RTP ToolBox™

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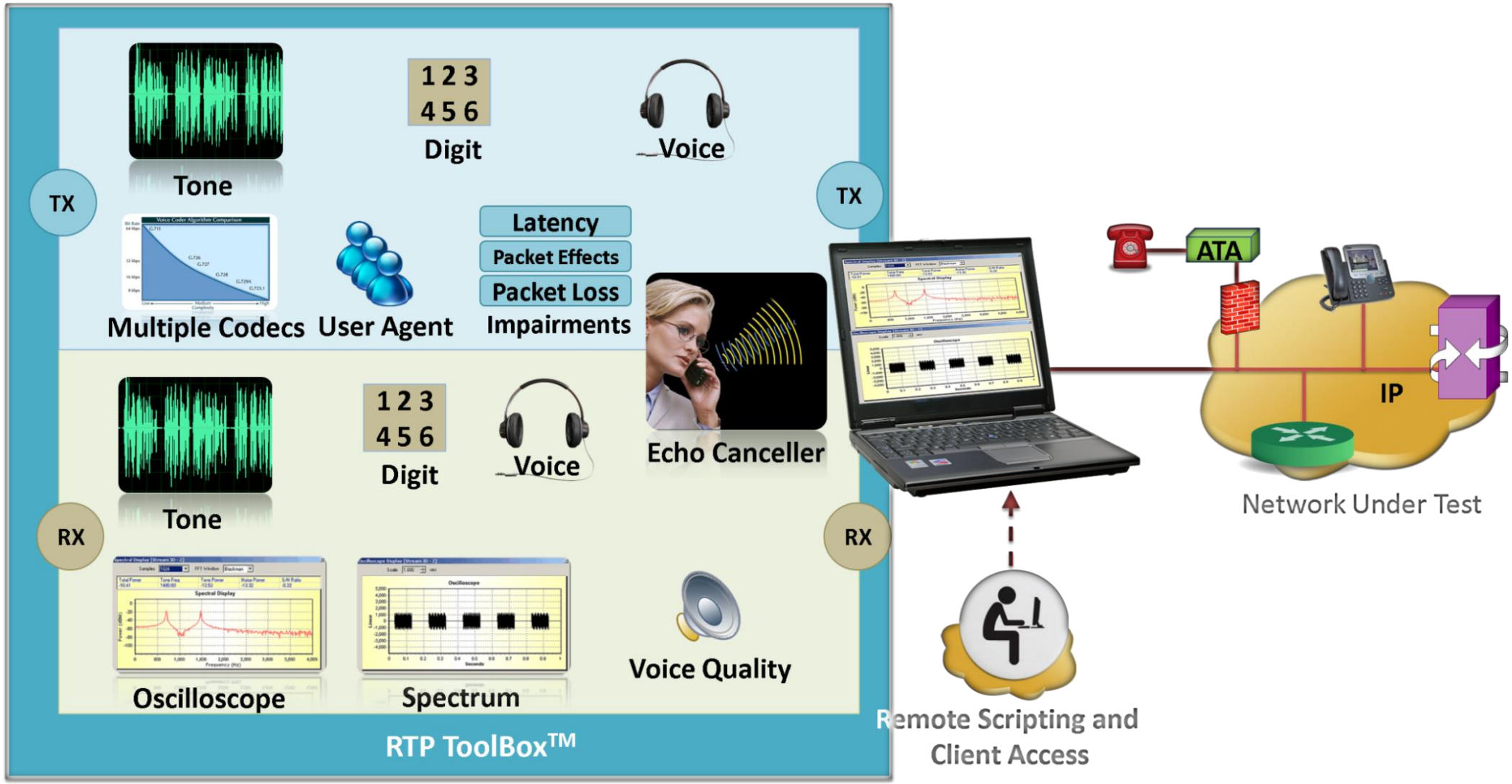
RTP ToolBox™


TX
Tone
1 2 3
4 5 6
Digit
Voice
Multiple Codecs
User Agent
Latency
Packet Effects
Packet Loss
Impairments
RX
Tone
1 2 3
4 5 6
Digit
Voice
Echo Canceller
Oscilloscope
Spectrum
Voice Quality
RTP ToolBox™
TX
RX
ATA
IP
Network Under Test
Remote Scripting and
Client Access

Applications

- Testing and developing enhanced voice features (VAD, Echo Cancellation, Codec, Digit Regeneration, Digit Generation, Fax over IP, Jitter Implementation etc) within end-user equipment (IP Phones, ATA, MTA etc)
- End-to-End network testing before and during VoIP deployment
- Testing media gateway telephony interfaces
- 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(32 k and 24 k), SPEEX_WB, G.711 App II (ALaw & MuLaw with VAD), iSAC, and AMR_WB
- Automated testing of Digital Signal Processing embedded into network elements
- Echo cancellation testing & compliance

Key Features

- Manually create and terminate RTP sessions independent of call-signaling protocols such as SIP, H323, MEGACO, or MGCP
- Monitor RTP and RTCP packets
- Automatic scan option to capture all incoming RTP and RTCP packets
- Monitoring RTP streams using scalable oscilloscope and spectrum analyzer
- Generation / detection of in-band and out-of-band digits / tones (DTMF, MF, user-defined)/ events per RFC-2833
- User-defined impairments: latency, packet loss, out of sequence and duplicate packets
- Multiple frame interval or Packetisation Time supported for almost all codecs

Key Features (Contd.)

- Sending and recording of voice files (.glw) with a synchronous Tx / Rx option
- G.168 testing for echo cancellation equipment
- 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 (Dial) & Call Reception
- Quality metrics with R-Factor and MOS Factors, Jitter Buffer Statistics, Degradation Factor, and Burst Metrics are graphically represented
- Supports client-server functionality (requires additional license) - C++ & TCL clients

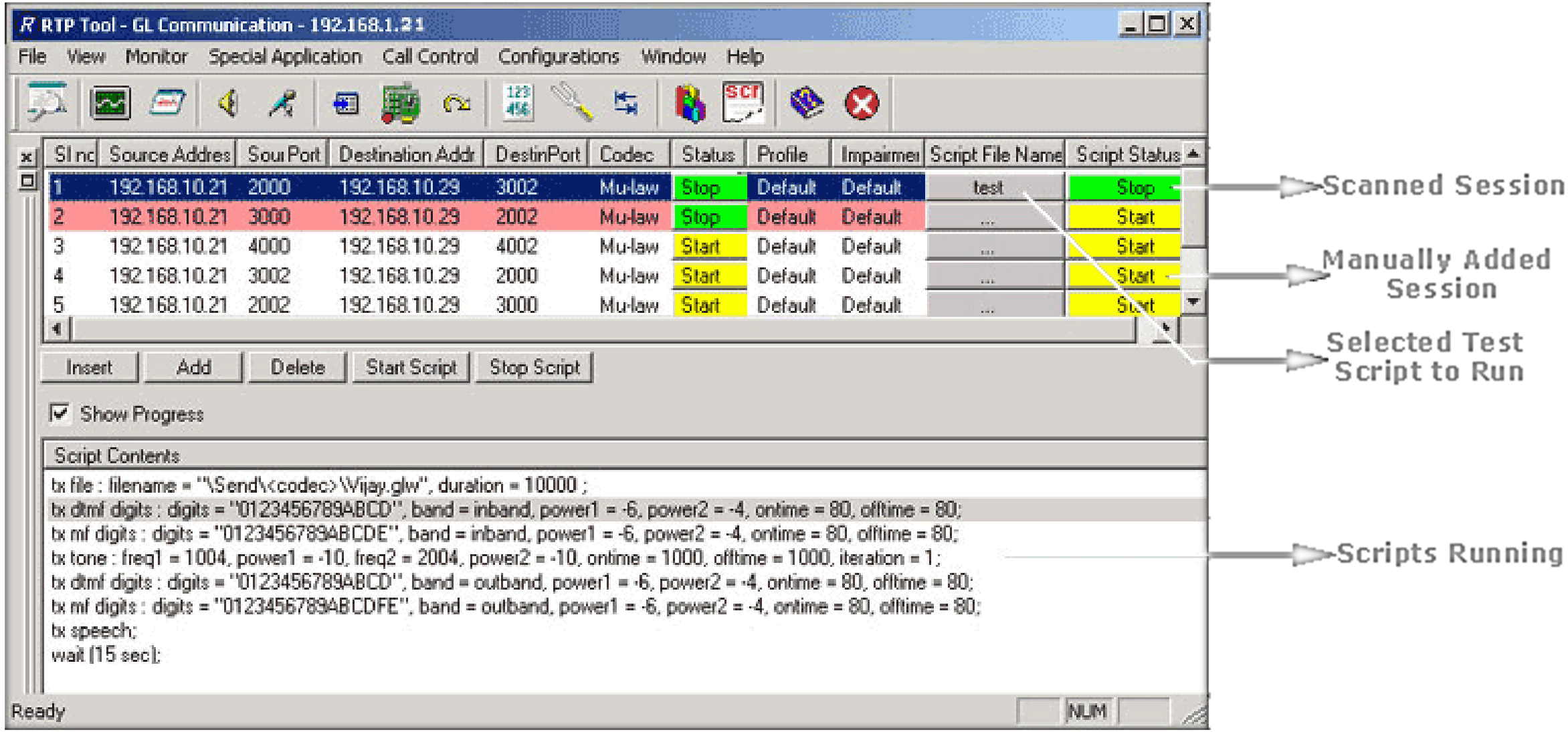
Supported Codecs

Codec Name	Sampling Rate	VAD Support
G,711 (Mu-law / A-law -64 kbps)	8000	No
G.726 (5 bit 40 kbps/4 bit 32 kbps/ 3 bit 24 kbps/2 bit 16 kbps)	8000	No
GSM 6.10 FR (13.2 kbps) GSM HR (5.6 Kbps)	8000	No
GSM EFR (12.2kbps, packet time fixed at 20 ms)	8000	No
G729 (8 kbps)	8000	No
G729B (8 kbps)	8000	Yes
AMR (4.75kbps, 5.15kbps, 5.9kbps, 6.7kbps, 7.4kbps, 7.95kbps, 10.2kbps, 12.2 kbps)	8000	Yes
EVRC (Rates - 1/8, ½ and 1), EVRC0	8000	No
SMV (Modes – 0, 1, 2 and 3)	8000	No
ILBC (for 20msec)	8000	No
ILBC_13_33 (for 30msec)	8000	No

Supported Codecs (Contd.)

Codec Name	Sampling Rate	VAD Support
SPEEX	8000	Yes
G722 / G722.1 (rates - 24 kbps and 32 kbps)	16000	No
SPEEX_WB	16000	Yes
G,711 (MuLAW_2 / ALAW_2)	8000	Yes
ISAC	16000	No
AMR_WB (optional codec)	16000	Yes
G726_40_VAD	8000	Yes
G726_32_VAD	8000	Yes
G726_24_VAD	8000	Yes
G726_16_VAD	8000	Yes
EVRCB, EVRCB0	8000	Yes
EVRC_C	16000	Yes

Create and Manage RTP Sessions



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Create RTP Sessions

- RTP session can be created in either of the following ways -
  - Manually
  - Using 'Auto scan for Incoming Session' feature

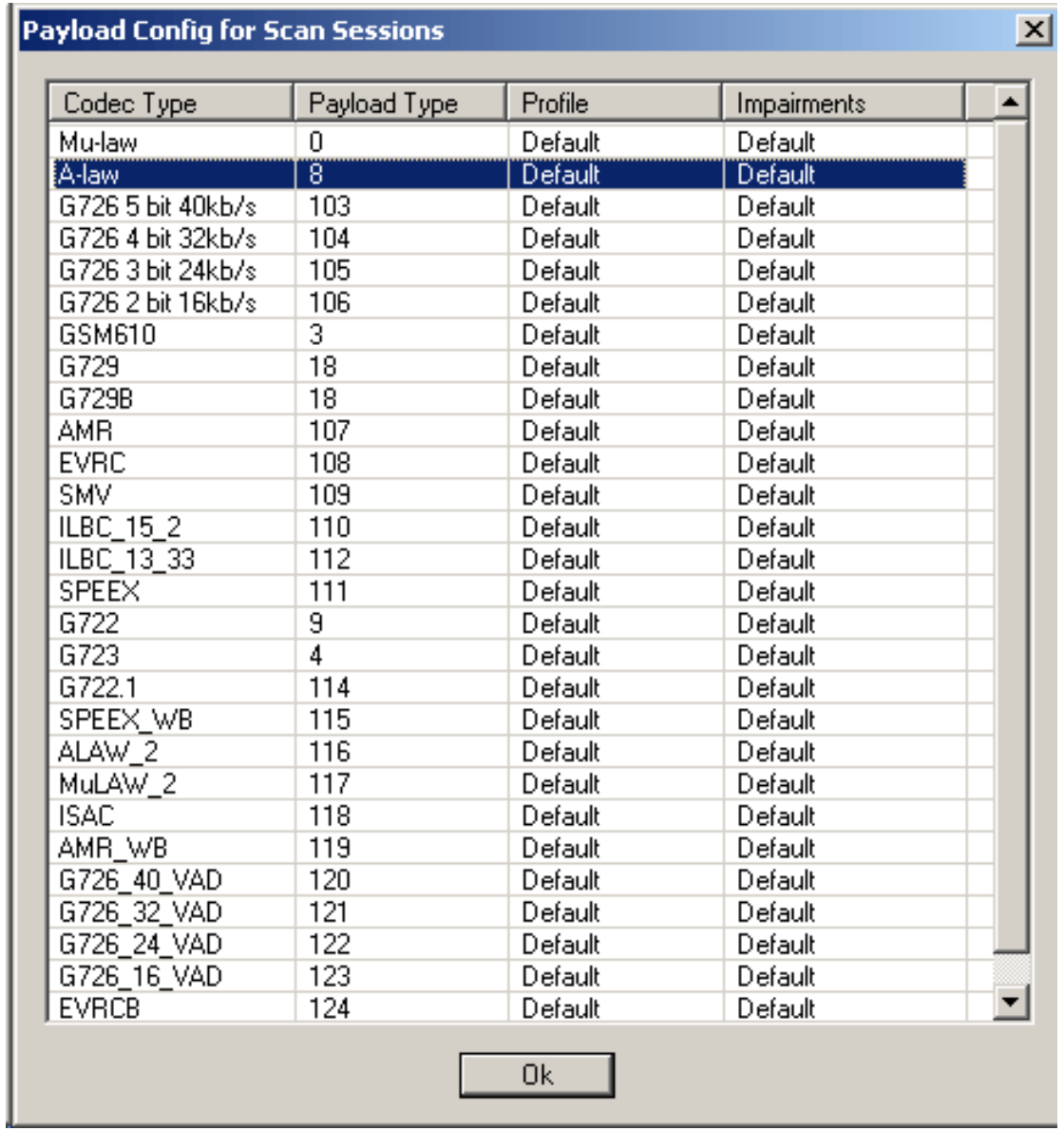
- Auto Scan Feature –
  - Monitors all incoming packets addressed to the machine on which RTP ToolBox™ is running
  - Automatically displays the RTP sessions on which the RTP packets are being transmitted if there are any RTP packets in the traffic

