Hardware and Software Study of Active Noise Cancellation

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ABSTRACT

Noise cancellation involves removing an unwanted noise while keeping the source sound. The source sound may consist of speech, music played from a device such as an iPod or a computer, or no sound at all. The objective of this project is to study the process of noise cancellation both as hardware and as software. The hardware will consist of building a noise cancelling circuit that uses headphones as an output, a microphone to pick up the noise to be cancelled and, if desired, a source sound. The software portion of the experiment will use MATLAB to simulate the hardware circuit as well as simulate other methods: filtering unwanted signal, a pi phase shift for inverting the noise signal, the least mean squares algorithm, and the recursive least squares algorithm. The hardware solution works well with periodic noise but has difficulty removing noise from non-periodic noise. The software solution using adaptive filtering works better than the hardware solution but only with periodic noise. Difficulties encountered include the kinds of noise that can be canceled and the time delay internal to the circuit.

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1. INTRODUCTION

Noise is any unwanted disturbance present in a signal. Noise cancellation can be used in areas where noise can be harmful to ones hearing, such as: engine rooms or aircraft runways. In signal processing, noise is data within the wanted signal that carries no real value. Typically, signals are stronger without noise which gives a better signal to noise ratio The study of cancelling noise from a wanted signal arises from need to achieve stronger signal to noise ratios. There are two techniques for cancelling noise: passive noise reduction and active noise cancellation. [7]

Headphones that utilize passive noise reduction often use material that blocks some sound waves from entering the user's ears. The best headphones for passive noise reduction are circum-aural (i.e. covering ears completely). Circum-aural headphones block more incoming sound waves due to more layers of high density foam. [7] The disadvantage of passive noise reduction is not all of the ambient noise will be cancelled. Some of the noise, no matter the method for passive noise cancellation, will make it to the user's ear. In contrast, active noise cancellation employs a different approach to noise reduction.

Headphones that utilize active noise cancellation apply different techniques. Active noise cancellation involves creating a supplementary signal that deconstructively interferes with the outside, ambient noise. [7] The disadvantage of active noise cancellation is the compromise made in audio quality as well as the price. Whether the process is done hardware or software with a DSP chip, the unwanted signal and source signal will share the same frequency content, leading to cancellation of both.

In theory, the best approach to cancelling noise would be to take the noise signal, invert it, and add the input and inverted signals such that they add deconstructively. This is the basic approach used for noise cancelling headphones. This project explores this basic idea of noise cancellation from a hardware and software approach.

2. HARDWARE APPROACH

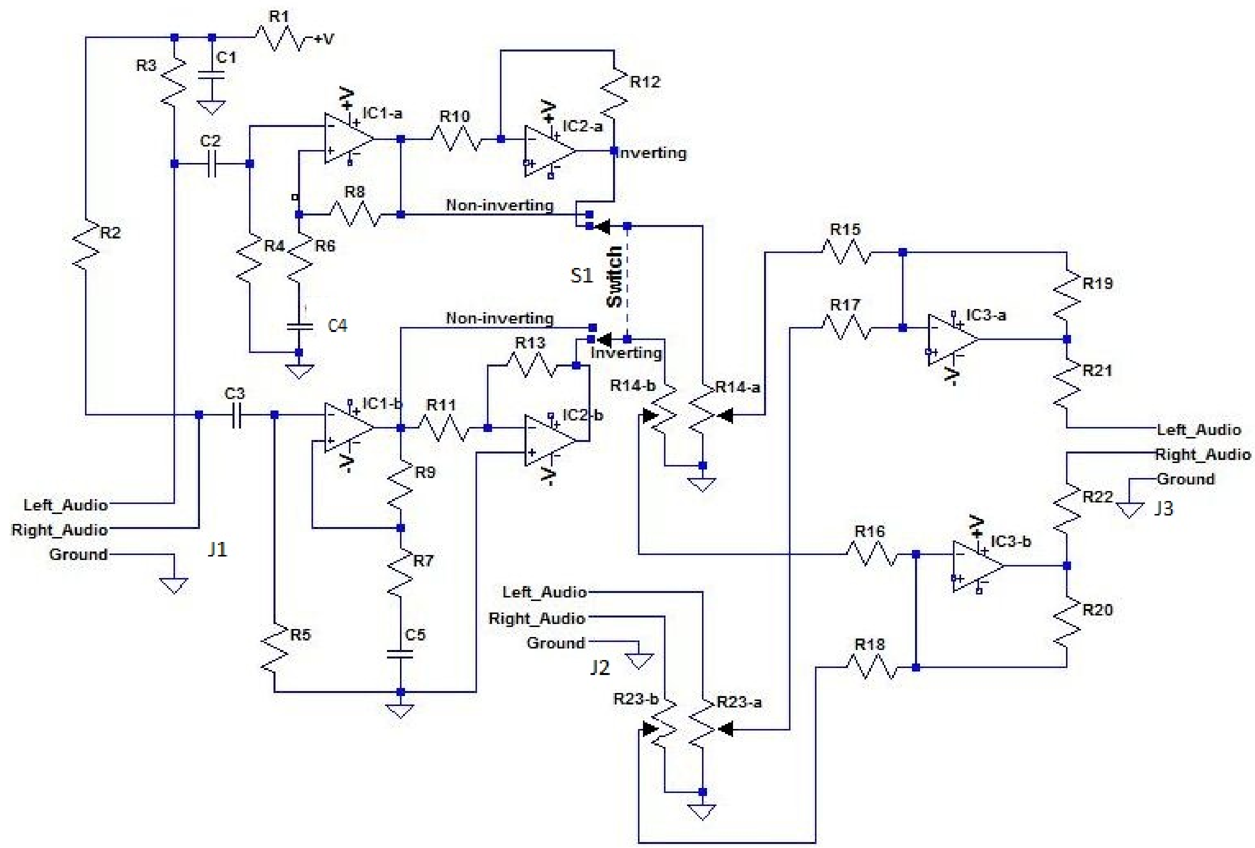

Figure 1: Circuit Diagram of Noise Cancelling Hardware

Table I: Selected Components Used for Noise Cancelling Hardware (Figure 1)

Label	Component	Value
R_1	Resistor	4.7kΩ
R_{2-3}	Resistor	2.2kΩ
R_{4-5}	Resistor	1MΩ
R_{6-7}	Resistor	1kΩ
R_{8-9}	Resistor	33kΩ
R_{10-13}, R_{15-18}	Resistor	10kΩ
R_{14}, R_{23}	Dual-Gang Potentiometer	100kΩ
R_{19-20}	Resistor	100kΩ
R_{21-22}	Resistor	47Ω
IC_{1-3}	Integrated Circuit	NE5532
C_1	Electrolytic Capacitor	33μF
C_{2-3}	Mylar Capacitor	0.01μF
C_{4-5}	Electrolytic Capacitor	10μF
S_1	DPDT Switch	
J_{1-3}	1/8 Inch Audio Jack	

2.1. Components:

R_1 and C_1 decouple the bias voltage from the power supply by being placed in a voltage dividing network. R_8 and R_6 were chosen to be 33kΩ and 1kΩ for a gain of 31dB for the amplifying stage. A large value for R_4 was chosen (1MΩ) to give a good ground reference for the amplifying stage. C_2/R_4 and C_4/R_6 are high-pass filters blocking any DC before the amplifier. Lower capacitor values are chosen for these, 0.01µF. The next stage is the unity gain inverting amplifiers. So 10kΩ resistors were chosen for R_{10} and R_{12} to keep unity gain. The gain of the last stage (summing stage) is set by R_{19} and R_{15}, which were chosen to be 100kΩ and 10kΩ. R_{17} is added to create a summing amplifier, chosen to be 10kΩ to interact with the 100kΩ potentiometer logarithmically. Since this is stereo there is a right and left, so configuration and set up is mirrored for the opposite side. [10]

2.2. Idea behind the Circuit:

The noise cancelling circuit is composed of three separate stages. The first stage ($IC_{1a\text{-}b}$) is a non-inverting amplifier stage. This stage simply takes the input from the audio jack (J1) and amplifies the signal. Amplifying this signal is crucial since the microphone produces a smaller signal that is difficult to work with. The gain of this stage is set by one plus R_8 and R_6 (33kΩ and 1kΩ for a gain of approximately 31dB).

The second stage of the circuit ($IC_{2a\text{-}b}$) is a phase inverting configuration that inverts the amplified signal from the first stage. This stage is supposed to invert the phase so that the output added with the ambient sound from outside the headphones deconstructively interferes.

The DPDT switch (S_1) was added to help with the timing delay issues the circuit has. That is the time at which the user hears the noise from outside the headphones must match the time at which the circuit is producing the inverting noise. The switch is to help with this timing issue. Switching the switch to non-inverting or inverting can help with the timing of the ambient noise the user hears. This is determined by the user, if the noise is quieter in the non-inverting position then the delay was significant enough to cause it to be out of phase with the ambient noise. The distance the microphone is from the user's ear also helps with the timing delays. These delays are produced from the signal traveling through the circuit and back to the user's ears.

The potentiometer (R14) is used to attenuate the signal from the second stage (microphone). Noise heard by the user will most likely be muffled due to the headphones covering a portion of the ear. Attenuating the signal produced by the microphone will help match the amplitudes of the ambient noise and the output of the circuit.

Last stage of the noise cancelling circuit is a summing, non-inverting, amplifying stage. The gain of this circuit is set by R19 and R15, and R17 is essential to making this stage a summing op-amp. Summing is only used when the user is also listening to music (plugged into audio jack, J2). Modified noise from the second stage and the music from J2 will be added together and sent to the output at J3. This allows the user to hear their music while the ambient noise is still being cancelled by deconstructive interference.

In summary, the noise signal is amplified and then inverted. This noise is produced by a microphone connected to audio jack, J1. Amplified and inverted noise is produced at the output of the circuit, thus, deconstructively interfering with the ambient noise.

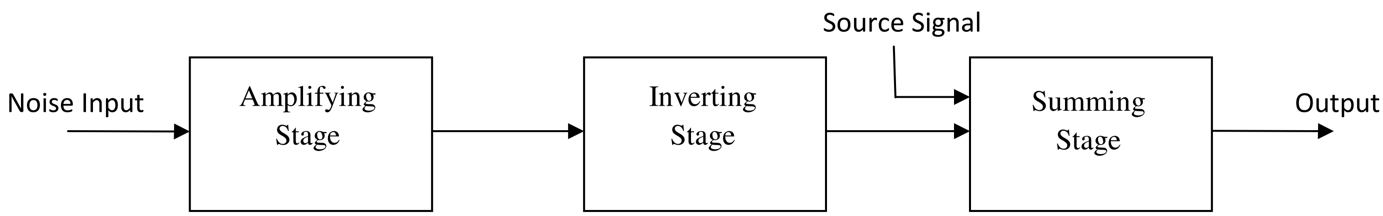

Figure 2: Block Diagram of Noise Cancelling Circuit

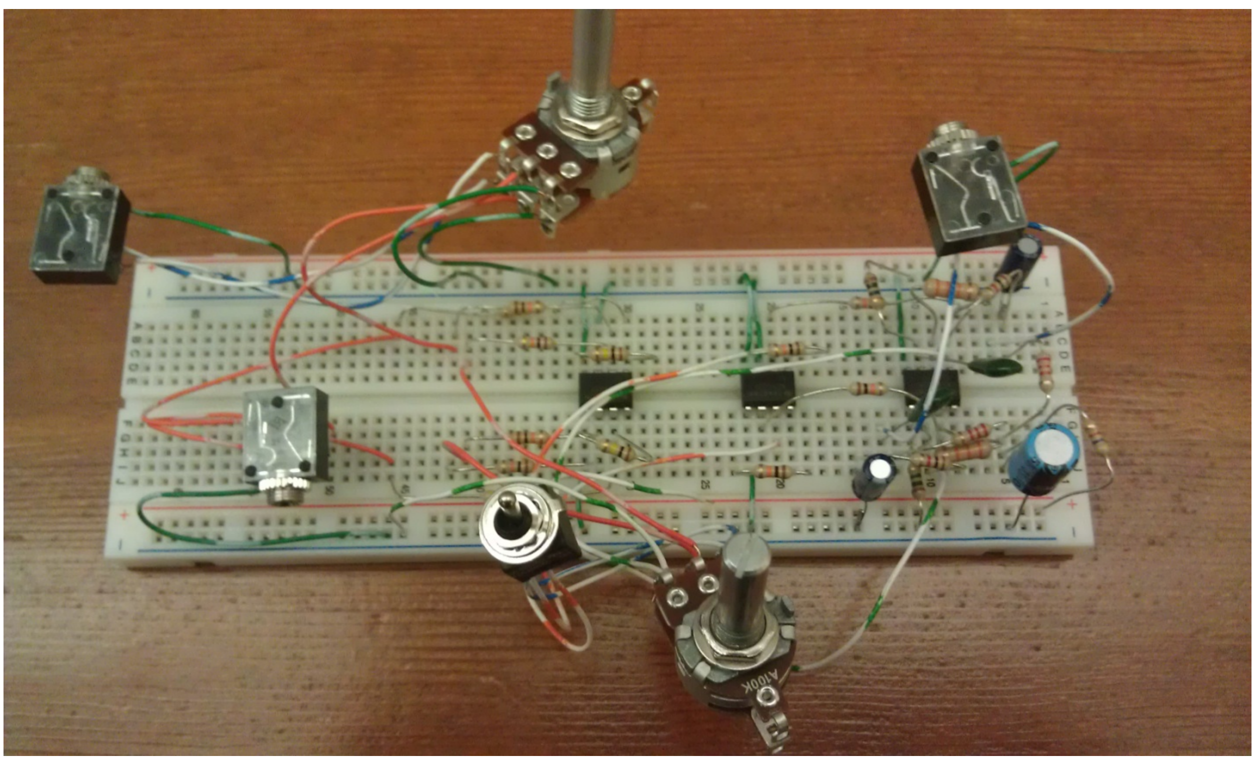

Figure 3: Completed Construction of Noise Cancelling Circuit

2.3. Testing and Results

The circuit is essentially a mirror image of itself. Due to being stereo noise, the circuit has a left and right side, contributing to the left and right outputs of headphones. This allows testing to be conducted on a single side.

First tested the left-side of the circuit amplifier stage with a 100 mV_{pp} 1 kHz sine wave inputted on J1. This value had to be small (100mVpp) due to clipping. Clipping occurred at an input of roughly 140mV_{pp}. The wave got amplified to a 6.75V_{pp} 1 kHz sine wave.

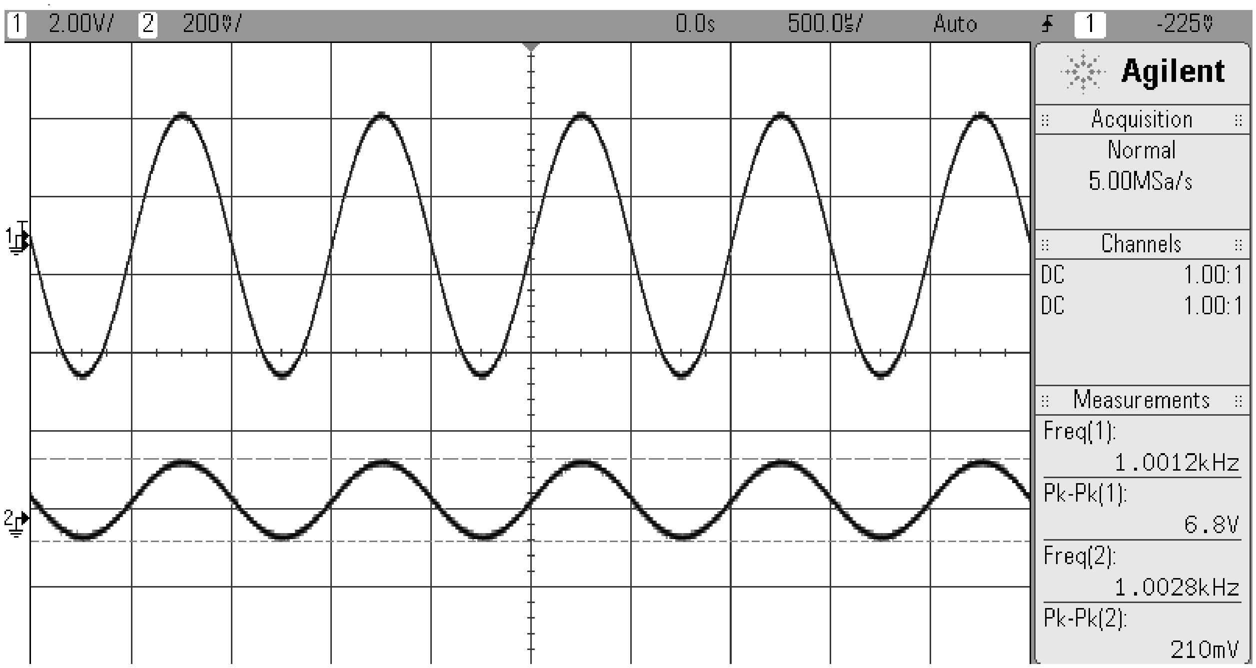

Figure 4: Left-Side Output Stage 1 (channel 1: output after amplification; channel 2: input signal)

Then the right-side amplifier stage was tested using the same input, 100 mV_{pp} 1 kHz sine wave. The signal was amplified to 6.85 V_{pp} 1 kHz sine wave. Clipping occurred at an input of 140 mV_{pp}.

