



## LOW PASS FIR FILTER DESIGN USING OPTIMAL LS METHOD

Jyoti Patel<sup>1</sup>, Silky Pareyani<sup>2</sup>

<sup>1</sup>M.Tech Scholar, ECE Department, GGCT, Jabalpur

<sup>2</sup>Asst. Professor, ECE Department, GGCT, Jabalpur

\*\*\*

**Abstract-** The paper represents the design of linear phase finite impulse response (FIR) digital filter under optimization framework, which uses least squares design method for minimizing magnitude response error and thereby reducing the ripple content. In this work we present least squares (LS) approach to design linear phase Finite Impulse Response (FIR) filter. Since the design of FIR digital filters is non-analytic, we aim at ideal zero-phase magnitude response and minimize the weighted error in passband and stopbands. The problem of least squares can then be solved non-iteratively by solving system of linear equations. Solution of which yields impulse response that is both real and symmetric. Frequency response of the proposed LS FIR filter shows a flat passband, and higher stop-band attenuation than traditional FIR design.

**Key Words:** Finite Impulse filter, Digital filter design, Evolutionary algorithms, Digital signal processing

### 1.INTRODUCTION

A digital filter is a system that performs mathematical operations on a sampled, digitized signal to reduce or enhance certain features of the processed signal. Digital filter scheme consists of a pre filter or anti-aliasing filter to perform filtering of an input signal using a low pass filter. This is required to restrict the bandwidth of a signal to satisfy the sampling theorem. An interface is needed between the analog signal and the digital filter, this interface is known as analog-to-digital converter (ADC). After the process of sampling and converting, a digital signal is ready for further processing using an appropriate digital signal processor. The output signal that is digitized is usually changed back into analog form using digital-to-analog converter (DAC). Digital filter is a major topic in the field of digital signal processing (DSP). Over the past few years the field of DSP has become so popular both technologically and theoretically. The major reason for its success in the industry is due to the use of the low cost and development of software and hardware. Applications of DSP are mainly the algorithms that are

implemented either in software using interactive software like MATLAB or a processor. In high-bandwidth applications FPGA, ASIC or a specialized digital signal processor are used for expediting operations of filtering. Digital filters are preferably used because they eliminate several problems associated with analog filters. There are two fundamental types of digital filters: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR).

#### 1. FIR FILTER:

Filters play an important role in digital signal processing applications. They are widely used in digital signal processing applications, such as digital signal filtering, noise reduction, frequency analysis, multimedia compression, biomedical signal processing and image enhancement etc. A digital filter is a system which passes some desired signals more than others to reduce or enhance certain aspects of that signal. It can be used to pass the signals according to the specified frequency pass-band and reject the other frequency than the pass-band specification.

The frequency response of an ideal low pass filter (LPF) is shown in the Figure 2. The ideal LPF is a filter having linear phase characteristics which does not affect low frequencies and discards high frequencies. In other words, it perfectly passes all of the frequencies from 0 Hz for example, to the cut-off frequency  $f_c$  without any attenuation, and completely eliminates all the frequencies above  $f_c$ . The term cut-off frequency refers to the frequency at which the response begins to fall off significantly. Mathematically, an ideal low pass filter has a magnitude response given by:

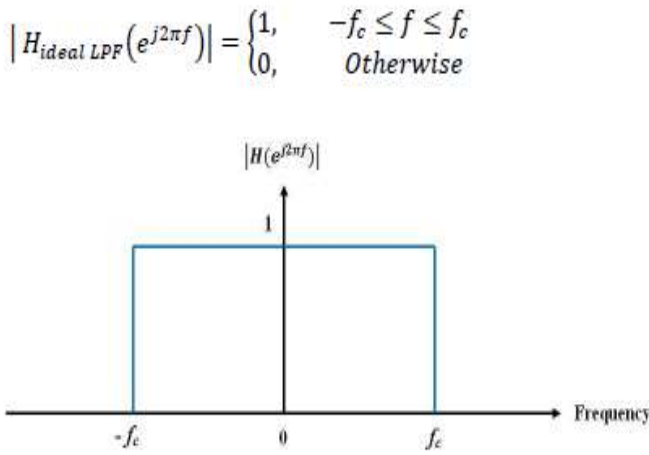


Figure 1: Ideal LPF response

It is helpful to investigate the impulse response of such filter. The impulse response is the output of the system in the time domain when the input is provided. Hence, the impulse response of the ideal LPF is

$$h_{ideal\ LPF}(t) = \begin{cases} \frac{\sin(2\pi f_c t)}{\pi t}, & \text{for } t \neq 0 \\ 2f_c, & \text{for } t = 0 \end{cases}$$