Manage RTP Sessions
Transmit and Receive Settings

- Configure the properties for sending / receiving RTP Traffic
- On transmitting session –
  - Type of codec needed, sampling rate, voice payload type, RFC 2833 payload type, comfort noise payload type, packetization time, SSRC, timestamp, and sequence number for the out going Traffic can be set
  - 'Quality Of Service (QoS)', i.e., IP Type of Service properties such as precedence, delay, throughput, and reliability values can be assigned to the stream
- On receiving session –
  - Specify a desired voice payload type for each codec, payload type to receive RFC 2833 events, and Comfort Noise Payload Type
  - Set the buffer used for delayed packets that arrive at receiving end (Supports both static and dynamic jitter buffers)

Payload Configuration

- Configures and saves Payload type, Tx Rx profile and Impairment's settings for the scanned sessions



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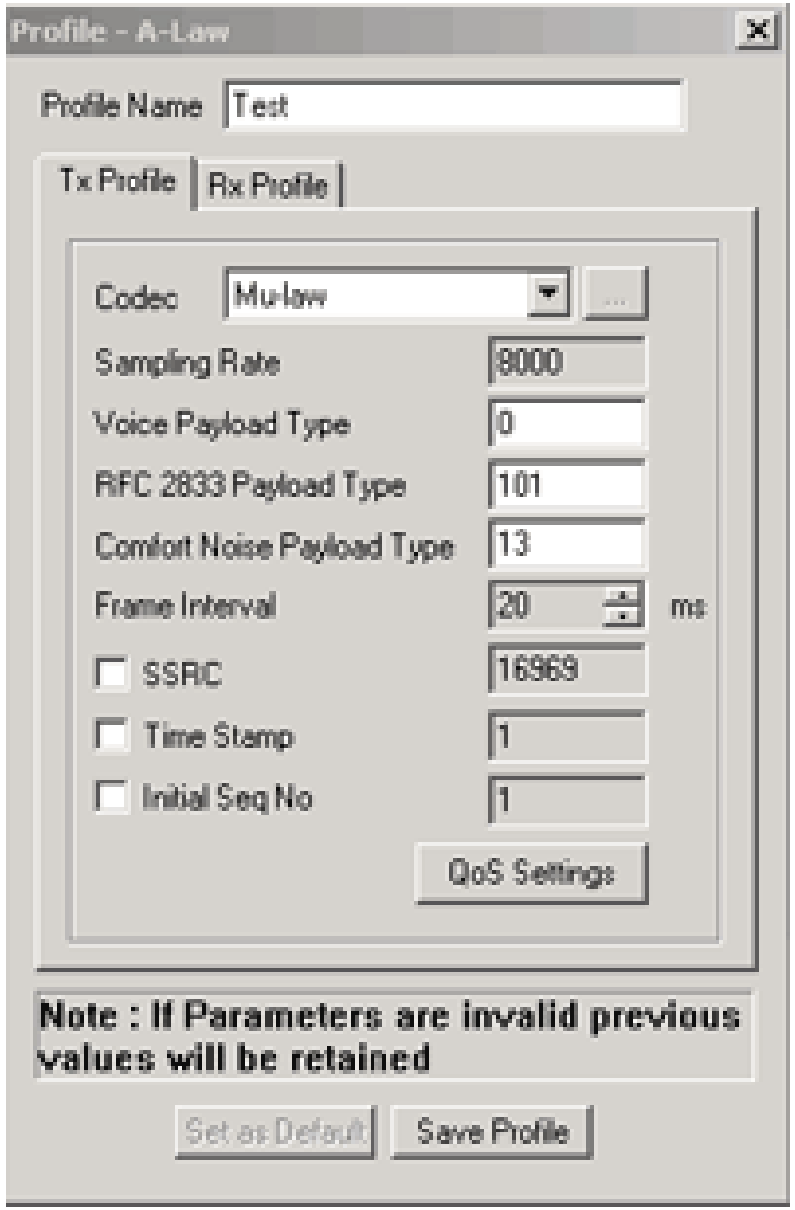
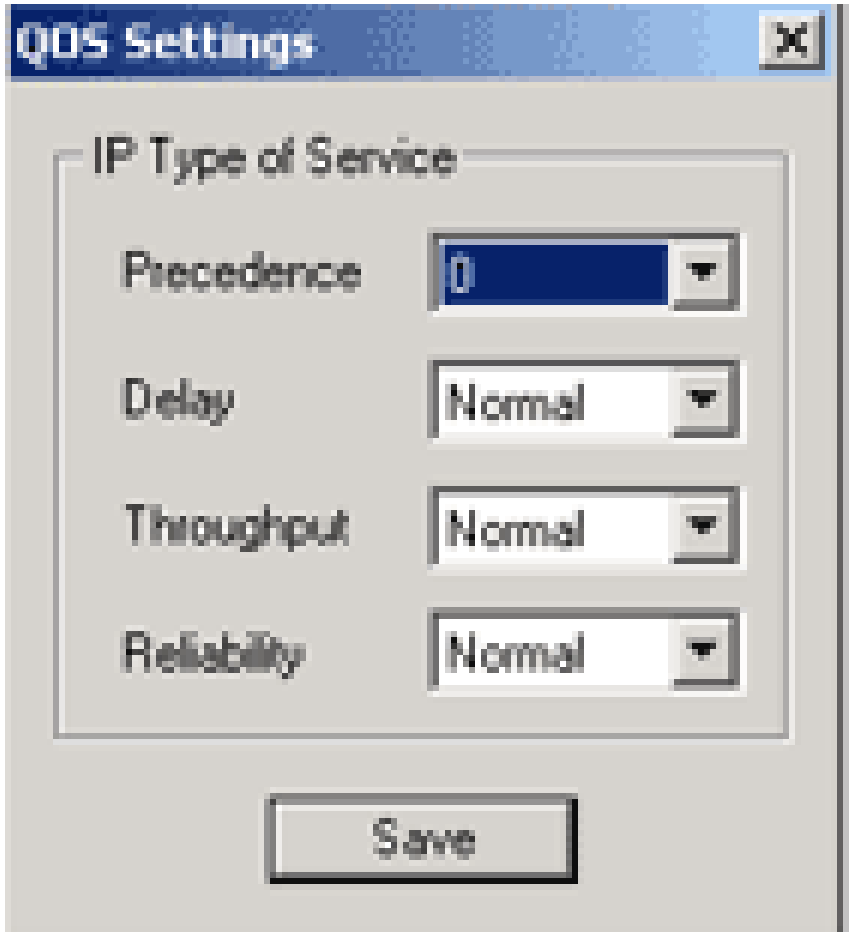
Transmit Settings



QOS Settings



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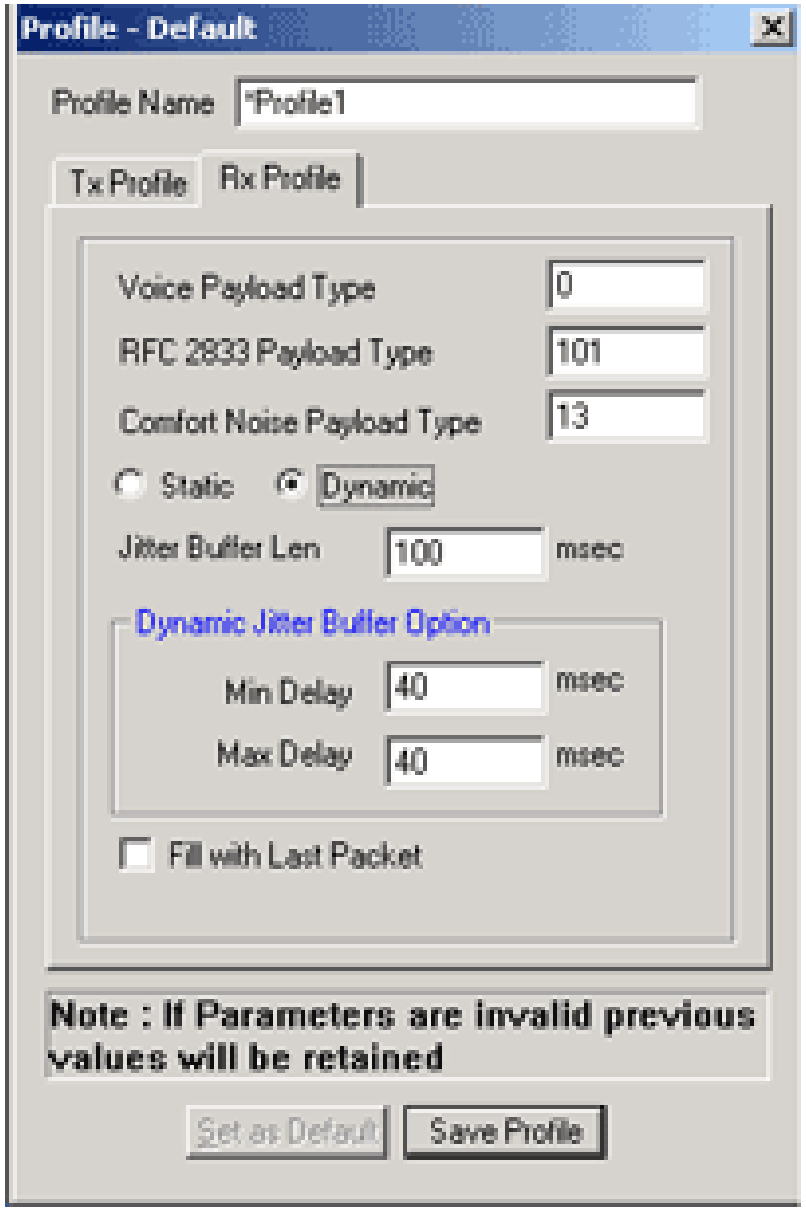
Receive Settings



RTP Traffic – Generate, Impair, Detect

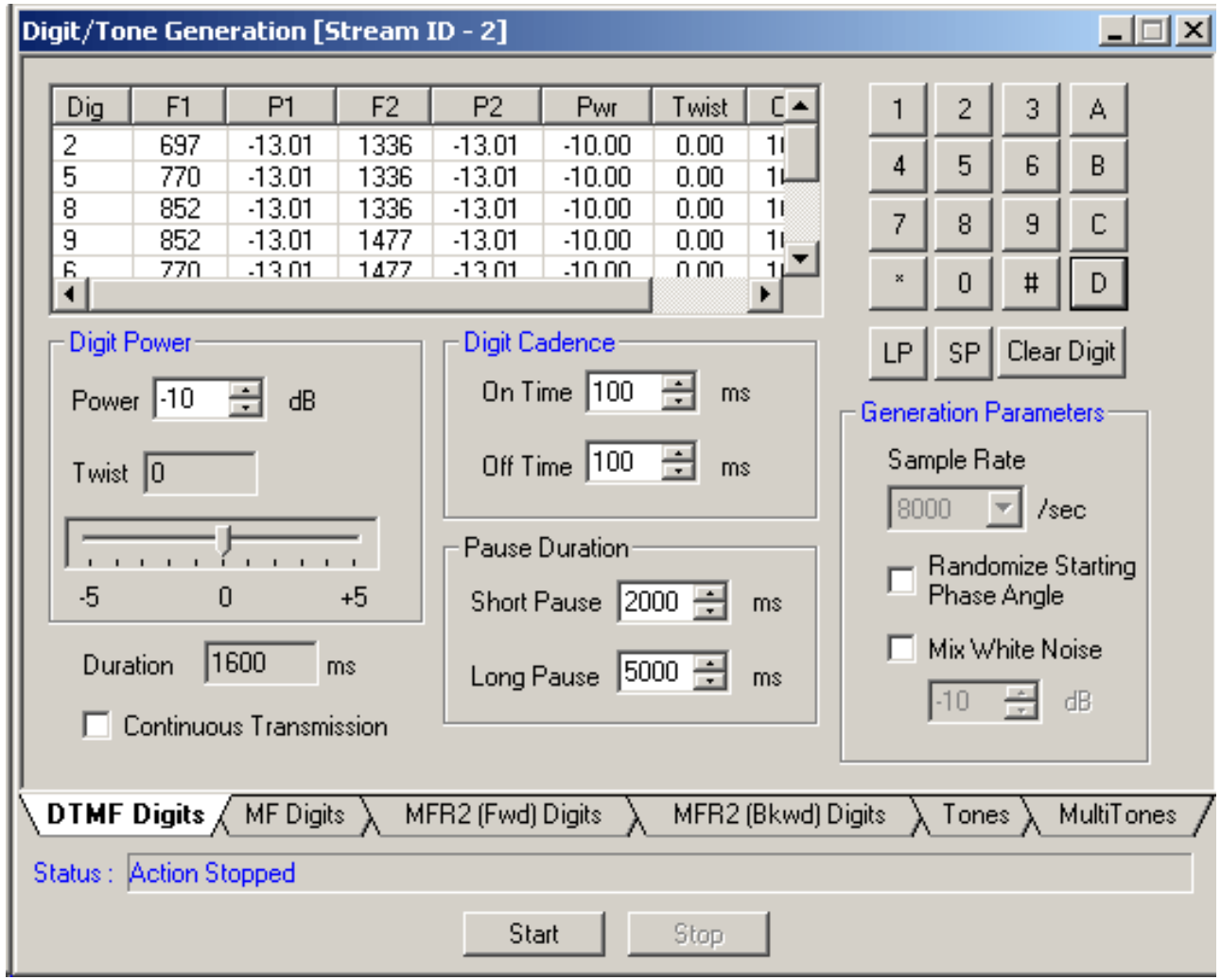
RTP ToolBox™ includes the following RTP generation and detection features –

