EE1L1 IP-1 Intermediate Report

”Booming Bass”

A4 - Amplifier 2

TU Delft

EE1L1 IP-1

Intermediate Report

"Booming Bass"

by

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Electrical Engineering, Integrated Project 1
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Contents

Nomenclature

Symbols

f Frequency, in Hertz [Hz]

ω Angular frequency, in radians per second [rad/s]

ω_h Angular frequency of the high-pass filter, filter 1

ω_l Angular frequency of the low-pass filter, filter 2

ω_p Angular frequency of the filter in the amplifier circuit

V_M Magnitude of a sinusoidal voltage signal, in Volt [V]

Circuit Elements

C Capacitance/Capacitor, in Farad [F]

C_1 Capacitor 1, element of filter 1

C_a Capacitor a, element of filter 2

C_x Capacitor x, element of amplifier circuit

L Inductance/Inductor, in Henry [H]

R Resistance/Resistor, in Ohm [Ω]

R_1 Resistance 1, element in the circuit before the two filters

R_2 Resistance 2, element of filter 1

R_a Resistance a, element of filter 2

R_b Resistance b, element of the filter in the amplifier circuit

R_c Resistance c, element of the filter in the amplifier circuit

Circuit Nodes

V_{in} Voltage input, the input signal of the circuit

V_{out} Voltage output, the output signal of the circuit

V_A Node A, the node between filter 1 and filter 2, and its voltage

V_B Node B, the node between filter 2 and the positive input of the amplifier, and its voltage

V_C Node A, the node between R_b, R_c and the positive input of the amplifier, and its voltage

1 Introduction

Sound systems are found in a lot of modern technological devices. They are an indispensable part of a lot of electrical devices, such as televisions, cars, mobile phones, and laptops, to name a few. Even standalone audio systems are a dime a dozen. It is difficult to imagine current technology without any sound systems. The main component of audio systems are the speakers. Speakers convert electrical signals into acoustic signals by creating small variations in air pressure. These differences in air pressure are perceived as sound waves, and with the right frequency, these waves become audible to us humans as sound. Summarizing, the speakers are the components that allow the audio systems to actually produce audible sound.

During the IP1 project, we will build a speaker. The aim of this project is to have us learn more about filters and to apply the theories that we learn in the Linear Circuits B course in a practical case. Another important goal is to learn more about working together in a group with shared goals and a shared product. Our intended final product is a working speaker with a decent sound output. The quality of the speaker will be determined by the quality of the output sound, and this will be tested by having it play different sound frequencies and music.

Speakers consist of different components: a power supply, a power amplifier, and multiple filter circuits that control the drivers. Our group has been divided into multiple subgroups, each working on a different component. Our task is to work on the power amplifier for our speaker, and as such, that component is what we will be discussing in this intermediate report. We are tasked with designing and building a working amplifier that can sustain the speaker. The main purposes of the amplifier circuit is to amplify the signal. The amplifier circuit starts with an input signal, an electrical sound waveform that is being sent from an electrical device connected to the input of the speaker. This input signal should then be filtered to isolate all relevant frequencies, while minimizing background noise and unwanted fragments within the input signal. This is an important step, since we want the signal sent to the filter sections of the speaker to be clear, and we want to reduce power consumption of the speaker. Secondly, the amplifier that we are using, a Texas Instruments LM3886 amplifier, can generate more noise when amplifying high frequency signals [2]. By filtering unwanted parts of the input signal before amplifying the signal, the power that is required to amplify the signal is also reduced and the power efficiency of the speaker increases [4]. After filtering the input signal, the remaining signal will be amplified and then sent to the filters. The input signal at the start of the circuit has a very low voltage of a maximum 0.4 V. The volume of the sound output is dependent on the voltage of the signal, and therefore it is essential to amplify the signal in order to get a respectable voltage to be able to output a decent audio volume. Each of these mentioned steps within the circuit come with certain restrictions and constraints, which will be discussed in more detail in the respective parts of the report.

In this report, we present our process as we work on the project. First, we will discuss the specifications, analyze our circuit and to set up equations defining it, and then fully design our circuit by calculating all relevant values for each element of the circuit using the equations obtained from the analysis. After we have finished designing our circuit, we will simulate it using a program called LT-Spice. We will critically compare those simulations to our expectations and specifications. If those simulations do not meet our expectations, we can make improvements to our design based on the issues found. When we are satisfied with our simulations and have a complete design, we can start building our circuit using the components available to us. Then, we will test the built circuit and compare it to the related simulations and specifications. If necessary, we can still make adjustments to the final design if the measurements during those tests do not satisfy the specifications. Finally, we will present our final design once we are satisfied with all measurement data we have obtained and all specifications have been met. All of these design processes will be discussed in detail in their associated subsections within the final report, and this intermediate report will discuss everything that we have completed so far as of time of writing.

2

Theory and Analysis (Design Methodology)

2.1. Specifications

The focus of our subgroup is to build a working amplifier that amplifies the input audio signal and filters unwanted input frequencies. In the IP1 project manual, figure 7.6 [1], we were given a circuit that we had to analyze and design by deriving appropriate values for each element. As can be found in chapter 2.6.4 of the manual [1], our design had to satisfy certain predetermined specifications:

1. The amplifier circuit has a non-inverting configuration.
2. The passband of the amplifier is 20 Hz - 40 kHz (-3 dB bandwidth). This means that the signal components with frequencies f < 20 Hz and f > 40 kHz will be suppressed, relative to the signal components with frequencies of 20 Hz < f < 40 kHz.
3. The voltage gain in the passband equals 25.
4. DC on the input signal should be blocked and may not be present at the output.
5. The DC offset of the opamp may be maximally amplified by 1.

