# ASHESI UNIVERSITY

# DESIGN AND IMPLEMENTATION OF A LOW-COST AND PORTABLE NOISE MASKING SYSTEM

**CAPSTONE PROJECT**

B.Sc. Electrical and Electronics Engineering

**Michael Osei**

**2021**

ASHESI UNIVERSITY

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## CAPSTONE PROJECT

Capstone Project submitted to the Department of Engineering, Ashesi University in partial fulfilment of the requirements for the award of Bachelor of Science degree in Electrical and Electronics Engineering.

**Michael Osei**

**2021**

# DECLARATION

I hereby declare that this capstone is the result of my own original work and that no part of it has been presented for another degree in this university or elsewhere.

Candidate's Signature:

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Candidate's Name:

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Date:

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I hereby declare that preparation and presentation of this capstone were supervised in accordance with the guidelines on supervision of capstone laid down by Ashesi University.

Supervisor's Signature:

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Supervisor's Name:

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Date:

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# Acknowledgements

Firstly, I give the Almighty God thanks for the gift of life, His eternal grace, mercy, favor, and strength throughout this project work.

Much appreciation and deep gratitude go to my supervisor, Mr. Francis Gatsi, whose supervision, encouragement, and academic advice helped me undertake this project.

Finally, my appreciation goes to the entire Engineering Department, the Ashesi Financial Aid team, as well as my parents for their hard work, resources, support, and prayers during my four years of stay at Ashesi University. This journey would have been challenging without their help.

# Abstract

Noise pollution has been given more awareness because of its adverse impacts on human health and comfort. The portable low-frequency noise reduction device built in this project can effectively solve low-frequency noise pollution problems using active noise cancellation. Active Noise Cancellation (ANC) technology has been a tremendous challenge in modern science. The main reason for ANC's slow adoption is its higher initial cost compared to passive methods. Thus, considering this, this project seeks to build a cost-effective and attractive ANC system to overcome this economic hurdle using essential hardware resources such as microphones, amplifiers, and speakers. ANC technology uses the destructive interference principle of waves to cancel out a signal by superposition with its inverted signal. This capstone report presents the possible results of implementing a cost-effective ANC system to facilitate an enclosure's sound level. Simulations in Proteus and Multisim and hardware testing results reveal that the noise can be decreased by $6 - 17$ dB for low frequencies. However, the system attenuated sounds of low frequencies ($\leq$ 1 kHz) much better than high frequencies. The noise source position and movement, intensity, and frequency of the noise signal influence these sound effects.

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# 1 INTRODUCTION

## 1.1 Background

The modern development in architecture, innovation, technology, and acoustics has subjected the average human to a rising and wide range of artificial sounds. From the neighbor's music to the noise from traffic, moving vehicles, and industrial equipment to a normal human conversation in an office, these disturbances can negatively influence an individual's comfort, psychological wellness, or communication effectiveness; hence, noise masking or cancellation systems are of vital significance. Audio processing applications have been evident in artificial reverberation, sound mixing, and editing, and noise cancellation.

Research reveals that people with prolonged exposure to noise are not too productive at the workplace. In a survey that examined some workers, 57% testified that "Background noise caused a major deterioration in their ability to concentrate. The longer they remained in the office, the more disrupted they were by the noise" [1]. Some people are not distracted by background sound or music while working; others prefer quiet environments to be more productive. Noise absorption materials are used in some workplaces to combat noise pollution. Some carpets mask sound due to their extra cushioning. Furniture is likewise deliberately positioned at specific vantage points to prevent sound from propagating from one end to another to ensure the privacy of an individual's conversation. Albeit these helpful strategies, the bulky material is not easily movable. A portion of these materials can be pretty costly. A standard technology is noise-canceling headphones whose speaker compartment employs both passive and active technologies.

## 1.2 Problem Definition

Noise is any unwanted signal resulting from a distorted signal that is being communicated [2]. Acoustic noise problems become progressively evident as more modern industrial equipment such as engines, fans, and transformers are innovated. Subsequently, sound masking is used in many applications, including speech privacy, the most common one.

Sound masking systems are unique audio systems designed to create spatially uniform sound levels [3]. From open offices, call centers with high activity levels, to research areas where distractions need to be limited, masking lessens employee distractions. In closed offices, additional masking can offer confidentiality more economically than developments in structural elements like noise-absorbing walls and doors. In the cities, it is used to conceal the traffic sound of nearby airports or freeways. Masking is employed in hospitals and medical facilities to ensure patients are not distracted by external sounds from traffic, helicopters, and the sounds of nursing staff, health equipment, and other patients. Masking ensures the confidentiality of patient information, as required by the Health Insurance Portability and Accountability Act (HIPAA) [4].

For in secure facilities, such as military contractor meeting rooms, corporate boardrooms, and government offices, it is employed to restrict purposeful eavesdropping attempts with advanced devices. In residential settings, such as homes and apartments, sound masking blocks out noise from aircraft fly-overs, street sounds, barking dogs, as well as boisterous neighbors. This creates peaceful sleeping and domestic tranquility. For households, masking devices are found in specialty catalogs or at internet sites [3]. In schools, students sometimes find it challenging to locate quiet places to study. Intermittent or constant noise can cause distractions while learning. The noise is generated either periodically or randomly.

This project is designed to build a portable and cost-effective noise masking device that blocks stray noise to create a suitable noise-free environment for the above issues.

## 1.3 Motivation

Masking is merely the concealing or suppressing of something where that thing is not changed but hidden. In this manner, the addition of an unnoticeable background sound eases distractions and the transparency of human speech or spatially constrained noise. Sound masking is explicitly tuned ambient background sound that targets the same frequency as human speech, reducing its intelligibility [3]. When people wear masks, they disguise the face of the wearer. Deodorants only mask body odors but do not eliminate them.

Similarly, one-way windows shroud the people on the opposite side, so they are invisible. Sound can mask different sounds to hide them. Envision, you are in an obscured room, and someone is flicking a torch on and off. The light is recognizable and disturbing. Now imagine, the lights are on. The same torch is being flicked on and off, but without notice, because it has been masked [5]. For each situation, the objective is to hide something that exists. At the point when unwanted sounds are concealed, the listener is less interrupted. It results in more satisfaction with one's environment and makes the individual more productive. Sound masking can deliver speech confidentiality in conditions where otherwise it cannot be acquired without unreasonable cost [6].

Active noise control systems [7]–[9] cancel the undesired noise by generating a secondary noise of equal amplitude and opposite phase of the primary noise to cancel both noises based on the principle of superposition. The ANC system is exceptionally effective for attenuating low-frequency noises where passive techniques are costly and ineffective. Most practical ANC systems can track variations of unknown noise characteristics and environment automatically. Whether or not done structurally with sound attenuating materials or active background level control, the essential function of masking decreases the scope of fluctuations in sound level [3]. It is by far the most affordable tool for providing privacy. It will profit students, workers, and generally individuals who wish for a serene noise-free working environment.

## 1.4 Aims and Objectives

As of late, a broad range of noise masking technologies has surfaced in the form of earbuds, noise-absorbing walls, loudspeakers, etc. In this capstone report, the different sound masking techniques are examined, and a model of a sound masking system is designed.

This project aims to:

- Review the various noise cancellation techniques: active noise cancellation, passive noise cancellation, analog or digital noise cancellation, and real-time noise cancellation.
- Develop an improved low-cost and portable electronic device that masks stray sound from the outside environment using active noise cancellation technique.

Students with a demand for steady, quiet study spaces will benefit significantly from this device. It will be cheap and affordable—people who want absolute silence during the night time or when resting will use this product.

# 2 LITERATURE REVIEW

A review of various existing ANC noise cancellation projects helped design and build a more simplified and effective working device. This chapter seeks to elaborate on how gaps identified with existing solutions advised my design considerations.

## 2.1 Introduction and Problem Statement

Noise pollution has become one of the significant environmental issues, and it becomes more and more evident through research and the industrial revolution. Regarding this issue, many have been involved in the design and implementation of noise cancellation technology. Two main methods have been employed for noise cancellation: active cancellation and passive cancellation [2]. While active noise cancellation techniques have proven to be very effective for attenuating low-frequency noises, passive methods tend to be relatively bulky, expensive, and ineffective. With passive noise cancellation, soundproofing mechanisms such as enclosures and silencers are employed for high noise attenuation over a broad frequency but are generally ineffective at low frequencies (below 1kHz) [9], [10]. ANC systems use the superposition principle and destructive interference to attenuate low-frequency noise where passive methods are either inefficient or costly. Therefore, Active Noise Control has garnered attention and is rapidly advancing because it allows enhancement in noise control for real-world applications, often with future benefits in size, flexibility, aesthetics, and cost.

According to the reviews, significant further research and development are still being made into active noise cancellation to improve commercialization system performance. However, the concept of active noise cancellation has existed since the beginning of the 20th century. In 1936, a German patent was issued to Paul Lueg for the basic ideas of active noise cancellation or control; he was the first to realize the possibility of attenuating background noise picked up with a microphone by superimposing a phase flipped wave [9]. In the 1950s, Harry Olson and Everet May successfully demonstrated Lueg's concept in rooms, ducts, and headsets [8]. With the invention of integrated circuits, opamp circuits, and small-scale microphones, ANC systems, such as headphones, became increasingly probable and versatile. Therefore, hands-on implementation of noise-canceling headphones provides a vital insight into the structure of passive and active noise cancellation.

## 2.2 Approach to Solving the Problem

Various experiments for reducing noise ranging from tonal to broadband have been performed using portable Analog ANC systems. In one of their papers, Ratnam et al. demonstrated the concept of active noise cancellation based on destructive interference. They proposed that for optimal noise cancellation to be achieved, the noise of equal amplitude but entirely out of phase is combined with the undesired noise to cancel out through destructive interference. The noise is sensed by a microphone and sent to an Analog ANC system which converts the signal received to anti-noise. This anti-noise combines with the initial noise inside the acoustic duct through destructive interference, leading to reduced noise. It is impractical to complete because of the components' physical requirements and the delay in propagating the sound from one end of the acoustic duct to another [11]. Figure 2.1 shows the details of the Analog ANC system.

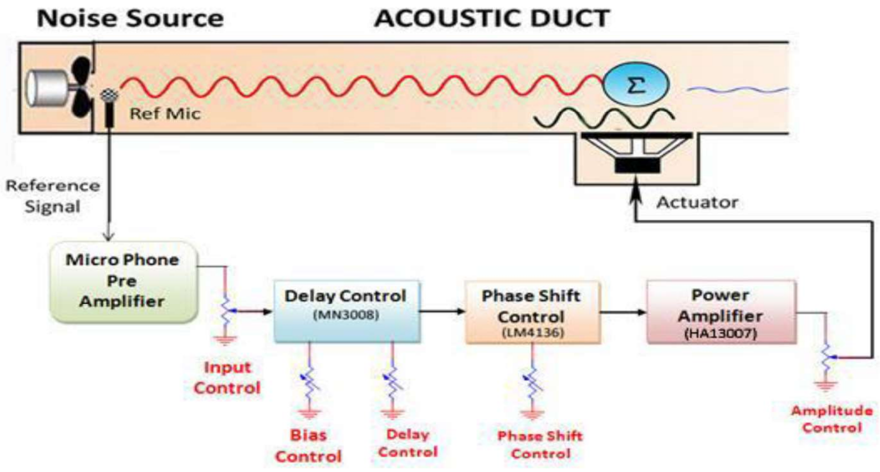


*Figure 2.1: Components of Typical Analog ANC System*

In one journal, Hassan and Howlader proved via an experiment that some ANC models could reduce the noise level in a seminar room below -50 dB through destructive interference [2]. Further work sought to introduce some add-on functions to increase ANC systems' overall value and make them more attractive and practical [12]. A large percentage of ANC techniques employed signal processors with algorithms, thereby making the noise-canceling device expensive. Some methods, such as the **Time Difference of Arrival (**TDOA) algorithm used to detect the noise source direction performed poorly on uncorrelated noise signals. Liu et al. have shown in Figure 2.2 that when the microphone closest to the source is chosen as a reference, the system's performance is improved by **5dB** [13]. It informed my decision to mount the microphones on the headphones to capture background noise signals effectively.

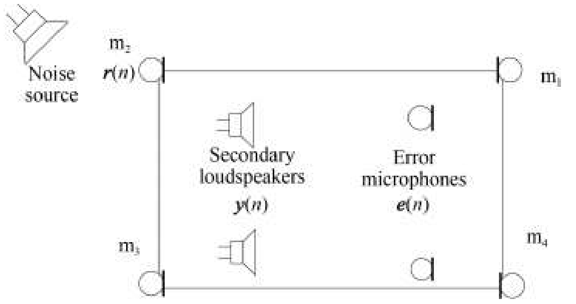


*Figure 2.2:* Integrated ANC system (m1, m2, m3 & m4 - microphone array)

Kannan et al. adopted a new approach to active noise cancellation. Instead of generating an anti-noise signal by applying the FXLMS algorithm to the reference signal, they separated the reference signal into a periodic signal and a random signal before canceling each one [14]. Though this approach turned out to be more effective than a basic adaptive system, it had a higher complexity and did not work well for low signal-to-noise ratio (SNR) values. It is, therefore, not surprising that Zahir et al. implemented a technique that combined the noise and anti-noise signals and showed the error calculation in the time domain. A microphone captured the real-time noise replacing the audio file approach [15]. The method can be used to cancel noise in real-time and was considered for my design.

