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DESIGNING CONFERENCE SYSTEMS WITH AEC

Objectives

This application note summarizes the basic concepts of Acoustic Echo Cancellation (AEC) and then discusses certain design guidelines. The final section focuses on simple troubleshooting and system optimization techniques for AEC systems. This document should provide useful information to system designers and installers on how to maximize your AEC designs to achieve the best performance.

AEC basic theory

Acoustic Echo: Reflected signals caused by acoustic coupling between a microphone and a loudspeaker in remote conferencing applications. The "Far End" signal is amplified and reinforced by loudspeakers at the "Near End." This signal gets reflected off walls, ceiling, floors and people before getting picked up by the microphone and transmitted back to the remote site (Far End.)

The following diagram illustrates the concept of acoustic echo.

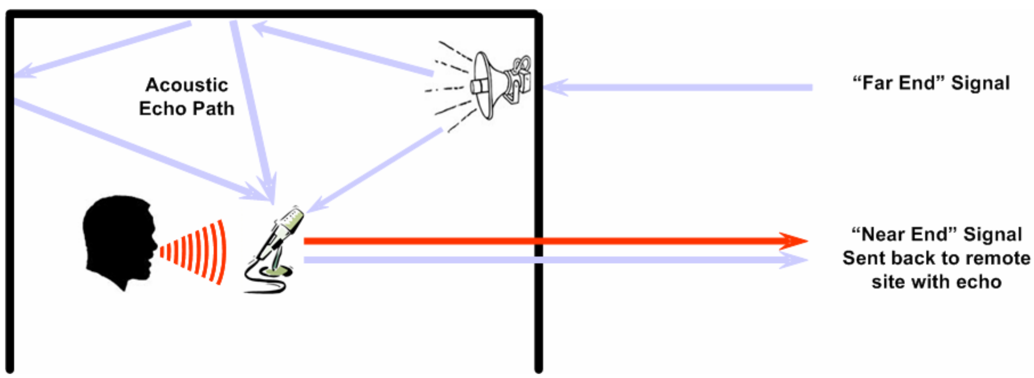

Figure 1 - Typical acoustic echo phenomena

Acoustic Echo Path: An acoustic reflection path that causes echo.
See the diagram above.

AEC Algorithm: By continuously tracking changes in the acoustic echo paths between the loudspeaker and the microphone, the AEC algorithm eliminates the echo.

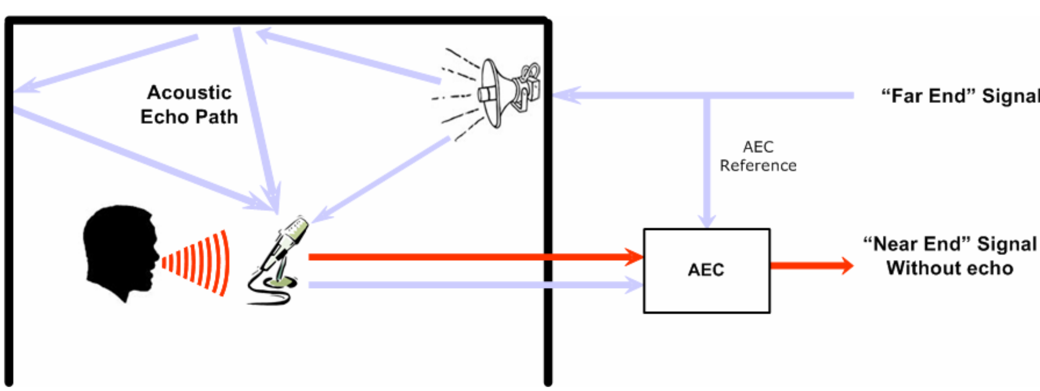

Figure 2 - Typical AEC system

The echo cancellation process:

1. The AEC reference signal is initially sampled by an adaptive filter
2. The parameters of the adaptive filter are adjusted during convergence
3. The resulting filtered signal is a model of the acoustic echo paths, 180 degrees out of phase.
4. Phase cancellation removes the echo when filtered signal is combined to the AEC input microphone signal
5. Non Linear processing removes residual echo if any
6. Finally, noise reduction is applied to remove static background noise

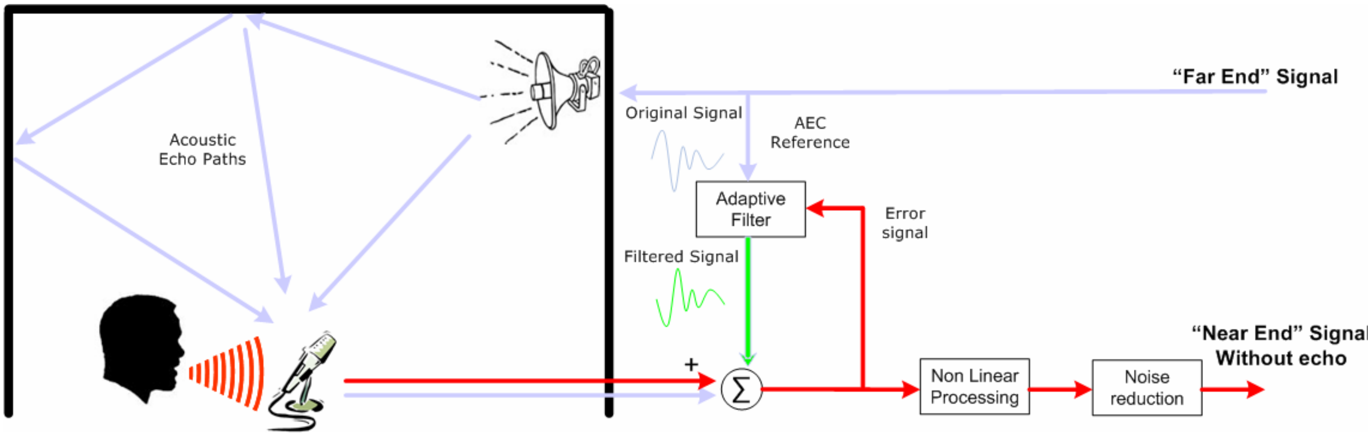

Figure 3 – Detailed AEC block diagram

Why is acoustic echo cancellation so complex?

- The acoustic echo signal picked up by the microphone is different from the AEC reference. It has been amplified, played back in an unknown acoustic environment where acoustic reflections modified the signal.
- An adaptive filter is required to perform what is called "system identification" of the acoustic echo paths, also referred as "training of the filter". This is the process of modeling a signal 180 degrees out of phase to cancel out the far end signal picked up by the microphone. The adaptive filter design is where most of the AEC complexity lies.
- Finally, acoustics of a room may be constantly changing (wireless microphone moving, people coming and leaving the room, etc.) and the AEC algorithm needs to adapt to those changes.

AEC Reference: The AEC reference is the signal that needs to be removed to prevent echo. In other words, it is the far end signal (Codec + Telco) but may also include local sources, such as DVD or CD player that you do not want to transmit back to the remote site.

Referring to the diagram above, the AEC reference is used to model the signal that needs to be removed at the microphone input channel. Based upon the model, the filter is created and continuously updated.

Wideband Echo Canceller: The AEC2W card is referred as a wideband echo canceller since the AEC algorithm operates from 20 to 20 kHz, even in double talk conditions.

A Wideband AEC algorithm is DSP resource intensive and that is why it requires being loaded on-board a dedicated DSP chip for each AEC2W card.

Note: *Echo cancellation is performed on a per channel basis on all Biamp system products. This technique allows each microphone to converge independently and consequently achieve the best results for each microphone channel. DSP processing power is not shared among AEC2w cards and the use of AEC will not affect the normal AudiaFlex DSP resources.*

Tail Length: It is the maximum echo time delay that will be removed by the AEC algorithm. The tail length of the AEC2w algorithm is 128ms.

