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Many Approaches for Obtaining Least Noisy Signal using Kaiser Window and Genetic Algorithm

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ABSTRACT
In a communication system in a signal is an information holding portion which is a route for the system. But this route may not be clean and plane route when data is transmitted from signal there will be some noise is added to the signal and signal turned into a noisy signal, so that is why we want to remove this noise from signal to get information from the signal. In this research, we find some approaches to remove this noise from the signal. These approaches are Wavelet filter based on Butterworth IIR filter and Kaiser Window FIR filter for the analysis of the signal. The newly designed Wavelet filter presents and idea for signal analysis and finding difficulties during any communication. The other techniques are FIR filter that mostly uses for videos. It is somehow valuable technique for filtering signal. This technique uses for complicated purposes like signal preconditioning and various communication application. Most of the FIR filters designing methods is based on Windows method, frequency sampling method and Optimal Sampling method these approaches has been proposed to obtain the clean signal from a noisy signal.

Keywords
DSP, OMRA, FIR digital Filter

1. INTRODUCTION
The signal that usually transmitted may contain useless things which are called noise. It is a very hot topic among researchers to clean a signal and make a smooth transmission. Meanwhile, earlier days both type that I have introduced FIR and IIR have away through several signs of progress in different eras.

1.1 Wavelet
Wavelet is functions that changed data into a different frequency and then study each module with a resolution and then matches to its scale. This approach is far better than an earlier approach where the signal covers break and sharp points. Wavelet was introduced freely in the field of physics, engineering, and geology.

1.2 Fir Filters:
FIR filters are one of two main types of digital filters used in Digital Signal Processing (DSP) applications, the other type being IIR. "FIR" signifies "Limited Impulse Response.If you put in a drive, that is, a solitary "1" test took after by numerous "0" tests, zeroes will turn out after the "1" test has advanced through the postpone line of the channel. In the normal case, the drive reaction is limited because there is no input in the FIR. An absence of criticism ensures that the drive reaction will be limited. In this way, the expression "limited motivation reaction" is almost synonymous with "no input.The design of digital FIR filters includes the design of filter transmission function numbers n, n=0, 1.., N that deliver goal frequency answer. A number of ways like window method, optimal sampling and frequency sampling have been presented for this purpose. An extensive amount of windowing methods were planned earlier, founded on changed limits like sampling frequency, cut off frequency, passband ripple, stopband attenuation, order of the filter, filter length, width of the lobes, etc. Best simple window role is rectangular window role where the signal is cut by increasing the impulse answer with a function with unit amplitude inside a known variety and overlooking the constants external the window. It was stated the occurrence of passes and ripples in frequency answer due to the great oscillation or side parts produced by sudden truncation. To minimize these things, windows that do not hold sudden breaks in their time and frequency area features are selected. To overcome the limits of rectangular window function, few windows were introduced for the implementation of FIR digital filters that might not have speedy break in time and frequency area characteristics. In this concern, most usually used windows are Hamming, Henning, Exponential window, Blackman, Chebyshev. As these methods suffer from the absence of plan flexibility, Kaiser Window has been used to get clean signals in the research work. Kaiser Window with varying passband and stopband ripples has been used. Kaiser window function was first suggested by Kaiser to design non-recursive digital filters by using the improved zeroth order Bessel function. Late, Kaiser Window has gone through some changes offered by different researchers.

2. BACKGROUND
2.1 Statistically Matched Wavelet Design:
This research offering the design of statistically coordinated wavelet filter rows that have been widely deliberate where it has been expressed as a reserved optimization problematic

2.2 Constructing Wavelets from Desired Signal Functions
In this research, writer offerings the best applications of orthonormal multiresolution studies (OMRA) use either Daubechies, Meyer’s, or Lemuria’s wavelets

2.3 Distributed Signal Processing via Chebyshev Polynomial Approximation
In this research combinations of the graph, multiplier operations are an inner class of linear operators for processing signals well-defined on graphs.

2.4 In this study Speech Denoising Using Different Types of Filters

Speech Recognition is a broader solution which refers to tools that can identify a speech without actuality targeted at single speaker such call system can identify the arbitrary voice.

These are some techniques that were used for noisy signals. But there are some flaws in these techniques so that's why these are not famous.

