Bahir Dar University

Bahir Dar Institute of Technology

Faculty of Electrical and Computer Engineering

Department of Electrical Engineering

Semester Project

Project Title: Active Noise Cancellation Device

Name	**Id**
1.Desalegn Kindie	**BDU-1201447**
2.Eyob Nigatu	**BDU-1201629**
3.Eyoel Melaku	**BDU-1201632**
4.Getnet Fentaw	**BDU-1201834**

Advisor Name: Mr. Yimesgen Getahun

Date:23/08/2016

BAHIR DAR, ETHIOPIA

Declaration

We are students of Bahir Dar University in Bahir Dar Institute of technology (BIT), faculty of Electrical and Computer Engineering. The information found in this project is our original work and all sources of materials that will be used for the project work will be fully acknowledged.

Name	Signature
1.Desalagn Kindie	___________________________
2.Eyob Nigatu	___________________________
3.Eyoel Melaku	___________________________
4.Getnet Fentaw	___________________________

Date of Submission:

This semester project has been submitted for examination with our approval as a university advisor.

Project Advisor Name	Signature	Date
Mr. Yimesigen Getahun	____________________________	____________________________

Acknowledgment

We express our sincere gratitude to God Almighty for His blessings, wisdom, and strength, which have enabled us to successfully complete this project. Additionally, we extend our thanks to our respected advisor, Mr. Yimesigen, for his invaluable direction and guidance throughout this endeavor. His insights and support have been instrumental in navigating the challenges of this project and ensuring its successful completion. We deeply appreciate his dedication and expertise, which have significantly contributed to our academic and personal growth.

Table of Content

List of Figure

List of Tables

Abbreviations

ADC--- Analog Digital Converter

ANC--- Active Noise Cancellation

ANR--- Active Noise Reduction

DAC--- Digital to Analog Converter

FIR -- Finite Impulse Response

FuLMS --- Filtered-u-Least Mean Square

FxLMS-- Filtered-x-Least Mean Square

IIR-- Infinite Impulse Response

LMS--- Least-Mean Square

MATHLAB-- Matrix Laboratory

NLMS--- Normalized Least Mean Square

PSD-- Power Spectrum Density

RLS--- Recursive Least Square

Abstract

Active Noise Cancellation reduces unwanted noise from the ambiance, enhancing the listening experience of mobile communications and multimedia applications. The active noise-reducing headphone stands out as one of the most successful applications of active sound control. This project aims to study the principles of Active Noise Cancellation using both analog and digital mechanisms, focusing on headset applications.

The objective is to design and implement active noise control systems, comparing analog and digital approaches. The method involves introducing both analog and digital controllers, starting with analog feedback controllers and proceeding to discuss digital control algorithms. For the analog case, a noise-cancelling headphone with a negative feedback system using a low-pass filter is designed. Analog filters are incorporated to pre-process the input signal, effectively suppressing noise components within specific frequency bands.

The trade-off in the analog design lies in the performance and stability of the system, given the use of a low-pass filter. Finally, the frequency response of the noise attenuation, audio signal, and the loop gain from the negative feedback system is calculated and simulated using MATLAB. A robust method for a noise cancellation system is the use of digital filtering. In this project, these types of filters are employed using the adaptive filter, which is applied using LMS (Least Mean Square) and NLMS (Normalized Least Mean Square) Algorithm to remove noise. The noise cancellation performance between the two algorithms is also explained, with NLMS demonstrating better performance and faster convergence speed compared to LMS.

Keyword: LMS, NLMS, ANC, negative feedback, MATHLAB

Chapter one

Introduction

In our fast-paced and mobile world, people engage in various activities while on the move. Listening to music is no exception. However, when enjoying music outside the comfort of home, we often encounter less predictable noise environments. Whether it’s the roar of airplane engines, the hum of bus or car motors, or the persistent buzz of lawnmowers, noise can disrupt our auditory experience.

Active Noise Cancellation is a method for reducing undesired noise. ANC is achieving by cancelling anti-noise wave through secondary sources. These secondary sources interconnect through an electronic system using a specific signal-processing algorithm for the particular cancellation scheme. Our project is to build a Noise-cancelling headphone by means of active noise control. Essentially, this involves using a microphone, placed near the ear, and electronic circuitry, which generates an "anti-noise" sound wave with the opposite polarity of the sound wave arriving at the microphone. This results in destructive interference, which cancels out the noise within the enclosed volume of the headphone. This project will demonstrate the approaches that we take on tackling the noise cancellation effects. The electronic circuitry is of two types. The first is low-pass filter and in the second case adaptive filter.

Noise Cancellation makes use of the notion of destructive interference. When two sinusoidal waves superimpose, the resulting waveform depends on the frequency amplitude and relative phase of the two waves. If the original wave and the inverse of the original wave encounter at a junction at the same time, total cancellation occur. The challenges are to identify the original signal and generate the inverse without delay in all directions where noises interact and superimpose.

1.1 Background about Noise

Any unwanted signal that interferes with the communication or measurement of an information carrying signal is termed as noise.

The noise we have considered in order to build the project are:

1.**Random Noise**

Random noise usually refers to electric or acoustic signal that consists of equal amounts of all frequencies. Power spectral Density of random noise:

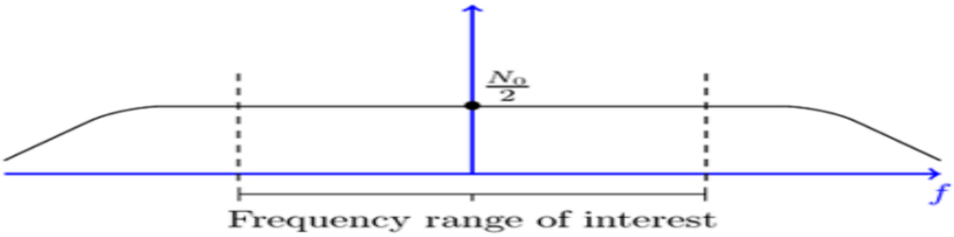

Figure1.1:Thermal noise (random noise)

2. **White noise:**

White noise is a random signal having equal intensity at different frequencies, giving it a constant power spectral density. The power spectral density (PSD) of additive white Gaussian noise:

$$s_x(f) = \frac{N_0}{2}$$

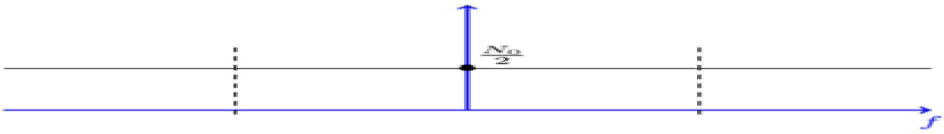

Figure 1.2: White noise

Noise Reduction

The consequences of exposing people to noise from various sources may vary from short term effects such as sleep disturbance to long term effects such as permanent hearing loss. To reduce the noise from source reaching our ear involves various methods which can be categorized into

1. **Passive Noise Control**

Passive noise control is a method in which the noise from the source is not allowed to reach the ear of the person. This is done by blocking the path of the noise using absorbing materials or by reflecting the noise in some other direction. Thermopolis or polystyrene, clothes and wood are some examples of the materials which absorb the noise and reduce the adverse effects occurring.

Figure 1.3: Passive Noise Reducing Headset

2. **Active Noise Control**

Active noise control is a very effective electronic method to reduce the effect of the noise in an environment. It is basically generation of anti-noise, equal in magnitude and opposite in phase with the noise. The anti-noise and the noise are destructively interfered to remove the effects of noise from the path of the noise. While passive and active noise cancellation may be applied separately, they are often combined to attain maximum effectiveness in noise cancellation.

