Two Channel Audio Mixer

by

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Senior Project

ELECTRICAL ENGINEERING DEPARTMENT

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Acknowledgements

I would like to thank my family and friends for their support during my time at Cal Poly, especially my parents and grandparents. I would like to thank Dr. Pilkington for his guidance as the project advisor. I would like to thank my dad for helping me build then enclosure for the mixer, his help was invaluable.

Abstract:

This device uses digital signal processing techniques to combine two audio signal inputs into a single audio signal output. The mixer controls the amplitude (volume) of each input signal, as well as the ratio of each signal's contribution to the master output. In addition, the mixer performs simple equalization of each audio signal with amplitude control of the Bass (20Hz-320Hz), Middle (320Hz-5120Hz), and Treble (5120Hz-20kHz) frequency ranges. The mixer can also high- and low-pass filter each input audio channel independently. Finally, the mixer provides the ability to monitor each audio channel without outputting through the master output. With this device, a user can obtain a single, mixed and equalized, audio signal derived from two audio input signals, providing an essential functionality for mixing two songs together, or combining two separate instrumental tracks together to create a more complex musical passage.

Chapter I. Introduction

The Two Channel Audio Mixer is a digitally implemented audio signal mixer intended for DJ mixing purposes. The mixer allows a user to manipulate two audio signals, eventually combining them in whatever proportion desired to form a single output audio signal. The mixer also allows the user to monitor each channel through a separate output channel that does not require simultaneous output through the master channel. These features combined with a 3-band Equalizer and High and Low-Pass Filter for each channel, provide the user with all the basic functionality expected from a DJ mixer. Pioneer's DJM-350 serves as a comparison benchmark for an existing digital two channel mixer. [3] The functionality of the proposed Two Channel Mixer loosely parallels this device. Inspiration for this project came from personal experience making DJ mixes, which lead to the desire for a better understanding of how mixers work, and an interest in designing a fully functional two channel mixer that one could use.

Chapter II. Background

The Two Channel Audio Mixer is a digitally implemented system that is built around a Digital Signal Processor (DSP). DSPs are specialized microprocessors that are designed to solve arithmetic and logical arguments very quickly. This enables them to compute and manipulate complex signals such as audio and video signals in a manner that makes them accessible in real-time. For the scope of this project the DSP will be processing audio signals. DSPs are especially good at digitally implementing filters, which require many multiplication and addition computations. For this project the DSP will be processing volume control, filtering and equalization, signal mixing and routing, and processing external control inputs.

A DSP system takes an analog signal and converts it to a digital signal that can be processed and manipulated. It then converts that digital signal back to an analog signal that is output. Analog to Digital Converters are used to transform an analog input signal to a usable digital representation for the DSP. ADCs work by sampling the analog signals using a sampling frequency that is twice as large as or larger than the frequency of the input analog signal, in keeping with the Nyquist Sampling Theorem. The signal is sampled, held, and quantized and encoded and then sent to the DSP for processing. Likewise when the DSP outputs a signal it is sent to a Digital to Analog Converter (DAC). This works in contrast to the ADC, creating an analog signal. The DAC works by decoding the digital signal, holding it, and finally filtering the signal. This is the basic signal flow for Digital Signal Processors.

With respect towards this project, using a Digital Signal Processor allows for real time manipulation of the input audio signals, which is necessary for properly implementing an audio mixer that can be used as a DJ platform. Also, since the DSP is by definition a digital platform that is programmed in software, it allows for easier changes to be made when implementing the project if problems arise, and quicker design fixes. If the project was implemented with analog hardware and a subsystem needed to be modified or changed it would require circuit redesign and time to swap the physical parts in and out of the system. This is another example of the practicality of using a DSP for implementing the Two Channel Audio Mixer. For more information on Digital Signal Processors and Digital Processing in general, consult page 465 of Ambardar’s *Analog and Digital Signal Processing* [6], as well as Analog Devices *A Beginners Guide to Digital Signal Processing* [7].

Chapter III. Requirements and Specifications

<table>
<tr><th colspan="2">2 Channel Audio Mixer</th></tr>
<tr><td>Dylan Kinney
EE 463-01</td><td>Dr. Wayne Pilkington
1. I agree to supervise this senior project. ______
2. The specifications are [1]-[2]:
☐ Abstract—Describes what project should do, not how.
☐ Bounded—Identify project boundaries, scope, and context
☐ Complete—Include all the requirements identified by the customer, as well as those needed to define the project.
☐ Unambiguous—Concisely state one clear meaning.
☐ Verifiable—A test can prove if system meets specification.
☐ Traceable—Each engineering specification serves at least one marketing requirement.</td></tr>
<tr><td colspan="2">ADVISORS: Please initial above, if you agree to supervise this senior project. Also, please check applicable boxes above. Comment below, if requirements or specifications require revision.</td></tr>
</table>

TABLE I

TWO CHANNEL AUDIO MIXER REQUIREMENTS AND SPECIFICATIONS

Marketing Requirements	Engineering Specifications	Justification
2	1. The device mixes two audio channels into one master output channel.	In keeping with expected operation of an Audio Mixer.
2,3	2. Each audio channel routes to a monitor channel.	A customer must preview incoming signals without outputting through the master channel.
1, 4	3. Total Harmonic distortion of audio signal ≤ 0.1%.	To preserve audio signal fidelity.
1,	4. The mixer has a frequency range of 20 Hz – 20kHz.	The mixer plays all audio frequencies humans can hear.
1,2,4	5. The mixer uses RCA connections for input channels and output channel.	Industry standard and makes setup easy for customers. [3]
1, 2, 3,4	6. The mixer uses a ¼" PHONE connection for the monitor channel.	Ease of use and familiarity for customer. [3]
2, 4	7. Desired power consumption of 20 W or less.	Power consumption similar to comparable 2-channel mixers on the market. [3]
2, 3, 5	8. Approximate dimensions of the mixer: 8.5” (W) x 11.9” (L) x 4.2”(H)	The mixer needs a similar size to current 2-channel mixers. [3]
3,4	9. Finished wood comprises the mixer’s external case providing desired style and appeal.	Fulfilling aesthetic expectations of the mixer.
1, 6	10. Implementation via EVAL-ADAU1446EBZ Evaluation Board, Digital	Digital implementation, platform must process 16-bit or 24-bit audio and handle required

	Signal Processor.	amount of inputs and outputs.
1, 3	11. Audio Inputs and Outputs on board require 1/8” stereo connection.	Constraint of the board, Audio Signals are input and output through 1/8” connections.
2	12. 3-Band Equalizers, High and Low Pass Filters, Master and Monitor Volume, controlled via knob potentiometers. Channel Volume controlled by slide potentiometers. Crossfader controlled by slide potentiometer. Monitor Channel Select controlled by push button(s).	This control surface is similar to regularly used DJ mixers [3], and provides ease of use for those using this mixer for the first time.
7	13. The mixer design requires NEC compliance	For public safety assurances. [4]
Marketing Requirements 1. The mixer should have quality sound fidelity. 2. The mixer should have easy operation for those familiar with DJ hardware. 3. The mixer should isolate audio signals in a monitor channel. 4. The mixer should cost less than comparable mixers on the market. 5. The mixer should aesthetically please. 6. Digital implementation of the mixer. 7. Ensure device does not harm the public.		

The Two Channel Audio Mixer has ten Engineering Specifications and six Marketing Requirements, seen above in Table 1. These Requirements and Specifications ensure the project will conclude with the desired functionality that the user of a consumer market (non-studio) DJ audio mixer would expect.

Marketing Requirement number one mandates the mixer remain faithful to the input audio signals, outputting a quality, minimally distorted signal. The second requirement ensures easy operation of the mixer by those with audio mixing experience, primarily DJ mixing. This allows minimal time learning the operation of the mixer and more time using the mixer as intended. The third marketing requirement recognizes an essential piece of functionality that all mixers must have, the ability to monitor input channels. Monitoring is a vital function of a mixer, allowing a DJ to set up transitions and listen to a channel without the audience or crowd hearing that channel. Thus, this functionality must be included in the mixer. The fourth marketing requirement allows for possible future business ventures. This requirement provides a competitive advantage if the mixer were ever to be sold commercially. The Pioneer DJM-350 [3] is the reference for a two channel mixer currently on the market. Aesthetics, while not directly related to functionality, is very important when making a product and is a point of pride for the mixer owner. This is the reason for marketing requirement number five. Requirement six stipulates *digital implementation of the mixer.* This allows for easier future modifications and more functionality. Also, using the DJM-350 as a baseline requires a digital implementation of the Two Channel Mixer. [3] The last requirement ensures the device does not violate safety codes. [4]

The Engineering Specifications are shaped around the marketing requirements to ensure that the desired device is created. The justifications for these specifications can be seen in Table 1 to the right of the Specifications. Mixing functionality and channel monitoring ensure the mixer is able to complete the most basic requirements of an audio mixer. Minimal distortion of the audio signals being routed through the mixer also ensures that the sound quality is high, an expectation of any user. A 20 Hz – 20 kHz frequency range maintains sound quality and audio fidelity by ensuring all audio input signals will be fully output from the mixer. This frequency range corresponds to that of human hearing. Specifications five and six detail which type of connections are to be used for the input and output channels as well as the monitor channel. RCA and 1/4" PHONE connections are industry standards [3]. The mixer is to have a power consumption of 20 Watts or less, which is comparable to similar mixers [3]. The physical dimensions of the mixer must be comparable to other two channel mixers for easy installation in a typical DJ booth. In keeping with aesthetic expectations of this market, the body of the mixer will be finished wood or metal.

Detailed Design Review Additions:

Wood was decided on for the case of the mixer. This was due to easier fabrication of the case when using wood as opposed to using a metal case. Machinery required to fabricate a metal case would not have been as easily accessed as tools needed to create a case from wood. A finished wood case would also help satisfy the aesthetic requirements of the design, resulting in a retro, tasteful appearance.

Engineering Requirement ten changed from a general *Digital Implementation via microcontroller or DSP* constraint to having a defined DSP and Evaluation Board. Analog Device's ADAU1446 SigmaDSP Digital Audio Processor was selected. This chip will be on the EVAL-ADAU1446EBZ Evaluation Board. This chip and accompanying evaluation board were selected because they are designed to process stereo audio signals. The high precision chip helps against distortion, and also has a sampling rate that will not alias the audio signal. The evaluation board also has enough GPIO pins and other inputs to implement the control interface for the mixer. This will be covered in more detail in the Design Alternatives section.

Engineering Requirement eleven was added because the Analog Devices evaluation board selected has 1/8" stereo audio inputs and outputs. This means the RCA inputs and outputs on the case must be adapted to a 1/8" audio line in order to interface with the board, and keep with the design constraints.

Engineering Requirement twelve specifies the user control surface. Each band of the 3-band Equalizer (for each channel) will be controlled by a knob potentiometer. The filters for each channel will also be knob potentiometers. The volume control for each channel will be vertically positioned slider potentiometers. The crossfader control will be a horizontally positioned potentiometer. The channel select for the monitor output will be controlled by a push button. The monitor output volume control and master output volume control will also be knob potentiometers. This control surface can be seen below in Figure 2.

The requirements and specifications table format derives from [1], Chapter 3.

Chapter IV – Design Alternatives

When considering what platform to use when digitally implementing the mixer, the decision was between using a microcontroller or a digital signal processor. The two largest constraints affecting the alternatives was the ability to process audio signals with minimal distortion and loss, and ability to accept the inputs required of the control surface. With the microcontroller approach interfacing with user controls such as potentiometers for the EQ and Filter control would simpler then doing so with a DSP, as they are designed to handle a very large range of peripherals. Likewise, the DSP alternative would be much more efficient at processing the audio signals, as this is what it is designed to do. Since the mixer requires not just one but two audio signals to be simultaneously processed, this all but eliminated the microcontroller alternative, as many microcontrollers would not be able to do this.

Even though a DSP could process two audio signals it would need to be able to interface with the control surface in order to be viable. Many DSPs are designed now to act more like microcontrollers, meaning they can handle peripherals and be programmed for a wide range of applications. The dsPIC and TI DSPs controllers were the first I came across while researching potential DSP controller platforms. With the dsPIC's I could not find an accompanying development or evaluation board that would handle the constraints of the project. TI had one controller that came closest to fulfilling the project needs. TI's TMS320C5515 DSP Evaluation Module met the minimum processing constraint in that it could process 16 bit audio. The board also had two stereo inputs and one stereo output. In addition the board had a headphone output. This could have allowed for the desired audio input and outputs for the mixer. However it was unclear if the board would be able to handle the required user control inputs, or if would have to interfaced with another microcontroller to accomplish this.

Analog Device's SigmaDSP Audio Processors were the last alternative I found. The ADAU1446 was much more oriented toward processing audio then the TI board mentioned above. The core allowed for 28-bit processing with 56-bit processing in double precision mode. This meant the DSP met the design requirements of handling 16-bit or 24-bit audio sources, without degrading the signal. The SigmaDSP core runs at 172 MHz, capable of running 3584 instructions per sample with a sample rate of 48 kHz. The 48 kHz sample rate ensures the audio signal will not be aliased. The evaluation board also has four stereo inputs and 8 stereo outputs. This more than satisfies the design constraints of needing two audio input channels and two audio output channels. The board also has 24 GPIO (General Purpose Input Output) connections that can be used to interface the user controls with the DSP. The evaluation board also comes with an external GPIO board with potentiometers, buttons, switches, a rotary encoder, and LEDs that could all be used for testing prototypes of mixer functionality before building the entire device. The ADAU1446 is also programmed via Sigma Studios, a graphical programming software. This was also preferred over the TI DSP, which was programmed by C/C++, as I do not have a lot of experience programming in C. The graphical block programming makes it much easier to visualize the audio signal flow. Sigma Studios also has a large variety of audio processing components such as programmable filters, simplifying the implementation process of the design. For these benefits the Analog Device's ADAU1446 DSP and evaluation board was the chosen design alternative.

Chapter V – Project Design

Level 0 Block Diagram:

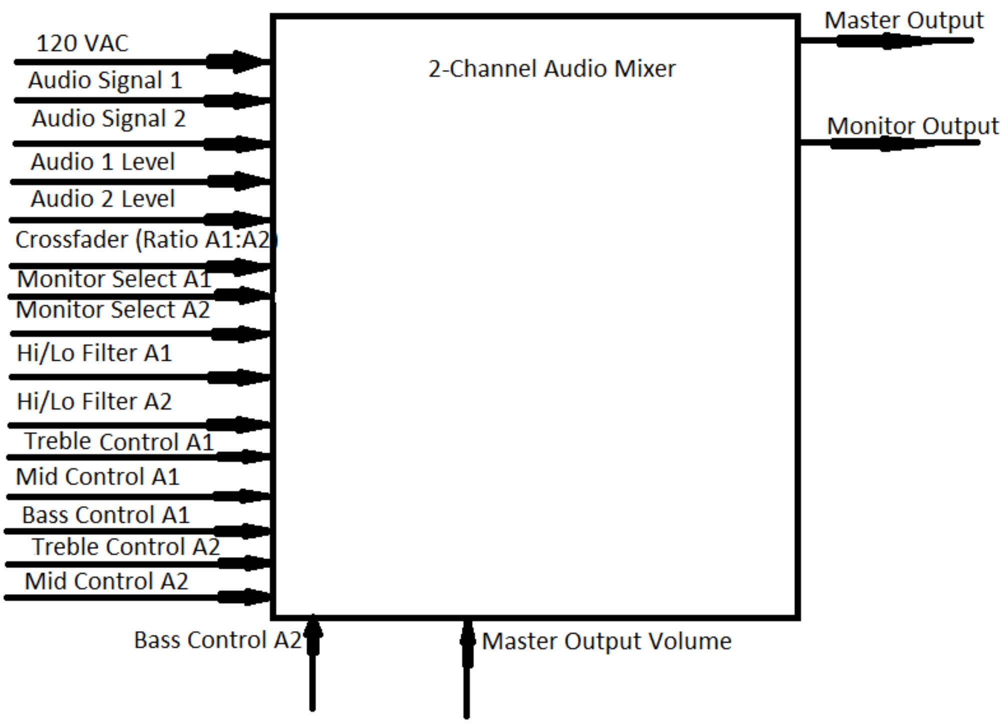

Figure 1: Two-Channel Mixer Block Diagram

Module	Two-Channel Audio Mixer
Inputs	-120 V AC, 60Hz. -Audio Signal 1, RCA connection. -Audio Signal 2, RCA connection. -Audio 1 Level: vertical slider, variable control. -Audio 2 Level: vertical slider, variable control. -Crossfader (Ratio of A1 and A2): horizontal slider, variable control. -Monitor Select A1: push button. -Monitor Select A2: push button. -Hi/Lo Filter A1: variable control. -Hi/Lo Filter A2: variable control. -Treble Control A1: variable control. -Mid Control A1: variable control. -Bass Control A1: variable control. -Treble Control A2: variable control. -Mid Control A2: variable control. -Bass Control A2: variable control. -Master Output Volume: variable control.
Outputs	-Master Output Audio Signal: RCA connection. -Monitor Output Audio Signal: 1/4" PHONE connection.
Functionality	Takes two audio signal inputs, filters and combines the signals, and creates a master audio signal output and monitor output audio signal.

