Traffic Simulation over IP, Ethernet, TDM, and Wireless with MAPS™

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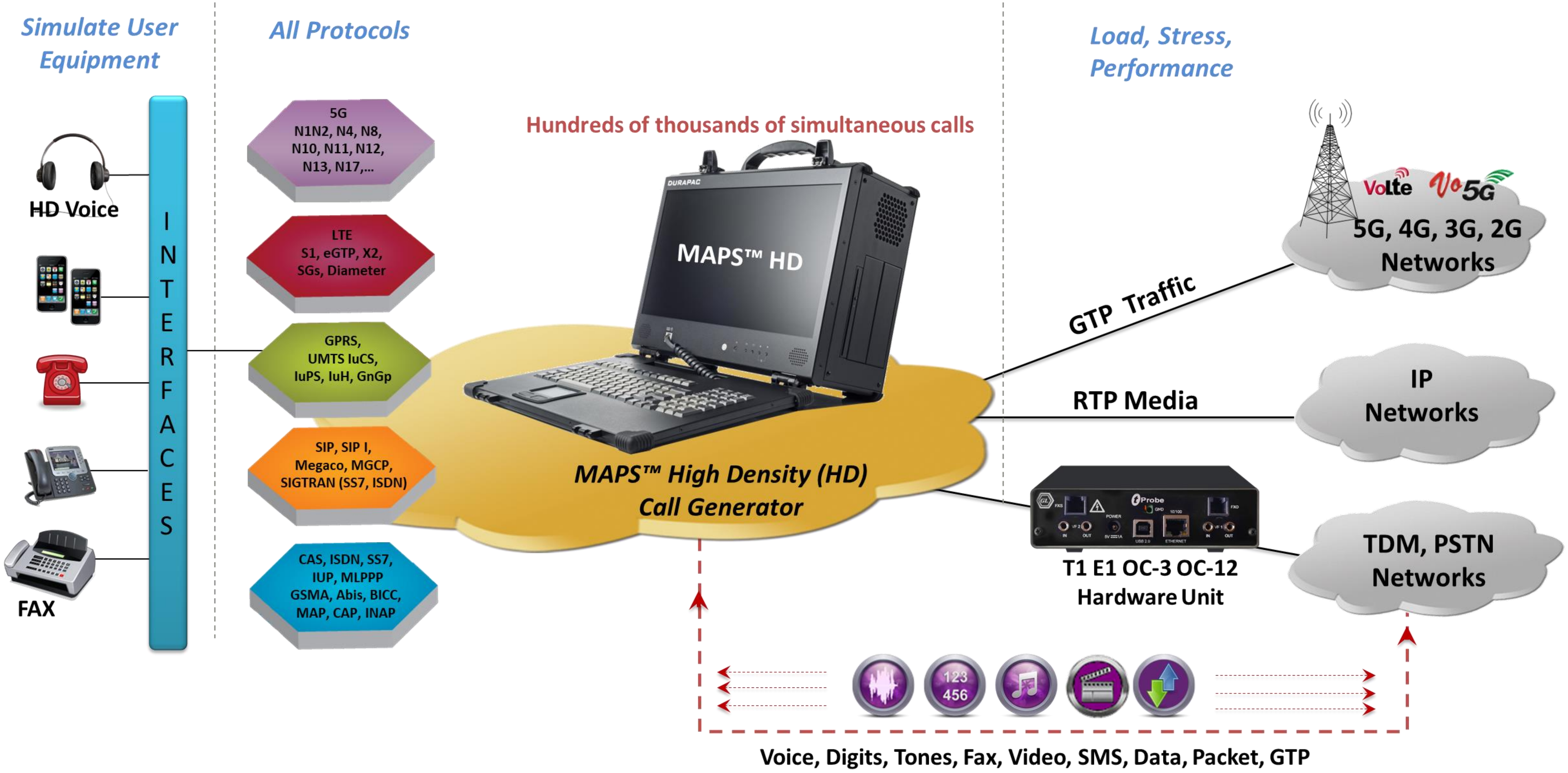
Traffic Simulation using MAPS™



Features

Traffic Types	
[GTP Mobile Traffic Simulation] Generate and verify user mobile data (Email, Web-HTTP, and FTP), gateway traffic, and packet traffic over (GTPv1 and GTPv2) GPRS Gb, UMTS (GnGp, IuPS), and LTE (S1, eGTP) network interfaces	• Stateless simulation of HDL Files, Hex string, and BER patterns, • GTP Mobile Traffic Core – Stateful HTTP traffic simulation • Simultaneous simulation of multiple GTP sessions per user • PacketLoad (HD GTP Mobile Traffic Core)- Stateful high density packet traffic generation • TCP/HTTP, UDP, and PCAP Replay • Mobile Traffic Core – Gateway • Mobile Traffic Simulation - GPRS Gb
RTP Traffic Simulation over SIP, SIP I, MGCP, MEGACO, UMTS, GSM, Diameter, and LTE networks	• Create, manage RTP sessions and generate and receive RTP traffic over the sessions with complete automation capability • Simulation of RTP Traffic such as Voice, Digits, Tones, IVR and Impairments • Automate the IVR testing process (call establishment and traffic generation / detection) process through scripts • All Voice Codecs supported including - G.711, G.711 App II with VAD, G.729, G.726, G.726 with VAD, GSM, AMR NB and WB, EVRC, SMV, iLBC, SPEEX NB and WB, and G722, G722.1.
	• Simulation of RTP Video Traffic (H.263 & H.264), Fax (Pass-thro & T.38)
	• RTP Voice Quality Measurements – MOS, R-Factor scores
	• Simulation of RTP FAX Traffic - G.711 Pass-thro and T.38 UDPTL
SMS Traffic Simulation over the GSM, UMTS, and MAP interfaces	• Ability to push / pull Short Messages over the network as if sent by thousands of mobile phones (Short Message Mobile Originated (SMS-MO)). MAPS™ can also transmit a Short Message to a mobile phone (Short Message Mobile Terminated (SMS-MT)).
TDM Traffic Simulation over ISDN, SS7, GSM, CAS interfaces	• Simulation of TDM Traffic such as digits, voice file, single tone, dual tones, Dynamic VF
	• Simulation of TDM Fax Traffic
	• TRAU GSM traffic over GSM Abis interface • Create, monitor, and terminate TRAU GSM traffic sessions

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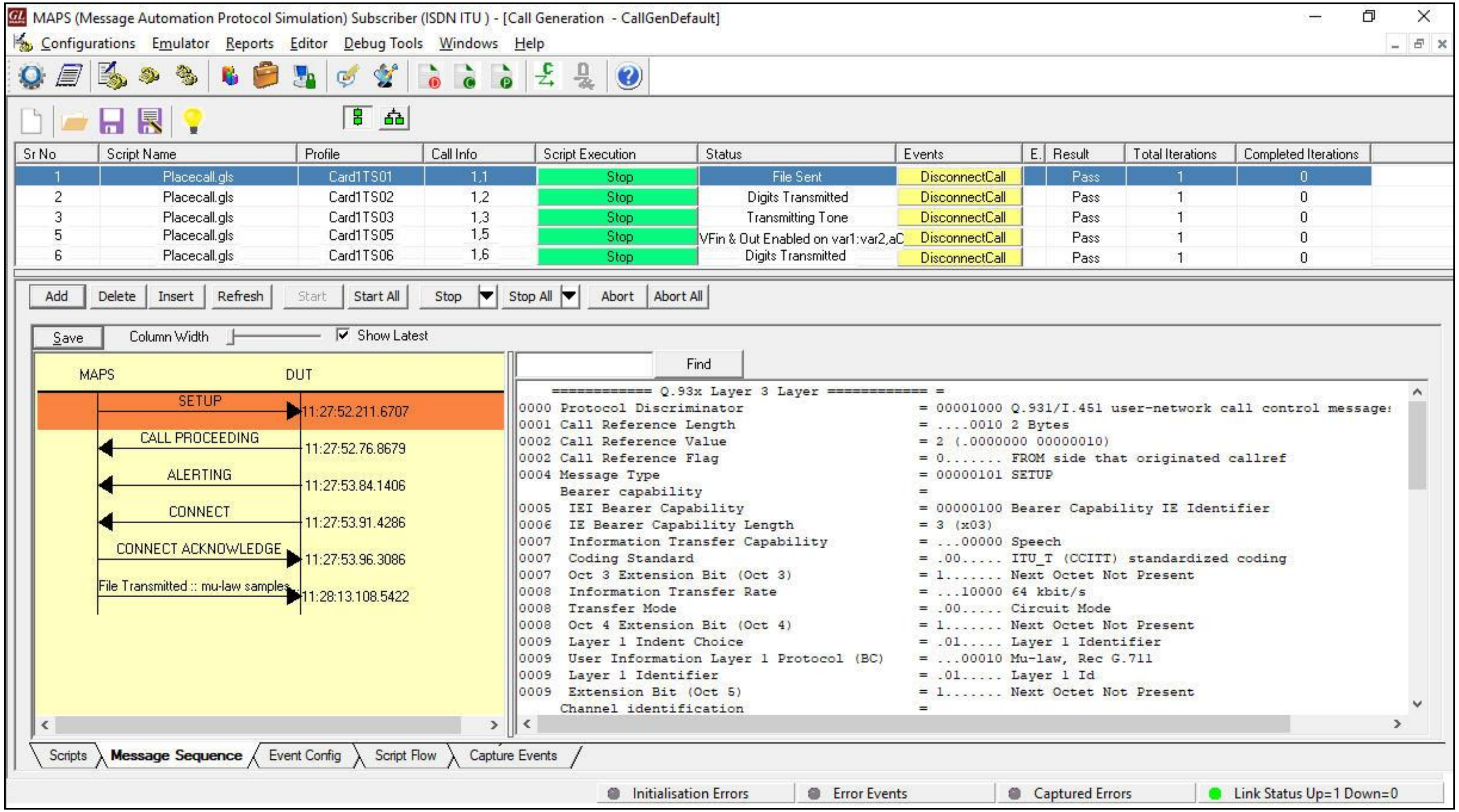
TDM Traffic Simulation

Analog and TDM Traffic Simulation

TDM Traffic Options	Licenses
File based Record/Playback (includes xx600)	XX610
Transmit/Detect digits (Place Call/ Answer Call) (includes xx600)	XX620
Multi-Channel TRAU Tx/Rx Emulation and Analysis	XX646
WCS Fax Emulation Software 2 Fax ports licences 8 Fax ports licences 30 Fax ports licences 60 Fax ports licences 120 Fax ports licences	XXFT0 XXXFT2 XXXFT3 XXXFT4 XXXFT5 XXXFT6

Voice, Digits and Tones

- With the purchase of additional license (xx610, xx620), MAPS™ supports transmission, detection and capture of DTMF/MF digits, voice files, single /dual tone over established calls on TDM and Analog networks
- The volume of calls can vary from few hundreds to thousands of calls depending on the T1 E1 or Analog platform of choice



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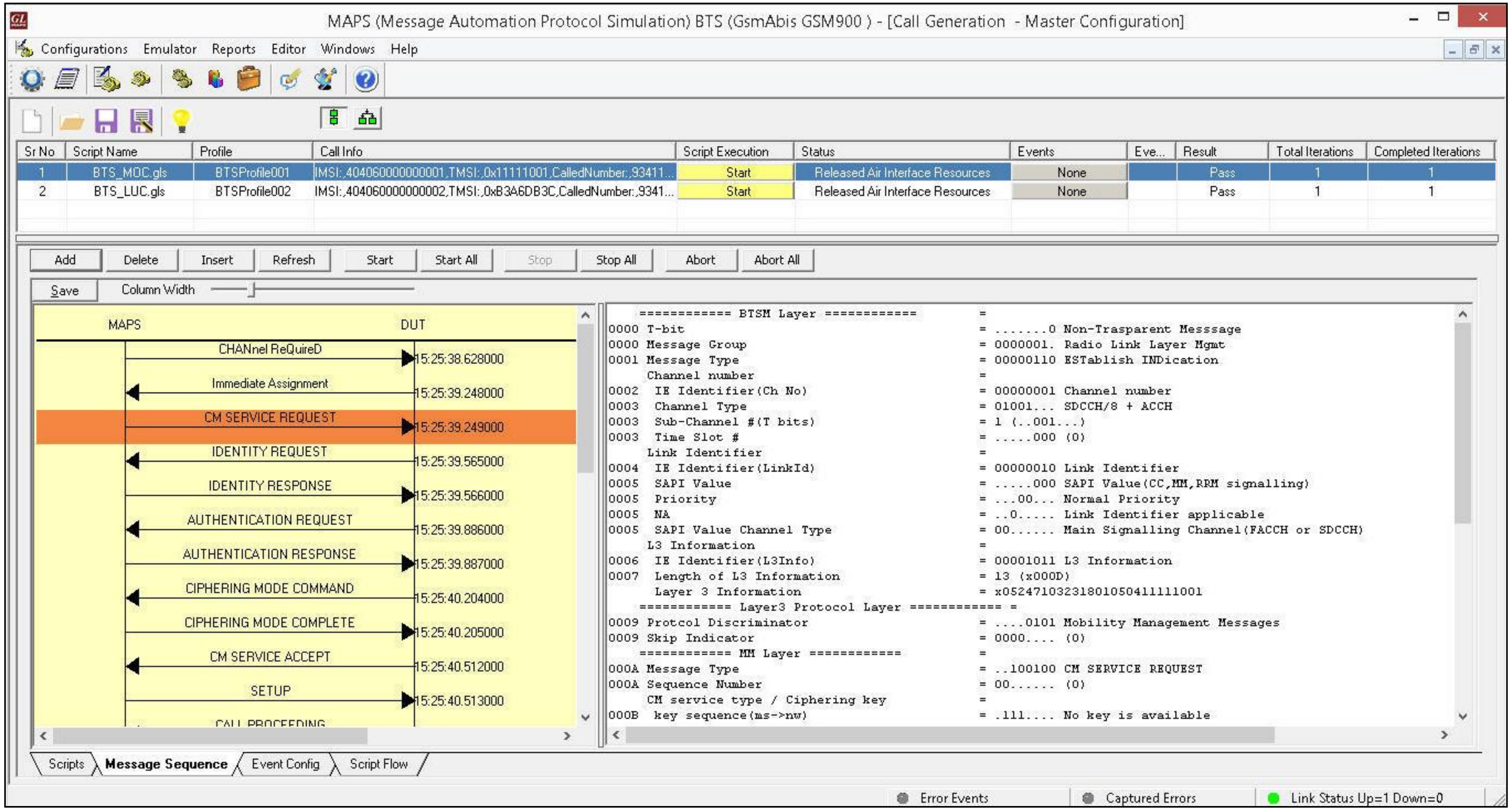
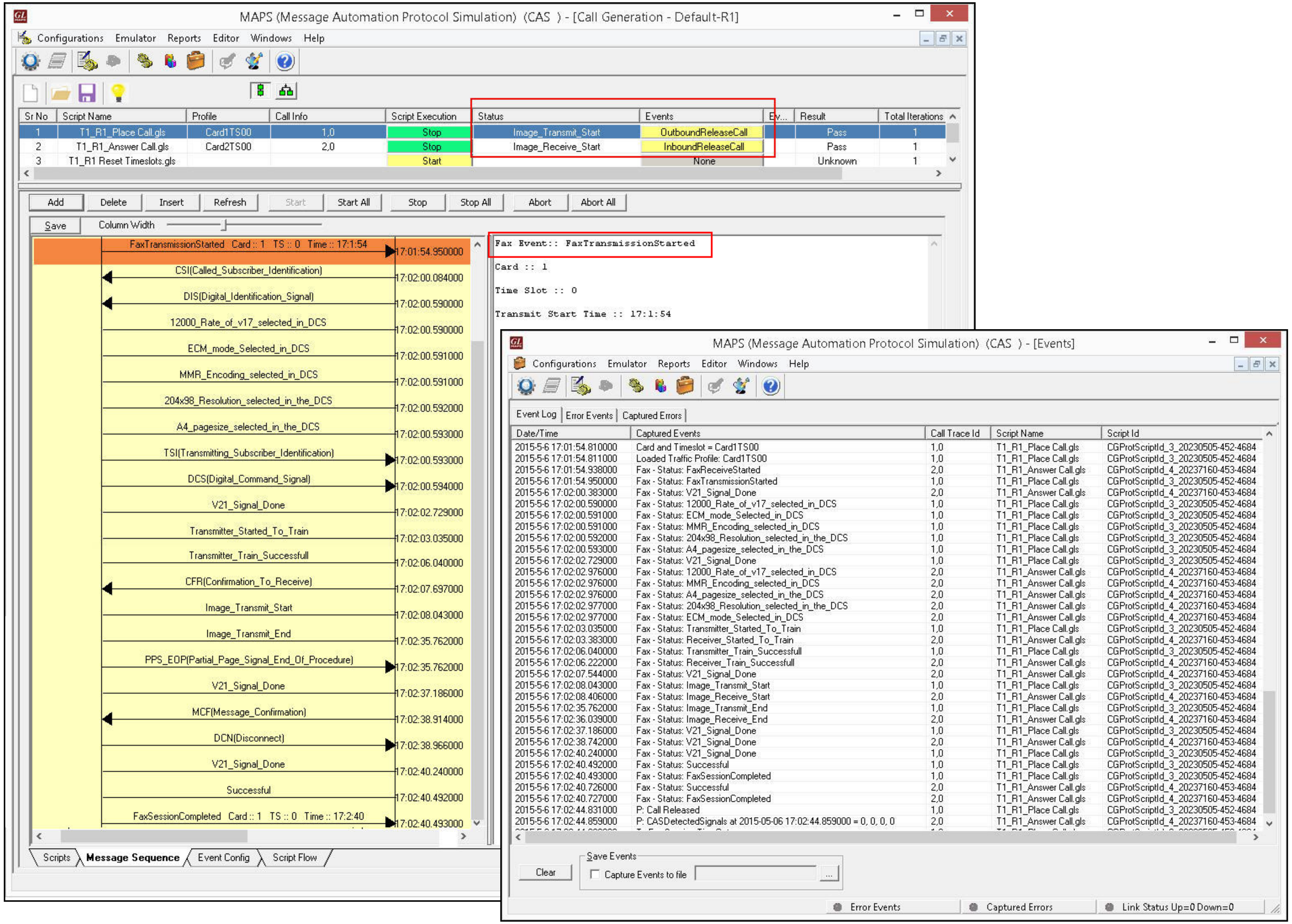
TRAU GSM Traffic

- For GSM, TRAU (Transcoder Rate Adapter Unit) traffic simulation (xx646) is included with options to create, monitor, and terminate TRAU GSM traffic sessions supporting transmit/receive DTMF digits, files, and tones over established GSM calls
- TRAU traffic simulation is applicable for MAPS™ GSM Abis application only

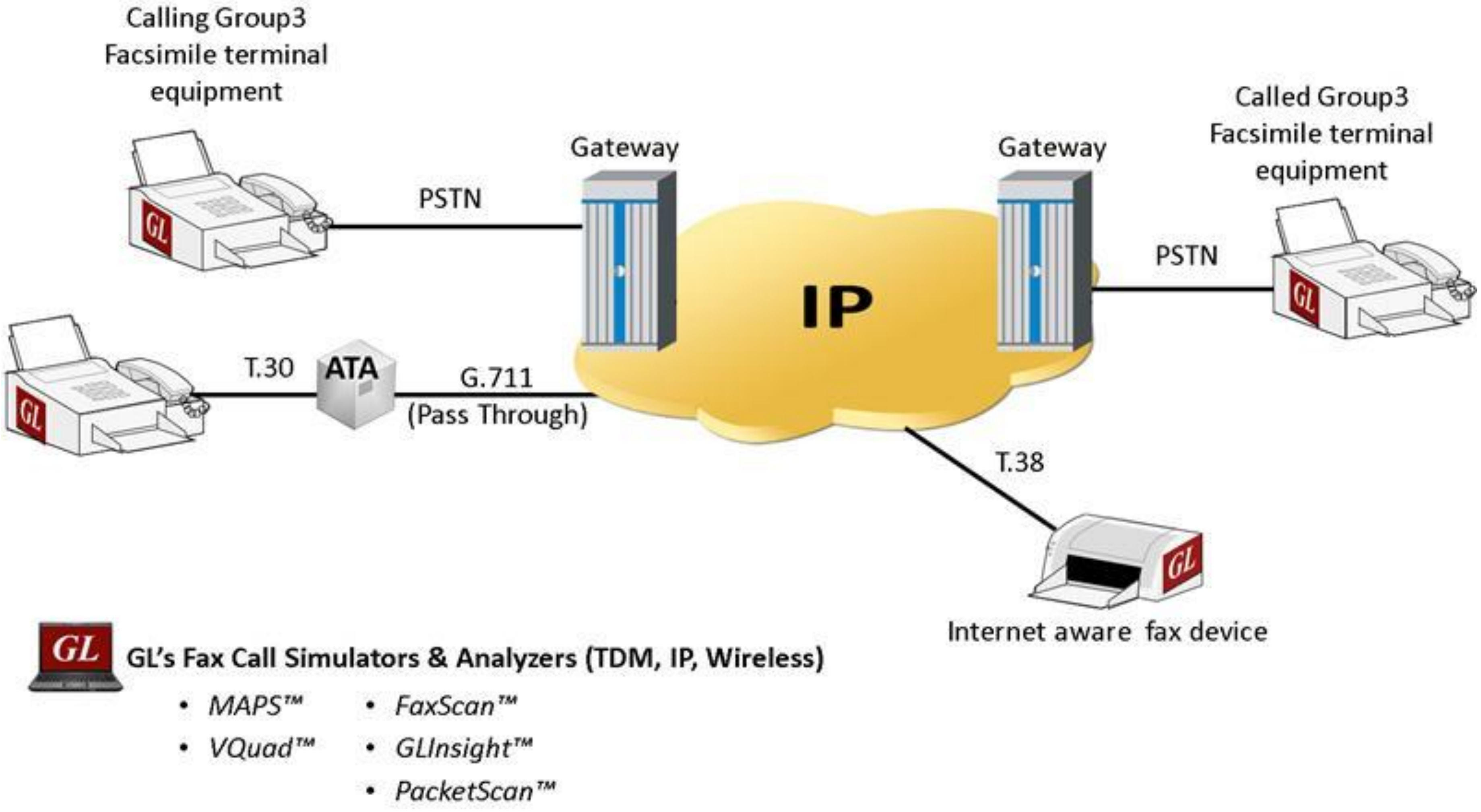


FAX Simulation over T1 or E1

- Fax Simulator is used to emulate complete real-time Fax calls over T1 or E1. It is available with MAPS™ CAS, MAPS™ ISDN, and MAPS™ SS7 emulators
- Fax Simulator can transmit and receiving single and bulk (100's) fax calls over many T1 E1 timeslots or through two-wire FXO and FXS lines
- Typical applications of our Fax Emulation software are - load testing of fax servers, qualification testing of T.38 Gateways, testing of ATAs (Analog Terminal Adapters), testing of fax machines, and many more

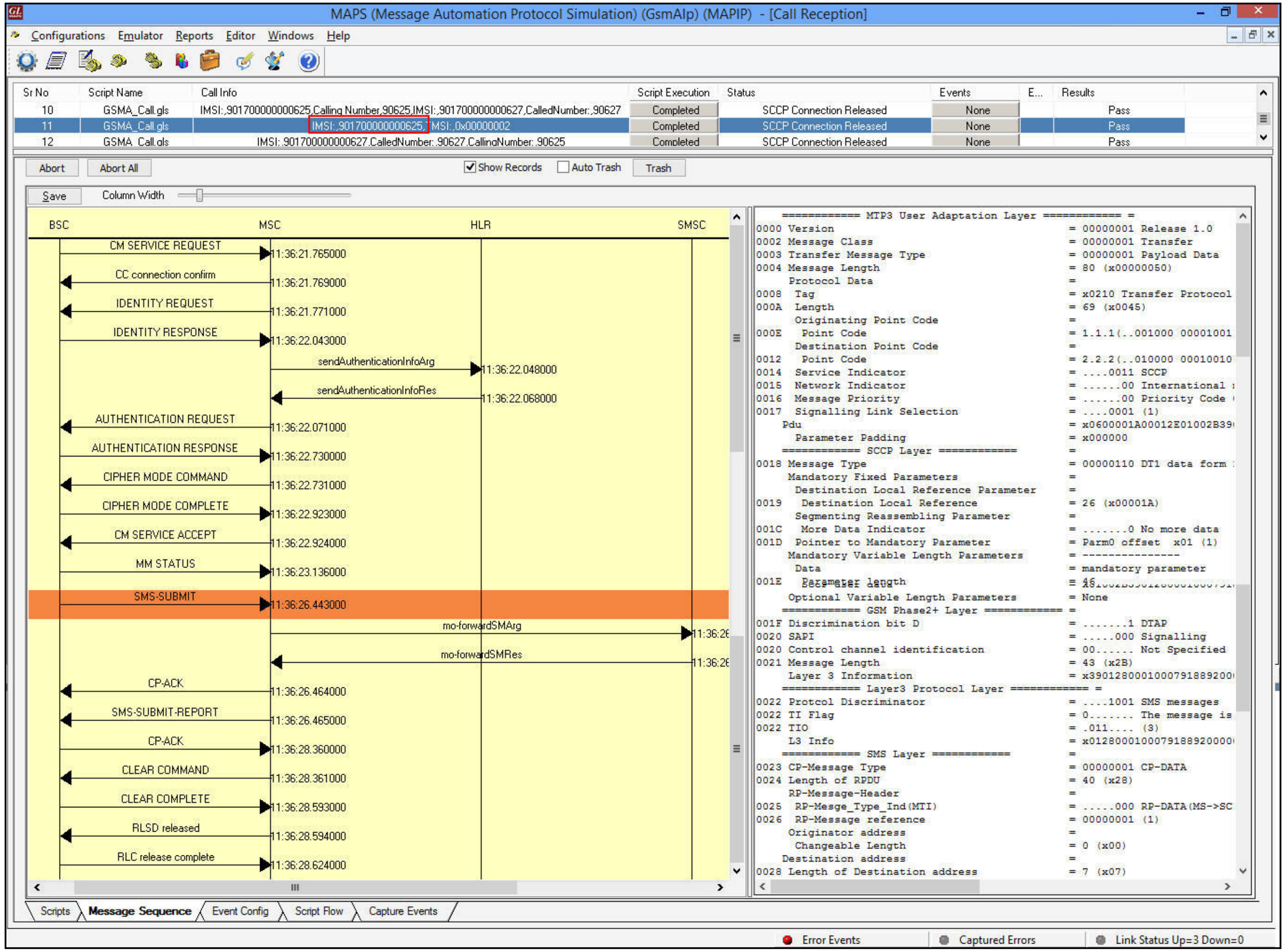


Fax Simulation over IP



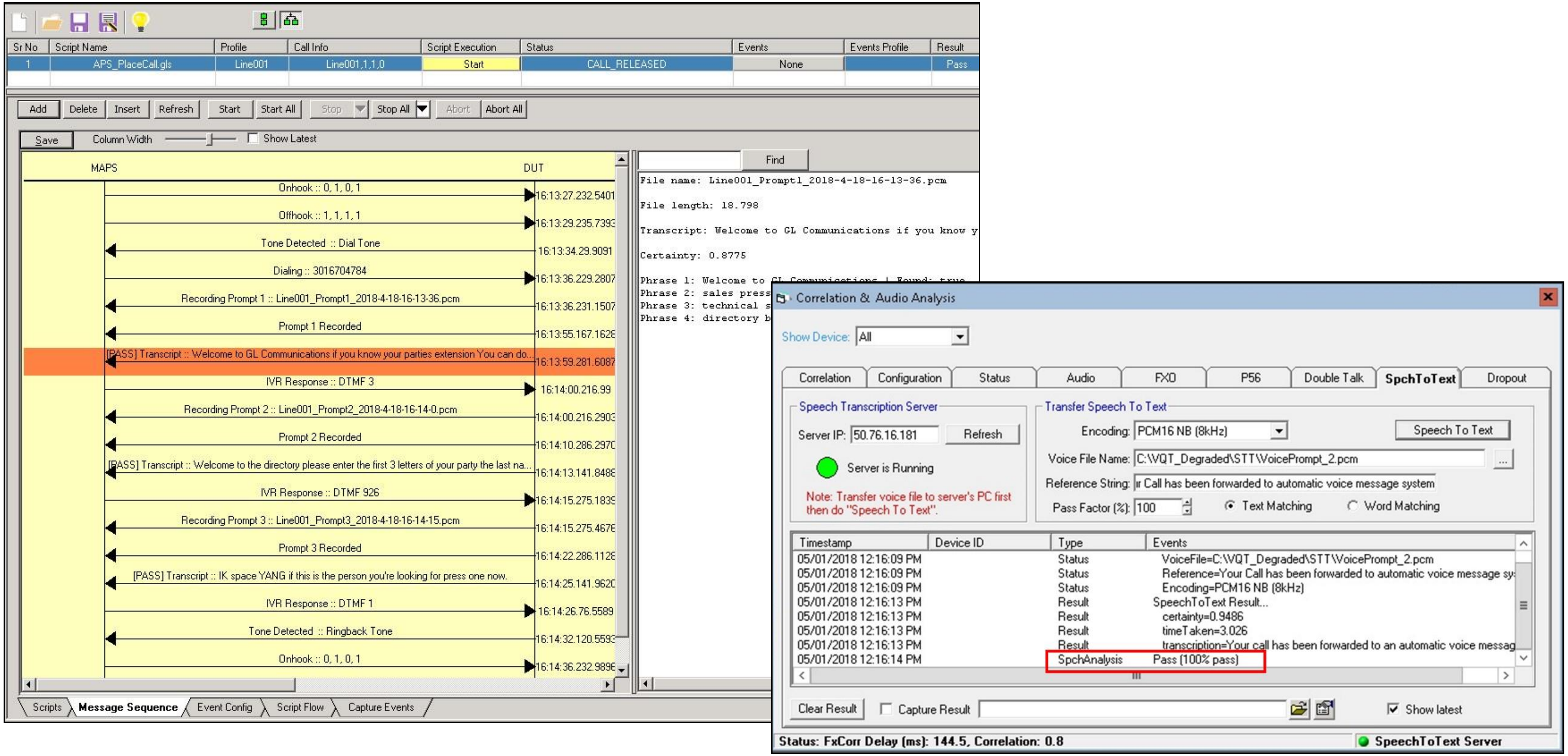
SMS Traffic

- MAPS™ also supports sending and receiving SMS (Short Message Service) using signaling channel simultaneously with other voice and data services over a GSM, UMTS, or MAP interfaces



IVR Test Solution

- GL's IVR test platforms can detect user-defined digits, send DTMF digits in response to voice prompts, tones, and play/record voice files, perform speech-to-text transcription, and analyze transcribed text for correctness, using a simple setup and automate the whole process through scripts



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TDM Traffic Commands

- Create TDM Sessions

  TxRx:create_tdmsession *(2, 2)*

- Send Action

  - Send Digits
    TxRx:tx _TDM dtmf digits: digits = "123456789*#,abcd", band = inband, power1 = -6, power2 = -4, ontime = 80, offtime = 80;

  - Send Tones
    TxRx: tx _TDM tone : freq1 = 400, power1 = - 8, duration= 5000;

  - Send File
    TxRx:tx _TDM file: filename = "C:\Program Files\GL Communications Inc\tProbe E1 Analyzer\A-Law Samples\count10.pcm";

TDM Traffic Commands (Contd..)

