Testing Voice Service
for
Next Generation Packet Voice Networks

Next Generation voice networks combine voice and data on the same transmission path. The advantages are many, but because of the technology employed in these new networks, service providers face new Quality of Service challenges: Packetized voice transmission adds nonlinear compression and the need for timely packet delivery from networks not originally set up for these conditions. Moreover, the voice encode/decode and related functions insert significant delays in the voice path, creating echo problems that would otherwise not be noticed.

To win and keep customers, Next Generation network providers must ensure their networks minimize these problems. The goal is to provide the same "toll quality" voice transmission that circuit-switched networks furnish today.

Sage Instruments provides a suite of voice quality tests to address this QoS challenge for Next generation networks. Moreover, recognizing that voice encode/decode functions are being pushed closer to the customer, Sage products support ***2- and 4-wire test access*** at the network edge, as illustrated in **Figure 1**.

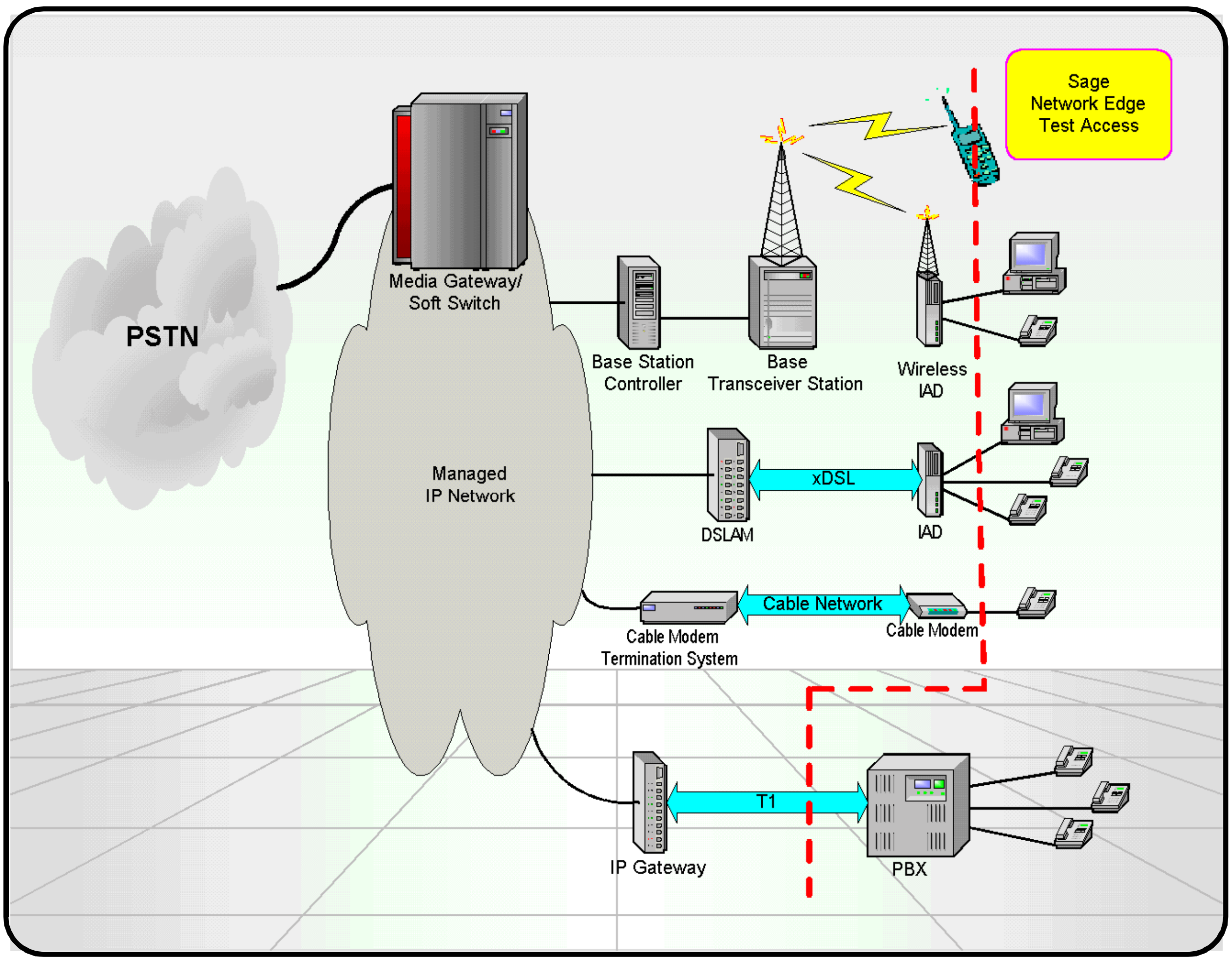

Figure 1 — Sage Test Access Points at Network Edge

Echo

Echo is one of the most important factors affecting voice quality. The degree of annoyance an echo presents depends on both echo loudness and round trip delay from the speaker to the echo point. In VoP (voice-over-packet) networks, the echo problem is aggravated, not because the packet network creates additional echos, but because the extra delay introduced by compressive codecs and packet networks make echo more irritating.

For example, based on **Figure 2**, an echo at -25 dB with less than 10 ms delay is probably not very objectionable to most people. It only adds some reverberant side-tones. But an echo of -30 dB at 100 ms delay will be very objectionable to almost every caller.

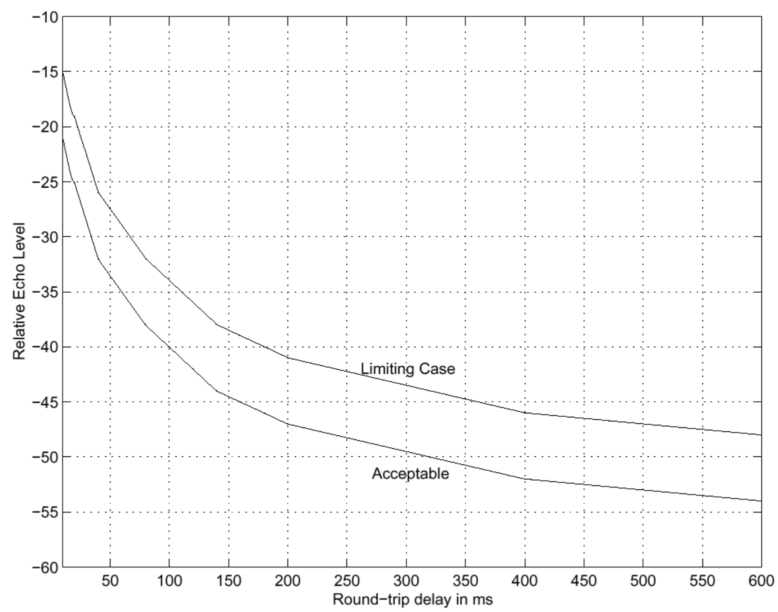

