

Avia™ Acoustic Echo Cancellation Notes

Applies to DSP-1282 and DSP-1283

Crestron Electronics, Inc.

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Contents

1	Teleconferencing/Videoconferencing
1	Room Considerations
2	Source: Microphone Inputs
2	Initial Set Up
3	Source: Dante™ Inputs
4	Routing
6	AEC Configuration Selections
7	Metering
7	Expanded Controls
8	Best Practices
9	AEC Check List

Teleconferencing with POTS and Videoconferencing with SIP

An AEC (Acoustic Echo Canceller) is used to remove echo on a POTS or SIP call. When AEC is not used, call participants may hear echoes over local loudspeakers as voices are picked up by the microphone(s) in the far room and returned in the local room. An AEC eliminates echo, improving intelligibility and call quality for successful communications.

Room Considerations

AEC performance and the quality that can be expected on SIP & POTS call is related to the preparation of the call environment.

A well designed acoustic environment will be inherently less prone to call echoing and problem audio. Reverberant rooms (aka "live-rooms") are usually designed with an abundance of hard and reflective surfaces that negatively impact intelligibility and make the AEC optimization more difficult. In some live rooms the AEC implementation may not provide acceptable results. Large glass windows and hard tile floors are examples of highly reflective surfaces that contribute to a room that's acoustically bright and live. An environment that is designed to address acoustic considerations will typically use materials and configurations to minimize reflective acoustic energy, attenuating echoes and ultimately improving AEC effectiveness.

Other factors contributing to AEC performance may include local mechanical noise from HVAC systems, ambient noise from sources in adjacent rooms or from outside sources such as street traffic, trains, and planes. All of these sources will contribute to the NC or Noise Criteria of an interior room. NC standards have been around since the 1950's (when very noisy HVAC systems first came on the

scene) and they have been refined to address a wide variety of contemporary room use cases. The Acoustical Society of America (ASA) and American National Standards Institute (ANSI) have published standards (ANSI/ASA S12.2-2008) for evaluating room noise. Their recommendation is for NC 25-30 for teleconferencing spaces.

Another benchmark measurement is the "RT60" of a space. Generally defined, this is the Reverberation Time for the average sound to decay by 60 decibels. A simple experiment is to clap your hands in your conference room, just one clap; if the room is somewhat reverberant the sound may remain noticeable for a second or two as it decays. If the clap decays in under 1 second the room is less reverberant. The objective is to have a space with an RT60 that's appropriate for your use. In conferencing environments where attention is paid to acoustic design the results will be obvious. It is recommended that a system designer review these issues prior to construction. They should demonstrate (through calculations) that a space will achieve the target NC and RT60. Once a space is completed, they can perform acoustic measurements to verify the targets were achieved.

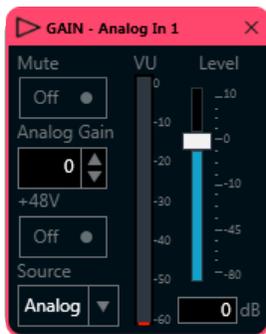
Location and configuration of microphones and speakers will also contribute to call quality. Best performance is achieved when mics are not pointed at the loudspeakers used for call playback; position mics as far away as possible from the loudspeaker. A close proximity of mic to loudspeaker makes it more difficult for the AEC to fully eliminate any call echo.

The noise criteria, reverberation time, placement of microphones and speakers, and AEC tuning will all have an impact on the ultimate call quality. With proper planning and execution the Avia AEC will provide audio that's loud and clear so that your conference call is successful.

Source: Microphone Inputs

Up to 12 microphone channels can be used with an Avia DSP featuring AEC. Each channel has a separate Mic/Line port on the rear panel of the DSP and a separate AEC object for configuration in the software tool. It is necessary to have separate controls for each microphone element because a number of set-up variables come into play when tuning to achieve optimum AEC results. Among the variables are the microphone pick-up pattern (omni-directional, cardioid, etc.), the mic location(s) around a table or suspended from the ceiling, and if you are combining various mic styles such as multi-element table "pucks" or single element goosenecks with overheads. (E.g. a goose neck personal mic is set up for the chairman at the head of the table and puck mics set up for the other participants.) The proximity of the mic(s) to the room loudspeakers and the volume level of the playback system will have an impact on the tuning of the AEC and final performance.

