A Design Approach For Noise Cancellation In Adaptive LMS Predictor Using MATLAB.

Pranjali M. Awachat[1] (Research Scholar[1]), S.S.Godbole[2] (Assistant Professor[2])

Electronics Engineering Department
G.H.Raisoni College of Engineering
Nagpur,

Electronics & Telecommunication
G.H.Raisoni College of Engineering
Nagpur,

Abstract

The main goal of this paper is to present a simulation scheme to simulate an adaptive filter using LMS (Least mean square) adaptive algorithm for noise cancellation. The main objective of the noise cancellation is to estimate the noise signal and to subtract it from original input signal plus noise signal and hence to obtain the noise free signal. There is an alternative method called adaptive noise cancellation for estimating a speech signal corrupted by an additive noise or interference. This method uses a primary input signal that contains the speech signal and a reference input containing noise. The reference input is adaptively filtered and subtracted from the primary input signal to obtain the estimated signal. In this method the desired signal corrupted by an additive noise can be recovered by an adaptive noise canceller using LMS (least mean square) algorithm. This adaptive noise canceller is useful to improve the S/N ratio. Here we estimate the adaptive filter using MATLAB/SIMULINK environment.

Key words: LMS algorithm, Noise cancellation, Adaptive filter, MATLAB/SIMULINK.

I. Introduction:

Noise is a nuisance or disturbance during communication and it is unwanted. Noise occurs because of many factors such as interference, delay, and overlapping. Noise problems in the environment have gained attention due to the tremendous growth of technology that has led to noisy engines, heavy machinery, high electromagnetic radiation devices and other noise sources. For noise cancellation with the help of adaptive filter and employed for variety of practical applications like the cancelling of various forms of periodic interference in electrocardiography, the cancelling of periodic interference in speech signals, and the cancelling of broad-band interference in the side-lobes of an antenna array. In sound signal or speech signal, noise is very problematic because it will difficult to understanding of the information. Speech is a very basic way for humans to convey information to one another with a bandwidth of only 4 kHz; speech can convey information with the emotion of a human voice. The speech signal has certain properties: It is a one-dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary, i.e. the frequency spectrum is not constant in time. Although human beings have an audible frequency range of 20Hz to 20 kHz, the human speech has significant frequency components only up to 4 kHz. The most common problem in speech processing is the effect of interference noise in speech signals. In the most of practical applications Adaptive filters are used and preferred over fixed digital filters because adaptive filters have the property on the other hand, have the ability to adjust their own parameters automatically, and their design requires little or no a priori knowledge of signal or noise characteristics. In this paper we have to used adaptive filter for noise cancellation. The general configuration for an Adaptive filter system is shown in Fig.1. It has two inputs: the primary input *d(n)*, which represents the desired signal corrupted with

c undesired noise, and the reference signal *x(n)*, which is the undesired noise to be filtered out of the system. The goal of adaptive filtering systems is to reduce the noise portion, and to obtain the uncorrupted desired signal. In order to achieve this, a reference of the noise signal is needed and is called reference signal x(n). However, the reference signal is typically not the same signal as the noise portion of the primary amplitude, phase or time. Therefore the reference signal cannot be simply subtract from the primary signal to obtain the desired portion at the output.

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

1. White noise,
2. Colored noise,

II. Adaptive Filter :

Concept of adaptive noise cancelling

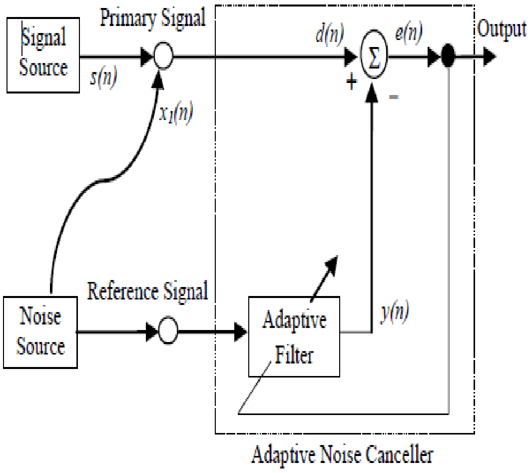

Fig.1.Adaptive Noise Cancellation System

Where

$s(n)$ - Source signal
$d(n)$ - Primary signal
$x_1(n)$ - Noise signal
$x(n)$ - Noise Reference input
$y(n)$ - Output of Adaptive Filter
$e(n)$ - System Output Signal

