International Journal of Advanced Engineering and Technology
www.newengineeringjournal.com
ISSN: 2456-7655
Received: 05-07-2021, Accepted: 24-07-2021, Published: 11-08-2021
Volume 5, Issue 2, 2021, Page No. 01-07

Identified optimal infinite impulse response filters for eight band audio equalizer using bacterial foraging algorithm (BFA)

Ishaku Thlama Balami[1], E C Anene[1], Abdullahi Muhammad[2], Mohammed Ahmed[1], Saidu Mohammed Kumo[1]
[1] Department of Electrical/Electronic Engineering, Abubakar Tafawa Balewa University Bauchi, Bauchi, Nigeria
[2] Department of Physics, Federal University Kashere, Akko Gombe, Nigeria

Abstract
This paper describes the impact of optimal identified low-order Infinite Impulse Response (IIR) filters from the higher-order counterpart. A 4th order IIR filter were design using classical method from first principle using Bilinear Transformation (BLT) method and were used in identifying the equivalent IIR filter in 2nd order using an optimization method. The reason for the identification is to minimize cost of implementation. An eight band audio equalizer was developed and analyzed. It was also observed that the Bacterial foraging optimization **(**BFO) designed filters equalizer output were slightly higher in gain than that of BLT with -12 dB that is 23 % boost and -14 dB 2.4 % attenuation respectively with respect to the original signal gain of -13 dB. This attenuation can be seen at also roll-off point of BLT with the higher level of ripples. The effectiveness of proposed BFA has been also established with fast convergence speed, high performance in digital equalization.

Keywords: butterworth, digital equalizer, bilinear transformation (BLT), infinite impulse response (IIR), bacterial foraging optimization (BFO)

Introduction

Equalization is a very well-known audio effect utilized in many devices used to play audio signals. Also known as Equalizer, it filters the incoming sound signal into a minimum of three basic tones: bass, mid-range and treble. It gives the user the choice of equalizing the tones according to preferred interest. Some frequencies are boosted while some other frequencies are cut and some others remain unaffected through the use of different gain settings which may be a positive boost if the gain is greater than one, negative cut if the gain is less than one attenuate or equal to zero unchanged. Equalizations refer to a filtering process where the frequency content of an audio signal is adjusted to make the source sound better, adjust for room acoustics, or remove noise that may be in a frequency band different from the desired signal. Most audio equalizers have a number of bands that operate on the audio signal in parallel with the output of each filter, added together to form the equalized signal. This structure is shown in figure 1. In this effect, all the filters are controlled independently. If low frequencies are boosted it gives a bass effect while boosting higher frequencies gives a treble effect (Rao *et al* 2010) [31]. Each gain constant in the audio equalizer is used to adjust the relative signal amplitude of the output of each band. The input signal is always added to the output such that if all the gain values are zero the signal is unchanged (Brown *et al* 1998 & Bal'azs, 2008) [6, 5]. The center frequencies and the number of bands in analog audio equalizers vary widely from one manufacturer to another. An eight-band equalizer with center frequencies at 60, 1711.5, 5024.5, 8337.5, 11650.5, 14963.5, 18276.5 and 19928 Hz is implemented in MATLAB/SIMULINK as in Figure 1 (Li, 2007 and Li and Jean, 2013) [22, 21].

Filters are signal conditioners. They function by accepting an input signal, blocking pre-specified frequency components, and passing the original signal minus those components to the output. In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range. There are two main kinds of filters, analog and digital. They are quite different in their physical makeup and how they work.

An analog filter uses analog electronic circuits made up from components such as resistors, capacitors and op amps to produce the required filtering effect. Such filter circuits are widely used in such applications as noise reduction, video signal enhancement, graphic equalizers in hi-fi systems, and many other areas. There are well-established standard techniques for designing an analog filter circuit for a given requirement. At all stages, the signal being filtered is an electrical voltage or current which is the direct analogue of the physical quantity (e.g. a sound or video signal or transducer output) involved. The most common filter Types are the Butterworth, Chebyshev, and Bessel types. Many other types are available, but 90% of all applications can be solved with one of these three. Butterworth ensures a flat response in the passband and an adequate rate of roll-off. A good "all-rounder" the Butterworth filter is simple to understand and suitable for applications such as audio processing. The Chebyshev gives a much steeper roll-off, but passband ripple makes it unsuitable for audio systems. It is superior for applications in which the passband includes only one frequency of interest (e.g., the derivation of a sine waves from a square wave, by filtering out the harmonics). The Bessel filter gives a constant propagation delay across the input frequency spectrum.