Figure 2 — Echo Tolerance Curve

Sage Echo Sounder Test

The Sage Echo Sounder test measures and characterizes network echoes on a real time basis. Using an advanced test signal to penetrate the network, the algorithm can report up to four signal reflections (echoes). For each echo, it displays real-time echo level and round trip delay.

Network Latency

With the delay sources present in a VoP network, the combined delay can quickly add up to a one-way delay greater than 200 milliseconds. Unfortunately, even with no echo, one-way delays of 150 ms to 200 ms can be a voice quality problem. This is because, in the normal give and take of a conversation, the interval between one talker stopping and the other talker starting is often as little as 200 ms. So, when end-to-end transmission delays approach this amount, it disrupts the normal flow of conversation. It can even affect a listener's perception of a talker's response. For example, the delay in response could be wrongly interpreted as reluctance, uncertainty, or surprise.

The primary sources of delay are:

Voice processing — these are delays introduced by bandwidth-saving devices such as vocoders (for voice compression), and voice activity detectors that squelch voice packet transmission during speech silent periods.

Packetizing — delays that occur as multiple vocoded voice frames are combined into larger, more efficient packets. The problem is, larger packets mean increased delay. For example, a packet size of 40 ms introduces a one-way delay of 40 ms.

Jitter-buffer — delays caused by the buffering of packets to reduce packet jitter and packet loss. VoP packets tend to arrive at their destination out of order and at non-uniform intervals. The receiving end's jitter-buffer waits for late arrivals, and then re-orders the packets. To work properly, the buffer interval must be several times larger than the packet size. For example, a jitter buffer of three 40-ms packets introduces a one-way delay of 120 milliseconds.

