Adaptive Cancellation of noise using firefly algorithm

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ABSTRACT

Noise is an interruption occurs in the signals that represents the unnecessary sound in the genuine data which leads to the failure in transferring the data at the destination. The distortion in the Audio signals is due to the noise present at the time of acquisition. Consequently, it is required to remove the noise from the signals completely to retain its originality. Considering this fact, a digital filter FIR has introduced in this paper which removes the noise effectively. Moreover, the proposed technique stated a firefly optimization algorithm to reduce the error rate while eliminating the noise from the audio file. The simulation analysis have performed to examine the existing LMS and NLMS adaptive filters and to study the effect of proposed algorithm. The performance of these algorithms has observed or compared through the error efficiency of individual.

Keywords*—Audio Signal, Noise, FIR, Firefly Optimization Algorithm*

I. INTRODUCTION

Audio signal processing or audio signals are the electronic signals can be analog and digital. These signals are utilized for speech improvement or speech gratitude purposes. Noise may be entered into the signal which may damage the signal evaluation or the actual eminence of the signal, before the processing of these signals [1]. It is an unnecessary sound that joins with the actual signal and violets the uniqueness of the signal. Consequently, audio signals must be filtered before advance processing. Digital signal processing is utilized for the improvement of the speech as well as for the noise diminution. Noise makes the audio signal intolerable and frustrating to listen to. Therefore, there are different methods and filters that have become the thought for the researchers. Noise has turn into the trouble when it is above the threshold hearing [2]. It is ok, if noise does not affect the hearing. But it may affect the quality if the range of noise is higher. Noise can be generated from various sources. The amount of noise in analog signals is higher than digital signals. Various types of noises are currently present in the signals such as broadband noise, narrower noise, impulse and irregular noise [3]. Consequently there are various sources from which the noise can enter into the audio signal. Some of the sources of the noises are errors throughout digital signal processing, original recording environment and it can be from sound cards [4]. Apart from the noise, focuses on the methods that require to be followed before further processing of the signal. Various types of digital filters have been utilized for the noise reduction in the signal [5]. On the basis of these filters, noise can be generally classified into two type i.e. single-channel technique and multi channel technique. From these two techniques multi channel technique is enhanced technique than single channel technique in terms of noise removal and speech enhancement. In some audio signals, only one technique can be applied and in some cases combination of the techniques can be applied according to noise or depends upon the type of signal. Each technique has their advantages and disadvantages [6]. Each technique is used depending on the nature of the signal like Non stationary signal analysis can be done during the Discrete Wavelet Transform technique.

Different Audio Noise reduction Techniques or system is used to enhance the signals quality by removing the noise from the signals. For removing the noise, most of the audio noise reduction systems use filters [7]. According to the corresponding frequency, filters remove the noise from the signals by altering the amplitude of the signals. Filters can be of two types such as fixed and tunable. When passband and stopband frequencies are fixed then filters are taken as fixed filters. Whereas, when passband and stopband frequencies are variable then it is taken as tunable filters. As the conventional techniques of noise reduction is not competent enough in giving the quality output. At the time of capturing of signal noise can get into the signal. Thus, different methods have been used from years for noise reduction. The earlier methods are not satisfactory, so different techniques and methods are projected till now [8]. Conventional methods do not provide efficiency and accuracy. This is the reason for presenting a new method for noise removal or reduction from the signal. The digital filter FIR has proposed in this paper to reduce the effect of noise from the selected signal or audio file. Moreover, the firefly optimization is proposed to remove the noise optimally with less error.

II. BACKGROUND

Noise is an error or unwanted random disturbance of a useful information signal in a communication channel. The noise is typically a summation of unwanted or disturbing or undesirable energy arising from either natural or man-made sources. Noise has inadvertently penetrated into our systems. Noise cancellation has become an important area of research and has been proactively applied in many areas. Several techniques have been proposed till now to remove the noise from the signal but still suffering from various issues. In the traditional method, LMS and NLMS adaptive algorithms are used for removal of noise in the signal but it does not provide results efficiently. Moreover, there can be chance of having noise after evaluation due to which an optimized FIR filter has proposed in case of any noise left in the signal.

III. PROPOSED WORK

The techniques such as LMS, NLMS were used for removing the noise from the signal but did not approach the effective results. Hence to fulfill the need a new technique has proposed which uses the firefly algorithm and Digital FIR filter (Bandstop Filter) for removing the noise from the digital audio signals. This study also performs the analysis by implementing the LMS and NLMS methods for noise removing for comparison purpose. For the purpose of evaluation of the proposed work the analysis is done on various audio samples. Then to prove the proficiency of the proposal the comparison is done by evaluating various performance parameters.

