Solving the Snoring Problem: Attenuation through Active Noise Control

Brent O. Reichman

A senior thesis submitted to the faculty of
Brigham Young University
in partial fulfillment of the requirements for the degree of

Bachelor of Science

Scott Sommerfeldt, Advisor

Department of Physics and Astronomy

Brigham Young University

April 2013

ABSTRACT

Solving the Snoring Problem: Attenuation through Active Noise Control

Brent O. Reichman
Department of Physics and Astronomy
Bachelor of Science

Active noise control (ANC) uses a control signal to effectively cancel out unwanted sound. Applying ANC to snoring presents an interesting challenge because of its unpredictable nature and the close distance between the source and the desired region of cancellation. This experiment focuses on two factors: How much attenuation can be achieved in a standard bed using different microphone and speaker setups and how large is the "zone of silence" that is created.

Keywords: Active Noise Control, Snoring, Feedforward Algorithm, Attenuation

ACKNOWLEDGMENTS

I would like to thank Dr. Scott Sommerfeldt for all of the help that he has given me with this thesis. He has been an incredible help when I have encountered problems in research and in writing. I would like to thank William Johnson for helping me first set up the ANC system. I would also like to thank Dr. Jean-Francois Van Huele and Dr. J. Ward Moody for their invaluable feedback on the writing of this paper. Last but not least, I would like to thank my wonderful wife Angela for her patience as I have put many hours and a few late nights into writing this paper.

Contents

List of Figures

List of Tables

Chapter 1

Introduction to Snoring and Active Noise Control

1.1 Snoring Noise

Snoring is for many people not only a nuisance but a serious problem. Not only does snoring disrupt the sleep of those nearby, it is an important sympton connected with obstructive sleep apnea and excessive daytime sleepiness. [1] Though the reports on snoring differ greatly, it has been reported that over 50% of people snore, and in extreme cases the levels of snoring reach up to 90-100 dB. [2, 3] Snoring is produced as certain areas- in particular the soft palate, the tongue, and the nasal cavity- vibrate during sleep. [1] Because much of the sound in snoring is carried by lower-frequency components of the snore, earplugs are often not effective in attenuating, or lessening, the sound.

1.2 Types of Noise Control

Noise control is the effort to reduce unwanted sound from an environment. Different methods of control may be divided into two categories- active and passive noise control. Passive noise control involves placing barriers to impede the transmission of sound. These barriers could be walls on the side of a highway or earplugs. Passive noise control is effective at blocking higher frequency sounds but in general less effective against low-frequency tones. Active noise control (ANC) uses a control signal as an "anti-noise" to effectively cancel out unwanted sound. The sound is picked up by a microphone and a wave of opposite phase is emitted, and the two waves cancel each other out. In this paper we dicuss the extent to which ANC is able to attenuate the sound of a person snoring. The first chapter is devoted to an explanation of the underlying principles of ANC- The mathematical underpinnings, physical limitations, and a brief history. The second chapter focuses on the setup used in the experiment, and the third chapter presents the results and the achieved attenuation.

1.3 History of Active Noise Control

Though noise cancelling headphones would appear to many to be a relatively new phenomenon, the principles and ideas surrounding the technology have been around for quite some time. In 1934 Paul Lueg of Germany registered a patent addressing ANC. Though Lueg was not an acoustician by profession, he correctly outlined the principles of ANC and possible uses. [4] The first experiments were carried out in 1953 and 1956 by Harry F. Olson. [5]

The ideas behind Lueg's patent lay for the most part dormant until the advent of the digital signal processor (DSP). The DSP is a microprocessor that enables operations to be performed on data at a very high rate. [6] In order to input a reference signal from a microphone and output a cancelling signal, a very high sampling rate and thus very quick calculations are needed. Though some experiments in ANC were performed before the output of the DSP, true progress and application was only possible after the advent of the DSP.

1.4 Addition of Sound Pressure Waves

Active noise control depends on the coherent addition of pressure, or sound, waves. The term coherent indicates that there is a constant phase relationship between the two waves. When the phase relationship is varying in time, i.e. the waves are incoherent, the total pressure cannot be calculated easily, and instead to determine the sound level one must add the intensities of the waves. However, if the waves are coherent then the phasors representing each wave can be added in order to obtain the total pressure. [7] The square of this pressure can then be used to find the intensity. In the case of coherent sinusoidal waves, the total pressure squared simplifies to

$$\overline{(p_1+p_2)^2} = \overline{p_1^2} + \overline{p_2^2} + 2\overline{p_1 p_2}\cos(\theta-\phi). \tag{1.1}$$

In this equation p_1 represents the amplitude of the first wave, the source, and p_2 represents the amplitude of the control signal. The symbols θ and ϕ represent the phase of the first and second waves respectively. The overlines represent an averaging over time. The goal of ANC is to produce a control wave to effectively "cancel out" the wave emitting from the primary source in a region. One can achieve this by making Eq. (1.1) equal to zero. This can be accomplished if $\overline{p_1^2} = \overline{p_2^2}$ and $\theta - \phi = \pi$.

If $p_1 = p_2$ and $\theta - \phi = \pi$, meaning the waves are completely out of phase with each other, then Eq. (1.1) reduces to zero. The ability to match the phase and amplitude of the control wave with the source wave determines the amount of attenuation that is possible to achieve. Attenuation is typically measured in dB. In acoustics dB are often measured in reference to 20 μPa or 1 pW. However, due to the properties of the logarithmic function used in calculating the sound pressure level (SPL) an attenuation, or change in dB, is the same whether the reference is one pW or 20

μPa or 1 V. For this experiment the data are all taken in V and all levels are in reference to 1 V. For acoustic purposes, a drop in 3 dB is just barely noticeable to a person's ears. In many areas of active noise control, the zone of silence is defined as the space in which a drop of 10 dB or more is observed.

1.5 Feedforward Algorithm

The feedforward algorithm drives our DSP and determines the necessary output to achieve maximum attenuation. To produce a wave that is exactly out of phase with the source sound we need to know how the wave will change between the source and the error microphone, the desired space of cancellation. The feedforward algorithm can determine how the sound changes and what the DSP must output, relative to the reference signal, to produce a cancelling wave.

A transfer function tells us how the wave properties change as it travels between two places. [8] A transfer function describes how the sound that is emitted by a car engine sounds when it arrives at the driver's ears in the driver's seat, for example. Figure 1.1 shows one of the more complex forms of the feedforward algorithm used in ANC. In this figure P and H are transfer functions that describe how the sound changes between the source and the error microphone and the control speaker and the microphone respectively. These transfer functions are represented as vectors. The symbols $\hat{\text{P}}$ and $\hat{\text{H}}$ represent the estimates of P and H used by the DSP. For the control system to accurately output the proper cancelling signal, $\hat{\text{P}}$ and $\hat{\text{H}}$ must be accurate estimates of P and H.

