

# Design of Adaptive Noise Canceller using LMS Algorithm (using Chirp Signal as Input)

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**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 signal corrupted by an additive noise or interference. This method uses a primary input signal that contains a analog signal having random input 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.

## 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 signals, and the cancelling of broad-band interference in the side-lobes of an antenna array. In analog (chirp) signal

or speech signal, noise is very problematic because it will difficult to understanding of the information. 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. 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 signals can be modeled using following White noise.

ADAPTIVE FILTER :

Concept of adaptive filter

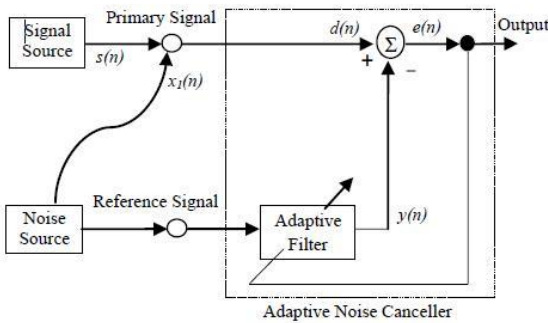


Fig.1. Adaptive Noise Cancellation System

S(n)- source signal  
 d(n)-primary signal  
 x1(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 random signal + 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

$$e^2=s^2+(x_1-y)^2+2s(x_1-y). \quad (2)$$

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

$$E[e^2]=E[s^2]+E[(x_1-y)^2]+2E[s(x_1-y)] \\ =E[s^2]+E[(x_1-y)^2] \quad (3)$$

The signal power  $E[s^2]$  will be unaffected as the filter is adjusted to minimize  $E[e^2]$ . Accordingly, the minimum output power is  $E[e^2] = E[s^2] + \text{min}[E[(x_1 - y)^2]]$  (4)

When the filter is adjusted so that  $E[e^2]$  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  $n$ . Moreover, when  $E[(x_1 - y)^2]$  is minimized  $E[(e - s)^2]$  is also minimized, since, from (1),

$$(e - s) = (x_1 - y). \tag{5}$$

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 (1), the output noise is given by  $(x_1 - y)$ . Since minimizing  $E[e^2]$  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.

LMS ALGORITHM:

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

$$M-1 \ y(n) = \sum w_i(n) * x(n-i); \tag{1}$$

$$I = 0$$

$$e(n) = d(n) - y(n) \tag{2}$$

$$w_i(n+1) = w_i(n) + 2\mu e(n)x(n-i) \tag{3}$$

In these equations, the tap inputs  $x(n), x(n-1), \dots, x(n-M+1)$  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.  $w_i(n)$  denotes the tap weight at the  $n$ th iteration. In equation (3), the tap weights update in accordance with the estimation error. And the scaling factor  $\mu$  is the step-size parameter  $\mu$  controls the

stability and convergence speed of the LMS algorithm. The LMS algorithm is convergent in the mean square if and only if  $\mu$  satisfies the condition:  $0 < \mu < 2 / \text{tap-input power}$

$$M-1$$

$$\text{where tap-input power} = \sum E[|u(n-k)|^2].$$

$$K=0$$

SIMULATION RESULTS:

In this section we evaluate the performance of LMS algorithms in noise cancellation setup Fig. 1. Input signal is chirp(analog) 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 first input signal to the adaptive filter is white noise. This demo uses the adaptive filter to remove the noise from the signal output. Over time, the adaptive filter in the model filters out the noise so you only hear the 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  $\mu$  is varied. Various outputs are obtained for various step size i.e.  $\mu = 0.007$ , system reaches steady state faster when the step size is larger. Fig.1. Original signal, Noisy signal and filter signal .

Figure:2 plot MSE

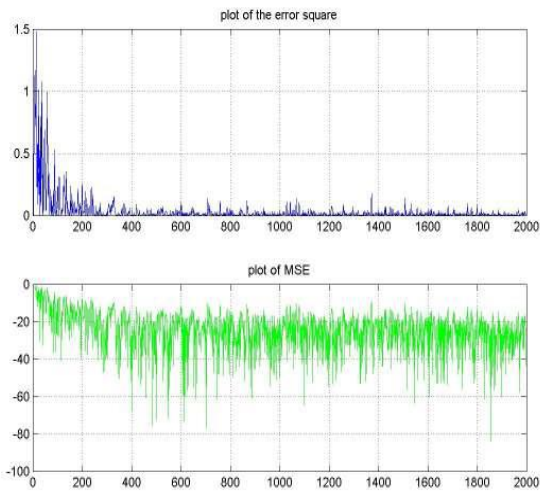


Figure:1 Original Signal

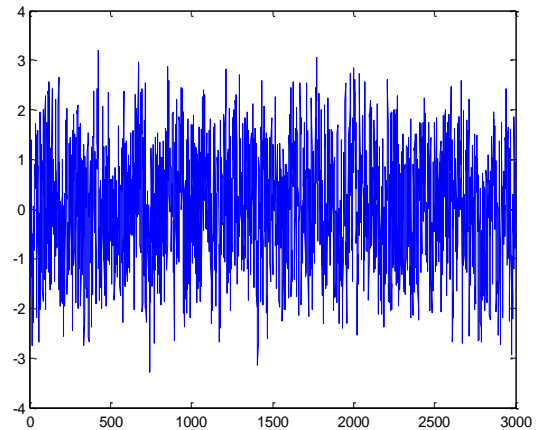
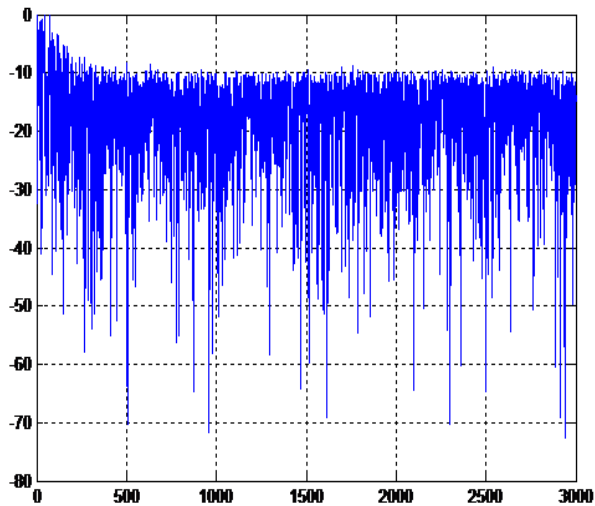


fig3:Time response of filtered signal



## REFERENCES:

- [1] Adaptive Signal Processing by John G Proakis, 3rd edition, Perntice Hall of India.
- [2] B. Widrow, "Adaptive noise canceling: principles and applications", Proceedings of the IEEE, vol. 63, pp. 1692-1716, 1975.
- [3] Adaptive Signal Processing by Bernard Widrow and Samuel D.Stearns; Pearson Education Asia, LPE.
- [4] Adaptive Filter Theory by Simen Haykin: 3rd edition, Pearson Education Asia,LPE.