“Auto-Tuned” Stereo

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LIST OF TABLES AND FIGURES

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I have taken efforts in this project. However, it would not have been possible without the kind support and help of many individuals. I would like to extend my sincere thanks to all of them.

I am highly indebted to the EE Department for their guidance and constant supervision as well as for providing necessary information regarding the project & also for their support in efforts for completing the project.

I would like to express my gratitude towards my parents & Audio Engineering Society for their kind co-operation and encouragement which help me in completion of this project.

ABSTRACT

The purpose of this project was to determine the capability of introducing delay and gain to audio signals to synthesize a “tuned” experience. The project uses audio operational amplifiers, field effect transistors, a class D amplifier, microphones, speakers, a 32-bit microcontroller for design and testing. When implementing the MOSFET into the all pass filter, the phase delay incurred by varying the transistor resulted in minimal change in phase. MOSFETs are used as variable resistors to vary the phase change and gain. By modifying the gain the synthesized distance from each speaker can be adjusted. Vary the phase change provides the remaining time delay matching. The result is calibrated sound system.

I. INTRODUCTION

The following report covers a brief theory of sound and design and implementation of an analog audio amplifier attempting to adjust the phase and gain of individual channels of a stereo system. Digital implementation will also be considered and pursued as an alternative for the analog design. The report will go into detail how phase is significant when dealing with audio signals, and which frequencies matter.

II. BACKGROUND

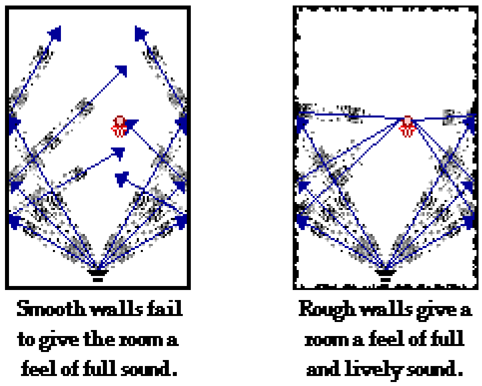

Figure 1 Sound Reflection Example

Sound and music are parts of the everyday sensory experience. The basis for an understanding of sound, music and hearing is the physics of waves. Sound is a wave that is created by vibrating objects and propagated through a medium from one location to another.

When a wave reaches the boundary between one medium another medium a portion of the wave undergoes reflection and a portion of the wave undergoes transmission across the boundary. Figure 1 is an example of single source reflection. The amount of reflection is dependent upon the dissimilarity of the two sources. Multiple sources reduce the effective of reflections towards the observer. Figure 3 shows the final configuration for this project.

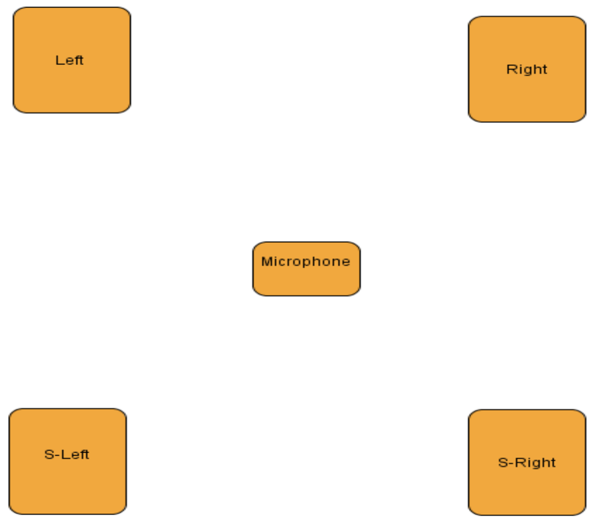

Figure 2 Four Audio Sources for Reflection Negation

Four separate and equally spaced sources create constructive interference inside of the four sources see in Figure 2. The constructive interference is dependent on the amplifiers performance, though with similar distance from each source, phase distortion is easier to correct for.

The audible range for human hearing is between 10Hz ~22kHz. For typical surround sound audio systems, the rear channels are 90º out of phase with the front channels. This creates a delayed effect for media designed to immerse the user with specific audio channels for the rear speakers. This project however is more concerned with all channels initially at the same phase with phase control relinquished to the amplifier.

Sound localization is the process of determining the location of a sound source. The brain utilizes subtle differences in intensity, spectral, and timing cues to allow us to localize sound sources. Human hearing can make valid measurements of phase at frequencies of 800Hz and below while any audio above 1kHz is typically undetectable when trying to determine phase.

The primary reason that human hearing is incapable of discerning phases at those frequencies is due to the wavelength at those frequencies. II.1 is the classical equation for calculation wavelength.

(II.1) $$\lambda = \frac{v}{f}$$

The speed of sound is 340.29 m/s at sea level, in dry air, and at 68º F. Choosing a specific frequency below 800Hz reduces the complexity of the design. Choosing 500Hz results in the following wavelength below:

(II.2) $$\lambda = \frac{v}{f}$$

(II.3) $$\lambda = \frac{\frac{340.29m}{s}}{500Hz}$$

(II.4) $$\lambda = 0.68058m$$

Using this wavelength, determining the phase difference of each channel is simplified by reducing the distance from the each speaker to multiples of wavelengths. Knowing that each wavelength has 360º degrees, calculating degrees of phase is done by equating fractional wavelengths then multiply that number by 360º. For example, for a two channel system, if speaker 1 is 3m away and speaker 2 is 2.5m away. Then using the below relations II.5-8, where K is a multiple of wavelengths, should equate in the phase difference of the two speakers.

(II,5,6) $$\frac{|D_1 - D_2|}{\lambda} = K \qquad K \times 360° = \theta°$$

(II.7,8) $$\frac{3m - 2.5m}{0.68058m} = 0.734667\lambda \qquad 0.734667 \times 360° = 264.48°$$

Note that K is fractional in this case, but it is possible to be larger than whole number multiples. If that were the case, it is appropriate to use the fractional portion, because that is the true phase difference between both sources.

Along with sound localization, sound intensity is also another consideration when determining distance from the source to observer. The louder a source sounds, the closer it appears to be, the opposite is true as well. Intensity will be the volume of the source in this instance. The intensity is proportional to the pressure squared. The pressure, vibrates air particles to produce a force on a membrane, is inversely proportional to the distance from the source. Using the below relationship (I.9,10) provides a connection between speaker volume and relative distance.

(II.9,10) $$p \propto \frac{1}{r} \qquad I \propto p^2$$

Utilizing a microphone it is possible to determine distance. The microphone outputs a voltage that corresponds to the membrane vibration. The louder the source becomes, the higher the output voltage becomes. Unfortunately using a single microphone can only measure intensity from a single source successfully with little to no background noise. This indicates that multiple

microphones will be needed if all channels are to be measured and modulated at the same time. It is possible to measure all channels with a single microphone, though they will have to be measured at separate times so they don't interfere with the others measurement. Figure 3 will be the measuring apparatus, using four identical and equally spaced microphones.

Figure 3 Measuring 4-Channels Simultaneously with 4 Microphones

III. REQUIREMENTS AND SPECIFICATIONS

Four-Channel Audio Amplifier

- Audio Frequency Range:10Hz~22kHz
- Current Output: 4 Amps
- Power Output: 20~45Watts
- 4 Channel Input
- 4 Channel Output
- Mute Function
- Standby Function
- Equal Channel Length
- 4 Ohm Speaker Load
- ±12V Supply
- 90º Maximum Phase Shift ≈ 500us at 500Hz

Atmel EVK1100 Development Board

- 512 kB Flash
- Up to 60 Mhz Operations
- Low Power Consumption
- 5V Tolerant I/O
- 8 Configurable Channels (8/10 bits)
- 2.54mm Wrapping Area

Ti DSK C5416 DSP processor

- 48kHz Sampling Rate
- 2 Channel Input
- 2 Channel Output
- 144 I/O Pins
- Fast Return From Interrupt

Electret Microphone Board

- 2.7V up to 5.5V VCC
- Onboard OPA344 Operational Amplifier

IV. DESIGN

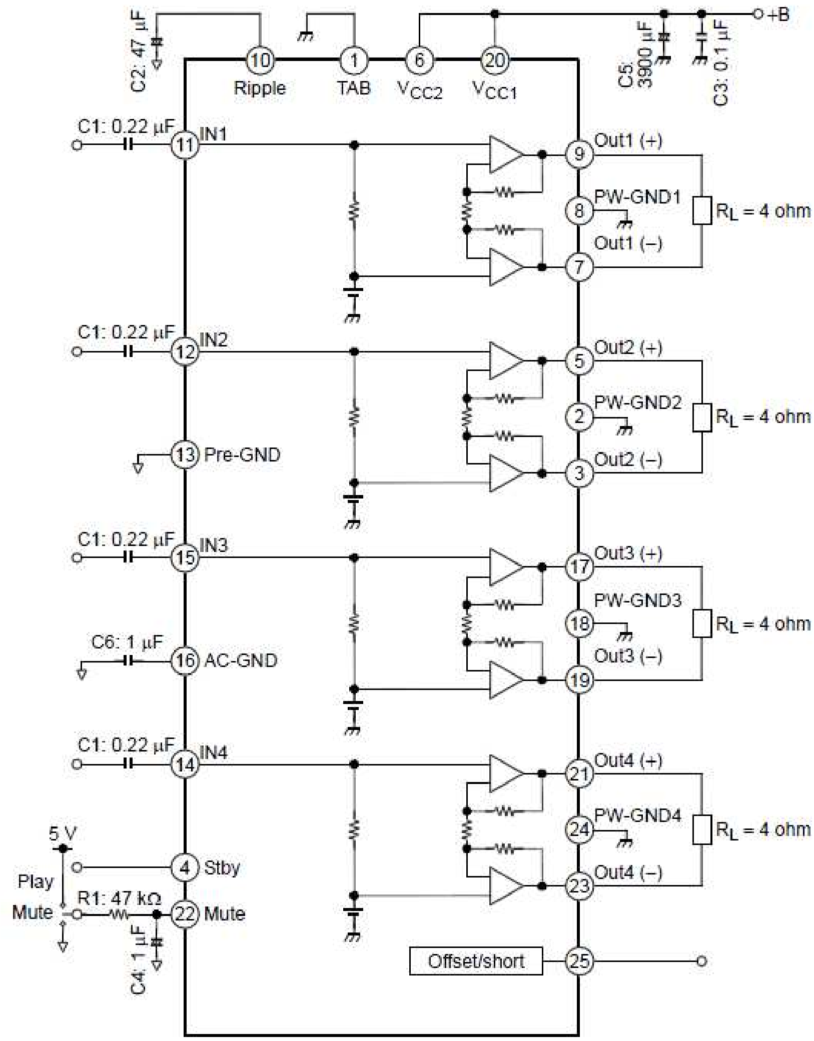

Figure 4 Test Circuitry for Class D Amplifier Provided by TB2939HQ Datasheet

Designing an audio amplifier requires input and output impedance compensation as well as moderate power output. A dedicated audio power amplifier is used to accommodate for the previously mentioned requirements. A class D amplifier manufactured by Toshiba, TB2939HQ, provides the appropriate input and output impedance loading solutions as well as moderate power output with 26dB gain. Using the manufacturer's suggested circuit in Figure 4 provides a low noise amplifier with programmable offset and mute functionality.

To design the phase delay component of the amplifier requires and Allpass filter. Figure 5 is the active variant of an Allpass filter.

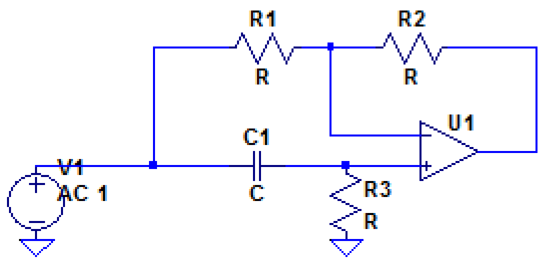

Figure 5 Active Allpass Filter Topology

The allpass filter has the following transfer function, magnitude response and phase response respectively:

(IV.1) $$H(s) = \frac{sRC - 1}{sRC + 1}$$

(IV.2) $$|H(j\omega)| = 1$$

(IV.3) $$\angle H(j\omega) = 180^\circ - \tan^{-1}(\omega RC)$$

In Figure 5, R1 and R2 are equal for a ~0dB gain and R3 and C1 determine the frequency at which the phase shifts 90º. For the application in this project, recommended design frequencies lay between 100-800Hz. As previously mentioned, 500Hz is an ideal frequency to design for. Using IV4-5:

(IV.4) $$\omega = \frac{1}{R_3 C_1} \qquad \omega = 2\pi f$$

(IV.5) $$500Hz = \frac{1}{R_3 C_1 2\pi}$$

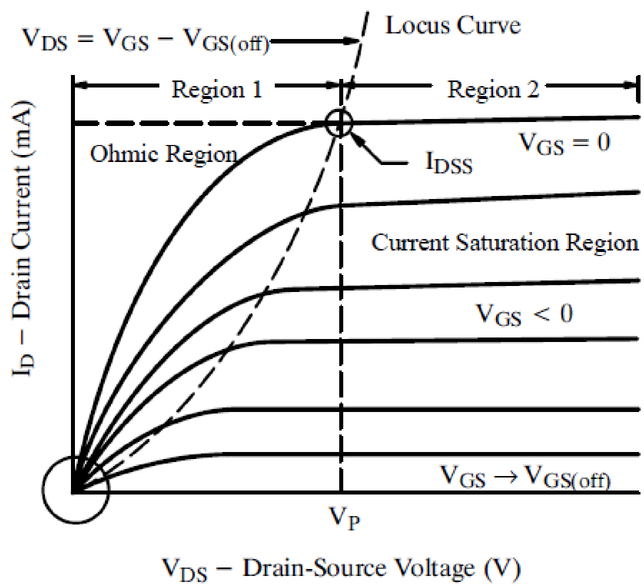

Figure 6 MOSFET Behavior Showing Drain Current vs Vds with Various Vgs in Ohmic and Saturation Regions

Designing this filter requires an R and C value to equate in a corner frequency of 500Hz. In this project R is determined by the behavior of the MOSFET. The impedance the MOSFET exhibits provides a variable behavior in its ohmic region where the current flowing from drain to

source is linearly related to drain source voltage. Figure 6 shows the region of operation the project will be focused on.

Now that the MOSFET is realized as variable resistor, we use the new variable allpass filter.

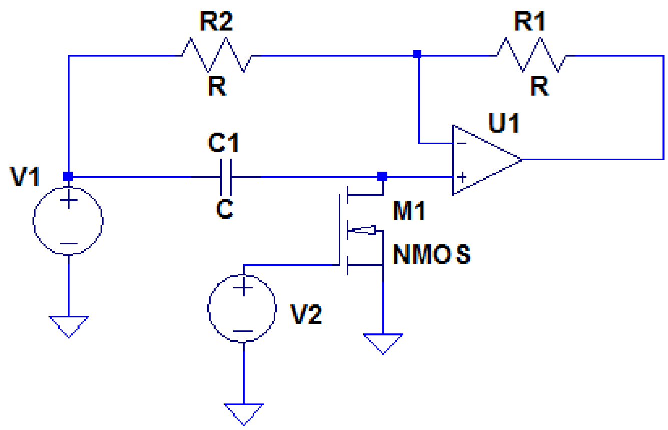

Figure 7 Active Allpass Filter with Varying Phase, NMOS as Variable Resistor

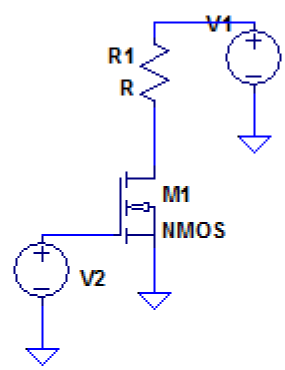

Figure 8 Simple Voltage Divider with Varying Resistance, NMOS

To regulate gain coming out of each channel, a voltage divider is used with an additional MOSFET in resistor configuration in Figure 8. Figure 8only provides attenuation for as low of impedance the MOSFET can reach. This configuration would suffice for as long as the other

channels needed to be attenuated. Although the possibility that could arise that signals would need to amplified as opposed to attenuated, the former requires the circuitry in Figure 9. Figure 9 requires additional MOSFETs and increases the potential of clipping when entering the class D amplifier, but provides greater accuracy when matching gains of each signal. Figure 8 provides more simplicity and circuit reliability, so it is the chosen design in this project.

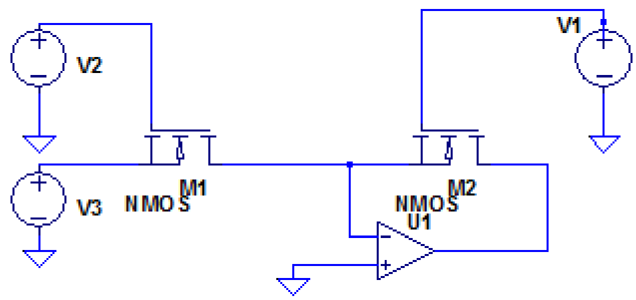

Figure 9 Active Non-Inverting Amplifier with Two NMOS as variable Resistors

Choosing an appropriate MOSFET with large enough effective resistance is critical. Choosing ALD1107 integrated IC with four NMOS transistors, reveals impedance with the trend IV.6

(IV.6)
$$R = (6800 \times Vgs) + 5000$$

With the lowest impedance being 5kΩ, now the previously unknown C1 in Figure 7 is calculated using IV.5.

