Lab 3 (Final): AM Receiver

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EE 157/L
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1 Introduction and General Background

For the third and final lab of EE157, I attempted to tackle assignment 1: to design and build a broadcast AM receiver. My rationale behind this was that I wanted to actually design simpler, low-level circuits such as amplifiers and demodulators and to understand how they work at a more fundamental level. This is as opposed to the FM receiver, my impression of which was sullied by the thought of just tacking on external circuits to the monolithic SA605 chip. No, I wanted to actually design some **circuits**, and I got my wish. In classic monkey's paw fashion, I experienced the frustration of designing low-level circuitry and having it not work to the design specifications due to some non-linearity, or just a poor design decision.

1.1 Superheterodyne Reciever

One method of organizing the reception of an AM signal into a block diagram is the so-called 'superheterodyne receiver' invented by Reginald Fessenden, a generic form of which is shown in Figure 1. This system first receives the RF signal through an antenna, and puts the signal through an 'RF filter' also commonly called an image rejection filter. This filter blocks any other bands that might cause images to be projected into the band of interest through the mixer. The signal is then amplified through the RF amplifier, which may also be called a 'low noise amplifier (LNA)' which boosts the received signal power while also having a small noise figure. This amplifier must have a low noise-figure because Friis' formula indicates that the noise figure of the first amplifier in a set of cascaded stages is the most significant source of noise for the system.

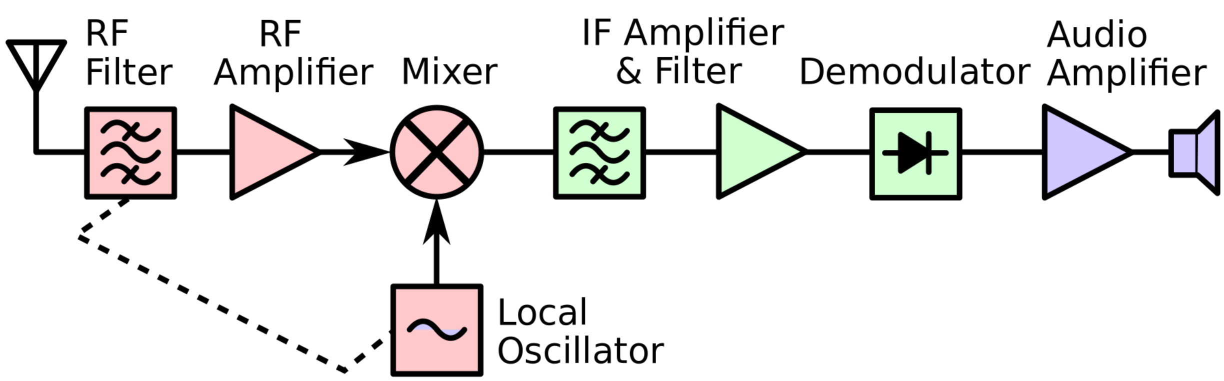

Figure 1: Generic single-conversion superheterodyne reciever
By Chetvorno - Own work, CC0, https://commons.wikimedia.org/w/index.php?curid=46222556

The signal is then mixed down from the radio frequency to the intermediate frequency (IF). The purpose of doing this is that better resonant circuit bandwidths, amplifier noises and gains can generally be achieved at lower frequencies. Mixing the received signal down to a lower frequency allows for the design of better performance circuits to operate on the signal than at higher frequencies. The signal then goes through a highly selective filter which filters out everything except for the band of interest, in this case the station we're trying to receive, and is then amplified to again boost the power of the signal. The signal is then demodulated from the IF.

Demodulation methods vary depending on the precise type of modulation used for the transmitted signal. Sub-types of amplitude modulation include: Double sideband full carrier (DSB-FC), double sideband suppressed carrier(DSB-SC), single sideband(SSB), vestigial sideband (VSB), etc.

All these types of modulation require different demodulation techniques. Fortunately, broadcast AM uses DSB-FC modulation which is the easiest to demodulate.

After demodulation, the message has been extracted from the carrier and all that's left to do is to amplify the message and use it to drive a speaker!