Figure 1.1 is so complex because sometimes it is desirable to estimate P and H in real time. The schematic shown is able to use the measured inputs and outputs in order to obtain reasonable estimates for $\hat{\text{P}}$ and $\hat{\text{H}}$. In figure 1.1, an $\oplus$ means the two signals are added and $\otimes$ corresponds to a multiplication operation associated with updating the adaptive filters. The arrow going through the boxes with W, $\hat{\text{P}}$, and $\hat{\text{H}}$ inside them indicate an update of the transfer function. The function

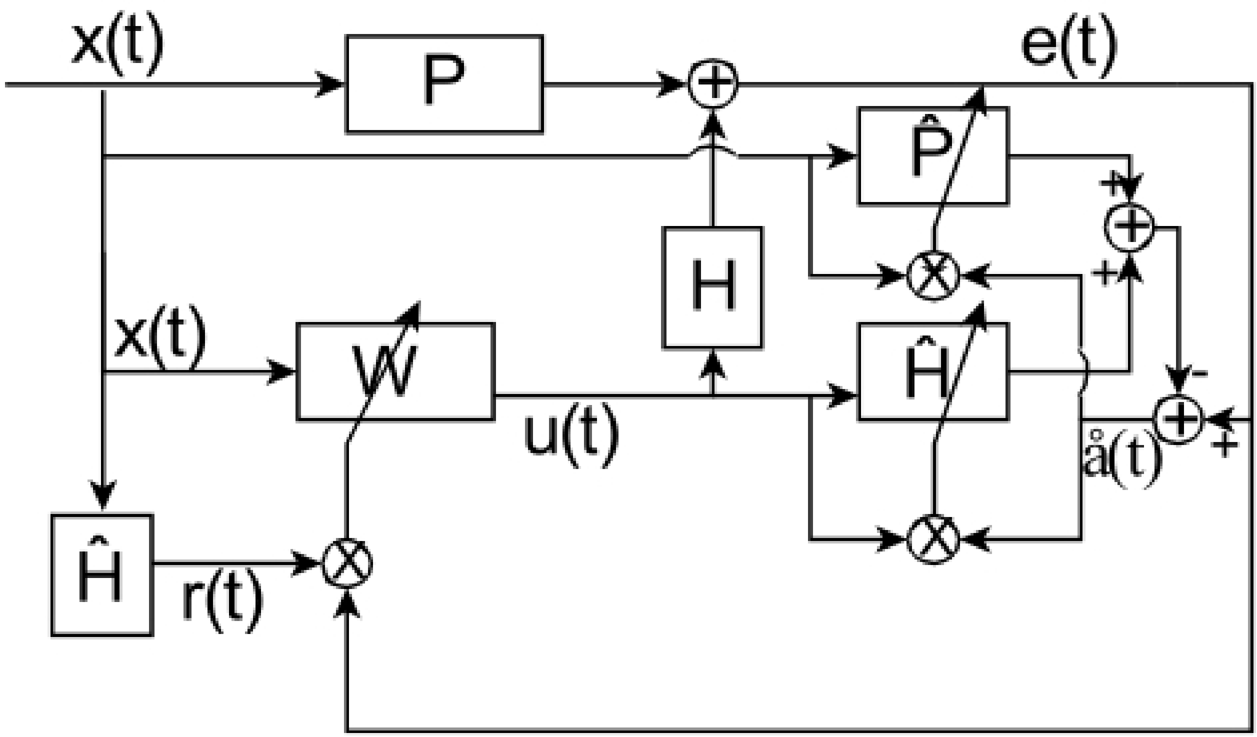

Figure 1.1 The feedforward algorithm including passive online system identification.

used to update W is [9]

$$W(t+1) = W(t) - \mu e(t) r(t). \tag{1.2}$$

To update the other equations we use $\hat{P}(t+1) = \hat{P}(t) + \alpha \dot{a}(t) x(t)$, where $\dot{a}$ is the actual measured error minus the predicted error obtained using $\hat{\mathrm{P}}$ and $\hat{\mathrm{H}}$. The update equation is the same for $\hat{\mathrm{H}}$. In the equations μ and α are convergence parameters. These convergence parameters are a weight given to the error and determine how much W, $\hat{\mathrm{H}}$, and $\hat{\mathrm{P}}$ change for some error. These parameters must be carefully chosen. If they are too small, the system won't converge to a solution quickly, and if they are too big the system will be unstable and won't converge at all.

Updating the transfer functions $\hat{\mathrm{P}}$ and $\hat{\mathrm{H}}$ in this manner is called "online system identification." It is useful because it can run and determine on its own how it must react in order to obtain the most attenuation at the error microphone. However, in many cases it can be better to use offline system identification. In offline system identification, one experimentally determines what the transfer function H is before running the system. If H is already known, there is no need to update $\hat{\mathrm{P}}$ and $\hat{\mathrm{H}}$. In fact, the transfer function $\hat{\mathrm{P}}$ is not needed at all because it is only used to update $\hat{\mathrm{P}}$ and $\hat{\mathrm{H}}$. All that is needed for proper control is $\hat{\mathrm{H}}$, which is just the experimentally determined transfer function, and the error signal e(t) to determine how one must update W in order to achieve the maximum control.

1.6 Physical Limitations of ANC

As was stated in the analysis of Eq. (1.1), in order to achieve the greatest attenuation, the waves being added together must have a phase difference of π. This is easier to achieve when the source is emitting a simple sine wave. If the control signal is delayed an integer number of periods, the wave is still exactly out of phase and is still cancelled. A simple Fourier transform of a typical snore, however, reveals that there are no main frequency components in the pressure wave, and can

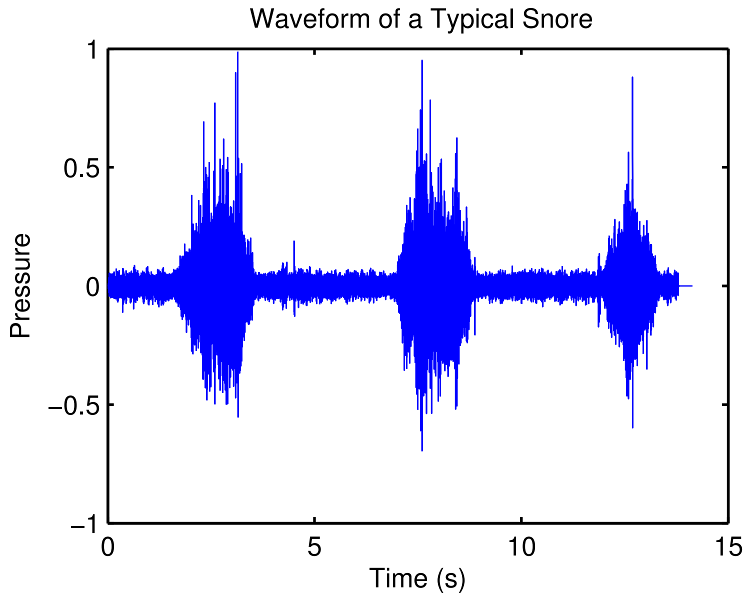

Figure 1.2 The waveform of a snore showing pressure as a function of time.

be seen in Fig. 1.3. In this situation, there is exactly one optimal solution for the algorithm.

The non-sinusoidal and unpredictable quality of a snore limits our possible speaker placement. If our control speaker is placed farther away from the error mic than the source, the wave emitted by the control will not arrive at the error mic until well after the wave from the source is passed. Thus causality and the unpredictable nature of snoring require our control speaker to be placed at least as close as the source to the error microphone. In addition to this, the algorithm takes time (about two milliseconds) to process the reference signal and output the control. Due to the delays in calculation, the control must be at least two feet closer to the error microphone than the source.

Aliasing is another problem that can complicate an ANC system. Because the DSP is sampling at a finite rate, there is a finite amount of frequencies it can detect. If the DSP is sampling at a rate f_s, the maximum frequency that can be detected is $\frac{f_s}{2}$, which is called the folding or Nyquist frequency. [10] In order to ensure that aliasing is not a problem, the reference signals to the DSP and its output should be filtered, and a safe value to avoid aliasing problems is $\frac{f_s}{2.5}$.

