EE321 – Digital Signal Processing
MATLAB Assignments: Set-6

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Submitted by

Manish Kumar - 200102108

Soham Karak - 200102107

Mohit Manoj - 200102106

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1 Bandpass Digital filter design

Given specifications :

$$\text{Stopband-1: } [0,\ 0.4\pi],\ \text{Attn.} = 40\ \text{dB}$$
$$\text{Passband-1: } [0.45,\ 0.55\pi],\ \text{Attn.} = 0.5\ \text{dB}$$
$$\text{Stopband-1: } [0.65\pi,\ \pi],\ \text{Attn.} = 50\ \text{dB}$$

1.1 Minimum Order IIR filter :

1.1.1 Magnitude response

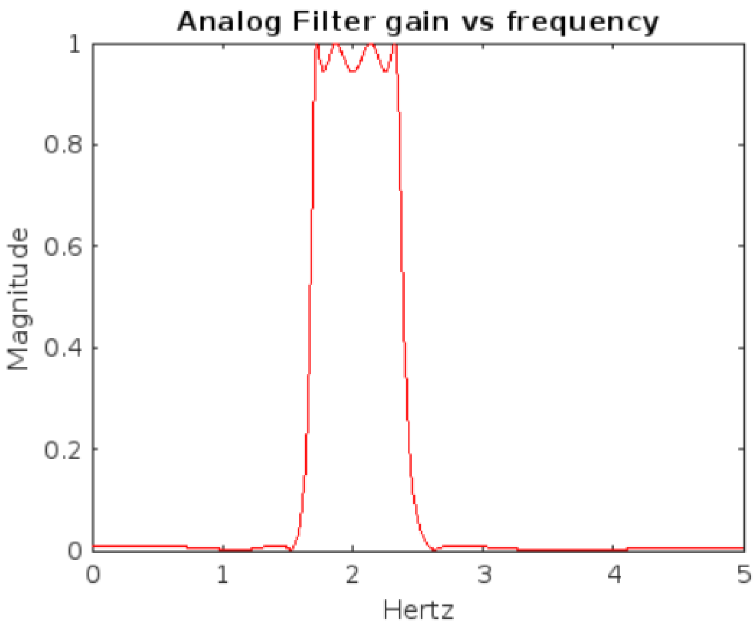

Figure 1: Magnitude Response of IIR filter

1.1.2 Phase Response

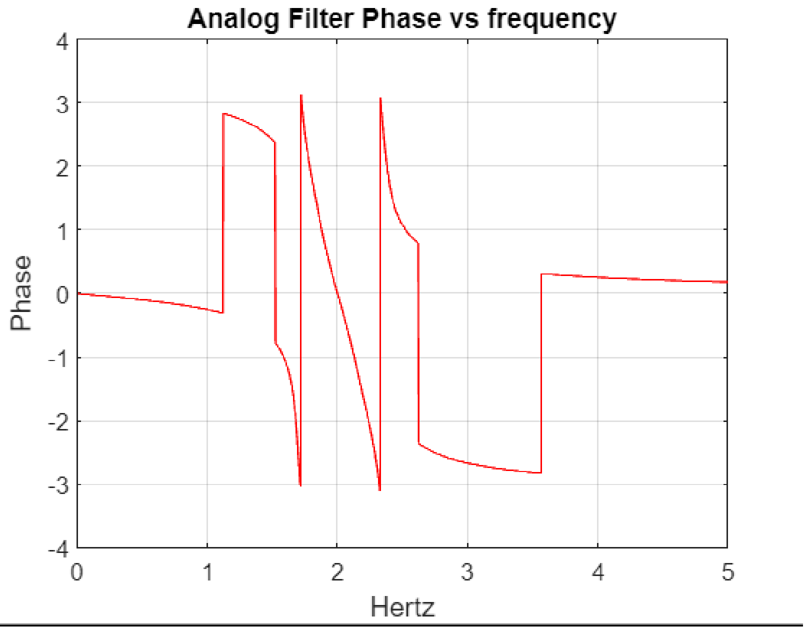

Figure 2: Phase Response of IIR filter

1.1.3 Log magnitude response

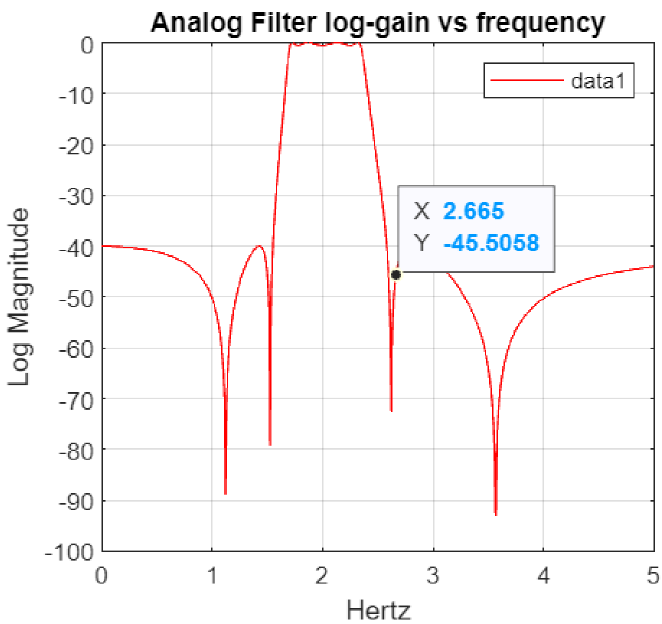

Figure 3: Log Magnitude Response of IIR filter

1.1.4 Group Delay response

