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6.101 Final Report
NTSC Transmission through Optic Fiber

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Abstract

The National Television Standard Committee was developed in 1941 and had no provision for color. In 1953 a second NTSC standard was developed to allow for color TV transmission. It was widely used up to 2010, when it gradually became replaced with different digital standards. For our final project, we will implement a scheme to transmit and receive NTSC video and audio signals through optic fiber. The transmission of video and audio signals across optic fiber have been done using digital transmission standards; our project will take an old standard and transmit it across a new medium. The project will be divided into a transmission unit and a receiving unit. The transmission unit will include an audio modulator, and a multiplexer. The multiplexer signal will be transmitted and received via an optical fiber medium. The receiving unit will consist of the necessary filters to demodulate and split the received signals into separate audio and video signals, which will be displayed on a television unit provided by the lab.

1 Design Overview

Our goal is to build a system that will transmit an NTSC signal from a camera through optic fiber. The output from the camera will be displayed on a TV unit. This is a way to emulate a real TV station that broadcasts at different frequencies.

The scheme that was followed is depicted in Figure 1. The acquired signal from the camera consists of two parts: a video signal and an audio signal. The audio signal is FM modulated on a carrier frequency of 16MHz; the video signal is passed into the system as-is. These signals are multiplexed, or combined into one signal, and transmitted through an optic fiber. At the receiving end of the optic fiber, a demultiplexer separates the audio and video signals. The audio signal is demodulated at this point. Finally, both signals are amplified to the proper output format and then sent to the TV.

Germain, Hugo and Khaled divided the project into three parts: modulation, demodulation, and transmission and receiving. Germain worked on the modulation component which includes frequency modulation of the audio signal. Hugo was responsible for the transmission and receiving component of the circuit. Khaled was responsible for the demodulation/amplification component, which includes the detection of the video and audio signals, amplification and outputting of final signals.

2 Block Diagram

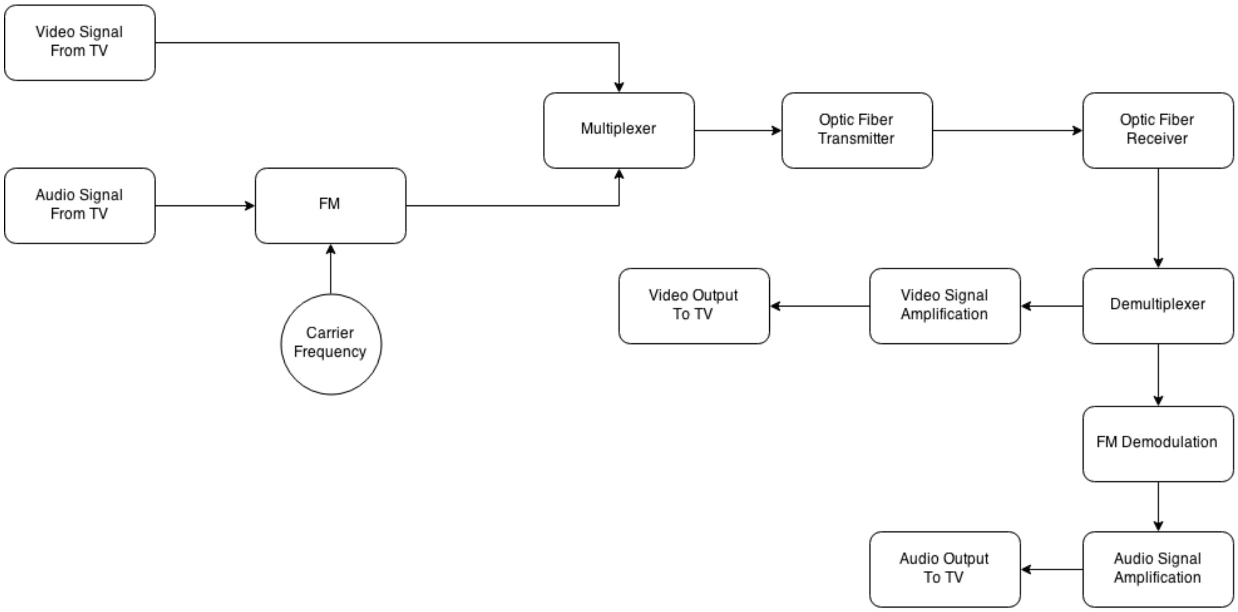

Figure 1: High level system block diagram detailing major sub-blocks and inter connections between them.

3 Modulation by Germain Martinez

The goal of the modulation part of the circuit is to convert the audio signal coming from the RCA connector of the camera into a frequency-modulated signal. This signal will then be combined with the video signal and then get sent across the optic fiber channel. The circuit that handles this entire operation is shown in Figure 2.

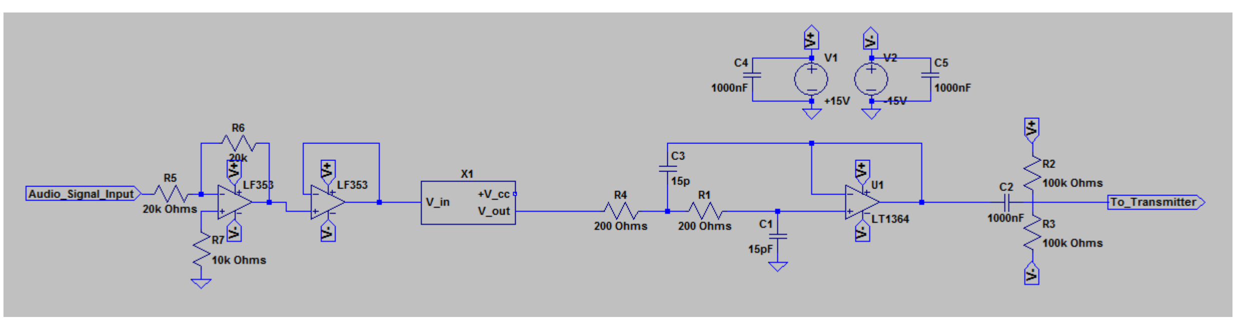

Figure 2: The circuit topology for the modulation block. X1 is the frequency modulation circuit, which is fully depicted in Figure 4.

The analysis of this circuit is broken down into three stages: the buffer stage, the modulation stage, and the filtering stage.

3.1 The Buffer Stage

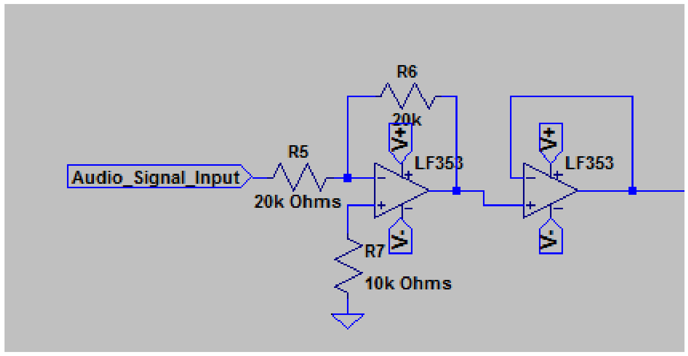

Figure 3: The buffer circuit.

Figure 3 shows the topology of the buffer circuit. This buffer circuit has two parts. The audio signal first passes into an inverting op-amp circuit that has unity gain. Inverting the audio signal has no effect on the resulting output since audio signals sound the same regardless of signal polarity. The output of the inverting buffer is connected to a non-inverting buffer circuit which also has unity gain. This circuit allows us to pass the audio signal into the modulation circuit without putting an excess load on the audio source or the modulation circuit. There are two unity-gain buffers being used in the circuit. Normally, only one buffer is necessary to achieve modularity in design. From testing the implementation of the circuit, we found that when we used only one non-inverting buffer there was distortion of the audio signal at the input of the buffer due to some non-ideal effects of using the crystal oscillator. To make sure that the audio signal was correctly modulating the carrier without getting distorted at the input, we included an extra unity-gain inverting buffer stage.

3.2 The Modulation Stage

To send the audio across the channel, this implementation utilizes frequency modulation. Frequency modulation is utilizing a faster signal, or carrier signal to send information from a slower signal by changing the frequency of the carrier signal. Typically, as the information signal increases in amplitude or voltage value, the carrier signal's frequency increases, and its amplitude stays constant. This implementation sends the audio signal across the channel using frequency modulation because these kinds of signals are more resistant to noise and distortion from the channel.

One way to achieve frequency modulation is to pass the information signal into the input of a voltage-controlled oscillator. A voltage-controlled oscillator is an oscillator, or a periodic signal generator circuit, whose signal frequency changes with a different input voltage. Passing the audio signal into the input of this circuit results in a frequency modulated sinusoidal signal at the output whose base frequency is dependent on the frequency of resonance of the circuit. At the core of the circuit is the crystal oscillator, which is depicted in the schematic as a box with two other parallel lines, one at each connection terminal. This circuit component is used to dictate the frequency of resonance of the circuit. A crystal oscillator consists of a specially-crafted piezoelectric, or a material that resonates at a

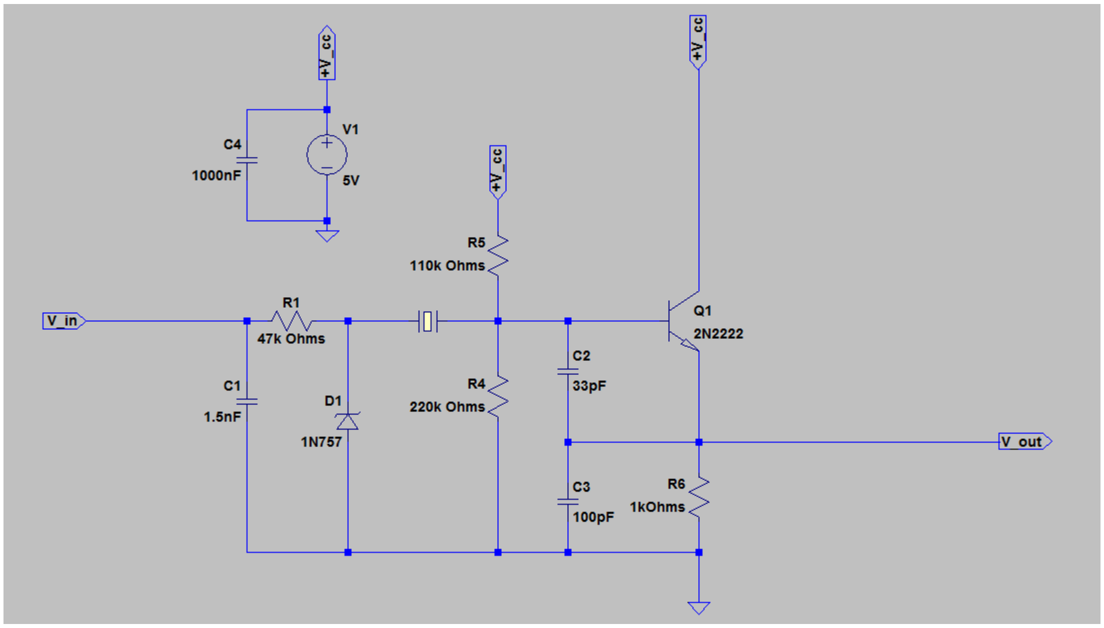

Figure 4: The modulation circuit.

precisely-defined frequency in the presence of an electric field. Crystal oscillators have a high Q-factor (on the order of about 10,000), so they have a low rate of energy loss. This means that they can generate signals with only one frequency for a long time, and they are more reliable for single-frequency generation than a standard tuned-RLC oscillator circuit. Crystal oscillators are characterized by their resonating frequency f, their series resistance R_{ser} and their parallel capacitance C_{para}. The crystal used in this circuit has the following parameters:[1]

$$f = 16MHz$$
$$R_{ser} = 40Ohms$$
$$C_{para} = 5pF.$$

A crystal oscillator alone is not going to generate a sustained carrier frequency. To generate the carrier, this implementation connects the crystal oscillator to the base of a common-collector amplifier circuit using a 2N2222 npn BJT. This configuration is called the Colpitts oscillator configuration, and is normally used with inductors, but this implementation replaces the inductor with a crystal. R_4 and R_5 act as a voltage divider and set a bias voltage of 3.3 V at the base of the BJT. This common-collector configuration has a capacitor connecting the base and emitter of the BJT together. This is a positive feedback configuration for the BJT, which means that the oscillations generated by the crystal oscillator grow in magnitude up to a value supported by the components. The amount of feedback and signal growth that occurs is dependent on the values of C_2 and C_3. Typical nominal values of the capacitors are between 20 pF and 100 pF for C_2 and between 100 and 150 pF for C_3. The amplitude of the sinusoidal signal generated by this circuit is about 1 V peak-to-peak, and the frequency is a stable 16 MHz. Both of these values were verified through simulation and construction. To generate a frequency-modulated signal from this circuit, one would typically use a varactor, or a diode specifically engineered to have a large junction capacitance. Typical diodes have junction capacitances that are on the order of 1-10 pF or less, while varactors have a junction capacitance of hundreds of picofarads. As the reverse-bias current across the varactor increases, so does its junction capacitance. The crystal 'sees' a change in impedance, or load, due to the changing capacitance of the varactor; this causes the frequency of the crystal to change from its nominal value slightly. At high frequencies, most diodes behave like varactors in reverse-bias. This implementation uses a 1N757 zener diode because of its high reverse breakdown voltage of -9.1 V and its high nominal junction capacitance of 78 pF. [2]

3.3 The Filtering Stage

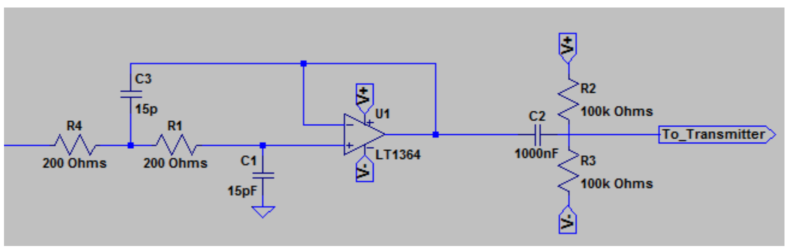

Figure 5: The modulation circuit.