SMOS (Sage Mean Opinion Score)
Sage's proprietary SMOS algorithm measures round-trip delay over a range of 0.0 to 5000.0 msec with an accuracy of ± 0.2 milliseconds.

Voice Clarity

The voice processing equipment that causes delay can also affect voice clarity. Although Vocoders and Voice Activity Detectors add to the overall transmission efficiency of a network, they may degrade voice transmission by adding noise and distortion: Vocoder compression algorithms can sacrifice quality and clarity to minimize bandwidth requirements

SMOS (Sage Mean Opinion Score)
Sage's SMOS test uses an artificial voice signal to test voice clarity. The MOS number provides an objective evaluation of the overall speech clarity associated with the call under test. A MOS score greater than 4.0 is toll quality; below 3.0 is unintelligible.

Circuit Loss

To achieve great voice quality, network providers must choose a fine balance between providing enough loss to attenuate echo and maintain circuit stability, while still being comfortably audible.

SMOS (Sage Mean Opinion Score)
The SMOS test reliably reports circuit loss over a range of -80 to +20 dB through live packet networks and real world compressive codecs.

Speech Clipping/Silence Suppression

In an effort to reduce network traffic, silence suppression is used to eliminate voice packet transmission during silent talker periods. However, some silence suppression algorithms may clip the leading edge of each active voice segment causing voice to sound choppy or abrupt.

PVIT (Packet Voice Impairment Test)
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.

Comfort Noise

Comfort Noise is inserted during silent periods to make conversation more "natural' and similar to a normal phone call. But if comfort noise is too loud, it sounds like a "bad connection," if too low, the circuit sounds "dead."

PVIT (Packet Voice Impairment Test)
For networks that employ silence suppression, PVIT measures the comfort noise level during quiet periods. “Noisy” silent periods are time stamped and recorded, and a running calculation presents the average comfort noise.

Packet Loss

In data traffic, a lost packet can be re-transmitted. But in voice traffic, a lost packet can't be resent later. Such packet losses can cause momentary voice breakups or dropouts, and even call disconnections.

A packet loss can occur due to the following reasons:

Packet arrives too late — If a packet arrives later than a jitter buffer can tolerate, this packet will be treated as a lost packet. Late packets usually occur during peak traffic periods because of queuing delays as a packet sits in queue behind other packets waiting for transmission. Newer equipment allows an administrator to specify that voice traffic has priority during network congestion. This prioritization is typically achieved through class of service features.

Packet misrouted — If a packet is mis-routed to nowhere and never arrives at its destination, it will of course be treated as a lost packet.

Errored-packet — Under certain harsh transmission environment (wireless or xDSL, for example), if the bit errors exceed the correctable amount inside a packet, this packet will be treated as a lost packet.

PVIT (Packet Voice Impairment Test)
Voice frame loss events are measured and time stamped. It also calculates and presents the % frame loss rate.

Figure 3, below illustrates two examples of packet loss.

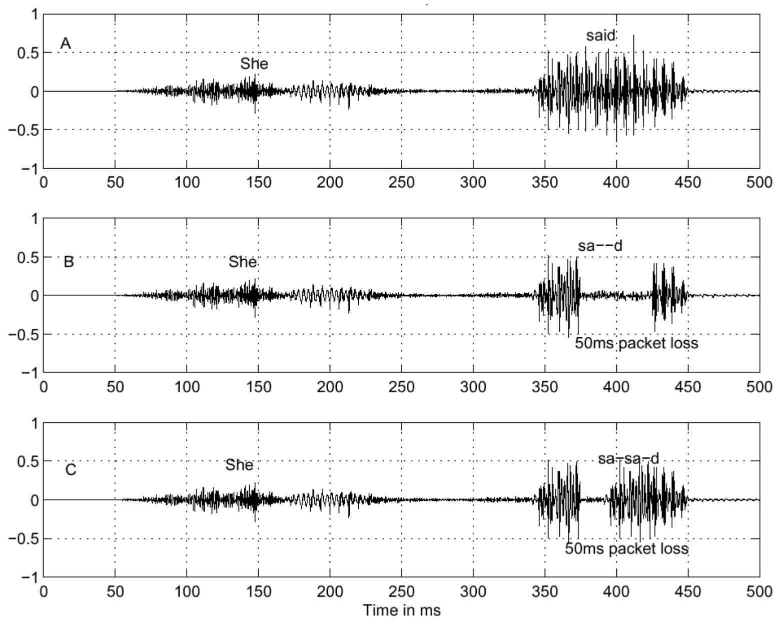

Figure 3 — Packet Loss Examples

A — A snapshot of male speech saying "She said."
B — A 40 ms block of speech signal centered at 400 ms time mark (horizontal axis) is replaced by "comfort "noise due to packet loss.
C — A 40 ms block of speech signal centered at 400 ms time mark is replaced by its previous 40 ms block of signal due to packet loss.

