



NetVanta Unified Communications Technical Note

Analog Gateway Implementation Guide: Understanding and Eliminating Echo

Introduction

This technical note describes analog gateway technology and approaches to reduce or eliminate audio problems that are found at a customer's location.

Analog Technology Review

Analog SIP PSTN gateways provide interconnectivity between analog CO lines/trunks and the UC server. Analog gateways convert two wire PSTN trunk signaling and audio into SIP signaling and digitized voice RTP media.

How they can be connected:

1. Directly to a Central Office
2. Connected through a Channel Bank to the Central Office
3. Connected directly to a PBX analog extension

Analog lines provide the following types of services:

1. Ability to dial another telephone number connected to the PSTN.
2. Ability to receive incoming calls.
3. Ability to receive the calling name and number (CLASS information)
4. Transport voice, DTMF, and fax information when connected to an outside party.
5. Provide answer supervision. Answer supervision is the ability to determine when the far end has answered an outgoing call.
6. Provide disconnect supervision. Disconnect supervision is the ability for a phone line to signal to the gateway that the far end has disconnected on an incoming or outgoing telephone call.

Understanding Echo

Echo is the phenomena where the person speaking hears a reflection of their own voice, slightly delayed. Two fundamental principles apply to echo:

1. The louder the echo, the more noticeable and annoying it is.
2. The more delayed the echo, the more noticeable and annoying it is.

The following is also true – echo is only noticeable if the end-to-end delay of the audio is more than 25 ms. Echo only occurs when there is two wire to four wire or four wire to two wire conversion). Any analog to digital conversion will be a conversion point. The echo can be described as a signal leakage between transmit and receive audio paths.

Echo is ALWAYS present in analog phone circuitry. SIP/VoIP networks exacerbate the pre-existing analog impairment. VoIP audio packet latency (due to packetization and network latency) causes the echo to become objectionable. In TDM and legacy PBXs networks the pre-existing echo is indistinguishable from the sidetone in the handset.

Let's understand the impact of the power level (or loudness) of the reflected echo. This is measured using the concept of Echo Return Loss (ERL). ERL is a measure of the ratio of the power of the original voice signal to the power of the returned echo. It is expressed in dB and its value is:

$$\mathbf{ERL} = \log_{10}(\mathit{VoicePower} / \mathit{EchoPower})$$

The higher the ERL the lower the echo is compared to the original signal. A target ERL of greater than 40 dB is required for the echo to be acceptable.

Echo Loudness Reduction Techniques

1. Match as closely as possible the source and destination impedances. A small mismatch in source and destination impedances (resistance at a given frequency) will have a significant impact on the ERL. Conceptually it is comparable to attaching a 4" fire hose to an ordinary garden hose. Once water flows from the fire hose to the garden hose, some of the water will be pushed back toward the fire truck.

$$L_r = 20 \log_{10} \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right|$$

Z_1 is the impedance towards the source and Z_2 is the impedance toward the load.

If the impedance is off by just 100 ohms, the ERL will be lowered by approximately 22db (usually resulting in a lower than acceptable ERL).

Most SIP PSTN gateways will give you a few options to select impedance matching on a line or gateway basis. Choose the one that provides the ERL with the largest number.

2. Reduce the SIP PSTN Gateway transmit audio. Reducing the transmitted audio power level will increase ERL by the same amount. If you reduce the original audio source, the echo tail may fall below the perceptible echo range or reduce the echo tail to a level where the echo canceller on the SIP PSTN gateway can have a better effect. However, you want to ensure that you don't reduce the transmit volume to a level that you cannot carry on a normal conversation.

3. Reduce the SIP PSTN Gateway receive audio. Reducing the received audio power level will also increase the ERL. As above you want to ensure that a normal conversation can be carried out.

Reducing the Effects of Delay

Generally speaking, delay is introduced from various sources. The following is a non-exhaustive list:

- Transmission delay (0.005 ms/km x Distance)
- Packetization (Approx 20 ms) and Voice compression delays (dependant on codec)
- Network Transmission delay (Routers, Switches and WAN connections)

To combat the effect of delay, echo cancellation is used in the analog SIP PSTN gateways.

