Signals and Systems

(EE-231)

DE-43 Mechatronics

Syndicate – A

Scientific Report – Assignment 3

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Title: Analysis of Filters and Noise Signals

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ABSTRACT:

Noise reduction and filter designing are important techniques for processing audio signals to improve their quality and clarity. Noise reduction involves identifying and removing unwanted sounds from an audio signal, such as background noise or interference. Filter designing involves creating filters that can be applied to an audio signal to shape its frequency response and modify its characteristics. These techniques are widely used in a variety of applications, including audio engineering, telecommunications, and speech recognition. In this report, we will explore the principles and techniques of noise reduction and filter designing and provide examples of how they are used in practice.

INTRODUCTION:

Audio signals play an important role in a wide range of applications, including entertainment, communication, and information processing. However, the quality of an audio signal can be degraded by various factors, such as background noise, interference, and distortions. In order to improve the quality and clarity of an audio signal, it is often necessary to apply various processing techniques, such as noise reduction and filter designing.

Noise reduction is a technique that involves identifying and removing unwanted sounds from an audio signal. This can be achieved through a variety of methods, such as spectral subtraction, Wiener filtering, and Kalman filtering. Noise reduction is important for improving the intelligibility and clarity of an audio signal and is widely used in applications such as speech recognition, audio engineering, and telecommunications.

Filter designing, on the other hand, involves creating filters that can be applied to an audio signal to shape its frequency response and modify its characteristics. Filters are used to remove or attenuate certain frequencies from an audio signal, or to enhance certain frequencies to improve its overall quality. Filter design is an important aspect of audio processing, and is used in a variety of applications, including audio engineering, telecommunications, and speech recognition.

In this report we will remove unwanted frequencies from a sound signal and analyse the effect of filtering bass and treble frequencies from the signal both graphically and experimentally.

Materials and Methods:

MATERIALS USED:

- Software: MATLAB ver. R2022b
- Audio file: (Sounds and Music Library, 2017)
- File format: .wav format

- File size: 1.69mb
- Audio duration: 1:11 minutes.

METHODOLOGY:

The following Commands were used in MATLAB for the manipulation of signals:

FUNCTION DETAILS:

1. **Audioread:**

   Audioread (filename) reads data from the file named filename, and returns sampled data, y, and a sample rate for that data, Fs.

   Syntax:

   [y, Fs] = audioread (filename, samples)

   Here 'filename' is the name and address of the file and 'samples' is the numbers of samples of the function.

2. **Sound:**

   Sound (y) sends audio signal y to the speaker at the default sample rate of 8192 Hz.

   Syntax:

   sound(y)

   Here y is the function that is to be played.

   audiowrite (filename, y, Fs)

   Here 'filename' is the name and address of the file, 'y' is the matrix and 'Fs' is the sample rate.

ADD-ONS USED:

- Signal Analyzer
- Filter Designer

PROCEDURE:

1) Create a new folder called Assignment.
2) Copy the audio file into that folder, similarly, create a new file in MATLAB and save it in that folder.

3) In **MATLAB editor,** start writing your code.
4) Read the audio file using **audioread** command.
5) Go to **Apps tab** and click on **FilterDesigner** and design a filter for basspass and treblepass.
6) Create a rect filter that only passes frequencies less than 2000. Apply this filter to the signal and plot the response in the time and frequency domain.
7) From the **APPS** tab click on **Filter Design**

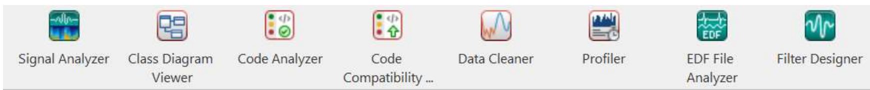

Figure 1 APPS tab

8) Apply the following settings for each filter:

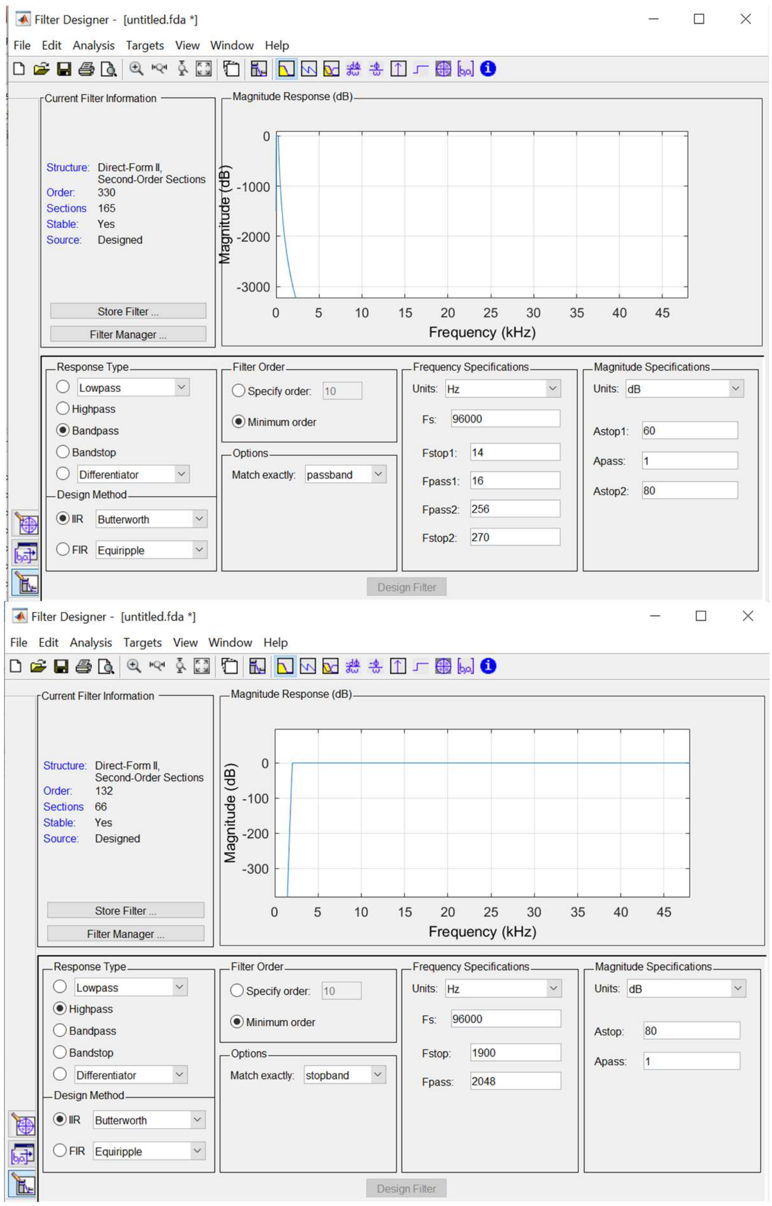

Figure 2. Filter settings

9) Click on **Run.**
10) Go to **Apps tab** and click on Signal Analyzer.
11) Analyze the signals and observe the results.
12) Using **sound** command listen to the signals before and after being filtered.
13) Write down your observations.

RESULTS:

The following results were obtained from our analysis:

Filtration of frequencies higher than 270Hz:

The following graph was obtained:

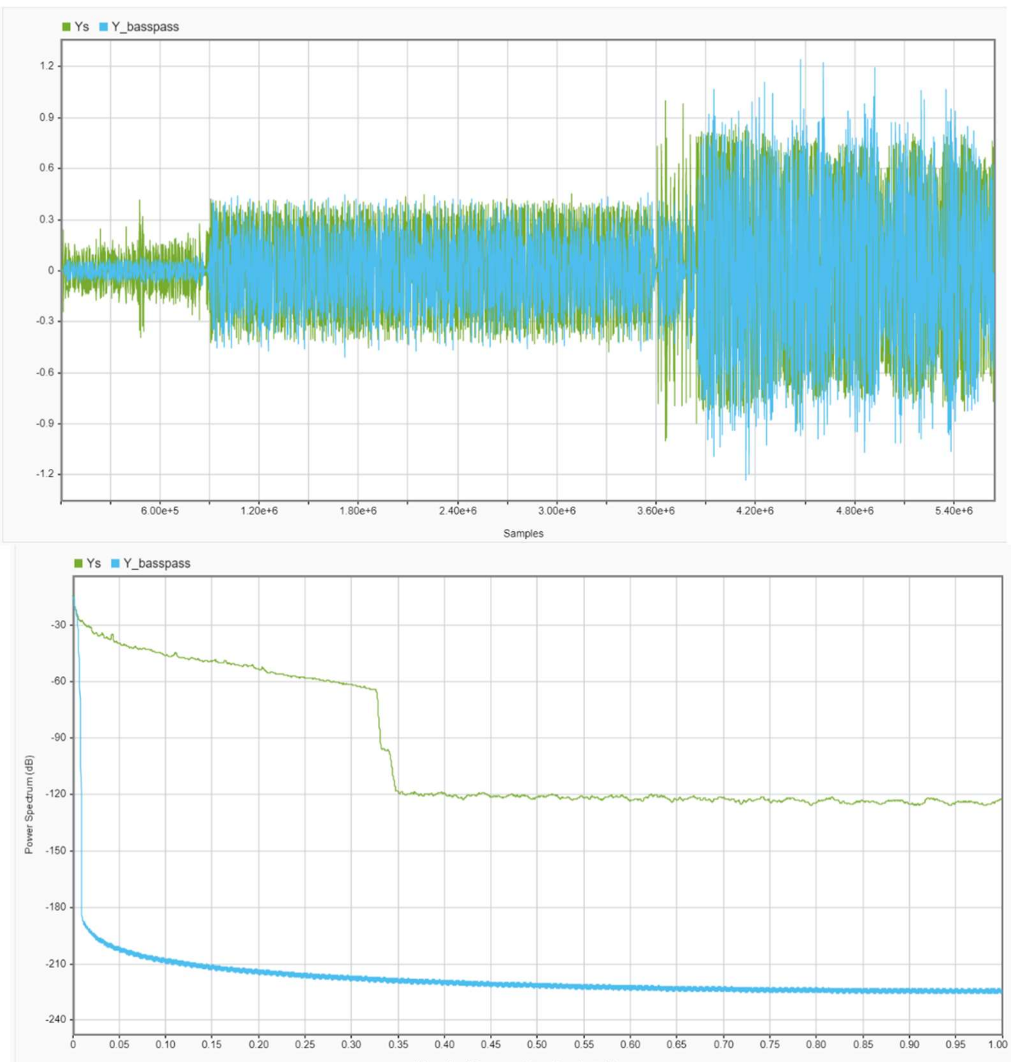

Figure 3. Signal analysis in Time and Frequency domain.

Graphical analysis:

The top graph shows the filtration of higher frequencies from our signal in time domain, where as the bottom graph shows the original (y0) and filtered signal (Y_low) in frequency spectrum. The graphs show what happens to the signal after it is passed through a **Rect** filter. It clearly shows that after filtration the high pitch frequencies of the filtered signal (Y_low) have been filtered out passing only lower frequencies.

Physical Analysis:

Practically when listening to the signal it can be observed that only the bass and lower pitched frequencies were audible. Only the background noise could be heard and the sound overall was faint.

Filtration of noise frequencies:

The following graph was obtained:

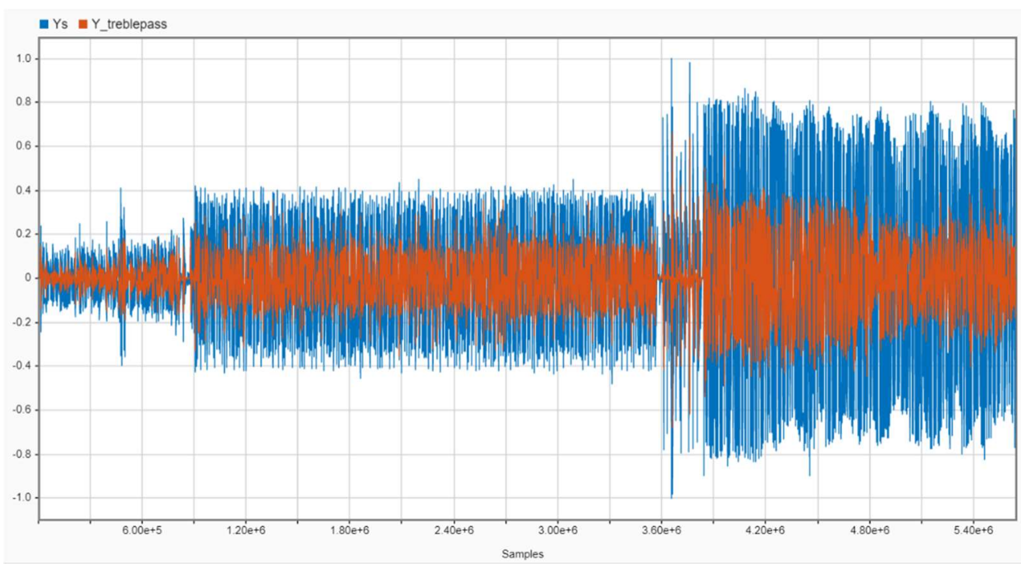

Figure 4.1 Lower frequencies filtered out (t domain)