Initial Set Up



In the Gain object, turn on phantom power for condenser mics; no power is required for dynamic mics. Set the analog gain to a level appropriate for the microphone that provides strong level but doesn't distort or clip the input.

Analog Gain adjustments are important to establish a good mic level prior to the ADC and pre-AEC processing. Too high of a level ("hot mic") and you'll risk clipping the input to the ADC which results in distorted audio. Once the audio becomes distorted it cannot be corrected later in the signal chain. The objective is to have a strong input signal while allowing for adequate headroom that keeps the conversation in a safe operating range when a participant gets too close to the mic or when there are many trying to talk at the same time.

If the mic is set too low there may not be sufficient levels for the AEC to compare the audio in your room to the far end room. The AEC has to be able to "hear" both sides of the conversation to eliminate undesirable echo and ambient noise. Set a gain that best captures all of the conference participants and accommodates talkers at different volume levels and different locations relative to the microphone.

Tip: Anytime you engage phantom power to a condenser microphone the input should be muted. The application of phantom power will generate a pop that will playback through the system if unmuted.

Tip: Variances between mic products and installations may require different Analog Gain settings. We find that this value is typically between 30 to 36 dB with a high quality condenser mic in close proximity (1 meter average) to the talkers.

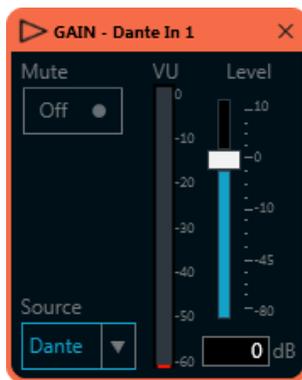
Tip: The Level Control slider control is post ADC and should be set to 0 with the appropriate upstream Analog Gain.

Tip: Once you have configured the channel strip and AEC you can save this configuration for reuse in additional rooms using the same type mics. The AEC operation may not be exactly the same due to differences in room size, mic and speaker locations but with a few adjustments you can usually optimize for the local room requirements.

Source: Dante™ Inputs

Dante channel routing to the AEC is available on the DSP-1283. The Dante source is routed to the input channel strip by selecting the Source drop-down; the default selection is analog. When a Dante source is selected the frame around the gain object will change color to orange; this visual cue helps identify the Dante channels routed to an input channel strip. Dante is a digital format so the Analog Gain control is removed along with the microphone phantom power selectors.

Signal Page Gain Object:

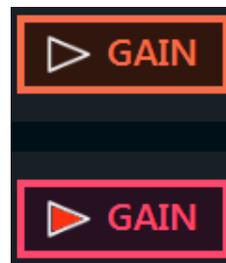


A Dante input channel is digital and the level control is the exclusive adjustment; mute is always provided for an input.

A maximum of 12 Dante channels can be routed through the numerically corresponding input channel strips. Input channels 1-12 will always select the same channel number on the Dante network. For example, if your Dante microphone is assigned to channel 3 on the Dante network- that signal is selectable on input strip 3 in the Avia tool. If a Dante mic is assigned to channel 24 on the Dante bus it is outside the range 1-12 for input strips with AEC in the Avia tool.

There is no mixing of analog and Dante signals on any input channel strip; they are mutually exclusive. An additional 8 Dante channels (sourced from the Dante matrix) can be routed through the Aux strips for EQ and dynamics processing, but not for AEC processing since this function is exclusive to the input strips.

On the signal page, the channel strip gain object displays:



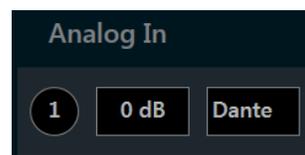
Dante input selection with an orange border.

Muted inputs indicated with a solid red ▶

Analog input selection with a red border.

System Page – Analog In:

Dante input to the channel strip indicated.

