

DRUM VOLUME CONTROL

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Abstract

It was our intention to create a system which would attenuate the sound pressure level, or quiet, the sound waves emanating from a drum. Basic operation of our system would include inheriting a sound signal, inverting this signal, and combining the new sound with the old sound to create destructive interference. Certain challenges prevented us from achieving significant attenuation, but various successes realized by the project and knowledge acquired through testing and experimentation essentially laid the groundwork for a fully functional design in the future. At present, our system can successfully detect a sound, recognize the specific frequencies associated with that sound, invert the signal, and re-combine the processed signal with the original without introducing significant distortion.

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Introduction

One problem that we have noticed over the years is that when musicians play in a small venue (a church or bar, for example), the drums have a tendency to overpower the rest of the musicians, causing an unpleasant listening experience for the audience. It was our intention to design a way to diminish, or attenuate, the volume of the drums in such settings, thus creating a more pleasant listening experience.

Various methods of volume control for drums are already in place, but there are various shortcomings with each of these methods. For example, many drummers use acrylic drum shields and foam dampening pads, but these methods do not offer a way to adjust the volume, which may be desired depending on the overall volume of the music being played or the person playing the drums. Another solution is to use electronic drums which do offer adjustable volume, but unfortunately do not offer the same “feel” and “response” as normal acoustic drums, which is a very desirable thing to have.

It is for this reason that we chose to approach the problem by utilizing Active Noise Cancellation, the technology employed by current models of noise-cancelling headphones. The basic theory behind this technology involves capturing a sound wave, inverting, or shifting the wave by 180° , and combining both waves to create destructive interference and attenuate the signal.

In determining specific objectives, we first determined what the scope of our project would be. We knew that the complexity of 3D spherical wave interference patterns meant that decreasing the volume of a complete drum kit would be infeasible, so we narrowed our focus to a single 12” drum. In theory, we would be able to apply our method to one drum and extrapolate the method for the entire kit, but of course, much more knowledge concerning interference patterns due to the other drums would have to be attained.

Additionally, the complexity of these interference patterns meant that, in the absence of many speakers to create cancellation effects everywhere, we would have to focus on cancellation in a single direction. We believed this to be possible due to previous knowledge of spherical wave patterns, and how multiple sources of spherical waves will create “nodal lines,” or directional lines along which cancellation occurs equally at all points. Once we determined the scope of our project, we set our more specific goals for our project, the first of which was to achieve attenuation of 6 dB SPL (sound pressure level) in three separate directions. Eventually we believed this to be too complex of a problem and narrowed our focus to attenuation in a single direction. Another goal associated with our project, though not explicitly mentioned in the beginning of the course, was to introduce minimal distortion within the processed drum signal. Distortion within our electronic components implies that different frequencies would be delayed by different amounts of time. If distortion were present within our system, recombination of our processed signal and original signal would result in cancellation of some frequencies, while others would constructively interfere and amplify, creating a new, possibly louder, sound.

The upcoming sections will describe in detail our original design for our system, changes made to this design, various testing procedures to verify functionality and results of those tests, various challenges and successes we experienced, conclusions made and recommendations future work on the project.

2 Design

2.1 Original Design Procedure

Our original design included five main components, excluding microphones and speakers: a low-pass filter, a high-pass filter, a pre-amplifier stage, an inverter, and a final amplifier stage (block diagram can be viewed in the appendix).

The first two components to be encountered first by the input signal after propagating through the microphone would be combined to form an effective band-pass filter, the purpose of which would be to filter out sounds not associated with the drum. In choosing the appropriate model for our filters, we decided to build eighth-order Bessel filters, which are constructed by cascading several copies of the Sallen-Key circuit (Figure 2) with unique values for resistor and capacitor components, specified by the corresponding table of filter coefficients (Figure 3). We chose the Bessel filter model for our design due to its exceptional linear phase attribute, which, as mentioned earlier, is very important to the success of the project since it will be less likely to introduce distortion. We decided to create an eighth-order filter because, in general, as we introduce more poles to a filter (and consequently, a higher order), certain qualities of the filter, such as cutoff frequency roll-off and linear phase, will improve greatly. As a result, we desired to build as high-order a filter as possible, but this required knowledge of coefficients k_1 and k_2 (see Figure 3) for filters of a certain order. Since the highest-order coefficients we had knowledge of were eighth-order coefficients, we settled on this for an adequate design.

The second component in our modular design was Pre-Amplifier stage. We recognized the output on a microphone would be too small to work with. Our initial design utilized a pre-amplifier stage to amplify the signals from the microphone before it was fed into the rest of the circuit. We assumed that this signal would be better to work with in the latter stages if it resided in a bigger power spectrum. This could potentially allow for the signal to drive latter stages of the design or if signal attenuation occurred in latter stages, there would be enough power generated on this stage for minimal attenuation to not have a significant effect on the signal.

The third stage of the circuit is meant to “phase-shift” the signal 180 degrees. This allows for the theoretical destructive interference we were trying to accomplish after the signal was fed through a speaker. The idea was that if we “phase-shift” the signal 180 degrees, we create the exact opposite signal and thus generate the exact opposite sound wave in the latter stages. We wanted to utilize an inverting amplifier to negate the signal, which would essentially be “phase-shifting” by 180 degrees. A basic inverter seemed logical for the desired 180 degree phase shift because other types of phase-shifting devices may not be as consistent for this purpose. The simplicity of the design itself is what allowed us to make accurate 180 degree phase shifting within our original tolerances for phase-shifting 180 degrees. Also, because we utilized an analog design, we thought it would be more practical to use an inverter instead of an actual phase-shifter for the purpose of 180 degree phase-shifting.

