Avoiding Frequency Cancellation Using Multiple Microphones in a Surveillance Camera

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March 23, 2021

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Printed in Sweden
Tryckeriet i E-huset, Lund

Abstract

In surveillance it is important that the recordings are accurate and in as high quality as possible. The surveillance surroundings and equipment can alter and change the audio and needs to be compensated for.

This master thesis project investigates whether using multiple microphones could improve the audio quality. The project was limited to only use analog electronics and constricted itself to the audible frequency range of 20 Hz to 20 kHz.

The projects started with a theoretical study of different concepts of multiple microphones, then one concept was implemented and evaluated. The chosen concept was how to avoid destructive interference using multiple microphones achieving a frequency response without cancelled frequencies.

After the prestudy, the concept was realised by filtering the signals from two microphones that were exposed to destructive interference in different frequency ranges. The filter was based on the Linkwitz-Riley crossover filter.

Simulations were used to evaluate the filter, witch was then realised in a prototype that was tested in an audio lab.

The test result showed that the concept functioned successfully. The filter produced a signal without the frequency losses that each of the microphones experienced.

Popular Science Summary

In the world of surveillance the main focus has for a long time been on video recording but now the interest of supplementing it with recorded sound is growing. As with video recording there are a lot of challenges with recording good quality sound in an efficient way to a low cost. The shape and placement of the camera can even be the causes that are degrading the sound quality of the recordings.

If the recording microphones are placed on cameras that are mounted on reflective surfaces, there is a high risk that the recorded sound will experience losses in the frequency band of 20 to 20000 Hz, that is the frequency range that a human can hear. In surveillance, the audible frequency range is of great interest and it is therefore important to keep it as unaffected by the recording equipment as possible.

Because it not always possible to change the placement of the microphone on the camera, another way of solving the problem is needed. This project has evaluated a solution to the frequency loss problem by using multiple microphones. If the microphones are placed with a small distance between them on the camera the frequency losses they experiences will be different. The next step of the project was then to built a filter that filters out the "bad" parts of the microphones frequency band and keeps the "good" parts and then add the selected pieces together, and thus end up with an audio recording with no major frequency losses.

The filter chosen was a Linkwitz-Riley crossover filter, commonly used in professional audio, to split an audio signal into two different frequency bands, for example a subwoofer and a treble. The filter is based on a lowpass and a highpass filter that has the same cut-off frequency. That is, at what frequency the filters begins to attenuate. To make the filter suite the project the filter was implemented "backwards", in other words the Linkwitz-Riley crossover filter was fed the two audio signals from the microphone and sent out one.

Acknowledgements

During the process of making this master thesis project I have had the opportunity to receive advice and guidance from both expected and unexpected directions. The people that have contributed to the mere existence and completion of this project, are too many to be all mentioned by name and thanked accordingly. To all of you I would like to say a big thank you! I hope you know who you are, you all deserve it.

However, there are some people I would like to mention.

First of all I would like to thank my supervisor at Axis, Johan Sunnanväder, for sharing his knowledge and enthusiasm and for being the best supervisor I could ever have wished for.

I would also like to thank my manager Anders Svensson and the department Product Audio at Axis for the great reception and never ending support.

Fredrik Rusek and Pietro Andreani, my supervisor and examiner at LTH, thank you for your support and all the help guiding me through this sometimes confusing examination process.

I would like to send a warm thank you to my friends, especially Freddie and Johannes who never run out of good advice.

And to my family, Simon and little Lovisa who decided to enter this world in the middle of the project. I could never have done this without you.

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Chapter 1
Introduction

This master thesis project has been realized in collaboration with Axis Communications AB, a company in the market of security and video surveillance.

The project started with the ambition to find and evaluate several ideas on how to improve the audio in a surveillance camera using multiple microphones. As the work progressed, the idea of evaluating several ideas was discarded, and one of the ideas was chosen to be the main focus of the project.

The subject chosen for the project was to examine how to achieve a frequency response of the recorded audio without frequency cancellation in a situation where the microphones of the camera experiences destructive interference. The goal of the project was to develop an analog filter that met the requirements. The filter was simulated and then a prototype was built and tested.

1.1 Motivation

Several of Axis cameras record audio in addition to video. At the moment most of the cameras only use one microphone to record the surrounding sound, which can cause problems because of the microphone's limitations or the surroundings impact on the recording.

Axis wanted to examine how using multiple microphones could improve the audio quality. Axis has previously studied the benefits of beam forming and wanted to continue the study of the use of multiple microphones with other concepts.

1.2 Background

In surveillance there are some aspects of audio recording that are more interesting than others.

Lossless recordings are important. To not lose interpretability it is desirable that the sound is not altered by the system recording it. To have an unaltered recording is also important when the sound is treated by algorithms that are designed to recognise specific sounds characteristics.

It is not always known in advance what kind of sound will be recorded. To have a system that can handle sound with a wide dynamic range and having a frequency range where all frequencies in the audible area are equally amplified, are also desired.

The placement of the camera is sometimes a source of disturbance. When the camera is placed on a reflective surface, the reflection of the sound can cause acoustic phenomena that can affect the audio quality negatively.

1.3 Limitations

This project will try to improve the audio without moving the microphones to an acoustically more favourable position, as a solution to the problem with cancelled frequencies. The method to solve the problem will instead be electronically analog.

The frequency range that will be handled in the project are the frequencies in the audible frequency range, 20 Hz to 20,000 Hz.

The focus of the project is on the functionality of the filter. Noise and other performance optimizations will be overlooked.

Chapter 2

Theory

This chapter gives a short explanation of the theory of the most important concepts in the project.

2.1 Human Hearing

The human ear can hear sound in the frequency range between 20 and 20,000 Hz, but the range decreases with age. Sound frequencies in that range are not perceived in the same way, the ear is more sensitive to frequencies between 500 to 5,000 Hz, which coincides with the frequencies speech is composed of [1].

2.2 Acoustics

Sound can be explained by physics and described by laws, but is also personally perceived. Sections 2.3 and 2.4 below present a short introduction to the physical aspect of sound.

2.3 Physics of Sound

Sound propagates as waves, but contrary to light, sound needs a medium to propagate in. The speed of sound is dependent on the medium, among other things. In air the speed is 344 m/s, $\mathrm{v_{air}}$.

Sound is made of periodic waves and can be described by its frequency, f, or equivalently by its wavelength λ. The relationship of the frequency and wavelength of the sound wave is dependent on the velocity v, see equation 2.1 [6].

$$f = \frac{v}{\lambda} \tag{2.1}$$

2.4 Interference

When sound waves mix, the amplitude of the resulting wave is either affected constructively or destructively, depending of the phase of the mixing waves. This is called interference.

Destructive interference occurs when some of the sound waves from a single sound source reflects and mix with sound waves that have different path length. What specific differences in path length that causes destructive interference is determined by the wavelength of the sound, see equation 2.2.

$$\Delta L = \left(n + \frac{1}{2}\right)\lambda \tag{2.2}$$

where n is an integer.

Combining the equations 2.1 and 2.2 gives the different frequencies when destructive interference occurs. $n = 0$ gives the lowest frequency, n = 1 the second lowest and so on for a given difference in travel distance [7].

Figure 2.1 shows the situation when the sound source is in the far-field of the two microphones. The sound waves can be considered to be parallel to each other, incident from directly above and is hitting one of the microphones on a dome camera. One sound wave hits *Mic 2* directly without any reflections, having the path length l. The second sound wave hits *Mic 2* after first reflecting in the surface beside the camera and has a path length of $l_1 + l_2$. That gives a $\Delta L = (l_1 + l_2) - l$. The smallest cancelled frequency due to destructive interference in *Mic 2* can be derived from equations 2.1 and 2.2 as mentioned above. See equation 2.3.

$$f = \frac{\mathrm{v_{air}}}{2\Delta L} \tag{2.3}$$

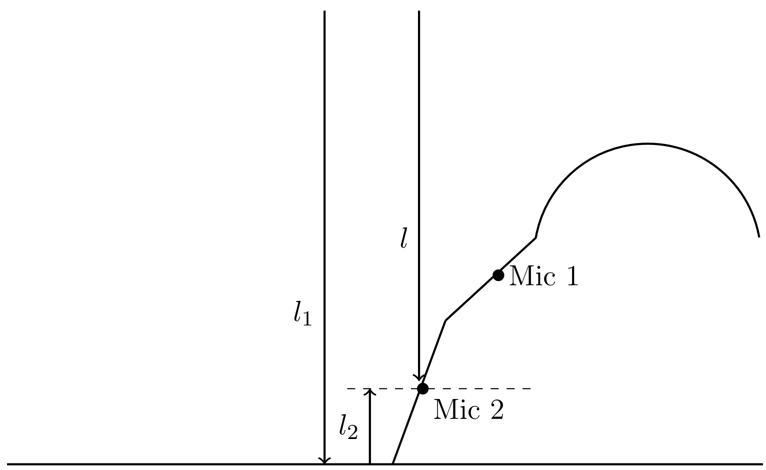


Figure 2.1: Sound waves hitting the camera and the reflective surface from above.

2.5 Microphones

A microphone is an electromechanical device that converts air pressure into an electrical signal. A microphone consists of a diaphragm that vibrates when it is hit by sound waves, and the electronics that convert the vibrations into an electrical signal [2].

The microphone used in this project is an electret microphone. An electret microphone uses the principle of a capacitor to convert the diaphragm movement in the microphone to an electronic signal. One of the capacitor plates is permanently charged, hence the name electret microphone. This type of microphone often contains a preamplifier and needs external power [4].

The electronic configuration varies between different types of microphones and gives them different characteristics. The characteristics most interesting for this project are explained in sections 2.6 to 2.8 below.

2.6 Dynamic Range

The dynamic range of a microphone is the difference between the loudest and the weakest sound a microphone can detect without distorting the signal. The weakest sound is limited by the microphones noise floor and is often expressed as Equivalent Input Noise, EIN. The loudest is usually referred to as the Acoustic Overload Point, AOP [3].

2.7 Directionality

The directionality of a microphone describes what directions of the incoming sound the microphone is most sensitive to, sensitivity is described in section 2.4. Two examples of microphones with different directionalities are the omnidirectional and the bi-directional microphone. The omnidirectional microphone is equally sensitive to sounds from all directions whereas a bi-directional microphone is sensitive to sounds only from the front and the back of the membrane [3].

2.8 Sensitivity

Sensitivity is a measurement of the electrical output as a function of the input sound pressure. It is often measured with a 1 kHz sine wave at a 94 dB sound pressure level (SPL), or 1 Pascal (Pa) pressure. That gives the sensitivity in units of mV/Pa.

