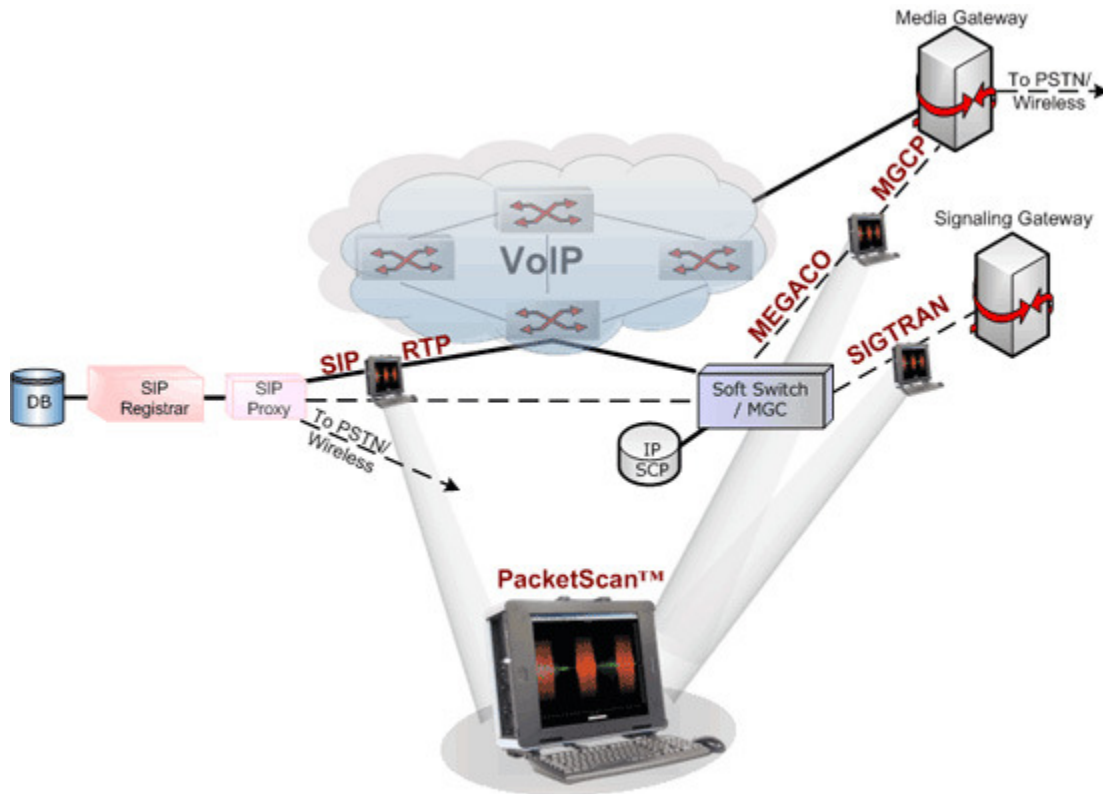


PacketScan™

SIP | H323 | Megaco | MGCP | RTP | RTCP | Video Analysis



Overview

PacketScan™ is a real-time VoIP analyzer that captures live IP traffic, and segregates them into SIP/H323 calls and collects statistics about the calls. Hundreds of calls can be monitored in real-time including detailed analysis of selected voice band streams. Applications include testing of IP phones, Gateways, IP Routers and Switches, and Proxies.

Users can perform a host of activities on the captured calls, allowing you to get an exact picture of QOS (quality of the service) and the technical adherence (adherence to the protocols specified by the standardizing authority) of the system under test.

PacketScan™ allows users to listen/record VoIP calls in real-time; perform power, frequency, spectral, tone and digit analysis with ease and precision. Its ability to monitor / record audio and video data of a session to files (in QuickTime *.qt format), allows users to perform powerful video analysis. The captured VoIP calls with video can be played back using 3rd party VLC

Viewer application. Detailed call statistics such as packet loss, gap, jitter, delay, RTP performance statistics, and unparalleled voice band statistics can be monitored simultaneously. Sophisticated filters permit zooming and recording of specific calls of interest.

It allows to record all or filtered traffic into a trace file. It supports Filtering and search capability which adds a powerful dimension to the SIP analyzer.

