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Application Notes

Title **Echo in Voice over IP Systems**

Series VoIP Performance Management

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Overview

This application note describes why echo occurs, what effects it has on voice quality, how echo cancellation and suppression work and how to troubleshoot echo problems.

Contents

Introduction

What is Echo?

Echo is an obvious and very annoying problem in telephony systems, and can occur in Voice over IP, Cellular and long distance connections.

There are two major types of echo:

- Talker Echo (Figures 1 & 3) occurs when a proportion of the talker's (i.e. person speaking) voice is reflected back to them. The talker hears a delayed copy of his or her own voice.
- Listener Echo (Figure 2) occurs when a talker's voice is reflected back to them and then re- reflected again towards the listener. The listener hears two or more copies of the talker's speech. Listener Echo is less common than Talker Echo.

Echo is a common problem in Voice over IP services - <u>not</u> because VoIP introduces echo, but because VoIP *increases delay* and makes echo more obvious and annoying.

There are techniques that can be applied to reduce echo problems, such as echo cancellation and echo suppression, however these are not always effective.

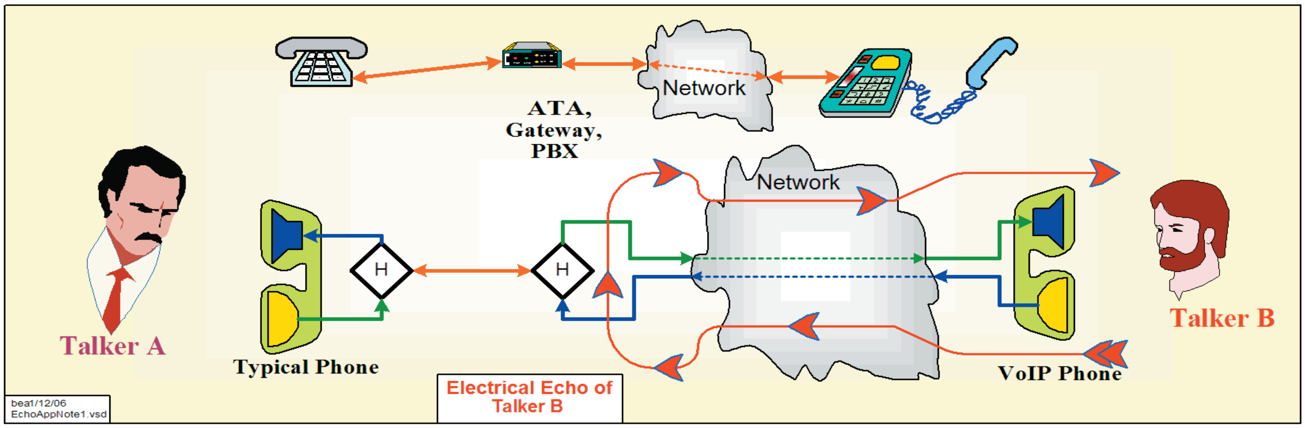


Figure 1: Hybrid Talker Echo

Sources of Echo

There are two common causes of echo:

- Reflections in 2-4 wire interfaces.
- Acoustic Echo

Some "echo" is deliberately introduced in telephone systems. In a typical telephone handset, a proportion of the speech energy from the microphone is fed back to the earpiece. This provides a natural way to control the loudness of the talker - if someone speaks very loudly; this results in a loud signal being fed back to their ear. This deliberate feedback is called "sidetone". Because the signal is fed back instantaneously, it does not sound like echo, which by definition is delayed with respect to the original speech.

Electrical Echo

Both Voice over IP and traditional digital telephone systems (PCM, ISDN) are "4-wire" in the sense that the signal in one direction is carried over a separate "pair" of wires and the signal in the other direction on a separate "pair" of wires. This means that the two signals are independent of each other.

Analog local loops typically used to connect to individual telephones, are "2-wire" as the signals going in both directions are carried over the same pair of wires. Where a 4-wire digital system connects to a 2-wire analog system, it is necessary to perform a *2-4 wire conversion* using either a *transformer hybrid or active hybrid.* This conversion function is typically built into Central Office or PBX line cards, or into channel banks.

