Echo and Echo Control

A White Paper

February 24, 2006

Issue 0.4, COMPAS ID 106541

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

1. Introduction

It's probably safe to assume that everyone reading this document has heard echo occur in one form or another. Echo, simply stated, occurs when you hear yourself speak. Not so simply stated, echo involves many factors such as delay, your system configuration (for example: analog trunks, type of phone, speakerphones, handsets, etc.), room acoustics, levels, etc. The purpose of this white paper is to explain the various kinds of echo, discuss the products and features that exist within Avaya to reduce echo, list some ways to isolate/troubleshoot an echo source, and outline options that may reduce echo based on trouble shooting findings.

2. What is Echo…echo…echo…

Echo occurs and exists on almost every voice call made. In fact, echo has always existed in traditional voice environments and is not new in the Voice over IP (VoIP) configurations. Furthermore, a certain amount of echo, called side tone, is very acceptable to most individuals. Side tone occurs when the speaker can hear himself/herself speak through the receiver. In this sense, side tone actually gives the speaker some reassurance that his/her conversation is going through and can be heard on the other end. However, when side tone becomes too loud, it is unacceptable.

What makes echo actually noticeable is delay. If there is enough delay between the talker's utterance and the time of arrival of the echo at the ear, the talker will begin to perceive the echo as problematic, and not as side tone. When the time it takes for the echo to return to the talker's ear (sometimes called the round-trip echo path delay) reaches around 30ms or greater, some users will begin to notice the echo as problematic or objectionable echo. With a little less delay, that is, a round-trip echo path delay of around 10-50ms, users may still find the experience objectionable but may perceive the echo as reverberation or they may report a tunnel-effect or muddiness in their side tone. Only with very little delay in the echo path will the echo be unnoticed and simply blend in with side tone.

As people have become more comfortable with Internet/Intranet networks, they are beginning to move their voice applications from the traditional, TDM circuit-switched environment to an Internet Protocol (IP) environment. (Just to clarify, an Intranet is a private network that uses Internet software and standards. It is typically used by employees and/or customers of a company who have been given authorized access to use that network). In general, longer delays exist over the Internet/Intranet. And while those delays are tolerable with respect to data, the increased delays in VoIP-based systems are the main reason that echo is now more perceivable than in the past (non-IP networks). Again, the VoIP-based telephony system is not creating the echo. The echo was always there. The delay introduced by the VoIP system causes the echo to be noticed. So, even with the lowest-delay options that a VoIP system can offer (that is, G.711 voice with 10ms packetization interval) the overall round-trip delay will still be on the order of 100ms which is still enough for any echo to be perceived as annoying. Any time a phone call is made to traverse an IP network (which means the voice samples are bundled into IP packets) a significant delay results and is unavoidable. (Note: The primary contributors to the 100ms delay include packetization, encoding and decoding, often resulting in jitter). This means that in those cases, it is much more important to make sure echo sources are controlled.

In general, there are several factors that one should always remember with respect to echo:

- Echo has always existed on voice calls
- Echo is caused by electrical and acoustical signal reflection points (2-to-4 wire hybrid circuits and the loudspeaker-to-microphone air-coupling path in speaker phones, headsets and handsets)
- Echo becomes perceivable when there are excessive delays in the network path

- It is always best to cancel/alleviate echo at its source
- The equipment used by the person hearing the echo is usually not the source of the echo (exception: excessive side tone levels).

2.1 Echo Sources

Echo sources are either acoustic or electrical-hybrid in nature. Factors that aggravate the perception of echo include signal level (round-trip echo return loss), round-trip delay, and the character of the echo path itself (reverberant, non-reverberant, linear, or non-linear).

2.1.1 Acoustic Echo

Acoustic echo occurs when a talker's voice is acoustically coupled (acoustically meaning "thru the air") from a loudspeaker back into a microphone. Acoustic echo can occur in analog and DCP telephones in speakerphone mode. It can also be heard based on the position of a speakerphone with respect to reverberating surfaces (for example, walls, ceiling, shelving, laptops placed too close to the speaker phone, etc.). Basically, acoustic echo can occur anytime there is an acoustic coupling between a loudspeaker and a microphone. Some degree of acoustic echo also occurs on telephone handsets and headsets.

In general it is always best to control or cancel echo at a point closest to the source of the echo. Therefore, a phone is responsible for controlling its own possible sources of acoustic echo, especially if it is going to be used in a higher-delay environment such as VoIP telephony.

Acoustic echo is a particularly difficult echo to control since it can often be very loud, reverberate for a long time, and/or change quickly and radically during the course of a conversation. A special acoustic echo canceller is needed to handle these behaviors. A good acoustic-echo-canceling speakerphone will satisfactorily eliminate the echo without causing a half-duplex experience, and therefore should be used in a VoIP telephony system. Avaya IP telephones with speakerphones have an acoustic echo canceling speakerphone and handset.

An important fact to note is that millions of phones and speakerphones used today were designed *before* VoIP telephony existed. These older speakerphones/phones were not designed for a high delay environment that may result when involved in a VoIP–based phone call. Therefore, less stringent control of echo by the terminal is available when calls involving "older" equipment are used in VoIP environments. In such cases, in-network echo controllers must be relied upon.

2.1.2 Electrical-Hybrid Echo

Hybrid echo, also called line echo or electrical echo, is typically caused by impedance mismatches in the analog telephone network (where impedance is the resistance to flow of an electrical current in a circuit). In a traditional telephone call involving analog telephone sets connected through the (digital) PSTN, carrier use 2-to-4 wire voice frequency hybrids on each end of the call to convert between the analog and digital signal domains. As voice signals are transmitted across the network and reach the hybrid (a device that converts a 2-wire telephone circuit to a 4-wire telephone circuit and vice versa), electrically generated echo can result because the energy in the 4-wire portion may reflect back on itself. That is, due to the impedance mismatch, the entire signal is not transferred through the hybrid. Instead, a portion of the electrical signal bounces back off the hybrid and travels back toward the talker. If the delay between the talker and the hybrid is large enough, then it will be perceived as echo. Previously, with traditional phone equipment, there is little delay and so the reflection is not noticeable as echo - it simply mixes with side tone. Figure 1 shows a call going from Phone A to Phone B. There are four potential echo generation points for this call. Caller A will probably experience side tone from Point 1, but not perceive echo. Point 2 shows the 2-wire to 4-wire hybrid (impedance mismatch), resulting in echo being sent back to Phone A. Depending on the round trip delay, this echo may not be noticeable, or will be perceived as side tone. Points 3 and 4 are the likely sources of echo, given the longer (greater than 30 ms) round trip delays.

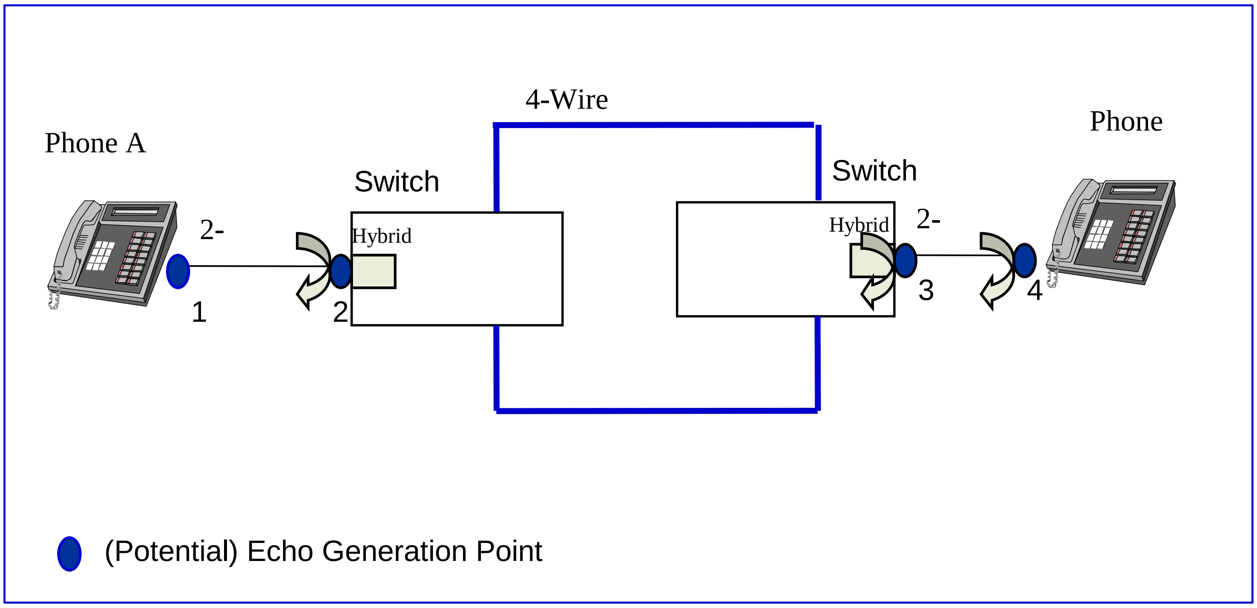

Figure 1. Example of Electrical-Hybrid Echo

In order to get the best transfer of signal through a hybrid, the hybrid must match the impedance of the 2-wire segment. However, more often than not, the impedance is not matched well. There are several reasons for this. First, there are multiple reflection points along the transmission paths. In addition, the loop length and wire gauge rarely matches the few impedance choices offered by the existing telephony gear. Furthermore the impedance choice is not automatically programmed - it is something that is manually administered.