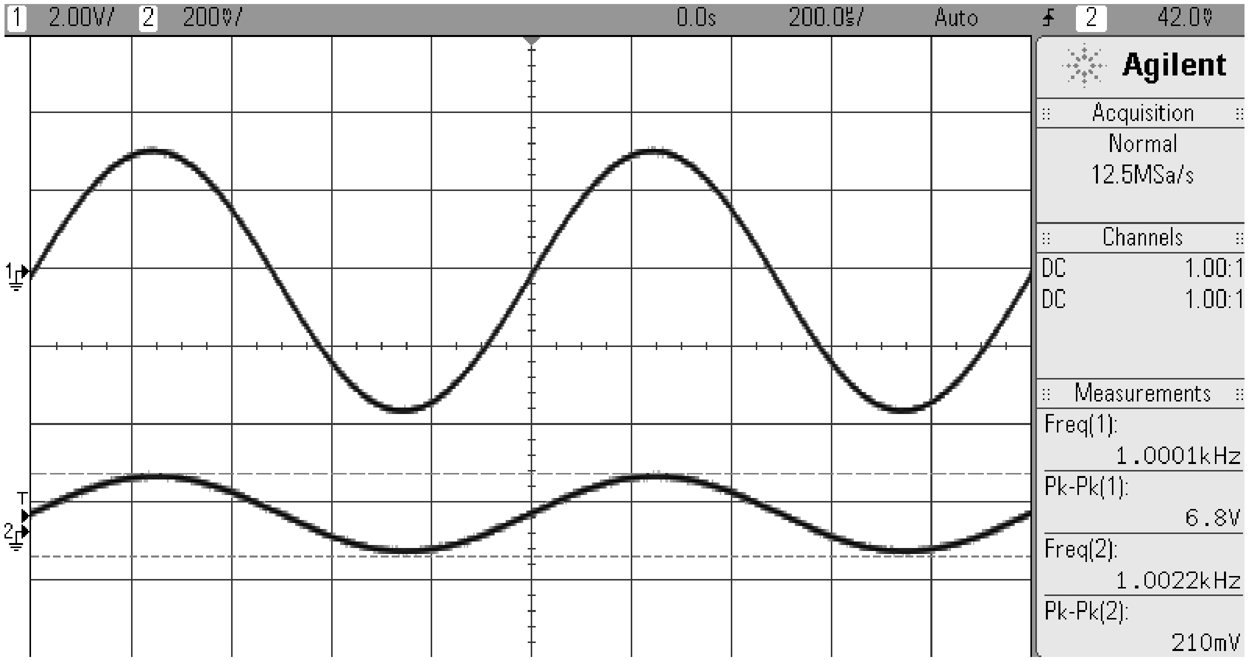

Figure 5: Right-Side Output Stage 1 (channel 1: output after amplification; channel 2: input signal)

Second stage inversion was then tested using the same signal, 100 mV_{pp}1 kHz sine wave. Left-side was first tested for inversion. Clipping occurred at 140 mV_{pp}.

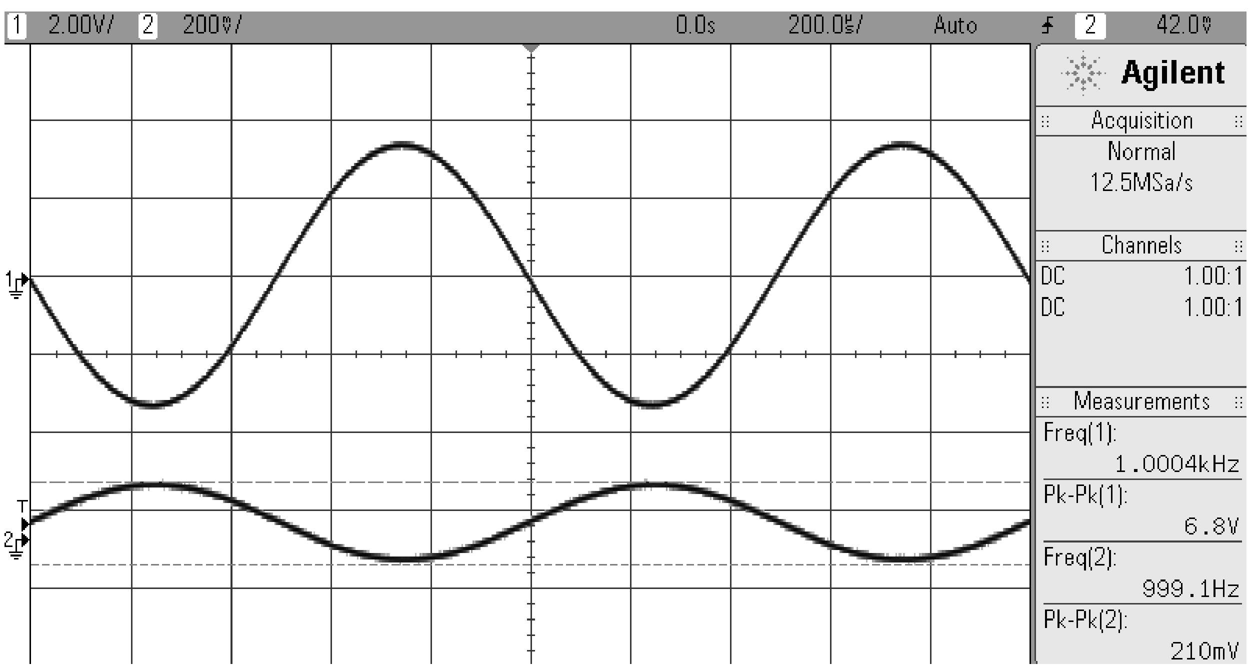

Figure 6: Left-Side Output Stage 2 (channel 1: output after inversion; channel 2: input signal)

Right-side, second stage inversion texted with the same input, 100mV 1 kHz sine wave. Clipping occurred at 140 mV_{pp}.

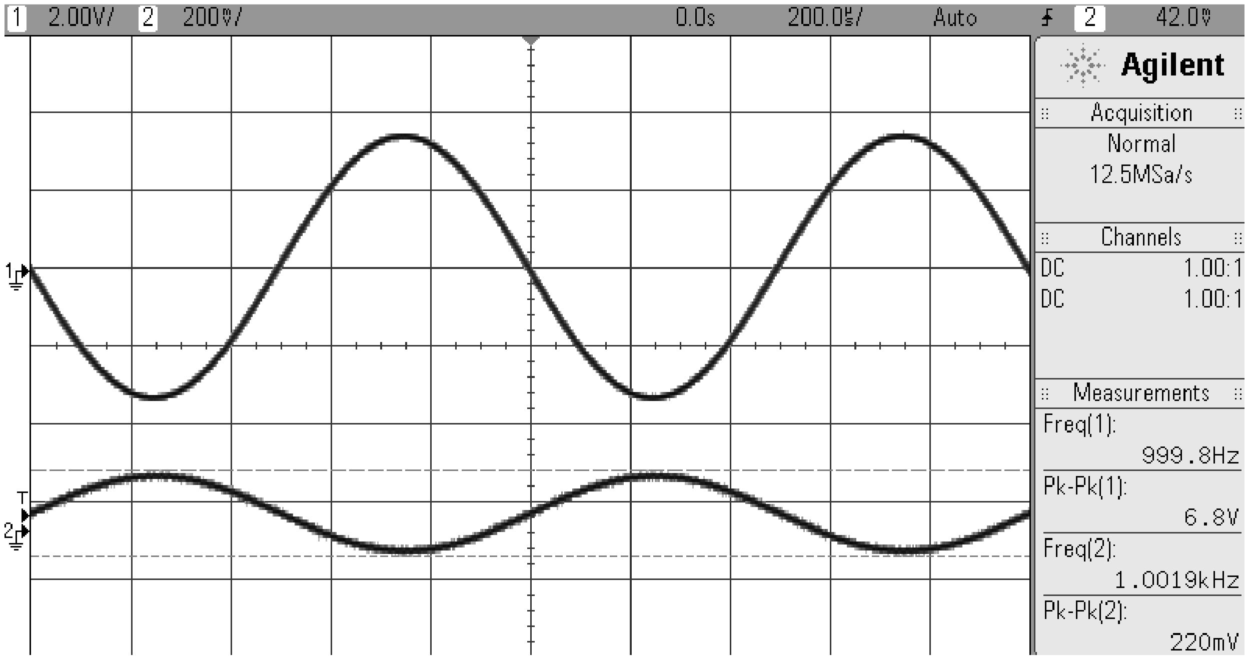

Figure 7: Right-Side Output Stage 2 (channel 1: output after inversion; channel 2: input signal)

With the input only on left-side and another source connected (J2), this is done to show that the summing stage of the hardware functions properly. We see an inverted signal amplified according to the attenuation of the potentiometer (R14) with the signal from J2 added to the input signal from J1. The sine wave appears noisy due to the input signal of white noise from J1 being added to a sine wave from J2.

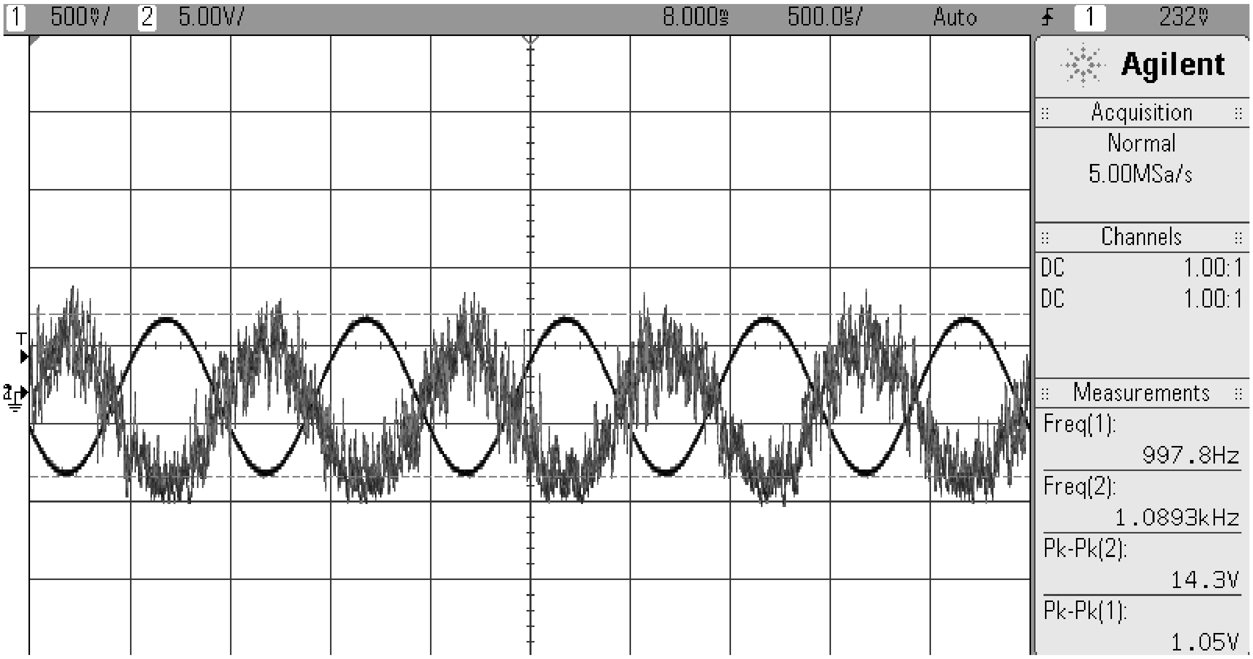

Figure 8: Left-side Output Stage 3 (channel 1: output after noise signal (white noise) summed with source signal (sine wave); channel 2: input source signal (sine wave))

With the input only on right-side and another source connected (J2). We see an inverted signal amplified according to the attenuation of the potentiometer (R14) with the signal from J2 added to the input signal from J1. Noisy sine wave is seen on output for the same reason it is seen on the left side output (figure 8).

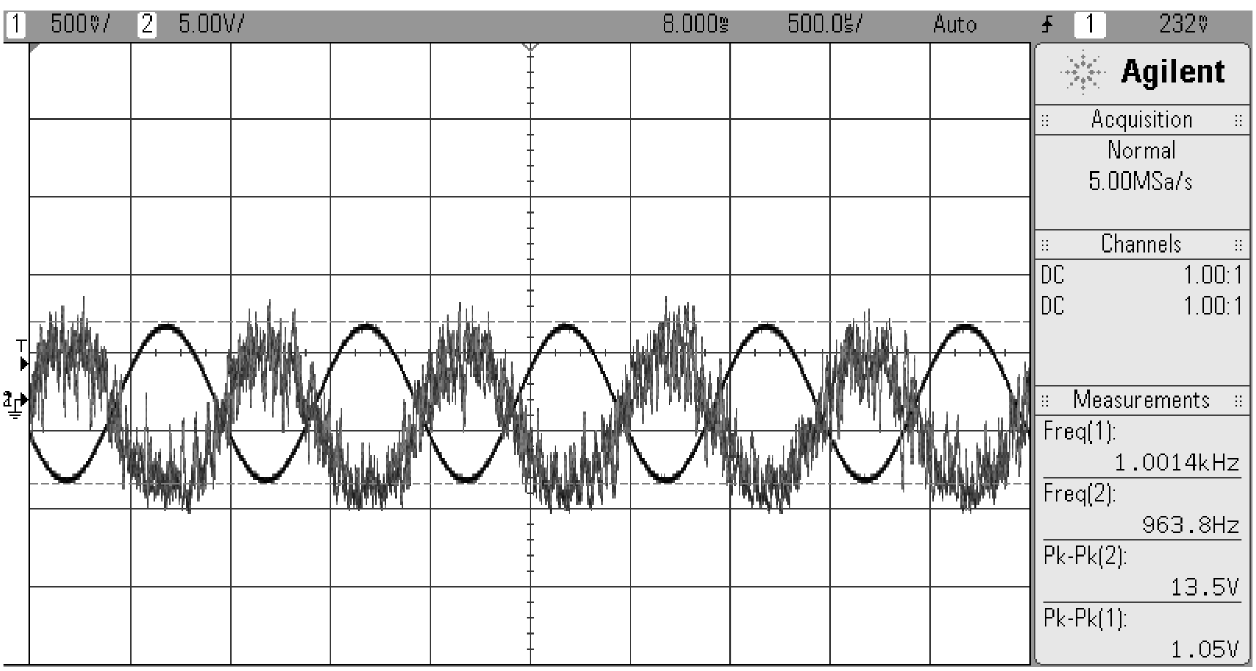

Figure 9: Right-Side Output Stage 3 (channel 1: output after noise signal (white noise) summed with source signal (sine wave); channel 2: input source signal (sine wave))

For field testing a sine wave, treated as noise, was played through PC speakers, a microphone placed near the user's ear was plugged into J1. Turtle Beach X12 Earforce headphones were used, these were plugged into J3 (output audio jack). The sine wave (noise) from the headphones is loud, louder than the outside sine wave. This is due to the attenuation of the input signal from the microphone. For configuring, start with the circuit in the non-inverted position, that is the switch (S1) must be switched to allow the first stage to enter the potentiometer (R14). Attenuate the microphone input using the potentiometer until the least amount of noise is heard from the headphones (typically as low as possible without completely shutting off channel). At this point switch to the inverting stage. This process is subjective to the user and may vary for different individuals.

This method was done several times using different sine waves and different configurations for the microphone. With the microphone placed as close to the PC speakers as possible, better results were achieved. The inverting stage (stage 2) produced a much quieter output than the non-inverting stage (stage 1) just as it had previously done for the microphone being closer to the user's ear.

