Bachelor Thesis

Pre-Amplifier and Noise Cancellation

for an Intelligibility-Enhancing Automatic Volume Control System

T. Timmer & Q. van Wingerden

Automatic Volume Control project

This page is intentionally left blank.

Bachelor Thesis

Pre-Amplifier and Noise Cancellation
for an Intelligibility-Enhancing Automatic Volume Control System

by

T. Timmer & Q. van Wingerden

to obtain the degree of Bachelor of Science
at the Delft University of Technology,
to be defended on Thursday July 4th at 9:00, 2019.

Project duration:	April 23, 2019 – July 5, 2019	
Thesis committee:	ir. E.W. Bol	TU Delft, Chair
	dr.ir. R.C. Hendriks	TU Delft, Supervisor
	dr. J. Martinez	TU Delft, Jury Member 1
	dr.ir. A. Koutrouvelis	TU Delft, Jury Member 2 and Assistant Supervisor

An electronic version of this thesis is available at http://repository.tudelft.nl/.

TU Delft

This page is intentionally left blank.

Abstract

This Bachelor graduation project has the goal to create a device which is able of automatic volume control, to be used for enhancing speech intelligibility. To tackle the intelligibility of speech through Public Address Systems (PA Systems), a *Intelligibility-Enhancing Automatic Volume Control* system was proposed.

The total system to be made must be able to alter a clean speech signal according to a noise estimation. Then the altered, enhanced, signal should be amplified before being sent to an existing Public Address System. A subsystem is added in order to dampen the outside noise in a car-like environment.

The whole project is divided into three parts: *Noise Statistics Estimation*, *Intelligibility Enhancement* and *Amplifier and Noise Cancellation*. These parts have been performed by three different subgroups. In this report, the *Amplifier and Noise Cancellation* is discussed. The other parts are explained in the respective reports [1, 2].

The *Amplifier and Noise Cancellation* group will amplify the enhanced audio signal with the use of a pre-amplifier. This group also introduces an additional noise cancellation subsystem for usage in enclosed spaces, like a car. It does so by inverting the recorded environment noise below 500 Hz, and adding this to the *to be amplified signal* before sending it to the PA System.

This thesis is divided in two main design sections: the design of the pre-amplifier and the design of the active noise cancellation circuit. At the heart of both circuits lies a LM386 Audio Operational Amplifier (Op-Amp) but they both have different objectives.

The pre-amplifier is designed to have a flat transfer function in audio range, 20 Hz to 20 kHz. The output level of the pre-amplifier is a standard level for consumer electronics, being 447 mV_{max}. The pre-amplifier inverts the signal from the microphone and the audio signal to achieve noise cancellation.

The noise cancellation circuit features a microphone amplifier and a Low Pass Filter (LPF). The microphone amplifier amplifies the signal so that the microphone circuit's output level is at the same level of the audio input of the pre-amplifier (200 mV_{pp}). The filter makes sure only sounds below 500 Hz are passed to the pre-amplifier.

With the inverting capabilities of the pre-amplifier and both signals being completely out of phase, a theoretical cancellation of sound signals is possible. Because of the LPF used in the microphone amplifier this cancellation is done for signals below 500 Hz.

At the end of the project, a system was built which met most of the requirements. Some of the requirements can not be satisfied due to incapability of the test equipment available. The system does amplify the signal to the desired amplitude and is capable of slightly cancelling noise in a car. However, improvements of the product are needed to function more optimally.

This page is intentionally left blank.

Preface

This bachelor graduation thesis closes the door of the three-year bachelor Electrical Engineering programme. The programme offered a lot of theoretical and even practical knowledge to the students. The journey embarked with the basic knowledge of circuit design, and it took us all the way to Maxwell's equations, gaining more and more knowledge along the way.

Not only did our knowledge increase, but our mindset has changed as well. During our studies, we learned to sometimes take a step back, regroup and go for it again, with a fresh view of the problem. There was plenty of room for gathering experience as well. Some of us joined a student D:DREAM team, whilst some took it upon themselves to follow an internship.

In this final examination, the combination of the above mentioned skills are tested. By successfully completing this final examination, the students can proudly say that they graduated the programme.

First of all, we would like to thank the Dean of EEMCS, John Schmitz, for providing the bachelor graduation project proposal. It was a challenging, but interesting proposal which left plenty of room for our own interpretation.

We would also like to thank our supervisor, Richard Hendriks, for taking the time to supervise our group whilst the proposal was not his own.

In addition, we want to thank our co-supervisor, Andreas Koutrouvelis, for co-supervising our group throughout the whole process.

Next, our gratitude goes to Gerard Janssen for providing us with feedback, a lot of insight into the problem and theoretical help during our work. The help he provided us with gave our sub-group purpose inside the larger system. He also helped to review this thesis. Without him the project could not have been completed to this extend.

Lastly, we would like to thank the other two subgroups for working together on this project. While the borders of the subgroups were sometimes not so clear, the collaboration turned out great. It was a very educational journey and we are satisfied with the presented project outcome.

T. Timmer & Q. van Wingerden
Delft, June 2019

Contents

1
Introduction

This project is done in cooperation with two other groups [1, 2], called the Noise Statistics Estimation group and the Intelligibility Enhancement group, respectively. These groups tackle different aspects of the same problem. Therefore the project introductions found in the theses are similar. This document will focus on the third group, Amplifier and Noise Cancellation.

1.1. Intelligibility-Enhancing Automatic Volume Control

Recognising speech in noisy environments can be challenging. This is best described by the cocktail party problem [3]. You may find yourself at a party wanting to talk to the person next to you. Every other attendee wants to talk to someone else and this results in a noisy environment. It is challenging to focus on the speech of the person next to you and the possibility occurs that the speech is not intelligible at all.

The same kind of problem may be present at a press conference or at a train station. You clearly want to hear what the speaker is saying through the Public Address System (PA System), but you are having trouble to do so. To tackle this problem, we propose an Intelligibility-Enhancing Automatic Volume Control system. This system will improve the intelligibility of speech in noisy environments. The full system overview is shown in Figure 1.1.

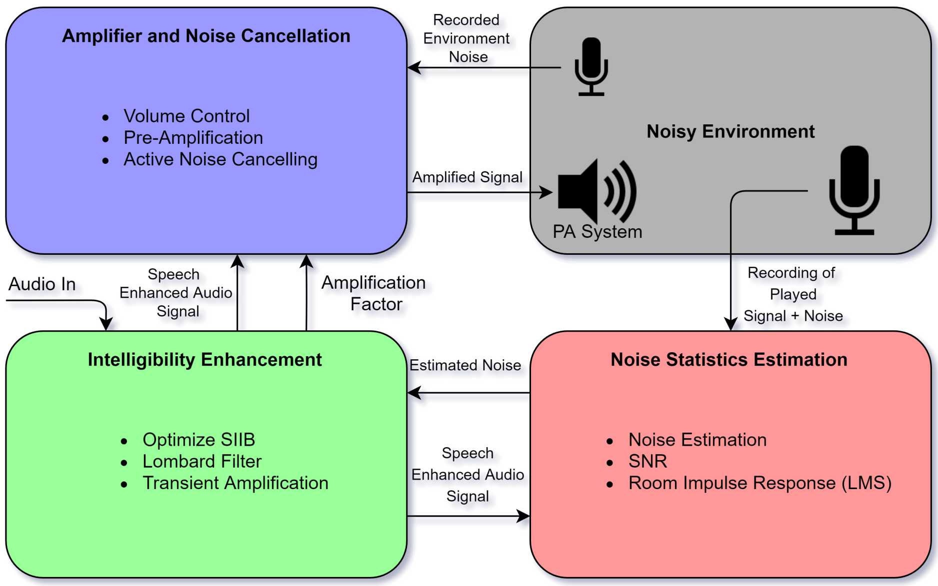


Figure 1.1: The Project Overview.

The *Noise Statistic Estimation* group is shown in Figure 1.1 in the pink box in the lower right corner. This group will record the playing signal, acoustic noise in the room and the added amplifier noise and will use the enhanced audio signal to calculate the Signal-to-Noise Ratio (SNR) and room impulse response. It will then pass the estimated noise towards the *Intelligibility Enhancement* group.

The *Intelligibility Enhancement* group is shown in Figure 1.1 in the green box in the lower left corner. This group will take the audio input signal and will enhance the speech in this signal by the use of multiple algorithms, controlled by the estimated noise from the *Noise Statistics Estimation* group. It will then pass this enhanced signal to the *Noise Statistics Estimation* group and to the *Amplifier and Noise Cancellation* group. It also passes an Amplification Factor (AF) to the *Amplifier and Noise Cancellation* group which will control the volume.

Last but not least, the *Amplifier and Noise Cancellation* group is shown in Figure 1.1 in the blue box in the upper left corner. This group will amplify the enhanced audio signal with the use of a pre-amplifier and will use the AF received to adapt the volume automatically to enhance intelligibility. This group also introduces an additional noise cancellation subsystem for usage in enclosed spaces, like a car. It does so by inverting the recorded environment noise below 500 Hz, and adding this to the enhanced audio signal before amplifying and sending it to the PA System.

1.2. State-of-the-Art

While pre-amplifiers are crucial in some (professional) audio systems [4], these systems are often not documented in researches. Nevertheless, the literature on amplification in hearing aids gives a valuable impression of how to tackle the problem of amplifying an audio signal and keeping it intelligible. Audio amplification problems arise mainly in designing hearing aids [5]. Even visually guided hearing aids are looked into [6]. Although these are different applications, in their essence they solve the same problem. Since technology is ever developing, the use of audio editing programs like Audacity [7] is ever rising, making audio editing easier and more available to the consumer. Also, the subject of automatic gain control has been studied in [8].

Regarding noise suppression inside cars, there are a lot of systems available. Each of these systems work in different ways. Not all of these systems are controlled through the audio system of the car. For example, noise suppression with the use of dampening fabrics has been researched [9]. It has been proven that, with the use of the correct textile materials, sound suppression is very effective. However, this is for non-knitted materials [9]. The disadvantage of non-knitted materials is that they aesthetically look worse than knitted materials. When knitting materials with the use of spacer structures, the sound suppression results look promising. There are even ideas to have the fabric move as an actuator to actively suppress noise [9].

Not only the textile industry is keeping itself busy with noise suppression in cars. The brand Bose, known for its QuietComfort noise-cancelling headphones, recently developed a car-version. It works with accelerometers at the wheels and in the frame to sense the vibrations of the car and subsequently filtering these out as noise. It also uses the sound system in the car to send out anti-noise [10]. This product is however not available in the market at the time of writing. Silentium has also developed a system, called Quiet Bubble™, to reduce the noise in a car during the different driving conditions [11]. These are just two recent examples of active noise control. Systems have already been made for the tonal engine noise suppression and interior noise suppression in aircraft [12]. Nissan put active noise control in a Bluebird vehicle in 1992, where amplifiers, loudspeakers and processors (separate from the car-audio system) were used [12]. Active engine noise systems are introduced by Honda (2005), Toyota (2005) and Lotus (2008) [12].

1.3. Thesis Outline

The thesis outline is as follows. Chapter 2 lists the requirements for the proposed *Intelligibility-Enhancing Automatic Volume Control* system as well as the requirements for the *Amplifier and Noise Cancellation* subsystem. An overview of the subsystem is provided in chapter 3.

The two main designs are described in more detail in the chapters thereafter. Chapter 4 describes the design process of the pre-amplifier. This includes an input analysis, a simulation, the selection of the Integrated Circuit (IC) and the implementation of the circuit and code used. Chapter 5 then describes the design process of the noise cancellation subsystem. It also includes an input analysis, the circuit design and the actual implementation of the circuit.

Chapter 6 connects both circuits and explains the test setup and results of both of these subsystems. Chapter 7 includes the discussion of the end product. The results are validated according to the Programme of Requirements. In chapter 8 the conclusions are drawn. This includes recommendations for future work.

2

Programme of Requirements

In order to clearly define the project boundaries and results, a Programme of Requirements (PoR) should be constructed. In this PoR the requirements of the final product are listed. First, the general PoR of the entire project will be listed. In specifying these requirements for the overall system, some assumptions were made regarding the noise conditions and the existing hardware in the near-end environment. This will not be discussed further in the document, as the focus of the document lies on the part covered by the *Amplifier and Noise Cancellation* subgroup. Second, the subgroup's Key Performance Indicators (KPIs) are listed. Then the Mandatory Requirements followed by the Trade-off Requirements are given.

2.1. General Requirements

These are the general requirements for the entire project. These requirements will always be kept in mind.

Assumptions

1. The near-end noise is uncorrelated with the speech signal.
    - This is a reasonable assumption when near-end noise is defined as a signal that consists of contributions of all noise sources except the loudspeaker.
2. The existing power amplifier needs an audio input at line level (447 mV_{max} [13]).
    - Typical value for consumer applications.
3. The voltage gain of the existing power amplifier is equal to 25.
    - A set gain of the existing system is needed, to determine how much the system needs to amplify the signal in order to reach a certain output level at the speaker.
4. The output power level of the existing power amplifier is less than 100 dBA.
    - From [14], the maximum permissible occupational noise exposure for 2 hours is 100 dBA. Assuming that PA System employees work for 8 h per day, this means the PA System can make announcements for 25 % of the time.
5. The input signal is pre-recorded and noise-free.
    - Typical for announcements, music and audio-books.

