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ISSN 2959-6157

Design and Hardware Implementation of FIR Digital Filters

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Abstract:

Digital Signal Processing (DSP) plays an important role in the College of Information Science and Technology & Artificial Intelligence areas such as communications, audio and video signal processing, etc. Filters as a central element solve the problems of noise suppression and signal extraction, which is signal separation and restoration. Finite impulse response (FIR) filters are often used in applications such as high-quality audio and image processing due to their linear phase, stability, and flexible frequency response design. This paper reviews FIR filter design methods, including window function methods, frequency discretization methods, and optimization algorithms, and investigates methods for their hardware implementation. Analyzing basic convolutional filter operations and the ways to increase the speed of filter operations by applying hardware platforms such as FPGAs and DSP processors will be explored to present innovative feasibility in the future development of new algorithms. The results of this study show that FIR filters have promising applications when supported by hardware technologies, especially in the implementation of higher-order filters that exhibit unique advantages.

Keywords: FIR digital filters; signal processing; hardware implementation; optimization algorithms; frequency response design.

1. Introduction

Digital signal processing (DSP) plays an important role in modern technology and is widely used in communication systems, audio and video processing, radar, and medical image processing. In these applications, filters are the most important signal processing tools, solving important tasks such as noise suppression, signal extraction and signal transformation. Depending on their impulse response characteristics, digital filters can be divided into finite impulse response (FIR) and infinite impulse response (IIR) filters.

Internationally, FIR filter research focuses on the optimization of design algorithms and hardware implementation techniques. In recent years, the combination of FPGA and ASIC has enabled more efficient hardware implementation of FIR filters. The combination of parallel computing, pipelined architectures and hardware gas pedals has dramatically improved the computational efficiency of FIR filters in real-time signal processing and the processing of large quantities of data. Furthermore, the development of FIR filters based on convolutional neural networks (CNN) has become a research focus for implementing FIR filters in CNN architectures to improve signal processing. Researches in this direction have particular potential in areas such as image recognition and speech analysis.

In China, research institutes and universities have made great strides in FIR filter research, especially with respect to algorithms in optimizing filter coefficients and the efficiency of hardware implementation. Much of our researches focus on developing energy-efficient, high-performance hardware architectures to meet the demand for mobile devices and embedded systems. And hardware development of proprietary chips is a key to facilitate the application of FIR filters. In cooperation with domestic industry, research results are gradually being put into practice.

This article provides a detailed overview of main features and benefits of digital FIR filters. First, it describes the main advantages of FIR filters over IIR filters in terms of inherent stability, linear phase characteristics and ease of hardware implementation. Then, three commonly used FIR filter design methods are introduced, including the window function method, the frequency sampling method and the optimization algorithm design, and the design ideas, application scenarios, advantages and disadvantages of each method are discussed. In addition, this paper discusses the specifics of implementing FIR filters in hardware, especially the design steps and challenges in FPGA and DSP platforms. Finally, this paper discusses the future development of FIR digital filters and their innovative potential in areas such as intelligent signal processing, low-power design, and adaptive filtering. It also envisions future possibilities in optimizing design efficiency and reducing hardware resource consumption.

2. Characteristics and Advantages of FIR Digital Filters

FIR Digital Filters are a kind of linear time-invariant system. Its impulse response ends within a finite time. Compared to IIR filters, FIR filters have the following unique advantages:

Firstly, FIR filters have line phase characteristics [1], which can achieve strict linear phase response by reasonable design. This ensures that all frequency components of the input signal pass through the filter with the same time delay, avoiding phase distortion. This is especially critical for applications that require accurate preservation of signal waveforms, such as high-fidelity audio processing and image processing.

Secondly, FIR filters have inherent stability. Since the poles of FIR filters are all located in the unit circle, their structure inherently ensure system stability. This characteristic enables the FIR filters to effectively handle numerical errors and environmental changes in practical applications without causing system instability. It is difficult for IIR filters fully ensure[2].

Lastly, FIR filters also have flexible frequency response design. By increasing the order of the filter, FIR filters are able to approximate any desired frequency response[3]. This flexibility gives it a significant advantage in the design of high-order low-pass, band-pass, band-stop and high-pass filters.

