EE1L1 IP-1 Intermediate Project Report

Booming Bass Sound System

Group A4 1 Filters

TU Delft

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Booming Bass Sound System

by

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1 Introduction

Speakers consist of multiple components: the power supply, the power amplifiers and the filters. The filters can be divided into three subgroups: low-pass filters, mid-pass filters, and high-pass filters. Their functions are, respectively, to pass low frequencies, mid frequencies, and high frequencies. A filter has this function to improve the clarity of the sound or protect the components in the speaker. This report will handle the measurements, simulation, designing and building of these filters, so that they can be combined with other components and thus make a qualitatively high speaker. A qualitatively high speaker is defined by its ability to have a high sound quality and an aggregate filter bank transfer function that is as flat as possible. To get the aggregate transfer function as flat as possible, the frequency overlap between filters should be as small as possible.

One of the constraints that the filter groups need to retain is that the frequency range is twenty hertz to twenty kilohertz. So, for instance, the low-pass filter passes the frequencies up to two hundred hertz, the mid-pass filter then take over from there and go up to two thousand hertz, and the high filters take over from that point and pass everything above. In addition, for each filter there is a maximum of two capacitors and inductors that can be used. Another requirement for the high-pass filter of the tweeter is that this filter should be at least a second-order filter.

There are a few steps that need to be taken to build the filter. The first step is getting a general understanding of how filters work. Next, the values of the components that are in the speaker need to be calculated. Then the filter can be designed, using the given measurements and a, with the other filter groups agreed, cutoff frequency. Afterwards, this filter will be simulated in LTSpice. If this works correctly the filter can be built and then measured using the given measuring setup. If this complies with the result that is desired, the filter is complete.

This report will go through each of these steps. Firstly, the loudspeaker analysis will be discussed, then the simulation results of the loudspeaker analysis, afterwards the design methodology will be discussed, then the simulation results of the design, then the measurement results of the design, and finally the conclusion of this report will be presented.

2

Loudspeaker analysis

2.1. Theory and Analysis

Our goal is to make a good sounding sound system. In order to achieve this, we must make filters to ensure that the right frequencies from the input signal are distributed to the right speaker elements. As is outlined in the IP-1 manual, speaker elements can be damaged or may destort when frequencies, which the speaker element was not manufactured to display, are played over it. It is essential for electrical engineers to know how the speaker behaves at different frequencies, in order to make filters tailored towards them.
As we are not provided with a scheme of the components in the speaker elements, we must make our own equivalent model of the speaker based on the model (2.1) given in the manual and the speaker impedance measurements provided to us by the project instructors. We will calculate the values of the components in the equivalent model through *speaker analysis*.

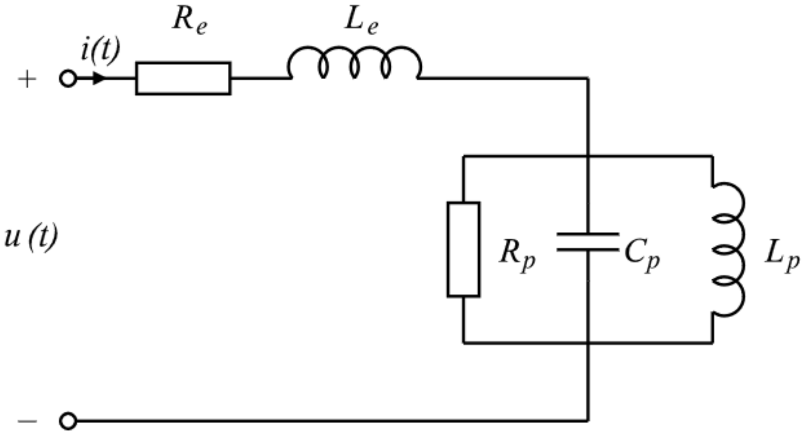

Figure 2.1: Equivalent speaker model 1 from the IP-1 manual

When doing speaker analysis, we look at the behavior of the circuit, which we can derive from the model in (2.1) even without knowing the values of the components, and compare these to the measurements of the speaker impedance. In this model, Re is the DC resistance over the voice coil, Le is the self-inductance of the voice coil, and Rp, Cp, and Lp are the electric equivalent of the mass-spring system of the speaker. We will refer to the combined impedance of Rp, Cp and Lp in parallel as Zeq,p and to the impedance of Re and Le in series as Zeq,e.

When the frequency of the circuit nears zero, the impedance of Le and Lp also nears zero according to equation (2.6). This causes Zeq,p to near zero too. As a result, the contribution of Zeq,p and

Le to the speaker's total impedance can be ignored at very low frequencies. The only component contributing to the system's impedance will then be Re.

One can observe that there is a peak in the system's impedance. This peak can be explained by the following equation of the equivalent impedance of Cp and Lp in parallel, and the equation for the resonant frequency in a parallel resonant circuit.

$$Zeq = \frac{\frac{L}{C}}{\frac{1}{j\omega C} + j\omega L} \tag{2.1}$$

$$\omega 0 = \frac{1}{\sqrt{LC}} \tag{2.2}$$

When we enter the resonant frequency in the formula for the equivalent impedance of Cp and Lp in parallel as $\frac{1}{\sqrt{LC}}$, the denominator becomes 0. This means that the impedance of Lp and Cp skyrockets at the resonant frequency of Lp and Cp, which leads to Zeq,p at the resonant frequency becoming equal to Rp and therefore also becoming purely resistive. Therefore, the impedance at the peak in the impedance graph will be equal to Re + Le + Rp, and the impedance peak will be at the resonant frequency.

At frequencies after the resonance peak, the impedance of Cp, and therefore the impedance of Zeq,p will be near zero, and thus, the electric equivalent of the mass-spring system can be replaced by a short circuit. The equivalent model of this circuit can be seen in figure (2.2), and will be referred to as model 2.

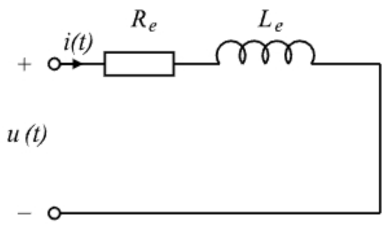

Figure 2.2: Equivalent speaker model 2 from the IP-1 manual

Finally, there is a correlation between the bandwidth of the speaker's impedance and the values or Rp and Lp according to the following equation:

$$Bw = \frac{1}{Rp * Cp} \tag{2.3}$$

2.2. Methodology

Using the theory in section 2.1, the values of the components in the filter's equivalent model can be calculated.

2.2.1. Obtaining Re

For R_e we read the total impedance at the lowest measured frequency, which is 10 Hz, of the impedance graph of the designated filter.

