MAPS™ SIP
SIP + RTP + MSRP Simulation

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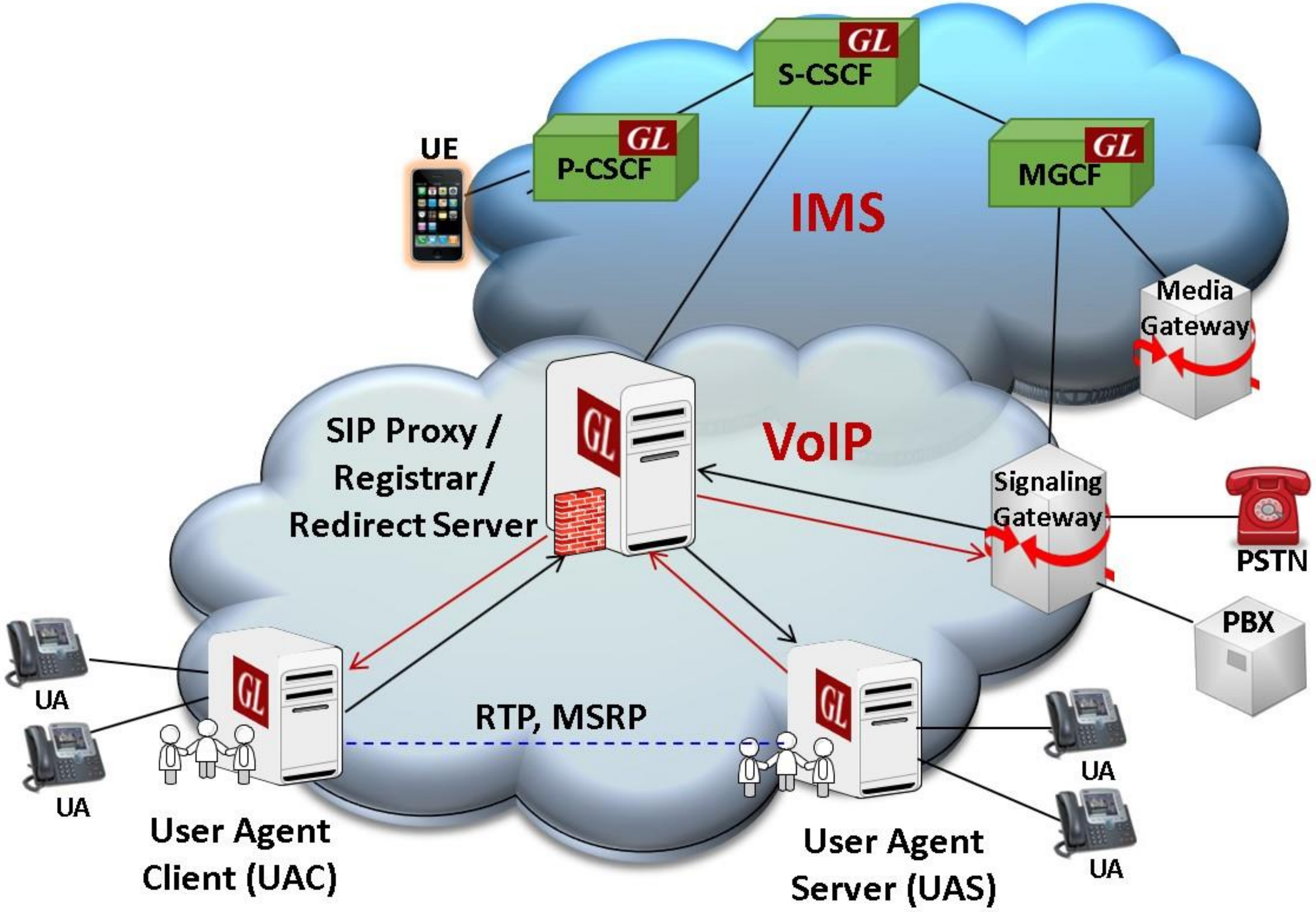
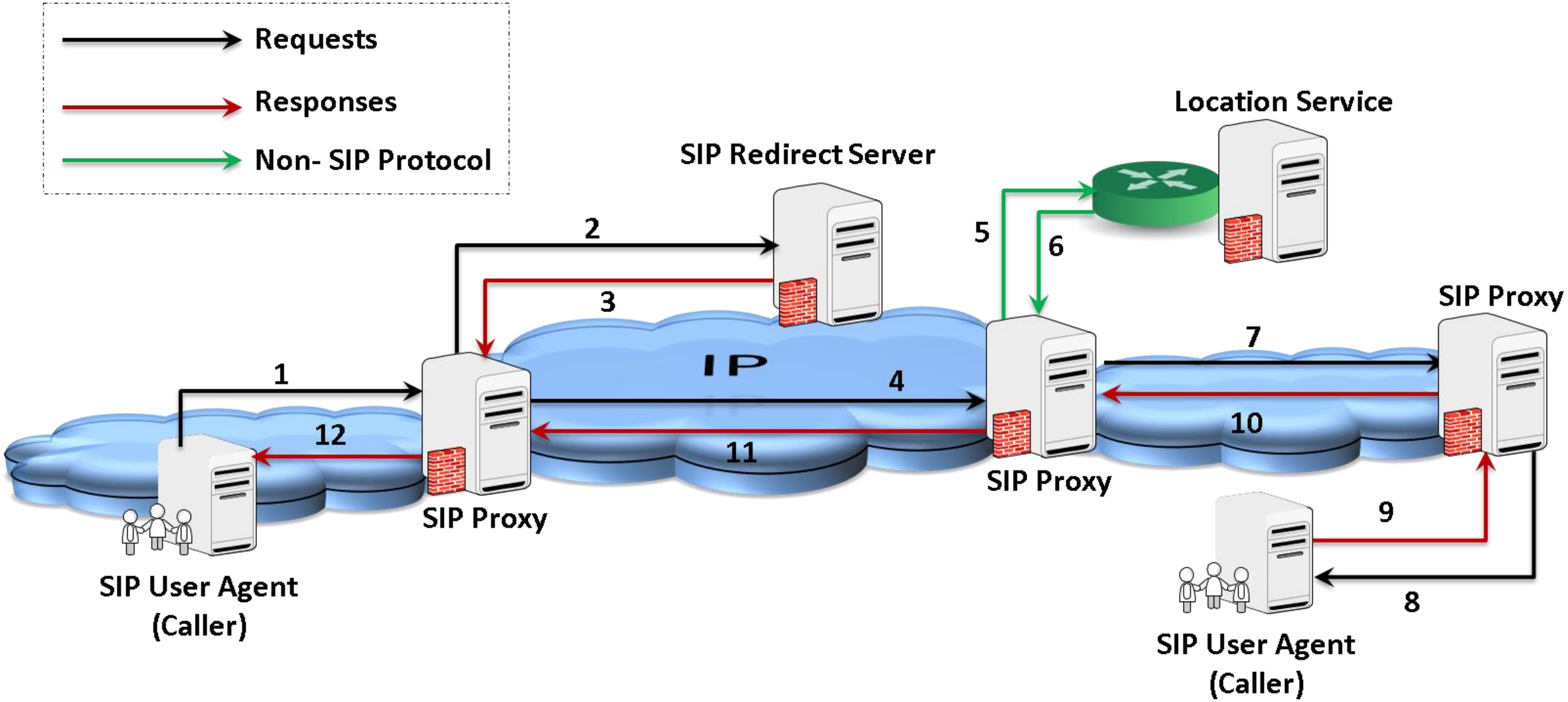
MAPS™ SIP



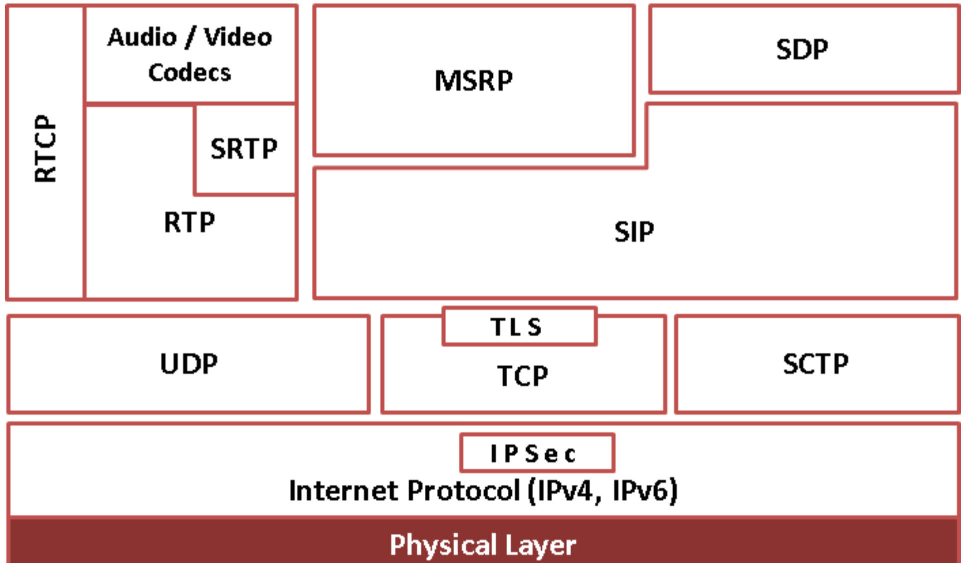
MAPS™ SIP
Normal RTP Traffic Generation
(2000 simultaneous calls)

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)

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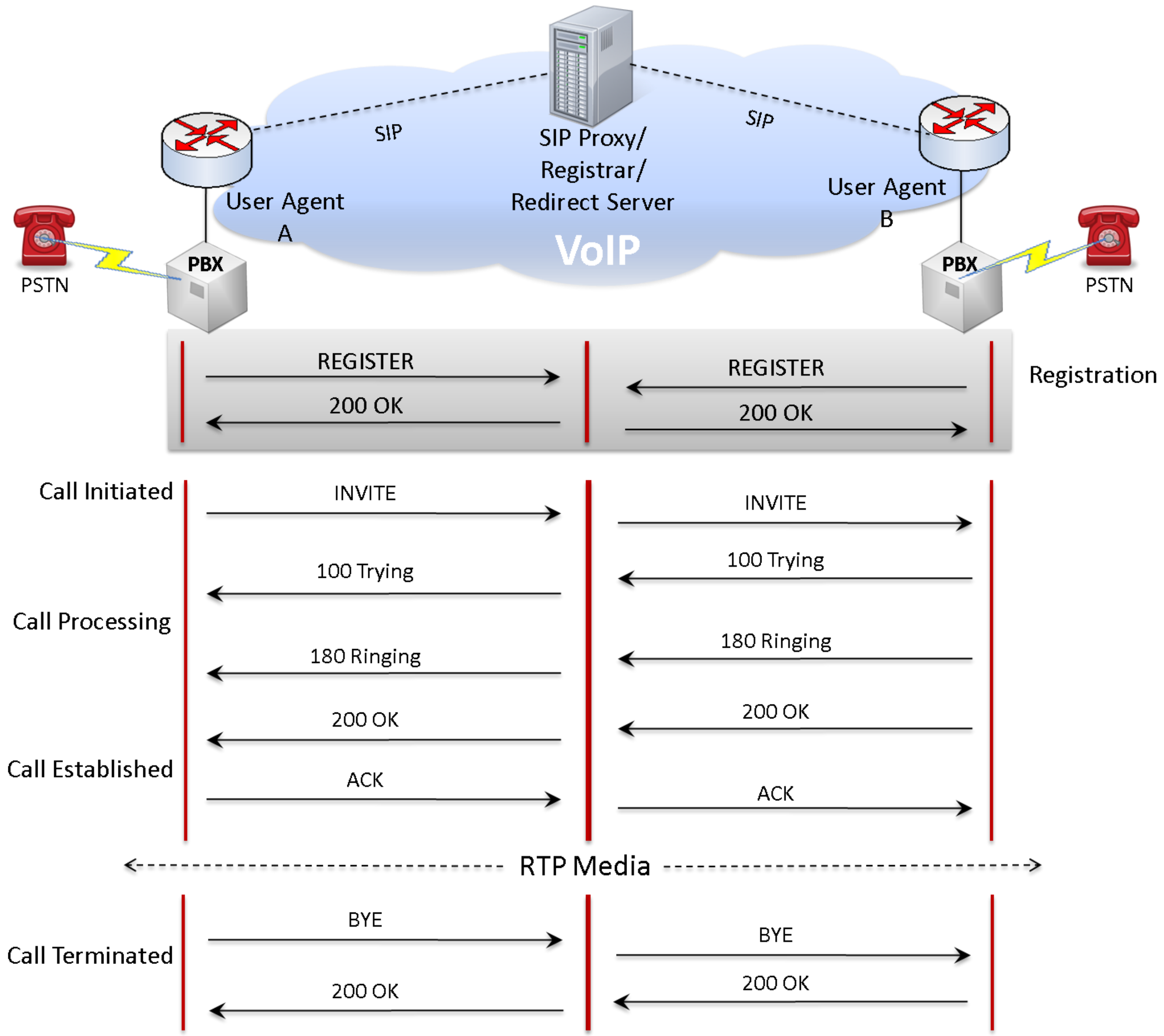
Generic SIP Call Flow



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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	• 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	• 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	• Supports various RTP traffic (PKS102) such as, digits, voice file, tones, IVR, FAX, and Video in IP networks
	• Supports almost all industry standard voice codec types - G.722, G.729, G.726, GSM, AMR, EVRC, EVS, OPUS, SMV, iLBC, SPEEX, and more. *AMR and EVRC variants require additional licenses
	• Supports 64-bit RTP core to enhance performance - handles increased call rate of up to 3000 calls with high volume traffic
	• Supports both G.711 Pass Through Fax Simulation (PKS200) and T.38 Fax Simulation over UDPTL (PKS211)
	• Transmit and receive pre-recorded video traces supporting video codecs like H.264, H.263, and VP8
	• Study packet effects through impairment generation –
	    • Latency (Uniform distributed & Normal distributed)
	    • Packet loss (Periodic & Random)
	    • Packet effects (Duplicate & Out of order)
	• Bulk Video call generation supported with H.264, H.263, and VP8 video codecs
	• Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL)
	• User-defined voice quality statistics for received RTP Traffic can be calculated and updated periodically during run-time to a csv file
	• Supports simulation of SIP/MSRP User Agents end-points in an NG9-1-1 network and send and receive communications over IP networks. MSRP sessions supports simulation of IM Only Calls, Audio and IM Calls, and Video and IM Call types

SIP Call Types

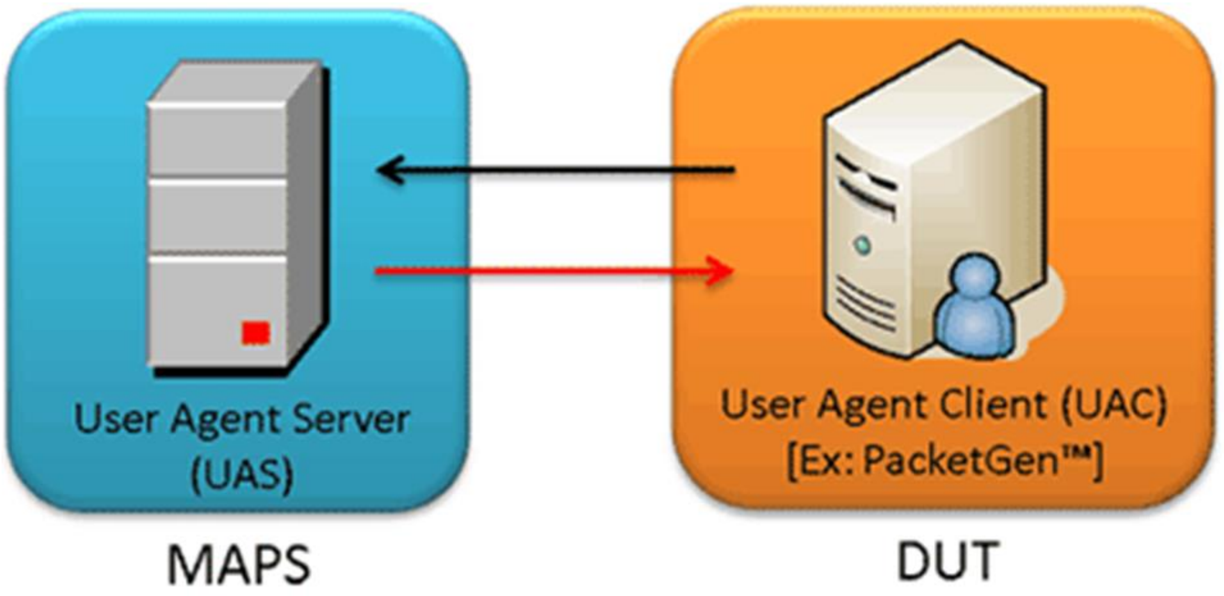
- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX)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