Our final stage in our original design before the signal was sent to the speaker was the Final Amplifier. This component was an op-amp circuit as well, and in fact was almost identical to our Pre-Amplifier stage, with the exception that our R_f value, which was used to control the gain

factor of the circuit (see Equation (1)), was replaced by a 50 k Ω potentiometer. The rationale for doing this was to allow the user to adjust the gain manually, and thus control the amplitude of the inverted signal which was responsible for cancelling the original wave. In this sense, the user had control over the attenuation of the original wave, and thus had volume control of the drum.

For this component, the same model op-amp was used as in the previous two components. R_f was replaced by a RV6NAYSD503A-P Clarostat 50 k Ω single-turn potentiometer, R_g was replaced by a 1.476 k Ω resistor, and, based on Equation (x), the circuit will have a gain of anywhere between 1.000 and 34.198, depending on the position of the potentiometer dial. See Figure 4 for the circuit diagram of the Final Amplifier.

After this stage, our processed signal would propagate through the speaker and be projected back at the drum to cancel the original sound wave. Note that in our original design, we determined that the “wet,” or processed, signal would propagate in a direction opposite to that of the “dry,” or original, signal.

2.2 Original Design Details

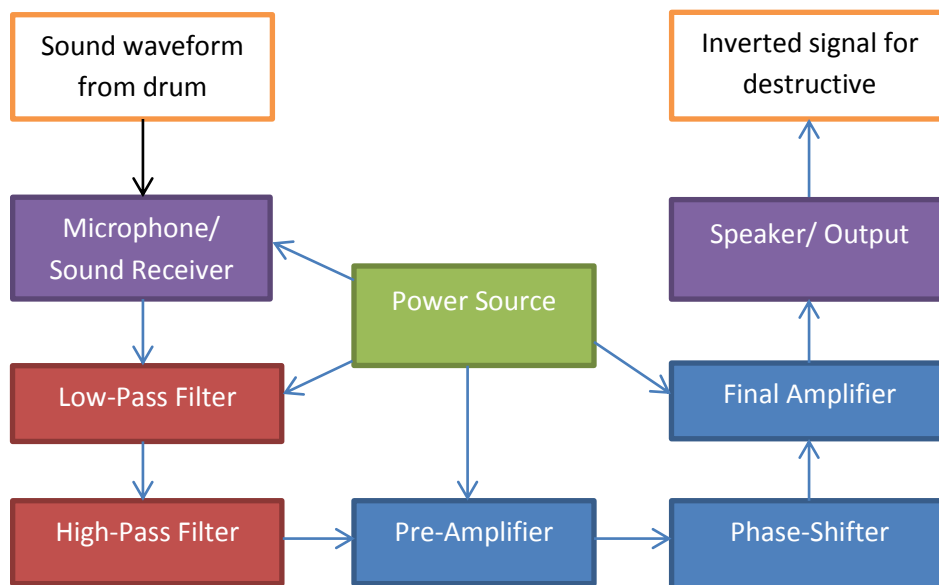


Figure 1: Original Design Block Diagram

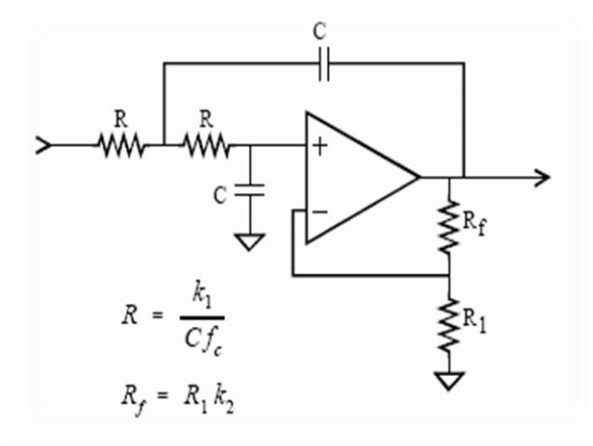


Figure 2: Sallen-Key Circuit

# poles		Bessel	
		k_1	k_2
2	stage 1	0.1251	0.268
4	stage 1	0.1111	0.084
	stage 2	0.0991	0.759
6	stage 1	0.0990	0.040
	stage 2	0.0941	0.364
	stage 3	0.0834	1.023
8	stage 1	0.0894	0.024
	stage 2	0.0867	0.213
	stage 3	0.0814	0.593
	stage 4	0.0726	1.184

Figure 3: Bessel Filter Coefficients

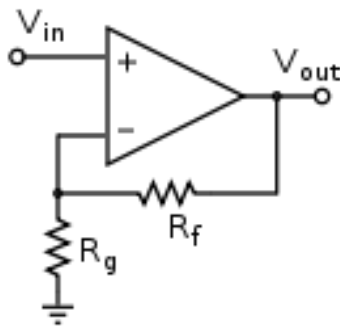


Figure 4: Pre-Amplifier/Final Amplifier

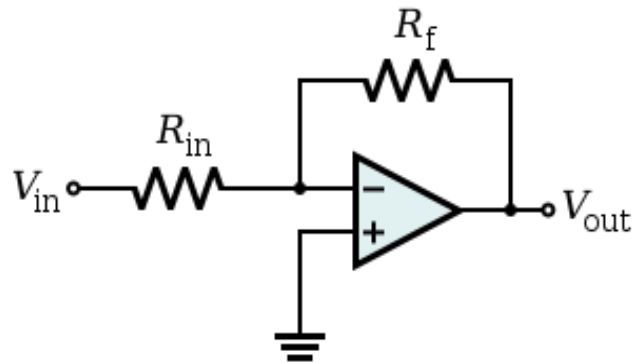


Figure 5: Inverter

In Figure 4, we see the circuit diagram for the Pre-Amplifier stage as well as the Final Amplifier. For the Pre-Amplifier, we have component values: $R_f = 33.44 \text{ k}\Omega$, $R_g = 1.989 \text{ k}\Omega$, and $V_{out} = 17.812 \cdot V_{in}$. The value of V_{out} was calculated using the transfer function specified as

$$V_{out} = V_{in}(1 + R_f/R_g) \quad \text{Equation (1)}$$

In Figure 5, we see the circuit diagram for the Inverter stage. We have component values: $R_f = 38.75 \text{ k}\Omega$, $R_{in} = 39.14 \text{ k}\Omega$, and $V_{out} = V_{in} \cdot (-0.99)$. This value of V_{out} was calculated using the transfer function for the inverter specified as

$$V_{out} = V_{in} * (-R_f/R_{in})$$

Equation (2)

2.3 New Design Procedure

Throughout testing our original design, we encountered a couple major challenges which made us reconsider the stability of our components, and eventually led us to choosing new components altogether.

