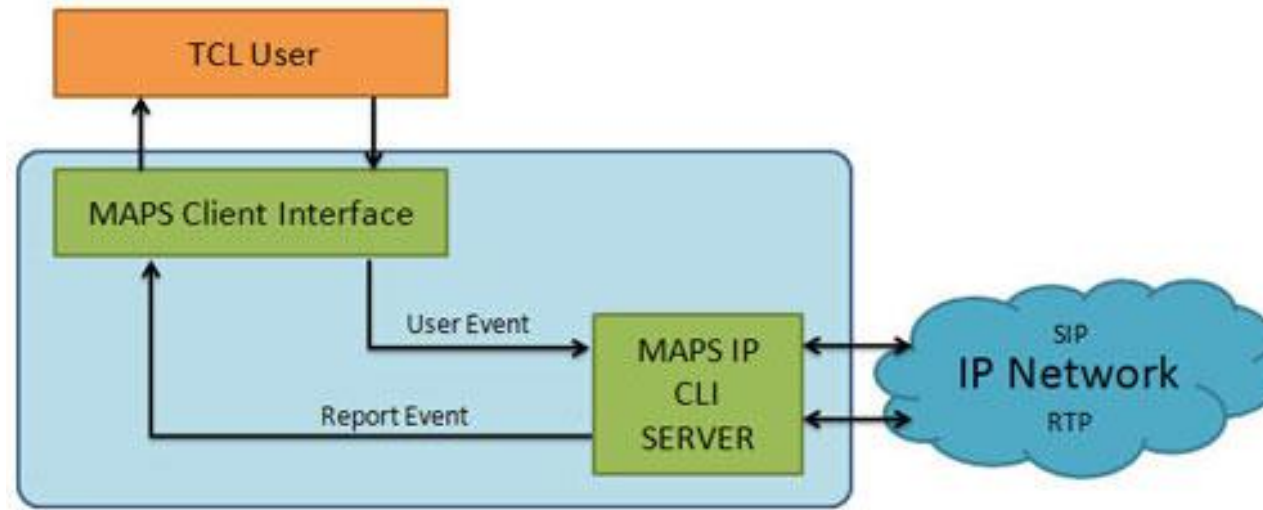
Active Calls Graph | Average Jitter Distribution | E-Model | RTP Packets Graph | **T.38 Analysis** | Call Graph | Call Summary

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 Java, VBScripts, TCL, Python and others
- The MAPS™ APIs allows for programmatic and automated control over all MAPS™ platforms. Each MAPS™ server can receive multiple client connections and offer independent execution to each client
- Likewise, a single client can connect to multiple MAPS™ servers, including servers running different protocols, permitting complex cross-protocol test cases

MAPS™ SIP CLI Test System



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

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=rtpmap:0 PCMU/8000
```

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

CDR Data 120 Secs

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

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

Quick Search: Trafficid [GO] View Records Per Page: 20 Sort Expression: STARTTIME DE

Trafficid	Probename	Calling Number	Called Number	Starttime	Duration	Payload1	Payload2	Cor
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
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
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
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
Summary								
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	
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
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
2062474	PacketProbe0	114@192.168.20.45	117@192.168.20.126	2013-02-12 15:28:12.000000	00:00:00.000000			
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			
2062476	PacketScan	114@192.168.20.45	117@192.168.20.126	2013-02-12 15:28:06.515	00:00:00.000000			
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			
2062468	PacketProbe0	19134163019@192.168.20.45	102@192.168.20.129	2013-02-12 15:23:09.000000	00:00:00.000000			
2062469	PacketScan	19134163019@192.168.20.45	102@192.168.20.129	2013-02-12 15:23:03.161	00:00:00.000000			
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
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
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

GL Communications Inc.
 Telecommunication Products and Consulting
 NetSurveyorWeb 3.7.10 - Real Time Monitoring System

Protocol Type: VOIP

Data | Filters / Views | Reports | Alarms | Admin

System Status as of 2014-05-19 08:05:32

Date Range: 2014-05-19 To 2014-05-19
 Hour Range: 00:00:00 To 23:59:59
 Enable Alarms:
 Query Execution Time: 0.14040 Seconds

Listening_MOS

MOS Category	Count	Percentage
Desirable LMOs	15	41.67 %
Reach Connection LMOs	12	33.33 %
Acceptable LMOs	9	25 %
Not Recommended LMOs	0	0 %

Conversational_MOS

Time (HH)	Desirable CMos	Reach Connection CMos	Acceptable CMos	Not Recommended CMos
0-1	0	3	1	0
5	4	0	5	0
8	11	0	0	0

Time (HH)

Thank you