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FIR Filter Design Using Mixed Algorithms: A Survey

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***Abstract*— In digital communication system, digital information can be sent on a carrier through changes in its fundamental characteristics such as phase, frequency and amplitude. The use of a filter plays an important part in a communication channel because it is effective at eliminating spectral leakage, reducing channel width, and eliminating interference from adjacent symbols (Inter Symbol Interference) ISI. It describe the developed and dynamic method of designing finite impulse response filter with automatic rapid and less error by an efficient genetic and neural approach. GA and Neural are powerful global optimization algorithm introduced in combinational optimization problems. Here, FIR filter is designed using Genetic, Neural approach by efficient coding schemes.**

We need to design these filters with some constraints imposed by requirements of the communication system in which we are going to use them. The use of optimization techniques have been proved to be quite useful towards the design of those digital filters with certain specifications. This paper reviews about the uses of optimization systems in digital filter design.

***Index Terms*— Genetic Algorithm, Artificial Neural Networks, Back propagation, FIR Filter, Optimization, DSP. FDA**

I. INTRODUCTION

A filter is a frequency selective circuit that allows a certain frequency to pass while attenuating the others. Filters could be analog or digital. Analog filters use electronic components such as resistor, capacitor, transistor etc. to perform the filtering operations. These are mostly used in communication for noise reduction, video/audio signal enhancement etc.

Filters constitute an essential part of DSP. Actually, their extraordinary performance is one of the main reasons which have made DSP so popular. Filter is essentially a system or network that improves the quality of a signal and/or extracts information from the signals or separates two or more signals which are previously combined Nowadays digital filters can be used to perform many filtering tasks are replacing the traditional role of analog filters in many applications.[1].

II. TYPES OF FILTER

A filter can be defined with reference to various fields such as chemistry, optics, engineering, turbulence modelling, engineering, computing, philosophy, and signal processing. Let us consider signal processing filters, filter can be defined as a device used for removing unnecessary part or parts of the signal. This removing of unnecessary parts of the signal is called as filtering process. These signal processing filters are classified into various types such as electronic filters, digital filters, and analog filters.

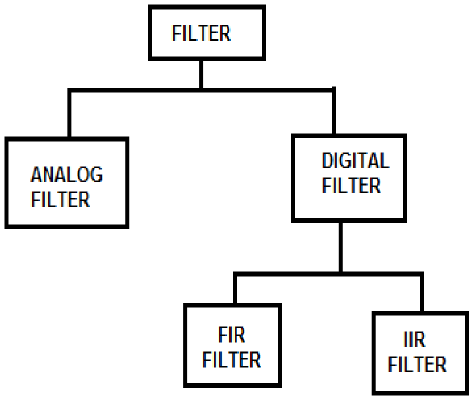

Figure 1: Classification of filters

III. DIGITAL FILTER:

Digital filters are used in wide variety of applications from signal processing, aerospace, control systems, defence equipments, telecommunications, system for audio and video processing to systems for medical applications to name just a few. Basically filter refers to a frequency selective device which extracts the useful portion of input signal lying within its operating frequency range and could be contaminated with random noise due to unavoidable circumstances. Analog filters are implemented with discrete components but the digital filters perform mathematical operations on a sampled, discrete time signal to reduce or enhance the desired features of the applied signal [2].

Digital filers are superior to their analog counterpart due to its wide range of applications and better performance. The advantages of digital filters over analog filters are small physical size, high accuracy and reliability. Digital filtering is one of the most powerful tools of Digital Signal Processing. Digital filters are capable of performance specifications such as ability to achieve multi-rate operation and exact linear phase that would, at best, be extremely difficult, if not impossible, to achieve with an analog implementation. In addition, digital filter characteristics are easy to change under software control. Digital filters are widely used in the fields of automatic control, telecommunications, speech processing and many more.

Digital Filter is an important part of digital signal processing (DSP) system and it usually comes in two categories: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR). FIR filter is an attractive choice because of the ease of design and stability. By designing the filter taps to be symmetrical about the centre tap position, a FIR filter can be guaranteed to

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have linear phase. Linear phase FIR filters are also required when time domain features are specified

