Nortel Communication Server 1000 and Business Communications Manager

Enterprise Voice Audio Quality

Troubleshooting

Contents

Abstract 3
Audio Quality Impairments........ 4
Defining the Problem 5
  Call Example 1 - IP set to IP set 6
  Call Example 2 - IP set to PSTN residential line 6
Impairments........ 7
  Head Set........ 7
  Telephone 8
  Side Tone........ 8
  IP Network 9
  Virtual/IP Trunk 9
  Gateway Configuration........ 9
  Call Server 10
  Loss and Level Plan........10
  Clocking 11
  Recorded Announcement / Message on Hold........11
  Analog Trunks........11
  Digital Trunks........12
  Public Switched Telephone Network........12
    Residential Lines 12
  Business to Business Calls........13
How to make Test Calls........14
How to perform captures over a longer period of time........14
Appendix A – Echo Tutorial 16
Appendix B – IP Audio Tools........20
Appendix C – TDM Audio Capture Tools........21
Appendix D – Audio Problem Tracking Sheet........22
Glossary 23

New in the release

For a detailed history of this document, see Revision History.

Revision History

Date	Revision
December 2007	Standard 01.01. This document is up-issued to support Nortel Communication Server 1000 and Business Communications Manager. The legal statement is updated, and a New in this release section is added.
April 2006	Standard 3.00. This document is up-issued to support Nortel Communication Server 1000 and Business Communications Manager. Updated document numbering.
March 2006	Standard 2.00. This document is up-issued to support Nortel Communication Server 1000 and Business Communications Manager. Updated formatting and platform support information.
February 2006	Standard 1.00. This document is issued to support Nortel Communication Server 1000 and Business Communications Manager.

Abstract

Although audio quality is subjective, tools and methods are discussed in this document that you can use to objectively investigate and resolve audio quality issues. This document provides an overview of common problems encountered when using Enterprise Voice, provides tools and techniques to resolve audio quality issues, and introduces common troubleshooting terminology used. In addition to reading this document, Nortel recommends that you consult the Engineering Guidelines for the phone, headset, and other connected audio devices.

Audio Quality Impairments

In this document, an impairment is defined as any reduction of the quality of the audio in a call. Audio quality can be impaired by problems with Enterprise Voice equipment, or by problems on the public switched telephone network (PSTN). The following are the most common impairments, applying to both Time Division Multiplexed (TDM) communications, and to Packet Based Audio communications (VoIP or VoFR for example).

- Low/High audio levels – a low signal level can impact audio quality, and a low signal-to-noise ratio can cause the noise in the call path to impair audio. A high signal level can lead to distortion.
- Distortion - any modification of the original audio, such as high audio level clipping, phase shifts, and frequency shifts.
- Noise – background acoustic noise, or noise electrically induced in the call path.
- Echo – an audible reflection of the speaker's voice, delayed in time.
- Delay – the length of time it takes for the talker's audio to reach the listener.
- Clock slip – differences in clocking references can cause clicks and pops due to audio deletion.
- Cross Talk – audio from a different call can be heard.
- No audio or one-way speech path – neither party can hear or be heard, or only one party can hear the other.

The following impairments only apply to Packet Based Audio communications:

- Packet Loss – loss of audio information due to data network issues.
- Jitter – variation in the delivery rate of packets across the network.
- Compression – the use of audio compression techniques to reduce bandwidth requirements. This includes Transcoding as well as Multiple Encodings.

Further sections in the document describe where these impairments can occur.

Defining the Problem

Defining the problem to be solved is usually the most difficult aspect of audio troubleshooting, due to the subjective and often intermittent nature of audio problems. A good approach is to interview a sampling of users, and use the resulting data to look for common characteristics that can help you define what conditions lead to audio problems.
Interview the users who reported a problem, and other users who did not report any problem. Use the following guidelines when interviewing users:

- Ask open ended questions such as "Describe what you heard". Do not ask "Did you hear echo?"
- Try to get details on the call path – who was called and how was the call established. For example, a call between an ACD agent and a residential home phone is a call path, but the real call path was PSTN caller dialed IVR system which transferred to a Branch Office via Virtual Trunk to an IP Phone.
- Look at the equipment on the user's desk. Note the phone type in use. Note the accessories in use (headset, amplifier, recording device etc.).
- Look for any patterns in the trouble reports. For example: Are problems more prevalent on inbound calls versus outbound? Are problems reported on calls between two phones on the local system? Is the time of day a factor?
- Make note of the background noise in the environment.
- Does the user use handsfree, handset or headset?
- If echo is described, try to determine when and for how long echo occurred in the call. Was echo heard only at the beginning of the call, throughout the call, or intermittently during the call?

