A Literature Survey based on Adaptive Algorithms

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Abstract-Thissurveypaper is reviewed in different concerns. It has been conducted to know about designing of adaptive filter and alsoto know where the adaptive algorithms are used in the various applications. The main goal of this survey paper is to study and performance of different adaptive filter algorithms for noise cancellation and echo cancellation.

Keywords-Adaptive algorithms,Adaptive Filter, Adaptive Noise cancellation System, Convergence rate,Noise

I. Introduction

A literature review clearly establishes the need of the work. It helps to address queries about development in these studies and allowit for many unresolved problems to arise and thus obviously express all boundaries about the progress of the research work. Digital signal processing systems are becoming more and more interesting due to the development in digital circuit design. There are some techniques which include digital systems for various filtering application. Digital systems are used to manage the information of the various input signal.Adaptive filters are made applicable in any kind of new environment. Adaptive filter is used for digital signal processing andalsomaintainvarious applications in time changing environment of input statistics. Corrupted signal is enhancedbycertain and uncertain noise which isreducedwith the help of adaptive filters. The applications of adaptive filters such as identification, inverse modelling, interference cancellation and prediction are the major essentials to solve the problem of noise and acoustic echo cancellation. Various algorithms are designed mainly LMS, NLMS and RLS algorithm for interference cancellation to get suitable adaptive filter. The performance measures of adaptive algorithm are misadjustment, rate of convergence, computational requirements, stability and numerical robustness,. Section 2 gives an overview of adaptive filter. The brief description of noise cancellation system is explained in section 3. The basic concept of an adaptive noise cancellation system is to allow the corrupted signal from a digital filter that influence to suppress the noise corrupted signal while leaving the original signal unchanged. Adaptive Noise Cancellation (ANC) completely constrict the low frequency noise for which passive methods are ineffective. Section 4 gives description about adaptive algorithms such as Least Mean Square (LMS), Data Sign LMS, Leaky LMS, and Recursive Least Square (RLS) .Section 5 concludes the main research work.

II. Adaptive Filter

It is an adaptive filter which can adjust itself its transfer function corresponding to the best adaptive algorithm conveyed by an error or corrupted signal. Most of the adaptive filters are digital filters because of the complicated of these adaptive algorithms. Adaptive filters are used for various applications because some parameters of the desired processing action are not known in progress [1] [2]. The adaptive filter uses feedback in the form of an error signal to filter its transfer function to associate the altering parameters. The adaptive process requires the use of a cost function which is a criterion for initial performance of the filter, to deliver an algorithm, which determines how to modify the filter transfer function to minimize the cost on the next iteration. Fig.1 shows the block diagram of adaptive filter [3].

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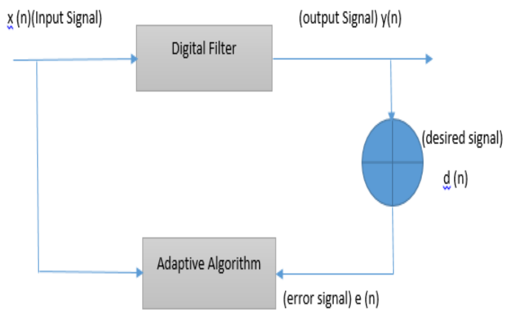