Table 2: Block Diagram I/O Key

Figure 1 and Table 2 represent the Level 0 design of the mixer. The mixer requires a 120 V AC input at 60 Hz for power, since this device in intended primarily for use in North America. Two audio signals are input to the mixer and are represented by Audio Signal 1 and 2. These are RCA connections. For each of these audio signals, there is a vertical slider volume level control, represented by Audio 1(2) Level. A crossfader input controls the ratio of Audio Signal 1 and 2 that is output by the master channel. This input would be a horizontal slider, with a 1:1 ratio in the middle position. Monitor Select A1 and A2 are push buttons that select the desired Audio Signal to be output through the monitor output audio signal. Hi/Lo Filter A1 and A2 are knobs that control how much each channel is high or low pass filtered. If the knob is in the middle position, no filtering is applied to the channel. Treble Control A1 and A2, Mid Control A1 and A2, and Bass Control A1 and A2 control their respective channels 3-band EQ, boosting or attenuating preset frequency ranges for each channel. The Master Output Volume controls the magnitude of the master output signal. The functionality of the Level 0 Block is based in part of the DJM -350 mixer. [3]

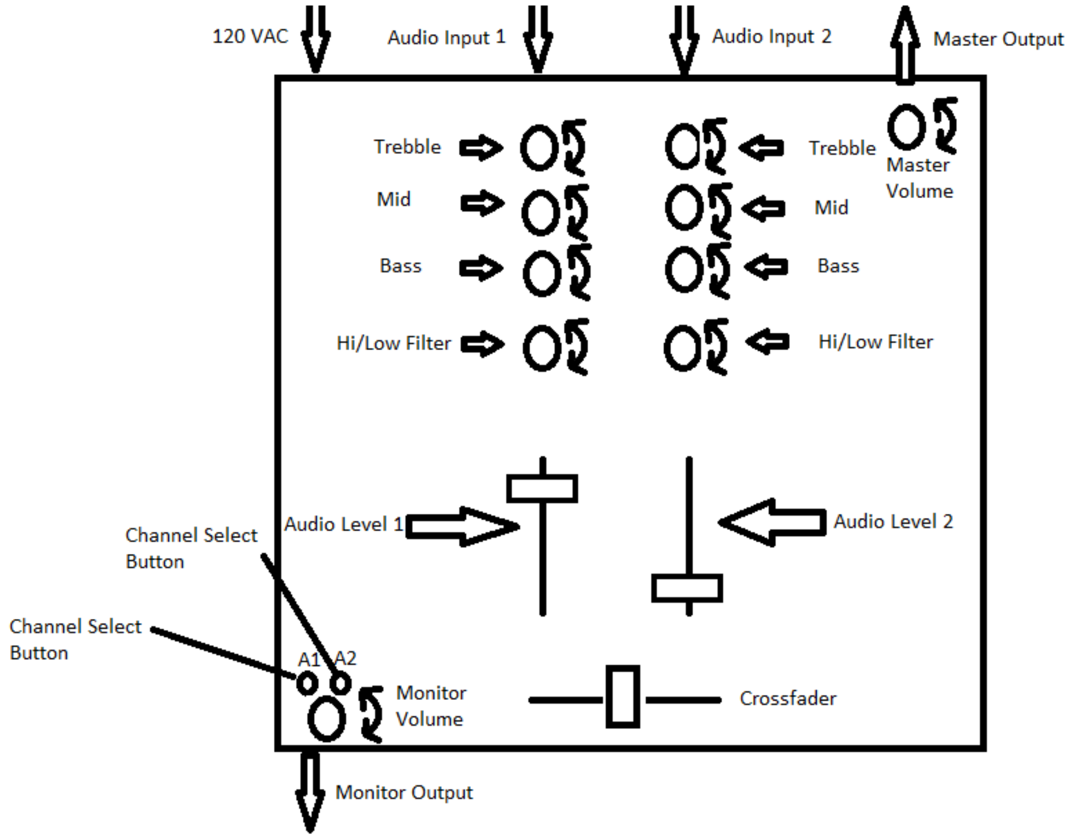

Figure 2: Two Channel Mixer Interface

Figure 2 above shows the desired layout of the user interface for the Two Channel Mixer. Also present in the figure is the approximate locations of external power and audio signal inputs and outputs to the mixer. With respect to user interface, each channel has duplicate controls (i.e. 3-band EQ, Hi/Low Filter, Volume control), while the crossfader affects both channels. The for each of the band control knobs on the EQ (Treble, Mid, Bass), the user will attenuate the specific frequency ranges by turning the knob to the left, and boost the range by turning the knob to the right. The direction is same for both channels. A knob also controls each channels high and low pass filters. When the knob is positioned in the center, no filtering is applied, when the control is turned to the left, the low pas filter will activate and the cutoff frequency will continue to decrease as the knob is turned farther left. Likewise when the knob is turned to the right the high pass filter will activate, and increase the cutoff frequency of the filter as the knob is turned farther right. Audio Level's 1 & 2 are controlled by sliders (or faders), that when at their zero position (at the bottom), will fully attenuate the respective audio signal, allowing no audio to pass. When the slider is at its maximum position (at the top), the audio signal will be maximally gain boosted. The crossfader control essentially applies a ratio control to both audio signals when they are mixed together. When in the middle, the crossfader applies a 1:1 ratio to both signals, passing both through. When the slider is all the way to the left or all the way to the right the left or right channel alone will be passed, respectively. The crossfader will follow a linear curve when suppressing the desired channel, from middle point to extreme left or right point. The

Monitor Output Channel select will be two push buttons the simply select which audio channel is output through the Monitor Output. This design will not be a concrete restraint as a single button or even a switch could be used if Audio Channel 1 is the default channel outputted to the Monitor Output, and Audio Channel 2 is selected once the push button or switch is closed. This constraint is left free in the case simplifications to design are required. The Monitor Volume and Master Volume controls would also be knob controlled, with similar attenuation and gain boost control as the EQ knobs.

Level 1 Block Diagram:

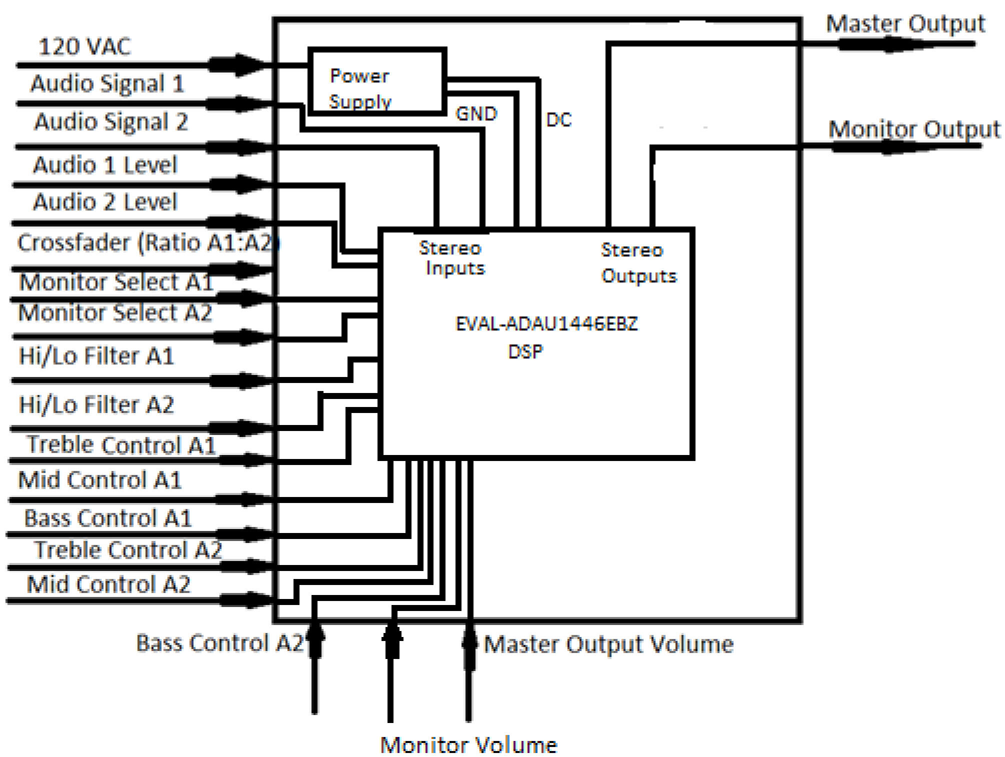

Figure 3: 2 Channel Mixer Level 1 Block Diagram

The Level 1 Block Diagram includes the Microcontroller and Power Supply subsystems. The power supply is provided with the Evaluation board which has a 120 VAC input, and outputs the required DC voltage and current levels needed to supply the internal electronics of the board, as well as the Ground return. The internal DSP Evaluation Board is powered by the

DC output from the Power Supply. In addition, the DSP Evaluation Board receives all the mixer control-input signals detailed in the Level 0 Block. DSP Evaluation Board subsystem outputs the Master and Monitor Output signals. Previously expected subsystems of current amplifiers for the Master and Monitor output channels are no longer needed as the evaluation board is designed to output stereo audio, which means current amplifiers should not be needed to drive the signal.

All of the user interface controls will be able to receive a DC voltage of 3.3 V from the GPIO expansion board. Ground will also be supplied to these controls as well. The potentiometers and another analog control inputs will interface with the GPIO pins of the board, and then into the auxiliary ADC of the board. If the ADC does not have sufficient channels for all inputs, external ADCs will be obtained from Analog Devices to accommodate the remaining control inputs, which will be converted and connected to the digital inputs of the board. This will be determined after prototype build and testing is complete in the upcoming months.

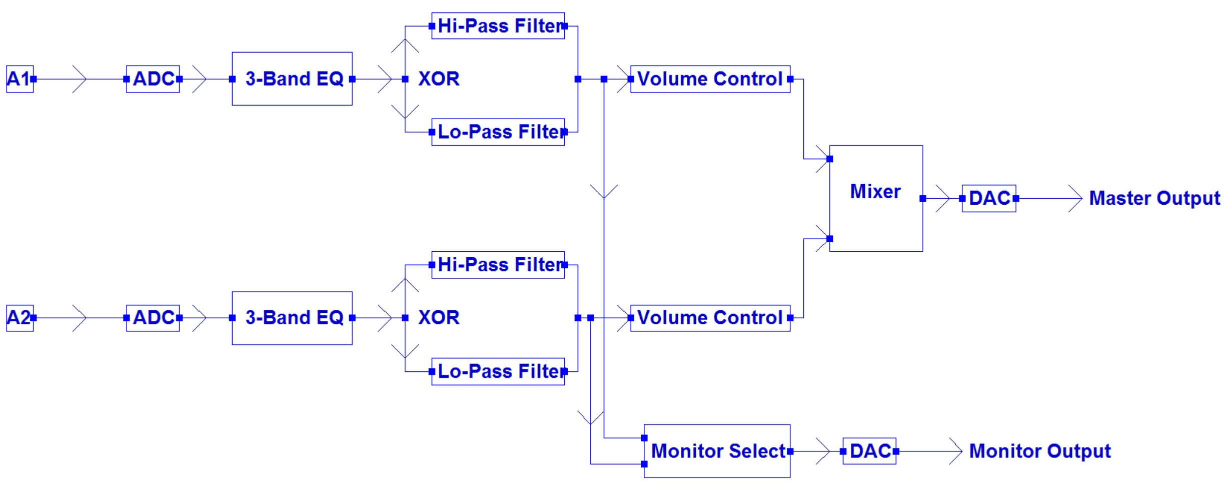

Figure 4: Audio Signal Flow of Mixer

Figure 4 above shows the designed audio signal flow of each channel inside the DSP. Each signal will enter the system through the stereo inputs on the board, where they will be subsequently converted to digital signals after passing through ADCs. Once digitized, each signal will flow into its respective 3-Band EQ for frequency specific attenuation or boost. After this, the signal will flow through either the high pass or low pass filter for its channel. As stated in the control surface description, this will be based on which direction the filter knob is turned. If the filter knob is in its center position, it will simply be passed through unfiltered. Next both

signals will branch, with one route going to the volume control and the other going to the monitor output control. Depending on the push button control, either Channel 1 or Channel 2 will be output, passing through and onboard DAC and output through the stereo output. The signal also flows through the volume control for each channel, where it is attenuated or boosted from 20 Hz to 20 kHz based on the user control. Both channel signals then input into the mixer block, where the crossfader effects are applied (a ratio is applied to the signals), and the two signals are combined. After this the single master signal is processed through a DAC and output from one of the stereo outputs.

The 3-Band EQ parameters are designed to apply attenuation or boost at 60 Hz, 1.3 kHz, and 10 kHz. These parameters will be applied to one of the 2nd order double precision EQ blocks in Sigma Studios. The frequencies are based off of frequency spectrum testing of white noise passed through Pioneer's DDJ-ERGO DJ controller. This provided an example of a successful on the market product to model after.

The high and low pas s filters will 2nd order filters based on provided Butterworth models in Sigma Studios. This is not a design constraint however and the filter type can be changed in order to best complete the mixer. Butterworth filters were decided on due to their clean passband that does not distort the desired signal with ripples. The low pass filter, when activated, will start at 20 kHz and incrementally have its cutoff frequency reduced down to 20 Hz passed of position of the knob potentiometer. Likewise the high pass filter will start at 20 Hz when activated and have its cutoff frequency incrementally increased based on position of the knob potentiometer.

Final Design and Implementation

The signal flow and audio processing was designed and implemented in Analog Devices SigmaStudio Graphical Design Tool Software. The design is essentially split into three stages and will be described as such to make description and comprehension easier. These stages include the Equalizer (EQ) stage, Filter stage, and Volume Control stage. Each of these stages can be bypassed in real-time via SigmaStudio, which can run in time with the ADAU1446 via a USB connected to the board in an I^2C command interface. This allows for each stage to be easily tested and will allow for external control of the Equalizer for each channel, which will be detailed later. These stages interface with external controls which route through the boards Multipurpose pins, and enter the Auxiliary ADC if they are analog signals or the General Purpose Input Output as digital signals. The Monitor Output Channel, or Cue Channel, routes from the node between the Filter and Volume Control stages. To simplify description, the audio signal input stage will be detailed with the EQ stage, and the output stage will be detailed with the Volume Control stage.