3. RESULT AND TECHNIQUES

Kaiser Window

Kaiser Window permits distinct control on the width of the foremost part and decrease of the side parts. Kaiser window is defined by the following equation:

$$w_k(n) = \int \frac{I0\{[1-(\frac{n-\alpha}{\alpha})2]^{1}/_{2}\}}{I0\ \beta}$$

$for\ 0 \le n \le N$ (1)

Here, $\alpha = \frac{N}{2}$ and I_0 is the first kind 0th order modified Bessel function. Here, (N+1) is the length of the window,

$$N = \frac{A-8}{A-82.285\Delta\omega} \quad (2)$$

If the transition width is $\Delta\omega = \omega_{stop\_band} - \omega_{pass\_band}$ (3)

and ωs is stopband edge frequency and ωp is passband edge frequency. β is the shape of the window which can be selected independently. There is ripple of $\pm\delta 1$in the passband and $\delta 2$in the stopband. For the FIR filter design using Kaiser Window, minimum ripple of $\delta 1$and $\delta 2$ a a is considered. Let the minimum ripple be represented by δ. If attenuation is defined in dB, $A = -20 log 10\delta$ (4)

The value of β can be found out by using the following equations:

$\beta = 0.1102\ A - 8.7\ for\ A > 50$

$\beta = 0.5842\ A - 21\ 0.4 + 0.07886\ A - 21\ for\ 21 \le A \le 50,$

$\beta = 0\ for\ A < 21$ (5)

Table1:Comparison of Kaiser Window with other Window.

Name of Window	Time-domain sequences, $\omega\ n\ ,0 \le n \le M$	Time-domain sequences, $\omega\ n\ ,0 \le n \le M$
Rectangular	1	Narrowest main lobe about $\frac{4\pi}{M+1}$& large side lobes about -13 dB.
Barlett (Triangular)	$1 - \frac{2\lvert N - \frac{M-1}{2}\rvert}{M-1}$	Medium main lob of $\frac{8\pi}{M}$ & Side lobs of -25 db. Leakage Factor 0.28%.
Blackman	$0.42 - 0.5\cos\frac{2\pi n}{M-1} - 1 + 0.8\cos\frac{4\pi n}{M-1}$	Large main lob $\frac{12\pi}{M}$&Very good side lobs-57 dB. Leakage Factor 0%.
Hamming	$0.54 - 0.46\cos\frac{2\pi n}{M-1}$ for n=0, 1,…,M-1	Medium main lob $\frac{8\pi}{M}$ &Good side lobs -41 dB. Leakage Factor 0.03%.
Kaiser	$w_k\ (n) = \int \frac{I0\{[1-(\frac{n-\alpha}{\alpha})2]^{1}/_{2}\}}{I0\ \beta}\ for\ 0 \le n \le N$ $=0 \quad elsewhere$ …………………………….. (1)	Kaiser Window function has an independent parameter α. By choosing the value of α and the filter length N arbitrary specifications can be achieved in Lowpass (LP), highpass (HP), bandpass (BP) and bandstop (BS) filters. By changing value of β and the length of the filter main lobe

3.1 Proposed Approach # 1

I. Introduction to IIR Filters:

Filters are defined as the electrical circuit that is used to remove unwanted frequency from the signal. When we talk about digital signal processing filters, have many applications and different types, so when we talk about the annoying frequency or unwanted frequency we talk about noise in signal or random noise. Also, we can say that extract wanted to part from the noisy signal and removed noise from the signal such component lying certain frequency range. The function of the filter is to separate signal and repair signal. When we talk about serration of signal, it is done when the signal is affected by the interfaces or other noisy factors. For example, let's suppose we have a device for me ruing the electrical activity of a baby's heart while when he is in the womb, the raw signal is likely to be ruined by the breathing and heartbeat of the mother. A further may be used to separate these signal so that they can be separately observed signal restoration is used when the signal is distorted in some way. For example, the deblurring of any picture learned with a wrongly focused lens or shaky camera.

II. Summary of Butterworth filter and Calculation of order and coefficients:

Butterworth filters is also a type of the signal processing. The purpose of the designing this filter is to answer which is flat as mathematically possible in the passband. It is also called flat hugely magnitude filter.

Butterworth filters have changing magnitude function with ω unlike others types of filters have.

Non-monotonic ripples in the pass band or stop band. The Butterworth filter has very slow roll-off, so it demands to implement a particular stop band specification, but also Butterworth filters have a more linear phase response in band pass than other types of filters have. The squared repose of the Butterworth filters is given as the function of the cut-off the frequency.

$$|H_a(j\Omega)|^2 = \frac{1}{(\frac{\Omega}{\Omega c})} 2n \quad (6)$$

The maximum passband edge attenuation is

$$|H_a(j\Omega_p)| = \frac{1}{\sqrt{1} + (\frac{\Omega p}{\Omega c})} 2n \quad (7)$$

$$(\Omega_p/\Omega_c)^{2n} = \varepsilon^2 \quad (8)$$

So the order of the filter can be calculated from the pass edge frequency

$$n = \frac{\log \varepsilon}{\log\Omega\, p - \log\Omega\, c} \quad (9)$$

The minimum stop band edge attenuation wrote;

$$|H_a(j\square_s)|^2 = \frac{1}{(1 + \frac{\Omega}{\Omega c})} 2n \quad (10)$$

The transfer equation H_a $(j\square_s)$ of the Butterworth filter is evaluated from the order of the filter.

$$H_a^*(j\square) = [\int_0^\infty ha(t)e - j\Omega tdt] \quad (11)$$

From the above-given expression the filter order and the coefficients are calculated.

