Signals and Systems Final Report : Sheet Music Player

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1 Abstract

The motivation behind this project was to create a device that would not only play sheet music, but also create an acoustically pleasing sound; moreover, we sought to reduce the dependency by implementing the project fully with hardware.

Our simulated "sheet music" is a grid of black and white boxes, with black representing notes. We propose to play it by sliding it over a set of photodiodes, which will, in turn, activate the oscillators. The signal from the waves would be modulated to enhance its sound with harmonics, so they sound more natural – like an instrument. For the scope of this project, we demonstrate a functional circuit for a single pitch, which may then be duplicated to create a wider range of pitches.

2 Process

2.1 Sensor

The first part of the circuit interprets the visual data through a sensor into an electronic signal. This is done by reading the sheet music with photodiodes. The corresponding circuit is shown in Figure 1.

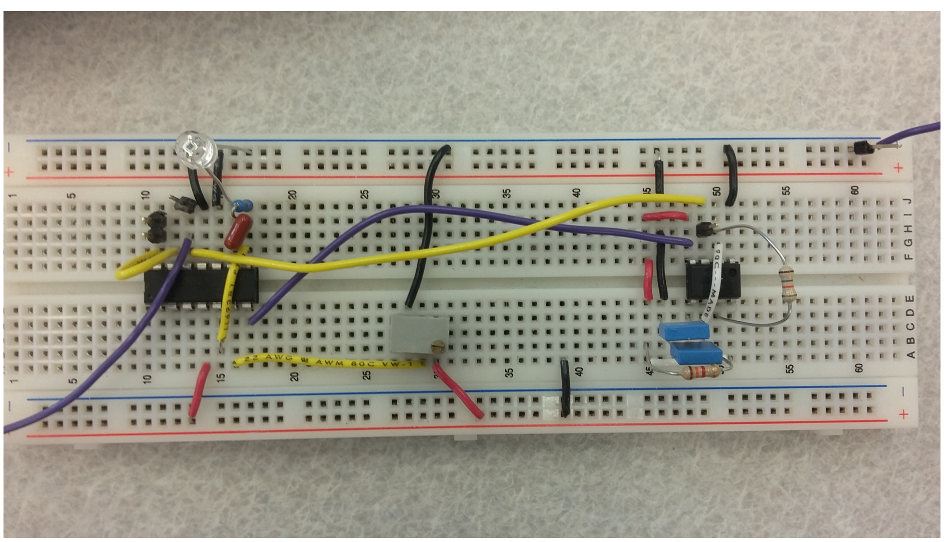


Figure 1: The circuit we used to get our input signal

A photodiode is used to generate current, which is run through an op-amp to push the signal to the rails. A potentiometer was installed to calibrate the voltage threshold at 800 mV, such that when the photodiode is covered by black paper, the op-amp will output 5V, and when it is covered by white paper, the op-amp will output 0V. We then use this output to activate the oscillator.

