University of Tehran
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Real-time Digital Signal Processing Laboratory

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Lab 3

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Contents

Abstract

In this Lab we shall implement an IIR bandstop filter with order 6 and the given frequencies, we do this with fdatool, then we shall go on to implement this filter in C.

In the next part we go on to designing a C program in which we generate a gaussian white noise, we do this with the central limit theorem.

In the last part we implement the echo system in MATLAB and complete the steps as instructed in the lab manual.

1 IIR Elliptic Filter

Here we design the IIR filter as follows.

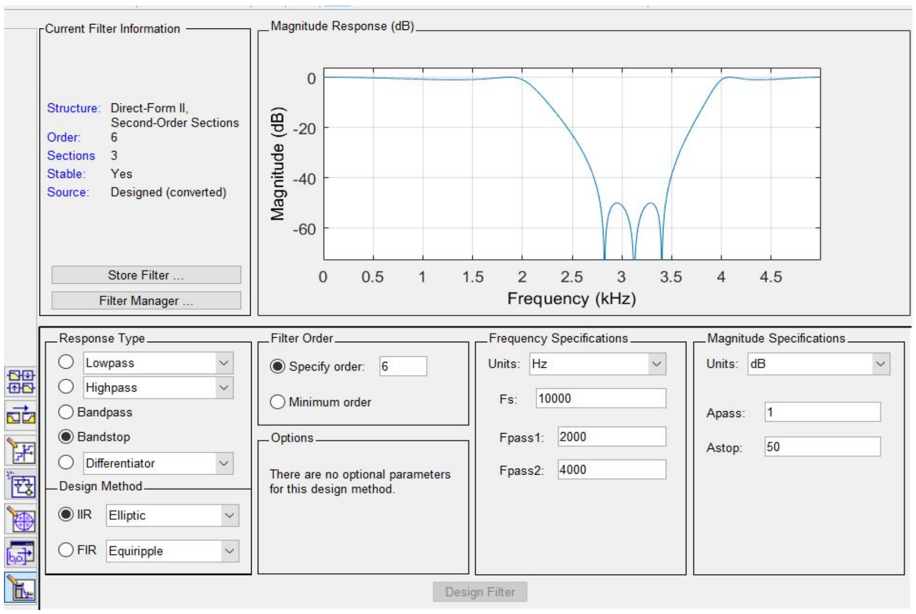


Figure 1: Filter Design

Now we plot the frequency response of this filter accordingly.

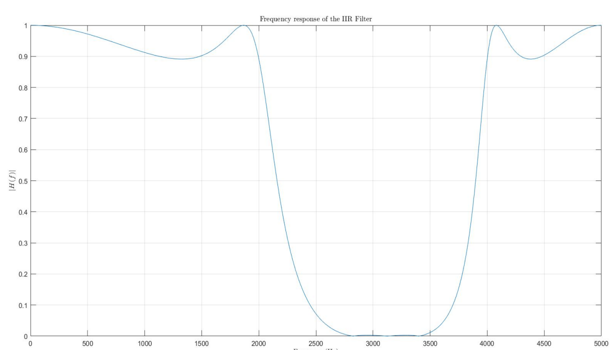


Figure 2: IIR filter frequency response

1.1 C Implementation of IIR Filter & White Noise

Here we designed a C program to implement an IIR filter, we did this with the aid of the hint included in the lab manual.

After implementing the IIR filter we designed a function in C to create white noise with the desired means and variances, to do this we used the **Central Limit Theorem** which I shall explain in the end of this report.

After applying the white noises to the IIR filter we get our desired results, the plots for the noises and the filter results can be seen as follows.

