2023

PRACTICAL CIRCUIT DESIGN WITH DISCRETE ANALOG COMPONENTS

EXPLORING ANALOG CIRCUITS, AM/FM RADIO MIXING & DETECTION METHODS, FILTERS, & TRANSISTORS

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INTRODUCTION

Why study radio?

CHAPTER 1: EXPLORING THE AD831 MIXER BOARD

Introduction to Analog Signal Processing: The Mixer Stage

Before we examine the demodulation process in radio receivers, it is important to first understand how receivers selectively tune into a desired radio station while discriminating or blocking unwanted channels. The model we use here is from the Superheterodyne radio design. When a desired station is tuned in, the circuits in the "front-end" (antenna and tuner) of the receiver combine our RF channel with a new signal that is produced by a what is called a *local oscillator* (LO) inside the radio circuitry itself. The RF signal and the LO signal are then *mixed*.

Figure 1: AD831 Mixer board used in this experiment.

Mixing refers to the process of multiplying two frequencies together. This is also called *heterodyning*. Heterodyning converts a broadcast signal from its carrier frequency to a local fixed frequency, so the RF circuit components do not have to be tuned manually every time we change channels. The purpose of combining two RF frequencies is to create a new, IF (intermediate frequency) that contains the audio signal for any channel we choose.

This new IF frequency can be used to process *all* the incoming stations. In the early days of radio, users had to manually tune two or three variable capacitors to hear a station. By using this new IF, the entire FM band can be processed with the same circuitry and only one tuner knob is required. In fact, when we tune in our station, we are also tuning the frequency of the LO because they are both part of the tuner (variable capacitor). In this way, the LO frequency is always 10.7MHz greater than the radio stations operating carrier frequency.

Since FM covers 88.1MHz -108.1MHz, the LO must operate within the range of 98.7-118.7MHz. Special components such as tuning diodes allow the LO to be adjusted to the exact frequency so that the difference between the LO and RF signal = 10.7MHz. Figure 1 shows the board I used to demonstrate this process.

It turns out that when sinusoidal waveforms are mixed, we end up with four major frequencies at the output. We have the two fundamental frequencies we started with as well as two new ones called the *Sum* (RF + LO) and *Diff* (RF-LO) frequencies. Several harmonics of these are created as well. I used a function generator to produce a sine wave at 28.4MHz and another one at 17.7Mhz. These signals were connected to the inputs of my mixer board. The output of the board produced a *Sum* frequency of **46.1Mhz** (28.4MHz + 17.7MHz) and a *Difference* frequency of **10.7MHz** (28.4MHz – 17.7MHz). This is called *frequency translation*. All incoming FM stations are translated down to a center frequency 10.7MHz and processed by the IF amplifier stages of the circuit.

After mixing the frequencies, the IF is amplified many times and then filtered to attenuate the unwanted frequencies/channels. Only after this process is complete, does the receiver use another circuit that eliminates the IF itself and recovers our original music, voice, or data that we wanted to hear in the first place. This recovery of our audio is called *demodulation* or *detection*. The use of the "superhet" model *greatly* simplifies the complexity of modern radio designs because the amplifier stage of the receiver can be tuned to operate at just one frequency, the IF.

Mixing Signals with the AD831 Double-Balanced Mixer Board

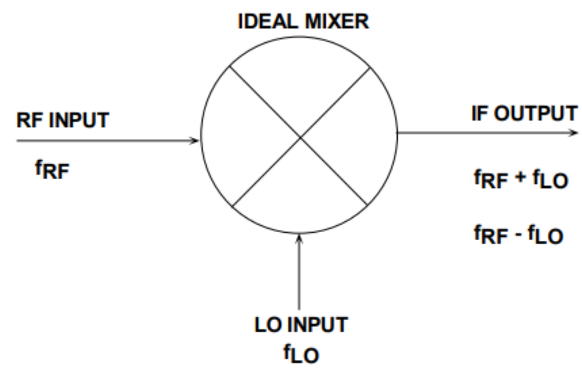

Figure 2: Simplified Mixer Model. I chose 28.4MHz for RF, and 17.7MHz for the LO.

The circuit that creates the new IF from the two fundamental frequencies is called the *mixer*. So, what is a mixer? Figure 2 shows a block diagram of a typical circuit with the input and output ports. Mixers are identified with a circle diagram with an "X" inside as shown. These devices have two inputs and one output.

STEP 1: The radio frequency (RF) that we wish to listen to is connected to the RF input (**f_{RF}**) of our mixer. This contains ***all*** the AM/FM/HAM band of radio frequencies our antenna is picking up, including the one we are interested in. The unwanted stations must be filtered out so we can listen to the specific one we want. To do that, the desired frequency we want is *frequency-shifted* to the IF described above. In AM, that IF is 455KHz and in FM it is 10.7MHz. **In this demo, I chose 28.4MHz as my desired channel or carrier frequency.** This is referred to as the RF that would normally be selected in the radio tuner circuit.

STEP 2: The radio has another circuit that works with the tuner to create some new frequencies, including our IF. The circuit that does this is called the *local oscillator*. Oscillators are often made from the combination of inductors and capacitors that can be tuned to a specific range of frequencies. These circuits produce a sinusoidal wave that is connected to the LO input of our mixer (**f_{LO}**). When we tune our radio to a specific channel, we change the frequency of the LO. **In this demo, I chose 17.7MHz as the LO.**

STEP 3: The mixing process involves multiplying the two signals together. The result is that several new frequencies are created. The two we are interested in are called the *sum* and *difference* frequencies. Figure 2 defines these frequencies as $f_{RF} + f_{LO}$ (sum frequency) and $f_{RF} - f_{LO}$ (difference frequency).

This process is more easily understood with a picture so let's examine Figure 3 and the scope image of my test circuit. The RF is a 28.4 MHz Sine wave (yellow), and the local oscillator input is 17.7MHz (blue). When the mixer circuit does its magic, the green wave is produced at a frequency of 10.7MHz. The green wave is the new IF that is used in FM.

The IF waveform is complex looking because it contains the two original frequencies we started with as well as the sum and difference frequencies described above. It also contains several harmonics. The important thing to understand is this new IF contains the music/voice we want. We just aren't ready to hook up the speakers just yet!

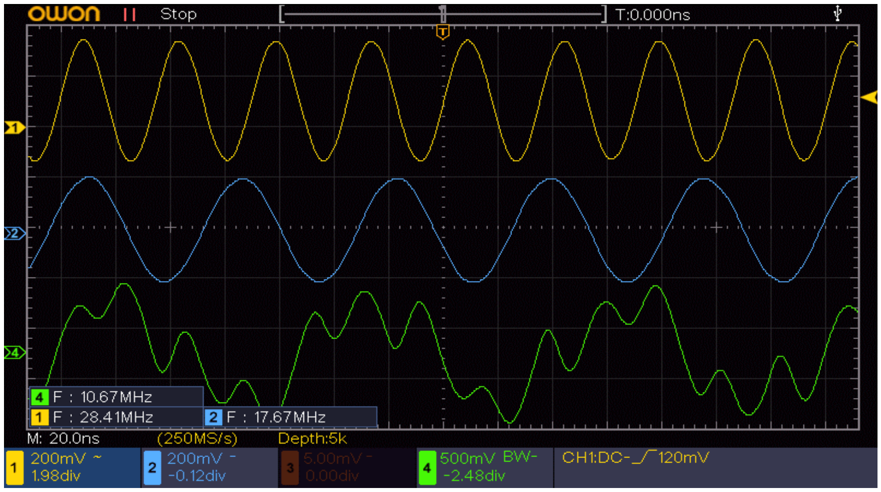

Figure 3: Scope Traces from RF (Blue), LO (Yellow), & Intermediate Output Frequency

STEP 4: After this IF is created, the signal is processed by the 1st IF amplifier circuit. A tuned transformer is used to pass only the frequencies near the IF (10.7 MHz) and it rejects all other frequencies. It is then sent to the 2nd IF amplifier which amplifies and cleans up the signal a bit more.

STEP 5: Our signal is getting close to being audible and contains only the channel we wish to listen to. But before it can go to the power amplifier stage, the IF must be removed so that only the music/voice we want will make it to the speakers. This stage of processing is called **demodulation/detection**. We will discuss that in a later section.

Examining The Output of the Mixed Frequencies in the Frequency Domain

Oscilloscopes are used in what is called the "*time domain".* This means the amplitude of signals are shown on the Y-axis and time is displayed along the X-axis. But when we want to understand the individual signals in complex waveforms, we need to switch to the "*frequency domain".* Modern scopes accomplish this by using FFT or *Fast Fourier Transform* to convert a signal from the time to frequency domain. Spectrum analyzers are much better at this as we will see shortly.

Notice the red trace in Figure 4 below. The spikes represent the frequency components that make of the complex orange trace (#3) in the top pane. Without FFT, we would not be able to discern the individual frequencies that make up the orange trace. My blue cursors measured the time between the start of the trace (A) and the first fundamental frequency at point B.

Notice, it calculated this to be 10.75MHz which is the expected output of my mixer board.

The other spikes represent the sum, difference, and harmonic frequencies that were produced in the mixing process.

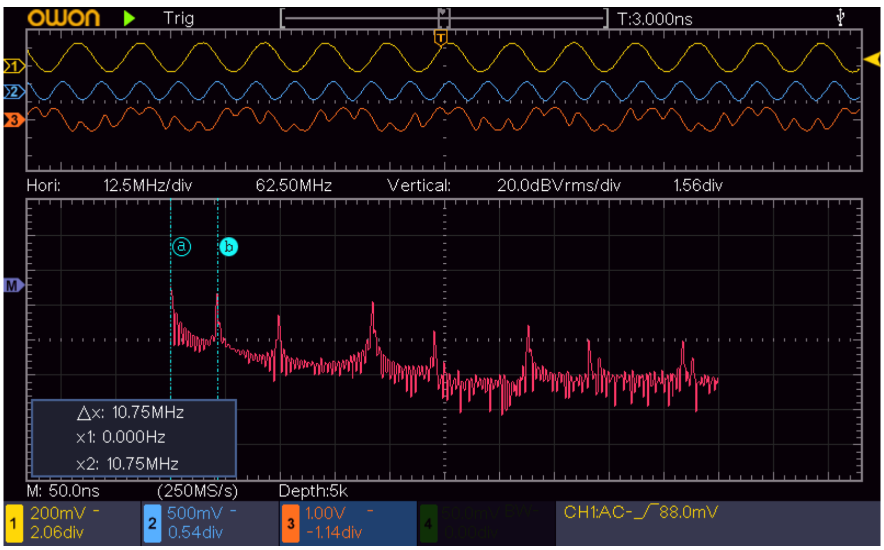

Figure 4: FFT scope function displays the fundamentals, sum, difference, and harmonic frequencies after mixing.