Some of these specifications are intrinsically solved by the given design and do not need further attention in our calculations: specification 1 is given by the orientation of the opamp, and specification 4 is solved by capacitor C_1, as DC input voltages are absorbed by that capacitor and not present thereafter. Specifications 2 and 3 are our main targets and will be the key goals of our calculations.

In order to reduce the passband of the opamp to 20 Hz - 40 kHz as desired by specification 2, our design has two filters at the very front of the design: a high-pass filter consisting of C_1 and R_2, and a low-pass filter consisting of R_a and C_a. Together, these two filters function as a passband filter, but analysis of these two filters can be separated to make the calculations simpler. Specification 3 demands that the voltage gain is (approximately) 25 over the entire circuit. A larger gain means the maximum output volume of the speakers is larger, as the voltage of the output becomes larger. The input voltage at the start of our circuit is 0.4 V. The power supply of the amplifier is approximately 18 V. If we take an amplification/gain of 25, the maximum output voltage will be approximately 10 V. We consider it acceptable if the output voltage is slightly larger than 10 V, and thus the gain is slightly more than 25. It is preferable that the voltage is slightly higher than 10 V than slightly lower, since the maximum voltage controls the maximum output volume, and in case the output volume is too loud, it can be controlled by adjusting the voltage of the input signal by changing the voltage of the signal from the electrical device from which we play the audio. The gain of the opamp is controlled by resistors R_b and R_c, and partly also by capacitor C_x, which combined with R_b functions as a high-pass filter and ensures that low frequency signals are amplified less.

In the end, once the entire speaker is finished, we want the plot of the volume of the output sound to be as flat as possible within the spectrum of frequencies that humans can hear, so between 20 Hz and 20 kHz. We consider it acceptable if the low frequencies near 20 Hz and the high frequencies near 20 kHz have a slightly lower output volume, as these frequencies are very difficult to hear and rare in most sound waves for audio files. Overall across the entire spectrum, we aim to keep the variation in output voltage below 3 dB, but it is okay if it is very slightly more than that near 20 Hz and 20 kHz, as long as the common frequency sounds are unaffected. To be able to achieve a relatively equalized plot of the output volume, our amplifier should also have an equalized output voltage across the spectrum. With our predetermined cutoff frequencies, we will have less amplification at 20 Hz and 20 kHz, especially the former of the two frequencies. We want to keep this within the acceptable margin of 3 dB.

2.2. Analysis

2.2.1. Analysis: Filter 1 - High-pass

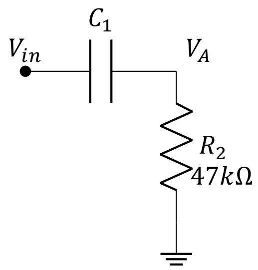

Figure 2.1: High-pass filter section of the circuit.

This circuit is designed to achieve a cutoff frequency of 20 Hz in order to partially achieve specification 2. As the analysis is conducted through ω, the following equation is required:

$$\omega = 2\pi f \tag{2.1}$$

Using Eq. (2.1) it can be found that the value of ω_h for the frequency cutoff is approximately 125 rad/s. In order to meet specification 2, the filter circuit is analyzed using voltage division. After derivation, this yields the following transfer function:

$$\frac{V_A}{V_{in}} = \frac{j\omega R_2 C_1}{1 + j\omega R_2 C_1} \tag{2.2}$$

From this transfer function, a function to calculate the cutoff frequency can be derived:

$$\omega_h = \frac{1}{C_1 R_2} \tag{2.3}$$

As both ω_h and R_2 are known, the value of C_1 can be calculated through making the use of Eq. (2.3). This yields the value of 170 nF for C_1.

2.2.2. Analysis: Filter 2 - Low-pass

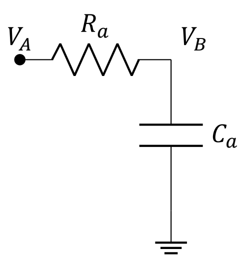

Figure 2.2: Low-pass filter section of the circuit.

This circuit is designed to achieve a cutoff frequency of 40 kHz in order to partially achieve specification 2. This design together with the previous design should meet the entirety of specification 2. The analysis is conducted through ω, requiring the use of Eq. (2.1). Using this equation, it can be found that the value of ω_l is roughly 250k rad/s. The analysis of this circuit is done by using voltage division. After derivation, this yields the following transfer function:

$$\frac{V_B}{V_A} = \frac{1}{1 + j\omega R_a C_a} \tag{2.4}$$

Similarly to filter 1, from this transfer function, a function to calculate the cutoff frequency can again be derived:

$$\omega_l = \frac{1}{C_a R_a} \tag{2.5}$$

This equation still has both R_a and C_a as unknowns. If we choose a realistic value for either one, the other value can be found using the Eq. (2.5). In order to minimize voltage loss over the two filters, the value for R_a to needs to be as large as possible relative to the value of R_2[3]. By doing so, the voltage output of the filters, V_B, approaches the voltage input, V_{in}, which allows us to choose a smaller value for the gain over the opamp circuit. We preferably do not want to compensate for the loss of voltage, since a higher gain over the opamp also gives a bigger chance of larger voltage variations at the output of our circuit. Because of this, minimizing the voltage loss over the filters is preferable. However, as according to Eq. (2.5), choosing a larger value for R_a would also require us to choose smaller values for C_a. We chose the value of R_a to be 4 kΩ, which gives us the value of 1 nF for C_a, which is available to us.

