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# MAPS™ SIP-I Emulator

SIP-ISUP Emulation over IP

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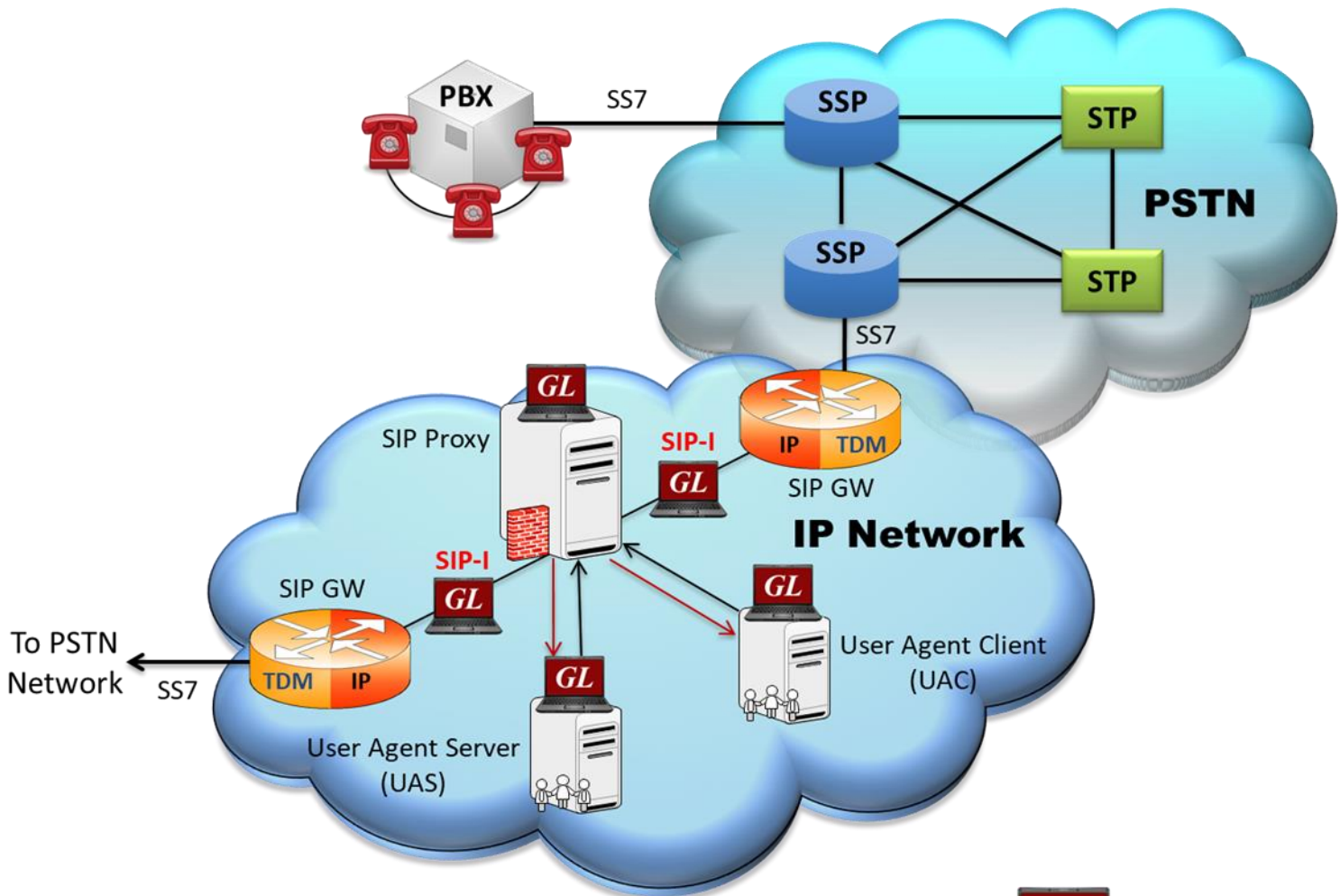
 ***GL Communications Inc.***

818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878

Phone: (301) 670-4784 Fax: (301) 670-9187 Email: [info@gl.com](mailto:info@gl.com)

