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Review study on “Design and Analysis of FIR Filters”

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***Abstract* - Digital filters may be more expensive than an equivalent analog filter due to their increased complexity, but they make practical many designs that are impractical or impossible as analog filters. Digital filters can often be made very high order and are often finite impulse response filters which allows for linear phase response. When used in the context of real-time analog systems, digital filters sometimes have problematic latency (the difference in time between the input and the response) due to the associated analog-to-digital and digital-to-analog conversions and anti-aliasing filters, or due to other delays in their implementation.**

The digital filter is the most basic building block used for most DSP applications; an efficient implementation of the filter is essential to save hardware. A Digital Filter is made up of three basic components: adders, coefficient multipliers and delays. Digital filters are commonplace and an essential element of everyday electronics such as radios, cellphones, and AV receivers.

Present review paper tried to improve performance using computation simulation but here MATLAB simulation has been performed to simplify the complicated computational calculation and improvise previous performance.

***Index Terms* - FIR filter, MATLAB, window function, frequency sampling, optimization, amplitude-frequency characterization**

INTRODUCTION

1.1 Introduction of digital filter

In signal processing, a digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. This is in contrast to the other major type of electronic filter, the analog filter, which is an electronic circuit operating on continuous-time analog signals.

A digital filter system usually consists of an analog-to-digital converter (ADC) to sample the input signal, followed by a microprocessor and some peripheral components such as memory to store data and filter coefficients etc. Program Instructions (software) running on the microprocessor implement the digital filter by performing the necessary mathematical operations on the numbers received from the ADC. In some high-performance applications, an FPGA or ASIC is used instead of a general-purpose microprocessor, or a specialized digital signal processor (DSP) with specific paralleled architecture for expediting operations such as filtering. [1]

Digital filters may be more expensive than an equivalent analog filter due to their increased complexity, but they make practical many designs that are impractical or impossible as analog filters. Digital filters can often be made very high order and are often finite impulse response filters which allows for linear phase response. When used in the context of real-time analog systems, digital filters sometimes have problematic latency (the difference in time between the input and the response) due to the associated analog-to-digital and digital-to-analog conversions and anti-aliasing filters, or due to other delays in their implementation.

Digital filters are commonplace and an essential element of everyday electronics such as radios, cellphones, and AV receivers. [2]

The digital filter is a discrete system, and it can do a series of mathematic processing to the input signal, and therefore obtain the desired information from the input signal. The transfer function for a linear, time-invariant, digital filter is usually expressed as

$$H(z) = \frac{\sum_{j=0}^{M} b_j z^{-j}}{1+\sum_{i=1}^{N} a_i z^{-i}}$$

Where, ai and bi are coefficients of the filter in Z-transform.

There are many kinds of digital filters, and also many different ways to classify them. According to their function, the FIR filters can be classified into four categories, which are lowpass filter, high pass filter, bandpass filter, and band stop filter.

According to the impulse response, there are usually two types of digital filters, which are finite impulse response (FIR) filters and infinite impulse response (IIR) filters.

According to the formula above, if ai is always zero, then it is a FIR filter, otherwise, if there is at least one none-zero ai, then it is an IIR filter. Usually we need three basic arithmetic units to design a digital filter, which are the adder, the delay, and the multiplier.

The following are several steps of designing a digital filter:

- Make sure of the property of a digital filter according to the given requirements.
- Use a discrete linear time-invariant system function to approach to the properties.
- Make use of algorithms to design the system function.
- Use a computer simulation or hardware to achieve it.

1.2 FIR filter

In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time. This is in contrast to infinite impulse response (IIR) filters, which may have internal feedback and may continue to respond indefinitely (usually decaying). [3]

The impulse response (that is, the output in response to a Kronecker delta input) of an Nth-order discrete-time FIR filter lasts exactly N + 1 samples (from first nonzero element through last nonzero element) before it then settles to zero.

FIR filters can be discrete-time or continuous-time, and digital or analog. The finite impulse response (FIR) filter is one of the most basic elements in a digital signal processing system, and it can guarantee a strict linear phase frequency characteristic with any kind of amplitude frequency characteristic. Besides, the unit impulse response is finite; therefore, FIR filters are stable system. The FIR filter has a broad application in many fields, such as telecommunication, image processing, and so on.

The system function of FIR filter is

$$H(z) = \sum_{n=0}^{L-1} h[n]z^{-n} ,$$

Where L is the length of the filter and h[n] is the impulse response.

