Final Project (Part 1): AM Radio

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INTRODUCTION

The goal of this project was to build a small-scale AM radio circuit, with a transmission section and a receiver section. Amplitude modulation (AM) radio is a type of radio transmission which uses a carrier wave at fixed and chosen high-frequency, and the amplitude of this carrier wave is modulated based on the lower-frequency message which is being transmitted (see Fig. 1). This amplitude-modulated signal is broadcast into the environment, and can be picked up an antenna elsewhere, where a receiver circuit can process it to extract the message. In this project, the signal will be broadcast using an LED, and received using a phototransistor.

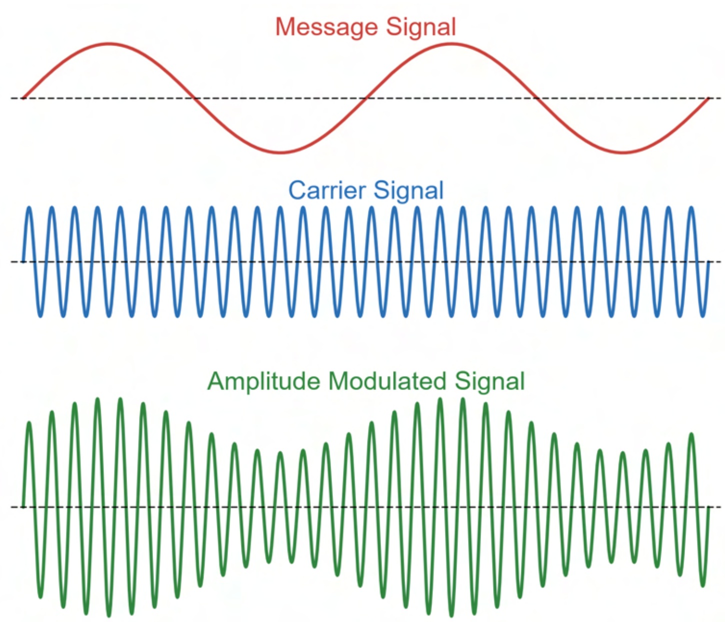

FIG. 1. AM radio signal example

The receiver circuit will pick up other frequencies from its environment in addition to the target frequency, so it must filter out all unwanted frequencies, in order to be left with only the desired amplitude-modulated signal. Furthermore, the receiver will typically receive a weak signal, so it must also amplify it in order to have sufficient voltage to work with. Once the signal is filtered and amplified, the receiver must finally convert the amplitude-modulated carrier signal back into the original message signal. In this project, this is achieved using a peak detector along with a capacitor and resistor to smooth between the peaks.

CIRCUIT DESIGN

Transmitter

First, the amplitude-modulated signal is created by multiplying the lower-frequency message with a high-frequency sine wave of fixed amplitude. In this project, this was achieved using a metal–oxide–semiconductor field-effect transistor (MOSFET). The MOSFET is a device which controls the current from its drain pin to its source pin based on the voltage between the gate and source pins. If the source is connected to GND, and the gate voltage is referenced to GND through a large resistor, then we can vary the gate voltage based on the message signal (see Fig. 2).

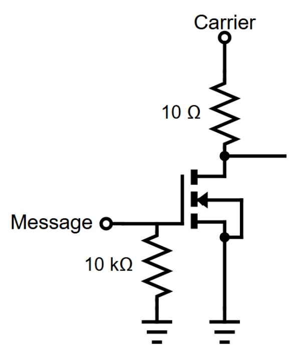

FIG. 2. Amplitude modulation MOSFET design. Typical carrier signal has $f \approx 1$kHz and amplitude 200 mV, with 0 DC. Typical message signal has $f \approx 50$Hz and amplitude 400 mV, with 2.3V DC. These values can vary and still produce desirable results.

Since the drain is connected to the carrier wave input signal through a resistor, then as the current from drain to source is modulated by the message signal, the voltage drop across the resistor will be modulated as well. Therefore, if the carrier wave is oscillating at a fixed amplitude, then the drain pin will contain the desired modulated signal.

This is easier to understand if we think about the MOSFET as a variable resistor. As long as we control the drain-source voltage so that we operate in the linear range (see Fig. 3), then the MOSFET essentially acts as a voltage-controlled resistor, with its resistance value depending on the message voltage at a given time. Understanding this, we see how the fixed resistor and the MOSFET variable resistor create a voltage divider, and as the MOSFET's resistance is varied, the voltage at the drain will change.

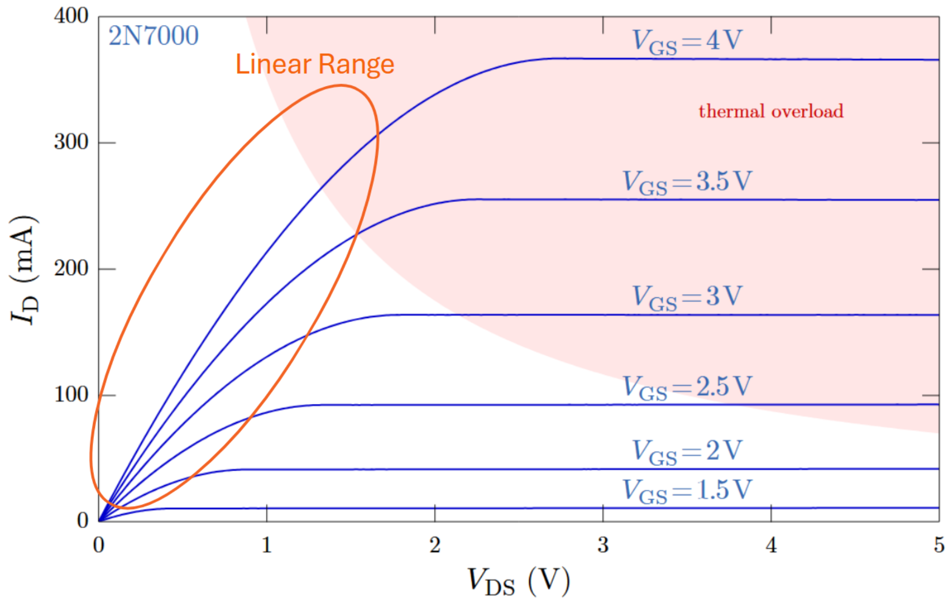

FIG. 3. MOSFET voltage vs. current graph

As we can see from Fig. 3, the gate-source voltage needs to be at least approximately 2V in order to access the linear range. Therefore, since our message signal has an amplitude of a couple hundred mV, we set its DC offset to about 2.3V.

