

Analysis of FIR Filter Design Techniques

¹Kanu Priya, ²Lajwanti Singh

¹Dept. of Electronics & Communication, GGSCMT/PTU Jalandhar, Kharar, India

²Dept. of Electronics & Communication, AICE/RTU Kota, Rajasthan, India

Abstract

This paper deals with some of the techniques that are used to design FIR filters. In the paper the windowing method, the frequency sampling methods and optimal filter design methods are discussed in detail with their merits and demerits. Each method has its own advantages and disadvantages.

Keywords

FIR, FIR Filter Design Methods, FIR Filters, FIR Filter Techniques

I. Introduction

FIR filters are digital filters with finite impulse response. They are also known as non-recursive digital filters as they do not have the feedback (a recursive part of a filter), even though recursive algorithms can be used for FIR filter realization.

FIR filters can be designed using different methods, but most of them are based on ideal filter approximation. The objective is not to achieve ideal characteristics, as it is impossible anyway, but to achieve sufficiently good characteristics of a filter. The transfer function of FIR filter approaches the ideal as the filter order increases, thus increasing the complexity and amount of time needed for processing input samples of a signal being filtered.

FIR filter design essentially consist of two parts [1]:

1. Approximation Problem
2. Realization Problem.

The approximation stage takes the specification and gives transfer function through four steps. They are as follows:

- A desired or ideal response is chosen, usually in the frequency domain.
- An allowed class of filters is chosen (e.g. the length N for a FIR filters).
- A measure of the quality of approximation is chosen.
- A method or algorithm is selected to find the best filter transfer function.

Realization part deals with choosing the structure to implement the transfer function which may be in the form of circuit diagram or in the form of a program [2].

II. FIR Filter Design Methods

Most FIR filter design methods are based on ideal filter approximation. The resulting filter approximates the ideal characteristic as the filter order increases, thus making the filter and its implementation more complex.

The filter design process starts with specifications and requirements of the desirable FIR filter. Which method is to be used in the filter design process depends on the filter specifications and implementation.

There are essentially three well-known methods for FIR filter design namely:

1. The window method
 2. The frequency sampling technique
 3. Optimal filter design methods
- II.a. Window method of FIR filter design [3]

A. Window Method of FIR Filter Design

The basic idea behind the Window method of filter design is that the ideal frequency response of the desired filter is equal to 1 for all the pass band frequencies, and equal to 0 for all the stop band frequencies, then the filter impulse response is obtained by taking the Discrete Fourier Transform (DFT) of the ideal frequency response. Unfortunately, the filter response would be infinitely long since it has to reproduce the infinitely steep discontinuities in the ideal frequency response at the band edges. To create a Finite Impulse Response (FIR) filter, the time domain filter coefficients must be restricted in number by multiplying by a window function of a finite width. The simplest window function is the rectangular window which corresponds to truncating the sequence after a certain number of terms.

This technique is based on designing a filter using well-known frequency-domain transition functions called windows. Rectangular windowing the time domain will result in a frequency spectrum with the width of the pass band close to the desired value but with side lobes appearing at the band edges.

Other windows, like the Blackman, have better stop-band attenuation and group delay but a wide transition band.

To suppress the side lobes and make the filter frequency response approximate more closely to the ideal, the width of the window must be increased and the window function tapered down to zero at the ends. This will increase the width of the transition region between the pass and stop bands, but will lower the side lobe levels outside the pass band [3].

B. Frequency Sampling Method for FIR Filter Design

The frequency-sampling method for FIR filter design is perhaps the simplest and most direct technique imaginable when a desired frequency response has been specified. It consists simply of uniformly sampling the desired frequency response, and performing an inverse DFT to obtain the corresponding (finite) impulse response [Ref 4]. The results are not optimal, however, because the response generally deviates from what is desired between the samples. When the desired frequency-response is undersampled, which is typical, the resulting impulse response will be time aliased to some extent. It is important to evaluate the final impulse response via a simulated DTFT (FFT with lots of zero padding), comparing to the originally desired frequency response.

C. Optimal Filter Design Methods

In Optimal Filter Design Method various methods are used to design the filter coefficients again and again until a particular error is minimized [1]. The various methods are as follows:

1. Least square method.
2. Equiripple method.
3. Maximally flat.
4. Generalised equiripple.
5. Constrained band equiripple.

In Least square method there is no constraint on the response between the sample points, and poor results may be obtained. The frequency sampling technique is more of an interpolation method rather than an

approximation method. Least square method [5-6], controls the response between the sample points by considering a number of sample points larger than the order of the filter. As the energy of the signal is related to the square of the signal, a squared error approximation criterion is appropriate to optimize the design of the FIR filters.

