Subject: Synthesis of Audio Signals
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Introduction

This project consists in three parts. The first two consist in emulating an already known sound using MATLAB as a tool. The chosen sound is a recording of a piano (piano.wav). The main method to make the synthesis is the additive synthesis.

This sound will be used first in the *synth.m* function to render the Bumble Bee piece for the SynthChallenge project. It was later used in the second part of the project which consists in mapping the sound into a two octave range scale, in this case is C major.

The third part is to create an arbitrary sound. I chose the FM synthesis as the main method for this part.

Analysis

Using the function *analysis.m* we can analyze the harmonics of the sound and their correspondent amplitudes. We can see the result in figure 1

I just took till the 7th harmonic to synthesize the final sound, because for higher harmonics the amplitudes are too small and I don't hear a real difference in the result.

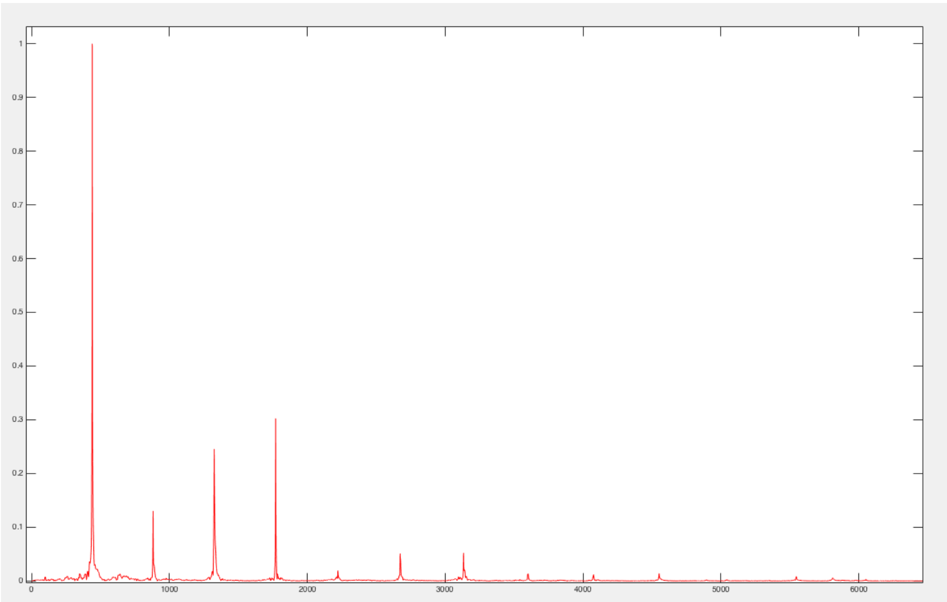

Figure 1: harmonics of the piano sound.

The second step is to determine the envelope. Since the fundamental frequency of the recording is 440 Hz, I tried to develop a band-pass filter to aisle the harmonic and to be

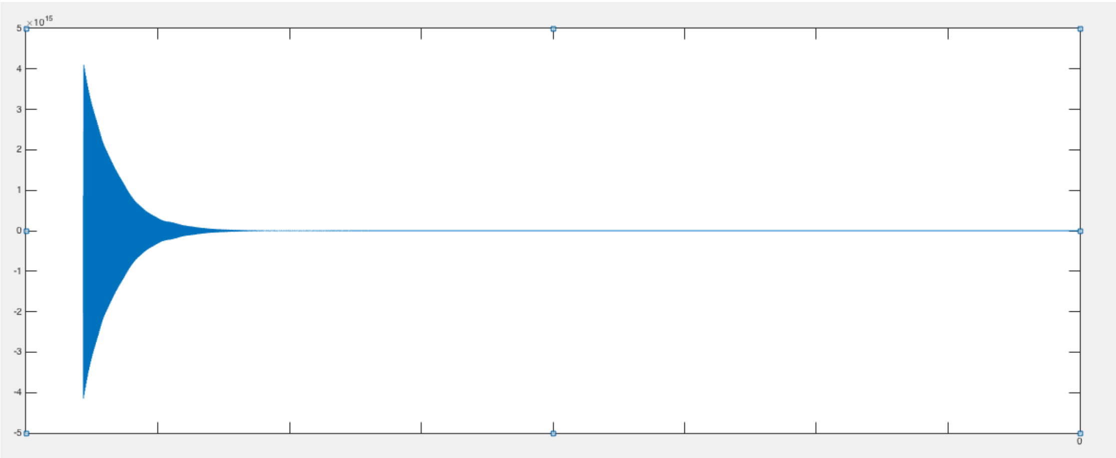

Figure 2: envelope of the first harmonic of the piano sound

able to see how it changes during time. Initially I could think first, since the piano is a percussive instrument, that the envelope shape would have an exponential function. The envelope shape is possible to be seen in figure 2.

The band-pass filter was made using first the MATLAB function *buttord* which does the computation for the Butterworth filter so we can use that result to create with the function *butter,* the B an A vectors to filter the signal with the function *filter*. We can see the code of this step of the project in the *signal_filtering.m* file in the appendix. In figure 3 it is possible to see a plot of the filter used.