A. *Finite Impulse Response (FIR)*

Digital filter is one whose impulse response is of finite duration [3]. The impulse response is "finite" because there is no feedback in the filter. If we put in an impulse (that is, a single "1" sample followed by many "0" samples), zeroes will eventually come out after the "1" sample has made its way in the delay line past all the coefficients. FIR (Finite Impulse Response) filters are implemented using a finite number "n" delay taps on a delay line and "n" computation coefficients to compute the algorithm (filter) function. The above structure is non-recursive, a repetitive delay-and-add format, and is most often used to produce FIR filters. This structure depends upon each sample of new and present value data. The number of taps (delays) and values of the computation coefficients are selected to "weight" the data being shifted down the delay line to create the desired amplitude response of the filter. In this configuration, there are no feedback paths to cause instability. The calculation of coefficients is not constrained to particular values and can be used to implement filter functions that do not have a linear system equivalent. More taps increase the steepness of the filter roll-off while increasing calculation time (delay) and for high order filters, limiting bandwidth. This can be stated mathematically as:

$$y(n) = \sum_{0}^{N-1} h(k)\,x\,(n-k)$$

where, y(n) = Response of Linear Time Invariant (LTI) system.

x(k) = Input signal
h(k) = Unit sample response
N = No. of signal samples

FIR filters are simple to design and they are guaranteed to be Bounded Input-Bounded Output (BIBO) stable. By designing the filter taps to be symmetrical about the centre tap position, an FIR filter can be guaranteed to have linear phase response. This is a desirable property for many applications such as music and video processing.

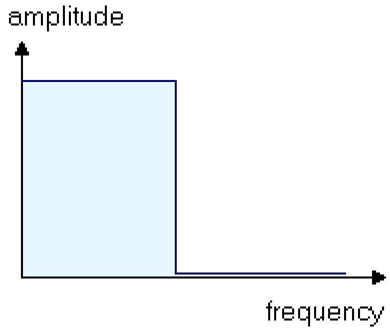
B. *Infinite Impulse Response (IIR) Filter*

IIR filter is one whose impulse response is infinite [4]. Impulse response is infinite because there is feedback in the filter.

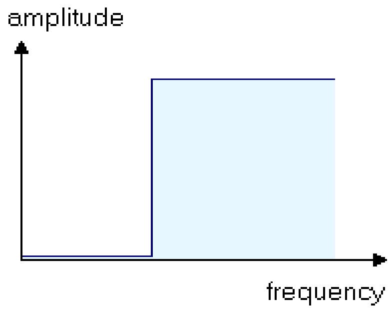
This permits the approximation of many waveforms or transfer functions that can be expressed as an infinite recursive series. These implementations are referred to as Infinite Impulse Response (IIR) filters. The functions are infinite recursive because they use previously calculated values in future calculations to feedback in hardware systems. IIR filters can be mathematically represented as:

M is the number of feed-back taps in the IIR filter and N is the number of feed-forward taps. IIR Filters are useful for high-speed designs because they typically require a lower number of multiply compared to FIR filters. IIR filters have lower side lobes in stop band as compared to FIR filters. Unfortunately, IIR filters do not have linear phase and they can be unstable if not designed properly. IIR filters are very sensitive to filter coefficient quantization errors that occur due to use of a finite number of bits to represent the filter coefficients. One way to reduce this sensitivity is to use a cascaded design.

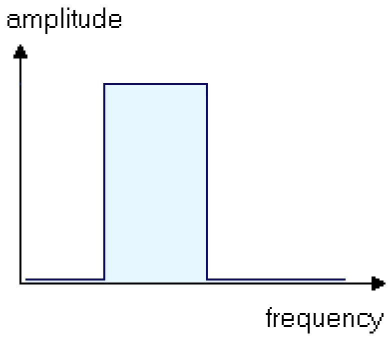
Figure 2 shows classification of filters on the basis of frequency.



(A) Low Pass



(B) High Pass



(C) Band Pass

Figure 2: Filter classification on frequency basis

A Low-Pass Filter is a filter that passes signals with a frequency lower than a certain cut-off frequency and attenuates signals with frequencies higher than the cut-off frequency. That show in figure 2(a).

B High-Pass Filter is an electronic filter that passes signals with a frequency higher than a certain cut-off frequency and attenuates signals with frequencies lower than the cut-off frequency. That show in figure 2(b).

C Band Pass Filter is an electronic device or circuit that allows signals between two specific frequencies to pass, but that discriminates against signals at other frequencies. That show in figure 2(c).

IV. LITERATURE REVIEW

Genetic Algorithm

Since the beginning of the nineteenth century, a significant evolution in optimization theory has been noticed. Classical linear programming and traditional non-linear optimization techniques such as Lagrange's Multiplier, Bellman's principle and Pontyagrin's principle were prevalent until this century. Unfortunately, these derivative based optimization techniques can no longer be used to determine the optima on rough non-linear surfaces. One solution to this problem has already been put forward by the evolutionary algorithms research community. Genetic algorithm (GA), enunciated by John Holland in the year 1975, is one such popular algorithm which is based on the concept of "survival of the fittest" by Charles Darwin [5]. Holland and his co-workers including Goldberg and Dejong popularized the theory of GA and demonstrated how biological crossovers and mutations of chromosomes can be realized in GA to improve the quality of the solutions over successive iterations.