Our first major challenge encountered was the general unreliability of the first microphone we used. Throughout our testing procedures, we noticed that at times our microphone would receive a signal and an oscilloscope measuring the voltage across the microphone would display a clean waveform; however, at some time later, the scope would not display this clean signal under the same operating conditions as before. Since our testing procedures involved comparing signal data from many different strikes of the drum, we recognized that our entire system would have to be completely repeatable; any inconsistencies in equipment would effectively invalidate measurements taken. Therefore, we made the decision to switch our current microphones with a Shure SM57 and a Shure SM48, two higher-quality dynamic microphones with increased reliability and sensitivity.

Another major challenge we experienced was the fact that our op-amp circuits could not source enough current to power our 8 Ω speakers. Due to this, the voltage at the output of our Final Amplifier stage would be clipped at a certain value once connected to our speaker, and we determined that the highest voltage the amplifier could output into the speaker without distorting the signal was 0.2 V, corresponding to an input power of 0.005 W into the speaker, a barely audible tone. Clearly, this low signal level would not be strong enough to appreciably attenuate the signal from the drum, so we decided to replace our speakers with a Behringer Eurolive B215A 400W powered amplifier/speaker combination, as well as a Marshall MG30DFX 30W powered amplifier/speaker combination to continue our experimentation. We chose these speakers primarily due to their large power ratings, which would allow us to amplify our signal sufficiently without fear of clipping, but also due to their broader frequency response as well as lower distortion ratings. Because they include powered amplifiers as well, they then replaced our old amplifier component in addition to our speakers.

We also replaced our original Pre-Amplifier with a powered mixer to eliminate the chance of our op-amp distorting higher-frequency signals, and decided that our filters were ultimately not necessary, due to the fact that their purpose (eliminating sounds not associated with the drum) would be easily realized during the demo by simply striking the drum in a quiet environment.

3. Design Verification

3.1 Original Design Verification

We began testing our design by verifying that our inverter functioned properly. These tests were to make sure that the component made a perfect inversion for each frequency and did not introduce any additional phase shift, as well as to make sure that the component would have essentially unity gain for the entire length of the input waveform. These tests were conducted by inputting a waveform from the function generator into the input of the inverter and measuring the input with channel one on the oscilloscope and the output of the inverter on channel two. At the time we took these measurements, we had not yet taken samples of our drum to discover what frequencies were present within the signal. Though the drum was tuned relatively low, we had reason to suspect that there would still be high frequency content present within the signal, so we estimated the drum to contain frequencies anywhere between 1 and 10 kHz. For this particular test, we varied the input waveform from the function generator between 1 and 7 kHz at 0.5 kHz intervals. Plots of various results are shown at the end of the chapter. As we can see, the inverter introduces a perfect inversion (phase shift of 180°) for all frequencies tested and achieves unity gain for the entire length of the waveform, with the exception of some small fluctuations that, ultimately, we deemed small enough to be considered negligible, though the output did not always fall within our 5% tolerance limit.

Similar tests were conducted for the Pre-Amplifier stage, as we would input a waveform from the function generator into the circuit and view the input and output of the circuit on channels one and two, respectively. The purpose of this testing procedure was to verify that the output would be a perfectly scaled version of the input for all points on the waveform. Two plots of this test are given in the Figures section at the end of the chapter, and we can see from those that the circuit accomplishes its task. Again, there are small deviations from a perfect scaling on the output, but we deemed these negligible once again; however, we were a bit concerned to see the fluctuations getting a bit larger, and noted the possibility that they may become prominent enough to introduce serious problems once the systems became interconnected.

This same testing procedure was applied to our Final Amplifier component, but for each frequency we varied the gain from its minimum to its maximum value and noted the effects. Plots of input and output waveforms are shown at the end of the chapter for a 1 kHz wave at minimum gain and at maximum gain. Again, we see small fluctuations in the output waveform, but we can confirm that these are simply due to the function generator being unable to output a perfectly smooth sine wave at such low voltages (see figure at minimum gain; input and output waves are essentially the same, but the input waveform is not a perfect sine wave).

One problem that we encountered with the Final Amplifier stage is that, for high gain values, a noticeable phase shift occurred for higher frequencies. As it happens, this phase shift does not become prominent for frequencies associated with the drum, but at the time this measurement was taken, we were unaware of the frequencies within the drum signal and so we deemed it necessary to test sufficiently high frequencies. A table of phase-shift values at minimum and maximum gain for different frequencies is given at the end of this chapter.

3.2 New Design Verification

We began by analyzing the specific frequency content within the drum. We did this by striking the drum with a microphone nearby, running the microphone into our powered mixer, and measuring the output of this mixer with the oscilloscope. Various tests revealed that the drum signal contained frequency content up to about 6.5 kHz, though much of the frequency content above 500 Hz was measured as well below 0 dB, sometimes as low as -20 or -30 dB (in this case, 0 dB is defined as a 1 V_{RMS} sine wave). We then analyzed the frequency spectrum between 0 and 500 Hz and noted all prominent frequencies, especially those above 0 dB, and re-tested our inverter for these specific frequencies. The most prominent frequencies we found were 193.5 Hz, 196 Hz, 212 Hz, 230 Hz, and 236 Hz, and we will show plots of inverter input and output signals at 193 Hz and 233 Hz at the end of this chapter. Results are very similar to those calculated earlier; no additional phase shift is seen and fluctuations in amplitude are minor, even though at times they fall outside of the 5% tolerance range.

