Communications Infrastructure Test Group Engineering

Technical White Paper

Hammer VoIP Test System Echo Detection and Analysis

ABSTRACT:

This Technical White Paper provides an overview of echo impairment of digital communications networks and how to use the Hammer VoIP Test System to detect and analyze echo. The information from the VoIP Test System can be used with Echo Canceller devices by wireless and digital network providers to take corrective action ensuring the highest voice quality across their network. .

Overview

Digital telephony subscribers are becoming more and more critical of their network. They expect near-wireline voice quality. A common impairment of digital telephone connections is *Echo*. Echo is when a delayed and distorted copy of a voice signal is reflected back to its sender interrupting communication.

Echo is a quality problem.

An important technology in network voice quality is Echo Cancellation. The Hammer VoIP Test System can be used to detect and analyze echo in audio samples captured from telephone connections made with any one, or combination of telephony protocols off a network. This information can be used by network operators to deploy and configure their Echo Cancellation resources to provide the highest call quality for their customers or be vendors of network elements (Gateways) implementing echo cancellation to test their devices.

This White paper discusses the following:

- Echo and Its Causes
- Echo Cancellation
- Hammer VoIP Test System Echo Detection and Analysis
- VoIP Test System Echo Test Example

The example in this document was created on a Hammer LoadBlaster 500™ equipped with the Hammer VoIP Test System 2.7 in conjunction with .Hammer TestBuilder 2.7. A passing familiarity with the Hammer, Hammer TestBuilder, a previous version of the VoIP Test System, the PSTN, digital networks, are assumed with the example.

Figure 1. shows a block diagram of telephone network architectures and echo discussed in this paper.

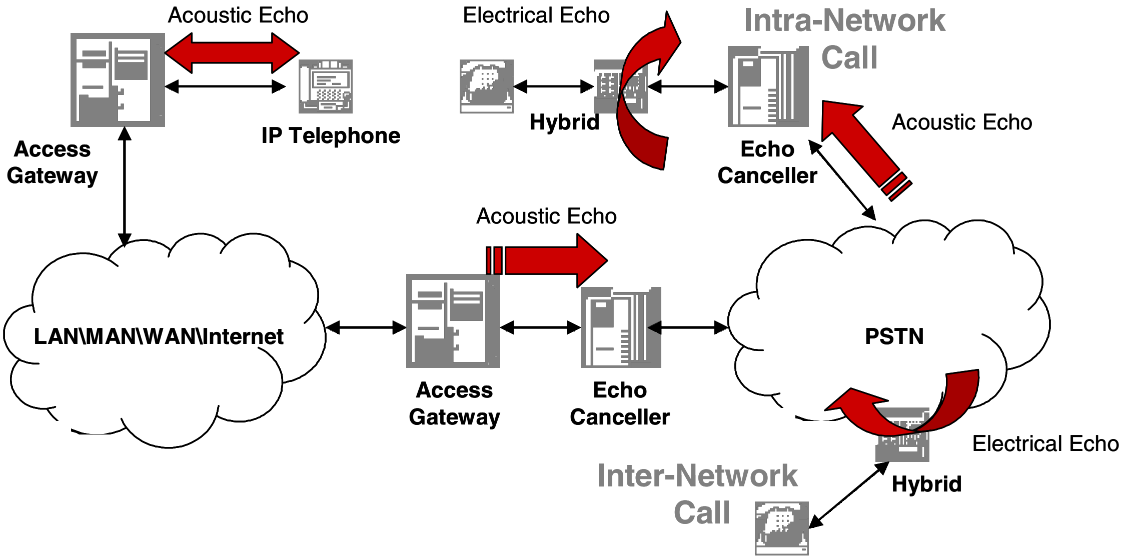

Figure 1. Network Architectures and Echo

SEQARABIC Echo

In digital networks an *Echo* occurs when hardware intended to amplify or relay the speech of one party picks up the speech from the other party and reflects it back to them.

Echo in Telephone Calls

Echo is perceptual. There is a component of echo in every telephone call. An echo is a signal with a delay and strength. The length of time it takes for a voice signal to be reflected back determines how much an echo impairs the quality of a telephone call. If an echoed signal's strength does not rise above the voice call's background noise, it is not noticed.

A beneficial echo is called *Sidetone*. It is an effect and applied technique allowing the speaker to hear their own voice in the handset's speaker, convincing themselves their voice is being heard. The benefits of Sidetone are felt when a speaker hears their own voice at a low volume within between 5 and 25 milliseconds (ms). Call quality begins to be is impaired when a speaker hears their speech repeated after approximately 30 ms. Too little sidetone at all makes a call unerring, as the speaker cannot hear themselves speak.

