

Lock-in Amplifier

Introduction

A lock-in amplifier is an analog circuit that picks out and amplifies a particular frequency of oscillation and rejects the other frequencies. Lock-in amplifiers are useful because they can extract very weak signals from noisy environments as long as the signal is at a precise and known frequency. We built a system that transmits audio by amplitude modulating light at a particular reference frequency. The receiving side then acts as a lock-in amplifier by demodulating at the reference frequency, so that we can extract the audio data. We will discuss the design choices that were made for this system during planning and construction, the performance that was achieved, and possible extensions of this project. Although we used AM transmission, we also looked into FM transmission and so we will present our suggestions for how our system could be adapted in that way.

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Motivation

Lock-in amplifiers are useful for taking measurements in scientific research. For example, part of an experimental set-up could be modulated at a given frequency so that the lock-in amplifier can filter to that frequency to separate signal from noise. Because of this, learning about lock-in amplifiers could be relevant to research that either of us does in the future. The use of light as a transmission channel is also interesting, because it could potentially be used to transmit noise from one side of a noise-proof window to the other without wires. In one of our tests, for instance, we transmitted audio through a water bottle.

Full Block Diagram and Overview

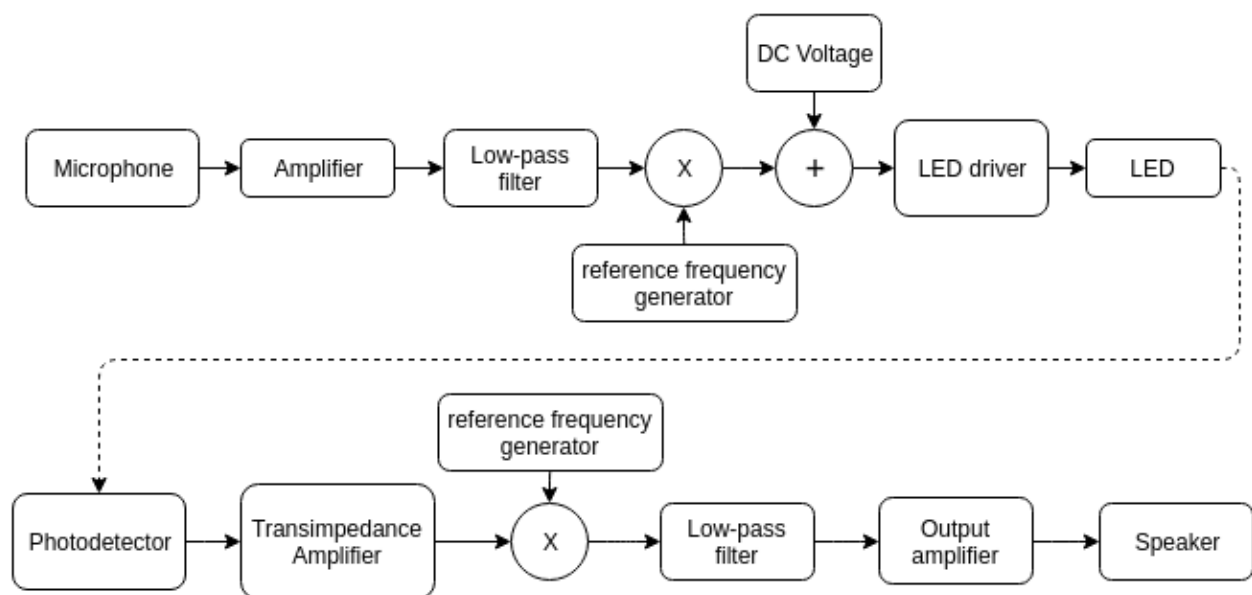


Figure 1: Block Diagram of Full System

In our system, the output of a microphone is mixed with a reference tone and used to drive an LED. The LED output is picked up by a photodetector, whose output is mixed with the same reference tone for demodulation in order to drive the speaker. A low-pass filter whose passband includes only the audio spectrum is required on the input of the modulating mixer in order to prevent noise at higher harmonics of the reference frequency from interfering with the transmission. Another identical low-pass filter is

required on the output of the demodulating mixer in order to extract the audio signal, which has now been centered around zero by the demodulating mixer.

We have logically divided our system into two parts: the low-frequency pipeline (the microphone, its amplifier, the LED and photodiode, the speaker, and its amplifier) and the high-frequency pipeline (the modulator and demodulator). As such, each of these subsystems was constructed in full separately and then integrated together.

Low-Frequency Pipeline Modules (evschell)

Microphone and Amplifier

The microphone that we used is from CUI Inc. part number 102-1720-ND. This part was chosen, because it picks up audio frequencies (20-20kHz), it fits directly into the breadboard, has a good SNR, and was easy to use. We used the interfacing circuit that is suggested in the datasheet and is shown below. The +Vs as shown in Figure 3, was provided by the 5 V port of the Protoboard power supply. The rest of this stage was powered with +15/-15/0 V from the Protoboard.

Initially, a 1 μF capacitor was used for C in Figure 2, but in testing of later stages of the system, it was discovered that this capacitor was high-passing out the lower frequencies of the audio spectrum. To correct this problem, the capacitor was increased to 100 μF .

The microphone produces a signal with a maximum amplitude of around 100 mVpp. We chose to amplify the signal directly after the input, because having a larger signal propagating through the system helps increase the SNR throughout the circuit. The amplifier boosts the signal amplitude, such that the peak to peak voltage is somewhere between 1 and 10 volts, depending on the volume of the input audio. This is implemented with an op-amp inverting amplifier using a 353 as shown in Figure 3. The gain was chosen to be around 150 when the modulator is included in the system and 30 when the modulator is not included in the system.

