Realistic Synthesis of Musical Instruments and Performance

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INTRODUCTION

For the past century, much research has been done on the physics of stringed instruments, to the extent that almost every aspect of stringed instruments are well understood. More recently, researchers and musicians alike have begun studying the synthesis of musical instruments. This research has also progressed tremendously, and various techniques have been developed for realistic computer synthesis of the sound of stringed instruments.

The Fourier transform is a complex mathematical formula that decomposes a signal into the frequencies that make it up. The equation shown below is the discrete form of the Fourier transform, which can be used in a number of applications through Fourier analysis.

$$X_k \overset{\text{def}}{=} \sum_{n=0}^{N-1} x_n \cdot e^{-2\pi i k n/N}, \quad k \in \mathbb{Z}$$

A musical instrument produces a unique sound wave, which can be analyzed through the Fourier transform. The Fourier transform of this sound wave reveals that one note actually contains many harmonics. These harmonics of varying amplitudes, when added together, are what constitute the unique timbre of an instrument.

The purpose of this research project is to synthesize a realistic violin sound through additive synthesis by using the Fourier transform. By decomposing a violin sample into its primary harmonics through Fourier transform, the violin sound can be synthesized by recreating the sound wave, which is accomplished by adding all the harmonics back together.

MATERIALS & METHODS

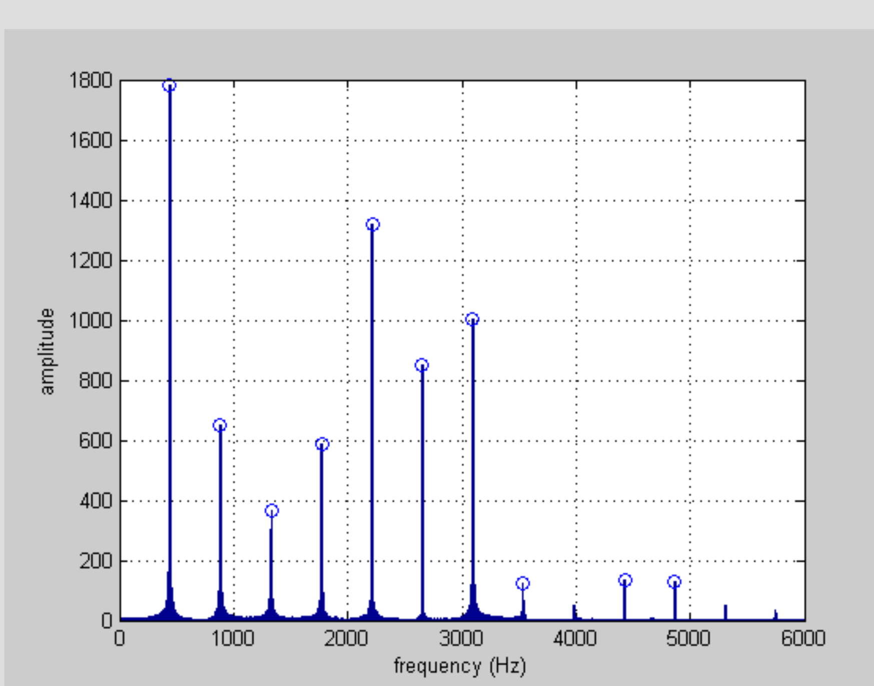
For this research project, I used violin samples created by the University of Iowa Electronic Music Studios, and conducted the Fourier analysis in Matlab. I originally considered creating original violin samples, but settled with the samples created by the Electronic Music Studios, as the professionally-recorded samples would provide the most accurate results.

The initial step of the project consisted of feeding a violin sample of A_4 (440 Hz) through Matlab. Then, the Fourier transform of a slice of the violin sample decomposed the sample into the various harmonics that composed the note, also listing the amplitude that corresponded to each harmonic. The peaks in the graph are the amplitudes of the primary harmonics that compose the note, and the amplitude corresponding to each peak is taken.

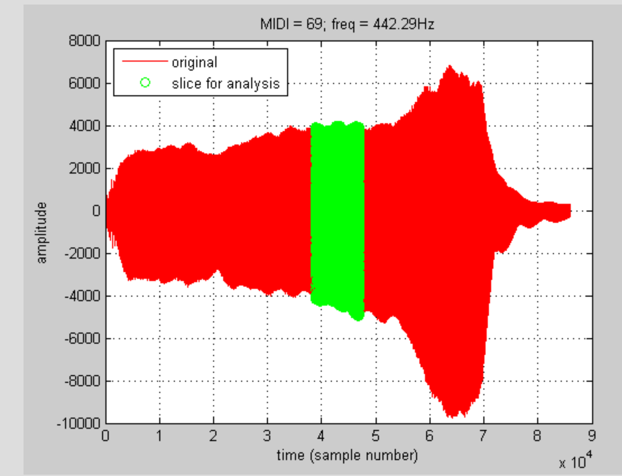
The next step was to graph the frequencies against each harmonic, and find the LSRL of the resulting data points through curve-fitting. The purpose of this step was to find a more accurate frequency of the note, to be used for synthesis later, as the LSRL would average out the error in the frequencies calculated from the sample.

