

Design of an Optimal Analogue Microphone System for Best Possible Capture of Incoming Acoustic Signals

Johan Sunnanväder
tfy11jsu@student.lu.se
audio@SunandWeather.com

Department of Electrical and Information Technology
Lund University
&
Axis Communications AB

Supervisor: Johan Wernehag

Assistant Supervisor: Henrik Dunér

Examiner: Pietro Andreani

June 1, 2016

Printed in Sweden
E-huset, Lund, 2016

Abstract

This project examines how to achieve the best possible capture of incoming acoustic signals in a surveillance camera without overspending. This will be done by designing a complete analogue microphone system to be integrated into a surveillance camera and identify the critical stages in terms of performance. The project will cover the complete analogue signal chain from incoming acoustic signals to analogue-to-digital conversion which covers the following steps: Evaluation of mechanical designs to eliminate undesired acoustic effects, choice of an optimal microphone, design of analogue signal processing such as amplification, filtering and signal level adjustments, evaluation of an optimal signal path for minimal interference and evaluation of an analogue-to-digital converter.

A complete microphone system has been successfully implemented with the desired performance in terms of frequency response, distortion, noise and dynamic range. This has been done by identifying the limiting factors and adapting the performance of the surrounding components accordingly.

Acknowledgements

I would like to thank my supervisor at Axis, Henrik Dunér, for sharing his experience in audio design as well as enthusiasm for the project. Thanks, to Anders Svensson for the great reception at Axis and to Simon Christensson for additional support. Also, a big thanks to everyone at Axis who have been willing to help and share their skills and knowledge.

I would also like to thank my supervisor at LTH, Johan Wernehag, for guidance and helpful comments.

Thanks, to Lars Nilsson of LBN LJUDTEKNIK for additional consultation.

Table of Contents

1	Introduction	1
1.1	Motivation	1
1.2	Acoustics	1
1.3	Microphone	2
1.4	Analogue Signal Processing	2
1.5	Analog to Digital Conversion	2
1.6	Power Supply	2
1.7	Complete Microphone System	2
2	Acoustics	3
2.1	Theory	3
2.1.1	Quality Factor	4
2.1.2	Helmholtz Resonator	4
2.2	Theoretical Limit	4
2.3	Conceptual Tests	5
2.3.1	Straight Tube	5
2.3.2	Funneled Tubes	6
2.3.3	Helmholtz Resonator	6
2.4	Conclusion	7
3	Microphone	9
3.1	Theory	9
3.1.1	Electret Microphones	9
3.1.2	Microelectromechanical Systems (MEMS) Microphones	10
3.1.3	Stokes' Law of Sound Attenuation	10
3.2	Microphone Specifications	11
3.2.1	Sensitivity	11
3.2.2	Directionality	11
3.2.3	Signal-to-Noise Ratio (SNR)	11
3.2.4	Equivalent Input Noise (EIN)	12
3.2.5	Frequency Response	12
3.2.6	Total Harmonic Distortion (THD)	12
3.2.7	Power Supply Rejection (PSR) and Power Supply Rejection Ratio (PSRR)	12
3.2.8	Acoustic Overload Point (AOP)	13

3.2.9	Dynamic Range	13
3.3	Microphone Measurements	13
3.3.1	Frequency Response	13
3.3.2	Sensitivity and SNR Measurements	13
3.4	Choice of Microphone	14
4	Analogue Signal Processing _____	17
4.1	Theory	18
4.1.1	Frii's Formula	18
4.1.2	Ohm's law	18
4.1.3	Noise Calculations	18
4.2	Operational Amplifier Specifications	19
4.2.1	Temperature Dependence	19
4.2.2	Phase Margin	19
4.2.3	Unity Gain Bandwidth	19
4.2.4	Common Mode Rejection Ratio (CMRR)	20
4.2.5	Noise	20
4.2.6	Input Bias Current	21
4.2.7	Input Offset Current	21
4.2.8	Supply Voltage Sensitivity	21
4.2.9	Total Power Dissipation	22
4.2.10	Input and Output Impedance	22
4.2.11	Slew Rate	22
4.2.12	Maximum Input Voltage and Maximum Output Voltage	23
4.2.13	Minimum and Maximum Operating Voltage	23
4.2.14	Crosstalk Attenuation	23
4.2.15	Total Harmonic Distortion (THD)	23
4.3	Choice of OP Amplifiers	23
4.3.1	Main Amplifier	23
4.3.2	Filter	24
4.3.3	Line Driver	24
4.3.4	Chosen OP Amplifiers	24
5	Main Amplifier _____	27
5.1	Theory	27
5.1.1	Amplifier Configurations	27
5.1.2	Noise	29
5.1.3	Filter	30
5.2	Amplifier Configuration	31
5.3	Connection Circuit for Main Amplifier	32
5.4	Calculations	33
5.4.1	Noise Performance	33
5.4.2	Total Noise Performance of Main Amplifier	38
5.4.3	Frequency Response	41
5.5	Simulation	42
5.5.1	Noise	42
5.5.2	Frequency Response	43

5.5.3	CMRR	43
5.6	Implementation	44
5.7	Measurements	44
5.8	Evaluation	45
5.8.1	Possible Improvements	47
6	Voltage Regulation of Power Supply	49
6.1	Initial Measurements	49
6.2	Noise Reduction	50
6.3	Calculations	51
6.3.1	Noise	51
6.3.2	Simulations	52
6.4	Measurements	53
6.4.1	Noise	53
6.5	Evaluation	54
6.5.1	Possible Improvements	54
7	Filtering	55
7.1	Theory	55
7.1.1	Filter Orders	56
7.1.2	Passive/Active Filters	56
7.1.3	Filter Configurations	56
7.1.4	Active Filter Topology	57
7.2	Shelf Filter	57
7.2.1	Calculations	59
7.2.2	Simulations	59
7.2.3	Measurements	60
7.3	Anti Aliasing Filter	61
7.3.1	Calculations	61
7.3.2	Simulations	61
7.3.3	Measurements	62
7.4	Evaluation	64
7.4.1	Possible Improvements	64
8	Signal Transport	65
8.1	Theory	65
8.1.1	Differential Signal Transport	65
8.1.2	Parasitic Effects of transmission lines	66
8.2	Line Driver	66
8.3	Calculations	66
8.4	Measurements	67
8.5	Evaluation	68
8.5.1	Possible Improvements	68
9	Signal Level Adaption	69
9.1	Theory	69
9.2	Resistive Pad	70

9.3	Calculations	70
9.4	Measurements	70
9.5	Evaluation	71
10	Complete Design _____	73
10.1	Measurements	73
10.1.1	Complete Electronics	73
10.1.2	Complete Electronics with Microphone	73
10.1.3	Complete Microphone System	75
10.2	Evaluation	76
10.2.1	Possible Improvements	78
11	Conclusion _____	79
11.0.1	Microphone	79
11.0.2	OP Amplifiers	79
11.0.3	Acoustics	80
11.0.4	Analogue Signal Processing Circuits	80
11.0.5	Codec	81
11.1	Complete Microphone System	81
11.1.1	Cost	83
11.2	Possible Improvements	83
11.2.1	Measurements	83
12	Final Remarks _____	85
	References _____	87

List of Figures

1.1	Block diagram of the audio chain.	1
2.1	Block diagram of the audio chain with the acoustics highlighted. . .	3
2.2	The frequency response of the calibrated reference microphone . . .	5
2.3	The frequency response of the calibrated reference microphone with a closed tube of length $l = 66,62 \text{ mm}$ placed in front of it.	6
2.4	The frequency response of the calibrated reference microphone with a closed tube of length $l = 66,62 \text{ mm}$ placed in front of it and a Helmholtz resonator placed at the middle of the tube.	7
3.1	Block diagram of the audio chain with the microphone highlighted. .	9
3.2	Schematic of an electret microphone.	10
3.3	Frequency response of the PUI Audio POM-3535L-3-R	14
3.4	Frequency response of the InvenSense ICS-40720 microphone.	15
4.1	Block diagram of the audio chain with the analogue signal processing electronics highlighted.	17
4.2	The noise of a standalone OP amplifier represented by an input voltage.	20
5.1	Block diagram of the audio chain with the main amplifier highlighted.	27
5.2	A non-inverting amplifier configuration	28
5.3	An inverting amplifier configuration	28
5.4	A differential amplifier configuration	29
5.5	A low pass filter realised with a single ended RC circuit	30
5.6	A low pass filter realised with a differential RC circuit	31
5.7	The instrumentation amplifier configuration to be used as the main amplifier.	32
5.8	The connection circuit for the main amplifier.	33
5.9	Schematic of the non-inverter amplifier with represented noise sources.	34
5.10	Schematic of the differential amplifier with represented noise sources.	36
5.11	The SNR of the main amplifier as a function of its gain.	39
5.12	The SNR degradation caused by the main amplifier as a function of the gain.	40

5.13	The SNR degradation caused by the main amplifier as a function of the OP amplifier voltage noise with a gain of 30 dB and an OP amplifier current noise fixed to 1 pA/√Hz.	41
5.14	The noise density of the main amplifier as a function of the frequency.	42
5.15	The simulated frequency response of the main amplifier from 1 mHz to 1 MHz.	43
5.16	The PCB layout of the main amplifier.	44
5.17	The measured frequency response of the main amplifier from 20 Hz to 20 kHz.	46
5.18	The measured frequency response of the main amplifier with microphone, inside the camera.	47
6.1	Block diagram of the audio chain with the voltage source highlighted.	49
6.2	The bias circuit schematic.	50
6.3	The bias circuit equivalent with noise sources.	51
6.4	The noise power from the voltage source on the bias voltage for different capacitor values.	52
6.5	The noise power from the voltage source on the bias voltage.	53
7.1	Block diagram of the audio chain with the filters highlighted.	55
7.2	An active second order low pass filter using the Sallen-Key topology.	58
7.3	A low pass shelf filter.	58
7.4	The frequency response of the shelf filter.	59
7.5	The measured frequency response of shelf filter.	60
7.6	The frequency response of the anti aliasing filter.	62
7.7	The measured frequency response of the anti aliasing filter.	63
8.1	Block diagram of the audio chain with the signal transport circuits highlighted.	65
8.2	Schematic of the line driver.	66
8.3	Measured frequency response of the line driver.	67
9.1	Block diagram of the audio chain with the signal attenuation circuit highlighted.	69
9.2	A voltage divider circuit.	70
10.1	The measured frequency response of complete electronics chain.	74
10.2	The measured frequency response of complete electronics chain with microphone, inside the camera.	75
10.3	The measured frequency response of complete microphone system.	76
11.1	The measured frequency response of the microphone as a standalone component, microphone inside the camera and after filtering.	82

List of Tables

3.1	Sensitivity and SNR for the MEMS (InvenSense) and the electret (PUI Audio) microphone.	14
5.1	Measured specifications for the main amplifier.	40
5.2	Measured specifications for the main amplifier.	45
5.3	Measured specifications for the main amplifier with the microphone, inside the camera.	45
6.1	Measured stability and noise of the power supply.	50
6.2	Noise and SNR degradation of the signal depending on the capacitor value.	53
7.1	Measured specifications for the anti aliasing filter.	62
8.1	Measured specifications for the line driver.	67
9.1	Measured specifications for the resistive pad.	71
10.1	Measured specifications for the complete electronics chain.	73
10.2	Measured specifications for the complete electronics chain with microphone inside the camera.	74
10.3	Measured specifications for the complete microphone system.	75
10.4	The maximum distance from the microphone system to the sound source until the received signal level equals the noise level.	76

The goal of this project is to achieve the best possible capture of incoming acoustic signals in a surveillance camera at the best possible price. In order to do so the whole analogue chain from incoming acoustic signals to the analogue-to-digital converter (AD converter) will be covered. All the steps is represented in the block diagram in figure 1.1.

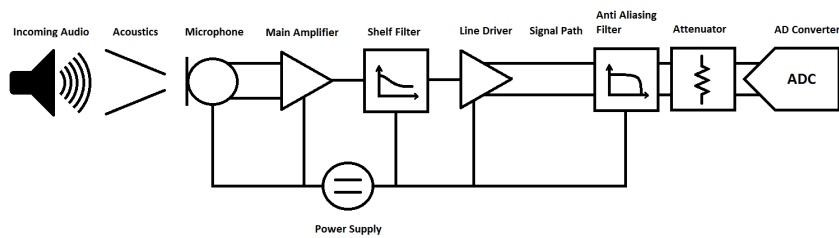


Figure 1.1: Block diagram of the audio chain.

1.1 Motivation

In the audio field, Axis has focused mainly on optimizing the audio electronics for line in and external microphones. This project is the first investment to achieve the highest possible audio quality from an internal microphone of the camera. This would make it possible to listen to sounds further away from the camera, easier to interpret sounds such as speech and would make it possible for audio detection algorithms to analyse the sound, without any external equipment.

1.2 Acoustics

A bad mechanical design can have a devastating effect on the sound quality. A poorly designed acoustic signal path can give rise to standing waves that can cause the frequency response to be heavily distorted. In this project different designs to eliminate the acoustic effects will be evaluated.

1.3 Microphone

The most crucial component in the system is the microphone. It will set the absolute noise floor of the system as well as the dynamic range. The following signal processing circuits will be designed according to the microphone to preserve the sound quality all the way to the AD converter.

Different microphone types will be tested and evaluated to be able to choose the best microphone for the project.

1.4 Analogue Signal Processing

In order to amplify the signal to a desired voltage swing, achieving the desired frequency response and transporting the signal to the AD converter (codec) a number of signal processing circuits will be required: a main amplifier to get the desired voltage swing, a number of filters to get the desired frequency response and a line driver to get the signal to the codec without being too heavily affected by interference.

1.5 Analog to Digital Conversion

The AD converter to be used should be able to convert the signal without deteriorating the signal more than necessary. It should also be able to handle the desired dynamic range. A suitable AD will be chosen and evaluated.

1.6 Power Supply

It is important that the power supply is stable enough not to cause any significant distortion or noise. The available voltage source will be examined and evaluated. If it proves to be too noisy a voltage regulator will be implemented.

1.7 Complete Microphone System

When all parts of the microphone chain is finished, the complete microphone system will be integrated in a camera and tested and evaluated. This test will give the specifications of the complete microphone system.

The acoustic design is critical when trying to achieve high sound quality. Since the microphone will be placed on the PCB, an acoustic transmission line must lead the acoustic signals into the camera. Even very short acoustic transmission lines can have a devastating effect on the frequency response.

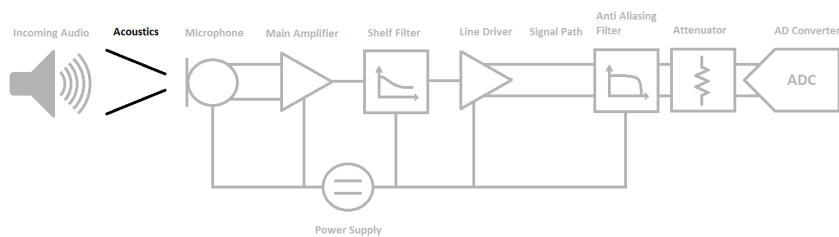


Figure 2.1: Block diagram of the audio chain with the acoustics highlighted.

2.1 Theory

A straight acoustic transmission line work in the same way as an organ pipe. A standing wave will form at the transmission line's resonance frequency as well as every harmonic which will induce peaks in the frequency response. For a closed pipe the wavelength of the fundamental standing wave will be

$$\lambda_{StandingWave} = 4 \cdot l_{pipe} \quad (2.1)$$

plus every harmonic: $\frac{3\lambda}{4}, \frac{5\lambda}{4}, \frac{7\lambda}{4}, \dots$

For an open pipe the fundamental wavelength will be

$$\lambda_{StandingWave} = 2 \cdot l_{pipe} \quad (2.2)$$

plus every harmonic: $\frac{3\lambda}{2}, \frac{5\lambda}{2}, \frac{7\lambda}{2}, \dots$

The wavelength can be translated to the frequency with the following formula:

$$f = \frac{v}{\lambda}, \quad (2.3)$$

where v is the velocity of sound in a given medium.

The resonance frequency may be terminated using a Helmholtz Resonator.

2.1.1 Quality Factor

The quality factor, Q , describes how effectively a frequency is attenuated (or amplified) at the desired frequency in relation to the surrounding frequencies. The quality factor is defined as follows.

$$Q = \frac{f_m}{f_2 - f_1}, \quad (2.4)$$

where f_m is the mid frequency of the dip (or peak) and f_1 and f_2 are the frequencies on either side at which the attenuation (or amplification) is reduced by 3 dB. [17]

2.1.2 Helmholtz Resonator

A Helmholtz resonator is essentially a cavity in which acoustic resonance occurs. It can be used to eliminate or single out an acoustic signal of a specific frequency. The resonance frequency is determined by the shape and dimensions of the resonator. [11]

2.2 Theoretical Limit

If one is using a straight tube the theoretical maximum and minimum length of the tube before the resonance peak frequency enters the audible frequency range of 20 Hz – 20 kHz can be calculated using equation 2.1 and 2.3:

Minimum length, if one desires to keep the resonance peak below the audible frequency range:

$$l_{20Hz} = \frac{\lambda_{20Hz}}{4} = \frac{v}{4f} = \frac{346.1 \text{ m/s}}{4 \cdot 20 \text{ Hz}} \approx 4.33 \text{ m}$$

Maximum length, if one wants to keep the resonance peak above the audible frequency range:

$$l_{20kHz} = \frac{\lambda_{20kHz}}{4} = \frac{v}{4f} = \frac{346.1 \text{ m/s}}{4 \cdot 20 \text{ kHz}} \approx 4.33 \text{ mm},$$

where the velocity of sound, v , is the velocity in air at 25 °C.

In conclusion, the length of the tube will have to be satisfy one of the following conditions:

$$l_{tube} \geq 4.33 \text{ m}$$

or

$$l_{tube} \leq 4.33 \text{ mm}.$$

2.3 Conceptual Tests

To test the theory, a few conceptual tests were made.

2.3.1 Straight Tube

In the audio laboratory the reference microphone is calibrated to compensate for the incessant distortion in order to get a flat frequency response (see figure 2.2).

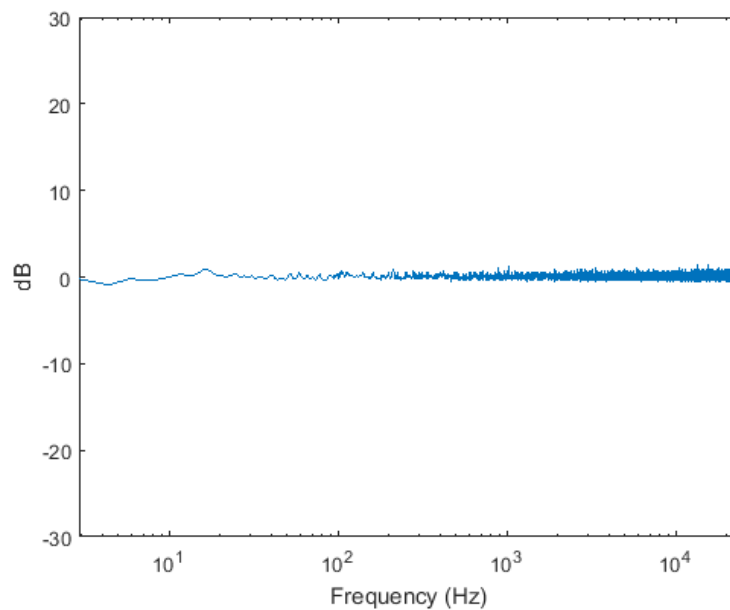


Figure 2.2: The frequency response of the calibrated reference microphone

A tube of length $l = 66,62 \text{ mm}$ is placed on top of the microphone forming a closed tube. The resonance frequency should, according to 2.1 and 2.3 be

$$f = \frac{v}{\lambda} = \frac{v}{4 \cdot l_{\text{pipe}}} = \frac{343 \text{ m/s}}{4 \cdot 66,62 \text{ mm}} = 1,287 \text{ kHz}.$$

So there should be a peak in the frequency response at that frequency as well as at the harmonics frequencies. The result is shown in figure 2.3.

The resonance frequency is according to the frequency response around 1150 Hz which is a bit lower than the calculated value. This is probably because the tube used in this test is slightly funnel shaped which slightly impacts the standing wave frequency.

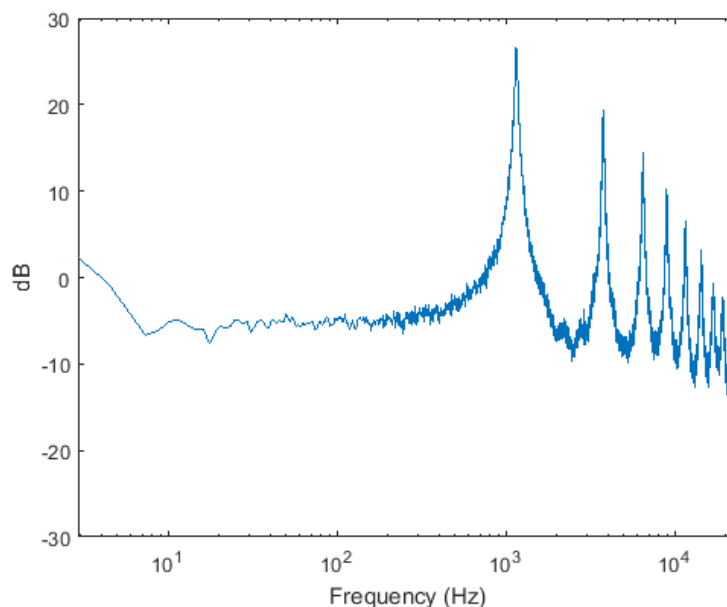


Figure 2.3: The frequency response of the calibrated reference microphone with a closed tube of length $l = 66,62 \text{ mm}$ placed in front of it.