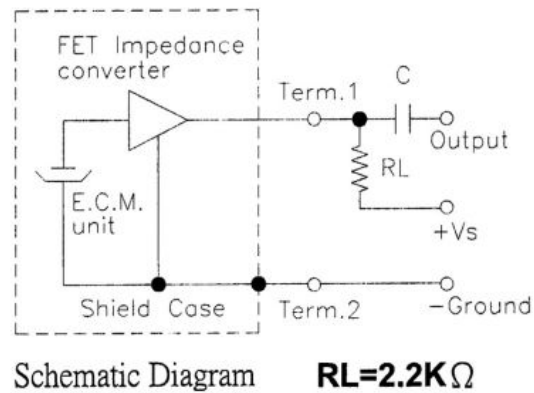


Figure 2: Microphone Circuit Suggested in CUI Inc. Datasheet

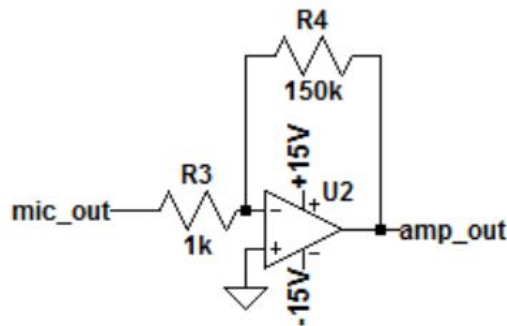


Figure 3: Input Amplifier

Low-pass filters

We built two identical low-pass filters. One cleaned up the microphone output before it was used for modulation, ensuring that we only had signals in the audio band. The other was used to clean up the demodulated signal before it was sent to the speaker. The audio spectrum extends to less than one decade away from our modulating frequency of 100 kHz, so we needed a relatively sharp roll-off in order to avoid harmonic distortion. We chose the cut-off to be 10 kHz instead of 20 kHz so that there is a lot of attenuation at the 100 kHz frequency at the cost of having more attenuation at the higher frequencies in the audio spectrum. In order to get the sharper roll-off, the low-pass filters were built using a Sallen-Key topology, which is a second order filter. Our schematic as shown in Figure 4 was built with a 353 op-amp and powered by +15/-15/0 V from the Protoboard.

When the filters were first constructed with the calculated resistor and capacitor values, the output started to attenuate around 8 kHz. In order to increase the frequency at which attenuation starts, the resistor values were changed to be slightly smaller. This ensures that more of the audio spectrum is within the pass-band.

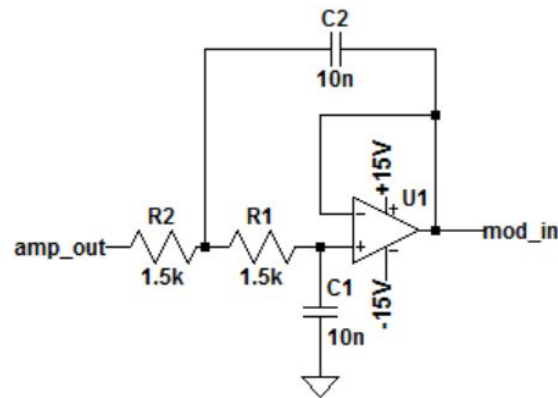


Figure 4: Sallen-Key Low-pass Filter

LED and Photodetector

Our system uses an LED and photodiode to send data over a light channel. This channel is where noise can be added to the system depending on how much we shield it from extraneous light sources, such as the room's lighting or the sun. On the transmission side, an LED-driving circuit or transconductance amplifier was needed to take the modulated voltage signal for transmission and convert it to a current signal. On the receiving side, a transimpedance amplifier was used directly after the photodiode to convert the current signal back to a voltage signal which could be fed into the demodulator.

One big aspect of the design of this section was part selection. The LED needed to be chosen so that its response time would allow it to keep up with the modulated input signal, which was originally planned to be 1 MHz. Later, a lower modulation frequency of 100 kHz was chosen. The LED that was used was manufactured by Vishay, part number TLHG5400. The photodetector also needed to be chosen so that its response time allowed it to keep up with the possible modulation frequency of 1 MHz. Additionally, it required a spectral sensitivity that included the wavelength at which the LED operates (562-575 nm for green light, as given by the LED datasheet). The photodetector was manufactured by Osram, part number SFH 213. The op-amps used were 353s and the BJT was a 2N3904. The power supply used was +15/-15/0 V from the Protoboard.

The transconductance amplifier topology that was selected is shown in Figure 5. This circuit converts voltage to current by applying the voltage over a resistor connected to the emitter of a BJT. An op-amp buffer was included to ensure a more linear conversion. The transimpedance amplifier used the same negative feedback op-amp design as was used in Lab 6 and is shown in Figure 6.

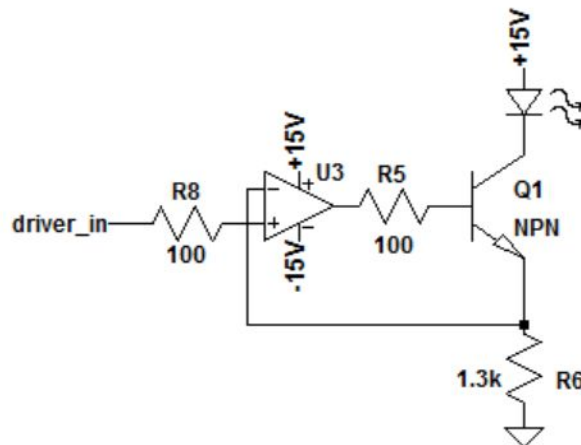


Figure 5: Transconductance Amplifier used to drive LED

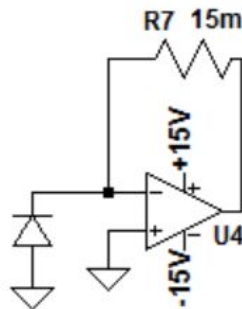


Figure 6: Transimpedance Amplifier used after photodiode

During construction, an inverting summing amplifier stage shown in Figure 7 was added before the transconductance amplifier in order to properly bias the LED. This summing amplifier takes the voltage signal from the low-pass filter and adds it on top of a DC voltage, which can be adjusted by the potentiometer. The LED was biased such that the peak to peak brightness variation was largest so that the AC signal was large and also such that the average brightness was as high as possible in order to assure that the system could communicate over the furthest possible distance. The voltage that worked