If brute force data collection must be used, try the following techniques:

- Configure the telephone with a special key or key press feature that creates a debug log with a date/time stamp of a problem call. On CSE1000 systems, you can use Malicious Call Trace or an Activity key. On BCM, you can use Feature 9*9 to provide a MTT Stamp in the system logs.
- Provide a select group of end users a data collection sheet that contains the following columns:
  Date/Time, Internal/External Call, Headset/Handset/Handsfree, Problem, Constant/Intermittent. If possible, provide check boxes to make it easier for users to provide feedback.
- Have the users email the completed form to their Help Desk

Review the data you collected by interviewing users and analyze the call path. Look for a pattern that helps you understand the problem.

Node Based Troubleshooting

In this document, a node is defined as is any device that can affect the audio characteristics of a call. By looking at the audio path as a sequence of clearly defined nodes you can verify that each node is correctly configured, and apply appropriate troubleshooting techniques.

Following are two examples of a call path and the nodes in the path.

Call Example 1 - IP Phone to IP Phone

Problem reported – Caller A finds the audio from Caller B loud and distorted. Caller A does not experience this with other callers. Caller B does not report any problems.

Node 1 – Caller A headset
Model – GN-Netcom – on adapter (not batteries)

Node 2 – Caller A IP Phone
Model – Nortel i2004/P2 running D98 software

Node 3 – IP Network
Model – obtain a diagram of the network

Node 4 – Caller B IP Phone
Model – Nortel i2004/P2 running D98 software

Node 5 – Caller B headset
Model – GN-Netcom – on adapter (not batteries)

Call Example 2 - IP Phone to PSTN residential line

Problem reported – Caller A (ACD agent) receives an inbound queue call from a home residence and hears a low level echo throughout the call.

Node 1 – Caller A headset
Model – GN-Netcom – on adapter (not batteries)

Node 2 – Caller A IP Phone
Model – Nortel i2004/P2 running D98 software

Node 3 – IP Network
Model – obtain a diagram of the network

Node 4 – Voice Gateway Card (VGW)
Model – 32 port SMC running 4.00.55 software

Node 5 – PRI card
Model – Dual Port PRI card

Node 6 – PRI Span
Model – Acme ILEC local PRI

Node 7 – Residential Phone Line
Model – Acme ILEC local loop

Node 8 – Residential Phone
Model – Cordless

Now that the call path is broken down into manageable pieces, we can consider a number of common impairments at different nodes in the call path, and how to resolve them.

Impairments

This section discusses the factors that can have an impact on audio quality.

Headset

Headsets provide flexibility and comfort for users. However, because each user can have a different brand and model of headset, it is difficult to predict the audio quality experienced by each user. Wireless headsets add additional issues because they are sometimes prone to delay and information loss. To minimize headset problems, take the following steps:

- Use the mid range on the side tone (if adjustable on the amplifier) and TX/RX levels whenever possible to stay in the linear operation of the amplifier(s). Once the telephone and headset audio levels are optimized, use the headset receive volume level slider on the headset amplifier (if applicable) for minor adjustments during calls.
- Headset microphone placement can also have a large impact on audio quality and overall noise levels. Follow the manufacturer recommendations. A general rule is 1 inch away from the corner of the mouth (just out of the breath stream from the nose and mouth).
- Verify the power source – batteries or AC wall adapter. If noise or hum is present, try batteries instead of the AC power.
- When troubleshooting echo or noise problems, use the MUTE button on the headset amplifier to help isolate the source of the echo/noise. For instance, muting the amplifier removes the headset as the noise source. Muting at the phone itself removes the amplifier and headset as possible sources.
- The Nortel Wireless i22xx handsets have a configurable Noise Mode which can improve the signal-to-noise ratio of the transmitted audio by reducing the

microphone sensitivity. However, when the user leaves the noisy environment for a quiet environment, the transmitted audio can be too low in volume.

PC based IP Telephony such as the MCS5100 Client or the i2050 utilize a Nortel tested and approved USB Audio Headset. This headset is recommended to ensure that the audio quality is the best possible since it does not rely on the PC Audio Card.

Telephone

All telephones are not manufactured to meet the same standards, and therefore audio characteristics vary from one telephone to another. The use of handsfree and cordless telephones add even more possibilities for impairments.

Check the receive volume level settings on the telephone and set it to mid range whenever possible. (Use the same method for handsfree operation). Using the mid range volume reduces the chance of acoustic coupling between the speaker and microphone.

If handsfree is in use, observe the background noise levels and placement of the phone in relation to the user – is a good signal-to-noise ratio possible? In other words, can people hear the speaker over background noise?

Most TDM handsfree telephones use an echo suppressor – loss is inserted to control acoustic echo generated at the handsfree telephone. Newer TDM and IP-based telephones use a Digital Signal Processor (DSP) to cancel the acoustic echo that is created. DSP-based cancellation is often smoother and less intrusive in the conversation flow than echo suppression.

A worn or faulty handset curly cord can introduce echo via EMI.

Cordless phones often use audio companding/compression to reduce the radio bandwidth required, automatic gain control, and noise cancellation techniques. These cordless phone characteristics can cause echo issues to occur because the echo tail path appears to be changing. To determine if this is a problem, test the connection using a wired 2500 telephone and see if the problem is resolved.