2.3.2 Funneled Tubes

If the larger opening of the tube is closed the resonance frequency will be lower than if the smaller opening is closed. This can be concluded by blowing in the pipe and listening when the resonance frequency is higher and when it is lower. In conclusion, funneling the shape of the pipe leading to the microphone seems to be an efficient way of tuning the resonance frequency. Perhaps if one desires to push the resonance frequency out of the audible spectrum it is possible to do by funneling if the pipe is short enough.

Although, it seems that the funneling only effects the resonance frequency slightly and may not be enough to push the resonance frequency out of the audible range.

2.3.3 Helmholtz Resonator

A possible way of eliminate the resonance frequency is by placing a Helmholtz resonator at the middle of the tube. This is tested and the result is shown in figure 2.4.

As seen in figure 2.4 the resonance frequency of the tube is efficiently eliminated, although it does not yield a flat frequency response. This is because the quality factor of the Helmholtz resonator is much higher than the tube's which

means that the frequency to be eliminated will be eliminated too well which causes a dip rather than a cancellation. Also, since the quality factor of the Helmholtz resonator is so high the frequency range where the frequencies are canceled is very small and therefore only the center of the resonance peak will be quenched.

Furthermore, the Helmholtz resonator seems to be hard to tune. This is mainly because its resonance frequency depends on many geometric parameters.

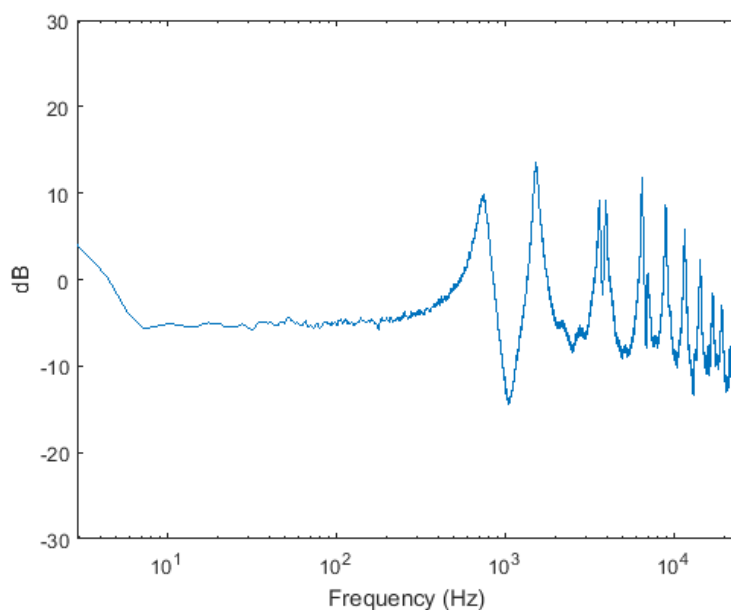


Figure 2.4: The frequency response of the calibrated reference microphone with a closed tube of length $l = 66,62 \text{ mm}$ placed in front of it and a Helmholtz resonator placed at the middle of the tube.

2.4 Conclusion

The acoustic effect of the tube seems to be very hard to get rid of with the evaluated methods. The tuning of the possible designs to get the right Q-factor and the right frequency would be very time consuming. Therefore, a more practicable solution will be implemented: the PCB on which the microphone will be placed will be modified to fit vertically to the camera wall, making it possible to place the microphone right up to the camera chassis which will give an acoustic transmission line shorter than 4.33 mm . This solution should cause no acoustic interference within the audible frequency range.

In order to achieve the best possible sound capture for surveillance the microphone must have the right specifications. A number of microphones with the specifications well suited for the application will be evaluated by testing in a sound isolated environment. The microphone with the most desirable performance is chosen for the project.

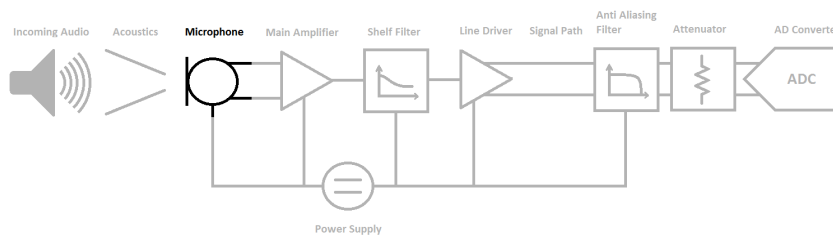


Figure 3.1: Block diagram of the audio chain with the microphone highlighted.

3.1 Theory

The different types of microphones that will be evaluated are electret and MEMS microphones. Here follows a short description of how the different microphones work. Also, some theory that will be used to choose the best microphone for the project.

3.1.1 Electret Microphones

An electret microphone is essentially a capacitive element that changes its capacitance according to the mechanical pressure applied, for example by sound waves, and a FET that works as an internal amplifier (see figure 3.2). The capacitive element has a built in charge which means that varying capacitance also varies the charge applied to the gate of the FET. This charge variation will then be amplified

and the corresponding electric signal can be collected at the output (+ and -). The required bias voltage is for the FET to become active. [2]

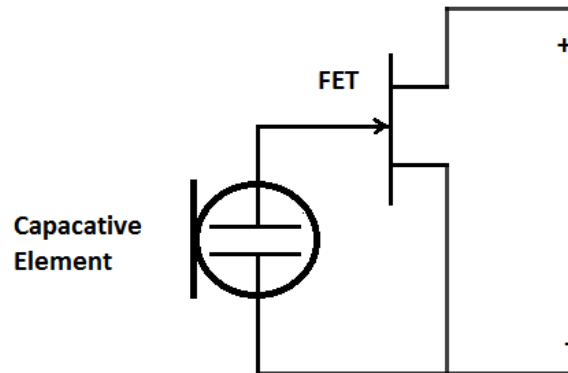


Figure 3.2: Schematic of an electret microphone.

3.1.2 Microelectromechanical Systems (MEMS) Microphones

An analogue MEMS microphone usually consist of a transducer element (the element that converts sound pressure to electrical signals) and an amplifier circuit. The transducer element is essentially a pressure sensitive capacitor which is connected to the amplifier circuit; both integrated into the same IC circuit. This makes it possible to create very small microphones. [12]

3.1.3 Stokes' Law of Sound Attenuation

Stokes' law gives the attenuation of sound depending on a number of factors:

$$\alpha = \frac{2\mu\omega^2}{3\rho V^3}, \quad (3.1)$$

where α is attenuation rate with which the amplitude of the sound decreases exponentially, μ is the dynamic viscosity coefficient of the medium in which the sound travels, ω is the sound frequency, ρ is the density of the medium and V is the speed of sound in the medium. [9]

3.2 Microphone Specifications

The following specifications are considered when choosing a suitable microphone.

3.2.1 Sensitivity

The sensitivity tells you the electrical output of the microphone for a given acoustic input. The value that is specified in the datasheet for a microphone is typically produced by applying a 1 *kHz* sine at 94 *dBu* which equals a pressure of 1 *Pa*.

Sensitivity is specified in *mV/Pa* or *dBV* with the following relationship:

$$Sensitivity_{dBV} = 20 \cdot \log_{10} \left(\frac{Sensitivity_{mV/Pa}}{Output_{REF}} \right), \quad (3.2)$$

where $Output_{REF} = 1 \text{ V/Pa}$.

A high sensitivity is not good for all applications since it has a higher probability of distorting the signal. [10]

In the case of surveillance cameras it is beneficial to have a high sensitivity since the sound source is often far away from the microphone and therefore weak. Higher sensitivity gives a more powerful signal for the incoming sound which means that it will be less sensitive to noise.

3.2.2 Directionality

The directionality says how well the microphone takes up sound depending on the angle of the incoming sound. [10]

The desired directionality for surveillance is often omnidirectional which means that the microphones captures the incoming signal equally well no matter which angle the sound comes from. This is because one wants to be able to cover the largest possible area.

3.2.3 Signal-to-Noise Ratio (SNR)

The SNR is the difference in power between a certain reference signal and the power of the noise. The reference signal is typically 94 *dBu* at 1 *kHz*. The noise is usually measured over the 20 *Hz* – 20 *kHz* interval since this is the frequency range that the human ear can detect.

The SNR is measured in *dB* and is calculated as shown below.

$$SNR = 10 \cdot \log_{10} \left(\frac{P_{signal}}{P_{noise}} \right) = 20 \cdot \log_{10} \left(\frac{V_{signal}}{V_{noise}} \right) \quad (3.3)$$

It is not unusual that the frequency spectrum is weighted according to the frequencies that the human ear is most sensitive to before the SNR is calculated. There are different weighing standards that can be used depending on the application. The most common is the A standard. The SNR will then be specified in *dBA*. [10]

The SNR is a critical aspect in surveillance. To be able to properly interpret a certain sound, the signal that this sound give rise to should have a higher power than the noise. One wants to be able to detect as weak signals as possible.

3.2.4 Equivalent Input Noise (EIN)

The EIN is the power of a sound that, at the input of the microphone, generates a signal with the same power as the noise of the microphone. It can be calculated as follows.

$$EIN = \text{acoustic overload point} - \text{dynamic range} \quad (3.4)$$

$$EIN = 94 \text{ dB} - SNR \quad (3.5)$$

That is to say the lowest acoustic signal that can be distinguished by the microphone for the specified frequency range. [10] One wants as high EIN as possible.

3.2.5 Frequency Response

The frequency response tells you how well the microphone picks up audio at different frequencies i.e. the output level of the microphone as a function of the frequency. [10] One usually wants a flat frequency response but the frequencies around 500 Hz to 5 kHz are usually most important since this is where most essential information, such as speech, is located. It is also possible that a peak for high frequencies is desired since high frequency sounds may be attenuated before reaching the microphone.

3.2.6 Total Harmonic Distortion (THD)

The THD tells you the amount of distortion at the output of the microphone (in percent) given a clean input signal. It can be calculated as follows.

$$THD = \frac{\sum_{i=1}^{\infty} Power(f_{harmonics})}{Power(f_{fundamental})} \quad (3.6)$$

The acoustic test signal is typically 105 dBu when deciding this specification. [10]

3.2.7 Power Supply Rejection (PSR) and Power Supply Rejection Ratio (PSRR)

This tell you how big impact fluctuations in the power supply have on the signal. If the PSR and PSRR are low the fluctuation in the power supply may create a great deal of distortion in the signal. The PSR is typically measured using a 100 mV_{pp} square wave at 217 Hz on top of the power voltage and the PSRR using a 100 mV_{pp} sine wave at 217 Hz on top of the power voltage. The specified value is often defined as the impact at 1 kHz and is not weighted. The PSRR is calculated as follows.

$$PSRR(dB) = 20 \cdot \log_{10} \left(\frac{\Delta V_{Supply}}{\Delta V_{out}} \cdot A_V \right) \quad (3.7)$$

[10]

The PSR and the PSRR is important if the power supply fluctuates a lot. But it is better to make sure that the power supply is silent since some microphones do not have any PSR or PSRR at all.

3.2.8 Acoustic Overload Point (AOP)

The AOP is defined as the acoustic signal power that generates a $THD = 10\%$. This is also known as the clipping point. [10]

3.2.9 Dynamic Range

The dynamic range is the difference between the maximum and the minimum signal level that the microphone can handle without the signal being distorted or drowning in noise. [10] In surveillance one wants to have as large dynamic range as possible.

3.3 Microphone Measurements

The following microphones have been chosen for evaluation:

- PUI Audio POM-3535L-2-R, \$2.47 (10k price)
- InvenSense ICS-40720, \$2.80 (5k price)

The most crucial specifications of the microphones will be verified and the microphone that performs best in accordance with the project will be chosen. The specifications that will be tested is frequency response, sensitivity and SNR (more comprehensive measurements will be done on the complete microphone system in order to acquire the full specifications of the finished design).

Basic power supply circuits were constructed in order to do measurements on the microphones. Batteries were used as a voltage source since they are more or less noise free.

3.3.1 Frequency Response

The microphones were placed in a quiet environment. White noise was played and the corresponding signal was analysed with a fast fourier transform (FFT) to get the frequency response. The result for the Electret (PUI Audio, POM-3535L-3-R) is displayed in figure 3.3.

The result for the MEMS (InvenSense ICS-40720) microphone is displayed in figure 3.4.

3.3.2 Sensitivity and SNR Measurements

To measure the sensitivity of the microphones a sinus tone of 1 kHz at 94 dBu is needed as this is the condition in which the parameter is specified. A specific point in space was calibrated to be exactly 94 dBu using a sound pressure meter. The microphones were placed in this point and the corresponding signal RMS value was measured. By using equation 3.2 the sensitivity was calculated. The result is shown in table 3.1.

To calculate the SNR the noise generated by the microphone itself is needed. To measure this the microphones were placed in a silent place and the output RMS noise was measured for the desired bandwidth, $20\text{ Hz} - 20\text{ kHz}$ (since the audio

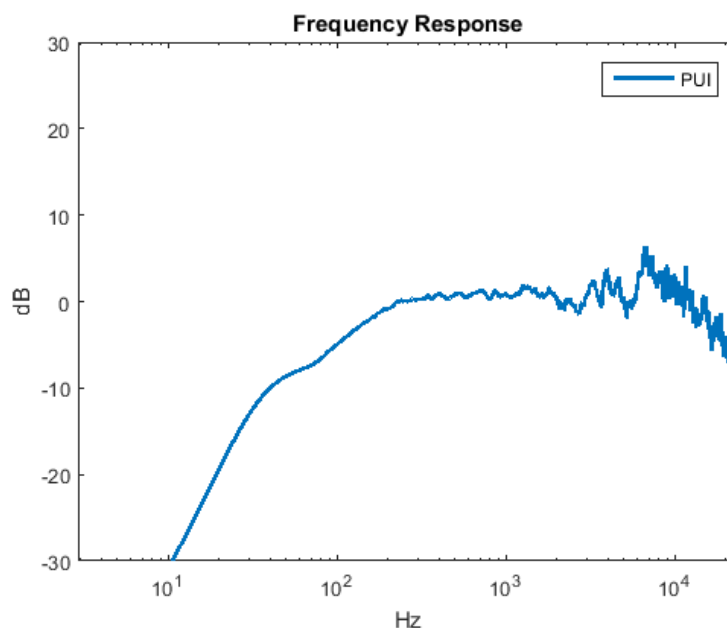


Figure 3.3: Frequency response of the PUI Audio POM-3535L-3-R

lab used was not silent for frequencies below 200 Hz the actual frequency span used was 200 Hz – 20 kHz). Using equation 3.3 the SNR was calculated. The result can be seen in table 3.1.

3.4 Choice of Microphone

The frequency response for the two microphones differ greatly as seen in figure 3.3 and 3.4. A flat frequency response is desired which means that there will be a need for filtering. Both microphones have a relatively flat frequency response in the middle of the frequency range but on the edges it differs. For the electret

	InvenSense	PUI Audio
Signal (mV_{RMS})	27.1	18.1
Noise (μV_{RMS})	9.7	6.30
Sensitivity (dB)	-31.34	-34.85
SNR (dB)	68.92	69.17

Table 3.1: Sensitivity and SNR for the MEMS (InvenSense) and the electret (PUI Audio) microphone.

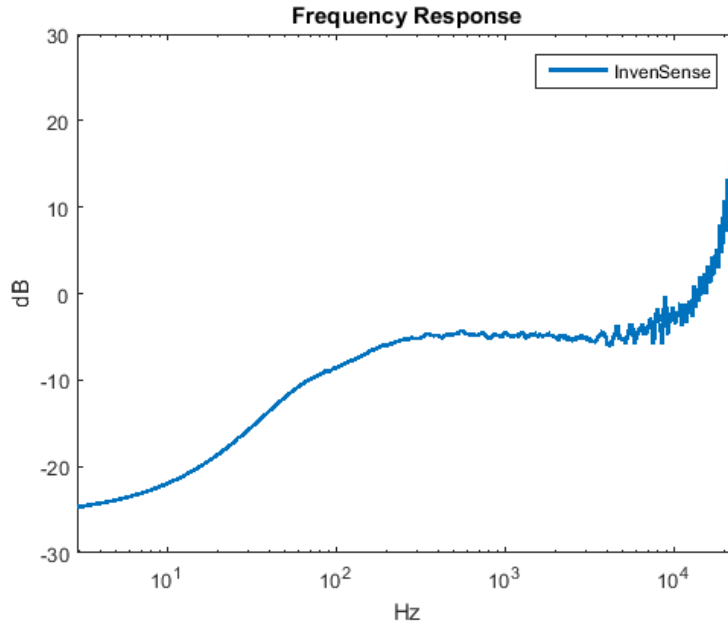


Figure 3.4: Frequency response of the InvenSense ICS-40720 microphone.

microphone (figure 3.3) the low frequencies falls off rapidly, making it hard to obtain low frequency signals without gaining the noise too much. The MEMS microphone (figure 3.4) also falls of for low frequencies but not as rapidly making it easier to obtain low frequency signals without gaining the noise to much.

At high frequencies the electret microphone falls off slightly while the MEMS microphone is heavily gained. A heavy gain is in our case preferred for two main reasons. It may be desired to have a high gain at high frequencies since high frequencies are more easily attenuated then lower frequencies, as seen in equation 3.1. Because the audio source is often far away from the microphone in surveillance applications, high gain at high frequencies may result in a more or less flat frequency response. If the peak at high frequencies is to be attenuated the noise will also be attenuated which gives a lower total noise. Furthermore, the electret seems to have some kinks at high frequencies which will be hard to get rid of with standard analogue filtering methods.

As seen in table 3.1 The SNR is slightly higher for the electret but the sensitivity is much higher for the MEMS. The SNR is an important factor since it determines the lowest possible signal that can be interpreted. Even though the SNR is slightly better for the electret, the high sensitivity of the MEMS will make it more resilient to noise in the analogue processing circuits which will probably yield a higher total SNR for the whole microphone system.

In conclusion, the InvenSense ICS-40720 MEMS microphone is chosen for the project due to its good frequency response and great sensitivity and SNR.

Analogue Signal Processing

The task of the analogue signal processing is to amplify the signal to a desired level, filter the signal to get the desired frequency response, efficiently drive the signal over the transmission lines and finally to adapt the signal level to the codec. This should be done by not adding too much noise and distortion and thereby maintaining the quality of the signal.

The desired signal amplitude is determined by the codec which has a maximum input swing rating that should be matched as well as possible. The filtering is designed to get a flat frequency response (except maybe for high frequencies where a higher gain might be desired). To get a flat frequency response the filter should compensate for the non-flat frequency response of the microphone, as well as for possible acoustic phenomena. The filter should also attenuate frequencies that are higher than the codec can sample to avoid aliasing. Since the signal will be transported across the camera to the codec there will be a need for a line driver that makes sure to minimize the effect of possible interference. The block diagram and the analogue signal processing chain can be seen in 4.1.

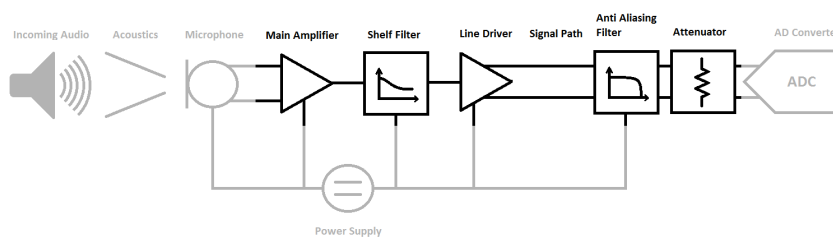


Figure 4.1: Block diagram of the audio chain with the analogue signal processing electronics highlighted.

The OP amplifiers will be operating with supply voltage of 12.5 V . The optimal voltage swing of the signal to the codec should be 2.5713 V_{pp}

4.1 Theory

In order to understand how to design the analogue processing circuit and to fully understand the specifications of the OP amplifiers some theory is required.

4.1.1 Frii's Formula

The noise factor (F) tells you how much the SNR deteriorates from the input to the output of a certain component or system. The F is defined as follows.

$$F = \frac{SNR_{In}}{SNR_{Out}} \quad (4.1)$$

For a whole system the total F can be calculated with the following formula.

$$F_{tot} = F_1 + \frac{F_2 - 1}{A_1} + \frac{F_3 - 1}{A_1 \cdot A_2} + \frac{F_4 - 1}{A_1 \cdot A_2 \cdot A_3} + \dots, \quad (4.2)$$

where A is the amplification of each stage.

The noise figure (NF) is the F in dB :

$$NF = 10 \cdot \log_{10}(F)$$

[16]

4.1.2 Ohm's law

Voltage, resistance and current relates as follows.

$$U = R \cdot I, \quad (4.3)$$

where U is voltage, R is resistance and I is current. [4]

4.1.3 Noise Calculations

To calculate the total noise given a certain noise density and a certain bandwidth the following equation is to be used.

$$N_{RMS} = \sqrt{\int_{f_{Low}}^{f_{High}} (PSD_f)^2 df} \approx PSD \cdot \sqrt{f_{range}}, \quad (4.4)$$

where PSD is the noise density (power spectral density). The approximation can be made if the noise can be considered white, since that means that the noise power is evenly distributed over the frequency range and therefore the integral can be approximated with a constant value. This equation assumes that a perfect filter (a brick-wall filter with filter coefficient 1) is used.