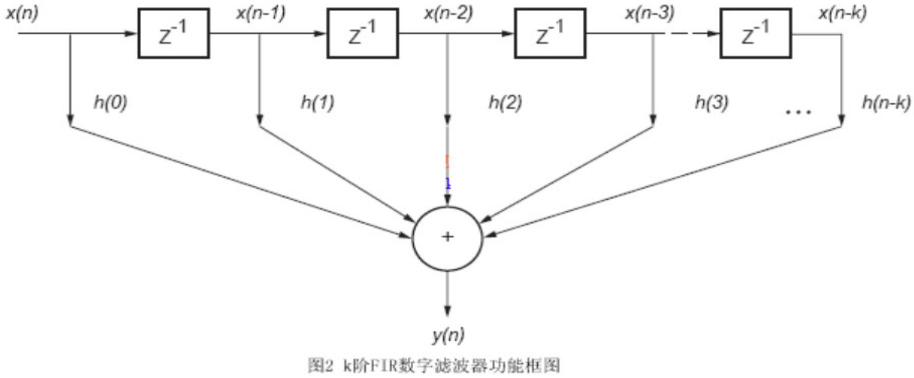

Fig. 1 Block Diagram of a Simple FIR Filter

Figure 1 is a block diagram of a simple FIR filter, where each box labeled z-1 represents a register cell with one clock cycle delay. The diagram shows the data channels and the operations must be done by the filter. Each stage of the filter holds a delayed input sample, the input connections and output connections of each stage are called taps. The set of coefficients {hk} is called the filter's tap coefficients. A M-order filter has M+1 taps. The output

ISSN 2959-6157

yFIR[n] is obtained by multiplying the sampled value of data flow at each clock edge n(time subscript) with the taps through shift registers, and summing them up. The addition and multiplication of filters must be fast enough to form y[n] before the next clock arrives. The size of each stage must be measured to fit the width of their data channel. In practical applications where accuracy is required, the Lattice structure can reduce the impact of finite word lengths, but increases computational cost. The general goal is to filter as fast as possible to achieve high sampling rates. The longest signal path through the combinational logic includes M-stage addition and one-stage multiplication operations. The FIR structure specifies the finite word length of each arithmetic cell of the machine and manages the flow of data during the operation process.

3. The Design Methods of FIR Digital Filters

The core of FIR design lies in determining the filter coefficients, which are the individual parameters that determine their impulse response. Common design methods include window function methods, frequency sampling methods, and optimization algorithms.

3.1 Window Function Methods

The most common FIR filter design method is the window function method, whose basic idea is to obtain an infinitely long impulse response sequence through the inverse Fourier transform of an ideal filter, and then truncate this sequence by selecting an appropriate window function[4]. The commonly used window functions include rectangular window, Hamming window, Hanning window and Blackman windows. The influence of different window functions on the frequency response of the filter is mainly reflected in the trade-off between main lobe width and the side lobe attenuation. In practical applications, such as processing biomedical signals like EEG, ECG, etc., FIR filters are used to design low-pass or band-stop filters by the window function method to help eliminate specific frequency noise, such as 50Hz power interference. Since biomedical signals are often sensitive to phase distortion, Hanning windows are often used in these applications to ensure a linear phase response of the signal[5]. The window function method also has a variety of uses in radar systems. The radar-received signals are accurately filtered and processed to compress the pulses, and pulse compression techniques in radar systems are used to improve range resolution. By transmitting longer FM signals (e.g., Linear FM pulses, LFM) and compressing them at the time of reception, shorter equivalent pulse lengths can be achieved. FIR filters designed by the window function method can be used for matched filter design for pulse compression, where the commonly used window functions include Hamming window and Blackman window. For the Hamming window: it has good side lobe suppression ability, which can reduce the influence of pseudo-signal (side flap) next to the main lobe of the compressed pulse, and is suitable for scenarios with a more complex noise environment. For the Blackman window: Due to its excellent side lobe attenuation (better than the Hamming window), it is suitable for radar systems that require a very low level of side lobe, helping to improve the signal-to-noise ratio for target detection. The compared filtering result can be seen in Figure 2.The figure below shows the filtering results, and the figure above shows the raw signal with burrs, which severely damages additional analysis.