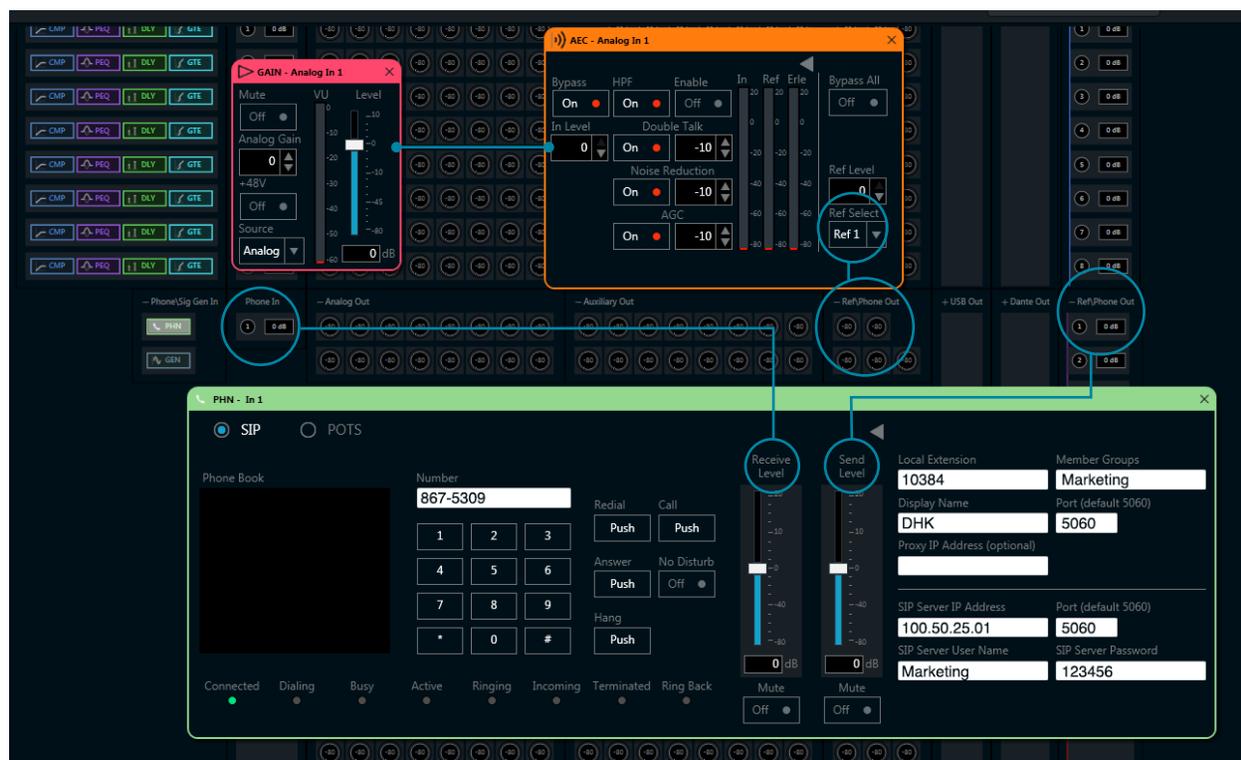


+48V would be indicated if an analog source was selected in the gain object.