## 2.3 Hardware Design

This section focuses on the hardware requirements of the noise-masking system. Most existing solutions made use of microphones in capturing the incoming noise signal. The researchers initially implemented a simple Analog ANC system to test three primary selected frequencies of noise in one article. As shown in Figure 2.3, this analog system consisted of a microphone module to capture the initial noise and a bandpass filter to filter the output. This signal was then given a 180 degrees' phase shift using an opamp inverter and then finally released into the initial noisy area using a speaker [15]. Figure 2.4 also shows the analog ANC circuit for reducing single tone and recorded signals. Filtering is fundamental to separate the desired signal from unwanted noise because it helps obtain the input signal's required information. This depends upon the configuration of the filter. Hence high pass filters were selected for my design.

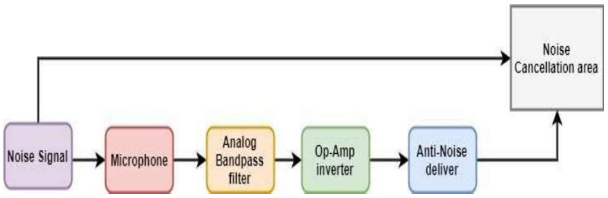


*Figure 2.3: Block diagram of the analog setup*

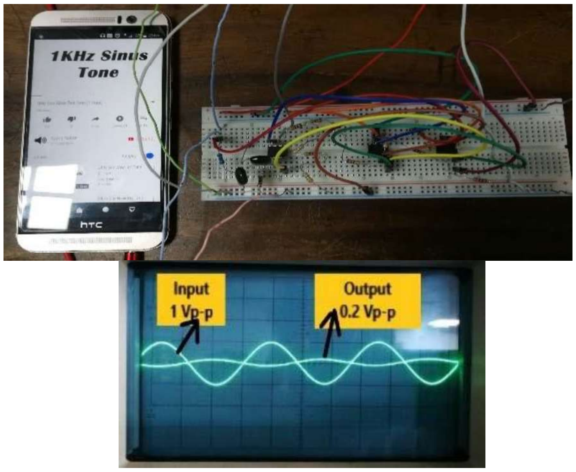


*Figure 2.4: Analog ANC circuit for reducing single tone and recorded signals*

## 2.4 Principles Considered

Most researchers focused on destructive interference in achieving a noise-canceling effect. The reviews portrayed how noise-canceling circuits operate generally. The theory behind opamp circuits, resistor networks, and filtering circuits was considered. Common circuit treatment principles, including Ohm's Law, Kirchoff's Laws, voltage and current division, and the principle of superposition, were used in conjunction with other mathematical circuit relationships for design analysis. It enlightened me on how voltage and current are shared among the circuit's various components to achieve a noise-canceling effect. Such principles were applied with basic components, operational amplifiers, and analog circuits to achieve the desired result and are attached as APPENDIX B.

## 2.5 Gaps Identified

- For Ratman's design, there is a delay between when the microphone senses the undesired noise at one end of the acoustic duct and reaches the other end leading to variations between the noise at the sending and receiving ends of the duct.
- Some models can reduce undesired noises of low frequency only, up to 100 Hz.
- Due to the computational complexity, some ANC systems were slow in detecting moving sources.
- Some existing devices employ two parallel ANC systems and a separator, which makes it complex to operate.
- Some experiments were not carried out on real individuals in noisy environments and cannot predict how the system will perform in real life.

# 3 SYSTEM REQUIREMENTS AND DESIGN CONSIDERATION

## 3.1 Introduction

This chapter discusses the proposed solution's architecture, the modules' costs, and how the individual components come together to form a noise cancellation system.

## 3.2 Project Requirements

The system features of the product or project must meet the user's requirements to one's satisfaction.

### *3.2.1 User Requirements*

- The packaging must be attractive and comfortable to handle.
- The user should be able to operate the device through manual attenuation of the feed-in signal.

### *3.2.2 System Requirements*

For the final deliverable to fulfill the user's expectations, the product must meet the following requirements.

- Microphones mounted on both ends of headphones to pick up ambient sound from surroundings.
- Must have potentiometers for volume control.
- Make use of amplifiers and filters in different configurations to cancel out noise entirely.
- Must be energy efficient with minimal power consumption.
- Attenuate sounds of very low frequencies ($\leq$ 1 kHz).
- Make use of headphones speakers to observe noise cancellation.

## 3.3 Design Specifications

The design specifications define the system design and features that ensure the final deliverable meets the user requirements. The requirements and ratings of the components selected are appended as APPENDIX A.

### *3.3.1 Block diagram of the proposed solution*

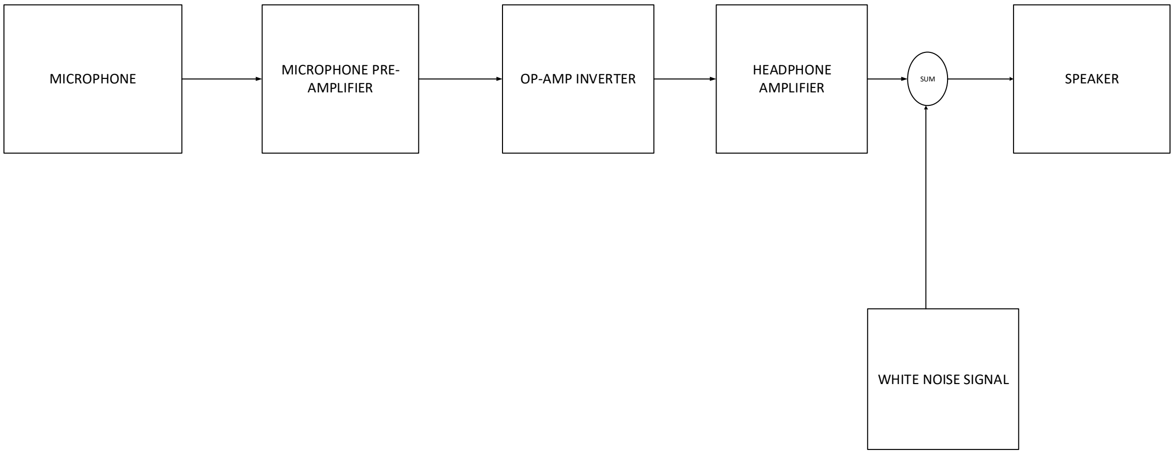


*Figure 3.1: Block Diagram of Proposed Design*

### *3.3.2 Design Components*

This section defines the hardware components used for the project. These components include:

- Microphone
- Pre-amplifier
- Phase Inverter
- Headphone Amplifier
- Speaker
- White Noise Signal Generator

## 3.4 Review of Sensors and Modules

This section briefly describes the individual components selected in designing the noise-canceling device, their purpose and the criteria for their selection.

### *3.4.1 Microphone*

A microphone is a device that captures audio and converts sound waves into electrical signals. The signal can be amplified as an analog signal or converted to a digital signal [16].

*Specifications of Microphone*

- It must be affordable and easily accessible.
- It must be miniature and effortlessly installed.
- It must be suitable for use in electronic systems.

*Types of Microphones Considered*

- **ELECTRET MICROPHONE**

An electrostatic capacitor-based microphone, also known as a *condenser microphone,* uses a permanently charged material, which requires an external power source to operate. It is primarily used in electronic systems such as telephones, headsets, and smartphones to facilitate recording and communication [17].

- **MOVING COIL MICROPHONE**

It is the most widely used type of dynamic microphones. It is often ideal for use on stage and several field recorders because of its durability, simple design and does not require an external power supply [17].

- **RIBBON MICROPHONE**

It is also a type of dynamic microphone which is known for its high fidelity. It is usually not considered dynamic because of how it operates. It is equally sensitive to sounds approaching from its front and rear, hence bidirectional. Such microphones are traditionally used in the studio and radio industry [17].

The three microphones above are compared in the Pugh Chart Matrix below.

*Table 3.1: Pugh Chart Showing Comparison of Microphones*

| Criteria | Dynamic (Baseline) | Weight | Moving-Coil | Electret | Ribbon |
|---|---|---|---|---|---|
| Cost | 0 | 4 | +1 | 0 | -1 |
| Size | 0 | 4 | 0 | +1 | -1 |
| Durability | 0 | 3 | +1 | 0 | -1 |
| Performance | 0 | 4 | 0 | +1 | -1 |
| Ease of Use | 0 | 3 | +1 | +1 | 0 |
| Sensitivity | 0 | 2 | 0 | +1 | -1 |
| Transient Response | 0 | 2 | -1 | +1 | 0 |
| **Total** | 0 | 22 | 8 | 15 | -17 |

The electret microphone was preferred because it functions on a different principle and best fits the design specifications. It contains a thin plastic diaphragm vibrating and alters the distance between its front and back plates in response to a sound wave. The varying capacitance created by the movement of the diaphragm is transmitted as an electrical signal. This microphone also features the complete $20\ Hz - 20\ kHz$ human frequency response.

*Figure 3.2: Electret Microphone*

### *3.4.2 Pre-amplifier*

A microphone pre-amplifier accepts a very low-level signal from the microphone and amplifies it without adding extra noise [16]. It provides the microphone signal with stable gain, which prevents distortion of the noise signal. The **NE5532** operational amplifier circuit was preferred as the pre-amplifier to the microphone.

### *3.4.3 Phase Inverter*

The inverting amplifier generates the inverted or anti-noise signal required for noise cancellation after the noise signal received by the microphone is amplified. The opamp inverter must have a gain of negative one since the amplitude of the amplified noise signal must not change. Also, a **NE5532** opamp was chosen for the phase inverter.

*Features of NE5532 Op-amp*

**NE5532** opamp is appropriate for pre-amps, inverting amplifiers, professional audio equipment, control circuits, and instrumentation due to the reasons below [18].

- It shows better noise performance compared to most standard operational amplifiers.
- It compensates for gains equal to one internally.
- It has improved output drive capability and considerably higher small-signal and power bandwidths.

- It has an ample supply voltage range and is readily available.

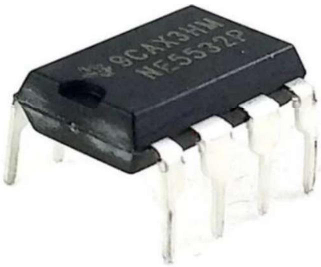


*Figure 3.3: NE5532 Opamp*

### *3.4.4 Headphone Amplifier*

It is a relatively low-powered and non-inverting amplifier that increases the low-voltage audio signal from a source device such as a smartphone or PC to an appropriate level. It can be converted into sound waves by the headphone's speakers.

The headphone amplifier to be selected for this project must have an adjustable gain of 6 dB and 17 dB, excellent performance for headphones 16-600 Ω, with extremely low THD 0.001% @ 1V RMS, a simple interface, and an analog volume control [19].

**NE5322** and **AD5322** operational amplifier circuits were considered for the headphone amplifier since both employ analog volume control. The Pugh Chart below shows a comparison between the two opamps.

*Table 3.2: Pugh Chart Comparing Specifications of Headphone Amplifier*

| Criteria | NE5532 (Baseline) | Weight | NE5322 | AD5322 |
|---|---|---|---|---|
| Availability | 0 | 4 | 0 | -1 |
| Gain Adjustability | 0 | 4 | +1 | -1 |
| Performance | 0 | 5 | +1 | 0 |
| Ease of Use | 0 | 3 | +1 | 0 |
| **Total** | 0 | 16 | 12 | -8 |

The **NE5322** is most suitable for use as a headphone amplifier. It has an internally frequency-compensated unity gain with a significant DC voltage gain of 100 dB. Also, the **NE5322** opamp has a wide bandwidth and a large output voltage [20].

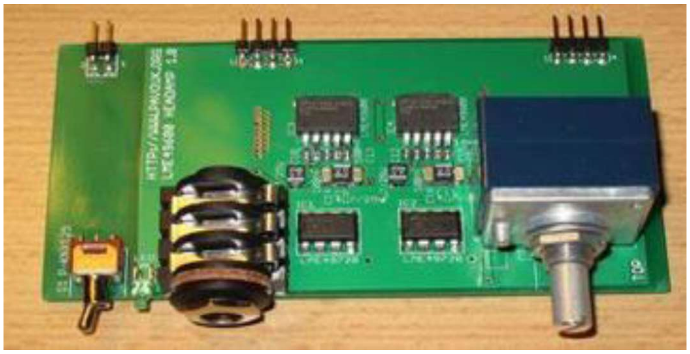


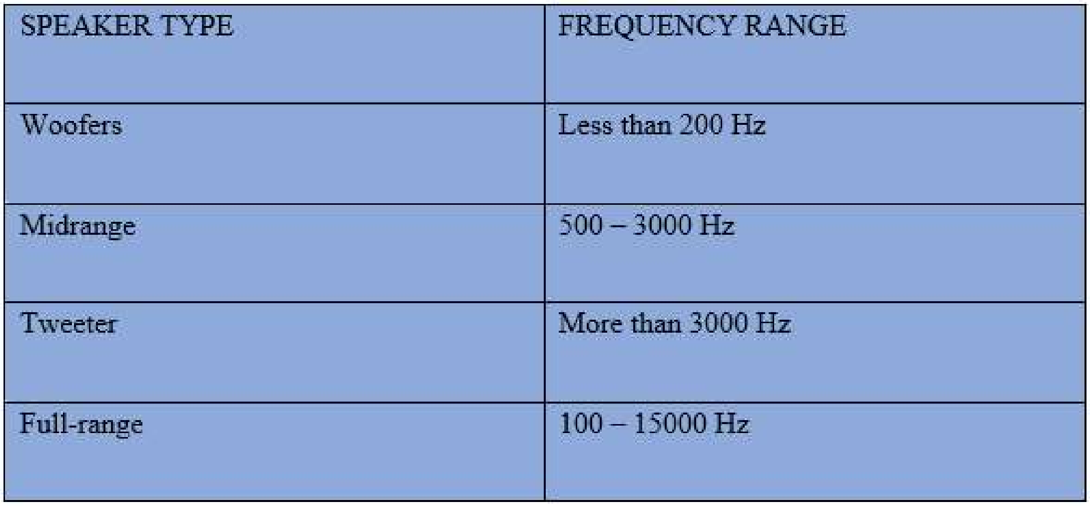
*Figure 3.4: Stereo Headphone Amplifier*

### *3.4.5 Speaker*

Microphones convert sound waves into electrical signals while speakers convert electrical signals into audible sound signals. The frequency range for which a speaker can effectively propagate sound waves is determined by its frequency response, a vital characteristic of speakers.

