DESIGN AND DEVELOPMENT OF HIGHLY EFFICIENT DIGITAL FILTER

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Abstract: In many applications for every signal it is desired to design a filter for specific frequency so it is very time consuming to develop a different filter for every desired selective frequency. For overcome this problem, the current designing of filter is quite easy to develop a frequency selective filter with few changes and also to separate two or more. One more easy benefit of this filter designing is to select one or more frequencies at a time present in the composite signal. By using an above MATLAB programs determine the coefficients and plot the frequency response of FIR filter to meet the specifications

Keywords: Fast Fourier Transform, Digital Signal Processing, Electrocardiogram, Finite Duration Unit Pulse Response, Infinite Duration Unit Pulse Response

1. INTRODUCTION

1.1 Digital Filters

In DSP the lots of techniques for designing of the digital filters for receiver process according to the filter specifications. Basically filter is a device to extract the information from the noisy signal. Digital filters are placed an important role in DSP, compared with analog filters and it is also used in every electronic device such as mobile phone, speakers and setup boxes. Digital filters are classified into two type finite duration unit pulse response (FIR) & infinite duration unit pulse response (IIR) filters. By using FIR digital filters, the impulse response is finite, so it can be used for Fast Fourier transform (FFT) algorithm to achieve the filtered signal. To use FIR filters in communication & video systems high performance in speed, area & power consumption is required [1].

The response of the FIR filter depends only on the present and past input Samples. The FIR filters circuit must be able to operate at high samples rates. It can be designed by impulse response truncation, windowing design method and by optimal filter designing technique. Practical FIR designs typically consist of filters that meet certain design specifications i.e., transition width and maximum passband/stopband ripples.

The main advantages of the FIR filters fir filters with exactly linear phase characteristics. It can be realized in both recursive and non-recursive structures. The FIR filters are free of limit cycle oscillations, when implemented on a finite word length digital system [3].

The disadvantage of FIR filters is that they often require a much higher order filter than IIR filter to achieve a desired performance. The application of FIR filters preconditioning, band selection, and low passes filtering. In digital system there is the requirement of multi rate digital signal processing when more than one sample rate is required. Decimation and interpolation are the two basic operations in multi rate digital signal processing. Decimation is used for reducing the sampling rate. It is process of down sampling followed by a mitigate aliasing filter [5].

1.2 Advantages of FIR Filter over IIR Filter

(f) FIR filter is finite
(g) FIR filter is non-recursive
(h) Impulse response of FIR filter eventually reaches zero
(i) FIR filter is not used in classical analog filters
(j) FIR filter has linear phase
(k) FIR filter is stable
(l) FIR filter only depend on inputs
(m) FIR filter consist of only zeros [6].

1.3 The Basic Steps of Design of Digital Filter

The designed traditionally process of digital filter can be divided into three steps:

1) According to the actual needs, determines performance and requirements of the filter.

2) Use a system function (namely transfer function), which cause and effect is stable, to approach the performance requirements. This function can be divided into two categories: namely IIR transfer function and FIR transfer function.

3) With a finite precision operation to achieve the transfer function.

The substance of FIR filters designed is to determine constants of the transfers sequence or impulse response which can meet requirements. Design methods are main window function method, frequency sampling method and the Chebyshev approximation method, etc. [10]

4) Filter specification: this may include stating the type of filter ex. Low pass filter, desired amplitude or phase responses and the tolerance (if any), we are prepared to accept, the sampling frequency and the word length to input data.

5) Coefficient calculation: at this step, we determine the coefficients of the transfer function H(z) which will satisfy the specifications given in (1). Our choice of coefficient calculation method will be influenced by the several factors, the most important of which are the critical requirement in step (1).

6) Realization: This involves the converting the transfer functions obtained in (1) into a suitable filter network or structure.

7) Analysis of finite word length effects: Here, we analyze the effect of quantizing the filter coefficients and the input data as

well as the effect of carrying out the filtering operation using fixed word lengths on the filter performance.

8) Implementation: This involves the production of the software code and/ or hardware and performing the actual filtering.

2. RELATED WORK

Ch. Sravani et al., (2016) [1] A digital filter plays an important role in today's world of communication and computations. The most common purpose for filtering is to remove the noise from desired signal. FIR filter are widely used over IIR because of its stability, linear phase characteristics. FIR filters from the basis of wireless system in many applications. The main objective of this paper is to design the minimum order FIR filter for the required specifications using different windowing techniques and to implement the 1-D and 2-D FIR filters using Simulink tool of MATLAB. The results of 1-D FIR is compared with the 2-D FIR to assess the performance of the filters.