Usually one needs to account for the filter not being perfect with a filter coefficient $BW_{coeff.}$. The equation then becomes

$$N_{RMS} = \sqrt{\int_{f_{Low}}^{f_{High}} (PSD_f)^2 df \cdot BW_{coeff.}} \approx PSD \cdot \sqrt{f_{range} \cdot BW_{coeff.}} \quad (4.5)$$

[3]

To calculate the peak to peak value of the RMS noise the following equation is used.

$$N_{pp} = 6.6 \cdot N_{RMS}, \quad (4.6)$$

where 6.6 is a constant used to approximate the RMS taking the standard deviation of white noise into account. [6]

4.2 Operational Amplifier Specifications

A number of operational amplifiers will be used to implement the analogue signal processing chain. In order to get the desired amplification and filtering as well as maintaining the signal quality it is important to choose OP amplifiers that are good enough. To do so one needs to know what specifications to consider. Among the OP amplifiers with adequate performance the cheapest one will be chosen.

4.2.1 Temperature Dependence

Usually the input offset voltage and input offset current varies slightly depending on the temperature. This is often represented in a curve in which it is shown how the current or voltage varies with the temperature. It is also possible that the temperature dependence is represented with an average temperature coefficient that gives the average voltage and current variation per temperature, usually as $\mu V/^\circ C$ and $nA/^\circ C$. [14]

Since Axis cameras are often placed outside and sometimes in very cold or very hot environments the temperature dependence should be low.

4.2.2 Phase Margin

The phase margin is the absolute value of the phase shift of the signal when the open-loop gain reaches unity gain. This parameter indicates whether the OP amplifier is stable. [14]

All OP amplifiers that are evaluated is assumed to be stable and therefore this parameter will be assumed to be ideal.

4.2.3 Unity Gain Bandwidth

The unity gain bandwidth is the frequency range from 0 Hz up to the frequency at which the open loop gain is unity. [14]

Since all considered OP amplifiers has a far greater frequency range than needed for audio applications this parameter will be assumed to be ideal.

4.2.4 Common Mode Rejection Ratio (CMRR)

CMRR is the ratio between the amplification of the differential signal (the desired amplification) and the amplification of the common mode.

$$CMRR = 20 \cdot \log_{10} \left(\frac{A}{A_{cm}} \right), \quad (4.7)$$

where A is the amplifiers signal amplification and A_{cm} is the amplifiers common mode amplification. [14]

This parameter is crucial for this project since some of the OP amplifiers will be used in a differential mode.

4.2.5 Noise

The noise specifications says how much noise an OP Amplifier adds to the signal. In order to maintain a low noise signal with a good SNR it is crucial for the OP amplifiers to have adequate noise performance.

The voltage noise generated by a standalone OP amplifier is often represented with a noise source at the input of the amplifier (see figure 4.2). It is often specified as a noise density and is denoted as nV/\sqrt{Hz} . The noise density is often specified for a certain frequency, usually 1 kHz , but may also be represented in a diagram of how the noise density varies with the frequency. Thus the total voltage noise from the OP amplifier depends on the bandwidth. It is possible that the total noise for a given bandwidth is specified for the OP amplifier.

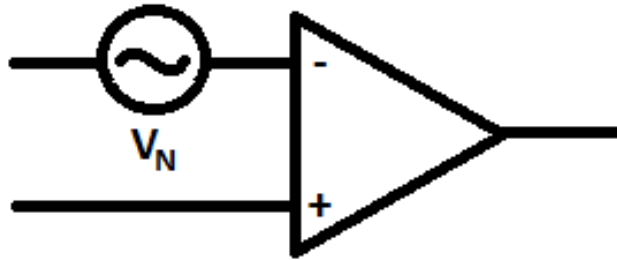


Figure 4.2: The noise of a standalone OP amplifier represented by an input voltage.

Another noise parameter that should be considered is the input current noise which can be seen as a current flowing into the OP amplifiers inputs. This noise is also specified as a noise density and is often denoted as fA/\sqrt{Hz} or pA/\sqrt{Hz} .

This noise should only be significant when one has a high source resistance. According to equation 4.3, a current flowing through a resistance creates a voltage. If the resistance is large the voltage will be significant.[5]

In this project the source resistance is designed to be low in all cases and therefore the input current noise should not be significant.

The noise is often divided into two regions. The region where flicker noise (or $1/f$ - noise) is dominant and the region where white noise is dominant. For low frequencies the flicker noise will be significant for an OP amplifier but as it is inversely proportional to the frequency it will decay with 3 dB/octave . The two regions are split by the so called corner frequency which is the frequency where the flicker noise has the same power as the white noise. [5]

The corner frequency varies between OP amplifiers and should therefore be taken into account. One wants the corner frequency to be as low as possible to minimize the noise for low frequencies.

In conclusion the voltage noise is the most important noise parameter for this project since it can not be improved externally and therefore sets the lowest possible noise floor for the analogue processing circuits. Furthermore the corner frequency for flicker noise should be taken into account to make sure that the low frequency noise is not too great. The current noise should be observed just to make sure it has a reasonable value; preferably lower than a few pA .

4.2.6 Input Bias Current

The input bias current is the average current that flows at the inputs of the OP amplifier. [14]

One typically desires a low input bias current but this parameter is still not crucial since a high input bias current can be handled by designing the feedback network in a clever way. Problem arises when the two inputs of the OP amplifier sees two different source resistances. Then the input bias current will give rise to two different voltages which will create an offset voltage at the output. By designing the feedback resistance so that the two inputs of the OP amplifier sees the same resistance the same voltage will be induced at both inputs thus having negligible effect on the output.

4.2.7 Input Offset Current

Input offset current is the difference in current between the two inputs of the OP amplifier. [5]

This value must be low since a difference in current between the two inputs will cause a voltage difference between the inputs which will create a large offset at the output when amplified.

4.2.8 Supply Voltage Sensitivity

The ratio between the change in input offset and the change in supply voltage. [5]

One typically desires a low supply voltage sensitivity but in this project it is not crucial since the power supply can be assumed to be fairly stable.

4.2.9 Total Power Dissipation

The power used by the OP amplifier that is not delivered to the load. [14]

This parameter will not be considered since the goal of this project is to get the best possible audio. Power dissipation is not the main concern.

4.2.10 Input and Output Impedance

It is important that the input impedance is high, especially for the main amplifier stage. The reason for a high input impedance is to not affect a previous stage or component by loading it to much. An infinitely high input impedance will be seen as an open circuit which is the ideal case since the previous circuits will not be loaded at all. The reason it is especially important for the main amplifier is because it will load the microphone which is the systems most critical component.

It is also important for the output impedance to be low. If the output impedance is high the current noise of the following stage will have a greater impact on the noise performance. A high output impedance may also degrade the overall performance of the following stages.

4.2.11 Slew Rate

This parameter says how fast the OP amplifier can vary the voltage. Usually specified as $V/\mu s$. [14]

It is crucial for this parameter to be adequate. If not, the signal will be distorted. To determine a suitable slew rate a sine wave at $f = 20 kHz$ is considered:

The slope of $\sin(x)$ is determined by

$$\frac{d}{dx}(\sin(x)) = \cos(x).$$

$\cos(x)$ is derived to find the maximum and the minimum value:

$$\frac{d}{dx}(\cos(x)) = -\sin(x)$$

$-\sin(x) = 0$ when $x = \pi/2$ and $x = 3 \cdot \pi/2$. These values of x are put into $\cos(x)$ to find the minimum and the maximum slope:

$$\cos(\pi/2) = 1$$

$$\cos(\pi/2) = -1$$

The signal is assumed to be $10 V_{pp}$. The time between the negative peak and the positive peak is $1/2f = 25 \mu s$ which gives the maximum slope of

$$\frac{10 V}{25 \mu s} = 0.4 V/\mu s.$$

This value is multiplied by 10 to make sure that all signals will remain undistorted at all time.

$$0.4 V/\mu s \cdot 10 = 4 V/\mu s$$

In conclusion the slew rate should be at least $4 V/\mu s$.

4.2.12 Maximum Input Voltage and Maximum Output Voltage

Specifies the maximum input and output voltage swing capability of the OP amplifier. [14]

4.2.13 Minimum and Maximum Operating Voltage

The minimum and maximum supply voltage with which the OP amplifier can operate. [14]

The supply voltage with which the OP amplifiers will be driven is 12.5 V. This must not be below the minimum operating voltage or exceed the maximum.

4.2.14 Crosstalk Attenuation

The ratio between the change in output voltage in a driven channel and the resulting change in output voltage of another channel. [14]

This specification is only important for multichannel OP amplifiers. When using an OP amplifier with several channels the crosstalk attenuation must be high.

4.2.15 Total Harmonic Distortion (THD)

THD is the ratio in percent between the signal and the sum of all harmonic distortions. [14]

This is an important parameter since the distortion should be kept low enough. Although, by reading a number of datasheets it becomes apparent that the THD is far below an acceptable level. According to a test by AxiomAudio.com 1 % is not audible until 8 kHz, and the reviewed OP amplifiers have at least 100 times lower THD in a worst case scenario. [1]

4.3 Choice of OP Amplifiers

Here follows the specifications that are most important for each stage in the processing chain.

4.3.1 Main Amplifier

The main amplifier stage will amplify a differential signal to a single ended signal with the desired voltage swing. The hard requirements, the requirements that have to be fulfilled, are the following: the OP amplifier has to be able to operate with a supply voltage of 12.5 V. The output swing must be, at the very least, enough to satisfy the whole acceptable voltage swing of the codec to utilize the full dynamic range.

The softer requirements most important to this stage are the following: the noise performance, since it is a limiting factor. Because this is the first stage in the chain the noise performance of this stage is by far the most important according to equation 4.2. Since the voltage noise density is the most affecting noise parameter this is the one that will be taken into account. The main amplifier

is fed a differential signal and therefore the CMRR must be high. The slew rate should be at least as calculated in section 4.2.11. The THD is also important but is assumed to be ideal.

4.3.2 Filter

The filter should equalize the signal in order to get the desired frequency response. The hard requirements for the filter are the following. The OP amplifier have to be able to operate with a supply voltage of 12.5 V. Both the input and the output swing of the filter must be at least 2.5713 V_{pp} .

Since this is the second stage the noise requirements are much weaker (see equation 4.2). The signal is now single ended and therefore the CMRR is much less important. The slew rate requirements however are the same as for the main amplifier stage. The THD requirements are also the same as for the main amplifier stage but still considered to be ideal.

4.3.3 Line Driver

The line driver will drive the signal from the analogue sound processing circuits to the codec. Since the signal will pass noisy components, such as processors and motors for the camera, a differential signal is desired in order to minimize the effect of interference. The hard requirements for the line driver are the following. The OP amplifier have to be able to operate with a supply voltage of 12.5 V. Both the input and the output swing of the main amplifier must be at least 2.5713 V_{pp} . This OP amplifier also has to have a differential output for the reason stated above.

Since this is the third stage the noise requirements are still much weaker than for the main amplifier (see equation 4.2). The slew rate and THD requirements are the same as for the previous stages. This OP amplifier should be able to deliver a relatively high current in order to be unaffected by the capacitive characteristics of the transmission lines to the codec.

4.3.4 Chosen OP Amplifiers

Main Amplifier and Filter

The OP amplifier chosen for the main amplification and for the filtering circuits is the Texas Instruments OPA1662(dual channel)/OPA1664(quad channel). The hard requirements are satisfied and thus the op amplifier can be used for this application.

This particular OP amplifier has an exceptionally low noise density of $3.3nV/\sqrt{Hz}$ at 1 kHz and a relatively flat noise density in relation to the input frequency. Assuming a perfect filter (brick-wall filter) at 20 kHz and no flicker noise, the total noise will according to equation 4.5 be

$$N_{RMS} \approx PSD \cdot \sqrt{f_{range}} \cdot BW_{coeff} = 3.3nV/\sqrt{Hz} \cdot \sqrt{20 kHz} \cdot 1 \approx 467 nV_{RMS},$$

which is very small in comparison with the microphones noise of $9.7\mu V_{RMS}$. The current noise will have significantly lower impact since the source resistances will be low, though it will be taken into account in the noise calculations.

The CMRR is typically 100 dB according to the datasheet for the supply voltage to be used which should be enough. The THD is 0.0001 % and the slew rate is 13 $V/\mu s$ which is far beyond what is needed. Since this OP amplifier has a bipolar input the input bias current is very high. This does not have to be a problem but it has to be taken into account in the design.

Furthermore this OP amplifier has the very low price of \$0.75 for a dual channel and \$1.10 for a quad channel when purchasing at least 1k units.

Line Driver

For the line driver the LME49724 is chosen. It fulfills the hard requirements and can deliver a current of 80 mA which is relatively high. This will reduce the effect of the capacitive characteristics of the transmission lines as well as reducing the effect of interference. It also has a very low voltage noise of 2.1 nV/\sqrt{Hz} , a high slew rate of 18 $V/\mu s$ and a negligible THD .

The price of \$1.42 (when buying 1k or more) is high compared to the OP amplifier for the other stages but it is still relatively cheap for a fully differential OP amplifier.

Main Amplifier

The main amplifier is the first active stage in the analogue signal processing chain and is placed right after the microphone. This stage will amplify the signal to a desired level. The signal has to be amplified in order to cover the full dynamic range of the AD converter in order to obtain as much useful data as possible. Amplifying the signal will also reduce the effect of noise in following filter stages and signal transportation.

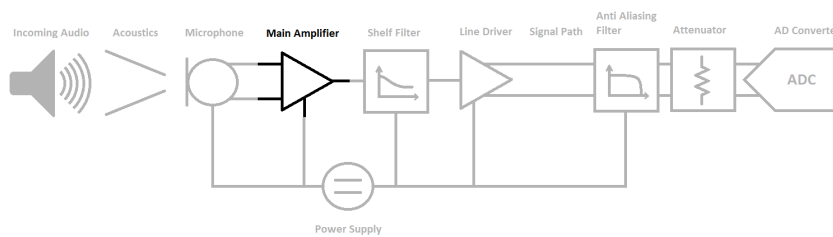


Figure 5.1: Block diagram of the audio chain with the main amplifier highlighted.

5.1 Theory

5.1.1 Amplifier Configurations

Non Inverting Amplifiers

A non-inverting amplifier (see figure 5.2) amplifies the signal without inverting it. The gain is set by the two resistors as follows.

$$A = 1 + \frac{R_2}{R_1} \quad (5.1)$$

The downside of this amplifier configuration is that it is impossible to get a gain lower than $A = 1$.

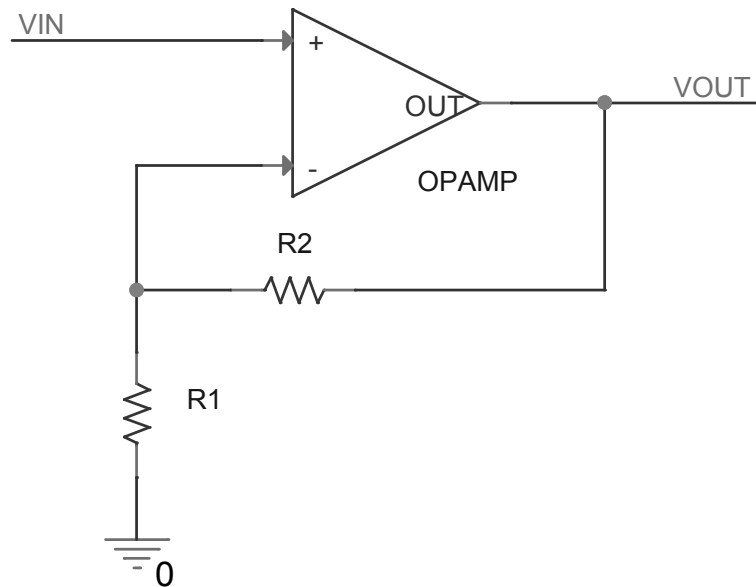


Figure 5.2: A non-inverting amplifier configuration

Inverting Amplifiers

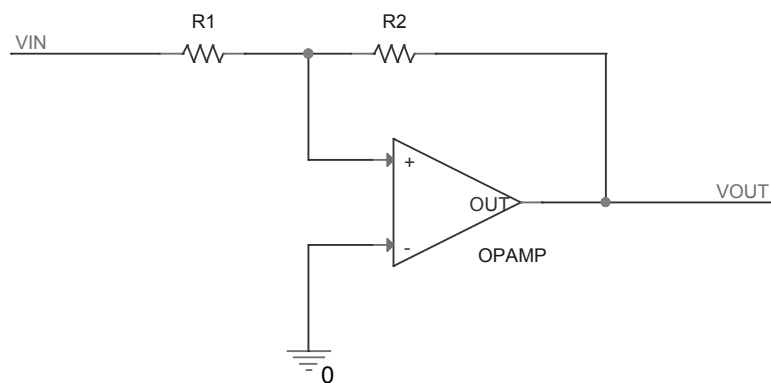


Figure 5.3: An inverting amplifier configuration

An inverting amplifier (see figure 5.3) amplifies and inverts the signal. The gain is set by the resistors as follows.

$$A = -\frac{R_2}{R_1} \quad (5.2)$$

With this configuration it is possible to obtain a gain less than one. The downside is that the input impedance will be R_1 . [18]

Differential Input Amplifiers

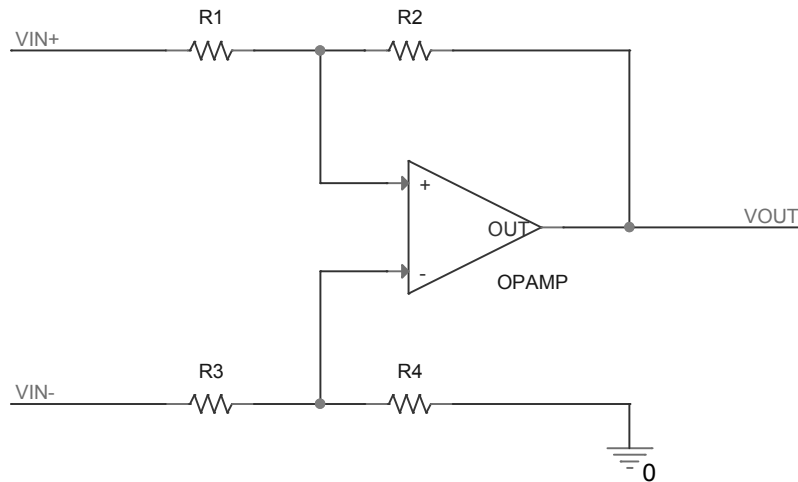


Figure 5.4: A differential amplifier configuration

The differential input amplifier (see figure 5.4) amplifies the difference between the two input signals. If $R_1 = R_3$ and $R_2 = R_4$, which is usually the case, the output is given by

$$V_{OUT} = \frac{R_3}{R_1}((V_{IN-}) - (V_{IN+})). \quad (5.3)$$

[18]

5.1.2 Noise

Thermal Noise (Johnson-Nyquist Noise)

The resistors will contribute with noise depending on temperature and resistor value. The noise density from a resistor can be calculated with the following formula.

$$v_n^2 = 4k_B T R, \quad (5.4)$$

where k_B is Boltzmann's constant $k_B = 1.38 \cdot 10^{-23} \text{ m}^2 \text{ kg s}^{-2} \text{ K}^{-1}$, T is the temperature and R is the resistance value of the resistor. [18]

Flicker Noise ($1/f$ Noise)

Noise of unknown origin with a $1/f$ characteristic. The frequency at which the thermal noise becomes more significant than the flicker noise is called the corner frequency, f_{corner} . This frequency is usually relatively low in comparison with the audible spectra of $20 - 20kHz$. Therefore it usually does not affect the total noise performance significantly. [20]

Noise Summation

If all noise sources can be assumed to be independent of all other noise sources and produce only white noise, the average of the noise sources can be summed as follows.

$$E_{tot} = \sqrt{E_1^2 + E_2^2 + E_3^2 + \dots} \quad (5.5)$$

[18]

5.1.3 Filter

A filter is a circuit or component that modifies the frequency response of a system.

RC Filter

The most standard passive filter is realised using a simple RC circuit (See figure 5.5).

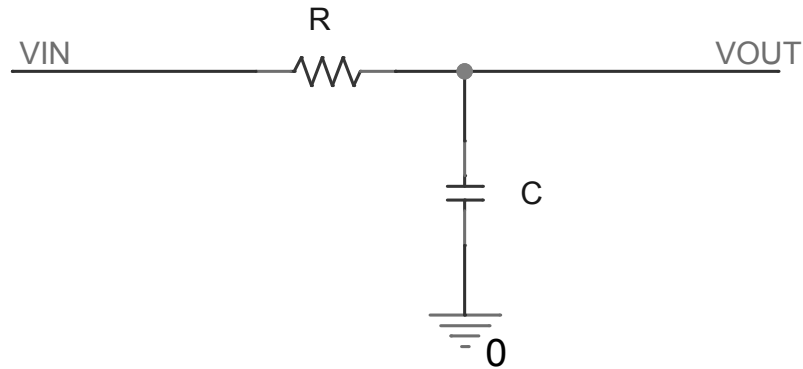


Figure 5.5: A low pass filter realised with a single ended RC circuit

The cutoff frequency, f_c , which is the frequency at which the gain has dropped $3 dB$, is calculated with the following formula.

$$f_c = \frac{1}{2\pi RC} \quad (5.6)$$

This equation can also be used for differential filters (see figure 5.6) but the capacitor C' across the two signal paths should have half the value of C i.e. $C = 2C'$.

