International Journal of Science and Research (IJSR)
ISSN (Online): 2319-7064
Impact Factor (2012): 3.358

An Opposition based Harmony Search Approach for Performance Improvement in Linear FIR Filter Design

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Abstract: ***In this paper, Improved opposition based harmony in linear phase of optimal design filters are used. In research optimization of RGA (real code GA), DE (Differential Evolution) and PSO (Particle Swarm Optimization). The Steps used are initialization randomly generated population of solutions, opposite solutions are also obtained and one is selected as a priori guess. In proposed OHS optimization method by multiplying with randomly generated variable, mean of Harmonic search and opposition based harmonics vector are used. Initialize harmony memory, each solution of harmony memory passes through memory consideration rule, pitch adjustment rule, opposition-based re-initialization generation jumping and then mean of actual solution and opposition-based solution are used, which gives the optimum result corresponding to the least error fitness in multidimensional search space of FIR filter design. Incorporation of different control parameters in the basic HS algorithm results in the balancing of exploration and exploitation of search space. Low pass, high pass, band pass, and band stop FIR filters are designed with the proposed OHS and other aforesaid algorithms individually for comparative optimization performance. A comparison of simulation results reveals the optimization effectiveness of the Proposed (Improved-OHS) over the other optimization techniques for the solution of the multimodal, non differentiable, nonlinear, and constrained FIR filter design problems.***

Keywords: RGA, DE, PSO, Proposed OHS, FIR Filters.

1. Introduction

Digital filters are a significant part of DSP. The extraordinary performance of filter is one of the key reasons that DSP has become so popular. It is necessary to have circuits capable of selectively one frequency or range of frequencies out of combined frequencies in a circuit. The circuit designed to perform separation and restoration of signals is called filter. Mostly filters are used for two purposes: signal ***separation*** and signal ***restoration***.

Signal separation- It is needed when a signal has been mixed with interference, noise, or other signals. Signal restoration is used when a signal has been changed in some way.

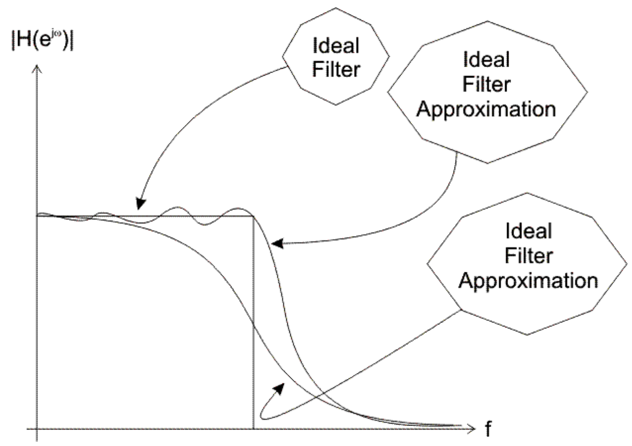
The easiest way to implement a digital filter is by *convolving* the input signal with the digital filter's *impulse response*. All possible linear types of filters can be made in this manner and this method of designing filter is called filter **kernel**.

Another way to design Filter is called Recursion. When a filter is implemented by using convolution technique, each sample in the output is calculated by *weighting* the samples in the input, and combined them together to design actual filter Response.
Recursion filter are extended version of this method. In this method of designing filter, previous value with respect to the inputs applied and this method contain a set of coefficients called Recursion Coefficient.

To obtain its impulse response, simply apply an impulse at input. The impulse responses of recursive filters are carried out which composed of sinusoids signals that exponentially decayed in amplitude and this makes the impulse responses *infinitely long*. However, the amplitude finally drops below the round-off noise of the system, and the remaining samples can be ignored. Because of its characteristics it is called as Infinite Impulse response filter and the filter designed by using convolution called Finite impulse Response filter.

Recursive filters are an efficient way of achieving a long impulse response, without having to perform a long convolution. They execute very rapidly, but have less performance and flexibility than other digital filters. Recursive filters are also called *Infinite Impulse Response* (IIR) filters, since their impulse responses are composed of decaying exponentials. This distinguishes them from digital filters carried out by convolution, called *Finite Impulse Response* (FIR) filters.

Digital filter is essentially a system which improves the quality of signals and extracts data (information) from signals or separates the Harmonics of the signal which are combined.



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.

