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Speech signal processing with adaptive noise canceller- A Review

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Abstract: Noise related problems in the environment have gained attention due to the enormous growth of technology. Excessive noise in the environment proves damaging to humans from a physical as well as psychological aspect. The problem of controlling noise level in the environment has been the focus of tremendous amount of research over years. This paper describes a study of adaptive filtering algorithms that can be applied to the input of standard receivers that are trained on noise free speech. Such methods are very useful in various applications which include communication systems biomedical engineering and industrial applications. In this review, we have classified adaptive filtering algorithms that is LMS(least mean square),NLMS(normalized least mean square) and RLS(recursive least square) according to their computational complexity, convergence speed, signal to noise ratio to demonstrate limitations as well as their effective contribution to the existing communication quality.

Keywords: ANC (Adaptive Noise Canceller), LMS, NLMS, RLS

I.INTRODUCTION

In all practical situations, the received waveform contains some form of noise component. Depending on the amount and type of noise, the quality of received signal can range from slightly degraded to being totally unintelligible. The problem of removing unwanted noise has been the subject of numerous research.[1] Several techniques have been proposed by various researchers like spectral subtraction adaptive noise cancelling. In digital communication Adaptive filters have advantage over traditional filters. An adaptive filter is a filter that adjusts its transfer function according to an optimizing algorithm. High speed digital data transmissions mostly suffer from inter symbol interference (ISI) and additive noise. The adaptive algorithms recursively determine the filter coefficients in order to eliminate the effects of noise and ISI.[1,2]

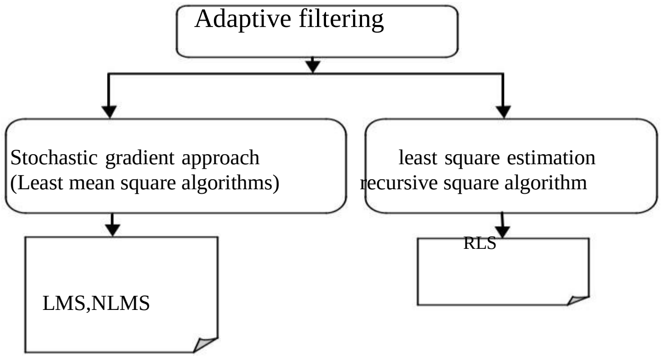

Fig.1. Hierarchy of adaptive filters

II. ADAPTIVE NOISE CANCELLER

The reference noise x(n) is input to the tranversal filter.The output of the filter is y(n) which is convolution of x(n) and filter tap weight W(n).The noisy signal d(n) consists of information signal S(n) corrupted by noise N(n).The y(n) and d(n) are compared to give error signal e(n).The coefficients of the adaptive filter are changed

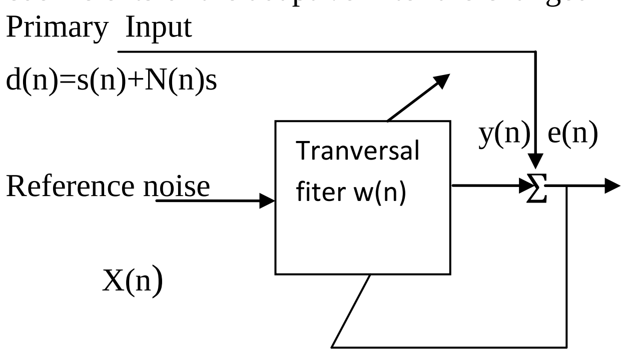

Fig.2. Adaptive noise canceller

III. LEAST MEAN SQUARE ALGORITHM (LMS)

Least mean square algorithm is the most widely used algorithm due its computational simplicity.LMS are the class of adaptive filters that produce least mean squares of the error signal. It is a stochastic gradient descent method in which the filter is only adapted to error at the current time.