With these assumptions, the general requirements can be determined.

Mandatory Requirements

The mandatory requirements need to be met in order for the overall system to be considered successful.

- The system must improve intelligibility such that the word recognition rate is at least 90 % in the presence of near-end noise.
- The system must be able to suppress near-end noise in the frequency band from 0 Hz to 500 Hz.
- The system must operate in Signal-to-Noise Ratios below 15 dB.
- The system must be able to process speech in the frequency band from 0 kHz to 8 kHz.
- The system must be able to play audio in the frequency band from 20 Hz to 20 kHz.
- The system must be able to process pre-recorded speech in advance (pre-processing).
- The system must not damage hearing.
- The system must not add more than 3 dBA noise to the enhanced speech signal.
- The system must have a maximum pre-processing delay of 5 times the duration of the input signal with a maximum of 20 minutes.
- The system must have a maximum latency of 100 ms without pre-processing.
- The system must not record near-end noise when the system is not broadcasting any audio.
- The system must not store recorded near-end noise longer than the system latency.

Trade-Off Requirements

The trade-off requirements are not necessary to be met, however, if the they are met, the end-user will become increasingly satisfied.

- The system should be plug-and-play.
- The system should be able to work on a 12 V power supply voltage.
- The sound coming out of the system should sound natural according to listening tests.
- The system must not add more than 1 dBA noise to the enhanced speech signal.
- The combined price of all the individual components should not exceed €100.

2.2. Subgroup Requirements

Here the Key Performance Indicators of the subgroup are listed. These are used to evaluate success of the product. After the KPIs, the mandatory requirements and trade-off requirements are given.

2.2.1. Key Performance Indicators

The KPIs are tested later in the document, chapter 7, according to the requirements given in the following subsections, subsection 2.2.2 and subsection 2.2.3.

Output Level

The output level of the system is a very important measurement. If the output level is not of adequate strength, the signal may be played too weakly, or too loud.

Latency

The latency of the system is also very important. When the latency is too high, the delay between what is played and what is measured may influence the performance of the system.

Power Consumption

In a world where energy is becoming more and more important, it would be wise to limit the power consumption of the system. The power consumption, expressed in Watt (W), is a good performance measurement.

Noise Sound

The system's output should be of high quality. This partially means that there should be as little noise as possible added by the system, as less noise improves the SNR.

Audio Quality

The audio quality, not to be confused with intelligibility, is perhaps the most important indicator. With a good audio quality, people are more eager to buy the product. This is however not easy to measure, listening tests can be deployed for this.

2.2.2. Mandatory Requirements

- The product must not introduce additional noise of more than 3 dBA.
- The product must be plug-and-play, meaning that it can be used on any system.
- The product must be able to work on 12 V.
- The pre-amplifier must have a flat transfer function $H(f)$ from 20 Hz to 20 kHz.
- The product must not infringe privacy.
- The system must not introduce delay of more than 10 ms.
- The pre-amplifier must have a standardised line level of −10 dBV (= 447 mV$_{max}$ [13]).
- The filters must not remove information bearing frequencies.
- Both the input and output of the product should be connected with 3.5 mm jack-plugs.
- The SNR must be improved by a minimal of 3 dB with the use of the noise-cancelling circuit.
- The product must fit into the dashboard cabinet of a car.
- The system must be able to suppress near-end noise in the frequency band from 0 Hz to 500 Hz.

2.2.3. Trade-off Requirements

- The sound coming out of the product should not be distorted.
- The subsystem should cost less than €80.
- The product should have a power consumption of less than 10 W.
- The system should have an option to play music.
- The system should have a bass-boost option.

2.3. Inputs

This system will have three inputs. These inputs are necessary to have the system function as intended.

- **Audio**, the enhanced input signal from the *Intelligibility Enhancement* group.
- **AF**, the Amplification Factor from the *Intelligibility Enhancement* group.
- **Car Sound**, the recorded noise in a car.

It is assumed that the input signal is already optimised for intelligibility (*Intelligibility Enhancement* Group) and that the frequency band in which they want to broadcast is the human hearing range (20 Hz to 20 000 Hz) [15]. The assumption is made that the microphone only picks up noise, due to the isolation of the prototype.

2.4. Outputs

Below, the output of the system is given.

- **To Speaker**, this is the signal which will be send to the existing PA System.

3

System Overview

This chapter gives an overview of the entire *Pre-Amplifier and Noise Cancellation* system design, which is discussed in detail in chapter 4 and chapter 5, respectively. In Figure 3.1 the flow of the signals is shown. This is the initial design, alterations were made during the design process. The system will consist of two amplifiers, one voltage regulator, one microphone, passive circuit elements, a dual digital potentiometer and one Microcontroller Unit (MCU).

The *audio* input is the enhanced speech signal from the *Intelligibility Enhancement* group or the music input. The amplitude of the incoming signal will be compared to line level [13] and the gain can be calculated accordingly. This gain calculation will then be mapped to a resistance value and the gain is changed automatically.

The AF input comes from the *Intelligibility Enhancement* group. This signal will be used to control the volume. This volume calculation will then be mapped to a resistance value and a digital potentiometer value will be changed accordingly, effectively changing the volume.

The last input *Car Sound* is a signal which comes from a microphone. The microphone will record sound in the car cabinet and it will then compare this sound to the sound played out by the speaker. This is done to receive only the recorded noise. Once the noise is known it is passed through a Low Pass Filter (LPF), inverted and then added to the signal being played. This will cancel out the noise.

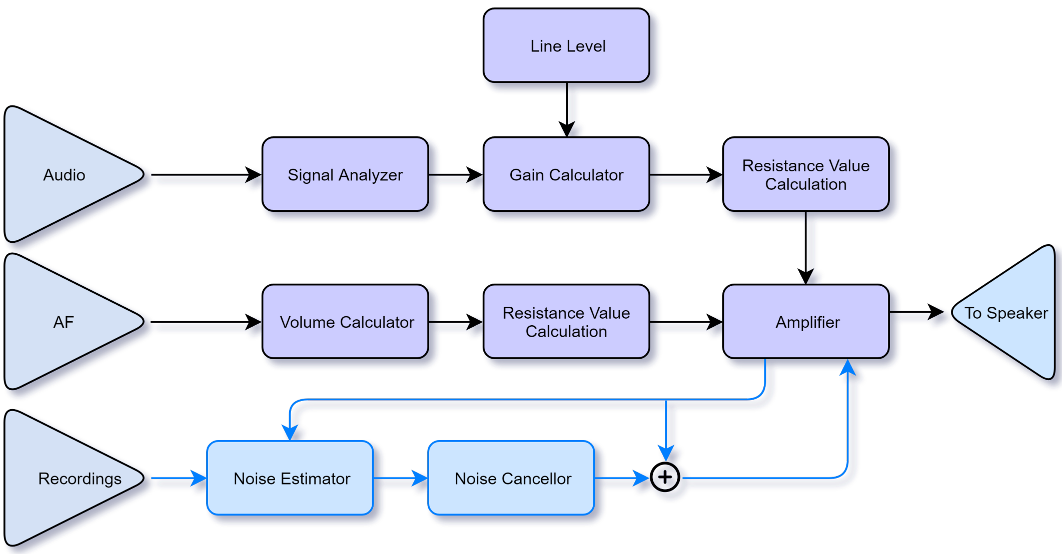


Figure 3.1: An overview of the amplifier system.

3.1. Power Supply

The product should work on 12 V because it has the car add-on. Since most new cars nowadays have USB ports, the product should also work on 5 V. Therefore a voltage regulator of 5 V is used to ensure the product will have this compatibility. This voltage regulator provides power to both the amplifiers, the digital potentiometer and to the MCU. There was no real requirement on the voltage regulator other than an output voltage of 5 V. This is because the current drain of the circuitry is so low that every voltage regulator suffices. The regulator used is the *L7805CV* from ST Microelectronics, which provides 5V at a maximum of current 1.5A [16].

4

Amplifier Design

The amplifier designed is a pre-amplifier. Such an amplifier converts the input amplitude to *line level* which then can be used by a power amplifier. Since the system will become a plug-and-play system, a pre-amplifier is needed in order to make the signal compatible with every PA System [4]. In section 4.1 the analysis of the input for the pre-amplifier is given, hereafter section 4.2 will lay out a simulation of the Operational Amplifier, section 4.3 compares different Integrated Circuits and section 4.4 explains the amplifier design process.

4.1. Input Analysis

In order to design the pre-amplifier, the input audio level of the source first has to be known. Therefore, an analysis was made by measuring audio output levels from multiple input source devices. This analysis resulted in the discovery that the output levels from these devices where non-identical, as can be seen in Figure 4.1.

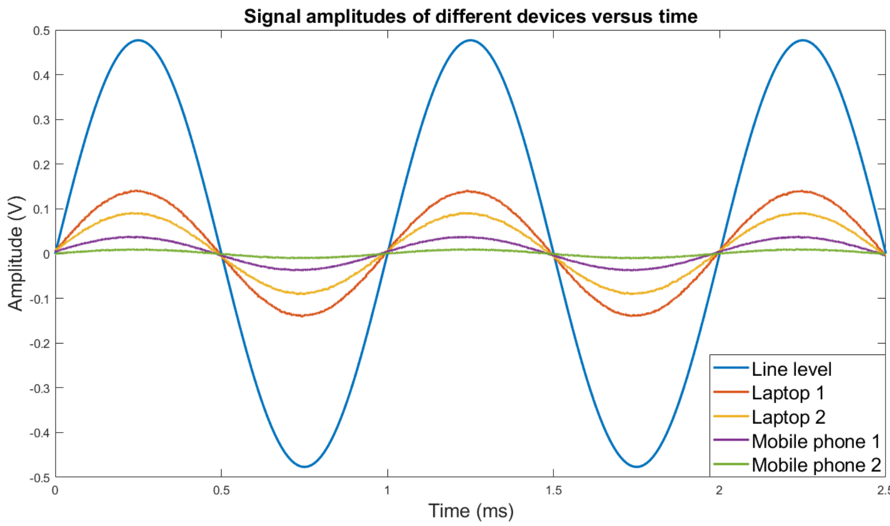


Figure 4.1: Audio level outputs of 4 different devices plotted against time, compared with line level.

Once the source outputs were properly defined, and line level was chosen as the reference amplifier output, the gain of the amplifier could be calculated. Dividing line level (V_{max} = 447 mV [13]) by the average of the audio level outputs from Figure 4.1, which was found to be 100 mV$_{max}$, a standardised gain of *4.5* was found.

4.2. Op-Amp Simulation

The initial thought was that a fully automatic variable gain amplifier could be used to control the output of the amplifier. The options for this were an inverting Op-Amp and a non-inverting amplifier. The gains realised by these amplifier setups are listed in Equation 4.1a and Equation 4.1b, respectively [17], where *GainControl* is the potentiometer value from Figure 4.3. *R1* is the value of resistor *R1*. The resulting gain is called $Gain_{inverting}$ and $Gain_{non\text{-}inverting}$, respectively.

$$Gain_{inverting} = -\frac{GainControl}{R1}. \tag{4.1a}$$

$$Gain_{noninverting} = 1 + \frac{GainControl}{R1}. \tag{4.1b}$$

Since the amplifier should realise the standardised gain of *4.5*, both of the options were viable to use. Therefore a simulation was first carried out. To make sure the gain could also be used to scale down the input voltage, if proven to be greater than line level, the inverting Op-Amp was first used in the simulation. The non-inverting amplifier would not be able to do this since the gain is always higher than 1, see Equation 4.1b. The simplified circuit of the simulation, via Simulink [18], is seen in Figure 4.2.

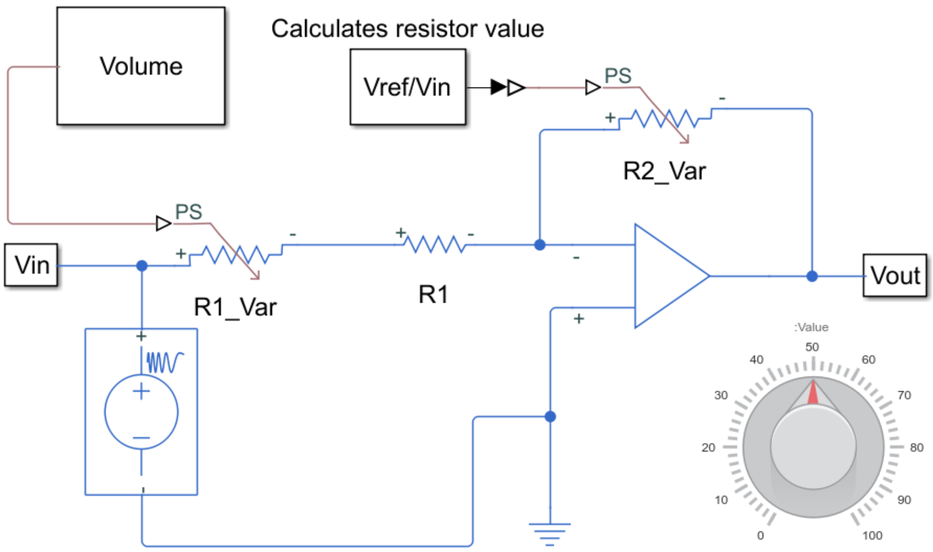


Figure 4.2: The Simulink Simulation Circuit with both variable gain and variable volume.