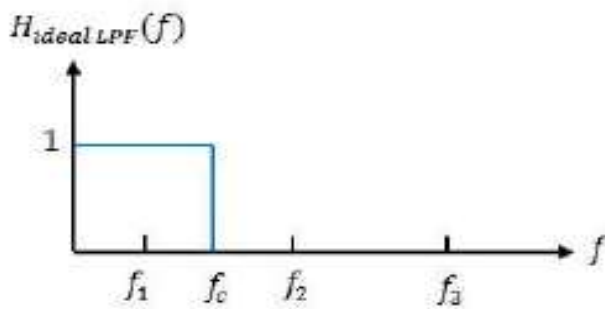


Figure 2: Ideal LPF output

FIR filters also known as non-recursive digital filters have a finite impulse response because after a finite time the response of FIR filter settles to zero. Block diagram of FIR filter is shown in Figure. The basic structure of FIR filter consists of adders, multipliers and delay elements as shown in Figure. The difference equation of nth order digital filter (FIR) can be represented as:

FIR channels otherwise called non-recursive computerized channels have a limited motivation reaction in light of the fact that after a limited time the reaction of FIR channel settles to nothing. Block graph of FIR channel is displayed in Figure. The fundamental construction of FIR channel comprises of adders, multipliers and defer components as displayed in Figure. The distinction condition of nth request computerized channel (FIR) can be addressed as:



Figure 3: Block diagram of digital filtering process.

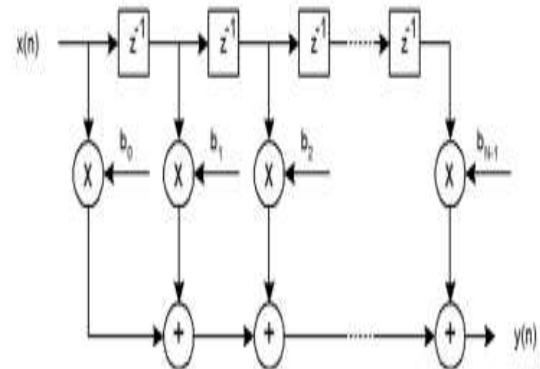


Figure 4: Basic structure of FIR filter.

## 2. FIR FILTER DESIGN

FIR filters have boosted the development of digital signal processing, beamformers, and so on, due to their stability and low coefficient sensitivity. Generally, the design of FIR filters follows two main principles, i.e., the specification on the response error and the low implementation complexity. In this brief, we describe the sparsity of the filter coefficients using the k-maximum function, which equals to -0-norm under mild conditions and has no restriction on the magnitude of nonzero coefficients. In order to avoid possible violation of specifications on response errors caused by frequency discretization, we estimate the frequencies at which the magnitude of the response error is maximized when constructing linear problems in the proposed algorithm. To address the nonlinearity and nonconvexity of the resulted optimization problem, we transform it into a piecewise linear concave optimization (PLCO) problem (1).

In designing FIR filter, most important parts are approximation and realization. Transfer function can be calculated in four steps after taking specification in approximation stage as, usually in the frequency domain, desired or ideal response is chosen. Filter class is chosen which is allowed (e.g. the tap for a FIR filter). Approximation quality is chosen. Lastly, best algorithm is selected which is used to find the transfer function. Implementation of the above transfer function in the form of circuit (blocks) or program (coding) is done by selecting the structure of filter, this stage is called as realization. Filter structure selection is important part in implementation on FPGA because of area and speed. Hardware implementation part in pre modulation cannot afford more area because of less space in on flight [4]. There are three types of FIR filter design techniques,