Speakers are classified as per their intended frequency range [21].

*Table 3.3: Comparison of Speaker Specifications*



| SPEAKER TYPE | FREQUENCY RANGE |
|---|---|
| Woofers | Less than 200 Hz |
| Midrange | 500 – 3000 Hz |
| Tweeter | More than 3000 Hz |
| Full-range | 100 – 15000 Hz |

Headphones operate similarly to speakers. While loudspeakers set all the air circulating in a room to produce audible sound, headphone speakers, on the other hand, only move the volume of air inside your ear canal. Headphone speakers are, therefore, miniature and much more intelligent. Their speaker enclosures usually have openings at the front or rear that allow air to circulate easily in and out of them to produce audible quality sound. Earbuds like headphones function similarly. Headphones are categorized into three main types [22].

- **Open-back headphones:** are open to the air at the rear as well as the front. Hence, they cannot block outside noise effectively.
- **Semi-open back headphones:** allow the passage of noise in and leak out sound through their speaker components.
- **Closed-back headphones:** are sealed around the back completely. Theoretically, no sound escapes or leaves from the outside. They are preferably used for mixing audio and recording.

For this project, closed-back noise-canceling headphones are used. They possess woofer characteristics, which make it simple to disconnect completely.

*Figure 3.5: Wired Headphone Speaker*

### *3.4.6 White Noise Generator*

In signal processing, white noise is a random signal having equal intensity at different frequencies, giving it a constant power spectral density [3]. It consists of all audible sounds with an equal intensity that masks other spontaneous sounds in an environment. It has strong masking capabilities because of its broadband spectrum. For instance, white noise is employed in the cities to help block out noise from traffic. Some sources of white noise include radio or television static, whirring fan, humming air conditioner, and hissing radiator.

For this project's purpose, the white noise will further mask the extra unmasked sounds after the cancellation stage at the headphone amplifier's output by summing the generated output sound with white noise. It enhances acoustic satisfaction by boosting the aural privacy of the space.

*Figure 3.6: White Noise Signal Generator*

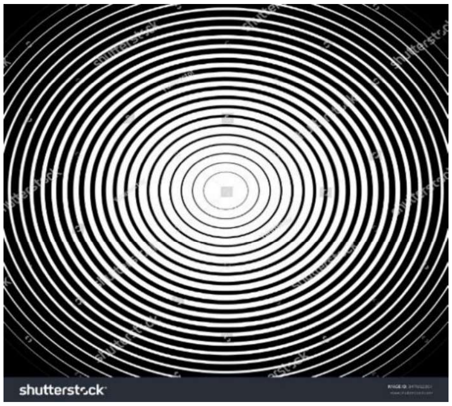


*Figure 3.7: White Noise (Oval texture)*

## 3.5 Circuit Schematic

This section shows how the individual hardware components are connected to build the noise cancellation system using Proteus and Autodesk Eagle.