The "Calculates resistor value" algorithm divided line level *(Vref)* by the input voltage *(Vin)* to change the resistor value *(R2_Var)*, which changed the gain of the amplifier and resulted in a change of output level *(Vout)*. Due to the fact that the gain had to be changed for even the smallest input fluctuations, it was discovered through this simulation that the implementation of a fully automatic variable gain was too unstable. Therefore, only an automatic volume control is proposed in which the AF will be the main control signal for altering the volume.

Due to the controlling of the volume rather than the gain, the decision was made to use a non-inverting Op-Amp. This was possible because the input voltage has been set to the averaged input analysis signal strength, $100\,\mathrm{mV}_{max}$. Only the volume has to be tuned automatically for output amplitude control. The final simulation circuit, in which a non-inverting Op-Amp is used is found in Figure 4.3. The in-simulation knob value determines the AF by being turned by the user. This was simulated in a range from *0* to *100*, where *50* is the middle point. The values of the Amplification Factor do not match the actual input value, but it is merely a way to simplify the volume control simulation. The Gain Control potentiometer is used to achieve the standardised gain of *4.5*. This potentiometer is still present in order to change the gain if needed. For example, it might be needed if there is no power amplifier present and the pre-amp has to act as a power amplifier. This is not the case in the given application.

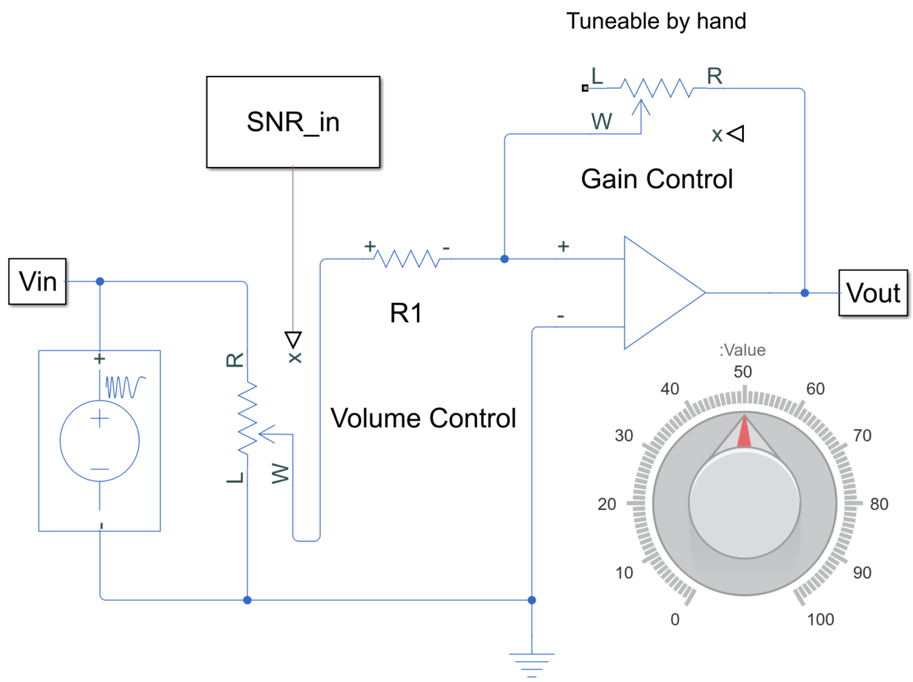


Figure 4.3: The Simulink Simulation Circuit with variable volume.

4.3. Integrated Circuit

The next step was to implement the simulation into hardware. This raised the choice between designing a transistor based circuit ourselves and using an Integrated Circuit (IC). The decision was made to use an IC in order to save time for the noise cancellation system. Also there is a wide range of high quality audio Op-Amps that will tackle our problem. Comparing multiple Operational Amplifiers was the first step in achieving this. An overview of the different Integrated Circuits compared is given in Table 4.1.

Table 4.1: Comparison of 3 different integrated Operational Amplifiers.

	LM741 [19]	**LM386** [20]	**LM358** [21]
Max supply voltage	$\pm 22\,\mathrm{V}$	$12\,\mathrm{V}$	$32\,\mathrm{V}$
Power dissipation	$100\,\mathrm{mW}$	$24\,\mathrm{mW}$	max $830\,\mathrm{mW}$
Typical application	general-purpose	audio amplification	general-purpose
Features	Overload Protection on In/Output	low distortion 0.2%	single or split voltage supply

In the end, the IC LM386 was chosen to use in the circuit, due to its low power consumption, low distortion and the fact that it has been optimised for audio signals. The only downside to this Operational Amplifier is the minimum gain of 20 , but this would make it possible to also use the designed system as a power amplifier if needed [20]. To tackle the high gain, the input signal will be lowered with a factor of *4.45*, resulting in a total circuit gain of *4.5*.

4.4. Implementation

Since the LM386 was chosen, first the minimal part circuit was built, which was taken from the data sheet [20]. A minimal part circuit is a circuit which uses the minimal number of parts in order to function. The values of the capacitors and resistors were chosen according to those available. This circuit turned out to be of poor quality, due to the coupling of the input and output grounds. Coupling of the input and output grounds leads to energy transfer between the two, which in turn leads to distortion in the output signal. Therefore the circuit had to be adjusted. The adjusted amplifier circuit design is shown in Figure 4.4.

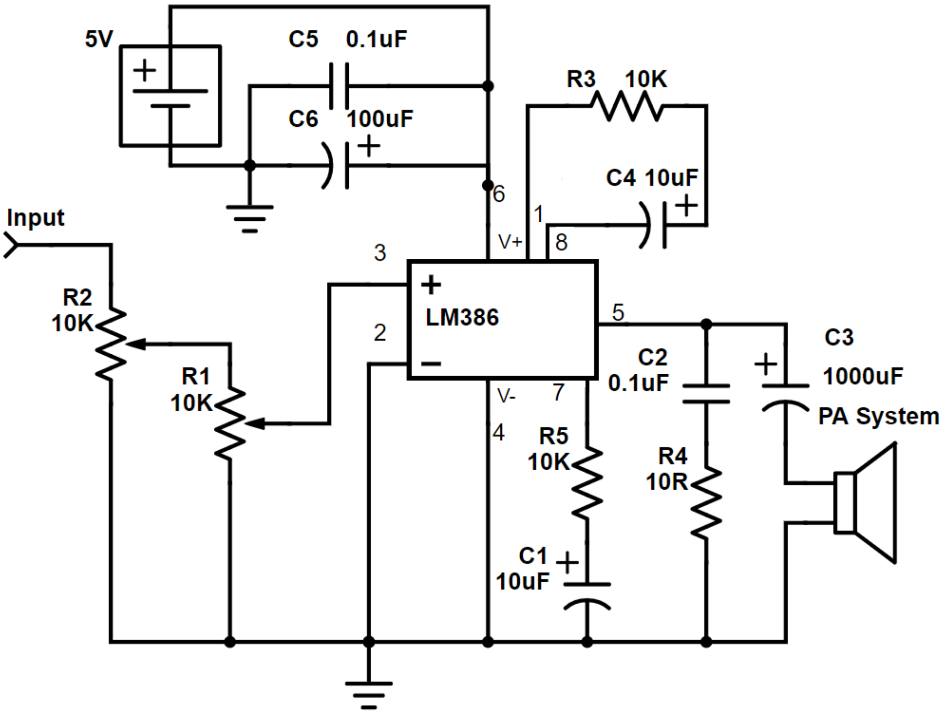


Figure 4.4: Amplifier circuit.

To decouple the input from the output, *R5* and *C1* are added. The value for the capacitor was taken from the data sheet of the LM386 [20, p. 6]. However, the resistor is added to tackle the possibility that the effective series resistance of the decoupling connection is too low, which might result in unwanted oscillations at the output. It might also be the case that this resistor is not needed, but it is placed just to get ahead of this problem.

R3 and *C4* are used to determine the gain. These are added in order to change this gain, which is achieved by changing the value of *R3*, if needed. This is described in the data sheet of the LM386 [20].

C5 and *C6* were used to decouple the power supply for both high and low frequencies respectively. The combination of *C2* and *R4* is called a Zobel Network. This was added to compensate for the impedance of a speaker "in order to minimize the effects of the voice coil's inductance" [22]. It also makes testing with speakers possible.

Lastly, *C3* was added in order to block Direct Current (DC) from making it to the PA System.

4.4.1. Development Board

To make the volume control automatic, the amplifier should have a Microcontroller Unit (MCU). This MCU will control a digital potentiometer to achieve the automatic volume control. The processor should:

- Calculate the resistance value for the volume control, based on the AF value.
- Use the calculated resistance value to control a digital potentiometer, Serial Peripheral Interface (SPI).

In Table 4.2 below, several development boards are compared to each other.

Table 4.2: Comparison of development boards.

Processor	Arduino Nano [23]	Raspberry Pi Zero [24, 25]	Nucleo-L432KC [26–28]
Clock	16 MHz	1 GHz	80 MHz
Memory	2 kB	512 MB	64 kB SRAM
Number of Ports	22 digital, 8 analog	40 pins	32 pins
Interface	Serial, SPI, I^2C	SPI, I^2C, Serial	Serial, I^2C, SPI
Operating Voltage	5 V	3.3 V	1.7 V to 3.6 V
Price	€20	€10	€14

By comparing these three development boards, the conclusion that the Raspberry Pi is overpowered can be made quickly. Not only does the device have a lot of compulsory memory, the clock-speed is also above necessary. Also, the Pi has to run an operating system, which has a long start-up time. This means that the Pi is not suitable for the application.

The Arduino Nano is a well-known development board used by many prototypers/developers. It has sufficient communication possibilities and has a lot of programming support. Comparing it with the Nucleo-L432KC, featuring a STM32L432KC MCU [29], it has a lower clock speed (so lower computational power) and less memory. Moreover, the Nucleo-L432KC, or another Microcontroller Unit from the same product family, can be integrated easily when finalising the project onto a Printed Circuit Board (PCB). This means that upgrading to an even faster MCU is really easy to do. Therefore the Nucleo-L432KC is the development board / MCU of our choice.

4.4.2. Volume Control Code

The chosen development board will map the Amplification Factor, calculated by the Intelligibility Enhancement group, to a resistance value of 0 to $10k\Omega$. This value will then be used to change the digital potentiometer value used for the volume control. The code for this mapping of the potentiometer was written in C++. The algorithm is rather simple, it compares the AF to the previous value. If the AF is found to be different (line 11) , the value of the potentiometer should be tuned to 100*AF (line 13 & 21). If it is the same (line 8) nothing needs to be done. The code can be found in Listing 4.1. This is a simplified version of the code only used to show how the value is calculated. The writing of the value to the digital potentiometer is simplified to line 21.

The Dual Digital Potentiometer used is the *DS1803-010*. This potentiometer can change between 0Ω to $10k\Omega$ [30] in 256 steps.

Listing 4.1: The simplified code for controlling the volume.

```
// Input Reading
 int newAF = 0;
 int currentAF = 0;
 int R = 5000;

 main() {
    while(true){
       if (newAF = currentAF){
          newAF = read('AF');
       }
       if (newAF != currentAF){
          // Calculate new value from the AF
          R = 100*newAF;
          // Make the old value of AF the new value
          newAF = currentAF;
          break;
       }
    }

    // Write the new value to the potentiometer
    write.Potentiometer(R);
 }
```

5

Noise Cancellation Design

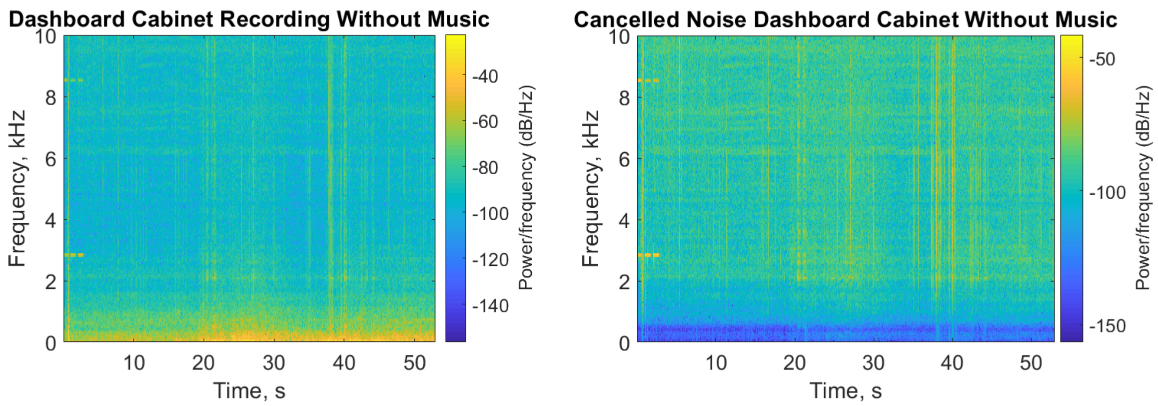
While driving in a car, one might experience low frequency noise induced by the tyres, engine or even wind [31]. This noise might cause annoyance, fatigue or even safety issues, for example not being able to hear sirens [9, 32]. To counter this, a noise cancellation addition is proposed to work together with the amplifier. In section 5.1 the analysis of the input needed for this cancellation is given, section 5.2 explains the choice of the microphone while section 5.3 goes through the design steps for the circuit.