To express sensitivity in units of dBV, the equation 2.4 is used.

$$Sensitivity_{\mathrm{dBV}} = 20 \cdot \log_{10}\left(\frac{Sensitivity_{\mathrm{mV/Pa}}}{Output_{\mathrm{REF}}}\right) \tag{2.4}$$

$Output_{\mathrm{REF}}$ is 1 V/Pa [3].

2.9 Decibel

Decibel is used to describe the ratio between the amplitude of two signals in a logarithmic scale. The definition of decibel for root power values is expressed in equation 2.5 below.

$$dB = 20 \cdot \log_{10}\left(\frac{input\ value}{reference\ value}\right) \tag{2.5}$$

Decibel can also be expressed as an absolute measure, when the first signal is compared to a known reference signal. Common standard reference signals are 1 Vrms with the unit dBV, and dB SPL that uses a reference signal with the rms pressure of 20μ Pa. dB SPL is often used in acoustics.

The equation for the ratio between two power values is expressed in equation 2.6.

$$dB = 10 \cdot \log_{10}\left(\frac{input\ value}{reference\ value}\right) \tag{2.6}$$

Due to the logarithmic behaviour, a doubling of the amplitude of a signal expressed in volts will be a gain of 6 dB, whereas if it is the power that is expressed the gain will be 3 dB [5].

2.10 Frequency Response

A microphone or an electronic filter alters the incoming signal in the frequency domain. To show how a system, e.g. a microphone, affects the incoming signal, a frequency response is used. Thus, the frequency response graph is a measure of how the system responds to different frequencies. Some systems amplify frequencies in the high region more than in the low or vice versa.

The frequency response is often divided into two specified responses. The amplitude response gives the relation between signal amplitude and frequency, but systems also have a phase diagram. As the name implies the phase response shows how the system affects the phase of the incoming signal [3].

2.11 Signal-To-Noise Ratio (SNR)

The signal-to-noise ratio is the measurement of the ratio between the power of the noise level and the power of a reference signal.

For microphones, the reference signal is usually 1 kHz, 94 dB SPL and SNR is often measured over a bandwidth of 20 kHz. SNR can also be an expression of the ratio of volts instead of power. Equation 2.7 shows how to calculate the SNR of both types of signals [3].

$$SNR = 10 \cdot \log_{10}\left(\frac{P_{\text{signal}}}{P_{\text{noise}}}\right) = 20 \cdot \log_{10}\left(\frac{V_{\text{signal}}}{V_{\text{noise}}}\right) \tag{2.7}$$

2.12 Analog Filter

An analog filter processes an analog signal in the frequency domain. Ideally the filter either stops or lets the specific frequencies pass, but due to physical limitations the unwanted signals are instead attenuated gradually. The frequency when the signal is attenuated with 3 dB is called cut-off frequency, F_c. The frequency range that the filter is designed to stop is called the stop band and the frequency range that is passed through is called pass band.

To describe how the filter processes the signal, a transfer function is used to visualize the relationship between the input and output signal. An example is the transfer function of the RCL circuit, see figure 2.2. The transfer function is expressed in equation 2.8.

$$H(s) = \frac{V_o}{V_{in}} = \frac{RCs}{LCs^2 + RCs + 1} \tag{2.8}$$

Here R is the resistance, C the capacitance, L the inductance. s is the complex frequency defined as s = $e^{i2\pi f}$ where f is the frequency of the incoming signal [8].

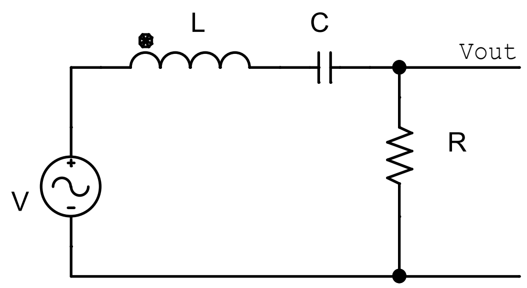


Figure 2.2: The circuit of an RCL filter.

Different filter characteristics will be discussed in section 2.14 to 3.3.

2.13 Buffer

To prevent two circuits from affecting each other they are separated with a buffer. A buffer is a non inverting amplifier with unity gain, also called a follower. It has high input impedance and low output impedance, thus preventing a circuit from being loaded by subsequent circuits [5].

2.14 Active/Passive Filter

Analog filters can be active or passive depending on whether active or passive components are used. Passive filters consist of resistors, capacitors and inductors. When working with lower frequencies, less than 10 MHz, the use of active components, such as operational amplifiers, is then preferred, as the inductive element of the filter becomes large and space consuming [8].

2.15 Q Factor

The quality factor, or Q factor, is a dimensionless parameter that describes the shape of the attenuation slope of the frequency response. The inverse of Q is also known as the damping ratio and is shown in equation 2.9 below.

$$\alpha = \frac{1}{Q} \tag{2.9}$$

The transfer function of the RCL circuit can be expressed with the Q factor.

$$H(s) = \frac{H_0}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2} \tag{2.10}$$

H_0 is the pass-band gain and $\omega_0 = 2\pi F_c$ is the angular frequency [9]. As mentioned in section 2.12, s is the complex frequency defined as s $= e^{i2\pi f}$.

2.16 Filter Order

The order of the filter affects the frequency response in many ways.

How fast the signal is attenuated is dependent on the order of the filter. A higher order filter gives the frequency response a steeper slope. The order of a filter can be increased by cascading lower order filters.

The relation between the order and the attenuation is described in the equation 2.11 below.

$$Attenuation = n \cdot 20 \text{ dB/decade} \tag{2.11}$$

where n is the order of the filter. A decade is an increase or decrease of the frequencies by a factor of 10.

The steepness of the slope can also be expressed in dB/octave. An octave is a doubling or halving of the frequencies. 20 dB/decade equals 6 dB/octave and the attenuation can therefore be expressed as below in equation 2.12 [8].

$$Attenuation = n \cdot 6 \text{ dB/octave} \tag{2.12}$$

The order of the filter affects how much the phase is shifted. An active first order filter has a phase shift of 90 degrees at all frequencies and a second order filter a phase shift of 180 degrees and so on. Every order adds a shift of 90 degrees.

The phase shift, ϕ, of a lowpass filter is expressed in equation 2.13. The phase response of a highpass filter is shifted 90 degrees from that of the lowpass filter as can be seen in equation 2.14.

$$\phi(\omega) = \tan^{-1}\left(\frac{\omega}{\omega_0}\right) \tag{2.13}$$

$$\phi(\omega) = \frac{\pi}{2} - \tan^{-1}\left(\frac{\omega}{\omega_0}\right) \tag{2.14}$$

ω is the angular frequency in radians per seconds and ω_0 is the center frequency of the phase response, also in radians per seconds. The center frequency marks the frequency when the phase has shifted 50 % of a whole shift. In the case of the high pass filter the center frequency is 45 degrees. F_{c} is usually the same frequency as the center frequency [15].

2.17 Sallen-Key

Second order filters can be realised in different ways. One of the most common designs is Sallen-Key and that is the one used in this project.

The Sallen-Key filter is an active filter using an operational amplifier as a buffer, in other words it doesn't amplify the signal. An other name for the Sallen-Key filter is *voltage controlled voltage source* [15].

2.18 Filter Configuration

Designing filters with complex poles makes it possible to achieve filters with specific characteristics. Three popular filter designs are Butterworth, Chebyshev and Bessel filters. Each of the filter designs has its benefits and drawbacks. The designs are specified by their Q value.

Butterworth configuration in filters is used for maximum flatness of the passband in the frequency response. On the other hand, it has a nonlinear phase response. Chebyshev filters have the steepest rolloff but has ripple in the passband. Bessel filters have the most linear phase response [5].

2.19 Crossover Filter

A crossover filter splits the signal into two or more signals with different frequency ranges. A typical application is in audio when it is necessary to divide the signal between two or more loudspeakers with different frequency characteristics. The crossover filter splits the signal in high and low frequencies using highpass and lowpass filters. The frequency where the filters intersect is called crossover frequency. If the filters have the same cut-off frequency, F_{c}, the crossover-frequency will be equal to F_{c}.

A frequency response of a crossover filter using Butterworth configuration can be seen in figure 2.3, as the solid line. The dashed lines shows the frequency response of the highpass and lowpass filters, each having a $F_{\mathrm{c}} = 1$ kHz, resulting in a crossover frequency of 1 kHz.

The Butterworth configuration is used to get the amplification of the frequencies around the crossover frequency as flat as possible. The frequency response of the crossover filter is however not flat around the crossover frequency. The problem with the extra amplification at the intersection point is due to the amount of gain drop of the lowpass and highpass filters at that point, and is explained in the next section, see section 3.3.

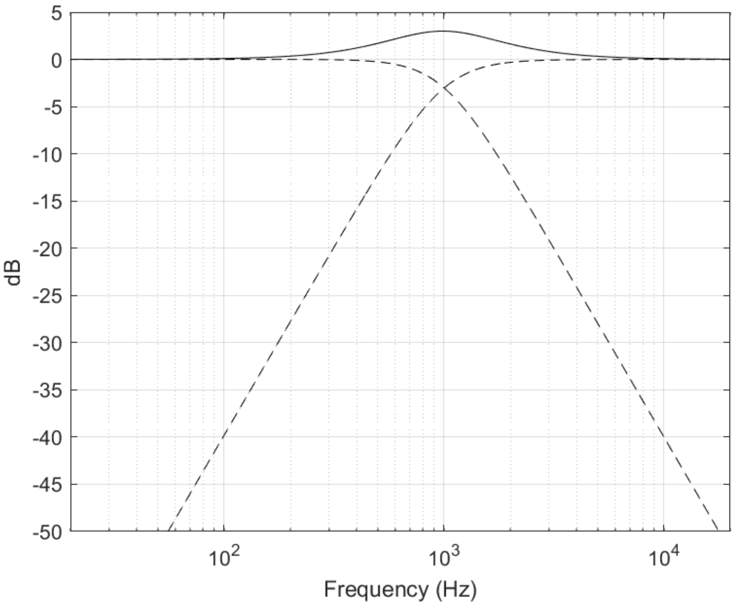


Figure 2.3: The solid line shows the 2nd order crossover filter using Butterworth filters. The dashed lines shows the HP and LP filters of the crossover filter.

Chapter 3

The Project, Teory and Simulations

The project was divided into the following different phases: prestudy, execution, and evaluation of the results. This method was chosen because it is well known at Axis and is used by them in their projects.