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

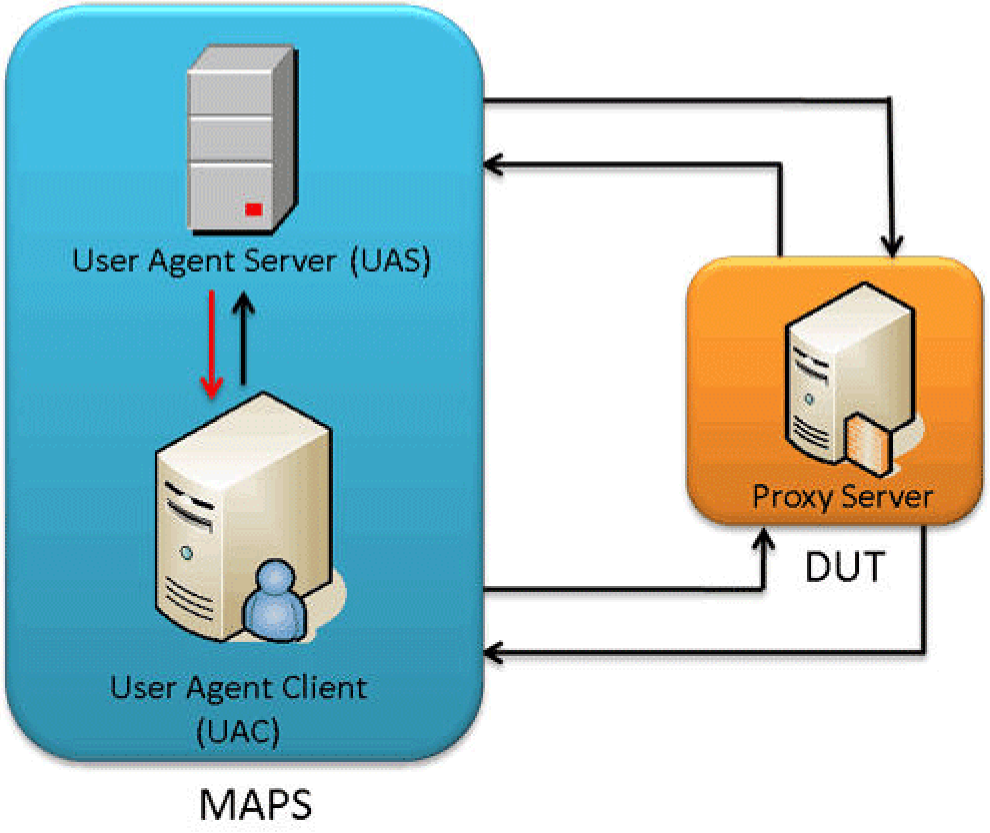


MAPS™ SIP Configured as UAC / UAS

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

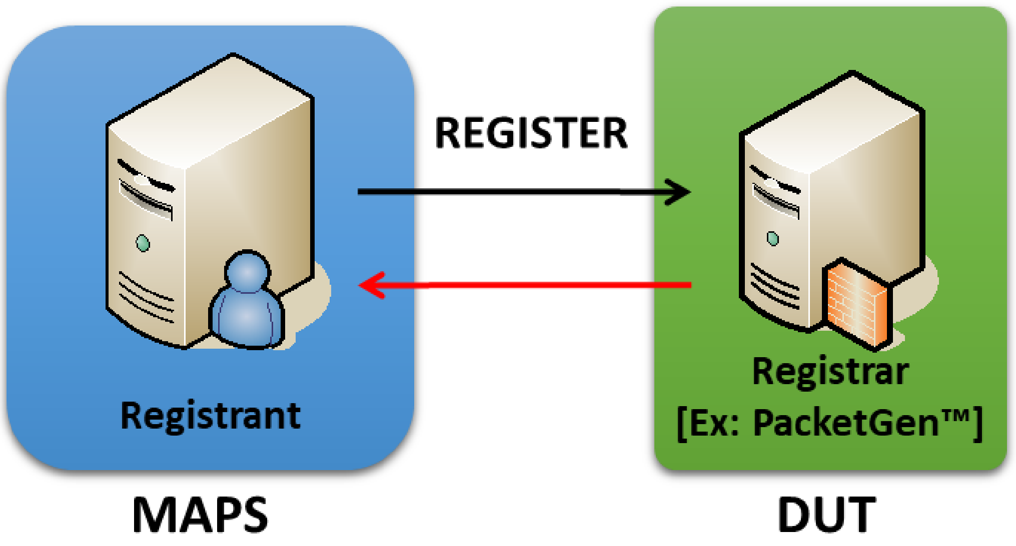


MAPS™ SIP Configured as Registrant

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

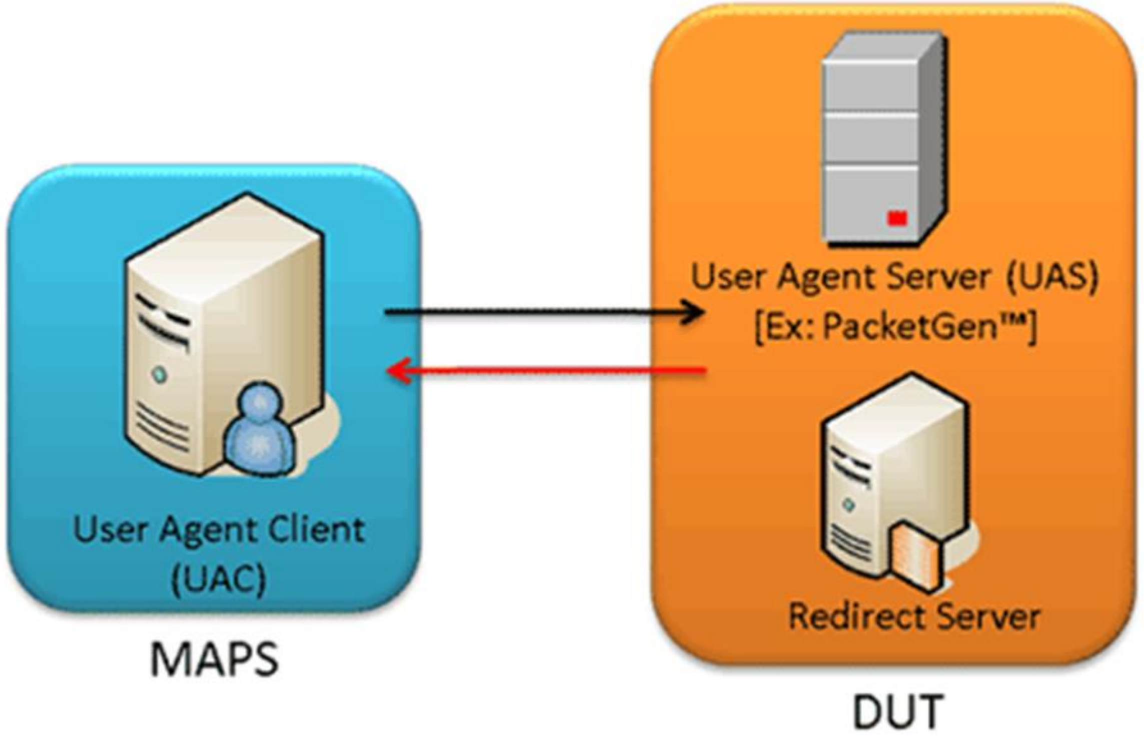


MAPS™ SIP Configured as UAC

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

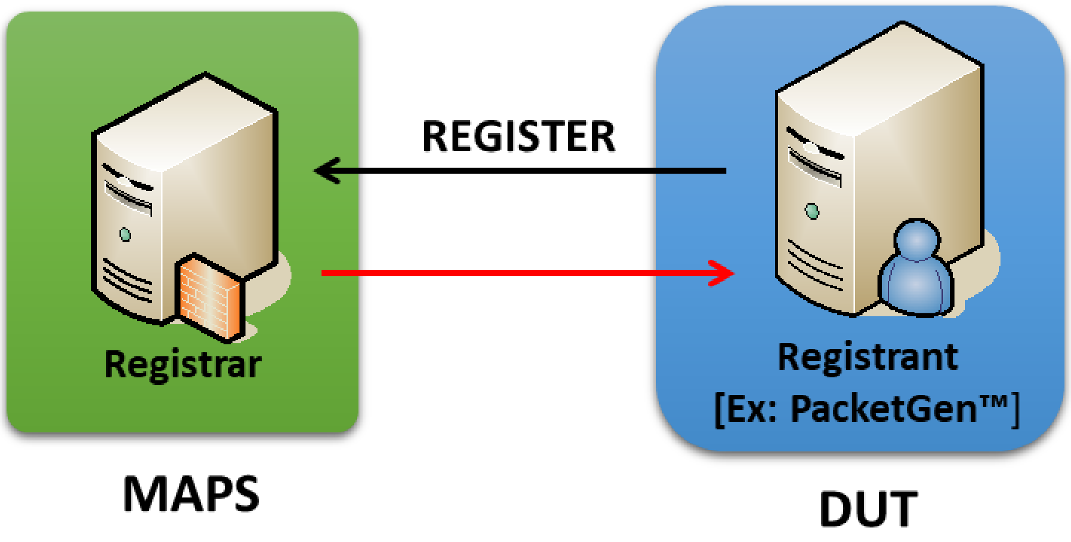


MAPS™ SIP Configured as Registrar

Testing Registrant

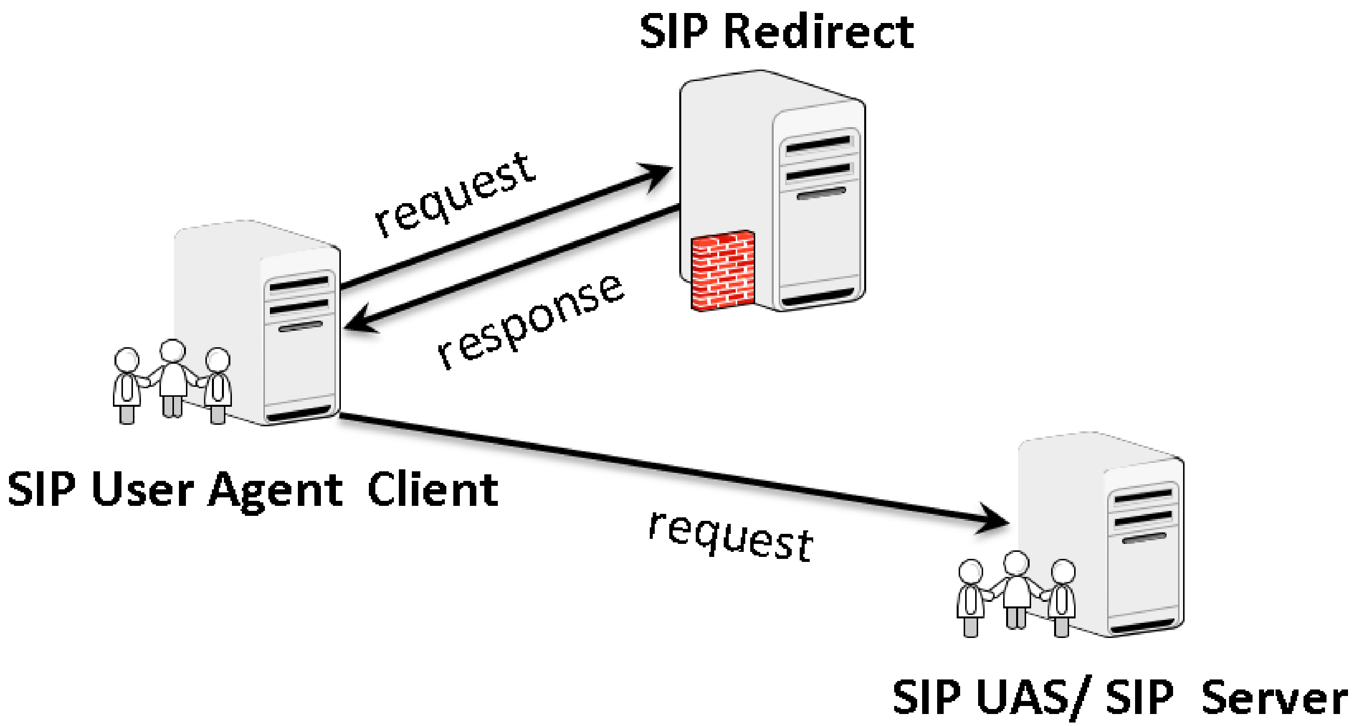
Scenario: MAPS™ acting as Registrar and testing Registrant

- 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)

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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)

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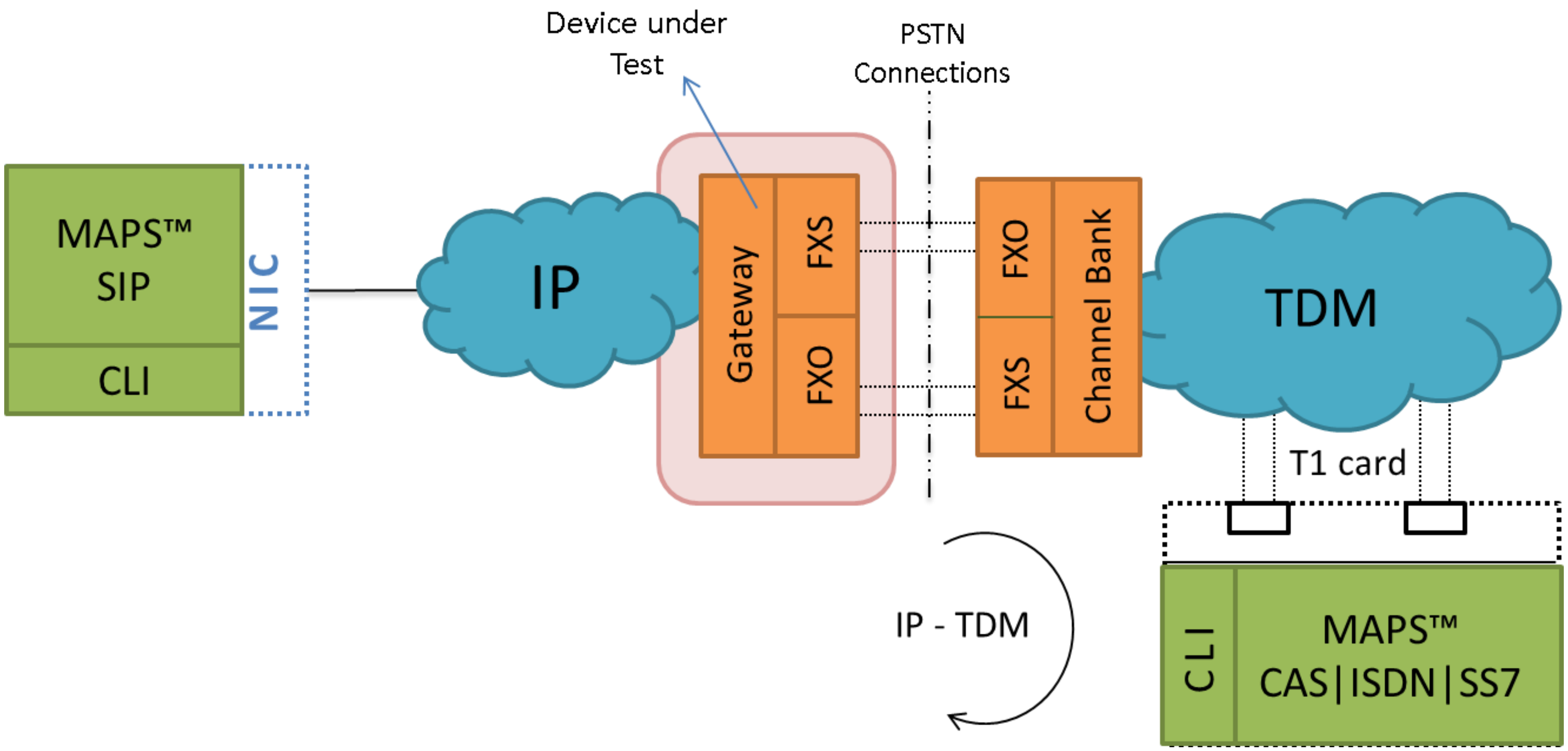
End-to-End Gateway Testing

- Evaluates Gateway / ATA product features such as call connectivity, call signaling, traffic generation, voice quality testing, codec, and hundreds of other features



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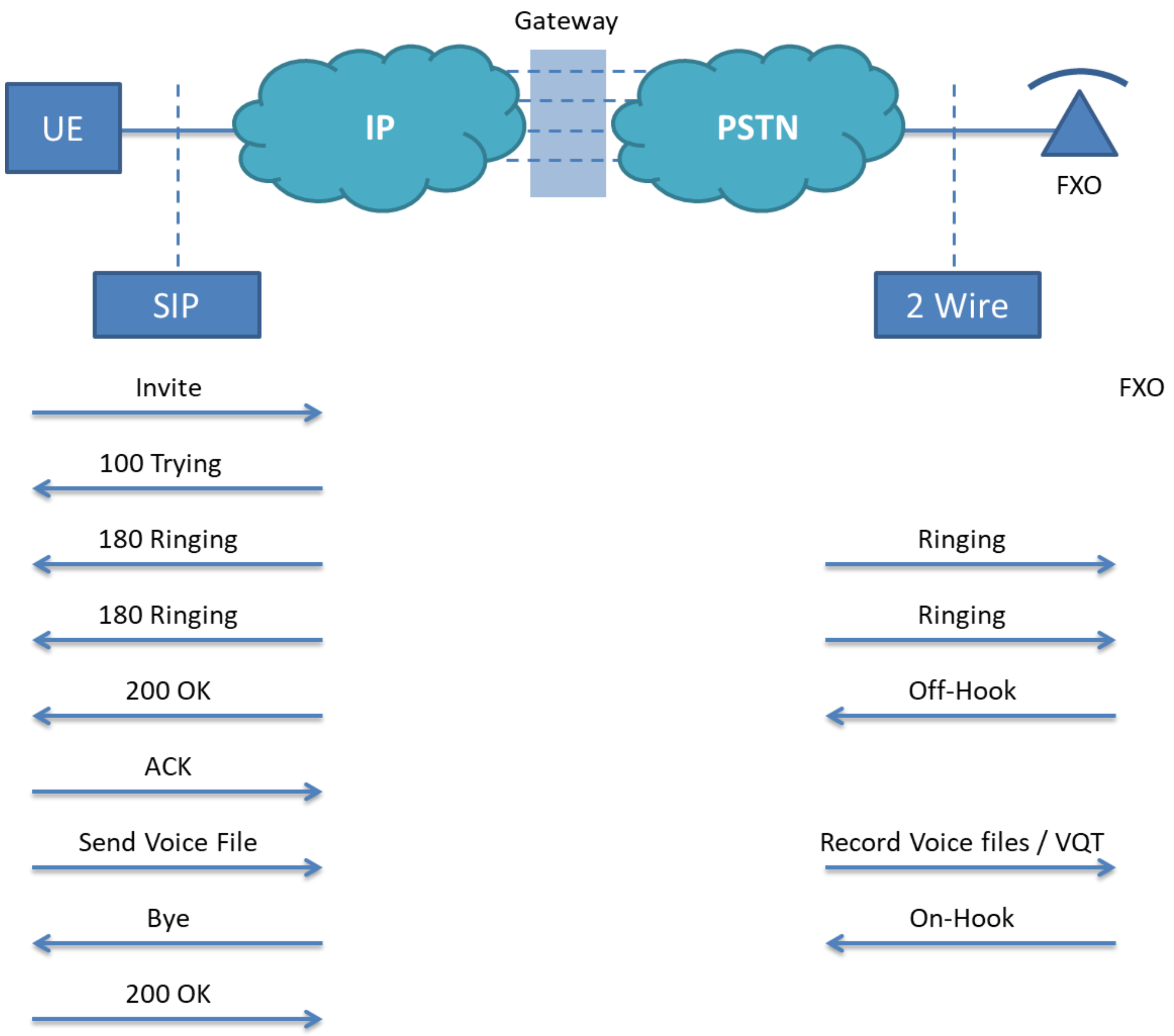
End-to-End Gateway Testing Call Scenario



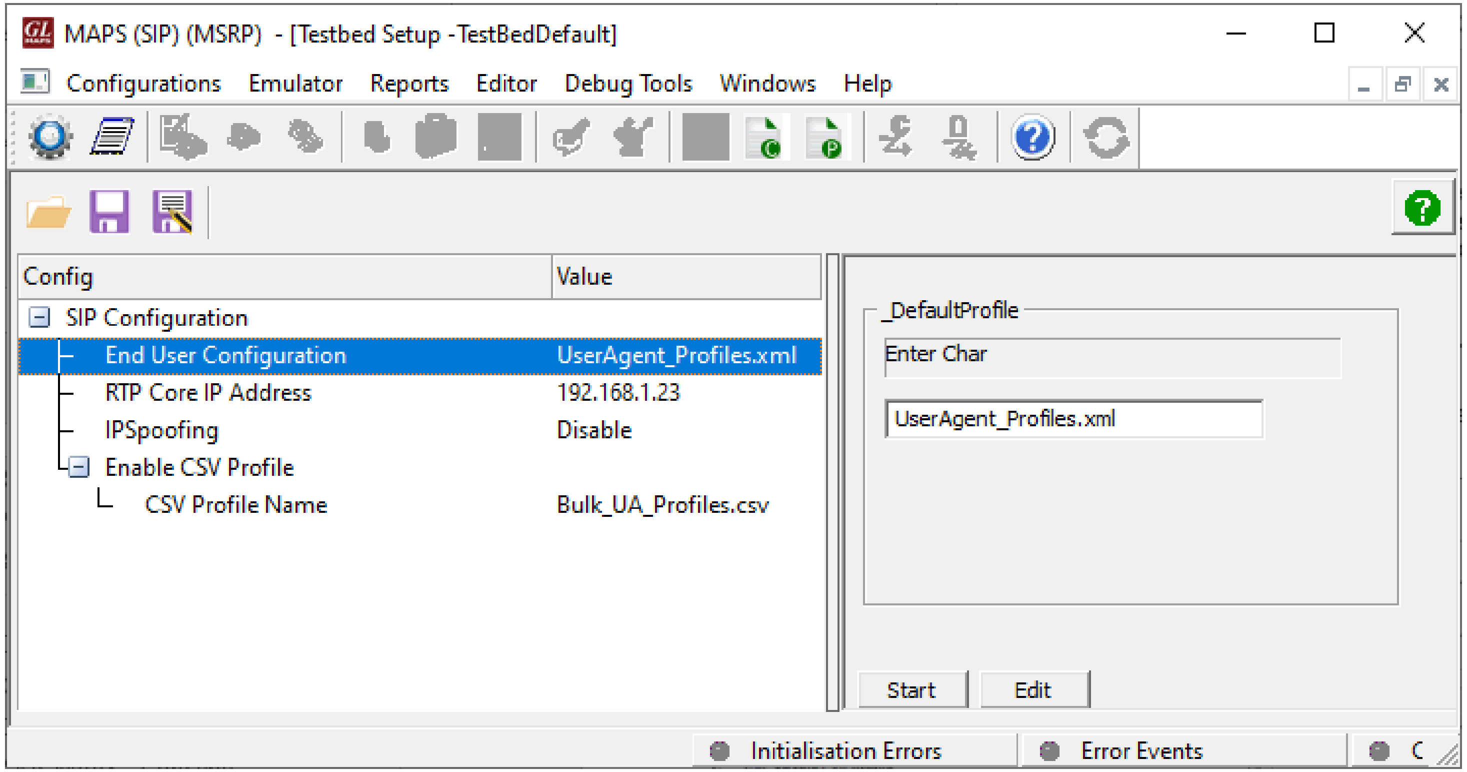
Test Bed Configuration

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

RTP Core IP Address:

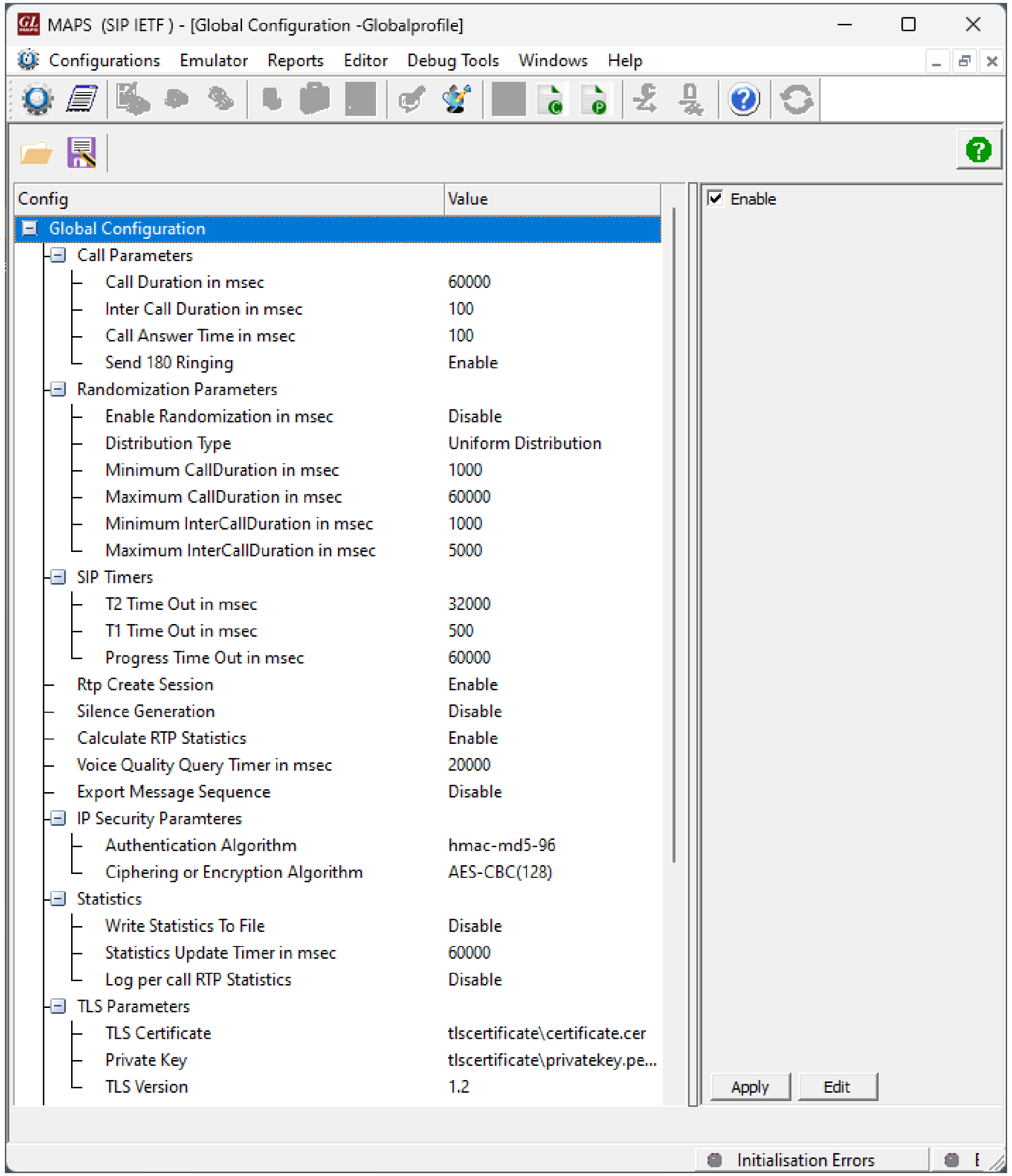
IP Address of the system on which the RTP Core should be invoked

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



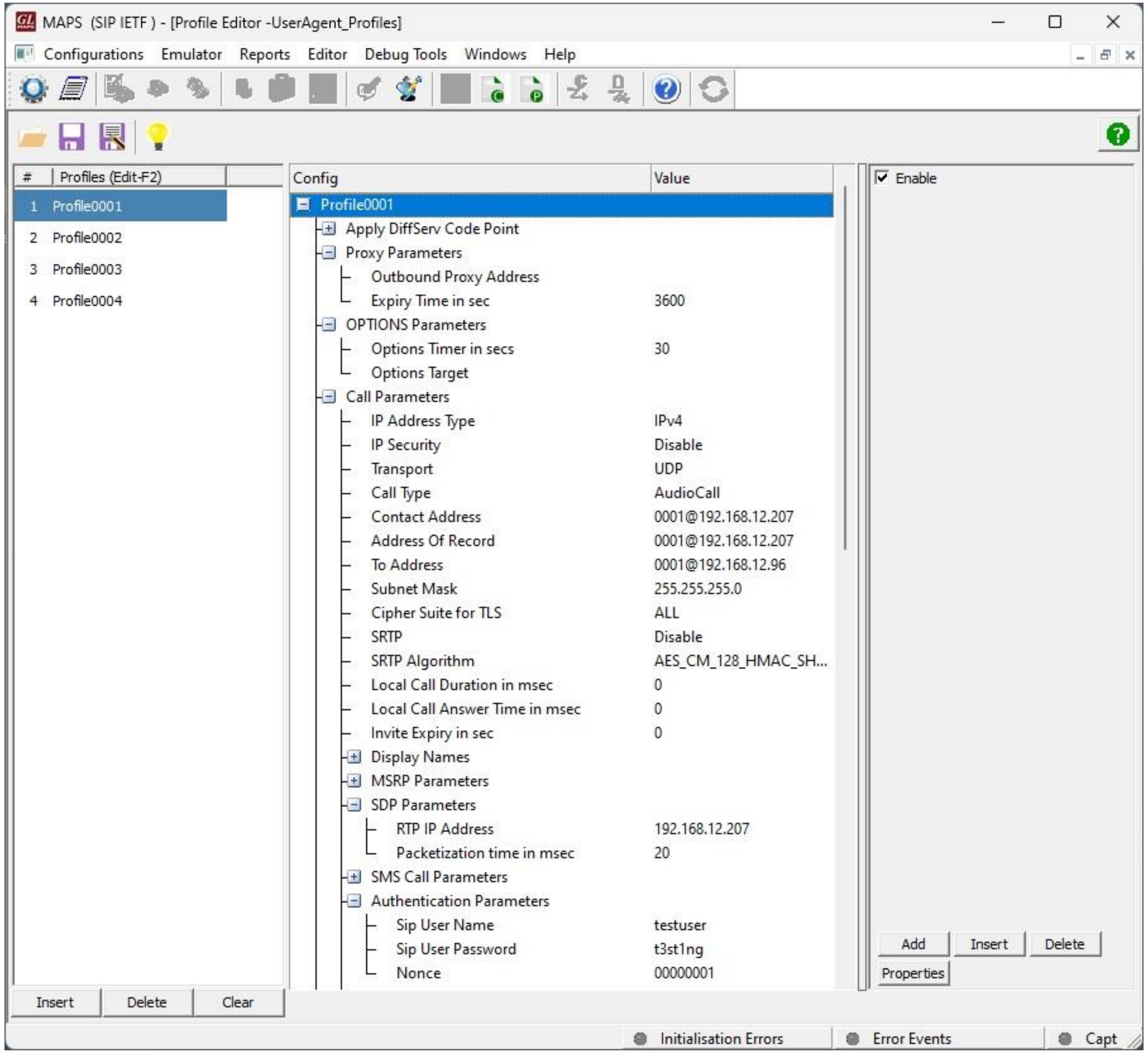
Global Configuration

- 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

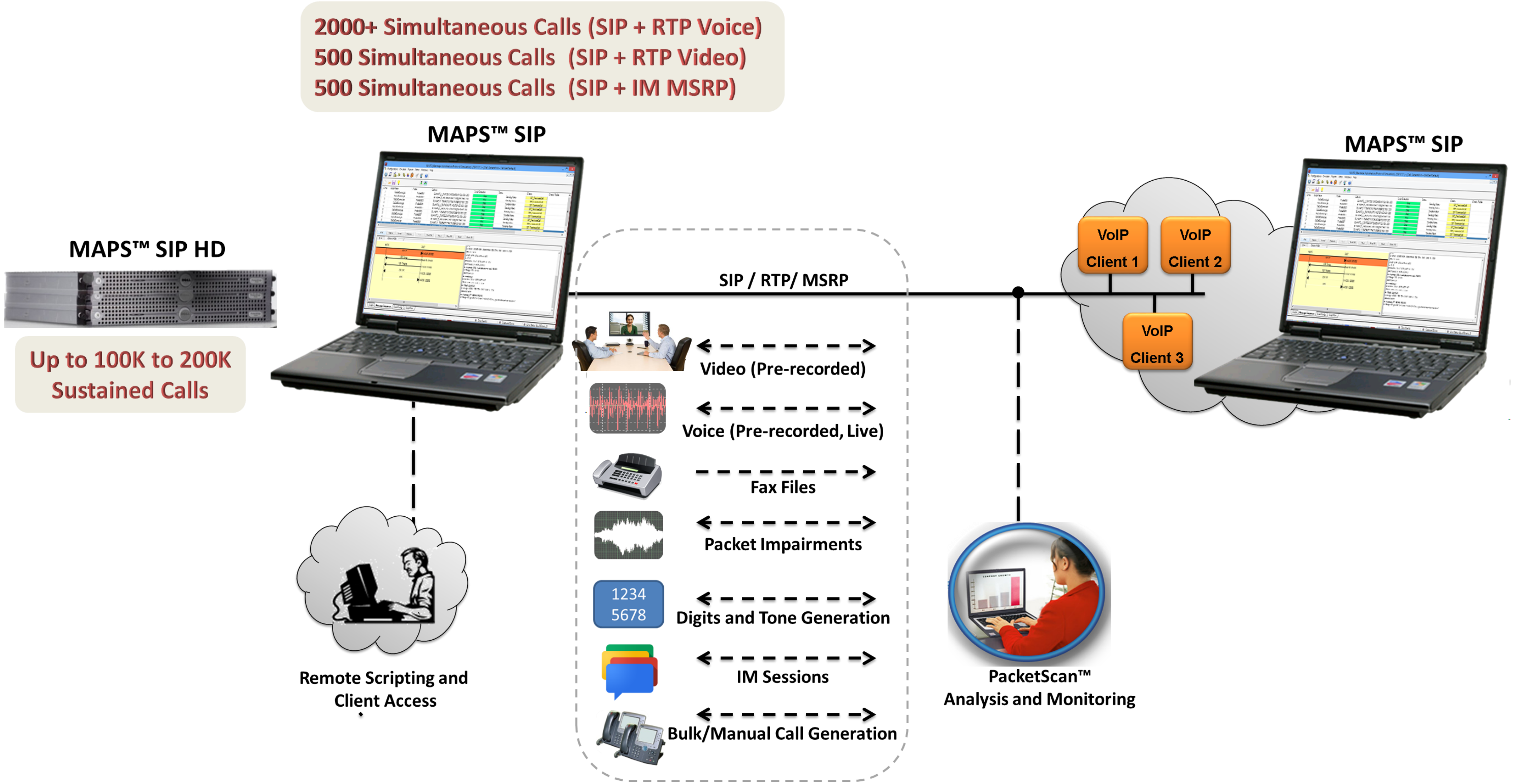


User Agents Configuration

- 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 in one shot



IP Traffic Simulation Capabilities and Performance



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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 @350 CPS	20000 @ 350 CPS	-	-

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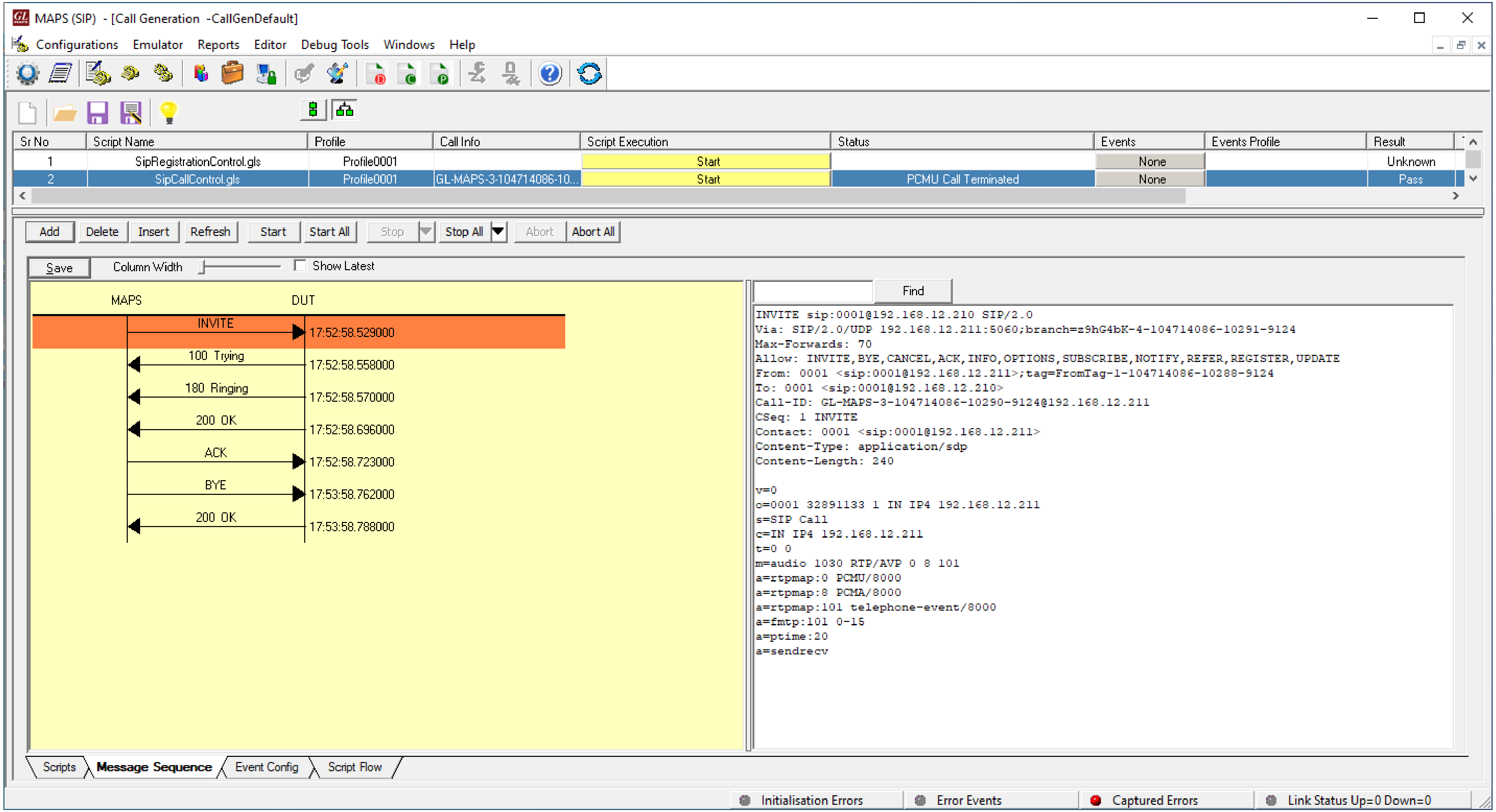
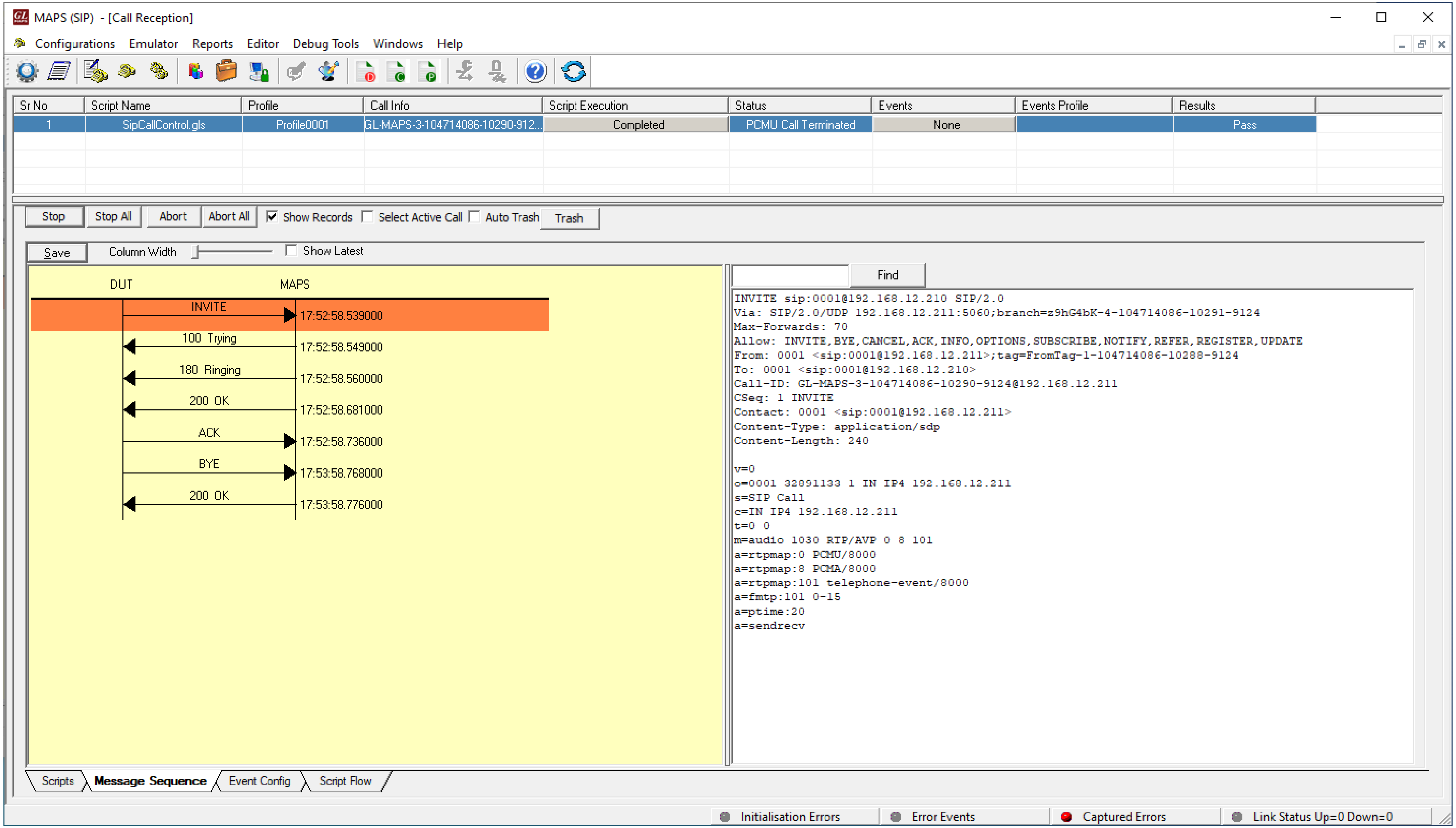
Call Generation with Voice Traffic

