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



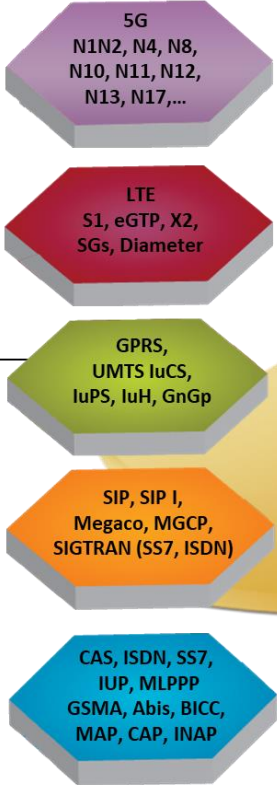
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Traffic Simulation using MAPS™

Simulate User Equipment



All Protocols

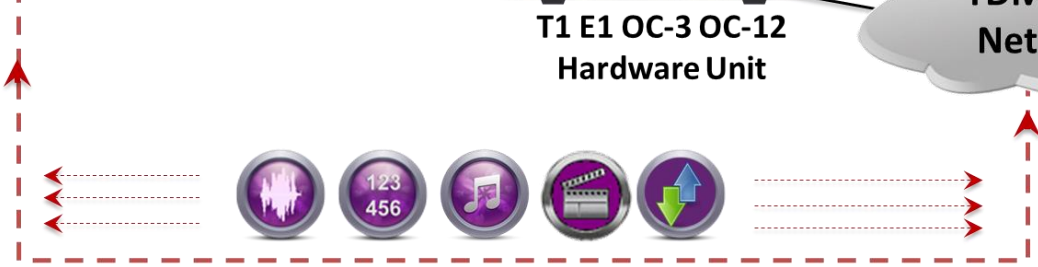
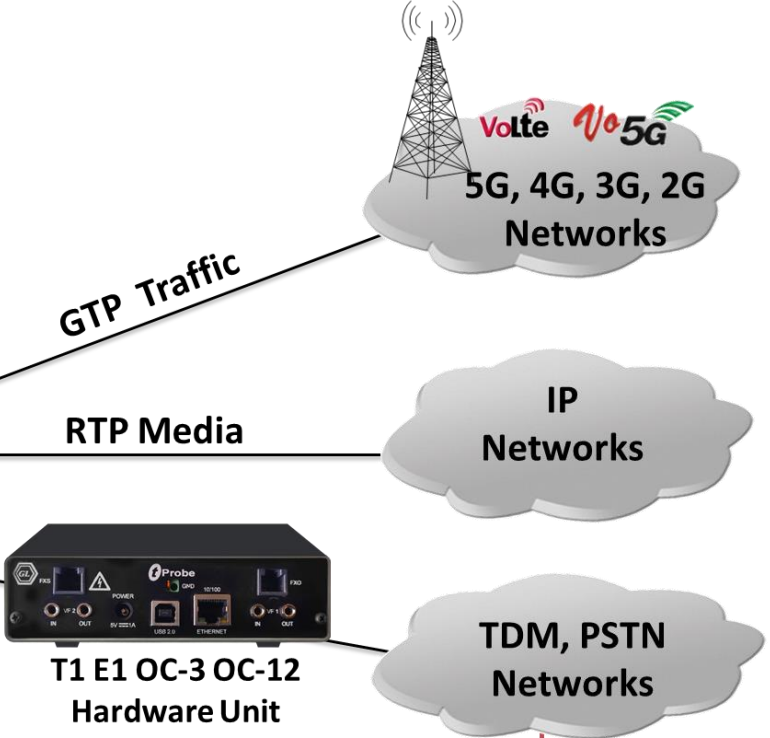


Hundreds of thousands of simultaneous calls



MAPS™ High Density (HD) Call Generator

Load, Stress, Performance



Features

Traffic Types	
<p><u>GTP Mobile Traffic Simulation</u> 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</p>	<ul style="list-style-type: none"> • 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
<p><u>RTP Traffic Simulation</u> over SIP, SIP I, MGCP, MEGACO, UMTS, GSM, Diameter, and LTE networks</p>	<ul style="list-style-type: none"> • 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. <hr/> <ul style="list-style-type: none"> • Simulation of RTP Video Traffic (H.263 & H.264), Fax (Pass-thro & T.38) <hr/> <ul style="list-style-type: none"> • RTP Voice Quality Measurements – MOS, R-Factor scores <hr/> <ul style="list-style-type: none"> • Simulation of RTP FAX Traffic - G.711 Pass-thro and T.38 UDPTL
<p><u>SMS Traffic Simulation</u> over the GSM, UMTS, and MAP interfaces</p>	<ul style="list-style-type: none"> • 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)).
<p><u>TDM Traffic Simulation</u> over ISDN, SS7, GSM, CAS interfaces</p>	<ul style="list-style-type: none"> • 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

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

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

- Table:** A table showing call execution results. The columns are Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, E, Result, Total Iterations, and Completed Iterations. The data rows are:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	E	Result	Total Iterations	Completed Iterations
1	Placecall.gls	Card1TS01	1,1	Stop	File Sent	DisconnectCall		Pass	1	0
2	Placecall.gls	Card1TS02	1,2	Stop	Digits Transmitted	DisconnectCall		Pass	1	0
3	Placecall.gls	Card1TS03	1,3	Stop	Transmitting Tone	DisconnectCall		Pass	1	0
5	Placecall.gls	Card1TS05	1,5	Stop	VFin & Out Enabled on var1:var2,aC	DisconnectCall		Pass	1	0
6	Placecall.gls	Card1TS06	1,6	Stop	Digits Transmitted	DisconnectCall		Pass	1	0
- Buttons:** Below the table are buttons for "Add", "Delete", "Insert", "Refresh", "Start", "Start All", "Stop", "Stop All", "Abort", and "Abort All".
- Message Sequence Diagram:** A diagram showing the interaction between MAPS and DUT. The sequence includes:
 - MAPS sends SETUP to DUT at 11:27:52.211.6707.
 - DUT sends CALL PROCEEDING to MAPS at 11:27:52.76.8679.
 - DUT sends ALERTING to MAPS at 11:27:53.84.1406.
 - DUT sends CONNECT to MAPS at 11:27:53.91.4286.
 - MAPS sends CONNECT ACKNOWLEDGE to DUT at 11:27:53.96.3086.
 - MAPS sends File Transmitted :: mu-law samples to DUT at 11:28:13.108.5422.
- Message Details:** A detailed view of a Q.93x Layer 3 Layer message. The message is a Q.931/I.451 user-network call control message. The details include:
 - 0000 Protocol Discriminator = 00001000 Q.931/I.451 user-network call control message:
 - 0001 Call Reference Length = ...0010 2 Bytes
 - 0002 Call Reference Value = 2 (.00000000 00000010)
 - 0002 Call Reference Flag = 0..... FROM side that originated callref
 - 0004 Message Type = 00000101 SETUP
 - Bearer capability =
 - 0005 IEI Bearer Capability = 00000100 Bearer Capability IE Identifier
 - 0006 IE Bearer Capability Length = 3 (x03)
 - 0007 Information Transfer Capability = ...00000 Speech
 - 0007 Coding Standard = .00..... ITU_T (CCITT) standardized coding
 - 0007 Oct 3 Extension Bit (Oct 3) = 1..... Next Octet Not Present
 - 0008 Information Transfer Rate = ...10000 64 kbit/s
 - 0008 Transfer Mode = .00..... Circuit Mode
 - 0008 Oct 4 Extension Bit (Oct 4) = 1..... Next Octet Not Present
 - 0009 Layer 1 Indent Choice = .01..... Layer 1 Identifier
 - 0009 User Information Layer 1 Protocol (BC) = ...00010 Mu-law, Rec G.711
 - 0009 Layer 1 Identifier = .01..... Layer 1 Id
 - 0009 Extension Bit (Oct 5) = 1..... Next Octet Not Present
 - Channel identification =
- Navigation:** At the bottom, there are tabs for "Scripts", "Message Sequence", "Event Config", "Script Flow", and "Capture Events".
- Status Bar:** At the very bottom, there are indicators for "Initialisation Errors", "Error Events", "Captured Errors", and "Link Status Up=1 Down=0".

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

The screenshot shows the MAPS (Message Automation Protocol Simulation) software interface. The title bar indicates the window is for "MAPS (Message Automation Protocol Simulation) BTS (GsmAbis GSM900) - [Call Generation - Master Configuration]". The menu bar includes "Configurations", "Emulator", "Reports", "Editor", "Windows", and "Help".

The main window displays a table of script executions:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Eve...	Result	Total Iterations	Completed Iterations
1	BTS_MDC.gls	BTSPProfile001	IMSI:.404060000000001,TMSI:.0x11111001,CalledNumber:.93411...	Start	Released Air Interface Resources	None		Pass	1	1
2	BTS_LUC.gls	BTSPProfile002	IMSI:.404060000000002,TMSI:.0xB3A6DB3C,CalledNumber:.93411...	Start	Released Air Interface Resources	None		Pass	1	1

Below the table are buttons for "Add", "Delete", "Insert", "Refresh", "Start", "Start All", "Stop", "Stop All", "Abort", and "Abort All".

The main area is split into two panes. The left pane shows a "Message Sequence" diagram between "MAPS" and "DUT". The sequence includes:

- MAPS to DUT: CHANnel ReQuireD (15:25:38.628000)
- DUT to MAPS: Immediate Assignment (15:25:39.248000)
- MAPS to DUT: CM SERVICE REQUEST (15:25:39.249000)
- DUT to MAPS: IDENTITY REQUEST (15:25:39.565000)
- MAPS to DUT: IDENTITY RESPONSE (15:25:39.566000)
- DUT to MAPS: AUTHENTICATION REQUEST (15:25:39.886000)
- MAPS to DUT: AUTHENTICATION RESPONSE (15:25:39.887000)
- DUT to MAPS: CIPHERING MODE COMMAND (15:25:40.204000)
- MAPS to DUT: CIPHERING MODE COMPLETE (15:25:40.205000)
- DUT to MAPS: CM SERVICE ACCEPT (15:25:40.512000)
- MAPS to DUT: SETUP (15:25:40.513000)
- DUT to MAPS: CALL PROCEEDING

The right pane shows a detailed protocol stack for the "CM SERVICE REQUEST" message:

```

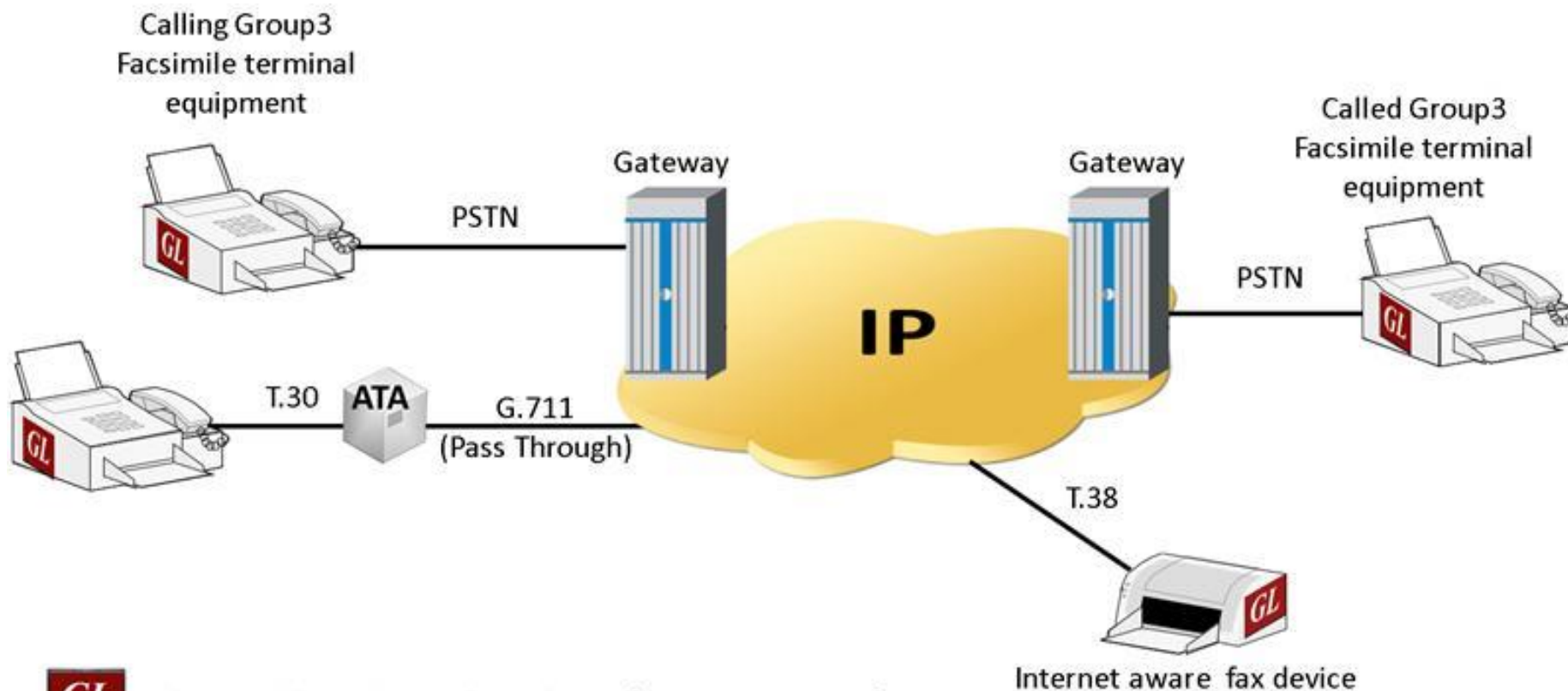
===== BTSM Layer =====
0000 T-bit = .....0 Non-Transparent Message
0000 Message Group = 0000001. Radio Link Layer Mgmt
0001 Message Type = 00000110 ESTablish INDication.
      Channel number =
0002 IE Identifier(Ch No) = 00000001 Channel number
0003 Channel Type = 01001... SDCCH/8 + ACCH
0003 Sub-Channel #(T bits) = 1 (...001...)
0003 Time Slot # = .....000 (0)
      Link Identifier =
0004 IE Identifier(LinkId) = 00000010 Link Identifier
0005 SAPI Value = .....000 SAPI Value(CC,MM,RRM signalling)
0005 Priority = ...00... Normal Priority
0005 NA = ..0.... Link Identifier applicable
0005 SAPI Value Channel Type = 00..... Main Signalling Channel(FACCH or SDCCH)
      L3 Information =
0006 IE Identifier(L3Info) = 00001011 L3 Information
0007 Length of L3 Information = 13 (x000D)
      Layer 3 Information = x05247103231801050411111001
===== Layer3 Protocol Layer =====
0009 Protocol Discriminator = ...0101 Mobility Management Messages
0009 Skip Indicator = 0000.... (0)
===== MM Layer =====
000A Message Type = ..100100 CM SERVICE REQUEST
000A Sequence Number = 00..... (0)
      CM service type / Ciphering key =
000B key sequence(ms->nw) = .111.... No key is available
    
```

At the bottom, there are tabs for "Scripts", "Message Sequence", "Event Config", and "Script Flow". The status bar shows "Error Events", "Captured Errors", and "Link Status Up=1 Down=0".

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



GL's Fax Call Simulators & Analyzers (TDM, IP, Wireless)

- MAPS™
- VQuad™
- FaxScan™
- GLInsight™
- PacketScan™

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

The screenshot displays the MAPS (Message Automation Protocol Simulation) interface. The top window shows a table of call reception events:

Sr No	Script Name	Call Info	Script Execution	Status	Events	Results
10	GSMA_Call.gls	IMSI: 901700000000625, Calling Number: 90625, IMSI: 901700000000627, Called Number: 90627	Completed	SCCP Connection Released	None	Pass
11	GSMA_Call.gls	IMSI: 901700000000625, Calling Number: 90625, IMSI: 0x00000002	Completed	SCCP Connection Released	None	Pass
12	GSMA_Call.gls	IMSI: 901700000000627, Called Number: 90627, Calling Number: 90625	Completed	SCCP Connection Released	None	Pass

The main window shows a message sequence diagram between BSC, MSC, HLR, and SMSC. The 'SMS-SUBMIT' message is highlighted in orange. The diagram shows the following sequence of messages:

- CM SERVICE REQUEST (BSC to MSC)
- CC connection confirm (MSC to BSC)
- IDENTITY REQUEST (BSC to MSC)
- IDENTITY RESPONSE (MSC to BSC)
- sendAuthenticationInfoArg (MSC to HLR)
- sendAuthenticationInfoRes (HLR to MSC)
- AUTHENTICATION REQUEST (BSC to MSC)
- AUTHENTICATION RESPONSE (MSC to BSC)
- CIPHER MODE COMMAND (BSC to MSC)
- CIPHER MODE COMPLETE (MSC to BSC)
- CM SERVICE ACCEPT (BSC to MSC)
- MM STATUS (BSC to MSC)
- SMS-SUBMIT** (BSC to MSC)
- no-forwardSMAArg (MSC to SMSC)
- no-forwardSMRes (SMSC to MSC)
- CP-ACK (MSC to BSC)
- SMS-SUBMIT-REPORT (MSC to BSC)
- CP-ACK (BSC to MSC)
- CLEAR COMMAND (BSC to MSC)
- CLEAR COMPLETE (MSC to BSC)
- RLSD released (BSC to MSC)
- RLC release complete (MSC to BSC)

The right pane shows the detailed structure of the SMS-SUBMIT message, including MTP3 User Adaptation Layer and SCCP Layer parameters.

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

The screenshot displays the IVR test software interface. The main window shows a call log with columns for Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, Events Profile, and Result. Below the log is a 'MAPS' section showing a sequence of events between the test environment (MAPS) and the Device Under Test (DUT). A 'Correlation & Audio Analysis' window is open, showing the 'Speech To Text' tab. This window includes a 'Speech Transcription Server' section with a 'Server IP' of 50.76.16.181 and a 'Server is Running' indicator. It also features a 'Transfer Speech To Text' section with an 'Encoding' of PCM16 NB (8kHz) and a 'Voice File Name' of C:\VQT_Degraded\STT\VoicePrompt_2.pcm. The 'Pass Factor (%)' is set to 100. A table at the bottom of the window shows the transcription results, with a 'SpchAnalysis' entry highlighted in red, indicating a 'Pass (100% pass)'.

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result
1	APS_PlaceCall.gls	Line001	Line001,1,1,0	Start	CALL_RELEASED	None		Pass

Timestamp	Device ID	Type	Events
05/01/2018 12:16:09 PM		Status	VoiceFile=C:\VQT_Degraded\STT\VoicePrompt_2.pcm
05/01/2018 12:16:09 PM		Status	Reference=Your Call has been forwarded to automatic voice message sy
05/01/2018 12:16:09 PM		Status	Encoding=PCM16 NB (8kHz)
05/01/2018 12:16:13 PM		Result	SpeechToText Result...
05/01/2018 12:16:13 PM		Result	certainty=0.9486
05/01/2018 12:16:13 PM		Result	timeTaken=3.026
05/01/2018 12:16:13 PM		Result	transcription=Your call has been forwarded to an automatic voice messag
05/01/2018 12:16:14 PM		SpchAnalysis	Pass (100% pass)

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 IncltProbe 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;

# RTP Traffic Simulation

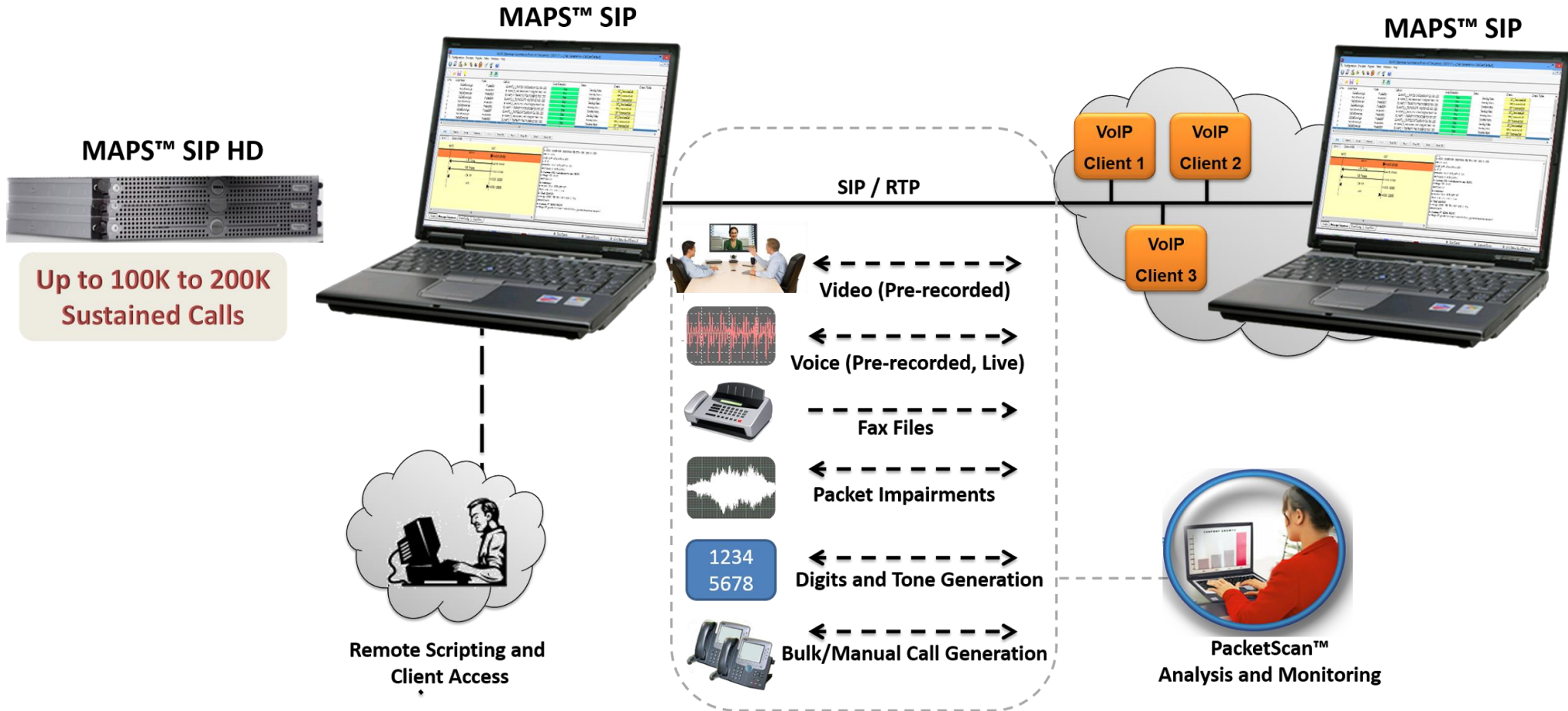


# RTP Traffic Simulation

| <b>RTP Traffic Options</b>                                                                                                                                 | <b>Licenses</b>                                              |
|------------------------------------------------------------------------------------------------------------------------------------------------------------|--------------------------------------------------------------|
| 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<br>2 Fax Ports Licences<br>8 Fax Ports Licences<br>30 Fax Ports Licences<br>60 Fax Ports Licences<br>120 Fax Ports Licences | PKS200<br><br>PKS202<br>PKS203<br>PKS204<br>PKS205<br>PKS206 |