We then tested that our combined system of our new microphone, mixer, and inverter to verify that our system could receive and invert an audio signal. To do this, we input waveforms of various frequencies into the Marshall amplifier/speaker with the SM57 microphone nearby, ran the microphone through the mixer, and then fed the output of the mixer into the input of the inverter. We then measured the input and output of the inverter on channels one and two of the oscilloscope, respectively. The reason I did not measure the output of the function generator on channel one is that, due to the time delay associated with the electronics, different frequencies propagating through will be phase-shifted by different amounts. Therefore, the output of the function generator and the output of the inverter would not appear to be 180° out of phase, and we would gain no real information from the test. However, we do find that the inverter works for signals being picked up by the microphone, and the oscilloscope's Quick Measure feature verifies that the frequency of the signal coming into the inverter is the same as the frequency of the signal coming out of the function generator (though you will not be able to tell by the graphs, as the photographs were taken by a cell phone and could not pick up the Quick Measure display). Various graphs of this test are shown at the end of this chapter.

In addition to these tests, we decided to verify whether or not air is a dispersive medium, which occurred to us as a possibility midway through the semester. A dispersive medium is one in which the speed of sound wave propagation varies with frequency, and this would cause obvious problems for our project. For example, for a speaker at a fixed location projecting the inverted drum signal, the original signal will travel some fixed distance to the speaker to be combined with the processed signal. If, however, different frequencies are travelling at different velocities, some frequency components will arrive at the speaker out of phase with the inverted signal and will potentially amplify instead of cancel.

However, we did not at the time know if air was, indeed, a dispersive medium. To test this, we would have to measure the relative phase differences of different frequencies travelling through the air. However, we would not be able to measure this without also introducing some additional phase delay associated with the electronics. In order to compensate for this, we began by measuring the phase delay associated with just the electronics by eliminating the phase delay of the air as well as we could. We did this by placing our microphone directly next to the Marshall

speaker (photos given at end of chapter), and input various waveforms into the speaker using the function generator. The output of our microphone was fed into the mixer, and we then measured the output of the function generator and the output of the mixer on separate channels of the oscilloscope. For each frequency tested, we measured the average phase shift associated with the system by recording the phase shift for ten trial runs and averaging the results. Using these values, we calculated the various slopes associated with the phase vs. frequency plots by using the equation

$$\Delta\text{phase}/\Delta\text{frequency} = \text{slope } (^{\circ}/\text{Hz}) \quad \text{Equation (3)}$$

and from these, we can calculate the various approximate time delays associated with each frequency by using the equation

$$\text{slope}/360^{\circ} = \text{time delay (s)} \quad \text{Equation (4)}$$

Tables of these measured values for various frequencies can be found in the Figures section at the end of the chapter.

We then conducted the same measurement, this time moving the microphone away from the speaker by 10 cm. Once again, we recorded the average phase shift between the output of the function generator and the output of the mixer, and then calculated the phase shift associated with air by using the equation

$$\text{Phase (d = 10 cm)} - \text{Phase (d = 0 cm)} = \text{Phase due to air} \quad \text{Equation (6)}$$

We also calculated what the expected phase shift would be in air, assuming a constant sound speed of 343.2 m/s. These results are tabulated at the end of the chapter as well.

We conducted the experiment once again, this time changing the distance between the microphone and speaker to be 20 cm, and calculated average phase values of the system, average phase values of the air (by using Equation (x) and using Phase (d = 20 cm) instead of Phase (d = 10 cm)), and expected phase values assuming a constant speed of sound. These results are also displayed at the end of the chapter.

As we can see, not only are the average phase values associated with the air quite a bit different than the expected values, but the time delays for different frequencies associated with the electronics are quite dissimilar as well. This means that, unfortunately, both air dispersion and non-linearity within the phase response of our system immensely complicated our project.

In view of this and the fact that we had such little time left in the semester, we concluded that we would not be able cancel out all frequencies within the drum signal. However, we determined that if we could cancel a single frequency or narrow band of frequencies, we may still yield valuable results.

To begin testing whether or not we could achieve cancellation of a single frequency, we started by aiming the Marshall speaker and the Behringer speaker directly at each other and input a signal from the function generator into the Marshall speaker. This signal was then collected by the SM57 microphone, placed in between the two speakers, and then fed into the mixer and

inverter. The output of the inverter was then fed into the Behringer speaker, and we then used the Senior Design Lab's sound level meter to measure attenuation at different points between the speakers. We did this by placing the meter at nine distinct locations between the speakers and measuring the SPL at each location for varying amplitudes of the inverted wave. Results indicated that for all frequencies, we would not achieve attenuation at all points between the two speakers, but we would instead achieve attenuation at a few select points and achieve amplification elsewhere. After briefly studying the very basic science behind sound wave propagation, we came to the conclusion that waves of the same frequency travelling in opposite directions, regardless of phase shift between them, will form standing waves between the sources with certain discrete points called "nodes," where complete cancellation will be achieved. As a result of this study, we determined to direct both the original signal and processed signal in the same direction, so as to achieve cancellation in a line, as opposed to a point.

We then attempted to cancel the drum signal and measuring the results with the sound level meter placed ten feet away from the drum. First, we needed to measure the sound pressure level produced by the drum itself. Our set-up for this experiment included placing a microphone near the drum, feeding the microphone into the mixer, feeding the output of the mixer into the inverter, and connecting the output of the inverter into the Behringer speaker. We then struck the drum by dropping our bouncy ball down the tube and observed the results on the meter. After sampling 30 separate readings, we then averaged these together and computed the standard deviation of values (average = 76.24 dB, standard deviation = 4.23 dB). Unfortunately, we felt that the variance in values of the sound level meter indicated that it was too sensitive to consistently provide accurate measurements, so we then proceeded to take measurements using the oscilloscope.