Note the brief intervals of time involved in determining sidetone. In digital networks (especially IP) with their inherent latencies it is very easy to have delays in excess of 30 ms.

Types of Echo

There are two primary sources of echo in telephone communications networks:

- Electrical Echo

- Acoustic Echo

Electrical Echo

Electrical echo, sometimes called "Line-Echo" or "Hybrid Echo" occurs in Public Switched Telephone Network (PSTN) connections. It results from an imperfect electrical circuit in the network

In a digital Central Office (CO) *Hybrids* are devices that convert between the 4-Wire Switching fabric and the 2-Wire cable of the local loop. The local loop is the twisted pair cable that connects a subscriber's telephone to the CO. The Switching fabric is where the local loop's lines are connected to long-distance trunk lines.

Note that local calls do not have a problem with electrical echo. This type of connection does not have a hybrid in the circuit.

A Hybrid is a sensitive device that requires tuning to ensure call quality. If it's not properly tuned to match the Two-wire cable, voice signal energy passing from the 4-wire to the 2-wire part of the network is reflected back on itself creating an Echo.

The Hybrid on the opposite side of the network produces the echo heard by the speaker. That is, the near-end Hybrid creates an echo of the far-end caller's voice signal reflecting it back to the far-end. The near-end caller's voice signal is reflected back by the far-end hybrid.

In all digital networks, electrical echo is not a problem. However, calls originated from the PSTN may introduce this type of echo into digital networks.

Mis-tuned hybrids are a frequent problem found in COs'. Given information on the performance of a connection, the Hybrid can be tuned to meet a carrier's call quality performance needs.

Acoustic Echo

Acoustic echo occurs in both PSTN and digital networks. It is a more complex signal than electrical echo. It is also much more difficult to eliminate than electrical echo.

Acoustic echo results from the following sources:

- Handset use and design
- Voice encoding and decoding devices (codecs)
- Network delay in digital networks

HANDSET ECHO

Handset echo is the result of poor handset design or component quality, or operating the handset in an acoustically unfriendly environment.

A handset's physical and electrical design contributes a lot to reducing echo. A properly designed handset minimizes the acoustic coupling between the speaker and microphone of the handset. This acoustic coupling is an open-air path that can exist between the microphone and the speaker. Frequently eliminating this pathway requires a compromise between echo suppression and a handset's aesthetics.

An extreme case of handset echo is called *Howling*. This is when a feedback situation develops between the between microphone and speaker. This frequently occurs in the use of "hands-free" telephones.

Some standards for handset design to exist. Although, these standards are primarily for digital wireless handsets and not wirelines. These standards include specifications for sidetone tolerance and echo suppression performance.

Another type of handset echo is called multipath echo. This echo can occur in acoustically unfriendly places like automobiles

echo occurs when speech indirectly enters the handset's microphone late after having been reflected off of walls, windows, floors, ceiling, and furniture. This reflected sound is called multipath audio or reverberation. Mutipath echo is heard by both parties of the call as an echo.

CODEC ECHO

Echo occurs on digital networks is the result of an unsuppressed acoustic or electrical echo made worse by digital encoding.

Digital handsets and digital network elements include a codec for the compressing, encoding and decoding of speech to reduce the network bandwidth used in a call. This encoding is typically imperfect reducing voice quality. There is also requisite decode of the speech at the other end of a digital call. This process of encoding and compressing, then uncompressing and decoding typically introduces milliseconds of delay to transmitted speech which can abet an echo.

NETWORK ECHO

Additional speech delays occur when a digital call is processed through multiple network elements in different technology networks. An encoded echo would be delayed at the intersection of two networks resulting in a signal reflection

An example of this might be a long-distance telephone call from an analog landline on one carrier's network to the other party's digital wireless telephone on another carrier's network. A call like this would initially be analog on twisted pair copper wire to the CO. At the CO it could be digitized and placed on the carrier's fiber optic backbone. The call may leave the original carrier as ISDN going to the wireless carrier's mobile switching center. From the mobile switching center to the Cell Site it might be Ethernet. Finally, from the Cell Site to the receiving party's wireless telephone it would essentially digital radio. A multipath echo introduced by the wireless telephone might arrive on the analog line side 200 ms after the original speech causing a significant impairment of call quality.

Further, in IP networks, a packetszed voice signal could have packets taking different routes to their destination. The delayed arrival of groups of packets in a signal having taken separate route results in momentary echo. This echo can quickly disappear as the traffic conditions and routing on the network change.

Measuring Echo

Echo is a reflected audio signal. It is measured in terms of signal strength in decibels (dB) and delay in ms. The more significant component is the delay.

