Teleconferencing with Sound Systems

This document provides an explanation of the processes and performance of the LecNet 2 solutions to the problems encountered when sound systems are used for teleconferencing. It is intended for system integrators, consultants and others involved in the system design.

*US Patent 5,414,776

LECTROSONICS®
Rio Rancho, NM, USA
www.lectrosonics.com

The problem

Teleconferencing with sound systems presents a unique challenge since the loudspeakers and microphones are located in the same acoustic space. The problem is acoustical coupling between microphones and loudspeakers as illustrated here.

Far side audio is delivered by the loudspeakers in the room and the microphones pick it up and return it to the far side. The delay through this process creates an echo heard on the far side of the teleconference.

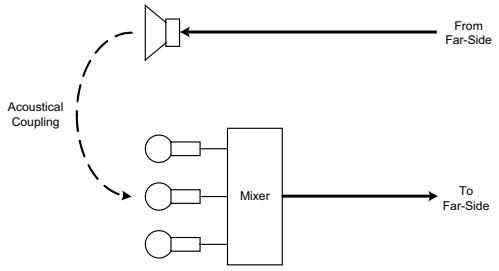


Fig. 1 Loudspeaker/Microphone acoustical coupling

Acoustical coupling can be even worse when the sound system is required to provide both teleconferencing and sound reinforcement. Sound reinforcement will deliver the far side audio signal to the listener's ears evenly around the room, which means that it is also delivered evenly to the microphones.

Two key concepts must be understood to continue a discussion of teleconferencing problems and solutions:

ERL (echo return loss)

ERLE (echo return loss enhancement)

ERL refers to the natural attenuation of the far side audio signal as it travels from the far side through loudspeakers and microphones in the near side room and back to the far side. The object of good teleconferencing room design is to maximize ERL within the constraints of the seating requirements and acoustical considerations. An effort is made to isolate loudspeakers and microphones in the basic room design in several ways:

- Place microphones close to the talker's mouths.
- Position loudspeakers and microphones to minimize coupling, considering the distances between them and their directional coverage patterns.
- Use mix-minus matrix routing to add additional isolation between loudspeakers and microphones.

While these efforts help the system performance, they are not enough to fully eliminate an audible echo at the far side of a teleconference. Additional processes are required to fully suppress and/or eliminate the echo.

ERLE (echo return loss enhancement) refers to additional circuits and processes used to further increase the echo return loss.

The LecNet 2 Solution

The LecNet 2 system uses two processes to significantly attenuate the echo:

- Acoustic echo canceller (AEC)
- Automatic mixing (gain sharing algorithm)

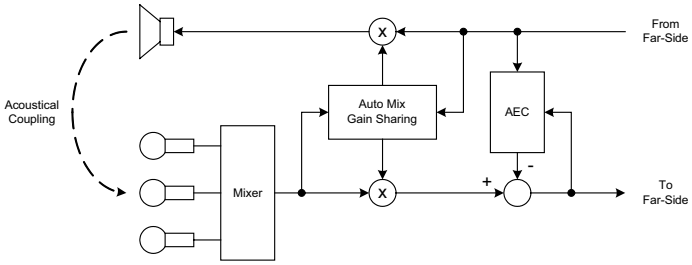


Fig. 2 ERLE processes to increase echo cancellation

Acoustic Echo Canceller (AEC)

Two digital acoustic echo cancellers are provided in the DMTH4 to further reduce the return of local signals to the far side. One operates on the telco connection and the other is dedicated to the video codec connection. Working in conjunction with the automixing process, echoes are minimized and not heard at the far side.

The echo canceller converges continuously when the level of the far side signal exceeds a minimum level, and the ratio of the far side signal to local room sound exceeds a minimum ratio. This dynamic control prevents divergence during periods of silence from the far side room or in "doubletalk" situations. The convergence takes place very quickly to keep up with the changes made by the automatic mixing algorithm and other changes that occur in the room. Setup is greatly simplified and any adjustments, such as level changes made with a remote control system, are accommodated automatically.

The convergence speed is adjustable in the control panel GUI to fine tune it to a particular situation. Faster convergence times can track changes in the room almost instantaneously, but the depth of echo cancellation will be reduced. Slower convergence times take a bit longer to fully converge, but produce greater echo cancellation. The ERLE value achieved by the echo canceller is displayed on the GUI and the effects of altering the convergence rate will be immediately visible and audible.

