Voice Quality Monitoring for VoIP Networks

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Introduction

Voice over internet protocol (VoIP) is the newest link in the telecommunications evolutionary chain. Routing both data and voice over a common, inexpensive IP network is realising significant financial advantages to those adopting the technology. Enterprises, call centres, long-distance carriers and home users are just some of the customers embracing the new approach to voice communications.

In order to realise the full potential of VoIP technology, it is essential that voice quality is maintained. However, with the migration from the public switched telephone network (PSTN) to mobile and now to VoIP telecommunication systems, the focus has been on ever decreasing bandwidth allocation for voice channels. This has created a number of new voice quality issues, and exacerbated existing issues. Further, with an ever increasing number of networks and service providers, it is essential to ensure voice quality is maintained at the boundaries between networks.

For service providers, voice quality is critical to their business. Customers are very intolerant of poor voice quality, and will move to service providers with superior quality. With superior voice quality, VoIP service providers will receive a direct benefit of increased revenue through:

- Increased migration of customers from PSTN and mobile networks
- Increased network usage by existing customers
- Decreased customer churn

This paper describes the voice quality issues faced by VoIP service providers, and how they can proactively manage voice quality within their network. Results are presented showing how the deployment of non-intrusive voice quality monitoring equipment enables critical early warning when there is a network problem, and allows service providers to pin-point and eliminate the causes of voice quality degradation. It also enables evaluation of the performance of voice quality enhancement equipment within the network.

Historical Perspective of Voice Quality

The PSTN Network

Within the PSTN network, there are only 2 significant sources of voice quality problems – delay and echo – and these are both well understood and managed. Excessive network delay, being the time taken for voice to travel from the talker's mouth to the listener's ear, can cause an adverse impact on the quality of a telephone conversation. Within the PSTN, delays are in general very short, and therefore not noticed. The major exception to this is for long distance calls transmitted by satellite link where delay can become annoying, or result in a "half duplex" conversation where each party has to indicate to the other when they have finished talking. Table 1 shows the impact of network delay on conversation quality.

End-to-End Delay (ms)	Affect on Conversation
< 100	Inaudible
100 – 150	Audible but not distracting
150 – 200	Distracting but conversation continues
> 200	Half-duplex conversation

Table 1 Affect of Delay on Conversations [1]

Echo, the second major cause of PSTN voice quality problems, occurs when the speaker hears a delayed version of their own voice. The echoed version of the speaker's voice is most commonly generated by the hybrid located in telephone handsets. Echo at a very short roundtrip delay (the total delay from the speaker's end of the connection to the listener, and then back to the speaker) does not affect the speaker's ability to conduct a conversation. However echo at a long roundtrip delay, say 200ms, is greatly disruptive for the speaker and can prevent the speaker from talking normally. Some indicative roundtrip delays are shown in Table 2. Typically, long-distance calls with a round-trip delay exceeding 25 ms will be routed through an echo canceller.

Roundtrip Delay (ms)	Affect on Speaker
< 20	Inaudible
> 25	Echo cancellers required
> 32	Interferes with speaker

Table 2 Affect of Roundtrip Delay with Echo [1]

Voice quality on the PSTN is so mature that it (i.e. "toll quality") has become the benchmark to which all other voice technologies are compared.

Mobile Telephony

The introduction of mobile (or cellular) telephony both introduced new voice quality problems and exacerbated existing problems. The introduction of voice coder/decoders (codecs) and a wireless link resulted in increased network delay, meaning that echo became an issue that had to be addressed on every mobile call. As a result, every mobile call passes through centralised echo cancellation equipment within the telephone exchange, previously only required for long distance PSTN calls.

In addition to the delay introduced, mobile telephony introduced a range of additional voice quality issues, the most significant ones being:

- A reduction in basic voice quality and increased distortion due to the use of low bit rate codecs.
- Interruptions and drop-outs due to wireless signal fading.
- Background noise due to the environment in which users could now make calls (pubs, train stations, cars, shopping centres etc)
- Double talk muting, where the other party may appear to drop out when both parties speak at once.

As a consequence, the voice quality of a mobile telephone network is never as good as the "toll quality" benchmark of the PSTN.

Voice Quality on Voice over IP

The introduction of VoIP into the telecommunications network has again both introduced new voice quality issues and exacerbated existing issues. The integration of VoIP networks with existing PSTN and mobile networks is illustrated in Figure 1, where MG represents the "media gateway" between the IP and PSTN networks.

