Abstract

Our project was building a simple guitar distortion amplifier that adds a controlled distortion effect to a guitar. A typical guitar amplifier allows one to control the amount of such distortion, control the tone of the resulting sound (i.e. the amplifier has some form of equalizer), as well as control the resulting volume levels. Thus the objective was to build a circuit that would:

- Amplify the weak guitar signal
- Add a controllable distortion effect to the guitar signal
- Have a controllable equalizer that would allow us to emphasise/de-emphasise certain frequencies (i.e. control bass/treble)
- Have a volume control

The processed signal was then outputted to a speaker, with a real electric guitar used as input. The resulting audio was recorded, with each of the controllable parameters adjusted.

Background

When a guitar string is plucked, the mechanical energy of the vibrating string is converted to an electrical signal by a transducer [4]. The transducer consists of a permanent magnet wrapped in a coil of copper wire, with this magnet magnetizing the guitar string creating a magnetic field aligned with the magnetic field of the permanent magnet. When the string vibrates, the moving magnetic field induces a current in the coil around the permanent according to Faraday's Law. [4]

Amplification

The electric signal produced by the guitar (via the transducer) is extremely low (order of magnitude 0.1V - which was observed in lab), and so this signal needs to be amplified before it can be processed. To achieve this, a simple non-inverting op-apm was used (throughout this circuit, all op-apms were powered by $\pm 15VDC$). R_1 was chosen to be 100Ω, R_2 chosen as $4.7k\Omega$, leading to a gain of about 47, and so the guitar signal was now around $5V$.

The Distortion Effect

Our first task was to come up with a method to add a distortion effect to our guitar signal. This effect ideally transforms the guitar sound into a 'fuzzier' and 'grittier' version of the original sound by clipping the input signal (i.e. amplifying the signal past its maximum, flattening out the peaks and troughs of the signal), while adding in harmonic/in-harmonic overtones as well as a sustain effect (making the notes continue to ring out even after the guitar string vibration ceases) [3]. In order to increase the input signal to the point that it is clipped, we use a non-inverting op-apm with R_1 and R_2 chosen such that the gain causes the guitar signal voltage to increase beyond the op-apm's power supply, leading to clipping. R_2 is chosen to be a $1M\Omega$ trimpot, with R_1 a $1k\Omega$ resistor, so that gain can be adjusted from 0 to 10^5 by adjusting the trimpot's resistance.

When the signal is clipped in this way, even a low input signal pushes the circuit to its maximum (i.e. causing clipping), with the output level remaining high even after input fades (due to gain) [5]. This is what leads to the sustain effect of our guitar amplifier.

Clipping is also what leads to the introduction of overtones; the effect of clipping leads to frequencies not originally in our signal being produced[3]. The way in which these overtones are introduced depends on the type of clipping used. 'Hard-clipping,' where the gain leads to the signal being higher than the op-apms power, leads to the peaks of our signal abruptly flatten, which results in a much harsher sound (the new frequencies may not share a harmonic relationship with the original signal) [3].

To create a 'warmer sounding' distortion effect, we will use 'soft clipping' ; this process gradually flattens the peaks of our signal, rather than doing it abruptly like hard clipping, leading to an introduction of a higher number of harmonics that share a harmonic relation to our original signal [3], leading to a less harsh sound. In order to create such an effect, we want different parts of our input signal to experience a different gain. To facilitate this type of soft clipping, we introduce switching diodes into our non-inverting amplifier circuit. The original idea to create this type of soft clipping came from wamplerpedals.com [1].

The gain of a non-inverting amplifier is determined by the ratio of the resistance inside to feedback loop, to the resistance of the connection from negative input of the op-apm to ground. In order to have different gain at different points of the input cycle, two switching diodes were placed in opposite directions in the op-apm feedback loop. At every instant that the current changes directions, both diodes have infinite resistance, so current flows through the trimpot and resistor connected to ground, leading to a gain of 10^5 (as explained previously). A few microseconds later however, the resistance of one of the diodes drops to zero (because there is now current flowing in the forward direction through it). Now current is flowing throught this diode and resistor connected to ground. This makes $R_2 = 0$, and the gain is now 1. At the next instant current changes direction, this diode now has infinite resistance (current flowing backwards), while the other diode's resistance does not drop to zero for a few microseconds (i.e. it is still infinity). In this instant, current is flowing through the trimpot and resistor connected to ground, and we have 10^5 gain. At the next instant, the process repeats and there is a gain of 1. This delay in the diode's resistance dropping from infinity to zero when a forward direction current is applied is due to it being a switching diode [2]. This leads to different gain being applied to different parts of the input signal, with the gain effect now that of 'soft clipping'; the peaks gradually flattened. Capacitors were added in the diode loop as well as between the outer resistor and ground to filter our high/low frequency noise respectively.

