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Research Article

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Use of Least Mean Square Algorithm in Eliminating Noise in Communication System

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Abstract Noise in communication systems has been a source of concern in Communication Engineering. This is due to the fact that it corrupts the original signal thereby altering the received signal. Consequently, the fidelity of the transmitted signal is low. Several methods are applicable in eliminating noise in communication, but the least mean square algorithm proves efficient due to its ability to estimate the noise in the communication system. The noise estimate from the LMS Algorithm is combined with the functionality of an adaptive filter to recover the original signal. The signal produced through such process is of high fidelity. That is the signal to noise ratio is usually high.

Keywords Adaptive Filter, Least Mean Square Algorithm, Signal to Noise Ratio (SNR), High Fidelity, Noise Canceller

Introduction

Communication is the process of transmitting intelligent signal from a source to a destination by means of an established link, between the source and the destination. The different communication equipments assembled together to establish the communication constitute the communication system. Below is a general block diagram for a communication system.

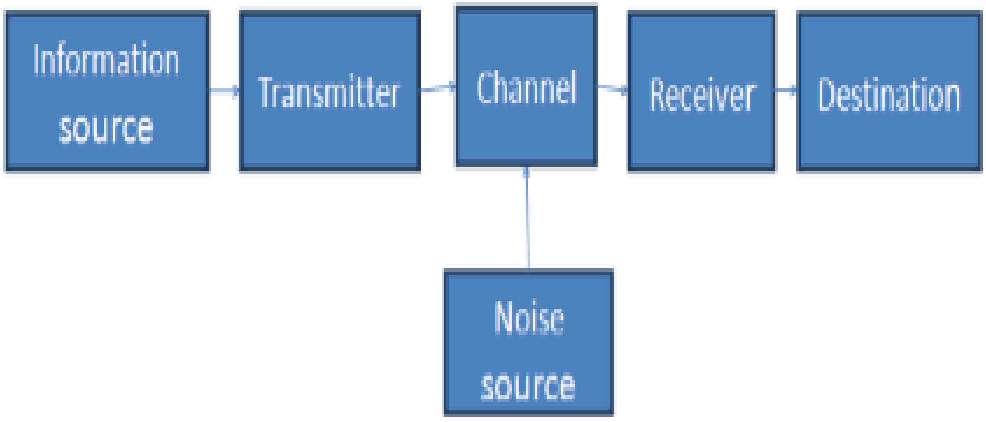

Figure 1: Block Diagram of a Communication System

1. *The Information Source* produces the required message signal to be transmitted. It selects a message from a set of possible messages to be transmitted to the receiving terminal [1].
2. *The Transmitter* converts the message, via modulation, into a form suitable for transmission over the communication channel. Modulation is the primary responsibility of the transmitter. In modulation, the message signal is superimposed upon the high frequency carrier signal.

3. *The Channel* provides a pathway between the output of the transmitter and the input of the receiver.
4. *Noise Source* introduces noise to the signal being transmitted, just before the signal gets to the receiver. The noise signal corrupts the original signal being transmitted. The information carried by the signal is degraded or lost in the presence of noise [2]. Consequently, the receiver misunderstands the signal received.
5. *The Receiver* reproduces the message signal from the corrupted signal via a process termed demodulation. Demodulation is the inverse process of modulation carried out in the transmitter.
6. *Destination* is the final block in the communication system which receives the message signal. It is the person or thing for which the message is intended.

The task of a communication system is to transfer a signal from some source to a remote user with a sufficient quality [3]. Noise and distortion are the main limiting factors in communication and measurement systems. Therefore, the modelling and removal of noise and distortion have been at the core of the theory and practice of communications and signal processing [4].

Noise cancellation is a variation of optimal filtering that includes creating an estimate of the noise by filtering the reference input, and then deducting this noise estimate from the primary input comprising both signal and noise [5]. To eliminate noise in communication system, an adaptive filter using Least Mean Square Algorithm will be suitable. Least Mean Square algorithm was first proposed by Widrow and Holf in 1960. It is one of the most common algorithms used to implement adaptive filtering [6]. Its operation is based on the adjustment of weight coefficients from sample to sample in such a way that the Mean Square Error is minimized. Least Mean Square algorithms are a class of adaptive filters used to mimic a desired by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal) [7]. It is simple to design, yet highly effective in performance and has made it popular in various applications [8].