An important final note is the fact that the echo canceller will never "diverge" (lose convergence). This unique algorithm will also converge on a continuous sine wave, which is especially important when DTMF tones are present in the room.

The plots in Fig. 3 illustrate the contribution to ERLE made by an AEC. These plots apply to any echo canceller, not just the LecNet 2 design. There is a minimum signal to noise ratio required before an echo canceller can converge. The greater the ratio, the faster the convergence, and the longer the time allowed for convergence, the deeper the cancellation.

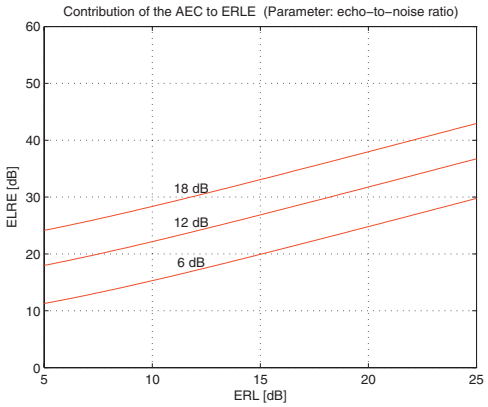


Fig. 3 AEC contribution to ERLE

The sloping lines indicate the ratio of far side audio to sound and noise present in the room, as measured at the microphone. The higher the far side signal is above the sound and noise in the room, the more the AEC can contribute. If the natural ERL of the room is 15 dB and the far side audio is 12 dB above the room sound and noise, the AEC can contribute an ERLE of about 26 dB.

Automatic Mixing

A proportional gain algorithm (US Patent 5,414,776) is used in the automatic mixing in the LecNet 2 system. Gain is allocated among all active channels in a seamless and continuous manner based upon channel activity. The algorithm operates in a natural, transparent manner and incorporates an adaptive AutoSkew™ process to eliminate artifacts such as comb filtering and abrupt gating that occur with conventional automatic mixing schemes. Audio from the far side of a teleconference participates in the local mixing algorithm just like a microphone in the local sound system.

The auto mixing algorithm is beneficial in echo cancellation as well as in sound reinforcement. The signals that are loudest and most active are emphasized and the others attenuated.

In a teleconference, recirculated sound (echo) from the far side is naturally attenuated by the ERL of the room (as measured at the microphone). Because of this, any such recirculated sound is even further attenuated by the auto mixer, effectively doubling the natural ERL of the room.

The auto mixing algorithm also eliminates abrupt gating of the participating signals to provide seamless audio in the teleconference and local sound reinforcement. Teleconference activity is full duplex with smooth transitions in the gain sharing among all active microphones and the far side signal.

The plot in Fig. 4 illustrates the contribution of the auto mixer to the ERLE. In essence, the auto mixer doubles the natural ERL of the system.

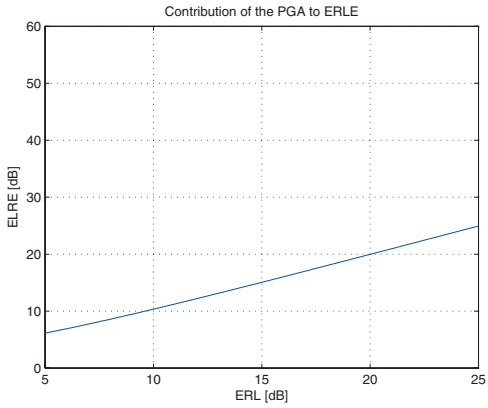


Fig. 4. Auto mixer contribution to ERLE

For example, when the natural ERL of the room is 15 dB, the auto mixer will add an additional 15 dB. The resulting 30 dB of echo return loss is actually adequate for many teleconferencing situations. This reinforces the importance of trying to isolate the loudspeakers and microphones from each other as much as possible in the sound system design.

AEC and Automatic Mixing Combined

The plots in Fig. 5 show the results of the AEC and auto mixer working together to maximize the echo return loss. The sloping lines indicate the ratio of far side audio to the room sound and noise, as measured at the microphone.