In planning FIR channel, most significant parts are estimation and acknowledgment. Move capacity can be determined in four stages subsequent to taking detail in estimate stage as, as a rule in the recurrence area, wanted or ideal reaction is picked. Channel class is picked which is permitted (for example the tap for a FIR channel). Estimation quality is picked. In conclusion, best calculation is chosen which is utilized to observe the exchange work Implementation of the above move work as circuit (squares) or program (coding) is finished by choosing the design of channel, this stage is called as acknowledgment. Channel structure choice is significant part in execution on FPGA in view of region and speed. Equipment execution part in pre adjustment can't manage the cost of more region as a result of less space in on flight [4]. There are three sorts of FIR channel plan strategies,

- a) Windowing technique
- b) Frequency sampling
- c) Optimal design technique

We cannot achieve minimum order of filter with window design technique because it is a simple and convenient design technique for higher order filters. Rectangular, Blackman, Hamming, Hanning, Kaiser, Flat-top and Gaussian are some of the design techniques which are mostly used [5].

Frequency sampling design technique is the simplest and most direct technique if the desired frequency response is specified. In this technique desired frequency response can be obtain by sampling the frequency response which is provided by the previous method [4]. There are many optimal design techniques where we can specify pass and stop bands. Some of these techniques are equi-ripple and least square methods. Most important type of optimal design technique is Parks – McClellan algorithm [6]. In this paper this algorithm is still optimized such that pass band error is reduced.

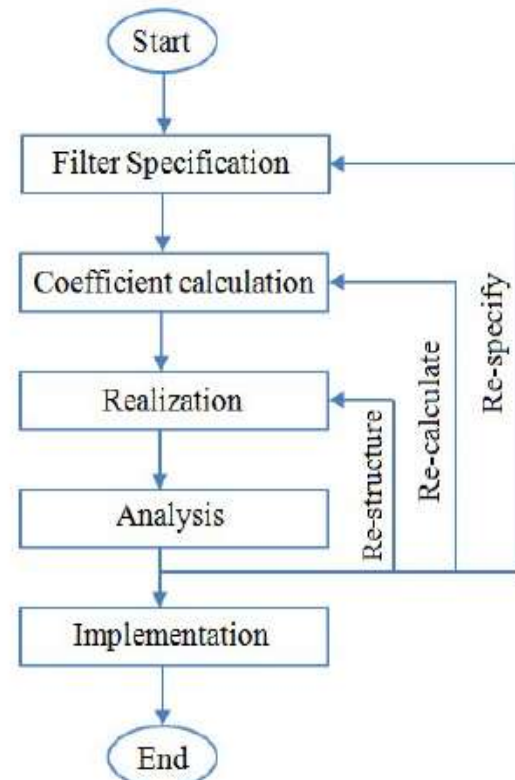


Figure 5: FIR filter design steps

### 3. PROPOSED METHOD

Different methods for designing of FIR filter design

1. Window method
2. Frequency sampling technique
3. Optimal filter design method

An optimal filter design method various method are used to design the filter coefficient again and again until a particular error is minimized.

To create a finite impulse response (FIR) the impulse response  $h(n)$  is convolved with input signal  $x(n)$  using

$$Y(n) = \sum_{i=1}^m x(n-i) \cdot h(i)$$

Where  $m$  is the number of points used to express the convolution.

In FIR designing, the following specifications are essential;

- (i) pass band edge normalized frequency
- (ii) stop band edge normalized frequency
- (iii) pass band Ripple  $\delta_p$
- (iv) Sampling frequency  $F_s$

To determine the pass band ripple and minimum stop band attenuation used the following expression

$$\delta_p = -20 \log(1 - \delta_p) \text{ db}$$

$$\delta_s = -20 \log(1 - \delta_s) \text{ db}$$

LS-optimal FIR Filter method employs the use of a minimum mean square error criterion given as

$$e = \sum_{i=1}^k W_i [H_i - D(W_i)]^2$$



where  $H(W)$  is the amplitude response of the desired amplitude response and  $W(w_i)$  is the weighting function.