2.4. Troubleshooting

There were many difficulties with this noise cancelling circuit design. The design itself is intuitive and straight forward. One of the biggest obstacles with this design is understanding that it does not perfectly cancel ambient noise. This cannot be achieved with the full range of frequencies heard (20-20 kHz). Market bought noise cancelling headphones typically work well for certain environments, working well where there are high or low frequencies. For example a pair of Bose Quiet Comfort 15, work well for frequencies up to 1 kHz, at which point their ability to cancel noise fails. [3]

Another trouble encountered was the DPDT switch. Several switches were burned out and lead to incorrect results. This was caused by overheating the switches while soldering the wires on the leads for connection to the breadboard. When switching to inverting stage (stage 2) the switch wasn't making correct contact leading to no noise cancellation and wasted time. With a more rigid DPDT switch this problem was quickly alleviated.

An additional hardware problem was finding microphones that were able to pick up the signal. Condenser microphones were used; these microphones were mounted to the headphones with each microphone on either side of the headphones. The thought behind this is that the microphones mounted directly to the headphones would help with the timing delays involved in the circuit. After testing, concluded that the condenser microphones were not picking up enough of the signal to allow the circuit to cancel. For the remainder of the experiment and testing two microphones were used. One microphone was an amplified microphone attached to the Turtle Beach X12 Earforce headphones. This microphone worked well for testing with a microphone that is place near the user’s ear. The other microphone used was a Logitech microphone; this microphone can be place in many different places.

Timing delays were the biggest problem with this noise cancelling circuit. The timing delay is associated with the time the signal takes to travel through the circuit and back to the user’s ear. Ambient noise outside the headphones is going to reach the user sooner than the signal traveling through the circuit. To offset this delay the DPDT switch was added to the circuit. The idea behind this switch is to switch which stage is going to be passed through the circuit, which is the amplifying stage or the inverting stage. Timing delay might be large enough to where the non-inverting signal is already out of phase with the ambient sound heard by the user. This scenario doesn't occur often; typically the inverting stage is best. The timing delay through the circuit is not significant enough to cause the non-inverting signal to be out of phase with the ambient noise heard by the user.

To alleviate the strain on the delay times, different microphone configurations were used. The first configuration was with the microphone placed near the user's ear; the second configuration was with the microphone placed near the noise source. With the microphone close to the user's ear, this would cause the time delay between the user hearing the signal and microphone detecting the signal to have the least amount of delay. This configuration worked well using the cancelling approach described above. A separate Logitech microphone was used and place close to the signal source (PC speakers). This configuration also worked well. The difference between these two configurations is the adjustment that will need to be done. The attenuation done by the potentiometer needs to be calibrated according to the position of the microphone.

2.5. Future Work

The first improvement to the circuit would be designing a printed circuit board (PCB). A PCB would help alleviate the circuit noise. The circuit noise being the open components on the breadboard (see Figure 2 above) touching or moving causing noise on the output that is clearly heard on the headphones. Using a surface mount PCB would miniaturize this circuit making it easy to add to the headphones.

Using higher quality audio op-amps would be a huge improvement to the circuit. NE5532 were used for this circuit because of the lower voltage that the IC needs to operate. Higher quality audio op-amps require higher voltages to turn on. This higher voltage would help the sound quality of the circuit; the higher voltage would eliminate low voltage clipping (see testing and results section for clipping present).

3. SOFTWARE APPROACH

MATLAB was used to simulate the hardware portion of the study. In order to simulate properly, the same microphone used in hardware must be used to record various noises. Five different types of tests were conducted. The first consisted of simulating the inversion of noise and adding two signals deconstructively. The second test consisted of creating filters so retain or remove specific frequencies. Test #3 involved using the fast Fourier transform to do a phase shift which will invert a signal similar to the method of test #1. For test #4 and test #5, adaptive filters were used; they consist of using the least mean squares and recursive least squares methods. The types of noise studied include: 100 Hz, 250 Hz, 440 Hz, 1 kHz, 10 kHz sinusoids, a quiet room, a room with talking students, hum from a computer, an electric fan, a fume extractor, a heat gun, typing on a keyboard, and jingling keys. The input/source signals consisted of six different speech .wav files that were recordings of different voices. For this report, only the tests involving a non-periodic heat gun noise and a periodic 1 kHz sinusoid will be shown to avoid redundancy. The desired/source signal used for this study will be one speech signal that consists of a male voice saying, "In a world we must defend." However, the following section will contain samples of other noise sources to give an idea of the frequency spectrum for different signals.