Another physical limitation comes from the difference in dropoff of the two sound waves.

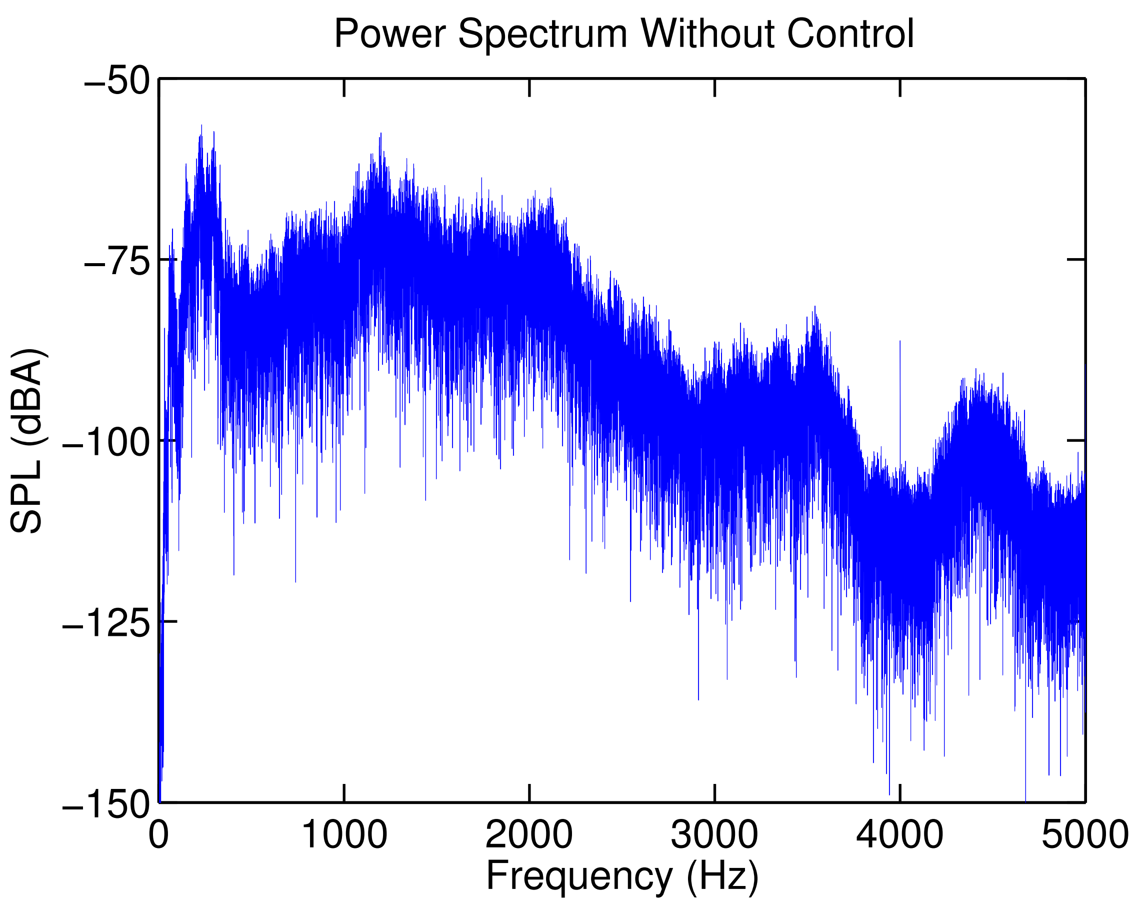

Figure 1.3 The power spectrum of a snore with no control added.

Ideally we want attenuation not only at the error microphone, but in a sizeable volume around the mic. However, as a pressure wave propogates, it drops off as $\frac{1}{r}$. If our control is much closer to the error microphone than the source, the dropoff rates will differ dramatically, and though we will get attenuation at the microphone, the volume where we achieve attenuation will be significantly smaller than if the two distances are relatively similar.

When the two limitations above are combined, we see a general view of how the experiment should be set up- The control must be placed as close as possible to the source while still allowing enough time to calculate the desired control wave from the reference input on the DSP. We then place as much distance as we can between the two speakers and our error mic in order to achieve the greatest zone of attenuation. In the confines of a bed, however, the spacing is limited. The typical queen and king-sized beds are 60 and 76 inches respectively. This gives us four or five feet possible in between the source and error mic in a realistic sleeping setting. With these limitations in mind, we will see the maximum attenuation achievable.

1.7 Previous Work on ANC and Snoring

While there has not been a plethora of research done in the area of ANC and snoring, it is not an unexplored area and there have been papers published. Sen M. Kuo in the Department of Electrical Engineering at Northern Illinois University in particular has published many papers. In one publication he achieved an average attenuation of 15-20 dB in his best runs [11]. However, the data shown concentrate on the range from 100-300 Hz. In this paper we show the entire spectrum over which we hope to attenuate the sound, as the sounds over 300 Hz continue to contribute quite heavily to the overall sound level of the snore.

1.8 Experiment and Results

My experiment builds upon and reinforces the work done previously. The setup is similar to some of the earlier work done, but with only one control speaker instead of two. The experiment was done in the small anechoic chamber at Brigham Young University. Using the ANC controller developed by Ben Faber, I was able to achieve attenuation between three and four dB. While this is less than the attenuation reported by Kuo et. al [11] , this includes the entire spectrum of sound measured. Though the attenuation is not as large as hoped, the tests show that attenuation can be achieved and that through improvements on the system ANC can be a viable option for reducing noise from snoring.

Chapter 2

Experimental Setup

2.1 Computational Work

Before beginning work with an actual ANC system, it is best to better understand the mathematical processes by constructing a computational model. I did this in Matlab at the request of Dr. Sommerfeldt. This model began with a simple sine wave in a system with known values of P and H. I then progressed to a more difficult model with online system identification, though still using a sine wave as the source. Finally I moved on to a model with online system identification and a random noise source. The code for this is found in Appendix A.

2.2 Overview of Equipment Used

For the experiment we used microphones from Larson Davis. The microphones were model 2551, a 1/2 inch prepolarized free-field microphone. The 1/2 inch microphone has a uniform response from 20Hz (The lower bound of human hearing) up to about 1kHz. The microphone requires a preamplifier, also from Larson Davis, model TMS426C01. This preamp is powered by an ICP© Signal Conditioner from PCB Piezotronics, model 483B07.

The speakers are from Mackie, model HR624. These speakers have a built in amplifier for the signals coming in and have a fairly uniform free-field frequency response, ± 1.5dB from 52 Hz to 20kHz.

The uniform response of the microphones and speakers enables the DSP to focus on the noises that are actually the loudest. We don't have to worry about whether there are some frequencies that are being missed because the microphone is not as sensitive at those frequencies. For the range of frequencies we are interested in (From about 20-1500 Hz) the response of the microphones and the speakers are flat, giving us a uniform response across all frequencies.

In order to limit the frequencies being processed by our DSP to avoid aliasing, the signals returning from the error mic and the signal from the DSP to the control speaker were both fed into a filter , the Krohn-Hite 3384.

Signals were generated using the HP 35670A Dynamic Signal Analyzer and from audio recordings on a MacBook Pro.

The DSP is controlled using the ANCRemote software developed by Ben Faber at BYU. Code Composer Studio 2 is used to connect to the DSP and load the program, at which point ANCRemote is used to turn the system on and off and change values such as the convergence parameter μ.