- Receive Actions

  - Monitor Digits

TxRx:monitor _TDM digits : band = inband, digittype = dtmf;

  - Monitor Tones

TxRx: monitor _TDM tones: "C:\Program Files\GL Communications Inc\tProbe E1 Analyzer\MTD Files\capture.mtd";

  - Record Files

TxRx:rx _TDM file: filename = "C:\Program Files\GL Communications Inc\ tProbe E1 Analyzer\A-Law Samples\Ajay.pcm" , duration = 30000 msec;

TDM Traffic Commands (Contd..)

- Start Fax Simulation

  TxRx:rawcommand " run task "FaxSimulatorT1:StartFaxSim";

- Transmit and Receive FAX

  - Transmit Fax

    TxRx:rawcommand “inform task * "TXFAX #1:1 TIFF_FILE 'WinClientServer\FAXSimulator\send\1.tif' CODEC_TYPE MULAW MODEM_TYPE 16 MIN_DATA_RATE 16800 MAX_DATA_RATE 33600 PAGESIZE_TYPE 1 RESOLUTION_TYPE 16 ECMENABLED 1";”trafficaction;

  - Receive Fax

    TxRx:rawcommand “inform task * "RXFAX #2:1 TIFF_FILE 'WinClientServer\FAXSimulator\Recv\rcvV34.tif' CODEC_TYPE MULAW MODEM_TYPE 16 MIN_DATA_RATE 16800 MAX_DATA_RATE 33600 PAGESIZE_TYPE 1 RESOLUTION_TYPE 16 ECMENABLED 1";” trafficaction;

Sample TDM Traffic Script

```
"OnCallConnected":
  TxRx:create_tdmsession(Cardno,TS);
  goto "TX-File";
  return;
"OnCallTerminated":
   goto "Stop Traffic";
   return;
"TX-File":
  TxRx:tx _TDM file: filename = "C:\Program Files\GL Communications Inc\Usb E1 Analyzer\A-Law Samples\Count10.pcm";
  Status="TX-File";
  EventLog ("Tx File Done");
  resume;
"Stop Tx":
  TxRx:stop _TDM tx file ;
  Status="Stop-TX";
  EventLog ("Stop all Tx Traffic");
  resume;

"Stop All":
  goto "Stop Traffic";
  resume;

"Stop Traffic":
  TxRx:stop _TDM tx file ;
  return;
```

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RTP Traffic Simulation

RTP Traffic Simulation

RTP Traffic Options	Licenses
RTP Soft Core for RTP Traffic Generation	PKS102
RTP IuUP Softcore	PKS103
RTP Video Traffic Generation	PKS106
RTP EUROCAE ED137	PKS107
RTP Voice Quality Measurements	PKS108
RTP Pass Through Fax Emulation 2 Fax Ports Licences 8 Fax Ports Licences 30 Fax Ports Licences 60 Fax Ports Licences 120 Fax Ports Licences	PKS200 PKS202 PKS203 PKS204 PKS205 PKS206

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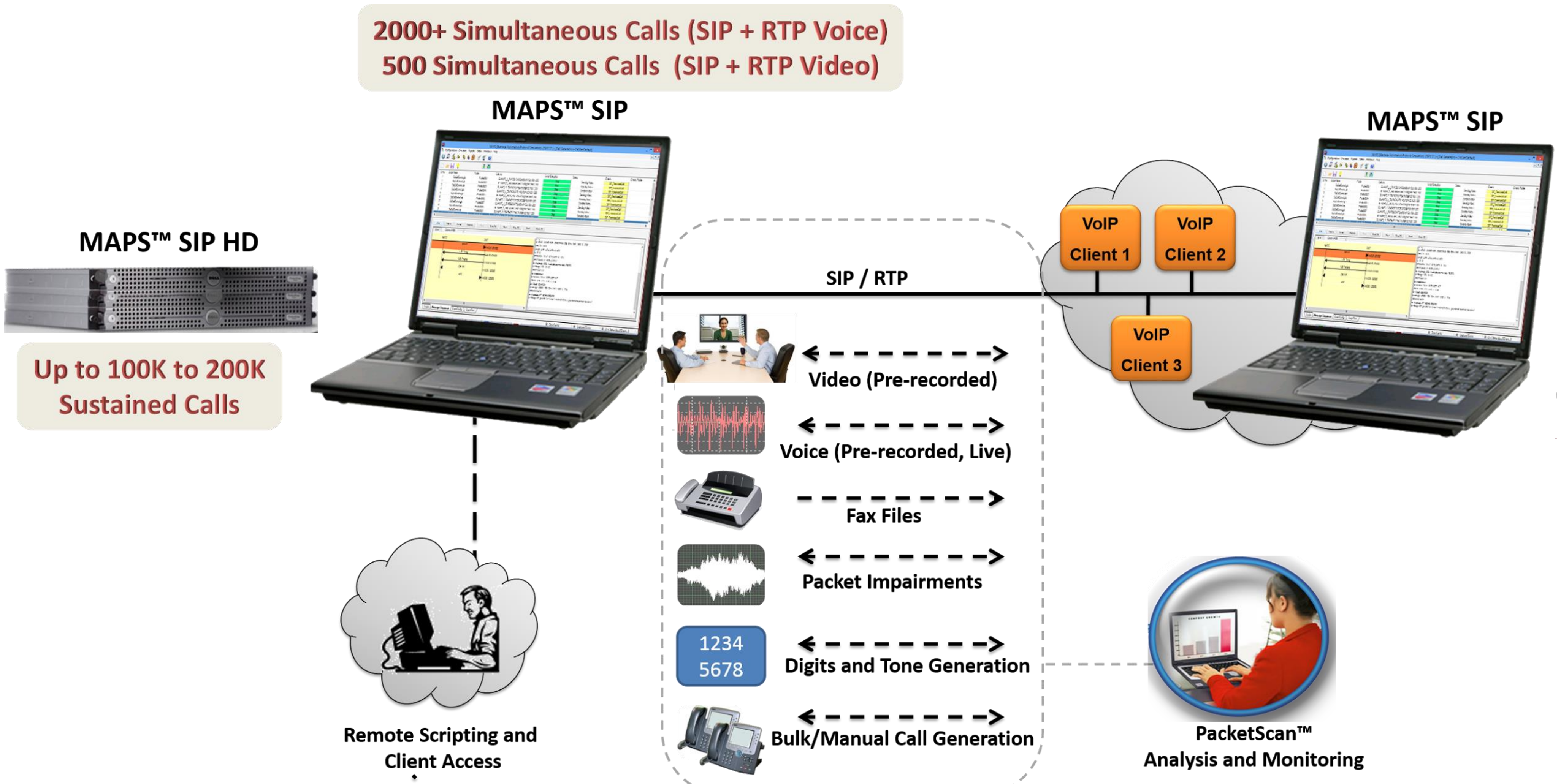
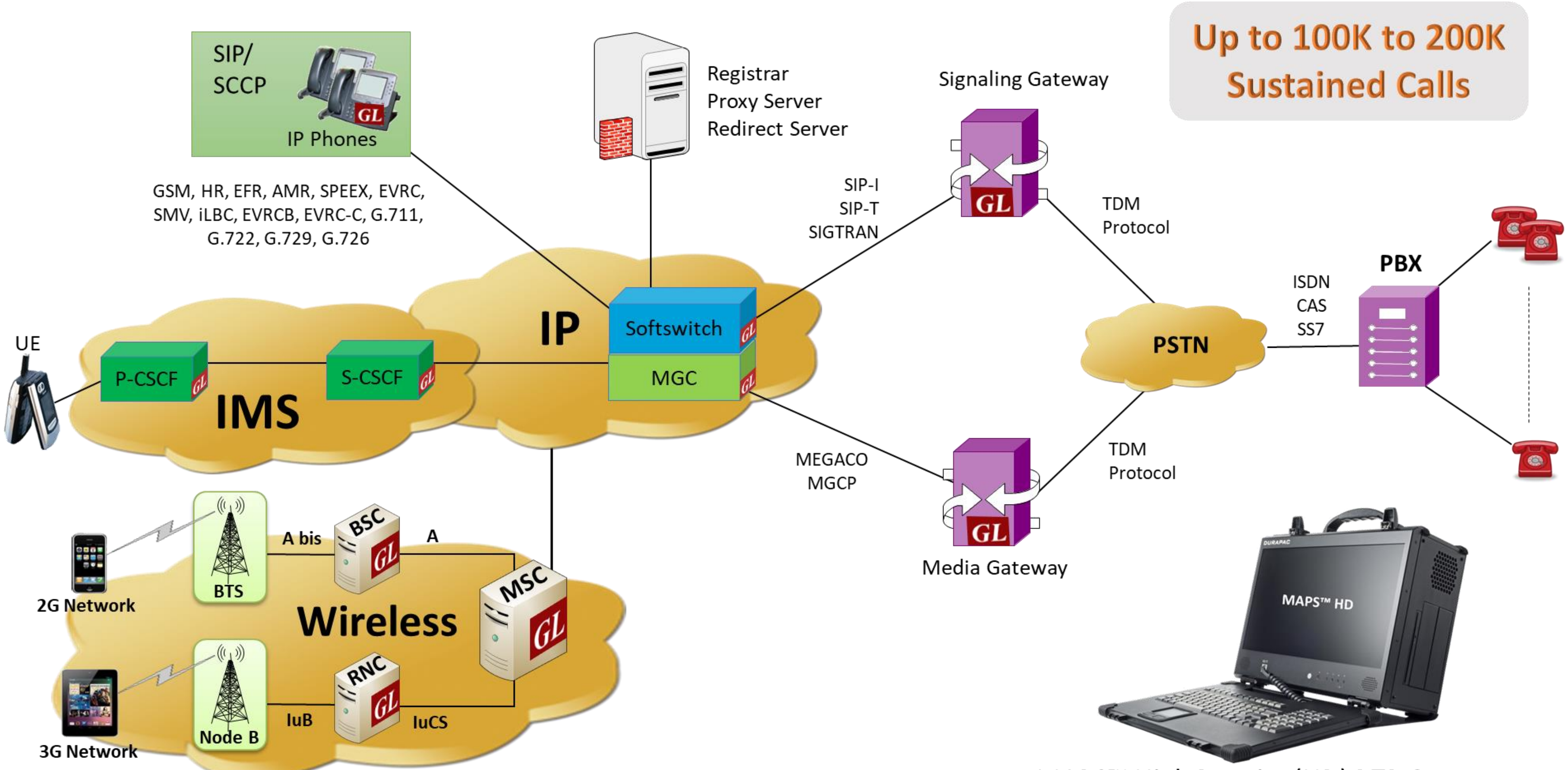
IP Traffic (RTP) Simulation Capabilities and Performance



High Density (HD) RTP Traffic Simulation



MAPS™ High Density (HD) RTP Generator (w/ 4x 1GigE or 2x 10 GigE HD NICs)

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RTP Traffic Capabilities and Performance

Product Version	Max Simultaneous Calls		
	Only Signaling	Signaling + RTP Voice Traffic	Signaling + RTP VideoTraffic
MAPS™ SIP 64-bit	70,000 Calls @ 250 CPS	2000 @ 250 CPS	500
MAPS™ SIP HD 64-bit	100,000 Calls @350 CPS	20000 @ 350 CPS	-

** The above performance is evaluated on a high-end Core i7 system with typical 12GB RAM.

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RTP Traffic Simulation

- Create, manage RTP sessions and generate and receive RTP traffic over the sessions with complete automation capability
- Transmit and receive pre-recorded video traces with video codecs like H.264, H.263 etc. **
- Transmit and receive pre-recorded voice files, and live voice
- RTP based Voice Quality (MOS and R-Factor) measurement for the received streams
- Customize codec options (payload type, ptime) over Tx/Rx sessions. All Voice Codecs are supported (Visit Voice Codecs webpage for more comprehensive information)
- Talk using Microphone - allows the user to generate live voice. "Play to Speaker" streams voice to a speaker.
- Transmit and receive FAX files in T.38 pass-through mode **
- Loopback real-time voice traffic (all received traffic is retransmitted as sent traffic)

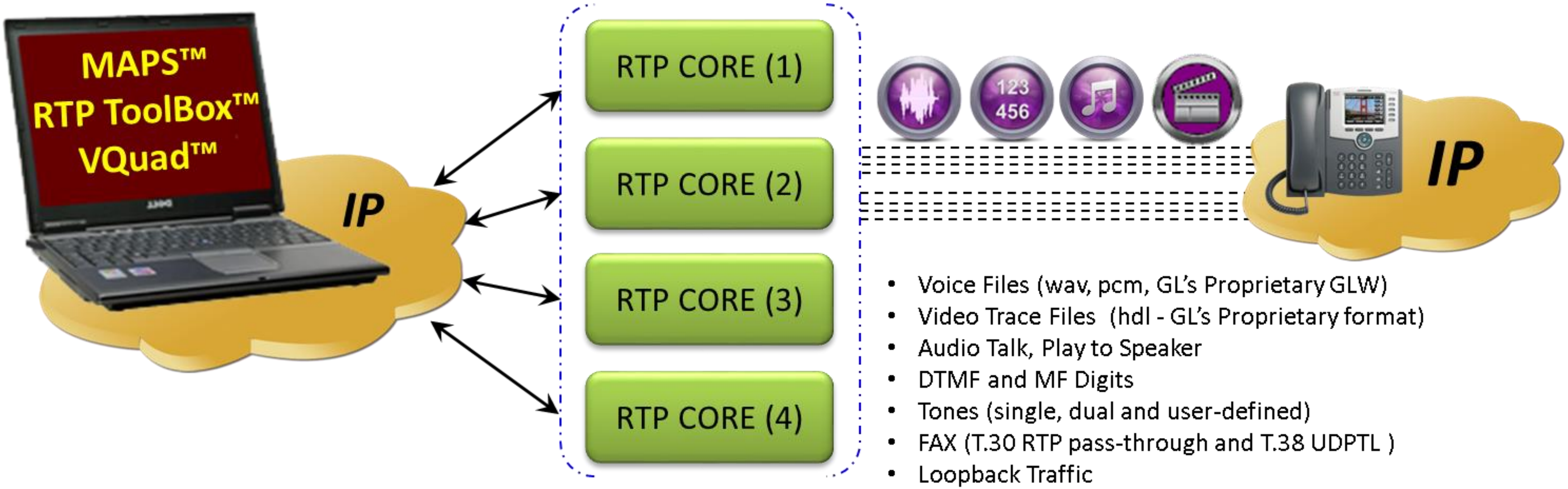
** Some of these features requires additional licenses – contact GL for more information

RTP Traffic Simulation (Contd.)

- Generation and Detection of RTP Events per RFC-2833 & RFC-4733 such as Answering Tone, Calling Tone, Special Dial Tone and other Call Progress Tones
- Generation of user-configurable impairments Latency, Packet Loss, Packet Effects over established RTP calls
- Supports RTP traffic implementation over Iu-UP (Iu User Plane Interface) layer of the UMTS IuCs Network**
- Supports RTP traffic as per ED-137B of EUROCAE standards used for voice communication in Air Traffic Control networks*
- Provides some vital statistics like total packets received and transmitted, Jitter, delay, lost packets, duplicate packets and out of order packets on each session
- Detailed statistical information of RTP and RTCP packets
- Jitter Buffer implementation for the received traffic to give near real time affect

** Some of these features requires additional licenses – contact GL for more information

RTP Core

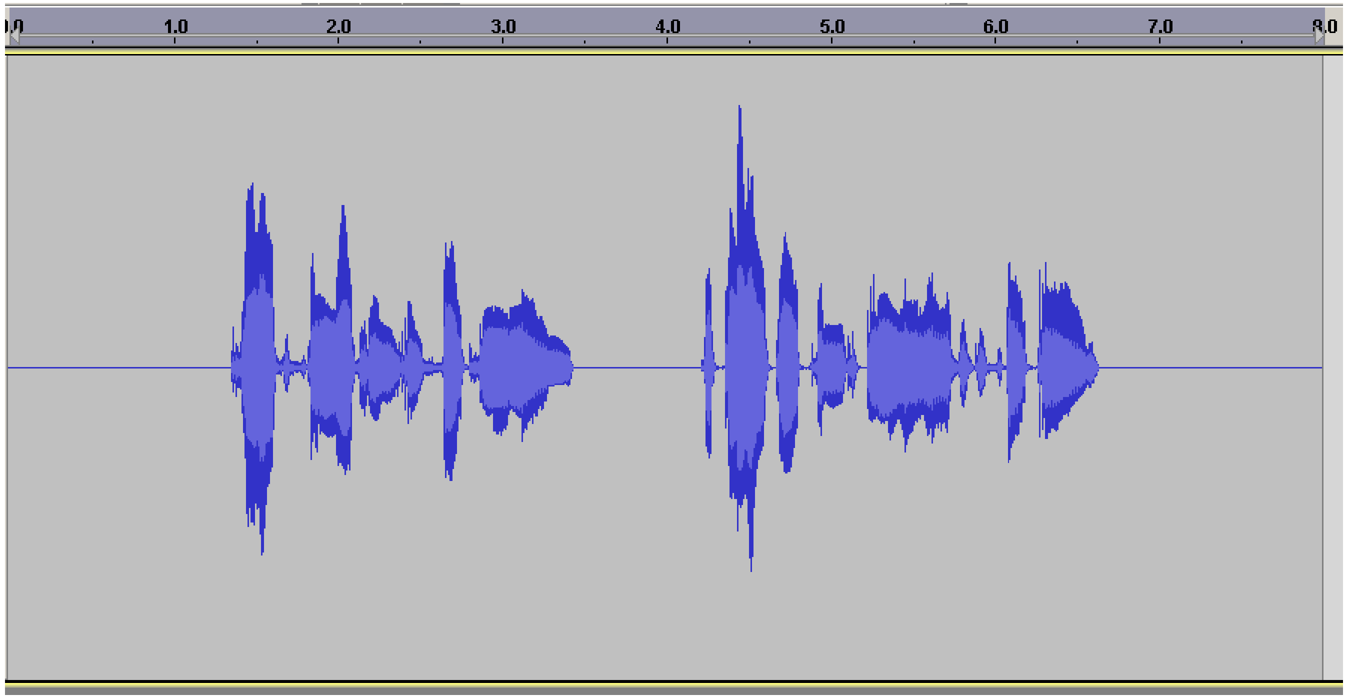


- Voice Files (wav, pcm, GL's Proprietary GLW)
- Video Trace Files (hdl - GL's Proprietary format)
- Audio Talk, Play to Speaker
- DTMF and MF Digits
- Tones (single, dual and user-defined)
- FAX (T.30 RTP pass-through and T.38 UDPTL)
- Loopback Traffic
- Voice Quality Metrics (LMOS CMOS)

RTP - Voice File

- POLQA relies on the use of specific files, tuned to suit the algorithm (supplied to GL by the ITU POLQA group)

- VQT also supports PESQ analysis

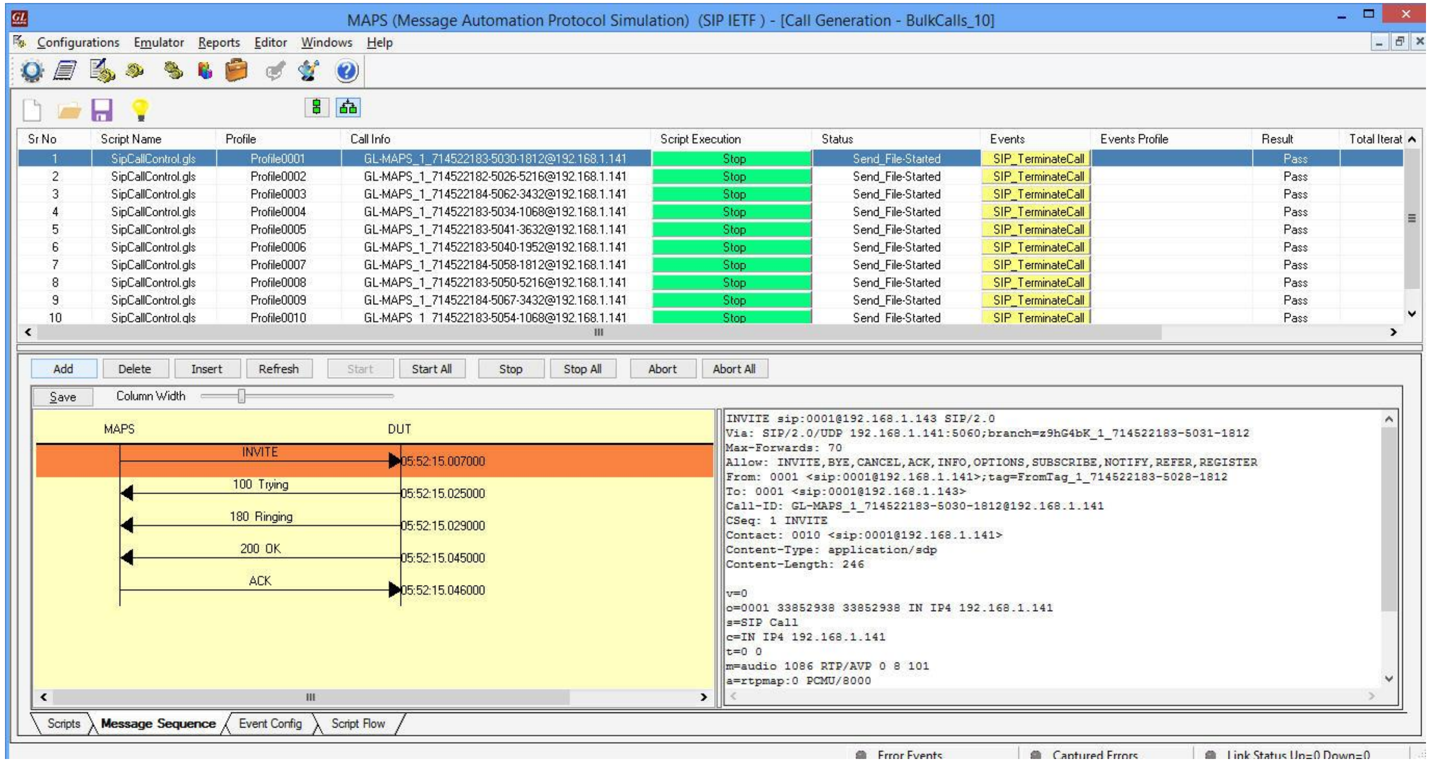


MAPS™ HD RTP

- Rackmount network appliance with 4x1GigE NIC
- Transport over UDP and TCP, IPv4 and IPv6, and TLS for secure transport
- Easily achieve up to 20,000 endpoints per appliance (5000 per port)
- Up to 350 calls per second (with RTP traffic)
- Scales to around 100,000 to 200,000 endpoints with use of Master Controller for single point of control
- Manage 10+ MAPS™ systems with single point of control from Master Controller

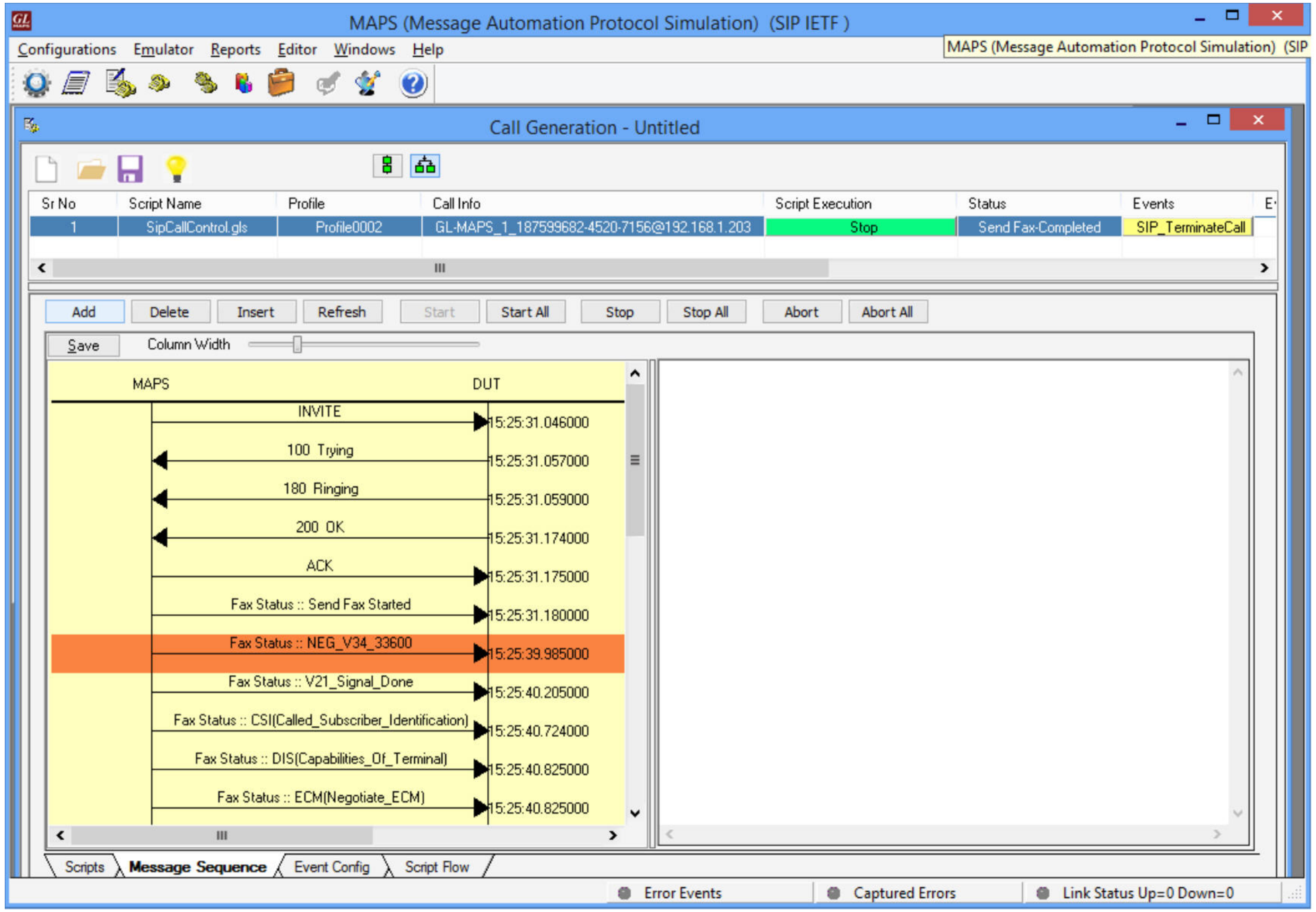
Bulk Voice Traffic Simulation

- Allows to specify a desired voice payload type to each codec for sending and receiving payload;
- Sampling rate of the codec is displayed for the selected codec
- Comfort noise generation is supported for A-law, µ-law and G.726 codecs for sending and receiving payload
- Allows to set the buffer used for delayed packets that arrive at receiving end (both static and dynamic jitter buffers are supported)
- Allows to set QoS (Type of Service) properties such as precedence, delay, throughput and reliability values to the outgoing stream