Starting with the Input and EQ Stage, which can be seen below in Figure 5, The stereo inputs that represent Channels 1 and 2 can be seen in the Input1 block. In that block Channel 1 is represented by 0 and 1, for the left and right channels respectively. Each channel is routed into t-connector blocks, which allow the signals to be routed to multiple inputs. Both channels are then

routed to their respective EQ blocks, which can be seen with the labeling of Crossover#, with # representing the specific channel number. In the Detailed Design phase of the project, the EQ Stage was going to be built around give equalizer blocks in SigmaStudio which could have each passband specified, and be controlled from within the SigmaStudio software via a volume slider. However these blocks did not allow for external control of the gain boost and attenuation of each passband, so it would not work as a part of the project, as the EQ needs to be controlled externally by a potentiometer. After testing of different methods of implementing the EQ with external control, it was determined that using a Crossover filter block with external volume control blocks and then mixing these signals would work as an EQ for the project. This works by using the Crossover filter to create three separate frequency bands. These bands constitute a low frequency band, a midrange frequency band, and a high frequency band. The low frequency band is low passed and has a corner frequency of 320 Hz. The midrange band is bandpassed and has corner frequencies of 320 Hz and 5120 Hz. The high frequency band is high passed and has a corner frequency of 5120 Hz. These bands are created in the Crossover block using a Linkwitz-Riley filter at 24 dB/octave. Generally this is done by combining two 2^{nd} Order Butterworth filters. The benefit of using a Linkwitz-Riley crossover is that there is a flat frequency response at the crossover frequency. The Crossover also provides a 10 dB boost to each band so that the user can add gain and attenuation to each band as desired. After each band is filtered the left and right channel are individually output and routed into a stereo volume block that can be externally. For Channel 1’s EQ volume control the Low band receives and external input from Auxiliary ADC Input port 2. The mid band is controlled by Auxiliary ADC Input port 3 and the High band is controlled by Auxiliary ADC Input port 0. Channel 2’s EQ volume controls receive the same Auxiliary ADC Input ports for the Low and Mid bands as Channel 1. The High band volume control for Channel 2 receives its input from Auxiliary ADC Input port 1. Each band volume control output is then routed into a 3-Channel Stereo Mixer block to be added back into one stereo signal. The output of this mixer block is sent to a switch block that can be controlled in real-time via SigmaStudio. This allows for the EQ stage to be bypassed if desired. The switch output is the beginning of the Filter Stage.

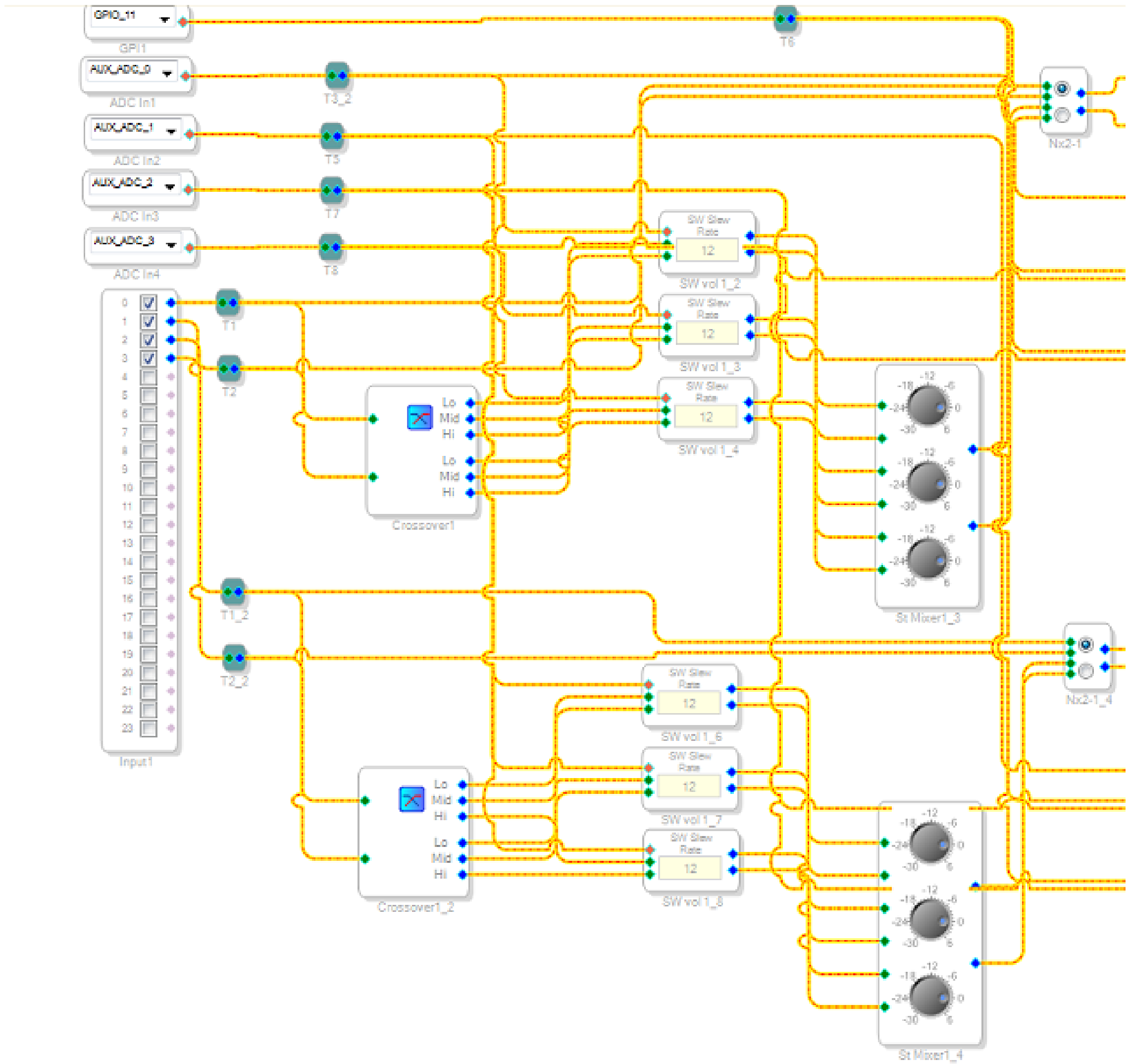

Figure 5: Input and EQ Stage in SigmaStudio

The Filter Stage of the project for each Channel starts at the output of the EQ switch. The stereo signal is split into its left and right components which are routed into State Variable Filter blocks. This will allow for real-time external control of both a high pass and low pass filter. The State Variable Filter block allows for variable control over the filters cutoff frequency and Q factor. For this project we are only concerned with variable control over the cutoff frequency for a low pass and high pass filter, so the Q-factor control inputs for each filter block are dummy controlled by GPIO_11, which receives no external signal. The filter block outputs a lowpass, higpass, and bandpass signal. The bandpass output is not used for this project. The lowpass filter is controlled by the transfer function $H(z)_{LP} = \frac{f(H(z)BP)}{(1-z^{-1})}$. Where $H(z)_{BP} = \frac{f(1-z^{-1})}{1+z^{-1}(f^2-2+qf)+z^{-2}(1-qf)}$. The highpass filter is controlled by the transfer function $H(z)_{HP} = \frac{(1-z^{-1})}{f}$. These transfer functions were obtained in SigmaStudio in the Algorithm Information, Content, section. One state variable filter block is used for each left and right channel, which are

then input into a Stereo Multiplexer block. This Mux block is able to receive an external signal, which toggles between the lowpass and highpass filter when activated. This is done using a button which is input into a multipurpose pin and routed through the GPIO. A toggle block is used which sends a digital signal to the Mux block when a rising edge input is recognized from the GPIO input. Channel 1 receives its toggle signal from GPIO_10 and Channel 2 receives its toggle signal from GPIO_9. The output of the multiplexer block (which is the left and right audio signals) is then routed to another switch. This switch, like the EQ switch, allows for the Filter Stage to be bypassed in SigmaStudio if desired. The output of this switch is the beginning of the Volume Control Stage. The Filter Stage can be seen below in Figure 6.

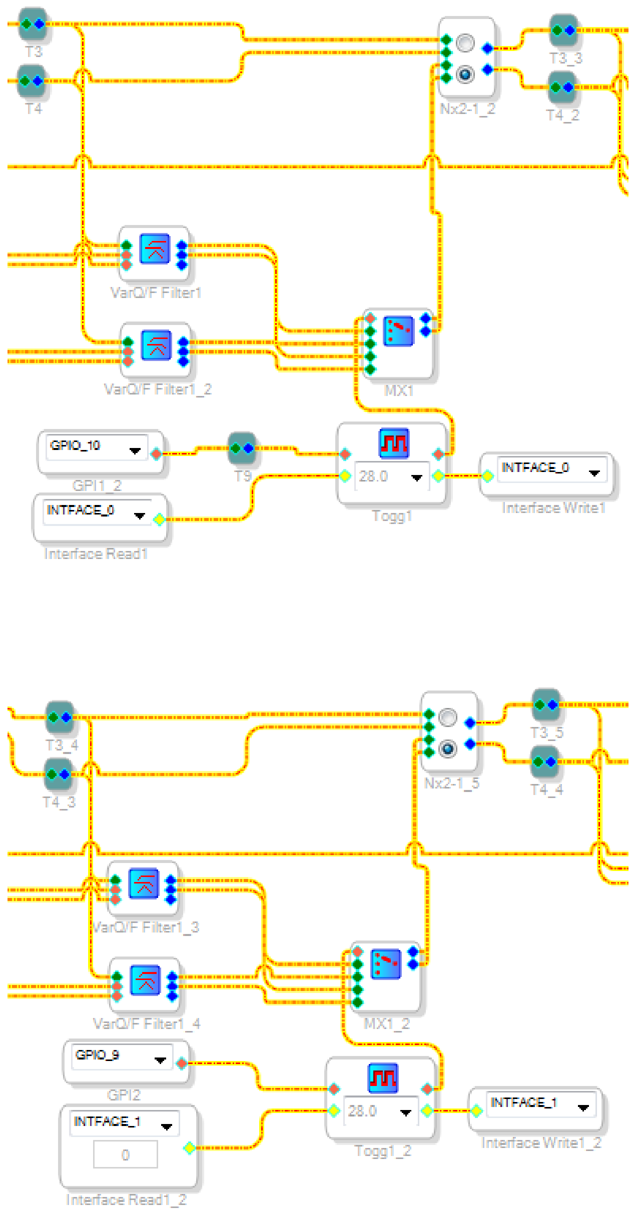

Figure 6: Filter Stage in SigmaStudio

The Volume Control Stage also includes the final mixing and output stage of the project, as well as the Monitor Output (Cue Channel) Stage. The output of the Filter Stage inputs into two T-connector blocks (for the left and right signal) which route to The Volume Control block for each channel as well as the Monitor Output Multiplexer block. The Volume Control section will be discussed first. The Volume Control Stage works the same way as the volume control did for the frequency bands output from Crossover block in the EQ stage. The external volume control block receives and external input that controls the volume of the channel. In Channel 1's case the external input comes from Auxiliary ADC Input 0 which is connected to Multipurpose pin 0 on the ADAU1446. Channel 2 receives its control input from Auxiliary ADC Input 1, which is connected to multipurpose pin 1. The output of the external volume control block is then routed to a switch in SigmaStudio that can bypass the channel if needed. The output of this switch for each channel is then input into a two channel stereo mixing block, which adds Channel 1 and 2 signals together and outputs a left and right audio signal. This stereo signal is the Master Output signal. The Master signal is routed to a gain block which multiplies the signal magnitude by 1.12 dB. This value was determined during magnitude matching testing on an oscilloscope and will be explained in more detail in the Chapter VII of the report. The output of the gain block is then routed to Output 1 and Output 2, which correspond to the mini-stereo output 0/1 on the evaluation board. The Monitor Output Channel works in similar fashion as the Filter Select in the Filter Stage. The Multiplexer receives Channel 1 and Channel 2 from the switch at the output of the Filter Stage. The Mux also receives a control input from a toggle block which activates on a rising edge. This rising edge is caused by a button that is connected to multipurpose pin 8, which routes to GPIO_8. This allows the user to control which Channel (1 or 2) they want to listen to via the Monitor Output. The Multiplexer block output routes to a gain block, which applies a 5 dB gain to the Monitor Channel. This is done to provide more drive to headphones plugged into the ¼" jack. This gain block is then routed to Output 2 and 3, which corresponds to mini-stereo output 2/3 on the evaluation board. The SigmaStudio representation of the Volume Control Stage can be seen below in Figure 7.

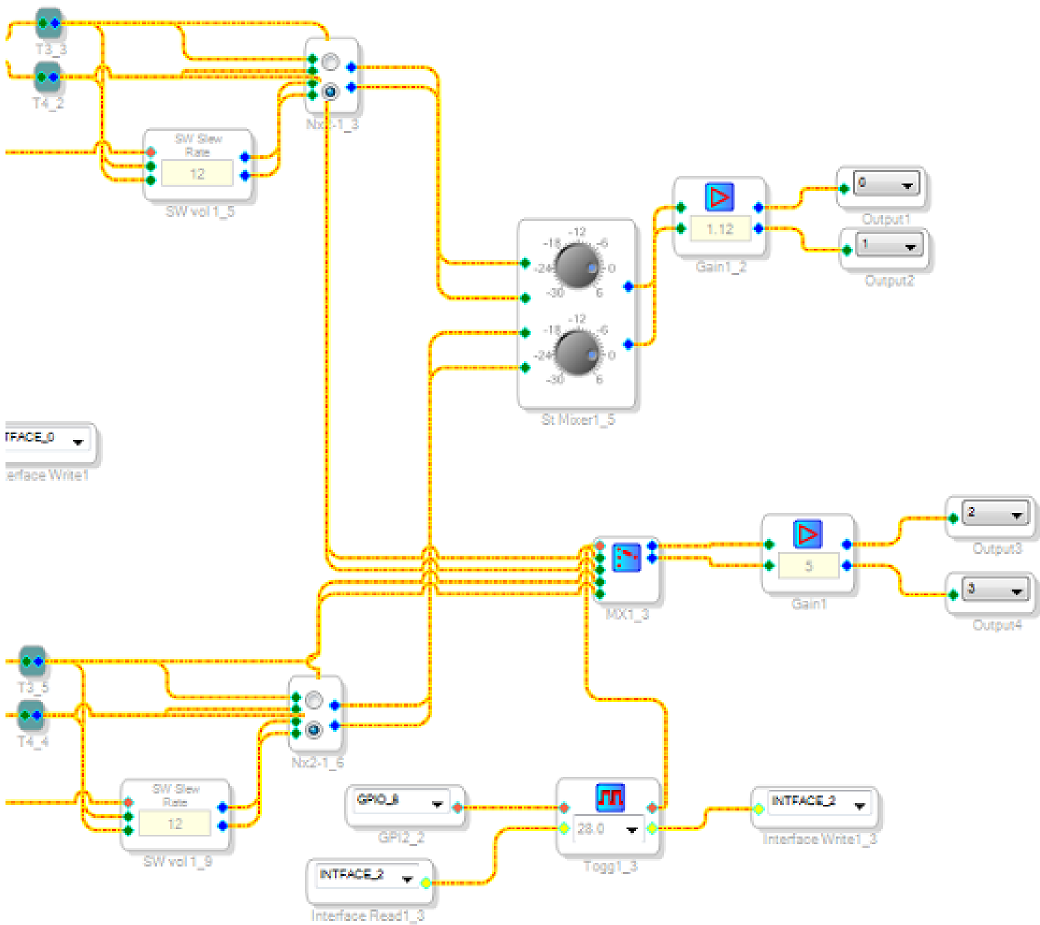

Figure 7: Volume Control Stage in SigmaStudio

As discussed in the Detail Design section, the external interface of the mixer is controlled by potentiometers and push buttons. The volume control for each channel is a 10 kΩ Silde Potentiometer [9] with a linear taper. The filters for each channel are controlled by 10 kΩ Rotary Potentiometer [10], which is also linear. The filter select for each channel are controlled by push buttons. The Monitor (Cue) Channel Select is also controlled by a push button. The Low, Mid, and High bands of each Channel's Equalizer were designed to be controlled by their own rotary potentiometer (same potentiometer as the filter) but unfortunately this was not possible. The ADAU1446 only has a 4 Channel Auxiliary ADC which allows for only four analog control signals to be input to the first four multipurpose pins. It was intended that this would be fixed by using external ADCs that used an SPI interface to communicate with the evaluation board. However the Serial Input and Output pins of the evaluation board do not interface with SPI so this was not possible, as they are designed for the more advanced TDM (Time Domain Multiplexing) interface which is primarily intended for audio purposes. To keep the functionality of the mixer intact a work around was created that would require the use of the real-time switching function in SigmaStudio. Each Channels EQ would use the potentiometers used to control the filters and the volume. This would require bypassing the filter and volume stages in

SigmaStudio in the channel where the equalizing was desired. This translates to Channel 1's Filter Control being the Low band Control, Channel 2's Filter Control being the Mid band Control, and Channel 1's Volume Control being the High band Control. This is the same for Channel 2 with the exception that Channel 2's High band Control is controlled by the Channel 2 Volume Control. While not perfect, this provides a viable workaround that allows for real-time equalizing of a desired channel. This also affects other controls that were originally intended to be implemented. To best keep the functionality of the mixer intact the most critical controls were kept, which are the EQ, Filters, and Volume Controls. The Master Output Volume control was not implemented, as this is not necessary as the Monitor Output is output to an active speaker system with its own volume controls. If this control was absolutely needed, it could be implemented in SigmaStudio. To minimize inputs the two Monitor Channel Select buttons were put into one single control. The functionality is exactly the same and it is actually easier to use the control now. The Monitor Output Volume Control was also not implemented as it was not possible with the input problem and could still be implemented digitally in SigmaStudio. The final control that was not included in the final design was that of the Crossfader. This decision was made independently of the input problem and was a style change in the design. To elaborate, many DJs (depending on style of mixing) do not actually use a Crossfader and functionality of the Crossfader was not seen as critical to the overall functionality of the system. For this reason it is not included in the final design.