For resistors, R , across the two signal paths the resistive value, R' , should be double the value of R i.e. $R = R'/2$. [15]

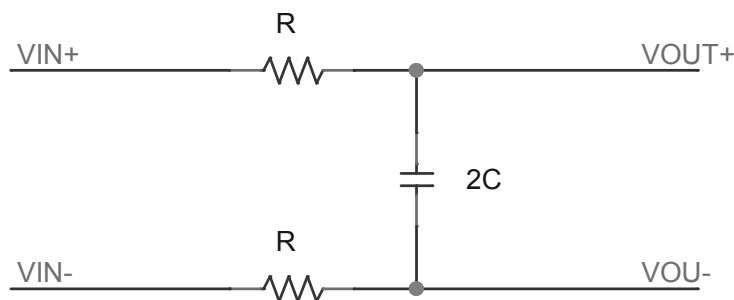


Figure 5.6: A low pass filter realised with a differential RC circuit

Ferrite Components

Ferrite components are made from ferrite which is a magnetic material and are used to suppress electromagnetic interference (EMI). They are designed to eliminate high frequency signals by absorbing their energy. Ferrite components are chosen based on mainly two parameters: the frequency at which the absorption peaks and the Q-factor. These parameters will be matched to the most significant EMI interference and thereby suppressing it as efficiently as possible. [8]

5.2 Amplifier Configuration

The main amplifier should have high input impedance so that the microphone performs well, low noise to not affect the SNR too much, high common mode rejection to not get the wrong DC-offset or be susceptible to common mode noise and a high enough gain.

The amplifier configuration that will be used for the main amplifier is that of an instrumentation amplifier (see figure 5.7).

This configuration converts a differential signal to a single ended signal which makes the filtering easier. It also has a much higher input impedance than a single OP amplifier configuration. This is because the input stage of the instrumentation amplifier is divided into two separate amplifiers both working in single ended mode, which gives a much larger input impedance than a differential configuration. A high input impedance makes sure that the microphone performs well. The instrumentation amplifier also has low noise, a very high common mode rejection ratio and a low drift, which means that the amplifier does not change its characteristics over time due to, for example, temperature changes.

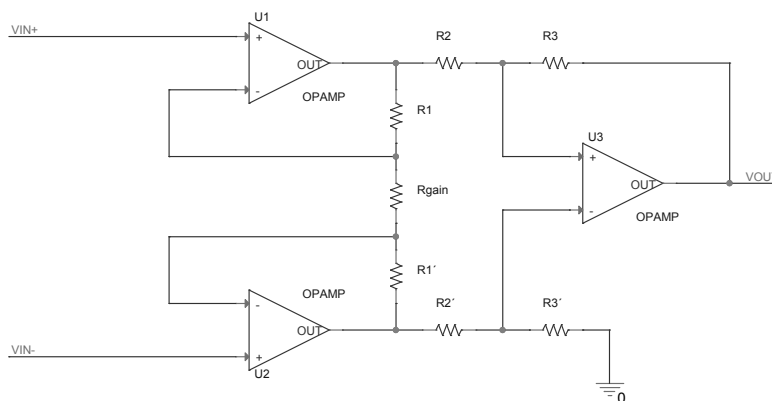


Figure 5.7: The instrumentation amplifier configuration to be used as the main amplifier.

Low resistor values are used in order to reduce the noise (see equation 5.4). The resistor R_{gain} will be a variable resistor since the required gain is hard to know since the microphone often will be placed far away from the sound source.

The amplifier is divided into two stages: the first two non-inverting OP amplifiers that amplify one half of the signal each, and the third, differential input, amplifier that amplifies the difference between the signals. The total gain, A , of the instrumentation amplifier is obtained by multiplying the gain of the two stages:

$$A = A_{stage1} \cdot A_{stage2} = \left(1 + \frac{2R_1}{R_{gain}}\right) \left(\frac{R_3}{R_2}\right), \quad (5.7)$$

The instrumentation amplifier in this project is designed so that only the first stage is amplifying. This is because one wants as high amplification as possible as early as possible according to Frii's formula (see equation 4.2).

5.3 Connection Circuit for Main Amplifier

In order for the main amplifier to perform as well as possible a certain connection circuit is required: a low pass filter to make sure that the amplifier is not saturated by interfering high frequency signals, a DC-block to be able to give the signal the desired bias voltage and ferrites in the signal path to eliminate certain electromagnetic interferences.

The low pass filter can be realised as a differential RC filter where the resistance of the microphone act together with a capacitance placed across the two input signals. The cutoff frequency of this low pass filter should be very high for two reasons: the source resistance of the microphone is uncertain since there is no minimum or maximum value in the microphones datasheet. If the microphone were to have a much higher source resistance in some cases the cutoff frequency

would drop. With a high cutoff frequency in the typical case there is a low risk of the cutoff frequency dropping enough to affect the audible frequency range. The second reason is that small capacitors are preferred since they usually have a dielectric material that does not affect the signal.

The DC-block is realised as a high pass filter at the input. The problem with this implementation is that a capacitor in the signal path is needed which would increase the flicker noise. Although, this will have a small effect if a large enough capacitor is chosen.

The schematic for the connection network can be seen in figure 5.8.

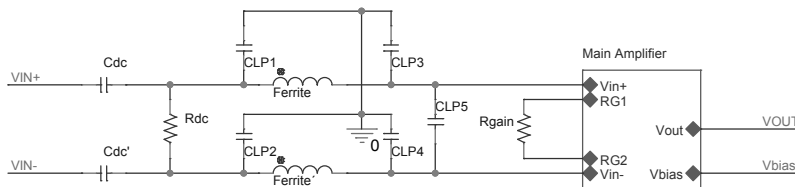


Figure 5.8: The connection circuit for the main amplifier.

The bias voltage is set by a buffered voltage divider that divides the supply voltage by using two equally large resistors (see chapter 6).

Another DC-block is placed after the main amplifier in order to make sure that the gain does not affect the bias voltage.

5.4 Calculations

Calculations on noise performance and frequency response will be done to be able to choose the right component values.

5.4.1 Noise Performance

One of the most important factors of the amplifier is its noise performance. Noise calculations that determine the output noise of the amplifier will be carried out to make sure there is no component that causes unnecessarily high noise. The noise performance will also be simulated to make sure the calculations are accurate.

Noise performance can be divided into three parts: the thermal noise, the voltage noise from the OP amplifier and the current noise from the OP amplifier. The current noise of the resistors will be neglected because of its relatively small contribution. To calculate the noise each contributing component will be observed individually while assuming all other components are ideal. The noise of each component will then be added together. The noise will be calculated for each stage and then added together for the main amplifier's total noise. The noise is assumed to be white. The noise that the noise sources produce will be denoted e and the output noise will be denoted E .

First Stage

The first stage is a non inverting amplifier. This configuration along with the noise sources is represented in the figure 5.9.

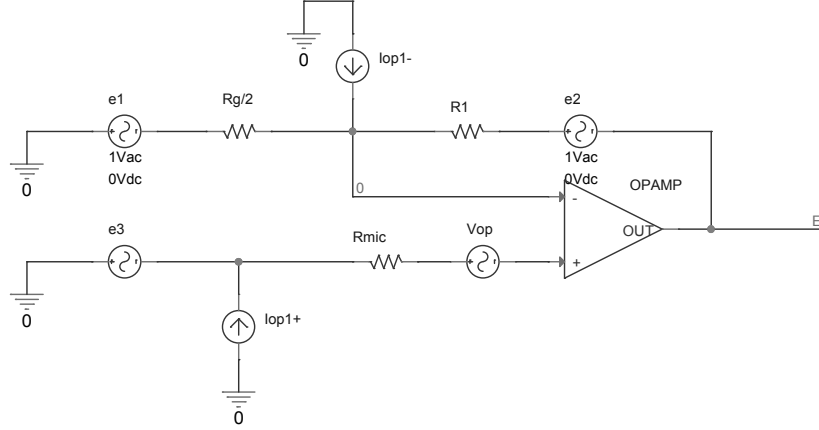


Figure 5.9: Schematic of the non-inverter amplifier with represented noise sources.

Thermal Noise of First Stage

The thermal noise is calculated using Johnson-Nyquist's equation 5.4.

Resistor R_g contributes with the following voltage noise density (V/\sqrt{Hz}):

$$E_1 = e_1 \cdot \frac{2R_1}{R_g}, e_1 = \sqrt{4kT \cdot \frac{R_g}{2}},$$

where T is the temperature assumed to be $300K$ and k is the Boltzmann constant. Resistor R_1 contributes with the following voltage noise density:

$$E_2 = e_2, e_2 = \sqrt{4kTR_1}$$

Resistor R_{mic} contributes with the following voltage noise density:

$$E_3 = e_3 \cdot \left(1 + \frac{2R_1}{R_g}\right), e_3 = \sqrt{4kTR_{mic}}$$

This gives a total output thermal noise density of

$$E_{th,OUT} = \sqrt{E_1^2 + E_2^2 + E_3^2} = \sqrt{4kT \left(\frac{R_g}{2} \left(\frac{2R_1}{R_g} \right)^2 + R_1 + R_{mic} \left(1 + \frac{2R_1}{R_g} \right)^2 \right)}$$

$$= \sqrt{4kTR_1 \left(1 + \frac{2R_1}{R_g}\right) + 4kTR_{mic} \left(1 + \frac{2R_1}{R_g}\right)^2}.$$

To acquire the input referred thermal noise density, $E_{th,IN}$, the output thermal noise density is divided by the amplification factor, A .

$$\begin{aligned} E_{th,IN} &= \frac{E_{th,OUT}}{A} = \sqrt{\frac{4kTR_1 \left(1 + \frac{2R_1}{R_g}\right) + 4kTR_{mic} \left(1 + \frac{2R_1}{R_g}\right)^2}{\left(1 + \frac{2R_1}{R_g}\right)^2}} \\ &= \sqrt{4kT \left(\left(\frac{R_g R_1}{R_g + 2R_1} \right) + R_{mic} \right)} \end{aligned}$$

OP Noise of First Stage

The noise caused by the OP amplifiers voltage noise is the following:

$$E_{OP1V} = V_{OP1} \cdot A = V_{OP1} \left(1 + \frac{2R_1}{R_g}\right)$$

The noise caused by the OP amplifiers current noise at the + port:

$$E_{OP1I+} = I_{OP1+} \cdot R_{mic} \cdot A = I_{OP1+} \cdot R_{mic} \cdot \left(1 + \frac{2R_1}{R_g}\right)$$

and for the – port:

$$E_{OP1I-} = I_{OP1+} \cdot R_1$$

The output noise density from the OP is the sum of the calculated noise factors:

$$\begin{aligned} E_{OP1,OUT} &= \sqrt{E_{OP1V}^2 + E_{OP1I+}^2 + E_{OP1I-}^2} \\ &= \sqrt{V_{OP1}^2 \left(1 + \frac{2R_1}{R_g}\right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g}\right)^2 + I_{OP1-}^2 \cdot R_1^2} \end{aligned}$$

The input referred noise is simply the noise sources that affects the input without amplification:

$$E_{OP1,IN} = \sqrt{V_{OP1}^2 + I_{OP1+}^2 \cdot R_{mic}^2}$$

Total Noise of First Stage

The total noise of the first stage is a summation of the noise of the two amplifiers in the first stage. Since the two stages are identical this can be done by multiplying the noise of one stage with $\sqrt{2}$. Thus, the noise of the first stage of the main amplifier is the following:

$$E_{stage1,OUT} = \sqrt{2} \cdot \sqrt{E_{th}^2 + E_{OP1}^2}$$

$$= \sqrt{2} \cdot \sqrt{4kT \left(\frac{R_g}{2} \left(\frac{2R_1}{R_g} \right)^2 + R_1 + R_{mic} \left(1 + \frac{2R_1}{R_g} \right)^2 \right) + \frac{V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_1^2}{}}$$

$$E_{stage1,IN} = \sqrt{2} \cdot \sqrt{4kT \left(\left(\frac{R_g R_1}{R_g + 2R_1} \right) + R_{mic} \right) + V_{OP1}^2 + I_{OP1}^2 \cdot R_{mic}^2}$$

Second Stage

The second stage is a differential input amplifier. This configuration along with the noise sources is represented in the figure 5.10.

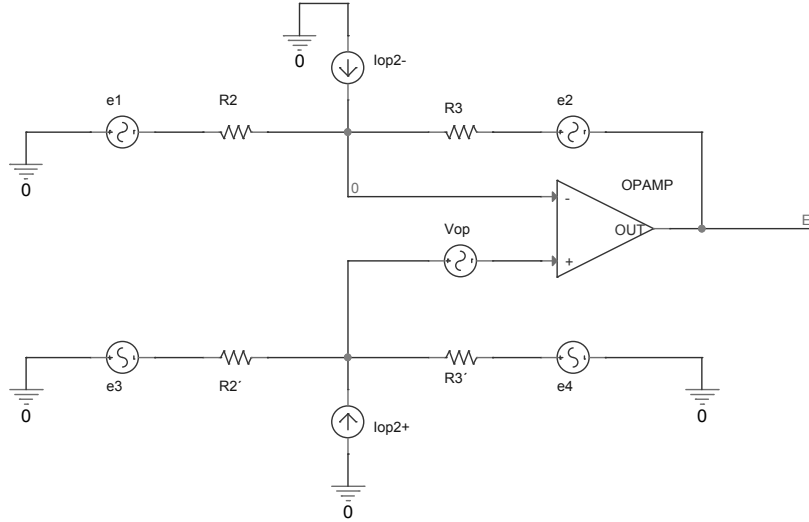


Figure 5.10: Schematic of the differential amplifier with represented noise sources.

Thermal Noise of Second Stage

The thermal noise is calculated using Johnson-Nyquist's equation 5.4.

Resistor R_2 contributes with the following voltage noise density (V/\sqrt{Hz}):

$$E_1 = e_1 \cdot \frac{R_3}{R_2}, e_1 = \sqrt{4kTR_2},$$

where T is the temperature assumed to be 300K and k is the Boltzmann constant.

Resistor R_3 contributes with the following voltage noise density:

$$E_2 = e_2, e_2 = \sqrt{4kTR_3}$$

Resistor R'_2 contributes with the following voltage noise density:

$$E_3 = e_3 \cdot \left(\frac{R'_3}{R'_3 + R'_2} \cdot \frac{R_3 + R_2}{R_2} \right), e_3 = \sqrt{4kTR'_2}$$

Resistor R'_3 contributes with the following voltage noise density:

$$E_3 = e_3 \cdot \left(\frac{R'_2}{R'_2 + R'_3} \cdot \frac{R_3 + R_2}{R_2} \right), e_3 = \sqrt{4kTR'_3}$$

This gives a total output thermal noise density of

$$\begin{aligned} E_{th,OUT} &= \sqrt{E_1^2 + E_2^2 + E_3^2 + E_4^2} \\ &= \sqrt{4kT \left(R_1 \left(\frac{R_3}{R_2} \right)^2 + R_2 + R_3 \left(\frac{R'_3}{R'_3 + R'_2} \cdot \frac{R_3 + R_2}{R_2} \right)^2 + R_4 \left(\frac{R'_2}{R'_2 + R'_3} \cdot \frac{R_3 + R_2}{R_2} \right)^2 \right)}. \end{aligned}$$

Since the second stage does not have any amplification the input referred thermal noise density will be the same as the output thermal noise density:

$$E_{th,IN} = E_{th,OUT}$$

OP Noise of Second Stage

The noise caused by the OP amplifiers voltage noise is the following:

$$E_{OP2V} = V_{OP2} \cdot A = V_{OP1} \left(\frac{R_2 + R_3}{R_2} \right)$$

The noise caused by the OP amplifiers current noise at the + port:

$$E_{OP2I+} = I_{OP2+} \cdot A = I_{OP2+} \cdot \frac{R'_2 \cdot R'_3}{R'_2 + R'_3} \cdot \frac{R_2 + R_3}{R_2}$$

and for the – port:

$$E_{OP2I-} = I_{OP2-} \cdot R_3$$

The total output noise from the OP is the sum of the calculated noise factors:

$$\begin{aligned} E_{OP2,OUT} &= \sqrt{E_{OP2V}^2 + E_{OP2I+}^2 + E_{OP2I-}^2} \\ &= \sqrt{V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_1^2} \end{aligned}$$

Since the second stage does not have any amplification and both terminals is used as inputs the input referred noise density will be the same as the output noise density:

$$E_{OP2,IN} = E_{OP2,OUT}$$

Total Noise of Second Stage

The total output noise of the second stage of the main amplifier is the following:

$$\begin{aligned}
 E_{stage2,OUT} &= \sqrt{E_{th}^2 + E_{OP2}^2} \\
 &= \sqrt{4kT \left(R_1 \left(\frac{R_3}{R_2} \right)^2 + R_2 + R_3 \left(\frac{R'_3}{R'_3 + R'_2} \cdot \frac{R_3 + R_2}{R_2} \right)^2 + R_4 \left(\frac{R'_2}{R'_2 + R'_3} \cdot \frac{R_3 + R_2}{R_2} \right)^2 \right)} \\
 &\quad + V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_1^2
 \end{aligned}$$

Since $E_{th,IN} = E_{th,OUT}$ and $E_{OP2,IN} = E_{OP2,OUT}$:

$$E_{stage2,IN} = E_{stage2,OUT}$$

5.4.2 Total Noise Performance of Main Amplifier

The total noise performance of the main amplifier is given by the sum of the noise from the two stages:

$$\begin{aligned}
 E_{MainAmp,OUT} &= \sqrt{E_{Stage1,OUT}^2 + E_{Stage2,OUT}^2} \\
 &= \sqrt{8kT \frac{R_g}{2} \left(\frac{2R_1}{R_g} \right)^2 + 8kTR_1 + 8kTR_{mic} \left(1 + \frac{2R_1}{R_g} \right)^2} \\
 &\quad + 2V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + 2I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + 2I_{OP1+}^2 \cdot R_1^2 + 4kTR_1 \left(\frac{R_3}{R_2} \right)^2 \\
 &\quad + 4kTR_2 + 4kTR_3 \left(\frac{R'_3}{R'_3 + R'_2} \cdot \frac{R_3 + R_2}{R_2} \right)^2 + 4kTR_4 \left(\frac{R'_2}{R'_2 + R'_3} \cdot \frac{R_3 + R_2}{R_2} \right)^2 \\
 &\quad + V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_1^2
 \end{aligned} \tag{5.8}$$

$$\begin{aligned}
 E_{MainAmp,IN} &= \sqrt{E_{Stage1,IN}^2 + E_{Stage2,IN}^2} \\
 &= \sqrt{4kT \left(\left(\frac{R_g R_1}{R_g + 2R_1} \right) + R_{mic} \right) + V_{OP1}^2 + I_{OP1}^2 \cdot R_{mic}^2 + 4kTR_1 \left(\frac{R_3}{R_2} \right)^2} \\
 &\quad + 4kTR_2 + 4kTR_3 \left(\frac{R'_3}{R'_3 + R'_2} \cdot \frac{R_3 + R_2}{R_2} \right)^2 + 4kTR_4 \left(\frac{R'_2}{R'_2 + R'_3} \cdot \frac{R_3 + R_2}{R_2} \right)^2 \\
 &\quad + V_{OP1}^2 \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_{mic}^2 \cdot \left(1 + \frac{2R_1}{R_g} \right)^2 + I_{OP1+}^2 \cdot R_1^2
 \end{aligned} \tag{5.9}$$

Using equation 5.8 the SNR of the main amplifier can be plotted as a function of the gain (see figure 5.11) using the output level of the microphone at 94 dBu , 1 kHz sine (see table 3.1) as signal level.

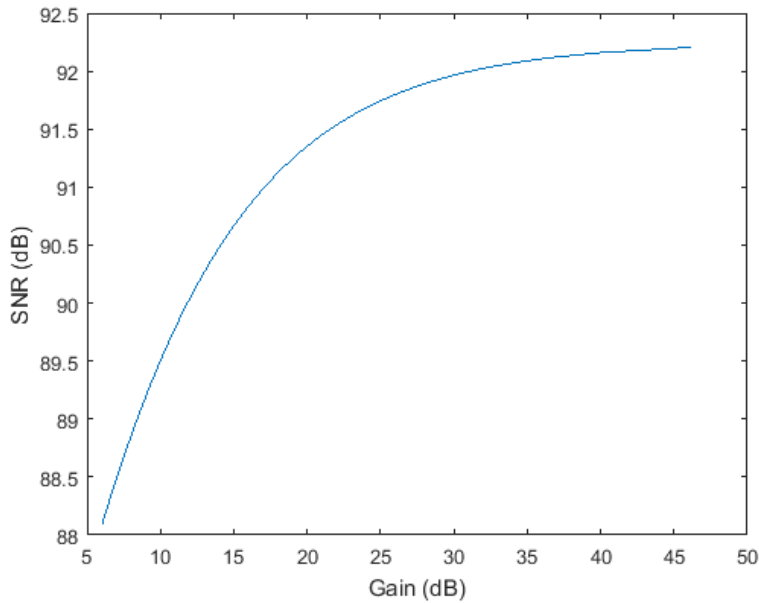


Figure 5.11: The SNR of the main amplifier as a function of its gain.

By summation of the calculated noise of the main amp using equation 5.5 it is possible to calculate the total theoretical SNR degradation caused by the main amplifier. It can be seen in figure 5.12.

Assuming a fixed gain of 30 dB it is also possible to calculate the theoretical SNR degradation of the main amplifier depending on the OP amplifiers voltage noise (current noise is kept constant since it has a very low impact when low resistor values are used). The result can be seen in figure 5.13.

To get a theoretical SNR degradation of less than 1 dB the voltage noise of the chosen OP amplifier must be less than $24.65\text{ nV}/\sqrt{\text{Hz}}$. To get a theoretical SNR degradation of less than 0.1 dB the voltage noise must be less than $7.326\text{ nV}/\sqrt{\text{Hz}}$. The practical implementation will have a higher noise due to external noise sources affecting the circuit so to achieve the desired SNR degradation a slightly better noise performance of the OP would probably be required.