After the modulated signal is created, it is passed through a non-inverting amplifier (see Fig. 4). Since we are operating in the linear range of the MOSFET, the amplitudes used in the first step were only a couple hundred mV. The benefit of amplifying this signal is that a higher amplitude signal will cause low-amplitude noise to create less signal distortion.

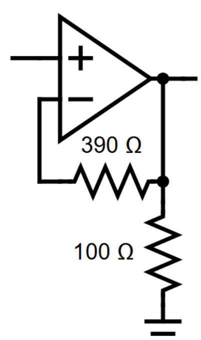

FIG. 4. Non-inverting amplifier, gain = 4.9. For future reference, this will be referred to as amp 1.

After the signal is amplified, it must be given a DC bias. This is because the signal will be broadcast from an LED, which has a set voltage drop across it when it is on (usually about 2V for a green LED, which was used in this project). Therefore, in this project, we used a voltage divider to create an offset of 2.5V, so that there is at least 500 mV of room in each direction for the signal to oscillate, which is sufficient in this case. This offset was achieved using a simple voltage divider, after the signal had been sent through a capacitor to remove any unwanted previous DC bias (see Fig. 5).

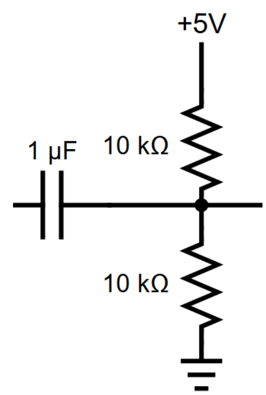

FIG. 5. DC biasing stage. Provides 2.5V DC.

Finally, before the signal is sent into the LED, it must be buffered through an op-amp. If this buffer is not present, then the LED will draw current from VCC (since the voltage divider runs from VCC to GND). If this happens, then it will affect the voltage drop across the resistor in the voltage divider, and will therefore lead to the DC offset being changed. Therefore, after the voltage divider, the signal is buffered, and then sent through the LED with a small current-limiting resistor in series (see Fig. 6).

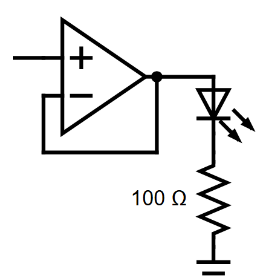

FIG. 6. Buffer and LED

Receiver

The signal is received via a phototransistor. The base pin of the phototransistor is disconnected, because in this case, the number of incoming photons act as the control for the collector-emitter current. If the emitter is grounded, and the collector is connected to VCC through a resistor, then the voltage drop across the resistor will vary based on the amount of incoming photons (see Fig. 7).

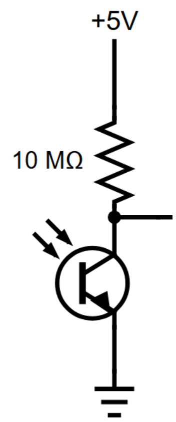

FIG. 7. Phototransistor setup

Through rough experimentation, we found that the signal from the LED varied the collector-emitter current by an extremely small amount. To compensate for this, we needed to use an extremely large resistor, as can be seen in Fig. 7 (since $V = IR$, we need a large resistance to get a reasonable voltage drop if current is small). This large resistor affects the next stage of the circuit, as follows: Generally, the signal from the collector can just be fed into the positive terminal of an op-amp, since the op-amp draws almost no current. However, since the physical op-amp actually does draw nano-Amps of current, then this will not work in this case. This is because nano-Amps of current flowing through the extremely large resistor in the previous step causes significant distortion in the signal. Therefore, before feeding the signal into the op-amp, the current must be given a lower-impedance path to GND, so the op-amp does not draw any current). Therefore, before the amplifier, we added a capacitor to block any unwanted DC, followed by a resistor to GND. The signal was then taken from above this resistor, and fed into the amplifier (see Fig. 8). This amplifier has very high gain, since the incoming signal from the phototransistor is quite small.

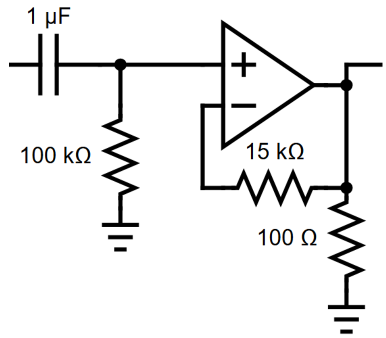

FIG. 8. DC blocking + amplification stage, gain ≈ 150. For future reference, this will be referred to as amp 2.

After the signal was amplified, it needed to be filtered, as discussed in the introduction. In this project, the carrier wave amplitude was chosen to be 1 kHz. Therefore, we used a band-pass filter, which had cutoff points of about 880 Hz and 1320 Hz. Therefore, any low-frequency signals (such as the 60 Hz flickering of the lights in the room) would be filtered out, and any high-frequency noise would be filtered out as well. The bandpass filter was simply constructed using a standard single-pole passive high-pass filter followed by a low-pass filter of the same type. Before and between these filters, the signal was buffered to prevent any impedance matching problems between circuit stages (see Fig. 9). An important note: Since there were few unwanted signals being picked up by the receiver in this project, then passive, single-pole filters were sufficient. However, if there is a scenario in which more precise tuning is needed, then one would likely have to increase the number of poles of the filters, and/or make them active as well. Alternatively, using a different type of filter entirely may be preferred (such as an LC tank filter).

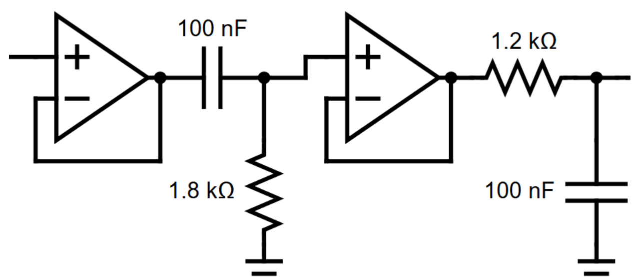

FIG. 9. High-pass filter ($f_c \approx 880$Hz) followed by low-pass filter ($f_c \approx 1320$Hz)

Once the signal is filtered, it needs to be amplified once more due to the fractional gain of the passive filters. Also, in this case, it needed to be amplified by a large amount, to prepare the signal to be able to handle the upcoming 0.6V diode drop (this diode being part of the peak-detector circuit). This stage is shown in Fig. 10).

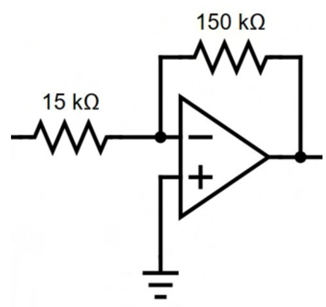

FIG. 10. Inverting amplifier, gain = 10. For future reference, this will be referred to as amp 3.