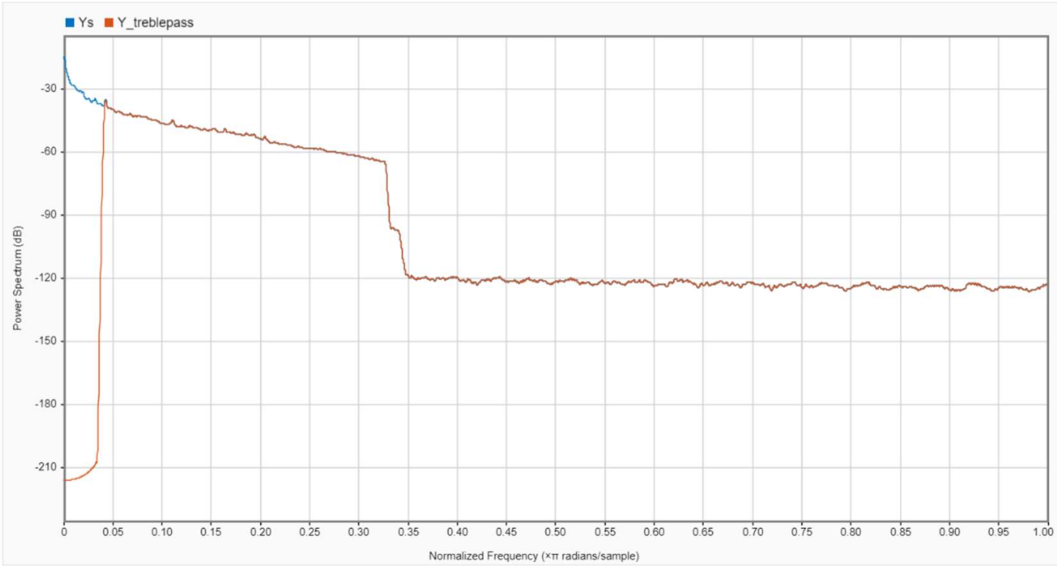

Figure 4.2 Lower frequencies filtered out (f domain)

Graphical Analysis:

The graph shows the difference between original and filtered signal. Lower frequencies being filtered out of the signal, while there is no effect on the rest of the signal.

Physical Analysis:

Practically when listening to the signal, the sound quality of the signal becomes better as most of the noise is filtered out.

DISCUSSION:

The signal was filtered, and the results were analysed both graphically and experimentally. Difficulties were faced when trying to resample the signal as sometimes the original signal got overwritten causing errors in measurements. The signals were manipulated in various methods and various outcomes were observed and recorded. Filtering a signal can improve the sound quality removing unwanted noise from the signal.

There are a few limitations to the study that should be considered when interpreting the results. For example, the filtered signals were observed with respect to a small number of people and the results were given based on their observations. A better result could have been obtained if a large number of people were to observe the results and provide their observations. Despite these limitations, the study provides valuable insights into the effectiveness of these .

CONCLUSION:

In conclusion, the study demonstrated the effectiveness of two filters designed to allow only specific frequency ranges to pass through. The first filter, which was designed to allow only frequencies of 16-256Hz to pass through, was found to be effective at blocking lower frequencies and allowing higher frequencies to pass through. The second filter, which was designed to allow only frequencies above 2000 to pass through, was found to be effective at blocking frequencies below 2000Hz and allowing higher frequencies to pass through.

Overall, the results of the study suggest that the filters were successful at achieving their intended purpose, and could potentially be used in a variety of applications where frequency selective filtering is required.

REFERENCES:

Sounds and Music Library. (2017, October 29). FUNNY and COMEDY SOUND EFFECTS I [Video]. YouTube. https://www.youtube.com/watch?v=kOP2tGkWU8U