EE1L1 IP-1 Intermediate Report

Loudspeaker filter design

Filter groups A4_2

TU Delft

EE1L1 IP-1

Intermediate Report

Loudspeaker filter design

by

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Contents

Nomenclature

Symbol	Description	Unit
R	Resistance	Ω
L	Inductance	H
C	Capacitance	F
I	Current	A
V	Voltage	V
P	Power	W
Z	Impedance	Ω
f_r	Resonance frequency	Hz
Q	Quality factor	(dimensionless)
Y	Admittance	S

Abbreviations	Description
AC	Alternating Current
DC	Direct Current
hpf	High Pass Filter
lpf	Low Pass Filter
ls	Loud speaker
Bw	Bandwidth

Constants	Value
π	3.14159 (used in sinusoidal calculations)
e	Euler's number, 2.718 (used in exponential equations)

Units	Meaning
μ	Micro (10^{-6})
n	Nano (10^{-9})
m	Milli (10^{-3})

1 Introduction

Loudspeakers are audio systems that are unavoidably found in today's day and age. You find them in phones, cars, shopping centers, etc. Taking in the fact that there are so many speakers around, it can be afterwards interpreted that audio systems in general are of high relevance in our society. The loudspeaker is a complex human craft with the ability to reproduce sounds from incoming sound signals. A loudspeaker creates variations in air pressure which is perceived as sound. This project is about utilizing passive filter design and loudspeaker analysis to comprehend this complexity, to afterwards put our understanding to practice, building a functioning loudspeaker. The project will be executed by nineteen TU Delft Electrical Engineering bachelor students. These students are split up into two groups, then they are split up again in ”component groups”. All component groups work on their own electrical speaker part. The electrical components of a loudspeaker consist of the power supply, the power amplifier, the low pass filter, the band pass filter and the high pass filter. Because the nineteen students are split up in two groups, there will eventually be one of each component. This report clarifies how the filter groups produce their components with a descriptive report about the process. Filter components are relevant for a loudspeaker due to the way loudspeakers create sounds. A loudspeaker can have more than one speaker, the loudspeaker built in this project contains three speakers, the woofer, the mid-toner and the tweeter. These speakers have different properties. Each speaker operates best at certain frequencies of sound. A speaker works well in a certain frequency range when the power consumption is relatively high compared to a reference point given by the creators of the acoustic speaker measurements. The purpose of the filters is to divide the incoming electrical signals, coming from the power amplifier which amplifies electrical signals, based on the signal's frequency, splitting them up in the desired range of the speakers. These signals are split up by being weakened by the filters when the frequency does not belong in the filter's frequency range. The woofer operates best in a low frequency range, the tweeter operates best in a high frequency range and the mid-toner operates best in the frequency range between the woofer and the tweeter. These derivations will be substantiated in chapter two. The filter groups are required to have mutual communication since the filter components are built based of each others properties. One property that dictates the design of these filters all together is the cutoff frequency. The cutoff frequency is the frequency at which the filter starts letting signals with a certain frequency through to excite the speaker, or stops signals with certain frequencies from exciting the speaker. The definition and the use of the cutoff frequency will be further elaborated in chapter two. We are provided with a limited variety of parts and a readily assembled speaker cabinet with the mechanical speaker components in place. The parts with which we may work consist of several inductors, capacitors and resistors with varying values. We can use the provided printed circuit boards to assemble a circuit together with a soldering tin and a soldering iron. We make use of the circuit simulation program called LTSpice. LTSpice is a computer software that can simulate electronic circuits to aid engineers in their circuit design. The relevant specifications, considering the filter groups, for the electrical components for the loudspeaker tell that the high pass filter and low pass filter must be of the second order and that the filters should be Butterworth filters. The specifications will be further discussed in chapter two. The report will divide our process of building the electrical speaker components into three parts. The analysis chapter, chapter one, elaborates on the work that precedes the design phase. Chapter three explains how running simulations is done and how to determine whether to build the simulated circuit or not. Chapter four distributes the speaker measurement result upon completion of the speaker, with all of the electrical components combined. Chapter five concludes this report. Organizationally, the project is a natural part of the Electrical Engineering degree. The goal is to exercise the teamwork skills of the bachelor students in order to prepare them for future careers.

2
Theory and Analysis (design methodology)

2.1. High pass filter

The high pass filter will be of the second order since that is one of the specifications given by the assignment. A second-order high pass filter consists of a capacitor in series with a parallel connected inductor and load impedance. This is demonstrated in figure 2.1

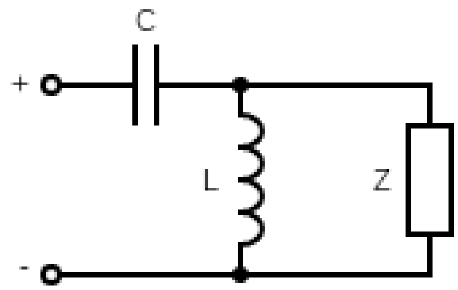

Figure 2.1: Second order HPF

As seen in figure 2.1, we must determine three values. The first value is the capacitance for capacitor C which from now on will be referred to as C_{hpf} with the subscript "hpf" referring to "high pass filter". The second value is the inductance of the inductor L which will be denoted as L_{hpf} for the remainder of this report. The third value is the tweeter impedance Z, which is currently continuing under the name $Z_{l}s$. Since $Z_{l}s$ is the impedance of the tweeter, the analysis starts at determining this impedance. The information given about the tweeter consists of the speaker impedance model, acoustic measurement data and impedance measurement data. Taking a look at the latter we find a graph displaying the absolute value of the impedance in Ohm on the vertical axis and the frequency in Hertz displayed on the horizontal axis. It shows a local maximum and an increase of impedance at the end of the graph. The speaker impedance model is shown in figure 2.2.

Speaker impedance

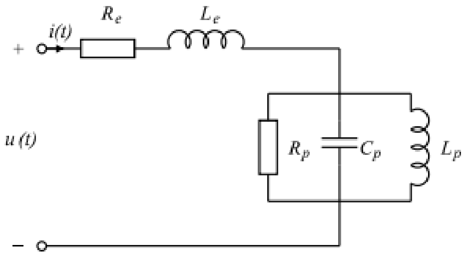

Figure 2.2: Tweeter impedance model

The model consists of five components without values. Using the speaker measurement data we can derive the following components:
R_e - The resistance of the voice coil (Ω)
Le - The self inductance of the voice coil (H)
Rp, C_p, L_p - The components that represent the electric equivalent of the mass-spring system of the tweeter in Ohms, Farad and Henry, respectively.

At low frequencies, we know that the impedance of inductors and capacitors are next to zero. The formula for the impedance of the inductor and capacitor are respectively:

$$Z_l = j\omega L \tag{2.1}$$

$$Z_c = \frac{1}{j\omega C} \tag{2.2}$$

- Z_l and Z_c representing the impedance of the inductor in Ohm (Ω)
- ω representing the angular frequency in rad/s
- L representing the inductance in Henry (H)
- C representing the capacitance in Farad (F)

At very low frequencies, the impedance is nearly equal to the resistance R_e because the value for ω here is almost zero. This leads to very low impedance values for the capacitor and the inductors, leaving the inductor to act as a short circuit, thus short circuiting the R_p. $R_e = 4.06\Omega$ From the definition of the resonance frequency, one can derive that the imaginary part of the tweeter impedance is equal to zero at the resonance frequency. At the resonant frequency, there is thus an impedance peak with a maximum of $R_e + R_p$. Subtracting R_e from the peak value results in a magnitude equal to the mass-spring system resistance. $R_p = 1.21\Omega$ At high frequencies, the impedance increases due to the self-inductance of the voice coil (L_e), the result is that the magnitude of the impedance of the tweeter is the sum of the impedance of the voice coil and the resistor R_e: $|Z_{ls}| = |R_e + j\omega L_e|$ at high frequencies ($f > 4000Hz$). We claim that all frequencies above $4000Hz$ are high because the graph of the speaker impedance magnitude starts increasing after the resonance drop at this frequency. At the highest possible frequency displayed by the graph, which corresponds to the value , the value of L_e is clear: $L_e = 53.3\mu H$