While the FFT function on the scope shows the discreet frequencies contained in the IF output, an instrument called a spectrum analyzer (SA) provides a better view into what is happening. The SA is the small device (*tinySA*) in the bottom of the image in Figure 5. It can be purchased [here].

Notice that the tinySA picks up many more harmonic frequencies than my FFT scope function does.

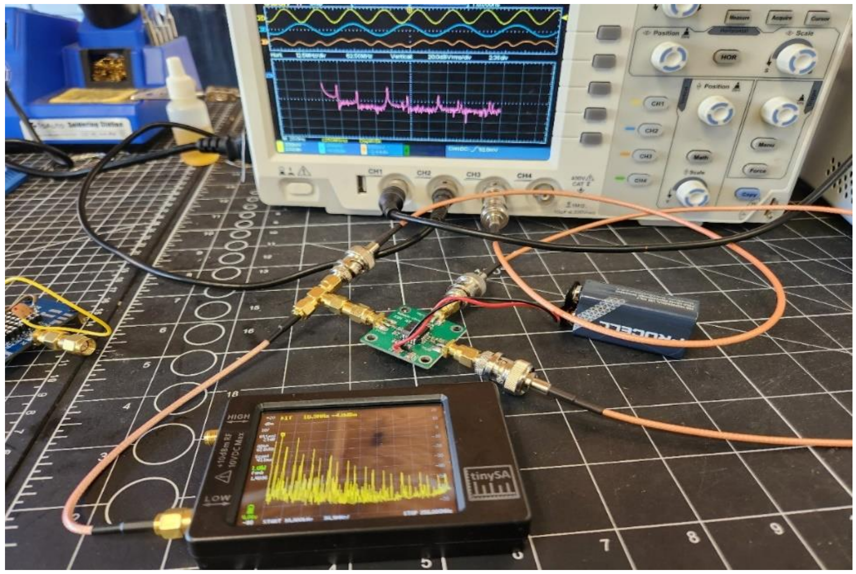

Figure 5: View of scope and spectrum analyzer showing the unfiltered intermediate frequency from the mixer output port.

For a closer look, check out Figure 6. I captured the screen of the tinySA. The output is very clean and more accurate than my scope. Also, the tinySA lets me move the cursor over and portion of the captured waveform to view its frequency. The height of the spikes represents the magnitude or strength of the signals. When we build filters, our goal is to make the unwanted "spikes" disappear or be of less magnitude. This is called *attenuation*. We can build filters that attenuate a band or pass a range of frequencies.

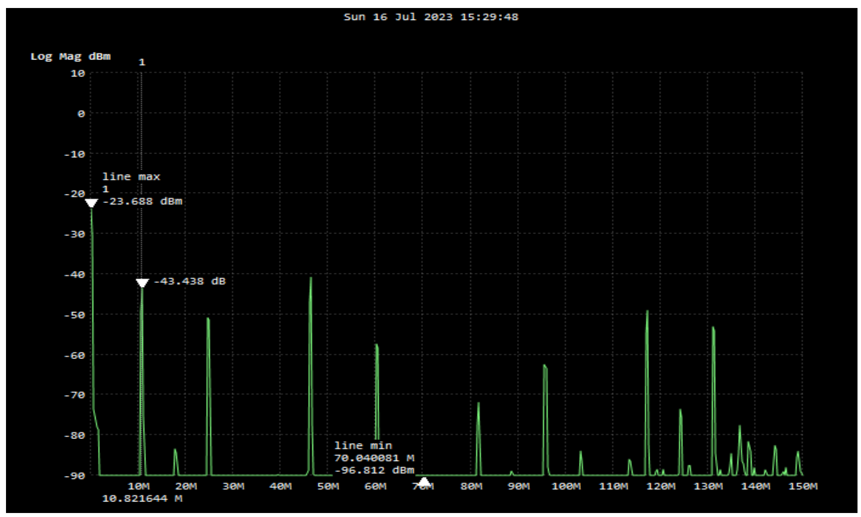

Figure 6: Here is the unfiltered output of my mixer board from the spectrum analyzer.

Spectrum Analysis of Filtered IF

Now I want to show you what happens when I add a cheap 10.7MHz ceramic filter to the output of my mixer board. This filter rejects or attenuates all frequencies above 10.7MHz and is called a low-pass filter. In FM radio, it is this IF frequency that we wish to work with at this point.

First let's check out the scope's FFT function. Figure 7 shows the 10.7MHz as a spike on the left end of the trace. Compared with Figure 4, you can see how the frequencies *greater* than my IF are reduced to microvolts. Only my desired spike is visible.

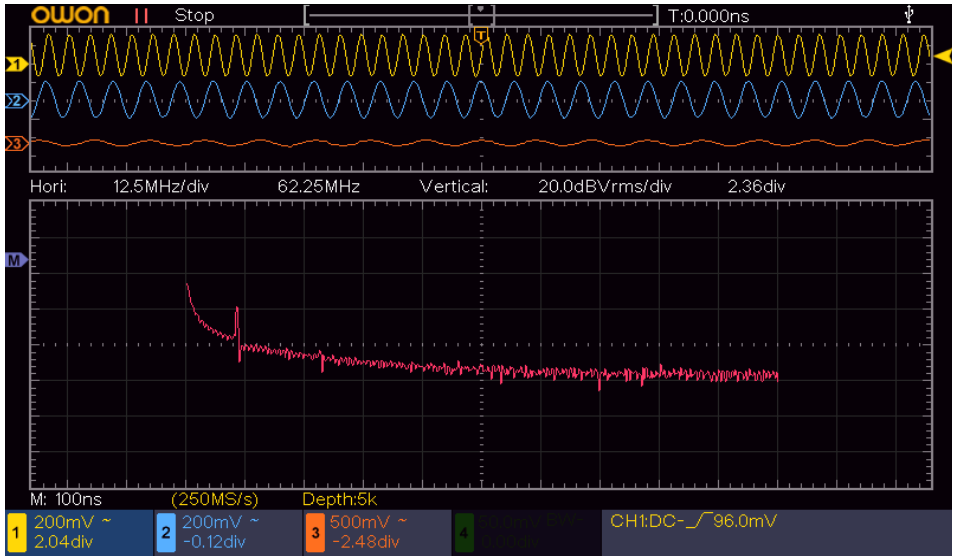

Figure 7: After filtering the output of my mixer board with a 10.7MHz ceramic filter only the 10.7MHz is passed through. The other frequencies are attenuated by the 2nd IF transformer (see step 4 above).

When we examine the mixer board output with the tinySA, we get the same results but with greater resolution. Figure 8 shows how the low-pass filter virtually eliminates the harmonics and other unwanted signals. However, you can still see a lot of unwanted frequencies. And even those these higher frequencies were greatly attenuated, perhaps we can do better.

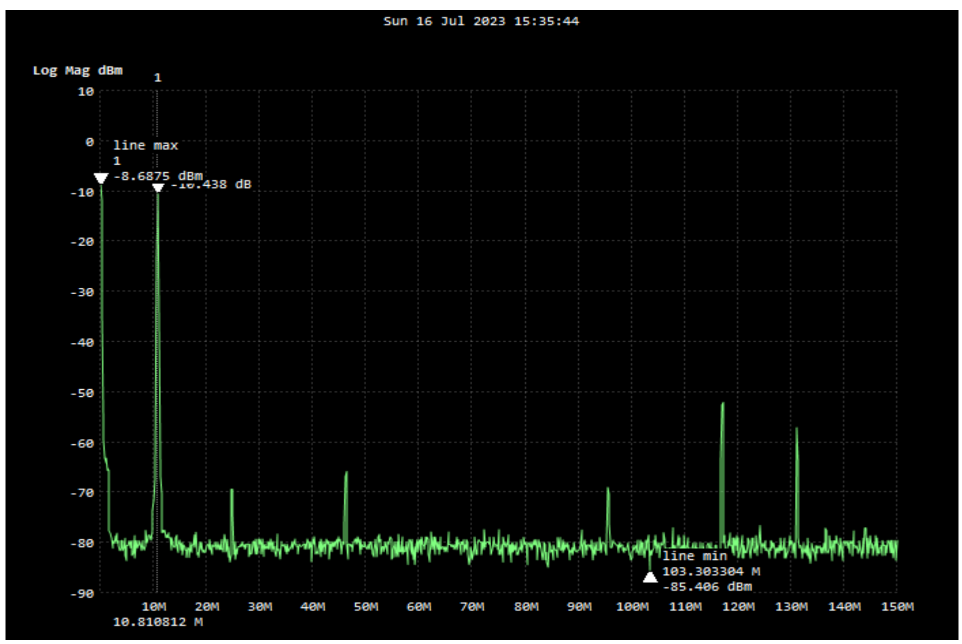

Figure 8: Filtered mixer board output shows 10.8MHz spike. The frequencies above that point are greatly attenuated.

Commercial Filter Evaluation

When I used the small ceramic filter, I could still see unwanted frequencies. So, I got to wondering what would happen if I tried a commercial filter. I purchased the 10.7MHz bandpass filter shown in Figure 9.

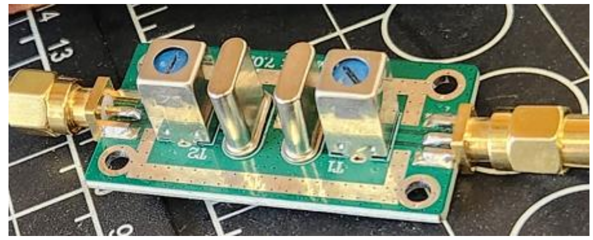

Figure 9: Commercial 10.7MHz filter used in this experiment.

I think you will agree that the results were terrific. Figure 10 shows the tinySA image of the output of my new mixer board. Now those high frequencies appear to be completely blocked. Keep this in mind when you purchase capacitors. Cheap ones have more resistance and inductance in them than the more expensive "*audio quality*" caps. One of the reasons the commercial filter board is so good is that it uses crystals and tuned transformers. Quartz crystals make great filters.