If the impedance mismatch is bad enough, it may cause poor signal transfer and thus may result in complaints of low volume levels (and hot side tone for non-VoIP calls). So, with traditional equipment one may get complaints of low volume levels, whereas with VoIP-based equipment, the complaints may consist of both low volume level *and* echo.

To solve the problem of electrical echo, telephony equipment must employ an echo canceller that is commonly referred to as either line, electrical, or network echo canceller. Usually it is simply referred to as an "echo canceller" and unless specified as being "acoustic" means it cannot handle acoustic echoes, but instead is designed to handle the electrical echoes caused by the analog phone network.

3. Echo Control Capabilities within Avaya

Echo control can be performed using:

- Analog echo cancellation - the isolation and filtering of unwanted echo signals from the main transmitted signal using electrical impedance-balancing circuits,
- Digital adaptive filters, or adaptive echo cancellers, and
- Analog or digital echo suppressors or attenuators which reduce or eliminate echo by injecting signal loss in the return path back to the talker's ear.

Avaya's products use a combination of digital echo cancellation and suppression. The only exception is the handset/headset mode of our IP phones, which currently use only echo suppression (following upgrades to IP phone software in the spring of 2005, these too, will use echo cancellers).

Long-distance providers and Inter-exchange Carriers (IXCs) typically have external echo cancellers included in their facility setup. Local Exchange Carriers (LECs) on the other hand, are not required to provide cancellation of echo from their facilities and endpoints when interfacing with a PBX (Private Branch Exchange).

LEC connections often consist of analog trunks that are Loop Start, Ground Start or Direct Inward Dialing (DID). Impedance under these conditions is poorly controlled, and may cause echo returns back to the Avaya call processing platform. Digital facilities (T1/E1) to the LEC are the preferred mode of trunk connectivity. Digital trunk connections eliminate the echo caused by analog impedance mismatches at the PBX interface. However, digital trunking to the LEC cannot eliminate echoes generated within the LEC or deeper within the external PSTN. Avaya has created a number of products to mitigate echo. These include circuit packs and media modules such as the:

- TN464GP/TN2464BP DS1 Circuit pack (DEFINITY® R9.5 or later with echo cancellation enabled), and the Media Module MM710 (FW V9 or later)
- TN464HP/TN2464CP DS1 Circuit pack (Communication Manager 3.0)
- Media Module MM711 and MM714
- TN2302AP IP Media Processor and the MM760
- TN2602AP IP Media Resource 320 (Communication Manager 3.0)

3.1 TN464/TN2464, MM710

The TN464GP/TN2464BP, TN464HP/TN2464CP DS1 circuit packs, and the MM710 media module provide T1/E1 ports for voice or data transmission. Voice time slots are transmitted through the system backplane over a TDM bus. Echo cancellers provided on these devices that can cancel voice echoes in the near end or far end transmission path. The circuit pack or media module is capable of canceling echoes with up to 96ms of tail-end delay for all DS0 channels in the T1 or E1 span. This means that for calls in which the reverberant duration of the physical echo path is within 96ms, the echo canceler will cancel or control the echo to provide a clear quality call. If the physical echo exceeds the 96ms tail limit, echo may be experienced since its delay, or dispersion, exceeds the 96msec "tail" range. Note that the round-trip delay of the voice path back to the talker's ear is not part of the physical echo path being canceled; the round-trip delay can only increase (large round-trip delay) or decrease (small round-trip delay) to the extent to that any remaining echo is noticeable to the talker.

The DS1 Circuit Pack administration form allows echo cancellation to be specified as:

- "inward," canceling echo coming into the port network through the interface across the T1/E1 circuit. This is commonly used when connected to a local exchange carrier (LEC), or

- "outward," canceling echo that originates within the port network and is coming to the T1/E1 circuit pack from the switch and going out toward the T1/E1 circuit. Outward can be used when connected to a private network (tie trunk).

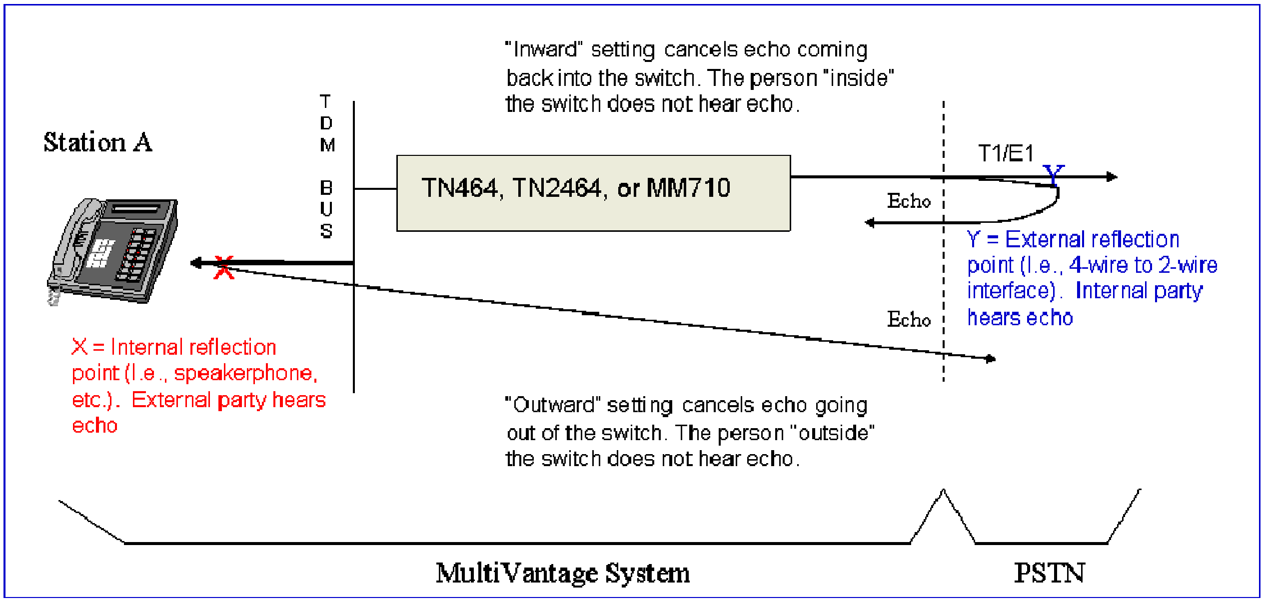

Figure 2. Example of Inward/Outward Echo Direction(s)

3.1.1 TN464GP/TN2464BP DS1 Circuit Pack and MM710 Echo Cancellation Configurations

The TN464GP/TN2464BP DS1 circuit packs and MM710 media modules offer a number of echo cancellation configurations. There is a potential for up to 15 configurations. Prior to V15 firmware for the TN464GP/TN2464BP and V9 firmware for the MM710, there were four configurations from which to choose. Starting with the V15 and V9 firmware vintages there are nine configurations. These configurations are combinations of many tunable low-level echo cancellation parameters designed to deal with returned echo signals with highly variable properties. Functions such as adaptation and digital adaptive filtering involve learning the characteristics of the voice signal and echo to determine how to perform the cancellation. Both initial and continual adaptation methods exist, known respectively as fast and slow convergence. The aggressiveness described in the available configurations is based on an echo suppression threshold – that is, the amount of time it takes to turn off reverse transmissions on a telephone line to reduce the effects of echo. The first four configurations contain more aggressive settings for echo control. The training period for these configurations is very small. If a configuration is described as "highly aggressive," it implies the echo controller will make strong use of echo suppression techniques in the return path of the voice call, which may result in clipped speech. For those who cannot tolerate clipping, Configurations 8 or 9 are recommended.

One very positive aspect of DS1 Echo Canceller is the ability to change an echo cancellation setting while a call(s) is taking place. That is, one can leave a call up, change the setting and immediately hear the effect.

The nine echo control (EC) configurations are as follows[4]:

1. **Echo Control Configuration 1 – Highly Aggressive Echo Control**
   This configuration can control very strong echo from a distant party. It (as well as Echo Control Configuration 4) provides the most rapid convergence in detecting and correcting echo at the beginning of a call. The initial echo fades faster than the other settings (generally in a small fraction of a second) regardless of the loudness of the talker's voice. EC Configurations 1 and 4 are the same except for loss. EC Configuration 1 has 6dB of loss and EC 4 has 0dB of loss. This makes EC Configuration 1 a good choice for consistently high network signal levels. EC Configuration 1 can cause low-volume complaints and/or complaints of clipped speech utterances, particularly during doubletalk when both parties speak simultaneously. Because EC Configuration 1 strongly relies on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints. (Prior to Communication Manager 2.0, EC Configuration 1 was the default configuration.)

   The 6dB of loss in EC Configuration 1 is in one direction only and depends on the setting of the "EC Direction" field on the DS1 Board form. If the direction is set to inward, then the 6dB of loss is inserted in the path out from the board towards the T1/E1 circuit. Conversely, if the setting is outward, then the 6dB of loss is inserted into the path from the T1/E1 circuit towards the TDM bus.