To reduce the impact of harmonic distortion and high-frequency noise on the frequency-modulated signal, a Sallen-Key Low-Pass filter was placed at the output of the modulation stage. The Sallen-Key filter is a second-order filter, so it provides an attenuation of -40 dB per decade. Ideally, the cut-off frequency is set to be

$$f_{-3dB} = \frac{1}{(2\pi * \sqrt{R_1 * R_4 * C_1 * C_3})} \approx 53MHz$$

. However, the effects of the protoboard and the LT1364 op-amp amplifier cause this cutoff frequency to decrease. To get an approximation of the true frequency behavior of the circuit, simulations were carried out in LTSPICE.

This Sallen-Key filter also acts as a buffer stage to isolate the modulation part of the circuit from other parts of the system. To remove any DC bias on the output signal, this circuit uses a decoupling capacitor C_2 which is connected at one end to a voltage divider R_2 and R_3 that sets the bias voltage to zero volts.

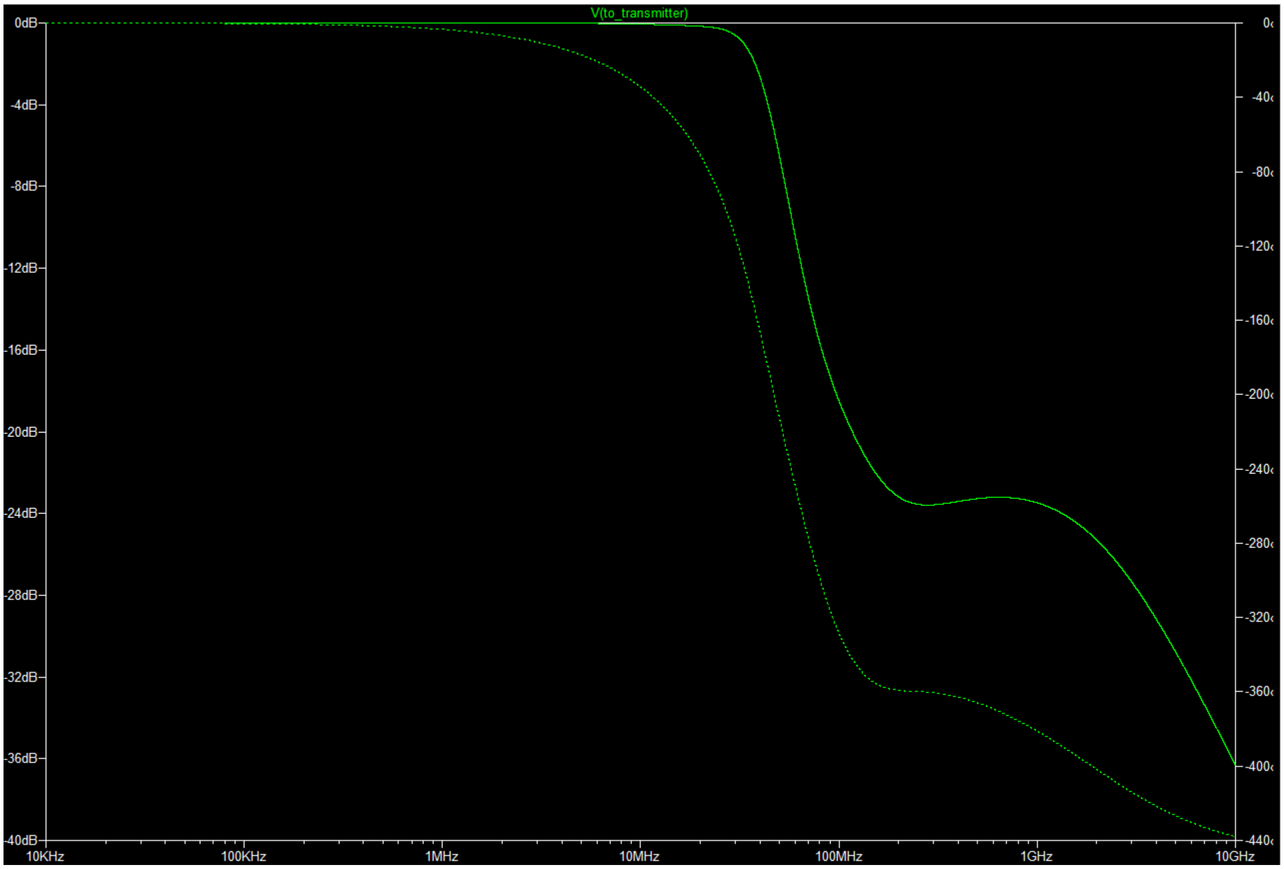

Figure 6: Simulated frequency response of the Sallen-Key Filter.

3.4 Discussion

In the process of creating this circuit, a few designs were considered. The first design considered was a traditional Colpitts oscillator circuit, shown in Figure 7. This design utilized an inductor and capacitor to generate the resonant frequency, which would be amplified by a common-base amplifier.

There were a couple of issues with this design. The first issue is that the inductor would need an inductance of 150 nH, which is tricky to make because it has to be small and compact. A 1 microhenry inductor can be made by putting about 20 turns of wire around a pencil, but smaller inductors are more difficult to make. These inductors are usually etched on a printed circuit board or premade by companies.

The main issue, however, is the Q-factor of this circuit. The Q-factor of the RLC tank circuit can be a couple of orders of magnitude smaller than that of a crystal oscillator if one is not careful, but even the best ones will still be an order of magnitude below that of a crystal at best. This means that the strength of the carrier signal that is produced from this circuit will decay much faster in time than the carrier signal in the crystal oscillator case. As a result of its higher Q-factor, the crystal oscillator is also much more resistant to frequency drift than the traditional Colpitts oscillator.

One trade-off that was made in this design was in the frequency modulation index. The

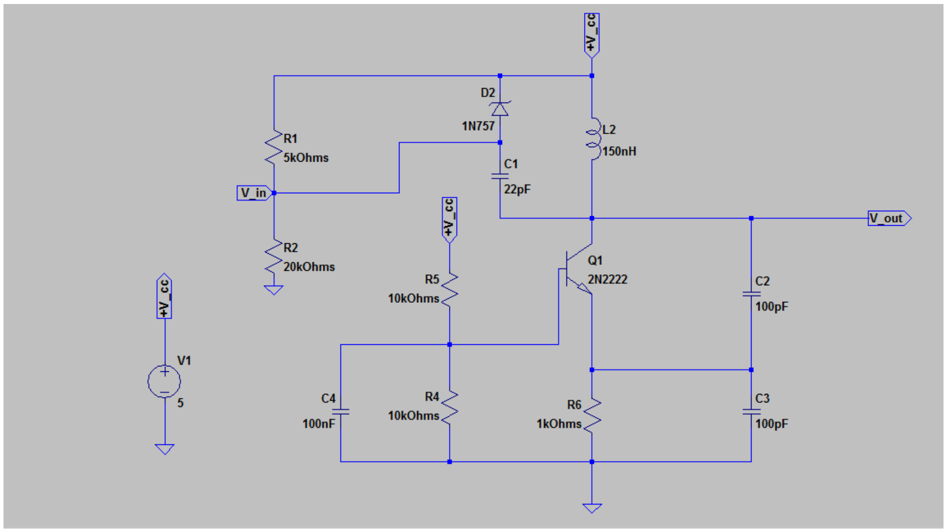

Figure 7: The Colpitts Oscillator.

frequency modulation index, h, is defined as the maximum amount the carrier frequency changes from its original frequency, Δf, divided by the largest frequency component found in the modulating signal, f_m. A larger modulation index results in a higher-quality demodulated signal. [3]

In simulation, the measured frequency deviation of the signal was found to be 7.5 kHz for a modulating signal running at 15 kHz. The modulation index of this implementation was found to be about 0.5. The crystal oscillator has a high Q-factor, which means that the carrier signal frequency can't deviate very much from the resonant frequency. As a result, the frequency modulation index suffers; the Colpitts oscillator is much better in this respect since its Q-factor is lower. The implementation of the crystal oscillator design was built on a through-hole protoboard. All parts were soldered onto the protoboard. This was done for better overall performance and stability. An implementation of the circuit was also built on a solderless breadboard. However, the output was attenuated and slightly distorted due to the high parasitic capacitances and inductances of the breadboard itself, so a soldered build was preferred. Figures 8 and 9 show captured oscilloscope measurements of the output signal in the soldered and solderless breadboards, respectively.

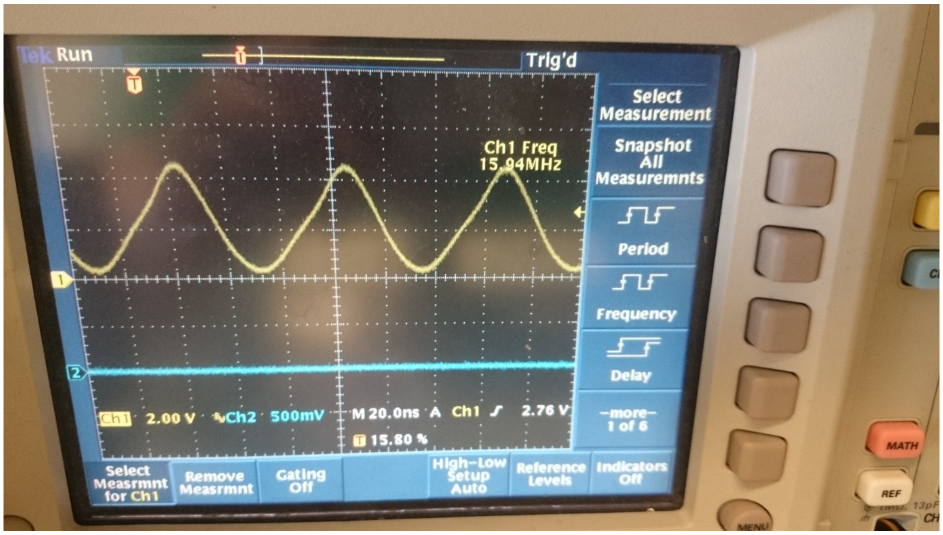

Figure 8: Measured FM signal, soldered breadboard.

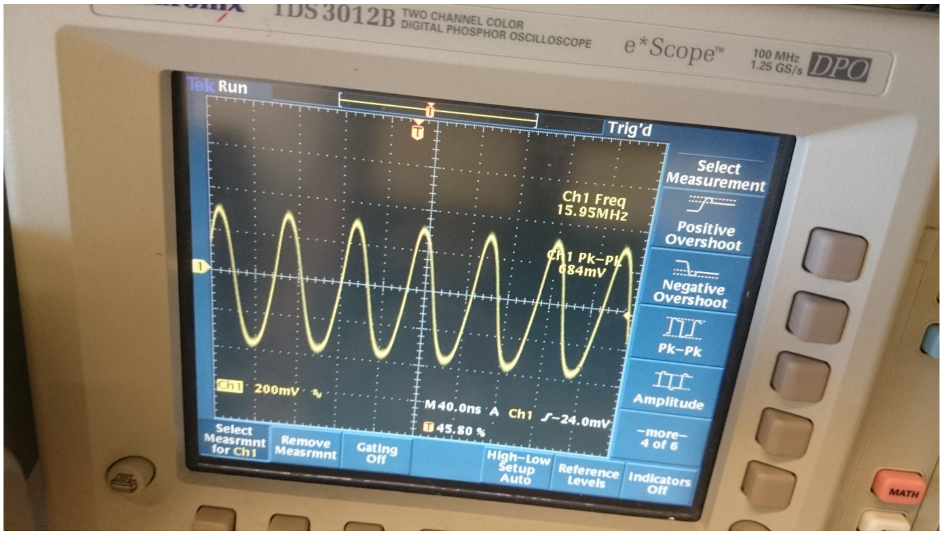

Figure 9: Measured FM signal, solderless breadboard.

4 Transmitting and Receiving by Hugo Malpica

After modulating the audio signal coming from the camera, both the modulated audio signals and the NTSC video signals needed to be combined, sent through the optic fiber and then recovered. Figure 10 shows the total system that transmitted and received these signals to then send onto the demodulation portion of the system.

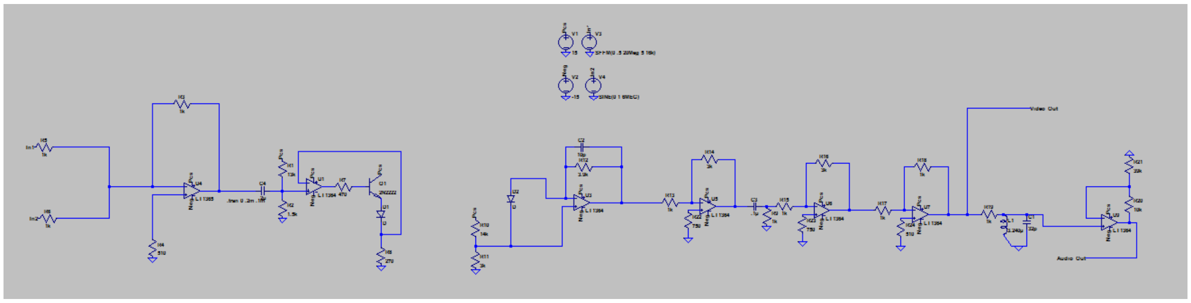

Figure 10: Full transmission and receiving circuit.

4.1 Summer Circuit

After the modulation of the audio signal, both the audio signal and the video signal (Figure 11 and Figure 12 respectively) needed to be combined so they could be transmitted through the optical fiber.