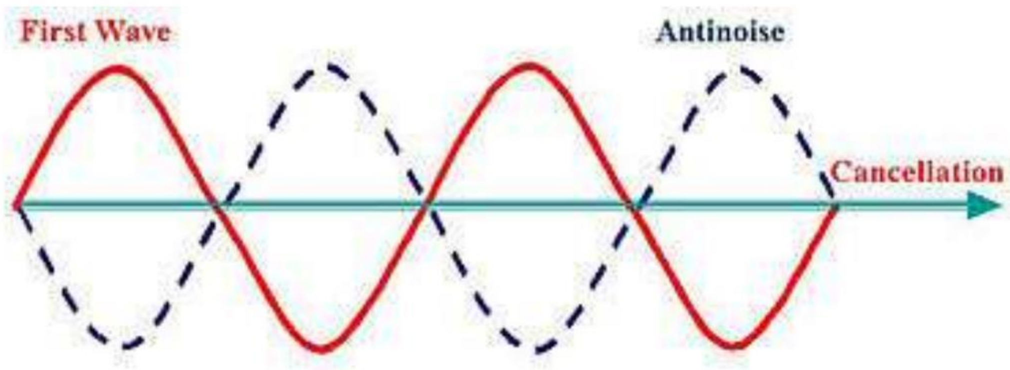

Figure 1.4: ANC

1.2 Statement of the Problem

The increasing use of machinery in modern life has created an environment saturated with unwanted noise. This noise pollution, generated by engines, transformers, compressors, and even everyday appliances, negatively impacts our lives in various ways.

Noise pollution occurs when unwanted sounds infiltrate the environment, affecting both humans and animals. Despite its pervasive presence, noise pollution is often overlooked as a health hazard. Here are the key aspects of this problem:

- Impact on Mental Health: Frequent or loud noise triggers anxiety and stress. The brain remains vigilant for signs of danger even during sleep. People living with noise pollution may experience irritability, frustration, and heightened sensitivity to stress. Environmental noise disrupts sleep, leading to difficulty falling asleep, staying asleep, or waking too early. Sleep quality and mood are affected.
- Impact on Physical Health: Direct Hearing Impairment: Severe noise exposure can directly cause hearing loss, abnormal loudness perception, tinnitus (persistent ringing in the ears), and distorted hearing.

- Indirect Health Effects: Short-term noise exposure may temporarily raise blood pressure and increase blood viscosity. Long-term exposure to noise is associated with higher rates of cardiovascular disease due to its impact on stress hormone levels and the nervous system.

In summary, noise pollution affects mental well-being, sleep quality, and physical health. Addressing this problem requires understanding its sources and implementing effective mitigation strategies. From hindering clear communication and sleep to causing stress and even hearing damage, excessive noise disrupts our well-being and overall quality of life. Furthermore, the need for personal quiet time often conflicts with the needs and activities of others, creating additional challenges. This project aims to address these issues by proposing an Active Noise Cancellation (ANC) solution that offers a way to mitigate unwanted noise without restricting others' activities or hindering essential machinery operations.

1.3 Objective

1.3.1 **General Objective**

The main aim of this semester project is to design and simulate active noise cancelling headphone using negative feedback system and adaptive filters .

1.3.2 **Specific Objective**

- To reduce unwanted or undesired noise.
- To demonstrate the performance and stability of our feedback system using MATHLAB.
- Implement digital ANC using LMS algorithm and NLMS algorithm to cancel different noises from an audio signal.
- To show the difference between active and passive noise cancelling.

1.4 Methodology

The methods that we have used to do our project are described below. The first thing is that we have prepared a proper time schedule that specifies the things we will do. Then we have gathered different information that is needed for our project from sources such as, books and internet. We then assemble each information to a meaningful and in supportive way to the project. Since we have already chosen the software that we are going to use for our project, i.e. Mat-lab, we have designed and analyzed the model. We designed our circuit in Simulink and simulate it using Mat-lab. To do so we have figured out the code by reading different sources about Mat-lab coding which are related to digital signal processing. Finally, we have analyze simulated result in accordance with the objectives listed.

We have read through different materials regarding our project and we collected the data. The next thing we have done is choosing the right procedure of implementation of the project. That is we have done our work based on classes of active noise cancellation. The first class is analog active noise cancellation and the second is digital active noise cancellation. In the analog case we have used a low pass filter to cancel low frequency noise. And the noise attenuation will be shown with MATLAB using bode plot of the designed circuit. In case of the digital ANC an adaptive filter we have implemented NLMS algorithm and LMS algorithms. And the noise cancellation mechanism is has been shown by MATLAB using both graphs and real-world noise and cancellation of this noise.

1.5 Significance of the Project

Active noise cancellation is essentially a way to reduce unwanted background noise by actually playing an anti-noise signal that cancels out the noise. Sound is a pressure wave with alternating periods of compression and rarefaction. The anti-noise signal is an inverted version of the noise signal. When the noise and anti-noise signals are played together, pressure periods of one occurs at the same time as rarefaction periods of the other such that the two interfere destructively. Noise cancellation headphones makes it possible to listen to music without raising the volume excessively. It can also help a passenger sleep in noisy vehicle such as an airliner. In the aviation environment, noise cancelling headphones increase the signal-to-noise ratio significantly more than passive noise attenuating headphones or no headphones, making hearing important information such as safety announcement easier. Noise cancelling headphones can improve listening enough to completely offset the effect of a distracting concurrent activity.

1.6 Motivation

Active noise cancellation is used to reduce the volume of unwanted noise propagating through air using measurement sensors such as microphones and output actuators such as loud speakers. The traditional approach to acoustic noise control uses passive techniques that include enclosures, barriers, and silencers to attenuated undesired ambient sound. Active noise cancellation efficiently attenuates low-frequency noise where passive methods tend to be ineffective or expensive when seeking to achieve the same level of results. Noise levels in human settings often come under scrutiny for physical health and psychological concerns.

Thus research of active noise cancellation, particularly for intense low frequency ambient noise, allows for an improvement of both physical health and mental well-being. There are individuals who are not overly conscientious about the impact the intensity of sound has on hearing. Understanding the difference between noise cancellation techniques such as active noise cancellation can have an impact on the future of one's hearing ability. The effects of low-frequency noise are of particular concern because of its pervasiveness due to numerous sources, efficient propagation, and reduced efficacy of many structures in attenuating low frequency noise compared with other noise. Low-frequency noise of a high intensity can even cause symptoms of respiratory impairment and aural pain. Using active noise cancellation, noise-cancelling headphones can be created to allow for an improved environment for communication such as ham radio operations in environments with ambient low frequency noise. This project aims to research active noise cancellation techniques, applications, and implementations to provide a better understanding of active noise cancellation and how it can be used to improve Ham Radio communication. Although active noise control can be obtained through the use of analog circuitry, using a digital signal processor allows for more advanced processing and potential for future development as new advancements in noise cancelling technology are discovered. The motivation behind the research presented in this thesis is to help contribute to the understanding of active noise cancellation techniques

1.7 Scope of the Project

The scope of this semester project is to build and simulate active noise cancellation techniques, applications, and implementations to provide a better understanding of active noise cancellation and how it can be used to improve acoustic spaces such as a noisy environment. In order to accomplish this task, background information on noise cancellation and its applications is discussed in detail. The main output of this work is the research of different techniques of active noise cancellation from scholarly sources and an explanation and simulation.

Chapter Two

Literature Review

Active control was first theorized by Paul Lueg in 1936 in U.S. Patent Number 2,043,416 [7]. His patent describes measuring the sound field with a microphone, electrically manipulating the resulting signal, and then feeding it to an electroacoustic secondary source [4]. Destructive interference is created from the superposition of the acoustic wave of the loudspeaker and the wave of the original source [4]. This concept is the basis for today‘s active noise control; however, at that time Lueg could not practically demonstrate his patent.

Seventeen years later, Harry Olson and Evert May published another paper which describes another system for active noise control [4]. In contrast with Lueg‘s paper which used prior knowledge of the signal from the detecting microphone (feed-forward control), Olson and May‘s strategy needed no prior knowledge of the sound field [4]. Instead, it used a feedback method to cancel sound by feeding back the signal from a much closer microphone to a second loudspeaker [4].

A few years later in 1956, William Conover was working on acoustic noise reduction from power distribution transformers at Generic Electric Company [8]. The magnetostriction in the transformer made it hum at even harmonics of the line frequency [8]. This periodic nature of the sound allowed Conover to avoid using a microphone and instead generate a reference signal which has the same frequency components as the primary noise. These reference signals were derived from full-wave rectified version of the line voltage and band pass filtered to obtain the even harmonics [8]. Only the amplitude and phase of this reference signal need to be varied by the controller of the system [4]. The objective of his design was to cancel the pressure in a particular direction away from the transformer, such as a nearby house [4]. He carried out the cancellation using a manual controller, which he adjusted to compensate for winds and temperature changes [4]. The suggestions in his paper to use an automatic control system as well as multiple secondary loudspeakers and monitoring microphones were very important in the future development of ANC [4].