Noise Samples

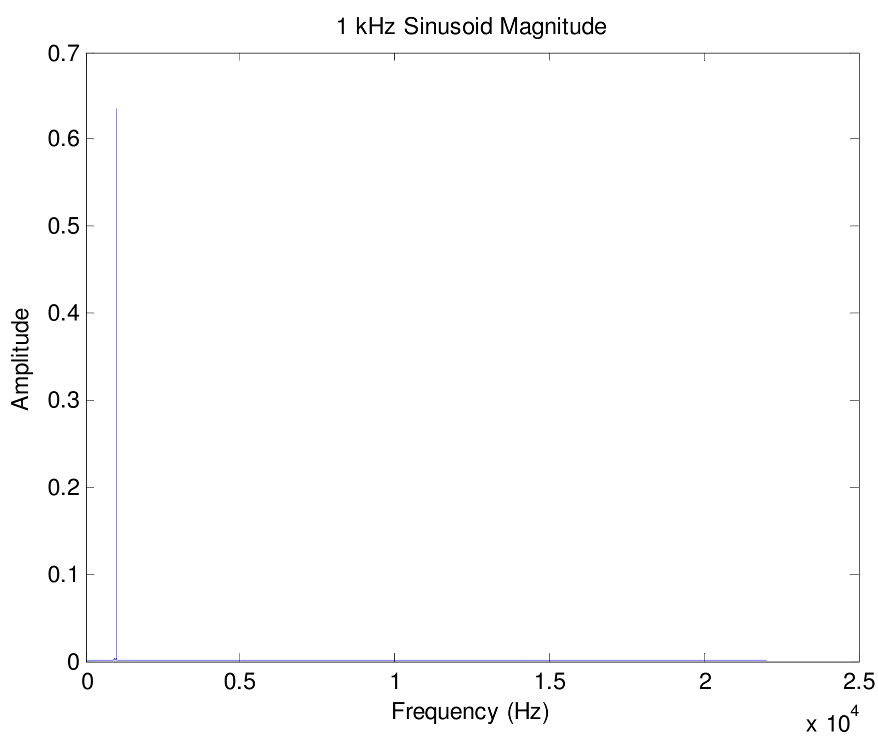

Figure 10: Magnitude Frequency Response of 1 kHz Sinusoid

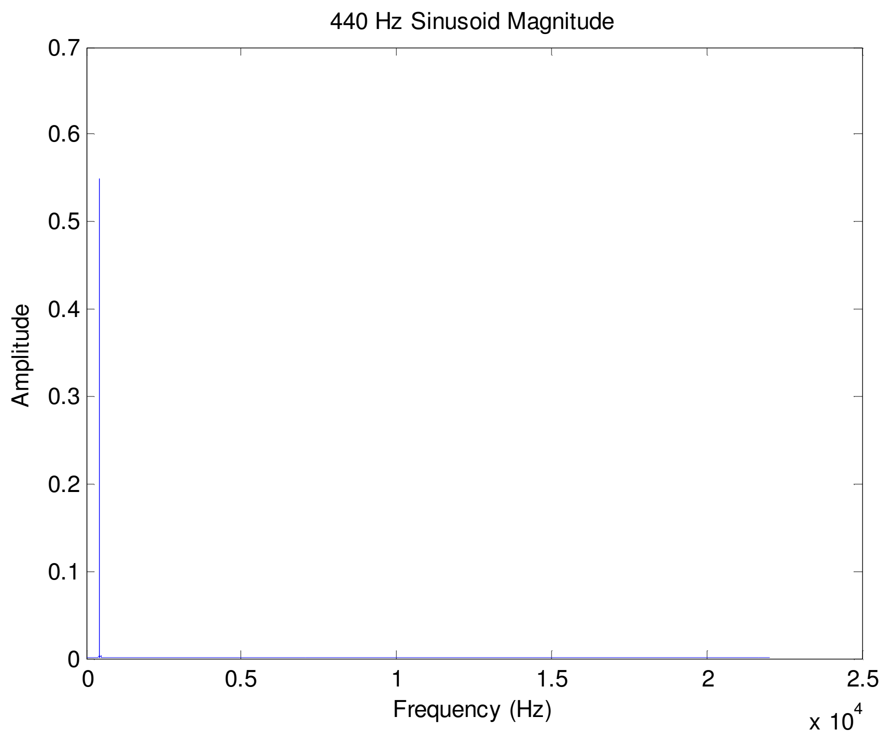

Figure 11: Magnitude Frequency Response of 440 Hz Sinusoid

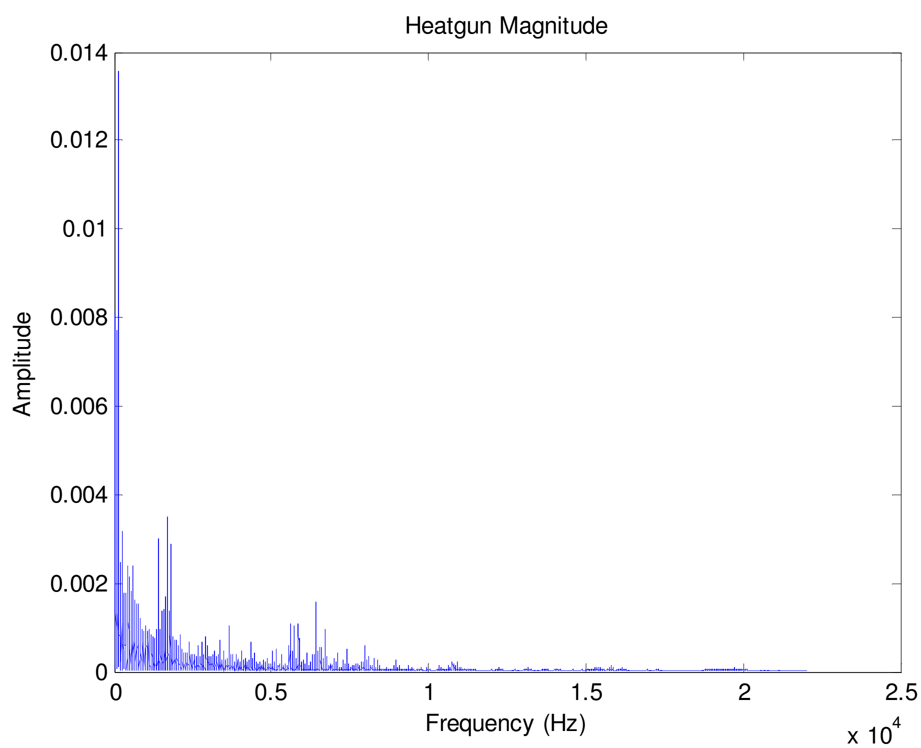

Figure 12: Magnitude Frequency Response of Heat Gun

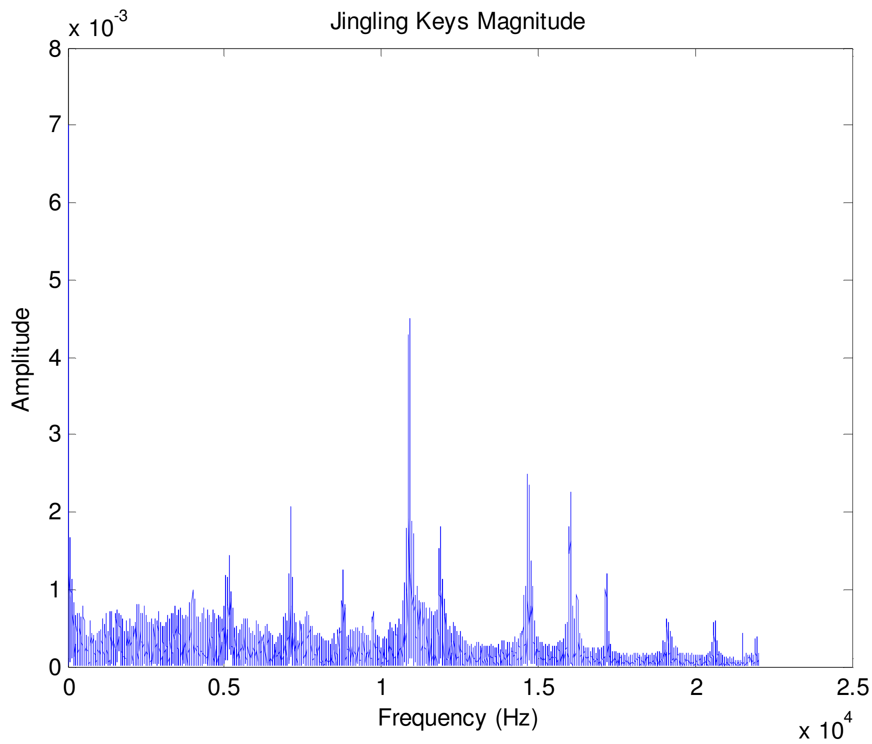

Figure 13: Magnitude Frequency Response of Jingling Keys

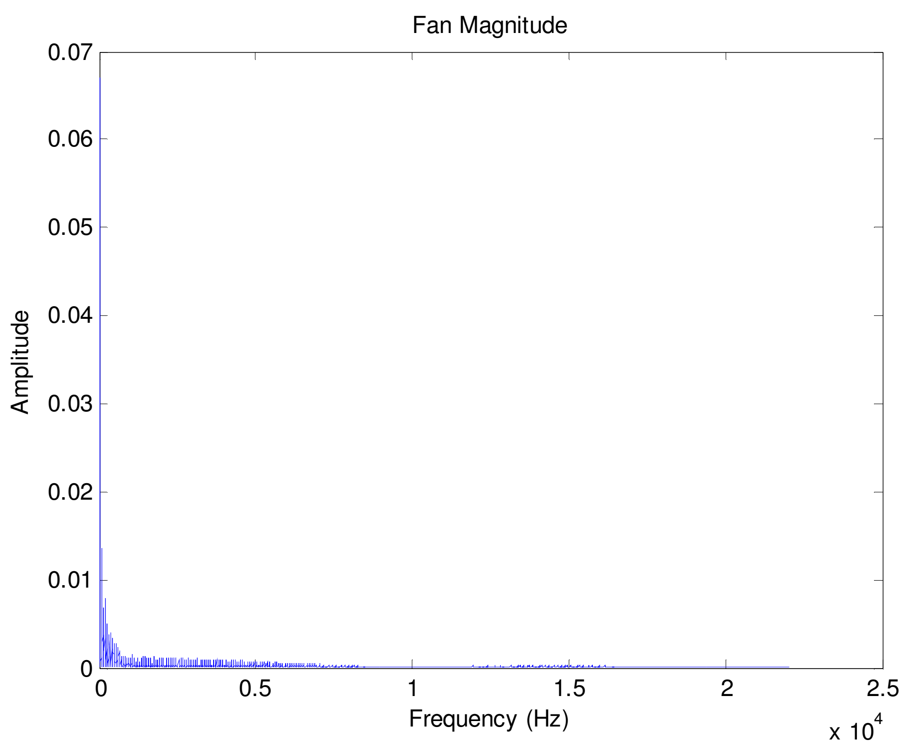

Figure 14: Magnitude Frequency Response of an Electric Fan

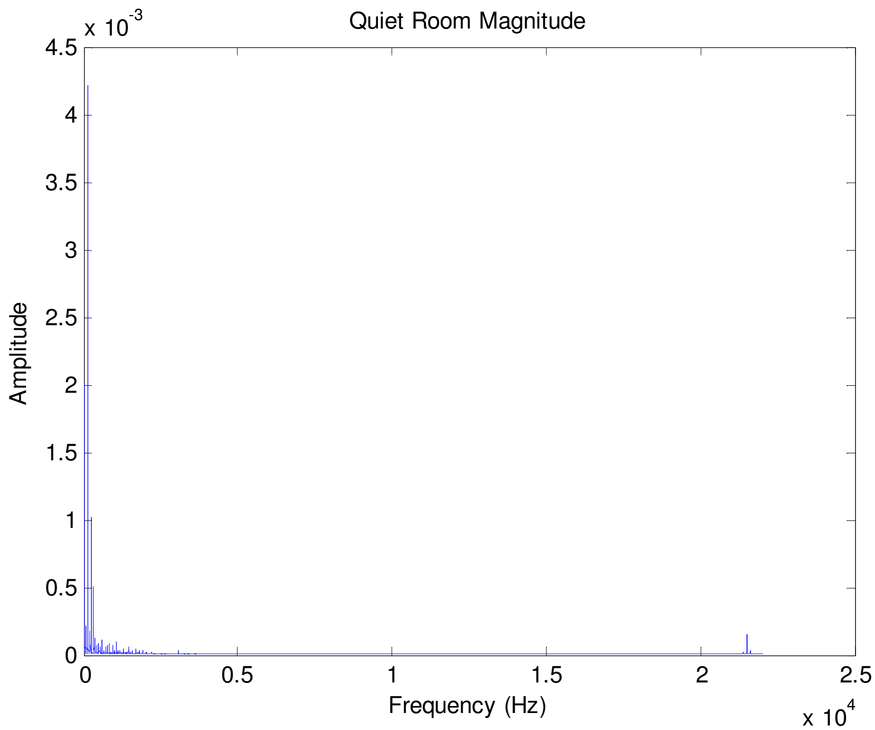

Figure 15: Magnitude Frequency Response of a Quiet Room

Speech Sample

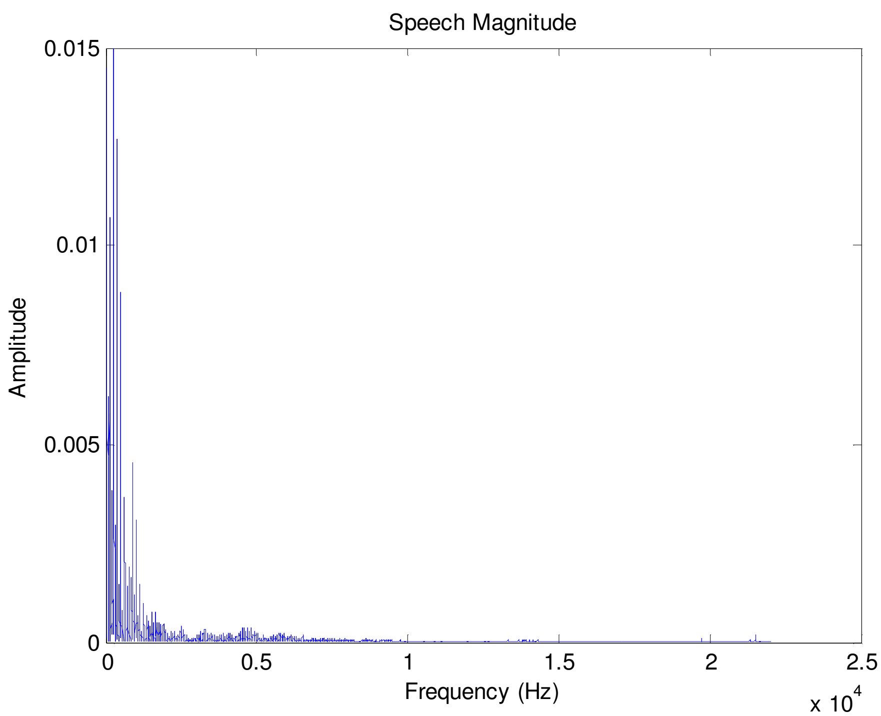

Figure 16: Magnitude Frequency Response of a Speech .wav File

From figures 10 and 11, it can be seen that the magnitude response of a sinusoid lies on its current frequency. To cancel the noise from a sine wave, a filter could be implemented at specific frequencies to cancel out (notch filter) or avoid (low-pass, high-pass) the noise of a sine wave. For magnitude responses like in figure 12 (heat gun) and 14 (fan), most of their signal lies in the lower frequencies. A high-pass filter with a cutoff frequency above 2 kHz could theoretically cancel most of the noise if the source signal would not be affected. However, from figure 15, the majority of the desired speech signal lies at lower frequencies. For noise like jingling keys from figure 13, a simple filter would not work well due to overlap with the frequency spectrum of the speech signal.

3.1. Test #1: Simulating the Hardware Circuit

The first test involved importing a noise .wav file and a speech .wav file. Since both sources were separated into two channels, MATLAB was used to add them together. Recalling the hardware portion, the first stage was used to amplify the noise, the second stage was used to invert the noise, and the third stage added the noise with its inverted signal.

Since this was a simulation, the first stage was neglected due to amplification not being required within the software method. The second stage was simulated by multiplying the noise signal by -1. Finally, the third stage added the inverted noise signal to the original signal which contained the speech and noise.

The desired speech signal was recorded using the hardware microphone in a quiet room. The duration of the signal is approximately 4.5 seconds with the actual speech starting at 1.25 seconds and ending at 3 seconds.

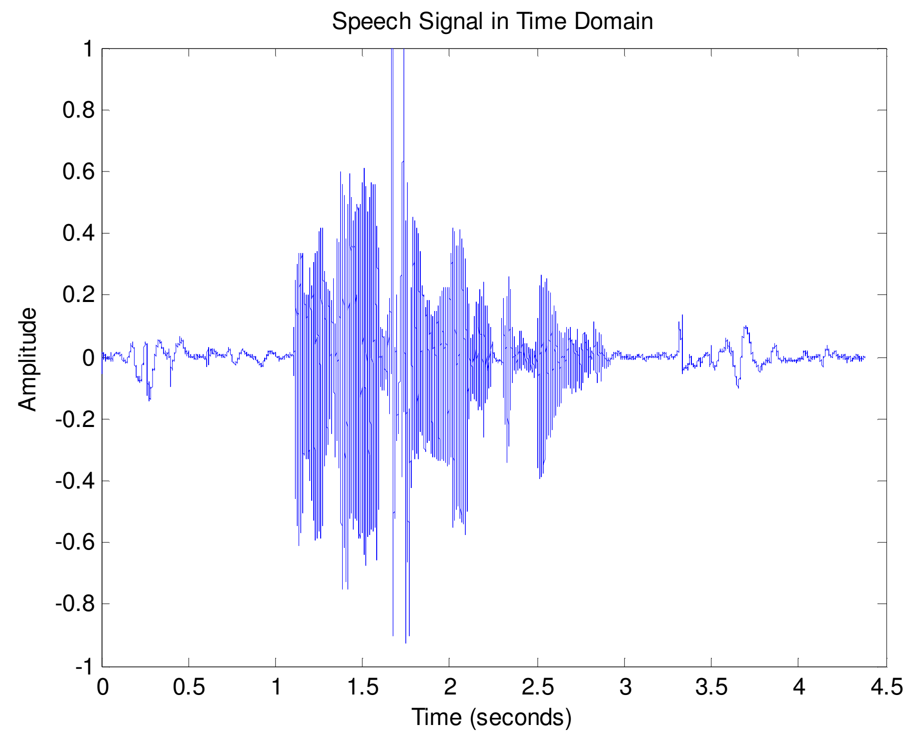

Figure 17: Speech Waveform

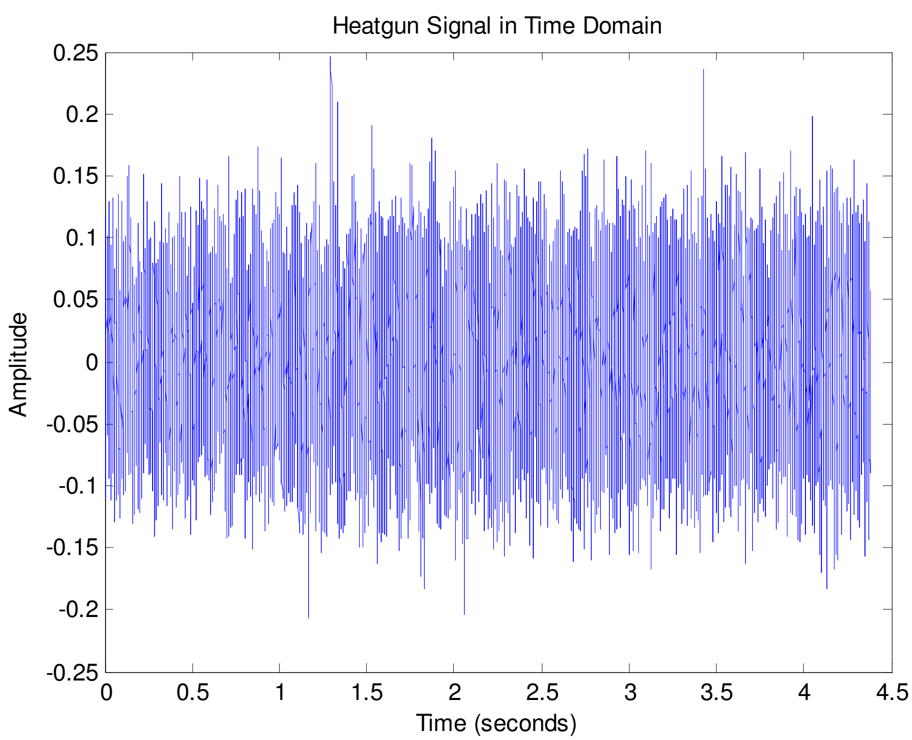

Figure 18: Heat Gun Waveform

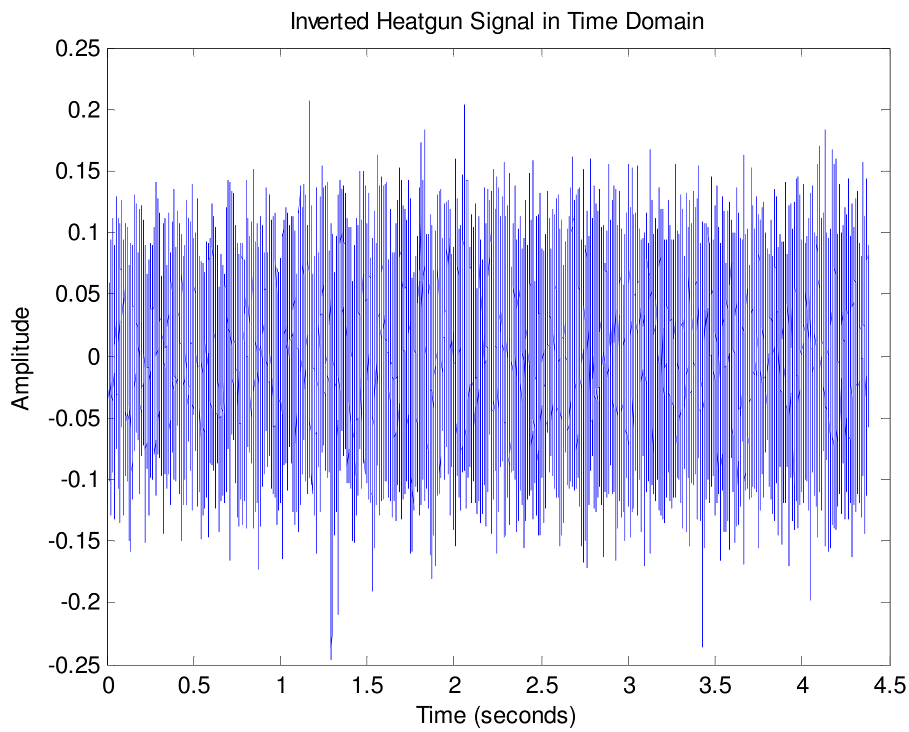

Figure 19: Inverted Heat Gun Waveform

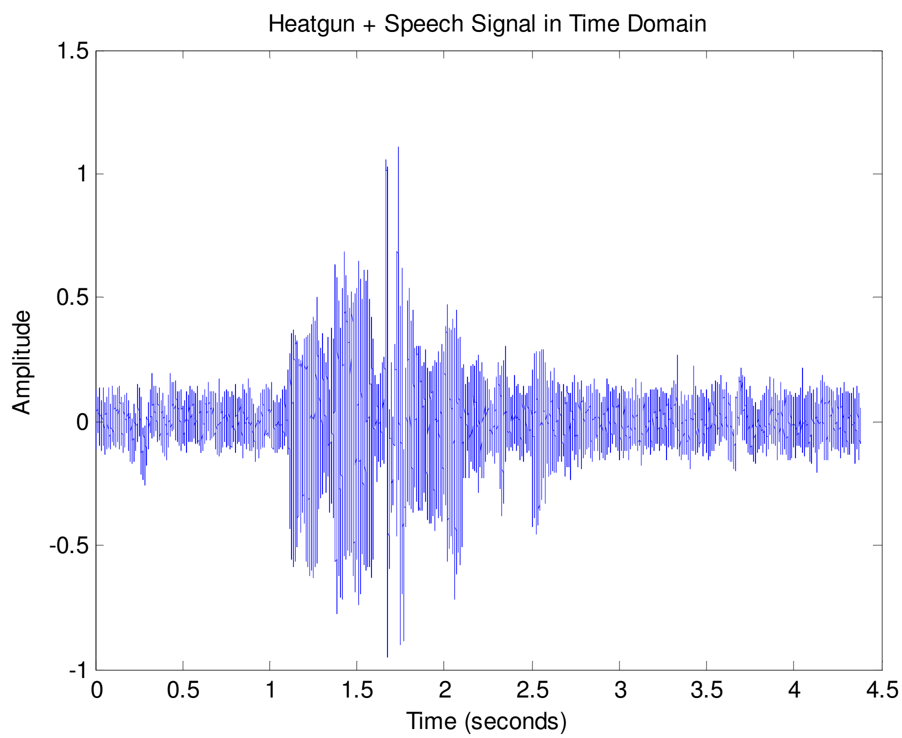

Figure 20: Combined Heat Gun and Speech Waveform

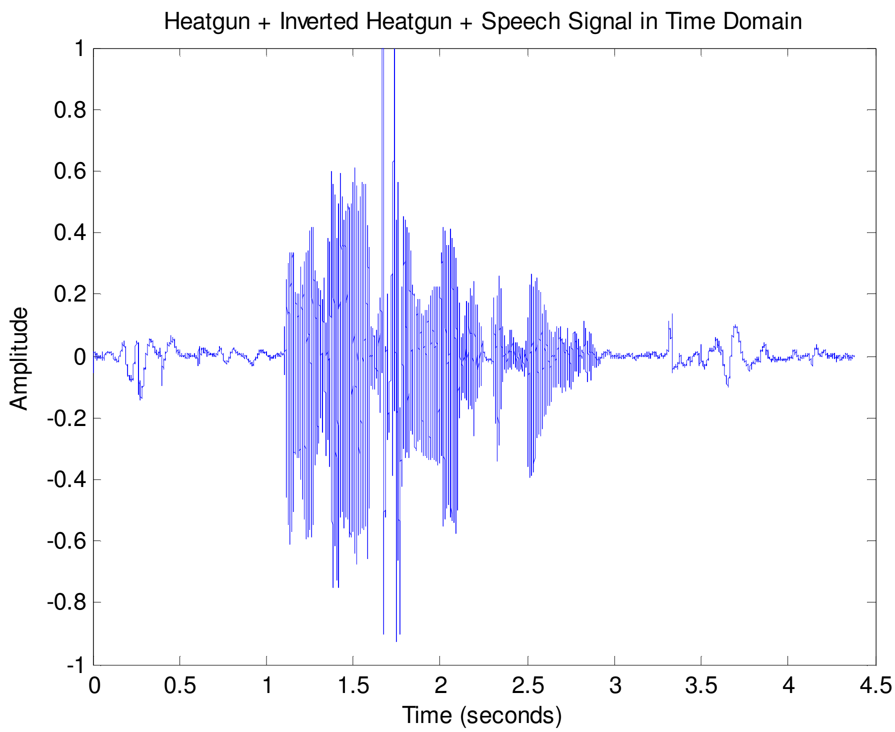

Figure 21: Summation of Heat Gun, Inverted Heat Gun, and Speech Output Waveform

As can be seen from the plots above, the noise was not only reduced but was perfectly cancelled. This was verified with the output .wav file generated by MATLAB. However, in the real world, one cannot simply multiply -1 to a signal. This problem was addressed during the next set of tests.

3.2. Test #2: Designing a Low-Pass

By using the fast Fourier transform (FFT) one can see where the frequencies of various signals lie. With this information, a filter can be built to remove or retain a specific range of frequencies. Fortunately, MATLAB contains both an FFT and an iFFT function (inverse fast Fourier transform) [6]. From the FFT plot of the speech sample in figure 16, one can determine that most of the signal is contained at frequencies lower than 2 kHz.

With this information, a low-pass filter (Butterworth) with cutoff frequency of 2 kHz was designed. The butter function of MATLAB generates the filter coefficients given the order of the filter and the cutoff frequency. A filter order of two was used to reduce computation time. When filtering the speech with heat gun noise, the filter failed to remove the noise due to the heat gun's signal being present at low frequencies. When the output signal was played in MATLAB, both the speech and the heat gun were clearly heard with a noticeable decrease in volume. The decrease in volume is due to the higher frequencies being filtered out. This same problem also occurred with the 1 kHz sinusoid noise since the cutoff frequency was 2 kHz.