The following step was to analyze the graphs of the amplitude and phase of the harmonics versus the frequencies of the harmonic, which were also calculated through the Fourier transform.

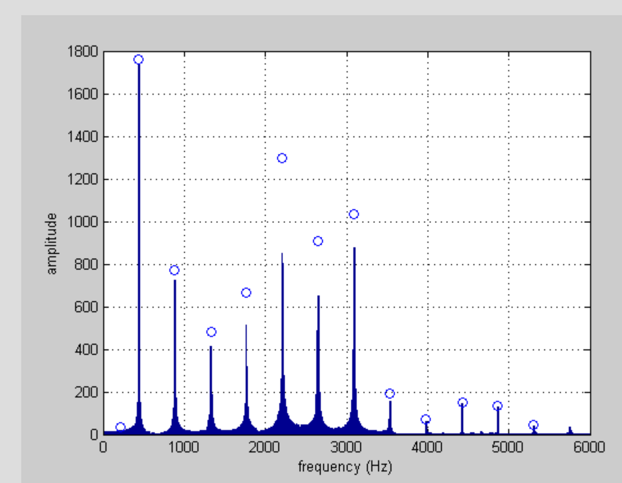
Finally, the amplitude and phases of the harmonics were superposed to create a complex wave, which was the representation of the synthesized violin sound.



