Introduction

In this lab, you will design three filters to pass different ranges of the audible frequency spectrum to flash LEDs in time to music — you'll have made your very own color organ!

To do this, you will select your desired corner frequencies and calculate the appropriate resistor and capacitor values to build filters with said corner frequencies.

The audible range is actually a somewhat small spectrum of frequencies, as demonstrated below:

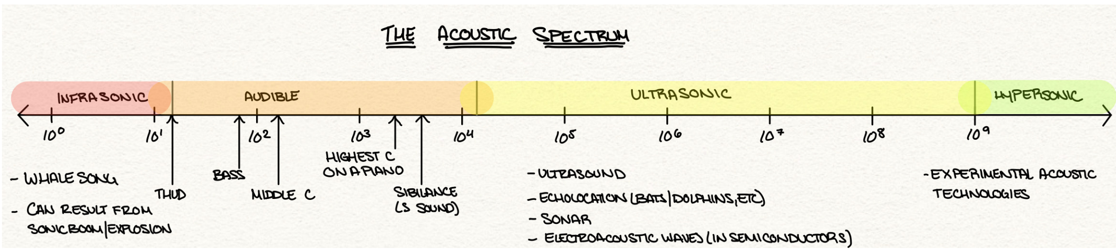


Figure 1: Sketch of the acoustic spectrum.

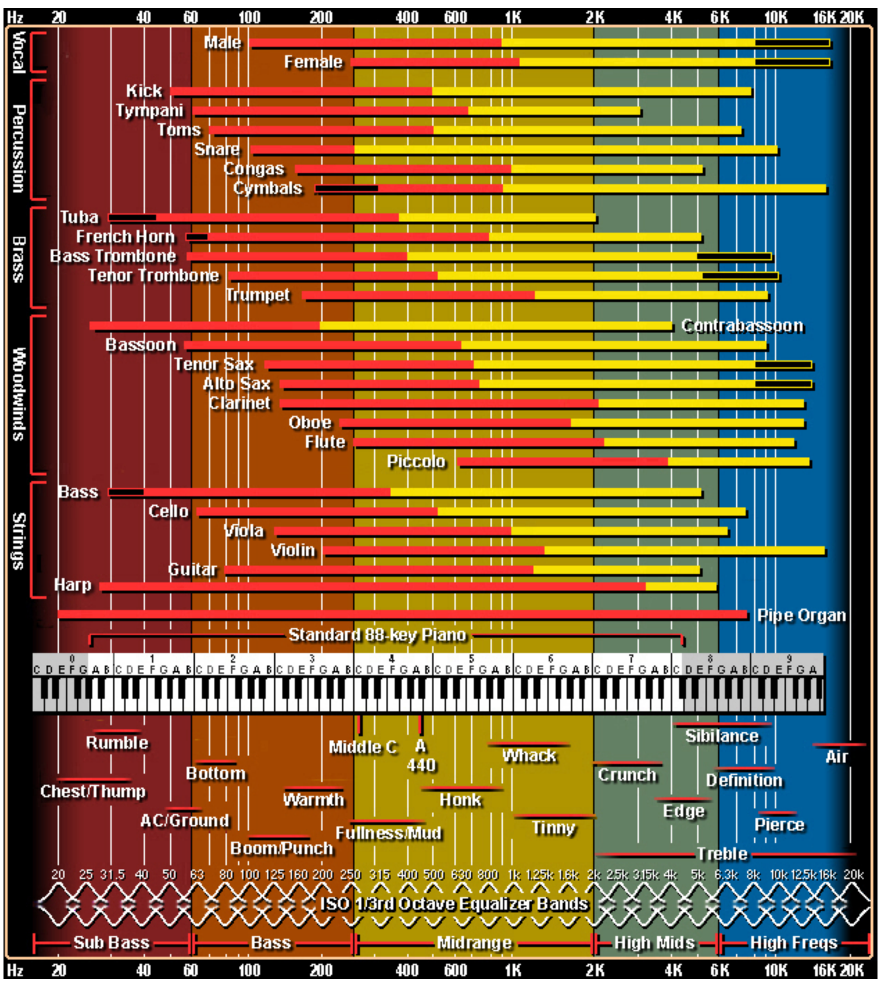


Figure 2: Expansion of the audible range of the acoustic spectrum.