Side Tone

Side tone is the name given to the small amount of audio from your telephone speaker that is picked up by the telephone microphone. The purpose of side tone is to provide feedback that the phone is operating, and to mask any echo. Due to the development of telephony standards, the frequency characteristics and loudness of the side tone is different than on older TDM telephones. This change can cause users to report a problem. If side tone concerns are reported:

Capture the bidirectional audio using Ethereal or a TDM recorder. Compare the captured audio with the user’s experience. If the issue is side tone, the problem is not audible in the Ethereal or TDM recordings.

Record the handset audio via a device such as the “THAT-1” from www.jkaudio.com

IP Network

Packetized telephony has had a significant impact on system design. Systems can now be distributed across the world, yet share the same 4-digit dialing plan and utilize direct media paths between endpoints. Data applications can tolerate packet loss and delay, while voice applications need reliable and timely packet delivery to provide a quality conversation. A discussion of the tools and techniques used to optimize the quality of a packet network is well outside this document, but some of the critical points are as follows:

- Review the L2 and L3 data device port statistics. Look for CRCs and other error indications.
- A data network cannot create echo. A poorly performing data network can increase the delay experienced and therefore make the echo more apparent. Reduce delay whenever possible via QoS techniques and by using the smallest packet payload the network can tolerate – for instance 20ms of audio per packet.
- Packet loss on the network can lead to distortion of the audio. Many Nortel products can provide statistics on the call quality and therefore the level of packet loss.
- External tools can be used to test and measure the packet path (Chariot, NetIQ, Vivinet and so on), or the actual audio packets can be captured and reviewed to determine if packets are missing (Ethereal or Sniffer).
- Wireless LANs (WLAN) require additional engineering before deployment which can greatly offset the cost of troubleshooting after deployment. Frequency (channel) interference, power and coverage must be engineered correctly to provide the best packet network quality.

Virtual/IP Trunk

A Virtual or IP Trunk is a media path between systems or phones that passes the audio in a packet stream.

- All of the recommendations discussed under the IP Network section apply to Virtual or IP Trunks.
- Additional considerations can arise around loss and level planning. See Loss and Level Plan on page 10 for more information.

Gateway Configuration

A gateway is a device that processes packets and forwards the audio and signaling to and from the TDM domain. The gateway is a critical point in the call path since it contains many individual functions and features:

- The Gateway must remove jitter that is introduced on the packet network. Ensure that the Jitter Buffer is configured to the appropriate level of no less than 2 times the payload – for instance, 20ms payload per packet = 40ms Minimum Jitter Buffer Depth.

Voice Activity Detection (VAD) can affect the audio quality because it attempts to detect and then simulate silence to save network bandwidth. Make test calls with VAD enabled and disabled, if the supported by the CODEC.

High compression rates such as G.729 can be detected by some users as distorted audio – for example, users report that the audio is not “crisp” or it does not have the “fidelity” they expect. Try using a compression rate such as G.711 to see if this corrects the problem.

Multiple compressions can cause distortion. For example, if the system records a Voice Mail message via a G.729 call, and then later replays the message via a G.729 call, a total of three compression cycles can result (since most Voice Mail systems also compress the audio to save storage space). Most users find the resulting audio quality unacceptable.

Enable the Echo Canceller on the Gateway under all situations, unless an external canceller is used on the TDM side of the trunks.

Call Server

The Call Server controls the set up and establishment of calls. It is possible that calls can occur where no speech is present, or the speech path is only one-way. To resolve this, perform the following:

Perform network packet traces on the T-LAN of the affected phone and the E-LAN of the Call Server.

If the situation can be duplicated, create a conference call with a third party to determine if they can hear and be heard.

Loss and Level Plan

The audio levels used on a call are critical to the call quality. If one or more callers is hard to hear, too loud, or if there is echo on a call, users can become frustrated or distracted. Check the following:

The communications equipment and the Central Office equipment must be compliant to the current TIA/ITU recommendations. These standards are designed to provide the highest signal to noise ratio possible, with echo controlled via loss insertion.

Calls that utilize a Virtual Trunk or traditional trunk can have Gain or Loss applied to the audio. Ensure that the default Loss Plan values are being used unless a custom Loss Plan is required. A default Loss Plan is normally designed to meet the TIA/ITU standards.

The complexities of Loss Plan configuration are outside of this document; for more information, refer to the following documents:

(CSE1000/Meridian) *Transmission Parameters 553-3001-182*

(BCM) *Programming Operations Guide N0008589*

Clocking

The clock reference used by the Communications System must track and lock to a high-precision clock reference. If this reference is not stable, clock slips can occur. A clock slip is the equivalent of packet loss in the TDM domain, and causes an audible click or pop. Clock slips can also result in echo since a TDM timeslot deletion is equivalent to a time shift, and therefore a change in the echo tail path occurs. Check the following:

- Verify that the PRI/DTI-based clock references are not slipping.
- Verify that each expansion cabinet or system is tracking to a higher precision source (ensure that it is not free running).