Adaptive Filtering

Fig. 1 shows the adaptive noise cancellation setup. In this application, the corrupted signal passes through a filter that tends to suppress the noise while leaving the signal unchanged. This process is an adaptive process, which means it cannot require a priori knowledge of signal or noise characteristics. Adaptive noise cancellation algorithms utilize two signal it can vary in (sensor). One signal is used to measure the speech + noise signal while the other is used to measure the noise signal alone. The technique adaptively adjusts a set of filter coefficients so as to remove the noise from the noisy signal. This technique, however, requires that the noise component in the corrupted signal and the noise in the reference channel have high coherence. Unfortunately this is a limiting factor, as the microphones need to be separated in order to prevent the speech being included in the noise reference and thus being removed. With large separations the coherence of the noise is limited and this limits the effectiveness of this technique. In summary, to realize the adaptive noise cancellation, we use two inputs and an adaptive filter. One input is the signal corrupted by noise (Primary Input, which can be expressed as $s(n)$ □ $x_1(n)$). The other input contains noise related in some way to that in the main input but does not contain anything related to the signal (Noise Reference Input, expressed as $x(n)$). The noise reference input pass through the adaptive filter and output $y(n)$ is produced as close a replica as possible of $x_1(n)$. The filter readjusts itself continuously to minimize the error between $x_1(n)$ and $y(n)$ during this process. Then the output $y(n)$ is subtracted from the primary input to produce the system output $e(n)=s(n)+x_1(n)-y(n)$. This is the denoised signal.

In the system shown in Fig. 1 the reference input is processed by an adaptive filter. An adaptive filter differs from a fixed filter in that it automatically adjusts its own impulse response. Thus with the proper algorithm, the filter can operate under changing conditions and can readjust itself continuously to minimize the error signal. The error signal used in an adaptive process depends on the nature of the application.

In noise cancelling systems the practical objective is to produce a system output $e(n)=s(n)+x_1(n)-y(n)$ that is a best fit in the least squares sense to the signal s. This objective is accomplished by feeding the system output back to the adaptive filter and adjusting the filter through an LMS adaptive algorithm to minimize total system output power.

In an adaptive noise cancelling system, in other words, the system output serves as the error signal for the adaptive process. It might seem that some prior knowledge of the signal s or of the noises x_1 and x would be necessary before the filter could be designed, or before it could adapt, to produce the noise cancelling s, x_1 and x signal y.

Assume that s, x_1 , x and y are statistically stationary and have zero means. Assume that s is uncorrelated with x_1 and x , and suppose that x is correlated with x_1 . The output e is

$e=s+x_1-y$. (1)

Squaring, one obtains

$e2=s2+(x_1-y)2+2s(x_1-y)$. (2)

Taking expectations of both sides of (2), and realizing that s is uncorrelated with x_1 and with y, yields

$E[e2]=E[s2]+E[(x_1-y)2]+2E[s(x_1-y)]$
$=E[s2]+E[(x_1-y)2]$ (3)

The signal power E [s2] will be unaffected as the filter is adjusted to minimize E[e2]. Accordingly, the minimum output power is

$mine[e2]=E[s2]+mine[(x_1-y)2]$ (4)

When the filter is adjusted so that E[e2] is minimized, $E[(x_1-y)2]$ is, therefore, also minimized. The filter output y is then a best least squares estimate of the primary noise no. Moreover, when $E[(x_1-y)2]$ is minimized[(e-s)2] is also minimized, since, from (1),

$(e-s)=(x_1-y)$. (5)

Adjusting or adapting the filter to minimize the total output power is thus tantamount to causing the output e to be a best least squares estimate of x_1 the signal s for the given structure and adjustability of the adaptive filter and for the given reference input. The output z will contain the signal s plus noise. From (l),the output noise is given by(x_1-y). Since minimizing E[e2] minimizes $E[(x_1-y)2]$ minimizing the total output power minimizes the output noise power. Since the signal in the output remains constant, minimizing the total output power maximizes the output signal-to-noise ratio.