the best was -9.5 V which effectively added +9.5 V, because this stage is inverting. This biased the current through the LED was 5.5 mA.

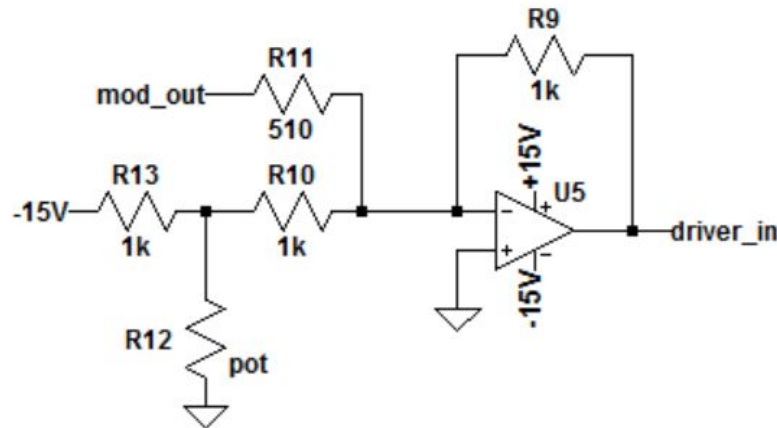


Figure 7: Inverting Summing Amplifier

Amplifier and Speaker

This is the output stage, which amplifies the signal before feeding it into a speaker. We used the class D amplifier that we made in class, so that we could devote more time to different parts of our project. For a speaker, we used one of the computer speakers in lab. It is shown below in Figure 8.

When this module was initially connected to the rest of the low-frequency pipeline, it became apparent that the lab kit power supply was not able to stably power the circuit. Whenever the speaker was connected, spikes of oscillations would occur on the signal and propagate back throughout essentially the entire system. The first attempted solution was adding more bypass capacitors to the power rails. Next, the project was moved to the protoboard power supply instead of the lab kit. The solution that finally eliminated the oscillations was moving just the amplifier section of the system to the 5V port of one of the heavy duty DC power supplies above the lab work stations. It is believed that the class D amplifier could have been causing shoot-through which affected the rest of the system, because it was connected to the same power supply.



Figure 8: Speaker

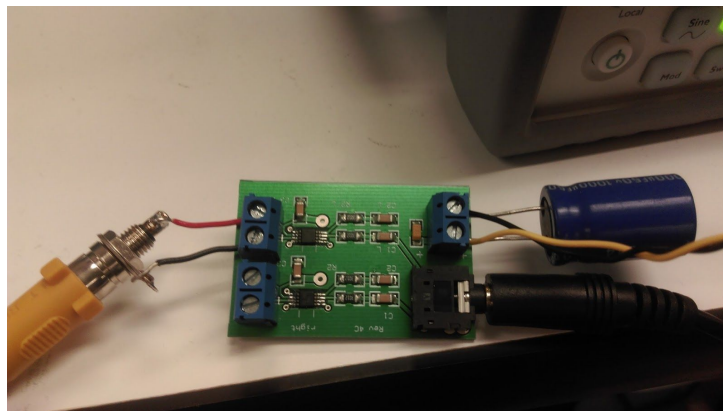


Figure 9: Class D Amplifier

High-Frequency Pipeline Modules (sadun)

Current switcher

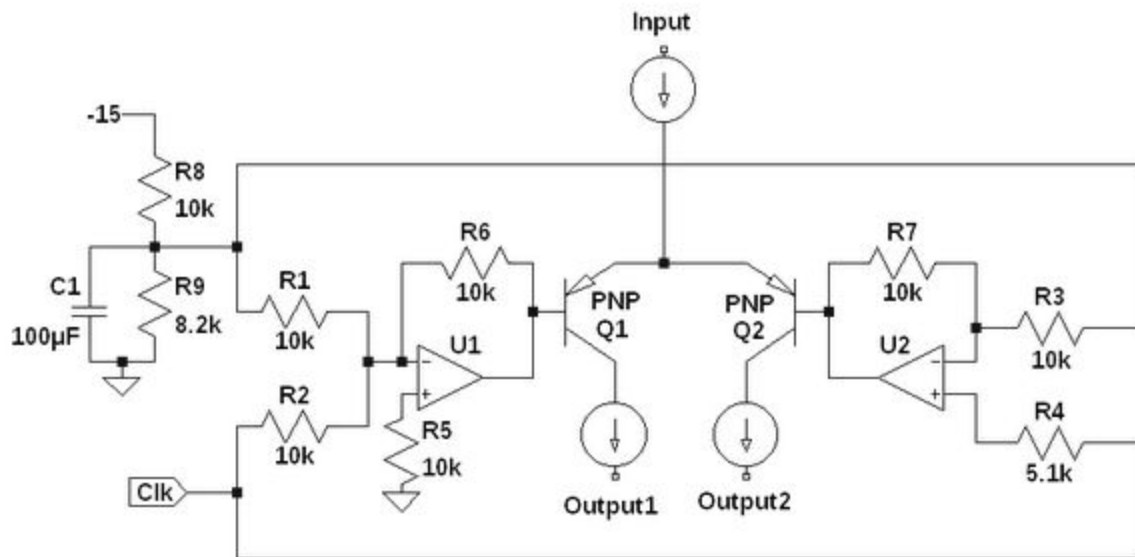


Figure 10: Current Switcher used in both modulator and demodulator.

