CRD1569-1

Dual Microphone Tuning Guide

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1 Introduction

This document provides an overview of the set up and tuning of the dual-microphone voice capture board, part of the CRD1569-1 Cirrus Logic Voice Capture Development Kit for Amazon AVS-Enabled Products. It enables users to tune the solution and produce functional system tunings.

2 Solution Overview

2.1 Reference Design Implementation

The reference design implementation is as shown in Figure 1 and comprises:

- web-based tool for tuning and diagnostics
- CS47L24 on voice capture board
- Raspberry Pi 3 with AVS Client Application and Sensory's TrulyHandsfree™ wake word engine

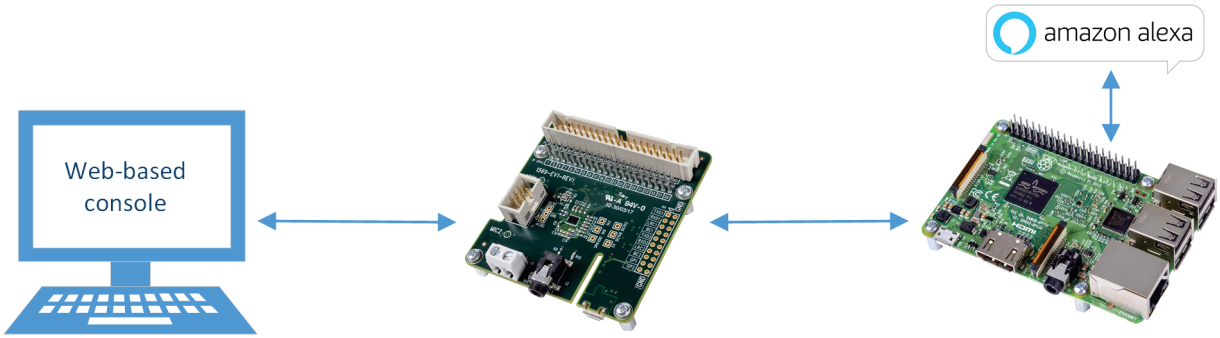

Figure 1 Reference Design Implementation

2.2 Solution Software Block Diagram

The reference solution software runs on the CS47L24 device's DSP3 and is structured as shown in Figure 2.

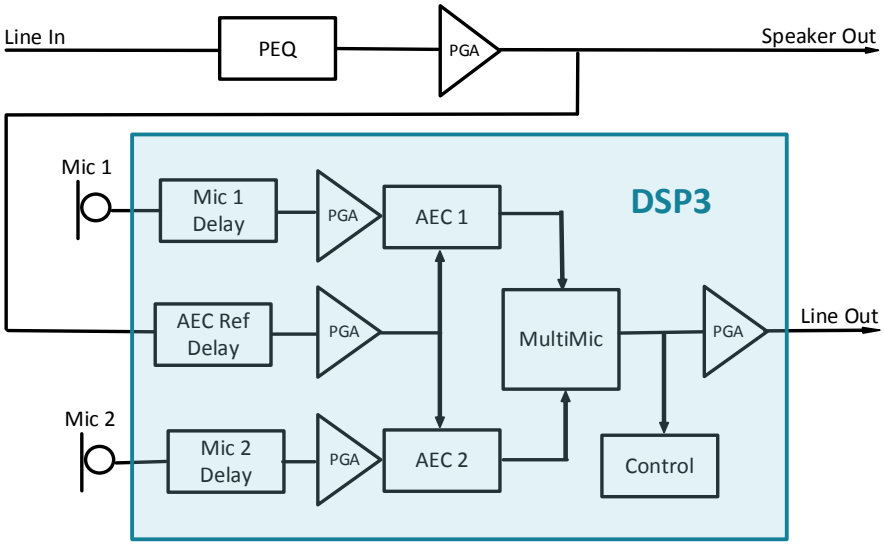

Figure 2 Solution Software Block Diagram

The solution software provides:

- gain control for line in signal
- gain control for microphones
- gain control for speaker out
- AEC reference path bulk delay
- microphone path bulk delay
- acoustic echo cancellation (AEC)
- multi-microphone processing
- SoundClear® software system control
- SoundClear software bypass mode

Additionally, there is a five-band EQ in the Rx path to shape the sound.

