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FIR Digital Filter and Its Designing Methods

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ABSTRACT

In this paper discuss about the digital filter. In the beginning, the windowing method and the frequency sampling methods are discussed in detail with their merits and demerits. Digital FIR filter plays very important role in communication channels. The digital filters are easily designed and also easy to use in several of signal filtering applications. In this paper discuss about the methods of FIR filter. The choice of method to design the filter depends heavily on the decision of designer whether to compromise accuracy of approximation.

Keywords— FIR,WLS ,DFT,IDFT,GWO

I. INTRODUCTION

Digital filters are fundamental building blocks for signal processing applications. A filter is a frequency-selective device. It passes signal components at some frequencies and attenuates signal components at other frequencies. Digital filters are essential parts of many digital signal processing systems, with control systems, systems for audio and video processing, communication systems and systems for medical applications. Due to the increasing number of applications involving digital signal processing and digital filtering the variety of requirements that have to be met by digital filters has increased as well. Accordingly, there is a need for supple techniques that can design digital filters satisfying sophisticated specifications. Digital filters are classified as Recursive and Non-Recursive filters.

Digital Filters

Recursive Filter   Non-Recursive Filter

The response of non-Recursive or FIR filters depend only upon Present and previous input of signal. The response of Recursive or IIR filters depends on present inputs, past inputs and previous outputs also.

For designing a filter we need a set of specifications either in discrete time or frequency domain. Digital filters operate on discrete time signals and consists of multipliers, adders and delay elements (shift registers) implemented in digital logic.

II. FIR FILTER DESIGN

FIR filters are filters having a transfer function of a polynomial in z - and is an all-zero filter in the sense that the zeroes in the z-plane determine the frequency response magnitude characteristic. In the FIR system, the impulse response sequence is of finite duration, i.e., it has a finite number of non-zero terms. FIR filters have the

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following advantages:-

• They can have an exact linear phase.

• They are always stable.

• The design methods are generally linear.

• They can be realized efficiently in hardware.

• The filter start up transients has finite duration.

The z transform of a N-point FIR filter is given by

$$H(z) = \sum_{n=0}^{N} h_n z^{-n}$$

Here h_n is the impulse response of the filter.

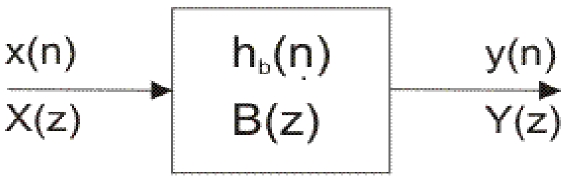

Fig 1: FIR filter

There are three methods for FIR filter design namely:

- The window method.
- The frequency sampling technique.
- Optimal filter design methods.

2.1 The Window Method

In this method, the desired frequency response specification $H_d(w)$, corresponding unit sample response $h_d(n)$ is determined using the following relation

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(w)e^{jwn}\,dw$$

Where $H_d(w) = \sum_{n=-\infty}^{\infty} h_d(n)e^{-jwn'cc]p}$

In general, unit sample response $h_d(n)$ obtained from the above relation is infinite in duration.

2.2 The Frequency Sampling Technique

In this method, the desired frequency response is provided as in the previous method. The given frequency response is sampled at a set of equally spaced frequencies to attain N samples. Thus, sampling the continuous frequency response $H_d(w)$ at N points essentially gives us the N-point DFT of $H_d(2pnk/N)$. Therefore by using the IDFT formula, the filter coefficients can be calculated using the following formula

$$h(n) = \frac{1}{N}\sum_{n=0}^{N-1} H(k)e^{j\left(\frac{2\pi n}{N}\right)k}$$

2.4 Optimal filter design methods

Many methods are present under this category. The basic idea in each method is to design the filter coefficients again and again until a particular error is minimized. The various methods are as follows:

- Least squared error frequency domain design
- Weighted Chebyshev approximation
- Nonlinear equation solution for maximal ripple FIR filters
- Polynomial interpolation solution for maximal ripple FIR filters

III. L2-NORM MINIMIZATION METHOD

L2-norm method, also known as the LSM (Least Square Minimization Method), is widely used for parameters estimation. In this method, a discrete-time transfer function is assumed and an error function is formulated on the basis of some desired amplitude and phase response. A norm of the error function is then minimized with respect to the transfer function coefficients. As the value of the norm approaches zero, the resulting amplitude or phase response approaches the desired amplitude or phase response. It is an iterative method and, as a result, it usually involves a large amount of computation.