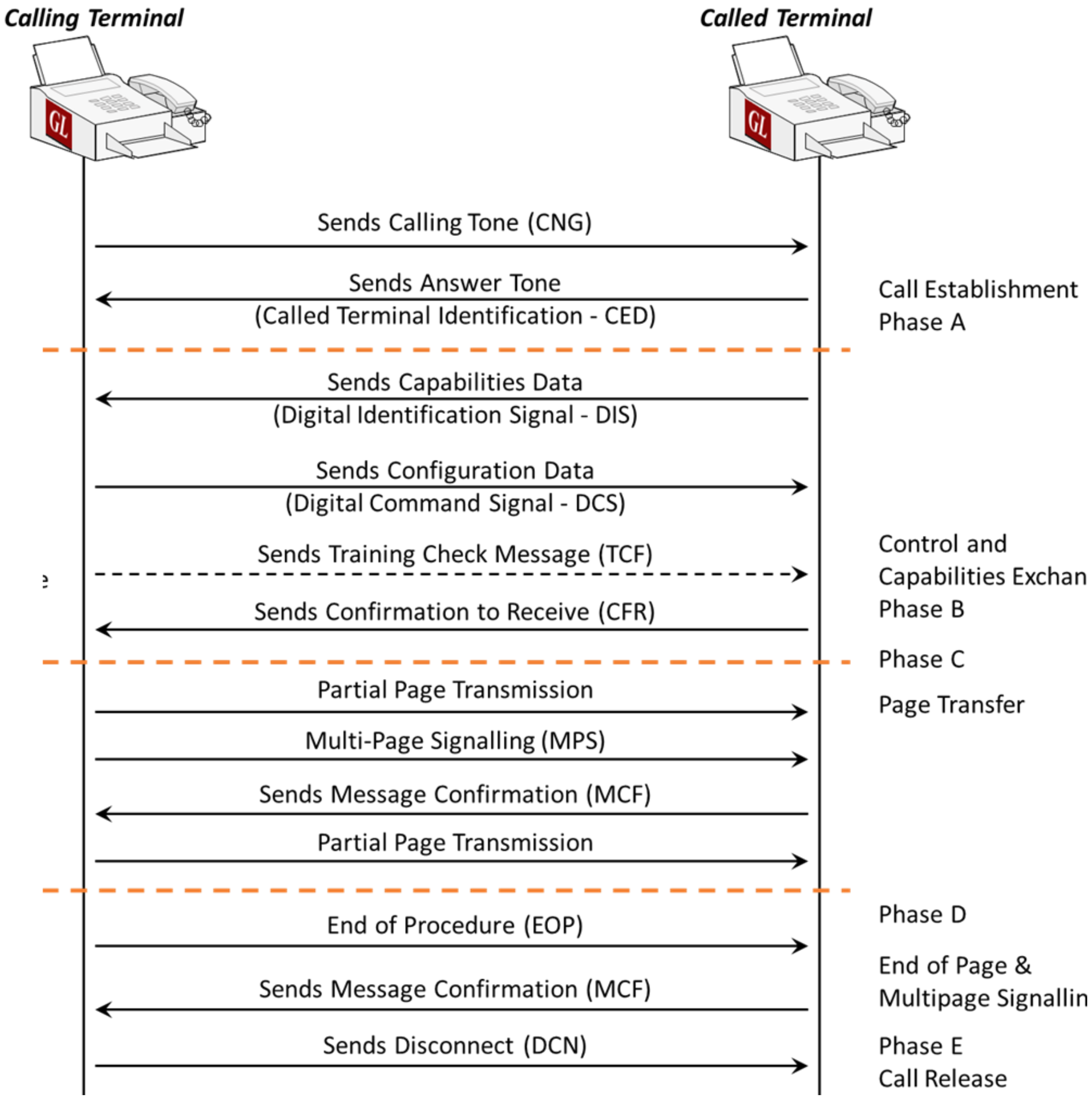


Bulk RTP FAX Simulation (T.30 pass through and T.38 UDPTL)

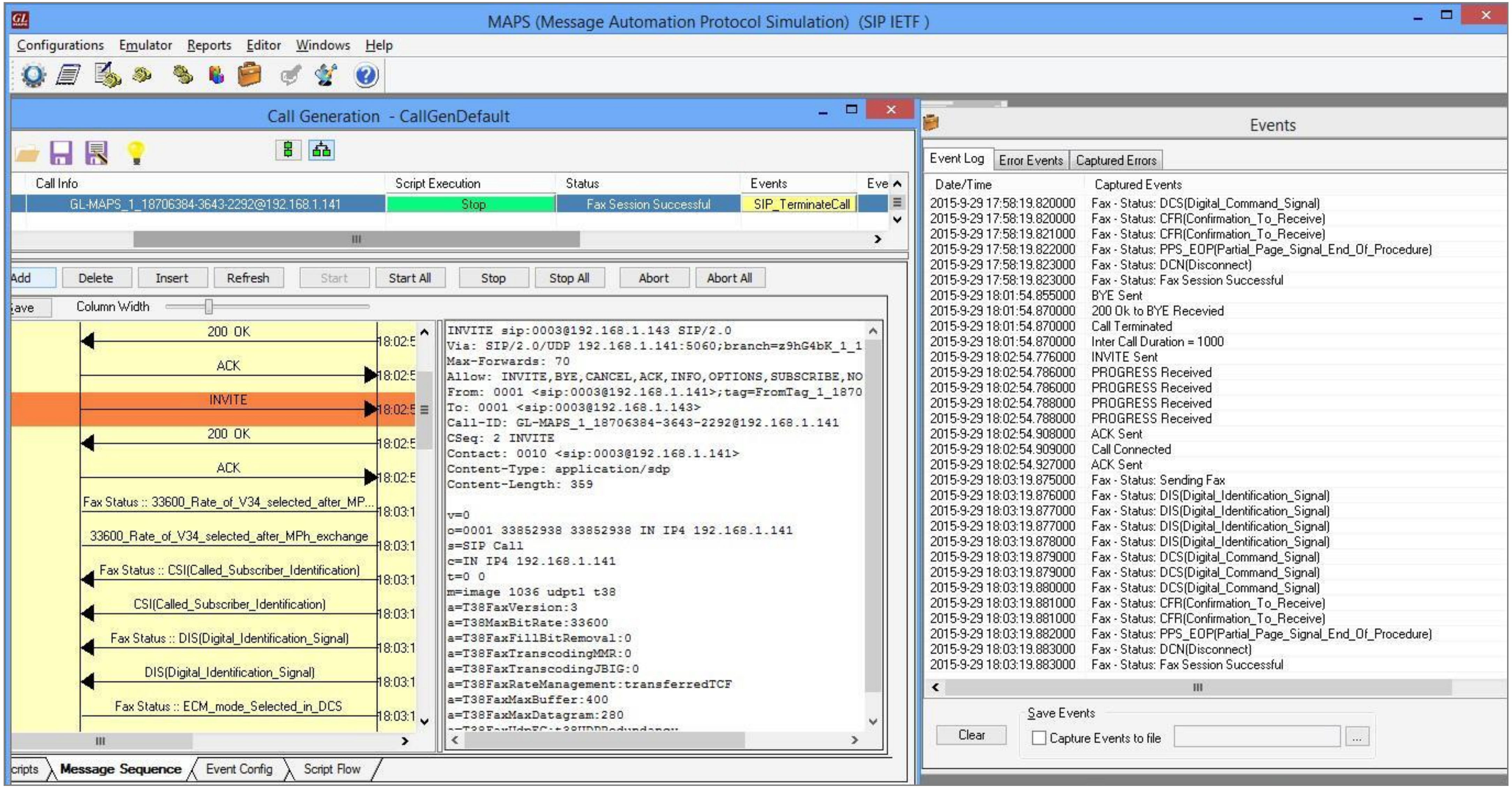
- RTP pass-through supports up to 120 Fax ports, whereas T.38 fax simulation over UDPTL supports unlimited channels, and constrained only by CPU capacity
- MAPS™ allows the user to initiate fax calls by sending call control messages using proper scripts and profiles. The profile allows necessary parameters of call control messages to be changed during runtime. The below figure depicts the T.30 fax call being generated using MAPS™ SIP



Call Scenarios - Fax T.30

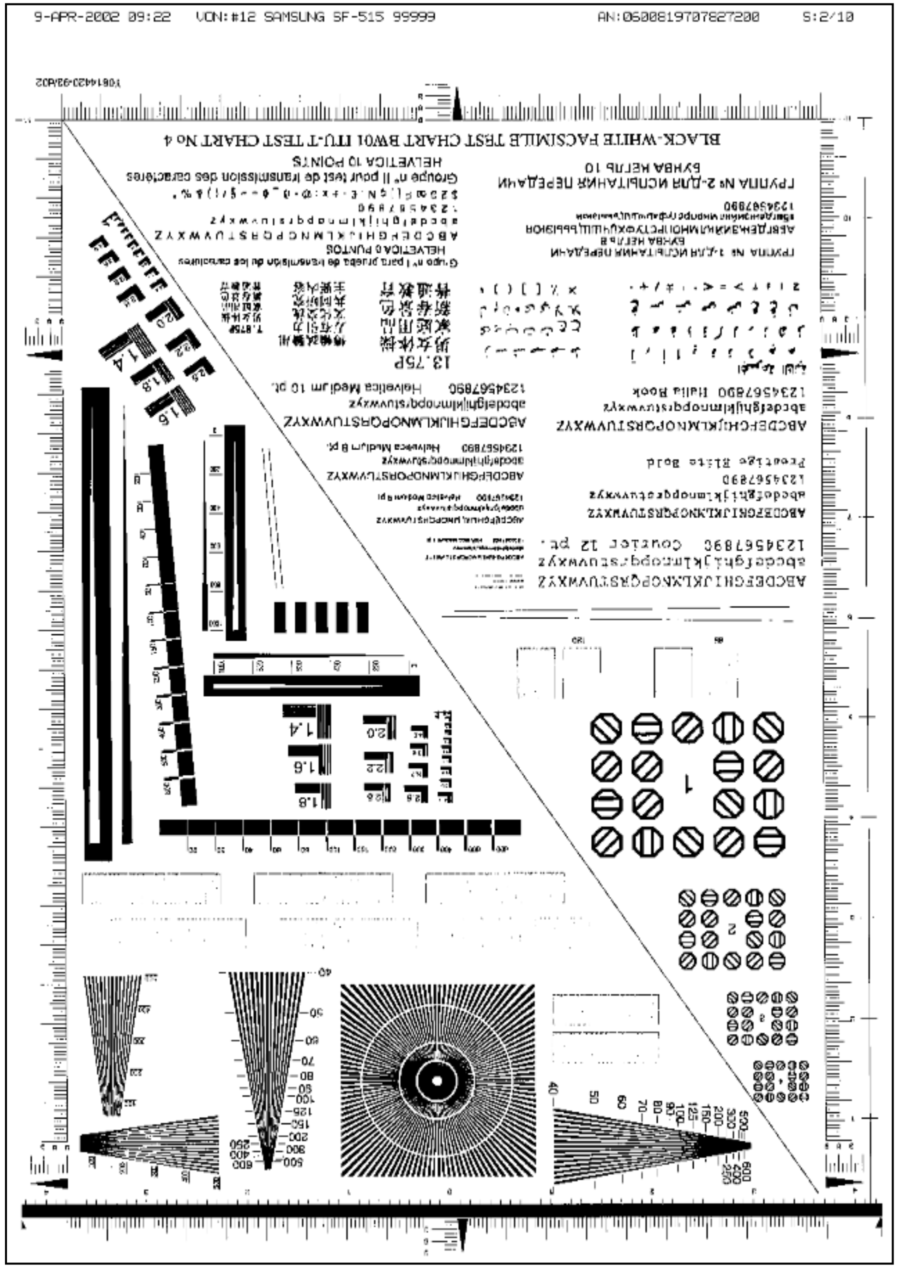


T.38 Fax Call in Progress and Related Events



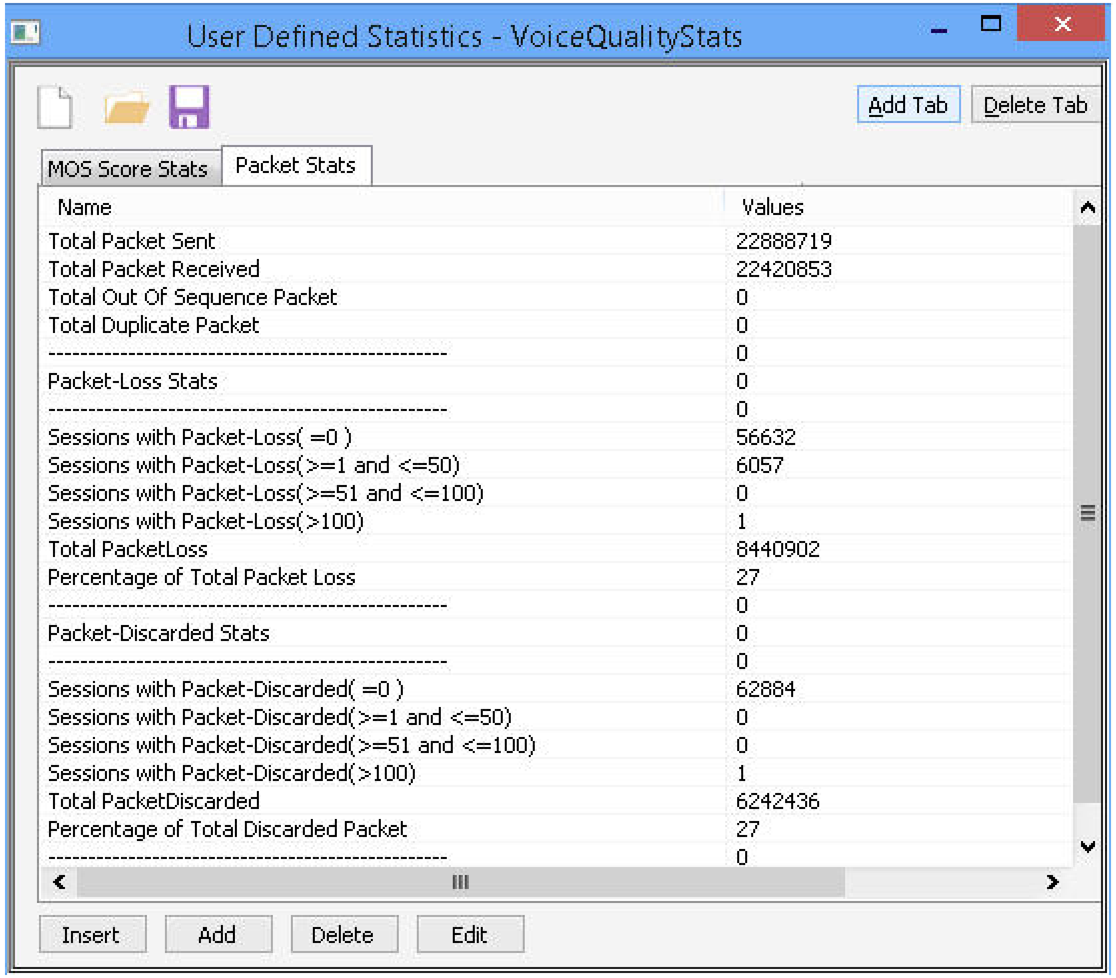
RTP - Fax File

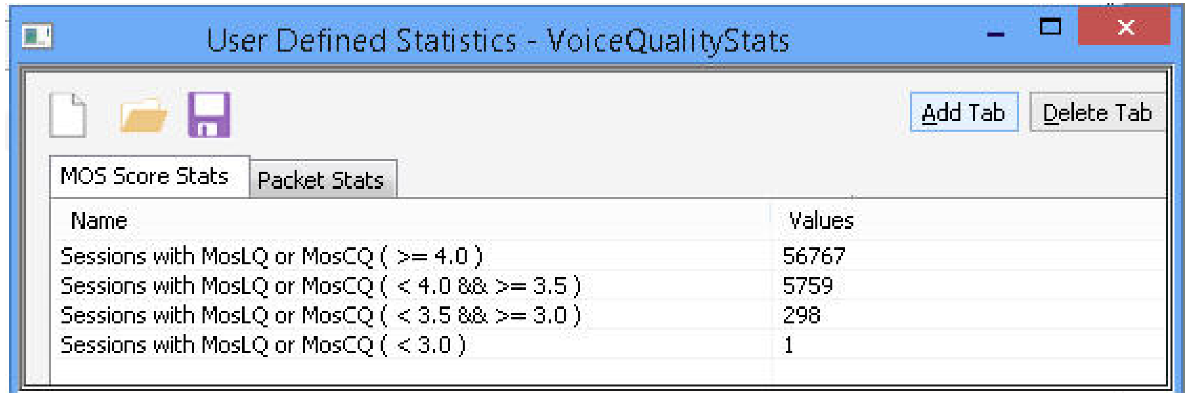
- GL provides several *.tif fax files for transmission/reception
- Files were designed by the CCITT (Consulting Committee for International Telephone and Telegraph) many years ago



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





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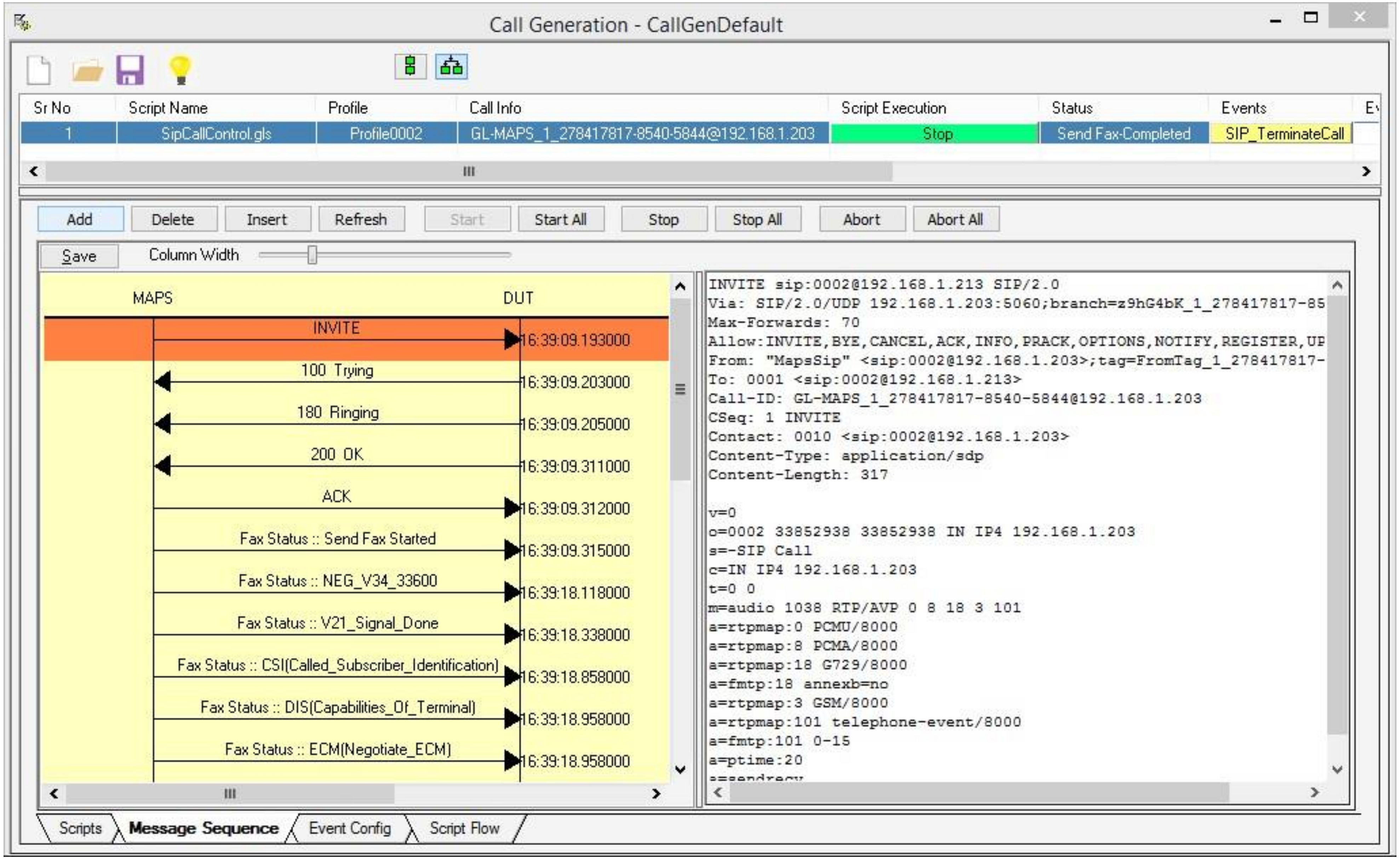
Call Generation with T.30 Pass through FAX Traffic Type



RTP Video Traffic Capabilities

RTP Video Call Generation Capability

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

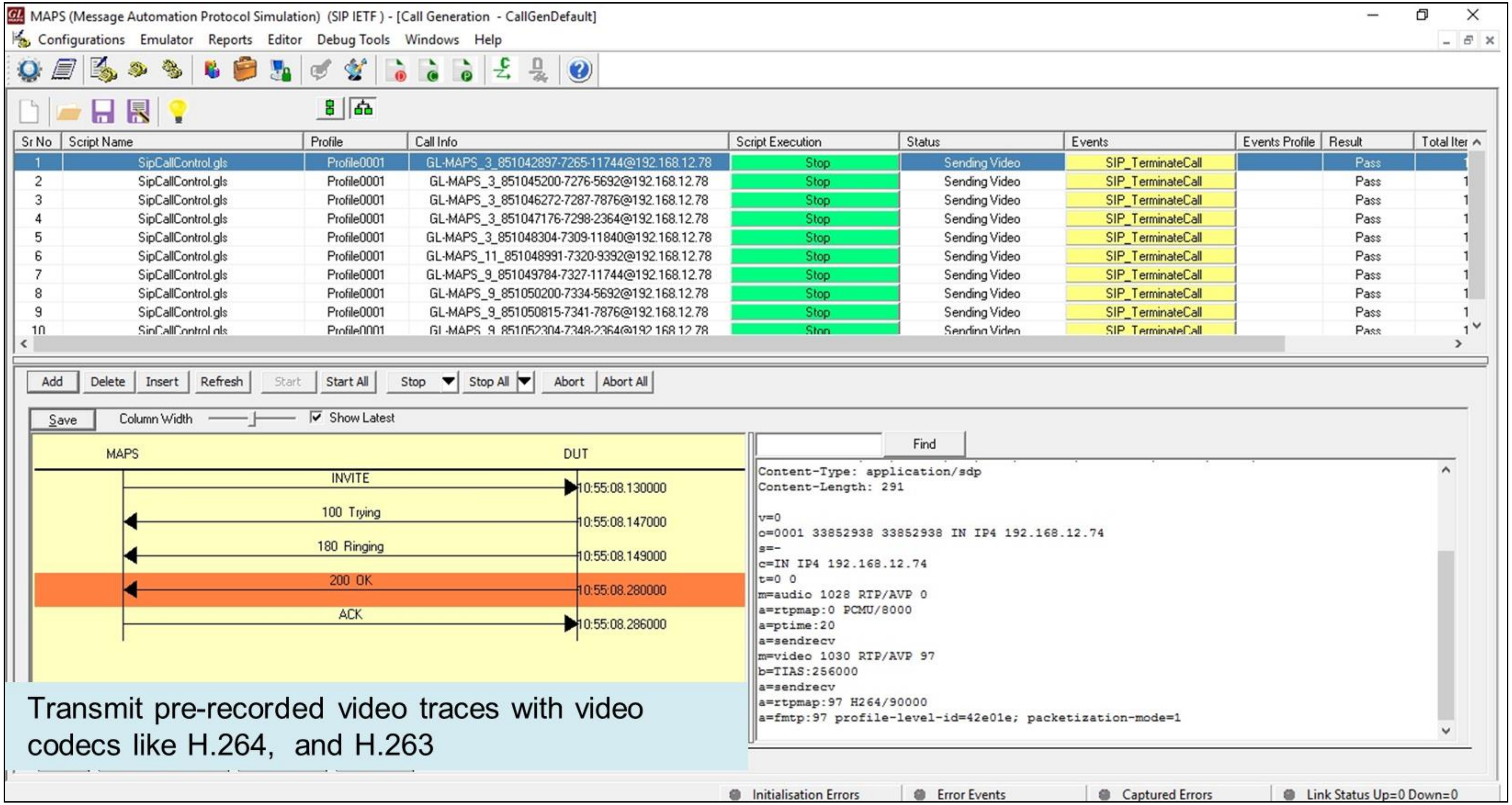
Voice Quality Statistics

- Provides statistics of CMOS and LMOS scores, Packet Lost / Discarded / Duplicates / Out-of-Sequence packets

Video Quality Metrics in PacketScan™

- Provide visibility into the captured video call, detail statistics of signaling, audio and video parameters

Bulk Video Traffic Simulation



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

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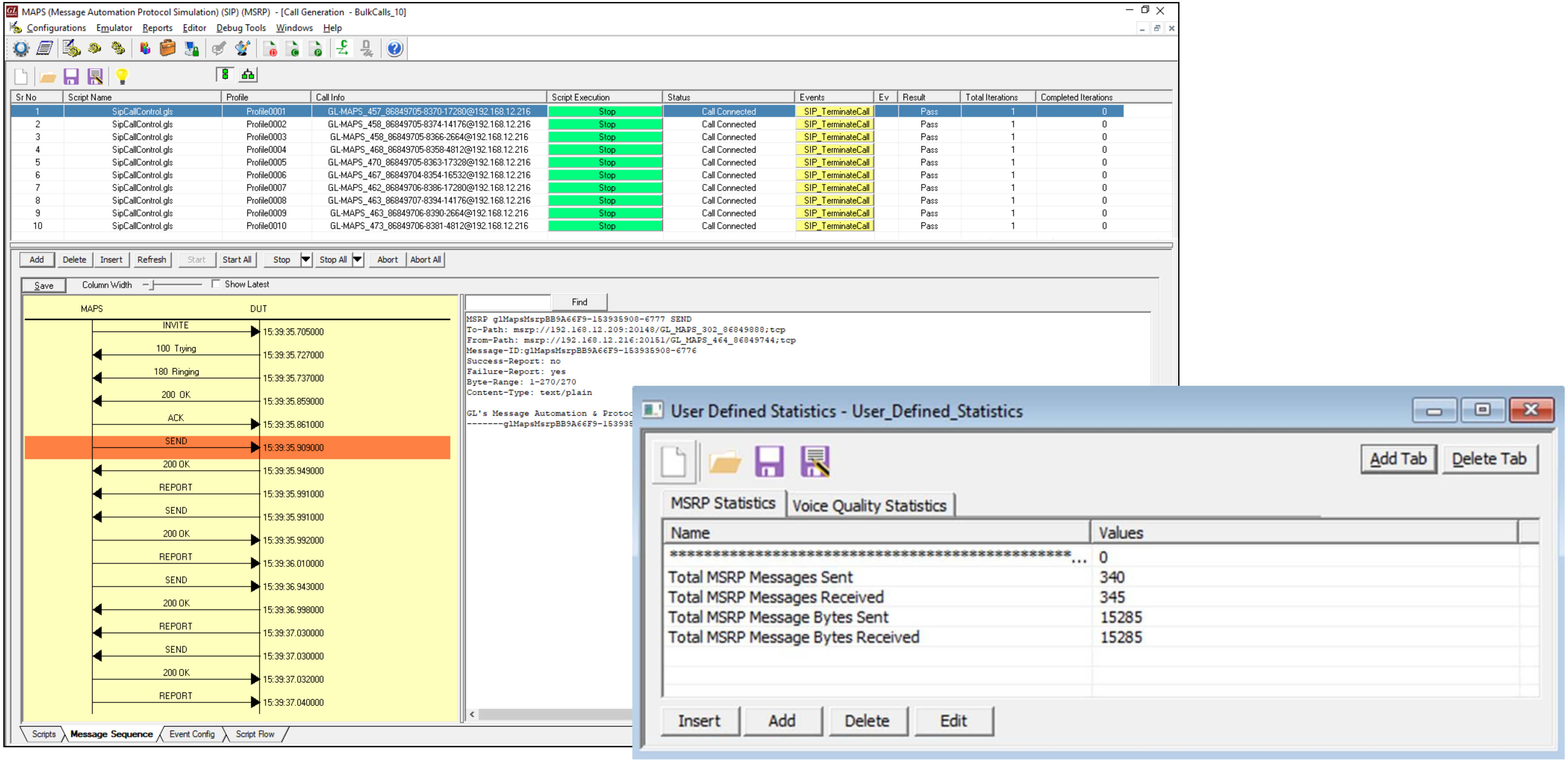
MSRP Traffic Simulation