3.1 Prestudy

The prestudy started with an evaluation of possible ideas for multiple microphone concepts. When four of the most interesting concepts were decided on, a literature study of the four concepts and a general theoretical study of audio and analog electronics took place.

During the prestudy the decision to only continue with one of the concepts was made, because of time constraints and to be able to do a more thorough investigation of the concept. The concept that was chosen was called Flat Frequency Response, and is explained in section 3.1.1 to 3.1.2.

At the end of the prestudy an execution plan was set up for the continuing work.

3.1.1 Theory Behind the Flat Frequency Response Concept

As mentioned in section 2.2, room acoustics can change and degrade the quality of the sound. To avoid losing some of the frequencies in the incoming sound due to destructive interference, the idea is to record the sound with several microphones that are exposed to the destructive interference differently. The microphones are placed on the camera with a small distance between them, so that different frequencies are cancelled. The microphones can therefore together cover the whole frequency band from 20 Hz to 20 kHz.

The microphones should be placed on the camera in such a way that the cancelled frequencies end up either in the area of low frequencies or high frequencies in the frequency response of the microphone. The parts of the frequency response that are affected by the destructive interference are then cut off by filters and the two remaining microphone signals are mixed. The resulting signal will be without any frequency losses.

As was mentioned in section 2.4, destructive interference gives rise to a number of cancelled frequencies. In this project the focus was on the first cancelled frequency, i.e. when $n = 0$ in equation 2.2. It is assumed that the following cancelled frequencies has less impact on the sound quality than the first.

3.1.2 Filter

During the prestudy it was decided to limit the number of microphones to two because the complexity increases with the number of microphones.

It was decided that the filter to be used to filter out the unwanted frequency bands and mix the two signals should be a crossover filter used backwards. Instead of splitting a signal into two frequency bands, the crossover filter to be implemented will have two inputs and one output. Despite the backwards implementation of a crossover, the filter implemented in this project will be referred to as a crossover filter.

Because of its favorable characteristics the crossover filter will be implemented as a Linkwitz-Riley crossover filter to get as flat a frequency response from the mixed microphone signals as possible. The Linkwitz-Riley filter is explained in section 3.3. Though the Linkwitz-Riley crossover filter is used for splitting signals between loudspeakers, the idea is that the theory of the filter can be used for our purpose as well.

The fourth order filter was chosen because of its 360 degrees phase shift and the 24 dB/oct slope. When the phase shift is 360 degrees the signal is not inverted, and there is no need for the microphone to be inverted. A steeper slope in the amplitude response is preferred because it means that the stopband has less influence.

3.1.3 Components

The microphone used in the project was the high sensitivity omnidirectional electret condenser microphone Primo EM273 [12]. High sensitivity is favourable in surveillance because the sound source is usually far away and therefore the incoming audio is weak, and more sensitive to noise.

The operational amplifier chosen for the project is OPA377 [13].

Axis uses the E-12 resistor and capacitor series, so the calculated filter parameters were adjusted to fit the E-12 values. If the parameters would prove to have been to coarsely chosen, more suitable components would be ordered and used instead.

3.2 Execution and Evaluation

This section will describe the execution of the project and the evaluation of the results thereof. The execution consisted of the following parts:

- Calculate the circuit parameters
- Simulate the circuit
- Build prototype
- Perform tests in audio lab

The simulations of the circuits and the built of the prototype were done in parallel and to make the error handling easier it was decided that the simulation and building of the crossover filter was to be made step by step. First the second order highpass and lowpass filters were going to be evaluated. Then the fourth order highpass and lowpass filters. As the last step the supportive circuits and a buffer to mix the two filtered signals, were added.

3.3 Linkwitz-Riley Crossover Filter

Even though they maximize the flatness of the passband, ordinary Butterworth filters with a gain drop of 3 dB at F_c are not enough to avoid the uneven amplification of the frequencies around the crossover frequency when used in crossover filters, as can be seen in figure 2.3. The parameters of the Butterworth filter need to be tweaked to perform well in crossover filters.

A filter that solves the problem of getting extra gain in the crossover region in the frequency response is the Linkwitz-Riley crossover filter. The filter is named after its inventors, Siegfried Linkwitz and Russ Riley.

The fourth order Linkwitz-Riley (LR) crossover filter, see figure 3.1, is a crossover filter that uses the Butterworth configuration with its component parameters set to a specific value that makes the frequency response flat. This configuration is now the standard for professional audio active crossover filters. The frequency response of the fourth order Linkwitz-Riley crossover filter can be seen in figure 3.2.

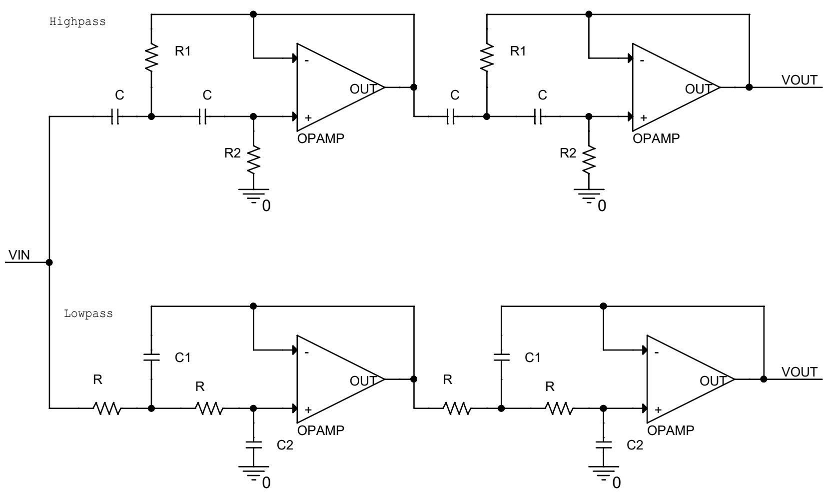


Figure 3.1: Fourth order Linkwitz-Riley crossover filter.

A crossover filter made of ordinary Butterworth filters has a drop of gain at the crossover frequency of -3 dB. When the two signals from the HP and LP filter are added it results in an increase in amplitude of 6 dB, as can be seen in equation 2.5 in section 2.9. The Linkwitz-Riley crossover solution has a drop of gain of -6

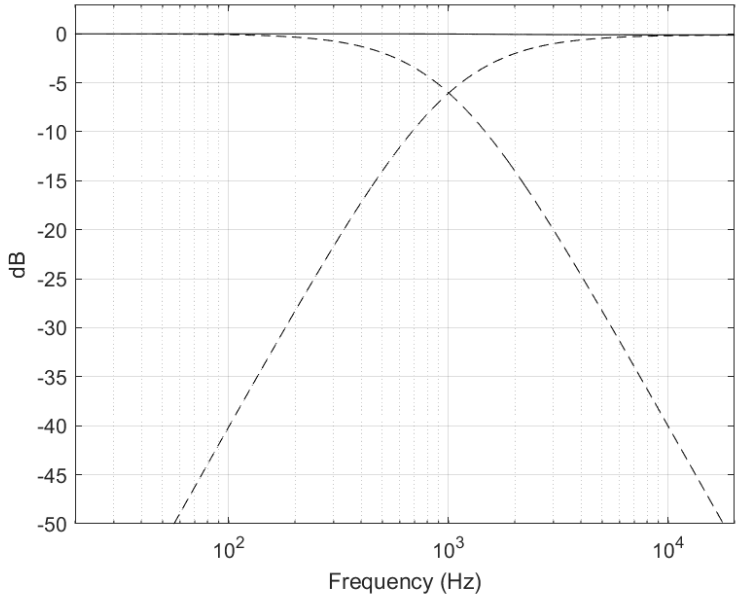


Figure 3.2: The solid line shows the crossover filter using Linkwitz-Riley filter.The dashed lines shows the HP and LP filters.

dB at the crossover frequency, making the gain of the crossover amplitude 0 dB, as can be seen in figure 3.2.

The frequency response of the fourth order Linkwitz-Riley filter has a steep 24 dB/octave slope and the output of the filter is in phase with the input. The fourth order filter is implemented by cascading two second order filters.

To calculate the parameters for the two second order lowpass and highpass filters the equations below are used. The parameters are set by ω_0, the Q value in table 3.1 and predefined values of C_1 and R_1. ω_0, expressed in equation 3.2, are set by the cut-off frequency, F_c, that are the same for the HP and LP filters.

As can be seen the equation are derived from the transfer function of the filters.

L(s), see equation 3.1, shows the transfer function of the second order lowpass filter. s is the complex frequency of the incoming signal.

$$L(s) = \frac{\omega_0^2}{s^2 + \frac{\omega_0}{Q_0}s + \omega_0^2} \tag{3.1}$$

F_c is expressed as the angular frequency ω_0, equation 3.2.

$$\omega_0 = 2\pi F_c \tag{3.2}$$

The relationship between the component value of the capacitance and resistance in the lowpass filter, ω_0 and Q_0 is shown in equation 3.3 to 3.5. C, C_1, C_2,

R, R_1 and R_2 can be seen in figure 3.1.

$$\omega_0 = \frac{1}{R\sqrt{C_1C_2}} \tag{3.3}$$

$$Q_0 = \frac{1}{2}\sqrt{\frac{C_1}{C_2}} \tag{3.4}$$

$$R = \frac{1}{2Q_0\omega_0C_2} \tag{3.5}$$

The second order highpass transfer function is shown in equation 3.6, where ω_0 is the same as in equation 3.2:

$$H(s) = \frac{s^2}{s^2 + \frac{\omega_0}{Q_0}s + \omega_0^2} \tag{3.6}$$

As in the highpass filter the capacitance and resistance of the transfer function of the filter are dependent on ω_0 and Q_0. The relationship between them can be seen in the equations 3.7 to 3.9.

$$\omega_0 = \frac{1}{C\sqrt{R_1R_2}} \tag{3.7}$$

$$Q_0 = \frac{1}{2}\sqrt{\frac{R_1}{R_2}} \tag{3.8}$$

$$C = \frac{2Q_0}{\omega_0R_2} \tag{3.9}$$

The values of Q_0 for each filter stage of a second and fourth order filter can be seen in table 3.1 [11].

	2nd order	4th order
Q_0 of stage 1	0.5	0.71
Q_0 of stage 2		0.17

Table 3.1: Q values for second and fourth order LR filter.

3.3.1 Calculation of Circuit Parameters

Before building the circuit in a simulation program the values of the components were calculated in MATLAB. The equations derived from the transfer function of the filters in the Linkwitz-Riley crossover filter in section 3.3 were used to calculate the component values.