The weighted least square error is represented by equation as

$$\varepsilon_{2} = \int_{0}^{\Pi} W(\omega)|H_d(\omega) - H(\omega)|^2 \, d\omega$$

Here

$H_d(\omega)$: The actual amplitude response.

$H(\omega)$: The ideal amplitude response.

$W(\omega)$: Nonnegative weighting function.

The weighting function can be used to allocate more importance to precise parts of the frequency response. For example, it is common to weight the stop band more heavily than the pass band. Once the length N and the type of the filter chosen, the goal is to find filter coefficients h(n) that minimizes ε_{2} .

IV. LITERATURE REVIEW

L. Litwin mentioned the concept of digital filter [3] as simply a discrete time, discrete amplitude convolver. Digital filters are basic building blocks in many digital signal processing systems. They have wide range of applications in communication, image processing, pattern recognition, etc. There are two major types of digital filters, that is, finite impulse response (FIR) filters and infinite impulse response (IIR) filters depending on the length of the impulse response.

Lawrence R.Rabiner in [4] discussed several techniques including windows method, frequency sampling method and equiripple designs for designing FIR filters. With the help of these techniques and filter characteristics, ease of design and methods of realization have been compared. Difficulty in calculation of Bessel function and Fourier series coefficient was the major disadvantage in window method technique. But transition bandwidth in the case of Frequency Sampling technique has been found to be 3/4th to that of Kaiser Window while equiripple design had smallest transition bandwidth. But if the ease of design parameter is

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considered, window method tends to be relatively easy to be used. The choice of technique depends heavily on the decision, whether to compromise accuracy of approximation or ease of design.

Thomas W. Parks and James H. Mcclellan presented Chebyshev Approximation method for the design of linear phase FIR filter, especially for the very long filters.

Lee, M. Ahmadi designed FIR filter using Genetic Algorithm (GA) in 1999 for obtaining optimal solutions which attracted most of the attention. Filters with binary, integer or real coefficients could be easily handled by the proposed method. The genetic algorithm comprised of three genetic operations, namely reproduction, crossover and mutation. These three operations were applied again and again and through natural selection and genetic operators, mutation and recombination, chromosomes with better fitness were found. Natural selection guaranteed that chromosomes with the best fitness would propagate in the future populations. It was tested for the design of filters with different amplitude and phase specifications.

Aimin Jiang and Hon Keung Kwan proposed a technique in paper [5] to design a Sparse FIR filter by

reducing the number of nonzero-valued filter coefficients by employing weighted least-square (WLS) approximation error constraints on the frequency domain. Both linear and non-linear phase FIR filter can be designed under this method. It is inspired by iterative shrinkage/thresh holing (IST) algorithm. The proposed design successively transforms the original non-convex problem to a series of sub problems which are simpler.

Niraja Singh and Pushpraj Tanwar design a Sparse FIR filter in paper [6] by reducing the number of nonzero valued coefficients so that the implementation complexity can be reduced. An efficient algorithm is presented in this research work. To design a Sparse FIR filter an iterative shrinkage/thresh holding (IST) based technique is employed. Many of the design approaches consider an FIR filter designs which are subjected to peak error constraints imposed on magnitude response. However, in this paper weighted least square (WLS) approximation error constraint imposed on

Frequency domain is considered on the designing of the filter.

Sabah M. Ahmed used the design technique that was implemented using Matlab in a form of interactive toolbox for FIR filter design using GA [7]. Simulation results for the filter design using GA were compared, and it was found that GA gave us the exact cut off frequency. Also the ripples in the pass band and in the stop band regions were attenuated successfully, but the problem was that the GA was inefficient in determining the global optimum in terms of convergence speed and solution quality.

Sangeeta Mandal et al. used Craziness based Particle Swarm Optimization Technique (CRPSO) [8],[9],[10], in 2012, 2015. CRPSO technique tried to find the best coefficients that closely match the ideal frequency response. A novel Craziness based Particle Swarm Optimization (CRPSO) technique was applied to the solution of the constrained, multimodal, non-differentiable, and highly nonlinear FIR band stop filter design problem to obtain the optimal filter coefficients. With almost same level of the transition width, the CRPSO produced the highest stop band attenuation and the lowest stop band and the pass band ripples as compared to those of PM algorithm, RGA and conventional PSO. Results obtained showed that the CRPSO did not show the problem of premature convergence .Then same technique was applied to FIR high pass filter and FIR low pass filter by SANGEETA MANDAL et al.,[8], [9] to obtain better results in comparison to genetic algorithm. This technique outperformed in the optimal characteristics of frequency spectrums.

