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Adaptive Noise Cancellation in Speech Signal

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Abstract : Speech has always been one of the most important carriers of information for people. It becomes a challenge to maintain its quality. In many application of noise cancellation, the changes in signal characteristics could be quite fast.So, To eliminate background noise from the main signal, adaptive filtering techniques should be used.The adaptive noise cancelling is an alternative method of estimating signals corrupted by additive noise or interference.The principle advantages of the method are its adaptive capability, its low output noise, and its low signal distortion. This paper describes the use of adaptive algorithms to reduce unwanted noisy signal, thus increasing communication quality.
Keywords: Adaptive filter, Noise cancellation, Adaptive algorithm.

I. INTRODUCTION:

In the process of digital signal processing, often to deal with some noise or time-varying signals. By using a two FIR and IIR filter of fixed coefficient cannot achieve optimal filtering. Under such circumstances, design of adaptive filters is necessary, to track the changes of signal and noise. Adaptive Filter uses the filter parameters of a moment ago to automatically adjust the filter parameters of the present moment, in order to achieve optimal filter. The adaptive filter uses feedback in the form of an error signal to refine its transfer function to match the changing parameters.

The goal of adaptive noise cancelling is to eliminate background noise from the main signal, which is composed of the desired signal and background noise that has been correlated with noise from a reference measurement. The technique therefore relies upon access to a reference signal, located at the source of noise fields, as well as the main or primary signal.

In general, noise that affects the speech signals can be modeled using any one of the following :

a. White noise,
b. Colored noise,
c. Impulsive noise.

a. White noise : White noise is a sound or signal consisting of all audible frequencies with equal intensity. It can be any value between 0 and 2π. Because of the broad-band spectrum, white noise has strong masking capabilities.

b. Colored noise : Any noise that is not white can be termed as colored noise. Colored noise has frequency spectrum that is limited within a range unlike white noise which extends over the entire spectrum. Colored noise can be generated by passing white noise through a filter with required frequency response.

c. Impulsive noise : Impulsive noise refers to sudden bursts of noise with relatively high amplitude. This type of noise causes click sounds in the signal of interest.

1.1 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. The adaptive filters are time-varying since their parameters are continually changing in order to meet a performance requirement.

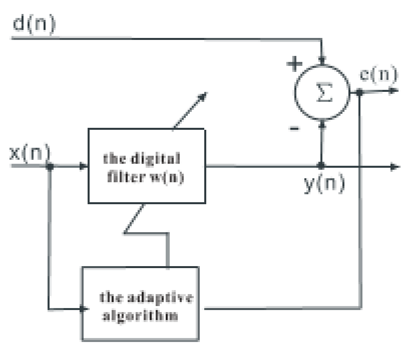

Fig.1 General setup of adaptive filter

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The general set up of adaptive filtering environment is as shown in fig.1,where n is the iteration number, x(n) denotes the input signal, y(n) is the adaptive filter output, and d(n) defines the desired signal.
The error signal e(n) is calculated as d(n)-y(n). 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.

1.2 Adaptive Noise Cancellation:
Adaptive noise cancellation is widely used to improve the Signal to Noise Ratio (SNR) of a signal by removing noise from the received signal. In this configuration the input x(n), a noise source N1(n), is compared with a desired signal d(n), which consists of a signal s(n) corrupted by another noise N0(n). The adaptive filter coefficients adapt to cause the error signal to be a noiseless version of the signal s(n) as shown in Fig.2.

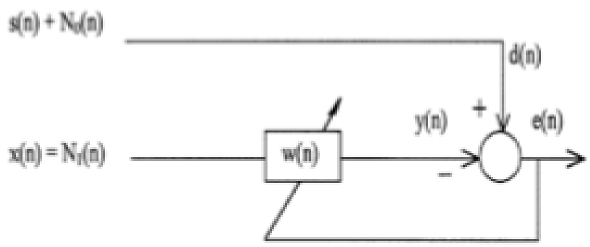

Fig.2 Adaptive noise cancellation configuration

The noise signals for this configuration need to be uncorrelated to the signal s(n). In addition, the noise sources must be correlated to each other in some way, preferably equal, to get the best results. The error signal should converge to the signal s(n), but it will not converge to the exact signal. In other words, the difference between the signal s(n) and the error signal e(n) will always be greater than zero. The only option is to minimize the difference between those two signals using certain error minimization techniques [4].

II. ADAPTIVE ALGORITHMS :

There are many algorithms used to adjust the coefficients of the digital filter in order to match the desired response as well as possible. This includes the following :
2.1 Least mean square (LMS) algorithm :

- This is introduced by Widrow and Hoff in 1959 is an adaptive algorithm.
- It incorporates an iterative procedure leads to the minimum mean square error.
- It does not require correlation function calculation nor it require matrix inversions. The least mean square algorithm is a linear adaptive filtering algorithm that consists of two basic process :

a] Filtering process:
This involves (i) computing the output of a transversal filter produced by a set of tap inputs, and (ii) generating an estimation error by comparing this output to a desired response.
b]Adaptive process:
This involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error.