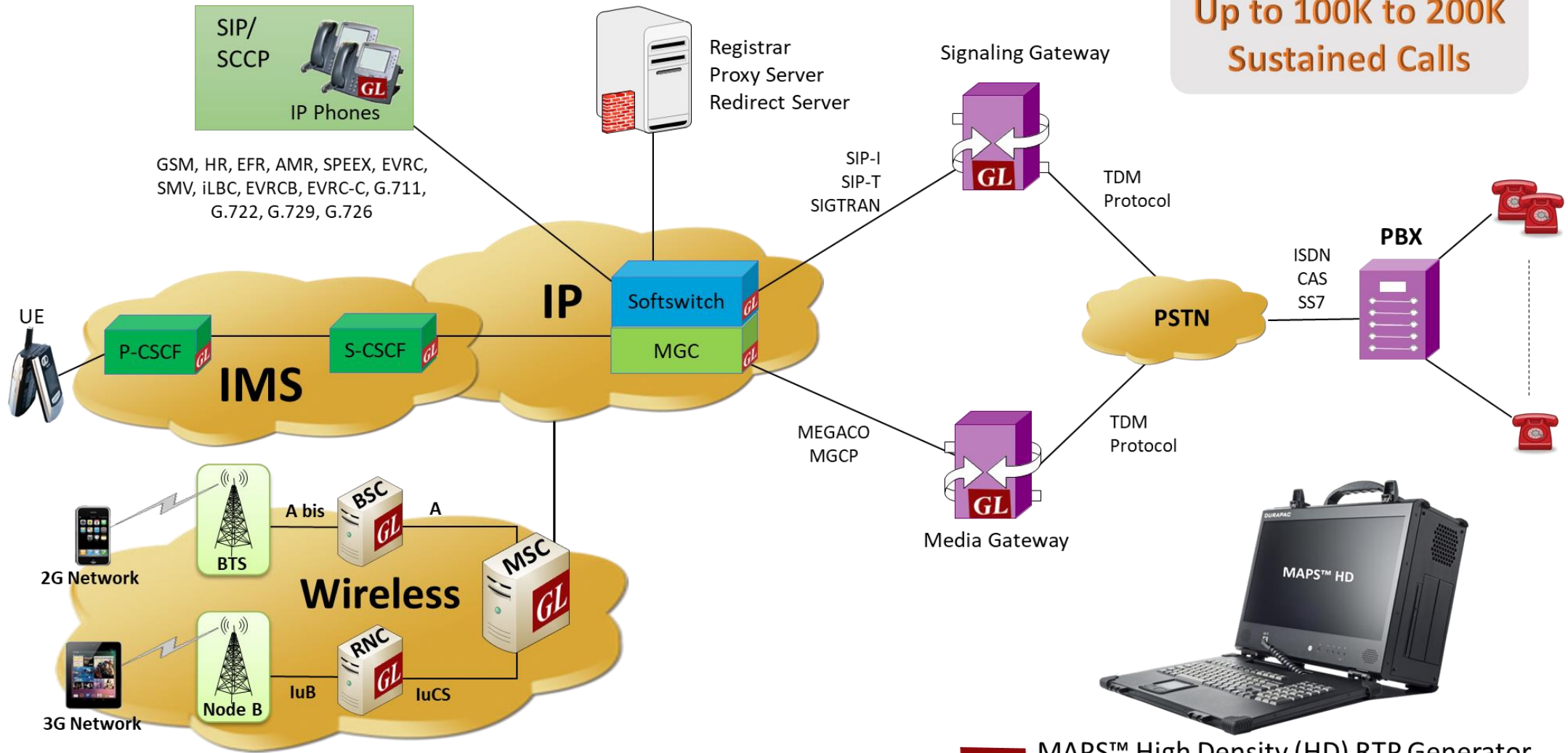
# IP Traffic (RTP) Simulation Capabilities and Performance

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



# High Density (HD) RTP Traffic Simulation

Up to 100K to 200K Sustained Calls



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



# RTP Traffic Capabilities and Performance

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

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

# 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)

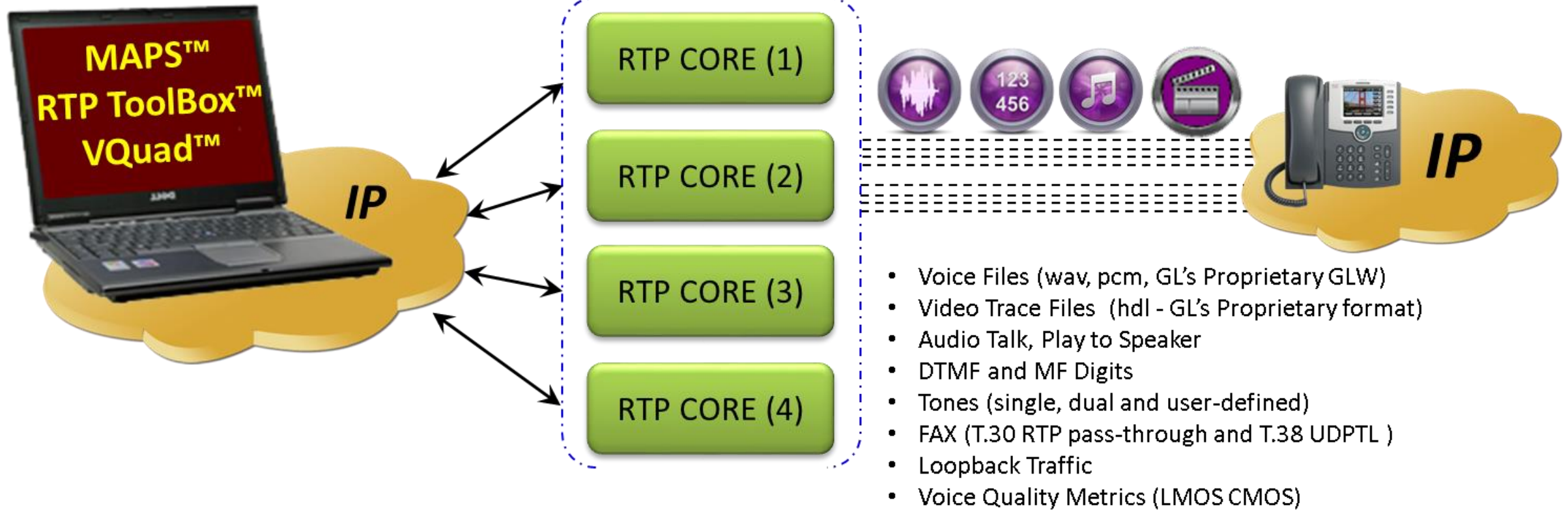
\*\* Some of these features requires additional licenses – contact GL for more information

# 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

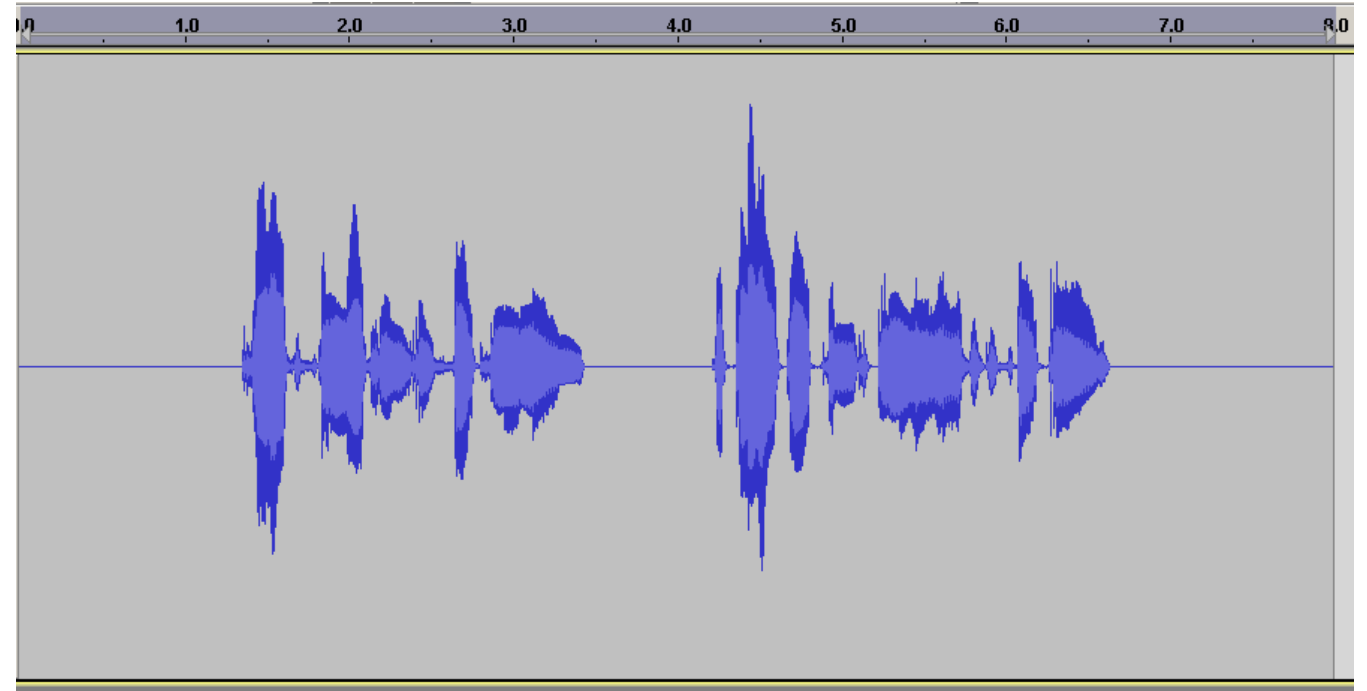
\*\* Some of these features requires additional licenses – contact GL for more information

# RTP Core



# 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,  $\mu$ -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

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface. The top window shows a table of call generation results with columns for Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, Events Profile, Result, and Total Iterat. Below the table is a control panel with buttons for Add, Delete, Insert, Refresh, Start, Start All, Stop, Stop All, Abort, and Abort All. The bottom window shows a detailed message sequence diagram between MAPS and DUT, including an INVITE message and subsequent responses like 100 Trying, 180 Ringing, 200 OK, and ACK. The right side of the bottom window displays the raw SIP message text.

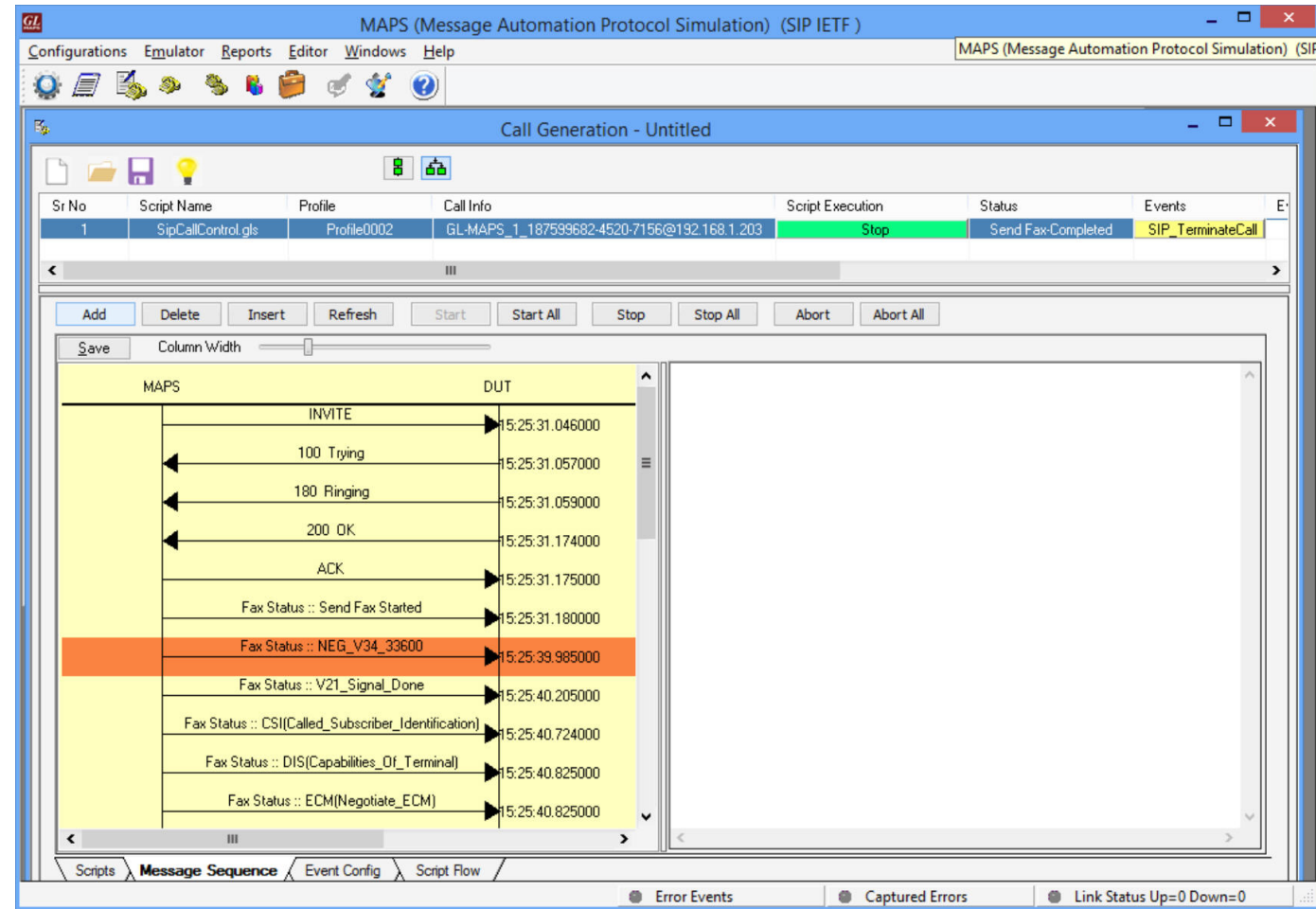
| Sr No | Script Name        | Profile     | Call Info                                   | Script Execution | Status            | Events            | Events Profile | Result | Total Iterat |
|-------|--------------------|-------------|---------------------------------------------|------------------|-------------------|-------------------|----------------|--------|--------------|
| 1     | SipCallControl.gls | Profile0001 | GL-MAPS_1_714522183-5030-1812@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 2     | SipCallControl.gls | Profile0002 | GL-MAPS_1_714522182-5026-5216@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 3     | SipCallControl.gls | Profile0003 | GL-MAPS_1_714522184-5062-3432@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 4     | SipCallControl.gls | Profile0004 | GL-MAPS_1_714522183-5034-1068@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 5     | SipCallControl.gls | Profile0005 | GL-MAPS_1_714522183-5041-3632@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 6     | SipCallControl.gls | Profile0006 | GL-MAPS_1_714522183-5040-1952@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 7     | SipCallControl.gls | Profile0007 | GL-MAPS_1_714522184-5058-1812@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 8     | SipCallControl.gls | Profile0008 | GL-MAPS_1_714522183-5050-5216@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 9     | SipCallControl.gls | Profile0009 | GL-MAPS_1_714522184-5067-3432@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |
| 10    | SipCallControl.gls | Profile0010 | GL-MAPS_1_714522183-5054-1068@192.168.1.141 | Stop             | Send_File-Started | SIP_TerminateCall |                | Pass   |              |

```
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

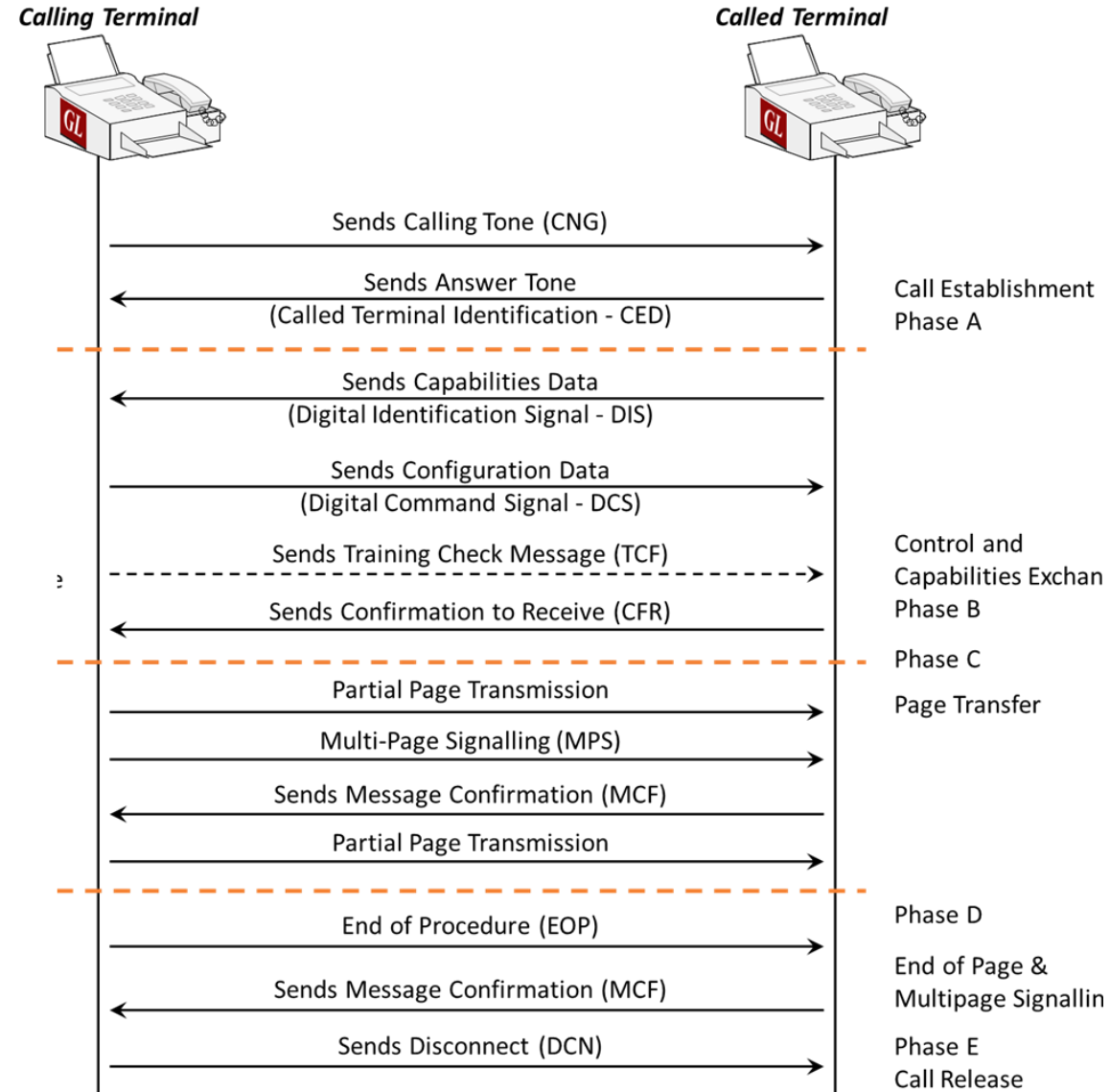
v=0
o=0001 33852938 33852938 IN IP4 192.168.1.141
s=SIP Call
c=IN IP4 192.168.1.141
t=0 0
m=audio 1086 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
```

# 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

MAPS (Message Automation Protocol Simulation) (SIP IETF)

Configurations Emulator Reports Editor Windows Help

Call Generation - CallGenDefault

Call Info Script Execution Status Events

GL-MAPS\_1\_18706384-3643-2292@192.168.1.141 Stop Fax Session Successful SIP\_TerminateCall

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

Save Column Width

200 OK 18:02:5  
ACK 18:02:5  
INVITE 18:02:5  
200 OK 18:02:5  
ACK 18:02:5  
Fax Status :: 33600\_Rate\_of\_V34\_selected\_after\_MP... 18:03:1  
33600\_Rate\_of\_V34\_selected\_after\_MPH\_exchange 18:03:1  
Fax Status :: CSI(Called\_Subscriber\_Identification) 18:03:1  
CSI(Called\_Subscriber\_Identification) 18:03:1  
Fax Status :: DIS(Digital\_Identification\_Signal) 18:03:1  
DIS(Digital\_Identification\_Signal) 18:03:1  
Fax Status :: ECM\_mode\_Selected\_in\_DCS 18:03:1

```

INVITE sip:0003@192.168.1.143 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.141:5060;branch=z9hG4bK_1_1
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NO
From: 0001 <sip:0003@192.168.1.141>;tag=FromTag_1_1870
To: 0001 <sip:0003@192.168.1.143>
Call-ID: GL-MAPS_1_18706384-3643-2292@192.168.1.141
CSeq: 2 INVITE
Contact: 0010 <sip:0003@192.168.1.141>
Content-Type: application/sdp
Content-Length: 359

v=0
o=0001 33852938 33852938 IN IP4 192.168.1.141
s=SIP Call
c=IN IP4 192.168.1.141
t=0
m=image 1036 udpt1 t38
a=T38FaxVersion:3
a=T38MaxBitRate:33600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:400
a=T38FaxMaxDatagram:280
--T38FaxT4ePC+28UPDRad...

```

Events

Event Log Error Events Captured Errors

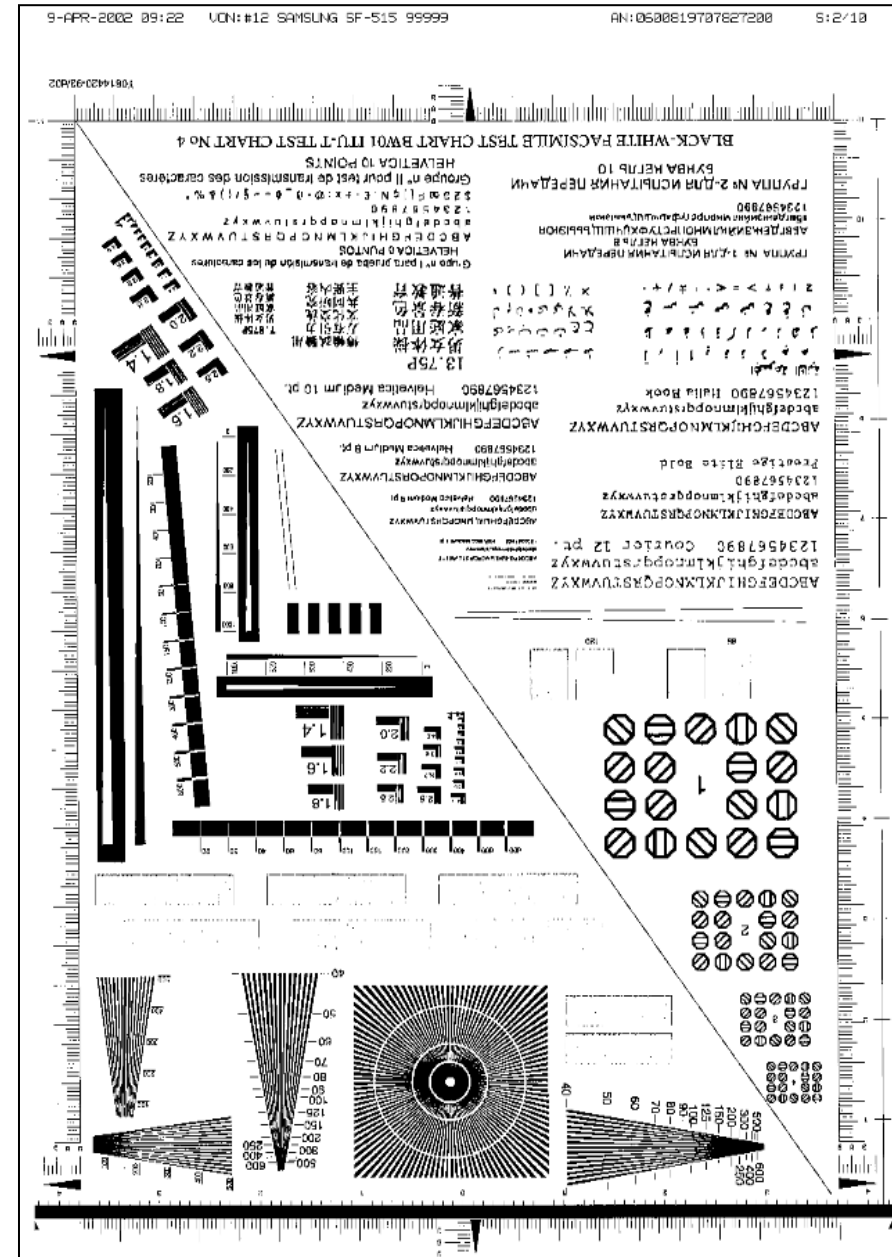
| Date/Time                 | Captured Events                                             |
|---------------------------|-------------------------------------------------------------|
| 2015-9-29 17:58:19.820000 | Fax - Status: DCS(Digital_Command_Signal)                   |
| 2015-9-29 17:58:19.820000 | Fax - Status: CFR(Confirmation_To_Receive)                  |
| 2015-9-29 17:58:19.821000 | Fax - Status: CFR(Confirmation_To_Receive)                  |
| 2015-9-29 17:58:19.822000 | Fax - Status: PPS_EOP(Partial_Page_Signal_End_Of_Procedure) |
| 2015-9-29 17:58:19.823000 | Fax - Status: DCN(Disconnect)                               |
| 2015-9-29 17:58:19.823000 | Fax - Status: Fax Session Successful                        |
| 2015-9-29 18:01:54.855000 | BYE Sent                                                    |
| 2015-9-29 18:01:54.870000 | 200 Ok to BYE Received                                      |
| 2015-9-29 18:01:54.870000 | Call Terminated                                             |
| 2015-9-29 18:01:54.870000 | Inter Call Duration = 1000                                  |
| 2015-9-29 18:02:54.776000 | INVITE Sent                                                 |
| 2015-9-29 18:02:54.786000 | PROGRESS Received                                           |
| 2015-9-29 18:02:54.786000 | PROGRESS Received                                           |
| 2015-9-29 18:02:54.788000 | PROGRESS Received                                           |
| 2015-9-29 18:02:54.788000 | PROGRESS Received                                           |
| 2015-9-29 18:02:54.908000 | ACK Sent                                                    |
| 2015-9-29 18:02:54.909000 | Call Connected                                              |
| 2015-9-29 18:02:54.927000 | ACK Sent                                                    |
| 2015-9-29 18:03:19.875000 | Fax - Status: Sending Fax                                   |
| 2015-9-29 18:03:19.876000 | Fax - Status: DIS(Digital_Identification_Signal)            |
| 2015-9-29 18:03:19.877000 | Fax - Status: DIS(Digital_Identification_Signal)            |
| 2015-9-29 18:03:19.877000 | Fax - Status: DIS(Digital_Identification_Signal)            |
| 2015-9-29 18:03:19.878000 | Fax - Status: DIS(Digital_Identification_Signal)            |
| 2015-9-29 18:03:19.879000 | Fax - Status: DCS(Digital_Command_Signal)                   |
| 2015-9-29 18:03:19.879000 | Fax - Status: DCS(Digital_Command_Signal)                   |
| 2015-9-29 18:03:19.880000 | Fax - Status: DCS(Digital_Command_Signal)                   |
| 2015-9-29 18:03:19.881000 | Fax - Status: CFR(Confirmation_To_Receive)                  |
| 2015-9-29 18:03:19.881000 | Fax - Status: CFR(Confirmation_To_Receive)                  |
| 2015-9-29 18:03:19.882000 | Fax - Status: PPS_EOP(Partial_Page_Signal_End_Of_Procedure) |
| 2015-9-29 18:03:19.883000 | Fax - Status: DCN(Disconnect)                               |
| 2015-9-29 18:03:19.883000 | Fax - Status: Fax Session Successful                        |

