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Review of methods of Adaptive Noise Cancellation Using LMS and NLMS Algorithms

Communication

KEYWORDS : Adaptive filter, ANC (Adaptive noise canceller), Least-mean square (LMS), NLMS (Normalised least mean square)

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ABSTRACT

The main goal of this paper is noise cancellation using adaptive filter. 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 which changes continuously like speech and corrupted by an additive noise or interference. The reference input is adaptively filtered and subtracted from the primary input signal to obtain the estimated signal. This paper studies the desired signal corrupted by an additive noise can be recovered by an adaptive noise canceller using adaptive filter and compare the algorithms which are LMS (least mean square) and NLMS (Normalised least mean square) for changing its coefficients according to the change in the signal

I. Introduction

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 [10]. Noise signal is reduced directly comparing with the original signal and reference signal using fixed filters. However, such an approach not work when the signal is changing ,where we use adaptive filters .Adaptive filter [6] compare the signal and adaptively changes the coefficients according to the change of the signal change .The term known as ANC(adaptive noise cancellation)[5].

Figure 1Adaptive Noise Canceller

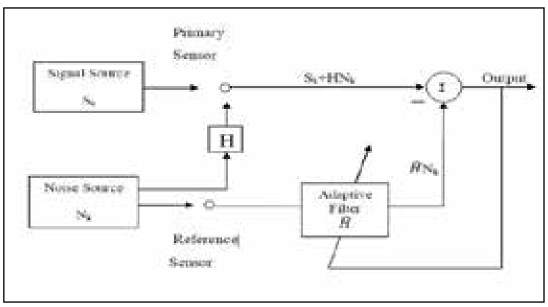


Figure 3. NLMS filter ANC

In this figure 1, the signal path from the noise source is passed to the primary signal as an unknown FIR channel H. The adaptive filter to the noise recorded at the reference signal, and then an adaptive algorithm is used to retain the adaptive filter to match or estimate the characteristics of the unknown channel H.

The adaptive algorithm, the refining procedure is designed to form the coefficients according to the change of the signal, based on some criteria, and the most popular criteria are the minimum mean square error (MMSE) and least square error. Such adaptive algorithms are the least mean square (LMS) algorithm[6], which was invented by Widrow and Hoff in 1960 and NLMS (Normalised least mean square) .

II .LMS of adaptive noise cancellation[5]

ANC having the dual-input and closed-loop adaptive feedback system. An Adaptive Noise cancellation utilizes a reference input, ideally containing just noise, which is passed through an adaptive filter and later subtracted from a primary input containing both the desired signal and components of the noise present in the reference input. The output becomes the primary signal with the noise attenuated or cancelled altogether. Adaptive filters are those with the ability to adjust their own parameters automatically with little or no previous information about the signal to be cleaned or noise to be cancelled. The correlation between the noise present in the reference channel and the primary input is important, the higher it is, the better the cancellation.[2] A diagram of the concept is pictured in Figure 2

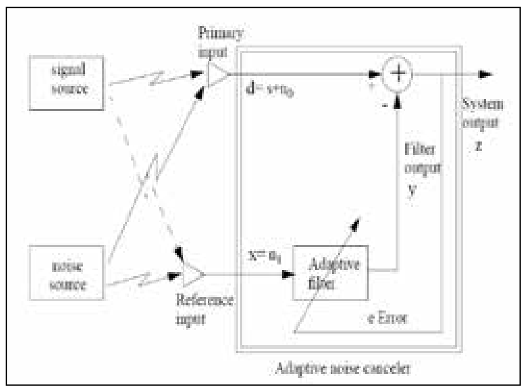


Figure 2: Block Diagram LMS of Adaptive Noise Canceller

In Figure 2, a primary source 's', that contains uncorrelated noise, n0, is transmitted to the upper left channel. The bottom left channel receives noise, n1, correlated with n1 in some unknown manner. This is the reference input. It is fed into the adaptive filter with the goal of replicating n1. As the noise characteristics are assumed to be unknown, an adaptive filter is necessary. The output of the adaptive filter is subtracted from the primary input, source plus n0, to produce z, which also serves as the error signal to the filter.
The output of the system, z, is define z = s + n0 - y. That is a best fit in the least squares sense to the signals. 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 n0 and n1 would be necessary before the filter could be designed, or before it could adapt, to produce the noise cancelling signal y. Assume that s, n0, n1, y are statistically stationary and have zero means. Assume that is uncorrelated with n0 and n1, and suppose that n1 is correlated with n0. The output

$z = s + n_0 - y$ (1)

Squaring, one obtains

$z^2 = s^2 + (n_0 - y)^2 + 2s(n_0 - y)$ (2)

Taking expectations of both sides and realizing that s is uncorrelated with n0 and with y, yields E[z2]=E[s2]+E[(n0-y)2]+2E[s(n0-y)]=E[s2]+E[(n0-y)2] (3)

The signal power E [s2] will be unaffected as the filter is adjusted to minimize E[z2]. Accordingly, the minimum output power is

min $E[z^2]=E[s^2]+\min E[(n_0-y)^2]$ (4)

Thus When the filter is adjusted so that $E[z^2]$ is minimized, $E[(no -y)^2]$ is, therefore, also minimized. The filter output y is then a best least squares estimate of the primary noise no. Moreover, when $E[(n_0 - y)^2]$ is minimized$[(z -s)^2]$ is also minimized,

$(z - s) = (n_0 - y)$. (5)

Adjusting or adapting the filter to minimize the total output power is thus tantamount to causing the output z to be a best least squares estimate of 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. The output noise is given by $(n_0 -y)$. Since minimizing $E[z^2]$ minimizes $E[(no - y)^2]$ 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.