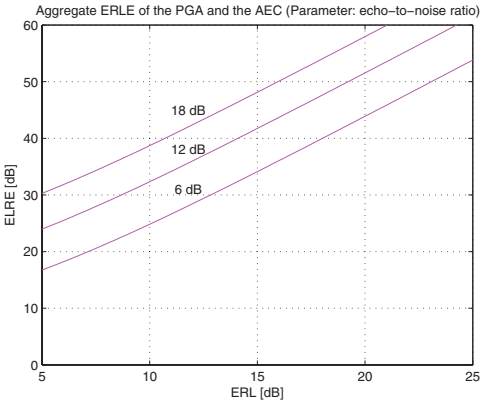


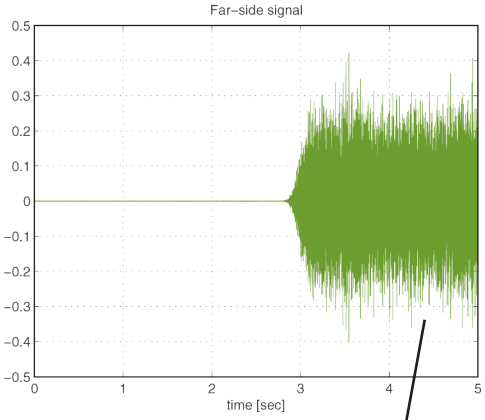
Fig. 5 AEC + Auto Mixer for maximum ERLE

When the natural ERL of the room is 15 dB, and the far side signal is 12 dB above the room sound and noise, over 40 dB of additional return loss is added by the AEC and auto mixer combined. This plus the natural ERL of the room results in an overall echo return loss of over 55 dB.

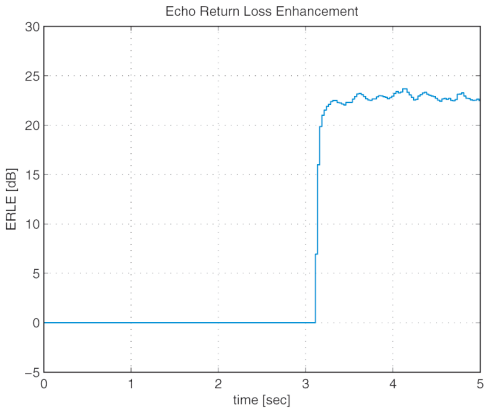
NOTE: These plots are based upon steady state conditions where the auto mixer skewing has settled and the AEC has fully converged.

AEC and Automatic Mixer Dynamics

The AEC and auto mixer operate independently but add to one another during dynamic signal activity to maximize ERLE. This example illustrates the activity during a 5 second time period with a brief period of silence between the near side and far side audio signals.



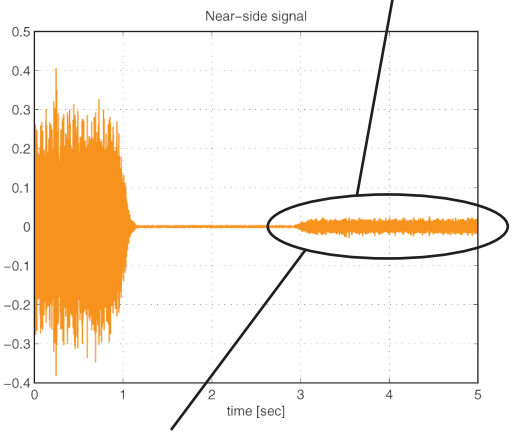
Far side audio signal



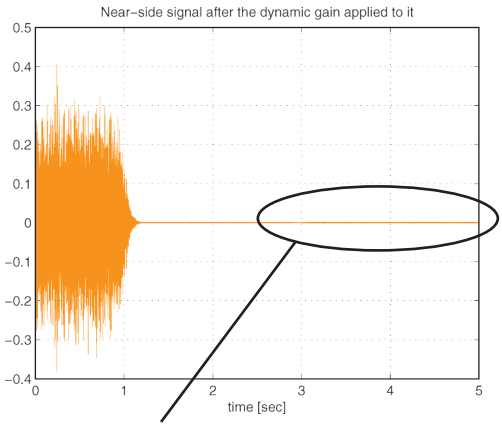
AEC convergence time

The AEC converges extremely fast when a far side audio signal first appears. The ERLE in this example is over 20dB within about 200ms. It is fast enough to follow the auto mixer activity.

This high speed convergence allows a single AEC to be used at the output to cancel the echo from the audio signal sent to the far side.