If the echoed signal is reflected in 25 ms or less it is either humanly imperceptible or considered sidetone. If the speaker hears their reflected signal after approximately 40 ms. or more, the call quality is impaired by echo.

The internationally recognized metric for measuring the level of signal reflected back is called the Echo Return Loss (ERL). The units for ERL are dB. The smaller the echo in a signal, the higher its ERL. The lower the ERL the larger the echo in a signal. The VoIP Test System measures ERL directly in voice signals.

Eliminating Echo

Echoes are eliminated two different ways from telecommunications channels, through *echo suppressors* and *echo cancellers*. Carriers position these devices within the network to ensure call quality. However, owing to their cost they are deployed sparingly.

Echo Suppressors

An echo suppressor detects human speech coming from one end of a connection, and suppresses all signals going the other way.

An echo suppressor is toggled by a voice recognition circuit. They can typically "trip" within 5 ms. to block a reflected signal. Unfortunately, this technique results in a half-duplex channel. This half duplex operation is not noticeable in voice communication, but can adversely effect data communications.

To handle data communications, telephone circuits with echo suppressors have an in-band signaling method to disable themselves. When the suppressor detects a pure tone at a specified frequency, they shutdown while the while the carrier is present.

Echo suppressors are found on the PSTN installed by Inter-Exchange Carriers (IXCs). IXCs typically have them installed at each end of their network to filter calls entering their network. See Figure 1.

Echo Cancellers

An echo canceller is a computer-based device that samples a call and "profiles" it. An echo will violate the profile. When an echo is detected it simulates the echo, estimates its magnitude, and then subtracts it from the audio signal it is sampling.

An echo canceller initially samples the signal on the channel to characterize the voice and the echo signals passing through it. The time the canceller takes to profile or model the signal is called the *convergence time*. This model is continuously updated during the life of the connection. As the signal passes through the canceller, it is compared to the signal model it has created. Any part of the signal that differs from the model is echo and is digitally subtracted from the signal before the signal is relayed on to its destination. Any echo that may pass through this initial subtraction is deleted by decreasing the signal's power, causing the echo to disappear into the background line noise.

Unfortunately, this technique cannot perfectly cancel acoustic echo. The information in the voice signal model has a finite lifespan called the *tail circuit delay*. An echo with a long delay (approximately 130 ms.) will have been aged out of the model and be interpreted by the echo canceller as part of the voice signal.

A common problem with echo cancellers is *Doubletalk*. Echo cancellers perform poorly in this situation. A typical telephone conversation is half-duplex; the parties alternate speaking and listening. Occasionally in a conversation both sides speak simultaneously. With two legitimate voice signals simultaneously active, an echo canceller may interpret either of the two parties' speech to be echo and subtract it from the connection. This result is an call audio impairment called *Speech-clipping*.

Echo cancellers are found in modern digital networks at their junction with other networks. See Figure 1.

Echo Control in Digital Networks

Echo cancellers are the superior solution for controlling echo in networks. They are less expensive than the analog-to-digital technology of echo suppressors, and they eliminate in-band signaling requirements. They are also digital. Echo cancellers are typically based on off-the-self Digital Signal Processors (DSPs) that are inexpensive and easy to program.

The following characteristics qualify an echo canceller:

- Circuit Tail Delay
- Convergence Time
- Double Talk Range

The ITU G.167 recommendation for acoustic echo controllers gives criteria for a number of performance characteristics. However, the state-of-the-art echo canceller should cancel across a tail delay of 128 ms; converge in 50 ms; and render speech clipping unnoticeable in double talk situations.

To further evaluate the performance of an echo canceller additional effects to test for are:

1. Residual Echo: Embedded echo. Echo resulting from too short a tail delay appearing later than expected.

2. Convergence Loss: Embedded echo reappearing in a signal in double talk situations sounding louder than a "typical" echo.

3. Changing Acoustics: Embedded echo resulting from changing the microphone volume over the duration of the signal simulating acoustic echo.

4. Howling Rejection: appearance and duration of a squeal when microphone /loudspeaker feedback is introduced.

5. Doubletalk Attenuation: Detectable lowering of speech signal volume during double talk situations.

6. Half-Duplex Communication: Elimination of one parties signal during doubletalk situations.

7. Audible Transitions: signal volume changes, noise, and attenuation of background noise between words in a voice signal.

Echo and the VoIP TS

The VoIP Test System provides a number of tools to for making voice call quality measurements including echo detection. One of these tools: VQScope allows you to detect and measure echo in voice signals recorded off a network.

