Design of FIR Filter using Window Method

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ABSTRACT

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.

Keywords: FIR filter, matlab, order, windowing

1. INTRODUCTION

In signal processing applications, the impulse response of Finite Impulse Response (FIR) filter is of finite duration because it settles to zero in finite time. This means that the impulse response of an Nth order discrete-time FIR filter exist for N+1 samples and then settles to zero. FIR filters can operate with discrete-time or continuous-time digital and analog signals [1]. FIR filter require no feedback which means that any rounding errors are not compounded by summed iterations, and this system can be easily implemented. The same relative error occurs in each calculation and is inherently stable, since the output of the filter is sum of finite number of finite multiples of input values. The system is easily designed to be linear phase by making the coefficient sequence symmetric. This property is desired for phase sensitive applications such as data communications, crossover filters and mastering [2]. FIR filters are very much used in Digital Signal Processing (DSP) applications, because their characteristics behavior is in linear phase and the feed forward implementation is useful in building stable high-performance filters. Figure 1 and 2 illustrates the direct-form FIR filter and transposed-form FIR filter implementation respectively. Direct-form and transposed-form FIR filter have similar complexity in hardware. But because of the high performance and power efficiency transposed-form FIR filters is more used than the direct-form FIR filter.

The operation of the direct-form FIR filter is described by the following equation, and the equation defines the output sequence y[n] in terms of the input sequence x[n].

\[ y[n] = h_0 x[n] + h_1 x[n-1] + h_2 x[n-2] + \ldots + h_{N-1} x[n-N+1] \]  

where, \[ x[n] \] is the input signal, \[ y[n] \] is the output signal, and \[ h_i \] is the filter coefficient. The following are the advantages of FIR filter over IIR filter [3].
2. RELATED WORKS

In this section, we review the prior work on the various design and implementation of FIR filters. Saeed et al [4] described the guidelines for a moving average filter based phase-locked loop, which can act as an ideal low-pass filter. The system is compared with different moving average filter based phase-locked loop and its tuning approach is evaluated. The control parameters are designed with two systematic methods such as proportional integral type loop filter in the phase-locked loop and using a proportional integral derivative type loop filter. Chao et al [5] developed a dynamic soft-sensing model combining finite impulse response and support vector machine to describe dynamic and nonlinear static relationships. The model parameters are then estimated within a Bayesian framework. The evaluation result from both the simulated and the industrial case describes the superiority to conventional static models in terms of dynamic accuracy and practical applicability. Pavel et al [6] introduced decomposition filter banks based on narrow-band linear-phase finite impulse response (FIR) filters consisting of inner and lateral FIR filters. The inner filters are optimal narrow band pass FIR filters based on isosextremal polynomials. The inner filters are supplemented by lateral narrow-band low and high pass FIR filters. This enables flexibility in the resulting frequency response of the filter bank. Keshab et al [7] presented a low-complexity algorithm and architecture to compute power spectral density using the Welch method. The Welch algorithm provides a good estimate of the spectral power at the cost of high computational complexity. A new approach is proposed to reduce the computational complexity of the Welch power spectral density computation. The architecture not only consumes less energy compared to the original method but also reduces the latency for 8 overlapping segments. Saha et al [8] discussed the opposition-based harmony search for the optimal design of linear phase FIR filters. The original harmony search algorithm is chosen as the parent one, and the opposition-based approach is applied. Random population is chosen during initialization and fitter one is selected as priori guess. In harmony memory, each solution passes through memory consideration rule, pitch adjustment rule, and then opposition-based re-initialization generation jumping. This gives optimum result corresponding to the least error fitness in multidimensional search space of FIR filter design. Chia et al [9] presented a practical method for designing fixed-point FIR filters. The method takes both the filter’s magnitude response and its hardware cost into consideration in the design process. The method constructs a basis set based on the fixed-point coefficients that have been synthesized already. The elements in the basis set are used to synthesize the undetermined fixed-point coefficients and the method employs some strategies to speed up the design process. Innocent et al [10] developed and tested frequency adaptive Phasor Measurement Unit (PMU) algorithms with wider linearity range than specified in IEEE std C37.118-1, with three different concepts encompassing robust state-of-the-art design approaches like FIR bandpass filtering, extended Kalman filtering, and discrete Fourier transform demodulation with FIR low pass smoothing. FIR-based PMU are linear phase with no overshoot in either phase or amplitude step responses and the extended Kalman filtering PMU is more computer-intensive but allows for a reduced group delay and better out-of-band interference rejection at the cost of a phase step response with overshoot. Herrick et al [11] presented a stable inversion of non-minimum phase systems with highly efficient computation for high-sampling rate applications. The stable filter that inverts the dynamics of a non-minimum system is based on cascading a stable pole-zero cancellation Infinite Impulse Response (IIR) filter with a high-order Finite Impulse Response (FIR) filter which inverts the unstable zero dynamics. The high-order FIR is realized based on efficient IIR filter implementation introduced by Powell and Chau then later modified by Kurosu. Fernando et al [12] developed a real-time, digital algorithm for PWM with distortion-free baseband. The algorithm not only eliminates the intrinsic baseband distortion of digital PWM but also avoids the appearance of sideband components of the carrier in the baseband even for low switching frequencies. The algorithm uses several FIR filters and a few multiplications and additions and is implemented in real time on a standard DSP. Wail [13] presented a stable explicit depth wavefield extrapolation using sparse frequency-space (f-x) FIR digital filter. The ideal impulse response of the wavefield extrapolation filter is non-sparse in nature, and designing such filters was formulated as an L1-
norm minimization that is convex with a quadratic constraint. Sparse filters were obtained by employing hard thresholding to the filter coefficients magnitude. The method is used to design high accuracy sparse extrapolation filters. Felipe et al [14] proposed an efficient estimator of optimal memory for discrete-time FIR filters in state-space. The crucial property is to measure and filter the output that involved with no reference and noise statistics. Testing by the two-state polynomial model has shown a very good correspondence with predicted values. Kwang-Jin et al [15] proposed a theory of harmonic filters and their mathematical models applicable to periodically time varying systems perturbed by a large periodic signal. The harmonic filters are based on a series of finite time delay and weighted sum operations, which allow selection or rejection of an input fundamental tone and/or its harmonics in a periodic manner.