(IV.7)
$$500Hz = \frac{1}{5000\Omega C_1 2\pi}$$

(IV.8) $$C_1 = 63.66nF$$

Using industry standard values, C_1 is 0.064uF. Assigning R1 and R2 to 10kΩ provides a steady response with an even ~0dB gain across the audio range of frequency.

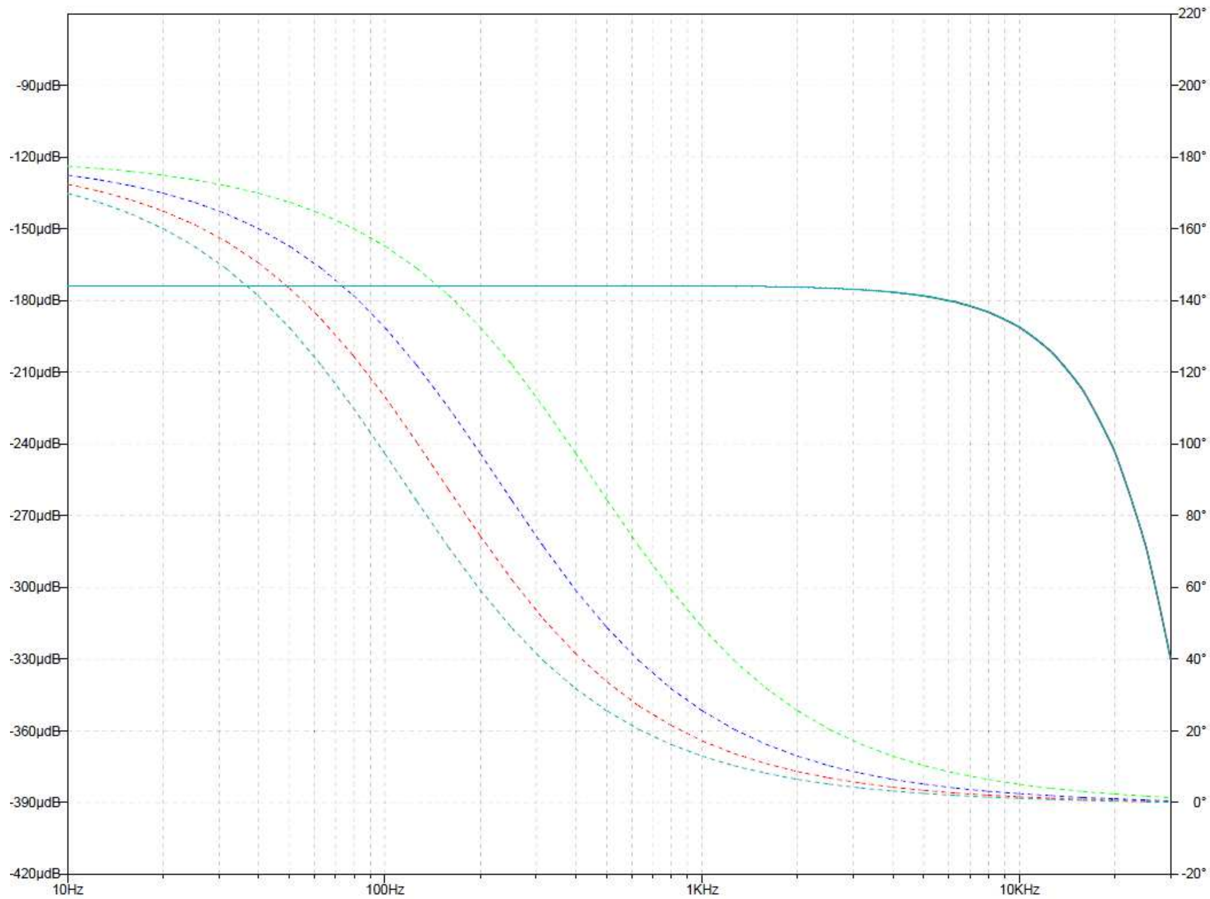

Figure 10 Magnitude and Phase Response of Allpass Circuit in Figure 7. Using MOSFET Impedance and C1 = 63.66nF: 5KΩ(Green) 10KΩ(Blue) 15KΩ(Red) 20KΩ(Teal)

Figure 7 shows that increasing the resistance from 5k-20kΩ causes the phase to shift nearly linearly around 500Hz and surrounding frequencies. Higher than 10kHz, the phase shows minuscule change. As stated in Section II, frequencies above 1kHz are nearly indistinguishable to the human ear. So there is no need for concern to change the phase at those negligible frequencies.

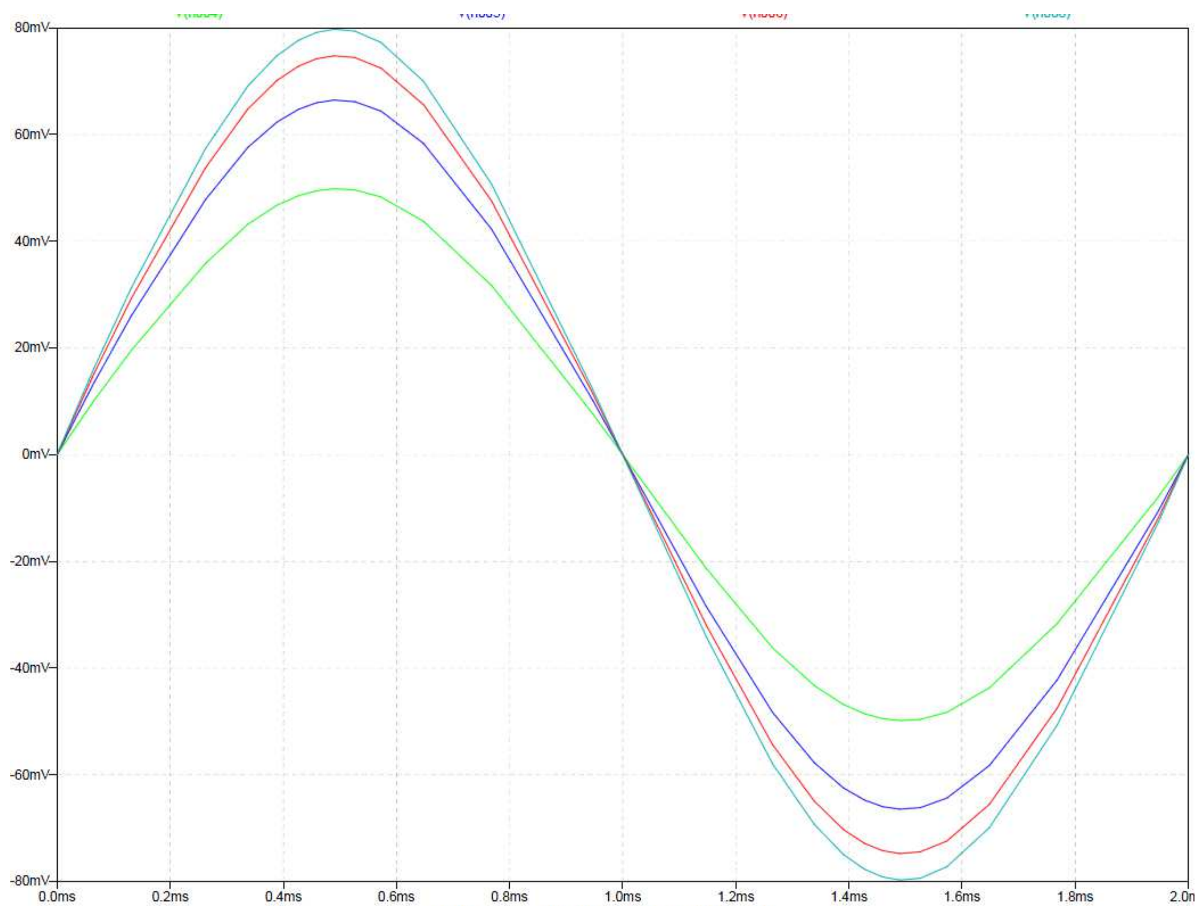

Figure 11 Simulated Voltage Division from Circuit in Figure 8 with Varying Resistance R3: 5KΩ(Green) 10KΩ(Blue) 15KΩ(Red) 20KΩ(Teal)

Vgs(V)	≈Impedance(Ω)
0	5000
0.1	5680
0.2	6360
0.3	7040
0.4	7720
0.5	8400
0.6	9080
0.7	9760
0.8	10440
0.9	11120
1	11800
1.1	12480
1.2	13160
1.3	13840
1.4	14520
1.5	15200
1.6	15880
1.7	16560
1.8	17240
1.9	17920
2	18600
2.1	19280
2.2	19960
2.3	20640
2.4	21320
2.5	22000

Table III Ohmic Region MOSFET Equivalent Impedance

Table I shows that the resistance needed for R_1 in Figure 8, so that the attenuation is sweeps over a broader range. Using IV.9 and 5kΩ for R_1 provides attenuation from 6dB to 1.78dB in its ohmic region.

(IV.9) $$\frac{Vout}{Vin} = \frac{R_{MOSFET}}{R_{MOSFET} + R_1}$$

From IV.9 and Figure 11, it is clear that the voltage division is non-linear, except that when dealing with similar resistance values and R_{MOSFET}'s limited range, the relationship with Vgs and impedance from Figure 11. is nearly linear. When Vgs is highly saturated, R_{MOSFET} becomes extremely large and essentially Vout ≈ Vin.

Figure 12 provides a basic block diagram of the communication between audio amplifier, EVK110 microcontroller, and the Electret microphone.

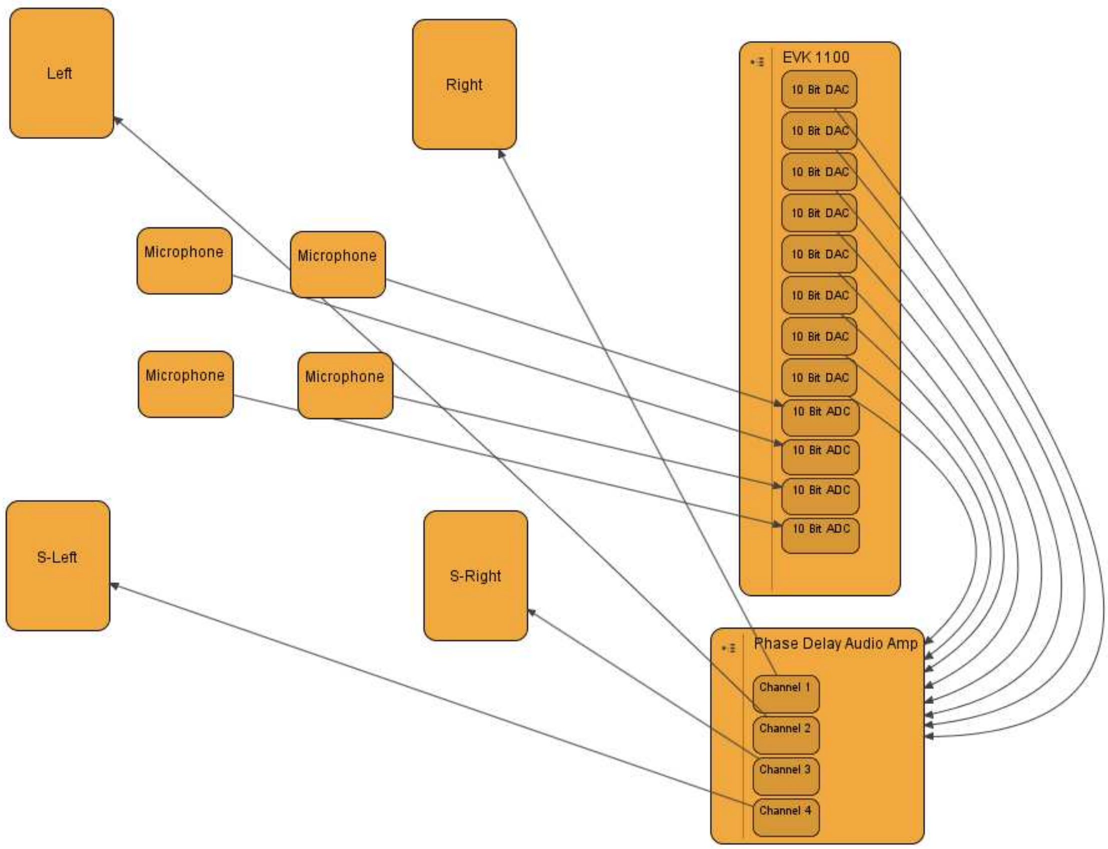

Figure 12 Block Diagram of Overall Project

They microphones feed analog voltages into the microcontroller's ADC, the program loaded onto the microcontroller determines the output voltage from each of the DACs. Those voltages range from 0~2.8V reliably. Each voltage is applied to a gate of each of the MOSFETS.

In order to saturate the MOSFET, the gate voltage must reach higher than 5.2V. Figure 13 is a non-inverting amplifier with the necessary gain of 6.02dB to swing the DAC voltages high enough to induce saturation

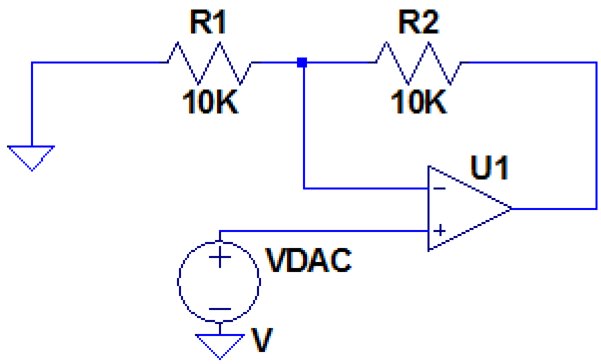

Figure 13 Non-Inverting Amplifier for Vgs input

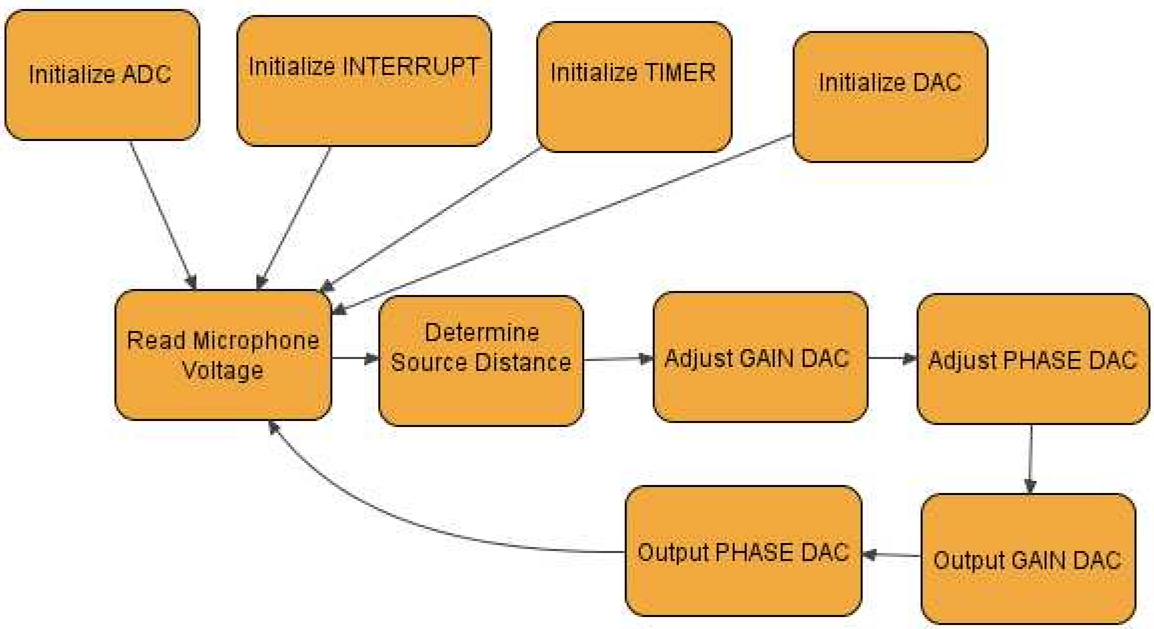

Figure 14 Chain of Project Operations

The order of operations between all three components is simplified in Figure 14. Initially the microphones output a voltage corresponding to the volume levels of speaker; then the microcontroller converts that analog voltage for comparison of each microphone. The appropriate Vgs levels are calculated to accommodate the gain and phase needs. These Vgs values are translated from digital back to analog values. The process then repeats itself as quickly as possible. Example code is for performing ADC is provided on page p46.

An alternative implementation of this project's objective can be envisioned digitally. Incorporation of Texas Instruments DSK 5416 signal processing board allows for "effective real time" delay and gain matching as the gain and time delay are added when sampling the audio then outputting. Figure 15 shows realization of the digital "auto-tuned" stereo. Figure 41 shows the final design schematic.