Chapter VI. – Physical Construction and Integration

The physical body of the mixer will be comprised of plywood screwed together to form dimensions listed under Engineering Requirement #8, in Table 1. Holes and notches will be cut into the top piece of wood so that the potentiometers for the user interface can fit through them. The positions of the knobs and slider seen in Figure 2 provide a visual of where these cuts will be made. Additionally, one hole will be cut in the vertical sidewall below the mixer controls for a ¼" PHONE connection. Likewise, at the opposite vertical side wall a whole will be cut to fit a female 120VAC power input. Three pairs of holes will also be cut on this vertical sidewall for RCA connections for Audio Signal 1 & 2, as well as the Master Output. The Evaluation Board will be fastened to the inside of the case via mounting screws which connect to its plastic leg supports. If it is possible to find potentiometers with mounting screws for cases, these will be used, otherwise and epoxy will be used to adhere the knobs and sliders to the wood. This is also true of the ¼" PHONE connection and RCA connections.

The power supply included with the evaluation board will be used to supply power to the system. It accepts an input of 100-240 VAC at 50-60 Hz and outputs a 6 V direct current at 3 A. This ensures at maximum power the system runs at 18W, which is below the 20 W system requirement.

As stated above the Mixer enclosure is made of wood. This was done with spare wood and was not purchased. Even though the wood was free with respect to the project it is still high quality wood. The wood is also very light weight, which while not a requirement of the project, is an added bonus to the overall system. The top board is plywood with a thickness of 3/8”. This board was cut to a width of 12.5” and a length of 13.875”. Changes in board dimensions were necessary to accommodate development board size and provide ample room for wiring and a breadboard. The size requirement was an approximation and not a hard requirement, which allowed for flexibility in changing the size when all appropriate equipment was obtained. The sides of the enclosure were created from ¾” thick wood boards. The length and width of the side boards were the same proportions as to accommodate the top board. The side boards were 5” tall. The bottom of the enclosure was made of the same ¾” thick board as the side. These pieces were cut as needed to create a bottom that fit within the dimensions of the side wall. The bottom piece was not lower than the sidewalls so that the inside of the case is actually ¾” higher than the bottom of the enclosure (the thickness of the wood). Two “feet” were created to for the enclosure as well. These were 5/8” x 5/8” and 7” long. These were positioned in the length direction of box. The box was completed by attaching the pieces together with #6 x 1 5/8” screws and wood glue.

The sides, bottom, feet, and top of the enclosure were all put together to ensure the dimensions all fit together. The top board was then taken off so drilling and cutting could be done. The top side board, which receives the power and audio inputs as well as outputting the master audio channel, was drilled all the way through to create a hole with a ½” diameter. Two of these holes were made for each channel (Channel 1, Channel 2, Master Output). These were located in the center of the side board and spaced approximated ½” horizontally from each other. A ½” hole was also cut to accommodate the USB that programs the board from the laptop. A rectangular cut was also made in this side wall so that the input of the power supply adapter could be accessed from outside the board. Small pieces of plywood were nailed over portions of this cut so that the power supply could not fall out of the case. A picture of this side wall can be seen below in Figure 8.

Figure 8: Top Side Wall

The bottom side (The side where the Monitor Output is) had a ½” hole drilled in it so that the Monitor Output Channel could be accessed with a ¼” PHONE connection. The extra ¼” was needed so that the rubber casing of the headphone wire could fit in the hole created. This can be seen below in Figure 9.

Figure 9: Bottom Side Wall

The top board required the most drilling and cutting as the user interface potentiometers and buttons would be located there. Eight 9/32” holes were drilled in the board where the rotary potentiometers would be placed. These were done in identical columns (one for each channel). The left column represents the controls for Channel 1, and was located 5.5” from the left side of the box. Likewise the right column was 5.5” from the right side and represents Channel 2. These

channels had a horizontal spacing of 1.5” from each other. Vertically each hole is spaced 1.5” from each other. ¼” holes were also drilled as ports for the push buttons for the Filter Select and Cue Channel Select controls. The Filter Select holes are located to the left and right (respective to Channel) of the fourth potentiometer hole located from the top down. The Cue Channel Select button is located 2” from the left side and 2.5” from the bottom side. Next the slide potentiometer slits were cut using a Sawzall. These cuts were 3” long vertically and located approximately 2” from the bottom potentiometer holes. A Sawzall blade was then used to manually (by hand) widen the slit so that the slide potentiometer could be used without it touching the sides of the wood. The top board can be seen below in Figure 10. This picture is of the completed mixer however with the potentiometers, buttons, and labeling present. The top three potentiometer holes for each channel are covered, as there are no individual EQ band potentiometers, so particles and liquids cannot enter the board.

Figure 10: Completed Top Board

After the case was assembled the inner construction of the system could begin. First the ¼" Stereo PHONE connector was attached to the case. This was done by using the bolt and washer included with the connector to secure the jack to a 2" diameter metal washer. This washer (with the jack attached) was then screwed into the side wall using three screws. Speaker wire was soldered to the Left, Right, and Ground stubs on the jack. These wires were then soldered to the corresponding channels on a striped mini-stereo cable, which could be connected to the output port of the evaluation board later. All solder joints and exposed wire were covered with insulating electrical tape to minimize exposure to hot leads. The completed ¼" connector can be seen below in Figure 11.

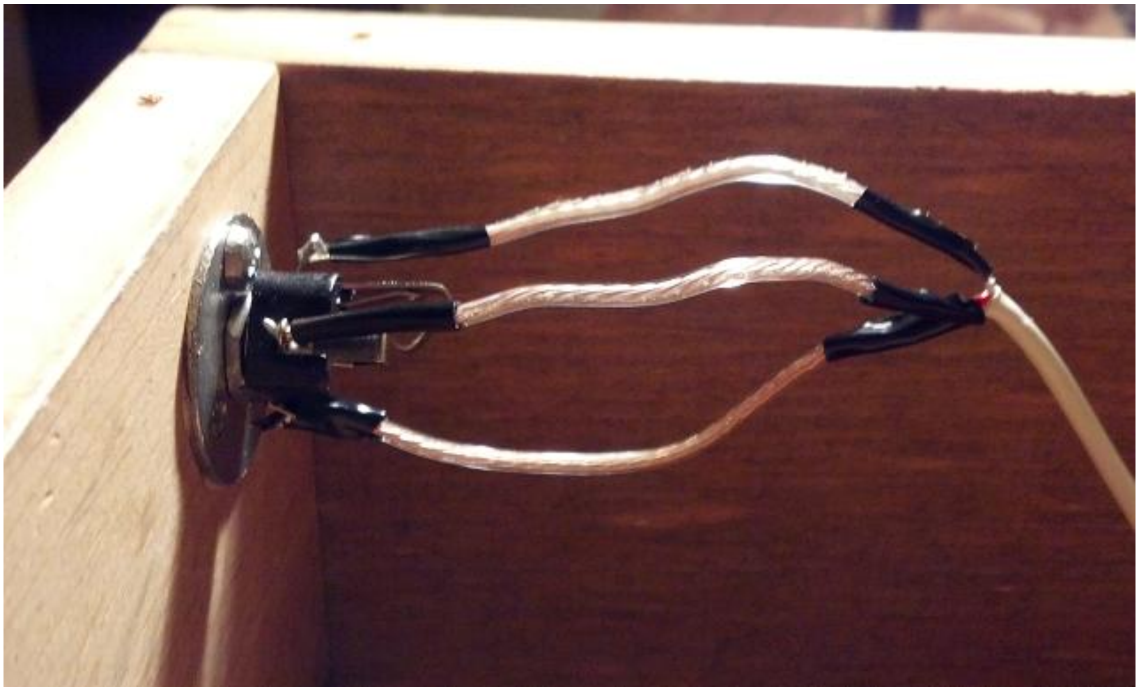

Figure 11: 1/4" Stero PHONE Connector and Wires

Next wooden blocks were used to secure the power supply adapter inside the enclosure. These would ensure that the adapter could not move out of position inside the box. These blocks were glued and screwed into the bottom of the box. Their location is below and to the right of adapter when using the top side wall as a reference frame for north. The adapter can be seen below in Figure 12.

Figure 12: Power Supply Adapter inside Enclosure

The ADAU 1446 Evaluation board was then mounted to the inside of the box. The board was positioned so as to allow room for wiring and the breadboard which the potentiometers and pushbuttons would be wired. The board was mounted by drilling holes in the bottom of the box in dimensions that corresponded to the size and location of the evaluation boards mounting screws. Flat tipped screws were inserted through these holes and the plastic leg mounts of the evaluation board were secured to these. The board was then placed over the mounts and secured with the plastic mounting screws included with the evaluation board. The installed ADAU1446 can be seen below in Figure 13.

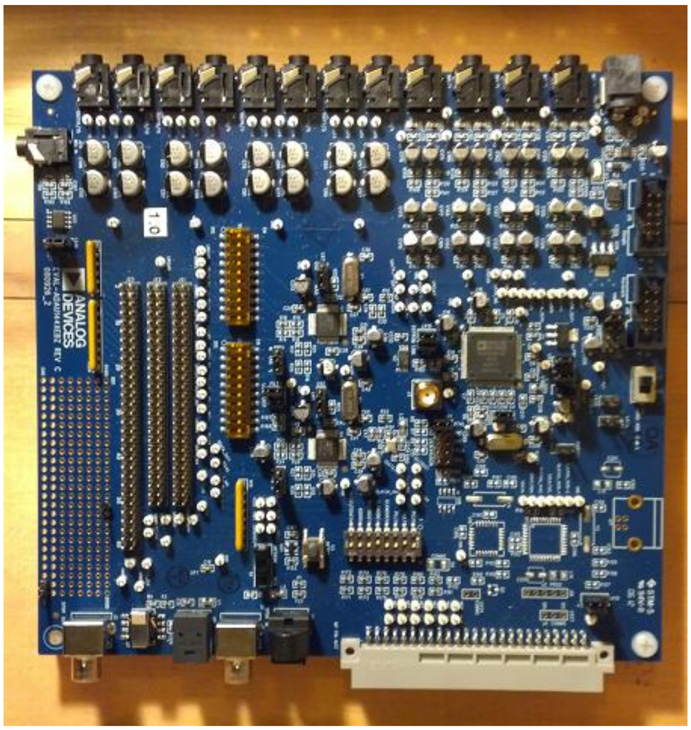

Figure 13: Mounted ADAU1446 Evaluation Board

Three Female-Female RCA adaptors were used to interface the RCA inputs from outside the mixer to the mini-stereo inputs of the Evaluation board. These were connected together with two blocks of wood and a long screw that was drilled through the plastic housing of each adapter, and the two wooden blocks that were to either side of the Channel 2 adaptor. This single unit was fitted into the previously drilled holes in the top side wall and secured to the side wall using four screws, two for each wood block. A visualization of this can be seen in Figure 14, as well as the previously discussed sub-systems inside the almost complete box. The white RCA connector corresponds to the left channel and the red RCA connector corresponds to the right channel.

Figure 14: RCA Adaptor Unit, Mixer Sub-Systems

Channel 1 and Channel 2 are connected to mini-stereo inputs 0/1 and 2/3 respectively, on the Evaluation Board. Master Output is connected to mini-stereo output 0/1 and Monitor Output is connected to mini-stereo output 2/3. These connections were made using RCA to mini-stereo wires, with the exception of the Monitor Output Channel, which used a mini-stereo wire that was stripped on one end and soldered to the ¼" Stereo PHONE jack as detailed earlier. The excess RCA to mini-stereo wire was bundled and zip-tied together to reduce wire clutter inside the enclosure. This was also done with the excess power supply wire that connects to the evaluation board. These wire bundles were placed to the left of the Evaluation board seen in Figure 14.

After this the potentiometers and pushbuttons were attached to the top where their interface holes and slots were located. Each potentiometer and pushbutton and previously had wires soldered to their connector leads. Those with excess wire showing were covered with

insulation electrical tape. The pushbuttons fit tightly enough into their drilled holes that the threads on the pushbutton housing could screw into the wood. The slide potentiometers were attached to the top board using 1/16" mounting screws that fastened to holes in the potentiometer metal housing. These slide potentiometers were then capped with Slide Potentiometer Knobs [11]. The Filter Control Rotary Potentiometers were attached to the top board using Gorilla Super Glue Gel. This was done as the threading on the potentiometer shaft was not long enough for the nut and washer to secure the potentiometer to the board. This was left to bond with the board for 24 hours.

The potentiometers and pushbuttons were connected the Evaluation board Multipurpose (MP) pins via wiring circuits on a breadboard. These circuits connected to the MP pins using J-hook connector cables. As recommended in Analog Devices', *Using Hardware Controls with SigmaDSP GPIO Pins* [12], the potentiometers were connected to the board with a 470 Ω (472 Ω recommended) resistors between the potentiometers output pins and the MP pin they were connecting with. This ensures and excessive current level does not harm the MP pin input. Likewise the pushbuttons used a 10 kΩ resistor between VDD and the top pushbutton terminal, with the bottom pushbutton terminal connected to GND. The input from the pushbutton to the MP pin was taken from the node between VDD and the top pushbutton terminal. VDD and GND are obtained from the Evaluation board via a soldered lead that connects to the breadboard, creating VDD and GND buses. VDD from the Evaluation board is equal to 3.3V. A diagram of the breadboard circuit can be seen below in Figure 15.