- Digit / tone generation
- Detection of digits and tones
- Record data to file
- Playback from file
- RTP Event Generation
- Capture RTP Events
- Play to Speaker
- Talk Using Microphone
- Synchronous Tx/Rx Utility
- Loopback
- Delay And Attenuate
- Impairments

Digit / Tone Generation

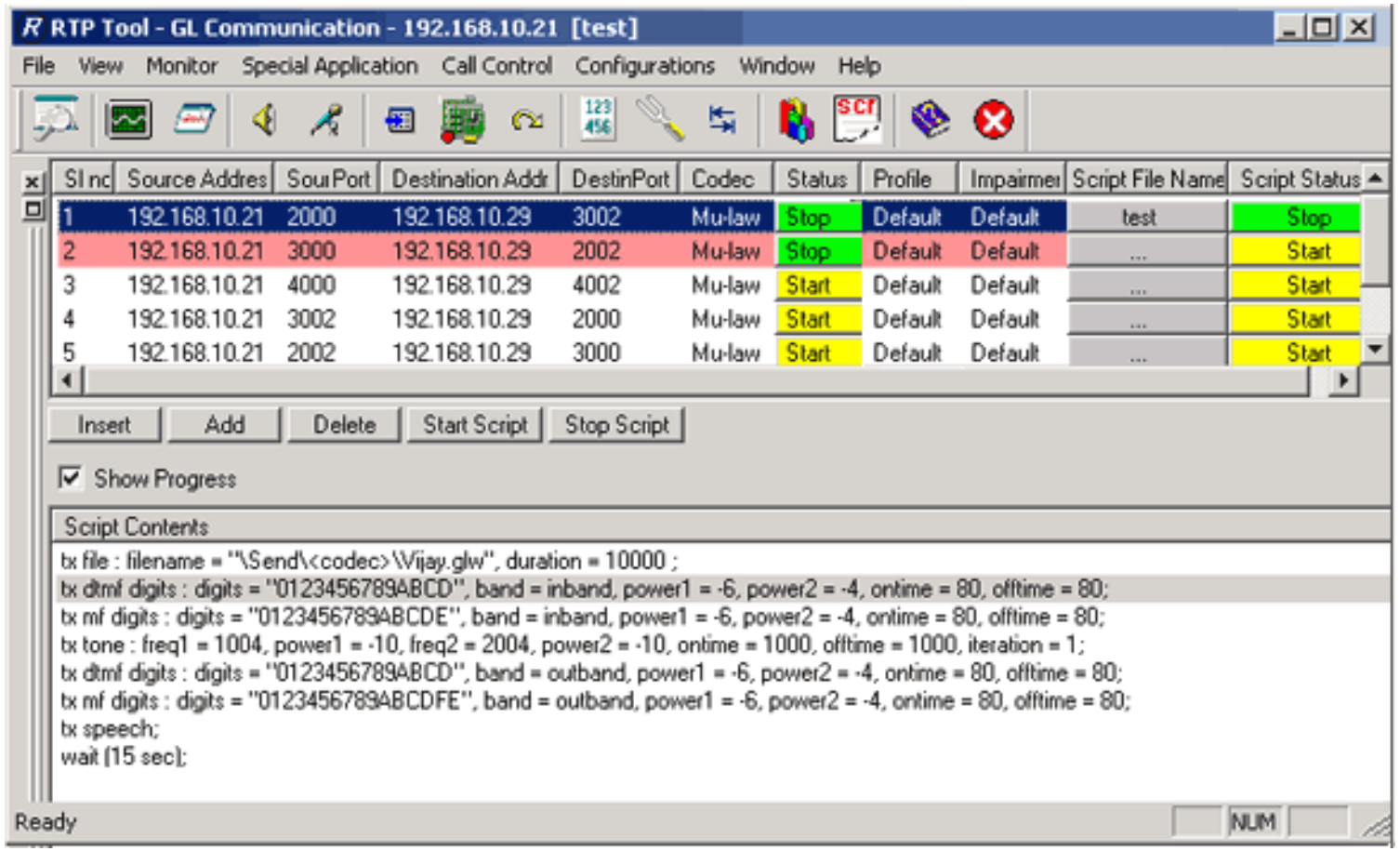
Generation and Detection of RTP Traffic

- Generates in-band digits and tones (continuous or at once)
- Supported tones include single, dual, and multi-tones
- Supported digits include DTMF, MF, and MFR2 forward and backward digits



Transmission of Digits and Tones

Generation and Detection of RTP Traffic



- Generation / Detection of in-band and out-of-band digits / tones (DTMF, MF, user-defined, etc.) / events per RFC-2833

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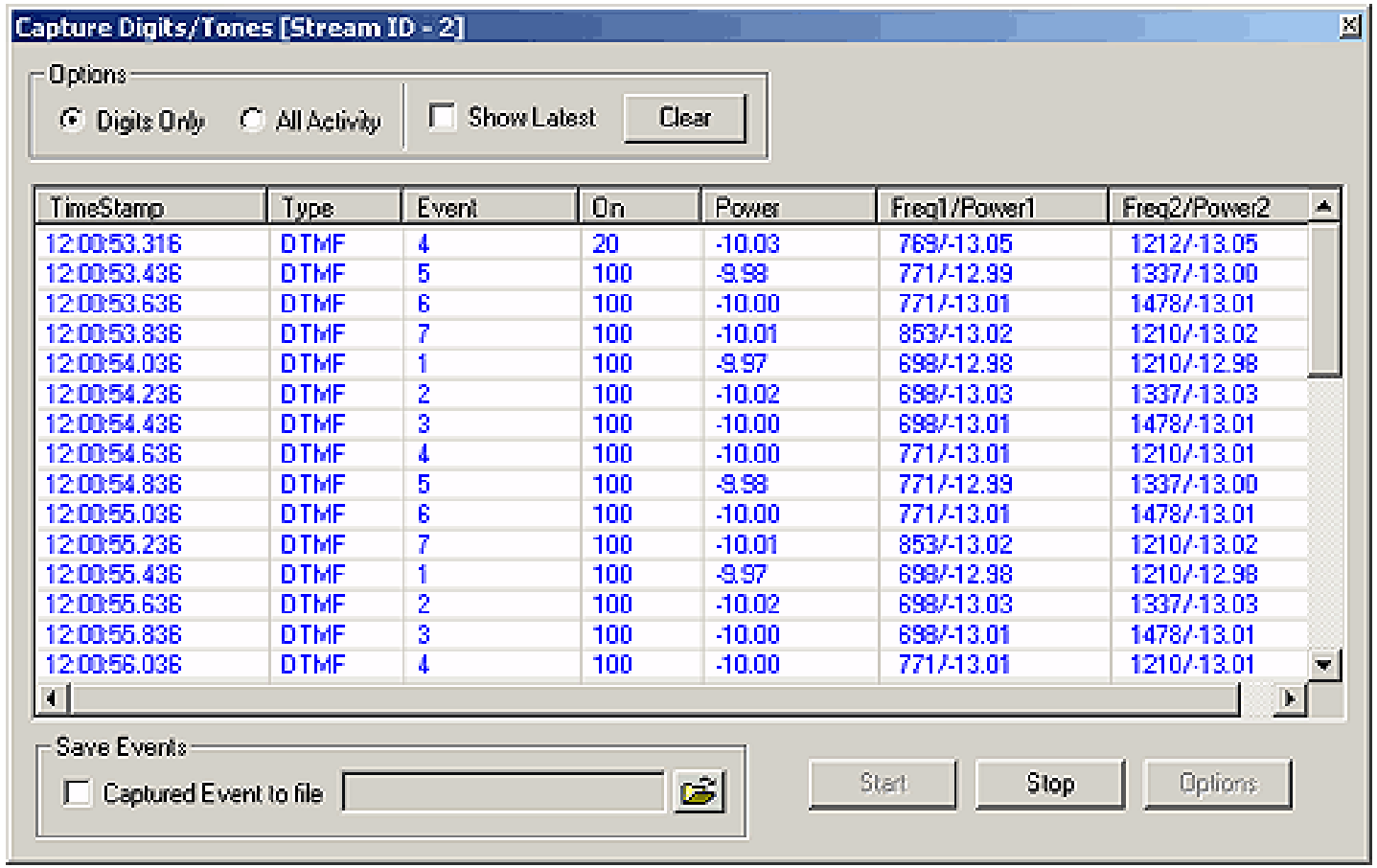
Detection of Digits and Tones

Generation and Detection of RTP Traffic

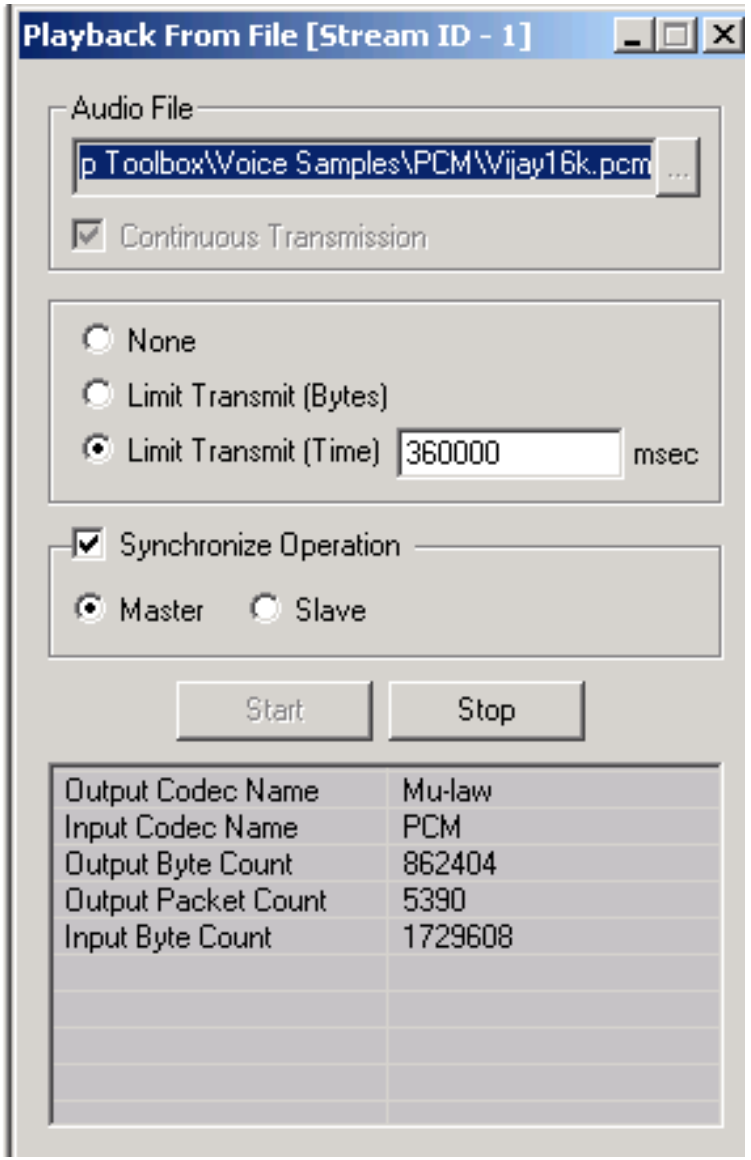
- Captures tones and digits in the traffic
- Displays additional information such as type of the signal, timestamp, event, power, and accept/ reject frequencies
- Completely supported for both in-band digits/tones and RTP digits/events per RFC-2833



Transmit / Record Voice File

Generation and Detection of RTP Traffic

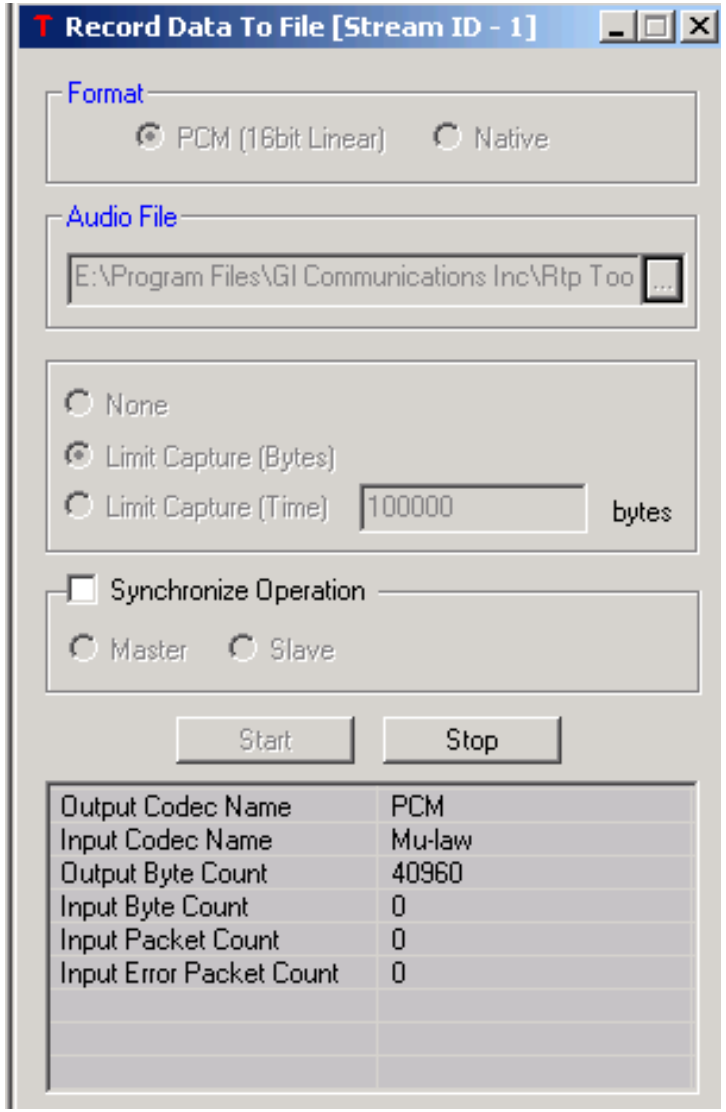
- PCM or Wave files can be sent over an established session using playback feature
- Can transmit a file on more than one session simultaneously