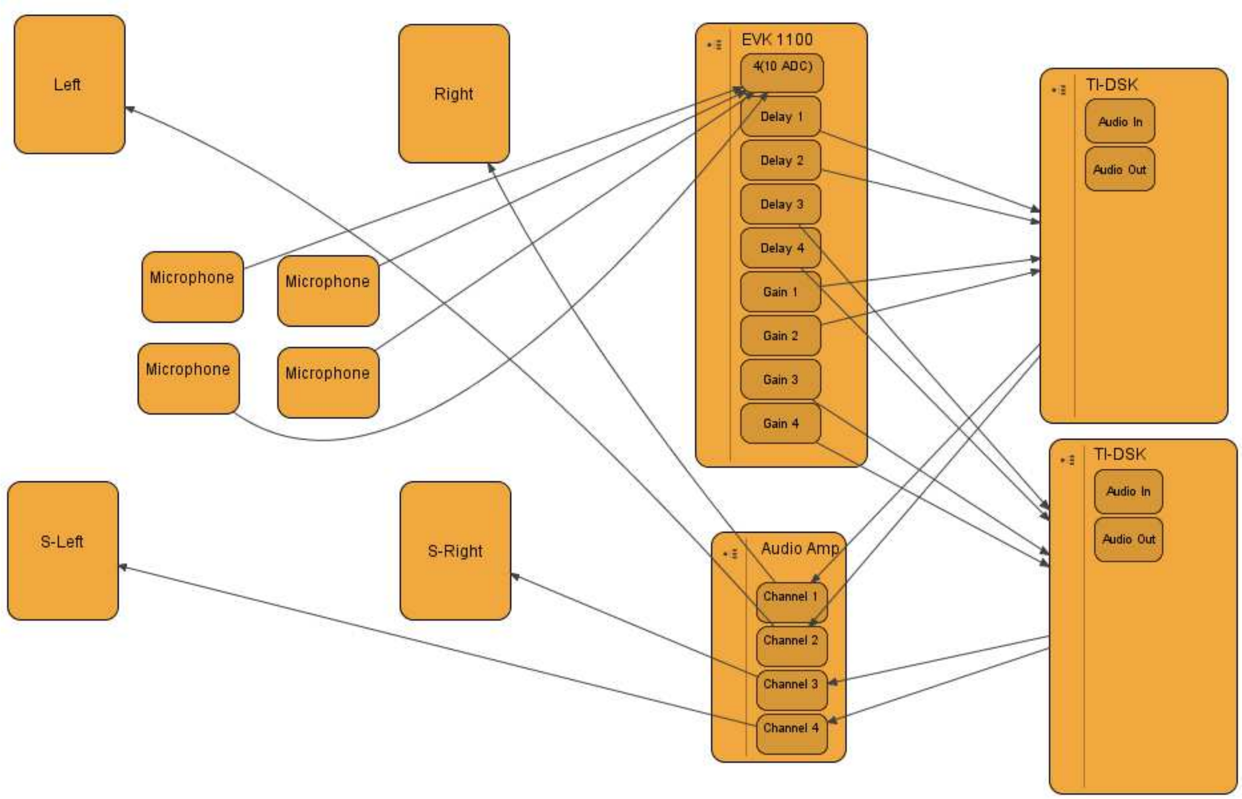

Figure 15 Block Diagram of Digital Realization of Project

Figure16 shows the order of instructions occurring between microphone, EVK100, and C5616 DSK. Example code digital delay and gain is given on p48-9.

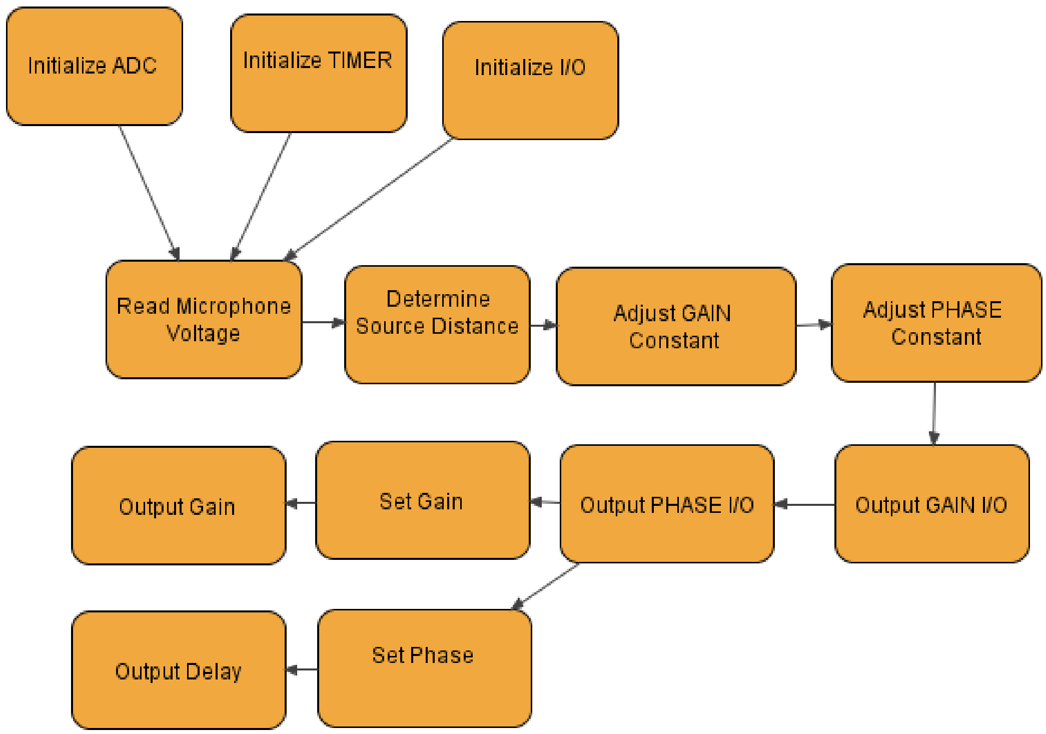

Figure 16 Chain of Operations for Digital Realization of Project

V. CONSTRUCTION

The construction of audio amp resulted in ordering a custom made PCB and prototyping on development bread board. Figures 17-20 display: 4-Channel audio amp, 2 microphone ADC instrument, 2-Channel MOSFET DAC control, EVK1100 implementation, analog 5-band equalizer.

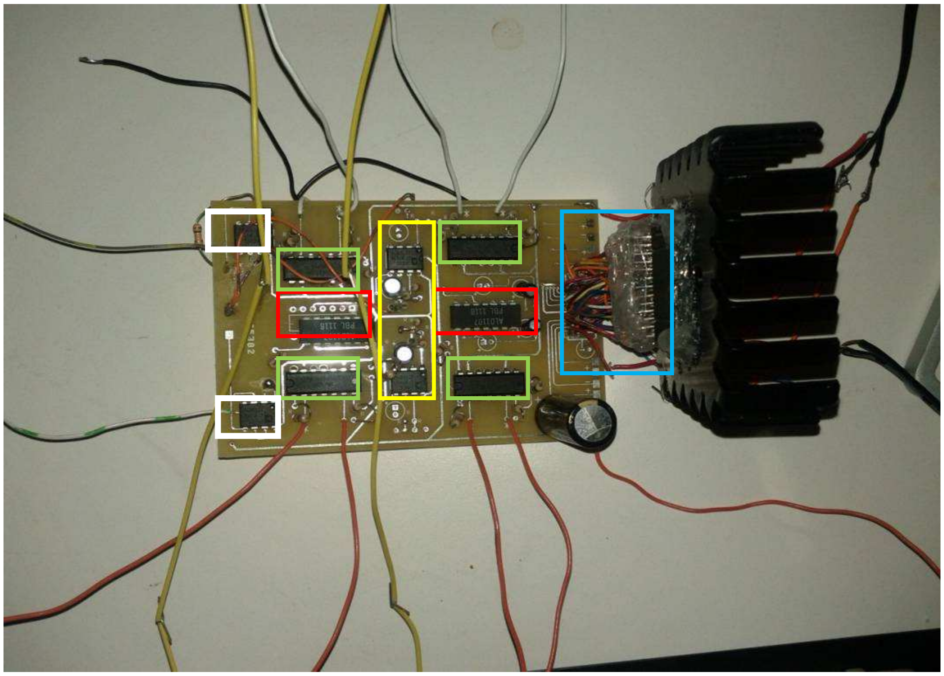

Figure 17 Constructed Audio Amplifier: Voltage Follower(White) Non-Inverting Amplifier(Green) MOSFET(RED) Allpass Filter(Yellow) Class D Amplifier(Blue)

Supply voltages supplied by Agilent dual power supply E3630A. Current output is limited to 1.5A, so two power supplies must be connected in series as to provide 3 A.

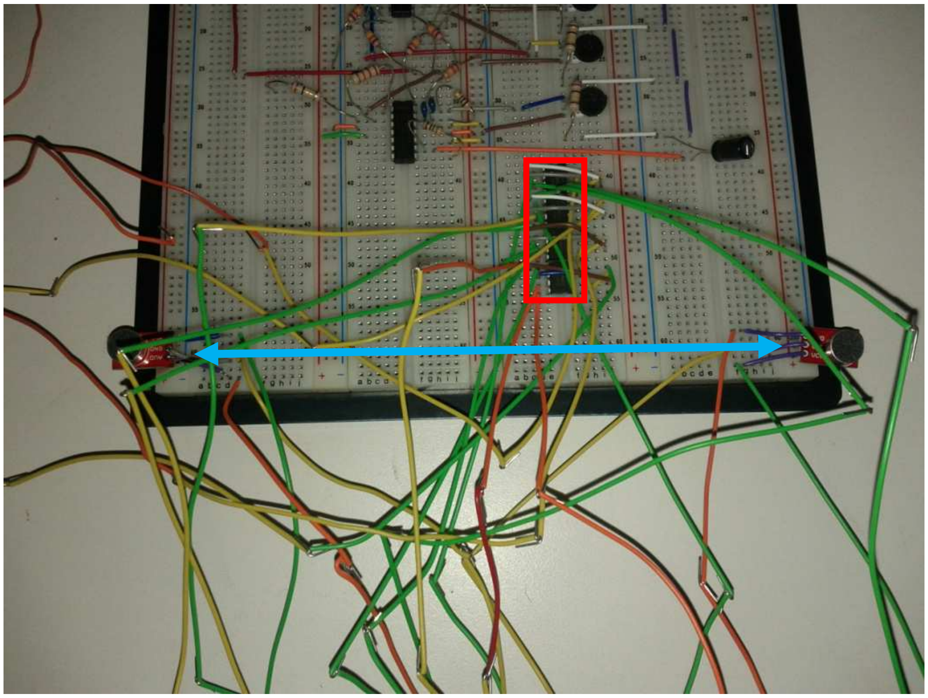

Figure 18 Four Constructed 10-Bit DACs (Red) Dual Microphone Measurement(Blue, ~20cm Separation)

Figure 18 shows DAC implementation and microphone measurement apparatus. This only represents a two channel setup. Four channels would require eight DACs with two additional microphones spaced 20cm apart.

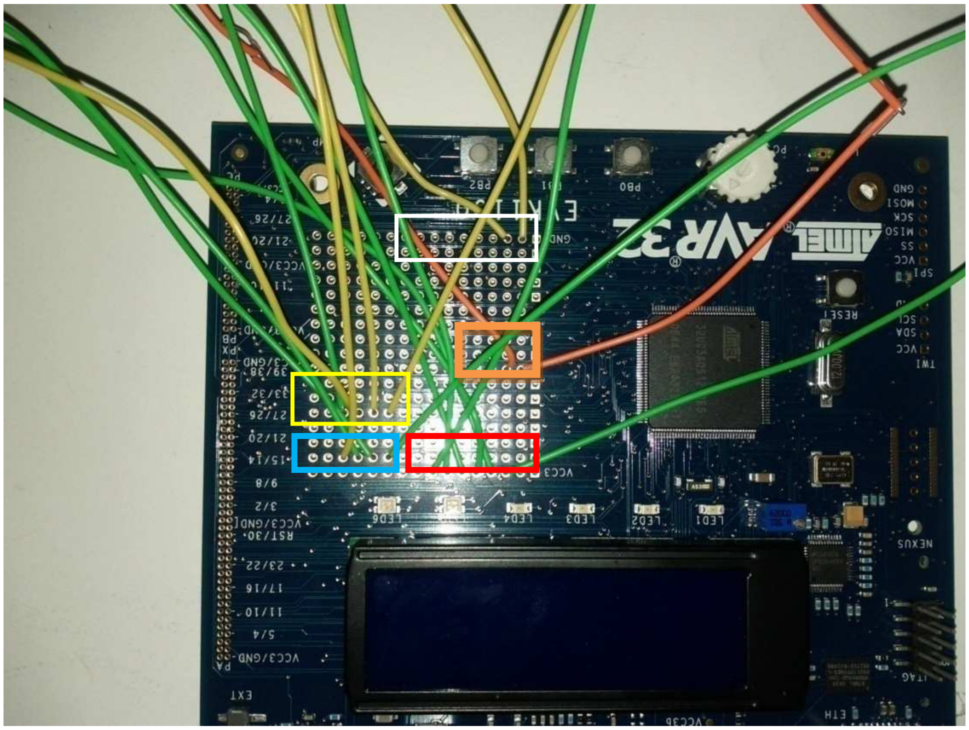

Figure 19 EVK1100 Development Board Example Setup: Serial Input(Blue) Serial Clock(Yellow) Two 10-Bit ADC (Orange) VDD/VREF (Red) VSS/GND (White)

Figure 19 is an ideal connection between DAC and microphone. Unfortunately the complete test did not occur as more time was spent trying to debug the amplifying and phase changing component.

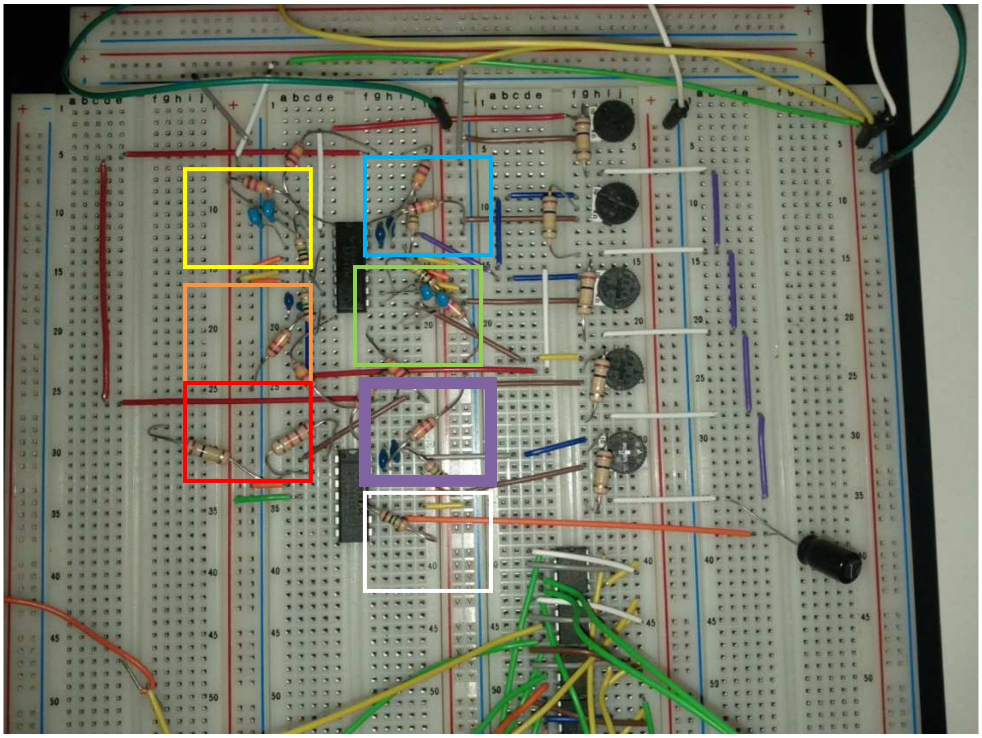

Figure 20 5-Band Analog Equalizer with 20dB gain, band set to: 32Hz(Yellow) 125Hz(Orange) 500Hz(Blue) 2kHz(Green) 8kHz(Purple) -14dB Gain Input(Red) 20dB Gain Output(White)

MOSFETs where not considered over potentiometers, due to simplicity of analog thumbwheels the equalizer remains completely analog.