Recorded Announcement / Message on Hold

RAN and MOH audio levels are often configurable within the RAN/MOH unit itself. Without a standard to calibrate the level, you must subjectively adjust the level using a cross reference of TDM and IP telephones if possible, and PSTN lines (if PSTN users hit the RAN/MOH). Check the following:

- A RAN/MHO message that is too quiet is less likely to cause a problem than one that is too loud.
- Verify the Terminating Impedance used on the connection between the RAN/MHO and the trunks. A mismatch can lead to noticeable echo if listeners talk during the RAN/MHO.

Analog Trunks

Analog trunks are often the weakest link in a call path. It is quite possible that a communication system can be installed with analog trunks for years with no reported problems, but the first call made over the trunks using an IP based telephone experiences echo. The echo was always there; however it is reflected back so fast that users do not perceive it unless an IP Phone is connected. To optimize the audio quality across the analog trunks, check the following:

- The Loss Plan engineering on the trunks is critical, because incorrect levels can lead to increased echo.
- Verify the communication system configuration of the trunks in the following areas:
  - Terminating Impedance – must match the CO configured value
  - Balance Impedance – determined by wire length and service type
  - Transmission Compensation – determined by CO compensation values

    ***Note*:** On the CSE1000, a trunk Class of Service of Non-Transmission Compensated (NTC) inserts 0dB Tx and 3dB of gain on Rx in the call path. If echo is strong or convergence is slow, change the CLS to Transmission Compensated (TRC) which inserts 4dB Tx and 1dB Rx losses.

Check the jumpers on the XUT/EXUT card, and ensure they match the settings recommended in the product documentation.

Measure each trunk for:

- Noise
- Echo Return Loss (ERL) – this is the level of echo reflection
- Loop current
- Circuit loss

Ensure that the trunk specifications are met as outlined in the (CSE1000/Meridian) ***Transmission Parameters 553-3001-182***

Digital Trunks

Digital trunks are very rarely a source of audio problems. However, if you suspect that the digital trunk is causing a problem, check the following:

Are the trunks digital all the way into the PSTN? It is possible that the digital trunk interfaces to a Line Concentrator which takes multiple analog lines and creates a digital trunk by multiplexing. If so, the issues identified for Analog Trunks on page 11 apply, such as impedance matching between the Line Concentrator and the CO, as well as noise on the trunks.

Individual timeslots in the trunk can fail when hardware failures occur on the CO and PBX side. Test each timeslot via controlled call testing.

Public Switched Telephone Network

After the audio reaches the PSTN, a number of possible audio impairments can occur, as follows:

Analog trunks can exist in the call path, and these trunks can be configured incorrectly, resulting in noise, audio level or echo issues.

Packetized trunks can be used, and therefore packet loss and jitter issues can be experienced if the trunks are not engineered correctly.

Loss and Level Plan standards have not been followed or are not up to date.

Residential Lines

Residential phone lines vary in quality. Quality is measured in a number of different ways, and certain characteristics are critical for modem/data connections that can also cause issues such as echo.

Check the following specifications, which must match the service provider test guidelines:

Line loss in dB – reported as –x dB or just x dB

Noise Level in dBrnC (dB with a specific “weighting” or averaging)

Power Influence – reported in dBmC is the effect of 60 Hz AC EMI on the line

Line length in feet or meters – ensure that the line length is the copper line length, not the distance to the CO. For example, there can be a line concentrator on a telephone pole half way to the CO.
If a line concentrator is used, make note.
Line resistance (DC Ohm reading) – have a short placed at the Central Office (CO) and measure from the residential side.
Balance – a relationship between loss and noise
On hook DC Voltage reading
Off Hook DC Voltage reading
Measure loop current when off hook
Make test calls. Listen to the audio quality.

Compare the test data to Table 1. Many POTS lines have taps or load coils installed to improve long distance performance, which can also be a source of echo.

Table 1: Residential line test comparison values

Parameter	**Min**	**Max**	**Troubleshooting Steps**
Line Loss	0 dB	-8.5dB	> 10dB is considered very poor
Noise Level	0 dBrnC	20 dBrnC	>30 dB is considered poor
Power Influence	0 dBmC	80 dBmC	> 90 dBmC is considered poor
Balance			60 dB or more ideal IF noise and Power Influence are ok.
On Hook V	-48 VDC	-54 VDC	Must be between values
Off Hook V	-8 VDC	-12 VDC	Must be between values
Loop Current	20 mA		Off hook current

Note: the values shown in Table 1 are North American standard values

Business to Business Calls

The communications equipment at the business you are calling can also affect the audio quality in all situations. Determine if a problem occurs when calling a specific business or extension at a business, and then determine what impairment is affecting the call.