Transmit / Record Voice File

Generation and Detection of RTP Traffic

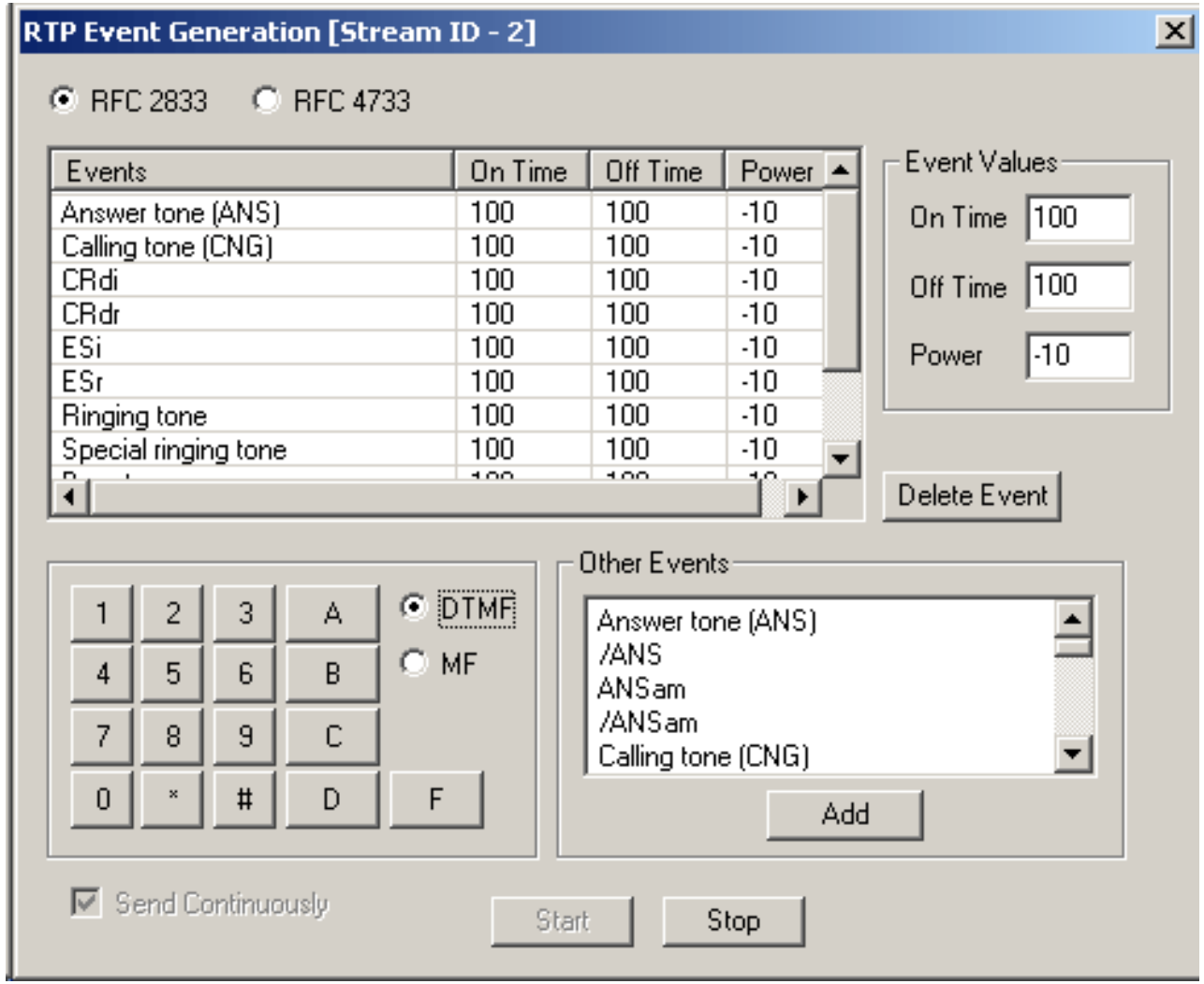
- Records the incoming voice data to file
- The recorded files can be compared using GL's Voice Quality Testing software, providing PESQ score



RTP Event Generation

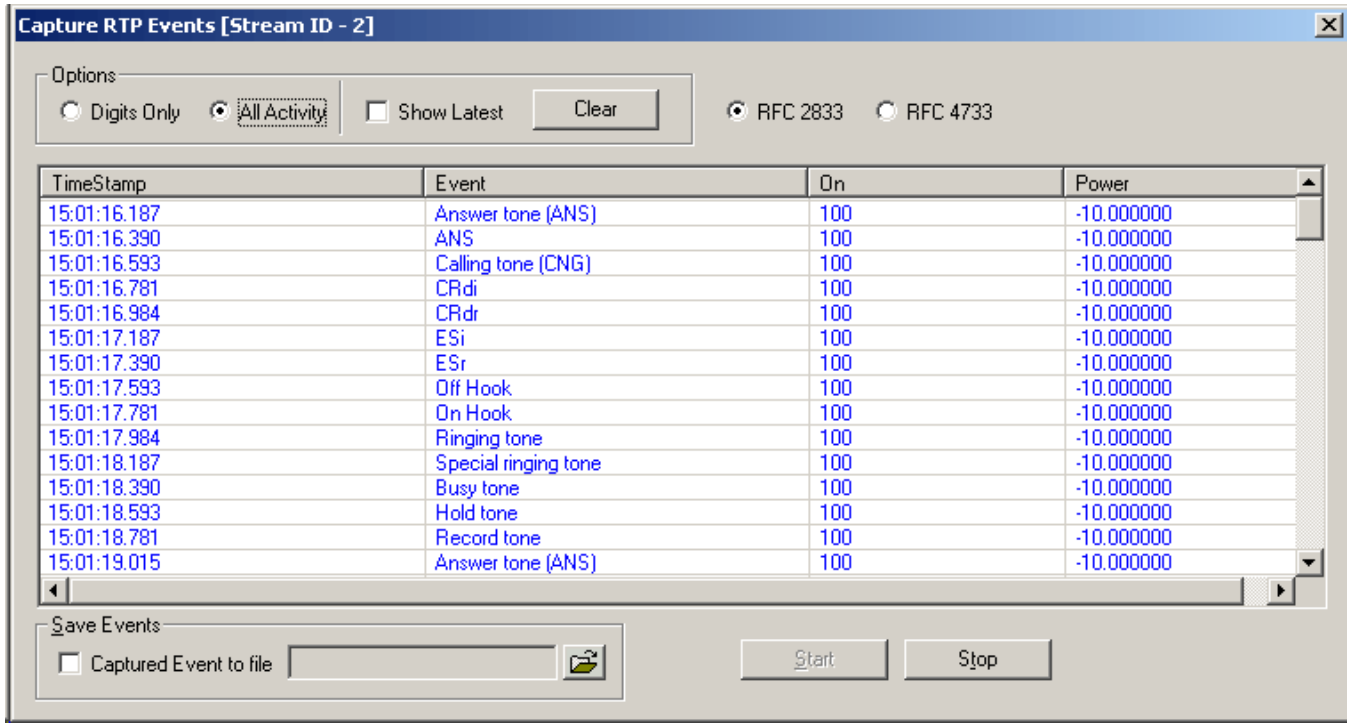
Generation and Detection of RTP Traffic

- Allows to transmit out of band digits on a session
- Supports 2 types of RFC events – 2833 and 4733



Capture RTP Events

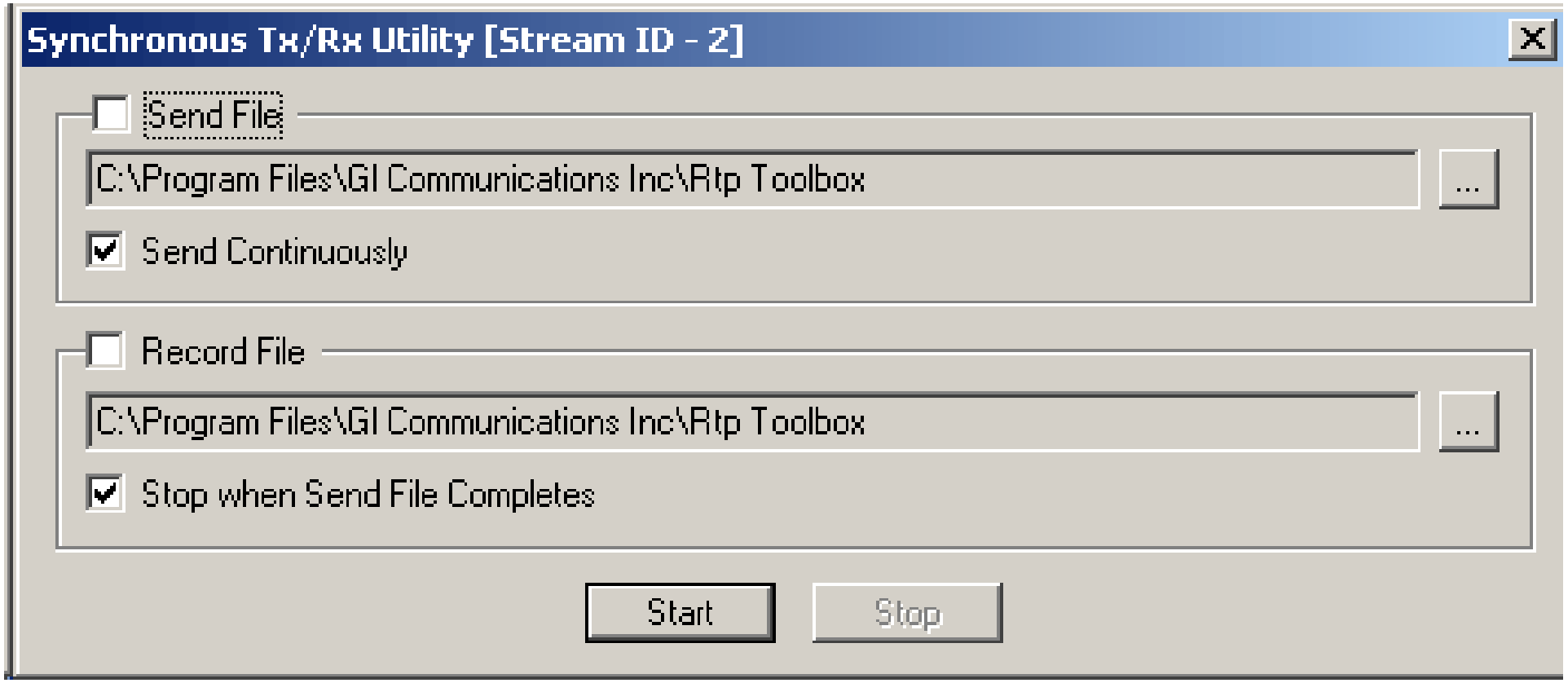
Generation and Detection of RTP Traffic



- Allows to detect out of band digits
- Supports two types of RFC events – 2833 and 4733
- Provides an option to view only digits being captured or to view all the events being captured

Synchronous Tx / Rx Utility and Loopback

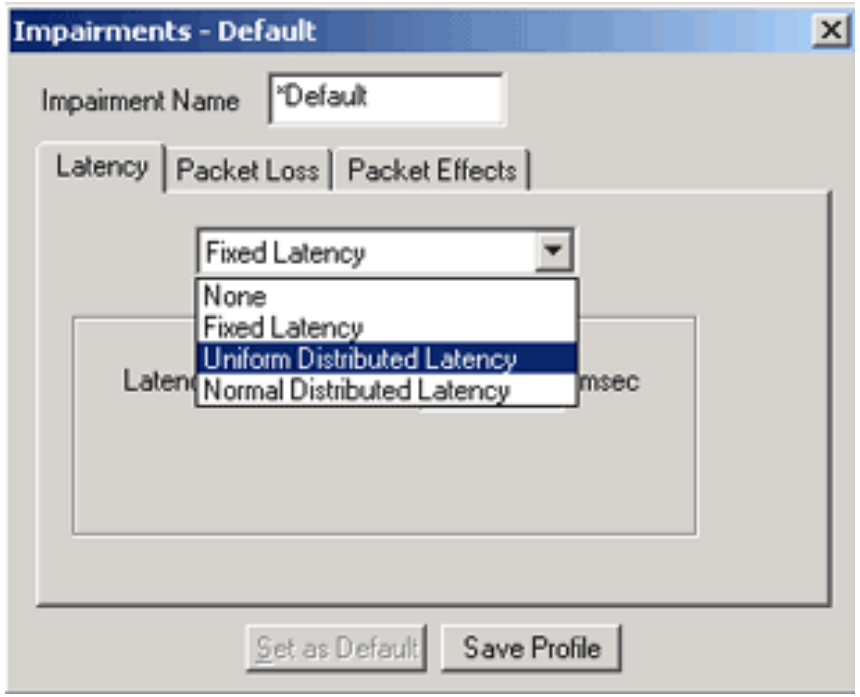
Generation and Detection of RTP Traffic

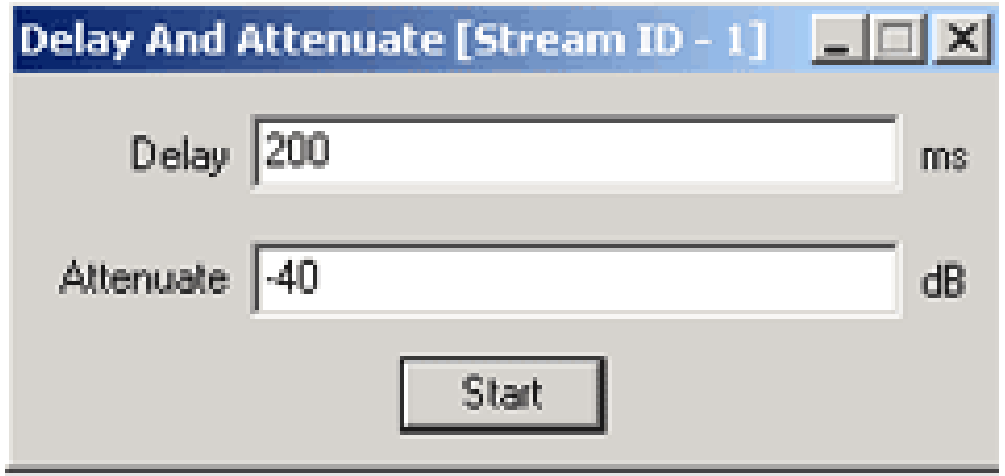


- Transmit and receives a file simultaneously on a session
- Loopback feature allows the user to loop back the incoming data on a session

Impairments

- Can introduce impairments manually and transmit on the RTP sessions
- Includes fixed latency, uniform/normal distributed latency, periodic/random/burst packet loss, out-of-order packets, and duplicate packets
- Delay value ranges from 0 to 2000ms; attenuate value ranges from -40 to 20dBm