2. **Echo Control Configuration 2 – Aggressive, Stable Echo Control**
   This configuration is nearly identical to EC Configuration 1 except that it does not inject an additional 6dB of signal loss, *and* convergence of the echo canceller is slower, but more stable than that provided by EC Configuration 1. If EC Configuration 1 is found to diverge during doubletalk conditions – noticeable by the sudden onset of audible echo - EC Configuration 2 should be used in place of EC Configuration 1. Because the echo canceller will converge somewhat slower, some initial echo may be noticeable at the start of a call, while the system is "training". EC Configuration 2 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 2 strongly relies on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints.

3. **Echo Control Configuration 3 – Aggressive, Very Stable Echo Control**
   This configuration is nearly identical to EC Configuration 2, but is even more stable. Because the echo canceller will converge somewhat slower, some initial echo may be noticeable at the start of a call. EC Configuration 3 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 3 strongly relies on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints.

4. **Echo Control Configuration 4 – Highly Aggressive Echo Control (Default)**
   Echo Control Configuration 4 is identical to EC Configuration 1 but does not provide the 6dB loss option as described above for EC Configuration 1. All other comments from EC Configuration 1 apply to EC Configuration 4. EC Configuration 4 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 4 strongly relies on echo suppression to help control echo, "pumping" of the distant party's background noise may occur, and lead to complaints. Starting with communication Manager 2.0, EC Configuration 4 is the default configuration.

5. **Echo Control Configuration 5 – Very Moderate, Very Stable Echo Control**
   Echo Control Configuration 5 departs significantly from EC Configurations 1 –4. The echo canceller is slower to converge and is very stable once it converges. Some initial echo may be

heard at the beginning of a call. EC Configuration 5 will not, in general, lead to complaints of clipped speech or pumping of the distant party's background noise.

6. **Echo Control Configuration 6 – Highly Aggressive Echo Control**
   Echo Control Configuration 6 is identical to EC Configuration 4, but reliance on the echo suppressor to control echo is about one-half that of EC Configuration 4. As a result, EC Configuration 6 will not clip speech as much as EC Configuration 4, but may cause somewhat more audible echo, particularly at the start of a call. Some pumping of the distant party's background noise may be perceptible.

7. **Echo Control Configuration 7 – Extremely Moderate & Stable Echo Control**
   Echo Control Configuration 7 provides very stable and transparent control of weak to low-level echoes. For connections having audible echo at the start of a call, the residual echo may linger for several seconds as the echo canceller converges. This configuration has the longest training period of all the configurations offered.

8. **Echo Control Configuration 8 –Aggressive, Very Transparent Echo Control 1**
   Echo Control Configuration 8 provides aggressive control of echo at the start of a call and more moderate control during the call. Unlike all prior settings, EC Configuration 8 uses "comfort noise" injection to match the actual noise level of the distant party's speech signal. The effect is one of echo canceller "transparency," in which there is very little (noticeable) clipped speech or noise pumping. In general there is greater echo control with full duplex. This configuration is very similar to EC configuration 9.

9. **Echo Control Configuration 9 – Aggressive, Transparent Echo Control 2**
   Echo Control Configuration 9 is nearly identical to EC Configuration 8, but provides somewhat more residual echo control at a slight expense of transparency. To many people, EC Configuration 8 and EC Configuration 9 will be indistinguishable.

3.1.2 TN464HP/TN2464CP DS1 Circuit Pack Echo Cancellation Configurations

The TN464HP/TN2464CP are newer DS1 circuit packs that differ slightly from the TN464GP/TN2464BP. One of the main differences is that the 96ms echo canceller chip that is on the TN464GP/TN2464BP has been replaced with a 128ms Echo Canceller chip. The TN464HP/TN2464CP is, therefore, potentially better at controlling echo paths of longer reverberant duration. Only three echo plans are offered with the TN464HP/TN2464CP.

Echo Cancellation Configuration 1 - TN464HP/TN2464CP

Configuration 1 is the recommended/default plan. Both comfort noise generation and residual echo suppression are turned on. During "single talk," background noise and residual echo from the far end may be suppressed and replaced with comfort noise. The comfort noise substitution reduces the perception of background noise/clipping, as observed by the talker. In this plan, the EC Direction is administered so as to cancel the talker's echo. For example, if the talker is inside a building, the EC Direction would be set to inward. Similarly, if a talker is outside a building, the EC direction should be set to outward. Since this plan turns on comfort noise and echo suppression, it is similar to EC plans 8 and 9 for the TN464GP/TN2464BP (firmware vintage 15 or greater) circuit packs.

Echo Cancellation Configuration 2 – TN464HP/TN2464CP

This plan has comfort noise generation turned off, and residual echo suppression turned on. This plan may work well in a quiet background environment. The talker may hear clipping in a noisy background environment. In this case, the EC Direction is administered to cancel the talker's echo. This plan may be a good compromise for a small percent of users who do not care for the comfort noise and prefer the silence during the residual echo suppression periods. Since the plan turns off comfort noise and turns on residual echo suppression, it is similar to EC configuration 1-6 for the TN464GP/TN2464BP (firmware vintage 15 or greater) circuit packs.

Echo Cancellation Configuration 3 – TN464HP/TN2464CP

This plan has both comfort noise generation and residual echo suppression turned off. This plan can be a good choice only if EC plans 1 and 2 do not satisfy the user's preferences. Situations that require configuration 3 should be very rare. For example, if the user does not care for the sound of comfort noise or the pumping/clipping of background noise. This configuration allows the user to hear sound from the earpiece as natural as possible. However, the user may hear residual echo during training periods, or all of the time if echo is sufficiently high and residual echo is always present. Convergence may be very slow. Since comfort noise and residual suppression are turned off, this configuration is similar to EC configuration 7 for the TN464GP/TN2464BP (firmware vintage 15 or greater) circuit packs. A gray area exists when background noise is higher than the threshold (for example, the remote signal comes in at a high level). Under this circumstance, this configuration is similar to EC configuration 5 or 6.

It is important to note that an EC configuration setting can be changed "on the fly." The change takes effect immediately. That is, it is not necessary to busyout/release the circuit pack – you simply change the setting on the DS1 Circuit Pack form. This can be done without disruption to existing calls - in fact, you will immediately hear the effect of the change.

Also important to note is that when there are both TN2302AP circuit pack(s) and TN464GP/TN2464BP or TN464HP/TN2464CP circuit pack(s) being used for a call, the echo canceller on the TN2302 is automatically turned off and the echo canceller on the DS1 circuit pack is used instead because it can accommodate echoes of much longer delay (96 ms).

3.1.3 Media Module 711, Media Module 714, G350 and G250

The MM711 (HW V20 and later) and MM714 are analog media modules that, like the MM710, are used in Avaya media gateways (G700). Echo tails of up to 8ms can be cancelled using one of these media modules. In addition, the G350 and G250 media gateways also have 8ms tails on the echo cancellers on their analog ports. These echo cancellers target the 2-to-4 wire hybrid echo reflections occurring locally at the board's connections to central office trunk lines.

3.2 TN2302AP IP Media Processor/MM760

The TN2302AP circuit packs, in addition to handling the exchange of bearer information between IP and TDM, provide echo cancellation capabilities and loss on connections that traverse the board. The TN2302AP circuit pack has an echo canceller tail of up to 32msec (V43 firmware or later) used to cancel echo coming from the TDM bus. The TN2302AP is a multi-purpose, acoustic and electrical echo controlling device that controls echo originating from local or PSTN analog trunks, PSTN digital trunks as well as acoustic echo originating from analog and digital telephones operating in handset, headset, and speakerphone modes. The switch will inform this circuit pack of the endpoint type used for a call and based on that, the TN2302AP will allocate the appropriate echo tail resources needed to cancel echo. For example, when the IP source is being routed over analog trunks, the full 32msec echo tail is used. If the IP call is routed to a DCP endpoint, only 16 of the 32msec echo tail will be used. If the DS1 with Echo Cancellation circuit pack (TN464GP/TN2464BP or TN464HP/TN2464CP) is being used, the system will bypass the echo canceller on the TN2302AP and use the 96 ms echo canceller on the TN464GP instead. This last example utilizes the longer echo tail length on the TN464GP/TN2464BP and restricts the two echo cancellers from being operational at the same time. The TN2302AP with V93 firmware (or later) allows technicians to tune some of the echo-controlling parameters directly on the circuit pack itself.