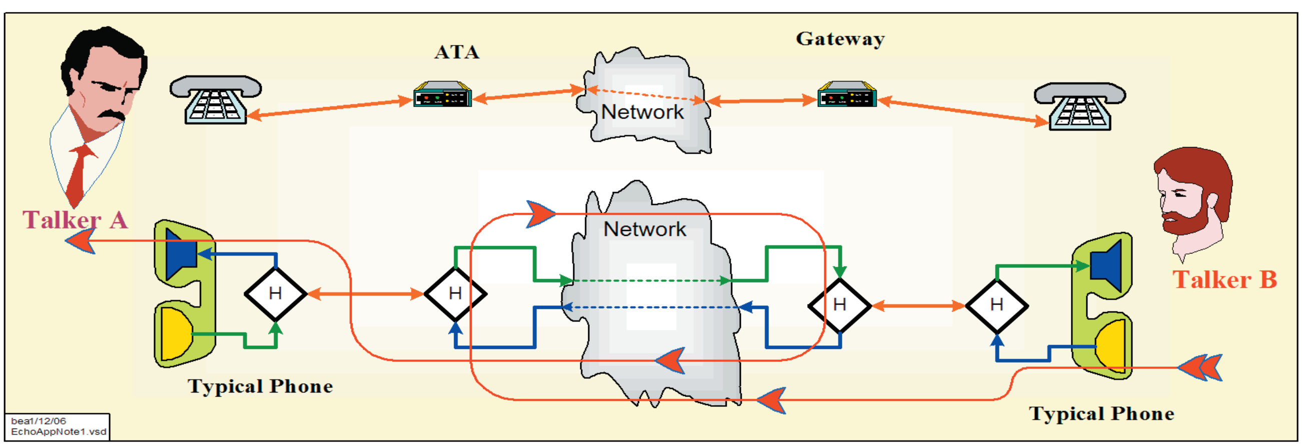


Figure 2: Listener Echo

The 2-4 wire conversion process typically relies on the hybrid being "balanced", which means that the loading presented by the 2-wire line matches that expected by the hybrid. If there is some mismatch, the transmit and receive signals on the 2-wire line cannot be properly separated and hence an echo occurs.

Echo is a very common problem on PCM-analog loop interconnections. However, with conventional telephone systems the delay is so short that the echo does not sound like echo, it sounds like sidetone.

Acoustic Echo

Acoustic echo (Figure 3) occurs when some proportion of the sound coming out of the "speaker" part of a telephone handset or headset can be heard by the microphone part of the handset or headset. This can be due to poor design, or even to the user holding the handset away from their ear.

Impact of Echo on VoIP Call Quality

Talker echo is the most common type of echo and results in a proportion of the talker's (person speaking) voice being reflected back to them. The discussion below primarily relates to this type of echo.

Echo is typically reported in terms of Echo Return Loss (ERL). This is the ratio between the original signal and the echo level expressed in decibels (dB). A higher ratio corresponds to a smaller echo, hence a 55 dB echo return loss would be a low echo level and 15 dB quite a high echo level.

The chart on the following page shows the relationship between delay and conversational quality for two conditions - firstly with a low level of echo (55 dB echo return loss) and secondly with moderate level of echo (35 dB echo return loss).

If round trip delay is very short, say less than 30 mS, then the talker cannot distinguish between the echo and the deliberately introduced sidetone.

If the delay is a little longer, say 50 mS, then the talker cannot hear the delayed copy of their speech as a distinct copy, however it does impact speech quality, resulting in a sound quality generally described as "hollow", "cave-like", "tunnel-like" or similar.

As the delay increases further, the echo becomes more obviously echo - and the combined effect of the loudness of the echo and its delay cause considerable annoyance.