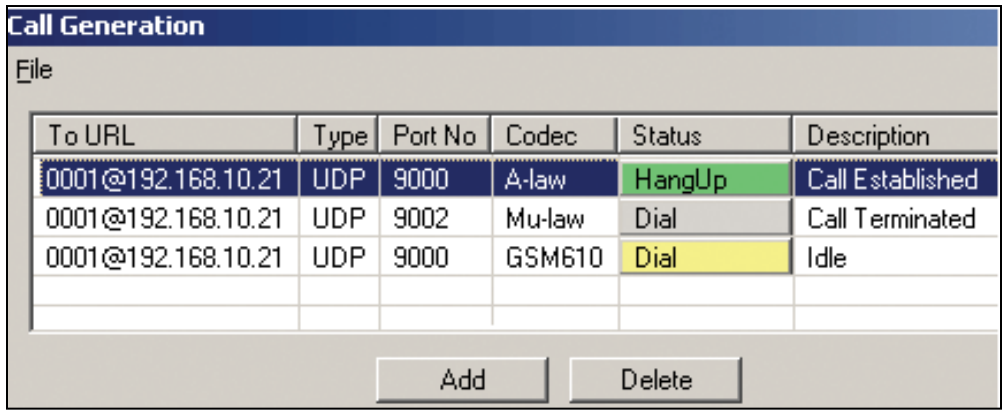


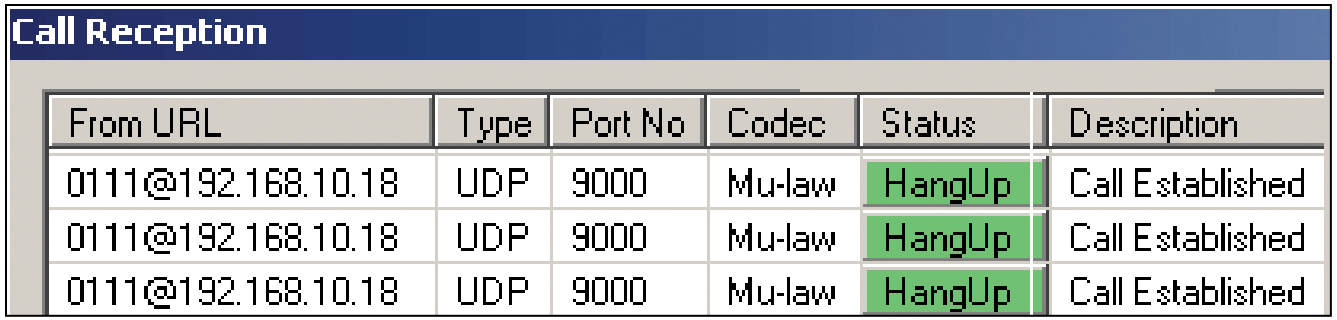
Sip Call Generation and Reception

- Users can configure and simulate a user agent (UA) for manual SIP call generation and reception using public URL and contact IP addresses
- Up to 8 User Agents can be configured using the CLI.
- Multiple SIP calls can be placed and received through a single user agent
- Supports transport type as either UDP or TCP
- Customize Codec options (payload type, ptime) for UA during Call Generation (Dial) & Call Reception
- All the calls at the application end will be answered automatically

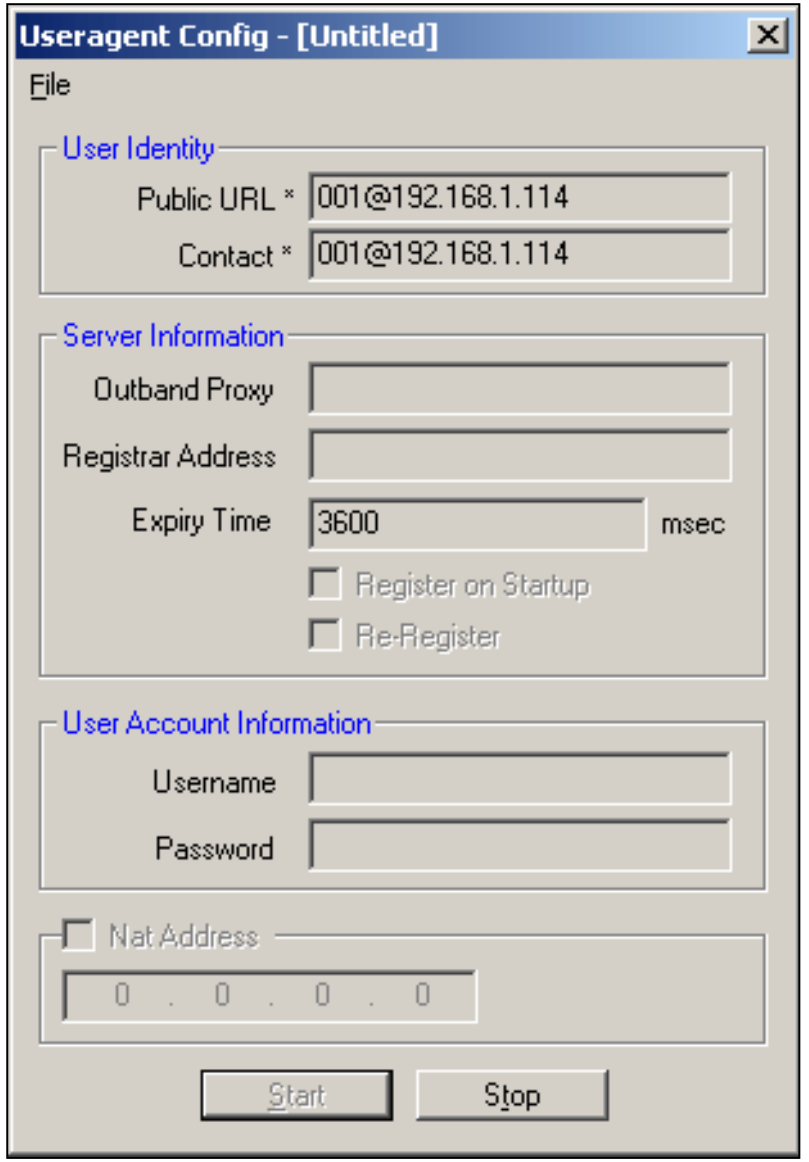
Sip Call Generation and Reception (Contd.)

- User Agent configuration requires Public URL and Contact field to be entered mandatorily.
- Other optional parameters include - Outbound Proxy, Registrar Address, NAT Address, Username and Password





User Agent Configuration



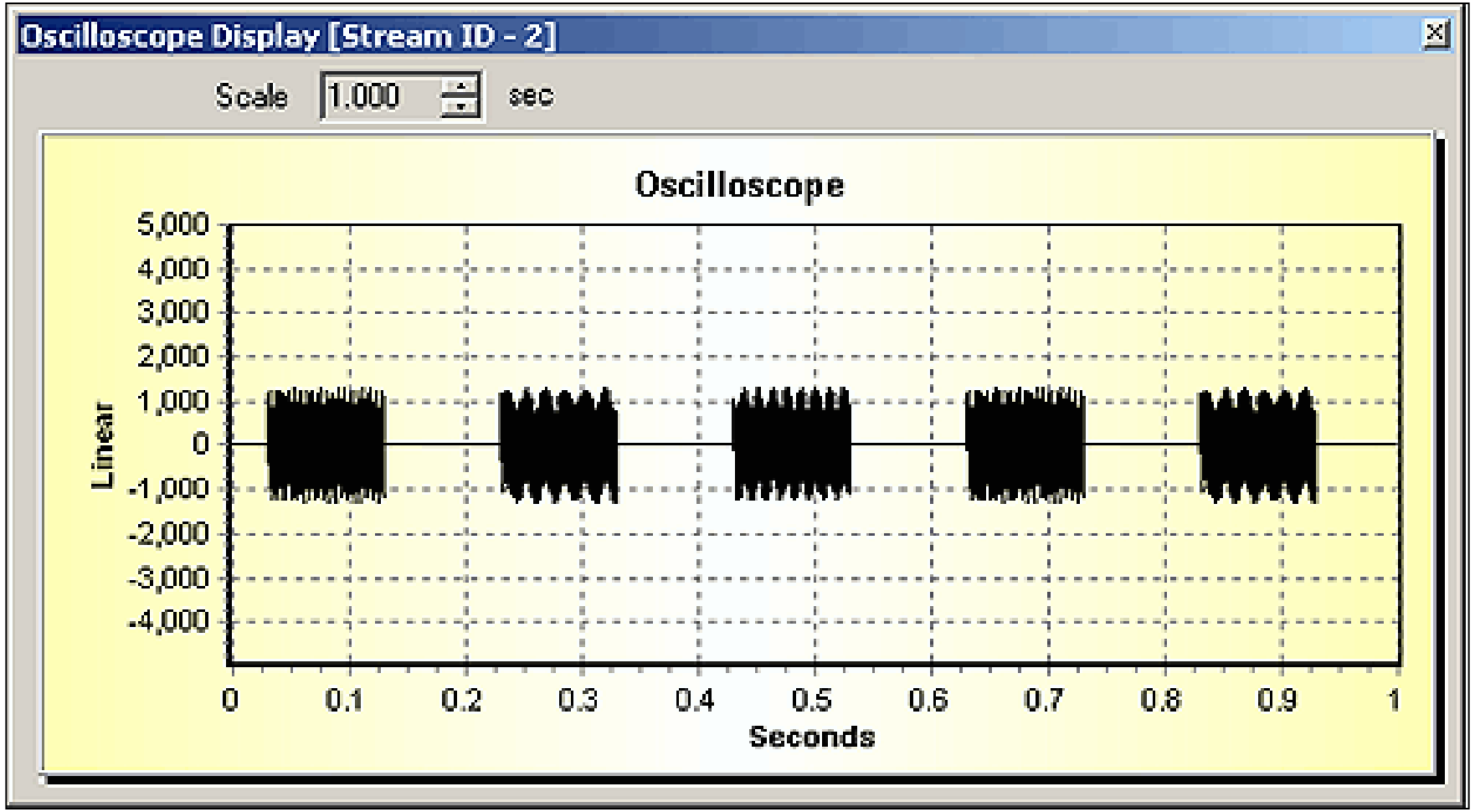
Analysis Tools

The RTP ToolBox™ has the following analysis tools –

- Graphical Analysis
  - Oscilloscope
  - Spectral Display
  - G.168 Waveform Viewer
- Statistics
  - RTP / RTCP Statistics
  - R Factor - Quality Metrics (R-Factor & MOS Factor), Degradation Factor, Burst Metrics, Delay Metrics, & Jitter Buffer Statistics

Oscilloscope

- The PCM codes (amplitude of the incoming signal) for any selected session are graphically displayed in real-time as a function of time



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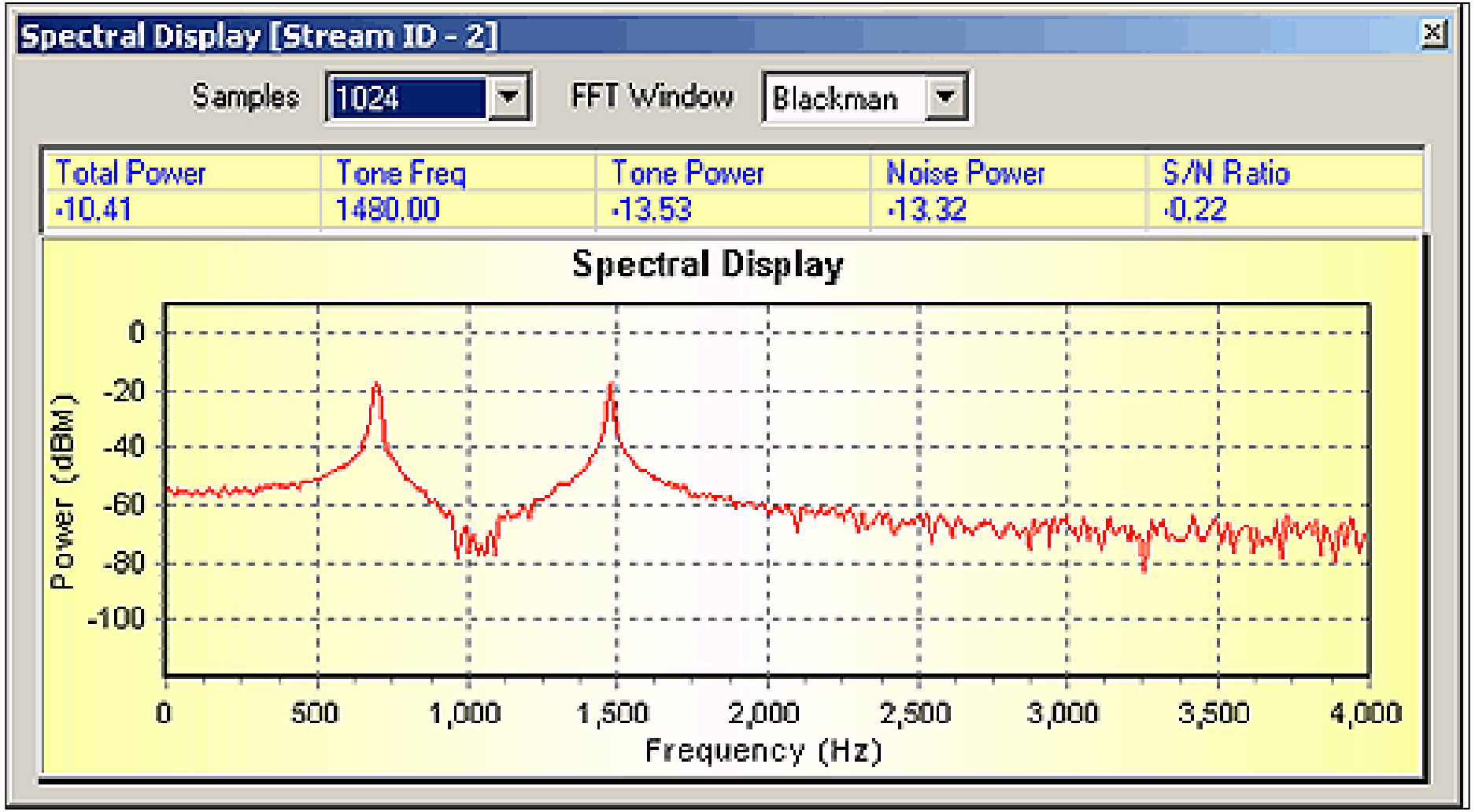
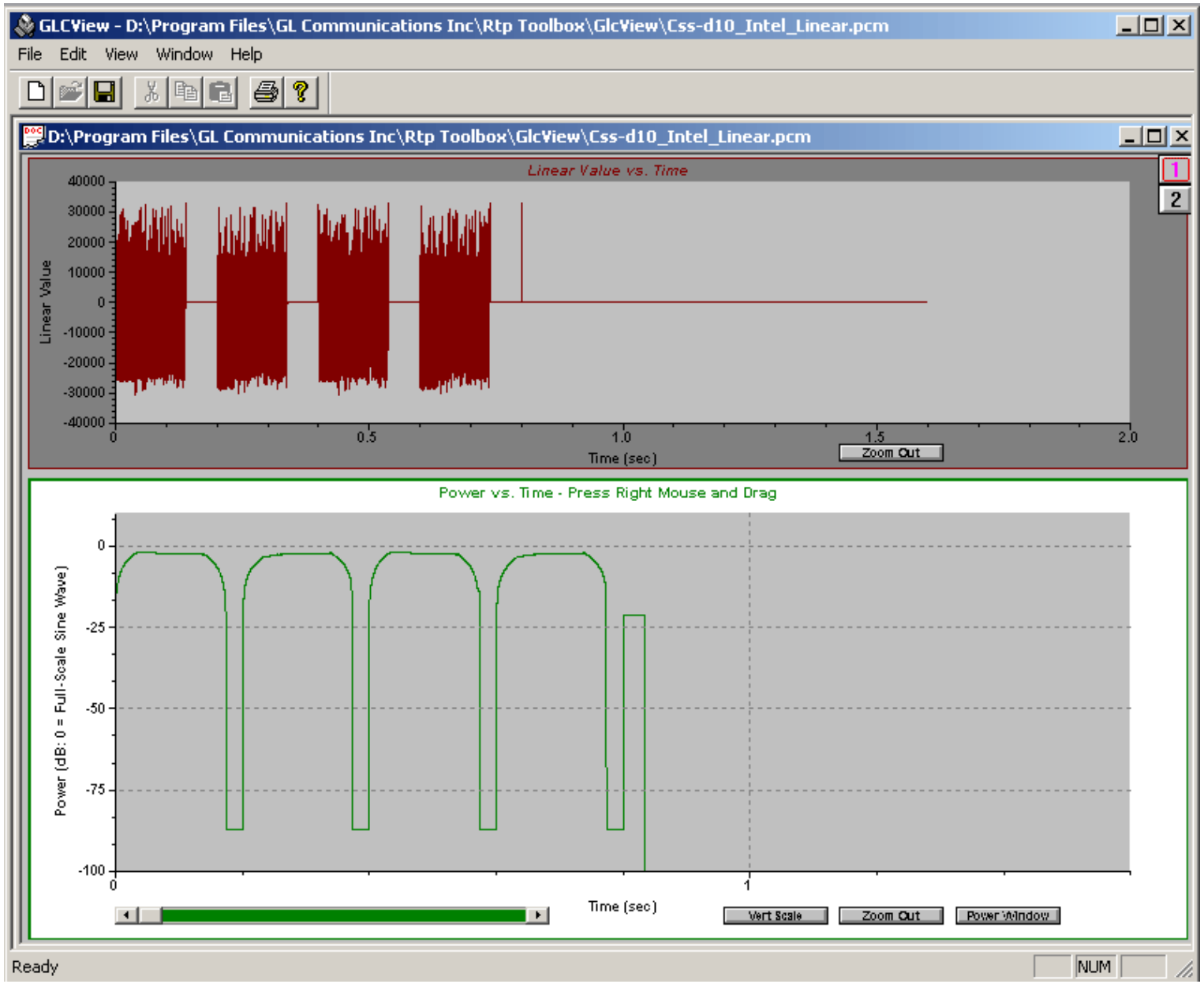
Spectral Display