2.2 Oscillator

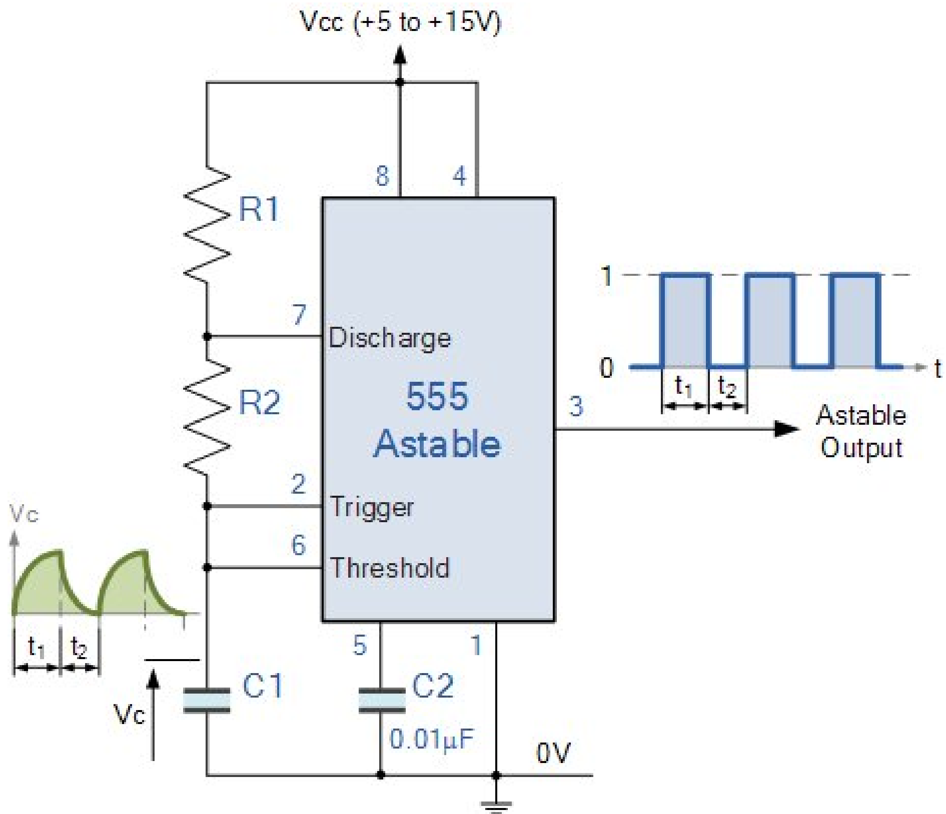


Figure 2: The 555 timer setup we are using

In order to generate the base frequencies for the notes we are using 555 timers (seen in the picture in previous subsection). The timer in our circuit is set up in the astable oscillator configuration at 230 hertz.

The governing equation of the 555 timer in the astable multivibrator circuit is as follows:

$$t_1 = log(2) \cdot (R_1 + R_2)C$$
$$t_2 = log(2) \cdot R_2 C$$

We chose our component values as $R_1 = 60k\Omega, R_2 = 13k\Omega, C_1 = .047\mu F$ to give the pulse wave as close to a 50% duty cycle so that it would be a musical square wave. In order to preserve the voltage against unintentional expenditure of current through a virtual voltage-divider, we buffered it through a unity-gain follower before passing the output to the next section of the circuit.

2.3 Frequency Doubler

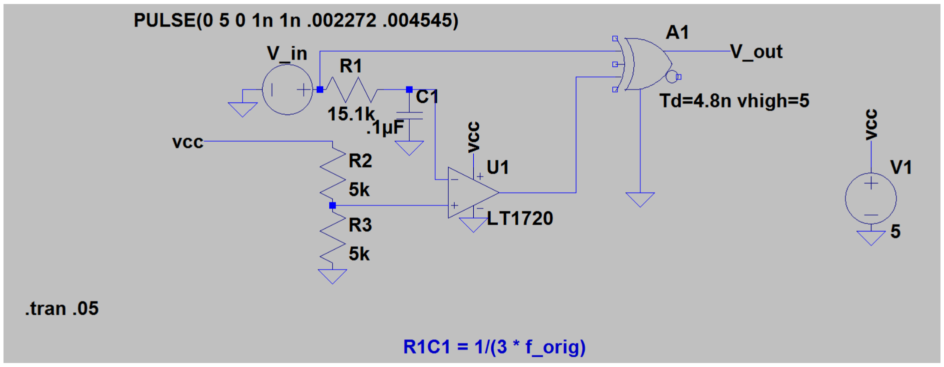
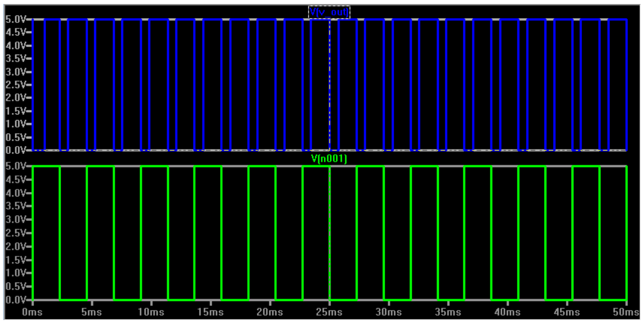