$$E\ [\ e(n)^2\ ] = E\ [(\ d(n) - y(n))^2\ ] \qquad (1)$$

The filter weight w(n) is changed iteratively

in iterative manner according to the error signal e(n).The e(n) is the estimated clean signal at the output. [2,6]

compute the updated filter coefficient w(n+1) to minimize the error iteratively. The algorithm updates the next tap filter using current tap weight and current gradient of cost function as given in equation (1)

$$w(n+1) = w(n) - \mu\Delta\xi(n) \qquad (2)$$

the gradient can also be expressed as

$$\Delta\xi(n) = \Delta(e^2(n)) = -2e(n)x(n)$$

$$W(n+1) = w(n) + 2\mu e(n)x(n) \qquad (3)$$

The convergence rate of LMS depends on step size μ. If μ is small then it may long time for convergence, if μ is too large the algorithm may never converge. The value of

μ should be computed based on the environmental effects on d(n). For algorithm to be stable the operation boundary is given by

$$0 < \mu < \frac{2}{\lambda_{max}} \qquad (4)$$

IV. NORMALIZED LEAST MEAN SQUARE ALGORITHM (NLMS)

to trace the desired signal d(n).The difference between d(n) and y(n) gives the error signal e(n) as shown in fig 2.[3]Then this e(n) is then fed to LMS algorithm to

A subclass of LMS algorithm that is commonly used is normalized least mean square (NLMS).These algorithms are upgrades to the standard LMS algorithms. The main drawback of the pure “LMS” is that it is sensitive to the scaling of its input.

This makes it difficult to choose a learning rate μ which guarantees the stability of the algorithm.The normalized least mean square algorithm (NLMS) solves this problem by normalizing with the power of the input. The normalization factor is inversely proportional to the instantaneous power. So when signal is weak, it increases step size proportionally and when signal is too large it reduces step size accordingly. It acts to increase both the stability and the convergence rate by allowing larger step size to be used. Therefore the weight update equation for NLMS is given by

$$w(n+1) = w(n) + 2 \frac{\mu}{\gamma + \| x(n) \|^2} e(n)x(n) \qquad (5)$$

The squared norm of the input signal is given by

$$\| X(n) \|^2 = \sum_{k=0}^{N} (x(n-k))^2 \qquad (6)$$

where N is the length of adaptive filter

The NLMS is most commonly used adaptive filtering due to their simplicity in their structure and implementations. They do add more complexity to the standard form of LMS, however normalized factor is computed once per iteration. Also the step size trade off are significantly helped by the added flexibility.[4]

V.RECURSIVE LEAST SQUARE ALGORITHM

The Recursive Least Squares (RLS) algorithm is based on the well-known least squares method . The least-squares method is a mathematical procedure for finding the best fitting curve to a given set of data points. This is done by minimizing the sum of the squares of the offsets of the points from the curve. The recursive least square (RLS) is a class of algorithms that operate in a different manner than LMS algorithm. The optimal coefficient vector is calculated in each iteration, providing the filter with better performance with respect to LMS algorithm. Recursive least squares adaptive filter is a algorithm which recursively finds the filter coefficients that minimizes the cost of the linear least squares related to the input signal.RLS algorithms are known for their excellent performance but at the cost of an increased computational complexity and some stability problems.[4,7]

The equation for RLS algorithm is given by

$$\xi(n) = \sum_{k=1}^{N} \lambda^{n-k} e^2(n) k \qquad (5)$$

Here k=1 is the time at which the RLS algorithm commences and λ is a small positive constant very close to, but smaller than 1. With values of $\lambda<1$ more importance is given to the most recent error estimates and thus the more recent input samples. RLS algorithms are known for excellent performance when working in time varying environments. these advantages come with the cost of an increased computational complexity and some stability problems.[9,10]

FAST TRANSVERSAL RLS ALGORITHM

FTRLS algorithm involves the combined use of four transversal filters for forward and backward predictions, gain vector computation and joint process estimation. The main advantage of FTRLS algorithm is reduced computational complexity as compared to the other available solutions.[8]

VI. CONCLUSION

In these algorithms LMS is the most popular algorithm used because of their low computational complexity. However LMS suffers from slow and data dependent convergence rate. The NLMS an equally simple but robust variant of LMS algorithm exhibits a better balance between simplicity and performance than LMS algorithm. Due to its good characteristics it is used widely in real time applications.RLS has faster convergence rate than LMS algorithm but at a cost of increased computational complexity.The SNR is better in RLS as compared to LMS and NLMS algorithms.

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