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CSE 3313-001 Fall 2022 Term Project

Project Report

Instructor: Dr. Jon Mitchell

Course Name: Discrete Signal Processing

Due Date: 12/12/2022

"I __Temitayo Aderounmu__ did not give or receive any assistance on this project, and the report submitted is wholly my own."

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CSE 3313 Term Project: Introduction

INTRODUCTION

Project Aim:

The term project involves the design and implementation task that is based on the knowledge we have gained throughout the semester in the lecture, projects and homework. The main task is for us to incorporate a Butterworth lowpass filter that will be designed using MATLAB. The goal for the project is to design a filter to remove unwanted noise from an audio file.

Project Overview:

Lowpass, high pass, and bandpass filters are just a few of the various filter types that can be employed to remove noise from an audio sample. Lowpass filters can be useful for cutting down on background noise since they let low-frequency signals pass through while attenuating (decreasing the amplitude of) higher-frequency signals. Contrarily, high pass filters attenuate lower-frequency frequencies while allowing high-frequency signals to pass through. Bandpass filters attenuate signals outside of this range while allowing a specific range of frequencies to pass through. A form of electronic filter called a Butterworth lowpass filter is made to pass low-frequency signals while blocking high-frequency ones. It bears the name Stephen Butterworth after the British physicist and engineer who initially introduced the filter in his 1930 paper, "On the Theory of Filter Amplifiers.".

Project Status:

The Project was completed on Monday, the 12th of December 2022. The design met all specifications, with the output audio being played in the background while the code is running.

SYSTEM DESIGN:

Audio Frequency Analysis

The project includes an audio sample that consists of a conversation in the presence of noise. The noise is thunder and a high-frequency tone.

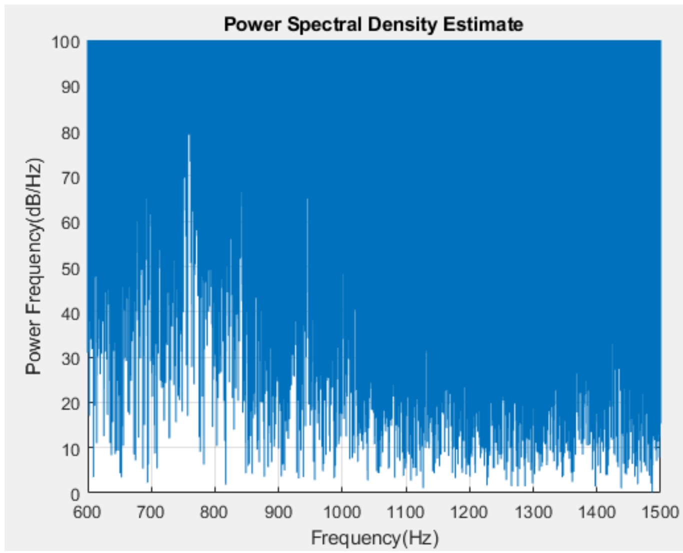

Figure 1. Audio Frequency Analysis. The Discrete Fourier Transformation [DFT] technique was implemented to determine the wanted and unwanted frequency ranges for the audio sample.

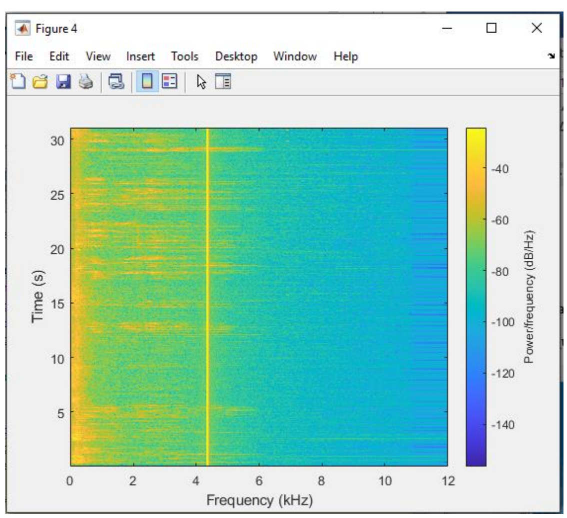

Figure 2. Audio Frequency Analysis. The spectrogram technique was implemented to determine the wanted and unwanted frequency ranges for the audio sample.

