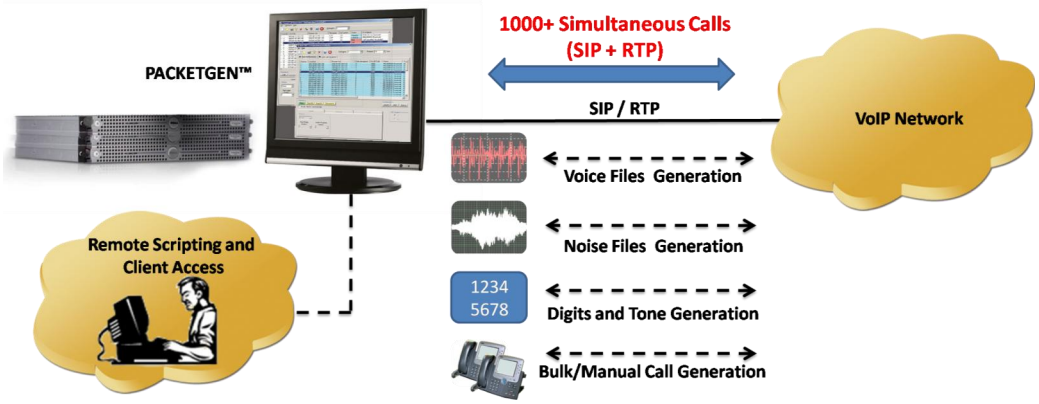
- SIP Bulk Call Generation with or without RTP
- RTP Traffic Generation (Voice, Fax, Digits, Tones)
- Test SIP Phones, Proxy Servers, Registrars, PSTN, & Media Gateway
- Scalable Distributed Architecture allows Higher Call Density
- G.711, G.726, G.729AB, GSM G.722, G.722.1, AMR, EVRC, SMV, iLBC
- SIP functionality – Registration, Call Hold, Call Forwarding, Authentication
- Scripting for RTP Traffic
- Load Testing with High Call Rates and Media Streams
- Manual and Bulk Call Generation
- Remote Access Capability
- RTP Impairments Generation
- Statistics, Events and Call Records

PacketGen™

(SIP Bulk Call Generator)



Overview

PacketGen™ is a PC-based real-time VoIP bulk call generator (including both SIP signaling and RTP generation) for stress testing and precise analysis of the VoIP network equipment. PacketGen™ is based on a distributed architecture, wherein SIP and RTP software cores can be modularly stacked in one or many PCs to create a scalable high capacity test system. PacketGen™ can be used to test basic functionality and verify proper protocol implementation in SIP based equipment such as SIP phones and Network servers, as well as Proxy Servers, Registrar servers, and PSTN and Media Gateways.

GL's PacketGen breaks ground with high density performance:

PacketGen™ on a Duo Quad Core PC can support 1000+ simultaneous calls with both SIP and RTP generation. This performance number is associated with using the G.711 codec, other codecs may provide higher call densities. PacketGen's distributed architecture achieves higher call density by interconnection multiple computer systems with SIP and RTP software cores on each.

Applications

- Stress Testing
- Manual and Bulk Call Generation
- Voice Quality Analysis
- Regression and Acceptance Testing
- Matrix Testing
- Protocol Compliance, Codec Compatibility Testing

Main Features

- Distributed architecture for SIP and RTP systems provides high call rates and media streams. Also, makes it scalable, i.e., easy to add additional load generation capacity.
- PacketGen™ breaks ground with high density performance by generating 1000+ simultaneous calls on a Duo Quad Core PC. Higher density is also achievable using multiple systems.
- RFC 3261 compliant, RFC 2833 digit generation/detection.
- Generates both SIP signaling and RTP traffic.
- Send / Record voice files on any (or all) RTP sessions. Also, provides the necessary voice quality algorithms.
- Audio Codecs supported are G.711 (A-law and U-law), G.711 App II (A-law and U-law with VAD), G.726, G729AB, GSM, G.722, G.722.1 (Wideband), AMR/AMR_WB (Narrow band/ Wideband), SPEEX/SPEEX_WB (Narrow band/Wideband), iLBC (20ms and 30ms), EVRC, EVRCB, SMV, and iSAC.
- Remote access capability using GUI, command line interface, or through Remote Desktop.
- Supported on 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/packetgen.html.