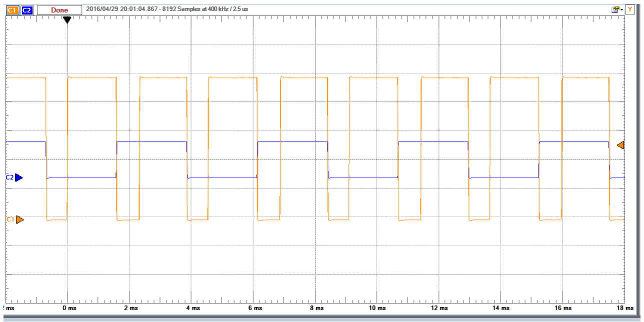
Figure 3: The frequency doubler with the XOR gate.

Because the 555 timer oscillator circuit generates a square wave, it is possible to double the frequency of this signal using an XOR gate and a delay. Namely, by introducing a delay of the input signal as another input to the comparator, there is an extra overlapping region in which the two differ, which creates the doubled frequency; the governing equation for the delay circuit is as follows:

$$R_1 C_1 = \frac{1}{3 * f_{orig}}$$



(a) Theoretical



(b) Measured

Figure 4: The input and output to the doubler circuit; the theoretical result was simulated in LTSpice.

As seen, the circuit doubles the frequency of the square wave. Because the XOR gate operates in the digital domain, this is not the case for the sine wave, which reaches its peak and trough briefly and gradually.

2.4 Low Pass Filter

After we have doubled the signal, we put it through a low pass filter. This changes the square wave signals to a cosine waves, which is what we want for modulation. To modulate the signals, we put them through an analog multiplier. Because square waves are sincs in the frequency domain, if we put square waves through the analog multiplier, we would not get a desired output.

If we multiply two cosine waves, however, we will get a relatively interference-free output. This is because a cosine wave in the frequency domain is two finite impulses; Modulating these finite impulses creates a clean frequency shift.

After multiplying the cosine waves, we get a modulated signal, shown below. In this figure, the blue signal is a perfect cosine wave from a function generator, and the orange signal is our measured modulated output.

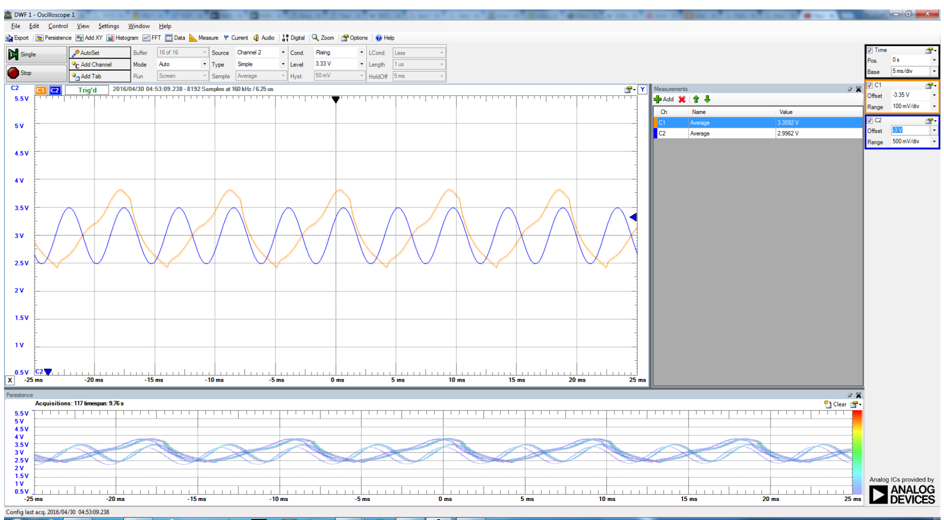


Figure 5: The modulated signal with an ideal sine wave as input, verifying that the multiplication operates as intended.