Packet Data Analysis (PDA)/ Traffic Analysis support the following two views

- Summary View (Call Quality Matrix) to display call information in graphical format
- Detail View (RTP Diagnostic View) to display call information in tabular format

These interfaces allows users to define parameters such as E-Model based MOS and R-Factor score, VQmon settings, & Dynamic Payload Mapping. It also supports real-time digit capturing like DTMF, MF and user-defined digits/tones.

What sets apart PacketScan™ is its ability to collect vital statistics about calls for theoretically infinite time. The ability of PacketScan™ to capture data is limited only by the hard disk capacity of the PC.

Main Features

- Monitor progress of up to 500+ simultaneous calls with bidirectional RTP traffic
- Supports SIP (SIP Session Initiation Protocol -2543 and -3261), Megaco3525, Megaco3015, MGCP, and H323 protocols.
- Supports decoding of MAC, IP, SIP, UDP, TCP, RTP, & RTCP
- Call Capturing based on Call Agents or Trigger Actions such as MOS, packet loss, latency, or called / calling numbers
- PacketScan™ supports standard codecs such as
 - G.711 (μ -law and A-Law - 64 Kbps)
 - G.726 (40/32/24/16 kbps)
 - GSM (13.2kbps)
 - G729 (8kbps)
 - G729B (8kbps)
 - G.722
 - G.722.1 (24 kbps and 32 kbps Wideband)
 - G.711 with VAD
 - AMR (Wideband codec - 6.60kbps, 8.85kbps, 12.65kbps, 14.25kbps, 15.85kbps, 18.25kbps, 19.85kbps, 23.05kbps, 23.85 kbps)

- AMR (Narrow band codec - 4.75kbps, 5.15kbps, 5.9kbps, 6.7kbps, 7.4kbps, 7.95kbps, 10.2kbps, 12.2 kbps)
- ILBC (15.2kbps and 13.33kbps)
- SPEEX (Wideband)
- SPEEX (Narrowband)
- ISAC (An optional Codec, must be purchased as a separate dongle extension)
- EVRC (Rates - 1/8, 1/2 and 1)
- SMV (Modes - 0, 1, 2 and 3 - Available if licenses are provided or owned, please call GL)
- H.263+ providing video capture and videoconference monitoring capability.

For more comprehensive information on these codecs click [here](#).

Full RTP Analysis with audio capture/playback supported for all common Codecs

1 GB circular buffer with capture and view filters

Decode AMR in all packing formats, & G.726 RTP in AAL2 or IP packing types

Search and filtering capabilities for both Real time as well as Offline Analysis.

It combines VoIP Protocol Analyzer (VPA) and Traffic Analyzer (TA) view.

Provides Summary, Detailed, Hex-dump, and Call detail records view of the captured traffic.

Summary view displays Frame number, Time, LEN, Error, Source TCP/IP/UDP Address, Destination TCP/IP/UDP address, Packet Type, and other SIP/RTP call details.

Statistics view displays statistics based on frame count, byte count, frames/sec, bytes/sec etc for the entire capture data. Statistics can be obtained for any fields or parameters in the protocol

Hex Dump view displays raw frame data as hexadecimal and ASCII octet dump

Detail view displays decodes of user-selected frame like Frame sync bits, control bits & data in detail. Further details of an individual (RTP) session of a call can also be obtained using Traffic Analyzer (TA) or the Packet Data Analysis (PDA) interface.

Packet Data Analysis (PDA) interface displays call information in graphical format (Traffic Analyzer Call Quality Matrix- Summary View) as well as in tabular format (RTP Diagnostic - Detail View)

User can get real-time call trace information based on H225 calls

Exports detailed and summary information to a comma delimited file for subsequent import into a database or spreadsheet

Can be deployed as a centralized monitoring system with associated relational or Oracle database

Ability to configure sipprot.ini file for customization of decoding options

Calculates minimum, maximum, and average Round Trip Delay (RTD) values for SIP calls

Graphs are provided for key values to give a pictorial representation of the statistics, there by allowing the user to monitor the network with ease

A host of counters gives the user an instantaneous snapshot of the VoIP traffic on the network

Calls and sessions are classified as Active, Completed or Failed giving the user an idea about the calls and its status in the network