According to Frii's formula 4.2 the stages after an amplifier (with gain higher than 0 dB) is less critical in terms of noise. This means that the requirements on the noise performance of the following stages are lower than for the main amplifier. The maximum allowed input referred noise of the following stages to degrade the SNR a maximum of 1 dB and 0.1 dB will be calculated. This will be done for two

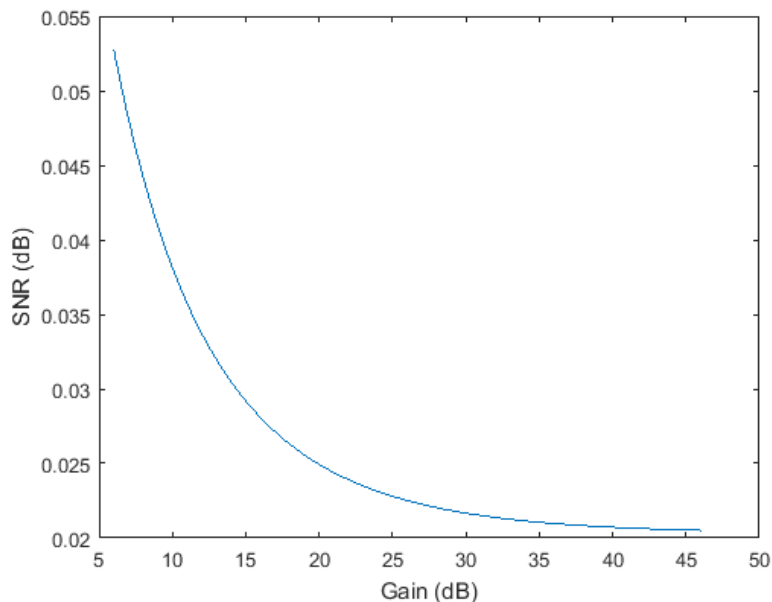


Figure 5.12: The SNR degradation caused by the main amplifier as a function of the gain.

different cases: when the gain is set for optimal capture of a 94 dBu signal, which is the signal power of standard SNR measurements, and when the gain is set for optimal capture of a 124 dBu signal, which corresponds to the acoustic overload point of the microphone. The latter can be considered the worst case scenario since it is rare for a surveillance camera to be exposed to such high acoustic signal levels. The result of the calculations is compiled in table 5.1.

In the worst case, when the gain is adapted for 124 dBu and the SNR is allowed to be degraded by no more than 0.1 dB , the allowed input referred noise of the following stages is $590\text{ }\mu\text{V}$ which is almost a hundred times higher than expected. It is therefore concluded that the noise of the following stages will not affect the SNR notably and therefore no noise calculations will be made for the later stages.

	Maximum Input Referred Noise (mV_{RMS})
94 dBu , 1 dB degradation	45.5
94 dBu , 0.1 dB degradation	41.4
124 dBu , 1 dB degradation	1.938
124 dBu , 0.1 dB degradation	0.590

Table 5.1: Measured specifications for the main amplifier.

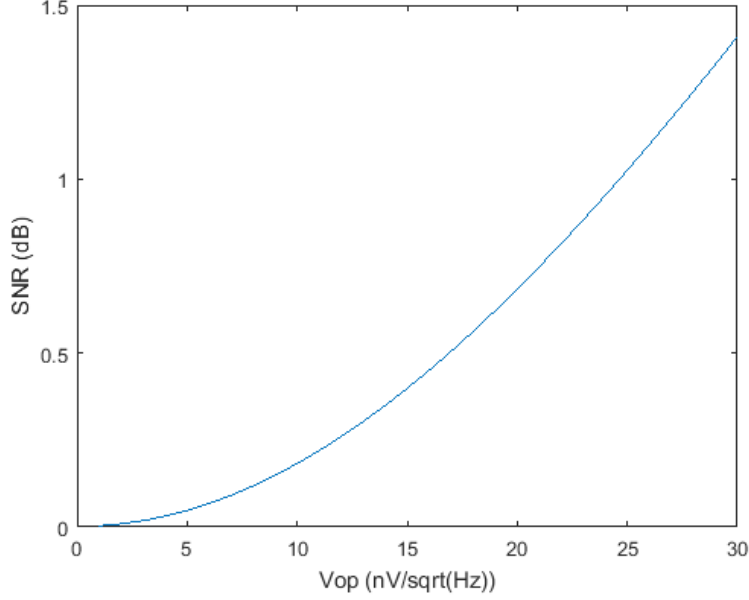


Figure 5.13: The SNR degradation caused by the main amplifier as a function of the OP amplifier voltage noise with a gain of 30 dB and an OP amplifier current noise fixed to $1 \text{ pA}/\sqrt{\text{Hz}}$.

5.4.3 Frequency Response

The frequency response for the main amplifier should be determined by the high frequency low pass filter at the input, since the OP amplifiers used has a very high bandwidth, and the DC-block. The low pass filter has a differential RC configuration and so the equation 5.6 can be used to calculate the cutoff frequency, f_c .

$$f_c = \frac{1}{2 \pi R_{mic} C_{equivalent}} = \frac{1}{2 \pi \cdot 325 \Omega \cdot 1180 \text{ pF}} = 415 \text{ kHz},$$

where R_{mic} is the resistance of each output of the microphone and is assumed to be 375Ω and $C_{equivalent}$ is the total equivalent capacitance if the differential low pass filter were to be represented as a single ended.

The cutoff frequency, f_c , of the DC-block should be well below 20 Hz to not affect the audible frequency range. The cutoff frequency for the DC-block is calculated below.

$$f_c = \frac{1}{2 \pi R_{dc} \cdot C_{dc}} = \frac{1}{2 \pi \cdot 100 \text{ k}\Omega \cdot 1 \mu\text{F}} = 1.59 \text{ Hz}.$$

5.5 Simulation

Simulations are made using PSpice to make sure that the design performs as desired.

5.5.1 Noise

The noise is simulated to make sure that the noise calculations are accurate enough. The noise of the main amplifier and its connection network is simulated using PSpice. The output noise for the lowest possible gain of $A = 2$ is displayed in figure 5.14.

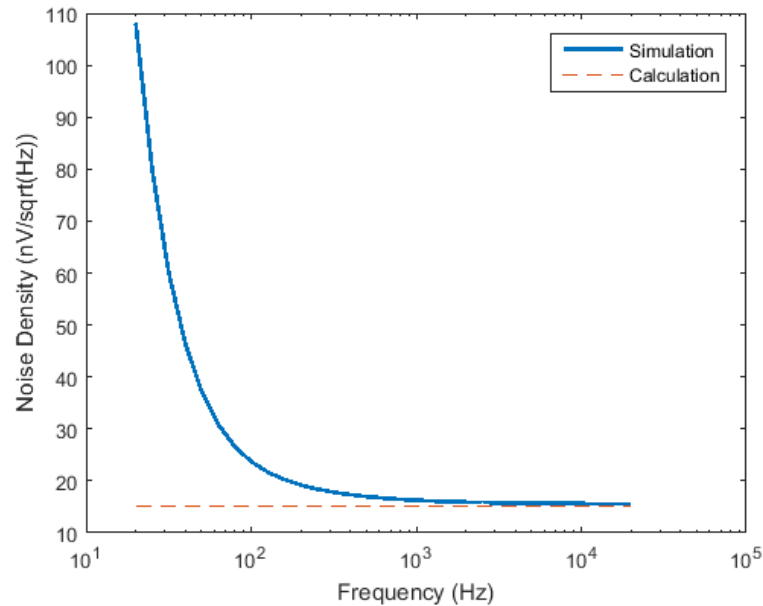


Figure 5.14: The noise density of the main amplifier as a function of the frequency.

The corner frequency, f_{corner} , is just above a hundred hertz which makes the approximation that the noise is white fairly good. For higher frequencies the noise density is observed to be $E_{MainAmp} = 15.37 \text{ nV}/\sqrt{\text{Hz}}$. Equation 5.8 gives the noise density of $E_{MainAmp} = 15.00 \text{ nV}/\sqrt{\text{Hz}}$ which is a difference of less than 2.5 % and is considered accurate enough. The reason the simulated data differs slightly from the calculated is probably because the simulation considers the current noise from the resistors which is neglected in the calculations.

5.5.2 Frequency Response

An AC analysis is made with PSpice to verify that the frequency response is as desired (see figure 5.15).

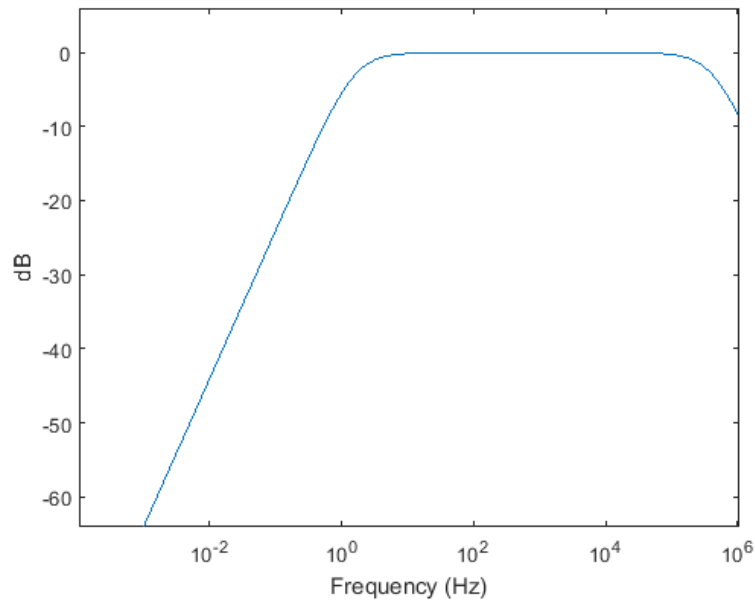


Figure 5.15: The simulated frequency response of the main amplifier from 1 *mHz* to 1 *MHz*.

As seen in figure 5.15 the cutoff frequency for the DC-block is $f_c = 1.59 \text{ Hz}$ and for the low pass filter it is $f_c = 415 \text{ kHz}$ which both matches the calculations.

5.5.3 CMRR

The CMRR was simulated by applying the same input signal on both inputs of the main amplifier. The input signals were 1 *kHz* sine waves with the amplitude of 500 *mV* with the gain set to $A = 2$. This yield a signal at the output with the amplitude of 3.16 μV which gives a common mode amplification of $A_{cm} = 3.16 \mu$. Using equation 4.7 the following CMRR is obtained.

$$CMRR = 20 \log \left(\frac{2}{3.16 \mu} \right) = 116 \text{ dB}$$

The actual CMRR will be less due to mismatch in components.

5.6 Implementation

Before the circuit can be realised a schematic with available components has to be drawn. This schematic will also include some components that is needed for a practical implementation such as decoupling capacitors for the power supply connections, connection pins and the microphone.

The PCB layout of the main amplifier is designed together with an experienced CAD engineer to achieve best possible performance. The layout can be seen in figure 5.16.

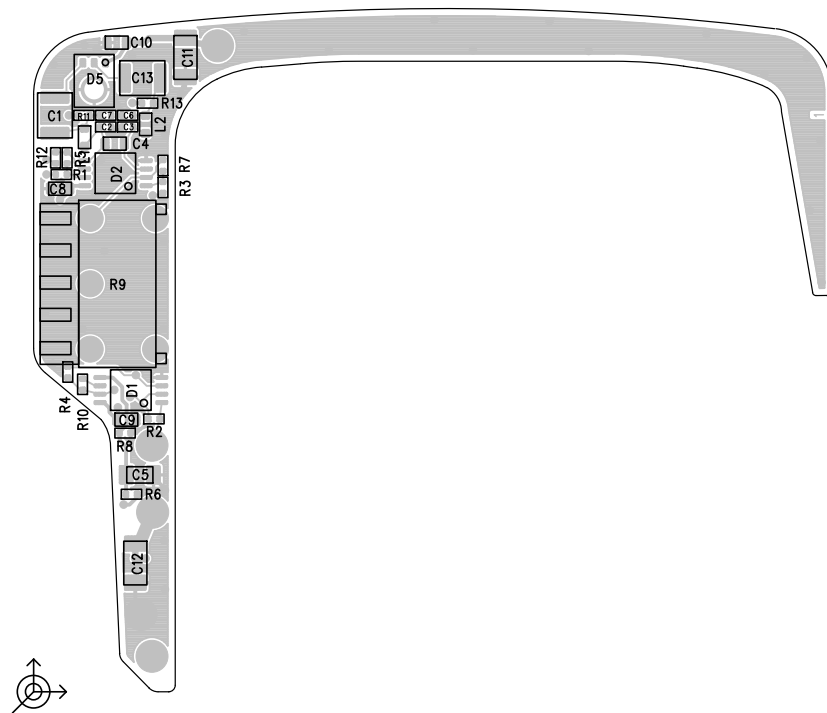


Figure 5.16: The PCB layout of the main amplifier.

5.7 Measurements

Measurements on the implemented product is made to see the actual performance of the main amplifier and the main amplifier together with the microphone, inside the camera. The measured specifications is shown in tables 5.2 and 5.3 and the frequency response is shown in figures 5.17 and 5.18.

The "Low Gain" values are measured with the lowest gain (6 dB) and the "High Gain" values are measured at around 38 dB gain to avoid clipping. All specifications are measured with the expected input signal level when the intended

	Low Gain	High Gain	Expected Best
Gain (dB)	6.009	45.367	6.021 – 46.064
CMRR (dB)	69.574	81.2	116
Noise (μV_{RMS})	8.0	56.86	2.0
THD (%)	0.0075	0.064	-
IMD (dB)	-82	-72	-
SNR (dB)	76	91.4	92.2

Table 5.2: Measured specifications for the main amplifier.

	Low Gain	High Gain	Expected Best
Noise (μV_{RMS})	30	1100	19.5
SNR (dB)	65.42	66.67	68.90
EIN (dBu)	28.38	27.33	25.5

Table 5.3: Measured specifications for the main amplifier with the microphone, inside the camera.

microphone is exposed to a 94 dB sine at 1 kHz (see table 3.1).

5.8 Evaluation

Gain

The gain range is almost exactly as expected. The slight deviation is probably due to deviations in the component values.

CMRR

The CMRR is much worse than the expected value. This is probably due to deviations in the component values since the CMRR is very sensitive to this. A CMRR above 80 dB is still considered good enough.

Noise Performance

The noise performance is not as good as expected. The additional noise can depend on several factors: the sound lab used is not ideal; the cables are affected by electromagnetic fields which increases the noise, especially since the output of the main amplifier is single ended. The noise performance is slightly better when the amplifier is placed inside a tin container. Also, the noise from the bias voltage, which will be present at the input, is not taken into account in the expected result.

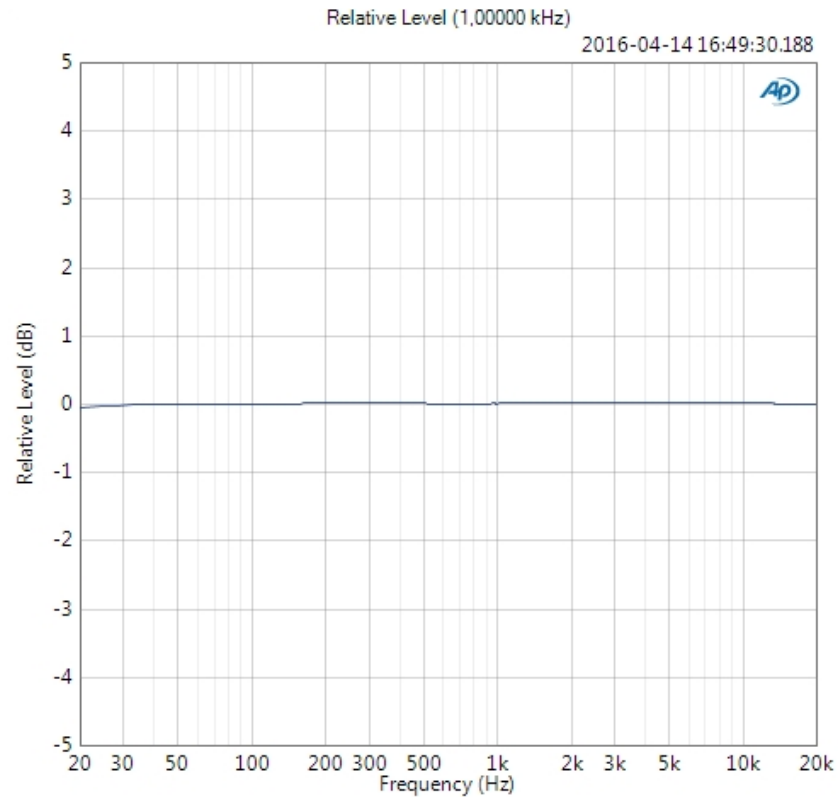


Figure 5.17: The measured frequency response of the main amplifier from 20 Hz to 20 kHz.

THD

The THD is below 0.1 % in the worst case which is good enough since a THD of 1 % is barely audible.

Frequency Response

The frequency response of the main amplifier is flat as expected but when the microphone is placed inside the camera the frequency response changes compared to when the microphone was measured as a standalone component. The following filters will be adapted to this new frequency response.

SNR and EIN

The SNR is almost 2 dB worse than expected. This is due to the noise being higher than expected since the signal is amplified correctly.

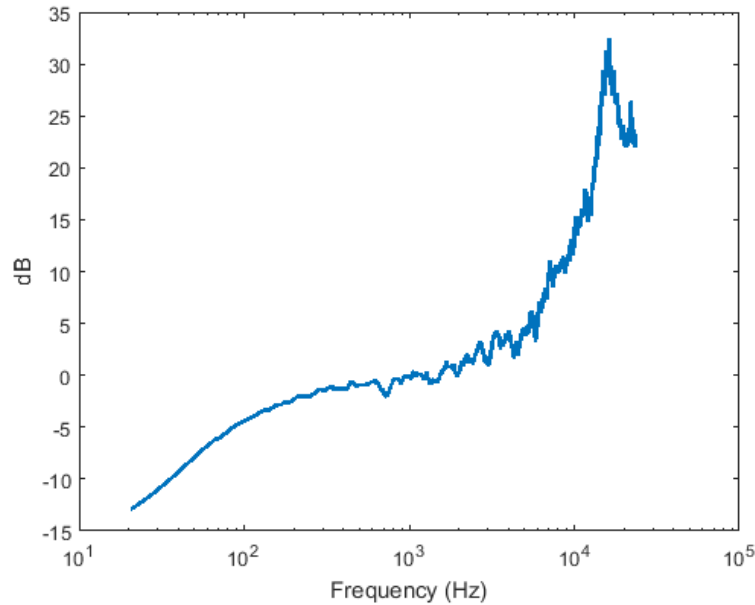


Figure 5.18: The measured frequency response of the main amplifier with microphone, inside the camera.

5.8.1 Possible Improvements

To avoid that the DC level varies with the gain a DC-block is placed after the main amplifier. One typically desires not to have capacitors in the signal path to avoid microphonic effects and to avoid too much flicker noise. This is possible to do by instead placing a capacitor in series with the gain resistor R_g in the main amplifier (see figure 5.7). Although, since the resistor values were chosen small for a good noise performance the capacitor would have to be very large which would become unpractical. By choosing higher resistor values it would be possible to use a reasonably large capacitor in series with the gain resistor and thereby avoiding the DC-block. Although, this would introduce more noise from the resistors.

Voltage Regulation of Power Supply

Two different voltages will be taken from the camera: $+3\text{ V}$ for the microphone and $+12.5\text{ V}$ for driving the OP amplifiers and to obtain the bias voltage. The power supply has to be stable for a number of reasons: the bias voltage is obtained by bisecting the power supply voltage using a voltage divider, which means that the noise of the bias voltage will be the noise of the power supply divided by two. If the power supply is not stable and noise free it is possible that the OP amplifiers become unstable. The power supply voltage must be fixed in order to get the correct bias voltage at all time.

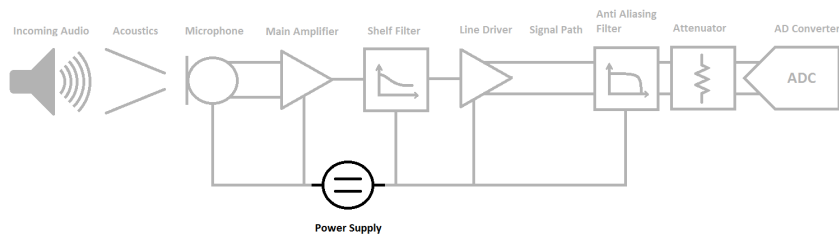


Figure 6.1: Block diagram of the audio chain with the voltage source highlighted.

The bias circuit can be seen in figure 6.2.

6.1 Initial Measurements

Before it is decided what regulation is required the stability and noise of the power supply is measured. The result can be seen in table 6.1.

The power supply seems to be stable but somewhat noisy. The microphone has a power supply rejection ratio of -45 dB which would result in a noise of less than $2\text{ }\mu\text{V}_{RMS}$ applied to the signal which would cause about 0.13 dB degradation of the SNR. The power supply rejection ratio of the OPA1662 is 110 dB and 95 dB for the LME49724 which should have no significant effect on the signal at all. The

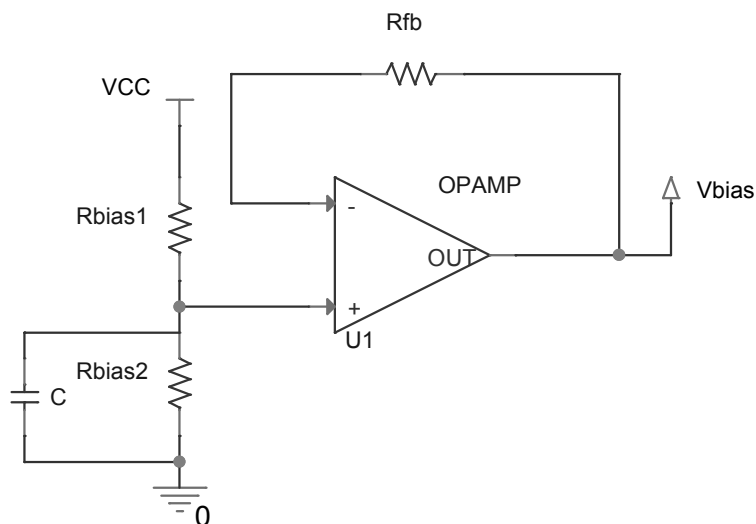


Figure 6.2: The bias circuit schematic.

