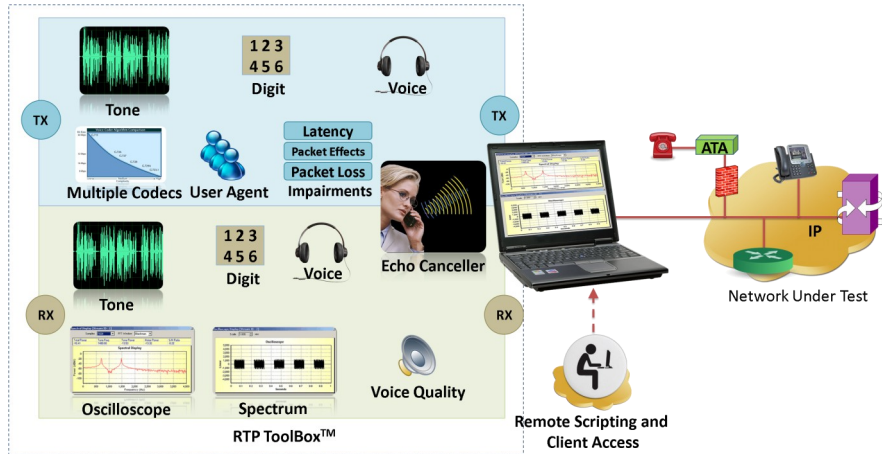


RTP ToolBox™ (RTP Simulation Tool)



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.

Main Features

- Create RTP sessions & Auto scan incoming RTP sessions.
- Supported Codec's: G.711 (Mu-law/A-law), G.729a, G.729b, G.726-5 bit 40kbps/4 bit 32 kbps/3 bit 24 kbps/2 bit 16kbps, G.726-40/32/24/16kbps with VAD, GSM (Full rate), AMR (All rates), EVRC, EVRC0, EVRCB, EVRCB0, EVRC_C, SMV, ILBC (20 msec), ILBC (30 msec), SPEEX, G.722, G.722.1, SPEEX_WB, G.711 App II (ALaw & MuLaw with VAD), iSAC, and AMR_WB
- 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.
- Set the RTP traffic properties (payload type, codec) and impairments during auto-scan.
- Monitoring RTP streams and captured data using scalable Oscilloscope and Spectrum Analyzer.
- Generation / Detection of in-band and out-of-band Digits / Tones (DTMF, MF, user-defined, etc) / Events per RFC-2833 & RFC-4733.
- Set delay and attenuate for incoming RTP traffic.
- User-defined impairments: latency, packet loss, out of sequence, and duplicate packets.
- Detailed statistical information of RTP and RTCP packets.
- Sending and recording of voice files (.glw) with a synchronous Tx/Rx option.
- G.168 testing for echo cancellation equipment.
- Talk and play to speaker options using PC sound card.
- Quality Metrics with MOS (G.107 based E-model/R-Factor), jitter buffer statistics, degradation factor, and burst metrics are graphically represented.
- Can run on any PC with Windows® XP (32 bit and 64 bit)/Vista (32 bit)/7 (32 bit and 64 bit) OS.

For more details, visit <http://www.gl.com/rtptoolbox.html>.

Monitor/ Create/ Terminate
RTP Sessions

Scan Automatically for RTP
Sessions

Provides Oscilloscope and
Spectrum Analyzers

Generates or Captures Single/
Dual / Multi-tones / DTMF / MF/
MFR2 Digits

Generates / Captures RFC-2833
& 4733 Digits & Events

Delay and Attenuate for
Incoming RTP Traffic

Graphically Represents Jitter
Buffer Statistics, and
Degradation Factor Statistics

Quality Metrics (R & MOS
Factors), Burst Metrics Graphs

Supports Synchronous Tx/Rx
of Voice Files



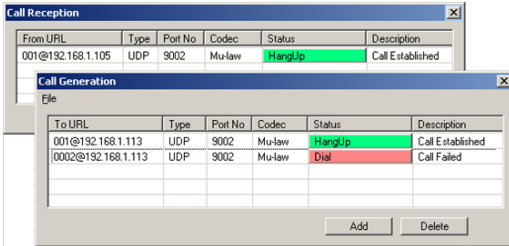
GL Communications Inc.