Table 1 lists the echo cancellation tail usage for the various endpoints, based on the 19 loss groups that Avaya Communication Manager software currently accommodates (see Section 3.6 and Appendix II for levels/loss plan details)

Table 1. TN2302AP Echo Cancellation Tail Usage

Phone/Trunk Port	Loss Group	Endpoint Definition	TN2302AP Echo Tail Usage (msec)
ONS	1	Analog line with Off Premises = n	16
Dgst	2	DCP line	16
ICS	3	BRI line	16
OPS	4	Analog line with Off Premises = y and R Balance = n	32
OPS_hi	5	Analog line with Off Premises = y and R Balance = y	32
CO 1	6	Analog CO/DID/DIOD trunk with Terminal Balanced = n and RA Trunk Loss = 0	32
CO 2	7	Analog CO/DID/DIOD trunk with Terminal Balanced = n and RA Trunk Loss = 2	32
CO 3	8	Analog CO/DID/DIOD trunk with Terminal Balanced = y	32
4W	9	Paging and 4-wire Tie including TN437 with Connect to Toll = n and STT Loss = norm	16
ATO	10	4-wire Tie with Connect to Toll = y	16
DCO	11	Digital CO/DID **If DS1 with Echo Canc. (TN464GP/TN2464BP or TN464HP/TN2464CP)**	32 **0**
DTO	12	Digital Tie/PRI with Connect to Toll = y	32
DTT	13	Digital Tie/PRI with Connect to Toll = n, STT Loss = norm, and DTT to DCO = norm	32
DTT wats	14	Digital Tie/PRI with Connect to Toll = n, STT Loss = norm, and DTT to DCO = low	32
S-TT	15	Digital Tie/PRI and 4-wire Tie except TN437 with Connect to Toll = n and STT Loss = low	16
2W TT 1	16	TN439 with Connect to Toll = n	16
2W TT 2	17	TN439 with Connect to Toll = y	16
IP_trunk	18	IP phone port type	0
IP_phone	19	IP Trunk port type	0

The MM760 is used in Avaya media gateways for VoIP. The functionality of the MM760 is the same as that of the TN2302. It is used when more VoIP resources are needed than provided by the motherboard on the G700 gateway. There is a "built-in MM760" on the motherboard of every G700, often simply called the "VoIP Engine." Similarly, there is "built-in" half-capacity MM760 on the G350 motherboard. The G250 also provides a reduced-capacity built-in MM760. The MM760 has a programmable echo tail of up to 32ms. The firmware is essentially identical between the TN2302AP and the MM760 modules. The release/version numbers differ:

- V2x VoIP Engines correspond to V7x TN2302AP circuit packs
- V4x VoIP Engines correspond to V9x TN2302AP circuit packs
- V5x VoIP Engines correspond to V10x TN2302AP circuit packs.

3.2.1 TN2602AP IP Media Resource 320

Communication Manager 3.0 introduces a new IP media resource circuit pack, the TN2602AP IP Media Resource 320. This circuit pack has the ability to handle up to 320 bi-directional channels. The echo canceller on the TN2602AP VoIP engine uses a window-based technique rather than the full tail echo cancellation functionality that exists on the TN2302AP. The echo canceller incorporates a bulk delay estimator that determines the best position for the 24 ms window in the specified tail length (up to 128ms). This window will be placed at that position, known as the bulk delay.

3.3 Traditional Phones/Speakerphones

Analog and DCP phones were designed to operate in networks with relatively short delays, minimizing any noticeable echo. When one DCP phone is talking to another DCP phone (on the same system), there is no analog trunk or hybrid (2-wire to 4-wire conversion), and therefore no impedance mismatch occurs that would otherwise induce echo. As shown in Figure 3, some audio quality settings can be adjusted for DCP (and IP) terminals on the *terminal-parameters 6400/607A1/4600/2420* administration form. This includes the amount of control that the user has over local volume adjustments as well transmit, receive and side tone levels, providing additional control over the bearer path.

```
display terminal-parameters                                     Page   2 of   3
                  6400/607A1/4600/2400-TYPE TERMINAL PARAMETERS

           Base Parameter Set: 1        Customize Parameters? y
        Note: Location-parameters forms assign terminal parameter sets.
        Note: LEVELS do not apply to the 4600 terminals.
 OPTIONS
                 Display Mode: 1           Handset Expander Enabled? y
         Volume for DCP Types: retain handset and speaker between calls
          Volume for IP Types: retain handset and speaker between calls
 PRIMARY LEVELS
          Voice Transmit (dB): +2.5             Voice Sidetone (dB): -11.0
           Voice Receive (dB): -2.0        Touch Tone Sidetone (dB): -25.0
     Touch Tone Transmit (dB): +1.0



 BUILT-IN SPEAKER LEVELS
          Voice Transmit (dB): 0.0               Voice Receive (dB): 0.0
                                           Touch Tone Sidetone (dB): -12.0
 6402/2402 BUILT-IN SPEAKER LEVELS
           Voice Receive (dB): -2.0        Touch Tone Sidetone (dB): -19.0
```

Figure 3. Display Terminal-Parameters Form

When used alongside an IP network with longer delays, some DCP phones (24xx and 64xx series) have been known to cause some echo to a remote speaker/listener, especially when operating in speakerphone mode. Parameter settings can be downloaded onto the 64xx telephones to help mitigate echo. The HIC-1

Headset Adapter Cord is recommended for DCP telephones. Additional headset information is covered in Section 4.6 Handset/Headset Echo later in this document.

3.4 IP Telephones/Speakerphones/Softphone

The Avaya IP telephones (46xx) provide acoustic echo cancellation when in speakerphone mode. Typically, these values are in the range of 50ms – 100ms. Echo suppression is provided for handsets and headsets (again, echo cancellation and suppression is available with IP phone software release R2.2 and higher). When using a headset with an IP telephone, Avaya recommends using the Plantronics HIP-1 Headset Adapter Cord. Avaya's Voice Monitoring Manager (VMM) displays Quality of Service (QoS) measurements for every call. Some environmental factors to keep in mind that may cause acoustic echo on the IP telephones include:

- The location of a user's phone with respect to the PC, fan, desk shelving, etc.
- Whether the IP phone is tilted versus flat (this can cause 3dB of loss bouncing from the desk on the 4620 and 4610 IP phones).
- Changes to the acoustic environment near the phone. Avoid moving objects close to the phone during a speakerphone call.

The Avaya IP Softphone provides customers with a PC-based IP-connected end-point with capabilities that are similar to traditional DCP terminals. The Avaya IP Softphone is supported only on the Windows 2000 and XP environments. The softphone does not support automatic acoustic echo cancellation. However, users do have the ability to create a user profile using the Audio Tuning Wizard feature. A user profile is dependent on factors such as the user's voice, background noise surrounding the terminal, the quality of the microphone and speaker devices, placement of the headset microphone (proper placement is under the chin), etc. Once a profile is created, another form exists, called the Audio Options form. This form allows users to modify both the Receive Gain and the Transmit Gain (volume) of a call. These gain volumes have default settings that may be adjusted during a call.

There are recommended sound cards and headset/handset devices for us with the Avaya IP telephones. Users must realize, however, that other factors will affect voice quality, as mentioned above. Approved sound cards, USB headsets and handsets (in no particular order) include:

- Creative Labs – Live
- Creative Labs – Audigy
- Creative Labs – Audigy 2
- Avaya AVD100 Mono PC Headset
- Avaya AVD300 Stereo PC Headset
- Avaya AVD400 Folding Stereo PC Headset
- Avaya AVD500 Multimedia Stereo PC Headset
- Clarisys i750 USB Internet Phone

3.5 ATM Environments

An ATM switch can replace a Center Stage Switch such that voice calls travel through an ATM switch to get between port networks and outside of the Communication Manager campus network. Converting the TDM based PCM voice data into ATM data involves packetization of the voice into cells (53 bytes each). This adds an additional 9msec of delay in the transport of the voice (or 18 msec round-trip) over the traditional CSS Avaya systems[5]. However, it is less delay than what is experienced over IP.

The ATM Expansion Interface (ATM-EI) circuit packs (the TN2305B and the TN2306B), include an echo canceller tail length of 8 msec. Given the ATM environment and characteristics, this has proven sufficient (in pure TDM environments) to cancel echoes from locally connected terminals and most echoes from external networks. Specifically, given that voice packets are sent in cells over a Layer 2 switch, there is very little latency. Another advantage of ATM is that the cells are shorter than IP packets, which also reduces the amount of delay over the network.

The 8 msec echo canceller is always on. When interacting with IP there has been no noticeable echoes reported. Given the canceller capabilities on the TN2305B/TN2306B circuit packs, as well as other echo controllers positioned in a customer configuration, ATM has had continuing positive results.

3.6 Levels/Loss Plan[7]

The perceived loudness during a call depends on factors such as the:

- speech level of the talker.
- hearing capability of the listener.
- efficiency of the transmit and receive electro-acoustic mechanisms of the endpoints.
- loss experienced within the network subsystems (for example, loops, trunks and switching systems).

Avaya allows users to administer the amount of loss or gain that is applied on calls (see Appendix 2 for additional details). The terms loss and gain are used to characterize acoustical and electrical signal level difference between points expressed in dB. Gain refers to the increase in signal level resulting in an increase of the acoustical signal towards a remote listener. Loss is a reduction in signal level resulting in a decrease of the acoustical signal towards a remote listener.

The loss plan administration provided by Avaya Communication Manager software is primarily intended to control signal losses in the telephones and gateways and *NOT* to control echo. However, in cases of severe echo, the administered loss can be changed to a different selection. In general, it is not advised to use loss plan administration in this way. For one thing, you would run out of capacity if you tried to apply the loss plan on a station-by-station basis. It is recommended that if changes are needed to the loss plan, you first consult with Avaya Services personnel. It is better to reduce the echo by strategic deployment of echo cancellers at the source of the echo.

