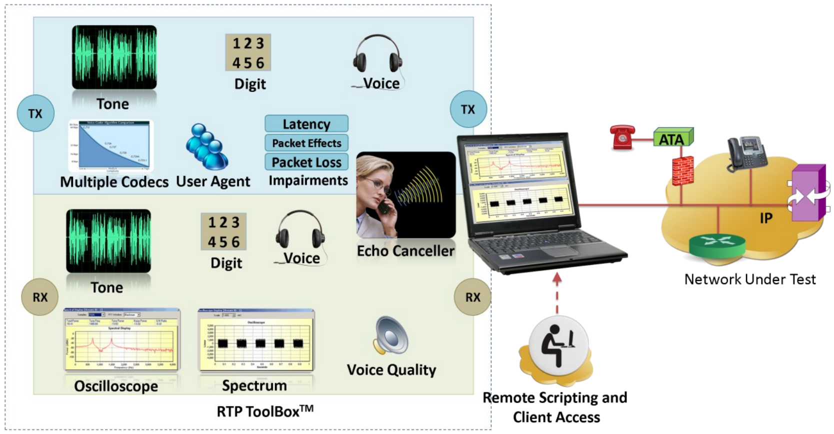
RTP ToolBox™ (RTP Simulation Tool)



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 [RTP ToolBox™](#) 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.

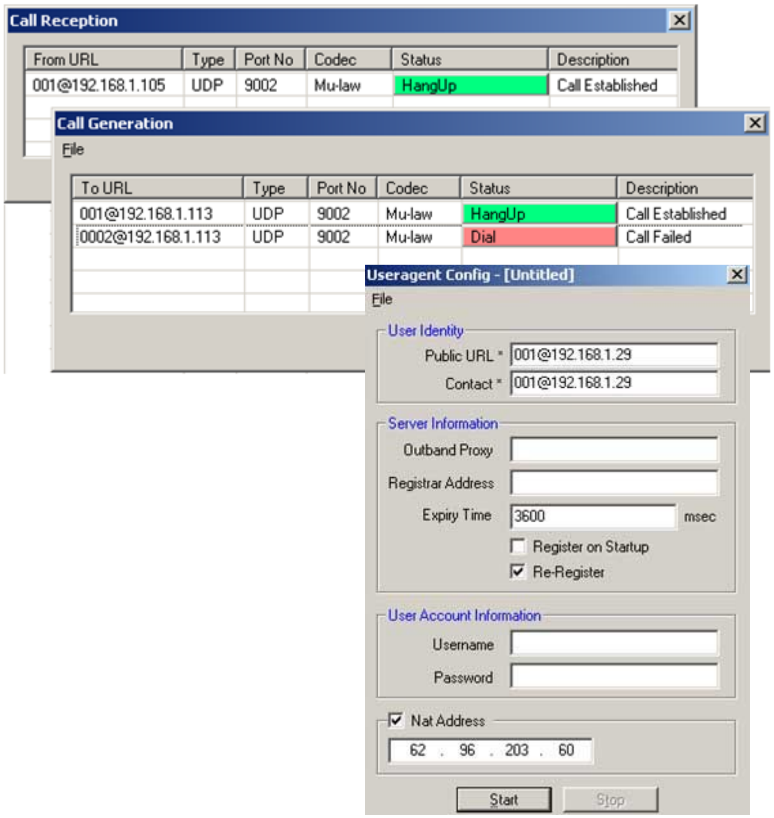
Reports

- 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.

SIP Call Generation & Reception Capability

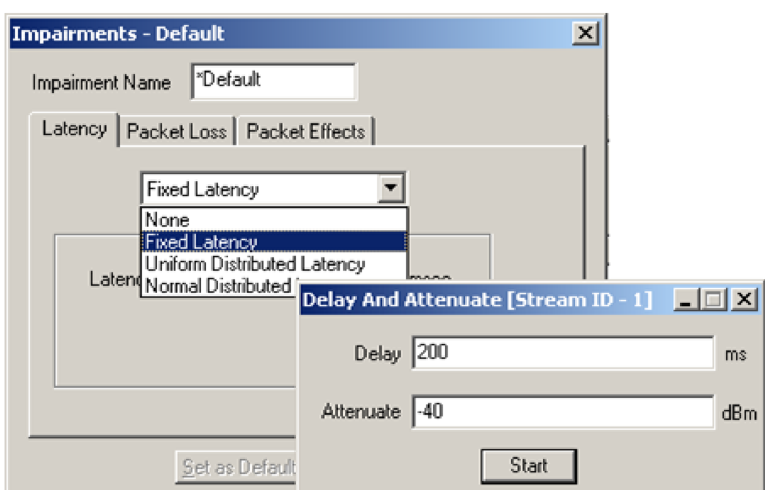
RTP ToolBox™ allows users to configure and simulate a user agent (UA) for manual call generation and reception. Multiple calls can be placed and received through a single user agent. All the calls at the application end will be answered automatically. Up to 8 User Agents can be configured using the CLI.