1.2 Mixing

In an abstract sense, a mixer implements a time-domain multiplication operation for two input signals. In the case of heterodyning a signal, those two signals are the received RF signal and the local oscillator (LO) signal. Because multiplication in the time-domain is a convolution operation in the frequency domain, the resulting effect is that the output of the mixer is the sum and difference of the two input signal frequencies via the 'sliding product' effect of convolution. The math behind this is fairly simple, take two sinusoidal input signals, the RF signal and the lo signal:

$$\begin{aligned} rf(t) &= cos(\omega_{rf}t) \\ lo(t) &= cos(\omega_{lo}t) \end{aligned} \tag{1}$$

the output of the mixer, the IF signal, is then a simple time-domain multiplication of the two signals and can be re-arranged through the multiplicative trig identities to be the following

$$\begin{aligned} if(t) = cos(\omega_{rf}t) * cos(\omega_{lo}t) \\ = \frac{1}{2}cos((\omega_{rf} + \omega_{lo})t) + \frac{1}{2}cos((\omega_{rf} - \omega_{lo})t) \end{aligned} \tag{2}$$

Which makes it clear that the two frequencies present on the output of the mixer are the sum and difference of the two input frequencies. One of these is the frequency in which we are interested, the IF, and the other is garbage that can be filtered out. You can mix either on the high or low side, meaning that the LO frequency can either be higher or lower than the RF. mixing on the high side has the effect of inverting the message sidebands, which is relevant for SSB or VSB demodulation but for DSB this inversion is inconsequential.

1.3 AM demodulation

There exists multiple ways to demodulate AM signals, and the different types of AM need to be demodulated using different methods. For example, envelope detection does not work for SSB, but does work for DSB-FC. SSB might require an additional heterodyning step using a diode product detector and a beat frequency oscillator. This being said, broadcast AM is modulated using DSB-FC, so arguably the simplest method of AM demodulation can be used: The envelope detector.

The envelope detector works on the principle that a the message in a DSB-FC signal (shown in Figure 2) is expressed as the envelope of the signal. If you can make a circuit that tracks the envelope of a signal, then you could extract the message. This is easily achieved in theory as the circuit only requires three components, a diode, a resistor and a capacitor, shown in Figure 3. The diode in this circuit, modelled using the 'switch' diode model, acts as a half-wave rectifier. The RC circuit then acts as a low-pass filter: the capacitor is charged from current through the diode during the positive half-wave cycle, and then, slowly relative to the carrier frequency, bleeds its charge through the resistor when the capacitor's voltage exceeds the input signal voltage. Thus, the circuit tracks the envelope of the signal and extracts the message.