4. General Echo Troubleshooting Guidelines

A major theme when it comes to mitigating echo is to solve the problem closest to the source of the echo (for example, 2-wire to 4-wire hybrid in analog trunks). One of the first questions is how to begin figuring out the source of echo. Remember that echo is when you hear yourself speak. While other voice quality issues such as popping and clipping may occur, that is not echo. Appendix 3 shows a sample "report" for troubleshooting echo. Basically, you want to:

1. Determine the source of echo.
2. Eliminate echo at the source (or as close to the source as possible).
3. Use echo cancellers, echo suppression, or loss (last resort!) to mitigate the echo.

Table 2 lists some common echo sources, and depending on the severity of the echo, references the section within this document that discusses how to mitigate that echo[3].

Table 2: Mitigating Echo

Echo Source	Persistant Echo	Come and Go	Beginning Only	Some specific point in time
Analog trunk	This is either too loud an echo or too far away for the echo canceller to handle. See Section 4.1.	Something about the echo source or the echo path is causing difficulty in the echo canceller's ability to consistently control it. See Section 4.2.	The echo canceller requires some time to learn about the echo source. During this training period some echo may be heard. If it is particularly troublesome, see Section 4.2.	Some event (e.g., another party added to a conference) is probably causing a change in the echo source(s) which requires the retraining of the echo canceller. During this time some temporary echo may be heard. If it is particularly troublesome, see Section 4.2.
DS1 trunk	See Section 4.3.	See Section 4.3.	See Section 4.3.	See Section 4.3.
Speakerphone	See Section 4.4.	See Section 4.4.	See Section 4.4.	See Section 4.4.
Handset/Headset	See Section 4.6.	See Section 4.6.	See Section 4.6.	See Section 4.6.

4.1 Mitigating Analog Trunk Echo

If an echo coming from an analog trunk is persistent, this usually means the echo is beyond the specifications of the echo canceller. It can be too loud or too far away. To reduce an echo that is too loud, try the following:

- It is possible that the impedance choice of the analog trunk has been chosen incorrectly. Determine the best impedance choice: 600 ohm vs. RC. Start with RC (the default) and if that does not reduce the echo, then try 600. Normally RC is used for longer loop lengths (greater than 3000 feet) and 600 ohm for shorter loop lengths. Trial and error may be necessary for those circuits that lie somewhere between. Impedance is administered in the "Trunk Termination:" field on the *trunk-group* form. If possible, create a new private trunk group so as not to subject the user to the experimentation. Try both impedance settings to see which provides the least echo. <u>Several</u> calls to the <u>SAME</u> destination should be tried at each setting to get an overall reading. That is, do not base your judgment on one phone call alone). Note that some of the newer VoIP gateway firmware (or media processing circuit packs) stores information about trunks, and by changing the impedance choice you may experience more temporary echo for the first call to a given trunk. For the second call, and calls thereafter there should be less.

```
display trunk-group 1                                          Page   1 of  20
                                TRUNK GROUP

Group Number: 1                    Group Type: did          CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: 0101


                          Country: 1
                                    Auth Code? n


TRUNK PARAMETERS
            Trunk Type: immed-start      Incoming Rotary Timeout(sec): 5
                                                   Incoming Dial Type: tone
     Trunk Termination: rc                   Disconnect Timing(msec): 500
       Digit Treatment:                                        Digits:
       Expected Digits:                               Sig Bit Inversion: none
     Analog Loss Group: 6                            Digital Loss Group: 11
   Extended Loop Range? n         Trunk Gain: high     Drop Treatment: silence

 Disconnect Supervision - In? y
```

Figure 4. Trunk Group Form Showing Trunk Termination Field

- If problems still persist, insert loss in 1dB increments (in both directions) until the problem is resolved. Typically the loss group for analog trunks is 6. The loss group for inter-gateway or IP trunk connections is 18. IP Phones use loss group 19. Thus loss should be inserted between loss groups 6 and 18, and between loss groups 6 and 19. See Table 1 for a list of loss groups and ***Appendix 2 – Loss Groups for VoIP*** for more explanation on this subject.
- A noisy analog trunk may cause sub-par performance of an echo canceller. If the analog trunk is noisy, has crosstalk or other impairments, take whatever steps possible to reduce or eliminate the noise or impairment on the trunk.
- If the problem still persists despite all efforts, the echo may be outside the window that is supported by the echo canceller (that is, it is too far away). See Section 4.5.

4.2 Decreasing Audible Echo Canceller Training Period

If the echo canceller is having difficulty training with a given echo source or change in echo source, there are some things that can attempted in an effort to minimize the temporary or occasional echo:

- First try all steps in Section 4.1 in an attempt to decrease the volume level of the echo. This alone may result in a significant enough improvement that no further steps are necessary.
- Turn off call classification if the feature is not being used (call classification can affect trunk gain and result in an extended echo training period at the early portion of the call)
- If the system is a G700 or G350, and analog trunks are the echo source, if possible use MM711 HW V21 FW V61 or greater or use an MM714. Both of these media modules have on-board echo cancellers that may alleviate the problem.

- Tail Saver Feature: The TN2302AP is capable of using echo canceller settings from the previous call (or a combination of previous call + some subset of calls) to determine an Analog Trunk ID. When Communication Manager software notices that specific analog trunk being used, the trunk ID is sent to the TN2302AP. The TN2302AP will then use the appropriate/stored echo tail length for that call, thereby eliminating the need to train for that call.

4.3 DS1 Trunk Echo

If the echo source has been determined to come from a DS1 trunk:

- Verify that the DS1 echo canceller has been turned ON on the DS1 Cricuit Pack Form

```
display ds1 03D20
                                DS1 CIRCUIT PACK

            Location: 03D20                           Name: uds1-tg23
            Bit Rate: 2.048                    Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                        Interface: user
   TN-C7 Long Timers? n                   Country Protocol: 1
Interworking Message: PROGress            Protocol Version: a
Interface Companding: mulaw                           CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4


      Slip Detection? n                 Near-end CSU Type: other
    E1 Sync-Splitter? n
   Echo Cancellation? y
        EC Direction: inward
    EC Configuration: 4

Command:
```

Figure 5. Page 1 of the DS1 Circuit Pack Form

- If the DS1 trunk card does not have an echo canceller, try upgrading to the TN464GP/TN2464BP circuit pack which has a 96 ms echo canceller tail length, or the TN464HP/TN2464CP which has a 128ms echo canceller tail length.

If an echo canceling DS1 trunk card is not an option:

- Insert loss in 1 dB increments (in both directions) until the problem is resolved. The loss should be inserted between the loss group associated with the DS1 trunk and loss group 18 (IP trunks or inter-gateway), and also between the loss group associated with the DS1 trunk and loss group 19 (IP phones). See Table 1 for the list of lost groups and ***Appendix 2 – Loss Groups for VoIP*** for more explanation loss groups.
- If problem persists, the echo may be too far away for the VoIP/Media Processor to handle, see Section 4.5.

4.4 Speakerphones

When talking over a speakerphone, the actual user of the problematic speakerphone will not be the one reporting the echo. The echo will be reported by a user on the other end of the call. That said, the user of the problematic speakerphone may not be aware of it, unless told by the callee/caller on the other end.

If the source of the echo is determined to be caused by a speakerphone:

- If the phone causing the echo is an IP phone, upgrade to the latest IP phone firmware. The IP phone itself is responsible for controlling its own echo.

- If the speakerphone causing the echo is a DCP or analog phone, upgrade the firmware on the TN2302AP to V92 or greater (V42 for G700/G350 VoIP engines). V92 contains enhancements for improved control of speakerphone echo.
- If the problem persists, you may want to modify your terminal parameters to return to the default volume settings. From there (if you still hear echo), insert loss in both directions until the problem is remedied. Note that for DCP speakerphones (loss group 2), the loss would be inserted between loss groups 2 and 18 (IP trunks or inter-gateway), and between loss groups 2 and 19 (IP phones).
- Also refer to the speakerphone guidelines for best acoustic arrangement.
- For some speakerphone models (for example, the 6408), improvements may occur by turning the volume control of the speakerphone down to nominal levels.

4.5 Echo Too far Away

If the echo source is so far away that it exceeds the window size of the echo canceller this would cause persistent echo on the call. If possible, have the customer write down the phone number used for that call to both get an idea of the destination of the call and attempt to re-create the problem. If the echo is indeed beyond the window of support for the echo canceller, then the echo should occur on every call to that same phone number and be persistent throughout the call for all calls to that number. This will usually be a long distance call, and this is more likely to occur with a less-reputable long distance provider, since the more reputable long distance carriers and service providers usually provide echo control at the edges of their network

- If a DS1 trunk is involved, try upgrading to the TN464GP/TN2464BP circuit pack, which has a 96ms echo canceller tail length, and make sure the DS1 echo canceling option is on.
- If the echo source is an analog trunk, try to get customer to switch to a DS1 trunk and deploy an echo canceling DS1 card.
- Some relief may be had by adding loss between the relevant loss groups (see Section 4.1 for analog trunk information, or Section 4.3 for information on DS1 trunks). See ***Appendix 2 – Loss Groups for VoIP*** for more explanation on this subject.
- Report the problem to the telephone phone service provider – most reputable service providers will usually provide echo cancellers and thus you would never have this problem, but there’s always the possibility that something is wrong with their echo canceller equipment. Regardless, it is always best to at least inform the carrier of their deficiency.
- If all else fails an external echo canceller may be necessary, or a customer may escalate the issue to the appropriate level.