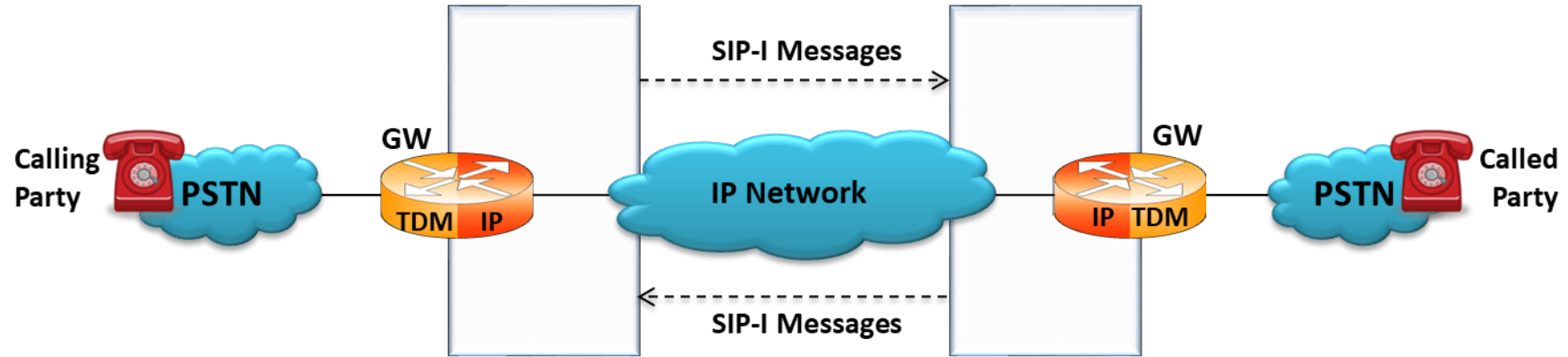
Website: <http://www.gl.com>

# Overview



 **MAPS™ SIP-I Emulator**  
**(SIP-I and SIP-T Protocol Emulation)**

# MAPS™ SIP-I



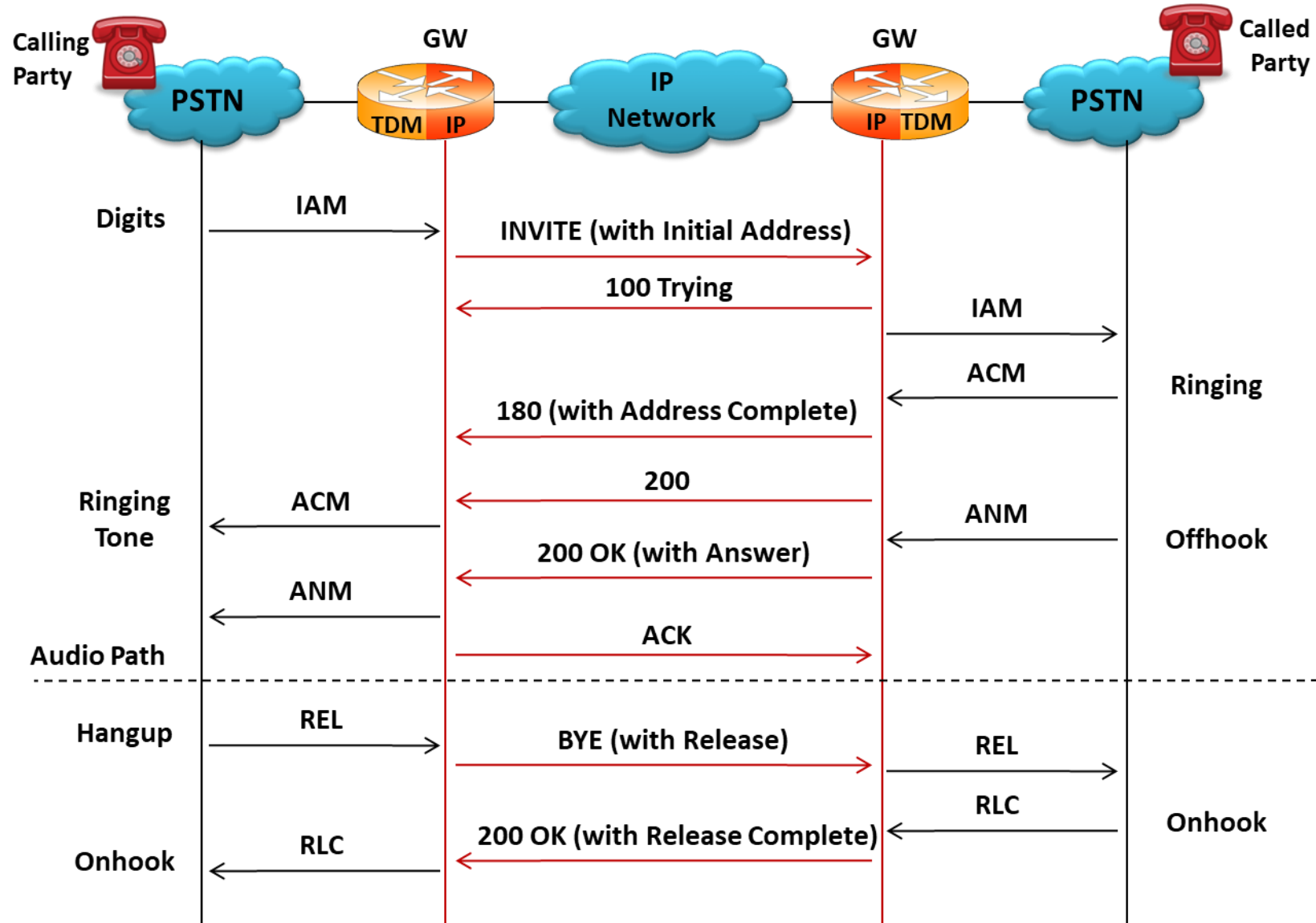
# Protocol Specific Features

<b>Signaling</b>	<ul style="list-style-type: none"><li>• Supports both UDP and TCP (Ipv4 and Ipv6)</li><li>• Generates and processes SIP-I valid and invalid messages</li><li>• Each SIP-I message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts</li><li>• Handles Retransmissions and Remote Retransmissions</li><li>• Supports Call hold, Call redirect, Bind Call Transfer, auto call rejection, and silence packets generation</li></ul>
<b>Traffic</b>	<ul style="list-style-type: none"><li>• Supports transmission and detection of various RTP traffic such as, digits, voice file, Fax, IVR, single tone, and dual tones in IP networks</li><li>• Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL)</li><li>• Supports both RTP G.711 Pass Through Fax Simulation and T.38 Fax Simulation over UDPTL</li><li>• RTP Voice Traffic Generation used in conjunction with SIP Signaling</li><li>• User-defined global statistics for received RTP audio quality metrics are updated periodically on run time</li><li>• Bulk Video Call Generation supported with H.264 and H.263 video codecs</li><li>• Supports almost all industry standard codec types</li></ul>
<b>Applications</b>	<ul style="list-style-type: none"><li>• Simulates Signaling Gateway, Softswitch with SIP-I (Profile C) support to test interworking of PSTN services over IP networks</li><li>• Fully integrated, complete test environment for SIP-I or SIP-T</li><li>• Inter-operability testing of networks</li></ul>

# Protocol Specific Standards

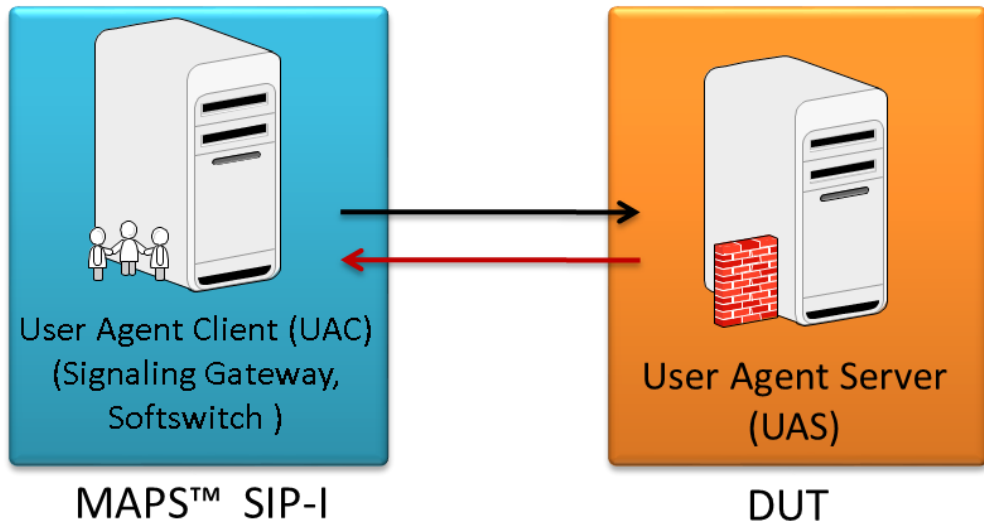
Available Standards	Standard / Specification Used
SIP-I (Profile-C)	ITU Q.1912.5 - Interworking between Session Initiation Protocol (SIP) and ISDN User Part ND1007:2001/07, PNO-ISC/SPEC/007- Interworking between Session Initiation Protocol (SIP) and UK ISDN User Part (UK ISUP)
SIP-T	IETF RFC 3372

# Typical SIP-I Call Procedure

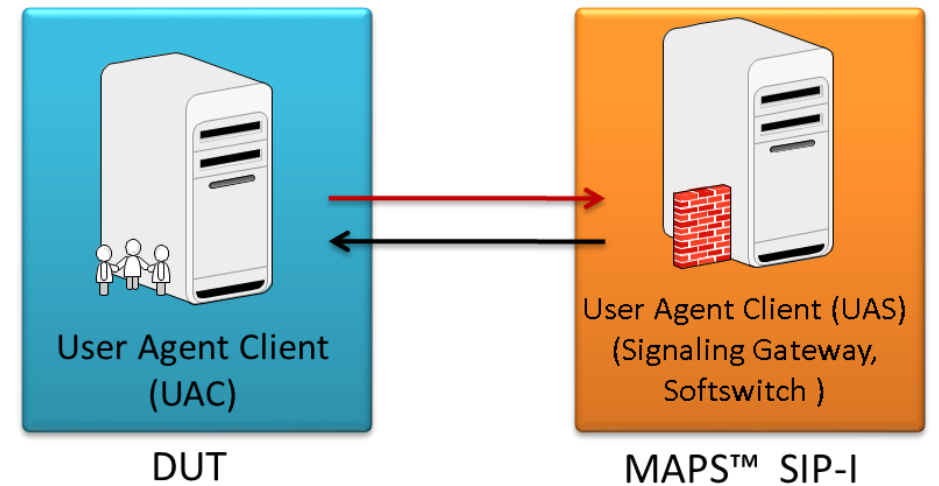


# Test Scenarios

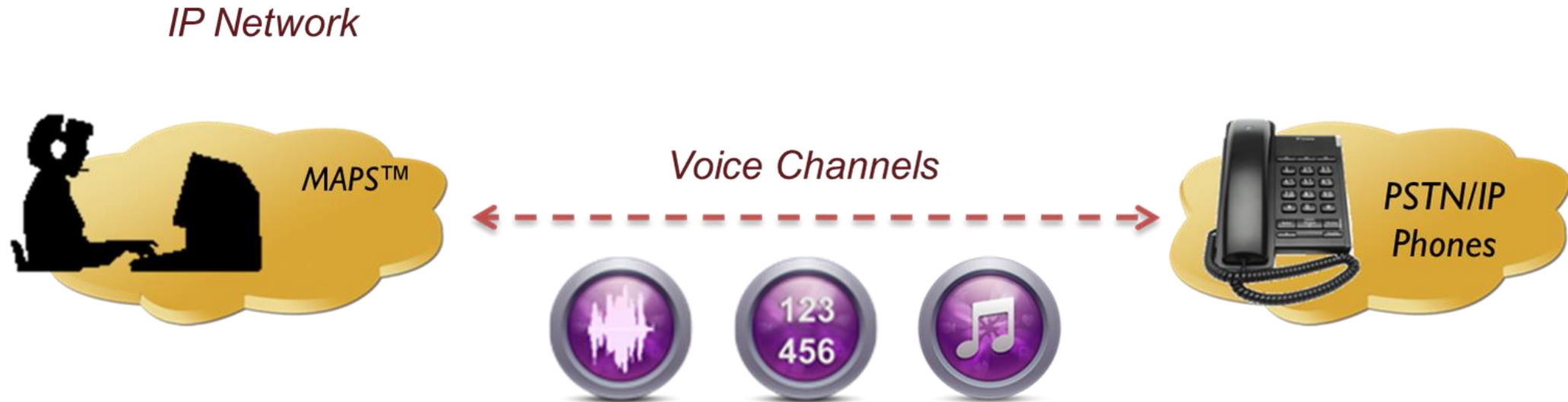
**MAPS™ SIP-I simulating UAC and testing UAS (DUT)**