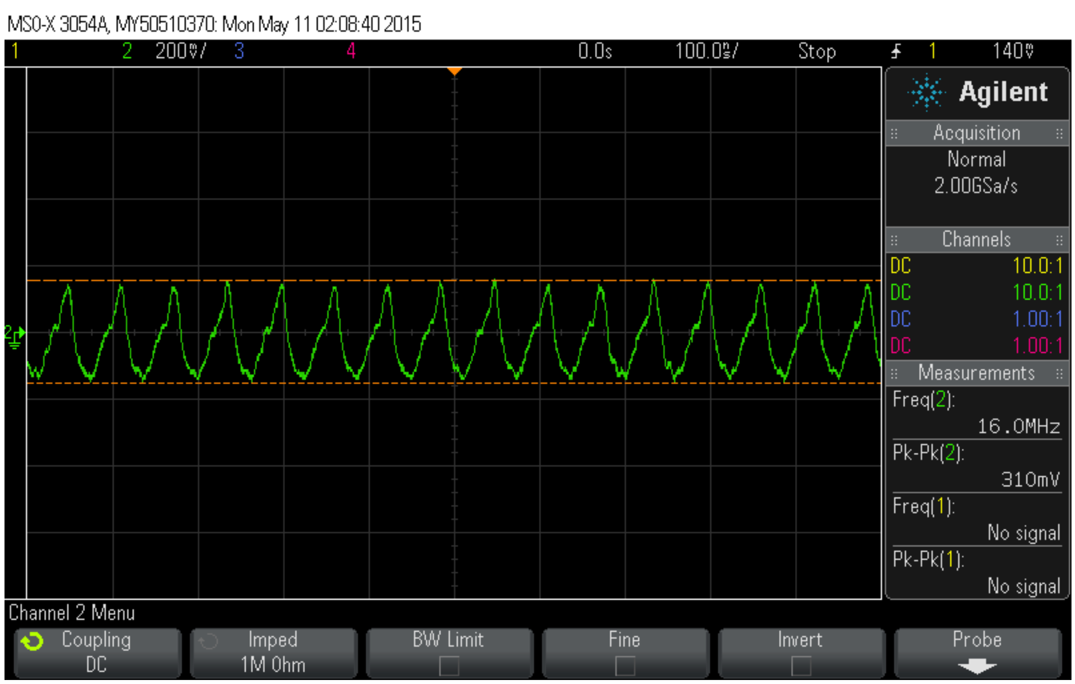

Figure 11: Modulated audio signal.

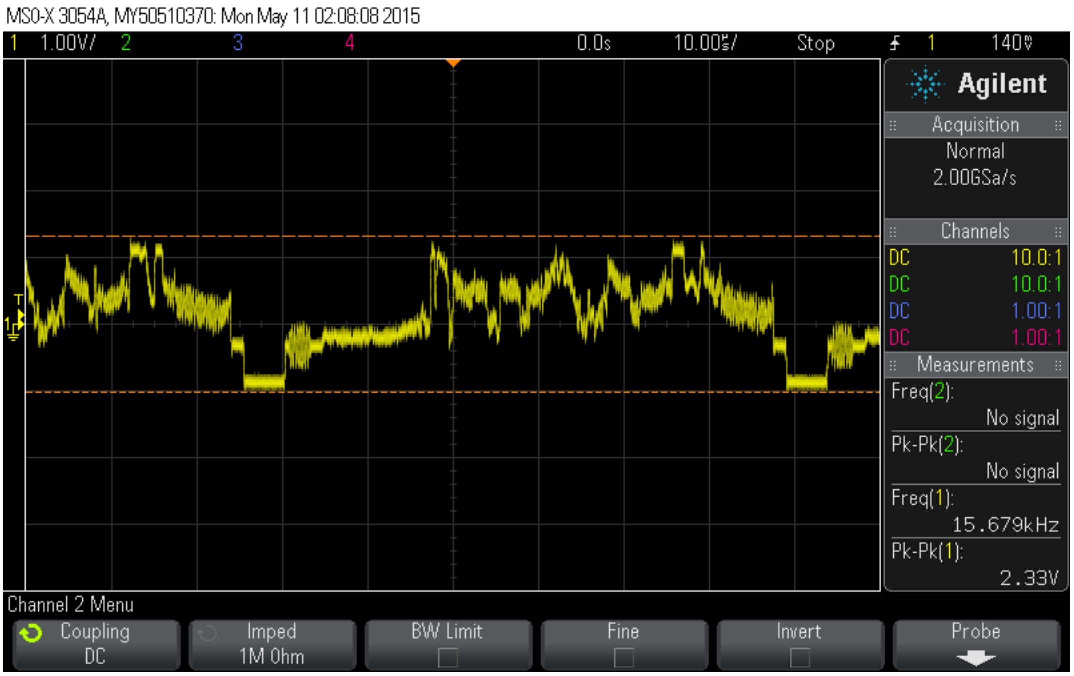

Figure 12: NTSC video signal.

To accomplish the task of adding the circuits, an op-amp summer circuit was used. Figure 13 shows the implementation of the summer circuit.

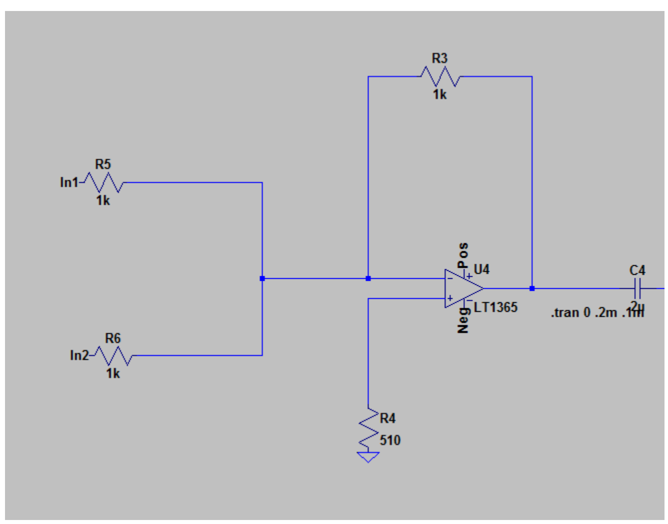

Figure 13: Summer Circuit

A summer circuit can be used in two configurations, with gain and without gain. To make sure our signals were transmitted properly, resistors values were chosen to be alike. The op-amp used was a LT1632. This op-amp was chosen due to its high bandwidth gain product of 45MHz, which was necessary to make sure the signals were not distorted due to low bandwidth. Figure 14 shows the output of the summer circuit.

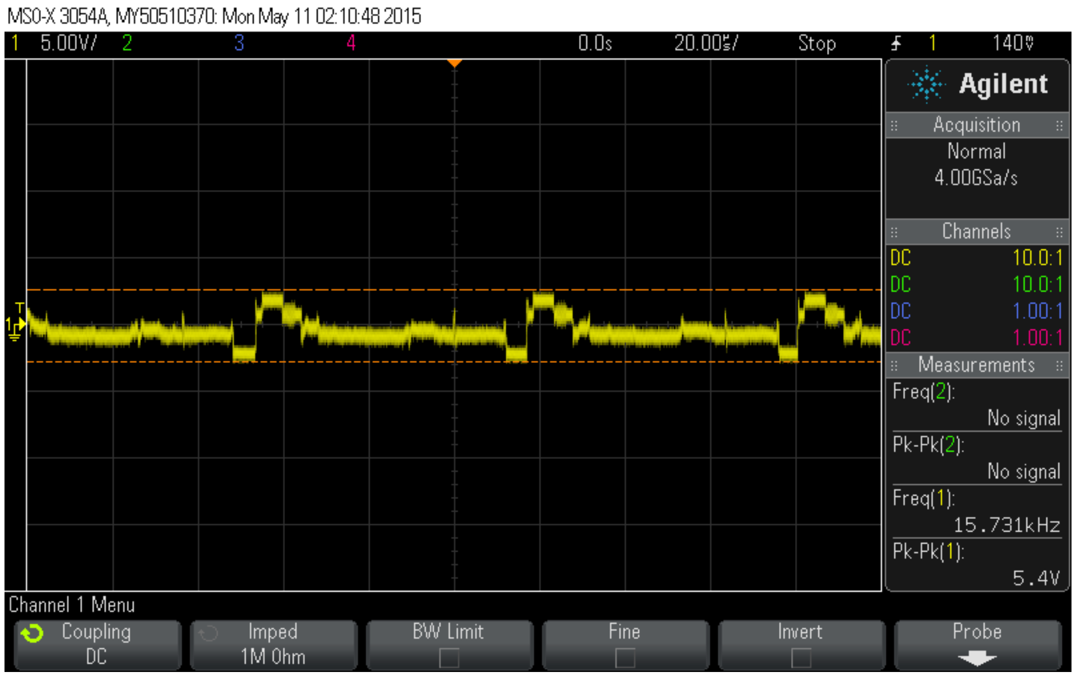

Figure 14: Output of the summer circuit.

4.2 Transmitter Driver

Once the signals were properly added together, the next step was driving the SFH757V transmitter, which was modeled by an LED. To ensure that our signal was being properly transmitted, the LED had to be driven with 20mA to ensure that it would be in the linear region as shown in Figure 15.

In order to properly drive the LED, a current source was needed. To make this current source, a 2N2222 BJT was used in a emitter follower configuration. At the emitter, the anode side of the transmitter was positioned and cathode was positioned to feedback to the negative input of the LT1632 op-amp. At the non-inverting side of the op-amp, the output of the summer circuit was offset by 6.6V, to ensure an operational region. The output of the op-amp fed into the base of the BJT while the collector was connected to 15V. At the cathode of the LED, a resistor of 270Ω was placed so that the current would always be around 20mA as shown in Figure 16. With this configuration, the signal in Figure 17 was transmitted via the optic fiber.

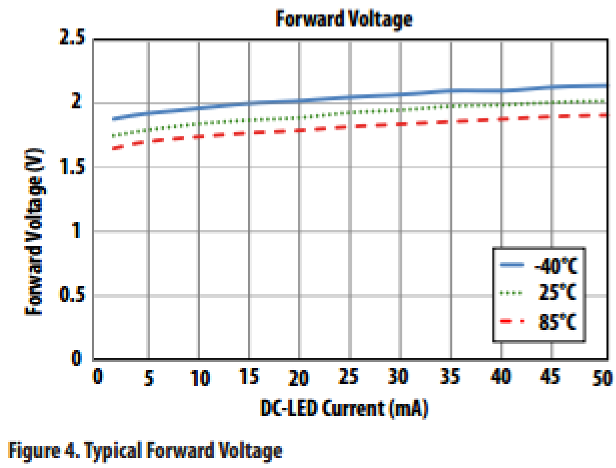

Figure 15: Typical Forward Voltage taken from the SFH757V datasheet. Belongs to Avago Technologies[8]

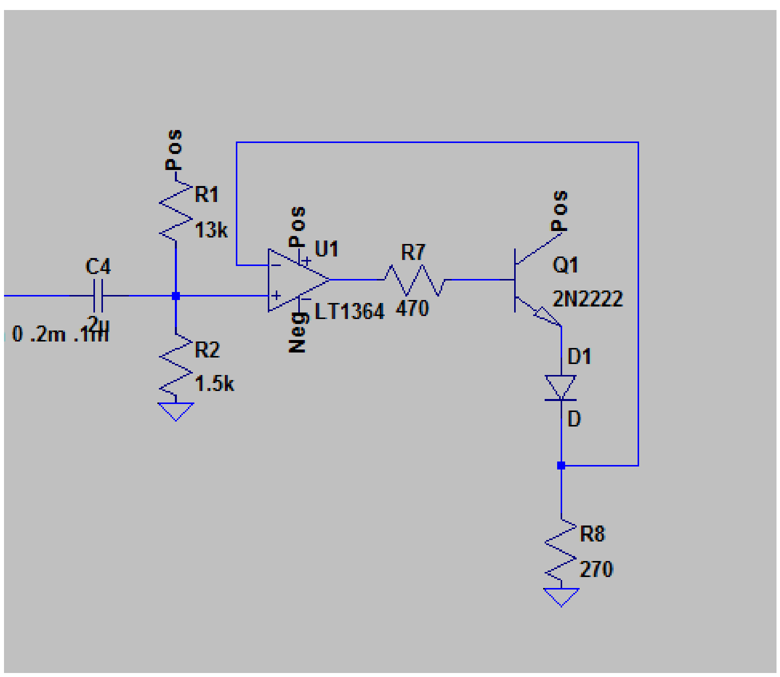

Figure 16: The current source circuit used to drive the transmitter in the linear region.

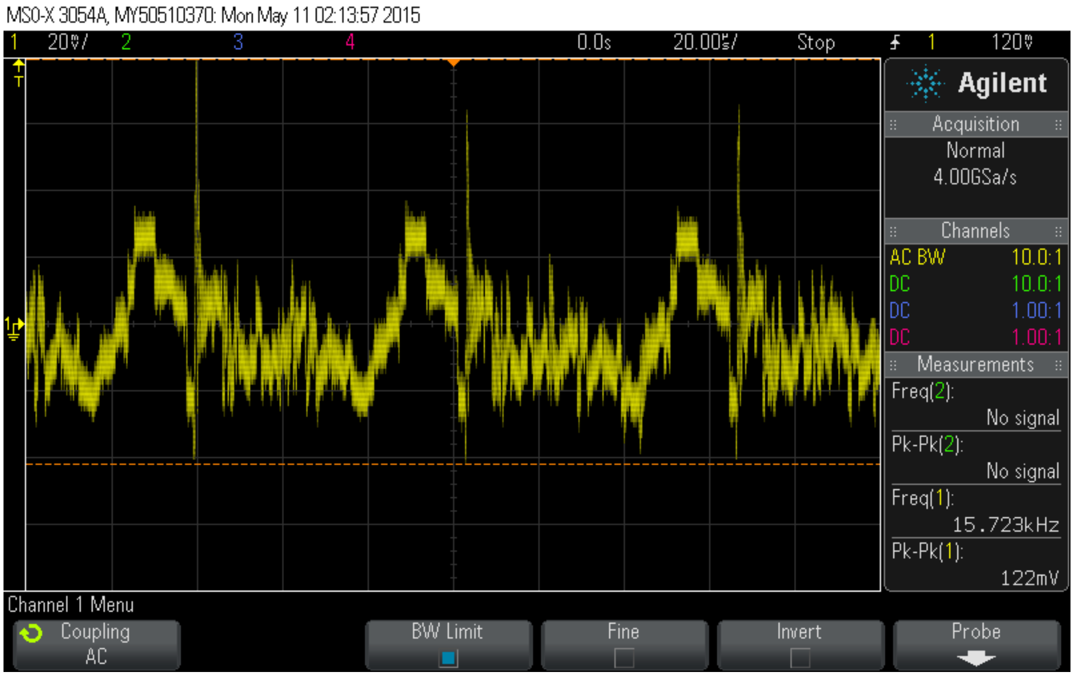

Figure 17: Signal being input into the anode of the transmitter.