When you build your own filters keep in mind that the length of your leads and the quality of your components are very critical when working with RF signals. Also, you should not use breadboards for high frequency work! They have a ton of stray capacitance and will disrupt your results. Also, be aware of using a scope probe around tuned circuits because the capacitance of the probes themselves will not let you accurately view your signals. This is especially true for front-end measurements around the antenna or tuner in a radio.

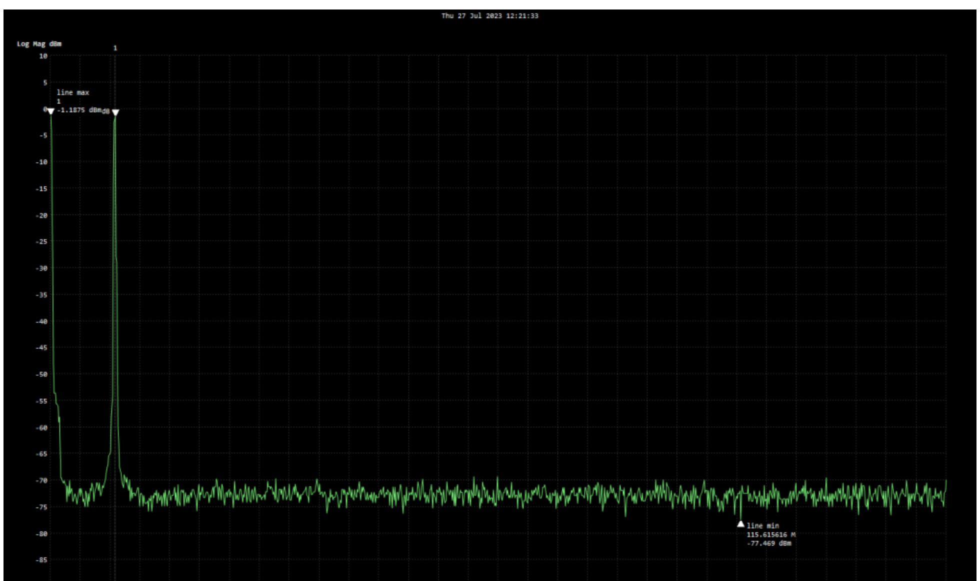

Figure 10: Results of commercial filter shows only the IF frequency is passed and all other signals are eliminated.

CHAPTER 2: Getting to Know R-C Filters

Introduction

Since we have been alluding to these cool filters, it is time to learn how we can use them to our advantage. This chapter will give you a small taste of how useful filters are. For example, when you try to read analog sensor data in industrial and automotive environments, eliminating unwanted electrical noise is a real challenge. Engine noise, alternator whine, fuel injector pulses, large 3-phase motors, and ignition spark all contribute to the collection of unwanted electro-magnetic interference (EMI) that plagues our sensor readings. These unwanted frequencies degrade and mask the sensor data we are attempting to collect, interpret, and display to the operator. The data must be robust. Safety systems and collision-avoidance sensors cannot simply work most of the time, for example. They must work all the time.

This chapter will demonstrate how simple resistors and capacitors can be combined to pass certain frequencies while blocking others. R-C filters are ubiquitous in electronics and must be understood before we learn how to use them in our circuits. If you are not familiar with filtering, I strongly recommend you read my paper on this subject before continuing. Let's not get ahead of ourselves here. You see, before we learn how to create and use filters, it would behoove us to get a basic understanding of how the electronic components we use for filters actually work. So, let's jump in!

Getting to Know Our Filter Components

Resistors

First, we will examine resistors. A resistor, as its name implies, resists something. In this case, they resist the flow of electrons in an electronic circuit. Due to the way they are made, electrons have difficulty moving through them. The flow of electrons in a conductor (like a copper wire) is what we call current. Current is signified by the letter "I" and is measured in amperes or just amps. The more current that flows in a circuit, the more work it can do for us. Imagine a small motor attached to a 9v battery. The small battery can only supply so many electrons because of its size. Consequently, I can only power a very small motor like one used on a robot. To run a larger motor, I will need a much larger battery because I need more current.

Figure 11: Typical Carbon Resistors

Sometimes, however, we need to cut down the voltage to a particular device. Say I have my 9v battery and I want to run a 5v light bulb. If I don't reduce the power going to the bulb it will overheat and burn out. So, I could use a resistor or two to divide the 9v down to 5v so the bulb will work properly.

Resistors (R) are made of various materials. Many are made of a mixture of carbon and other ingredients. The amount of carbon will determine how many electrons are blocked. Resistors are made in many different values of resistance. Their values are expressed in ohms (Ω). The definition of an ohm goes like this: 1Ω is the amount of resistance in a circuit with 1 volt and 1 amp of current running through it. We calculate R using Ohm's Law: R = Voltage / Current. Since voltage is referred to as "E" and current is referred to as "I", our formula becomes $R = E/I$. In

Figure 1, you can see that the resistors are identified by colored bands around them. The color pattern tells us their resistance value.

Finally, to give some perspective on this, when R = 1,000,000Ω that is a huge amount of resistance. On the other hand, a resistor of 10Ω is very small. If you connect an LED to a battery, you would typically use a 220-470Ω resistor to limit the current to keep the LED from overheating and burning out. You need to know that resistors convert electrical energy into heat. So, they come in different sizes (½ watt, 1 watt, etc. so they can withstand that heat.

Next, we need to understand a few things about capacitors.

Capacitors

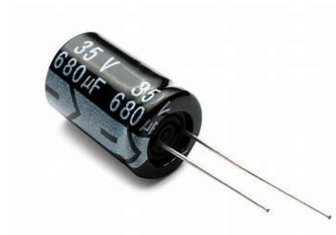

Figure 2:Typical Electrolytic Capacitor

Figure 2 shows a typical capacitor. This one has a positive (+) and negative (-) lead and must be connected correctly. Many other capacitors are not polarized and can be connected in either direction. When we apply a DC voltage to a capacitor from a battery it will begin to charge up. That means electrons will be stored in the plates of the capacitor. After it is charged it functions like a small battery. Please remember that capacitors store electrical charges. Resistors do not store any energy.

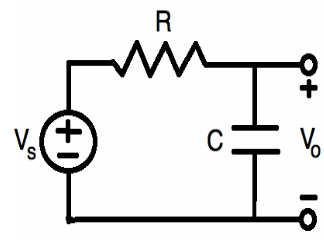

Figure 3A: Simple R-C Circuit

When you connect a capacitor to a battery, it charges up almost instantly. However, there many times when we want to control the charge and discharge rate of a capacitor. For example, electric vehicles use a pre-charge circuit that slowly lets voltage build up on the motor before the driver accelerates so it doesn't burn the motor and other components. Figure 3A shows such a circuit. There is a resistor that is in series with the capacitor. The resistor is used to slow down the flow of electrons from the battery as they enter the capacitor. This slows down the charging time of the capacitor. This type of circuit is also used in electronic filters as we will see shortly.

I want to leave you with a visual image of how this charging works. If I connected a capacitor across the battery and an LED or bulb across the capacitor as in Figure 3B, the LED would glow full bright as soon as I pushed my switch (S1) because electrons can flow from the battery, through R1 and through the LED and then return to the battery. In this condition, the LED is being powered by the *battery*. While S1 is closed, some of the battery's energy flows through C1 and it begins to charge. At some point, the capacitor cannot store any more electrons, so it becomes an energy storage device.

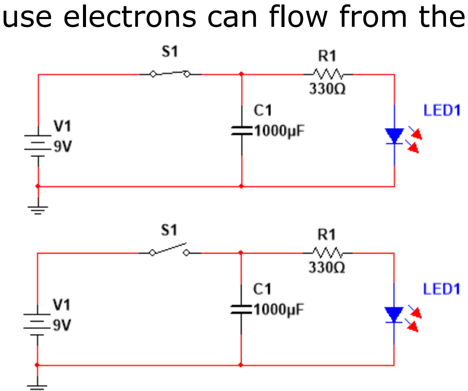

Figure 3B: Simple Capacitor and LED Circuit. When S1 is opened, C1 powers the LED for a second or two until its stored energy is depleted.

If I were to remove the battery power from the circuit by opening the switch, the stored energy in C1 will be discharged or returned into our circuit and current will flow through the LED and back to C1 just as it did when powered by the battery. While C1 has a charge, it will keep the LED glowing until the charge in C1 becomes 0V. So, you would notice that the LED would glow for a short time and then slowly dim out as C1 loses its charge. The size of the capacitor determines how long it takes to become fully charged and discharged. Keep in mind that when electrons/current flows we can do work. Otherwise, we cannot.

This circuit could be used to dim the interior lights of your car, for example. When you open the car door, the battery feeds current to the interior light and charges up a capacitor. When you close the door, the battery is no longer supplying power but the capacitor is. It keeps the lights on for a short time. But, since the capacitor can only store just so much energy, its voltage decreases and the lights slowly dim. Simple. No need for a computer or software. You can just use some discreet parts to do the job. You can choose how long to leave the lights on before they dim out by choosing the resistor-capacitor values. This type of circuit is used in millions of products and that is why I want you to understand how they work. I am sure you will find many uses for these components as you study electronics.

R-C Charging over time

Let's look at the scope image of this charging process. Without the use of a resistor in our circuit, the capacitor would charge almost instantaneously as current rushes into it. Without a charge on it, a capacitor acts like a piece of wire… a short circuit. For an instant, current rushes in. As the capacitor charges, the voltage increases and the current decreases.

We have seen how resistors are used to control current flow. By adding the resistor in series with my capacitor (C) in Figure 3A, it slows down the flow of the current and creates a delay so it will take a bit of time to charge the capacitor.

Figure 4 shows how an R-C circuit controls the charge time. This image displays the charge curve for a 2200Ω (2.2k) in series with 1000µF (microfarad) electrolytic (polarized) capacitor.

When I connect the battery, current begins to flow from the resistor into the capacitor. With the values I have chosen, it takes about 10-11 seconds for the capacitor to fully charge. The cool thing is that you can choose how long it takes for this charging to take place by simply changing the value(s) of R, C, or both. There are many online calculators to help you. Let's look at this in more detail.