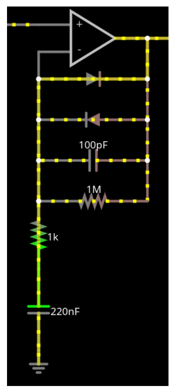


Figure 1: Falstad diagram of soft clipping diode circuit

The Equalizer

The function of the equalizer is to control the tone of the output sound, i.e. how much of the high, mid and low frequencies are heard. A typical guitar amplifier has knobs to control this effect; a 'treble' knob (a knob that controls how much the high frequencies are emphasised) and a 'bass' knob (a knob that controls how much the low frequencies are emphasised), as well as other knobs to control the mid-range frequencies (everything other than treble and bass). In order to create this effect, we created a band-pass filter, using variable resistors to adjust the high and low cut-offs as needed.The bass frequency range is approximately $20Hz - 160Hz$ while the treble frequency range is $2.5kHz$ and beyond, the mid ranges everything in between. The initial idea was to create two filters, one controlling each of these frequency ranges:

To control the bass (the low frequencies) a high-pass filter was used. We want this filter to be controlled by a potentiometer such that at one extreme of the potentiometer, no low frequencies are allowed to pass (i.e. high-pass cutoff frequency of $\approx 160Hz$), while at the other end of the potentiometer, all frequencies are allowed to pass (i.e. high-pass cutoff frequency less than $20Hz$). Using a $1\mu F$ capacitor along with the fact the cutoff frequency is determined by $\frac{1}{2\pi RC} = \frac{1}{2\pi R(10^{-6})}$, a $22k\Omega$ trimpot was chosen; this value was chosen as it was readily available in the lab, with its max resistance leading to a cutoff frequency of $\frac{1}{2\pi(10^{-6})(22000)} \approx 7Hz$. In order for the higher end of the cutoff to be $\approx 160Hz$, we determined that having the trimpot at 1000Ω would be sufficient. As it is tricky to adjust the trimpot to this exact value, a 1000Ω resistor was added in series to the trimpot, so that when the trimpot resistance was set to 0, our total resistance was 1000Ω, leading to $\frac{1}{2\pi(10^{-6})(1000)} \approx 160Hz$. Although this does increase our max resistance to $23k\Omega$, that simply lowers our other cutoff frequency to $\approx 6Hz$, which is still less than $20Hz$, so all possible bass that a human can hear is still audible. Hence, by adjusting the potentiometer of the high-pass filter, we now control the bass in our output sound.

To control the treble (the high frequencies) a low-pass filter was used. The setup was analogous to the high-pass filter, where now at one extreme of the potentiometer we want a low-pass cutoff frequency around $2kHz$, and the the other extreme of the potentiometer we want an infinite low-pass cutoff frequency. A 100Ω trimpot would achieve this, as $\frac{1}{2\pi(10^{-6})(100)} \approx 1.6kHz$, while $\frac{1}{2\pi(10^{-6})(0)} \approx \infty$, hence by adjusting the potentiometer of the low-pass filter, we now control the treble in our output sound.

As we only had two filters in our bandpass circuit, one controlling frequencies from $7Hz - 150Hz$ and the other $1.6kHz$ to ∞Hz, our equalizer was not affecting the midrange frequencies at all. In order to account for this without having to add another filter, while still using the high-pass filter to control the lower frequencies and using the low-pass filter to control the higher frequencies, the fixed resistors for both the filters were swapped: this meant the high-pass cutoffs were now 7Hz to 1.5kHz, the low-pass cutoffs 150Hz to ∞. Adjusting the trimpots now let us control the amount of high and low frequencies (the extreme ends of each still corresponding to 0 or max low/high frequencies respectively), while letting us control midrange frequencies as well, similar to a real equalizer.