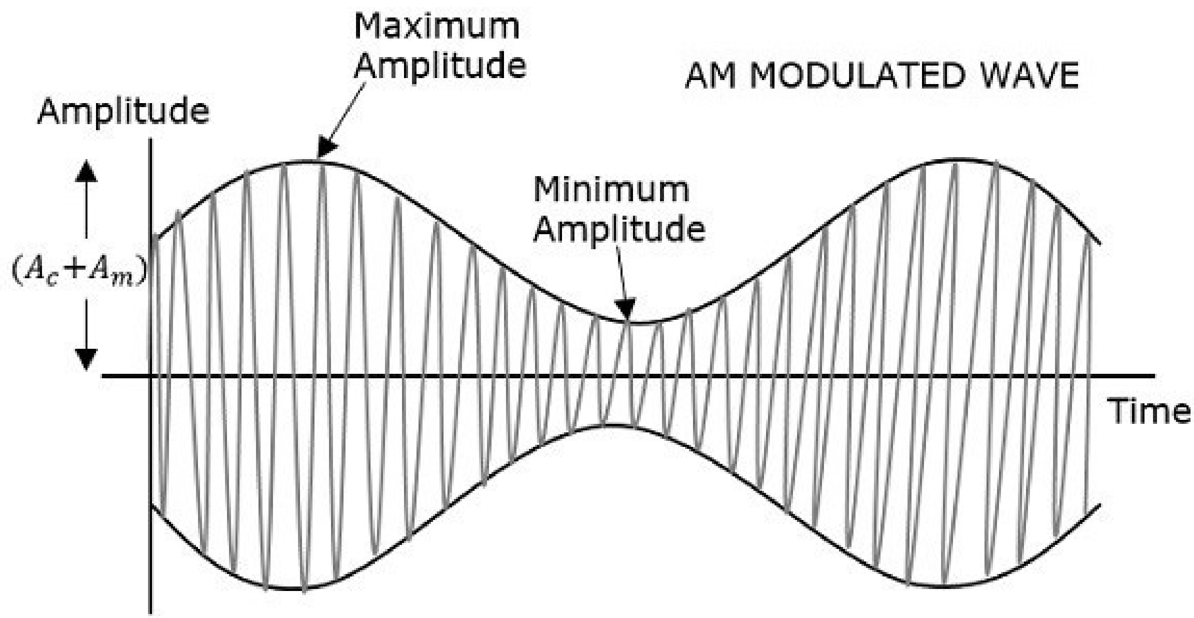

Figure 2: A double sideband full carrier signal

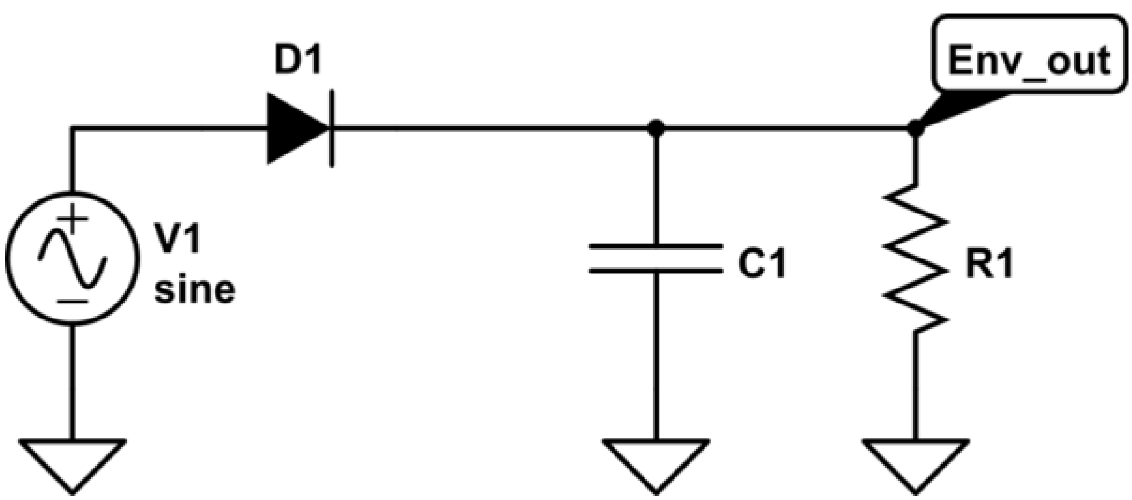

Figure 3: A simple, passive envelope detector circuit

2 Design Objective and Methodology

The stated goal of this project was to create an AM receiver that will receive the local AM station KSCO on 1080kHz. As stated previously, the station broadcasts a DSB-FC AM signal, so my receiver would have to receive, amplify, demodulate and play the signal through a speaker. The goal was to create a single conversion superheterodyne receiver, meaning that there would be one IF, and the mixer would be implemented using the SA602 mixer chip. The circuit would need to receive a signal at about -100dBm to -90dBm, approximate values for the signal strength cited by Prof. Petersen in the lab. The receiver should have as low of a noise figure as possible, and amplify the signal sufficiently to be heard well over the chosen speaker.

The methodology I used to approach the design of this circuit was to work back to front in terms of the block diagram. This will make more sense after the block diagram is discussed in the next section. The thought was that if I worked back to front, testing the completed circuits as I moved back, I would know what specs would be required from the frontmost amplification stages, i.e. what gain levels would be required from the amplifiers and mixer for the signal to be heard over the speaker. This came into play especially after I had tested my envelope detector and audio amplifier, so I then knew what power level the input signal would need to be.

I also planned ahead of time how I would divide up the PCBs. I had planned to put the back-end circuit on one board (demod, audio amplifier, speaker), put the IF amplification stage on another board, and on a third board put the front-end, the band select filter, mixer and antenna and low noise amplifier. Due to time restrictions, I ended up combining the front end and IF amplification onto one board, which worked out rather nicely.

3 Block Diagram-Level Design

Before designing any circuits, I first laid out a block diagram of my receiver, shown in Figure 4. This diagram shows several design decisions made up front about the receiver. Firstly, that the IF was chosen to be 455kHz. The main motivating factor behind this decision was that we were being offered 455kHz ceramic filters with 25kHz bandwidth by Prof. Petersen, which meant I would get a well performing and reliable circuit to act as the band select filter, without having to do research into what part to buy.