The following graph illustrates the ranges where the AEC algorithm performs echo cancellation:

- The blue hatched region represents the operating range of the echo canceller
- The grey region (below the noise floor level) represents signals considered as ambient noise; therefore no echo cancellation is to be performed.

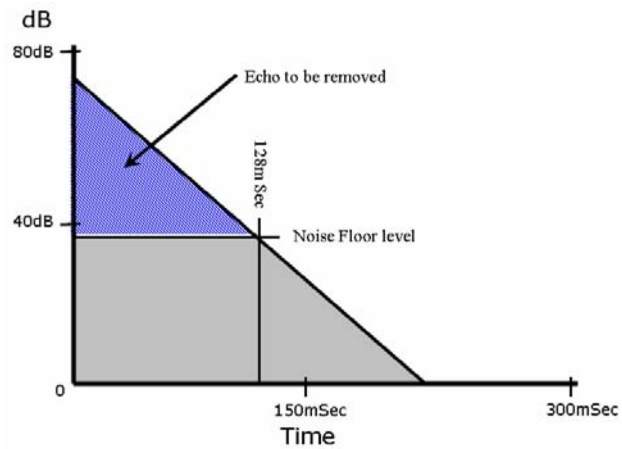

Figure 4 – Operating range of the AEC algorithm

Convergence rate/ Loss of convergence: Convergence refers to the adaptation process inside the AEC algorithm. The convergence rate indicates how quickly the AEC algorithm models the room and adapts to echo path changes. Loss of convergence occurs when the AEC algorithm cannot track acoustic echo path changes therefore echo is heard by the far end.

An algorithm "converges"when it efficiently modifies the parameters of the adaptive filter and echo is removed.

Residual echo: Residual acoustic echo signal that was not removed by the AEC algorithm.

Noise reduction: Automatically removes steady state background noise (e.g. air handling, projector fan, etc.) from the AEC input signal. Noise reduction is a useful tool to improve the audio quality of the signal transmitted back to the remote site.

Double talk: This occurs when speech is generated on both ends of the line. It is the most demanding state for the AEC algorithm.

Far end: The side on which the AEC algorithm does not operate.

Near end: It refers to the side upon which the canceller is designed to operate. Remember that while the AEC operates at the near end, only the far end benefits from acoustic echo cancellation. This explains why AEC systems are required on both ends for a complete conferencing solution.

Echo Return Loss (ERL): Amount of acoustic echo loss (in dB) between the AEC reference and the microphone. Negative values indicate a loss and are usually desired for best performance.

Echo Return Loss Enhancement (ERLE): Amount of echo attenuation (in dB) introduced by the AEC algorithm.

Total Echo Reduction (TER): It represents the sum of the ERL + ERLE and indicates the total echo reduction that was introduced by the room acoustics (ERL), the AEC algorithm and Non Linear Processing (ERLE).

Non Linear Processing (NLP): NLP is equivalent to a sophisticated ducking technology used to suppress any residual echo that the AEC algorithm did not cancel. Three NLP settings are currently available: Soft (set by default), Medium and Aggressive.

Designing systems with AEC

A successful design usually involves a combination of good Audia programming design skills as well as good overall system design skills (including microphone placement, speaker placement, room acoustics, etc). Although an open architecture DSP like AudiaFlex can help overcome some problems, the best designed and performing systems take all of these elements into consideration.

<u>DSP programming guidelines for Audia:</u>

The following section highlights programming guidelines for AudiaFlex platforms fitted with AEC2w cards. Refer to Appendix A (last page of this application note) for a screenshot of the template .dap file.

- **Microphone Inputs**

Always meter your inputs whether they are AEC inputs or not.

Keep unity gain structure across the entire DSP chain (Inputs to outputs).