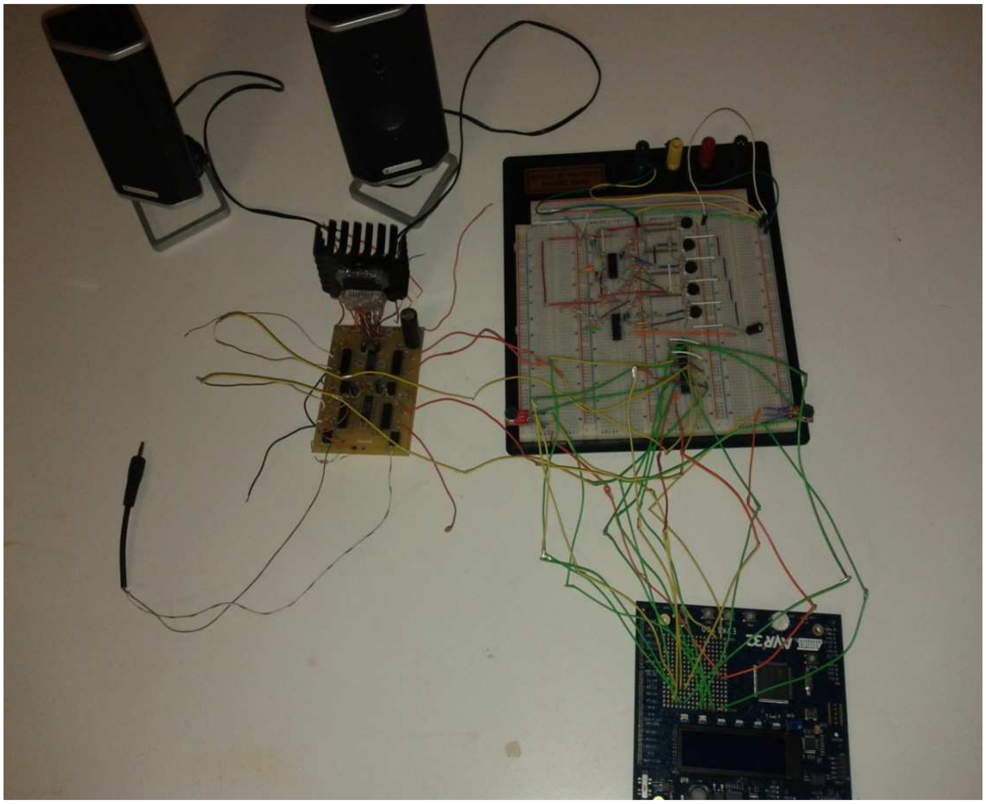

Figure 21 Overview of 2 Channel Project Implementation

This project uses ExpressPCB© and ExpressSCH© to design the board layout. Schematic and PCB layout can be found on p40 and p43-4. All components were through hole, so to make testing and debugging simpler on a development bread board. Hot glue was chosen as an electrical insulator and heat sink adhesive to keep the class D amplifier within heat constraints.Due to lack of time, constructing the dual DSK board configuration was incomplete and unavailable to be photographed. Example algorithm code for DSK board implementation is on p60

VI. TESTING

Equipment

- Agilent Oscilloscope DSOX2014A
- Agilent Dual Power Supply E3630A
- Agilent Multimeter 34401A
- Development Bread Board
- Scope Probes
- Various Connection Cables(Banana, Grabber, BNC, etc.)
- Variable Resistor Box
- Soldering Iron
- Hot Glue Gun

Materials

- Assorted Wire
- Solder
- Solder Flux
- Various Resistors ½ Watt Rating
- Various Capacitors 10-50V Rating
- ALD1107 nMOSFET
- OPA35 Audio Amplifier
- U747 Operational Amplifier
- ICL7842 Audio Amplifier
- TB2939HQ Class D Audio Amplifier

Voltage Follower

Figure 22 uses OP275 to construct a voltage follower. The inverting input connects to both input and output with the non-inverting input grounded providing the circuit in Figure 22. Figure 10 shows data collected for Figure 22.

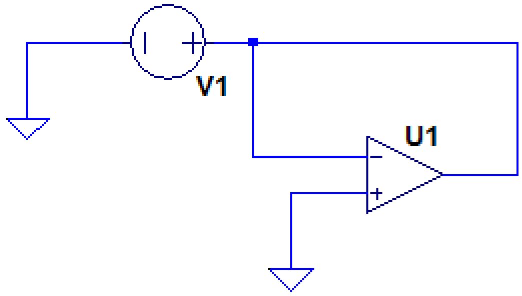

Figure 22 Voltage Follower Circuit, to Prevent Input Loading

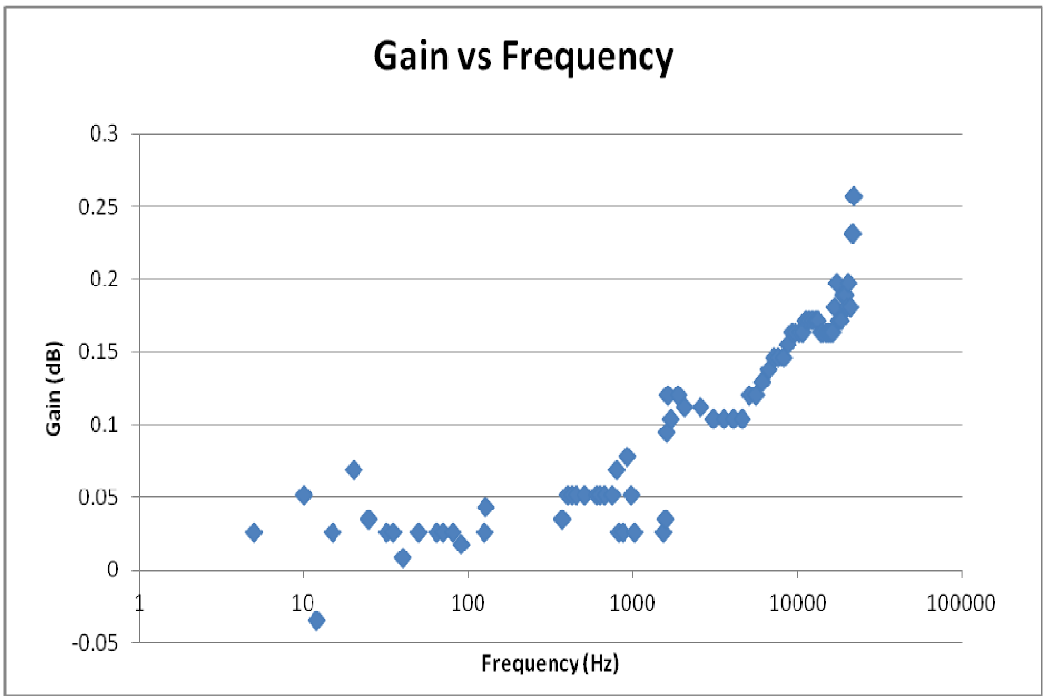

Figure 23 Voltage Follower Magnitude Response: 100mVpp Input

Figure 23 shows the voltage follower behaving within ~15% of its required specifications. Though at higher frequencies the amplifier starts to become less reliable, but still within acceptable range in desired audio range.

Allpass Filter & Voltage Divider

Using OP275 and ALD1107 ICs with dual voltage supplies at ±10V and the circuit diagram provided in Figure 7and 8 creates an isolated Allpass filter with attenuation. Testing the Allpass (filter resulted in the following captures seen in Figures 24-29.

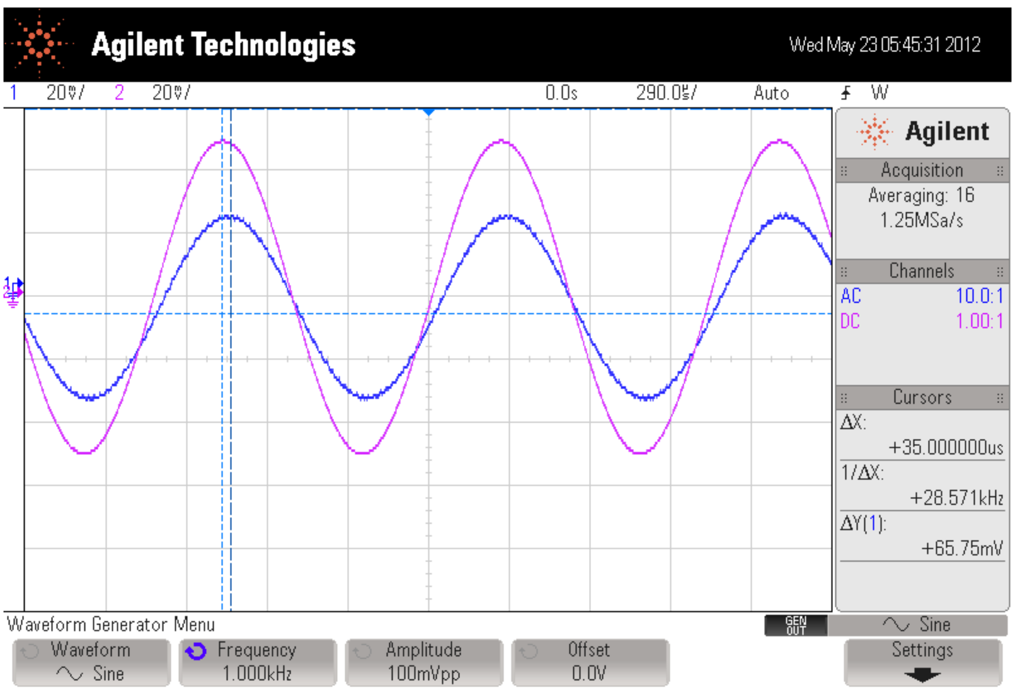

Figure 24 Output at Voltage Divider: 1kHz input 100mVpp 35us Delay

Figure 24 represents 0V at the gate of the MOSFET, providing an equivalent resistance of 5kΩ. Even though a phase change at this frequency is nearly negligible, it is important to note overall performance of the allpass filter.

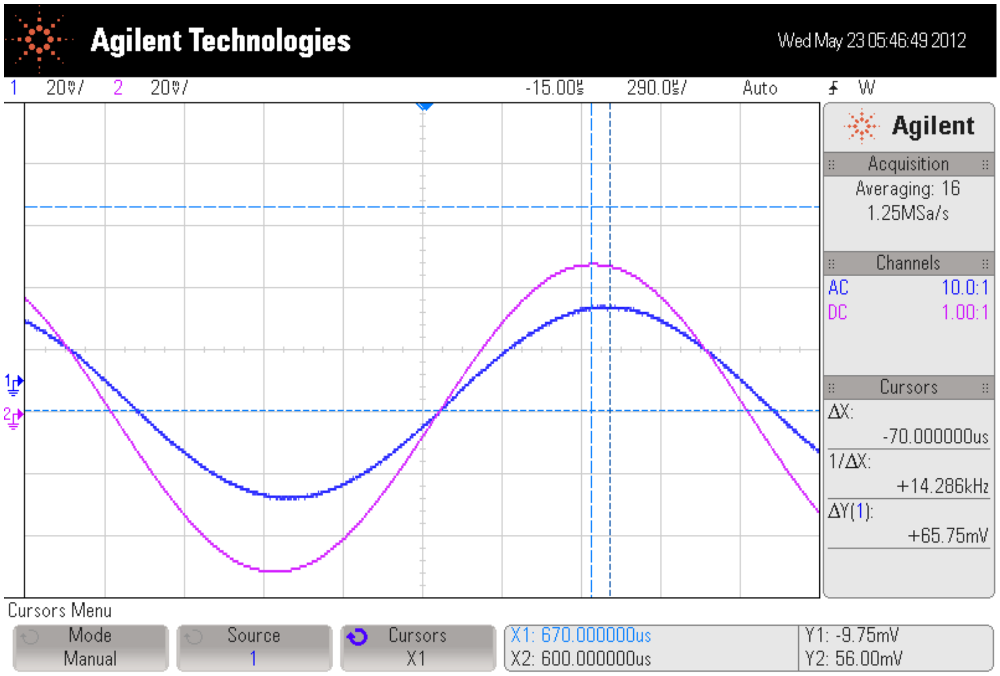

Figure 25 Output at Voltage Divider: 1kHz input 100mVpp 70us Delay

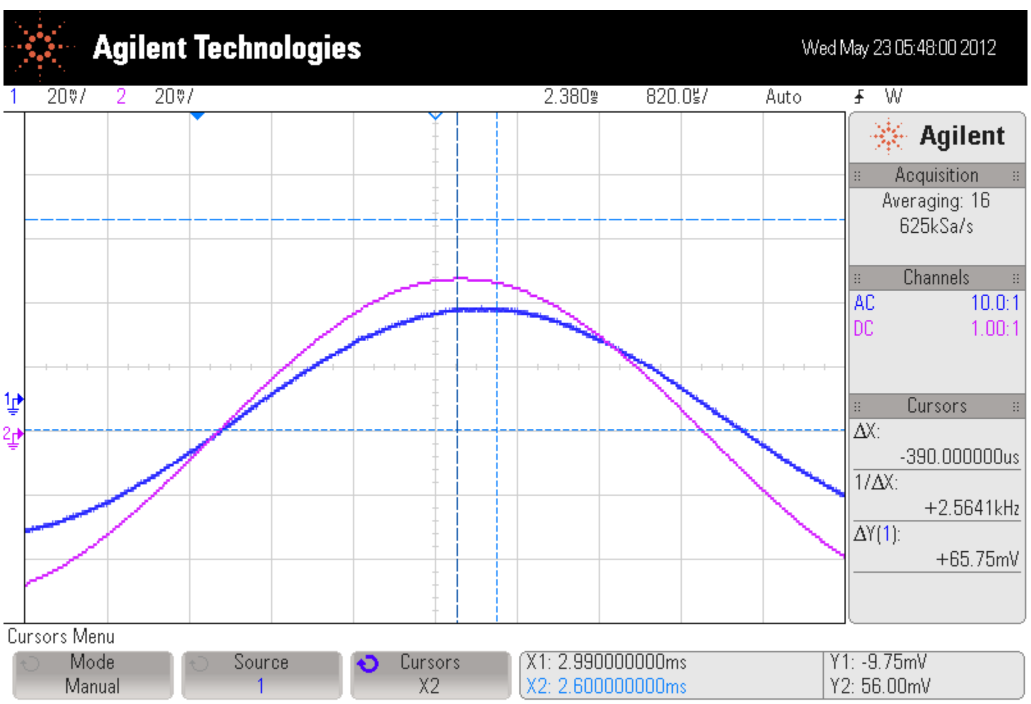

Figure 26 Output at Voltage Divider: 500Hz input 100mVpp 390us Delay

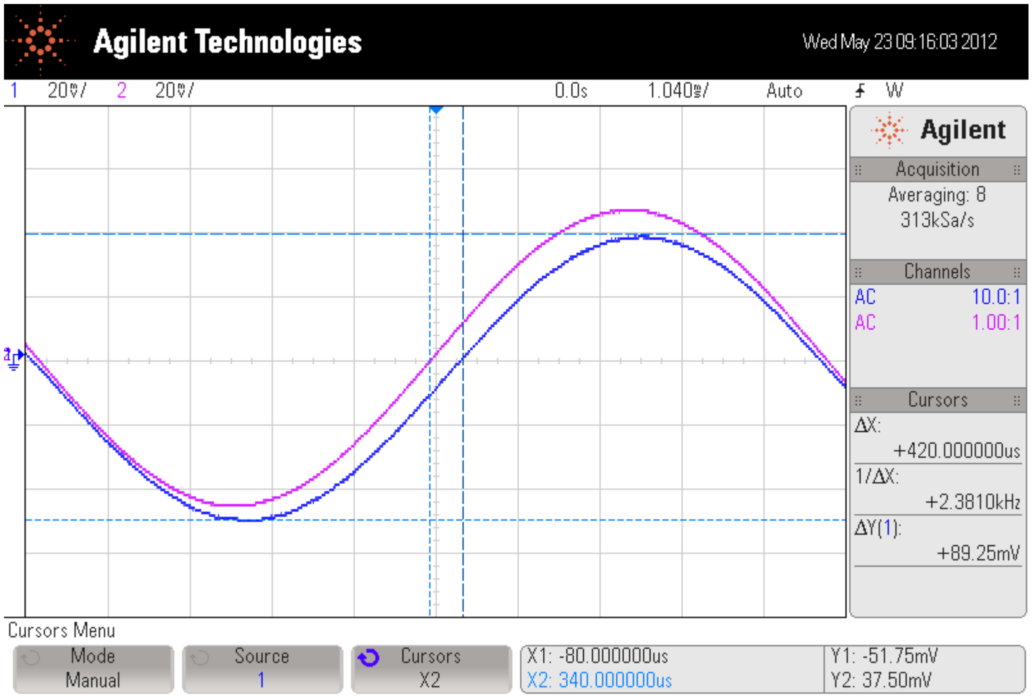

Figure 27 Output at Voltage Divider: 500Hz input 100mVpp 420us Delay

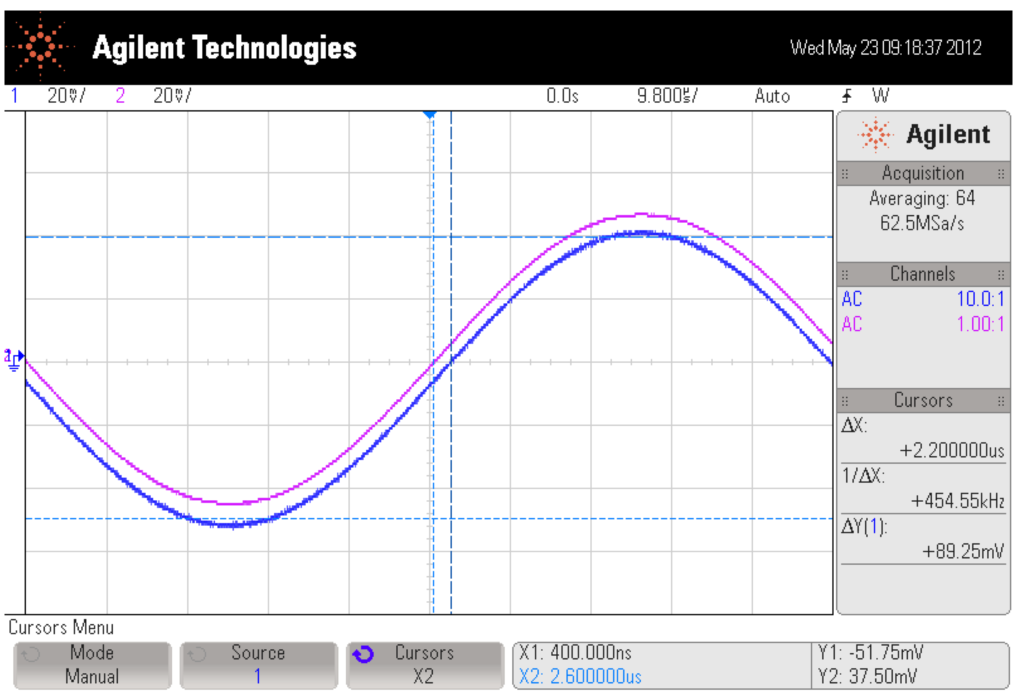

Figure 28 Output at Voltage Divider: 10kHz input 100mVpp 2.2us Delay

	500Hz		1000Hz		10kHz	
Vgs (V)	Delay (us)	Gain (dB)	Delay (us)	Gain (dB)	Delay (us)	Gain (dB)
0	390	-6	35	-5.8	2.2	-1.4
Saturation	420	-2	70	-4	-	-

Table IV Phase and Gain Characteristics Varying with Vgs Extremes. 10kHz Saturation Data Had no Effect

The practical tests showed relatively similar behavior from the simulation, that being increase in resistance results in phase change. As well as increasing Vgs to lessen the attenuation of signal at the voltage divider. The change is quite small, so the effectiveness of these allpass filters is questionable.