Echo Detection

VQScope used in conjunction with the VoIP Test System's PSQM scoring capability can be used to detect impairment in a transmitted and recorded audio sample in a TestBuilder created test.

Echoes are initially detected through their high PSQM scores. PSQM uses an algorithm (defined in ITU-T P.861) that determines how much a received audio sample sounds like the original sample, taking into account the characteristics of human hearing.

Under PSQM, lower scores are better. Scores can range from 0.0 (Perfect) to 6.5 (Poor) and higher. Audio with scores above 6.5 may still be humanly intelligible, but would fail most carrier voice quality standards and cause automated voice recognition applications to fail.

Hammer TestBuilder can create tests performing PSQM scoring of calls.

SEQARABICIn a PSQM scored call, a connection is made between two scripts executing on separate Hammer channels. PSQM consists of a quick in-band tonal synchronization of the playing and recording scripts, and then transmission of a single audio sample. The received audio sample is then scored to see to what degree it differs from the originally transmitted audio sample.

TestBuilder also has the ability to create PSQM tests simulating the important doubletalk condition, to stress echo cancellers.

Figure 2. shows the sequence of call events to score an audio sample. Caller A is the prompt (audio sample) playing script. Caller B is the prompt recording script. The received audio sample‘s scoring is handled by the B-Side.

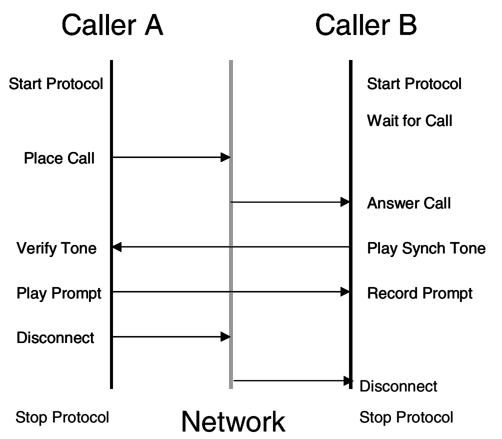

Figure 3. PSQM Scoring Call Sequence

A higher than expected PSQM score indicates call impairment.

Note that benchmark PSQM scores for the test equipment and System-Under-Test (or network) are needed before actual testing begins. These benchmarks must be established under ideal, controlled conditions. They will later be taken into account with the test results to determine the actual degree of call impairment

Echo Analysis

Analyzing acoustic samples is like detective work. Echo is just one of the sources of call impairment in digital networks The analyst needs to be open-minded about what they might find. Also, to narrow the analysis, the analyst must understand the signal path the samples followed through the System-Under-Test, before beginning the analysis.

Crosstalk (intrusion of a foreign signal), packet loss, and jitter (variable end-to-end packet transit time) are some of the other reasons for a high PSQM score. Fortunately, call impairments have recognizable characteristics that can easily be exposed with analysis using VQScope VQScope includes three tools used in a three-step process for analyzing the record of a call impaired by echo: Acoustic Spectrum, The Echo Return Loss (ELR) engine and its own built-in Echo Canceller (EC) engine. Both the ELR and EC tools were developed in accordance the ITU-T G.165 recommendation for Line Echo Cancellers.

Subjective listening tests can also be performed within VQScope.

VQScope allows you to do a quick visual inspection of a recorded audio sample’s acoustic signature with its Acoustic Spectrum tool. This inspection identifies many types of impairments based on the observed waveform.

For example, a high PSQM score resulting from packet loss would show a “cut-out” of several milliseconds looking like silence in an unexpected position within the signal’s waveform. This cut-out would be clearly shown by the Acoustic Spectrum tool.

Echo also has a characteristic signature, although echo is not as obvious to detect as packet loss. When echo impairment is suspected the VQScope ERL tool is used to confirm its presence..

Echo is typically responsible for a large amount of signal loss. The ERL tool graphically shows the signal loss between the recorded audio sample and the original signal. With the ERL the temporal location and magnitude of signal loss in the recorded audio sample are determined. Repetitive instances of signal loss within a signal are characteristic of echo.

The EC tool provides an analysis of any echo located in the recorded audio sample. It performs exactly like an echo canceller. Using an Empirx developed ITU-G 165 compliant algorithm. It graphically displays the location and amount of echo it detects in the recorded audio signal. If the signal has previously been through a network’s echo canceller(s) it can be used to detect any residual echo. It also provides metrics on the amount of echo present including the maximum strength of the echo it detected in the signal (Echo Depth).

By understanding the signal path of the System-Under-Test and using VQScope, an analyst can quickly diagnose echo and other call impairments in a three-step process.