Transfer Function for Low-Pass Filter:

$$H(z) = \frac{0.0167 + 0.0335z^{-1} + 0.0167z^{-2}}{1 - 1.602z^{-1} + 0.669z^{-2}}$$

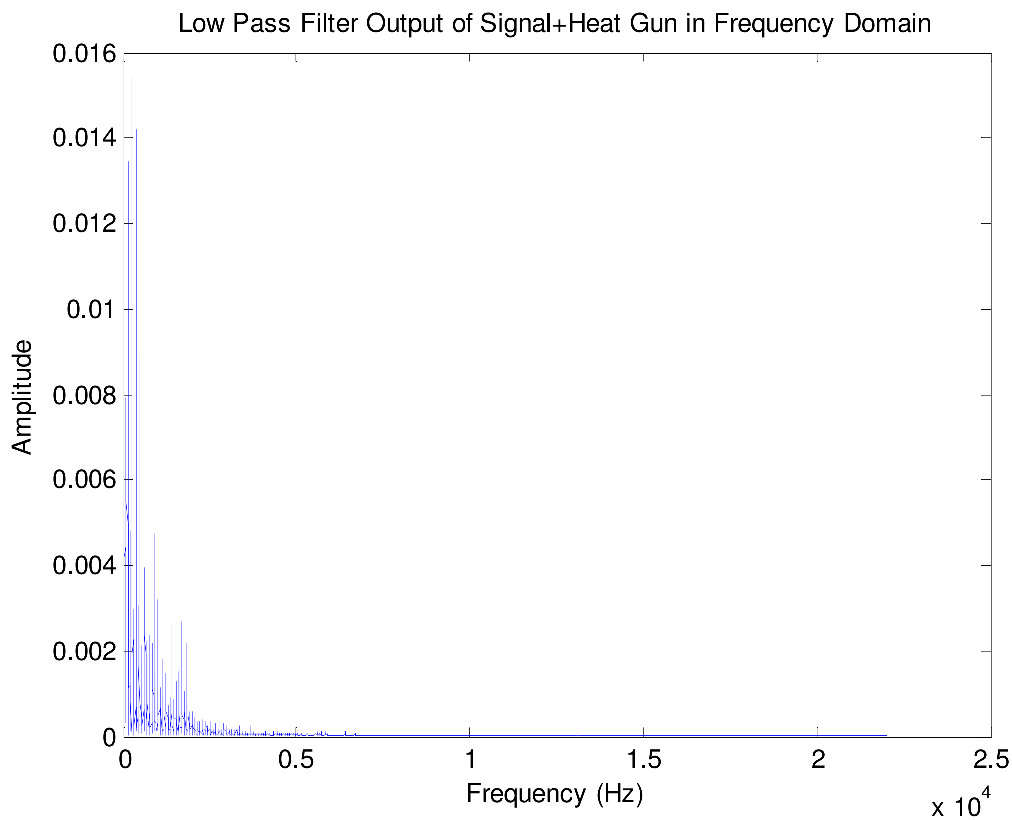

Figure 22: Magnitude Frequency Response of Low-pass Filter Output

When using sinusoids of various frequencies as the noise source, the low-pass filter successfully eliminated any noise above 2 kHz. However, in the real world, a low-pass filter would not be acceptable to cancel out noise. Using this method for noise cancellation would be very situational and designed for a specific purpose.

3.3. Test #3: Frequency Domain Phase Shift

The problem encountered during the first simulation test was how to invert a signal using a method that could be applied to the real world. The solution is to add a π phase shift. When working in the frequency domain, a π phase shift is similar to inverting a signal. These functions were used to convert the time domain signals to the frequency domain where the π phase shift could be done.

$Given\ a\ noise\ x(t)$

$Find\ \tilde{x}(t) such\ that\ x(t) +\ \tilde{x}(t) =\ 0$

$Using\ FFT$

$Let\ \tilde{X}(f) = X(f) \cdot e^{j\pi}$

$\left|\tilde{X}(f)\right| = |X(f)|$

$\angle\tilde{X}(f) =\ \angle X(f) + \pi$

$Using\ iFFT$

$x(t) + \tilde{x}(t) = 0$

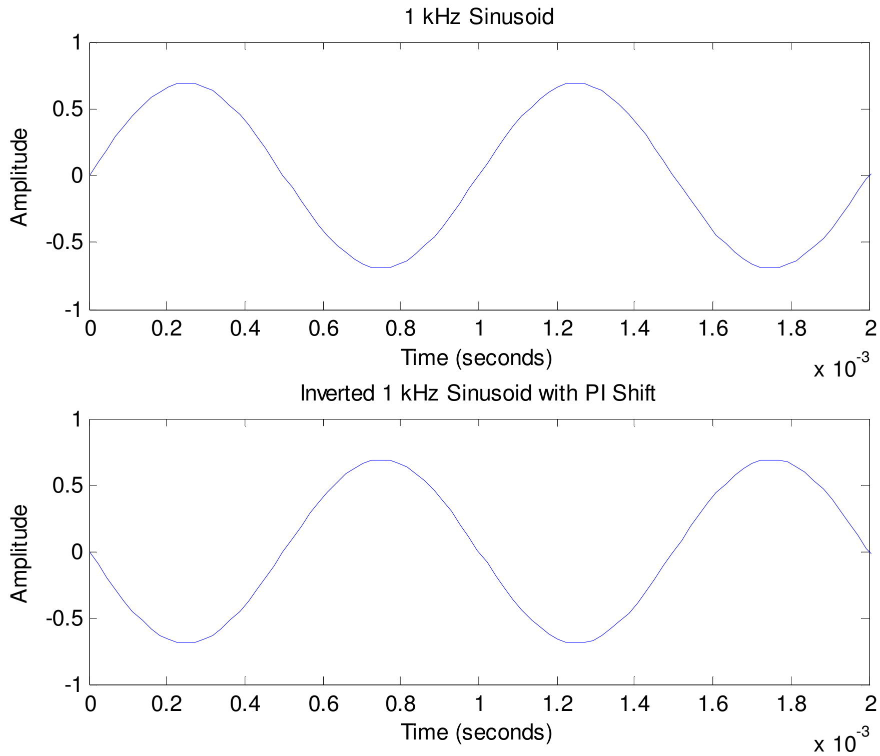

Figure 23: 1 kHz Sinusoid and its Inverse

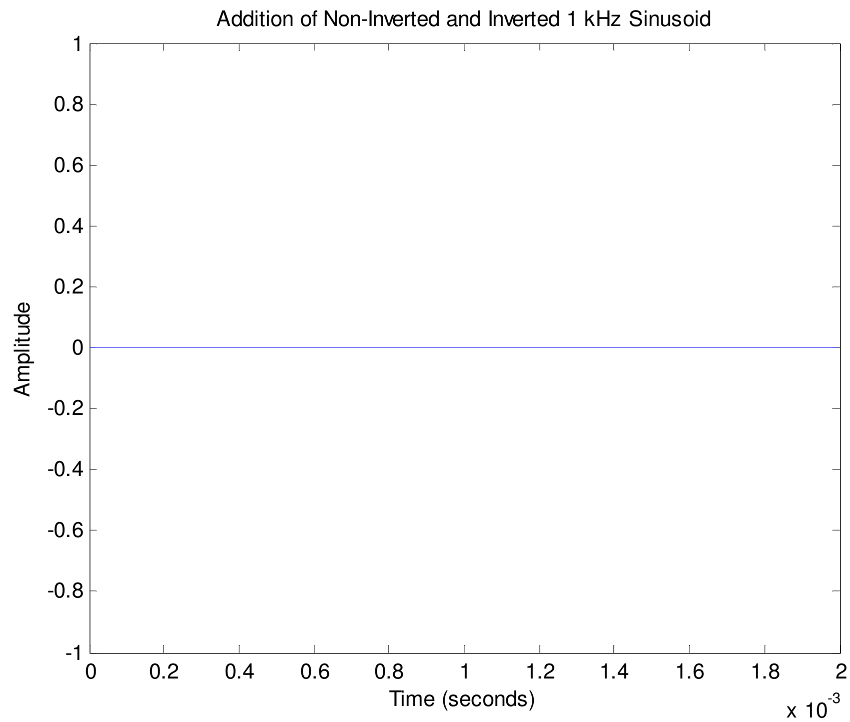

Figure 24: Summation of Non-Inverted and Inverted Sinusoid

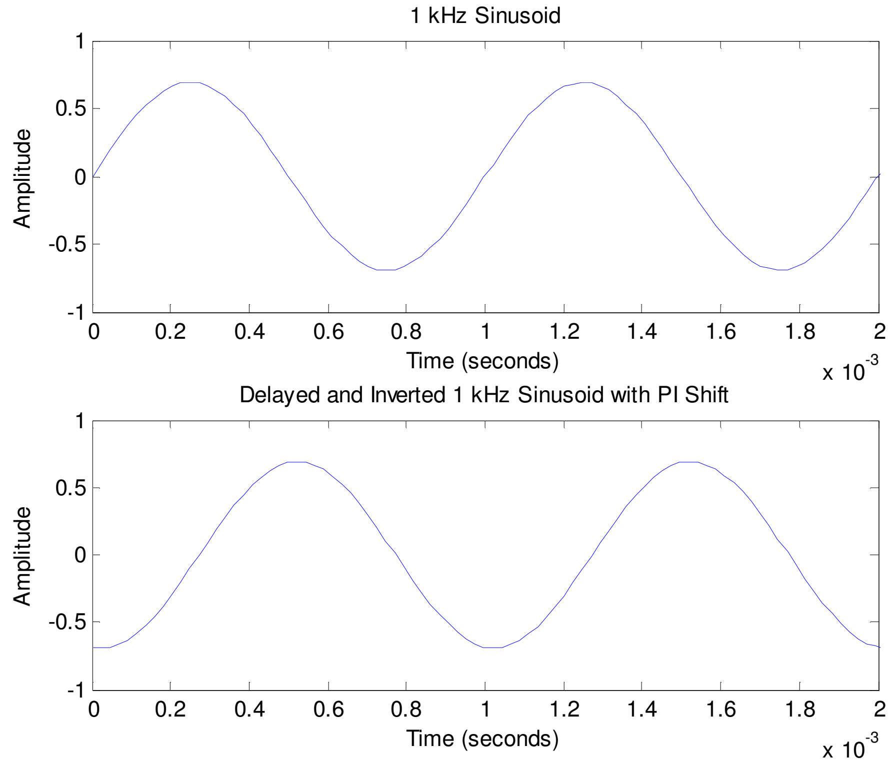

Figure 25: 1 kHz Sinusoid and a Delayed (1 second), Inverted Sinusoid

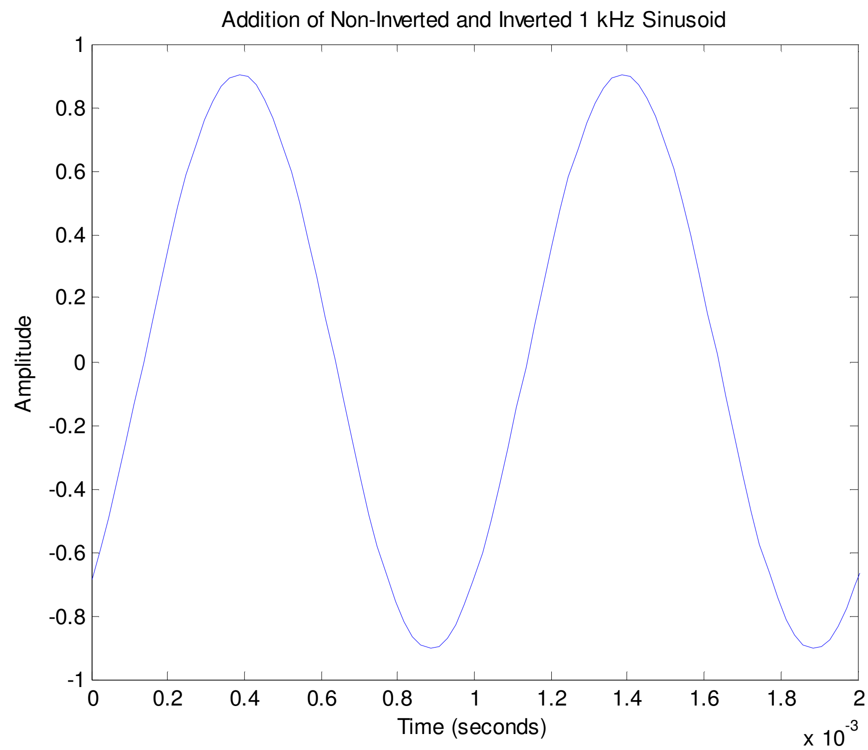

Figure 26: Summation of 1 kHz Sinusoid and a Delayed (1 second), Inverted Sinusoid