818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878, U.S.A

(Web) <http://www.gl.com/> - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) gl-info@gl.com

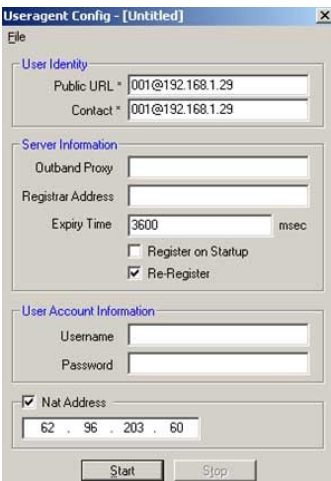
SIP Call Generation & Reception Capability

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, so received calls shows the status as Hang-Up indicating that call has been established. Up to 8 User Agents can be configured using the CLI.



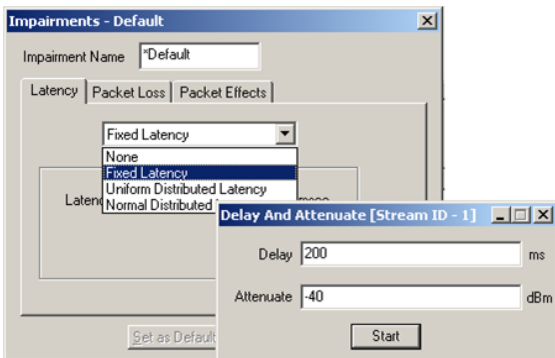
User Agent Configuration

The available options for user agent configuration include Public URL, contact IP addresses, Outbound Proxy, Registrar Address, NAT Address, & Re-register. The fields such as Outbound Proxy, Registrar Address, NAT Address, Username and Password are optional. In addition, more codec parameters such as Payload type, Packetisation time can be customized for each UA using codecoptions.ini file. The Call Generation (Dial) & Call Reception features use the codec information in the profile during negotiation.



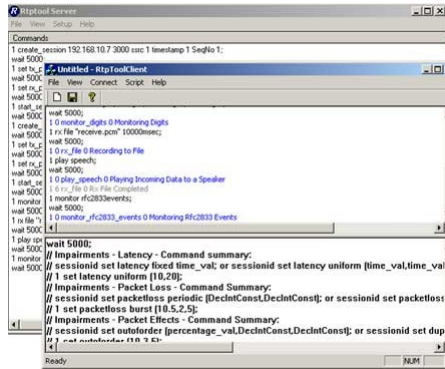
Impairments, & Delay / Attenuate

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.



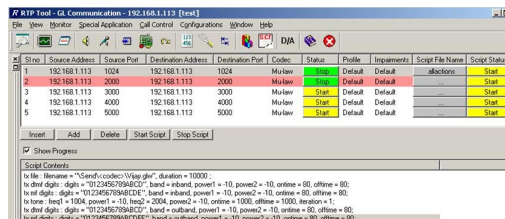
Server-Client Functionality (requires additional license)

RTP ToolBox™ can be configured as server-side application, to enable remote controlling of the application through multiple command-line 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.



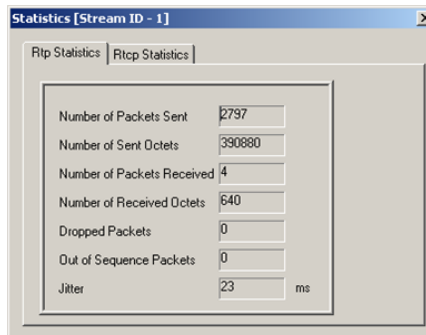
Script Processing

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.



RTP/RTCP Packet Statistics

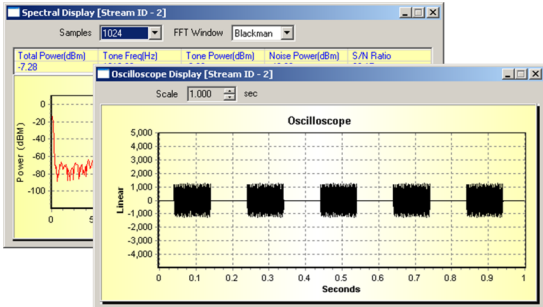
Statistics reports of RTP and RTCP packets transmitted on a session such as number of packets sent/received, dropped packets, out of sequence packets and more. Sender and receiver reports are also displayed using RTP/RTCP statistics applications.



