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Implementation of Adaptive Noise Cancellation using LMS and NLMS Algorithm

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Abstract— ***The main goal of this paper is to present the simulation scheme for noise cancellation by using an adaptive filter using as LMS (Least mean square) & NLMS (Normalized Least Mean Square) adaptive algorithm. The main objective of the noise cancellation is to remove the noise from original input signal plus noise signal and hence to obtain the noise free signal. Use of Adaptive filter may filter out or suppressed noise from noisy signal getting the clean speech .This process is known as the adaptive noise cancellation. Noise corrupted speech is taken from standard noises database, these noise signal having different SNR .The SNR at the output is improved by both algorithms. This adaptive noise canceller is useful to improve the S/N ratio and reduces MSE. Here we validate the results for adaptive filter using soft computing tool i.e. MATLAB. At the end of communication system information signal may be introduced with some noise along with the information signal. Noise may additive or convoluted. Adaptive filter finds better solution to deal this problem. The problem of controlling the noise level in the environment has been the focus tremendous amount of research over the years. This describes a study of techniques for noise cancellation which can be applied at the input to standard receivers trained on noise-free speech. Such methods have been used in various applications, including communication systems, biomedical engineering, and industrial applications.***

Keywords— ***Adaptive noise cancellation, least mean Square (LMS), Mean Square Error (MSE), normalized least mean Square (NLMS), Signal to noise ratio (SNR).***

I. INTRODUCTION

In all practical situations, the received speech waveform contains some form of noise component. The noise may be a result of the finite precision involved in coding the transmitted waveform (quantization noise), or due to the addition of acoustically coupled background noise. Depending on the amount and type of noise, the quality of the received waveform can range from being slightly degraded to being annoying to listen to, and finally to being totally unintelligible. The problem of removing the unwanted noise component from a received signal has been the subject of numerous investigations. The pioneering work of Wiener and others give an optimum approach for deriving a filter that tends to suppress the noise while leaving the desired signal relatively unchanged. The design of these filters requires that the signal and the noise be stationary and that the statistics of both signals be known a priori. In practice, these conditions are rarely met.

The classical approach to noise cancellation is a passive acoustic approach. Passive silencing techniques such as sound absorption and isolation are inherently stable and effective over a broad range of frequencies. However, these tend to be expensive, bulky and generally ineffective for cancelling noise at the lower frequencies. The performance of this system is also limited to a fixed structure and proves impractical in a number of situations where space is at a premium and the added bulk can be a hindrance. The shortcomings of the passive noise reduction methods have given impetus to the research and applications of alternate methods of controlling noise in the environment. Various signal processing techniques have been proposed over the year for noise reduction in the environment. The explosive growth of digital processing algorithm and technologies has given an impetus to the application of these techniques to the real world. Speech enhancement aims to improve speech quality by using various algorithms. It may sound Simple, but quality means a one of the following.

- Clarity and intelligibility,
- Pleasantness, or
- Compatibility with some other method in speech processing.

Intelligibility and pleasantness are difficult to measure by any mathematical algorithm. Usually listening tests are employed. However, since arranging listening tests may be expensive, it has been widely studied how to predict the results of listening tests. No single philosopher's stone or minimization criterion has been discovered so far and hardly ever will [1] [2].The central methods for enhancing speech are the removal of background noise, echo suppression and the process of artificially bringing certain frequencies into the speech signal. Enhancing of speech degrades by noise, or noise reduction, is most important. This project deals with the removal of background noise. Many DSP Algorithms has developed a large family of noise reduction and speech enhancement products suitable for most applications from speaker-phones to audio conferencing, and from motorbike communication to Internet phones and car kits. On application basis two categories of algorithms can be recognized when it comes to speech enhancement. The first category uses multiple microphones for recording (adaptive microphone arrays) while the second uses only a single microphone, and therefore, can also be used to clean up existing recordings.[3]Other algorithms of speech enhancement for noise reduction on basis of method can be categorized into three fundamental classes: filtering techniques, spectral restoration, and model-based methods. [2]

