Proactively Test your VoIP Service to Improve Customer Satisfaction
Tektronix’ SIP Loopback Test Agent can monitor and troubleshoot VoIP service quality as perceived by subscribers by remotely testing to SIP end devices. No field test equipment is required. This test provides over 60 VoIP QoS measurements that allow operators to test their VoIP service delivery, proactively monitor their end-subscriber’s service, troubleshoot, and isolate VoIP issues throughout the service life cycle.

VoIP Service Delivery
A remote loopback test can be performed on the VoIP end device to create a "birth certificate" that confirms the service is ready to go.

Proactive Monitoring
Use proactive loopback testing to cover your full subscriber base to verify that any changes to your network are having no impact on your end-subscriber’s service as compared to the day it was installed.

Troubleshooting and Fault Isolation
With probes deployed at key network segments, isolate network faults by performing loopback tests to intermediary points throughout the network.

Return on Investment
The reduction of customer service calls due to proactive testing and faster resolution of issues can more than pay for the DirectQuality® system.
True Voice-Path Testing

Tektronix’ loopback tests conduct measurements of true user-perceived service quality. This allows you to identify and resolve problems before your subscribers know about them.

Using a widely used SIP Loopback implementation, Tektronix can perform cost-effective, remote tests to the customer premises with Tektronix PowerProbes that are deployed in key network core or hub site locations. The Speech & DTMF Loopback Test Agent on the PowerProbes can perform tests to SIP-Loopback enabled devices such as IP Phones, Softphones, ATAs, MTAs, and intermediary devices that can terminate VoIP calls.

These end devices can act as a packet or audio reflector. The packet reflector will assess media transmission performance over the VoIP network to the subscriber’s premises. The audio loopback feature tests the device’s internal circuitry (including the codec) to evaluate the true user-perceived speech-quality, including the impairments caused by the codec’s D/A & A/D speech conversion and compression.

Network Fault Isolation with the SIP Loopback Test Agent

Remotely isolate distribution and access network faults that occur between an intermediary point (for example, a Session Border Controller) and another SIP end device served by the same branch by performing a loopback test to each device with our SIP Loopback Test Agent as shown below.
Speech Quality
PESQ LQ MOS
VOES MOS
R-Factor (CQ and LQ)
Speech Power, Loss & Distortion

Noise
C-Message Noise
Wideband Noise
Noise Gain
C-Notch Noise Gain
Signal-To-Noise Ratio

Voice Transmission
Frame Muting Ratio
Comfort Noise
Clipping Events (Front-End, Back-End, & In-Between)
Clipping Ratio (Front-End, Back-End, & In-Between)
Average Clipping Duration (Front-End, Back-End, & In-Between)
Hang-Over Events
Average Hang-Over Time

RTP Statistics
Packets Sent & Received
Packet Loss, Burst, & Gaps
Packets Out of Order & Discarded
RTCP Reporting

Jitter
Average Jitter
Jitter Buffer Size
Jitter Buffer Usage

Delay
Round-Trip Delay

Frequency Response
Loss (1100Hz, 2100Hz)
RSL (1100Hz, 2100Hz)

DTMF Detection & Validation
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Create Reports by origin, destination, city, region, or breakout for any testing period for network monitoring, troubleshooting, and trending

In the QoS Analysis View, User-Defined Service Level Classes are Used to Present Results in Highly-Identifiable Pass / Fail Categories

Fax Tone Detection
CNG Tone Detection & Duration
CED Tone Detection & Duration

Connection Status
Call Disposition Code
SIP Return Code
Answer Seizure Ratio (ASR)
Call Completion Ratio (CCR)
Call Loss Ratio (CLR)
Network Effectiveness Ratio (NER)
Answer Bid Ratio (ABR)
Negotiated Media

NOTE: Test measurement availability varies according to the network protocol the PowerProbe is used with.
Benchmark Competing Technologies with Industry-Standard Speech Quality Algorithms

The Speech and DTMF agent incorporates standards-based VQES and PESQ algorithms that provide quality measurements that are ideal for the benchmarking of competing technologies and services.

**VQES Algorithm**

Monitors the end-to-end quality of your voice services using MCI Labs’ statistics-based Voice Quality Evaluation System (VQES) algorithm. It calculates VQES MOS and Unsatisfied User Ratio, as well as conducting a full connectivity performance analysis.

**PESQ Algorithm**

Assesses the end-to-end quality of voice services using the ITU-T PESQ algorithm, to implement the PESQ Listening Quality MOS, frame muting for packet-loss detection, distortion, and voice clipping.

**DirectQuality Web-Based OSS**

**Advanced Test Automation**

DirectQuality anticipates measurement requirements and will generate and execute testing plans based on your QoS objectives. Automate test plans or start tests on-demand.

**Color-Coded Service Levels**

DirectQuality simplifies the monitoring of service faults by displaying results using user-definable Service Level Classes. Service violations can be forwarded to fault management systems via SNMP.

**Business-level QoS Reports**

DirectQuality provides a set of business-driven report templates with high-level and drilldown views.

About Tektronix:

Tektronix Communications provides network operators and equipment manufacturers around the world an unparalleled suite of network diagnostics and management solutions for fixed, mobile, IP and converged multi-service networks.

This comprehensive set of solutions support a range of architectures and applications such as LTE, fixed mobile convergence, IMS, broadband wireless access, WiMAX, VoIP and triple play, including IPTV.

For Further Information:

Tektronix maintains a comprehensive, constantly expanding collection of application notes, technical briefs and other resources to help engineers working on the cutting edge of technology.

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