Genetic algorithm is an optimization method which resembles the natural selection. A set of vectors which can act as a potential solution of the problem at hand is called genome (chromosome). A set of genomes is called population. GA creates new generations by applying some genetic operators to the individuals of population. A typical GA [9] can be summarized as follows:

1. **Initialization**: Generate initial population and compute score of each individual.
2. **Selection**: Select two individuals for mating.
3. **Crossover**: Mate two selected individuals and generate offspring.
4. **Mutation**: Mutate the offspring.
5. **Evaluation**: Calculate scores of offspring.
6. Repeat step 2-5 until a predefined number offspring is generated.
7. **Replacement**: Insert new offspring into the population.
8. Repeat steps 2-7 while termination criterion is not met.

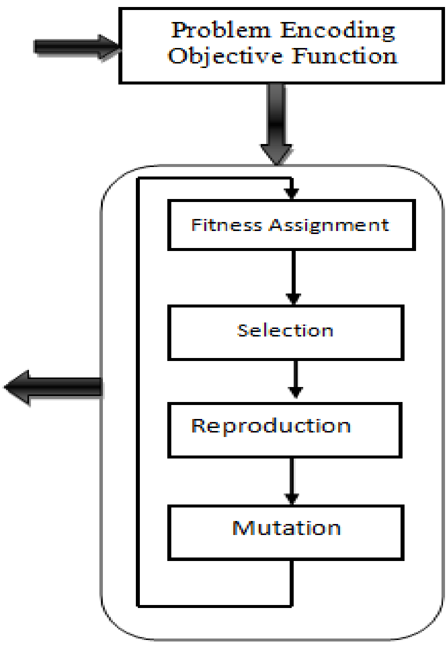


Figure 3: Flow Chart of GA

The encoding of the genome and defining an evaluation function or fitness function are the most important parts of GA design process. The structure of the genome must represent a solution to the problem of interest. Evaluation function, on the other hand, compares the performance of genomes to a goal and assigns a score to them. GA uses scores to rank the genomes in population.

V. MERITS AND DEMERITS OF GENETIC ALGORITHM

Genetic algorithm has key advantages over other widely-used techniques such as traditional algorithm, frequency sampling method and window method. It produces less ripples in pass band region and less ripple in stop band region and it has good transition band.

However the convergence time is large and some time pre mature convergence is occur. Which generate difficulties?

Progress of Genetic Algorithm in Digital Filter Design

Genetic Algorithm (GA) based design techniques are widely popular for synthesizing finite impulse response (FIR) filters. An effective design method for minimum phase digital FIR filters using GA has also been described in [6]. While obtaining the optimal-pass band and stop-band responses, the mean squared error (MSE) function is used and to optimize the transition band response the mean absolute error (MAE) is utilized.

Optimizing the design of infinite impulse response (IIR) filter has been achieved using GA, as reported in [6]. The IIR filter design under the mixed criterion of H2 norm and ∞ norm is proposed in [6] and GA is introduced to realize the filter design based on such criterion. It has been shown that the filter designed by GA is superior to conventional Butterworth filter in terms of either the optimization capability of design method or the performance of designed filter. Using these techniques, the signal to noise ratio (SNR) is improved and the frequency domain performance approaches to theoretical one.

A new method for designing recursive and non-recursive frequency sampling filter has been published in [7]. The use of a hybrid real-coded GA for optimizing transition sample value has been investigated which yields the maximum stop band attenuation. A modification allows the coefficient word length to be optimized concurrently, thereby reducing overall number of design steps and simplifying the design process. The techniques are able to consistently optimize filter with up to six transition samples. The techniques presented in this paper could form the basis for integrating several of the optimizations. Investigation into increasing this integration by using a binary coded GA to optimize nonlinear phase, quantized coefficient FIR filter are introduced, with an analysis of the difficulty of the problem from a GA perspective [7].

