Assignment 6 – Adaptive Filtering

Objective

In this assignment, our basic objective was to develop an FIR filter based on an IIR response using Least-Mean-Square algorithm. We implement a kind of adaptive filtering while updating the filter coefficients for every iteration based on the error obtained between the desired and actual output. The process diagram is as below:

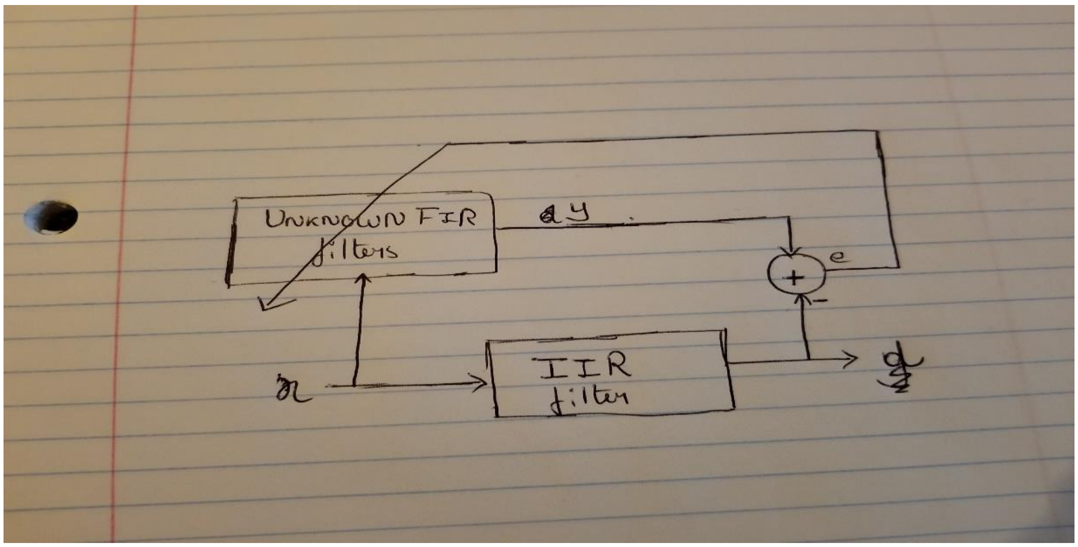


Fig 1: Process Diagram

Background

FIR filters are more precise and efficient than the IIR filters. Moreover, we can implement linear phase filtering using FIRs. We don’t get ant phase shifts while implementing FIR filters and phase can be independently corrected when we use them.

The only disadvantage of implementing FIR filters is their resolution.

But as it is easy to fine tune the parameters of FIR filter, we chose it.

Least Mean Square is an adaptive filtering method used to match the response of the unknown filter while updating the coefficients of the adaptive filter based on the error between desired signal and actual signal.

It is dependent on delta which is the function of convergence gain factor (a), power of the signal and length of the adaptive filter (L). Basically, for a given signal, we can vary a and L to obtain proper magnitude and phase response.

Expectations

I expected similar results as Weiner-Hopf filtering method and the results obtained were indeed similar.

Selection for parameters

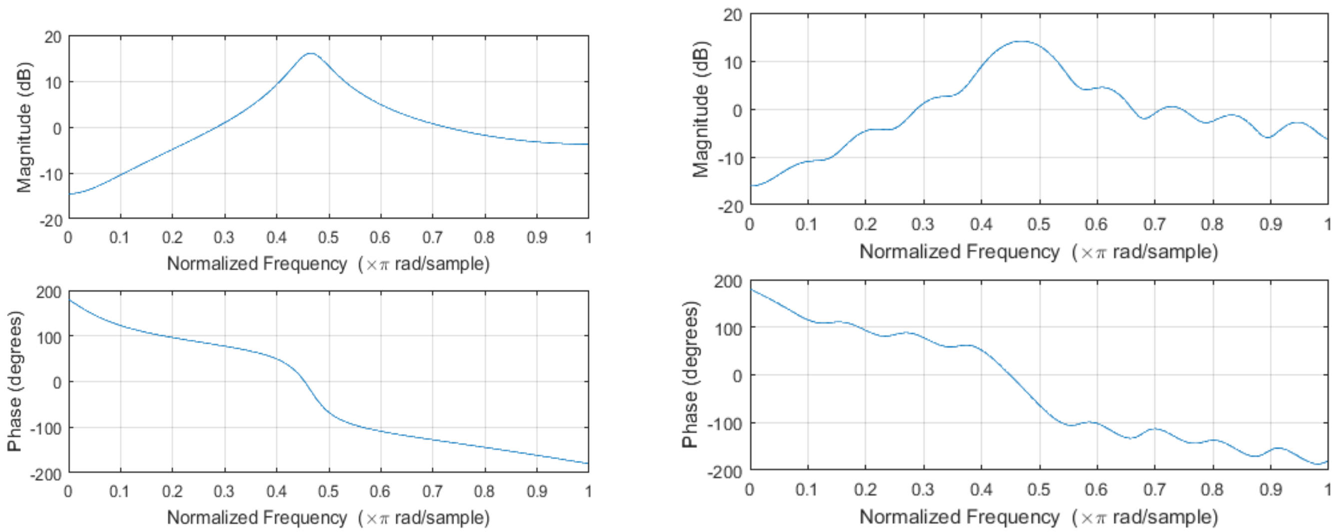
While selecting parameters, we need to keep in mind that power of the signal is constant and only two variables we can change are convergence gain factor (a) and length of the filter (L). These two have opposite effect on the magnitude and the phase response of the filter as a is in numerator while L is in denominator.

