SIGNALS AND SYSTEMS

DEPARTMENT OF MECHATRONICS ENGINEERING-41-B

PROJECT REPORT

Audio Signal Processing (Noise Reduction)

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Submitted to:

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1. Project Title:

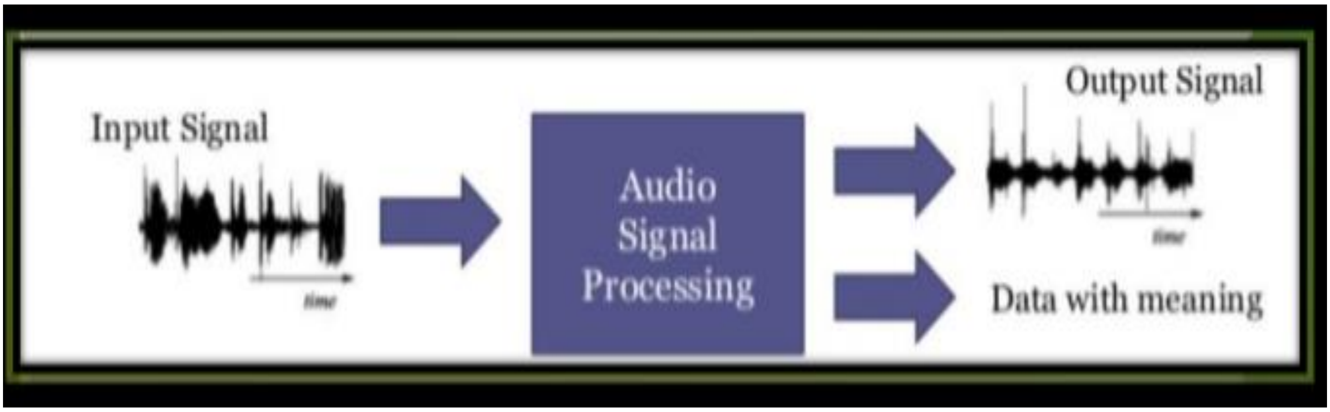
Audio Signal Processing (Noise Reduction)

2. Problem:

In this project our main objective is to observe the digital signal processing steps. At first creating the time domain signal of given audio signal (having some noise) using MATLAB command and then convert the time domain signal into frequency domain signal. Then we must filter this noisy signal and observed which filter give us our desired output by comparing with original time domain signal.

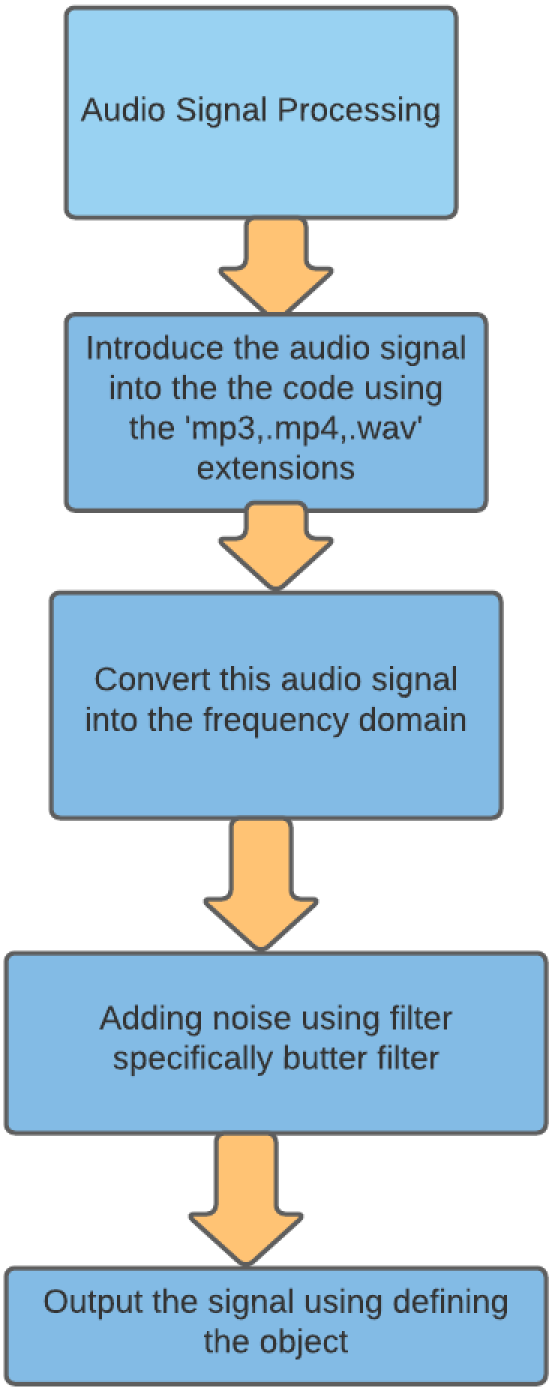
3. Working:

In this we have done audio signal processing specifically noise reduction in an audio Signal. We first introduce an audio signal having extension .mp3, .wav etc. Then we apply Fourier transform. The Fourier transform will convert our time domain given signal into frequency domain audio signal. Now we add noise using filter such as, as in is this project we introduce beep sound as f1.wav and f2.wav extensions. And at the end we play original signal and output signal and, hence completing our project.



<u>Figure. I</u> Demonstration of audio signal processing.

4.Flow Diagram



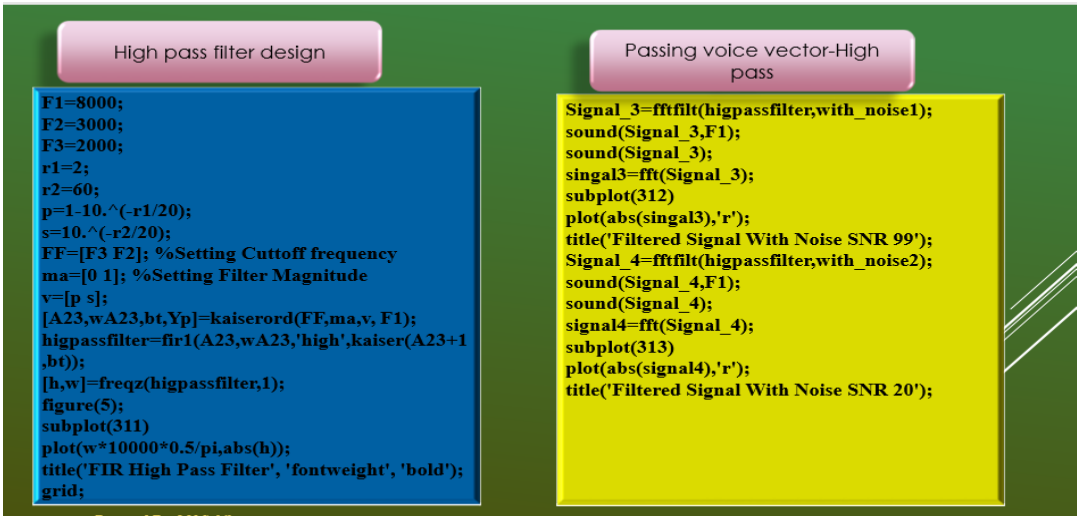


Figure.V Demonstration of high pass filter design and passing voice vector high pass design.

Figure.VII Demonstration of band pass filter design and passing voice vector band pass.

5. Some Important terms

I. SIGNAL:

Anything that convey information like speech, Image, Video, etc.