2.2.2. Obtaining Le

To obtain the inductance Le we first calculate the impedance ZLe at the highest measured frequency, that being 20 kHz (=40kπ rad/s), knowing that R_e is the real part of |Z| and that Z_{Le} is the imaginary part of |Z| according to model 2 (2.2). Z_{Le} is obtained through the following equations:

$$|Z| = \sqrt{(Re(Z))^2 + (Im(Z))^2} \tag{2.4}$$

This can be rewritten as:

$$Im(Z) = Z_{\text{Le}} = \sqrt{|Z|^2 - (Re(Z))^2} \tag{2.5}$$

Filling in the values for Re(Z) and |Z| at 20kHz gives us a value for Z_{Le} at 20kHz. Then we can rewrite the following equation into equation (2.7) to finally obtain Le by filling in the value of Z_{Le} for Z_L and $40k\pi$ for ω.

$$Z_L = j\omega L \tag{2.6}$$

$$L = \frac{Z_L}{j\omega} \tag{2.7}$$

2.2.3. Obtaining Rp

To calculate Rp we again rewrite equation (2.4) to first obtain Re(Z), using Z_{Le}, which can be obtained through equation (2.6) by filling in the previously calculated value for Le and the resonant frequency for ω, to obtain Im(Z) at that frequency.

$$Re(Z) = \sqrt{|Z|^2 - Im(Z)^2} \tag{2.8}$$

Filling in |Z| and Im(Z) for the resonant frequency in the equation gives us Re(Z). For the resonant frequency it holds that:

$$Re(Z) = Re + Rp \tag{2.9}$$

So to obtain Rp we subtract Re from the calculated Re(Z).

It is also possible to calculate Rp by subtracting Rp from |Z|, as the imaginary part of Z is very small at the resonant frequency, consisting only of Z_{Le}, allowing us to simplify |Z| = Re(Z) which we can use in equation (2.9).

2.2.4. Obtaining Cp

To get the capacitance Cp we can use equation (2.3). For this equation we must first know the bandwidth, which we can only obtain after knowing |Z| at the cutoff frequency. The following holds for Z_{cutoff}:

$$Z_{cutoff} = Re + \frac{Rp}{\sqrt{2}} \tag{2.10}$$

Now we can determine the bandwidth, as the bandwidth is the difference between the cutoff frequencies on both sides of the resonant frequency. The cutoff frequencies can be found by manually searching for the frequencies where |Z| = Z_{cutoff} in the speaker impedance data provided by the instructors and then picking the two frequencies closest to the resonant frequency.
By rewriting equation (2.3) we can then obtain Cp through:

$$Cp = \frac{1}{Rp * Bw} \tag{2.11}$$

Note that Bw should be in angular frequency.

2.2.5. Obtaining Lp

For Lp, equation (2.2) can be rewritten to:

$$L = \frac{1}{C\omega_0^2} \tag{2.12}$$

Lp is then obtained by filling in the resonant frequency and the previously calculated value of Cp for C.

2.2.6. Verifying the equivalent model

After calculating these values, a simulation is made in LTspice to ensure that the calculated values are correct. When the values are correct, the impedance graph of the simulation and of the measurements should be the similar, and should be the same at the lowest, resonance, and highest frequency.
The circuit should be made in LTspice using an independent voltage source with an AC amplitude of 1. Then, an AC sweep of the type decade is simuluated, with a start frequency of 10 Hz and a stop frequency of 20 kHz, to match the graph provided by the tutors. You should also make sure that a linear representation of the impedance is shown.
The graph should be of the impedance, and therefore, we measured by taking the ratio of $\frac{V_{(\text{n001})}}{I_{(R_\text{e})}}$

2.3. Calculations and measurement results

By using the methodology as stated in section 2.2 and the impedance graphs (fig. 2.3, 2.4, 2.5) provided to us by the project instructors, the following values have been calculated for each filter.

Filter	Re (Ω)	Le (H)	Rp (Ω)	Cp (F)	Lp (H)
High	4.0616	53.26×10^{-6}	1.2365	142.619×10^{-6}	97.84×10^{-6}
Band-pass	4.1423	1.759×10^{-4}	5.0495	1.751×10^{-3}	1.553×10^{-3}
Low	4.0542	181.6×10^{-6}	8.8998	1.3346×10^{-3}	3.591×10^{-3}

Table 2.1: The calculated component values for each filter

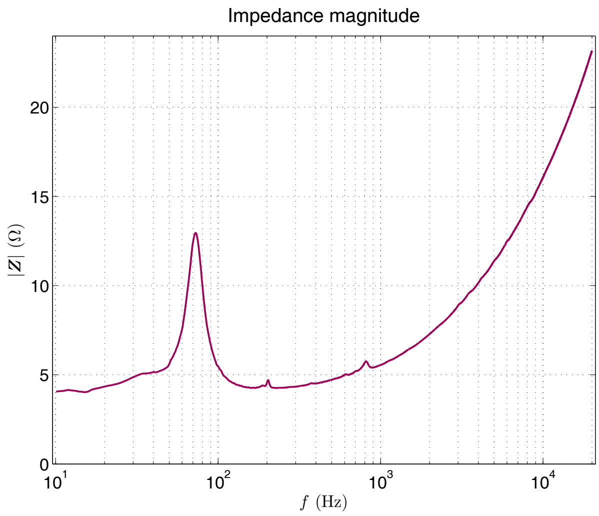

Figure 2.3: Measured loudspeaker impedance of the woofer as a function of applied frequency. This will help us calculate the values of the components in the loudspeaker model (fig. 2.1) for the woofer.

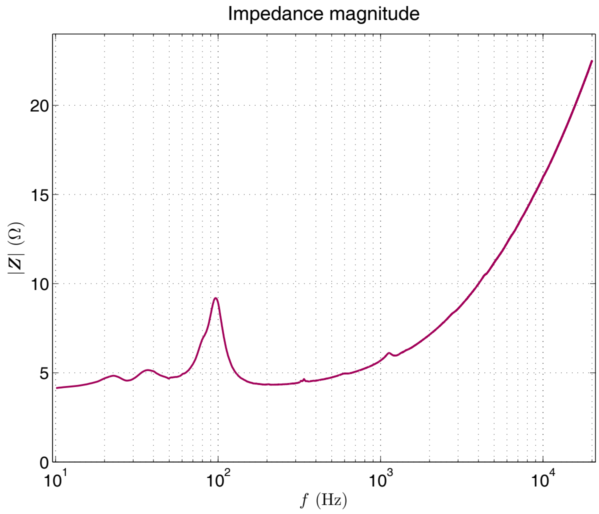

Figure 2.4: Measured loudspeaker impedance of the midtoner as a function of applied frequency. This will help us calculate the values of the components in the loudspeaker model (fig. 2.1)for the midtoner.