3.2 Proposed Approach # 2

Step 1: Filters are applied using Kaiser Window function with altered values of passband and stopband ripples (passband ripple varies from 0.01 to 0.40 and stopband ripple ranges from 0.09 to 0.49). Ruined signal is then filtered using the applied filters. Filtered signals are kept in a matrix. This matrix is termed as early population. Each row of the matrix is covering a filtered signal named as a chromosome.

Step 2: For the purpose of fitness standards of chromosomes, following equation is followed.

$$\beta = \frac{\text{Corrupted Signal}}{\text{Filtered Signal}} - 1 \quad (12)$$

Lastly, the value of $10\log_{10}(\beta)$ has been used as the suitability value.

Step 3: Created on fitness values a set of filtered signals has been nominated from the original population using Roulette Wheel Selection way.

Step 4: Single point Crossover is done with 100% possibility in between the nominated set of chromosomes and off mechanisms are made.

Step 5: Change is done on the offspring chromosomes with 25% likelihood.

Step 6: Replacement of parent signals by off-spring signals with better suitability values than the parent signals.

Step 7: Signal with highest fitness value has been obtained as best offspring signal.

Step 8: Recap Step 4 to Step 7 N (N=10) times.

4. RELATED WORK

Genetic Algorithm Based Approach

This is another approach for filtering. This algorithm is based on the idea of evolutionary standard presented by Charles Darwin. This algorithm helps to find the challenging type of search algorithm which is called as Heuristic Search Algorithm. This algorithm demands extra information about the problem that we want to find straight in more promising track. They show a smart operation of that casual search to find some problem through random selection. It does not mean that it is based on random selection, but they accomplish before the information to effort the search to the area of better routine within searched space then GA encrypts all the data into a simple string called chromosome which has permanent length. Every chromosome has its fitness value. GA is suitable for resolving optimization problem. A typical GA can be described as follows:

Step 1: GA starts with a set of imaginable solutions or chromosomes called initial population. Fitness rate of every individual is calculated.

Step 2: A set of chromosomes is before nominated using a convinced kind of selection process.

Step 3: These chromosomes are used to produce a new population using inherited operator crossover followed by another genetic operator mutation.

Step 4: Fitness rate of each chromosome is calculated.

5. CONCLUSIONS

Nowadays, noises are common in communication channels, and the recovery of the transmitted signals from the communication pathway without any noise is considered as one of the tough jobs. This is accomplished by the denoising techniques that eliminate noises from a digital signal, and we have many of de-noising algorithm I discuss two approaches FIR and IIR.FIR and IIR both approaches use for denoising signals and have its benefits.