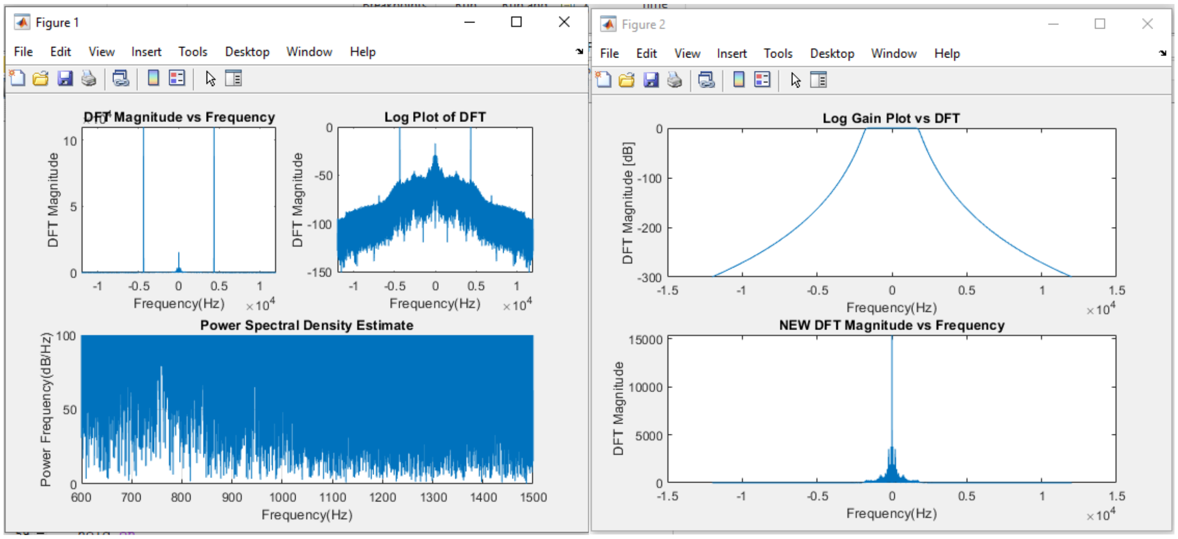

Figure 2. Plots. Subplots using MATLAB.

Filter Design

I used MATLAB filter design commands to design a digital filter that will reduce the noise from the inputted audio sample.

```
%% Filter Design
%Using MATLAB filter design commands to design digital filter that will
%reduce noise from the audio sample.

wp = 1700;
ws = 2300;
Dp = 1;
Ds = 40;
Atten = 0.5;

%Find N
vlu = (10.^(Ds/10)-1)/(10.^(Dp/10)-1);
N = (log10(vlu))/(2*log10(ws/wp));
Nnxt = round(double(N))+1;
FC1 = wp/(10.^(Dp/10)-1)^(1/(2*Nnxt));
FC2 = ws/(10.^(Ds/10)-1)^(1/(2*Nnxt));

%Find H(s)
Hs = 20*log10(sqrt(1./(1+(faxis/FC1).^(2*Nnxt))));
```

Filter Implementation

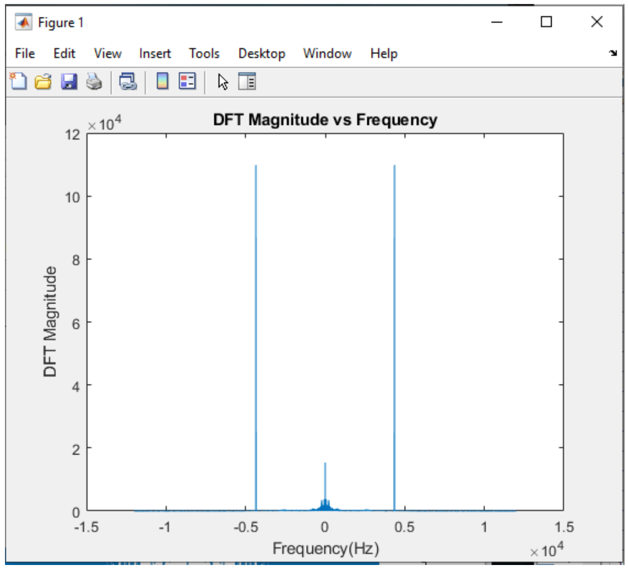

Figure 3. Plot of the Discrete Fourier Transformation [DFT] before filtering. In the plot the zero frequency is at the center of the plot and the horizontal axis illustrates frequency in Hz.