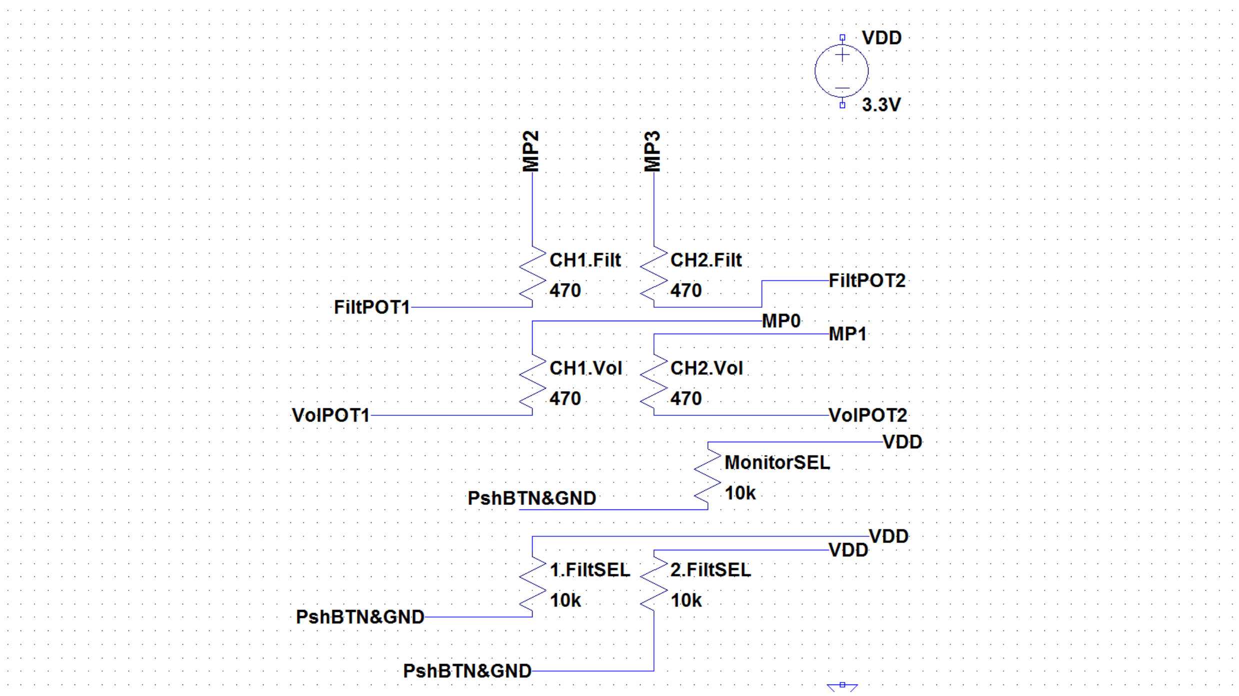

Figure 15: Potentiometer & Pushbutton Board Interface Circuit

With the potentiometers and pushbuttons wired to the Evaluation board Multipurpose pins the breadboard was secured to the inside of the enclosure, below the Evaluation board shown in Figure 14, via an adhesive strip that could be peeled off the bottom of the breadboard. The top of the mixer box was then placed on the rest of the enclosure, with care taken as not to damage any of the wiring, aligning the screw holes and screwing in the top of the box. Each interface component was labeled with permanent marker, as well as the audio channel inputs and master output. Finally a *Modern Walnut* oil-based wood stain was applied to the outside of the mixer box to give the wood a uniform color, adding aesthetic appeal to the Two Channel Audio Mixer.

Chapter VII. – Integrated System Tests and Results

Project Verification Test Plan:

The plan for a System Level test on a very basic level entails passing a simple sine wave through the system, with no EQing or filtering being applied to the waveform, and measuring via an Oscilloscope the input waveform and the output waveform, checking for distortion. This can also be done with more complex waveforms and even musical pieces, though the latter might be more difficult to analyze.

Subsystem Verification Test Plan:

To test the EQs and filters, a recorded piece of white noise will be passed through the mixer. For each band of the EQ the boost and attenuation will be tested on this piece of white noise. This will be recorded digitally and then be analyzed using AN-879 Analyst plug-in in Sonar X2 Producer. With this dB vs. Frequency is plotted and analyzed in real-time to ensure the EQ bands are applying the desired boosts and attenuations. This method will also be applied when testing the filter, making sure that the cutoff frequencies of the high and low pass filters are changing as designed. An example of what this will look like can be seen below in Figure 5. The Analyst plug-in can also be used to test the volume controls of each channel and verify the proper dB boost or attenuation is being applied. Testing the Monitor Output select buttons will consist of playing two different songs on channel 1 and 2 and listening through the Monitor Output to ensure the correct song is received.

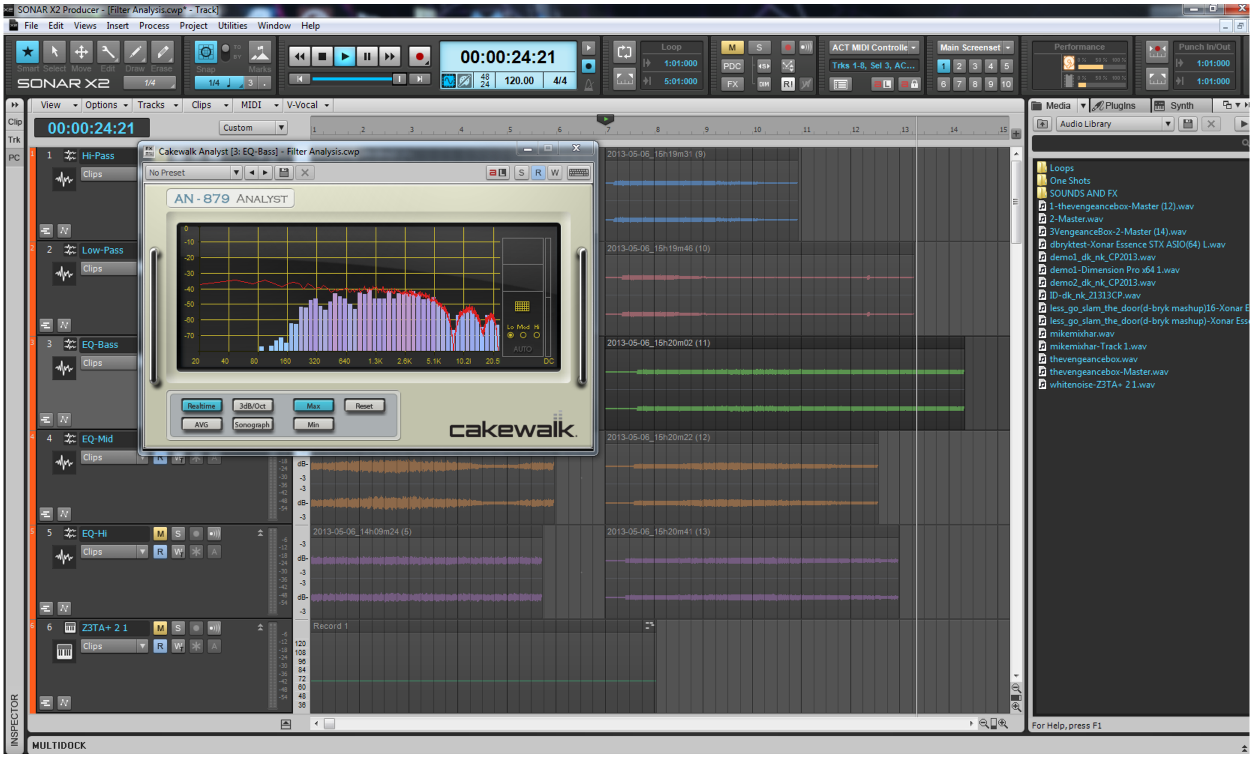

Figure 16: EQ Bass Test Example

Figure 5 gives an example of what testing will look like using Sonar's Analyst plug-in. In the picture the Bass is maximally attenuated, which can be seen as the analyst was unable to measure any frequencies under 80 Hz.

The first test that was performed on the mixer system was to pass a sine wave through the mixer, comparing the input waveform with the output waveform, and looking to see if there is any distortion in the output waveform. A sine wave of 500 Hz with an amplitude of .5 V (1 V_{PP}) was generated via a function generator. The input waveform was monitored on Channel 2 of the oscilloscope and the output waveform was monitored on Channel 4 of the oscilloscope. The input peak to peak voltage (V_{PP}) was measured between 940-960 mV. The output waveform (on Channel 4 of the oscilloscope) voltage was measured between 840-860 mV. There was no distortion to the waveform however, which showed the mixer was not adding any unwanted noise to the signal. The output signal was delayed when compared to the input signal, which is to be expected as it takes time to process the signal in the DSP core. To ensure that the amplitude of the input signal was the same as the output signal, a gain block was added to the SigmaStudio project, and was adjusted until the input voltage matched the output voltage magnitude. This resulted in a 1.12 dB amplification of the input signal. A capture of the amplitude matched input and output signals can be seen below in Figure 17. In addition to this test the output was also measured with the Channel Volume fully attenuated, which can be seen in Figure 18. The output with the High Pass and Low Pass filters at their maximum cutoff frequencies were also captured and can be seen in Figures 19 and 20 respectively.

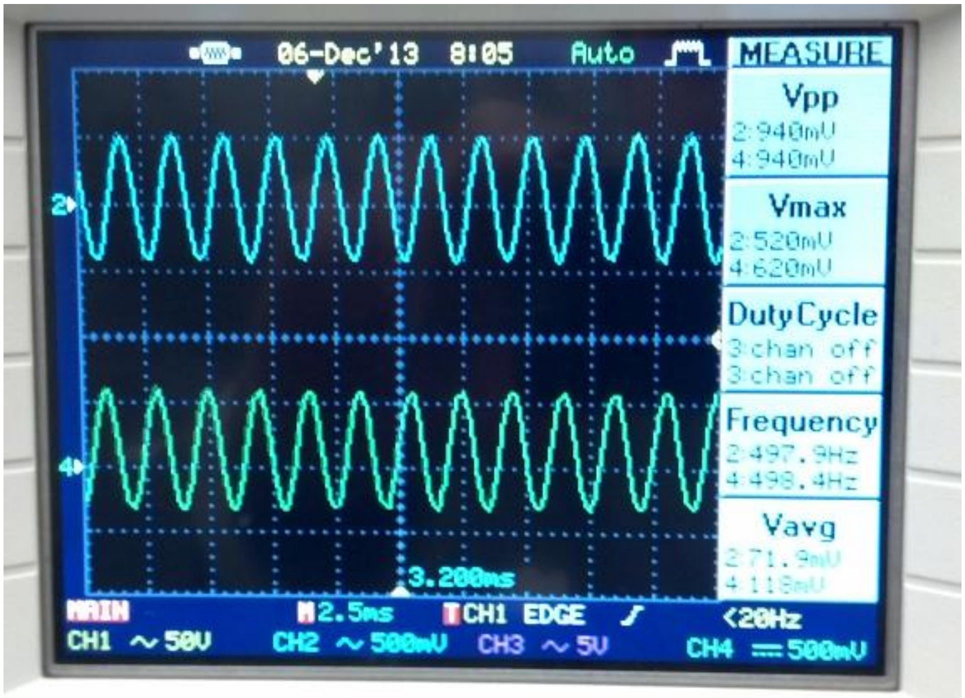

Figure 17: CH2 Input Waveform, CH4 Output Waveform

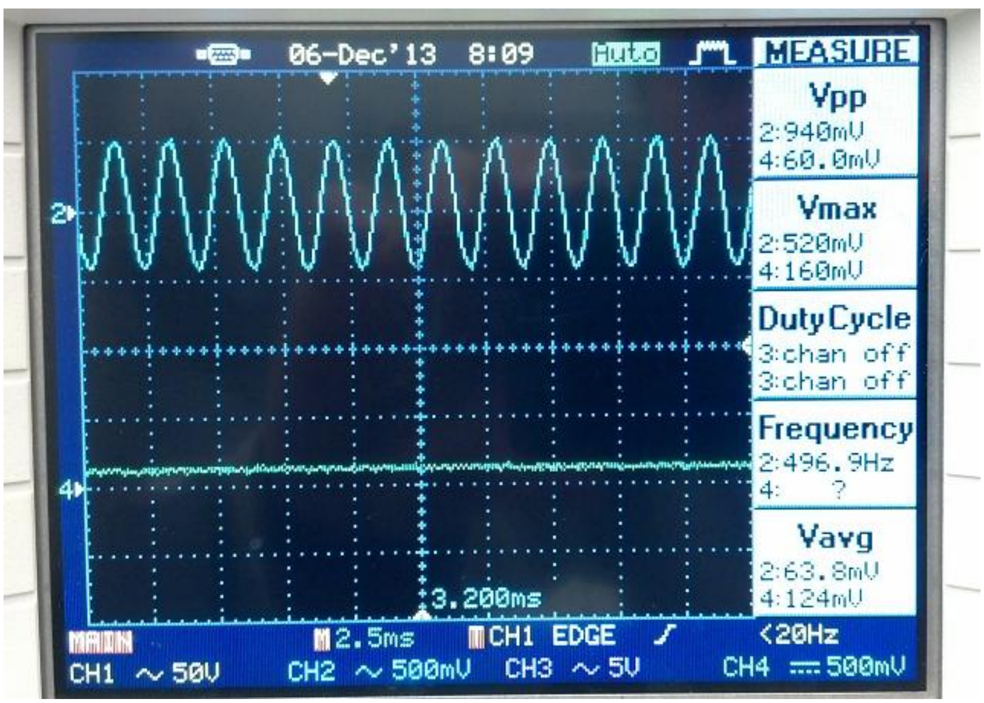

Figure 18: Channel Volume Fully Attenuated, CH4 Vpp = 60 mV

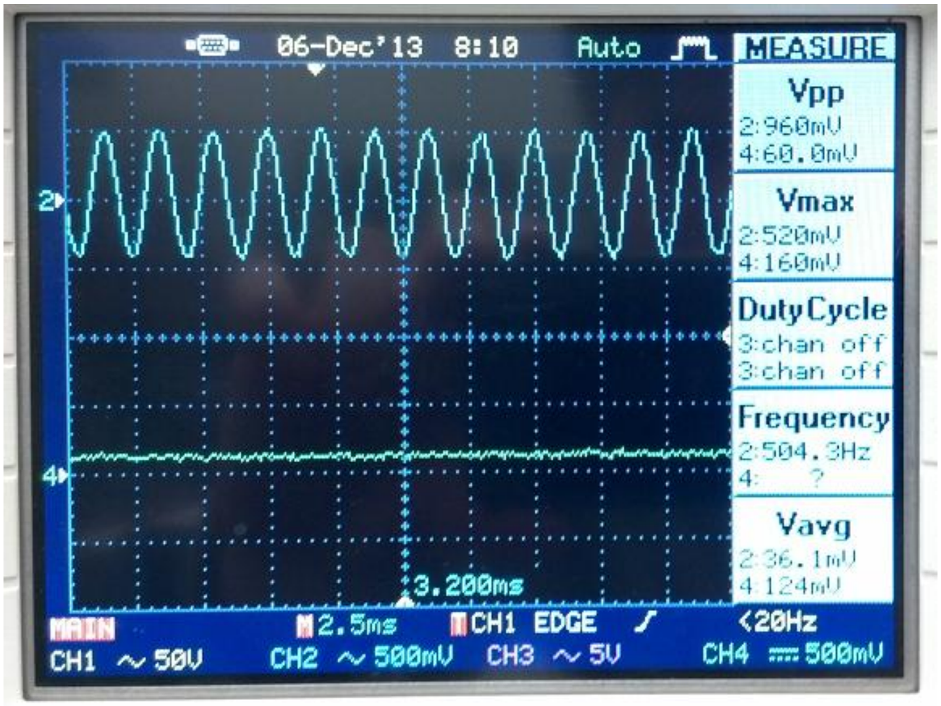

Figure 19: High Pass Filter Maximum Cutoff Frequency, CH4 Vpp = 60 mV

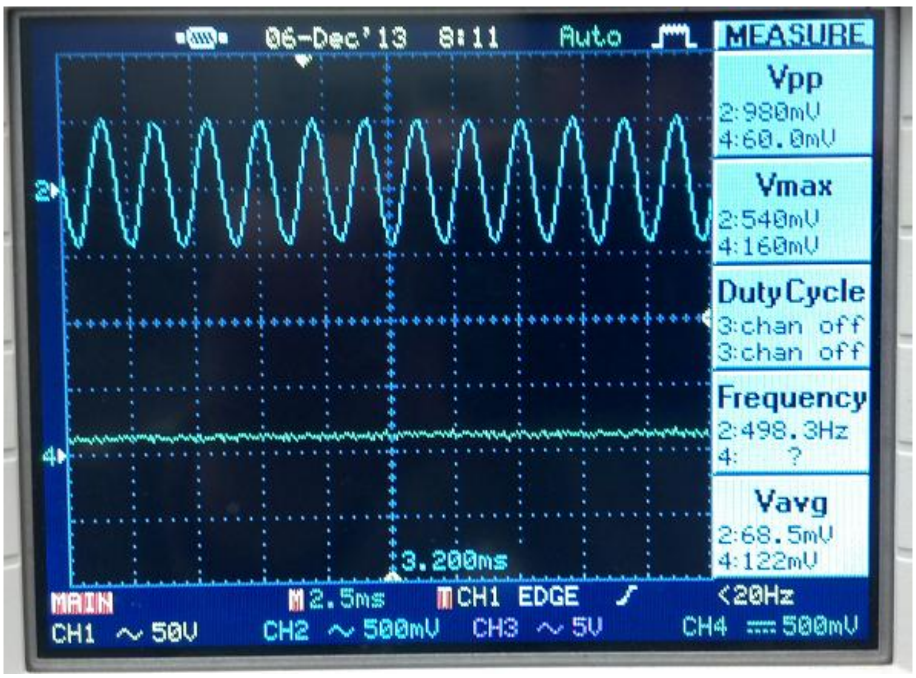

Figure 20: Low Pass Filter Maximum Cuttoff Frequency, CH4 Vpp = 60 mV

The Channel Volume, High Pass Filter, and Low Pass Filter output waveform captures show that the elements can fully attenuate the input signal waveform to a negligible value as desired, which would translate to be able to attenuate the input signal to inaudible levels when it

is desired to do so. This entire test process was not done with music as it became very difficult to analyze such complex waveforms.