Echo Detection Example

The following section is an example of how to setup, execute and evaluate an echo test using a Hammer LoadBlaster 500 with TestBuilder and the VoIP Test System.

The example shall include the following sub-sections:

1. Setup and configure the hardware.
2. Setup and configure the test.
3. Execute the test.
4. Analyze test results for echo.

For sub-sections two through four a tutorial format is used to instruct engineers unfamiliar with VQScope how to use the tool on a mouse-click by click basis.

Hardware Setup and Configuration

The System-Under-Test for this example is two Access Gateways which convert between digital telephony (ISDN) and IP telephony then back again. Figure 3 below summarizes the System-Under-Test.

Note that two sets of digital telephony connections are needed. One set connects the Hammer to itself in a "loop-back" connection. This connection is used to benchmark the test equipment. The second set connects the Hammer to the Gateways.

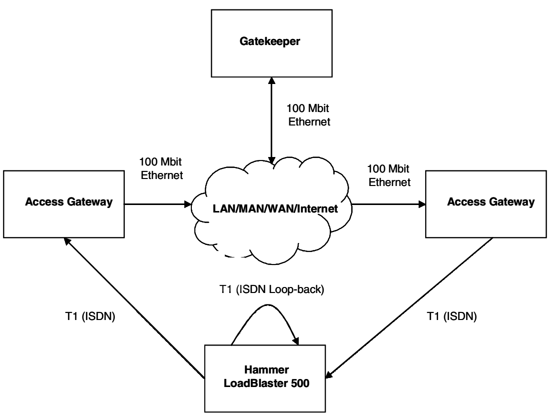

Figure 4. Echo Example Block Diagram

The details of the setup are omitted for the sake of brevity. However, they include:

1. Cabling the Hammer and the Gateways together.
2. Setting-up the Gatekeeper to recognize and route telephone calls between the Gateways.
3. Configuring the Gateways. This includes enabling their Echo Cancellation capability.
4. Configuring the Hammer's telephony hardware. This primarily involves using the Configurator to setup ISDN for use with the Gateways.
5. Configuring the Hammer's software to interface with the Gateways. This primarily involves programming the PhoneBook to place calls through the Gateways.
6. Verify the Voice Quality Server and Hammer Telephony Server are running on the Hammer in the Configurator application.

Test Setup and Configuration

TestBuilder with the VoIP Test System comes delivered with several tests for performing PSQM scoring. This example uses the test: `Voice Quality Test` located in the TestBuilder `CallProfileTests` folder.

Open the test using TestBuilder displaying the test's ladder diagram. In this example doubletalk will be simulated. Select each of the Voice Quality Play icons in the test's ladder diagram. Select Properties from the pop-up menu. In the Voice Quality Properties window check the Enable Doubletalk checkbox. Enable the PSQM Method's radiobutton.

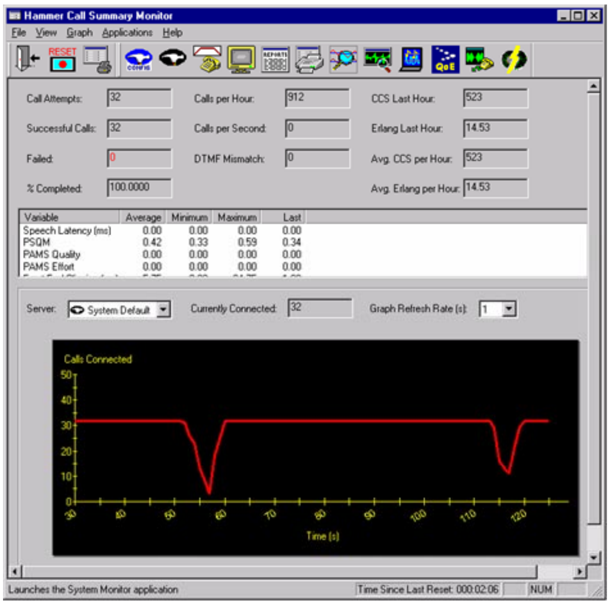

Figure 5. Test Example Ladder Diagram

Figure 4 shows the TestBuilder ladder diagram for the test with the Voice Quality Play Properties window open.

Test Execution

The scheduling and execution of the Echo Test calls is through Test Profiler. The execution of the calls can be observed on the System Monitor and the Call Summary Monitor.

The tests need to be run in two series. The first series is to benchmark the Hammer hardware. This will determine the base PSQM score for the System-Under-Test. The second series is the actual "live" test of the calls through the Gateways.

TEST PROFILER

In the Test Profiler, select the `PSTN Switched Circuit Telephony` category of tests and the test named: `Voice Quality Test`.

