

SIP Bulk Call Generation with or without RTP



RTP Traffic Generation (Voice, Fax, Digits, Tones)



SIP Phones, Proxy Servers, Registrars, PSTN and Media Gateway testing



Scalable Distributed Architecture



G.711, G.726, G.729ab & GSM codecs.



Voice Quality algorithms (PAMS, PSQM, PESQ MOS scores)



Scripting for RTP Traffic



Load testing with high call Rates, Media Streams with real world traffic



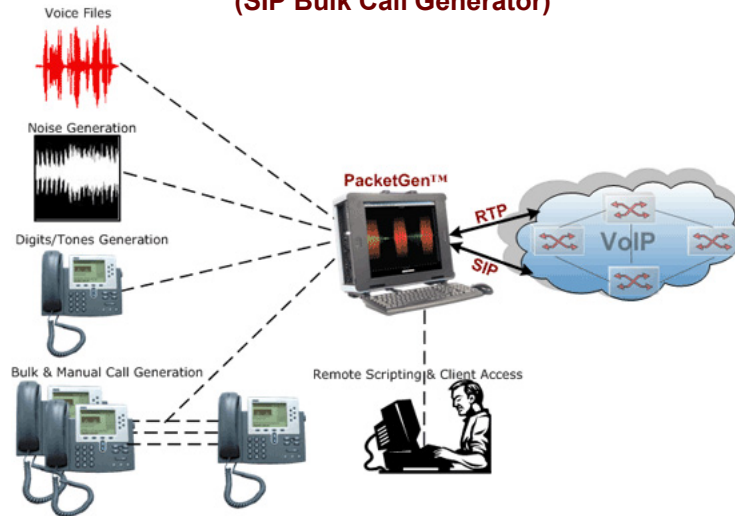
Remote Access Capability



RTP Impairments Generation



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. An optional hardware RTP can support 120 real-time voice calls from real phones, or fax calls from fax machines. Calls can also be made to IP phones and to IP Analog Telephone Adapters. **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.

PacketGen™ Specifications

- Windows 2000/XP, 2 GHZ, 512 MB RAM, 40 GB Hard Drive, 10/100/1000 Ethernet Port, Parallel or USB port for License, Headphones and Microphone.
- RFC 3261 compliant, RFC 2833 digit generation/detection.
- 500 simultaneous calls per SIP Core per PC (20 to 50 cps/PC) (Without RTP)
- 200 Simultaneous calls (with RTP) per PC running single SIP/RTP Software Core. Distributed architecture permits 150 Simultaneous calls (with RTP) per each additional PC

Main Features

- Distributed architecture for GUI, SIP and RTP systems (provides high call rates and media streams). Also, makes it scalable – easy to add additional load generation capacity.
- Generates both SIP signaling and RTP traffic.
- Full SIP functionality – Registration, Call hold, Call forwarding, Authentication etc.
- Manual and Bulk call generation with complete flexibility on each call session.
- Send/Record Voice files on any (or all) RTP sessions. Also, provides the necessary Voice Quality algorithms, thus providing the ITU standard PAMS, PSQM, PESQ MOS scores.
- Perform various actions like Send/Detect digits/tones (both Inband and RFC2833), Talk and Playback etc. on any (or all) RTP sessions to simulate real world traffic.
- Audio Codecs supported are G.711 (A-law and U-law), G.726, G729ab and GSM.
- Powerful scripting capability for RTP traffic generation, which allows user to simulate/test IVR kind of systems. Allows for conditional commands as well as script looping.
- Automatic generation of impairments such as Latency, Packet Loss, Out of order packets, Duplicate packets etc. over the RTP for any (or all) established calls.
- Remote Access Capability using GUI or Command Line Interface.
- Provides Statistics, Events and Call Records.



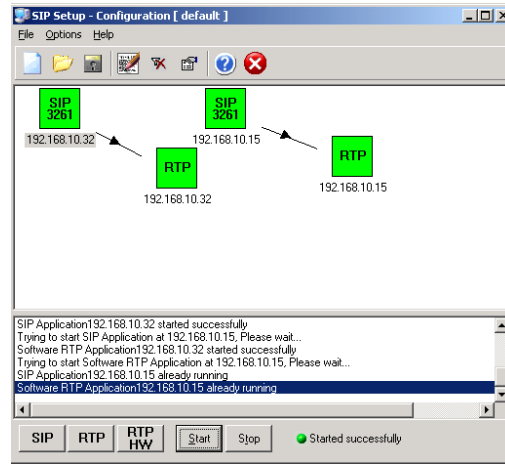
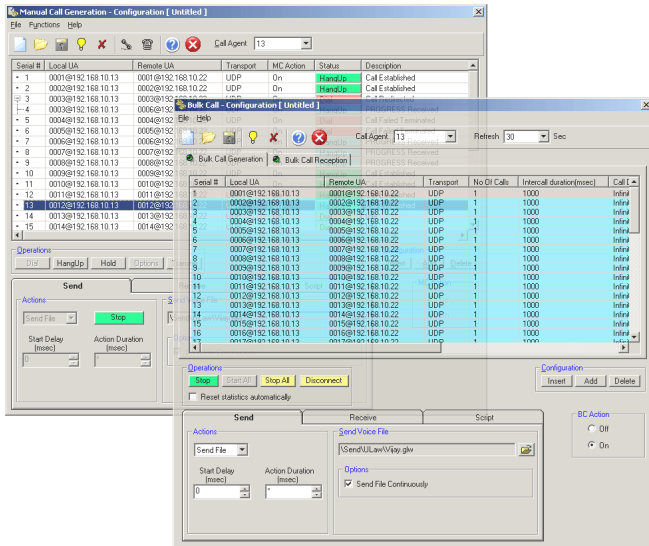
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Sip Setup and Configuration