The final stage is the half-wave rectifier and peak detector. Before sending the signal through the diode to cutoff the negative side of it, we gave it a slight DC bias using a voltage divider (about 0.5 V), so better prepare it for the 0.6V diode drop (see Fig. 11). The reasoning for this is because some parts of the signal were just below 0.6V after the previous amplifier, so they needed to be raised slightly. In general, the amount of DC in the final signal is not important, since it can easily be eliminated.

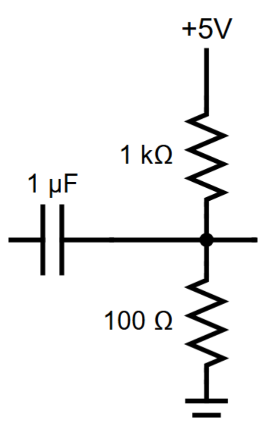

FIG. 11. DC biasing stage. Provides ∼0.5V DC.

Finally, the signal is sent through the peak-detector circuit, consisting of a diode to cutoff the negative portion followed by a capacitor and resistor to smooth between the peaks (see Fig. 12).

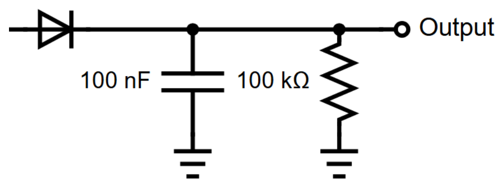

FIG. 12. Peak detector stage

One important note is as follows: The capacitor and resistor values in the peak-detector must be chosen strategically. If the product RC is too small, then there will be little smoothing, and the final signal will look choppy. If the product RC is too large, then the signal will be smoothed well near any message peaks, but will be extremely distorted near the troughs. This is because if the smoothing is very flat, then none of the points near the message troughs will be detected, since the gradual smoothing will cause the signal to run above any peaks near the troughs. This is best explained through Fig. 13.

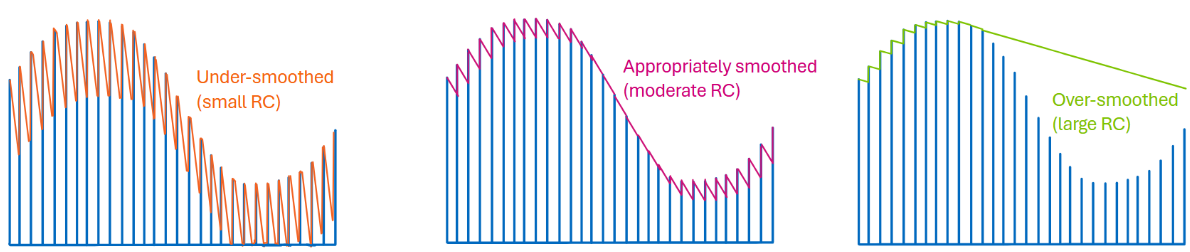

FIG. 13. Peak detection, varying RC time constants

We can see from the Fig. 13 that using this method of smoothing, there will still be some roughness to the final signal. If we needed a more smoothed out final signal, it would be best to use an alternative method of smoothing, or to send the signal through further processing after being roughly smoothed via this method.

Summary

Shown in Fig. 14 is a concise summary of the entire AM radio circuit. Fig. 15 is the full schematic of the circuit, with each stage labeled accordingly. Fig. 16 shows the full breadboard of the circuit, with each stage labeled.

