Frequency Dependent Optical Compressor

NYU Music Technology: Analog Electronics Final Project Report

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Introduction

This circuit uses an LED to control the amount of audio compression on an adjustable range of frequencies. This compressor is an example of an optical compressor and uses light to detect the amplitude of an incoming signal. The user can then adjust the average brightness of the LED which directly affects the amount of compression – compression being a reduction of high amplitudes, overall decreasing the difference between quiet and loud parts of an audio signal. Unlike most optical compressors, this compressor allows the user to choose a range of frequencies to compress. For example, the compressor can compress frequencies only below 1kHz (compresses only middle and bass frequencies).

See demo video linked below:

https://youtu.be/p-JJz5How9w

Circuit Analysis

To better understand this single-band, variable frequency, optical compressor, the components of the circuit must be broken down. In this section I have chosen to divide the circuit into four parts; utilities, precision rectification / DC bias, RC filtering, and optical compression. I would also like to outline the overall processes of this circuit. Below you will find a variation on a block chart which outlines independent and parallel processes.

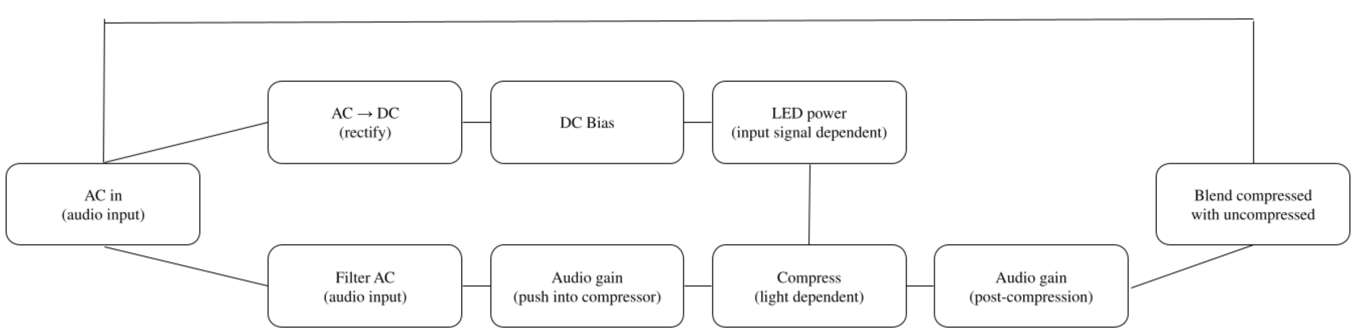

fig.1: Block diagram of single-band, variable frequency, optical compressor

Utilities

Utilities are parts of a circuit that exist for protection or allow the circuit to be divided into functional sub-circuits. The first utility sub-circuit that I used was the audio safety circuit. This sub-circuit exists as the very beginning and very end of the circuit. Consisting of a 220 Ohm resistor and a large 10uF capacitor, respectively, this sub-circuit resists large spikes in current via the resistor then blocks DC voltage due to the decoupled nature of capacitors. The safety circuit on the beginning protects the following circuit while the safety circuit at the end protects speakers, headphones, or any other audio reproduction devices that may be damaged by DC voltages and large current spikes. The second utility sub-circuit used is the buffer or voltage follower. Buffers consist of one op-amp – in this case a TL072 – where the non-inverting input is the input and the output feeds back into the inverting input. This prevents signal backflow and decouples the impedance of the sub-circuit before and after. This means that impedance after the buffer is not dependent on impedance before the buffer.

Precision Rectification / DC Bias

Rectification is converting an AC signal into a DC signal. This is extremely important in this circuit, and in any optical compressor circuit. The manner in which the LED is used requires DC voltage while the input signal is AC. While AC voltage will power an LED, the entire input signal will not be represented and will result in a very bad sounding compressor. The rectification in this circuit is achieved using a precision rectifier circuit (fig.2).