The first step was to evaluate the circuit and a convenient F_c of the filter of 1 kHz was chosen. A F_c of 1 kHz was chosen because it is close to the expected F_c in the final filter adjusted to the microphones' frequency response, and to simplify the evaluation of the circuit. The performance of the filter is not affected by changing F_c in this way. Q_0 was set to 0.5 when the values of the components in the second order filters were calculated, then 0.71 for the fourth order filters as was suggested in table 3.1. Because the component values in the equations are dependent of each other the resistance R in the low pass filter was set to 1 $k\Omega$ and the capacitance C in the high pass filter was set to 1 nF.

To solve for C_2 in the LP filter, equation 3.5 needed to be rewritten as:

$$C_2 = \frac{1}{2Q_0\omega_0 R} \tag{3.10}$$

Equation 3.9 was rewritten to solve for R_2 in the HP filter, as:

$$R_2 = \frac{2Q_0}{\omega_0 C} \tag{3.11}$$

3.3.2 Simulations of the Circuit

To ensure that the theory works the circuits were built and run in Cadence PSpice. The amplitude and phase response of the second and fourth order HP and LP filters and then the complete Linkwitz-Riley filter were evaluated. The supportive circuit that filters and divides the supply voltage for the microphone was evaluated for noise.

Second Order Highpass Filter

As the first step towards the finished prototype the second order HP filter was simulated. The filter used the design suggested by Linkwitz and Riley, described in section 3.3. At first the filter was simulated using the parameters calculated in section 3.3.1. When the filter was working as desired the component values were adjusted to fit the E-12 series and simulated again.

Figure 3.3 shows the circuit. As the input, a voltage source providing a sine wave with a frequency of 1 kHz and an amplitude of 1 V was used. An offset of 1.5 V was needed to keep the signal positive at all times. The need of positive voltage is due to the fact that most of Axis' cameras can only supply positive voltage and that limits the operational amplifier to only operate on positive signals. The filters use the operational amplifier OPA377 as the active element. The capacitor C in the filter stops the offset DC component in the input voltage VIN and the biasing of 1.5 V is lost. Therefore the operational amplifier needs to be biased with 1.5V.

The circuit was simulated and an AC analysis was run first with the calculated values and then with the values from E-12 series. The values from the calculations and the adjusted values can be seen in table 3.2. To get the resistance of 160 kΩ R1 and R2 were replaced with two resistors with the values of 150 kΩ and 10 kΩ placed in series.

The amplitude response of the AC analysis of the second order highpass filter implemented with the calculated values can be seen in figure 3.4a. F_c = 1 kHz has dropped 6dB as was intended. The rolloff is 12 dB/oct as expected.

The phase is shifted 180 degrees, as can be seen in the phase response in figure 3.4b. The phase plot is limited to show only the frequencies in the range of 20–20 kHz in which the phase has not yet shifted 180 degrees.

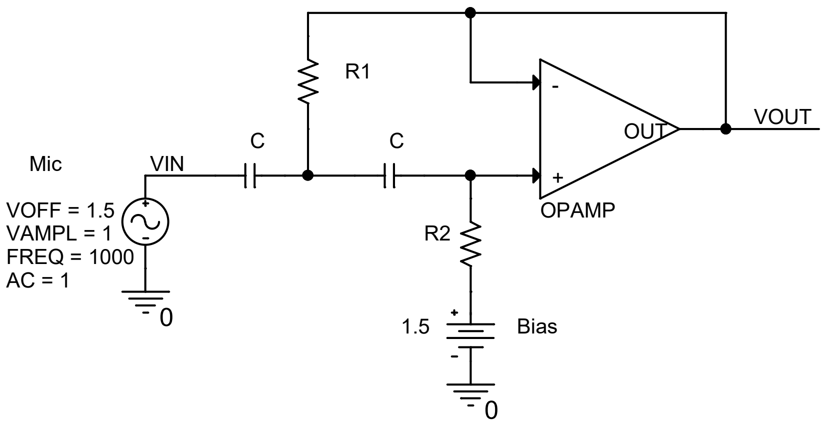
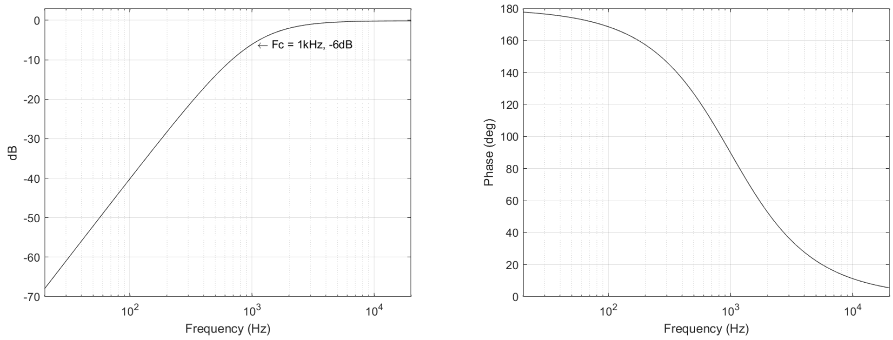


Figure 3.3: Second order highpass filter.

	Calculated values	E-12 values
C	1 nF	1 nF
R1	159 kΩ	160 kΩ
R2	159 kΩ	160 kΩ

Table 3.2: The component values in the second order highpass filter.



(a) The amplitude response showes that F_c has dropped 6 dB.

(b) Phase response. The phase is shifted 180 degrees.

Figure 3.4: Amplitude and phase response of the simulated second order highpass filter. $F_c = 1$ kHz. The filter is using the theoretical component values.

	Calculated values	**E-12 values**
R	1 kΩ	1 kΩ
C1	159 nF	160 nF
C2	159 nF	160 nF

Table 3.3: The component values in the 2nd order lowpass filter.

The AC analysis from the simulation of the circuit implemented with the components from the E-12 series shows no difference from the previous analysis.

Second Order Lowpass Filter

The second order lowpass filter was simulated in the same way as the highpass filter, except for some differences of the circuit, see figure 3.5. As is mentioned in section 3.3, the output of a second order filter is 180 degrees out of phase. The phase of the input of the lowpass filter is therefore inverted. The operational amplifier does not need to be biased since the input of the lowpass filter has a resistor that does not affect the DC.

The calculated values of the resistors and capacitors and the values adjusted to fit the E-12 series can be seen in table 3.3. The capacitor value of 160 nF was realised with a 150 nF and a 10 nF capacitor in parallel.

As for the highpass filter, an AC analysis was run. The amplitude and phase responses of the AC analysis of the second order lowpass filter implemented using the calculated component values can be seen in figures 3.6a and 3.6b. The gain at $F_c = 1$ kHz has dropped 6 dB and the the rolloff is 12 dB/oct. Phase has shifted 180 degrees, as was expected. Because the input signal to the lowpass filter is inverted the phase responses of both the highpass and lowpass filter are in phase.

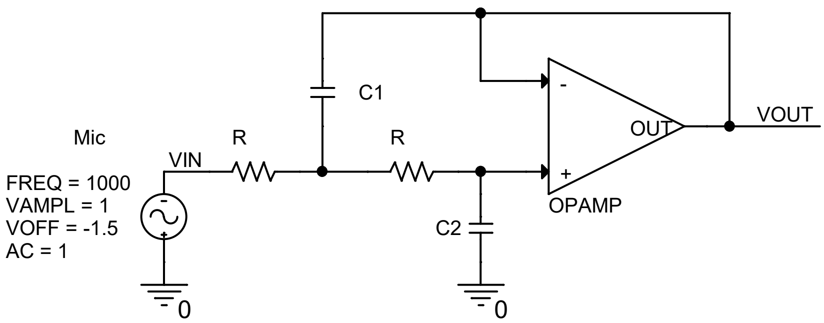
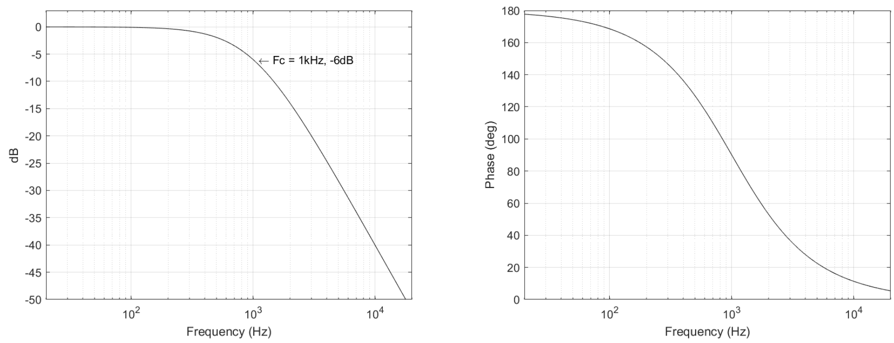


Figure 3.5: Second order lowpass filter.



(a) Amplitude response. F_c of 1 kHz has dropped 6 dB.

(b) Phase response. The phase is shifted 180 degrees.

Figure 3.6: Simulated second order lowpass filter. The filter is implemented using the theoretical component values.

The frequency response of the filter when the component values had been changed to fit the E-12 series did not differ from the frequency response of the filter with calculated values.

Fourth Order Highpass filter

The fourth order highpass filter was implemented by cascading two second order highpass filters, see figure 3.7. The component values were calculated with Q = 0.71, the Q value for 4th order filter. The values were thereafter adjusted to fit the E-12 series. The component values can be seen in table 3.4.

The frequency response of the simulated filter was given by an AC analysis. Figure 3.8a shows the amplitude response of the simulated fourth order HP filter. Because of the adjustments of the component values F_c has moved slightly from 1 kHz to 1.03 kHz. The rolloff is 24 dB/oct, the same as the theoretical value.

The phase response of the simulated fourth order highpass filter is shown in figure 3.8b. The phase has shifted 360 degrees, though the plot is limited to show

	Calculated values	E-12 values
C	1 nF	1 nF
R1	112 kΩ	120 kΩ
R2	226 kΩ	220 kΩ

Table 3.4: The component values in the fourth order highpass filter.

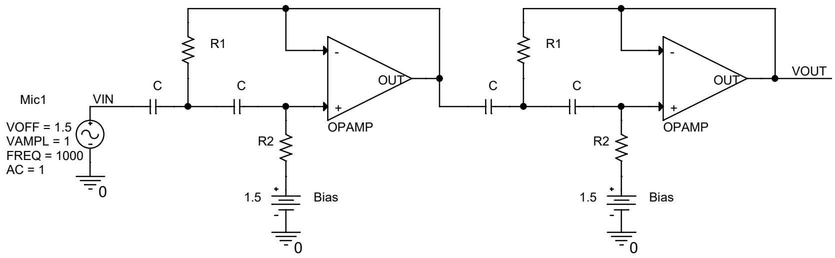


Figure 3.7: Fourth order highpass filter.