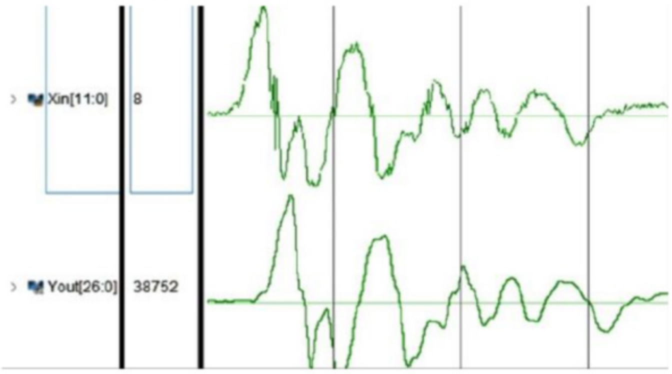

Fig. 2 The improved DA-based FIR filter is used to compare filtering results[6]

3.2 Frequency Sampling Methods

The frequency sampling method, as a method for designing FIR filters, has a large number of studies exploring its application and optimization in signal processing. This method works by specifying the desired frequency response in the frequency domain and converting it into time-domain filter coefficients by Inverse Discrete Fourier Transform (IDFT). In the frequency sampling method, the transition band sampling values must be determined in order to maximize the attenuation in the stopband. The transition band sampling values are usually obtained by the look-up table method, but the values obtained by the look-up table method are not guaranteed to be optimal. Evolutionary Programming (EP) is a multi-intelligent stochastic optimization technique that can find global optimal solutions for complex problems [7].

Studies have shown that the frequency sampling method can provide high accuracy in multiple application scenarios, especially in areas with strict frequency response requirements, such as communications, audio processing, and image processing. For example, in audio signal processing, researchers have found that this method is able to effectively generate band-pass filters and accurately filter out noise from specific frequency bands, thereby enhancing the clarity of audio signals. In addition, recent studies have also proposed various improvement strategies, such as combining weighting processing, to reduce frequency leakage problems and improve filter performance.

3.3 Optimization Algorithms

With the improvement of computing power, the application of optimization algorithms in FIR filter design is becoming more and more widespread. Common optimization algorithms include least squares method, Chebyshev approximation, and genetic algorithm, etc. The core idea of optimization algorithms is to obtain the filter coefficients that satisfy the design requirements through iteration and minimization of error functions. In recent years, researchers have successfully designed FIR filters with different complexity requirements through these algorithms. For example, the least squares method is widely used in the design of low-pass filters, as it can minimize the error between the target frequency response and the actual filter response. In addition, the Chebyshev approximation method can provide optimized frequency response control, providing better bandpass or bandstop filter designs while reducing computational complexity.

There are two principles for designing FIR filters based on Chebyshev. One is the equal ripple design, which means that the frequency error of the filter has the same ripple-like distribution in the passband and the stopband, and the equal ripple design minimizes the quantization error. This design strategy allows the Chebyshev design to better control the error distribution in both the passband and the stopband to achieve a more consistent filtering performance. The other is the Chebyshev polynomial approximation method, where the alternation theorem is the core of the efficient implementation Chebyshev approximation algorithm. The alternation theorem extends the real cases to the complex cases, deriving an efficient exchange algorithm for designing complex FIR filters in the Chebyshev sense. By transforming the complex error functions, the Remez exchange algorithm can be used to compute the optimal complex Chebyshev approximation. If the complex Chebyshev errors alternate, the algorithm converges to the optimal solution.In all other cases, the algorithm converges to the best Chebyshev approximation on the required frequency bands.This is the optimal solution of the Chebyshev polynomial approximation method [8].

4. Hardware Implementation of FIR Filters

The advancement in hardware technology has implemented FIR filters not limited to software simulation, and the need for hardware acceleration is increasing. Hardware implementation of FIR filters is usually based on platforms such as FPGAs, DSP processors, or microcontrollers.