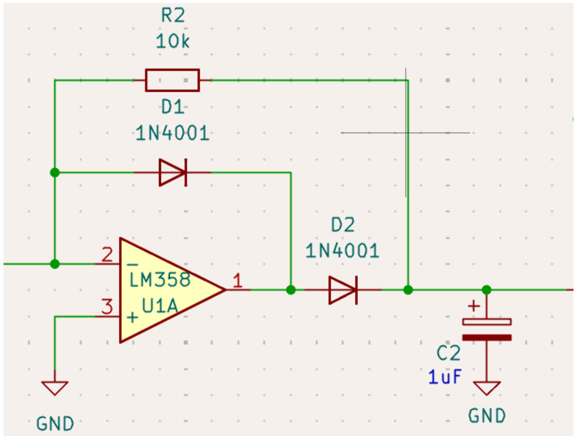

fig.2: Half-wave rectifier sub-circuit

This circuit consists of one TL072, two 1N4001 diodes, a 10kOhm resistor, and a 1uF capacitor. The first diode (in feedback path) is only active given a positive voltage. This allows for positive voltage to accurately follow the positive input and function normally. With negative voltage, diode one doesn't allow for voltage to pass. Diodes must receive an input voltage of at least +0.6V to conduct. Ultimately, positive voltages remain unchanged while negative voltages will be 0V. The last step in the rectification process is to smooth the positive voltages. This is done using a 1uF capacitor which will hold the charge of the positive voltages and release them smoothly. Smoothing the signal will move the signal closer to DC. The result is a DC signal that increases while amplitude of the AC input increases. Precision rectification does not result in an “absolute value” style conversion of the signal but rather takes only the positive voltages. This is not as accurate as a full-wave rectifier that also converts the negative voltage to its absolute value.

The second part of this sub-circuit is the DC bias. This is important because the rectified signal may not be powerful enough to turn on an LED. This circuit uses a white LED which has a forward voltage drop between 3-5V. The solution to this is to inject an adjustable DC voltage into the rectified signal. Injecting a DC is done by taking 9Vin from the power rails and using a potentiometer to divide the voltage (0V to +9V). This is placed in series with the rectified circuit. Adding a constant voltage to a varying voltage is a linear function and won’t change the rectified signal other than increasing the total voltage by a constant. The DC bias can be increased or decreased depending on how much current the LED needs but also serves as a threshold control by varying the average brightness. Increasing brightness → less resistance → more compression. The entire rectification and DC bias sub-circuit can be seen below (fig.3)

