A Review of Novel Adaptive Filtering Approach for Speech Enhancement

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Abstract:-Current Method of speech enhancement has been developed with adaptive filtering approach. The removal of unwanted signal i.e. noise from speech signals have applications ranging from cellular communications to front ends for speech recognition system.
Adaptive Filtering techniques are used in many applications such as cancellation of echo, adaptive equalization, adaptive noise removal , and beam forming applications .This survey paper is an optimal survey of adaptive filtering using Least mean squares algorithm(LMS), Normalized LMS algorithm, and Recursive least squares(RLS) algorithm has been evaluated for noisy speech.

Keywords— *Adaptive filtering, LMS, NLMS, RLS, Speech Enhancement.*

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I. INTRODUCTION

In real time situations speech signals are corrupted by several different forms of noise such as speaker sound, background noise like door slam fan running in background, car noise, TV noise and also they are concern to distortion caused by communication channels; examples are low-quality microphone, room reverberation, etc. In all such situations extraction of high resolution signals is an important task.Filtering techniques are mainly classified as adaptive and non adaptive filtering techniques. In real time environment the analytical nature of all speech signals are non-stationary; as a result non-adaptive filtering may not be suitable for such practical situations. Speech enhancement improves quality of signal by suppression of noise and reduction of distortion.

The speech enhancement tells about the growth of communication system. Enhancement means improvement in the value or quality of something. When we applied it to speech this simply means the improvement in the quality of degraded speech signal by using different signal processing tools. Speech enhancement describe that, it refers not only to noise reduction but also to separation of different independent signals and dereverberation. This is a very difficult problem due to two reasons. First, the characteristics as well as nature of the noise signal can change severely in time and between applications. It is defined differently for each application.Two criteria are mainly used to measure performance of speech: intelligibility and quality of speech signal. It is very difficult to satisfy both at same time. Speech enhancement is an important area of speech signal processing where the aim is to improve the pleasantness andintelligibility of a speech signal. The most common approach in speech enhancement is noise removal where evaluation of noise characteristics can cancel noise components and preserve only the clean speech signal.

There are many approaches in the literature to know about speech enhancement.Adaptive filtering has become one of the important and effective approaches for the speech enhancement. Adaptive filters allow to detect time varying potentials and to identify the dynamic variations and changes of the speech signals. Excluding, they improve their behavior or nature according to the input signal. Therefore, they can recognize shape variations in speech samples and thus they can obtain a improve signal estimation. The first adaptive noise cancelling system at Stanford University was evaluated and implemented by two students in 1965.Their work was undertaken as another part of a project for a course in adaptive systems given by the Electrical Engineering Department. Adaptive noise cancelling technique has been successfully applied to a number of different applications from 1965. Different methods have been reported so far in the literature to improve the performance of speech processing systems; some of the most important ones are: LMS filtering [1], spectral subtraction [4]-[5], thresholding [2]-[3], Wiener filtering.Other way LMS-based adaptive filters have been popularly used for speech enhancement [6]–[8]. In a current study, however, a steady state convergence analysis with deterministic reference inputs showed that the steady-state weight vector is biased for the LMS algorithm, so that the adaptive evaluation excludes the Wiener solution. To handle this drawback another strategy was considered for evaluating the coefficients of the linear expansion, namely, the block LMS (BLMS) algorithm [10], in which the coefficient vector is updated only one time and every occurrence based on a block gradient estimation.A major advantage of the transform domain or the block LMS algorithm is that the input speech signals are shortly uncorrelated. Recently Jamal Ghasemi et.al [9] introduced a new approach based on Eigenvalue spectral subtraction for enhancement of speech, [12] authors explains usefulness of speech coding in voice banking, a new method for voicing detection of speech samples and pitch estimation or evaluation of pitch frequency.. This method is based on the spectral analysis of the speech multi-scale product [11].LMS is replaced with its Normalized version, NLMS. In real time applications of LMS filtering, a main parameter is step size. If the step size for LMS is large, the convergence rate of the LMS algorithm will be fast, but the steady-state mean square error (MSE) will increase. Other side, if the step size is small, convergence rate will be slow because the steady

state MSE will be small due to step size. The step size provides a tradeoff between the steady-state MSE and the convergence rate of the LMS algorithm. The performance of the LMS algorithm may be improved by making the step size variable rather than fixed.
This new approach with variable step size is known as variable step size LMS (VSSLMS) algorithm [13].RLS algorithm tracks the time variations to the optimal filter with relatively very high convergence speed of the process which is used in many practical applications such as cancellation of echo, sound control, and radar.