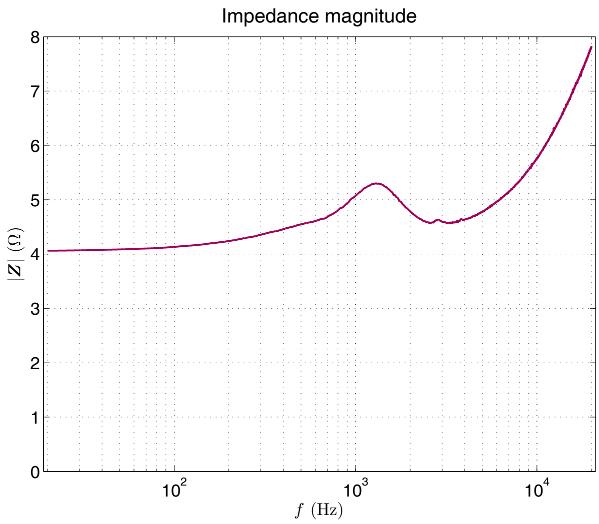

Figure 2.5: Measured loudspeaker impedance of the tweeter as a function of applied frequency. This will help us calculate the values of the components of the in the loudspeaker model (fig. 2.1) for the tweeter.

2.4. Simulations

Using the LTspice set-up described in section 2.2.6 , the following simulation results are obtained.

2.4.1. Low-pass filter

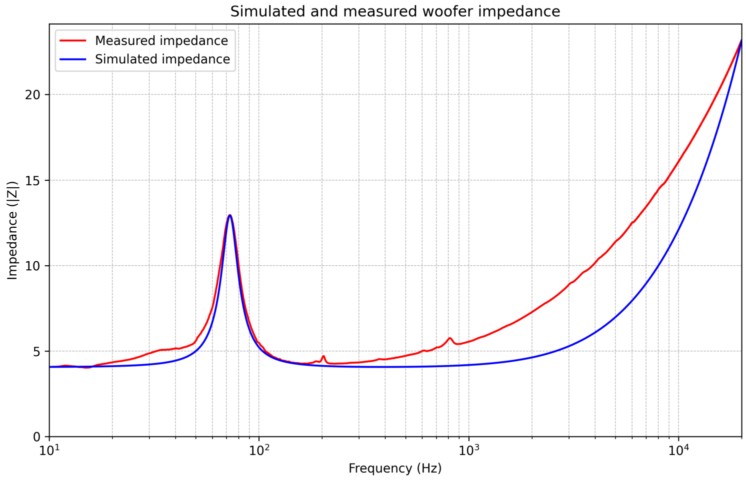

Figure 2.6: The impedance of the woofer simulated with LTSpice and the measured impedance of the woofer which was provided by the tutors

The simulated and measured impedances are shown above in figure (2.6). The simulated impedance only matches the measured impedance up until about 200 Hz. This is due to numerous reasons:
First, the calculated values are based on measurements, which means that they are not exact and that there may be an error in the measured impedance at some points, leading to a deviation in the calculated values. Secondly, the model we created is a simplification and can therefore not include all of the frequency-dependent behavior of the woofer.
The difference between the measured and simulated impedance is especially significant from 1kHz onward as the discrepancy of the measured impedance relative to the simulated impedance becomes greater than 50%, but is already noticeable after around 400Hz. However, these measurements are accurate enough to aid in the design of a designated low filter, as they are accurate within the frequencies for which we intend to use the woofer. The simulated and measured impedance at the points of 10 Hz, 72.7 Hz and 20 kHz match, as is expected, for the values of the components in the equivalent circuit of the woofer were calculated using the measurements at those frequencies.

2.4.2. Band-pass filter

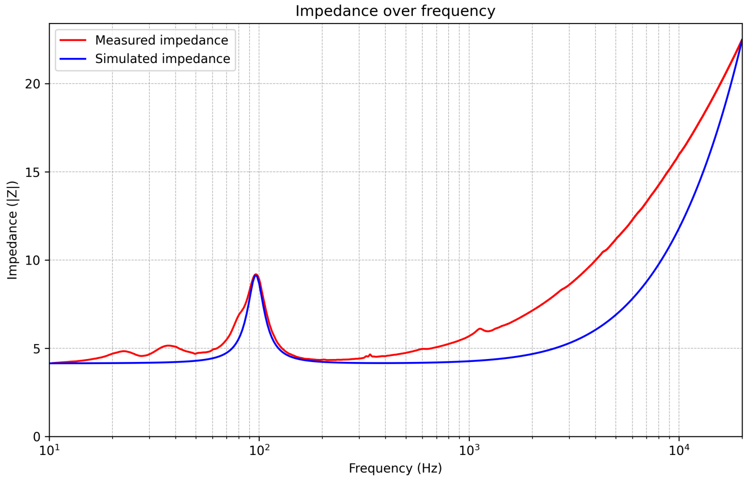

Figure 2.7: Bandpass impedance simulated using LTSpice and provided measured impedance

The measured and simulated impedance graphs show clear correlation aswell as points of deviation. Both curves follow the same general trend, with a resonant peak around 100 Hz and increasing impedance at higher frequencies. The measured impedance consistently exceeds the simulated impedance, particularly at higher frequencies eventually leading to a clearer divergence close to 1kHz, likely due to measurement errors, or material imperfections. This divergence however becomes less insignificant closer to 20kHz until the measured and simulated impedance have the same trajectory and line. The resonant peak in the measured data is broader and slightly higher, suggesting greater damping or energy loss in the real system compared to the simulation. At lower frequencies, the curves align closely, indicating the model performs well in this range, while at mid and high frequencies, deviations become more pronounced, possibly due to additional loss mechanisms or limitations in the simulation model.

2.4.3. High-pass filter

The simulated impedance matches the shape of the measured impedance. However, the blue curve lies lower than the red curve in the graph in some places. This deviation starts around 50 Hz and becomes bigger in between 50 Hz and 500 Hz,and then becomes smaller in between 500 Hz and 1347.3 Hz (the resonance peak), where the two curves match up again. After the resonance peak, the two curves deviate again. At 20 000 Hz the two curves match up again. These points of the matching of the two curves are expected, because at these frequencies the values of the components of the model are calculated. The deviation between the curves is at most 0.3 Ohm. The difference in the simulated and the measured impedance can have multiple reasons.
Firstly, the calculated values of the components are based on measurements. These measurements can have errors in them, or deal with uncertainties in the measurement equipment. This can lead to a deviation in the calculated values for the components and can thus lead to a deviation in the curve of the impedance.
Secondly, the model we created is a simplification of reality. Thus, this model cannot include all of the frequency-dependent behavior of the tweeter. However, these measurements have a low enough deviation, and are thus accurate enough to help design the high-pass filter.