The front end of an AEC input would typically include RMS meter(s) (target 0dBu), parametric equalizer(s) or filters, and leveler(s) to prevent excessive level changes.

AEC Advanced Features

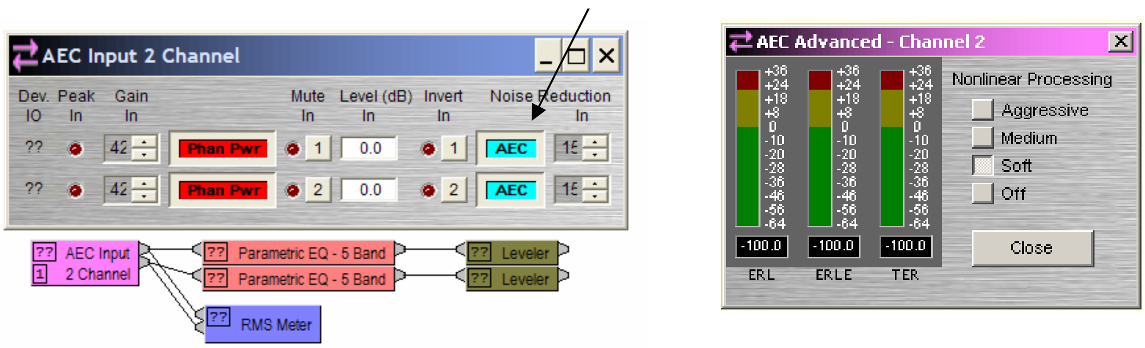

Figure 5 - Typical AEC microphone input and Advanced AEC settings

Right click on the AEC toggle button to access Advanced Features of the AEC algorithm. (*Remember that these settings are on a per channel basis and need to be updated on all inputs for consistent results*.)

This dialog box allows one to meter ERL, ERLE and TER values as well as adjusting NLP settings.

The following table summarizes the influence of ERL values on the system performance.

ERL Values	Most probable cause	System performance	Solution
Positive Values 0 < ERL < +8 dB	• Volume of amplifier is too high • Inadequate microphone/ speaker placement • Input gain of far end signal too low	• Average or poor performance results • AEC algorithm may not be able to converge properly • Residual echo is being heard	• Lower the volume of the amplifier • Adjust input gain • Check gain structure
Negative Values -16 < ERL < 0 dB	• Acoustic echo loss between the loudspeaker and the microphone.	• Best performance results	No modification to system required
Extreme Values -16 < ERL < -30dB	• Microphone gain too low or reference signal is too high • Gain structure is not uniform	• Average or possible poor performance results • Residual echo is being heard	• Check gain structure • Increase microphone input gain • Decrease level from far end

- **Line inputs**

Set all line input gains for nominal 0dBu reading on an RMS meter.
The front end of auxiliary and codec inputs would typically include parametric equalizers and level control blocks.

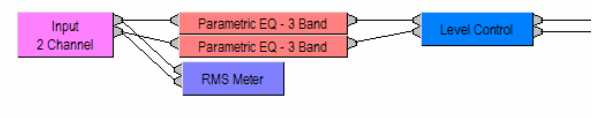

Figure 6 - Typical line input section

NOTE: *Some video codecs may have built in AEC feature. Make sure to disable the AEC inside the codec for best results.*

- **Telephone inputs**

The "Far end" signal typically comes from the phone line or video codec in a conference system.
Remember to meter your input signal and adjust gain for nominal 0dB reading on an RMS meter.
A parametric EQ may help improve the poor audio quality of the typical telephone signal and a leveler will prevent excessive level changes heard from different calling sites.