Pankaj R. Ambilduke et al., (2016) [2] in many practical applications of DSP, there is a problem of changing the sampling rate of a signal, either increasing it or decreasing it by some amount. For example, Telecommunication system transmits and receives different types of signals (e.g. fax, speech, video, etc.), so there is a requirement to process the various signals at the different rates with corresponding bandwidth of the signals. Digital filters such as IIR filter, FIR filter and CIC filter is designed and taking their performance in case of magnitude response, step response, impulse response, pole-zero plot, filter coefficients, storage requirements, hardware requirements, number of stages and simulated waveforms for same input specifications. Finally compare them and discuss the advantages and limitations of these filters. These filter structures designed in the Simulink model in Matlab 2012a environment.

Divya Goyal and Ritesh Goel, (2015) [3] the aim of this paper is to design FIR filter of the order of 50 using various Window function Methods. This paper presents the importance of filter in signal processing. Digital filter are of two types (1)FIR (2)IIR.There are Various methods for Filter designing, this paper discuss the Window Method. Low pass FIR filter distinguishing Hamming, Kaiser, Blackman, Tukey and Rectangular Window Function Methods are presented. Design of FIR filer is done in MATLAB by FDATOOL. Low pass filter is designed with Sampling frequency 5000 Hz and Cut-off frequency 1000 Hz. Magnitude and Phase Responses of low pass Filter using various window techniques are demonstrated. Authors found that Filter designing by window method is easy and fast.

Prachi Kamble et al., (2015) [4] FIR Filter is very important type of Digital Filters which is a vital element in Digital Signal Processing. In this paper authors design FIR Filter structure and implement it for audio application. FIR filter is a type of digital system that filters discrete-time signal and the main signal, main objective performing frequency domain filtering by processing sample data. It is use in various applications like Speech recognition, Speech synthesis, digital audio, Telecommunication, seismic signal processing (noise elimination), and several other areas of signal processing. This FIR Filter is designed with the help of MATLAB SIMULINK (Win_2012) and XILINK System Generator (ISE_Win_14.2).

Suraj R. Gaikwad et al., (2014) [5] in this paper authors propose the design and development of digital filters for audio application using Xilinx System Generator. Digital Filters are important elements in Digital Signal Processing (DSP). In this case, authors can decimate the high frequency signal (128 KHz) up to respective low frequency signal (3.4 KHz). Also authors introduced some designs of digital filters using MATLAB Simulink model and Xilinx System Generator and compare their respective results in case of computation and storage requirement. The proposed technique is centered on the FIR filtering technique for decimation of audio signal.

C. Uthayakumar and B. Justus Rabi (2014) [6] Windowing methods are used for the design of Finite Impulse Response (FIR) filter. This paper concerns with the design and implementation of FIR filter using a rectangular window method. A window is a finite array consisting of coefficients selected to satisfy the desirable requirements. Simulations are carried out for the FIR filter using rectangular window and the best order of FIR filter is proposed using MATLAB. For the filter design the cut-off frequency is desired to be fixed and simulations are made for the FIR filter with order 4, order 8, order 12 and order 18, for low pass filter, high pass filter, band pass filter and band stop filter. Finally the results are analyzed for its performance.

Fei Zhang et al., (2014) [7] To make automatic external defibrillators (AEDs) easy to use by the public who is not familiar with emergency treatment and electrocardiogram (ECG) analysis, it is critical to have an accurate shock able rhythm recognition algorithm. This paper presents a novel com positive algorithm by combining a slope variability analyzer with a band-pass digital filter so as to accurately distinguish shock able rhythms from non-shock able rhythms for automatic external defibrillators (AEDs). A total of 35 ECG records from the widely recognized Creighton University Ventricular Tachyarrhythmia Database (ClJDB) were used to test the performance of the proposed algorithm. The obtained sensitivity of 94.2% and the specificity of 96.6% both satisfy requirements by the AHA rules on the arrhythmias detection for AEDs, and show a higher performance comparing with the previous HILB algorithm and the slope variability method only. As a conclusion, the proposed com positive algorithm would potentially provide a useful tool for AED systems with a higher accuracy and lower computation requirements.