ANC technique has been successfully applied to many applications, such as acoustic noise reduction, adaptive speech enhancement and channel equalization. In this paper use a simulink model in acoustic noise cancellation

III. LMS ALGORITHM:

The LMS algorithm is a widely used algorithm for adaptive filtering. The algorithm is described by the following equations:

$$y(n)=\sum_{i=0}^{M-1} w_i(n)*x(n-i); \qquad (1)$$

$$e(n)=d(n)-y(n) \qquad (2)$$

$$w_i(n+1)=w_i(n)+2ue(n)x(n-i); \qquad (3)$$

In these equations, the tap inputs x(n),x(n-1),……,x(n-M+1) form the elements of the reference signal x(n), where M-1 is the number of delay elements. d(n) denotes the primary input signal, e(n) denotes the error signal and constitutes the overall system output. wi(n) denotes the tap weight at the nth iteration. In equation (3), the tap weights update in accordance with the estimation error. And the scaling factor u is the step-size parameter u controls the stability and convergence speed of the LMS algorithm. The LMS algorithm is convergent in the mean square if and only if u satisfies the condition: 0 < u < 2 / tap-input power

where tap-input power $= \sum_{K=0}^{M-1} E[|u(n-k)^2|]$.

IV: SIMULATION AND RESULTS:

In this section we evaluate the performance of LMS algorithms in noise cancellation setup Fig. 1. Input signal is speech signal whereas Gaussian noise was used as noise signal. The LMS adaptive filter uses the reference signal and the desired signal, to automatically match the filter response. As it converges to the correct filter model, the filtered noise is subtracted and the error signal should contain only the original signal. The desired signal is composed of colored noise and an audio signal from a .wav file. The first input signal to the adaptive filter is white noise. This demo uses the adaptive filter to remove the noise from the signal output. When you run this demo, you hear both noise and a person recorded voice. Over time, the adaptive filter in the model filters out the noise so you only hear the recorded voice (Original signal).The two signals were added and subsequently fed into the simulation of LMS adaptive filter. The order of the filter was set to M = 40. The parameter μ is varied. Various outputs are obtained for various step size *i.e.* μ = 0.002, 0.04 system reaches steady state faster when the step size is larger. Fig.2. Original signal, Noisy signal and filter signal for LMS step size i.e.μ= 0.002.