- Implementation of the LMS Algorithm :
  Each iteration of the LMS algorithm requires distinct steps in this order:

1.Filter output :
y[n] = $\sum_{k=0}^{M-1} u(n-k) w_k^*[n]$ (1)

2.Estimation error :
e[n] = d[n] – y[n] (2)

3.Tap-weight adaptation :
$$w_k[n+1] = w_k[n] + \mu u[n-k] e^*[n] \quad (3)$$

- Results of LMS algorithm :

The LMS algorithm was simulated using Matlab. Fig.3 shows the input speech signal which is collected from the computer system. Fig.4 shows the desired echo signal derived from the input signal. Fig.5 shows the adaptive filter output which will reduce the echo signal from the input signal. Fig.6 shows the mean square error signal calculated from the filter output signal. The step size was set to 0.02. The MSE shows that as the algorithm progresses the average value of the cost function decreases.

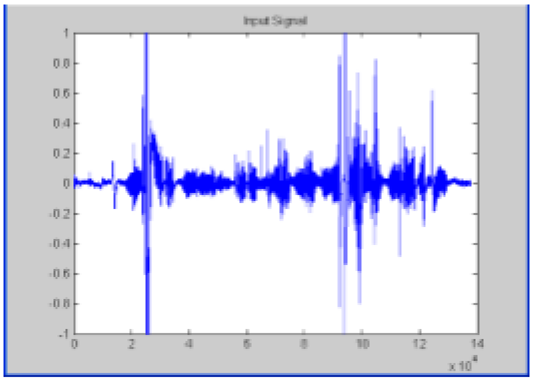

Fig.3 Input signal

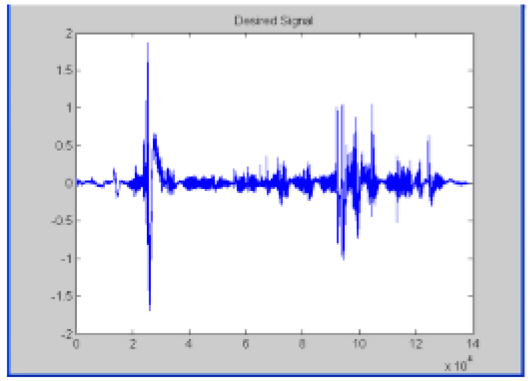

Fig.4 Desired signal

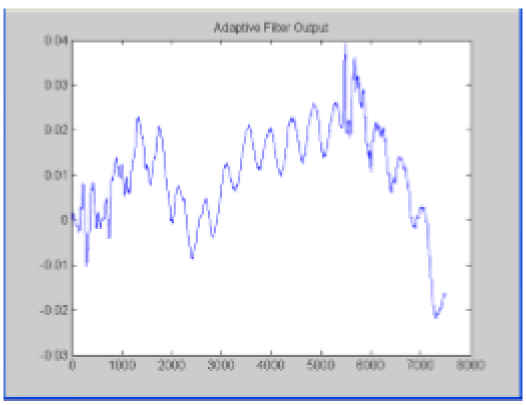

Fig.5 Adaptive filter output

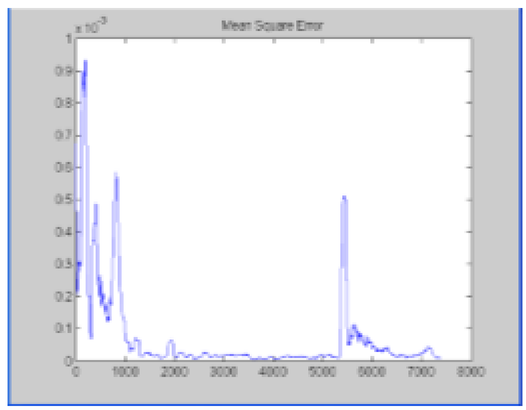

Fig.6 Mean square error

2.2 Normalised Least mean square (NLMS) algorithm :

- One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance.
- The normalised least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value.
- Step size value is calculated by using the formula : Step size=1/dot product (input vector, input vector). This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n). This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, R :

tr [R] = $\sum_{i=0}^{N-1} E[x^2(n-i)]$

= E { $\sum_{i=0}^{N-1}[x^2(n-i)]$ } (4)

- The recursion formula for the NLMS algorithm is stated in equation :

w(n+1) = w(n) + $\frac{1}{x^T(n)x(n)}$e(n)x(n) (5)

- Implementation of the NLMS Algorithm:
  Each iteration of the NLMS algorithm requires these steps in the following order :
  1.The output of the adaptive filter is calculated as :

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = w^T(n)\, x(n) \tag{6}$$

2.An error signal is calculated as the difference between the desired signal and the filter output :

$$e[n] = d[n] - y[n] \tag{7}$$

3. The step size value for the input vector is calculated :

$$\mu(n) = \frac{1}{x^T(n)x(n)} \tag{8}$$

4. The filter tap weights are updated in preparation for the next iteration :

$$w(n+1) = w(n) + \mu(n)\, e(n)\, x(n) \tag{9}$$

- Results of NLMS algorithm :

The NLMS algorithm was simulated using Matlab.Fig.7 shows the input signal.Fig.8 shows the desired signal.Fig. 9 shows the adaptive filter output. Fig. 10 shows the mean square error.The step size was set to 0.1.

