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A Review on Application of Adaptive Algorithms in Signal Processing

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ABSTRACT: Various fields like audio, video, medical and cellular industries are now a days dependent on digital technology i.e. the Digital Signal Processing. Digital signal processing basically deals with the digitized representation of signals or data and use of digital subsystems to study, manipulate, save or collect meaningful data from these signals. For filtering applications, Digital Signal Processing (DSP) techniques include digital systems. It is a powerful device for signal processing and control applications in time variation environment of input statistics. To reduce the signal distortion inspired by predictable and unpredictable noise by use of adaptive filters. Its applications such as identification, inverse modeling, prediction and interference cancellation are essential to explicate the problem of acoustic echo and noise cancellation. If accurate information of the signals that is to be processed is available, the designer can easily choose the appropriate algorithm to process the signal. In this review paper, we have investigated researches done in the field of signal processing. We have focussed on the applications of adaptive algorithms in denoising ECG signals, audio signals, acoustic echo cancellation, EEG/ERP signals, etc. In some of the research work, Particle Swarm Optimization (PSO), Modified PSO, Artificial Bee Colony (ABC) algorithms have also been used and therefore, discussed in this paper.

KEYWORDS: Adaptive filter, LMS, NLMS, RLS, Acoustic echo, PSO, MPSO, ECG, EEG/ERP, SNR, MSE, Convergence rate.

I. INTRODUCTION

Digital Data Processing techniques are ever changing due to the heavy requirement of the data transmission. The devices are becoming smart and therefore are capable of capturing the two signals and are capable of transmitting the same data over the different channels of communication. High definition video and audio signals are maligned with some spurious signals called as noise. The desirable characteristics of the signal received at the receiving end have to be processed to get the high fidelity signal. There is a need for more sophisticated and efficient digital processing techniques. In this review paper, we have comprehended the recent work and then summarized.

II. RELATED WORK

Vivek Joshi and et.al have proposed adaptive noise canceller for removing noise in ECG signal using modified particle swarm optimization technique (MPSO). MPSO technique provides better performance than any other optimization technique. Various parameters like SNR, Peak reconstruction error and Mean square error have been evaluated.ECG stands for electrocardiogram i.e. it is an electrical signal generated in the heart. ECG signals have various types of noises like Power line interference, electrode contact noise etc. These noises can be reduced by using adaptive filter algorithms by adjusting filter coefficients according to problem specification. There are several complexes as well as nonlinear problems which can be solved by using special optimization techniques i.e. conventional PSO algorithms which start by initializing random swarm of M particles each particle is having R unknown particles to be optimized, here velocity and position of a particle are updated. The conventional PSO faces the problem for large signal length, so the PSO is modified by giving inertia weight parameters. In MPSO the position of each particle in the swarm is updated. It is seen that conventional adaptive filter algorithms provide one solution after each iteration whereas PSO gives no. of the possible solution in single iteration cycle.Adaptive Noise Canceller (ANC) is basically implemented

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using RLS & LMS algorithms but PSO provides faster convergence rate, here, at each iteration MSE (mean square error) is calculated.Performance parameters of ANC using MPSO technique has been evaluated which shows that MPSO performance is better than PSO. The simulation results also show 19 % improvement in SNR, 91% decrease in Peak Reconstruction Error (PRE) and 99% reduction in Mean Square Error (MSE) using MPSO technique [1].

Atar Mon and et.al have proposed three conventional adaptive algorithms LMS, NLMS, and RLS for ANC are analyzed based on single channel broadband feedforward. For faster convergence NLMS is modified and simulation results show the high quality of Noise Cancelation and minimized mean square error.Noise is an unwanted signal that interferes with another signal in communication. Noise and degradation are the major factors that limit the capacity of audio signal processing, therefore, it is required to model and remove the effect of noise and distortions. For this adaptive filtering is required, they have the property to modify the values of their parameters i.e. their transfer function, during the processing of the input signal, in order to generate a signal at the output without undesired components, degradation, noise and interference signals. All the work is carried out in MATLAB. Here NLMS is modified and implemented using sine function in ANC. Background noise is Gaussian noise, 300hz of which is added to the audio signal. Filter length 10 is used to obtain better performance.MSE is compared for all four (LMS, NLMS, modified NLMS and RLS) algorithms which indicate that convergence rate of modified NLMS is faster than NLMS [2].