- Filtering Techniques
- Spectral Subtraction Method
- Wiener Filtering
- Signal subspace approach (SSA)
- Spectral Restoration
- Minimum Mean-Square-Error Short-Time Spectral Amplitude Estimator (MMSE-STSA)
- Speech-Model-Based

II. AN ADAPTIVE FILTERING PROCESS

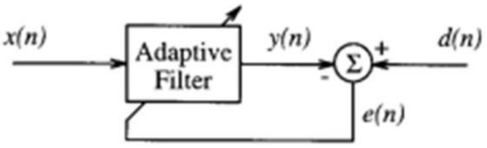

FIGURE 1: THE GENERAL ADAPTIVE FILTERING PROBLEM

Figure 3.1 shows a block diagram in which a sample from a digital input signal x (n) is fed into a device, called an adaptive filter, that computes a corresponding output signal sample y(n) at time n. For the moment, the structure of the adaptive filter is not important, except for the fact that it contains adjustable parameters whose values affect how y (n) is computed. The output signal is compared to a second signal d (n), called the desired response signal, by subtracting the two samples at time n. This difference signal, given by

$$e(n) = d(n) - y(n) \qquad \text{….. (1)}$$

is known as the error signal. The error signal is fed into a procedure which alters or adapts the parameters of the filter from time n to time (n + 1) in a well-defined manner. This process of adaptation is represented by the oblique arrow that pierces the adaptive filter block in the figure. As the time index n is incremented, it is hoped that the output of the adaptive filter becomes a better and better match to the desired response signal through this adaptation process, such that the magnitude of e (n) decreases over time. In this context, what is meant by "better" is specified by the form of the adaptive algorithm used to adjust the parameters of the adaptive filter. In the adaptive filtering task, adaptation refers to the method by which the parameters of the system are changed from time index n to time index (n + 1). The number and types of parameters within this system depend on the computational structure chosen for the system. We now discuss different filter structures that have been proven useful for adaptive filtering tasks [14].

III. FILTER STRUCTURE

An adaptive filter is a self-designing and time-varying system that uses a recursive algorithm continuously to adjust its tap weights for operation in an unknown environment [1 and 14]. Figure 3.1 shows a typical structure of the adaptive filter, which consists of two basic functional blocks: (i) a digital filter to perform the desired filtering and (ii) an adaptive algorithm to adjust the tap eights of the filter. The digital filter computes the output y (n) in response to the input signal u (n), and generates an error signal e (n) by comparing y (n) with the desired response d (n), which is also called the reference signal, as shown in Figure 3.1. The performance feedback signal e (n) (also called the error signal) is used by the adaptive algorithm to adjust the tap weights of the digital filter. The digital filter shown in Figure 3.1 can be realized using many different structures. The commonly used transversal or finite impulse response (FIR) filter is shown in figure below

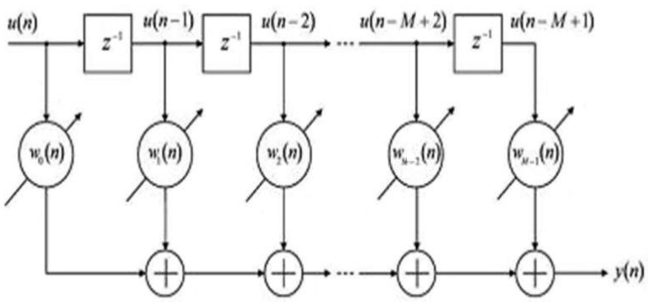

Figure 2: An M-tap adaptive transversal filter

m = 0,1,...,M-1, indicated by circles with arrows through them, are the filter tap weights at time instance n and M is the filter length. These time-varying tap weights form an M ×1 weight vector expressed as

$$w(n) \equiv [w_0(n), w_1(n) \dots, w_{M-1}(n)]^T \quad \dots (2)$$

Where, the superscript T denotes the transpose operation of the matrix. Similarly, the input signal samples, u (n-m), m = 0,1,...,M-1, form an M ×1 input vector

