RTP ToolBox[™] (RTP Simulation Tool)



Overview

GL's RTP ToolBox[™] testing and simulation tool is designed not only to monitor RTP and RTCP packets, but also to allow users to manually create and terminate RTP sessions, independent of call-signaling protocols such as SIP, H323, MEGACO, or MGCP.

This tool can be used for testing and developing enhanced voice features (VAD, echo cancellation, codec, digit regeneration, digit generation, fax over IP, jitter implementation, and more) within end-user equipment (IP Phones, ATA, MTA etc), testing media gateway telephony interfaces, end-to-end network testing before and during VoIP deployment, automated testing of digital signal processing embedded into network elements.

For more information, please visit <u>RTP ToolBox™</u> webpage.

Main Features

Capacity

- Create RTP sessions & Auto scan incoming RTP sessions; supports IPv6 addressing
- Can run on any PC with Windows[®] 7 /8 (32 bit and 64 bit) OS
- G.168 testing for echo cancellation equipment
- User-defined impairments: latency, packet loss, out of sequence, and duplicate packets
- Talk and play to speaker options using PC sound card

Call Generation

- Call generation and reception ability provides UA simulation (up to 8 UAs through CLI).
- Customize codec options (payload type, ptime) for UA during Call Generation & Reception.
- Multiple frame interval or Packetisation Time supported for almost all codec s.
- Generation / Detection of in-band and out-of-band Digits / Tones (DTMF, MF, user-defined, etc) / Events per RFC-2833 & RFC-4733.

Traffic Handling

- Set the RTP traffic properties (payload type, codec) and impairments during auto-scan.
- Sending and recording of voice files (.glw) with a synchronous Tx/Rx option.
- Set delay and attenuate for incoming RTP traffic.

Reports

- Monitoring RTP streams and captured data using scalable Oscilloscope and Spectrum Analyzer.
- Detailed statistical information of RTP and RTCP packets
- Quality Metrics with MOS (G.107 based E-model/R-Factor), jitter buffer statistics, degradation factor, and burst metrics are graphically represented.



818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878, U.S.A (Web) <u>www.gl.com</u> - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) <u>info@gl.com</u>

SIP Call Generation & Reception Capability

RTP ToolBox[™] allows users to configure and simulate a user agent (UA) for manual call generation and reception. Multiple calls can be placed and received through a single user agent. All the calls at the application end will be answered automatically. Up to 8 User Agents can be configured using the CLI.

The available options for user agent configuration include Public URL, Contact IP Addresses, Outbound Proxy, Registrar Address, NAT Address, & Re-register. In addition, more codec parameters such as Payload type, Packetisation time can be customized for each UA using Codec Options feature.

From URL	Туре	Port No	Codec	Status		Descriptio	n	
01@192.168.1.105	UDP	9002	Mu·law	HangUp)	Call Estab	lished	
Call Generation								
Eile								
To URL		Туре	Port No	Codec	Status		Descripti	on
001@192.168.1	1.113	UDP	9002	Mu-law	HangUp		Call Estal	blished
0002@192.168	1.113	UDP	9002	Mu-law	Dial		Call Faile	d
1				seragent C	opfig - [] In	titled]		
1				File	oning [oni	aacaj		-
				C Server Info Outband Registrar A Expiry	ontact * 00 mation I Proxy ddress y Time 36 V	1@192.168.1. 00 Register on St Re-Register	29 Nartup	msec
				User Accou Use Pas IV Nat Add	unt Informatic ername seword dress	n . 60		-

Impairments, & Delay / Attenuate

Users can manually introduce impairments and transmit on the RTP sessions. This includes introducing fixed latency, uniform/normal distributed latency, periodic/random/burst packet loss, out-of-order packets, and duplicate packets. Users may also apply delay and attenuate to the incoming data on a session.

Impairments - Default		×
Impairment Name Tefault		
Latency Packet Loss Packet	Effects	
Fixed Latency None Fixed Latency		
Laten Normal Distributed	Delay And Attenuate [Strea	m ID - 1]
	Delay 200	ms
	Attenuate 40	dBm
<u>S</u> et as Default	Start]

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Voice Codec Options

The Call Generation (Dial) & Call Reception features provides various codec parameters in the TX/RX profiles during negotiation.