Nitika Gulbadhar and et.al [3] have presented various adaptive algorithms for acoustic echo cancellation along with their merits and demerits. In acoustic applications, the noise reflected from the surrounding environment reduces the audio and speech signal quality and it becomes impossible to recover the transmitted signal. To remove noise and enhance the quality of signal the technique which is used is called Acoustic Noise Cancellation. **Least Mean Square Algorithm (LMS)-**LMS algorithm is used to calculate Mean square error and is based on steepest descent method and gradient search technique. This method can be easily implemented and has low computational complexity. The step size controls the convergence rate, smaller value of step size leads to more convergence time whereas the larger value of stepsize diverges the algorithm and degrades the filter performance. LMS algorithm has fixed step size for every iteration which is its main disadvantage. **Normalized Least Mean Square Algorithm (NLMS)-**NLMS was developed to overcome the problem of LMS. It is more stable to unknown signal as step size depends on the input vector. It has faster convergence rate compared to LMS due to variable stepsize. **FX-LMS Algorithm-It** is seen that LMS algorithm did not perform well in ANC framework. the presence of actuators, A/D converters, D/A converters and anti-aliasing filters causes significant changes in signal. So, there is a need of this secondary path function as a change in signal demands. **Average Adaptive Filter-**When higher convergence rate is required average adaptive filter is used. AAF belongs to stochastic gradient algorithm. **Leaky Least Mean Square Algorithm-**It is a standard least mean square algorithm which differs only because of the cost function. In this method, the coefficient overflow problem is avoided. **Recursive Least Square Algorithm-**It minimizes the cost function, here, each iteration requires $4N^2$ multiplication operations and $2N^2$ additions which provide greater computational complexity. Performance improvement in terms of SNR for various filters has been shown. Also, the comparison in terms of computational complexity, convergence rate, stability, and robustness have been evaluated 2.9 seconds and higher stability.

Suman and et.al have proposed Delta rule algorithm which is used for learning complex patterns in the Artificial neural network. They have also implemented adaptive filters using LMS and RLS algorithms and compared their results. Active noises are random real-time noises, they can be canceled by using sound absorbing material, they occur in the frequency range between 50-250hz, here the wavelength is too high. So this technique cannot cancel these types of noises as the material required is too heavy.So to cancel these noises Active noise control (ANC) system is used which is based on the principle of superposition, on the basis of which it cancels the unwanted noise. An anti-noise signal of equal amplitude and opposite phase is generated and combined with unwanted noise for the cancellation of noise. All the simulation work is done in MATLAB. It is seen from the results that RLS is better than both LMS and Delta rule algorithms but the computational complexity of RLS is more than the other two algorithms. As delta rule algorithm requires more computations than LMS and its residual noise is less than LMS it is found to be more efficient in this work [4].

Smita Dubey and et.al [5] have used NLMS adaptive filter algorithm to denoise the ECG signal. ECG signal is collected from the MIT-BIH database. The results show that by applying NLMS algorithm, signal to noise ratio (SNR) and least mean square error has been improved. **ECG i.e.** Electrocardiogram is an important biomedical tool for diagnosis of heart disorder. It picks up the electrical impulses generated by the polarization and depolarization of atrium and ventricles. These waveforms are used to measure the rate and regularity of heartbeats as well as the size and

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position of the chambers, the presence of any damage to the heart, and the effects of drugs or devices used to regulate the heart. LMS algorithm is used to calculate the mean square error and is based on stochastic gradient descent method. It minimizes the cost function of the coefficients, its main drawback is that it is sensitive to the scaling of its input which makes it very difficult to choose step size that defines the stability of algorithm. NLMS overcomes the disadvantage of LMS by normalizing the power of input also, the step size can be chosen independently of input signal power and no. of tap weights which provide NLMS higher convergence rate and steady error rate compared to LMS. All the work is carried out in MATLAB. The results show that on updating the filter weights and coefficients depending upon the multiple input vectors more efficient result is obtained which provides a decrease in mean square error and hence reduced the signal distortion.