5.1. Input Analysis

To efficiently design this noise-cancellation, the frequency behaviour of the noise should be known. Since the main implementation of this noise-cancellation will be inside a car, recordings were made at different locations in a Mitsubishi Spacestar while driving. These recordings were done multiple times, with and without music playing through the car's integrated speaker system. These recordings were then analysed using Matlab [33].

Tyre noise is divided into multiple frequency bands, up until 2 kHz [34]. Vehicle engine noise is concentrated in low frequencies [35], around 100 Hz. It has been found that the integrated car loudspeakers are capable of cancelling noise fields for frequencies up to 200 Hz [36] or even 500 Hz [37]. This means that the noise at the location of the driver has to be measured and filtered with a Low Pass Filter to only retain the frequencies under 500 Hz.

Results of one of the recordings in the car can be found in Figure 5.1a, the other 3 recordings can be seen in Appendix A. From these results, it can be seen that the power is mainly distributed between 20 Hz and 600 Hz. Therefore, during the analysis, noise was first considered as every sound below 500 Hz, this noise was filtered out, inverted and added to the recorded signal, the result of this can be found in Figure 5.1b.



(a) Unedited recording.

(b) Edited recording.

Figure 5.1: Recording of the sound in a car whilst driving without music.

5.1.1. Time Difference of Arrival

Tests were done to measure the time delay of a signal between two locations in the car; namely where the driver is and where the product will be. These tests were done in order to calculate the phase difference between the listener and the product. If this phase difference is known, the anti-noise phase can be adjusted accordingly. This adjustment will improve the cancellation capabilities by making sure the anti-noise reaches the listener at the same time as the real noise, thus being completely out of phase [38]. The time delay tests were done in the same car as the noise measurement described in section 5.1. The *time difference* test consisted of two pulse-trains, which were played after each other. There was a 10 s pause between pulse train *y1* and pulse train *y2*, so that the microphone could be moved to another place. Several tests have been carried out at several positions. The tests between the dashboard cabinet and the driver's head were the most important tests, since the product will most likely be placed there.

The result of one of these measurements can be seen in Figure 5.2. Note that this measurement was done while the engine was running, which is clearly visualised by the noise between the pulses.

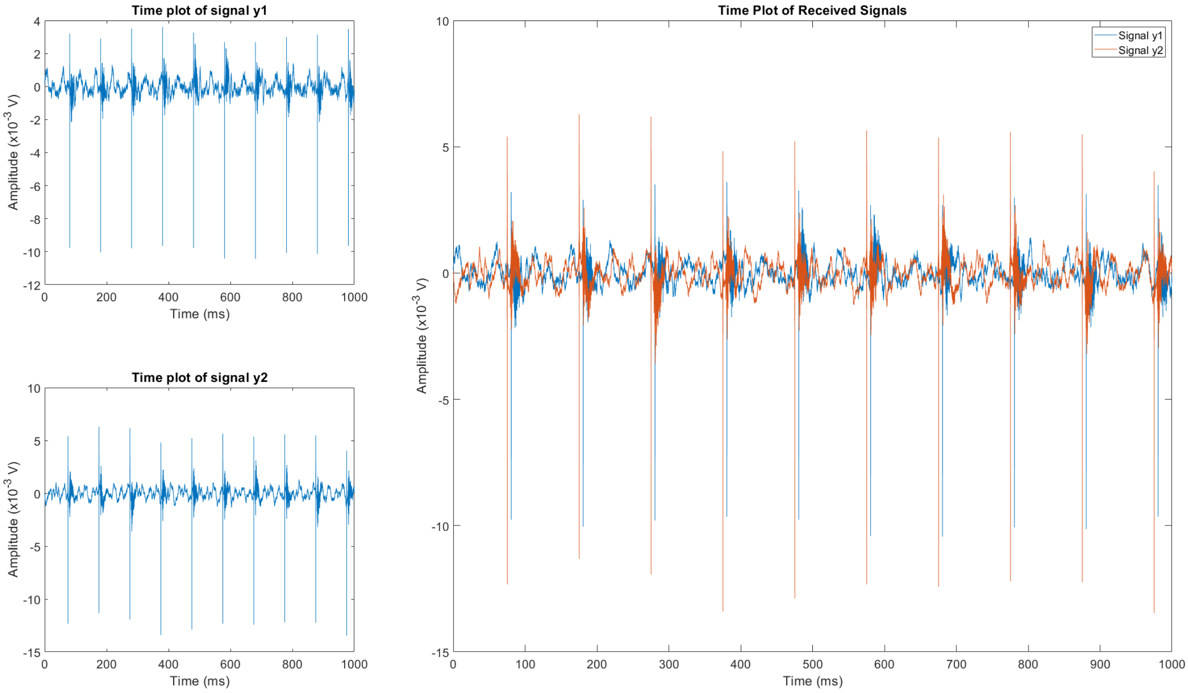


Figure 5.2: The graphs of the pulse trains at 2 different positions.

MATLAB [33] was used to process the signals. The time delay, the distance and the sample difference between the peaks of the pulse-train were calculated. From these values, the phase difference of the signal can be derived. The average time delay between the dashboard and the driver was found to be 3 ms. The signal strength is weaker at the drivers location. This is due to the fact that the driver is further away from the speaker. The dampening between the two locations is neglected for the initial implementation because it is only useful when it can be measured real time.

5.2. Microphone

Once this time delay was measured, the microphone had to be chosen. An electret microphone was chosen in order to implement it in the prototype, for their low cost and small size. This kind of microphone is also used in headsets, phones and recorders. Microphone outputs are of low amplitude, therefore its output should be amplified [4]. In Table 5.1, the comparison of two of these microphones is shown.

Table 5.1: Comparison of Microphones.

Microphone	**Ben's Electronics**	**KECG2740PBJ**
Sensitivity	-56 ± 2dB	-40 ± 3dB
Impedance	2200 Ω	2200 Ω
Operating Voltage	4.5 V to 10 V	2 V to 10 V
Max Current consumption	0.5 mA	0.5 mA
Sensitivity Reduction	−3 dB	−3 dB
Frequency Range	20 Hz to 20 kHz	20 Hz to 20 kHz
Diameter	9.3 mm	5.4 mm

As can be seen in Table 5.1, the microphones compared are nearly the same, the only difference is their sensitivity, diameter and operating voltage. The *KECG2740PBJ* has a higher sensitivity, which means that it will have a higher output level, but it might clip faster than the one from *Ben's Electronics*. The operating voltage range of both microphones are up to 10 V and therefore, with the 5 V power input, this would not cause any issues. The *KECG2740PBJ* is chosen for its higher sensitivity, so the output can be amplified with a smaller gain.

5.3. Circuit Design

Due to the fact that an amplifier circuit has been designed previously in chapter 4, a similar approach was taken and the LM386 was chosen again as the amplifier for microphone amplification. This IC was chosen because the functionality of the Integrated Circuit was already known. The circuit of the microphone amplifier can be seen in Figure 5.3. The schematics are based on the data sheet of the LM386 [20, Fig. 14].

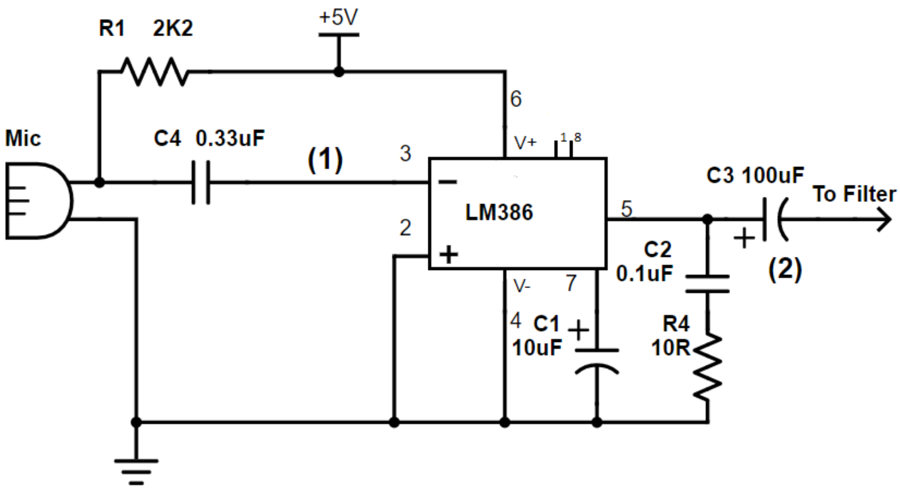


Figure 5.3: The LM386 Microphone Amplifier Schematics.

After the recorded audio signal, it is important to only retain the information below 500 Hz. Putting a Low Pass Filter in the circuit makes sure only the signals under 500 Hz are passed through. The filter can be placed in two places. The first place is before the Op-Amp, but after the power line of the microphone, noted by (1) in the circuit. The second place is all the way at the end of the circuit (2). The preference goes to place (2). This is because the signal is of sufficient strength to filter, but also to not unnecessarily amplify the noise of the added components.

5.3.1. Filter Design

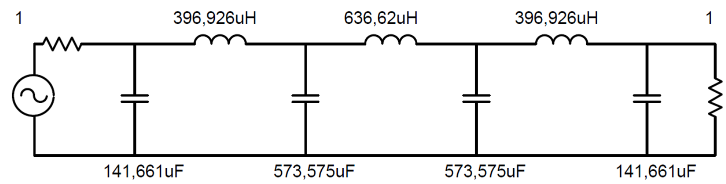
There are multiple ways to design filters. MATLAB [33] can be used, but there is also a (free) dedicated filter design software available. The filter designed in this section is designed with the help of this software, called Elsie [39] and later with MATLAB [33].

The filter that is needed has the following requirements:

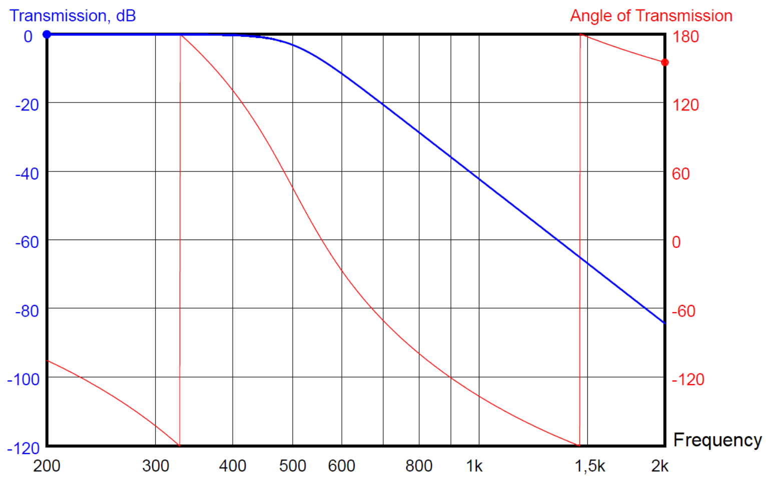
- Flat transfer- and phase-function below 500 Hz.
- Cut-off Frequency of 500 Hz.
- Stopband -12dB/decade.

The software Elsie [39] has an automatic design tool for filters. According to the requirements, the transfer function has to be as flat as possible below the cut-off frequency. This requirement is given in order to make sure the amplifier does not alter the frequency bands and intelligibility enhancing methods of the *Intelligibility Enhancement* group. A Butterworth filter is the most suited filter family because of it's flat pass-band characteristics. Chebyshev filters have an unwanted ripple present. When automatically designing a Butterworth LPF, with Elsie and order $N = 7$ (which was the highest order Elsie was capable of designing), the schematic and transfer function is as seen in Figure 5.4. Since the phase behaviour of the filter alters the phase of the anti-noise signal per frequency, the phase-behaviour of the filter should be as flat as possible.

This filter has a flat transfer function (noted in the figures by transmission) and the -3dB point is neatly around 500 Hz. However, there are a lot of components needed and the phase behaviour (noted in the figures by angle of transmission) is far from ideal.



(a) The circuit schematic.