How to make Test Calls

Making test calls can pose a problem because people act differently when they know they are being recorded, which can impact test results. The following guidelines can help:

- Take detailed notes of the call path, endpoints, volume levels used, dialed numbers, time of day, and so on. All of these things can assist in narrowing down the list of possible problems.
- Synchronize the time on all PCs involved with the test.
- When using Ethereal or TDM Recording, state the information into the audio file. For example "This is Call 4 from i2004 to residential line 555-1212 using headset on i2004. I am experiencing a low level echo whenever…."
- Do *not* call automated telephony systems such as IVR or Voice Mail and attempt to perform echo tests just because you do not have another body to answer your test calls. Many non-Nortel IVR and Voice Mail systems are great echo sources, and do not represent a real-world situation. Also, talking over top of a greeting does not allow the ECAN time to converge and train on the echo.
- Speak normally, and do not keep saying "Check…check" since this does not provide a realistic signal. Instead, read something, talk about the weather, sports, politics, or whatever provides the normal conversation dynamics.
- Name all the files in a logical fashion. For example, name the Ethereal, GL-Com and PBX D-channel files all with the same information such as "Call5-i2004toResidential".
- Keep the files small and manageable. Keep Ethereal files to 10MB or less for ease of handling and review. Use file compression, such as zip, to save space.

How to identify problems over a longer period of time

Identifying intermittent audio problems can be difficult; use the following techniques to improve your chance of success:

- Select a group of users that are willing to be part of the "test" group
- Select a PRI that can be used for outbound calls by the test group. The ideal PRI has a D-channel and does not use NFAS.
- Create a new route in the route data block that allows the test group to dial an access code to target the PRI selected above.
- Install the GL-Com or equivalent PRI recorder on the PRI
- If the audio problems are being reported by IP Phone users, add a new zone to the Node and move these test group into the new zone
- Change the TN configuration on a Media card to include the new zone on a single card.
- Connect a packet capture tool (such as Ethereal) to the network on a port that mirrors the media card with the new zone
- Configure Ethereal to provide a continuous capture, creating a new file every 10MB.

Instruct the test group on how to dial the ACOD of the PRI, or configure a Speed Dial key for them to do the same

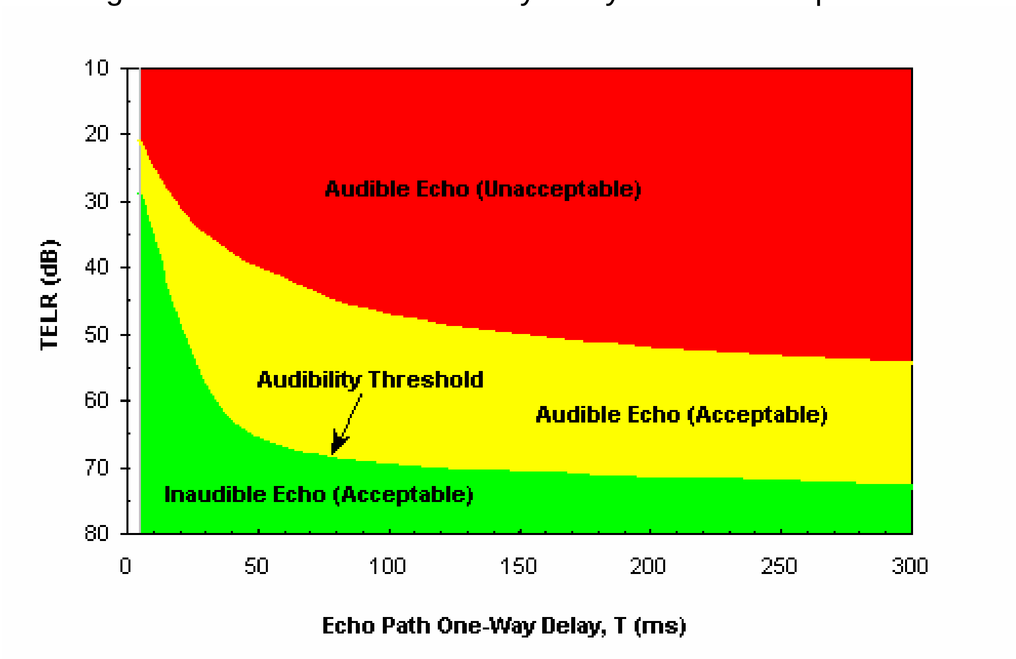
When a problem call is reported, check the PRI and packet capture files for the timeframe reported.

Appendix A – Echo Tutorial

Echo has been present on the public network since the first telephone call. On most telephone calls over a short geographical distance, the audio delay is so low that if an echo is present, you can not detect it.

Figure 1 shows the effects of one way delay and echo amplitude on the user's perception.