Real-time audio monitoring of RTP streams using Audio Playback and Write to File features

Audio Write to file allows user to save the RTP streams as wave files, for later playback or for analysis using more sophisticated audio manipulating tools

Call Quality Of Service (QOS) for all calls with E-Model based (G.107) Mean Opinion Score (MOS) and R-factor with individual and summary statistics presented in graphical and tabular formats

Recommended PC

- Windows 2000/XP/Vista Operating System
- 2 GHZ, 512 MB RAM, 40 GB Hard Drive, 10/100/1000 Ethernet Port, Parallel or USB port for License, Sound Card, Headphones and Microphone

Supported Protocols

A brief overview of the protocols supported by PacketScan™ is given below:

- **Session Initiation Protocol (SIP) - RFC 3261 and 2543**

SIP can be used with other IETF protocols to build complete multimedia architecture. Typically, these architectures will include protocols such as the Real-time Transport Protocol (RTP) (RFC 1889) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) (RFC 2326) for controlling delivery of streaming media, the Media Gateway Control Protocol (MEGACO) (RFC 3015) for

controlling gateways to the Public Switched Telephone Network (PSTN), and the Session Description Protocol (SDP) (RFC 2327) for describing multimedia sessions.

- **Media Gateway Control Protocol (MGCP) - RFC 2705/3435 (3991)**

MGCP is a protocol for controlling Voice over IP Gateways (or Call Agent endpoints) from external call control elements. It assumes a call control architecture where the call control "intelligence" is outside the gateways and handled by external call control elements. The Call Agent can create, modify and delete connections in order to establish and control media sessions with other multimedia endpoints. Also, the Call Agent can instruct the end points to detect certain events and generate signals. The endpoints automatically communicate changes in service state to the Call Agent.

- **Media Gateway Control (MEGACO) - RFC 3525 and 3015**

Megaco, also known as H.248 is a signaling protocol, is used between Media Gateway and Media Gateway Controller (Call Agent). Megaco/H.248 is architecturally quite similar to MGCP, however Megaco/H.248 supports a broader range of networks.

- **H.323**

H.323 provides the foundation for audio, video and data communication on packet based IP network. H.323 specifies functions provided by other ITU_T as well as IETF standards under one umbrella. First, ITU_T- H225 provides callsetup/disconnect and terminal to gatekeeper signaling. ITU_T-H245 protocol adds terminal control functions that are used to negotiate terminal capabilities, channel usage and other end-to-end functions. The IETF RTP and RTCP provide information transport as well as session management. Finally, ITU_T standard voice and video encoding algorithm provide the analog to digital conversion and signal compression required for bandwidth optimization.

PacketScan™ (Real-Time VoIP Traffic Analysis/Monitoring Tool)

Summary, Details and HEX/ASCII Views

The **PacketScan™** main screen allows real-time as well as offline protocol analysis. The analyzer displays summary, detail, call trace, statistics and hex dump view in different panes. The summary pane displays various information such as Frame Number, Time, Length, Message Types, IP source and destination address and so on. User can select a frame in summary view to analyze and decode each frame in the detail view. The Hex dump view displays the frame

Enhanced Trace Saving Options

Users can control the captured trace files by saving the trace using different conventions such as trace files with user-defined prefixes, trace file with date-time prefixes, and slider control to indicate the total number of files, file size, frame count, or time limit. This feature also allows the captured frames to be saved into a trace file based on the filtering criteria set using display filter feature.

Configuring INI Files

Users can edit the SipProt.ini INI files to customize the decoding options. The following parameters can be edited as per user requirements

- **[#RTP_PAYLOAD_RFC4733]:** This header indicates the RFC to be followed for out of band events
- **[#RFC2833]:** This indicates out of band payload type
- **[#RFC2190]:** This field indicates payload type for H.263
- **[#WITHOUT_PAYLOAD]:** This field indicates the static payload range