Echo cancellation or echo suppression is a technique that looks for echo tails within the received audio stream. When an echo tail is revealed it will substitute the “tail” with background noise. Echo cancellation involves a DSP (Digital Signal Processor) analyzing transmit audio and subtracting it from the received audio. However, echo cancellers can only work within a limited audio range. The echo cancellers expect that the echo audio be at least 15 dB lower than the original audio. If it is not within this range, the echo canceller cannot distinguish whether it is reflected audio or part of the conversation. Echo cancellers also work within a specified range of delay. The echo tail delay must be within the SIP PSTN gateway specifications. If the echo tail delay is outside the time range (measured in milliseconds) it will not be able to reduce the echo.

Ensure the SIP PSTN gateway has sufficient echo tail to deal with your specific application. Check the gateway specifications to see what the maximum amount of delay that can be accommodated. Also ensure that the level of ERL is high enough to ensure that the echo canceller can distinguish between echo and actual audio.

Reducing Acoustic Echo

Acoustic echo is the echo produced by sound waves traveling between the mouth and handset.

Eliminate the potential that the far end is using a poor quality telephone or a speaker phone. Keep in mind that some environments are worse for echo than others. For example, if a speaker phone is used in a room with high ceilings and an empty room then there will be more challenges with echo than in a small room with wall treatments to absorb the echo.

Test Procedure

Pre-deployment Measurements

It is advisable to test and assess the analog CO lines characteristics prior to deploying an analog gateway. Measuring and capturing the results will ensure that you have a snapshot of the behavior of the analog lines allowing you to determine if conditions have changed after you have installed the analog gateway. An analog gateway should be installed with good quality lines.

When you measure the lines you are assessing the lines for the following items:

- Echo
- Noise
- Electrical characteristics

Test all of the analog CO line prior to deploying an Analog Gateway.

General Procedure for testing Analog Voice lines:

- Measure in-use line current
- Measure in-use line voltage
- Measure db line loss from CO line to Central Office
- Measure ERL – Echo Return Loss.

Measurement	Acceptable Range	Line 1	Line 2	Line 3	Line 4	Line 5	Line 6	Line 8
On-hook DCV	-24 to -50 Vdc							
Off-hook DCV	-5 to -15 Vdc							
Loop Current	23 - 35 DCma							
Circuit Loss	-10 to -3 dBm							
Using 1KHz Tone (dBm) phone number	-10 to -3 dBm							
Echo Return Loss (dB)								

To reduce echo that the SIP Telephone user hears, the following techniques are used:

- Input level – This is typically the gain settings for the audio that is being received from the PSTN.
- Output level – This is the gain level for increasing or decreasing audio levels as it leaves the SIP PSTN gateway to the central office.
- Echo cancellation tail – This is the sliding time window that listens for echo and eliminates it. Echo cancellation algorithms look for the “echo tail” as a portion of the audio that it. If the echo volume is too high the echo algorithm cannot distinguish echo from the actual audio.

Methodology and Procedure

The test procedure takes advantage of two generally available sources of audio tones in the public telephone network. The first is a digital milliwatt telephone number which when called generates a 1 KHz tone at a constant power of 0 dBm (.775 Volts over a resistance of $600\ \Omega$). If that number is called the difference between 0 dBm and the power of the received signal is the receive loss for the trunk. This is the **Receive Loss Test**.

If a silence number is called, a signal transmitted and the power of the received signal measured, the result is a measure of the echo being returned. This is the basis of the **Echo Return Loss Test**. It should be noted that using the above techniques the Transmit Loss over a trunk cannot be measured directly. In general the Transmit and Receive Loss for a trunk are the same. The test procedure assumes that this is true.

The characteristics of the trunks should be the same for a given location. However, there are cases where they vary so they must be individually measured. For normal operation gateways are configured so that outgoing calls are part of a hunt group which chooses a trunk based on a hunt algorithm. This is not convenient when you want to always have a call made on the trunk you are trying to test. There are techniques to force an outgoing call out through a particular trunk which are gateway dependant. See [AudioCodes Test Call Routing](http://kb.adtran.com) technical note, available at <http://kb.adtran.com>, for an example of how to do this for AudioCodes gateways.