III. Implementation of the LMS algorithm

The LMS algorithm [4] is extremely simple since it minimizes the instantaneous square error instead of the mean square error, using a simple gradient-based optimization method[1].

Each iteration of the LMS algorithm requires the following distinct steps in the given order[3,8] :-

1. Initialy, set each weight Wn (i), where i= 0,1,..,N-1, to an arbitary fixed value such as 0.
2. Compute filter output :-

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

3. **Compute the error estimate :-**

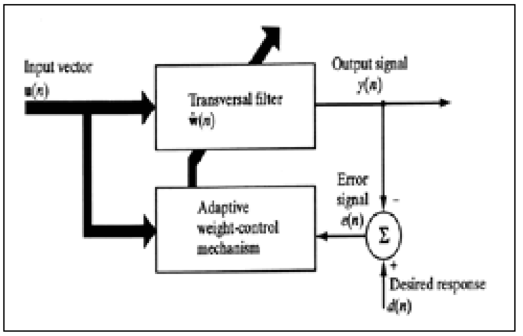
$$e(n)=d(n)-y(n) \quad (7)$$

4. **Update the next filter weights :-**

$$w(n+1)=w(n)+2\mu e(n)x(n) \quad (8)$$

IV. NLMS (Normalised least mean square)

The Normalized Least Mean Square (NLMS) [7] algorithm is a modified form of Least Mean Square(LMS) algorithm . In the standard LMS algorithm, when the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. The Lagrangian methodology [1] has been used in order to proposed a nonlinear adaptation rule defined in terms of the product of differential inputs and errors which means a generalized as the normalized (N)LMS algorithm. The NLMS filter is preferred over LMS algorithm because of its particular characteristics of faster convergence, besides a lower mean squared error . The NLMS filter as an adaptive noise canceller is shown in figure .3.



We may view the NLMS algorithm as a time-varying step-size algorithm, calculating the convergence factor μ

$$\hat{\mu}(n)=\frac{\tilde{\mu}}{\left|u(\upsilon)^2\right|} \quad (9)$$

The Filter weights are updated by the [9]

$$\hat{w}(n+1)=\hat{w}(n)+\frac{\tilde{\mu}}{\left\|u(\upsilon)\right\|^2}u(n)e*(n) \quad (10)$$

V .CONCLUSION

Adaptive noise cancellation using LMS and the modified form of LMS i.e.NLMS have been studied and analysed on the basis of their merits and demerits.

REFERENCE

[1] S. Haykin, Adaptive Filter Theory. Englewood Cliffs, NJ:Prentice- Hall, 1986. | [2] B. Widrow, Adaptive filters 1: Fundamentals,|| Stanford Electronics Lab., Stanford Univ., Rep. SUSEL-66-126, Dec. 1966 | [3] Simon Haykin, "Adaptive Filter Theory", Prentice Hall, 4th edition | | | [4] Adaptive Signal Processing by John G Proakis, 3rd edition, Perntice Hall of India. | | [5] G.Amjad Khan1, Dr. K.E. Sreenivasa Murthy2 ,J.Satyanarayana3 "Noise cancellation in Speech Signals by Using a Constrained Stability LMS Algorithm" International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 Vol. 2, Issue 1, Jan-Feb 2012, pp. 426-430. | | [6] Jafar Ramadhan Mohammed1, Muhammad Safder Shafi2, Sahar Imtiaz2, Rafay Iqbal Ansari2, and Mansoor Khan2 "An Efficient Adaptive Noise Cancellation Scheme Using ALE and NLMS Filters" International Journal of Electrical and Computer Engineering (IJECE) Vol.2, No.3, June 2012, pp. 325~332 ISSN: 2088-8708 _325. | | [7] Pranjali M. Awachat, S.S.Godbole "A Design Approach For Noise Cancellation In Adaptive LMS Predictor Using MATLAB" International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 www.ijera.com Vol. 2, Issue4, July-august 2012, pp.2388-2391 2388. | | [8] Mamta M. Mahajan e al. " Design of Least Mean Square Algorithm for Adaptive Noise Canceller" INTERNATIONAL JOURNAL OF ADVANCED ENGINEERING SCIENCES AND TECHNOLOGIES Vol No. 5, Issue No. 2,2012 172 –176. | | [9] Cristina Gabriela SĂRĂCIN1, Marin SĂRĂCIN2, Mihai DASCĂLU3, Ana-Maria LEPAR4 "ECHO CANCELLATION USING THE LMS ALGORITHM" U.P.B. Sci. Bull., Series C, Vol. 71, Iss. 4, 2009 ISSN 1454-234x. | | [10] Jashvir Chhikara and Jagbir Singh "Noise cancellation using adaptive algorithms" International Journal of Modern Engineering Research (IJMER) Volume 50, Nov 4,2009. | | [11] RAJ KUMAR THENUA AND S.K. AGARWAL"SIMULATION AND PERFORMANCE ANALAYSIS OF ADAPTIVE FILTER IN NOISE CANCELLATION" International Journal of Engineering Science and Technology Vol. 2(9), 2010, 4373-437. |