IV. METHODOLOGY

The proposed work has been performed analysis of existing filters such as LMS and NLMS with respect to the proposed digital FIR filter. Filters are usually used to eliminate the noise from the signal. An additional term used for them are denoising or speech denoising. The chief focus of this theory is the digital filters. Audio can be tainted at the time of acquisition. So, the task of the filters is to satisfy the noise so that the original signal maintains its quality.

The methodology for the proposed work has shown in the below block diagram.

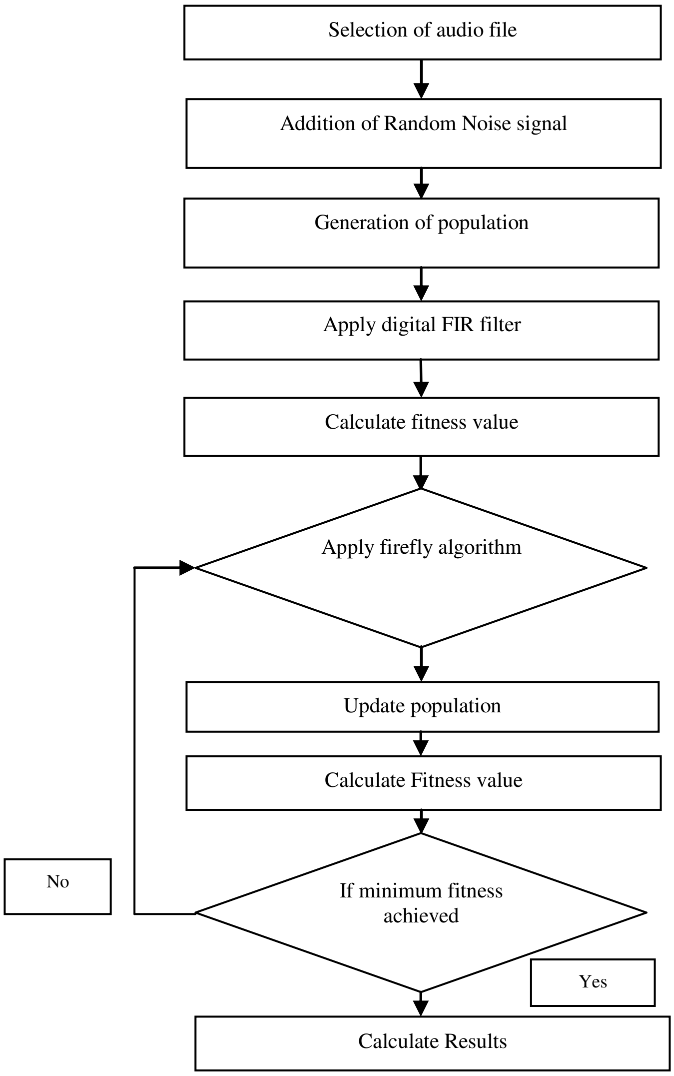

Fig 1. Framework of the proposed work

V. RESULTS AND DISCUSSIONS

This section provides the overview to the results that has been obtained after implementing the proposed work. the proposed work implements the firefly algorithm for removing the noise from the signals. For the comparison purpose it also analyzes the traditional LMS and NLMS techniques for noise removal. the following graph shows the comparison of LMS, NLMS and firefly on the basis of error rate. the very step in obtaining the desired output is to take the original signal as an input. thus, below figure 2 shows the original signal used for the evaluation. This signal is free from noise and any kind of disturbance.

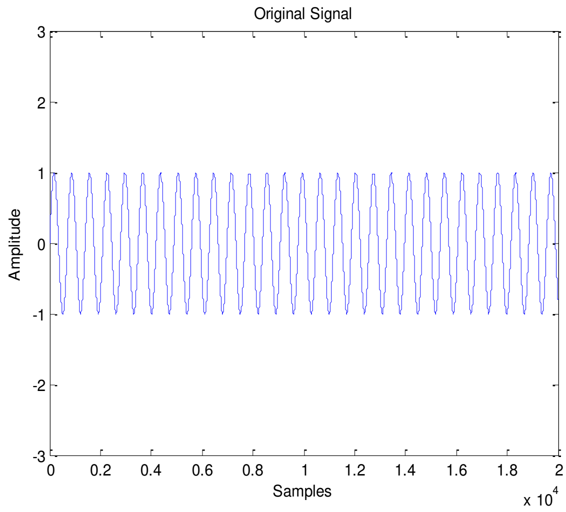

Fig 2. original signal

The figure 3 depicts the noisy signal which will be collaborated with the original signal and then filtration process will be applied onto this signal to check the performance of the individual technique.