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.



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.



Voice Codec Options

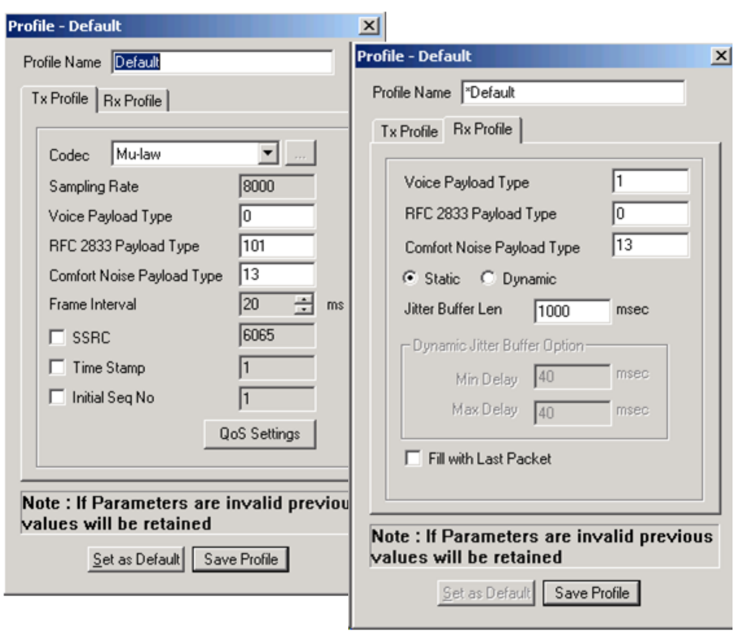
The Call Generation (Dial) & Call Reception features provides various codec parameters in the TX/RX profiles during negotiation.

- 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, ½ 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.

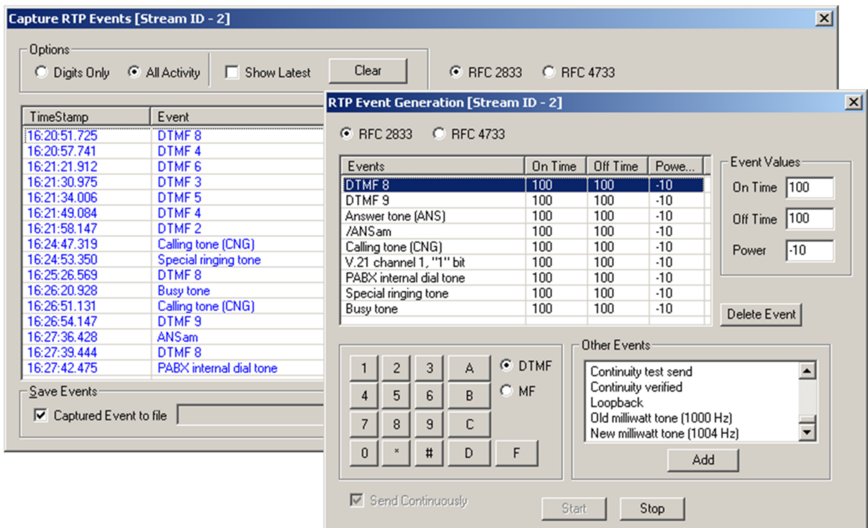


RTP Traffic Generation

Transmit/Record Tones, Digits

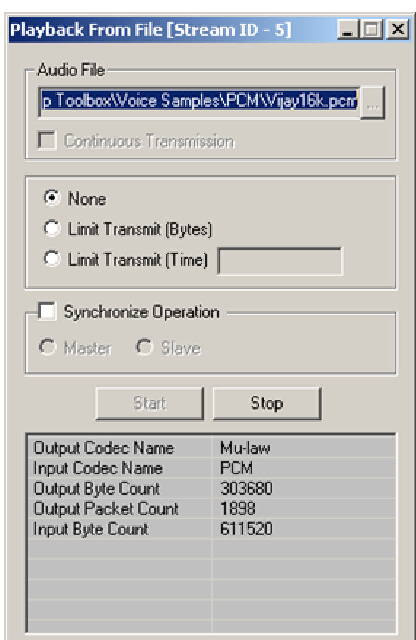
RTP ToolBox™ can be used to generate in-band digits and tones. The supported tones include single, dual, and multi-tones. Supported digits include DTMF, MF, and MFR2 forward and backward digits. The generation of RTP Events/Digits per RFC-2833 & RFC-4733 is available.