**MAPS™ SIP-I simulating UAS and testing UAC (DUT)**



# RTP Traffic Simulation



## *Tx*

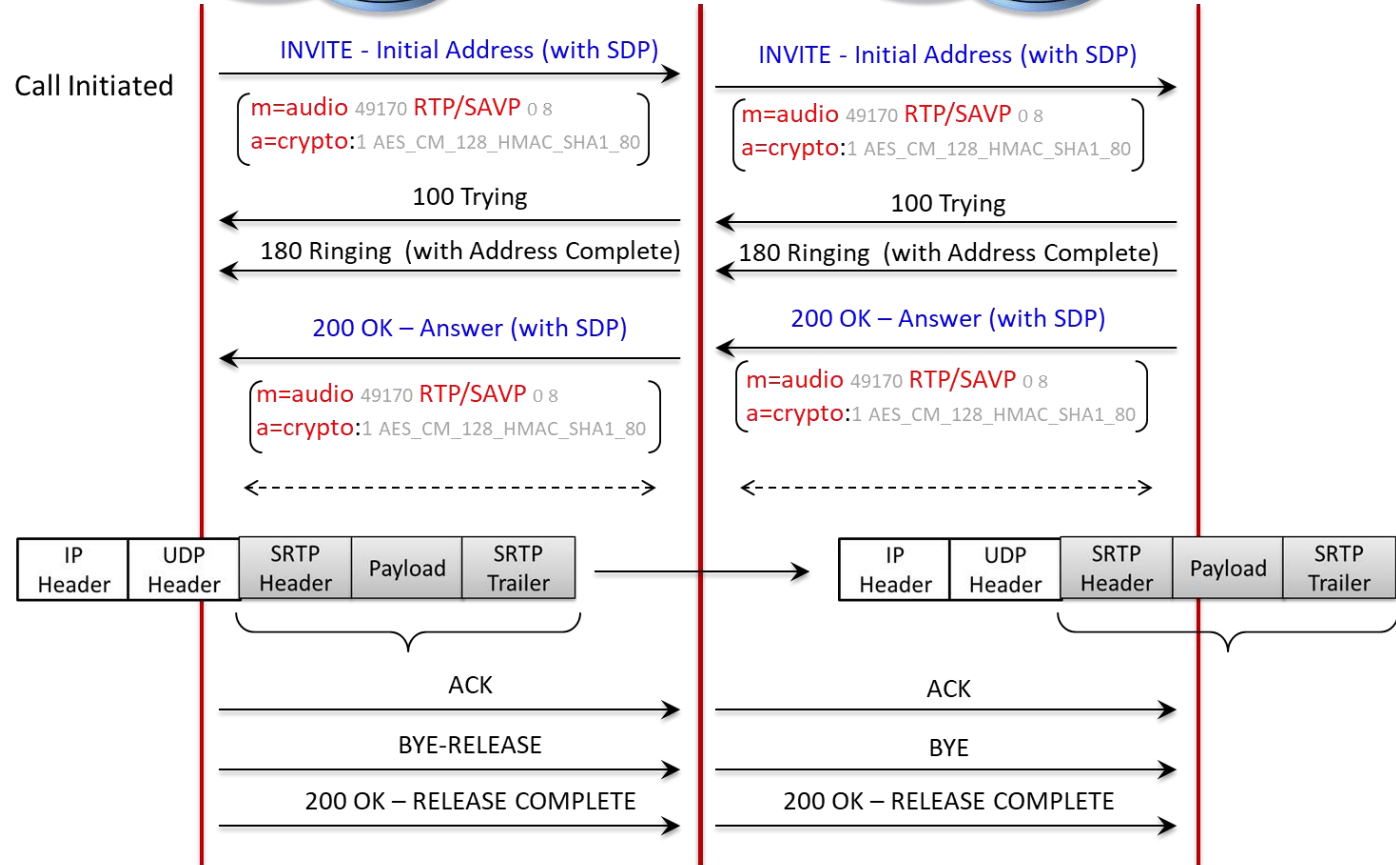
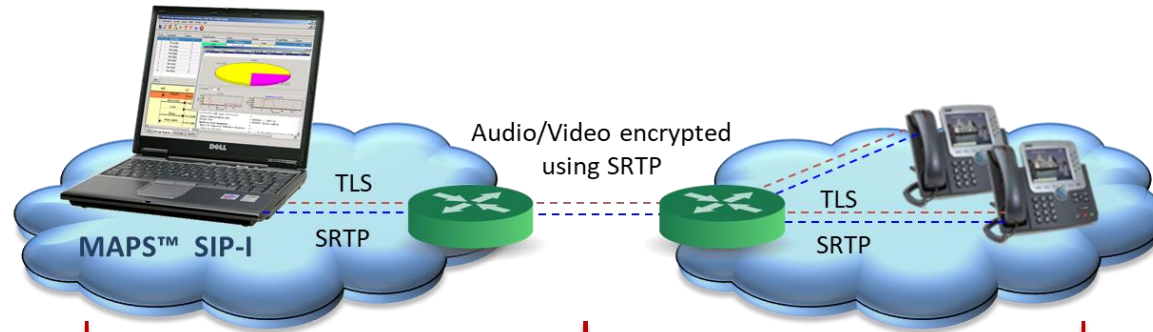
- *Pre recorded GLW files*
- *DTMF, MF Digits*
- *User Defined Tones*
- *Insert Voice*
- *FAX T.30*

## *Rx*

- *GLW files*
- *DTMF, MF Digits*
- *User Defined Tones*
- *FAX T.30*



# Secured RTP Traffic Simulation



# Secured RTP Traffic Simulation

**GL MAPS (Message Automation Protocol Simulation) (Sip1 ITU) - [Call Reception]**

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Even...	Results
1	SipCallControl.gls	Profile0001	GL-MAPS_4_504878194-2961-1636@192.168.12.212	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 -- INITIAL ADDRESS → 15:21:14.145.8792
100 Trying ← 15:21:14.148.3683
180 Ringing -- ADDRESS COMPLETE ← 15:21:14.149.2221
200 OK -- ANSWER ← 15:21:14.191.5561
ACK → 15:21:14.198.4357
BYE -- RELEASE → 15:21:23.245.4077
200 OK -- RELEASE COMPLETE ← 15:21:23.247.2967
```

**Find**

```
--unique-boundary-1
Content-Type: application/sdp
v=0
o=9880325901 33852938 33852938 IN IP4 192.168.12.212
s=SIP Call
c=IN IP4 192.168.12.212
t=0 0
m=audio 1024 RTP/SAVP 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=crypto:1 AES CM 128 HMAC SHA1 80 inline:vSjhjHUAAC9ImCMdQAAAL0iYYx1AAAAviJhjHUA
a=sendrecv

--unique-boundary-1
Content-Type: application/ISUP; version=gr394; base=gr394
Content-Disposition: signal;handling=required