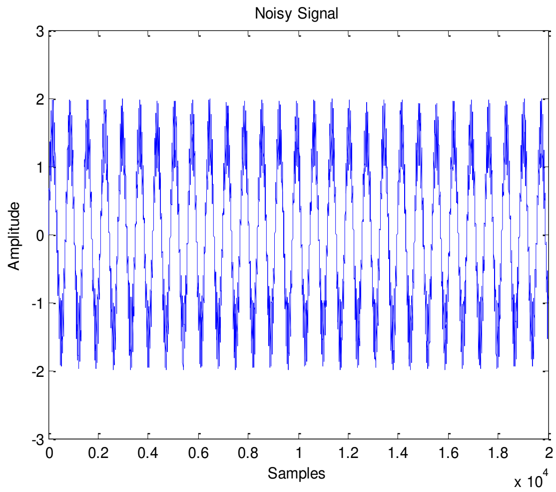

Fig 3 Noisy signal

The signal which contained a noise has been filtered through the FIR filter shown in the below figure 4.

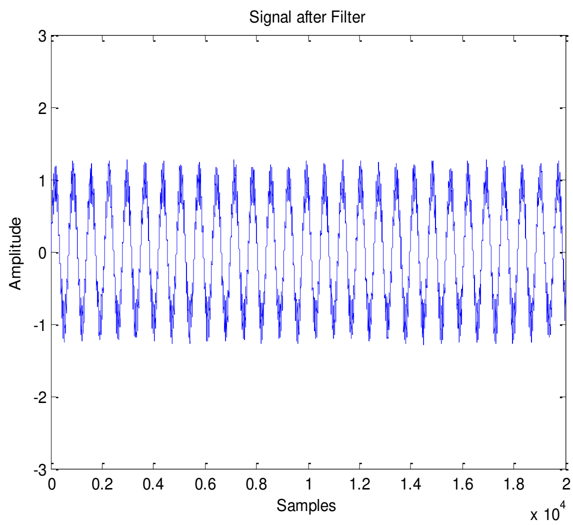

Fig 4 signal acquired after applying filter

The figure 5 exemplifies the best fitness value achieved through the proposed work. It has been measured through the mean square error which shows that if minimum fitness vale is achieved, the signal will be optimized with less noise.

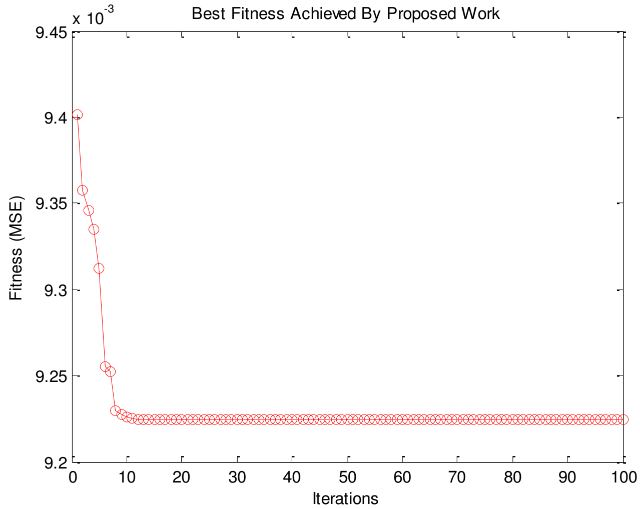

Fig 5 Best Fitness achieved by the proposed work

The error efficiency of the proposed work has shown in the figure 6 which is quite less in comparison with other existing methods shown in the figure 7. The Error efficiency has measured through the amplitude with respect to number of samples varying by 0.2 value.