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SIP Setup and Configuration

The SIP Setup screen controls the foundation of the desired test environment. The user has the flexibility to configure multiple SIP and RTP instances on a local system and/or remote systems. Each SIP and RTP instance provides additional call density capabilities, thus allowing a true distributed architecture. In addition, true RTP Load sharing is provided within PacketGen™.

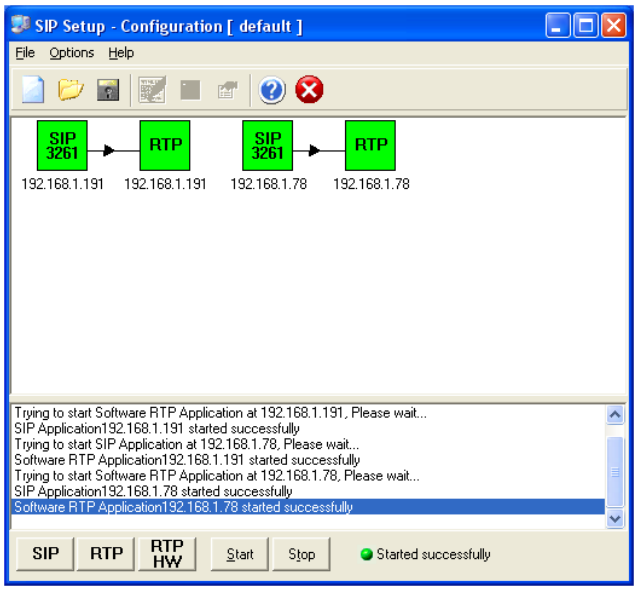

Figure: SIP Setup Configuration

SIP Registration

PacketGen™ provides facility to register a single or a bunch of User Agents simultaneously. Each Registration gives flexible configuration options like Registrar server address, Address of Record, Expiry time etc. Also, each Registration can be configured for automatic registration refresh, after the existing registration expires. A quick configuration utility helps to configure hundreds of registrations easily.

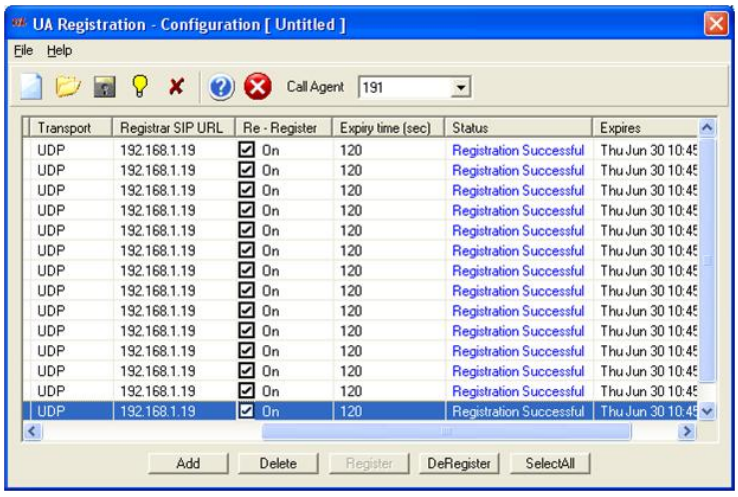

Figure: UA Registration Configuration

Manual and Bulk Call Generation

PacketGen™ supports both Manual and Bulk Call Generation, with complete flexibility on each individual call session such as quick configuration utility, current status of each configured session, traffic generation and QOS measurements, call processing options including hold and call transfer.