The resonance peak allows one to determine the remaining values by making use of the bandwidth . The definition of the bandwidth of an impedance is the difference in frequency where the magnitude of the impedance is equal to the factor of $\frac{1}{\sqrt{2}}$ times the magnitude value at the resonance frequency. Simply using this will not yield the correct magnitude due to the resistor R_e which elevates the entire impedance graph by its value. Utilizing a different formula:

$$Magnitude_c = |Z_{ls}| - \left(R_p - \frac{R_p}{\sqrt{2}}\right) \tag{2.3}$$

With $Magnitude_c$ representing the magnitude at which the corner frequencies are found. This magnitude ($|Z_l s| = 4.94328\Omega$) yields: $f_1 = 906.5Hz$ and $f_2 = 1796.8Hz$

$$B_w = f_2 - f_1 = 890.3Hz \tag{2.4}$$

The value for C_p can be determined by yet another formula:

$$C_p = \frac{1}{Bandwidth \cdot R_p} = \frac{1}{890.3 \cdot 1.21} \tag{2.5}$$

$$C_p = 147.6\mu F$$

For the a standard second order high pass filter the formula for the Resonance frequency is known:

$$\omega_0 = \sqrt{\frac{1}{L_p C_p}} \tag{2.6}$$

$$L_p = 94.4\mu H \tag{2.7}$$

Now all the speaker impedance values have been determined, allowing the initiation of the filter design. The speaker impedance:

$$Z_l = R_e + L_c + R_p||L_p||C_p \tag{2.8}$$

The Zobel network

The speaker measurement data has shown an increase of impedance for high frequencies due to the self-inductance of the voice coil. This occurrence may produce issues in later stages of design, since a high impedance that grows uncontrollably can affect the voltages and currents within the filter that we wish to regulate. A Zobel network is implemented to compensate for this growth. The Zobel network consists of a resistor and capacitor in series, connected to the speaker in parallel.

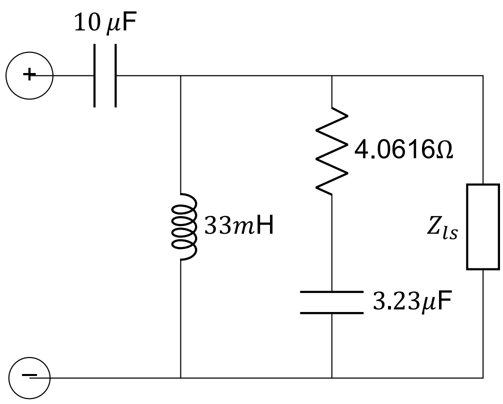

Figure 2.3: Second order high pass filter with Zobel network

Where R_z equals the resistance of the Zobel network and C_z refers to the capacitance of the Zobel network. These values are determined by their formulas: $R_z = R_e$; $C_z = \frac{L_e}{R_e^2}$, Therefore $C_z = 3.23mF$ and $R_z = 4.06\Omega$.

The cutoff frequency

Since the purpose of this filter is to weaken the electrical signals, coming from the power amplifier, which do not belong within the frequency range of the tweeter. It must be discussed what the cutoff frequency of the high pass filter should be. The cutoff frequency of the high pass filter is the frequency at which the gain is halved. The gain is the scale of the voltage over the speaker impedance (V_o) divided by the input voltage (V_i). On the decibel scale this is represented by a gain of $-6dB$.

$$\frac{V_o}{V_i} = \frac{1}{2} = 0.5 \tag{2.9}$$

Figure 2.4: Formula for the quality factor

$$Q = R_e \sqrt{\frac{C_{hpf}}{L_{hpf}}} \tag{2.13}$$

$$20 \log_{10} 0.5 = -6dB \tag{2.10}$$

To determine at what frequency the gain should be $-6dB$. It is necessary to take yet another look at the measurement data. This data also contains information about the acoustic measurements of the speakers. One of the graphs displays the power response in dB on the y-axis and the frequency in Hz on the x-axis. To determine the frequency range of the speaker using this graph it was agreed upon to consider all the frequencies within the range of the bandwidth to be the frequency range of the speakers. The bandwidth is determined by the cutoff frequencies of the tweeter's power response. For power, the cutoff frequencies are found at $-3dB$ below the reference point where the magnitude is zero.

$$\frac{P_2}{P_1} = \frac{1}{2} = 0.5 \tag{2.11}$$

$$10 \log_{10} 0.5 = -3dB \tag{2.12}$$

If this is how one determines the frequency ranges of the speakers, the graphs that displays the power response in dB on the y-axis and the frequency in Hz on the x-axis yields the ranges displayed in table 2.1.

Speaker	woofer	mid-toner	tweeter
Frequency range(Hz)	$65 < f < 125$	$86 < f < 1671$	$1468 < f < 9326$

Table 2.1: Frequency ranges of speaker types

The table shows areas where the frequency ranges of the speakers overlap. Within these overlapping ranges, a cutoff frequency is chosen. For the high pass filter a cutoff frequency of $1900Hz$ is chosen. Mind that the power response of the speakers is lower for both the mid-toner and the tweeter, causing the sound to be of lower volume if the gain remains to be one. This is why it is better to cross the transfer function for the band-pass filter and the high pass filter at a gain of a value slightly above $-6dB$. The value of the deviation of the magnitude has been guessed by the team to be around $+0.5dB$ with respect to $-6dB$. With the cutoff frequency for the high pass filter and the speaker impedance determined, the filter design is continued. Using LTSpice, the circuit is simulated and tested with values for C_{hpf} and L_{hpf}. During this testing phase, it is discovered that the value for C_{hpf} influences the position of the cutoff frequency and the value L_{hpf} should be adjusted to the capacitance accordingly: Q represents the quality factor which should be of the value $\frac{1}{\sqrt{2}}$ due to the Butterworth specification. This specification implies a maximum flat curve of the transfer functions which display the voltage gain. To find a capacitance that gives the right cutoff frequency. LTSpice can be used to test the capacitances available in the lab. After trial and error, a suitable capacitance is found for C_{hpf}. $C_{hpf} = 10\mu F$ This value grants a cutoff frequency at the desired $1900Hz$. According to figure 2.4 the value of the inductance is determined. $L_{hpf} = 33mH$.

2.2. Band pass filter

To design a functional mid-range speaker, also referred to as the mid-toner, it is necessary to construct a band-pass filter that eliminates frequencies outside the desired range. While there are numerous design approaches for creating such a filter, the impedance of the mid-toner will always play a critical role in these circuits. For this reason, the impedance model of the mid-toner holds significant importance. The data provided for the mid-toner includes the impedance model, acoustic measurement data, and impedance measurement data. The impedance model of the mid-range speaker is depicted in Figure 2.5.

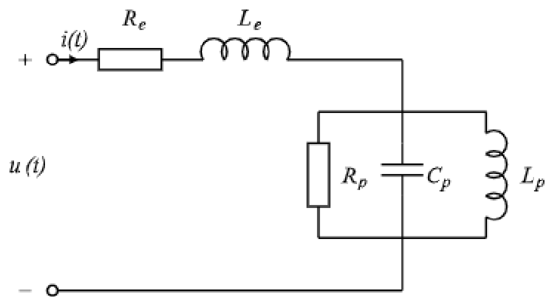

Figure 2.5: Mid-toner impedance model

- R_e representing DC resistance of the voice coil
- L_e representing the self inductance of the voice coil in Ohms (Ω)
- R_p, C_p, L_p representing electric equivalent of the mass-spring system in Ohms (Ω)

For the obtainment of the values for the components in Figure 2.5, the impedance measurement data must be utilized, specifically the Excel sheet containing the data of the graph displaying the absolute value of the impedance in Ohms on the vertical axis and the frequency in Hertz on the horizontal axis.

-To determine R_e, the impedance data plotted against frequency (Hz) in the Excel sheet will be examined. The value of R_e can be determined by analyzing the impedance at a very low frequency.

$$R_e = 4.14\Omega$$

-To determine R_p, the value of the impedance peak at the resonance frequency in the Excel sheet must be examined. Subtracting R_e from this value yields the resulting value, R_p.

$$R_p = 5.05\Omega$$

-To determine L_e, the impedance at very high frequencies must be examined. Subtracting R_e from this value gives the reactance of the inductor, which can then be used to calculate L_e.