4.3 Transimpedance Amplifier

To recover our signal from the SFH250V transmitter, which is modeled as a photodiode, a transimpedance amplifier was used. The light that travels through the optic fiber hits the photodiode which produces a current. The transimpedance amplifier is a current amplifier, that produces a voltage in response to a current input.[6]

In order to get the best possible signal back, our photodiode had to configured in reverse bias. This means that the cathode side has a higher voltage than the anode side. This was accomplished by biasing the voltage at the cathode by 2.6V which led in the non-inverting side of the LT1364 op-amp, which has a gain bandwidth product of 70MHz. The anode simply connected to the inverting input of the op amp. From the inverting input to the output of the op-amp a 3.9kΩ resistor was chosen experimentally. Initially a 10kΩ resistor, a typical value for a transimpedance ampifier was usitlized, but during testing, it kept producing a distorted signal and the picture would not sync meaning that the video kept rolling. Increasing the resistor just made the distortion worse so the resistance was dropped. The 3.9kΩ resistor produced the best picture quality and the slowest rolling.

In addition the the resistor, a 10pF capacitor was added for stability. The instability is caused by a lagging phase shift in the feedback path and it becomes more unstable when combined with the lagging phase shift of the op amp. The capacitor allows for a path for the signal to travel quickly which helps deal with the lag but this reduces the usable bandwidth which is why an op-amp of high gain bandwidth product is necessary. The schematic of the transimpedance amplifier can be found in Figure 9 and the input to the transimpedance amplifier can be found in Figure 19.

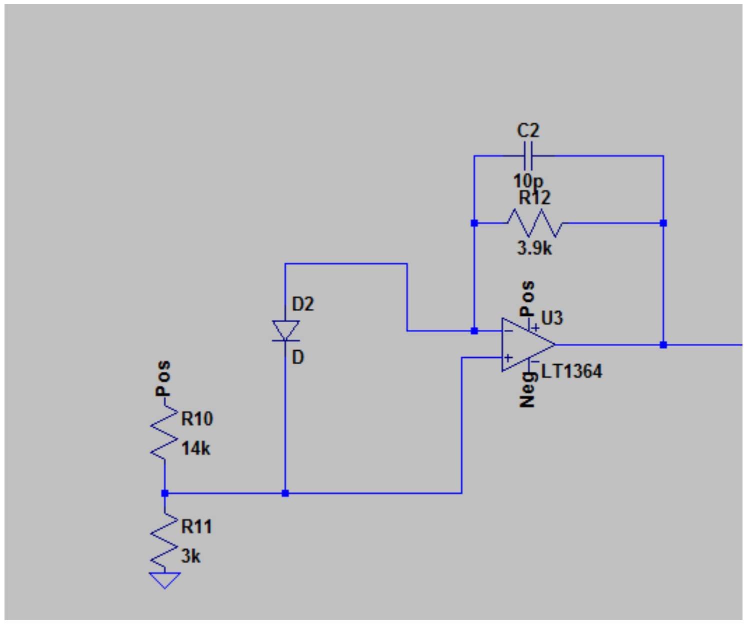

Figure 18: Transimpedance amplifer circuit with the receiver in reverse bias

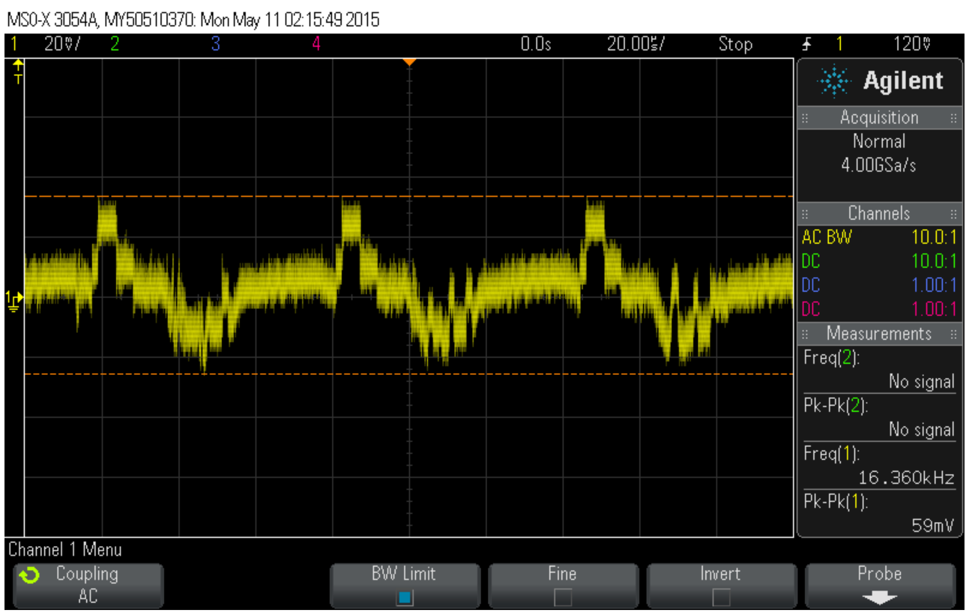

Figure 19: Input of the transimpedance amplifier

4.3.1 Distortion

Another cause for the distortion, is caused by unmatched poles. In Figure 20, the output of the transimpedance amplifier, some spikes are visible that are not visible in Figure 12 and Figure 19. The product of these unmatched poles first become visible in Figure 14 and with the amplification they go through along with more unmatched poles, steep spikes can be clearly seen in Figure 20 and the sync pulse which produces a still video image on the

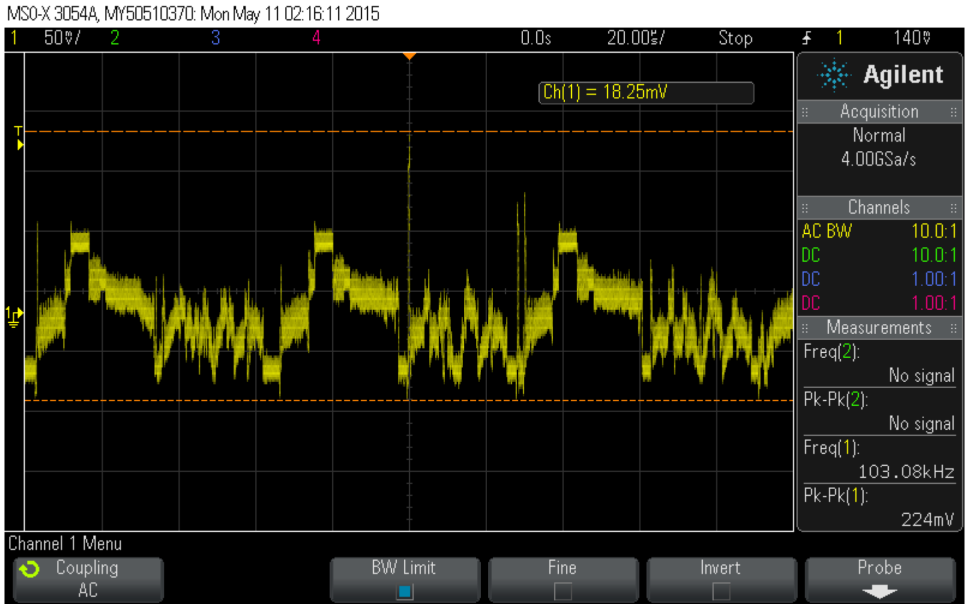

Figure 20: Output of the transimpedance amplifier

television set, is clearly slanted and noisy.

If a frequency sweep is run from a function generator, what one sees is that there is a bulge instead of a constant signal that is present from 3MHz to 4MHz which is the within the range of frequencies that the NTSC video signal is composed of. The result is the occurrences of spikes and other noise that is now added to the pure NTSC video signal.

4.4 Amplification

Once we recovered the combined signal, amplification of the signal was needed in order to ensure that the signal had enough voltage to be read by the television and so that it could easily pass though the demodulation circuit. The amplification stages were simply inverting op-amps. This meant that we had a gain of the value of the division of the second resistor over the first resistor. Using 3kΩ resistors with 1kΩ, the gain was simply of 3. More gain stages were needed in order to combat the gain bandwidth product. More gain meas less bandwidth so the op-amps used for these amplification stages are LT1364. Figure 21 shows the schematic of the amplification stage, while Figures 22, 23 and 24 show the outputs of the separate amplifiers, the first two being of gain 3, while the last is just a unity-gain amplifier that inverts our signal so that it can be in the correct position and allowed to be read by the television. If the signal if inverted when input into the television, you will obtain a heavily distorted grey and black image.

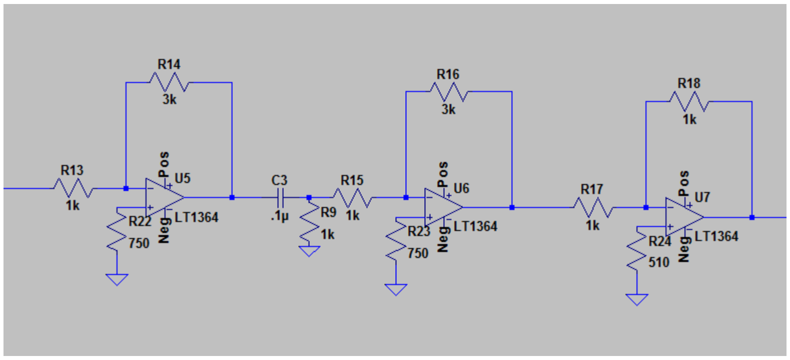

Figure 21: The amplification stage of the circuit.

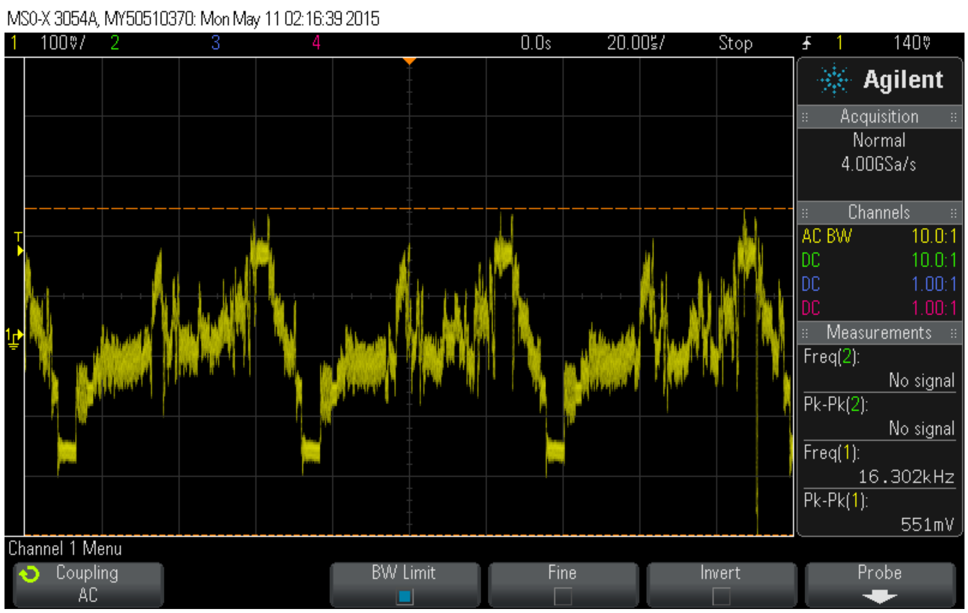

Figure 22: Output of the first amplifier.

Although this signal contains the audio signal, this combined signal shown in Figure 24, can be directly input into the television monitor even though it is combined with the 16MHz modulated signal. This is because NTSC is typically in the range of 8.4MHz. Any higher frequency, would not be picked up well by the television. The difference of picture quality is denoted in the Results section.

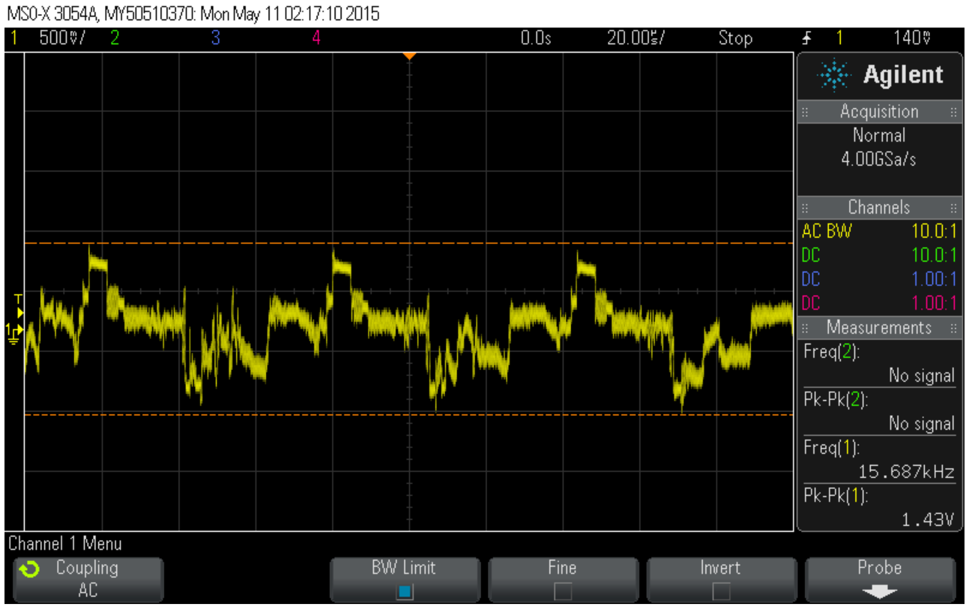

Figure 23: Output of the second amplifier.