- Displays data received in spectral domain (Spectral Amplitude vs. Frequency)

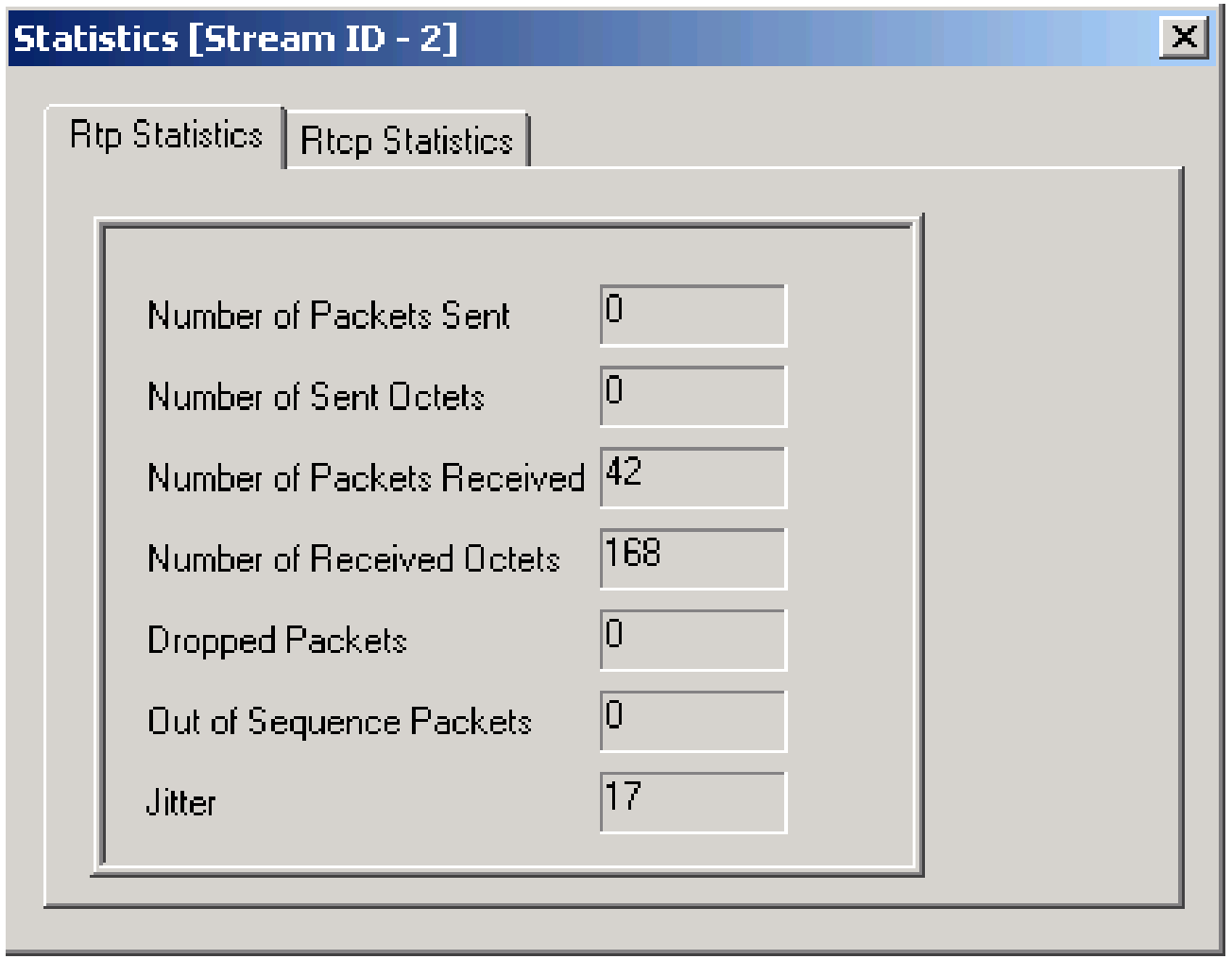


G.168 Waveform Viewer



RTP / RTCP Statistics

- Statistics reports of RTP and RTCP packets transmitted on a session
- RTP Statistics includes Number of Packets Sent, Packets Received, Dropped Packets, Out of Sequence Packets, and Jittera
- Sender Reports and Receiver Reports are displayed using RTCP statistics



Estimating Speech Quality of Packets

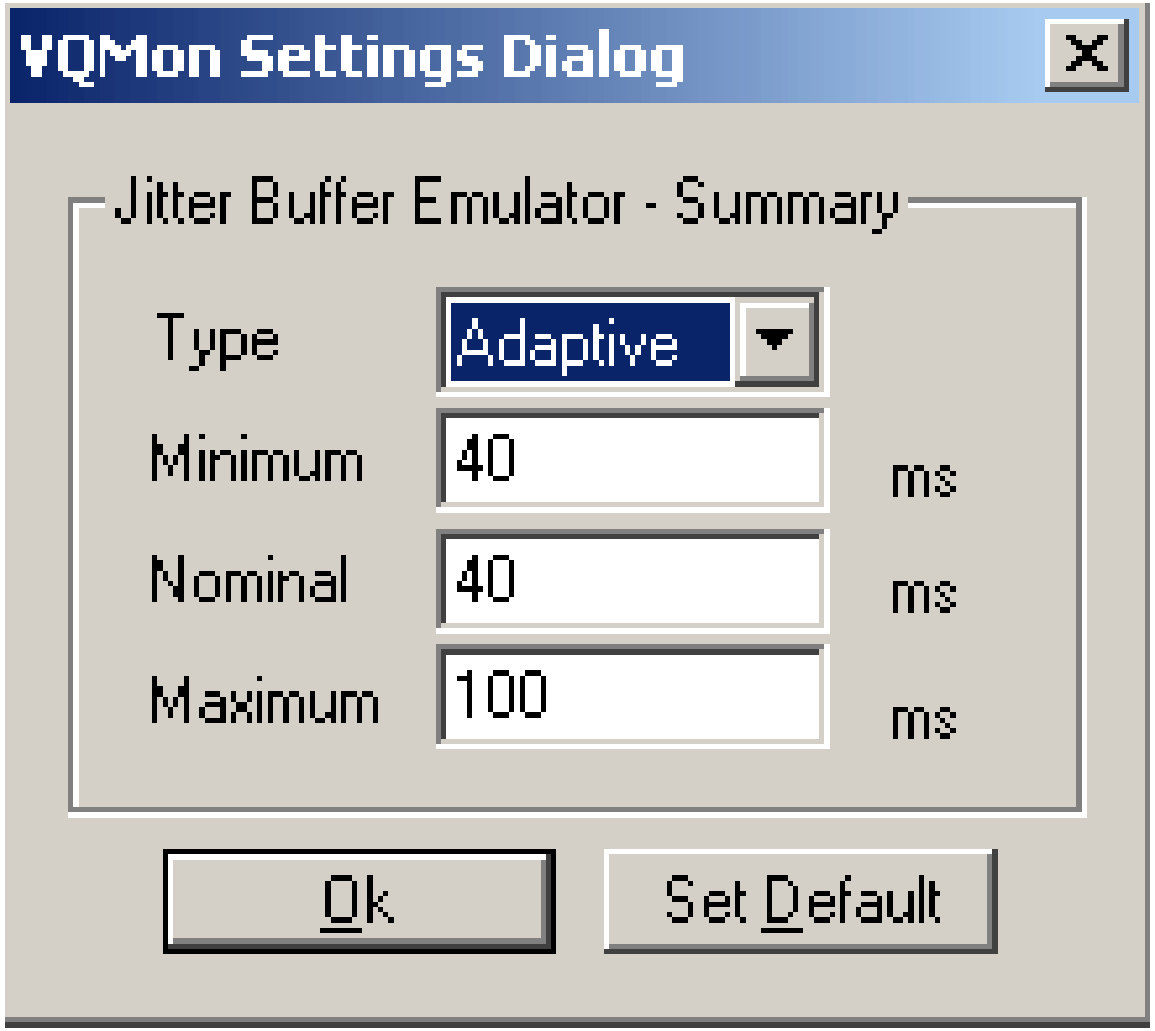
Codec Name	MOS-LQ	MOS-CQ	MOS-PQ	R-LQ	R-CQ	VQmon-Nom MOS	VQmon-Nom R-factor
G.711 μ-law	4.2	4.18	4.45	93	92	4.2	93
G.711 A-law	4.2	4.18	4.45	93	92	4.2	93
G.722	3.91	3.91		96	95	3.91	96
G.722.1 (32 K)	4.04	4.01		100	99	4.09	102
G.722.1 (24 K)	3.91	3.91		96	95	3.98	98
G.729A/G.729AB	3.91	3.88	3.8	82	81	3.91	82
GSM-FR	3.57	3.53	3.63	73	72	3.57	73
G.726-40k	4.16	4.14	4.13	91	90	4.16	91
G.726-32k	4.04	4.01	3.89	86	85	4.04	86
G.726-24k	3.35	3.3	3.52	68	67	3.35	68
G.726-16k	2.82	2.77	3.2	57	56	2.82	57
G.726-40k with VAD	4.16	4.14	4.13	91	90	4.16	91
G.726-32k with VAD	4.04	4.01	3.89	86	85	4.04	86
G.726-24k with VAD	3.35	3.3	3.52	68	67	3.35	68
G.726-16k with VAD	2.82	2.77	3.2	57	56	2.82	57

Estimating Speech Quality of Packets (Contd.)