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 the 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™.

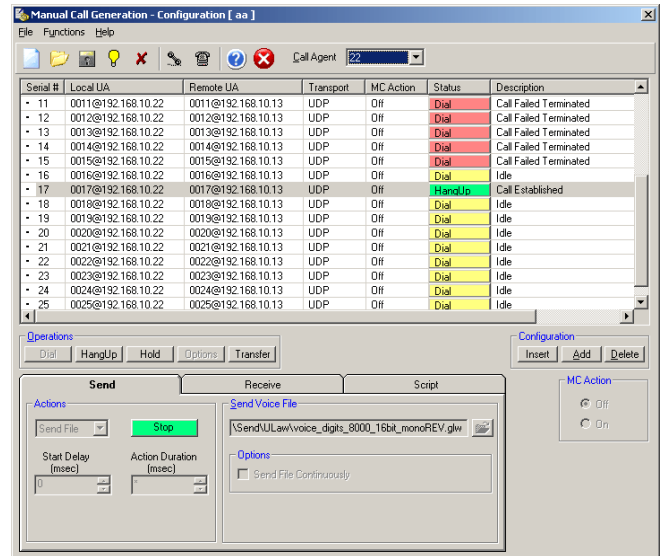
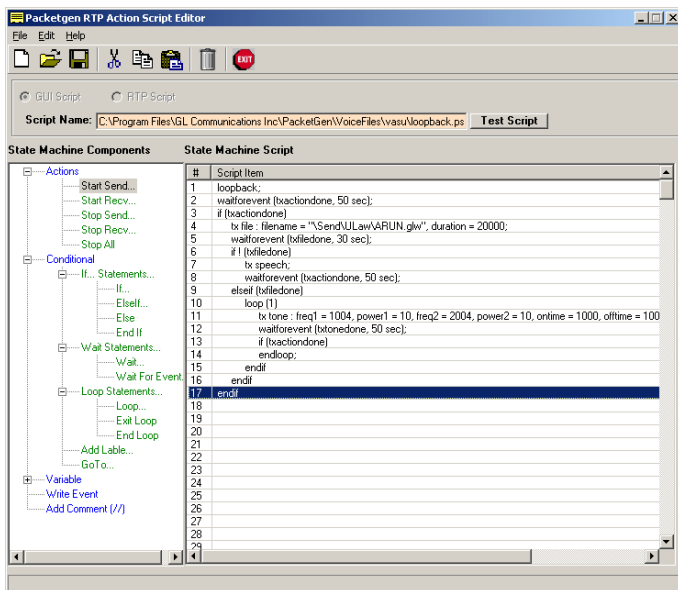


Manual and Bulk Call Generation

PacketGen™ supports both Manual and Bulk Call Generation, with complete flexibility on each individual call session like Quick Configuration Utility, Current Status of each configured session., Traffic Generation and QOS Measurements, Call processing options including Hold and Call Transfer.

RTP Traffic options

PacketGen™ allows the user to send a voice file, record to a voice file, generate Loopback traffic (all received traffic will be re-generated as send traffic), send DTMF/MF in-band/out-band digits, send user-defined single/dual Frequency Tones, generate real-time voice using the default audio device (microphone).



PacketGen™ 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 user to create/test IVR kind of applications. Scripts can be created using the RTP Script Editor, shown beside, which allows an intuitive, point and click script setup.

Auto-Action Functionality

PacketGen™ Auto-Action provides a quick and easy method to configure signaling as well as traffic actions, once the call session is established. Configuration is call session based, thus each call may be configured for unique activities. Signaling options include Call Transfer, Call Reject (User-Defined Error), Hold and Re-Direct. Traffic options include Transmit/Record Voice, Generate/Detect Tones, Digits and Noise and Send/Receive Fax