2.3 Algorithm used in calculation

The algorithm used in our system is the standard algorithm described in section (1.3). However, the system uses offline system identification . The transfer function Ĥ is determined by emitting noise for a short period of time before the snoring noise has started to determine the impulse response. I found that μ values of approximately $5.0 \cdot 10^{-8}$ were effective.

2.4 Preliminary tests

Many tests were performed as the I began to understand the ANC system to see the robustness of the system in different circumstances. I found that one precaution that needed to be taken was to guard against aliasing. If the sampling frequency is not high enough on the DSP then some frequencies may be interpreted as higher ones. Even though we played a pure sine wave at 250 Hz, the DSP would output a signal at 250 Hz and 500 Hz. Typically the maximum frequency that you can accurately measure is $f_s/2$ where f_s is the sampling frequency of the DSP. However, to be safer it is better to set the maximum frequency at $f_s/2.5$. As our system is taking data at $f_s = 4000Hz$, we set our maximum frequency at 1500 Hz. This was done by putting the reference signal and the error signal through a low-pass filter set at 1500 Hz before they went into the DSP.

Another difficulty came from lower frequency tones. In each run there were strong low-frequency tones. Though these tones, on the order of less than 1 Hz, were well below the level of human hearing, the DSP still detected these tones and the feedforward algorithm responded greatest to these frequencies, leaving other frequencies in the audible range less uncontrolled. High-pass filters were used with a cutoff frequency at 20 Hz, the lower end of the audible range for humans.

2.5 Experimental Setup

As stated in Section 1.6 there are many physical limitations to ANC. These limitations constrain the possible distances at which the control speaker can be placed. The control speaker must be placed with enough distance between it and the source that there is enough time to calculate the control wave, but it must also be far enough away from the error microphone in order to maximize the size of the zone of silence around the microphone. In order to accomodate each possibility, various distances between the error microphone and the control speaker were tried.

Each trial resembled Fig. (2.1). The distance in between the source and the error microphone

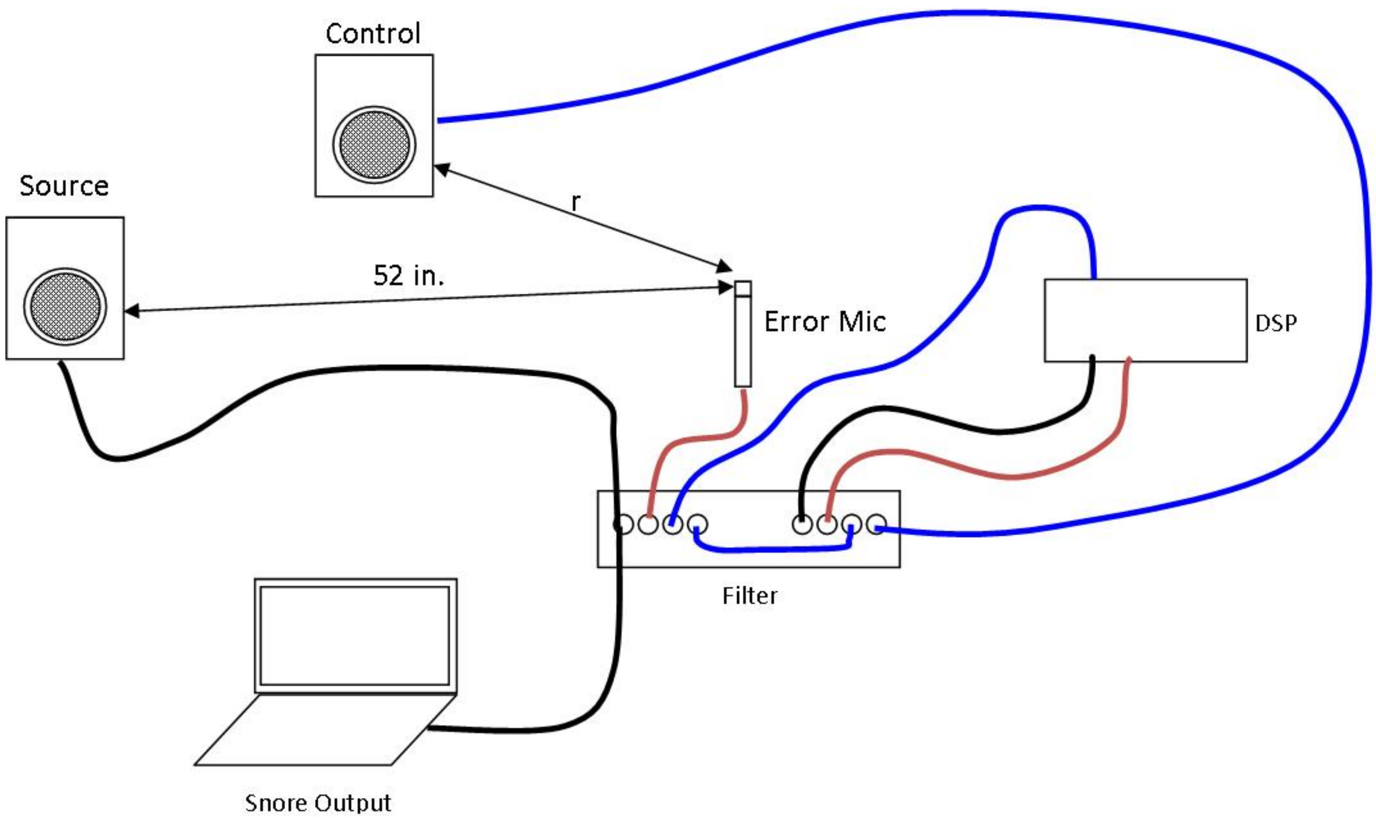

Figure 2.1 Our setup including filters.

was kept constant at 52 inches. This reflects the average king bed size (86 inches wide) minus one foot on each side, which should reflect the maximum space available in a king bed in between the snorer and the other person. The value of *r* was changed but the countrol speaker was kept close to the line in between the source and the error microphone. The different r values tested in this setup were 20, 23, 25, 27, 29, 31, and 33 inches.

Following these tests, additional tests were performed at a distance of 27 inches using a different number of taps, or coefficients. These taps correspond to the number of coefficients kept in $\hat{\mathrm{H}}$ and W by the DSP. There are two types of taps used by the DSP, control taps and SysID taps. The SysID taps correspond to $\hat{\mathrm{H}}$ and are obtained by characterizing the impulse response of the system before control begins. The control taps correspond to W and are updated according to Eq. 1.2. In the next set of experiments we varied the number of control and SysID taps in order to achieve maximum attenuation. In general, the greater number of taps you have the more accurately the system can characterize the sound and response. However, a greater number of taps also increased the computation time needed.

The experiment was carried on in the small anechoic chamber at BYU. This provided an isolated environment ideal for a controlled experiment. An ideal though not necessarily realistic setup has one source, the sound recorded on my computer, which is teed off to two different places- the source speaker and the reference input on the DSP. The DSP then takes the reference signal and calculates its output wave, which is then sent out through the filter to the control speaker. The error microphone records the pressure and that information is relayed through the filter back to the DSP, which is then updated according to the algorithm in section 1.5. The signal from the error microphone is also recorded to calculate the sound pressure level (SPL) to compare the results of each trial.