Due to the lack of capable technology in the 1930‘s and 1950‘s, ANC was not possible until modern computers became available. Research and development of active noise reduction (ANR) headphones truly began in 1978 after Dr. Amar Bose felt the need to develop headphones that masked the low rumbling of plane engines and other cabin noises. Bose Amar, "Head-phoning,"[2] this was a patent given to Bose on his work on noise cancelling headphones. His headphones have a small cavity between the diaphragm and the ear canal with a microphone in the cavity closely adjacent to the diaphragm providing a feedback signal that is combined with the input electrical signal to be reproduced by the headphones to provide a combined signal that is power amplified for driving the diaphragm. In 1980, the well-known filtered-x least mean squares algorithm was developed by Morgan and also independently by Widrow in 1981 and now figures prominently in recent active noise control research [12]. [1] Chu Moy, ―Active Noise Reduction Headphone Systems‖, 2001, In his research he describes that the noise suppression headphones evolved to noise cancellation technologies that actively nullify noise with anti-noise. Anti-noise is simply an inverted version of the noise signal, so that when noise and anti-noise meet, they cancel each other. The effect is a manifestation of the superposition principle. When two identical sound waves combine that are 180 degrees out of phase, the result is destructive interference. If the anti-noise is not a perfect replica of the noise waveform or is not exactly 180 degrees out of phase, the destructive interference will weaken the noise, but not cancel it. Noise Cancelling Headphones employ electronic circuits that are integrated with the headphone design to generate anti-noise. They are called active noise reduction systems. ANR systems create anti-noise by first sampling the noise with a microphone and then, either by inverting the noise from outside by using inverting amplifier, which is then mixed with the desired audio signal before being played back in the headphone (Open-loop System) or by sampling both the sound emitted by the headphone transducer as well as the noise in the ear cup and inverting it, then feeding back this signal to the desired audio signal input as an error correction for the noise.

In our project, we've introduced several novel features to enhance the active noise cancellation system compared to previous works. Specifically, we've integrated both analog and digital filters, leveraging an adaptive filter with the Normalized Least Mean Squares (NLMS) algorithm instead of the Least Mean Squares (LMS) algorithm used in earlier studies. Our goal is to effectively eliminate noise interference that disrupts the desired signal.

Utilizing the Normalized Least Mean Squares (NLMS) algorithm instead of the conventional Least Mean Squares (LMS) presents several benefits. Notably, NLMS demonstrates a swifter convergence rate, particularly when the input exhibits gradual variations. In terms of stability, NLMS inherently maintains a more robust equilibrium compared to LMS, which is prone to instability due to its sensitivity to input scaling. This sensitivity complicates the selection of an appropriate learning rate that ensures consistent stability. Despite NLMS necessitating increased computational efforts, the resultant expedited convergence and diminished error in relation to the unknown system justify the additional resource expenditure. In essence, NLMS surpasses LMS by offering accelerated adaptation, enhanced stability, and more effective convergence, rendering it superior in specific applications.

Chapter Three

System design and Mathematical model

3.1 ANC by Analog Mechanism

This type of headphone typically includes a loudspeaker, an error microphone, and an analog control unit. The error microphone is generally placed as close as possible to the ear canal, since the objective of the active control is principally to minimize the perceived sound pressure. The system is a close loop system and can attenuate the noise below 1 KHz.

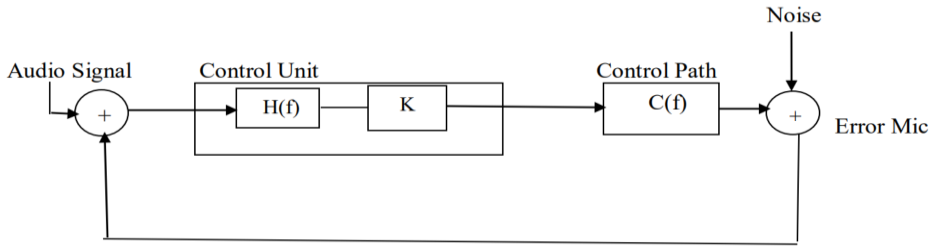

Figure 3.1: Block diagram of the feedback control system with audio input

$$P_o(f) = P_i(f)/ (1 + KC(f)H(f))$$

Where $P_o(f)$ is the sound pressure inside the analog hearing protector without control, K is the amplifier gain, H(f) is the frequency function of the compensation filter, and C(f) is the frequency function of the control path, i.e., the transfer path comprising the loudspeaker, headset cavity, and error microphone. By letting the amplifier gain K assume large values, the magnitude of the denominator in the above Equation becomes large and the sound pressure under control approaches zero.

The simplified system design for the analog active noise cancellation headphone using model will be like the figure below:

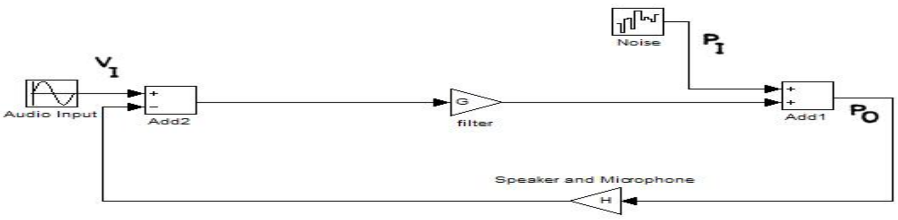

Figure 3.2: The simplified version of our Mat lab Simulink's model

VI is the audio signal, which we want to hear through the speaker. PI is the noise signals which we want to cancel. P0 is the signal, which the ear hears. H represents closed loop gain of the system.

3.2 Digital ANC System Model

A very basic single channel active noise control system consists of:

- A reference microphone sensor- to sample the disturbance to be cancelled.
- An electronic control system to process the input signal and then generate the control signal. This is basically an adaptive filter
- A loudspeaker driven by the control signal to generate the anti-noise
- An error microphone to provide feedback to the controller so that it can adjust itself to minimize the resulting sound output. The system thus described is known as an ―Adaptive System‖ as it can adapt itself according to the change in the reference signal which is feedback to the controller through the error microphone.

3.2.1 **Basic Structure of Digital ANC Systems**

Modern active sound control systems consist of a control source used to introduce a controlling signal in to the acoustic system. This disturbance reduces the unwanted noise originating from the primary sources by adding to it a signal of same magnitude but opposite in phase. The control signals that drive the control actuators are generated by an electronic controller, which use as inputs, the feedback error from the error microphone. Active noise control systems are mostly used in the low frequency range usually below 500-600 Hz. At higher frequencies, passive control measures generally become more cost effective.

Two major types of active noise control system will be considered here:

- Adaptive filtering - the coefficients are updated adaptively to reduce the error output
- Waveform synthesis-a type of feed-forward control that is suited only to periodic noise.

3.2.1.1 Adaptive Control

An adaptive controller is a controller that can change its nature in response to changes in the process and the disturbances occurring dynamically. An adaptive controller is a controller with adjustable parameter and a mechanism for adjusting the parameters. The adaptive control mechanism includes a digital filter with dynamically changing coefficients as per the requirement and with the changes in the environment.