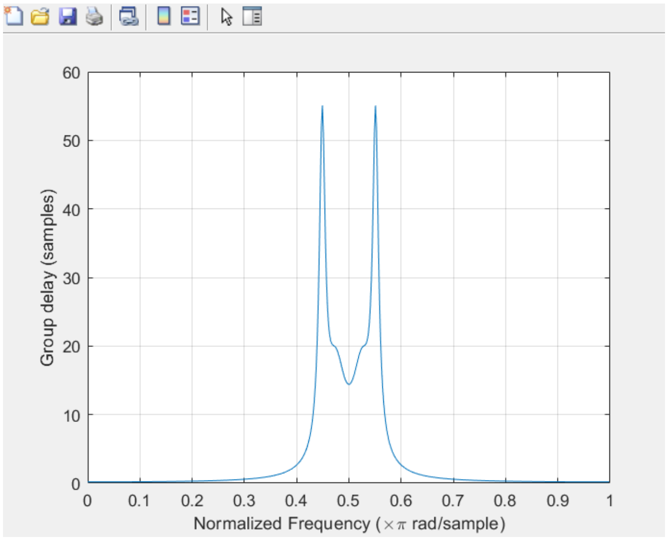

Figure 4: Group Delay response of IIR filter

1.2 Minimum Order FIR filter

1.2.1 Magnitude response

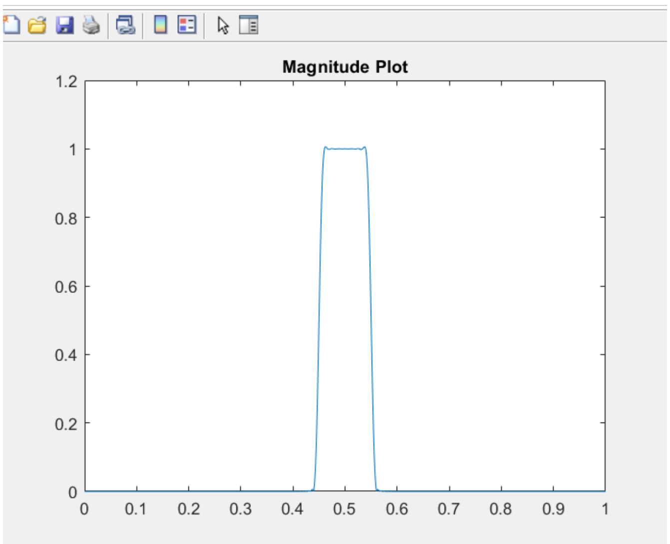

Figure 5: Magnitude Response of FIR filter

1.2.2 Log magnitude response

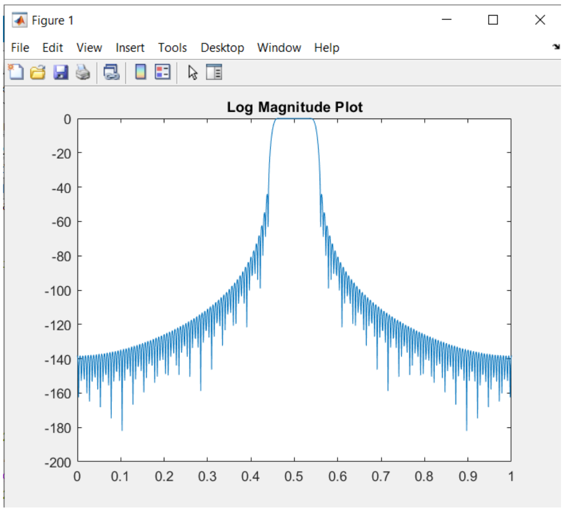

Figure 6: Log Magnitude Response of FIR filter in dB

1.2.3 Group Delay response

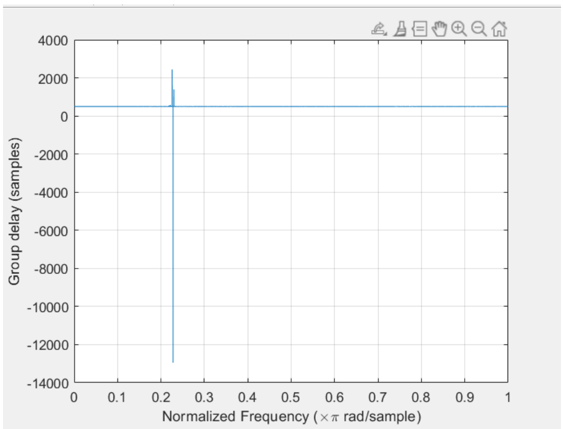

Figure 7: Group Delay response for FIR

1.3 Comparison :

An IIR filter has lower order than the FIR filter for the same specification. The plots from filterDesigner are attached in the zip file.

1.4 Impulse Response of Filters :

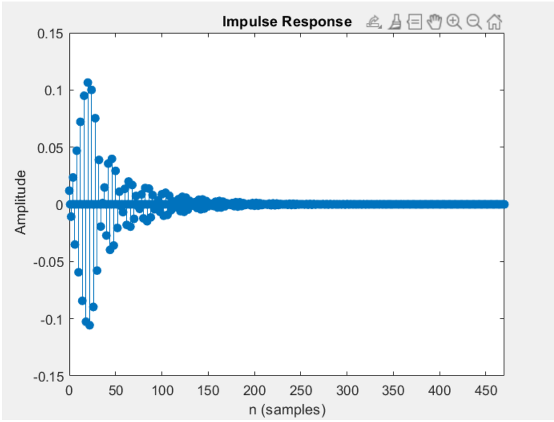

Figure 8: Impulse Response of IIR filter

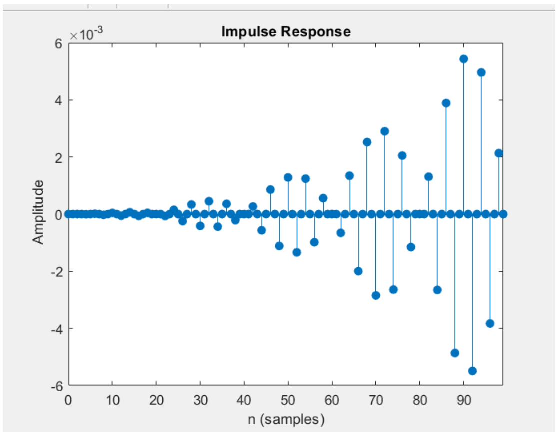

Figure 9: Impulse Response of FIR filter

2 Echo Removal

2.1 Intro

Here, we are given an audio file containing ”good morning” sound which is contaminated with an echo and we need to filter the echo using digital filter.

2.2 Theory

The audio file contains discrete samples so lets consider $y[n]$ to be the contaminated sound (with echo present) and let $x[n]$ be the original sound (no echo), that we need to recover from $y[n]$.

By definition, an echo is just the time delayed and attenuated form of original signal. So, we assume the following relation :

$$y[n] = x[n] + \alpha x[n-N]$$

where α denotes the attenuation and N denotes the delay (measured in samples). Clearly, we have $0 < \alpha < 1$.

Taking the Z-Transform :

$$\begin{aligned}
Y(z) &= X(z) + \alpha X(z) z^{-N} \\
Y(z) &= X(z)[1 + \alpha z^{-N}] \\
\frac{Y(z)}{X(z)} &= 1 + \alpha z^{-N} \\
H(z) &= \frac{X(z)}{Y(z)} = \frac{1}{1 + \alpha z^{-N}}
\end{aligned}$$

It is clear from the transfer function that if we want to get $x[n]$ from $y[n]$ the we need to find the values of α and N.