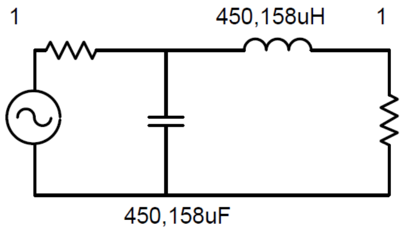
(b) The transfer and angle.

Figure 5.4: The Seventh Order Butterworth LPF created by Elsie.

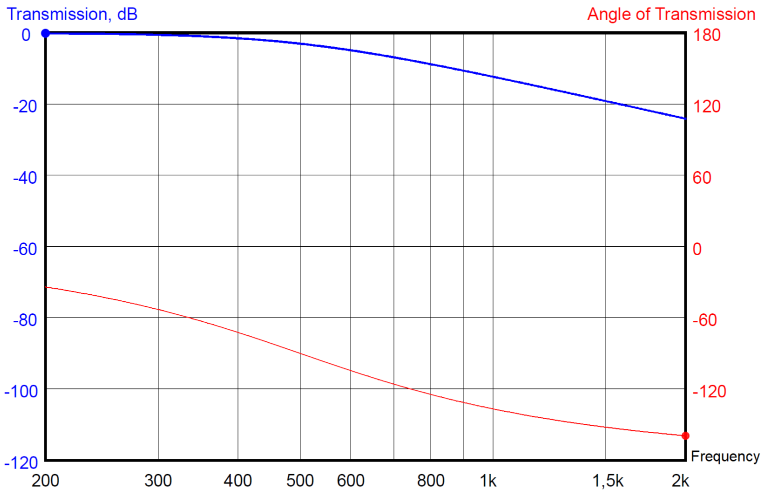
Tuning the filter back to order $N = 2$, the filter in Figure 5.5 is designed by Elsie. The stop-band slope of a Butterworth filter is determined by the order of the filter. The slope of the stop-band is $6 \cdot NdB/octave$ where N is of course the filter order.

It can be seen, comparing Figure 5.5b with Figure 5.4b, that the transmission plot is not that steep and also starts descending sooner. It should be noted that the filter still meets the requirements for the transfer function mentioned in the beginning of the section.

The phase behaviour is, however, a lot better. It is still not flat below 500 Hz but it is not as aggressive as in Figure 5.4b. There are also a lot less components needed to construct the filter.



(a) The circuit schematic.



(b) The transfer and angle.

Figure 5.5: The Second Order Butterworth LPF created by Elsie.

The results of the filter were not optimal, but due to time constraints, a better filter could not be designed. The components values Elsie calculated for Figure 5.5a were too specific. This means that some adjusting had to be done in order to be able to build the filter.

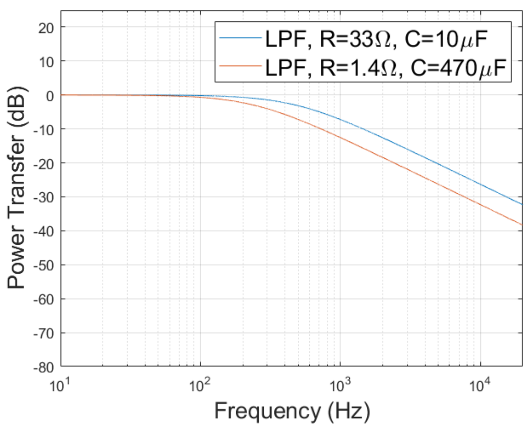
The filter was built with available component values, L= 0.44 mH and C= 0.47 mF, but it proved to be of no success. This was probably caused by insufficient knowledge of the input and output impedance of the filter. A first order ($N = 1$) RC LPF was introduced after consulting the manual from EPO-1 [40], where such a filter has been designed for another cut-off frequency, and evaluated through MATLAB [33].

When designing such a RC filter, the constant RC (note that R is the value of the resistor and C is the value of the capacitor) can be found by Equation 5.1b which follows from Equation 5.1a [17, p.638], when rewriting ω_c as $2 \cdot \pi \cdot f_c$, where f_c is the cut-off frequency.

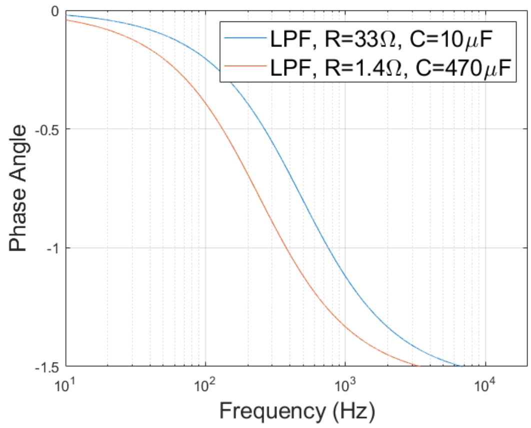
$$\omega_c = \frac{1}{RC}. \tag{5.1a}$$

$$RC = \frac{1}{2\pi f_c}. \tag{5.1b}$$

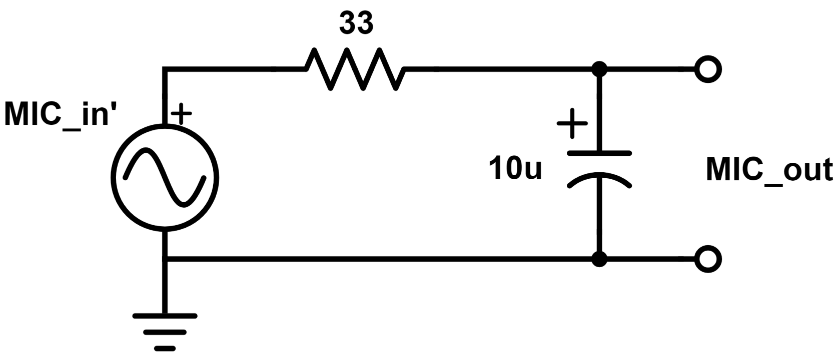
When using a cut-off frequency of 500 Hz, RC is found to be 318.3×10^{-6}. When comparing this ratio with the available component values, one configuration was found, namely a resistor value of 33Ω and a capacitor value of 10 µF. This configuration is only slightly different from the optimal configuration. The transfer function and phase behaviour of this configuration are compared to a configuration with a higher ratio. These comparisons are found in Figure 5.6a and Figure 5.6b respectively. As can be seen in these figures, the higher ratio LPF has a lower cut-off frequency and therefore will not suffice. The filter schematics from the used filter can be seen in Figure 5.6c. Although this filter is not ideal, the cut-off frequency (-3dB-point) is 481 Hz and almost has a flat transfer function in the pass-band.



(a) Transfer function for two possible configurations.



(b) Phase angle for two possible configurations.



(c) The circuit design.

Figure 5.6: RC Low Pass Filter design.

6

The Prototype

In this chapter, building, testing, adjusting and validating of the prototype are discussed. The tests are done according to the KPIs as described in chapter 2. To achieve realistic results, a signal generator was used to mimic the input signals, and an oscilloscope was used to measure the in- and outputs. The used oscilloscope is a *Tektronix TDS 2022C Two Channel Digital Storage Oscilloscope (200MHz, 2GS/s)* [41]. The used signal generator is a *Tektronix AFG3021C Single Channel Arbitrary Function Generator (250 MS/s, 25 MHZ)* [42].

First, the Pre-Amplifier and its characteristics will be presented in section 6.1. Then, in section 6.2 the results of the Microphone Amplifier are discussed. The combined circuitry has also been tested in section 6.3. Finally, in section 6.4, the noise cancelling capabilities are presented.

6.1. Pre-Amplifier

The transfer function of the Pre-Amplifier was tested with a sine wave input signal of 200 mV_{pp}. As can be seen in Figure 6.1, the transfer function is almost flat from 20 Hz to 20 kHz. The signal generator is able to generate signals from 10 Hz and higher. Therefore there is no value at $f = 0$ Hz. This can be seen by the incline in the beginning of Figure 6.1.

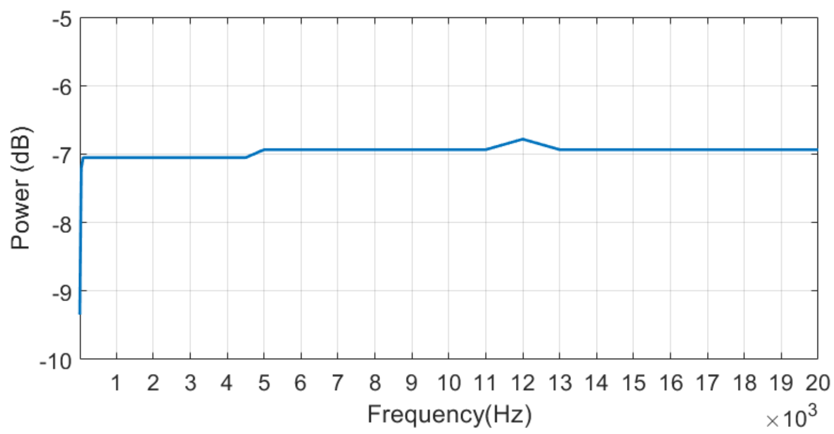
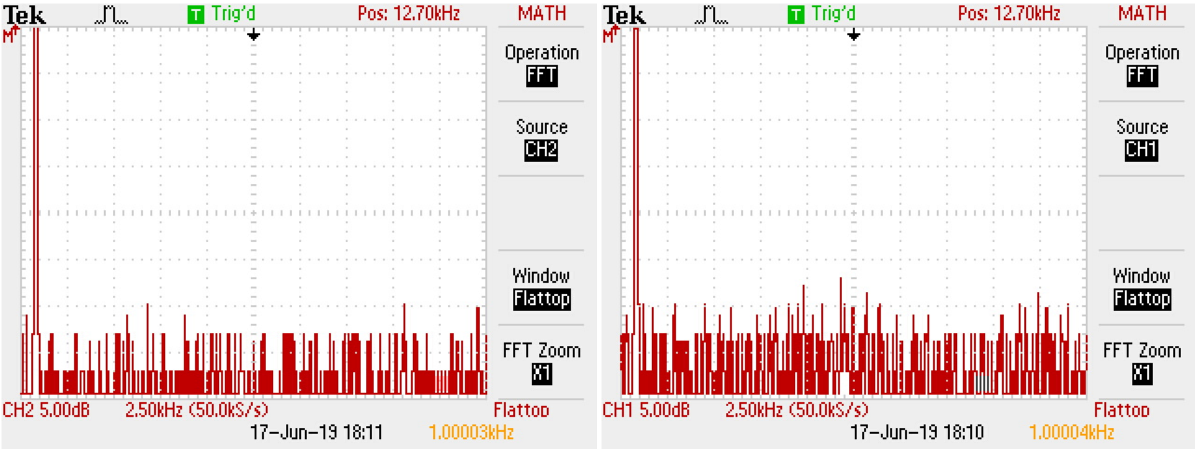


Figure 6.1: The Transfer Function of the Pre Amplifier.

This measurement was done when the amplifier was tuned for line level with an input of 200 mV_{pp}, as determined as average in section 4.1. In order to tune the amplifier, the value of potentiometer *R1* has to be changed. The peak-level of the output signal is, on average, -7 dB V, which corresponds to 447 mV_{max} (*line-level* [13]).

6.1.1. Noise Behaviour

The original plan was to test the amplifier with a Spectrum Analyzer for its noise behavior. However, after consulting *ing. X. van Rijnsoever*, the Spectrum Analyzers available were not capable of testing the frequency band in which the amplifier operates (20 Hz to 20 kHz). He then offered a solution which would make it possible to approximate the noise behaviour. Using the Fast-Fourier Transform (FFT), the power distribution before and after the amplifier could be measured. The results of this can be seen in Figure 6.2. The exact noise addition is not clear, but it can be seen that noise is added. The noise addition is estimated at 3 dB.



(a) Input FFT Pre-Amplifier with frequency range 0-20kHz.

(b) Output FFT Pre-Amplifier with frequency range 0-20kHz..

Figure 6.2: FFTs of the Pre-Amplifier implementation.

6.2. Microphone Amplifier

The transfer function of the microphone amplifier has also been tested. The test procedure was the same as in section 6.1, but the input signal was 13 mV_{pp} instead of 200 mV_{pp}. It can be seen in Figure 6.3 that the transfer declines when the frequency gets higher. This is because of the RC LPF at the end of the amplifier.

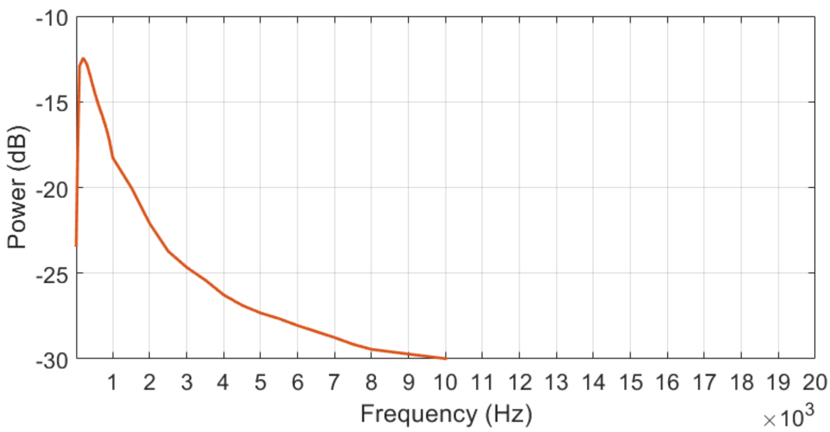


Figure 6.3: The Transfer Function of the Microphone Amplifier.