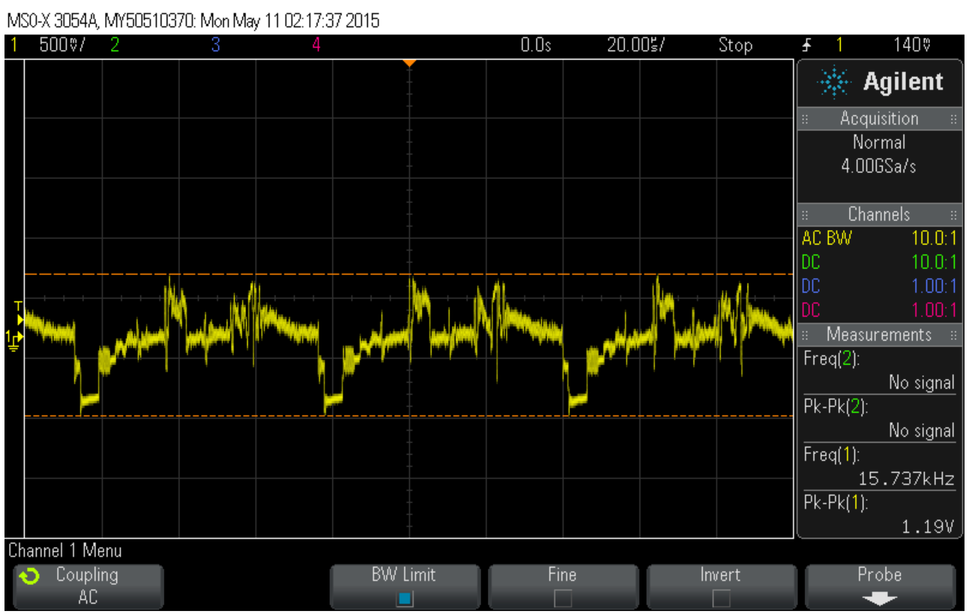

Figure 24: Output of the unity gain amplifier

4.5 Filtering

The same signal that is directly output into the television set, is also sent to a RLC filter that has a high Q and is set to be around 16MHz so that the NTSC video signal would be filtered out and the modulated audio signal would be left. The filtered signal is then put into a non-inverting amplifier with a gain of 1.256. This value was supposed to be of 4.9 when originally calculated but the resistors were switched by mistake but it produced the desired signal so it was kept. As in previous stages, the LT1364 was used due to its high gain bandwidth product. Figure 25 shows the schematic for the RLC circuit as well as for the final amplification stage. Figures 26, 27, 28 and 29 show the output signals.

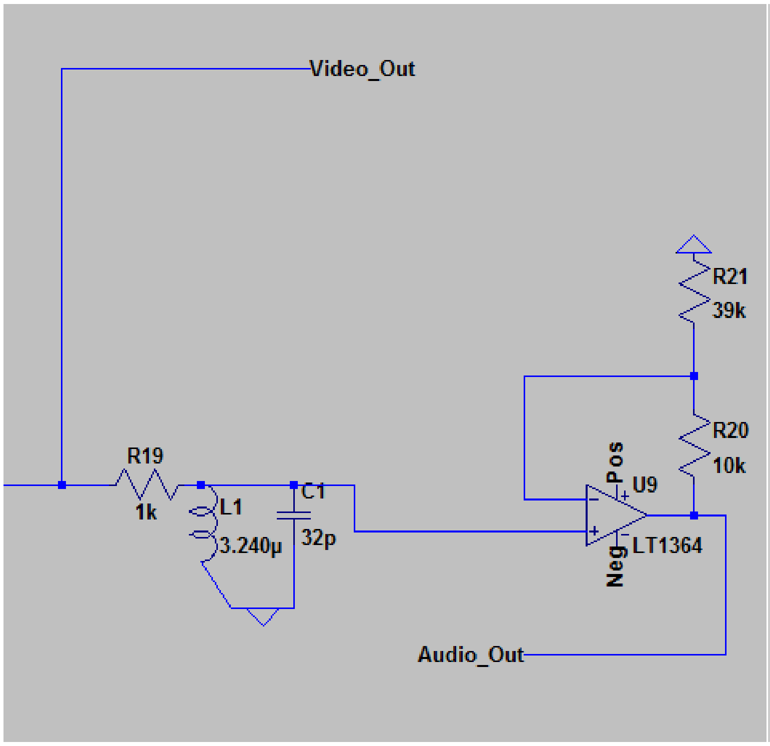

Figure 25: RLC filtering and final amplification stage schematic.

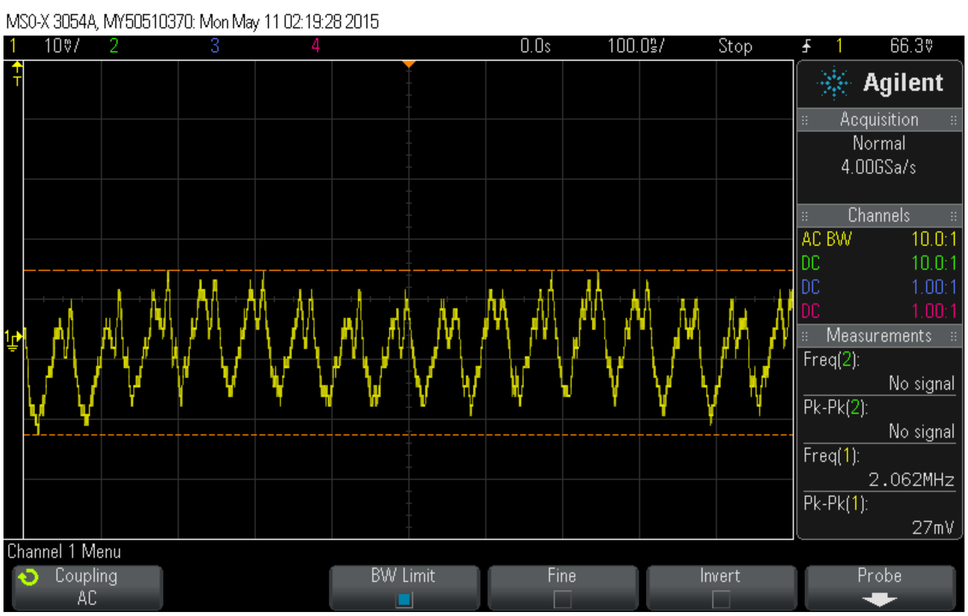

Figure 26: Output of the RLC filter.

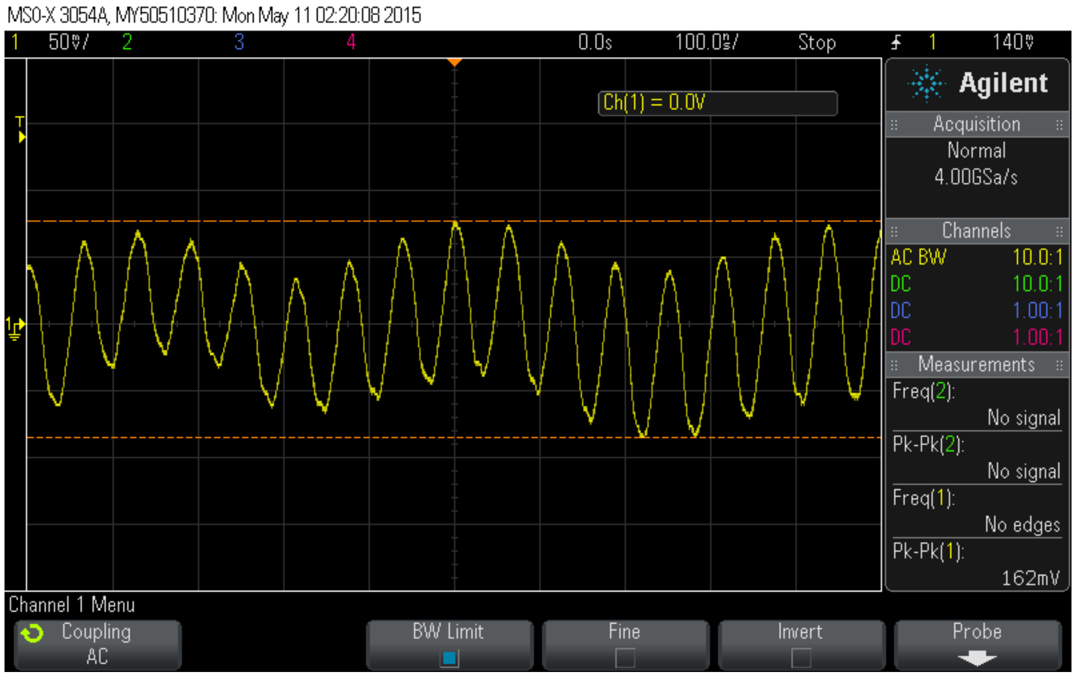

Figure 27: Output of the final amplifier. This signal will go into the demodulation circuit to be demodulated.

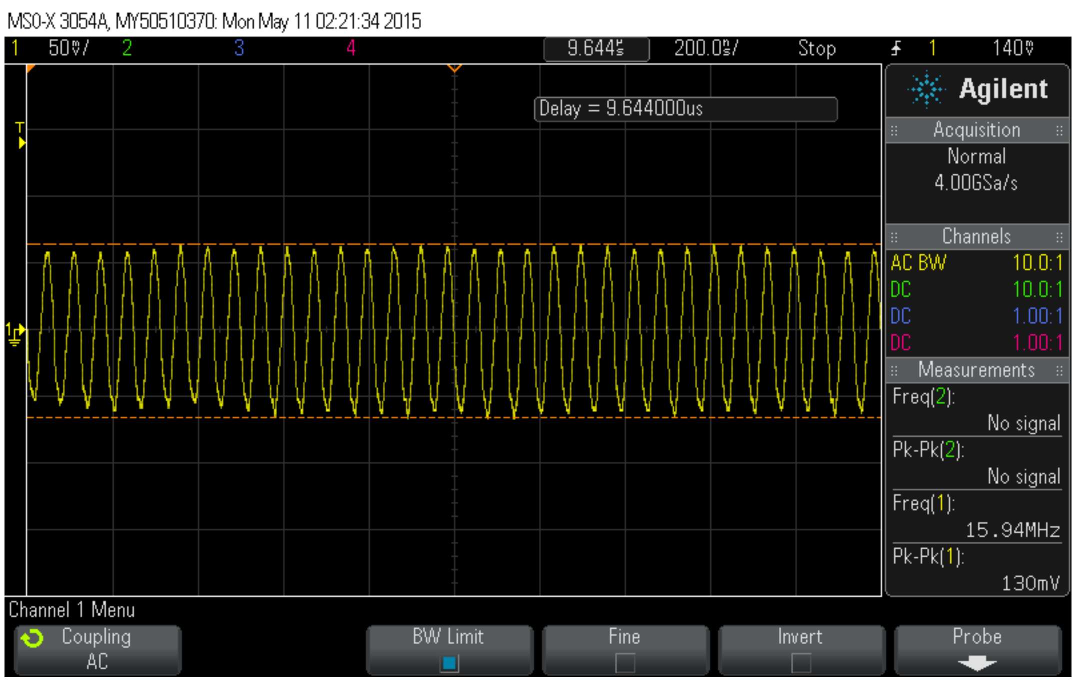

Figure 28: Pure 16MHz portion of modulated audio signal.

One thing that is noticeable in the signals is the fact that there is a slight oscillation in the 16MHz carrier in itself as seen in Figure 28 and Figure 29. The modulated audio signal, as shown in Figure 27, is then sent to the demodulation module of the system to be

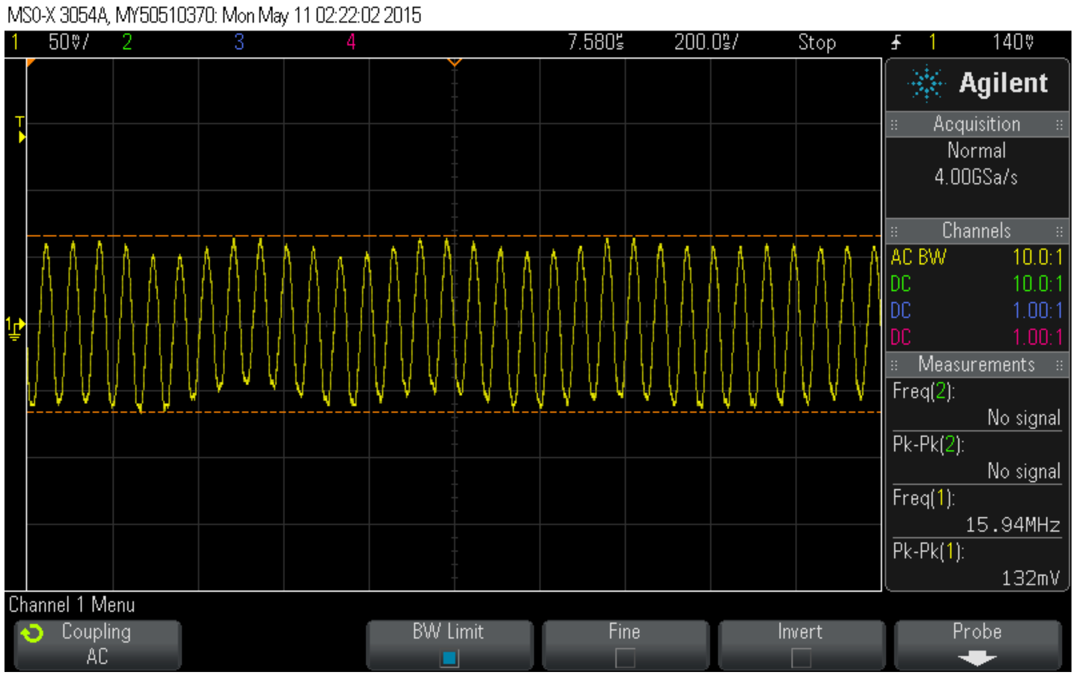

Figure 29: Oscillation portion of the modulated audio signal.

demodulated.