The FIR filter can be implemented by obtaining the coefficients and order of the filter that meet desired specifications, which can be in the time-domain and/or the frequency domain. Mostly frequency domain method is used to design FIR Filter. Matched filters present a cross-correlation between the applied input signal and a known pulse-shape. The FIR convolution is a method of cross-correlation between the applied input signal and a time-reversed of the impulse response. Therefore, A matched-filter's impulse response is "designed" using sampling the known pulse-shape and using those samples in reverse order as the coefficients of the filter.

The linear time invariant (LTI) system and the filter are identical and are often used to perform spectral shaping or frequency selective filtering. The behavior of filtering action is determined by frequency response characteristics, which depends on the choice of system parameters that represents the coefficients of the difference equations. Thus, by suitable selection of the coefficients, we can design frequency selective filters which pass signals with frequency components in some bands while attenuate signals containing frequency components in other frequency bands.

The designing of filters based on the requirement of ripples in the pass band, the stop band attenuation and the transition width of the filters.

In the designing of filter the designer always has to compromise with the some other design specifications. The conventional optimization method and other classical optimization algorithms are not sufficient to produce optimize results and non uniform objective functions of FIR filters, the objective function can never converge to the global minimum solution. When a specific frequency response is required various designing methods are used.

- Window design method
- Frequency Sampling method
- Weighted least squares design
- Parks-McClellan method
- The Remez exchange algorithm is commonly used to find an optimal equiripple set of coefficients.

On the basis of frequency Response band passed or Rejected. The filter is classified into various types.

- Low-pass filter – in which frequencies are passed below cutoff frequency and rest of the frequencies are attenuated.
- High-pass filter – in which frequencies are passed above cutoff frequency and rest of the frequencies are attenuated.
- Band-pass filter – only frequencies above lower cutoff and below upper cutoff frequencies are passed and rest are attenuated.
- Band-stop filter or band-reject filter – only frequencies below lower cutoff and above upper cutoff frequencies are passed and rest are attenuated.
- Notch filter – It rejects one specific frequency - an extreme band-stop filter.
- Comb filter – It has multiple regularly spaced narrow pass band which appears like comb.
- All-pass filter – all frequencies are passed, but the output phase gets modified.
- Cutoff frequency – It is the frequency above or below which the filter will not pass signals.
- Transition band- It is a band of frequencies between a passband and stopband.
- Ripple is the unwanted content present in passband and stop bands of the response.
- The order of a filter is the degree of the approximating polynomial and in passive filters corresponds to the number of elements required to build it. Increasing order increases roll-off and brings the filter closer to the ideal response.

Genetic Algorithm seems to be attracted a considerable attention and standard GA (also known as real-coded GA (RGA)) shows good performance for finding the promising regions of the search space, they are ineffective in determining the global optimum solution and prone to revisiting the same suboptimal solution. In order to reduce the problem of RGA, orthogonal genetic algorithm (OGA), hybrid-Taguchi GA (TGA) are proposed. Tabu search, Simulated Annealing (SA), Bee Colony algorithm (BCA), differential evolution (DE), particle swarm optimization (PSO), opposition-based BAT (OBAT) algorithm, some variants of PSO like PSO with Quantum Infusion (PSO-QI), adaptive inertia weight PSO, chaotic mutation PSO (CMPSO), Novel PSO (NPSO), Gravitational search algorithm (GSA), seeker optimization algorithm (SOA), some hybrid algorithms like DE-PSO have also been used for the filter design problems with varying degree of comparative optimization effectiveness. Most of the above algorithms show the problems of fixing algorithm's control parameters, premature convergence, stagnation, and revisiting of the same solution over and again. In order to overcome these problems, in this paper, a novel optimization algorithm called Improved opposition-based harmony search (OHS) is employed for the FIR filter design.

Tizhoosh introduced the concept of opposition-based learning (OBL). This notion has been applied to accelerate the reinforcement learning and the back propagation learning in neural networks. The main idea behind OBL is the simultaneous consideration of an estimate and its corresponding opposite estimate (i.e., guess and opposite guess) in order to achieve a better approximation for the current candidate solution. In the recent literature, the concept of opposite numbers has been utilized to speed up the convergence rate of an optimization algorithm, for example, opposition-based differential evolution (ODE). This idea of opposite number OBL be incorporated during the harmony memory (HM) initialization and also for generating the New Harmony vectors during the process of HS. OBL has been utilized to accelerate the convergence rate of the HS. Hence, our proposed approach has been called as opposition-based HS (OHS). OHS uses opposite numbers during HM initialization and also for generating the new HM during the evolutionary process of HS.