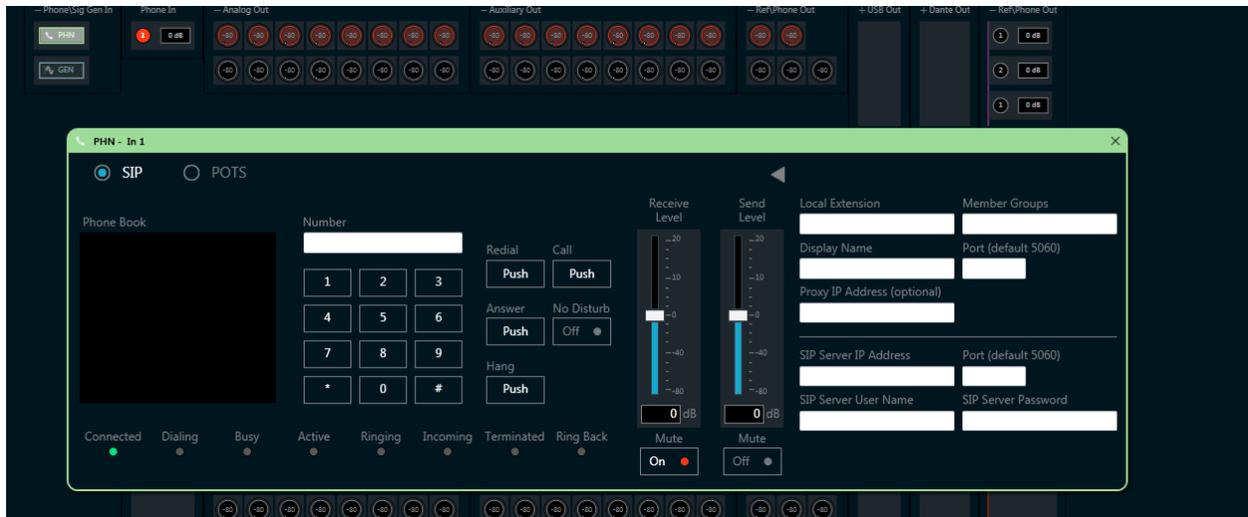
Routing

A basic system requires a microphone input signal in the local room and a signal from the far end room. The near end room mic signal should be connected to an input on the DSP, processed by the AEC, and sent out to the far end room on the Phone Out output in the matrix.

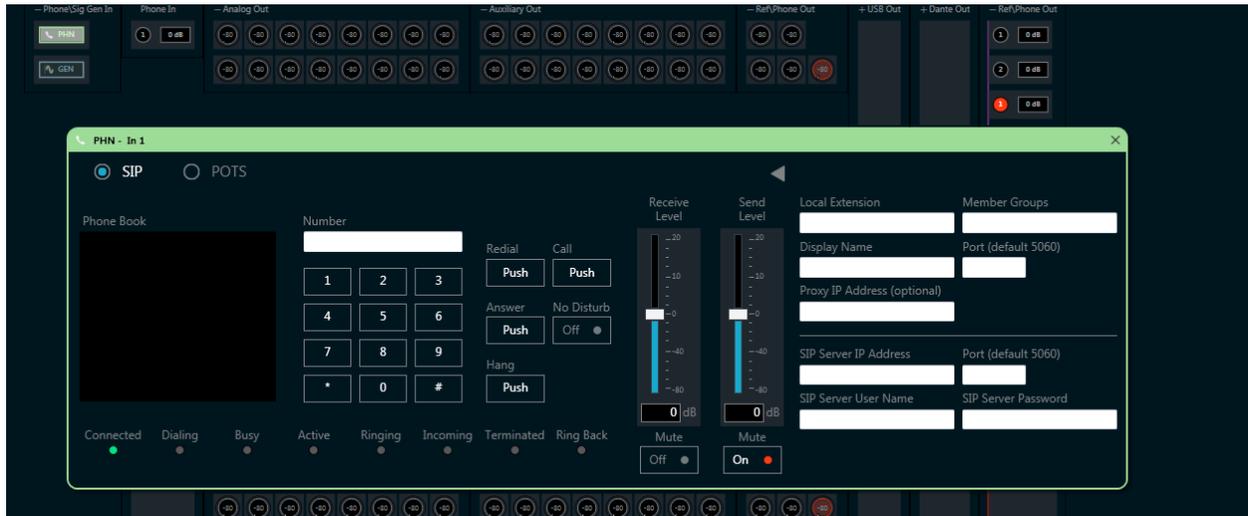
The signal coming from the far end room is connected in the Phone In of the matrix and should be routed to the desired analog output for playback in the local room. This signal must also be routed to one of the reference channels (Ref 1 or Ref 2) for the AEC to sample the content and identify the echo to be canceled.



PHN object muting of the Receive Level results in matrix and Phone In gain mute:



PHN object muting of the Send Level results in matrix and Phone Out gain mute:



AEC Configuration Selections

Enable: This selection routes the signal to the AEC algorithm.