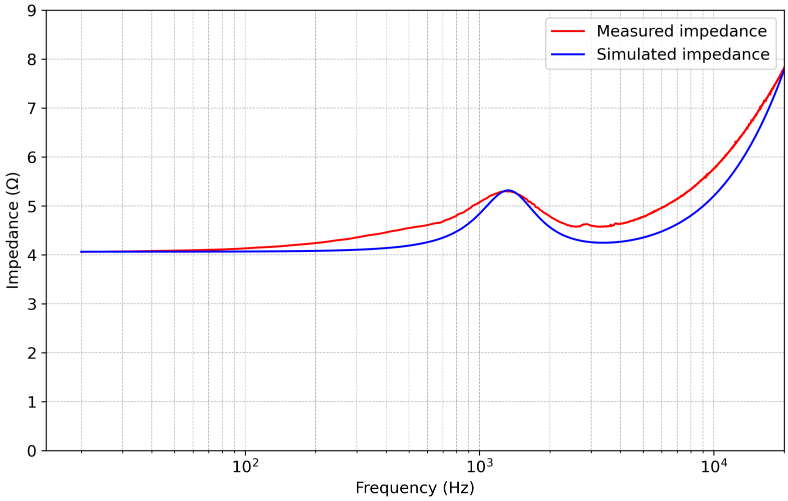

Figure 2.8: Measured loudspeaker impedance (red curve) as a function of applied frequency, compared to our model (blue curve). Over the region of interest (20 - 20 000 Hz) the deviation between the model and the measurement is less than 0.3 Ohm.

2.5. Conclusions

The simulated results of the calculated values deviate compared to the measured impedance. These graphs will never be exactly the same, because the calculated values are based on measurements. This means that these values are not exact. Furthermore, there may be an error in the measured impedance leading to a deviation of the calculated values.

3

Passive Filter Design

3.1. Introduction

In loudspeaker systems, filters play a critical role in ensuring that audio signals are properly distributed among the various drivers, such as the tweeter, midrange driver, and woofer. Each of these components is designed to handle a specific range of frequencies. Filters serve as a vital intermediary, directing manageable and appropriate frequencies to the corresponding drivers, thereby optimizing performance and protecting the system from damage.
For example, a high-pass filter removes low frequencies from the signal, allowing only higher frequencies to reach the tweeter. This is essential because tweeters are not mechanically capable of reproducing low-frequency sounds and could be damaged if exposed to them. Similarly, low-pass filters restrict higher frequencies, ensuring that only bass frequencies are sent to the woofer, which is specifically designed for low-end sound reproduction. Midrange drivers, which handle frequencies between the bass and treble ranges, often rely on band-pass filters to allow only the desired midrange frequencies through.
This division of audio signals into distinct frequency bands is crucial for enhancing the overall sound quality of a loudspeaker system. By ensuring that each driver operates within its designed frequency range, filters prevent distortion and improve the clarity of the reproduced sound. Additionally, this targeted signal delivery protects the components from receiving frequencies they cannot handle, which could lead to inefficiency or even physical damage.
There are a few requirements for the designing and building of the filters. Firstly, the high-pass filter has to be at least of the second order. Secondly, each filter can only use a maximum of 2 capacitors and a maximum of 2 inductors in its design.

3.2. Design Methodology

3.2.1. Zobel Network

A Zobel network makes the impedance of the combination of the speaker element and the Zobel network completely real and consists of a RC-network. Not all speaker elements need a Zobel network.
Equations with regards to a Zobel network found in the manual:

$$C_{Zobel} = \frac{Le}{Re^2} \tag{3.1}$$

$$R_{Zobel} = Re \tag{3.2}$$

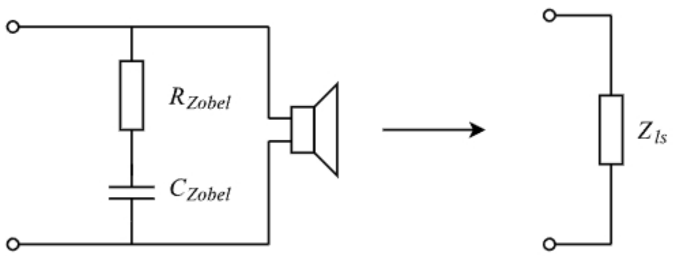

Figure 3.1: A Zobel network

3.2.2. Cut-off frequencies

When making a filter bank it is important to decide a clear cut-off frequency. By doing this it is agreed to which frequency each filter works at. The decision for the cutoff frequency was decided by looking at the acoustic amplitude response functions of the filters (Figures 3.2, 3.3, 3.4). The cutoff frequency for the high-pass and band-pass filters was agreed at approximately 1500 hertz. This was done because the power response is best at that point for both filters. For the low-pass and band-pass filters the cut-off frequency was placed at 232 hertz for the same reason.

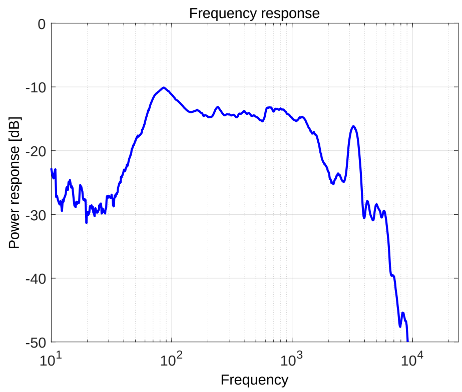

Figure 3.2: The acoustic amplitude response of the woofer as a function of applied frequency.

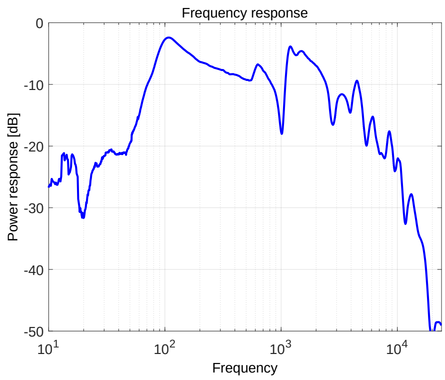

Figure 3.3: The acoustic amplitude response of the midtoner as a function of applied frequency.

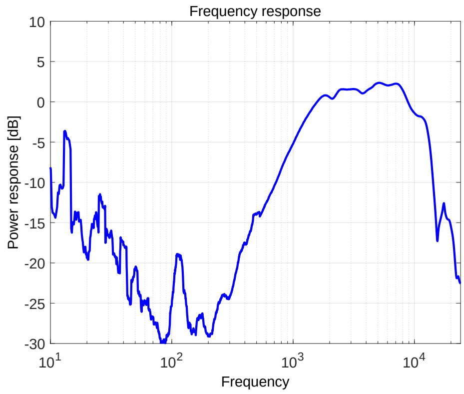

Figure 3.4: The acoustic amplitude response of the tweeter as a function of applied frequency.