2.2.3. Analysis: Amplifier

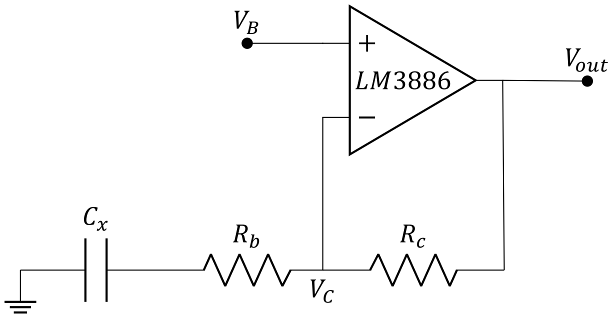

Figure 2.3: Amplification section of the circuit.

This circuit is designed to meet specification 3. The amplifier that we are using is a Texas Instruments LM3886 model amplifier. This section of the circuit should primarily focus on achieving a voltage gain of 25, since the passband part of the specification is handled by the filter part. As we have previously already minimized the loss over the filters, and early simulations of the filters have shown us that the loss over the filters is less than 1%, we do not need to compensate for any voltage loss and a gain of 25 is sufficient. The analysis of this part of the circuit is once again done by voltage division, which after derivation yields the following formula to calculate the gain over the amplifier:

$$\frac{V_{out}}{V_C} = \frac{j\omega C_x R_c}{j\omega C_x R_b + 1} + 1 \tag{2.6}$$

$\frac{V_{out}}{V_C}$ is the same as the gain over this circuit. From Eq. (2.6), it can be reasoned that if $j\omega C_x R_b$ is sufficiently large, the +1 in the denominator can be ignored as it will have negligible effect. This is the case for all relevant frequency levels for our circuit. If the +1 in the denominator can be ignored, the fraction can be simplified into the following equation, which has been used to calculate the gain as a ratio between R_c and R_b:

$$Gain = \frac{R_c}{R_b} + 1 \tag{2.7}$$

We want our circuit to have our circuit to have a gain of 25, so using Eq. (2.7) we can easily see that the ratio between R_c and R_b has to be 24:1. Eq. (2.6) can also be transformed into the form of a transfer function, which shows that this part of the circuit also functions as a high-pass filter, which is something we will have to consider when choosing values:

$$\frac{V_{out}}{V_C} = \frac{j\omega(C_xR_c + C_xR_b) + 1}{j\omega C_xR_b + 1} \tag{2.8}$$

We can also find the cutoff frequency of this equation, as it will be the value of the pole. All that needs to be done now, is to make sure that the cutoff frequency of Eq. (2.8) does not affect the passband. The pole gives the following equation:

$$\omega_p = \frac{1}{C_xR_b} \tag{2.9}$$

To not affect the passband of the filters, ω_p must be significantly lower than 125 rad/s. Thus it can be concluded that C_xR_b from Eq. (2.9) needs to be significantly large as to create such an ω_p. In order to do so, we chose a value of 100μF for C_x and a value of 1kΩ for R_b, which we considered significantly large. This would give us a cutoff frequency of 10 rad/s for ω_p, which we also find acceptable. If for any reason we would have to change this cutoff, we would prefer to make the cutoff frequency smaller, and this could be done by increasing the value for C_x and R_b, while making sure the gain is not significantly affected. After having decided these values, we can calculate the value of R_c using Eq. (2.7), which gave us a value of 24kΩ.

2.3. Full circuit

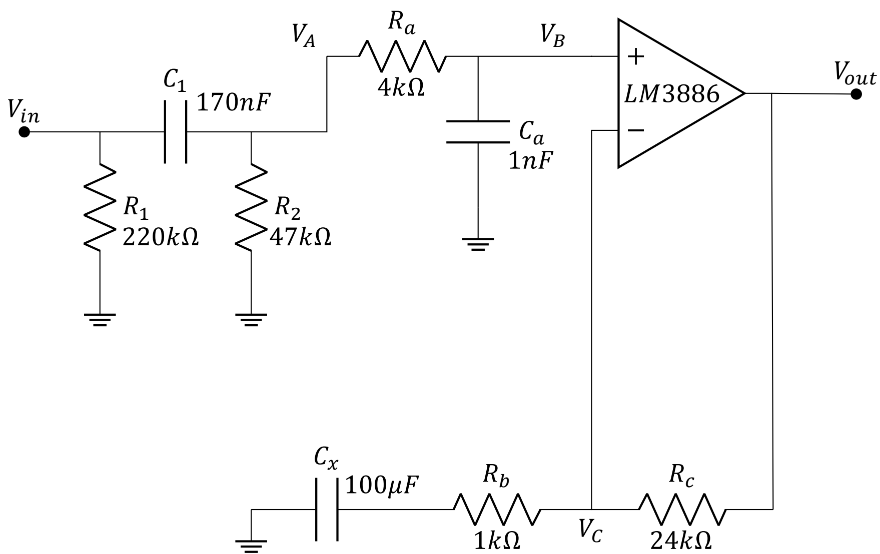

Figure 2.4: Full circuit as analyzed. Original circuit obtained from the manual [1].

We have now derived all parts of our circuit, and have found values for all of the elements within it. The entire circuit containing all of these values can be found in figure 2.4. Now that we have a complete circuit, we can simulate it to make sure that it satisfies all of our specifications, and if necessary, we can still make adjustments.