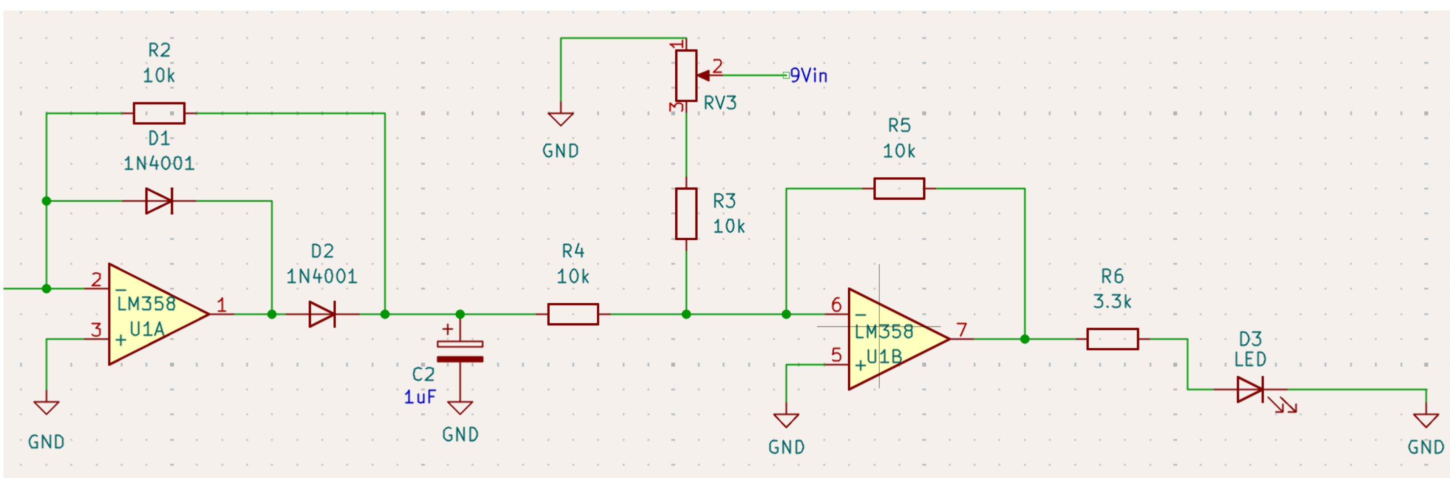

fig.3: Half-wave precision rectifier and DC bias sub-circuit

RC Filtering
RC filtering is used in the compression part of this circuit in order to compress a range of frequencies rather than the whole broadband signal. My initial design had three bands of set frequencies that could be compressed at different levels. Due to the shipping time of more TL072s or TL074s, I was only able to include one band. I decided to make this band variable, as will be described later.

This sub-circuit is in between and in series with both of the buffers/voltage followers and consists of one 10k audio taper potentiometer and one 1uF capacitor. I oriented these to create a low pass filter – potentiometer on input and capacitor to ground. In this circuit, the filtering is not for frequency shaping but rather for selecting what range of frequencies to compress. Since the potentiometer can sweep from ~1 to 10k Ohms, the frequency cutoff maximum and minimum are shown below.

Upper Frequency Cutoff:

$$Fc = 1/(2piRC)$$
$$Fc = 1/(2pi * 1 * 0.000001)$$
$$Fc = 159235.67hz$$

Lower Frequency Cutoff:

$$Fc = 1/(2piRC)$$
$$Fc = 1/(2pi * 10000 * 0.000001)$$
$$Fc = 15.92hz$$

As can be seen in the equations above, the highest Fc with this given capacitor potentiometer configuration is nearly 16kHz. This measurement means that – if the potentiometer is at ~1 Ohms – all frequencies from 20Hz (lowest significant frequency in audio) to ~16kHz will be compressed. The same principle applies to the lowest Fc. All frequencies below ~16Hz will be compressed. It is true that this frequency is not audible to humans but this is the closest workable potentiometer value we had. The schematic for this section can be seen below (fig. 4).

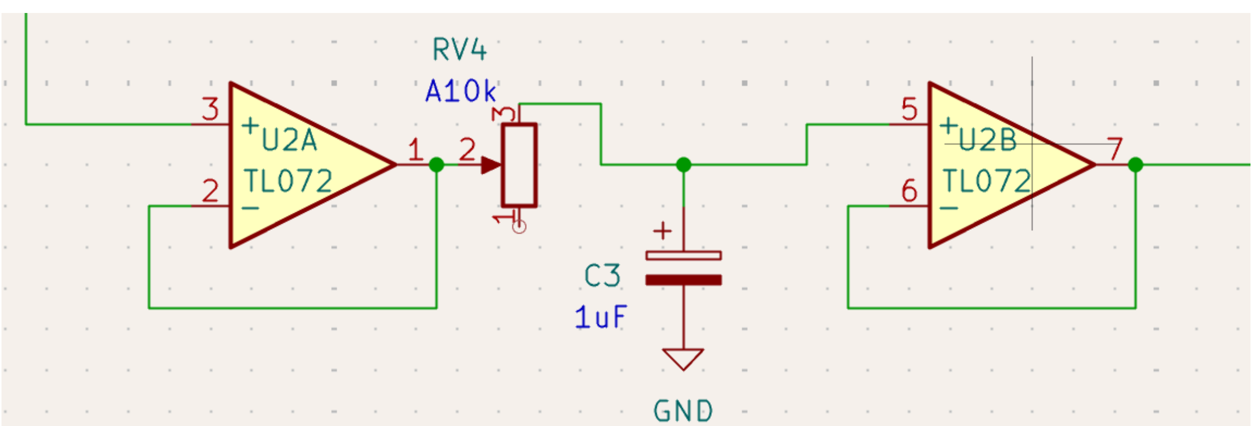

Fig.4: Variable RC filter sub-circuit

Optical Compression
The most important part of the circuit is the optical compression. Half of this sub-circuit is explained in the rectification and DC bias section of this paper. This section will focus on the part of the compression circuit that passes the audio signal – not the rectified signal.