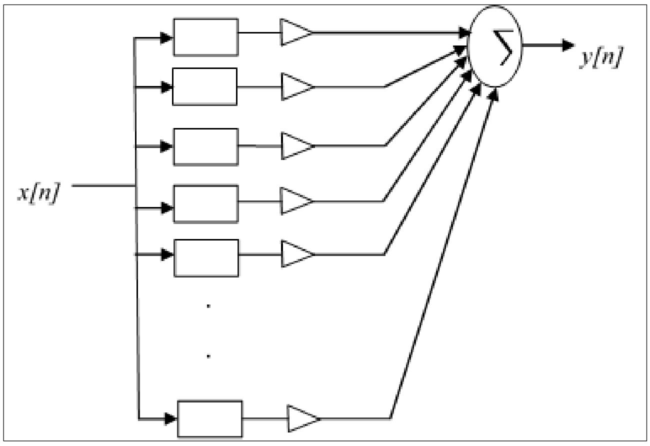

Fig 1: N-band IIR Equalizer Data-Flow Diagram

A digital filter is a mathematical algorithm implemented in hardware and/or software that operates on an input signal to produce a digital output signal for the purpose of achieving a filtering objective. Digital filters can have characteristics which are not possible with analog filters, such as a truly linear phase response. Unlike analog filters, the performance of digital filters does not vary with environmental changes, for example, thermal variations. This eliminates the need to calibrate periodically. Digital filters are broadly divided into two classes, namely infinite impulse response (IIR) and finite impulse response (FIR) filters. Either type of filter, in its basic form, can be represented by its impulse response sequence. Digital Filters are designed by using the values of both the past outputs and the present input, an operation brought about by convolution. If such a filter is subjected to an impulse, then its output need not necessarily become zero. The impulse response of such a filter can be infinite in duration. Such a filter is called an Infinite Impulse Response filter or IIR filter. The infinite impulse response of such a filter implies the ability of the filter to have an infinite impulse response. This indicates that the system is prone to feedback and instability. The content of this work is the design of IIR filters using the classical design method and optimal design using MATLAB. The design of IIR filters proceeds through a vastly different set of steps. The design of IIR filters is closely related to the design of analog filters, which is a widely studied topic.

An analog filter is usually designed and a transformation is carried out into the digital domain. Two transformations exist – the impulse invariant transformation and the bilinear transformation. In this work, the focus is on designing fourth-order IIR filters to meet a set of specifications. The designed IIR filters are characterized by a significantly lower order than the corresponding FIR filters. There it is shown that the best IIR filter is less complex than the optimum FIR filter. The price to pay is the non-linear phase of the IIR filters, which is unavoidable. The Infinite Impulse Response filter type that is adopted in this work is Butterworth. Butterworth filters are causal in nature and of various orders, Butterworth or maximally flat filters have a monotonic amplitude-frequency response which is maximally flat at zero frequency response and the amplitude-frequency response decreases logarithmically with increasing frequency. The Butterworth filter has minimal phase shift over the filter's bandpass when compared to other conventional filters.

The Digital Filter Design problem involves the determination of a set of filter coefficients to meet a set of design specifications. These specifications typically consist of the width of the passband and the corresponding gain, the width of the stopband(s) and the attenuation therein; the band edge frequencies (which give an indication of the transition band) and the peak ripple tolerable in the passband and stopband(s). The Bilinear Transformation method was the method used to map the analog classical fourth-order design to the digital classical fourth-order design filters, which overcomes the effect of aliasing that is caused due to the analog frequency response containing components at or beyond the Nyquist frequency.