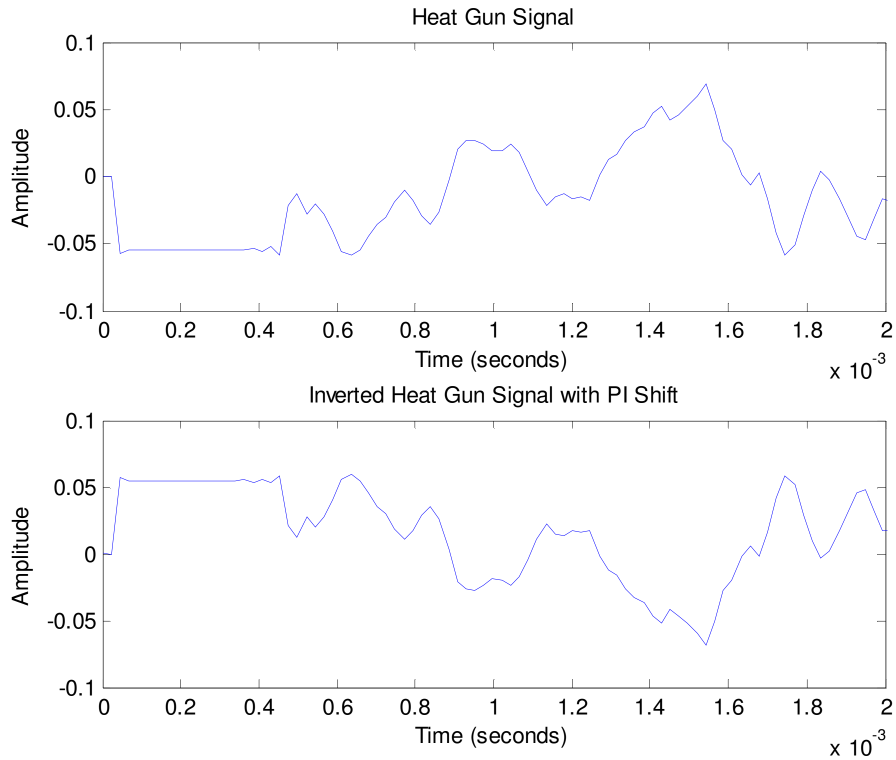

Figure 27: Heat Gun and its Inverse

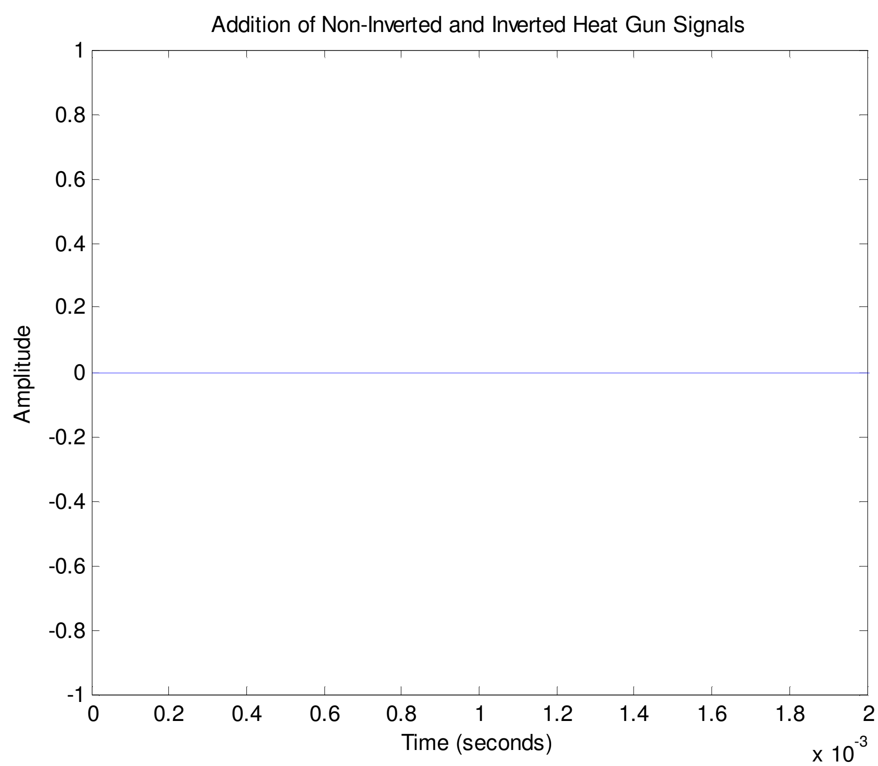

Figure 28: Summation of Heat Gun and its Inverse

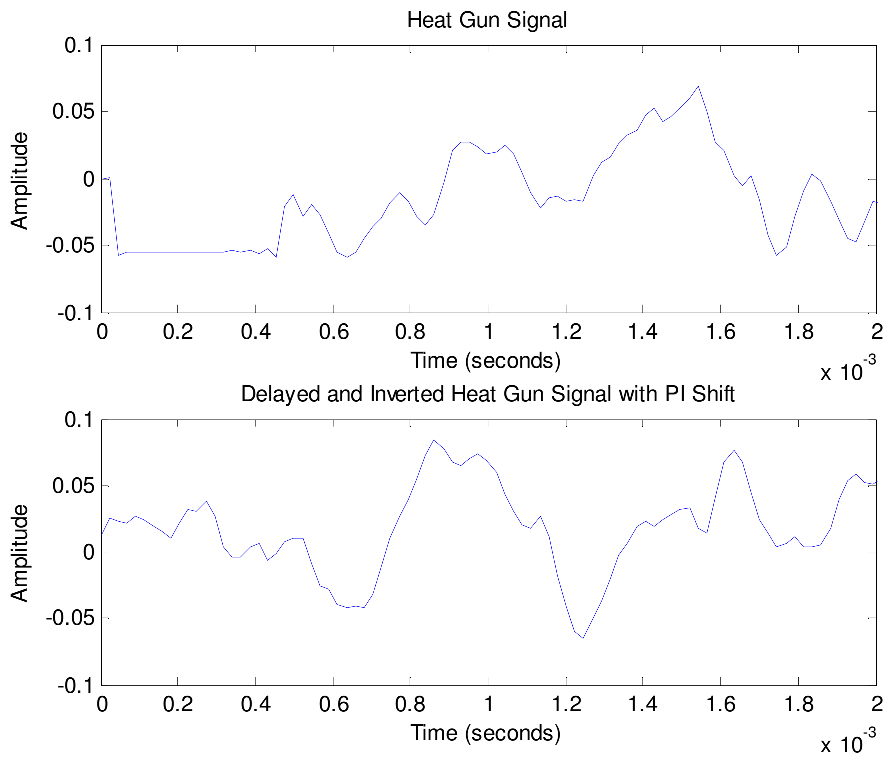

Figure 29: Heat Gun Signal and a Delayed (1 second), Inverted Heat Gun Signal

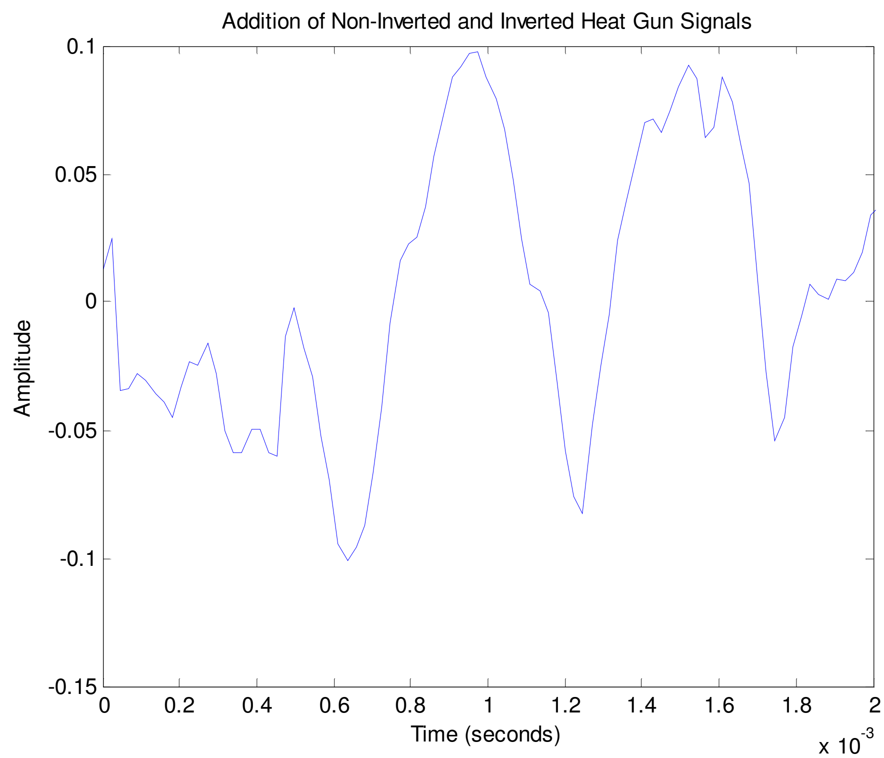

Figure 30: Summation of Heat Gun Signal and a Delayed (1 second), Inverted Heat Gun Signal

As can be seen from the plots above, adding an input signal with the same signal but π shifted in the frequency domain will output no signal. This method will work for both periodic (sinusoid) and non-periodic (heat gun) signals. However, figures 23 and 27 do not take into account the time delay that would be encountered in a physical circuit when using a microphone. Delay is not an issue for software, but in order to simulate actual hardware, delay should be considered. When adding an input signal with a time delayed version of the same signal, a π shift in the frequency domain will no longer zero the output signal as seen in figures 26 and 30. This method also implies that one will have the source signal and noise signal in separate channels otherwise both the source and noise will be cancelled. If there were a way to estimate the time delay, this method could work, but the estimated time delay would have to be perfect.

3.4. Test #4: Least Mean Squares

This section discusses the use of adaptive filters. If there were no time delay to take into account, the methods applied in the previous tests would be all that is needed. However, in the real world, one will not likely have the a priori information required to cancel out the noise that will be heard (i.e. one won't know the type of noise that will be heard).

Adaptive filters update their filter coefficients iteratively using algorithms like least mean squares or recursive least squares [1]. For noise cancellation in this project, an adaptive filter will take two inputs. The first input is the desired signal and it consists of the signal s(n) and noise X(n); the second input is a reference noise x(n) that relates in some way to the noise heard in the desired signal. The reference noise in this case will be the delayed version of the noise heard with the desired signal. The output e(n) is the error signal which consists of the desired signal with cancelled noise.

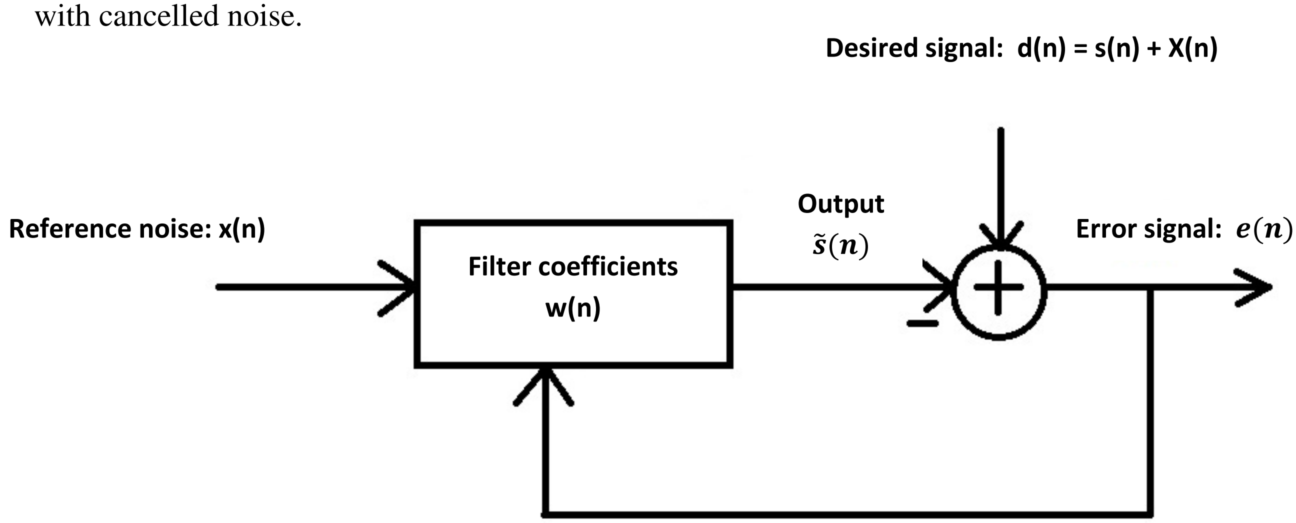

Figure 31: General Adaptive Filter Block Diagram

A common noise cancellation adaptive filter uses the least mean squares (LMS) method. The goal of the LMS method is to find the filter coefficients required to reduce the mean square error of the error signal [8]. The error signal is the difference between the desired d(n) and output y(n). The filter will only adapt to the error at the current time. The algorithm for the least mean squares method is shown below.

Least Mean Squares Algorithm:

$Parameters$:

$p \ = \ filter\ order$

$u \ = \ step\ size$

$n = length\ of desired\ signal = \#\ of\ iterations$

$Initialization$:

$\boldsymbol{w} = [w(0) = 0, w(1) = \ 0, \ldots, w(p) = 0]$

$Computation$:

$For\ n \ = \ 0, 1, 2, \ldots$

$\boldsymbol{x}(n) \ = \ [x(n), x(n-1), \ldots, x(n-p+1)]^T$

$\tilde{\boldsymbol{s}}(n) = \ \boldsymbol{w}(n) \cdot \mathbf{x}(\mathrm{n})$

$\boldsymbol{e}(n) \ = \ \boldsymbol{d}(n) - \tilde{\boldsymbol{s}}(n)$

$\boldsymbol{w}(n+1) \ = \ \boldsymbol{w}(n) \ + \ u \cdot \boldsymbol{e}^T(n) \ \cdot \ \boldsymbol{x}(n)$

The filter order used will affect the quality of the noise cancellation. A higher filter order will eliminate noise faster at the expense of cost. For this project, a filter order of two was used to highlight the reduction of error over time in the following plots as well as reduce computation time. The step size used was .012; the step size between filter coefficients will affect how well the noise is cancelled. A small step size is ideal because there will be less room for error between steps. The parameter, n, is dependent on the signal input to MATLAB and will increase or decrease depending on the length of the signal.

Mean Squared Error Plots:

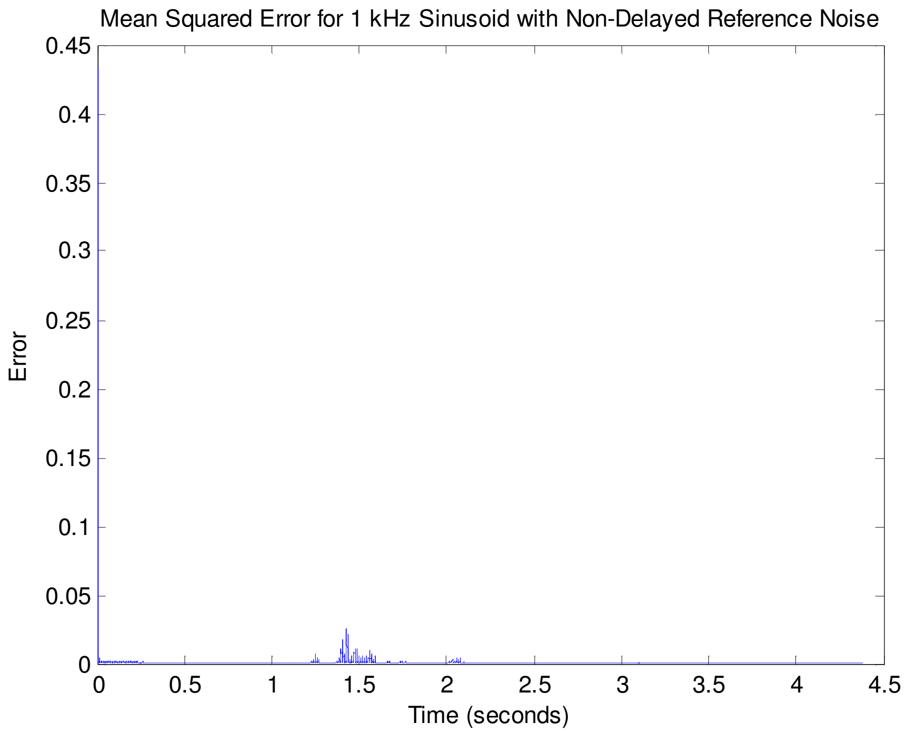

Figure 32: LMS Mean Square Error for 1 kHz Sinusoid (No Delay)

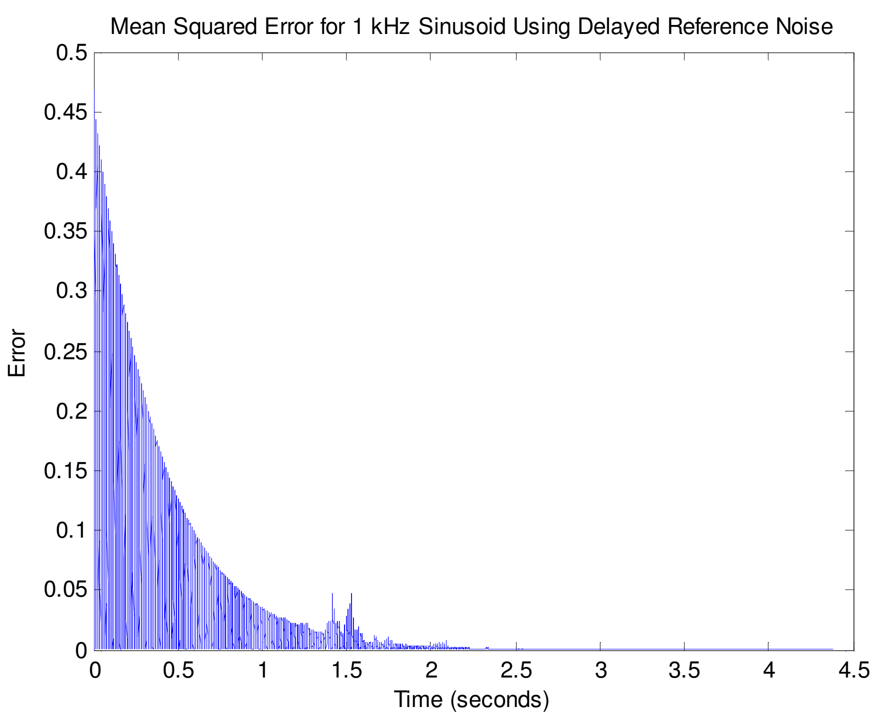

Figure 33: LMS Mean Square Error for 1 kHz Sinusoid (1 second Delay)

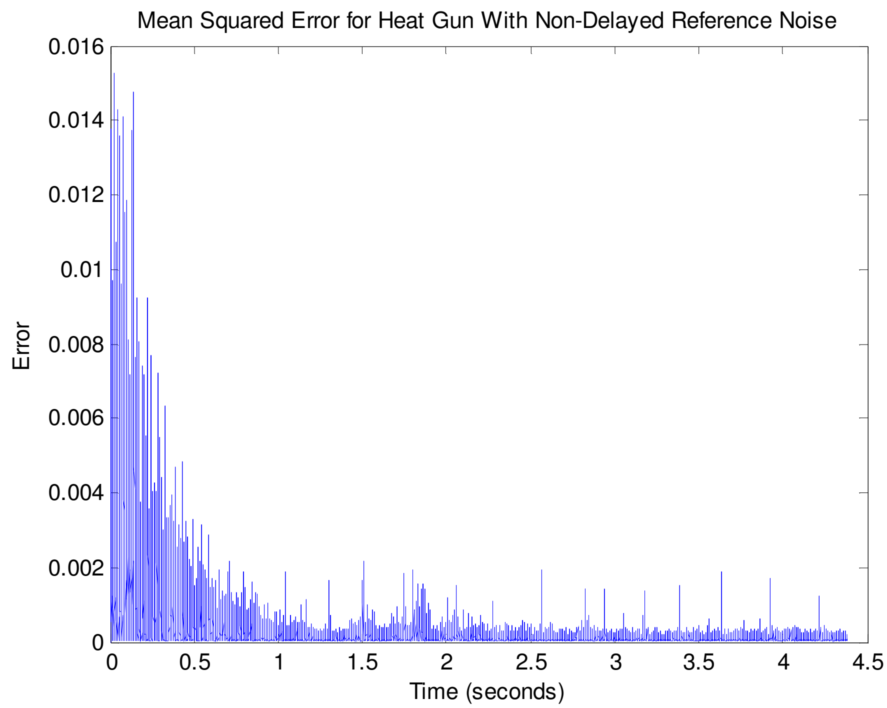

Figure 34: LMS Mean Square Error for Heat Gun Signal (No Delay)

Figure 35: LMS Mean Square Error for Heat Gun Signal (1 second Delay)

The plots above show the mean squared error between the output signal (speech with removed noise) and the source signal (speech with no noise). The mean squared error is a useful tool that estimates the difference between the true values of the quantity being measured and a new set of values relating to the original. The mean squared error is calculated with the following equation:

$$MSE = \frac{1}{N}\sum_{n=1}^{N} e^2(n)$$

From these tests, it is clearly seen that using a non-delayed reference input will result in a mean squared error that decreases over time. This can be said regardless of the periodicity of the noise signal. From figure 35, it can be observed that using the same periodic signal as both the noise and as the reference signal will result in an almost perfect output. Figure 36 shows that the adaptive filter will take more time to minimize the error when using a delayed reference noise. Similarly, the same non-periodic signal as both the noise and reference will result in the adaptive filter minimizing the error but not to the extent of a periodic noise. Lastly, when using a delayed non-periodic reference noise, the least mean squares algorithm fails to reduce the noise to a noticeable amount. An increased filter order was shown to reduce the noise faster unlike an increase in step size which lowered the sound quality of the error signal.