	100 k Ω load	600 Ω load
Voltage Level +12.5 V (V)	12.53	12.50
Voltage Level +3V (V)	2.994	2.987
Noise +12.5 V (μV_{RMS})	174	176
Noise +3 V (μV_{RMS})	306	215.1

Table 6.1: Measured stability and noise of the power supply.

bias voltage circuit have a power supply rejection ratio of only 6 dB. Therefore it is important that the +12.5 V is filtered to remove the noise.

6.2 Noise Reduction

The +3 V for the microphone is filtered using a 0.1 μF capacitor to further reduce the effect of the power supply noise on the signal. The +12.5 V is also decoupled using 0.1 μF to make sure that the OP amplifiers is kept stable. The bias voltage is filtered using 10 μF creating a low pass filter together with a resistor in the voltage divider. Since the resistor value is 1 k Ω the low pass filter should have a cutoff frequency of 15.9 Hz and the 20 dB/decade attenuation of a first order low pass filter.

6.3 Calculations

6.3.1 Noise

The total noise of the bias circuit is calculated assuming no noise from the power supply. The bias circuit along with the noise sources can be seen in figure 6.3.

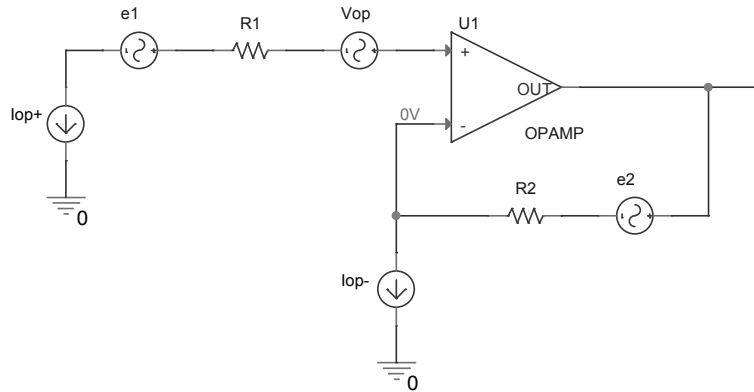


Figure 6.3: The bias circuit equivalent with noise sources.

Thermal Noise

$$E_1 = e_1 = \sqrt{4kTR_1}$$

$$E_2 = e_2 = \sqrt{4kTR_2}$$

$$\Rightarrow E_{th} = \sqrt{4kTR_1 + 4kTR_2}$$

OP Amplifier Noise

$$E_{OPV} = V_{OPA} = V_{OP}$$

$$E_{OPI+} = I_{OP+}R_2 \quad A = I_{OP+}R_2$$

$$E_{OPI-} = I_{OP-}R_1$$

$$\Rightarrow E_{OP} = \sqrt{V_{OP}^2 + I_{OP+}^2 R_2^2 + I_{OP-}^2 R_1^2}$$

Total Noise Density

$$E_{tot} = \sqrt{4kTR_1 + 4kTR_2 + V_{OP}^2 + I_{OP+}^2 R_2^2 + I_{OP-}^2 R_1^2}$$

Total Noise

The total noise is calculated considering only the audible range and excluding the flicker noise.

$$Noise = E_{tot} \sqrt{20000 - 20} = 0.740 \mu V_{RMS}$$

This should degrade the SNR with about 0.02 dB.

6.3.2 Simulations

Noise

The noise added to the bias voltage by the bias circuit is simulated to make sure it matches the calculated value. The simulated noise density can be seen in figure 6.4.

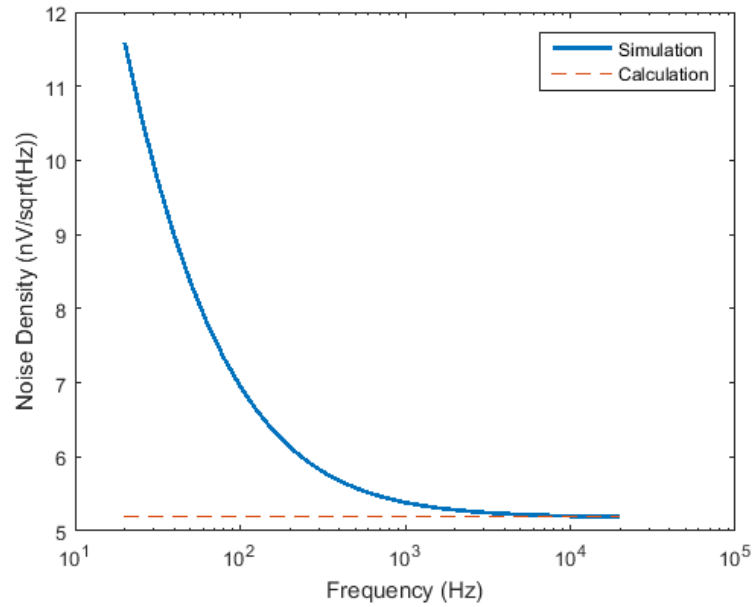


Figure 6.4: The noise power from the voltage source on the bias voltage for different capacitor values.

To find out the effect of the noise from the voltage source, simulations of the bias circuit are made using an input representing the noise density. The circuit is then simulated using different values of the capacitor to find out which one is most ideal for the implementation. The simulations can be seen in figure 6.5.

By integrating the curves in figure 6.5 the total noise applied to the bias voltage is obtained for each value of C . The result, as well as its impact on the SNR given the noise and signals level of the microphone, can be seen in table 6.2.

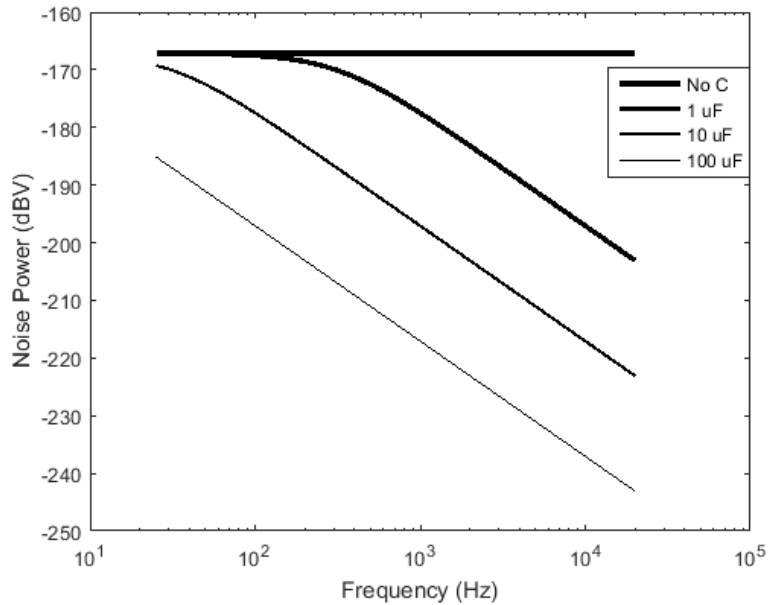


Figure 6.5: The noise power from the voltage source on the bias voltage.

	Noise (μV_{RMS})	SNR degradation (dB)
No C	87.989	19.20
1 μF	6.7081	1.69
10 μF	0.9059	0.03
100 μF	0.09440	0.0004

Table 6.2: Noise and SNR degradation of the signal depending on the capacitor value.

Using a 10 μF capacitor is good enough since it degrades the SNR by only 0.03 dB . Moreover, having a capacitor greater than 10 μF would be unpractical.

6.4 Measurements

6.4.1 Noise

The noise of the bias voltage is measured to be 8.0 μV_{RMS} which is considerably higher than expected. This would degrade the SNR with about 2.25 dB .

6.5 Evaluation

The measured noise is higher than expected. This is probably caused by external interference. When measuring the noise output from the main amplifier, a value of $8 \mu V_{RMS}$ is obtained (see table 5.2) even though the main amplifier has a minimum gain of 2. This suggests that the bias noise is significantly less than $8 \mu V_{RMS}/2 = 4 \mu V_{RMS}$ since the bias noise is placed directly on top of the signal before amplification. The reason the bias noise is much less in this case is probably because the bias voltage in this case only travels on the PCB and not in external transmission lines making it less sensitive to interference. The actual noise will probably be even lower when integrated in the electronics since the transmission lines will be much shorter. Also, the electronics will be placed inside the metal casing of the camera which will further reduce the noise.

6.5.1 Possible Improvements

The bias voltage is a critical component since its noise will be placed right on top of the signal. For the test implementations in this project the bias voltage will travel through several external transmission lines making it susceptible to interference. For a real implementation it is advised to keep the transmission lines for the bias voltage as short and well shielded as possible.

In order to achieve a desired frequency response the signal will be filtered. One typically desires a flat frequency response for forensic audio. It is in this case possible that a peak for high frequencies is desired since high frequencies attenuates faster than low frequencies (see equation 3.1) and the camera is usually placed far away from the sound source.

The filters will have to compensate for the microphones non flat frequency response as well as possible acoustic effects due to the camera housing.

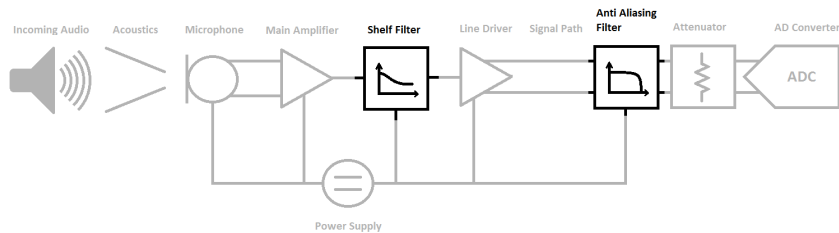


Figure 7.1: Block diagram of the audio chain with the filters highlighted.

7.1 Theory

The transfer function for a first order low pass filter has the following form:

$$A(s) = \frac{A_0}{1 + as}, \quad (7.1)$$

where a is a constant determined by the component values and A_0 is the filter gain.

More specifically the transfer function of a passive first order RC low pass filter is the following.

$$A(s) = \frac{\frac{1}{RC}}{s + \frac{1}{RC}} = \frac{1}{1 + sRC}, \quad (7.2)$$

where $s = j\omega$ and ω is the angular frequency. In this case $a = RC$.

For a second order low pass filter that allows complex poles the transfer function has the following form.

$$A(s) = \frac{A_0}{(1 + as + bs^2)}, \quad (7.3)$$

where a and b are constants determined by the component values.

It is possible to achieve higher order filters by cascading second or first order filters. The transfer function for higher order filters can be written in the following form:

$$A(s) = \frac{A_0}{(1 + a_1 s + b_1 s^2)(1 + a_2 s + b_2 s^2) \dots (1 + a_n s + b_n s^2)}, \quad (7.4)$$

where a and b are constants determined by the component values of each filter. a and b also determines which configuration the filter is in.

7.1.1 Filter Orders

The order, n , of the filter determines how fast a signal will be attenuated. For passive RC filters the first order filter attenuates the signal 20 dB/decade before/after the cutoff frequency. The attenuation increases with the order of the filter as follows.

$$\text{Attenuation} = n \cdot 20 \text{ dB/decade} \quad (7.5)$$

The order also determines how flat the frequency response will be for the unattenuated signal band before/after the cutoff frequency. [17]

7.1.2 Passive/Active Filters

A filter can be passive, which means that it contains only passive components, or active, which means that it contains one or more active components, such as OP amplifiers, that requires a power supply. With an active filter it is possible to obtain complex poles in the system without the need for inductive components, which are in general big and expensive. With complex poles it is possible to further tweak the filters to obtain the desired performance. [17]

7.1.3 Filter Configurations

There are several different filter configurations that each have their benefits and drawbacks. The filter configurations are determined by the values of a and b . A specific filter configuration will require specific values of a and b . To implement a specific filter the component values are to be chosen to match the required a and b . [17]

Butterworth Filter

A Butterworth filter configuration is used when the pass band flatness is the most important aspect. This filter attenuates the pass band signal as little as possible before the cutoff frequency. The drawback is that it does not have a linear phase shift, which will cause distortion. [17]

Chebyshev Filter

A Chebyshev filter is chosen when one desires the maximum possible attenuation above the cutoff frequency. This filter has the sharpest possible transition between the pass band and the stop band. The drawbacks are that the pass band is not flat up until the cutoff frequency nor does this filter have a linear phase shift. [17]

Bessel Filter

The Bessel filter is used to achieve minimum distortion since it has a linear phase shift. Due to this feature the flatness and attenuation slope suffers. [17]

7.1.4 Active Filter Topology

First Order Low Pass Filter

A first order filter can be implemented using a simple RC circuit but with an OP amplifier as a buffer to reduce the output resistance and reduce the load from the following stages.

Second Order Low Pass Filter

One of the most simple ways to implement a second order filter is to use the Sallen-Key topology (see figure 7.2). [17]

Shelf Filter

Sometimes one does not desire to attenuate the signal to minus infinity but to a certain level. In that case a shelf filter can be used. A shelf filter can be implemented by placing a filter on the feedback of an OP amplifier. This implementation can be seen in figure 7.3.

The low frequency signal is amplified to a desired value while the high frequency signals see the amplifier configuration as a unity gain follower. Therefore only the desired low frequency range is amplified while the rest is unaffected creating a shelf in the frequency response.

7.2 Shelf Filter

A shelf filter is used to compensate for the microphones characteristics as well as for the acoustic effects of the camera in order to get a flat frequency response for low frequencies. The shelf filter that will be used has the configuration shown

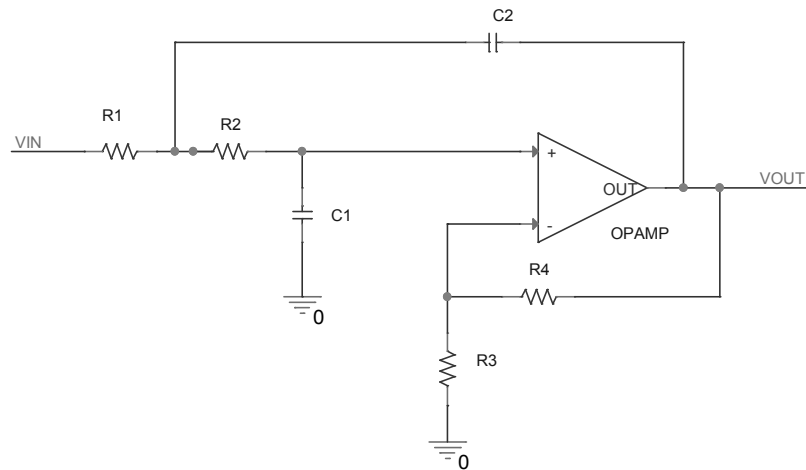


Figure 7.2: An active second order low pass filter using the Sallen-Key topology.

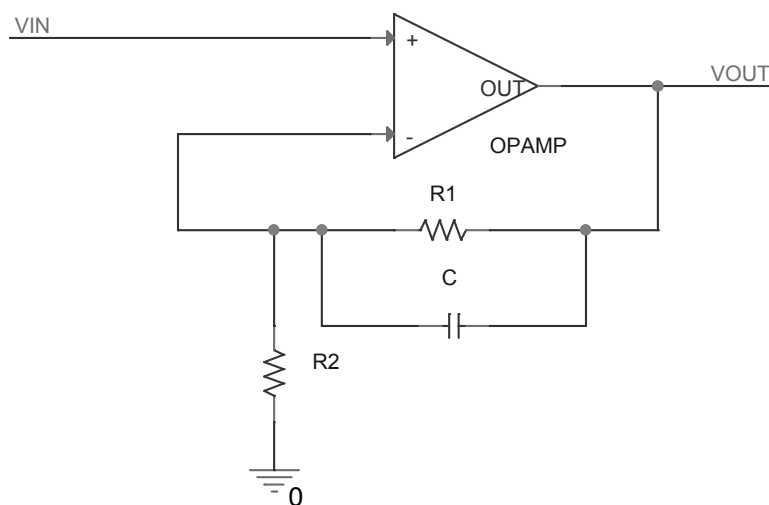


Figure 7.3: A low pass shelf filter.

in figure 7.3. The gain and cutoff frequency of the filter will be based on the frequency response of the microphone inside the camera shown in figure 5.18. The gain for the low frequency signals is set to 11 *dB* and the cutoff frequency set to 130 *Hz*.

7.2.1 Calculations

Frequency Response

The component values are chosen by solving the following equations 7.2 and 5.1:

$$f_c = \frac{1}{2\pi R_2 \cdot \frac{CR_1}{C+R_1}}$$

and

$$Gain_{LowFreq} = 1 + \frac{R_1}{R_2}$$

to match the desired values. When C is chosen to be $1 \mu F$ the resistors is to be $R_1 \approx 4.7 k\Omega$ and $R_2 \approx 1.5 k\Omega$ rounded to the nearest standard component value.

7.2.2 Simulations

Frequency Response

The simulated frequency response of the shelf filter can be seen in figure 7.4.

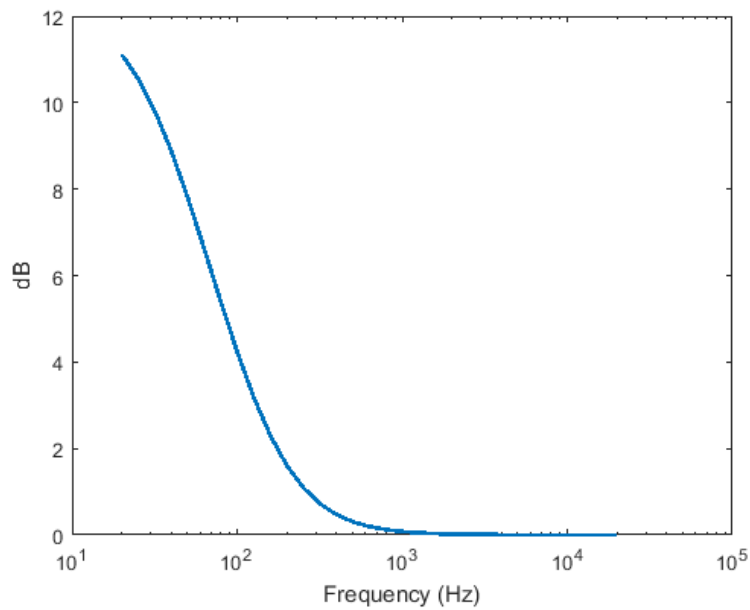


Figure 7.4: The frequency response of the shelf filter.

The frequency response simulation shows a cutoff frequency close to $130 Hz$ and a gain for low frequencies of about $11 dB$.

7.2.3 Measurements

Frequency Response

The measured frequency response of the shelf filter can be seen in figure 7.5.

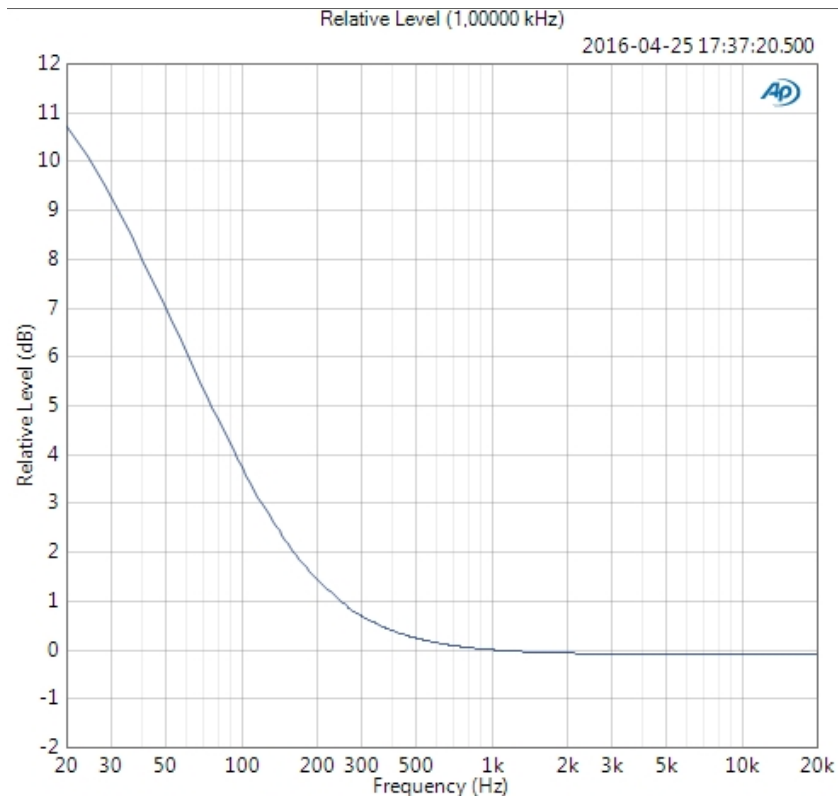


Figure 7.5: The measured frequency response of shelf filter.

The cutoff frequency seems to be close to the desired $f_c = 130 \text{ Hz}$ and the gain for low frequencies is 10.6 dB .

Noise

The measured noise is $1.694 \mu V_{RMS}$ which gives an SNR of 104.06 dB with the intended input signal. This is high enough to barely affect the total noise performance at all.

THD

The THD is measured to be 0.099% which is considered acceptable since it is under 0.1% .

7.3 Anti Aliasing Filter

To make sure that the aliasing does not occur due to interfering high frequency signals as well as to remove noise outside the audible frequency range an anti aliasing filter is implemented. It will be placed as close to the codec as possible to remove as much interference as possible that might have been added in the signal path.