3
Simulation Results

3.0.1. Simulation: Filter Circuit

Using LTSpice, we have simulated all major parts of our design. First, we simulated the filter section of the circuit, using an input voltage of 0.4 V, the maximum voltage that we will obtain as audio input. A combination of both the filter circuits, and the circuit used for this simulation, can be found in the appendix, figure A.1. This simulation gave us the following bode plot:

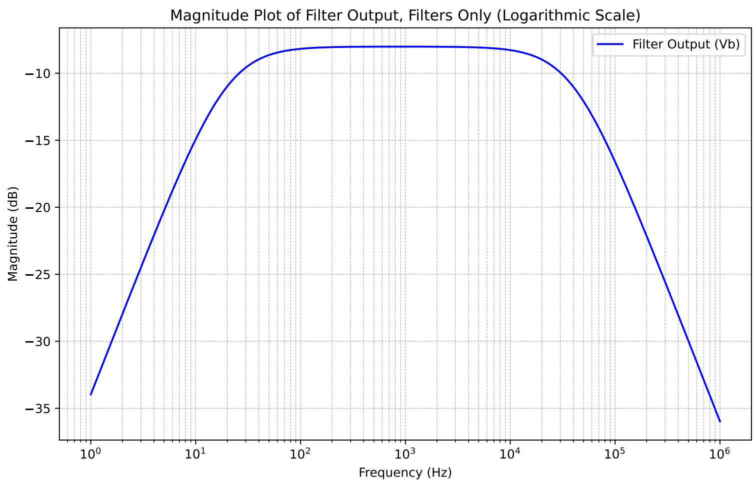

Figure 3.1: Bode plot simulation from LTSpice of the filter section of the circuit, as seen in figure A.1. Input voltage of 0.4 V. A slightly larger version of this plot can be found in the Appendix, figure A.2.

In LTSpice, it is possible to change the y-axis of the plot to voltage instead of decibel, and from that plot, we could clearly see that the voltage output of the filter section of the circuit was still 0.4 V. In order to prove this using the bode plot as shown in figure 3.1, the following equation was used to obtain the output voltage using the decibel values:

$$V_M = 10^{\frac{dB}{20}} \tag{3.1}$$

By filling in the magnitude at the peak of the simulation into this equation, approximately -8 dB, we also obtain an output voltage of 0.4 V. From this observation, we can conclude that the voltage loss over the filters has been minimized properly and is fully negligible, and thus we concluded that a gain of 25 over the amplifier part of the circuit is sufficient and suitable. Furthermore, the cutoff frequencies as previously calculated with Eq. (2.3) and Eq. (2.5) are also properly visible within this simulation plot as the -3 dB points of the graph, as both 20 Hz and 40 kHz give an output magnitude of approximately -11 dB. So, we can conclude that this part of the circuit accurately satisfies specification 2.

3.0.2. Simulation: Amplifier Circuit

As we have successfully minimized the voltage loss over the filters, a gain of 25 over the amplifier is sufficient, and our previously designed amplifier circuit is suitable. Therefore, we did not need to make any adjustments to it, and used that circuit for our simulations of the amplifier part of our design. Once again using an input voltage of 0.4 V, which is the maximum output voltage of the filter part of the circuit, this simulation gave us the following bode plot:

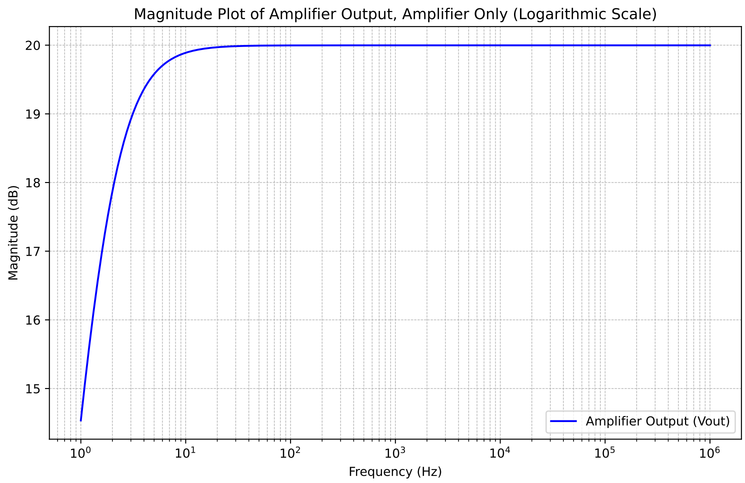

Figure 3.2: Bode plot simulation from LTSpice of the amplifier section, as seen in figure 2.3. Input voltage of 0.4 V. A slightly larger version of this plot can be found in the Appendix, figure A.3.

With a gain of 25 over an input voltage of 0.4 V, we expect to see an output voltage of 10 V over the amplifier part of the circuit at the peaks of the plot. By changing the y-axis in LTSpice to voltage instead of decibel, it was clearly visible that this is the case for this circuit, and also by using Eq. (3.1) with the magnitude value at the peak, 20 dB, we calculated an output voltage of 10 V. A derived in Eq. (2.9), we expected a cutoff frequency of 10 rad/s over this circuit. From this plot, it can be seen that the cutoff frequency at -3dB is approximately at 1.5-1.6 Hz. Using Eq. (2.1) to convert it to rad/s, we found that it is equal to approximately 10 rad/s as expected, so all of our expectations and specifications were met. Wrapping up, we concluded that the output gain over this circuit is acceptable and satisfies specification 3.

3.0.3. Simulation: Full Circuit

To fully get a grasp of what our entire circuit would theoretically do, we simulated our entire circuit design. By doing this simulation, we could fully observe whether everything looks according to our expectations and whether our specifications over the entire circuit have actually been met. So, we simulated our entire circuit, combining both of the circuits used to obtain figure 3.1 and 3.2, in order to obtain a bode plot for our entire circuit. This simulation yielded us the following bode plot:

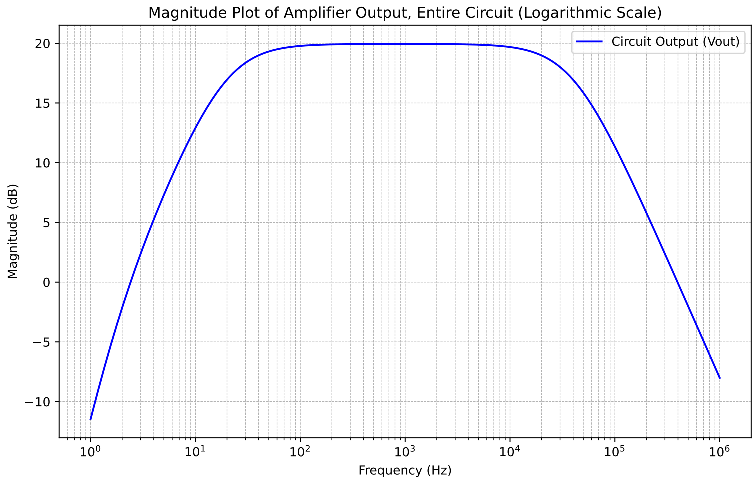

Figure 3.3: Bode plot simulation from LTSpice of the complete circuit, as seen in figure 2.4. Input voltage of 0.4 V. A slightly larger version of this plot can be found in the Appendix, figure A.4.

The gain of the output voltage is completely as expected, and has a peak of 20 dB corresponding to an output voltage of 10 V as previously calculated and observed. The -3 dB cutoff frequencies, visible at 17 dB in the plot, are also at 20 Hz and 40 kHz as expected. From this plot it is also visible that at the high end of the spectrum, near frequencies of 20 kHz, the output signals of these corresponding frequencies have a slightly lower voltage with a maximum deviation of approximately -1 dB. This is well within the deviation range we defined as acceptable. By the same token, at the lower end of the spectrum, near frequencies of 20 Hz, the output signal of these corresponding frequencies have a slightly lower voltage with a maximum deviation of approximately -3 dB. As also visible in the plot, a deviation of approximately -1 dB is obtained at signal frequencies of about 40-50 Hz. Since most people cannot properly hear frequencies below 40 Hz well, we have determined that this deviation is acceptable as well. Once the entire speaker has been built, it is possible that low frequency signals of less than 40-50 Hz have slightly bigger voltage deviations than -3 dB, but due to the previous mentioned reason of those frequencies being barely audible to most people, we have still decided not to change our circuit. We conclude that the obtained bode plot of our fully designed circuit satisfies both specification 2 and 3. Consequently, we have decided to build this designed circuit without making any more adjustments.

4 Measurement Results

During the construction phase of this project, not all values for each element and component that we have initially calculated were available. Thus, some slight modifications have been made to our design based on the accessibility of the elements. Due to this, we have made changes to R_a, R_c and C_x:

- R_a has been changed from 4 kΩ to 3.9 kΩ. This slightly increases the cutoff value for the lowpass filter, which means that the upperbound of the bandpass increases. This is no issue for our specifications.
- R_c has been changed from 24 kΩ to 27 kΩ. This increases our expected gain from 25 to 28, according to Eq. (2.7). We find this acceptable, as this only increases our maximum audio volume output, while the supply of our power amplifier should be able to handle this.
- C_x has been changed from $100\mu F$ into $1000\mu F$. This decreases the cutoff frequency of the amplifier circuit, which should also have no negative effect on our output.

All of these changes have been compiled into figure 4.1:

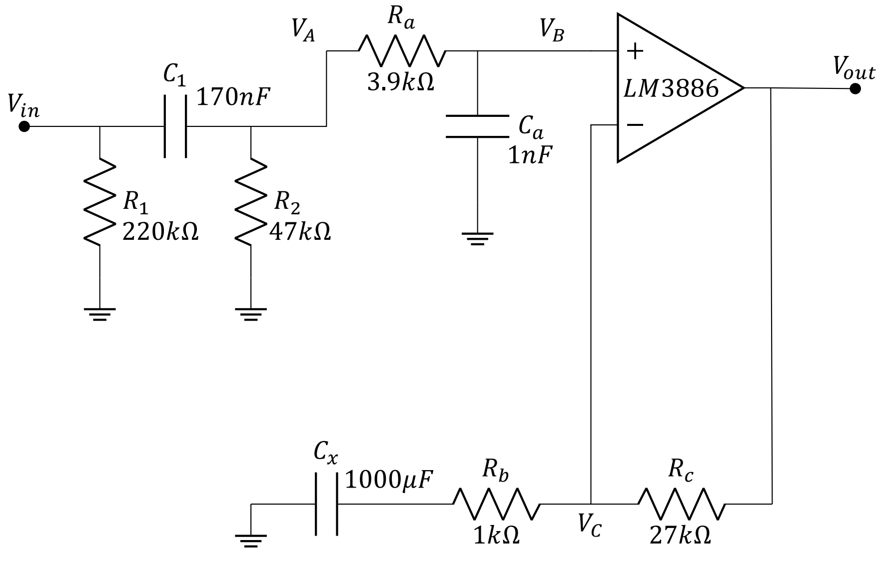

Figure 4.1: Full circuit as we have built, based on 2.4. As some parts were not available with exact values, we have slightly changed some elements.

We have built this circuit and have made measurements using an input voltage V_{in} of 0.2 V. For our first measurements, we have decided to use this input voltage as it is half of the maximum voltage anticipated to be used as an input to our system. Based on the measurements obtained, the following plot has been generated:

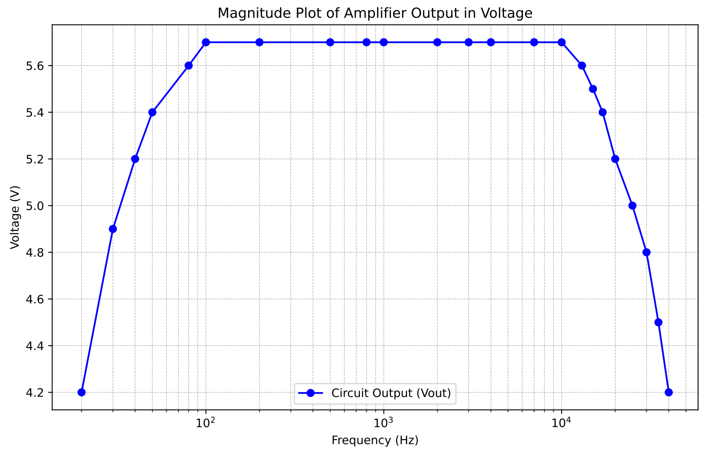

Figure 4.2: First measurement that has been done with our built circuit on December 19, 2024. Plot shows the output voltage of the entire circuit, when using an input audio signal with a voltage of 0.2 V. Measurements are from the circuit as seen in figure 4.1. A slightly larger version of this plot can be found in the Appendix, figure A.5.