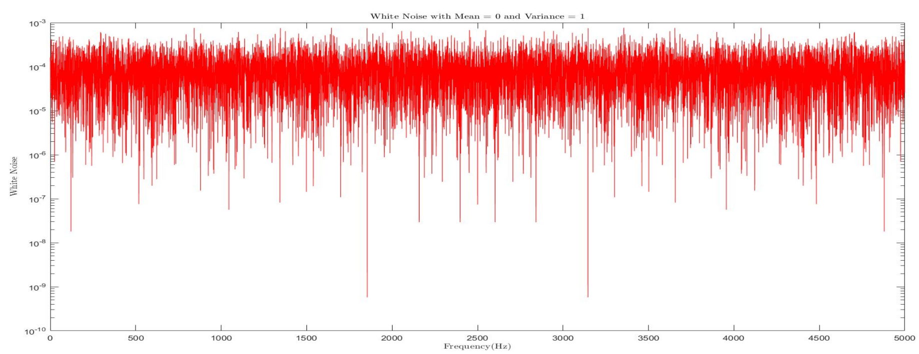


Figure 3: White Noise with Mean = 0 and Variance = 1

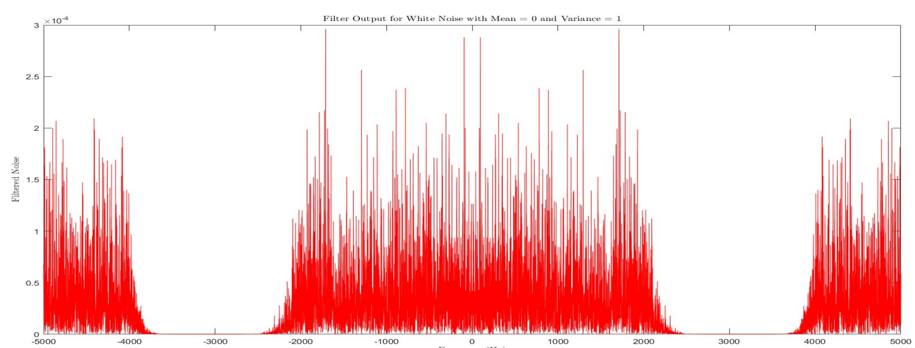


Figure 4: Filter Output for White Noise with Mean = 0 and Variance = 1

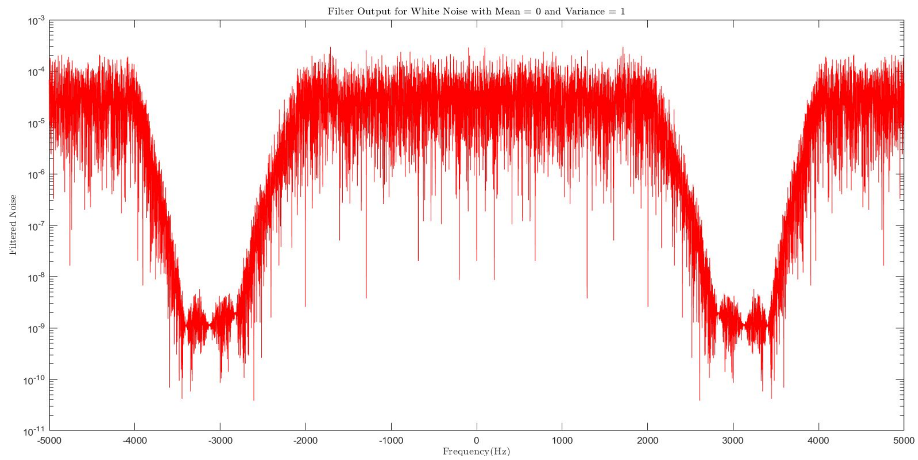


Figure 5: Filter Output for White Noise with Mean = 0 and Variance = 1 Logarithmic Scale