Chapter 3

Results and Attenuation

3.1 Computation Results

The results from the computational models give us an upper limit for the attenuation possible. The tests also provide more information about what parameters make the ANC system run as effectively as possible. It was discovered that in all three situations, attenuation is possible. Given enough time, each configuration was able to lower the pressure by a factor of at least 10^{-10}, the equivalent of 200 dB. However, in each case the time it took to achieve this attenuation was different. As the system became more complex it took additional time to arrive at the solution. In addition, I found that the value for the convergence parameter α was much higher than the value for μ.

3.2 Comparing Results Between Trials

After each trial it is necessary to easily compare runs. Preferably such a comparison would give us a single number to attach to each run to make the attenuation easy to see. One possible metric to use is the overall sound pressure level (SPL) calculated from the rms pressure. While using this method produces one single number to compare, it does not take into account the fact that the ear responds differently to higher and lower frequencies. In order to take the strength of each frequency into account, we need to first perform a fourier transform on each run. We then calculate the dB at each frequency given by our transform relative to one Volt. Since we are looking at the attenuation, or change between two trials, we can look at the change in dB relative to 1 V and compare the two trials.

In order to better estimate how the ear would hear each frequency we need to use the A-weighting curve. The A-weighting curve shown in figure 3.1 illustrates the response of the ear from 20 Hz, the lower limit to our hearing, to 2 kHz, the upper limit of our testing in this experiment. Figure 3.1 shows that our hearing drops off dramatically as the frequency of the sound lowers. To account for the way in which we hear sound we simply add this A-weighting curve to the number of dB calculated at each frequency.

The SPL given after accounting for A-weighting are expressed in dBA. Using the SPL given in dBA we can calculate L_A, the overall A-weighted SPL. Since we have the SPL at each frequency, we can then add these levels together using the formula [7]

$$L_{tot} = 10\log_{10}\left(\sum_n 10^{\frac{L_n}{10}}\right). \tag{3.1}$$

This gives us our L_A, a single number we can compare across each trial.

3.3 Data for Individual Trials

Fig. 3.2 shows the fourier transform of our snore, taken with no control being used. We can see a few prominent peaks, most pronounced around 250 Hz and just over 1 kHz. The sound also drops off quite sharply after the 2 kHz mark, which is good because for our first trials we were limiting our output to only 1500 Hz. As we are concentrating mainly on the region up to 2000 Hz, another graph showing the spectrum from 0 to 2000 Hz is shown as well.

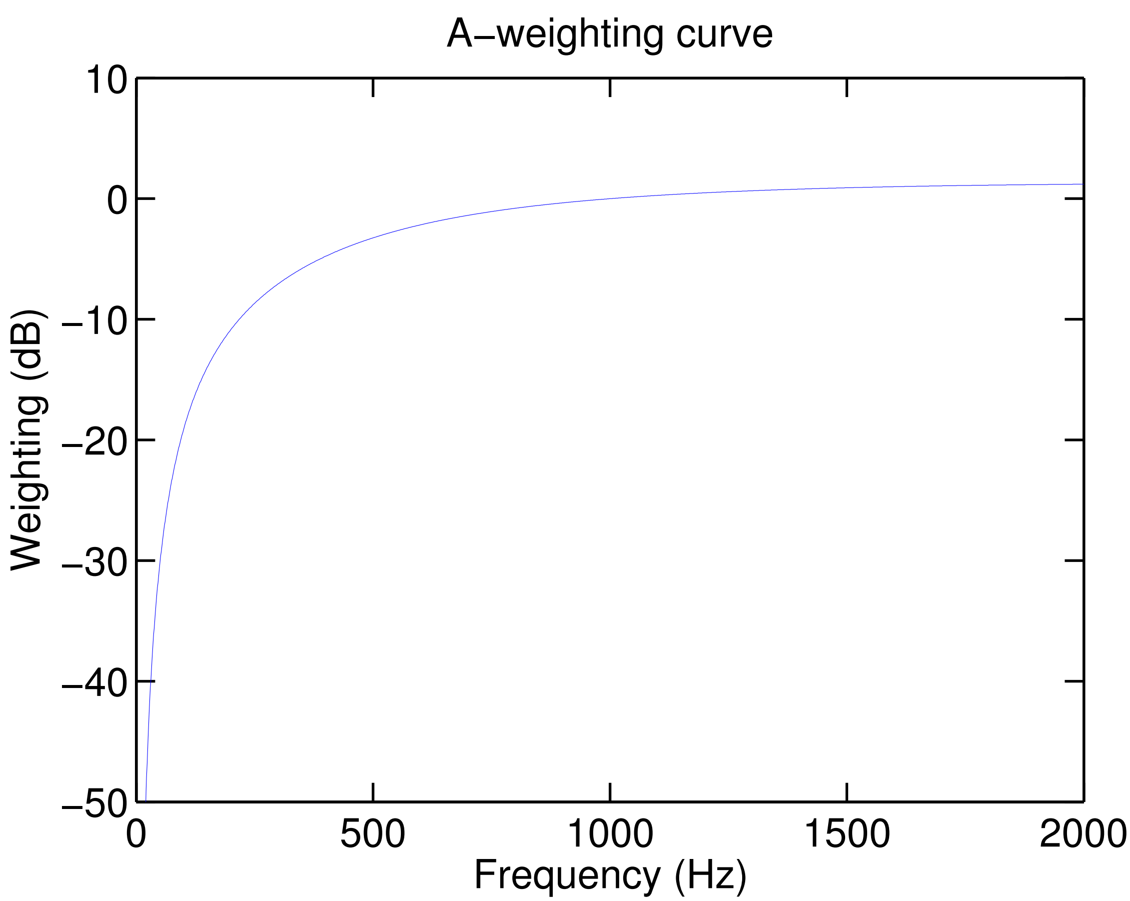

Figure 3.1 This portion of the A-weighting curve shows how the ear responds to frequencies from 20 Hz to 2 kHz.

