ECE20007 Final Project: Audio Equalizer

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ECE20007 Final Project

In ECE20007, you will complete a final project with your lab partner that will count as your final assessment for this lab. You will work with your lab partner to complete the project, but you are expected to work out the math, design your own topology, and complete the report as an individual assignment.

You will have two weeks of lab time to work on the project and your report. The second week will consist of performing a demonstration to your teaching assistant to verify your project works correctly.

These labs will be written and submitted as a formal lab report, in which you will be expected to include an abstract, introduction, theory, procedure, results, and a conclusion.

The final project will equate to 10% of your final grade. You will be expected to demonstrate that your project works in lab during the second week of the project. The demonstration will count as a multiplier to the results section of your report, which is 40% of the report grade. If you are able to properly demonstrate half of the project in the demonstration, you will be able to receive up to half of the credit for the results section, or 20% of the total grade.

$$Project\_Grade = (Demo\_Grade \cdot Results\_Grade) + Rest\_Of\_Report \tag{1}$$

The main purpose of these projects is to get you thinking in a more experimental mindset, testing different ways of presenting data, and thoroughly explaining your process and results. As you move forward in science and engineering, effectively developing and explaining a project or product will become increasingly important, especially when justifying funding or employment. For these projects, the general expectations and best practices will be laid out in this document, but it will be up to you to determine a good process to complete the goals at hand. Below is the general outline for how the projects will work:

Before Lab, and early lab 1:

1. Complete the assigned reading for the project and watch any videos associated with the project. You may want to continue doing general research to get a better understanding on the concepts being used. Make sure to cite your sources.
2. Read through the tasks you will perform with the purpose of effectively understanding what the end goal of the experiment is.
3. Develop a general schematic, along with general steps for how you will approach the tasks of the experiment. This can be changed once you talk with TAs in lab, but you should try to have a good plan ahead of time.

During Lab:

1. Work through your plan to complete the tasks.
2. Take thorough notes, measurements, screenshots, and pictures so that you will have enough information to work on your report after you are done.

After Lab:

1. Review your notes and measurements from lab, and verify you have enough data to justify your explanations and goals. If you do not, you may have to go back to lab to take more measurements.
2. Develop your lab report with the expected sections for that report. If you have at least a rough draft completed before the second lab session of the project, you are welcome to ask for feedback on your formatting, explanations, and your report in general prior to finishing and submitting.

Final Project Intro

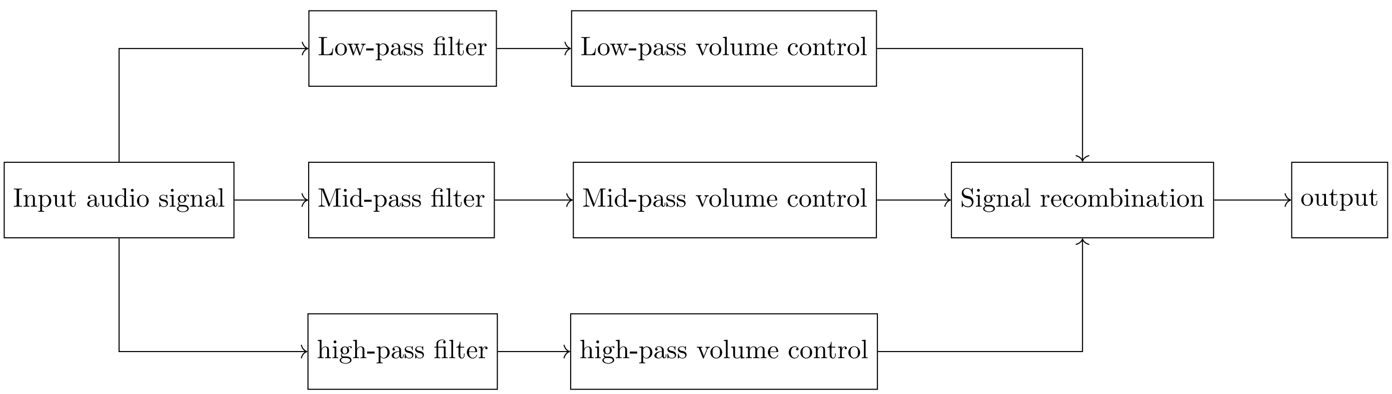
Throughout this course, you have learned about several principles and elements related to analog signal processing, namely filters and amplifiers. At this point in the course, you may be asking yourself, “Now, what can I actually do with these concepts?” Well, processing analog signals is the answer!