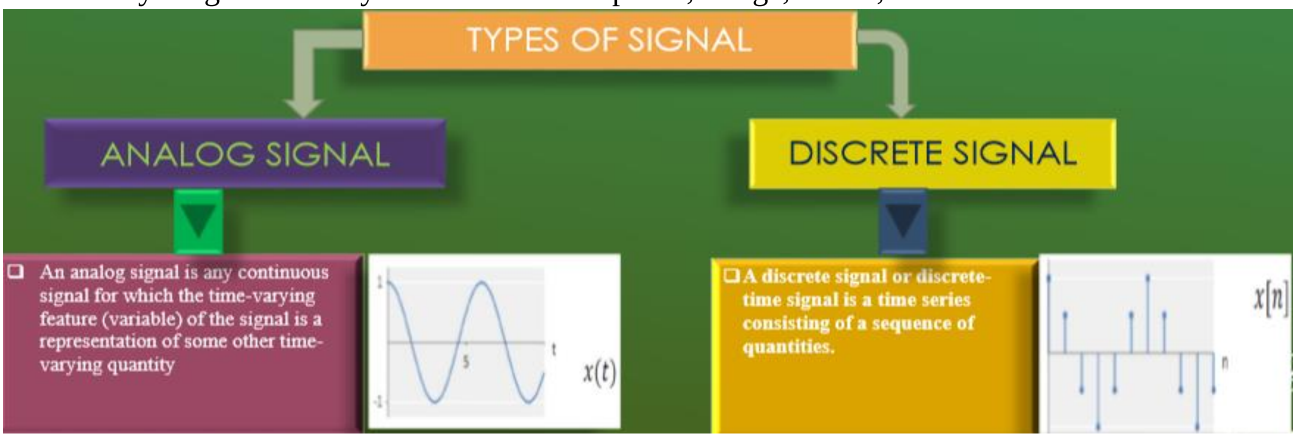
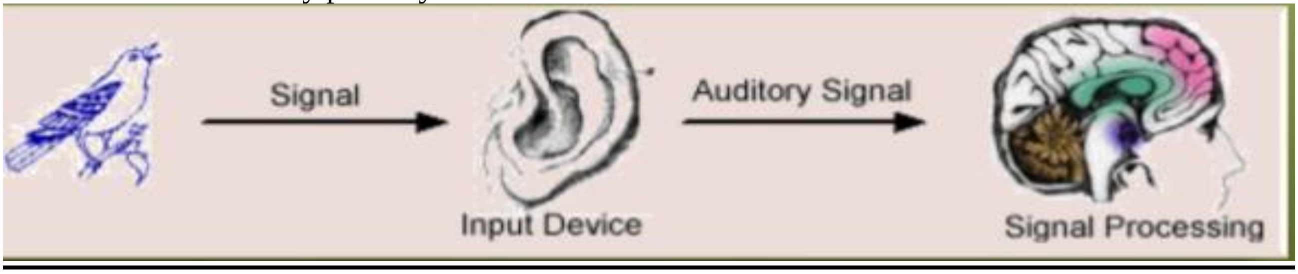


Figure. I. Types of Signal

II. What is Signal Processing?

When signal is passed through a system then the system is basically processed. In other words, the operations which are performed by system are usually referred to as signal processing. Processing means operating in some fashion on a signal to extract some useful information e.g., we use our ears as input device and then auditory pathways in the brain to extract the information.



I. Butterworth filter

The *Butterworth filter* is a type of signal processing *filter* designed to have a frequency response as flat as possible in the passband. It is also referred to as a maximally flat magnitude *filter*.

II. Sampling Frequency

The **sampling frequency** (or **sample rate**) is the number of **samples** per second in a Sound. For example: if the **sampling frequency** is 44100 hertz, a recording with a duration of 60 seconds will contain 2,646,000 **samples**. ... To get the **sampling frequency** of a selected Sound, click Info or choose Get **sampling frequency**.

III. Band-pass

A **band-pass filter** or **bandpass filter** (BPF) is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range.