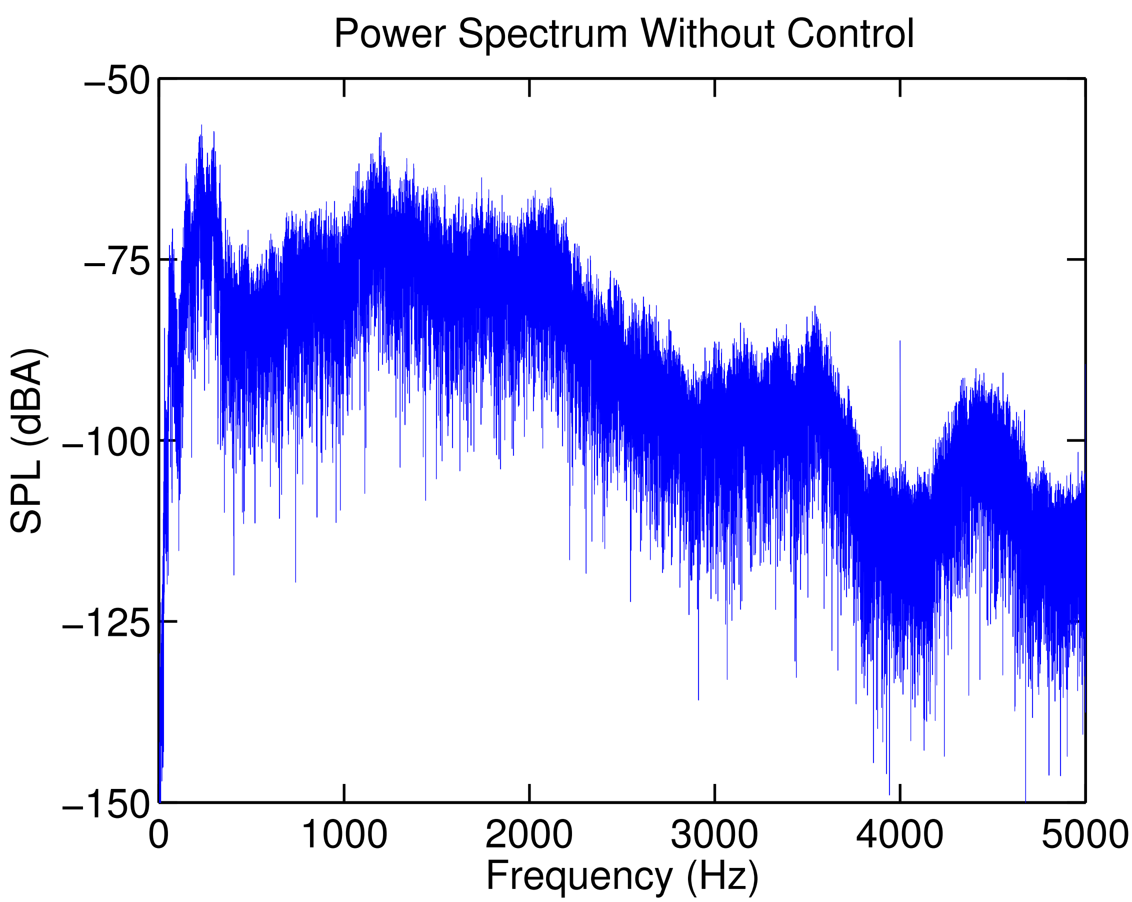

Figure 3.2 The power spectrum of a snore with no control added.

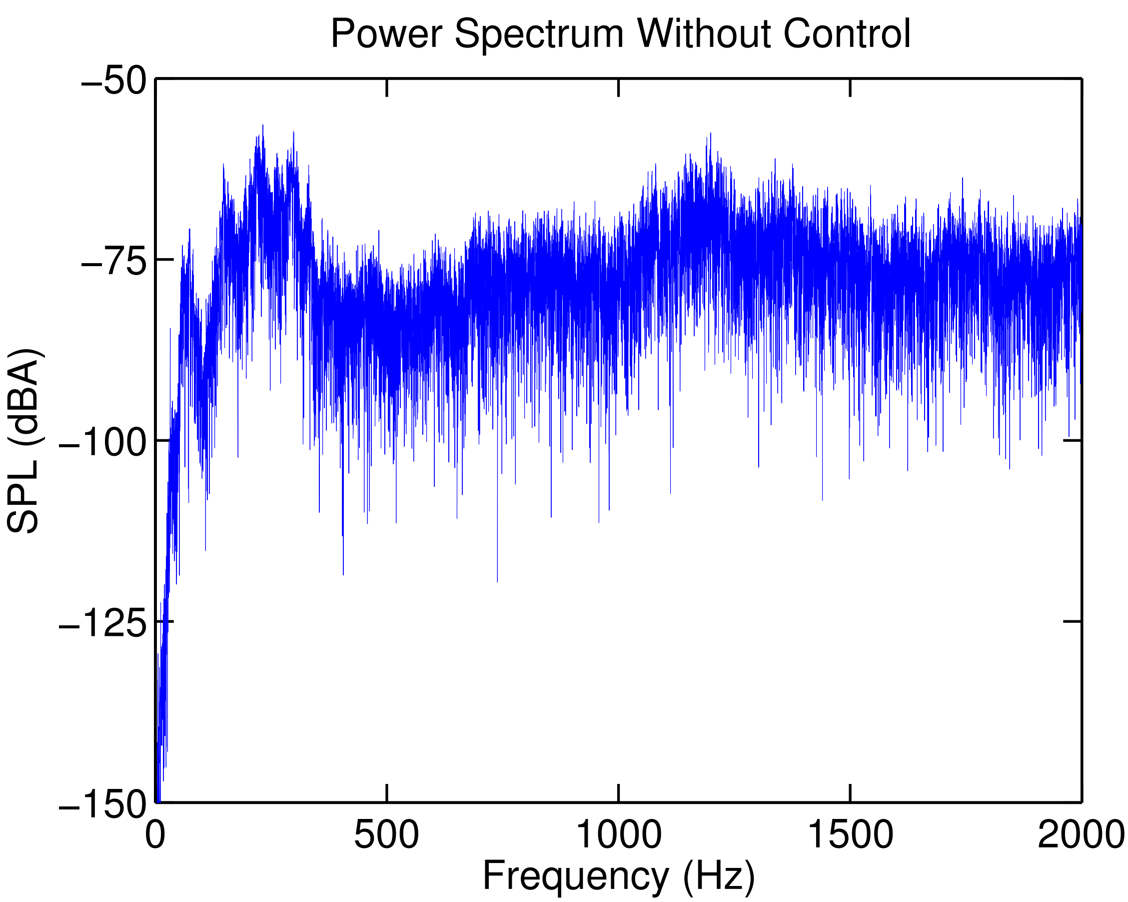

Figure 3.3 The power spectrum of a snore with no control added from 0 to 2000 Hz.