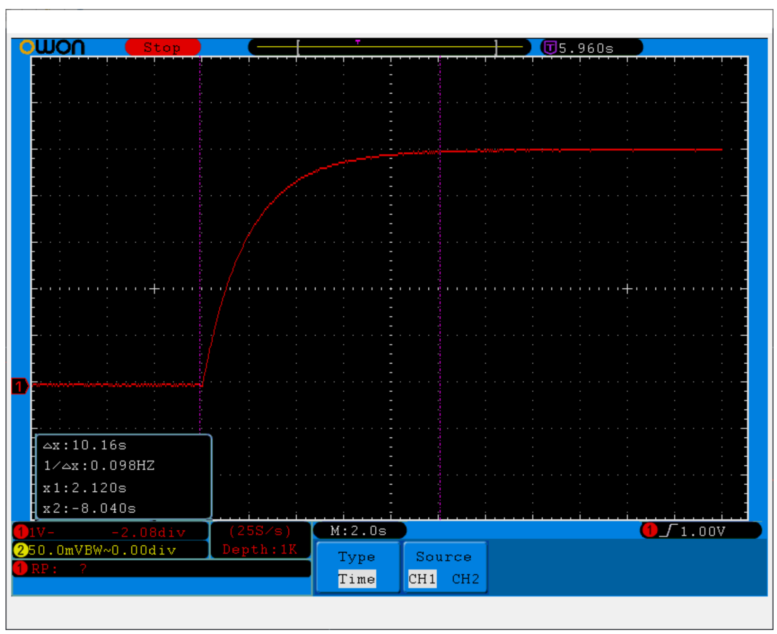

Figure 4: Voltage Increases from zero to near battery voltage in 5-time constants (2.2 x 5 = 10-11 sec).

RC Time Constant

To determine how long it takes to charge a capacitor in a DC circuit you simply multiply the resistor value times the capacitor's value. The result is called the *RC Time Constant*. It is symbolized by the Greek letter Tau (T). The formula is:

$T = R\ x\ C$ where T is in seconds, R is in ohms and C is in Farads. A farad is a huge quantity so capacitors are created with values in microfarads (µF), nanofarads (nf), or picofarads (pf). For reference, 1000µF = .001Farards or 1,000,000nF or 1,000,000,000pF.

Example: The RC time constant for my circuit is 2200Ω x .001F = 2.2 Seconds. This means that after 2.2 seconds, my capacitor will be charged to about 63% of the applied voltage. In this case, it is based on the voltage of my battery. During the next time constant from 2.2 – 4.4 sec., it will charge another 63% and so on. It takes 5 RC time constants to fully charge the capacitor so in my case it takes about 11 seconds (5 x 2.2 sec). When the voltage in the capacitor reaches 99% of the applied or battery voltage, we say it is fully charged.

If I removed the battery in Figure 3A and replaced it with a switch, I could discharge the capacitor by closing the switch and allowing current to dissipate through the resistor. It would take the same amount of time to discharge the capacitor through my resistor as it took to charge it in the first place. So, after 11 seconds, the capacitor would be out of energy.

Remember I said the capacitor is like a small battery? Well, the electrons that were stored in the capacitor's plates now flow through the resistor and any component connected to the circuit. The voltage slowly diminishes until it reaches 0 volts. At that point, the capacitor is fully discharged.

On the oscilloscope image in Figure 5, you can see that the voltage goes from 9V (battery voltage) down to zero at the same rate as when the capacitor charged. The voltage exponentially decays until it reaches 0 volts. At this point there is no more energy stored in the capacitor. Again, it takes 5-time constants to discharge. In our case that is 2.2 x 5 ~ 11sec.

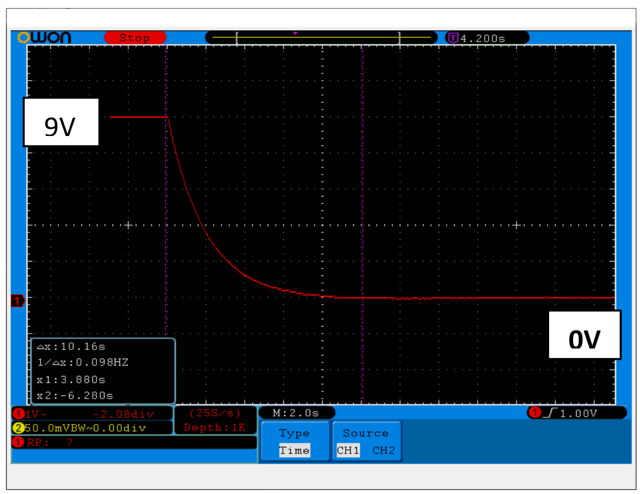

Figure 5: Discharge curve for the same capacitor.

Introduction to R-C Filters

Now I want to show you how filters work. By combining resistors and capacitors (or resistors and inductors/coils), we can filter out unwanted frequencies or EMI noise. This is very handy when we want to filter out interference that is disrupting the readings of a sensor. Whether we are working with a microcontroller, automotive application, or a musical device, these filters are crucial when we need clean sensor signals or clean sound.

This circuit works on the principle that capacitors react differently to low frequencies than to high ones. In other words, their resistance to current flow is inversely proportional to frequency. At low frequencies, their resistance is high and as frequencies increase, their resistance decreases. Resistors do not react to changes in frequencies. Only capacitors and inductors do. When capacitance/inductance *and* frequency are considered together, the resistance to current flow is called *reactance* and it is designated as X_c. The reactance of an inductor is designated L_c. Thus, capacitors and inductors are called *reactive components* because of their unique response to changes in frequency. Now let's look at a simulation in National Instrument's Multisim application to see how we can take advantage of these reactive devices.

First, we will explore low-pass R-C filters. These circuits allow low frequencies to pass through but limit or block high frequencies. This is the circuit that can help us remove high frequency noise in our microcontroller power supply circuits, for example. For this demonstration, I will apply an AC signal to the filter network to simulate analog sensor data. Let's see how our filter reacts to changing frequencies!

Circuit Simulation for a Low-Pass Filter in Multisim

Our Circuit Setup:

- For this demo, we will use a 1000Ω (1K) resistor and 15nF (.015μF) ceramic capacitor to filter out high frequencies.
- Source is an Input Signal: 10V P-P Sine wave (red Trace on scope)
- Output of RC filter shown in yellow trace on scope.
- Figure 6 shows my RC test setup in the simulator.
- Figure 7 is a simple schematic of the circuit.

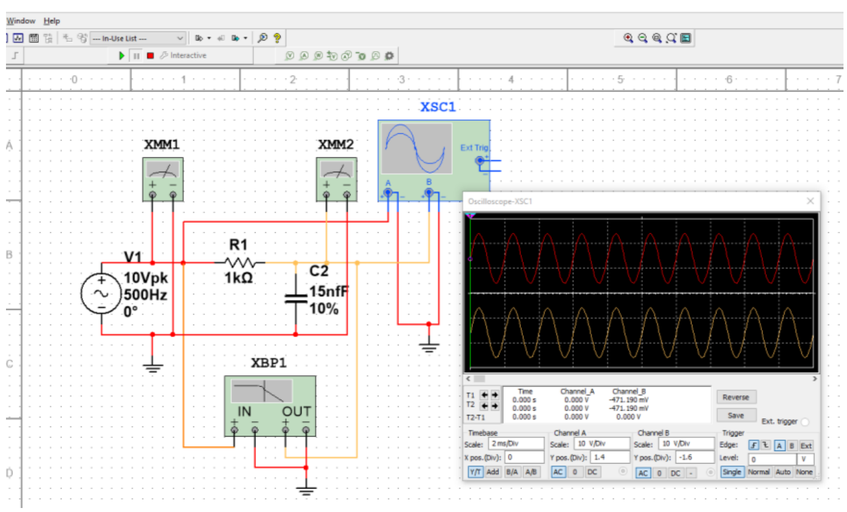

Figure 6: Multisim Layout of My RC Low pass Filter Circuit

Let's figure out how the circuit affects my output signal. At low frequencies, the reactance(resistance) of C is high. Recall the reactance of a capacitor and frequency are inversely related. So a low frequency sound "sees" the resistor as an easier path to the output because the capacitor has very high reactance.

When the frequency is high, C provides an easier path for the electrons to travel and they are routed to ground (B-) and are not passed to the output. That is because the “resistance” in C is less than the resistance in R. This circuit could be used as a Bass control on an amplfier. Low frequencies pass through and higher frequencies are blocked.

To reverse the filter design, just reverse the position of R and C. Then high frequencies would be passed and low frequencies would be blocked. Thus, you would have a *high-pass* filter.

As you will soon see, we can set the threshold by which low frequnciess are passed and high frequencies are blocked. The terms “low” and “high” are arbitray and determined by the circuit designer. These circuits are used in speaker crossover networks to route high sounds to the tweeter and low sounds to the woofer or sub-woofer.

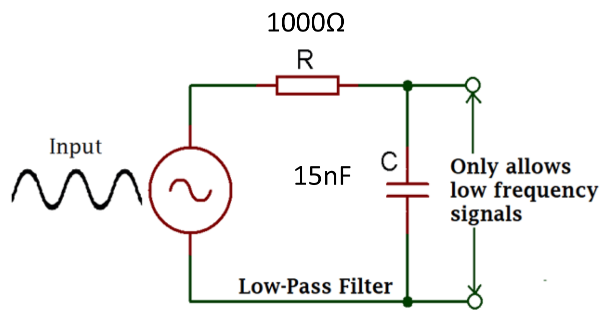

Figure 7: Simple R-C Low-Pass Filter Circuit

Computing Output Voltage from a Low Pass RC filter at Specific Frequencies

Example 1: Calculate the output voltage of the filter at a frequency of 1000Hz.

1. Compute capacitive reactance at 1000Hz:

$$X_c = \frac{1}{2\pi f c} = 1 \div 2\pi \times 1000 \times (15 \times 10^{-9}) = 1 \div 6280 \times .000000015 = \underline{\mathbf{10{,}615\Omega}}$$

2. $V_{OUT} = (V_{IN} \times X_C) \div \sqrt{R^2 + Xc^2} = 10 \times 10{,}615 \div \sqrt{1000^2 + 10615^2} = \underline{9.95V}$
   The yellow scope trace shows ~9.92V output from the filter (Figure 8).