To test the Filter and Equalizer sub-systems white noise was generated and played through the system. Each sub-system (Filter, EQ) was activated independently and the output was recorded in Sonar X2. In Sonar X2 the recording could be analyzed using the AN-879 Analyst plug-in, which provides dB vs Frequency graphs. While using the Analyst it was immediately noticed that the system is providing ground noise at 60 Hz. This is unwanted and is something that can be planned to fix in the future improvements section of the Conclusion. External Filtering could help attenuate the 60 Hz hum, as well as adding ground loop protection to the system.

The High Pass Filter was the first element tested. As cutoff frequency is increased it can be seen that lower frequencies attenuate first as the filter works its way from the Low frequency range, through the Mids, and finally to the High Frequency ranges. Figure 21 below shows the High Pass at maximum cutoff. If the 60 Hz ground noise is ignored it can be seen that the almost all frequencies are attenuated by -40 dB, with many frequencies attenuated by -50 dB. This shows that the filter is working as intended and attenuated across the whole frequency band when the cutoff frequency is at its maximum.

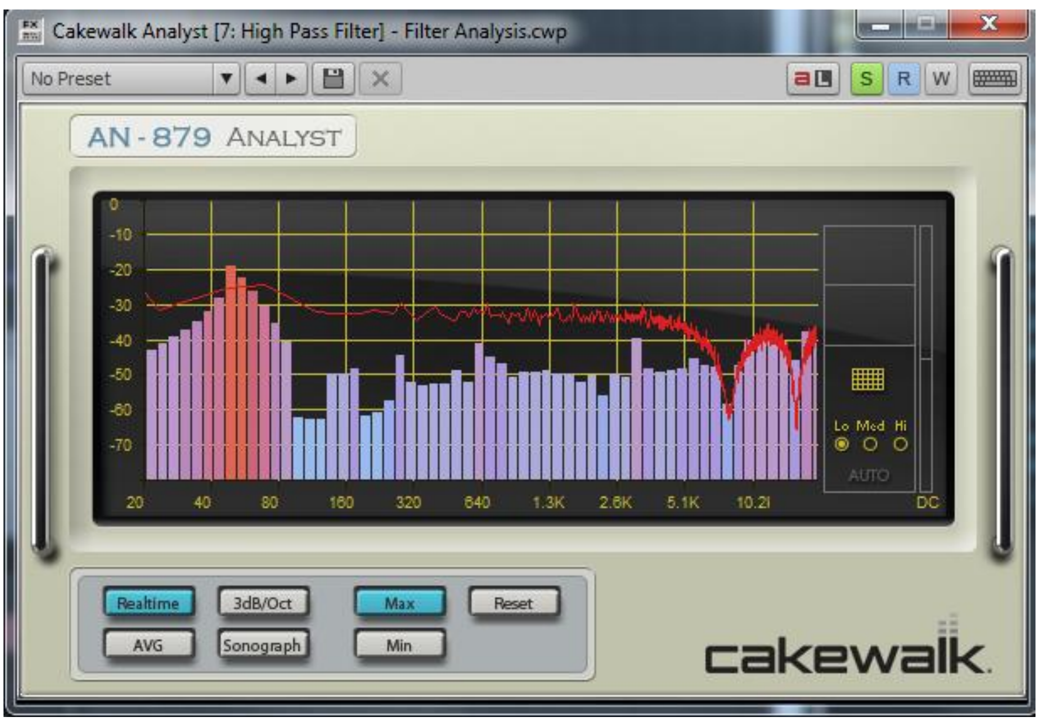

Figure 21: High Pass Filter at Maximum Cutoff Frequency

The Low Pass Filter was the next element tested. Like the High Pass Filter test, the Low Pass Filter was also taken to its maximum cutoff frequency, which in this case would be a low frequency of 20 Hz, and the attenuation was observed. Again the 60 Hz noise needs to be

ignored. Figure 22 shows the Low Pass Filter with the cutoff frequency at its maximum. It can be seen that the Highs are attenuated down to as much as -70 dB. The Mid range frequencies are also greatly attenuated with most below -50 dB and many below -60 dB. The Low range is hard to analyze due to the 60 Hz noise but it can be seen that around 100-140 Hz there is attenuation in the range of -40 to -55 dB. This shows that the Low Pass Filter is performing as desired for the system.

Figure 22: Low Pass Filter at Maximum Cutoff

The Low Frequency band was the first band tested in the EQ sub-system. The High and Mid band controls are completely attenuated for an easier analysis. The Low band starts completely attenuated as well, which can be seen in Figure 23. Again the 60 Hz noise makes analyzing low end frequency ranges harder but that the frequency ranges up to 320 Hz, where the corner frequency for this band is, is attenuated by at least -40 dB and -50 dB in most cases. Figure 24 shows the band at maximum gain boost and the difference is very noticeable. From 20 to 40 Hz the level is over -30 dB, which is a +10 dB change. Again that value is hard to analyze because of the ground noise. The biggest difference can be seen at 100 Hz where the level went from almost -50 dB to nearly -20 dB, a difference of +30 dB. The dB level of the range from 20 Hz to 320 Hz has clearly risen so it is clear that this band both attenuates and boosts the 20-320 Hz range.

Figure 23: EQ Low Band Maximally Attenuated

Figure 24: EQ Low Band Maximum Boost

The Middle frequency range (320 Hz to 5120 Hz) was tested next. The same methodology was used as for the Low frequency range. When the band is at maximum attenuation it can be seen that much of the passband is under -50 dB of attenuation, with the

upper-mid range near -60 dB of attenuation. This can be seen in Figure 25. This indicates that the EQ is properly attenuating this frequency band. When the band is at maximum boost the range is brought up to approximately -30 dB, a boost of 20 to 30 dB. This shows that the Mid Range properly boosts its designed frequency range. Figure 26 shows this. Like the Low Range band, the Mid Range EQ band properly attenuates and boosts its frequency range.

Figure 25: EQ Mid Band Maximum Attenuation

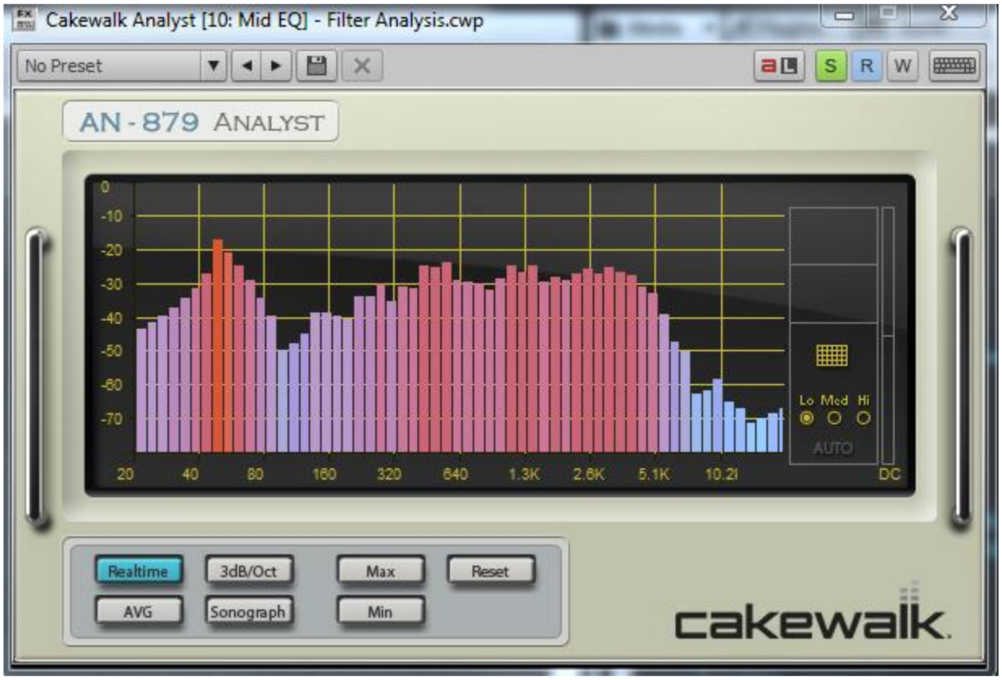

Figure 26: EQ Mid Band Maximum Boost

Finally the High range frequencies band of the EQ was tested, using the same methodology as the previous two bands. When the band is maximally attenuated the frequency range (5120 Hz to 20 kHz) drops to below -70 dB, showing that the control properly attenuates the band. This is seen below in Figure 27. Likewise when the band is boosted to its maximum the range rises to approximately -40 dB, an increase of 30 dB. Figure 28 illustrates this. The High frequency rang band can be properly attenuated and boosted.

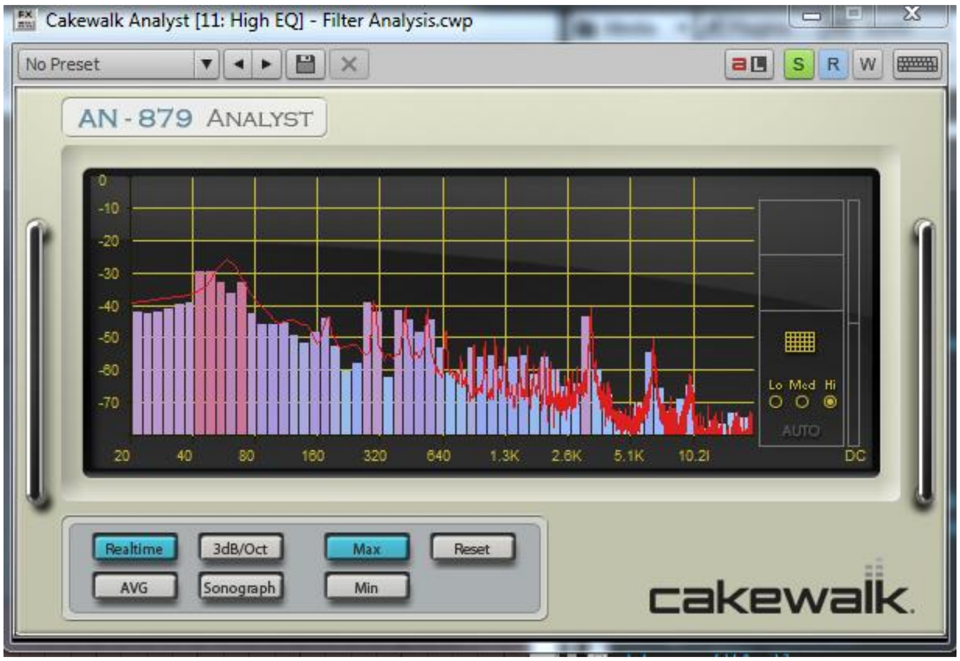

Figure 27: EQ High Band Maximum Attenuation

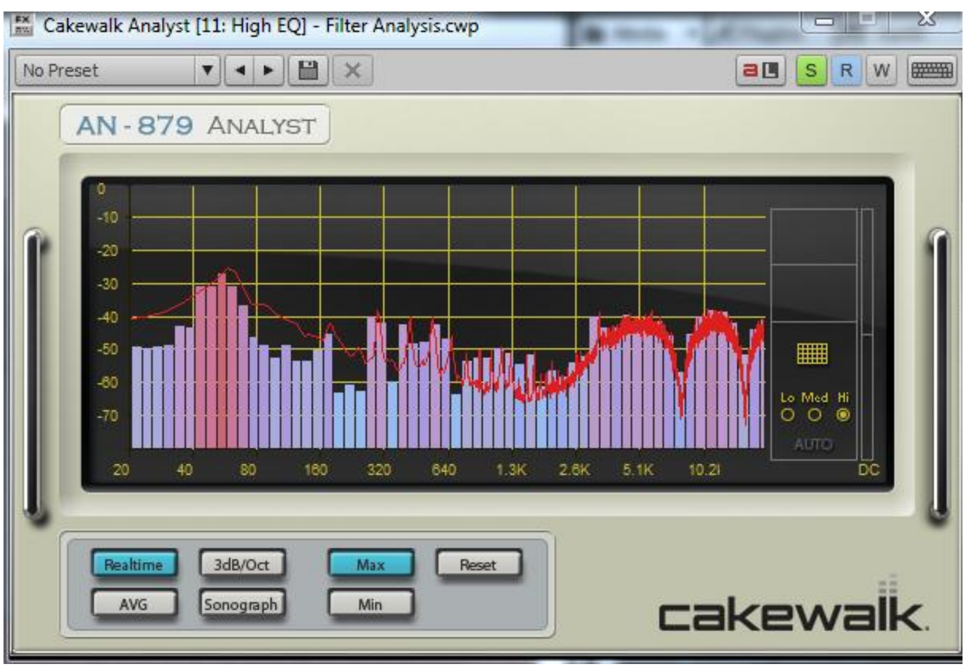

Figure 28: EQ High Band Maximum Boost

This test shows that the Equalizer sub-system works as intended and actively boosts and attenuates each frequency range. The average difference between maximum gain boost and maximum attenuation for all the ranges is around 30 dB, which shows they have the same uniform performance as desired.

To test the Monitor Output Select button the output was observed via listening. The Channel defaults to Channel 2 and when the button was pressed it switched to output Channel 2. This test indicated that the Output Select function performed correctly and as expected. The Filter Select buttons for each Channel were also tested by activating the buttons and listening to see if the filter type changes. This is easy to hear because when the High Pass Filter is acting as an All Pass (Cutoff Frequency near 20 Hz) and the Low Pass Filter is engaged, the Low Pass is at its maximum cutoff frequency and it attenuates the Channel. This also holds true for when the Low Pass Filter is acting as an All Pass (Cutoff Frequency near 20 kHz) and the High Pass Filter is activated. Both Channel 1 and Channel 2 have working Filter Select controls.

The presence of the noise at 60 Hz made testing for the true Total Harmonic Distortion impossible for the system. With all the stages bypassed in SigmaStudio, the THD was tested for in Room Equalization Wizard, which generated a sweep from 20 Hz to 20 kHz and measured the SPL of the system, which came at to 82.9 dB. The presence of the noise at 60 Hz resulted in THD of 35.7%. To get a better idea of the systems THD without the interference a sine wave at 1000 Hz at -20 dB was generated and passed through the system. At a low gain setting this resulted in a THD of approximately 0.174%, which is much closer to the required 0.1%. This can be seen in the Figure 29 below. The presence of the 60 Hz noise and its harmonics can also be seen in the Figure, with a dB spike in the frequency chart at 60 Hz. The test did reveal the frequency response of the system which can be seen in purple in the figure, the system performed well with a flat frequency response. The test begins to alias as it passes 20 kHz as it had a sampling rate of 44.1 kHz.