Save Events

Clear  Capture Events to file

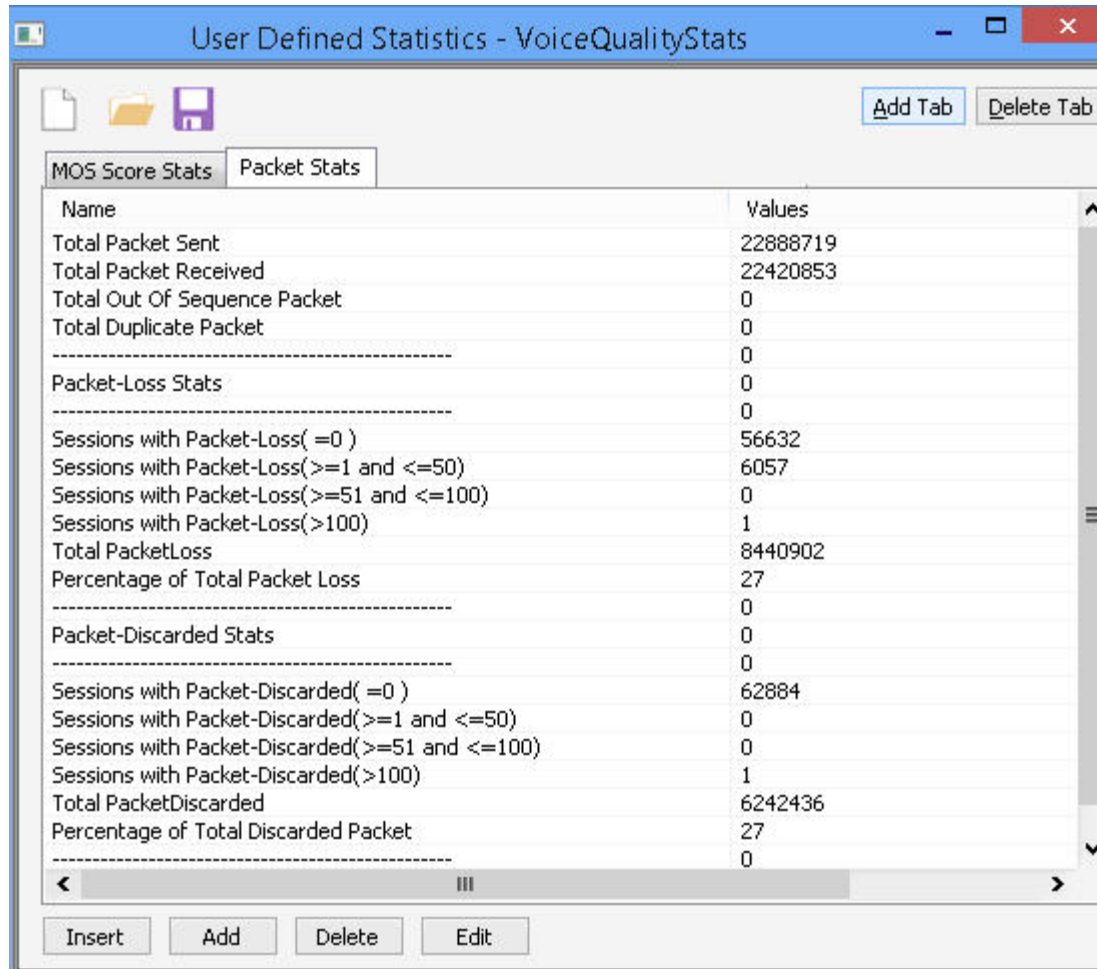
# RTP - Fax File

- 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



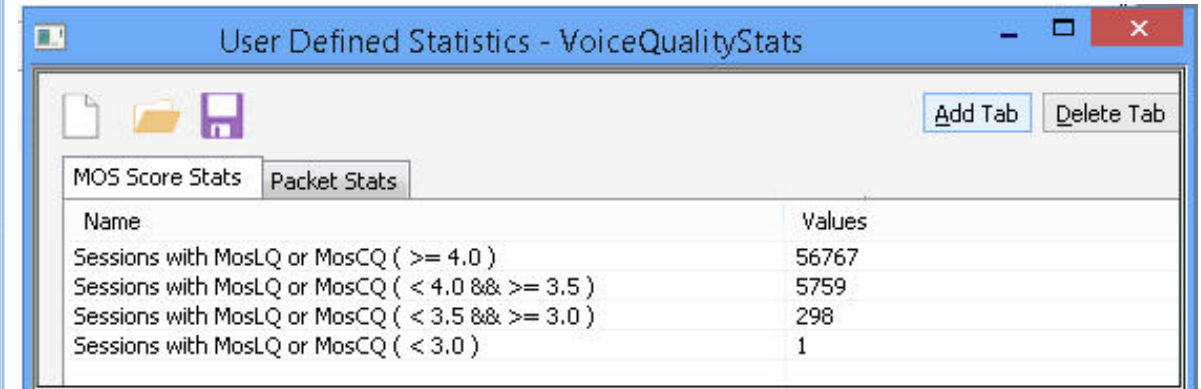
# RTP Voice Quality Measurements

- 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



The screenshot shows a window titled "User Defined Statistics - VoiceQualityStats" with a "Packet Stats" tab selected. The window contains a table with two columns: "Name" and "Values". The table lists various statistics related to packet loss and discarding.

| 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        |



The screenshot shows a window titled "User Defined Statistics - VoiceQualityStats" with a "MOS Score Stats" tab selected. The window contains a table with two columns: "Name" and "Values". The table lists statistics related to MOS scores.

| 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      |

# Call Generation with T.30 Pass through FAX Traffic Type

Call Generation - CallGenDefault

| Sr No | Script Name        | Profile     | Call Info                                   | Script Execution | Status             | Events            |
|-------|--------------------|-------------|---------------------------------------------|------------------|--------------------|-------------------|
| 1     | SipCallControl.gls | Profile0002 | GL-MAPS_1_278417817-8540-5844@192.168.1.203 | Stop             | Send Fax-Completed | SIP_TerminateCall |

Column Width

| MAPS                                                | DUT             |
|-----------------------------------------------------|-----------------|
| INVITE                                              | 16:39:09.193000 |
| 100 Trying                                          | 16:39:09.203000 |
| 180 Ringing                                         | 16:39:09.205000 |
| 200 OK                                              | 16:39:09.311000 |
| ACK                                                 | 16:39:09.312000 |
| Fax Status :: Send Fax Started                      | 16:39:09.315000 |
| Fax Status :: NEG_V34_33600                         | 16:39:18.118000 |
| Fax Status :: V21_Signal_Done                       | 16:39:18.338000 |
| Fax Status :: CSI(Called_Subscriber_Identification) | 16:39:18.858000 |
| Fax Status :: DIS(Capabilities_Of_Terminal)         | 16:39:18.958000 |
| Fax Status :: ECM(Negotiate_ECM)                    | 16:39:18.958000 |

```

INVITE sip:0002@192.168.1.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_278417817-85
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UP
From: "MapsSip" <sip:0002@192.168.1.203>;tag=FromTag_1_278417817-
To: 0001 <sip:0002@192.168.1.213>
Call-ID: GL-MAPS_1_278417817-8540-5844@192.168.1.203
CSeq: 1 INVITE
Contact: 0010 <sip:0002@192.168.1.203>
Content-Type: application/sdp
Content-Length: 317

v=0
o=0002 33852938 33852938 IN IP4 192.168.1.203
s=-SIP Call
c=IN IP4 192.168.1.203
t=0 0
m=audio 1038 RTP/AVP 0 8 18 3 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
==end==

```



# 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

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface. The main window shows a table of call simulation results. Below the table is a control panel with buttons for 'Add', 'Delete', 'Insert', 'Refresh', 'Start', 'Start All', 'Stop', 'Stop All', 'Abort', and 'Abort All'. A detailed SIP message trace is visible, showing the sequence of messages between MAPS and DUT (Device Under Test).

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

**SIP Message Trace:**

```
MAPS DUT

INVITE 10:55:08.130000
100 Trying 10:55:08.147000
180 Ringing 10:55:08.149000
200 OK 10:55:08.280000
ACK 10:55:08.286000
```

**SIP Message Content:**

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

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

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

# MSRP Traffic Simulation

MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - BulkCalls\_10]

Configurations Emulator Reports Editor Debug Tools Windows Help

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

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

Save Column Width Show Latest

MAPS DUT

```

15:39:35.705000 INVITE ->
15:39:35.727000 100 Trying <-
15:39:35.737000 180 Ringing <-
15:39:35.859000 200 OK <-
15:39:35.861000 ACK ->
15:39:35.909000 SEND ->
15:39:35.949000 200 OK <-
15:39:35.991000 REPORT <-
15:39:35.991000 SEND ->
15:39:35.992000 200 OK <-
15:39:36.010000 REPORT <-
15:39:36.943000 SEND ->
15:39:36.998000 200 OK <-
15:39:37.030000 REPORT <-
15:39:37.030000 SEND ->
15:39:37.030000 200 OK <-
15:39:37.040000 REPORT <-

```

Find

```

MSRP_g1MapsMsrpBB9A66F9-153935908-6777 SEND
To-Path: msrp://192.168.12.209:20148/GL_MAPS_302_86849888;tcp
From-Path: msrp://192.168.12.216:20151/GL_MAPS_464_86849744;tcp
Message-ID: g1MapsMsrpBB9A66F9-153935908-6776
Success-Report: no
Failure-Report: yes
Byte-Range: 1-270/270
Content-Type: text/plain

GL's Message Automation & Protocol
-----g1MapsMsrpBB9A66F9-153935908-6777

```

Scripts Message Sequence Event Config Script Flow

User Defined Statistics - User\_Defined\_Statistics

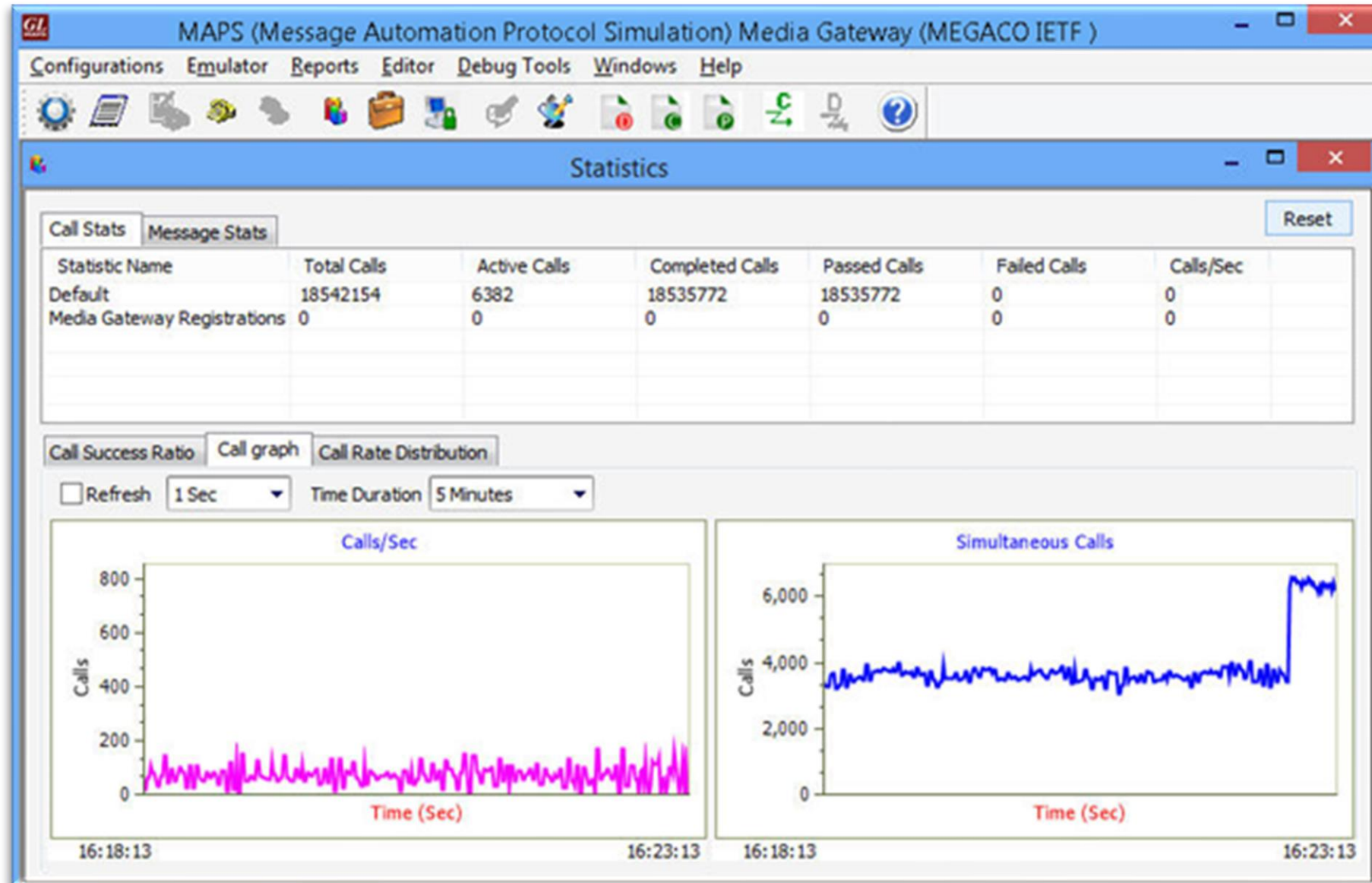
Add Tab Delete Tab

MSRP Statistics Voice Quality Statistics

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

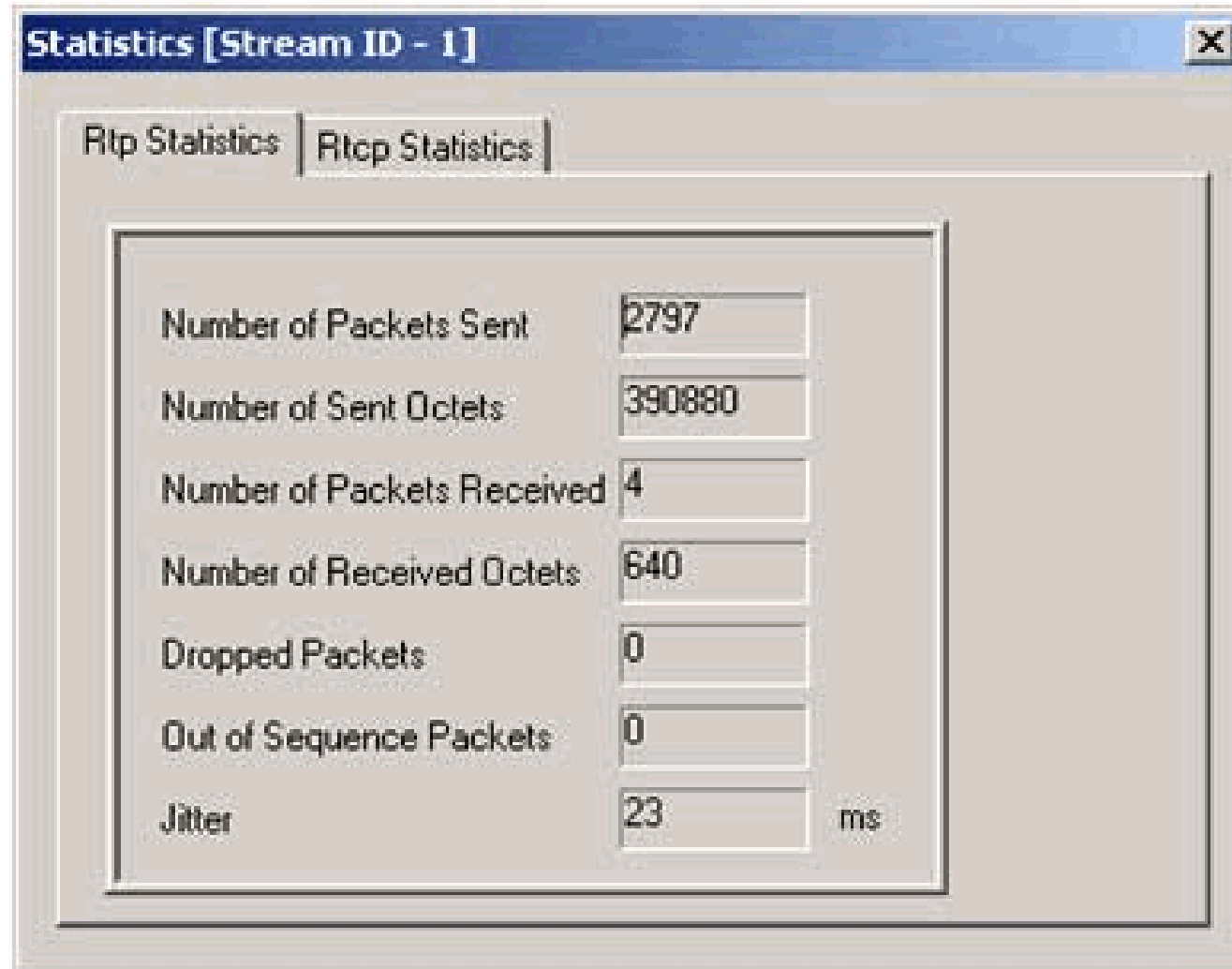
Insert Add Delete Edit

# 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

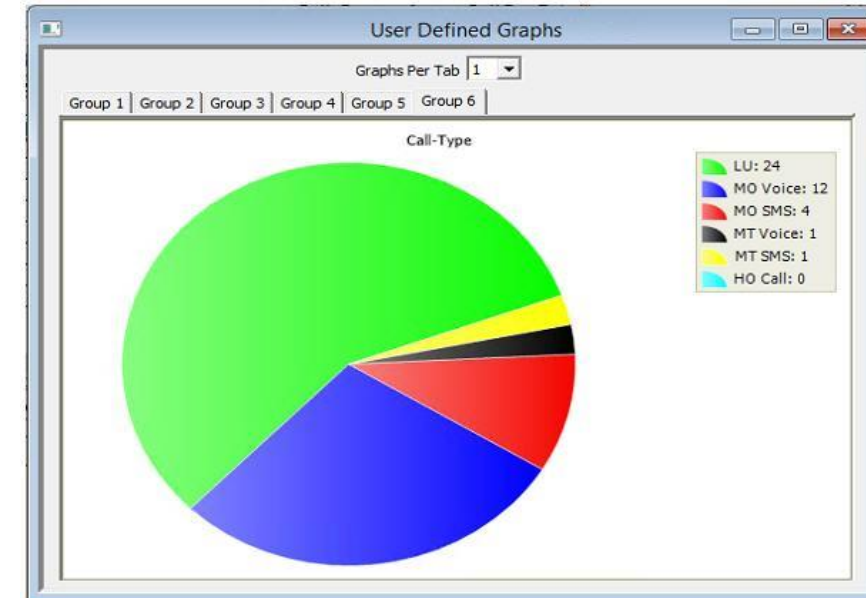
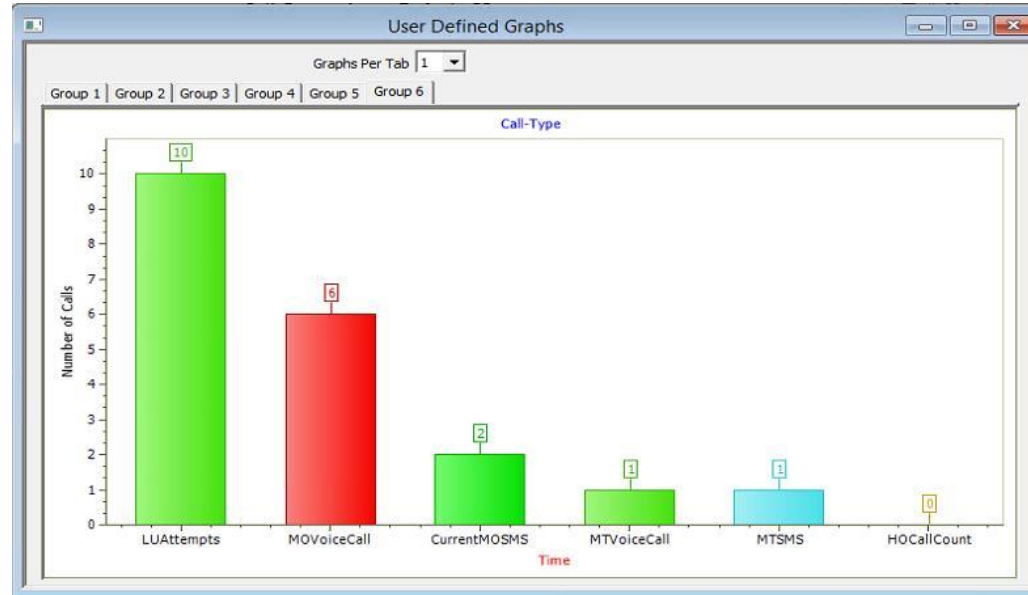