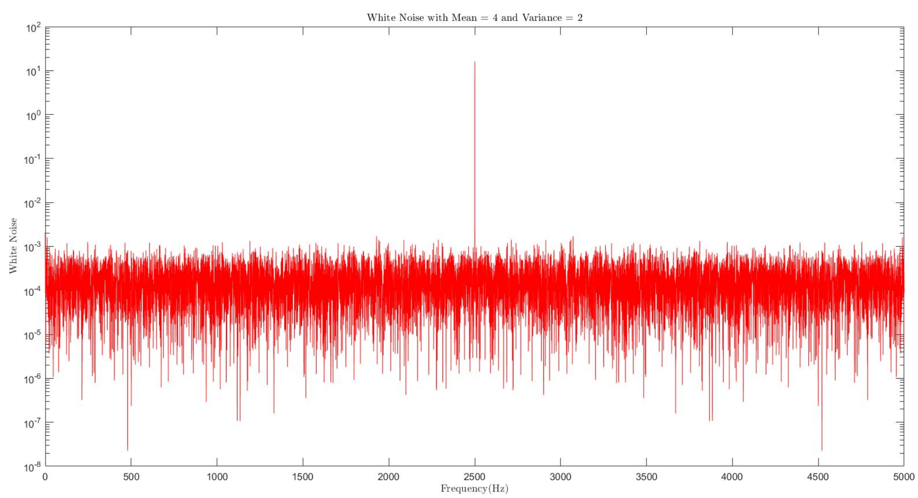


Figure 6: White Noise with Mean = 4 and Variance = 2

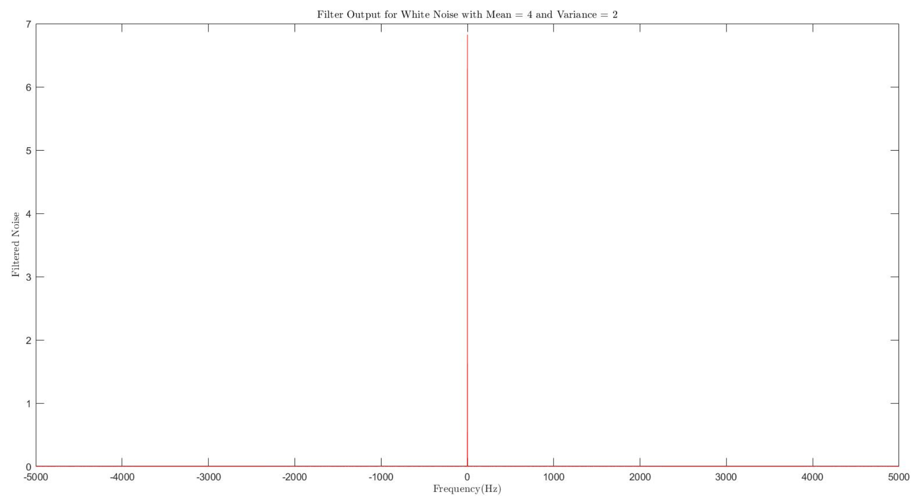


Figure 7: Filter Output for White Noise with Mean = 4 and Variance = 2

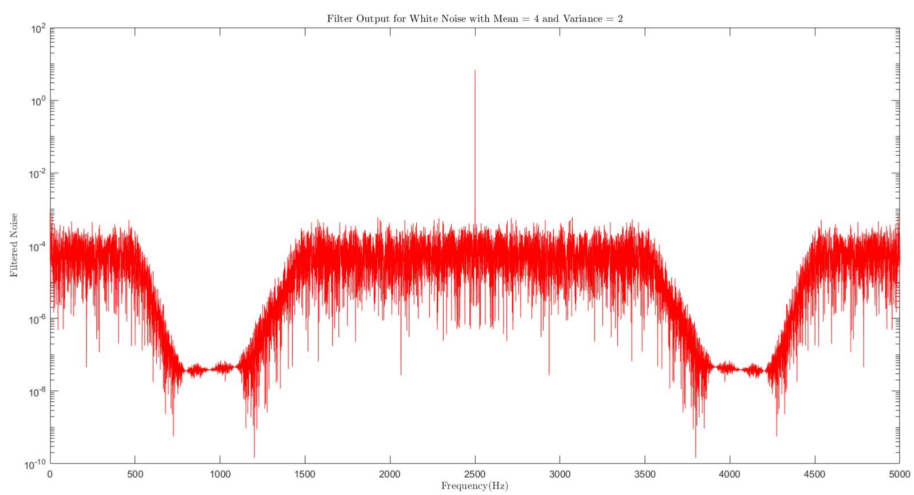


Figure 8: Filter Output for White Noise with Mean = 4 and Variance = 2 Logarithmic Scale