The inductance L_e is calculated using the following formulas:

$$|Z_{ls}| = |R_e + j\omega L_e| = 22.49\Omega$$

$$L_e = \frac{\sqrt{Z_{ls}^2 - R_e^2}}{2\pi f} = \frac{\sqrt{22.49^2 - 4.14^2}}{2\pi \cdot 19999.7}$$

Where Z_{ls} represents the total impedance, R_e is the DC resistance of the voice coil, ω is the angular frequency given by $\omega = 2\pi f$, L_e is the inductance of the voice coil, j is the imaginary unit, and f is the frequency at which the impedance is measured.

$$L_e = 0.1759\,\text{mH}$$

-To determine C_p, we first need to obtain the bandwidth (B_w). This is done by identifying the frequencies at which the impedance is equal to $Z_{ls} = R_p + \left(R_e \cdot \frac{1}{\sqrt{2}}\right)$. By subtracting the lowest frequency from the highest and multiplying the result by 2π, we can calculate B_w in Hz. Then, by rearranging the bandwidth equation, we can derive C_p.

$$Z_{ls} = 4.14 + \left(5.05 \cdot \frac{1}{\sqrt{2}}\right) = 7.71\,\Omega$$

$$B_f = 2\pi(f_2 - f_1) = 2\pi(105.3 - 87.3)$$

$$B_f 2\pi = \frac{1}{R_p \cdot C_p}$$

$$C_p = \frac{1}{B_f \cdot 2\pi \cdot R_p} = \frac{1}{18 \cdot 2\pi \cdot 5.05}$$

$$C_p = 1.751\,\text{mF}$$

-To determine L_p, we first need to identify the frequency at which the impedance peak is maximum. Then, using the resonance frequency equation, we can rearrange the variables to solve for L_p, the inductance.

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With the mid-toner impedance model validated by the simulation results shown in Figure **??**, the design phase of the band-pass filter can now commence. There are numerous approaches to designing such a circuit; however, the chosen design for this project combines a second-order high-pass filter and a second-order low-pass filter, as illustrated in Figure **??**.

The initial step in the design process is to determine the frequency range for the band-pass filter. This was achieved by analyzing the acoustic amplitude and phase response of the woofer, mid-toner, and tweeter. Specifically, the frequency range was identified where each speaker's power response lies between its peak value and 3 dB below the peak. Additionally, the phase alignment was evaluated to ensure smooth transitions within these frequency ranges. Based on this analysis, the band-pass filter was designed to operate within the 220 Hz to 1500 Hz range.

Determination of C_{Hp} and L_{Hp}:

To calculate the values for the high-pass filter components C_{Hp} and L_{Hp}, the following equations were used:

$$\omega_o = \frac{1}{\sqrt{C_{Hp} \cdot L_{Hp}}}, \quad Q = R_{\text{load}} \cdot \sqrt{\frac{C_{Hp}}{L_{Hp}}}.$$

The cutoff frequency (ω_o) was converted from 1500 Hz to radians per second. For R_{load}, the impedance of the speaker within the chosen frequency range was approximated as $4.14\,\Omega$. A quality factor (Q) of $\frac{1}{\sqrt{2}}$ was selected to ensure a smooth transition between filters. This value was chosen based on guidance from the I.P manual.

By solving the first equation for L_{Hp} and substituting it into the second equation, the calculated values are:

$$C_{Hp} = 124\,\mu\text{F}, \quad L_{Hp} = 4.22\,\text{mH}.$$

Determination of C_{Lp} and L_{Lp}:

The process for determining the values for the low-pass filter components C_{Lp} and L_{Lp} followed the same methodology as for the high-pass filter. However, the cutoff frequency was set to 220 Hz. The calculated values are:

$$C_{Lp} = 18.1\,\mu\text{F}, \quad L_{Lp} = 0.62\,\text{mH}.$$

see figure 2.6 for the ideal circuit, with the calculated values.

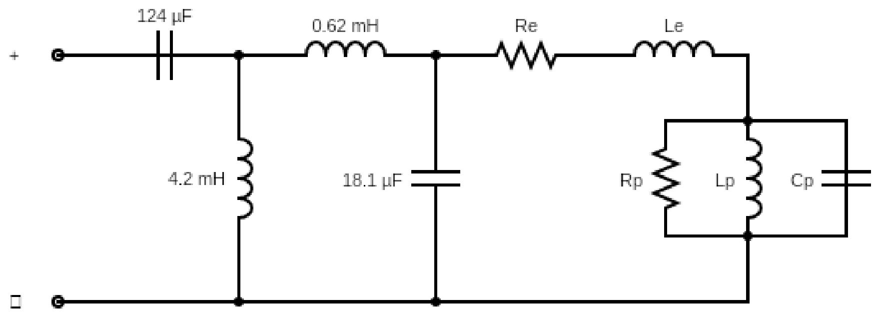

Figure 2.6: Ideal circuit for the mid-toner impedance model, showing the band-pass filter design with theoretical values.

Adjustments for Practical Component Availability:
While the calculated values represent the ideal components for the chosen frequency range, practical limitations dictated by available components required adjustments. After iterative testing and optimization, a resistor was added in series with the circuit to attenuate the magnitude as needed, and another resistor was introduced in series with the capacitor of the low-pass filter to shift the cutoff frequency appropriately. The final component values selected were:

$$C_{Hp} = 150\,\mu\text{F}, \quad L_{Hp} = 2.2\,\text{mH}, \quad C_{Lp} = 22\,\mu\text{F}, \quad L_{Lp} = 0.68\,\text{mH}, \quad R_1 = 1\,\Omega, \quad R_2 = 2.2\,\Omega.$$

These adjusted values achieve a frequency response close to the desired range while accounting for real-world constraints. see figure 2.7 for the applicable circuit.

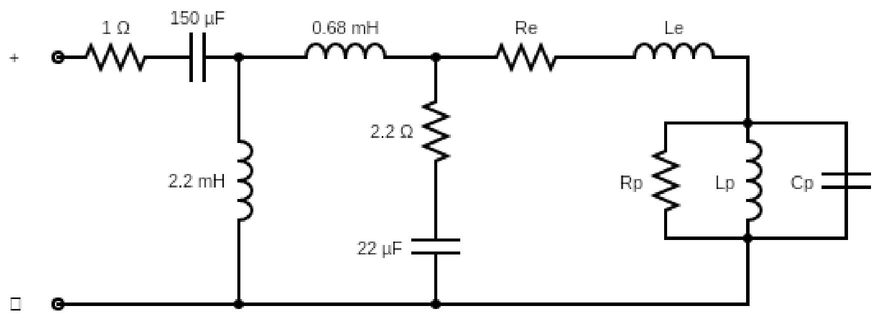

Figure 2.7: Ideal circuit for the mid-toner impedance model, showing the band-pass filter design with applicable values.

2.3. Low pass filter

To make a sub-woofer/ low frequency speaker work, we need a working low pass filter. Since there are multiple ways to make said low pass filter, we have to check what filter design is realistic for us to obtain our desired cutoff frequency. This cutoff occurs when the the maximum value of the gain (at resonant frequency) drops 3 decibel. To find the best design that meet our requirements we use ltspice to simulate the filter, but for that to be accurate we also need to model the sub-woofer, with the needed impedance and acoustic measurement data. (note: all the measurement data can be found in appendix B)

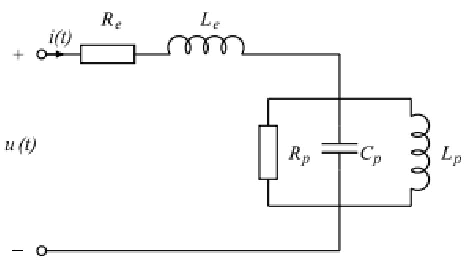

Figure 2.8: sub-woofer impedance model

With the given model (fig 2.8) and the measurement data that was given to us, we can now obtain the values of R_e,R_p,C_p,L_p and L_e. To obtain the value of R_e, we analyze the impedance of the speaker at very low frequency. When done we've gotten a value of

$$R_e = 4.0542\Omega$$

When obtaining the value of R_p, we should look at the impedance of the speaker at the resonant frequency. Since the impedance at that frequency is mostly equal to R_e and R_p we can obtain a value for R_p. When done we've got

$$R_p = 8.8998\Omega$$

Since we have got the R_e we can calculate L_e with the following formula

$$|Z_{ls}| = |R_e + j\omega L_e|$$

This formula applies only when at very high frequency, since we know ω (where $\omega = 2\pi * f$) and can choose a very high frequency and can read the impedance at that frequency, we can calculate L_e. the value we get from using the formula and a bit of algebra is:

$$L_e = 0.1816mH$$

To calculate C_p and L_p we need to obtain the bandwidth of the speaker. We do this in the following manor: at first we need to calculate the impedance value for the bandwidth with

$$R_B = \frac{R_p}{\sqrt{2}}$$

Subtract that value from the resonant frequency and then read at that impedance level the frequencies that occur. When done you can subtract the 2 frequencies of each other and you get B_w. The calculations that we got for R_B and B_w are:

$$\frac{8.8998}{\sqrt{2}} = 6.293\Omega$$

$$R_B = 12.954 - 6.293 = 6.578\Omega$$

$$B_w = 90.3 - 56.6 = 33.7Hz$$

Now to calculate C_p and L_p we use the following formula:

$$B_w = \frac{1}{R_p \cdot C_p} \quad \omega_0 = \sqrt{\frac{1}{L_p \cdot C_p}}$$

Where ω_0 is equal to the resonant frequency. With a bit of algebra we can get the value of C_p and with that we can also get L_p. The value's that we have gotten with the calculations are:

$$C_p = 3.334mF \quad L_p = 1.43749mH$$

With all these values we can finally simulate the speaker model with the right values, which we use to design the filter to our desire.
At first we wanted to make a first-order low-pass filter because of its simplicity and easier calculation. The only drawback was that the power transfer was less ideal leading to a lower gain. That is why we chose a second order circuit, though the calculations are a bit tougher, the gain we can get from it is far higher than that we can get from a first order circuit. With some calculations, using some values of the first-order circuit and a bit of trial and error, we have chosen to use a total inductor value of $2.88mH$ and a total capacitor value of $200\mu F$. To get these values, we put a $2.20mH$ and a $0.68mH$ in series and two $100\mu F$ capacitors in parallel. Just as shown in figure 2.9.