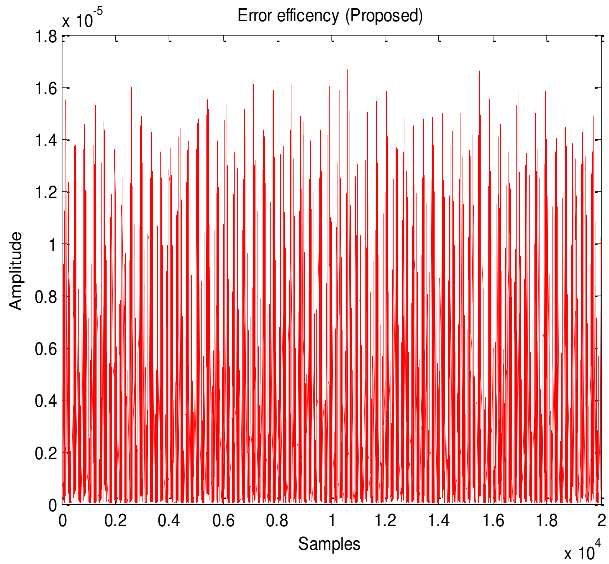

Fig 6 error efficiency of the proposed work

From the graph it is observed that the error rate of firefly noise removal technique is lower as compare to the LMS and NLMS methods. The error rate of LMS method exists between 0 and 1, for NLMS it ranges from 2 to 3. But in case of firefly, the error rate is noticed between 0 and 0.1. Hence it is proved that the firefly is a quite effective method to remove the noise from the signals.

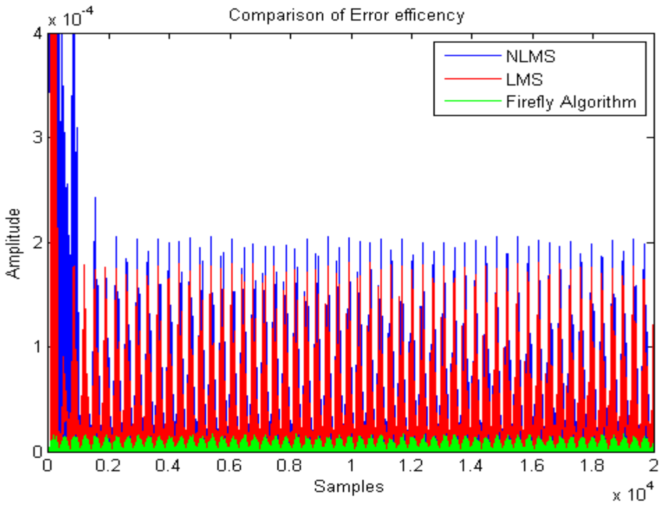

Fig 7 Comparison graph on the basis of Error efficiency

The graph below shows the comparison of signals after applying the LMS, NLMS and proposed work for noise removal. The data is calibrated in the form of amplitude of the signals on y axis and signal samples on x axis. The signals that are received after applying the NLMS technique are more variant as compare to the LMS and firefly method and the noiseless signal which is received after applying LMS method is much better than NLMS but not same as for firefly. Hence the signal is denoised by firefly algorithm is much better and noise less as compare to other methods.

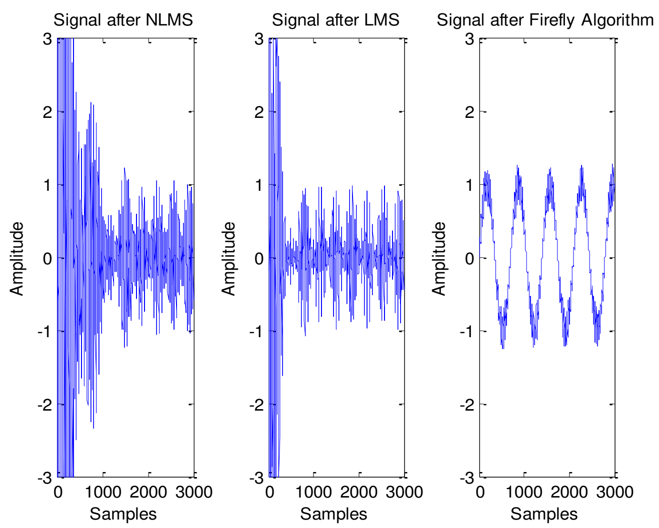

Fig 8 Signal after applying NLMS noise removal technique.

VI. CONCLUSION AND FUTURE SCOPE

As Noise can corrupt the actual behavior of the signal due to which it is required to remove the noise from the signal. Several filters have been used till date from which LMS and NLMS adaptive filters are concerned in this report. The main idea is to describe the application of LMS and NLMS adaptive filter. These techniques have been used to remove noise from the signal and their results have been investigate which concludes that LMS algorithm outperforms and produce good results in noise deduction problem.

For the future reference several filters can be enhanced which may reduce the complexity in cancelling of noise from the signal. Moreover filters based upon adaptive filtering can be used for more practical usage which may result into efficient output signal.

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