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ASSIGNMENT REPORT

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Contents

1 Introduction

1.1 Background

Equalization is the process of adjusting the balance between frequency components within an electronic signal. An audio equalizer is an audio device with multiple frequency controls for adjusting sound tone quality. To be specific, an equalizer can be used to compensate for deficiencies in a sound pickup or to reduce extraneous sounds, such as noise. Besides, it can also be used to improve instrument clarity. For example, boosting a sounds harmonics gives the impression of more presence and brightness and decresasing a sounds harmonics gives the impression of a dull, less dazzling sound.

1.2 Motivation

The audio output quality of loudspeakers may not be uniform due to size, mechanical and cost constraints, even though they are usually designed to have a fairly uniform response across the frequency spectrum. Therefore, audio equalizer is performed to flatten this response, or to shape it according to listener preferences.

In the early years, analog filters were mainly used to implement the audio equalizer. However, the proliferation of digital audio sources, such as the Internet and the USB, gives an opportunity to apply digital signal processing to audio signals. Actually, many advantages, which include simplification of design and verification, greater flexibility and reliability, and significantly superior sound quality, allow the creation of digital equlizer that can outperform its analog counterparts.

1.3 Objective

The objective of this project is to design and simulate an digital audio equalizer using Matlab. It includes both implementation and evaluation of the audio equalizer.

1.4 Report Outline

This report mainly constists of two parts: the implementation and the performance analysis of an audio equalizer. In the implementation, the procedure of creating a Graphical User Interface (GUI) by using Matlab's App Designer and the construction of a FIR filter by using Matlab's DSP toolbox is included. Meanwhile, the performance analysis focuses on displaying the equalizer simulation result and elaborating the difference among outcomes when different filter algorithms and configurations are applied.

2 Design And System Construction

The entire audio equalizer consists of 3 major blocks: audio input, digital audio equalizer and audio output. The audio input is implemented through Matlab's internal audioread() function. The audio output, which includes the DAC and reconstruction filter, is built on Matlab's sound() function.

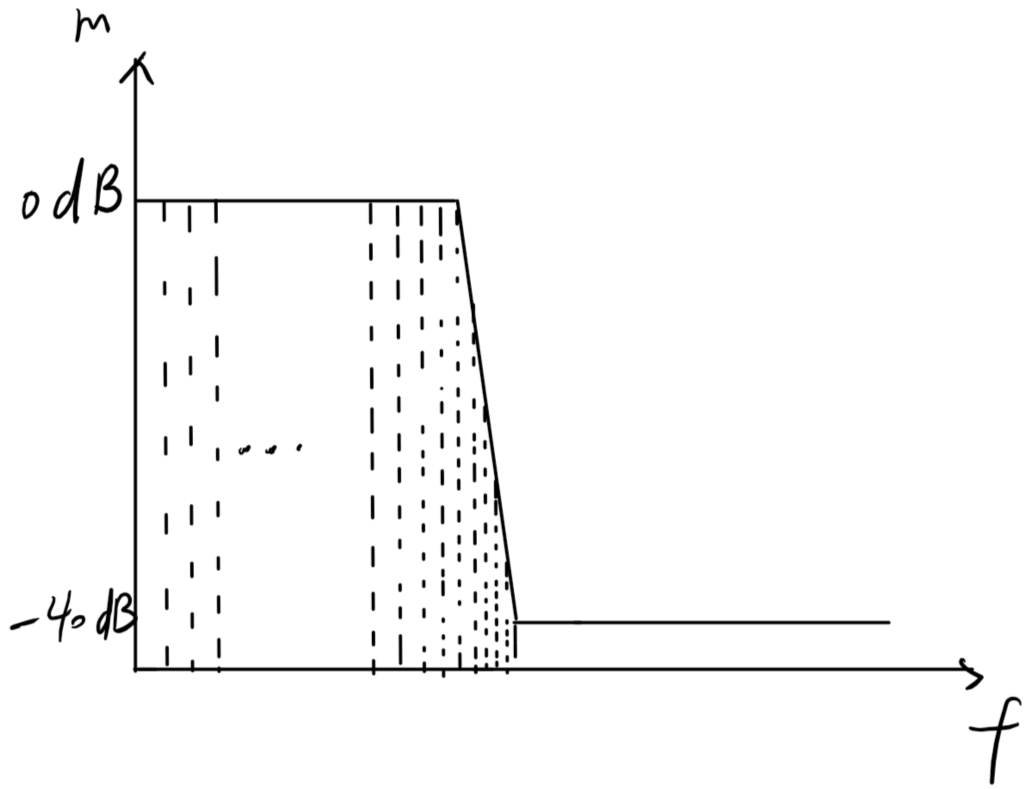
The GUI is designed by using Matlab's App Designer. As shown in the figure above, each frequency band in the equalizer is controlled by a slider. The 'Apply' button collects the values set by sliders and pass to fir2() function to generate the filters' coefficients b. The 5 'plot' buttons plot the figure of the different response of the filter.