6. AKNOWLEGMENTS

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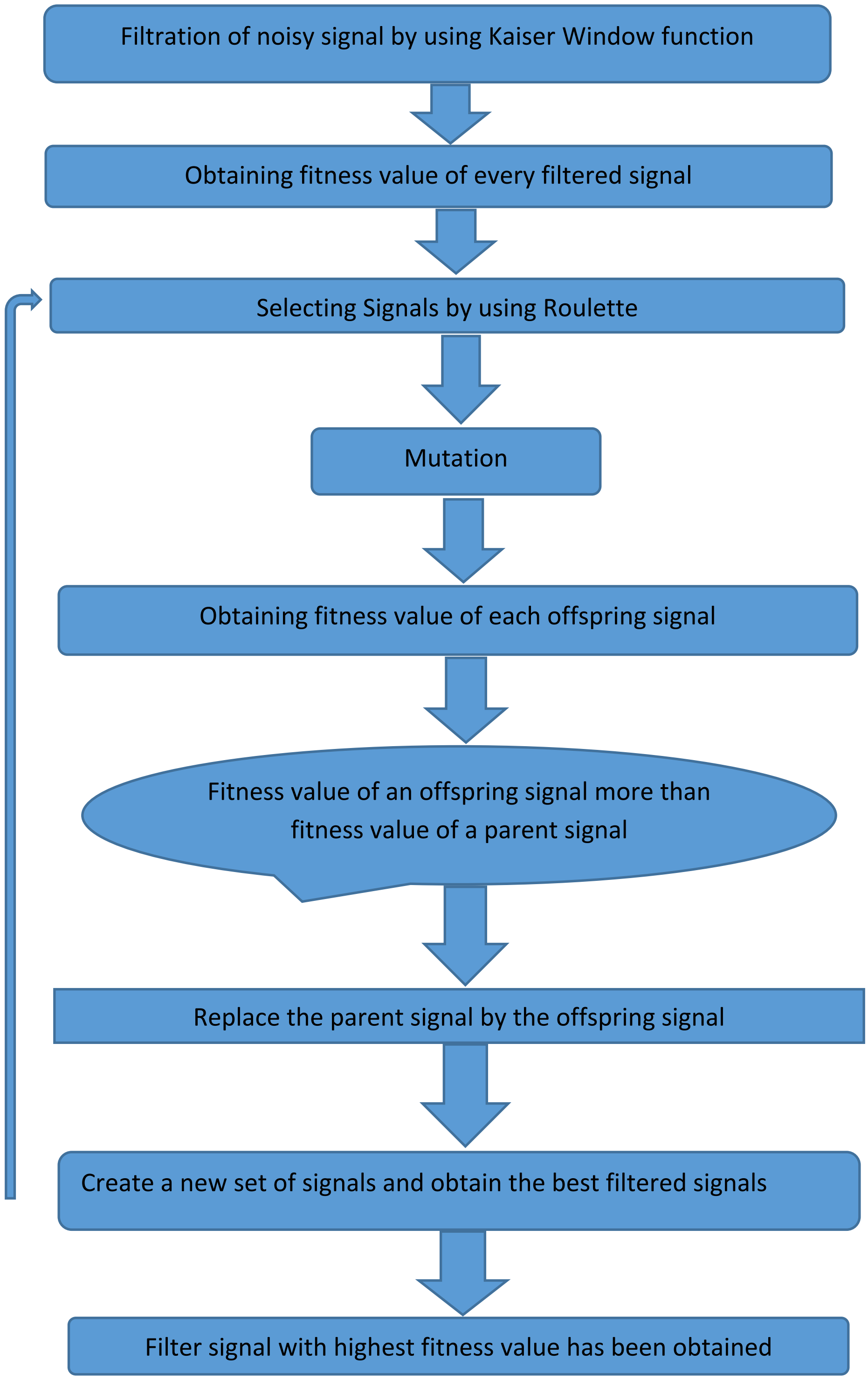

Figure1: Flowchart of Proposed Algorithm

7. REFERENCE

[1] Mahesh S.Chavan, R.A.Aggarwala, M.D.Uplane, ‘Interference reduction in ECG using digital FIR filters based on Rectangular window,' WSEAS Transactions on Signal Processing, Issue 5, Volume 4, May 2008, pp.340-49.

[2] L.R.Rabiner and B.Gold, Theory and Application of Digital Signal Processing. Englewood Cliffs, NJ: Prentice-Hall, 1975.

[3] Alyson K. Fletcher, Vivek K Goyal and Kannan Ramchandran, 2003. "Iterative Projective Wavelet Methods for Denoising," *Proc.Wavelets X, part of the 2003 SPIE Int. Symp. On Optical Science & Technology*, Vol. 5207, pp: 9-15, San Diego, CA August.

[4] C. K. Chui, “An Introduction to Wavelets,” Wavelet Analysis and Its Application, Vol I, Academic Press, Inc., 1992

[5] Major Joseph 0. Chapa, “Constructing Wavelets from Desired Signal Functions” Center for Imaging Science, Rochester Institute of Technology, Rochester, NY 14623, USA.

[6] https://en.wikipedia.org/wiki/Digital_filter.

[7] J. Barros, ―On the Use of the Hanning window for harmonic analysis in the standard framework‖, IEEE Transactions on Power Delivery, 2006.

[8] http://www.cs.utexas.edu/~mooney/cs343/slide-handouts/heuristic-search.4.pdf Sharma, S. 2009. Book on Digital Signal Processing (with Matlab Programs)

[9] Ye, W.B., Yu, Y.J., H. Zhao.et.al. 2013. Sparse FIR Filter Design Based on Genetic Algorithm, In Proceedings of International Symposium on Circuits and Systems.

[10] Claudia Schremmer, Thomas Haenselmann and Florian Bomers, 2001. "A Wavelet Based Audio Denoiser," *In Proc.*

[11] Awad, A.M. 2010. Adaptive Window Method for FIR Filter Design, In Proceedings of IEEE Conference on Open Systems (ICOS).

[12] Table 1 and Figure 1 modified by the author and adapted by Poulami Das, Subhas Chandra Panja, Sudip Kumar Naskar and Sankar Narayan Patra.