===== ISUP Layer =====
0000 Message Type = 00000001 Initial address
Mandatory Fixed Parameters =
Nature Of Connection Indicators Parameter =
0001 Satellite indicator = 00 no satellite circuit in the connection
```

Scripts **Message Sequence** Event Config Script Flow

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

# Testbed Configuration

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface. The window title is "MAPS (Message Automation Protocol Simulation) (Sipl ITU) - [Testbed Setup - TestBedDefault]". The menu bar includes "Configurations", "Emulator", "Reports", "Editor", "Debug Tools", "Windows", and "Help". The toolbar contains various icons for file operations and simulation control. The main area is divided into two panes. The left pane shows a tree view of configuration options under "SIP Configuration":

Config	Value
SIP Configuration	
End User Configuration	UserAgent_Profiles....
RTP Core IP Address	192.168.1.31
IPspoofing	Disable

The right pane shows a "Enable" checkbox which is checked. At the bottom of the right pane are "Start" and "Edit" buttons. The status bar at the bottom of the window shows "Initialisation Errors" and "Error Events" indicators.

# Profile Configuration

GL MAPS (Message Automation Protocol Simulation) (Sipl ITU) - [Profile Editor - UserAgent\_Profiles]

Configurations Emulator Reports Editor Debug Tools Windows Help

Profiles (Edit-F2)

#	Profiles (Edit-F2)	Config	Value	Enable
1	Profile0001	Profile0001		<input checked="" type="checkbox"/>
2	Profile0002			
3	Profile0003			
4	Profile0004			
5	Profile0005			
6	Profile0006			
7	Profile0007			
8	Profile0008			
9	Profile0009			
10	Profile0010			

Apply DiffServ Code Point

Proxy Parameters

- Outbound Proxy Address
- Expiry Time in sec: 3600

OPTIONS Parameters

- Options Timer in secs: 30
- Options Target

Call Parameters

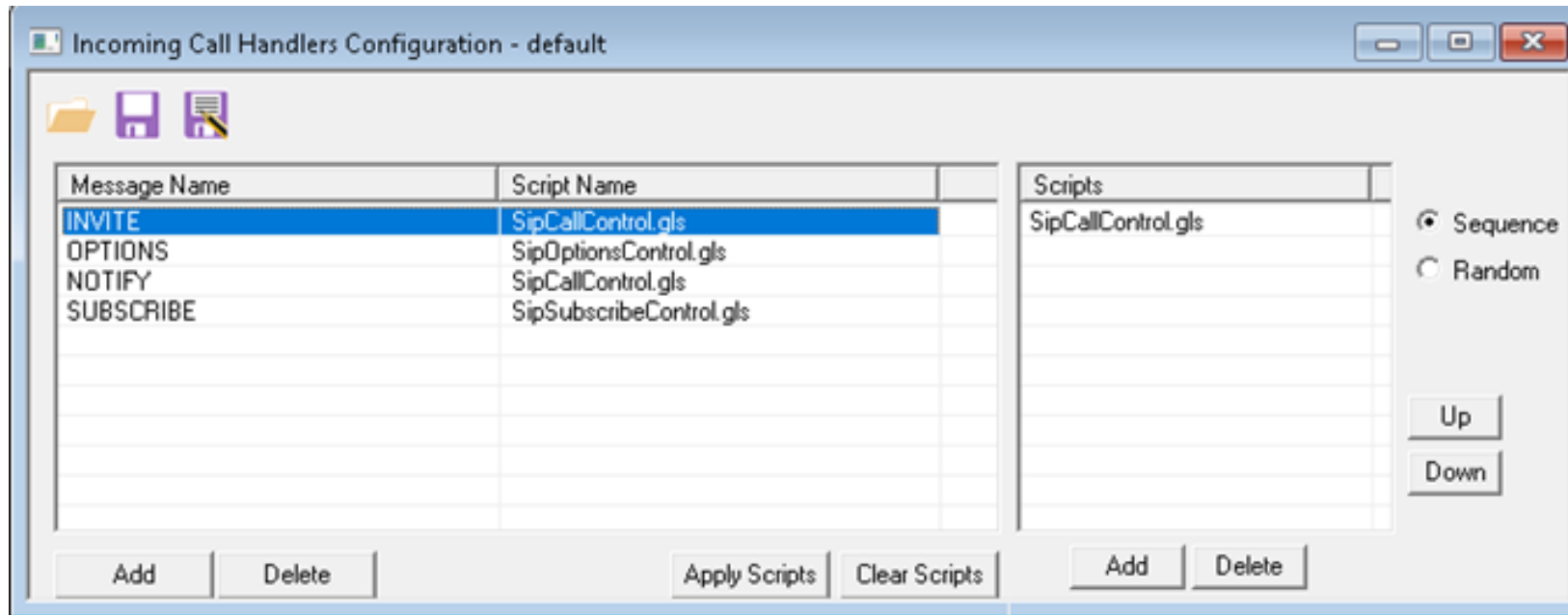
- IP Address Type: IPv4
- Transport: UDP
- Call Type: AudioCall
- Contact Address: 9880325901@192.168...
- Address Of Record: 9880325901@192.168...
- To Address: 8431908401@192.168...
- Subnet Mask: 255.255.255.0
- Cipher Suite for TLS: ALL
- SRTP: Disable
- SRTP Algorithm: AES\_CM\_128\_HMAC...
- Local Call Duration in msec: 0
- Local Call Answer Time in msec: 0

Add Insert Delete Properties

Insert Delete Clear

Initialisation Errors Error Events Captured Errors

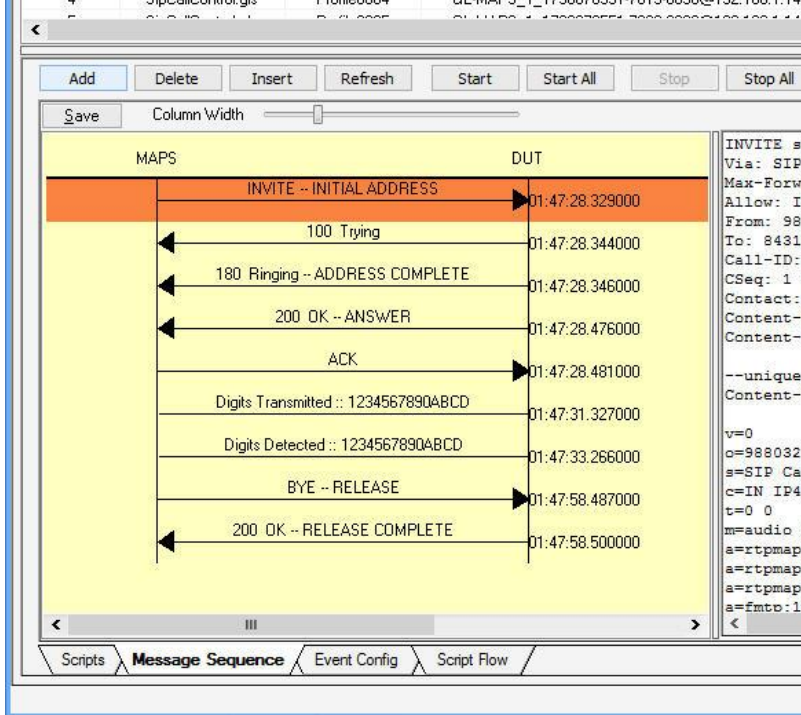
# Incoming Call Handler Configuration



# Call Generation and Reception

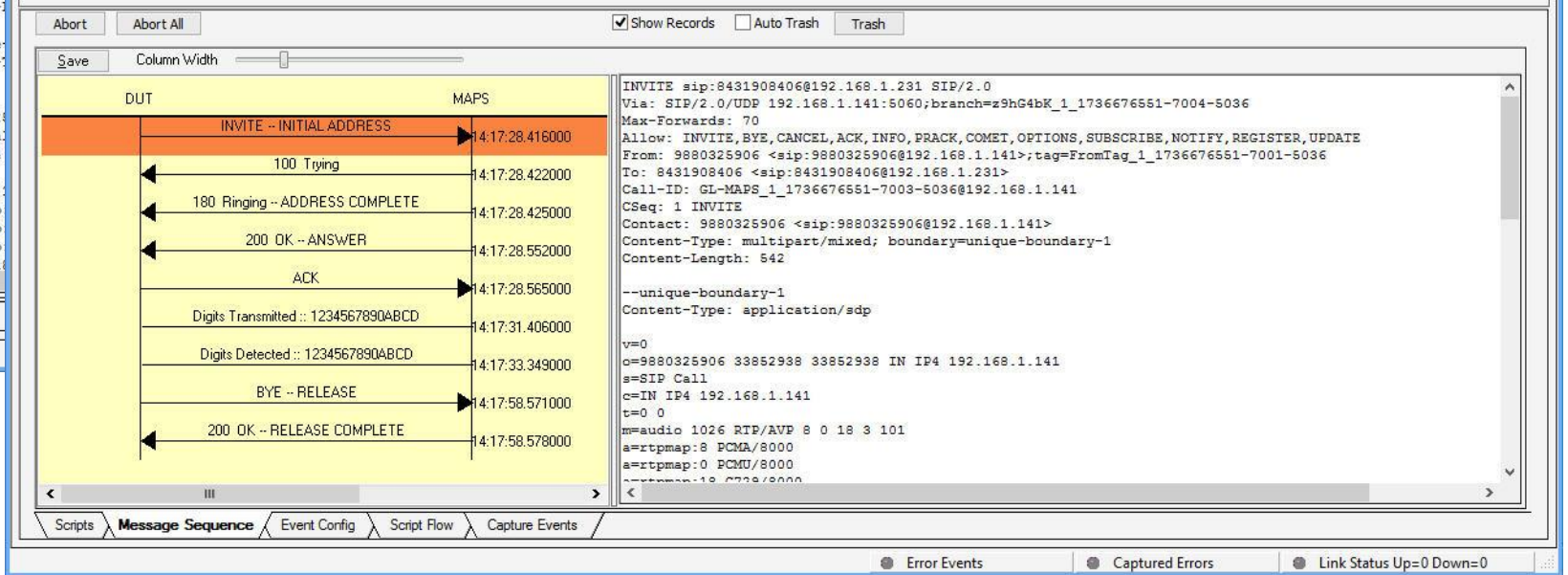
MAPS (Message Automation Protocol Simulation) (Sipl ITU) - [Call Generation - CallGenDefault]

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Ev...	Result	Total Iterations	Completed Iteration
1	SipCallControl.gls	Profile0001	GL-MAPS_1_1736676552-7031-4784@192.168.1.141	Start	Call Terminated	None		Pass	1	1
2	SipCallControl.gls	Profile0002	GL-MAPS_1_1736676551-7011-7996@192.168.1.141	Start	Call Terminated	None		Pass	1	1
3	SipCallControl.gls	Profile0003	GL-MAPS_1_1736676551-6999-6172@192.168.1.141	Start	Call Terminated	None		Pass	1	1
4	SipCallControl.gls	Profile0004	GL-MAPS_1_1736676551-7015-6036@192.168.1.141	Start	Call Terminated	None		Pass	1	1



MAPS (Message Automation Protocol Simulation) (Sipl ITU) - [Call Reception]

Sr No	Script Name	Call Info	Script Execution	Status	Events	Even...	Results