For high speed low complexity filter design, it is common practice to constrain the filters" coefficient

to be power of two or a sum of power of two terms (p2), avoiding the full multiplication [7]. Tapped interconnection of different sub filters are sometimes used to enhance ripple and stop band attenuation performances. An extension of the simple cascade architectures, suitable for hardware implementation, is the polynomial sharpening techniques. The design of p2 sharpening filter based on a specific genetic algorithm has been proposed in the above article. The proposed scheme optimizes both the FIR sub-filter and the sharpening polynomial coefficient expressed as p2 terms. This allows getting better performances than the classical p2 design techniques when FIR filters with long impulse response are involved. Using this specific genetic algorithm with a particular free parameters encoding around a set of suitable leading values, allows obtaining a very high reduction of the computational cost. It has been shown in [7] that optimizing both the polynomial and the filter coefficient allow obtaining very good performances; sometimes better that the simple infinite precision sharpening techniques.

VI. DESIGNING TECHNIQUES OF FIR FILTERS

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

A. *Kaiser window*

Kaiser window is a well known flexible window and widely used for FIR filter design and spectrum analysis, since it achieves close approximation to the discrete pro late spheroidal functions that have maximum energy concentration in the main lobe. With adjusting its two independent parameters, namely the window length and the shape parameter, it can control the spectral parameters main lobe width and ripple ratio for various applications. Side lobe roll-off ratio is another spectral parameter and important for some applications. For beam forming applications, the higher side lobe roll-off ratio means, that it can reject far end interferences better . For filter design applications, it can reduce the far end attenuation for stop band energy. And for speech processing, it reduces the energy leak from one band to another.

B. *B. Optimal Filter Design Methods*

Optimization is the act of obtaining the best results under given circumstances. Optimization can be defined as the process of finding the condition that gives the maximum or minimum value of the function. If x* corresponds the minimum value of function f(x), the same point also corresponds to maximum value of the function –f(x). Thus optimization can be taken to mean minimization since the maximum of the function can be found by seeking of the negative of the same number

VII. ARTIFICIAL NEURAL NETWORKS

ANN has been wide utilized in the appliance of communication systems. The ANN is networks of simple process components known as neurons. They're connected to every different by weights. Every vegetative cell multiplies the incoming signals by the corresponding weights and sums then up. If these add or the activation price is over threshold, the vegetative cell changes its output. The network may be trained to adjust its weights within the learning section. Still, the network is in a position to perform some task additional simply than a traditional pc due to huge property and parallel operations of all the weather. It resembles brain in 2 respects: A neural network acquires information through learning. A neural network's information is hold on inside interneuron association strengths called conjugation weights. Artificial Neural Networks area unit being counted because the wave of the longer term in computing. they're so self-learning mechanisms that do not need the standard skills of a technologist. Currently, only a few of those neuron-based structures, paradigms really, area unit being employed commercially. The power and quality of artificial neural networks are incontestable in many applications including speech synthesis, diagnostic issues, medicine, business and finance, robotic management, signal processing, pc vision and lots of different issues that constitute the class of pattern recognition. for a few application areas, neural models show promise in achieving human-like performance over additional ancient AI techniques.

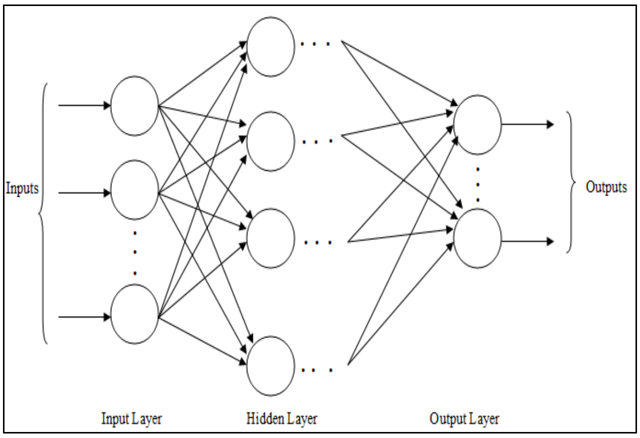


Figure 4: General Structure of Neural Network

An Artificial Neural Network (ANN) also known as "Neural Network (NN)" is a computational model based on the structure and function of biological neural network. In other words ANN is computing system which is made up of a number of simple processing elements (the computer equivalent of neurons, Nodes) that are highly interconnected to each other through synaptic weights. The number of nodes, their organization and synaptic weights of these connections determine the output of the network. ANN is an adaptive system that changes its structure/weights based on given set of inputs and target outputs during the training phase an produces final outputs accordingly. ANN is particularly effective for predicting events when the network have a large database of prior examples to draw. The common implementation of ANN has multiple inputs, weight associated with each input, a threshold that determine if the neuron should fire, an activation function that determine the output and mode of operation. The general structure of a neural network has three types of layers that are interconnected: input layer, one or more hidden layers and output layer as shown in Figure 3.
There are some algorithms that can be used to train an ANN such as: Back Propagation, Radial-basis Function, an Support