In order to become widely adopted, VoIP providers must demonstrate that voice quality comparable with PSTN can be achieved, particularly for demanding business users. The use of even lower bit rate codecs,

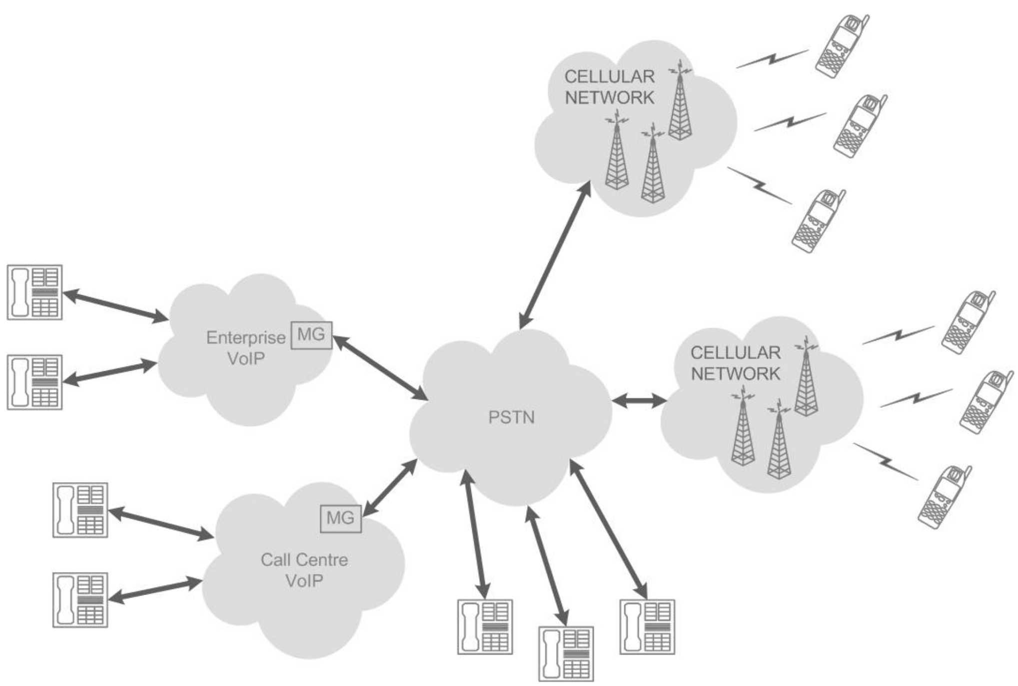

Figure 1 The PSTN, VoIP and Cellular Networks

and the inherent long and variable network delays with an IP network have resulted in increased pressure on voice quality. Table 3 gives a summary of the significant voice quality issues that can exist in a VoIP network. In Table 3 VQE refers to voice quality enhancement equipment, incorporating such functions as echo cancellation, noise reduction and level control, and CPE refers to customer premises equipment.

Voice Quality Fault	Potential Causes	Impact
Distorted, garbled, warped voice	Codec, IP network	Listener may not understand or recognise the speaker
High echo levels	CPE, VQE	Speaker hears own speech, normal conversation is difficult
Front-end clipping	Codec, VQE	Listener doesn't hear the first syllable of speech bursts
Doubletalk muting	VQE	Each party may think other party has dropped out, uncertainty as to whether other party has heard. Creates confusion, repetition.
Silence or comfort noise suppression	Codec, VQE	Speaker gets the feeling that the listener is no longer on the phone
Low or varying speaker volume	CPE, Codec, VQE	Listener has to strain to hear what is being said, and will often shout when it is their turn to talk. The speaker is often asked to repeat what was said or talk louder.
Background noise	Environment, Codec, VQE, IP network	Listener has to strain to hear what is being said, and will often shout when it is their turn to talk. The speaker is often asked to repeat what was said or talk louder.
Long delays from end-to-end	IP network	Users are constantly speaking over each other. Conversation is difficult and tends becomes half-duplex
Varying bursts of noise and/or distortion	IP network, codec	Listener strains to hear what is being said. Speaker often has to repeat what they said.
Occasional, unexpected quiet periods during speech	IP network, codec	Listener asks the speaker to repeat.
Unexpected sounds such as beeps and tones	VQE, codec	Listener strains to hear what is being said. Speaker often has to repeat what they said.

Table 3 VoIP Voice Quality Issues

As can be seen from Table 3, there are several potential sources of voice quality problems in VoIP networks. One of these issues, codec performance, is illustrated in Table 4. Table 4 gives an indicative "Mean Opinion Score" (MOS) for various codecs. The MOS is the voice quality rating on a 1 to 5 scale, 1 being poor and 5 being excellent. In Table 4, G.711 is the wideband codec used on the PSTN (A-law and μ-law), with a MOS 4.1 regarded as "toll quality".