4.6 Discussion

Initially when coming up with what would be the Summer circuit, which uses frequency division multiplexing to separate the signals and then adds them together, the first thought that came to mind was to use any sort of scheme that added signals together. Using this mindset, a time division multiplexing adder was simulated. Although it did add the signals like it was supposed to there was a problem. Without digital circuitry, it was near impossible to separate the signals at some later point. Since this class focuses on analog electronics, that would have gone against the spirit of the class so the idea was scrapped and the summer was implemented.

For the transmission portion of the circuit, what was initially implemented was a scheme that would select a resistor with a range of voltages to bias the LED at a range of currents that were within the allowed linear region. Although this worked, it did attenuate the signal and added noise since sometimes the voltages ranged a bit outside of the allowed region and when passed through the resistor, would generate a current outside of the linear biasing zone. To combat this, a current source was implemented as shown in the Transmitter Driver section.

5 Signal Conditioning and FM Demodulation by Khaled Moharam

This section of the project is concerned with conditioning and demodulation of the FM signal to extract an audio output with high fidelity with respect to the input audio. The signal conditioning uses amplitude limiting techniques and filtering to eliminate distortion. The output of the signal conditioning block is fed to FM demodulation block. FM demodulation uses slope detection with noise filtering and amplification.

5.1 Signal Conditioning

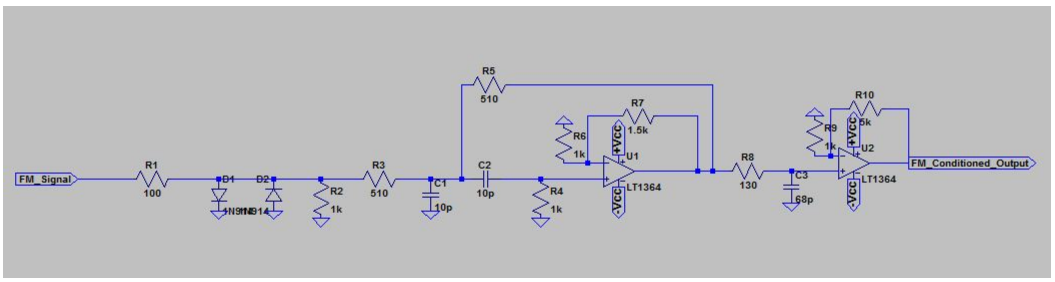

Figure 30: Overall Signal Conditioning System

FM audio signals transmitted from noise susceptible sources through noise susceptible channels suffer AM distortion that is amplitude variation of the FM signal, which in the FM demodulation stage causes distortion of the output audio signal. However, it first appeared to me that AM noise would be minimal due to the small dimensions of designed circuits and the transmission medium. However, for the sake of completion and suppressing any possible source of noise-which will be evident in the actual system implementation- I decided to design a signal conditioning circuit.

Signal conditioning circuit is divided into three parts:

1. Diode limiter: The diode limiter circuit clips the amplitude of the incoming FM signal at approximately 1.6v pk-pk. This eliminates AM distortion and outputs an FM square wave with constant amplitude.

2. Band pass filter: The band pass filter extracts a FM sinusoidal signal with an amplification of 2.5.

3. Low pass filter: The low pass filter further amplifies the FM sinusoidal signal and further suppresses any signal harmonics.

5.2 Diode Limiter

The diode limiter circuit is made of a voltage divider network with two back to back diodes in parallel with R2. The moment the voltage across R2 reaches the threshold voltage of the diode that is approximately 0.8 volts for D1 or -0.8 volts for D2, the output voltage

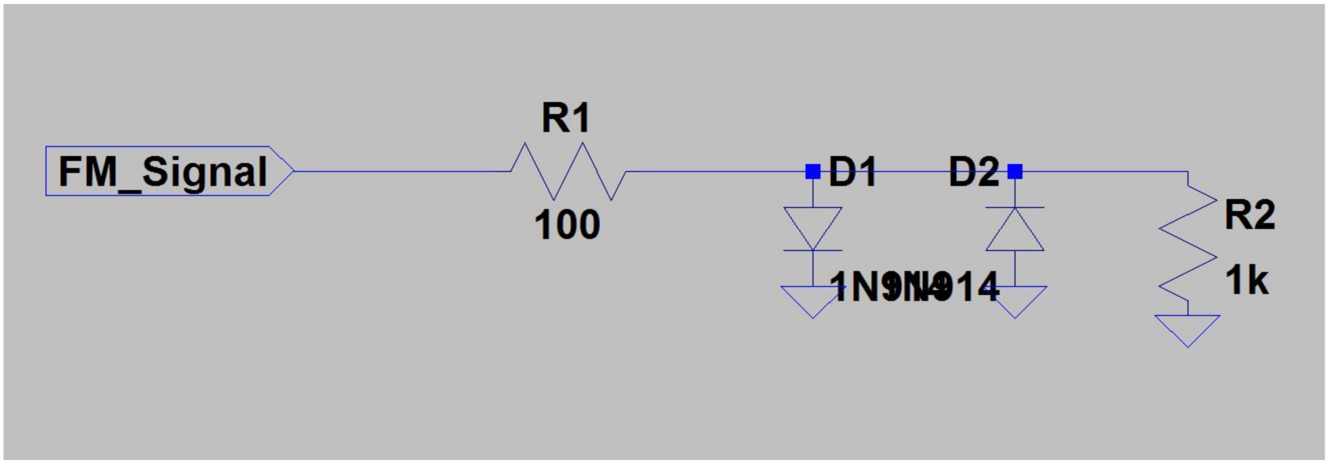

Figure 31: Diode Limiter

clips at 0.8 or -0.8 volts. This eliminates the AM distortion noise. The values in R1 and R2 determine the clipping onset voltage. There is a source of non-ideality that is, the clipped output voltage is not fixed at 0.8 voltage; however, output voltage slightly increases with respect input voltage due to diode properties. This has minimal effect on the output signal since the output voltage variation is minimal as well as having minimal AM noise.

The performance of the circuit was as expected. It outputs a square wave at large voltage input values. However, for input voltages that are less than 2 volts of peak amplitude, the non-idealities become more noticeable and the output waveform is not completely a square wave; however, the peak amplitude is limited at around 0.8 volts. This non-ideality is not a problem as the bandpass filter is able to extract the 16 MHz fundamental frequency.

5.3 Band-pass Filter

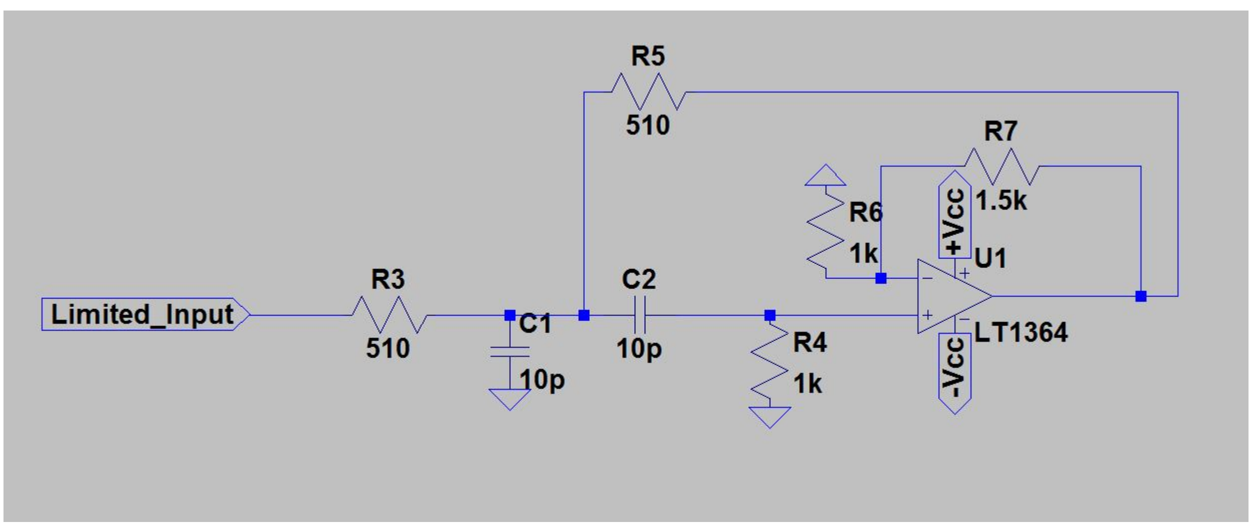

Figure 32: Bandpass filter

The bandpass filter is a second order Sallen-Key topology with a low gain of 2.5. The initial idea was to have a filter centered at 16 MHz, which is the frequency of the FM carrier, and have a gain of 10. The high gain was necessary to amplify the extracted 16 MHz sinusoidal to be at suitable amplitude for the next stage of demodulation. But given the limited bandwidth of 70MHz for the LT1364 op-Amps, the design was tricky. At a gain of 10, the center frequency would at 7 MHz. The design decision was then to split the gain process into two different stages, where the second stage will be discussed in the next section. The gain was reduced to a value of 2.5, however, the circuit simulations showed a center frequency that is far away from 16 MHz. The trick to fix this anomaly is to design the bandpass filter at a much higher center frequency than 16 MHz. I designed the filter above to be centered at 30 MHz. The simulations result at this design criterion was a bandpass filter centered at 15 MHz. Further tweaking showed that it's very difficult to push the center frequency to further approach the 16 MHz requirement. So a second order bandpass filter centered at 30 MHz was designed to give a filter of the required property with a center frequency near 16 MHz. There is a desirable side effect in having a center frequency that is not perfectly 16 MHz; this side effect will be discussed in the FM demodulation section. Figure 33 shows the output of the bandpass filter.

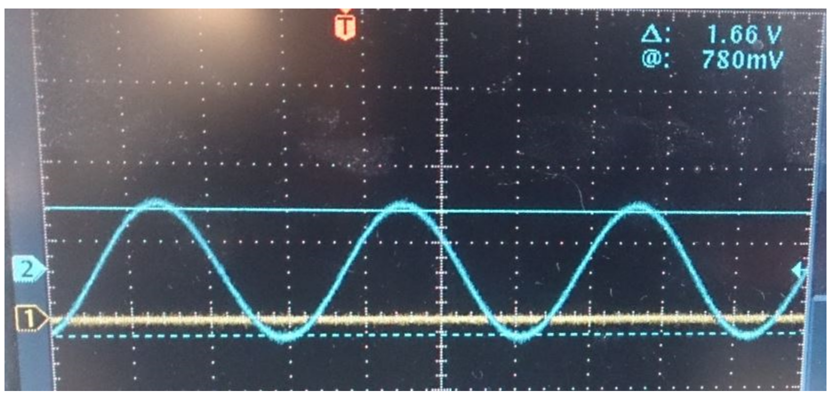

Figure 33: Bandpass filter output

5.4 Low-pass Filter

Given the low gain of the bandpass filter, another stage of amplification is required. I decided to use a simple low pass filter topology with a gain of 6 to have a simple amplification topology assisted with low pass filtering that eliminates most of the higher harmonics of the fundamental 16 MHz carrier frequency. Figure 35 shows the bode plot of the bandpass filter cascaded with low pass filter. The center frequency is 13.5 MHz with a high quality factor such that the gain at 16 MHz is 17 Db. The output of the low pass filter is amplitude limited FM sinusoidal that is ready for the demodulation stage.

5.5 FM Demodulation

There are several approaches to FM demodulation. In my initial planning, I focused on studying two methods:

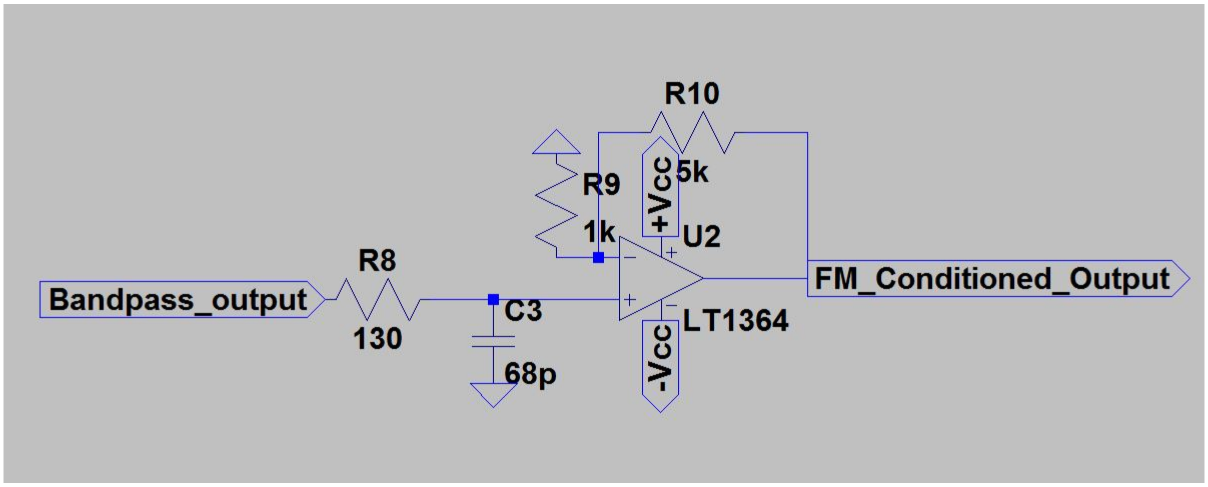

Figure 34: Low pass filter

Figure 35: Bode plot of cascaded bandpass and lowpass filters

1. Quadrature demodulation
2. Slope detection

The ultimate goal is to output a clean audio signal with maximum fidelity to the input signal.