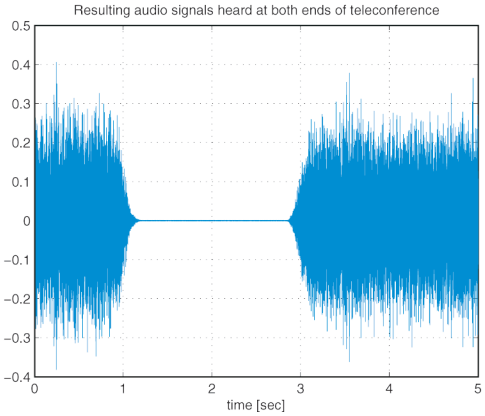
An echo signal is created in the near side room through the sound system loudspeakers and mics



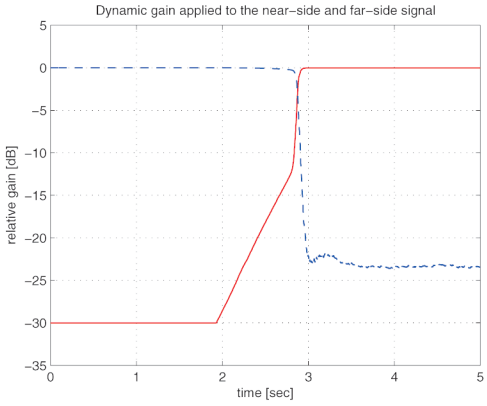
The return loss enhancements eliminate the echo

In addition to the AEC, the auto mixer also attenuates the echo because it is lower in level than the far side audio signal itself. The auto mixer doubles the natural ERL of the system.

The ERLE of the AEC and auto mixer together remove all audible echo from the teleconference.



The signals heard at both sides of the teleconference after processing



Auto mixer gain transition from near side to far side

The gain applied by the auto mixer shifts from the near side to the far side, controlled by the dynamics of the audio signals.

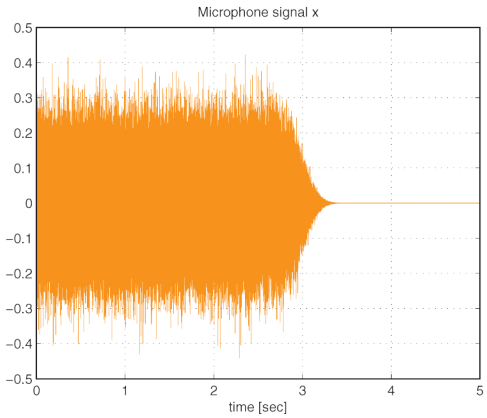
Note that the gain on the near side signal holds steady (“last mic hold”) until the far side signal begins to appear, then shifts quickly to the far side.

The near side and far side signals are not altered by the processing.

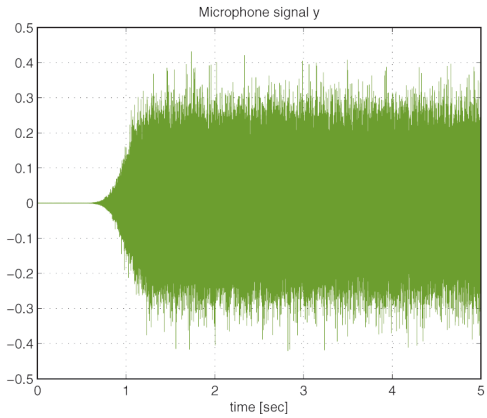
Dynamic activity during a “double talk” situation

In this example, the near side audio signal lasts for about 3 seconds, while the far side audio signal overlaps it and lasts for about 4 seconds. This overlapping situation is often referred to as “double talk.”

When the near and far side audio signals are equal in level, the AEC convergence freezes and maintains the cancellation at its previous setting until the far side audio signal is again dominant.