20–20 kHz where the phase has not yet shifted 360 degrees.

Fourth Order Lowpass filter

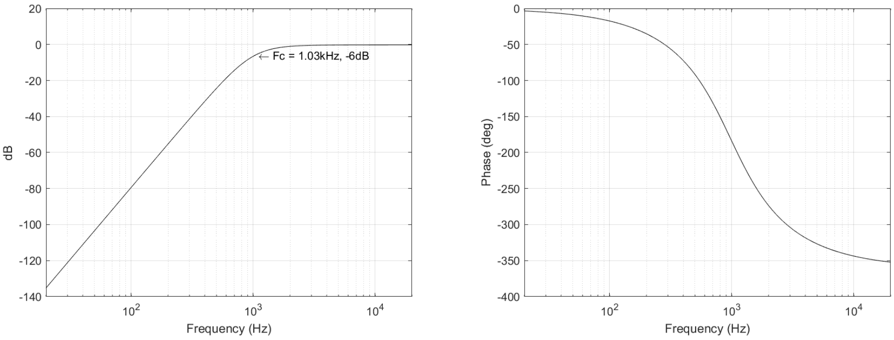
In the same way as with the fourth order highpass filter, the fourth order lowpass filter was implemented by cascading two second order lowpass filters, as can be seen in figure 3.9. The values of the resistors and capacitors were calculated and adjusted to fit the E-12 series, see table 3.5.

The frequency response of the filter was given by an AC analysis. The F_c of the simulated fourth order lowpass filter at -6 dB has shifted to a value of 1.1 kHz, see figure 3.10a. The rolloff is 24 dB/oct. The phase response of the simulated circuit has shifted 360 degrees, see figure 3.10b, thus the phase shift of the lowpass filter is in phase with the phase shift of the highpass filter. The plot with a limitation of 20–20 kHz don't show the whole shift.

The phase response of the simulated circuit has shifted 360 degrees, see figure 3.10b, thus the phase shift of the lowpass filter is in phase with the phase shift of the highpass filter. The plot with a limitation of 20–20 kHz don't show the whole

	Calculated values	E-12 values
R	1 kΩ	1 kΩ
C1	226 nF	220 nF
C2	112 nF	100 nF

Table 3.5: The component values in the fourth order lowpass filter.



(a) The amplitude response. The gain of F_c of 1.03 kHz has dropped 6 dB.

(b) Phase response. The phase is shifted 360 degrees.

Figure 3.8: Simulated fourth order highpass filter with components adjusted to fit the E-12 series.

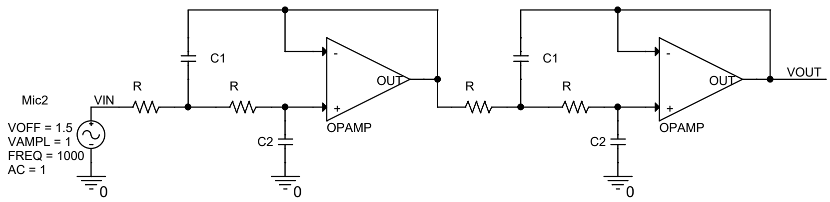


Figure 3.9: Fourth order lowpass filter.

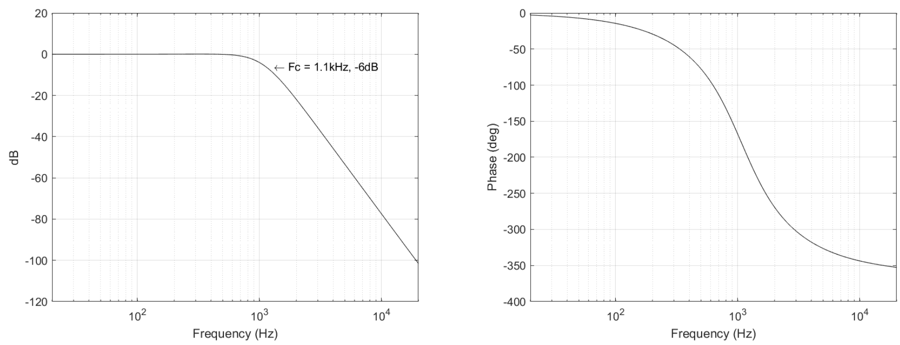
shift.

Final Filter Circuit

The signals from the fourth order highpass and lowpass filters were then summed by a buffer and a supportive circuit for the microphone was added, see figure 3.11. At this stage the bias voltage was changed from 1.5 V to 3 V, as it was convenient when making the measurements.

According to the data sheet, the microphone needed to be connected with a load resistor and a DC blocking capacitor. In the highpass filter where a capacitor is the first component, only the resistor, Rmic $= 5.6\ \Omega$, was added. In the lowpass filter a resistor and a capacitor, C3 $= 10$ nF, were added. To prevent the microphone's supportive circuit to load the lowpass filter a buffer was placed after the microphone circuit. Like the buffer in the output stage, the operational amplifier following the microphone was biased at 3 V.

After the measurements of the cancelled frequencies of the microphones on the camera were made, see section 4, a new crossover frequency for the crossover filter was calculated. The crossover frequency should be placed between the frequencies where the cancellation occurs. That places the F_c of the highpass and lowpass filters in such a way that the unwanted frequency ranges are cut off. The new F_c



(a) The amplitude response. The gain of -6 dB has moved to 1.1 kHz.

(b) Phase response. The phase is shifted 360 degrees.

Figure 3.10: Simulated fourth order lowpass filter with components adjusted to fit the E-12 series.

	Calculated values	E-12 values
C	1 nF	1 nF
R1	45 kΩ	47 kΩ
R2	90 kΩ	100 kΩ

Table 3.6: The component values in the fourth order highpass filter in the final crossover filter.

was chosen to be 2.5 kHz.

New component values for the fourth order highpass and lowpass filters were calculated and then adjusted to fit the E-12 series. To reduce the complexity of the circuit the number of components was kept as low as possible. Because of the limited amount of values to choose from in the E-12 series the F_c of the highpass and lowpass could not be matched exactly. The component values giving the smallest difference of F_c is listed bellow.

The component values of the highpass filter with $F_c = 2.3$ kHz is shown in table 3.6, and the component values of the lowpass filter with $F_c = 2.4$ kHz is seen in table 3.7.

An AC analysis was run. The amplitude response of the finished crossover filter with the test $F_c = 1$ kHz is shown in figure 3.12a. The F_c of the two filters are not matched, which results in the gain of -5.3 dB at the crossover frequency of 1.08 kHz. Because of unmatched F_c the output of the crossover filter is not flat. The phase response is shown in figure 3.12b, and as can be seen the phase has shifted 360 degrees.

The amplitude response of the crossover filter with a F_c that is adjusted for the frequency cancellations of the microphones is shown in figure 3.13a. The crossover frequency is 2.3 kHz and the gain has dropped -5.5 dB. It results in a gain of 0.5 dB at 2.3 kHz in the filter output. The calculated crossover frequency of 2.5 kHz

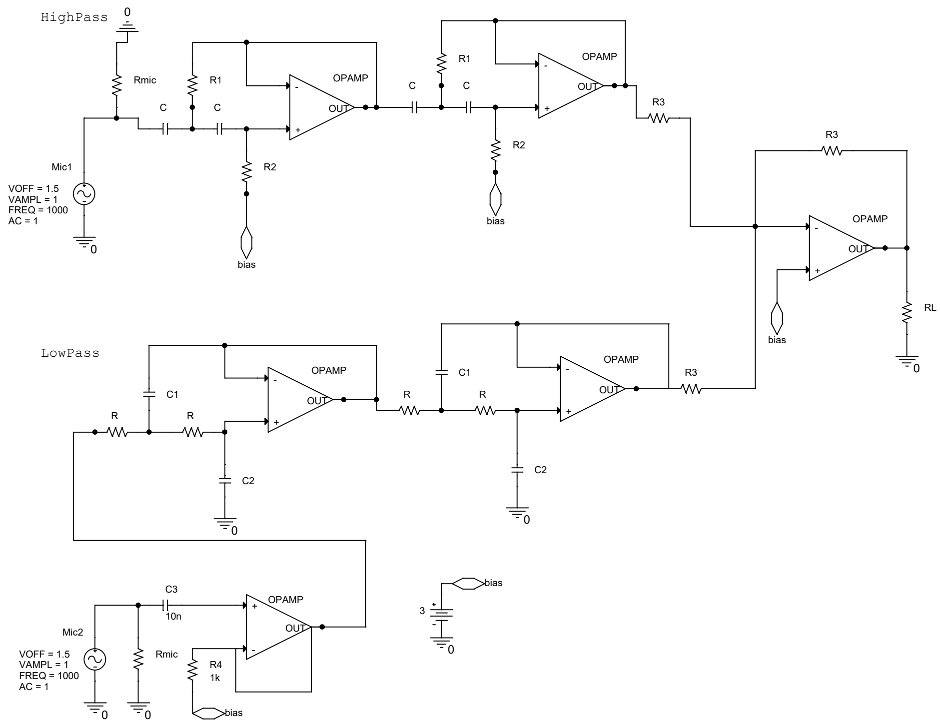
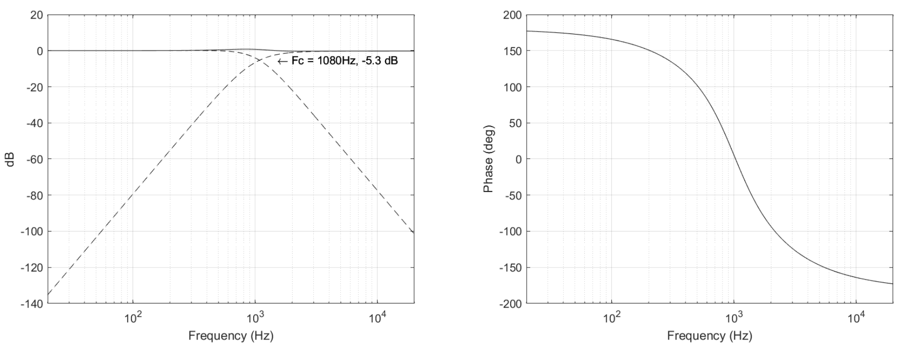


Figure 3.11: The final circuit of the filter.