2.3 Tunable Parameters

The following parameters are tunable:

	Units	Minimum	Maximum
Software Enable			
SoundClear Enabled	boolean	0	1
Pass-Through Tx Gain	dB	-INF	24
Solution Tuning Parameters			
Microphone Input Gain	dB	-INF	24
Line Input Gain	dB	-INF	24
Line Out Gain	dB	-INF	24
AEC Reference Delay	mSec	4	12
Microphone Bulk Delay	mSec	0	12
Microphone Spacing	mm	0	512
Double-Talk Detection Threshold	—	0	1
Tx Minimum Speech Threshold	dB	-INF	6
Speaker Out Gain	dB	-32	16
CODEC Tuning Parameters			
Rx Path Equalizer — 5 Bands	dB	-12	+12
Real Time Diagnostics			
Audio Levels			
Microphone In 1	dB	-100	0
Microphone In 2	dB	-100	0
Line Out	dB	-100	0
Line In	dB	-100	0
Audio Mode			
Tx & Rx	boolean	0	—
Rx Only	boolean	1	—
Tx Only	boolean	2	—
Silence	boolean	4	—
ERLE	dB	0	60
DTD	—	0	1

3 Reference Design Software Tuning Process

The software tuning process assumes that the speaker/amplifier configuration is in a known fixed configuration relative to the microphones, and that the targeted speaker volume is defined and available.

The software tuning process flow is:

Step 1: Configure multi-microphone parameters

Step 2: Balance Rx path gains and Rx EQ

Step 3: Adjust Tx path microphone input gain and line out levels

Step 4: Configure AEC parameters for optimal performance

Step 5: Verify system performance

These steps are shown in the Control Console Configuration panel, as shown in Figure 3, and are covered in the following sections.

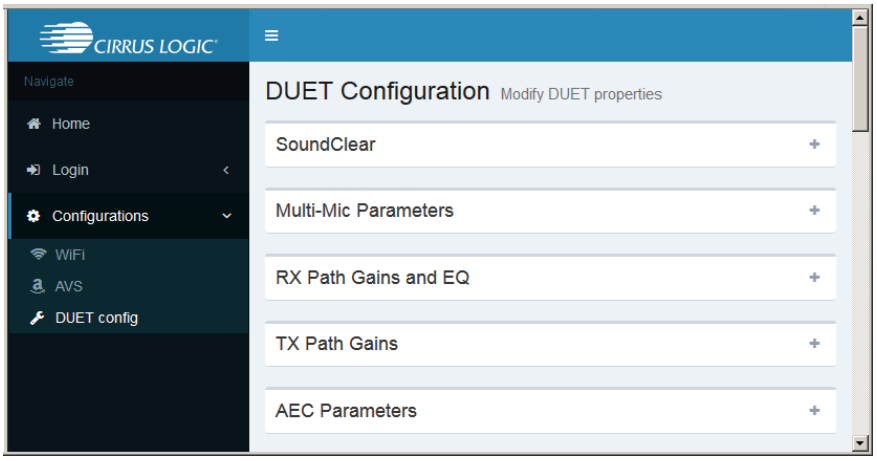

Figure 3 Control Console Configuration Panel

3.1 Multi-Microphone Setup

On the voice capture board, the physical microphone spacing is 25 mm.

The best performance is achieved with a physical microphone spacing of 25 mm and a tuned acoustical setting of 27 mm.

The microphone spacing should only be changed if the end-user design has a different physical microphone spacing; the spacing is changed in the *Multi-Mic Parameters* box in the Control Console Configuration panel, as shown in Figure 4.