This sub-circuit consists of one 10k resistor as *Rin*, a photocell (~8 to ~20kOhms) coupled to the white LED as *Rf*, and a TL072. There are two non inverting op-amps before and after the inverting (compressing) op-amp. These consist of one TL072, one 10k audio taper potentiometer, and one 10k resistor. This is shown below in fig. 5.

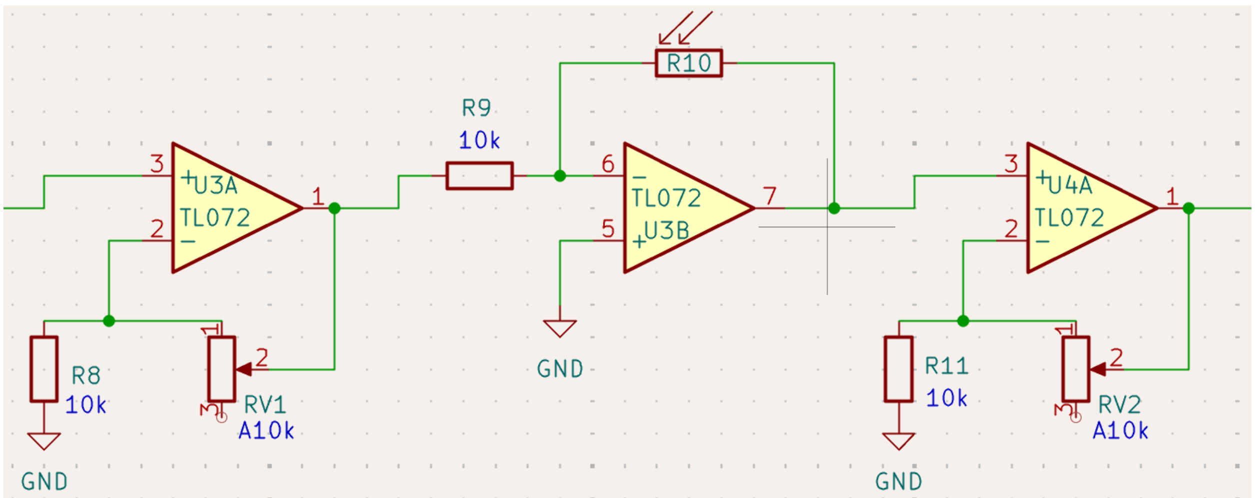

fig.5: Gain reduction and input/output gain sub-circuit

Since compression is essentially a reduction of the peaks of an audio signal, it would be considered a non-linear function. It is important that some kind of threshold is set so that the whole signal is not reduced. To achieve this, a non inverting op-amp circuit is required with a photocell (light dependent resistor) coupled to the LED as *Rf*. This coupled LED and photocell apparatus is called vactrol. The vactrol will decrease in resistance as the LED gets brighter, and vice versa for lower levels of light due to the nature of photocells. A lower resistance in *Rf* means less gain because the op-amp is not trying to match the gain of a signal that is far lower than the input. As a result, a brighter LED means lower resistance at *Rf* which decreases gain of the incoming audio signal. All of this is a result of the amplitude (rectified audio signal) of the same audio signal. Simply put, the input audio signal is reacting to itself.