	Calculated values	E-12 values
R	1 kΩ	1 kΩ
C1	45 nF	47 nF
C2	90 nF	100 nF

Table 3.7: The component values in the fourth order lowpass filter in the final crossover filter.

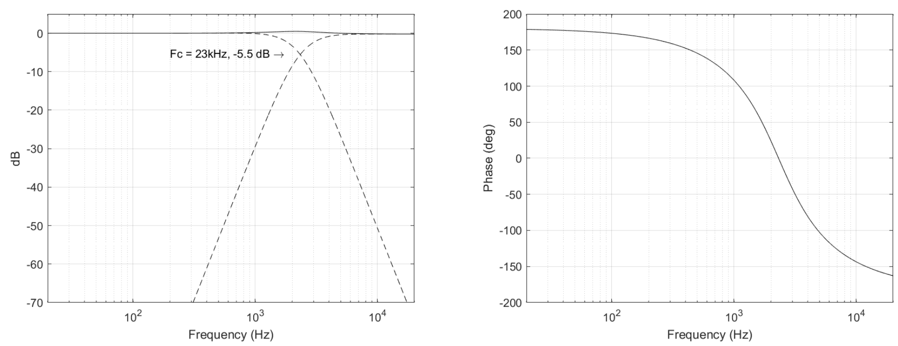
has moved 0.2 kHz. The phase response can be seen in figure 3.13b. The phase has shifted 360 degrees.



(a) The amplitude response. The gain has droped 5.3 dB at 1.08 kHz.
(b) Phase response. The phase is shifted 360 degrees.

Figure 3.12: The simulated fourth order crossover filter with $F_{\mathrm{c}} = 1$ kHz.

The amplitude response of the crossover filter with a F_{c} that is adjusted for the frequency cancellations of the microphones is shown in figure 3.13a. The crossover frequency is 2.3 kHz and the gain has dropped -5.5 dB. It results in a gain of 0.5 dB at 2.3 kHz in the filter output. The calculated crossover frequency of 2.5 kHz has moved 0.2 kHz. The phase response can be seen in figure 3.13b. The phase has shifted 360 degrees.



(a) The amplitude response. Extra gain of 0.5 dB at 2.3 kHz.
(b) Phase response. The phase is shifted 360 degrees.

Figure 3.13: The simulated fourth order final filter adjusted for the microphones.

Voltage Divider With Lowpass Filter

The microphones needed to be supplied with 5 V. The voltage source that supplies the voltage to the prototype card does not produce a clean noise free voltage and needs to be filtered because the signals from the microphones are weak and therefore sensitive to noise.

The circuit in figure 3.14 is a voltage divider with a filter and a buffer as the output stage. The voltage divider produces 5 V and the lowpass filter has a $F_{\mathrm{c}} = 1.2$ Hz. The operational amplifier in the buffer needs to be supplied with 6 V, but the amplifier OPA377 has a high Power Supply Rejection Ratio, PSRR, which means it can supress noise on supply voltage.

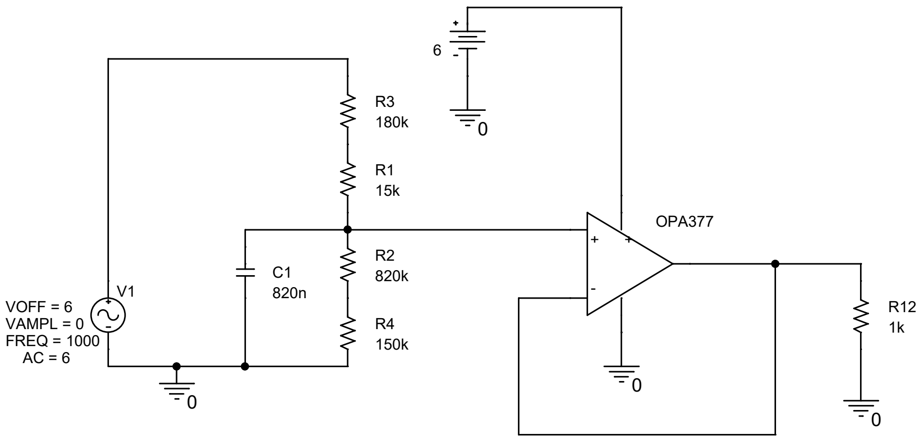


Figure 3.14: The voltage divider with lowpass filter.

To calculate the component values for the voltage divider equation 3.12 was used. The resistances were then adjusted to the E-12 series. The values in the E12 series were not close enough to the calculated resistance so a decision to use several resistors was made.

$$V_{out} = \frac{R2 + R4}{R3 + R1 + R2 + R4} V1 \tag{3.12}$$

V1 is the supply voltage of 6 V and V_{out} is the 5 V. The resistances are taken from the circuit in figure 3.14.

The component values of the filter were then calculated, using equation 3.13, where F_{c} is the cut-off frequency of the lowpass filter. C is the capacitance $C1$ and R is the resistance $(R1 + R3) \parallel (R2 + R4)$.

$$F_{\mathrm{c}} = \frac{1}{2\pi RC} \tag{3.13}$$

R and C needed to be of a large value to get a low F_{c} and to fit the E-12 series. The values of the components in figure 3.14 gave F_{c} of 1.2 Hz.

The noise density of the circuit was simulated to make sure that the high frequencies were filtered out and that the circuit's noise contribution was not exceeding the noise floor of the microphone. The noise floor, or EIN, of the output of the circuit was calculated by taking the integral of the noise density in the interval 20 Hz to 20 kHz.

The noise density of the filtering voltage divider can be seen in figure 3.15. Noise is present in the lower frequencies but is quickly attenuated. The output noise was calculated to be $V_{\mathrm{rms}} = 0.947\ \mu$V and was compared to the output noise of the microphone that was, according to its data sheet, $V_{\mathrm{rms}} = 1.4\ \mu V$.

Random, uncorrelated noises add quadratically according to the following equation, 3.14.

$$Noise_{\mathrm{tot}} = \sqrt{V_{\mathrm{rms,noise1}}^2 + V_{\mathrm{rms,noise2}}^2} \tag{3.14}$$

The total noise contribution of the filtering voltage divider and microphone is thus $V_{\mathrm{rms,noise}} = 1.69\ \mu$V, just slightly higher than 1.4 μV.

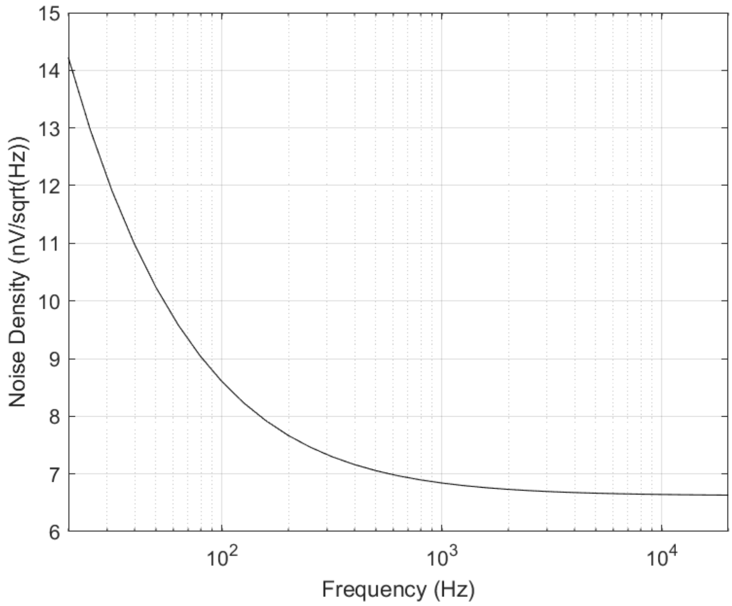


Figure 3.15: Noise density of the filtering voltage divider.

Chapter 4

The Built and Testing of the Prototype

The development of the prototype was divided into two parts. First the placing of the microphones on the camera body to measure the frequency response of the microphones. Then the construction of the actual filter itself.

Microphones on Camera

To be able to measure the destructive interference registered by the microphones, two microphones were put on a typical dome camera from Axis. The camera was then mounted on an reflective plate, see figure 4.1. The uppermost microphone was placed where the microphone on dome cameras is normally placed. It is known that the frequencies that are cancelled when the microphone is placed in that position are around 1 kHz. The second microphone was then placed closer to the bottom plate so that, according to the equations in section 2.4, the cancelled frequencies would be in a higher region.

Figure 4.1: The dome camera with two microphones. The arrows marks the placement of the microphones.

To see which frequencies were cancelled out, the microphones were tested with white noise in a quiet box in Axis' audio lab. The box is designed to resemble an anechoic chamber.

Before the testing could start the box was calibrated with a calibration microphone, *Earthworks M30*. The calibration was made to compensate for resonances in the box that would otherwise interfere with the measurements. The camera with the inserted microphones was then placed in the box, with the loudspeaker at an angle of 90 degrees to the reflective surface that the camera was mounted on. White noise was played as the measuring signal and the microphones' outputs were analyzed by the software *Spectra PLUS-SC*. Other equipment used in the measurements of the microphones were the sound card *M-Audio M-Track plus MKII* and the amplifier *Behringer iNuke NU1000DPS*.

The amplitude response of the topmost microphone can be seen in figure 4.2 The response has a clear drop of gain in the close region of 1.5 kHz, where the first cancelled frequency was estimated to be. The amplitude response also shows a small drop of gain around 5 kHz. Calculations of the estimation of the second cancelled frequency, using equations 2.1 and 2.3, gives a cancellation of frequencies at 5 kHz.

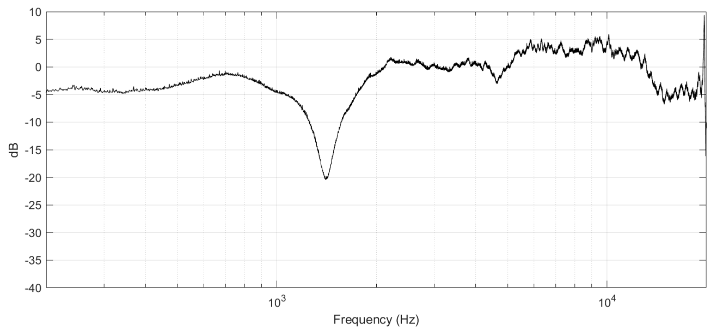


Figure 4.2: Amplitude response of the topmost microphone.

Figure 4.3 shows the amplitude response of the bottommost microphone. Like the amplitude response of the upper microphone a clear drop in gain is visible around 3.7 kHz. The first cancelled frequencies are in a higher region than the topmost microphone, as was expected. As in the measurements of the topmost microphone, the higher order cancelled frequencies can be seen in the amplitude response as well. The calculation estimate of those frequency cancellations are around 10 kHz and 17 kHz, which is consistent with the measurements.