Firstly, the coefficients need to be quantized and stored. In real hardware, the filter coefficients need to be quantized to accommodate the fixed or floating-point computing capabilities of the hardware. The loss of accuracy during quantization is a significant challenge in hardware implementations and requires careful design to maintain the performance of the filter. Secondly, the core operation of the FIR filter is the convolution operation [9], which is the input signal multiplication and summation of the filter coefficients point by point. To enhance the speed of operation and processing efficiency, pipelining and parallel computing techniques are usually used in hardware design to optimize the data flow. Finally, resource optimization in hardware implementation involves optimization of processing unit utilization, power management and circuit area. These optimization strategies need to be fully considered in the early design process to minimize cost and energy consumption while meeting performance requirements.

5. The Future Development of FIR Filters

With the development of technology, the application of FIR filters in several fields is expanding, and raising the demand in various areas, so innovative designs and applications are necessary. In traditional FIR filter design, the main challenge lies in balancing filter order with computational complexity. With the improvement of hardware

ISSN 2959-6157

performance and the development of new algorithms, FIR filters offer a wide scope for innovation in the implementation of higher order filters. For example, by introducing the adaptive filtering technique [10], the filter coefficients can be dynamically adjusted to suit different signal environments, thus enhancing the flexibility and adaptability of the filter. In addition, using deep learning algorithms to optimize the filter design, combined with the data-driven method, the computation amount can be significantly reduced while maintaining the filter performance. In the future, focus can be placed on error control [11], especially on hardware platforms with fixed-point operations, to ensure that the filter maintains accuracy while computing efficiently. With the development of algorithms, research can explore new design methods, such as filter coefficient optimization strategies based on evolutionary algorithms, genetic algorithms, or particle swarm optimization. These new design methods can improve design efficiency and achieve more accurate frequency response control. In the context of 5G and the Internet of Things, we can explore how to efficiently deploy FIR filters in distributed systems to optimize their signal processing capabilities in low-latency, distributed environments.

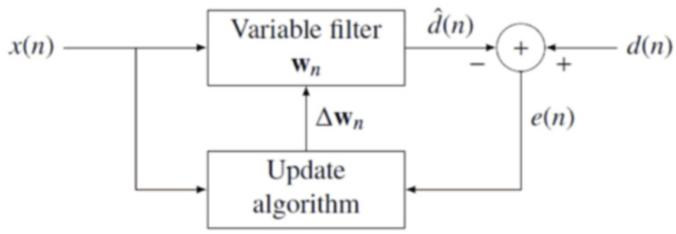

Fig. 3 Block diagram of the adaptive FIR filter

The operation of the adaptive filter can be seen in Figure 3.It involves two basic processes: the filtering process and the adaptive process. The filtering process is the convolution of the input signal with the filter coefficients, used to produce an output response to a series of input data. The adaptive process is the adaptive adjustment of the filter parameters by specific algorithms aimed at continuously reducing the mean square error between the response signal and the desired signal [12].

6. Conclusion

This paper analyzes the design methods, material implementation, innovative feasibility, and scientific progress of digital FIR filters. By analyzing different design techniques and hardware implementation strategies, it can be concluded that FIR filters have unique advantages in applications where high accuracy and stability are strictly required and have the potential for wider application as hardware technology support develops. Future research and applications will focus more on the flexibility and adaptability of filter design. Firstly, advanced algorithms such as deep learning are bringing new breakthroughs in filter development, and these techniques enable filters to adjust their parameters dynamically to adapt to changing signal environments. At the same time, technologies such as low power consumption and self-adaptability will be important guidelines for hardware implementation of filters, especially in scenarios that require efficient signal processing, such as IoT and mobile devices. FIR filters will gradually be integrated with big data analytics, machine learning and hardware acceleration technologies. It will make these things more useful in future communication, signal processing, and smart devices and create opportunities for innovative applications.

Authors Contribution
All the authors contributed equally and their names were listed in alphabetical order.

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