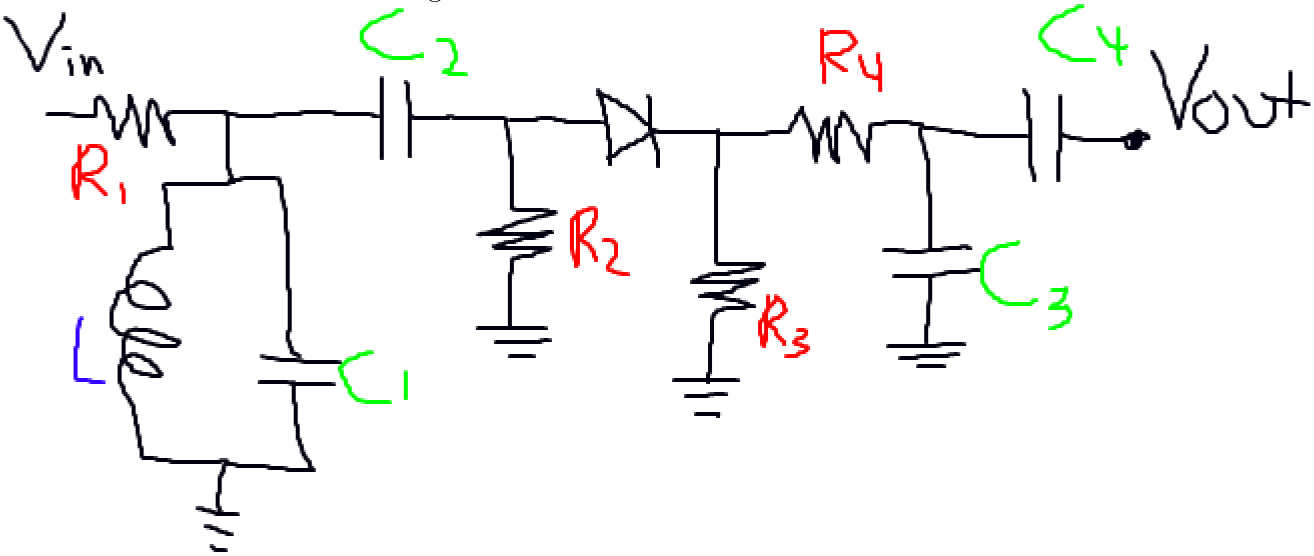
Building an Amplifed FM Radio and Measuring Noise

—–By David Galbraith—–

Abstract:
For this lab, we made an FM radio that took a frequency-modulated signal and turned it into music. Then, we made an amplifier that made the music louder. Finally, we used amplification to get measurements of Johnson-Nyquist noise, and we wrote Python programs to verify the Central Limit theorem and the change in standard deviation with increasing sample size. Along the way, we measured the velocity of propagation of electrical signals in wires, getting $1.9 * 10^8$ meters per second, and we measured the impedance of our cables at 50Ω.

Figure 1: FM Demodulation Circuit



$R_1 = 33\Omega; R_2 = 150\Omega, R_3 = 150\Omega, R_4 = 150\Omega, L = 1\mu\text{H}; C_1 = 22000\text{ pF}; C_2 = C_3 = C_4 = 10000\text{ pF}.$

Circuit: On the far left you see V_{in}, which is where the signals come in from the outside. Then to the right of that there is a bandpass filter consisting of L and C_1. A bandpass filter works by exploiting the resonance properties of the LC circuit. In an LC circuit, current travels in an oscillating manner, as charge builds up on one plate of the capacitor, then discharges through the inductor, creating a magnetic field that powers a continued current until the charge builds up on the other plate of the capacitor, and the process repeats itself in the opposite direction. These direction changes happen at regular intervals, so the system is a harmonic oscillator. The number of direction changes per second is called the "resonant frequency" of the circuit, and it can be calculated from the formula $\omega = (2\pi\sqrt{LC})^{-1}$, where ω is the resonant frequency, L is the inductance of the inductor in Henrys, and C is the capacitance of the capacitor in farads. If there is input current, then the system becomes a driven harmonic oscillator. If the input current is at or around the resonant frequency, it can easily join the flow of current through the LC circuit, so the circuit presents a low impedance to it. Since $V = IR$ by Ohm's law, this low impedance means that there will be a low voltage between terminal A and the ground below at near-resonant frequencies, so most of the voltage will go on to the rest of the circuit. On the other hand, at frequencies away from resonance, the filter presents a high impedance, since the input current will often be going in one direction while the LC circuit current is going in the opposite direction. As a result of this high impedance, there can be a large accumulation of voltage between terminal A and the ground below at non-resonant frequencies, so not much voltage will flow on into the rest of the circuit. The LC bandpass filter for this lab is centered at roughly 1 MHz using an inductance of 1 microHenry and a capacitance of 22000 picofarads, giving a resonant frequency of $(2\pi\sqrt{LC_1})^{-1} = 1073022$ Hz. Another relevant number for a filter is the bandwidth. The bandwidth gives a measurement of the range of frequencies the filter allows through. One half-bandwidth above and below the resonant frequency, the output of the filter is 70.7% of the input. More than one half-bandwidth above and below resonance, the output rapidly drops off; less than one half-bandwidth above and below, it is closer to 1. The bandwidth of an LC filter is given by

Also on the left side of the circuit but not a part of the bandpass filter, we have R_1 and C_2.

These components are used for biasing, which is when you optimize a signal to work with a certain device. R_1 decreases the amplitude of the input signal, because that is what resistors in alternating currents do, while C_1 puts a phase shift between the voltage and current of the input signal, because that is what capacitors in alternating currents do. These changes to the signal help it work properly with the diode. The value of C_1 was fixed *apriori* by the fact that 10000 pF was the only capacitance value we had available, having already used our only 22000 pF capacitor in the LC filter. We found a usable value for R_1 by trial and error, putting in every resistor value we had until the circuit worked. R_2 serves to discharge the input current when the diode is functioning as an open circuit; its value could have been chosen arbitrarily as long as it was big enough to actually discharge the voltage efficiently enough that the circuit does not catch fire.

The remainder of the circuit serves as an envelope detector. An envelope detector takes a high-frequency signal as input and sends to the output the envelope of that signal. The envelope of an oscillating signal is basically the curve given by connecting the peaks of each of its waves. This circuit works as an envelope detector because the diode only allows current to flow in the direction that it is pointing on the diagram. When current is flowing through the diode, it acts like a short circuit, so the output signal of the diode is equal to its input signal. At other times, its output signal is zero. A "textbook" envelope detector consists only of the diode, R_3, and C_3, so only consider those parts for the moment. When the diode is not acting as an open circuit, its output is a DC signal whose voltage equals the amplitude of the input voltage; under these conditions, this voltage accumulates on C_3. When the diode is acting like an open circuit, C_3 discharges through R_3, so V_{out} drops until the diode goes back to letting charge through. As a result of this process, V_{out} is an AC signal that measures the amplitude of V_{in}. That's why it's called an "amplitude-modulated signal". The combination of R_4 and C_4 acts as a low-pass filter on this signal, The additional component C_4 serves the purpose of biasing, modifying the input for optimal performance with our speaker system. As before, C_4 was fixed by our small selection of capacitors.