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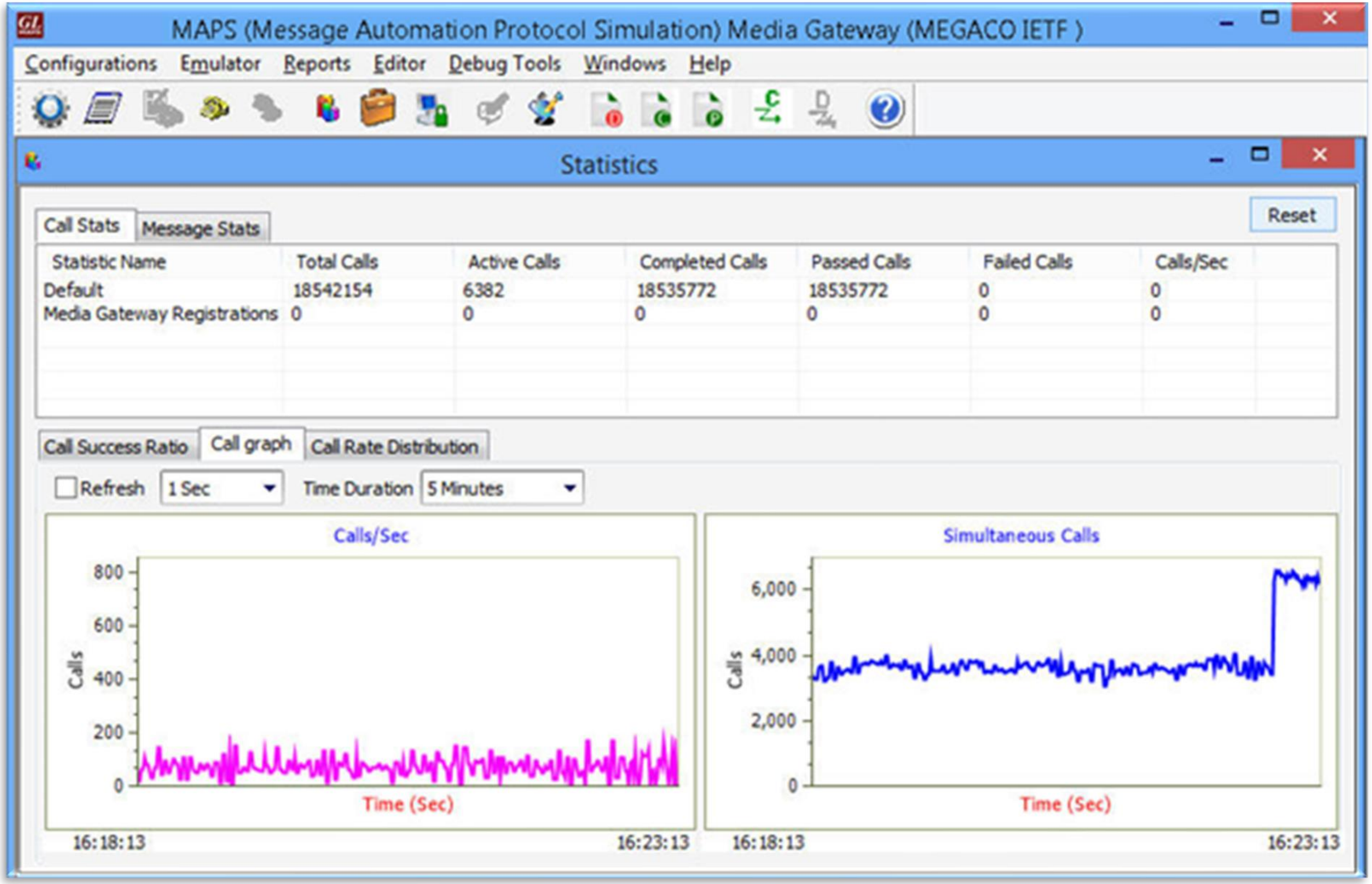
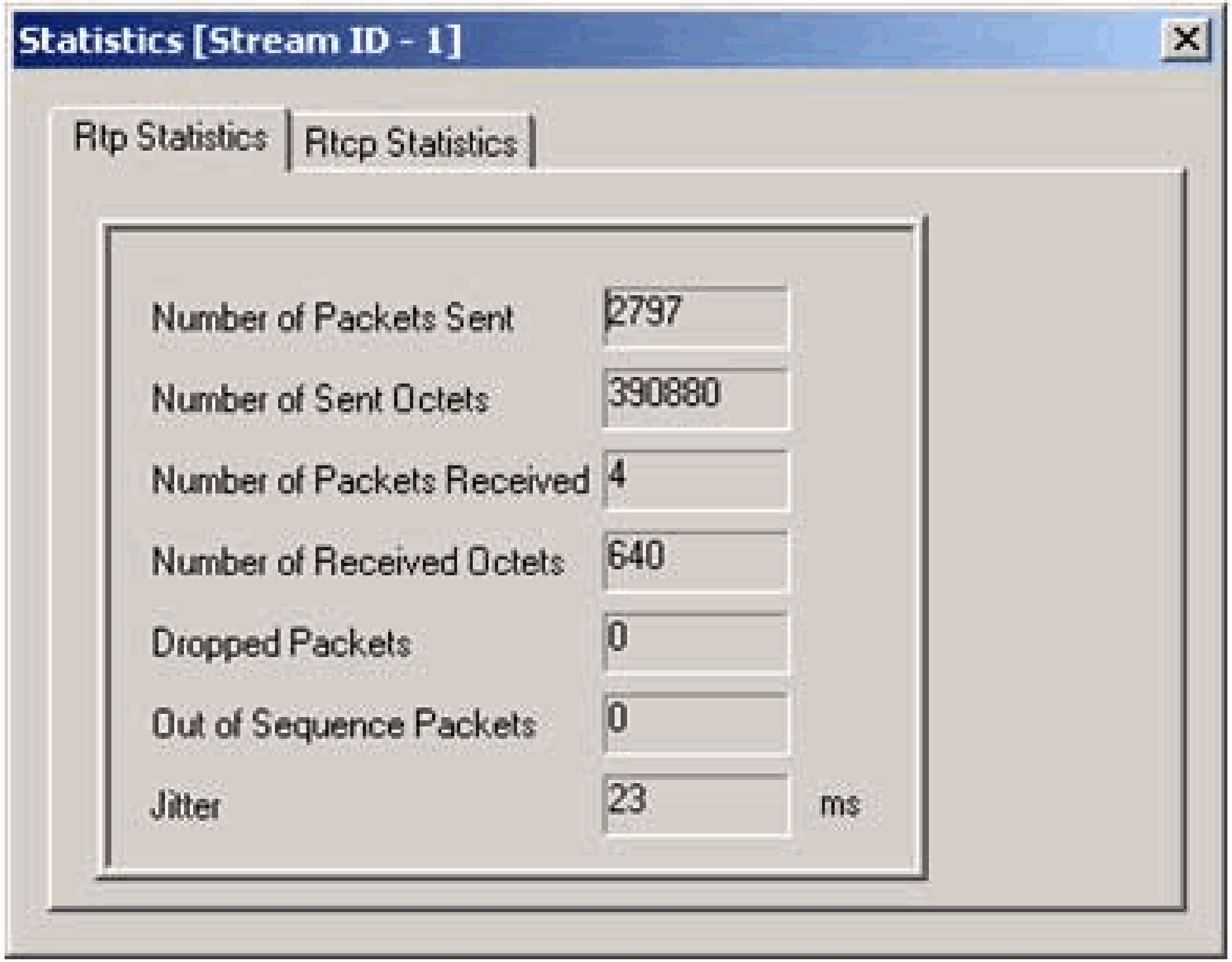
Bulk Call Simulation Results



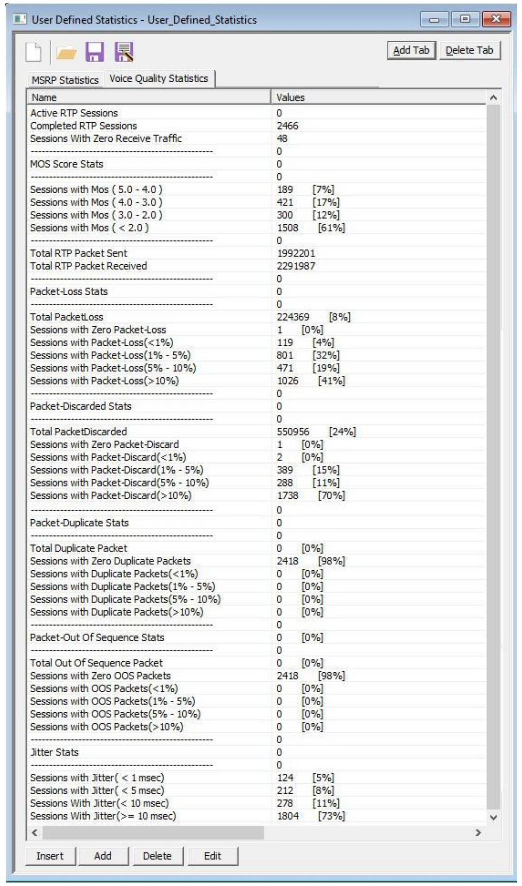
RTP/RTCP Packet Statistics

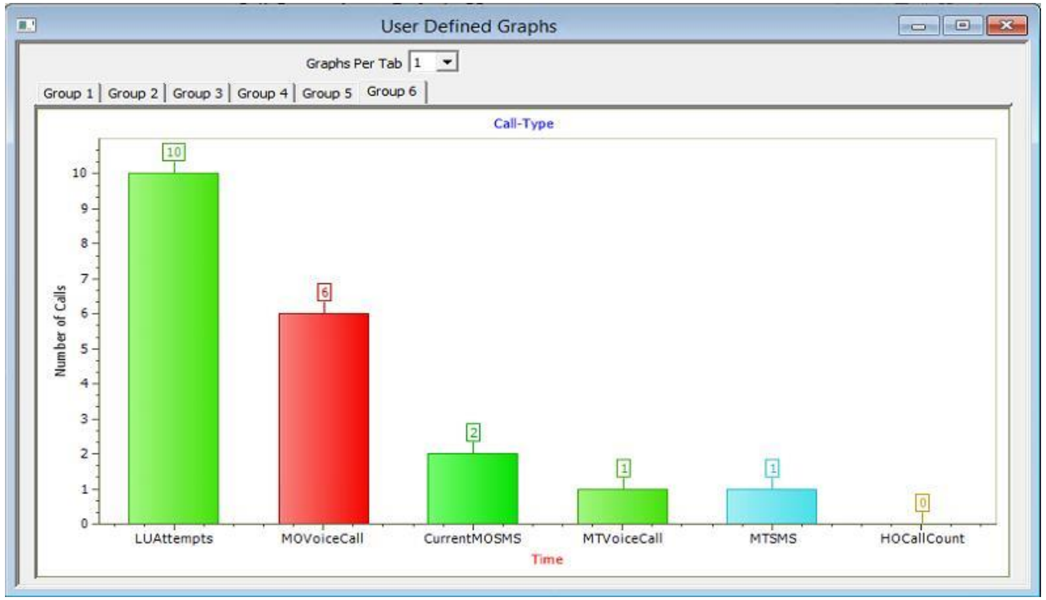
- Statistics reports of RTP and RTCP packets transmitted on a session such as number of packets sent/received, dropped packets, out of sequence packets and more. Sender and receiver reports are also displayed using RTP/RTCP statistics applications

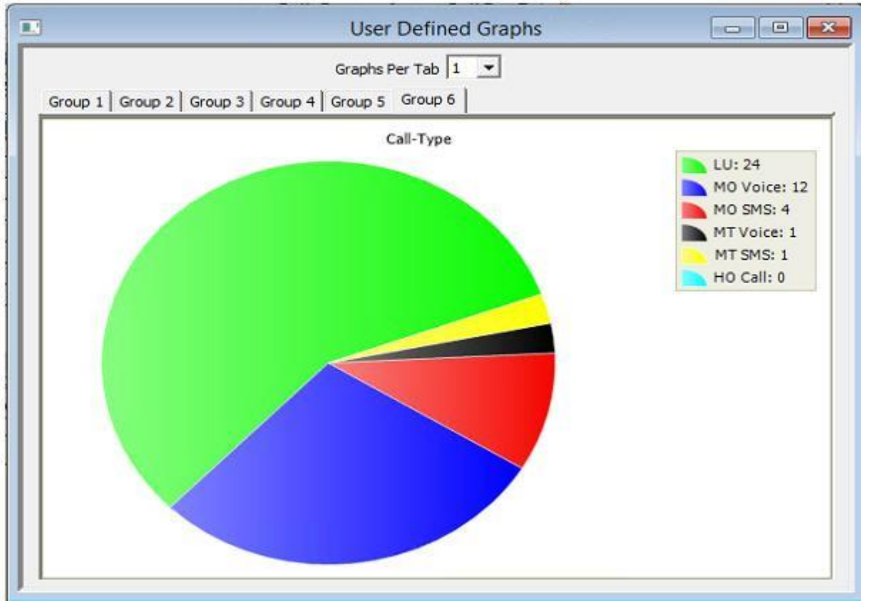


Speech Quality Metrics (R Factor and MOS)

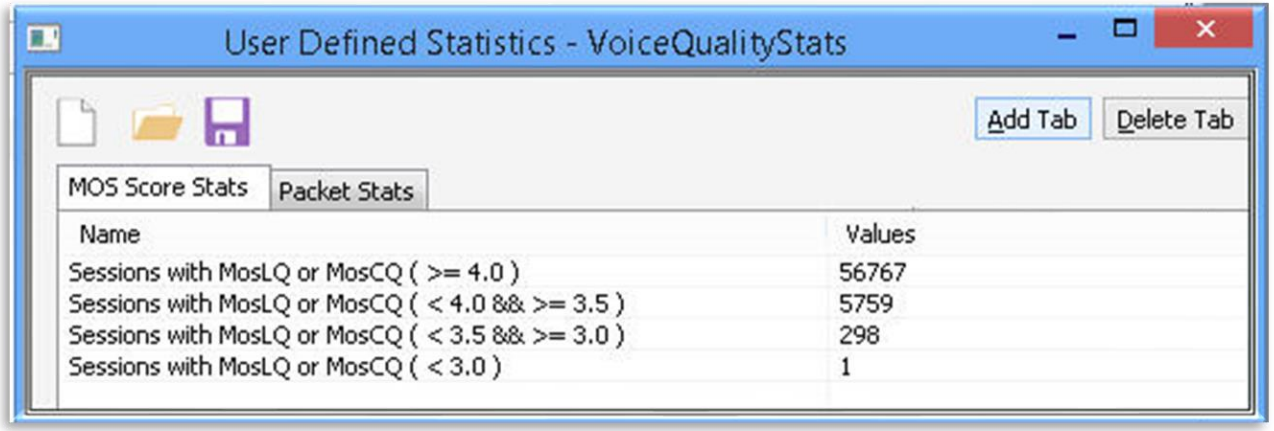
- Quality metrics include R-Factor, Listening and Conversational Quality MOS scores, Packet Loss, Discarded Packets, Out of Sequence Packets and Duplicate Packets
- R Factor graph will display statistics such as, R-Listening, R-Conversational, R-G107 and R-Nom. MOS Factor graph will display statistics such as MOS CQ, MOS PQ and MOS Nom. Estimates are based on the ITU G.107 E Model



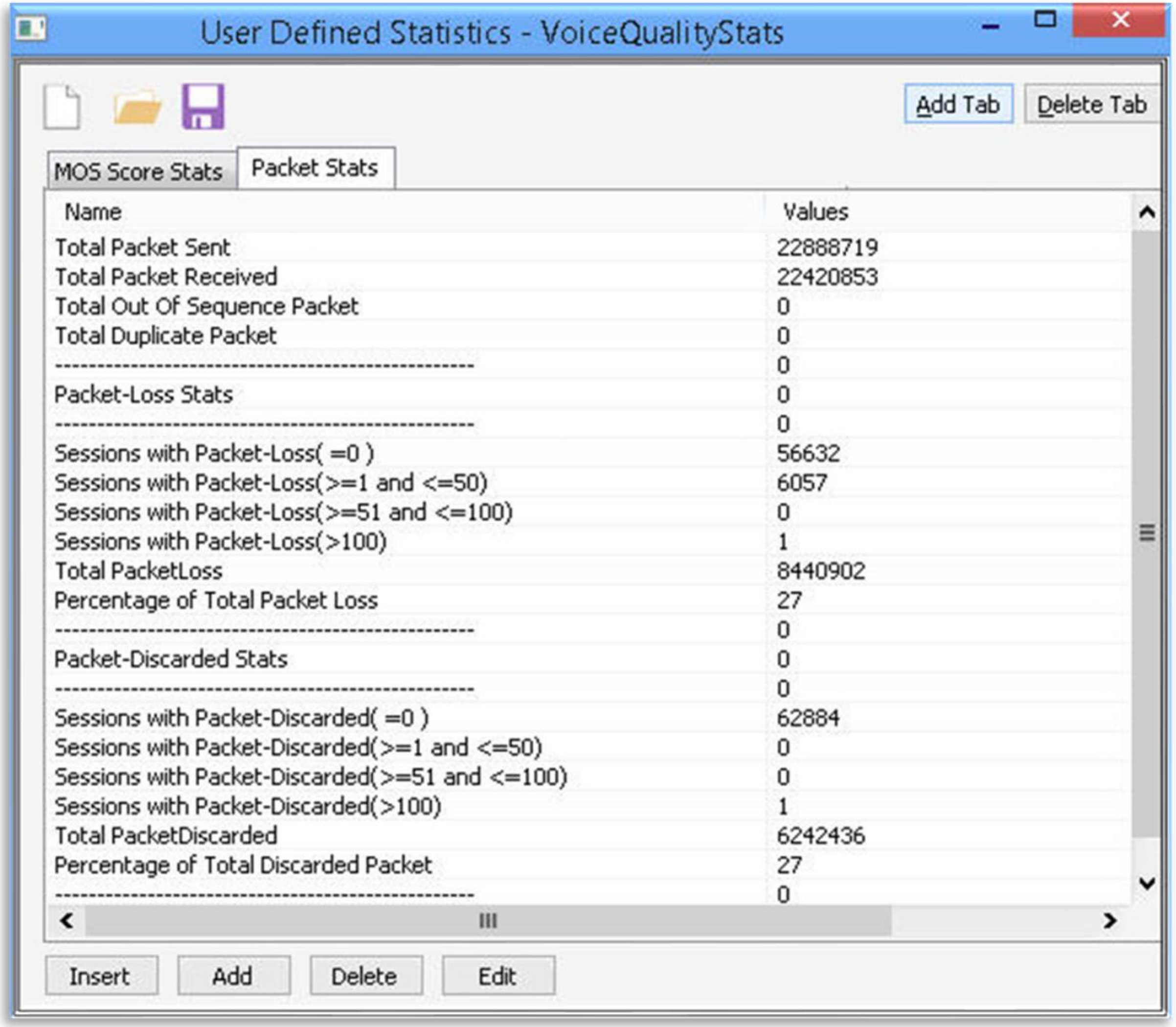




Speech Quality Metrics (R Factor and MOS) (Contd.)



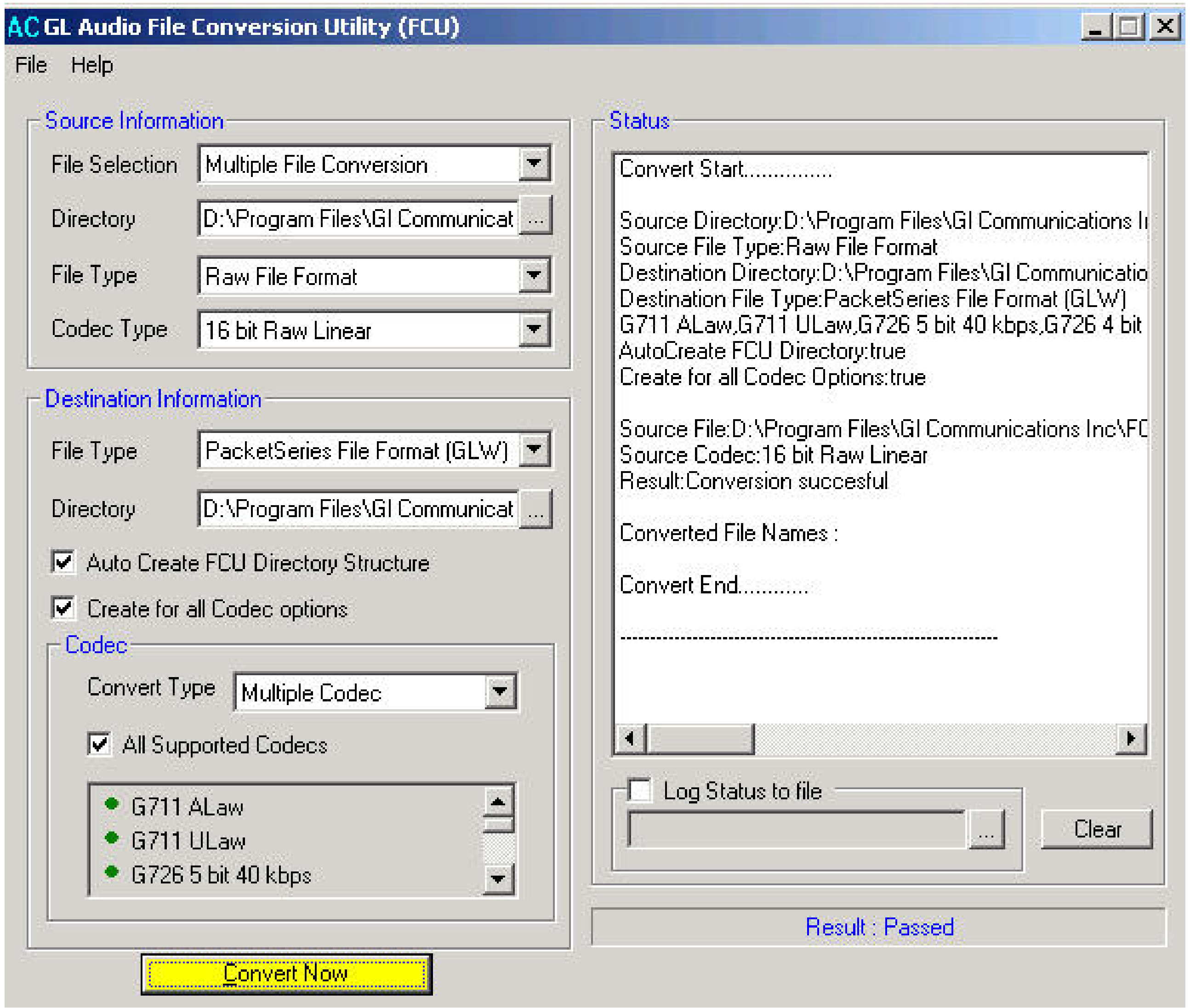
Name	Values
Sessions with MosLQ or MosCQ (>= 4.0)	56767
Sessions with MosLQ or MosCQ (< 4.0 && >= 3.5)	5759
Sessions with MosLQ or MosCQ (< 3.5 && >= 3.0)	298
Sessions with MosLQ or MosCQ (< 3.0)	1



Name	Values
Total Packet Sent	22888719
Total Packet Received	22420853
Total Out Of Sequence Packet	0
Total Duplicate Packet	0
--	0
Packet-Loss Stats	0
--	0
Sessions with Packet-Loss(=0)	56632
Sessions with Packet-Loss(>=1 and <=50)	6057
Sessions with Packet-Loss(>=51 and <=100)	0
Sessions with Packet-Loss(>100)	1
Total PacketLoss	8440902
Percentage of Total Packet Loss	27
--	0
Packet-Discarded Stats	0
--	0
Sessions with Packet-Discarded(=0)	62884
Sessions with Packet-Discarded(>=1 and <=50)	0
Sessions with Packet-Discarded(>=51 and <=100)	0
Sessions with Packet-Discarded(>100)	1
Total PacketDiscarded	6242436
Percentage of Total Discarded Packet	27
--	0

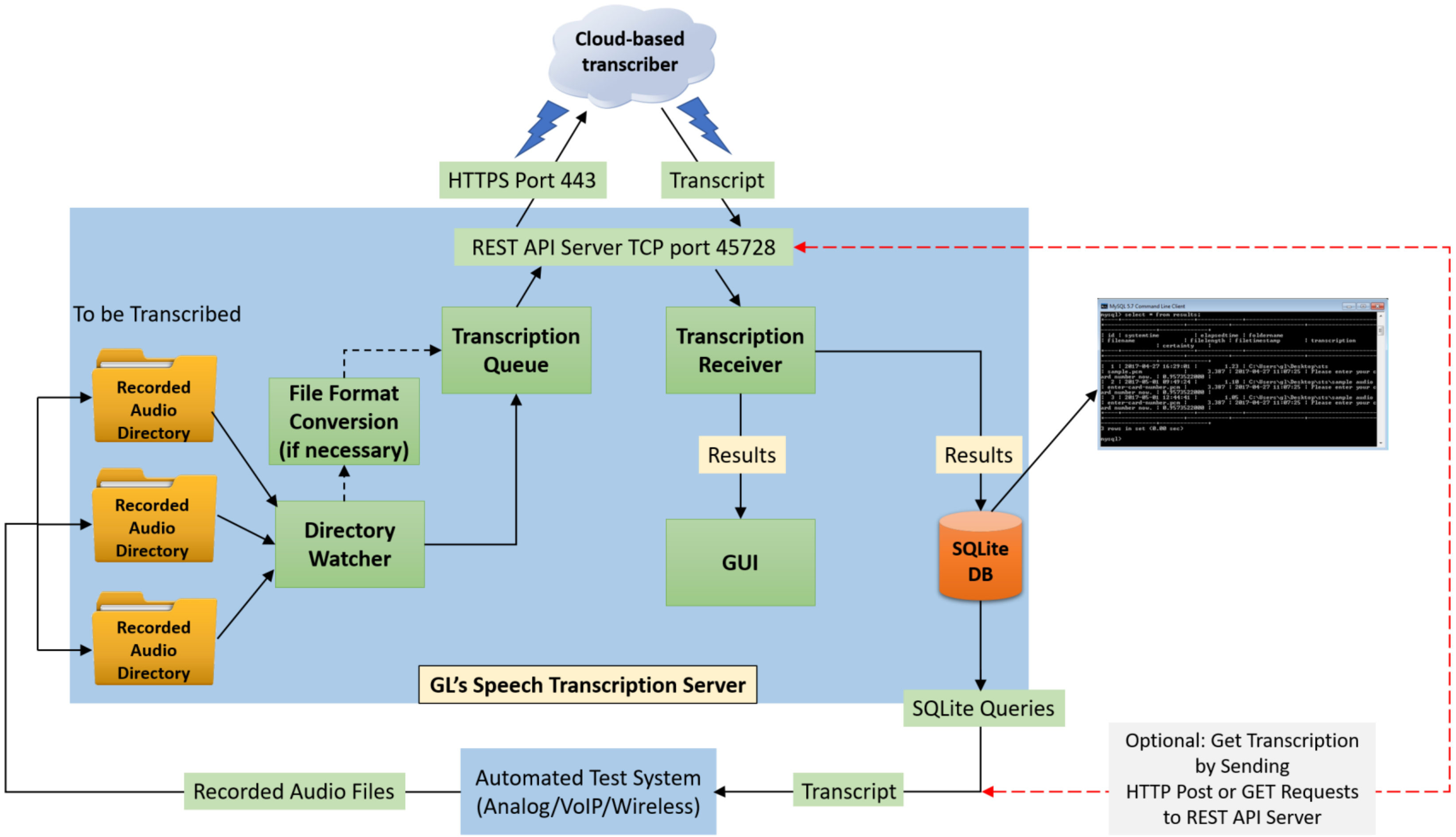
Audio File Converter Utility (AFCU)

- GL's Audio File Conversion Utility (Audio FCU) is generally used in conjunction with GL Packet Series products to further enhance send and record voice file capabilities
- This utility supports almost all industry standard voice codec data formats, that helps to convert recorded voice files from their native codec format to a GL standard format



Speech Transcription Server

- The Speech Transcription Server can be used for confirming voice prompts (announcements) and aid in testing Interactive Voice Response (IVR) systems as well as voice transportation over any network
- Network providers use the application to record the voice prompts associated with the IVR, perform a Speech to Text conversion on the recording to confirm the prompt was proper (based on what the prompt should be), and thus confirming their IVR functioning



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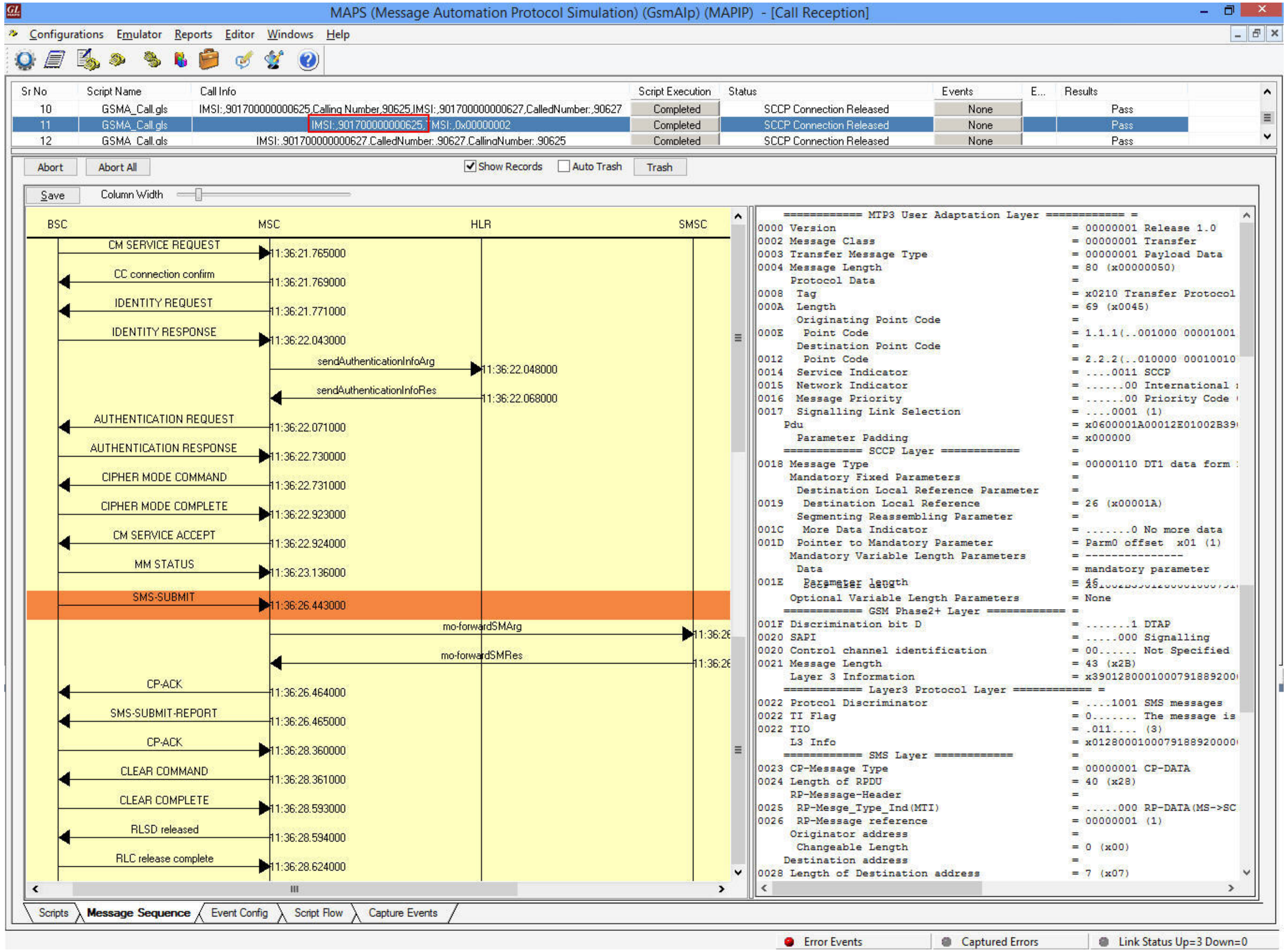