Vector learning, etc. The Back Propagation is the simplest but it has one disadvantage that it can take large number of iterations to converge to the desired solution [8]. In Radial Basis Function (RBF) network the hidden neurons compute radial basis functions of the inputs, which are similar to kernel functions in kernel regression. Speech has popularized kernel regressions, which he calls a General Regression Neural Network (GRNN) [3]. General Regression Neural Network (GRNN) is a variation of Radial Basis Function (RBF) network that is based on the Nadaraya – Watson kernel regression. The main features of GRNN are fast training time and it can also model nonlinear function. GRNN being firstly proposed by Sprecht in 1991 is a feed forward neural network model base on non linear regression theory. It approximates the function through activating neurons. In GRNN transfer function of hidden layer is radial basis function.

$$y'i = \frac{\sum_{i=1}^{n} yi * exp - D(x - xi)}{\sum_{i=1}^{n} exp - D(x - xi)}$$

$$D(x - xi) = \sum_{k=1}^{m} \left(\frac{xi - xik}{\sigma}\right)^2$$

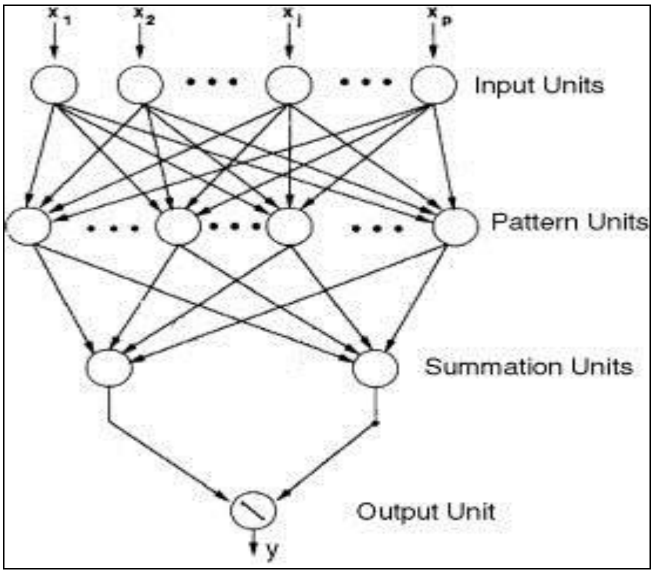


Figure 5: Generalized regression neural network

VIII. CONCLUSION

This paper suggests the neural network technique for designing linear phase FIR filter. Based on the various algorithms of neural network we concluded that the designed model of FIR filters using neural network are have better performance than the conventional design method of FIR filter. Carrying out literature review is very significant in any research project as it clearly establishes the need of the work and the background development. It generates related queries regarding improvements in the study already done and allows unsolved problems to emerge and thus clearly define all boundaries regarding the development of the research project

REFERENCES

[1]. Kaiser , J.F., "Non recursive digital filter design using I0-sinh window function", in proc. IEEE Int. Symp. Circuits and systems (ISCAS'74). 20-23, San Francisco, Calif, USA, 1974.

[2] Ricardo A. Losada, "Digital Filters with MATLAB", The MathWorks, Inc., May 2008.

[3] Rabiner Lawrence R., "Techniques for Designing Finite-Duration Impulse-Response Digital Filters", IEEE Transactions on Communication Technology, Vol. com -19, April 1971

[4] L. R. Rabiner, M.T. Dolan and J.F. Kaiser, "Some Comparisons between FIR and IIR Digital Filters", Vol- 53, Feb.1973

[5] W. Northcutt, "The Darwin Awards 3: survival of the fittest", Plume, 2004.

[6] LiXue, Z. Rongchun and Wang Qing, "Optimizing The Design of IIR Filter Via Genetic Algorithm", In Proceedings of in Proceedings of IEEE Int. Conf. on Neural &Signal Processing, pp. 476-479, December 14-17, 2003.

[7] P Gentili, F. Piazza and A. Uncini, "Improved Power of Two Sharpening Filter Design by Genetic Algorithm", IEEE Transaction on Filter Design, pp. 1375-1378, March 1996.

[8] P. Ramesh Babu, "Digital Signal Processing", 4th Edition, Scitech Publication Pvt. Ltd, 2007.

[9] Design of FIR Filter using Genetic Algorithm, Samrat Banerjee, Sriparna Dey, Supriya Dhabal, IJETR, Volume-3, Issue-6, June 2015

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