Another consideration shown by the diagram is the necessary gain from each amplifier stage. After having built and tested the envelope detector and audio amplifier, I discovered that the circuit had good output levels for input signals of -30dBm or greater. This figure then became my target for power gain in the previous stages. The SA602 datasheet states that it the mixer gives +18dB of conversion gain at 45MHz, but I went with a more conservative guess of 12dB. The ceramic filter datasheet says to expect an insertion loss of -5dB, which is listed on the diagram. This leads to the goal of achieving a gain of 60dB with the IF amplifier stage to bride the gap between -90dBm out of the band select filter and -30dBm into the envelope detector.

I'll note the lack of inclusion of a low noise amplifier before the SA602. This is because the SA602 datasheet states that it can receive signals as low as -119dBm, so I figured that -100dBm would be no problem, assuming the signal to noise ratio of the received signal was high enough.

4 Envelope Detector

The first circuit I actually designed and built was the envelope detector. My first attempt at doing this was to design the passive circuit shown in Figure 3. This passive circuit is simple in theory, but presents a large practical flaw: The model assumed for the diode is an simple switch without taking into account its forward operation voltage. This wouldn't be a problem if the incoming signal had an mean-squared voltage of greater than 0.7V, but the modulated signal is not going to be more than half a volt to a volt. This means that the signal has a hard time turning the diode on, and the signal is either not passed or is highly distorted, especially for higher modulation factors where this 0.7V operating voltage eats into the envelope.

An engineering schematic of my envelope detector demodulation circuit is shown in Figure 5. The biggest improvement of this circuit over the passive circuit is that the rectifier diode is biased by the transistor in such a way that negates the diode's operating voltage and allows the full positive half-signal to be passed through the circuit with little distortion. The circuit operates as

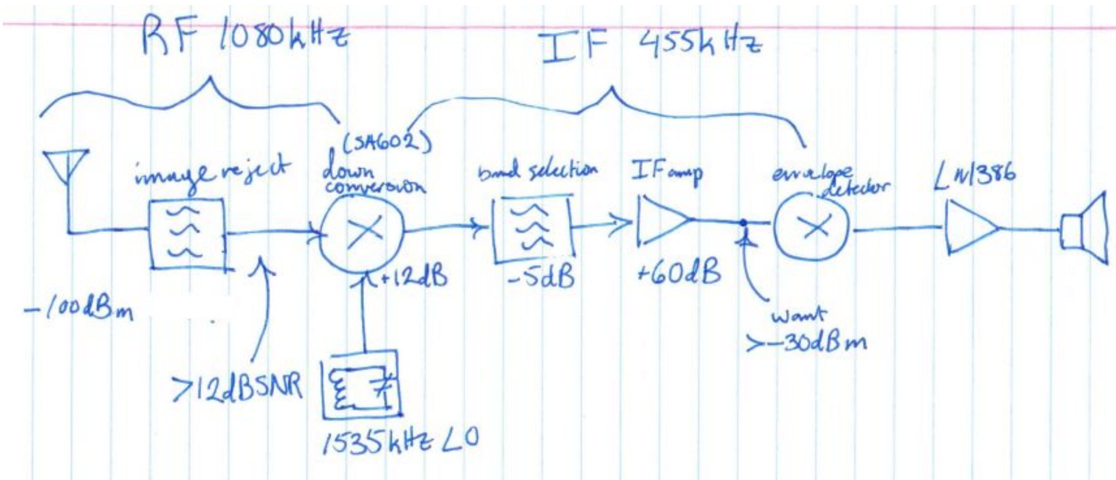

Figure 4: Block diagram of my planned AM radio receiver

follows. The voltage divider biases the diode anode at a little over the sum of the operating voltages of the diode and the BJT's BE voltage. During the positive half-wave cycle of the input signal, the base of the transistor is greater than the required BE voltage, so the transistor behaves as an emitter follower, passing the signal. The signal then goes through the RC low pass filter which filters out the IF. In the negative half-wave cycle, the transistor base voltage dips below 0.7V and the transistor enters cutoff and the signal is no longer passed through to the emitter. The current requested by the signal source is then provided by the diode from Vcc.