The difference between the two test series is in the Calling Direction Box. For the first "benchmark" test assign the "loop-back" spans. These are the spans that connect the Hammer to itself. In the second "live" test assign the spans that connect the Hammer to the two Gateways.

In the Test Profiler's Calling Direction box, configure the test for at most a single ISDN span on each of the A and B sides. PSQM scoring is a CPU-intensive process. Scoring with more than a span of channels on a LoadBlaster 500 would require a remote VQS be assigned.

Finally, select `Blast` for the Calling Profile. The Number of Callers ideally should be set to the maximum number of calls supported by the receiving Gateway, within the PSQM single span restriction.

BENCHMARK TEST

Execute the first test, using the loop-back spans using Test Profiler.

The execution of the test can be observed from the Hammer System Monitor on a per channel basis. From the Call Summary Monitor PSQM scores for the calls can be observed.

Note the average PSQM score in the Call Summary Monitor for approximately 10 minutes of calls or until the average PSQM score stabilizes. This average score is the benchmark PSQM score for the Hammer. The System-Under-Test cannot produce a lower (better) score than this.

A typical PSQM score for a Hammer LoadBlaster 500 with an ISDN loop-back is: 0.30.

LIVE TEST

Halt the benchmark test and reopen the TestBuilder ladder diagram for the test. In the `Voice Quality Play Properties` window for each of the Voice Quality Play icons, set the Method Threshold number to 1.68 times the benchmark PSQM score. Adjusting the threshold in this way retains the received audio samples for calls likely to have impairment.

For example, if the benchmark test score is 0.30, the PSQM threshold should be set to 0.50 for sample data collection.

Execute the second "live" test, the Gateway test using Test Profiler with the Gateway connected spans.

Note the average PSQM score in the Call Summary Monitor until the average PSQM score stabilizes. Observe the difference between the "live" average score and the benchmark average score. More importantly, note the relationship between the Threshold number previously set and the Average and Maximum PSQM scores.

A live subjective listening test can also be performed during the test using the channel monitor to hear both sides of the call, including the doubletalk signal being transmitted.

Analysis is performed after the test is complete on the saved files produced as work products by the VoIP Test System. This analysis identifies the source of high PSQM scores. Data will only be saved on calls whose PSQM score exceeded the Threshold.

If no PSQM scores are exceeding the threshold. The threshold may need to be set downward and the test re-run to collect data. The tester may have to accept there is no call impairment in the System-Under-Test, if the live PSQM scores are close to the benchmark average.

Figure 5 shows the Call Summary Monitor for a typical "live" test. Note the PSQM scores. In the figure shown call PSQM scores have exceeded the 0.50 threshold set for the test. The example continues using a saved audio sample with a PSQM score of 0.57.

Test Results Analysis

Analysis is performed on calls with higher than expected PSQM scores. The analysis is a three-step process. At any step a diagnosis can be reached ending the process.

The three steps are:

1. Spectral Analysis
2. Signal to Noise Ratio Analysis
3. ERL

All of the analysis takes place using the single window of the VQScope application in the VoIP Test System

SAMPLE DATA

Open the VQScope application. From its File menu select Open. The Input Test Reference and Signal Files window appears to load the audio sample files for the analysis.

The first file to get is the Test Signal. This is the received recorded audio sample from the PSQM scoring. The samples with scores exceeding the threshold are located in a folder named after the test.

For this example the test the data is located in:

```
C:\Hammer\Vqs\Save\Voice Quality Test
```

In this folder will be located pairs of files containing the sample data. The file names are made up of fields separated by pound signs (#). The information in the third and forth field from the left is the most important. These fields are the audio sample name and the PSQM score.

An example of the first four fields found in the file name for this example might be as follows:

```
H9999#1234567#voipboy1p1.pcm#0.57#
```

Field three and four indicate the file contains the transmitted and received audio sample `voipboy1p1` which received a PSQM score of 0.57.

From the Input Test Reference and Signal Files window browse to the stored sample directory and select the sample with the highest PSQM score (fourth field) to be the Test Signal.

Select only files in the folder that does not have a postfix. The file with the postfix `pcm.phf` is not an audio sample used in analysis.

In the window's Reference Signal box browse and click on the file in the default directory that has the same audio sample name as the one found in third field of the Test Signal file previously selected.

In this example, it would be `voipboy1p1.pcm.`

Click OK, and VQScope will load with recorded sample data and the reference signal ready to begin the analysis.