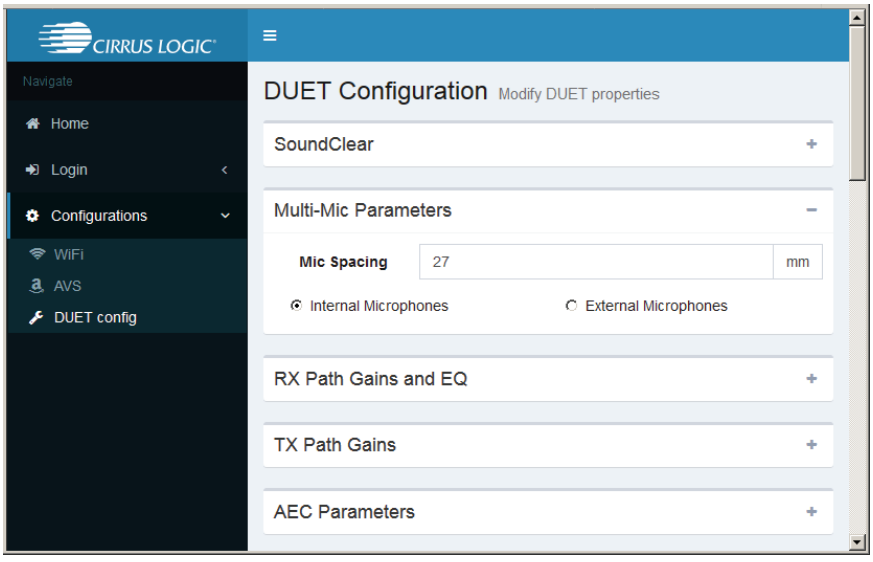

Figure 4 Control Console Configuration Panel Multi-Mic Parameters Box

Select the *Internal Microphones* radio button to use the on-board CS7250B microphones and the *External Microphones* radio button to enable the digital microphone interface available on the DMIC2 IF header J3 on the voice capture board.

3.2 Rx Path — Line In Gain

Set the volume of the user speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

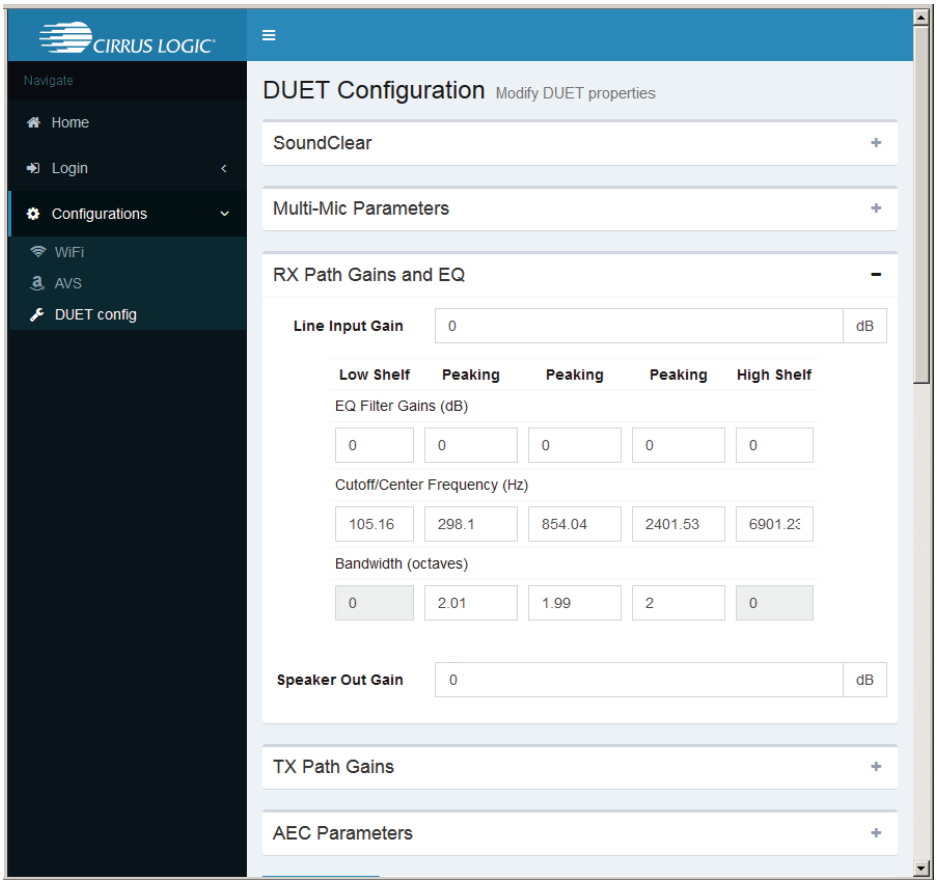

Figure 5 Control Console Configuration Panel Rx Path Parameters Box

The goal is to optimize the line-in path dynamic range without clipping:

- during playback, in the *Rx Path Gains and EQ* box in the Control Console Configuration panel, shown in Figure 5, adjust the *Line Input Gain* until the *Line In* value on the *Audio Levels* diagnostics tab reads ~ -10 dB, as shown in Figure 6.
- during playback, monitor the clip counters to ensure that the Line In signal does not clip – if clipping occurs, as shown in Figure 7, adjust the *Line In* control until the peak signal levels do not exceed ~ -10 dB, and reset the clip counter by clicking on the blue *Line In* button next to the clip counter/indicator
- clip detectors are latched and will remain highlighted in pink until reset