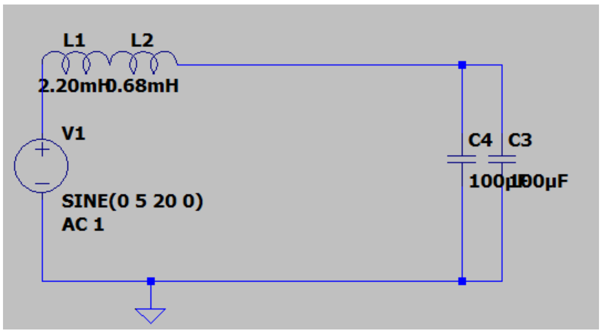

Figure 2.9: low pass filter simulation model

With this filter model, we can now put the filter and speaker model together in the simulation and then get the gain and phase over the whole circuit, which we will compare against the measurements done when the filter is built and if needed change the values of the inductors and/or capacitors.

3

Simulation results

3.1. High pass filter

LTSpice provides the graph of the transfer function. LTSpice graphs both the magnitude and phase plots of the transfer function. In the frequency domain of the tweeter, phase plots are not as useful as magnitude plots since phase differences at high frequencies are not perceivable by the human ear. The transfer function is displayed in 3.1 and has a desired cutoff frequency at $1912Hz$.

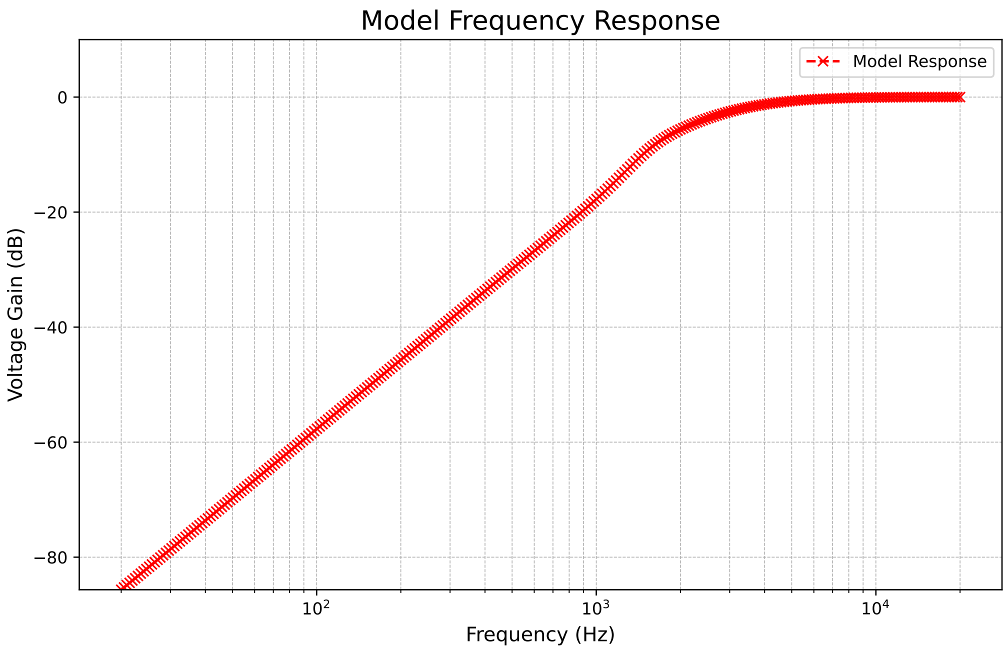

Figure 3.1: Simulated high pass filter transfer function

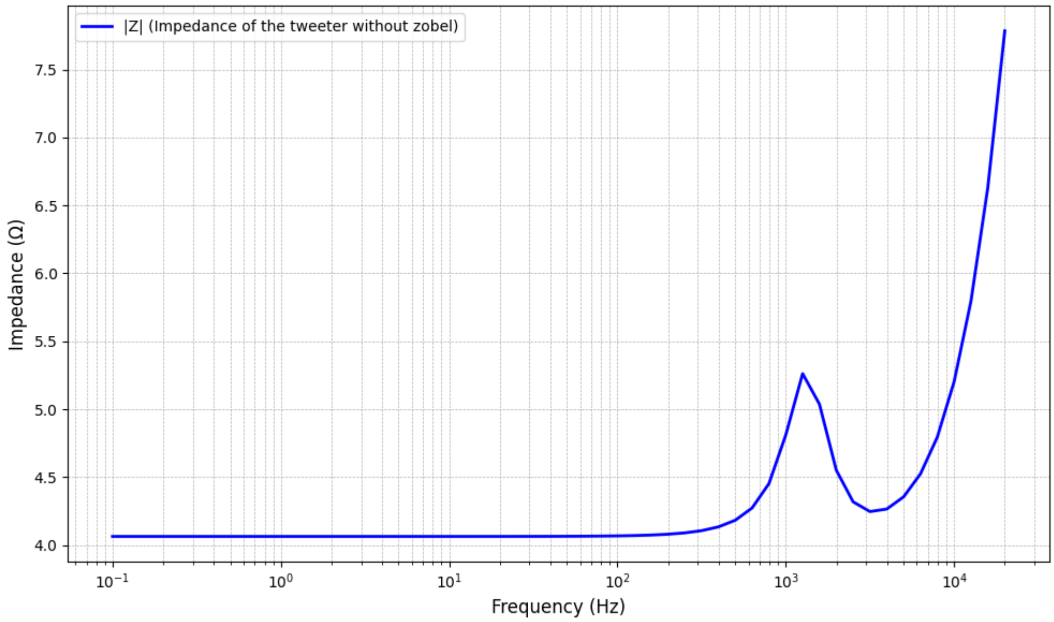

Figure 3.2: LTSpice simulation of the impedance of the tweeter without the Zobel network.

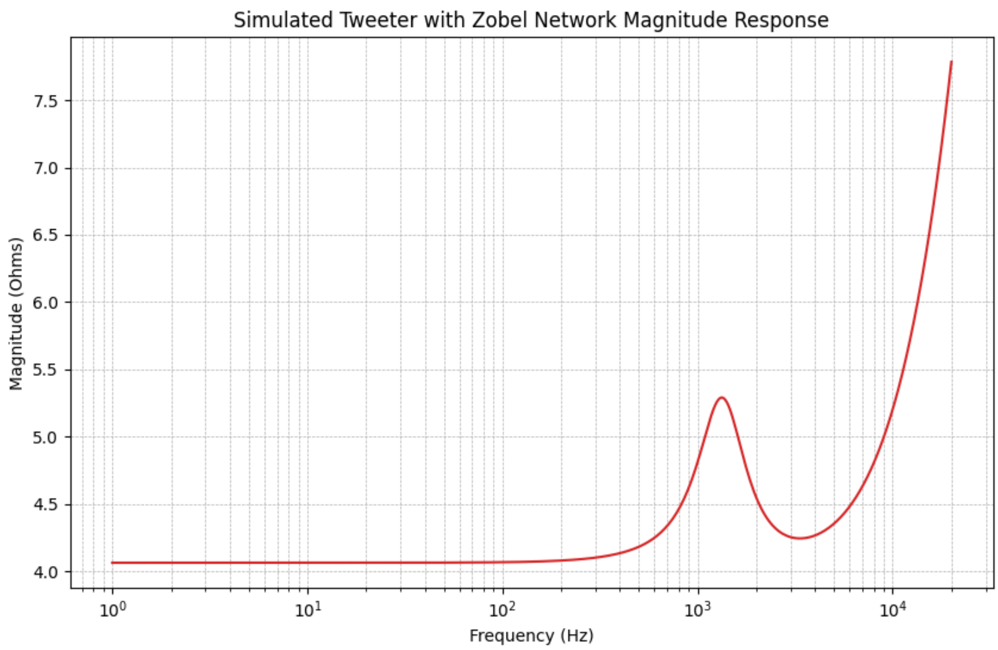

Figure 3.3: LTSpice simulation of the impedance model of the tweeter with the Zobel network.