Fig 1 Block diagram of adaptive filter

III. Noise Cancellation System

Noise is disturbance unwanted signal during signal communication. Noise can occur because of many elements like delay, overlapping and interference. These problems in the environment are obtained because of the excessive improvement of technology that has accelerated to noisy engines, and other noise sources. Noise cancellation system operating for various applications such as to cancel the periodic interference in speech signals, to cancel the periodic interference in electrocardiography. Adaptive filtering has been largely used in many practical applications. Important results have been obtained in noise and interference cancelling for biomedical applications [4]. In the process of signal processing for noise or time varying signals, Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) fixed coefficient filters cannot achieve best filtering. So, we must design adaptive filters, to provide the changes of signal and noise signal. Adaptive filter technology gives better performance as compared to conventional methods. The general configuration of Noise Cancellation System [5] is shown in Fig.2. It consists of two input signals, the signal d(n) which is corrupted by an undesired noise x1(n) , and desired signal s(n) and the other reference signal x(n), that is to be filtered out of the system. The main goal of Noise Cancellation system is to reduce the noise signal, and to get the de-noised signal. A reference signal x (n) is required for filteration. Though, the reference signal is commonly not the same signal as the noise portion of the primary amplitude, phase or time. So, the reference signal cannot be subtract directly from the input signal to get the desired result at the output .Noisewhicheffectson the speech signal can be considered as White noise or Colored noise.

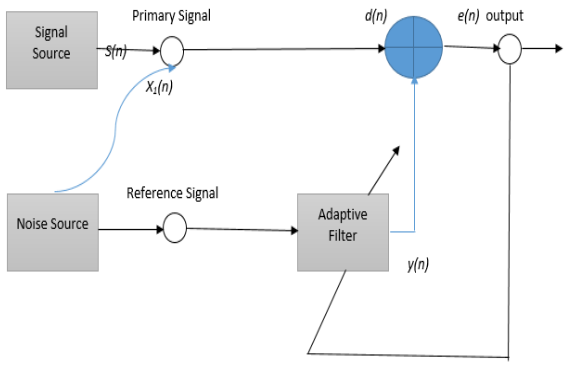

Fig2 Adaptive noise cancellation system

Where s(n) –primary signal, d(n)- corrupted signal, x(n)-noise reference input, x1(n) -noise signal, y(n)-output of adaptive filter, e(n) -system output signal.

Adaptive Noise Cancellation system utilize two signals, One signal that is used to measure the corrupted signal while the other signal is used to measure the noise signal alone. This technique adaptively adjusts its filter coefficients so as to remove the noise from the corrupted signal. This technique requires high coherence between the noise component in the corrupted signal and the noise in the reference signal. Unfortunately this is a limiting factor, as the microphones need to be space apart in order to prevent the speech being constituted in the noise reference signal and thus it being removed. In summary, we use two input signals and an adaptive filter to realize an adaptive noise cancellation system. One input signal is the corrupted by noise which can be expressed as:

$$d(n)=s(n)+x_1(n) \tag{1}$$

The other input signal which is noise reference input signal pass through an adaptive filter and output y(n) is produced as close a replica as possible of x1(n).The filter readjusts itself its filter coefficients continuously to minimize the error between and during this process. Then the output is subtracted from the corrupted signal to produce the system output. It is expressed as:

$$e(n)=s(n)+x_1(n)-y(n) \tag{2}$$

This is the de-noised signal.