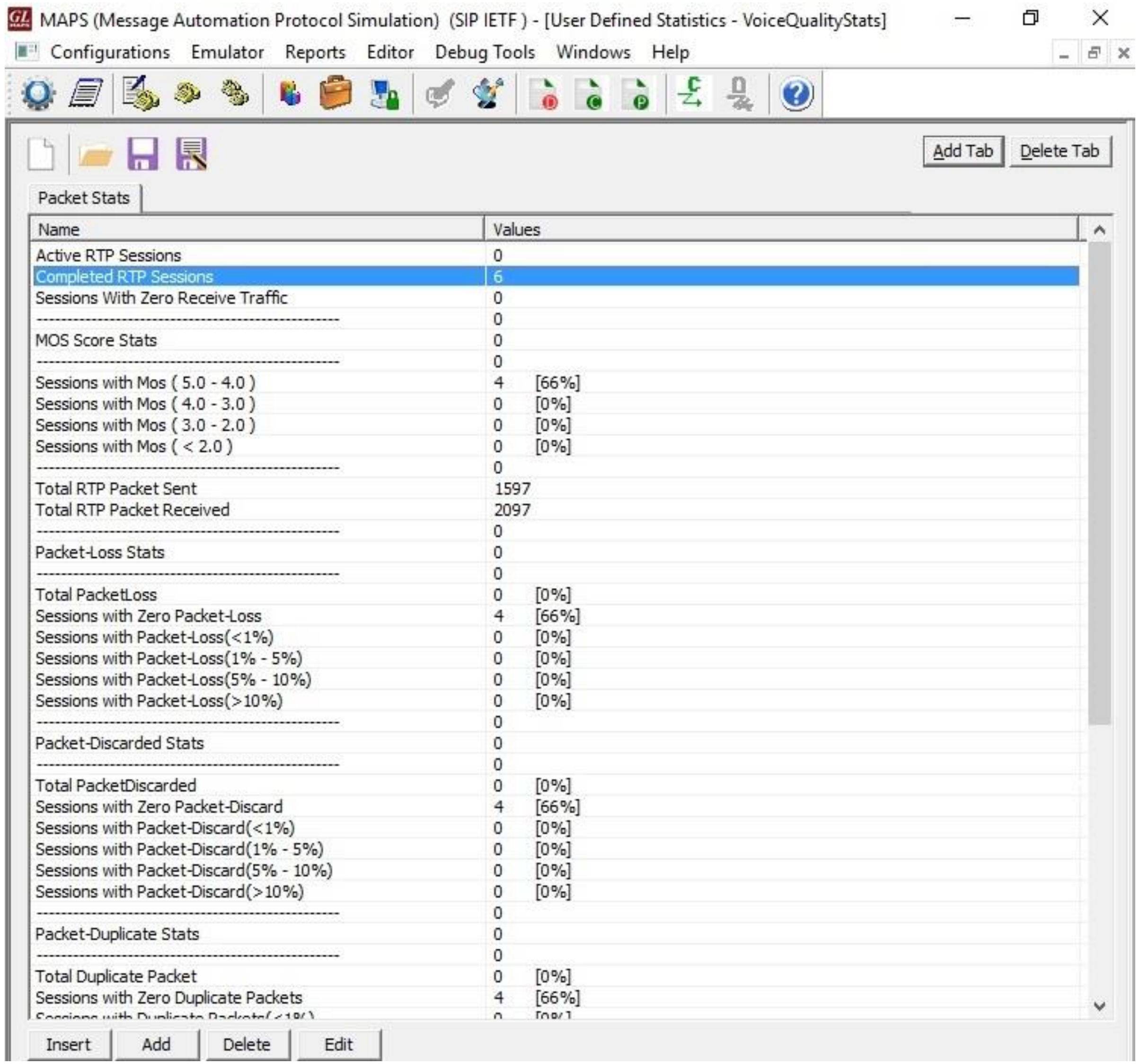


Call Generation with IVR Traffic

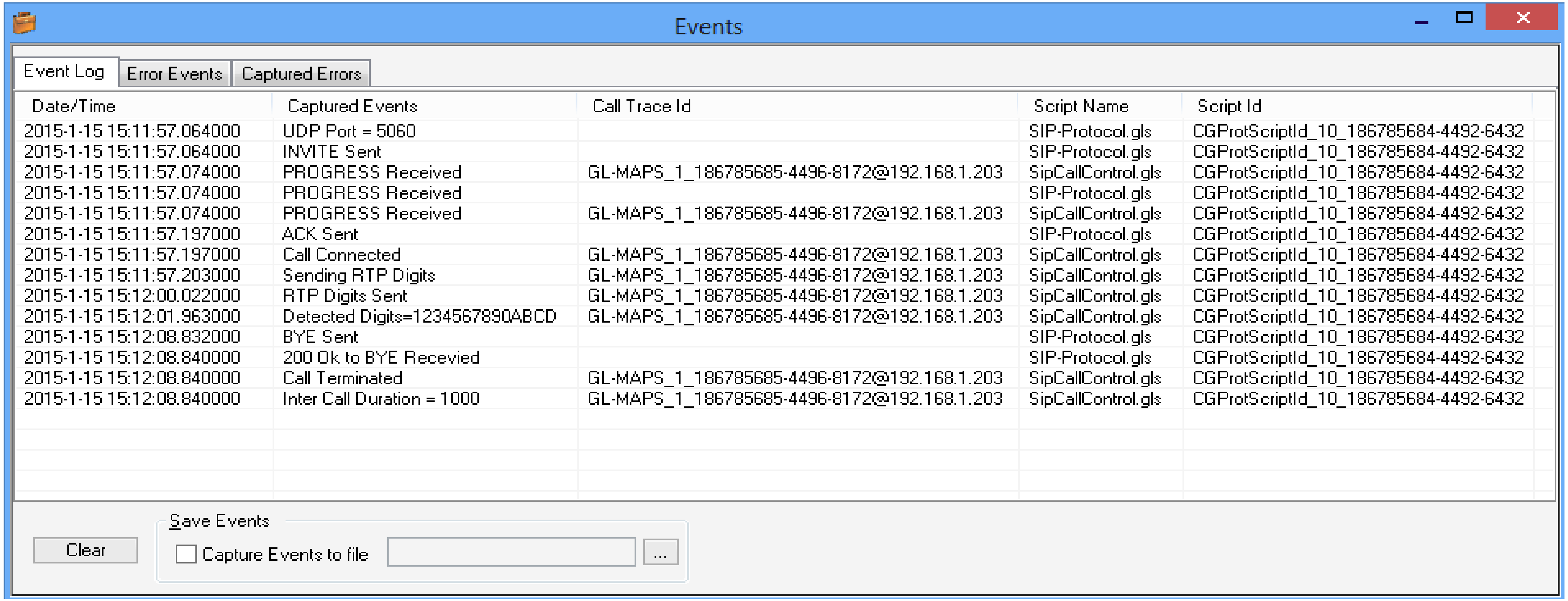


RTP Voice Quality Measurements

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



Event Log



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 Recevied		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

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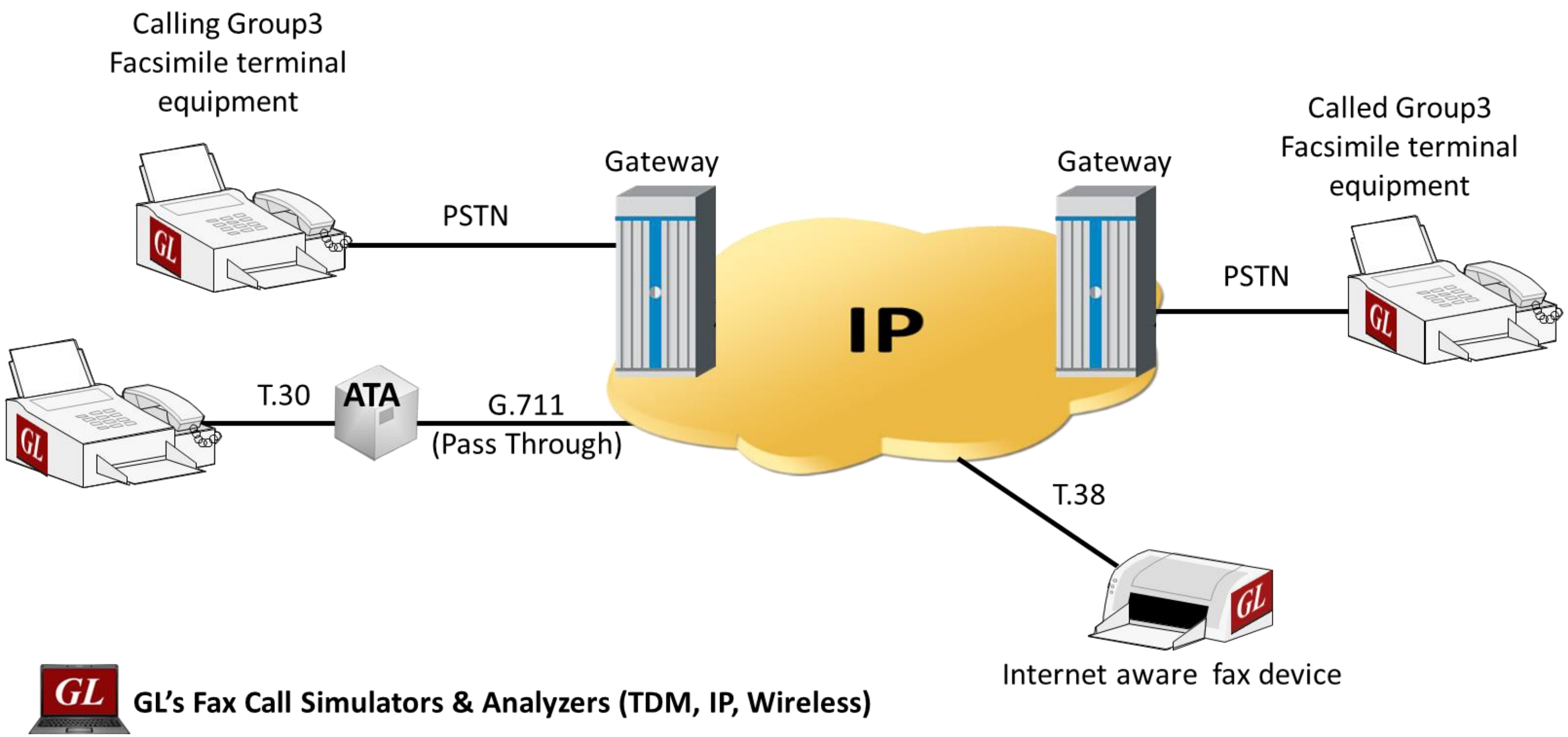
Fax Simulation over IP



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

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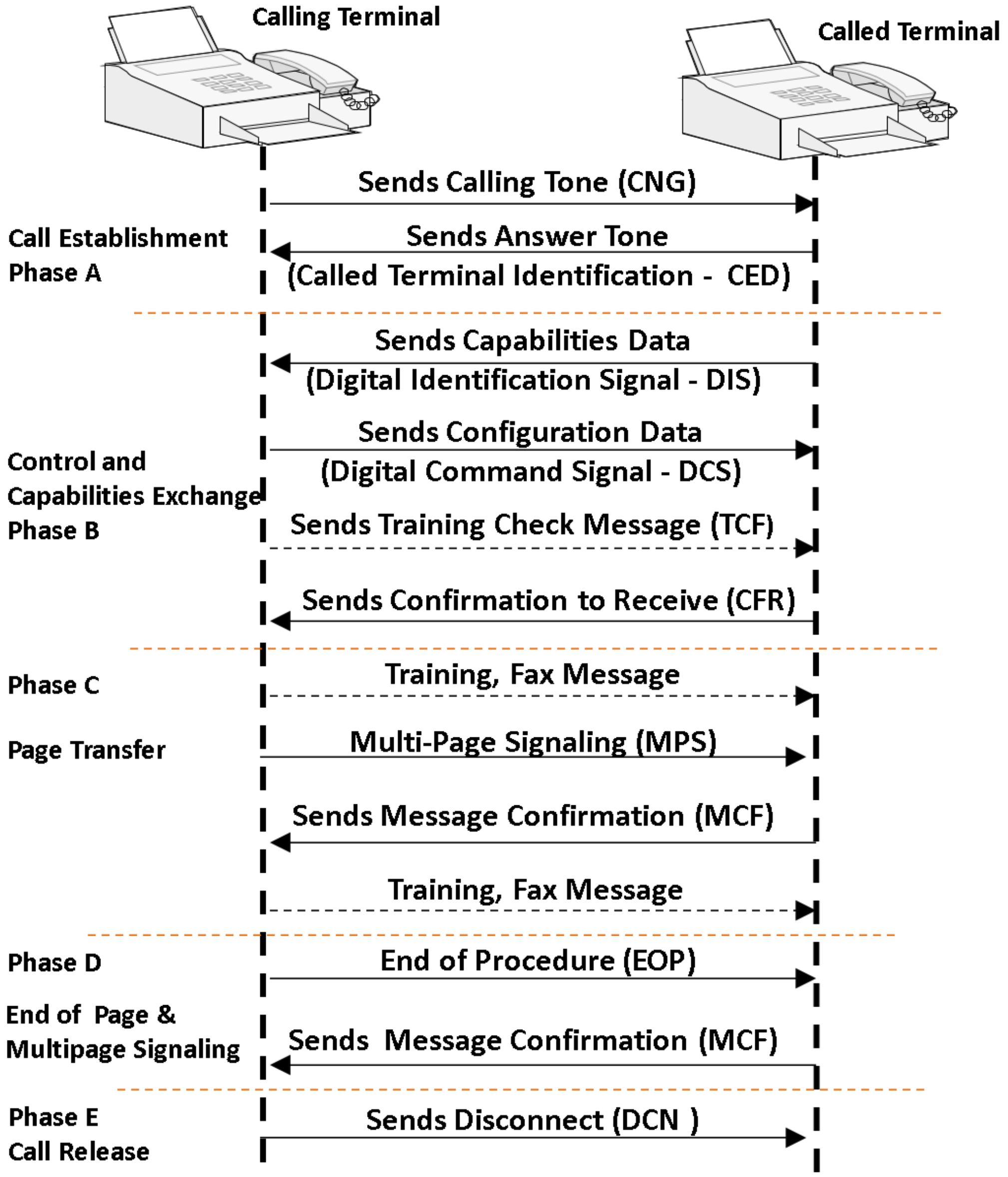
Call Scenarios - Fax T.30



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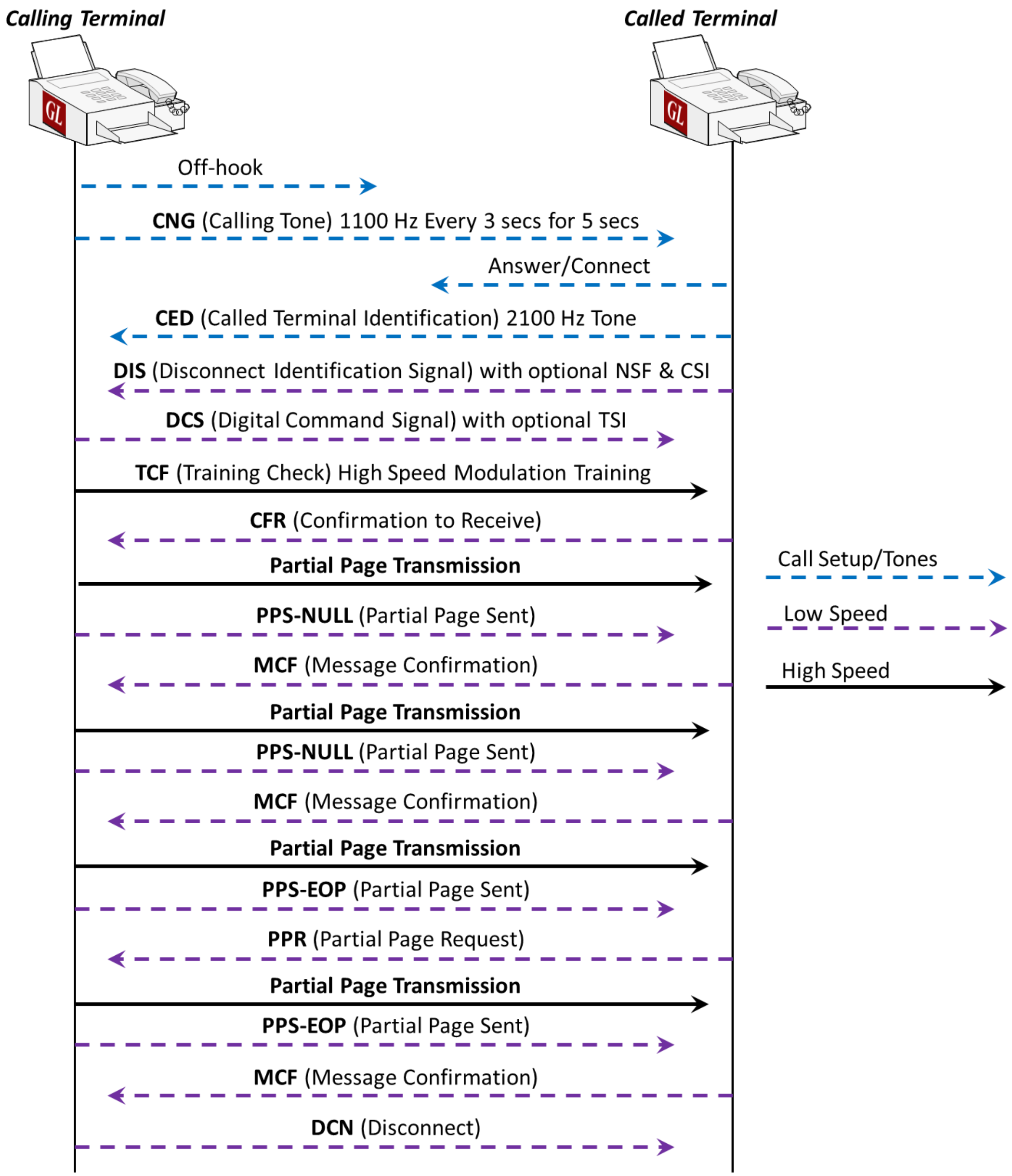
T.38 Fax Emulation over IP using MAPS™



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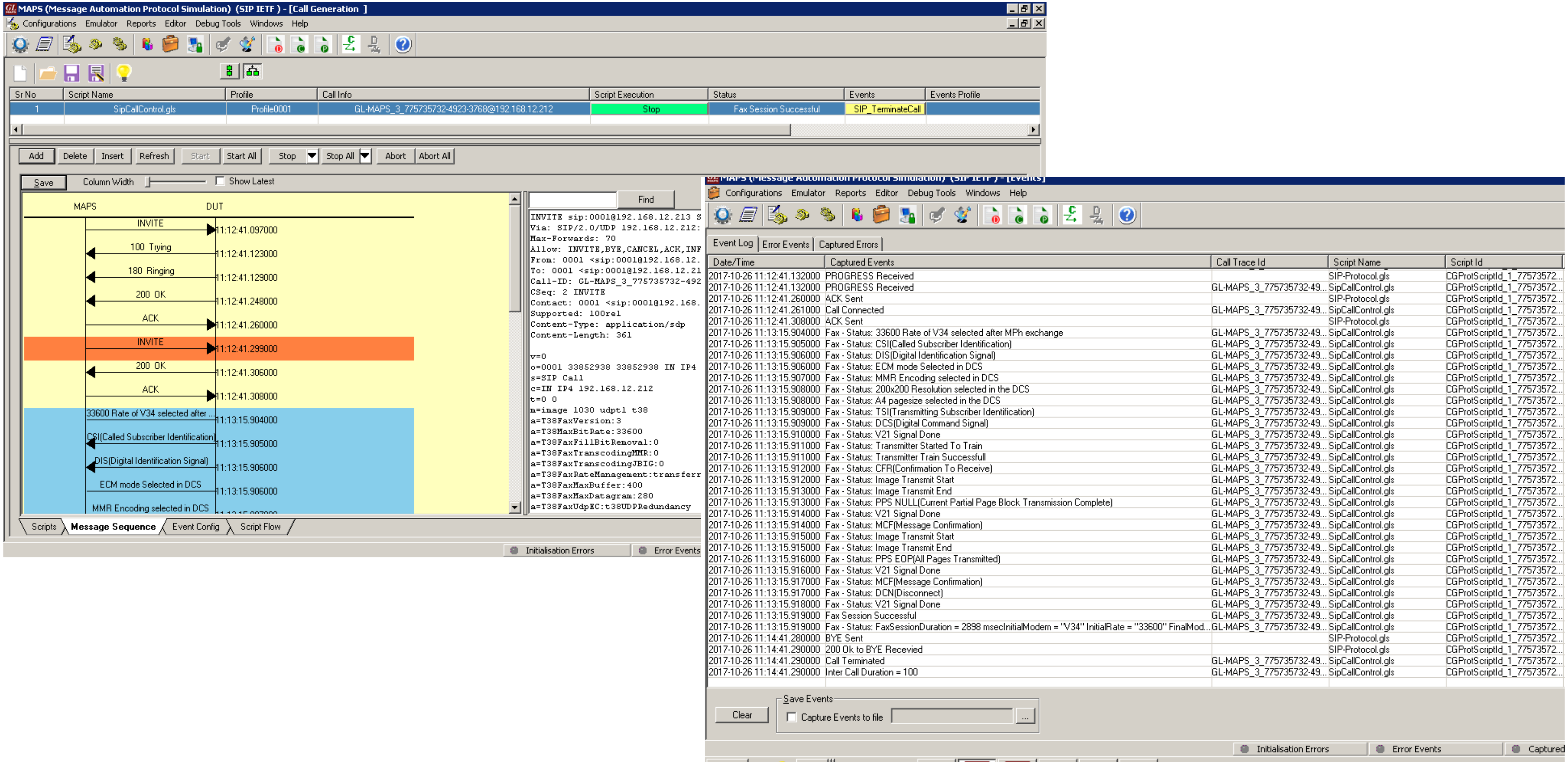
T.38 Fax Call in Progress and Related Events



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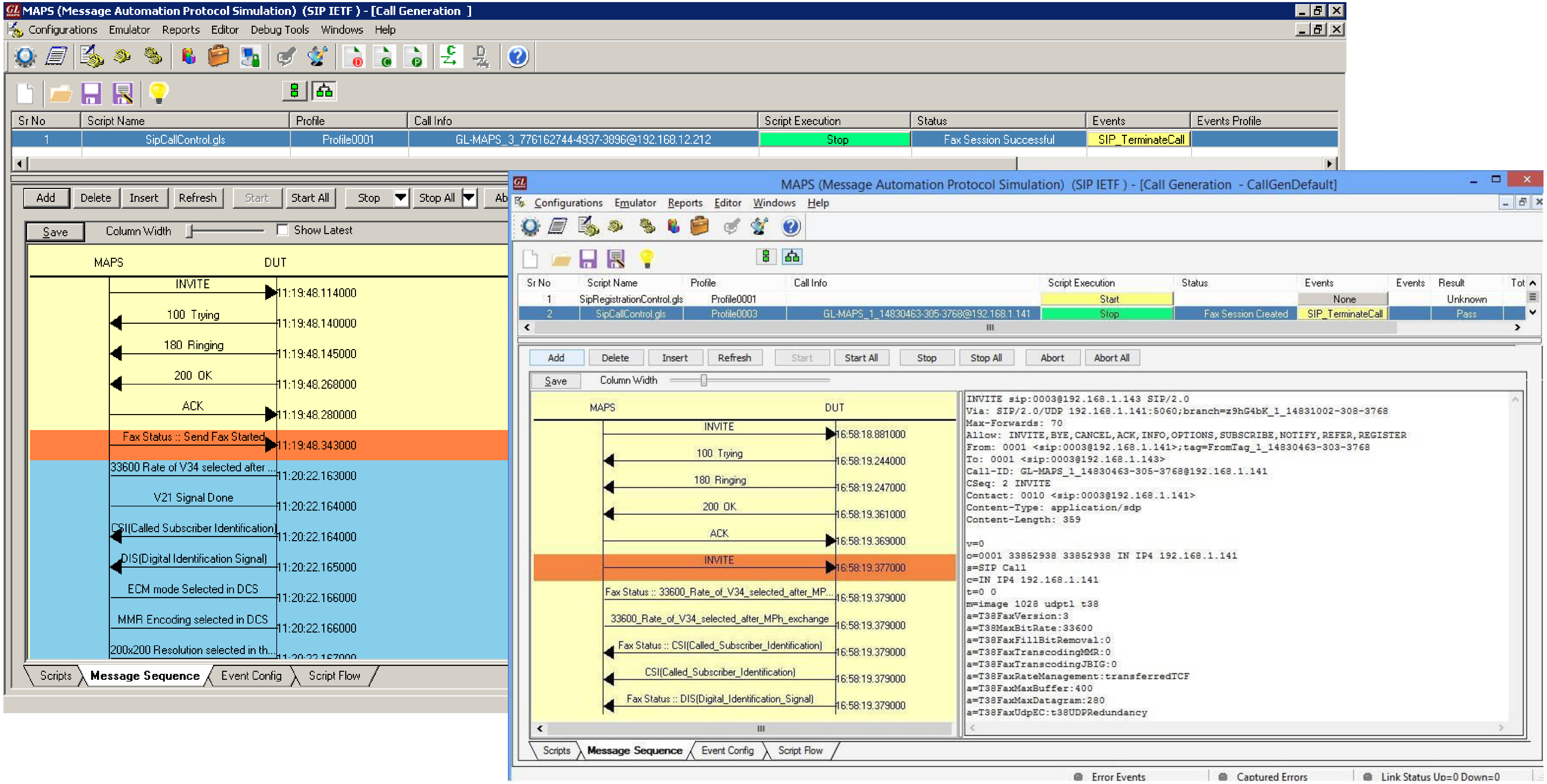
Call Generation with FAX Traffic