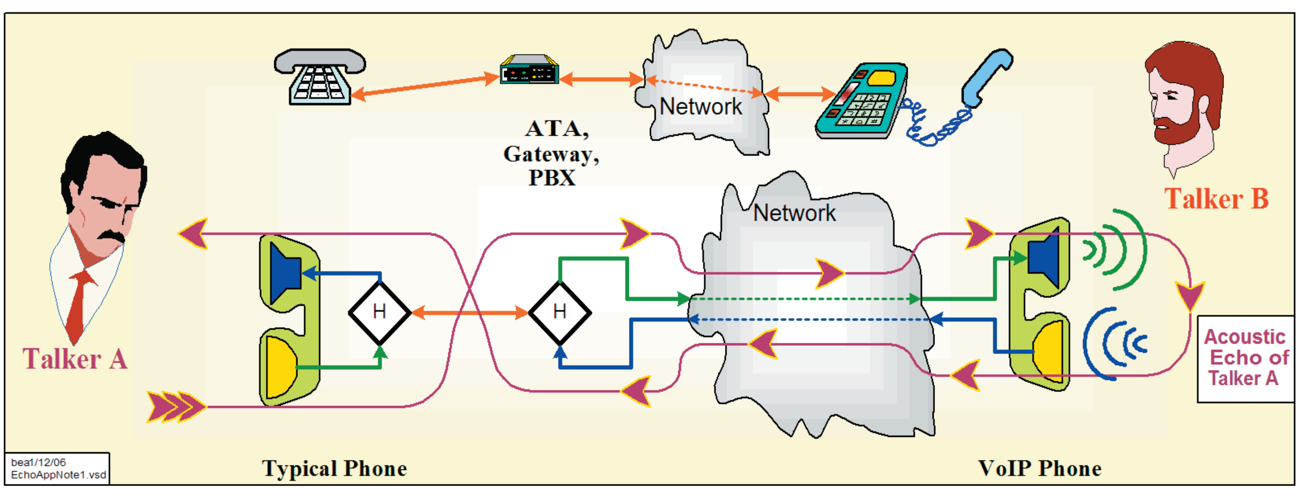
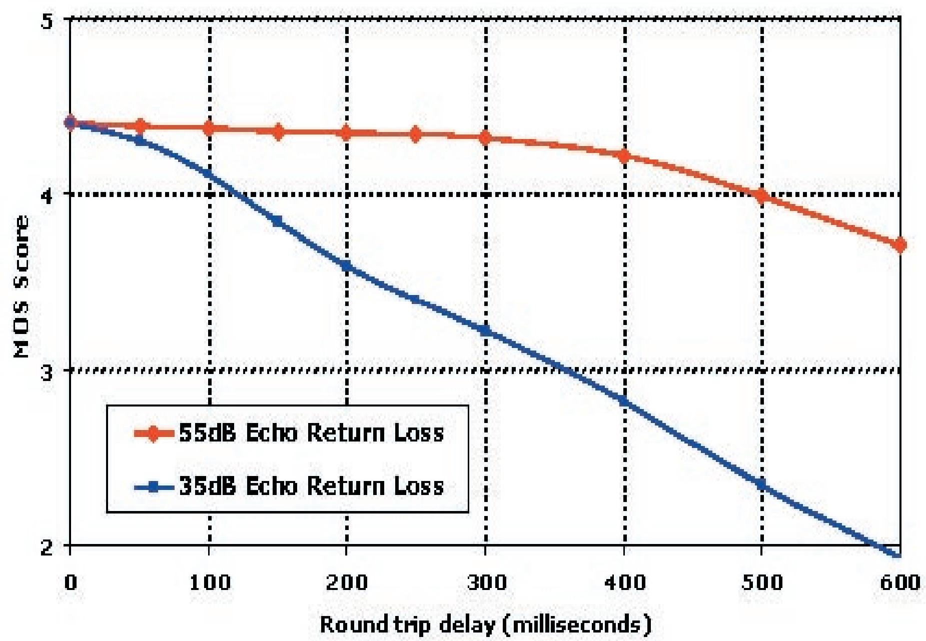


Figure 3: Acoustic Talker Echo.



Echo Suppression and Cancellation

Echo Suppression (or NLP)

Low to moderate levels of talker echo cannot be easily heard while the talker is actually speaking but are much more obvious during the gaps in speech (silence periods). An early approach to masking echo problems was to detect when these silence periods occur and to replace the silence with artificial background noise. Echo suppression is often called non-linear processing (NLP).

Echo Cancellation

Echo cancellation is a more sophisticated approach to removing the echo that may be present on telephone connections. An adaptive signal-processing algorithm monitors the speech signals going in each direction and attempts to learn the characteristics of the echo - i.e. if an echo is present, then what are its associated delay and amplitude? As the echo cancellation algorithm attempts to learn the characteristics of the echo path, the echo is reduced more and more as the learned characteristics become more accurate. The adaptation process is temporarily suspended during doubletalk, i.e. when both users are speaking simultaneously.