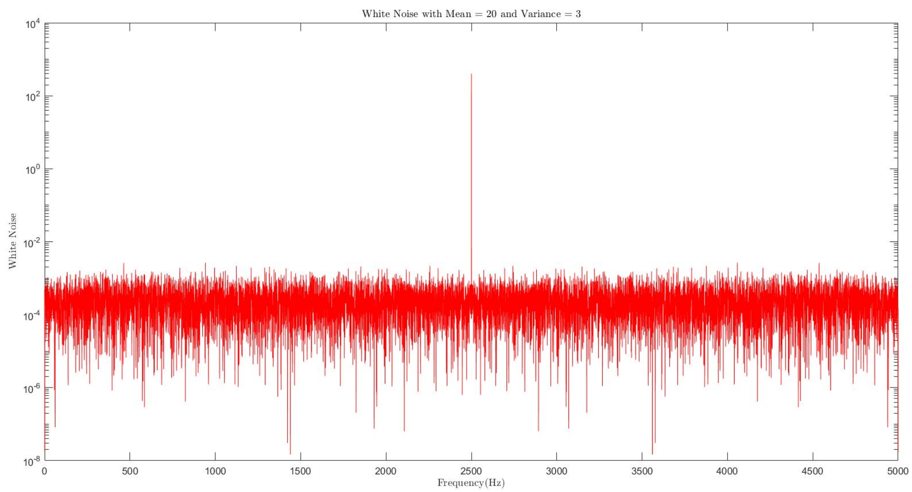


Figure 9: White Noise with Mean = 20 and Variance = 3

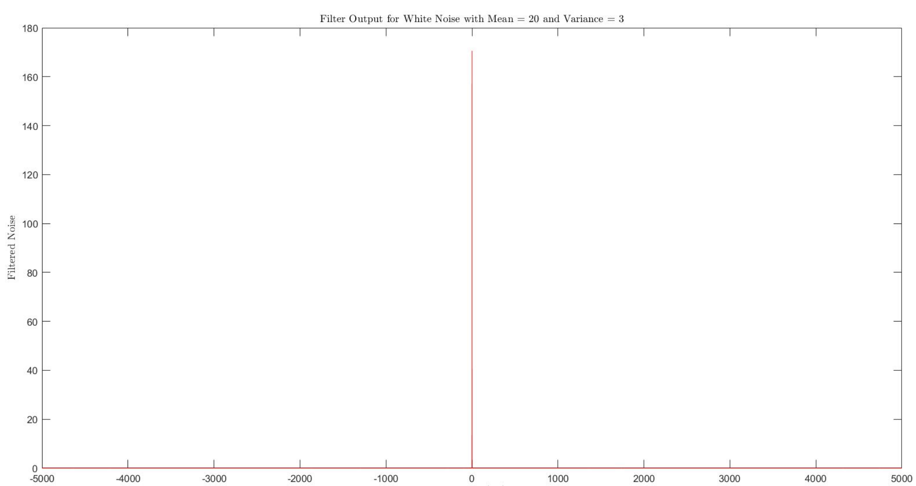


Figure 10: Filter Output for White Noise with Mean = 20 and Variance = 3

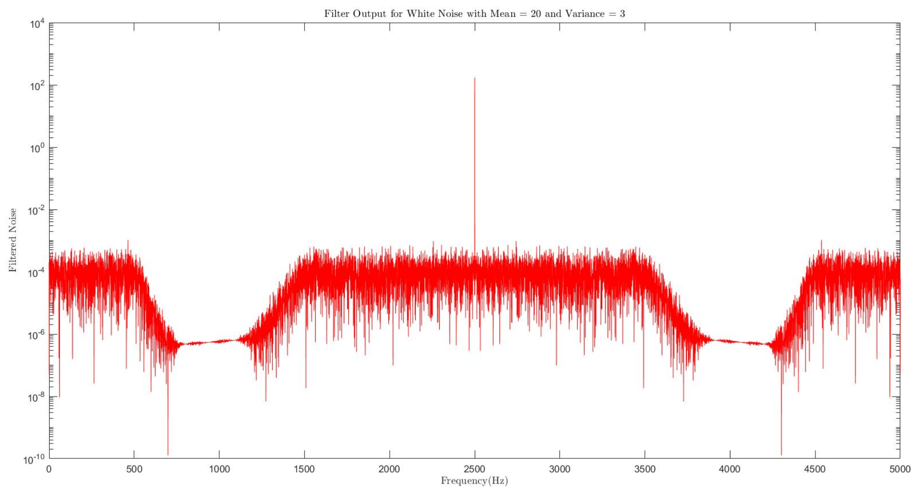


Figure 11: Filter Output for White Noise with Mean = 20 and Variance = 3 Logarithmic Scale

2 Echo Generating System

2.1 Transfer Function

Here we obtain the transfer function of the echo generating system.