Cases B and C will be correctly measured by PVIT as a packet loss event with duration of 40 ms and time stamp around 400 ms.

Note that short, continuous packet losses are worse for voice quality than random packet loss spread over time. In other words, losing ten sequential packets is worse than losing ten packets evenly spaced over an hour time span. PVIT's time/date stamped events show whether packet losses are bursty or random.

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 (Packet Voice Impairment Test)

PVIT time stamps each voice frame slip, and measures its duration. It also continually calculates the net slips (positive or negative). **Figure 4**, below, illustrates the two types of slips (jitter).

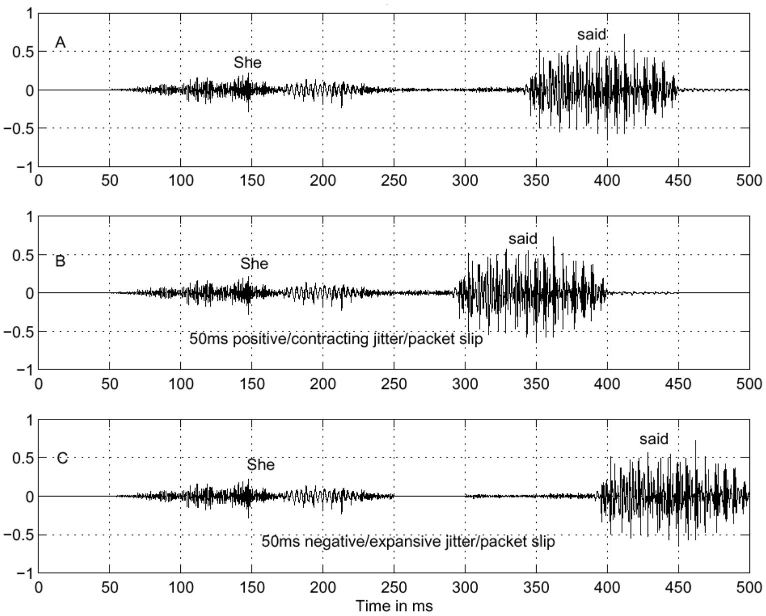

Figure 4 — Jitter/Slip Examples

A — A snapshot of male speech saying “She said ”

B — A 50 ms block of signal centered at 300 ms time mark is annihilated (deleted),causing the relative delay between the word “She ”and the word “said ”to be shrunk by 50 ms. This will be measured by PVIT as a positive packet slip (jitter)of 50 ms with a time stamp around 300 ms.

C — A 50 ms block of silence is “forcefully ”inserted into the signal centered at 275 ms time mark, causing the relative delay between the word “She ”and the word “said ”to be increased by 50 ms. This will be measured by PVIT as a negative packet slip (jitter)of 50 ms with a time stamp around 275 ms.

Codec Type

Codec type detection serves several purposes:

1. You can use it to establish a reference MOS number. For example, is MOS 4.15 a good number for this network under test? Without codec type information, one cannot answer such question. But if the codec type is known, one can then compare the measured MOS with the “theoretical” ideal number in Table 2. For example, if the codec is G.711 PCM, then MOS 4.15 is a very bad number, indicating there might be serious packet loss or coding problems. But if the codec is a typical 8kbps vocoder, then MOS 4.15 is just about right.

2. Verify Service-Level-Agreement (SLA) and network configuration. If a network is configured (based on SLA) to use G.711 PCM, not G.726 ADPCM, such SLA configuration can be verified by SMOS’s codec type detection.