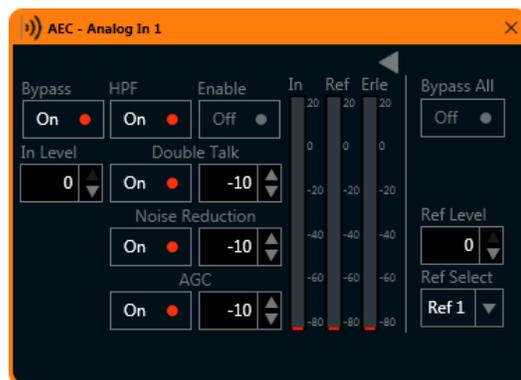
Bypass: This selection will stop AEC processing.

HPF: The High Pass Filter is used to block low frequency noise and mechanical vibrations that can have a negative impact on call quality. Low frequency street noise, HVAC rumble and equipment noise generated in the room may be mitigated by engaging the HPF.

Double Talk: This is a condition under which both ends of a call are talking at the same time. To improve intelligibility and to prevent echo, use the Double Talk function. The value set for double talk is the amount of attenuation applied to the outgoing audio. Setting this value too low can potentially result in audio artifacts or distortion.

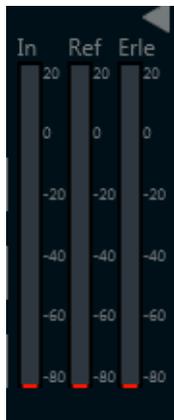
Noise Reduction: Reduces background noise coming from the local room during a call. By turning on Noise Reduction and setting the attenuation value, call quality and intelligibility can be improved. The algorithm determines what is speech versus what is extraneous noise being picked up by the mics. The value set for noise reduction is the amount of suppression applied to the noise present in the local room that's on the outgoing audio. Setting this value too low can potentially result in artifacts or audio distortion.

AGC: Automatic Gain Control is used to regulate audio levels for consistency. By turning on the AGC (after the AEC levels are established) and setting a target, the signal that is below the threshold will be increased in level to meet the target; a signal above the threshold will be attenuated to meet the target. The AGC can be useful for bringing up the audio level for a soft talker or bringing down the level for the loud guy. An AGC target level set too high can result in distortion or clipping, while setting the target too low may not deliver sufficient level to AEC.



AEC tuning occurs in this window. The AEC algorithm is not active when Bypass is on. The AEC object does not have audio routed to it when Enable is off. Both selections are required for active AEC processing.

Metering



In: Indicates the level of the input signal sent to the AEC algorithm.

Ref: Indicates the level of the signal used as a reference for the AEC algorithm.

Erle: The Echo Return Loss Enhancement (ERLE) meter indicates the amount of attenuation being applied to the signal as the echo is attenuated, canceled, and removed from the audio.

A value of -20 means that 20 dB of echo has been removed from the audio.

Expanded Controls

When the AEC object window is expanded several additional controls are exposed:

Bypass All: Stops all AEC processing. This applies to all AEC strips processed in the same DSP unit.

Ref Level: Used to control the level of the signal that is the reference for the AEC. If the level of the incoming signal from the far end is high, it may help to attenuate the reference to the AEC to prevent clipping. The Ref Level must be strong. Weak signal limits the AECs ability to "hear" the conversation so it can identify and cancel the echo.

Ref Select: Ref Select routes the output of the PHN object to the AEC on either reference channel Ref 1 or Ref 2. When selecting a reference for the AEC, make sure that only the desired reference signals are being routed and mixed into the reference channel.

The only signal that can be on the Reference channel that's routed to the AEC is the output of the PHN object. Any other signal will kill the AEC.

Once the microphone audio passes through the input channel strip and the AEC, it must be routed to the Phone Out cross- point in the matrix. If proper gain has been set up in the input strip then we look for an optimum setting of 0dB at the cross point



Routing to the analog output (optionally to Dante out if going to a Dante enabled unit) and to a reference channels.

The Phone In level control sets the audio from the far end.