Once we were assured of a more accurate and consistent way of measuring data, we were able to begin trying to attenuate the drum signal. Once again, since we were not going to be able to cancel all frequencies, our goal was to cancel out a single frequency (or narrow band of frequencies) that was most prominent within the drum. We then determined the phase shifts associated with our system for each of the frequencies calculated earlier, and then based on those phase shifts, calculated the distances the speaker would have to be placed from the microphone in order to cancel out these frequencies, using the equation

$$(343.2/\text{frequency}) * [(\text{phase shift})/360^\circ] = \text{distance} \quad \text{Equation (7)}$$

Note that in this equation, we assume the speed of sound is constant at 343.2 m/s. Based on our experimental evidence from earlier, we know that this is not the exact speed for all frequencies; however, for an average value it works well since we will be varying the distance of the speaker over a meter, which is well over the range of distance values we calculated (values tabulated at the end of this chapter).

We then varied the distance of our speaker from our microphone from one to two meters away from our microphone in 10 cm intervals, testing for four distinct volume levels of our speaker at each location, and recording voltage graphs along with FFT plots for each location and volume of the speaker.

Unfortunately, results did not yield any noticeable attenuation of the signal. For most locations and volumes of the speaker, the spectral results indicated that no new frequency content was introduced, but in most cases the frequencies originally present in the drum were amplified. Photos are shown at the end of this chapter to further illustrate that, regrettably, there was no noticeable attenuation.

3.3 Figures for Chapter 3

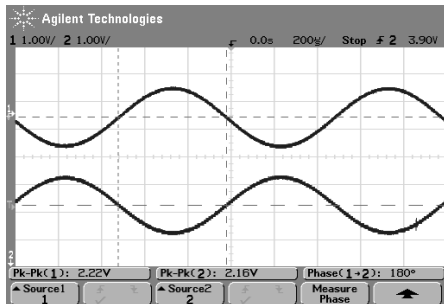


Figure 6: Inverter Test at 1 kHz, 2 Vpp

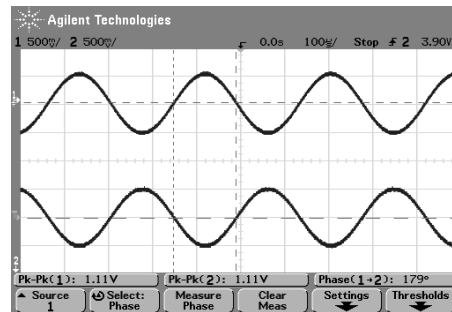


Figure 7: Inverter Test at 3.5 kHz, 1 Vpp

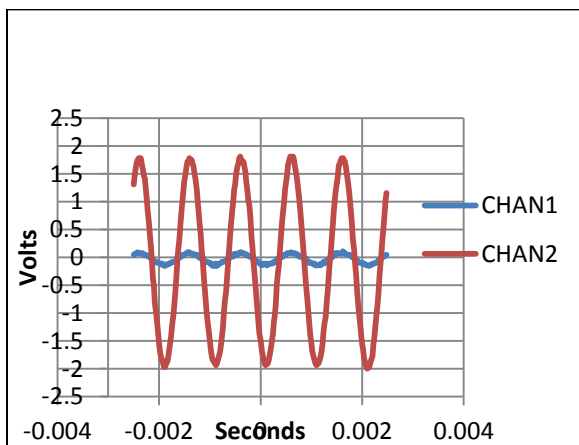


Figure 8: Pre-Amplifier Test at 1 kHz, 0.2 Vpp

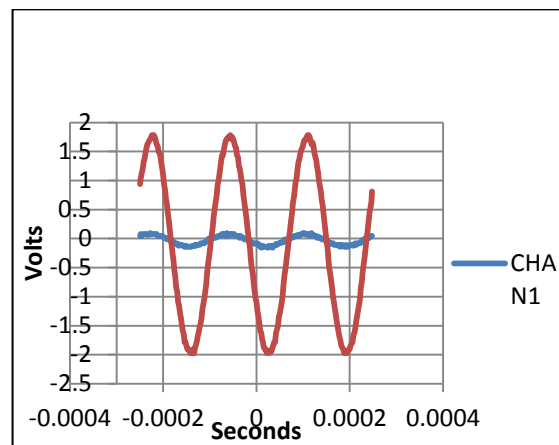


Figure 9: Pre-Amplifier Test at 550 Hz, 0.2 Vpp

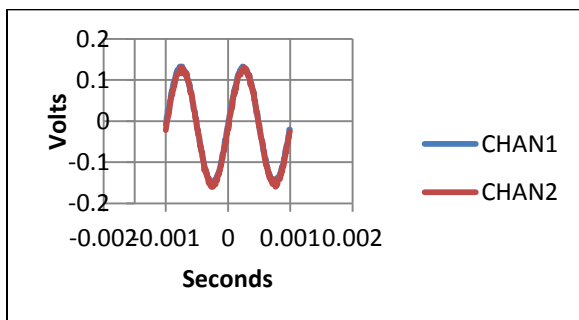


Figure10: Final Amplifier Test at Minimum Gain

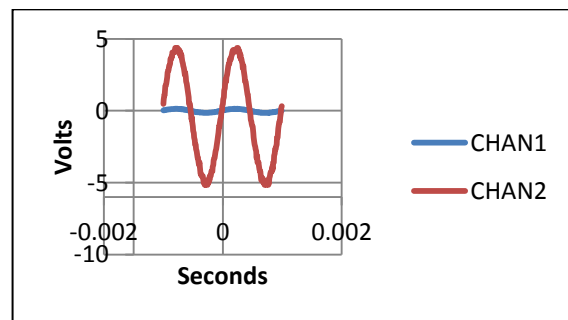


Figure 11: Final Amplifier Test at Maximum Gain

Table 1: Phase Shifts Associated with Final Amplifier

Frequency	Minimum Gain	Maximum Gain
1 kHz	0°	2°
2 kHz	0°	4°
3 kHz	-1°	8°
4 kHz	-1°	9°
5 kHz	-4°	13°
6 kHz	0°	15°
7 kHz	0°	18°
8 kHz	7°	17°
9 kHz	0°	19°
10 kHz	0°	30°