6. The Graphs of the audio signal processed:

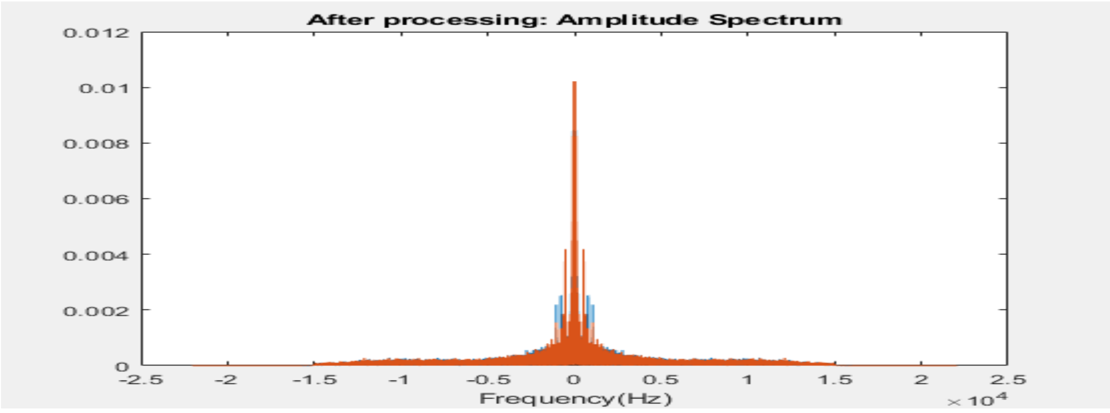


Figure.IX Demonstration of amplitude spectrum after processing

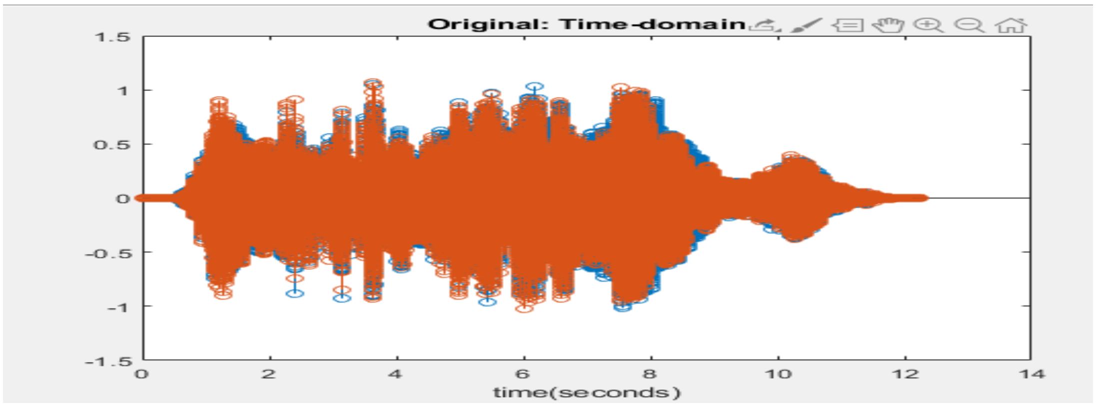


Figure.X Demonstration of original time domain in seconds

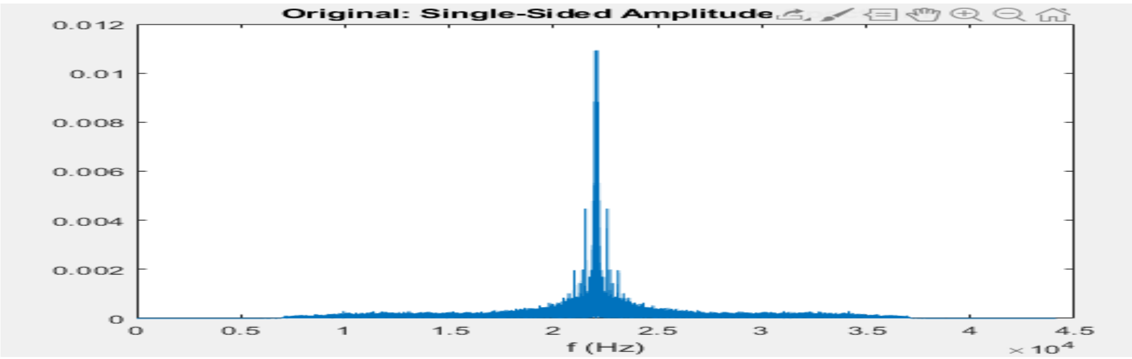


Figure.XI Demonstration of single sided amplitude in hertz

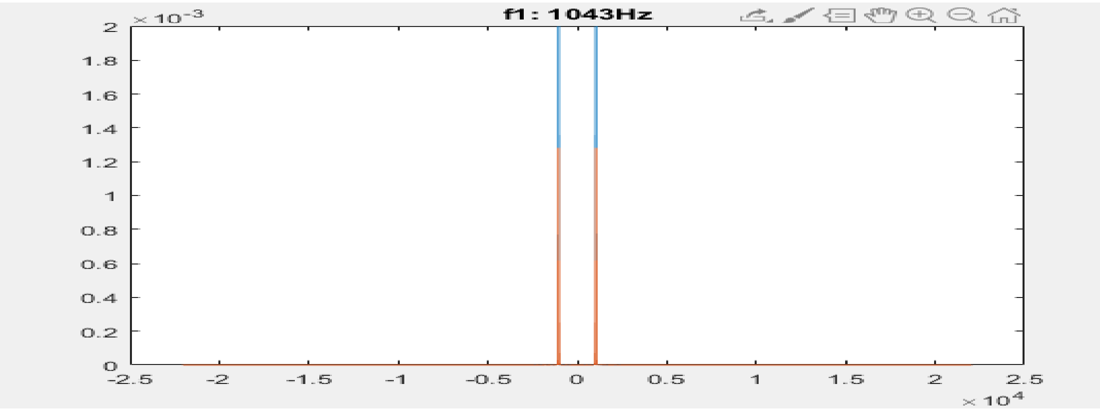


Figure.XII Demonstration at f1:1043Hz

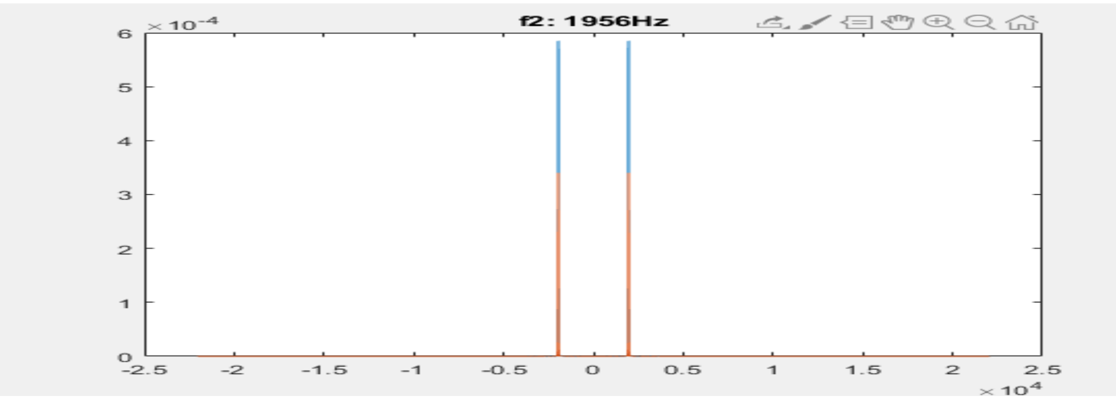


Figure.XIII Demonstration at f2:1956Hz

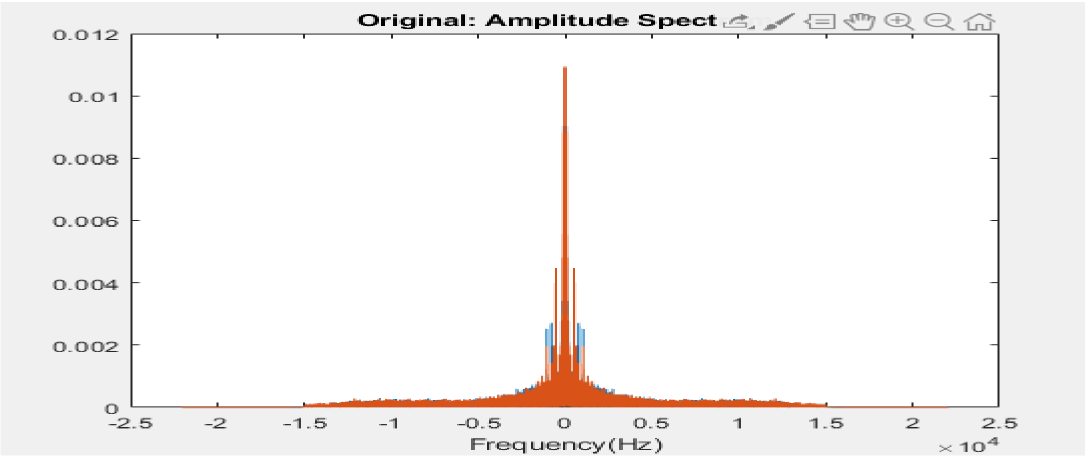