Codec	Bit-rate (kbps)	Indicative MOS
G.711 PCM	64	4.1
G.726 ADPCM	32	3.85
G.728 LD-CELP	16	3.61
G.729 CS-ACELP	8	3.92
G.729 x 2 Encodings	8	3.27
G.729 x 3 Encodings	8	2.68
G.729a CS-ACELP	8	3.7
G.723.1 MP-MLQ	6.3	3.9
G.723.1 ACELP	5.3	3.65

Table 4 VoIP Codecs [2]

It is clear that codecs with lower bit-rates degrade voice quality, and furthermore, if a speech signal needs to pass through multiple codecs, the voice quality will degrade with each codec as shown in the table. Such a scenario often occurs as a connection traverses two or more networks.

A second significant source of voice quality issues highlighted in Table 4 is VQE equipment, which can lead to severe degradation in voice quality due to residual echo and double talk muting. Both codec and VQE performance can lead to performance degradation that cannot be determined by observation of network delay/jitter and packet loss statistics.

Proactive Management of Voice Quality

Proactive management of voice quality provides network operators with a number of significant benefits, including:

- Improved fault diagnosis and resolution.
- Assessment of compliance with service level agreements (SLAs), which may specify a number of mandatory voice quality performance metrics to which the network operator is bound.
- Assessment of impact of changes made to network equipment or configuration.
- Overall network health monitoring.
- Monitoring of voice quality at boundaries between networks.

While there are several options for implementing voice quality management in the network, this paper focuses on the use of non-intrusive voice quality monitoring probes, which can be deployed throughout the PSTN, mobile and VoIP networks as illustrated in Figure 2.

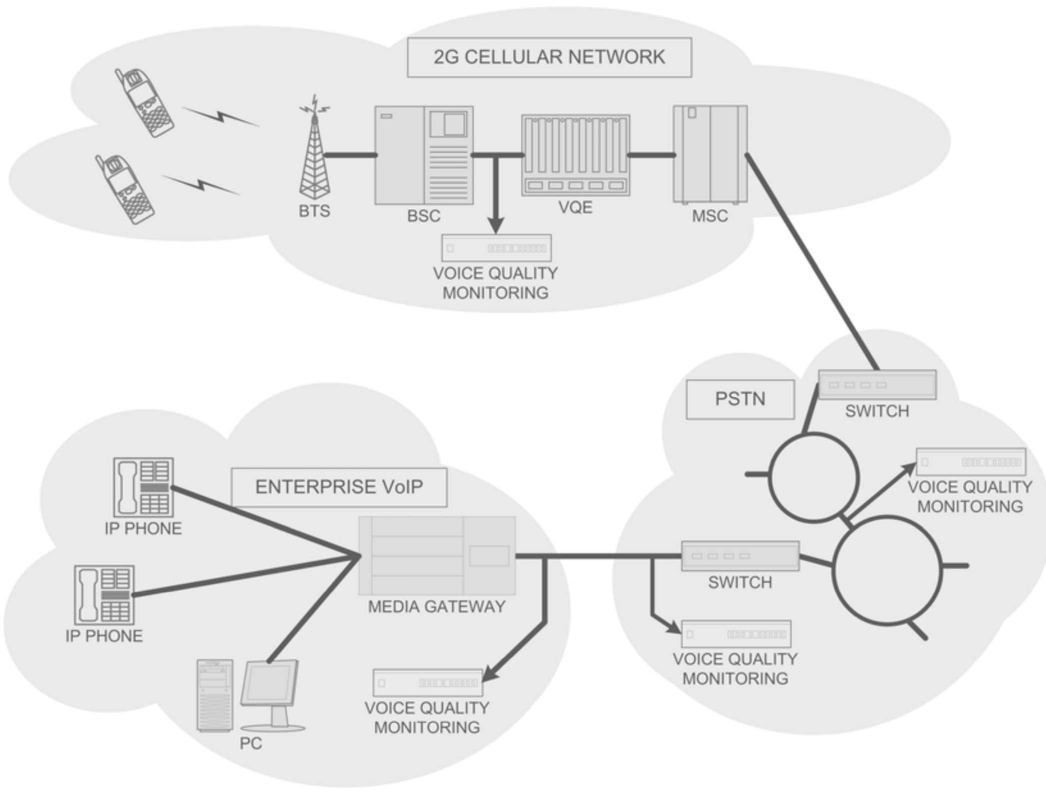

Figure 2 Non-Intrusive Voice Quality Monitoring

The non-intrusive probes, deployed in the transmission path, provide significant advantages over alternative approaches to network testing, including:

- Fully automated testing. No need for equipment at network endpoints or manual intervention.
- Voice quality is measured on the users' actual voice, not using indirect measures such as packet delay/jitter and packet loss.
- Voice quality is measured around the clock, enabling rare events to be detected.
- Potential for assessing every call leads to greater accuracy of results.
- Correlation with network load and time of day are easily established.
- As the voice quality monitoring is non-intrusive, network capacity is maintained.