4.6 Handset/Headset Echo

It is important to remember that the actual user of the problematic handset or headset will not be the one reporting the echo. The echo will be reported by someone on the other end of the call. Therefore, the user of the problematic handset or headset may not be aware of it, unless told by the callee/caller on the other end.

Handset echo is usually a low-level acoustic echo, caused by the direct coupling of handset speaker to handset microphone. In a VoIP-based call, this echo may be noticeable although it will usually be low in level, and it will be most noticeable when the phone listen volume level is set rather high (on the phone which is causing the echo). If the phone listen volume level on the phone of the person hearing the echo is too high (that is, too much gain), echo may also be affected.

If the source of echo is determined to be caused by a handset:

- Ask the user of the problematic handset to lower his/her phone listen volume level.
- If the phone (handset) causing the echo is an IP phone, upgrade to the latest IP phone firmware. Earlier versions of firmware did not always control handset echo well. The newer versions of firmware employ some form of handset echo control.
- If the phone handset causing the echo is a DCP or analog phone, upgrade the firmware on the TN2302AP to V92 or greater (V42 for G700/G350 VoIP engines). V92 contains enhancements for improved control of handset echo.

Headset echo can be acoustic or electrical echo. In most cases, it will probably be a low level echo and usually can be controlled. However, there is one known case of a headset (M12) that, when on handset mode, can cause a very extreme, loud echo when the phone listen volume is set too high.

If the source of echo has been determined to be caused by a headset:

- Make sure the headset is an Avaya-branded headset. If the echo source is determined to be an Avaya M12 headset adaptor (in handset mode), install the special patch chord provided by manufacturer.
- Ask the user of the problematic headset to lower his/her phone listen volume level.
- If the headset causing the echo is connected to an IP phone, upgrade to the latest IP phone firmware. Earlier versions of firmware did not always control headset echo well. The newer versions of firmware employ some form of headset echo control.

4.7 Some Known Echo Issues

This section focuses on some known echo issues experienced by some Avaya customers. These are being included here to help troubleshoot similar echo issues that may be encountered.

4.7.1 Test Scenarios to Assess TN2302AP Bearer Path Performance

The type of connection being used on a 2-party call will affect the loss, delay, return loss as well as parameters and initialization of the echo canceller in the TN2302AP circuit pack. The following sub-sections represent some common connections in which echo may occur.[9] When testing each connection type:

1. Originate and answer the call
2. Observe and note the environment. Assure repeatability of the physical configuration and settings
3. Use the status station and list trace commands to verify connectivity. Where IP segments are involved, measure segment delay. Data network administrators may be needed to accurately assess data network performance.
4. Minimally, perform "talk with quiet far end" and "doubletalk" tests (for example, mutual counting in the local language where one party is counting forward and the other party is counting backwards in order to differentiate each voice).

5. Note results.

4.7.2 DCP-to-DCP Terminals within same PN and other TDM Endpoints

These connections will have no TN2302AP circuit pack involvement. They could serve as a baseline for a very low delay circuit, approaching best-case performance with no echo canceller and minimum delay, but they will not be affected by changes in the TN2302AP circuit pack.

4.7.3 IP-terminal to DCP

One of the configurations causing some residual echo towards the IP network is illustrated in Figure 6 (assuming a firmware load older than V78). In this example, a DCP phone (e.g. 24XX series) is in speakerphone mode at "A". Other users are at "B". The important characteristics of this configuration are[2]:

- The DCP phone at "A" operates in speakerphone mode.
- The IP interconnection through the TN2302AP circuit pack and LAN introduces a significant delay, much more than when traditional TDM interconnections are used. With TDM interconnections, the echo is not observable because of its short round-trip delay, which is the situation for which "traditional" e.g. DCP phones, have been designed.
- Users at the other side at "B" hear echo on a DCP, Analog or IP phone when the echo canceler on the TN2302AP allows residual echo propagation into the IP network.
- The source of echo is a too-low 'Echo Return Loss' for the "traditional" phone in speakerphone mode.

Note that the same echo problem may occur when:

- Either or both of the Port Networks (PN) are replaced with a gateway (G350, G700 etc.).
- G700 gateways are stacked together and interconnected through an Octaplane. The Octaplane should be considered an IP network.

The DCP speakerphone mode uses a technique called *echo suppression* to reduce echo. When the user at "A" is listening to a talker at "B", the speakerphone effectively reduces the sensitivity of the speakerphone microphone by switching in loss to "B." This 'echo return loss' reduces the amount of acoustic echo observed by "B." Conversely when the user at "A" is speaking and the user at "B" is listening, the 'echo return loss' is switched out and the speakerphone microphone is restored to full sensitivity.

This 'echo return loss' switching causes echo reduction and works very well when the switching is implemented properly. However, the loss switching causes the conversation to be half-duplex: speaking simultaneously causes fast 'echo return loss' switching, essentially wiping out pieces of the speech signal in both directions. In short delay network configurations such as when two DCP phones are both connected to a single Port Network or Gateway, or when they are connected via an intermediate analog or digital TDM trunk, the echo suppression works very well. However, with long delays in the network a noticeable echo is observed.

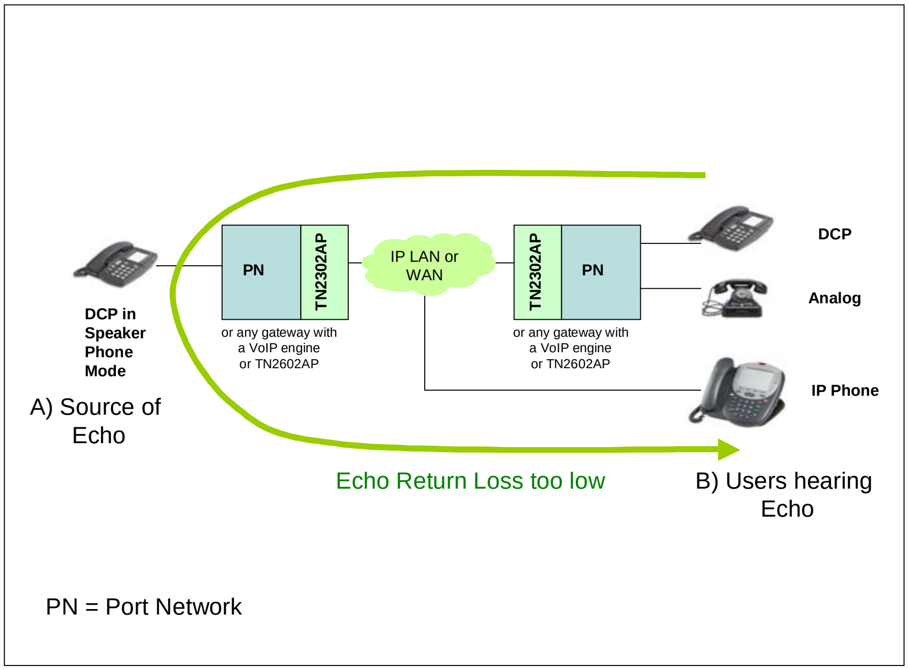

Figure 6. IP Interconnected Systems

The IP Media Processor and VoIP Engines have an echo canceller, which reduces the echo from the DCP phone to the IP network. When no 'echo return loss' switching takes place, the echo canceller works very well. However, when loss is switched in or out of the voice path, the echo canceller needs to converge on a speech signal, which is very different than before the switch action occurred. The convergence takes some time, during which echo is passed through to the IP network, which has resulted in a short observable echo or distorted signal. At this time (TN2302AP firmware version 92), echo has been reduced to an acceptable level by applying some acoustic echo canceling techniques in the TN2302AP / VoIP engines. The TN2302AP/VoIP engines already deployed need to be upgraded with the latest firmware incorporating the echo cancellation improvement.

4.7.4 DCP-to-DCP across different port networks

In this case two TN2302AP circuit packs will be involved. It is important to note that the TN2302AP in the second port network controls the echo heard by the user in the first port network and vice-versa. Upgrade both circuit packs to get a complete test of a new firmware release.

NOTE: If the two port networks are connected via a TN464GP/TN2464BP with echo cancellation turned on, the echo canceller resources in the TN2302AP circuit pack are NOT used. The exception to this is if the port networks are IP connected, in which case the TN2302AP circuit pack's echo cancellation capabilities will be used.

4.7.5 IP terminal to DS1 trunk

Where the DS1 trunk has echo cancellers enabled via the TN464GP/TN2464BP circuit packs, the system will rely on the DS1 board to cancel echo that comes from outside the local environment. When no echo canceller is enabled on the DS1, the TN2302AP circuit pack may cancel echoes on short-delay DS1 calls.

In Figure 7, a DS1 interconnect is used between systems. The large delay is incurred due to the IP connection to the IP phone. As mentioned earlier in this document, the TN464GP/TN2464BP and MM710 DS1 boards have echo cancellers. The DS1 boards connecting the two systems in the figure should have their echo cancellers set to operate "outwards". If the DS1 echo cancellers perform comparably to the TN2302AP echo canceller, the echo will likely be acceptably low. Otherwise the echo canceller firmware needs to be upgraded.

