Testing VoIP with diversifEye™

Voice over IP (VoIP) is one of the three elements of the complete converged IP network service offering of Triple Play. VoIP communication is made possible by two key protocols SIP and RTP. SIP is used to establish the call and RTP is the means in which the audio content is packetized and transmitted. VoIP will have to share the existing bandwidth with other applications such as Video and data.

To ensure VoIP performance and quality, network delay must be insignificant. Packet latency or delay should not exceed 250ms, after this point sound quality will be perceived as poor. Ensuring QoS settings are correct on the network elements helps, however other components such as choice of codecs are as important.

diversifEye™ the converged IP communications test solution of choice enables users to evaluate VoIP performance through emulating user and server client side agents. Along with fundamental bulk call testing diversifEye™ also determines per individual call performance, busy hour call attempts, number of call rejects, MOS (R factor and PESQ) scoring and how the choice of codec effects performance plus a host of additional statistics.

2007 FROST & SULLIVAN Global Product Innovation Award

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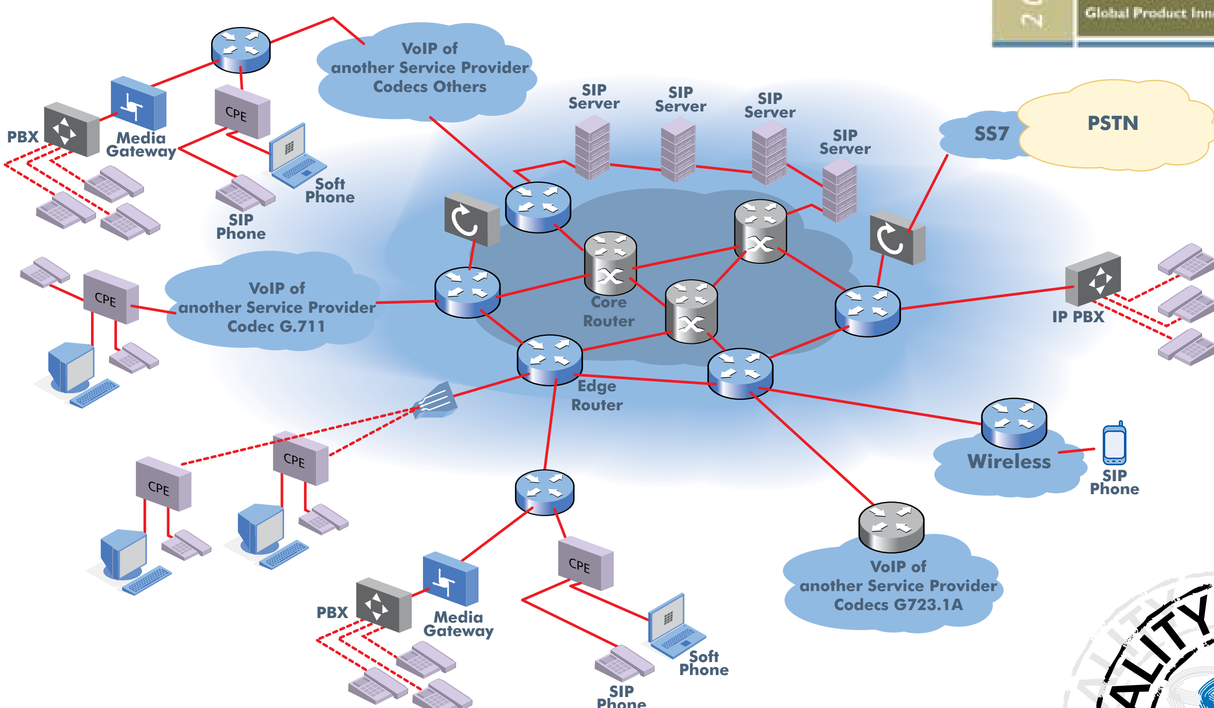