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

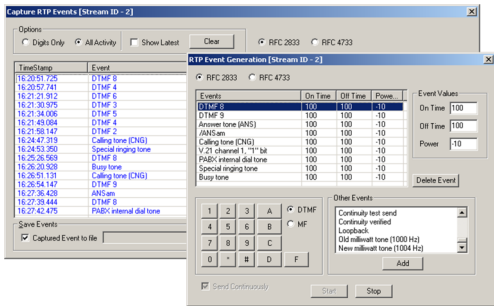
Oscilloscope and Spectrum Analyzer

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).



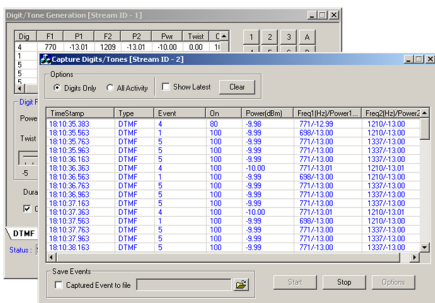
Generation/Detection of Digits/Tones/RTP Events Generation

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.



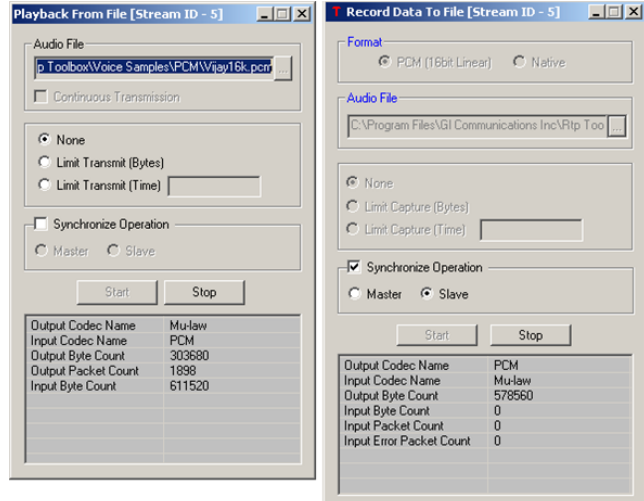
Detection

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.



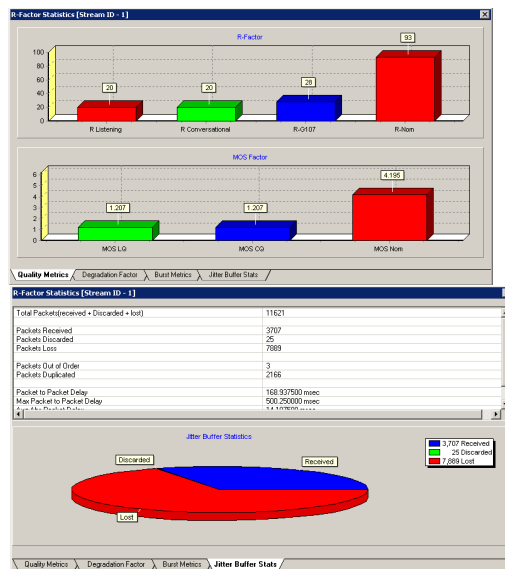
Transmit/Record Voice File

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, PAMS and PSQM score. The ability to send and record files also allows G.168 testing for echo cancellers.



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.



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

Creating / Monitoring RTP Sessions

Tx -Profile

RTP ToolBox™ supports the following codecs:

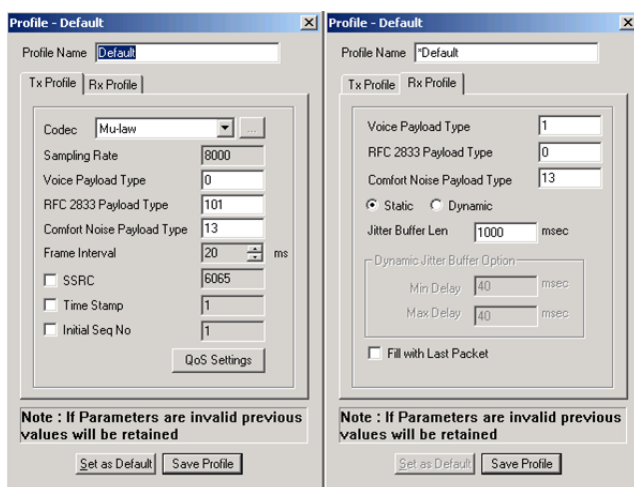
- G.711 (A-law / Mu-law - 64kbps)
- G.711 App II (A-Law and Mu-Law with VAD Support)
- G.729, G.729B (8 kbps)
- G.726 (5 bit 40kbps/4 bit 32kbps/3 bit 24kbps/2 bit 16 kbps)
- G.726 (40/32/24/16 kbps with VAD)
- GSM (13.2 kbps)
- AMR (4.75kbps, 5.15kbps, 5.9kbps, 6.7kbps, 7.4kbps, 7.95kbps, 10.2kbps, 12.2 kbps) (optional codec)
- AMR_WB
- EVRC (Rates - 1/8, 1/4 and 1) , EVRC0 (optional codec)
- EVRC_B (Rates - 1/8, 1/4, 1/2, and 1), EVRCB0 (optional codec)
- EVRC_C (optional codec)
- SMV (Modes - 0, 1, 2 and 3- Available if licenses are provided or owned, please call GL)
- iLBC (for 20msec)
- iLBC_13_33 (for 30msec)
- SPEEX
- G.722
- G.722.1 (32 k and 24 k)
- SPEEX_WB
- iSAC (optional codec)

Sampling rate of the codec is displayed for the selected codec.

Comfort noise generation is supported for A-law, μ -law and G726 codecs for sending and receiving payload.

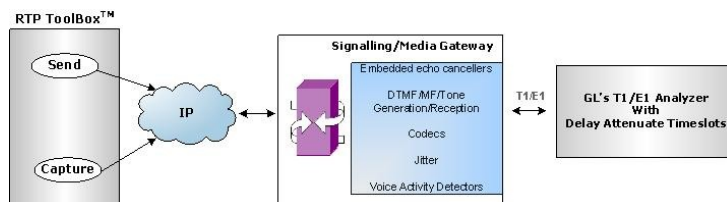
Rx -Profile

User can specify a desired voice payload type to each codec for receiving data; comfort noise generation is supported for A-law, μ law and G726 codecs. SID packets for these codecs are carried with separate payload type, which will be different from voice payload type (By default it is 13). Users can set the buffer used for delayed packets that arrive at receiving end (both static and dynamic jitter buffers are supported), and, after the session is established, transmit inserting required impairments.



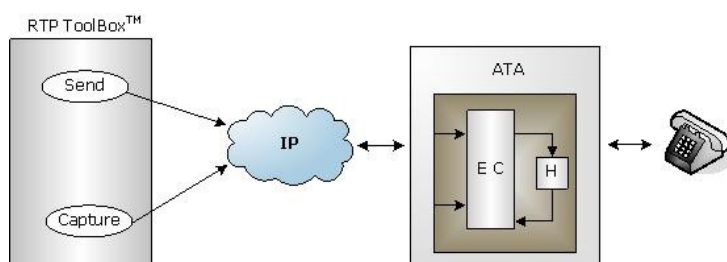
Media Gateway Testing using RTP ToolBox™

- 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, PAMS and PSQM
- 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.



Buyers Guide

- [PKB100](#) - RTP ToolBox™ Application
- [PKB110](#) - RTP Toolbox™ with Client-Server Application
- [PCD103](#) - AMR codec for RTP Toolbox™
- [PCD104](#) - EVRC codec for RTP Toolbox™
- [PCD105](#) - EVR_B codec for RTP Toolbox™
- [PCD106](#) - EVR_C codec for RTP Toolbox™
- [PCD102](#) - ISAC codec for RTP Toolbox™

Related Software

- [PKS120](#) – Message Automation & Protocol Simulation (MAPS)
- [IPN100](#) – IPNetSim™ - 1Gbps of through bandwidth
- [PKB105](#) – G.168 Echo Canceller Test Compliance Suite
- [PKBT67](#) – Automated Echo Canceller Testing – T1 Version
- [PKS100](#) – PacketGen™ with PacketScan™
- [PKS110](#) – Packet H. 323
- [VQT004](#) – VQT with PESQ, PAMS and PSQM



818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878, U.S.A

(Web) <http://www.gl.com/> - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) gl-info@gl.com