# Speech Quality Metrics (R Factor and MOS)

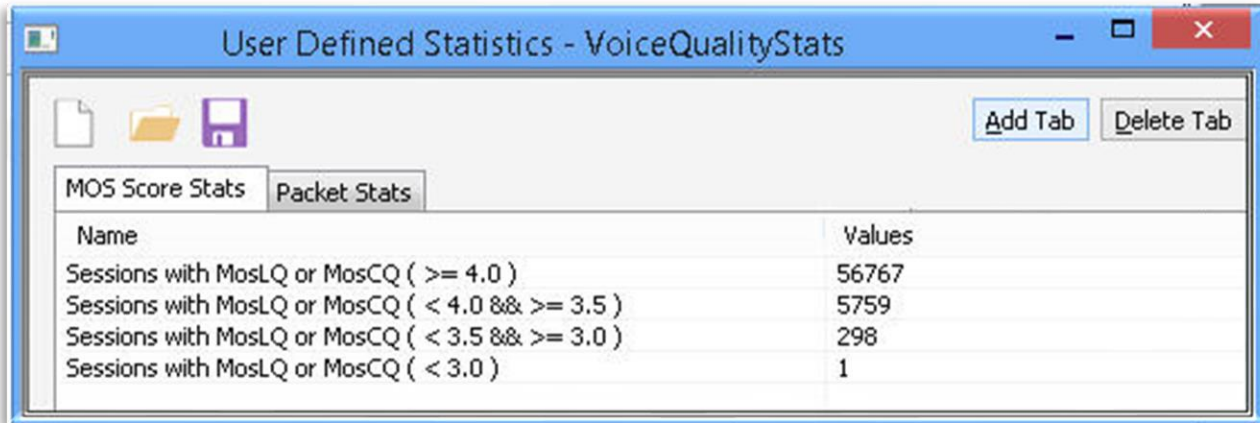
- Quality metrics include R-Factor, Listening and Conversational Quality MOS scores, Packet Loss, Discarded Packets, Out of Sequence Packets and Duplicate Packets
- R Factor graph will display statistics such as, R-Listening, R-Conversational, R-G107 and R-Nom. MOS Factor graph will display statistics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

User Defined Statistics - User\_Defined\_Statistics

| Name                                      | Values       |
|-------------------------------------------|--------------|
| Active RTP Sessions                       | 0            |
| Completed RTP Sessions                    | 2466         |
| Sessions With Zero Receive Traffic        | 49           |
| MOS Score Stats                           | 0            |
| Sessions with Mos ( 5.0 - 4.0 )           | 189 [7%]     |
| Sessions with Mos ( 4.0 - 3.0 )           | 421 [17%]    |
| Sessions with Mos ( 3.0 - 2.0 )           | 300 [12%]    |
| Sessions with Mos ( < 2.0 )               | 1508 [61%]   |
| Total RTP Packet Sent                     | 1992201      |
| Total RTP Packet Received                 | 2291987      |
| Packet-Loss Stats                         | 0            |
| Total PacketLoss                          | 224369 [9%]  |
| Sessions with Zero Packet-Loss            | 1 [0%]       |
| Sessions with Packet-Loss(<1%)            | 119 [4%]     |
| Sessions with Packet-Loss(1% - 5%)        | 801 [32%]    |
| Sessions with Packet-Loss(5% - 10%)       | 471 [19%]    |
| Sessions with Packet-Loss(>10%)           | 1026 [41%]   |
| Packet-Discarded Stats                    | 0            |
| Total PacketDiscarded                     | 550956 [24%] |
| Sessions with Zero Packet-Discard         | 1 [0%]       |
| Sessions with Packet-Discard(<1%)         | 2 [0%]       |
| Sessions with Packet-Discard(1% - 5%)     | 389 [15%]    |
| Sessions with Packet-Discard(5% - 10%)    | 288 [11%]    |
| Sessions with Packet-Discard(>10%)        | 1738 [70%]   |
| Packet-Duplicate Stats                    | 0            |
| Total Duplicate Packet                    | 0 [0%]       |
| Sessions with Zero Duplicate Packets      | 2418 [98%]   |
| 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            | 2418 [98%]   |
| 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)           | 124 [5%]     |
| Sessions with Jitter( < 5 msec)           | 212 [8%]     |
| Sessions With Jitter(< 10 msec)           | 278 [11%]    |
| Sessions With Jitter(>= 10 msec)          | 1804 [73%]   |

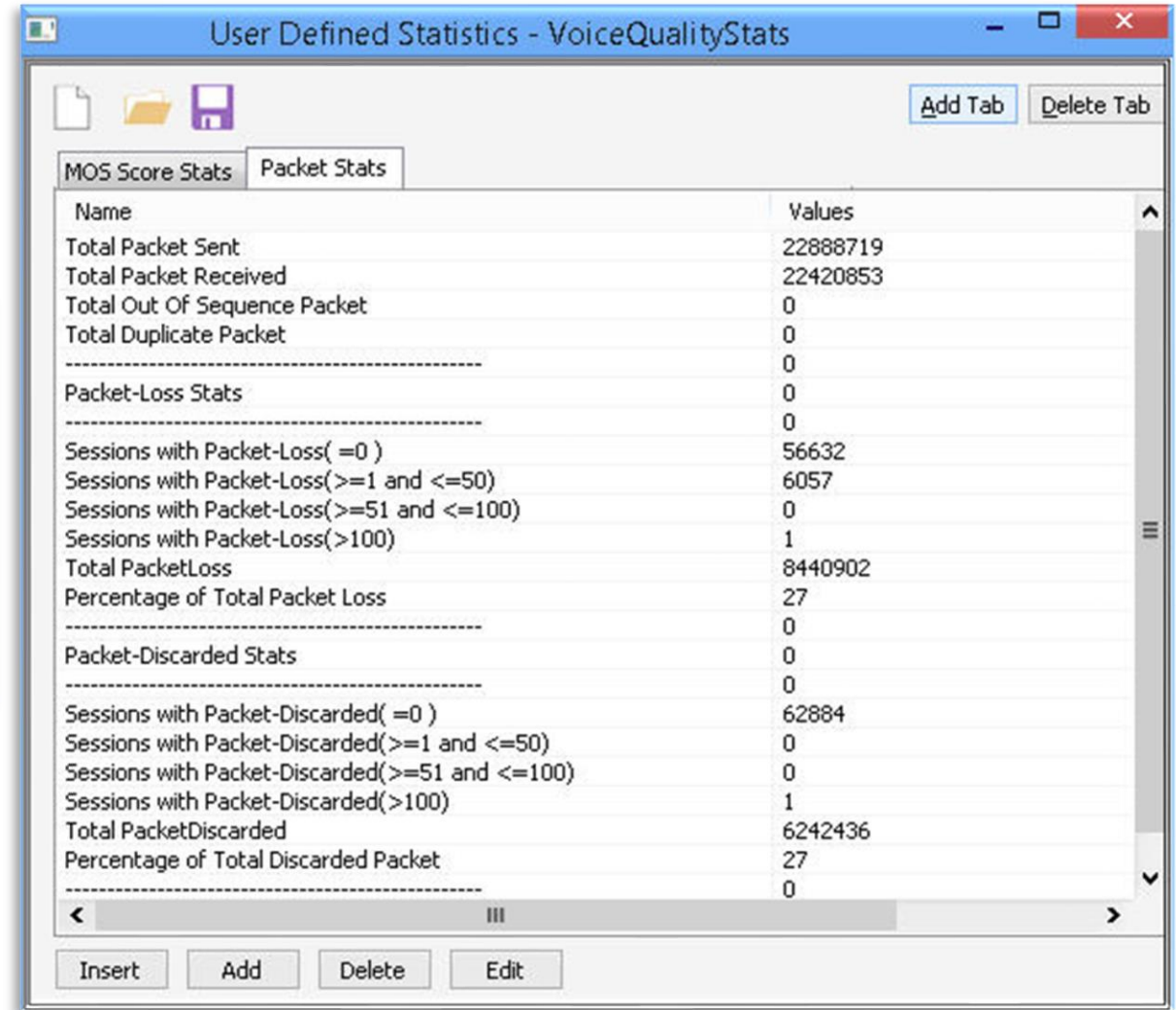


# Speech Quality Metrics (R Factor and MOS) (Contd.)



The screenshot shows a window titled "User Defined Statistics - VoiceQualityStats" with a tab labeled "MOS Score Stats". The window contains a table with the following data:

| Name                                                   | Values |
|--------------------------------------------------------|--------|
| Sessions with MosLQ or MosCQ ( $\geq 4.0$ )            | 56767  |
| Sessions with MosLQ or MosCQ ( $< 4.0$ && $\geq 3.5$ ) | 5759   |
| Sessions with MosLQ or MosCQ ( $< 3.5$ && $\geq 3.0$ ) | 298    |
| Sessions with MosLQ or MosCQ ( $< 3.0$ )               | 1      |

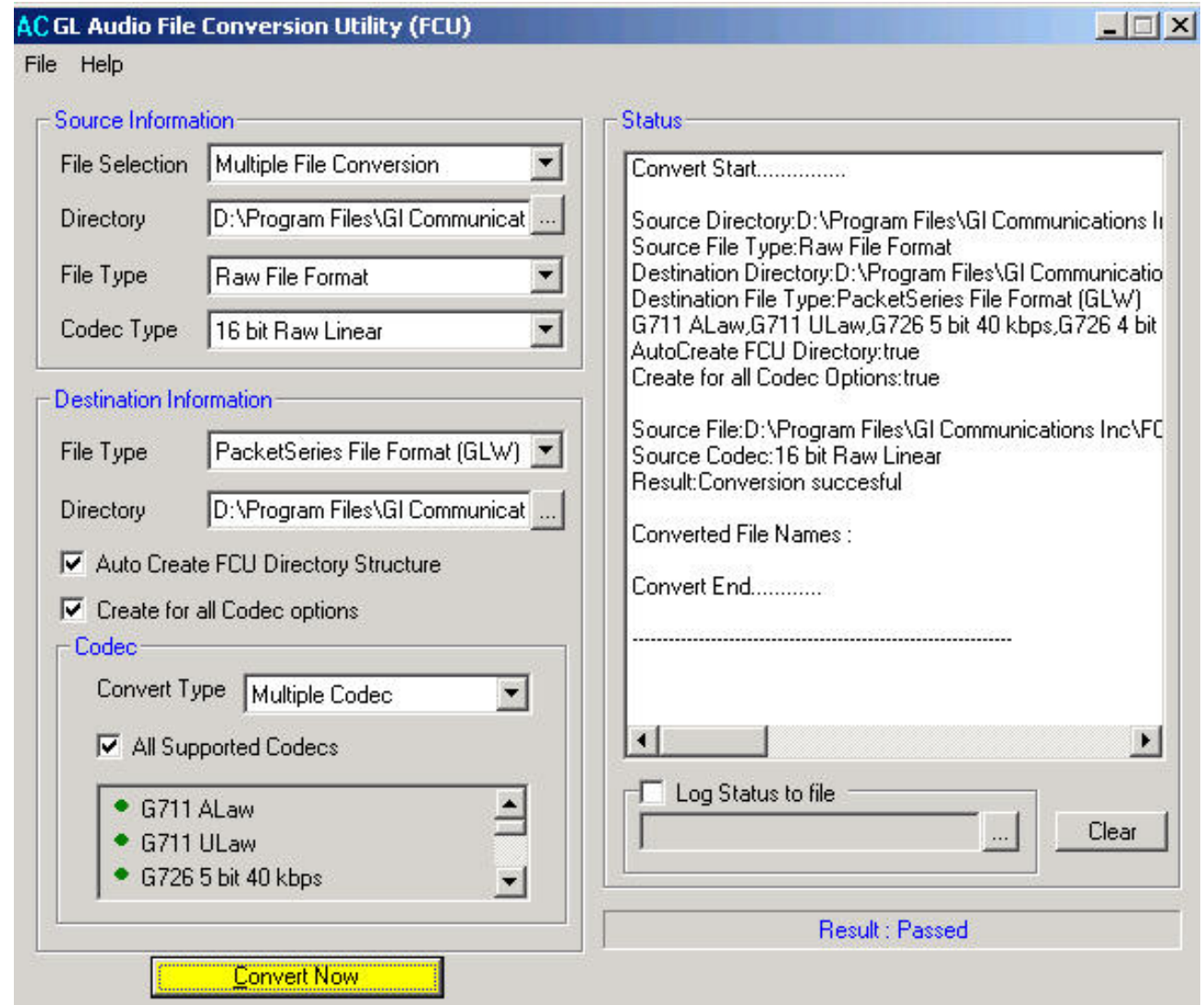


The screenshot shows a window titled "User Defined Statistics - VoiceQualityStats" with a tab labeled "Packet Stats". The window contains a table with the following data:

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

# Audio File Converter Utility (AFCU)

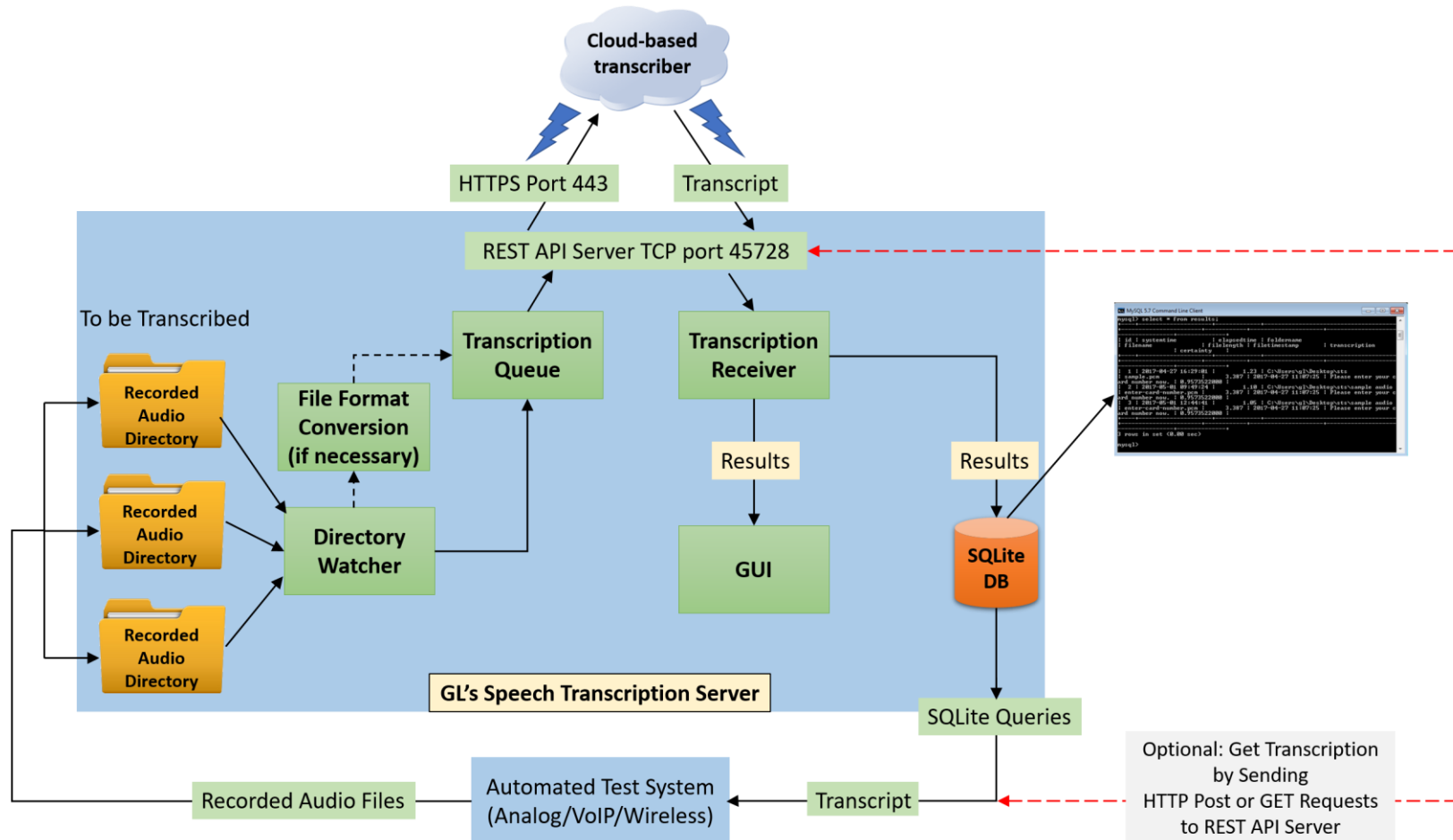
- GL's Audio File Conversion Utility (Audio FCU) is generally used in conjunction with GL Packet Series products to further enhance send and record voice file capabilities
- This utility supports almost all industry standard voice codec data formats, that helps to convert recorded voice files from their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



# SMS Traffic Simulation

- 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

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface for a call reception simulation. The window title is "MAPS (Message Automation Protocol Simulation) (GsmAlp) (MAPIP) - [Call Reception]".

**Call Records Table:**

| Sr No | Script Name   | Call Info                                                                               | Script Execution | Status                   | Events | E... | Results |
|-------|---------------|-----------------------------------------------------------------------------------------|------------------|--------------------------|--------|------|---------|
| 10    | GSMA_Call.gls | IMSI: 901700000000625, CallingNumber: 90625, IMSI: 901700000000627, CalledNumber: 90627 | Completed        | SCCP Connection Released | None   |      | Pass    |
| 11    | GSMA_Call.gls | IMSI: 901700000000625, IMSI: 0x00000002                                                 | Completed        | SCCP Connection Released | None   |      | Pass    |
| 12    | GSMA_Call.gls | IMSI: 901700000000627, CalledNumber: 90627, CallingNumber: 90625                        | Completed        | SCCP Connection Released | None   |      | Pass    |

**Message Sequence Diagram:**

The diagram shows the interaction between BSC, MSC, HLR, and SMSC. Key events include:

- CM SERVICE REQUEST (BSC to MSC)
- CC connection confirm (MSC to BSC)
- IDENTITY REQUEST (BSC to MSC)
- IDENTITY RESPONSE (MSC to BSC)
- sendAuthenticationInfoArg (MSC to HLR)
- sendAuthenticationInfoRes (HLR to MSC)
- AUTHENTICATION REQUEST (BSC to MSC)
- AUTHENTICATION RESPONSE (MSC to BSC)
- CIPHER MODE COMMAND (BSC to MSC)
- CIPHER MODE COMPLETE (MSC to BSC)
- CM SERVICE ACCEPT (BSC to MSC)
- MM STATUS (BSC to MSC)
- SMS-SUBMIT (BSC to MSC) - **Highlighted in orange**
- mo-forwardSMArg (MSC to SMSC)
- mo-forwardSMRes (SMSC to MSC)
- CP-ACK (MSC to BSC)
- SMS-SUBMIT-REPORT (MSC to BSC)
- CP-ACK (BSC to MSC)
- CLEAR COMMAND (BSC to MSC)
- CLEAR COMPLETE (MSC to BSC)
- RLSD released (BSC to MSC)
- RLC release complete (MSC to BSC)

**Protocol Stack View:**

The right pane shows the protocol stack details for the SMS-SUBMIT message:

- MTP3 User Adaptation Layer:**
  - 0000 Version = 00000001 Release 1.0
  - 0002 Message Class = 00000001 Transfer
  - 0003 Transfer Message Type = 00000001 Payload Data
  - 0004 Message Length = 80 (x00000050)
  - 0008 Tag = x0210 Transfer Protocol
  - 000A Length = 69 (x0045)
  - 000E Originating Point Code = 1.1.1(...001000 00001001)
  - 0012 Point Code = 2.2.2(...010000 00010010)
  - 0014 Service Indicator = ...0011 SCCP
  - 0015 Network Indicator = ...00 International
  - 0016 Message Priority = ...00 Priority Code
  - 0017 Signalling Link Selection = ...0001 (1)
  - Pdu = x0600001A00012E01002B39
  - Parameter Padding = x000000
- SCCP Layer:**
  - 0018 Message Type = 00000110 DT1 data form
  - 0019 Destination Local Reference = 26 (x00001A)
  - 001C More Data Indicator = ...0 No more data
  - 001D Pointer to Mandatory Parameter = Parm0 offset x01 (1)
  - 001E Mandatory Variable Length Parameters = mandatory parameter
  - Data = 45...
  - Optional Variable Length Parameters = None
- GSM Phase2+ Layer:**
  - 001F Discrimination bit D = ...1 DTAP
  - 0020 SAPI = ...000 Signalling
  - 0020 Control channel identification = 00... Not Specified
  - 0021 Message Length = 43 (x2B)
  - Layer 3 Information = x39012800010007918892000
- Layer3 Protocol Layer:**
  - 0022 Protocol Discriminator = ...1001 SMS messages
  - 0022 TI Flag = 0... The message is
  - 0022 TIO = .011... (3)
  - L3 Info = x0128000100079188920000
- SMS Layer:**
  - 0023 CP-Message Type = 00000001 CP-DATA
  - 0024 Length of RPDU = 40 (x28)
  - RP-Message-Header =
  - 0025 RP-Message\_Type\_Ind (MTI) = ...000 RP-DATA (MS->SC)
  - 0026 RP-Message reference = 00000001 (1)
  - Originator address = 0 (x00)
  - Changeable Length =
  - Destination address =
  - 0028 Length of Destination address = 7 (x07)

# 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

**Correlation & Audio Analysis**

Show Device: All

Speech Transcription Server  
 Server IP: 50.76.16.181 Refresh  
 Server is Running  
 Note: Transfer voice file to server's PC first then do "Speech To Text".