3.2.3. First and Second Order Filters

3.2.3.1. Low-pass filter

Before starting the design of a low-pass filter, it was checked whether or not a Zobel correction network was necessary. Using the data of the equivalent model of the woofer from table 2.1 and equations (3.1) and (3.2) the following values for R_{Zobel} and C_{Zobel} for the woofer were obtained:
R_{Zobel} = 4.0542Ω
C_{Zobel} = 11.05μF
With these values a circuit of the woofer with the correction network in parallel was constructed, shown below:

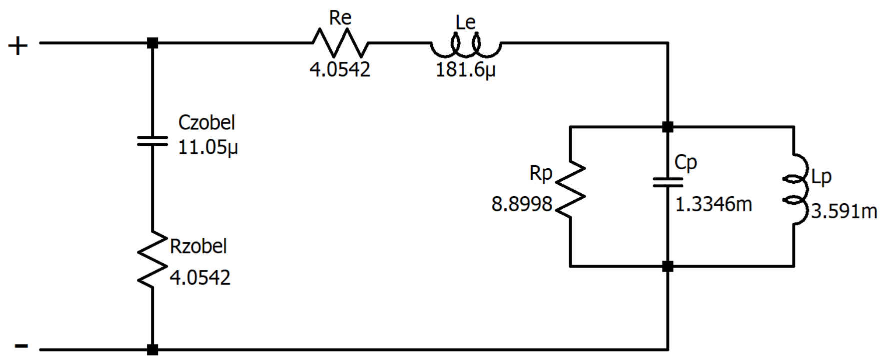

Figure 3.5: Woofer circuit diagram with Zobel correction network, constructed in LTSpice

When performing an AC sweep through LTSpice, and comparing the impedance data of the simulated woofer impedance with/without the Zobel correction network the following graph is obtained:

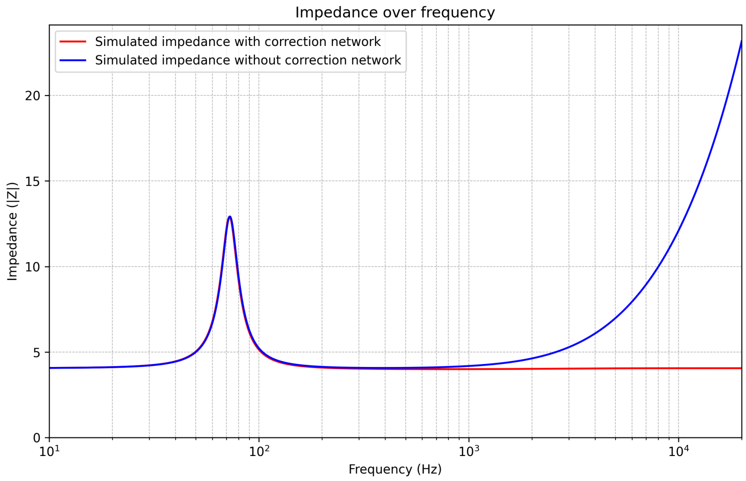

Figure 3.6: The simulated impedance of the woofer with and without the zobel correction network implemented

From the above figure it is evident that a Zobel correction network is not needed for the low-pass filter. The impedance with and without the correction network only diverges past around 1kHz, which is beyond the operating range of most woofers and beyond our planned crossover frequency of low to mid.

3.2.3.2. Band-pass filter

A band-pass filter consists of a low-pass filter and a high-pass filter, and for our filter we chose to use two second-order filters for the band-pass filter. This makes sure that the roll off is steeper, and the cut-off frequency selection is more specific.

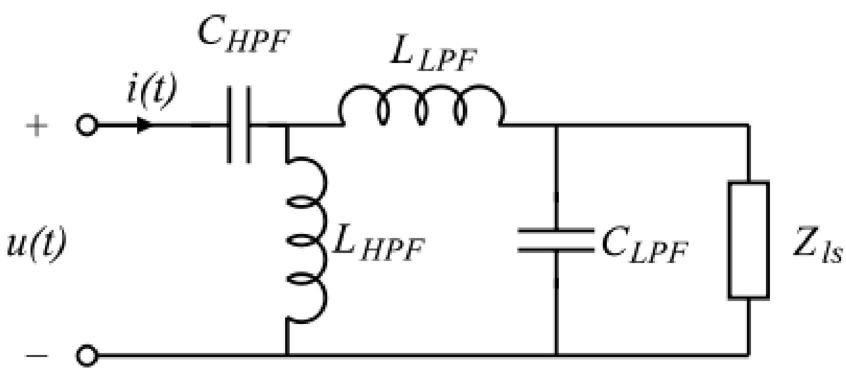

Figure 3.7: Second-order band-pass filter

3.2.3.3. First-order high-pass filter

In the manual of this project, there was an assignment about a first-order high-pass filter, and therefore, we have derived the transfer function for this order.

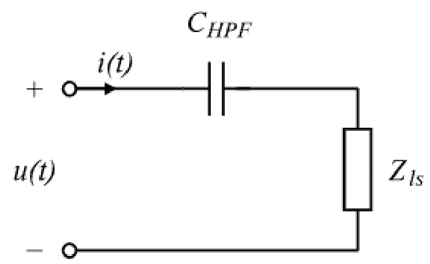

Figure 3.8: A first-order high-pass filter

For the first-order high-pass filter, the following equation applies for the output voltage, with Z_{ls} being the impedance of the loudspeaker element:

$$U_{\text{out}} = \frac{Z_{ls}}{Z_{ls} + \frac{1}{j\omega C}} \cdot U_{\text{in}} \tag{3.3}$$

$$H_{\text{HPF}}(\omega) = \frac{U_{\text{out}}}{U_{\text{in}}} = \frac{Z_{\text{ls}}}{Z_{\text{ls}} + \frac{1}{j\omega C}} = \frac{Z_{\text{ls}} \cdot j\omega C}{1 + Z_{\text{ls}} \cdot j\omega C} \tag{3.4}$$

The power transfer is formulated by:

$$G_{\text{HPF}}(\omega) = |H_{\text{HPF}}(\omega)|^2 \tag{3.5}$$

Thus, the power transfer for a first-order high-pass filter becomes:

$$G_{\text{HPF}}(\omega) = \frac{Z_{\text{ls}}^2 \omega^2 C^2}{1 + Z_{\text{ls}}^2 \omega^2 C^2} \tag{3.6}$$

The -3 dB frequency is formulated by:

$$f_{\text{-3dB}} = \frac{1}{2\pi Z_{\text{ls}} C} \tag{3.7}$$

So, the angular frequency will be: $\omega = \frac{1}{Z_{ls} C}$. Filling in this value into the equation of the power transfer gives $G_{\text{HPF}}(\omega) = \frac{1}{2}$. This proves that equation (**??**) is indeed the -3 dB frequency, because the power dissipated by the speaker has reduced by half.