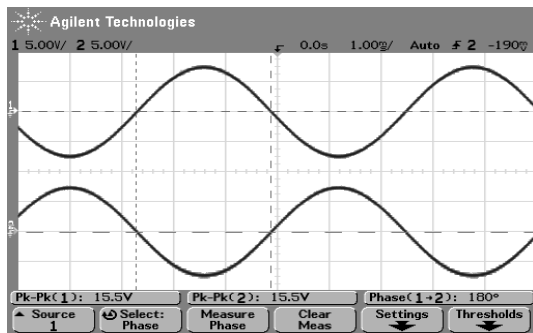


Figure 12: Inverter Re-Tested at 193 Hz, 15 Vpp

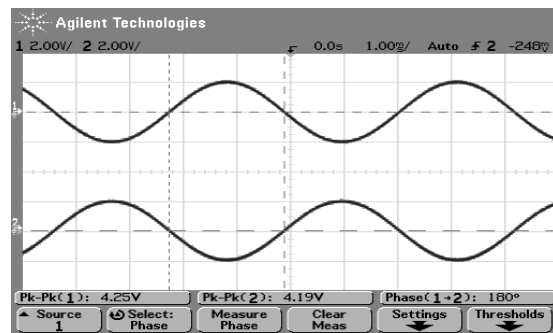


Figure 13: Inverter Re-Tested at 233 Hz, 4 Vpp

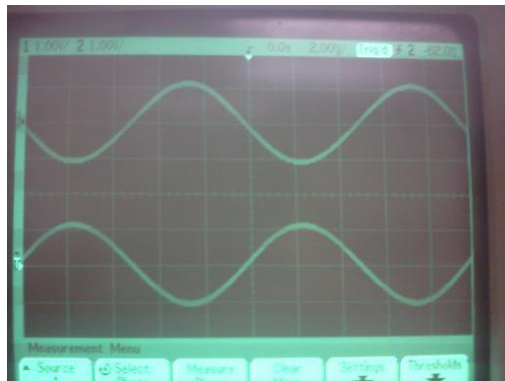


Figure 14: Inverted Signal from Microphone at 100Hz

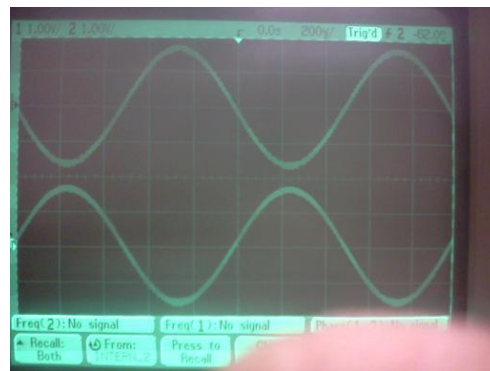


Figure 15: Inverted Signal from Microphone at 1 kHz



Figures 16 and 17: Images showing distance between microphone and speaker is zero (or as close as it can be)

Table 2: Average Phase Shifts, Phase vs. Frequency Slopes, and Time Delays of Electronic System

frequency	1	2	3	4	5	6	7	8	9	10	Avg	Slp	t
100Hz	78	81	78	77	77	81	80	78	79	80	79	.86	2.4
200Hz	163	161	167	164	167	166	165	164	166	165	165	.21	.57
300Hz	184	188	185	185	186	185	186	184	185	185	185	.31	.86
400Hz	215	217	218	216	216	218	214	215	216	216	216	.17	.47
500Hz	234	232	235	233	231	233	233	233	232	233	233	.19	.54

Table 3: Average Phase Shifts, Phase Delays Due To Air, and Expected Phase Due To Air for d = 10 cm

Frequency	Average	Phase delay due to air	Expected phase
100Hz	2.3	283.4	10.49
200Hz	150.3	345.5	20.98
300Hz	170.3	345	31.47
400Hz	232.9	16.8	41.96
500Hz	264.8	31.9	52.45

Table 3: Average Phase Shifts, Phase Delays Due To Air, and Expected Phase Due To Air for d = 20 cm

Frequency	Average	Phase delay due to air	Expected phase
100Hz	2.3	283.4	10.49
200Hz	150.3	345.5	20.98
300Hz	170.3	345	31.47
400Hz	232.9	16.8	41.96
500Hz	264.8	31.9	52.45

Table 4: Average Phase Shifts for Prominent Drum Frequencies; Distances from Microphone To Speaker to Cancel Frequencies

Frequency	Average	Distance (m)
193.5Hz	328	1.615
196Hz	327.1	1.602
212Hz	348.8	1.556
230Hz	207.9	0.836
236Hz	283.1	1.14