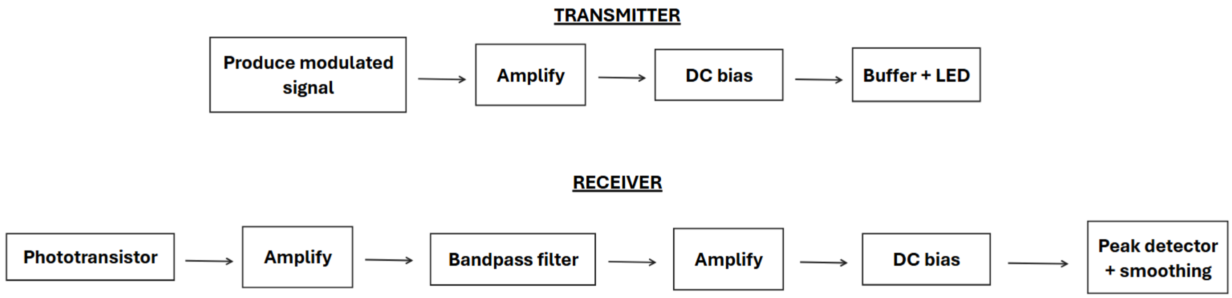

FIG. 14. Summary of AM radio circuit stages

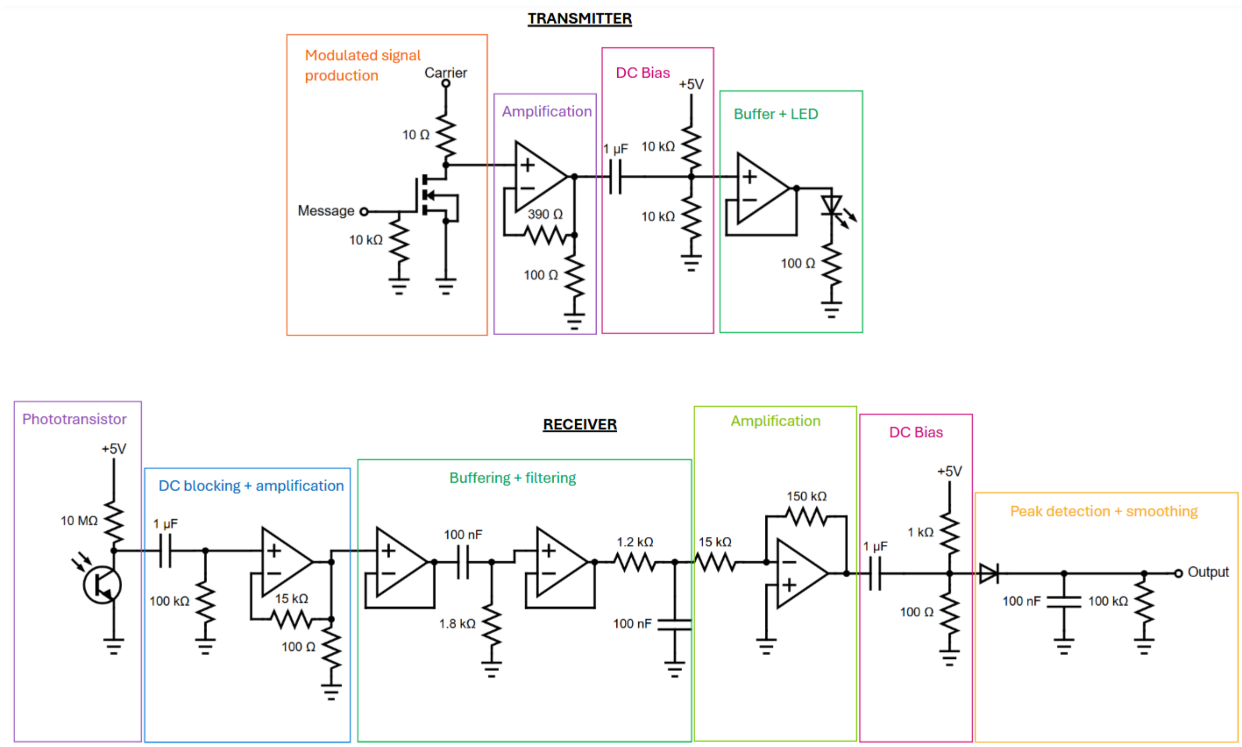

FIG. 15. Full circuit schematic with labels

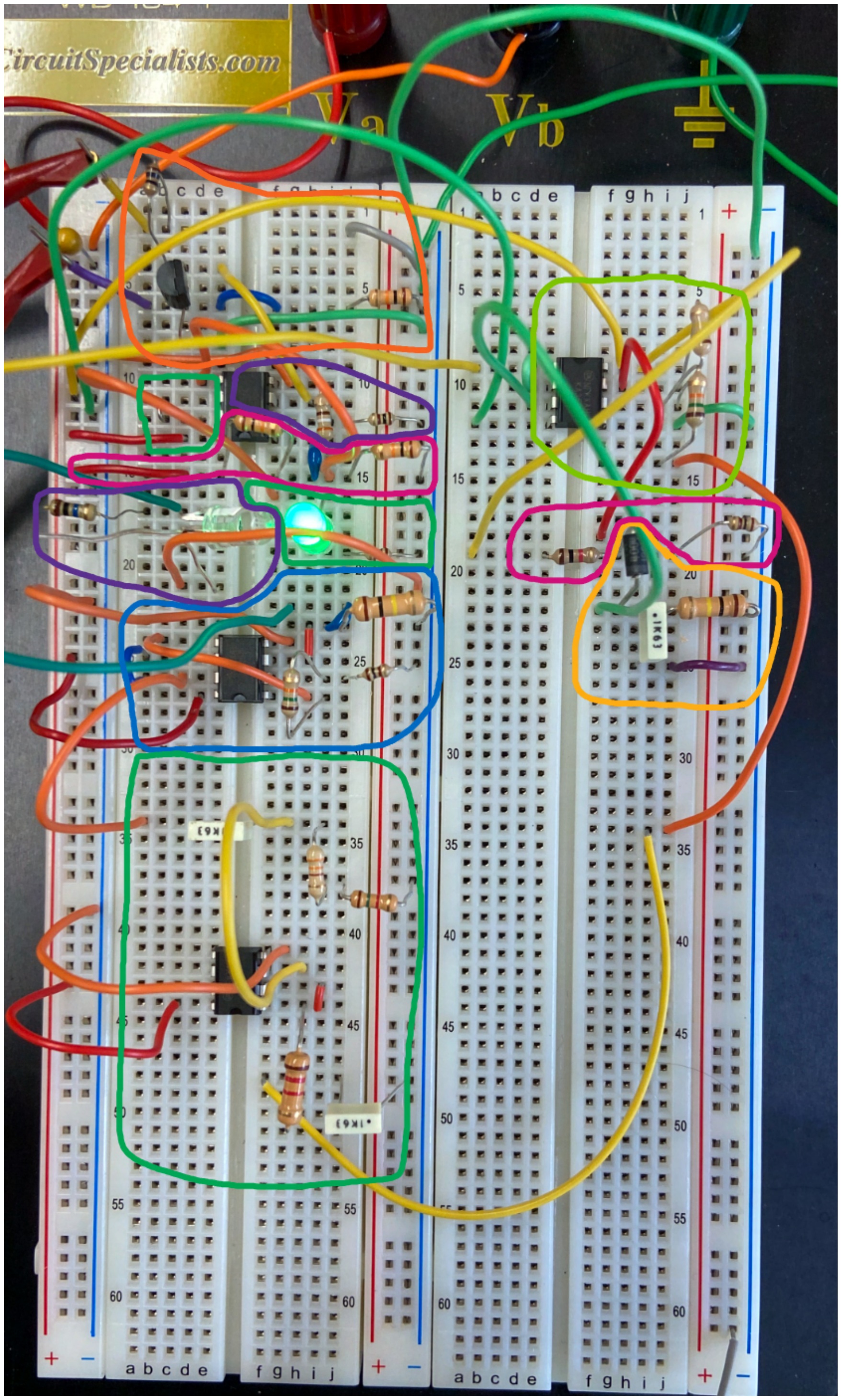

FIG. 16. Full breadboard of AM radio. Stages are labeled using the same color scheme as Fig. 15.

RESULTS

Transmission Stage

Initially, we set the carrier signal to 1 kHz, and the message signal to 40 Hz (sinusoidal message shape). We see that this produces the expected waveform shape (see Fig. 17); specific measurements will be discussed in the following sections. When observing the general shape of the output of the first circuit stage, we see that it has the expected sinusoidally modulated signal.

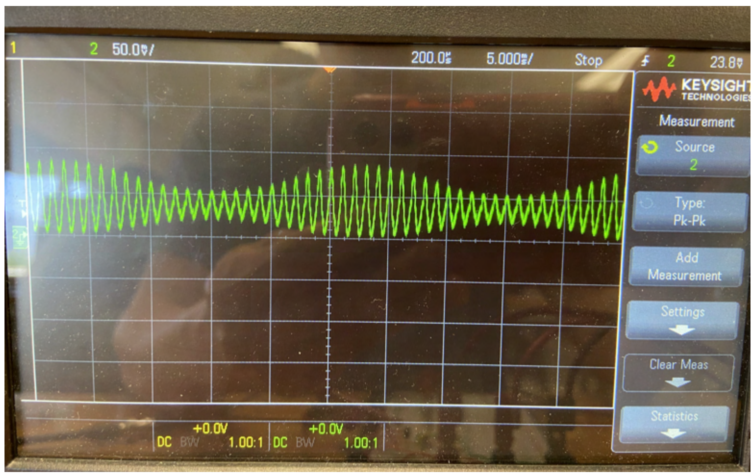

FIG. 17. Modulated signal. $f_{\text{carrier}} = 1$ kHz, $f_{\text{message}} = 40$ Hz.