3.2.3.4. Second-order high-pass filter

In the manual there was another assignment about the second-order high-pass filter to derive the output voltage of this circuit.
The equivalent impedance of Z_{ls} and L is denoted by Z_{eq}. This gives the equation of the output voltage:

$$U_{out}(\omega) = \frac{Z_{eq}}{Z_{eq} + \frac{1}{j\omega C}} \cdot U_{in} = \frac{-Z_{ls} j\omega L}{Z_{ls} j\omega L + \frac{1}{j\omega C} \cdot (Z_{ls} + j\omega L)} = \frac{-Z_{ls}\omega^2 LC}{-Z_{ls}\omega^2 LC + Z_{ls} + j\omega L} \tag{3.8}$$

Using equation 2.2 we get that $LC = \frac{1}{\omega_0^2}$. Then dividing equation 3.8 by Z_{ls} and using the equation

$$\frac{L}{Z_{ls}} = \frac{1}{Q\omega_0} \tag{3.9}$$

gives:

$$U_{out}(\omega) = \frac{-\frac{\omega^2}{\omega_0^2}}{1 - \frac{\omega^2}{\omega_0^2} + j\frac{\omega}{Q\omega_0}} \cdot U_{in} \tag{3.10}$$

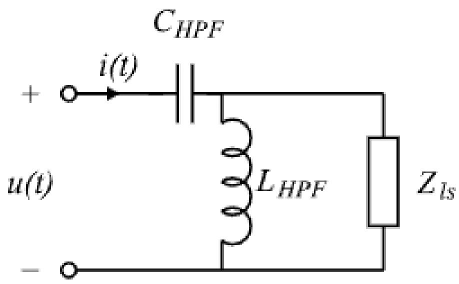

Figure 3.9: A second-order high-pass filter

3.2.4. Building the filters

Not all values of the components are available in the lab. We approximated our ideal design with the available components listed in the IP1-manual, and simulated this design in LTspice.

The circuit should be made in LTspice using an independent voltage source with an AC amplitude of 1. Then, an AC sweep of the type decade is simulated, with a start frequency of 20 Hz and a stop frequency of 20 kHz. You should also make sure that a linear representation of the impedance is shown.
The graph should be of the impedance, and therefore, we measured by taking the ratio of $\frac{V_{(n002)}}{V_{(n001)}}$
After simulating, the values of the components that are going to be used, are clear. Table 3.1 shows the values of the used components for each filter. The filter is built on a printed circuit board with an

Filter	Re (Ω)	Le (H)	R_{Zobel} (Ω)	C_{Zobel} (F)
High	4	0.55×10^{-3}	4	15×10^{-6}
Band-pass	4.1423	1.759×10^{-4}	n.a	n.a
Low	4.0542	181.6×10^{-6}	n.a	n.a

Table 3.1: The component values used in the final design for each filter

input, an output, and two ground plug-ins. We soldered our circuit together.

3.2.5. The measurement set-up

For measuring the complex impedance of a speaker, the project instructors use the Matlab program LS-Measure. This program produces a signal to measure the response at all frequencies simultaneously. Appendix H of the IP1-manual tells you more about the measurement system and set-up.

3.3. Simulations

3.3.1. Low filter

Using LTSpice, the voltage transfer of the low filter is plotted. This results in figure (3.10).

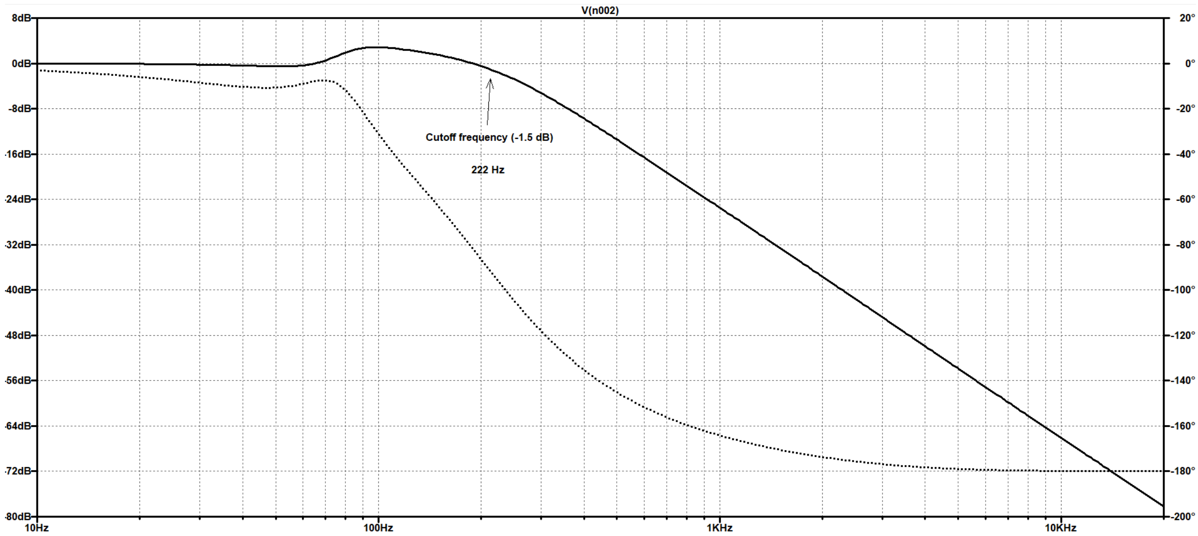

Figure 3.10: Simulated voltage response of the low filter

The cutoff frequency is close to the intended cutoff frequency of 232 Hz, and the 'power bulge', although slightly overlapping with the acoustic amplitude peak, is covering the right frequencies in order to flatten out the final acoustic amplitude response of the woofer.

3.3.2. Band-pass filter

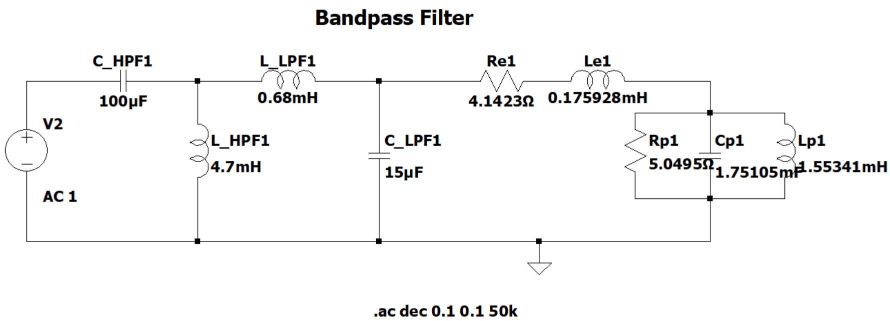

Figure 3.11: Bandpass Realistic Component Schematic

Using the calculated values from the Table 1, the most ideal components were soldered as in the realistic component schematic figure 3.11. The final cutoff frequencies for the midpass filter for these components were: 232.15Hz, 1575.87. Close to the intended cutoff frequencies.

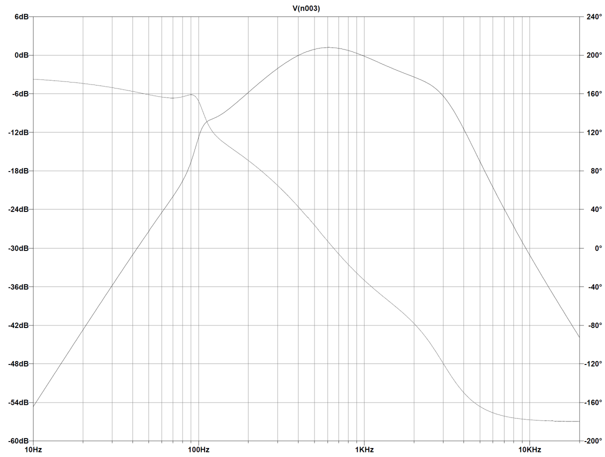

Figure 3.12: Simulated voltage response of the bandpass filter

Using LT Spice, the resulting voltage response was then simulated.