Figure 1: The effects of one way delay and echo amplitude



TELR = Talker Echo Loudness Rating

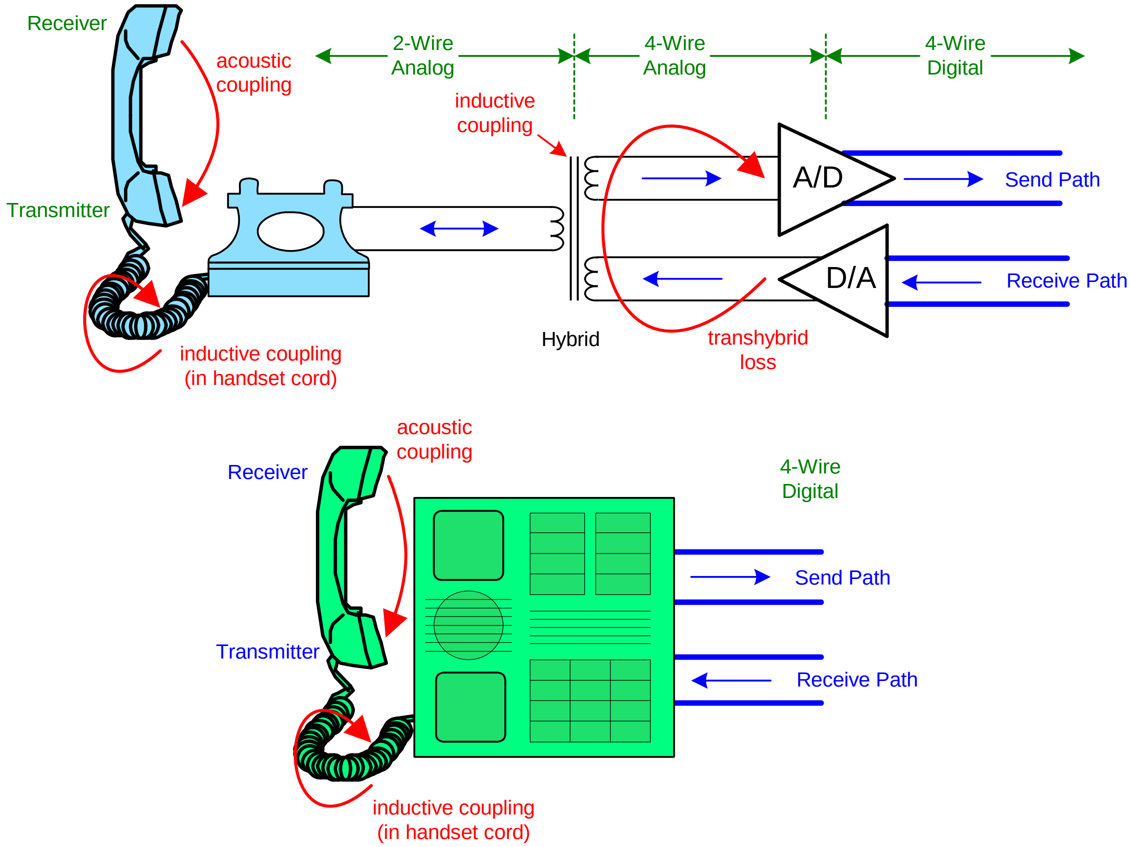
Packet technologies such as VoIP introduce additional delay in the call path due to packetization, transmission and buffering, which can cause the echo to become noticeable.

For example, consider a G.729 call made with 30ms packet payload:
G.729 Compression (15ms) --- LAN/WAN(10ms)-----GatewayJitterBuffer(60ms)-----Public Network

The one-way delay is 85ms plus any delay experienced on the PSTN. This results in an echo reflection being delayed by 170ms; if uncorrected, this is perceived by users.

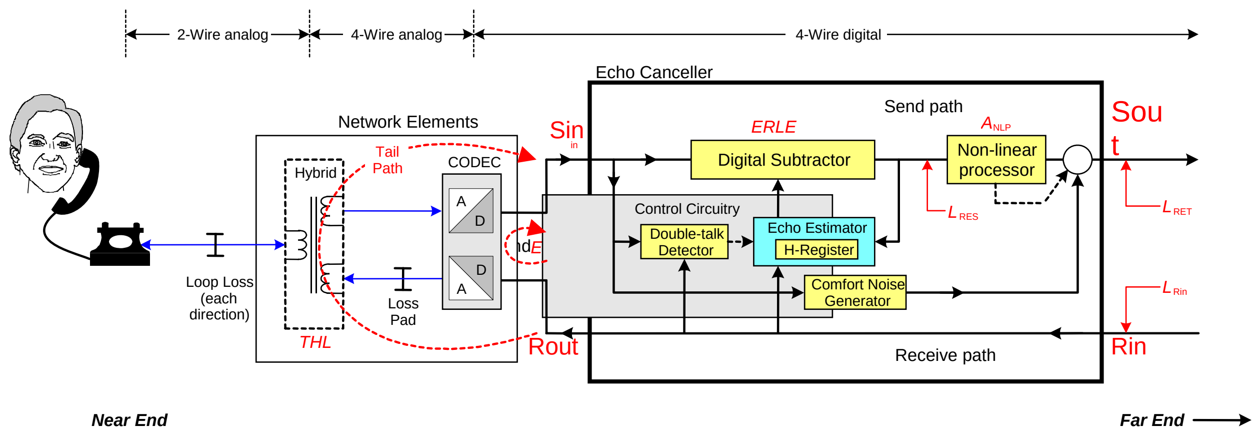
Figure 2 shows where echo is generated in the call path.