	THD %	Desired THD %
20Hz - 20 kHz Sweep	35.70%	0.10%
1000 Hz Sine Wave	0.174%	0.10%

Table 3: Total Harmonic Distortion Test Results

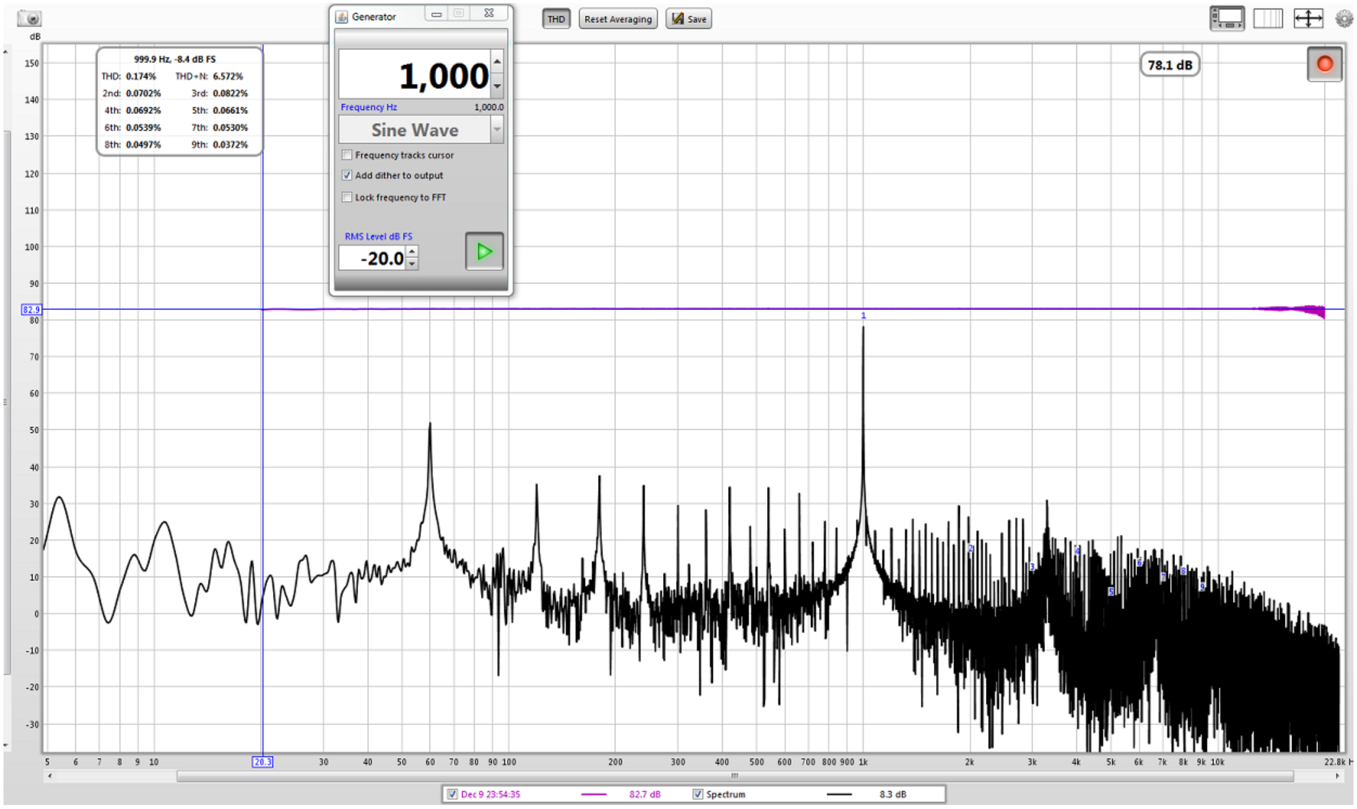

Figure 29: Total Harmonic Distortion Test

These tests showed that the system as a whole and the individual sub-systems worked as designed. The input of a sine wave into the system while monitoring the input and the output of the system showed that the system did output an undistorted sine wave of approximately equal magnitude. This test also showed that the Volume and Filter controls would attenuate the signal when fully activated. The filter and equalizer sub-system analysis uncovered a 60 Hz noise, likely cause by ground noise, which is a small problem for the system. However these tests also showed that the filter cutoff frequency does change when activated and will attenuate the signal when at its maximum cutoff frequency relative to whether it is a high or low pass filter. The tests also showed that equalizer components properly boost and attenuate the audio signal, with a level difference of approximately 30 dB between maximum boost and attenuation. The pushbutton tests were simple but did demonstrate that the system could switch between filter types and select which channel was desired to output through the Monitor Output. The ground noise made the testing of Total Harmonic Distortion difficult, causing a THD of 35.7% due to the presence of the noise. When testing with a 1000 Hz Sine Wave, a THD of 0.174% was recorded, this was much closer to the target THD of 0.1%. With exception to the ground noise, these tests demonstrate the working order of the system and its sub-systems.

VIII. Conclusions

The Two Channel Audio Mixer correctly mixes two separate audio channel inputs, creating one Master Output Channel. The most basic function of a mixer by definition is to mix to signals together, and the system achieves this specification. This mixer also allows for each input audio channel to be routed to a Monitor Output Channel. This allows for a DJ to listen to one channel without it being output through the Master where an audience could hear it. This is a critical specification as it allows the user to synch up two tracks for proper mixing. The Total Harmonic Distortion of the output audio signal was not less than or equal to 0.1%. This was due to the introduction of a 60 Hz noise most probably caused by ground noise. Alternatives for improving this flaw in the future could include researching how to best dispel ground noise hum. An alternative could be exploring how a filter external to the system with a bandstop at 60 Hz and high Q-factor could help eliminate the noise. The tests for THD did however show the frequency response of the system, and verified that the system has a frequency range of 20 Hz to 20 kHz as desired. This allows the system to replicate audible audio signals. The mixer satisfied the project specification that it uses RCA inputs and outputs to interface audio signals. This was done for Channel 1, Channel 2, and the Master Output Channel. The mixer was also able to implement the Monitor Output Channel with a ¼" PHONE interface, allowing for the user to connect headphones to the channel. It terms of power consumption, the use of the ADAU1446 power supply adaptor ensured that the system would at maximum use 18 W of power. This is 2 W below maximum allowed power consumption for the system. The dimensions of the completed box ended up being 12.5" (W) x 13.875" (L) x 5.5" (H). This is larger than the desired approximate lengths in specification number 8. However those dimensions were made approximate to allow flexibility in final size as not all the parts were known when the initial dimensions were made. The mixer case was constructed of wood that was sanded and stained, fulfilling the project specification, and adding aesthetic appeal. The tenth specification of the project was also fulfilled as the mixer was implemented via the ADAU1446 and its EVAL-ADAU1446EBZ Evaluation Board. The use of RCA to RCA adaptors allowed for the RCA external inputs and Master output to interface with the 1/8" mini-stereo connections on the evaluation board. The use of RCA to 1/8" (mini-stereo) wire allowed the adaptors and the board to be connected. With partial exception to the 3-Band Equalizer the Volume Control, Filtering, and Monitor Output selection were implemented as designed. The Volume Controls for each channel were successfully implemented with slide potentiometers, and the Filter Controls were implemented with rotary potentiometers. The Monitor Output Channel select was also successfully implemented with a pushbutton. Master Volume, Monitor Volume, and Crossfader were not included in the finalized implementation of the project. The Crossfader was not desired as it is not a critical component, and the Master Volume and Monitor Volume controls can be implemented and controlled in real time in SigmaStudio. The 3-Band Equalizer was not able to be implemented as desired as unfortunately the development board could only take four analog signal inputs as it only had a four channel Auxiliary ADC. This also made implementing the Master Volume and Monitor Volume not possible. ADCs connected in SPI with the board were

the designed solution for this problem but unfortunately the Serial I/O of the board was not designed for SPI Interfacing. The work around described in the design section was implemented so that the 3-Band EQ for each channel could still be controlled externally in real time with two rotary potentiometers and one slide potentiometer. This was successful. Proposed alternatives for fixing the problem in the future include finding ADCs that can interface with the Evaluation Board correctly, or experimenting with a second microcontroller where the analog inputs from the extra potentiometer controls are converted to a digital form and sent to the Evaluation Board for processing. Finally the project followed safety protocols by insulating solder joints and exposed portions of metal wiring where shock could occur. The system is also properly grounded through the evaluation board power supply. The system is enclosed in wood and is not intended to be opened by the user. It is required that if someone open the enclosure they first disconnect all power from the system, so as to reduce the risk of electrocution.

Future improvements of the program include fixes to the problems mentioned previously, but also added functionality. This includes adding LED outputs to the board that indicate which channel is playing through the Monitor output. LED indicators could also show which Filter is selected (High or Low Pass) next to the Filter control. Another improvement could be to have an LED level meter next to the Volume Controls that shows the dB level of each channel, and can indicate if clipping is occurring. An improvement to increase the durability of the system would be to print a PCB for the potentiometer and pushbuttons interface circuitry. Finally an advanced improvement that would add great functionality to the board would be creating an ASIO that would allow the mixer to interface with DJ software on computers. This would open the mixer up to those who are laptop DJs.

While the Two Channel Audio Mixer has flaws, it provides the basic functionality of a DJ mixer, allowing for two audio inputs to be manipulated as desired and mixed together. The final Two Channel Audio Mixer works and provides a fun and powerful system to use.

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Appendicies:

A. Specifications

Marketing Requirements	Engineering Specifications	Justification
2	1. The device mixes two audio channels into one master output channel.	In keeping with expected operation of an Audio Mixer.
2,3	2. Each audio channel routes to a monitor channel.	A customer must preview incoming signals without outputting through the master channel.
1, 4	3. Total Harmonic distortion of audio signal $\leq 0.1\%$.	To preserve audio signal fidelity.
1,	4. The mixer has a frequency range of 20 Hz – 20kHz.	The mixer plays all audio frequencies humans can hear.
1,2,4	5. The mixer uses RCA connections for input channels and output channel.	Industry standard and makes setup easy for customers. [3]
1, 2, 3,4	6. The mixer uses a ¼" PHONE connection for the monitor channel.	Ease of use and familiarity for customer. [3]
2, 4	7. Desired power consumption of 20 W or less.	Power consumption similar to comparable 2-channel mixers on the market. [3]
2, 3, 5	8. Approximate dimensions of the mixer: 8.5” (W) x 11.9” (L) x 4.2”(H)	The mixer needs a similar size to current 2-channel mixers. [3]
3,4	9. Finished wood comprises the mixer’s external case providing desired style and appeal.	Fulfilling aesthetic expectations of the mixer.
1, 6	10. Implementation via EVAL-ADAU1446EBZ Evaluation Board, Digital Signal Processor.	Digital implementation, platform must process 16-bit or 24-bit audio and handle required amount of inputs and outputs.
1, 3	11. Audio Inputs and Outputs on board require 1/8” stereo connection.	Constraint of the board, Audio Signals are input and output through 1/8” connections.
2	12. 3-Band Equalizers, High and Low Pass Filters, Master and Monitor Volume,	This control surface is similar to regularly used DJ mixers [3], and provides ease of use for

	controlled via knob potentiometers. Channel Volume controlled by slide potentiometers. Crossfader controlled by slide potentiometer. Monitor Channel Select controlled by push button(s).	those using this mixer for the first time.
7	13. The mixer design requires NEC compliance	For public safety assurances. [4]
Marketing Requirements 1. The mixer should have quality sound fidelity. 2. The mixer should have easy operation for those familiar with DJ hardware. 3. The mixer should isolate audio signals in a monitor channel. 4. The mixer should cost less than comparable mixers on the market. 5. The mixer should aesthetically please. 6. Digital implementation of the mixer. 7. Ensure device does not harm the public.		

Table 1: Copy of Table 1

B. Parts List and Costs

	Cost Breakdown
Labor (at $15/hr)	$4770
Parts Total	$853.98
-Analog Devices EVAL-ADAU1446EBZ	$760.92
-Power Supply	*Included in price of Evaluation Board
-Case Material	Wood: Owned Screws: $1.44 Modern Walnut Stain: $9.99
-Interfacing Equipment	Rotary Potentiometers: $1.72 Slide Potentiometers: $5.90 Slide Knob: $1.90 Pushbuttons: Owned RCA-RCA Adaptors: $7.95 ¼” Stereo Jack: $2.50
-Misc. Components	RCA- 1/8” mini-stereo wires: $29.83 18 AWG Tinned Speaker Wire: $8.09

	1/8” mini-stereo wire: Owned 470 Ω Resistors: $2.50 10 kΩ Resistors: $1.34 Test-Hook Set: $19.90 Breadboard: Owned
Total	$5623.98
Estimated Optimistic	$4330.92
Estimated Pessimistic	$4730.92

Table 4: Parts and Costs Breakdown

An extra 80 hours of Labor was added to the original estimate price of $3570. This increase alone caused the Labor price to rise higher than the pessimistic estimated amount of $4730.92. The new parts total was actually under budget at $853.98. The original parts estimate was $960.92. This combined with the new labor costs equaled a total of $5623.98. The time it would take to complete the project was clearly underestimated.

C. Schedule – Time Estimates & Actuals

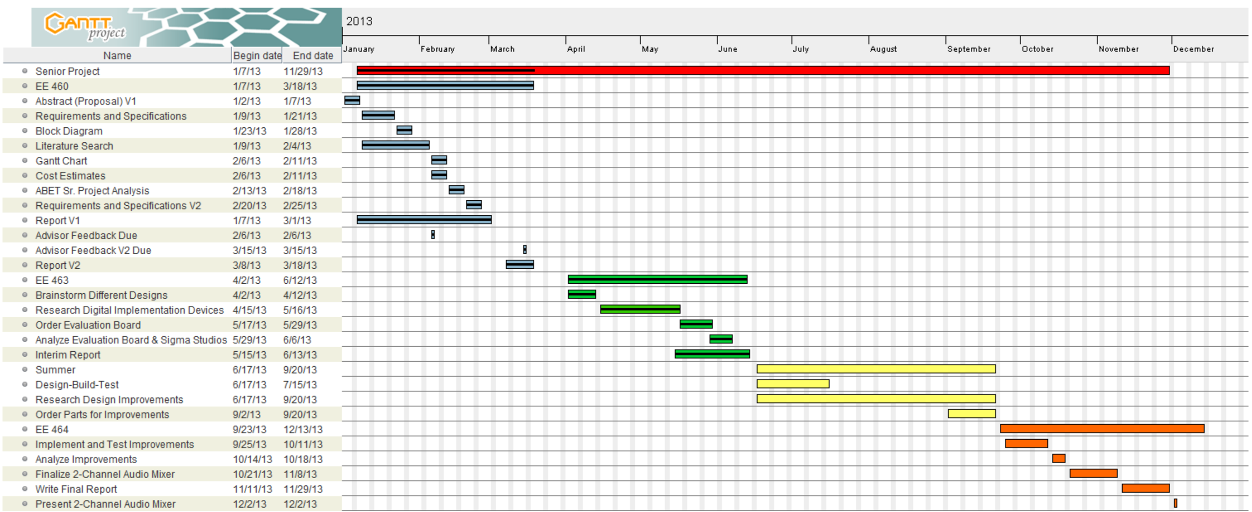


Figure 30: Updated Project Timeline

An extra 80 hours of Labor was added to the project timeline. These occurred primarily in the last few weeks of November and the first week of December. Construction of the actual system took much longer than anticipated and took place primarily during Thanksgiving week.

