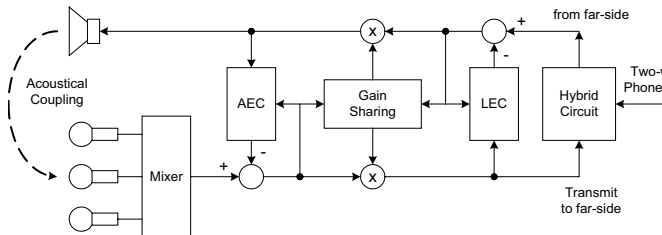


Improved Echo Canceller

Teleconferencing with sound systems creates a difficult challenge in providing clear communications without echo caused by coupling between loudspeakers and microphones sharing the same acoustic space.



DM Series teleconferencing signal flow

Beginning with Ver. 2.0 of LecNet 2 firmware for the DMTH4 telephone interface, improved algorithms are used in the acoustic and line echo cancellers. No hardware modifications were made, so the firmware can be downloaded and installed into earlier DMTH4 processors.

The benefits of the new firmware include:

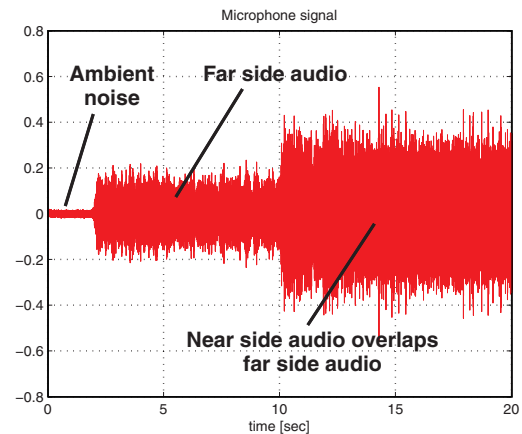
- Automatic convergence rate
- Convergence regardless of system ERL
- “Boa Constrictor” algorithm prevents divergence even during double talk
- Improved metering accuracy
- More robust line echo canceller

Automatic Convergence

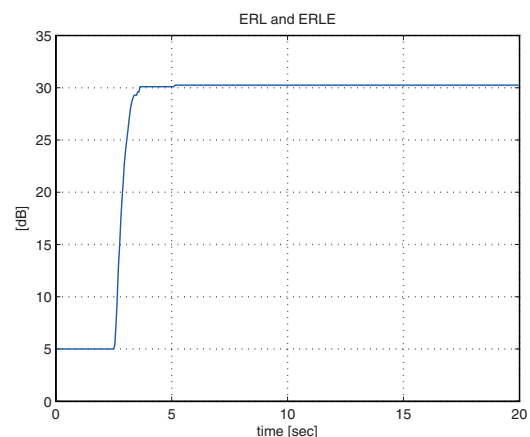
Conventional echo cancellers use a fixed convergence setting that imposes a trade-off between the speed of convergence and cancellation depth. Fast convergence results in limited cancellation depth, so residual echo can still be heard. Greater depth of cancellation requires a slower convergence rate, but may result in audible echo until full convergence is achieved. A fixed convergence setting forces the user into a compromise.

The new algorithm in Ver. 2.0 firmware adjusts the convergence rate automatically, eliminating the compromise imposed by a fixed setting. The process accurately and reliably measures residual echo content in the far side transmit signal. When the content is high, the algorithm allows faster convergence, and when the content is low, the convergence slows down to achieve greater depth of echo cancellation.

A manual adjustment is also provided in the control panel GUI to bias the algorithm slightly toward a faster or slower convergence. With very high local noise levels, the convergence can be adjusted slightly to nudge the echo canceller toward greater depth. In good, quiet local conditions the convergence rate can be adjusted slightly to encourage faster convergence. The manual adjustment is subtle and provided only for fine tuning.



Audio activity over a 20 second time period



Convergence over the 20 second time period shown above

The illustrations above depict audio activity and echo canceller convergence during a 20 second time period. Audio activity for the first two seconds consists of ambient noise in the room, followed by audio from the far side of the teleconference entering the microphones from the loudspeakers. After ten seconds someone begins speaking into a near side microphone, overlapping the far side audio. The lower plot shows how quickly the echo canceller converges, and how it remains converged even during the double talk situation in the last ten seconds of the conversation.

Converge Regardless of System ERL

ERL (echo return loss) refers to the amount of attenuation that occurs in the signal path from far side to local loudspeakers to local microphones and back to the far side. The separation between loudspeakers and microphones and the acoustics of the space determine the total amount of attenuation of the echo heard at the far side.

ERLE (echo return loss enhancement) refers to the additional circuitry and processes used to increase the natural ERL of the system. The primary enhancement is the addition of a DSP-based echo canceller.

In some situations ERL can actually be negative, when there are few or even a single microphone at one side, and a large system with many microphones and loudspeakers at the other side. Loudspeaker/microphone coupling in the large system can generate more signal transmitted on the phone line than was received from the other side, resulting in a negative ERL value.

The improved echo canceller in Ver. 2.0 firmware will converge even with a negative ERL in the system.

“Boa Constrictor” Algorithm Prevents Divergence

An adaptive filter, the core of every echo canceller, will naturally diverge during double talk or when there is excessive noise in the system. Conventional echo canceller designs often employ a dedicated logic module to try to prevent divergence from taking place. The Ver. 2.0 firmware implements a special algorithm that prevents divergence and further enhances the depth of convergence through a unique process.

As the system is running, updated values for the adaptive filter are compared with the current values before being applied. If the new values will increase the depth, they are loaded into the filter. If not, they are discarded. This happens every few milliseconds, so in effect, the process is continuous and the echo canceller remains converged under all but the most extreme circumstances.

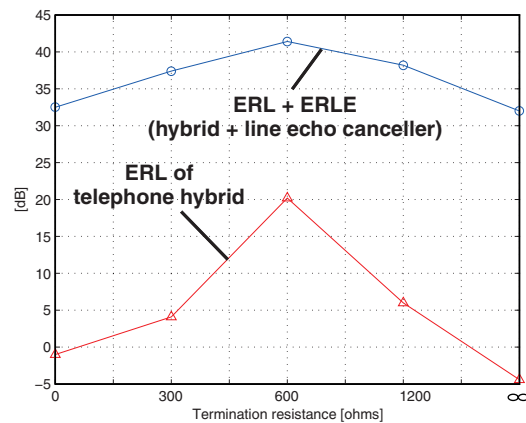
Because of the variable convergence factor, the echo canceller can take advantage of even very brief pauses in speech to converge at a fast rate and make small improvements. Convergence increases in depth continuously with every brief opportunity in much the same manner as a boa constrictor confines its victim. Every time the prey exhales, the snake’s grip tightens, and never lets go. The process takes place continuously and yields impressive echo cancellation depth.

Improved Metering Accuracy

Perhaps a “fringe benefit” but a noticeable result of the new algorithm is improved accuracy of the meters in the control panel GUI. The accurate measurement of residual echo content in the transmit signal provides the data needed to update the meters more accurately than in the previous firmware.

More Robust Line Echo Canceller

The line echo canceller benefits from the Ver. 2.0 improvements as well since it utilizes the same algorithms as the acoustic echo canceller. It is very tolerant of gross mis-matches in the termination impedance of the telephone line as is shown in the diagram.



Return loss in the line echo canceller

The same benefit gained with the acoustic echo canceller in dealing with poor or negative ERL also applies to the line echo canceller. The echo canceller will converge even in extreme cases where the telephone line is shorted or not terminated.