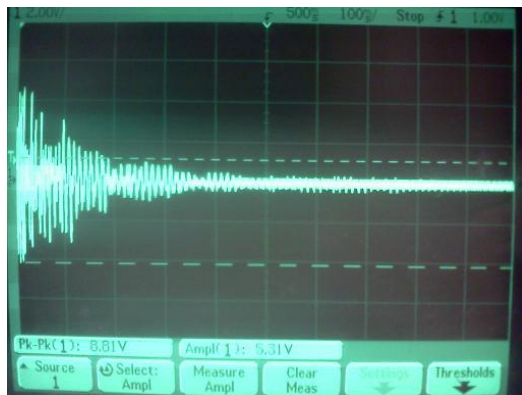


Figure 18: Drum by Itself Sampled for 1 s

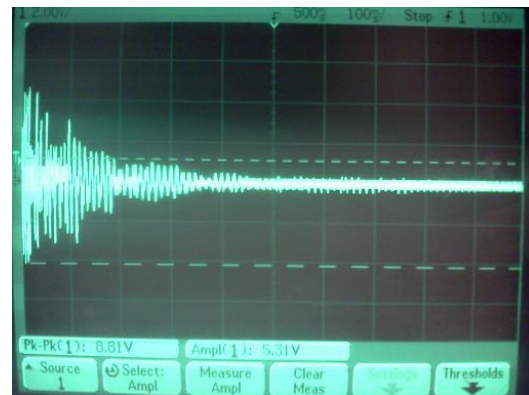


Figure 19: Drum Sampled with $d = 1.4$ m with Arbitrary Gain

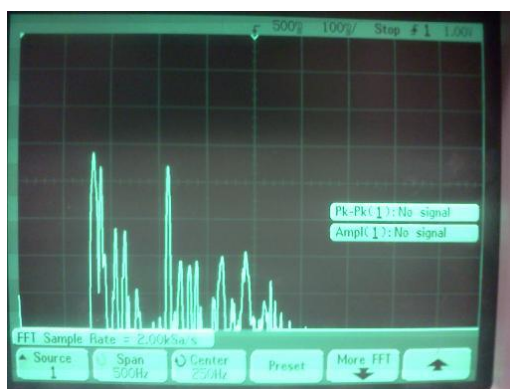


Figure 20: FFT for Drum by Itself

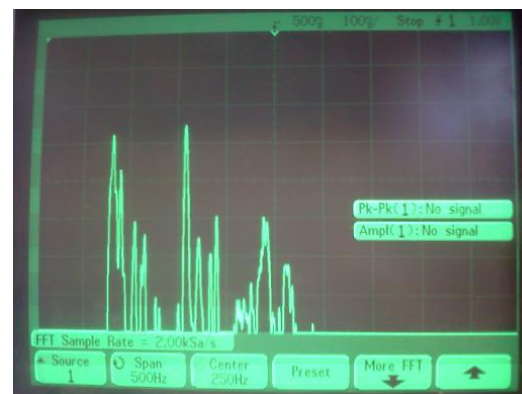


Figure 21: FFT for Drum with $d = 1.4$ m with Arbitrary Gain

4. Costs

4.1 Parts

The parts included in the cost calculation will include only the parts that were used in the final demonstration. The more expensive parts of the project were not paid for, but rather borrowed from various people; specifically, the cables, microphone, mixer and speaker were provided to us. If we neglect these items in the cost calculation, our cost would come down to \$9.02.

Module of System	Part	Manufacturer	Retail Cost (\$)	Bulk Purchase Cost (\$)	Quantity	Actual Cost (\$)
Microphone	SM57	Shure	124.00	17.00	1	132.68
Interconnects						
	XLR Cable	Cablewholesale	3.69	3.69	1	3.95
	1/4" cable	Cablewholesale	1.85	1.85	1	1.98
	Alligator Clips	Radioshack	3.19	3.19	1	3.41
Pre-Amplifier	402 VLZ3 Mixer	Mackie	99.99	99.99	1	106.99
Inverter						
	LM741CN op-amp	National	0.63	0.63	1	0.67
	20J40KE 40k resistor		2.98	2.31	2	6.38
Amplifier	BEH B215XL Speaker	Behringer	196.99	196.99	1	210.78
Total						466.33

Mics

Bulk: <http://www.tradetang.com/for-sale/Microphone-SM57-LC-SM57LC-sm57-15pcs-Dynamic-Legendary-Instrument-Vocal-with-box/132848-2655860.html>

Retail: <http://www.sweetwater.com/store/detail/SM57>

Interconnects

XLR cable: <http://www.cablewholesale.com/specs/microphone-cable/10xr-01206.htm>

¼" Cable: <http://www.cablewholesale.com/specs/audio-extension-cable/10a1-61225.htm>

Alligator Clips:

<http://www.radioshack.com/product/index.jsp?productId=2062227&camp=PPC%3AGoogle>

Pre-Amplifier

<http://www.musiciansfriend.com/pro-audio/mackie-402-vl3-compact-audio-mixer/630342000000000?src=3WWRWXGB&cagpspn=pla&gclid=CNKtvKuz4q8CFSQCQAodd3ZjDg>

Inverter

Op-Amp: <http://www.ece.illinois.edu/ecestores/>

Resistors:

<http://www.alliedelec.com/search/searchresults.aspx?dsNav=Ntk:Primary|40+kilohm+resistor|3|,Ny:False,Ro:0&dsDimensionSearch=D:40+kilohm+resistor,Dxm:All,Dxp:3&Term=40+kilohm+resistor&fromsearch=true&SearchType=0>

Amplifier

http://www.americanmusical.com/Item--i-BEH-B215XL-LIST?src=Y0802G00SRCHCAPN&gclid=CMu_3qgp4q8CFQ0DQAodtA1RAw

4.2 Labor

The labor costs for the project are calculated according to the following formula:

$(\$/\text{hour}) * (3) * (\text{hours to complete}) = \text{total labor cost}$

We estimate that all of us (three people) contributed to about 150 hours spent working on the research, design, implementation, and testing. If we give ourselves a salary of about \$35 per hour, our total cost of labor comes to: \$15,750.

5. Conclusion

5.1 Accomplishments

Though we did not achieve perfect attenuation of the drum signal, we believe that the various accomplishments we made, along with the knowledge we attained, laid the groundwork for a much more successful project in the future. One of these successes is that the system is capable of detecting and inverting sound, which ultimately is the basis of the entire project, and without which sound cancellation would be impossible. In addition to this, the techniques we developed in measuring phase delays of certain frequencies within the system and the air, the knowledge that both the original signal and processed signal must travel in the same direction, and the knowledge that, through DSP technology, it is possible to implement linear-phase filters with control over the time delay of the filter, is crucial to the work we would perform in the future to create a functional design. Also, it is comforting to know that, since the most prominent frequencies of the drum are located within a relatively narrow band, then cancellation of those frequencies may still be possible if the problems we experienced near the end of the project were not simply due to air dispersion or non-linear phase responses in the electronics.

