Noise Suppression in speech signals using Adaptive algorithms

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Adaptive Filtering is a widely researched topic in the present era of communications. When the received signal is continuously corrupted by noise where both the received signal and noise change continuously, then arises the need for adaptive filtering. The heat of the adaptive filter is the adaptive algorithm. This paper deals with cancellation of noise on speech signals using three algorithms-Least Mean Square (LMS) algorithm, Normalized Least Mean Square (NLMS) and Sign-Data Least mean Square (SDLMS) algorithm with implementation in MATLAB. Comparisons of algorithms are based on SNR. The algorithms chosen for implementation which provide efficient performances with less computational complexity.

Keywords* : *Adaptive noise canceller, Least Mean Square, Normalized Least Mean Square , Sign-Data Least mean Square.

1.INTRODUCTION

In this modern world we are surrounded by all kinds of signals in various forms. Some of the signals are natural, but most of the signals are man-made. Some signals are necessary (speech); some are pleasant (music), while many are unwanted or unnecessary in a given situation. In an engineering context, signals are carriers of information, both useful and unwanted. Therefore extracting or enhancing the useful information from a mix of conflicting information is a simplest form of signal processing. More generally, signal processing is an operation designed for extracting, enhancing storing, and transmitting useful information. The distinction between useful and unwanted information is often subjective as well as objective. Hence signal processing tends to be application dependent. In contrast to the conventional filter design techniques, adaptive filter do not have constant filter coefficients and no priori information is known. Such a filter with adjustable parameters is called an adaptive filter. The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics. Although both FIR and IIR filters is by far the most practical and widely used. The reason being that FIR has adjustable zeros, and hence it is free of stability problems associated with adaptive IIR filters that have adjustable poles as well as zeros. However the adaptive FIR filters are not always stable and their stability depends critically on the Algorithm.

2. ADAPTIVE FILTERING

Adaptive filtering can be considered as a process in which the parameters used for the processing of signals changes according to some criterion. Usually the criterion is the estimated mean squared error or the correlation. The adaptive filters are time-varying since their parameters are continually changing in order to meet a performance requirement. In this sense, an adaptive filter can be interpreted as a filter that performs the approximation step on-line. Usually the definition of the performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design. The general setup of adaptive filtering environment is shown in figure1 below, where k is the iteration number, x(k) denotes the input signal, y(k) is the adaptive filter output, and d(k) defines the desired signal. The error signal e(k) is calculated as d(k)-y(k). The error is then used to form a performance function or objective function that is required by the adaptation algorithm in order to determine the appropriate updating of the filter coefficients. The minimization of the objective function implies that the adaptive filter output signal is matching the desired signal in some sense.

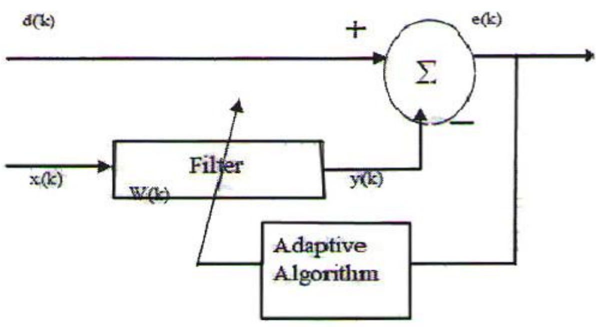

Figure 1: General setup of adaptive filter

3.ADAPTIVE NOISE CANCELLER

Adaptive filter is widely used as noise canceller. In an adaptive noise canceller figure2 (3) two input signals, d(k) and x(k), are applied simultaneously to the adaptive filter. The signal d(k) is the contaminated signal containing both the desired signal, s(k) and the noise n(k), assumed uncorrelated with each other. The signal, x(k) is a measure of the contaminating signal which is correlated in sole way with n(k), x(k) is processed by the digital filter to produce an estimate y(k), of n(k). An estimate of the desired signal, e(k) is then obtained by subtracting the digital filter output y(k), from the contaminated signal.

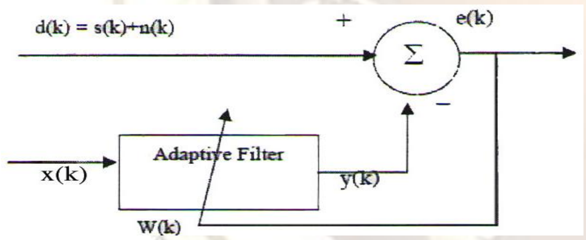

Figure 2 : Adaptive Filter as noise canceller

3.1 LMS ALGORITHM

LMS algorithm was first proposed by Widrow and Hoff in 1960. It is the most widely used for adaptive filter algorithm. It is simply approximate version of steepest descent method and is used for adjusting a set of adaptive filter's coefficient. The process of this algorithm, it send filter input that is specified by x(k) passing through adaptive filter. It give filter output is specified by y(k). the process in adaptive filter is assumed by this equation.