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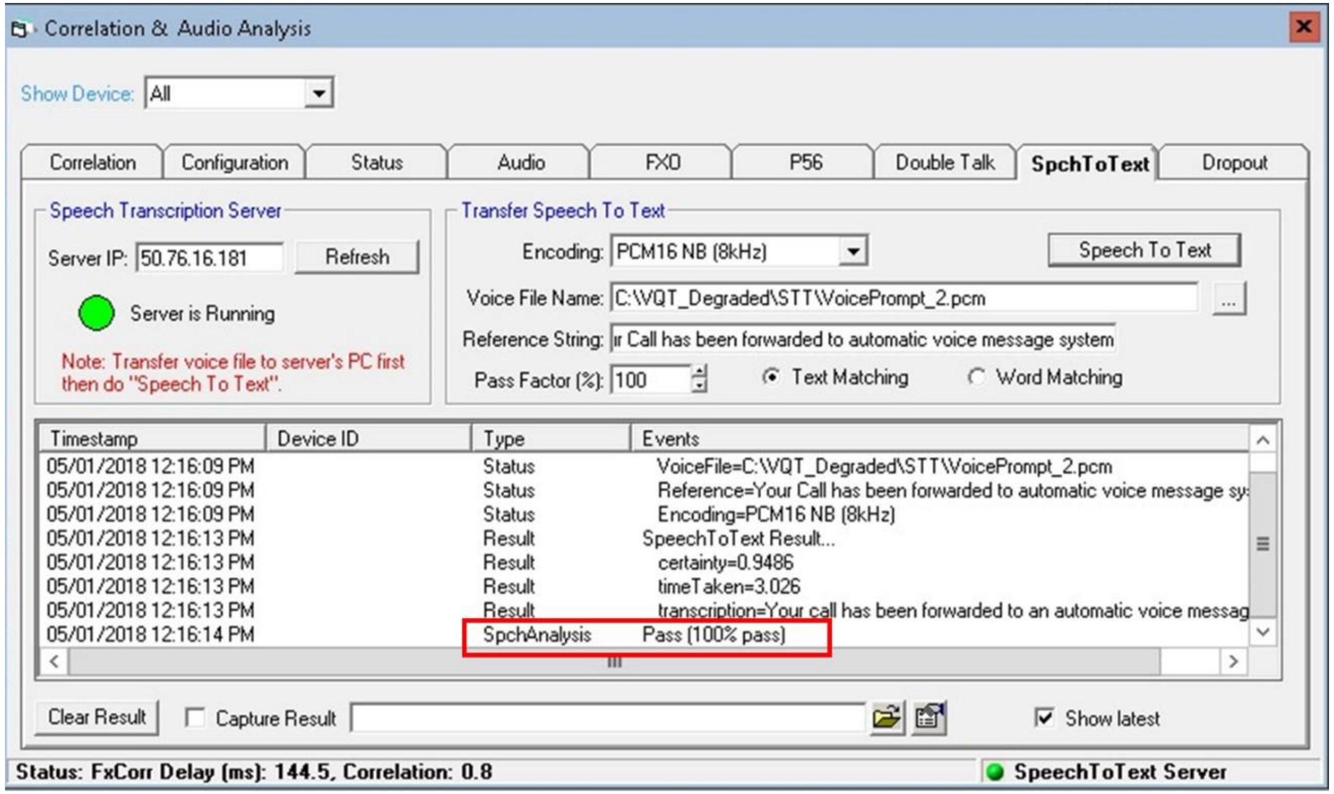
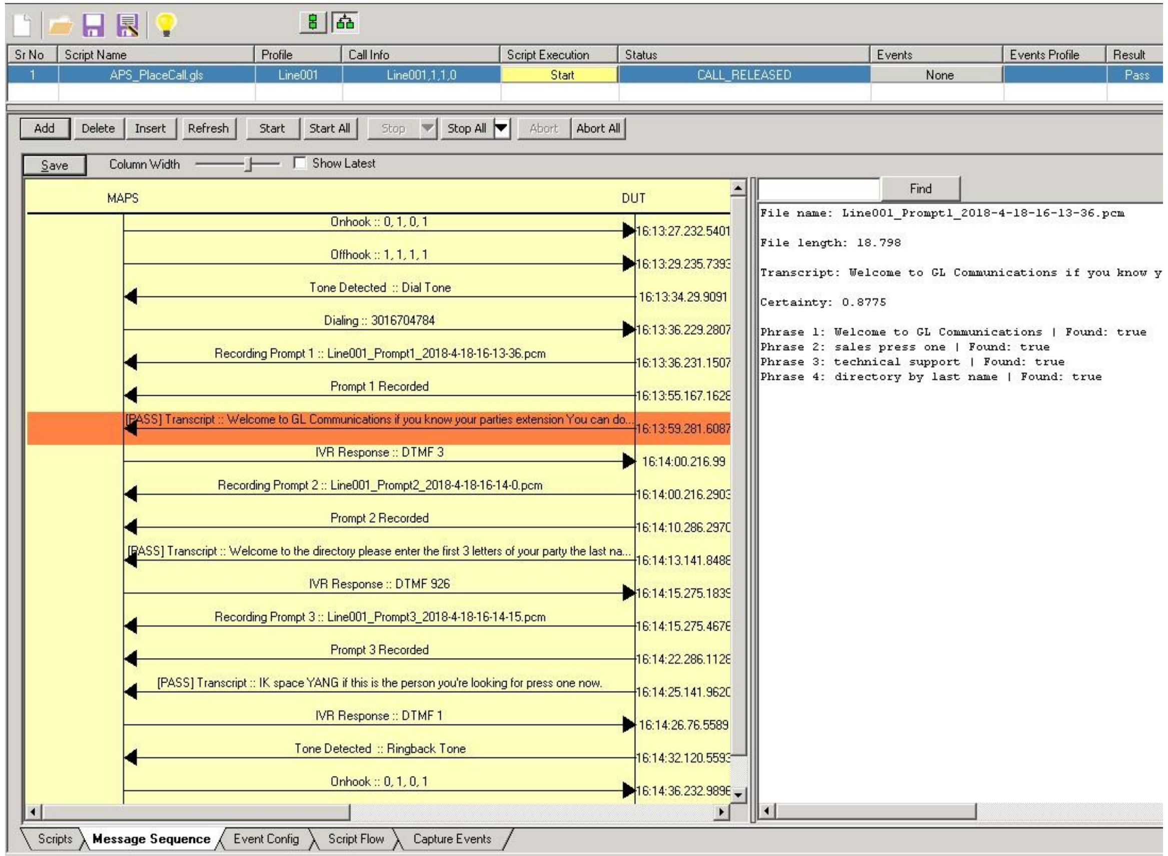
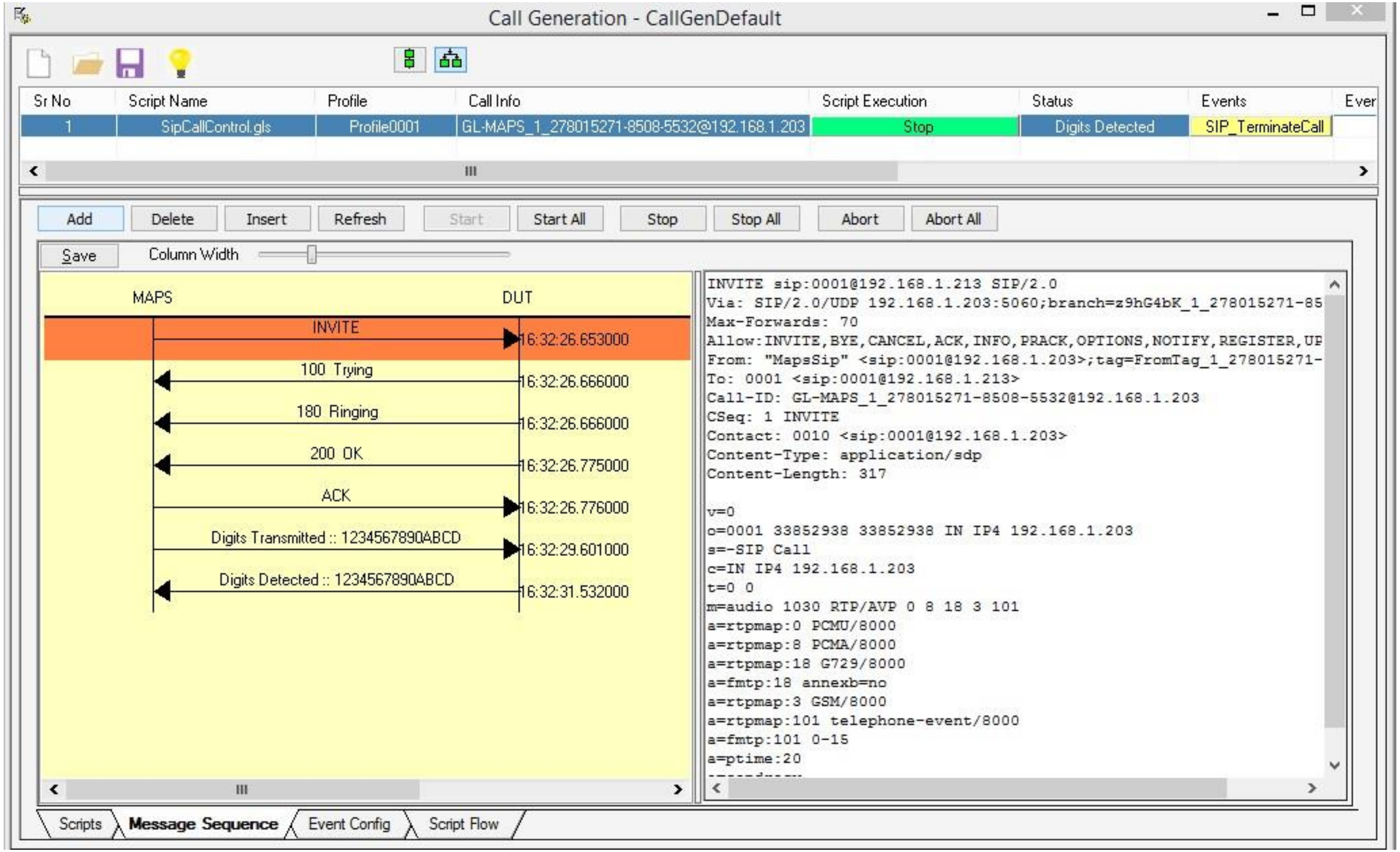




Call Generation with IVR Traffic Type



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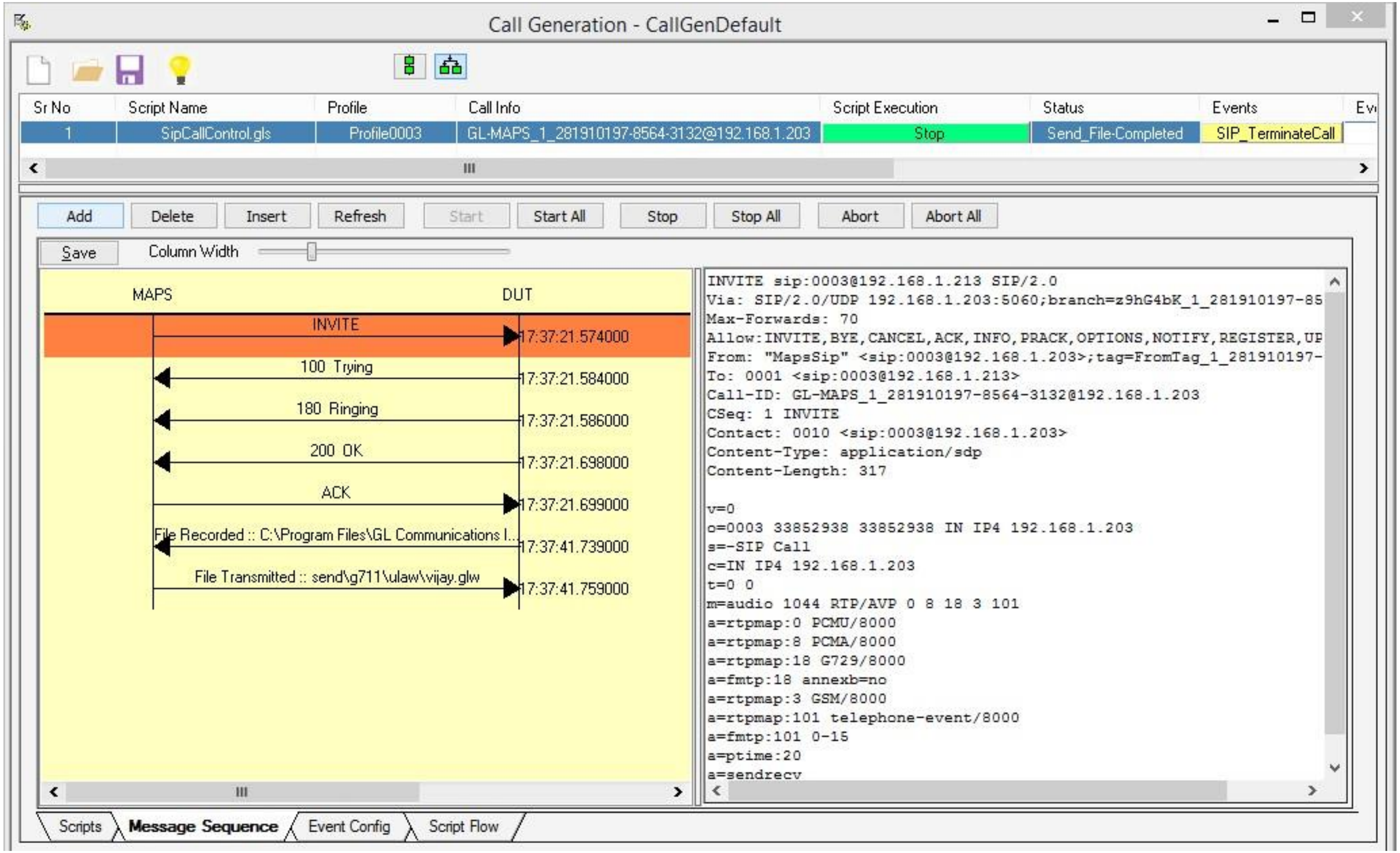
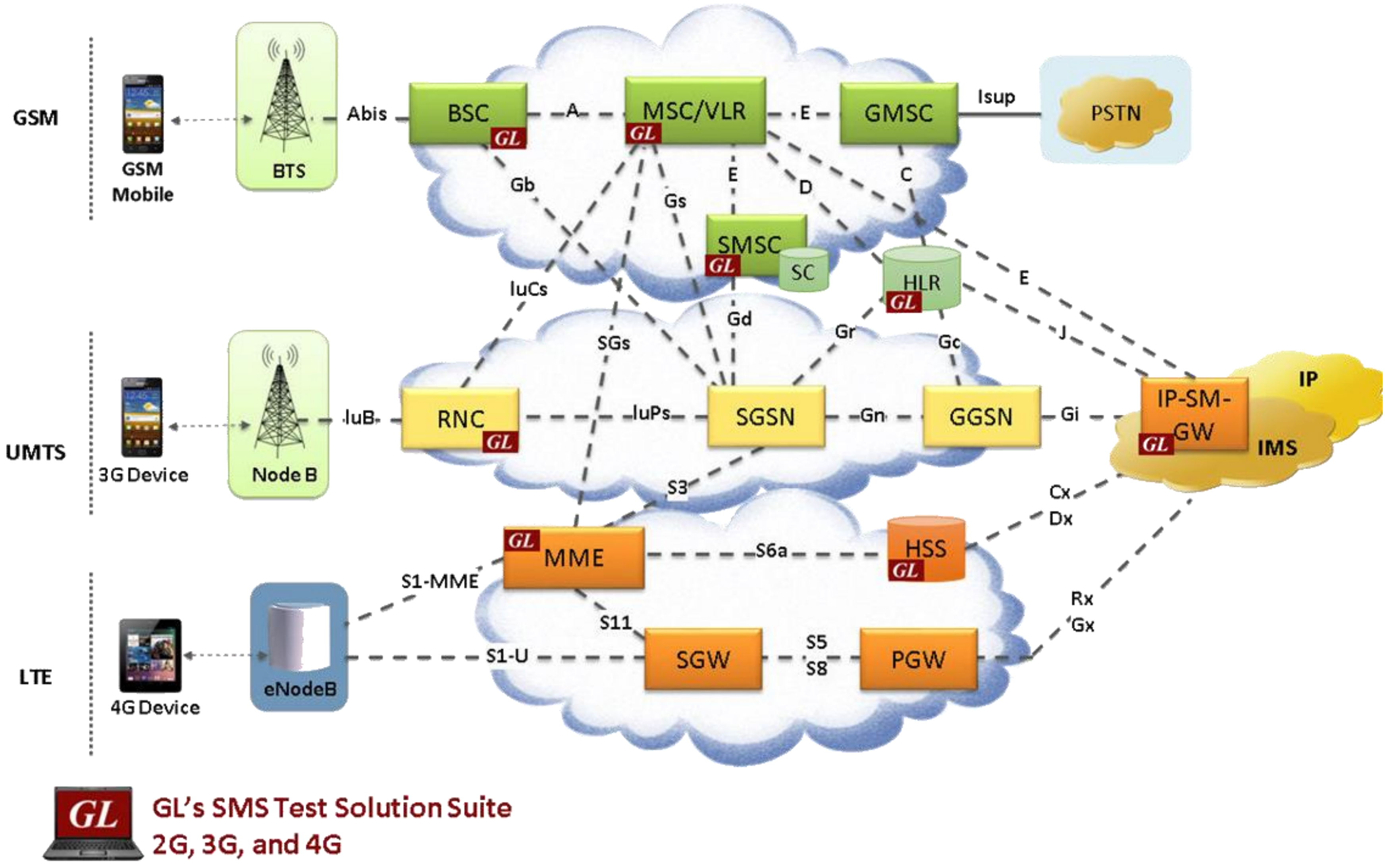
Call Generation with File Traffic Type



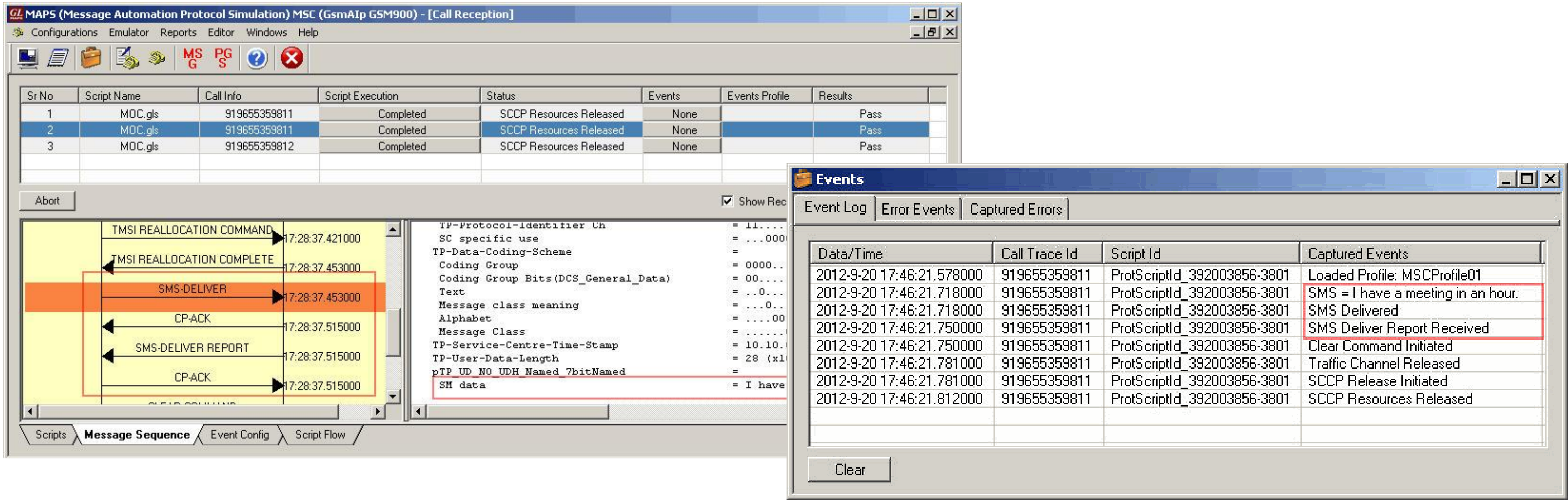
GL Communications

Short Message Service (SMS) Test Solutions



Short Message Service (SMS) Test Solutions

- MAPS™ supports testing following SMS types:
  - **Short message Mobile Terminated (SMS-MT):** It is the ability of a network to transmit a Short Message to a mobile phone. The message can be sent by phone or by a software application
  - **Short message Mobile Originated (SMS-MO):** It is the ability of a network to transmit a Short Message sent by a mobile phone. The message can be sent to a phone or to a software application
- MAPS™ for GSMAoIP, MAP, MAPIP supports short message service simulation



GL Communications

IP Traffic Commands

- Create RTP Session
  TxRx:create_session (MediaIPAddress, MediaPort);

- Start Session
  TxRx: start_session(PeerMediaIPAddress,PeerMediaPort Codec Payload,Packetizationtime);
- Send Actions
  - ➢ Send Digits
    TxRx: tx _Rtp digits: digittype = dtmf, digits = "1234567890ABCD*#", band = inband, power1 = - 6, power2 = - 4, ontime = 80, offtime = 80;

  - ➢ Send File
    TxRx: tx _Rtp file: filename = “\Send\G711\ULAW\Vijay.glw", duration = 30 sec;

  - ➢ Send Tones
    TxRx:tx _Rtp tone : freq1 = 1004, power1 = -6, freq2 = 2004, power2 = -4, ontime = 80, offtime = 80, iterations = 10;

  - ➢ Transmit RTP Speech
    TxRx: tx _Rtp speech;

  - ➢ RTP Loopback
    TxRx: loopback _Rtp;

IP Traffic Commands (Contd.)

- Receive Actions
  - Monitor Digits

    TxRx: monitor _Rtp digits: band = inband, digittype = mf;
  - Record Files

    TxRx: rx _Rtp file: filename = "C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_1.glw", duration = 30 sec;
  - Monitor Tones

    TxRx: monitor _Rtp tones : freq1 = 1000, freq2 = 2000;
  - Play RTP Speech

    TxRx: play _Rtp speech;
- Stop All RTP Transmission and Reception

  TxRx:stop _Rtp tx * ;

  TxRx:stop _Rtp rx * ;
- Stop Session

  TxRx:stopsession;

- Send and Receive Actions
  - Transmit Fax

    TxRx:SendFax(TxMinDataRate,TxMaxDataRate,TxFaxFileName);
  - Receive Fax

    TxRx:RecvFax(RxMinDataRate,RxMaxDataRate,RxFaxFilename);

Sample RTP Traffic Script

```
TxRx:create_session (MediaIPAddress,MediaPort);
TxRx:start_session(PeerMediaIPAddress,PeerMediaPort
Codec Payload,Packetizationtime);
ActiveUserEvent: "Talk","Stop Traffic";
 wait;
 "Talk":
        if(State == "CALL ESTABLISHED")
                //Tx Speech Action
                TxRx:tx _Rtp speech;
                ActiveUserEvent: "Listen","Stop Tx";
        endif
resume;
 "Stop Traffic":
TxRx:stop _Rtp tx * ;
TxRx:stop _Rtp rx * ;
TxRx:stopsession;
 exit;
```

GL Communications

GTP Traffic Simulation

GTP Traffic Mobile Options

GTP Mobile Traffic Options	Licenses
PacketCheck™	ETH100
Mobile IP Traffic Core (< 1Gbps) – PacketLoad - HD Mobile IP Traffic Core (up to 4 Gbps)	ETH101
Mobile IP Traffic Core - Gateway	ETH102
Mobile Traffic Core - Gb	ETH103

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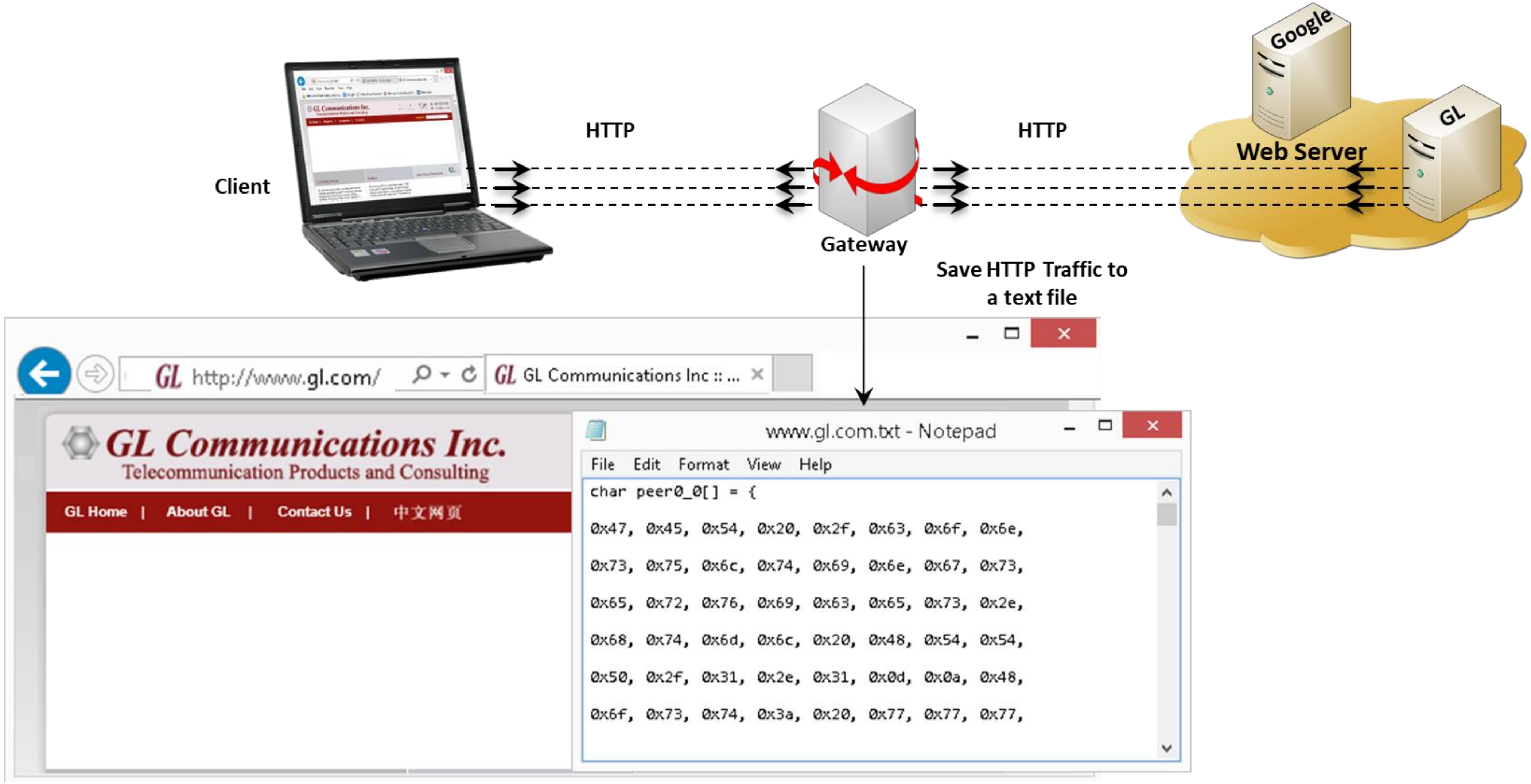
Mobile Traffic Simulation

- **Packet Traffic Simulation - GTP (ETH100)**
  - Supports stateless end-to-end data generation and verification at the other end over GTP (GPRS Tunnelling Protocol)
  - The IP traffic can be generated as Sequence Number, Hex string, BER patterns, or playback captured Ethernet traffic (*.HDL) files
- **Mobile Traffic Core - GTP (ETH101)**
  - Supports stateful user-plane packet simulation services between any two nodes (GTP-U protocol entity) in UMTS (SGSN, GGSN, RNC), and LTE (SGW, PDNGW) networks
  - It allows simultaneous simulation of multiple sessions per user. Currently, supports HTTP traffic simulation with the base requirements such as port number, server IP address, and pre-canned HTTP traffic file
  - Each GTP traffic is identified by a Tunnel ID, UDP port (2152 is default for GTP-U traffic), and the multiple HTTP connections are differentiated by Connection ID
  - Also supports generation and verification of data traffic such as Email, FTP, HTTP, and more

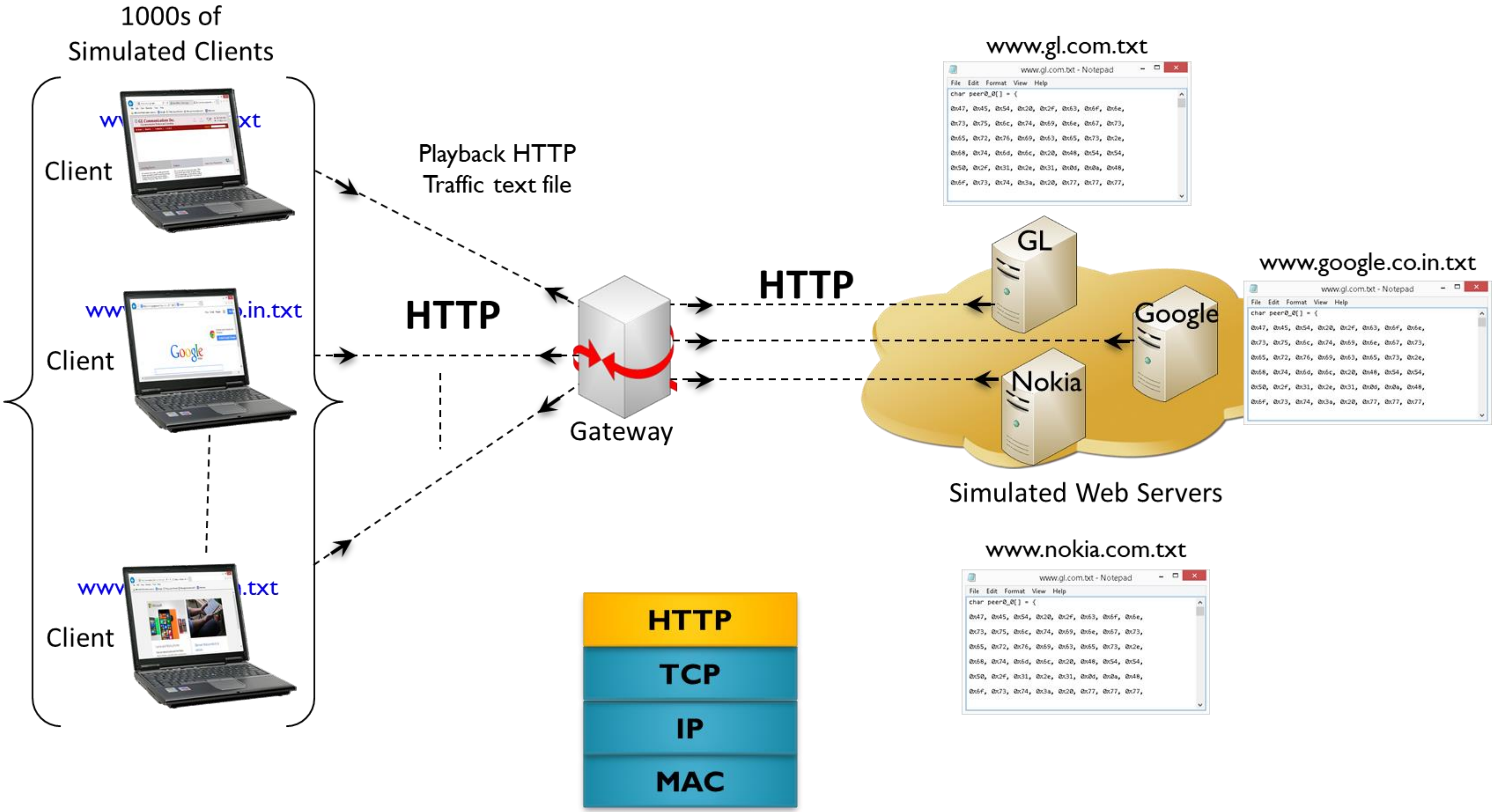
Mobile Traffic Simulation (Contd.)

