ELECTRICAL AND COMPUTER ENGINEERING
UNIVERSITY of MICHIGAN

Soundless Music

Michigan Engineering

EECS 452 Digital Signal Processing Design Lab
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Objective

The purpose of our project is to design and build a portable device which will transform music into a multi-sensory experience. The device analyzes incoming audio signals in real time. The frequency and beat characteristics of the signal are interpreted and used to vibrate a bank of motors in a way that best represents the incoming audio signal.

Background and Inspiration

Music is an integral part of today's society. Music and its profound effects can be seen in virtually every culture, socioeconomic class, and age group. Unfortunately, not everyone has the ability to experience music due to hearing disabilities. Our project goal was to develop a device which would enable the hearing impaired to enjoy music though the use of vibrating motors.

Device Overview

- Audio signal from external device is sampled (e.g. mp3 player)
- Digital signal processing work is completed on the Texas Instruments C5515 eZDSP Development Board
- Motor logic is calculated and output via the C5515 GPIO pins
- Current amplification circuit is used to allow GPIO pin output to enable individual motor vibration
- Motors are mounted to a forearm sleeve
- Mid-range and high frequencies are represented by two motors each, and three motors represent low frequencies (See Figure 1)

TI C5515 eZDSP Board

Vibration Motor

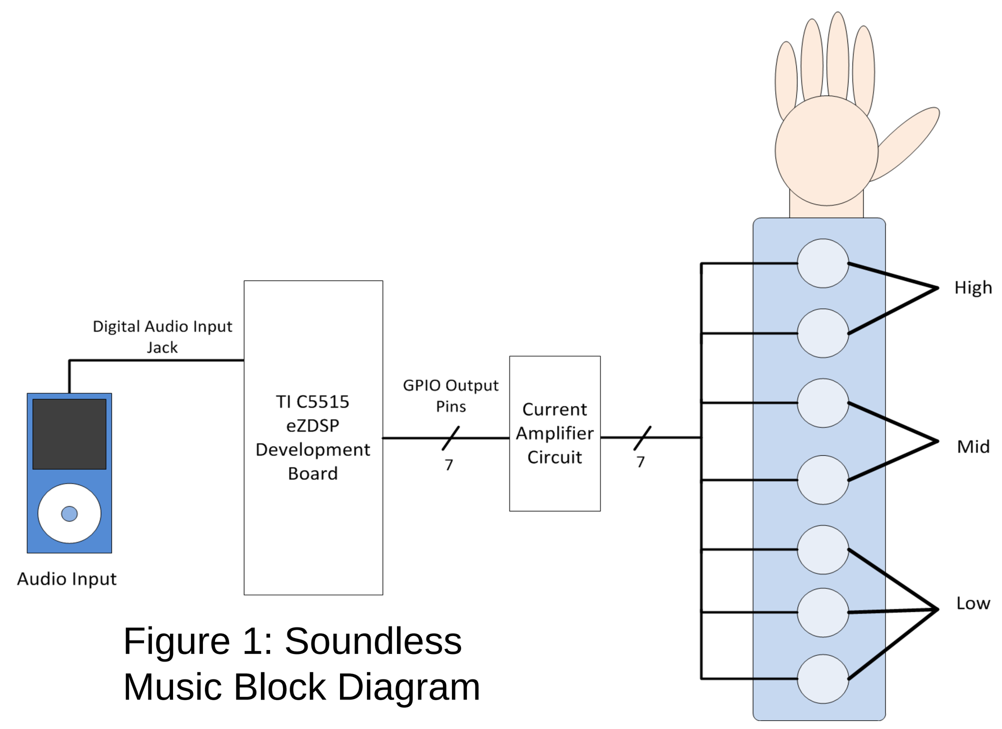


Figure 1: Soundless Music Block Diagram

Implementation: Frequency

The Soundless Music device samples audio signals acquired from the audio jack of an mp3 player. The audio signal is sampled at 9600 Hz. Analysis of the incoming audio signal occurs in real-time with the use of interrupts. A 512 point radix-2 Fast Fourier Transform (FFT) is performed when the device receives 512 points of signal data. The power spectral density (square of the FFT magnitudes) of the signal is found for three frequency bands:

