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Design of FIR Digital Filters Using Evolutionary Algorithms:

A Comparative Study

by

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Supervised by

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Thesis presented to

The Department of Telecommunications Engineering

In partial fulfillment of the requirements for the degree of

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To

my fiance,

my parents and all my family members

DECLARATION

I, Haneen Salem Bany AL-Domi, recognize what plagiarism is and I hereby declare that this thesis entitled, "**Design of FIR Digital Filters Using Evolutionary Algorithms: A Comparative Study**", which is submitted to the department of **Telecommunications Engineering at Hijjawi Faculty for Engineering Technology**, for the partial fulfillment of the requirements for the degree of Master of Science is my own work. I have not plagiarized from any sources. All references and acknowledgments of sources are given and cited in my thesis. I have used the conventional citation and referencing. Each significant contribution and quotation in this thesis from work of other people has been attributed and referenced.

Haneen Bany AL-Domi

December 2016

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ABSTRACT

Design of FIR Digital Filters Using Evolutionary Algorithms: A Comparative Study

FIR filter design is a multi-modal optimization problem. The traditional gradient based optimization techniques such as windowing, Park and McClellan (PM), etc., are ineffective for digital filter design. Therefore many evolutionary techniques and approaches such as Pattern Search (PS), Particle Swarm Optimization (PSO), Improved Particle Swarm Optimization (IPSO), Seeker Optimization Algorithm (SOA) and Hybrid Seeker Optimization (HSO) have been proposed to find the best solution of FIR filter design problem. Thus, when designing FIR filter, one need to adopt one of these approaches.

The aforementioned FIR filter design techniques are used to design linear phase FIR filter. These techniques are different in terms of controlling passband, stopband and transition width of the filters. Thus evolutionary techniques are proposed to overcome these techniques.

Furthermore, PSO suggests a new definition for the swarm updating and velocity vector so the solution quality will be better. PS, PSO, IPSO, SOA and HSO algorithms lead to find minimum number of taps and thus reducing the number of multipliers and adders that are used in the FIR filter design.

In this work, we plan to perform a detailed comparison between the evolutionary techniques and the conventional gradient techniques. In addition, a comparative study between different evolutionary algorithms will be illustrated. Here, software packages such as MATLAB and ScopeFIR are used to illustrate and testify our work.

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CHAPTER 1

FIR FILTERS AND THESIS MOTIVATION

1.1 Introduction to DSP and Filters

Digital Signal Processing (DSP) has a lot of features, algorithms and techniques that makes it better than analog ones. DSP provides flexibility as well as very good performance in terms of attenuation and selectivity. Thus, DSP has a wide use in communication field applications, biomedical engineering, military applications, etc.

One of the most important applications of DSP is the digital filter design. In general, a digital filter presents a system which is used to improve the signal quality and retrieve the desired information from it.

A digital filter has two types: The first is Infinite Impulse Response (IIR) and the second is Finite Impulse Response (FIR) filters. The response of FIR filter when applying an impulse signal at its input will vanish during a finite time. However, IIR filter response to an impulse input never disappears and will extend to infinity. The implementation of FIR filters is further simpler than IIR filters. However, FIR filters are slower than IIR filters. Overall, FIR filters are a preferable choice because they provide stability as well as design simplicity. Thus, in this work we shed the light on the design of FIR filter types using different approaches and evolutionary algorithms.

1.2 FIR Filters Characteristics and Design

FIR filters possess many beautiful properties compared to IIR filters which are guaranteed stability, possibility of exact linear phase characteristic at all frequencies and lower coefficient sensitivity.

There are different methods that are currently used for FIR filters design. Some of these methods are frequency sampling and the windowing. The easiest and popular method is the windowing method.

The widowing method uses many types of window functions such as Butterworth, Chebyshev and Kaiser. The window selection depends on the requirements of the passband and stopband, stopband attenuation, ripples and the transition width. Once the window function is selected ideal impulse response is multiplied with it. The length of the impulse response of ideal filter can be limited by these windows to a finite window to implement the actual response. However, the windowing method doesn't form the optimal design because it suffers from some weaknesses. For example, windowing method is not able to provide sufficient control of the frequency response in the various frequency bands and other filter parameters such as transition width. Filters which are designed by windowing method will have large error on discontinuous sides of the ideal frequency response and frequencies away from the discontinuity will have smaller one [1].