Class D Amplifier

Using the test circuitry provided by the TB2939HQ datasheet, shown in Figure 4, with a 12V supply, signal generator, and oscilloscope creates the testing environment for the class D amplifier. Figure 28-33 display the scope images from testing the class D amplifier. Applying a heat sink to the amplifier ensures optimal operating range, as the max output current is 4Amps.Tieing the 12V supply to the active low mute and standby functions disables, allowing regular operation.

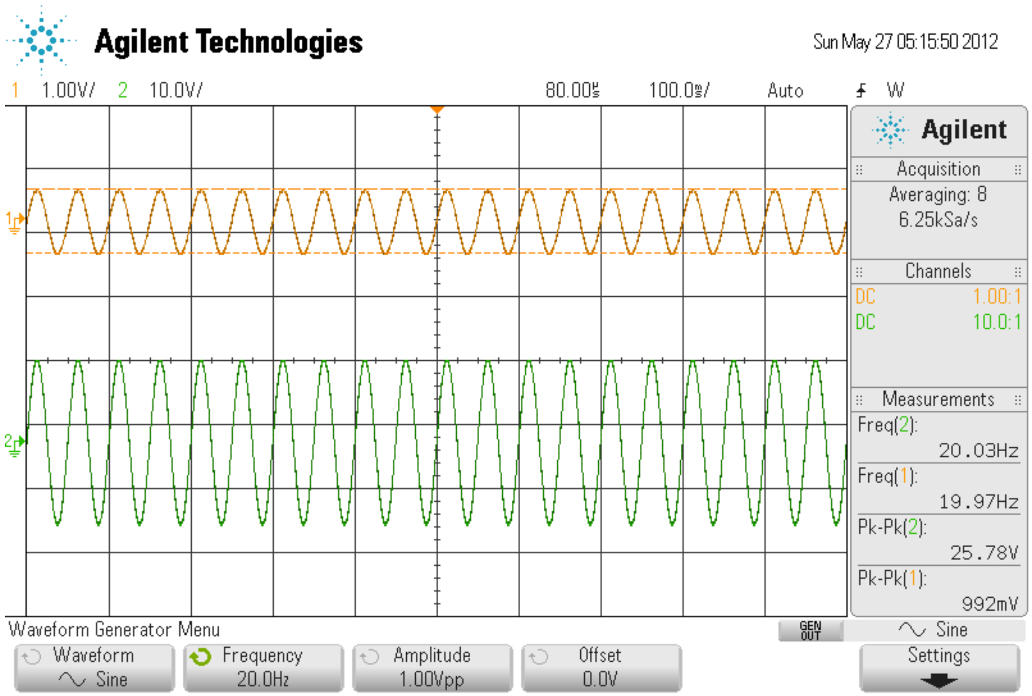

Figure 29 Class D Amplifier Freq. Response: 20Hz

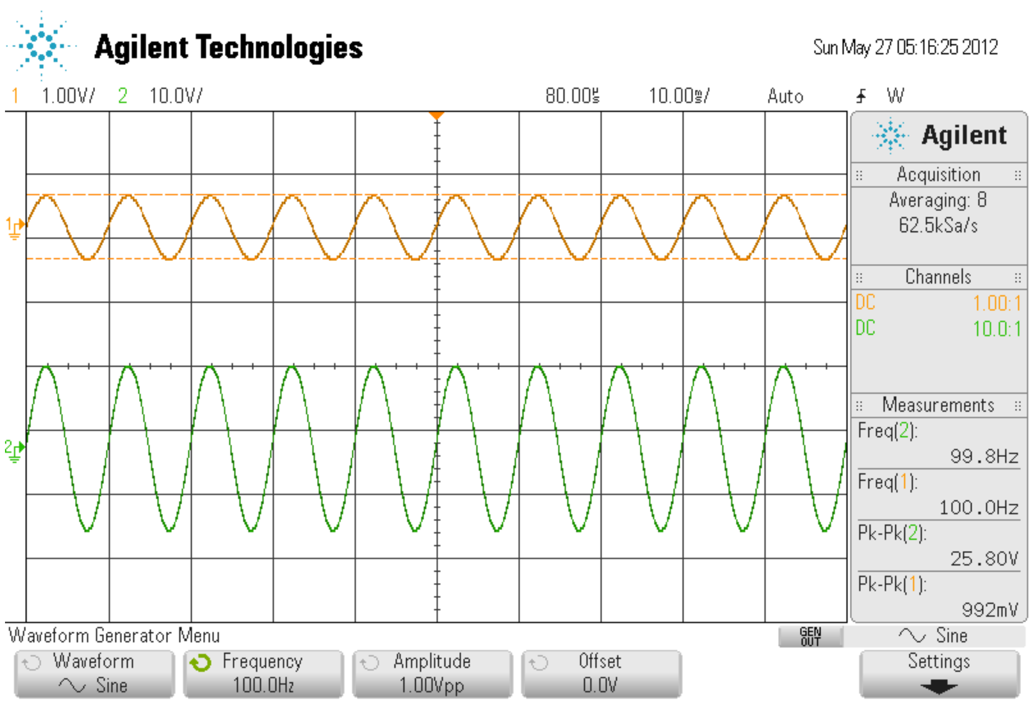

Figure 30 Class D Amplifier Freq. Response: 100Hz

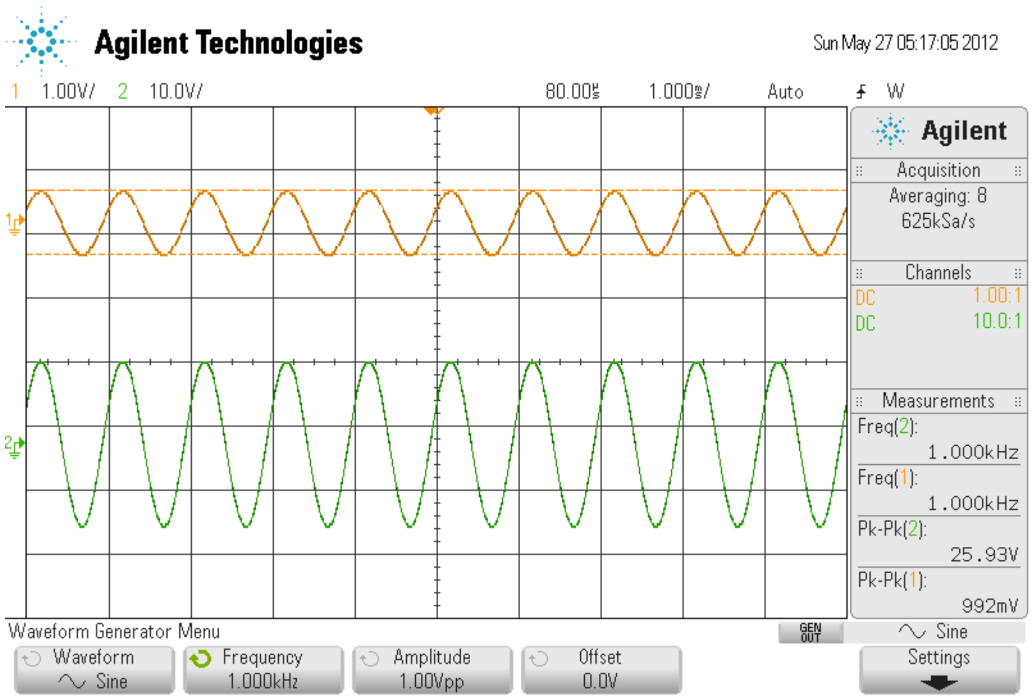

Figure 31 Class D Amplifier Freq. Response: 1kHz

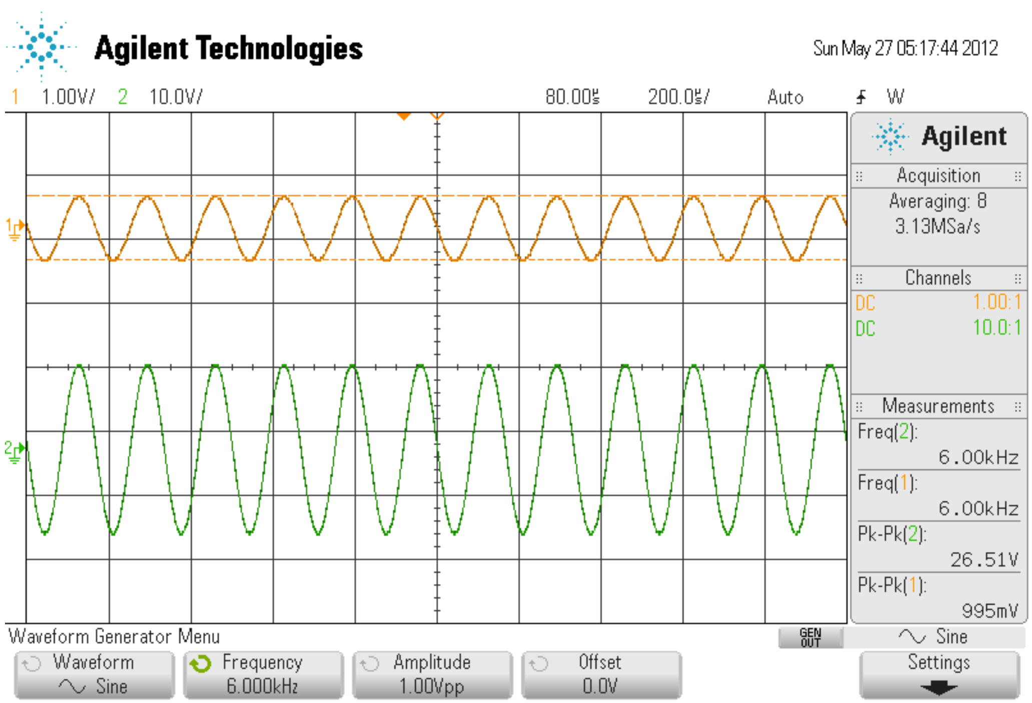

Figure 32 Class D Amplifier Freq. Response: 6kHz

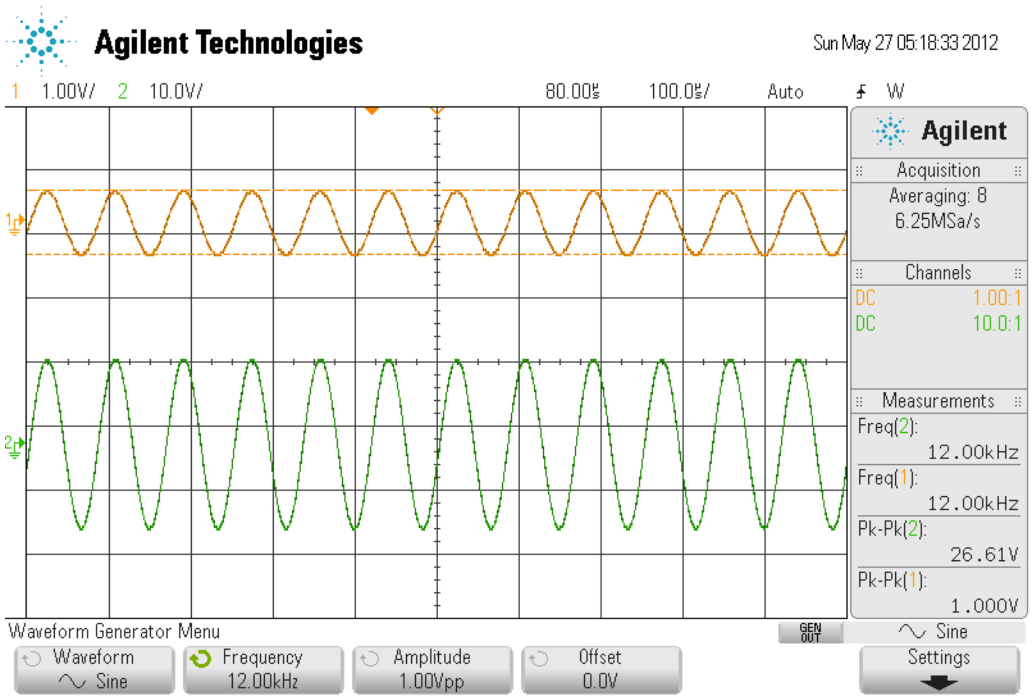

Figure 33 Class D Amplifier Freq. Response: 12kHz

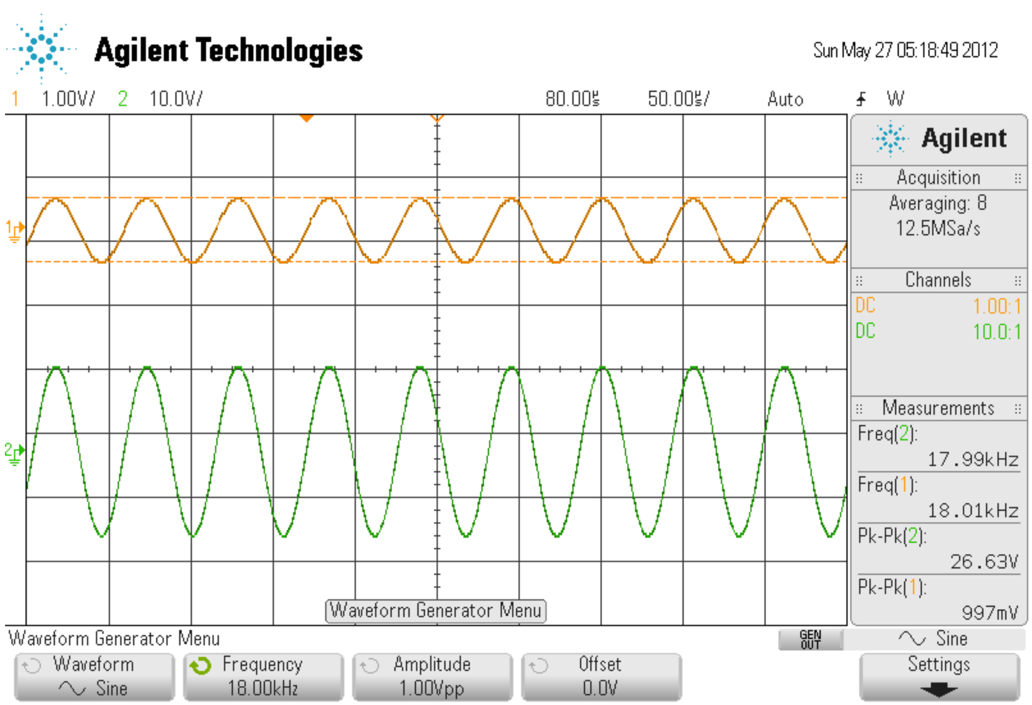

Figure 34 Class D Amplifier Freq. Response: 18kHz

f (Hz)	input(V)	output(V)	Gain (dB)
20	0.992	2.578	8.295425
50	0.992	2.578	8.295425
100	0.992	2.58	8.302161
250	0.995	2.58	8.275933
500	0.992	2.583	8.312255
1000	0.992	2.593	8.345817
2000	0.997	2.611	8.362234
4000	0.997	2.641	8.461465
6000	0.995	2.651	8.511733
8000	0.997	2.661	8.526994
10000	0.997	2.666	8.5433
12000	1	2.661	8.500897
14000	1	2.666	8.517203
16000	1	2.61	8.33281
18000	0.997	2.663	8.53352
20000	1	2.663	8.507423

Table III Frequency Response of Class D Amplifier. NOTE: Output Value must be reduce by 1/10, Gain taken into consideration.