The key operating component of both the modulator and the demodulator is the current switcher shown above, which takes a current source and switches the current back and forth between two sinks according to a clock input. When the base voltage on Q1 is significantly higher than the base voltage on Q2, the current flows exclusively through Q2, and vice versa. This is the core concept of a Gilbert cell mixer (figure below), used in RF circuits as a phase detector or a differential demodulator, which we adapted for this lower-frequency, single-ended purpose.

The rest of the schematic above is just biasing and base driving. R8, C1, and R9 create a stable reference voltage. The resistors around U1 create a summing amplifier, and the resistors around U2 an inverting amplifier, so that the base voltages V_{B1} and V_{B2} have a low common-mode oscillation and express the clock signal differentially.

In theory, it is not necessary to suppress the common-mode voltage, as if the clock is a square wave of sufficiently high amplitude (i.e. >100 mV - we used 1 V peak-to-peak), then Q1 and Q2 will essentially be either fully on or fully off regardless of fluctuations in the reference voltage. But in testing, this differential topology performed significantly

better. The current-switching is fairly non-ideal at the 101 kHz frequency that we used, due to the limitations of the 2N3906 BJT's and the LM6132 op-amp, and when the two outputs are resistors to ground, the voltages observed at the collectors of the BJT's show significant ringing. Still, the waveform is consistent, and can be low-passed to eliminate the worst of the ringing.

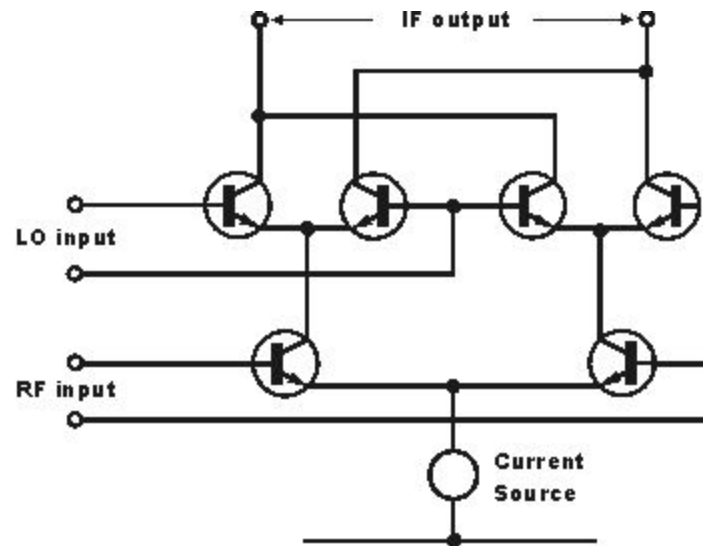


Figure 11: Gilbert Cell Mixer for differential radio use

Modulator

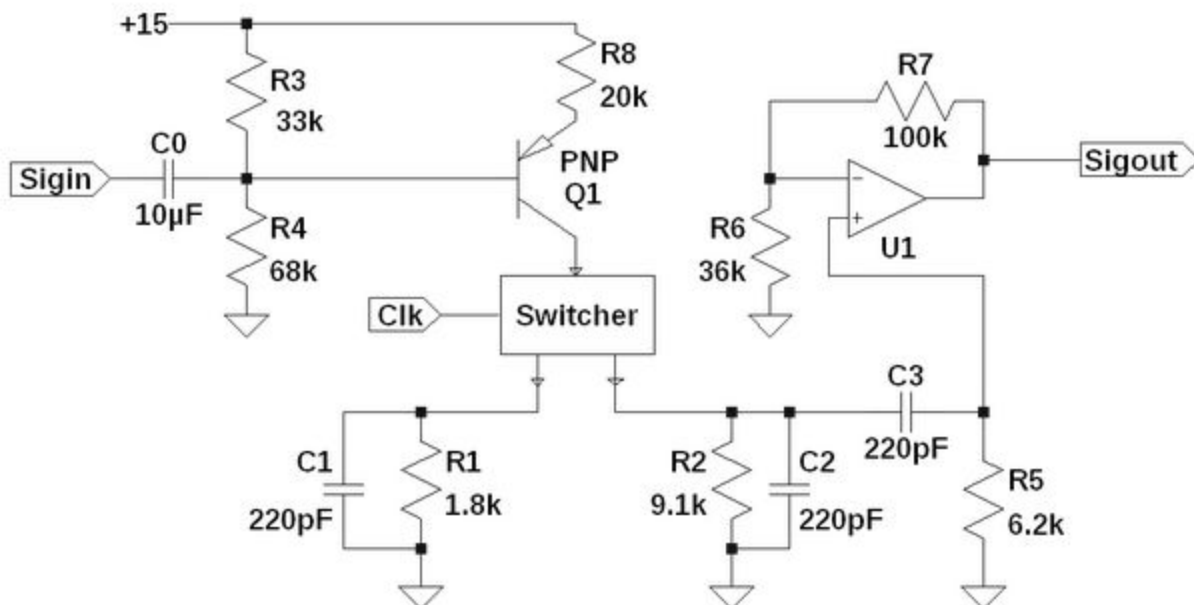


Figure 12: Modulator

To mix a 20 kHz-maximum audio signal up to a center frequency of 101 kHz, we used the modulator topology above with the current switcher at its core. The input signal is coupled through C0 to a positive bias voltage set by R3 and R4, allowing R8 and Q1 to serve as a common-emitter transconductance amplifier. Thus the AC current coming out of each leg of the current switcher is not only proportional to the audio signal voltage entering C0, but also pulses with the clock.