Results of Voice Quality Monitoring

While the non-intrusive voice quality monitoring probes illustrated in Figure 2 provide measures of many aspects of voice quality, the value of two of these measures will be illustrated here. Figure 3, shows MOS measurements over 1000 calls, and the distribution of those MOS scores. Degradation in MOS as illustrated by the trend curve demonstrates the presence of a network fault. Upon correction of the fault, voice quality returns to normal. In the absence of voice quality monitoring, such a fault may go undiagnosed for an extended period of time, leading to customer dissatisfaction.

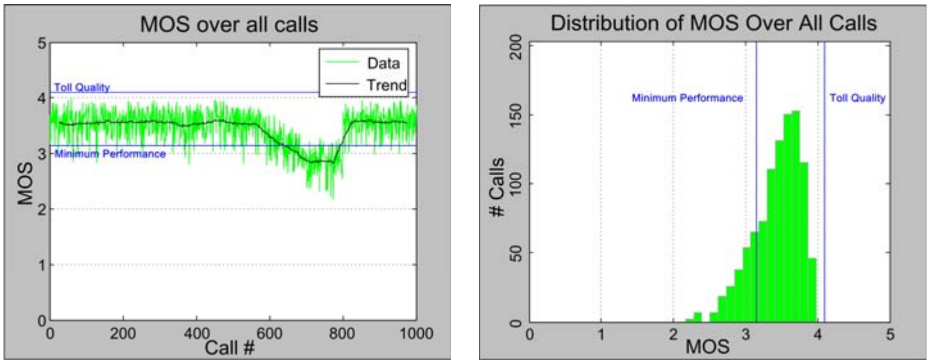

Figure 3 Mean Opinion Score Results and Distribution

Figure 4 shows the results of residual echo measurement. The presence of occasional high levels of echo may suggest occasional divergence of the echo canceller. The data also shows a potential equipment fault, leading to a persistent degradation. Again, proactive management of voice quality enables network operators to identify and resolve such problems.

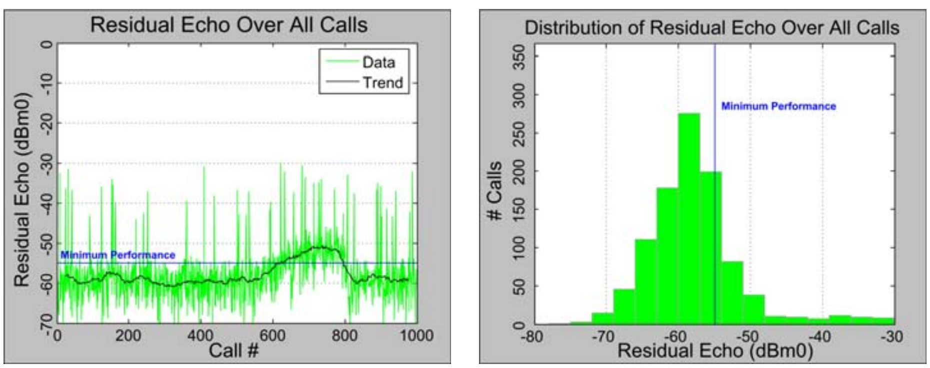

Figure 4 Residual Echo Results and Distribution

Conclusion

The emergence of VoIP has provided significant cost savings for those adopting the technology. However, in order to maximise the benefits for existing users, and accelerate adoption of VoIP, it is necessary to provide assurance of voice quality to users of the technology.

There are significant voice quality challenges inherent in VoIP networks, relating to packet delay/jitter and packet loss within the network, and to the performance of low bit rate codecs and voice quality enhancement equipment.

Proactive management of voice quality through the deployment of non-intrusive voice quality monitoring probes has been shown to deliver significant benefits to network operators, ultimately leading to increased revenue through greater customer satisfaction.

References

[1] Oliver G, “Understanding and Managing Delay in the Global Voice Network”, PMC-Sierra webinar, Dec 2003

[2] “Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation”, Cisco Systems Inc., 2003

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