### *3.5.1 Noise Canceling Schematic*

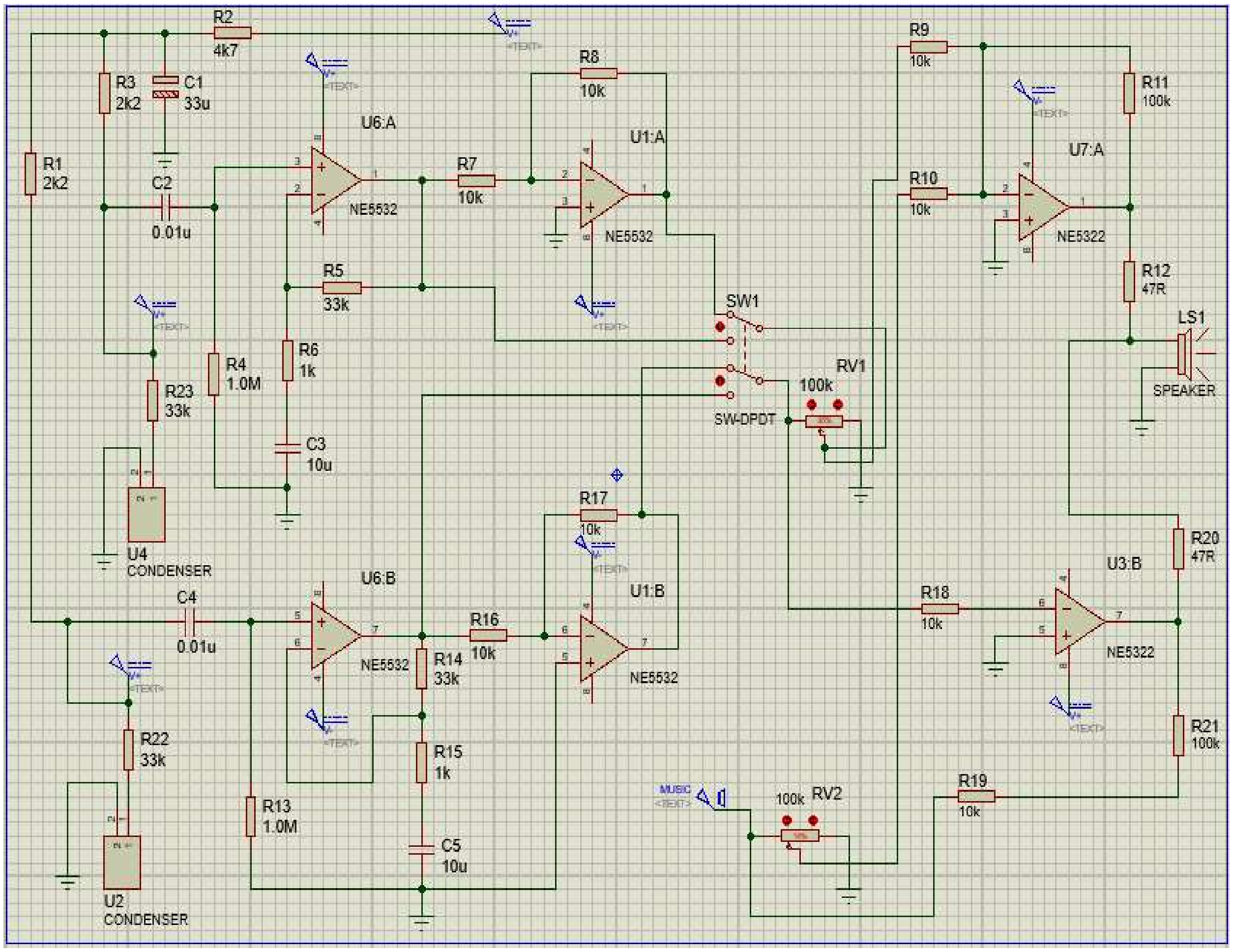


*Figure 3.8: Schematics of the Noise Cancelling Circuit*

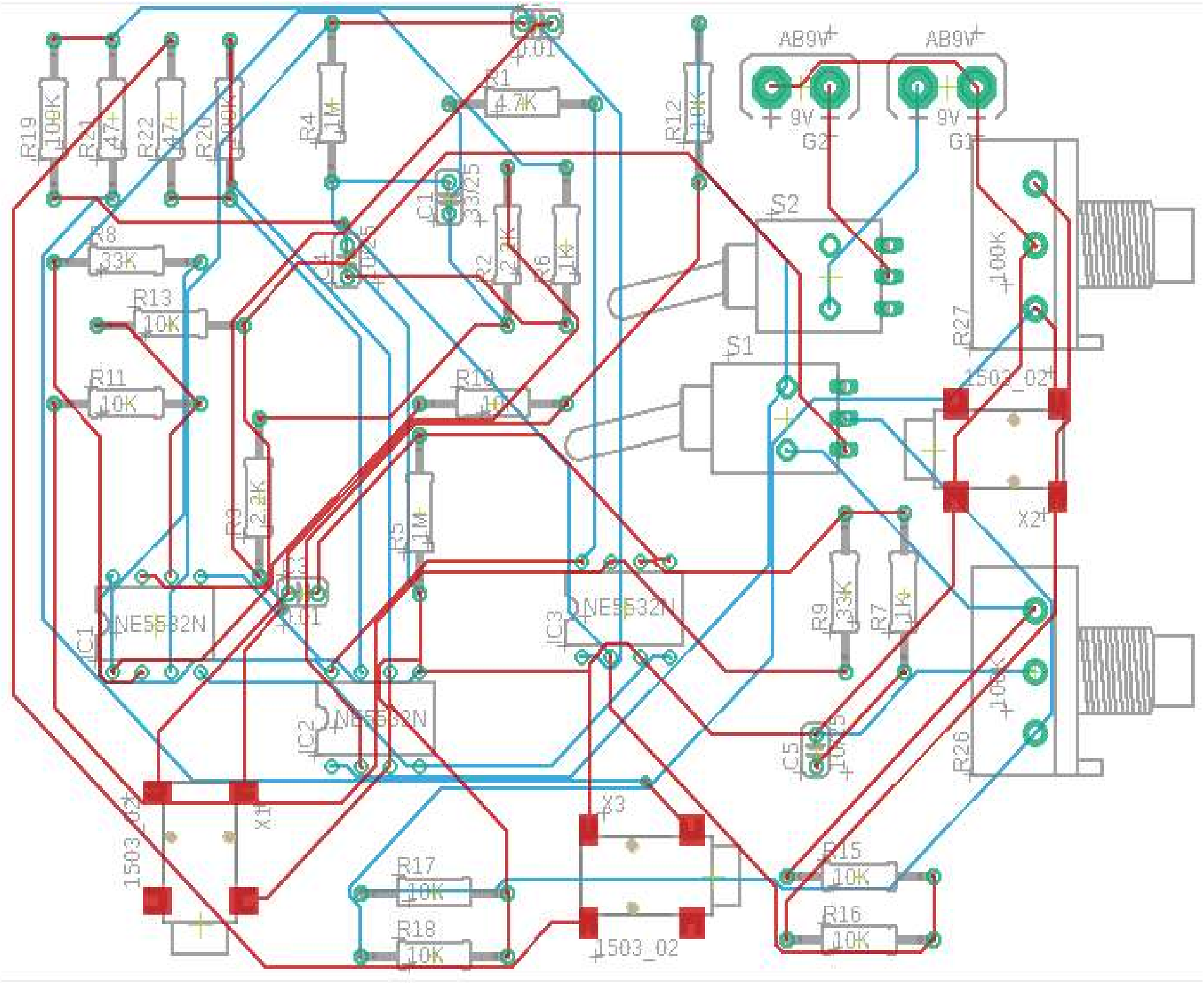


*Figure 3.9: PCB View of Noise Cancelling Circuit*

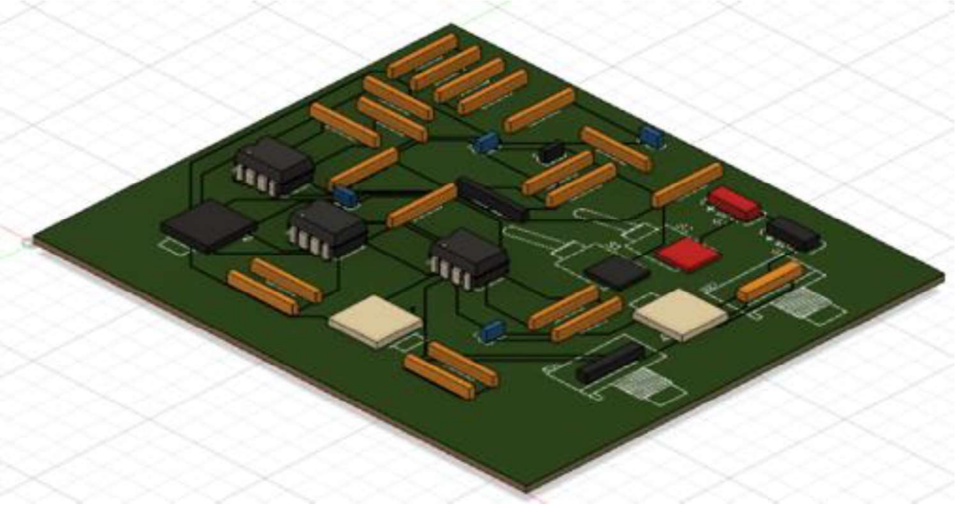


*Figure 3.10: 3D View of Noise Cancelling PCB*

### 3.5.2 White Noise Schematic

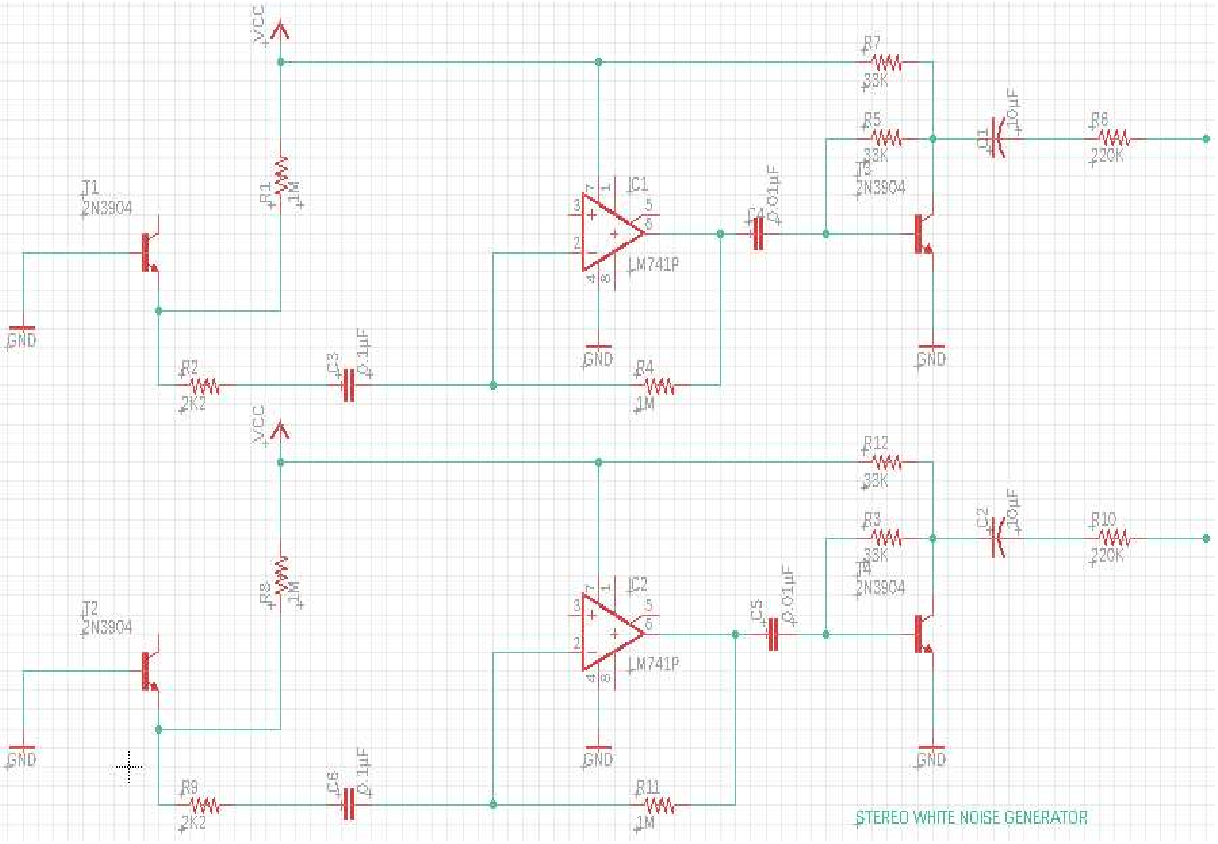


Figure 3.11: Schematics of White Noise Generator Circuit

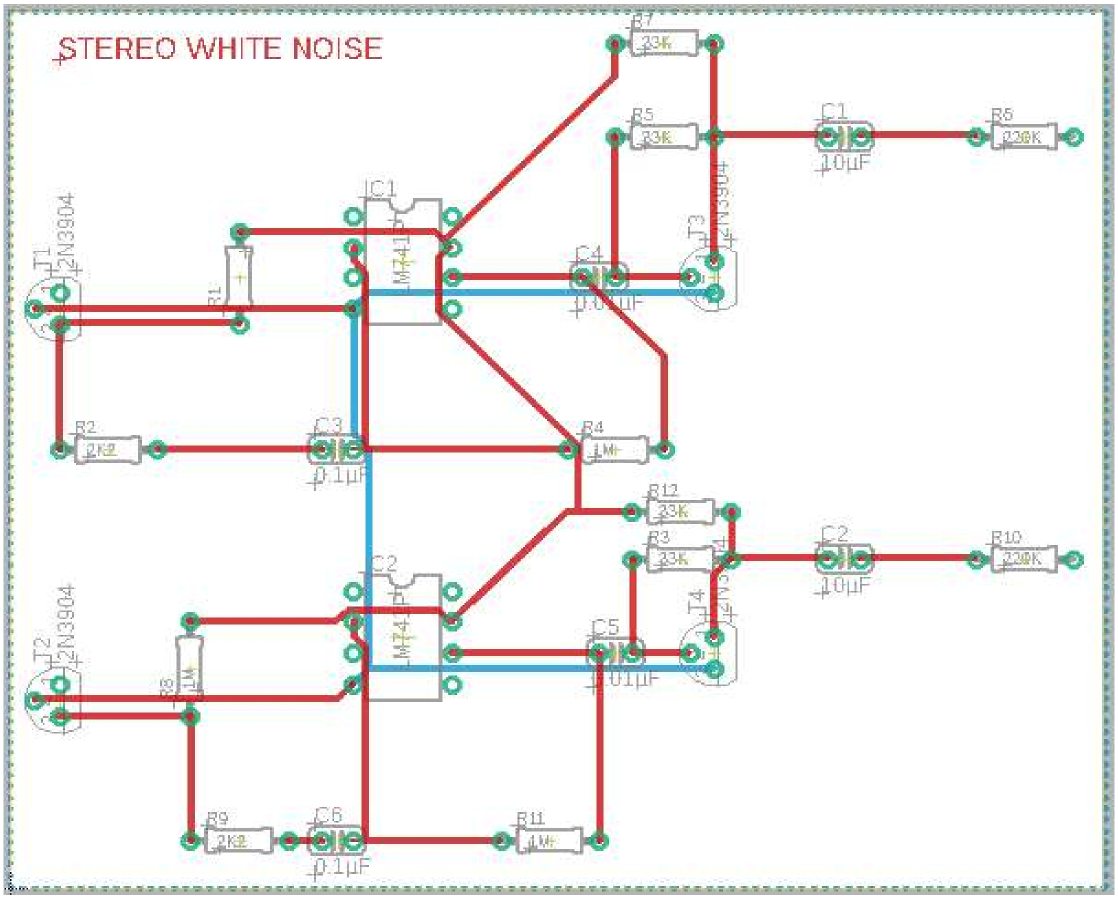


*Figure 3.12: PCB View of White Noise Generator Circuit*

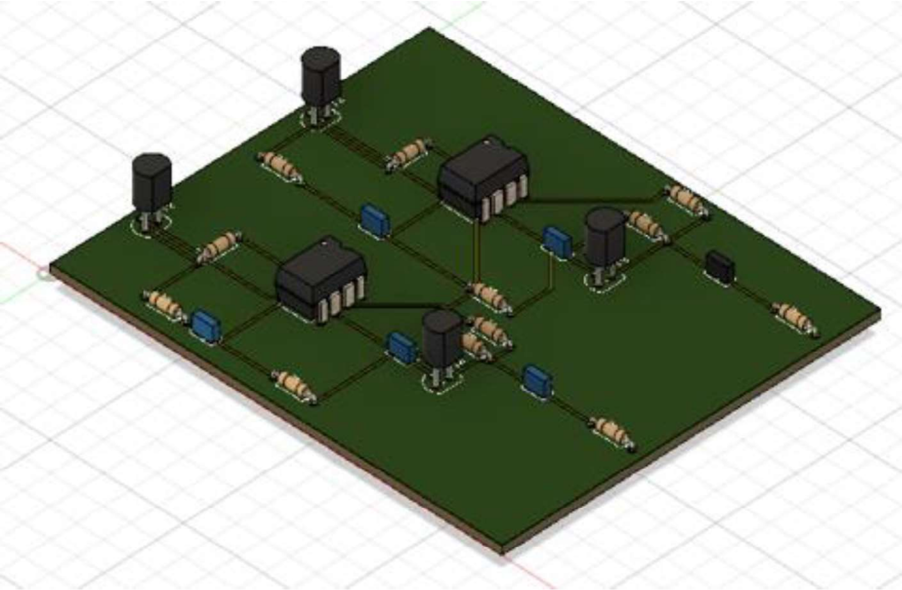


*Figure 3.13: 3D View of White Noise Generator PCB*

### 3.5.3 SolidWorks Model of Prototype

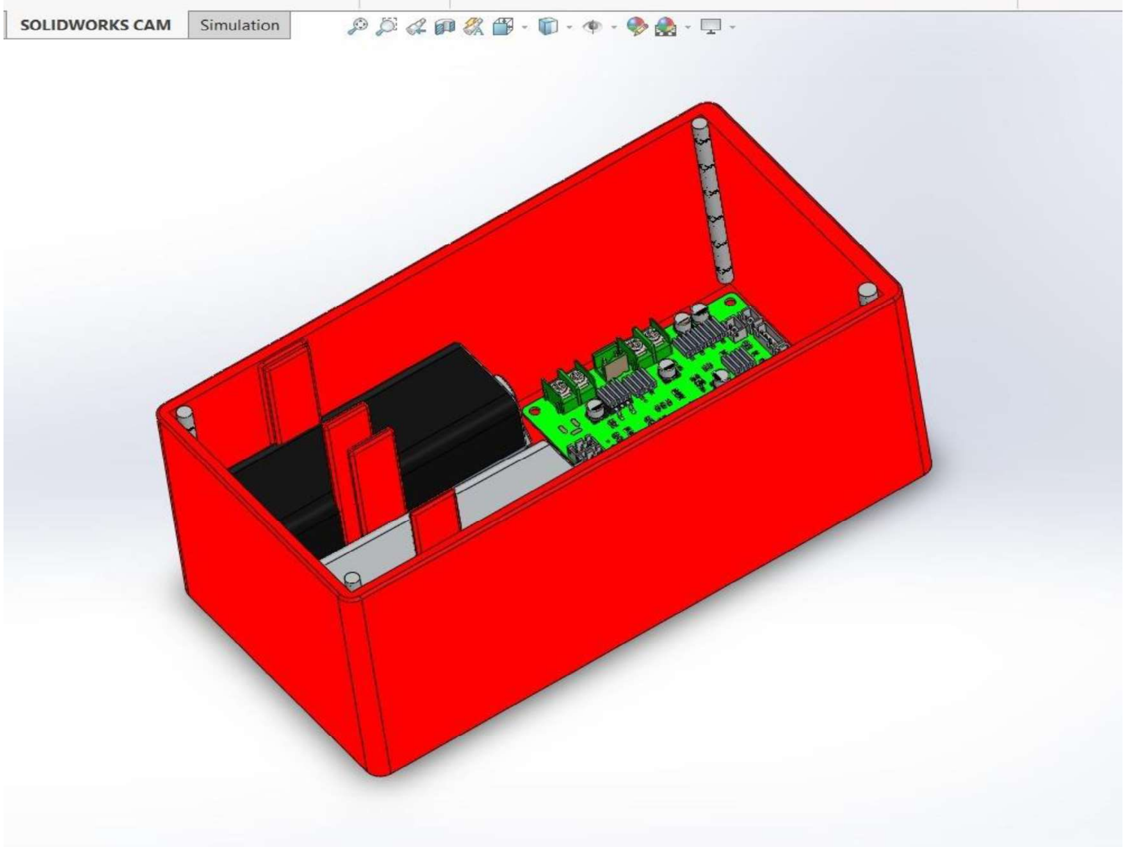


*Figure 3.14a: SolidWorks model of the prototype*

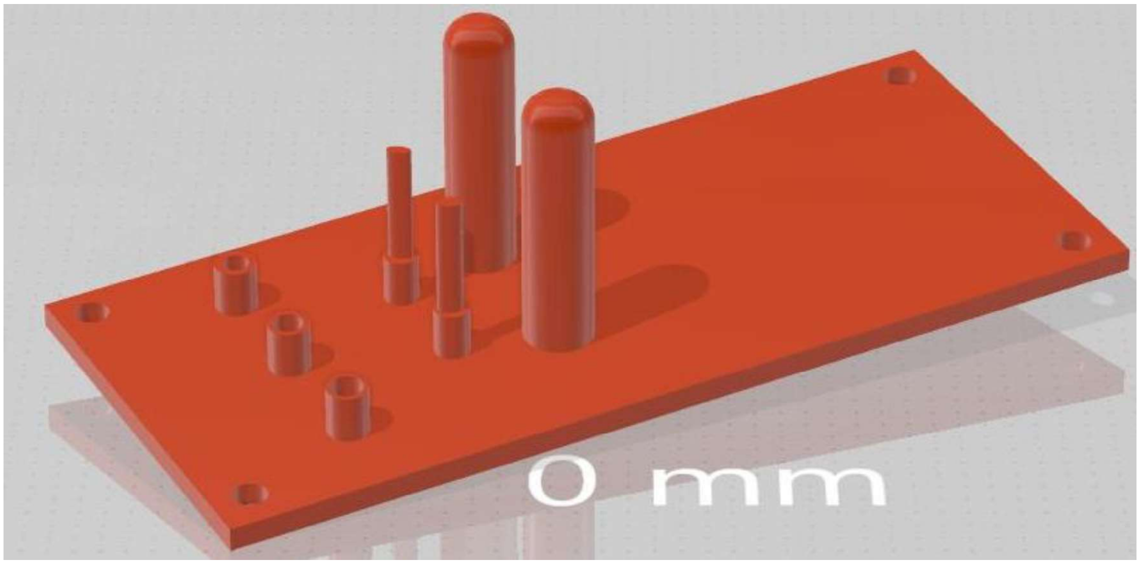


*Figure 3.14b: SolidWorks model of the prototype*

# 4 IMPLEMENTATION

## 4.1 Introduction

- A review of academic journals on noise cancellation helped analyze the strengths and shortcomings of related works and establish where improvements were necessary.
- A critical analysis of related works inspired the functional block diagram of my proposed solution. Schematic diagrams were then created from the block diagram.
- The criteria used in selecting my design components included cost, market availability, durability, and performance. The above criteria helped compare the chosen components and their alternatives.
- The circuits were simulated using Proteus and Multisim and then tested on a breadboard to see how well they functioned.
- All components were assembled on a PC board to build the noise masking device.