Looking at 3.2 and 3.3 one can conclude that the Zobel network does not deliver the expected results since the graph is not flat after the resonance peak., yet it remains a harmless addition to the circuit since the resonance peak is now smoother than before.

3.2. Band pass filter

By simulating the calculated speaker model in LTSpice (as shown in Figure 3.4), we obtained the impedance and phase response depicted in Figure 3.5. This simulation was based on the calculated component values for the speaker, and the resulting plot shows the impedance variation with frequency.

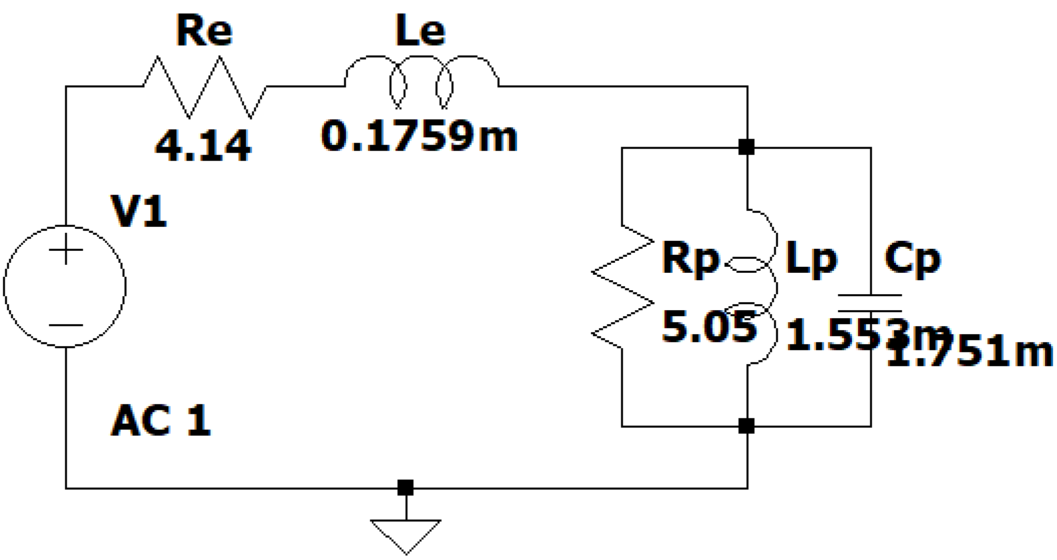

Figure 3.4: Midrange speaker model constructed in LTSpice, with its calculated values.

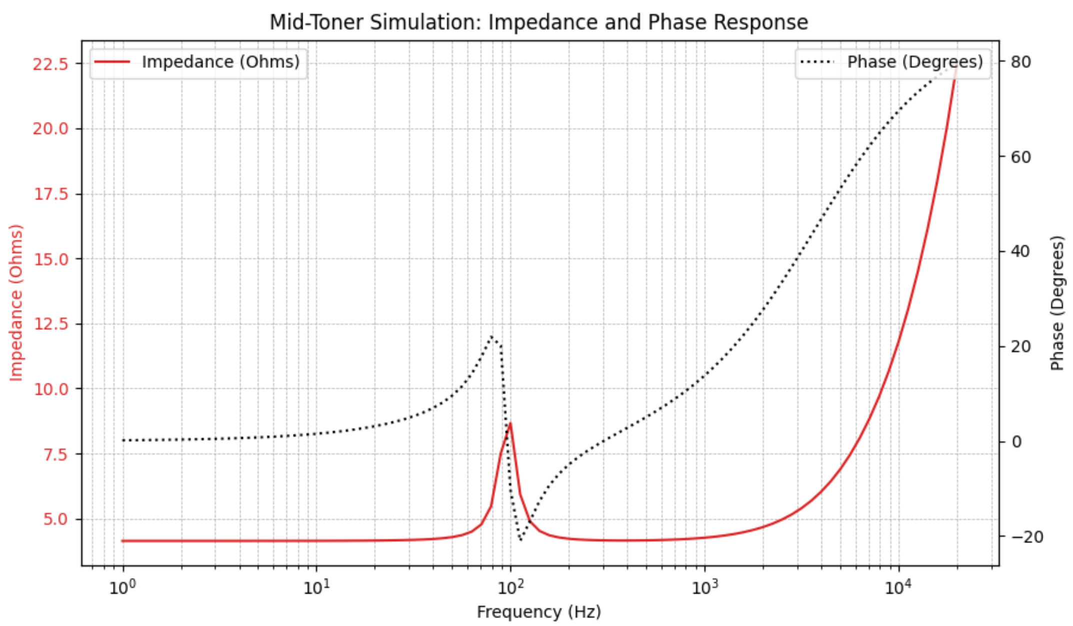

Figure 3.5: Impedance and phase response of the mid-range speaker obtained from the LTSpice simulation. The plot shows both the impedance and phase variation with frequency.

Next, by simulating the bandpass filter we designed (referenced in Figure 3.6), we obtained the magnitude and phase response shown in Figure 3.7. The results from the simulation illustrate the behavior of the filter with respect to both the magnitude and phase responses across a range of frequencies.

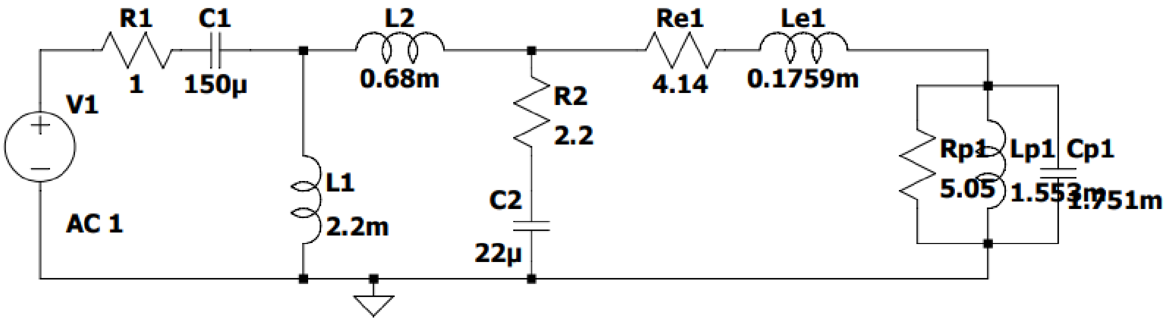

Figure 3.6: Bandpass filter circuit designed in LTSpice, showing the component values and configuration used for simulation.

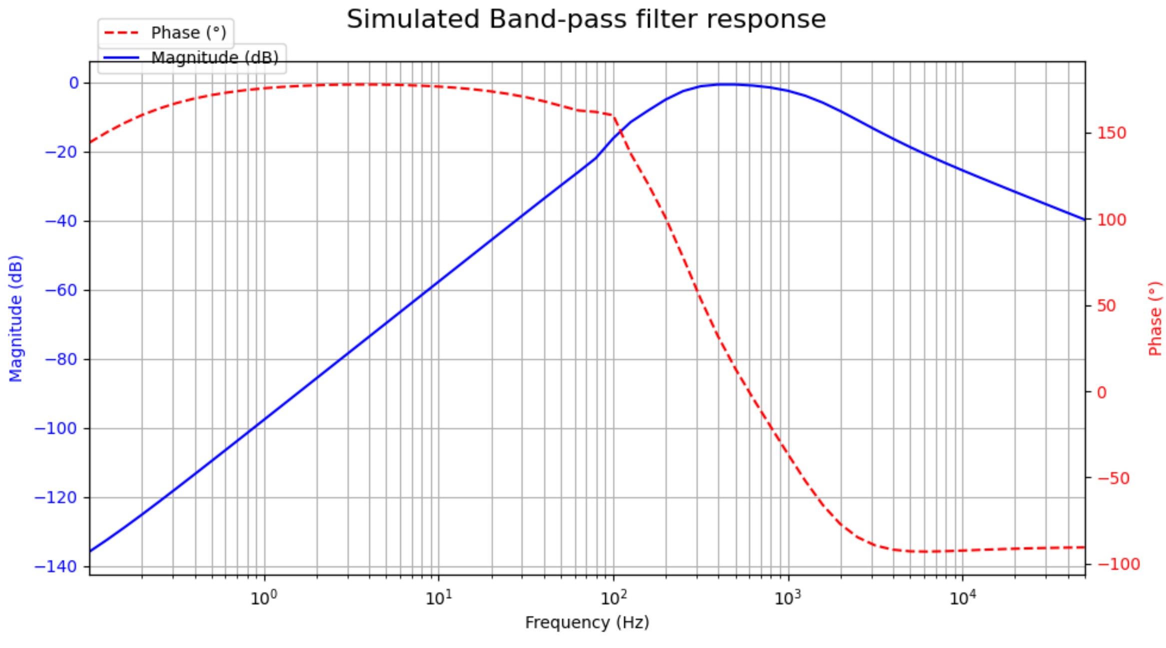

Figure 3.7: Magnitude and phase response of the bandpass filter obtained from the LTSpice simulation. The plot illustrates how the filter performs across different frequencies.