1.3 Related Works

Many methods have been proposed to design FIR filter. One of the most early methods is proposed by Parks and McClellan (PM). This method is the most popular for optimum FIR filter design because of its computational efficiency and its flexibility. PM method can specify the ratio between passband and stopband ripple (δ_p, δ_s). Unfortunately, one cannot determine a specific value for δ_p, neither for δ_s. On the other hand, the floating-point coefficients given by PM which will require quantization if hardware implementation is desired. These reasons lead to develop new optimization algorithms to find optimum taps (or coefficients) of FIR filters. These optimization algorithms, called evolutionary algorithms, will be explored to design optimum FIR filters with the largest stopband attenuation and better parameters control. The upcoming paragraphs will list some of these algorithms.

Particle swarm optimization (PSO) algorithm was developed by Kennedy and Eberhart in 1995 [4, 5]. PSO can be used as a general optimization method. It is easy to implement and few parameters are used to control it. Some research work has been done in exploring the flexibility of FIR filter design using PSO [6, 8]. PSO algorithm is developed based on social behavior of a swarm of bees, fish schooling and bird flocking and multi-dimensional optimization problems. To improve the efficiency many adjustments of the conventional PSO have been made.

Genetic Algorithm (GA) is another approach which confirms itself to be more efficient from obtaining the local optimum while retaining its moderate computational complexity. Unfortunately, GA was not successful in terms of convergence speed and the quality of the solution [6]. It depends on survival of the fittest concept [10]. Henceforth, GA was described in [13] to design FIR filters. The linear phase is determined by using symmetric filter coefficients. Fixed point implementations are obtained for linear-phase FIR filters by orthogonal GA (OGA) which is based on an experimental design technique in [14]. Stable digital filters with minimum phase for finite impulse response and infinite impulse response is designed by GA in [15].

FIR digital filter design is a subject of interest for researchers in various fields and applications. Many researchers have studied the performance and optimization of FIR filter using different global algorithms.

In [11], the authors described how we can use the PSO algorithm in electromagnetic optimizations. In addition, authors from UCLA present the results that show the swarm behavior in the PSO algorithm.

The authors in [12] studied the effect of Particle Swarm Optimization with Quantum Infusion (PSO-QI) on FIR and IIR filter design. The results show that PSO-IQ has an effect on stopband and passband of the filter response.

Simulated annealing (SA) algorithm is described in [16] to design digital filters with coefficient values which are presented as the sum of a power of two. Linear phase for digital filter procedure by this algorithm is presented. Then the algorithm is applied to design FIR filters.

1.4 Motivation

In this work, we focus our attention on using evolutionary algorithms to design FIR digital filter. These algorithms aim at optimizing the stopband and the passband to have equiripple FIR filters. Fortunately, evolutionary algorithms found to have great importance in filter design compared to traditional algorithms as they provide great improvement of filter response.

In addition, we use one of the most efficient software such as MATLAB to obtain the optimal coefficients of the designated FIR filter. Here, we implement the evolutionary algorithms using such a great software. Furthermore, to verify the obtained results another commercial software called ScopeFIR is used. Afterwards, a comparison between the results obtained via MATLAB and ScopeFIR are compared.

1.5 Problem Statement

Different approaches and techniques both traditional and evolutionary have been used to design FIR filters. Needless to say, most approaches used nowadays are based on evolutionary algorithms. Here, we focus our attention on conducting a comparison between traditional algorithms such as windowing and PM and evolutionary algorithms (PS, PSO, IPSO, SOA, and HSO) in terms of convergence, transition bandwidth and accuracy.

Furthermore, a comparison between the evolutionary algorithms (PS, PSO, IPSO,SOA and HSO) themselves has been performed. As a result, one who is in-charge in designing FIR filters will be able to select one of these algorithm to optimize his design.

CHAPTER 2

FILTER'S BASICS

2.1 Introduction

A filter is a device or a process which can perform signal processing functions to enhance a desired signal by removing unwanted frequency components or features from it. This means removing some frequencies in order to suppress interfering signals and reduce unwanted noise. Filters are designed to separate the undesirable signal frequencies from the desirable, or change frequency content that leads to changes the signal waveform. Filters usually serve two purposes the first is signal separation and the second is signal restoration. When a signal has been polluted with noise, interference or with another signal, signal separation is indispensable and when a signal has been distorted in a way or another, signal restoration is needed.