One useful application of these principles is making a combination audio equalizer and amplifier. More likely than not, you’ve encountered these (sometimes unknowingly) in your life. Some examples of where these can be found inside of including, but not limited to: phones, Bluetooth speakers, car radios, and musical instrument amplifiers. Sometimes you can adjust them manually using knobs, but in our current era, a lot of this is contained within digital algorithms. In our case, we will be creating one using manual knobs.

In this project, the objective is to take many of the concepts you have learned in this course to create something that demonstrates functionality. In our case, an audio equalizer. You will demonstrate your knowledge of signal filters, operational amplifiers, oscilloscope techniques, and breadboard usage. The desired implementation is a circuit that can have an audio source (computer, phone, Mp3, etc.) plugged into it, which can then have the signal adjusted for low, middle, and high frequency boosts or reductions. After, it will be able to be plugged into a speaker that will play the processed sound.

Basics of an Audio Equalizer

A typical audio equalizer is divided into five separate stages: the input stage, the filtering stage, the adjustment stage, the recombination stage, and the output stage. At the input stage, the signal is distributed to separate filters, which will accept frequencies that they are designed for. These frequencies then get filtered, adjusted, and recombined. After recombination, they are fed as an output into an amplifier of some sort. A block diagram reference is shown below:



1 Basics of a Power Amplifier

There are multiple different types of amplifiers that are commercially available, ranging from digital to analog, as well as having multiple different architectures for both. The specifics of the architecture are out of the scope of this course, but in our case we will be utilizing a low-power multistage transistor amplifier. Specifically, we will use a class AB amplifier, which has a similar architecture and functionality to the operational amplifiers that the earlier labs in this course use.

What separates an operational amplifier to a power amplifier is simple: output current capacity. As an example, the LM324 outputting a 5Vpp sine wave has a maximum output current of around 5.9mA due to its large resistance on its output terminal. A power amplifier, like the LM386 that we will be using in this lab, is capable of outputting around 223mA of current under the same load, which is over 1400 times more output power!

This is why something like a power amplifier gets chosen to provide more power to an output. Concert venues will oftentimes have speaker systems that require multiple thousands of watts to drive them, and you can't do that with something like an LM324. Fortunately for you, this course is not interested in making you design your own. We're only looking for 400mW of power for this project, and an LM386 power amplifier chip puts out just that!

2 A Quick review of Passive Filters

This course covers only a small glimpse into passive filters in the filters lab. In summation, passive filters are just filters that rely entirely on circuit reactivity instead of a powered element. As a quick definition, circuit reactivity is just properties of a circuit that return sent power back to its source. Essentially, the components are "reacting" to the change in frequency. The term is used in this way more heavily in the power systems field of electrical engineering, but it still holds true for small and/or low voltage systems like this. Passive first-order filters of the high-pass and low-pass type were the subset of filters covered earlier in this class in the filters lab, specifically for RC (resistor-capacitor) topologies. There are many other topologies out there, but they are not covered by nor required for this course. If you feel like pushing your project further by using more complex filters, feel free to do so! However, not every TA may be able to help you debug your circuit.

3 A Glimpse into Active Filtering

Active filtering is a subset of filtering that involves an operational amplifier component. Having the op amp does a few things for a circuit: controllable gain, better output isolation, and a higher-order cutoff rate. Essentially, this raises the order of the differential equation governing the circuit while also keeping something similar to a first order topology. These do have easier calculations than a typical second order circuit. A specific direction, if one is interested in researching these, it the Sallen-Key topology. These utilize two resistors and capacitors, and yield a second-order response.

4 Buffering and Why You Should Do It

Finally, buffering is probably the most important part of this project. One thing that isn't directly mentioned in this course is terminal impedances, which is another form of loading. Loading has been discussed in a variety of perspectives throughout the lab, so feel free to look back on previous labs to refresh on loading. You can think of this problem as input loading, where the back end of the circuit begins to affect the front end. For first-order filters, this is not the biggest deal, but sometimes it may put you just outside of the acceptable tolerances. In more practical terms, you may notice if you try to change the cutoff of one filter, it will likely change the cutoffs of the other two filters. The way you get around this is by creating voltage-followers through buffer implementation.

Buffers are an easy way to isolate an input signal from an output signal. Essentially what happens is that the buffer takes the input signal and electrically recreates it on the output, and the result is two separate circuits. The analysis is not relevant here as it is covered in ECE20008 and ECE20002, but it is important to know. If you cannot seem to get your cutoffs right, try implementing a buffer before the input of the filter.