Best Practices

- AEC is a dynamic function and always precedes all other processing.
- Routing a local mic used in a SIP or POTS system into the matrix and through any Aux strip will result in convergence issues and a failed AEC.
- The spectrum analyzer object is a great tool for finding resonant room modes - all rooms have them. Most modes will be related to room dimensions and the path length ("the wave") of the audio from the playback loudspeaker system bouncing off of reflective surfaces. Depending on room design including dimensions and placement of mics and loudspeakers, the modes can build to the point they impact system intelligibility.
- Use the spectrum analyzer to gather room information that's useful in configuring the EQ objects in the output strips for loudspeakers. Use EQ cut to minimize ring or build-up with incremental inputs to tame a resonant mode. Too aggressive of a room tuning can overshoot your target. It is best to ease in small amounts of EQ and test your new settings using the spectrum analyzer to confirm your EQ input is achieving the desired room result.
- Check the Avia library for pre-made channel strips with various microphones. The strips library will grow as new mics are tested and optimized. These can provide a solid starting point for system set up and optimization.
- Check your gain structure: If you don't set proper gain structure throughout the signal path the AEC performance will suffer or possibly not work at all. Usually around 30 to 36 dB of analog gain is required to bring a condenser mic to a useable level. If too little gain is applied, the signal will be weak and it will impact AEC effectiveness. If too much gain is applied you may clip the signal and you'll have no headroom for the dynamics of live operation.
- Post AEC- a clipped signal can't be corrected downstream. Spending time to correctly adjust the AEC will contribute to good audio results. Ensure that the levels through the downstream signal chain are also optimized for top performance.

DSP1282 & DSP1283 AEC Check List

1. Source selection – connect the microphone:

- Make your balanced mic connection to the DSP rear panel Mic/Line input
- You can also select a Dante source if using the DSP-1283
- Confirm the source selection in the gain object; it's either an analog input from the rear panel or Dante channel from the matrix
- Mute on - Analog condenser mics - turn on phantom power
- Mute off - Set Analog Gain to an appropriate value
 - Use the VU meter as a reference to avoid clipping
 - Set gain- usually about 30 to 36 dB of gain is appropriate for most mics

2. Route the mic to the Phone Out output at a level of 0 dB in the cross point:

- The mic can also be routed to the USB output if an external codec is being used to initiate and manage the call
 - If an external codec is being used, disable its AEC so that it does not interfere with the AEC in the Avia DSP

3. Open the AEC object in the input strip:

- Turn Bypass off for the AEC object in the input channel
 - Also ensure that Bypass All is turned off in the AEC global controls
- Turn Enable on
- Turn the HPF on
- Turn the Double Talk on and set the value to -10
- Turn the Noise Reduction on and set the value to -20
- Turn the AGC off. You don't want to be chasing the AGC - it is dynamic and should not be on during AEC level tuning
- Set the In Level to 0
- Set the Ref Level to 0
- Choose the Ref Select (Ref1 or Ref2) for the incoming call audio

4. Repeat for each microphone to be used in the call. Each AEC must use the same Ref Select channel.

5. Route the Phone In to a matrix Analog Out to play incoming call audio from the far room on the near room loudspeakers.

- The Receive and Send Level controls in the PHN object are the same controls as the input and output trims on the matrix for the Phone In and Phone Out channels
- Set the cross points and trims on the matrix or the controls in the Analog Output/ channel strip to adjust the level being sent to the near room amplifier and speakers
- Phone In can also be routed to a Dante output (if using a DSP-1283) for routing to a Dante enabled product
- If an external USB codec is being used to manage the call, route the USB In from the matrix channel and call source to either:
 - the analog output feeding the external device
 - the USB channel feeding the external device

6. Route the Phone In to one of the two reference channels. Optimum 0 dB at the cross point. This is the reference signal used by the AEC algorithm.

- If an external USB codec is being used to manage the call (Crestron RL for example), route the USB In from the USB channels that the call is on to the reference channel
 - If the call is coming in as a stereo signal (2 channels) from the USB codec, route both the left and right channels to the same reference channel
 - Make sure that the call AEC is off in the external Codec

7. Make the call:

- If using the SIP call option in the Phone object make sure that all of the SIP information such as server IP Address, extension, username, and password are entered and correct
- If using the POTS call option in the Phone object dial the number from the key pad
- If using an external Codec refer to the appropriate user manuals and guides for disabling the AEC in that device
 - In this situation, the Avia DSP and its AEC is used for the call audio processing only, typically output on 2 channels USB. The third party Codec is used for the session initiation and call management

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Avia™ Acoustic Echo
Cancellation Notes
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