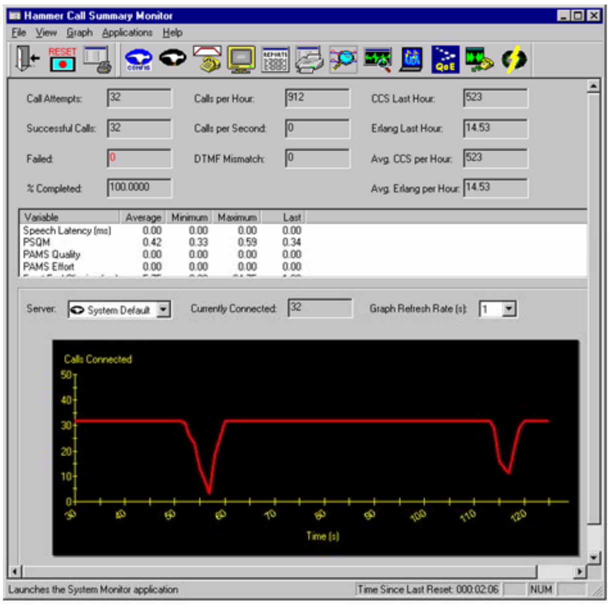

Figure 6. Call Summary Monitor for Live Test

SPECTRAL ANALYSIS

The first step is to perform a spectral analysis by visually comparing the sonic signature of the transmitted audio sample with the audio sample received. This is done with the `Layered View` View of VQScope. Figure 6 shows the sonic signature of the impaired call.

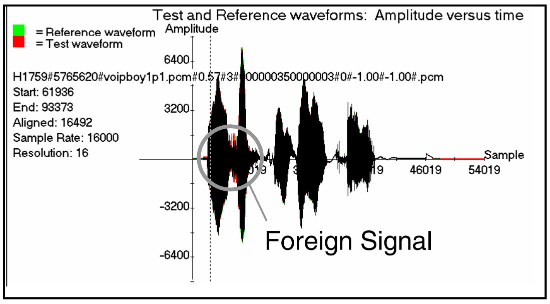

Figure 7. Sonic Signature of Impaired Call

The figure shows the test and reference audio samples overlaid upon each other. The graph is signal amplitude in Hz over time. The time axis is in terms of samples where the sample rate is 16K Hz.

A circle has been placed around an area of the graph and labeled "Foreign Signal". This is a signal that was not present in the reference sample transmitted, but appears in the recorded sample. This foreign signal appears as an "overlap" or a "shadow" of a different color on the reference signal's graph. "Overlaps" or "shadows" are characteristic of echo. They can also be crosstalk.

Note that this analysis would also be able to detect call impairments resulting from packet loss or jitter. This graph can provide an explanation for high PSQM scores without having to perform any further analysis.

In addition to a visual analysis, a subjective listening test can also be performed on the recorded audio sample to detect impairment by clicking on `Play` in the Test Signal graph window.

SIGNAL TO NOISE RATIO ANALYSIS

The second step is to perform a signal to noise ratio analysis of the impaired call. This is done using the `Echo Return Loss` View of the VQScope. Figure 7 Shows the Signal to Noise distribution of the recorded signal.

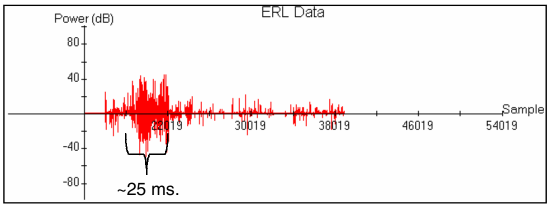

Figure 8. ERL Graph of Impaired Call

Figure 7 shows the strength of the foreign signal in the recorded signal. The graph is power in dB over time. The time axis is in terms of samples where the sample rate is 16K Hz.

Attention should be paid to the section of the signal that contains the foreign signal detected in the previous analysis step.

A brace has been placed around an area of the graph and labeled "~25 ms". Duration can be approximated by dividing the sample numbers shown on the graph ticks by the 16,000 Hz sample rate to arrive at millisecond duration. From the graph a foreign signal of about 40 dB with duration of about 25 ms can be estimated to exist in the received audio sample.

Foreign signals with a low enough amplitude can sometimes be "buried" in the received signal and go undetected. This graph can also be used to uncover any additional foreign signal that might have gone undetected in the Spectral Analysis. Note from the Figure 7's graph, that the remainder of the recorded signal is relatively "clean". . Most echoes repeat within a signal. There is no repetition of the discovered foreign signal. This would argue against echo impairment of the call

ERL ANALYSIS

The third step is to perform an ERL analysis of the impaired call. This is done using the `Echo Canceller` View of the VQScope. Figure 8 Shows the residual (echo?) signal that the EC tool could remove from the recorded signal.