Inverting Op-Amp Maximum Gain:

$$Gain = -(Rf/Rin)$$
$$Gain = -(20000\,/\,10000)$$
$$Gain = -2$$

Inverting Op-Amp Minimum Gain

$$Gain = -(Rf/Rin)$$
$$Gain = -(8/10000)$$
$$Gain = -0.0008$$

To control input and output gain, I made two variable non-inverting op-amps each with a maximum gain of 2 and minimum gain of 1. The non-inverting op-amp circuit adjusts how much of the signal is being passed to the compressor. Increasing the input gain ultimately results in more compression if the DC bias for the LED is static. The inverse is true for less input gain. The non-inverting op-amp, after the signal is compressed, is the makeup gain. This is to bring the compressed signal to a desired level after the transients have been reduced. A signal that is only compressed is far quieter than the original signal so normalizing the levels between the two is important.

Non-Inverting Amplifier Maximum Gain

$$Gain = 1 + (\mathrm{Rf/Rin})$$
$$Gain = 1 + (10000/10000)$$
$$Gain = 2$$

Non-Inverting Amplifier Minimum Gain

$$Gain = 1 + (\mathrm{Rf/Rin})$$
$$Gain = 1 + (1/10000)$$
$$Gain = 1.0001 \approx 1\ (no\ gain)$$

Lastly, I added a blend knob with the middle pin connected to the audio out and the outer pins connected to the compressed signal and dry input signal. This for a blend of the compressed low frequencies with the uncompressed high frequencies. In the future, I would like to use a dual gang potentiometer to also filter out the low frequencies of the dry signal so that all low frequencies below *Fc* are compressed. This is the major flaw of my current circuit. The compression is very subtle because of the blend of the dry low end. The full schematic is shown below (fig.6)

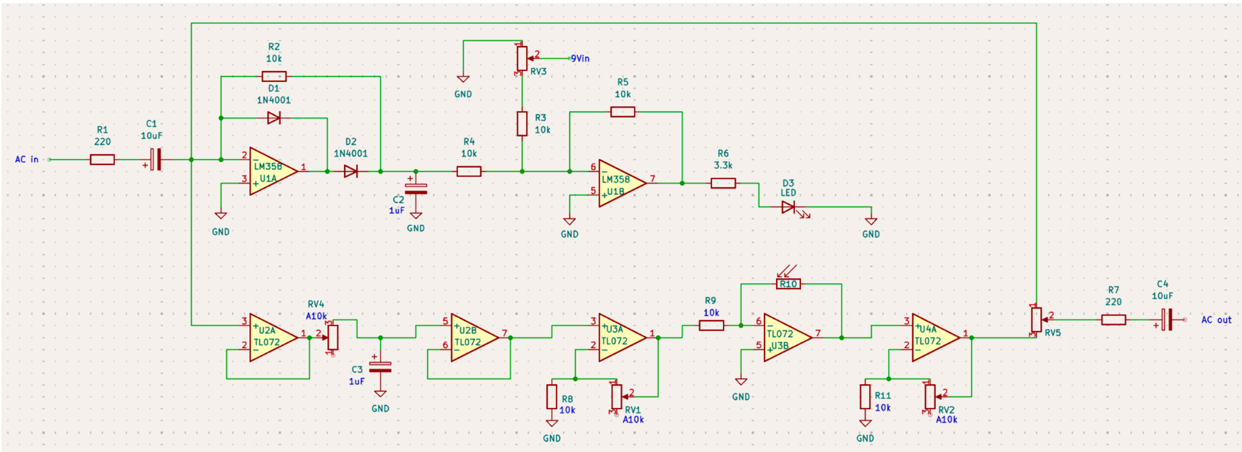

fig.6: Full schematic

Conclusion

This project was definitely not what I planned and I am certainly sad that I wasn't able to get all the parts in time. Regardless, I am very happy that I could make some sort of frequency dependent optical compression work. I feel that the challenging part of my original idea was executed here and I enjoyed debugging and testing this circuit. Despite certain finicky potentiometers and vactrol configurations (result of either low tolerances or large value ranges) there were few major hiccups other than changing my project idea at the last minute. I hope to finish this project over winter break with the full set of features.