Figure 2: The most common sources of echo on analogue and digital telephones



There are industry-wide standards to cover the acoustic and electrical properties that telephones must meet, but not all manufactures adhere to these standards. As a result, an echo source can be present in the call path. An echo can be corrected using an Echo Cancellation module (ECAN). Figure 3 illustrates the inner workings of an ECAN.

Figure 3: An ECAN

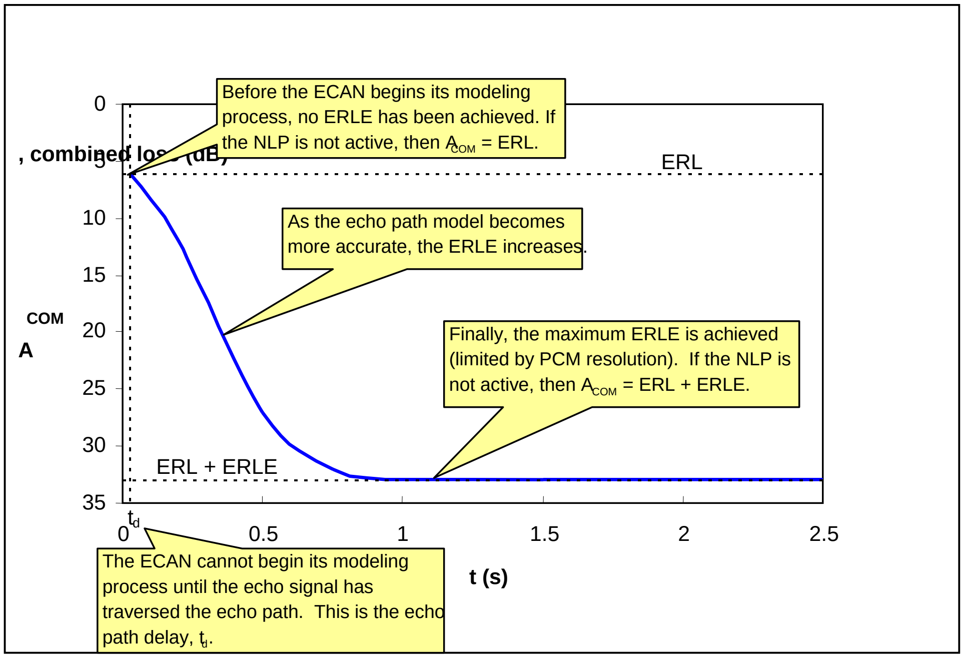


ECAN operates on a simple premise. The audio at the Rin point is compared to the audio at the Sin point. Through the power of a Digital Signal Processor (DSP) the difference in the audio streams are analyzed and a model of the Tail Path is created. This model is then used to digitally subtract the echo from the Sin stream. Any echo that remains is called Residual Echo and is at the Sout stream.
There is another feature in the ECAN called the Non-linear processor (NLP). Consider the NLP as a smart volume control that can quickly attenuate or disengage to allow audio to pass. If the VoIP user is talking and the Near End user is silent (Rin is high and Sin contains only echo), the NLP engages and therefore prevents the residual echo from being heard by the Far End. If the Near End person speaks, the NLP must disengage to allow the audio to pass.

However, there are times when echo can be heard at the very beginning of a call, such as when the Far End says “Hello”. The reason for this small burst of echo is that it takes a period of time for the ECAN to create the model of the Tail Path. If the Tail Path is highly complex, such as in a conference call, the time to converge on the Tail Path can be longer. Figure 4 illustrates the steps the ECAN must go through before convergence is complete.

Note: The Echo Return Loss (ERL) on a purely digital call is higher when compared to a call across analog trunks or to a residential phone line. This is why purely digital calls rarely experience initial echo.

Figure 4: Steps required for convergence.



ACom – Combined Loss showing the ERL and ERLE added together
ERL – Echo Return Loss – the level of the echo measured at the Sin point
ERLE – Echo Return Loss Enhancement – the reduction in ERL due to the cancellation process

For more information on Echo Cancellation, please see the ITU-G.168 guidelines which is the industry standard on ECAN specifications.

For a complete version of this Echo White Paper click on the following link: http://navigate.us.nortel.com/Bay+Documents/White+Paper/nn111662_pss5.pdf?DMW_FORMAT=pdf

Appendix B – IP Audio Tools

Due to the growth of VoIP systems in the marketplace, new tools are being developed every day to monitor, measure and report on packet performance, and therefore audio performance.

The tool that is most often used by Nortel Enterprise Technical Support is the network packet capture tool. You can find an example of this tool at www.ethereal.com. Ethereal has the built-in capability to extract G.711 RTP audio packets and align them into an audio file that can be listened to.

To capture the VoIP audio stream, perform a capture on a hub or a data switch that has a port mirrored to facilitate the trace.