- Mobile Traffic Core – Gateway (ETH102)
  - Supports simulation of Gateway and transfer user plane data from RNC to GGSN
  - Handles GTP tunnels on both direction of SGSN. It can also act as GGSN for user-plane traffic by encapsulating IP traffic over GTP
  - This module is supported with MAPS™ GnGp, MAPS™ LTE S1, MAPS™ LTE eGTP-c
- Mobile Traffic Simulation - GPRS Gb (ETH103)
  - Supports simulation of Mobile traffic over Gb interface between BSC and SGSN
  - Transmits the pre-canned HTTP file (*.txt) between BSC and SGSN nodes. It multiplexes both signaling and traffic over Gb interface.
  - This module is supported in MAPS™ GPRS Gb

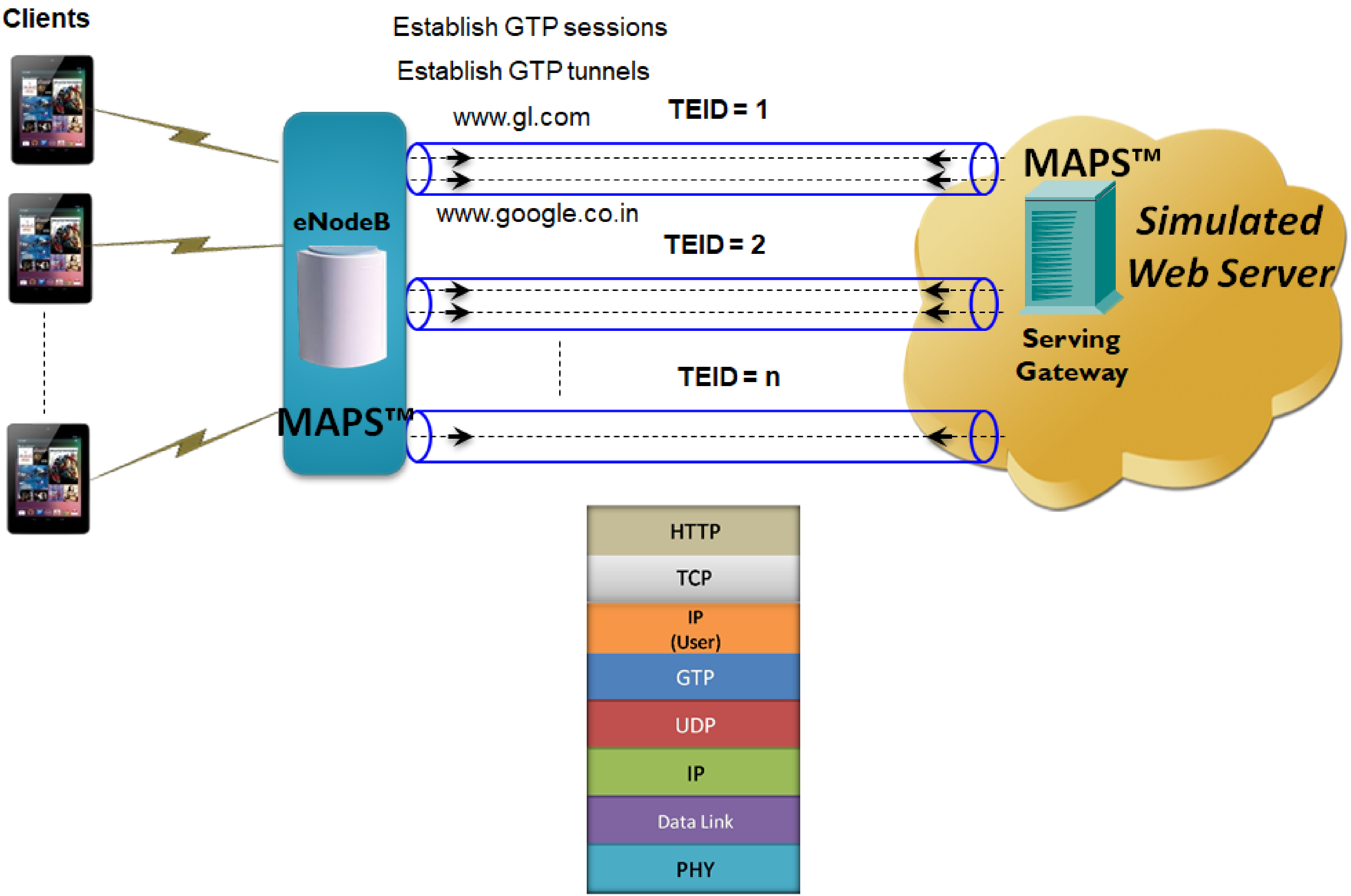
Mobile IP Core Server



Mobile Traffic over IP

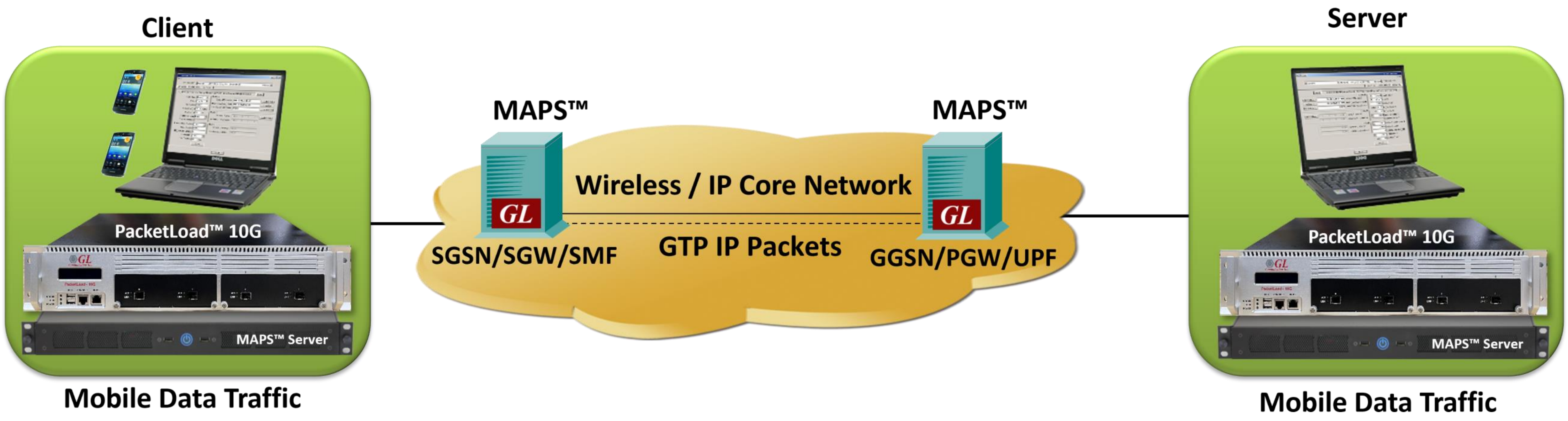


Mobile Traffic over GTP



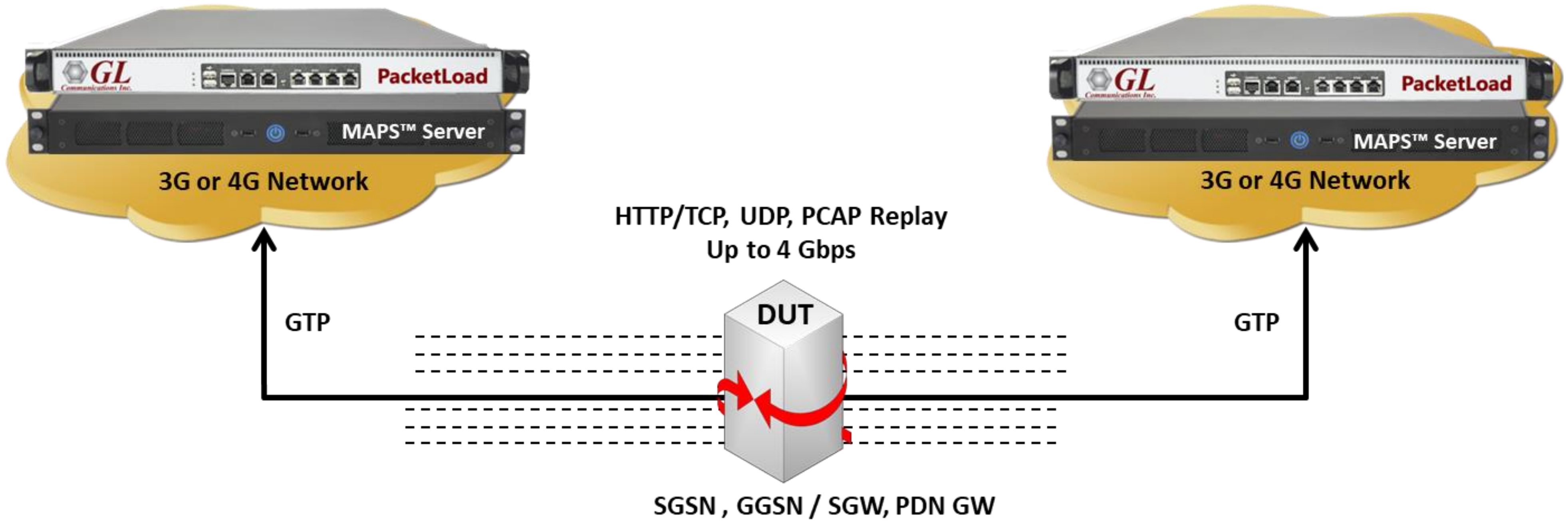
User Plane Traffic Simulation for LTE, UMTS, and GPRS Networks

- GTP Traffic Simulation modules within MAPS™ supports user-plane traffic simulation in LTE, UMTS, and GPRS networks
- The solution includes Mobile IP traffic (ETH101 / PacketLoad™), Packet traffic (BERT, Hex String, and HDL File) and Mobile Gateway traffic (ETH102) types of traffic simulation. These modules also support generation and verification of data traffic such as Email, FTP, Web (HTTP), Video, and more

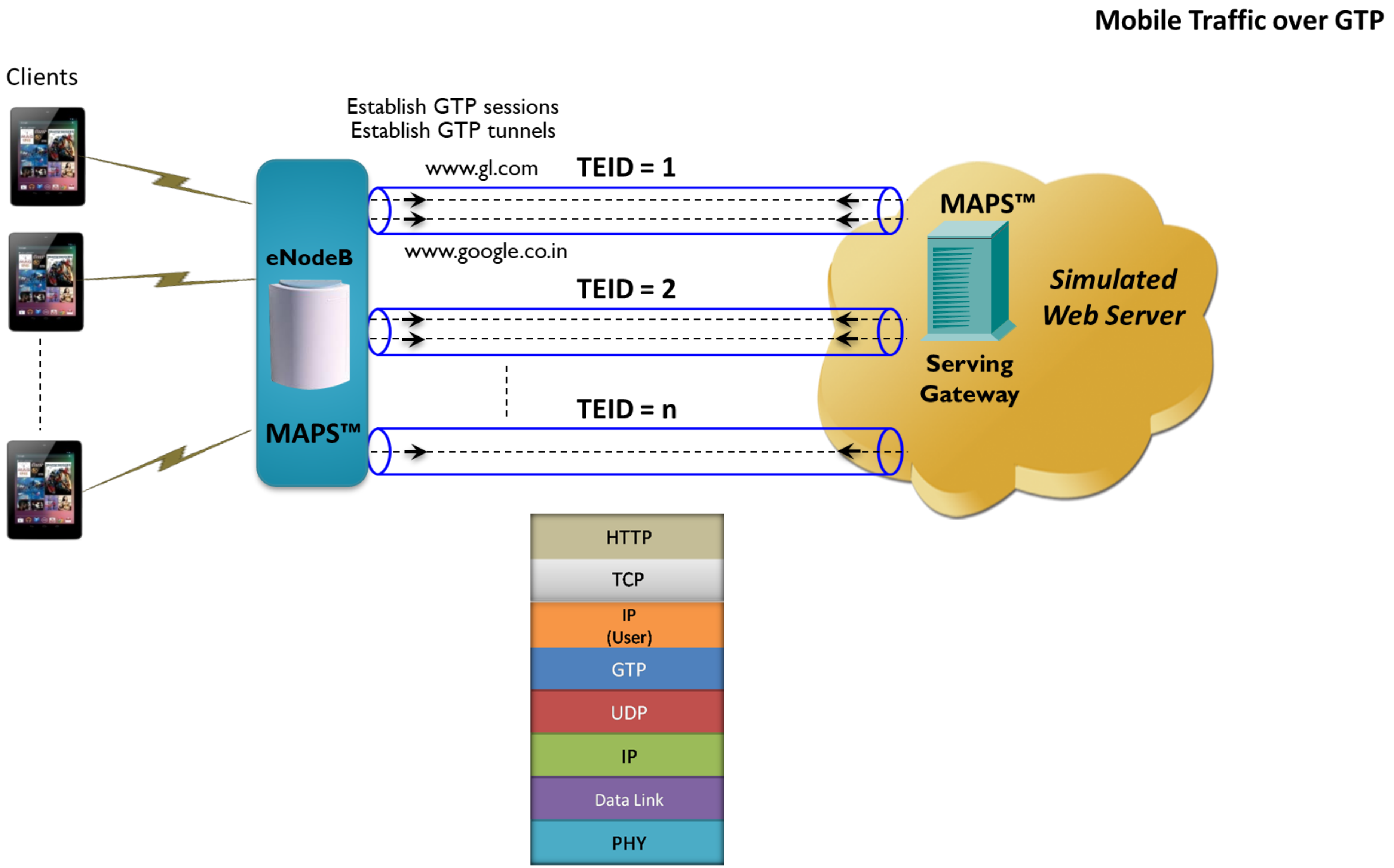


High Density Mobile Traffic Core - GTP (PacketLoad™)

- Provides high density (up to 4 Gbps) stateful TCP/HTTP, UDP, and PCAP Replay traffic simulation solution
- Supported in UMTS (SGSN, GGSN, RNC), and LTE (SGW, PDNGW) networks
- PacketLoad™ is a 1U network appliance includes 4 x 1GigE ports supporting total capacity of up to 4 Gbits/sec stateful packet traffic generation

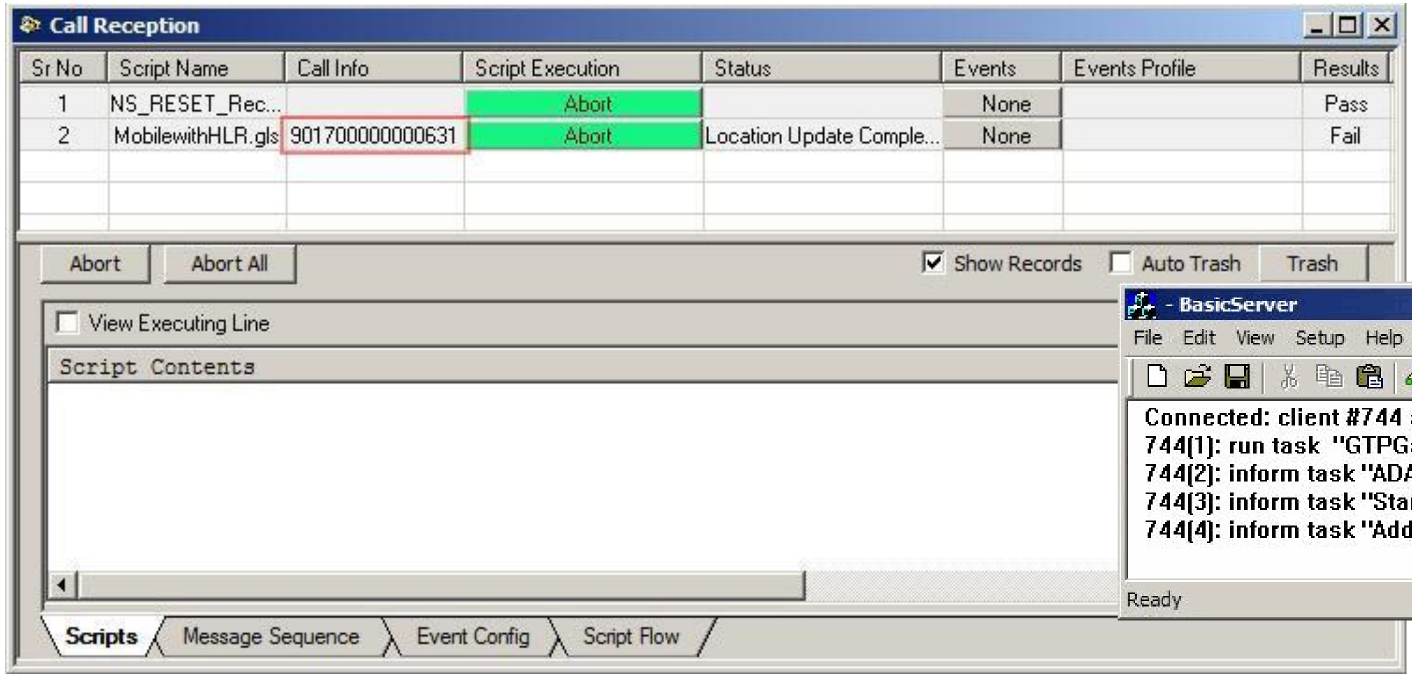


Mobile Traffic Simulation – ETH101

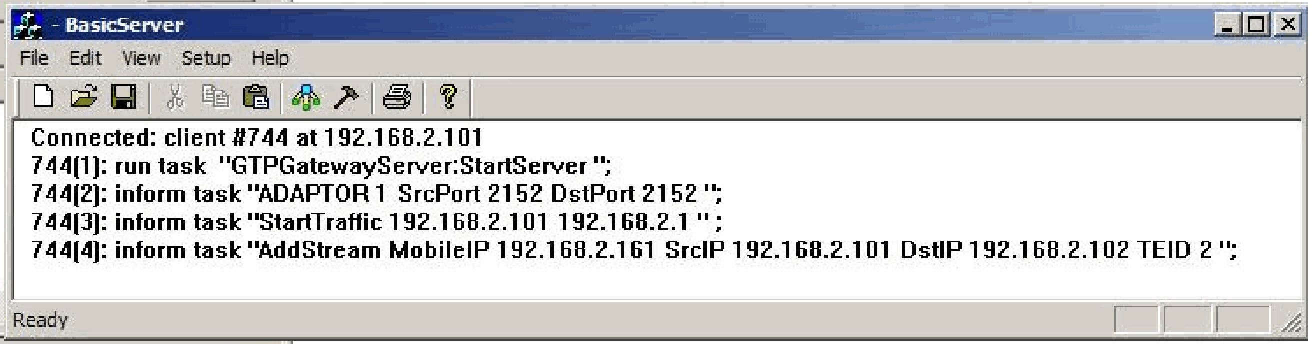


Mobile Traffic Simulation – ETH101 (Contd.)

SGSN Call Reception



GGSN Server Log



- Notice the **Internet** indication for iPhone and Android
- Notice the **SIM IMSI number** for each connected phone logged in **SGSN Call Reception** window
- **GGSN Server** entry for each new phone connected and IP address of each phone log
- Browse a website using the phone browser (www.google.com is good to start)

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Mobile Traffic Simulation – ETH102

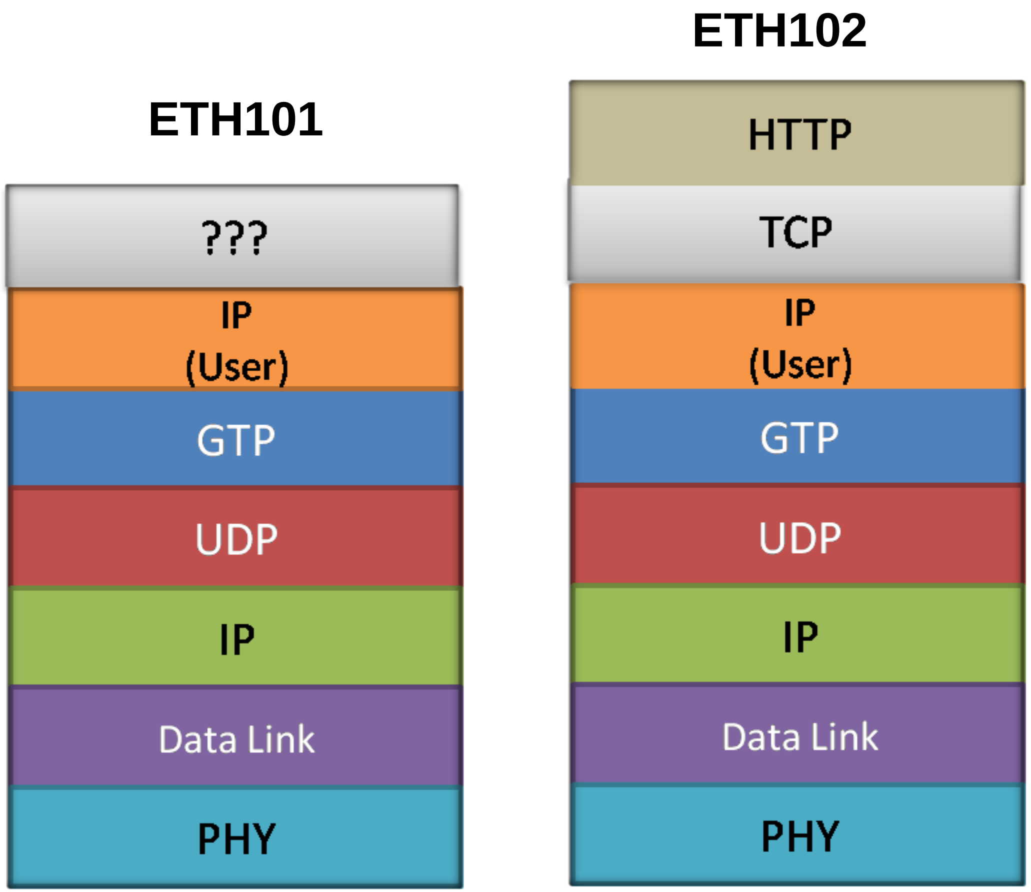
???: any IP-based protocol used to carry packet radio service

HTTP: Hyper Text Transfer Protocol

GTP: GPRS Tunnelling Protocol

UDP: User Datagram Protocol

IP: Internet Protocol



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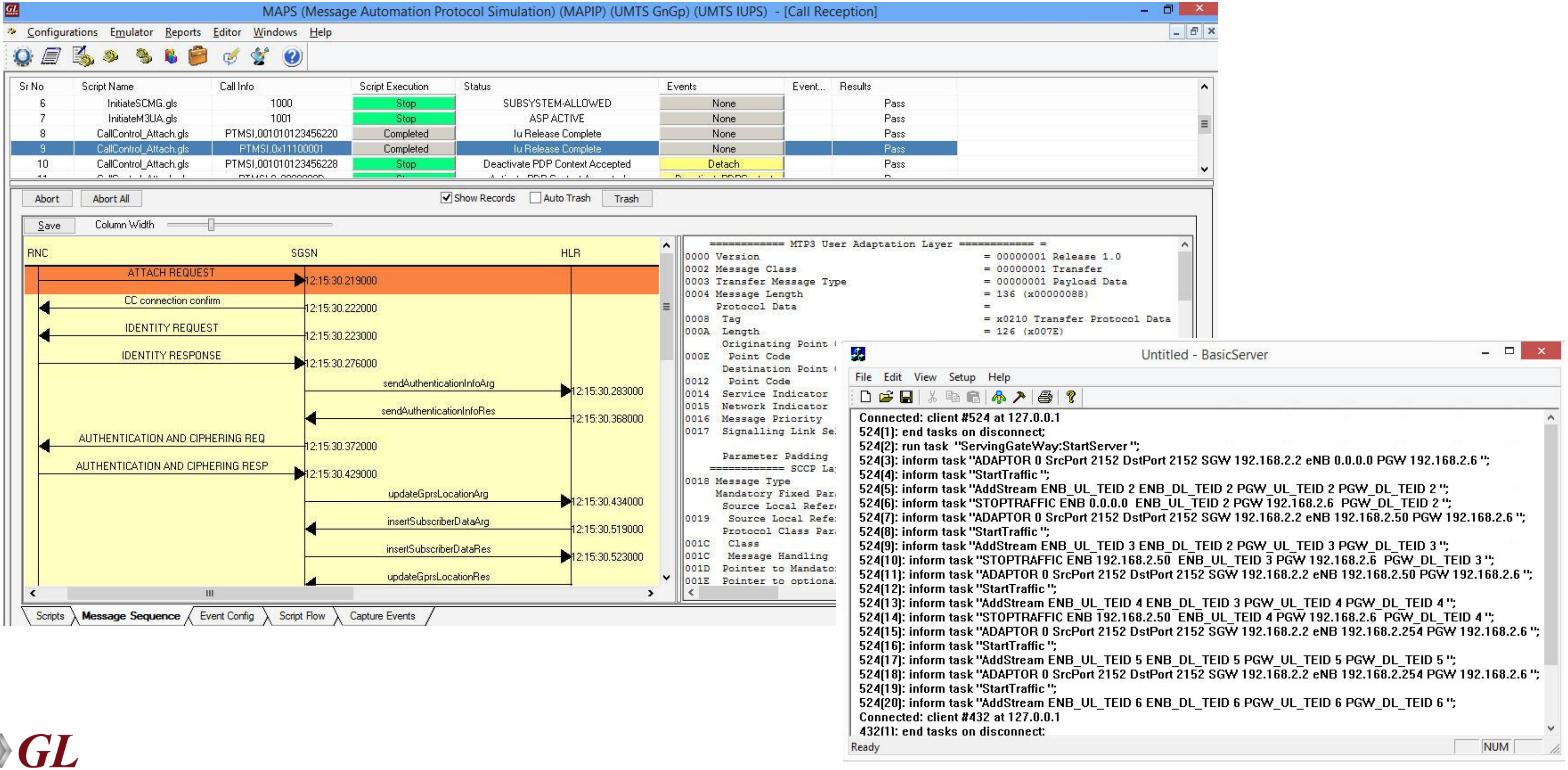
Mobile Traffic Simulation (ETH102) – Call Generation and Reception