#### 4. FIR FILTER DESIGN PROCEDURE

In practice, design of frequency selective digital FIR filters involves five steps, which can be summarized as follows:

- 1- Filter specification: As discussed in the previous chapter, this includes specify the filter type, such as LPF, HPF, BPF or BSF, with preferred amplitude response, as well as pass band, stop band and sampling frequency.
- 2- Coefficient calculation: The main goal here is to calculate the transfer function of a filter and then determining its coefficient by a proper method to satisfy the specifications in step (1) with a minimum of computational processes.
- 3- Realization: As discussed before, this includes converting the transfer function into appropriate filter structure.
- 4- Analysis: This includes analyzing, simulating and testing with real data to examine whether the filter meets the performance requirements. If not, return to step 2 or reduce the performance requirements.
- 5- Implementation: This includes implementing the actual filter obtained in software form, hardware or both.

#### 5. RESULTS AND DISCUSSION

To show the effect of the window function on FIR filter design and according to the equations, a typical relation between the magnitude frequency response of the desired or ideal LPF  $H_d(e^{jw})$  window function  $W_d(e^{jw})$ . The Fourier transform of the window

$$H_w(e^{jw}) = H_d(e^{jw}) * W(e^{jw})$$

It can be observed that the width of the transition band is determined by the width of the main-lobe of the window function

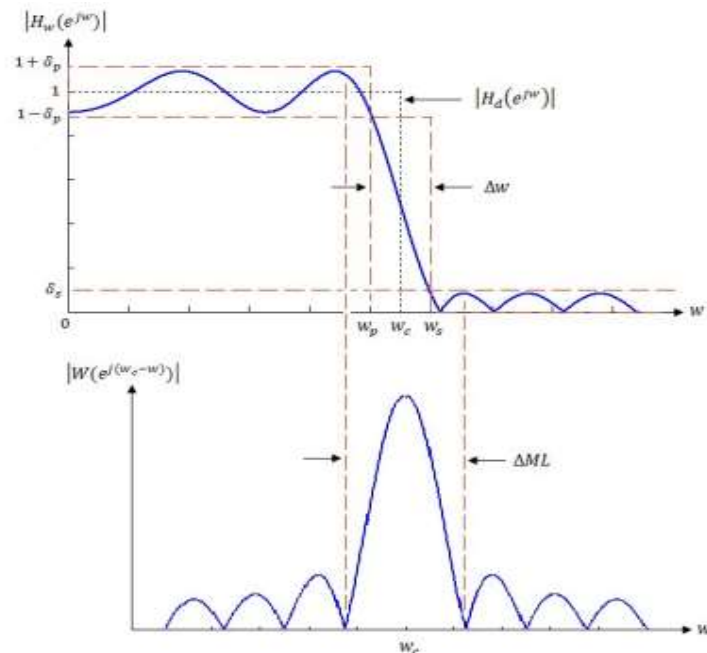
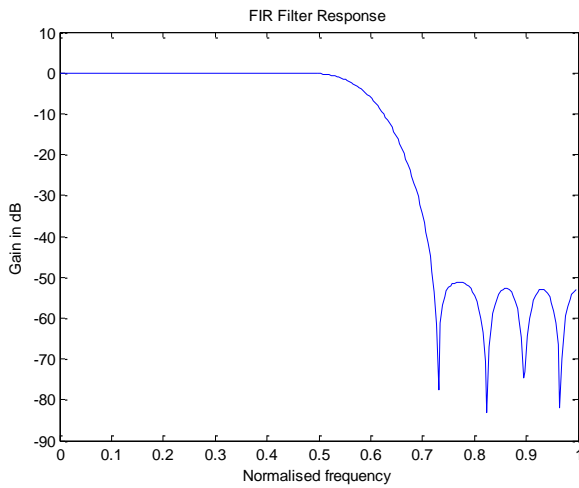