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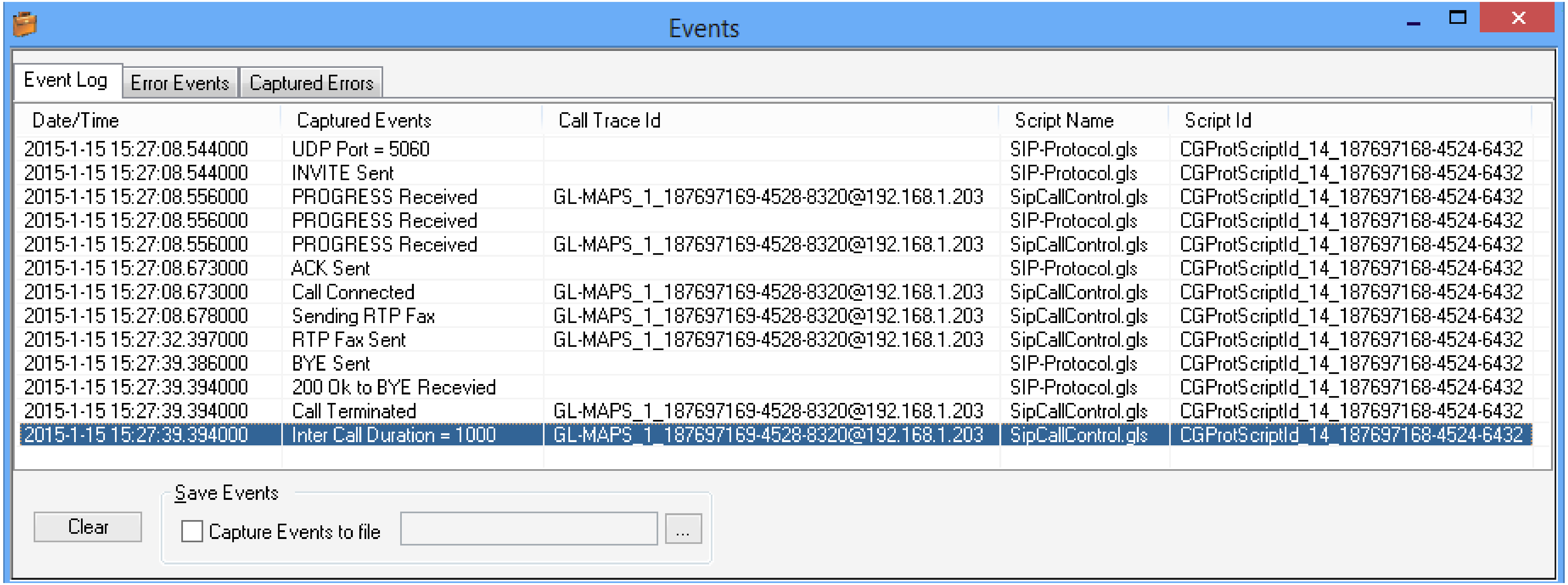
FAX Traffic Events



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 Recevied		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

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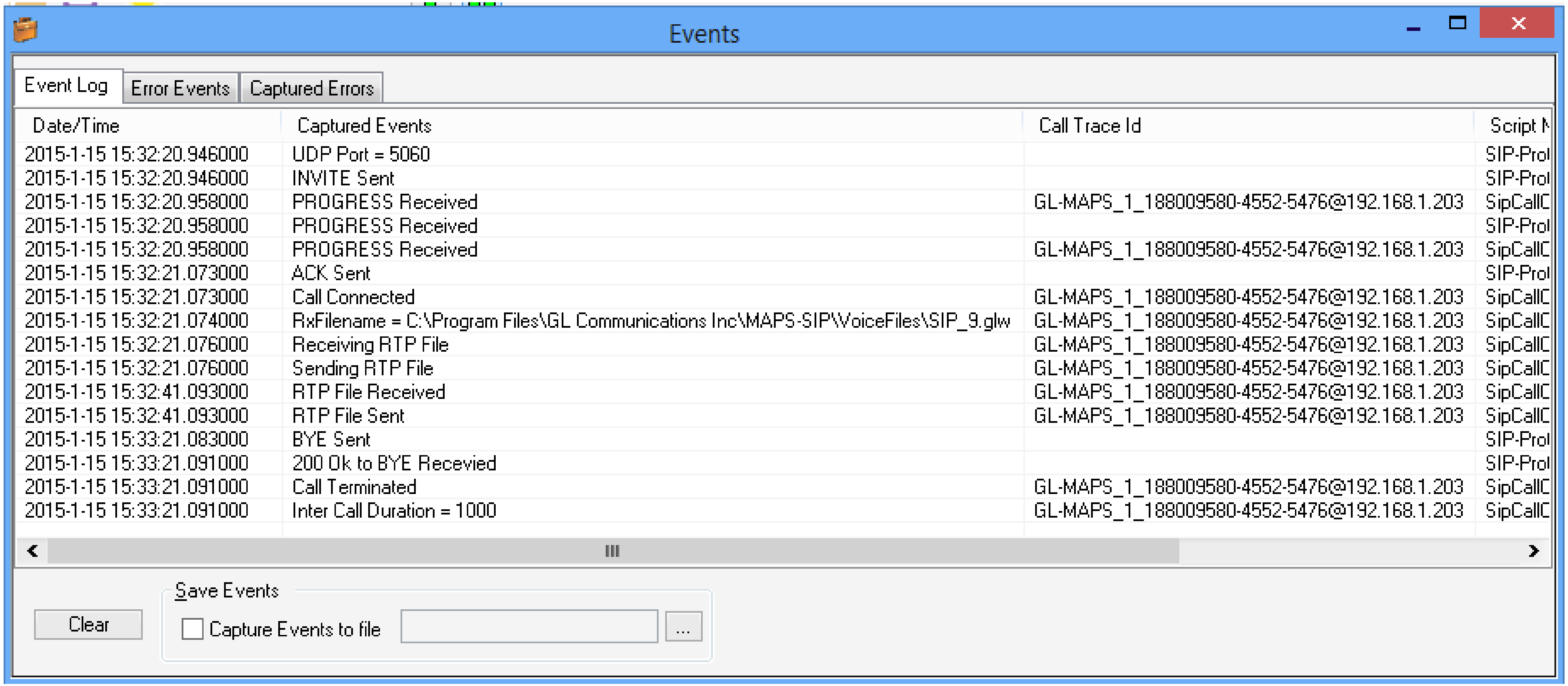
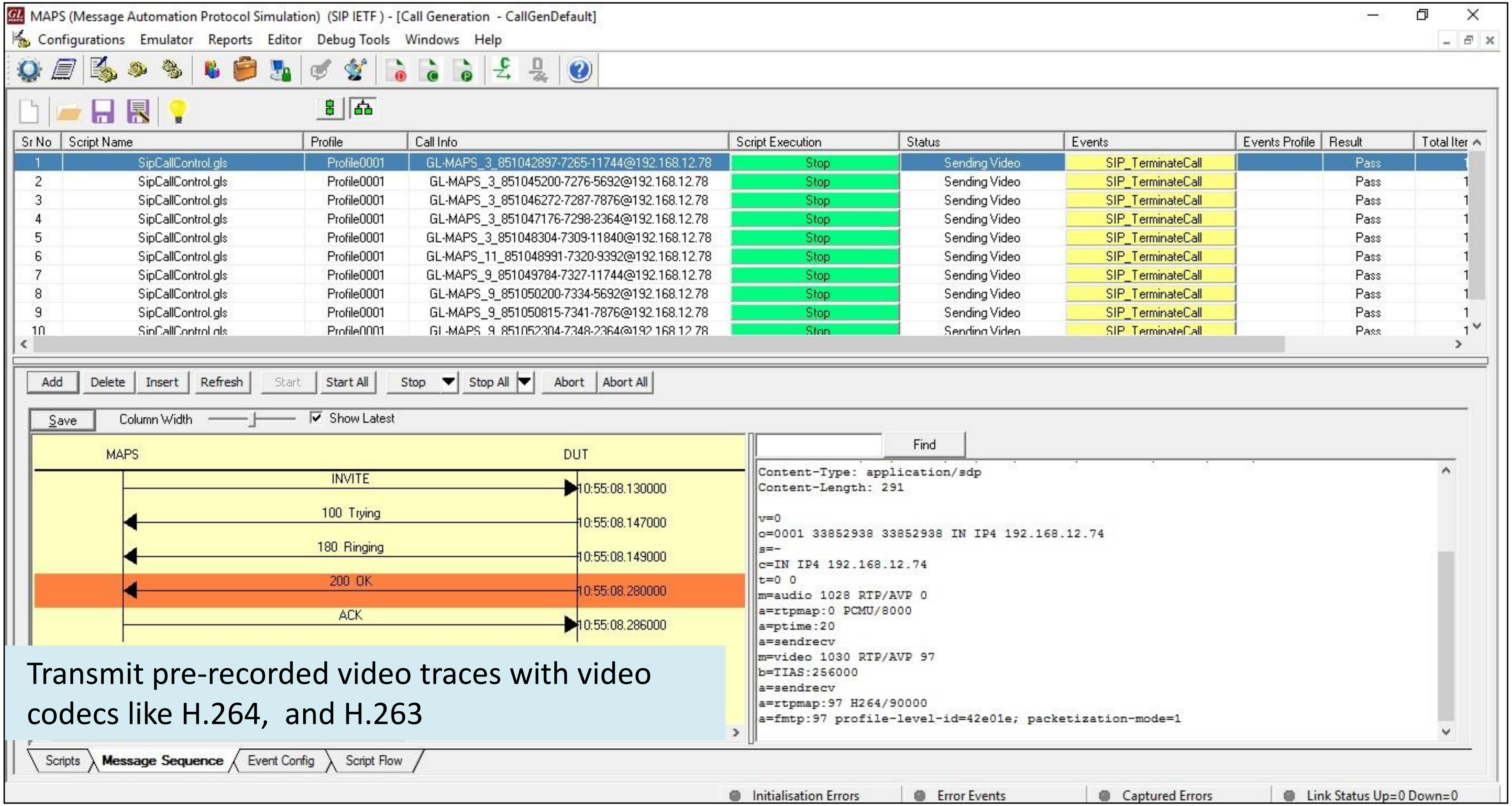
File Traffic Events



Video Call Generation



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

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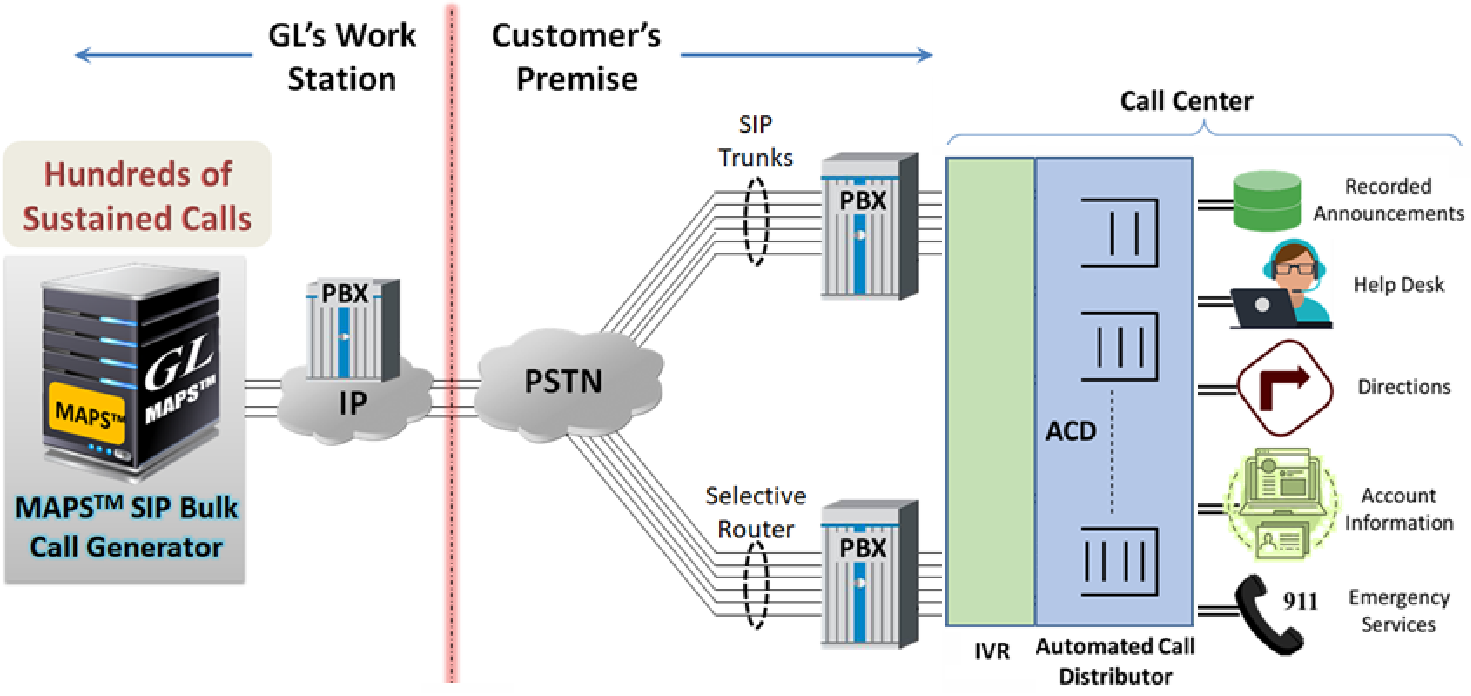
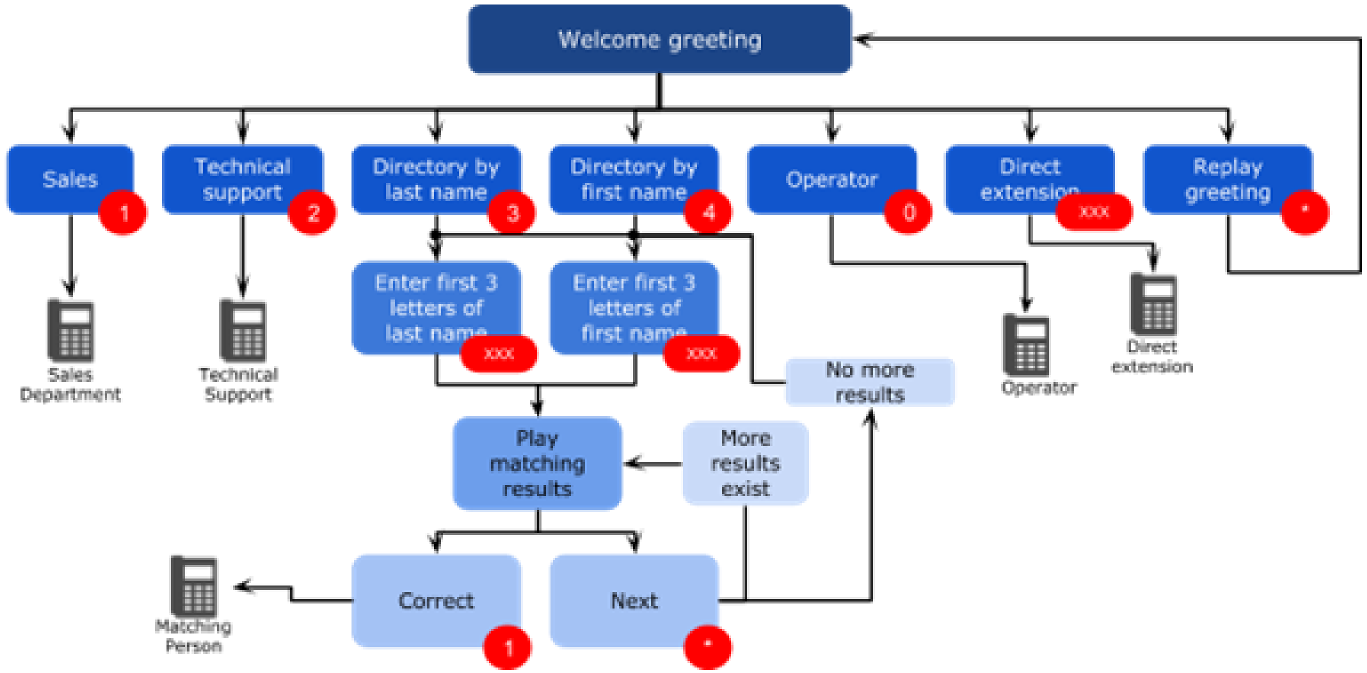
Speech to Text Interactive Voice Response (IVR)