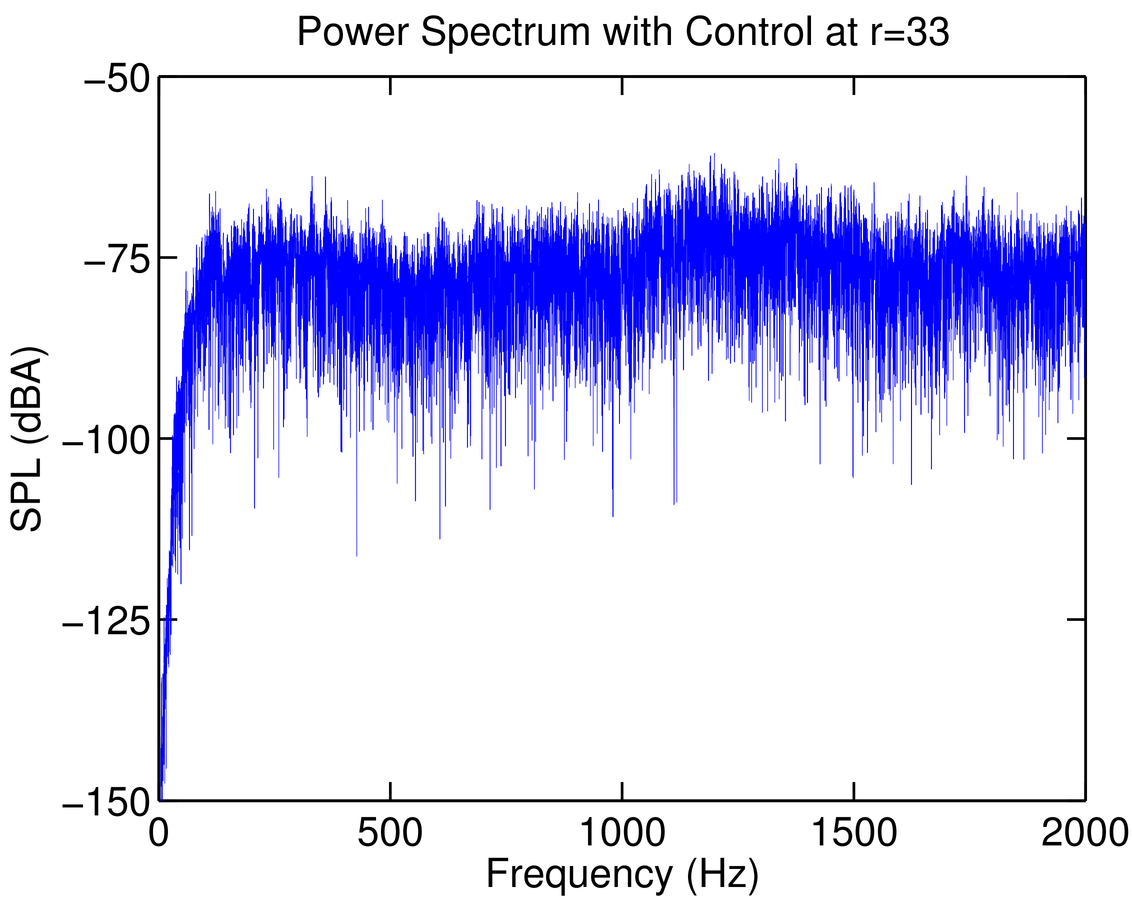

Figure 3.4 The power spectrum of a snore with the control speaker at $r = 33$ in.

These graphs enable us to see what the ANC system is doing to control the output. I include spectra from two of our runs, the smallest and largest values of r, which adequately represent the range of results.

As we can see from Fig. 3.4 the two most prominent peaks visible in Fig. 3.3 have been brought down in magnitude. The rest of the spectrum could still be brought down more, but the fact that those two peaks have been brought down is inspiring and tells us that we are on the right path. It also shows that the ANC system tends to attack the largest peaks first when trying to maximize the attenuation.

Figure 3.5 shows the spectrum with the control speaker located at $r = 20$ in. As you can see

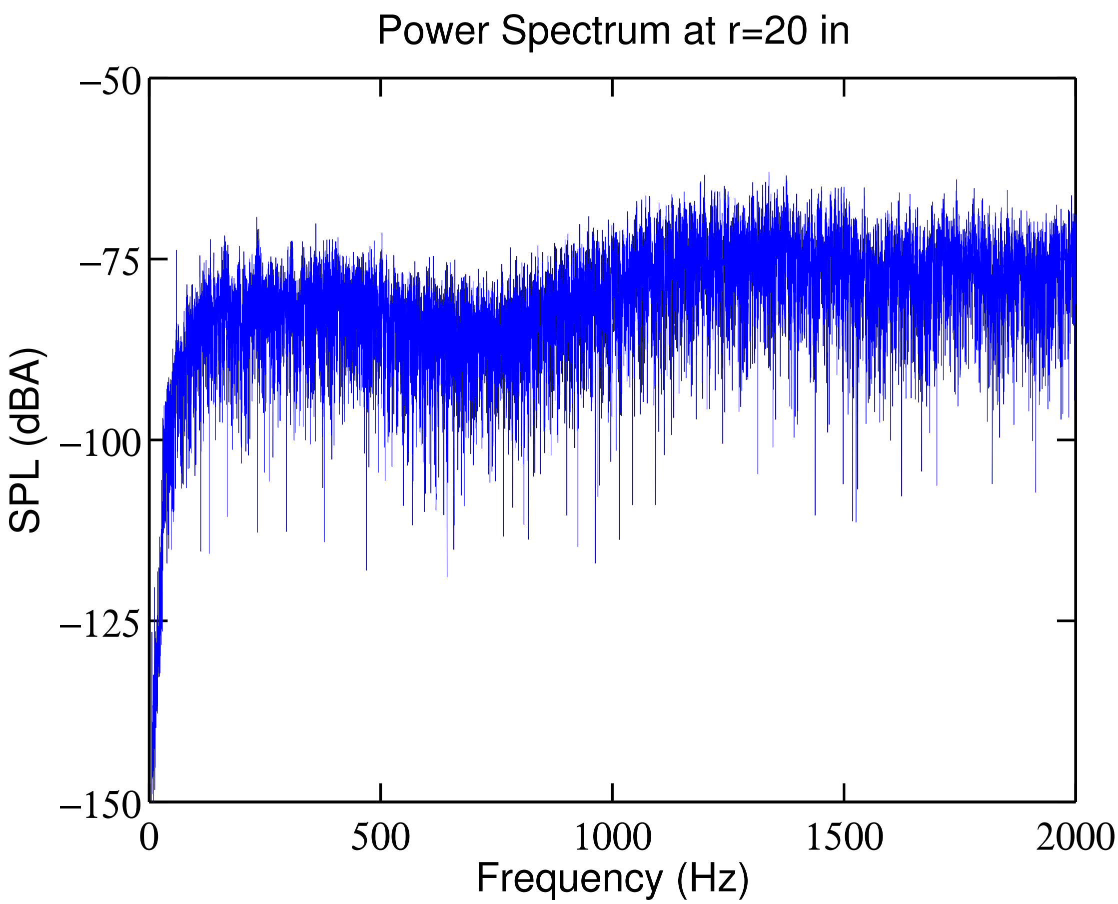

Figure 3.5 This shows the power spectrum of a snore with the control speaker located at $r = 20$ in.

r (in)	Attenuation (dBA)
20	3.2
23	2.3
25	3.2
27	3.0
29	1.4
31	1.5
33	1.6

Table 3.1 The attenuation as the distance r was varied.

the attenuation is even greater for many of the regions. The SPL below 1000 Hz sees a big dropoff. This attenuation is not seen all the way up until 1500 Hz, our cutoff frequency for our control, but it is still even more promising.

3.4 Attenuation Achieved

Table 3.1 shows the attenuation achieved for varying r. As a reference, a drop in about 6 dB corresponds to the pressure being cut in half. A change of about 3 dB is what the human ear can percieve as a difference in sound level. Except for the measurement at $r = 23$ in, which can probably be attributed to random error, the attenuation level is fairly constant from $r = 20$ to $r = 27$. After that the attenuation drops off to an almost unnoticeable change when r is greater than 27 inches.

Table 3.1 shows that there is some attenuation that can be achieved at all the distances tested, and further distances should be tested in order to find the limits of the system, in particular how large r can be before there is no attenuation at all.