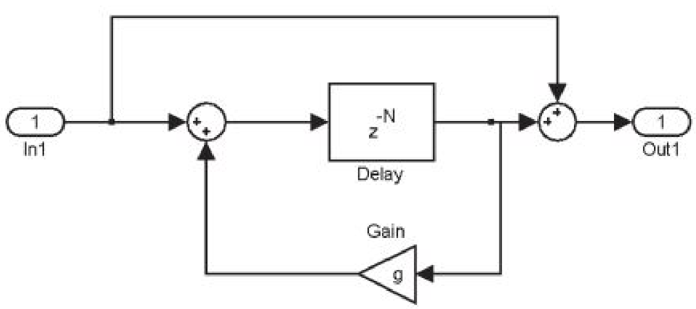
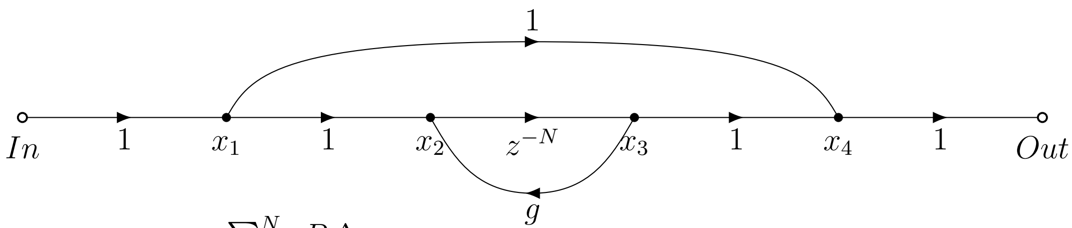


Figure 12: Echo Generating System

We shall use the Mason theorem to obtain the transfer function, to do so we need the **Signal Flowgraph** which is depicted below.



$$H(z) = \frac{\sum_{i=1}^{N} P_i \Delta_i}{\Delta}, \quad N = 2, \quad P_1 = z^{-N}, \quad P_2 = 1$$

$$\Delta = 1 - gz^{-n}, \quad \Delta_1 = 1, \quad \Delta_2 = 1 - gz^{-N}$$

$$H(z) = \frac{1 + (1-g)z^{-N}}{1 - gz^{-N}}$$

$$1 - gz^{-N} = 0 \xrightarrow{\text{Poles must be on left side}} 0 < g < 1 \longrightarrow \mathit{Stable}$$

2.2 Implementation in MATLAB

We implemented the filter produced from the transfer function in MATLAB accordingly, then I recorded a voice in MATLAB and fed it to the filter with different gains.

It is easily heard and observed that with the gain increasing the echo increases and if it increases more than a certain amount (about $g = 1$) the voice becomes horrible and intolerable and nothing is distinguishable from it.

3 Central Limit Theorem

We know that $X_1, X_2, \ldots, X_n$ are random variables with $\mathbb{E}(X_i) = m_i$ and $Var(X_i) = \sigma_i^2 < \infty$, if we define $a = \lim\limits_{n\to\infty} \frac{1}{\sqrt{n}} \sum_{i=1}^{n} m_i$ and $b = \lim\limits_{n\to\infty} \frac{1}{n} \sum_{i=1}^{n} \sigma_i^2$, it can be proven that $Z_n = \frac{\sum_{i=1}^{n} X_i}{\sqrt{n}}$ converges to Z in distribution where Z is defined as below.

$$Z_n \xrightarrow{d} Z, \quad Z \sim \mathcal{N}(a, b)$$

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*