3.3.3. High-pass filter

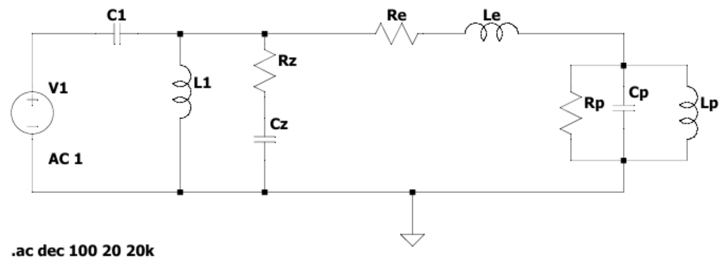

Figure 3.13: The design of the second-order high-pass filter using a zobel network.

The simulation of the second-order high-pass filter (figure 3.13) was done via nodal analysis in LTSpice.

Component	Value	Unit
C1	15.0×10^{-6}	F
L1	0.55×10^{-3}	H
Rz	4.00	Ω
Cz	3.30×10^{-6}	F

Table 3.2: The values of the components in the high-pass filter circuit.

Table (3.2) shows the values of the used components. In figure (3.14), the Bode plots of the design of the high-pass filter are visible. The -3 dB cut-off frequency is a frequency of 1489 kHz. The most important part is that the transfer function is as flat as possible for high frequencies. This can be seen in the graph of the magnitude.
In the graph of the phase, it can be seen that the phase starts at 20 Hz at about 180 degrees. Then, the phase starts to descend slowly until 100 Hz, where it starts to descend more quickly. At 1000 Hz, the phase starts to descend even more quickly. At the -3 dB frequency (1489 kHz), the phase starts to descend less quickly. The phase is at a value of around 10 degrees at 20 000 Hz.

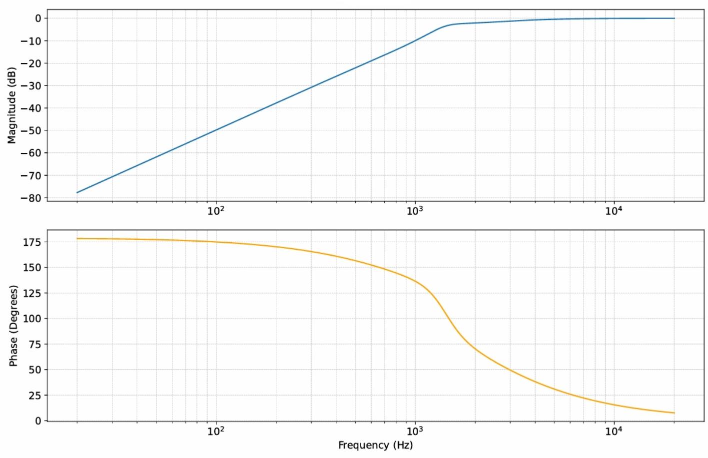

Figure 3.14: Simulated Bode plots of the high-pass filter design. The magnitude (blue curve) as a function of the applied frequency. The -3 dB cut-off frequency is 1489 kHz. And the phase (yellow curve) as a function of the applied frequency.

3.4. Measurements

3.4.1. Low-pass filter

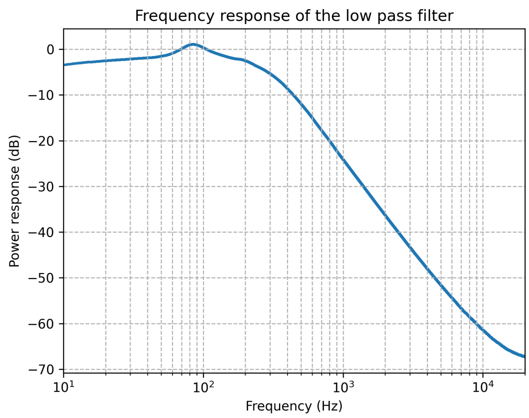

Figure 3.15: Low filter voltage transfer

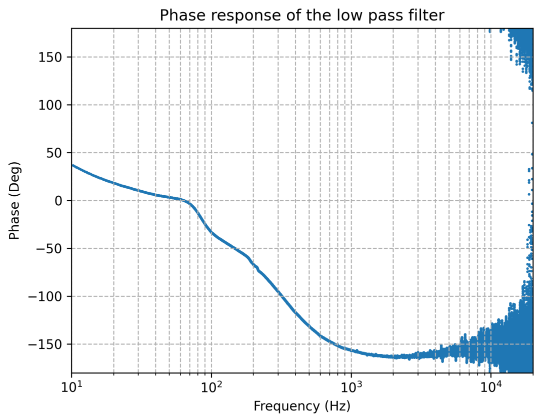

Figure 3.16: Low filter phase transfer

It is visible that the measured voltage transfer of the low filter is not the same as the simulated voltage transfer. The most important reason for this is that there wasn't accounted for the inductors and capacitors internal resistance in the simulations. By assuming that the inductors and capacitors were perfect, the simulations portray a result that is too good to be true. Apart from that, there also wasn't accounted for the error margin of the inductance and capacitance of the components used in the filter during simulation.

3.4.2. Band-pass filter

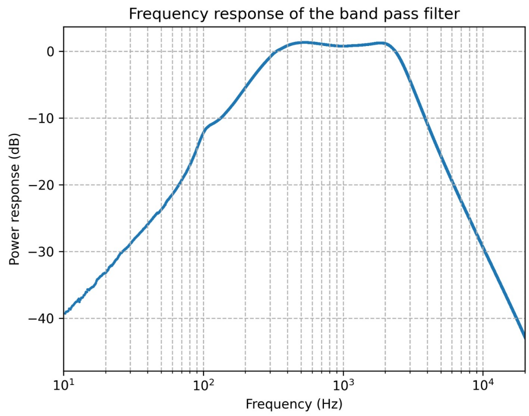

Figure 3.17: Frequency response of the bandpass filter

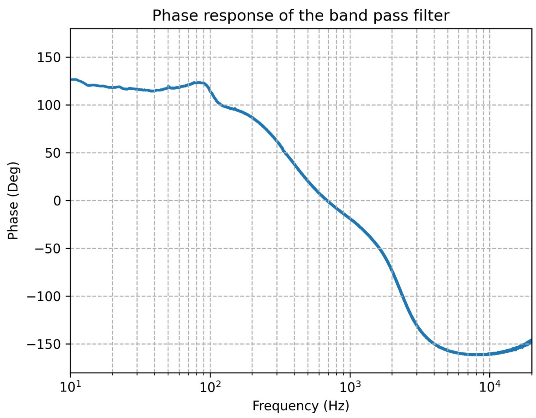

Figure 3.18: Phase response of the bandpass filter

In figure (3.17) the frequency response is shown, and you can see that the frequency response between the cut-off frequencies is very flat and then in drops quite steep. This is ideal for a band-pass filter. The phase response(3.18), however, is not ideal; the phase at the cut-off frequencies goes from around 75 deg to -160, this is a huge difference and noticeable to the human ear. Since the phase shits is more than 180 degrees the signals will cancel each other out, but we found that by reversing the poles, and adding 180 degrees to the phase, the phase lines up to the low-pass and high-pass filters.