In circuit theory, a filter is an electrical network that changes the signal amplitude and/or signal phase characteristics with respect to frequency. Typically, a filter doesn't change the component frequencies of that signal and it doesn't add new frequencies to the input signal. It can change the amplitudes of the various frequency components and/or their phase relationships.

Usually, filters are used in electronic systems to confirm signals in some frequency ranges and refuse signals in other frequency ranges.

Fig. 2.1 shows a filtering example where one can see a useful signal at frequency f_1 has been polluted with an undesired signal at f_2. The polluted signal has very low gain at f_2 if it is passed through a circuit compared to f_1. Thus, the unwanted signal will be removed and the useful (desired) signal will remain. Note that in this simple example, we are not concerned with the gain of the filter at any frequency other than f_1 and f_2. As long as f_2 is sufficiently attenuated relative to f_1, the performance of this filter will be acceptable. In general, however, a filter's gain may be specified at several different frequencies or over a band of frequencies.

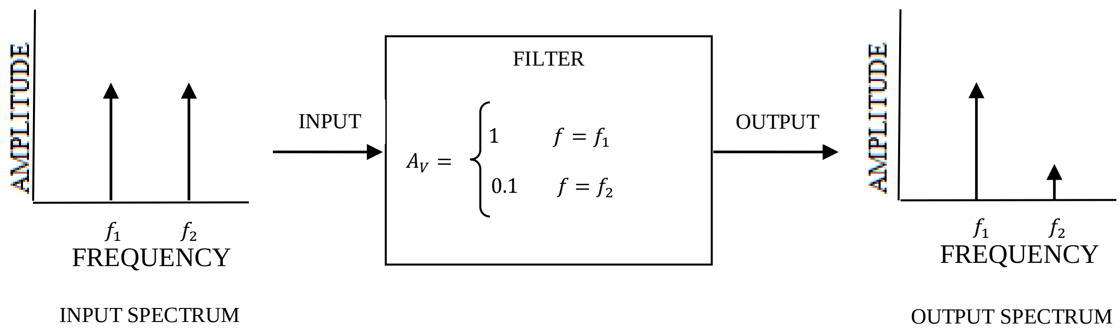

Fig. 2.1: Block diagram showing LP filtering action [1].

The transfer function (TF) or network function is used to describe mathematically the frequency-domain behavior of a filter. TF is the ratio of the Laplace transforms of output signal and input signal. So the voltage transfer function of a filter H(s) can be written as [1]

$$H(s) = \frac{V_{out}(s)}{V_{in}(s)}\,. \tag{2.1}$$

2.2 Classification of Filters

Filters can be classified as shown in Fig. 2.2. Here, one can see many kind of classifications, one classifies filters whether they are analog or digital. The second classifies filters whether active or passive. Based on the filter’s impulse response the filters are classified into FIR or IIR, etc. Hereinafter, we will describe these classifications.

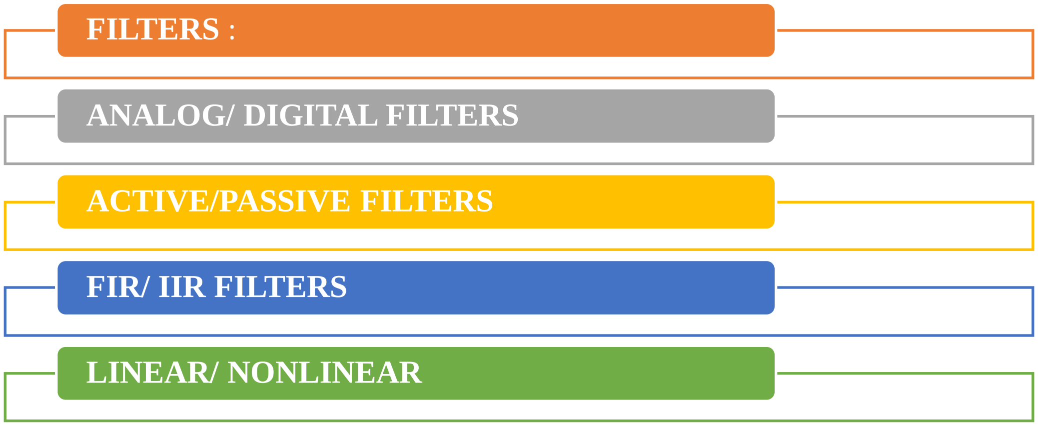

Fig. 2.2: Classifications of filters.