5 Volume Control

Volume control is a simple concept in the form of analog filters. It is simply a way to reliably reduce the amplitude of the outgoing waveform after it has been processed. Your first thought may be, "I can just insert a potentiometer at the end of this circuit and be done with it," which would be almost correct. Before we get into the proper way to do it, let's talk about why that isn't the best way around it.

As mentioned earlier, loading is one of the more problematic concepts that you will experience in this lab. Another practical example of this lab is the potentiometer loading the input of the end-stage summation amplifier, which is covered in the next section. The summation amplifiers have resistor-based gain control, and through Kirchoff's Voltage Law, we can assume that a potentiometer before the input resistor will modify what voltage the input terminal is receiving. To get around this, we can combine an op amp and a potentiometer, because op amps also carry an isolation property to them, albeit less intense than a buffer would demonstrate.

6 Signal Recombination

Summation amplifiers are just normal inverting amplifiers with multiple inputs on the negative terminal. These types of amplifiers are a basic way to add multiple signals on the input to a combined output, while also ignoring superposition. An example topology is shown in the op-amp lab, specifically in the inverting amplifier reading.

The reason why the summing amplifier (somewhat) ignores superposition is because of the "virtual ground" principle that inverting amplifiers hold. Op amps naturally try to adjust both of their inputs to be the same to each other when they are setup with negative feedback, so if the positive terminal is grounded, the negative terminal will try and compensate to ground. Because of this, there are not any other paths to conduction. In a non-ideal world, there are brief moments when the voltage on that terminal will quickly shoot up and fall back down, but it's nothing that will affect the gain calculation seriously, so you can calculate each input line's gain individually as normal.

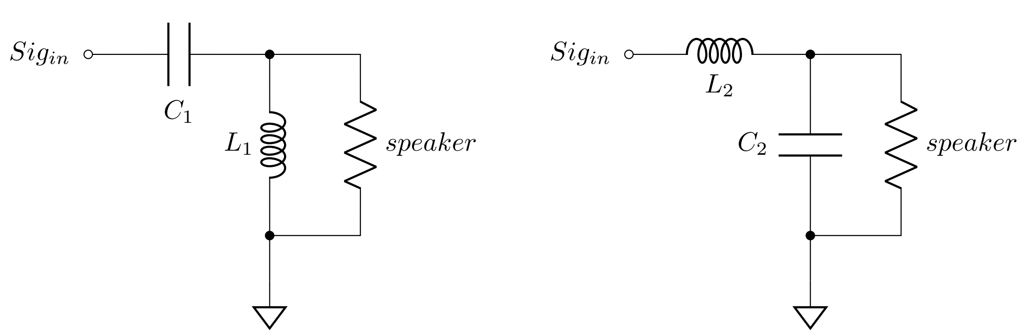
Below, you'll see a concept defined as "output ripple," the difference between your highest and lowest point, in V_{RMS}.

7 Crossover Filters and Speakers

Crossover filters are an easy way to isolate different speakers that have different frequency ranges. In our lab, you'll notice two speakers bound to a metal chassis. One of these is of a smaller diameter, the tweeter, and the other of a larger diameter, the woofer. You may have an intuitive notion that these accept different frequencies, which is correct. The tweeter is a smaller speaker that needs less magnetic force to drive, which means a lower inductance and a lower turn count on the inductor inside of the speaker. Conversely, the woofer requires a higher magnetic force to drive, which means a higher inductance and a higher turn count.

Because inductors act as reactive components and behave like filters, they will naturally filter out signals that are wildly out of their operating range. However, things that are close to their operating range do not get filtered out, and when they pass through, they sound bad. If you buy a cheap Bluetooth speaker with multiple cones inside of it, one of the reasons why it will sound bad is because they lack extra filtering, which is where crossover filters come into play.

Crossover filters are usually second or higher order filters since the goal is to reduce the number of unwanted frequencies as much as possible. Since they are higher order, we will just give the cutoffs without derivations, as they are typically gone over in higher-level labs. Since this is a shared-cutoff circuit, you only need to make two calculations: one for the inductors, and one for the capacitors. They are shown below:



(a) Crossover Filter 1 (b) Crossover Filter 2

Figure 1: Symmetric High pass and Low pass filters to provide a crossover at -3dB. Equations to get the proper values can be used in Formula 2.

$$L = \frac{R_{spk}}{\pi f_c}, \qquad C = \frac{1}{4\pi f_c R_{spk}} \tag{2}$$

8 Specifications

In your audio equalizer, implement these specifications:

Specification	Requirement
Speaker Impedance	8 Ohm
Bass filter -3dB cutoff	320Hz $\pm$10%
Mid filter -3dB cutoffs	320-3200Hz $\pm$10%
Treble filter -3dB cutoff	3200Hz $\pm$10%
V_{amp} with all volumes turned to minimum settings	$<$15mV_{RMS} @ 100 Hz, 1000 Hz, 10000 Hz
V_{amp} with all volumes turned to maximum settings	100 mV_{RMS} $\pm$ 10% @ 100 Hz, 1000 Hz, 10000 Hz
Maximum ripple	15mV_{rms}
Output power	Greater than 400 mW from 100 Hz to 10000 Hz

Input voltage should be 320mV RMS, or what is generally used for commercial, non-amplified audio signals.