Aye Than Mon et al., (2014) [8] deals with band pass filter which is located between up converter and modulator of uplink model for C band small satellite communication system that work digitally. In this paper, FIR equiripple method is used. The selection of FIR equiripple filter depends on the nature of the problem and specifications of the desired frequency response. The main theme of designing a digital FIR filter is to provide the better settlement solution, to improve an efficiency of the desired signal of the system and to allow adjustment of the compromise between the over shoot reduction and transition region width for practical application of the small satellite uplink system. The realization structure of this filter with a specific and symmetric filter coefficient is analyzed and the symmetric coefficients of the filter structure are that this filter is stable, it is also linear and it has a constant group delay. And then the magnitude response and phase response of this filter are analyzed and the simulation results are also described using FDA tool that is one of the Computer Aided Design tool available with MATLAB which enables design of the digital filter blocks faster and more accurate. With the performance evaluation of the equiripple filter design, the output results are completely suitable for the proposed small satellite uplink model and so Equiripple filter design is found to be the most suitable and optimized method to meet the desired specification. The second part of the paper is to analyze the antenna deign analysis. A multiple-feed system or a beam former is used to generate several beams from this common aperture.

Alia Ahmed Eleti and Amer R. Zerek (2013) [9] The work reported in this paper deal with of Finite Impulse Response FIR digital filter design by using window method. The window method is easiest to design FIR, but lacks flexibility especially when the passband and stopband ripples are different. The digital filter is used to filter discrete time signals with the ability to modify the frequency response of the filter at any time and it used in many application such as data compression, biomedical signal processing, communication receivers, etc. Using MATLAB package software programs are developed for designing FIR digital filter and good results are obtained.

Rakhi Thakur and Kavita Khare (2013) [10] Signal processing ranks among the most demanding applications of digital design concepts. It is a mature technology domain wherein the demands for enhanced performance and reduced resource utilization have risen exponentially over the years. Field Programmable Gate Array (FPGA) design technology has becoming the preferred platform for evaluating and implementing signal processing algorithms. The advantages of the FPGA approach to digital filter implementation include higher sampling rates than are available from traditional DSP chips, lower costs than an application specific integrated circuit (ASIC) for moderate volume applications, and more flexibility than the alternate approaches. Since many current FPGA architectures are in-system programmable, the configuration of the device may be changed to implement different functionality if required. This paper describes an approach to the implementation of digital filter based on field programmable gate arrays (FPGAs) which is flexible and provides performance comparable or superior to traditional approaches, low power, are efficient re-configurable digital signal processing architecture that is tailored for the realization of arbitrary response Finite impulse response (FIR) filters.

3. PROPOSED WORK

Toolbox of MATLAB signal processing provides various window functions, designing functions of filter and realizable functions of filter. The methods of Window function is the most commonly used method which design FIR digital filter. It has very important function in the design of FIR digital filters, properly choose window function, can improve the performance of design of the digital filter.

4. RESULTS AND DISCUSSION

The fact that well defined equations are often available for calculating the window coefficients has made this method successful. Having this it makes simply possible to design FIR band pass filter for different windowing techniques as Hamming, Hanning, Rectangular, blackman window for the range of different frequencies up to MHz.

Phase and magnitude response is obtained with the help of filter visualization tool for the windows mentioned above and then found the efficient values of frequencies which has to be selected or extract from the composite signal. This is very useful for the type of application where a selective frequency is required for further use or to separate different desired frequencies from a composite signal.

Table 4.1 Efficiency of the signal which is filtered for different range of frequencies:

Range of frequency	**Composite signal**	**Desired frequency to be filtered**	**Filtered frequencies**	**Difference (Hz)**
Up to Hz	200, 300, 400, 500	200, 300	200.3, 198	+0.3, -2
For kHz	2, 10 ,15 , 18	10, 15	9.976, 1.514e+4	-0.024, +140
For MHz	0.2, 1, 1.5, 3	1, 3	9.915e+5, 3.004e+6	-8500, +4000

With the help of table it is cleared that designed FIR filter is very much suitable for low frequency and for very high frequency it is going some less efficient. One thing that we observe is that the difference between outputs of all windows is only having different magnitude of desired frequency components.