Building of the Prototype Board

To be able to evaluate the performance of the filter with real components a prototype was built on a prototype card using surface mounted components. The prototype was made in Axis' soldering lab. The layout of the card can be seen in figure 4.4.

The layout mimics the circuit from the simulation, see figure 3.11, except for the filtering voltage divider that is placed in the lower right corner of the prototype. The highpass and lowpass filters are in the upper left corner with the buffer as output stage. The supportive circuit for the microphone is placed below

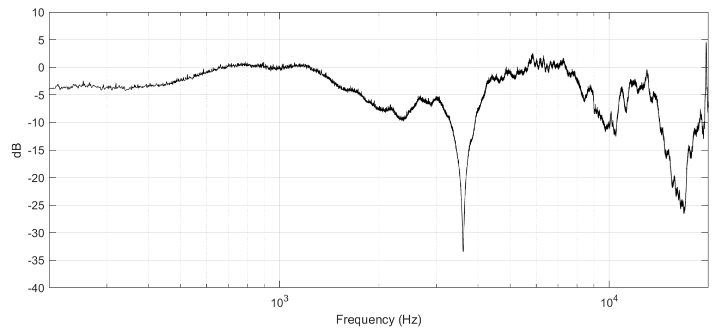


Figure 4.3: Amplitude response of the bottommost microphone.

the lowpass filter in the lower left corner. The card is fed 3 V and 6 V and has pins for the microphone output, marked *In*, the 5 V to power the microphones marked *5V Mic* and the output of the entire filter marked *OUT*.

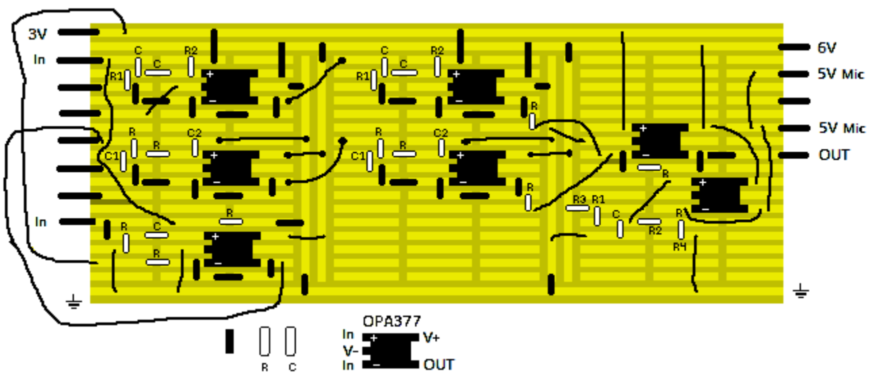


Figure 4.4: The layout of the filter prototype.

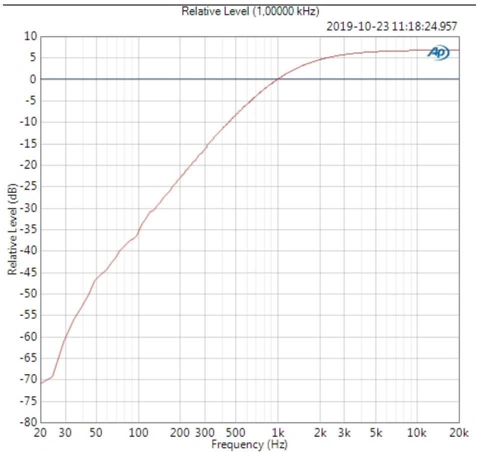
The prototype filter was built in stages after each part had been evaluated through simulation, see section 3.3.2. When each part had been built on the prototype board, tests were run in an audio lab at Axis, to evaluate the frequency response. First the filter was tested with a test signal provided by an audio analyzer and when the filter had been rebuilt to fit the microphones' frequency response, the whole system of microphones and filter was tested together.

The prototype was evaluated in steps, as was the method the simulation had used. First the second order lowpass and highpass filters frequency response was evaluated, then the fourth order filters and so on.

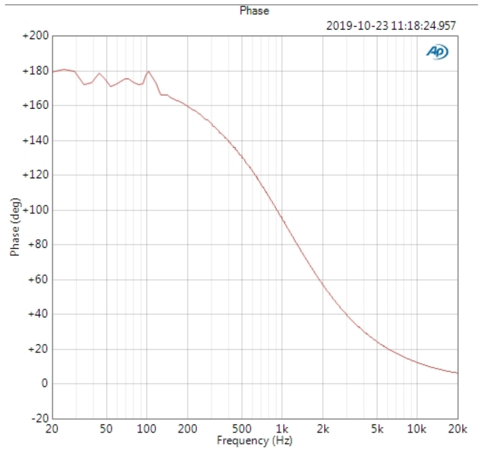
To perform the audio testing of the filters, the equipment used was the audio analyzer *Audio Precision apx526* and the power supply *Aim-TTi CPX400DP*, providing the prototype card with 3 and 6 V. The prototype card was connected to the analyzer that provided it with a sine sweep from 20 Hz to 20 kHz as input signal. The amplitude responses from the frequency analysis made by the audio analyser puts a reference line at 0 dB where the plot crosses 1 kHz.

Second Order Highpass Filter

The measured amplitude response of the prototype second order highpass filter with $F_c = 1$ kHz can be seen in figure 4.5a. As in the simulation the gain has dropped 6 dB at 1 kHz.



(a) The measured amplitude response.



(b) The measured phase response.

Figure 4.5: The frequency response of the second order HP filter on the prototype card.

The phase response of the second order HP filter can be seen in figure 4.5b. The phase makes a shift of 180 degrees as in the simulations. The noise floor of the filter is visible in the frequency range under 200 Hz, and is an effect of the filter attenuating those frequencies.

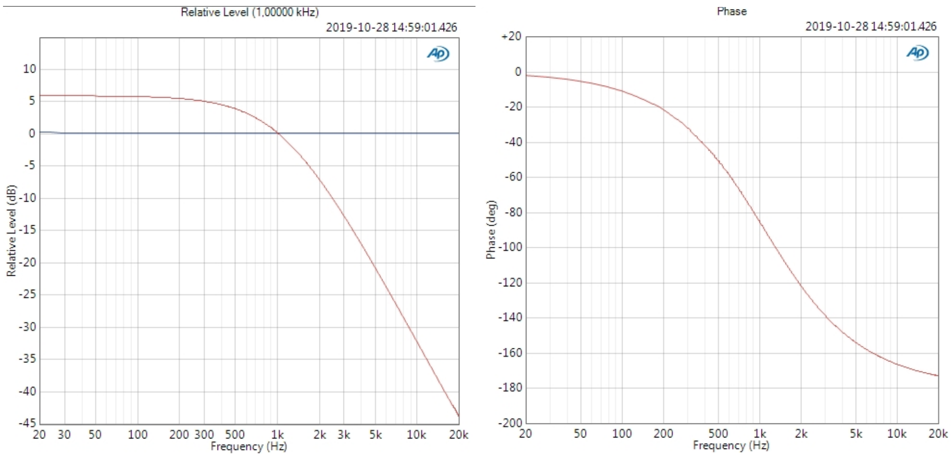
Second Order Lowpass Filter

Figure 4.6a shows the amplitude response of the measurements of the prototype second order lowpass filter with $F_c = 1$ kHz. The drop of gain at 1 kHz is 6 dB as was expected.

The phase response of the second order lowpass filter can be seen in figure 4.6b. The filter shifts the phase 180 degrees, from 0 to -180 degrees, contrary to the simulated filter were the phase shifted from 180 to 0 degrees. This is because the input signal to the lowpass filter wasn't inverted during the frequency analysis in the audio lab, as it was during simulation.

Fourth Order Highpass filter

The amplitude response of the prototype fourth order highpass filter, see figure 4.7a, shows that the filter has a drop in gain of 6 dB at approximately 1 kHz as



(a) The measured amplitude response. **(b)** The measured phase response.

Figure 4.6: The frequency response of the second order LP filter on the prototype card.

was expected. As in the the second order highpass filter the noise floor is visible in the measurement.

The phase response of the fourth order highpass filter can be seen in figure 4.7b. As in the amplitude response the noise floor is visible.

Fourth Order Lowpass filter

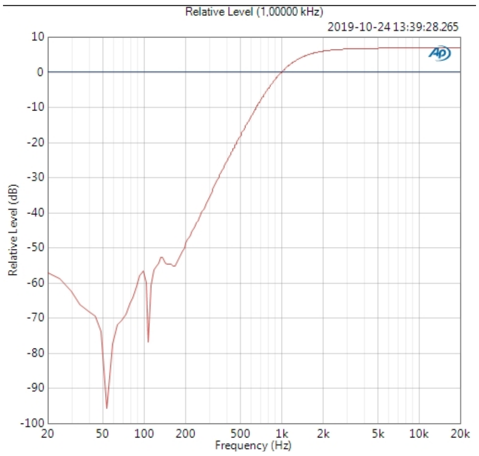
The amplitude response of the prototype fourth order lowpass filter can be seen in figure 4.8a. Because of the adjusted components in the filter, F_c at -6 dB has moved. The simulated F_c was 1.1 kHz which the real measurements also shows.

The phase response of the prototype fourth order lowpass filter shows that the phase has shifted 360 degrees, see figure 4.8b. The lowpass filter is in phase with the highpass filter.

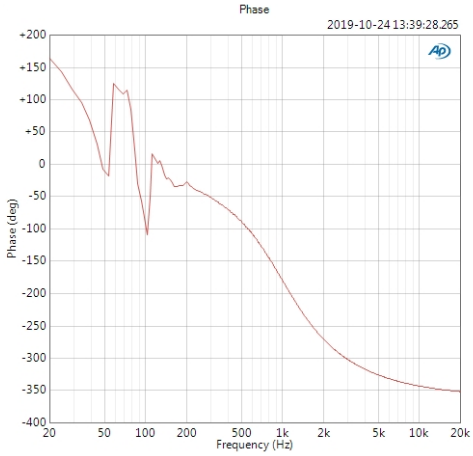
Final Filter Circuit

The measured amplitude response of the prototype crossover filter, $F_\text{c} = 1$ kHz, tested with test signals from the audio analyzer can be seen in figure 4.9a. Just as in the simulation there is an uneven amplitude gain around 1kHz. Due to the uneven relation between the component values in the filter circuits the F_c of the highpass and lowpass filter do not match exactly. The unevenness of the amplitude response is however relatively small. The phase of the crossover filter shifts 360 degrees as can be seen in figure 4.9b.

