

## Watch VoIP Portable Expert Analysis Application

Voice over IP service providers are facing many challenges when rolling out Triple Play networks. Video and Voice deployment will only be successful in a high quality network environment. AnaCise Watch VoIP and IPTV Portable product suites are meant for testing and optimizing the Triple Play environment. Watch VoIP is the application that assists the service provider during installation, testing, verification and troubleshooting of the VoIP service at client site. The Watch VoIP portable application is an easy to use Quality of Service optimisation package tailored for Triple Play networks. Being a powerful application you may indeed also benefit from Watch VoIP Portable when using it in the SIP based of the network access part.

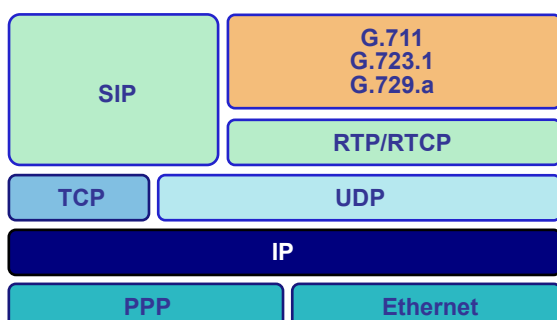


Watch VoIP Portable can perform active or passive real-time monitoring at the VoIP access network, including traffic stream analysis, and is able to view specific call statistics, call trace, immediate listening of VoIP RTP audiostream and analyze those to obtain QoS statistics, such as packet loss, jitter (Variations in delay of packet delivery), R-Factor, G.107, MOS, etc., per call basis analysis.

# WatchVoIP

### Key Features

- Non-intrusive voice quality monitoring and evaluation based on the industry standard ITU-T G.107 E Model ( MOS and R-Factor ).
- Call flow in Ladder Diagram , every SIP call/transaction is individually recorded and traced in graphic call flow to present all of signaling point reiterated with IP address/port number, delta time and an alarm icon to indicate the violation of SIP signaling.
- Support SIP based signaling analysis per RFC-3261 in full details to display complete protocol decoding with plain mnemonics in English.
- RTP voice stream analysis in real time including RTP Jitter, Inter-packet delay, Packet loss, Codec Type, SSRC identify and Session base throughput...etc., per the industry standard RFC-1889 and audio playback of G.711, G.723.1, G.729.a Codecs.
- Supports Peer to Peer or Through Proxy mode in IP phone emulation.
- Easy of use, full Window based Graphic User Interface in Intuitive drill-down methodology.

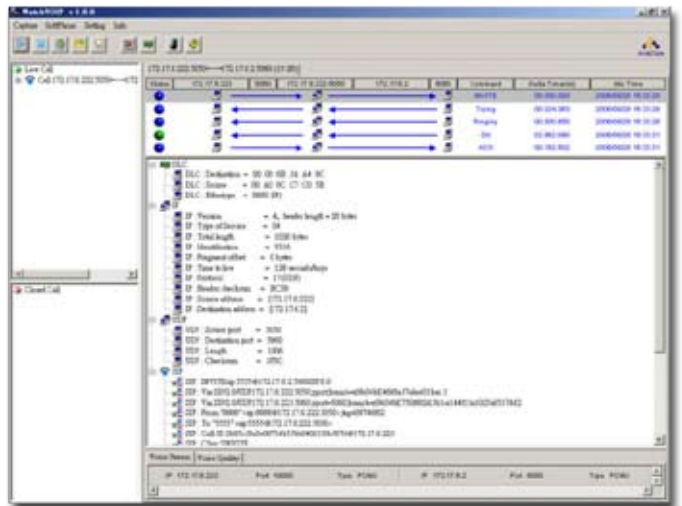


Watch VoIP Portable is focused on the SIP environment. The VoIP protocol stack is based on SIP, G.711, G.723.1, G.729.a, RTP/RTCP and other protocols, Despite a complicated protocol stack, Watch VoIP Portable provides a user-friendly interface, with easy to use tools for fast troubleshooting.