Figure.XIV Demonstration of amplitude spectrum

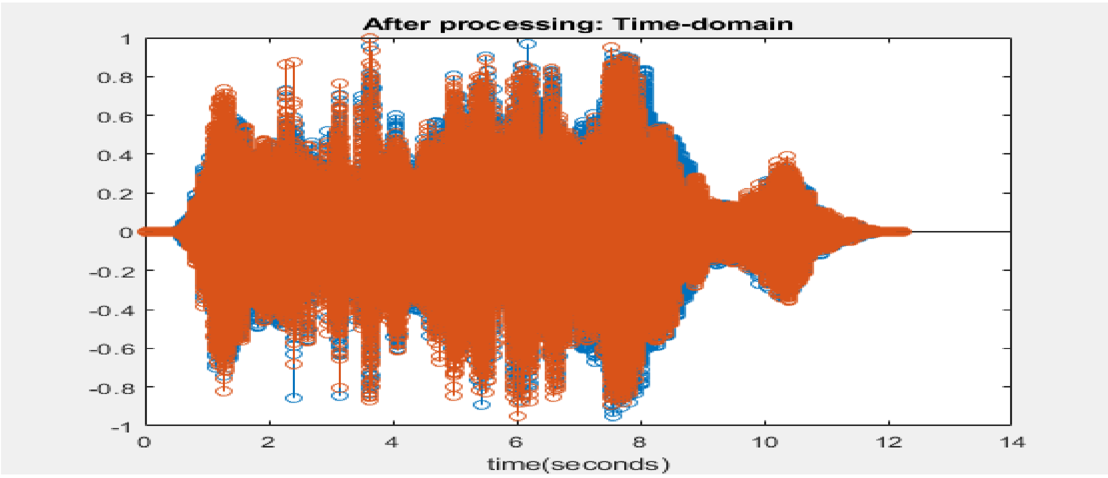


Figure.XV Demonstration of time domain (after processing)

7. Conclusion

The conclusion of the project is we have learned the code. We know how the code works and how to use MATLAB for signal processing and specifically audio signal processing and more specifically noise reduction in an audio signal. We can process any signal of such type as we have processed above. Our program gives desired results as we successfully removed the noise. The graphs of our code are shown above showing that our code successfully solved the problem given to it. At last, it is concluded that using MATLAB we can process any signal.

8. References:

All the data and help are taken from the following websites. We accessed the links numerously so not mentioning the dates here:

1. https://www.mathworks.com/help/audio/audio-processing-algorithm-design.html
2. https://www.music.mcgill.ca/~gary/307/week1/matlab.html
3. https://www.mathworks.com/matlabcentral/fileexchange/69330-audio-signal-processing