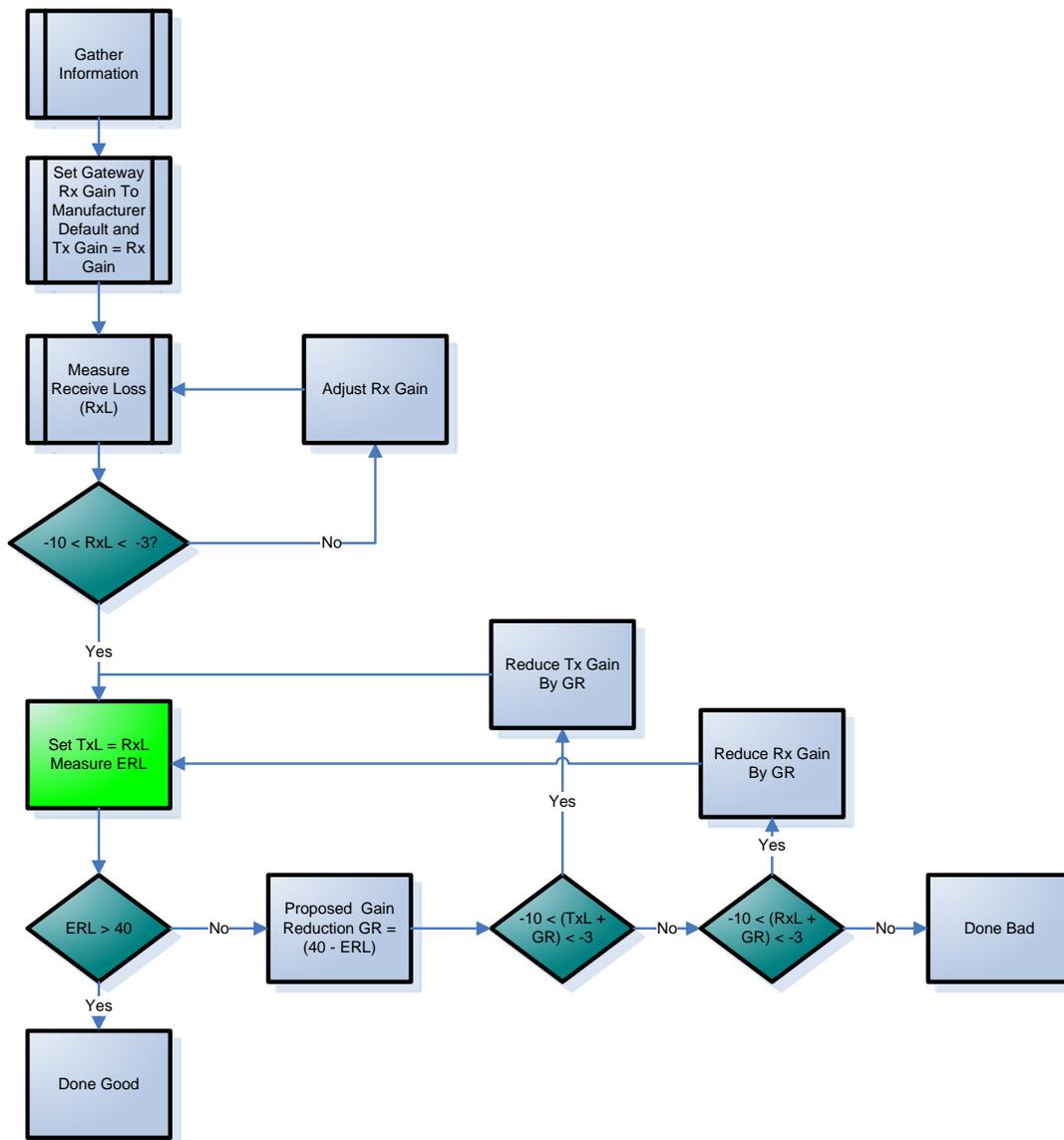
Another note: unfortunately what is ‘**transmit**’ and what is ‘**receive**’ from a gateway perspective depends on whether you look at from the SIP or PSTN side. And of course the gateway manufacturers do not have a common perspective. The following is the terminology used by the gateway vendors and this document:

Gateway	PSTN to SIP	SIP to PSTN
Mediatrix	Input Offset	Output Offset
Vega	Tx	Rx
AudioCodes	Input Gain	Voice
Objectworld	Rx	Tx

The most important measure for echo is the ERL. However before measuring ERL it is important to have the trunk loss levels in an acceptable range. This allows the echo canceller to operate on the echo and achieve the desired results.