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

RTP ToolBox[™] supports the following codecs:

- G.711 (A-law / Mu-law 64kbps), G.711 App II (A-Law and Mu-Law with VAD Support)
- G.722 (64 kbps) , G.722.1 (32 kbps and 24 kbps)
- G.729, G.729B (8 kbps)
- G.726 , G.726 (40/32/24/16 kbps with VAD)
- GSM 6.10 FR (13.2 kbps), GSM-HR (rate 5.6kbps)
- GSM-EFR (12.2kbps, packet time fixed at 20ms)
- SPEEX, SPEEX_WB (packet time fixed to 20msec)
- iLBC, iLBC_13_33
- SMV (Modes 0, 1, 2 and 3- Available if licenses are provided or owned, please call GL)
- AMR (4.75kbps, 5.15kbps, 5.9kbps, 6.7kbps, 7.4kbps, 7.95kbps, 10.2kbps, 12.2 kbps), AMR WB (optional codec)
- EVRC ($1/8, \ensuremath{\rlap{}^{1}_{2}}\xspace$ and 1) , EVRC0 (optional codec)
- EVRC_B (1/8, ¼, ½, and 1), EVRCB0 (optional codec)
- EVRC_C (optional codec)

For more information, please visit Voice Codec webpage.

Profile - Default	×
Profile Name Default	Profile - Default
Tx Profile Rx Profile	Profile Name Sefault
Codec Mu-law	Tx Profile Rx Profile Voice Payload Type 1 RFC 2833 Payload Type 0 Comfort Noise Payload Type 13 © Static Dynamic Jitter Buffer Len 1000 Dynamic Jitter Buffer Option Min Delay 40 Max Delay 40 Fill with Last Packet
Note : If Parameters are invalid previou values will be retained	Note : If Parameters are invalid previous values will be retained
	Decias Derauic Save Profile



RTP Traffic Generation

Transmit/Record Tones, Digits

RTP ToolBox[™] can be used to generate in-band digits and tones. The supported tones include single, dual, and multi-tones. Supported digits include DTMF, MF, and MFR2 forward and backward digits. The generation of RTP Events/Digits per RFC-2833 & RFC-4733 is available.

The RTP ToolBox[™] application allows capturing tones and digits in the traffic. It also displays additional information about the captured signal such as type of the signal, timestamp, event, power, and more. This is completely supported for both in-band digits/tones and RTP digits/events per RFC-2833 & RFC-4733.

Options C Digits Only	All Activity Show Latest	Clear @ RFC 283	13 О П	IFC 4733		X	
TimeStamp	Event DTMF 8	RTP Event Generation [Stream	n ID - 2]			1	×
16:20:37.441 16:21:21:912 16:21:30.975 16:21:30.006 16:21:49.084 16:21:58.147 16:24:47.319 16:24:47.319 16:25:26.569 16:26:20.928 16:26:51.131 16:26:54.147 16:27:36.428	D TIMF 4 D TIMF 6 D TIMF 5 D TIMF 5 D TIMF 5 Calling tone (CNG) Special ringing tone D TIMF 8 Busy tone Calling tone (CNG) D TIMF 9 ANSam	Events DTMF 9 DTMF 9 Answer tone (ANS) /ANSam Calling tone (CNG) V.21 channel 1, "1" bit PABX internal dial tone Special inging tone Busy tone	On Tim 100 100 100 100 100 100 100 100	e Off Time 100 100 100 100 100 100 100 100 100 10	Powe10 -10 -10 -10 -10 -10 -10 -10 -10 -10	Event Values On Time 100 Off Time 100 Power -10	
16:27:33:444 16:27:42:475 Save Events ✓ Captured Eve	PABX internal dial tone	1 2 3 A C 4 5 6 B C 7 8 9 C 0 * # D ☑ Send Continuousty	DTMF MF F	Continuity Continuity Loopback Old milliwa New milliw	test send verified tt tone (1000 att tone (100 Add	▲ 1Hz) 4 Hz) ▼	

Transmit/Record Voice File

The application can also send voice files (*.wav and *.pcm) & record the incoming voice data to file, limited to desired no. of bytes and time. These files can be compared with GL's optional Voice Quality Testing software, providing PESQ score. The ability to send and record files also allows G.168 testing for echo cancellers.

Playback From File [Stream ID - 5] 📃 🗐 🗙	👖 Record Data To File [Stream ID - 5] 🛛 📃 🗙
Audio File p Toolbox/Voice Samples\PCM\Vijay16k.pcm	Format © PCM (16bit Linear) C Native
Continuous Transmission	Audio File
© None	C:\Program Files\GI Communications Inc\Rtp Too
C Limit Transmit (Bytes)	C None
Synchronize Operation	C Limit Capture (Cytes)
C Master C Slave	V Synchronize Operation
Start Stop	C Master 📀 Slave
Output Codec Name Mu-law Input Codec Name PCM Output Bute Count 303680	Start Stop
Output Packet Count 1898 Input Byte Count 611520	Output Codec Name PCM Input Codec Name Mu-law Dutrut Rute Count 570500
	Input Byte Count 0
	Input Error Packet Count 0



Oscilloscope and Spectrum Analyzer

The PCM codes (amplitude of the incoming signal) for any selected session are graphically displayed in real-time as a function of time. The data received on a specified timeslot can be viewed in the spectral domain (Spectral Amplitude vs. Frequency). A Fast Fourier Transform (FFT) is applied to successive sample sets of the incoming data and displayed in graphic form. The FFT length can adjust the frequency resolution (from 32 points to 8192 points).