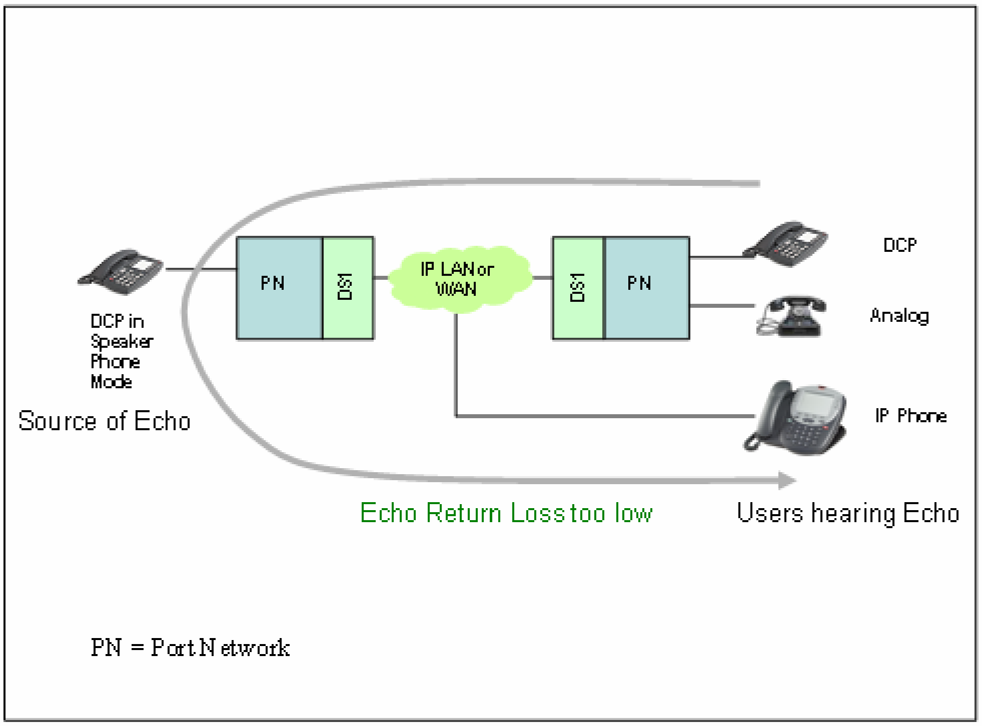

Figure 7. DS1 Interconnected Systems

4.7.6 IP terminal to IP trunk

With shuffling enabled, the IP terminal on the near end will communicate directly to the TN2302AP circuit pack on the far-end switch (if DCP or analog terminals), or directly to the IP station. Use the *status station* or *list trace* commands to verify connectivity is as expected. Results will be similar to other IP-terminal connections. Differences are likely to include additional delays introduced on the IP trunk, which probably runs over a longer distance. The Network Region Design may affect codec selection. Connectivity will need to be verified on both switches.

4.7.7 IP terminal to CO (analog) trunk

Echo from CO Trunks differs from that returned from DCP terminals. The source is typically electrical return from two-to-four-wire conversion circuits occurring at one or more points along the trunk line. This level of echo may be significant, but it tends to vary little over time. CM takes advantage of this stability to reduce the training period for echo cancellers on these calls.   Each call is set up with a port ID for the analog trunk involved. If the TN2302AP has established a call on this trunk before, it uses values retained from the last connection as a starting point.   If not, a longer initial training period is required.   At the end of each call, the cache for the particular analog trunk port is updated.   The first call after an upgrade or a reset of the board to each trunk group member may result in a longer audible training period.

5. Other Times You May Hear Echo

While the intent of this paper is to discuss echo control for 2-party calls, there are other times when one may hear echo. Wireless phones incur additional delay over IP phones, for example. As they have become more popular, cell sites continue to "pop up," making the transition from one cell site to another more seamless to the user.

Echo during multi-party conference calls may occur depending on the factors involved. For example, whether one party is on a cell phone, the country that someone is calling from, the acoustics in a given room, etc. So, a person could be on a 2-party call with no perceived echo and then conference in another party. Now echo is evident to one or two parties involved. What has happened is that the signals from two of the three parties have been added together and then forwarded on to the third party. Consequently, the echo signals of the two parties originating from the signal of the third party, also gets added. This may result in a very loud echo return to the third party. To compensate for this, extra loss needs to be introduced for two of the three parties on the call. A way to achieve this without adjusting the loss plans mentioned earlier, is to have the last person who called before the echo became apparent to mute their phone.

6. Summary

Echo control is about balancing and making compromises in one's network. There are tradeoffs that each customer must consider. Given the lower costs of sending voice over an IP network, customers are increasingly merging their voice and data systems over IP networks. However, customers must realize that the amount of time it takes to transmit, encode and decode the voice packets from one endpoint to another over an IP network is longer than transmitting voice over a traditional circuit-switched system. This increased delay may result in customers hearing echo which previously was not perceived. Avaya has many products in place to sufficiently control echo in a customer's environment and continues to research and develop such products on an on-going basis.

7. Glossary

TERM	DEFINITION
Crosstalk	Occurs when you hear one or two other people talking on your phone line that you did not call.
DCP	Digital Communications Protocol
DID	Direct Inward Dialing. The ability to dial inside a company directly without going through an attendant.
DS1	Digital Service Level 1
Echo Cancellation[6]	A technique that allows for the isolation and filtering of unwanted signals caused by echoes from the main transmitted signal. An echo cancellation device puts a signal on the return transmission path that is equal and opposite to the echo signal. Echo cancellation allows full duplex modems to send and receive on the same frequency. Network-based echo cancellation can interfere with modems, which perform their own echo cancellation. To avoid cancellation problems, modems capable of echo cancellation (such as V.32 modems) send a unique answer tone with a phase reversal every half second. The network echo cancellers detect the phase reversal in the answer tone and disable themselves. (Contrast with echo suppression.)
Echo Canceller Delay	The signal delay introduced by the echo canceller device or algorithm. This delay increases the total round-trip echo path delay. Echo canceller delays should ideally be (and usually are) quite small. In particular in IP networks, it is important to have a small delay as the codec/packetization already adds a significant delay.
Echo Canceller Tail Length (ECTL)	The maximum time delay, measured in milliseconds (ms), of echo path delay that the echo canceller is able to handle for eliminating echo. An echo canceller's job is to essentially remember the outgoing voice stream and cancel any reflected voice that matches within a certain amount of time (which is the echo tail length).
Echo Suppression[6]	The process of turning off reverse transmission on a telephone line to reduce the annoying effect of echoes in telephone connections, especially on satellite circuits. An active echo suppressor impedes full-duplex data transmission. More simply stated, adjusting the gain/loss characteristics to reduce the perception of echo (playing with volume).
Ground Start (GS)	A way of signaling on subscriber trunks in which one side of the 2-wire trunk (typically the "Ring" conductor of the Tip and Ring) is momentarily grounded to get dial tone.
Half-Duplex	A circuit designed for data transmission in both directions, but not at the same time. With respect to voice, when two people attempt to talk at the same time, a system is unable to pass both voice directions at the same time, which results in clipping of speech syllables.
Impedance[6]	The total opposition (that is, resistance and reactance) a circuit offers to the flow of alternating current. It is measured in ohms. The lower the ohmic value, the better the quality of the conductor. Low impedance will

	help provide safety and fire protection and a reduction in the severity of common and normal mode electrical noise and transient voltages.
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IXC	Inter-Exchange Carrier. Long distance carriers that include all facilities-based inter-LATA carriers. The term generally applies to voice and data carriers but not to Internet carriers.
LEC	Local Exchange Carrier (aka local telephone companies)
Loop Start (LS)	A way of obtaining dial tone in which a line is seized by bridging though a resistance the tip and ring (both wires) of a telephone line.
Non-Linear Processing (NLP)	Used to reduce residual echo to the required (inaudible) level. So called because the echo signal that couldn't be dealt with through the linear echo canceller is left over as a residual "non-linear" signal. An echo suppressor is a type of non-linear echo processor, because the echo suppression algorithm itself does not necessarily operate on the speech signals in linear relationship with the level of echo.
Ohm	Unit of resistance that allows one ampere of current to pass at the electrical potential of one volt.
PN	Port Network
Side tone	The affect of hearing one's voice in a telephone handset. This is intended as an indication that one's telephone is working.
TDM	Time Division Multiplexing
VoIP	Voice over Internet Protocol

Appendix I – Additional Echo Notes

Miscellaneous Echo Sources and Non-Linear Echo

There can be other sources of echo that do not necessarily fall under the above two categories. An example is electrical echo caused by poorly designed or faulty electronics used in telephony gear. For example, some headsets can cause significant echo. Often a line echo canceller can successfully model the echo caused by such devices, and thus control or minimize the echo. But sometimes the echo is one that cannot be modeled easily (e.g., M12 headset in handset mode), and thus although some improvement may be provided by the line echo canceller, it is best to simply fix or replace the problematic equipment.

There are more mathematical definitions of the word "non-linear" but simply put, a *non-linear echo source* is one that cannot be modeled easy. An example of this is the M12 headset adaptor in handset mode. It produces no echo at low volume levels, but very loud echo at high volume levels. There are other ways for an echo source to be considered non-linear, but in this case it is non-linear because it behaves differently at one volume level compared to another. A linear echo source will produce the same relative echo regardless of signal level. A linear echo source is what is expected by a line echo canceller.