1	SipCallControl.gls	GL-MAPS_1_1736676551-7003-5036@192.168.1.141	Completed	Call Terminated	None		Pass
2	SipCallControl.gls	GL-MAPS_1_1736676551-7015-6036@192.168.1.141	Completed	Call Terminated	None		Pass
3	SipCallControl.gls	GL-MAPS_1_1736676552-7031-4784@192.168.1.141	Completed	Call Terminated	None		Pass
4	SipCallControl.gls	GL-MAPS_1_1736676551-6999-6172@192.168.1.141	Completed	Call Terminated	None		Pass
5	SipCallControl.gls	GL-MAPS_1_1736676551-7011-7996@192.168.1.141	Completed	Call Terminated	None		Pass
6	SipCallControl.gls	GL-MAPS_1_1736676551-7008-3028@192.168.1.141	Completed	Call Terminated	None		Pass
7	SipCallControl.gls	GL-MAPS_1_1736676552-7035-6036@192.168.1.141	Completed	Call Terminated	None		Pass
8	SipCallControl.gls	GL-MAPS_1_1736676552-7028-6172@192.168.1.141	Completed	Call Terminated	None		Pass

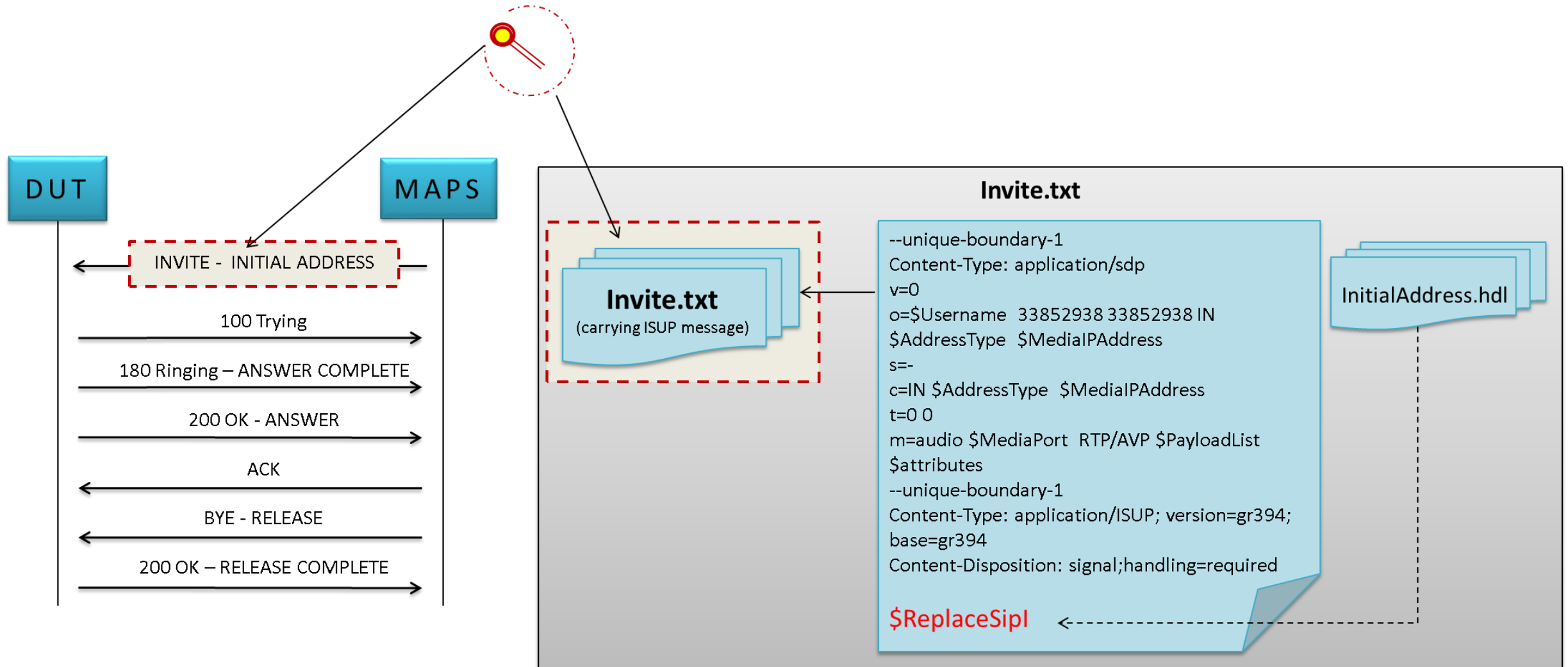


# Event Log

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2017-11-14 12:34:03.010000	RtpCoreSntpAlgorithm =	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:34:03.010000	RtpCoreSntpKey =	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:34:03.115000	200 Ok to Invite Sent	GL-MAPS_4_494846934-1580-2952@19...	SIP-Protocol.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:34:03.132000	Call Connected	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:34:03.166000	Sending RTP File	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:34:28.181000	RTP File Sent	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:35:03.159000	BYE Received	GL-MAPS_4_494846934-1580-2952@19...	SIP-Protocol.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:35:03.163000	200 OK to BYE Sent	GL-MAPS_4_494846934-1580-2952@19...	SIP-Protocol.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:35:03.164000	Call Terminated	GL-MAPS_4_494846934-1580-2952@19...	SipCallControl.gls	ProtScriptId_1_494846989-3...
2017-11-14 12:35:10.724000	Script Initialized	ProtScriptId_0_494914719-3478-3508	SipCallControl.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.728000	INVITE Received	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.728000	Loaded Profile: Profile0001	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.728000	SDP Successful	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.731000	PROGRESS Sent	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.734000	PROGRESS Sent	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.856000	200 Ok to Invite Sent	GL-MAPS_3_494914624-1600-1660@19...	SIP-Protocol.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:10.873000	Call Connected	GL-MAPS_3_494914624-1600-1660@19...	SipCallControl.gls	ProtScriptId_0_494914719-3...
2017-11-14 12:35:22.453000	Script Initialized	ProtScriptId_0_494926447-3487-3932	SipCallControl.gls	ProtScriptId_0_494926447-3...
2017-11-14 12:35:22.462000	INVITE Received	GL-MAPS_3_494926384-1611-2140@19...	SIP-Protocol.gls	ProtScriptId_0_494926447-3...
2017-11-14 12:35:22.463000	Loaded Profile: Profile0002	GL-MAPS_3_494926384-1611-2140@19...	SIP-Protocol.gls	ProtScriptId_0_494926447-3...
2017-11-14 12:35:22.464000	SDP Successful	GL-MAPS_3_494926384-1611-2140@19...	SIP-Protocol.gls	ProtScriptId_0_494926447-3...
2017-11-14 12:35:22.471000	PROGRESS Sent	GL-MAPS_3_494926384-1611-2140@19...	SIP-Protocol.gls	ProtScriptId_0_494926447-3...
2017-11-14 12:35:22.474000	PROGRESS Sent	GL-MAPS_3_494926384-1611-2140@19...	SIP-Protocol.gls	ProtScriptId_0_494926447-3...