$$y(k) = w(k)^T \ x(k)$$

where $w(k)^T$ is transpose and k is time index. It compares the filter output to the desired signal that is specified by d(k) and gets the error by this equation

$$e(k) = d(k) - y(k)$$

The LMS algorithm try to adjust the adaptive filter coefficient that make e(k) be minimized, it is shown in function mean square error (MSE), it use the error to adjust the next coefficient of adaptive filter by this equation.

$$w(k+1) = w(k) + \mu e(k) \times (k)$$

3.2 NORMALIZED LMS (NLMS)

Normalized LMS (NLMS) algorithm is another class of adaptive algorithm used to train the coefficients of the adaptive filter. This algorithm takes into account variation in the signal level at the filter output and selecting the normalized step size parameter that results in a stable as well as fast converging algorithm. The weight update relation for NLMS algorithm is as follows

$$w(k+1) = w(k) + \mu e(k) \times (k)$$

The variable step can be written as

$$\mu e(k) = \frac{\mu}{p + x^t(k) \times (k)}$$

Here μ is fixed convergence factor to control small adjustment. The parameter p is set to avoid denominator being too small and step size parameter.

3.3 SIGN-DATA LMS ALGORITHM

New algorithms that made use of the signum (polarity) of the input signal have been derived from the LMS algorithm for the simplicity of implementation, enabling a significant reduction in computing time, particularly the time required for "multiply and accumulate" (MAC) operations. In this paper we consider the Signed Regressor Algorithm (SRA), for which the weight update relation is given as

$$w(k+1) = w(k) + \mu \operatorname{sgn}\{x(k)\} \times e(k)$$

where sgn{.} is the well known signum function.

Among the three adaptive algorithms presented above, the SRA has a convergence rate and a steady-state error that are slightly inferior to those of the LMS algorithm for the same parameter setting. But, the computational complexity of SRA is much less compared to LMS algorithm. The advantage of the NLMS algorithm is that the step size can be chosen independent of the input signal power and the number of tap weights. Hence the NLMS algorithm has a convergence rate and a steady state error better than LMS algorithm. On the other hand some additional computations are required to compute μ(n).

4.SIMULATION RESULTS

The parameter of clean speech sample figure 3 considered for testing of the algorithms were: duration 2 second, PCM 22.050kHz.

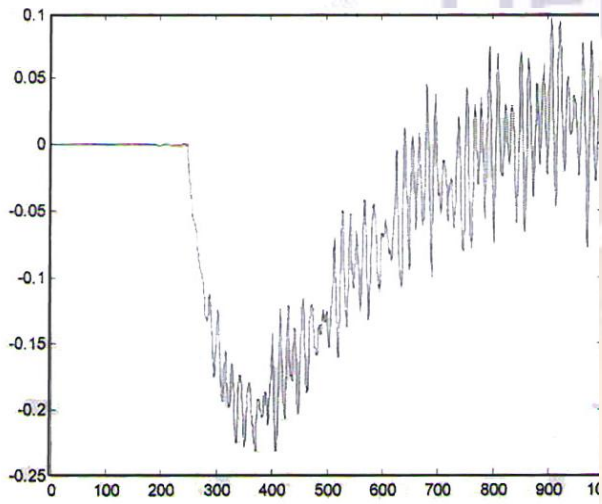

Figure 3: Original Speech signal (1000 samples)

A random noise is generated using randn function and added to the original speech signal. The SNR of the signal corrupted with noise was –1.9966dB (noise power= -4.7357e-004). A linear combination of the generated noise and the original signal is used as primary input for the filter. The below figure 4 shows original speech signal corrupted by noise.

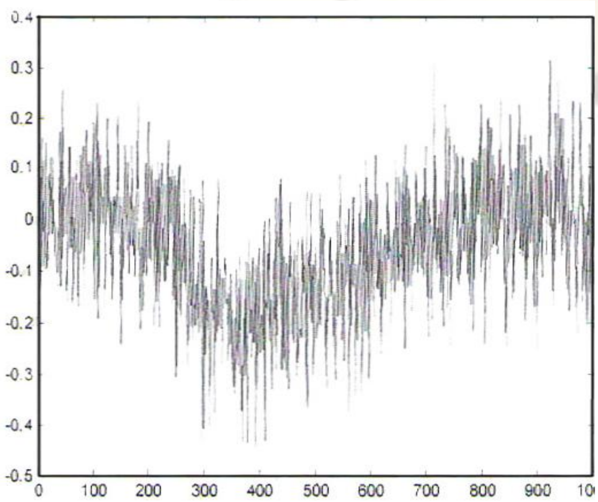

Figure 4: Signal Corrupted by noise

Denoised speech signal obtained by using LMS filter is shown in below figure 5. The output signal to noise ratio of signal denoised with LMS algorithm was 9.8793dB (noise power = -1.442e-004)