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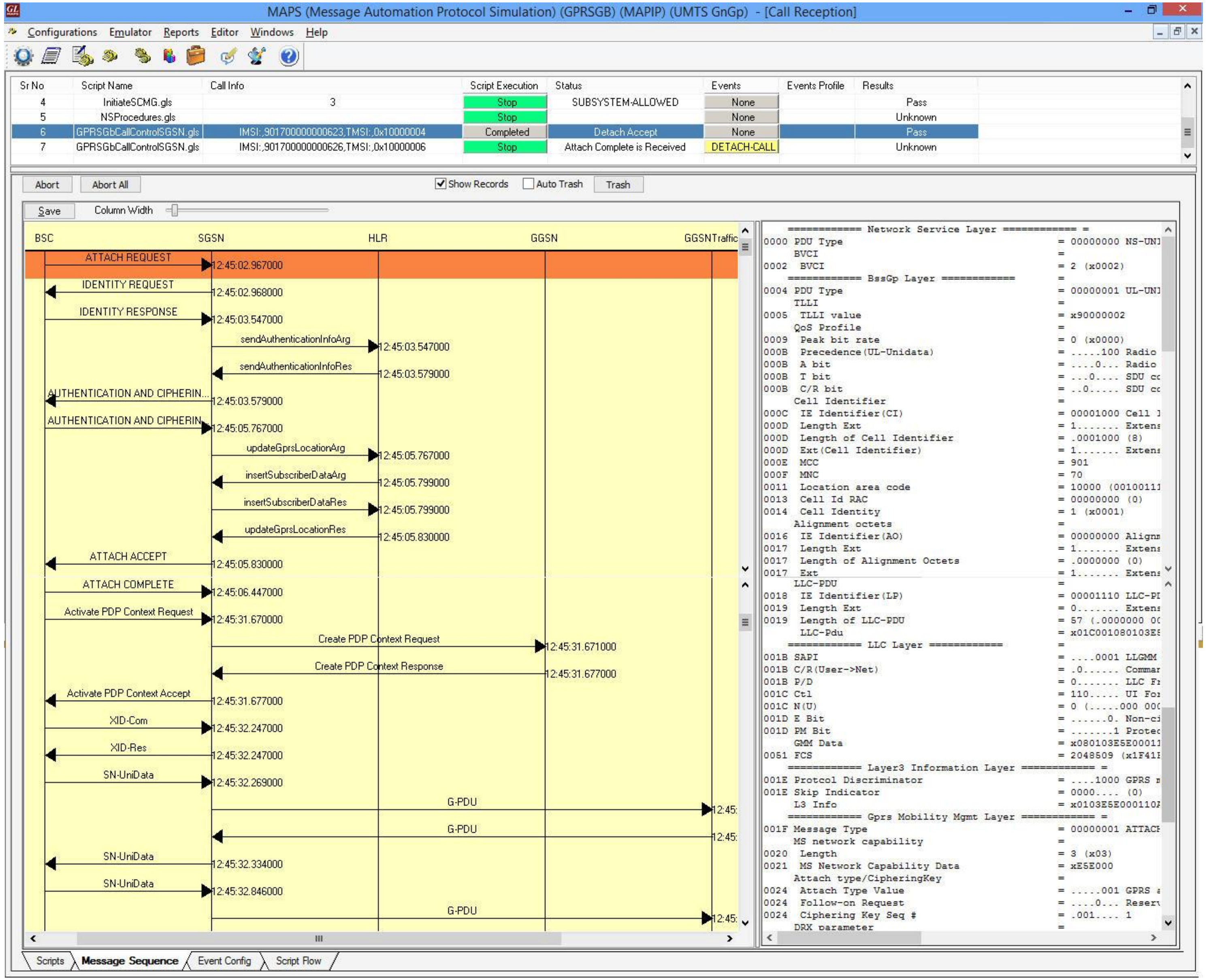
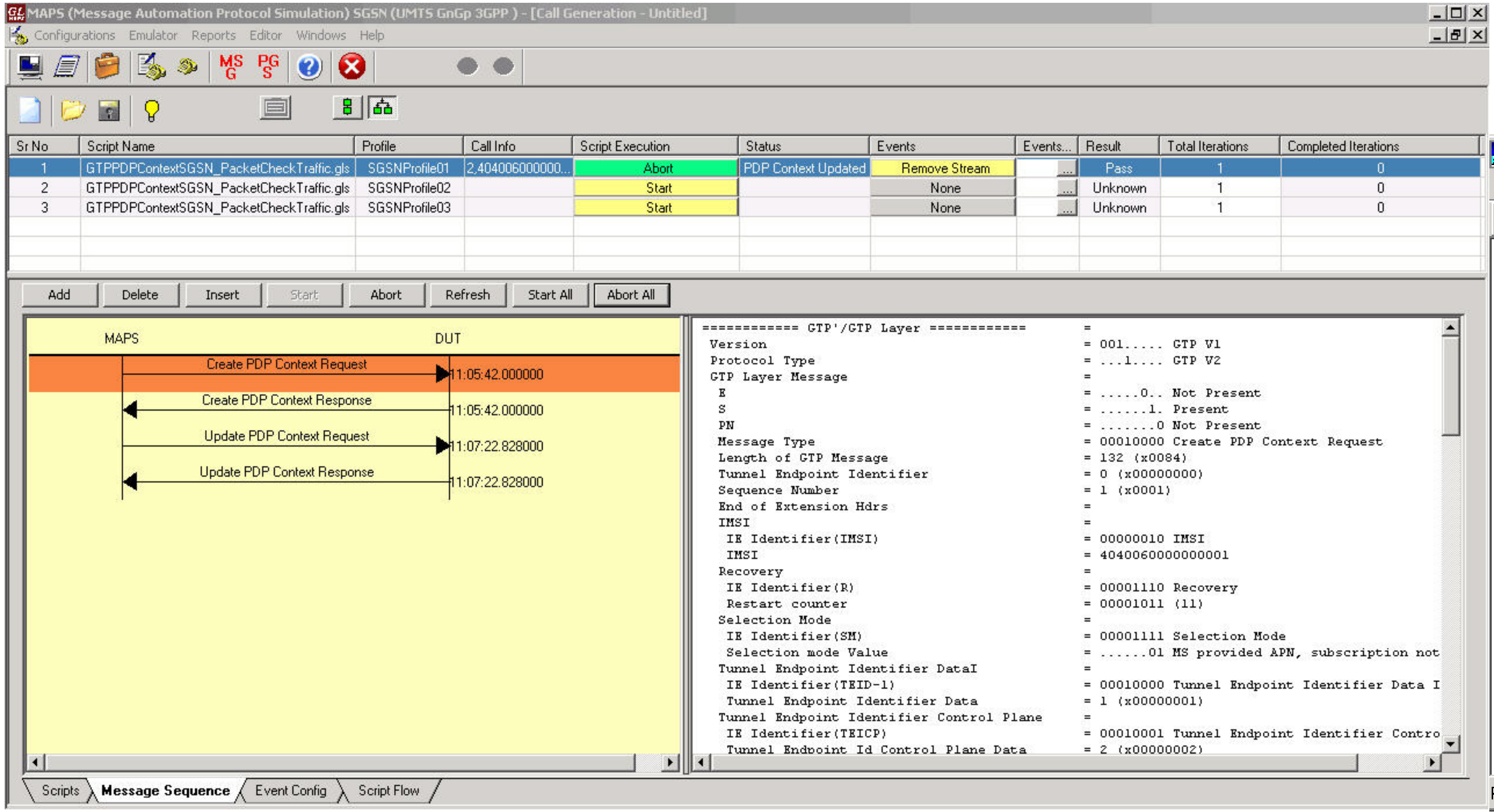
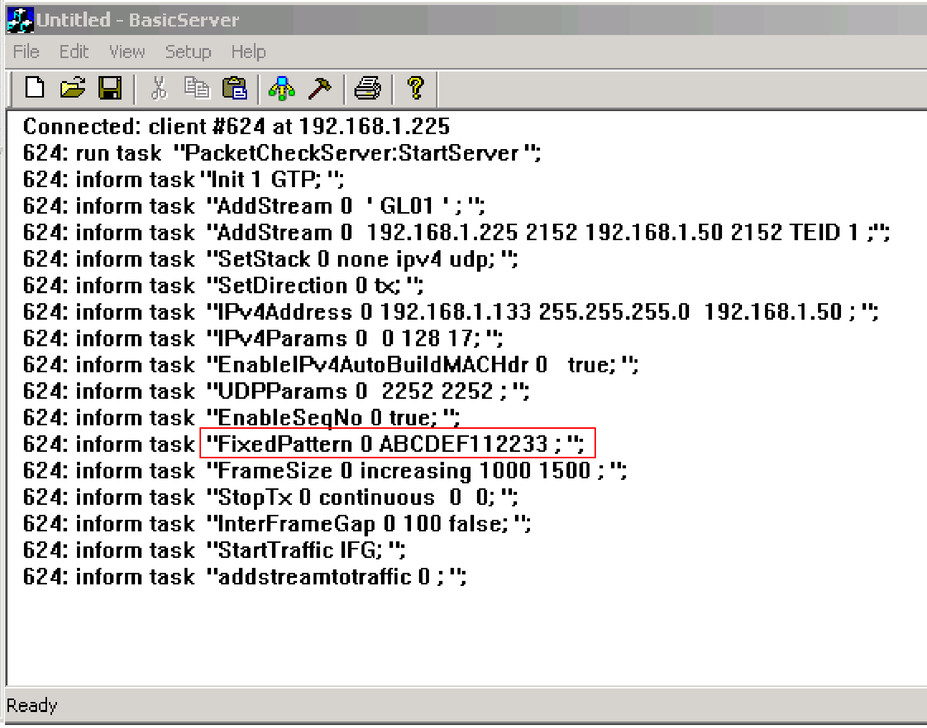
Mobile Traffic Simulation - GPRS Gb (ETH103)



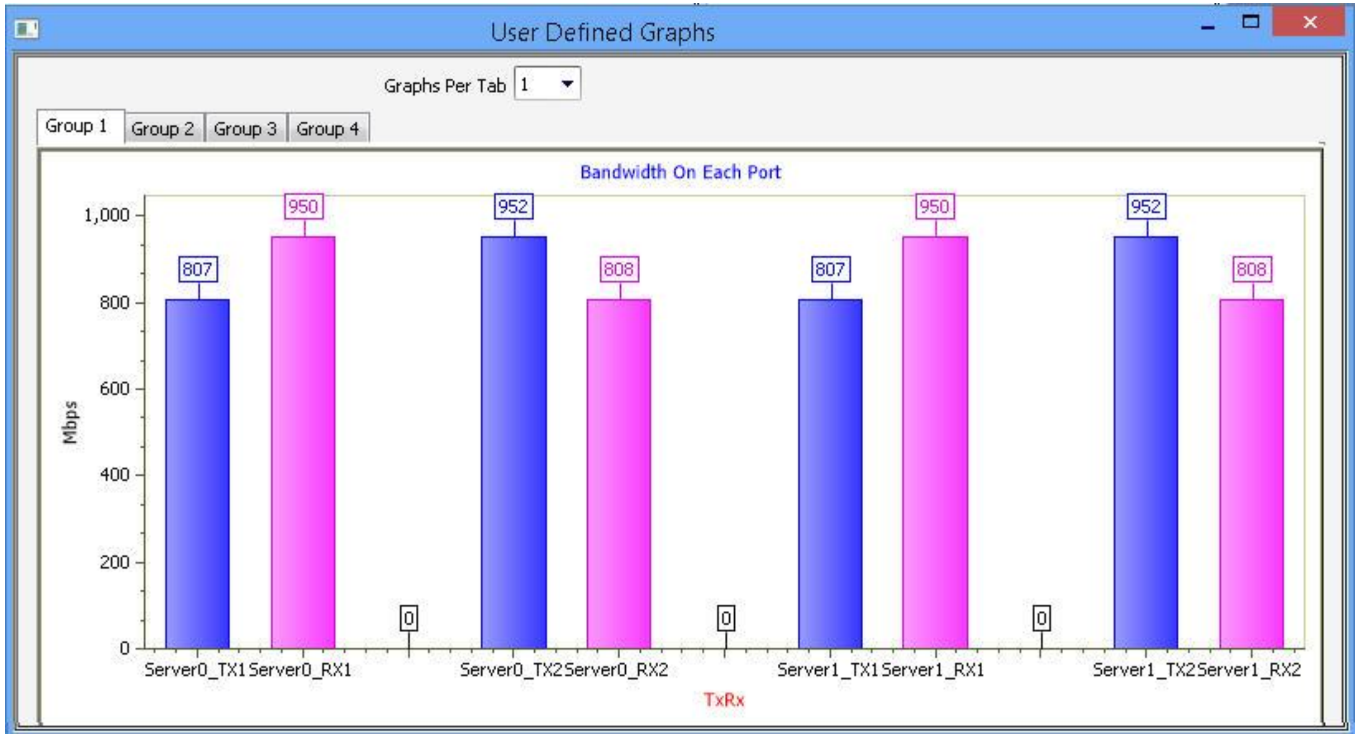
Packet Traffic Simulation – ETH100

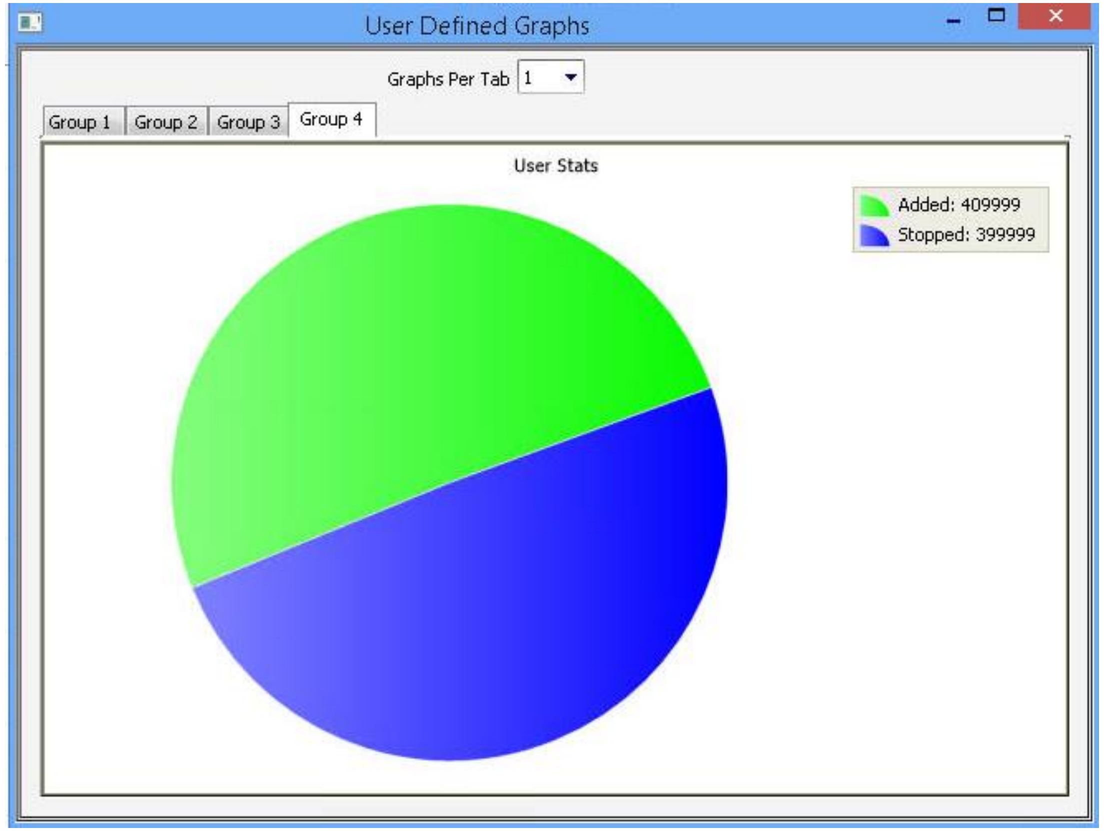
- **GTP User - Traffic Simulation (ETH100)** - This module is used to handle pre-defined GTP user-plane traffic such as HDL, Sequence Number, BERT, Hex in LTE and GPRS/UMTS networks



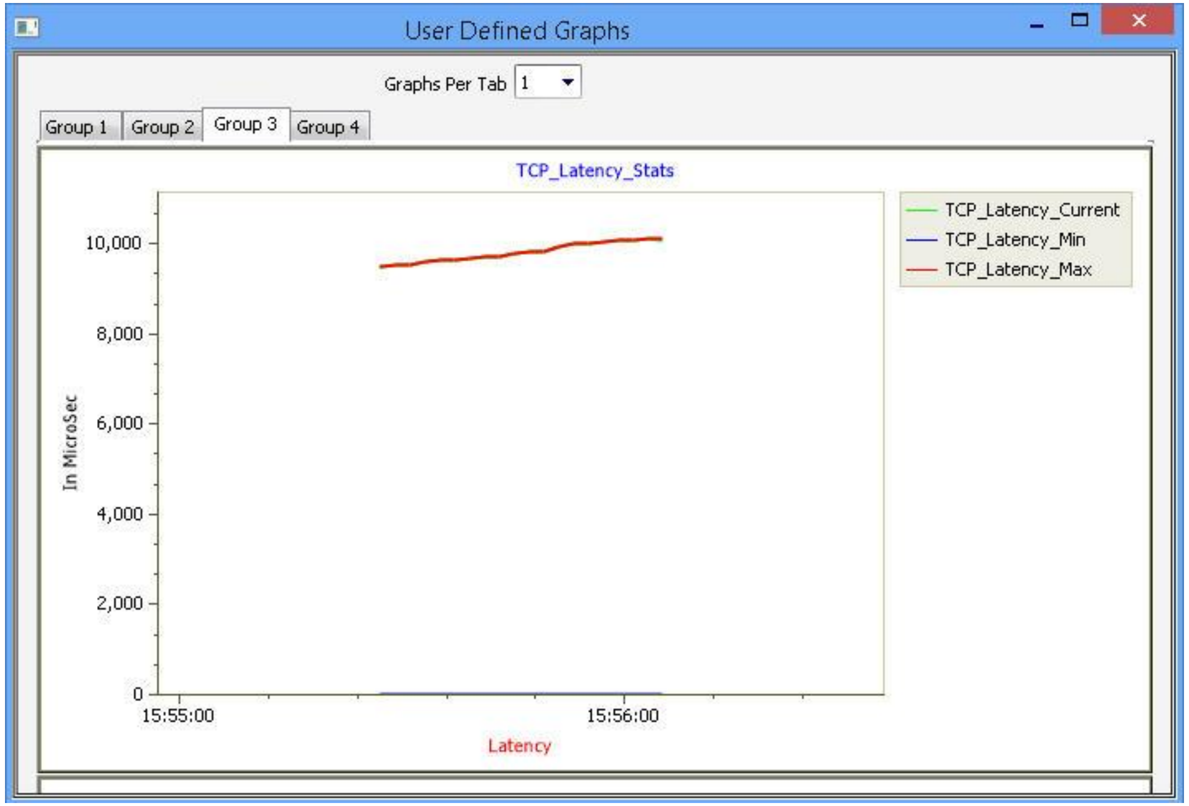


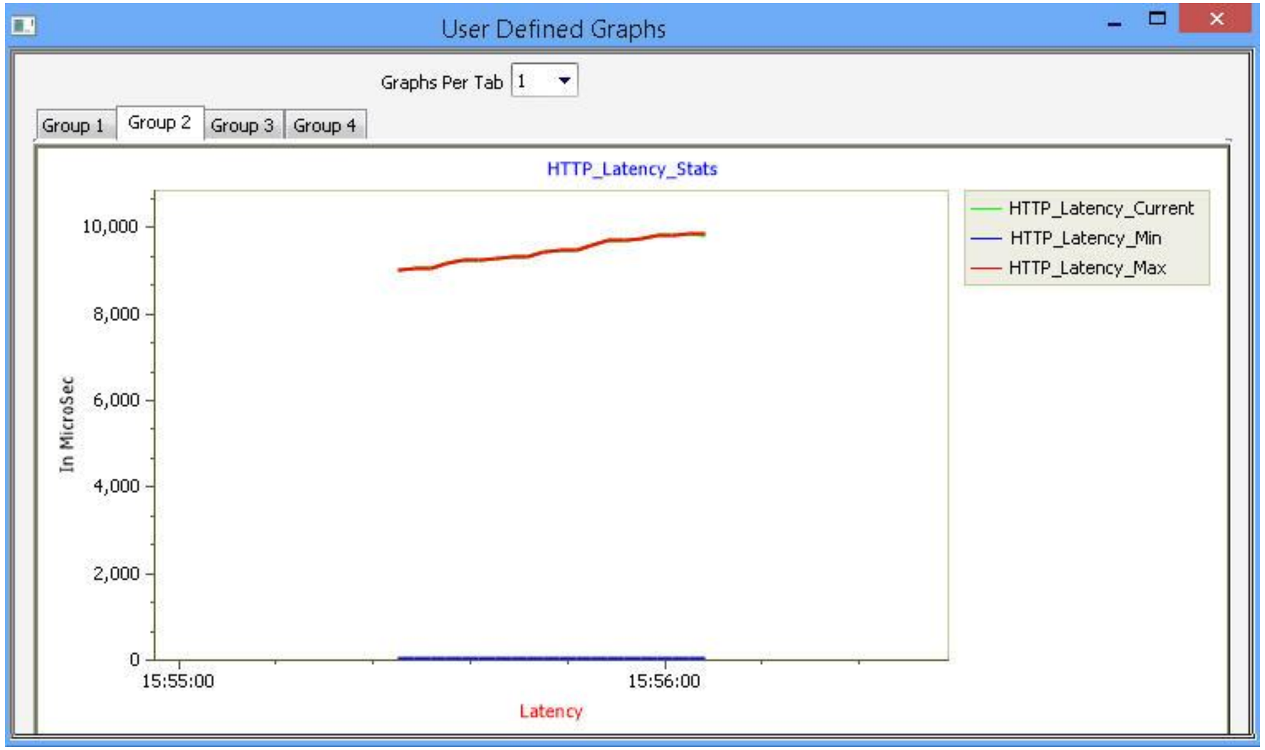
High Density Packet Traffic Statistics



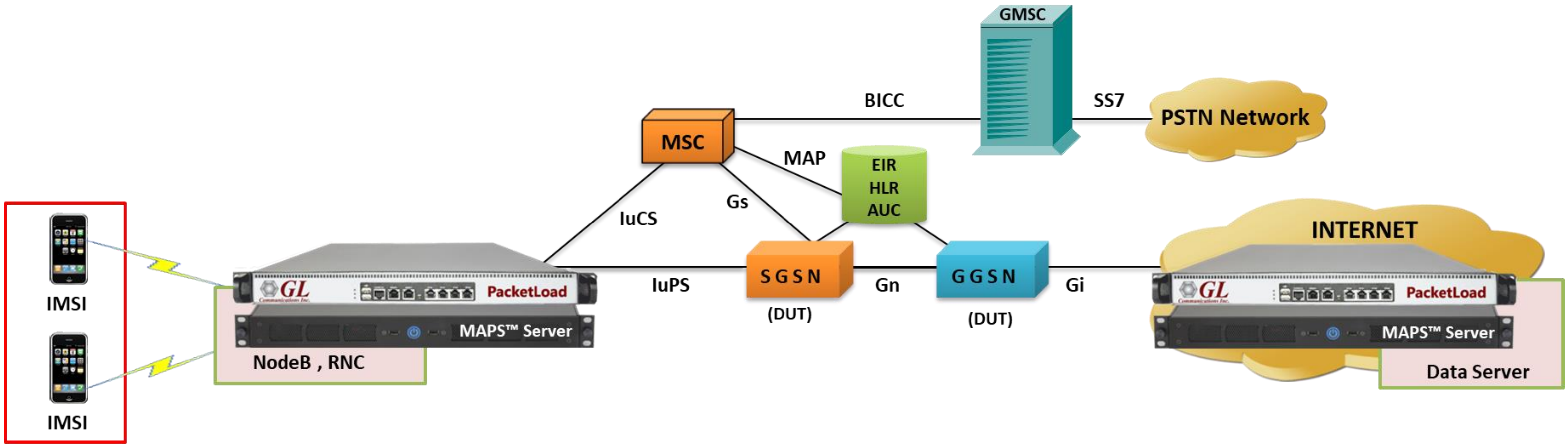


High Density Packet Traffic Statistics (Contd.)

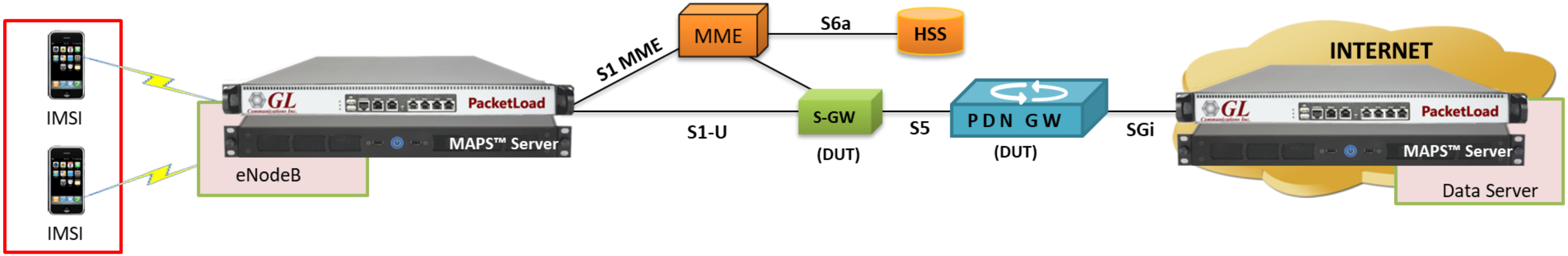




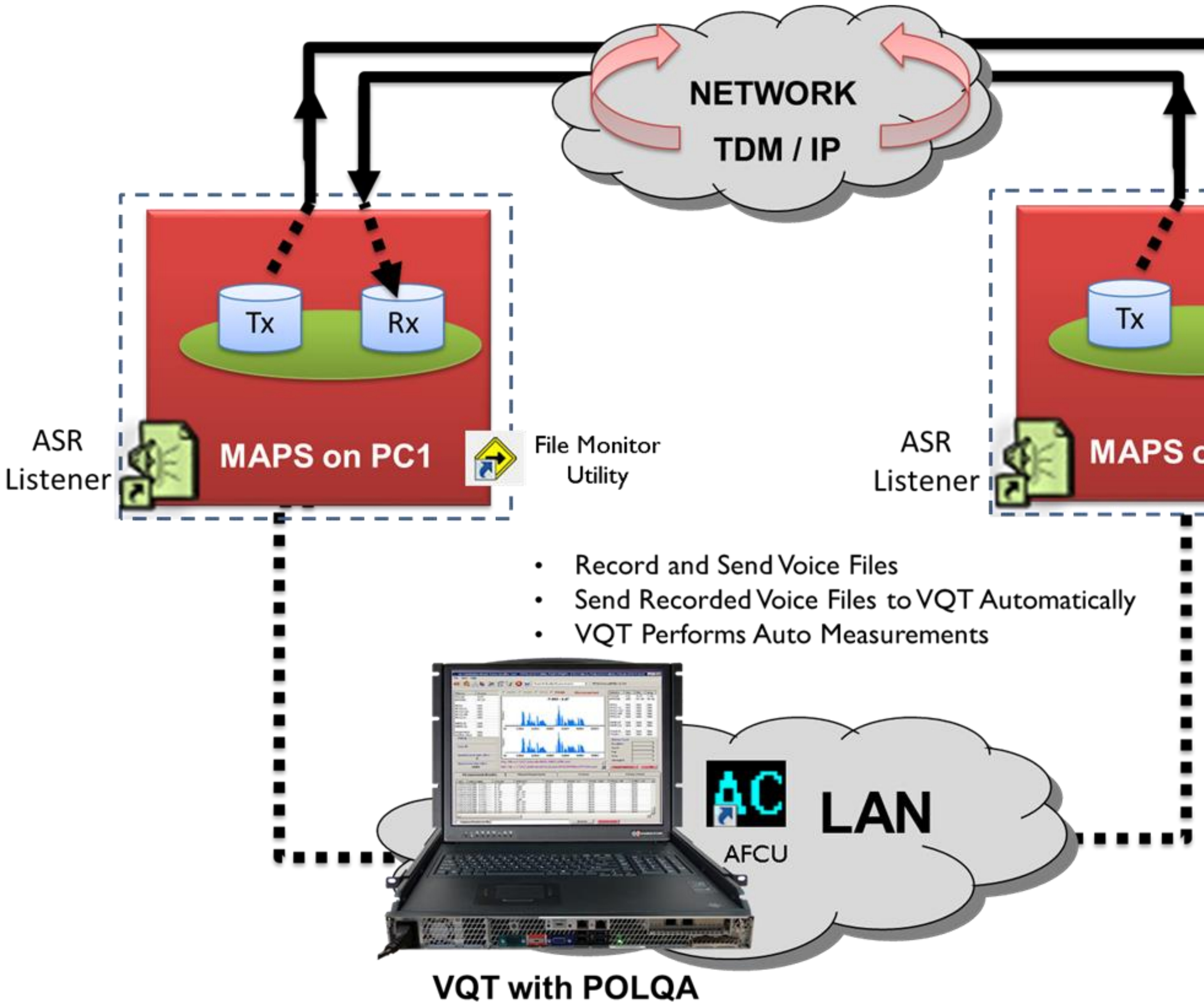
Test Scenario – 3G using PacketLoad™ along with MAPS™ Server



Test Scenario – 4G using PacketLoad™ along with MAPS™ Server



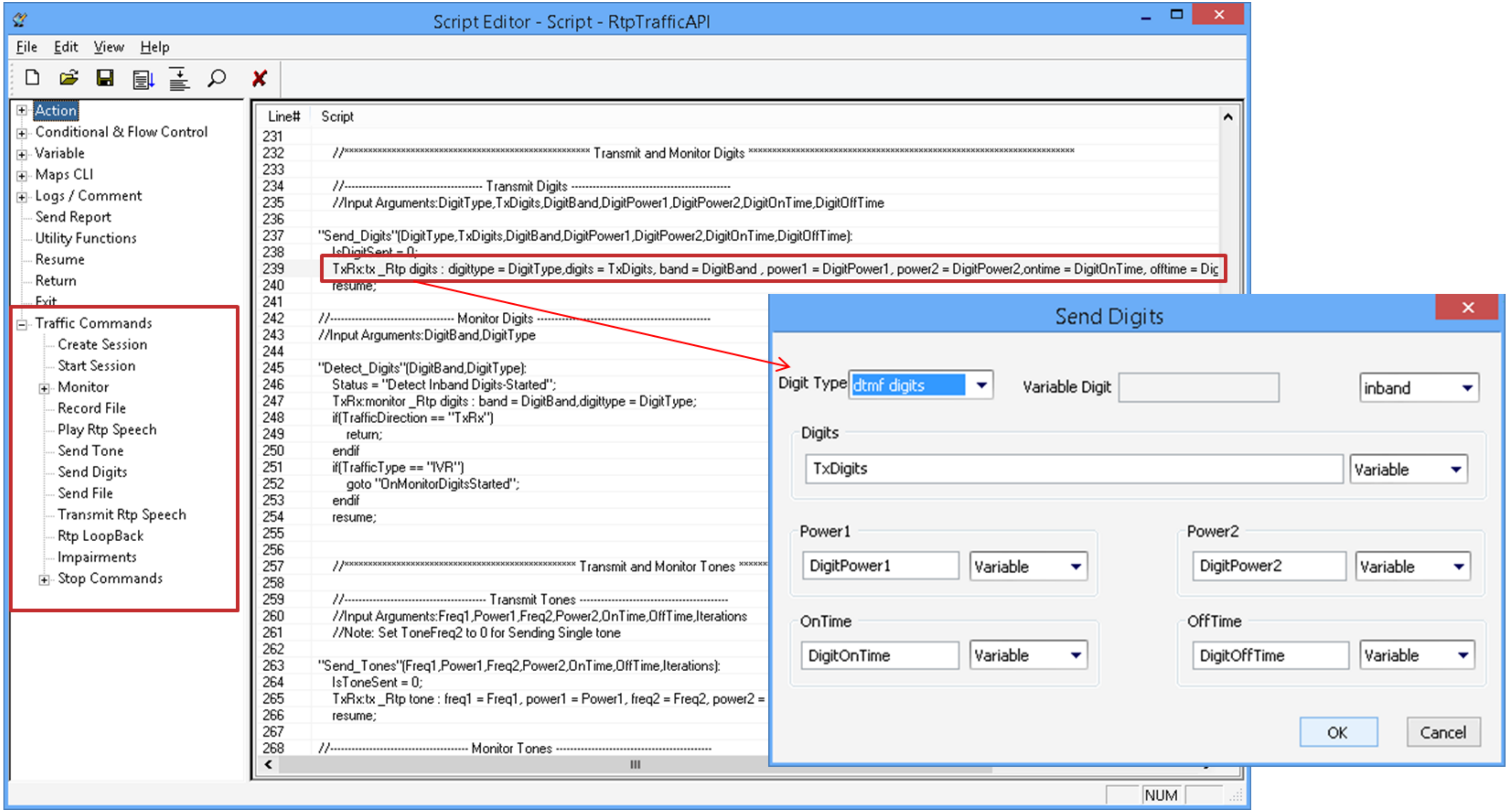
VQT Analysis



- Record and Send Voice Files
- Send Recorded Voice Files to VQT Automatically
- VQT Performs Auto Measurements