Transfer Speech To Text  
 Encoding: PCM16 NB (8kHz) Speech To Text  
 Voice File Name: C:\WQT\_Degraded\STT\VoicePrompt\_2.pcm  
 Reference String: Your Call has been forwarded to automatic voice message system  
 Pass Factor (%): 100 Text Matching Word Matching

| Timestamp              | Device ID | Type         | Events                                                                          |
|------------------------|-----------|--------------|---------------------------------------------------------------------------------|
| 05/01/2018 12:16:09 PM |           | Status       | VoiceFile=C:\WQT_Degraded\STT\VoicePrompt_2.pcm                                 |
| 05/01/2018 12:16:09 PM |           | Status       | Reference=Your Call has been forwarded to automatic voice message system        |
| 05/01/2018 12:16:09 PM |           | Status       | Encoding=PCM16 NB (8kHz)                                                        |
| 05/01/2018 12:16:13 PM |           | Result       | SpeechToText Result...                                                          |
| 05/01/2018 12:16:13 PM |           | Result       | certainty=0.9486                                                                |
| 05/01/2018 12:16:13 PM |           | Result       | timeTaken=3.026                                                                 |
| 05/01/2018 12:16:13 PM |           | Result       | transcription=Your call has been forwarded to an automatic voice message system |
| 05/01/2018 12:16:14 PM |           | SpchAnalysis | Pass (100% pass)                                                                |

Status: FxCorr Delay (ms): 144.5, Correlation: 0.8 SpeechToText Server

| Sr No | Script Name      | Profile | Call Info     | Script Execution | Status        | Events | Events Profile | Result |
|-------|------------------|---------|---------------|------------------|---------------|--------|----------------|--------|
| 1     | APS_PlaceCallgls | Line001 | Line001.1.1.0 | Start            | CALL_RELEASED | None   |                | Pass   |

MAPS DUT

- Onhook :: 0,1,0,1 16:13:27.232.5401
- Offhook :: 1,1,1,1 16:13:29.235.7393
- Tone Detected :: Dial Tone 16:13:34.29.9091
- Dialing :: 3016704784 16:13:36.229.2807
- Recording Prompt 1 :: Line001\_Prompt1\_2018-4-18-16-13-36.pcm 16:13:36.231.1507
- Prompt 1 Recorded 16:13:55.167.1626
- [PASS] Transcript :: Welcome to GL Communications if you know your parties extension You can do... 16:13:59.281.6087
- IVR Response :: DTMF 3 16:14:00.216.99
- Recording Prompt 2 :: Line001\_Prompt2\_2018-4-18-16-14-0.pcm 16:14:00.216.2903
- Prompt 2 Recorded 16:14:10.286.2970
- [PASS] Transcript :: Welcome to the directory please enter the first 3 letters of your party the last name... 16:14:13.141.8486
- IVR Response :: DTMF 926 16:14:15.275.1835
- Recording Prompt 3 :: Line001\_Prompt3\_2018-4-18-16-14-15.pcm 16:14:15.275.4676
- Prompt 3 Recorded 16:14:22.286.1126
- [PASS] Transcript :: IK space YANG if this is the person you're looking for press one now. 16:14:25.141.9620
- IVR Response :: DTMF 1 16:14:26.76.5589
- Tone Detected :: Ringback Tone 16:14:32.120.5593
- Onhook :: 0,1,0,1 16:14:36.232.9896

File name: Line001\_Prompt1\_2018-4-18-16-13-36.pcm  
 File length: 18.798  
 Transcript: Welcome to GL Communications if you know your parties extension You can do...  
 Certainty: 0.8775  
 Phrase 1: Welcome to GL Communications | Found: true  
 Phrase 2: sales press one | Found: true  
 Phrase 3: technical support | Found: true  
 Phrase 4: directory by last name | Found: true

# Call Generation with IVR Traffic Type

Call Generation - CallGenDefault

| Sr No | Script Name        | Profile     | Call Info                                   | Script Execution | Status          | Events            | Ever |
|-------|--------------------|-------------|---------------------------------------------|------------------|-----------------|-------------------|------|
| 1     | SipCallControl.gls | Profile0001 | GL-MAPS_1_278015271-8508-5532@192.168.1.203 | Stop             | Digits Detected | SIP_TerminateCall |      |

Save Column Width

| MAPS                                 | DUT                    |
|--------------------------------------|------------------------|
|                                      | INVITE 16:32:26.653000 |
| 100 Trying                           | 16:32:26.666000        |
| 180 Ringing                          | 16:32:26.666000        |
| 200 OK                               | 16:32:26.775000        |
| ACK                                  | 16:32:26.776000        |
| Digits Transmitted :: 1234567890ABCD | 16:32:29.601000        |
| Digits Detected :: 1234567890ABCD    | 16:32:31.532000        |

```

INVITE sip:0001@192.168.1.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_278015271-85
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UP
From: "MapsSip" <sip:0001@192.168.1.203>;tag=FromTag_1_278015271-
To: 0001 <sip:0001@192.168.1.213>
Call-ID: GL-MAPS_1_278015271-8508-5532@192.168.1.203
CSeq: 1 INVITE
Contact: 0010 <sip:0001@192.168.1.203>
Content-Type: application/sdp
Content-Length: 317

v=0
c=0001 33852938 33852938 IN IP4 192.168.1.203
s=-SIP Call
c=IN IP4 192.168.1.203
t=0 0
m=audio 1030 RTP/AVP 0 8 18 3 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20

```

# Call Generation with File Traffic Type

| Sr No | Script Name        | Profile     | Call Info                                   | Script Execution | Status              | Events            | Evt |
|-------|--------------------|-------------|---------------------------------------------|------------------|---------------------|-------------------|-----|
| 1     | SipCallControl.gls | Profile0003 | GL-MAPS_1_281910197-8564-3132@192.168.1.203 | Stop             | Send_File-Completed | SIP_TerminateCall |     |

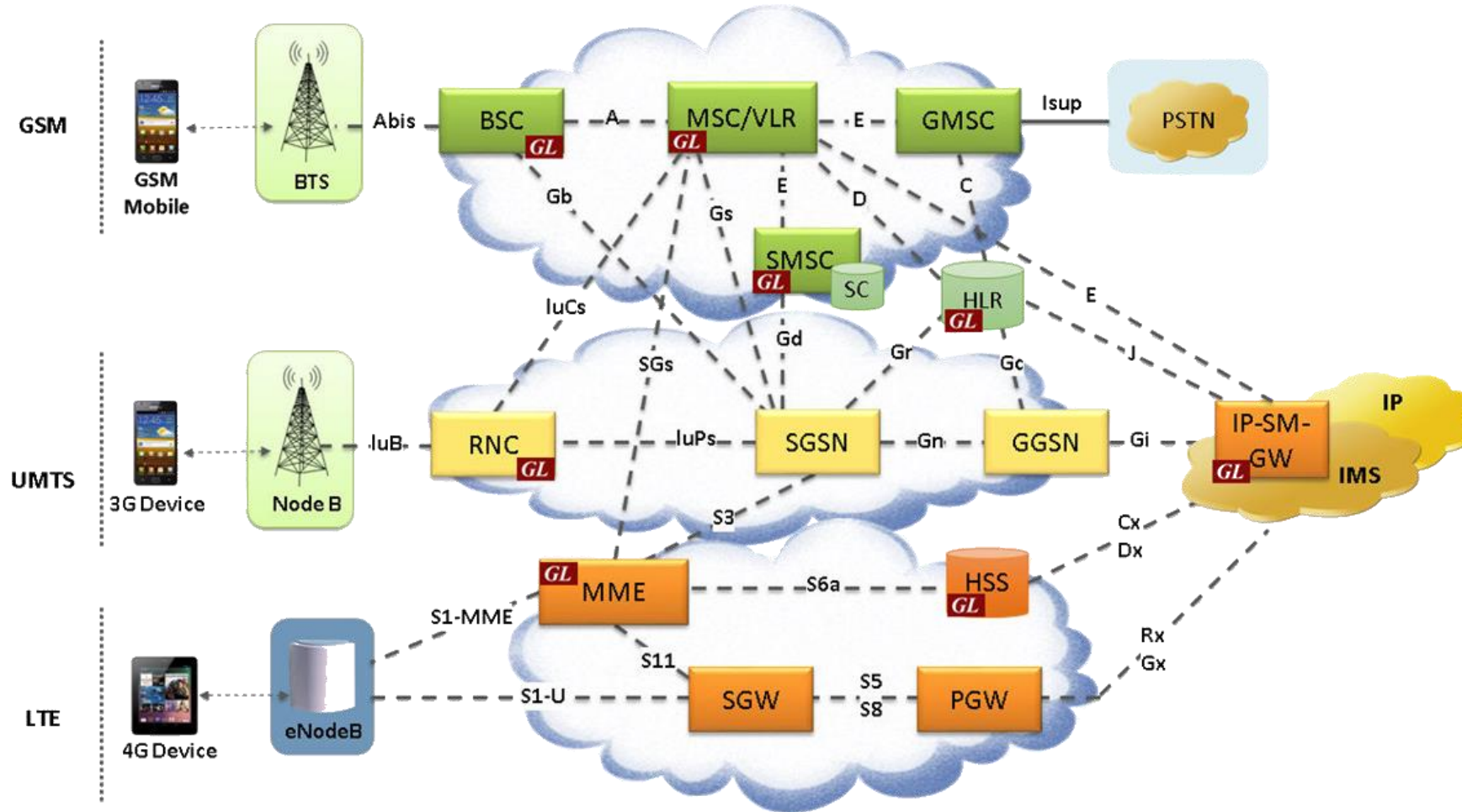
MAPS DUT

- INVITE → 17:37:21.574000
- 100 Trying ← 17:37:21.584000
- 180 Ringing ← 17:37:21.586000
- 200 OK ← 17:37:21.698000
- ACK → 17:37:21.699000
- File Recorded :: C:\Program Files\GL Communications L... ← 17:37:41.739000
- File Transmitted :: send\g711\ulaw\vijay.glw → 17:37:41.759000

```
INVITE sip:0003@192.168.1.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_281910197-85
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UP
From: "MapsSip" <sip:0003@192.168.1.203>;tag=FromTag_1_281910197-
To: 0001 <sip:0003@192.168.1.213>
Call-ID: GL-MAPS_1_281910197-8564-3132@192.168.1.203
CSeq: 1 INVITE
Contact: 0010 <sip:0003@192.168.1.203>
Content-Type: application/sdp
Content-Length: 317

v=0
o=0003 33852938 33852938 IN IP4 192.168.1.203
s=-SIP Call
c=IN IP4 192.168.1.203
t=0 0
m=audio 1044 RTP/AVP 0 8 18 3 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

# Short Message Service (SMS) Test Solutions



**GL** GL's SMS Test Solution Suite  
2G, 3G, and 4G

# 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

| Sr No | Script Name | Call Info    | Script Execution | Status                  | Events | Events Profile | Results |
|-------|-------------|--------------|------------------|-------------------------|--------|----------------|---------|
| 1     | MOC.gls     | 919655359811 | Completed        | SCCP Resources Released | None   |                | Pass    |
| 2     | MOC.gls     | 919655359811 | Completed        | SCCP Resources Released | None   |                | Pass    |
| 3     | MOC.gls     | 919655359812 | Completed        | SCCP Resources Released | None   |                | Pass    |

| Data/Time                 | Call Trace Id | Script Id                   | Captured Events                    |
|---------------------------|---------------|-----------------------------|------------------------------------|
| 2012-9-20 17:46:21.578000 | 919655359811  | ProtScriptId_392003856-3801 | Loaded Profile: MSCProfile01       |
| 2012-9-20 17:46:21.718000 | 919655359811  | ProtScriptId_392003856-3801 | SMS = I have a meeting in an hour. |
| 2012-9-20 17:46:21.718000 | 919655359811  | ProtScriptId_392003856-3801 | SMS Delivered                      |
| 2012-9-20 17:46:21.750000 | 919655359811  | ProtScriptId_392003856-3801 | SMS Deliver Report Received        |
| 2012-9-20 17:46:21.750000 | 919655359811  | ProtScriptId_392003856-3801 | Clear Command Initiated            |
| 2012-9-20 17:46:21.781000 | 919655359811  | ProtScriptId_392003856-3801 | Traffic Channel Released           |
| 2012-9-20 17:46:21.781000 | 919655359811  | ProtScriptId_392003856-3801 | SCCP Release Initiated             |
| 2012-9-20 17:46:21.812000 | 919655359811  | ProtScriptId_392003856-3801 | SCCP Resources Released            |

# 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;
```

# GTP Traffic Simulation

# GTP Traffic Mobile Options

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

# 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

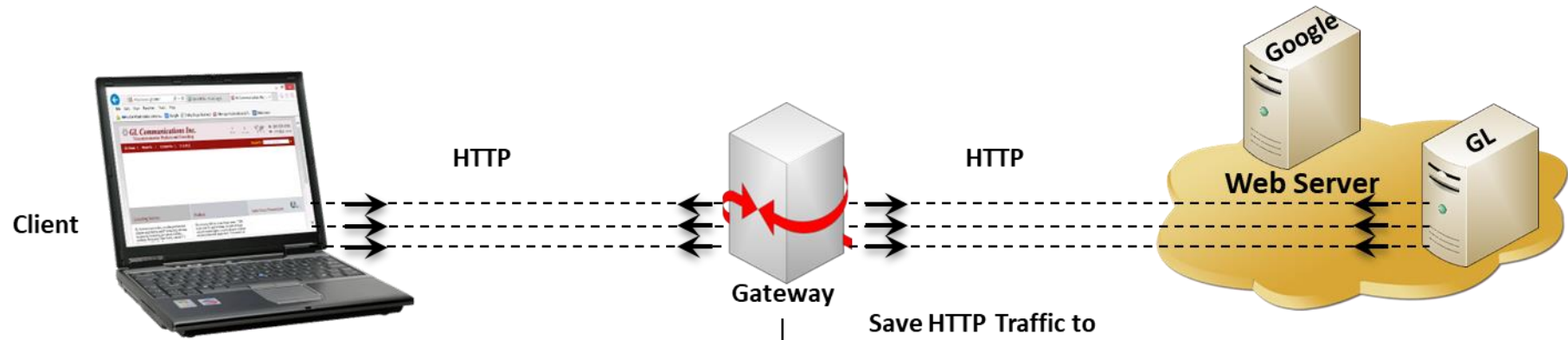
- **Mobile Traffic Core - GTP (ETH101)**

- 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

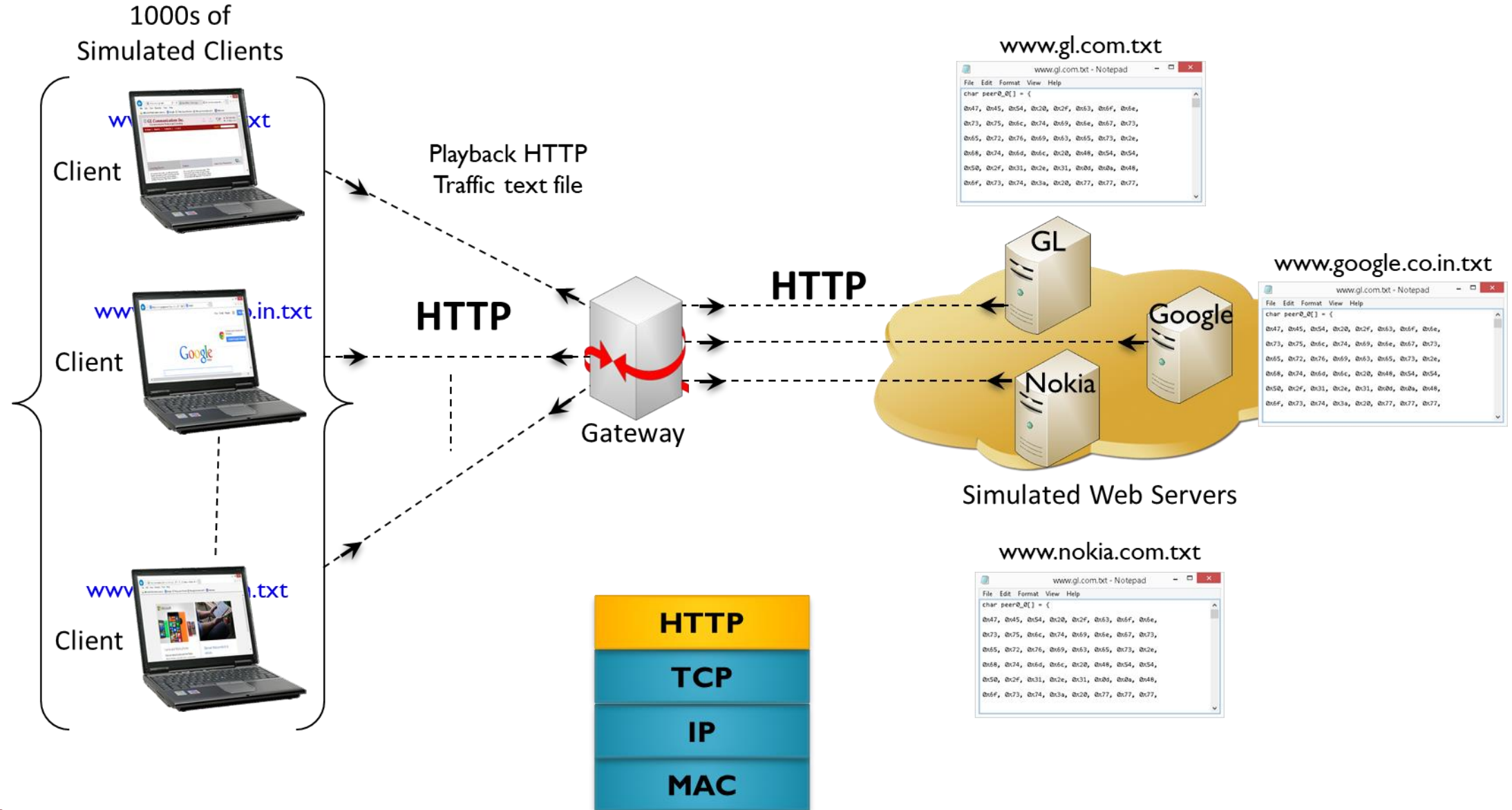
# Mobile IP Core Server



The screenshot shows a web browser window displaying the website for 'GL Communications Inc. Telecommunication Products and Consulting'. The browser's address bar shows 'http://www.gl.com/'. Overlaid on the bottom right of the browser window is a Notepad window titled 'www.gl.com.txt - Notepad'. The Notepad window contains the following text:

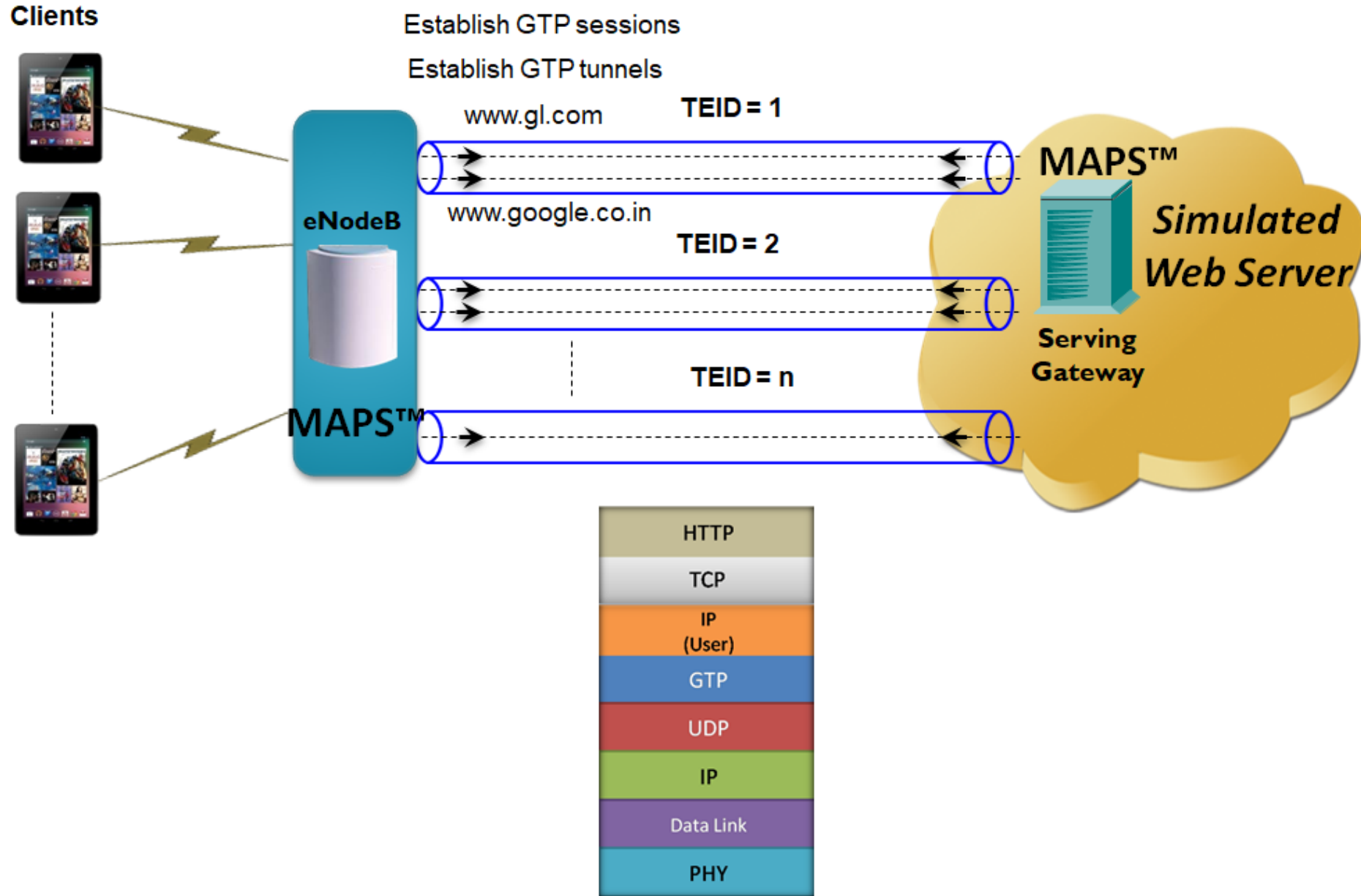
```
File Edit Format View Help
char peer0_0[] = {
0x47, 0x45, 0x54, 0x20, 0x2f, 0x63, 0x6f, 0x6e,
0x73, 0x75, 0x6c, 0x74, 0x69, 0x6e, 0x67, 0x73,
0x65, 0x72, 0x76, 0x69, 0x63, 0x65, 0x73, 0x2e,
0x68, 0x74, 0x6d, 0x6c, 0x20, 0x48, 0x54, 0x54,
0x50, 0x2f, 0x31, 0x2e, 0x31, 0x0d, 0x0a, 0x48,
0x6f, 0x73, 0x74, 0x3a, 0x20, 0x77, 0x77, 0x77,
```

# Mobile Traffic over IP



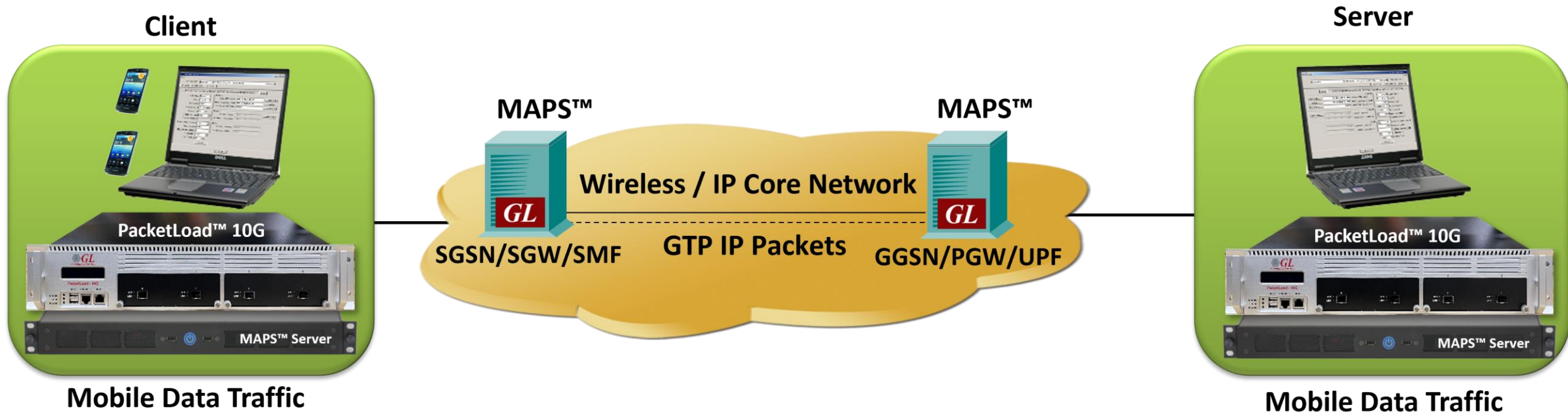


# Mobile Traffic over GTP



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

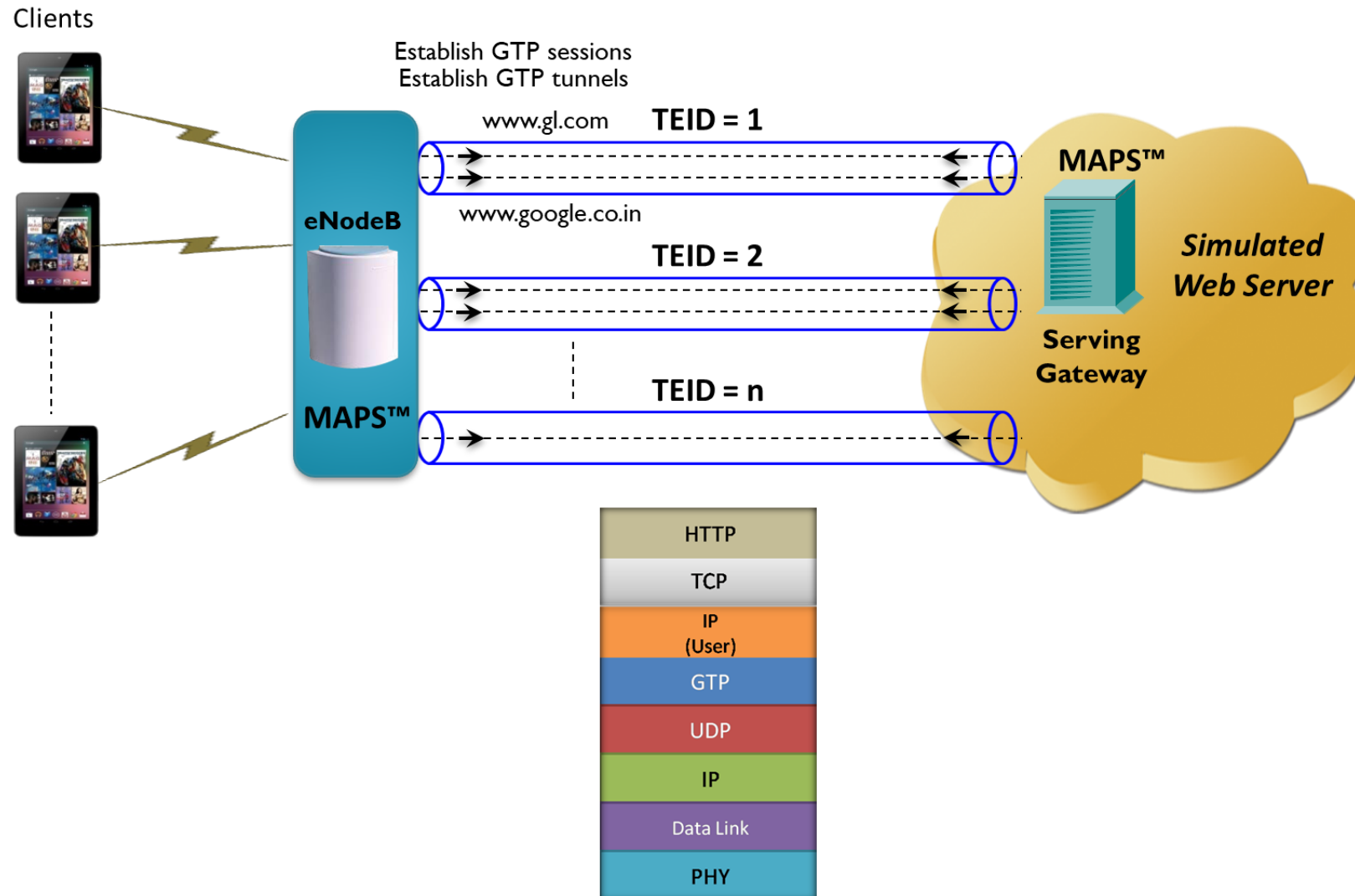
- GTP Traffic Simulation modules within MAPS™ supports user-plane traffic simulation in LTE, UMTS, and GPRS networks
- The solution includes Mobile IP traffic (ETH101 / PacketLoad™), Packet traffic (BERT, Hex String, and HDL File) and Mobile Gateway traffic (ETH102) types of traffic simulation. These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