7.3.1 Calculations

Frequency Response

The anti aliasing filter should have a cutoff frequency just over 20 kHz to not attenuate the audible frequency range too much. A Bessel filter configuration is desired since it is important to have a linear phase shift to not distort the signal. A certain amplification is also desired to be able to adapt the signal level to the codec. The codec has an oversampling of 128 times and the signal should be attenuated by at least 80 dB at this point. This will require a second order low pass filter. The Sallen Key configuration shown in figure 7.2 is used.

By writing the filter characteristics on the form shown in equation 7.3:

$$A = \frac{A_0}{(1 + 2\pi f_c(C_1(R_1 + R_2) + (1 - A_0)R_1C_1)s + (2\pi f_c)^2 R_1R_2C_1C_2s^2)}$$

it is possible to identify the constants a and b :

$$a = 2\pi f_c(C_1 * (R_1 + R_2) + (1 - A_0)R_1 C_2)$$

$$b = (2\pi f_c)^2 R_1 R_2 C_1 C_2$$

The component values should be chosen so that $a = 1.3617$ and $b = 0.618$ which are the constant values for a second order Bessel filter. [17] C_1 is chosen to 22 nF and C_2 is chosen to be 47 nF which is larger than $\frac{4C_1b}{a^2}$ which is required to get real resistor values. This gives the resistor values of $R_1 = 39 \Omega$ and $R_2 = 270 \Omega$ rounded to the nearest standard component value.

The component values of R_3 and R_4 are calculated using equation 5.1 to get the desired gain of 5.34 dB . $R_3 = 2 k\Omega$ and $R_4 = 4.7 k\Omega$.

7.3.2 Simulations

Frequency Response

The simulated frequency response of the anti aliasing filter can be seen in figure 7.6.

The frequency response simulation shows a cutoff frequency close to 20 kHz and an attenuation of 40 $dB/decade$ above the cutoff frequency. This will yield an attenuation of 84 dB 128 times above the sampled frequency range (which is the oversampling of the codec) which should be enough to prevent aliasing.

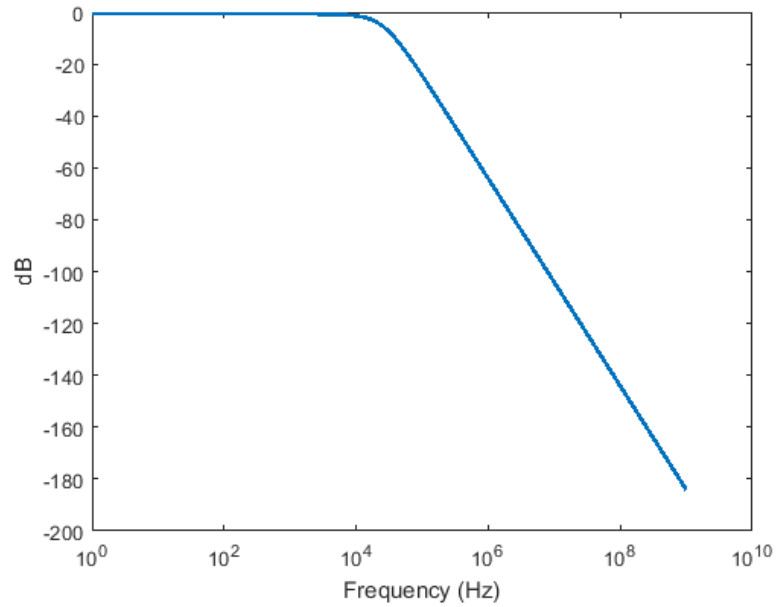


Figure 7.6: The frequency response of the anti aliasing filter.

7.3.3 Measurements

The measured specifications is compiled in table 7.1.

Frequency Response

The measured frequency response of the anti aliasing filter can be seen in figure 7.7.

The cutoff frequency is slightly above 20 *kHz*.

	Measured	Expected
Gain (<i>dB</i>)	5.389	5.352
CMRR (<i>dB</i>)	69.88	-
Noise (μV_{RMS})	12	-
THD (%)	0.096	-
SNR (<i>dB</i>)	104.48	-

Table 7.1: Measured specifications for the anti aliasing filter.

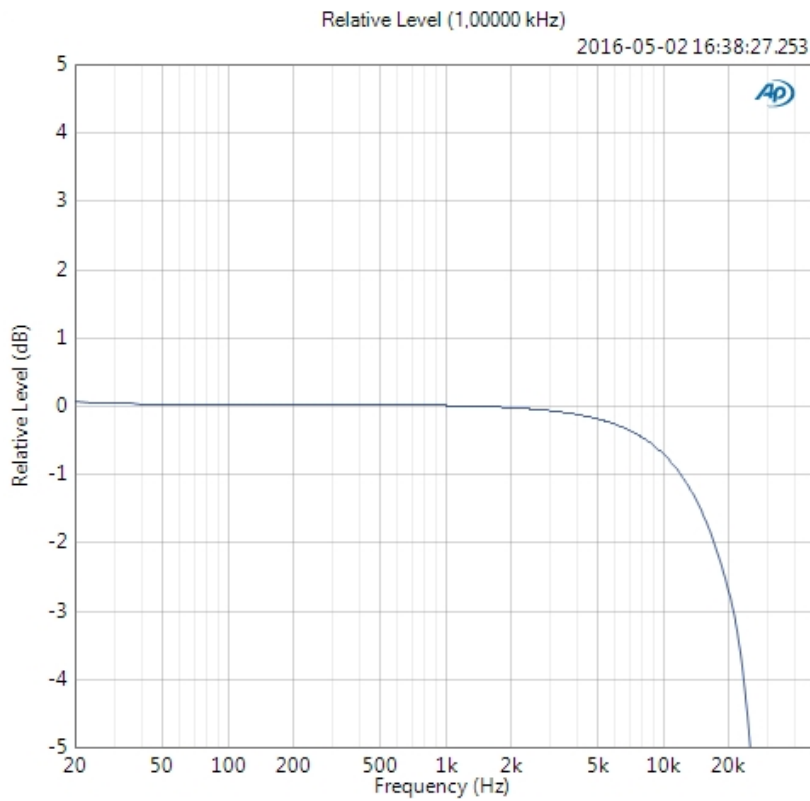


Figure 7.7: The measured frequency response of the anti aliasing filter.

Gain

The gain is close to the desired. The small deviation is because of deviations in component values.

Noise

The measured noise is $12 \mu V_{RMS}$ which gives an SNR of 104.48 dB with the intended input signal. This is high enough to barely affect the total noise performance at all.

THD

The THD is measured to be 0.096% which is considered acceptable since it is under 0.1% .

CMRR

The CMRR is less than desired. This is because of deviations in component values. To avoid this problem components with lower error tolerance should be used.

7.4 Evaluation

Both filters perform as desired with low noise, low distortion and a desired frequency response.

7.4.1 Possible Improvements

Since the acoustic performance can differ greatly between cameras the filters would have to be tuned specifically for each camera to achieve optimal performance. The shelf filter can easily be tuned to achieve a new gain for low frequencies and a new cutoff frequency just by changing the component values. The cutoff frequency of the anti aliasing filter can also easily be adjusted to start attenuating lower frequencies by changing component values.

It is of course possible, and likely, that these filters will not be enough to achieve the desired frequency response for all cameras but since the signal is converted to single ended by the main amplifier, adding additional filters should be easy and require a small amount of components.

Signal Transport

The signal must be transported from the microphone to the codec. The environment between the microphone and the codec is usually very noisy because it often travel near noisy components such as processors and the camera motors. In order to make the transmission line as noise resilient as possible a line driver is used. The line driver converts the single ended signal to a differential signal while being able to output a relatively large amount of current. The signal is also amplified to further reduce the impact of induced noise.

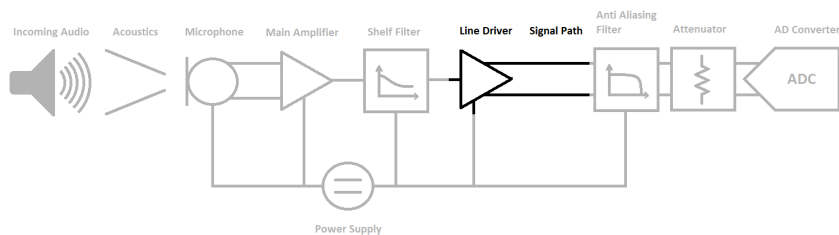


Figure 8.1: Block diagram of the audio chain with the signal transport circuits highlighted.

8.1 Theory

8.1.1 Differential Signal Transport

The signal to be transported is divided into two equally large signals. One of these signals are inverted. Both of these signals will travel together to the desired location through a transmission line. When the signals reaches its destination the inverted signal is inverted back and added to the non-inverted signal. Since the two signals travel together both will be exposed to roughly the same interference. When one of the signals is inverted back and added to the non-inverted signal the interferences will be cancelled out resulting in an interference free signal. [7]

8.1.2 Parasitic Effects of transmission lines

A transmission line will have a certain amount of parasitic properties. A transmission line can be seen as a chain of small series inductors, series resistors, parallel capacitors and parallel resistors. These parasitic effects may attenuate the signal, affect the frequency response and make the signal more susceptible to noise. These effects can be reduced by having a line driver that is able to supply a large amount of current. [13]

8.2 Line Driver

By using a line driver with a high output current capacity the single ended signal is converted to differential and the parasitic effects of the transmission lines are reduced. The signal is also gained in the line driver to further reduce the effect of possible interference noise. The schematic of the implementation used can be seen in figure 8.2.

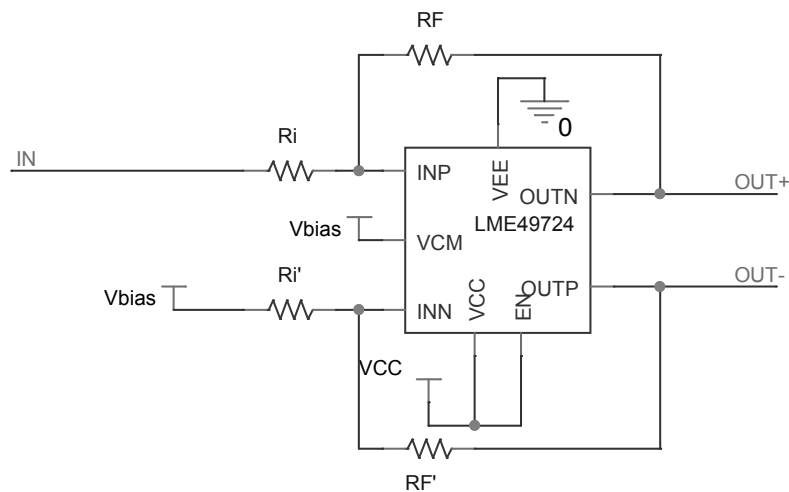


Figure 8.2: Schematic of the line driver.

8.3 Calculations

Gain

The desired gain is 3.5 dB. The component values are calculated using the following equation.

$$Gain = \frac{R_F}{R_i}$$

$$R_F = 150 \Omega \text{ and } R_i = 100 \Omega$$

8.4 Measurements

The measured specifications can be seen in table 8.1.

	Measured	Expected
Gain (dB)	3.5 dB	3.5 dB
Noise (μV_{RMS})	4.229	-
THD (%)	0.071755 (with main amplifier)	-
SNR (dB)	115.46	-

Table 8.1: Measured specifications for the line driver.

The frequency response can be seen in figure 8.3.

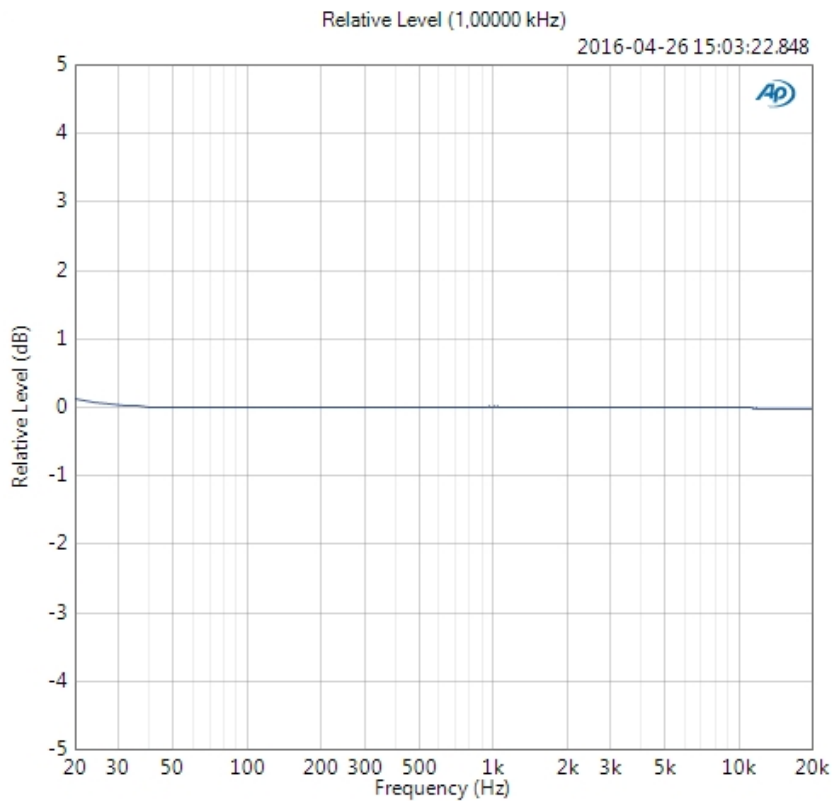


Figure 8.3: Measured frequency response of the line driver.

8.5 Evaluation

Gain

The gain is as desired.

Noise

The measured noise is $4.229 \mu V_{RMS}$ which gives an SNR of 115.46 dB with the intended input signal. This is high enough to barely affect the total noise performance at all.

THD

This measurement is made together with the main amplifier in order to get the correct bias voltage of the input signal. The THD is measured to be 0.071755% which is considered acceptable since it is under 0.1% .

8.5.1 Possible Improvements

The line driver is sensitive to component variations. If component values deviate from the intended there will be a DC-offset at the output which will ultimately cause a loss of dynamic range in the codec due to an incorrect bias voltage. This problem can be reduced by choosing components with a lower deviation tolerance or possibly by tuning the circuit using a trim potentiometer.

The line driver is used to transport the signal safely from the microphone electronics to the codec through noisy environments. The OP amplifier used for the line driver is relatively expensive and it is possible that this component is not needed if, for example, the microphone is placed close to the codec. In that case, the line driver can be removed and the signal fed to the codec single ended.

Signal Level Adaption

In order to adapt the signal level to the codec, which requires a bias voltage of 1.5 VDC and can handle a maximum voltage swing of 2.5713 V_{pp} on each input, an attenuator is required. By gaining the signal to a certain level it is possible to divide both the bias voltage and the signal voltage to the desired level with only one voltage divider.

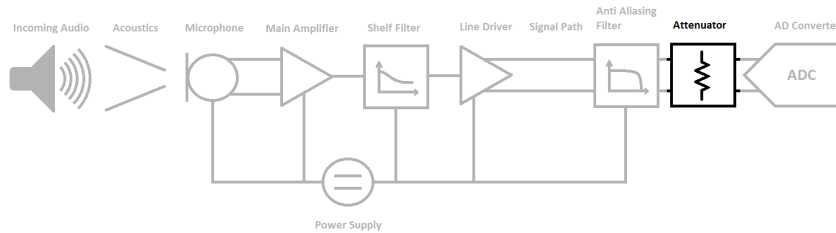


Figure 9.1: Block diagram of the audio chain with the signal attenuation circuit highlighted.

9.1 Theory

Voltage Division

Voltage can be scaled down and divided using resistors. When two or more resistors are placed in series over a voltage source the voltage will drop a certain amount over each resistor based on its relative value to the other resistors. The voltage drop over a given resistor, R_i , can be calculated using equation 9.1:

$$V_{OUT} = V_{IN} \cdot \frac{R_i}{R_1 + R_2 + \dots + R_n}, i = 1, 2, \dots, n. \quad (9.1)$$

[19]

9.2 Resistive Pad

Since both the signal level and the bias voltage should be scaled equally it is possible to use a simple voltage divider using resistors; a resistive attenuator also called resistive pad (see figure 9.2).

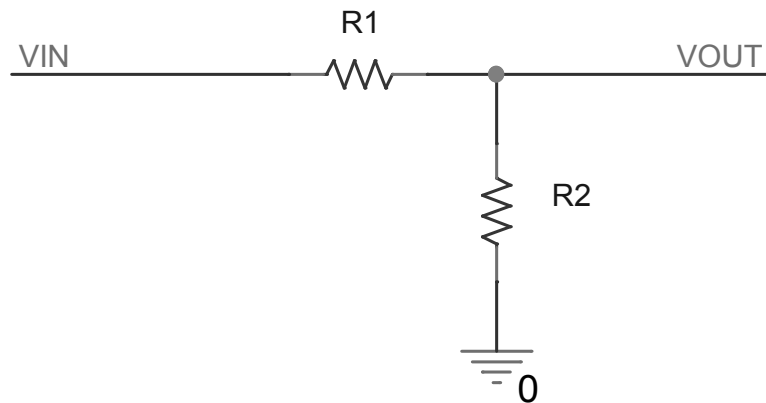


Figure 9.2: A voltage divider circuit.

9.3 Calculations

The intended level of the signal before the level adaption is $10.7 V_{pp}$ for both components of the differential signal and the bias voltage is $6.25 V_{DC}$. The maximum acceptable signal level after the voltage adaption is $2.5713 V_{pp}$ and the required bias is $1.5 V_{DC}$. The codec has an input impedance of $21 k\Omega$ which has to be taken into account.

Using equation 9.1 the resistor values are calculated to be $R_1 = 1.5 k\Omega$ and $R_2 = 488 \Omega$ (R_2 will be implemented with two resistors; 470Ω and 18Ω in series).

9.4 Measurements

The measured specifications can be seen in table 9.1.

	Measured	Expected
Gain (dB)	-12.325	-12.351
CMRR (dB)	85.925	-
Noise (μV_{RMS})	1.129	-
THD (%)	0.000142	-
SNR (dB)	118.896	-

Table 9.1: Measured specifications for the resistive pad.

9.5 Evaluation

Gain

The attenuation is slightly lower than desired. This is because the pad is not yet connected to the codec. The codec will add a resistance of $21\text{ k}\Omega$ in parallel with the output of the pad, increasing the attenuation slightly.

Noise

The measured noise is $1.129\text{ }\mu V_{RMS}$ which gives an SNR of 118.9 dB with the intended input signal. This is high enough to not affect the SNR notably.

THD

The THD is measured to be 0.000142% which is very low. The THD is mostly caused by the nonlinearities in the active components. Since the resistive pad only has passive components the THD is very low.

CMRR

The CMRR of 85.925 dB is good enough but could be better if components with lower deviation tolerance were used.

The components are connected as seen in figure 4.1 to form the complete electronics. The complete electronics chain is measured. Then the microphone is added and the electronics is placed in the camera and measured again. Finally, the electronics with microphone is connected to the codec inside the camera. The last measurements is then made on the signal when it has been converted to digital audio and converted back to analogue using the codec.

10.1 Measurements

10.1.1 Complete Electronics

The complete electronics is the main amplifier, shelf filter, line driver, anti aliasing filter and the resistive pad, in that order. The measured specifications are shown in table 10.1.

	High Gain	Expected
Gain (<i>dB</i>)	2.787 – 42.131	2.51 – 42.55
CMRR (<i>dB</i>)	80.863	-
THD (%)	0.081261	-
IMD (<i>dB</i>)	-57.08	-
Differential DC-Offset (<i>mV</i>)	12.24	-

Table 10.1: Measured specifications for the complete electronics chain.

The frequency response of the complete electronics chain can be seen in figure 10.1.

10.1.2 Complete Electronics with Microphone

The microphone is added and the electronics is placed inside the camera. The measured specifications can be seen in table 10.2.

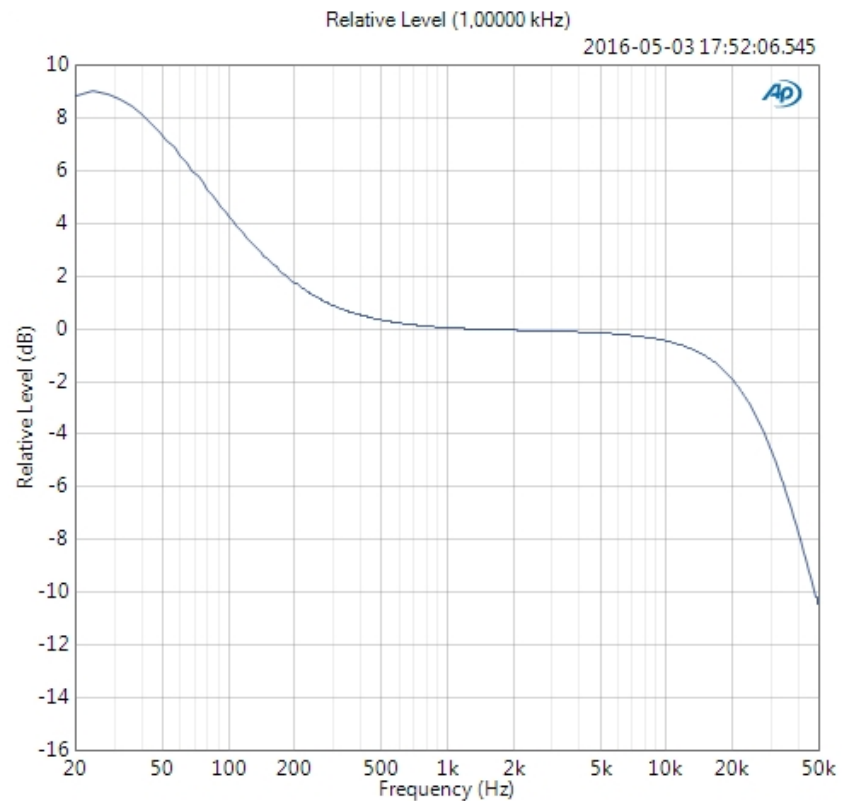


Figure 10.1: The measured frequency response of complete electronics chain.