Save/Load All Configuration Settings

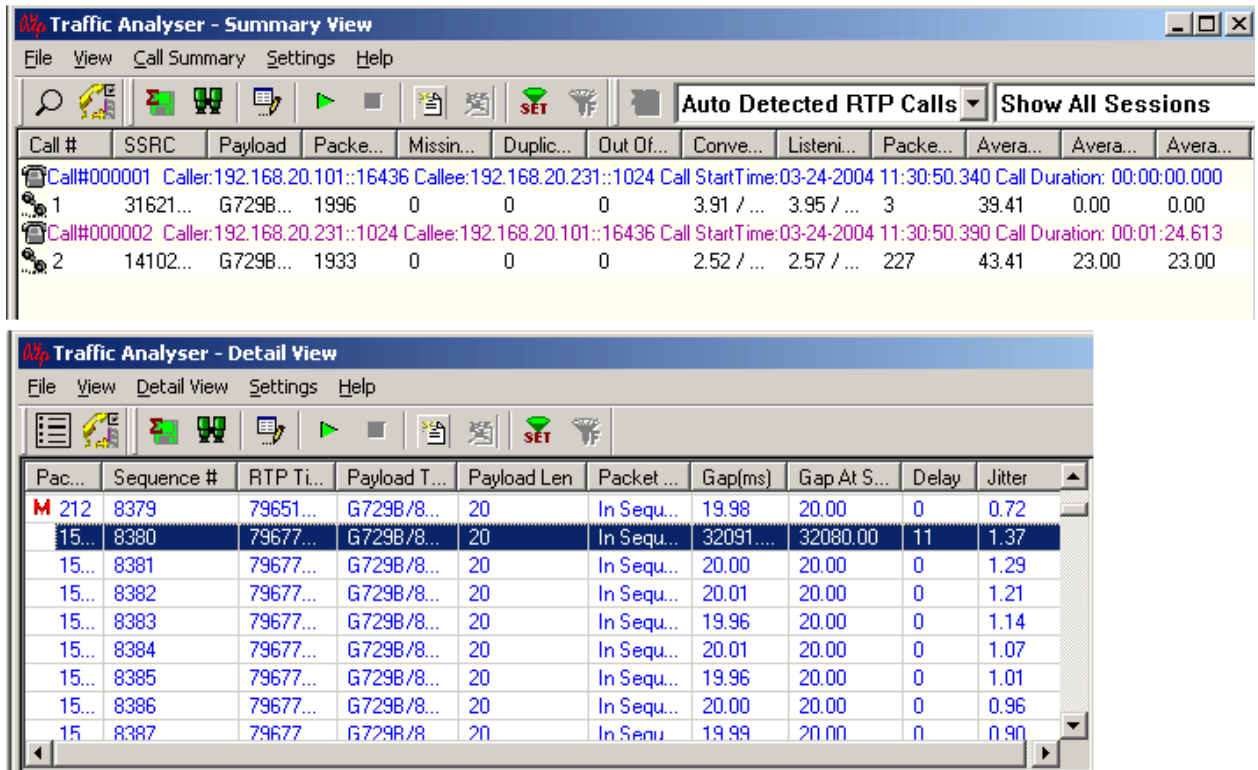
Protocol Configuration window provides a consolidated interface for all the important settings required in the analyzer. This includes various options such as protocol selection, startup options, stream/interface selection, filter/search criteria and so on. All the configuration settings done in any of these options can be saved to a file, loaded from a configuration file, or user may just revert to the default values using the default option.

Packet Data Analysis (PDA) / Traffic Analysis (TA)

PacketScan™ provides in-depth real-time and post-process data investigation. The PDA view assists in any comparisons that are to be made between the two RTP sessions. Vital aspects from the RTP frame needed for close analysis are included provides in tabular and graphical formats. Information provided within the Traffic summary and detailed views include:

- Includes host of graphs such as Gap, Jitter, Gap Distribution, Jitter MOS, Quality, Wave and Spectral Display for media stream analysis of Jitter, Delay, Packet-loss, Sequencing, etc.
- Detailed statistical information are provided for Inband (DTMF & MF)events, RFC 2833 events, RTP/RTCP packet count and reports per direction, Duplicate and Missing Packets
- "Call Trace" for SIP Sessions