Unfortunately, we do not know the input signal amplitude of the microphone amplifier as stated in section 5.2. This means that there will be no fixed solution to make sure the output of the microphone amplifier has the same peak amplitude as the audio signal going in the pre-amplifier. The solution for this is a potentiometer in a voltage-divider configuration.

Here, the value of the division can be adjusted to the specific application of the system. The potentiometer used has a value of $10k\Omega$.

The output signal amplitude of the microphone can however be calculated. This involves taking the microphone sensitivity and converting it to mV Pa^{-1} [43]. The pressure excited on the microphone is however still unknown. This means that, because of an irregular sound input, the signal amplitude can not be calculated. The output level of the used microphone is 7 mV Pa^{-1} to 14 mV Pa^{-1} [43, 44].

The microphone amplifier should be inverting, to make sure the noise signal is added inverted to the pre-amplifier. When testing the amplifying circuit with the microphone attached, it turned out that the output would be very distorted when put in inverting mode. This is most probably due to the fact that the microphone needs a power supply, resistor and capacitor to work, as can be seen in Figure 5.3. Inverting the microphone output will therefore be done in a later stage of the product.

6.3. Combined Amplifier

In this section, the testing and validation of the combined circuitry, found in Figure C.1 is discussed.

6.3.1. Power Management

When connecting both the pre-amplifier and the microphone amplifier to a 5 V power supply, the circuitry drains 100 mA. Using $P = U \cdot I$ (Power = Voltage * Current), the power the circuits consume is 500 mW. Suppose that a generic 9 V battery has a capacity of 500 mAh. Assuming the voltage remains constant, the battery will last a minimal of 9 h.

When supplying the voltage regulator with a voltage of 5 V, the output of the regulator is 4.2 V. This is not ideal, since the minimal voltage needed to amplify (without clipping) is 6 V. The solution is to supply a minimal of 6 V to the voltage regulator.

6.3.2. Inverting Pre-Amplifier

Since the microphone amplifier does not invert the microphone signal, see section 6.2, the pre-amplifier is chosen to be the inverting stage. This means that both the enhanced audio signal and the microphone signal will be inverted. Since inverting the audio signal won't affect the quality/intelligibility and the microphone will only pick up noise, this approach is taken.

6.3.3. Combined Testing

In Figure 6.4 the test-setup for this section can be seen. There are two signal generators, one oscilloscope, a potentiometers and the two amplifier circuits from section 5.3 and section 4.4. Note that the figure is a simplified version of the real test-setup.

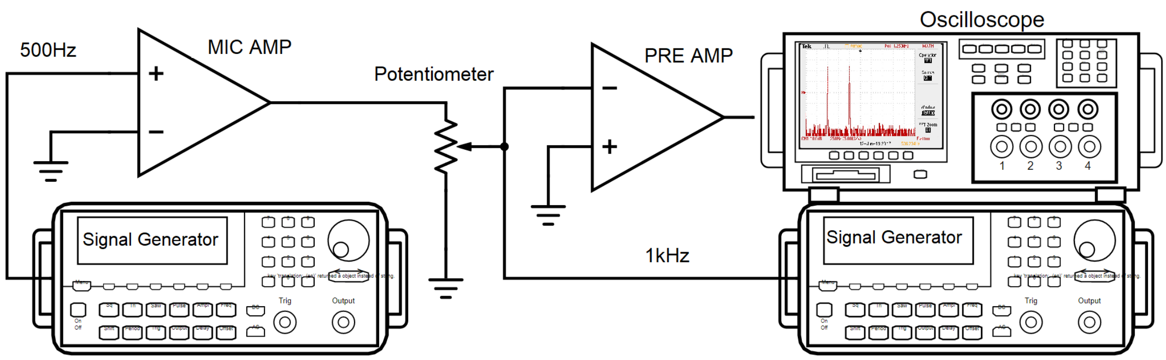


Figure 6.4: The Schematic Test Setup of the Combined Circuit.

Below in Figure 6.5 the Fast-Fourier Transform of the output of the system can be seen. The microphone input was generated by the signal generator, being 13 mV$_{pp}$, 500 Hz. The audio input was generated by an identical signal generator. This signal is 200 mV$_{pp}$, 1000 Hz. It can be seen that both signals are represented equally in the output signal.

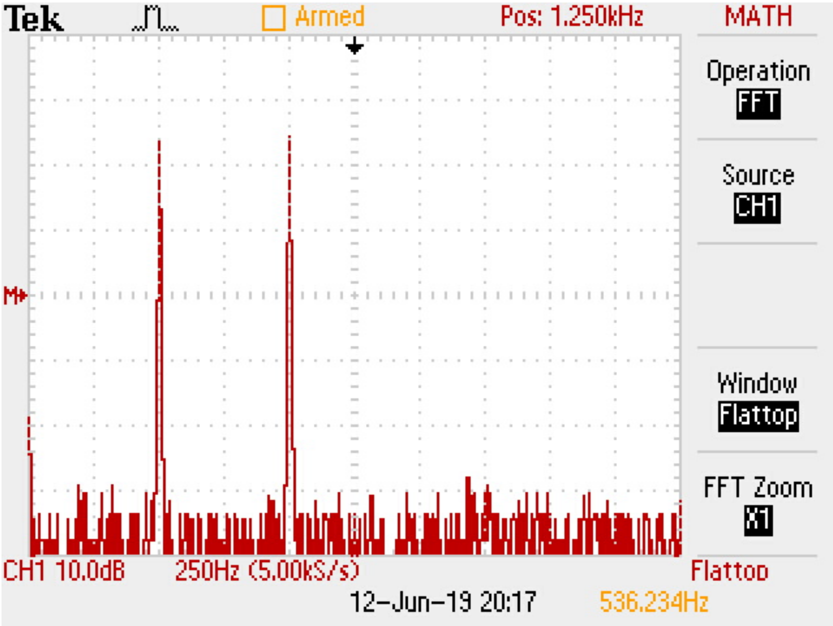


Figure 6.5: Fast-Fourier Transform of the output of the system.

6.3.4. Latency Testing

Using the delay function of the Tektronix Oscilloscope, the delay between two signals can be measured. On *CH1*, the output of the pre-amplifier was connected. The input of the amplifier was connected to *CH2*. The input is the same 1000 Hz 200 mV_{pp} sine wave as usual. The latency of the amplifier is 8 µs, as can be seen in Figure 6.6.

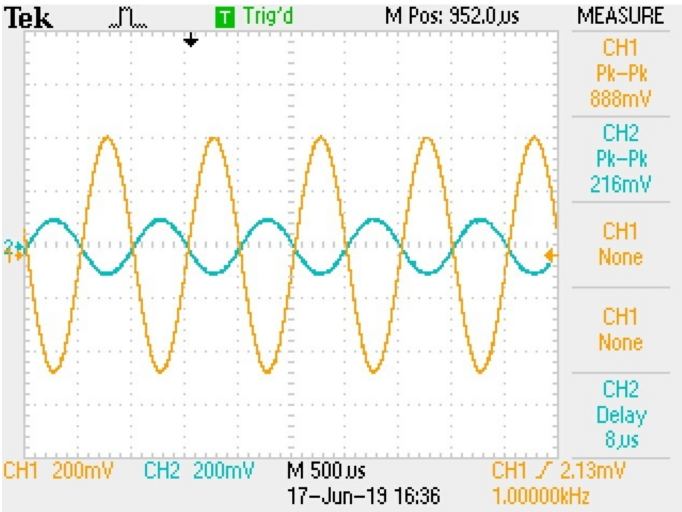
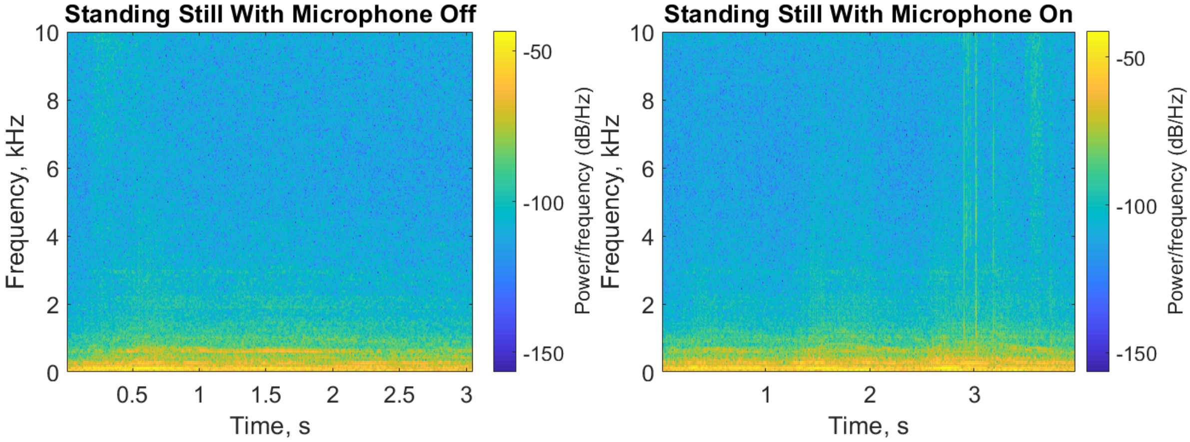


Figure 6.6: The input and output of the LM386 Pre-Amplifier. *CH1* is the output, *CH2* is the input.

Since the same chip was used in the microphone amplifier, the total delay of the system is two times the delay of the pre-amplifier. This means that the total delay is 16 µs. This was confirmed by testing.

6.4. Noise Cancelling

The noise cancelling feature was the last feature to test. Since the input analysis of this part was measured in a Mitsubishi Spacestar, this test was executed in that car as well. The same device was used for recording the difference with and without the noise cancelling feature. Once the prototype was connected, the engine was turned on. The recorder then recorded the audio with and without the prototype on, these recordings can be found in Figure 6.7. Afterwards, the car was driven around a parking lot, with and without the prototype turned on. The audio was recorded with music and the microphone on as well. Lastly, only driving with the audio system turned off completely was recorded for reference. Spectrograms of all the recordings can be found in Appendix B. It can be seen that the noise power distribution changes with the device on. The noise cancelling feature is only tested at the driver's head.



(a) Standing Still Without Cancelling.

(b) Standing Still With Cancelling.

Figure 6.7: Standing Still With Motor Running, Without Cancelling and With Cancelling.

7 Discussion

This chapter provides the discussion of the results described in chapter 6 compared to the requirements. According to chapter 2, the mandatory requirements of the Amplifier and Noise Cancellation system are:

1. The Pre-Amp must have a flat transfer function $H(f)$ from 20 Hz to 20 kHz.
2. The Pre-Amplifier must have a standardised line level of -10 dBV (= 447 mV$_{max}$ [13]).
3. The product must not introduce delay of more 10 ms.
4. The product must not introduce additional noise of more than 3 dBA.
5. The product must be able to work on 12 V.
6. The filters must not remove information bearing frequencies.
7. The system must be able to suppress near-end noise in the frequency band from 0 Hz to 500 Hz.
8. The product must fit into the dashboard cabinet of a car.
9. Both the input and output of the product should be connected with 3.5 mm jack-plugs.
10. The product must be plug-and-play, meaning that it can be used on any system.
11. The product must not infringe privacy.
12. The SNR must be improved by a minimal of 3 dB.

7.1. Pre-Amplifier

According to the results, the transfer function of the Pre-Amplifier meets Requirement 1, it has an almost flat transfer function from 20 Hz to 20 kHz. The amplifier is also able to amplify the input signal (200 mV$_{pp}$) to the line-level peak voltage (447 mV$_{max}$) and thus the amplifier also meets Requirement 2. The delay introduced by the amplifier was found to be 8 µs, which also ensures Requirement 3 is satisfied. Whether Requirement 4 is satisfied is not determined exact because this could not be measured using a spectral analyser. It is therefore approximated with the FFT-plots shown in subsection 6.1.1. Since the noise addition is not exactly quantifiable, Requirement 4 can not be tested. However, according to the approximation Requirement 4 is met.

7.2. Power Supply

The voltage regulator has only been tested for voltages from 5 V and higher since it could be easily tested with the power supply available. The situation when the input voltage switches rapidly is not tested. This for instance happens when the device is switched off. Also, a low dropout regulator is needed if the device has to work on 5 V from a USB port because then the voltage regulator output is 4.2 V, as stated in subsection 6.3.1. Since Requirement 5 states that the product must work on 12 V, this requirement is met.