This paper describes the comparative optimal designs of linear phase low pass (LP), high pass (HP), band pass (BP) and band stop (BS) FIR digital filters using other aforesaid algorithms and the proposed OHS approach individually. The OHS does not prematurely restrict the searching space. A comparison of optimal designs reveals better optimization usefulness of the proposed algorithm over the other optimization techniques for the solution of the multimodal, non differentiable, highly nonlinear, and constrained FIR filter design problems

2. Problem Identification

1. The problems of fixing algorithm's control parameters, premature convergence, stagnation, and revisiting of the same solution over and again.
2. The multimodal, non differentiable, highly nonlinear, and constrained FIR filter design problems.
3. Standard GA (also known as real-coded GA (RGA)) shows a good performance for finding the promising regions of the search space, but they are inefficient in determining the global optimum and prone to revisiting the same suboptimal solution.
4. PSO destroys the simplicity of the algorithm and leads to an undesirable computational overhead

3. FIR Filter Design

Digital filters are classified as finite impulse response (FIR) or infinite impulse response (IIR) filter depending upon whether the response of the filter is dependent on only the present and past inputs or on the present and past inputs as well as previous outputs, respectively.

$$H(z) = h(0) + h(1)z^{-1} + \cdots + h(N)z^{-N} \qquad (1)$$

or

$$H(z) = \sum_{n=0}^{N} h(n)z^{-n}, \qquad (2)$$

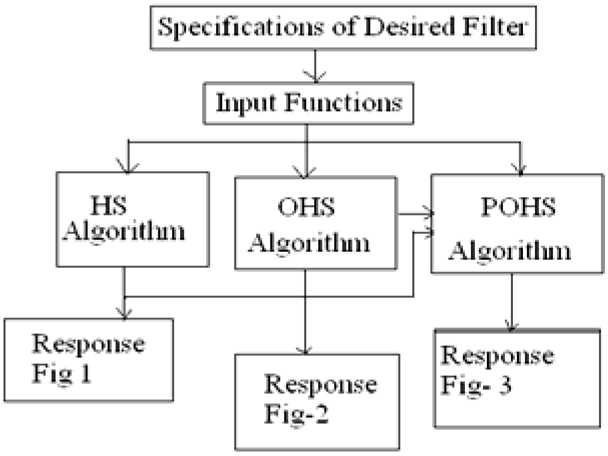
Where h (n) is called impulse response. The difference equation representation is

$$y(n) = h(0)\,x(n) + h(1)\,x(n-1) + \cdots + h(N)\,x(x-N)$$

The order of the filter is N, while the length of the filter (which is equal to the number of coefficients) is N+1.The FIR filter structure are always stable and can be considered to have linear phase response. The impulse responses h(n) are to be calculated in the designed process and the values of h (n) will calculate the type of the filter. Other desired characteristics are short filter length, short frequency transition beyond the cut-off point, the ability to control the attenuation in the stop band. In many applications -3dB frequency has become a recognizable parameter for defining the cut-off frequency f_c (the frequency at which the magnitude reached an absolute value of 0.5). The effect of using the 3 dB measure is that it varies with filter length since the sharpness of the transition width is a function of the order of filter. Additionally, as the filter order increases, the transition width decreases, and -3dB approaches fc asymptotically. In any filter design, some of these parameters are fixed while others are optimized. In this paper, the Improved OHS is applied in order to obtain the actual filter response. The filter are designed with h(n) individuals or In this paper, the HS and OHS is applied in order to obtain the actual filter response as closer to the ideal filter. The designed FIR filter with h(n) individuals or particles/solutions is positive symmetric and of even order. The length of h (n) is N+1; that is, the number of coefficients is also N+1. In each iteration, solutions are updated. Fitness values of these updated solutions are evaluated using the new coefficients and the new error fitness function. The solution produced after a certain number of iterations and/or after the error fitness value below a certain limit, which assumed to be the optimal result. The error is used to evaluate the error fitness value of the solution. It compares the error between the magnitudes of frequency responses of the ideal and the actual filters. An ideal filter has a magnitude of one on the pass band and a magnitude of zero on the stop band. The error fitness function/value is minimized using the evolutionary algorithms such as RGA, PSO, DE, OHS and Improved OHS individually. The individual that have minimum error fitness values represent the better filter, that is, the filter has better frequency response.