We also see that the process is robust. If we vary the frequency of the message (see Fig. 18) or the carrier (see Fig. 19), we observe that the modulated waveform responds accordingly.

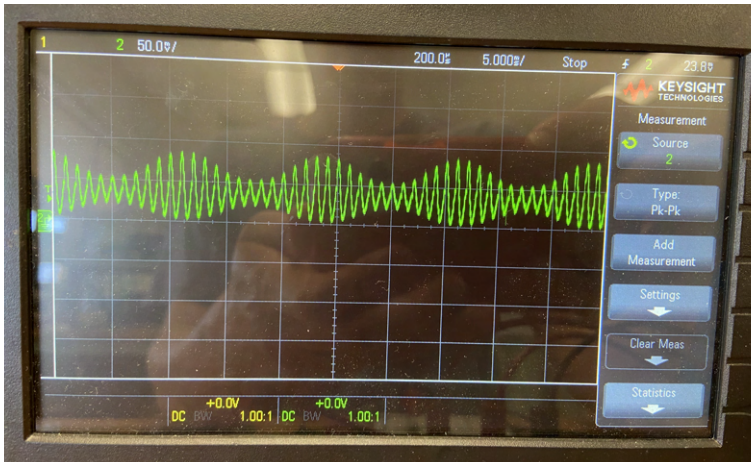

FIG. 18. Modulated signal. $f_{\text{carrier}} = 1$ kHz, $f_{\text{message}} = 80$ Hz.

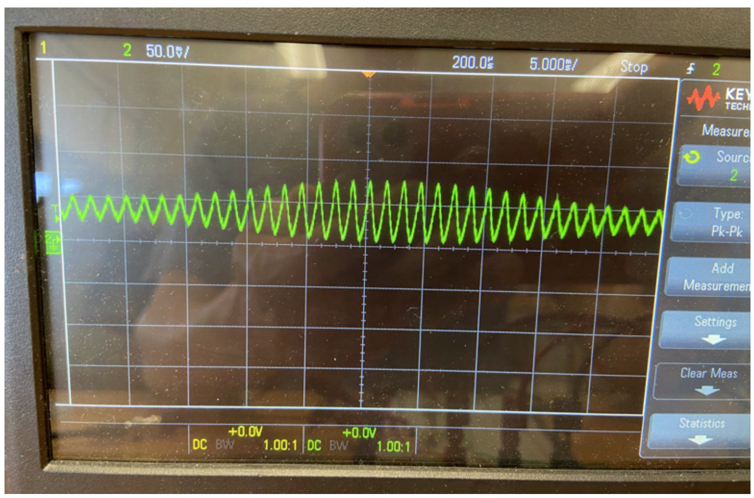

FIG. 19. Modulated signal. $f_{\text{carrier}} = 700$ Hz, $f_{\text{message}} = 20$ Hz.

Before this modulated signal gets sent through amp 1, we measure that it has a peak-to-peak voltage of roughly 160 mV. At the output of the amplifier, we measure that the signal has a peak-to-peak voltage of roughly 750 mV. This corresponds to a gain of roughly 4.7, which is within 10% of the expected gain of 4.9.

Observing the signal after it is DC biased (before being sent through the LED), we see that it has a DC offset of about 2.4 V, which is within 10% of the expected 2.5 V offset. The final signal which is sent through the LED is shown in Fig. 20.

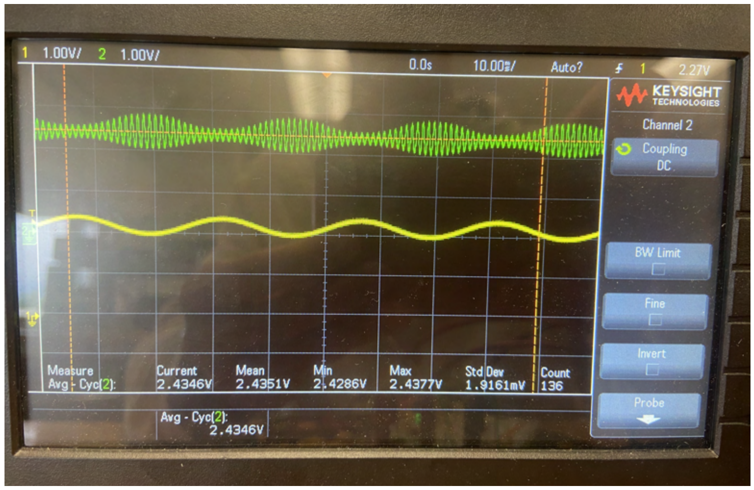

FIG. 20. Signal which is sent through LED (green), along with original message signal (yellow).

Receiver Stage (Phototransistor + Amplifier)

After this signal gets broadcast from the LED, the phototransistor picks up the signal along with other environmental signals and noise. Observing the phototransistor output signal after its DC gets blocked, we measure its peak-to-peak voltage to be about 5.8 mV (see Fig. 21). At the output of amp 2, we measure the peak-to-peak voltage to be roughly 770 mV (see Fig. 22). This corresponds to a gain of roughly 130.

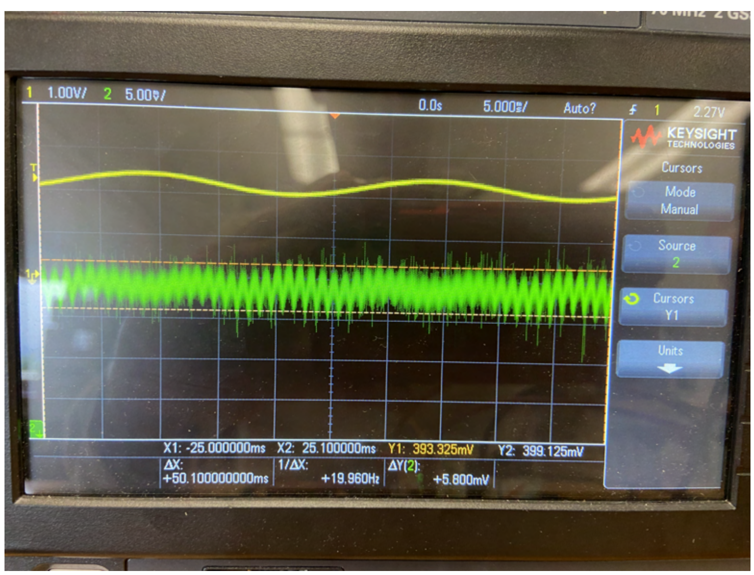

FIG. 21. Input signal to amplifier 2.