7.3. Noise Cancellation

The filter used in the noise-cancellation should not remove information bearing frequencies, according to Requirement 6. Unfortunately, due to the RC filter used, limited time and limited resources, the current filter does not have a transfer function as intended. However the circuit does not remove information bearing frequencies and thus meets Requirement 6. For improvement, a different or higher order filter can be implemented. The noise cancellation subsystem is however able to suppress near-end noise in the frequency band from 0 Hz to 500 Hz, and thus satisfies Requirement 7. This was working in the test setup, where the signals were simulated by the signal generators.

7.4. Complete Prototype

The complete implemented prototype consists of a box with the following dimensions 112 mm x 160 mm x 62 mm *(LxWxH)*. This ensures that the product fits in a dashboard cabinet of a car, Requirement 8. In this box, there are two 3.5 mm jack-plugs, for the input and output audio, satisfying Requirement 9. Due to these jack-plugs, an audio device can be easily connected to the system, which in turn will be easily connected to an existing system, making it plug-and-play (Requirement 10). Since the prototype does not record any sound, except the car noise, Requirement 11 is also met. The prototype is also capable of playing music, even though the volume will not change automatically since that feature is developed according to the Amplification Factor which is only for the speech enhancement.

In the car test itself, fully noise cancelling was not achieved. Requirement 12 has to be tested more in order to get an exact value for the noise reduction, which can not be determined with the current testing methods. While listening and driving in the car, variance between turning the prototype on and off could be heard, however the reduction quality was not as high as expected. Some sounds were clearly more quiet, while some were a bit louder. Time-delay between multiple locations in the car were measured in order to see if a delay was needed for the noise-cancelling circuit, subsection 5.1.1. Since the delay was small, and time was running low, it was left out of the current prototype. However implementing this delay would further improve the cancellation, since the anti-noise is now not completely out of phase and thus the noise is not being dampened completely.

The Trade-Off Requirements discussed in chapter 2 are stated as follows:

13. The sound coming out of the product should not be distorted.
14. The subsystem should cost less than €80.
15. The product should have a power consumption of less than 10 W.
16. The system should have an option to play music.
17. The system should have a bass-boost option.

Requirement 13 is not easily tested, but when playing the signal through a speaker via the prototype, distortion could only be heard when the volume was turned way up above line level (measuring: 0.77 V_{max}). Since the product won't play above line level, the requirement is met. The cost of the hardware used for the subsystem came to a total of €70, so Requirement 14 is met, which is great for consumers. Requirement 15 is met with 500 mW, see subsection 6.3.1. The option to play music, Requirement 16 is also met, since the amplifier has a flat transfer function from 20 Hz to 20 kHz. The prototype does not have a bass-boost option, Requirement 17, but the LM386 does have the option for this. It was not implemented due to time constraints and due to the fact that the main purpose of the amplifier will be with speech and not music.

8

Conclusion

This chapter concludes the project. It summarises and reflects on the research, states improvements together with expansion possibilities. This thesis described the design of an *Amplifier and Noise Cancellation* system, to be implemented in an *Intelligibility-Enhancing Automatic Volume Control* System. The subsystems of the *Amplifier and Noise Cancellation* are designed, implemented and tested individually and as an entire system all together. It can be concluded that the designed system fulfils all the testable mandatory requirements. More testing, with other equipment, should be done in order to validate the other requirements.

Improvements

Sadly, not everything works as intended, yet. There are a few problems that could have been solved if more time was available. To improve the system, the filter of the noise-cancelling circuit can be upgraded for better response. Also, the phase behaviour of the system has to be measured real time and tuned to make sure the sounds that are going through the speakers are true anti-noise signals, meaning that there is a phase shift of $180°$ between the sounds that the car makes and the anti-noise sounds that come out of the speakers.

Furthermore, the noise behaviour of the system is not known precisely. If the noise behaviour of the system is known, the requirements would be better defined and validated.

Next, the signal analyser part of Figure 3.1 can be made automatically. This was the intention in the first place, but it would take a lot of time away. The focus lied on making a proof-of-concept. When the system is analysing it's own input signal, the volume of the product can be controlled automatically when packed in a stand-alone-package.

Expanding the product

To improve the response of the microphone, multiple microphones could be used to form a microphone array. This estimates the noise at different spots in the car, making sure a more optimal cancellation can be achieved. With this addition, the delay between the microphone and the listeners can be monitored and used for the cancellation. The company Bose [10], has implemented such a solution. In addition, they have accelerometers which measure vibrations in the car, but that is beyond the scope of this project.

When driving, the box where the system is located in vibrated and this was subsequently picked up by the microphone. There should be a dampening system, a layer of rubber for example, that makes sure the box does only pick up sounds coming from the engine, tyres and wind. This was also where the assumption of only picking up noise was based on.

The system now consists of multiple prototype soldered Printed Circuit Boards. This is not very robust and prone to breaking. That is why a custom made PCB is a very good idea. This makes sure the wiring of the system is always correct in addition to making the system a lot more reliable and robust. Furthermore, the system will decrease in size, meaning that there is more room for customers left in their dashboard cabinets.

List of Acronyms and Abbreviations

Notation	Description	Page List
AF	Amplification Factor	2, 6, 7, 10, 11, 13, 27
DC	Direct Current	12
FFT	Fast-Fourier Transform	22–24, 26
I^2C	Inter-Integrated Circuit	13
IC	Integrated Circuit	3, 9, 11, 17
KPI	Key Performance Indicator	4, 5, 21
LPF	Low Pass Filter	i, 7, 15, 17–20, 22
MCU	Microcontroller Unit	7, 8, 13
Op-Amp	Operational Amplifier	i, 9–11, 17
PA System	Public Address System	i, 1, 2, 4, 6, 9, 12
PCB	Printed Circuit Board	13, 28
PoR	Programme of Requirements	3, 4
SNR	Signal-to-Noise Ratio	2, 5, 6, 26
SPI	Serial Peripheral Interface	13
SRAM	Static Random Access Memory	13

Bibliography

[1] M. Rijkeboer and W. Sewnarain, "Noise statistics estimation based on adaptive filtering," BSc. thesis, TU Delft, The Netherlands, 2019.

[2] E. Riemens and B. Luppes, "On the enhancement of intelligibility," BSc. thesis, TU Delft, The Netherlands, 2019.

[3] M. A. Bee and C. Micheyl, "The "Cocktail Party Problem": What Is It? How Can It Be Solved? And Why Should Animal Behaviorists Study It?" *J Comp Psychol.*, vol. 122, no. 3, p. 235–251, Aug 2008. [Online]. Available: https://www.ncbi.nlm.nih.gov/pubmed/18729652

[4] Neumann, "Why do I need a preamp," 2019, published 24/04/2019. [Online]. Available: https://www.neumann.com/homestudio/en/why-do-i-need-a-preamp

[5] Kahrs, M., & Brandenburg, K. (Eds.)., *Applications of Digital Signal Processing to Audio and Acoustics*. Boston: Kluwer Academic Publishers, 2002, vol. 437. [Online]. Available: https://doi.org/10.1007/b117882

[6] G. Kidd Jr., "Enhancing Auditory Selective Attention Using a Visually Guided Hearing Aid," *J Speech Lang Hear Res.*, pp. 3027–3038, oct 2017. [Online]. Available: https://www.ncbi.nlm.nih.gov/pmc/articles/PMC5945072/

[7] F. R. SCOTT, *Signature*. McGill-Queen's University Press, 1964. [Online]. Available: http://www.jstor.org/stable/j.ctt1w6t8n2

[8] A. Tang, F. Yuan, and E. Law, "A New 2.4GHz CMOS Low-Noise Amplifier with Automatic Gain Control," in *2006 49th IEEE International Midwest Symposium on Circuits and Systems*, vol. 2, Aug 2006, pp. 200–203.

[9] R. Monaragala, "11 - knitted structures for sound absorption," in *Advances in Knitting Technology*, ser. Woodhead Publishing Series in Textiles, K. Au, Ed. Woodhead Publishing, 2011, pp. 262 – 286. [Online]. Available: http://www.sciencedirect.com/science/article/pii/B9781845693725500115

[10] R. S. Aouf, 2019, bose noise cancelling technology, published 29 January 2019. [Online]. Available: https://www.dezeen.com/2019/01/29/quietcomfort-bose-noise-cancelling-technology-cars/

[11] Silentium, 2017, information about noise cancelling products of Silentium. [Online]. Available: https://www.silentium.com/

[12] S. Elliott, "A review of active noise and vibration control in road vehicles," 2008. [Online]. Available: https://eprints.soton.ac.uk/65371/

[13] E. Winer, *The Audio Expert: Everything You Need to Know About Audio.* Focal Press, 2013, ISBN 978-0-240-82100-9.

[14] "Occupational Noise Exposure Regulation," Occupational Safety and Health Act, accessed: 2019-06-19. [Online]. Available: https://www.osha.gov/laws-regs/regulations/standardnumber/1910/1910.95

[15] M. Long, *Architectural Acoustics*, 2nd ed. Boston: Academic Press, 2014, ISBN 978-0-12-398258-2.

[16] *L78 Positive voltage regulator ICs*, ST Microelectronics, November 2016, l7805CV.

[17] C. K. Alexander and M. N. Sadiku, *Fundamentals of Electric Circuits*. McGraw Hill Education, 2013, International Edition, ISBN 978-1-259-07139-3.

[18] "MATLAB - Simulink," R2019a, The MathWorks, Natick, MA, USA.

[19] *LM741 Operational Amplifier*, Texas Instruments, May 1998, rev. October 2015.

[20] *LM386 Low Voltage Audio Power Amplifier*, Texas Instruments, May 2014, rev. May 2017.

[21] *LMx58-N Low-Power Dual-Operational Amplifiers*, Texas Instruments, Jan. 2000, rev. Dec 2014.

[22] W. S. Gontowski, "IC Audio Power Amplifiers and Zobel Networks: One Size Does Not Fit All," ST Microlectronics. [Online]. Available: https://www.eetimes.com/document.asp?doc_id=1274906#

[23] "Arduino Nano," 2019, Retrieved 28-05-2019. [Online]. Available: https://store.arduino.cc/arduino-nano

[24] "Raspberry Pi Zero," 2019, Retrieved 28-05-2019. [Online]. Available: https://www.raspberrypi.org/products/raspberry-pi-zero/

[25] "BCM2835 Datasheet," 2012, Retrieved 28-05-2019, errata on https://elinux.org/BCM2835_datasheet_errata. [Online]. Available: https://www.raspberrypi.org/documentation/hardware/raspberrypi/bcm2835/BCM2835-ARM-Peripherals.pdf

[26] "NUCLEO-L432KC," ST Microelectronics. [Online]. Available: https://www.st.com/content/st_com/en/products/evaluation-tools/product-evaluation-tools/mcu-mpu-eval-tools/stm32-mcu-mpu-eval-tools/stm32-nucleo-boards/nucleo-l432kc.html

[27] "Data brief - NUCLEO-XXXXKX - STM32 Nucleo-32 boards," ST Microelectronics, 2019. [Online]. Available: https://www.st.com/resource/en/data_brief/nucleo-l432kc.pdf

[28] *STM32 Nucleo-32 boards (MB1180) - User manual*, ST Microelectronics, 2018. [Online]. Available: https://www.st.com/content/ccc/resource/technical/document/user_manual/e3/0e/88/05/e8/74/43/a0/DM00231744.pdf/files/DM00231744.pdf/jcr:content/translations/en.DM00231744.pdf

[29] *Ultra-low-power Arm Cortex-M4 32-bit MCU+FPU, 100DMIPS, up to 256KB Flash, 64KB SRAM, USB FS, analog, audio*, ST Microelectronics, May 2018, rev. 4.

[30] *DS1803 Addressable Dual Digital Potentiometer*, Maxim Integrated, accessed on: 15-5-2019. [Online]. Available: https://datasheets.maximintegrated.com/en/ds/DS1803.pdf

[31] D. Vigé, "13 - vehicle interior noise refinement – cabin sound package design and development," in *Vehicle Noise and Vibration Refinement*, X. Wang, Ed. Woodhead Publishing, 2010, pp. 286 – 317. [Online]. Available: http://www.sciencedirect.com/science/article/pii/B9781845694975500133

[32] Mark A.W. Andrews, "Ask the brains," *SA Mind*, vol. 20, no. 7, p. 74, January 2010.

[33] "MATLAB," R2019a, The MathWorks, Natick, MA, USA.

[34] K. Iwao and I. Yamazaki, "A study on the mechanism of tire/road noise," *JSAE Review*, vol. 17, no. 2, pp. 139 – 144, 1996. [Online]. Available: http://www.sciencedirect.com/science/article/pii/0389430495000046

[35] N. Krishnamurthy and J. H. L. Hansen, "Car noise verification and applications," *International Journal of Speech Technology*, vol. 17, no. 2, pp. 167–181, Jun 2014. [Online]. Available: https://doi.org/10.1007/s10772-013-9215-3

[36] H. Chen, P. Samarasinghe, and T. D. Abhayapala, "In-car noise field analysis and multi-zone noise cancellation quality estimation," in *2015 Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA)*, Dec 2015, pp. 773–778.