- 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

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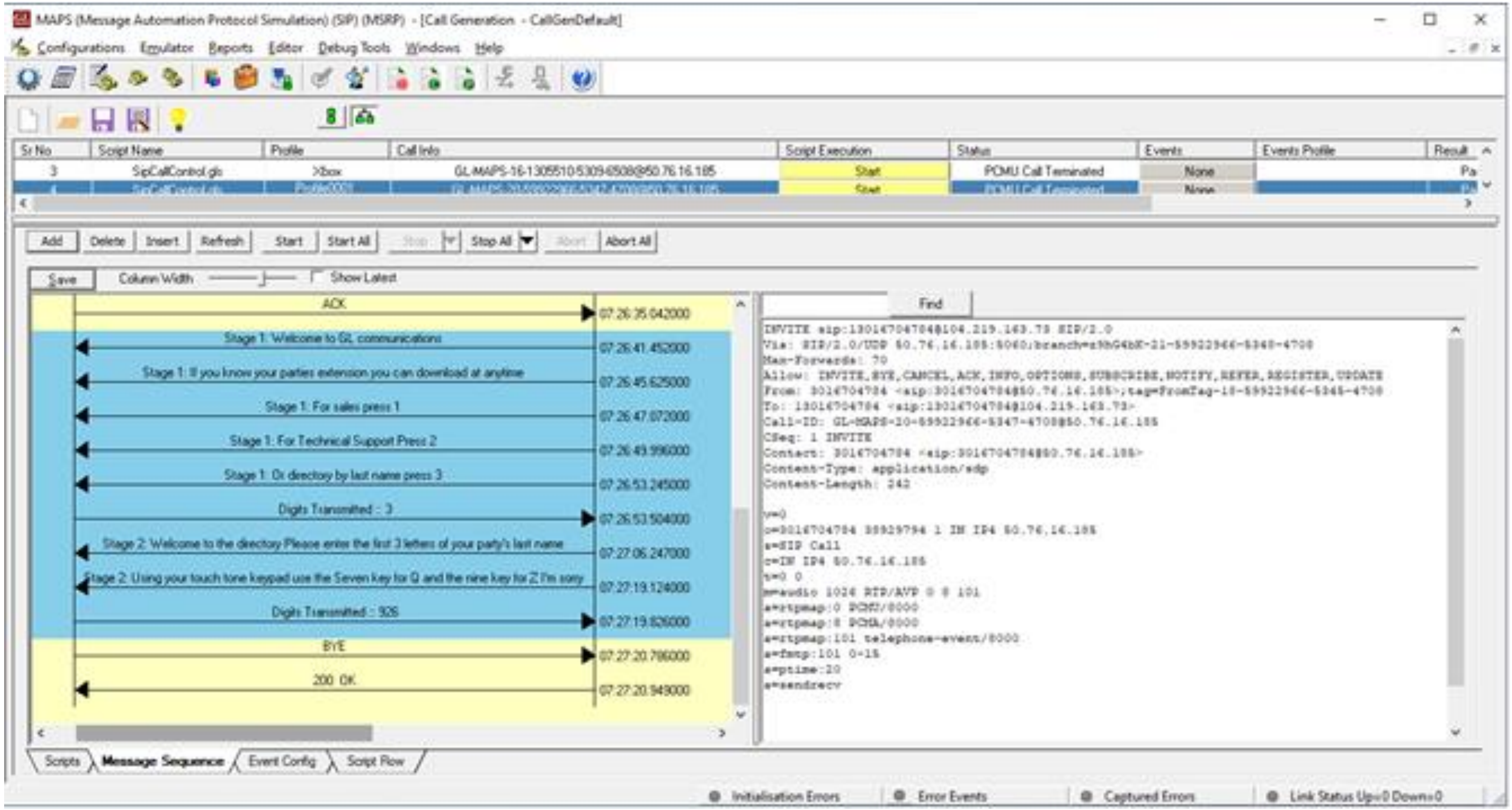
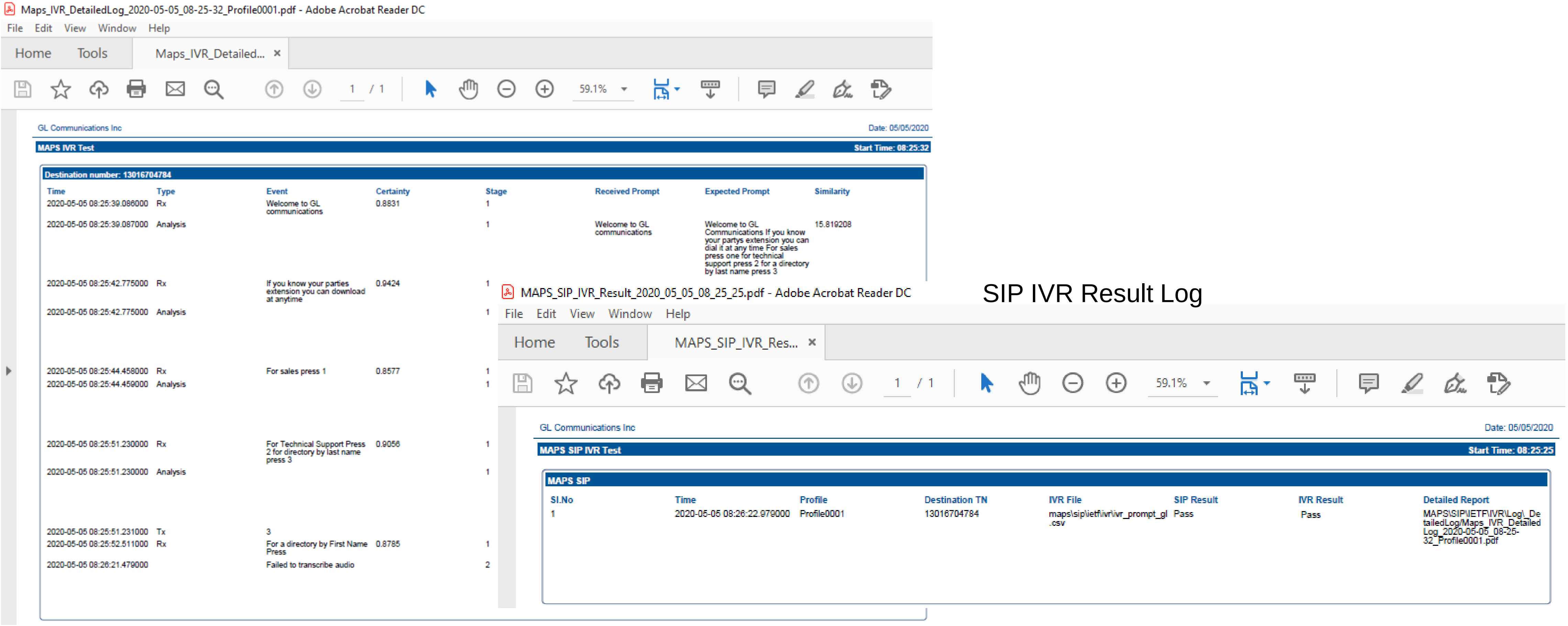
IVR Call Simulation



IVR Call Simulation Reports

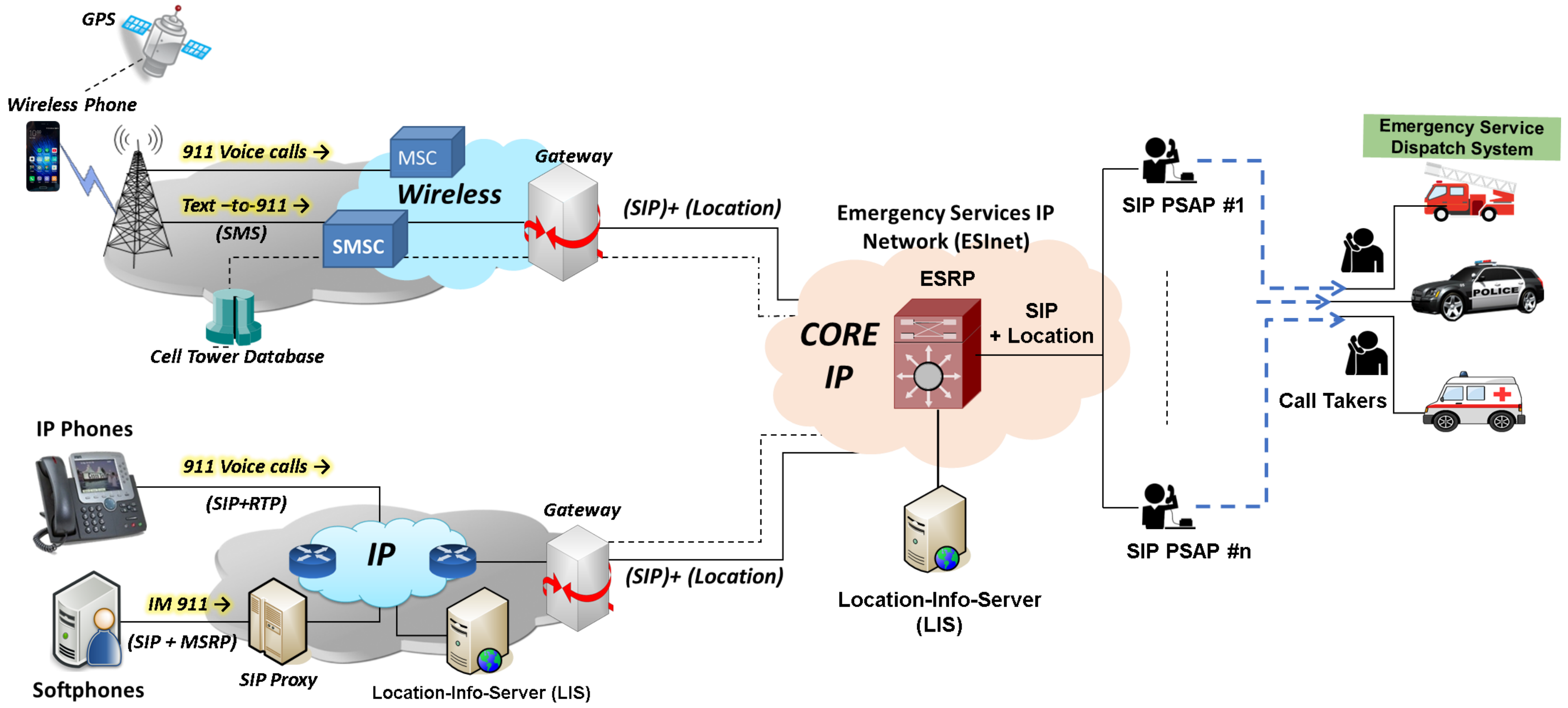
SIP IVR Detailed Log

SIP IVR Result Log



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



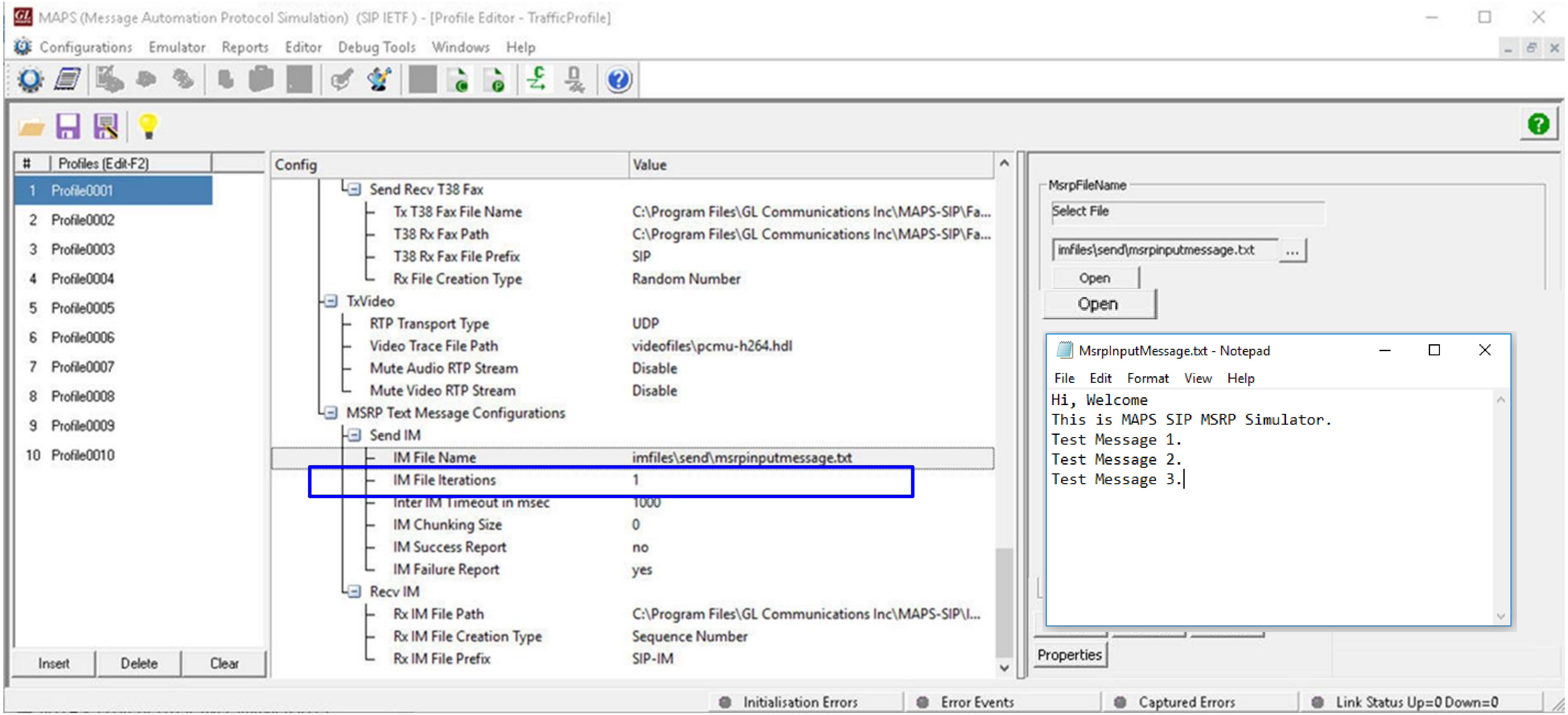
GL Communications

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

GL Communications

MSRP Traffic Configuration



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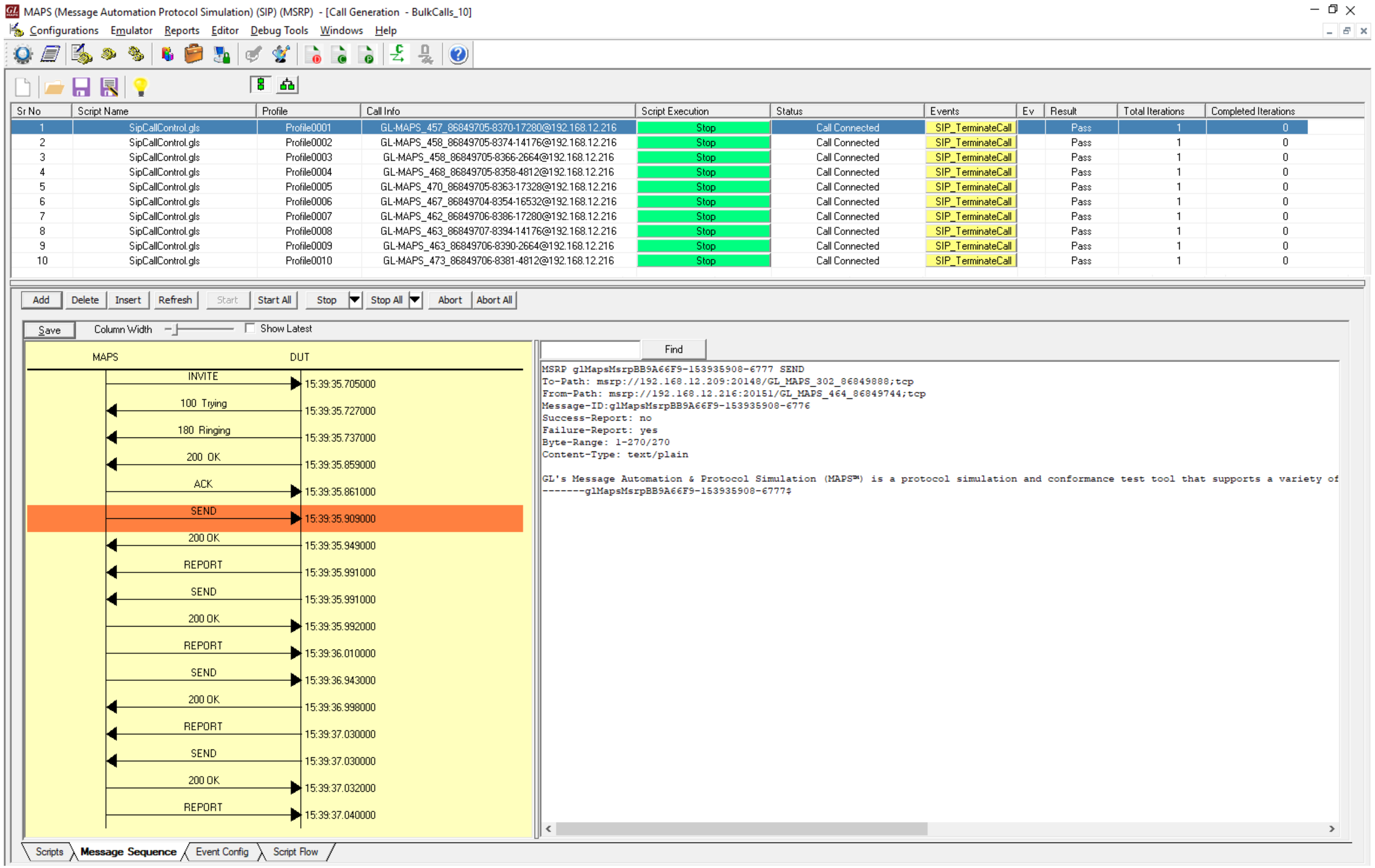
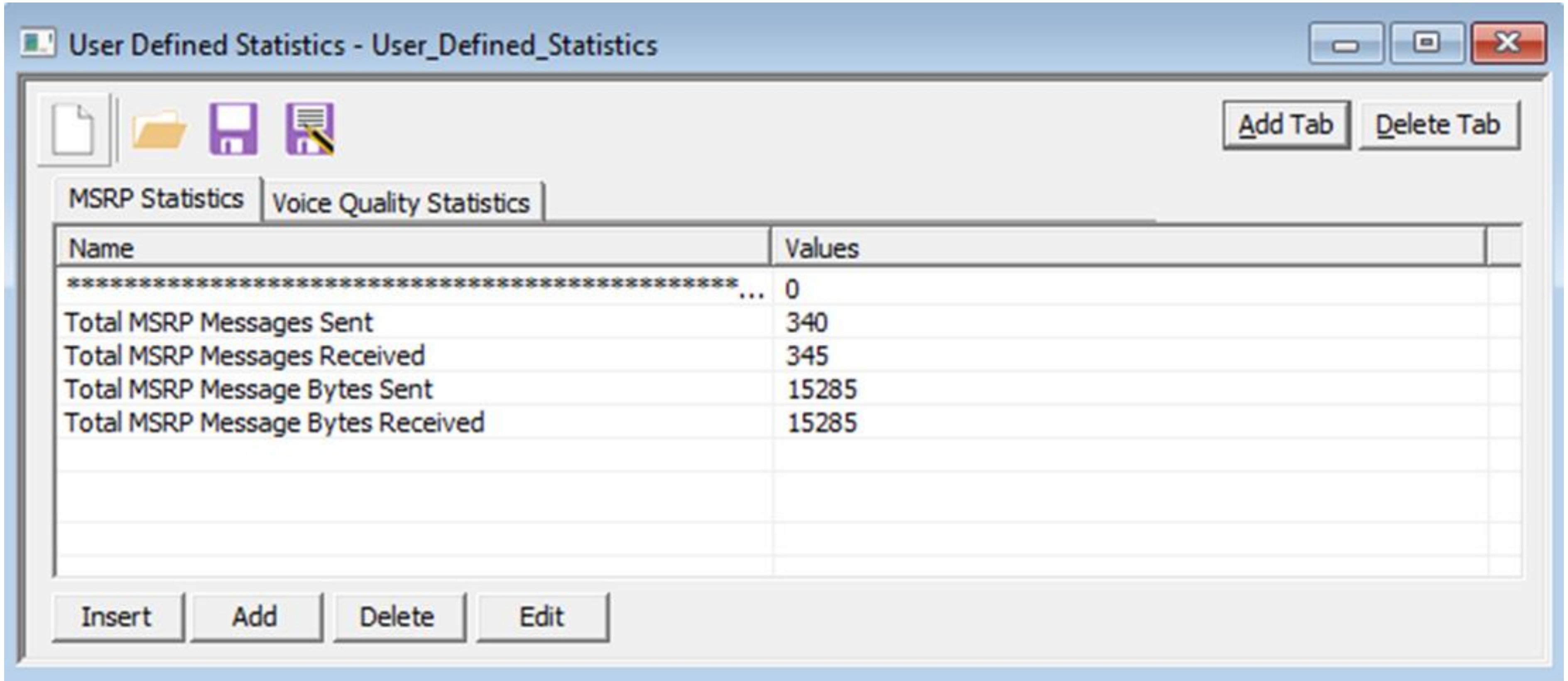
MSRP Call Generation