## VoIP Control Data Record (CDR) Interface

The Watch VoIP CDR interface provides instant analysis of the protocols and transactions interacting in the call flow. Watch VoIP traces and builds the flow of all calls involving RTP, RTCP and SIP(RFC-3261) protocols used in association with the VoIP call, a process with many possible steps involved, and just as many possible problems, it decodes all signaling in a call trace flow format for easy troubleshooting of the messages. All timestamps are displayed for resolving of timing related issues. The call process are indicated using easy to understand icons that makes it simple to follow all steps in the call. You will see source and destination IP address of every telephone conversation and the transaction parameters involved in the CDR, time stamped with detailed analysis of the SIP protocol.

- Support for SIP (RFC-3261) signaling.
- Associates signaling, voice streams per call.
- Provides clear flow of the signaling messages.
- Identification of all signaling endpoints participating in the call.
- Provides accurate timestamps of all signaling messages.
- Measurement of call setup timing.
- Off-line Analysis with support for Sniffer and Ethernet file formats.



## IP Phone Emulation

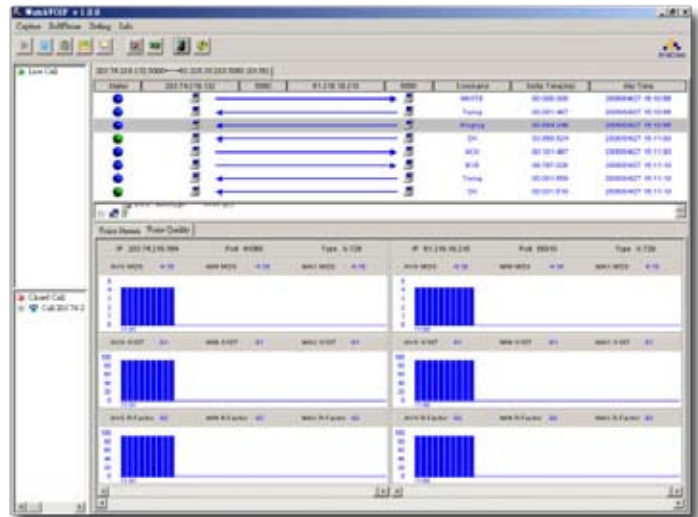
Watch VoIP provides the ability to conduct active test calls using the integrated IP phone feature. The phone is able emulate most of the common IP phone functions. The engineer can use the IP Phone to link back to the Service Center or test server directly in order to conduct the test.

With IP Phone Emulation there is no need to bring along any physical IP test phones to the customer site, and even more it is very easy to test the customer IP phone in order to verify basic phone failure or poor VoIP quality caused by the actual physical handset.

## RTP Packet Analysis

For the end user, in the Triple Play network, large delays are not acceptable as they may cause echo and missing details in the conversation. In a VoIP conversation you tend to interrupt your conversation partner constantly when having long delays. Jitter often causes strange sound effects and packet loss cause interrupts. Watch VoIP Portable provides detailed packet analysis of the RTP stream. When troubleshooting or optimizing the network Watch VoIP collect all information from the Device under test (DUT) for assessment of packet delay variation, packet loss, jitter, and other parameters in order to establish the benchmark and aid the optimization process.

