University of Tehran
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Real-time Digital Signal Processing Laboratory

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Lab 1

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Contents

Abstract

In this Lab we will attempt to remove a monotone from an audio file using an FIR filter designed via C programming, to verify our results we shall continue to utilize MATLAB and check the essential plots and audios before and after filtering.

1 Filtering a monotone from an audio signal

1.1 Part 1

In this part we read the *mefsin.wav* file using MATLAB and checked its sampling frequency, the result is as follows.

```
The sampling frequency of mefsin.wav in hertz is :
      16000
```

Figure 1: Sampling frequency

So the sampling frequency is :

$$f_s = 16000Hz$$

1.2 Part 2

In this part we draw the spectrum of the audio file with the help of the Fast Fourier transform (fft) in the desired range.

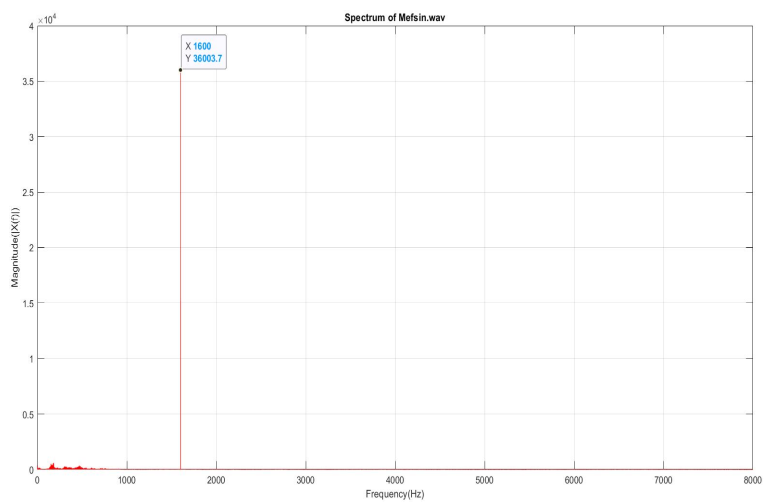

Figure 2: Audio spectrum

As we can see, there is a great peak in $f = 1600Hz$, this is undesirable and we wish to remove it.

1.3 Part 3

In this part we will attempt to obtain the coefficients for our FIR filter, we do this as follows:

$$f_{peak} = 1600Hz, \quad f_s \equiv= 2\pi, \longrightarrow \frac{f_{peak}}{f_s} = \frac{\theta}{2\pi} \longrightarrow \theta = 2\pi\frac{1600}{16000} = \frac{\pi}{5}$$

For more knowledge, we shall take a look at the FIR filter in the $\mathcal{Z}$ domain and time domain respectively.

$$H(z) = (1 - z_1 z^{-1})(1 - z_2 z^{-1}) \xrightarrow[z_2=e^{-j\theta}]{z_1=e^{j\theta}} (1 - e^{j\theta}z^{-1})(1 - e^{-j\theta}z^{-1})$$

$$\longrightarrow H(z) = 1 - 2\cos(\theta)z^{-1} + z^{-2}$$

$$\text{Filter Coefficients}: \ h[0] = 1, \quad h[1] = -2\cos(\theta), \quad h[2] = 1$$

1.4 Part 4

In this part we designed a C program to represent the FIR filter which performs the filtering as we require.

To verify our results, we plot the filtered signal in MATLAB accordingly.

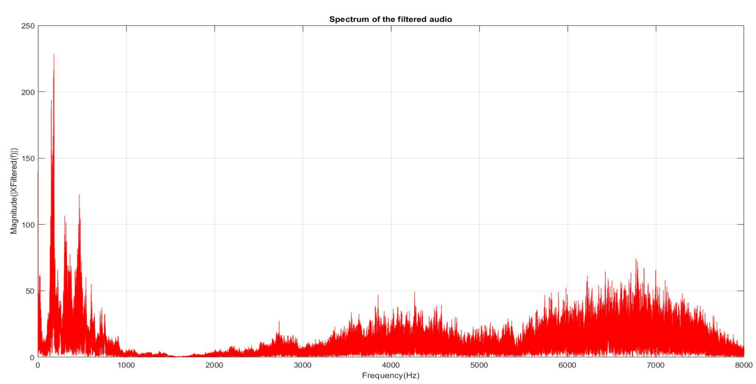

Figure 3: Filtered audio spectrum

As we can observe, the peak in $f = 1600Hz$ has been obliterated, hence we expect the audio file to be free of the disturbing noise, to verify this we generate the output file in *.wav* format using *audiowrite*.

As expected the output audio sound is clear and noise free.

References

[1] Vahid Shah-Mansouri, *Real-time Digital Signal Processing Laboratory lab notes, Fall 01*

[2] Mohammad Ali Akhaee, *Digital Signal Processing lecture notes, Spring 01*