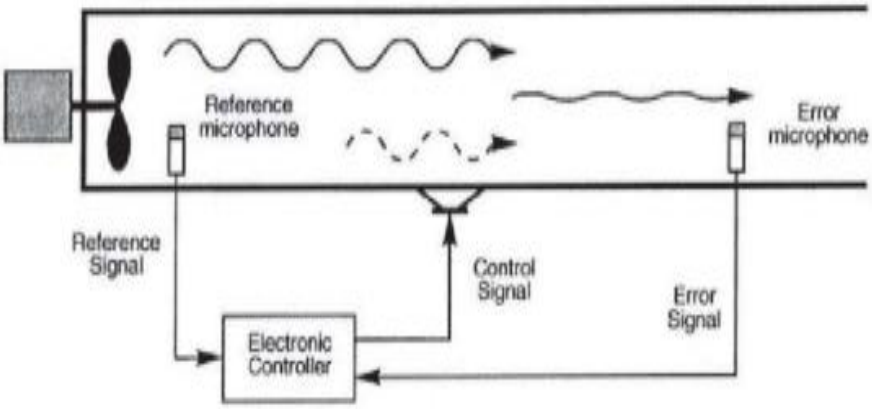

Figure 3.3: Digital ANC system block diagram with single channel

3.2.2 **Digital Filter**

The heart of the control system is a digital filter with specific properties, which synthesizes the anti-noise.

The two main requirements of the digital filter we have used here is:

(i) FIR Filter

(ii) Adaptive Filter

3.2.2.1 Adaptive Filter

The term adaptive signifies that that the filter weights are not fixed, rather these are adjustable according to the variation in external environment. The weights are randomly initialized and according to an adaptive algorithm the controller updates its weights so as to minimize the residual noise at the listener side. The significances of an adaptive filter can be summarized as filter parameters such as bandwidth and resonant frequency change with time. The coefficients vary with time and are adjusted automatically by an adaptive algorithm. Coefficient adaptation

- The purpose of determining the coefficients of the filter model is to maximize the statistical correlation between the reference signal and the coefficients.
- This in turn is done by minimizing the correlation between the error signal and the filter state as is relevant to the coefficients.
- When the adaptive filter is working, the error signal decreases in magnitude, and this slows down the movement of the coefficients. The filter is observed to be converging to a solution.

3.2.2.2 FIR Adaptive Filter

- The controller filter may take a number of forms, the most common of which is the finite Impulse Response (FIR) filter. A FIR filter may be represented as shown in the Figure below. z-1 represents a delay of one (input) sample and wi represents filter weight.
- FIR filters are ideally suited to tonal noise problems, i.e. noise with harmonics, where the reference signal is a few sinusoids and where the control signal does not in any way corrupt the reference signal.

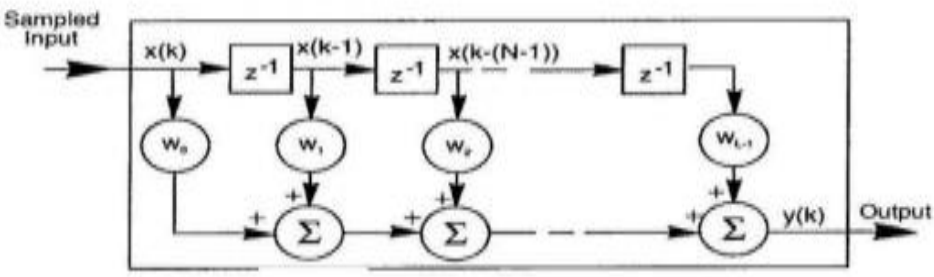

Figure 3.4: FIR filter architecture

When there are resonances in the system or if there is some noise feedback from the control source to the reference sensor, resulting in the corruption of the reference signal, the FIR filter is not the best choice and then the Infinite Impulse Response (IIR) filter is often chosen for its ability to directly model the poles in the system resulting from such effects. The main advantage of the IIR filter is that it uses fewer weight coefficients than required by a FIR filter for a complex system, thus reducing computational load. But this advantage comes at the cost of instability, slower convergence and also the possibility of convergence to a local minimum in the error surface increases instead of a global minimum. The three parameters that affect the performance of a digital filter in an active noise cancelling system are: the type of filter, the filter weight values and the number of weights. The adaptation algorithm of the controller is responsible for tuning the adaptive filter so that the resulting control signal minimizes the error signal received by the controller.

Since the characteristics of the acoustic noise source and the environment are time varying, the frequency content, amplitude, phase, and sound velocity of the undesired noise are non-stationary. An ANC system must therefore be adaptive in order to cope with these variations. Adaptive filters adjust their coefficients to minimize an error signal and can be realized as (transversal) finite impulse response (FIR), (recursive) infinite impulse response (IIR), lattice, and transform domain filters. The most common form of adaptive filter is the transversal filter using the least mean-square (LMS) algorithm.

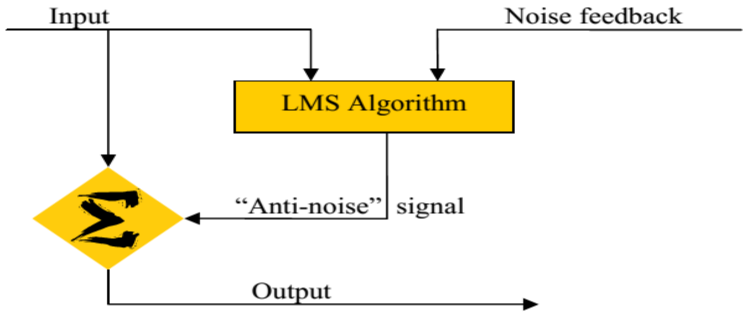

Figure 3.5: Sample ANC flow chart with LMS

3.3 The LMS Algorithm

The Least Mean Square (LMS) is an adaptive algorithm, LMS algorithm uses the estimates of the gradient vector from the available data. The LMS incorporates an iterative procedure that makes corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error.

The Least-Mean-Square algorithm in words:

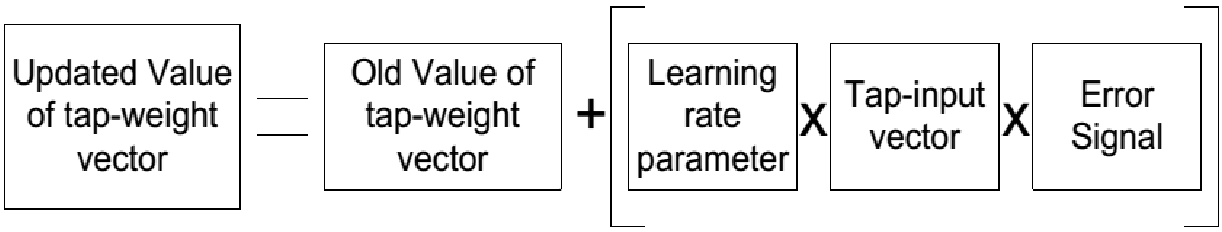

Figure 3.6: The LMS Algorithm in words

The LMS Algorithm consists of two basic processes:

- Filtering process:

1. Calculate the output of FIR filter by convolving input and taps.

2. Calculate estimation error by comparing the output to desired signal.

- Adaptation process:

Adjust tap weights based on the estimation error. The LMS algorithm is obtained by substituting the instantaneous error approximations into the basic steepest-descent Weiner filter algorithm.

The implementation steps for the LMS algorithm –

- Filter output:

$$\mathrm{y(n)}=\sum_{k=0}^{m-1} \boldsymbol{n(n-k)w_k^*(n)}$$

- Estimation error:

$$\boldsymbol{e[n] = d[n] - y[n]}$$

- Tap-weight adaptation:

$$\boldsymbol{w_k(n+1) = w_k[n] + \mu u(n-k)e^x[n]}$$

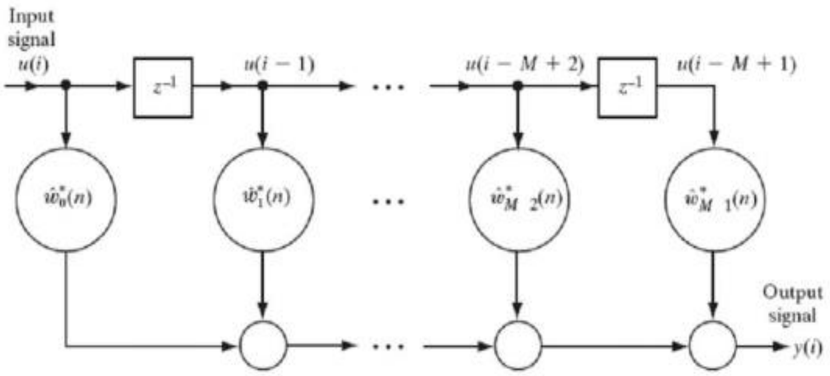

Figure 3.6: Tap weight adaptations

3.4 The NLMS Algorithm

By normalization the step size with respect to the energy in the reference signal vector in the LMS algorithm the algorithm known as NLMS algorithm is obtained. It is one of the important techniques to maintain the effective speed of convergence while maintaining the desired steady state response. Thus, NLMS algorithm solves the problem of instability of LMS algorithm due to variation in the power of the reference signal. The co-effects adjustment of Normalized LMS algorithm is given by:

$$\boldsymbol{w(n+1) = w(n) + \beta \frac{x(n)}{\mathcal{E} + ||x(n)||^2} e(n)}$$