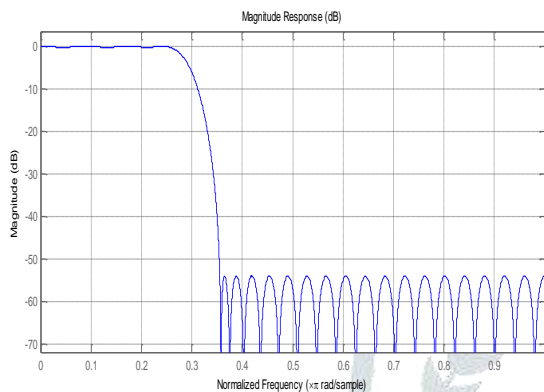
Figure 6: The effect of the window function in the frequency domain

Table 1 Comparison of Pass band and stop band Ripple

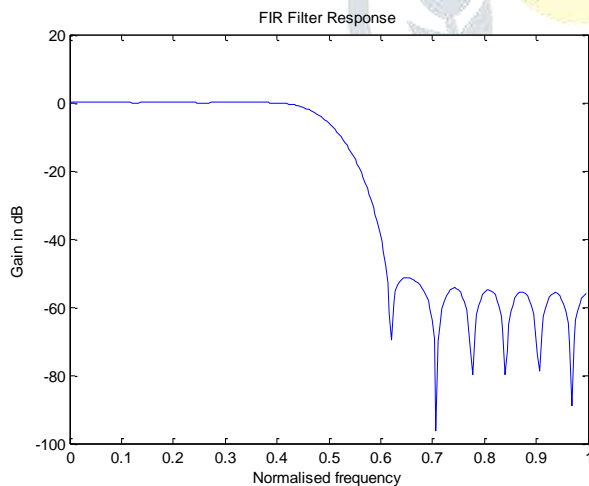
s.no.	Order Of Filter	Algorithm	Pass band Ripple	Stop band Ripple
1.	M= 50	Existing Algorithm (Ref 1)	0.009	0.145
		Proposed Algorithm	0.0002	0.0019
2.	M= 100	Existing Algorithm(Ref 1)	0.000123	0.0012
		Proposed Algorithm	0.00034	0.0154
3.	M= 150	Existing Algorithm (Ref1)	0.0089	0.01
		Proposed Algorithm	0.000037	0.0034



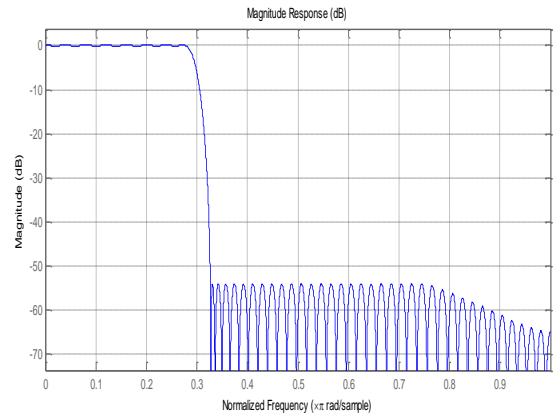
**Figure 7: Frequency response (in db) of Low Pass FIR Filter using ExistingAlgorithm (For M= 50)**



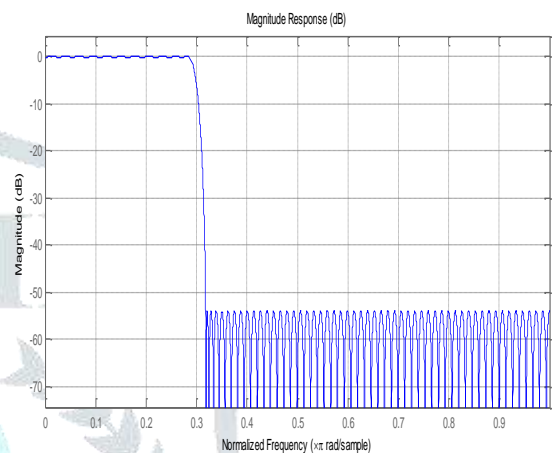
**Figure 8: Frequency response (in db) of Low Pass FIR Filter using Proposed Algorithm (For M= 50)**