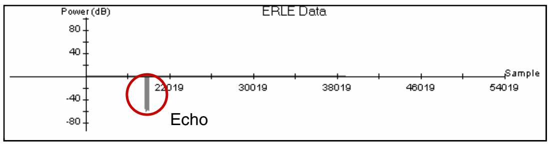

Figure 9. Residual Signal Echo Cancellation

The figure shows the strength and duration of the residual echo in the received recorded signal. The graph is power in dB over time. The time axis is in terms of samples where the sample rate is 16K Hz. Not shown on this figure (but shown in the display) is the maximum strength of the extracted signal: -63.2 dB.

A circle has been placed around an area of the graph and labeled "Echo". The dense graph pattern displayed is characteristic of an echo and meets all the criteria of echo listed in ITU-G 165.

Figure 9 shows the entire VQScope window used in this analysis. Note the PSQM score and additional voice quality metrics at the bottom of the window. In addition, the Reference and Test (received and recorded) audio samples are shown in separate windows.

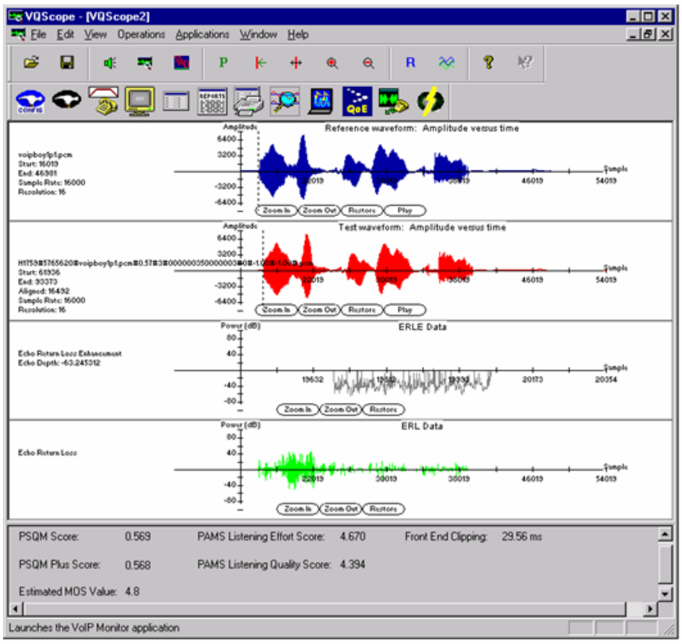

Figure 10. VQScope Window of Impaired Call

Final Analysis

For the purposes of this example, a final more detailed spectral analysis was performed to verify the results. This is not shown.

The Cool Edit™ off-the-shelf sound editing software was used to inspect the audio frequency spectrum of the fragment of foreign signal in detail.. This analysis verified that that the "echo" found in this example has the acoustic spectrum of the doubletalk signal transmitted by the receiver while recording.

From this, we can determine the Gateway detected the reflected doubletalk signal and successfully cancelled it. However, approximately 25 ms. of echo "leaked" through the canceller. There is also the possibility the echo was present earlier in the signal, but went undetected. The 25 ms of echo is probably results from the time the Gateway's echo canceller took to converge on and eliminate the doubletalk signal.. Further tests with other audio samples would be needed to accurately determine the Gateways: tail delay, convergence time and full double talk range.

HYPERLINKSEQARABICHY PERLINKSEQARABICSEQARABICSEQAR ABICSummary

All voice calls on digital networks require some form of echo cancellation. These cancellers are deployed on either side of the compressed signal paths. To achieve high voice quality telephone calls, understanding the echo effect, how the telecom industry deals with it, and how the Hammer can be used to detect and analyze echo are very important.

This Technical White Paper described how to use the Hammer VoIP Test System perform echo detection and analysis.

It discussed electrical and acoustic echo and their sources. In addition, it discusses echo cancellation and the criteria for evaluating it.

It discussed testing for echo. A good testing strategy, a Hammer LoadBlaster, a TestBuilder built test, and the VQScope application are the tools needed to detect and analyze echo in telephone calls.

Cool Edit™ is a trademark of the Syntrillium Corp.

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Finally, it gave a step-by-step tutorial of how to hunt a real echo in a network System-Under-Test.

The information in this white paper can be used to expand the Hammer's capability to perform voice quality testing on new classes of communication and non-communication devices that previously could not easily be tested with TestBuilder.

Technical white papers are a service provided by Communications Infrastructure Test Group Engineering. They are intended to inform our colleagues and customers on the Hammer's capabilities. They are for the exclusive use of Empirix customers and employees only.