This circuit worked quite well for down to very small amplitudes with little distortion, though for modulation factors of over 80% with higher power amplitudes, the signal started to distort a bit. I'm not sure what was happening in the circuit to cause this distortion, but I would guess that the transistor was leaving forward active for the top part of the envelope's wave cycle, as the waveform appeared flattened.

5 Audio Amplifier

The second part of the circuit I designed was the audio amplifier. The periferal circuit I used is shown in Figure 6, and is a slightly modified version of one found in the LM386 datasheet. The $10\mu F$ capacitor bypassing pins 1 and 8 indicates that the LM386 is configured for gain = 200. I set it this high because I didn't anticipate giving the amplifier such a high power signal or such a high-frequency signal that the output would start to distort, and I wanted as much gain as I could get from the amplifier. pin 7 is bypassed to ground, and the unused input is also shorted to ground to prevent a floating potential screwing up the differential input. The input is fed by a 10k potentiometer, giving variable control of the delivered input signal power.

The output of the amplifier is fed into a speaker with several accompanying passive components. The $470\mu F$ capacitor acts to decouple the amplifier from the speaker to prevent a large DC current from going through the speaker. It is made as large as possible as to not block any lower frequency signal components from being blocked from going to the speaker. The 10k resistor is acting as a bleeder resistor to allow an alternate current path for capacitor to discharge when the speaker is disconnected from the circuit. If this weren't there, when the speaker was plugged back in, the capacitor would discharge through it and create a loud pop potentially damaging the speaker and is just annoying. The Zobel network (RC shunt on the output of the amplifier) serves to linearize

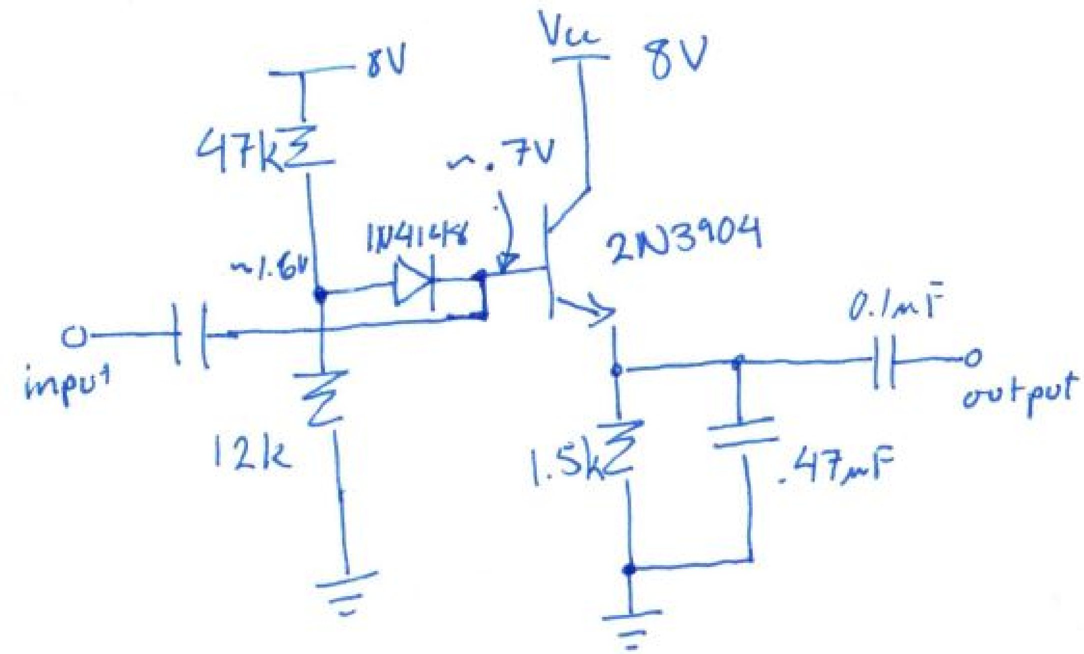

Figure 5: Engineering schematic of my biased envelope detector

the high frequency impedance of the amplifier, by compensating for the inductive reactance of the speaker by giving an alternate 10 ohm load seen by high frequencies to ground.