## 4.2 Experimental and Engineering Design Analysis

The circuit in the noise-canceling system can be divided into four distinct stages, with each containing an opamp circuit that uniquely alters the analog signal. The main calculations considered in designing this system are attached as APPENDIX C.

The first section in the circuit schematic is a non-inverting pre-amplifier, the second a phase-inverter, and the third is a summing amplifier. The last and final section consists of a white noise generator. These four main sections are interrelated. The signal is modified as it passes through each section of the circuit. The noise-canceling headphones contain two identical channels that run parallel to each other to maintain the stereo sound. Noise masking is achieved by feeding a sinusoidal analog audio signal through the various sections of the circuit.

### *4.2.1 Pre-amplifier*

The first sub-circuit is a non-inverting opamp circuit. It behaves as a pre-amplifier. The noise signal from the condenser microphone (input jack J1) is weak and must be amplified to a suitable level. The gain of this pre-amplifier is about $34\ dB$ and is determined by the ratio between resistors $R_5\ and\ R_6$ in the feedback path. The amplitude of the audio signal as it leaves this circuit is controlled by the values of $R_5\ and\ R_6$. A pair of high-pass filters are built to block any direct current (DC) from passing through the pre-amplifier. This is achieved by $C_2/R_4\ and\ C_3/R_6$. After amplification, the output signal is sent to the input of the phase-inverter.

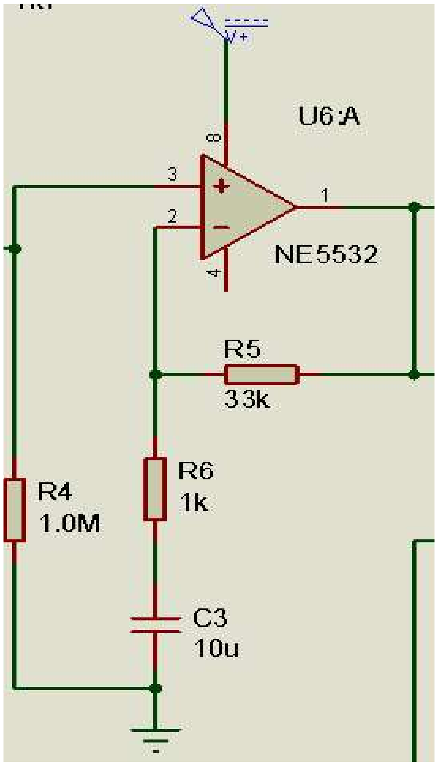


*Figure 4.1: Pre-Amplifier*

### *4.2.2 Phase-inverter*

The second sub-circuit is an inverting opamp circuit. This circuit accepts the output signal from the pre-amplifier and inverts its phase. The inversion is done by changing the polarity of the signal's voltage as it leaves the opamp. Resistors $R_8$ $and$ $R_7$ control the amplification of the signal within this circuit. This circuit has a gain of 1 since the values of $R_8$ $and$ $R_7$ are equal. It ensures that the amplitude of the input wave is maintained and reduces any distortion of the audio signal. This sub-circuit uses a DPDT toggle switch that allows switching on or off the noise canceling circuit by selecting either the phase-inverted or raw non-inverted amplified signal. It is essential for testing and debugging. The output of this circuit (desired signal) is fed directly via a $100\ K\Omega$ potentiometer to the input of the third stage. The potentiometer helps adjust the gain of the output signal at that point, allows setting the amplitude of the microphone signal, and permits the user to fine-tune the noise canceling. For ideal noise cancelation, the amplitude of the phase-inverted signal must be equal to that of the unwanted noise signal. The output signal from the potentiometer is fed into the headphone amplifier.

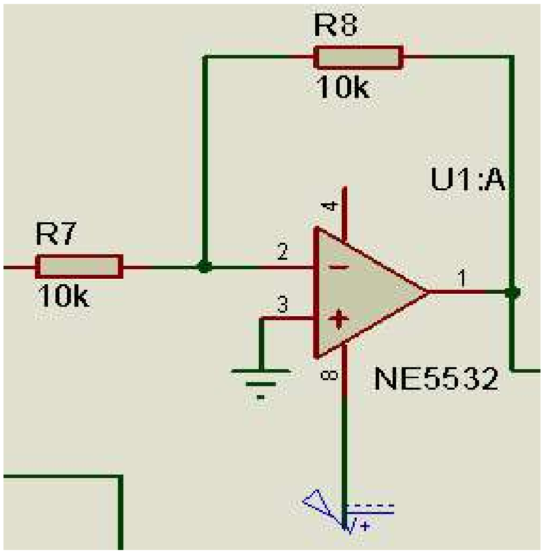


*Figure 4.2: Inverting Op-Amp Circuit*

### 4.2.3 *Headphone Amplifier*

The third sub-circuit is an inverting amplifier which has a summing feature by the inclusion of $R_{10}$. It combines the inverted noise signal with an auxiliary input (music). The auxiliary input supplied by input jack J2 passes through a $100\ K\Omega$ potentiometer, which acts as volume control and regulates the music's amplitude through attenuation. The ratio between resistors $R_{11}\ and\ R_9$ determines the gain of this summing amplifier. The summing amplifier combines the music audio signal with the inverted noise signal and plays both through the same speaker. The output of this amplifier is mixed with the white noise signal, the last part of the noise-canceling circuit before the signal is sent to the headphone speaker.

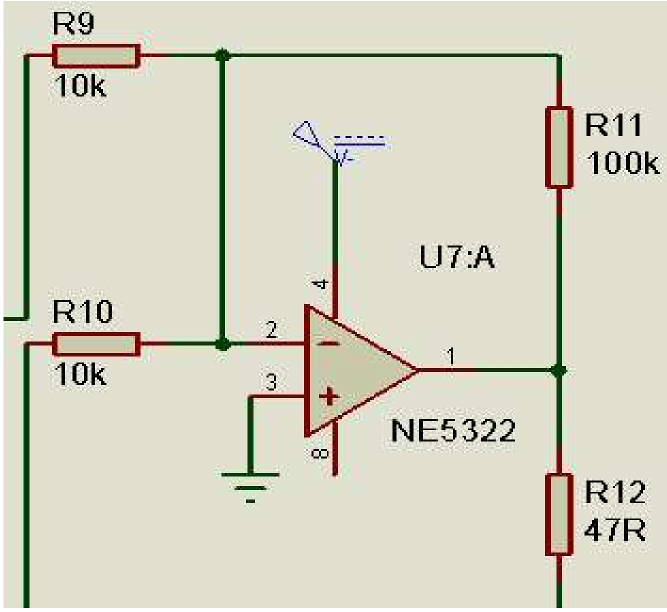


*Figure 4.3: Summing Amplifier*

### 4.2.4 *White Noise Generator*

The last sub-circuit is a white noise generator. Due to the broadband spectrum of white noise, it has strong masking capabilities. The junction base-emitter of the 2N3904 transistor generates the white noise. The LM741 amplifier amplifies the white noise. Filtering of the white noise is achieved by $C_1/R_6\ and\ C_2/R_{10}$. The white noise signal generated is mixed with the headphone amplifier's output to mask the extra or non-canceled noise after the initial cancellation process.

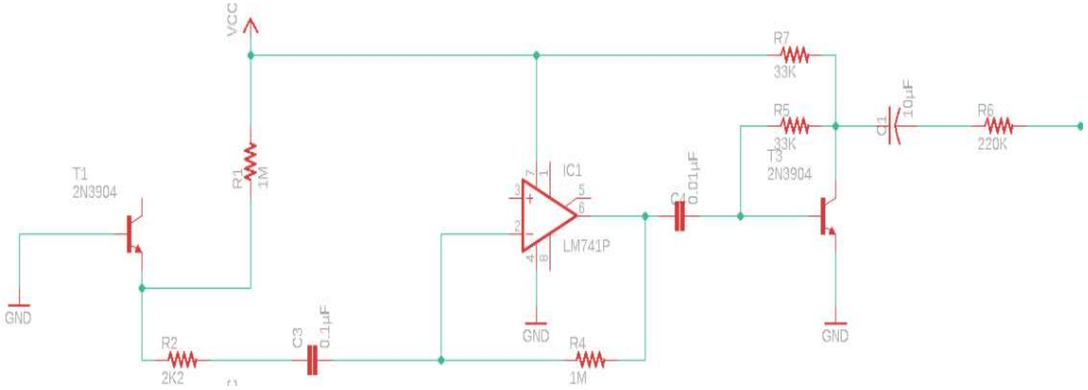


*Figure 4.4: White Noise*

# 5 RESULTS AND ANALYSIS

This chapter focuses on all test cases, results, and functionality. The noise-canceling system was first simulated, and the physical device was built afterward. For the hardware testing, I observed the system's effects on two sound sources in a room: a person talking (human voice) with sound (noise) from a rotating ceiling fan in the room. After several tests and simulation, the working principle of the noise cancellation device is outlined below:

- The electret microphones pick up ambient sound from the external environment.
- The microphones' source signal is inverted to produce a similar signal with a 180° phase shift.
- While one side of the headphone speakers produces the source sound, the other side produces the inverted sound.
- The source, inverted signals, and music are fed into the human ear concurrently through the headphone speakers to monitor the cancellation.
- After the initial cancellation process, a white noise signal is employed to mask the extra non-canceled noise.

The project was practically tested by scoping the output of each noise-canceling sub-circuit with an oscilloscope. It was observed the headphones eliminated any low-frequency noise in the room. Though the headphones receive the noise waves, music, and inverted noise waves simultaneously, destructive interference mutually cancels out the noise and its inverted waves, allowing only the music to be played through the headphone speakers. The generated white noise masked any extra ambient noise. A speech signal has a frequency range of 300 – 3400 Hz; however, the audible frequency of the human ear ranges from 20 Hz to 20 kHz [23]. The ANC device canceled entirely any speech signal with a frequency less than or equal to 1 kHz.

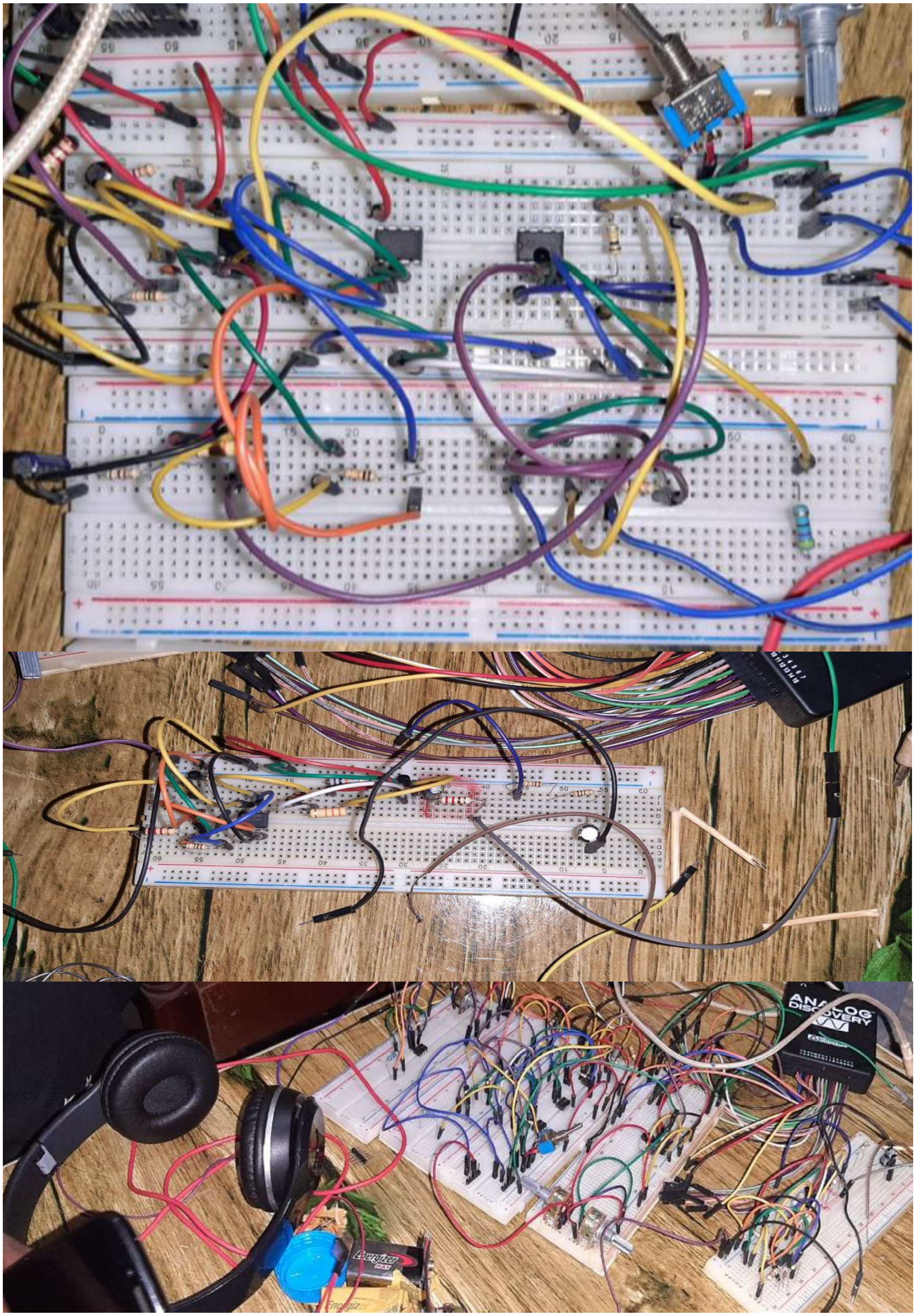


*Figure 5.1: Snapshot of the Testing of the Noise Cancellation and Masking System*