2.5 Analog Multiplier

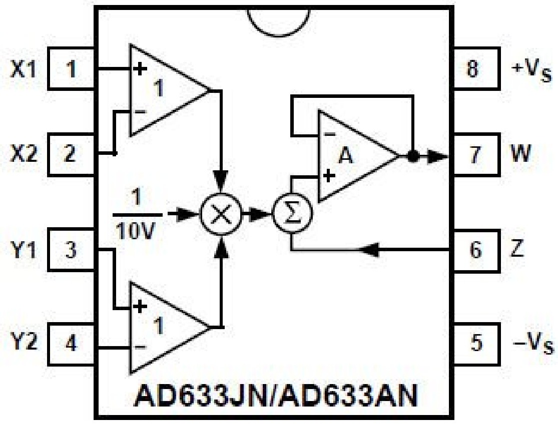


Figure 6: AD633 Integrated Circuit for Analog Multiplication.

Modulation and Demodulations are both convolutions in the frequency domain, meaning that they correspond to multiplications in the time domain. After unsuccessful attempts to implement Log-Antilog, MOSFET-based, and BJT-based multiplier, we decided to use an integrated circuit, AD633, for the multiplication. One decided advantage was that it was possible to eliminate the DC offset in the process of multiplication.

$$V_W = \frac{(X_1 - X_2)(Y_1 - Y_2)}{10V} + Z$$

In the multiplication process, we have thus far assumed a perfect cosine wave in terms of the carrier signal. As this is not the case, the dc offset – among other noises – preserves the content of the original signal proportionally, which is undesirable. Because the AD633 is a differential multiplifer, we were able to reduce the noise introduced by the DC offset.

To demodulate our signal, we ran the signal through the analog multiplier again with the half-frequency wave as our carrier wave. After the demodulation, the signal would be enhanced with the 920 Hz harmonic, as well as some residue of the 230 Hz carrier signal, which is acceptable for a musical superposition. The results will be discussed in greater detail in the following. section.

3 Results

Our end result was a signal that produced a much richer sound than that of our original oscillation – although there is still a buzzing sound in our final output due to slight mismatches in our resistors. You can hear a sample of the audio our circuit outputs in our final video, linked here.

3.1 Frequency Analysis

We will discuss the frequency content of our signal at several key points in our circuit.

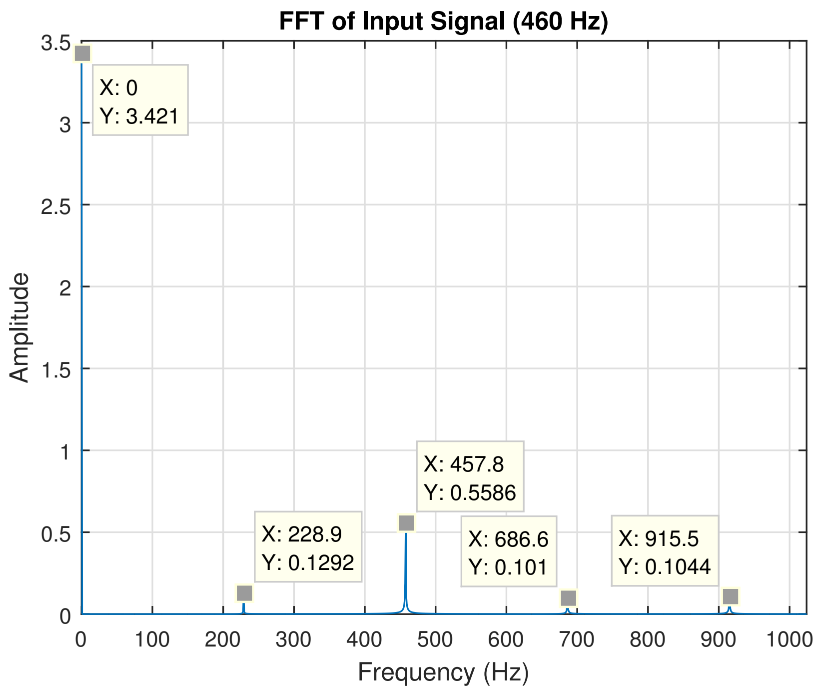


Figure 7: The frequency content of the input signal; as seen, the most prominent frequency is at 460 Hz.