S.M Rafi and et.al [6] have proposed a new particle swarm optimization technique for the design of a two-channel linear phase quadrature mirror filter bank(QMF) in the frequency domain. Here the PSO is modified by the concept of Scout Bee from Artifical Bee Colony for designing the quadrature mirror filter bank. The results clearly show improvement in proposed PSO technique over the earlier one. In past decades, it is seen that multi-rate filter banks have received attenuation due to their various applications in engineering fields. Among various multi-rate filter bank, a two-channel filter bank also called QMF bank was used to reduce or eliminate the effect of aliasing error in subband coding of speech signals. Particle swarm optimization technique is used for solving optimization problems, here the swarm is composed of some volume fewer particles with velocities, each of which represents a feasible solution in the solution space. In the proposed method, the concept of Artificial Bee Colony is used in original PSO. In ABC, honey bee colony consists of employed, onlooker and scout bees. In ABC, employed and onlooker bees handles the process of exploitation and scout bees maintain exploration. Similarly, in the proposed method, the original PSO method is modified by introducing Scout particles in PSO.The proposed PSO method has been implemented on Genuine Intel (R) CPU T2300 @ 1.66 GHz, 1 GB RAM.The simulation results show that the proposed PSO technique results in low peak reconstruction error and low amplitude distortion. Results also show that proposed PSO method is a good alternative for designing QMF bank.

Yang Liu and et.al have proposed LMS adaptive filtering algorithm for the reduction of noise from audio signal or speech signal. Here the signal is filtered in the time domain and the filter coefficients are calculated adaptively by LMS algorithm further NLMS algorithm is also used. In today's scenario noise is an integral so the speech signals cannot be recorded in pure form and are contaminated by some background noises. So these audio or speech signals should be filtered with the DSP tools before they are transmitted or stored. Recently noise reduction is in great demand as no. of audio application are increasing. Noise reduction can be done in many different ways such as beam forming, adaptive filtering, temporal filtering, spatial-temporal filtering, etc. In this proposed work the noise is reduced from the audio signal by using LMS adaptive filter algorithm. The desired audio signal is restored by passing the noisy speech signal through a FIR filter whose coefficients are estimated by minimizing the MSE.Simulation results show that by using NLMS algorithm performance quality of the noisy audio signal is improved. Therefore, the proposed method is quite effective in reducing the noise from audio signal especially when white Gaussian noise is in use [7].

Raghavendra Sharma and et.al have performed the comparative study of various existing adaptive filter algorithm i.e. LMS, NLMS, and RLS for denoising the sound signals of musical instruments, here, the noise added to the sound signal is additive white Gaussian noise. Results show that RLS algorithm is better than LMS and NLMS in terms of parameters such as peak signal to noise ratio, convergence rate, least time for denoising the sound signal. In today's world, the sound of musical instruments in the form of digital signal is becoming popular, but it is seen that signal obtained after transmission is always corrupted with noise and it needs to be processed before it is used for any application. There are several ways to recover original signal from such noisy distortions. Each algorithm performance is compared by computing the peak signal to noise ratio (PSNR) besides hearing perception. In this paper, various adaptive noise cancellation algorithms have been applied to sound samples produced by musical instruments(shehnai, dafli, and flute) sampled at 44.1K samples per second. The performance of all three algorithms is compared on the basis of mean square error along with the various percentage of white Gaussian noise introduced in the signal. Among the three, NLMS and RLS are not sensitive to stepsize and are more accurate and have faster convergence rate. It is seen that results obtained from RLS are better with respect to the speed of convergence, accuracy, MMSE and PSNR values [8].

Mitul Kumar Ahirwal and et.al have proposed an improved method for filtering EEG/ERP signals. Adaptive Noise Canceller (ANC) has been implemented through five versions of PSO i.e. particle swarm optimization technique. Also,