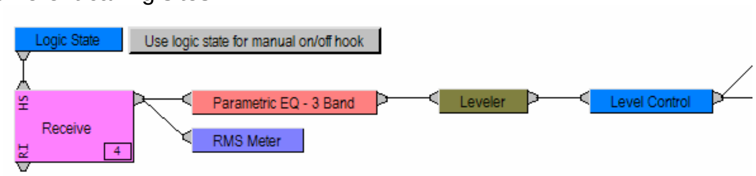

Figure 7 - Typical phone input section

NOTE*: A logic state is an efficient way to manually control the Hook State (HS) of the receive block (On or Off Hook).*

- **Muting of microphones**

Mute the AEC microphone input signals inside the AudiaFlex and not at the microphone.

Muting before the AEC input (e.g. muting locally for a push to talk microphone) affects the convergence rate of the AEC algorithm as the adaptive filter needs to be "trained" each time the microphone is un-muted.
Here is a typical design to internally mute microphone with contact closures.

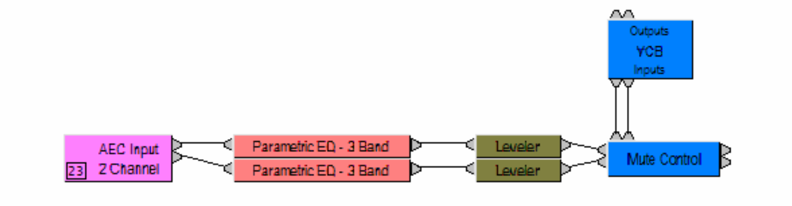

Figure 8 - Typical Muting of microphone

- **Auto Mixer**

An Automixer is strongly recommended for all AEC microphone inputs for the following reasons:

- Automatic system gain adjustments based on the Number of Open Microphones (NOM)
- Improve Signal to noise ratio by keeping unused microphones gated off
- Ease the complex work of the AEC algorithm by gating unused microphone channels

If local speech reinforcement is not a requirement, the mix output of the Automixer will be sufficient for your application.

On the other hand, if local speech reinforcement is required, we recommend using the direct outputs of the Automixer for a mix minus design on multiple loudspeaker zones. It will efficiently optimize the gain of the system before feedback.

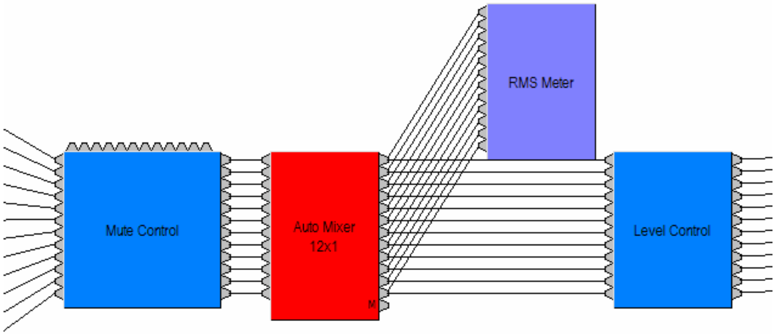

Figure 9 - Typical mute + Automixer for a Mix-Minus design

NOTE: *RMS meters on the outputs of the Automixer are a useful tool for checking whether a microphone gate is ON/ OFF.*

- **Level Control**

The optimal location for input level change is <u>after</u> the Automixer and <u>before</u> the Matrix mixer.
If level control were to be done before the Automixer, it may affect the Adaptive Threshold Sensing (ATS) of the Automixer i.e. how gates turn ON/OFF.

We generally recommend restricting the range of the user-level control to ensure that no major level changes will affect the AEC algorithm performance.