3. FIR FILTER IMPLEMENTATION

FIR filters are used in applications where exact linear phase response is needed, and the filter is normally implemented in a non-recursive way to provide a stable filter. The design or FIR filter consists of two stages namely: approximation and realization. In the approximation stage based on the specification and the transfer function is obtained through the following four steps.

i. Desired response is chosen in the frequency domain
ii. Length of the FIR filter is chosen
iii. Measure of quality of approximation is chosen
iv. Algorithm is chosen to find the best filter transfer function

The realization stage deals with choosing the structure to implement the transfer function which is in the form of circuit design or in the form of a program. The system to convert ideal impulse function like a sink function into a FIR filter design is named as windowing method. This method uses multiplying the infinite impulse response with a finite length sequence called the window function and the function is zero outside the chosen interval. The window function is represented by \( w(n) \), and is represented by,

\[
w(n) = w(-n) = \begin{cases} 
\text{nonzero; } & |n| \leq (N - 1)/2 \\
0 & |n| > (N - 1)/2
\end{cases}
\]  

where, \( N \) represents the width of the window function \( w(n) \), and \( n \) is an integer of value, \( 0 \leq n \leq N - 1 \). For the rectangular window the signal is given as,

\[
w(n) = \begin{cases} 
1; & -\frac{N-1}{2} \leq n \leq \frac{N-1}{2} \\
0 & \text{elsewhere}
\end{cases}
\]  

In this system the truncation operation increases the signal bandwidth and the spectrum will spread. This phenomenon is called spectral spreading. The signal bandwidth is inversely proportional to the signal duration. Larger the window, smaller the bandwidth and smaller spectral spreading results in less signal distortion. Figure 3 shows the frequency response of different order FIR low pass filter. It is observed that the order of the filter increases the magnitude response that reaches the cut-off frequency, \( fc = 1000 \text{ Hz} \). At low frequency the output is obtained maximum, and at higher frequency the output is all most zero.

Figure 3 Frequency Response of Low Pass FIR Filter

Figure 4 shows the frequency response of the high pass filter for different order. For this also the cut off frequency, \( fc \) is fixed as 1000 Hz. When the order of the filter increases the magnitude response reaches the cut-off frequency. At low frequency the output is almost zero, and at high frequency the output is maximum.
Figure 4 Frequency Response of High Pass FIR Filter
Figure 5 shows the frequency response of a band pass FIR filter with different orders. At low frequency and high frequency, the output is nearly zero, and at frequencies between 750 Hz and 1250 Hz, the output is maximum.

Figure 5 Frequency Response of Band Pass FIR Filter
Figure 6 shows the frequency response of a band pass FIR filter with different orders. At low frequency and high frequency, the output is maximum, and at frequencies between 750 Hz and 1250 Hz, the output is very minimum.

Figure 6 Frequency Response of Band Stop FIR Filter

4. CONCLUSION

In this paper, the design of FIR filters with order 4, order 8, order 12, and order 18 was implemented and analyzed its performance for low pass filter, high pass filter, band pass filter and band stop filter. Also the performance of these filters is analyzed.
References