The echo path itself may be non-linear which could cause a linear echo source to appear non-linear to an echo canceller. A good example of this is an IP network. An IP network contains various impairments, e.g., packet loss and jitter, which make the echo path non-linear. Therefore, echo control must occur such that echo is prevented from entering the IP network, since once it has entered the IP network, it will be harder if not impossible to cancel or control due to the non-linearities introduced by the IP network. This is another good reason why IP phones must control/cancel all possible sources of echo (speakerphone, handset, and headset). Some other functions that can cause an echo path to be non-linear, and thus will prevent an echo canceller from controlling echo well are things like speech coding and silence suppression (typical functions used in Voice-over-IP).

Echo Canceller Notes

An echo canceller attenuates echo by generating an internal estimate or replica of the actual echo and subtracting it from the return signal. However, some period of time of active signal (speech) is needed for the echo canceller to learn about the echo source. During this learning period, or training period as it is sometimes called, a brief amount of echo may be heard.

Due to technical and theoretical limitations, (even after training) steady state echo canceller attenuation is limited and thus some extra loss (suppression) is usually switched on temporarily to bring the residual echo to an inaudible level.

An acoustic echo canceller will usually employ some additional switched loss during training time and in general during the call (since it is often retraining due to a changing acoustic environment). And although the switched loss may take away from the full-duplex nature of the call, it is a necessary evil as acoustic echoes are more difficult to cancel, and users are usually more tolerant of some loss of duplex-ness when in speakerphone mode.

Line echo cancellers on the other hand are expected to provide a more full-duplex environment since users on handsets do not expect or tolerate half-duplex conversations, and therefore line echo cancellers are usually designed with less of a tendency to suppress or clip a signal. This is at least part of the reason why the training period of line echo cancellers may be noticeable whereas usually not true for acoustic echo canceling speakerphones (another part of the reason is that acoustic echo cancellers often employ more complex algorithms which can help hide the fact that training is in process).

Appendix 2 – Loss Groups for VoIP calls

Earlier sections of this document talked about signal levels and loss plans administerable within Avaya Communication Manager software. A bit more detail on loss is provided here, though again we caution that loss plans provided within Avaya software are primarily intended to control signal losses in the telephones and gateways and *NOT* to control echo.

It is generally accepted that a connection with an end-to-end loss (called Overall Loudness Rating) of 10dB, which approximates a normal conversation between a talker and listener spaced 1 meter apart, will provide a high degree of satisfaction for the majority of users. Exceptions to this include people with a hearing impairment as well as the loss plan for Italy, which is typically louder than loss plans for other countries.

In an IP telephony network, the end-to-end loss of 10dB is implemented as 8dB in the speaker's telephone, 0 dB in the IP network, and another 2dB loss in the listener's telephone. To account for personal preference or the presence of background sound, listeners may adjust relative to the 10dB loss value by changing the volume control on their telephone. The IP telephony loss values are globally identical and specified in ITU and various local standards.

To accommodate loss/gain values from a global perspective, loss plans exist within Avaya software to support various countries and or locations. These are found on the *system-parameters country-options* forms. Each country-specific loss plan has a set of default values. The default values have been found to offer the highest voice quality for the associated country. In order to ensure that the signal levels are controlled properly within the scope of the voice network consisting of Avaya systems, the appropriate country-dependent loss plan should be administered. Use the *trunk-group* form to change/display the loss group a trunk is assigned to (page 1). In addition to the ability to administer loss for 2-party calls, Avaya Communication Manager also allows country-dependent conference call loss administration. Loss is applied depending on the number of parties involved in the conference.

To accommodate multi-country deployment, the loss plan may be administered per location. For example, let's say that a switch within the United States has six locations. Five of those locations can be administered to use the loss plan created for the United States (country code 1**)**, and the sixth location may be administered to use the loss values of Belgium (country code 8). This flexibility is very attractive in today's global markets.

In an attempt to clarify confusion that may exist with loss groups as it pertains to echo and VoIP-based calls, the following examples have been provided. By "VoIP call" we mean any call that uses a media processor. This includes a TN2302AP, a TN2602AP, or G700/G350/G250 VoIP Engine resource.

Note that the examples shown below may use a G700 VoIP Engine port or a TN2302AP port, but they are interchangeable in terms of the loss plan. Also note that an inter-gateway call traversing a G700 Octaplane is treated no differently than an inter-gateway or IP trunk call traversing an IP network (the Octaplane is really just a high speed IP bus). The same loss group applies for any inter-gateway call or IP trunk call.

IP Phone calls to non-IP endpoint

Let's take the simpler case of IP phone to analog trunk. The connection is made up of:

IP phone ←→ IP network ←→ MedPro port ←→ Analog trunk port
Loss Group 19 ←---------------------→ Loss Group 6

The loss group for IP phones is 19, the loss group for analog trunks is usually 6. Therefore to insert loss for any IP phone calls to analog trunks, you would insert loss in the loss plan between loss groups 19 and 6.

Station-to-Station Inter-gateway or IP trunk calls

A more confusing case is that in which traditional phones are connected by way of an IP trunk or inter-gateway connection. An example is a DCP-to-DCP call in which the DCP stations reside on different gateways (i.e., an inter-gateway call between G700s which can be within the same stack). The connection looks like this:

DCP Phone 1	←→	VoIP Engine Gateway 1	←-→	Octaplane or IP Network	←→	VoIP Engine Gateway 2	←→	DCP Phone 2
Loss Grp 2	←→	Loss Grp 18				Loss Grp 18	←→	Loss Grp 2

For this kind of call, the choice of loss groups is not obvious. The loss group associated with a VoIP port on an inter-gateway call is 18. That for a DCP station is 2. So to affect the loss for the call depicted above you must insert loss between loss groups 2 and 18. Note that the loss will be symmetrically placed on both ends of the call. Also note that loss inserted between loss group 2 and itself will only affect calls between DCP stations residing on the SAME gateway (or port network).

Station-to-Trunk Inter-gateway or IP trunk calls

Another example is a DCP station calling out to an analog trunk which resides on another gateway. This connection would look like this:

DCP Phone 1	←→	VoIP Engine port Gateway 1	←-→	Octaplane or IP Network	←→	VoIP Engine Port Gateway 2	←→	Analog Trunk
Loss Grp 2	←→	Loss Grp 18				Loss Grp 18	←→	Loss Grp 6

In this case, if you were battling analog trunk echo problems, you would insert loss between loss group 18 and loss group 6 (since this is the echo path of interest). Note that this is not a symmetrical case (like the DCP-to-DCP case shown previously). Also note that loss inserted between loss group 2 and loss group 6 will only affect calls between DCP stations and analog trunks which reside on the SAME gateway.

Appendix 3 –Echo Troubleshooting Report

1. Attempt to determine the source of the echo (keeping in mind that the person reporting the echo is not the source).
2. Determine the severity of the echo.
   a. Only at beginning of call (implies that the echo canceller is training based on talker's voice).
   b. Through-out entire call
   c. Intermittent during call
   d. Occurs only when a specific number(s) is dialed
3. What type of equipment is being used?
   a. Endpoint type (for example, IP phone, DCP, Analog, IP Softphone, IP Agent)
   b. Headset? If yes, is it Avaya-approved?
   c. Was a speakerphone being used during the call?
   d. Are the most recent versions of firmware and software being used?
4. What is the call flow? For example, is the call going over an analog trunk? DS1 trunk? Via a gateway?

Based on answers to the above questions, echo will fall into one of two basic categories: acoustic echo or electrical-hybrid echo.

Acoustic echo: Characteristics of the physical environment (hardware itself, excessive delay, the room the phone is located, etc.) that cause reflected voice signals back to the source. Keep in mind the human ear perceives echo if the reflected echo is within 30 ms of the originating stream (on average).

Electrical-hybrid echo: Impedance mismatch as the media stream travels along the voice path (4-wire to 2-wire conversions).

Troubleshoot based on input above. For example:

If the call flow is interfacing a trunk, first try to eliminate hybrid echo. Turn on echo cancellation (in the right direction) on the trunk interface (TN464GP/TN2464BP or equivalent).
http://info.dr.avaya.com/cgi-bin/g3fs/download?jej/echo_setup.pdf

If the call flow is IP on one or both ends, consider acoustic echo. Excessive delay on the network (use VMM as a tool to measure), acoustic environment of the room that a speakerphone is used in (i.e., move the phone around, try moving to a different room, etc.) may be the cause. If the call is ip-tdm or hairpinned, the TN2302APs have a default echo tail of 16 ms - this can be changed to 32 ms maximum via the TN2302AP debug monitor.

In all echo cases, echo suppression can be utilized to reduce the awareness of echo. Essentially, playing with the loss plan is an example (be careful!!!). Also, note that the older Agere chipset used in the IP hard phones have something called automatic gain control, which was removed with the newer Broadcom chipset. But, with IPT 2.0.14 friendly fix or 2.1.1 and later firmware, there is now an enhancement to somewhat mimic AGC. Ensure that "SET AUDIO 2" parameter is in place on the settings file to activate.