Codec Name	MOS-LQ	MOS-CQ	MOS-PQ	R-LQ	R-CQ	VQmon-Nom MOS	VQmon-Nom R-factor
AMR NB 7.95k	3.69	3.65	3.7	76	75	3.69	76
EVRCB	3.98	3.98		84	84	3.98	84
SMV	3.61	3.57		74	73	3.88	81
Speex WB	4.14	4.16		106	105	4.16	106
Speex NB	4.14	4.16		91	90	4.16	91
iLBC 13.3k	3.88	3.84	3.79	81	80	3.88	81
iLBC 15.2k	3.95	3.91	3.82	83	82	3.95	83

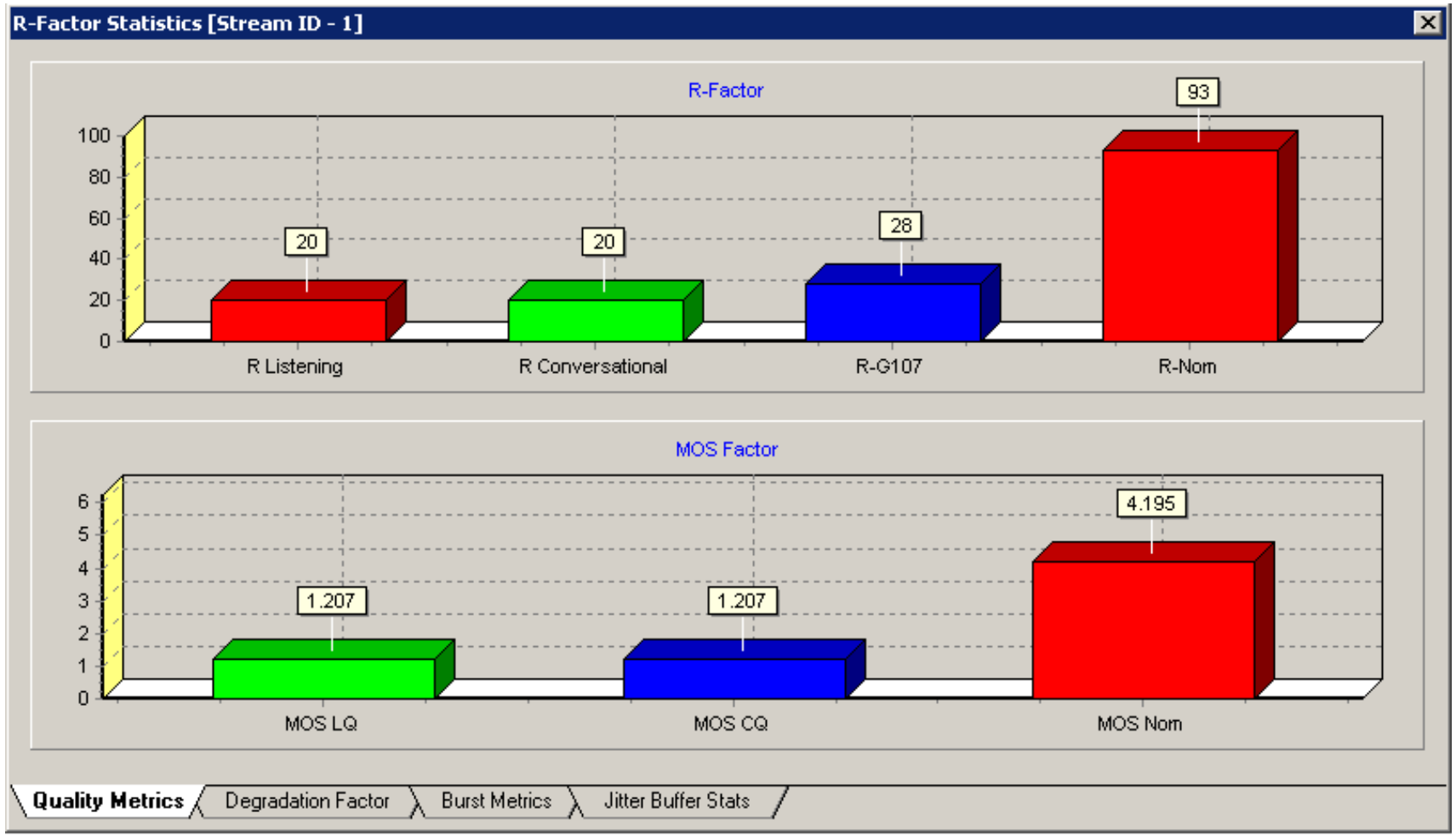
Jitter Buffer Emulator Settings

- Depending on the Jitter Buffer Emulator settings, the R-Factor and MOS are calculated



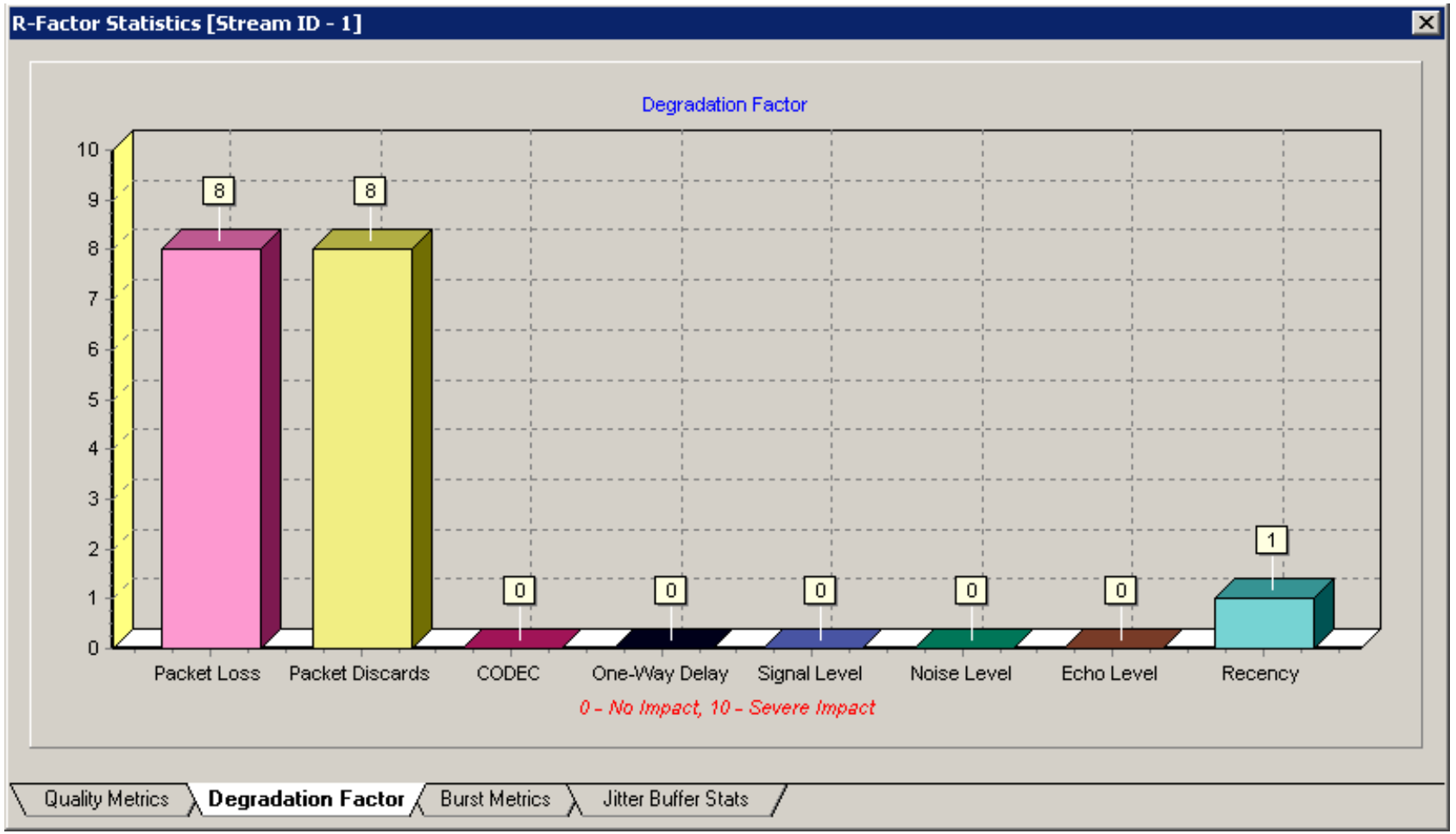
Quality Metrics

- Quality metrics displays for R-Factor and MOS Factor graphs
- R Factor graph – displays statistics such as, R-Listening, R-Conversational, R-G107 and R-Nom
- MOS Factor graph – displays statistics such as MOS CQ, MOS LQ and MOS Nom

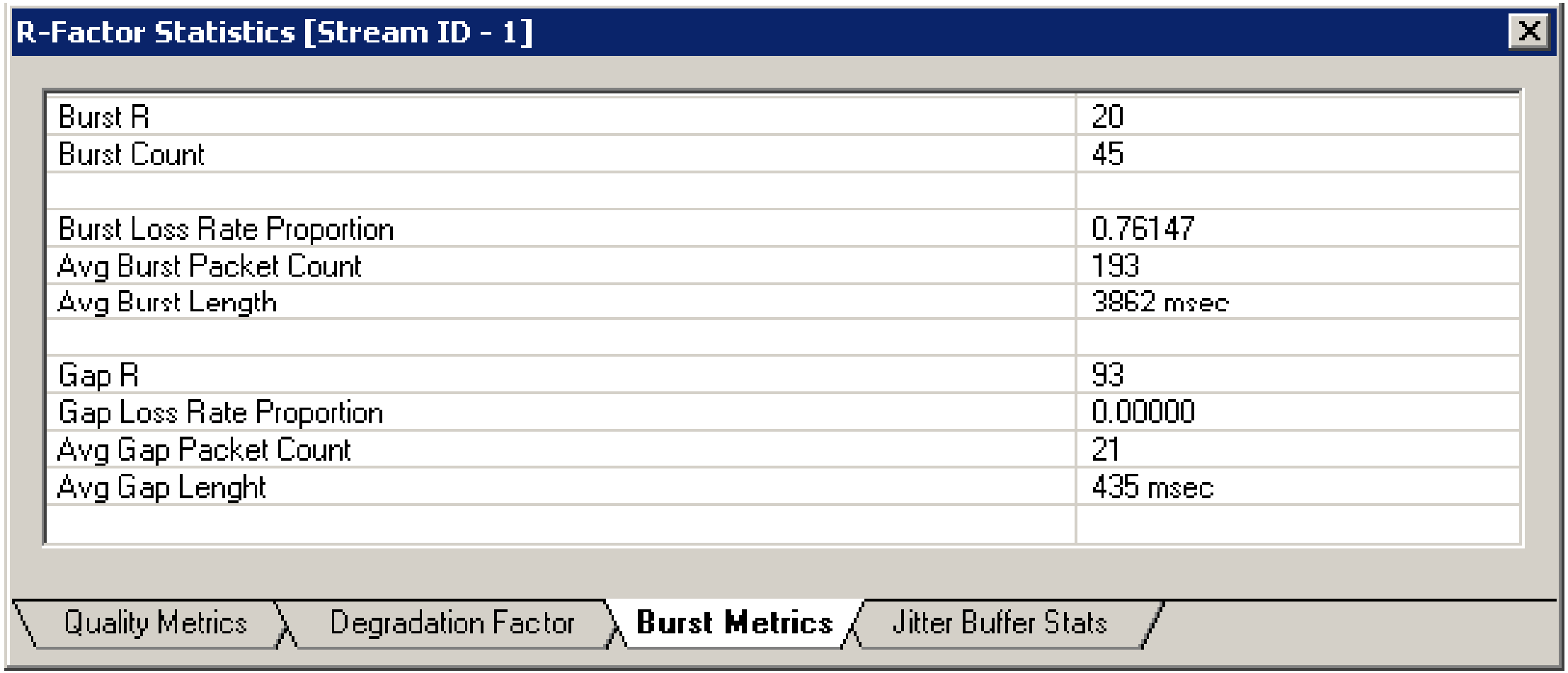


Degradation Factor

- Displays a bar graph to indicate different degradation factor statistics.
- For voice quality measurement, various degradation factors are included such as network packet loss (Packet Loss), jitter buffer packet discards (Packet Discards), voice CODEC selection (CODEC),
- one-way delay, voiced segment signal level (Signal Level), silence period noise level (Noise Level), echo level, and recency.