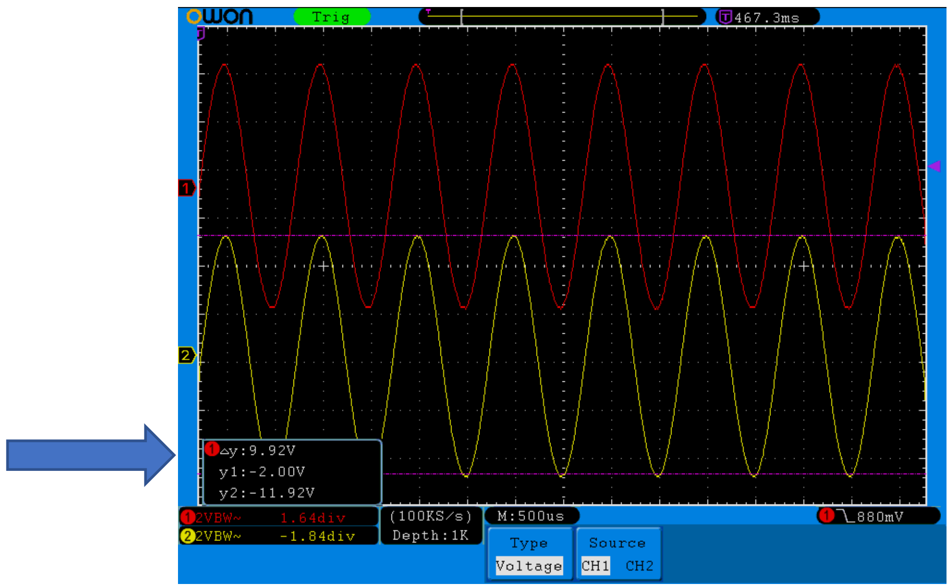

Figure 8: Input & Output Signal at 1KHz

Example 2: Calculate the output voltage of the filter at a frequency of 20,000Hz.

3. Compute capacitive reactance at 20,000Hz:

$$X_c = \frac{1}{2\pi f c} = 1 \div 2\pi \times 20000 * (15 \times 10^{-9}) = 1/ \times .000000015 = \underline{\mathbf{530.78\Omega}}$$

4. $V_{OUT} = (V_{IN} \times X_C) \div \sqrt{R^2 + Xc^2} = 10 \times 530.78 \div \sqrt{1000^2 + 530.78^2} = \underline{\mathbf{4.67V}}$

The yellow scope trace shows ~4.64V output from the filter (Figure 9).

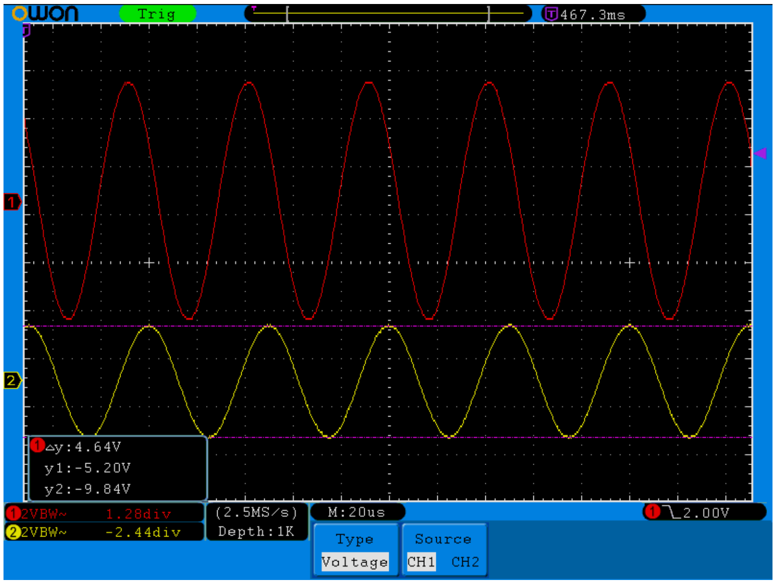

Figure 9: Input & Output Signal at 20KHz

Example 3: Calculate the CUTOFF frequency of our filter.

We have learned how the filter responds to changes in frequency. As the frequency of our input signal increases, the reactance of the capacitor declines. This allows more current to flow through the capacitor to ground which reduces our output voltage.

The corner/cut-off frequency (f_c) formula is $f_c = \frac{1}{2\pi RC}$

5. Using our 1000Ω resistor and 15nF capacitor filter, compute f_c:
6. f_c = 1 ÷ 6.28 x 1000 x .000000015 = **<u>10,615Hz.</u>**

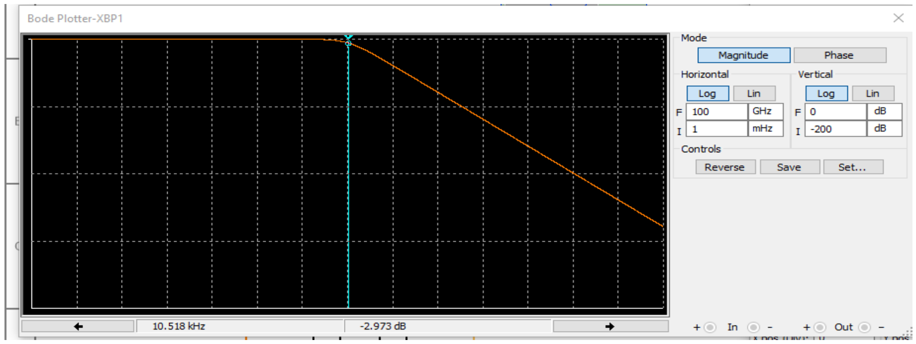

Figure 11: Bode Plot showing the corner frequency of 10,518Hz at ~-3db

You can see in Figure 11 that the cut-off frequency at -3dB is 10.518KHz which is close to our calculations. The actual tolerances of our components and the calibration of the instruments account for the discrepancy in the calculated and observed values.

When the resistance of the resistor and the resistance(reactance) of the capacitor are equal, the output of the filter voltage is 70.7% of the input voltage. Or, the output is reduced by about 30%. This is equivalent to a -3dB decline in output voltage.

The -3dB level was chosen by electrical engineers because it is the point where the output power level is 50% less than the input. This is used by convention in many applications and is often referred to as the ***half-power* level**. If you are familiar with voltage dividers, you know that when two resistors of equal value are in series with a voltage, the output across R_2 will by ½ the input voltage. When we compare two power levels where P_{out} / P_{in} = .5, this corresponds to a -3dB loss or attenuation in the power level (dB_{out} = 10Log (.5) = -3dB). But is our case, we are *not* measuring power. We are measuring voltage across a resistance and a reactance. Therefore, the associated power ratio is equal to the square of the voltage ($P = V^2 / R$) so we use a different formula. **The formula for comparing two voltages or two currents in dB is:**

$$dB = 20\text{Log}_{10} (V_{out} / V_{in})$$

The decibel scale is logarithmic and when the output voltage is compared to the input voltage ***at the cutoff frequency***, output voltage is always .707 x V_{in} .

When the dB is negative, it means the output is less than the input signal. This cut-off frequency defines the boundary between frequencies passed and frequencies blocked. The frequencies beyond the cut-off are attenuated significantly (Figure 6). In a single pole/stage RC filter, the output voltage decreases at about 20dB/decade, where a decade represents a 10x increase in frequency. For example, Table 1 shows that at 10KHz, our output voltage is 7.28V and at 100KHz it is 1.06V which is -19.5dB. Notice the roll off is linear past the cutoff frequency.

Remember, in our RC filter, we only have one reactive component. Namely the capacitor. The resistor does not react to frequency changes. When the frequencies are low, the reactance of the capacitor is very high. The capacitor acts like a super high value resistor. No current goes through it. Our signal goes through the resistor. But, when frequencies increase, the reactance (resistance) of the capacitor greatly diminishes.

This now provides an easy path to ground. So, the electrons go to ground and very little voltage is measured at the output of the filter. Also, the RC combination consumes some power. That is why the output voltage of a passive RC filter is always lower than the input. This type of filter does not amplify a signal.

If you want to amplify a signal you would use an active filter. Active filters are built from operational amplifier integrated circuits and would be used in audio and Ham radio applications where the signal will drive speakers or headphones.

CHAPTER 3: Getting to Know High Pass R-C Filters

Capacitive Reactance in a High Pass Filter

The circuit in Figure 3-1 is called a high-pass filter because it allows high frequencies to pass and blocks low frequency signals. This is just the opposite to the low-pass circuit we have been working with. As we have discovered, both the resistor and capacitor resist current flow. But only the capacitor reacts differently to changes in frequency. That is why we call its "resistance" *capacitive reactance (X_c)*.

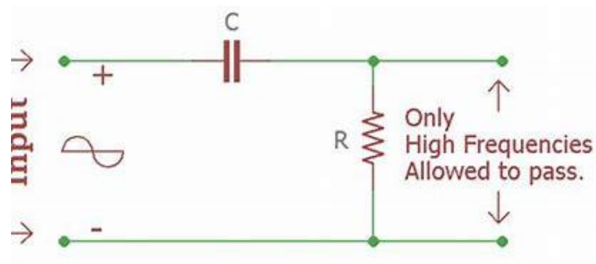

Figure 3-1: Simple R-C High-Pass Filter Circuit

If you have worked with voltage dividers, you can see how the resistor and cap form a voltage divider network. The difference is *this* divider responds to changes in frequency. Thus, we have a *frequency-dependent voltage divider*. In our circuit, only the capacitor reacts to changes in frequency and so we have what is called a *single-pole* high-pass filter.

Simulation of Our Circuit in Multisim

Our Setup:

- We will use our same components we have been using: the 1K resistor and 15nF (.015µF) ceramic capacitor. But this time the signal is connected in series with the capacitor, not the resistor.
- AC Input Signal: 10V P-P Sine wave (Red Trace on scope)
- Output of RC filter shown in yellow trace (Figure 18).

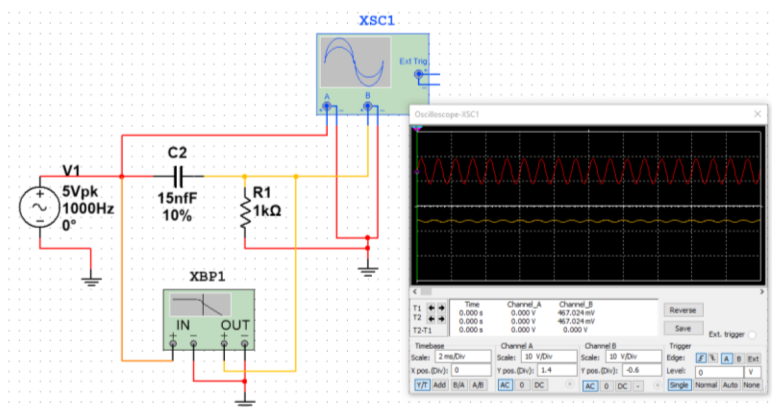

Figure 3-2:High Pass Filter in Multisim (5V Pk = 10V P-P)

Notice I have redesigned the circuit by swapping the positions of the resistor and capacitor. Now, the input signal is connected to the capacitor. Since the reactance of the capacitor at low frequencies is very high, it filters out those low signals. In Figure 3-2, the input signal is 1000Hz. Here the output is opposite to that of the low pass filter. Notice how the output signal (yellow) in Figure 3-2 is extremely small. This is because it is resisting the low frequencies in the signal. Again, just the reverse of the low pass filter design. Swapping the components is all that is needed to make a high or low pass filter. Let's see what happens as we increase the input frequency.