Figure 6 Audio Levels Tab

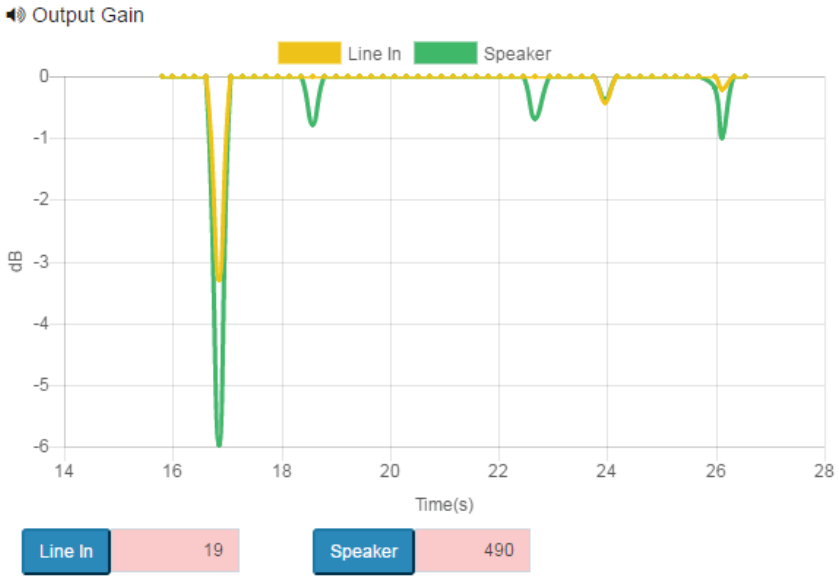

Figure 7 Audio Levels Tab – Output Gain with Clipping

3.3 Rx Path — PEQ

Set the volume of the user speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

The Rx EQ filter response can be viewed by selecting the RX EQ diagnostics tab. Figure 8 shows an example with all the filters enabled.

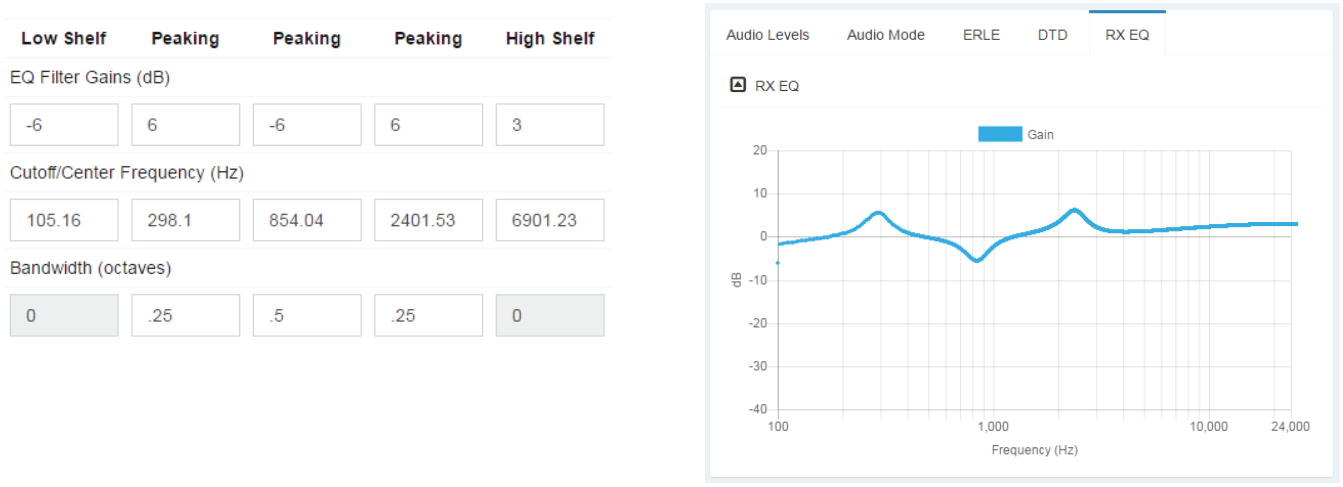

Figure 8 Example Rx Tuning with EQ Tab Display

The goal is to shape the music playback sound according to user preference, by adjusting the filter values in the *RxPath Gains and EQ* box in the Control Console Configuration panel, shown in Figure 8.

Additional benefits of Rx EQ include:

- reduction of distortion at resonant frequencies in transducers or case
- sound-shaping for meeting frequency response masks

3.4 Rx Path — Speaker Out Gain

Speaker Out Gain can be used to change both the analog gain of the amplifier on the voice capture board and the headphone out jack on the voice capture board.