R2 and C2 convert the current into a voltage and low-pass it with a corner frequency of 80 kHz, then C3 and R5 high-pass it with a corner frequency of 116 kHz. Finally, U1 amplifies the modulated signal with a gain of about 3 and buffers it to produce a low output impedance. (There is no particular reason why the low-pass and high-pass filters were designed to exclude each other; it has roughly the same effect as a more conventional band-pass design, but with less gain.)

Demodulator

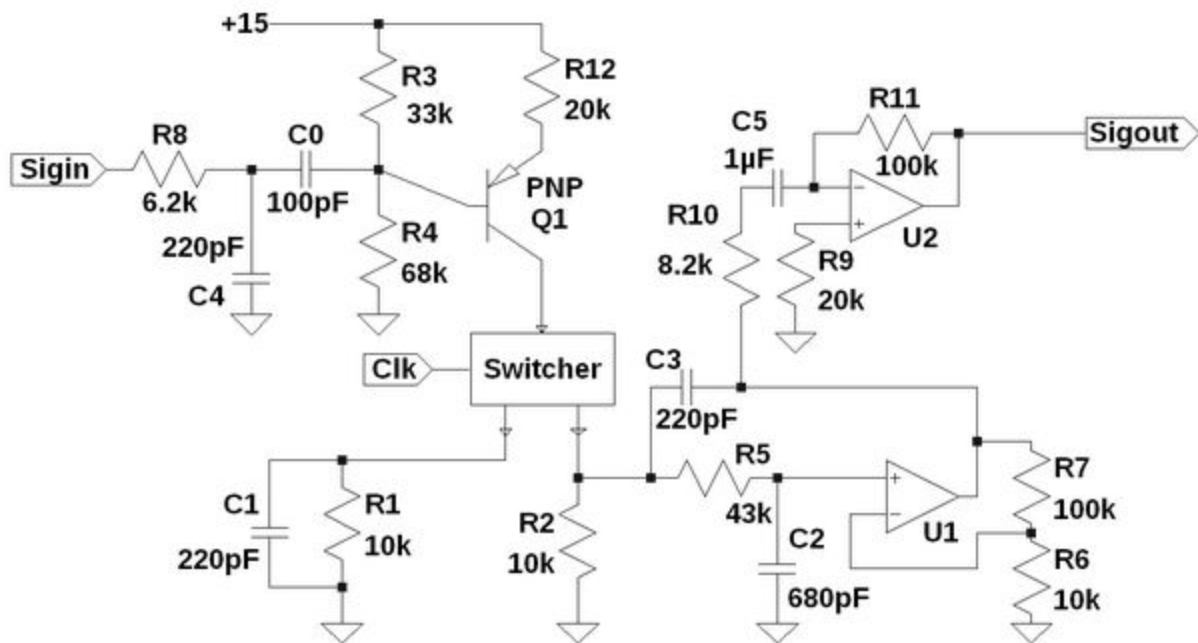


Figure 13: Demodulator

In theoretical frequency domain, the process of demodulation is almost identical to the process of modulation - mixing, followed by a low-pass filter. Our demodulator was designed from that perspective, and has a topology almost identical to the modulator

with the input and output filters changed out. The input filter is now a band-pass filter with corner frequencies of 72 and 116 kHz, and the output filter is now a second-order modified Sallen-Key 20 kHz low-pass filter with a 20 Hz high-pass gain stage. (The high-pass was necessary to avoid amplification of the DC component, which was causing clipping.)

However, our current switcher is not an ideal mixer, and even an ideal mixer can perform inefficient demodulation if its clock is out of phase with the signal it receives. The consequences of these imperfections will be discussed in the next two sections.

System Integration and Testing

Because the system as a whole is one big signal pathway, testing each component individually was not difficult. If any component was incorrectly functioning, it was easy to spot by replacing inputs and outputs with function generators and oscilloscopes.

However, it was harder to deal with the problem of gradual SNR deterioration, not due to malfunctioning or incorrectly-assembled components, but due to suboptimal design. We ran into a number of issues in testing and integration:

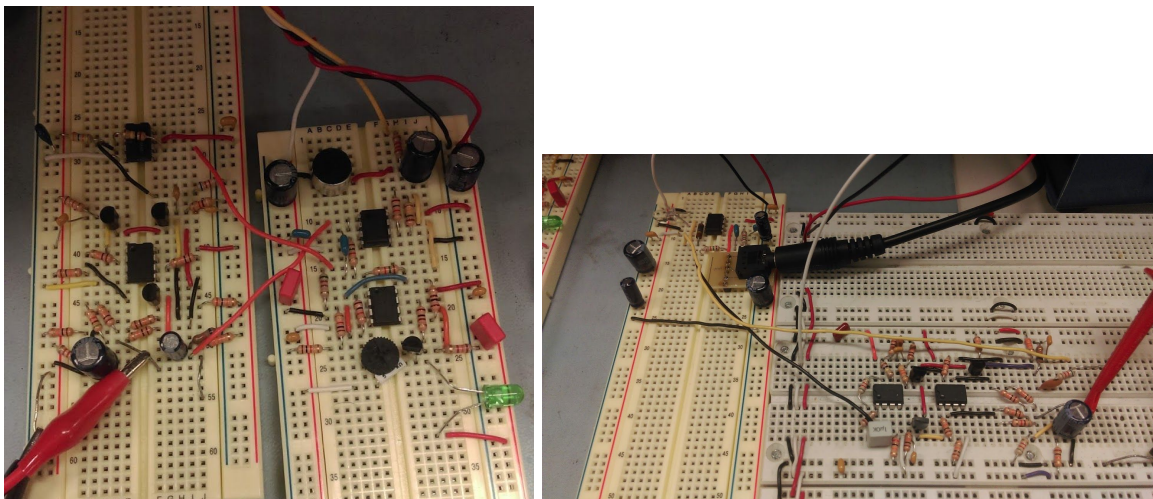


Figure 14: The transmitting (left) and receiving (right) sides of the full assembly