Sanity check question: *Why might this fact cause some difficulties in this lab?*

Note: Acoustic waves are *not* electromagnetic waves: sound waves are mechanical and therefore need a medium through which to propagate, whereas EM waves do not need a medium[1]: they can propagate through the vacuum of space.

You will be targeting the bass, midrange, and treble sections depicted in figure 2 above, which we define as follows:

Bass	20-500 Hz
Midrange	1,000-5,000 Hz
Treble	6,000-20,000 Hz

Ultimately, these frequency ranges are **guidelines**: the goal of this lab is to independently light up your LEDs (with little to no overlap — two LEDs should not light up at the same pure frequency, and "dead zones" should be minimal/imperceptible). Also, you may notice that these ranges do not align exactly with those displayed in figure 2. After completing Part 1 of the lab, you will see why that is.

Lab

Part 1: Frequency Response of the Speaker-Microphone System

The system you are building today is not limited to just the color organ circuit: you must also consider the ability of your speaker to reliably reproduce the desired frequency at a volume large enough to excite the microphone, and the ability of your microphone to respond to the desired frequency. You must also consider that in some ranges, the signal will be highly attenuated, if picked up at all, and compensate for that when you design your color organ (*How can I add gain? Which frequencies should I choose? How can I create sharp cutoffs to minimize both overlap and dead zones?*). To gain the necessary information to design a working color organ, we first identify the speaker-microphone system's frequency response. We will do this empirically: you will sweep over a range of audio frequencies and record the amplitude of the received wave at that frequency.

Now you are ready to complete the first part of the lab! Go to the jupyter notebook and complete Part 1.

Part 2: *Bass*-ic Color Organ

Now we are ready to begin building the actual color organ circuit! The finished product will look something like this:

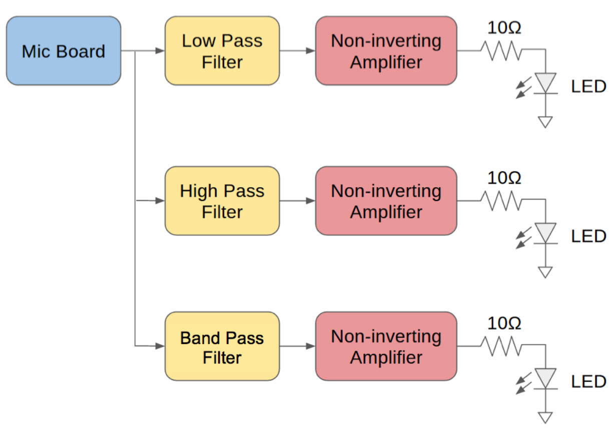


Figure 3: High-level overview of completed color organ.

[1]This bothered early scientists, so they came up with the concept of the aether (subsequently decommissioned in 1897).

Not only is this a larger circuit than those you have built in previous labs, but you will also be extending it (in a manner of your choosing) at the end of Color Organ Part II (lab 5), so **be sure to plan ahead when constructing your circuit.**

> **2.1.** As a layout exercise, sketch on a piece of paper how you plan to allocate the space on your breadboard.

The first filter you will build will isolate the bass frequencies, so it is a low-pass filter. You will use only one capacitor, so it is first-order.

Recall the Bode plots of first- vs. second-order poles.

> ***Something to ponder:*** *How could you use second-order behavior to improve your color organ?*

Part 3: A *Treble*-some Color Organ

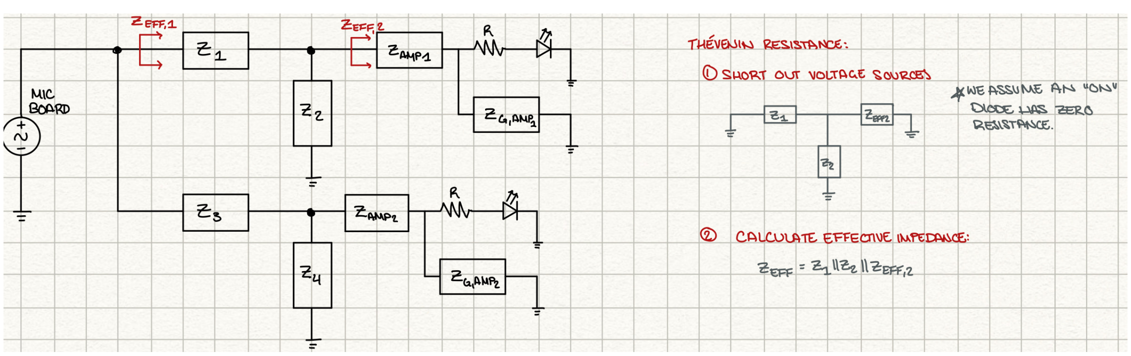
Now, you will build a first-order high-pass filter to isolate the treble frequencies.