- **Analog/Digital filters:** Filters are classified as analog or digital according to the type of signal which can be either analog or digital. A system which can perform mathematical operations on samples of a discrete-time signal to reduce or improve some aspects of that signal is called a digital filter. However, the analog filter is a system which can perform mathematical operations on continuous time analog signals. A digital filter may be used to process analog signal by digitizing it. Afterwards, the digitized signal can be manipulated mathematically. Upon completion, a new analog signal is reconstructed back.
- **Active/Passive filters:** Filters are classified as active or passive based on the components which will be used for filter design. An active filter which uses active components such as an amplifier is a type of analog filter. Amplifiers can be used to enhance the performance of a filter. In addition to amplifiers, this filter can use the passive components such as resistors and capacitors. On the other hand, passive filters are made from passive components only such as inductors, resistors and capacitors.
- **FIR and IIR filters:** Filters are classified as FIR and IIR based on the impulse response type. FIR is a filter which has a finite duration impulse response which will die during a

finite time. In contrast to FIR, the IIR filter impulse response never decays and extends to infinity.

- **Linear and nonlinear filters:** Linear filter is a signal-processing device whose output is a linear function of its input in contrast to the nonlinear filters.

2.3 Digital Filters

Digital filtering is one of the best tools of DSP. DSP provides higher performance compared to analog processing. In addition, the digital filter characteristics can be changed easily under control programs. Therefore, these filters have a wide use in communications applications such as echo cancellation in modems, speech recognition, and noise cancellation.

There are two main types of digital filters: finite impulse response (FIR) and infinite impulse response (IIR). The impulse response of any dynamic system indicates the reaction of this system in response to external change. As was previously mentioned, the impulse response of an FIR filter goes to zero within a finite amount of time. On the contrary, IIR filter has an impulse response that never dies out, never decays and extends to infinite length of time. Henceforth, an FIR filter has a number of useful features such as guaranteed stability, low coefficient sensitivity and provides ease in design. Therefore, we will focus on FIR filters and its design.

FIR filter has finite duration impulse response and also knows as non-recursive, feed forward or transversal filters (i.e., has no feedback). It can be used to implement digitally almost any kind of frequency response. This type of filters is implemented by using a series of delays, multipliers, and adders to get the filter's output. Fig. 2.3 shows an FIR filter block diagram of length N. The operating on prior input samples causes delays.

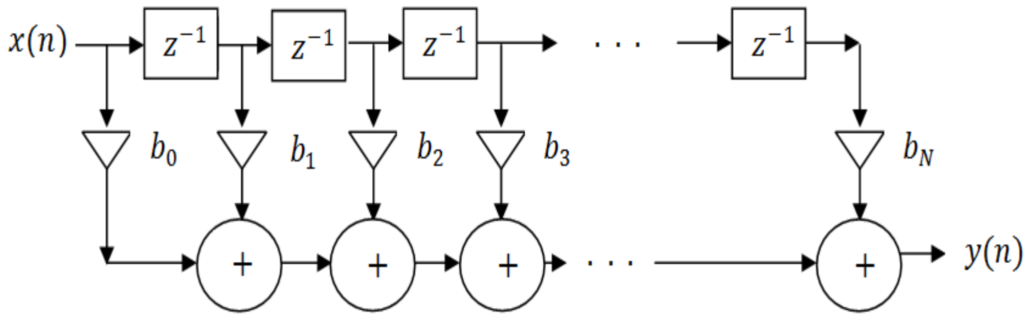

Fig. 2.3: Block diagram of a FIR filter [1].