[37] H. Chen, P. Samarasinghe, T. D. Abhayapala, and W. Zhang, "Spatial noise cancellation inside cars: Performance analysis and experimental results," in *2015 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Oct 2015, pp. 1–5.

[38] R. Wolfson, *Essential University Physics*, 2nd ed. Edinburgh Gate, Harlow, Essex, CM20 2JE: Pearson Education Limited, 2015, vol. 1, International Edition, ISBN: 978-1-78448-045-5.

[39] "Elsie," 2003-2018, An uncommonly easy-to-use electrical filter design and analysis program written by engineers for engineers, Student edition, V2.84. [Online]. Available: http://www.tonnesoftware.com/elsie.html

[40] G. Janssen, J. Creemer, D. Djairam, M. Gibescu, I. Lager, N. van der Meijs, S. Vollebregt, R. Roodenburg, J. Hoekstra, and X. van Rijnsoever, *Lab Courses EE Semester 1 - Student Manual. Course Labs of EE1C11, EE1P11, and EE1M11*, 2015-2016.

[41] *TDS2000C and TDS1000C-EDU Series Digital Storage Oscilloscopes*, Tektronix. [Online]. Available: https://images-na.ssl-images-amazon.com/images/I/B12oT7ls1sS.pdf

[42] *Arbitrary/Function Generators AFG3000C Series Datasheet*, Tektronix. [Online]. Available: https://www.tek.com/datasheet/afg3000c-arbitrary-function-generator-datasheet

[43] J. Eargle, *The Microphone Book*. Elsevier, Burlington, 2005, ISBN 02405-1961-2.

[44] *Electret Condenser Microphone*, Kingstate Electronics Corp, October 2004.

A
Car Sound Recordings For Input Analysis

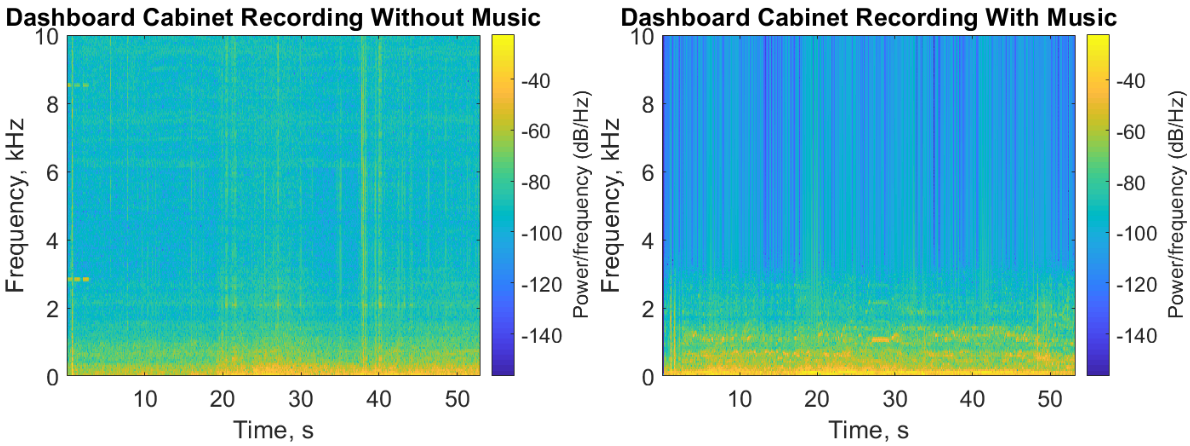


Figure A.1: Dashboard Cabinet Recordings Without And With Music Playing.

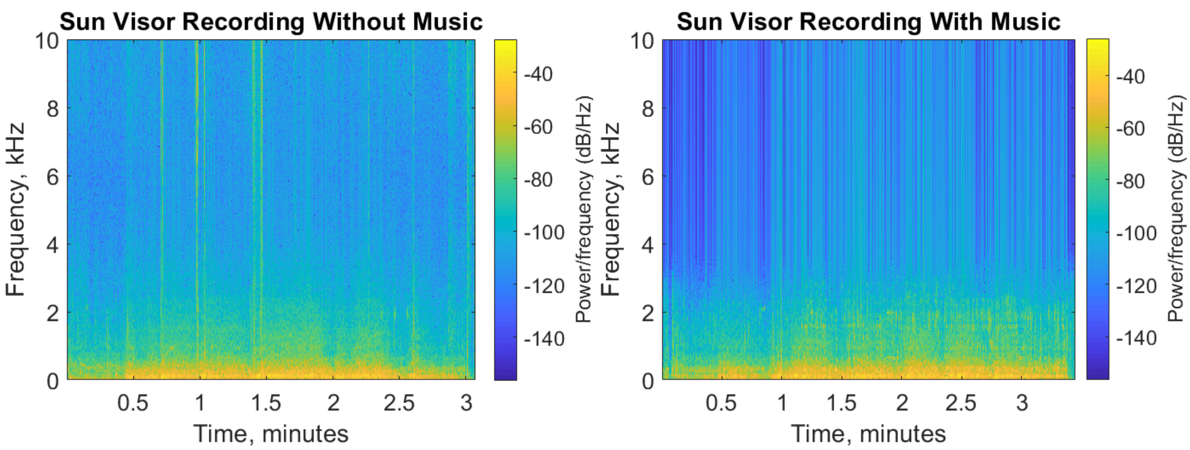


Figure A.2: Sun Visor Recordings Without And With Music Playing.

B

Car Sound Recordings For Noise Cancellation

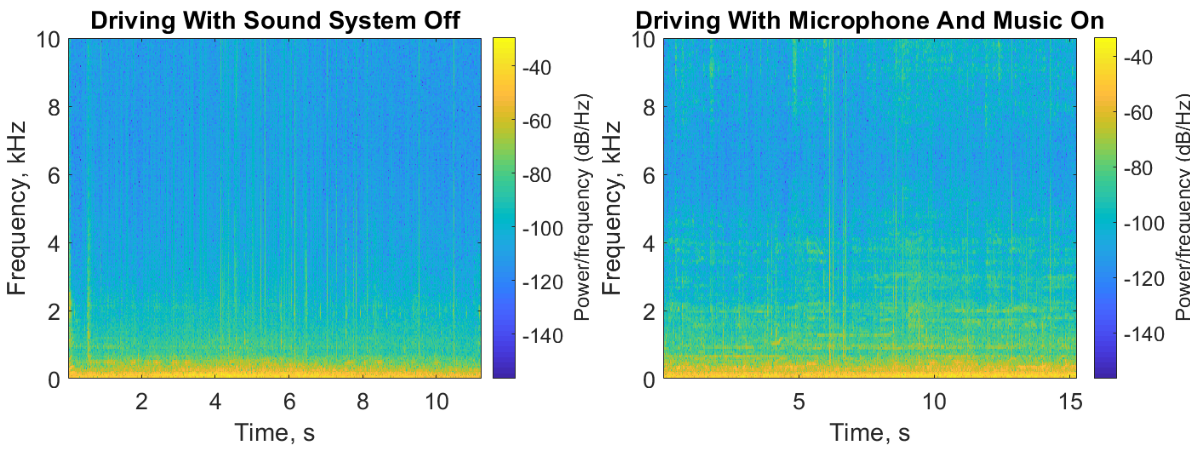


Figure B.1: Driving With Sound System Off And Music On respectively.

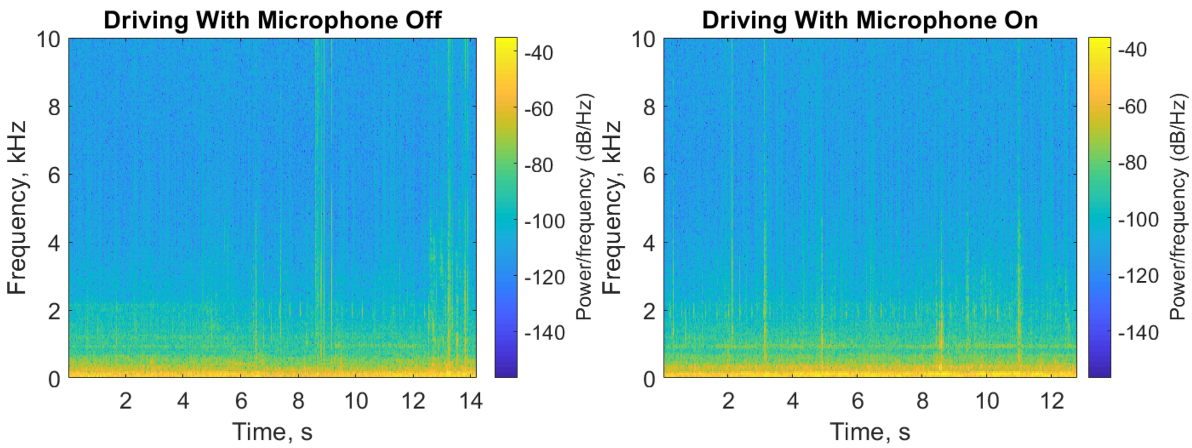


Figure B.2: Driving Without Cancelling and With Cancelling.

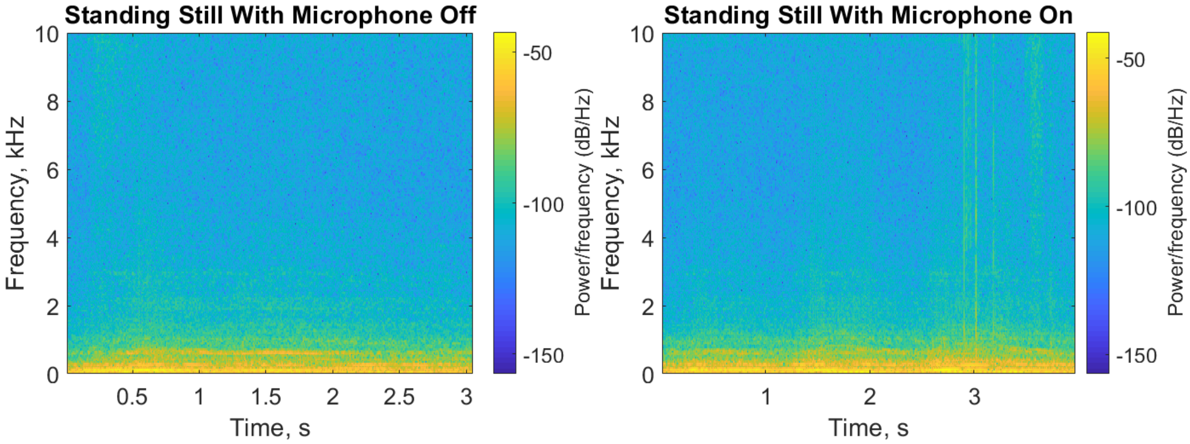


Figure B.3: Standing Still With Motor Running, Without Cancelling and With Cancelling.

C

Total Circuit

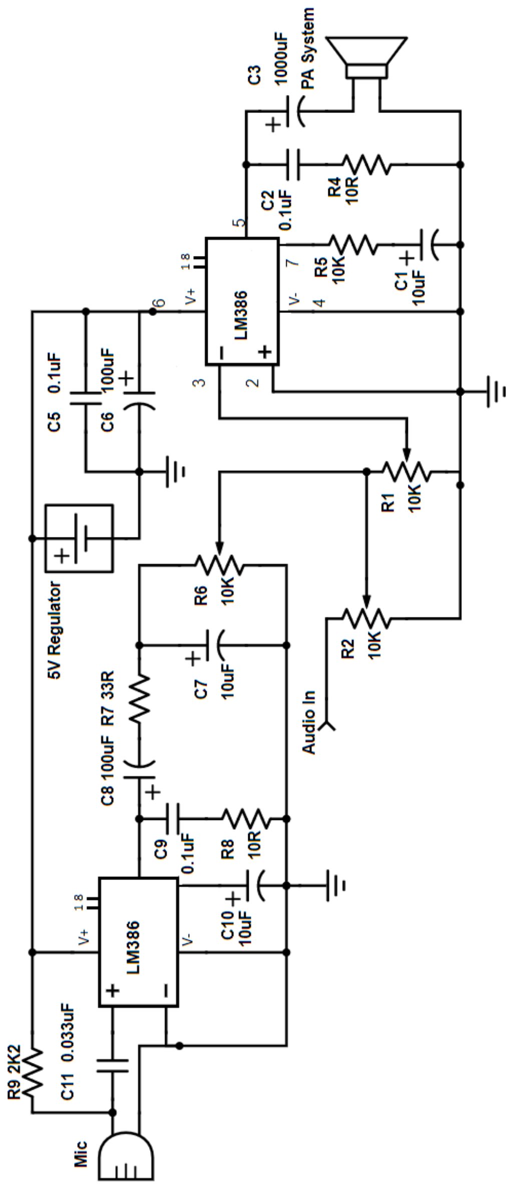


Figure C.1: Total Circuit Of The Prototype.