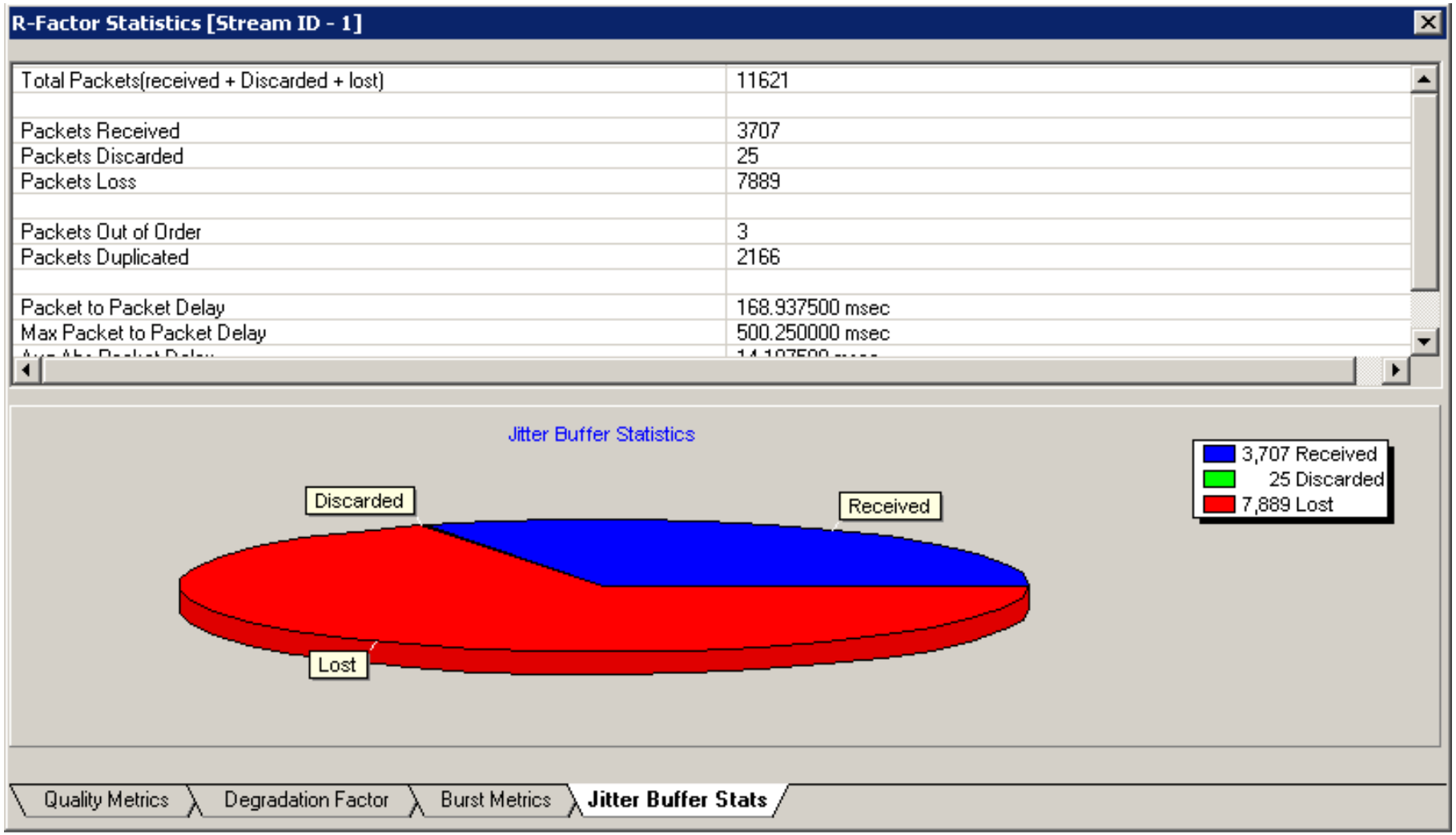


Burst Statistics



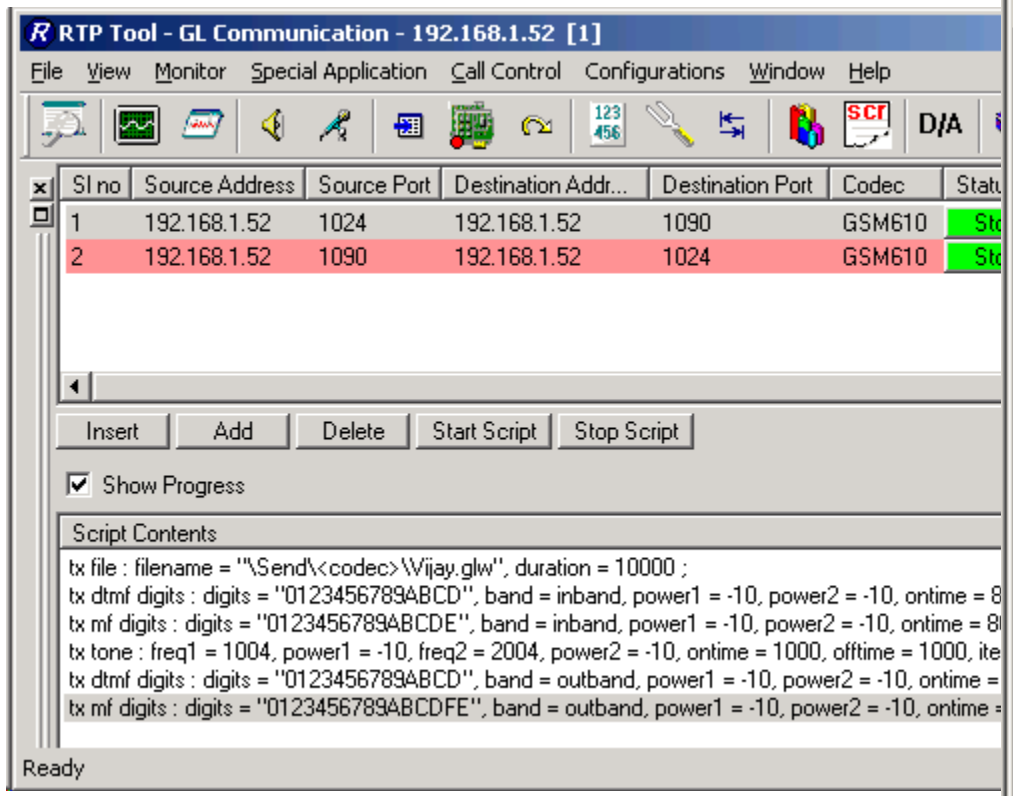
Jitter Buffer Statistics

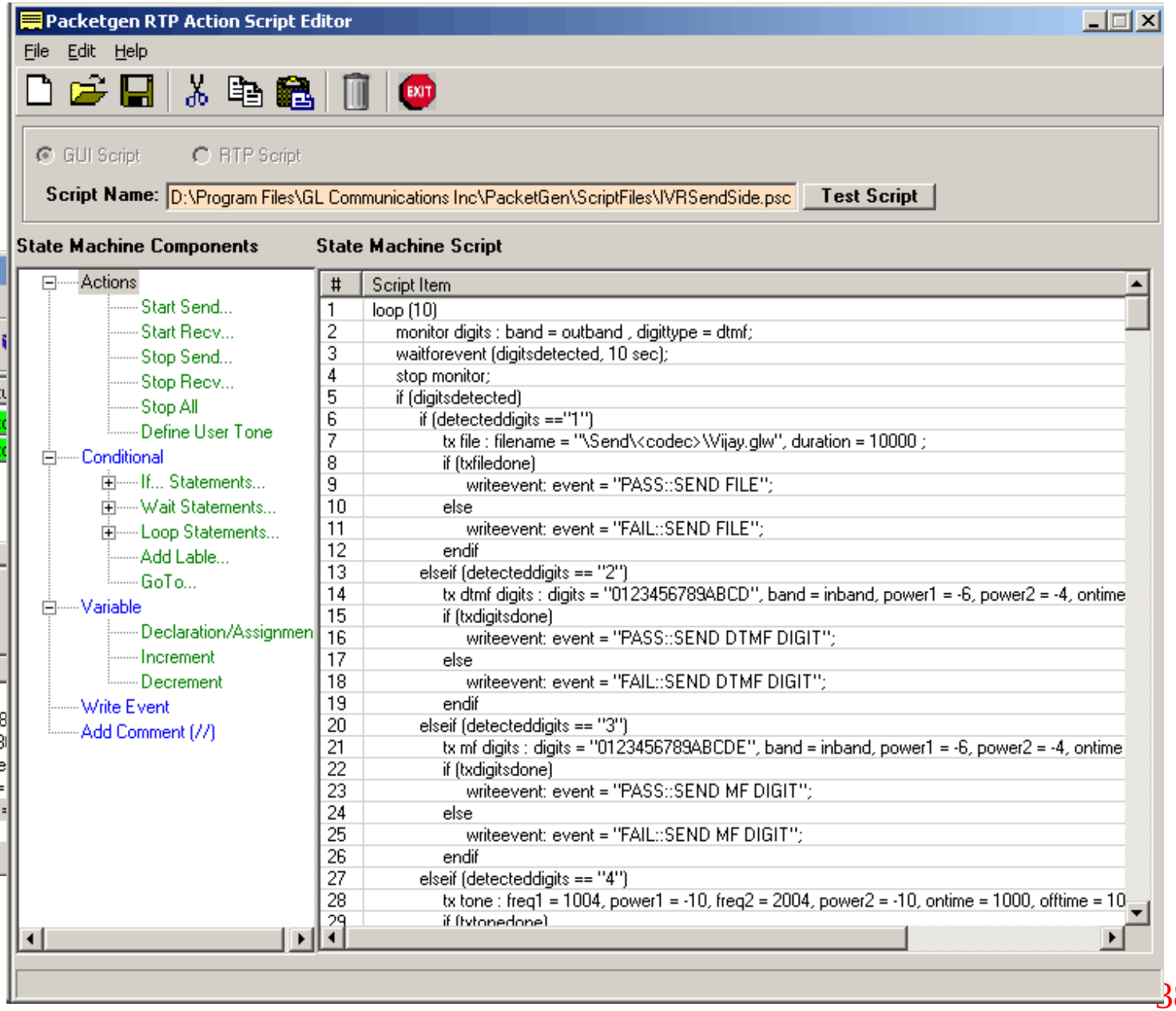
- Displays a pie chart indicating number of packets received, discarded, lost, out of order, duplicated, and more



Scripting Traffic Actions

- Simple user interface to create scripts
- Automated script control of traffic
- Conditional statements, stack multiple actions





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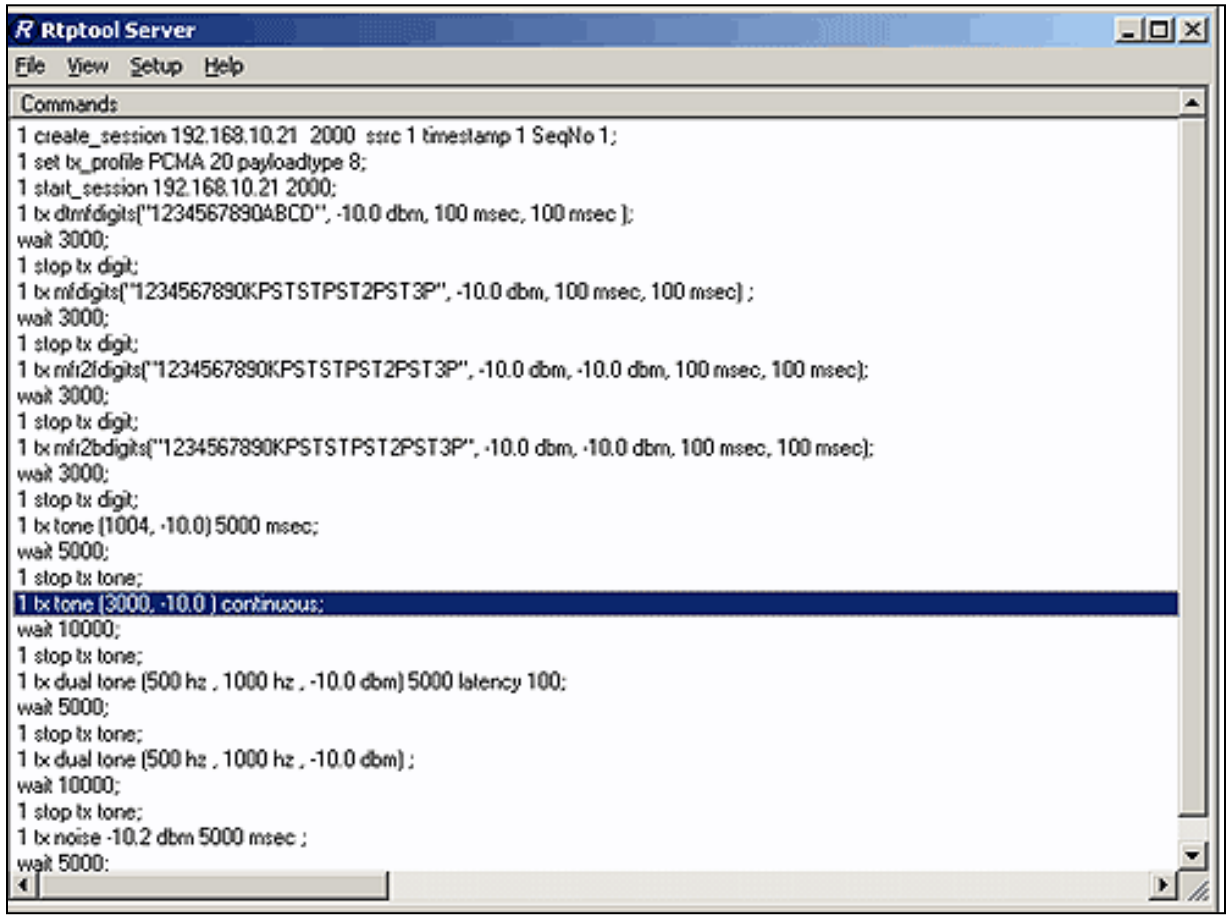
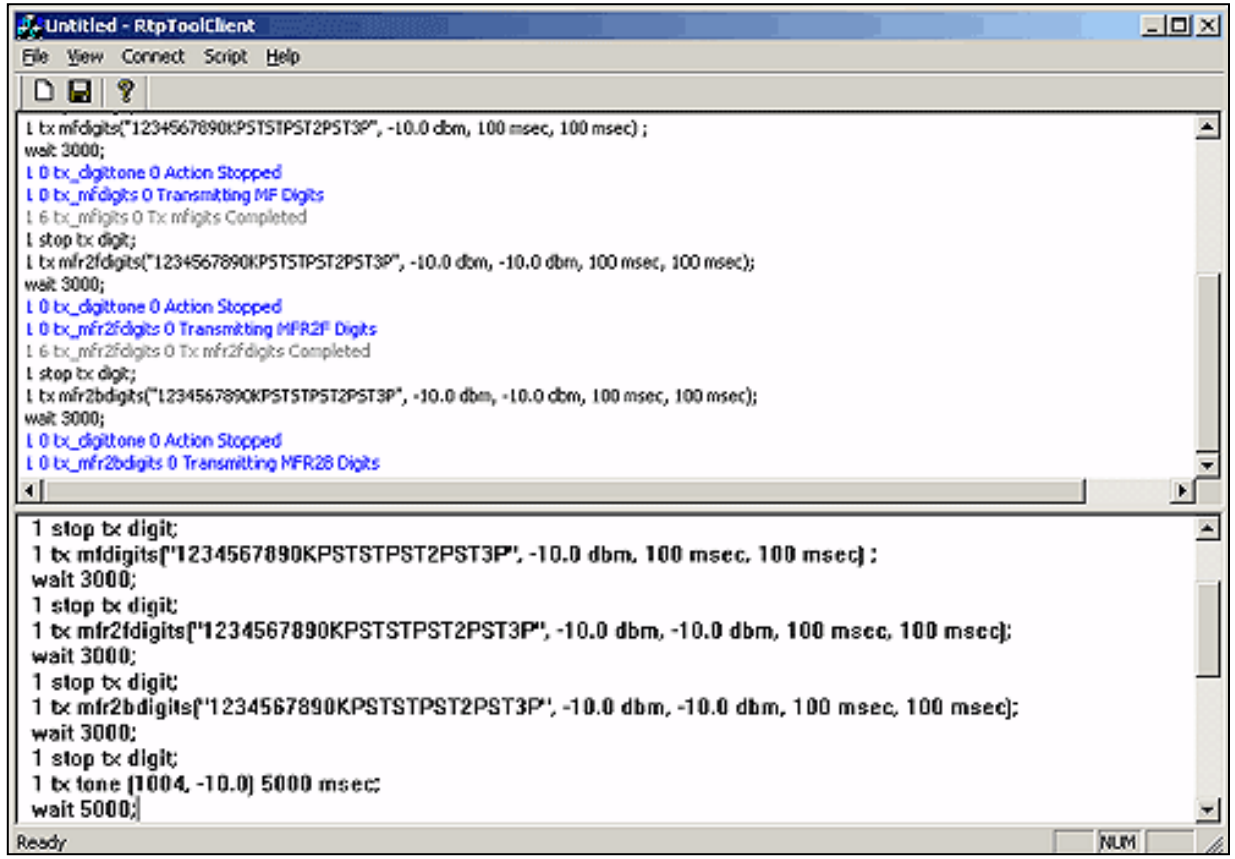
Server Client Functionality

- RTP ToolBox™ can be configured as server-side application
- Enables remote controlling of the application through multiple command-line based clients
- Supported clients include C++ and TCL based clients
- Perform all functions such as creating RTP sessions, Digit/Tones/Event generation and reception, Setting impairments, Creating session profiles and so on remotely
- Generates and receives SIP calls through commands

Server Client Functionality





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Thank you