2.3 Finding Parameter values

2.3.1 For N :

For this, we first find the autocorrelation function of the input signal.

$$\begin{aligned}
R_{yy} &= \sum_{n=-\infty}^{\infty} y[n] \cdot y[n-k] \\
&= \sum_{n=-\infty}^{\infty} (x[n] + \alpha x[n-N]) \cdot (x[n-k] + \alpha x[n-k-N]) \\
&= R_{xx}(k) + \alpha R_{xx}(N+k) + \alpha R_{xx}(k-N) + \alpha^2 R_{xx}(k)
\end{aligned}$$

which takes its maximum values at $k = \pm N$ and $k = 0$. Hence, the echo appears at sample point n = N.

Plotting the autocorrelation in MATLAB and finding the peaks, we obtain N = 0.15*Fs = 1200, where Fs = 8000Hz and is the sampling frequency of the audio signal.

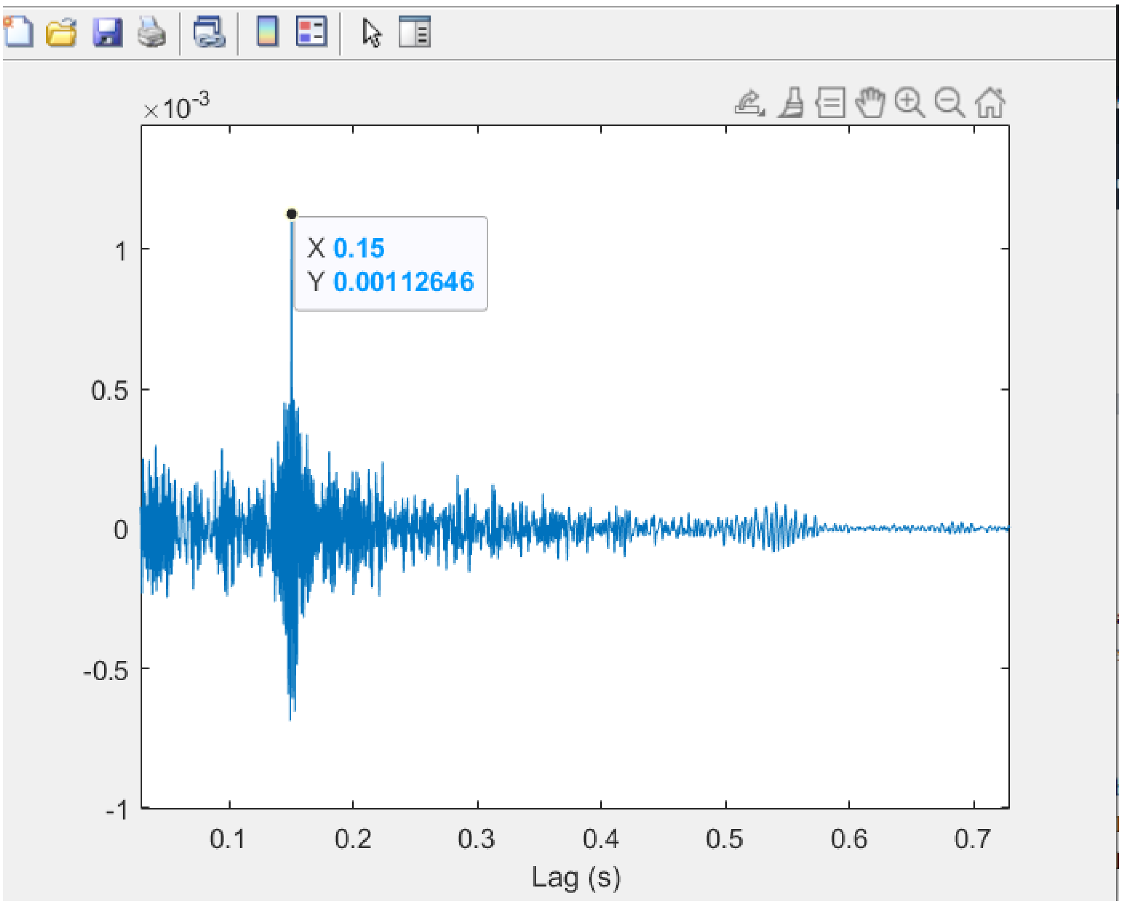

Figure 10: Autocorrelation function of the signal $y[n]$

2.3.2 For α :

Here we assume that $x[n] = 0$ for $n < 0$. Hence $x[n] = y[n]$ for $0 <= n < N$. This means that for the interval $[0, N]$ the original and contaminated signals are same.