The simulation results for the impedance and phase response of the mid-toner speaker (Figures 3.5 aligned well with the theoretical predictions. However, some discrepancies at higher frequencies, due to non-ideal component behavior, which were not accounted for in the theorie

For the band-pass filter design, theoretical component values were calculated, but practical limitations required adjustments due to component availability. These adjustments influenced the cutoff frequencies. Initially, we aimed for a range between 220 Hz and 1500 Hz, but the simulated cut-off frequencies shifted to 180 Hz and 1600 Hz. The final component values ($C_{Hp} = 150\,\mu$F, $L_{Hp} = 2.2$ mH, etc.) were selected to closely match the desired frequency response, reflecting the need for iterative testing and optimization in practical circuit design.

3.3. Low pass filter

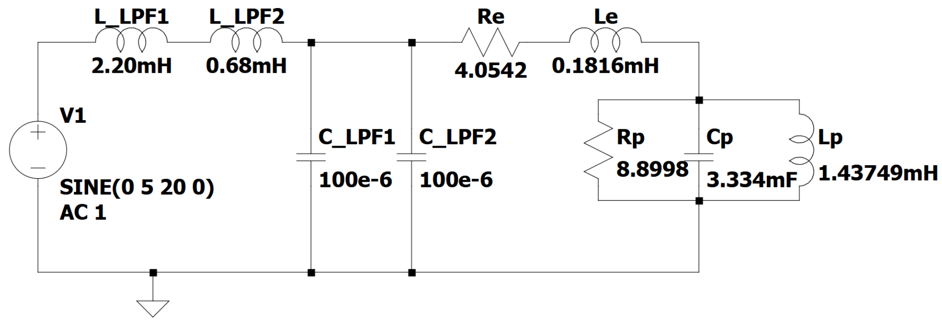

Figure 3.8: Circuit used in the LTSpice simulation

The model of the speaker and filter design have been simulated in LTspice. The schematic of this simulation can be found in figure 3.8. The results of this simulation are shown in figure 3.9. The magnitude is shown in blue, while the dotted red line represents the phase difference. The cutoff frequency is displayed using a dotted blue line, in our simulation this was at 150,86Hz.

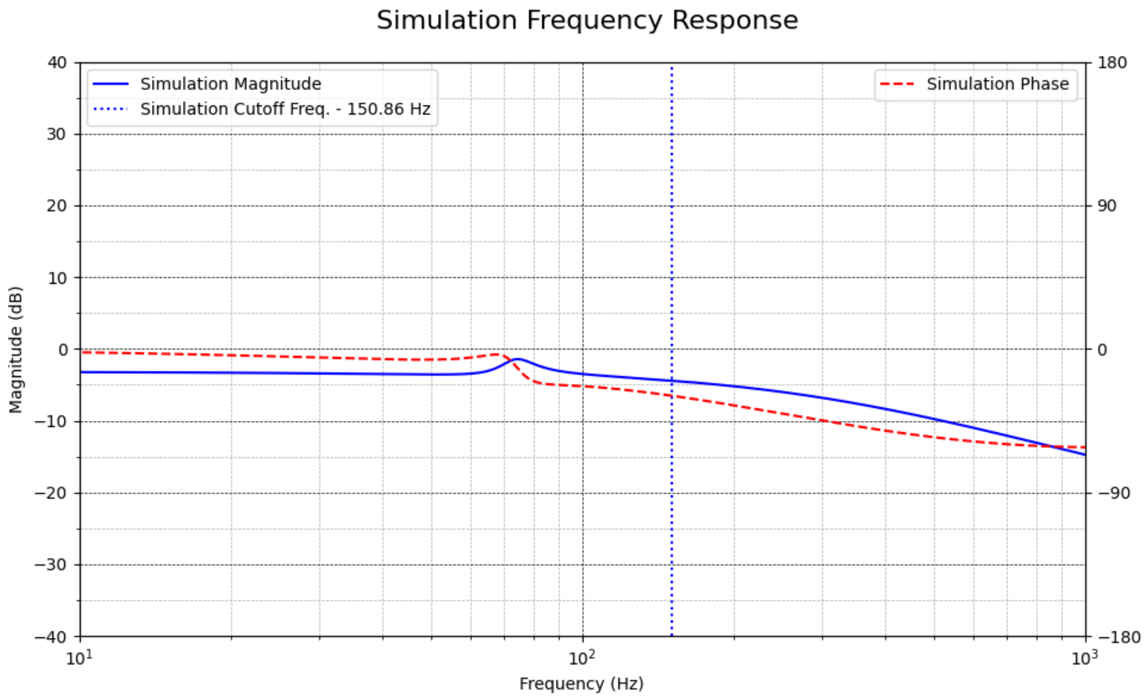

Figure 3.9: Frequency response of the LTspice simulation using internal resistances for the inductors and capacitors

The simulation in figure 3.9 also accounts for the internal resistances of the inductors $L_{LPF1,2}$ and the capacitors $C_{LPF1,2}$. While this would normally lead to a better simulation, this is currently not really the case, The internal resistances of the components of the speaker are unknown, therefore, our simulation cannot account for all the resistances and produces faulty data. Figure 3.10 shows the simulation run without accounting for these internal resistances.

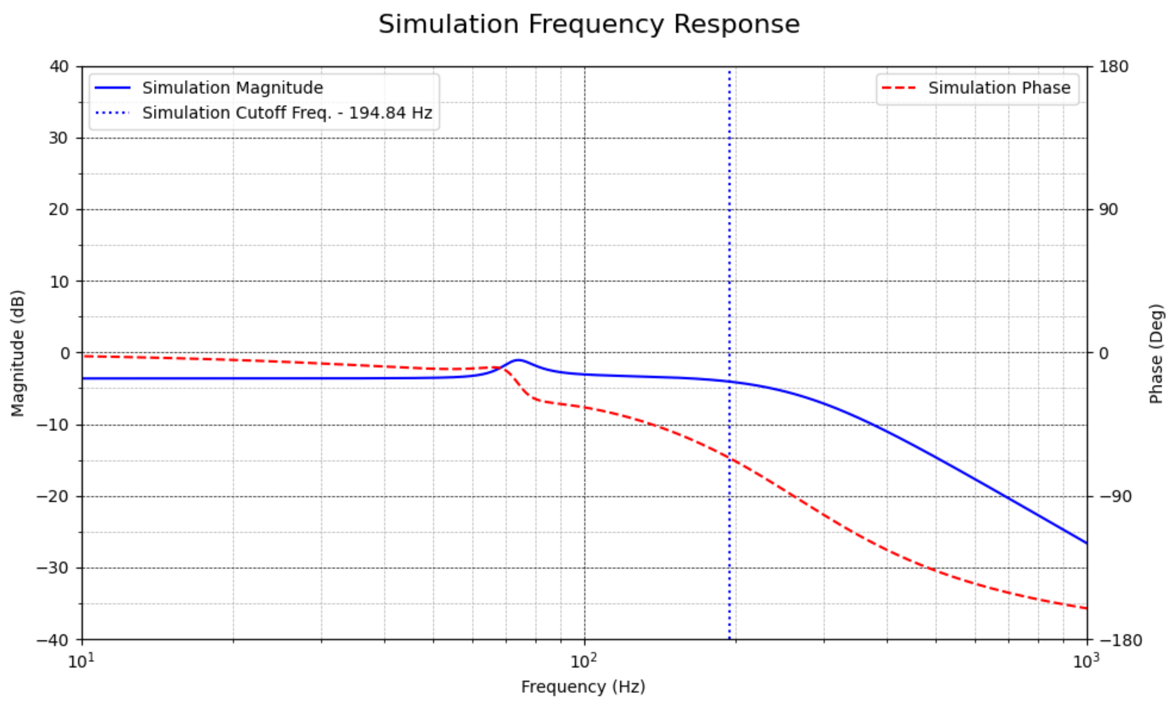

Figure 3.10: Frequency response of the LTspice simulation without internal resistances

A notable difference between figure 3.9 and 3.10 is the cutoff frequency. Without simulating the internal resistances, the cutoff frequency aligns better with our desired frequency of 200Hz. Later, we will learn that figure 3.10 more closely simulates the real filter.