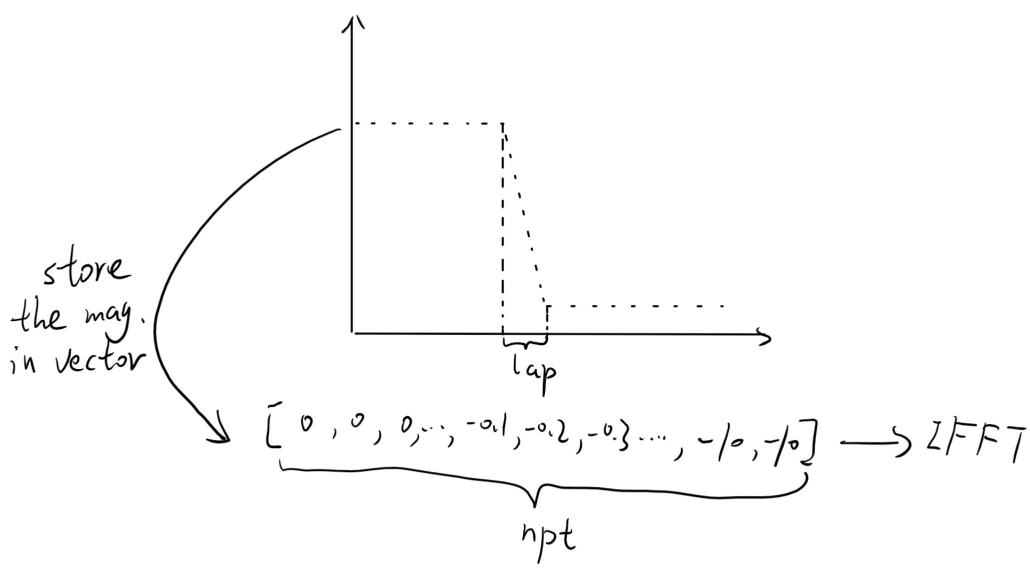
3 Algorithm

The FIR filter function used in this equalizer is b=fir2(n,f,m,npt,lap,wind), where n, f, m, npt, lab and wind represent filter order, frequency interval, magnitude corresponding to each frequency interval, number of grid points, length of region around duplicate frequency and window function.

fir2() uses frequency sampling method to design FIR filter. The function interpolates the desired frequency response linearly onto a dense, evenly spaced grid of length npt, which is default set to 512. fir2 also sets up regions of lap points around repeated values of f to provide steep but smooth transitions. To obtain the filter coefficients, the function applies an inverse fast Fourier transform to the grid and multiplies by window.

For example, to obtain the coefficients of a low pass filter like the figure below, the function samples the desired frequency response of the filter and then maps the sampling points to a grid. The number of sampling points depends on the value of npt. lap, as shown in the second figure below, decides the width between band edges. Less the value of lab, steeper the transitions from edge to edge. The sampling points are stored in a vector and sent to inverse fast Fourier transform function to get the filter coefficients b.