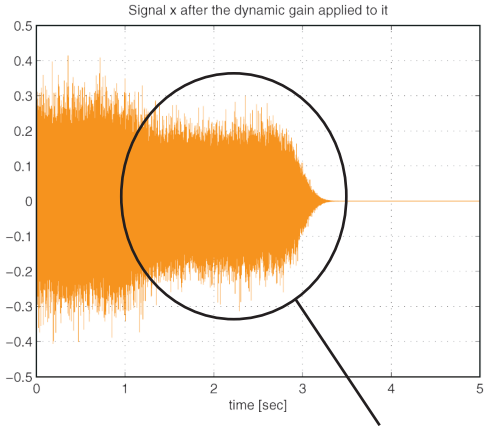


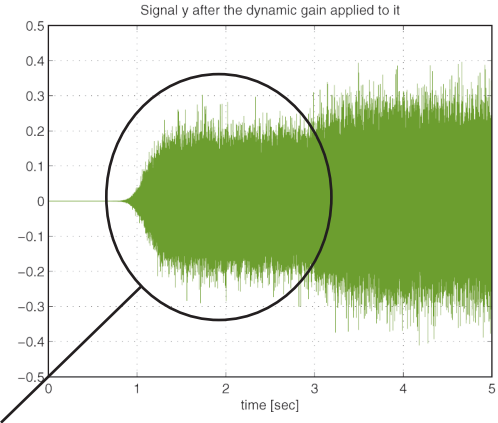
Near side audio continues for about 3 seconds.



Far side audio overlaps the near side by about two seconds.

The original audio signals prior to processing are shown here.

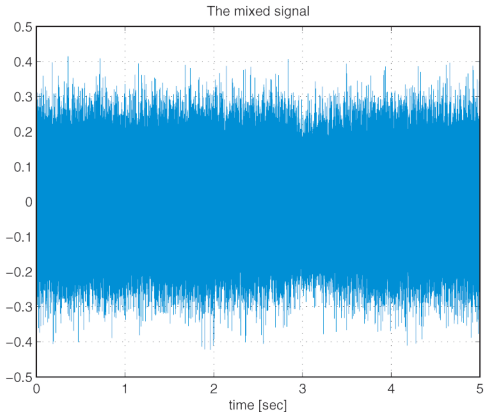




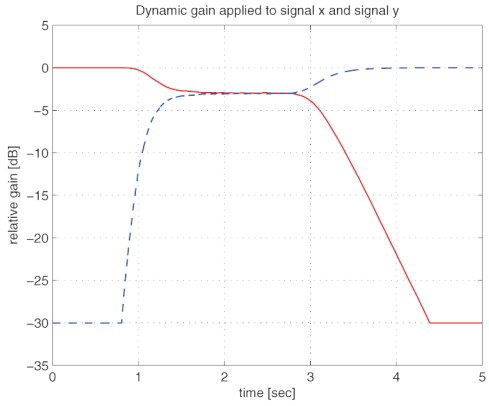
Both near and far side signals are attenuated during the “double talk” period so the sum is the same as a single signal by itself

During the “double talk” period, both near and far side audio signals are attenuated 3dB by the auto mixer so that their sum is the same as one signal by itself.

The AEC convergence is frozen during the “double talk” period and continues cancellation with its previous value until the instant the far side audio signal is again higher than the near side.



The mixed audio heard at both sides of the teleconference



The gain shifts from the near side to the far side over the 5 second time period as depicted here.

The processed audio signal heard at both ends of the teleconference is constant in level after the processing is applied.

The attack time (gain increase) for the far side is controlled by the far side audio signal itself. The gain applied to the near side audio signal falls gradually after the signal stops for a smooth transition.

Conclusion and Benefits

LecNet 2 provides a very effective approach to teleconferencing with sound systems. Good system design is enhanced by automatic mixing and mix-minus zoning, and an advanced DSP algorithm adds sophisticated acoustic echo cancellation. The combination of these processes working together simultaneously eliminates echo heard at the far side of a teleconference and provides sound reinforcement without feedback and ringing at the near side.

The specific benefits are:

- Centralization - a single echo canceller concentrates extensive DSP horsepower into a very sophisticated algorithm rather than dividing the resources across many inputs
- Scalability - a single echo canceller works equally well with any number of inputs, making the sound system expandable - literally hundreds of inputs are possible for a very cost effective solution
- Extremely fast AEC convergence compliments the exclusive Proportional Gain Automatic Mixing Algorithm* to produce excellent total echo return loss
- Does not require calibration or “training” for proper operation
- Easily integrates with other brands of audio equipment as well as earlier LecNet audio components
- Multiple DMTH4 digital telephone hybrids can be used to bridge multiple phone lines and codecs

LECTROSONICS®
581 Laser Road • Rio Rancho, NM 87124 USA • www.lectrosonics.com
Tel (800) 821-1121 or (505) 892-4501 • Fax (505) 892-6243 • sales@lecrosonics.com

10 October 2006