3.4.3. High-pass filter

The comparison of the measured and the simulated frequency response of the high-pass filter is portrayed in figure (3.19). We can see that the measured corner frequency matches the simulated corner frequency. The shape of the two graphs is also approximately the same. At low frequencies, the measured magnitude oscillates. Despite these oscillations, the graph gradually increases, and the oscillations decrease as the frequency gets higher. At around 300 Hz, the oscillations stop, and the graph still increases.

In the simulation, we work with ideal circumstances, such as ideal capacitors and inductors.These do not account for the internal resistance that the capacitor and the inductor have. In reality, these circumstances are never ideal and there are internal resistances, and that is why the simulated and measured magnitudes will never be exactly the same.

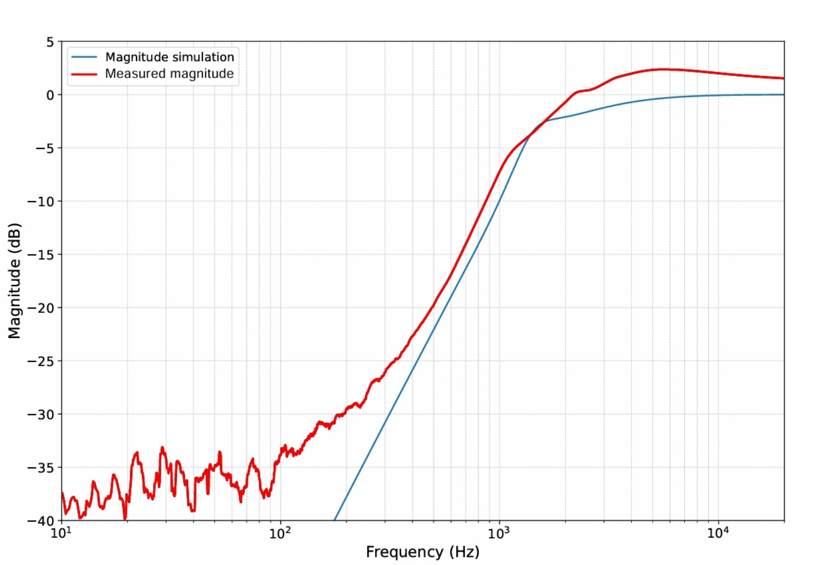

Figure 3.19: Measured frequency response magnitude (red curve) as a function of applied frequency, compared to our simulation (blue curve). The measured corner frequency matches the one of the simulation.

Figure (3.20) shows the measured phase response of the high-pass filter. The curve shows a lot of oscillation before 800 Hz, which dies down gradually as the frequency gets higher. After 800 Hz, the phase response does not have any oscillations. The shape of this curve is the shape we expected, because it descends as the frequency gets higher. However, these results still suffice and will work well for the high-pass filter in the filter bank because the oscillation stops before the cut-off frequency of 1500 Hz.

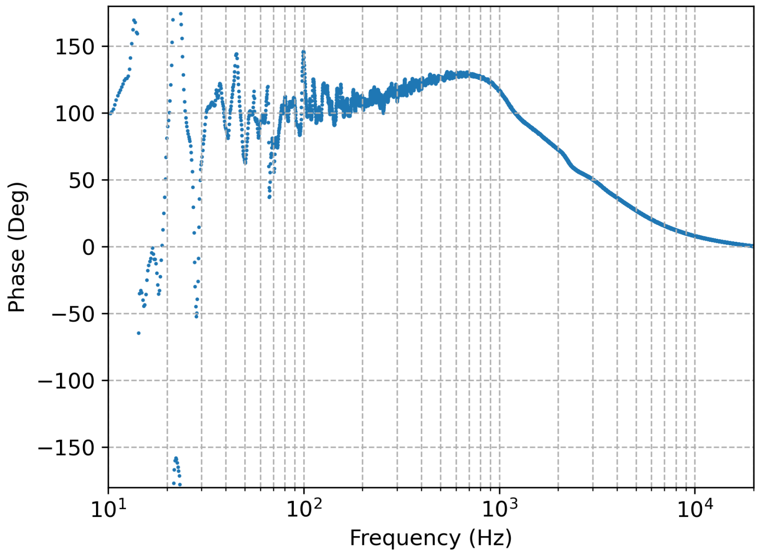

Figure 3.20: Measured phase response (blue curve) as a function of applied frequency.

3.5. Conclusions

There were some deviations from the measured results and the simulated results. A probable reason for this is that the capacitors and inductors internal resistances were not accounted for in the simulation. Another reason is that there was not accounted for the error margin of the inductance and capacitance of the components used in the filter during the simulation.

4
Conclusion

In conclusion, we started with the loudspeaker analysis to obtain values for the components of the equivalent model of the loudspeaker. These calculated values were then simulated to verify that these values were correct. When the graphs of the measured impedance and the simulated impedance with the calculated values were about the same, the values are correct. These graphs will never be exactly the same, because the calculated values are based on measurements. This means that these values are not exact. Furthermore, there may be an error in the measured impedance leading to a deviation of the calculated values.

Then, we started designing our filter. This started with determining what the cut-off frequencies would be based on the acoustic measurements provided to us. Our cut-off frequencies are 232 Hz and 1500 Hz.
The high-pass filter implemented a Zobel network to make the impedance of the combination of the speaker element and the Zobel network completely real. The other filters did not implement a Zobel network.
The circuits of the filters were simulated and some values were adjusted because not all values of components were available in the lab. We concluded that the simulated results were good based on the cut-off frequencies and the shape of the curves.
Finally, we measured the power functions of our filters and compared them to our simulation results.

The measurements and simulations varied which was likely attributed to the non-ideal behaviour of components. This differnce however, did not serve to undermine the correlation from simulation to measurement, which indicates likely that these discrepancies were purely based on real world factors affected the physical circuits or the measurement setup.

We satisfied our requirements, because our filter range is 20 Hz to 20 000 hertz. The low-pass filter only passes low frequencies up to 232 Hz, the band-pass filter only passes frequencies between the cut-off frequencies, and the high-pass filter only passes high frequencies after 1500 Hz. In addition, for each filter there is a maximum of two capacitors and inductors used. Another requirement for the high-pass filter of the tweeter was that this filter should be at least a second-order filter, and this requirement was satisfied as well, because the high-pass filter is a second-order filter.