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V. CONCLUSION

The report has described the various techniques involved in the design of FIR filters. Every method has its own advantages and disadvantages and is selected depending on the type of filter to be designed.The window method is basically used for the design of prototype filters like the low-pass, high-pass, band-pass etc. They are not very suitable for designing of filters with any given frequency response. On the other hand, the frequency sampling technique is suitable for designing of filters with a given magnitude response.Digital filter plays a very significant role in different digital signal processing applications. The digital filters are simply designed and also easy to use in numerous of signal filtering applications. The preference of technique to design the filter depends heavily on the decision of designer whether to cooperation accuracy of approximation.

An FIR filter can be intended using weighted least square error (WLS) method because WLS give better accuracy.

VI. REFERENCES

[1] Antoniou, "Digital signal processing: signals, systems and filters, "McGraw-hill ,New York,2005.

[2] E. C. Ifeachor and B. W. Jervis , "Digital Signal Processing," A Practical Approach, Prentice Hall, 2002.

[3] L. Litwin , "FIR and IIR digital filters," IEEE Potentials 0278-6648, pp.28–31, 2000.

[4] Lawrence R.Rabiner, "A new optimal window," IEEE Trans. Signal Processing, vol.39, pp. 1753–1769, Aug. 1991.

[5] A. Jiang and H. K. Kwan, "WLS design of sparse FIR digital filters," IEEE Trans. Circuits System .I, Reg. Papers , vol.60, pp. 125-135, Jan. 2013.

[6] Niraja Singh and Pushpraj Tanwar, "FIR (sparse) filter design using IST algorithm," IJIET, vol. 2, Issue 2, pp.386- 394, April 2013.

[7] Sabah M. Ahmed, "Design of FIR Filters With Arbitrary Amplitude And Phase Specifications Using Genetic Algorithm, "IEEE Potentials 0-7803-8294-3, pp. 648-651, 2004.

[8] E. S. Group, "FIR Filter Design Techniques," no. 1, pp. 1–12.

[9] Y. Kumar, "Design of Optimum Digital FIR Low Pass Filter Using Hybrid of GA & PSO Optimization," vol. 3, no. 9, pp. 685–692, 2013

[10] S. Chakraborty, "Design and Realization of Digital FIR Filter using Dolph-Chebysheb Window," vol. 4, no. 7, pp. 987–996, 2013.

[11] P. Yadav and P. Kumari, "Review Paper on FIR Filter Design," Int. J. Innov. Res. Technol., vol. 1, no. 12, pp. 593–596, 2015.

[12] N. Mishra, "Literature Review on FIR Filter Optimization," vol. 4, no. 1, 2015.

[13] S. A. Al-taie, "International Journal of Emerging Technologies in Computational and Applied Sciences

[14] (IJETCAS)," Int. J. Emerg. Technol. Comput. Appl. Sci. (IJETCAS), pp. 289–293, 2013.

[15] D. Mandal, S. P. Ghoshal, RajibKar and Sangeeta Mandal, "Craziness based Particle Swarm Optimization algorithm for FIR band stop filter design," Swarm and Evolutionary Computation,(Elsvier),vol. 7, pp. 58–64,2012.

[16] D. Mandal, S. P. Ghoshal, RajibKar and Sangeeta Mandal, “Design of optimal linear phase fir high pass filter using craziness based particle swarm optimization technique ,”Computer and Information Sciences, (Elsevier), vol. 24, pp. 83-92, 2012.

[17] Borah, Amlan Deep, Abir J. Mondal, Deboraj Muchahary, and Alak Majumder. "FIR low pass filter design using Craziness base Particle Swarm Optimization Technique," International Conference in Communications and Signal Processing (ICCSP), pp. 0870- 0874, 2015.

[18] D. Mandal, S. P. Ghoshal, RajibKar and Sangeeta Mandal, “Novel Particle Swarm optimization for Low Pass FIR Filter Design, ”WSEAS Transactions on Signal Processing ,Issue 3, vol.8, July, 2012.

[19] Rani, Surekha, Balwinder Singh Dhaliwal, and Sandeep Singh Gill. "Differential Evolution Biogeography Based Optimization for Linear Phase Fir Low Pass Filter Design," Chemical and Process Engineering Research 31 , pp. 81-88 , 2015.

[20] H. D. Helms, “Digital filters with equiripple or minimax responses,”IEEE Trans. Audio Electroacoustic, vol. AU-19, pp. 87–94, Mar. 1971.