- **Matching of component input and output voltage levels.** It was found, for instance, that the planned microphone output level of 1 V led to about 20% modulation depth, and that the modulator could accept larger voltage inputs without problems. Turning up the gain on the microphone amplifier increased effectiveness.

- **Switching noise from the class D amplifier.** The class D amplifier used to drive the speaker had a switching frequency of 500 kHz, and would cause dramatic spikes in its power supply which would also propagate to the breadboards. Getting a separate power supply helped, but did not eliminate, this problem. However, we do not think this affected audio quality - only waveform quality on the oscilloscope.
- **Radio pickup on long leads.** In the original design, nicknamed the “cheating” topology, there was only one clock, a Schmitt-trigger oscillator that fed both the transmitting switcher and the receiving switcher. However, this required a long wire bridging the free-space gap between transmitter and receiver, and we found that whichever side wasn’t on the clock source side was seeing tremendous fluctuations in the switching output due to radio interference. Thus, we decided to have separate clocks for the transmitter and receiver apparatus, which necessitated the use of function generators to keep frequency consistent.
- **Phase drift between clocks.** Two unsynchronized function generators at almost the same frequency will drift relative to each other. When the demodulating clock is in quadrature with the modulating clock, demodulation produces no output, so we would see our system performance fade in and out at a frequency of about 1 Hz. This could be fixed by manually dynamically tuning the function generator frequency, or by building a phase-locked loop (see “Possible Extensions” below).

(Amusingly, when the function generators were set to be mismatched by a frequency in the audio range like 220 Hz, a clear sinusoidal tone was “demodulated” by the generators shifting in and out of phase, and any audio input was mixed with the 220 Hz tone to produce an interesting voice effect.)

- **An inadequate demodulator output stage.** The demodulator output originally did not have the extra 12x amplifier stage, and was originally a first-order passive lowpass filter, not a second-order active one. Even in individual component testing, this produced unacceptably weak output, with the 101 kHz oscillations dozens of times larger than the audio-frequency signals that we hoped to extract. Even in the final design, the output is not fully cleaned up, with the final pass being done by the speaker amplifier’s input stage.

Performance and Noise Analysis

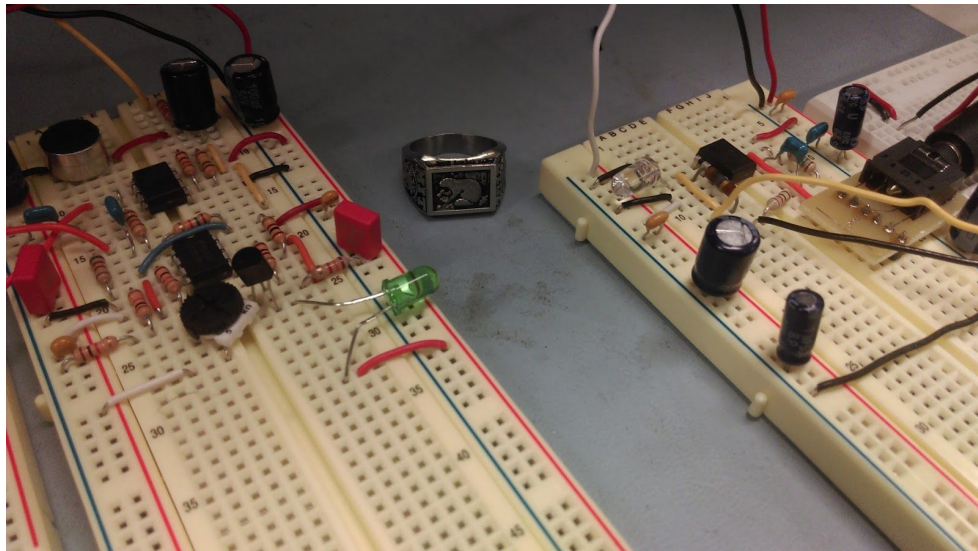


Figure 15: Experimental Set-up for Transmission

Qualitatively, we observed that our system worked when the LED and photodiode were separated by distances on the order of 1.5 inches, transmitting speech over the optical channel. (To test whether the information was transmitting by other means, we blocked the optical channel, and the system stopped working.) However, especially as compared to the low-frequency pipeline on its own, there was a strongly noticeable hiss, just as is characteristic of a real AM radio. (But there was no 60 Hz hum, like there was in the low-frequency pipeline!)

In order to measure the performance of our system, we placed the speaker at least 3 feet away from the circuitry, next to a recording laptop, and recorded the output of the system under various conditions. (In Figures 16 and 17 below, Liz was singing into the microphone, and the speaker output was pulsing as the function generators drifted in and out of phase.)