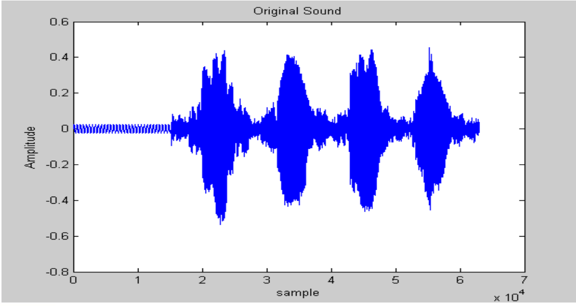

Figure:2 Original Signal

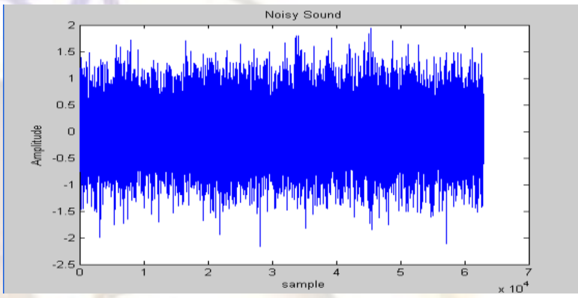

Figure:3 Noisy Signal

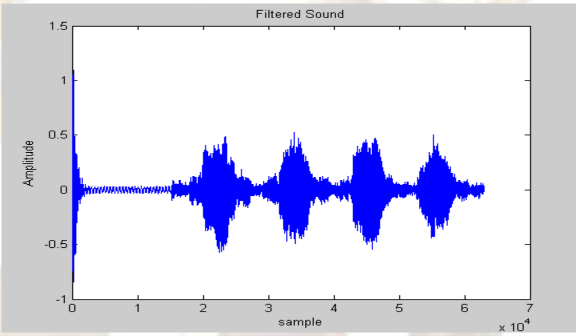

Figure: 3 Filter signal for LMS step size i.e μ=0.002.

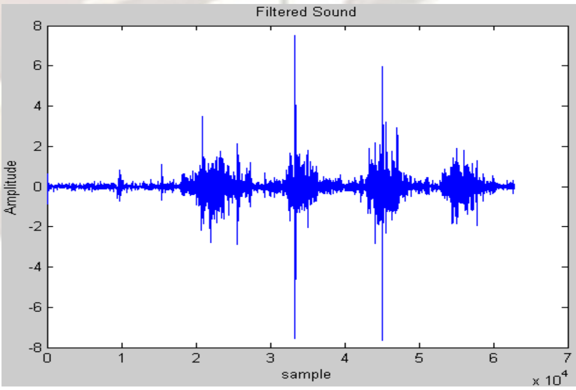

Figure: 4 Filter signal for LMS step size i.e.μ=0.04.

References :

1) Soni Changlani & M.K.Gupta, "Simulation of LMS Noise Canceller Using Simulink", International Journal On Emerging Technologies, 2011, ISSN : 0975-8364, pp 50-52.
2) Ying He, Hong He, LiLi, YiWu and Hongyan Pan"The Applications and Simulation of Adaptive Filter in Noise Cancelling." International conference on computer Science and Software Engineering-2008.
3) Ondracka J., Oravec R., Kadlec J., Cocherová E., "Simulation Of RLS And LMS Algorithms For Adaptive Noise Cancellation In MATLAB, Department Of Radio electronics FEI STU Bratislava, Slovak Republic UTLA, CAS Praha, Czech Republic.
4) Raj Kumar Thenu & S.K. Agarwal , "Hardware Implementation Of Adaptive Algorithms for Noise Cancellation", International Conference On Network Communication And Computer, 2011, pp 553-557.
5) Raj Kumar Thenu & S.K. Agarwal , "Simulation And Performance Analysis Of Adaptive Filter In Noise Cancellation", International Journal of Engineering science And Technology, Vol. 2(9), 2010, pp4373-4378.
6) Soni Changlani & Dr.M.K.Gupta, "The applications And Simulation of Adaptive Filter In Speech Enhancement", International Journal of Computer Engineering And ArchetectureVol. 1,No. 1, June 2011, pp95-101.
7) V.R.Vijaykumar, P.T.Vanathi & P. Kangasapabathy, "Modified Adaptive Filtering Algorithm For Noise Cancellation In Speech signals", Electronics And Electrical Engineering, Elektronika IR Elektrotechnika,ISSN1392-1215,No. 2(74),2007,pp17-20.
8) Lilatul Ferdouse, Nasrin Akhter, Tamanna Haque , Nipa3 and Fariha Tasmin Jaigirdar, "Simulation And Performance Analysis of Adaptive Filtering Algorithm In Noise Cancellation, International Journal Of Computer science Issues, Vol.8, Issue 1, January 2011, pp 185-192.
9) Mamta M.Mahajan & S.S.Godbole, "Design Of Least Mean Square AlgorithmFor Adaptive Noise Canceller", International Journal Of Advanced Engineering Science And Technologies,2011, Vol. No. 5, Issue No. 2,pp 172-176.