November 22, 2004

One Useful Way To Test VoIP Quality

As a part of a multi-vendor IP Telephone review in conjunction with CT Labs, we used Sage Instruments' neat little 935AT portable unit to perform end-to-end call tests to assess overall speech quality. Cool stuff. Here's more.

Sage Instruments' 935AT was one of the key pieces in CT Labs' test setup for our IP Phone Reviews. It allowed us to:

- Automate perceived speech quality tests (done with SMOS test);
- Run a speech clipping/silence suppression test (done with PVIT test);
- Check for comfort noise level (done with SMOS test);
- Perform a latency test (done with SMOS test);
- Do an effective bandwidth test ("percentage of bandwidth available" frequency response of the 300--3400 Hz band -- done with SMOS test).

The 935AT (pictured) comes with common 2-wire and 4-wire analog interfaces as well as a T1 interface with dual direction drop and insert capability and many PCM span diagnostics. Access to testing is via standard MF/DTMF/DP signaling features; there's also built in analysis functions for simplified troubleshooting.

We would imagine it would be a useful piece of equipment in any telecom closet or interconnect installation van.

Sage Instruments SMOS and PVIT tests

All of automated tests were performed with two Sage 935AT Communication Sets. There are two basic suites for our industry:

The SMOS test, which uses a special technique to estimate the perceived quality of speech. As defined by Sage, an SMOS number above 4.0 is considered to be toll quality. An SMOS number between 3.0 to 4.0 is considered to be communication quality (intelligible but unnatural, or could be annoying and lack of speaker recognition, etc). An SMOS number below 3.0 is unacceptable for voice communication.

The SMOS test also measures latency. Sage's proprietary SMOS algorithm measures round-trip delay (latency) over a range of 0.0 to 5000.0 msec with an accuracy of +/- 0.2 milliseconds. The measured delay includes delay from a number of sources including voice processing, packetizing the voice into frames, and the jitter buffer process.

The PVIT test measures speech clipping (useful for verifying voice activity detectors), comfort noise, frame slips, and noise floor performance. You run this test for 15 minutes and it produces the following measurements:

- *Frame slip - Voice Packet Jitter/Jitter Buffer Resizing:* Jitter occurs when a network jitter buffer dynamically readjusts the buffer size. Buffer resizing balances the conflicting aims for less frame loss and shorter end-to-end transmission delay. A positive (contracting) slip occurs when a voice frame is deleted. A negative (expansive) slip occurs when a filler frame is inserted. PVIT time stamps each voice frame slip and measures its duration.

- *Voice Clip-Speech Clipping/Silence Suppression:* PVIT provides detailed diagnostic information on key packet network characteristics that impact voice clarity. It reports a voice clip when the signal corruption is at the leading edge of a voice segment. Each voice clip event is measured for duration and time stamped. A running calculation also presents the average duration of all voice clip events.

- *Noise hit-Comfort Noise:* If, during any individual silence period during the test, the noise level exceeds -45 dBm (45 dBrnC), a Noise Hit event will be reported by PVIT.

The layout for our CT Labs IP Phone Test Bed is pictured below. Each phone connected via 310-type wires (they look like 1/4 inch phono plugs) from Sage 901A Audio Adapter to the respective inputs on the Sage 935AT.

The Shunra device in the middle let us play with network conditions.

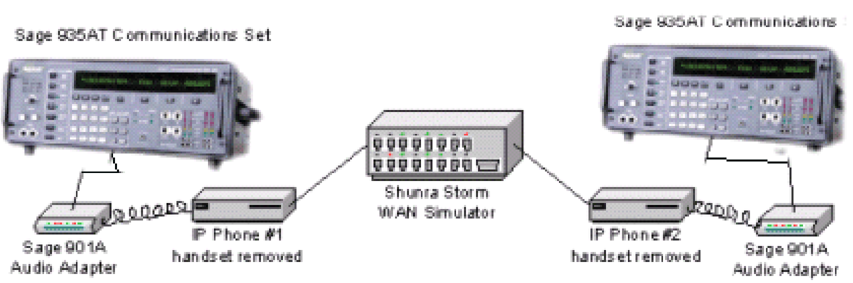

Figure 1 -- Test Setup, IP Phones Speech Quality Test