This circuit performed admirably and by testing it in combination with the envelope detector, I experimentally determined that my input signal to the envelope detector should be between -30 and -20dBm. This then became the figure to shoot for when designing the front end of my receiver and informed my subsequent design process.

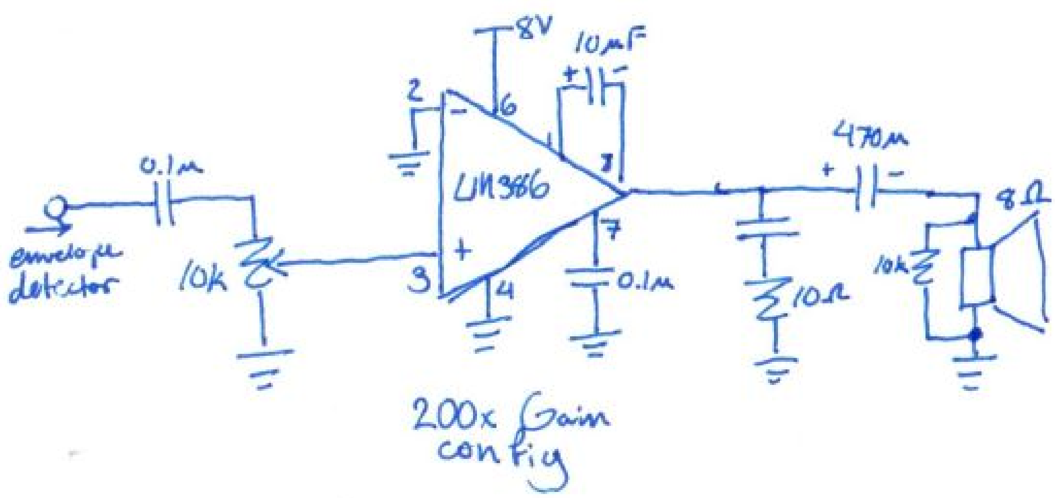

Figure 6: Engineering schematic of my LM386 audio amplifier peripherals

6 IF Mixer and Band Selection Filter

After the receiver back end was designed and built, I started designing the mixer and band select filter section, shown in Figure 7. The mixer uses an SA602 chip, which implements a Gilbert cell analogue mixer. pins 6 and 7 are the base and emitter of a biased BJT which should be connected to a tuned resonant circuit to implement an oscillator. For my local oscillator, I used a Colpitts LC oscillator with a tuning capacitor in parallel with the inductor to tune the precise oscillation frequency. As constructed, the oscillator operates between 1450kHz and 1560 kHz depending on the value of the tuning capacitor. This contains the desired 1535kHz, so it's just a matter of tuning the capacitor to the correct value to get a stable oscillation of 1535kHz.

The circuit shown in Figure 7 also includes an input impedance matching network. This was tuned to match the 50ohm source impedance of a coaxial transmission line with the 3kOhm differential input impedance of the SA602. the $0.1\mu F$ shunt capacitor blocks a DC path from pins 1 and 2 to ground, which would otherwise disrupt the internal transistor biasing. The output impedance of the SA602 is 1.5kOhms, which matches perfectly with the 1.5kOhm input impedance of the ceramic filter. The ceramic filter also has an output impedance of 1.5kOhms, which is mis-matched with the 50 ohm impedance of the SMA connector. I had originally tried to implement another impedance matching network to solve this problem, but for an unknown reason the network was not passing a signal. When I have more time, I will come back and further explore this issue out of interest.

The SA602 has a stated gain of 18dB at 45MHz, and the ceramic filter has an insertion loss of approximately 5dB. This gives a net gain of 13dB for the circuit. From testing, using the signal generator as an input and reading the output with the spectrum analyzer, I was reading a net gain of only 5dB. This sub-optimal gain is most likely due to the lack of an impedance matching network on the output of the ceramic filter, so it should be easy to remedy with some further experimentation. For 1080kHz input signals of less than -50dBm and greater than -90dBm, the mixer performs well and outputs the desired IF of 455kHz. If the signal power increases beyond -50dBm, third order products begin appearing on the spectrum analyzer. This indicates that the spurious-free dynamic range of the mixer is from -90dBm to -50dBm. Notably, in testing the oscillator frequency shows up as a constant -100dBm amplitude regardless of the amplitude of the input signal. I'm not sure what is causing this oscillator bleed-through, so this is another thing I would like to look at through further testing.