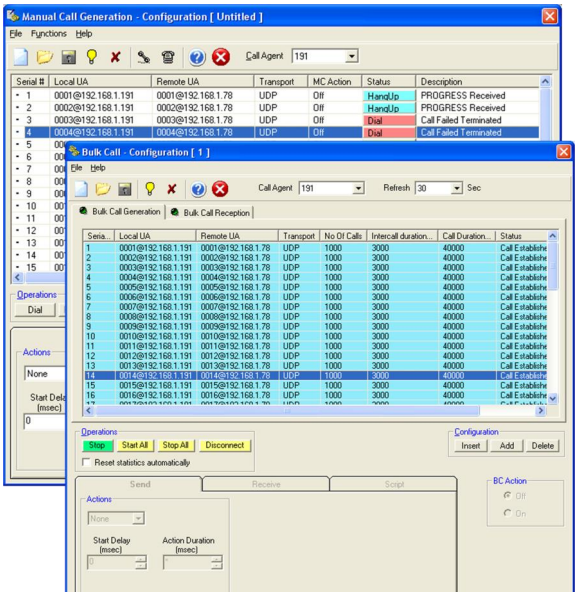

Figure: Manual and Bulk Call Generation

Auto Signaling Action

This feature provides a quick and easy method to configure signaling actions, to be performed automatically as soon as the call session is established. Configuration is based on call sessions, thus each call may be configured for unique activities.

Signaling options include Call Transfer, Call Reject (User-Defined Error), Hold and Re-Direct.

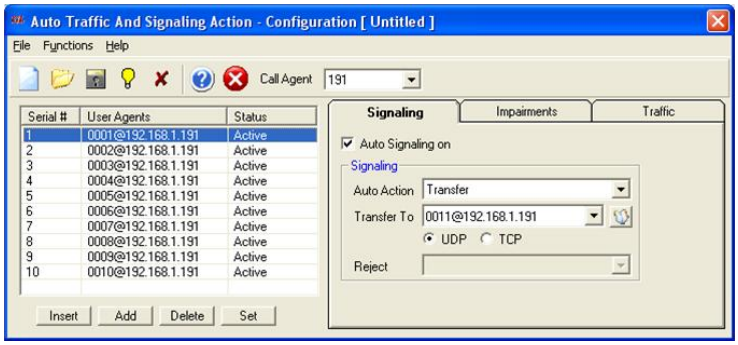

Figure: Auto Traffic and Signaling Actions

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Traffic Generation (RTP)

Once the call is established, PacketGen™ can generate and handle multitude of traffic, either manually or automatically. It facilitates the mechanism to test various network conditions and responses. Traffic options include Send Actions, Loop Back, Receive Actions, and Power Measurement.

Auto-Action feature provides a quick and easy method to configure signaling as well as traffic actions, once the call session is established.

Advanced traffic options like codec parameters, ptime and Rx jitter buffer control are provided.

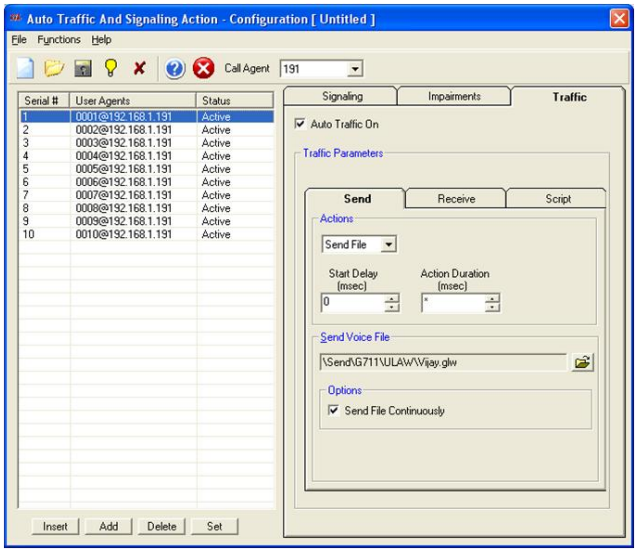

Figure: RTP Traffic Generation

RTP Impairment Generation

PacketGen™ allows user to configure various impairments on outgoing RTP streams. These categories of impairments can be generated.

