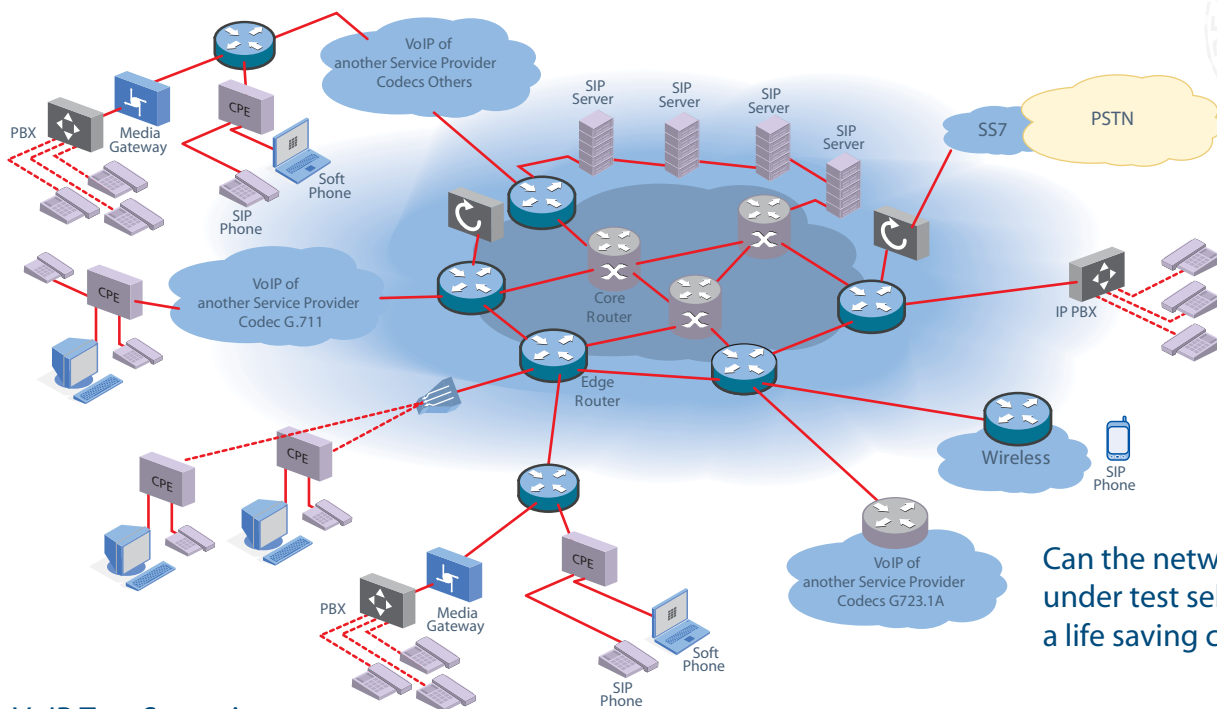
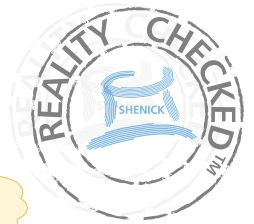




Testing VoIP with diversifEye™

Network benchmarking tests such as throughput or busy hour calling attempts are fine for network limitation measurements, but offer no information or valid data in terms of actual end point quality of experience on the per individual call basis. diversifEye's Per flow architecture delivers these crucial performance measurements.

To improve or add to existing test methodologies, it's necessary to include Quality of Experience performance measurements. Essentially, yes we want to see how the network will cope at various stages of the call lifecycle, and with various user scenarios, but more importantly how will these performance measurements impact the experiences on an individual emulated end point?



Can the network or device under test select and prioritize a life saving call?

Sample VoIP Test Scenarios

Per Call Analysis -

Emulate various call scenarios with differing codecs and numbers, verify if each and every call type is identified correctly in the network and prioritized accordingly.

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.

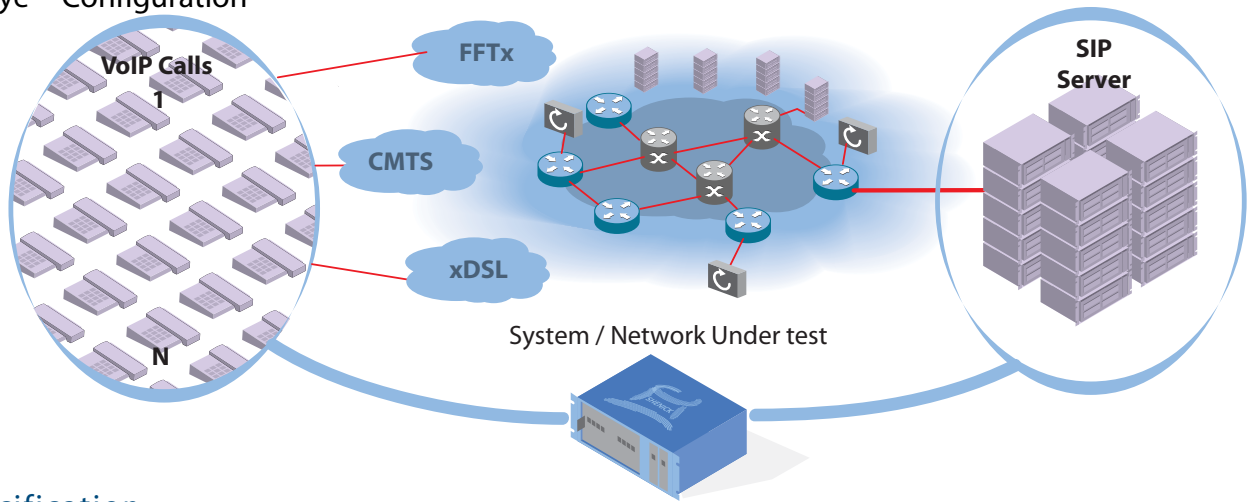
VoIP quality in Triple Play service -

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.



In the real world, voice conversations are generally two way or bidirectional therefore it should be considered as a pre-requirement for our any testing that the flows are stateful i.e. understand and respond to transport layer requests. Traffic management policy may include TCP window shaping, for improved VoIP quality. It's essential that the emulated end point on the test scenario is capable of conversing or negotiating this window re-sizing.

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 behavior
 - 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, Call session types include voice or multimedia with video
- SIP Digest support for Authentication (Registration, Invite, Bye)
- SIP Batches for many User Agents with configurable Call Lists
- Support for SIP enabled VoD and HTTP sessions
- Configurable CODEC template with flexible Sample Period, Frame Size and Packet Rate
- SIP IPv6 transition test strategies, dual IPv4/IPv6 stack functionality on emulated UACs
- Real Voice Samples available and configurable
- Flexible port assignments for both SIP and RTP flows, RTCP Support, Independent SIP and RTP Statistics
- B2BUA (RFC3261) implementation for NAT boundary traversal
- Real-time Voice Quality metrics

diversifEye Summary 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).
- Telepresence, measure performance on all 4 unique media flows associated with each and every emulated end-point.
- IPv6 migration test strategy, Dual-Stack Lite implementation.
- 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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