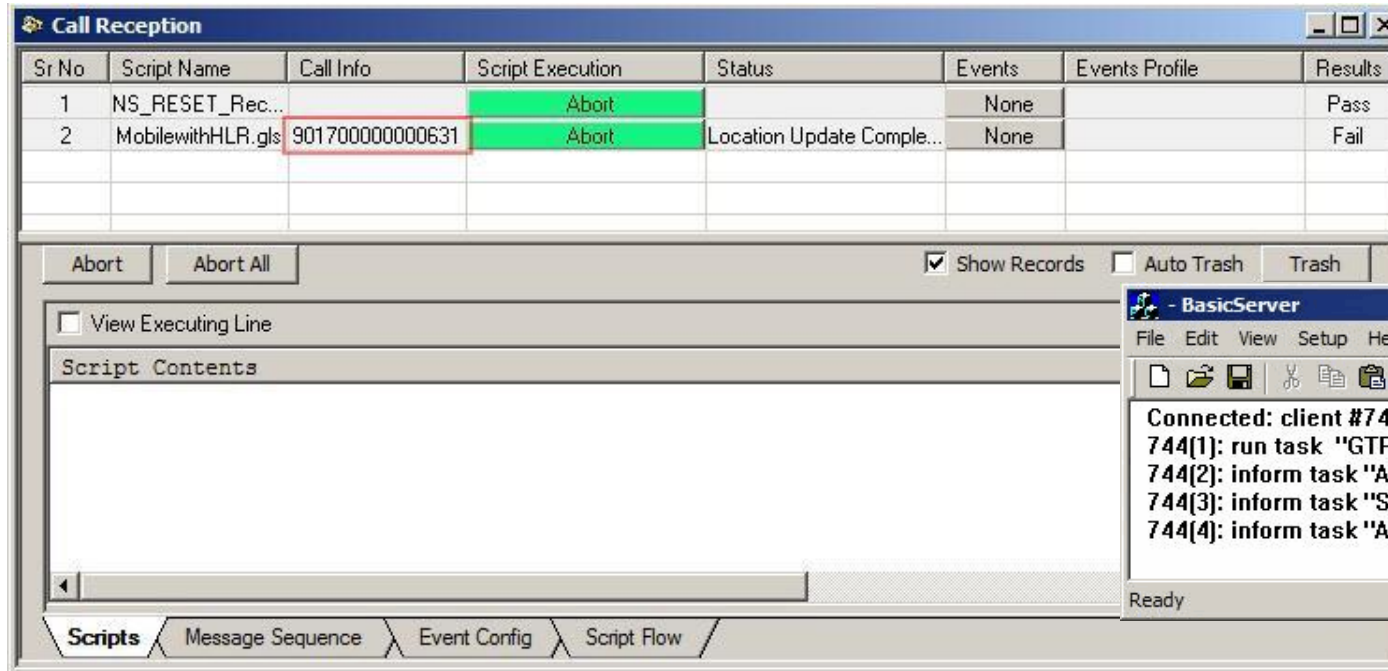
# Mobile Traffic Simulation – ETH101

Mobile Traffic over GTP



# Mobile Traffic Simulation – ETH101 (Contd.)

## SGSN Call Reception



| Sr No | Script Name       | Call Info       | Script Execution | Status                    | Events | Events Profile | Results |
|-------|-------------------|-----------------|------------------|---------------------------|--------|----------------|---------|
| 1     | NS_RESET_Rec...   |                 | Abort            |                           | None   |                | Pass    |
| 2     | MobilewithHLR.gls | 901700000000631 | Abort            | Location Update Comple... | None   |                | Fail    |

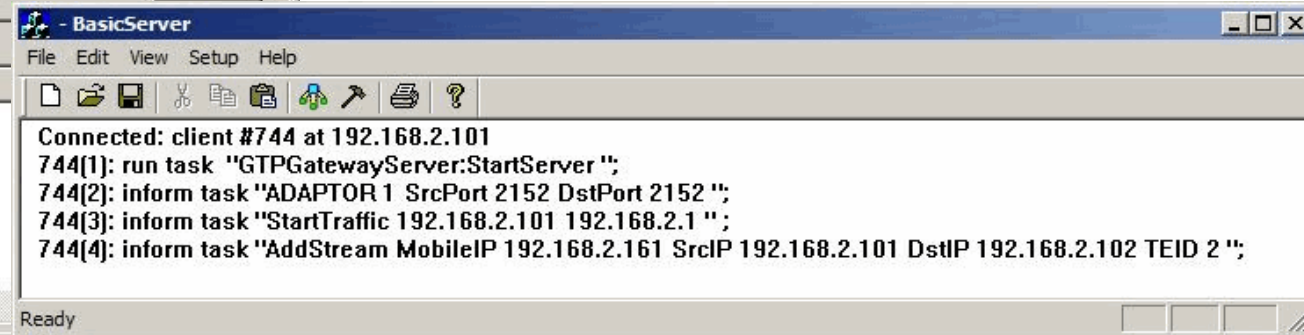
Buttons: Abort, Abort All, Show Records (checked), Auto Trash, Trash

View Executing Line:

Script Contents: [Empty text area]

Navigation: Scripts, Message Sequence, Event Config, Script Flow

## GGSN Server Log



```
Connected: client #744 at 192.168.2.101
744(1): run task "GTPGatewayServer:StartServer ";
744(2): inform task "ADAPTOR 1 SrcPort 2152 DstPort 2152 ";
744(3): inform task "StartTraffic 192.168.2.101 192.168.2.1 ";
744(4): inform task "AddStream MobileIP 192.168.2.161 SrcIP 192.168.2.101 DstIP 192.168.2.102 TEID 2 ";
```

Ready

- 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](http://www.google.com) is good to start)

# Mobile Traffic Simulation – ETH102

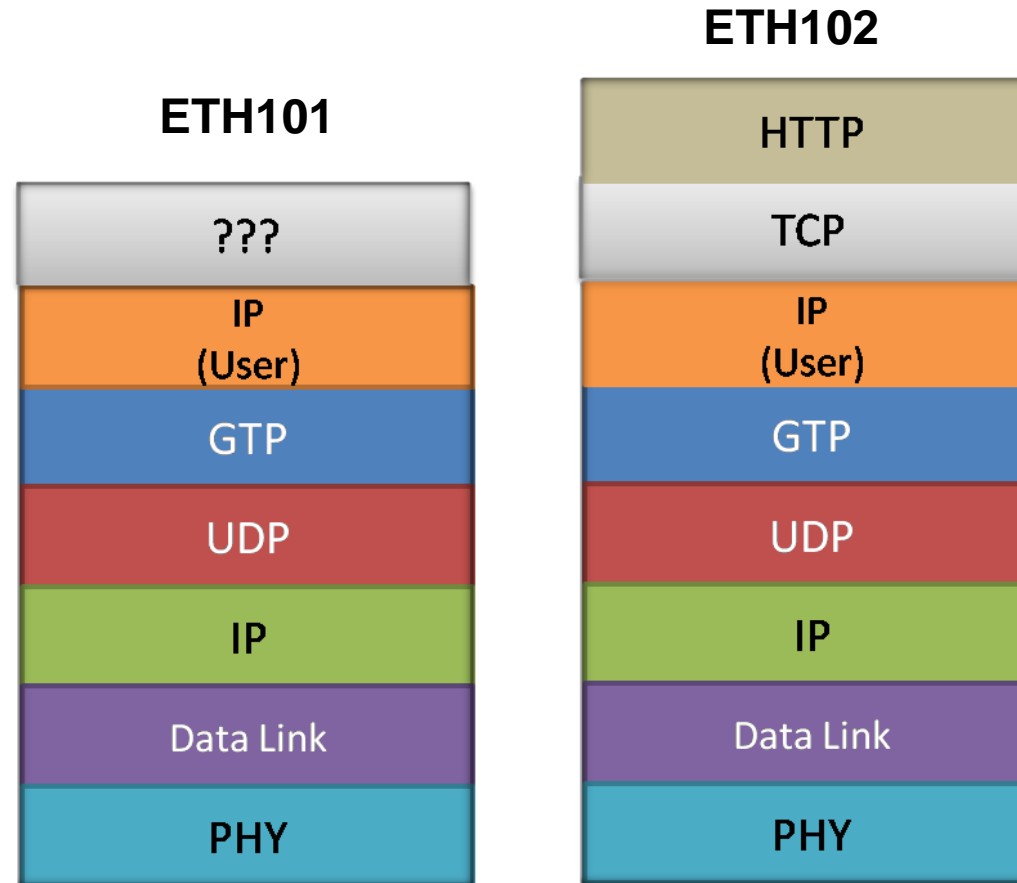
???: any IP-based protocol used to carry packet radio service

HTTP: Hyper Text Transfer Protocol

GTP: GPRS Tunnelling Protocol

UDP: User Datagram Protocol

IP: Internet Protocol



# Mobile Traffic Simulation (ETH102) – Call Generation and Reception

MAPS (Message Automation Protocol Simulation) (MAPIP) (UMTS GnGp) (UMTS IUPS) - [Call Reception]

Configurations Emulator Reports Editor Windows Help

| Sr No | Script Name           | Call Info             | Script Execution | Status                          | Events | Event... | Results |
|-------|-----------------------|-----------------------|------------------|---------------------------------|--------|----------|---------|
| 6     | InitiateSCMG.gs       | 1000                  | Stop             | SUBSYSTEM-ALLOWED               | None   |          | Pass    |
| 7     | InitiateM3UA.gs       | 1001                  | Stop             | ASP ACTIVE                      | None   |          | Pass    |
| 8     | CallControl_Attach.gs | PTMSI_001010123456220 | Completed        | Iu Release Complete             | None   |          | Pass    |
| 9     | CallControl_Attach.gs | PTMSI_0x11100001      | Completed        | Iu Release Complete             | None   |          | Pass    |
| 10    | CallControl_Attach.gs | PTMSI_001010123456228 | Stop             | Deactivate PDP Context Accepted | Detach |          | Pass    |

Abort Abort All Show Records Auto Trash Trash

Save Column Width

RNC SGSN HLR

ATTACH REQUEST 12:15:30.219000

CC connection confirm 12:15:30.222000

IDENTITY REQUEST 12:15:30.223000

IDENTITY RESPONSE 12:15:30.276000

sendAuthenticationInfoArg 12:15:30.283000

sendAuthenticationInfoRes 12:15:30.368000

AUTHENTICATION AND CIPHERING REQ 12:15:30.372000

AUTHENTICATION AND CIPHERING RESP 12:15:30.429000

updateGprsLocationArg 12:15:30.434000

insertSubscriberDataArg 12:15:30.519000

insertSubscriberDataRes 12:15:30.523000

updateGprsLocationRes

==== MTP3 User Adaptation Layer =====

```

0000 Version = 00000001 Release 1.0
0002 Message Class = 00000001 Transfer
0003 Transfer Message Type = 00000001 Payload Data
0004 Message Length = 136 (x00000088)
Protocol Data =
0008 Tag = x0210 Transfer Protocol Data
000A Length = 126 (x007E)
000E Point Code
0012 Point Code
0014 Service Indicator
0015 Network Indicator
0016 Message Priority
0017 Signalling Link Se.

Parameter Padding
===== SSCP La
0018 Message Type
Mandatory Fixed Par.
Source Local Refer
0019 Source Local Refer
Protocol Class Par.
001C Class
001C Message Handling
001D Pointer to Mandato
001E Pointer to optiona

```

Scripts Message Sequence Event Config Script Flow Capture Events

Untitled - BasicServer

File Edit View Setup Help

```

Connected: client #524 at 127.0.0.1
524(1): end tasks on disconnect;
524(2): run task "ServicingGateWay:StartServer ";
524(3): inform task "ADAPTOR 0 SrcPort 2152 DstPort 2152 SGW 192.168.2.2 eNB 0.0.0.0 PGW 192.168.2.6 ";
524(4): inform task "StartTraffic ";
524(5): inform task "AddStream ENB_UL_TEID 2 ENB_DL_TEID 2 PGW_UL_TEID 2 PGW_DL_TEID 2 ";
524(6): inform task "STOPTRAFFIC ENB 0.0.0.0 ENB_UL_TEID 2 PGW 192.168.2.6 PGW_DL_TEID 2 ";
524(7): inform task "ADAPTOR 0 SrcPort 2152 DstPort 2152 SGW 192.168.2.2 eNB 192.168.2.50 PGW 192.168.2.6 ";
524(8): inform task "StartTraffic ";
524(9): inform task "AddStream ENB_UL_TEID 3 ENB_DL_TEID 2 PGW_UL_TEID 3 PGW_DL_TEID 3 ";
524(10): inform task "STOPTRAFFIC ENB 192.168.2.50 ENB_UL_TEID 3 PGW 192.168.2.6 PGW_DL_TEID 3 ";
524(11): inform task "ADAPTOR 0 SrcPort 2152 DstPort 2152 SGW 192.168.2.2 eNB 192.168.2.50 PGW 192.168.2.6 ";
524(12): inform task "StartTraffic ";
524(13): inform task "AddStream ENB_UL_TEID 4 ENB_DL_TEID 3 PGW_UL_TEID 4 PGW_DL_TEID 4 ";
524(14): inform task "STOPTRAFFIC ENB 192.168.2.50 ENB_UL_TEID 4 PGW 192.168.2.6 PGW_DL_TEID 4 ";
524(15): inform task "ADAPTOR 0 SrcPort 2152 DstPort 2152 SGW 192.168.2.2 eNB 192.168.2.254 PGW 192.168.2.6 ";
524(16): inform task "StartTraffic ";
524(17): inform task "AddStream ENB_UL_TEID 5 ENB_DL_TEID 5 PGW_UL_TEID 5 PGW_DL_TEID 5 ";
524(18): inform task "ADAPTOR 0 SrcPort 2152 DstPort 2152 SGW 192.168.2.2 eNB 192.168.2.254 PGW 192.168.2.6 ";
524(19): inform task "StartTraffic ";
524(20): inform task "AddStream ENB_UL_TEID 6 ENB_DL_TEID 6 PGW_UL_TEID 6 PGW_DL_TEID 6 ";
Connected: client #432 at 127.0.0.1
432(1): end tasks on disconnect;

```

Ready NUM

# Mobile Traffic Simulation - GPRS Gb (ETH103)

MAPS (Message Automation Protocol Simulation) (GPRSGb) (MAPIP) (UMTS GnGp) - [Call Reception]

Configurations Emulator Reports Editor Windows Help

| Sr No | Script Name               | Call Info                             | Script Execution | Status                      | Events      | Events Profile | Results |
|-------|---------------------------|---------------------------------------|------------------|-----------------------------|-------------|----------------|---------|
| 4     | InitiateSCMG.gls          | 3                                     | Stop             | SUBSYSTEM-ALLOWED           | None        |                | Pass    |
| 5     | NSProcedures.gls          |                                       | Stop             |                             | None        |                | Unknown |
| 6     | GPRSGbCallControlSGSN.gls | IMSI: 90170000000623,TMSI: 0x10000004 | Completed        | Detach Accept               | None        |                | Pass    |
| 7     | GPRSGbCallControlSGSN.gls | IMSI: 90170000000626,TMSI: 0x10000006 | Stop             | Attach Complete is Received | DETACH-CALL |                | Unknown |

Abort Abort All Show Records Auto Trash Trash

Save Column Width

BSC SGSN HLR GGSN GGSNTraffic

```

sequenceDiagram
 participant BSC
 participant SGSN
 participant HLR
 participant GGSN
 participant GGSNTraffic

 BSC->>SGSN: ATTACH REQUEST 12:45:02.967000
 SGSN->>HLR: IDENTITY REQUEST 12:45:02.968000
 HLR-->>SGSN: IDENTITY RESPONSE 12:45:03.547000
 SGSN->>HLR: sendAuthenticationInfoArg 12:45:03.547000
 HLR-->>SGSN: sendAuthenticationInfoRes 12:45:03.579000
 SGSN->>HLR: AUTHENTICATION AND CIPHERING 12:45:03.579000
 HLR-->>SGSN: AUTHENTICATION AND CIPHERING 12:45:05.767000
 SGSN->>HLR: updateGprsLocationArg 12:45:05.767000
 HLR-->>SGSN: insertSubscriberDataArg 12:45:05.799000
 HLR-->>SGSN: insertSubscriberDataRes 12:45:05.799000
 HLR-->>SGSN: updateGprsLocationRes 12:45:05.830000
 SGSN->>BSC: ATTACH ACCEPT 12:45:05.830000
 BSC->>SGSN: ATTACH COMPLETE 12:45:06.447000
 BSC->>SGSN: Activate PDP Context Request 12:45:31.677000
 SGSN->>GGSN: Create PDP Context Request 12:45:31.671000
 GGSN-->>SGSN: Create PDP Context Response 12:45:31.677000
 SGSN->>BSC: Activate PDP Context Accept 12:45:31.677000
 BSC->>SGSN: XID-Com 12:45:32.247000
 SGSN-->>BSC: XID-Res 12:45:32.247000
 BSC->>SGSN: SN-UniData 12:45:32.269000
 SGSN->>GGSN: G-PDU 12:45:32.269000
 GGSN-->>SGSN: G-PDU 12:45:32.269000
 BSC->>SGSN: SN-UniData 12:45:32.334000
 SGSN->>GGSN: G-PDU 12:45:32.334000
 GGSN-->>SGSN: G-PDU 12:45:32.334000
 BSC->>SGSN: SN-UniData 12:45:32.846000
 SGSN->>GGSN: G-PDU 12:45:32.846000
 GGSN-->>SGSN: G-PDU 12:45:32.846000

```

```

===== Network Service Layer =====
0000 PDU Type = 00000000 NS-UNI
0001 BVCI =
0002 BVCI = 2 (x0002)
===== BasGp Layer =====
0004 PDU Type = 00000001 UL-UNI
0005 TLLI value = x90000002
0006 QoS Profile =
0007 Peak bit rate = 0 (x0000)
0008 Precedence (UL-Unidata) =100 Radio
0009 A bit =0.... Radio
000A T bit = ...0.... SDU cc
000B C/R bit = ..0.... SDU cc
000C Cell Identifier =
000D IE Identifier (CI) = 00001000 Cell I
000E Length Ext = 1..... Extens
000F Length of Cell Identifier = ..0001000 (8)
0010 Ext (Cell Identifier) = 1..... Extens
0011 MCC = 901
0012 MNC = 70
0013 Location area code = 10000 (0010011)
0014 Cell Id RAC = 00000000 (0)
0015 Cell Identity = 1 (x0001)
Alignment octets =
0016 IE Identifier (AO) = 00000000 Alignm
0017 Length Ext = 1..... Extens
0018 Length of Alignment Octets = ..0000000 (0)
0019 Ext = 1..... Extens
LLC-PDU
001A IE Identifier (LP) = 00001110 LLC-PD
001B Length Ext = 0..... Extens
001C Length of LLC-PDU = 57 (.0000000 0)
001D LLC-Pdu = x01C001080103E8
===== LLC Layer =====
001E SAPI =0001 LLGMM
001F C/R (User->Net) = ..0..... Commar
0020 P/D = 0..... LLC Fr
0021 Ctl = 110..... UI For
0022 N(U) = 0 (.....000 00)
0023 E Bit =0. Non-ci
0024 PM Bit =1 Protec
0025 GMM Data = x080103E8E00011
0026 FCS = 2048509 (x1F41)
===== Layer3 Information Layer =====
0027 Protocol Discriminator =1000 GPRS s
0028 Skip Indicator = 0000... (0)
0029 L3 Info = x0103E8E000110F
===== Gprs Mobility Mgmt Layer =====
002A Message Type = 00000001 ATTACH
002B MS network capability =
002C Length = 3 (x03)
002D MS Network Capability Data = xE8E000
002E Attach type/CipheringKey =
002F Attach Type Value =001 GPRS s
0030 Follow-on Request =0... Reserv
0031 Ciphering Key Seq # = ..001.... 1
0032 DRX parameter =

```

Scripts Message Sequence Event Config Script Flow



# 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

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface. The main window shows a table of script executions:

| Sr No | Script Name                              | Profile       | Call Info         | Script Execution | Status              | Events        | Events... | Result  | Total Iterations | Completed Iterations |
|-------|------------------------------------------|---------------|-------------------|------------------|---------------------|---------------|-----------|---------|------------------|----------------------|
| 1     | GTPPDPContextSGSN_PacketCheckTraffic.gls | SGSNProfile01 | 2.404006000000... | Abort            | PDP Context Updated | Remove Stream |           | Pass    | 1                | 0                    |
| 2     | GTPPDPContextSGSN_PacketCheckTraffic.gls | SGSNProfile02 |                   | Start            |                     | None          |           | Unknown | 1                | 0                    |
| 3     | GTPPDPContextSGSN_PacketCheckTraffic.gls | SGSNProfile03 |                   | Start            |                     | None          |           | Unknown | 1                | 0                    |

Below the table, a message sequence diagram shows the interaction between MAPS and DUT:

- MAPS sends Create PDP Context Request to DUT at 11:05:42.000000.
- DUT sends Create PDP Context Response to MAPS at 11:05:42.000000.
- MAPS sends Update PDP Context Request to DUT at 11:07:22.828000.
- DUT sends Update PDP Context Response to MAPS at 11:07:22.828000.