Here ' β' is the new step size and "||x(n)|| " is the L2 norm which reduces the sensitivity of LMS by affecting the step size in a negative gradient direction and "ε" is a small positive real value which avoids division by zero in case $x(n)$becomes zero. The NLMS algorithm converges when 'β ' obeys the inequalities.

$$\mathbf{0 < \beta < 2}$$

A block diagram illustrating the working of filtered-x NLMS algorithm for an ANC system is shown in Figure

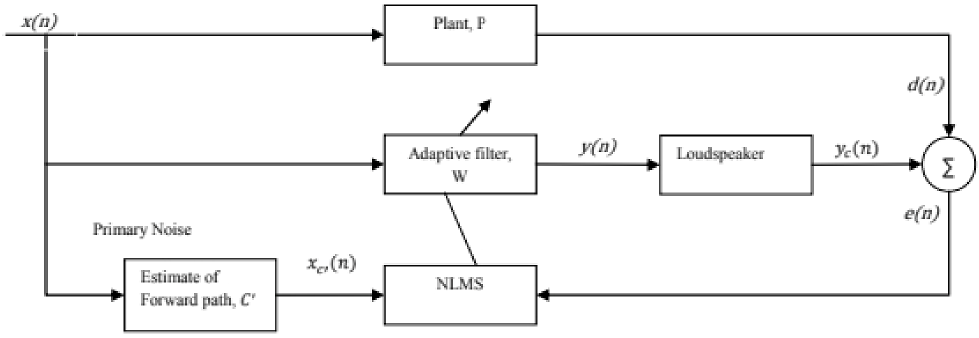

Figure 3.7: Block diagram of ANC system using NLMS

In Figure above the input signal $x(n)$ is filtered by an estimate of the forward path, which produces a filtered reference signal $x_{c'}(n)$. This filtered reference signal then becomes input for the weight adjustment algorithm NLMS. $d(n)$is the desired signal and it is propagated through the primary physical path P . By filtering $x(n)$with the adaptive FIR- filter W, the output from the adaptive filter is obtained. This output is denoted as $y(n)$. The output Y(n) is the input to the canceling loudspeaker and error signal e(n) and is then achieved in error microphone by acoustic interference $y_c(n)$ of which is loudspeaker output Y(n) filtered by the forward path, with the desired signal d(n) . The algorithm update equation will be as follows:

$$\boldsymbol{w(n+1) = w(n) + \beta \frac{x_{c'}(n)}{\mathcal{E} + \| \ x_{c'}(n) \|^2} e(n)}$$

Chapter Four

Result and Discussion

4.1 Result Analysis for Analog ANC

Simulation with a simple low pass filter our first step to do the analysis is to assume H is flat, i.e. H(s) =1. Block G is a single pole low-pass filter with cut off frequency at 500Hz and DC gain of 20, i.e. G(s)=20*(2π*50 0)/(s+2π*500). We chose 500Hz is because we want audio signal has a bandwidth at least 10 KHz. The whole system is as follow:

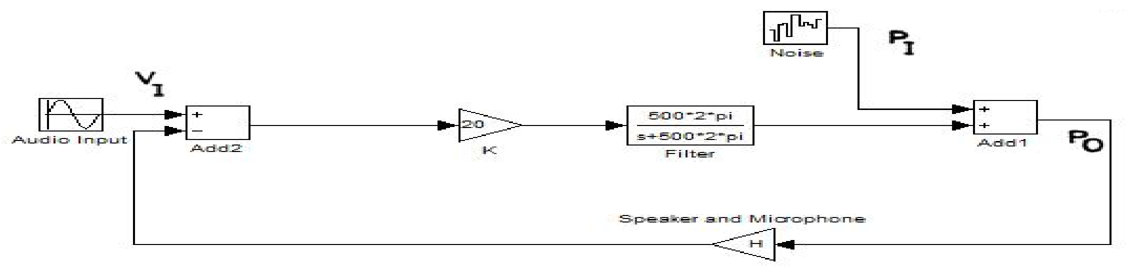

Figure 4.1: The mat lab simulation model

By plugging G and H into equation 4 and 5, we have:

P0/PI =(s+1000*π)/(s+21000*π) …………………… (6)

P0/VI = (20000*π)/ (s+21000*π) …………………. (7)

GH = (20000*π)/ (s+1000*π) ……………………… (8)

Generally, our design parameters are as table shown below.

Parameters	Design
Filter type	Single Pole Value
Cut off Frequency	500Hz
DC gain value	20
Feedback value	Flat

Table 1: Design parameter

Results

Our simulation result of the low pass filter frequency response is depicted as follows:

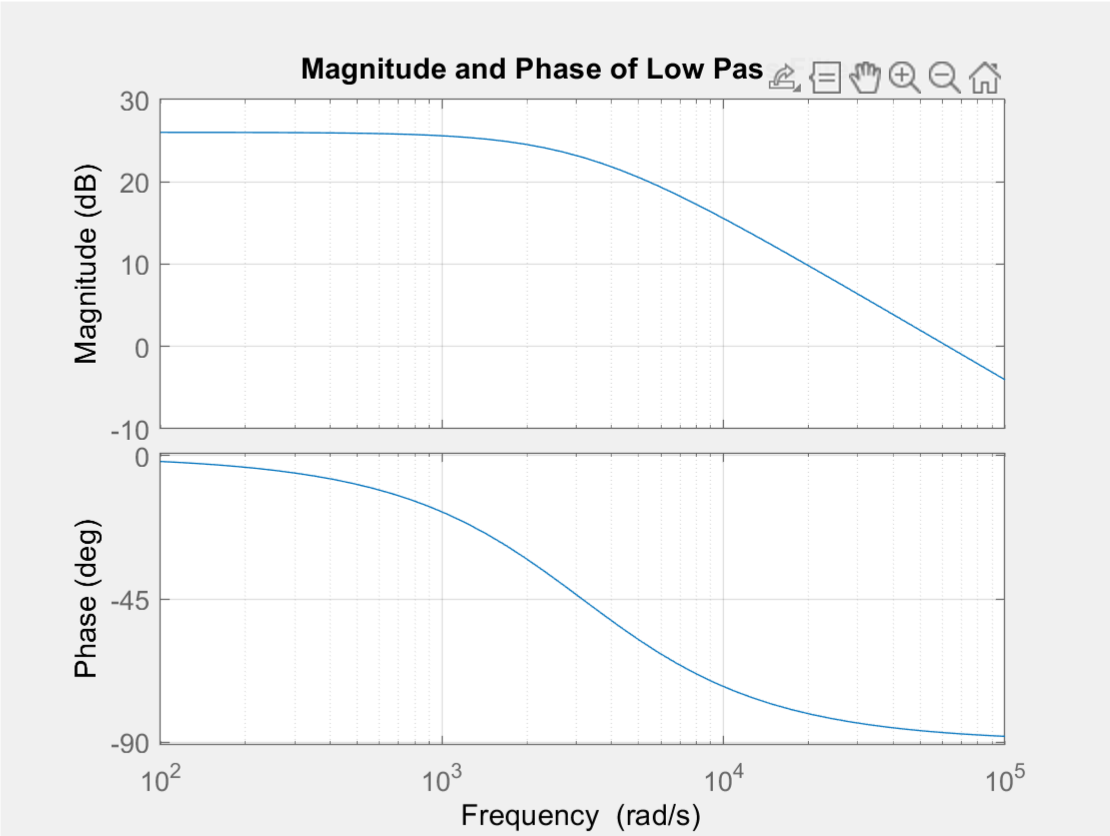

Figure 4.2: Magnitude and Phase of Low Pass Filter

The frequency response for noise attenuation (eq 6), audio signal (eq 7) and loop gain (eq 8) is as depicted below:

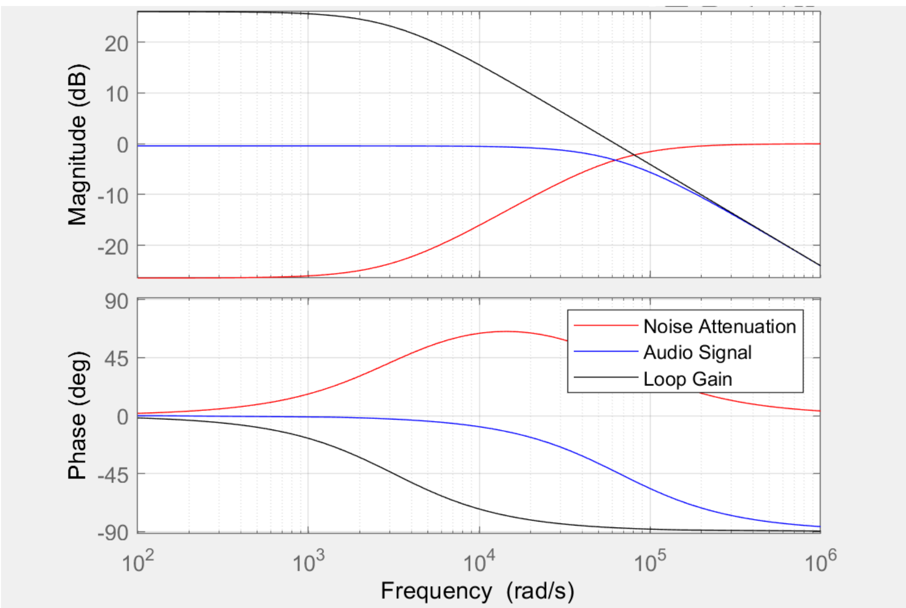

Figure 4.3: The frequency response of noise attenuation, audio signal& loop gain simulation result

We can see the system have good noise attenuation at low frequency from 100Hz. It can achieve more than 10dB attenuation before 3.4K Hz. Meanwhile, the audio signal is not reduced by the system until 10K Hz. Also, the loop phase response never goes below -90 degrees, that means the system is always stable and has at least 90 degrees phase margin.

4. 2 Result Analysis for Digital ANC

Simulation Tool used: MATLAB is used as the simulation tool for the project. The function adaptfilt.lms () and adaptfilt.nlms () is used for creating the adaptive filter working on LMS algorithm and NLMS algorithm respectively. This function gives the characteristics of the adaptive filter according to the input like step size, filter length etc. provided. The performance of the LMS algorithm has been assessed for noise cancellation. The MATLAB tool R2020a has been used for analysis. First a recorded voice signal is taken and then different noises i.e. AWGN, is used to corrupt the voice signal and the corrupted signal were filtered adaptively and results have been analyzed.

To evaluate the performance of the LMS and NLMS filter algorithms, simulations were conducted using specific parameters. Figures (4.7) and (4.8) will depict the results obtained with a step size of 8, a sample size of 2000, and a learning rate (μ) of 0.02. These chosen specifications will allow for a controlled comparison of the algorithms' behaviour during the noise reduction process.

Results:

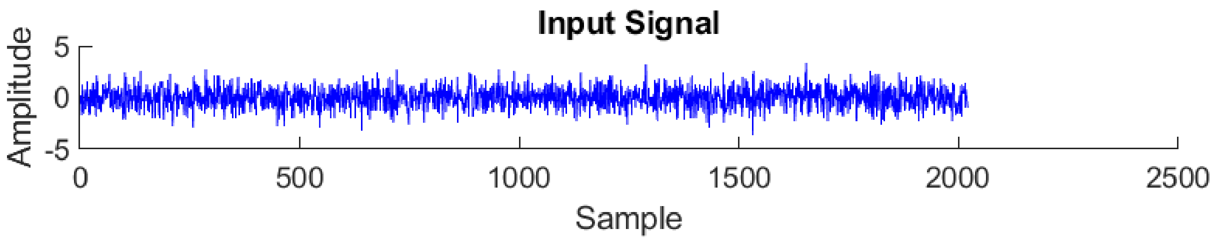

Figure 4.4: Input Signal

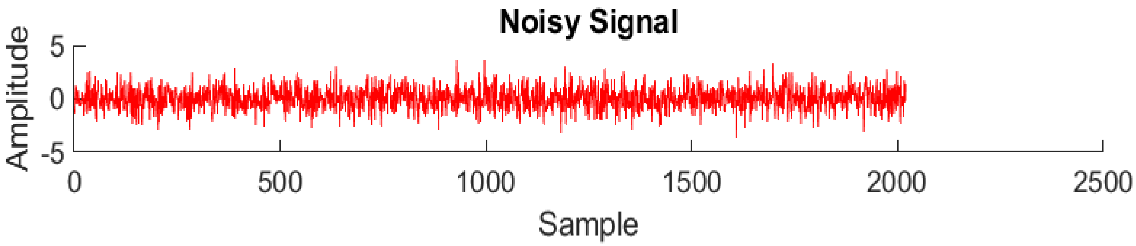

Figure 4.5: Noisy Signal

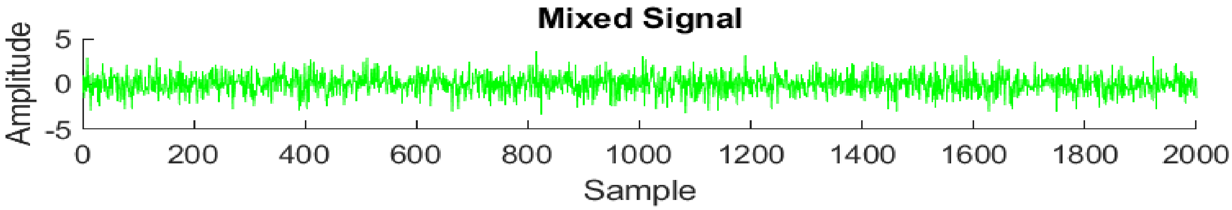

Figure 4.6: Mixed signal

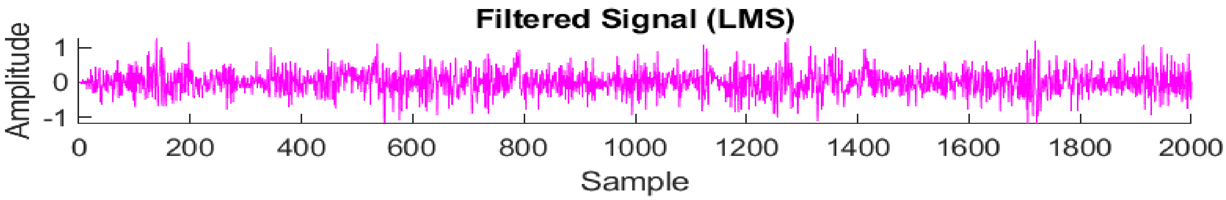

Figure 4.7: Results of filtered signal with LMS

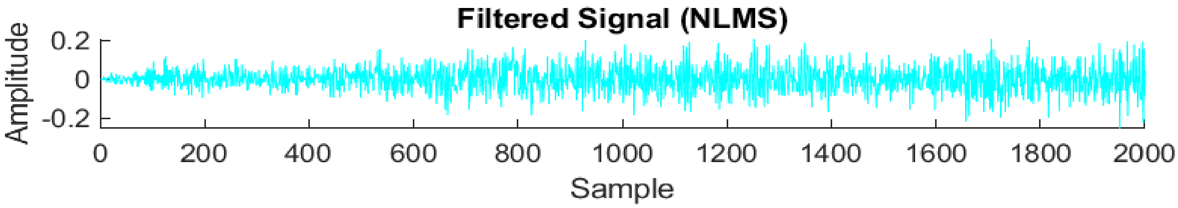

Figure 4.8: Results of filtered signal with NLMS

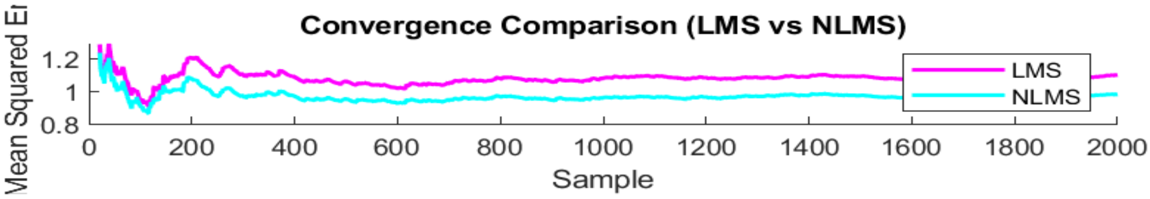

Figure 4.9: Convergence comparison between LMS VS NLMS using mean square error

4.3 Discussion

4.3.1 Analog Active Noise Cancellation

In Figure 4.3, the frequency responses of noise attenuation, audio signal, and loop gain reveal crucial insights into the system's performance for analog noise reduction. Notably, the graph illustrates a pronounced noise attenuation effect, particularly evident at lower frequencies starting from 100Hz. This substantial attenuation, exceeding 10dB before 3.4kHz, underscores the system's effectiveness in suppressing undesirable noise components, crucial for enhancing signal clarity and quality.

Simultaneously, the analysis of the audio signal's response unveils a significant aspect of the system's operation. Remarkably, the system preserves the integrity of the audio signal, with minimal impact observed until higher frequencies around 10kHz. This characteristic underscores the system's ability to maintain the fidelity of the audio output, ensuring that desired audio quality is retained throughout the noise reduction process. Collectively, the findings from Figure 4.3 reaffirm the system's capability in analog noise reduction, with robust noise attenuation, preservation of audio signal integrity, and inherent stability, positioning it as a reliable solution for various noise reduction applications.

4.3.2 Digital Active Noise Cancellation

In this section, we analyse the performance of the LMS (Least Mean Squares) and NLMS (Normalized Least Mean Squares) algorithms for noise cancellation based on the simulation results. The comparison is conducted across various metrics including convergence behaviour, filter coefficient evolution, and frequency response.

4.3.2.1 **Convergence Behaviour**

The convergence behaviour of the LMS and NLMS algorithms is a critical aspect of their performance. From the figure (4.9), we observe that both algorithms exhibit convergence towards lower MSE values as the number of samples increases. However, NLMS demonstrates faster convergence, particularly in scenarios with rapidly varying input power. This behaviour can be attributed to NLMS's ability to adaptively adjust the step size based on the input signal power, leading to more efficient convergence compared to the fixed step size used in LMS.

4.3.2.2 **Signal to Noise Ratio (SNR)**

The analysis of noise reduction performance focused on comparing the Signal-to-Noise Ratio (SNR) improvement achieved by the LMS and NLMS algorithms. Here, the results clearly demonstrate the NLMS algorithm's edge in noise cancellation. This advantage is evident in the significantly higher SNR improvement observed for the NLMS filtered signal compared to the LMS output. This superiority stems from the adaptive nature of the NLMS algorithm. Unlike LMS, which relies on a fixed step size for updating filter coefficients, NLMS dynamically adjusts these coefficients based on the characteristics of the incoming signal. This dynamic adaptation allows NLMS to more effectively converge on the optimal filter configuration for a given noise scenario. Consequently, the NLMS filtered signal exhibits a lower level of residual noise, signifying a more successful removal of unwanted noise components. The below table shows SNR of both algorithms:

Algorithm	Filter Order	Step Size	SNR(before filtering) db	SNR(After filtering) db	SNR Improvement db	MSE
LMS	8	0.02	10.0123	-0.4368	-10.4491	1.0787
NLMS	8	0.02	10.0123	-0.0700	-10.0823	0.9971

Table 4.1 : SNR and MSE output for LMS and NLMS

Table (4.1), presents the performance metrics of LMS and NLMS algorithms for noise cancellation. Both algorithms were evaluated with a filter order of 8 and a step size of 0.02. Before filtering, both LMS and NLMS exhibited similar Signal-to-Noise Ratio (SNR) values, approximately 10.0123 dB. However, after filtering, NLMS outperformed LMS, achieving an SNR of -0.0700 dB compared to LMS' -0.4368 dB. This resulted in a greater SNR improvement for NLMS (-10.0823 dB) compared to LMS (-10.4491 dB). Additionally, the Mean Squared Error (MSE) values for both algorithms were comparable, indicating similar levels of noise reduction effectiveness.

In conclusion, the NLMS algorithm, with its adaptive nature and dynamic coefficient adjustment, demonstrates superior noise cancellation capabilities compared to LMS, as evidenced by the higher SNR improvement achieved in the filtered signals.

Chapter Five

Conclusion and Recommendations for Future Work

5.1 Conclusion

In conclusion, our project focused on designing and comparing active noise cancellation (ANC) systems using both analog and digital approaches. We began by exploring analog methods, employing feedback systems and low-pass filters to mitigate unwanted noise. Through MATLAB Simulink simulations, we visualized the performance of these analog filters, highlighting their efficacy in reducing background noise.

Transitioning to the digital realm, we implemented algorithms such as the Least Mean Squares (LMS) and Normalized Least Mean Squares (NLMS) to further refine our noise cancellation capabilities. Through comparative analysis, we found that the NLMS algorithm demonstrated superior noise reduction performance.

Additionally, we have demonstrated the difference in their noise-cancelling performance and their noise-controlling performance using the simulated graphs that we obtained from the MATLAB simulation. Moreover, we utilized MATLAB to simulate the code for the two algorithms: the LMS and the NLMS algorithm.

Comparing analog and digital approaches, we identified distinct strengths and weaknesses. Analog systems offer simplicity and stability, while digital methods provide greater precision and adaptability.

In conclusion, the primary goal of using active noise-cancelling headphones is to diminish undesirable noise. They serve as an optimal solution for safeguarding the operator's hearing from noise-induced damage and facilitating clear communication between operators and customers. The design of these headphones incorporates a straightforward negative feedback system. There are two configurations for active noise-cancelling headphones: the open-loop system, which positions the microphone outside the ear cup, and the closed-loop system, which places the microphone inside the ear cup. Both setups aim to effectively reduce background noise and enhance auditory protection. Overall, our project underscores the importance of active noise cancellation in enhancing auditory experiences and improving productivity in various environments.

5.2 Recommendations for Future Work

Moving forward, we recommend several enhancements to our headset design to further improve its performance and capabilities. Firstly, incorporating more precise microphones and Analog-to-Digital Converters (ADCs), as well as Digital-to-Analog Converters (DACs), will elevate the accuracy of noise measurements, thus enhancing the effectiveness of our Active Noise Cancellation (ANC) system.

Additionally, integrating all design components into a dedicated embedded system will minimize feedback interference, ensuring smoother operation. We suggest implementing the project in assembly language to optimize computational resources and enhance overall performance.

To validate the stability and robustness of our ANC headset, we propose conducting extensive tests with real-world noise scenarios, including music sources and expanding to two channels. This will allow us to assess its efficacy across diverse environments accurately.

Furthermore, we recommend implementing and comparing four variations of adaptive algorithms, including FxLMS, FuLMS, Feedback ANC, and Hybrid ANC. By exploring these algorithms, we anticipate gaining valuable insights into their respective strengths and weaknesses, thus informing future iterations of our ANC headset.

In conclusion, while our current ANC headset effectively addresses artificial and real-world noise within a frequency range of 100 to 800 Hz and incorporates music sources, we believe these recommendations will significantly enhance its performance and user experience. By continuing to innovate and refine our design, we aim to deliver a truly immersive and distraction-free listening experience for users.

Reference

[1] Chu Moy, Active Noise Reduction Headphone Systems, 2001,http://headwize.com/tech/anr_tech.htm

[2] Bose, Amar, "Head phoning," US Patent No. 4,455,675, 6/19/84.