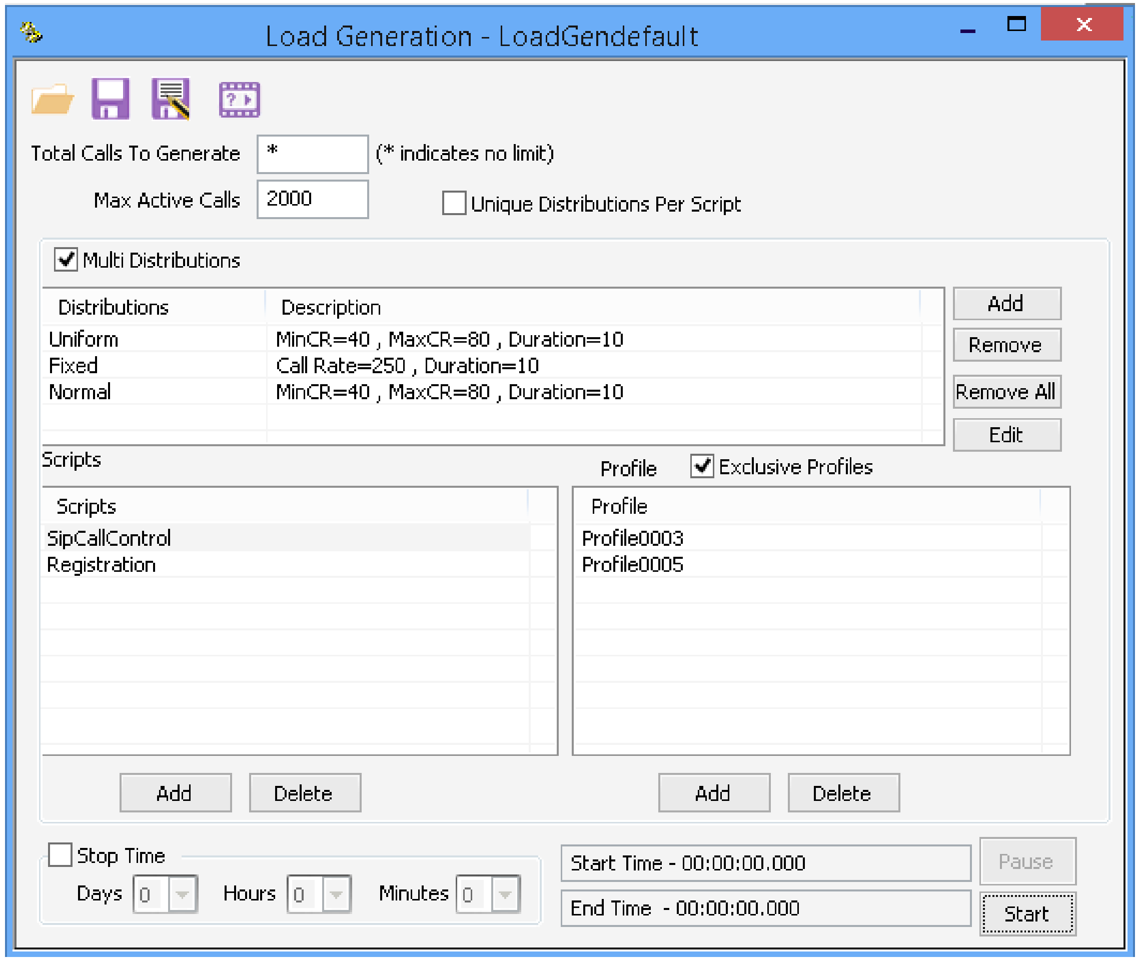


MSRP Statistics

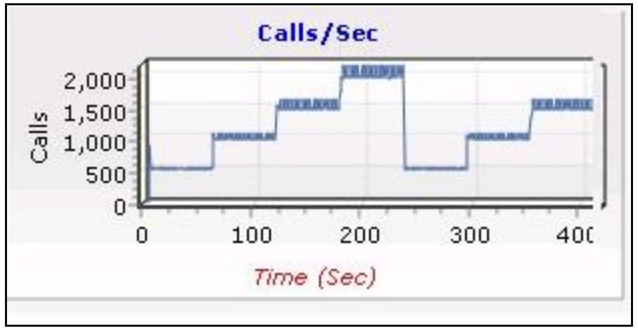


GL Communications

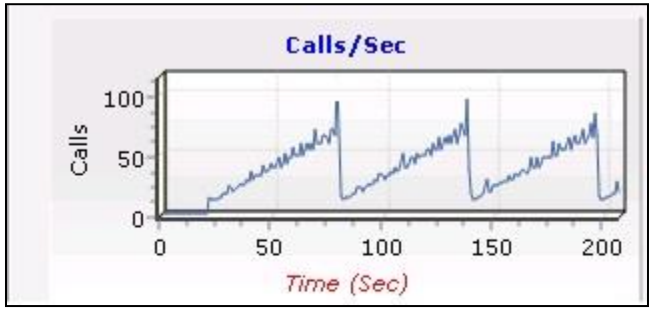
Load Generation



Step Statistical Distribution



Ramp Statistical Distribution



Saw-tooth Statistical Distribution

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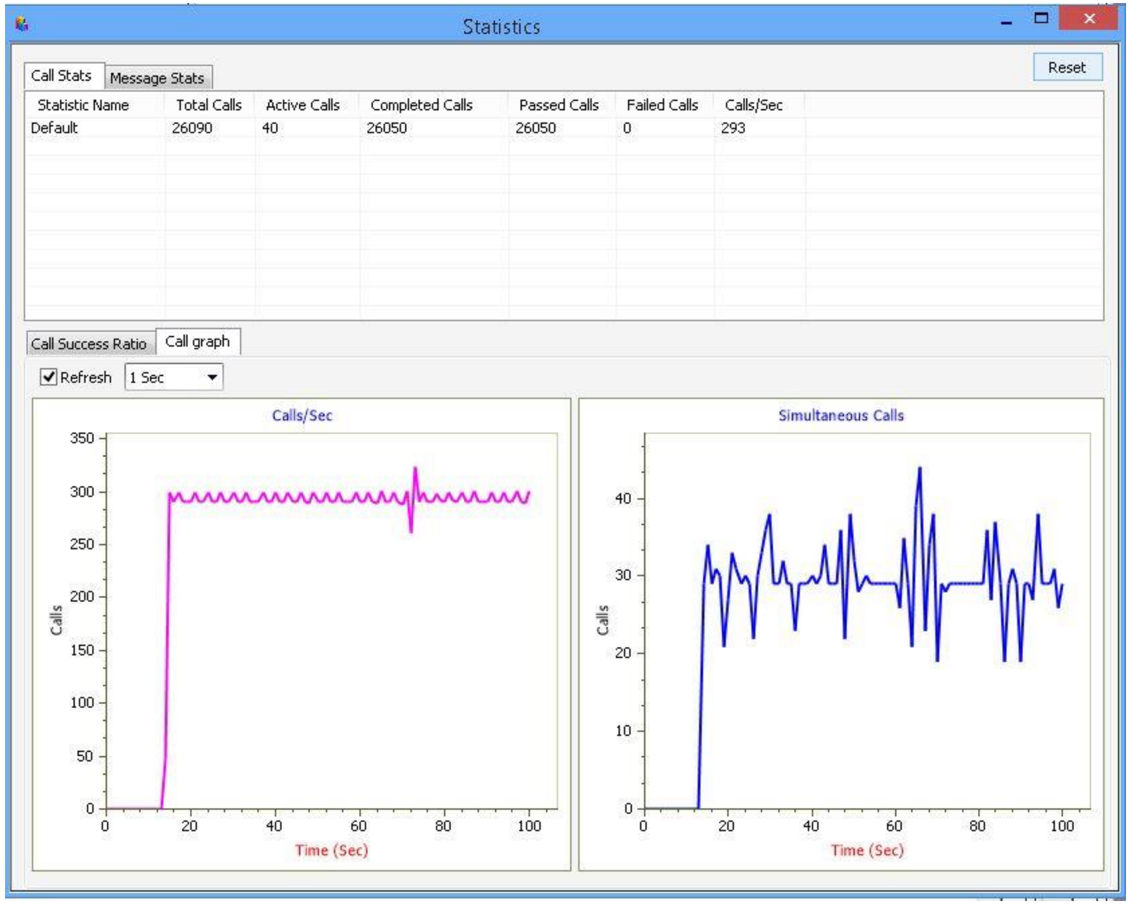
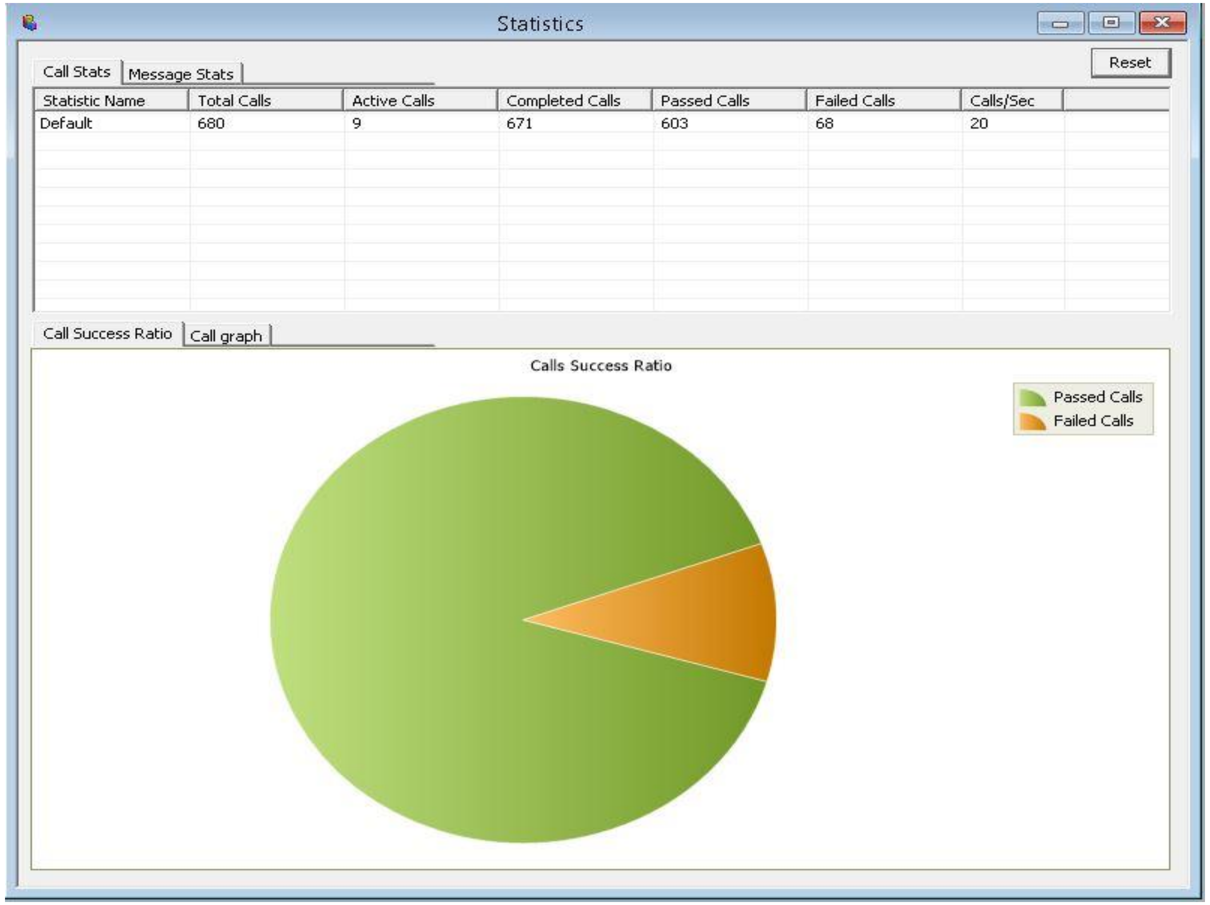
Success Call Ratio Statistics

Call Graph



Call Stats



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Message Statistics

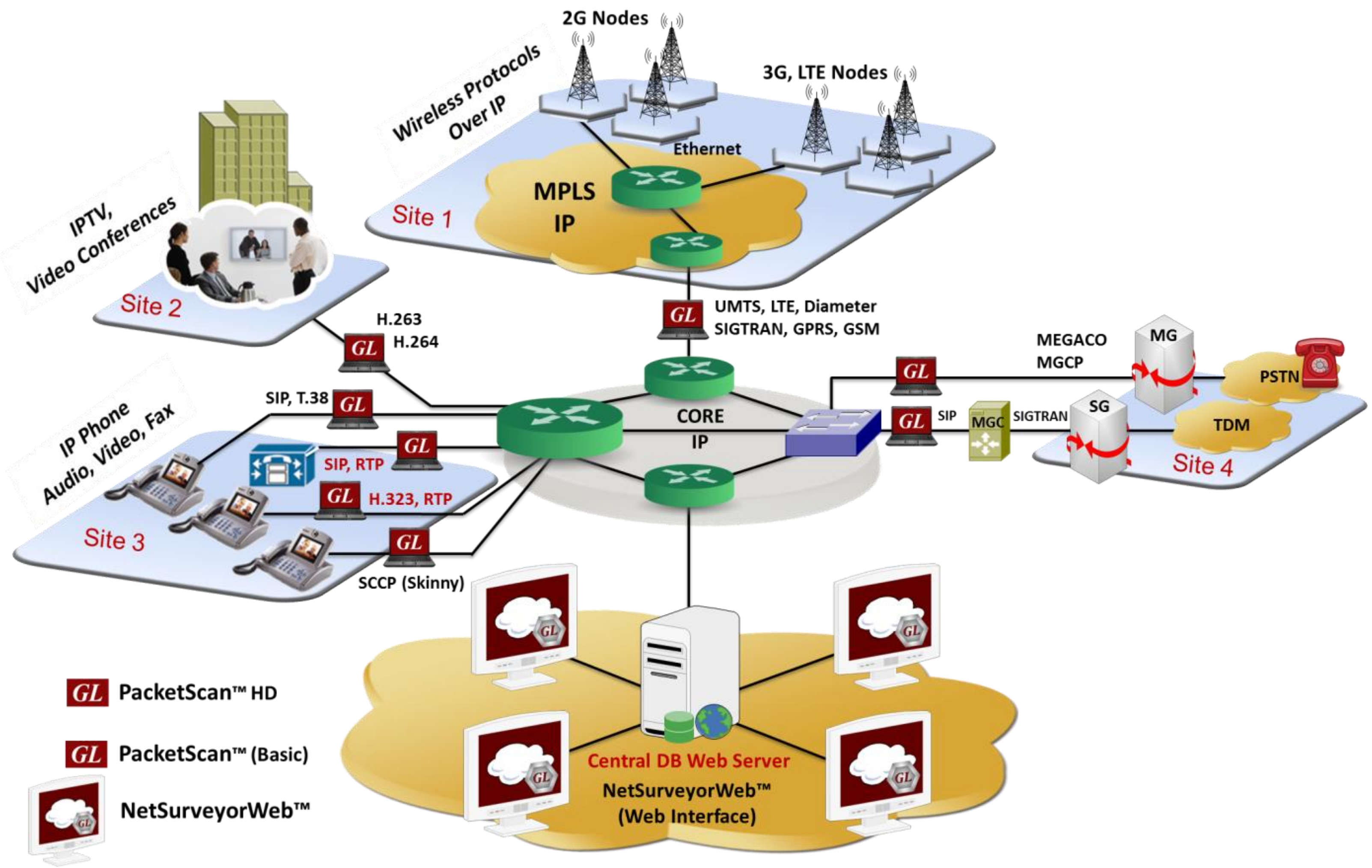
Statistics

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

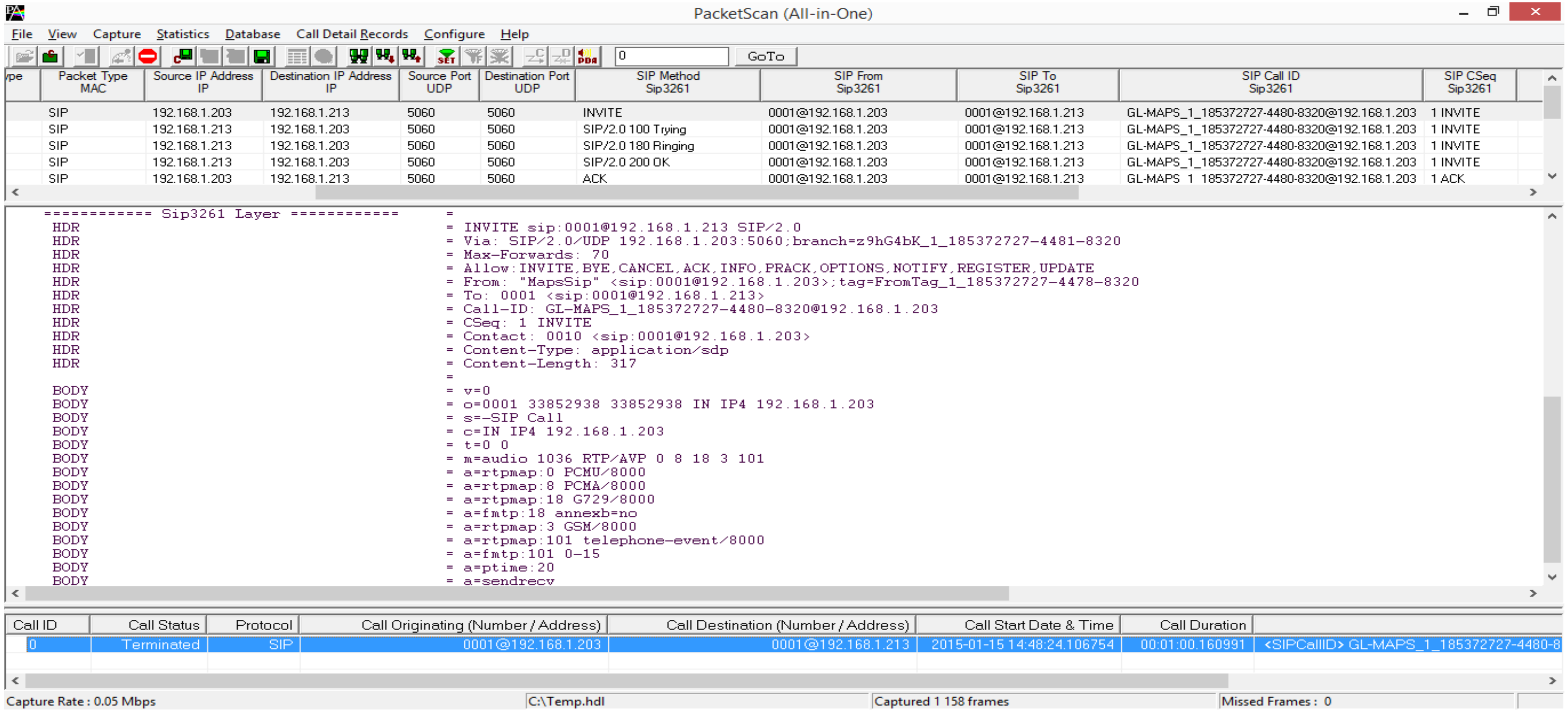


What the software does?

- 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



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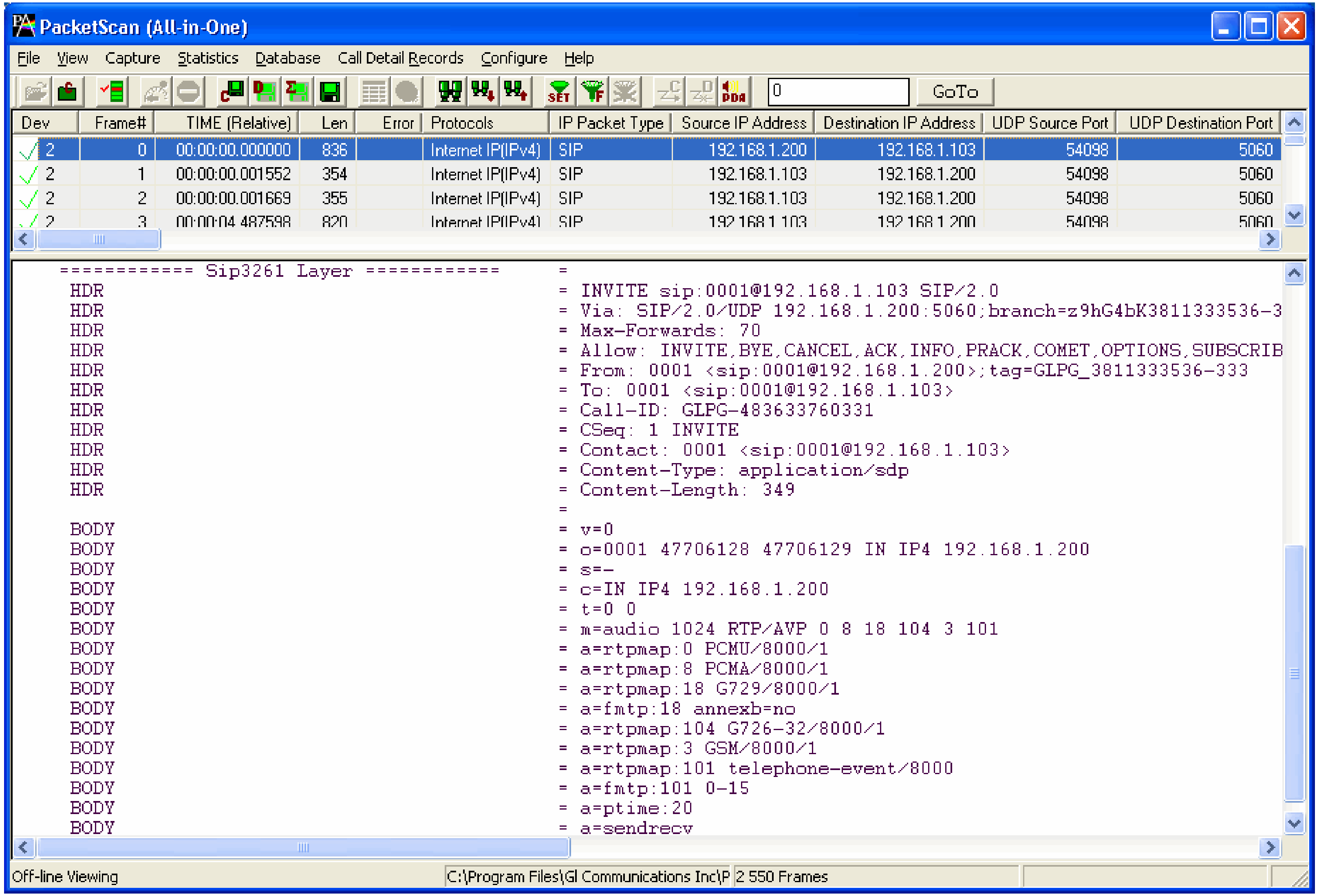
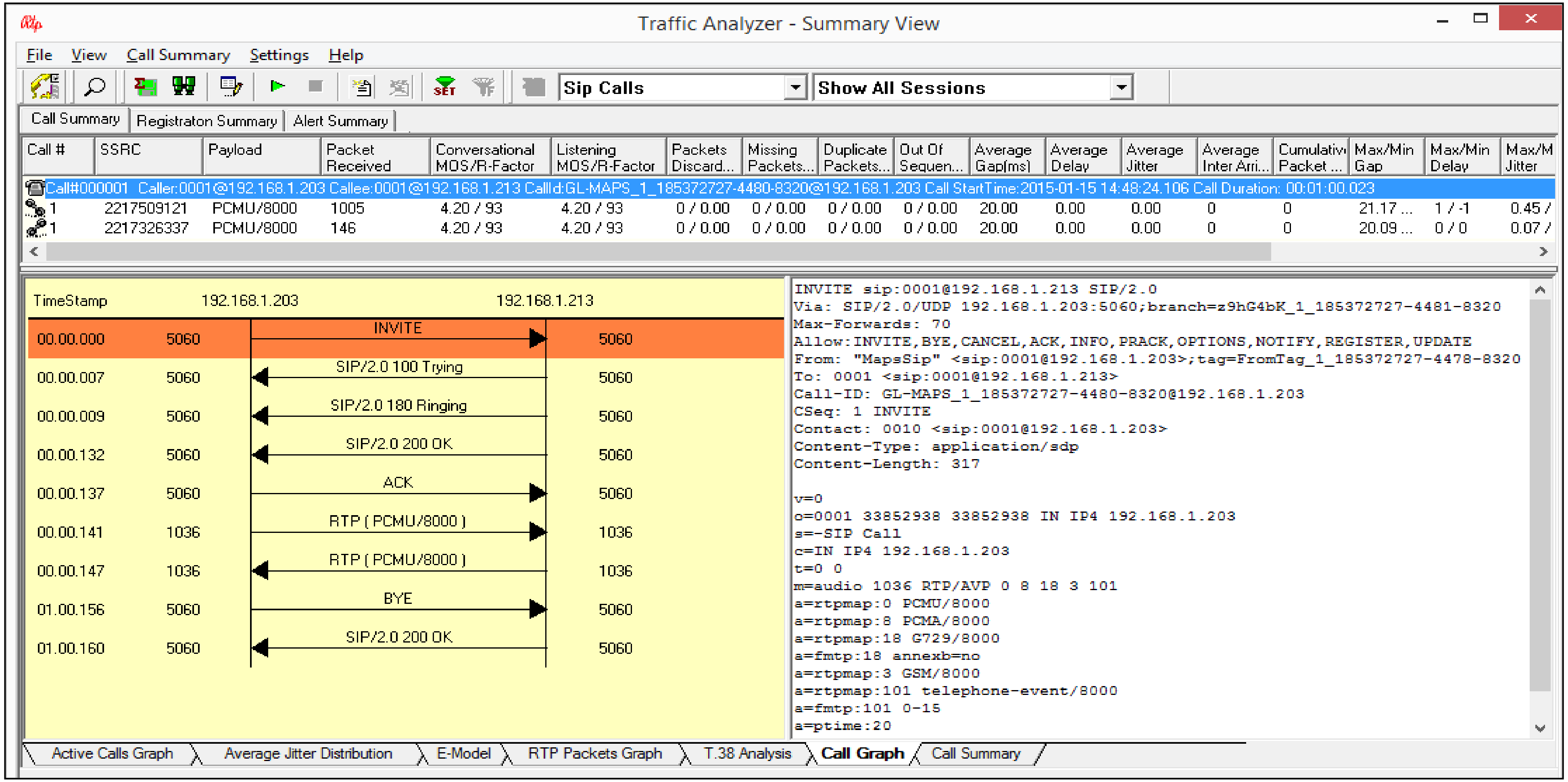
SIP Decode in PacketScan™