To the right, a GTP Layer data dump is shown:

```

===== GTP'/GTP Layer =====
Version = 001..... GTP V1
Protocol Type = GTP V2
GTP Layer Message =
E =0.. Not Present
S =1. Present
PN =0 Not Present
Message Type = 00010000 Create PDP Context Request
Length of GTP Message = 132 (x0084)
Tunnel Endpoint Identifier = 0 (x00000000)
Sequence Number = 1 (x0001)
End of Extension Hdrs =
IMSI =
IE Identifier(IMSI) = 00000010 IMSI
IMSI = 4040060000000001
Recovery =
IE Identifier(R) = 00001110 Recovery
Restart counter = 00001011 (11)
Selection Mode =
IE Identifier(SM) = 00001111 Selection Mode
Selection mode Value =01 MS provided APN, subscription not
Tunnel Endpoint Identifier Data I = 00010000 Tunnel Endpoint Identifier Data I
IE Identifier(TEID-I) = 1 (x00000001)
Tunnel Endpoint Identifier Data =
Tunnel Endpoint Identifier Control Plane =
IE Identifier(TEICP) = 00010001 Tunnel Endpoint Identifier Contro
Tunnel Endpoint Id Control Plane Data = 2 (x00000002)

```

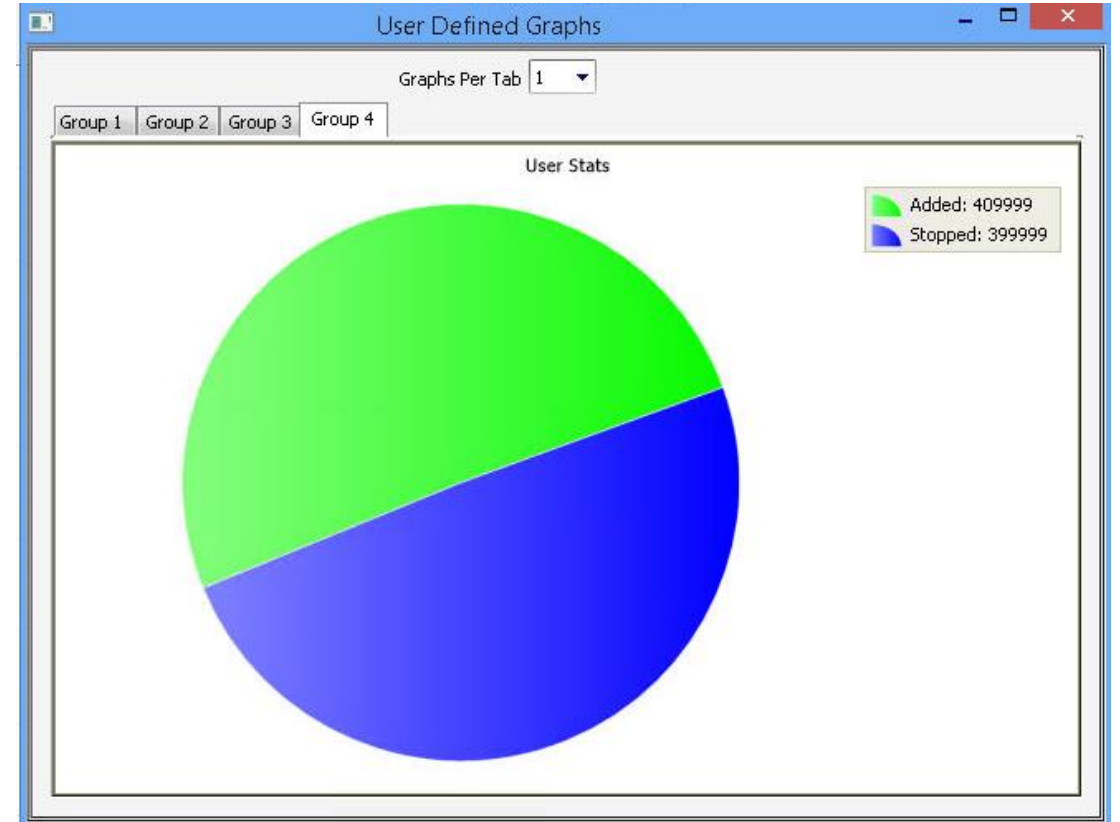
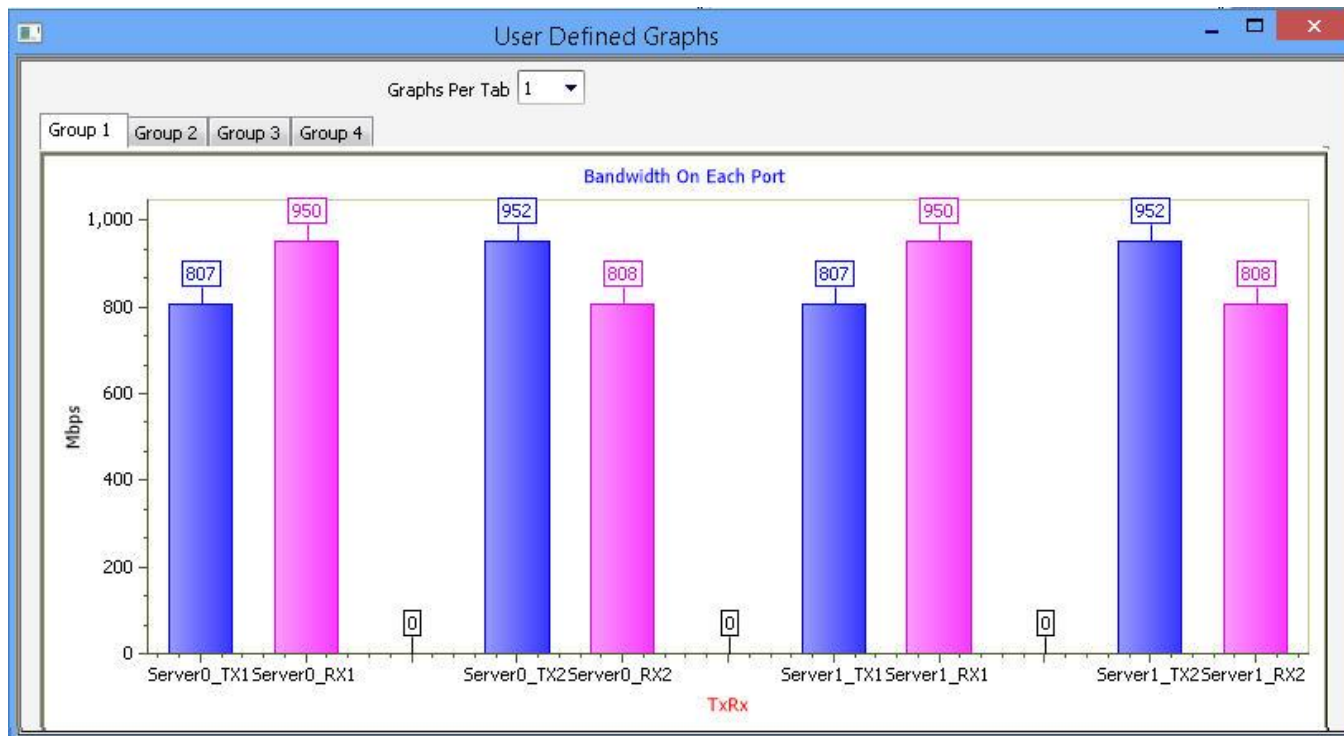
On the far right, the BasicServer console window shows the following output:

```

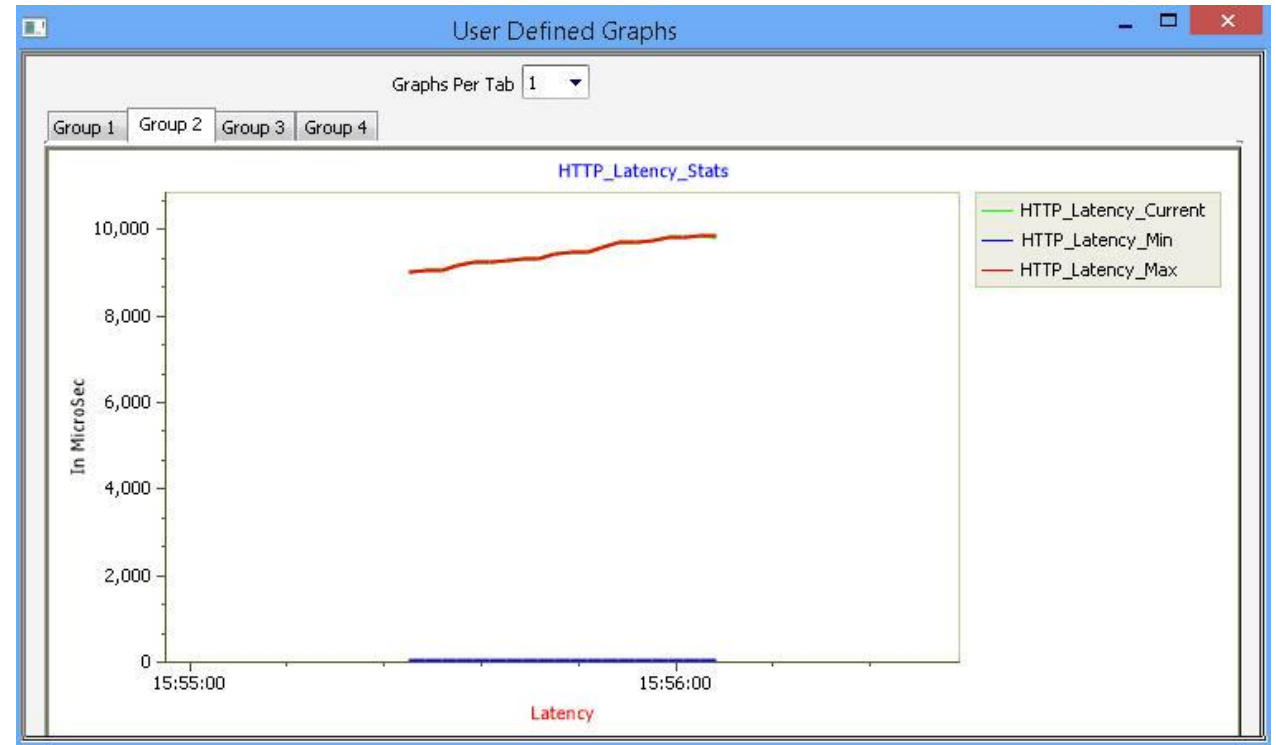
Connected: client #624 at 192.168.1.225
624: run task "PacketCheckServer:StartServer";
624: inform task "Init 1 GTP;";
624: inform task "AddStream 0 'GL01'";
624: inform task "AddStream 0 192.168.1.225 2152 192.168.1.50 2152 TEID 1 ";
624: inform task "SetStack 0 none ipv4 udp;";
624: inform task "SetDirection 0 tx;";
624: inform task "IPv4Address 0 192.168.1.133 255.255.255.0 192.168.1.50 ";
624: inform task "IPv4Params 0 0 128 17;";
624: inform task "EnableIPv4AutoBuildMACHdr 0 true;";
624: inform task "UDPPParams 0 2252 2252 ";
624: inform task "EnableSeqNo 0 true;";
624: inform task "FixedPattern 0 ABCDEF112233 ";
624: inform task "FrameSize 0 increasing 1000 1500 ";
624: inform task "StopTx 0 continuous 0 0;";
624: inform task "InterFrameGap 0 100 false;";
624: inform task "StartTraffic IFG;";
624: inform task "addstreamtotraffic 0 ";

```

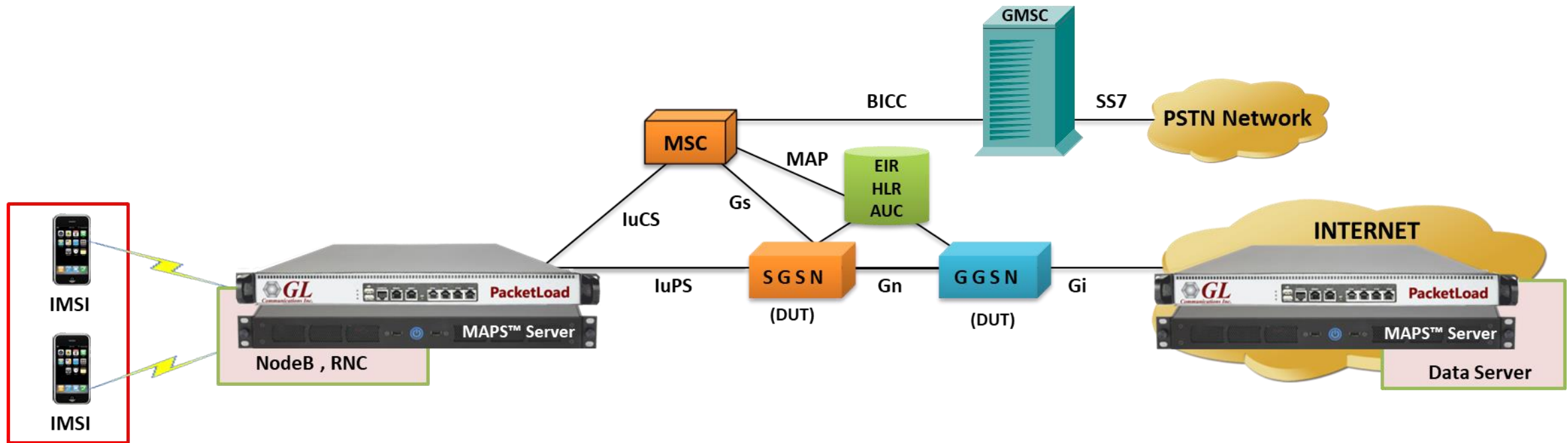
# High Density Packet Traffic Statistics



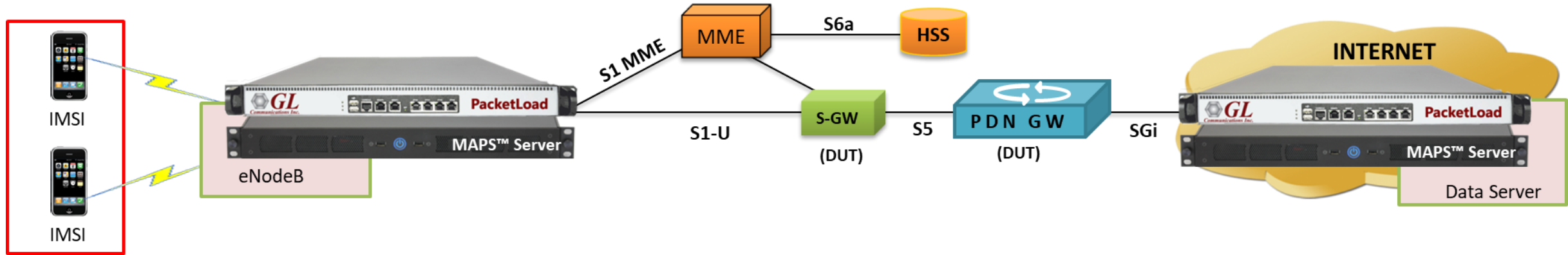
# High Density Packet Traffic Statistics (Contd.)



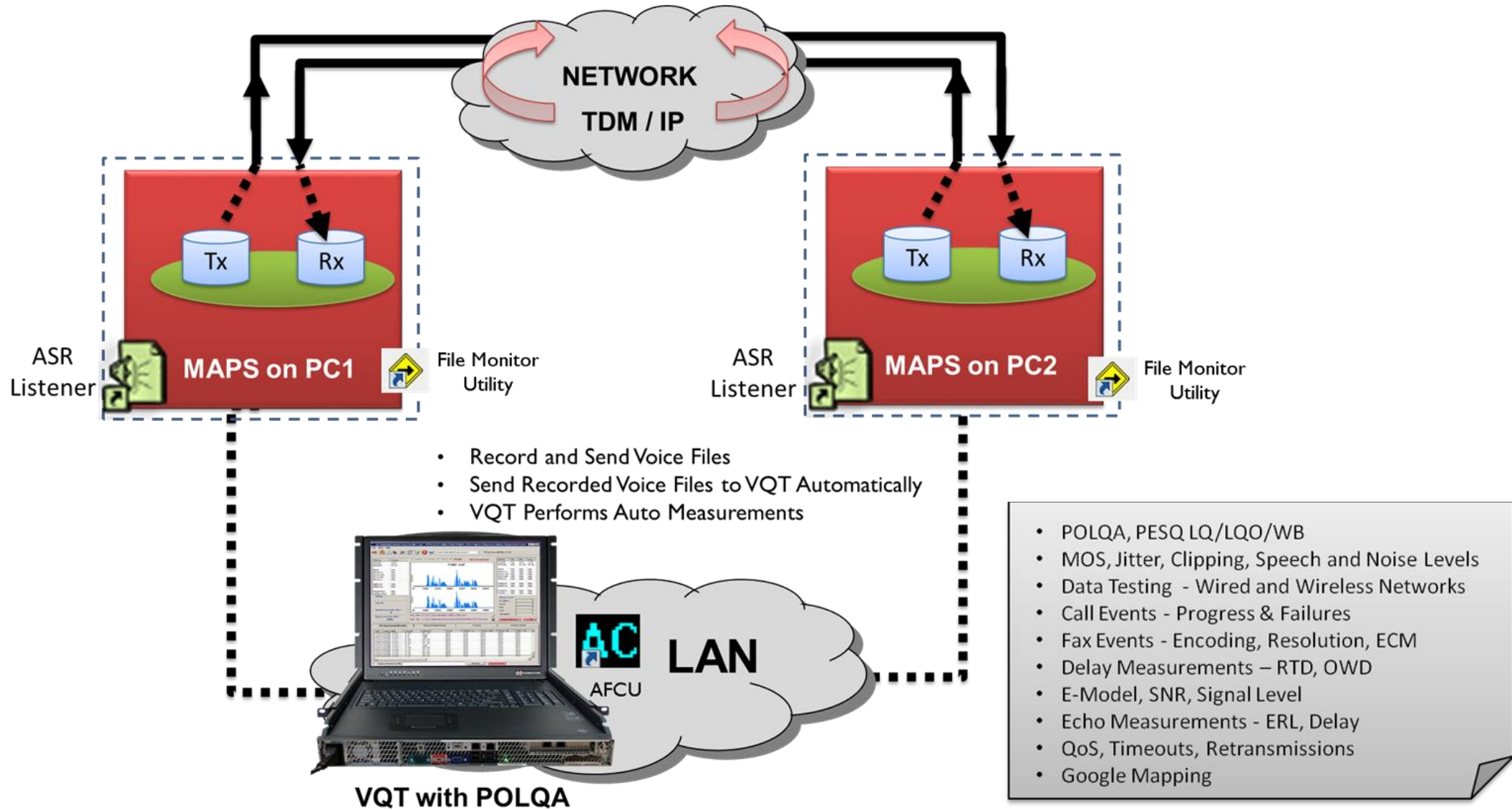
# Test Scenario – 3G using PacketLoad™ along with MAPS™ Server



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



# VQT Analysis



# Create Traffic Events using Script Editor

The image shows a screenshot of a 'Script Editor' window titled 'Script - RtpTrafficAPI'. The editor contains a script with the following content:

```
Line# Script
231
232 //***** Transmit and Monitor Digits *****/
233
234 //----- Transmit Digits -----
235 //Input Arguments:DigitType,TxDigits,DigitBand,DigitPower1,DigitPower2,DigitOnTime,DigitOffTime
236
237 "Send_Digits"(DigitType,TxDigits,DigitBand,DigitPower1,DigitPower2,DigitOnTime,DigitOffTime);
238 IsDigitSent = 0;
239 TxRx:tx _Rtp digits : digitype = DigitType,digits = TxDigits, band = DigitBand , power1 = DigitPower1, power2 = DigitPower2,ontime = DigitOnTime, offtime = Dig
240 resume;
241
242 //----- Monitor Digits -----
243 //Input Arguments:DigitBand,DigitType
244
245 "Detect_Digits"(DigitBand,DigitType);
246 Status = "Detect Inband Digits-Started";
247 TxRx:monitor _Rtp digits : band = DigitBand,digitype = DigitType;
248 if(TrafficDirection == "TxRx")
249 return;
250 endif
251 if(TrafficType == "IVR")
252 goto "OnMonitorDigitsStarted";
253 endif
254 resume;
255
256 //***** Transmit and Monitor Tones *****/
257
258 //----- Transmit Tones -----
259 //Input Arguments:Freq1_Power1,Freq2_Power2,OnTime,OffTime,Iterations
260 //Note: Set ToneFreq2 to 0 for Sending Single tone
261
262 "Send_Tones"(Freq1_Power1,Freq2_Power2,OnTime,OffTime,Iterations);
263 IsToneSent = 0;
264 TxRx:tx _Rtp tone : freq1 = Freq1, power1 = Power1, freq2 = Freq2, power2 =
265 resume;
266
267 //----- Monitor Tones -----
```

A red box highlights the 'Send Digits' command in the script (line 239). A red arrow points from this box to a 'Send Digits' dialog box. The dialog box has the following fields:

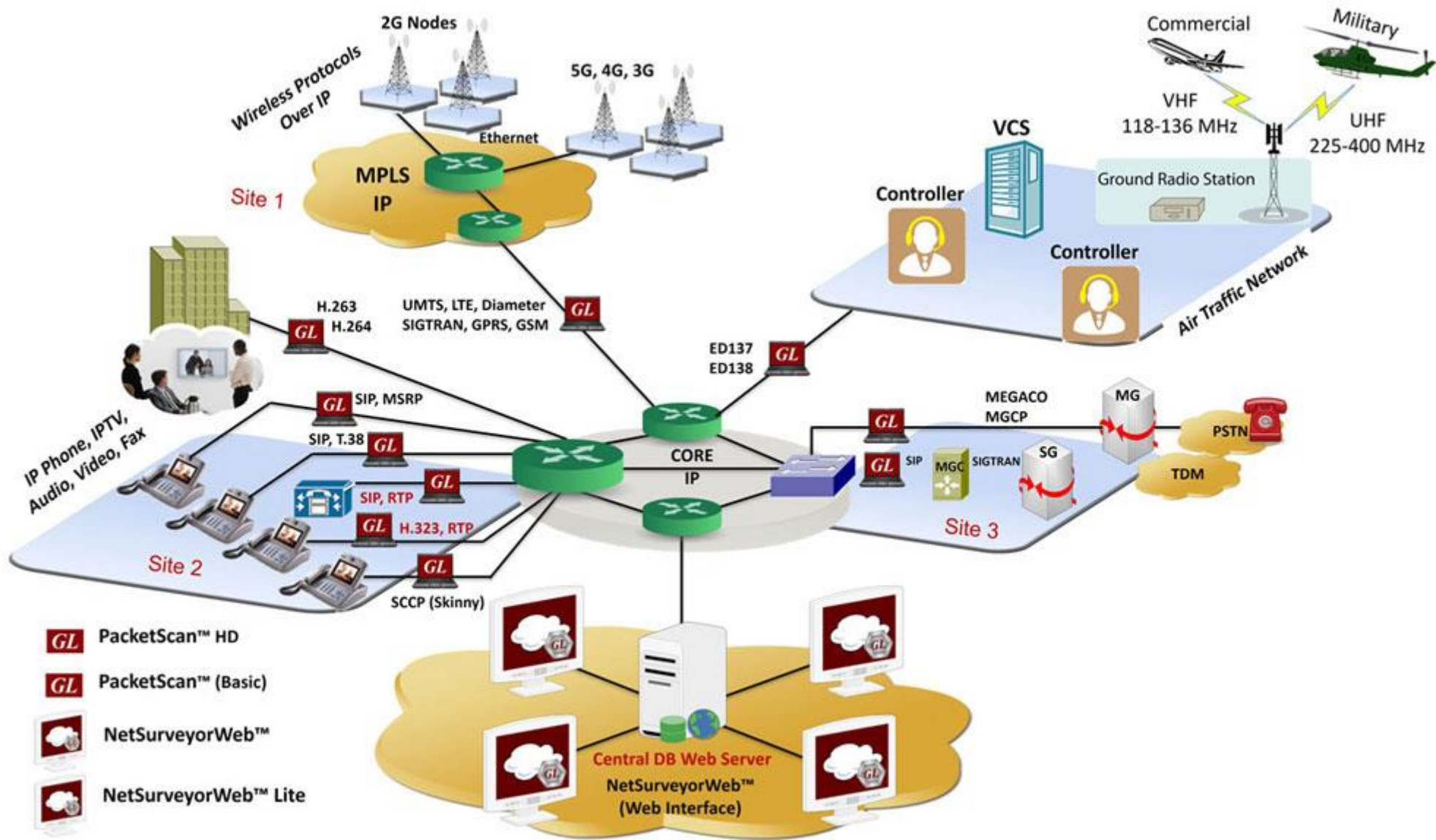
- Digit Type: dtmf digits
- Variable Digit: inband
- Digits: TxDigits
- Power1: DigitPower1
- Power2: DigitPower2
- OnTime: DigitOnTime
- OffTime: DigitOffTime

The 'Send Digits' dialog box also has 'OK' and 'Cancel' buttons at the bottom right.