The following is a summary of the flow chart below. The flow chart procedure should be followed for each trunk:

1. Set the Rx gain to the manufacturer's default. Set the Tx gain to the same level so that the gain in both directions is the same.
2. Perform the Receive Loss Measurement Test and note the result.
3. Adjust the Rx Gain so that the $-10 < \text{Rx Level} < -3$ dBm.
4. Perform the Echo Return Loss Test.
5. If $\text{ERL} < 40$ adjust the Tx Gain appropriately. If this would put the Tx Level outside of the -10 to -3 range then adjust the Rx Gain by the appropriate amount.



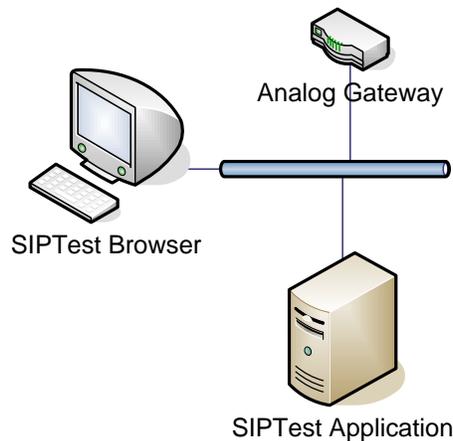
Test Flow Diagram

SIPTester

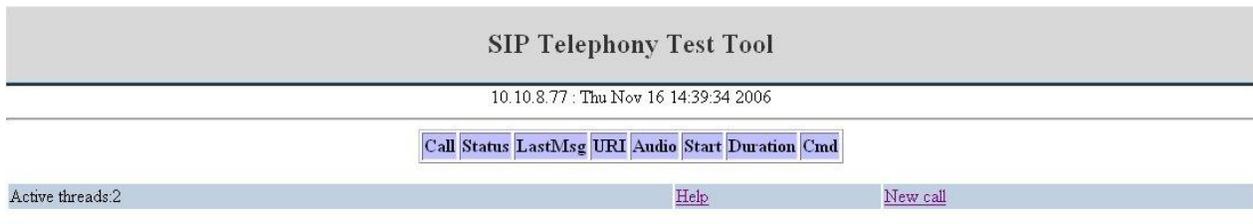
NetVanta uses the SIP Tester application to perform the Receive Loss and Echo Return Level tests. It is a Windows® application that provides the following capabilities:

1. Initiation of outgoing telephone calls to test numbers
2. Real time audio measurements including dBm power level and frequency
3. Real time recording of send and receive audio streams

To install SIP Tester, unzip the application and extract it to a known place (e.g., C:\Program Files\SIPTester). SIPTester does not modify the system registry and is completely self contained within the install directory. You may run the application by executing C:\Program Files\SIPTester\run.bat (for example).



The application runs within a command window. By default, SIPTester uses ports 8000 and 8001. Port 8000 is for SIP communications while port 8001 provides access to a control web page. To access the configuration window open a browser and navigate to `http://<<IP address of computer running SIPTester>>:8001`. You should then see this screen:



Disclaimer: this is a SIP telephony test tool, there is no warranty or guarantee expressed or implied.
(c) 2006 Objectworld Communications Corp.

To initiate a call select **New Call** and configure the parameters according to the test that you are going to perform.

SIP Telephony Test Tool

Make a call

SIP Number to Call:	<input type="text"/>
<input type="button" value="Call"/>	
<input checked="" type="radio"/> Quiet <input type="radio"/> Sine <input type="radio"/> WhiteNoise <input type="radio"/> Speech	
Freq (Hz):	<input type="text"/>
Level (dBm):	<input type="text"/>
Call Duration (seconds):	<input type="text"/>

NOTE: *In the sections that follow, unless there are explicit reasons not to, should be routed through the UC server. In other words, the number dialed should be of the form <<number>>@<<IP address of UC server>>. The UC server will then extend the call to the gateway. Some gateways, in particular the default configuration for AudioCodes, will not work properly if the call is made <<number>>@<<IP address of gateway>>, since they always return responses to the proxy (UC server) not back to the call initiator. There is no downside to making the call through UC server for testing purposes.*

Information Required

The following information needs to be gathered in order to perform the tests.