From this figure, it can be concluded that the circuit behaves approximately as expected by our simulations and expectations. The voltage gain straightens out between 100 Hz and 10 kHz. However, between 40 Hz and 20 kHz, the voltage only has a minimal deviation of 0.5 V, which is correspondent to a difference of approximately -1 dB. The same behavior could be seen in the simulations on figure A.4. The cutoff frequencies are also the same: the -3 dB is roughly equal to -1.4 V as can be calculated using Eq. (3.1). The voltage gain changed as a result of changing the value for R_c. The increase of 3 kΩ means that the circuit is expected to have a gain of 28 instead of 25, as previously mentioned. The following figure shows the gain calculated from our measurement:

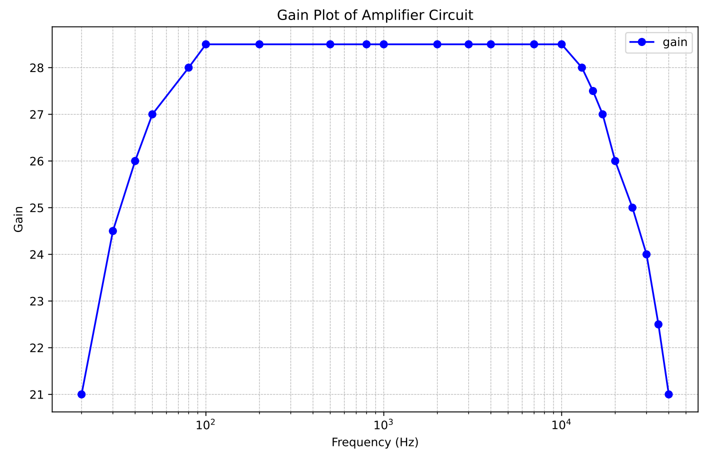

Figure 4.3: First measurement that has been done with our built circuit on December 19, 2024. Plot shows the gain of the entire circuit. Measurements are from the circuit as seen in 4.1. A slightly larger version of this plot can be found in the Appendix, figure A.6.

The calculated gain at the peak of the plot is approximately 0.5 higher than expected. This deviation in gain compared to our expectations can be explained by the margin of deviations of the values for the resistors R_b and R_c. We consider this gain deviation acceptable. Overall, we are satisfied with our current measurements for the circuit.

5

Conclusion

The simulations of our circuit look good and satisfy all the specifications mentioned. C_1 makes sure that no DC currents can pass the filters, and any DC currents will pass through R_1 to the ground. Combined, filter 1 and 2 create a bandpass filter with cutoff frequencies at 20 Hz and 40k Hz. The voltage loss over these filters has also been minimized by making R_a's value as large as possible, relative to R_2. The leftover signal is then amplified by the amplifier with a gain of 25 in our simulations, determined by the ratio between R_c and R_b. The V_{out} signal that we have simulated is also according to our expectations and satisfies our specifications.

Due to accessibility limitations, we have slightly altered the designed circuit when building it. Consequently to these changes, the output gain has been increased compared to our simulations, but we find this acceptable, as this only increases our maximum output sound volume, while it is still within the margins of the supply for the amplifier. The measurements that we have done on our circuit so far look very promising and satisfy the specifications for our circuit. In the future, we will do more measurements to further test our built circuit, with more different input signals. Once we are fully satisfied with our circuit, we start building the entire speaker by combining our amplifier together with the power supply and audio filters of the other subgroups.

Theoretically, we would want bode plot of the output audio signal that goes to the filter components, which are not a part of this report, to be as flat as possible. However, the elements and electrical components (drivers and amplifier) that are available to us are not capable of sustaining this[2]. As such, the specifications say that we should keep our cutoff frequencies at 20 - 40k Hz. This does result in frequencies at the far ends of the spectrum to have lower voltages, and thus the audio output of the speaker will have lower audio volumes at these ranges. We consider this acceptable for this project, as these frequencies are the most difficult for us humans to hear and most people would not be able to hear these frequencies properly anyway. For that reason, these frequencies are quite rare in audio waves, and may sometimes be considered background noise. Furthermore, the benefits of keeping our cutoff frequencies at 20 - 40k Hz far outweigh the potential benefits of extending the bandwidth beyond this range in order to make the output signal's bode plot more flat.

Future research on these kind of speakers may want to explore the potential of expanding the bandwidth of the filters. To be able to do so, accessibility to more powerful electrical components is required, as the amplifier needs to be able to handle larger frequency signals, and the components need to be able to sustain the power consumption and other downsides of this bandwidth expansion. The speaker drivers and components that are available to us during this project or not capable enough, and we were discouraged to attempt to implement these ideas.

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A

Appendix

A.1. Additional Circuits

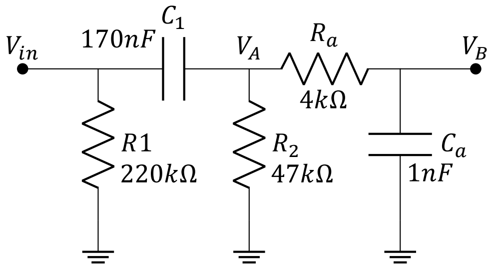

Figure A.1: Full circuit of both the filters combined.