$$u(n) \equiv [u(n), u(n-1), \dots, u[u-M+1)]^T \quad \dots (3)$$

With these vectors, the output signal y (n) of the adaptive FIR filter can be computed as the inner product of **w** (n) and **u** (n), expressed as

$$y(n) = \sum_{m=0}^{M-1} w_m(n)u(n-m) = w^T(n)u(n) \quad \dots (4)$$

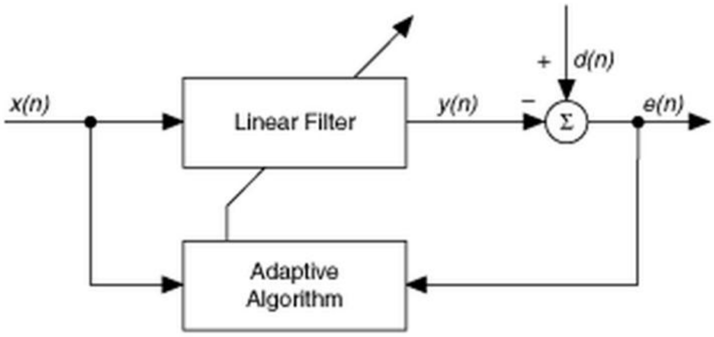

Fig.3 Adaptive Noise Cancellation Process

IV. LEAST MEAN SQUARE(LMS) ALGORITHM

Algorithm is a type of adaptive filter known as stochastic gradient-based algorithms as it utilises the gradient vector of the filter tap weights to converge on the optimal wiener solution. It is well known and widely used due to its computational simplicity. It is this simplicity that has made it the benchmark against which all other adaptive filtering algorithms are judged. LMS algorithm is a type of adaptive filter known as stochastic gradient- based algorithms as it utilises the gradient vector of the filter tap weights to converge on the optimal wiener solution. It is well known and widely used due to its computational simplicity. It is this simplicity that has made it the benchmark against which all other adaptive filtering algorithms are judged.

With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula.

W(n+1)=w(n)+2μe(n) x(n)…………(5)

Here x(n) is the input vector of time delayed input values, x(n) = [x(n) x(n-1) x(n-2) .. x(n-N+1)]T. The vector w(n) = [w0(n) w1(n) w2(n) .. wN-1(n)]T represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output diverges.

Implementation of the LMS Algorithm:

Each iteration of the LMS algorithm requires 3 distinct steps in this order:

1. The output of the FIR filter, y(n) is calculated using equation 2.

   Y(n)=$\sum_{i=0}^{N-1} w(n)\, x(n-1) = wt(n)x(n) \dots \dots \dots . (6)$
2. The value of the error estimation is calculated using equation 3.

   e(n)=d(n)-y(n) …………………………...…..(7)
3. The tap weights of the FIR vector are updated in preparation for the next iteration, by equation 4.

   W(n+1)=w(n)+2μe(n) x(n)…………… (8)

The main reason for the LMS algorithms popularity in adaptive filtering is its computational simplicity, making it easier to implement than all other commonly used adaptive algorithms. For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplications (N for calculating the output, y(n), one for 2μe(n) and an additional N for the scalar by vector multiplication.

V. NORMALIZED LEAST MEAN SQUARE (NLMS) ALGORITHM

One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. 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 normalized 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 following 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.

$Tr[R] = \sum_{i=0}^{N-1} E[x2(n-1)]$

$= E[\sum_{i=0}^{N-1} x2(n-i) \ldots\ldots\ldots\ldots\ldots (9)$

The recursion formula for the NLMS algorithm is stated in equation 6.

$W(n+1) = w(n) + 1/x^T(n)\, x(n)\quad e(n)\, x(n) \ldots\ldots\ldots\ldots (10)$

Implementation of the NLMS algorithm

The NLMS algorithm has been implemented in Matlab. As the step size parameter is chosen based on the current input values, the NLMS algorithm shows far greater stability with unknown signals.