3.5. Test #5: Recursive Least Squares

Another popular noise cancellation adaptive filter involves using the recursive least squares method (RLS). Unlike the LMS method, the RLS method finds the filter coefficients that minimize a weighted linear least squares function relating to the input signals instead of trying to reduce the mean square error. It is deterministic (i.e. no randomness is involved of future states of the system).The RLS method is faster than the LMS method at the expense of more complex computations [11].

Recursive Least Squares Algorithm

$Parameters$:

$p\ =\ filter\ order$

$\lambda = forgetting\ factor$

$\delta = value\ to\ initialize\ P(0)$

$n = length\ of desired\ signal = \#\ of\ iterations$

$Initialization$:

$\boldsymbol{w} = [w(0) = 0, w(1) = 0, \ldots, w(p) = 0]$

$Computation$:

$For\ n = 0, 1, 2, \ldots$

$\boldsymbol{x}(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$

$\tilde{\boldsymbol{s}}(n) = \boldsymbol{x}(n) \cdot \boldsymbol{w}(\mathrm{n}-1)$

$\boldsymbol{e}(n) = \boldsymbol{d}(n) - \tilde{\boldsymbol{s}}(n)$

$\boldsymbol{g}(n) = \boldsymbol{P}(n-1) \cdot \boldsymbol{x}(n)\{\lambda + \boldsymbol{x}^T(n) \cdot \boldsymbol{P}(n-1) \cdot \boldsymbol{x}(n)\}^{-1}$

$\boldsymbol{P}(n) = \lambda^{-1} \cdot \boldsymbol{P}(n-1) - \boldsymbol{g}(n) \cdot \boldsymbol{x}^T(n) \cdot \lambda^{-1} \cdot \boldsymbol{P}(n-1)$

$\boldsymbol{w}(n) = \boldsymbol{w}(n-1) + \boldsymbol{e}(n) \cdot \boldsymbol{g}(n)$

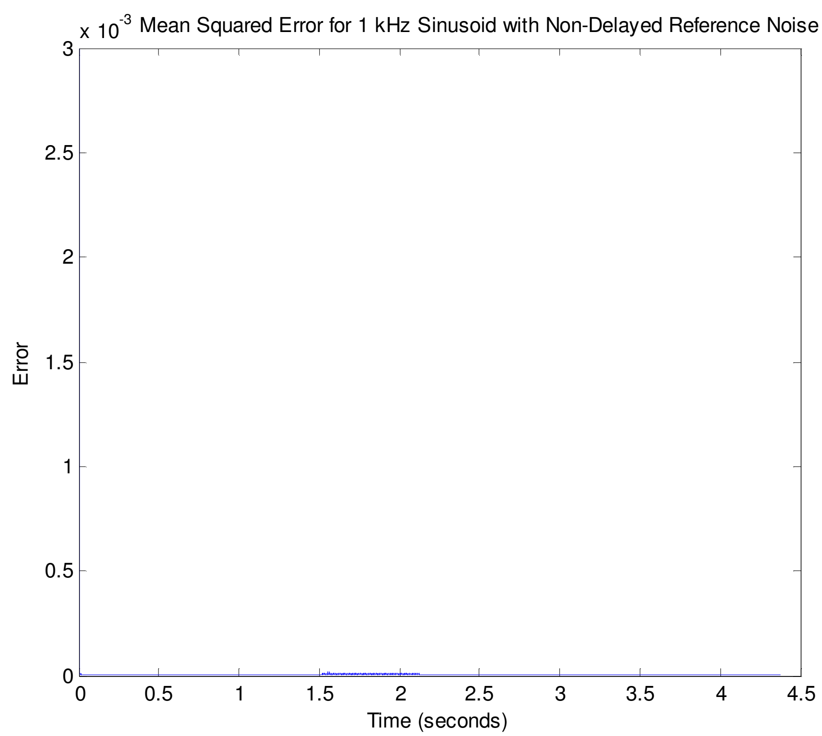

Figure 36: RLS Mean Square Error for 1 kHz Sinusoid (No Delay)

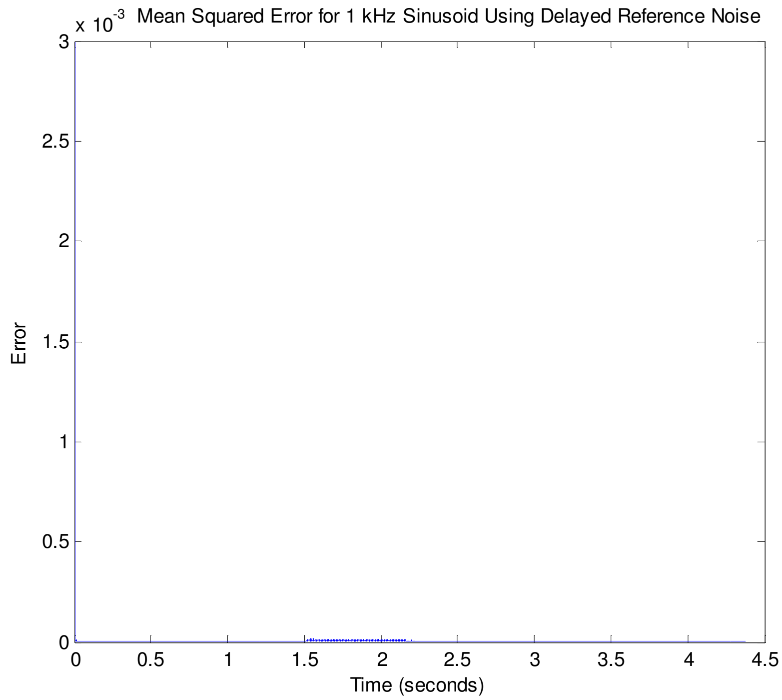

Figure 37: RLS Mean Square Error for 1 kHz Sinusoid (1 second Delay)

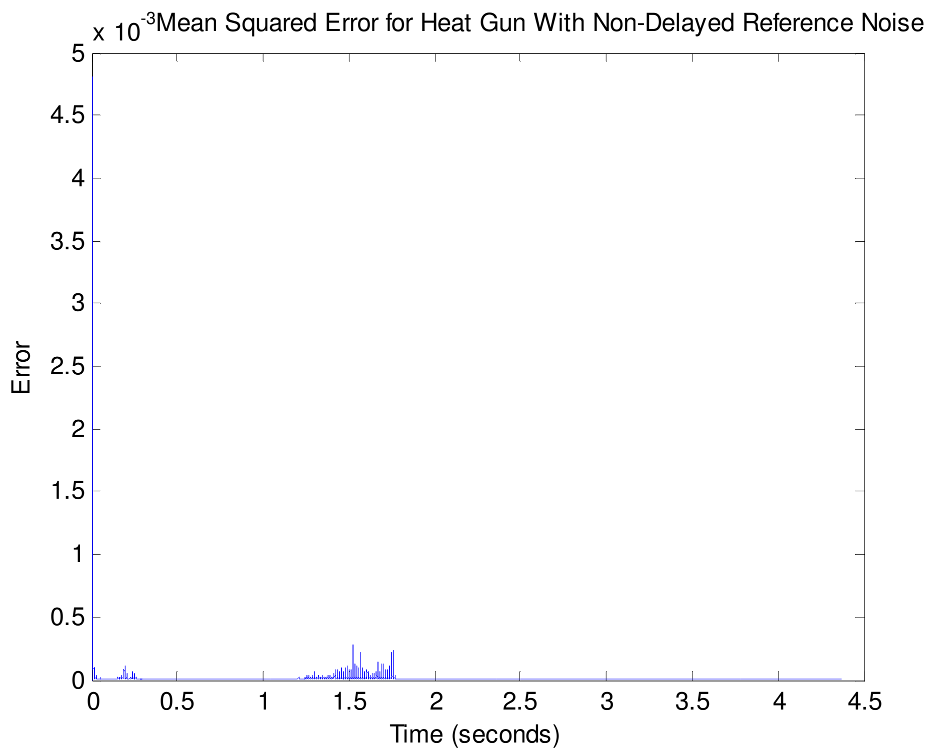

Figure 38: RLS Mean Square Error for Heat Gun Signal (No Delay)

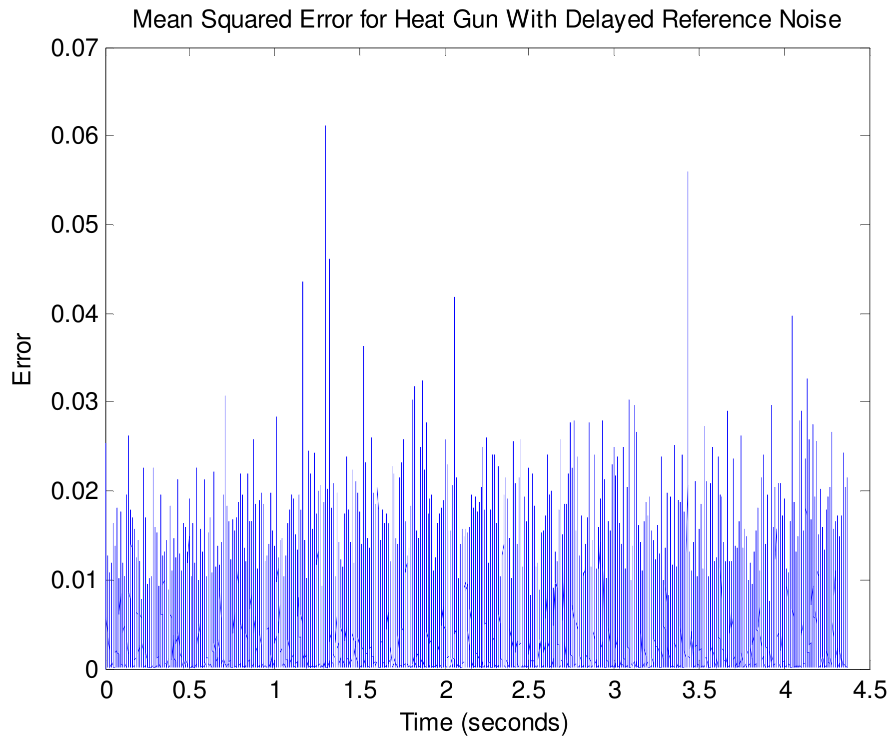

Figure 39: RLS Mean Square Error for Heat Gun Signal (1 second Delay)

The RLS method proves to be better than the LMS method when cancelling periodic noise. When using a 1kHz sinusoid noise signal, the .wav file generated contains almost no 1 kHz tone (figure 39 and 40) unlike the .wav file generated by the LMS which requires some time to cancel out the 1 kHz tone (figure 35 and 36). However, for non-periodic noise, the RLS does not work as well. Given a reference that contains no delay from the original noise source, the RLS cancels noise very well during pauses in the speech signal. However, when the reference noise is a delayed version of the noise, the RLS fails to cancel any noise.

3.6. Future Work

Adaptive filters are currently the best way to cancel noise because they adapt to the noise in an environment. An improvement to this project would be to optimize the RLS code such that when given a reference noise that is a delayed version of the input noise signal, the filter will successfully cancel the noise or at least minimize it. A possible extension for this project would be to devise a new algorithm that will cancel a delayed non-periodic reference noise signal. Another possibility involves estimating the time delay so that a pi phase shift could work. MATLAB could also be used to cancel the noise actively in real-time instead of taking predetermined sources.

4. CONCLUSION

Comparison between hardware and software approaches for noise cancelling shows that software approach is the best approach. Many of the downfalls of hardware are not present in the software. Timing delays are almost non-existent in the software, computing of the algorithms in the MATLAB are extremely fast. The noise cancelling for the periodic signals using the RLS algorithm cancels the noise from the signal almost completely, creating a clear signal.

The hardware and software approaches for this project are different in nature. Hardware is an active process, meaning the noise is being cancelled while it is being produced. The hardware uses a microphone to actively cancel noise, while the software uses preassembled noise and signal files with MATLAB to remove noise; a possible improvement for this project involves using MATLAB to actively cancel noise in real-time. The difference between both approaches makes it difficult to compare the two. It is seen that software works best for periodic signals, but doesn't perform well with non-periodic signals. This is due to the accuracy of software.

Both hardware and software were successful in reducing noise. Hardware produces quieter noise, but does not completely cancel the noise. Software will completely remove noise that is periodic using the RLS algorithm. Hardware performs better for non-periodic signals unlike the software approach that creates a noisier signal.

5. REFERENCES

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8. “Least Mean Squares (LMS) Adaptive Filter.” National Instruments. n.p. Web. 17 April 2012

9. “Least Mean Squares Filter.” Wikipedia. n.p. Web.17 April 2012

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11. “Recursive Least Squares Filter.” Wikipedia. n.p. Web. April 25 2012

APPENDIX A

MATLAB CODE

The following code was created to find the FFT of a signal after the user imports a .wav file of a signal. MATLAB will plot the magnitude and phase in the frequency domain as well as the waveform in the time domain. The fast Fourier transform is an efficient way of calculating the discrete Fourier transform. In order for the FFT to work, the length of the data is calculated to be the next power of two for the input signal. By doing so, the FFT will increase or decrease the number of data points to match the length of the data. If more points are required, the data will be padded with zeros; if fewer points are required, the data will be truncated.

```
% Get FFT of signal
% Import .wav file of signal and run code

% Time to use for plotting in time domain
% 0-5 seconds with 1/fs increments
t = [0:1/fs:5];
% Get the length of the data
L = length(data);
% Increase length to the next power of 2 for efficient FFT
NFFT = 2^nextpow2(L);
% Normalize by the length
Y = fft(data,NFFT)/L;
% Frequency to use plotting x-axis
f = fs/2*linspace(0,1,NFFT/2+1);
% Calculate the phase angles in radians
A1 = angle(Y(1:NFFT/2+1));

% Magnitude Frequency Domain
figure(1)
plot(f,2*abs(Y(1:NFFT/2+1)));
xlabel('Frequency (Hz)')
ylabel('Amplitude')
title('Magnitude Response')

% Phase Frequency Domain
figure(2)
plot(f, A1)
xlabel('Frequency (Hz)')
ylabel('Phase (radians)')
title('Phase Response')

% Time Domain Waveform
figure(3)
plot(t(1:length(data)),data)
xlabel('Time (seconds)')
ylabel('Amplitude')
title('Time Domain Waveform')
```

The following code is used to design a filter. Given the filter order and the cutoff frequency, the code will find the filter coefficients corresponding to a Butterworth filter. After the signal is filtered, the output will be played in MATLAB.

```
% Low-Pass Butterworth Filter

% Initialization:
% order = order of the filter
% cutoff_freq = cutoff frequency of filter
% fs = sampling frequency
% signal = source input
% noise = noise to add to source input
primary = signal + noise;

% Get filter coefficients
% Bk = numerator coefficients
% Ak = denominator coefficients
[Bk, Ak] = butter(order, cutoff_freq);

% Filter the primary signal using the filter coefficients found
output = filter(Bk, Ak, primary);

% Play the output
wavplay(output, fs)

end
```