Adjust the Speaker Out Gain to change the output gain to meet the target output volume level without clipping the Speaker Out or Line In signals.

Note that adjusting the Speaker Out Gain will cause the Line In levels to be adjusted – if necessary, readjust the Line In controls as described in section 3.2

3.5 Tx Path — Microphone Gain and Min Speech Threshold

Set the volume of the user speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

Switch to the Audio Levels diagnostics tab to monitor the microphone input levels and verify that the inputs are not clipped during the loudest playback conditions.

The goal is to optimize the microphone input signal level without clipping microphone inputs during Alexa playback:

- verify that the *Mic 1* and *Mic 2* values on the *Audio Levels* diagnostics tab are not being clipped
- during playback, adjust the *Mic Input Gain* until the *Mic In* values on the *Audio Levels* diagnostics tab read ~ -10 dB, as shown in Figure 9

Note that the *Mic Input Gain* controls the gain on both Mic 1 and Mic 2.

For the reference design the *TX Min threshold* does not need adjustment and should be left at -38 dB.

Figure 9 Audio Levels Tab – *Mic In* Values ~ -10 dB

3.6 Tx Path — Line Out Gain

The Line Out Gain control adjusts the output level sent to the wake word engine. The reference design uses a pre-set tuned value of 6 dB.

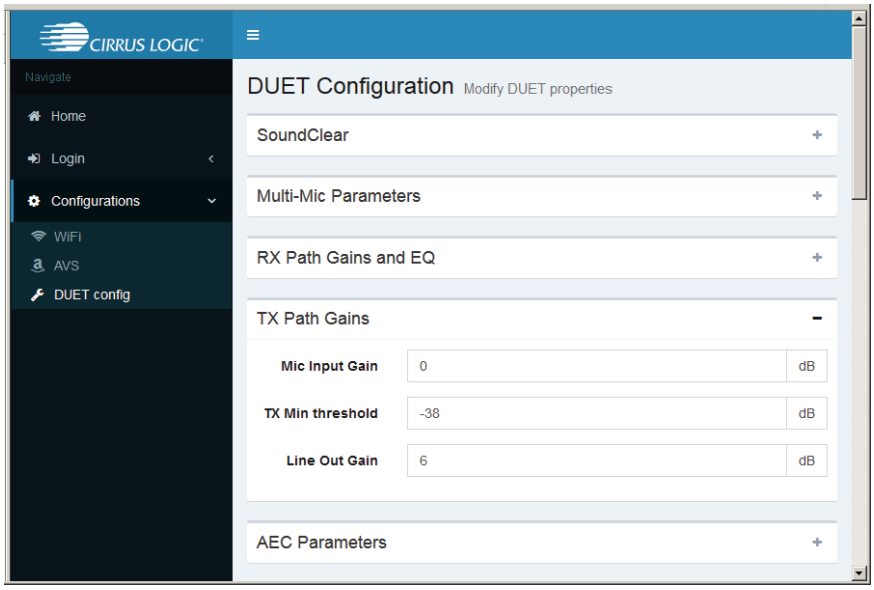

Figure 10 Control Console Configuration Panel Tx Path Parameters Box

3.7 AEC Setup — Setting AEC Bulk Delays

Set the volume of the speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

Systems with a variable distance between the speaker and microphone should always set the bulk delay with transducers positioned at the minimum expected spacing.

The goal is to align the echo signal and reference signal as close as possible for the best AEC performance and response times using the *AEC Reference Delay* and *Mic Bulk Delay* buffers to compensate for system-introduced latencies.