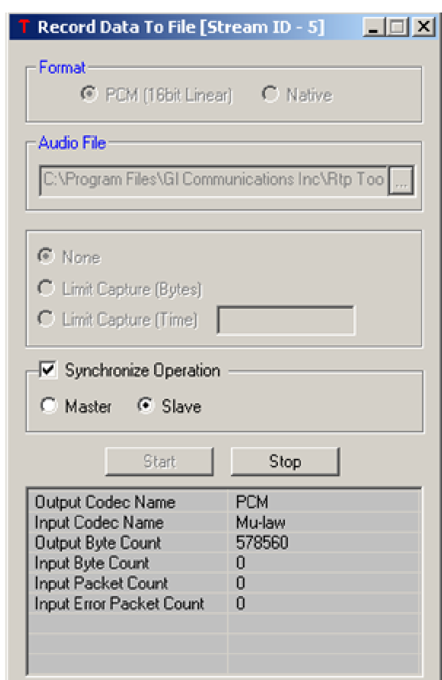
The RTP ToolBox™ application allows capturing tones and digits in the traffic. It also displays additional information about the captured signal such as type of the signal, timestamp, event, power, and more. This is completely supported for both in-band digits/tones and RTP digits/events per RFC-2833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



Transmit/Record Voice File

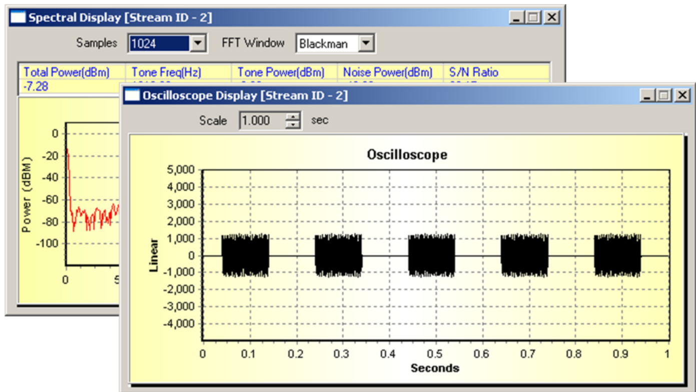
The application can also send voice files (*.wav and *.pcm) & record the incoming voice data to file, limited to desired no. of bytes and time. These files can be compared with GL's optional Voice Quality Testing software, providing PESQ score. The ability to send and record files also allows G.168 testing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





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).

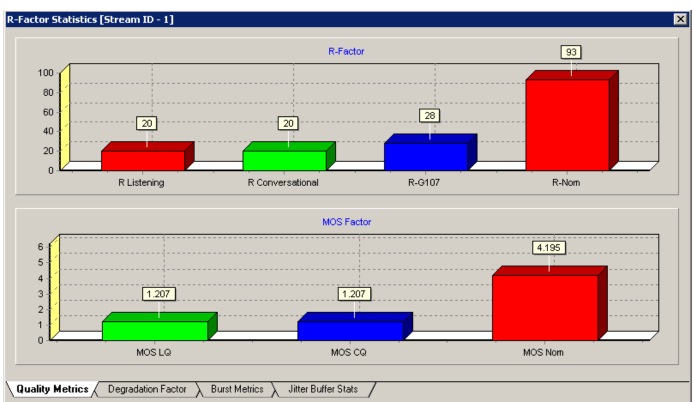


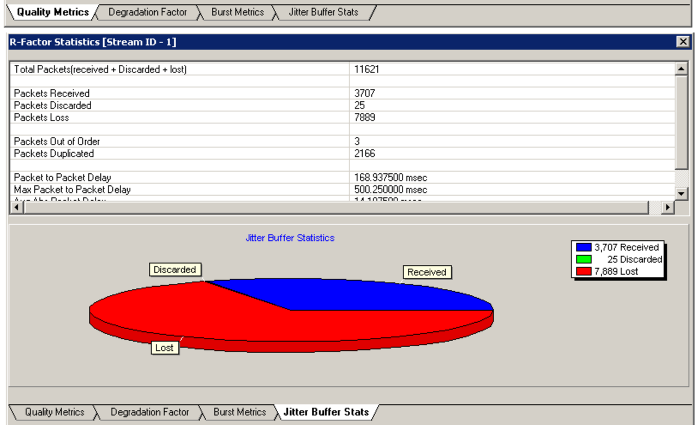
RTP/RTCP Packet Statistics

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.