Now the input signal is 30,000Hz. Figure 3-3 shows the yellow output is nearly the same voltage as the input. Again, just opposite of the low pass circuit.

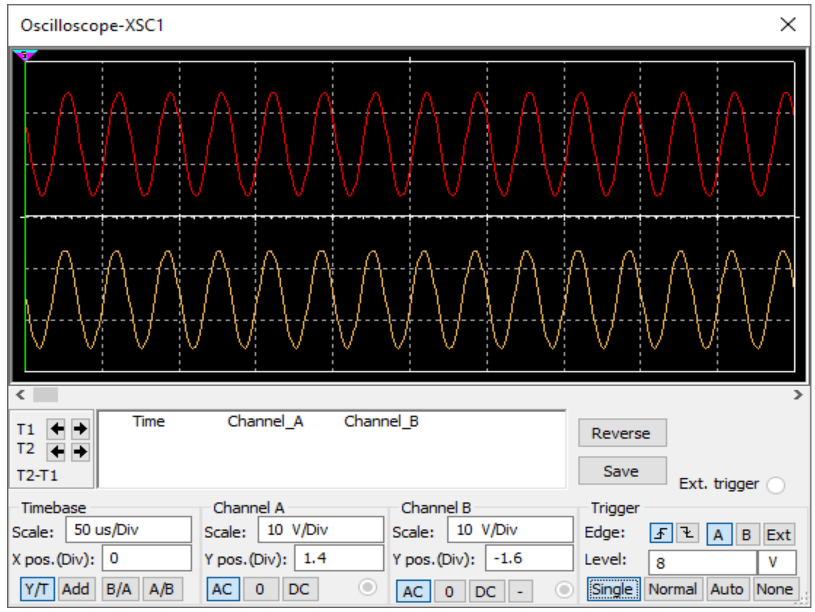

Figure 3-3: High Pass Filter with 30kHz Input Signal

The Bode plot in Figure 3-4 shows the characteristics of our filter. When the input signal approaches the cut-off frequency, only then are those higher tones allowed to pass.

However, notice we still have a gradual ramp. To change that we can build a 2-stage high pass filter. We will see that soon.

At higher frequencies, capacitors cannot charge and discharge fast enough to keep up with the sine wave. Their reactance is low and they appear as a short circuit or a wire to the rest of the circuit.

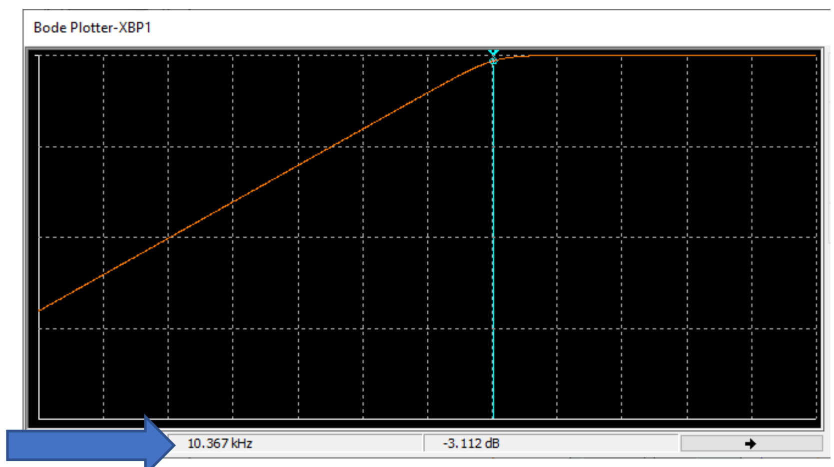

Figure 3-4: Bode Plot for High-Pass Filter. Cut-off Frequency at -3db is ~10,367Hz

Remember, the resistor-capacitor values not only determine the cutoff frequency, but they also determine how fast the capacitor charges and discharges. The cut-off or corner frequency occurs when the resistance of the resistor and the reactance of the capacitor are equal.

CHAPTER 4: Exploring the AM Radio Detector Circuit

Figure 12 below shows a basic detector circuit. I am mixing a 100KHz RF carrier sine wave with a 1KHz AM modulated sine wave at P1 with a signal generator. This simulates the mixer board we discussed in chapter 1. When the AC modulated sine wave signal swings positive, D1 begins to conduct. It takes 0.3V to turn on or forward bias D1. The output of D1 contains only the positive half of the waveform in the same way diodes are used to rectify AC to DC.

Figure 13 shows the scope traces from the test points in the simulation. The red trace is the 100KHz RF or carrier wave and the output of D1 is shown by the yellow trace from PR2 in the envelop detector network. This network uses a low-pass filter (C1 & R1) to smooth out the rectified output of D1.

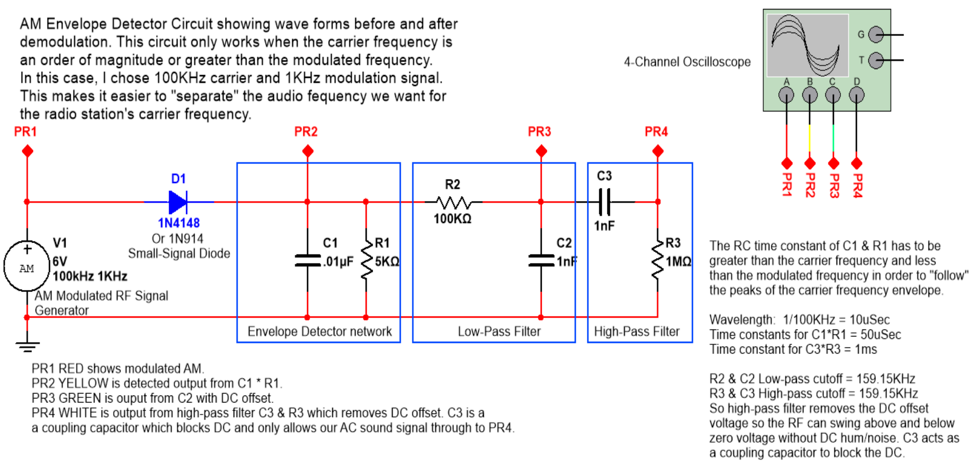

Figure 12: Proto-typical design of an AM detector where the positive half of the modulated waveform is used to recover the audio.

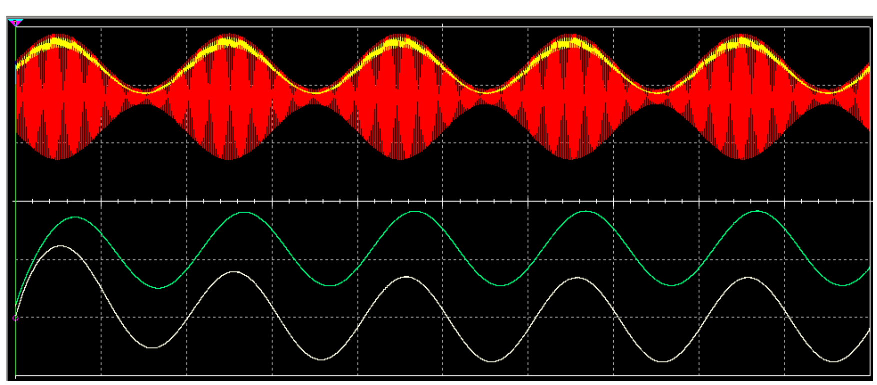

Figure 13: Scope image of the RF carrier (red) at P1, the recovered audio information (yellow) at P2, the green trace at P3 is the output of the high-pass filter, & the final audio signal (white) at P4 after DC offset is removed.

CHAPTER 5: Connecting Our Speakers

Connecting my AM Detector Circuit to an Amplifier


Figure 14: Image of my Class-D audio amplifier.

Now, I want to show you how to connect an AM detector circuit to an audio amplifier so we can evaluate the quality of the sound produced by the detector. The challenge is, when you take a fully functioning circuit and connect it to another circuit or another stage of your project, problems often arise. In my case, when I connected the detector to my small Class D amplifier, it sounded distorted and not very loud. This is caused by the amplifier ***loading down*** the detector circuit. That just means that the amplifier has a low input impedance and is trying to pull a lot of current from my detector. So once again, I fixed something that was not broken! To fix this, I added another stage between the detector and the amplifier to make them play nicely together. Let's see what I came up with.

The detector circuit cannot supply enough current to the amplifier. So, I need help. I decided to try using a bi-polar junction transistor configured as an "**emitter follower**". I could have also used an operational amplifier to do the same thing. I chose the common 2N3904 NPN transistor. The BJT will function as a "go between" or *buffer* between the two stages of my circuit. The EF has a high input impedance and a low output impedance. This means it can drive a heavy load (in this case the amplifier) without affecting the detector/input circuit. This way, the AM detector can work as it was designed and the EF stage will not diminish its amplification or filtering functions. Sounds like that is exactly what we need! Let's see how to set this up.

Requirements

- Supply voltage (Vcc) = 6V.
- The EF should draw a minimal amount of power.
- The circuit should not amplify the input signal. The amplifier will handle that.
- We want our AC signal to be accurately reproduced at the output of the transistor. We will take the output of EF from the emitter and not the collector. If we take it from the collector, the input and output waveforms will be amplified and be 180° out of phase. Since I am working with audio, I want the signal to mirror the input.
- The input and output of the EF will be coupled via a capacitor to block any DC.

Characteristics of Emitter Follower Configurations

- It has HIGH input impedance.
  - It won't load down the source/input signal from the previous stage.
- It has LOW output impedance.
  - This means it can "drive" heavy loads such as a speaker, relay, or amplifier.
- It has no amplification. This is called "unity" gain. The gain of the circuit = 1.
  - It is often used to act as a buffer to connect two circuits together.
  - It is used to isolate two stages of a circuit, so the second stage does not interfere with the output of the first stage.
- The EF only works if the transistor is always "ON".