For an echo cancellor to operate, it has to keep some history of the sampled speech signal that was the original source of the echo. This history uses significant amounts of memory, usually a scarce resource in the digital signal processing (DSP) chips used in VoIP systems. If the echo delay is greater than the length of this history kept by the echo canceller, then it will be unable to cancel the echo.

The time taken for the echo canceller to learn the characteristics of the echo is called the *convergence time.* Sometimes a severe echo can be heard for a few seconds at the start of a call - this is due to the time taken for the echo canceller to converge and cancel the echo and is therefore called *convergence echo*.

Implementation of Echo Cancellation and Suppression

Echo cancellation and echo suppression are usually implemented together, and are able to reduce quite significant levels of echo.

The Echo Return Loss Enhancement (ERLE) represents the improvement in echo level introduced by the echo canceller. For example:

If echo return loss (ERL) is 25 dB, and echo return loss enhancement (ERLE) is 30 dB, then residual echo return loss (ERL + ERLE) is 55 dB.

The echo canceller may reduce echo levels by 25 - 35 dB and the addition of echo suppression (NLP) can further improve this.

Note that ERL can potentially reach 0 dB and some echo cancellers can only improve the echo level loss by 30 dB, which may still allow the echo to be audible.

Echo cancellers are commonly implemented in VoIP gateways and typically are configured to cancel echoes from the "trunk" side of the gateway (i.e. the non-VoIP side). Echo cancellers may be also used in IP phones to control acoustic echo from the handset. This is common in full-duplex speakerphones. However, acoustic echo cancellation is not always implemented in IP handsets or softphones.

Echo Measurement and the VoIP Performance Management Framework

Echo may be measured using specialized test tools that analyze audio signals, or may be estimated by the echo cancellers typically integrated into VoIP gateways. Specialized test tools are obviously able to make more accurate measurements. However, these tools would only be practical for troubleshooting once a problem has been identified. Due to the nature of echo, problems can occur on an apparently *ad hoc* basis and hence it is desirable to detect echo problems on live calls as well as collect data for post-analysis.

The emerging protocols that fit within the VoIP Performance Management Framework are able to support the detection and reporting of echo problems affecting live calls. A key element in this process is RTCP XR [1].

During calls, endpoints with Telchemy's VQmon/EP exchange RTCP XR VoIP metrics reports. These reports contain the estimated Residual Echo Return Loss (RERL) after the effects of echo cancellation, as well as the network round trip delay. The RERL value is an estimate made by the echo canceller. This approach allows the estimated echo level on *every call* to be reported.

For example, consider an IP phone connected to a remote trunking gateway, and the gateway connects to the traditional telephone network. Say that some echo is occurring on the telephone network and the IP phone user is experiencing talker echo. The echo canceller in the trunking gateway will attempt to cancel the echo (or at least reduce the level of the echo) from the telephone network. RTCP XR VoIP metrics reports from the trunking gateway to the IP phone will report the estimated residual echo level. This allows the IP phone to incorporate the estimated echo level into its calculations of call quality. The IP phone may then report the call quality using SIP. Then the conversational call quality metric reported by the phone would incorporate the estimated echo level reported by the trunking gateway.

Diagnosing Echo Problems

General

The key to diagnosing echo problems is to realize that the echo is originating at the other end of the telephone connection relative to the person complaining about it.

Unfortunately in moving to VoIP, the round trip delay is often significantly increased - this can make any existing echo problems much more annoying. VoIP did not introduce the echo, it simply made it much more obvious.

Echo Source Identification - circuit problems

When connecting VoIP calls to analog local loops (two-wire connections) attached to a PBX, CO, or analog phone ports on VoIP gateways, it may be possible to identify the specific ports that are causing echo problems. The simplest approach is to analyze user reports of echo problems to see if there is some commonality in the remote line - i.e. if users A, B, C and D are reporting hearing echo problems then ask, "Was this occurring only when they were making calls to user E?" The VoIP performance management framework can be a very useful aid in this type of diagnosis by analyzing actual measured data.