The following code should be used in conjunction with the function on the previous page. After importing two .wav files, a source and noise signal, and setting the filter coefficients found earlier, this code will apply the filter to the incoming signal. An "output.wav" will be generated which can be played to hear the final output.

```
% Using Filter Coefficients to Filter an Input Signal

% Set signal = source signal (no noise)
% Set noise = noise signal
% Set fs = frequency sampling rate
% Note: signal and noise should be vectors of the same length
% Set Bk = filter coefficients given by filter_response
% Set Ak = filter coefficients given by filter_response
% Note: MATLAB does not set filter coefficients automatically

% Adding source and noise signals
input = signal + noise;

% Use filter function with filter coefficients
output = filter(Bk, Ak, input);

% Create a .wav file to listen to the output
wavwrite(output, fs, 'output.wav')
```

The following code takes two signals as its input (i.e. data and data1). The FFT function of MATLAB is used to introduce a π phase shift in the frequency domain for the second signal. Afterwards, the iFFT function of MATLAB will bring the second signal back to the time domain where it will be added to the first signal. If the noise cancels out, the plot of figure 2 will be at approximately at zero. It is suggested to use the same signal for data and data1 to ensure the correct steps are being taken.

```
% Adding a PI Shift in Frequency Domain

% Import .wav sound files
% Set data = first noise source
% Set data1 = second noise source
% Make sure the lengths of data and data1 are equal

% Initialization
t = [0:1/fs:1]; % Time Range: 0-1 second
L = length(data);
NFFT = 2^nextpow2(L);
% Divide by L to denormalize
Y = fft(data,NFFT)/L;
Y1 = fft(data1,NFFT)/L;

% Separate magnitude and phase of first signal
c = abs(Y);
d = angle(Y);
e = c.*exp(j*d);
% Separate magnitude and phase of second signal
c1 = abs(Y1);
d1 = angle(Y1);

% No delay; using original c and d
% f = c.*exp(j*(d+pi));

% Using a delay and shifting by pi; new c1 and d1
f = c1.*exp(j*(d1+pi));

% iFFT to bring back to time domain
% Have to multiply by L to denormalize
b = L*real(ifft(e));
b1 = L*real(ifft(f));

% Final Output in Time Domain
b2 = b + b1;

% Plotting signals
freq = fs/2*linspace(0,1,NFFT/2+1);
freq1 = fs/2 * linspace(0,1,NFFT);

figure(1)
subplot(211)
plot(t, b(1:length(t)))
xlabel('Time (seconds)')
```

```
ylabel('Amplitude')
title('First Signal in Time Domain')

subplot(212)
plot(t, b1(1:length(t)))
xlabel('Time (seconds)')
ylabel('Amplitude')
title('Second Signal in Time Domain')

figure(2)
plot(t, b2(1:length(t)))
xlabel('Time (seconds)')
ylabel('Amplitude')
title('Addition of First and Second Signals in Time Domain')
```

The following code applies the least mean squares algorithm. The inputs are the source signal (signal), the noise signal (noise), and the reference noise (ref). The code will then generate a plot of the mean square error which can be used to see the estimated difference between the original source signal and the final noise cancelled signal.

```
% Least Mean Squares
% Importing Data
% Vectors must have same length and sampling frequency
% signal = source input
% noise = noise to add to source input
% ref = signal that correlates to noise signal

% Initialization
% The least mean squares will perform better with higher orders which would
% increase cost. It is ideal to find the minimum order with the least amount
% of noise.
order = 2;
t = [0:1/fs:5]; % Time Range: 0-5 seconds
N = length(signal);
primary = signal + noise; % Add the signal and noise together
% Initializations for LMS algorithm
R = zeros(order,1);
desired = zeros(order,1);
w = zeros(order,1);
mu = .012; % Step-size

% LMS Algorithm
fori=1:N-1
    R=[ref(i);R(1:length(R)-1)];
y(i)=w'*R;
desired(i) = primary(i)-y(i);
e(i) = desired(i);
w= w+mu*R*conj(e(i));
end

% Filtered Output Signal
% plot(t(1:N-order+1), e)
% xlabel('Time (seconds)')
% ylabel('Amplitude')
```

```
% Calculate Mean Squared Error
MSE = e' - signal(1:length(e));
MSE = MSE.^2;

figure(1)
plot(t(1:length(MSE)), MSE)
xlabel('Time (seconds)')
ylabel('Error')
title('Mean Squared Error')

% wavplay(signal, fs)
% wavplay(primary, fs)
% wavplay(ref, fs)
% wavplay(desired, fs)
```

The following code applies the recursive least squares algorithm. The inputs are the source signal (signal), the noise signal (noise), and the reference noise (ref). The code will then generate a plot of the mean square error which can be used to see the estimated difference between the original source signal and the final noise cancelled signal. A .wav file will be generated for the input and output signal so the user can hear the original signal with noise and the output which is noise cancelled.

```
% Recursive Least Squares
% Importing Data
% Vectors must have same length and sampling frequency
% signal = source input
% noise = noise to add to source input
% ref = reference signal (e.g. delayed noise)
% lambda = forgetting factor,
% M = filter order
% delta = initial value for P; use a value in the 10^-3 range

% Output arguments:
% xi = output
% w = final filter coefficients

% Initialization
lambda = 1;
M = 5;
delta = .005;
w=zeros(M,1);
primary = signal + noise;

% eye(M) gives an MxM identity matrix
% dividing by delta replaces the 1's in the identity matrix with delta^-1
P=eye(M)/delta;

% Input Signal Length
N=length(ref);
```

```
% Set output as desired signal
xi=primary;

% Loop, RLS
for n=M:N
uvec=ref(n:-1:n-M+1);
k=lambda^(-1)*P*uvec/(1+lambda^(-1)*uvec'*P*uvec);
% xi is output
xi(n)=primary(n)-w'*uvec;
% Recursive equation to minimize cost function (difference between the
% desired and input signals)
w=w+k*conj(xi(n));
P=lambda^(-1)*P-lambda^(-1)*k*uvec'*P;

end


% Calculate Mean Squared Error
MSE = xi - signal(1:length(xi));
MSE = MSE.^2;

% Time to use for plotting in time domain
% 0-5 seconds with 1/fs increments
t = [0:1/fs:5];

figure(1)
plot(t(1:length(MSE)), MSE)
xlabel('Time (seconds)')
ylabel('Error')
title('Mean Squared Error')

% Output the input .wav file
wavwrite(primary, fs, 'input.wav')
% Output the new .wav file
wavwrite(xi, fs, 'output.wav');
```

APPENDIX B

COMPONENT SPECIFICATION

TEXAS INSTRUMENTS

NE5532, NE5532A
SA5532, SA5532A

www.ti.com

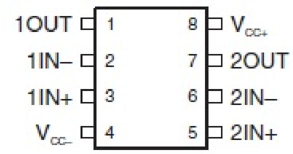
SLOS075I–NOVEMBER 1979–REVISED APRIL 2009

DUAL LOW-NOISE OPERATIONAL AMPLIFIERS

FEATURES

- **Equivalent Input Noise Voltage: 5 nV/$\sqrt{\text{Hz}}$ Typ at 1 kHz**
- **Unity-Gain Bandwidth: 10 MHz Typ**
- **Common-Mode Rejection Ratio: 100 dB Typ**
- **High DC Voltage Gain: 100 V/mV Typ**
- **Peak-to-Peak Output Voltage Swing 26 V Typ With $V_{CC\pm}$ = ±15 V and R_L = 600 Ω**
- **High Slew Rate: 9 V/µs Typ**

NE5532, NE5532A . . . D, P, OR PS PACKAGE
SA5532, SA5532A . . . D OR P PACKAGE
(TOP VIEW)



DESCRIPTION/ORDERING INFORMATION

The NE5532, NE5532A, SA5532, and SA5532A are high-performance operational amplifiers combining excellent dc and ac characteristics. They feature very low noise, high output-drive capability, high unity-gain and maximum-output-swing bandwidths, low distortion, high slew rate, input-protection diodes, and output short-circuit protection. These operational amplifiers are compensated internally for unity-gain operation. These devices have specified maximum limits for equivalent input noise voltage.

ORDERING INFORMATION(1)

T_A	PACKAGE(2)		ORDERABLE PART NUMBER	TOP-SIDE MARKING
0°C to 70°C	PDIP – P	Tube of 50	NE5532P	NE5532P
			NE5532AP	NE5532AP
	SOIC – D	Tube of 75	NE5532D	N5532
		Reel of 2500	NE5532DR	
		Tube of 75	NE5532AD	N5532A
		Reel of 2500	NE5532ADR	
	SOP – PS	Reel of 2000	NE5532PSR	N5532
			NE5532APSR	N5532A
–40°C to 85°C	PDIP – P	Tube of 50	SA5532P	SA5532P
			SA5532AP	SA5532AP
	SOIC – D	Tube of 75	SA5532D	SA5532
		Reel of 2500	SA5532DR	
		Tube of 75	SA5532AD	SA5532A
		Reel of 2500	SA5532ADR	

(1) For the most current package and ordering information, see the Package Option Addendum at the end of this document, or see the TI web site at www.ti.com.
(2) Package drawings, thermal data, and symbolization are available at www.ti.com/packaging.

RECOMMENDED OPERATING CONDITIONS

			MIN	MAX	UNIT
V_{CC+}	Supply voltage		5	15	V
V_{CC-}	Supply voltage		–5	–15	V
T_A	Operating free-air temperature	NE5532, NE5532A	0	70	°C
		SA5532, SA5532A	–40	85	

ELECTRICAL CHARACTERISTICS

$V_{CC\pm} = \pm15$ V, $T_A = 25°C$ (unless otherwise noted)

	PARAMETER	TEST CONDITIONS(1)		MIN	TYP	MAX	UNIT
V_{IO}	Input offset voltage	$V_O = 0$	$T_A = 25°C$		0.5	4	mV
			T_A = Full range(2)			5	
I_{IO}	Input offset current	$T_A = 25°C$			10	150	nA
		T_A = Full range(2)				200	
I_{IB}	Input bias current	$T_A = 25°C$			200	800	nA
		T_A = Full range(2)				1000	
V_{ICR}	Common-mode input-voltage range			±12	±13		V
V_{OPP}	Maximum peak-to-peak output-voltage swing	$R_L \geq 600\ \Omega$, $V_{CC\pm} = \pm15$ V		24	26		V
A_{VD}	Large-signal differential-voltage amplification	$R_L \geq 600\ \Omega$, $V_O = \pm10$ V	$T_A = 25°C$	15	50		V/mV
			T_A = Full range(2)	10			
		$R_L \geq 2\ k\Omega$, $V_O \pm 10$ V	$T_A = 25°C$	25	100		
			T_A = Full range(2)	15			
A_{vd}	Small-signal differential-voltage amplification	f = 10 kHz			2.2		V/mV
B_{OM}	Maximum output-swing bandwidth	$R_L = 600\ \Omega$, $V_O = \pm10$ V			140		kHz
B_1	Unity-gain bandwidth	$R_L = 600\ \Omega$, $C_L = 100$ pF			10		MHz
r_i	Input resistance			30	300		kΩ
z_o	Output impedance	$A_{VD} = 30$ dB, $R_L = 600\ \Omega$, f = 10 kHz			0.3		Ω
CMRR	Common-mode rejection ratio	$V_{IC} = V_{ICR}$ min		70	100		dB
k_{SVR}	Supply-voltage rejection ratio ($\Delta V_{CC\pm}/\Delta V_{IO}$)	$V_{CC\pm} = \pm9$ V to ±15 V, $V_O = 0$		80	100		dB
I_{OS}	Output short-circuit current			10	38	60	mA
I_{CC}	Total supply curent	$V_O = 0$, No load			8	16	mA
	Crosstalk attenuation (V_{O1}/V_{O2})	$V_{01} = 10$ V peak, f = 1 kHz			110		dB

(1) All characteristics are measured under open-loop conditions, with zero common-mode input voltage, unless otherwise specified.
(2) Full temperature ranges are: –40°C to 85°C for the SA5532 and SA5532A, and 0°C to 70°C for the NE5532 and NE5532A.

APPENDIX C

COMPONENT COSTS

PART	PRICE
DPDT Switch	$15.96
Audio Jacks x 3	$8.07
Various Resistors	$9.52
Various Capacitors	$6.28
100kΩ Audio Potentiometers	$7.38
Condenser Microphones x 3	$9.57
NE5532 x 6	$2.46
Total	$59.24

APPENDIX D

ANALYSIS OF SENIOR PROJECT DESIGN

Please provide the following information regarding your Senior Project and submit to your advisor along with your final report. Attach additional sheets, for your response to the questions below.

Project Title: Hardware and Software Study of Noise Cancellation

Students' Names: Riggi Aquino and Jacob Lincoln **Students' Signature:**

Advisor's Name: **Advisor's Initials:** **Date:**

• **Summary of Functional Requirements**

- The hardware design accepts inputs for noise and source signals. The noise is then amplified from a microphone. The amplified noise from the microphone then gets inverted. The inverted, amplified signal gets summed (with the source signal if available). This signal is outputted on headphones, which then cancels the noise by deconstructively interfering with the outside ambient noise that was also heard by the input microphone.
- The software design accepts separate files for its source and noise signals. After importing the .wav files into MATLAB, either the least mean squares or the recursive least squares algorithm can be applied. Once applied, MATLAB will create a .wav file of the output which should contain the source signal with cancelled noise.

• **Primary Constraints**

- The most difficult part of the hardware portion of the project was the timing delays involved with the circuit. Matching the time delays between when the user hears the outside ambient noise and when the microphone signal propagates through the signal to the output of the headphones was the most challenging. For this a DPDT switch was added to switch between inverting and non-inverting signal. This helped alleviate the timing delays issues involved by allowing the user to better match the signals for deconstructive interference.
- The software portion of this project was difficult when using an input that consists of the delayed version of the noise from the desired (source + noise) signal. It is desired to cancel this noise because in the real world, non-periodic noise will most likely be encountered and a delay of the noise must be taken into account within the hardware.

• **Economic**

- Originally, the overall cost of the project was believed to be under $100.
- The final overall component cost of the project was $59.24.
- A bill of materials is provided in appendix C.
- Materials needed for the project were: breadboard, oscilloscope, digital multi-meters, dual DC power supply, wire, components listed in appendix C, and computer with MATLAB.
- The expected development time was approximately six months. The first two months were allotted for research and design. The next 4 months were spent building, implementing, and testing the circuit and software. The deadline for the project was May 29th. No delays were involved.

• **If manufactured on a commercial basis:**

If this project was marketed on a commercial basis without aid from an outside source then:

- Estimated number of devices sold per year: 40 units
- Estimated manufacturing cost for each device: $59.24
- Estimated purchase price for each device: $75.99
- Estimated profit per year: $670.00
- Estimated cost for user to operate device: $20.00 for headphones and power source

• **Environmental**

- Use of silicon and copper to develop the circuit used will diminish the resources available.
- The pollution from factories due to power consumption will harm the environment (i.e. green-house gases), therefore directly impacting several species.

• **Manufacturability**

• This project wasn't built to the point of manufacturability. Had it been developed to be manufactured, a case would be developed in order to hide the inner circuitry.

• **Sustainability**

- If no power supply is available, batteries are required to power the circuit.
- The ICs used for the circuit contain silicon, which is a limited resource. Copper wire was used for making connections on the breadboard.
- Miniaturization of the circuit would lead to fewer resources required when implementing the circuit. Redesign of the circuit for lower voltages needed would lead to less power consumption by the circuit.
- Lower voltages leads to lower sound quality. Lower voltages could lead to lower voltage clipping resulting in distorted sound.

Ethical

- Loud noises are destruction to the human ear. The use of a circuit to cancel periodic noise would help reduce the destruction of the users hearing, such as using the circuit in an environment where there is a cyclic engine.

- **Health and Safety**
  - Prolonged usage of headphones may cause some discomfort and slight loss of hearing. The user must take responsibility to avoid listening to loud sources of audio for extended periods of time.
- **Social and Political**
  - There are no major political issues relating to this product. Social issues may include using the product to avoid hearing noises like a fire alarm or ambulance when driving.
- **Development**
  - Every lab electrical engineering students were given specific steps from the lab manual. For this project there was no manual given. This idea of having to come up with steps and testing without the help of a lab manual was the greatest tool learned from this experiment. The use of the multi-meter to test continuity helped greatly in testing the ground and power connections.
  - New techniques were learned for the software portion of this project. For example, importing and working with .wav files in MATLAB as well as using the functions internal to MATLAB like the FFT function. It was also learned how to implement different algorithms to cancel out noise.