An FIR filter has a number of useful properties which sometimes make it preferable to an infinite impulse response (IIR) filter. FIR filters:

Require no feedback. This means that any rounding errors are not compounded by summed iterations. The same relative error occurs in each calculation. This also makes implementation simpler. [2-3]

Are inherently stable, since the output is a sum of a finite number of finite multiples of the input values, so can be no greater than $\sum b\{i\}$ times the largest value appearing in the input. It can easily be designed to be linear phase by making the coefficient sequence symmetric. This property is sometimes desired for phase-sensitive applications, for example data communications, seismology, crossover filters, and mastering. [3]

The main disadvantage of FIR filters is that considerably more computation power in a general-purpose processor is required compared to an IIR filter with similar sharpness or selectivity, especially when low frequency (relative to the sample rate) cutoffs are needed. However, many digital signal processors provide specialized hardware features to make FIR filters approximately as efficient as IIR for many applications.

1.3 IIR filter

The infinite impulse response (IIR) filter is recursive structure, and it has a feedback loop. The precision of amplitude frequency characteristic is very high, and IIR filters are not linear phase.

The infinite impulse response (IIR) filter is a recursive filter in that the output from the filter is computed by using the current and previous inputs and previous outputs. Because the filter uses previous values of the output, there is feedback of the output in the filter structure. The design of the IIR filter is based on identifying the pulse transfer function G(z) that satisfies the requirements of the filter specification. This can be undertaken either by developing an analogue prototype and then transforming it to the pulse transfer function, or by designing directly in digital.

1.4 Comparison of FIR and IIR

The digital filters can be divided broadly into two types – finite impulse response (FIR) and infinite impulse response (IIR) filters. The main point is that the two types of filter are very different in their performance and also in their design. [3-7]

Under the same conditions as in the technical indicators, output of the IIR filter has feedback to input, so it can meet the requirements better than FIR. The storage units are less than those of IIR, the number of calculations is also less, and it's more economical.

The phase of FIR filter is strictly linear, while the IIR filter is not. The better the selectivity of IIR filter is, the more serious the nonlinearity of the phase will be.

The FIR filter is non-recursive structure, finite precision arithmetic error is very small. While IIR filter is recursive structure, and parasitic oscillation may occur in the operation of IIR filter.

Fast Fourier Transformation can be used in FIR filter, while IIR cannot.

The IIR filter can make use of the formulas, data and tables of the analog filter, and only a small amount of calculation. While FIR filter design may always make use of the computer to calculate, and the order of FIR filter could be large to meet the design specifications.

II-SIMULATION OF ELECTRONIC COMMUNICATION SYSTEM

The concept of telecommunications and electronic systems simulation

Telecommunication is the exchange of signs, signals, messages, words, writings, images and sounds or information of any nature by wire, radio, optical other electromagnetic systems.

Telecommunication occurs when the exchange of information between communication participants includes the use of technology. It is transmitted through a transmission medium, such as over physical media, for example, over electrical cable, or via electromagnetic radiation through space such as radio or light. [10]

System simulation technology refers to computer simulation technology, which developed since 1970 combining modern computers and simulation software. Computer simulation has high precision, versatility, good repeatability, rapid modeling, and low-cost advantages.

III-DESIGN OF FIR FILTER

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.

FIR filters are particularly useful for applications where exact linear phase response is required. The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter.

FIR filter design essentially consists of two parts.

- Approximation problem
- Realization problem

The approximation stage takes the specification and gives a 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.

Optimization

Historically, the window function method was the first method for designing linear-phase FIR filters. The frequency sampling method and optimized equi-ripple method were developed in the 1970's and have become very popular since then.

Recent times have witnessed a wide application of evolutionary optimization by researchers, in design of digital FIR filters, based on frequency domain specifications. A significant growth has been reported in the field of evolutionary optimization-based FIR filter design. Optimization-based techniques are used to solve the filter design problem by framing the design task as an error function which is further solved to determine the filter coefficients that satisfies the desired specifications. However, the nonlinear, non-differentiable, non-convex, multimodal nature of the associated optimization problem makes the design task quite challenging. In this regard, a number of evolutionary optimization- based techniques have been applied for FIR filter design.

Lacking precise control of the specified frequencies, like ωp and ωs, is the most serious disadvantage of the window function method in the design of a lowpass FIR filter. The frequency sampling method is better than the window method in the aspect that the real-valued frequency response characteristic Hr(ω) is specified, which can be either zero or unity at all frequencies, except the transition band. [12-15]

The Chebyshev approximation method offers completely control of the filter requirements. As a result, this method is more preferable than the other two. It is based on the Remez exchange algorithm, which minimizes the error with respect to the max-norm.

IV-CONCLUSION

On the basis of present study, it can be concluded that FIR filter design essentially consists of two parts.

1. Approximation problem
2. Realization problem

The approximation stage takes the specification and gives a transfer function using some method or algorithm to find the best filter transfer function.

The 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.

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

The above three the major advantages of using window method is their relative simplicity as compared to other methods and ease of use. The fact that well defined equations are often available for calculating the window coefficients has made this method successful.

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