Here are the steps to extract the audio from a capture file:

1. Open the Ethereal file
2. You can easily filter the file with "**ip.addr==x.x.x.x"** where the address is one of the addresses of interest.
3. Find a UDP packet to/from the devices of interest
4. Right click on the UDP packet --> Decode As ---> select RTP from the right hand drop down box AND select BOTH in the center UDP source box
5. Click OK
6. Scroll through the file to verify that all UDP audio packets have been correctly tagged as RTP
7. From the top menu select Statistics --> RTP --> Show All Streams
8. A window appears showing all RTP streams in a roughly chronological order. Select the first direction by left clicking, then the reverse audio path by shift-left clicking.
9. Click Analyze
10. Note the packet loss and jitter values reported in the Analyze window. Are they what you expect?
11. To save the audio for listening, click on Save Payload
12. Save the audio in 2 files - a "forward" and a "reverse" as identified by the check boxes which represent the 2 audio streams. Use an "au" extension on the file name.
13. When you click OK on the Save Payload page, the window minimizes to the task bar. Click on it to restore.

Note: You cannot listen to a combined or "both" (forward and reverse) audio file and hear any echo. To hear the echo, combine the forward and reverse files, creating a stereo file. This allows you to hear the original audio in one ear, and the echo in the other.

Appendix C – TDM Audio Capture Tools

Capturing the audio from the data network only provides half of the information required; a TDM recording device is required.

Nortel Enterprise Technical Support uses T1 recorders manufactured by GL Communications www.gl.com. The Ultra T1/E1 or Laptop T1/E1 product allows for the monitoring of multiple TDM trunks with the capability to record the audio on the trunks on a per timeslot basis.

To record audio at a TDM telephone, numerous analogue systems exist such as those manufactured by www.jkaudio.com.

To record the audio in a digital format directly from the TDM stream, see the AuxBox series from www.algosolutions.com.

The list of manufacturers and products provided above does not represent a complete list, and Nortel does not endorse any specific product.

Appendix D – Audio Problem Tracking Sheet

Table 3 contains the most common information required to track down audio issues. Provide a copy to each person who is willing to track issues.

Table 3: Problem tracking sheet

Date	Time	Incoming/Outgoing	Caller Id	Problem Observed
Jan 13	2:40pm	In	214-555-1212	Echo at the start of the call

Modify the Problem Observed column – or add more – if you are trying to capture a specific issue so that the user can just put a check mark in the column. A simpler form yields more data than a complex one.

Glossary

Companding – an analog to digital conversion technique that increases the signal to noise ratio by using a non-linear scale. This gives low level audio a higher digital resolution than loud audio signals.

Echo Canceller (ECAN) – a Digital Signal Processor (DSP) based device that compares the original audio with the reflected (echo) audio and attempts to mathematically “subtract” the echo. The ECAN must first calculate or converge on the delay and amplitude characteristics of the echo.

Echo Return Loss (ERL) – ERL is a measurement in dB of how strong an echo is when compared to the original audio signal. If an echo reflection is at the same amplitude as the original audio, the ERL is 0dB. If the echo is half as loud, the ERL is 3dB which indicates “3dB lower that the original”. Most Echo Cancellers require that the ERL be at least 6dB or greater for correct operation.

Echo Suppressor – a technique of echo removal by inserting additional loss in the echo path. For example, when Caller A is talking, an echo is being created from the phone of Caller B, and the echo suppressor inserts loss in the path from Caller B so Caller A cannot discern the echo.

Gateway – a card or device that coverts packetized audio (such as VoIP packets) to TDM (such as a PRI).

Impairment -- a term that was introduced in the ITU E-Model. An impairment is any audio characteristic that affects the quality of the call. The E-Model provides a method of calculating audio quality with known impairments in the call path.

Non-linear – a system is described as linear if a given input results in a given predictable output. A non-linear system’s output is not predictable. Most analog electronic circuits are designed to operate in a linear fashion (amplifiers and hybrids, for example), but if the input signal is too strong, non-linear (distorted) output can result.

Side Tone – most telephones are designed to play back a small portion of the transmitted audio in the ear piece. This audio gives the talker the sense that the phone is operating correctly as well as providing other psychological benefits. Side tone can also mask echo that has a very small delay.

SLIC – a “Subscriber Line Concentrator”. There are two different ways of using this technology. The first is to concentrate multiple residential lines into a digital trunk such as a T1. The other method works on a T1 that is servicing a PBX, breaking it down into

24 analog trunks connected to the CO. This second method is often used to make use of existing wiring or CO equipment.

Transcoding – the conversion of audio from one compression bit rate to another. For example, a call traverses a G.729 8kbps trunk, then across a G.726 32kbps trunk.

Nortel Communication Server 1000 and Business Communications Manager

Enterprise Voice Audio Quality Troubleshooting

Publication: NN43001-705
Document Status: Standard
Document version: 01.01
Document Date: 7 December 2007

To provide feedback or report a problem in this document, go to www.nortel.com/documentfeedback

NORTEL