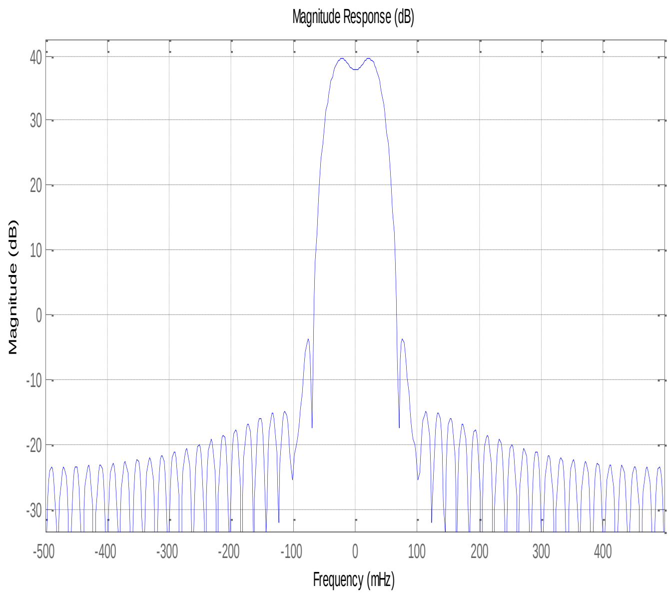
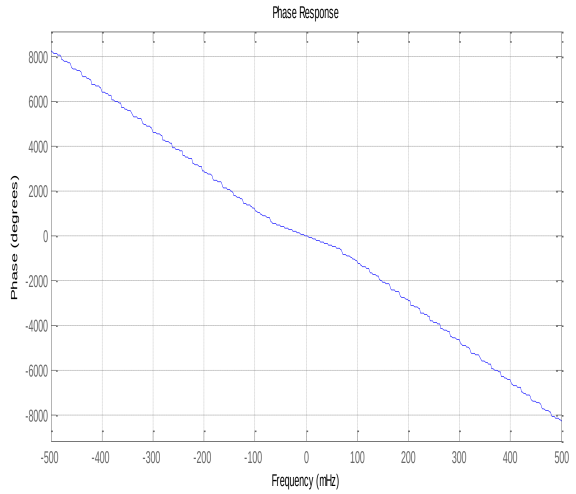


Figure 4.1: Magnitude response and Phase response of hamming window



5. CONCLUSION AND FUTURE SCOPE

The digital filter is used to filter discrete time signals with the ability to modify the frequency response of the filter at any time and it used in many application such as data compression, biomedical signal processing, communication receivers, etc. Using MATLAB package software programs can be developed for designing FIR digital filter and good results are obtained. Toolbox of MATLAB signal processing provides various window functions, designing functions of filter and realizable functions of filter. The method of Window function is the most commonly used method which design FIR digital filter. It has very important function in the design of FIR digital filters, properly choose window function, can improve the performance of design of the digital filter, or under meeting the requirement of design circumstances reducing the order of FIR digital filters. Designed FIR filter is very much suitable for low frequency and for very high frequency it is going some less efficient.

Existing work can be extended by designing FIR filters with order 4, order 8, order 12, and performance evaluation for band pass filter.

REFERENCES

1. Ch. Sravani, L. Saranya, Ch. Rojarani, P. Suresh Kumar, “Design And Implementation Of Digital Fir Filter & Its Performance Assessment Using Matlab Simulink”, International Journal of Electronics and Communication Engineering. ISSN 0974-2166 Volume 9, Number 1 (2016), pp. 7-12.

2. Pankaj R. Ambilduke, Manish Kumar Gurjar, “PERFORMANCE COMPARISON FOR EFFICIENT DIGITAL DECIMATOR FILTER STRUCTURES”, IJRET: International Journal of Research in Engineering and Technology eISSN: 2319-1163 | pISSN: 2321-7308.

3. Divya Goyal and Ritesh Goel, “Review of Low-Pass FIR Filter Design Using Window Method”, International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE) Volume 4, Issue 7, July 2015.

4. Prachi Kamble, Nayan Deote, Nitesh Wanjari, “Implementation of FIR Filter Structure for Audio Application Using Xilinx System Generator”, nternational Journal of Advanced Research in Computer Science and Software Engineering.

5. Suraj R. Gaikwad, Gopal S. Gawande, “Design and Implementation of Efficient FIR Filter Structures using Xilinx System Generator”, International Journal of scientific research and management (IJSRM).

6. C. Uthayakumar and B. Justus Rabi, “Design of FIR Filter using Window Method”, IPASJ International Journal of Electronics & Communication (IIJEC)

7. Fei Zhang, Pengye Li, Fan Jiang and Dakun Lai, “A Shockable Rhythm Detection Algorithm for Automatic External Defibrillators by Combining a Slope Variability Analyzer with a Band-Pass Digital Filter”, 2014 IEEE Workshop on Electronics, Computer and Applications.

8. Aye Than Mon, Zaw Min Naing, Chaw Myat New, Hla Myo Tun, “Development Of Digital Band Pass Filter And Antenna Design For C Band Small Satellite Uplink Model”, INTERNATIONAL JOURNAL OF SCIENTIFIC & TECHNOLOGY RESEARCH VOLUME 3, ISSUE 7, JULY 2014

9. Alia Ahmed Eleti and Amer R. Zerek, “FIR Digital Filter Design By Using Windows Method With MATLAB”, 978-1-4799-2954-2/13/$31.00 ©2013 IEEE

10. Rakhi Thakur and Kavita Khare, “High Speed FPGA Implementation of FIR Filter for DSP Applications”, International Journal of Modeling and Optimization, Vol. 3, No. 1, February 2013.