A.2. Simulation Figures

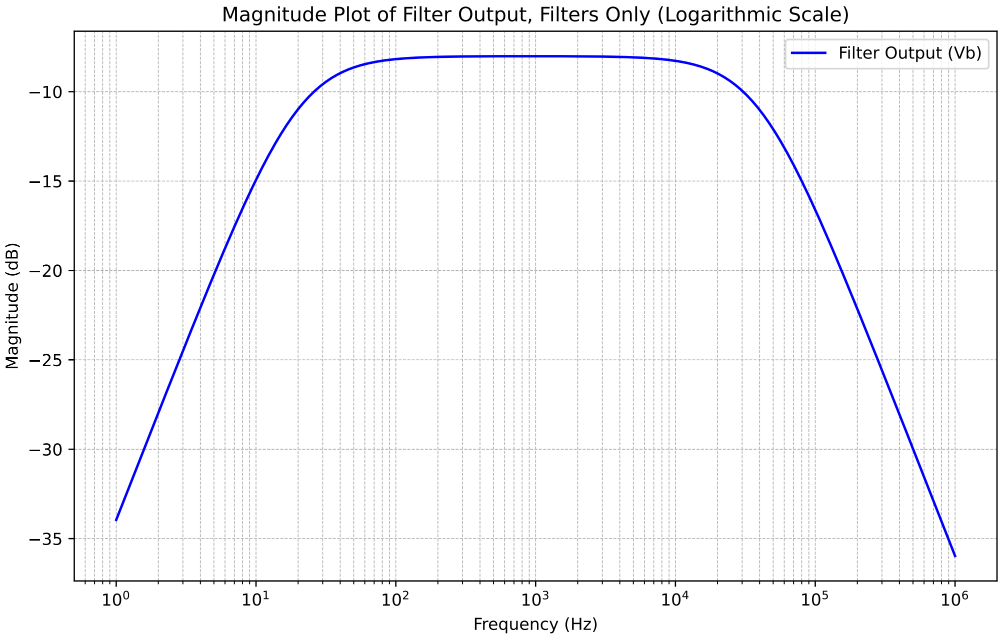

Figure A.2: Bode plot simulation from LTSpice of the filter section of the circuit, as seen in figure A.1.

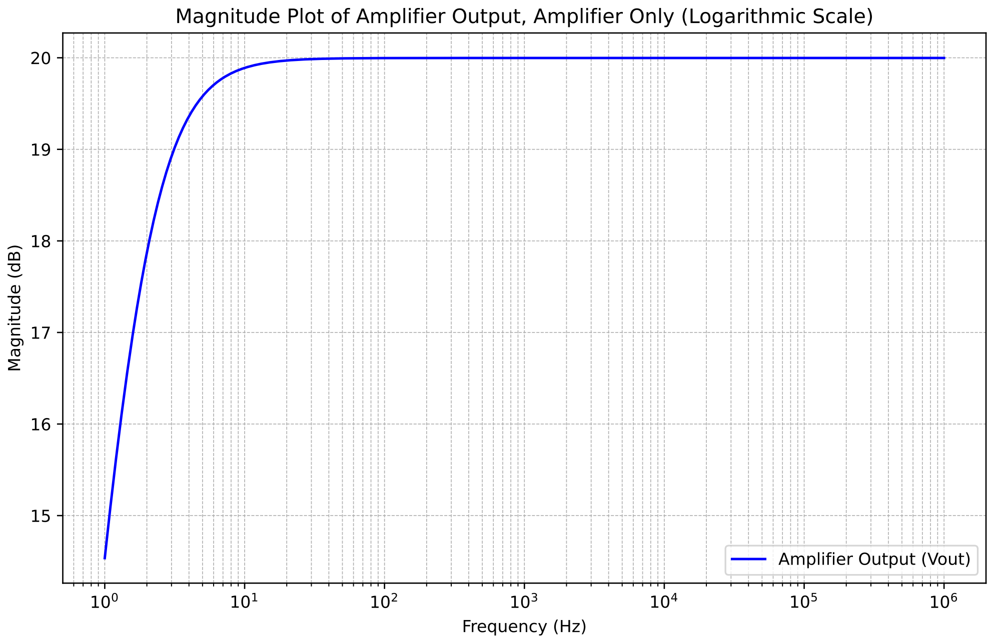

Figure A.3: Bode plot simulation from LTSpice of the amplifier section, as seen in figure 2.3.