5.5.1 Quadrature Demodulation

Quadrature demodulation is multiplying the input FM signal with 90 degrees phase shifted copy of itself. The output is fed into a low pass filter that outputs an audio de-

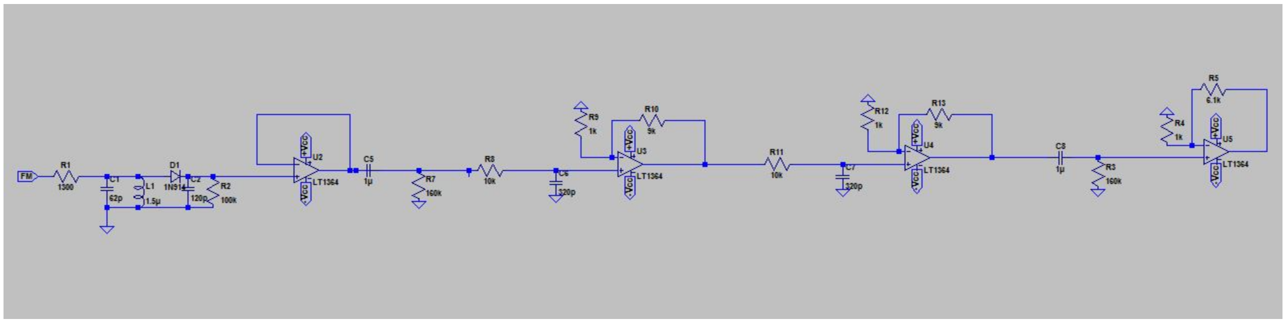

Figure 36: FM demodulation circuit

modulated signal. The implementation requires a phase shifting RLC circuit and an analog multiplier. The RLC is a simple design, but there are many different possibilities for the analog multiplier. I studied a design found in a paper written in 1968 that uses 3 differential pairs; however, due to the fact it's hard to have matched transistor pairs and a colleague of mine in class who was working on a similar project had trouble implementing a slope detector, so I decided to investigate how to implement a slope detector.

5.5.2 Slope Detection

Another simpler method but proved tricky to implement is slope detection. Slope detection transforms the FM signal into an amplitude varying FM signal such that amplitude variations are proportional to frequency variations of FM signal. The amplitude varying FM signal is fed through a peak envelope detector that outputs audio signal. The FM demodulation circuit used an RLC slope detector with a peak envelope detector. The output of the peak detector feeds is fed into a buffer to isolate it from the rest of the circuit. The buffer output passes through a high pass filter that eliminates DC bias. It's then amplified using two cascaded low pass filters each with a gain of 10. The low pass filter eliminates any high frequency distortion and the high gain amplifies the audio to audible levels.

5.6 Slope detector and envelop detector

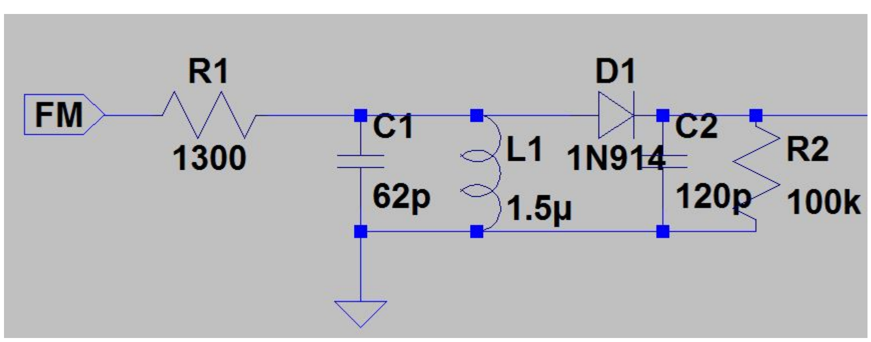

Figure 37: Slope detector and Peak detector

The idea of slope detection is to vary signal amplitude proportionally to input frequency.

The best way to achieve slope detection is to use a bandpass filter with sufficiently high Q-factor and a center frequency that is slightly below the FM carrier frequency. Given that the carrier frequency lies on the high slope line of the bandpass transfer function, the frequency variations of the FM signal cause amplitude variations that outputs a signal with an envelope that resembles the modulated audio signal. However, the design choice for the quality factor and center frequency is not arbitrary. It depends on the modulation index of the FM signal and equations that are derived from the linearization of the transfer function of the bandpass filter around the carrier frequency point. Choosing design parameters such that the frequency variations happen to occur on the most linear part of the transfer function is necessary for minimum distorted output. The following equations offer design guidelines for slope detector:

$$\delta\omega = maximum\ frequency\ deviation\ from\ carrier\ frequency$$

$$\omega_m = maximum\ modulating\ frequency$$

$$\omega_c = filter\ center\ frequency$$

$$\omega_o = carrier\ frequency$$

$$BW = \frac{100\Delta\omega}{3}$$

$$\omega_c = \frac{\omega_o + BW}{2\sqrt{2}}$$

$$C = \frac{1}{BW * R}$$

$$L = \frac{1}{{\omega_c}^2 * C}$$

The LC tank given above was designed for a carrier frequency of 16 MHz, filter frequency of 15.4 MHz, optimal performance for modulation index is 4. However, the slope detector worked perfectly for modulation index as low as 0.5. The diode peak detector is a simple topology similar to the one used in lab 1 with a time constant that is a fraction of the period time of 16 MHz carrier frequency. Figure 38 shows the output of the slope detector at a modulation index of 0.5. Figure 39 and 40 show the output envelope for modulation indices of 4 and 0.5, respectively.

From the figures, the output envelope at modulation index of 0.5 is a bit smaller than it's counterpart at modulation index of 4. It would be expected that a FM signal with small modulation index would not be detectable; however, as mentioned in the previous section on the bandpass filter, there is a side effect of having a bandpass filter that is not centered around 16 MHz. This permits the bandpass filter to act as slop detector. So in reality, we have two cascaded slope detectors.

Given that the output as shown in the figures is very small in amplitude and highly distorted, further amplification and filtering is implemented in the following stages.

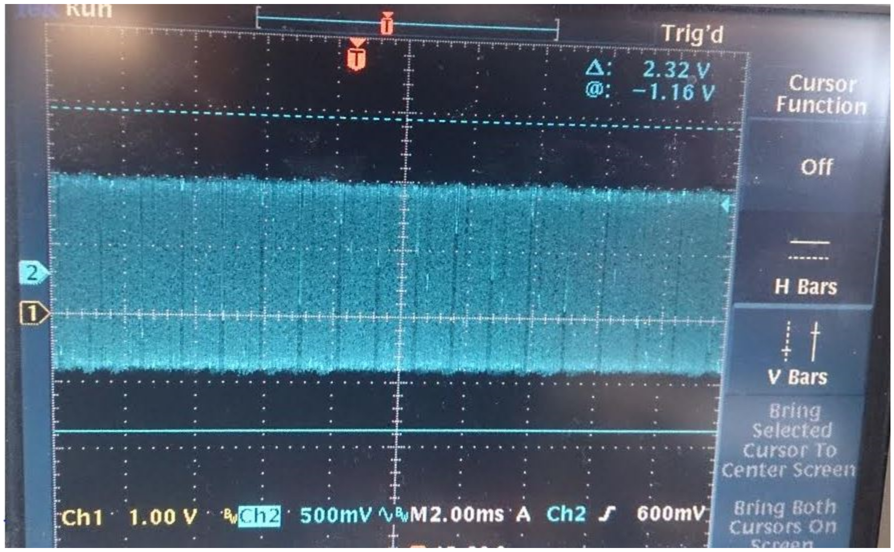

Figure 38: Output of Slope detector at modulation index of 0.5

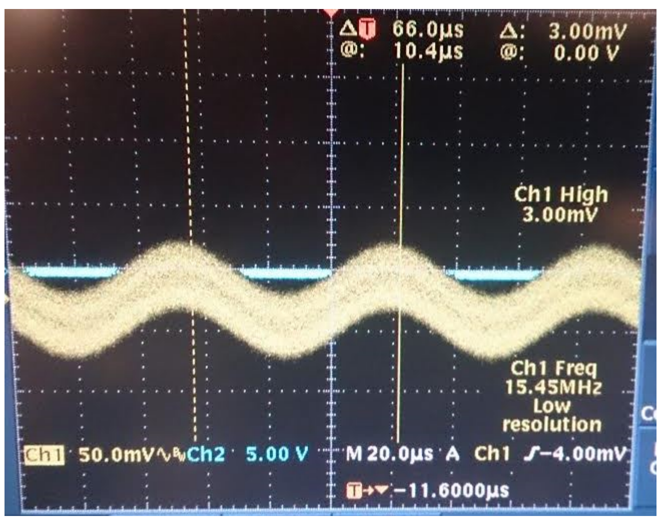

Figure 39: Output envelope at modulation index = 4

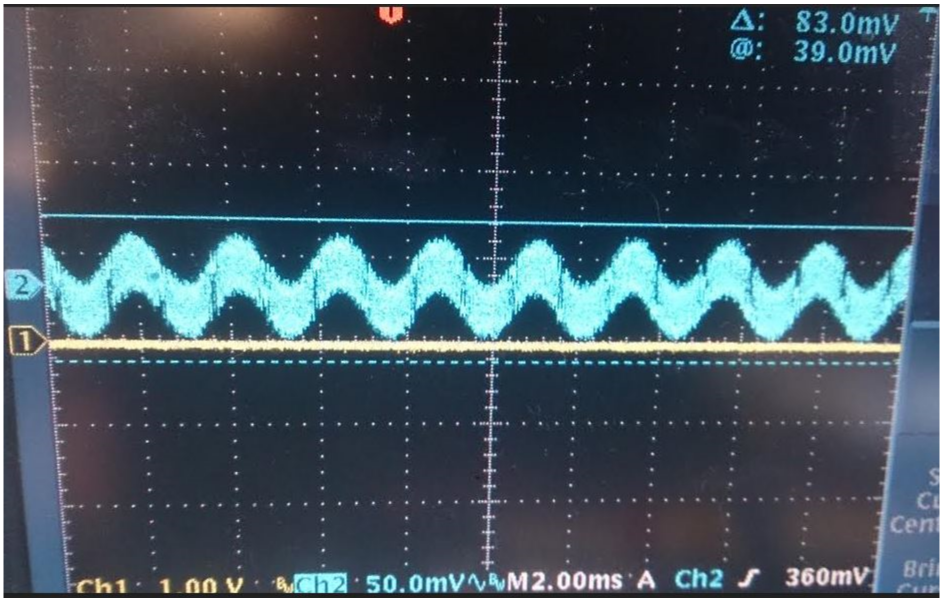

Figure 40: Output envelope at modulation index = 0.5

5.7 Unity Gain Buffer

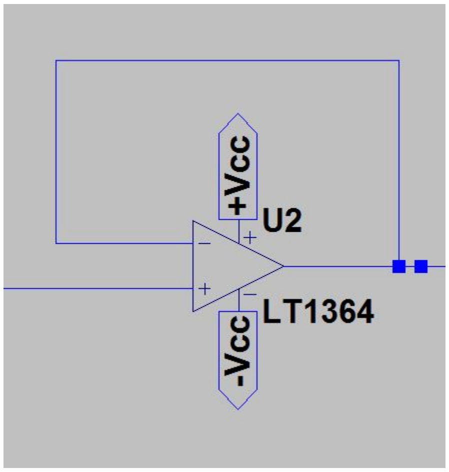

Figure 41: Unity Gain Buffer

The unity buffer isolates the RLC slope detector and the envelope detector form any loading due to the filtering and amplification stages that follow. A simple unity gain buffer was implemented.

5.8 Low-pass Filter

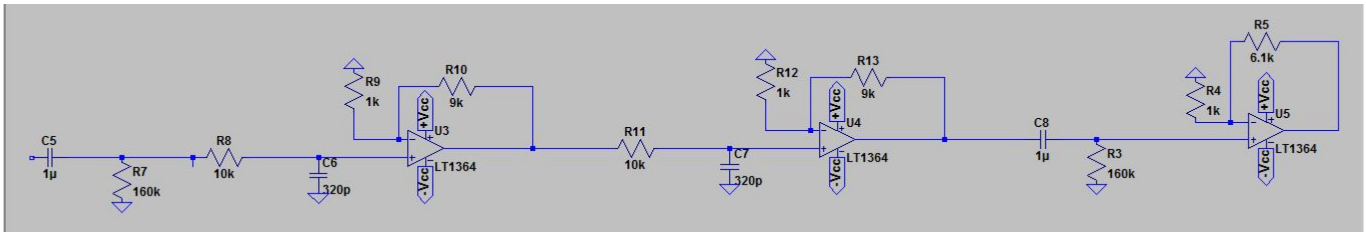

Figure 42: Lowpass filtering with amplification stage

This stage serves two purposes:

5.8.1 DC offset correction

The DC offset correction is implemented using a passive first order high pass filter at the output of the buffer with cutoff point of 10 Hz. Another passive first order high pass filter is found at the output of the second active lowpass filter before the final output amplification stage. The High pass filters eliminated most of the DC signal. There was only a DC bias of 50 mV in the output signal. It didn't affect the quality of the audio signal since the output peak amplitude could go up to 8 volts of peak amplitude.

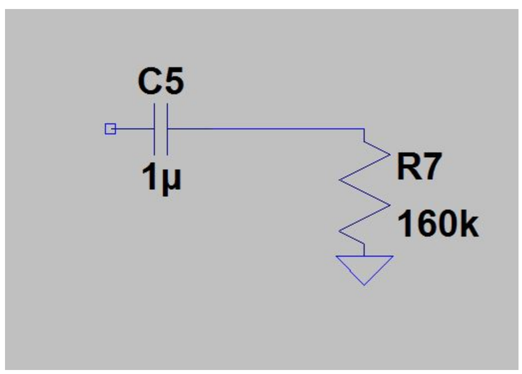

Figure 43: DC offset correction high pass filter

5.8.2 Lowpass filtering and amplification

Two cascaded 1st order low pass sallen-key filters are used for filtering out high frequency noise and harmonic distortion. Each filter is designed for a cutoff frequency of 50 KHz and gain of 10. A simpler topology of two cascaded first order filters each at a gain of 10 was preferred over the more sophisticated 2nd order low pass filter with a gain of 100. This is because at high gains second order sallen-key filters offer a design challenge for the criterion of stability. There are online tools that help in choosing component values for 2nd order sallen-key filters but I found using cascaded 1st order filters produce the same effect and at the same time being immune to instability issues.