4

Measurement results

4.1. High pass filter

Having built the circuit, the actual transfer function is derived. This transfer function shows that the actual transfer function closely resembles the modeled circuit's transfer function. The deviation lies outside of the tweeter's frequency range, rendering this deviation harmless to the final product. It is therefore in vain to attempt to correct this deviation. Since the high pass filter was built from parts available in the lab, the values in the hpf circuit do not differ from the values used in the simulation.

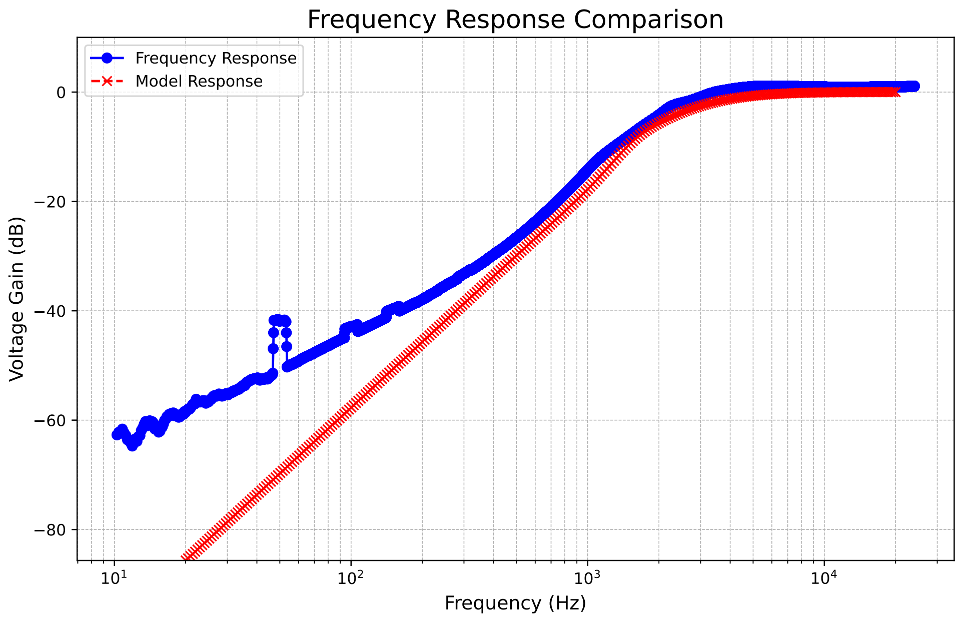

Figure 4.1: The modeled transfer function in comparison with the actual transfer function

4.2. Band pass filter

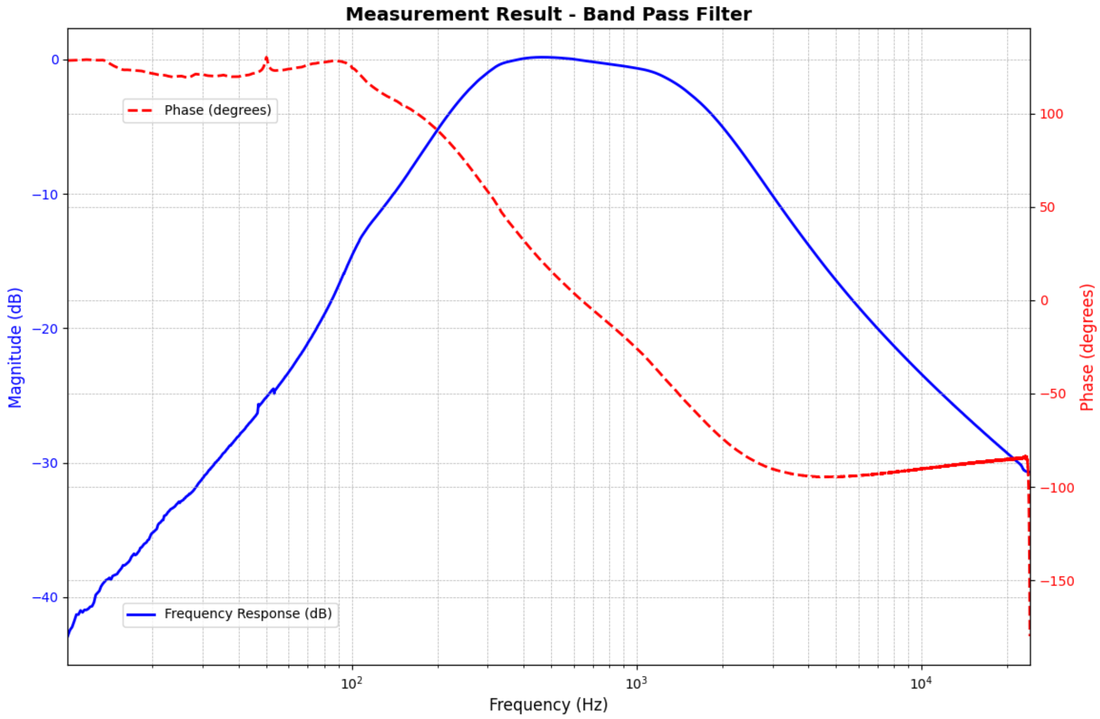

Figure 4.2: **Measured Phase and Magnitude Response of Practical Band-Pass Filter.** This plot shows the phase and magnitude response of the mid-range speaker after constructing the practical circuit. The results reflect the actual performance of the filter, including the effects of component tolerances, measurement conditions, and any adjustments made during the design and construction process. The plot provides insight into how the circuit's frequency response aligns with the expected performance, highlighting any deviations due to practical factors such as parasitic inductances, capacitances, and component imperfections.

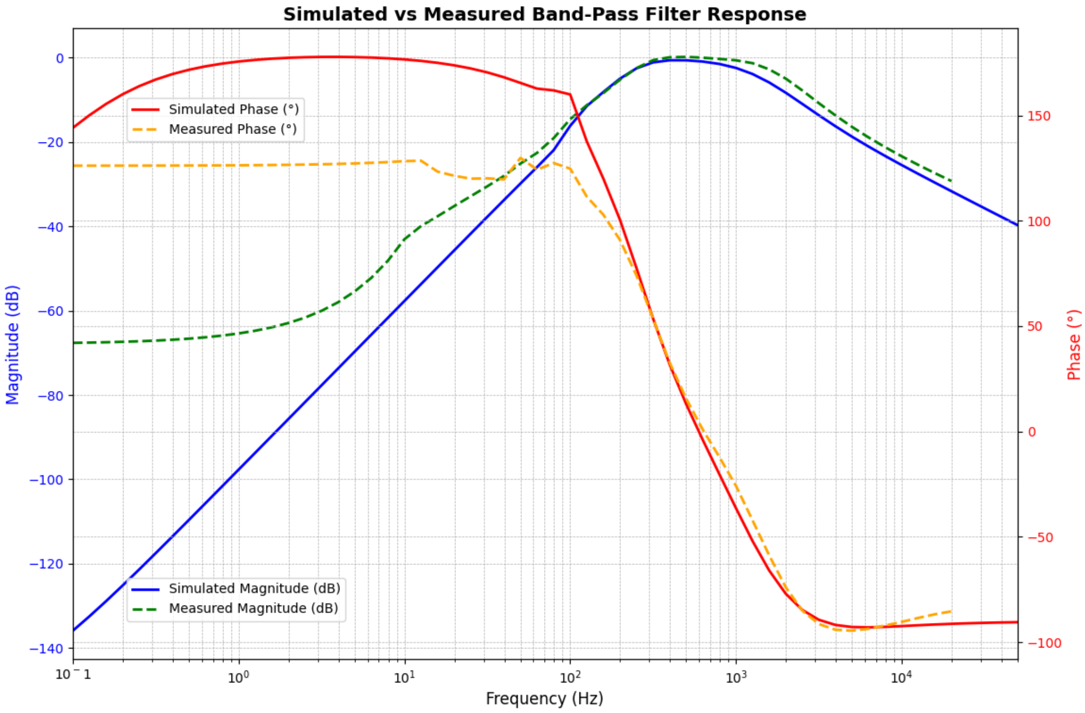

Figure 4.3: **Comparison of Measured vs. Simulated Phase and Magnitude Responses.** This plot compares the measured and simulated phase and magnitude responses of the mid-range speaker filter. Both the measured and simulated data are based on practical components used in the design. The plot highlights the similarities and differences between the two sets of results within the 220 and 1500 frequency range.