The only criteria we have to follow is that delta should be between 1/(10*P*L). To satisfy this criteria, we need to have a varying from 0.001 to 0.999, while L can not be larger than the length of the signal. Thus, by using trial and error method we found out the L = 20 and a = 0.99.

We also observed that the it’s not only about keeping the ratio between a and L constant as the filter length (L) is changing with the ratio too and thus the number of coefficients of the filter and also the value of the coefficients.

Results

With a = 0.99 and L = 20, we achieved the magnitude and phase response of both the filters identical to each other as shown in the figures below,



Original filter | Adaptive filter

Fig 2: Magnitude and Phase Response

We could see that both the magnitude and phase response are similar to each other.

Also, as the filter response is same the desired and actual output are also identical. Also, we could see that there’s a delay in the actual signal which means it takes time for the filter to adapt to the unknown filter which is obvious.

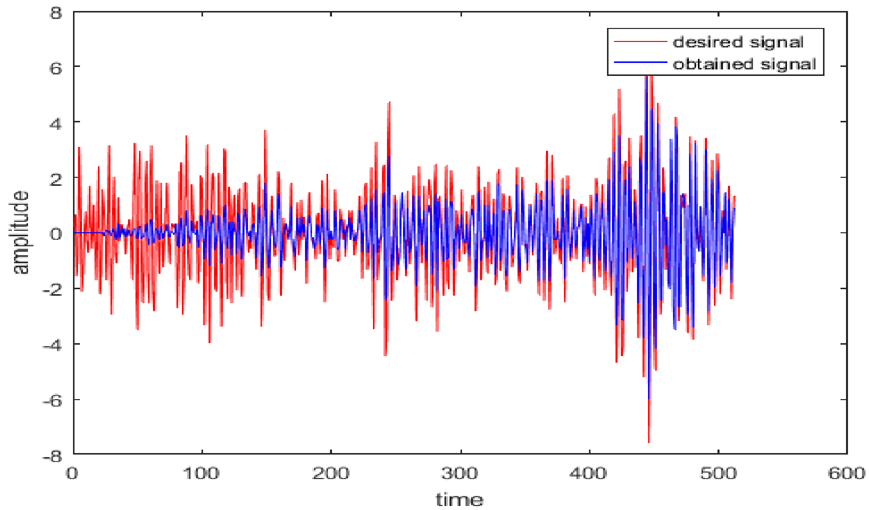


Fig 3: Desired signal v/s actual signal

We can also see that the frequency of both the signals match. Frequency spectrum of both the signals are plotted below:

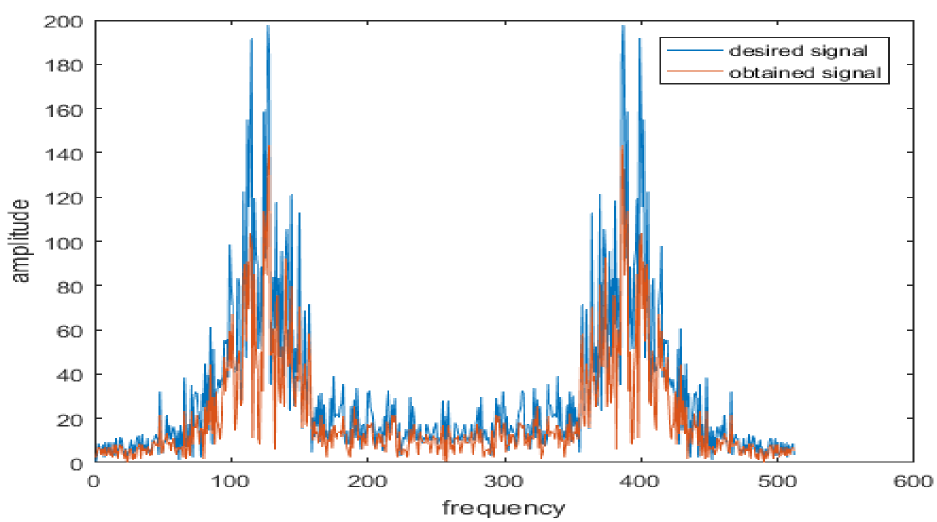


Fig 4: Frequency spectrum: Desired v/s actual

Conclusion: Thus, we successfully setup an adaptive filter using LMS algorithm where we altered the parameters a and L using trial and error method to find the best match between magnitude and phase response.