II. ADAPTIVE FILTER

Adaptive filter is a computational device that attempts to model the relationship between two signals in an iterative manner practically. Adaptive filters are self learning or self adjusting filters. As the speech signal into the filter continues, the adaptive filter coefficients adjust themselves to get the desired result, such as recognizing an unknown filter or cancelling unwanted noise signal from input signal. Most adaptive filters are digital filters that perform digital signal processing due to the complexity of the optimizing algorithms.Filters used for noise cancellation purpose can be adaptive fixed. The design of fixed filters is based on prior knowledge of both the speech signal and the noise signal. Other side, adaptive filters have the ability to adjust their own parameters automatically, and their design requires too little or no prior knowledge of signal or noise characteristics of speech signals.Adaptive filtering is a process in which the parameters used for the processing of signals changes according to some criteria. Usually the criterion is the estimated mean or the correlation. The adaptive filters are time varying because their parameters are continually changing in order to meet a performance requirement. Adaptive filter can be considered as a filter that performs the approximation step on line. Usually the definition of the performance criteria needs the reference signal existence that is usually hidden in the approximation step of fixed filter design.Adaptive filtering algorithms which have the adjusting mechanism for the filter coefficients are in fact closely related to classical optimization techniques, all calculations are carried out in an off-line manner. Adaptive filter is sometimes expected track the optimum behavior of a slowly varying environment due to its real time self adjusting characteristics.

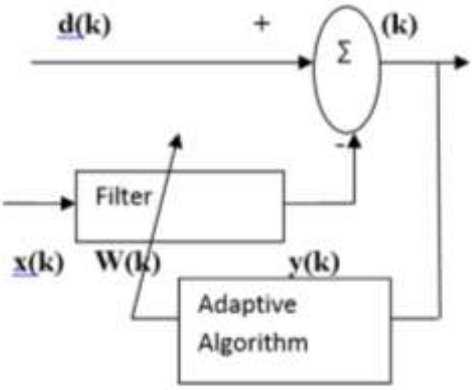

Fig 1: General set up of adaptive filter

III. ADAPTIVE ALGORITHMS

To obtain the best performance of an adaptive filter needs usage of the best adaptive algorithm with low computational complexity and a fast convergence rate.Adaptive algorithm is a process of adjusting the parameters of an adaptive filter to minimize a cost function chosen for present taskThe LMS algorithm is the most popular and frequentlyused adaptive algorithm. Other adaptive algorithms such as Normalized LMS, and RLS that have been applied and developed to increase the speed of adaptive process .This section describes these algorithms and their development process to gain a better understanding of adaptive filtering techniques.

A. Least mean square algorithm

The LMS algorithm was developed by Bernard Widrow. It was the first widely used adaptive algorithm. It was still popularly used in adaptive digital signal processing and adaptive antenna arrays, primarily due to its simplicity, easy implementation and better convergence properties.The LMS Algorithm is successful algorithms due to its efficient storage requirement and computational complexity. LMS algorithm is based upon the steepest descent approach. Basic LMS algorithm updates the filter coefficients after each and every sample of speech sample. An LMS algorithm is the best choice for many real-time systems. There are many types of LMS based algorithms, which include the Normalized LMS, Variable step size LMS, complex LMS, block LMS algorithm and the Time sequenced LMS algorithm.LMS adjusts the adaptive filter taps .LMS is improving them by an amount proportional to the instantaneous estimate of the error surface gradient .it is not requires correlation function calculation or matrix inversions. Due to this it is very simple and easier algorithm compare to others.Adaptive filter based on least mean square will remove the interference signal from measured signal by using a reference signal. The filtration of signal isobtained by designing a least mean square adaptive filter with a specific order and step size that will ensure the adaptation to converge after few seconds of adaptation of the filter.

i) Operation of the least mean square algorithm:

LMS algorithm involves two basic processes.
1. Filtering process 2.Adaptive process
A filtering process which involves two steps. Primary step includes computing the output of a transversal filter produce by a set of tap input and second step includes generation of estimation error by computing this output to a desired response.An Adaptive process, which involves the automatic updating the tap weight according to the estimation error of the filter .Since a feedback loop around the LMS algorithm, establish by the combination of these two processes as illustrated in fig 2.The filtering process is performing by transversal filter so that LMS algorithm is built around transversal filter. Adaptive control mechanism is performing by adaptive filter as explain in fig 2.A hot over the symbols is used for the tap-weight vector to differentiate from the value obtained by using the steepest descent algorithm. The result in the form of three basic relations as follows:

1] Filter output: $Y(n) = \hat{w}(n).u(n)$ ----- (3.1)
2] Estimation error: $E(n) = d(n) - y(n)$----- (3.2)
3] Tap-weight adaptation:
$\widehat{W}(n+1) = \hat{w} + \mu u(n)e^*(n)$ ---------- (3.3)

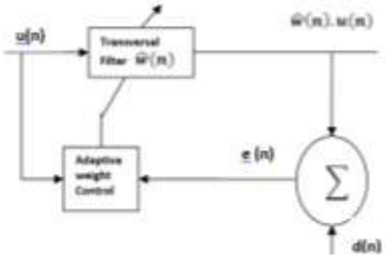

Fig 2: Basic block diagram of Adaptive Processing

The computation of the estimation error e(n) which is based on the present evaluation of the tap-weight vector ŵ(n) is given by equation (3.1) and (3.2). Note that the correlation that is applied to the current estimate of the tap-weight vector, ŵ(n) is represented by the second term μu(n) e*(n), on the right-hand side of equation (3.3) .The correlative procedure is started with an initial guess ŵ(0). The algorithm described by equation (3.1) to (3.3) is the complex form of the adaptive least-mean-square (LMS) algorithm. At each time updating, it is also necessary to know a prior knowledge of the recent values u(n), d(n) & w(n).

ii) Implementation steps for the LMS algorithm:
1. Define the desired response. Set each coefficients weight to zero.
2. Move all the samples in the input array one position to side right position, then load the present data sample k into input array. Calculate the output of the filter by multiplying every element in the filter array of filter coefficients by the corresponding element in the input array and all results are added up to give the output corresponding to data that was earlier loaded into input array.
3. Before filter coefficients can be updated error must be calculated, simply find difference between the desire responses and filter output.
4. To update filter coefficients multiply the error by μ, called as learning rate parameter and then multiply result by filter input and sum up this values of result to previous filter coefficients value.

B.Normalised least mean square algorithm
One of the primary drawbacks of the LMS algorithm is having a fixed step size parameter for each and every iteration. LMS needs knowledge of the statistics of the input signal prior to commencing the adaptive filteringoperation. In practice this is rarely possible.. Even if we assume that the only one signal to be input to the adaptive echo cancellation system is speech, there are many other factors such as amplitude and signal input power which will affect its performance**.**
The normalised least mean square algorithm (NLMS) is an algorithm which bypasses this issue by calculating maximum step size value. Since it is called as Extension of LMS algorithm. Step size value is calculated by using the following mathematical formula.
Step size=1/dot product (input vector, input vector)

Step size is one of the important parameter in adaptive algorithms. It is proportional to the inverse of the total assumed energy of the coefficients of the input vector x(n)of the instantaneous values. This addition of the expected values of the input samples is also equal to the dot product of the input vector with itself, and auto-correlation matrix, R which is trace of the input vectors.
$tr[R] = \sum_{i=0}^{N-1} E[x^2(n-i)]$
$= E[\sum_{i=0}^{n-1} x^2(n-i)]$(3.4)
The recursion formula for NLMS algorithm shown in equation (3.5)
$w(n+1) = w(n) + \frac{1}{x^T(n)\,x(n)} e(n)x(n)$(3.5)

C.Variable Step size least mean square algorithm (VSSLMS)
The idea of improving performance of LMS was, choosing larger step sizes in the starting stage to achieve higher convergence rate, and then choosing smaller step sizes to get less steady-state error. Modifying the secant integral function, it was easy to obtain equation (3.6), which was an infinitely differentiable function.
$Y = 1 - \sec h(x) = 1 - 2/(e^x + e^{-x})$ (3.6)
A new step size could constructed by replacing *x*and *y* with *e*(*n*) and *u*(*n*)
$\mu(n) = \beta\{1 - \sec h\,[\alpha\, e(n)^\gamma]\}$(3.7)
When the algorithm was almost steadyα and γ affected the error and β affected the convergence rate .Algorithm described as follows,