Bacterial foraging optimization (BFO) method

The optimization technique developed by Prof. K. M. Passino inspired by the social foraging behavior of *Escherichia Coli* which is used in Identifying the unknown second other filter coefficients. BFO algorithm based on foraging strategies of E Coli bacterium cells that tend to eliminate poor foraging strategies. BFO formulate the foraging behavior by bacteria such that it maximizes their energy intake per unit time. It consists of mainly four steps are chemotactic, swarming, reproduction and elimination/dispersal respectively. Natural selection of those bacteria that have strong foraging strategies and elimination of those that have poor foraging strategies occurs.

Problem Statement

Consider an IIR filter with an input-output relationship given by:

$$y(k) + \sum_{i=1}^{M} b_i y(k-i) = \sum_{i=0}^{L} a_i x(k-i) \qquad (1)$$

Where $x(k)$ and $y(k)$ are the filter's input and output, respectively, M ($\geq L$) is the filter order. The transfer function of this IIR filter can be written as:

$$H(z) = \frac{A(z)}{B(z)} = \frac{\sum_{i=0}^{L} a_i z^{-1}}{1 + \sum_{i=1}^{M} b_i z^{-1}} \qquad (2)$$

The parameters$a_0, a_1, a_2, \cdots, a_L, b_1, b_2, \cdots, b_M$ appearing in Equation (1) and (2) are the filter coefficients, and they determine the characteristics of the filter. The design of this filter can be stated as the optimization problem of objective function $J(w)$.

$$\min J(\omega) \qquad (3)$$

where $\omega = \{a_0, a_1, a_2, \cdots, a_L, b_1, b_2, \cdots, b_M\}$ is the filter coefficient vector. The aim is to minimize the cost function $J(w)$ by adjusting w.

$$J(\omega) = \frac{1}{N}\sum_{k=1}^{N} (d(k) - y(k))^2 \qquad (4)$$

Where $d(k)$ and $y(k)$ are the desired and actual responses of the filter, respectively and N is the number of samples used for the calculation of objective function.

The goal of this filter mathematical modeling in system identification is to adjust the digital filter coefficients to match an unknown system transfer function. In other words, the parameters of the filters are successively adjusted by filter modeling problem using the optimization algorithms until the error between the output of the filter and the unknown system is minimized (Ranjit & Sandeep, (2012), and Durmus & Gün (2011)) [34, 11].

Significance of the Study

Digital equalizers offer many advantages over analog equalizers in terms of quality, typical problems of analog equalizers like group delay distortion, inter-band interference, difficulty to achieve a perfectly flat frequency response, poor global system controllability, may be completely eliminated or attenuated within some desired mathematical precision by using digital signal processing (DSP) techniques.

Based on the Bacterial foraging Optimization design technique not only provides the highest stopband attenuation but also the quality output in terms of ripples and transition width which are much better than others.

This approach eliminates algebraic and programming mistakes, the framework to design a digital IIR filter with near-linear phase over the passband and minimized quality factor, subject to constraints on the magnitude response.

Justification of the Study

The justification of this research lies in the following:

Minimization of an objective function (typically the mean square error between desired response and estimated filter output) is often performed by gradient-based iterative search algorithms. However, when the error surface (objective function) is multimodal and/or non-smooth, gradient-based optimization methods often cannot succeed in converging to the global minimum.

One of the advantages of the BFA approach is that it performs a very thorough search since it has many cascaded loops (chemotaxis, swarming, reproduction and elimination dispersal). Since this increases the computational complexity of the BFA approach, the negative impact on the execution time is inevitable.

Research Gap

From the foraging discussion, many kinds of research have applied heuristic optimization for the design and identification of digital filters. Use of bacteria foraging algorithm in the identification of digital filter in equalizer applications hard not been done. This research work presented focused on this application to identify a second-order IIR digital filter from a fourth-order classically designed equivalent. This will minimize the cost of real-time implementation in terms of computation and protocol resources.