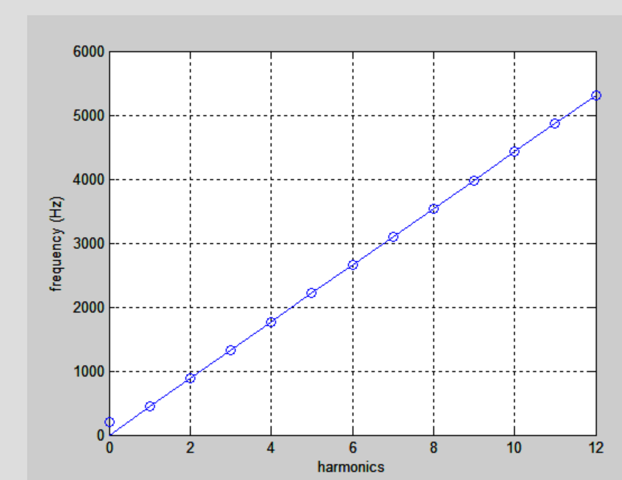
RESULTS



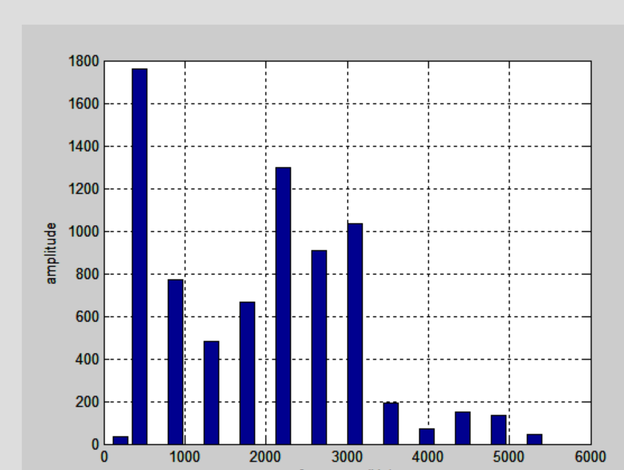
Acquired Time Signal



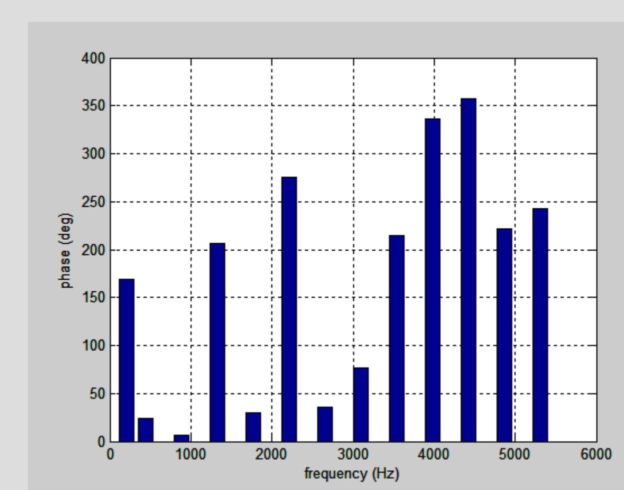
Fourier Transform



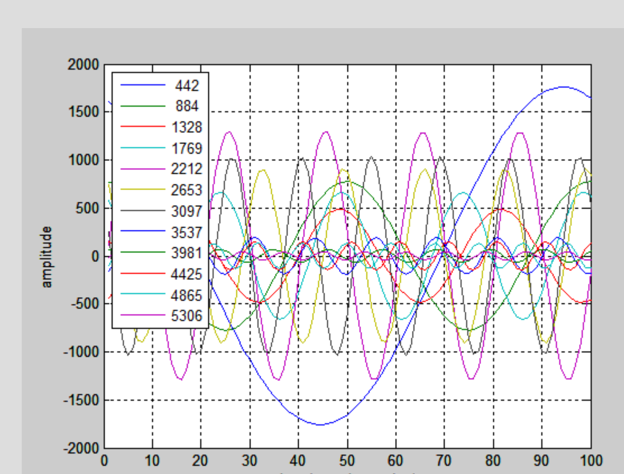
Linear Curve-fit



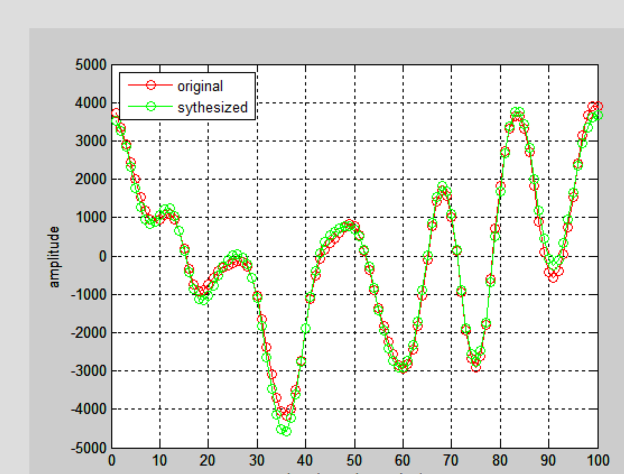
Amplitude vs. Frequency



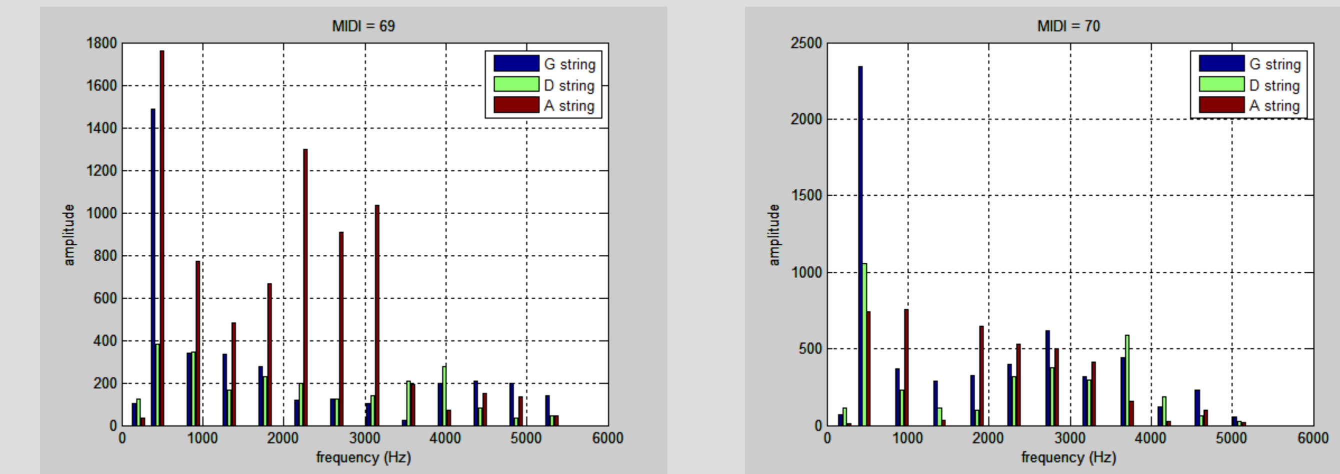
Phase vs. Frequency



Individual Harmonics



Comparison Between Original and Synthesized



Comparison of the Same Note Between Different Strings

SUMMARY

Through the usage of Fourier transform, we wrote a program in Matlab to synthesize a relatively realistic violin sound.

Further exploration can be carried out in several ways. The Fourier transform is a powerful tool to analyze not just one violin, but any musical instrument. Through similar methods, we can compare the decompositions of instruments, or one violin against another violin. This may lead to discoveries that explain what causes the difference in timbre between two of the same instrument or two different instruments.

Also, sound synthesis itself is not limited to just additive synthesis through the Fourier transform. Further exploration can involve the usage of the waveguide synthesis model, or the development of other techniques.

ACKNOWLEDGEMENTS / REFERENCES

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http://theremin.music.uiowa.edu/MIS-Pitches-2012/MISViolin2012.html (sound samples)

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