**Figure 9: Frequency response (in db) of Low Pass FIR Filter using Existing Algorithm (For M= 100)**



**Figure 10: Frequency response (in db) of Low Pass FIR Filter using Proposed Algorithm (For M= 100)**



**Figure 11: Frequency response (in db) of Low Pass FIR Filter using Proposed Algorithm (For M= 150)**

**6. CONCLUSIONS**

In this present work least square error design method is presented for the optimal design of FIR filter. The analysis shows that as the order of the filter is increased the ripple content in the stop band diminishes and can be seen with a greater amount for lower orders. The results when compared with the design using existing method shows that the ripple content disappears in a similar way but the only difference that ripples diminish suddenly using least square error design and in latter case ripples smoothen the response and give a constant response in stop band. Further least square error design can be modified using optimization algorithms. Least square error designs can be extended in future to design optimal FIR filters using optimization algorithms for both linear phase and Non-linear phase as well as for multi-objective designs. Various works are being done to extend least squares error design orthogonally using Orthogonal Least Square (OLS) algorithm to design linear and Non-linear digital filters.

**7. REFERENCES**

[1] Xiangming Xi and Yunjiang Lou, (2021), Sparse FIR Filter Design With k-Max Sparsity and Peak Error constraints, IEEE transactions on circuits and systems—II: express briefs, vol. 68, no. 4, April 2021, page no. 1497-1501.

- [2] Apoorva Aggrawal et.al, (2018), Design of optimal band stop FIR Filter using L1- norms based RCGA” Ain shams engg journal, volume 9, page no. 277-289.
- [3] Wen bin ye, xin lou, and ya jun yu (2017) Design of low power multiplierless linear phase FIR filter IEEE Access, vol 2 page no. 23466-23472.
- [4] Alia Ahmed Eleti et.al, (2013), FIR filter Design by using window method with MATLAB, IEEE 14<sup>TH</sup> conference STA-2013.
- [5] Zhang, M., & Kwan, H. K. (2017, April). FIR filter design using multi objective teaching-learning-based optimization. In Electrical and Computer Engineering (CCECE), 2017 IEEE 30th Canadian Conference on (pp. 1-4).
- [6] Dash, J., Dam, B., & Swain, R. (2017), optimal design of linear phase multi-band stop filters using improved cuckoo search particle swarm optimization. Applied Soft Computing, 52, 435-445.
- [7] Shao, P., Wu, Z., Zhou, X., & Tran, D. C. (2017), FIR digital filter design using improved particle swarm optimization based on refraction principle. Soft Computing, 21(10), 2631-2642.
- [8] Pak, J. M., Kim, P. S., You, S. H., Lee, S. S., & Song, M. K. (2017), Extended least square unbiased FIR filter for target tracking using the constant velocity motion model. International Journal of Control, Automation and Systems, 15(2), 947-951.
- [9] Dash, J., Dam, B., & Swain, R. (2017), Design of multipurpose digital FIR double band filter using hybrid firefly differential evolution algorithm. Applied Soft Computing
- [10] Raj, P. J., & Vigneswaran, T. (2016, March). A paradigm of distributed arithmetic (DA) approaches for digital FIR filter. In Electrical, Electronics, and Optimization Techniques (ICEEOT), International Conference on (pp. 4668-4672). IEEE.
- [11] Pak, J. M., Ahn, C. K., Shmaliy, Y. S., Shi, P., & Lim, M. T. (2016), Switching extensible FIR filter bank for adaptive horizon state estimation with application. IEEE Transactions on Control Systems Technology, 24(3), 1052-1058.
- [12] Dwivedi, A. K., Ghosh, S., & Londhe, N. D. (2016), Low power FIR filter design using modified multi-objective artificial bee colony algorithm. Engineering Applications of Artificial Intelligence, 55, 58-69.
- [13] Aggarwal, A., Rawat, T. K., & Upadhyay, D. K. (2016), Design of optimal digital FIR filters using evolutionary and swarm optimization techniques. AEU-International Journal of Electronics and Communications, 70(4), 373-385.
- [14] Kuyu, Y. C., & Vatansever, F. (2016), A new intelligent decision making system combining classical methods, evolutionary algorithms and statistical techniques for optimal digital FIR filter design and their performance evaluation. AEU-International Journal of Electronics and Communications, 70(12), 1651-1666.