- The V_{base} must be 0.7V above the $V_{emitter}$ to keep it on.
- $I_{emitter\ (emitter\ current)}$ = 10mA maximum to avoid exceeding current rating of 200mA for the 2N3904.
- The EF can only "source" or supply current. It cannot "sink" current. This means it acts as a source of power and not a source for ground or 0 Volts.

Exercise: Biasing an NPN BJT as an Emitter Follower (EF)

1. Refer to Figure 15 & calculate the value of the emitter resistor (RE). We want Collector-Emitter current to be 10mA (.01 amps) and Vcc= 6V. The value is:

   $R_E = \frac{6}{.01} =$ **600Ω.**

   I have a **560Ω** on hand so I will use that. The parts we use have a tolerance of 10-20% so you don't have to be exact. For example, I=E/R, $\frac{6}{560}$ = .01 amp. (10mA) so we are good to go.

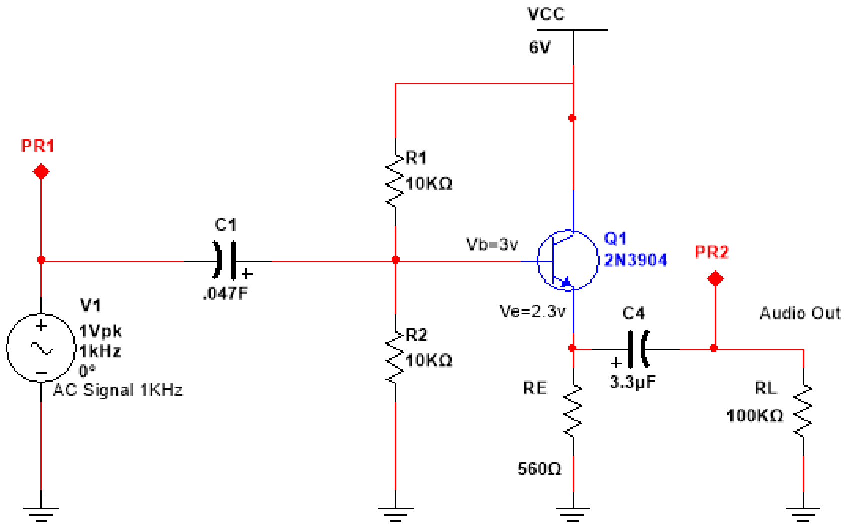

Figure 15: Typical Emitter-Follower circuit used to drive audio amplifier from AM Detector output.

2. Set the DC operating point of the transistor's base (input) so it will always be turned on. Since we are dealing with an AC signal, the radio frequency will swing positive and negative many times per second. So, we need to make sure the transistor can accommodate the entire range of the signal voltage. The easiest way to do that is make sure the V_{base}= $\frac{Vcc}{2} = \frac{6}{2} = 3volts.$
   a. Since we just want to cut the Vcc in half, any two identical resistors will work. Use the voltage divider formula: $Vout = Vin * \frac{R2}{R1+R2}$ or an online calculator.
   b. Try it and see. What matters is we should pick resistors that keep base current to a reasonable level. The datasheet shows the maximum collector current = 200mA and the maximum emitter-base voltage = 6V. Experience has shown that 10K Ω (10,000 ohm) resistors work well. If you want a higher input impedance, use 100K (Brown-Black-Yellow) resistors. We will check the actual voltage and currents in the simulations later.
3. The last thing we need to calculate are the capacitor values. The input capacitor (C1) blocks DC voltage so it will not affect our base voltage we just computed. Also, the output coupling capacitor (C2) blocks any DC from affecting the input to the audio amplifier.
4. Calculate values of C1 & C2.

a. C1 & R2 form a high-pass filter and must pass the *lowest* frequency you wish to hear. This is called the *cutoff frequency*. This represents the frequency where the signal's strength = -3dB or half power. For now, let's assume I want to pass all frequencies $\geq$ 1000Hz. High-pass filters reject frequencies below the chosen cutoff frequency and pass frequencies above the cutoff frequency.

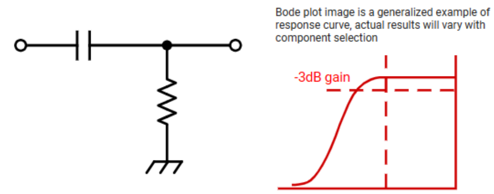

Figure 16: Typical R-C high-pass filter showing frequncies above cutoff are passed and those below are attenuated or blocked.

So, $C_1 = \frac{1}{2\pi\ RF}$ where R= R1||R2 = 5000Ω and F= cutoff frequency = 1000Hz.

The "||" means the resistors are in parallel. Why? Well, the AC signal can come into my circuit through three paths. It goes from R2 to ground, through Q1 and into R_E to ground, and through R1 to Vcc. Yes, to an AC signal, a DC source is like a ground so that means R1 & R2 are electrically in parallel. Since they are the same values, we divide their sum by 2 to get 5000Ω.

> The path through the emitter of Q1 is affected by the internal/intrinsic resistance of the transistor. This is called "little re". So, re= $\frac{26mV}{Ic}$ or $\frac{.026V}{.01}$ = **2.6Ω**. I_c= collector current. I chose 10mA in this circuit, but you need to use different values to match your design. Can you see why I ignore this path when computing input impedance? The resistance is so small that it is common to not include it in your calculations. But you always do what your boss says despite what you read here!

b. $C_1 = \frac{1}{6.2*1000*5000}$ = 3.2^{-8} Farads or **.03μF** (3.2^{-8} * 1,000,000).

c. **I chose a <u>.047μF</u> since it is a common value.** Capacitors have a 20% tolerance, so we are good to go.

5. Now let's calculate C2. But first we need to know the output impedance of the emitter-follower. There are several ways to calculate this. Let's keep it simple. The amplification of Q1 is called beta (β). Beta is the ratio of collector current to base current ($\beta = \frac{Ic}{Ib}$).

   a. For bipolar junction transistors (BJT) like Q1, designers use a value of 100. This is a *rule of thumb* and will work just fine. BJT's vary widely in this value. You can measure it with many DMM's. The datasheets show a range of 100-250 for common transistors.

   b. The output impedance is roughly equal to *re* + $\frac{R1||R2}{\beta}$ $\cong$ 3Ω + $\frac{5K}{100}$ = 53Ω. If I omit re, the output is 50Ω. Again, no difference between 3.2μF and 3.0μF!

   c. $C_2 = \frac{1}{2\pi*RF} = \frac{1}{6.2*1000*50}$ = 3.2^{-6} Farads or **<u>3.2μF</u>** (3.2^{-6} * 1,000,000). I had a 3.3μF on hand so I used that.

Finally, the output impedance must include the load (in ohms) of the amplifier I am connecting it to. According to the datasheet, the input impedance of my little amplifier is about 10K Ω.

When you choose a load resistor, there is another rule-of-thumb that says we should make *RL= 10X expected load*. So, in my case I made RL = 100KΩ (10K * 10). This is important to remember when using a simulator or when building a test circuit on a breadboard.

Since RE and RL are in parallel and their combined resistance =556Ω. That means the filter's cutoff frequency ~ 87Hz.

Refer to Figure 17 below for a summary of the circuit and how to calculate the component values.

Biasing the Emitter-Follower Summary & Details

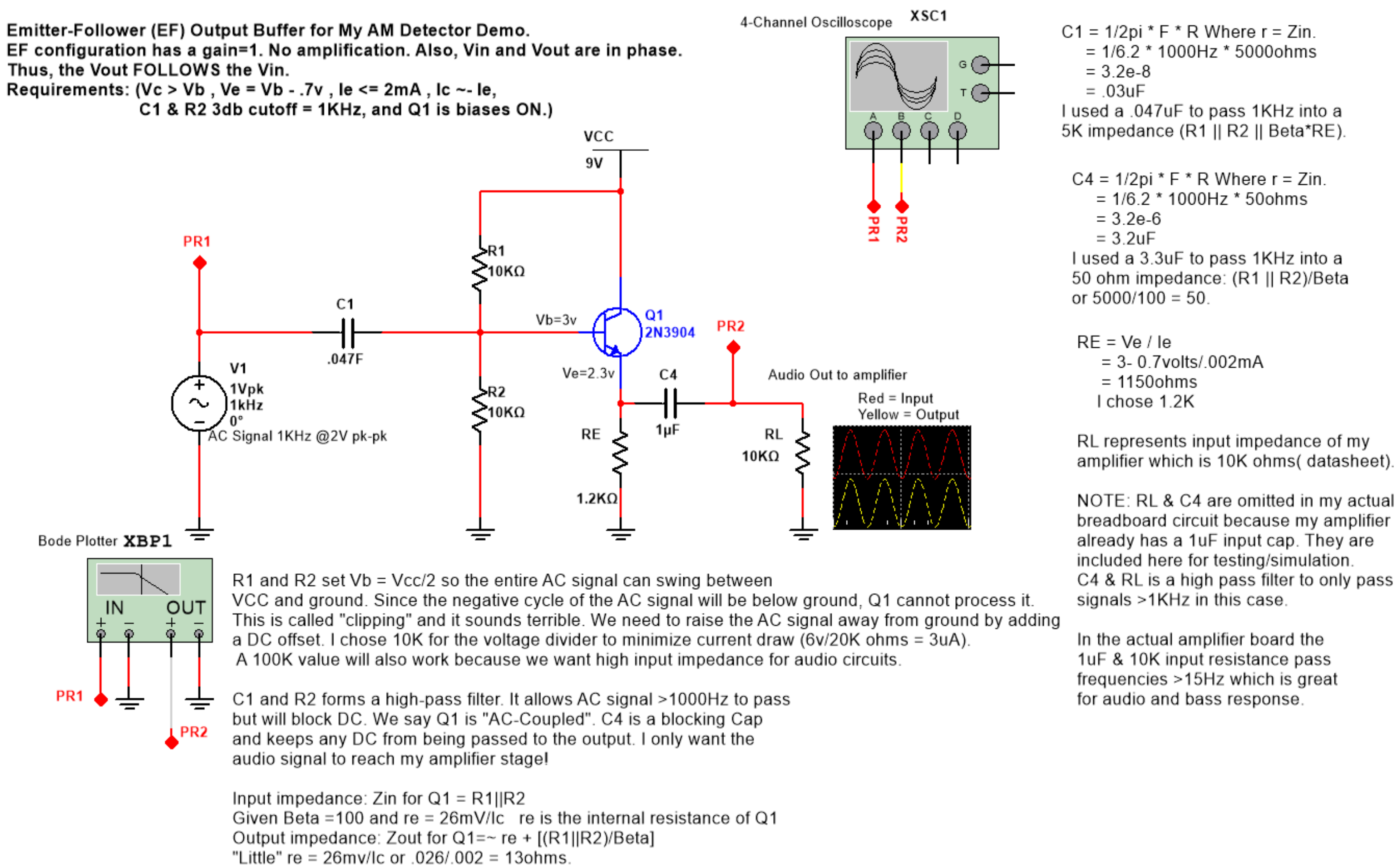

Figure 17: Detailed view of how to bias an emitter-follower. The values were chosen to accomodate the audio signals in this

The output cutoff frequency of RL & C4 is shown in Figure 18. You can see that when a 1uF and 10K are combined, they form a high-pass filter that passes frequencies at ~15Hz at -3dB as the simulation predicted.