- Low: 37.5 Hz to 262.5 Hz
- Mid-range: 412.5 Hz to 1275 Hz
- High: 1500 Hz to 4800 Hz

The integral of the power spectral density is calculated for each frequency band and a threshold is used to determine if there is significant frequency content of the signal within that particular band. The number of times the power spectral density threshold is crossed during the most recent half-beat is counted for each frequency band. If the threshold is crossed at least once in a particular frequency band, the first vibration motor for that band vibrates. If the threshold is crossed at least three times, the second vibration motor vibrates in addition to the first. For the low frequency band, all three vibration motors vibrate if the threshold is crossed more than five times.

Implementation: Beat

To determine the beat of an audio clip, data is sampled and broken into an array of overlapping windows. The FFT of each window is calculated to obtain the frequency content present in each window. The result of each FFT is filtered into a series of small frequency bands and the sum of magnitudes in each band is calculated. This results in a spectrogram matrix containing frequency content over time (See Figure 2).

After the spectrogram is calculated, the FFT of each horizontal frequency bin is calculated. This allows us to see any periodicity in frequency content, which is directly related to the beat in much modern music. To find the beat of the audio signal, we look for the maximum FFT value over the range of 1 to 6 Hz (See Figure 3). This is typically the range that any distinct beat will fall into. The frequency with the highest magnitude in the specified range is the beat rate of the input signal. The motors are updated two times for every one beat.

To get accurate beat information, approximately three to five seconds of data is required. Due to hardware limitations, we were forced to hardcode the beat information. In future iterations of Soundless Music, this will be computed in real-time. The beat detection algorithm was adapted from "Rhythm Detection in Recorded Music" by Joseph E. Flannick[1].