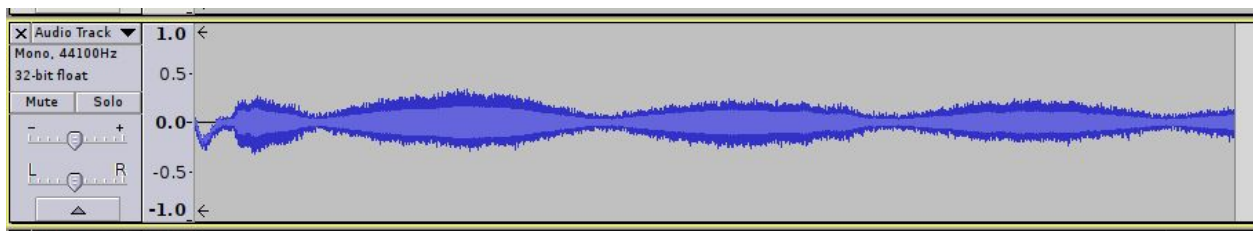


Figure 16: Waveform with a human making a clear tone

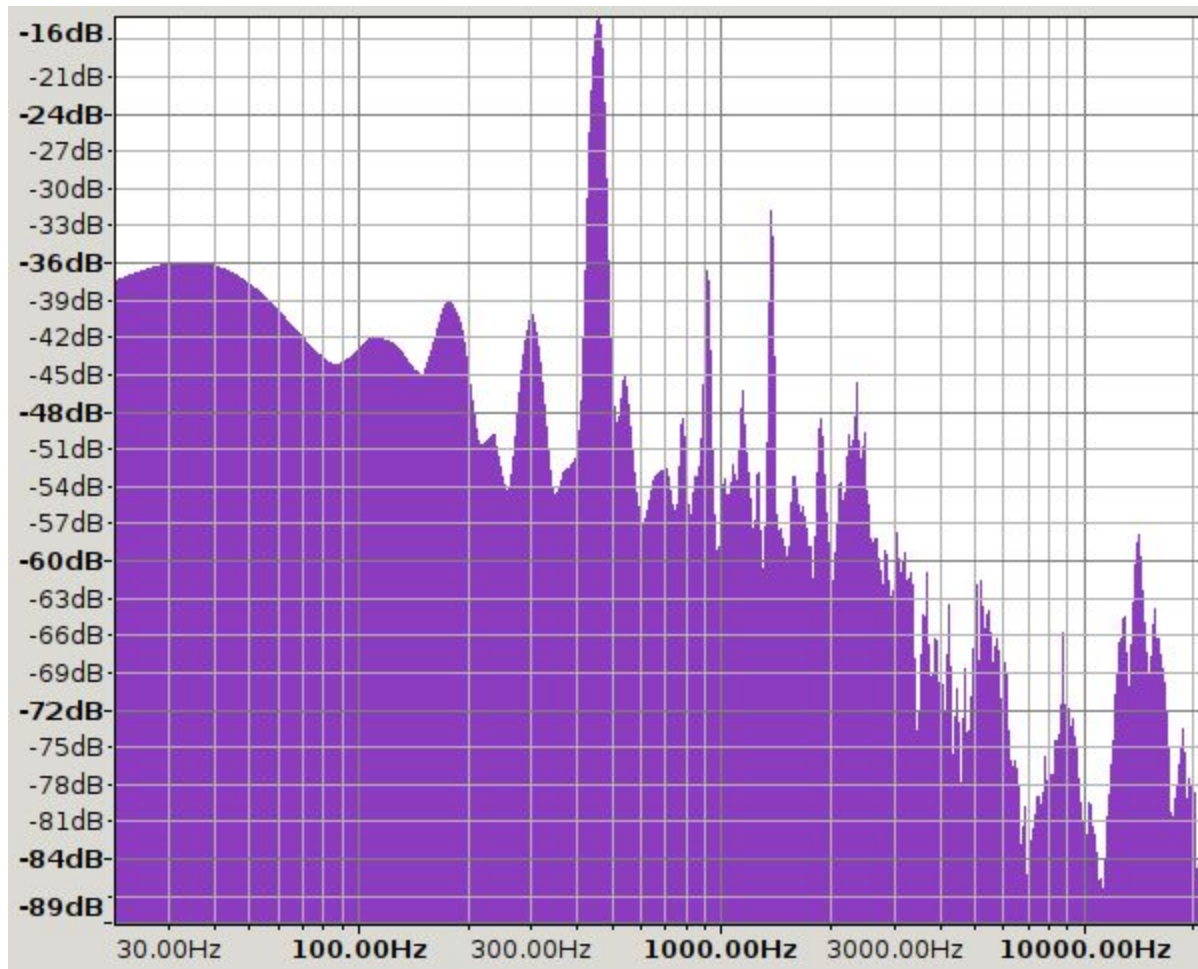


Figure 17: Speaker output with a human making a clear (~440 Hz) tone

The success of the setup was highly sensitive to the alignment of and distance between the photodiode and LED. We were not able to achieve the impressive signal-from-noise extractions that lock-in amplifiers are known for, but we were able to achieve a successful proof-of-concept that could easily be improved upon.

Possible Extensions

Incremental Improvements

Some of the ways in which one could build off of this project include making the frequency or bandwidth tunable, or redesigning the modules to achieve higher levels of precision. In particular, the band-pass filter components in the modulator and demodulator were somewhat haphazardly chosen, and a steeper cutoff and a narrower bandwidth (especially on the demodulator input) would, at the cost of potential audio

quality, significantly increase actual audio quality. Additionally, further examination of the current switcher would probably reveal design changes that would increase stability and decrease ringing.

Reference Oscillator

In our final design, we used two function generators to act as the clocks for the modulator and demodulator. However, the clock on the transmission side could be made using a reference oscillator, such as the topology shown in Figure 18. Because of its built-in 50% duty cycle, the Schmitt trigger topology lends itself nicely to this application. In this case, the clock on the receiving side would be replaced by a voltage-controlled oscillator in a phase-locked loop which would dynamically “keep in sync” with the clock from the receiving side by adjusting the frequency when phase differences were detected. (The mechanism of this feedback could be done in multiple ways, possibly involving building a second demodulator just for quadrature phase detection.)