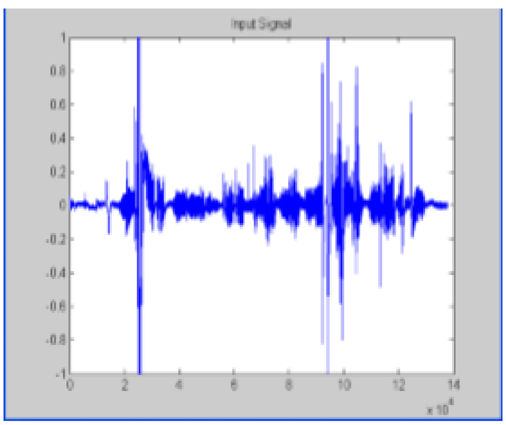

Fig.7 Input signal

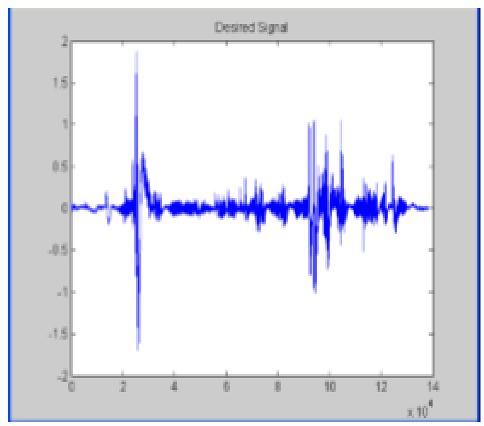

Fig.8 Desired signal

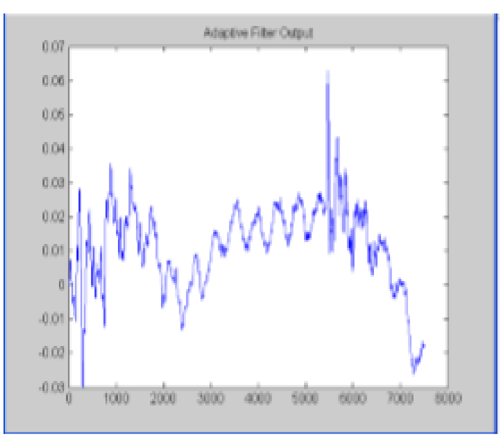

Fig.9 Adaptive filter output

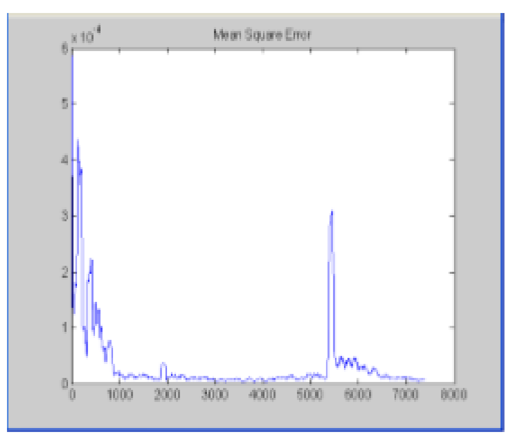

Fig.10 Mean square error

III. THE SUMMARY OF ADAPTIVE ALGORITHM PERFORMANCE :

Table 1 shows that NLMS algorithm is having the advantage over the LMS algorithm incase of Mean square error.

Table 1 :Summary of adaptive algorithm performance

ALGORITHM	ITERATIONS	FILTER ORDER	MEAN SQUARE ERROR
LMS	7500	1025	0.001
NLMS	7500	1025	0.0004

IV. CONCLUSION:

From this paper it is clear that Adaptive filtering is an important basis for signal processing and adaptive noise cancellation is an alternative way of cancelling noise present in a corrupted signal. Also different algorithms of noise cancellation are discussed by considering various factors.

V. REFERENCES:

[1] Allam Mousa, Marwa Qados, Sherin Bader: Speech Signal Enhancement Using Adaptive Noise Cancellation Techniques. Canadian Journal on Electrical and Electronics Engineering Vol. 3, No. 7, September 2012 .

[2] Radhika Chinaboina, D.S.Ramkiran, Habibulla Khan: Adaptive Algorithms For Acoustic Echo Cancellation In Speech Processing. IJRRAS 7(1), April 2011.

[3] Ying He, Hong He, Li Li, Yi Wu: The Applications and Simulation of Adaptive Filter in Noise Cancelling. International Conference on Computer Science and Software Engineering, © 2008 IEEE, Tianjin University of Technology, China.

[4] V. R. Vijaykumar, P. T. Vanathi, P. Kanagasapabathy: Modified Adaptive Filtering Algorithm for Noise Cancellation in Speech Signals. ISSN ELECTRONICS AND ELECTRICAL ENGINEERING,2007.