The Remez/Parks McLellan method produces a filter which just meets the specification without overperforming. Many of the window method designs actually perform better as you move further away from the passband, and means they are using more filter coefficients than they need. This means that they are using more filter coefficients than they need.

To design equiripple linear-phase FIR filters with linear constraint method is carried out by using the Remez exchanges algorithm. A technique is derived to convert a linearly constrained problem to an equivalent unconstrained one. The key step is to modify the original desired frequency response such that the constraint is reduced to a null constraint for the new target frequency response. Then the filter constrained by such constraint can be designed without any constraint by a set of bases obtained by transforming the original basis by the null space [7].

III. Merits and Demerits

A. Merits and Demerits of Windows Method

The simplest technique used to develop digital FIR filters is windowed filters. The use of windows often involves choosing the lesser of two evils. Windowed filters are easy to use and scalable and can be computed on the fly by the DSP. This latter point means that a tunable filter can be designed with the only limitation on corner frequency resolution being the number of bits in the tuning word [2]. The availability of well defined equations for calculating window coefficient has made this method preferable.

It offers very little design flexibility especially in low pass filter design. Kaiser window offers very low order to meet given specification. The best digital filter design results comes for using the Kaiser window from the windowing design technique, which has parameter s that allows adjustment of the compromise between the overshoot reduction and transition region width spreading. But when Kaiser window is compared with optimal filter design method, equiripple filter design found to be most suitable and optimized method to meet given specification.

B. Merits and Demerits of Frequency Sampling Technique

1. Unlike the window method, this technique can be used for any given magnitude response.
2. This method is useful for the design of non-prototype filters where the desired magnitude response can take any irregular shape.

The frequency response obtained by interpolation is equal to the desired frequency response only at the sampled points. At the other points, there will be a finite error present.

As we have mentioned above, the design process starts with the specification of desirable FIR filter.

C. Merits and Demerits of Optimal Filter Method

Optimal filter design techniques give best filters for given length of the FIR filter, but these techniques are more complex.

IV. Conclusion

Each of the given methods has its advantages and disadvantages.

Thus, it is very important to carefully choose the right method for FIR filter design. Due to its simplicity and efficiency, the window method is most commonly used method for designing filters. The sampling frequency method is easy to use, but filters designed this way have small attenuation in the stopband.

The paper briefly describes the various technique used to design FIR filter and there merits and demerits. The major advantage of window technique is its simplicity. The availability of well defined equations for calculating window coefficient has made this method preferable. But it offers very little design flexibility especially in low pass filter design. Kaiser window offers very low order to meet given specification. The best digital filter design results comes for using the Kaiser window from the windowing design technique, which has parameter s that allows adjustment of the compromise between the overshoot reduction and transition region width spreading. But when Kaiser window is compared with optimal filter design method, equiripple filter design found to be most suitable and optimized method to meet given specification.

References

- [1] Prof. Gopal S.Gawande, "Performance analysis of FIR Digital Filter Design Techniques", International Journal of Computing and Corporate Research, Vol. 2, Issue 1 January 2012.
- [2] Arojit roy chaudhary, "FIR filter design tech", M.Tech. credit seminar report, Electronic Systems Group, EE Dept, IIT Bombay, submitted November 2002.
- [3] Digital Signal Processing by Emmanuel Ifeakor, Barrie Jervis.
- [4] Julius O. Smith, "Spectral Audio Signal Processing".
- [5] J.G. Proakis, D.G. Manolakis, "Digital Signal Processing-Principles", Algorithms and Applications New Delhi: Prentice-Hall, 2000.
- [6] Magdy T. Hanna, "Design of Linear Phase FIR Filters with a Maximally Flat Passband", IEEE Trans. Circuits Syst. II, 43 (2), pp. 142 – 147, 1996.
- [7] Soo-Chang Pei, Peng-Hua Wang, "Design of Equiripple FIR Filters With Constraint Using a Multiple Exchange Algorithm", IEEE Transactions on Circuits and Systems—I: Fundamental Theory and Applications, Vol. 49, No. 1, January 2002.



Kanu Priya had completed her Bachelor's degree in Electronics and Communication Engineering from Bhai Gurdas Institute of Engineering & Technology, Sangrur, India in 2006, and Pursuing the Masters of Engineering degree in Electronics and Communication Engineering from National Institute of Technical Teachers' Training & Research, Punjab University, Chandigarh, India.

She is an Assistant Professor in the Department of Electronics & Communication Engineering,, Guru Gobind Singh Collage of Modern Technology, Kharar, Punjab, India. Her current research and teaching interests are in Digital Signal Processing, Signal Systems and VLSI. Ms. Kanu Priya is life member of ISTE.