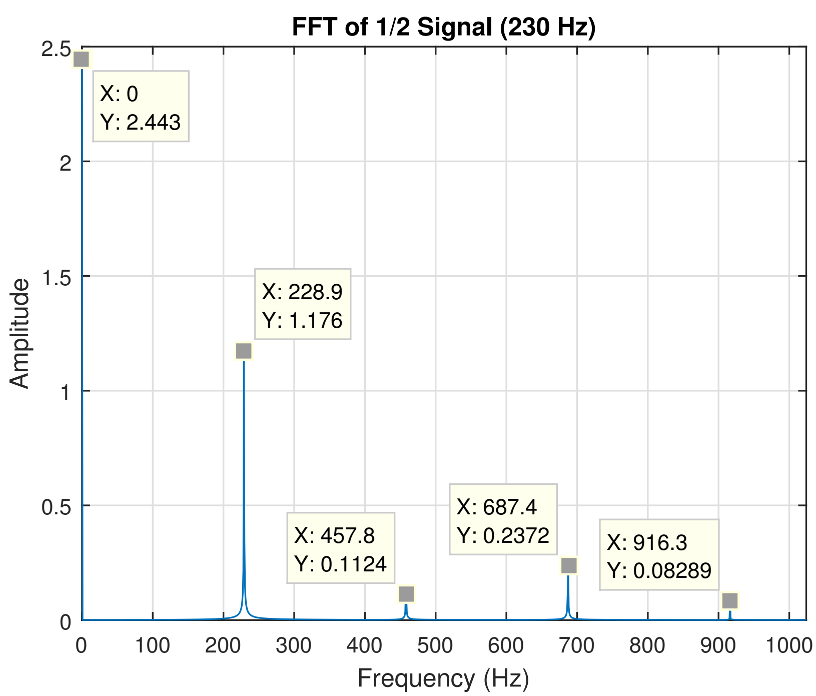


Figure 8: The frequency content of the carrier signal, at half the frequency. As seen, the most prominent frequency is at 230 Hz.

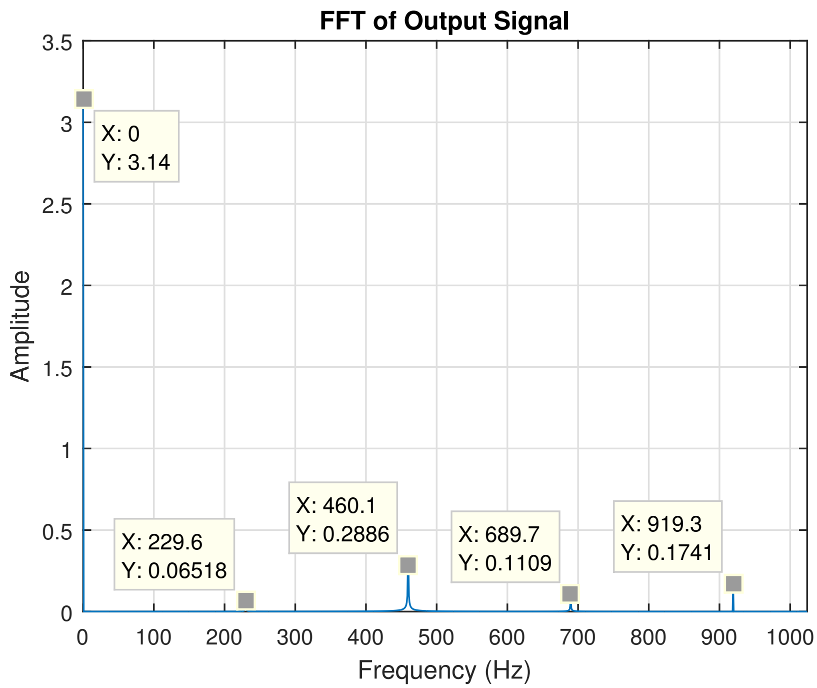


Figure 9: The frequency content of the output signal.

As intended, the amplitude at the first harmonic of the input frequency (960 Hz) is greatly enhanced compared to the original circuit. Moreover, by eliminating the DC offset, the residue of the carrier signal was greatly reduced.