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the comparative study of the performance of PSO and its various version has been done. Parameters like SNR (in dB), the correlation between resultant and template ERP and mean square error are also observed.EEG i.e. Electroencephalography is the recording of intrinsic electrical activity in the brain while ERP's (event-related potential) are the weak signals and buried in signals of spontaneous EEG with very low SNR. EEG responses are of two types namely, time locked or phase-locked responses also called ERP as they appear for the short time interval and secondly, time locked but not phase locked called event-related desynchronization (ERDS). In this paper ANC based PSO technique and its five versions is proposed for ERP filtering from EEG signals. PSO technique is used for solving optimization problems and it is a population based stochastic optimization technique. PSO consists of a swarm of particles and each particle represents a possible solution. Five versions of PSO that have been discussed are : **Constantly weighted inertia PSO (CWI-PSO):** By CWI-PSO constant rate for exploration and exploitation is achieved. It uses a fixed constant value which remains unchanged in all iterations also it is problem specific so it is difficult to find a constant value for a particular problem.**Linear decay inertia PSO (LDI-PSO):** LDI weighted approach is a good example for a trade-off between exploitation and exploration and there is a reduction in exploration area after each iteration. This technique requires maximum and minimum inertia values to define before starting the process.**Constrictions factor inertia PSO (CFI-PSO):** Implementation in a controlled manner for convergence of the particles, method for preventing explosion and ensuring convergence needs some constriction coefficients, which are all provided by CFI-PSO technique.**Nonlinear inertia PSO (NLI-PSO):** It is an alternative of LDI-PSO. **Dynamic inertia PSO (DI-PSO):** It is the most frequently used PSO technique among all five. If the problem for which optimum solution to be found is of dynamic nature, then DI-PSO may perform well due to its similar dynamic nature. Results show that performance is dependent on parameter selection and PSO variants. It is seen that the NLI and LDI variants of PSO have more score in fidelity parameter evolution and in shape measure as compared to another variant. Therefore, it is concluded that the ANC based on NLI and LDI variants of PSO technique can be very effectively used in ERP filtering from the EEG signal [9].

Dervis Karaboga and et.al have presented a comprehensive survey of advances in ABC and its applications. During a decade various algorithms have been developed which depends on different intelligent behaviors of honey bee swarms, among which ABC is most widely used to solve real world problems. There are various kinds of the swarm in the world but it is not possible to call them intelligent as their intelligent level could vary from the swarm to swarm. The key feature of swarm system is self-organization which results in collective behavior by means of local interactions among simple agents. It is seen that according to Millonas, the swarm must satisfy following principles:

1. It should be able to do simple space and time computations.
2. It should be able to respond to quality factors in the environment.
3. It should not commit its activities along excessively narrow channels
4.Should not change its mode of behavior upon every fluctuation of the environment.
5. Must be able to change behavior mode when needed.

From this study, it is observed that the growth of this field has exceeded the expectations [10].

Ali T. Al-Awami and et.al [11] in this paper have presented a new modified PSO technique and illustrated its superiority over PSO technique with its applications in Adaptive channel equalization. Adaptive equalization is important as it is used to mitigate the effect of Intersymbol Interference (ISI) in a digital communication system where adaptive algorithms are used to adjust the equalizers coefficients. Many adaptive algorithms have been developed such as LMS, as the output is a linear function of the input, use of the linear adaptive algorithm is occasionally successful. Alternatively, heuristic techniques have also been employed for Adaptive Equalization and in particular, the use of particle swarm optimization (PSO) in adaptive IIR phase equalization and in a recent work on interference cancellation in CDMA systems. Paper has presented the application of PSO to channel adaptive equalization. Results show that PSO not only improves the convergence time of the equalizer but also improves its BER performance. Results obtained here provides us with ample encouragement to further explore the use of PSO technique in other adaptive filtering application such as echo cancellation in telephony and ANC in industrial settings.

Sanjay K .Nagendra and et.al have focused on reducing unwanted echo by using adaptive filtering techniques, thus improving the signal quality. Various techniques and adaptive filtering algorithms have been examined and all the work is carried out in MATLAB. Acoustic echo cancellation occurs in telecommunication system when the audio source and sink operate in full duplex mode .the signal interference caused by acoustic echo distracts both users and reduce the signal quality. These echo signals can be canceled using adaptive filtering. Adaptive filters are dynamic filters which

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iteratively change their characteristics to achieve an optimal desired output.LMS algorithm is implemented using MATLAB. During the simulation, the echo signal was generated by defining an appropriate impulse response then convolving this with an audio input wave file. In this work, LMS algorithm is used and corresponding to it desired signal, adaptive filter output signal, estimation error and mean square error are plotted. The future work can be done using the same audio file as input on different algorithms such as NLMS (Normalized mean square algorithm), RLS (recursive least square algorithm), VSSLMS (variable step size LMS) and VSSNLMS (variable step size NLMS) [12].

III. CONCLUSION

The research in Digital Data Processing field has been revolutionized by heavy need pressed by the end users and industry. The international research trend is evident from the review of the some of the work done globally. There is a common trend of using Adaptive Algorithms like LMS, NLMS, RLS, PSO, and MPSO. It can be concluded that MPSO gives better performance for ECG signals. Also, RLS algorithm is more efficient than LMS as it provides faster convergence rate but it requires more complex hardware.

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