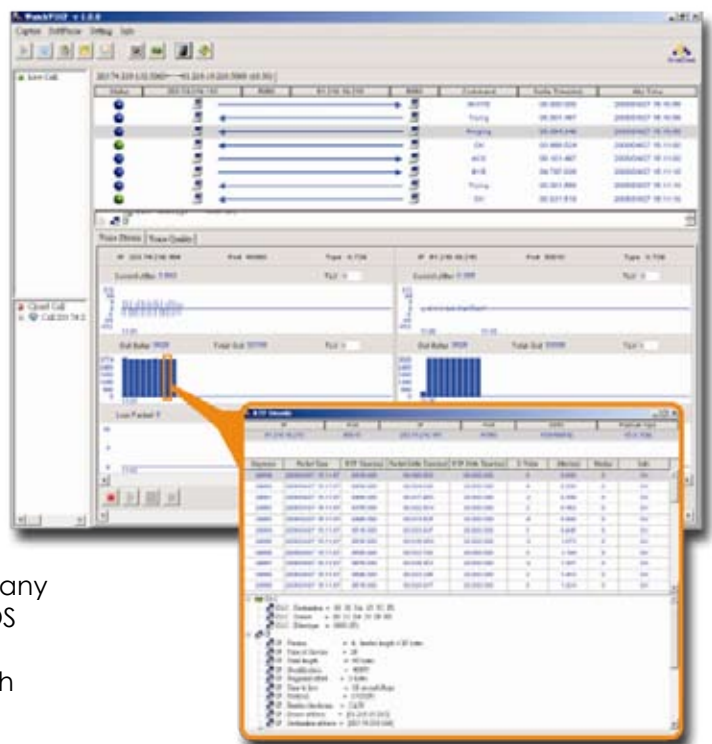
- RTP packet stream analysis.
- Evaluating delay variation, packet loss, jitter and benchmarking values of the DUT.
- Audio playback of G.711, G.723.1, G.729.a



## Non-intrusive Sound Quality Testing

Watch VoIP Portable is using MOS (ITU-T P.800), G.107 and R-Factor, in order to evaluate and rate the voice telephone traffic quality. Watch VoIP incorporates an extended version of the E Model analyzing the effects of time varying IP network impairments and provides a more accurate estimate of user opinion. Watch VoIP is a non intrusive flexible and user friendly application that allows you to analyze the call quality in many ways. It allows you to set up the analysis using 3 parameters, chosen between R-factor to Mean Opinion Score (MOS), Listener Quality (MOS-LQ), Mean Opinion Score PESQ (MOS-PQ), Conversational Quality (MOS-CQ), G.107 and R Conversation and evaluate those against each other.

- Objectively evaluates the media quality of any live traffic-based calls according to the MOS scale (ITU-T P.800), based on the network industry standard ITU-T G.107 E Model (which provides R-Factor evaluation) and voice quality evaluation.



## Watch VoIP Portable Specifications

### Standards supported

- ITU-T G.107 E Model(MOS and R-Factor evaluation)
- SIP(RFC-3261) signaling
- Audio playback of G.711, G.723.1, G.729.a Codecs

### System Requirements

- Pentium 4 1.8GMHz or faster CPU
- Min 512MB RAM(1024MB recommended)
- Min 2GB free hard disk space
- CD-ROM drive
- Min 1024x768 VGA monitor
- Sound card
- Operating system support: Microsoft Windows 2000 (SP4), Microsoft Windows XP with SP2

## Optional WatchINet Hardware Specifications

### Interface module

- USB 2.0 x 1 port
- Dual 10/100 Mbps RJ-45 Ethernet ports with full wire speed operations for monitoring and traffic simulation

### Power

- USB bus powered from the notebook PC; Max. 5W Power consumption

### Traffic Simulation Mode

- Layer-2~7 packet generation at speed up to 100Mbps with flow control
- Generation of up to 256 streams with unicast, multicast or broadcast modes
- Transmission of packet captured, or programmable packet length, sequential packet and pre-defined
- Packets stored in on-board memory or external file in matrix mode>Error generation – CRC, alignment, over/under size, dribble, IP checksum, etc.

### Weights

- 95mm x 76.6mm x 19.6mm ; 200g

### LED Status

- Auto negotiation, Link/Tx/Rx/Error



### Built-in TAP Mode

- Bi-direction monitoring packet capturing of network traffic at full wire speed
- Built-in hardware filter with up to 8 bytes offset patterns for IP address, TCP/UDP port number as well as protocol types
- Real-time display of traffic statistics:Packet/Byte/Utilizatio/CRC/Runts/Jabbers/Alignments/Broadcast/Multicast
- All of measurements with traffic analysis, protocol decoding & expert diagnostics are passed to Third-parity LAN analysis software



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