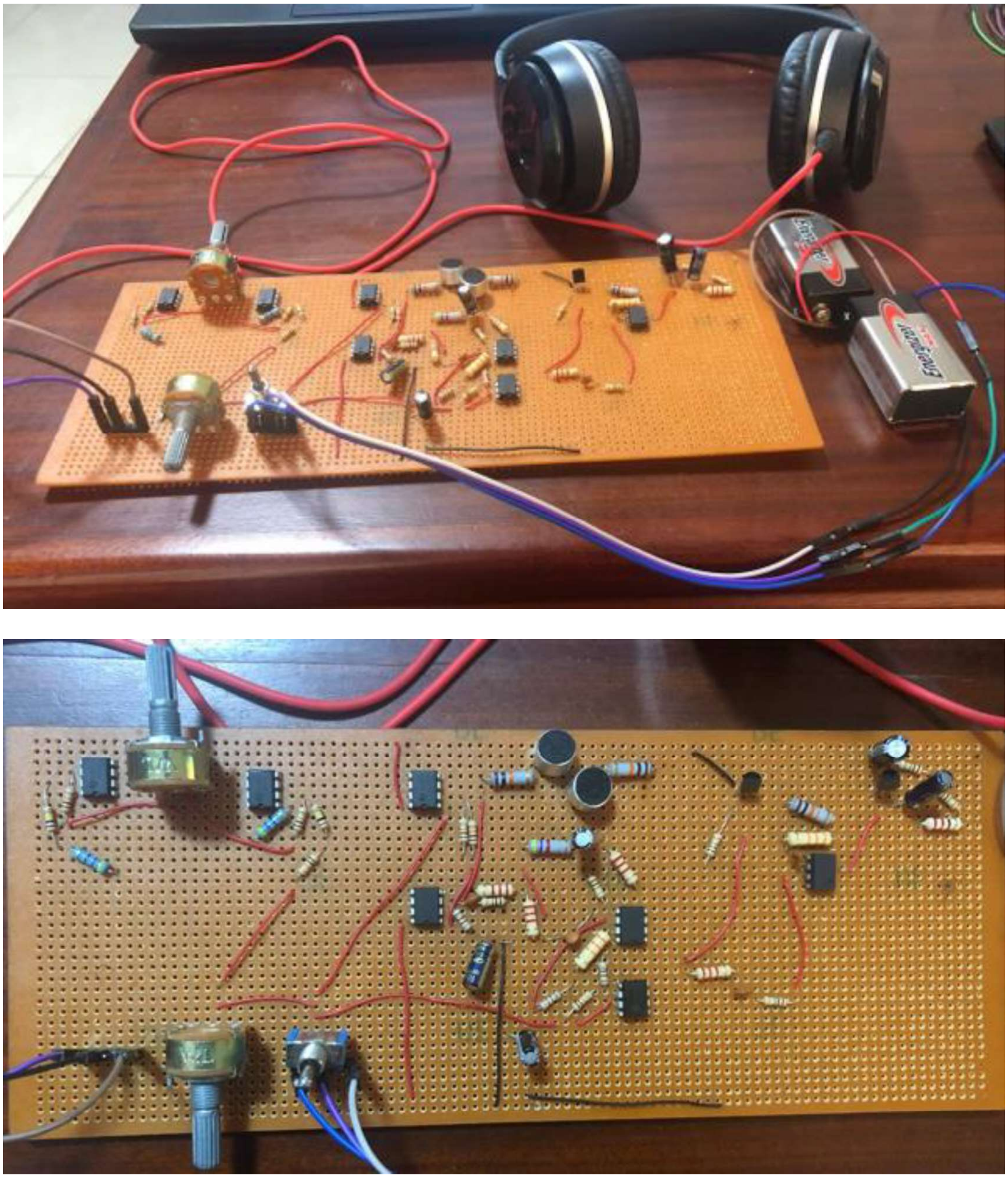


*Figure 5.2: PCB Design of Prototype*

## 5.1 Test Analysis

The system was placed in a quiet room with little background noise generated by a ceiling fan for testing purposes. The microphone elements were mounted on the headphones. After plugging both the microphone and headphone jacks, I put on the headphones, which eliminated echoes effectively. The toggle switch was flipped to observe its effect on sounds at different frequencies. However, it was observed the system attenuated sounds of low frequencies (≤ 1 kHz) much better than high frequencies. Music from an audio player connected to jack J2 was introduced into the apparatus. Adjusting the potentiometers ensured fine-tuning of the audio signal with little or no distortion. The output sound was finally played through the headphone's speakers.

## 5.2 Hardware Test Results

This section describes the results obtained after testing the physical device. The results generated from the working system are displayed below.

### *5.2.1 Source or Noise Signal*

The ambient sound picked up from the environment is converted to electrical signals by the electret microphones.

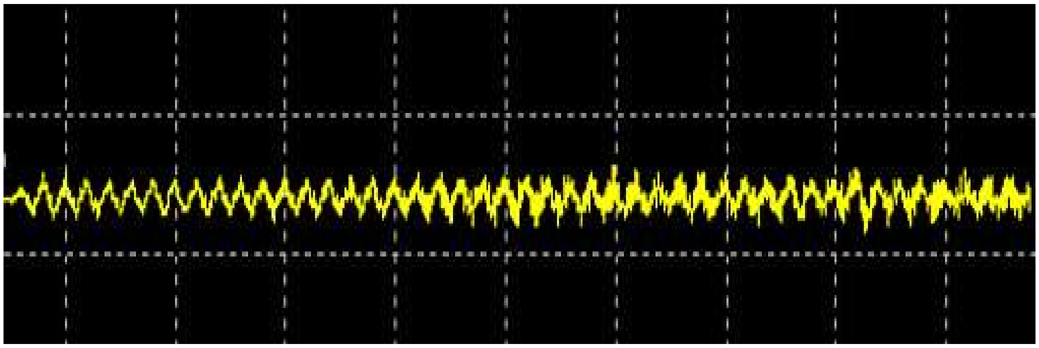


*Figure 5.3: Oscilloscope View of Noise Signal*

### *5.2.2 Amplification of Signal*

The purpose of the first sub-circuit is to amplify the noise signal from the microphone. The efficiency of the pre-amplifier is displaced in the graphs below. In Figure 5.3a and 5.3b, the noise signal had a small amplitude and was greatly amplified.

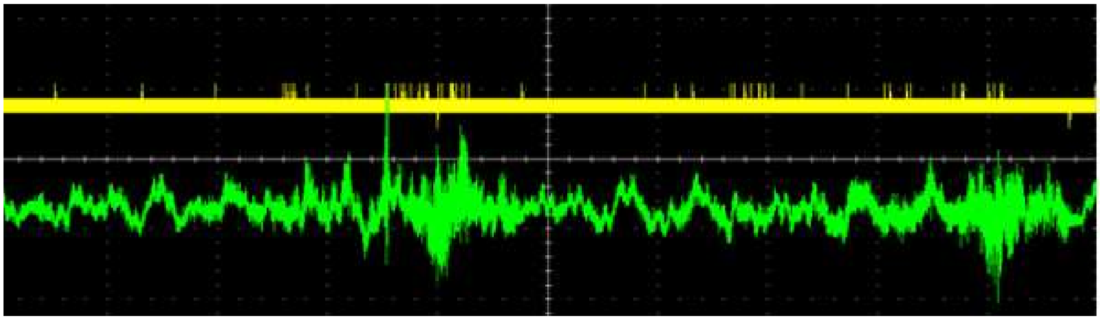


*Figure 5.4: Oscilloscope View of Amplified Noise Signal*

### *5.2.3 Inversion of Signal*

The second opamp achieves inversion of the phase of the amplified noise signal. Figures 5.4a and 5.4b illustrate the efficiency of the phase-inverter sub-circuit. The amplified noise signal undergoes a 180° phase-shift to generate the inverted signal.

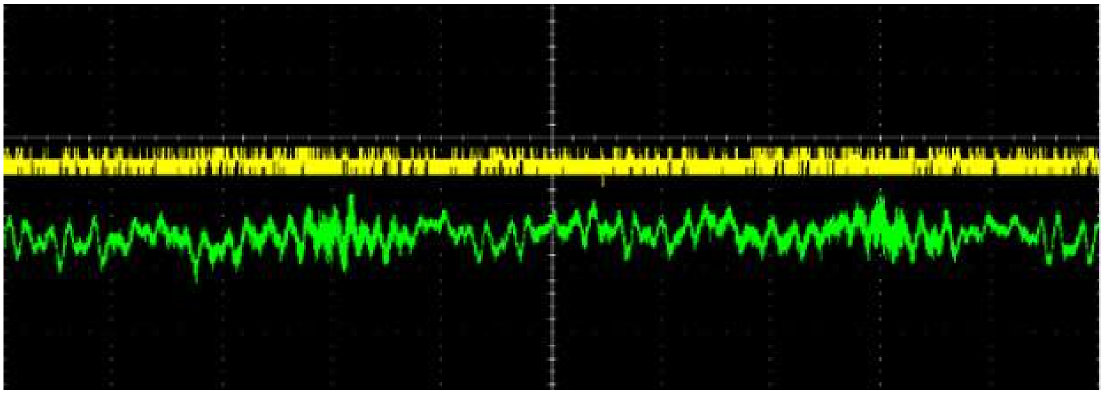


*Figure 5.5: Oscilloscope View of Inverted Noise Signal*

### *5.2.4 White Noise Signal*

Extra or unmasked sounds are canceled by introducing a white noise signal to the headphone amplifier's output.

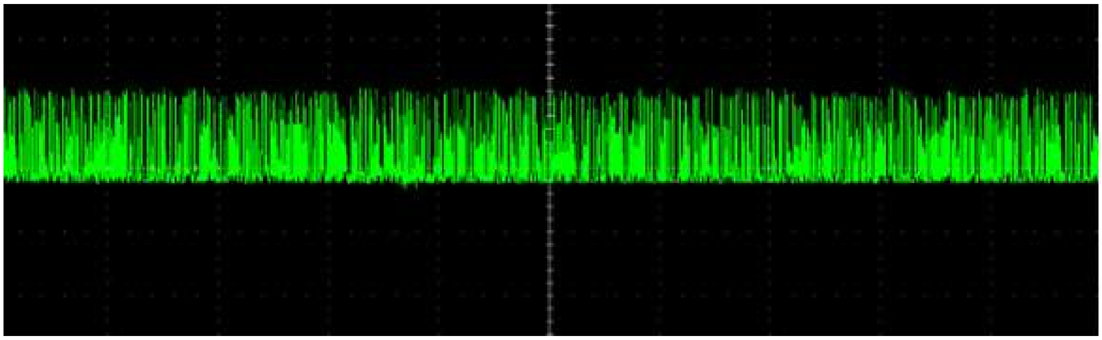


*Figure 5.6: White Noise Signal*

### *5.2.5 Summing of Signals*

The effectiveness of the summing amplifier in combining the inverted noise signal with auxiliary input (music) and white noise is displayed below.

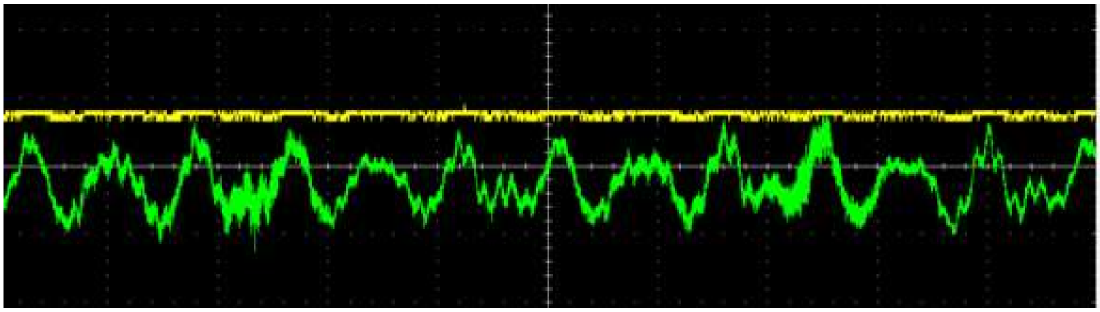


*Figure 5.7: Oscilloscope View of the Final Output*

## 5.3 Simulation Results

This section describes the results obtained after simulating the circuits using Multisim. The results generated are displayed below.