5.2 Uncertainties

Once again, major uncertainties within this project include air being a dispersive medium and non-linear phase responses within the electronics. The unfortunate aspect of these uncertainties is that, since we do not know the exact time delay or velocity associated with each frequency, we would have to measure these values for each frequency with very accurate, precise equipment. The reason for this is that if we were trying to accomplish the same thing for a device of a much higher frequency (say, a snare drum or cymbal, for example), then the most prominent frequencies would have very short wavelengths, corresponding to much larger amplification or attenuation effects for small deviations of phase difference beyond the desired 180° .

Also, one additional uncertainty that we have not mentioned is the countless reflections off walls and other objects that sound waves inevitably encounter. We were able to disregard this for our experimentation because all of our experiments were conducted in the same room with a shape of constant form (i.e. the walls did not move, etc.) and we assumed that the effect of inaccuracies in our measurements due to reflections would be removed because of this. However, if you were to apply this technology in various locations, it would be cause for concern.

5.3 Ethical considerations

5.4 Future work

Given more time to work on this project, we would begin by acquiring all the knowledge we could on implementing linear-phase filters using DSP technology, such that we could have full control over the time delay associated with each filter. We would then make note of every single prominent frequency within the signal of the drum and test our system and the air (in exactly the way we did before) to record total phase delays associated with those frequencies. We would then calculate the relative time

delays for each frequency, make note of the longest time delay, and then implement a linear-phase filter for each frequency within the drum signal, and delay each frequency such that each delay would equal the longest delay. In this sense, we would be able to maintain our speaker in a single location and effectively cancel out all frequencies at any distance in the direction the speaker is pointing.

References

Appendix A Requirement and Verification Table

Original System Requirements, Testing and Verification:

System	System Requirements	System Tests and Verification
Low-Pass Filter	Magnitude response falls to half power (-3dB) at 7.1kHz within a margin of 100Hz	Measure output voltage of filter with oscilloscope, input system with signal from function generator, verify that voltage is half-power (or $[1/2]^{1/2}$ magnitude voltage) of the highest voltage within the pass-band between 7000 and 7200Hz.
	Linear phase between 5kHz-7kHz with margin for error of $\pm 5\%$ (slope) (explained below)	Measure output voltage of filter with oscilloscope, input system with signal from function generator, vary frequency of signal, measure delay (distance) between output signal and input signal at ten points (frequencies) between 5-7kHz, determine slope by calculating delay/frequency. Verification is identical slope throughout frequency range within $\pm 5\%$.
High-Pass Filter	Magnitude response falls to half power (-3dB) at 4.9kHz within a margin of 100Hz	Measure output voltage of filter with oscilloscope, input system with signal from function generator, verify that voltage is half-power (or $[1/2]^{1/2}$ magnitude voltage) of the highest voltage within the pass-band between 4800 and 5000Hz.
	Linear phase between 5kHz-7kHz with margin for error of $\pm 5\%$ (slope) (explained below)	Measure output voltage of filter with oscilloscope, input system with signal from function generator, vary frequency of signal, measure delay (distance) between output signal and input signal at ten points (frequencies) between 5-7kHz, determine slope by calculating delay/frequency. Verification is identical slope throughout frequency range within $\pm 5\%$.

Pre-Amplifier	Output voltage will be equal to the input voltage scaled by a factor of fifty-one within a margin for error of $\pm 5\%$ at all points within the duration of the signal	Measure output voltage with oscilloscope, input system with signal from function generator, verify that ten individual points of output voltage for five individual frequencies are equal to the input voltage at those points scaled by a factor of fifty-one with a margin for error of $\pm 5\%$.
Phase-Shifter	Output voltage will simply be the negated version of the input voltage without any gain or attenuation factor within a margin for error of $\pm 5\%$ at all points within the duration of the signal	Measure output voltage with oscilloscope, input system with signal from function generator, verify that ten distinct points of output voltage for five distinct frequencies are equal to the input voltage scaled by a factor of -1.
Amplifier	Output voltage will be able to be adjusted up to one thousand times the input voltage	Measure output voltage with oscilloscope, input system with signal from function generator at very low voltage, verify that ten distinct points of output voltage for five distinct frequencies are able to reach a level one thousand times the level of the input voltage by adjusting our potentiometer.
Drum Striking Apparatus	Consistent volume (voltage level) within $\pm 10\%$	Though not an electrical system, it imperative we receive a consistent volume from the drum. We will measure this via a microphone and oscilloscope, and measure ten distinct voltages (at identical times for each waveform) for ten distinct drum strokes. Verification will consist of corresponding voltages being identical within a margin for error of $\pm 10\%$.
Overall System	Up to 6 dB sound pressure amplitude attenuation at a distance of 10 ft in directions at angles of 45° , 0° , and -45° with respect to the axis perpendicular to plane of playing area (in other words, 6 dB of attenuation if the drum or drummer is looking at you straight on, or at	Place microphones a distance of ten feet away from drum in three directions (directly in front of drum and at angles of 45° from direction drum is facing). Measure signals from drum without sound cancellation technology by observing microphone signals on oscilloscope. Verify that signal received by microphones with sound cancellation technology can be attenuated up to 6dB (half amplitude). First verify that each individual direction can achieve this without activating all three devices, then verify that we can achieve this in all three directions. We will

	angles of 45° in either direction).	verify each test by analyzing ten “drum strokes,” measuring ten distinct points within the signals obtained from those strokes, and compare them with the signals measured from the drum alone. Test success will equal an output signal of half the amplitude of original signal.
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