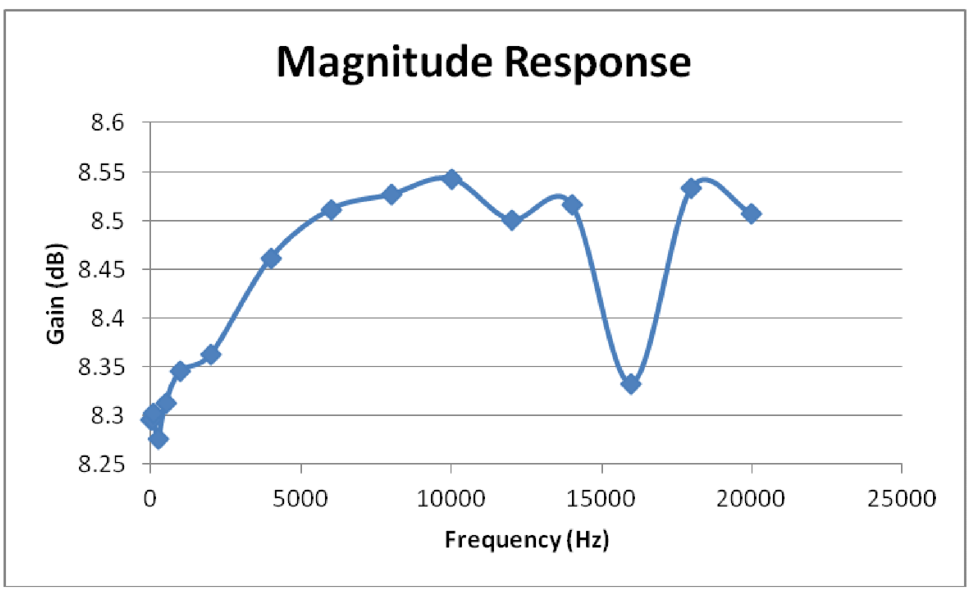

Figure 35 Magnitude Response With 1V input, resulting in 8.4dB gain

Even though realizing that the probe used was not correctly compensated for with a ratio of 10:1 when the correct value was 1:1, the expected gain is incorrect from that given in the datasheet. This is to be discussed further later.

Microphone

To measure peak voltage and develop a relationship between volume and distance, Figure 36 is constructed using the Electret microphone with Vcc set to 3V.

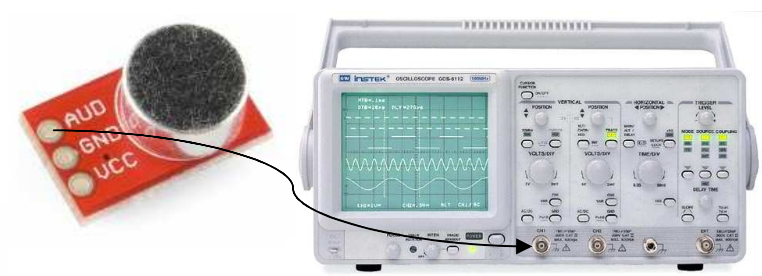

Figure 36 Test Setup, with Oscilloscope with single trigger enabled with close to zero threshold

Resulting data average at a distance of 1m from source with increasing volume (intensity). Table IV show various readings and their average value. Figure 37 gives a visual representation of the performance.

	Vout (V)					
Volume(Intensity)	Test 1	Test 2	Test 3	Test 4	Test 5	Average
0	0	0	0	0	0	0
5	0	0	0	0	0	0
10	0	0	0	0	0	0
15	0	0	0	0	0	0
20	0.24	0.22	0.24	0.23	0.23	0.232
25	0.38	0.3	0.28	0.31	0.3	0.314
30	0.38	0.38	0.4	0.4	0.36	0.384
35	0.38	0.22	0.24	0.5	0.3	0.328
40	0.38	0.3	0.24	0.3	0.46	0.336
45	0.54	0.42	0.44	0.4	0.46	0.452
50	0.78	0.32	0.36	0.4	0.38	0.448
55	0.72	0.4	0.5	0.42	0.36	0.48
60	0.7	0.38	0.38	0.48	0.46	0.48
65	0.74	0.72	0.72	0.64	0.64	0.692
70	0.84	0.64	0.82	1.01	0.95	0.852
75	1.1	1.18	1.12	1.05	1.001	1.0902
80	1.14	1.041	1.021	1.121	1.181	1.1008
85	1.141	1.181	1.081	1.281	1.221	1.181
90	1.361	1.481	1.221	1.181	1.321	1.313
95	1.101	1.281	1.181	1.021	1.489	1.2146
100	1.281	1.362	1.412	1.257	1.201	1.3026

Table IV Average Microphone behavior 1 meter from source with increasing volume.

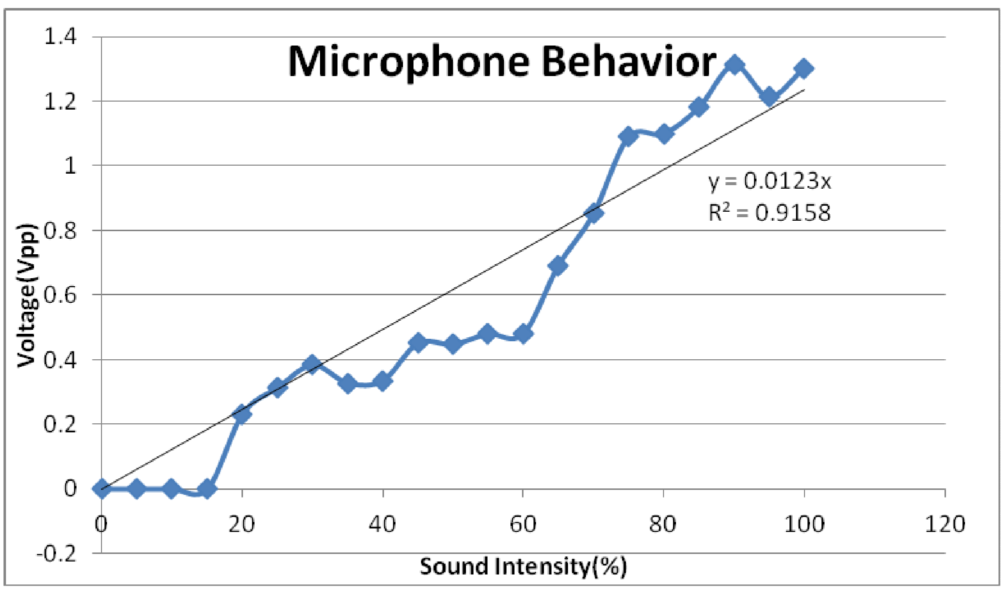

Figure 37 Sound Intensity versus Voltage(Vpp), 1m from source 1kHz Sinusoid

The behavior in Figure 37 can be applied to the Matlab simulation function to determine distance from listener and audio source. The function takes a known distance and its value for 50% intensity at that point. It then mimics the behavior of random intensities, trying to match gain and delay using another function that relates speed of sound and wavelength to a delay in discrete time.

MOSFET Impedance

Using the circuit provided by Figure 8 with R1 equal to 10kΩ, sweep Vgs from 0 to saturation to obtain the following plots in Figures 38 and 39.

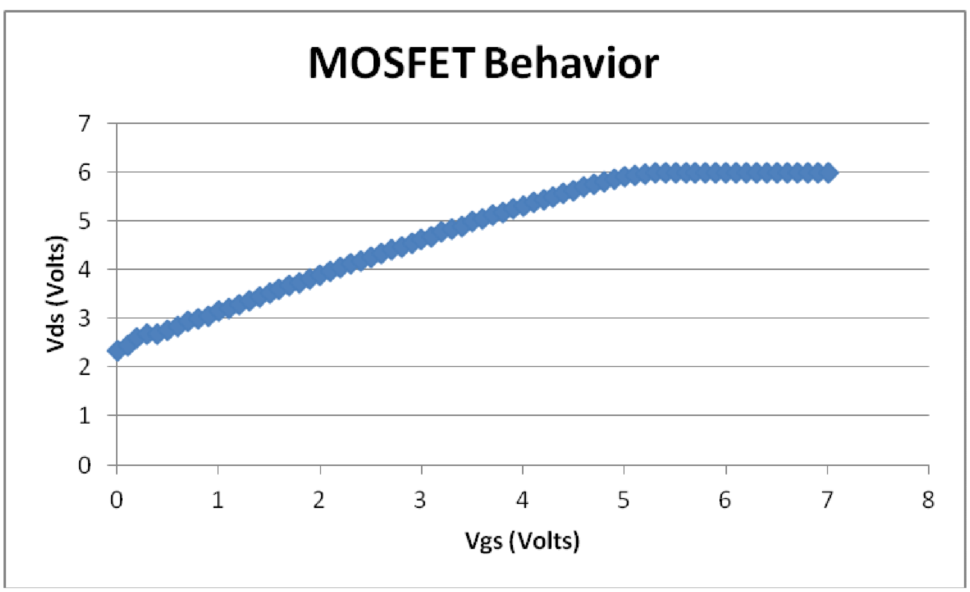

Figure 38 Vout with Increasing Vgs and 6V supply

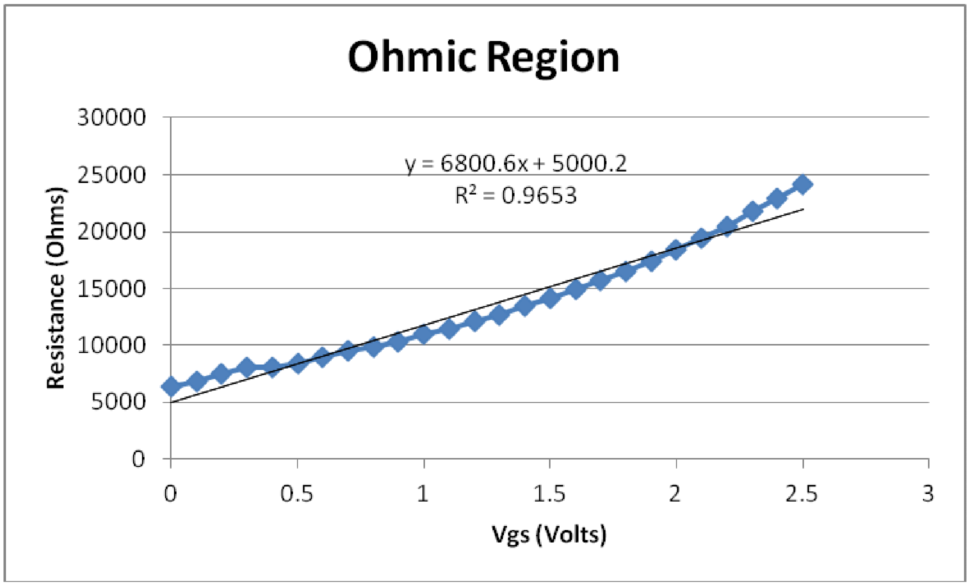

Figure 39 Linear Region, with Impedance approximately equal to trend line

Figure 39 provides the relationship mention earlier, IV.6. This is useful when trying to remain in the linearly region with linearly incrementing in code. It is more efficient for the processor to increment rather than stop and make a comparison.

5-Band Equalizer

The frequencies of the 5-Band equalizer are at: 32,125,500, 2000, and 8000Hz. To test the performance of the equalizer a frequency sweep must be conducted using the circuit depicted in Figure 41.

Sweeping frequencies with the values “10101” across the five potentiometers provides attenuation around the but not on the odd number bands. This results in gain seen around 32Hz, 500hz, and 8kHz. Figure 40 show the frequency response of the 5-band equalizer.

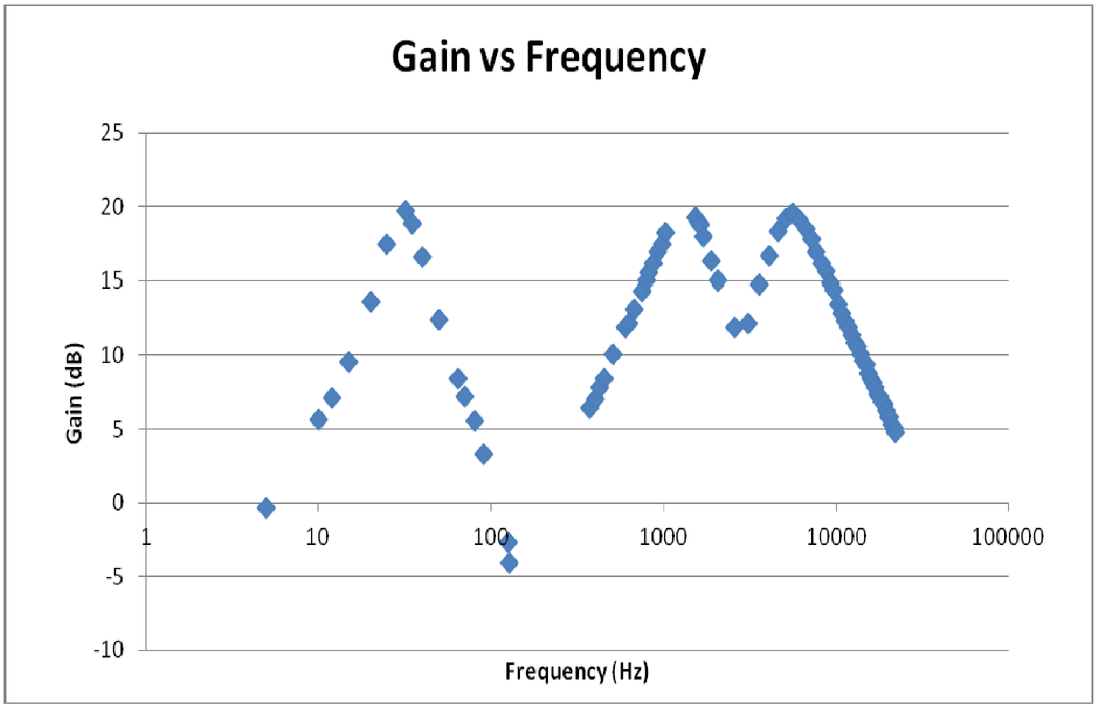

Figure 40 Magnitude Response of 5-Band Equalizer

VII. Conclusion and Recommendations

Due to the lack of significant phase change and less than ideal attenuation, it is more likely for a purely digital solution to overtake the initial analog effort. From the test data gather from the Allpass circuit shown in Section VII, it is apparent that the maximum time delay between 0V and saturation is 30 microseconds. At gate saturation the impedance become massive, but in the linear range there was no discernable difference in phase. To continue down the route of analog, would most likely require more circuitry with a higher chance of malfunction. In order to have significant phase delay at lower audible frequencies, multiple Allpass filters would have to be used. According to the test results from Section VI to delay the signal a quarter wavelengths, approximately 300 filters would have to be cascaded for a maximum delay of 10ms. The increase in filter amount would bring the number of MOSFETs being used. The increase in MOSFETs would limit the processing speed and I/O ability of the microcontroller. This number is far too high for practical use. Reducing gains of each channel offsets the required time delay, but this circuit only incorporates attenuation.

A solution to this dilemma would be to incorporate a digital signal processor with enough processing power to sample four signals at 48kHz. The process would incorporate changing the gain of each signal with a specific multiplier. Depending on delay, the index where the multiplier is applied changes, shown in Matlab simulation code on p60.

Unfortunately, the data recorded when measuring the magnitude response of the class D amplifier shown in Figure 35 is unreliable, as the expected gain when dealing with a supply voltage of 12V should have been approximately 26dB. This discrepancy begs to suspect that the amplifier was damaged internally. This was in timing, as the amplifier initially functioned correctly, though some connection may have been shorted between channels when measuring

each node when debugging earlier. With no signal present when testing, high DC voltage was present across various nodes. This is most likely to attribute to the failure of the IC, though the microphone behavior was as expected and the Allpass filter functioned but with minimal difference.

Though unable to complete the analog implementation of the project, a digital realization was not far from being fulfilled. Example code on p48-9 shows the algorithm in which distance, gain constant, and delays would be calculated the applied to each channel. By using a digital signal processor in combination with amplifier, provides a versatile solution to delay and gain matching. The new restriction when using a DSP is the reading of microphone voltage quickly enough as to not slow the sampling rate of the incoming audio signal to be delayed and multiplied a by some gain constant. If the restriction of “real-time” processing were lifted, then speaker distance, delay, and gain would only need to be measured once and set once, assuming the speaker is not moved.

In conclusion this project has further revealed the reality of digital dominance over analog circuitry. As more complex processes arise, an analog realization is much more difficult to design and perfect. While digital processing alleviates design, it is also very applicable to more than just audio. Overall this report represents the steps take to design a circuit, realize that circuit in physical form to later discover alternatives with more flexibility.

VIII. BIBLIOGRAPHY

[1]Dr. Ronald S. North Carolina State University, *Sound Localization.* 2001 Available: http://www.ise.ncsu.edu/kay/msf/sound.htm [Accessed: February 5, 2012].

[2 Dr. Burford J. Furman San Jose State University, *Sound Intensity*, 2005. Available: http://www.engr.sjsu.edu/bjfurman/courses/ME120/me120pdf/SoundIntensity.pdf. [Accessed March 23, 20012]

[3] Dr. Julius Smith, *Allpass Filters*, 2012. Available: http://ccrma.stanford.edu/~jos/pasp/Allpass_Filters.html [Accessed: January 5, 2011].