### *5.3.1 Source or Noise Signal*

A purely sinusoidal analog signal is captured and recorded by the condenser microphone. The noise signal is passed to the input of the pre-amplifier circuit for amplification.

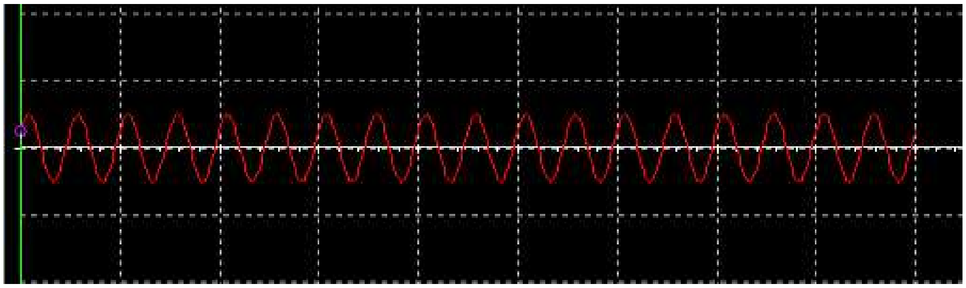


*Figure 5.8: Oscilloscope View of Noise Signal*

### *5.3.2 Amplification of Signal*

The original noise signal (blue waveform) is amplified in magnitude by the pre-amplifier. The noise signal has the same frequency as the amplified noise signal (green waveform) but different amplitudes.

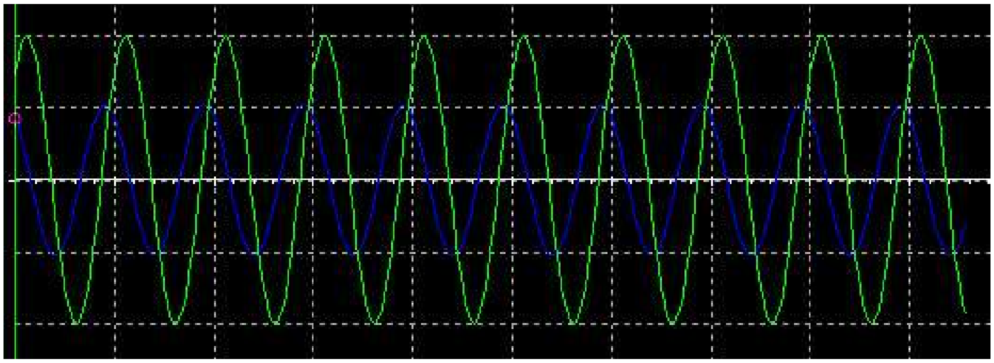


*Figure 5.9: Oscilloscope View of Amplified Noise Signal*

### *5.3.3 Inversion of Signal*

The phase inverter phase-flips the amplified noise signal through an angle of 180°. The output of the inverting amplifier is illustrated in the figure below.

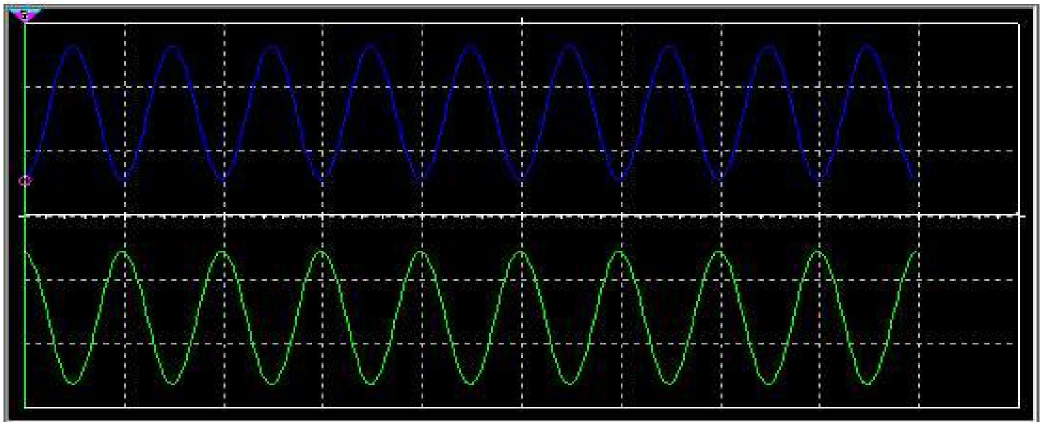


*Figure 5.10: Oscilloscope View of Inverted Noise Signal*

### 5.3.4 Auxiliary Input

An output microphone captures music from the computer audio system in Multisim. The music signal with a frequency of 50 Hz is fed as one of the inputs to the inverting summing amplifier.

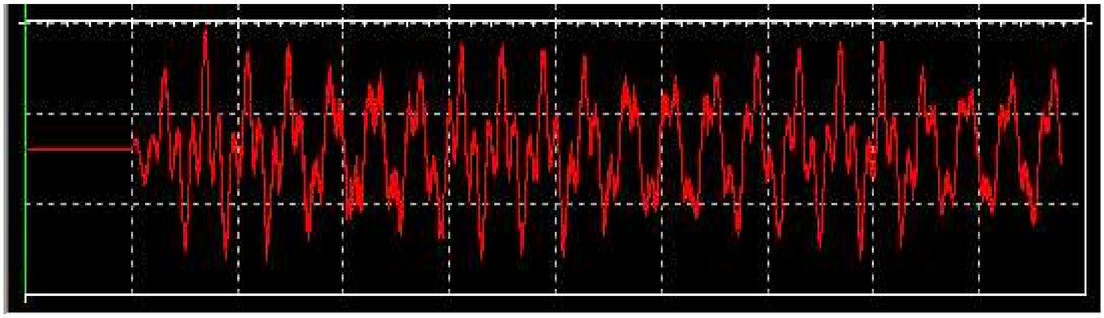


*Figure 5.11: Oscilloscope View of Auxiliary Input (music)*

### 5.3.5 White Noise Signal

Figure 5.12 shows the output of the simulated white noise circuit. Its broadband spectrum provides strong masking capabilities.

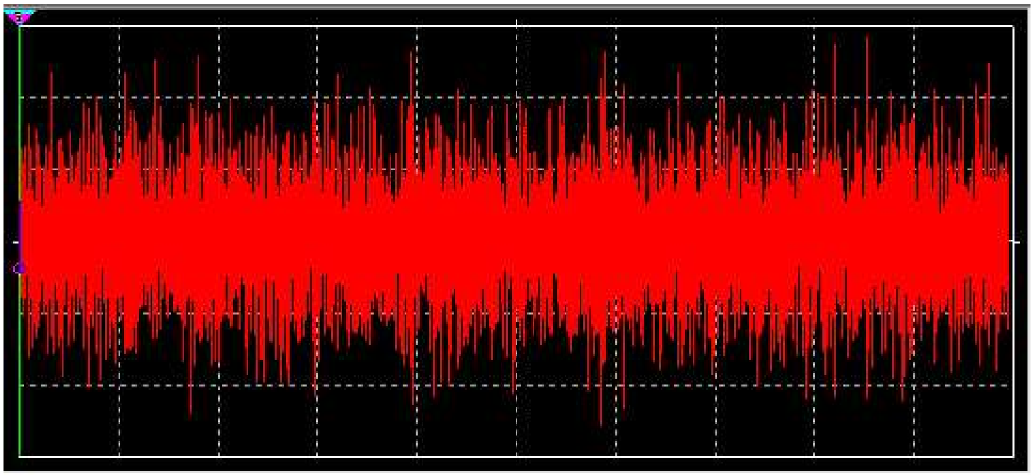


Figure 5.12: Oscilloscope View of White Noise Signal

### 5.3.6 *Summing of Signals*

The summing amplifier combines the noise signal, inverted noise signal, and music. The listener hears only the music from the headphone speakers since the noise and its inverted signal cancel out by destructive interference. The output of the summing amplifier has the same frequency as the music.

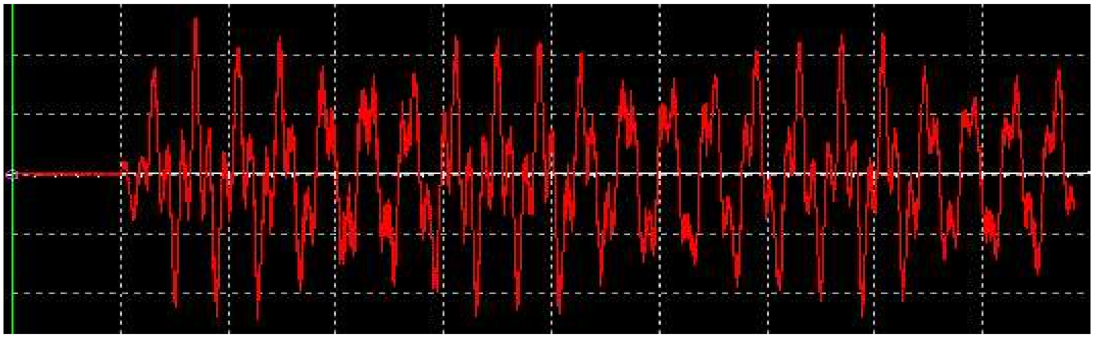


Figure 5.13: Oscilloscope View of Final Output

## 5.4 Statistical Analysis

A sample of the data points from the output waveforms was exported into Excel, with R Studio used for the statistical analysis. A Welch Two Sample t-test was used for the analysis since the samples came from two independent populations with equal variance, i.e., the simulated population and real-time population. The null and alternate hypotheses for testing the difference between the two samples or groups are clearly defined.

- Null hypothesis ($H_o$): the means of the two groups are equal ($\mu_1 = \mu_2$).
- Alternate hypothesis ($H_a$): the means of the two groups are not equal ($\mu_1 \neq \mu_2$).

The test results for the statistical analysis have been appended as APPENDIX D.

The difference in means for the data sample is 124.460588 (5.328685, 129.789273) from the noise signal data. The confidence interval also shows that the true difference in means occurs between ($-293.59196$ *and* $44.67079$). Hence, 95% of the time, the means of the two samples are not equal. The p-value of $0.044$ is less than 0.05. Thus, we can reject the null hypothesis of no difference and conclude with a high degree of confidence that a significant difference exists.

The difference in means for the data sample is 924.4492349 (−0.0209201, −924.4283148) from the final output signal data. The confidence interval also shows that the actual difference in means is between ($857.2321$ *and* $991.6664$). Hence, 95% of the time, the true means of the two samples are not equal. The p-value of $2.2e-16$ is much smaller than 0.05. Therefore, we can reject the null hypothesis of no difference and confidently conclude that the alternate hypothesis is true.

The statistical results prove that there is some degree of error between the hardware results and simulation results. This can be attributed to loose connections, power losses, aliasing, and microphone adaptability on the part of the hardware.

# 6 CONCLUSION

Noise is undesirable mainly because it makes conversation and other daily activities difficult. Health problems such as hearing loss may also arise from excessive exposure to noise. As was the project's aim, a low-cost and portable electronic noise masking system was modeled, designed, and built to mask stray sound from the external environment. Images of the working circuit have already been shown in the previous chapter. This prototype employs white noise to mask the residual noise after the outside noise has been canceled. Since it is difficult to capture all the external noise, hence the noise is not entirely canceled. Also, the inverter cannot invert the noise completely, and it is tough to thoroughly combine the noise and anti-noise (inverted noise). Therefore, white noise is employed. The prototype was tested by observing its effects on various noise sources such as human voice and a ceiling fan. In all cases, low-frequency sounds were predominantly attenuated.

Two 9V energizer batteries power this electronic circuit and require frequent replacement since they are non-renewable. This system works perfectly well with low-frequency signals as compared to high frequencies. It was noticed that noise masking is a privacy tool and can be used in many applications, especially speech privacy. The designed system can be used in offices, hospitals, financial institutions, and residential settings to eliminate stray noise.

## 6.1 Limitations

- All incoming sounds were not eliminated entirely due to the intensity and frequency of the sound and sound source position.
- Low-frequency sound waves were attenuated completely, while most higher-pitched sounds were masked with the introduction of the white noise signal.

- The circuit sometimes acts as a noise amplifier when the noise canceling is turned off, giving the input signal a static-like effect.
- The circuit requires much fine-tuning to ensure the original noise and phase-inverted noise signals have the same amplitude.

## 6.2 Future Work

- Embedding the noise-canceling circuitry in the headphones enhances miniaturization.
- A microphone filter can be included in the circuit to reduce the static-like effect of the microphone when the device is off.
- The efficiency of the ANC system can be enhanced by adaptively attenuating the fed-in signal.
- The power source should be rechargeable to limit frequent replacement of batteries and cut down operating costs.