The following graph demonstrates the signal waveform at each step along the circuit:

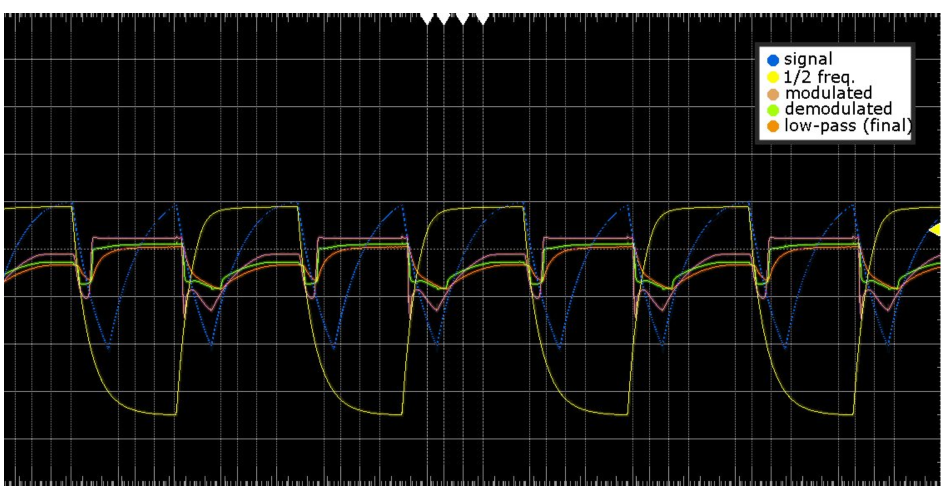


Figure 10: The output signals from our circuit, superimposed to see how they vary in their respective stages.

3.2 Comparison

To verify our results, we compared them to theoretical results using MATLAB. As shown in Figure 9, a signal at the identical frequency was generated via MATLAB and followed the same process, mathematically, as the circuit. The compared results are nearly identical.

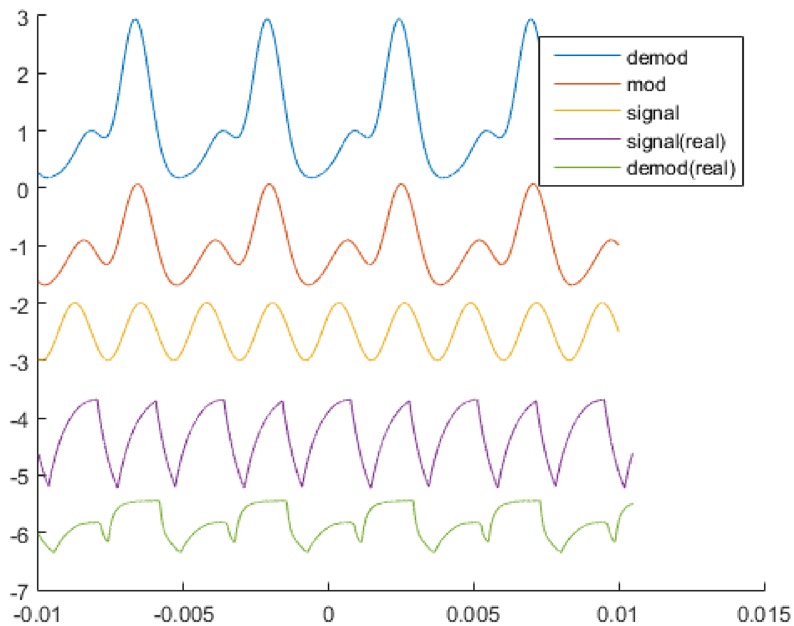


Figure 11: The comparison of our measured results with the theoretical result.

In the above Figure 9, the top three signals represent a theoretical, ”perfect” system. The bottom two signals are the real signal we created and our output after modulation and demodulation; the similarity between the theoretical and the measured signals are noteworthy.

4 Conclusion

We have demonstrated that it was indeed possible to create a system that responds to black-and-white grid sheet-music and output a corresponding signal, which is then manipulated in the frequency domain to enhance its acoustic qualities; upon this proof of concept, the full system is merely a duplication of the thus produced circuit at the desired frequencies.