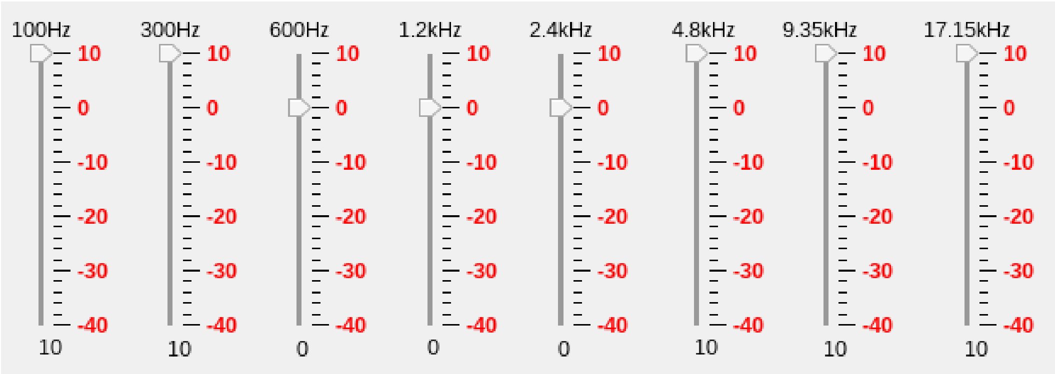


4 Results And Analysis

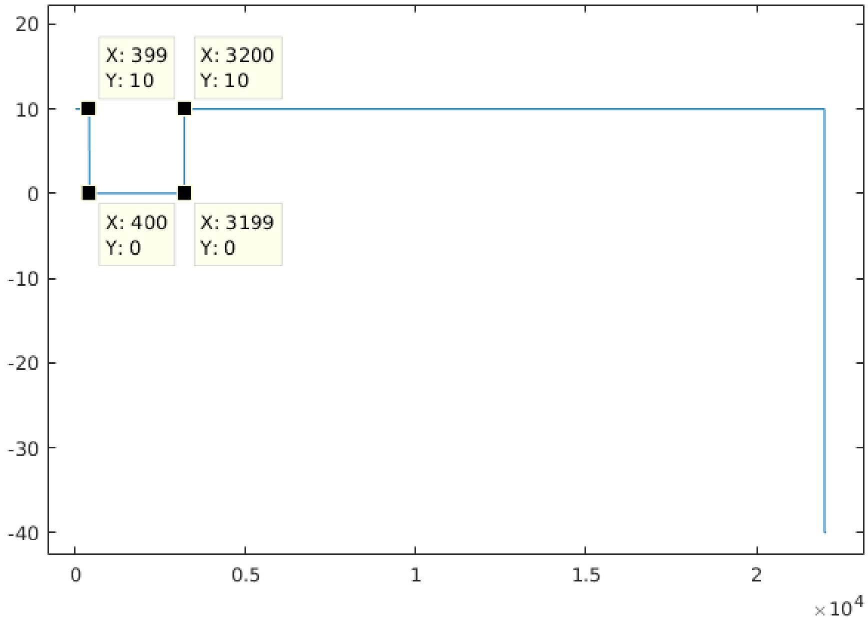
4.1 Results

Simulation setting:

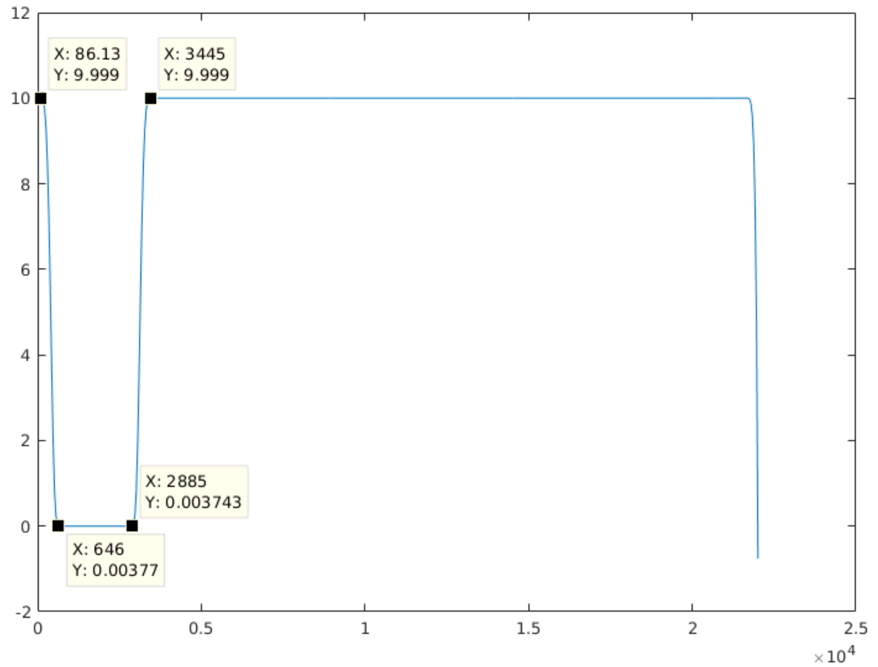
a) Signal components below 500 Hz and above 4000 Hz being enhanced.



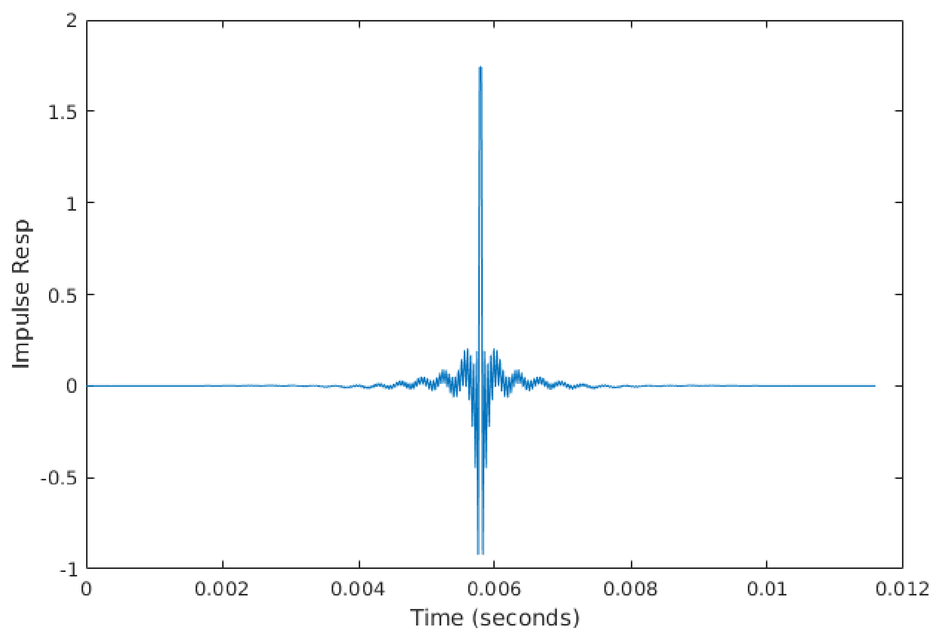
Ideal frequency response:



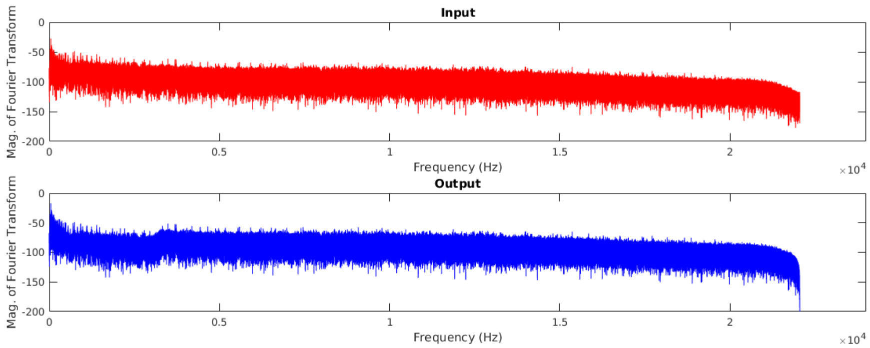
Actual frequency response:



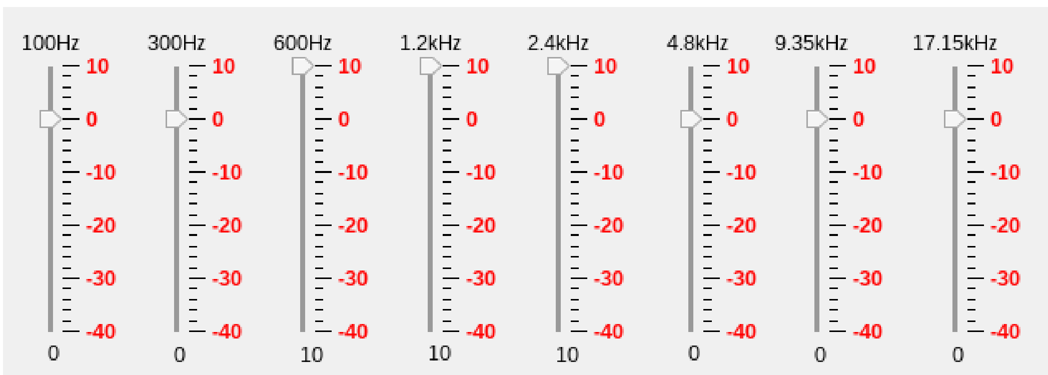
Impulse response:



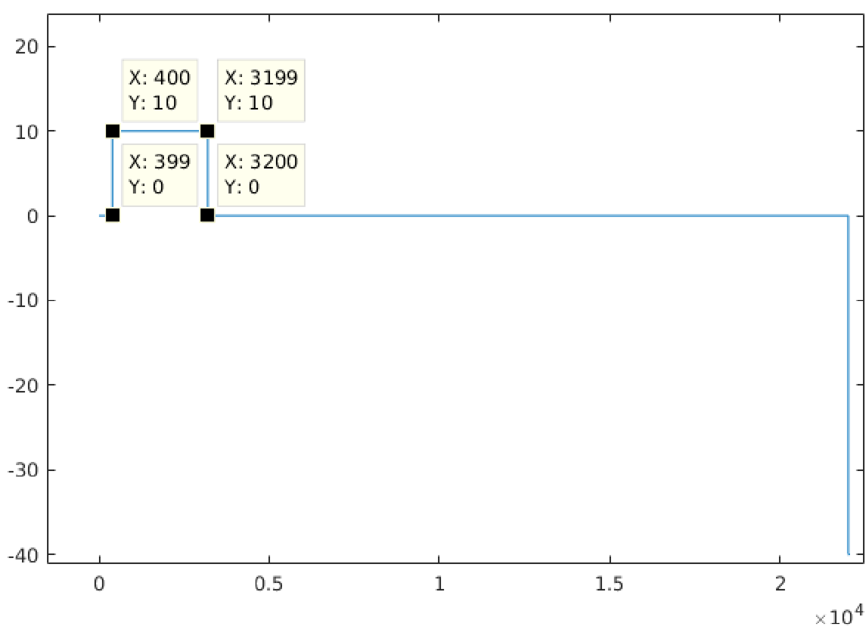
Frequency spectra:



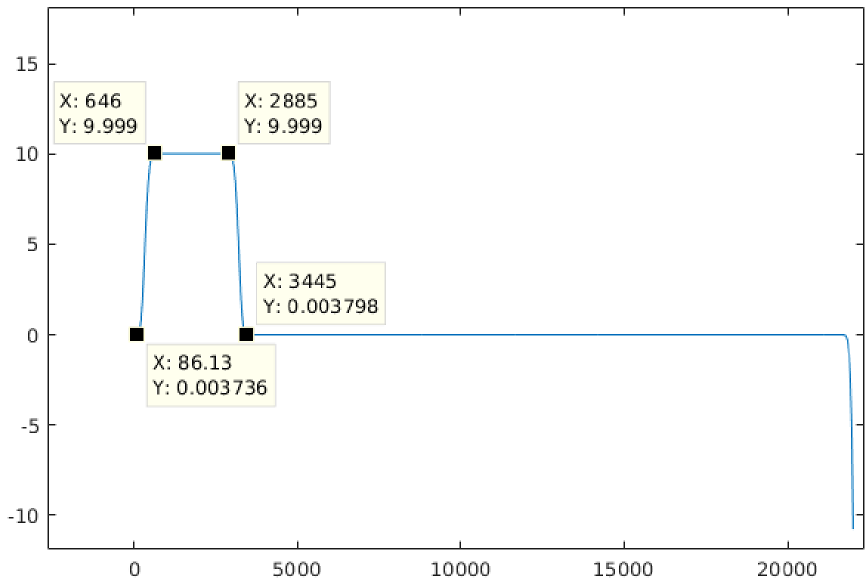
b) Signal components between 500 Hz and 4000 Hz being enhanced.



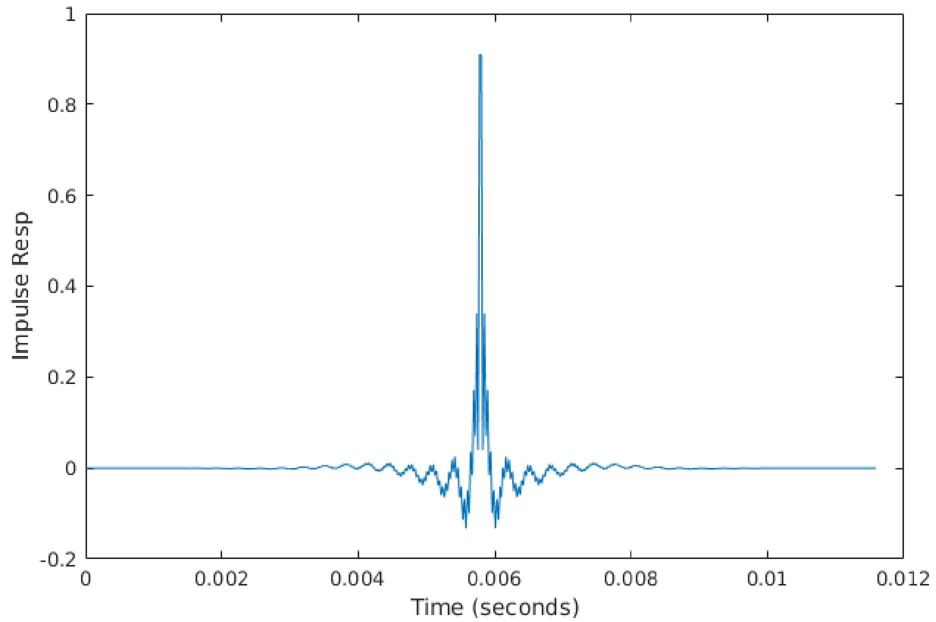
Ideal frequency response:



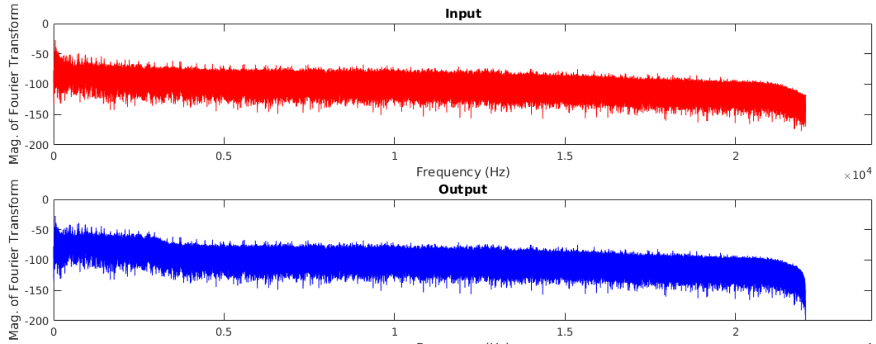
Actual frequency response:



Impulse response:



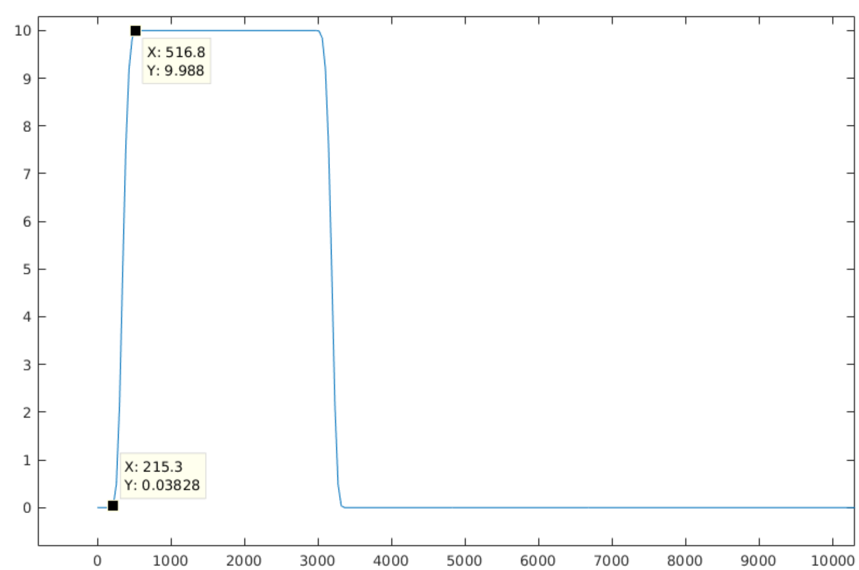
Frequency spectra:



4.2 Analysis

The figures above shows two different configurations of the equalizer. By setting the gain of band 1,2,6,7,8 to +10dB, signal components below 400 Hz and above 3200 are enhanced. On the second configuration, signal components between 400 Hz and 3200 Hz are enhanced.

As shown in the figure above, the transitions of an ideal filter frequency response is steep. For actual frequency response, the transitions between different band edges is more smooth. The transition band for setting a is about 560 Hz and this is the main factor which limits the filter's performance on low frequency band. For the bandwidth of low frequency band is narrow, which is 200 Hz typically, a wide transition band can cause significant influence on these section. One way to improve the performance is to increase the filter order. The figure on below shows the frequency response with 800 filter order. the transition bandwidth now decrease to around 300 Hz.



The obtained spectrum shows the result of the filtered signal. The shape of the output spectrum corresponds to the shape of the FIR filter's frequency response.

5 Conclusion And Recommendations

In summary, the FIR filter basically meets the specification of an audio equalizer. The result shows that the equalizer performs better on higher frequency band compared to lower frequency band. Generally, the frequency sampling method is a simple way to implement FIR filter. However, this method requires longer filter length to achieve a more precise solution. In comparison to other design method, such as window design method, least squared method and the Parks-McClellan method, this method can have overall more error. Thus, it is recommended to implement the digital equalizer by using other method for a better solution on low frequency band.

6 References

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https://www.mathworks.com/help/signal/ref/fir2.html

1. Wikipedia: Window Function

https://en.wikipedia.org/wiki/Window_function

1. StackExchange: Difference between frequency sampling and windowing method

https://dsp.stackexchange.com/questions/31905/ difference-between-frequency-sampling-and-windowing-method