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.

Jitter Buffer Statistics, Quality Metrics (R&MOS), Degradation Factor, Burst Metrics

Jitter Buffer feature allows setting the buffer used for delayed packets that arrive at receiving end. Both static and dynamic jitter buffers are supported. Quality metrics include various graphs for G.107 based E-model/R-Factor score reporting. **R-Factor Statistics** will display statistics such as - R-Listening, R-Conversational, R-G107, and R-Nom. The MOS graph will display statistics such as MOS CQ, MOS LQ and MOS Nom. It also supports Burst Metrics and Degradation Factor statistics.





Client-Server Functionality (requires additional license)

RTP ToolBox[™] can be configured as server-side application, to enable remote controlling of the application through multiple commandline based clients. Supported clients include C++ and TCL based clients. User can remotely perform all functions such as creating RTP sessions, Digit/Tones/Event generation and reception, Setting impairments, Creating session profiles & so on. User can also generate and receive SIP calls through commands. The RTP sessions associated with the SIP call are created automatically.



Script Processing

RTP ToolBox[™] provides easy to use interface to execute scripts (*.psc files) on selected sessions. Scripts can also be run on multiple sessions at the same time and its progress can be viewed in the Script Contents pane by highlighting the currently executing command of the script. For enhanced testing, users can also write IVR (Interactive Voice Response) scripts.

Sino	Source Address	Source Por	t Destination Address	Destination F	Port Codec	Status	Profile	Impairments	Script File Name	Script Statu
1	192.168.1.113	1024	192.168.1.113	1024	Mu·law	Stop	Default	Default	allactions	Start
2	192.168.1.113	2000	192.168.1.113	2000	Mulaw	Stop	Default	Default	1	Start
3	192.168.1.113	3000	192.168.1.113	3000	Mu·law	Start	Default	Default		Start
4	192.168.1.113	4000	192.168.1.113	4000	Mu·law	Start	Default	Default		Start
5	192.168.1.113	5000	192.168.1.113	5000	Mu·law	Start	Default	Default		Start
Inset			otop oonp	1						
Inser Sh	ow Progress								27	
Inser Sh Script	ow Progress									



Media Gateway Testing using RTP ToolBox™

- Complete G.168 Compliance Testing (All 13 Tests) Tests 1, 2A, 2B, 2C, 3, 4, 5, 6, 7, 8, 9, 10A, 10B, 11, 12, 13, 14, 15.
- Voice Quality Testing using PESQ
- Codec Testing and Verification



G.168 Compliance Test for EC within ATA

G.168 Tests which can be performed on an ATA using RTP ToolBox[™] include Tests 1, 2A, 2B, 2C, 3, 4, 5, 6, 7, 8, 9, 10A, 10B.





Buyer's Guide

Item No	Product Description
<u>PKB100</u>	RTP ToolBox™ Application
<u>PKB110</u>	RTP Toolbox [™] with Client-Server Application
PCD103	AMR codec for RTP Toolbox™ (requires additional license)
PCD104	EVRC codec for RTP Toolbox™ (requires additional license)
PCD105	EVRC_B codec for RTP Toolbox [™] (requires additional license)
PCD106	EVRC_C codec for RTP Toolbox™ (requires additional license)
-	
Item No	Related Software
Item No PKS120	Related Software Message Automation & Protocol Simulation (MAPS)
Item No PKS120 IPN100	Related Software Message Automation & Protocol Simulation (MAPS) IPNetSim™ - 1Gbps of through bandwidth
Item No PKS120 IPN100 PKB105	Related Software Message Automation & Protocol Simulation (MAPS) IPNetSim™ - 1Gbps of through bandwidth G.168 Echo Canceller Test Compliance Suite
Item No PKS120 IPN100 PKB105 PKBT67	Related Software Message Automation & Protocol Simulation (MAPS) IPNetSim™ - 1Gbps of through bandwidth G.168 Echo Canceller Test Compliance Suite Automated Echo Canceller Testing – T1 Version
Item No PKS120 IPN100 PKB105 PKBT67 PKS100	Related Software Message Automation & Protocol Simulation (MAPS) IPNetSim™ - 1Gbps of through bandwidth G.168 Echo Canceller Test Compliance Suite Automated Echo Canceller Testing – T1 Version PacketGen™ with PacketScan™

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