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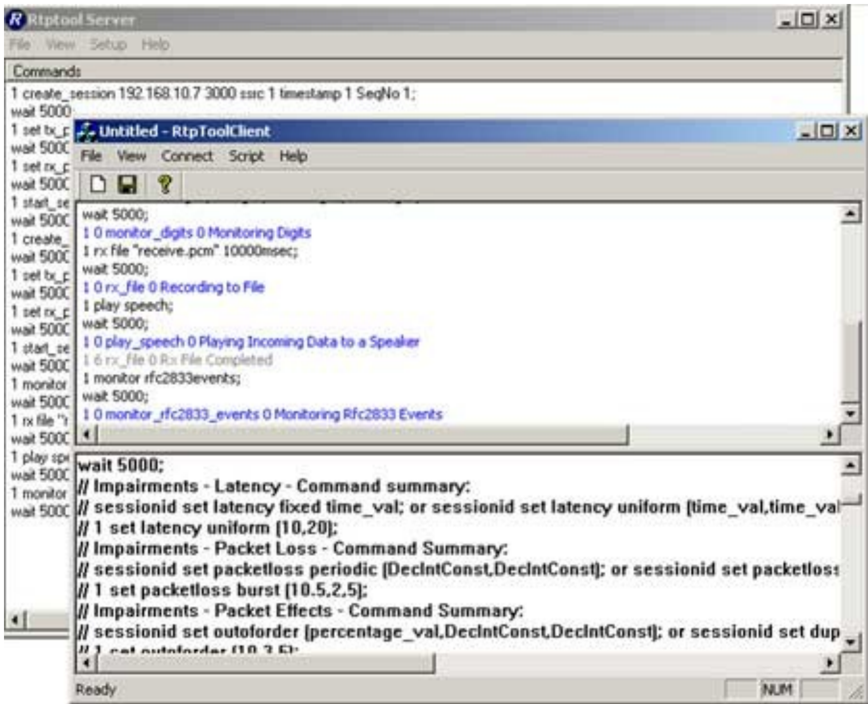
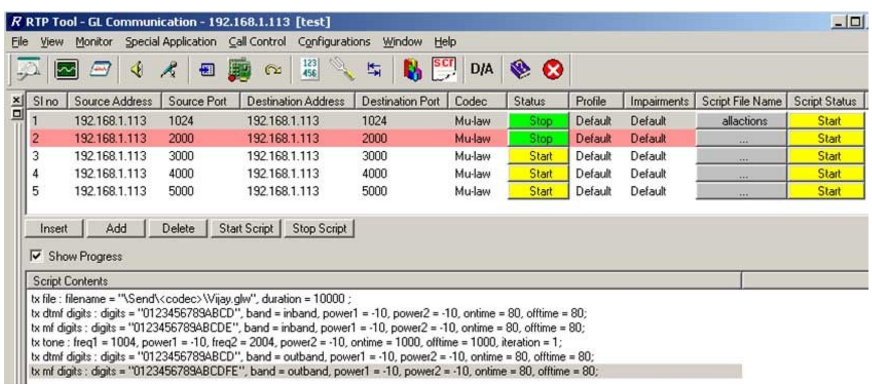
Client-Server Functionality (requires additional license)

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.



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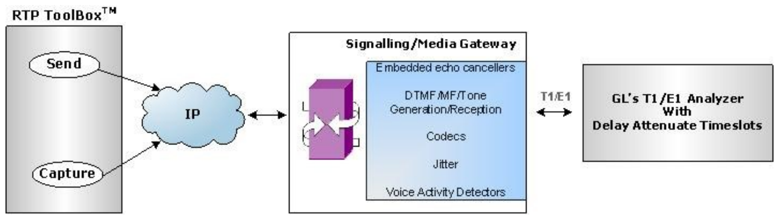
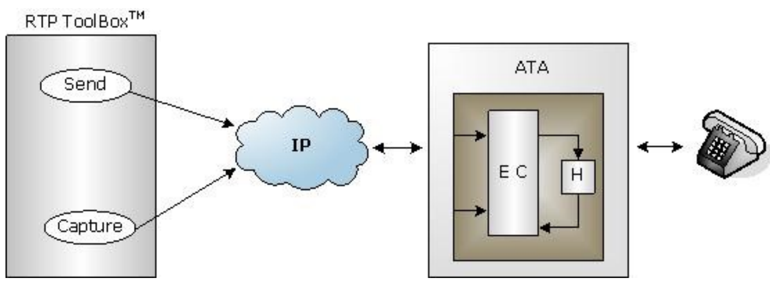
Media Gateway Testing using RTP ToolBox™

- 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
PKB100	RTP ToolBox™ Application
PKB110	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
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

For more information, please visit RTP ToolBox™ webpage.