- **Matrix Mixer**

While setting cross points inside the matrix mixer; use the signal path identifier to check whether routing is correct or not. To enable the signal path identifier, right click on a connection line and select “**Persistent Signal Path Identifier**” -> “**Normal**”.
Route to the AEC reference signals that you do not want to transmit back to the remote site, such as:

- Far end signal from the Telco input
- Far end signal from the Video codec
- Locally reinforced Aux sources if any

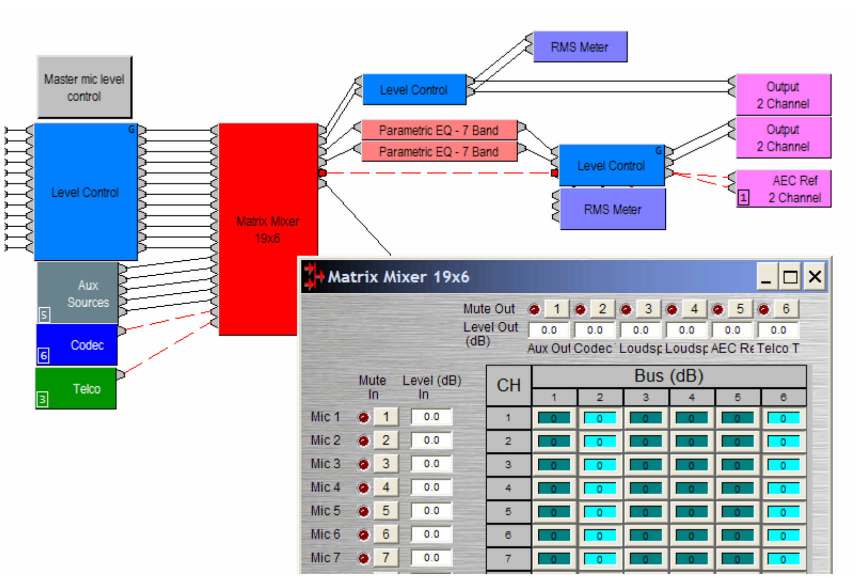

Figure 10 - Using the signal path identifier to check proper routing to the AEC reference

NOTE: *Double check for signal routing to the AEC reference to prevent unnecessary troubleshooting.*

- **Output level control for sound reinforcement**

Level control for the sound reinforcement signal needs to be "*synchronized"* with the level control to the AEC reference. Any level changes made to the sound reinforcement feed needs to be reflected to the AEC reference.

Illustration on where level control should happen internally.

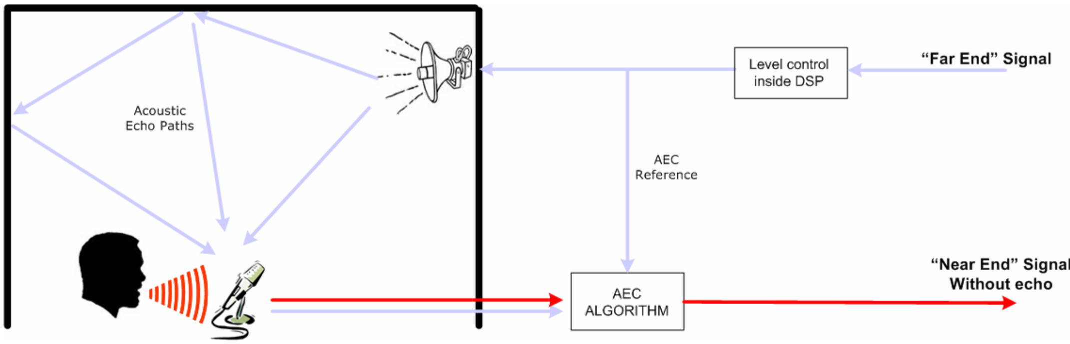

Figure 11 - Illustrating Output level control options

To best achieve this requirement, a ganged level control block can be used to control the feed to the loudspeaker and the AEC reference. See example below illustrating this.