7 IF Amplifier

Lastly, to bridge the front end and back end of the receiver, I designed the IF amplifier section. This is where I planned the majority of the receiver gain to come from. I designed the amplifier section to achieve approximately 60dB of gain, to bring the signal from -90dBm to -30dBm. To accomplish this I implemented 3 cascaded JFET amplifier stages with inter-stage impedance matching. To do this, I implemented 3 identically biased JFET amplifiers shown in Figure 8. The J310 is set up in a self-biasing JFET topology, where the designed drain source current is 13mA, and the base emitter voltage is set to about 2V. through the voltage divider set up by the drain and source resistors. Additionally, there is a bypass capacitor across the source resistor to increase AC gain. Also shown are tapped-C impedance matching networks on the input and output. The passive component values are listed as generic because they depend on what impedances I'm trying to match to. For example, the input of the first stage would try to match the 150k gate resistor to a 50 ohm co-ax or signal source impedance, but an inter-stage matching network would try to

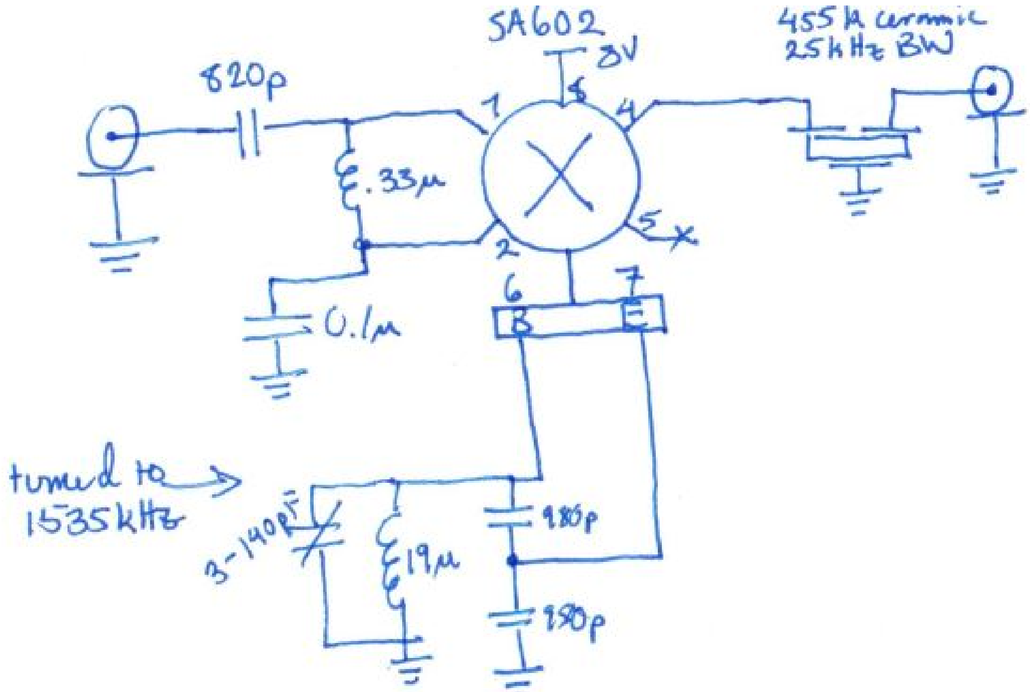

Figure 7: Engineering schematic of my SA602 mixer and band-select filter

match the approximately 470 ohm output impedance of the amplifier stage with the 150k input of the subsequent stage. The values for the passive network components were then determined using SimSmith.

Through testing these stages with the signal generator and spectrum analyzer, a single stage was found to produce approximately 25dB of power gain for a 455kHz signal. However, the addition of a second stage with inter-stage impedance matching only increased the gain by a further 10dB. The addition of a third stage seemed to not increase the power gain at all. This issue is something I wish to explore further on my own time, as I think there is a lot of knowledge to be gained from debugging this. As I see it, there may be several contributing factors to this issue.