For N order causal discrete-time FIR filter, a weighted sum of the recent input values shows the output sequence y[n] which is given by [1]

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N], \tag{2.2}$$

which can be written as [1]

$$y[n] = \sum_{i=0}^{N} b_i \,.\, x[n-i], \tag{2.3}$$

where $x[n]$ is the input signal, $y[n]$ is the output signal, N is the filter order, and b_i is the value of the impulse response at the ith instant for $0 \le i \le N$, and it represents the filter coefficient if the filter is a direct form FIR filter. The impulse response of this filter can be written as [1]

$$h[n] = \sum_{i=0}^{N} b_i \,.\, \delta[n-i] = \begin{cases} b_n \,, & 0 \le n \le N \\ 0 \,, & \text{otherwise} \end{cases}. \tag{2.4}$$

2.4 Properties of FIR Filters

There are a lot of useful properties which sometimes make an FIR filter better than an IIR filter:

- The rounding errors are not doubled by the summed iterations because FIR filters require no feedback.
- FIR filters are stable, since the output is a total sum of a finite number of multiples of the input values, so the maximum output value cannot be greater than $\sum_{\mathrm{i}=0}^{\mathrm{N}} b_i$.
- FIR filters can be designed to be linear phase easily by making the coefficients symmetric.

- There is a high flexibility in forming their magnitudes, depending on linear phase characteristics.
- An FIR filter has only zeros (no poles) so it is called an all-zero filter.
- An FIR filter needs more memory and/or calculation to reach a given characteristic of filter response but its delay feature is much better.

2.5 Types of Filters

The filters can be classified into a number of different bandforms describing which frequency bands the filter passes (the passband) and which it rejects (the stopband).

2.5.1 Lowpass Filters

A lowpass Filter is a filter that passes the signal whose frequency is lower than the cuttoff frequency (ω_c) and attenuates the signal whose frequency is higher than (ω_c). The amount of attenuation depends on filter characteristics. A lowpass filter is useful as it limits the upper frequencies range of, for instance, an audio signal.

For an ideal lowpass filter, the response can be written as follows [1]

$$H[\omega] = \begin{cases} 1\,, & 0 \leq \ \omega \leq \omega_c \\ 0\,, & \text{otherwise} \end{cases}, \tag{2.5}$$

where ω_c is the cuttoff frequency.

2.5.2 Highpass Filters

In contrast to the lowpass filter, the highpass filter passes the signal whose frequency is higher than the cuttoff frequency (ω_c) and attenuates the signal whose frequency is lower than ω_c. Amplitude response with a frequency higher than the cuttoff frequency (ω_c) is increased when a highpass filter is used.

For an ideal highpass filter, the response can be written as [1]

$$H[\omega] = \begin{cases} 0\,, & 0 \le \omega \le \omega_c \\ 1\,, & \text{otherwise} \end{cases}, \tag{2.6}$$

where ω_c is the cuttoff frequency.

2.5.3 Bandpass Filters

A bandpass filter is an intermediate filter that combines between lowpass and highpass filters. The most common application for this type is in wireless transmitters and receivers. To make sure that the frequency is within a certain rate, the difference between the upper and lower cuttoff frequencies is called the filter's bandwidth.

For an ideal bandpass filter, the response is given by [1]

$$H[\omega] = \begin{cases} 1\,, & \omega_l \le \omega \le \omega_h \\ 0\,, & \text{otherwise} \end{cases}, \tag{2.7}$$

where w_l and w_h is the lower and upper cuttoff frequencies, respectively.

2.5.4 Bandstop Filters

In contrast to the bandpass filter, the bandstop filter passes the signal whose frequency is higher and lower than the cutoff frequencies (ω_l and ω_h). As a result, the signal whose frequency is within the range between the lower and higher cuttoff frequencies is attenuated.

For an ideal bandstop filter, the response can be written as follows [1]

$$H[\omega] = \begin{cases} 0\,, & \omega_l \le \omega \le \omega_h \\ 1\,, & \text{otherwise} \end{cases}, \tag{2.8}$$

where ω_l and ω_h is the lower and upper cutoff frequency respectively.

2.6 FIR Filter Design Using Windows

A window function is defined in signal processing as a mathematical function which is equal to zero outside of certain interval. There are many types of window functions such as rectangular, Hamming, Hanning, Blackman and Kaiser window. There are a lot of applications for window functions, one of them is filter design.