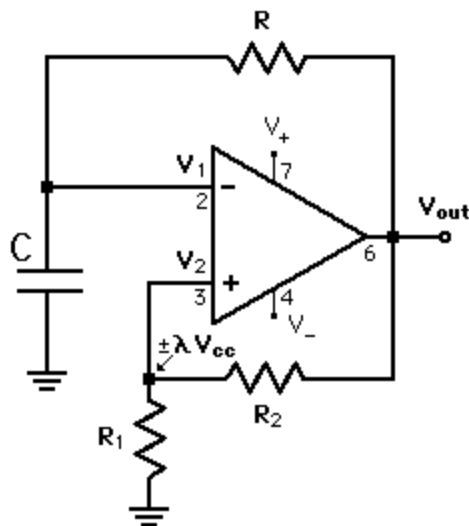


Figure 18: Schmitt Trigger Topology for Reference Oscillator

Frequency Modulation

Our stretch goal was to implement our system with FM in addition to AM. We did not complete this extension of the project, however we did start to make a plan for how it could be implemented with more time. This plan is based off of the block diagram shown in Figure 19 below.

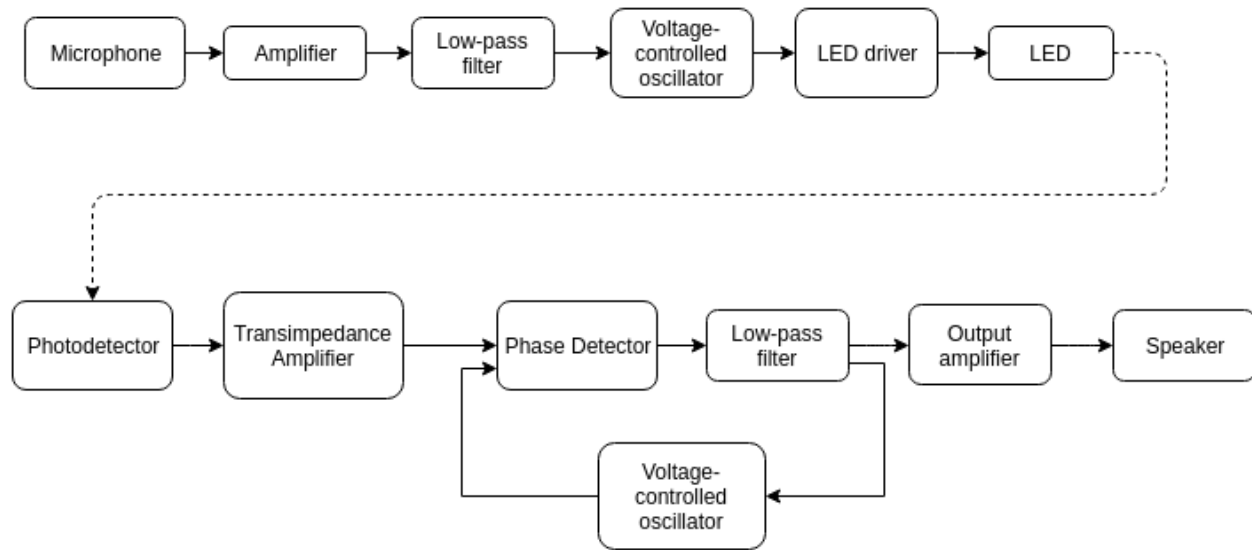


Figure 19: Block Diagram of FM System with Phase-Locked Loop

In the FM topology, like in “Reference Oscillator” above, a phase-locked loop is used to ensure that the receiving voltage-controlled oscillator (VCO) dynamically matches the transmitting VCO. However, in this case, the microphone output is not mixed with anything, but is used to drive the transmitting VCO, and the output is taken directly from the VCO controller, rather than from the demodulator/phase detector.

VCO: We would need a voltage-controlled oscillator whose frequency could fluctuate up or down by approximately 20 kHz as the input voltage fluctuated over its range. The center frequency should be somewhere in the hundreds of kilohertz. We did not decide upon a single topology for the VCO’s, but some options we considered include a voltage-controlled current source with a 555 and a 555 in astable operation with a variable voltage threshold.

Phase-locked loop: The receiver side would require a feedback circuit that would take in the noisy signal from the photodiode and use it to adjust the frequency of a VCO to match. Most implementations of a phase-locked loop rely on a phase detector topology which is similar to the Gilbert-cell mixer we built for the AM topology, but a challenge would be ensuring that the output of the photodiode was well-biased and well-sized to the input of the Gilbert cell. An additional constraint is that the output of the phase-locked loop must be of the right magnitude to drive the VCO. If we used the same VCO here and on the transmitting side, this would mean that our PLL would have to produce a signal of the same voltage range as the signal coming out of the microphone amplifier.

Conclusion

As expected with amplitude modulation, the output sound included the distinctive hiss that is associated with AM radio. However in some tests, we did note a reduction of 60 Hz in the output signal when adding the modulator. We anticipated the largest source of interference to be the overhead lights oscillating at 60 Hz. Performance could be marginally improved by making changes throughout the system to eliminate noise wherever possible, however changing to FM transmission would likely yield a higher increase in performance.

Resources

Figure 2 is from the CUI Inc. datasheet for the microphone. Figure 11 is from Radio-electronics.com.

(<http://www.radio-electronics.com/info/rf-technology-design/mixers/mixer-gilbert-cell.gif>)

Figure 18 is from Hyperphysics.

(<http://hyperphysics.phy-astr.gsu.edu/hbase/electronic/ietron/square.gif>)