[4] *Design with Operational Amplifiers and Analog Integrated Circuits,* Sergio Franco, McGraw Hill Publishing, 2001

[5] *Microelectronic Circuit Design*, Richard C. Jaeger & Travis N. Blalock, McGraw Hill Publishing, 2011

[6] Datasheets: ALD1107, TB2939HQ, OP275, ICL7842, UA747CN, EVK1100

APPENDIX A. Senior Project Analysis

Project Title: “Auto-Tuned” Stereo

Student’s Name: Daniel Robert Montes **Student’s Signature:**

Advisor’s Name: Bryan Mealy **Advisor’s Initials:**

Date:

• **Summary of Functional Requirements**

My project in its primary function adjusts the phase and attenuates each signal as to bring all signals close to identical as possible. It then amplifies those signals through a low power audio amplifier to create a tuned audio experience.

• **Primary Constraints**

The greatest limiting factor is introducing enough phase change with minimal circuitry. Using the design implemented in the project only provided a theoretical phase change of 40º, with practical results of 10º. In order to have higher performing analog circuitry, the number of MOSFETs would have to increase. This would lead to the increase of circuit complexity, increase of probability to fail, increase cost, and reduce processing speed of microcontroller as it now has more I/O to program and reset every cycle.

• **Economic**

This project has little human capital impact, except for in research and sales. Manufacturing would most likely be conducted by machine to drive cost and production time

down. Like almost all electronics though, this doesn't offer any new job opportunities except for those in research and production design. The larger population, that doesn't have technical training, will have little to gain profession wise. This project presents no initial financial capital, though there is the potential for companies, entrepreneurs, and private investors backing the project because they see some sort of fiscal opportunity. Once there is interest, the financial capital will follow. As it exists now, as a single prototype, the project has little to no real capital. Without a patent most prototypes have little to no capital. If this project were to be patented, then the value would increase in the event of a buyer making an offer to buy the idea of the underlining technology. This project offers no natural capital as it does not utilizes renewable energy sources, nor does it generate energy or byproducts for useful in any other capacity.

The time when cost is greatest, would be during research and development. During research, time is typically spent without any prototypes or schedule for development. In research, the goal is to investigate the requirements and goal, and narrow the technology and techniques to be used during development. Though research maybe the most costly period, soon to follow is the most beneficial time of production. Assuming that researching has removed all non-ideal possibilities and now the design and development process will be much faster and more efficient.

The inputs require a signal with amplitude 1Vpp or less. The input is split into left and right channels, then those are broken into left and right front and surround. The project cost is a factor of price of custom PCB, discrete ICs, passive elements, and manufacturing materials. I have paid almost the entirety of the project, close to 95% of the cost was covered by myself. My original estimate for the project was around $150. The final cost of the project concluded to be around $115.

Bill of Materials		
Resistors		$Cost
10kΩ	24	2
5KΩ	4	0.5
120KΩ	10	1.1
1MΩ	11	1
20kΩ	1	0.1
10kΩpot	5	0.7
100kΩ	6	0.7
Capcitors		
63nF	4	1.9
0.22uF	4	2.3
0.1uF	1	0.2
3900uf	1	0.2
47uF	1	0.4
1uF	1	0.2
22nF	2	0.5
5.6nF	2	0.5
1.5nF	2	0.5
330pF	2	0.6
82pF	2	0.7
470uF	1	0.9
IC		
ALD1107	2	3.5
OP275	4	6.5
UA747CN	4	3.4
TB2939HQ	1	7.9
ICL7842	2	0
Microcontroller		
EVK1100	1	0
Microphone		
Electret	2	19
Speakers		
4Ω	4	60
MISC		
Heatsink		0
Wires		0
Total		115.3

Table V Bill of Materials Used and Purchased

The project has no means of generating a profit, so there is no individual who profits except those from whom I have purchased items towards the development of this project.

Products emerge when there is a void in the market and when demand is high. Products typically stay relevant for eight to twelve months. No maintenance costs exist in this project, because it does not directly deal with earth forces or extreme conditions. The operations cost would be supply a large enough voltage for the amplifier to work. The cost of supplying power is essentially the only cost to operate this project.

Original estimated development time 4.5 months. Actual development time 5.25 months. Now that the project is finished, an alternative design will be pursued purely in software, seek more projects with audio background.

•If Commercially Manufactured

Depending on price if it was set below $130, I would estimate that volume of sales would be in the range of 70-100 thousand per year. If the device was anything like the finished project, the price would be above 250 to manufacture. Eliminating the microcontroller development board and design with internal microcontroller with less peripherals, only essential ports and functions; then cost could fall around to $100. Stated as before, $130 would be an ideal price range with a large enough profit to reinvest into research and reduced manufacturing. With an estimate of 85,000 products sold with a $30 profit from each, would result in profit margin of $2.55 million. With the amplifier running around 30W of power consumed, and a kWh rate of $0.1 for 4 hours a day; this would result into $3.64 per year to operate.

• **Environmental**

The environmental impacts in manufacturing the projects as it exists today come from the energy and chemicals spent manufacturing each component on the board. Manufacturers try their best to stay within regulations regarding chemical safety and environmental awareness, but there remains a small percentage of toxicity leaking from all products and waste byproducts stemming from the manufacturing plants. Once the components are manufactured, they are sold and shipped. Shipping these miniscule items by fossil fueled vehicle damages our atmosphere as well as the excessive packaging which at most times does not get recycled. Lastly comes the construction and testing of the project. Soldering each component on requires flux and solder. Both release toxic fumes and residue that not biodegradable and cost energy to filter out of air. Testing the board means having to turn on meters and supplies which all use energy without any way to return that energy back.

The project uses audio primarily, so the air quality determines the characteristics of a sound wave. A more polluted atmosphere creates a less than appealing sound reproduction. If the air quality were to decline, then the effectiveness of the project would become more of an issue. The only species that could have some adverse impact would be one with sensitivity to noise. Species who rely on sound for navigation might become confused or worse depending on their reaction to sound.

• **Manufacturability**

The largest issue with this project was user error. When developing the layout for the PCB, I made an error in the spacing for the class D power amplifier. Unfortunately I realized the pins where far too close and small. Already a week behind schedule, I had no choice but to work

around the mistake by adding a wire modification to lengthen the leads of the IC and insulating the wires with hot glue and securing the IC's location. As to be expected practical testing revealed a less than optimal performing circuit, but with the remnants of proper function nonetheless.

• **Sustainability**

Sustaining the amplifier in working condition proved to be somewhat difficult at first as the test supply voltages kept overloading because they were current limited. By combining supplies I was able to overcome that current limit issue, though the supply ripple regulatory capacitor began to overheat and was destroyed as the voltage become too great, it was rated at 12V but with 12V on the supply, the boundary was too close to the constraint. As discussed earlier in the report a purely digital realization of this project us much preferred. The gain and phase can be applied more accurately with reduced cost of components and manufacturing. Though a digital signal processors have their own expense as well.

APPENDIX B.

SCHEMATIC

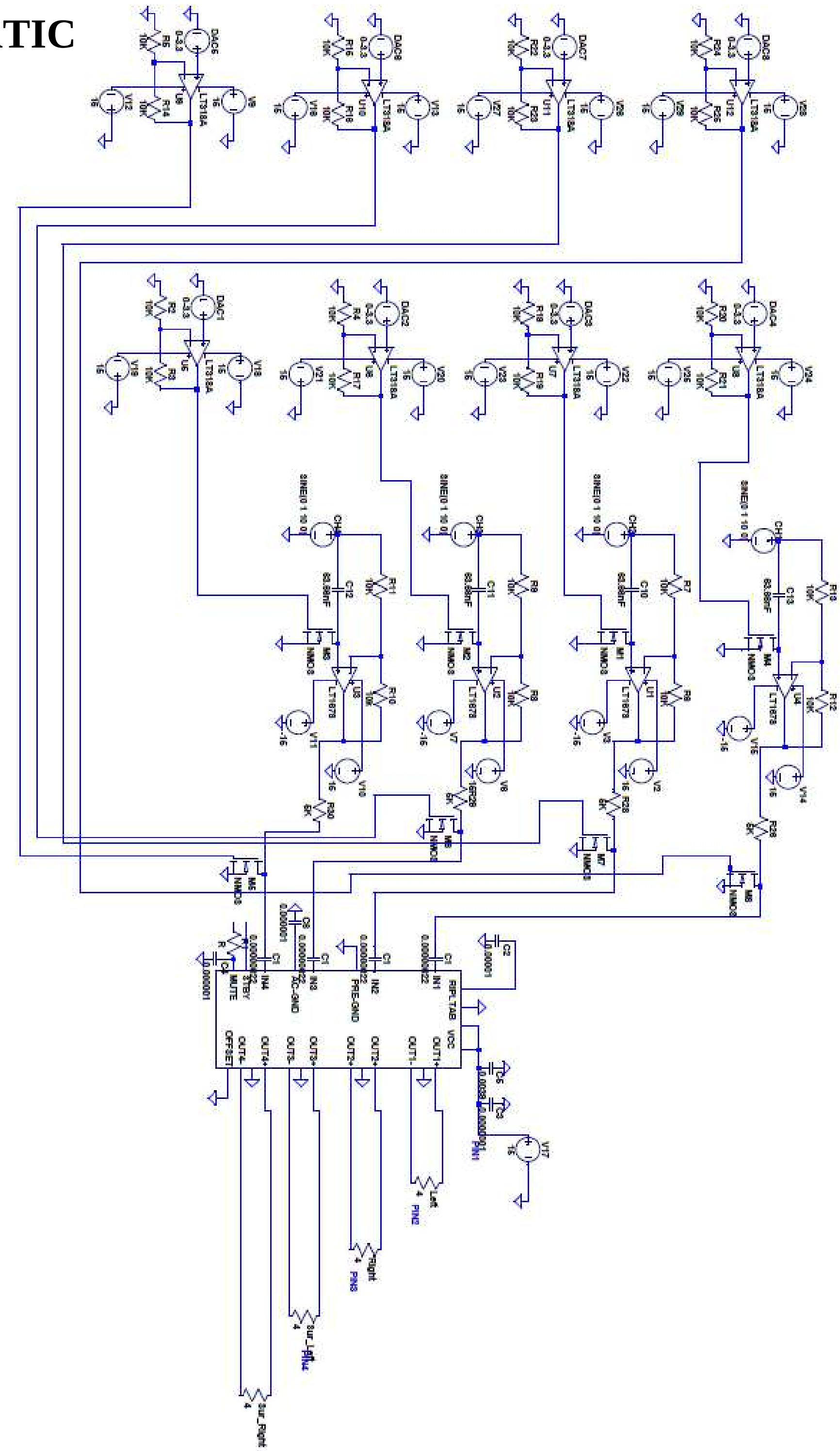

Figure 41 Complete Audio Amplifier Design

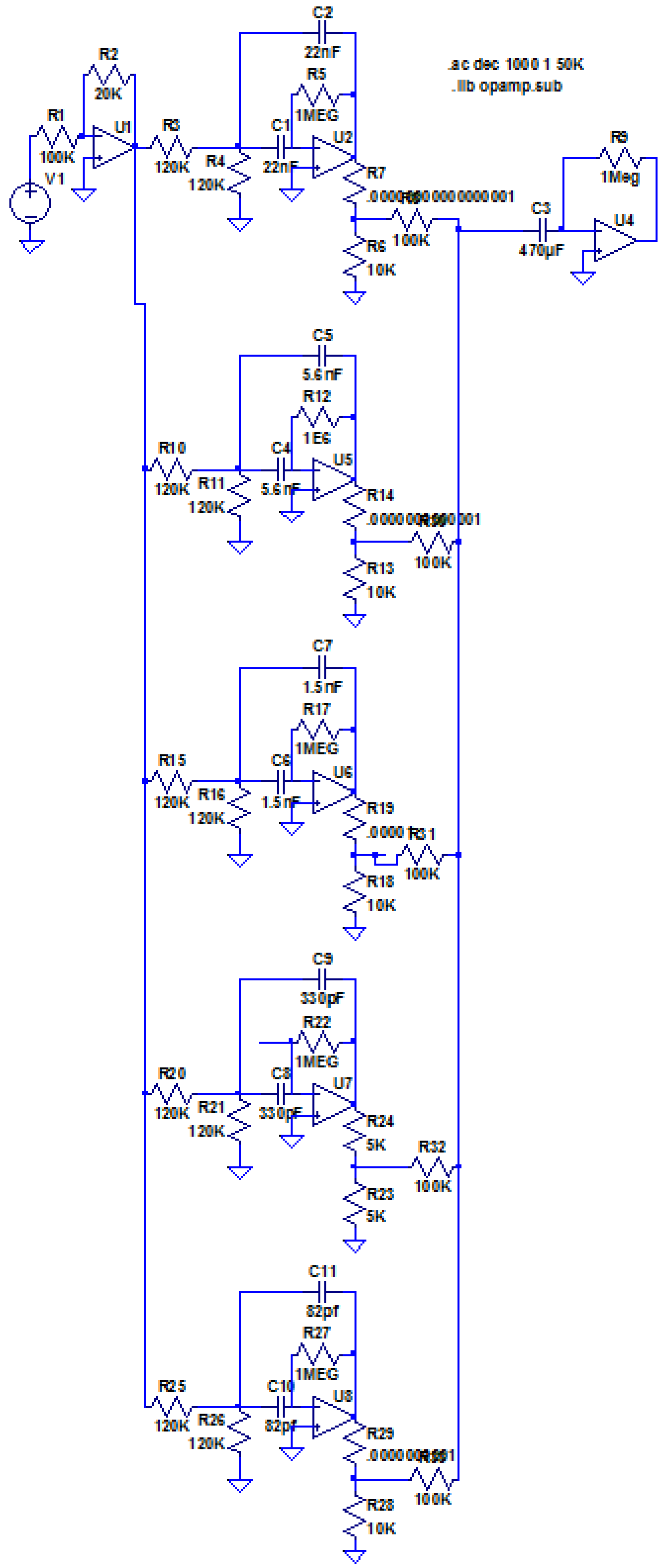

Figure 42 5-Band Equalizer: 32Hz 125Hz 500Hz 2kHz 8kHz

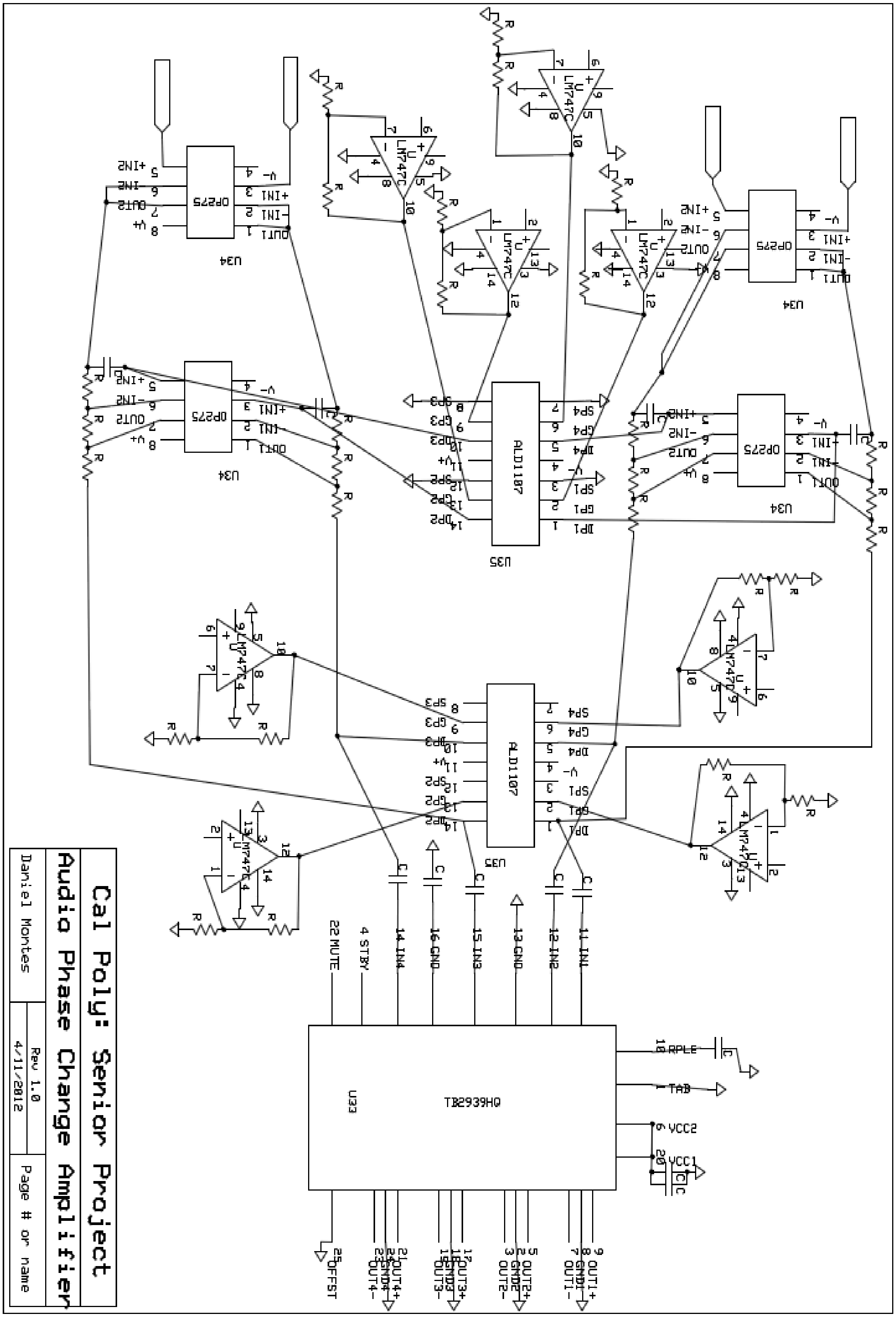

Figure 43 ExpressSCH© Project Layout

APPENDIX C.
BILL OF MATERIALS, COST, SCHEDULING

Bill of Materials		
Resistors		$Cost
10kΩ	24	2
5KΩ	4	0.5
120KΩ	10	1.1
1MΩ	11	1
20kΩ	1	0.1
10kΩpot	5	0.7
100kΩ	6	0.7
Capcitors		
63nF	4	1.9
0.22uF	4	2.3
0.1uF	1	0.2
3900uf	1	0.2
47uF	1	0.4
1uF	1	0.2
22nF	2	0.5
5.6nF	2	0.5
1.5nF	2	0.5
330pF	2	0.6
82pF	2	0.7
470uF	1	0.9
IC		
ALD1107	2	3.5
OP275	4	6.5
U747	4	3.4
TB2939HQ	1	7.9
ICL7842	2	0
Microcontroller		
EVK1100	1	0
Microphone		
Electret	2	19
Speakers		
4Ω	4	60
MISC		
Heatsink		0
Wires		0
Total		115.3

