Project Title: Implementation of Signal Processing for Audio Equalization
Student Name: Mitch Giffin
Advisor: Dr. Stewart
Memo: Functional Description

Introduction
This project will allow the user to tailor an audio signal to a desirable sound by passing an audio signal through C++ based digital filters with variable gain. Varying the gain of each filter will allow the user to amplify or attenuate the frequencies in that range as needed.

Inputs/Outputs
The inputs to the system will be an analog audio signal and a user defined mode switch. The output will be the same audio signal adjusted to the user's preference.

Modes
There will be two separate modes available for the user to select, active or passive. In active mode the audio signal will pass through the digital filtering circuitry allowing the user to modify the signal. In passive mode the audio signal will bypass the filtering circuitry and proceed directly to the output. In this mode, the input and output are identical.
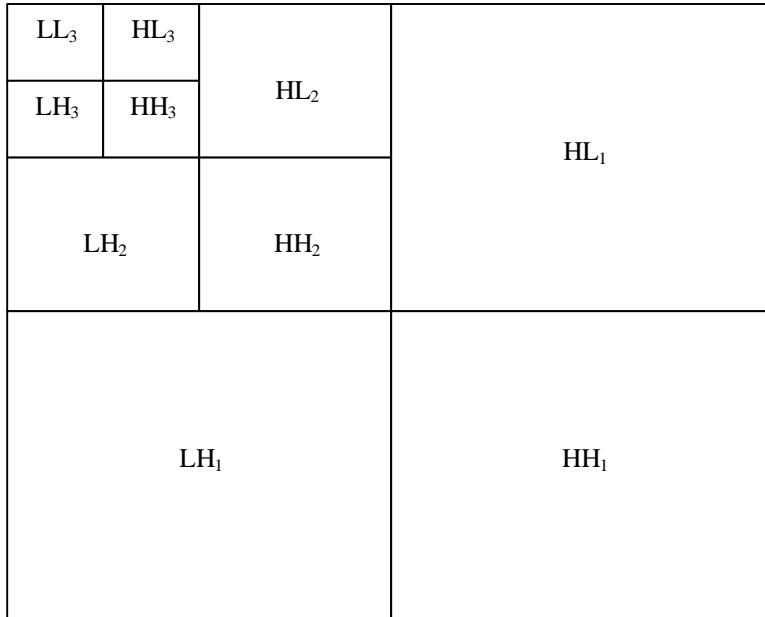
Methods
Three methods will be used and compared to attain audio equalization.

♦ The first method will consist of a bank of 15 digital filters followed by a summer. As stated above, the filters will have variable gain allowing the user to amplify or attenuate frequencies within the specified range.

♦ The second method is functionally similar to the first although equalization will be done in the frequency domain. An FFT of the input audio signal will be taken and equalization will be done by again passing the signal through 15 digital filters. Equalization will be accomplished by multiplication of specific frequencies to change the amplitude. The output of each filter will be passed to a summer and the output of the summer will have an IFFT performed on it to return the signal to the time domain.

♦ The final method will utilize lossy wavelet compression, the basis for JPEG 2000 data compression. With this method, the audio input will pass through a high and low pass filter. The frequencies at the output of these filters will be down-counted by a factor of two, which cuts the number of samples in half, and then passed through high and low pass filters identical to the first pair. Again the filter outputs will be down-counted by two and the process continues until the desired amount of filters is reached. For this project 16 filters with variable gain will be used representing 16 frequency ranges. The data compression comes in to effect during the down-counting and also when frequency amplitudes fall below a specified threshold value and will thus be deemed inaudible. Recombination of the signal is done using the same

method in reverse. Figure 1 shows an example of subband labeling and how the signal will be broken down.

Figure 1

LL_3	HL_3		

(subband labeling diagram: LL_3, HL_3, LH_3, HH_3, HL_2, LH_2, HH_2, HL_1, LH_1, HH_1)

Basic Block Diagram

Passive

| Analog Audio Signal | | Digital Filters with Variable Gain | | Transformed Audio Signal |

Active

Mode Switch