# Traffic Analysis over IP Networks



# PacketScan™ 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>



# Voice Traffic Analysis over IP

The screenshot displays the 'Packet Data Analyzer - Summary View' interface. At the top, there is a menu bar with 'File', 'View', 'Call Summary', 'Protocol Configurations', 'GUI Configurations', and 'Help'. Below the menu is a toolbar with various icons and a dropdown menu set to 'SIP'. A status bar on the right indicates 'Total : 824'. The main area is divided into two sections: a call log table and a detailed packet analysis pane.

**Call Log Table:**

| Call # | Caller             | Callee             | StartTime               | Duration     | VoiceQuality_L | VoiceQuality_R | Cor |
|--------|--------------------|--------------------|-------------------------|--------------|----------------|----------------|-----|
| 1      | 0001@192.168.12.92 | 0001@192.168.12.94 | 2023-06-01 15:01:34.419 | 00:01:00.023 |                |                |     |
| 2      | 0002@192.168.12.92 | 0002@192.168.12.94 | 2023-06-01 15:01:34.482 | 00:01:00.033 |                |                |     |
| 3      | 0003@192.168.12.92 | 0003@192.168.12.94 | 2023-06-01 15:01:34.533 | 00:01:00.045 |                |                |     |
| 4      | 0004@192.168.12.92 | 0004@192.168.12.94 | 2023-06-01 15:01:34.583 | 00:01:00.037 |                |                |     |
| 5      | 0005@192.168.12.92 | 0005@192.168.12.94 | 2023-06-01 15:01:34.623 | 00:01:00.049 |                |                |     |

**Packet Analysis Pane:**

The pane shows a sequence of frames between two IP addresses: 192.168.12.92 and 192.168.12.94. The frames are as follows:

- 00.00.000 - Frame 0: INVITE (5060) → 5060
- 00.00.020 - Frame 1: SIP/2.0 100 Trying (5060) ← 5060
- 00.00.029 - Frame 2: SIP/2.0 180 Ringing (5060) ← 5060
- 00.00.153 - Frame 9: SIP/2.0 200 OK (5060) ← 5060
- 00.00.163 - Frame 11: ACK (5060) → 5060
- 01.00.177 - Frame 3984: BYE (5060) → 5060
- 01.00.187 - Frame 3985: SIP/2.0 200 OK (5060) ← 5060

The detailed analysis on the right shows the decoded SIP message structure:

```
===== MAC Layer =====
Destination Address = x6C626D3E2B30
Source Address = x54BEF737BC79
Length/Protocol Type = x0800 Internet IP(IPv4)
===== IPv4 Layer =====
Version = 0100.... (4)
Internet Header Length (In 32 bit words) =0101 (5)
Differentiated Services Codepoint = 000000.. Default
Explicit Congestion Notification =00 Not-ECT (Not ECT-C)
IP Hdr No TCP SegmentationOffload =
Total Length = 761 (x02F9)
Identification = 15592 (x3CE8)
Reserved Bit = 0..... Not Set
Don't fragment = .0..... Not Set
More fragments = .0..... Not Set
Fragment Offset = 0 (...00000 00000000)
Time To Live = 128 (x80)
Protocol = 00010001 UDP
Header Check Sum = x0000
Source IP Address = 192.168.12.92 (xCOA80C5C)
Destination IP Address = 192.168.12.94 (xCOA80C5E)
===== UDP Layer =====
Source Port = 5060 (x13C4)
Destination Port = 5060 (x13C4)
Length (Header + Data) = 741 (x02E5)
```

A red arrow points from the 'INVITE' frame in the packet list to the 'SIP/2.0 100 Trying' frame in the detailed analysis pane. A blue text box with an arrow pointing to the analysis pane contains the text: "Displays decoded information of the selected SIP message".

# T.38 Analysis over IP

The screenshot displays the Packet Data Analyzer (PDA) interface in Summary View. The window title is "Packet Data Analyzer - Summary View". The menu bar includes File, View, Call Summary, Protocol Configurations, GUI Configurations, and Help. The toolbar contains various icons, including a dropdown menu set to "SIP" and a "Show Fax Calls" button highlighted with a red box. Below the toolbar is a "Call Summary" section with tabs for Call Summary, Registraton Summary, and Alert Summary. A table shows call details for call # 1, including source and destination IP addresses (192.168.1.244 and 192.168.1.60), conversational and listening times, and packet statistics.

| Call # | Ssrc_L    | ConversationalMos_L | ConversationalR_L | ListeningMos_L | ListeningR_L | PacketsDiscarded_L | PacketsDiscarded(%)_L |
|--------|-----------|---------------------|-------------------|----------------|--------------|--------------------|-----------------------|
| 1      | 390089559 | 4.20                | 93                | 4.20           | 93           | 0                  | 0.00                  |

The main area shows a list of packets between 192.168.1.244 and 192.168.1.60. The packet at timestamp 00.27.343... is highlighted in orange and selected. The protocol stack view on the right shows the T.38 Layer details for this message, including fields like seq-number, Contents, primary-ifp-packet, Length, IFPPacket, Preamble, type-of-msg, Choice Index, t30-indicator, Extensibility Marker, Contents, error-recovery, secondary-ifp-packets, Iteration Count, and FCS. A blue arrow points from the selected packet to the T.38 Layer details, with a text box stating "Displays decoded information of the selected FAX message".

```
===== T.38 Layer =====
UDPTLPacket =
seq-number = SEQUENCE
Contents = INTEGER
primary-ifp-packet = 3
Length = Open Type
IFPPacket = 1
Preamble = SEQUENCE
type-of-msg = CHOICE
Choice Index = 0
t30-indicator = ENUMERATOR
Extensibility Marker = 0
Contents = 0 no-signal(0)
error-recovery = CHOICE
Choice Index = 0
secondary-ifp-packets = SEQUENCE OF
Iteration Count = 1
secondary-ifp-packets = Instance 0
primary-ifp-packet = Open Type
Length = 1
IFPPacket = SEQUENCE
Preamble = 0
type-of-msg = CHOICE
Choice Index = 0
t30-indicator = ENUMERATOR
Extensibility Marker = 0
Contents = 0 no-signal(0)
===== MAC Layer =====
Padding octets = x401188E4C0A8
FCS = x013CCA38 (Invalid FCS. Correct FCS is xA72500)
```

Displays decoded information of the selected FAX message

# Video Quality Metrics in PacketScan™

Traffic Analyzer - Summary View

File View Call Summary Settings Help

Sip Calls Show Video Sessions Only

Call Summary | Registrar Summary | Alert Summary

| Call #                                                                                                                                                                                  | SSRC       | Payload         | Packet Received | Conversational MOS/R-Factor | Listening MOS/R-Factor | Packets Discard... | Missing Packets... | Duplicate Packets... | Out Of Sequen... | Average Gap(ms) | Average Delay | Average Jitter | Average Inter Arri... | Cumulativ Packet ... | Max/Min Gap   | Max/Min Delay | Max/Min ... |
|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------|-----------------|-----------------|-----------------------------|------------------------|--------------------|--------------------|----------------------|------------------|-----------------|---------------|----------------|-----------------------|----------------------|---------------|---------------|-------------|
| ▼ Call#000001 Caller:0003@192.168.1.203 Callee:0003@192.168.1.213 CallId:GL-MAPS_1_28456180-17497-5384@192.168.1.203 Call StartTime:2015-06-29 18:43:21.808 Call Duration: 00:01:00.050 |            |                 |                 |                             |                        |                    |                    |                      |                  |                 |               |                |                       |                      |               |               |             |
| 1                                                                                                                                                                                       | 1140062209 | PCMA/8000       | 3100            | 4.20 / 93                   | 4.20 / 93              | 0 / 0.00           | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 19.39           | 0.00          | 1.00           | 1                     | 0                    | 37.95 / 1.98  | 17 / -25      | 3.31 / 0.00 |
| 1                                                                                                                                                                                       | 1155789825 | PCMA/8000       | 3100            | 4.20 / 93                   | 4.20 / 93              | 0 / 0.00           | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 19.39           | 0.00          | 1.00           | 1                     | 0                    | 37.92 / 2.04  | 17 / -24      | 3.23 / 0.00 |
| 1                                                                                                                                                                                       | 1136487937 | h263-2000/90000 | 2816            | n/a                         | n/a                    | n/a                | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 64.62           | -70.00        | 70.00          | n/a                   | 0                    | 83.04 / 39.08 | 16 / -273     | 273.99 /    |
| 1                                                                                                                                                                                       | 1157928193 | h263-2000/90000 | 2816            | n/a                         | n/a                    | n/a                | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 64.62           | -70.00        | 70.00          | n/a                   | 0                    | 83.02 / 38.99 | 16 / -274     | 274.04 /    |
| ▼ Call#000002 Caller:0003@192.168.1.203 Callee:0003@192.168.1.213 CallId:GL-MAPS_1_28527647-17505-5428@192.168.1.203 Call StartTime:2015-06-29 18:44:33.264 Call Duration: 00:01:00.013 |            |                 |                 |                             |                        |                    |                    |                      |                  |                 |               |                |                       |                      |               |               |             |
| 2                                                                                                                                                                                       | 1134240769 | PCMA/8000       | 3099            | 4.20 / 93                   | 4.20 / 93              | 0 / 0.00           | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 19.39           | 0.00          | 1.00           | 1                     | 0                    | 38.03 / 2.03  | 18 / -24      | 3.38 / 0.00 |
| 2                                                                                                                                                                                       | 1150519041 | PCMA/8000       | 3100            | 4.20 / 93                   | 4.20 / 93              | 0 / 0.00           | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 19.39           | 0.00          | 1.00           | 1                     | 0                    | 37.98 / 2.01  | 17 / -24      | 3.21 / 0.00 |
| 2                                                                                                                                                                                       | 1137218817 | h263-2000/90000 | 2816            | n/a                         | n/a                    | n/a                | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 64.62           | -70.00        | 70.00          | n/a                   | 0                    | 83.10 / 37.99 | 15 / -274     | 274.01 /    |
| 2                                                                                                                                                                                       | 1151167489 | h263-2000/90000 | 2816            | n/a                         | n/a                    | n/a                | 0 / 0.00           | 0 / 0.00             | 0 / 0.00         | 64.62           | -70.00        | 70.00          | n/a                   | 0                    | 83.17 / 38.88 | 16 / -274     | 274.07 /    |

| Signalling Parameters          | Value                               | Audio Parameters                        | Value               | Video Parameters            | Value               |
|--------------------------------|-------------------------------------|-----------------------------------------|---------------------|-----------------------------|---------------------|
| Caller                         | 0003@192.168.1.203                  | Src RTP Channel                         | 192.168.1.203: 1034 | Src Media Type              | h263-2000/90000     |
| Callee                         | 0003@192.168.1.213                  | Src Media Type                          | PCMA/8000           | Src SSrc                    | 1137218817          |
| CallId                         | GL-MAPS_1_28527647-17505-5428@19... | Src SSRC                                | 1134240769          | Src Packet Count            | 2816                |
| Call Status                    | Terminated                          | Src Packets Count                       | 3099                | Src Missing Packets / (%)   | 0 / 0.00            |
| Call Initiated Time            | 2015-06-29 18:44:33.264             | Src Packets Lost / (%)                  | 0 / 0.00            | Src Duplicate Packet / (%)  | 0 / 0.00            |
| Call Established Time          | 2015-06-29 18:44:33.390             | Src Duplicate Packets / (%)             | 0 / 0.00            | Src Out of Sequence / (%)   | 0 / 0.00            |
| Call Stop Time                 | 2015-06-29 18:45:33.404             | Src Out of Sequence Packets / (%)       | 0 / 0.00            | Src Video Frame count       | 928                 |
| Call Duration                  | 00:01:00.013                        | Src Conversational MOS/R-Factor         | 4.20 / 93           | Src Frame Rate(Frames/sec)  | 15                  |
| Call Terminator                | Caller                              | Src Listening MOS/R-Factor              | 4.20 / 93           | Src AvgDelay                | -70.00              |
| Call Failure Reason            |                                     | Src Discarded Packets / (%)             | 0 / 0.00            | Src AvgGap                  | 64.62               |
| Session Request Delay (msec)   | 11.705                              | Src Average Inter Arrival Jitter (RTCP) | 1                   | Src MDI (DF:MLR)            | 186.89 : 0          |
| Session Disconnect Delay (m... | 6.567                               | Src Average Jitter                      | 1.00                | Src AvgMDI(DF:MLR)          | 130.50 : 0          |
| Post PickUP Delay (msec)       | 7.567                               | Src Average Delay                       | 0.00                |                             |                     |
| Total Signaling Frames         | 7                                   | Src Average Gap                         | 19.39               | Dest Video Channel          | 192.168.1.213: 1032 |
|                                |                                     | Dest RTP Channel                        | 192.168.1.213: 1030 | Dest Media Type             | h263-2000/90000     |
|                                |                                     | Dest Media Type                         | PCMA/8000           | Dest SSrc                   | 1151167489          |
|                                |                                     | Dest SSRC                               | 1150519041          | Dest Packet Count           | 2816                |
|                                |                                     | Dest Packets Count                      | 3100                | Dest Missing Packets / (%)  | 0 / 0.00            |
|                                |                                     | Dest Packets Lost / (%)                 | 0 / 0.00            | Dest Duplicate Packet / (%) | 0 / 0.00            |
|                                |                                     |                                         |                     | Dest Out of Sequence / (%)  | 0 / 0.00            |

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

# 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

The screenshot displays the NetSurveyorWeb web interface. At the top, the header includes the user 'GI', a 'Refresh' button, and dropdown menus for 'Protocol' (set to 'VOIP (SIP & H323)') and 'Type' (set to 'CDR'). A 'System Status' indicator shows the date and time as '2018-02-12 12:05:12' with a red status light.

The main navigation menu on the left includes sections for 'Quick CDR' (All Calls, Failed Calls, Passed Calls, Poor LMOS, Good LMOS, Longer Duration Calls, Voice Calls), 'Custom CDR' (CDR), 'Test' (Test), 'Test KPI' (Test KPI), 'Default KPIs' (Basic KPIs), 'Protocol Specific', 'Config', and 'MailBox'.

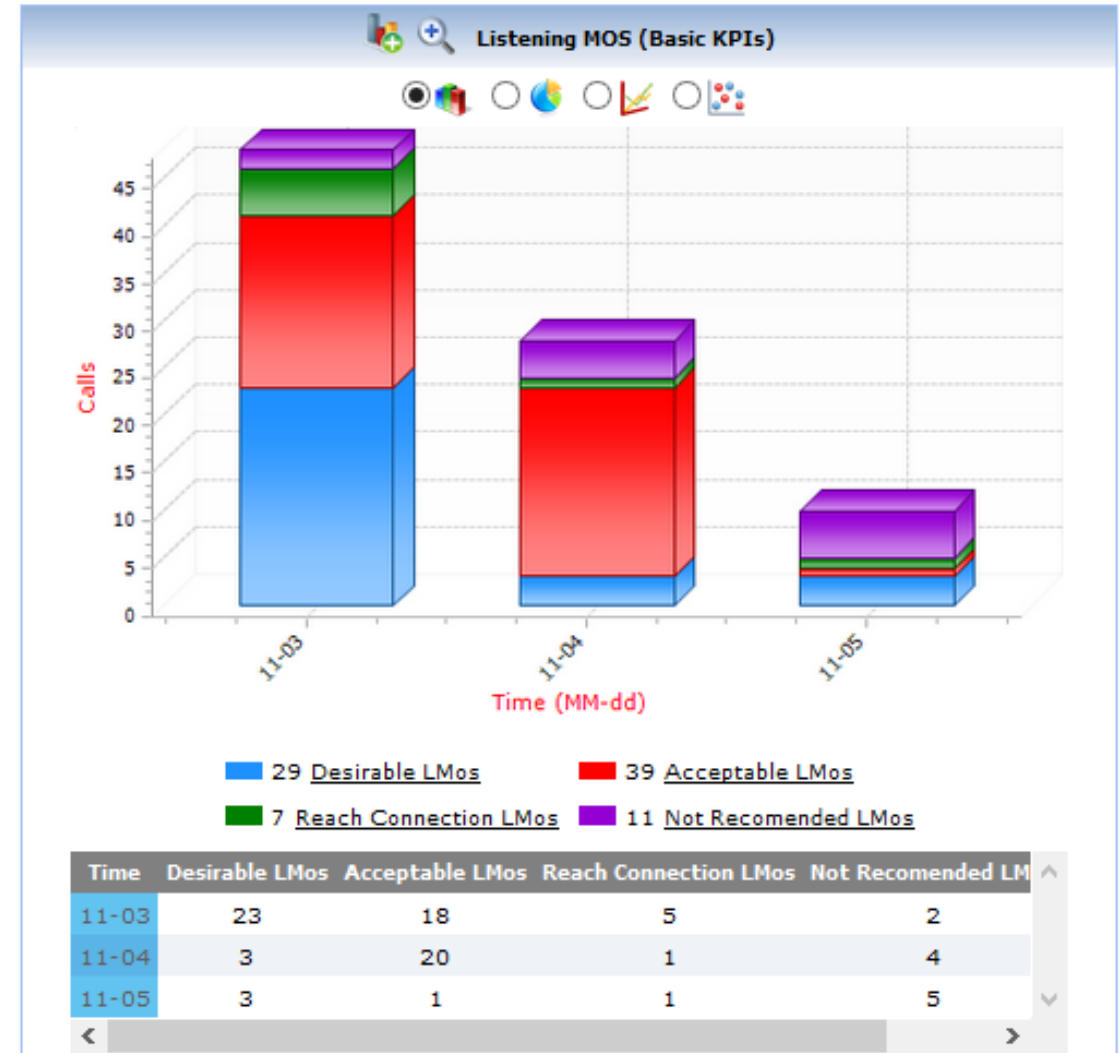
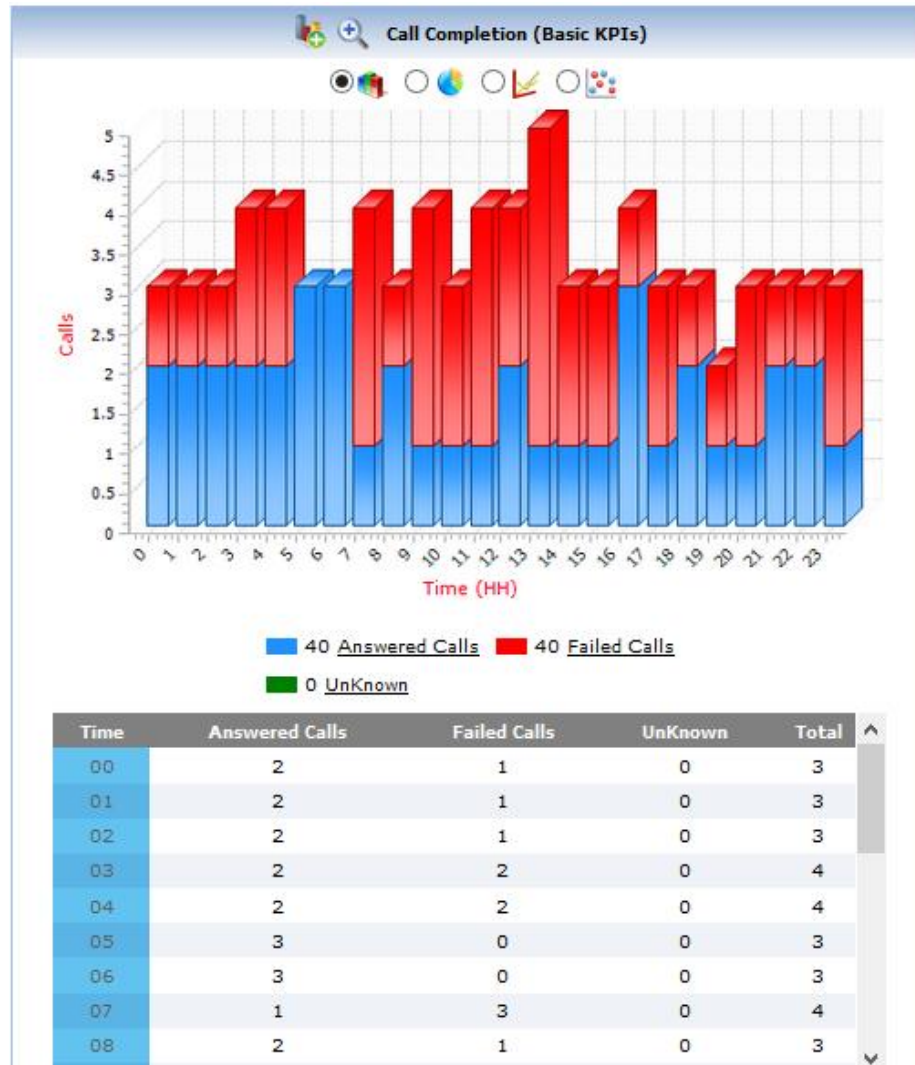
The central area shows a 'Quick CDR \ All Calls' view with a date range filter from '2018-01-01' to '2018-02-12' and a time filter from '00:00:00' to '23:59:59'. Below this, a 'Quick Search' field contains 'Trafficsumid' and a 'GO' button. The 'Query Execution Time' is noted as '0.19715 Seconds'.

The main content is a table of call records with the following columns: SI No, Calling Number, Called Number, Starttime, Duration, Call Success, and Failure Cause. The table contains 15 rows of data, each with a 'Call Flow' icon and a checkbox.

| SI No | Calling Number                                    | Called Number                                | Starttime               | Duration     | Call Success | Failure Cause |
|-------|---------------------------------------------------|----------------------------------------------|-------------------------|--------------|--------------|---------------|
| 1     | 001013012041639@ims.mnc001.mcc001.3gppnetwork.org | 3012041689@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.667 | 00:00:18.118 | 1            | 0             |
| 2     | 001013012041638@ims.mnc001.mcc001.3gppnetwork.org | 3012041688@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.666 | 00:00:18.118 | 1            | 0             |
| 3     | 001013012041637@ims.mnc001.mcc001.3gppnetwork.org | 3012041687@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.665 | 00:00:18.117 | 1            | 0             |
| 4     | 001013012041636@ims.mnc001.mcc001.3gppnetwork.org | 3012041686@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.663 | 00:00:18.117 | 1            | 0             |
| 5     | 001013012041635@ims.mnc001.mcc001.3gppnetwork.org | 3012041685@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.662 | 00:00:18.116 | 1            | 0             |
| 6     | 001013012041634@ims.mnc001.mcc001.3gppnetwork.org | 3012041684@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.661 | 00:00:18.115 | 1            | 0             |
| 7     | 001013012041633@ims.mnc001.mcc001.3gppnetwork.org | 3012041683@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.660 | 00:00:18.114 | 1            | 0             |
| 8     | 001013012041632@ims.mnc001.mcc001.3gppnetwork.org | 3012041682@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.659 | 00:00:18.011 | 1            | 0             |
| 9     | 001013012041631@ims.mnc001.mcc001.3gppnetwork.org | 3012041681@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-06 14:35:15.658 | 00:00:17.990 | 1            | 0             |
| 10    | 001013012041640@ims.mnc001.mcc001.3gppnetwork.org | 3012041690@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.865 | 00:00:09.629 | 1            | 0             |
| 11    | 001013012041639@ims.mnc001.mcc001.3gppnetwork.org | 3012041689@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.864 | 00:00:09.629 | 1            | 0             |
| 12    | 001013012041638@ims.mnc001.mcc001.3gppnetwork.org | 3012041688@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.863 | 00:00:09.629 | 1            | 0             |
| 13    | 001013012041637@ims.mnc001.mcc001.3gppnetwork.org | 3012041687@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.863 | 00:00:09.628 | 1            | 0             |
| 14    | 001013012041636@ims.mnc001.mcc001.3gppnetwork.org | 3012041686@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.862 | 00:00:09.628 | 1            | 0             |
| 15    | 001013012041635@ims.mnc001.mcc001.3gppnetwork.org | 3012041685@ims.mnc001.mcc001.3gppnetwork.org | 2018-02-02 16:48:10.862 | 00:00:09.628 | 1            | 0             |



# Reports



**Thank you**