- Latency
- Packet Loss
- Packet Effects

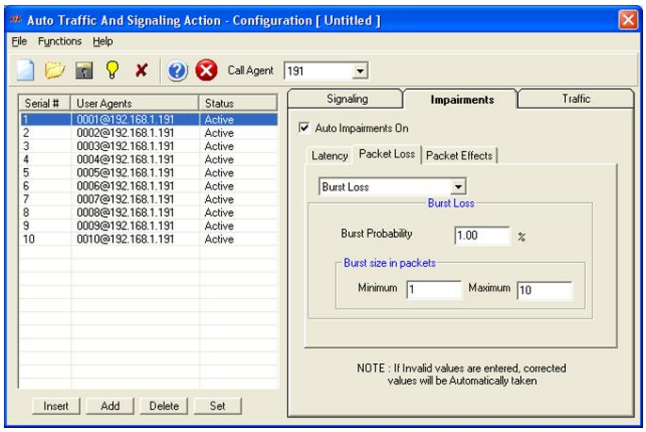

Figure: RTP Impairment Generation

Statistics & Events

PacketGen™ provides detailed statistics for each User Agent, SipCore as well as for the entire system. Included in the statistics are complete/incomplete calls, failed calls (based on user-defined thresholds) and type of generated traffic. Call Statistics window provide detailed call wise statistics per SipCore.

System statistics window provides the overall call statistics such as active calls in progress, completed calls, number of successful calls, attempted calls, and so on for each SipCore.

All events and statistics can be exported and saved for record or review at a later time.

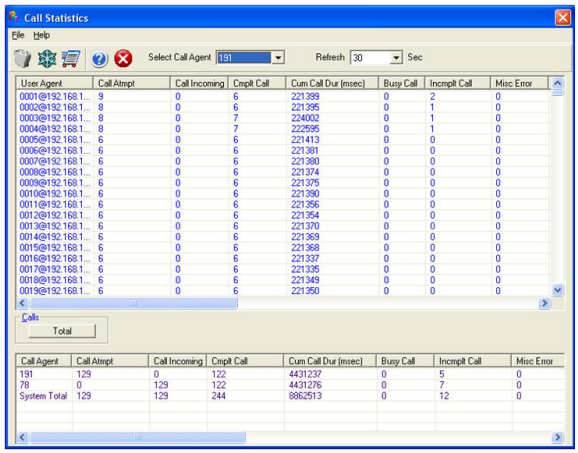

Figure: Statistics and Events

RTP Action Scripting

PacketGen™ provides a powerful scripting capability to control RTP traffic. Scripting features includes loops, conditional statements, wait for events, timers etc., Scripting gives the user greater control over the RTP traffic being generated allowing users to create/test IVR kind of applications. Scripts can be created using the RTP Script Editor, which allows an intuitive, point and click script setup.

The set of script elements included in the script editor allows user to perform all the traffic generation / reception actions as done using PacketGen™ main graphical user interface.

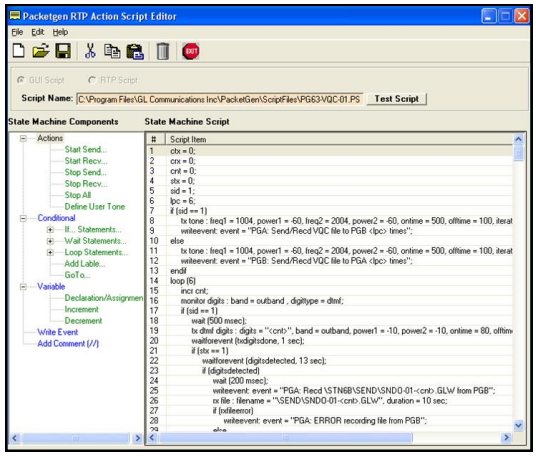

Figure: RTP Action Scripting

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Audio File Converter Utility (AFC)

PacketGen™ now transmits and records voice files using a GL proprietary file format (.glw). The accompanying GL Audio File Converter Utility (AFC) will automatically convert any voice file, into *.glw file format and vice versa. This allows the ability to send/receive voice files at a higher density with multiple codecs (the file is predefined with the desired codec). It also allows for Discontinuous Transmission/Reception.