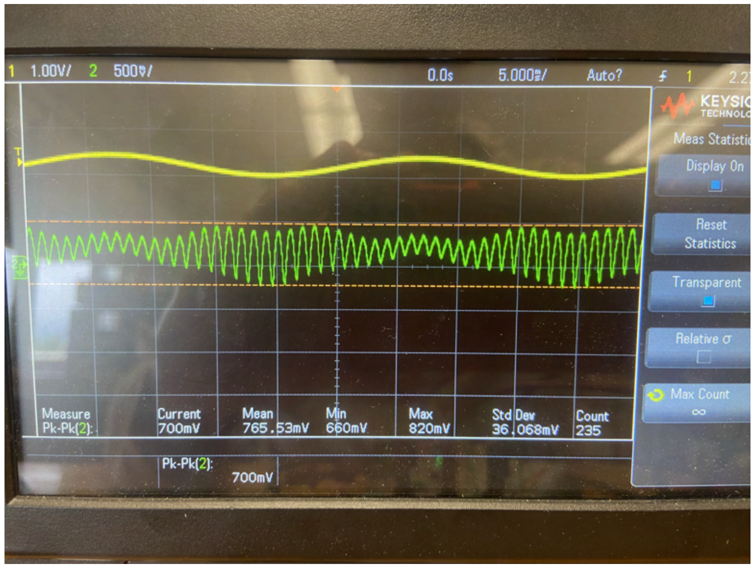

FIG. 22. Output signal of amplifier 2.

This measured gain is not quite within the 10% of the expected gain of 150. This may be due to the large amount of noise in the pre-amplified signal. Looking at Fig. 21, we can see that there is significant noise in the signal, which may have led to inaccurate measurements of the signal's amplitude. For example, if the noise adds just 0.3 mV on each side of the signal, then the actual signal amplitude would be closer to 5.2 mV, which corresponds to an amplifier gain of about 150. Therefore, we see how just a small amount of noise can have a large effect on the measured gain, due to the fact that the initial incoming signal from the phototransistor is small itself.

Observing the post-amplified signal from the phototransistor in Fig. 22, we see that its general shape looks like a superposition of multiple sinusoids. This is expected, since we have a high frequency carrier being modulated at a low frequency, along with the room lights emitting a sinusoidal signal at a different low frequency (60 Hz). Therefore, the incoming signal will include multiple different low frequencies, leading to a signal which appears to be a superposition of sinusoids. This will be investigated further in the following section.

Filtering stage

Next, this signal gets sent through the filtering stage. At the output of the filtering stage, we would expect to see our original amplitude-modulated signal (similar to the signal in Fig. 20), since the filters should eliminate any other signals coming from the environment (such as the 60 Hz room lights).

Observing the filter output, we see that its shape does indeed appear to be quite similar to the original modulated signal, as expected (see Fig. 23), however this will be investigated more quantitatively below, through the use of the Fast Fourier Transform (FFT), which is a tool used for displaying the various frequencies present in signal.

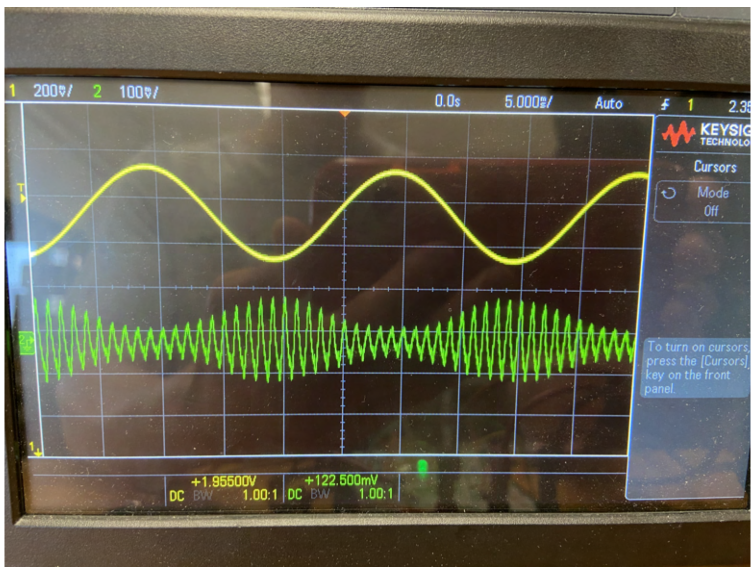

FIG. 23. Output signal of bandpass filter.

Performing an FFT on the pre-filtered signal (see Fig. 24), we see that there is a peak at 1 kHz (which is expected, as this corresponds to the carrier frequency). and that there is also a large peak around the low 0 to 100 Hz range (which corresponds to the message frequency as well as the frequency of the light in the room). Interestingly, there is also a peak at 2 kHz, however this can be explained by the fact that when working with non-linear circuit components (such as diodes or amplifiers), it is possible for higher-order harmonics of the signal to be produced. For example, the slight noise which can be seen near the troughs in Fig. 23 may contain sharp peaks, and these can contain higher-order harmonics.

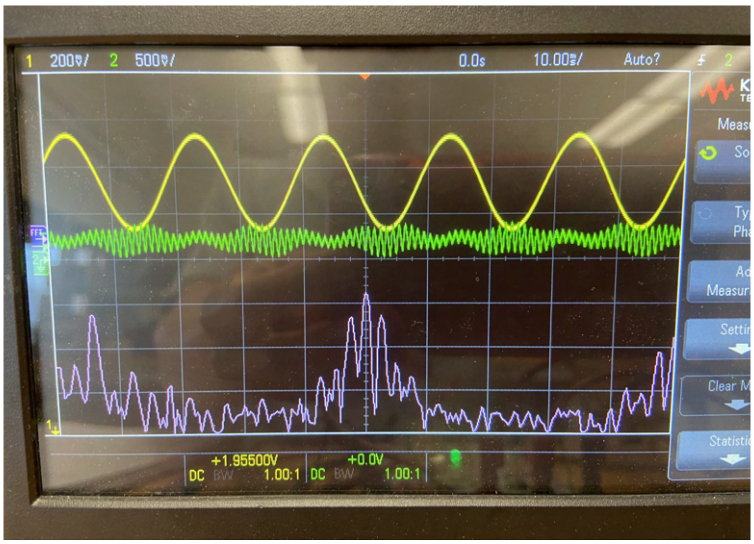

FIG. 24. Fourier transform (purple) of pre-filtered signal. The range of frequencies shown on the graph is 0 to 2 kHz, centered at 1 kHz. $f_{\text{carrier}} = 1$ kHz and $f_{\text{message}} = 100$ Hz.