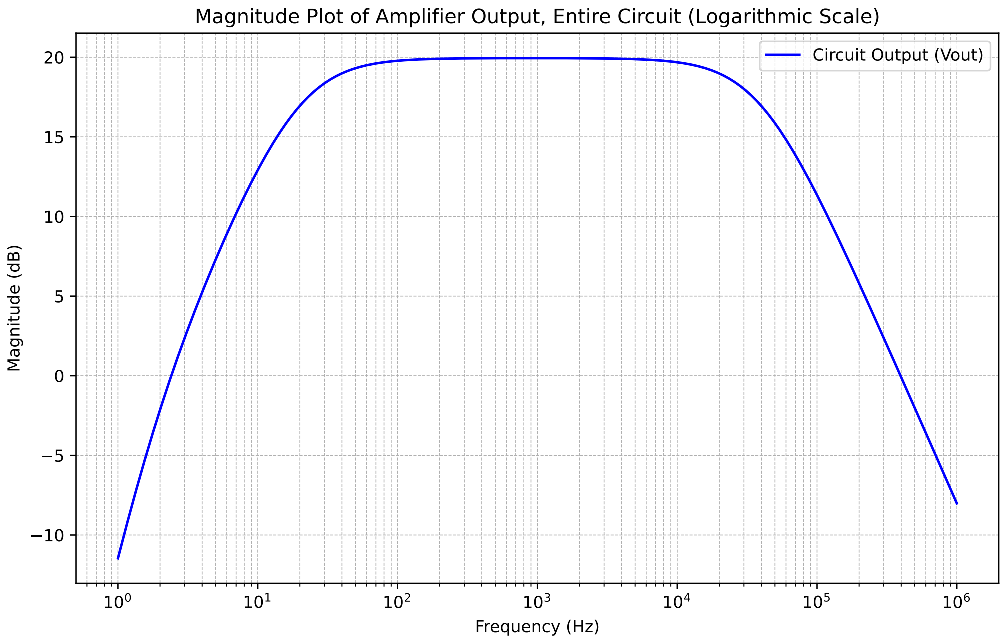

Figure A.4: Bode plot simulation from LTSpice of the complete circuit, as seen in figure 2.4.

A.3. Measurement Figures

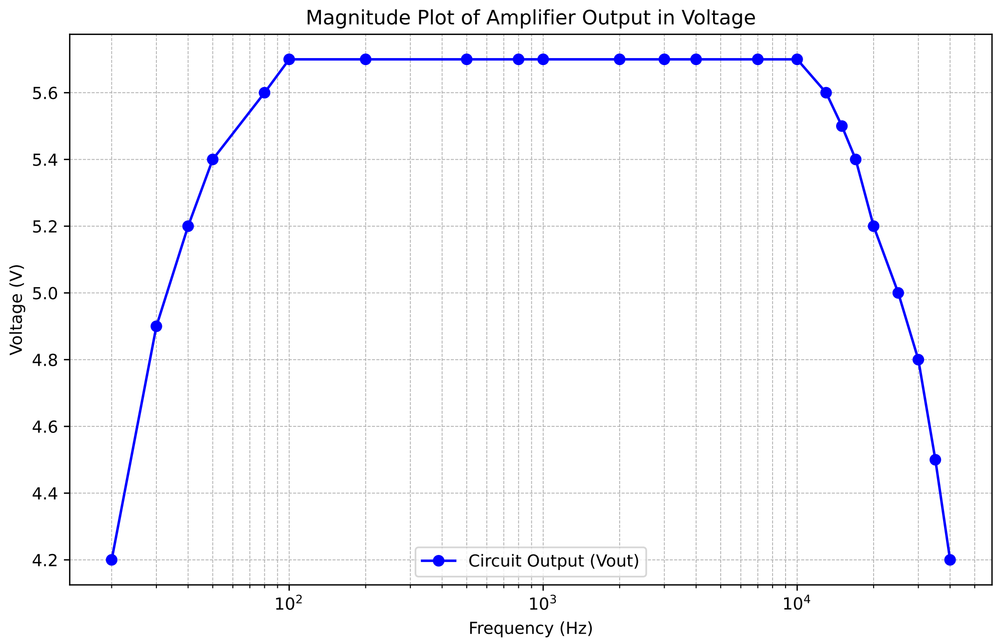

Figure A.5: First measurement that has been done with our built circuit on December 19, 2024. Plot shows the output voltage of the entire circuit, when using an input audio signal with a voltage of 0.2 V. Measurements are from the circuit as seen in figure 4.1.

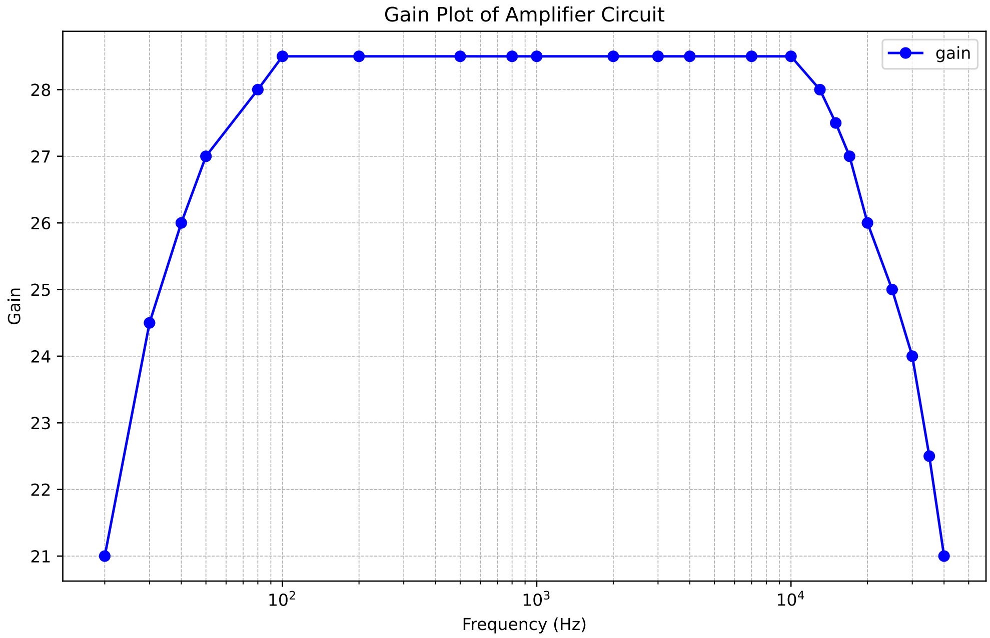

Figure A.6: First measurement that has been done with our built circuit on December 19, 2024. Plot shows the output voltage of the entire circuit, when using an input audio signal with a voltage of 0.2 V. Measurements are from the circuit as seen in figure 4.1.