Save Events  
  Capture Events to file

# Generating SIP-I Messages

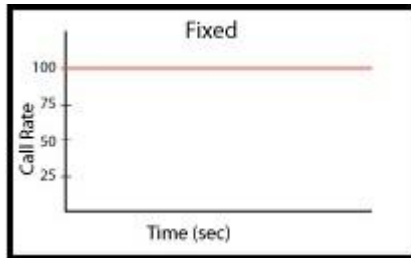




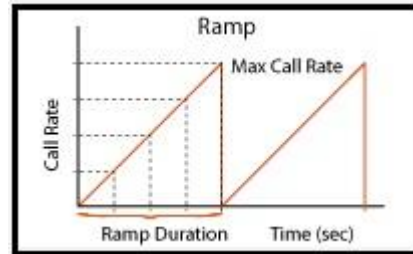
# Load Generation

- Stability/Stress and Performance testing using Load Generation
- Different types of Load patterns to distribute load
- User can load multiple patterns for selected script
- User configurable Test Duration, CPS, Maximum and Minimum Call Rate etc.

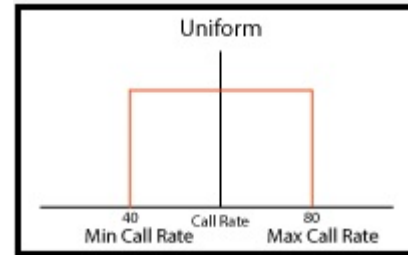
**Fixed**



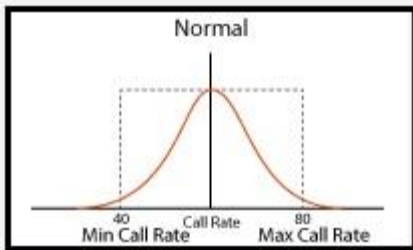
**Ramp**



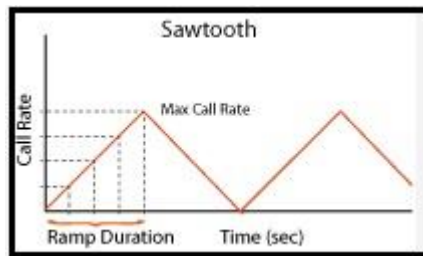
**Uniform**



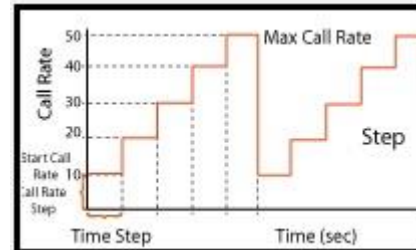
**Normal**



**Saw-tooth**



**Step**



The screenshot shows the 'Load Generation - default' window. It includes fields for 'Total Calls To Generate' (set to '\*'), 'Max Active Calls' (set to 1000), and a checkbox for 'Unique Distributions Per Script'. A table lists distributions: Uniform (MinCR=40, MaxCR=80, Duration=10), Fixed (Call Rate=50, Duration=10), and Normal (MinCR=40, MaxCR=80, Duration=10). Below are sections for 'Scripts' (containing 'SipCallControl') and 'Profile' (listing Profile0001 through Profile0011). At the bottom, there are controls for 'Stop Time' (Days, Hours, Minutes) and 'Start Time' (00:00:00.000) and 'End Time' (00:00:00.000) with 'Pause' and 'Start' buttons.

# Bulk Call Simulation using CSV File

The image shows a bulk call simulation setup in MAPS (Message Automation Protocol Simulation) software. On the left, an Excel spreadsheet named 'Bulk\_UA\_Profiles.csv' contains a list of contacts with columns for 'Contact', 'AddressOfRecord', and 'To'. The 'To' column contains email addresses like '1000@192.168.12.26'.

The main window displays the MAPS interface with a table of simulation results:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	E	Result	Total Iterat
1	SipCallControl.gls		GL-MAPS_1_714522183-5030-1812@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
2	SipCallControl.gls		GL-MAPS_1_714522182-5026-5216@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
3	SipCallControl.gls		GL-MAPS_1_714522184-5062-3432@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
4	SipCallControl.gls		GL-MAPS_1_714522183-5034-1068@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
5	SipCallControl.gls		GL-MAPS_1_714522183-5041-3632@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
6	SipCallControl.gls		GL-MAPS_1_714522183-5040-1952@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
7	SipCallControl.gls		GL-MAPS_1_714522184-5058-1812@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
8	SipCallControl.gls		GL-MAPS_1_714522183-5050-5216@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
9	SipCallControl.gls		GL-MAPS_1_714522184-5067-3432@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	
10	SipCallControl.gls		GL-MAPS_1_714522183-5054-1068@192.168.1	Stop	Send_File-Started	SIP_TerminateCall		Pass	

Below the table, a message sequence diagram shows the interaction between MAPS and DUT (Device Under Test). The sequence includes an INVITE message from MAPS to DUT at 05:52:15.007000, followed by 100 Trying, 180 Ringing, and 200 OK responses from DUT to MAPS, and finally an ACK message from DUT to MAPS at 05:52:15.046000.

To the right of the diagram, a SIP log shows the details of the INVITE message:

```

INVITE sip:0001@192.168.1.143 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.141:5060;branch=z9hG4bK_1_714522183-5031-1812
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER
From: 0001 <sip:0001@192.168.1.141>;tag=FromTag_1_714522183-5028-1812
To: 0001 <sip:0001@192.168.1.143>
Call-ID: GL-MAPS_1_714522183-5030-1812@192.168.1.141
CSeq: 1 INVITE
Contact: 0010 <sip:0001@192.168.1.141>
Content-Type: application/sdp
Content-Length: 246

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=audio 1086 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
    
```

# Customizations - Statistics and Reports

MOS, R-Factor

Packet Loss

Packets Discarded

Duplicate Packets

Out-Of-Sequence

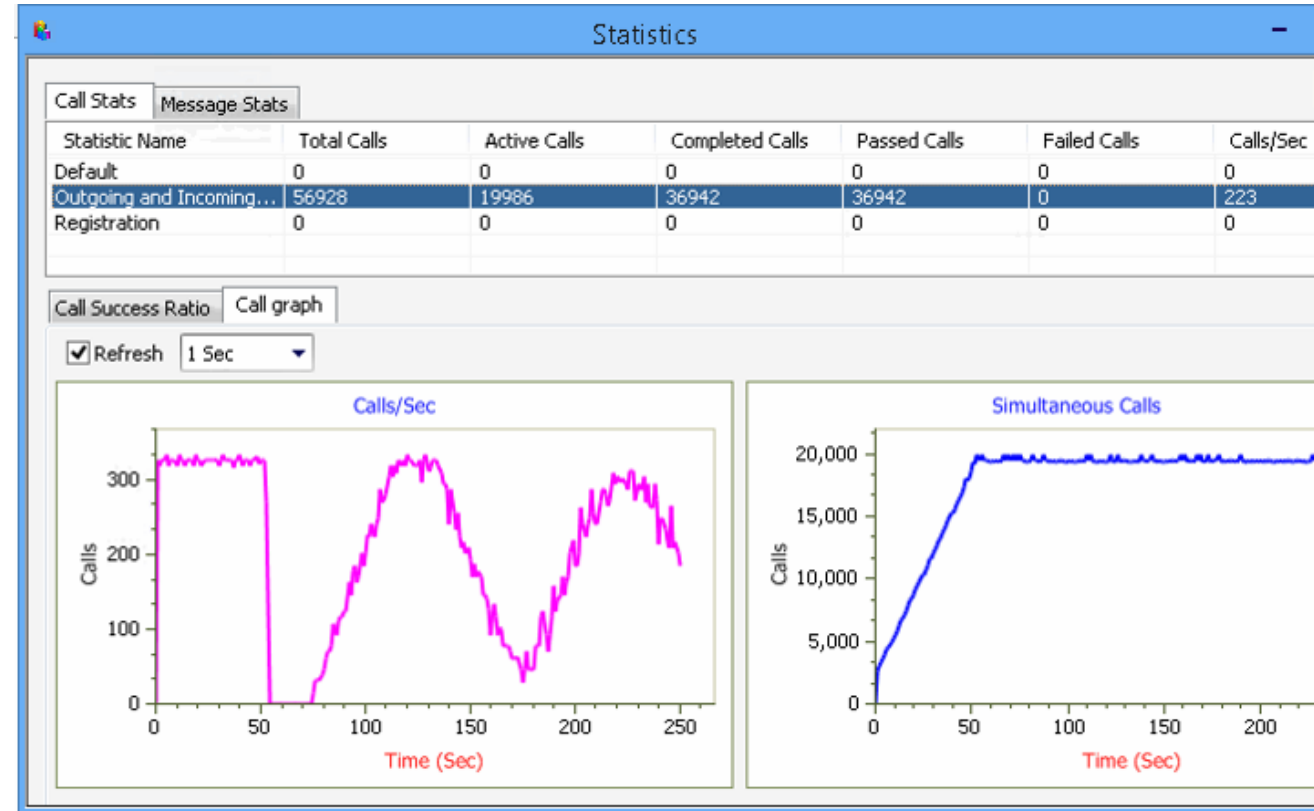
Packets

Jitter Statistics

User Defined Statistics - VoiceQualityStats

Packet Stats

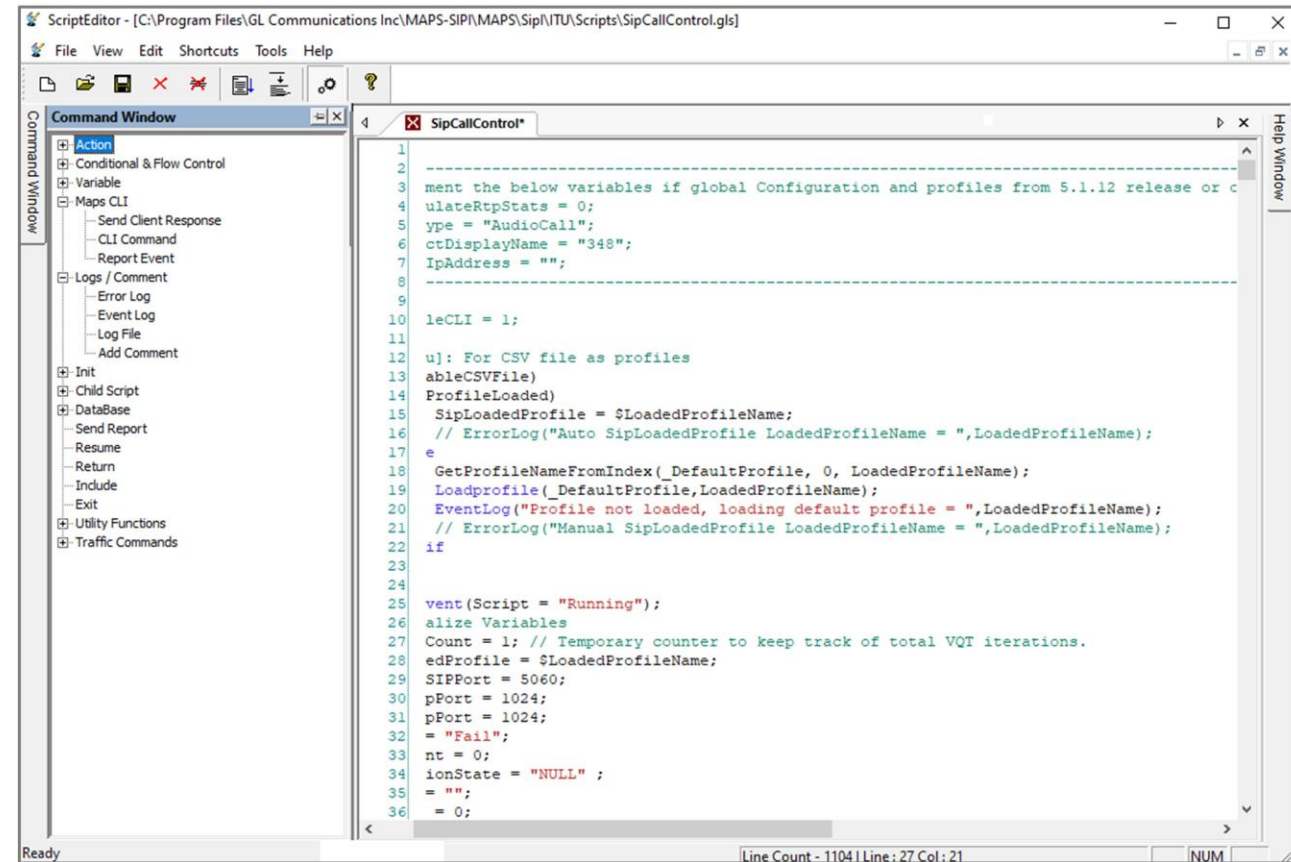
Name	Values
Active RTP Sessions	1987
Completed RTP Sessions	1548093
Sessions With Zero Receive Traffic	0
MOS Score Stats	0
Sessions with Mos ( 5.0 - 4.0 )	612618 [39%]
Sessions with Mos ( 4.0 - 3.0 )	852971 [55%]
Sessions with Mos ( 3.0 - 2.0 )	73446 [4%]
Sessions with Mos ( < 2.0 )	9058 [0%]
Total RTP Packet Sent	4485008797
Total RTP Packet Received	4481760883
Packet-Loss Stats	0
Total PacketLoss	4072 [0%]
Sessions with Zero Packet-Loss	1534967 [99%]
Sessions with Packet-Loss(<1%)	13126 [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	3738934 [0%]
Sessions with Zero Packet-Discard	1464299 [94%]
Sessions with Packet-Discard(<1%)	41479 [2%]
Sessions with Packet-Discard(1% - 5%)	37232 [2%]
Sessions with Packet-Discard(5% - 10%)	4843 [0%]
Sessions with Packet-Discard(>10%)	240 [0%]
Packet-Duplicate Stats	0
Total Duplicate Packet	0 [0%]
Sessions with Zero Duplicate Packets	1539942 [99%]
Sessions with Duplicate Packets(<1%)	0 [0%]
Sessions with Duplicate Packets(1% - 5%)	0 [0%]
Sessions with Duplicate Packets(5% - 10%)	0 [0%]
Sessions with Duplicate Packets(>10%)	0 [0%]
Packet-Out Of Sequence Stats	0 [0%]
Total Out Of Sequence Packet	0 [0%]
Sessions with Zero OOS Packets	1539942 [99%]
Sessions with OOS Packets(<1%)	0 [0%]
Sessions with OOS Packets(1% - 5%)	0 [0%]
Sessions with OOS Packets(5% - 10%)	0 [0%]
Sessions with OOS Packets(>10%)	0 [0%]
Jitter Stats	0
Sessions with Jitter( < 1 msec)	1450779 [93%]
Sessions with Jitter( < 5 msec)	93031 [6%]
Sessions With Jitter( < 10 msec)	4841 [0%]
Sessions With Jitter(>= 10 msec)	350 [0%]



Call Stats provide a running tabular log of system level stats, tracked stats include Total Calls, Active Calls, Completed Calls, Passed Calls, Failed Calls, Instantaneous Calls/Sec

# Customizations - Call Flow (Scripts)

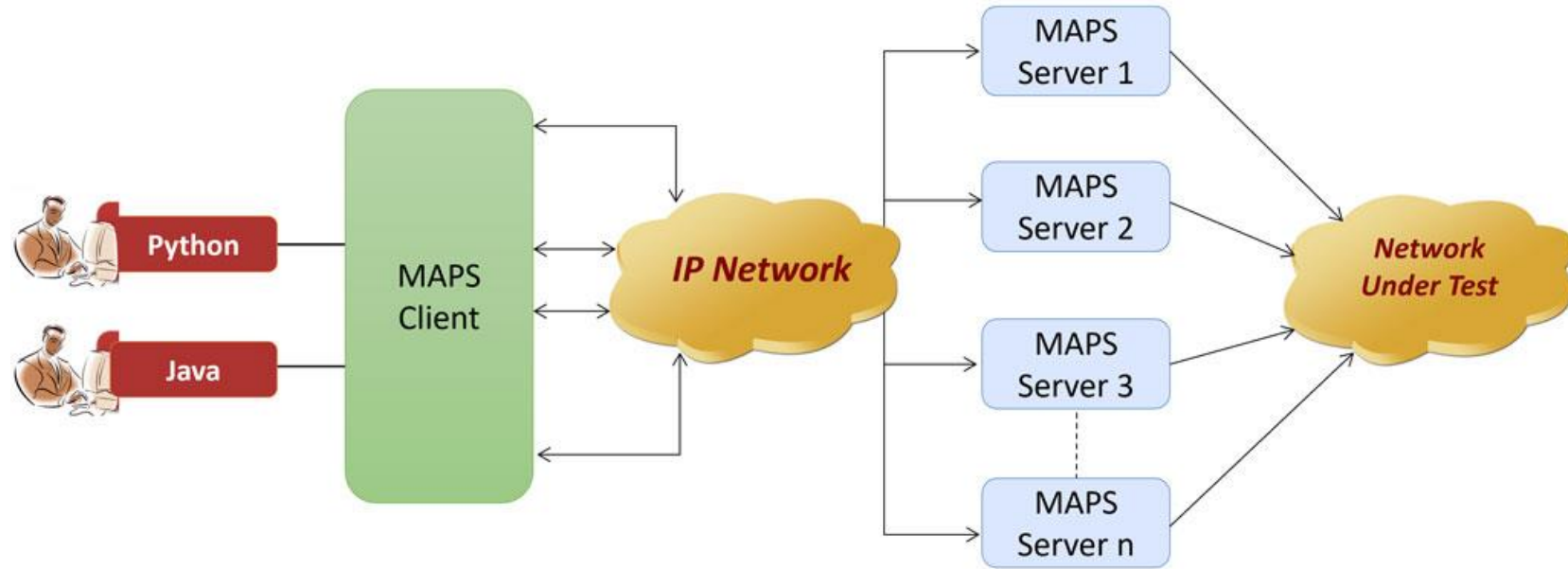
- Scripts are written in our proprietary \*.gls scripting language. They represent generic state machines intended provide protocol/signaling logic for a call and establish bearer traffic
- Each instance of a script corresponds to a single transaction/call, i.e if you place 500 calls in parallel you will have 500 script instances running at once. If you place 500 calls in series the same script will execute and terminate 500 times
- It is possible to create your own scripts, but almost never necessary! We attempt to provide all necessary scripts out of the box