By comparing both the measured and simulated magnitude and phase responses, as shown in Figure 4.3 , we can draw several conclusions regarding the performance of the practical band-pass filter. Both the measured and simulated results are based on practical components, with simulations being conducted using the same component values used in the actual circuit. The comparison allows us to assess how well the practical filter matches the simulated response, and provides insight into the accuracy of the simulations in predicting real-world performance.

The observed differences between the measured and simulated responses can be attributed to:

- **Components used:** The actual values of resistors, capacitors, and inductors deviate slightly from their nominal values, which impacts the overall frequency response.

Despite these discrepancies, the measured response generally aligns with the simulated results especially in the frequency ranges chosen, confirming that the practical filter design performs close to the expected response. The minor differences observed provide opportunities for further refinement, particularly in minimizing the impact of component tolerances in future designs.

4.3. Low pass filter

The low pass filter is build using two *Audyn ERA/100/100* 100 capacitors and two inductors, of the types *LU32/068/071* and *HQR32/2.2/60*.

Figure 4.4 shows the frequency response of the low pass filter, measured in a test setup. In the setup, the filter was connected to a power amplifier and an subwoofer. The power amplifier played a range of frequencies, while the magnitude and phase of the output to the subwoofer was measured.

After testing for a first time, we made a small revision, since the inductors were close together, there was a mutual inductance between the two inductors. A rebuild and second measurement gave the desired results.

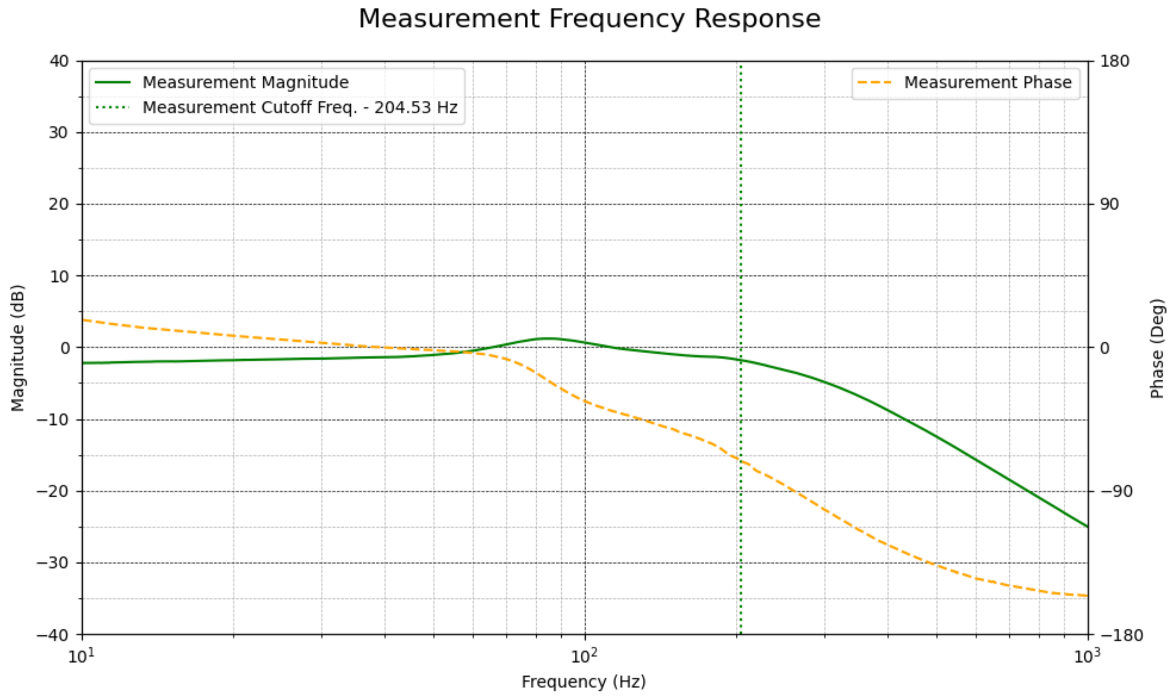

Figure 4.4: Frequency response of the measurements of the low filter

For a better comparison between the simulations and the measurements, the graphs of figures 3.10 and 4.4 are combined in figure 4.5.

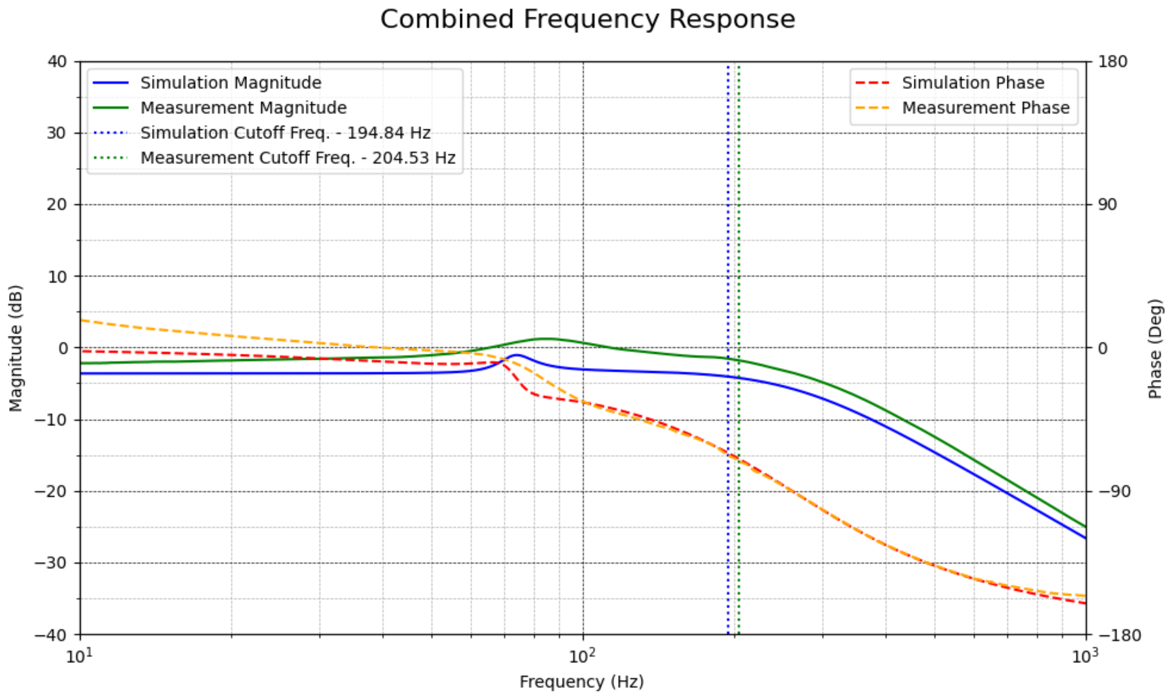

Figure 4.5: Frequency response of the simulation and measurements of the low filter

Shown by figure 4.5 there is around a 2dB magnitude difference over the whole range between the magnitude of the simulation and the magnitude of the measurement. There is an exception around the resonance frequency, here the peak of the measurement data is less steep and the resonance frequency has moved up by about 10Hz.

The difference in phase of the simulation and measurement after 100Hz is insignificantly small. under 100Hz, there is a bigger difference, primarily around the resonance peak.

Using a tone generator connected to the input and an oscilloscope to the output of the filter, it is possible to measure the gain at specific frequencies and compare these to the expected results. There is a 1.009% gain of the input signal measured at 20Hz. This transforms into a magnitude of 0.082dB. This is nearly identical to the simulated magnitude of 0.0804dB. This difference is also negligible at other tested frequencies. Therefore we believe that our filter closely resembles the simulation and any differences found in figure 4.5 are primarily the result of inconsistencies in the subwoofer.

5
Conclusions

5.1. High pass filter

The high pass filter simulations have shown that the Zobel network does not work as intended. It instead smooths out the tweeter impedance curve which is a harmless addition. The high pass filter imitates the behavior of the simulated model and is therefore successfully built.

5.2. Band pass filter

From figure 4.3, it is clear, that -application reaches the
-limitations -future comparison between the filters

5.3. Low pass filter

The filter is build using a second order circuit. After simulating, building and measuring, it performs as a low pass filter with a cutoff frequency of 204.54Hz. The filter closely mimics the simulated circuit. After measuring the filter without a load attached, we came to the conclusion that any discrepancies found between the simulation and the measurement are the result of inconsistencies in the subwoofer.

6
Bibliography