When performing an FFT on the filtering stage output (see Fig. 25), we still observe a peak at 1 kHz, which is expected since the filters are designed to pass the 1 kHz carrier frequency. However, we also still observe a peak at 2 kHz, which strengthens the theory that this frequency is baked into the signal as noise rather than being picked up externally. If it was external, it would have been filtered out easily since the 2 kHz signal would simply be superimposed with the LED signal. However, if the 2 kHz is instead baked in as noise, then it would be much harder to filter out. Also, interestingly, we still observe a peak at the low frequencies, even though they should have been filtered out. Therefore, we decide to observe the low frequencies more precisely to try to figure out what may be causing this unexpected result.

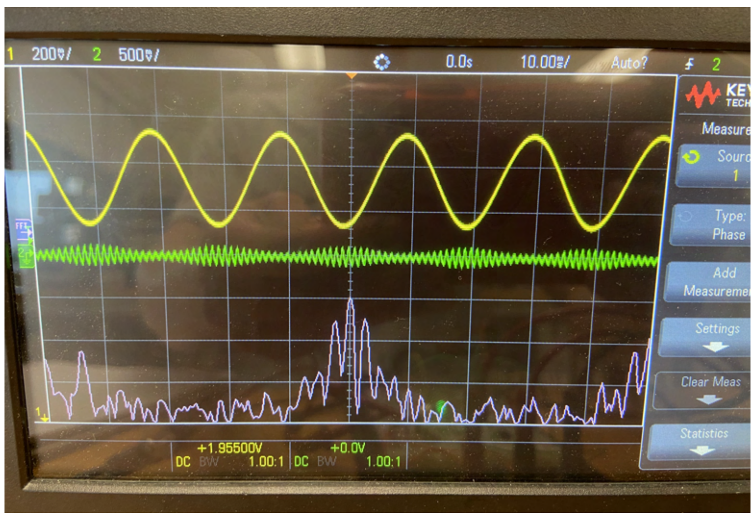

FIG. 25. Fourier transform (purple) of filter output. The range of frequencies shown on the graph is 0 to 2 kHz, centered at 1 kHz. $f_{\text{carrier}} = 1$ kHz and $f_{\text{message}} = 100$ Hz.

When performing this more precise FFT on the pre-filtered signal once again (see Fig. 26), we gain a better understanding of the peak at low frequencies. We can now see that there are actually three peaks in this area; one at 60 Hz (which corresponds to the room light frequency), one at 100 Hz (which was the message frequency in this test), and one at 120 Hz. The peak at 100 Hz may initially seem expected, however recall that the message is only supposed to be encoded in the modulated carrier wave, so there should be no message frequency present. Therefore, this implies that there may be some sort of leakage, allowing the message frequency to pass into the LED. This may be due to parasitic capacitance, which can be present in MOSFETs between the gate and the other pins. This may cause some small but finite amount of the 100 Hz signal to "leak" into the source or drain, and therefore make it into the LED, and into the rest of the circuit. This theory is supported by the fact that even when probing the output of the MOSFET at the beginning of the circuit, the 100 Hz still shows up on the FFT. However, this is likely not an issue, as this frequency should simply get filtered out by the filtering stage (and is likely small to begin with). Regarding the peak at 120 Hz, it may initially seem unexpected, since there should be no 120 Hz being picked up from anywhere in the environment. However, this is the exact ripple frequency of any full-wave rectified signals using the 60 Hz AC wall power. Therefore, this 120 Hz may be coming from the power supply, and therefore may be present all throughout the rails of our circuit. We can learn more about this peak by looking at the FFT of the filter output (see Fig. 27).

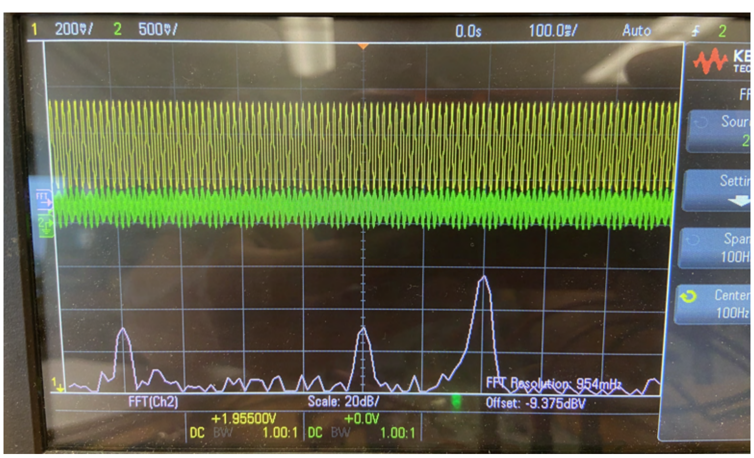

FIG. 26. Fourier transform (purple) of pre-filtered signal. The range of frequencies shown on the graph is 50 Hz to 150 Hz, centered at 100 Hz. $f_{\text{carrier}} = 1$ kHz and $f_{\text{message}} = 100$ Hz.

When observing the FFT of the filter output (see Fig. 27), we see that both the 60 Hz and 100 Hz peaks have disappeared, as expected. However the 120 Hz peak is being created within the circuit. In fact, to learn more about the 120 Hz, we probed various points of the circuit, and found that it only starts to show up at the output of amp 2. This leads to the following hypothesis: The power supply produces very small 120 Hz ripples. Since the phototransistor is drawing from the power rail (through the 10M resistor), then the incoming signal from the LED is actually being modulated at a very small scale by this 120 Hz. However, this 120 Hz is sharp, like a sawtooth wave rather than a sine wave, so therefore, it is baking some sharp imperfections into the output signal of the phototransistor. When this signal is amplified by 150 times (via amp 2), this 120 Hz becomes noticeable and shows up on the FFT. Similarly to how the sharp 2 kHz noise was baked into the signal, this sharp 120 Hz noise is as well, and therefore cannot be filtered out.

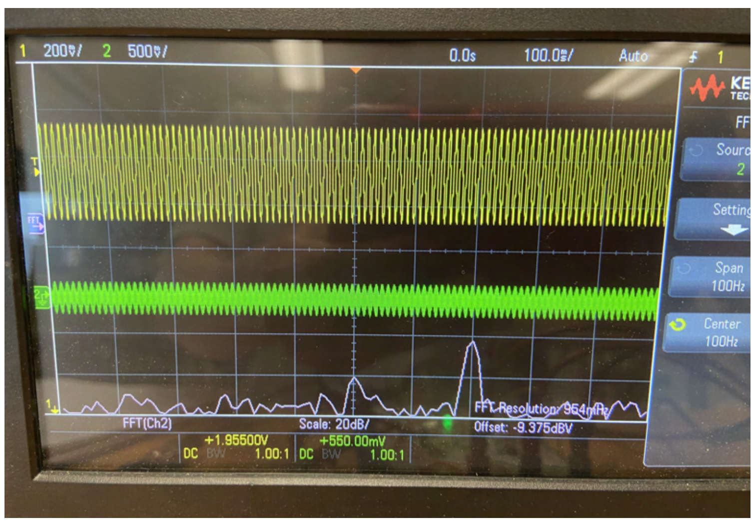

FIG. 27. Fourier transform (purple) of filter output. The range of frequencies shown on the graph is 50 Hz to 150 Hz, centered at 100 Hz. $f_{\text{carrier}} = 1$ kHz and $f_{\text{message}} = 100$ Hz.