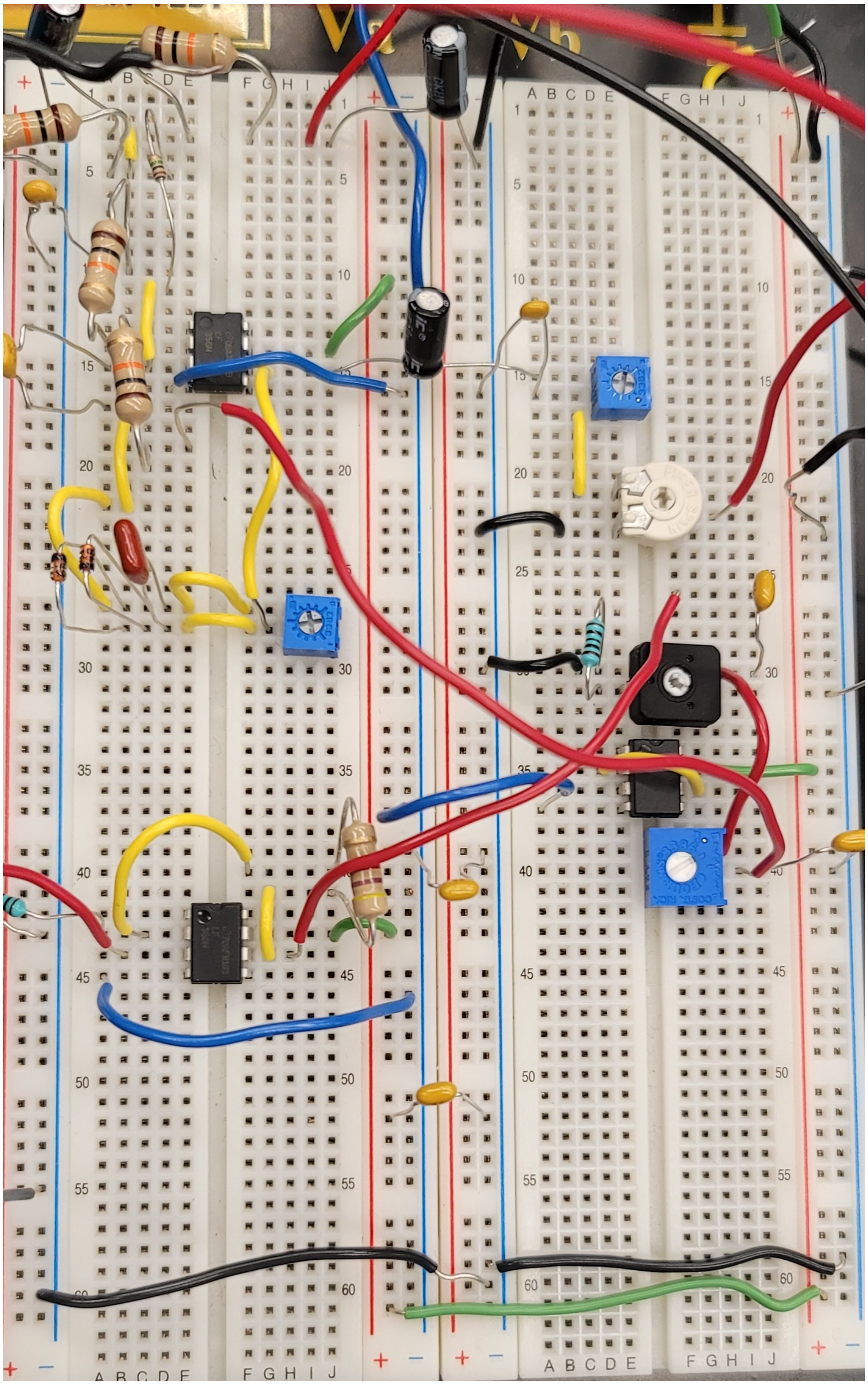
Initially, the equalizer was connected at the end of our circuit (i.e. after the portion of the circuit controlling the over-drive distortion effect). This led to an unexpected observation: almost no gain was heard in the output signal. After some investigation, we realized that the addition of the overtones previously explained was part of the distortion effect heard, and our equalizer appeared to be removing this effect by removing some of those frequencies. We then connected the equalizer *before* the distortion gain control, which fixed the issue.

Volume Control

This was the simplest component of the circuit. A high resistance trimpot ($100k\Omega$) is connected across our processed output signal and ground. Lowering the resistance of the trimpot shorts our output signal to ground, while increasing the resistance allows output signal to flow to whatever output device (scope or speaker) is connected. This let's us control the amplitude of our output voltage, higher amplitude (higher resistance of trimpot) corresponding to increased volume and vice versa.

Final Result

The following is an image of our completed circuit. The op-apm in the bottom left is our first amplification op-apm, that increases the guitar's input voltage from $\approx 0.1V to \approx 5V$. This signal is then sent to our band-pass filter on the bottom right of the breadboard (this was initially the end of our circuit as previously explained, hence the unusual placement), with the top trimpot the high-pass filter controller, the bottom trimpot the low-pass filter controller. The output of the equalizer was then sent to the top left; our soft-clipping op-apm. Finally, the output of this op-apm was sent out to a speaker, with it connected across a $100k\Omega$ trimpot to ground, to act as the volume control.



A rough draft of the Falstad simulation can be found here `https://tinyurl.com/y52bsotb` (rough draft as small adjustments were made to the physical circuit).

Connecting our guitar to input, output to a speaker, the following video was taken displaying our circuit in action. I am narrating and demonstrating what happens when the circuit is adjusted (i.e. what happens when each of the trimpots is adjusted)while my lab partner Adrian is playing the guitar. Adjustment of the overdrive (i.e. gain) effect, adjustment of volume, and adjustment of the equalizer (i.e. output of high and low frequencies) is observed. The video has been attacehd with the Quercus submission, and can also be found at the following link (it is being shared from my google drive): `https://drive.google.com/file/d/1w0Iv3ihA9TXET5QsHHuXjUFAI1qOq9UD/view?usp=sharing`

References

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