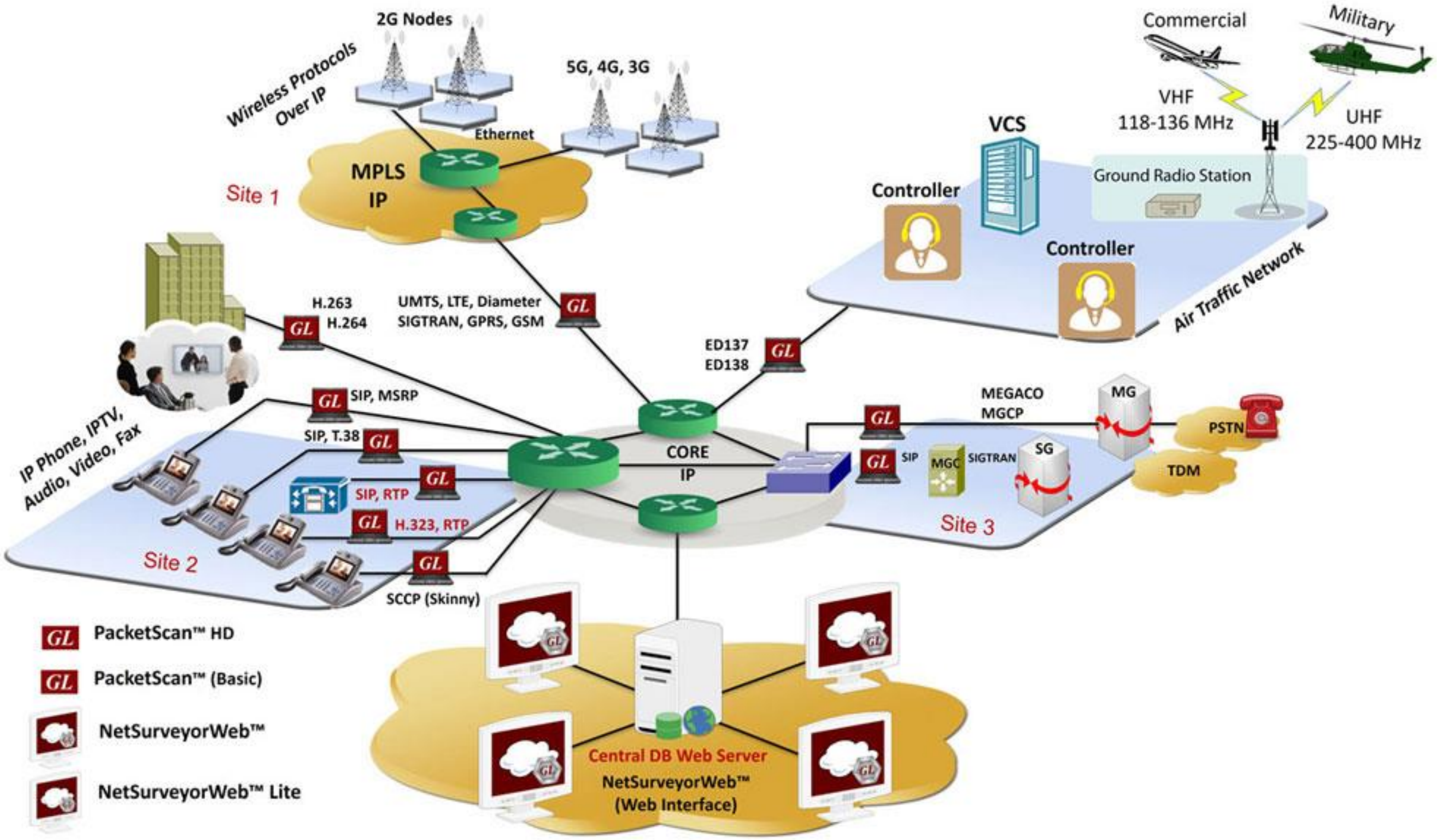
- POLQA, PESQ LQ/LQO/WB
- MOS, Jitter, Clipping, Speech and Noise Levels
- Data Testing - Wired and Wireless Networks
- Call Events - Progress & Failures
- Fax Events - Encoding, Resolution, ECM
- Delay Measurements – RTD, OWD
- E-Model, SNR, Signal Level
- Echo Measurements - ERL, Delay
- QoS, Timeouts, Retransmissions
- Google Mapping

Create Traffic Events using Script Editor



Traffic Analysis over IP Networks

PacketScan™ Analysis



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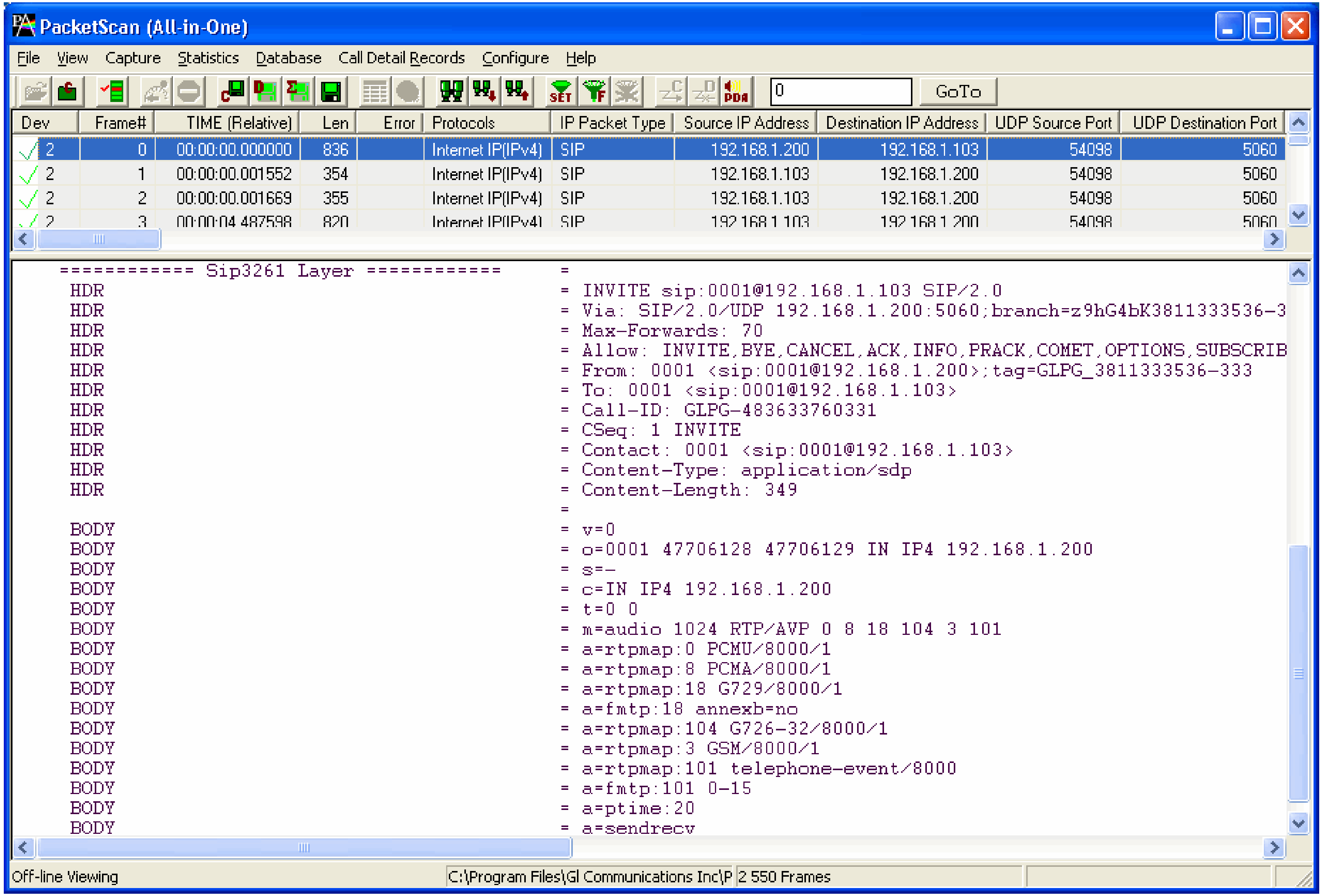
What the software does?

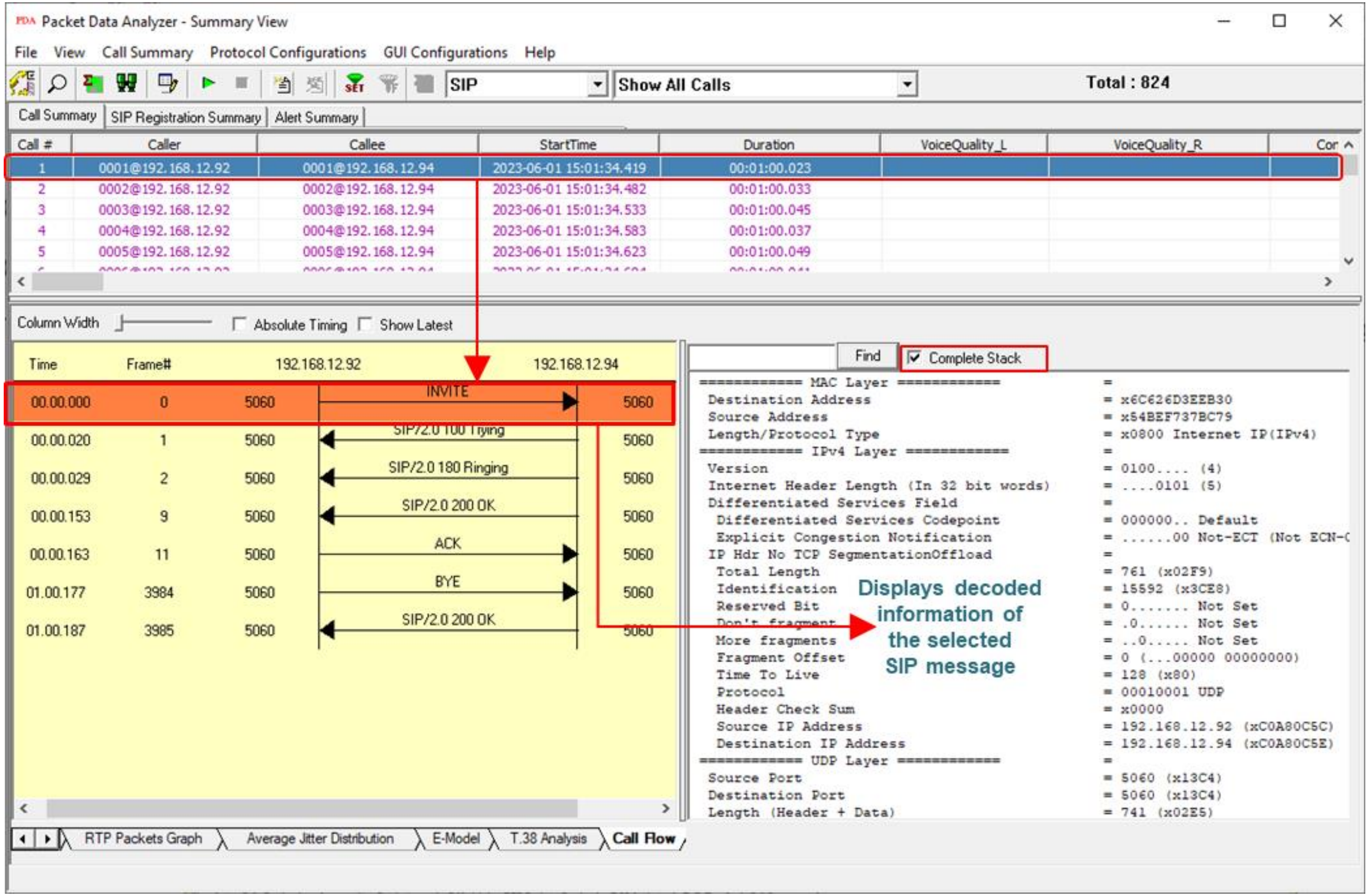
- Captures, segregates, and monitors packets; perform voice quality testing in real-time over VoIP network
- Unlimited traffic and signaling capturing capability; captured VoIP calls with video can be played back using 3rd party applications
- Can be deployed as a Probe for a centralized monitoring system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

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

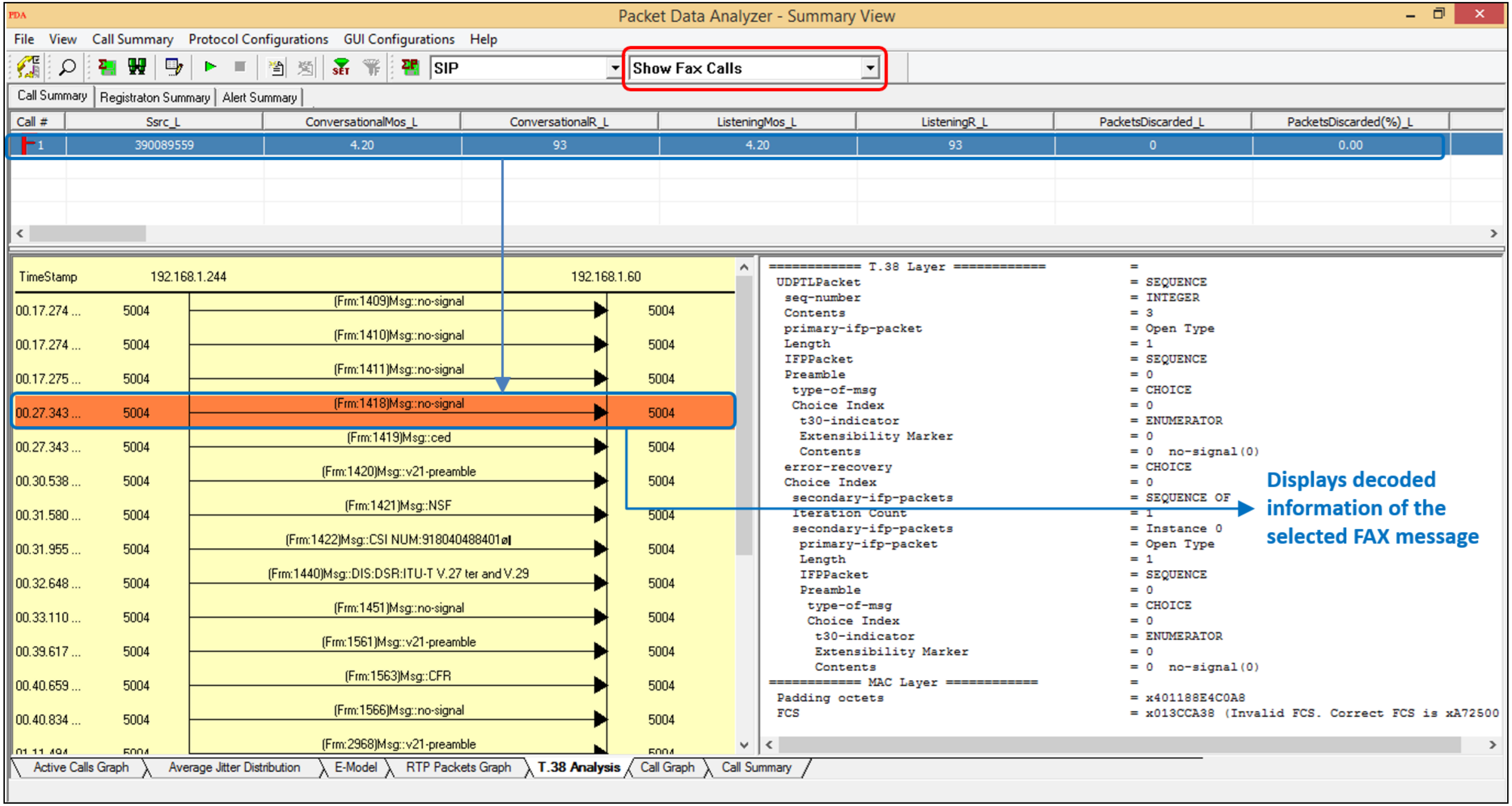
SIP Decode in PacketScan™



Voice Traffic Analysis over IP

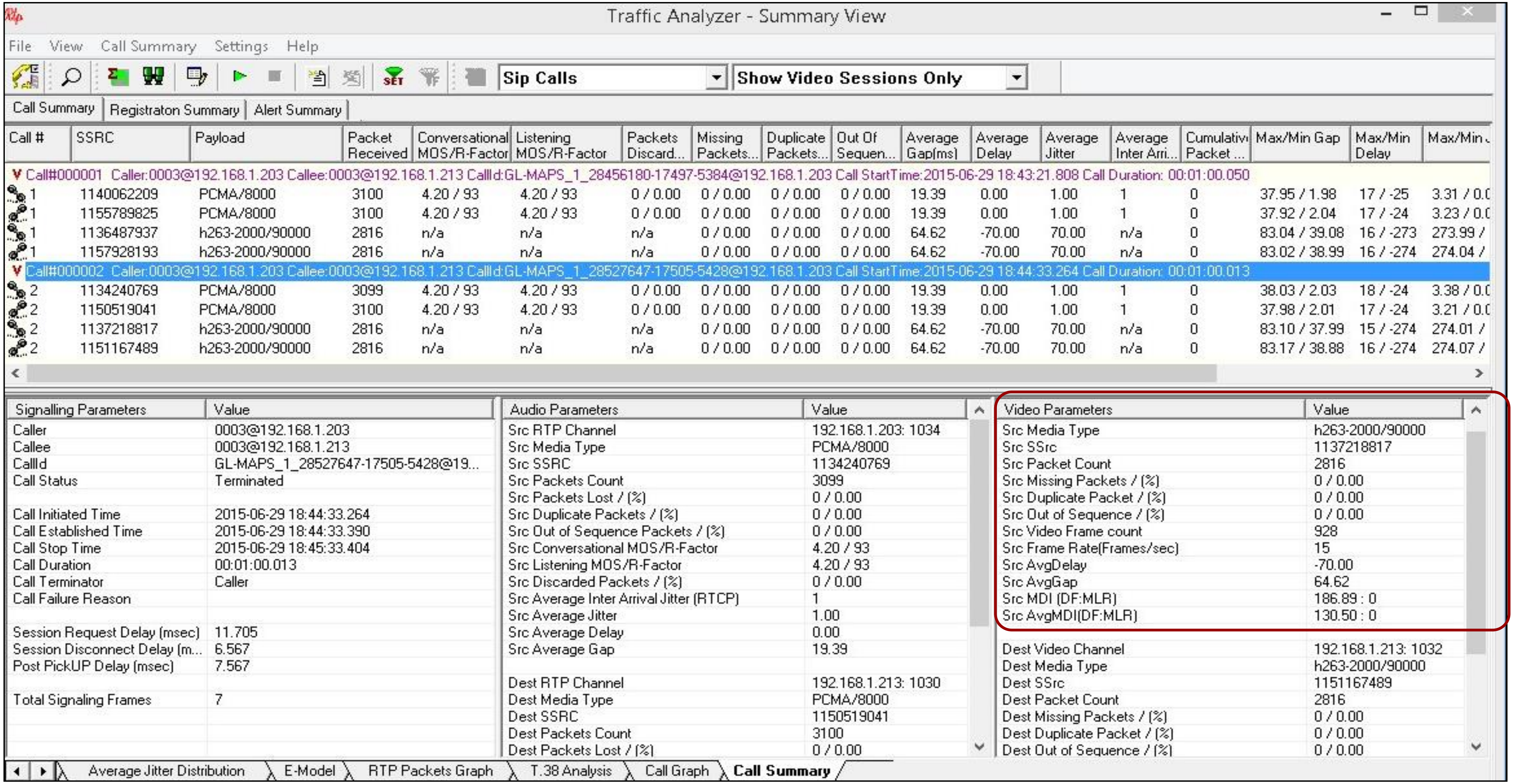


T.38 Analysis over IP



GL Communications

Video Quality Metrics in PacketScan™

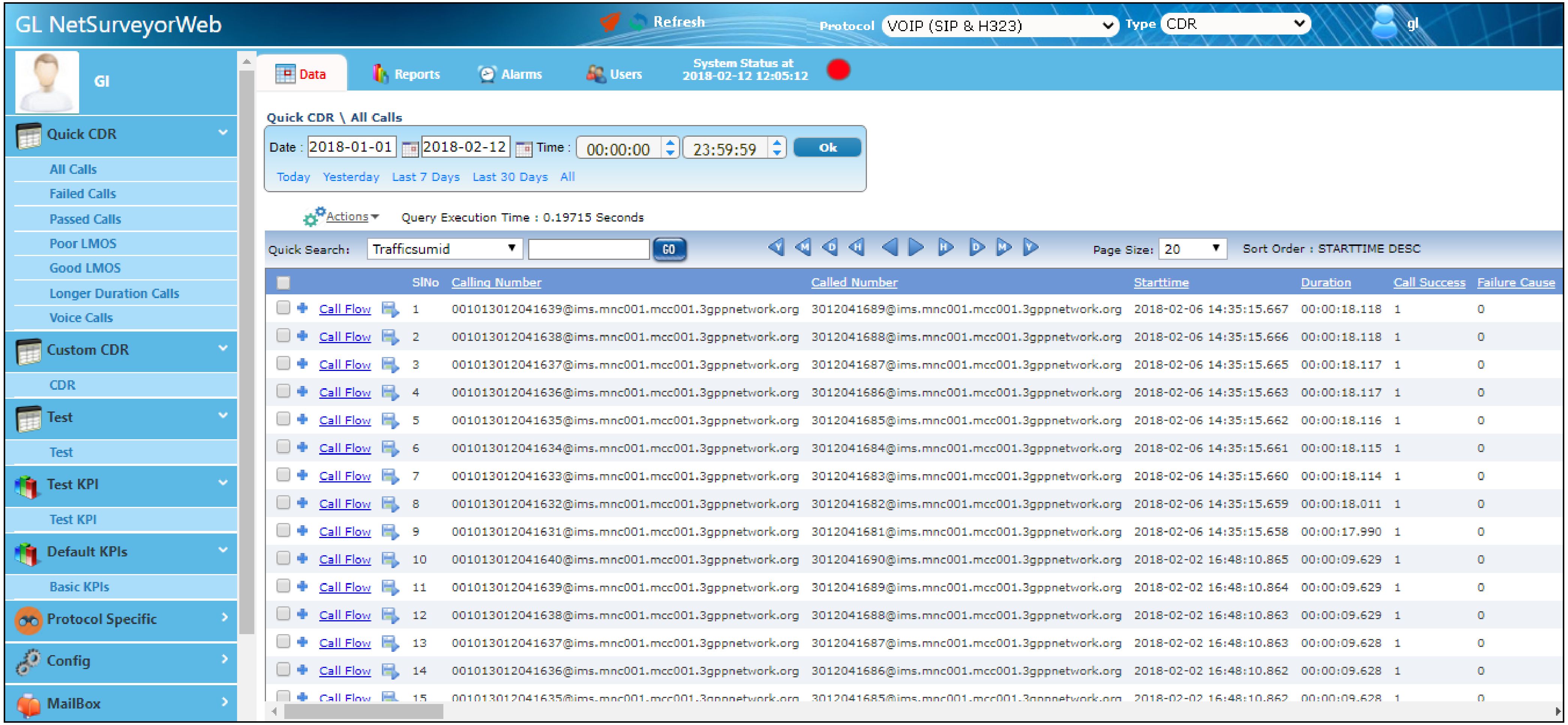


Video Quality Metrics in PacketScan™

- PacketScan™ with Video QoS capability addresses customers long felt need of Video Call Quality in IP networks
- Support Video QoS for H.263+ and H.264 video codec;
  - Source/Destination Video Channels
  - Average Delay/Gap
  - Packet Counts
  - Codec Type
  - Missing Packets
  - Delay, Gap
  - Video Frame Count
  - Media Delivery Index (MDI- (Delay Factor : Media Loss Rate))
  - Out Of Sequence count, Duplicate Packets count, and Frame Rate

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



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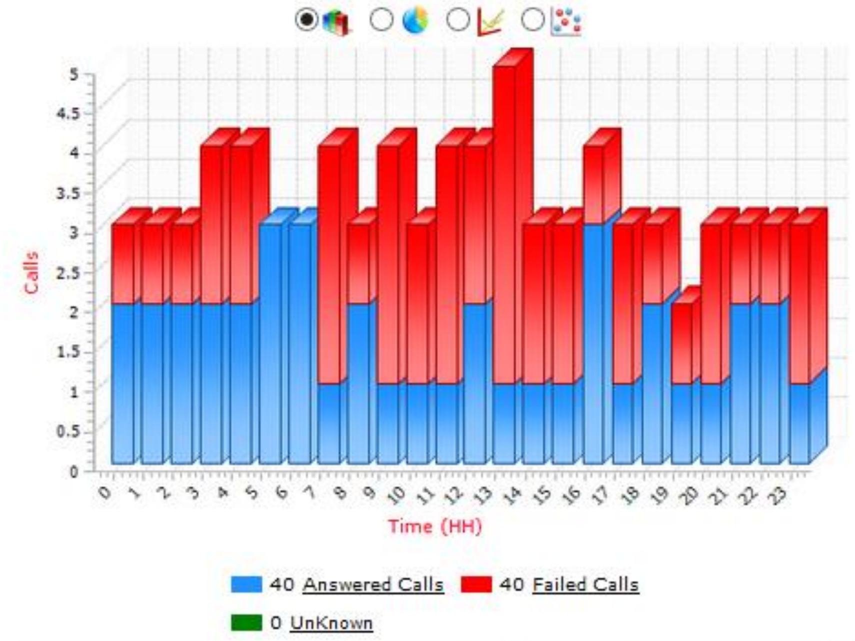
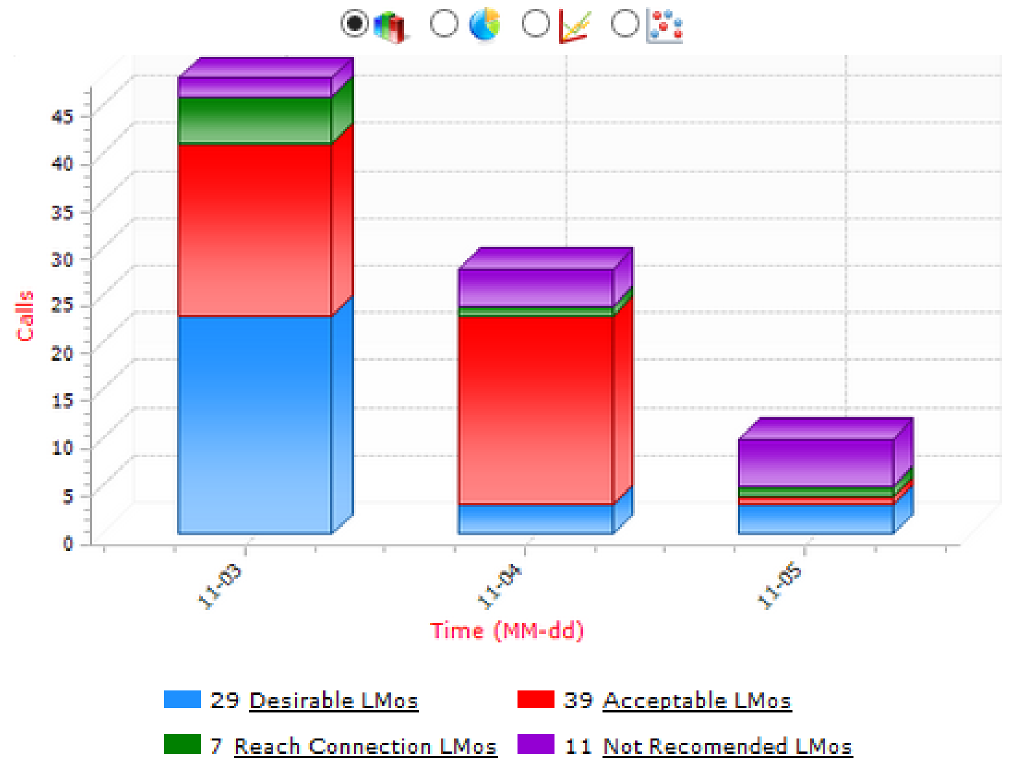
Reports

Call Completion (Basic KPIs)



Time	Answered Calls	Failed Calls	UnKnown	Total
00	2	1	0	3
01	2	1	0	3
02	2	1	0	3
03	2	2	0	4
04	2	2	0	4
05	3	0	0	3
06	3	0	0	3
07	1	3	0	4
08	2	1	0	3

Listening MOS (Basic KPIs)



Time	Desirable LMos	Acceptable LMos	Reach Connection LMos	Not Recomended LM
11-03	23	18	5	2
11-04	3	20	1	4
11-05	3	1	1	5

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Thank you