4. Proposed Methodology

The improvisation of filter response is done in the following way:



In proposed algorithm of Improved oppositiobased Harmony search algorithm. Harmony memory size (HMS) are initialize as done in harmony search algorithm or the number solution vector . Check for harmony memory Considering rate (HMCR) whose values lies between 0 and 1. Check for Pitch Adjustment Rate (PAR) whose value should be between 0 and 1. On the basis of random selection of harmony vector from HMS. If variable compared having fitness value less than worst fitness variable, the old harmony repalced by New Harmony.

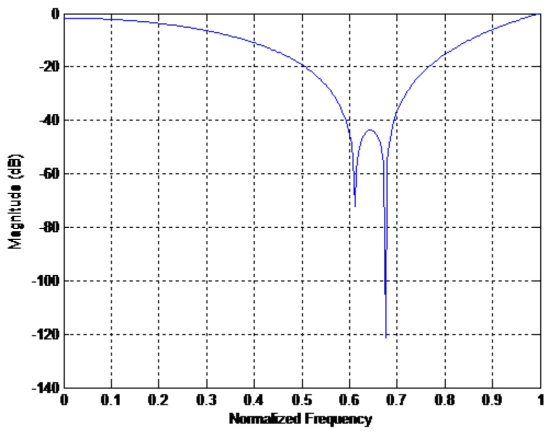


Figure 1

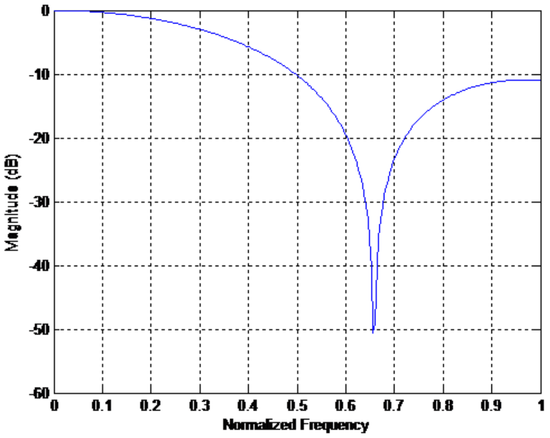


Figure 2

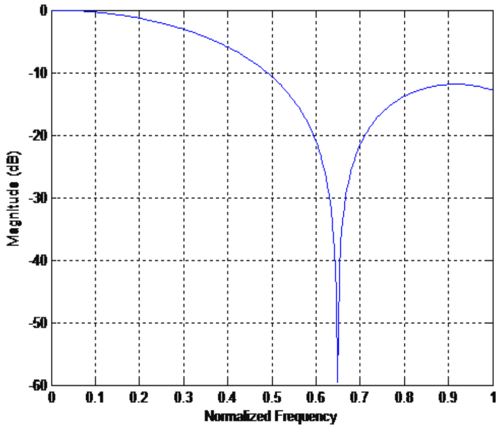


Figure 3

Set Harmony Index by the Random Selection of Harmony

Extract Value of harmony from Memory using Harmony Index
Create CMMask (using HMCR)
Create NHMask = 1 - CMMask
Create PAMask = (using PAR CMMask)

Create NewHarmony = (using CMMask old harmony and NHMask);

Call filter transfer function and check the improved response

5. Expected Outcome

1. It is revealed that Improved-OHS has the ability to converge to the best quality near optimal solution as compared to PM, PSO, OHS etc.
2. Improved-OHS possesses the best convergence characteristics in moderately less execution times as compared to the above listed algorithms.
3. Improved -OHS demonstrates better performance in terms of magnitude response, minimum stop band ripple, and maximum stop band attenuation with a very little declining in the transition width.
4. Improved-OHS may be used as a good optimizer to obtain the optimal filter coefficients in any practical digital filter design problem of digital signal processing systems.

6. Conclusion

In this paper, an improved opposition-based harmony Search (Improved-OHS) algorithm is applied to the solution of the constrained, problem of multimodal FIR filter design, yielding optimal coefficients of filters. Comparison of the results of Park McClellan, Real Genetic Algorithm (RSA), PSO, DE, and OHS algorithm has been done. It has been revealed that Improved-OHS has the ability to converge the best quality near optimal solution and possesses the best convergence characteristics in quite less execution times among the above mentioned algorithms. The simulation results clearly indicate that the Improved-OHS gives better performance in terms of magnitude response, lowest stop band ripple, and maximum stop band attenuation with a very little distortion in the transition width. Thus, the Improved-OHS may be used as a good optimizer to obtain the optimal filter coefficients in any practical digital filter design problem of digital signal processing systems.

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