Descriptions

The list below contains more in-depth descriptions of the specifications you must implement. In your final demonstration and report, you should be able to show that you have met all of these numerically and/or graphically.

1. **Speaker Impedance:** The speaker impedance is the magnitude of the resistive and inductive characteristic of the speaker that you'll be using. As some background, these come in some common multiples: 2, 4, 8, or 16 Ohms if they're newer, and 4, 8, 16, 40 or 80 Ohms if they're older. The speakers provided in lab have one of these impedance characteristics. Demonstrate that they're the correct ones. There are Z meters available for use to do this on your lab benches, or you can use a direct method with a waveform generator and oscilloscope. For reference, 1kHz is a common test frequency for audio equipment.

2. **Bass filter -3dB cutoff:** This is the cutoff where the lower frequency filter becomes mostly inaudible. This course uses the -3dB point of a frequency analysis to measure it. It must be within 10% of the listed cutoff value.

3. **Mid filter -3dB cutoff:** This is the cutoff where the middle frequency filter becomes mostly inaudible. This course uses the -3dB point of a frequency analysis to measure it. The higher and lower cutoffs must be within 10% of the listed values.

4. **Treble filter -3dB cutoff:** This is the cutoff where the higher frequency filter becomes mostly inaudible. This course uses the -3dB point of a frequency analysis to measure it. It must be within 10% of the listed cutoff value.

5. V_{amp} **with all volumes turned to minimum settings:** This is the amplitude of the voltage feeding into the power amplifier with your volume knobs turned all the way down. At the listed frequencies, demonstrate that your equalizer reduces these to less than 15mV RMS. These should be inaudible when you play sounds.

6. V_{amp} **with all volumes turned to maximum settings:** This is the amplitude of the voltage feeding into the power amplifier when your volume knobs are turned all the way up. At the listed frequencies, demonstrate that your equalizer keeps these at 100mV, within the 10% tolerance.

7. **Maximum ripple:** This is the difference between the highest voltage and the lowest voltage in your frequency spectrum when all of your knobs are turned all the way up. This is to ensure that no frequencies are louder than the rest at a equalization setting. With your whole frequency spectrum, show that the highest peak is not greater than 15mV above the lowest trough.

8. **Output power:** Demonstrate that the power amplifier outputs more than 400mW of power to your speakers at the given frequency range.

Prelab

Design **and simulate** a circuit topology for the adjustable equalizer and volume control subsystems that should, in theory, meet the design requirements. This includes circuit selection, component calculations, and supply voltage calculations for you amplifiers. Specifically, the LM386.

In Lab

9 The Equalizer

1. Between the two generated prelabs (one made by you, and one by your lab partner), agree on which system topology you would like to use. Before doing so, make sure that you receive and evaluate feedback from the TA staff.
2. Construct and test each filter and volume control circuit independently. Once constructed, do not disassemble them.
3. Construct the power amplifier and integrate the filters into it with a summation amplifier.
4. Verify that everything works as intended.
5. Once everything is working, record measurements and/or plots that demonstrate the device is within the design specifications.
6. Connect the amplifier to the tweeter and subwoofer speakers provided in the lab. Verify that your design works and that you can hear and adjust the sound being produced on its output.
7. Using your computer or phone as an audio source, demonstrate your device and its relevant subsystems to your lab TAs. Make sure that you have all of your prior measurements and a full schematic available for review during your demonstration.

10 Bonus Credit (+15%)

1. Experimentally determine the performance specifications for the provided tweeter and woofer. This will mainly involve finding the cutoff frequencies of the speakers. You are welcome to use your own personal set of speakers, too.
2. Select a crossover filter frequency that will pass frequencies in each speaker that are within the speakers' operating ranges.
3. Using your selected frequency, design, construct, and implement a crossover filter between the power amplifier and the speakers that will pass the correct frequencies to each filter.
4. Measure a frequency response of the filter system. Include this filter system within your in-lab demo.