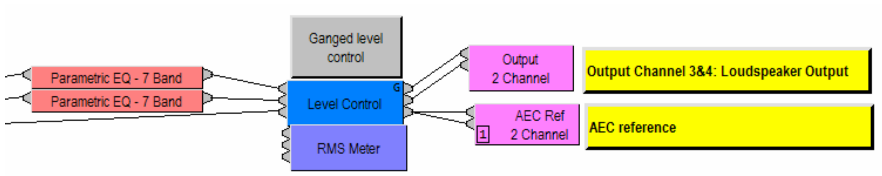

Figure 12 - Typical ganged level control block for feed to AEC reference and Sound reinforcement

- **AEC reference**

For best system performance, the AEC reference level reading on an RMS meter shall be within -3 and +3dBu.
We generally do not recommend muting or changing routing of the far end signal to the reference in the middle of a call. Doing otherwise increases the convergence rate since the adaptive filter needs to converge each time the signal is re-routed to the AEC reference.

<u>System design guidelines:</u>

- **Microphone / Loudspeaker placement**

Direct loudspeaker to microphone coupling should be avoided for best system performance. This general design guideline is no different than a typical installation where the designer will try to optimize gain before feedback.

Ceiling microphones, while commonly used, can be problematic for the following reasons:

- Worst conditions for acoustic coupling between the microphone and the speaker. ERL values will very likely be positive since acoustic echo loss between the loudspeaker and microphone is minimal.
- Poor gain before feedback
- Poor voice pickup and Signal to Noise Ratio if microphone located at proximity of Air duct, light ballast, etc.

- **Room Acoustics**

Acoustic treatment matters! Whether you are using AEC or not, rules of physics still apply. Similarly, an AEC system is not a solution to improve intelligibility in a highly reverberant and noisy environment.

Remember that surfaces such as carpets, acoustic tiles, and bookshelves are good surfaces to absorb, dampen or at least attenuate acoustic echo. On the other hand, highly reflective surfaces (glass, bricks..) will tend to reflect most of the acoustic signal and create reverberation that may exceed the tail length limit of the AEC algorithm.

Step by step procedure: Setting up an AEC system

Each installation is unique therefore each system will require unique settings and adjustments. Nevertheless, the following steps will provide good performance results for most typical AEC system installations:

1. Turn down your power amplifiers
2. Adjust your microphone input gains so that your RMS meters are showing 0dB approximately when someone is talking into the microphones.
3. Adjust your gain structure throughout the entire system for unity gain. This corresponds to our previous recommendations for having RMS meters across the entire DSP chain. Slowly bring up the levels of your amplifiers until you have sufficient signal
4. Do a test call and then adjust microphone levels as required.
5. Check ERL values in the advanced parameters dialog box. A countdown is good practice to test the AEC algorithm with strong attack syllables.
6. Once operational, do minor level changes as required but do not change the volume of the amplifier.

Troubleshooting AEC systems

The following simple troubleshooting techniques should be used whenever required. If you have any questions or need help troubleshooting your installation, do not hesitate to contact a member of the technical support team.

Residual Echo:

- Verify that routing to the AEC reference is correct
- Meter the signal feeding the AEC reference and make sure it is within recommended range -3dBu to +3dBu
- Adjust NLP settings from soft to medium. If echo is still being heard, switch to aggressive.

Positive ERL values:

There is probably too much acoustic coupling between the microphone and the speaker as explained in the AEC basics section. Possible causes may be:

- Amplifier volume is too high
- Poor microphone/speaker placement
- Input gain of the microphone is too high
- Gain structure is not optimized

Automixer is not gating ON&OFF properly:

- This symptom indicates that the Adaptive Threshold Sensing (ATS) is not working properly. Make sure that all microphone inputs have nominal 0dB gain when talking into the microphone.
- The combination of a leveler and high microphone input gain is an efficient mean to get a consistent signal that will always trigger the gate of the Automixer

Conclusion

This application note has covered most of the basic concepts about AEC as well as recommended design guidelines for implementing AEC in your systems. We hope that you now have a better understanding on how AEC works and are able to make use of this document for your future projects.

If this application note did not answer your questions on the topic of "Designing conference systems with AEC", please contact Biamp Technical Support Group by phone at 1-800-426-1457 (US & Canada)/ +1-503-641-7287 (International) or by email at support@biamp.com.

Appendix – AEC conference system template