User Agent	Call Atmpt	Call Incoming	Complt Call	Cum Call Dur (msec)	Busy Call	Incomplt Call	Misc Error
009@192.168.10.24	0	0	22	50409	0	1	0
010@192.168.10.24	0	0	22	50458	0	1	0
011@192.168.10.24	0	0	22	50777	0	1	0
012@192.168.10.24	0	0	22	50742	0	1	0
013@192.168.10.24	0	0	22	50813	0	1	0
014@192.168.10.24	0	0	22	50832	0	1	0
015@192.168.10.24	0	0	22	50893	0	1	0
016@192.168.10.24	0	0	22	51029	0	1	0
017@192.168.10.24	0	0	22	50987	0	1	0
018@192.168.10.24	0	0	22	50982	0	1	0
019@192.168.10.24	0	0	22	50999	0	1	0
020@192.168.10.24	0	0	22	50992	0	1	0
021@192.168.10.24	0	0	22	51131	0	1	0
022@192.168.10.24	0	0	22	51112	0	1	0
023@192.168.10.24	0	0	22	51048	0	1	0
024@192.168.10.24	0	0	22	51041	0	1	0
025@192.168.10.24	0	0	22	50981	0	1	0
026@192.168.10.24	0	0	22	50824	0	1	0
027@192.168.10.24	0	0	22	50902	0	1	0

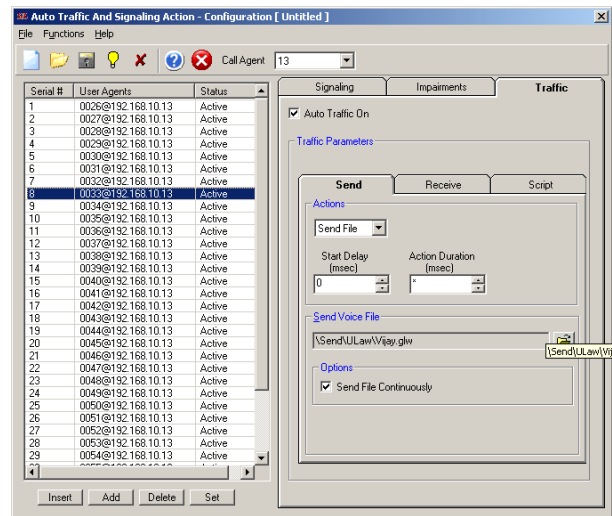
Call Agent	Call Atmpt	Call Incoming	Complt Call	Cum Call Dur (msec)	Busy Call	Incomplt Call	Misc Error
32	2449	0	2015	4581813	0	345	0
14	0	2098	1888	817363	0	92	0
System Total	2449	2098	3883	5383596	0	437	0

PacketGen™ Events

PacketGen™ provides the events screens such as Call Records, Captured Events, Captured Error Events, Tone/Digit Detection, Bulk Call Events, Events Search, and Error Log. Some of the events can be seen as shown in the figure to the right.

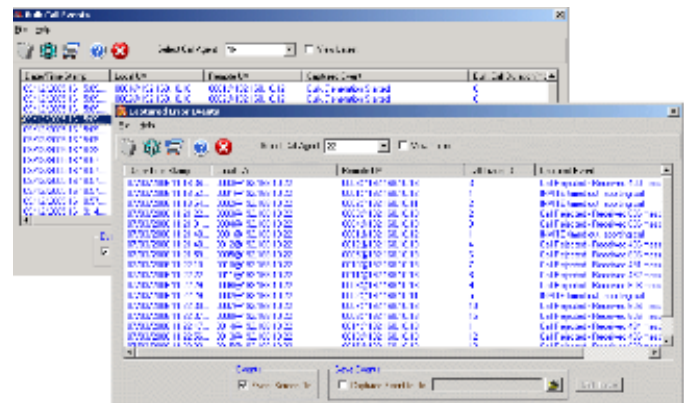
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D:\WINDOWS\system32\cmd.exe
D:\Program Files\GL Communications Inc\PacketGen>pgcli init;
PacketGen CLI Ready
D:\Program Files\GL Communications Inc\PacketGen>pgcli load sipsetup con;
load sipsetup config"cli";
Success
D:\Program Files\GL Communications Inc\PacketGen>pgcli start sipsetup co
start sipsetup config;
SIP Application192.168.10.22 started successfully
Software RTP Application192.168.10.22 started successfully
Success
D:\Program Files\GL Communications Inc\PacketGen>pgcli connect 192.168.
11 1g11;
connect 192.168.10.22 "g1" "g1";
Fail
Failed to Connect to Server - 192.168.10.22
D:\Program Files\GL Communications Inc\PacketGen>
    
```



PacketGen™ Statistics

The PacketGen™ Statistics screen provides detailed statistics for each user agent as well as for the entire system. All events and statistics screens are presented in a similar, very easy to read manner, which can be exported and saved for record or review at a later time. Included in the statistics are complete/incomplete calls, failed calls (based on user-defined thresholds) and type of generated traffic



PacketGen™ 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.

Buyers Guide:

- PKS100- [PacketGen™ \(includes PacketScan™\)](#)
- PKS101- [SIP Core \(additional\)](#)
- PKS102 - [RTP Soft Core \(additional\)](#)
- PKS201 - [RTP Hardware Core \(120 Port\)](#)

Related Software

- PKV105- [SIGTRAN Analyzer](#)
- PKS110 - [Packet H. 323](#)
- PKV100 - [PacketScan™ \(Online and Offline\)](#)
- PKV101 - [PacketScan™ - Offline](#)
- PKB100- [RTP ToolBox™](#)