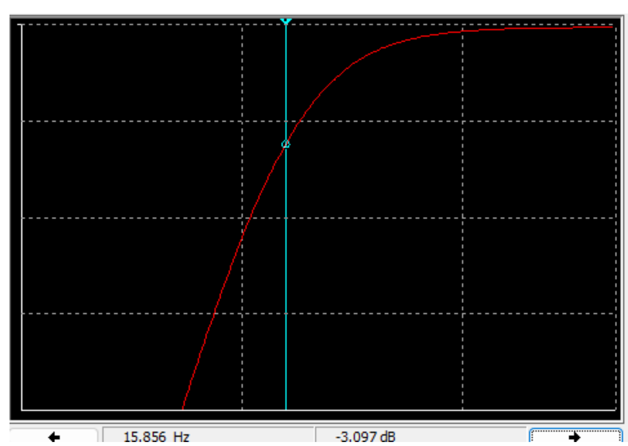

Figure 18: Bode plot shows that the cutoff for the high-pass filter from C4 & RL4 is 15.8Hz at -3dB.

CHAPTER 6: Bench Testing the Breadboard Version

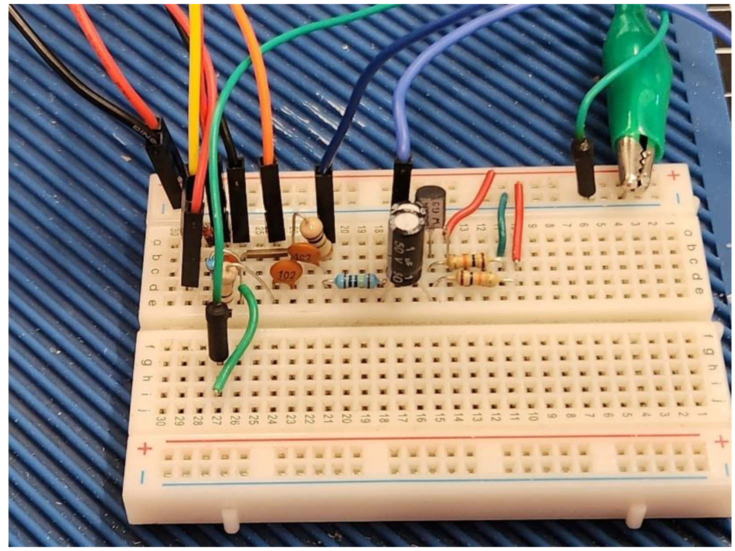

Figure 19: Detector test circuit with emitter-follower buffer output.

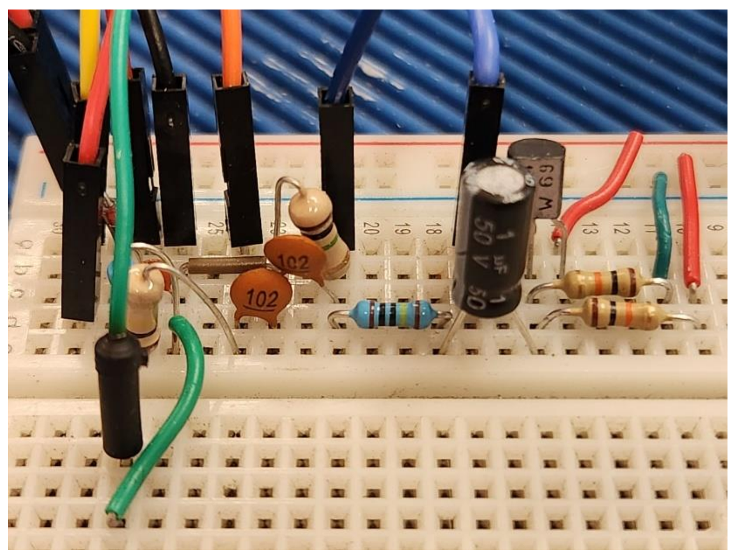

Figure 20: Close-up of test circuit.

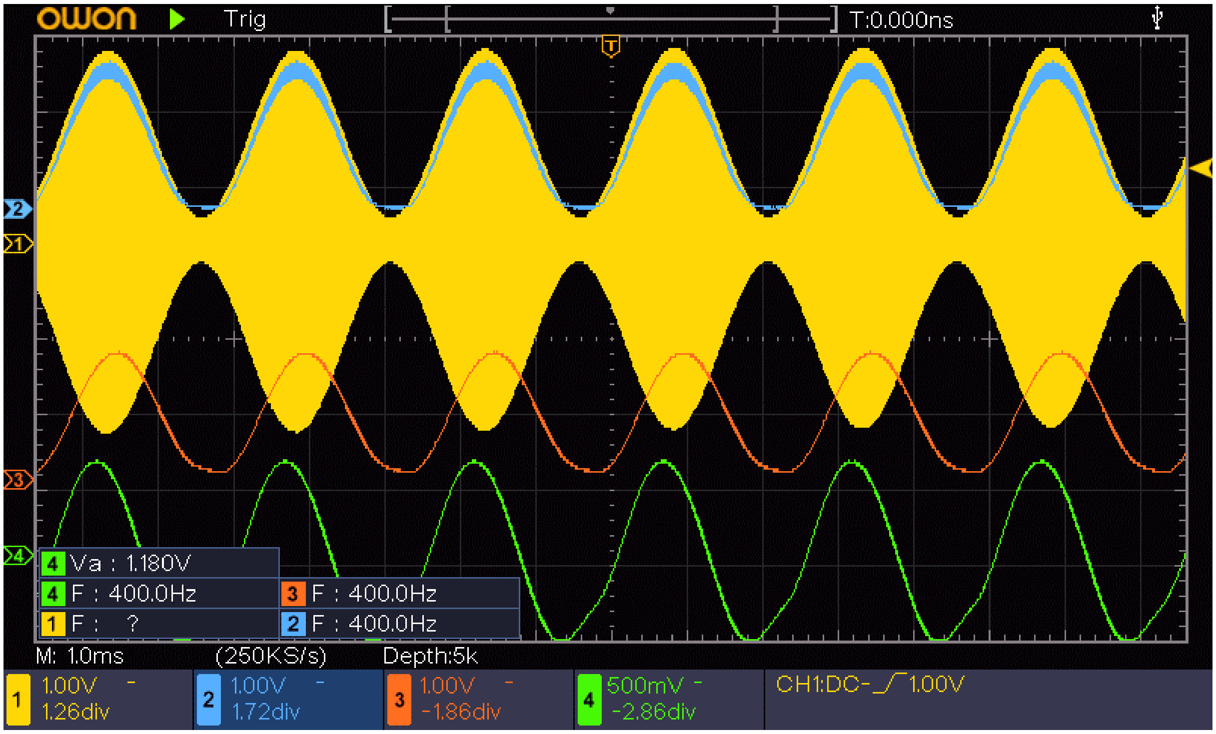

Figure 21: Breadboard results. Yellow = RF carrier, Blue = output of low-pass filter, Orange = output of the high-pass filter, Green = audio signal after DC is removed.

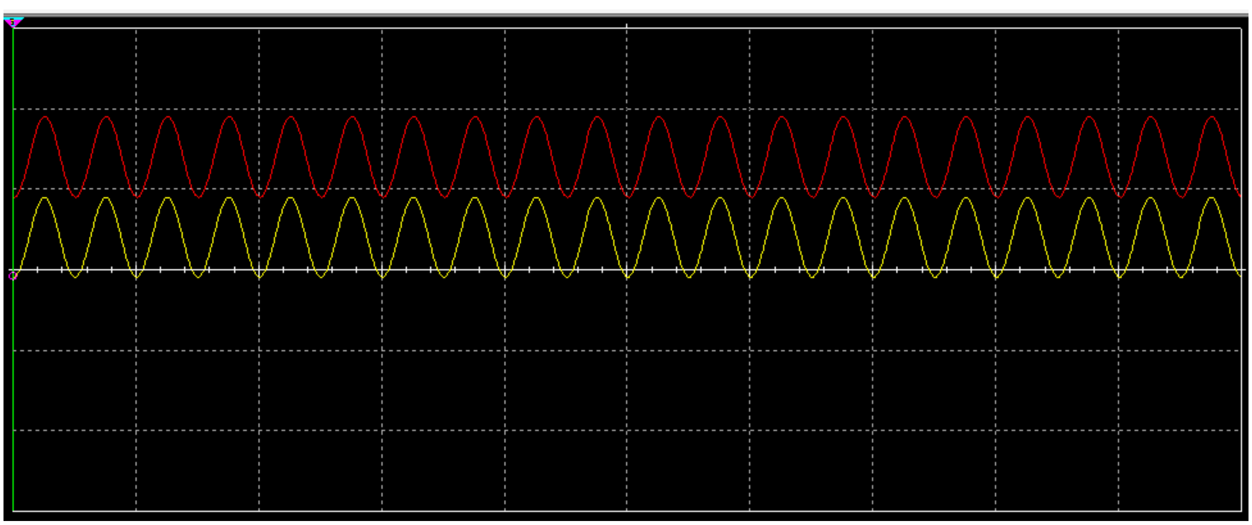

Figure 22: The red trace is the input signal. The yellow is the output. Notice the signals are in phase and nearly the same amplitude.