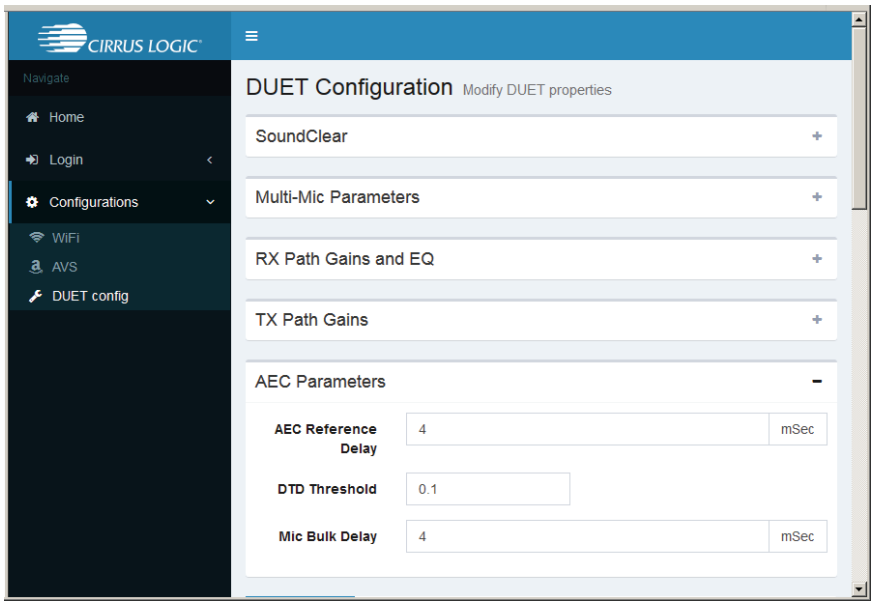

Figure 11 Control Console Configuration Panel AEC Parameters Box

3.7.1 Setting AEC Reference Signal Bulk Delay

An AEC reference signal bulk delay setting that is greater than the actual acoustic delay will cause non-convergence of AEC.

The setting is dependent on the speaker-to-microphone distance; for the reference design, the *AEC Reference Delay* should always be set to 4 mSec.

3.7.2 Setting Mic Bulk Delay

A mic bulk delay setting that is less than the actual reference signal latency will cause non-convergence of AEC.

The setting is dependent on the speaker-to-microphone distance; for the reference design the *Mic Bulk Delay* should always be set to 4 mSec.

3.7.3 Verifying AEC Performance

Set the volume of the speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

The goal is to make sure that the AECs are cancelling echo during playback, so that only the near-end talker’s voice is passed through to the AVS.

Understanding the ERLE Measurement

The output is shown on the *ERLE* tab, in Figure 12. The following points aid understanding of this display.

- during Alexa playback, the AECs cancel echo — typically 25+ dB
- ERLE varies due to differences in the level of the music or speech playback
- during double talk, that is, when Alexa is playing music and the near-end talker speaks, the measured ERLE will drop
- during no Alexa playback, AECs do not adapt
- speaker distortion will lower ERLE performance
- clipped microphones will cause drops in ERLE similar in nature to the way double talk changes ERLE
- bulk delays that are too long will cause the AECs to not converge — 0 ERLE
- bulk delays that are too short will cause slow convergence and lower the ERLE

Figure 12 ERLE Display

3.7.4 DTD Threshold

The DTD Threshold controls the balance of barge-in relative to the amount of echo.

Set the volume of the speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

The goal is to make sure that the near end talker’s voice is passed through to the AVS during playback without causing any echo during Alexa playback.

- set the threshold as low as possible without causing echo in single-talk Rx

3.8 System Verification

Using the *Audio Modes* tab, shown in Figure 13, to verify system performance.

Figure 13 Audio Mode Tab

Set the volume of the speakers to the target volume by commanding Alexa to ‘set volume to 11’.

Play full-scale continuous audio from the far-end by commanding Alexa to speak or play music.

Verify proper system response during all audio modes during ST, RX, and DT (TX & RX) as shown in Figure 13.

4 Definitions and Terminology

AEC	Acoustic Echo Canceller
Acoustic Loss	Echo loss as measured from the speaker to the microphone
dB	Decibels
DT	Double Talk – both Tx and Rx at the same time
DTD	Double Talk Detector
ENR	Ratio of Echo to Near End Speech
ERLE	Echo Return Loss Enhanced
FE	Far End – the side Alexa is on
NE	Near End – the side the human user is on
PEQ	Parametric Equalizer
PGA	Programmable Gain Amplifier
RX	Receive
SNR	Signal to Noise Ratio
ST	Single Talk – only Tx or only Rx at any given time
THD	Total Harmonic Distortion expressed in %
TX	Transmit
VAD	Voice Activity Detector

5 Bibliography

The following documents provide additional reference information:

CRD1569-1 User Guide – describes the Control Console.

6 Revision History

Revision History

Revision	Changes
1.0 APR ‘17	• Initial Tuning Release on WISCE™/eAudition platform.
1.1 JUN ‘17	• Initial release for CRD-1569-1 platform.
1.2 JUL '17	• Updated UI graphics and descriptions.

Contacting Cirrus Logic Support
For all product questions and inquiries, contact a Cirrus Logic Sales Representative.
To find one nearest you, go to www.cirrus.com.