[3] Shaw, E. A. G. and Thiessen, G. J. ―Acoustics of circumaural earphones,‖ Journal of the Acoustical Society of America, 1962, 34(9), 1233-1246.

[4] S. Elliot and P. Nelson, ―Active noise control,‖ IEEE Signal Processing Magazine, vol. 10, pp.12–35, 1993.

[5] Signal Processing/Steepest Descent Algorithm - Wikibooks, open books for an open world

https://en.wikibooks.org/wiki/Signal_Processing/Steepest_Descent_Algorithm June 1, 2018

[6] https://en.wikipedia.org/wiki/Nyquist_stability_criterion June 1, 2018

[7] P. Lueg, ―Process of silencing sound oscillations.‖ U.S. Patent No. 2043416, 1936

[8] Conover, W. (1956). "Acoustic Noise Reduction from Power Distribution Transformers." Generic Electric Company Technical Report.

[9] Franklin, G.F., Powell, J.D., and Emami-Naeini, A., Feedback Control of Dynamic Systems, 3rd ed., Addison-Wesley, Reading, MA, 1994.

[10] EECS 452, Winter 2008, Active Noise Cancellation Project, Kuang - Hung liu, Liang-Chieh Chen, Timothy Ma, GowthamBellala, Kifung Chu

[11] Eriksson, Larry J., "Active Sound Attenuation System with On-Line Adaptive Feedback

Cancellation," US Patent No. 4,677,677, 6/30/87.

[12] Morgan, D.R., & Widrow, B. (1980). "Adaptive Noise Cancelling: Principles and Applications." Proceedings of the IEEE, 63(12), 1692-1716.

Appendix

MATLAB Code for LMS and NLMS Noise Cancellation Comparison GUI:

```
function noise_cancellation_comparison_gui
    % Create GUI
    fig = uifigure('Name', 'Noise Cancellation Comparison',
'Position', [100, 100, 1600, 800]);

    % Create UI components
    btn_run_comparison = uibutton(fig, 'Text', 'Run
Comparison', 'Position', [50, 700, 150, 30],
'ButtonPushedFcn', @run_comparison);
    ax_input_signal = uiaxes(fig, 'Position', [50, 550, 500,
100]);
    ax_mixed_signal = uiaxes(fig, 'Position', [50, 425, 500,
100]);
    ax_filtered_lms = uiaxes(fig, 'Position', [50, 300, 500,
100]);
    ax_filtered_nlms = uiaxes(fig, 'Position', [50, 175, 500,
100]);
    ax_lms_mse = uiaxes(fig, 'Position', [600, 550, 300,
200]);
    ax_nlms_mse = uiaxes(fig, 'Position', [600, 325, 300,
200]);
    tbl_performance = uitable(fig, 'Position', [600, 100, 500,
200]);

    % Callback function to run noise cancellation comparison
    function run_comparison(~, ~)
        % Parameters
        N = 2000; % Number of samples
        M = 8; % Filter order
        mu = 0.02; % Step size
        sigma2 = 0.1; % Input noise power

        % Generate input signal and noise
        x = randn(N+M, 1); % Input signal
        v = sqrt(sigma2) * randn(N+M, 1); % Gaussian noise
        x_noisy = x + v;

        % Initialize filter coefficients
        w_lms = zeros(M, 1); % LMS filter
        w_nlms = zeros(M, 1); % NLMS filter

        % Filter signal using LMS and NLMS algorithms
        y_lms = zeros(N, 1);
        y_nlms = zeros(N, 1);

        % Track convergence for LMS and NLMS
        lms_mse = zeros(N, 1);
```

```
        nlms_mse = zeros(N, 1);

        % Initialize error vectors
        e_lms = zeros(N, 1);
        e_nlms = zeros(N, 1);

        for n = 1:N
            % Mix noise with input signal
            x_mixed = x_noisy(n:n+M-1);

            % LMS
            y_lms(n) = w_lms' * x_mixed;
            e_lms(n) = x(n+M) - y_lms(n);
            w_lms = w_lms + mu * e_lms(n) * x_mixed;
            lms_mse(n) = mean(e_lms(1:n).^2);

            % NLMS
            y_nlms(n) = w_nlms' * x_mixed;
            e_nlms(n) = x(n+M) - y_nlms(n);
            w_nlms = w_nlms + (mu / (x_mixed' * x_mixed)) *
e_nlms(n) * x_mixed;
            nlms_mse(n) = mean(e_nlms(1:n).^2);
        end

        % Compute SNR before filtering
        snr_before = 10 * log10(var(x) / var(v));

        % Compute SNR after filtering
        snr_after_lms = 10 * log10(var(x) / var(e_lms));
        snr_after_nlms = 10 * log10(var(x) / var(e_nlms));

        % Compute SNR improvement
        snr_improvement_lms = snr_after_lms - snr_before;
        snr_improvement_nlms = snr_after_nlms - snr_before;

        % Plot input signal
        plot(ax_input_signal, 1:N+M, x, 'b');
        xlabel(ax_input_signal, 'Sample');
        ylabel(ax_input_signal, 'Amplitude');
        title(ax_input_signal, 'Input Signal');

        % Plot mixed signal
        plot(ax_mixed_signal, 1:N, x_noisy(M+1:end), 'g');
        xlabel(ax_mixed_signal, 'Sample');
        ylabel(ax_mixed_signal, 'Amplitude');
        title(ax_mixed_signal, 'Mixed Signal');

        % Plot filtered signal (LMS)
        plot(ax_filtered_lms, 1:N, y_lms, 'm');
        xlabel(ax_filtered_lms, 'Sample');
        ylabel(ax_filtered_lms, 'Amplitude');
```

```
        title(ax_filtered_lms, 'Filtered Signal (LMS)');

        % Plot filtered signal (NLMS)
        plot(ax_filtered_nlms, 1:N, y_nlms, 'c');
        xlabel(ax_filtered_nlms, 'Sample');
        ylabel(ax_filtered_nlms, 'Amplitude');
        title(ax_filtered_nlms, 'Filtered Signal (NLMS)');

        % Plot mean squared error (MSE) for LMS
        plot(ax_lms_mse, 1:N, lms_mse, 'm', 'LineWidth', 1.5);
        xlabel(ax_lms_mse, 'Sample');
        ylabel(ax_lms_mse, 'Mean Squared Error');
        title(ax_lms_mse, 'Mean Squared Error (LMS)');

        % Plot mean squared error (MSE) for NLMS
        plot(ax_nlms_mse, 1:N, nlms_mse, 'c', 'LineWidth',
1.5);
        xlabel(ax_nlms_mse, 'Sample');
        ylabel(ax_nlms_mse, 'Mean Squared Error');
        title(ax_nlms_mse, 'Mean Squared Error (NLMS)');

        % Populate performance table
        data = {'LMS', snr_before, snr_after_lms,
snr_improvement_lms, lms_mse(end);...
                'NLMS', snr_before, snr_after_nlms,
snr_improvement_nlms, nlms_mse(end)};
        columnNames = {'Algorithm', 'SNR Before Filtering
(dB)', 'SNR After Filtering (dB)', 'SNR Improvement (dB)',
'Final MSE'};
        tbl_performance.ColumnName = columnNames;
        tbl_performance.RowName = [];
        tbl_performance.Data = data;
    end
end
```

MATLAB Code for Filter Transfer Functions and Frequency Response.

```
% Clear all variables, close all figures, and clear command
window
clear all;
close all;
clc;

% Define transfer functions
G = tf([20000*pi], [1 1000*pi]); % Low pass filter
N = tf([1 1000*pi], [1 21000*pi]); % Noise attenuation
A = tf([0 20000*pi], [1 21000*pi]); % Audio signal
L = tf([0 20000*pi], [1 1000*pi]); % Loop gain

% Plot magnitude and phase of the low pass filter
figure;
bode(G);
```

```
grid on;
title('Magnitude and Phase of Low Pass Filter');

% Plot frequency response of noise attenuation, audio signal,
and loop gain
figure;
bode(N, 'r', A, 'b', L, 'k');
grid on;
legend('Noise Attenuation', 'Audio Signal', 'Loop Gain');
title('Frequency Response Comparison (Noise Attenuation,
Audio Signal, and Loop Gain)');
```