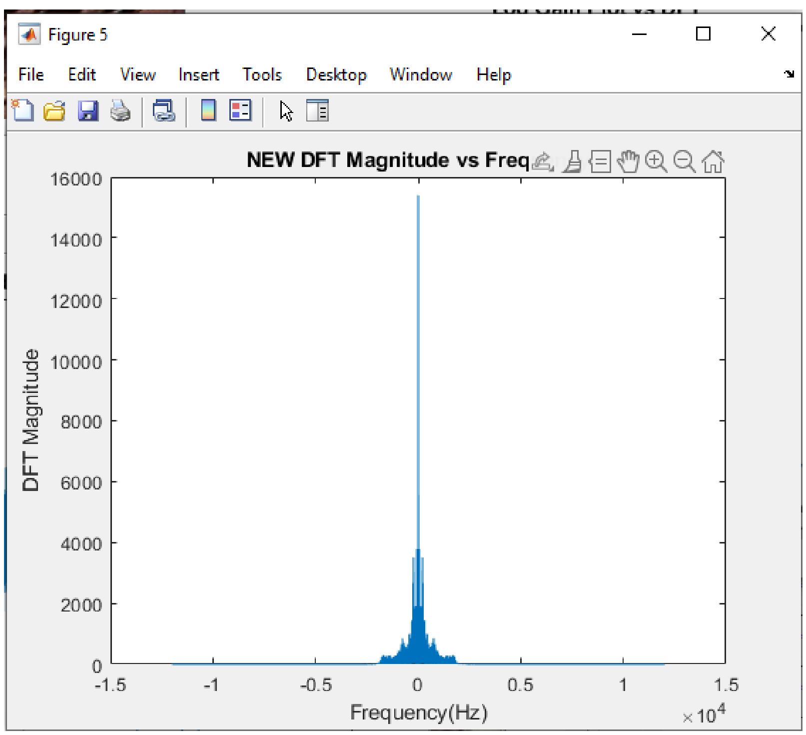

Figure 4. Plot of the Discrete Fourier Transformation [DFT] after filtering. In the plot the zero frequency is at the center of the plot and the horizontal axis illustrates frequency in Hz.

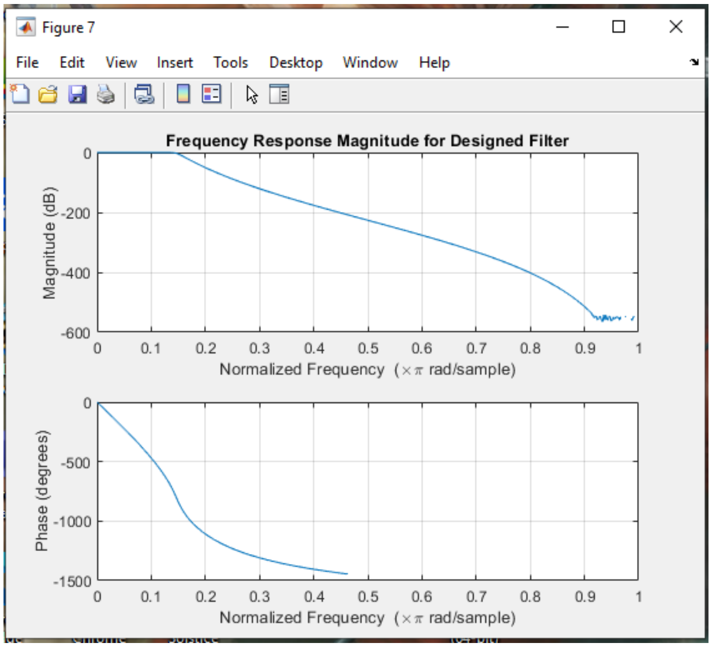

Figure 5. Plot of the magnitude of the frequency response of the designed filter. In the plot the horizontal axis illustrates digital frequency in rad/s.

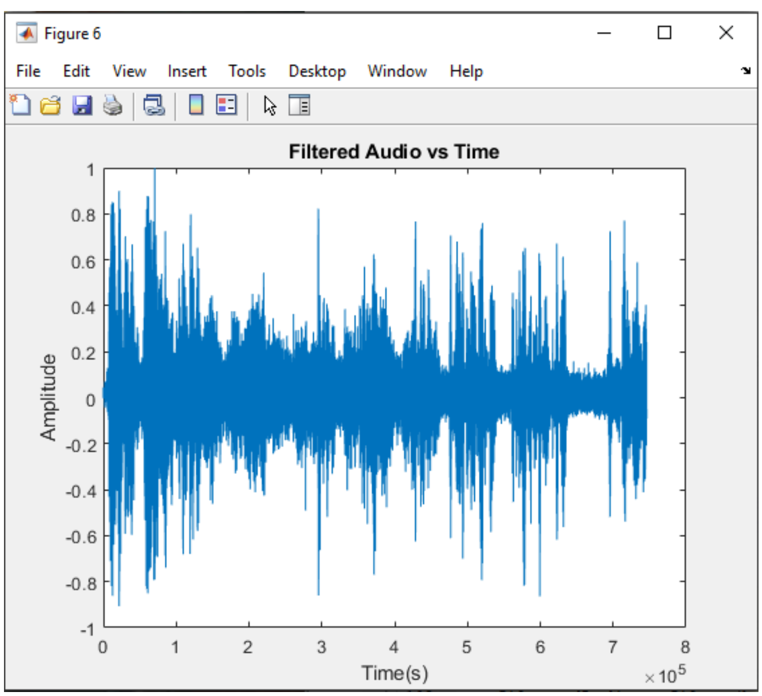

Figure 6. Plot of the audio signal that has been filtered in the time domain.