The Auto FCU (part of AFCU) is generally used in conjunction with GL's VQT application and converts degraded voice files from their native codec format to a standard format used by VQT. The Command line interface (CLI) in AFCU allows the users to load, start, and stop Auto FCU configurations, convert single file (raw / wav / glw file) of one codec to another file format of a different codec using the commands.

Audio Stream Utility

The existing "Playback" feature is used to play the selected call to speaker on the local computer (SIP/RTP core). To allow these calls to be heard from remote systems, GL has introduced Audio Stream Utility with PacketGen™. This utility automatically streams the voice of a selected call to a speaker on a remote system.

Voice Quality Testing using PacketGen™

PacketGen™ can be used to establish calls and send/record voice files over the IP network. These voice files are then analyzed using GL's Voice Quality Testing (VQT) application as per ITU algorithms. Voice Quality testing can be completely automated using Packet-Gen™ CLI, RTP Action scripting, along with ASR Listener, FCU and VQT software.

Command Line Interface

In addition to the GUI, PacketGen™ can also be operated through a Command Line Interface (CLI). All the functionalities of the PacketGen™ GUI are supported, except the configuration functions. Users can thus operate PacketGen™ from a DOS based console (instead of the GUI) or easily integrate PacketGen™ into their own applications.

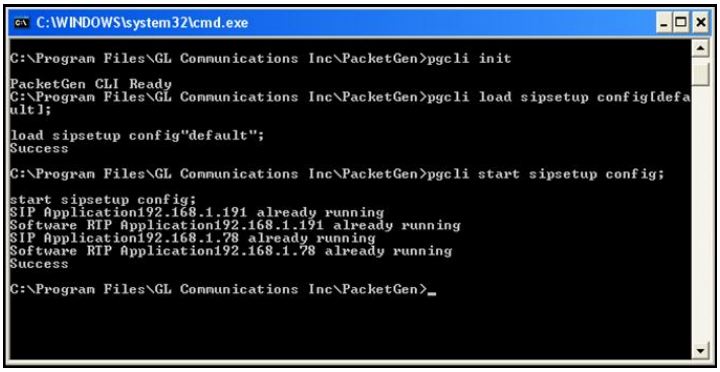

Figure: Command Line Interface

Buyer's Guide

PKS100 – PacketGen™ (includes PacketScan™)

PKS101 - SIP Core (additional)

PKS102 - RTP Soft Core (additional)

PCD102 - iSAC Codec for PacketGen™

PCD103 - AMR Codec for PacketGen™

PCD104 - EVRC Codec for PacketGen™

PCD105 - EVRC-B Codec for PacketGen™

PCD106 - EVRC-C Codec for PacketGen™

Related Software

PKS110 - Packet H. 323

PKV100 - PacketScan™ (Online and Offline)

PKB100 - RTP ToolBox™

PKS120 - MAPS™ SIP Emulator

PKS121 - SIP Conformance Test Suite (Test Scripts)

PKS122 - MAPS™ MEGACO

PKS123 - MAPS™ MEGACO Conformance Suite (Test Scripts)

PKS124 - MAPS™-MGCP

PKS125 - MAPS™ MGCP Conformance Suite (Test Scripts)

PKS130 - MAPS™ SIGTRAN

PKS135 - MAPS™ ISDN-SIGTRAN (ISDN over IP)

PKS140 - MAPS™ LTE-S1 interface

PKS142 - MAPS™-LTE eGTP

PKV107 - LTE Analyzer

PKV105 - SIGTRAN Analyzer

IPN400 - IPNetSim™ - 1Gbps w/ 4 links through bandwidth

FXT002 - GL Insight™ - Single Fax Analysis – IP

MDT002 - GL Insight™ - Single Modem Analysis – IP

PKS150 - TDM / VoIP Gateway w/ Analog and Digital Interfaces

VQT004 - Voice Quality Testing (PAMS, PSQM, PESQ)

VQT013 - VQuad™ with SIP (VoIP) Call Control

VBA032 – Near Real-time Voice-band Analyzer

PKB070 - Audio Processing Utility

*Specifications are subject to change without notice.

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