After the filter had been adjusted to have an $F_\text{c} = 2.5$ kHz and the voltage dividing lowpass filter, see section 3.3.2, the microphones were connected to the prototype card and the frequency response of the filtered microphones was measured. The measurement was done in the same way as described in section 4 with



(a) The measured amplitude response.

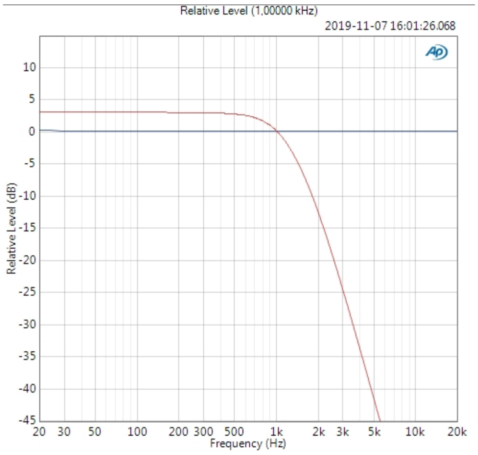


(b) The measured phase response.

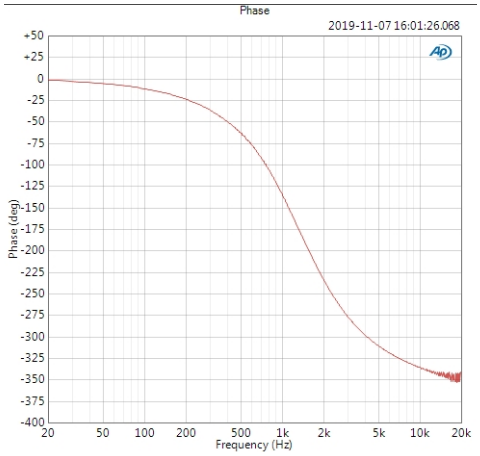
Figure 4.7: The frequency response of the prototype fourth order HP filter.

white noise as measuring signal. The output of the filter was connected to the *Spectra PLUS-SC* and an analysis of the frequency response was done.

The amplitude response of the crossover filter with signals from real microphones can be seen in figure 4.10. The drop of gain that was clearly seen in the frequency responses of the microphones, see figures 4.2 and 4.3, are evened out. Figure 4.11 shows the amplitude response of the filter output, as a black line, together with the amplitude responses of the unfiltered microphones, red and blue lines. In the figure the data has been smoothed and the frequency range adjusted to 20 – 20 kHz, for clarity.

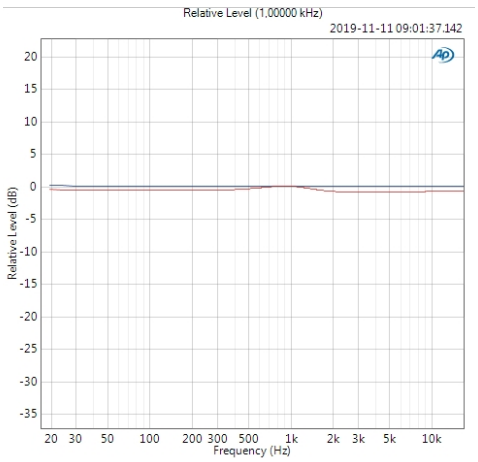


(a) The measured amplitude response.

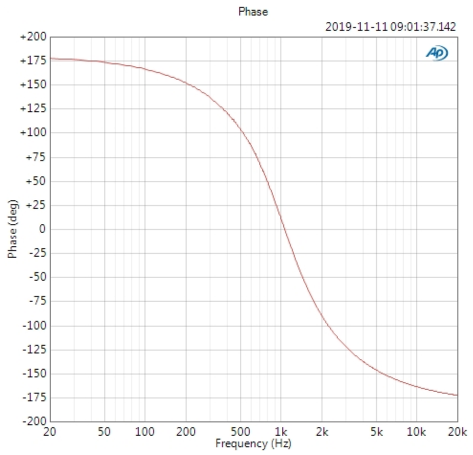


(b) The measured phase response.

Figure 4.8: The frequency response of the prototype fourth order LP filter.



(a) The measured amplitude response.



(b) The measured phase response.

Figure 4.9: The frequency response of the final circuit with $F_c = 1$ kHz. Tested with test signal.

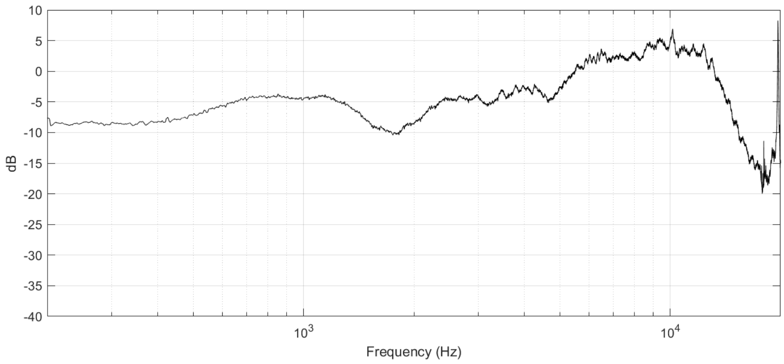


Figure 4.10: Amplitude response of the crossover filter with microphones as input signal.

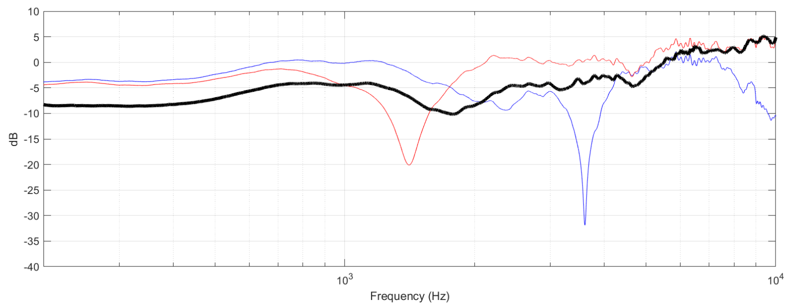


Figure 4.11: Amplitude response of the microphones, red and blue, together with amplitude response of the filter output, black.

Chapter 5
Discussion and Conclusions

In this chapter the results from the simulation and testing of the circuits are discussed. First the different parts of the filter are discussed, then the final filter.

5.1 Linkwitz-Riley Crossover Filter

The Linkwitz-Riley crossover filter was chosen for this project first of all because of it's flat frequency response but also because it's complexity is low and the filter configuration is easy to implement.

The original Linkwitz-Riley crossover filter divides one signal into two frequency bands, but the approach of this project was to use the filter to combine two signals from two microphones.

Simulation of the Crossover Filter

The results of the simulations of the crossover filter shows that the project's filter implementation works well for the purpose of the project.

The simulated frequency analysis of the crossover filter shows that the adjustments of the components to the E-12 series of resistors and capacitors causes some unwanted gain around the crossover frequency. This is because the relationship between the components in the filter is changed and that leads to a relocation of the F_c of the highpass and lowpass filters. To achieve a totally flat amplitude response of the crossover filter the F_c of the highpass and lowpass filter must match. To avoid this unwanted gain, components with values closer to the theoretical values could be chosen.

The extra gain in the crossover region was however considered small and not likely to be audible and the decision to continue with the chosen components was made.

Tests In The Audio Lab

The results of the tests in the audio lab also shows good performance.

The noise floor of the system is visible in the highpass filter, though this is expected.

As in the simulated amplitude response of the crossover filter, the uneven gain around the crossover frequency can be seen in the test results as well. The gain is however small and the amplitude response can be considered flat.

5.2 Filtering Voltage Divider

The purpose of the filtering voltage divider was to supply a clean 5 Volts to the microphones. The simulation of the noise density shows that the contributing noise floor of the filtering voltage divider is lower than the the noise floor of the microphones.

The resistance in the voltage dividing circuit are large and contributes therefore to noise. The noise contribution of the circuit could be improved by choosing smaller resistances. One of the reasons why smaller resistances were not used were that the filter would then need larger capacitors.

As this circuit is an assisting circuit to the microphones and not a part of the crossover filter, the result is considered good enough.

5.3 Prototype

Camera With Microphones

The frequency analysis of the microphones confirmed the theoretical calculations of the cancelled frequencies. It shows that the destructive interference is predictable on this sort of camera that is placed on reflective surfaces.

As the theory predicts, the measurments of the microphones shows drops of gain where the second frequency cancellation should occur. The bottommost microphone has much more noticeable cancellation than the topmost microphone. As the focus was on only the first cancellation, the reason for the differences of gain drops was overlooked, as the impact of the second cancellation was considered small.

The frequency analysis of the microphones was made with noise coming from an angle of 90 degree to the reflective surface. No other angles were tested. The filter is believed to be able to handle frequency cancellations occurring when sound waves are coming from other angles as well. The change of the difference in path length which affects the cancelled frequency, as in equation 2.3, is small. The changes of the cancelled frequencies could be handled by the filter.

Prototype Card

The prototype card was built with components commonly used by Axis. Because the component values deviated from the calculated values, the crossover frequency of the crossover filter was altered. That resulted in some uneven gain around the crossover frequency that in theory would be flat.

The result of the filter measurements are however considered good enough. The deviation from the theoretical result is small enough to not be audible.

5.4 Final Filter

The aim of the project was to find an analog audio solution to avoid frequency cancellation in the audible sound range. The method was to use multiple microphones.

The result of the testing shows that the chosen filter worked well and that the goal is met. The crossover filter manages to compensate for the frequency cancellations that are clearly seen in the frequency response of the microphones. Though the filter had some discrepancies between the theoretical performance and the real performance, the filter performed good enough.

Chapter 6

Future Work

The project focused on proving the concept of attaining a flat frequency response using two microphones. Not on optimizing the performance or keeping the noise as low as possible.

The filter was constructed to handle two precalculated frequencies. As it is now, the filter can handle some alteration of the cancelled frequencies. In the future, an evaluation of how the filter could adapt to a wider range of cancelled frequencies, within the limits of a reasonable manufacturing cost.

This project investigates the performance of the crossover filter under limited conditions. To make sure that the filter can perform under broader conditions, more tests would have to be made.

In this project the focus was on the first cancelled frequency. Cancelled frequencies in the higher region can be seen in the results of the microphone measurements, see figures 4.2 and 4.3, but they seems to have a much smaller impact on the sound quality than the first cancelled frequency. The reason for this needs further examination.

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Printed by Tryckeriet i E-huset, Lund 2021

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