Reconstruction Stage (Amplifier + Peak Detector)

Next, the output of the filter goes into amp 3. We measured that the peak-to-peak amplitude of the pre-amplified signal was about 220 mV, while the amplitude of the post-amplified signal was about 2.2 mV. This corresponds to a gain of 10, which is exactly the expected gain of this amplifier.

After the signal is amplified, it gets sent through the diode to half-wave rectify it (after getting slightly DC biased upwards). If the smoothing capacitor after the diode is removed, we get the output shown in Fig. 28. We see that this has the expected half-wave rectified shape (where the bottom half is cut off), but we also see that there is some noise in the signal. This may involve the 2 kHz or 120 Hz noise discussed earlier. However, this noise should not affect the final output, since the only feature of this signal which matters for production of the final output is the positions of the peaks. As long as the peaks are placed as expected, then we should get the expected final output.

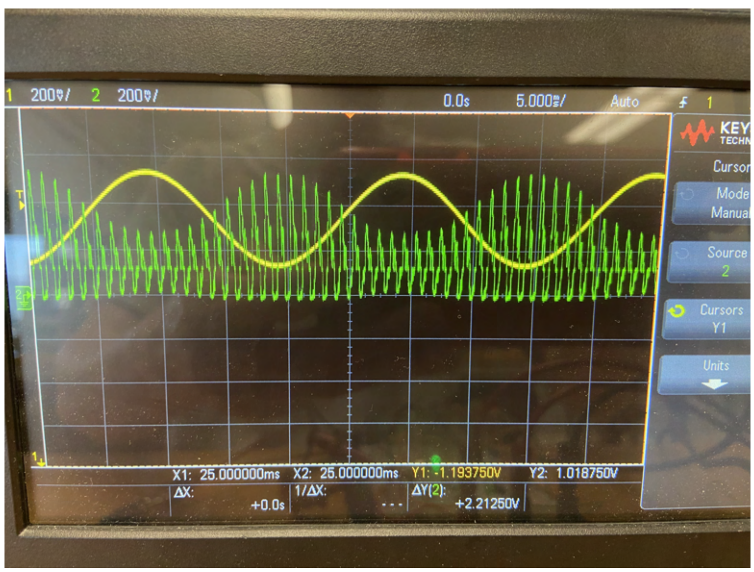

FIG. 28. Output of diode, with no smoothing capacitor present.

Finally, when the smoothing capacitor is present, we get our final reconstruction of the original signal (see Fig. 29).

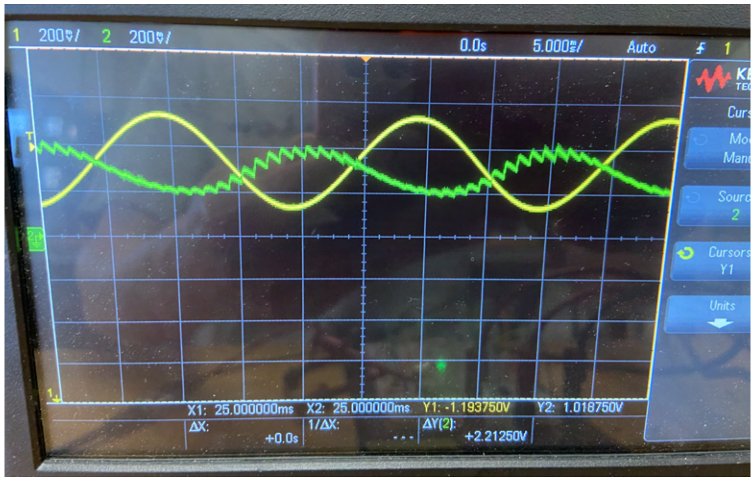

FIG. 29. Final message reconstruction (green) along with original message signal (yellow)

As discussed in Fig. 13, the capacitor and resistor values in the smoothing section are important. Through experimentation, we found that we obtained the most reasonable smoothing using $R = 100\ \mathrm{k}\Omega$ and $C = 0.1\ \mu\mathrm{F}$. Interestingly, this corresponds to an RC time constant of 10 ms, which is 10 times the period of our carrier. While it may initially seem like this is quite large, it is important to remember that the optimal time constant depends on the amplitude of the half-rectified signal. If the amplitude is large (so that the derivative is very large in some places), then the optimal RC time constant will be smaller, since the slope has to be great enough to follow that large derivative in some places. However, if the amplitude is smaller, then the maximum derivative will be smaller than in the previous case, so the RC time constant can be larger. Because the optimal RC time constant depends on the amplitude, and because we arbitrarily amplified the signal just enough to get good signal, then it makes sense that the optimal RC time constant would not necessarily correspond to the period of the carrier frequency.

Analyzing this final output, we see that its frequency almost perfectly matches the frequency of the original message signal (see Fig. **??**). We also see that when using this message frequency of 50 Hz, the final output is about 150° out of phase from the original signal, as opposed to the 180° phase difference which may initially seem expected. However, this seemingly random phase difference actually makes sense; since capacitors induce phase shifts, then the signal's phase got shifted several times as it passed through the circuit.

However, the phase of the final signal is irrelevant in a radio; the goal is for the final output to have the same shape as the input, and therefore contain the same information. However, as mentioned in Fig. 13, the simplified method of smoothing used in this project does not allow for perfect reconstruction of the original signal, which is why our final output still contains some small roughness.

CONCLUSION

In conclusion, this project successfully demonstrated the basic principles of AM radio communication using analog components. The system was able to encode a low-frequency message onto a high-frequency carrier wave, transmit it over an LED, receive it from a phototransistor, and process/decode it using amplifiers, filters, and a peak detector circuit.

The reasoning, design, and analysis of each stage of the circuit was addressed theoretically, and tested experimentally. Ultimately, the output wave closely approximated the input message, validating the design choices. Future improvements could include more sophisticated filtering, better noise control, and/or alternative demodulation methods to decrease signal imperfections and increase signal quality.