$E(n) = d(n) - y(n) = d(n) - WX^T(n)$(3.8)
$\mu(n) = \beta\{1 - \sec h\,[\alpha\, e(n)^\gamma]\}$(3.9)
$W(n+1) = W(n) + 2\,\mu\, e(n)\, X(n)$ (3.10)

Thus, the step size gives the tradeoff between the steady state MSE and convergence rate.

Algorithm	Weight updates recursion
LMS	$w(n+1) = w(n) + \mu e(n)\, x(n)$
NLMS	$w(n+1) = w(n) + \frac{\mu}{\gamma + x^T(n)x(n)} e(n)x(n)$
VSSLMS	$w(n+1) = w(n) + \mu(n)e(n)x(n)$

D. Recursive least square Algorithm
An algorithm which finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals recursively called as Recursive least square algorithm. RLS is in contrast with LMS and other algorithms. In the derivation of the RLS, the input signals areknown as deterministic,while for LMS and other similar algorithms are considered as stochastic. The RLS shows extremely fast convergence since RLS have high computational complexity.RLS Algorithm is potential alternative to overcome a drawback of slow convergence in colored environments .RLS uses the least squares method. Least squares develop a recursive algorithm for the adaptive transversal filter. RLS tracks the time variation to the optimal filter coefficient of the process with very fast convergence speed. Practical applications such as speech enhancement, channel equalization, echo cancellation,sound control and radar uses adaptive algorithm like RLS

algorithm. RLS has stability problems compared with the other adaptive algorithms.

RLS algorithm is summarized as follows; RLS algorithm has been derived using the matrix inversionlemma. The Weighted Least-Squares algorithm used for matrix inversions

Initial Value$P(0)=\delta^{-1}I$ (3.11)

Error signal equation

$e(n) = d(n) - w^T(n-1)x(n)$ (3.12)

Filter gain vector

$k(n) = \frac{P(n-1)x(n)}{\lambda + x^T p(n-1)(x(n)}$(3.13)

Filter coefficient adaptation

$w(n) = w(n-1)+k(n)e(n)$ (3.14)

Inverse correlation matrix(3.15)

$P(n) = \lambda^{-1}P(n-1) - \lambda^{-1}k(n)x^T P(n-1)$s

Equation (3.12) error signal describes filtering operation of RLS. Equation (3.14) filter coefficient adaptation describes Adaptive algorithm process. The tap –weight vector *w(n)* is updated by incrementing its old value. It is updated by an amount equal to the product of filter gain vector *k(n)* and error signal *e(n)*in the process. The inverse correlation matrix and filter gain vector are shown in equation (3.15) and (3.13) respectively, both are update the value of gain vector itself.

RLS algorithm has many drawbacks like the potential divergence behavior in finite-precision environments and high computational complexity, which is of order of N2, where N represents the filter length. Lost symmetry and positive definiteness of the matrix *P(n)* caused stability problems. Implementations exist based on QR decomposition of matrix *P(n)* square-root factorization. Variants of fast transversal algorithms with computational complexity of order N have beenproposed for model systems for long impulse responses and non-stationary environments many of these variants suffer from stability problems it is happen when implemented in finite precision.

Table 1: Comparison of various algorithms

Algorithms	Advantages
LMS	Low computational complexity, simple program
NLMS	Better numerical properties
RLS	Better numerical properties, Fast convergence, and goodtracking ability.

IV. Conclusion

It was verified how adaptive algorithms are used to adjust the coefficients of a digital filter to achieve a desired and time-varying response in several practical situations as well as applications .Impact was given on the description of different adaptation algorithms.LMS is an adaptive algorithm based on a stochastic gradient descent method .LMS algorithm adapts the evaluation based on the current error. NLMS is a extended part of the LMS algorithm with a normalized step length. RLS is an adaptive algorithm based on the idea of least square that are used to minimize the impact of old measurements.

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