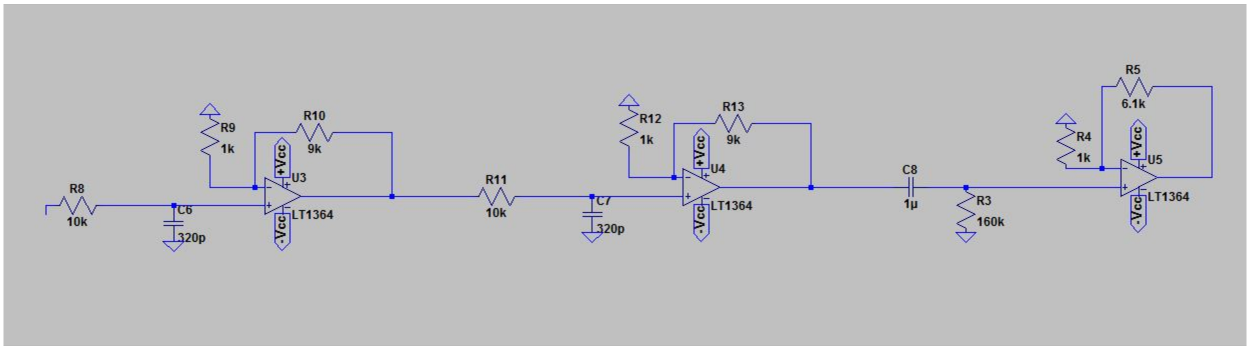

Figure 44: Lowpass filtering with amplification

The output of the cascaded 1st order filters goes through another passive first order high pass filter to remove any DC offset due to non-idealities of the filters.

A final op-Amp amplification stage is added to count for the very low modulation index of the generated FM signal by my teammate Germain. The low modulation index FM produces a very low amplitude envelope that a final gain stage of amplification is necessary to produce intelligible audio output levels. However, in the testing stage where the FM function of the function generator was used, the output levels were high enough to have a loud audio output coming from the speaker, although the modulation index was varied to go as low as 0.5. Figure 45 shows the audio output of the demodulation output when an FM signal with carrier frequency of 16 MHz and a modulating frequency of 1KHz is used.

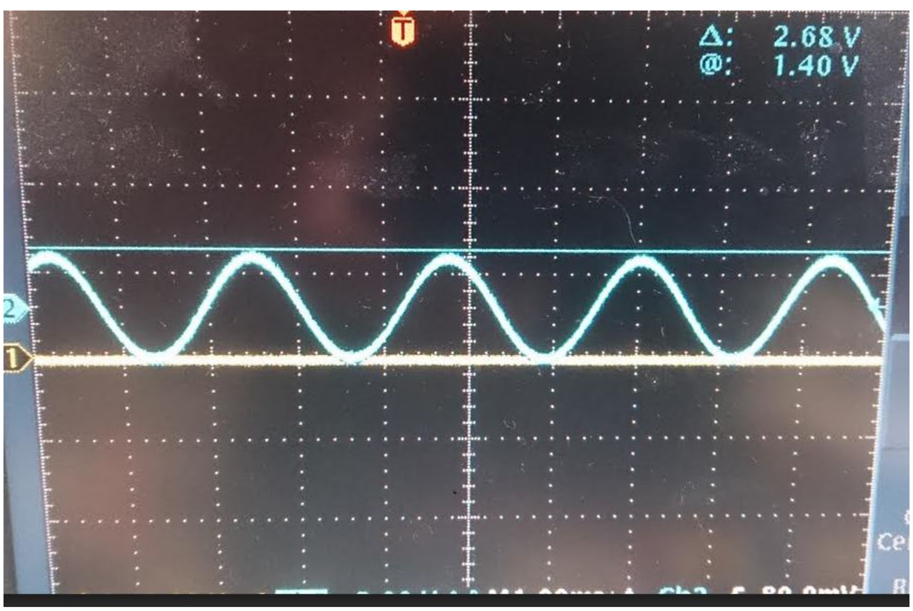

Figure 45: 1KHz output

5.9 Discussion

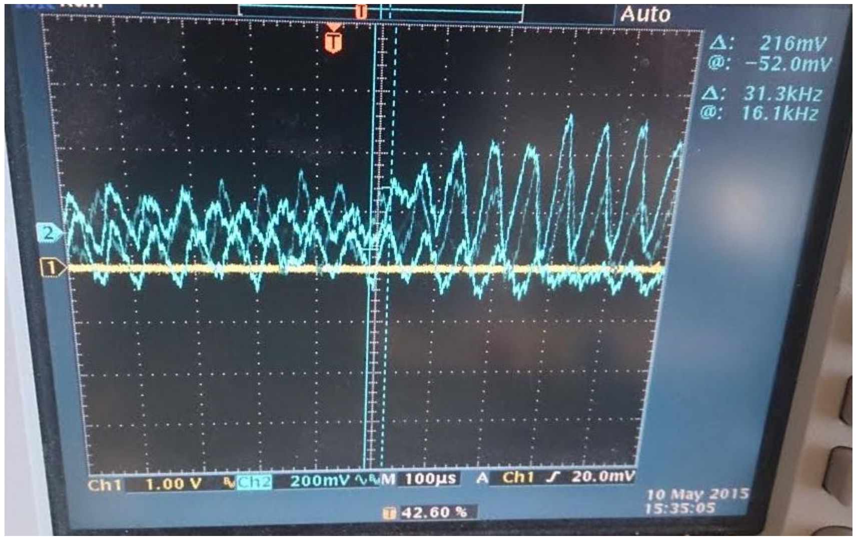

Figure 46: Mains noise

This project design was a great exercise in finding a simple elegant solution with minimum components to the complicated problem of FM demodulation. However, it was not without difficulties. Two main issues were faced during designing and execution of the project. The first approach in designing the slope detector was to use a low pass filter with a cutoff frequency a decade below the carrier frequency. The first attempt with first order low pass filters produced no useful output at a carrier frequency with a modulation index of 5. This is due to having a very low slope of 20Db/decade which is not enough to cause any noticeable amplitude variation of the FM signal. However, using low frequency carrier of 100 KHz with modulation index of 5, the output was discernable. This is not surprising. A modulation index of 5 with a maximum modulation frequency of 15 KHz means a frequency deviation of the carrier frequency up to 75 KHz. This is a huge variation relative to the original carrier frequency. It's then no surprise that there is a significant envelope developed on the top of the FM signal. Going up for the MHz frequency range produces not noticeable envelope.

An approach to fix this was to use a second order filter. A slope of 40 Db of decade would certainly produce noticeable envelope at carrier frequencies of as high as 16 MHz. But these attempts were to no avail.

The only way to fix this is to have a very high slope at which the carrier frequency is centered. A bandpass filter with sufficiently high Quality factor will do the job. And it indeed did the job.

The second main problem that faced us during project execution is mains noise or what is usually called mains hum – see figure 46. Connecting audio input to function generator didn't cause this problem; however, using the camera's mic would introduce pick up mains noise, then it would be modulated along with input audio signal on top of carrier frequency. However, this problem didn't affect the fidelity of the audio output. I tried two methods to eliminate the mains noise:

1-Notch filter: I designed a notch filter centered around 120Hz. It would do a good job eliminating the 120 Hz mains noise, but the harmonic multiples would still be present and the audio output is totally lost.

2-High pass filter: I designed a high pass filter centered at 200 Hz. This suppresses the 120 Hz mains noise, but the harmonic multiples would persist and the audio signal is completely lost.

6 Results

It is possible to transmit NTSC audio and video through optic fiber to output to a television set. This can be quantified by the quality of the signal being output to the television set. In the following Figure 47 shows the pure NTSC camera signal being output directly into the television while Figure 48 shows the NTSC camera signal being output into the television from the optic fiber after amplification. The difference in picture quality is not to noticeable in a still image.

In Figure 48 we can see that there is more lines or dots that seem to make it seem as the video is distorted. Also since the image is rolling in Figure 48, even though both pictures were taken from the same angle and height, the body in the picture in Figure 48 seems lower than the body in the picture in Figure 47.

The audio output was of good quality. There was some distortion due to mains noise. But the output sound was loud and clean enough to be clearly intelligible.

Figure 47: NTSC video signal directly output to the television.

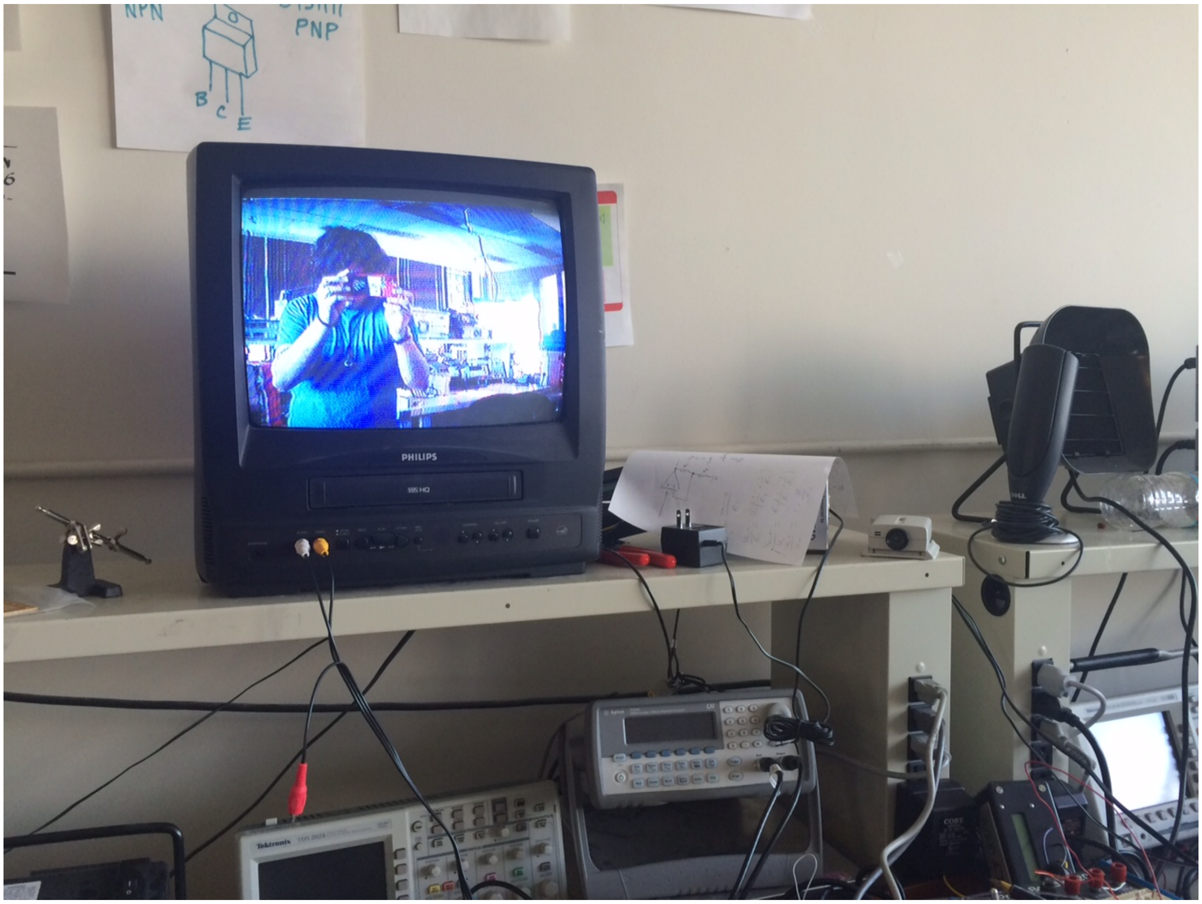

Figure 48: NTSC video signal output to the television after fiber optic transmission.

7 Further Implementation

One of the things that was thought of was how this project could be taken further. This is a list of improvements that could be made to this design:

1. Stop the rolling of the image displayed on the television set
2. Eliminate the 120Hz noise and its harmonics.
3. Add several more audio channels

8 Acknowledgments

We would like to acknowledge and give our thanks to Jim Hom, Joe Sousa, Dan Weber and Mary Caulfield. Jim was instrumental in giving us ideas that eventually led to our final project and offered his expertise in design and debugging when we needed it the most. We thank Joe Sousa for his insight and aid in debugging complex systems. We thank Dan for being our project advisor and guiding us through the design process. We thank Mary for her invaluable help with writing this report. Last but not least, we thank all the 6.101 students and staff for a great semester.

Disclaimer All pictures shown in this report belong to the students who's section they appear in. This means that Germain's pictures appear in his section, Hugo's pictures appear in his section and Khaled's pictures appear in his section. The pictures in the results page belong to Hugo. Unless given credit otherwise, such as in section Transmitter Driver, for the Typical Forward Voltage figure, these figures and pictures belong to these students.

References

[1] *Crystal Unit.* HC49-S. Rev. 1.1. SPC Multicomp. Oct. 2008.

[2] *Silicon 500mW Zener Diodes.* 1N757. Rev. C. Microsemi Corporation. Oct. 2005.

[3] Der, Lawrence. *Frequency Modulation (FM) Tutorial.* Silicon Laboratories, Inc.

[4] *LT1364/1365 Dual and Quad 70 MHz, 1000 V/us Op Amps.* LT1364. Rev. A. Linear Technology Corporation. 1994.

[5] Horowitz, Paul, and Winfield Hill. *The Art of Electronics.* Cambridge: Cambridge UP, 1989. Print.

[6] Horowitz, Paul, and Winfield Hill. *The Art of Electronics.* New York: Cambridge UP, 2015. Print.

[7] *Receiver with analog output for polymer optical fiber applications.* SFH250 and SFH250V. Avago Technologies. October 2011.

[8] *Transmitter for Polymer Optical Fiber Applications.* SFH757 and SFH757V. Avago Technologies. December 2011.