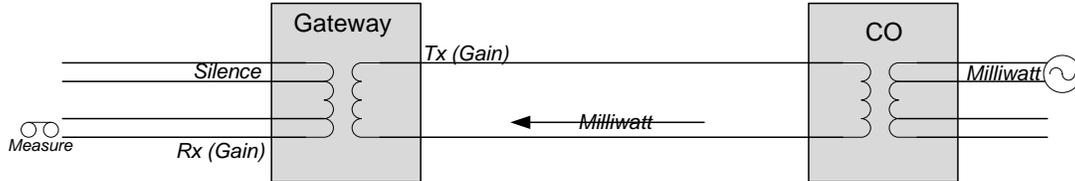
1. Locate your digital milliwatt and silence numbers for your local area. You can use resources such as <http://www.sandman.com/colookup.asp>.
2. Gather Information about the current configuration of your gateway.

Information	Value
Local CO Milliwatt telephone number	
Local CO Silence telephone number	
Gateway Current Input Gain Setting	
Gateway Current Output Gain Setting	
Gateway Current DTMF Gain Setting	

Measurement Procedures

Receive Loss Measurement

The line loss test will initiate a call to the local CO Digital Milliwatt (0 dBm) telephone number and measure the audio level in dBm.



Procedure:

1. Enter the milliwatt test number followed by the gateway IP address in the **SIP Number to Call** field. For example, **6135992111@192.168.1.11**.
2. Select the **Quiet** radio button.
3. Enter the **Call Duration** in seconds. (e.g. **20**).
4. Press **Call**.
5. The progress status will show the current status and any audio measurements.
6. Record the Audio level measurement (-55.78 dBm in the screen below) while the call is ongoing. Depending on how the call is terminated this value may not be retained at the end of the call. This is the Receive Loss in dBm.

SIP Telephony Test Tool

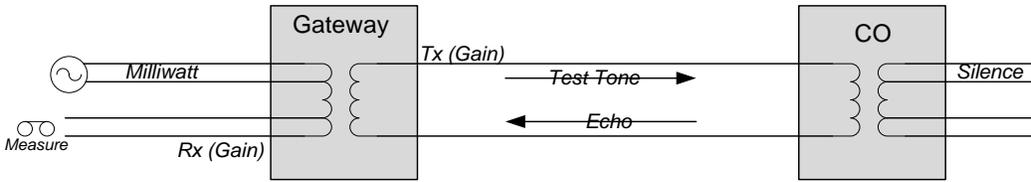
10.10.8.77 : Thu Nov 16 14:48:22 2006

Call	Status	LastMsg	URI	Audio	Start	Duration	Cmd
2451@10.10.8.178	connected	1 INVITE	Q	-55.78 dBm freq=500	14:48:16	5	<input type="button" value="release"/>

Active threads:6 [Help](#) [New call](#)

Echo Return Loss Measurement

The Echo Return Loss Test will send a digital milliwatt tone through the gateway to a silence source at the Central Office. The measured value of the receive audio will be the power of the echo reflected back to the SIP Test.



Procedure:

1. Enter the silence test number followed by the gateway IP address in the **SIP Number to Call** field. For example, **6135992112@192.168.1.11**.
2. Select the **Sine** radio button.
3. Enter 1004 in the **Freq (Hz)** field.
4. Enter the **Call Duration** in seconds. (i.e. **20**).
5. Press **Call**.
6. The progress status will show the current status and any audio measurements.
7. Record the Audio Level as above. This is the power level of the echo signal (EP). In dBm the ERL is $0 - EP$ since the power of the transmitted signal is 0 dBm. For example if $EP = -55.78$ dBm then the ERL is 55.78 dB.

Test Results

You can use the following table for recording measurements and calculating what changes should be made:

Measurement	Acceptable Range	Line 1	Line 2	Line 3	Line 4	Line 5	Line 6	Line 8
On-hook DCV	-24 to -50 Vdc							
Off-hook DCV	-5 to -15 Vdc							
Loop Current	23 - 35 DCma							

First Pass								
Gateway Tx Gain								
Gateway Rx Gain								
Receive Loss (dBm)	-10 to -3 dBm							
Echo Return Test Measurement (dBm)	< 40							
ERL (dB)								

Second Pass								
Gateway Tx Gain								
Gateway Rx Gain								
Receive Loss (dBm)	-10 to -3 dBm							
Echo Return Test Measurement (dBm)	< 40							
ERL (dB)								

Third Pass								
Gateway Tx Gain								
Gateway Rx Gain								
Receive Loss (dBm)	-10 to -3 dBm							
Echo Return Test Measurement (dBm)	< 40							
ERL (dB)								