Now, consider $y[N+1]$, which we can write as :

$$y[N+1] = x[N+1] + \alpha x[(N+1) - N]$$
$$y[N+1] = x[N+1] + \alpha y[1]$$

Now, if we assume $x[N] \approx x[N+1]$ i.e the samples are relatively smooth then, we can write :

$$y[N+1] = x[N] + \alpha y[1]$$
$$y[N+1] = y[N] + \alpha y[1]$$
$$\alpha = \frac{y[N+1] - y[N}{y[1]}$$

Clearly ,we already know N and $y[n]$ is the given audio signal. So we find the value of α using this and get $\alpha = 0.269$.

2.4 Conclusion

The echo removal filter is successfully designed and we are able to recover the original sound from it. We also present the graph of the signal after applying the filter.

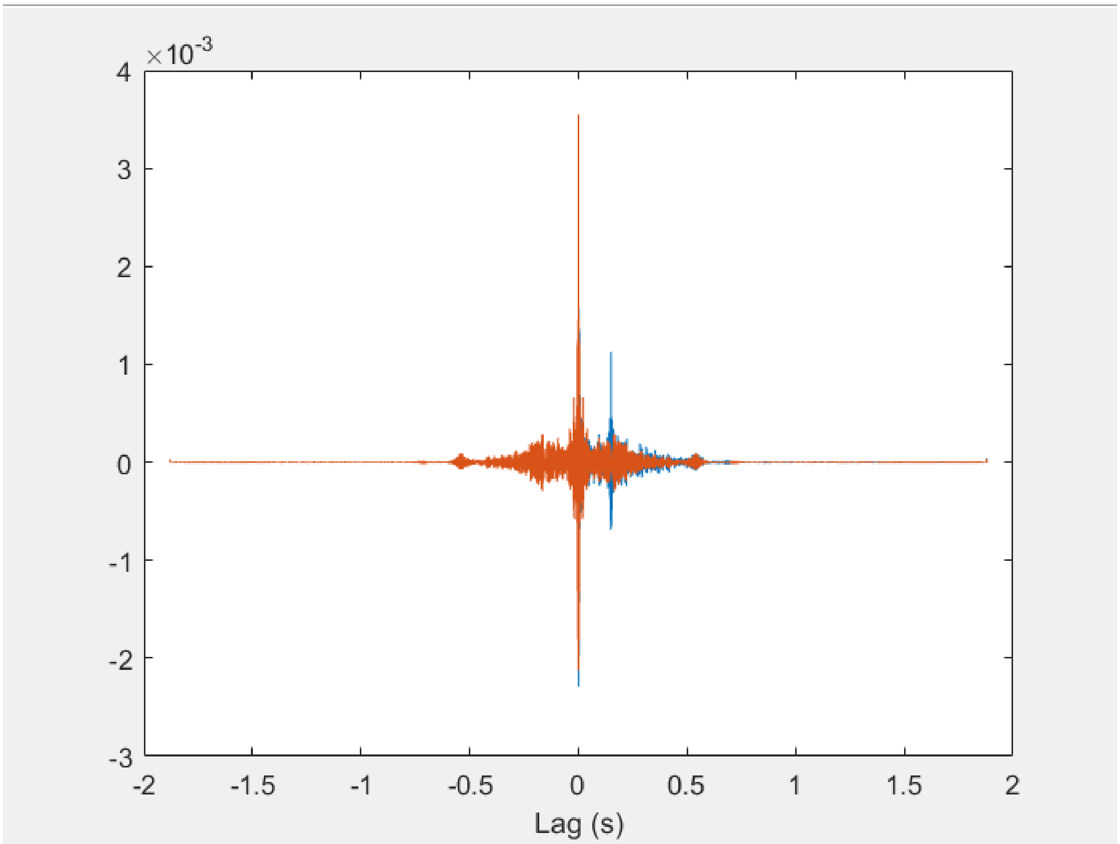

Figure 11: Autocorrelation function of the filtered signal