The block diagram of a typical adaptive filter is shown below

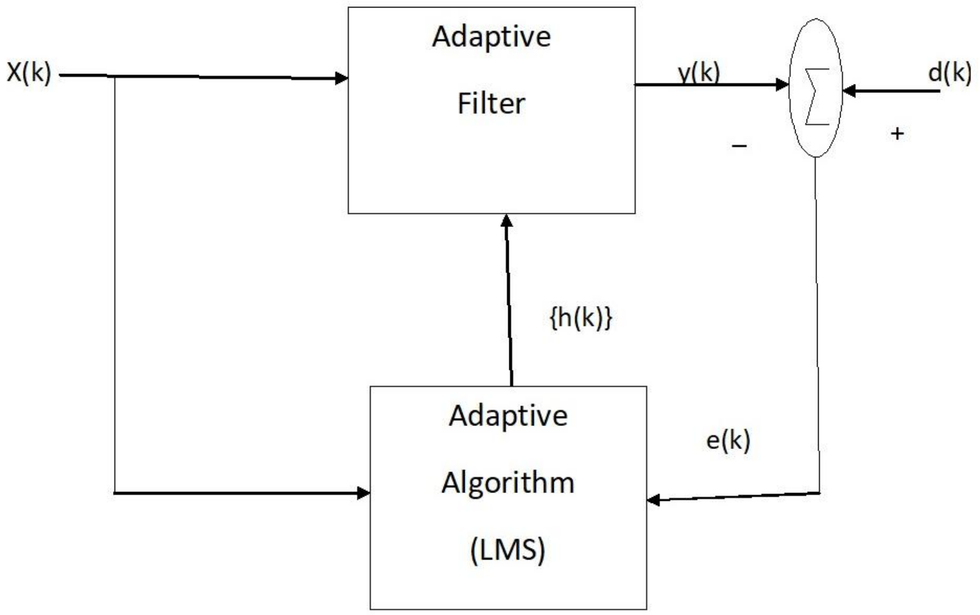

Figure 2: Block Diagram of Adaptive Filter

X(k) : Input Signal
Y(k) : Filtered Output
d(k) : Desired Response
h(k) : Impulse Response of Adaptive Filter

Methodology

The Least Mean Square (LMS) changes its filter coefficients based on the desired signal by finding the least mean square of the error signal , e(n), which is estimated by taking the difference between the desired signal, d(n), and the filtered signal y(n) [6].

To illustrate the concept of adaptive filtering of noise using Least Mean Square Algorithm, consider the operation of a simple noise canceller.

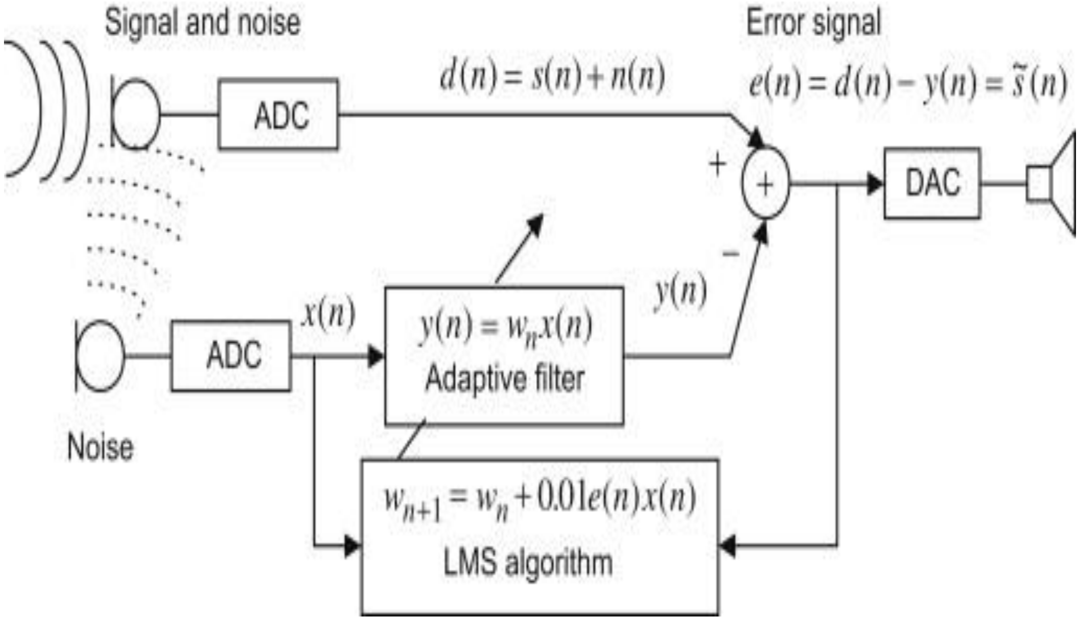

Figure 3: Noise Canceller Using a One Tap Adaptive Filter

The first microphone with Analogue to Digital Converter (ADC) is used to capture the desired speech s (n). The ADC produces a signal with noise due to the noisy environment.

That is d(n)=s(n)+n(n) -- (1)

The second microphone is placed where only noise is picked up, consequently the second Analogue to Digital Converter (ADC) channel captured noise x(n), which is fed to the adaptive filter .

The corrupting noise n(n) in the first channel is uncorrelated to the desired signal s(n), therefore separation between them is possible . The noise x(n) from the second channel is correlated to the corrupting noise n(n) in the first channel ,because both come from the same noise source .The noise signal x(n) is not correlated to the desired speech signal s(n).

Assume the corrupting noise in the first channel is linear filtered version of the second channel noise ; since it has a different physical path from the second channel noise ,and the noise source is time varying , in order to estimate the corrupting noise n(n) using an adaptive filter . The adaptive filter contains a digital filter with adjustable coefficients and the Least Mean Square Algorithm to modify the values of the coefficient(s) for filtering each sample. The adaptive filter then produces an estimate of noise y(n), which will be subtracted from the corrupted signal d(n)=s(n)+n(n). When the noise estimate y(n) equals or approximates the noise n(n) in the corrupted signal , that is

Y(n) approximately equals n(n),

The error free signal e(n) is given as

e(n)=s(n)+n(n)-y(n)=s~(n)---------------------------------(2)

From equation (2) it is seen that the clear speech s(n) is approximated , hence the noise is eliminated.

Conclusion

The Least Mean Square (LMS) Algorithm used as an adaptive filter improves the Signal to Noise Ratio (SNR) of the received signal by generating an estimated version of the noise to counter the noise. A perfect signal may not be received, but certainly a high fidelity signal is received.

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