IV. Adaptive Algorithms

The adaptive algorithm provides various application of interference elimination, and then with the help of these algorithms we can alter the signal characteristics could be faster. The adaptive filters like Least Mean Square, sign Least Mean Square and Normalized Least Mean Square aremajorlyused in the different signal processing application, because it is very easy to implement and widely used for simple computation. The RLS algorithm is the "ultimate" algorithmto illustrate the best convergence behavior [6]. A new modified adaptive algorithm is given byChansarkar,M.M.Desai and U.B. in 1997.An estimated recursive implementation algorithm is known as the Robust Recursive Least Square algorithm (RRLS). The RLS algorithm is dreadful with respect to persistent bounded data interruption. Simulation results are describedto demonstrate the efficacy of the RRLS algorithm [7].The new Variable-Step-Size (VSS-LMS) algorithm were evaluated in 2001.The results intimate that better performance can be achieved than the previous algorithm .The fixed-step-size (FSS) algorithm can be applied to sub band adaptive echo elimination. Simulation result shows that the proposed algorithm yields a lower steady-state misadjustment as well as a lower residual MSEwhen compared with FSS sub band LMS algorithm. The suggested algorithm results in a slower convergence rate but also a lower steady-state failure alteration in comparison with the NLMS sub band algorithm. It isobserved that the improved system performance is achieved with a different step size adaption for each and with every individual sub band [8].The adaptive algorithm is used for channel estimation, interference cancellation, and channelequalization in digital signal processing system.The Least Mean Square algorithm is the most important algorithm form various adaptive algorithms. Its convergence speed is decided by the step square algorithm is used for obtaining both the residual fault level & also provide highest speed of merging. Several Variable Step-Least Mean Square algorithms have reviewed and a modified of this algorithm developed in 2007 [9]. Entirely outcomes designate that convergence stability and tracking ability are superior to other adaptive algorithms [10]. P.Radhika, Chunduri. V.M.NarenSimha&MonpurAshwin are used various techniques in 2014 for elimination of unsolicited entities from various signals. The power line interference from all sensitive monitoring equipment's can be eliminated by implementing various techniques with different error nonlinearity-based on adaptive filters. The proposed algorithm is best for applications such as biotelemetry etc. These systems are usingsimple addition and give speed up over the other LMS-based realizations [11]. The LMS adaptive filter family is very attractive for implementation of low-cost real-time systems due to its low computational intricacy and robustness [6] [12].

A. LMS Algorithm

This is mostly used for different applications such as channel equalization, echo cancellation and noise cancellation. The equation below is LMS algorithm for updating the tap weights of the adaptive filter for each iteration.

$$w(n+1) = w(n) + \mu e(n) x(n) \tag{3}$$

Where x(n) is the input vector of time delayed input values and w(n) is the weight vector at the time n. μ is the step size parameter. This algorithm is used due to its computational simplicity. It requires 2N+1 multiplications and additions but it has a fixed step size for each iteration.

B. Data Sign LMS algorithm

In a high speed communication the time is critical, thus faster adaptation processes is needed

$$\operatorname{sgn}(a) = \begin{cases} 1 & a > 0 \\ 0 & a = 0 \\ -1 & a < 0 \end{cases} \quad (4)$$

For data Sign algorithm [7] weight update coefficients equation is:

$$w(n+1) = w(n) + 2\mu e(n) sign(x(n)) \quad (5)$$

By introducing the signal function and setting a value of power of two, the hardware implementation is highly simplified. It improves the convergence behavior, requires less computational complexity and also provides good result but throughput is slower than LMS Algorithm.

C. Leaky LMS Algorithm

It introduces a leakage coefficient into LMS algorithm so it becomes as:

$$w(n+1) = (1 - 2\mu\gamma) w(n) + 2\mu e(n) x(n) \quad (6)$$

Where $0 < \gamma << 1$ .The effect of introducing the leakage coefficient γ is to force any undamped modes to become zero and also forceto the filter coefficients to become zero if either e(n) or x(n) is zero.

D.RLS (Recursive Least Square) Algorithm

This algorithm[13] attempts to minimize the cost function in Equation (7). k= 1is the time at which the RLS algorithm commences and is a small positive constant very close to, but smaller than 1. With values of 1 more recent input samples, this results in a scheme that places more emphasis on recent samples of observed data and tends to forget the past

$$\zeta(n) = \sum_{k=1}^{n} \lambda^{n-k} e_n^k(k) \quad (7)$$

V. Conclusion

In this paper a review has been carried out about the Adaptive filtering algorithm with respect to the noise cancellation problem. Now-a days we have many adaptive algorithms available each having its various types of properties. The survey of designed adaptive filterwas conducted where we get LMS adaptive algorithm is the best for its stabilityand also for its high speed capability, convergence rate. To achieve minimum mean square error at a high convergencerateis the main task of this algorithm. When compared to LMS algorithm, RLS algorithm offers a faster convergence and lower error at steady state. But, this RLS algorithm is more computationally complex and if proper design procedures are not followed, RLS algorithm may diverge away resulting in instability.

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