> ***Sanity check question:*** *Do we need to put buffers in front of each filter now that we have two connected in parallel? I.e., does placing the filters in parallel affect their respective cutoff frequencies?*

To approach this question, we'll consider the topic of impedance "looking in" to a particular node or point in a circuit.

"Looking in"?? Electrons can't see!

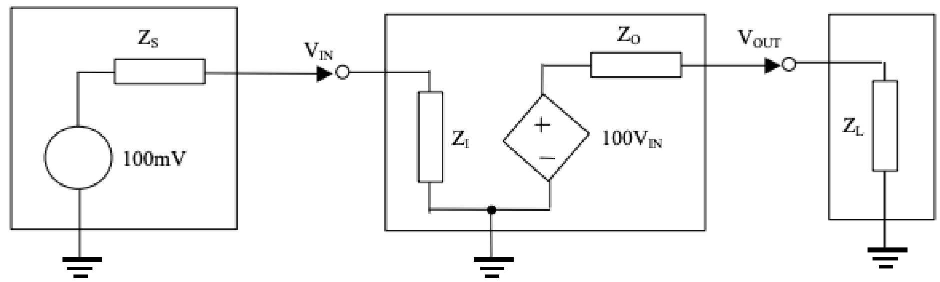
True, they can't. But, the concept of impedance "looking in" to a specific node is a very useful one (and one you'll see very often in circuits classes like EE 105 and EE 140, especially in transistor circuits). The impedance "looking in" to a node or part of a circuit is simply the Thèvenin equivalent impedance of that part of the circuit. Here's another quick refresher on Thèvenin:



Loading

Recall the impedance characteristics of the ideal op-amp: its input impedance is infinite and its output impedance is 0. This allows it to act like an ideal voltmeter at the input and supply infinite current at its output. But what happens if we don't have those characteristics?

Let's assume we have a noninverting amplifier with a gain of 100, with input impedance Z_I and output impedance Z_o, as in the schematic below. The middle box represents the amplifier.



We now see that V_{in} depends on the source output impedance Z_S and the amplifier input impedance Z_I, and V_{out} depends on the amplifier output impedance Z_O and the load impedance Z_L. Recalling the voltage divider equation,

$$V_{out} = (100V_{IN})\frac{Z_L}{Z_L + Z_O}$$

But $100V_{IN}$ is our desired V_O! To keep that approximately correct and avoid "loading" the output and reducing the voltage noticeably from what we expect, we need Z_L to be considerably larger than Z_O to keep $\frac{Z_L}{Z_L+Z_O}$ as close to 1 as possible.

This is why you set the "output load" on the signal generator to "High-Z": by doing so, you are telling it to expect a high-impedance load. The function generator has a 50-ohm output impedance, while the oscilloscope probes are high-impedance, so when the function generator is set to "High-Z," you can probe it with the oscilloscope and see the output voltage you expect (the one you explicitly set). If you set the function generator to "50 Ohm", it expects a 50 Ohm load. Since this is equal to its output impedance, V_{out} would be halved, so the function generator compensates for this by doubling its output voltage in 50 Ohm mode.

Buffers

You can think of a buffer as providing an impedance transformation between two *cascaded* circuits. When you observe an undesired loading effect between two circuits, placing a buffer between them changes the load impedance of the first circuit to a very high value and the source impedance of the second to a very low value in accordance with (approximately) ideal op-amp characteristics. This allows you to build very modular circuits easily, without having to do lots of ugly algebra (like you did with the passive bandpass filter you saw in your homework).

Something to ponder: *When/where would you use buffers in your color organ? What improvements could you make that would require you to add additional buffers?*