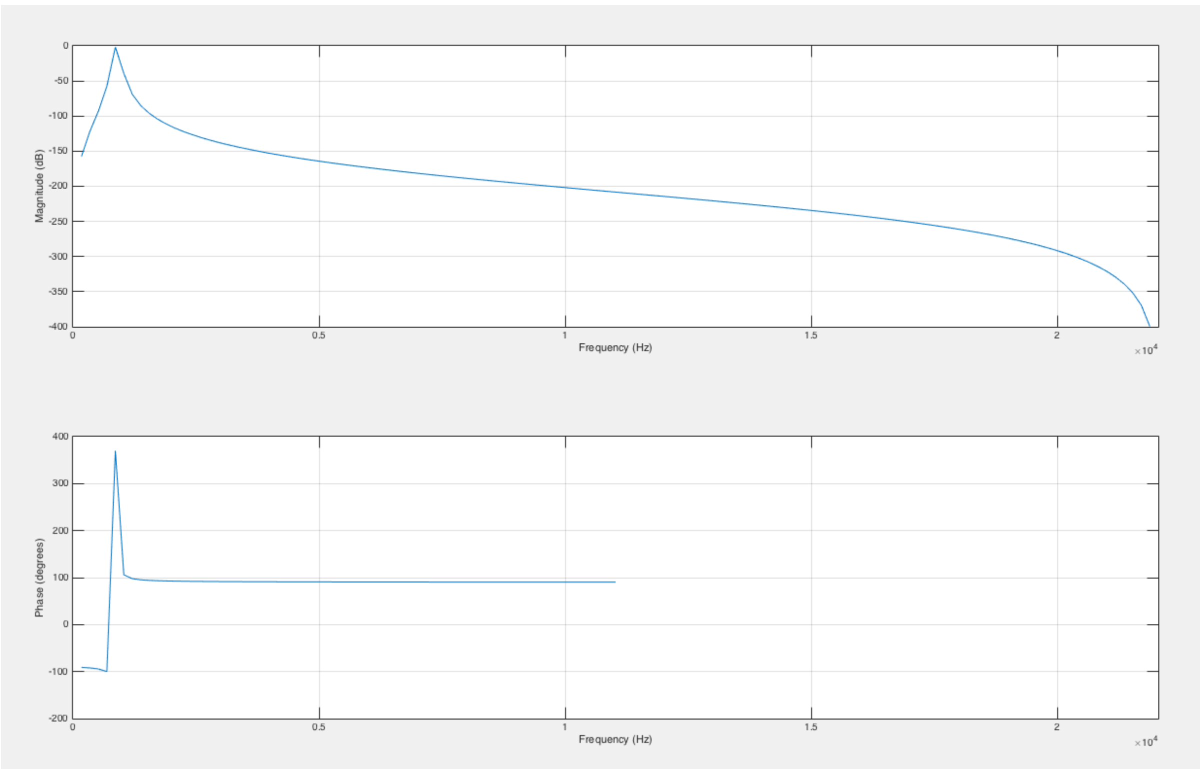

Figure 3: filter used to observe the envelope.

Synthesis

To synthesize the sound, I create a vector with the multiples of the harmonics and another one with their correspondent amplitudes and add them to create the sound multiplying the result by the exponential envelope. The decay of the exponential was finally chosen subjectively hearing the differences and picking the one that sounded the best.

I decided to add to the sound a reverb effect to try to make it less artificial, so I looked for different impulse responses of different chambers, and I finally chose one called SMALL_CHURCH. This choice, again, was made subjectively. The sound with and without the reverb effect is possible to hear it running the *test_piano.m* file.

Problems and possible solutions

The synthesized sound is not really like the recording. Maybe another step during the creation I should have done is to use different envelopes for each harmonic. That would have implied to analyze separately each one, filtering it and finding their correspondent decay.

Another problem I found is that for the octaves different from the 4^{th} one (the central one), it is possible to hear it with *two_octaves.m* file, the sound it's not so similar to a piano. I think that is because the real instrument has a different physical construction according to the different registers (like different quantity strings per note, different length of strings, with different resonance and different dampers) and in that case the tone quality it will be different.

Arbitrary Sound

The former method to synthesize this arbitrary sound is FM synthesis. This method consists in changing the pitch of a wave according to the frequency of a modulator signal. It is a simple method to create harmonic (modulator and carrier are multiples) and inharmonic sounds (modulator and carrier are not multiples and it is a way to create percussive sounds).

The equation 1 is a FM equation, where A is the peak amplitude of the signal, fc is the frequency of the carrier wave, fm is the frequency of the modulator signal and I is the modulation index that indicates by how much the modulated variable varies around its unmodulated level.

$$Signal = A * \sin(fc * t + I * \sin(fm * t)) \qquad (1)$$

In the figure 4 it is possible to observe a diagram of the basis of the frequency modulation.

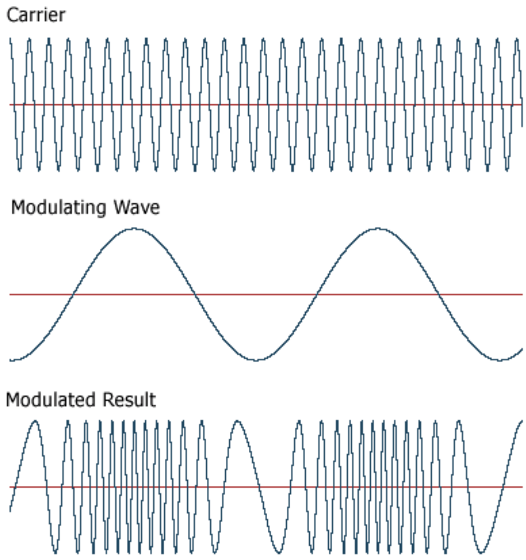

Figure 4: Diagram of FM basis.

After trying with different relationships among the carrier, the modulator, the modulation index and varying the duration, I choose to create a percussive sound, similar to a gong, and with a rounded and impure decay.

The frequency chosen of the carrier is 440 Hz, while the frequency of the modulator is 410 Hz, slightly inferior to make an inharmonic sound. The modulation index is 5.
Since again I am working with a percussive sound, the envelope chosen is an exponential, and the decay was chosen long enough to hear the changing frequency.