System Specification

A simple block shown in Figure 1 highlights the implementation of an 8-band graphic equalizer, which consists of eight IIR filters connected in parallel. Table 1 gives a summary of the frequency range of the filters, with the sampling rate of 40 kHz.

Table 1: Cutoff frequencies for the 8-bands

S/N	Band Numbers	Range (in Hz)
1	LPF1	0-60
2	BPF2	50-3373
3	BPF3	3363-6686
4	BPF4	6676-9999
5	BPF5	9989-13312
6	BPF6	13302-16625
7	BPF7	16615-19938
8	HPF8	19928

Filter to its digital equivalent. The two basic ways to perform the desired transformation is to use the bilinear transformation or impulse-invariant technique.

From Table 2, the normalized 4th order Butterworth is given as;

$$H(s) = \frac{1}{(s^2+0.765s+1)(s^2+1.848s+1)}$$

To transform analog Lowpass filter H(s) with unity cutoff frequency to Lowpass filter H(s) with a cutoff frequency $f_c = 60Hz$ and $\omega_c = 2\pi f_c$ substitute $s = \frac{s}{\omega_c}$; $\omega_c = 377.04 rad/sec$

Table 2: Normalised Denominator Polynomials for Butterworth filters

n	Normalised Denominator Polynomials in Factored Form
1	(1+s)
2	$(1+1.414s+s^2)$
3	$(1+s)(1+s+s^2)$
4	$(1+0.765s+s^2)(1+1.848s+s^2)$
5	$(1+s)(1+0.618s+s^2)(1+1.618s+s^2)$
6	$(1+0.518s+s^2)(1+1.414s+s^2)(1+1.932s+s^2)$
7	$(1+s)(1+0.445s+s^2)(1+1.247s+s^2)(1+1.802s+s^2)$
8	$(1+0.390s+s^2)(1+1.111s+s^2)(1+1.663s+s^2)(1+1.962s+s^2)$
9	$(1+s)(1+0.347s+s^2)(1+s+s^2)(1+1.532s+s^2)(1+1.879s+s^2)$

To obtain the transfer function of each Bandpass filter, the above equation was used for different values of W andω_0^2.

To transform analog Lowpass filter H(s) with unity cutoff frequency to Highpass filter H(s) with a cutoff frequency $f_c = 19928\, Hz$ and $\omega_c = 2\pi f_c$ substitute $s = \frac{\omega_c}{s}$; $\omega_c = 125227\, rad/sec$.

Table 3: Designed Analogue Filters Transfer Functions

Design Case	Transfer Function
Case1(LPF1)	$H(s) = \frac{1}{s^4 + 2.61313s^3 + 3.41421s^2 + 2.61313s + 1}$
Case2 (BPF2)	$H(s) = \frac{0.048919s^2}{s^4 + 0.312789s^3 + 0.050389s^2 + 0.00023s + 5.3965E-07}$
Case3 (BPF3)	$H(s) = \frac{0.059612s^2}{s^4 + 0.345285s^3 + 00.269017s^2 + 0.036152s + 0.01096264}$

Case 4 (BPF4)	$H(s) = \frac{0.090142s^2}{s^4 + 0.424596s^3 + 0.80704s^2 + 0.152196s + 0.12848542}$
Case 5 (BPF5)	$H(s) = \frac{0.178366s^2}{s^4 + 0.597265s^3 + 2.000138s^2 + 0.54404s + 0.82971351}$
Case 6 (BPF6)	$H(s) = \frac{0.090142s^2}{s^4 + 0.424596s^3 + 0.80704s^2 + 0.152196s + 0.12848542}$
Case 7 (BPF7)	$H(s) = \frac{0.059612s^2}{s^4 + 0.345285s^3 + 00.269017s^2 + 0.036152s + 0.01096264}$
Case 8 (HPF8)	$H(s) = \frac{0.048919s^2}{s^4 + 0.312789s^3 + 0.050389s^2 + 0.00023s + 5.3965E - 07}$