Echo Source Identification - acoustic echo

Acoustic echo is often due to cheap handsets or headsets, or to users who habitually hold their phone handset away from their ears. If the problem can be localized to a particular phone, headset, or user, then the best solution may be to replace the phone and potentially use acoustic echo cancellation software in the phone (available in some phones/ software builds). If echo problems occur with IP phone to IP phone calls, then immediately suspect acoustic echo – there is no path for the other types of echo.

Uncancelled echo level may be very high

An echo canceller typically decreases echo by 30dB or more. However if the echo level is severe to begin with, there may still be some audible echo after cancellation. This condition may also result in convergence echo.

No echo canceller

Some VoIP systems do not contain echo cancellers (e.g. an IP phone). And those that do, may have the echo canceller oriented in a direction opposite to the echo. For example, say that the user of an IP phone has a tendency to loosely rest the handset on their shoulder rather than pressing it to their ear. This may allow some acoustic feedback from the handset speaker to the microphone resulting in an echo being heard by the remote user. The call may be going through a VoIP gateway that contains an echo canceller. However, the echo canceller in the gateway would typically be oriented to cancel echoes originating from the PCM/ non-VoIP side of the connection and not from the IP side. Therefore, it would have no effect on the locally generated echo heard by the remote user.

Echo longer than tail

Echo cancellers are configured with some maximum echo path delay or echo tail length. It is expensive to make this tail length long because it consumes expensive DSP memory and so, it is often set to some compromise value, typically 16 to 128 milliseconds. If the actual echo path delay is longer than this value, the echo canceller will be unable to cancel the echo.

This problem may occur when a VoIP call is connected to a PCM trunk that is either connected to another VoIP network, a cellular network or some other network with delay (e.g. satellite).

Echo path distortion

Sometimes the path taken by the echo may introduce significant non-linearities that greatly reduce the effectiveness of echo cancellers, which generally assume that the echoed signal has a linear relationship with the original signal.

Signal level too high or low

If the local signal level is too high or too low, then the echo canceller may not operate correctly. Low signal levels can also cause echo suppressors (NLPs) to suppress quieter parts of speech because the echo canceller cannot distinguish between low level speech from the *local* user and echo. This can lead to gaps in the speech heard by the remote user.

Doubletalk

Some echo cancellers can have some difficulty dealing with calls that have excessive doubletalk - either due to the nature of the discussion (e.g. an argument) or high round trip delay causing the users to talk at the same time. This is because echo cancellers rely on periods of one-way speech to learn the echo characteristics.

Summary

Echo is a common problem in Voice over IP services - not because VoIP introduces echo but because it increases round trip delay, which makes echo more obvious. It is possible to troubleshoot echo problems in many cases, and deploying the VoIP performance management framework (i.e. RTCP XR VoIP Metrics, and SIP QoS reports) can help to identify in-service problems.

Notes

Acronyms

ATA	Analog Telephone Adapter
CO	Central Office
dB	Decibels
DSP	Digital Signal Processing
ERL	Echo Return Loss
ERLE	Echo Return Loss Enhancement
IP	Internet Protocol
ITU	International Telecommunications Union
ISDN	Integrated Services Digital Network
ms	Milliseconds
NLP	Non-Linear Processing
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
QoS	Quality of Service
RERL	Residual Echo Return Loss
RTCP XR	RTP Control Protocol Extended Reports
RTP	Real Time Protocol
SIP	Session Initiation Protocol
VoIP	Voice Over Internet Protocol
VQmon/EP	VQmon End Point
VQmon/SA	VQmon Stream Analysis

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References

[1] IETF RFC3611 RTP Control Protocol Extended Reports (RTCP XR) - VoIP Metrics T. Friedman, R. Caceres, A. Clark

[2] ITU-T Recommendation G.168 Digital Network Echo Cancellers

About Telchemy, Incorporated

Telchemy, Incorporated is the global leader in VoIP Performance Management with its **VQmon**® and **SQmon**™ families of call quality monitoring and analysis software. Telchemy is the world's first company to provide voice quality management technology that considers the effects of time-varying network impairments and the perceptual effects of time-varying call quality. Founded in 1999, the company has products deployed and in use worldwide and markets its technology through leading networking, test and management product companies. Visit www.telchemy.com.

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