```
ScriptEditor - [C:\Program Files\GL Communications Inc\MAPS-SIP\MAPS\Sip\ITU\Scripts\SipCallControl.gls]
File View Edit Shortcuts Tools Help
Command Window
Action
Conditional & Flow Control
Variable
Maps CLI
  Send Client Response
  CLI Command
  Report Event
Logs / Comment
  Error Log
  Event Log
  Log File
  Add Comment
Init
Child Script
DataBase
  Send Report
  Resume
  Return
  Include
Exit
Utility Functions
Traffic Commands
SipCallControl*
1
2 -----
3 ment the below variables if global Configuration and profiles from 5.1.12 release or c
4 ulateRtpStats = 0;
5 ype = "AudioCall";
6 ctDisplayName = "348";
7 IPAddress = "";
8 -----
9
10 leCLI = 1;
11
12 uj: For CSV file as profiles
13 ableCSVFile)
14 ProfileLoaded)
15 SipLoadedProfile = $LoadedProfileName;
16 // ErrorLog("Auto SipLoadedProfile LoadedProfileName = ",LoadedProfileName);
17 e
18 GetProfileNameFromIndex(_DefaultProfile, 0, LoadedProfileName);
19 Loadprofile(_DefaultProfile,LoadedProfileName);
20 EventLog("Profile not loaded, loading default profile = ",LoadedProfileName);
21 // ErrorLog("Manual SipLoadedProfile LoadedProfileName = ",LoadedProfileName);
22 if
23
24
25 vent(Script = "Running");
26 alize Variables
27 Count = 1; // Temporary counter to keep track of total VQT iterations.
28 edProfile = $LoadedProfileName;
29 SIPPport = 5060;
30 pPort = 1024;
31 pPort = 1024;
32 = "Fail";
33 nt = 0;
34 ionState = "NULL" ;
35 = "";
36 = 0;
Line Count - 1104 | Line: 27 Col: 21
```

# MAPS™ API Architecture

- API wraps our proprietary scripting language in standard languages familiar to the user:
  - Python
  - Java
- Clients and Servers support a “Many-to-Many” relationship, making it very easy for users to develop complex test cases involving multiple signaling protocols



# CLI Support

## MAPS™ CLI Server

```
MapsCLI - Untitled
File Edit View
View Latest Command
E :: 2020-3-18 12:17:31.875000 : Start "TestBedDefault.xml";
E :: 2020-3-18 12:17:31.979000 : LoadProfile "UserAgent_Profiles.xml"
E :: 2020-3-18 12:17:32.198000 : Apply Global Configuration # "_EnableCLI"=1;
E :: 2020-3-18 12:17:32.198000 : IncomingCallHandler # "INVITE"="SipCallControl.gls","IsApiClient"="True";
E :: 2020-3-18 12:17:42.260000 : UserEvent 200003 "SetVariable" # "CodecOption"="Profile0001";
E :: 2020-3-18 12:17:42.373000 : UserEvent 200003 "SetVariable" # "PacketizationTime"="22";
E :: 2020-3-18 12:17:42.480000 : UserEvent 200003 "SIP_AcceptCall";
E :: 2020-3-18 12:17:42.591000 : UserEvent 200003 "GetCallStatus";
E :: 2020-3-18 12:17:43.683000 : UserEvent 200003 "SendFile" # "TxFileName"="voicefiles\Send\G711\LLAW\Wijay.glw","TxFileDuration"
E :: 2020-3-18 12:17:53.749000 : UserEvent 200003 "SIP_TerminateCall";
E :: 2020-3-18 12:17:53.855000 : UserEvent 200003 "GetMessageCount";
E :: 2020-3-18 12:17:53.964000 : UserEvent 200003 "GetMessageInfo" # "Index"=0;
E :: 2020-3-18 12:17:54.073000 : UserEvent 200003 "GetMessageInfo" # "Index"=1;
E :: 2020-3-18 12:17:54.188000 : UserEvent 200003 "GetMessageInfo" # "Index"=2;
E :: 2020-3-18 12:17:54.294000 : UserEvent 200003 "GetMessageInfo" # "Index"=3;
E :: 2020-3-18 12:17:54.404000 : UserEvent 200003 "GetMessageInfo" # "Index"=4;
E :: 2020-3-18 12:17:54.510000 : UserEvent 200003 "GetMessageInfo" # "Index"=5;
E :: 2020-3-18 12:17:54.624000 : UserEvent 200003 "GetMessageInfo" # "Index"=6;
E :: 2020-3-18 12:17:54.729000 : StopScript: 200003;
ServerLog:errCode = 0,errString = connection has been gracefully closed for ClientId =3
CAPS NUM
```

## Sample Python CLI Script

```
Python 3.7.5 Shell
File Edit Shell Debug Options Window Help
Python 3.7.5 [tags/v3.7.5:5c02a39a0b, Oct 15 2019, 00:11:34] [MSC v.1915 64 bit
(AMD64)] on win32
Type "help", "copyright", "credits" or "license()" for more information.
>>>
= RESTART: C:\Program Files\GL Communications Inc\MAPS-SIP\PythonClient\example
e\SIP\SipBasicAnsCall.py
SERVER INITIALIZED
WAITING FOR INCOMING CALL..
CALL CONNECTED
RTP Action pass
12:17:42.072 <- INVITE
INVITE sip:9880325901@192.168.12.105 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.104:5060;branch=z9hG4bK 14 155540690 7463 3712
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,PRACK,COMET,OPTIONS,SUBSCRIBE,NOTIFY,REGISTER,
UPDATE
From: 8431908401 <sip:8431908401@192.168.12.104>;tag=FromTag-11-155540690-7461-3
712
To: 9880325901 <sip:9880325901@192.168.12.105>
Call-ID: GL-MAPS-13-155540690-7463-3712@192.168.12.104
CSeq: 1 INVITE
Contact: 8431908401 <sip:8431908401@192.168.12.104>
Supported: 100rel
Content-Type: multipart/mixed; boundary=unique-boundary-1
Content-Length: 472

--unique-boundary-1
Content-Type: application/sdp

v=0
o=0431900401 39095205 1 IN IP4 192.168.12.104
s=SIP Call
c=IN IP4 192.168.12.104
t=0 0
m=audio 1024 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/3000
a=rtpmap:8 PCMA/3000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv

Ln: 216 Col: 4
```

**Thank you**