Identification

System identification is done by mathematically modeling an unknown system using its input-output data. This is achieved by varying the parameters of the developed model so that for a set of given inputs, its output matches that of the system under consideration. For a plant whose behavior is not known, an adaptive system can be modeled and its parameters can be continuously adjusted using an adaptive algorithm. By the use of such adaptive algorithms, the required parameters can be obtained such that the output of the plant and the model are the same for the same set of inputs, which is the goal of system identification. The identified model depicts the characteristics of the given system. The problem of system identification in many instances can be formulated as a problem of adaptive IIR filtering. Different adaptive algorithms can be applied to adjust the feedforward and feedback parameters of the recursive system (Deivaseelan & Babu, 2012) [8]. In the system identification configuration, the adaptive algorithm searches for the system coefficients such that its input/output relationship matches closely to that of the unknown system.

Optimal IIR Filter Coefficients

As stated in Equation 30, objective function and the bacterium having a minimum function (J) which is retained for the next generation. For swarming, the distances of all the bacteria in a new chemotactic stage are evaluated from the global optimum bacterium till that point. To speed up the convergence, a simple heuristic rule to update one of the coefficients of the BFA algorithm was formulated as in Table 6 in chapter 4. The flow chart of the iterative algorithm is shown in Figure 7.

Results

The transfer function of fourth-order Butterworth analogue filters is presented in Table 3.

Table 4 gives the transfer function with optimal coefficients calculated using the BFO algorithm. The numerator and denominator polynomials contain the optimal values of the filter's transfer function coefficients.

As can be seen, Table 4 is second-order functions identified from the classically designed fourth-order filters.

Table 4: Filters Transfer Functions Calculated using BFO Algorithm

Design Case	Frequency Bands(Hz)	Transfer Function
Case1(LPF1)	60	$H(z) = \frac{-0.5216}{1 - 0.2935z^{-1} - 0.0506z^{-2}}$
Case2 (BPF2)	50-3373	$H(z) = \frac{0.9255 - 0.9255z^{-1}}{1 - 0.1437z^{-1} + 0.6893z^{-2}}$
Case3 (BPF3)	3363-6686	$H(z) = \frac{0.3890 - 0.3890z^{-1}}{1 + 0.2297z^{-1} - 0.1387z^{-2}}$
Case 4 (BPF4)	6676-9999	$H(z) = \frac{0.4547 - 0.4547z^{-1}}{1 + 0.2128z^{-1} - 0.0352z^{-2}}$
Case 5 (BPF5)	9989-13312	$H(z) = \frac{0.2567 - 0.2567z^{-1}}{1 + 0.1909z^{-1} - 0.0630z^{-2}}$
Case 6 (BPF6)	13302-16625	$H(z) = \frac{0.4450 - 0.4450z^{-1}}{1 + 0.1462z^{-1} + 0.1360z^{-2}}$
Case 7 (BPF7)	16615-19938	$H(z) = \frac{0.0064 - 0.0064z^{-1}}{1 - 0.2344z^{-1} - 0.0482z^{-2}}$
Case 8 (HPF8)	19928	$H(z) = \frac{-0.2261 + 1.3208z^{-1} - 0.2261z^{-2}}{1 - 0.0418z^{-1} + 0.6251z^{-2}}$