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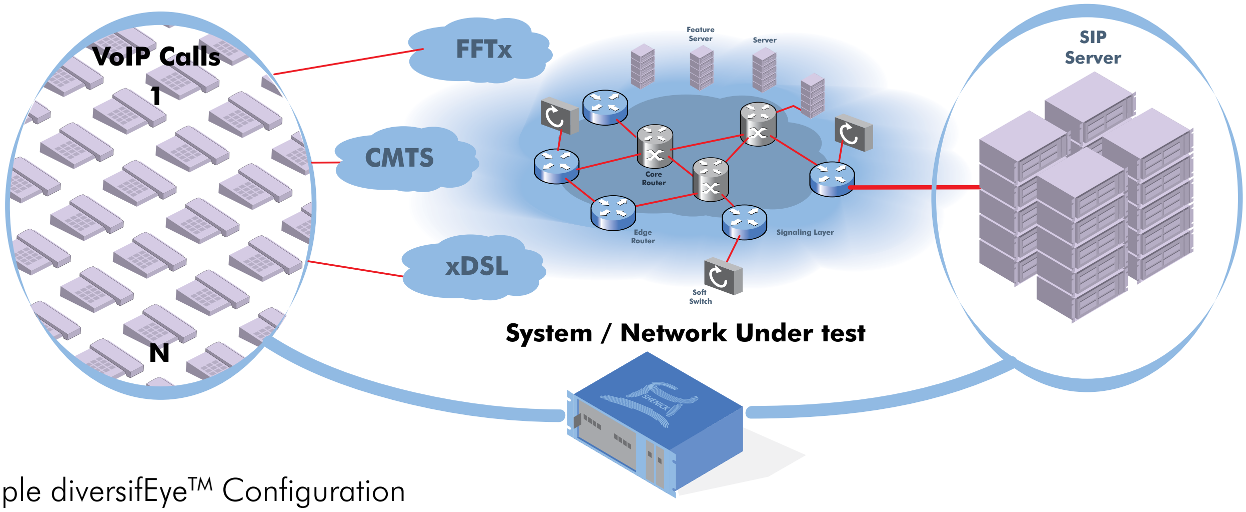
VoIP Test Scenarios

SIP Server Call Rate – Emulate and analyze the call rate of the SIP server, with thousands of VoIP users on the client side. Clients are unable to send or receive RTP media on half duplex connections. Reassess the call rate by allowing RTP media and full duplex connections.

VoIP quality in Triple Play service-
Access network test to support Triple Play (DSLAM/FTTH/CMTS, Residential Gateways, BRAS etc.). Determine VoIP performance under converged IP scenarios, determine threshold performance by increasing VoIP and other IP applications to a point where packet loss effects call quality and establishment rates.

DDoS Attack – Effect of Network Attacks on VoIP quality and performance. Examine impact of SIP based attacks along with DDoS, Viruses on individual and collective VoIP calls.



Sample diversifEye™ Configuration

Software Specification

- RFC 3261 and RFC 2327 (Session Description Protocol) Support
- User Agent Client and User Agent Server Support
- External SIP Servers and IP Phones supported
- Calls are routed for:
  - UAC to UAC to test network performance
  - UAC to UAS to test network and server performance (External or diversifEye™ UAS)
- Three modes of operation:
  - Normal behaviour
  - Sustained Calls – Low number of SIP packets, Large number of RTP packets
  - High Call Rate – High number of SIP packets, Low number of RTP packets
- SIP Client will Register, Initiate and Accept SIP Calls
- SIP Digest support for Authentication (Registration, Invite, Bye)
- SIP Batches for many User Agents with configurable Calling Lists
- SIP Attack Emulation
- Configurable CODEC template with flexible Sample Period, Frame Size and Packet Rate
- G.711, G.729, G.723a and others templates pre-configured
- Real Voice Samples available and configurable
- Independent SIP and RTP Statistics
- RTCP Support
- Flexible port assignments for both SIP and RTP flows

KEY FEATURES AND BENEFITS

- Access Node Independent.
- Latency, Jitter, Packet Loss Quality of Service metrics for overall network/device under test.
- Quality of experience metrics available on a per individual VoIP end-user.
- Choice of CODECs - G.711, G.729, G.723a. Configurable CODEC template available.
- Generate SIP based attacks coupled with full DDoS attack emulation, SYN/RST/UDP/ARP floods, Reflective DDoS attacks, Ping of Death, Teardrop, etc. Generate both attack traffic and regular application flows concurrently.
- Management and Configuration interface for external SIP servers.
- Captured file replay functionality (TCP/UDP).
- Single Integrated Test Solution, avoids the inconvenience and expense of multiple unintegrated test platforms and software applications.
- Test VoIP under multiple converged IP network scenarios such as Triple Play. Test VoIP performance and quality of experience under concurrent Voice, Video and Data application generation conditions.

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