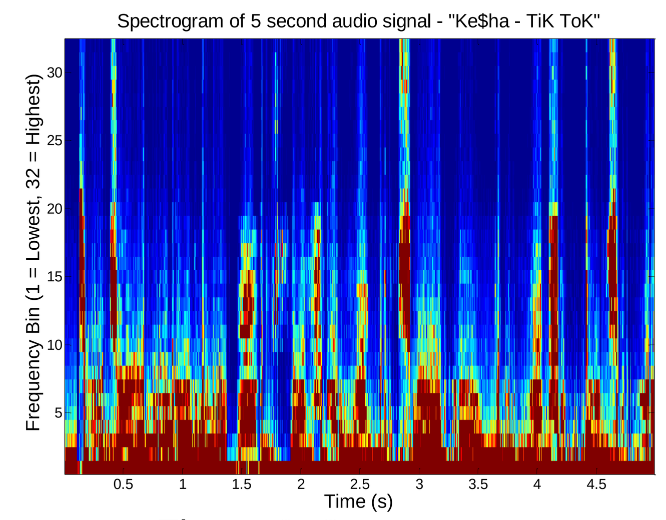


Figure 2: Spectrogram

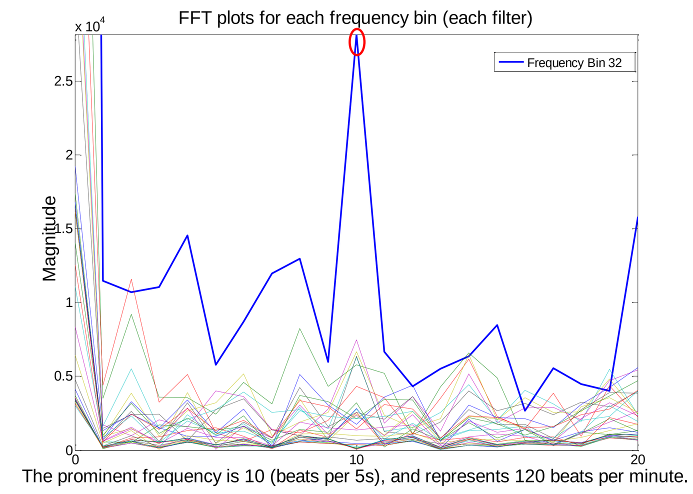
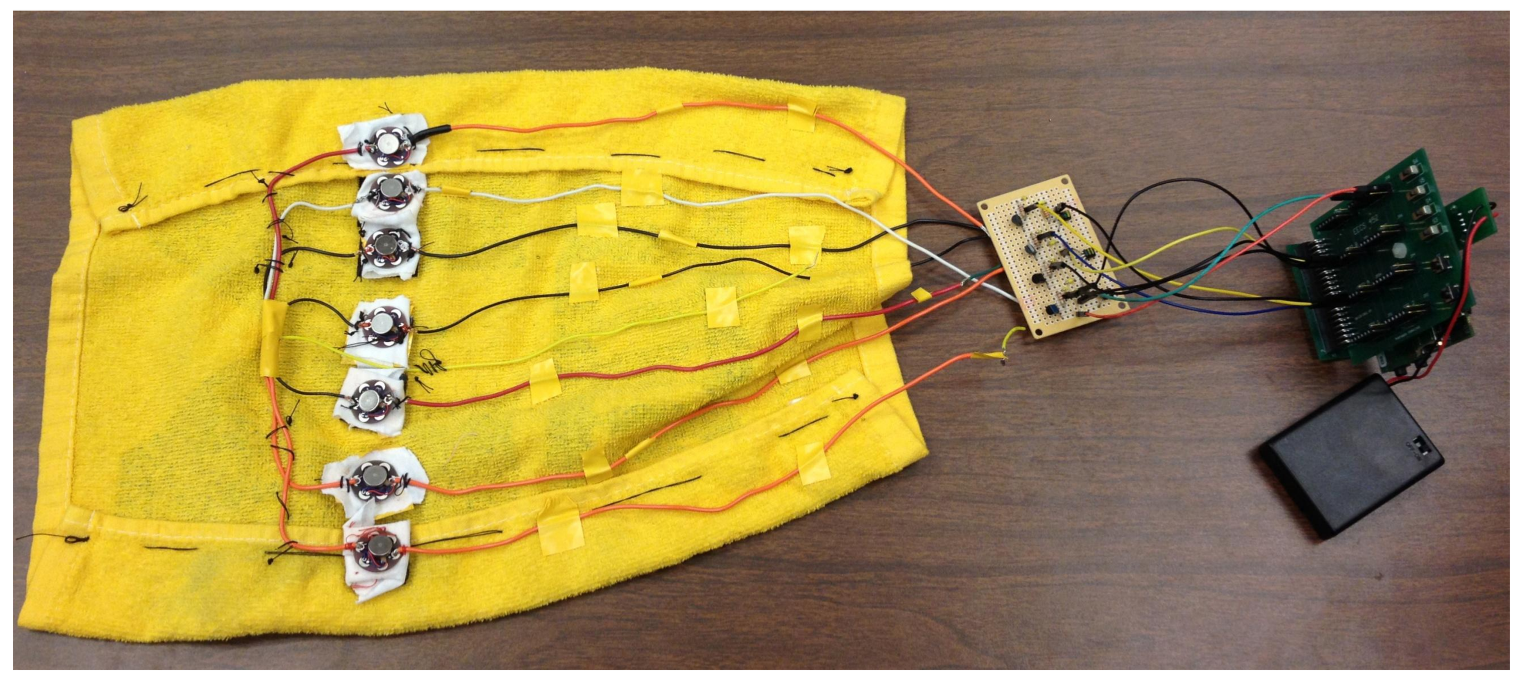


Figure 3: FFT of spectrogram over time

Prototype



Future Work

- Implement beat detection algorithm on upgraded device hardware
- Create a two-sleeve system and use two-channel audio

Acknowledgements

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[1]http://www.sju.edu/~rhall/Rhythms/joe.pdf