D. Analysis of Senior Project Design

TABLE V
APPENDIX D — ANALYSIS OF SENIOR PROJECT DESIGN

Project Title: 2 Channel Audio Mixer

Student's Name: Dylan Kinney **Student's Signature:** Dylan Kinney

Advisor's Name: Dr. Wayne Pilkington **Advisor's Initials: Date:**

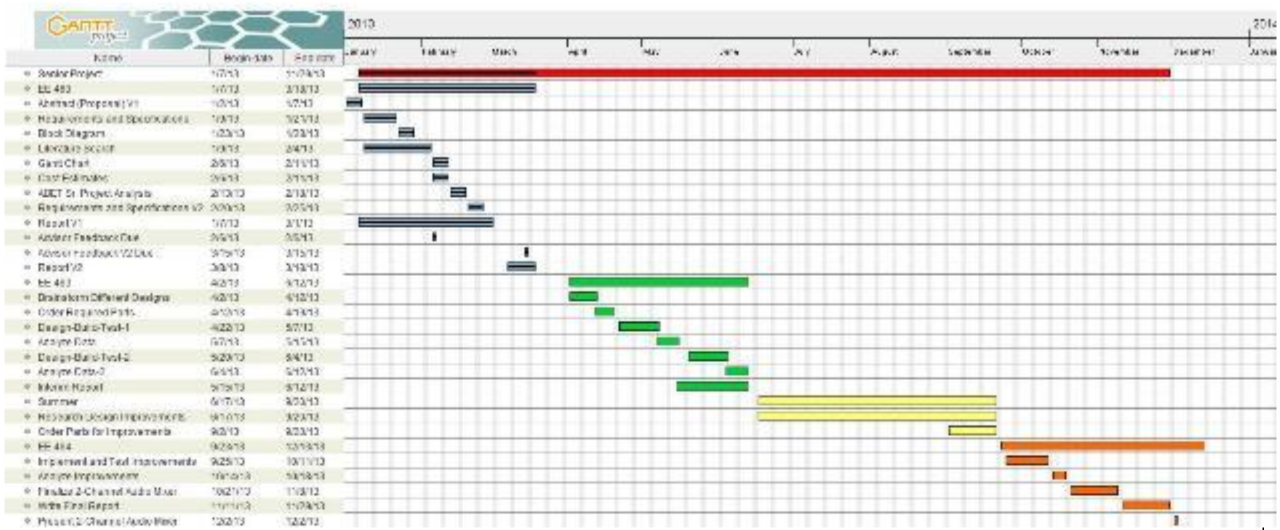
• **1. Summary of Functional Requirements**
This device takes two audio signal inputs, combines the signals, and creates a single audio signal output. The mixer controls the amplitude (volume) of each signal, as well as the ratio of each signals contribution to the master output. In addition, the mixer performs simple equalization of each audio signal with amplitude control of the Bass (20Hz-320Hz), Middle (320Hz-5120Hz), and Treble (5120Hz-20kHz) frequency ranges. The mixer can also high and low pass filter each audio channel. The mixer also allows for monitoring of each audio channel without outputting through the master output. A user obtains a single, desired, audio signal when faced with two audio signals, providing an essential functionality when one wishes to mix two songs together, or add two separate instrumental tracks together to create a larger piece. More detailed specifications are listed above in Table 1.

• **2. Primary Constraints**
The main constraints of the project are the audio signal quality required and the cost of the device to be competitive with existing alternatives. The platform must process 16 or 24 bit audio, and also have enough inputs for real-time user manipulation and mixing of the audio signals. This primarily affects the selection of a platform to digitally implement the Two Channel Audio Mixer. Options for digital implementation include an audio processing oriented microcontroller, a digital signal processor, or an FPGA.

• **3. Economic**
• Human Capital required for the project includes research time, design time, and labor necessary for proper implementation. This time correlates to financial capital by subtracting from the ability to make money by other means during said time. Digital implementation requires a platform that must be purchased. Parts for user controls and a power supply, as well as materials for a case will also require a significant financial contribution. The parts required for building the mixer also have manufactured and real capital costs to the suppliers who made them, and the tools they used. These materials, such as wood or metal for the case and silicon for electronics, also require natural resources to manufacture, which contributes to the project's natural capital. Other natural capital includes metals and chemicals used to make plastics.
• The costs of the project accrue primarily during the Spring of 2013 with two design, build, and test cycles occurring. These cycles require all of the above mentioned capital. Costs also accrue during the Fall of 2013 with improvement implementation and completion of the project. Benefits of the project accrue during the testing and analytical occurrences of the project's lifecycle, as well as upon project completion.
• Required inputs of the experiment can be seen above in the Block Diagram section of the report. Costs can be seen below in the Cost Breakdown table. The project is financed by my family and me. Moderate cost estimates of parts come to $700, this is primarily due to uncertainty of digital implementation platform.

	Cost Breakdown
Labor (at $15/hr)	$3570
Parts Total	$700
-Digital Platform	$50-$300
-Power Supply	$25-$100
-Case Material	$50
-Interfacing Equipment	$100
-Misc. Components	$150
Total	$4270
Optimistic	$4070
Pessimistic	$4570

• If the project reaches the market it would sell for $300-$500 over cost. Profit would go to investors and those responsible for project design and implementation. The above cost breakdown is indicative of the price to design and assemble one prototype model. Real costs for commercial sales will be different, primarily with regard to the price of labor.
• Products emerge approximately one fiscal quarter after final design and implementation. Products should exist for at least 5-10 years in proper working order. Maintenance and operation costs include fixing broken units, shipping replacement parts, and modifications to software.



Upon project completion, focus shifts towards emerging technological advances as a means of product improvement.

• **4. If manufactured on a commercial basis:**
• An estimated 1000 to 10,000 devices sold per year if manufactured on a commercial basis.
• Estimated manufacturing price of product of $100-$500.
• Estimated purchase price for each device of $800- $1000.
• Estimated worst case profit per year of $300,000 with 1000 units sold, $500 manufacturing price, and $800 sales price. Estimated best case profit per year of $9,000,000 with 10,000 units sold, $100 manufacturing price, and $1000 sales price.
• Initial cost for user of $800-$1000 because of product purchase. Estimated cost for user of $15 for RCA cables. Estimated user cost of $100-$250 for monitoring headphones. Total estimated cost of $915-$1265 for user. These are one-time costs and would not require further purchases for users.

• **5. Environmental**
• Environmental impacts include raw resources needed to manufacture the mixer, sources of energy needed to power manufacture device, and fossil fuels consumed during device distribution. Metals or wood are used in construction of the device's case. Semi-conductive and conductive elements are used in electronics purchased. Chemicals are used for manufacturing insulation and IC housing parts. Coal and Gasoline are the primary fuels for power plants providing power to manufacturer. Gasoline is used in the distribution of devices. Electricity consumption from coal and gasoline plants occurs during product use.
• Directly uses lumber or metals for product casing. Silicon is used for semiconductor devices. Copper, silver, and cold is also consumed in electronics parts. Indirectly uses coal and raw fossil fuels are consumed to create electricity for device use and manufacture.
• Device could harm ecosystems where lumber is taken from.
• Species such as birds, squirrels, and other woodland creatures are affected if harvested lumbar is not replanted by logging companies.

• **6. Manufacturability** Device is not intended for manufacturing so component selection would need changing in order to maximize profits. A warehouse and distribution service is also required for large scale sales of the device.

• **7. Sustainability**
• Maintaining the device would require a network of employees available for customer service and the repair of

devices.
• If the project uses wood from lumber companies that replant what they harvest, then the case of the mixer would be sustainable, over a period of years. If a metal case is designed, then the device would be using a limited resource and therefore not qualify as a sustainable device. The silicon and precious metals used in the electronics are also limited resources.
• Using certified environmentally friendly or recycled parts in the project. Also, discarding old devices in a responsible way and even using a cradle-to-cradle design method would improve the devices' sustainability.
• Certain parts of the design, such as electronic components, are processed in a very specific way and environmentally friendly alternatives may not exist currently. Sustainable redesign would require more expertise then just the initial designer.

• 8. Ethical

The design of the device in no way advocates ill will towards any person, and is meant for recreation and entertainment purposes, primarily for DJing parties and events. This use brings happiness to many people as well as the user of the device. While party and club atmospheres might also bring negative ramifications such as overindulgence and substance abuse, in keeping with a Utilitarian ethical framework, these incidents are minor when compared to the overwhelming happiness brought to those affected by the device.

With respect towards the IEEE Code of Ethics this project in no way violates any of the Codes. The design of the device will take the utmost precaution in ensuring public safety and immediately inform the public if any safety issues arise. This is in keeping with the number one code of the IEEE Code of Ethics.

• 9. Health and Safety

Lab safety guidelines are followed when designing and testing the device, minimizing many safety concerns, primarily that of electrocution. When manufacturing the device, workplace safety protocols are in effect to minimize workplace injury. Harm from machines used in assembly, and electrical danger, need analytical solutions before device is manufactured. When using the mixer, the user must take caution when around electricity, especially if liquids such as water and other beverages are present. The design of the mixer will respect NEC Safety Codes.

• 10. Social and Political

• Describe social and political issues associated with design, manufacture, and use. Social awareness of the environmental impact of the device effects device design. The manufacture of the device is also affected by this. The use of the product encourages lifestyles that may result in public resistance. While this resistance is misplaced and inappropriate, it must be accounted for. Ultimately the users and those affected by the device are responsible for their own actions and not the concern of the designers or manufacturers. Politically, large international companies in competition with the device may use political leverage to impede the devices' manufacture and distribution. Also, social outcry with regards to event or club lifestyles may result in political action, though this is not a primary concern.
• The project impacts the designers, manufacturers, distributors, and users of the device. The project also impacts any companies in competition with the device by negatively affecting their sales. Direct stakeholders are initial investors and designers of the device. Once in production, those employed by the manufacture and distribution of the device are stakeholders. Indirect stakeholders are other companies and those people who did not consent on the transaction between the designers and users.
• The project benefits stakeholders and employees of manufacture and distribution who profit from sales and make a living. The project benefits users who now have a quality mixer. Indirectly the project harms the competition when negatively impacting their sales. The project negatively impacts investors if the device does not make a profit.
• All stakeholders benefit equally in terms of being associated with the creation of the device. Pay would not be

equal, but based on level of investment and involvement. For example designers would make more than manufacturers. Pay differences could be conceived as inequities but people involved received profit proportional to their contributions.

• **11. Development**

• Through the design, testing, and implementation of the Two Channel Audio Mixer, experience in planning and executing a design project will be gained. Personal knowledge of programming embedded systems and digital signal processing will be expanded. Learning how improvements for a design based on analysis of test results enhances a final product is crucial.

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E. Program Listing

SigmaStudio Graphical Programming Tool – Used for designing the mixer and interfacing with the ADAU1446 Evaluation Board.

Sonar X2 and AN-879 Analyst – Digital Audio Workstation and VST plug-in used to record and analyze the output of the system. Used in Filter and Equalizer sub-system testing.

Room Equalization Wizard – Used to test the Total Harmonic Distortion and Frequency response of the system.

F. Compiler Output Information

Compiler Output:

Sigma Studio Version 3.9. Build 2, Rev 1220

Analog Devices Compiler for the 3rd generation SigmaDSP core.

Build date = 7/24/2013 at 11:32 PM

Summary

(Note: Estimates are based on a 48 kHz sample rate)

Instructions used:

618 (out of a possible 4096)

Modulo Data RAM used (X Memory):

285 (out of a possible 8192)

Non Modulo Data RAM used (X Memory):

0 (out of a possible 8192)

Parameter RAM used (Y Memory):

214 (out of a possible 4096)

Instance	Mips	Inst (max)	Data	Coeff	Other
Crossover1_2	114	114	72	84	
Crossover1	114	114	72	84	

MX1_2	18	18	7	0
MX1	18	18	6	0
MX1_3	18	18	6	0
Beginning	16	16	0	1
SafeLoadCode	15	15	0	7
SW vol 1_4	13	13	7	1
SW vol 1_9	13	13	7	1
SW vol 1_5	13	13	7	1
SW vol 1_2	13	13	7	1
SW vol 1_3	13	13	7	1
SW vol 1_6	13	13	7	1
SW vol 1_8	13	13	7	1
SW vol 1_7	13	13	7	1
VarQ/F Filter1_4	13	13	5	0
VarQ/F Filter1	13	13	5	0
VarQ/F Filter1_2	13	13	5	0
VarQ/F Filter1_3	13	13	5	0
Togg1	10	10	4	1
Togg1_2	10	10	4	1
Togg1_3	10	10	4	1
Input1	8	8	4	0
St Mixer1_4	8	8	2	3
St Mixer1_3	8	8	2	3
Gain1	6	6	2	2
Nx2-1_2	6	6	2	2
Gain1_2	6	6	2	2
Nx2-1_4	6	6	2	2
Nx2-1_3	6	6	2	2

Nx2-1_6	6	6	2	2
St Mixer1_5	6	6	2	2
Nx2-1_5	6	6	2	2
Nx2-1	6	6	2	2
End	5	5	0	0
Interface Read1_2	2	2	1	1
Interface Read1_3	2	2	1	1
Interface Read1	2	2	1	1
ADC In3	2	2	1	0
ADC In2	2	2	1	0
ADC In1	2	2	1	0
ADC In4	2	2	1	0
GPI1	2	2	1	0
Output2	2	2	0	0
GPI2	2	2	0	0
GPI1_2	2	2	0	0
Output1	2	2	0	0
Output4	2	2	0	0
GPI2_2	2	2	0	0
Output3	2	2	0	0
Interface Write1	2	2	0	-1
Interface Write1_2	2	2	0	-1
Interface Write1_3	2	2	0	-1

Subroutines called:

Total		618	618	285	214

(%)   17%   15%   3%   5%

Files written:

program_data.dat - load file for downloading code using ADI loader

hex_program_data.dat - load file for downloading code using microcontroller

ParamAddress.dat - Parameter RAM locations for schematic instances

G. Hardware Configuration

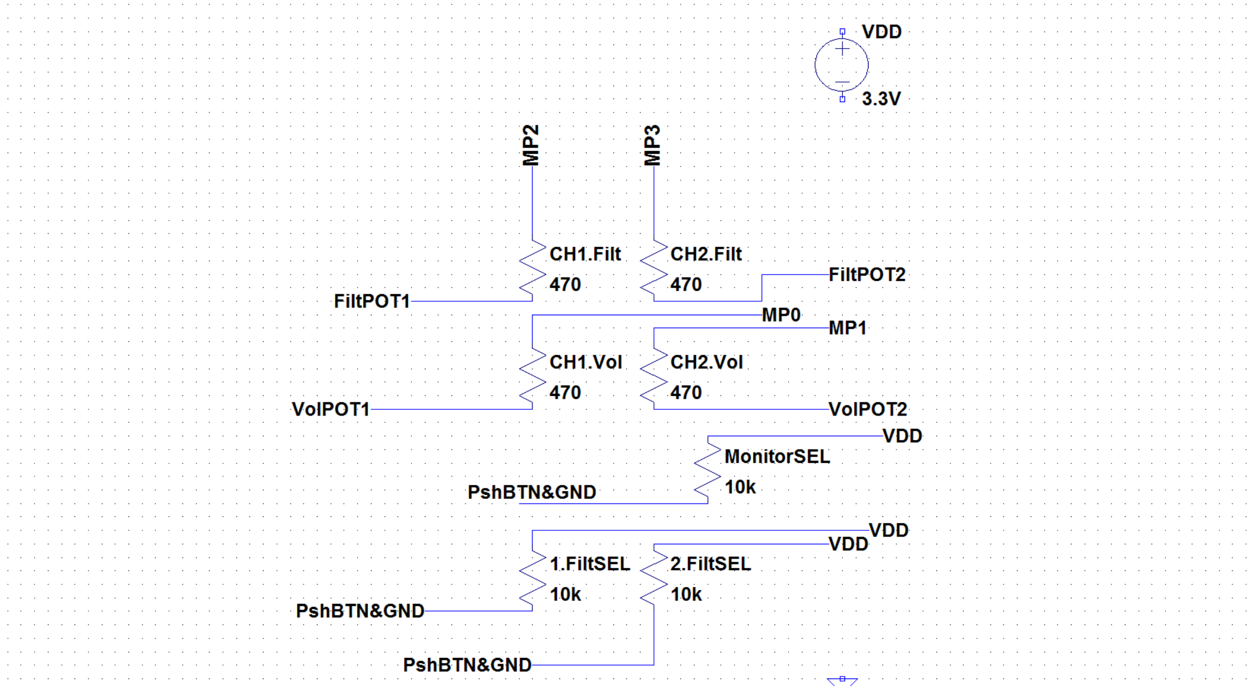

Figure 15: Potentiometer & Pushbutton Board Interface Circuit

Also see Figures 10 and 14 to see the physical locations of the user interface, as well as the location of hardware inside the mixer enclosure.