One of these issues may be improper inter-stage impedance matching. I calculated all of the passive component values using SimSmith, but I didn't have exact values for all of the components at hand, so I had to make due with approximate values. Especially when matching large impedances to small impedances, such as the 470 or 50 to 150k, even small changes in capacitor and inductor values presents a large drop in power transference according to my testing in SimSmith. Another issue may be inaccuracies in my evaluation of the input and output impedance for the circuit. Since the gate has an input impedance of more than 10M ohms, I'm not worried about inaccuracies on the input of each stage. However, I did use an approximation for the stage output, in that the output impedance is roughly equal to that of the drain resistor. In order to create a better model for the stage output impedance, I would have to analyze the circuit's small signal model, and that would be one of my first steps to figure out this issue.

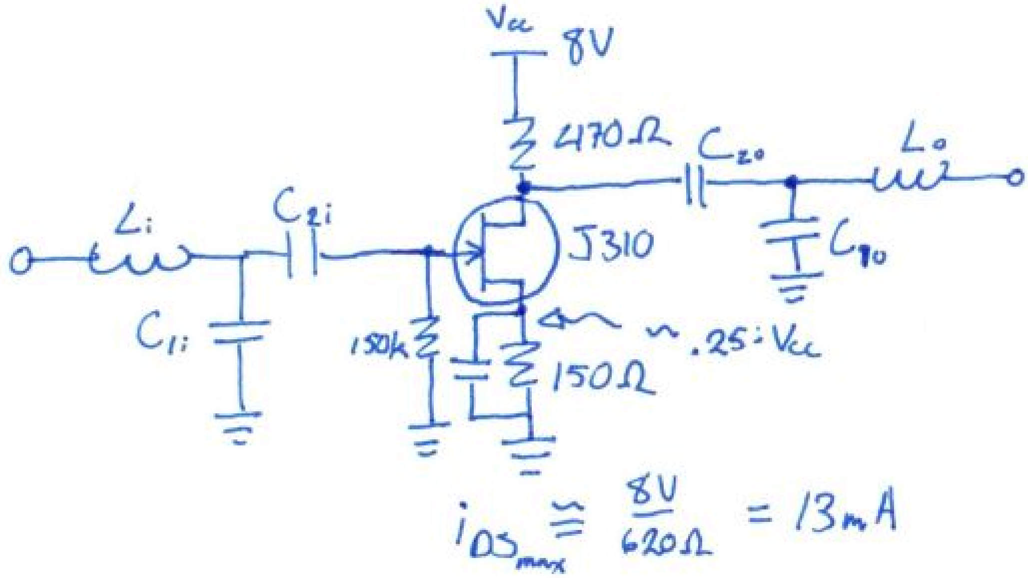

Figure 8: Engineering schematic of a single IF amplifier stage with impedance matching in and out

8 Conclusion and Further Work

This lab was an incredible experience as an introduction to experimental design engineering. Before taking this course, my biggest flaw as a budding engineer was getting stuck in the theory of a design or problem and never actually **trying** anything. Because of this course, I learned the value of trying out a preliminary design to get real feedback and to see results you as the designer would never have anticipated while thinking about a problem theoretically. I will say that in discovering this 'just try it' attitude, I went a bit too far in the other direction and made quick and dirty modifications without keeping a good record of what exactly I was doing and why. Continuing with this thread, I also learned the importance of taking good experimental notes and keeping an accurate and up-to-date engineering schematic, because I realized while doing my oral evaluation with Petersen that my engineering schematics weren't up to date with the small experimental modifications I had done. This made it difficult to recall all of the changes I had made, what exactly they were and why I made them. I also learned the importance of using engineering symbols in my schematic rather than wiring diagram symbols, because it makes the diagram much more readable for a reader who didn't design the circuit, or for me after I come back to it after completely forgetting my thought process. For example, in some of my diagrams I used the box/numbered-pin symbol for my SA602, which made it very hard to look back and, at a glance, get an intuitive understanding of what my circuit was doing. This made explaining my circuit schematics to Prof. Petersen difficult, as he had a hard time looking at my circuit and telling what was happening.

I fully intend to re-visit this project on my own time, and work out a few unresolved issues in my current circuits. For future reference, those issues are: Non-optimal gain in successive IF amplifier stages, non-optimal gain in the SA602 and band-select filter, general impedance matching issues. I also fully intend to come back and design a low-noise amplifier and loop-stick antenna circuit so that I can play around with that. If I complete all of these objectives, the next goal I have is to make the receiver tuneable so I can receive the entire AM band.