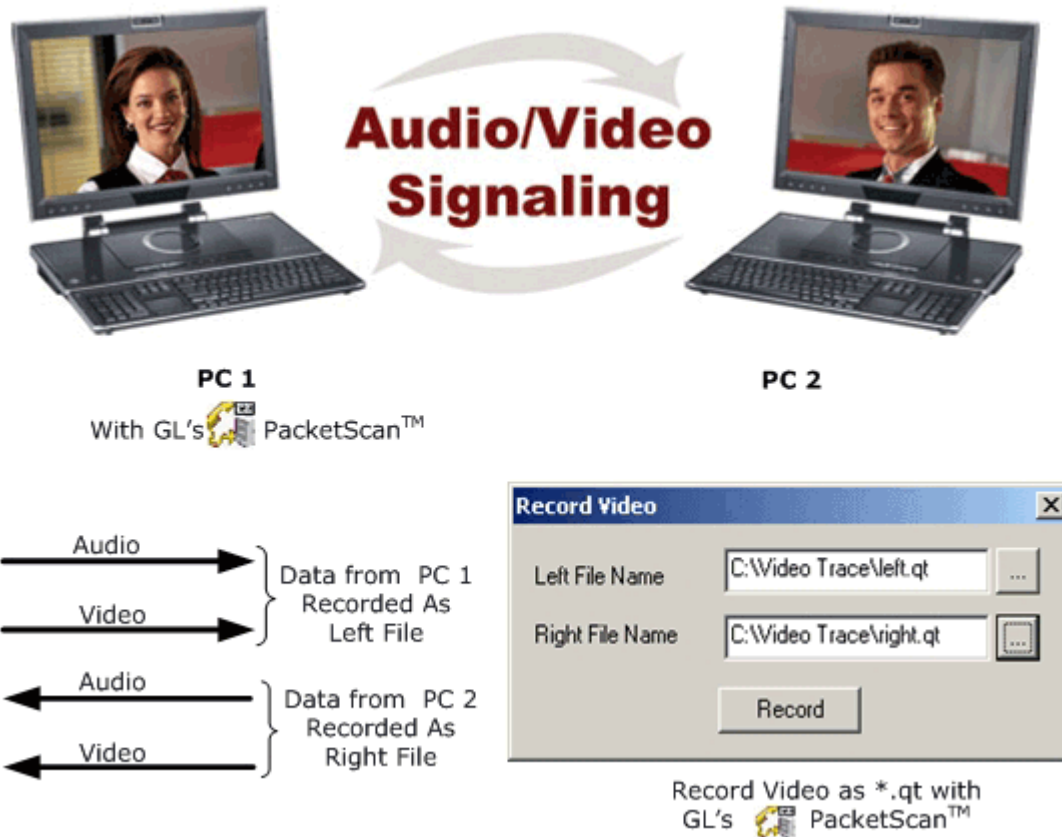
- Users can set E-Model based MOS and R-Factor parameters, VQmon settings, & Dynamic Payload Mapping
- Long-Term Activity Reporting
- Supports triggers and action feature to further filter calls. It allows users to specify the formats and the type of calls to be saved as output files *.hdl and/or *.wav format



Real-Time Digit Capturing

PacketScan™ provides a means for digit examination and capturing. An easy-to-read window indicates the direction of the captured digits along with the duration, power and frequencies. DTMF, MF and User-defined digits/tones are fully supported.

The video call can be recorded to a file in QuickTime format (*.qt) by selecting a SIP/Auto Detected RTP call. PacketScan™ may be installed in either of the system to record the video data as depicted in the image below.



The recorded video data in Quick Time Format (*.qt) can be viewed by VLC Viewer (3rd party application).

Record video option is available for both Auto Detected RTP Calls and SIP Calls. Supported Video Codecs are:

- H263+
- H263++ CIF 190 kbps
- H263++ CIF 350 kbps
- H263++ CIF 512 kbps
- H263++ QCIF 128 kbps
- H263++ QCIF 64 kbps
- H263++ QCIF 80 kbps.

H.263 is a video codec designed by the ITU-T as a low-bitrate encoding solution for videoconferencing.

QCIF - Quarter Common Intermediate Format, a videoconferencing format that specifies data rates of 30 frames per second (fps), with each frame containing 144 lines and 176 pixels per line.

CIF - A video format used in videoconferencing systems that specifies a data rate of 30 frames per second (fps), with each frame containing 288 lines and 352 pixels per line.