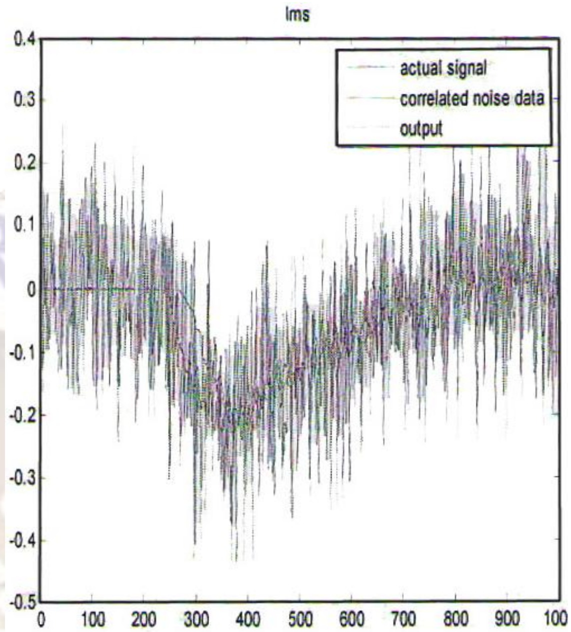

Figure 5: Denoised speech signal using LMS

Denoised speech signal obtained by using NLMS filter is shown in below figure 6. The output signal to noise ratio of signal denoised with NLMS algorithm was 13.9213 +31.4159i (noise power =9.6401e-005).

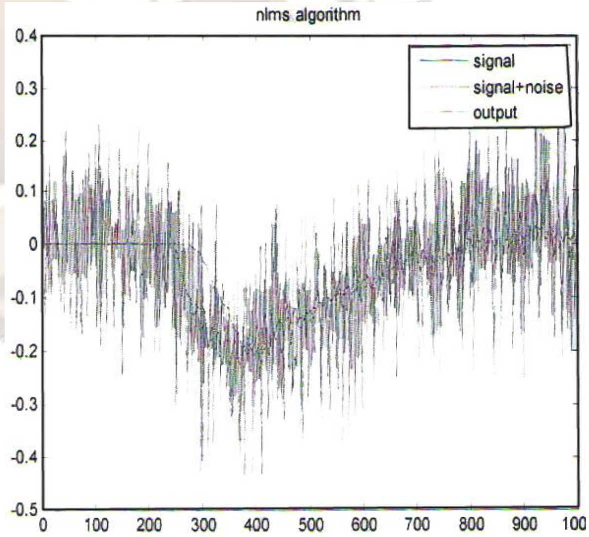

Figure 6: Denoised speech signal using NLMS

Denoised speech signal obtained by using SDLMS filter is shown in below figure 7. The output signal to noise ratio of signal denoised with SDLMS algorithm was –9.5117 +31.4159i (noise power = 0.0010).

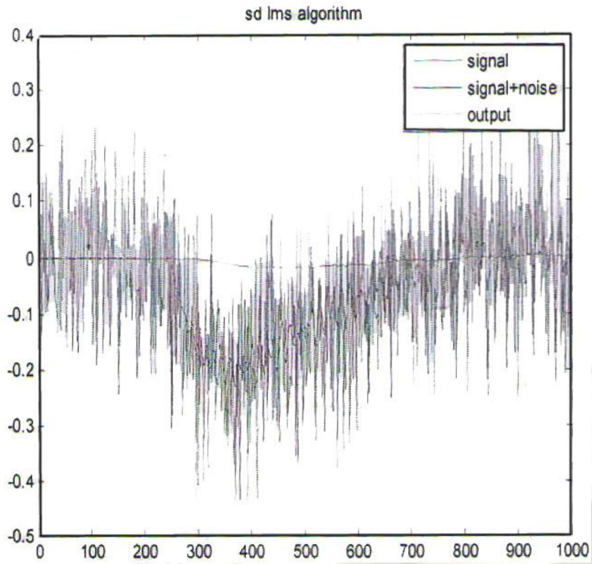

Figure7: Denoised speech signal using SDLMS

Algorithm used	SNR in dB
LMS	9.8793
NLMS	13.9213 +31.4159i
SDLMS	-9.5117 +31.4159i

Table

References

[1] Adaptive Filter Theory by Simon Hyakin; 3rd edition, Pearson Education, Asia, LPE.
[2] S. Koike, "Analysis of Adaptive Filters Using Normalized signed Regressor LMS Algorithm", IEEE Transactions on Signal processing, Volo.47, No.1, pp.2710-2723, 1999.
[3] E.Eweda, "Analysis and design of a signed regressor LMS algorithm for stationary and non-stationary adaptive filtering with correlated Gaussian data", IEEE Transactions on Circuits and Systems, Vol.37, No.11, pp.1367-1374m 1990.
[4] S. Olmos, L. Sornmo and P. Laguna, "Block adaptive filter with deterministic reference inputs for event-related signals: BLMS and BRLS", IEEE Trans. Signal processing, Vol.50, pp.1102-1112, May, 2002.