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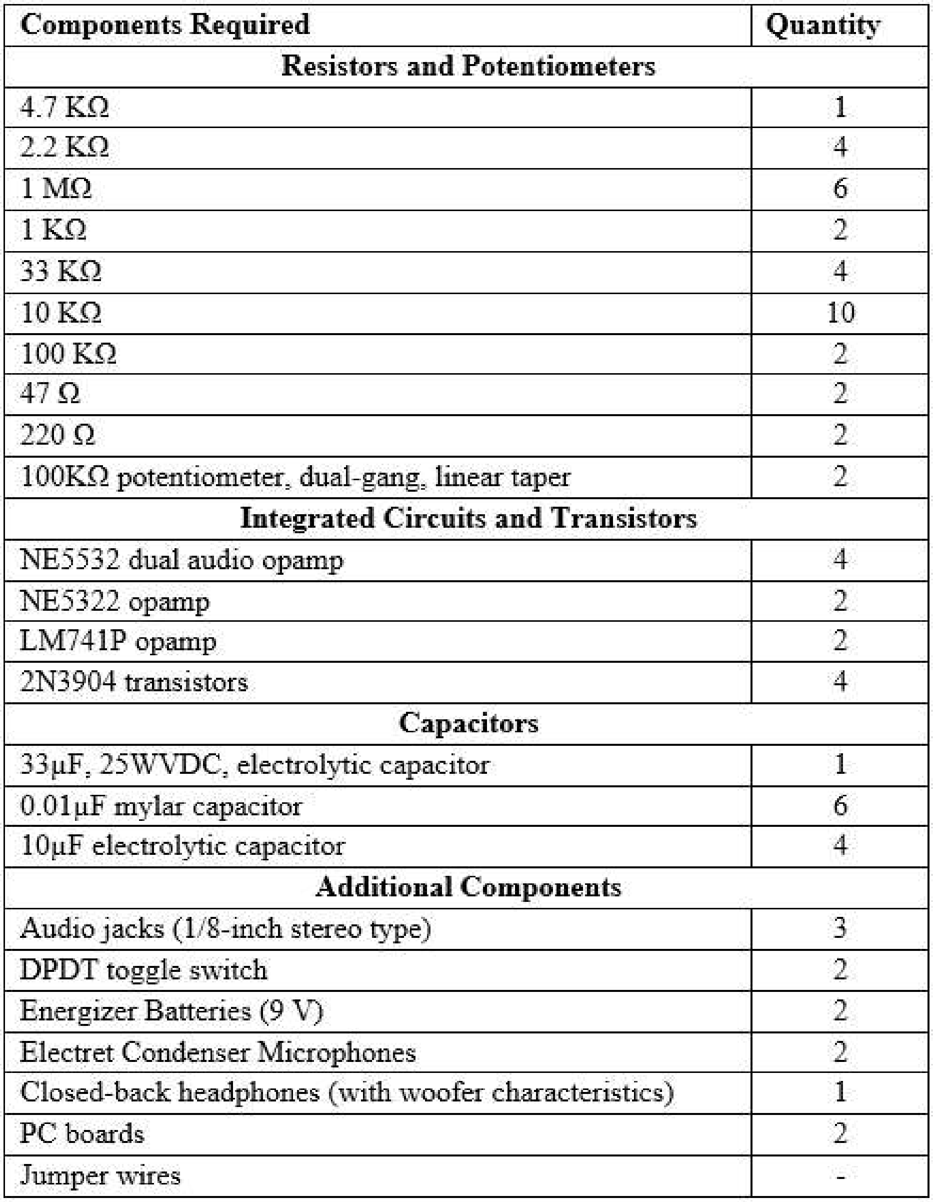
# APPENDIX A

*Table 3.4: Specifications of Hardware Components*



| **Components Required** | **Quantity** |
|---|---|
| **Resistors and Potentiometers** | |
| 4.7 KΩ | 1 |
| 2.2 KΩ | 4 |
| 1 MΩ | 6 |
| 1 KΩ | 2 |
| 33 KΩ | 4 |
| 10 KΩ | 10 |
| 100 KΩ | 2 |
| 47 Ω | 2 |
| 220 Ω | 2 |
| 100KΩ potentiometer, dual-gang, linear taper | 2 |
| **Integrated Circuits and Transistors** | |
| NE5532 dual audio opamp | 4 |
| NE5322 opamp | 2 |
| LM741P opamp | 2 |
| 2N3904 transistors | 4 |
| **Capacitors** | |
| 33µF, 25WVDC, electrolytic capacitor | 1 |
| 0.01µF mylar capacitor | 6 |
| 10µF electrolytic capacitor | 4 |
| **Additional Components** | |
| Audio jacks (1/8-inch stereo type) | 3 |
| DPDT toggle switch | 2 |
| Energizer Batteries (9 V) | 2 |
| Electret Condenser Microphones | 2 |
| Closed-back headphones (with woofer characteristics) | 1 |
| PC boards | 2 |
| Jumper wires | - |

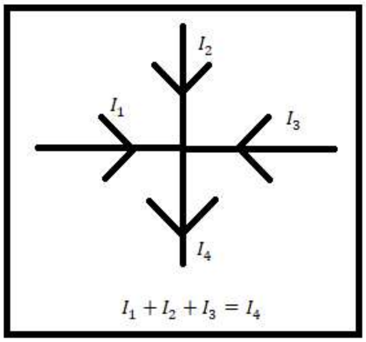
# APPENDIX B

Ohm's Law

$$\boxed{V = IR}$$

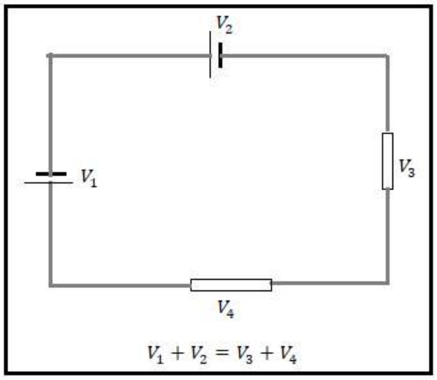
Kirchoff's Laws

- Junction Rule



$$I_1 + I_2 + I_3 = I_4$$

- Loop Rule



$$V_1 + V_2 = V_3 + V_4$$

*Resistors in Series*

$$R_{Total} = \frac{V}{I} = \frac{V_1+V_2+V_3+V_4}{I} + \cdots = \frac{V_1}{I_1} + \frac{V_2}{I_2} + \frac{V_3}{I_3} + \frac{V_4}{I_4} + \cdots$$

$$R_{Total} = R_1 + R_2 + R_3 + \cdots$$

*Resistors in Parallel*

$$\frac{V}{R_{Total}} = I = I_1 + I_2 + I_3 + I_4 + \cdots = \frac{V_1}{R_1} + \frac{V_2}{R_2} + \frac{V_3}{R_3} + \frac{V_4}{R_4} + \cdots$$

$$\frac{1}{R_{Total}} = \frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} + \frac{1}{R_4} + \cdots$$

## Voltage Division: Resistors in Series

$$V_T = IR_T$$

$$R_T = R_1 + R_2 + R_3 + \cdots$$

$$I = \frac{V_T}{R_T} = \frac{V_T}{R_1+R_2+R_3+\cdots} = \frac{V_n}{R_n}$$

$$V_n = \frac{R_n}{R_1+R_2+R_3+\cdots} V_T$$

## Current Division: Resistors in Parallel

$$Let\ G = \frac{1}{R}$$

$$V = I_T R_T = \frac{I_T}{G_T}$$

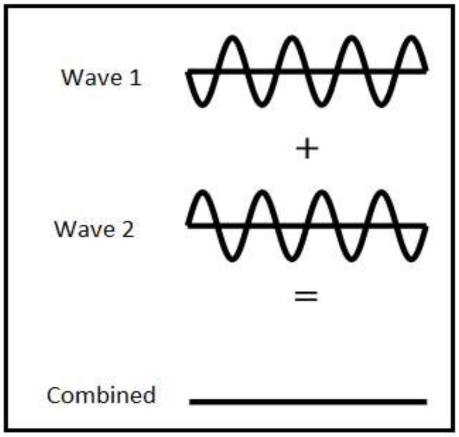
$$V = \frac{I_n}{G_n}$$

$$I_n = \frac{G_n}{G_t} I_t$$

$$I_n = \frac{G_n}{G_1+G_2+G_3+\cdots} I_t$$

## Principle of Superposition

- Destructive Interference



# APPENDIX C

Electret Microphone

$Bias\ Voltage = 2 - 10\ V$

$Operating\ Voltage = 9\ V$

$Operating\ Frequency = 20\ Hz$

$Impedance = 33\ K\Omega$

$Current: I = \frac{V}{Z} = \frac{9\ V}{33\ K\Omega} = 0.2727\ mA$

$Low\ pass\ filter\ cut - off\ frequency: f_c = \frac{1}{2\pi C_1 R_2} = \frac{1}{2\pi(33\mu F)(4.7\ K\Omega)} = 1.026\ Hz$

Non-inverting Pre-amplifier

$Gain = 1 + \frac{R_8}{R_6} = 1 + \frac{33\ K\Omega}{1\ K\Omega} = 34\ db$

$High\ pass\ filter\ cut - off\ frequency: f_{C_1} = \frac{1}{2\pi C_2 R_4} = \frac{1}{2\pi(0.01\mu F)(1\ M\Omega)} = 15.92\ Hz$

$f_{C_2} = \frac{1}{2\pi C_3 R_6} = \frac{1}{2\pi(10\mu F)(1\ K\Omega)} = 15.92\ Hz$

Unity-gain Phase Inverter

$Gain = \frac{R_8}{R_7} = \frac{10\ K\Omega}{10\ K\Omega} = 1\ db$

Inverting Headphone Amplifier

$Gain\ (for\ microphone\ signal) = \frac{R_{11}}{R_9} = \frac{100\ K\Omega}{10\ K\Omega} = 10\ db$

$Gain\ (for\ audio\ signal) = \frac{R_{11}}{R_{10}} = \frac{100\ K\Omega}{10\ K\Omega} = 10\ db$

White Noise Generator

$Supply\ Voltage = 9\ V$

$High\ pass\ filter\ cut - off\ frequency: f_c = \frac{1}{2\pi C_8 R_{29}} = \frac{1}{2\pi(10\mu F)(220\ \Omega)} = 72.34\ Hz$

Headphone Speakers

$Operating\ Voltage = 9\ V$

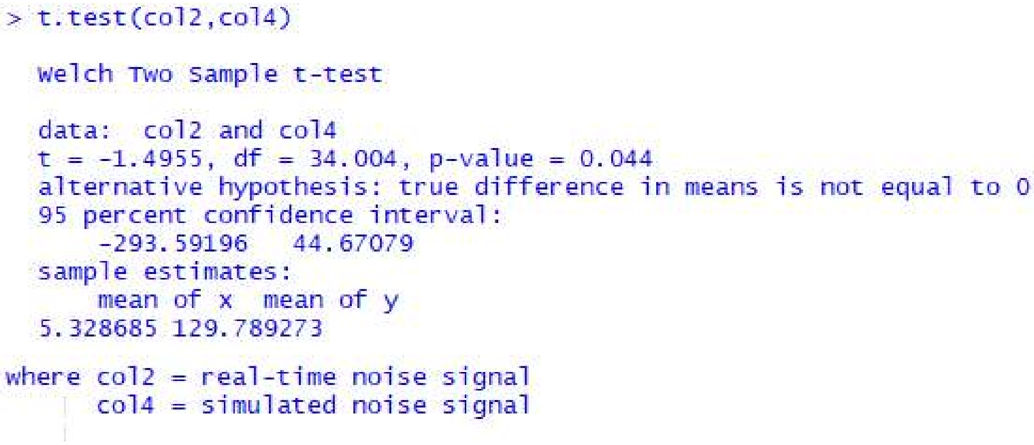
$RMS\ Voltage\ Supply = 10\ V$

$Overload\ Protection\ Impedance = 47\ \Omega$

$Alternating\ Current\ through\ the\ speaker, I = \frac{V}{Z} = \frac{9\ V}{47\ \Omega} = 191.49\ mA$

# APPENDIX D

Noise or Input Waveform



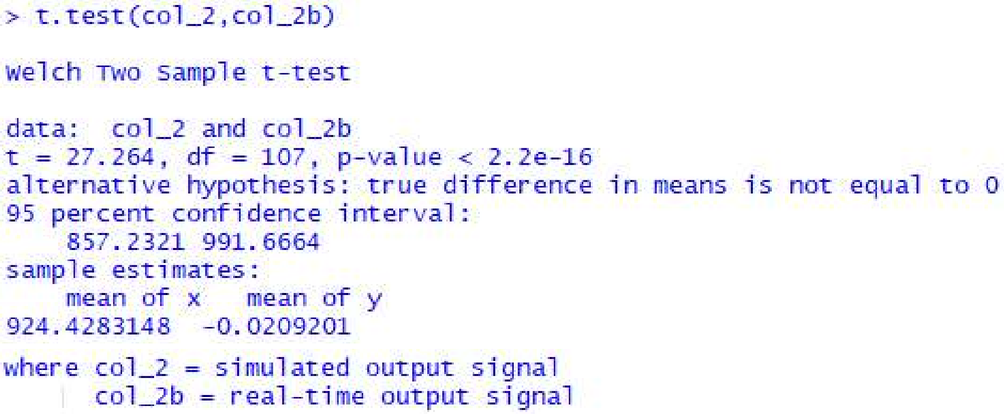
```
> t.test(col2,col4)

  Welch Two Sample t-test

  data:  col2 and col4
  t = -1.4955, df = 34.004, p-value = 0.044
  alternative hypothesis: true difference in means is not equal to 0
  95 percent confidence interval:
      -293.59196   44.67079
  sample estimates:
      mean of x  mean of y
  5.328685 129.789273

where col2 = real-time noise signal
      col4 = simulated noise signal
```

Final Output Waveform



```
> t.test(col_2,col_2b)

Welch Two Sample t-test

data:  col_2 and col_2b
t = 27.264, df = 107, p-value < 2.2e-16
alternative hypothesis: true difference in means is not equal to 0
95 percent confidence interval:
    857.2321 991.6664
sample estimates:
    mean of x   mean of y
924.4283148  -0.0209201

where col_2 = simulated output signal
      col_2b = real-time output signal
```