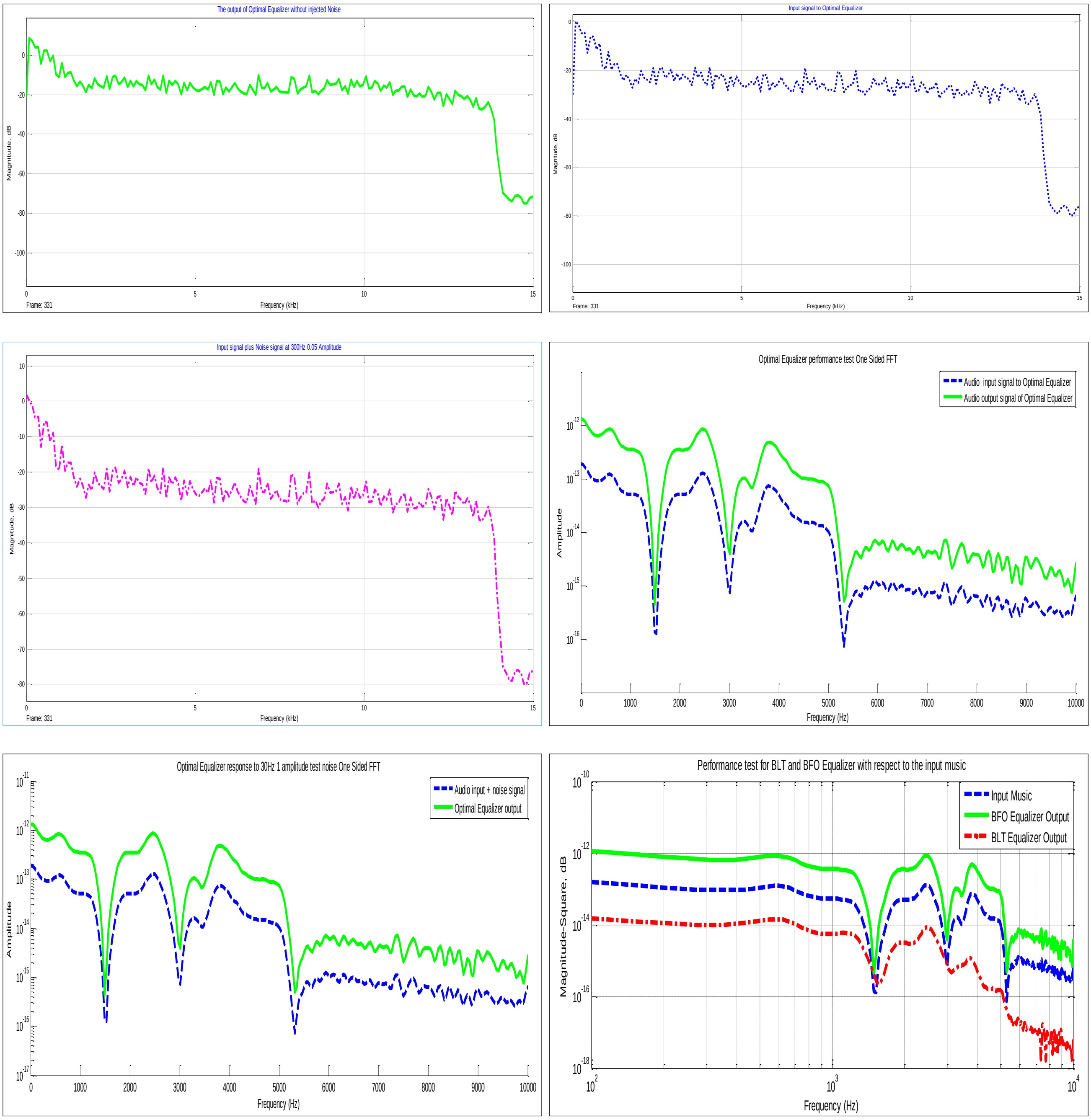

Fig 2: Audio Signal Graph for BLT and BFO Designed Equalizers

Discussion

Plots showing the gain and frequency responses of input signal BLT and BFA equalizers were obtained as shown in Figure 21. Using zoom command and closely examining the frequency response plot, it was noted that the BFA equalizer output had slightly higher gain across the frequencies of 10 kHz, while the BLT equalizer output is slightly below the original signal. This gain difference was -12dB that is 23% (-0.0470) boost for the BFA equalizer, -14dB 2.4% (0.004) attenuation for the BLT equalizer against the original signal with the gain of -13dB (0.2042). It was also observed that at the roll-off point the BLT equalizer output has slightly higher ripples if compared to the original signal and the BFA output signal.

Conclusion

The effectiveness of proposed BFA has been also established for the identification of all IIR digital filter bands. The proposed BFA possess fast convergence speed in term of a number of function evaluations to achieve the global solution. Results obtained for the BFA justify the potential of the proposed algorithm for the design of IIR digital filter. Therefore, the system performance of the digital equalizer boosts the low and high-frequency components.

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