	Low Gain	High Gain
THD (%)	0.80	0.67
SNR (dB)	66.58	67.17
EIN (dBu)	27.42	26.88
Differential DC-Offset (mV)	11.6	13.05
DC-Level (V)	1.535, 1.545	-

Table 10.2: Measured specifications for the complete electronics chain with microphone inside the camera.

The frequency response of the complete electronics chain with microphone, inside the camera, can be seen in figure 10.2.

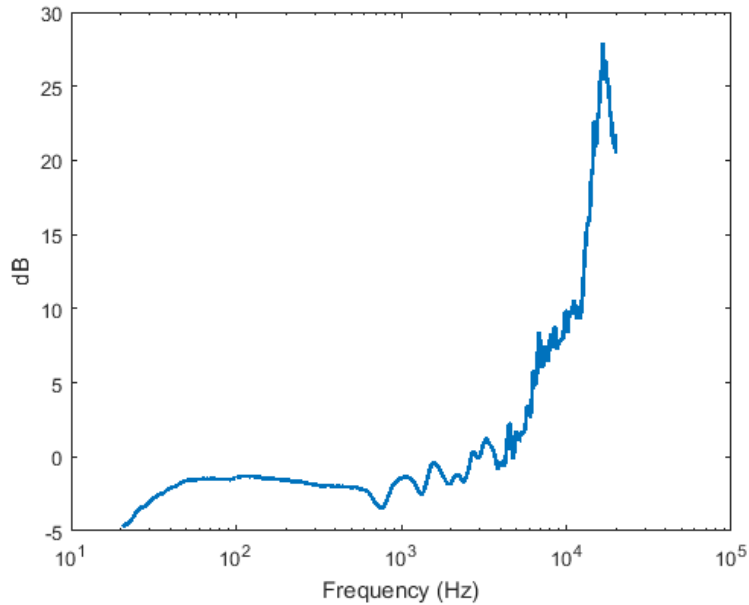


Figure 10.2: The measured frequency response of complete electronics chain with microphone, inside the camera.

10.1.3 Complete Microphone System

The complete electronics with microphone is now connected to the codec inside the camera to form the complete microphone system. The measured specifications can be seen in table 10.3.

	Low Gain	High Gain
THD (%)	0.77	0.75
SNR (dB)	61.67	67.07
EIN (dBu)	32.33	26.93
IMD (dB)	-33	-33

Table 10.3: Measured specifications for the complete microphone system.

The frequency response of the complete microphone system can be seen in figure 10.2.

Assuming an omnidirectional sound source, from which the sound is propagating as a sphere, the theoretical distance where the received signal level equals the noise level, can be calculated using the measured SNR. The result is shown in table 10.4.

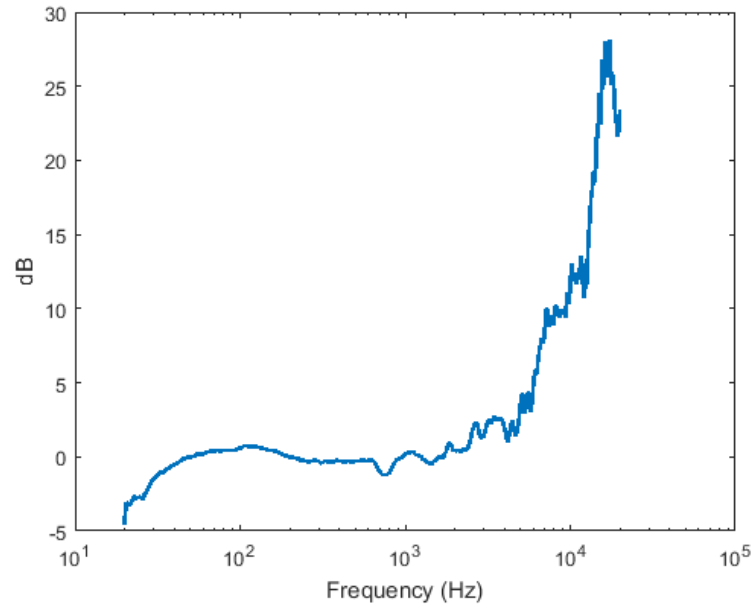


Figure 10.3: The measured frequency response of complete microphone system.

Sound Pressure Level	Distance (m)
70 <i>dBu</i> (Conversation)	11.94
94 <i>dBu</i> (Standard test level)	47.51
140 <i>dBu</i> (Aircraft, 50 <i>m</i> away)	671
194 <i>dBu</i> (Maximum sound pressure at 1 <i>atm</i>)	15 023

Table 10.4: The maximum distance from the microphone system to the sound source until the received signal level equals the noise level.

It should be noted that it is possible to interpret signals below the noise level since most desired signals appear on a limited frequency range which can be separated from the noise.

10.2 Evaluation

Gain

The gain is almost as expected. The small deviation is due to component inaccuracies but since the gain span is relatively large small deviations are accepted.

Frequency Response

The frequency response is not exactly as expected since it dips for low frequencies compared to how the shelf filter should behave (see figure 7.3). This is because of the DC-block that is placed after the main amplifier to make sure that the bias voltage does not vary with the gain. The frequency response dips because this DC-block has a cutoff frequency that is too high and therefore it attenuates the signal in the audible range slightly. The cutoff frequency can be lowered by using a higher resistor value but this would make it harder for the signal to acquire the correct bias value, thereby making the DC-block lose its purpose. To get optimal performance the optimal resistor value should be found so that the frequency curve is as unaffected as possible while the signal still obtains the correct bias voltage. Another possibility is to avoid the DC-block completely as discussed in section 5.8.1.

CMRR

A CMRR of more than 80 dB could be considered good enough for the electronics chain.

THD and IMD

The *THD* of the electronics is below 0.1 % which is far below what is considered audible. The *IMD* of -57 dB corresponds to 0.0014 % which should not be audible. The *THD* and *IMD* of the complete microphone system is probably limited by the distortion in the power amplifier and the speaker with which the measurements are made. Therefore these measurements can not be considered reliable. Although, the *THD* is guaranteed to be below the measured value which is below 1 % which is barely audible.

SNR and EIN

The SNR is, as expected, not degraded but improved by the rest of the electronics. This is probably because the system has become less sensitive to interference because of shorter transmission lines, line driver and metal shielding by the camera.

DC-Offset and DC-Level

The differential DC-offset (the difference in DC between the two differential signal components) of 13.05 mV corresponds to a loss in dynamic range by $\frac{13.05 \text{ mV}}{2.571 \text{ V}} = 0.5$ % assuming an average DC-level of the desired 1.5 V. The measured DC-level is 1.535 V for positive input and 1.545 V for negative which is slightly higher than desired. This is because these measurements were made without the codec connected. When the codec is connected an input resistance of 21 k Ω is introduced which will lower the DC-level slightly more.

The DC-offset still seems to vary slightly with the gain. To reduce this effect, as well as the DC-offset, the components will have to be matched better or trimmed using a potentiometer.

10.2.1 Possible Improvements

The main amplifier PCB layout is designed together with an experienced CAD engineer and created on a four-layer PCB for optimal performance. The following electronics are constructed on test PCB boards for easy modification possibilities. When the optimal schematic design is found these components should also be constructed with an optimal PCB layout to improve the total performance.

The conclusions regarding each part of the chain as well as for the complete microphone system is presented in this chapter.

11.0.1 Microphone

The chosen microphone, the Invensense ICS-40720, performs well. It has a remarkably high SNR for its price of \$2.80 (*1k* price). $SNR = 68.92\text{ dB}$ without any weighting. It has a fairly flat frequency response that does not fall off too much for low frequencies which makes it easy to compensate using filters and a peak for high frequencies which is suitable for listening to sounds far away or through some attenuating medium such as glass. It also has a high sensitivity which makes it easy to design analogue signal processing electronics that does not degrade the SNR notably.

In conclusion, the Invensense ICS-40720 has outstanding performance for surveillance applications. The only observed downside is the price of \$2.80 which is fairly high, which could be a problem if the budget for the audio solution is very low.

11.0.2 OP Amplifiers

The chosen OP amplifiers, the Texas Instruments OPA1662 and the Texas Instruments LME49724, performs as expected. Since the OP amplifiers generally has very good specifications even very small interferences will have a great impact on the measurements making it hard to determine if the specifications are accurate. Although, judging by the overall performance of the electronics the OP amplifiers seems to be well suited for the application.

The OPA1662 has a remarkably low price for its performance of \$0.75 (*1k* price) for a dual OP amplifier. In total, seven OPA1662 are used in the electronics for a total cost of \$3 which covers all OP amplifiers in the electronics except the line driver. To further reduce the price it is possible to replace one of the OPA1662 with the OPA1664 which is the same OP amplifier but in a quad configuration costing only \$1.11 (*1k* price) reducing the price to \$2.61. As stated in section 5.4.2 the requirements on noise performance of the filters are much lower than for the main amplifier which makes it possible to use a cheaper OP amplifier with a much worse noise performance. Although, buying the same OP amplifier for the whole circuit would make it easier to negotiate a bulk deal.

The LME49724, which is used as the line driver, is much more expensive costing \$1.42. This component converts a single ended signal to a differential making it a special purpose OP amplifier and is therefore more expensive and cannot be replaced as easily as an ordinary OP amplifier. For low budget audio solutions, or in system where the microphone system is well shielded or exposed to little interference, it would be possible to remove this component since it is not necessarily crucial to the performance. If the line driver is removed the signal will remain single ended, making the anti aliasing filter single ended and thereby reducing the number of OPA1662 OP amplifiers to six.

In conclusion, the analogue signal processing circuits implemented in this project performs as desired for a total cost of \$4.11 which can easily be reduced to \$3.72 without notable impact. In an ideal situation, such as the electronics being placed closed to the codec and being well shielded, it would be possible to remove the line driver, reduce the number of op amplifiers to six and thereby reducing the total cost of the analogue signal processing circuits to \$1.82. In case of mass production a bulk deal would most certainly be negotiated reducing the price further. Note that these prices only include the price of the OP amplifier since they are most certainly the most expensive components in the circuits.

11.0.3 Acoustics

It has become apparent that the acoustic impact on the frequency response is complex and unpredictable. Even though the microphone is placed as close to the chassis as possible, which should, theoretically, not impact in audible frequency range due to the acoustic conductor being too short, still gives quit a substantial and unexpected impact on the frequency response.

In conclusion, the impact of the acoustics is hard to predict. To achieve the desired frequency response a unique filter circuit would probably be required for each chassis or microphone placement, and to be certain how to design the filters the acoustic impact should be measured since it will probably differ from the expected. An alternative is to use advanced simulation tools for acoustics, but it will probably be unnecessarily expensive.

11.0.4 Analogue Signal Processing Circuits

With the analogue signal processing circuits the desired frequency response is achieved, the signal is made more resilient to interference, a variable gain ranging from $6dB$ to $45 dB$ is implemented, a signal level adaption to the codec which yields a dynamic range of close to 100 % of what 16 bits allows, all while keeping the THD below 0.1 % and with a noise performance that barely affects the SNR of the complete microphone system.

There are many ways to further improve the performance of each stage, some of which are discussed in their respective chapters. The filters, line driver and signal attenuator are constructed on test PCB:s. When the circuits are optimized they should be implemented on a custom made PCB to further improve the performance. The perhaps most important thing to consider when constructing the custom PCB is to keep the transmission line for the bias voltage as short and

well shielded as possible. In the measurements it becomes apparent that the bias voltage is much noisier than it should be, which is especially bad since the noise of the bias voltage is placed right on top of the signal before it is amplified. By making the anti aliasing filter fully differential the bias voltage will not be needed after the line driver which would make it possible to keep the transmission line for the bias voltage very short. If the bias voltage for some reason is needed after the line driver it is suggested that a second bias voltage buffer is used to supply the circuits that are less noise sensitive.

In conclusion, the analogue signal processing circuits does what they are supposed to without degrading the signal significantly. Although, the optimal performance of the design will not be achieved until implemented on a custom designed PCB.

11.0.5 Codec

The codec used is the Analog Devices ADAU1361 which is used with a sampling rate of 48 kHz and the resolution of 16-bit, which should give a good enough dynamic range for most applications, especially since the signal level adaption circuits makes it possible to take advantage of the whole range. It should be noted that the input impedance is relatively low and will probably have to be taken into account to achieve an accurate signal level adaption. The signal is measured before and after the codec and there is no significant difference. The price of the codec is \$2.74.

In conclusion, the codec seems good enough since it does not seem to cause any unwanted effects. The main reason this codec is used is because many of Axis' products already uses it which makes the microphone system design in this project compatible with other Axis products.

11.1 Complete Microphone System

The goal of this projects is to create an analogue microphone system for best possible capture of incoming acoustic signals. The limiting component is identified as the microphone. Therefrom the rest of the system was designed to get a desired frequency response, interference resilience, amplification and signal level to codec without degrading the signal notably.

The frequency response in the different stages of the implementation can be seen in figure 11.1.

The low frequencies are gained to get a flat response for low frequencies. The dip of about 3 dB for the lowest frequencies is because of the DC-block placed after the main amplifier. This dip should be easy to get rid of if required, as discussed in section 5.8.1, but since this dip is probably not audible it is considered good enough. Because of the acoustic effects the high frequencies are substantially gained. This can be considered a positive feature for surveillance applications since the camera is often placed far away from the sound source and high frequencies are more easily attenuated than low frequencies. This could be desired when listening to sound coming from far away or when listening through a window or through the transparent dome in a dome camera. It could also be desired if the camera

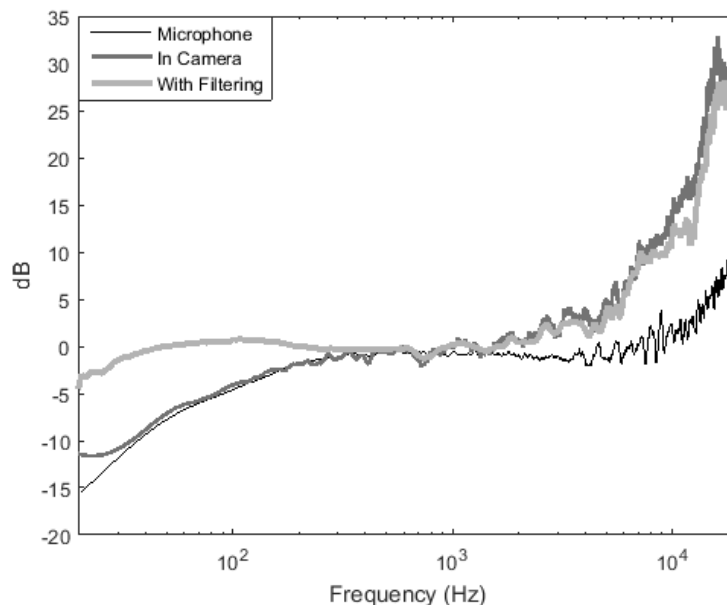


Figure 11.1: The measured frequency response of the microphone as a standalone component, microphone inside the camera and after filtering.

is placed in an environment that act as an acoustic resonator for low frequencies which could make the high frequencies "drown" in the low frequencies.

The downsides of having high gain for high frequencies is that the SNR is degraded, since the noise is also gained for high frequencies, and the dynamic range is reduced for sounds containing frequencies in the top of the audible frequency range. How much the high frequencies should be gained for optimal performance is very hard to determine because of the great diversity of camera applications and placements. Although, if the camera was to be placed in an environment where a flat frequency response is desired it is easy to attenuate the high frequencies digitally which should restore the SNR, leaving only the loss of dynamic range as the problem.

If one is sure that, for a certain implementation, the frequency response is desired to be flat, it should be possible to attenuate the high frequencies and get a fairly flat frequency response just by changing a few components in the anti aliasing filter, pushing down the cutoff frequency.

The amplification is as desired according to the measurements. The signal level seems to be well adjusted to the codec and the bias voltage and DC-offset seems to be within a reasonable error margin not to affect the dynamic range for any gains.

The SNR is improved when the complete microphone system is connected

together indicating an improved noise resilience.

In conclusion, the microphone system performs as desired within a reasonable error margin. However, some measurements, such as the THD and IMD, could not be considered reliable because of inferior measurement equipment.

11.1.1 Cost

The total price of the microphone system, not including resistors, capacitors, ferrites or PCB, is \$9.96 which can be reduced to \$9.57 using a quad OP in the main amplifier, and in a best case scenario reduced to \$7.4 if the line driver can be removed and the anti aliasing filter single ended. By negotiating bulk deals this price can probably be even further reduced.

11.2 Possible Improvements

Possible improvements for each stage is discussed in their respective chapters. Creating a custom PCB with a well designed layout will probably improve the overall performance further, especially in terms of noise resilience. Further improvements of the overall design may depend on the application such as desired frequency response and what casing is to be used for the camera. It could also be a question of priorities, such as low budget, which could mean choosing a less expensive microphone for example.

11.2.1 Measurements

It should be noted that all measurements in this project has been made under the influence of electromagnetic interference and acoustic noise. To get reasonable values on the noise performance the noise measurements were made from 200 Hz to 20 kHz which should not make a notable difference for the total noise performance. All measurements were made in the same laboratory to make sure to get consistent results.

To measure the THD and the IMD of the complete microphone system a special low distortion speaker element and a low distortion power amplifier would be required. The measurements made could therefore only be considered the roof of the THD and IMD and not the actual values. The positive aspect of having a noisy measurement environment is that it is possible to see the noise resilience of the circuits.

An ideal measurement environment should be completely noise free and equipped with ultra low distortion speaker elements and power amplifier. Since this was not available at all time it was considered better to make all measurements in the same environment to see the relative effect of each stage.

Final Remarks

The quest for the perfect audio can go on for as long as there is time. There will always be ways to improve or adjust for the listener. Especially for surveillance there are seemingly countless possible situations, all with a slightly different audio requirement. Furthermore, there can be many different opinions on what the optimal audio is. It could be the solution that sounds best to the human ear, the solution that makes it the easiest to interpret important information or the solution that makes it the easiest for a computer to interpret the sound.

This thesis is a conceptual demonstration of the possibility to achieve a desired audio performance without spending an unnecessary amount of money by highlighting the critical components which limits the performance and should be given extra attention, and where in the microphone system it is possible save money.

References

- [1] Ian Colquhoun Alan Lofft and Tom Cumberland. How much distortion can we hear with music? <http://www.axiomaudio.com/distortion> [2016-02-29].
- [2] Analog Devices. Using comparators to detect accessories in portable audio applications, 2008. <http://www.embedded.com/print/4015894> [2016-03-01].
- [3] Analog Devices. Calculating Spectral Noise Density to RMS Noise, 2010. <https://www.youtube.com/watch?v=ywChrIRIXWQ> [2016-02-29].
- [4] David J. Griffith. Introduction to electrodynamics, fourth edition, 2013.
- [5] Analog Devices. Mt-047: Op amp noise, 2009. <http://www.analog.com/media/en/training-seminars/tutorials/MT-047.pdf> [2016-02-29].
- [6] Analog Devices. Calculating rms noise to peak-to-peak noise, 2010. <https://www.youtube.com/watch?v=-KcODSYXIZA> [2016-02-29].
- [7] Douglas Brooks. Differential signals: Rules to live by, 2001. https://www.ieee.li/pdf/essay/differential_signals.pdf [2016-05-25].
- [8] Fair-Rite Products Corp. How to Choose Ferrite Components for EMI Suppression. <http://www.newark.com/pdfs/techarticles/FairRite/HCFCEMIS.pdf> [2016-03-21].
- [9] G. G. Stokes. On the theories of the internal friction in fluids in motion, and of the equilibrium and motion of elastic solids, 1845.
- [10] InvenSense. Microphone specifications explained, 2013. <http://www.invensense.com/wp-content/uploads/2015/02/MICROPHONE-SPECIFICATIONS-EXPLAINED1.pdf> [2016-02-29].
- [11] Izhar, Farid Ullah Khan. An improved design of helmholtz resonator for acoustic energy harvesting devices, 2016.
- [12] Jerad Lewis. Analog and Digital MEMS Microphone Design Considerations, 2013. <http://www.analog.com/media/en/technical-documentation/technical-articles/Analog-and-Digital-MEMS-Microphone-Design-Considerations-MS-2472.pdf> [2016-05-24].

-
- [13] Jim Browne. Cables, transmission lines, and shielding for audio and video systems, 2002. <http://audiosystemsgroup.com/TransLines.pdf> [2016-05-13].
 - [14] Jim Karki. Understanding operational amplifier specifications, 1998. <http://www.ti.com/lit/an/sloa011/sloa011.pdf> [2016-02-26].
 - [15] Ken Chan. Design of Differential Filters for High-Speed Signal Chains. <http://www.ti.com/lit/an/slwa053b/slwa053b.pdf> [2016-03-22].
 - [16] Paul H. Young. Electronic communication techniques, 2012.
 - [17] Ron Mancini. Op amps for everyone, 2002. http://web.mit.edu/6.101/www/reference/op_amps_everyone.pdf [2016-06-01].
 - [18] Texas Instruments. Noise Analysis in Operational Amplifier Circuits, 2007. <http://www.ti.com/lit/an/slva043b/slva043b.pdf> [2016-03-17].
 - [19] Theodore P. Pavlic. Ece 209: Circuits and electronics laboratory, 2011. http://www.tedpavlic.com/teaching/osu/ece209/support/circuits_sys_review.pdf [2016-05-25].
 - [20] Thomas H. Lee. The design of cmos radio-frequency integrated circuits, 2004.