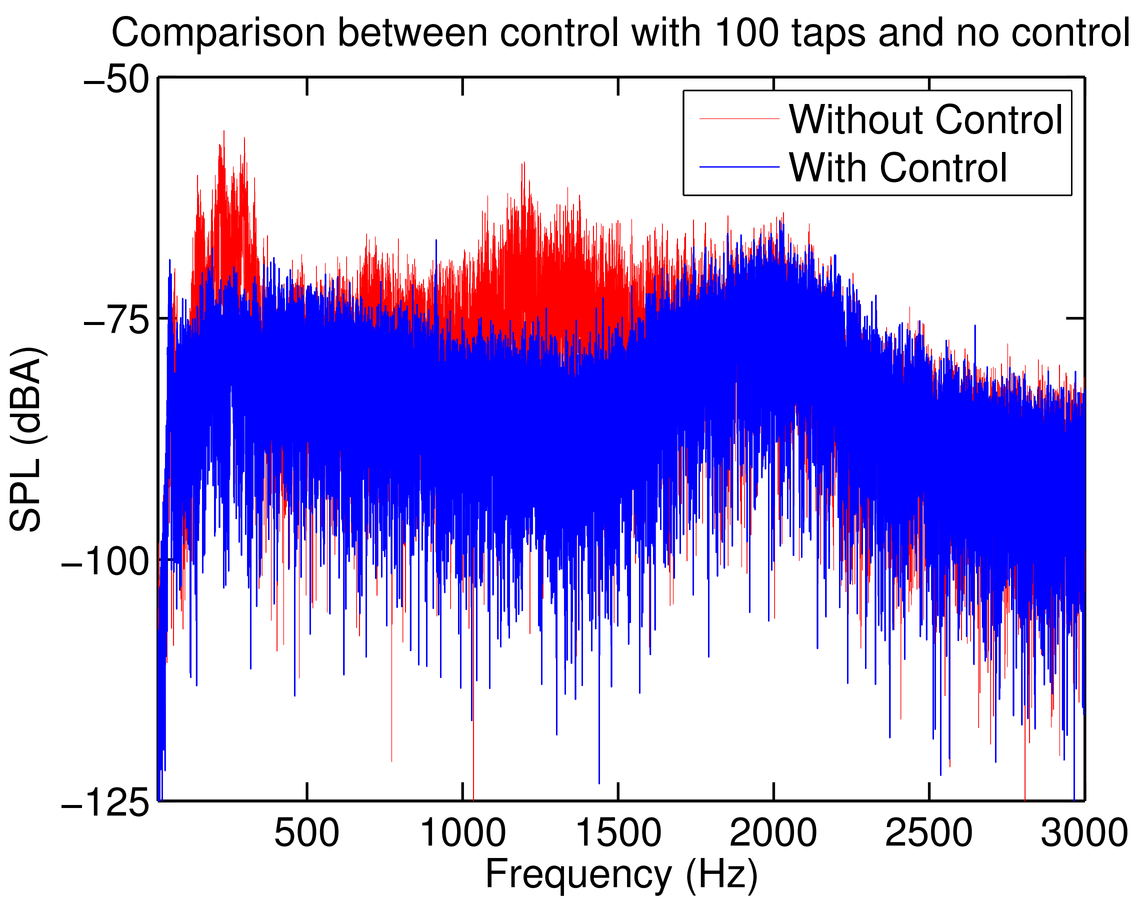

Figure 3.6 This shows the power spectrum of a snore when the DSP is storing 100 control and 100 sysID taps.

Number of Taps (Control, SysID)	Attenuation (dBA)
100, 100	5.9
125, 125	5.6
125, 175	4.5

Table 3.2 The attenuation as the number of taps was varied.

Number of Taps (Control, SysID)	Attenuation (dBA)
100, 100	9.0
125, 125	7.8
125, 175	7.1

Table 3.3 The attenuation as the number of taps was varied, examining only the range from 20 to 1500 Hz.

The tests with a greater number of taps showed a much greater attenuation achieved, as shown in Fig. 3.6. However, Table 3.3 shows that the attenuation did not increase unvaryingly as the number of taps increased. When the number of both control taps and SysID taps were the same, the attenuation was the largest. When higher numbers of control taps were tried, the system did not have sufficient computational time in between each run and quit working.

Using these three runs with a greater number of taps, it is important to consider the attenuation, not only in the broadband spectrum, but also specifically in the area in which the DSP is effective. Looking at Fig. 3.6, the attenuation is concentrated mainly in the region from 20 to 1500 Hz. Using Eq. 3.1, but taking into account only the frequencies in this range, we get an even greater attenuation more in line with many of the tests performed by Kuo et. al [11].

3.5 Discussion of Results

Throughout the course of the experiment various improvements were made to provide cases of more and more attenuation. Though preliminary tests with attenuation of 3 dBA were slightly discouraging, an attenuation of 6 dBA is an audible difference. Though more work must be done to discover the sound field in the area directly surrounding the error microphone, where a person's ears may be located, the data is a promising start indicating that attenuation of snoring through ANC could be a viable option.

3.6 Suggestions for Further Study

Improvements can still be made to the system. One possible change could be to analyze the energy density instead of the pressure level. In the near field the energy density can be used to better describe the energy content, and working with that metric may prove to be more effective in noise cancellation at short distances. Another possible improvement could be to use a multiple speaker setup, as was the case with some of the papers mentioned before. As mentioned in the previous section, more data must be gathered about the area directly around the error microphone. The attenuation achieved now is a promising start, and it is hoped that with more work and improvements ANC could provide a either a complement to passive noise control or a stand-alone method to reduce snoring noise.

Appendix A

Matlab Code Simulating an ANC System Using Offline System Identification

```
clear; close all;
length=50;
seconds=5;
rate=1000;
base=1:length;
mu=.005;
alpha=.01

ntrans=zeros(1,length);
n=ntrans';

ptrans=zeros(1,length);
ptrans(4)=.99;
```

```
p=ptrans';

htrans=zeros(1,length);
htrans(3)=.99;
h=htrans';

wtrans=zeros(1,length);
w=wtrans';

utrans=zeros(1,length);
u=utrans';

rtrans=zeros(1,length);
r=rtrans';

etrans=zeros(1,length);
e=rtrans';

for t=1:seconds*rate
    for x=1:49;
        n(length+1-x)=n(length-x);
        e(length+1-x)=e(length-x);
        u(length+1-x)=u(length-x);
        r(length+1-x)=r(length-x);
    end
```

```
    n(1)=rand(1)*2-1;
     u(1)=w'*n;
     e(1)=p'*n+h'*u;
    r(1)=h'*n;
    w=w-mu*e(1)*r;
    subplot(3,1,1)
plot(base,n)
title('n')
subplot(3,1,2)
plot(base,e)
title('e')
subplot(3,1,3)
plot(base,w)
title('w')
pause(.002)
end
```

Bibliography

[1] Z. Liu, X. Luo, H. Lee, and C. Lu, "Snoring source identification and snoring noise prediction," Journal of Biomechanics **40,** 861–870 (2007).

[2] P. G. Norton and E. V. Dunn, "Snoring as a risk factor for disease: an epidemiological survey," British Medical Journal **291,** 630–632 (1985).

[3] S. R. Chakravarthy and S. M. Kuo, "Application of Active Noise Control for Reducing Snore," ICASSP 2006 Proceedings **5,** 305–308 (2006).

[4] D. Guicking, "On the invention of active noise control by Paul Lueg," J. Acoust. Soc. Am. **87,** 2251–2254 (1990).

[5] S. D. Snyder, in *Active Noise Control Primer*, R. T. Beyer, ed., (Springer Verlag, 2000).

[6] S. A. Dyer and B. K. Harms, in *Advances in Computers*, M. C. Yovits, ed., (Academic Press, 1993), Vol. 37.

[7] L. E. Kinsler, A. R. Frey, A. B. Coppens, and J. V. Sanders, in *Fundamentals of Acoustics*, 4th ed., B. Russiello, ed., (John Wiley & Sons, Inc., 2000).

[8] J. Blauert and N. Xiang, *Acoustics for Engineers*, 2nd ed. (Springer, 2009).

[9] S. M. Kuo and D. R. Morgan, "Active Noise Control: A Tutorial Review," Proceedings of the IEEE **87,** 943–973 (1999).

[10] P. R. Cook, in *Springer Handbook of Acoustics*, 1st ed., T. D. Rossing, ed., (Springer, 2007).

[11] S. M. Kuo and R. Gireddy, "Real-Time Experiment of Snore Active Noise Control," IEEE Internation Conference on Control Applications (2007).

Index