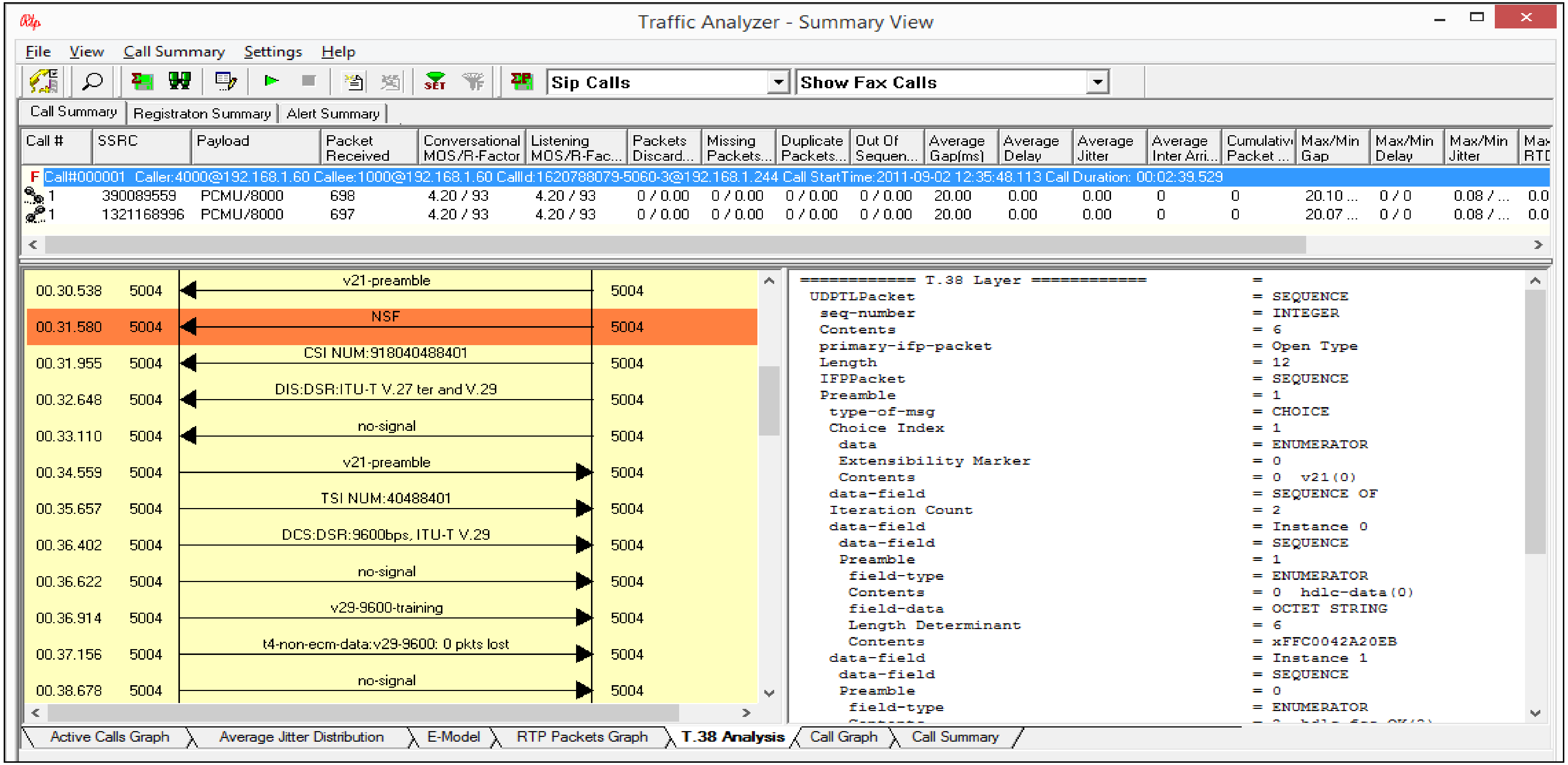


PacketScan™ PDA with SIP Call Summary

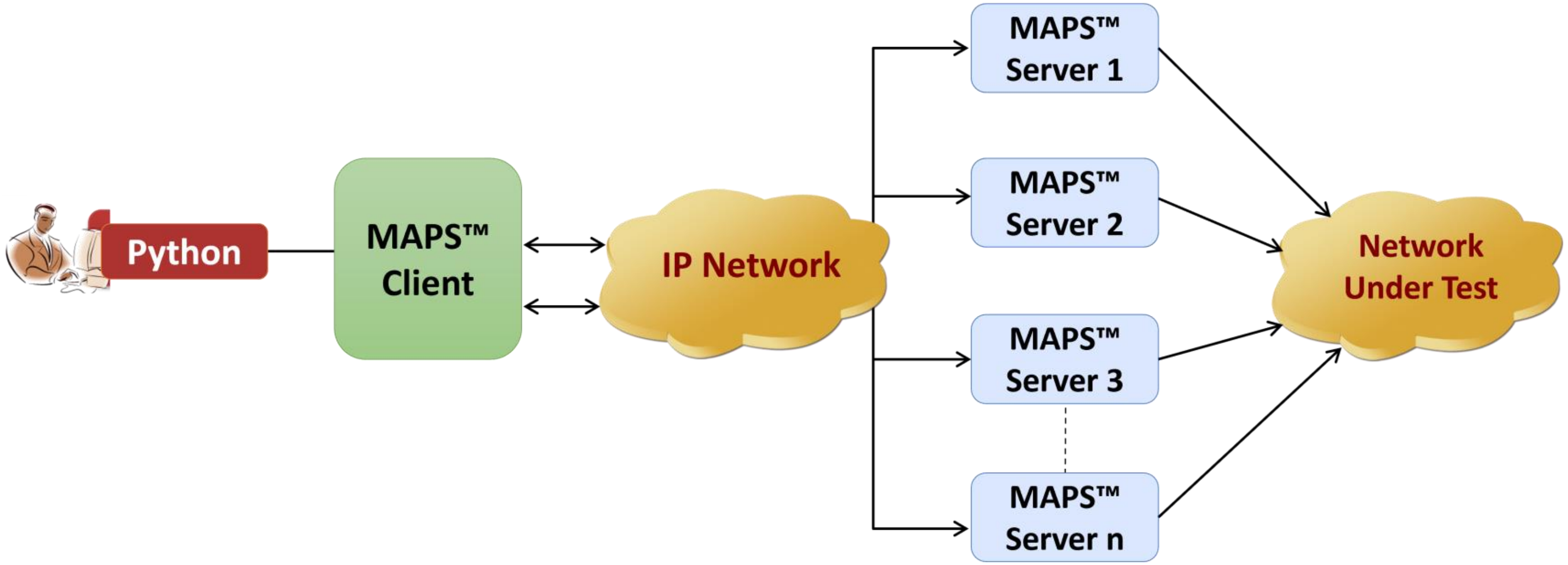


GL Communications

PacketScan™ Fax T.38 Analysis



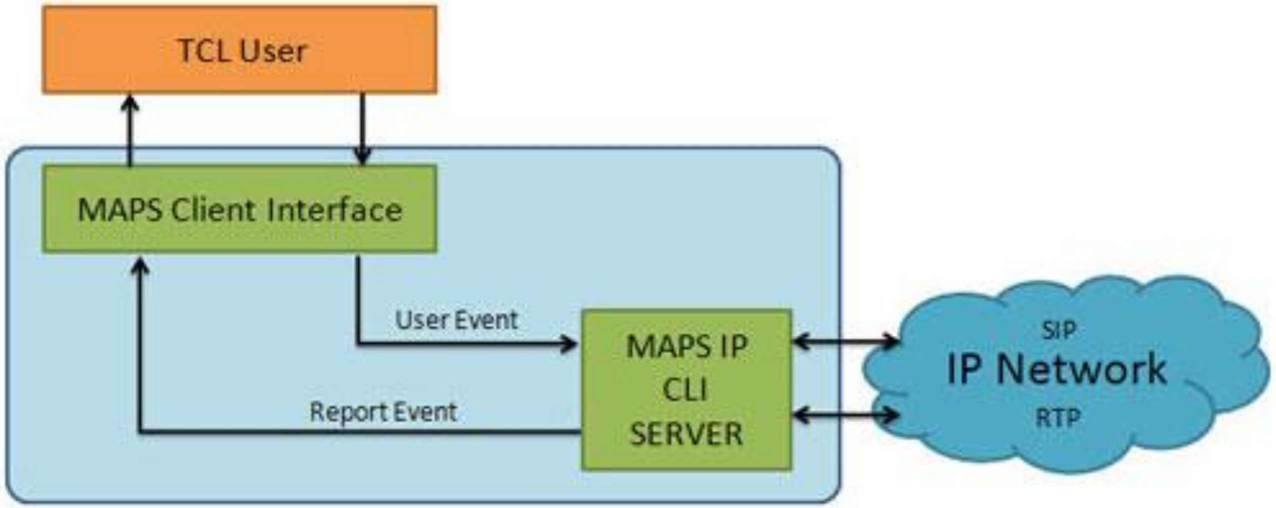
MAPS™ Command Line Interface



- MAPS™ can be configured as server-side application, to enable remote controlling through multiple command-line based clients. Supported clients include Java, VBScripts, TCL, Python and others
- The MAPS™ APIs allows for programmatic and automated control over all MAPS™ platforms. Each MAPS™ server can receive multiple client connections and offer independent execution to each client
- Likewise, a single client can connect to multiple MAPS™ servers, including servers running different protocols, permitting complex cross-protocol test cases

GL Communications

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

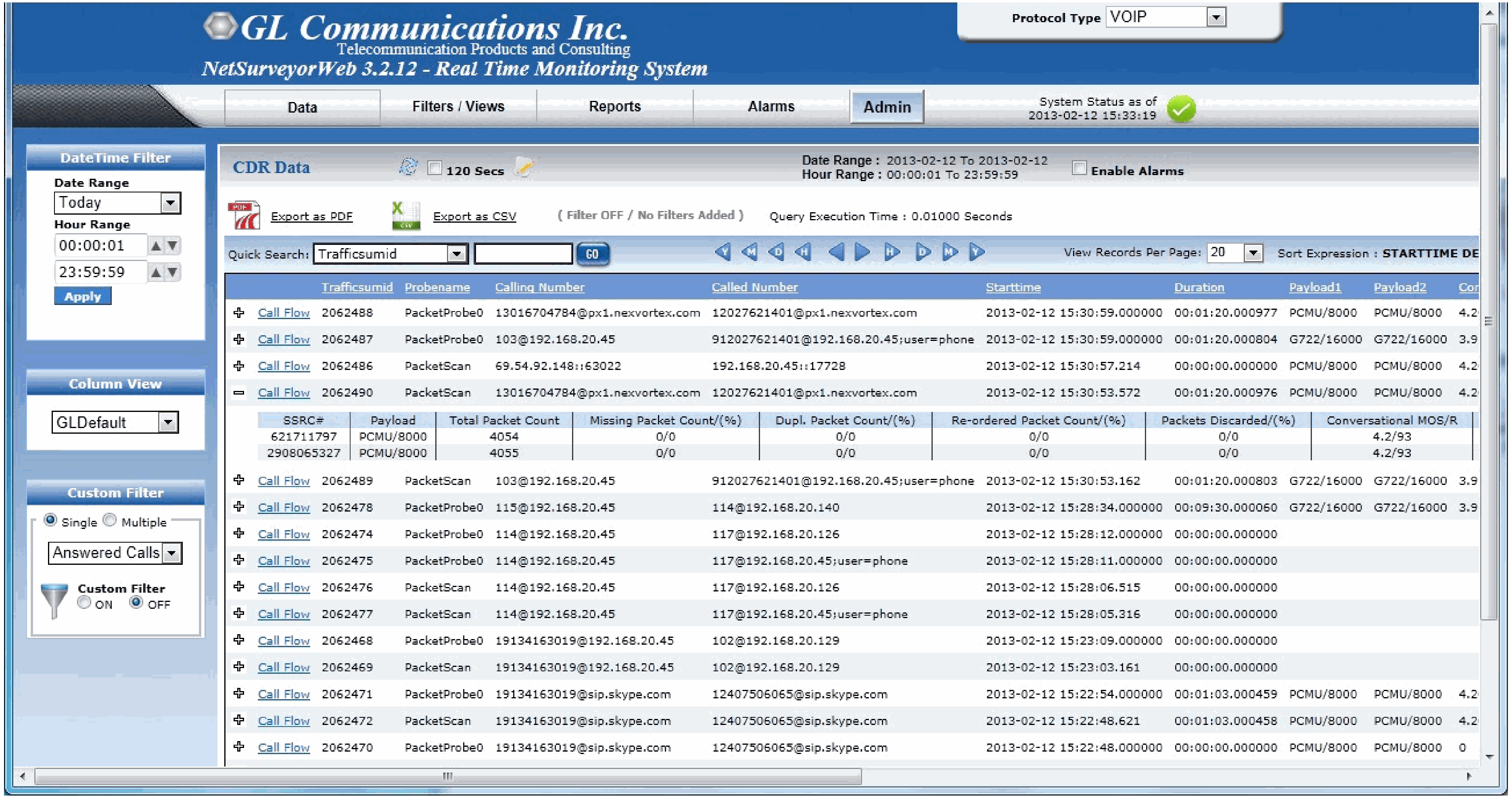
```
MapsCLI (SIP IETF)
File  Edit  View
View Latest Command
5 :: 2020-7-3 13:06:18.333000 : Start "TestBedDefault.xml" ;
5 :: 2020-7-3 13:06:18.442000 : LoadProfile "UserAgent_Profiles.xml"
5 :: 2020-7-3 13:06:18.770000 : Apply Global Configuration # "_EnableCLI"=1;
5 :: 2020-7-3 13:06:18.771000 : StartScript 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
```

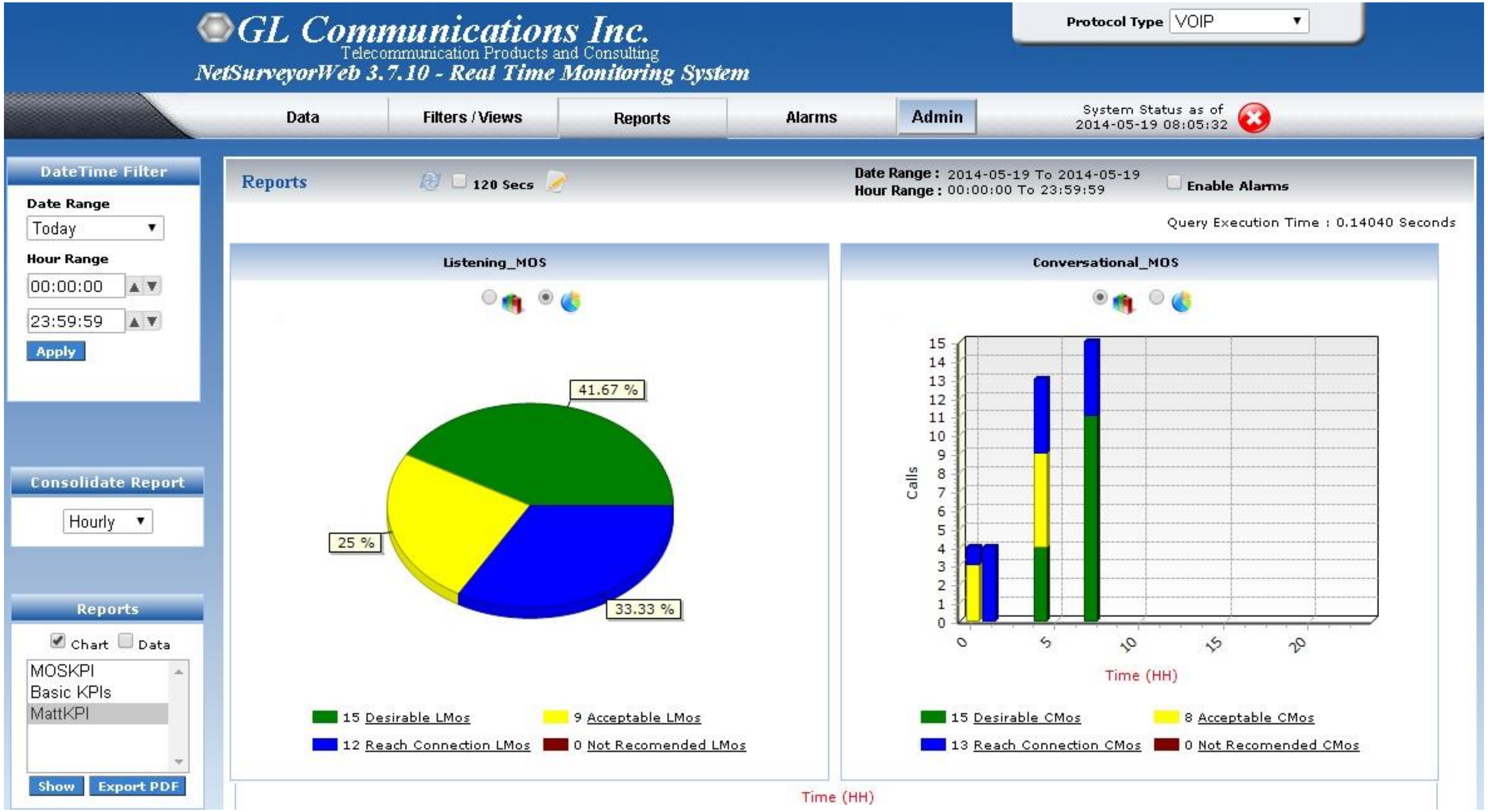
GL Communications

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



NetSurveyorWeb™ – Reports



Thank you