Table V Bill of Materials and Project Cost

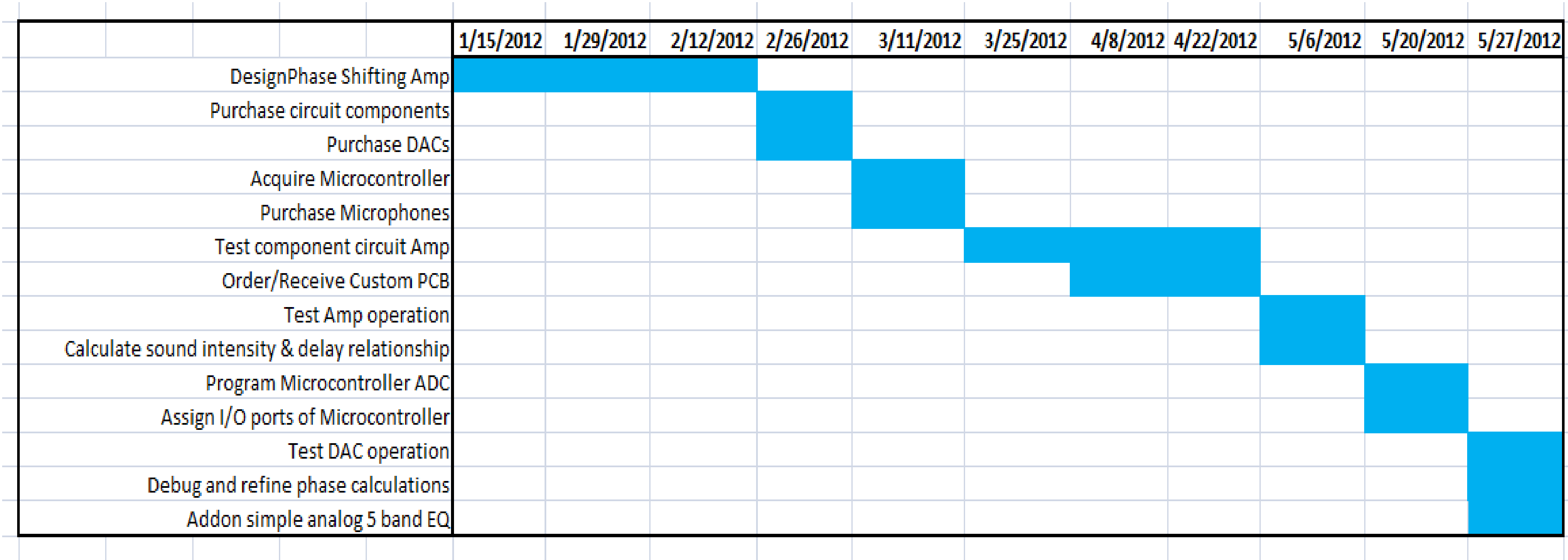

Figure 44 Time Schedule, Gantt Chart

APPENDIX D.

PRINTED CIRCUIT BOARD ARTWORK

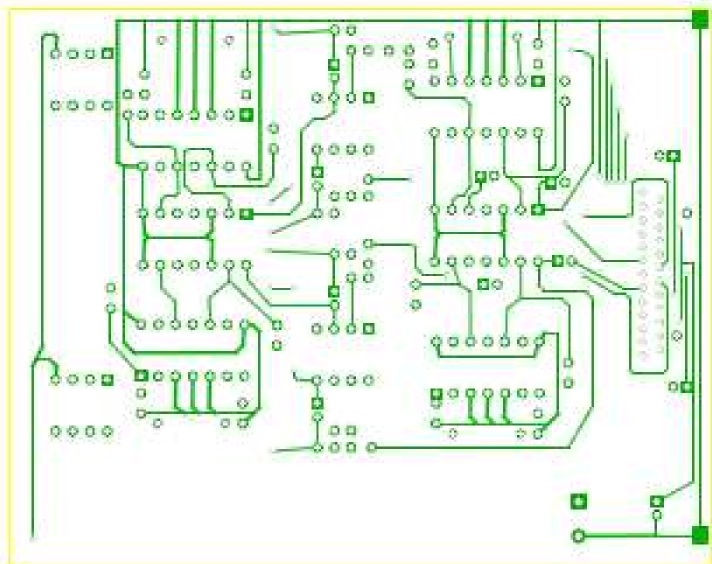

Figure 44 Bottom Layer Copper, PCB Layout

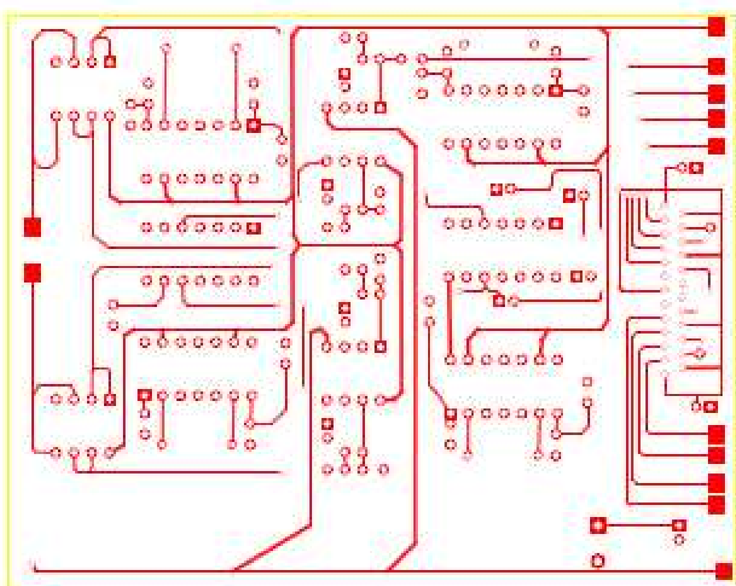

Figure 45 Top Layer Copper, PCB Layout

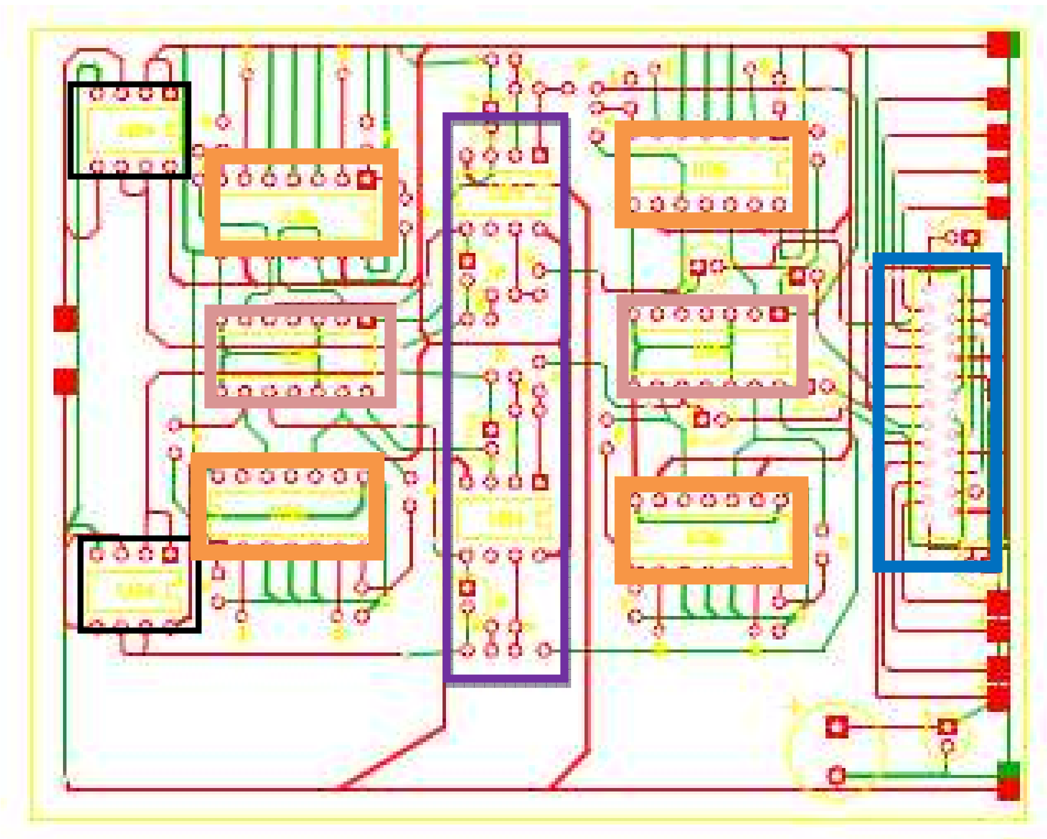

Figure 46 Complete PCB Layout: Voltage Follower(Black) Non-Inverting Amplifier(Orange) nMOSFET(Pink) Class D Amplifier(Blue)

APPENDIX E.

BASIC PROGRAMING LIST

```
/***************************************************************************
 *ADC READ MICROPHONE
 */

#include "board.h"
#include "print_funcs.h"
#include "gpio.h"
#include "pm.h"
#include "adc.h"
#include "delay.h"
#define VREF 3



/*! \name ADC channels choice
 */
//! @{
#if BOARD == EVK1100
// Connection of MIC_1
#  define MIC_1_Channel               0
#  define MIC_1_PIN                   AVR32_ADC_AD_0_PIN
#  define MIC_1_FUNCTION              AVR32_ADC_AD_0_FUNCTION
// Connection of MIC_2
#  define MIC_2_Channel                2
#  define MIC_2_PIN                    AVR32_ADC_AD_2_PIN
#  define MIC_1_FUNCTION               AVR32_ADC_AD_2_FUNCTION
#endif
//! @}

/*!
 * \brief main function : do init and loop to display ADC values
 */
int main( void )
{
 volatile avr32_adc_t *adc = &AVR32_ADC; // ADC IP registers address

#if defined(MIC_1_Channel)
signed short adc_value_MIC1 = -1;
#endif
#if defined(MIC_2_Channel)
signed short adc_value_MIC2 = -1;
#endif

// switch to oscillator 0
pm_switch_to_osc0(&AVR32_PM, FOSC0, OSC0_STARTUP);

// init debug serial line
init_dbg_rs232(FOSC0);

// GPIO pin/adc-function map.
static const gpio_map_t ADC_GPIO_MAP ={

#if defined(MIC_1_Channel)
                                {MIC_1_PIN, MIC_1_FUNCTION},
#endif
#if defined(MIC_2_Channel)
                                {MIC_2_PIN, MIC_2_Function},
#endif
                                                    };

// Assign and enable GPIO pins to the ADC function.
gpio_enable_module(ADC_GPIO_MAP, sizeof(ADC_GPIO_MAP) / sizeof(ADC_GPIO_MAP[0]));
```

```
// configure ADC
// Lower the ADC clock to match the ADC characteristics (because we configured
// the CPU clock to 12MHz, and the ADC clock characteristics are usually lower;
// cf. the ADC Characteristic section in the datasheet).

AVR32_ADC.mr |= 0x1 << AVR32_ADC_MR_PRESCAL_OFFSET;

adc_configure(adc);

  // Enable the ADC channels.
#if defined(MIC_1_Channel)
                                        adc_enable(adc, MIC_1_Channel);
#endif
#if defined(MIC_2_Channel)
                                       adc_enable(adc,MIC_2_Channel);
#endif

// do an infinite loop
while (true)
{
// launch conversion on all enabled channels
adc_start(adc);


// get value for MIC 1

#if defined(MIC_1_Channel)

         adc_value_MIC1 = adc_get_value(adc, MIC_1_Channel);

#endif

            temp_MIC1= adc_valueMIC1;//holds temp mic value for comparison

// get value for MIC 2

#if defined(MIC_2_Channel)

         adc_value_light = adc_get_value(adc, MIC_2_Channel);

#endif

             temp_MIC2= adc_value_MIC2//holds temp mic value for cmparison

//compares"volume" on each mic
#if temp_MIC2 >temp_MIC1;

LED0= ~00001;//LED 1 indicate left side louder
#endif

//compares"volume" on each mike
#if temp_MIC2 >temp_MIC1;

LED0= ~10000;//LED 5 indicates Right side louder
#endif

// slow down operations
delay_ms(10);                                                                                    }

// Disable the ADC channels.
#if defined(MIC_1_Channel)
adc_disable(adc, MIC_1_Channel);
#endif
#if defined(MIC_1_Channel)
adc_disable(adc, MIC_1_Channel);
#endif
while (true);

return 0;
```

%SIMULATED PROGRAMMING BEHAVIOR FOR TI DSK C5416

```
function [delay_filter_hn] = delay_filter(FILE, Dk_delays_msec,gains,Fsample)
% function [delay_filter_hn] = delay_filter(Dk_delays_msec,alphak_gains,Fsample)
%
% Input Parameters:
%   Dk_delays_msec = delay time in milliseconds
%   alphak_gains = fraction gains for delay
%   Fsample = sampling frequency
%
% Output Parameters:
%   Delay_filter_hn = impulse of delay filter
% Number of delays
M = length(Dk_delays_msec);


% Convert from milliseconds to seconds
Dk_delays_sec = Dk_delays_msec./1000;

% index of delays
idx = Dk_delays_sec.*Fsample;

% set to zero the amount of the largest index plus one
delay_filter_hn = zeros(1,idx(M)+1);

% set the first value to 0;
delay_filter_hn(1) = 0;

for j=1:M
   % Set every calculated index to the fractional gain
   % Plus one to make up for Matlab starting index of 1
   delay_filter_hn((idx(j)+1)) = gains(j);
end

hw = delay_filter_hn;

%Apply delay to audio out
[xn,FS,N] = wavread(FILE);
xn = xn';
newFILE = ['delay_', FILE];
a = size(xn);
if (a(1) == 1)
   wavwrite(eq_fftconv(xn,hw)', FS, N, newFILE);
else
   yn(1,:) = eq_fftconv(xn(1,:), hw);
   yn(2,:) = eq_fftconv(xn(2,:), hw);
   wavwrite(yn', FS, N, newFILE);
end
```

%MATLAB FUNCTION TO DETERMINE GAIN NECESSARY FOR DELAY_FILTER FUNCTION

```
function [Ich, loc, locm] = spherical_convrs(speaker1, speaker2, speaker3, speaker4, mic)



loc(1) = sqrt((speaker1(1)-mic(1))^2 + (speaker1(2)-mic(2))^2 + (speaker1(3)-mic(3))^2);
loc(2) = sqrt((speaker2(1)-mic(1))^2 + (speaker2(2)-mic(2))^2 + (speaker2(3)-mic(3))^2);
loc(3) = sqrt((speaker3(1)-mic(1))^2 + (speaker3(2)-mic(2))^2 + (speaker3(3)-mic(3))^2);
loc(4) = sqrt((speaker4(1)-mic(1))^2 + (speaker4(2)-mic(2))^2 + (speaker4(3)-mic(3))^2);

locm = loc*0.3048;

%0.5 Intensity at 10ft = 3.048m
for i=1:4
   Ich(i) = 1/(locm(i)/2.15526)^2;
   if Ich(i) > 1
      Ich(i) = 1;
   end
end

end
```

%MATLAB FUNCTION TO DETERMINE DELAY NECESSARY FOR DELAY_FILTER FUNCTION

```
function delay = delay_calc(locm)

%speed of sound
s = 340.29;

%calculate delay in ms
for i=1:4
delay(i) = (locm(i)/s)*1000;

end

end
```