Convergence speed and relative computational simplicity make the NLMS algorithm ideal for the real time adaptive echo cancellation system

As the NLMS is an extension of the standard LMS algorithm, the NLMS algorithms practical implementation is very similar to that of the LMS algorithm. Each iteration of the NLMS algorithm requires these steps in the following order.

1. The output of the adaptive filter is calculated.

   $$Y(n) = \sum_{i=0}^{N-1} w(n)\, x(n-i) = w^T(n)\, x(n) \ldots\ldots\ldots\ldots\ldots (11)$$

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

   $$e(n) = d(n) - y(n) \ldots\ldots\ldots\ldots\ldots (12)$$

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

   $$\mu(n) = 1/x^T(n)\, x(n) \ldots\ldots\ldots\ldots\ldots (13)$$

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

   $$W(n+1) = w(n) + \mu(n)\, e(n)\, x(n) \ldots\ldots\ldots\ldots\ldots (14)$$

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm. This is an acceptable increase considering the gains in stability and echo attenuation achieved.

4. Performance Parameter

The method discussed in the earlier chapters has been implemented quantitatively and qualitatively. In this chapter, based on the above method, speech parameters are calculated. In the above discussed method the speech enhancement is performed using Adaptive filter and filtering algorithms. The quality of the enhanced speech is evaluated on the basis of the following parameters.

1. MSE

The Mean Squared Error (MSE) measured in dB is one of the ways to determine the difference between values implied by an estimator and the true values of the quantity being estimated. MSE corresponds to the expected value of the squared error loss. MSE measures the average of the squares of the "errors". The difference between the values implied by estimator and the quantity which is to be estimated gives us the error value of the signal. The randomness of signal or the inability of the system to produce accurate values at the output gives rise to error in the signal.

2. SNR

Signal-to-noise ratio is a measure that compares the level of a desired signal to the level of background noise. Signal-to-noise ratio is defined as the power ratio between a signal and the background noise .It is measured in dB and is represented as SNR

VI. EXPERIMENTAL RESULT

LMS and NLMS algorithm for the adaptive filtering process is implemented by using software computing tool (MATLAB 7.10.0) .Input to the adaptive filter is the wave file of two second having sampling rate 16KHz .Different wave files of the different SNR is applied as the input and Improved SNR to that wave file get .The performance of the Adaptive algorithm is evaluated on the basis of SNR and MSE.

Table I Experimental result by NLMS implementation

			NLMS Implementation(0.1)	
Sr. No.	**Noises**	Level	SNR	MSE
2	Airport	sn0	17.98	0.00002239
3		sn5	19.123	0.00001877
4		sn10	20.531	0.00001346
5		sn15	20.5052	0.00001347
6	Babble	sn0	17.22	0.00002742

7		sn5	19.61	0.00001664
8		sn10	19.688	0.00001602
9		sn15	20.153	0.00001458
10	Car	sn0	19.493	0.00001712
11		sn5	19.786	0.00001580
12		sn10	19.764	0.00001576
13		sn15	20.166	0.00001447
14	exhibition	sn0	19.07	0.00001926
15		sn5	19.92	0.00001529
16		sn10	20.22	0.00001433
17		sn15	20.188	0.00001440
18	Restaurant	sn0	18.853	0.00001895
19		sn5	19.5471	0.00001675
20		sn10	20.211	0.00001428
21		sn15	20.15	0.00001467
22	Station	sn0	19.55	0.00001709
23		sn5	20.13	0.00001476
25		sn15	20.25	0.00001438
26	Street	sn0	19.30	0.00001808
27		sn5	19.65	0.00001621
29		sn15	20.32	0.00001413
30	Train	sn0	18.56	0.00002065
31		sn5	19.40	0.00001688
32		sn10	20.24	0.00001427

VII. CONCLUSION

The Signal to noise ratio (SNR) and Mean square error (MSE) of the filtered signal is obtained .The resulting values of the SNR and MSE is found to be better for Adaptive filter using NLMS algorithm compared with the LMS algorithm.

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