3. Troubleshoot codec transcoding problem. For a “hybrid ”long distance network, codec transcoding is not only a problem for voice quality, it also poses a dilemma for network management and SLA. More specifically, a call may start with G.726 ADPCM encoding. But in “middle ”of the network, it may become G.723.1 vocoder. At the end, it becomes G.726 ADPCM again. The presence of G.723.1 transcoding in such case can be verified with

SMOS's codec type detection. In this case, SMOS will report the G.723.1 vocoder, instead of the ADPCM.

Table 1 summarizes all the codec types that SMOS can detect and report:

Codec Type Report Symbol	***Description***	***Reference MOS Range***
VCD4K	Sub-4kbs vocoders	[3.0,3.8)
VCD8K	5-8kbps vocoders	[3.8,4.2)
VCD16K	12-16kbps vocoders	[4.2,4.35)
ADPCM16	16kbps G.726 ADPCM	[3.4,3.6]
ADPCM24	24kbps G.726 ADPCM	[3.9,4.1]
ADPCM32	32kbps G.726 ADPCM	[4.2,4.3]
ADPCM40	40kbps G.726 ADPCM	[4.3,4.4]
ADPCM	G.726 ADPCM with unknown data rates	[3.5,4.3]
PCM	G.711 μ/A-law PCM or pure analog	[4.45,4.60]
UNSURE	Too much distortion, not sure	N/A

Table 1 — Codec Types Reported

Effective Bandwidth

This parameter quantifies the attenuation distortion (frequency response) of the system under test, over the 300 to 3400 Hz band.

If a system under test uses waveform codecs (G.711 PCM or G.726 ADPCM), its measured effective bandwidth should be higher than 0.9. Anything below 0.85 signifies either excessive loop attenuation distortion (or poor analog circuitry design if testing through analog connection) or excessive band-limiting digital filtering.

On the other hand, if the system uses non-waveform low-bit-rate vocoders, then one must be careful in interpreting the value as shown in Table 2. Generally, the effective bandwidth should be maintained above 0.7. Anything below 0.65 indicates poor analog circuitry design (if testing through an analog connection) or excessive digital filtering.

Codec type	Theoretical Effective Bandwidth Reading
G.711 PCM@64kbps	1.00
G.711 PCM robbed-bits	1.00
G.711 PCM@56kbps	1.00
G.726 ADPCM@40kbps	1.00
G.726 ADPCM@32kbps	0.99
G.726 ADPCM@24kbps	0.97
G.729@8kbps	0.77
G.723.1@6.3kbps	0.88
Cell-phone VSELP@8kbps	0.85
Cell-phone EFR-ACELP@7.4kbps	0.87

Table 2 **—** Theoretical Effective Bandwidth Readings (Codec Only)

Call Completion Time

Because a VoP call often involves signaling hand-offs between IP networks and IP-to-PSTN networks, it is important to assess whether unreasonable call-setup delays are taking place.

The most complete picture of call set-up would measure the elapsed time between the last dialed digit, and an audible answer from the called end. The SMOS test does exactly that: It measures the elapsed time from the last dialed digit to detection of far end responder answer tone. It reports the call completion time in seconds, and tenths of a second, and includes the date and time of day at which the call was completed, according to the test system's internal clock.

What Sage Test To Use?

Use **Tables 3** and **4**, below, as a quick reference to the Sage Instruments test function that best fits the parameter to be tested.

Parameter	***Sage Instruments' Test Function***
Echo/Return Loss/Canceller Operation	Echo Sounder, Echo Generator
Network Latency	Echo Sounder SMOS (Sage Mean Opinion Score)
Voice Clarity	SMOS (Sage Mean Opinion Score)
Circuit Loss	SMOS (Sage Mean Opinion Score)
Speech Clipping/Silence Suppression	PVIT (Packet Voice Impairment Test)
Comfort Noise	PVIT (Packet Voice Impairment Test)
Packet Loss	PVIT (Packet Voice Impairment Test)
Packet Jitter/Jitter Buffer Resizing	PVIT (Packet Voice Impairment Test) SMOS (Sage Mean Opinion Score)

Table 3 **—** VoP Parameter vs Sage Test Function

Parameter	***Sage Instruments' Test Function***
Report Codec Type	SMOS (Sage Mean Opinion Score)
Effective Bandwidth	SMOS (Sage Mean Opinion Score)
Call Completion Time	SMOS (Sage Mean Opinion Score)

Table 4 **—** Other Useful Parameters for In-Service Testing