IMPLEMENTATION OF NOISE CANCELLATION FROM AUDIO SIGNAL USING LMS ALGORITHM

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ABSTRACT

This paper presents a approach for noise reduction or cancellation of noise through Least Mean Square algorithm. Filtering method is used for this approach is adaptive filtering. when the signal introduced by noise it get distorted and disturbed by the noise and hence reduced audio quality. Here to overcome this problem we use this filtering technique. This technique helps to improve audio quality and hence minimize losses at the end of result. The qualitative result is based on human hearing acuity of frequency of signal and in this various real world audio signal sources with noise are also discussed.

Keywords: Frequency, Noise Cancellation, Adaptive filtering, Least Mean Square Algorithm (LMS)

I INTRODUCTION

Acoustic noise becomes more utter as increases in number of industrial equipment such as transformer, blowers and engines that are in use. In this paper, noise is define as any kind of undesirable signal whether it is done by electrical vibrations or any kind of media. signal are superimpose with each other and create an effect with noise.

The main objective of this paper is to build a noise cancelling of audio signal by adaptive filters and LMS algorithm. Basically noise cancellation is a technique to reduce the undesirable noise from the original signal. An adaptive filter is use for filtration of noise from any audio signal using feedback. In many applications, adaptive noise cancellation is an effective method for recovering a signal corrupted by additional noise.

II BASIC CONCEPT OF NOISE

In general term, noise is simply an unwanted signal or sound. Noise in the sound means any variations in air pressure that is detected by human ear. Signal that is transmitted over the media is combination of both useful and unwanted signal. Let's take an example by diagram in which original signal and unwanted noise signal are superimpose with each other and create an effect with noise.

Input Signal

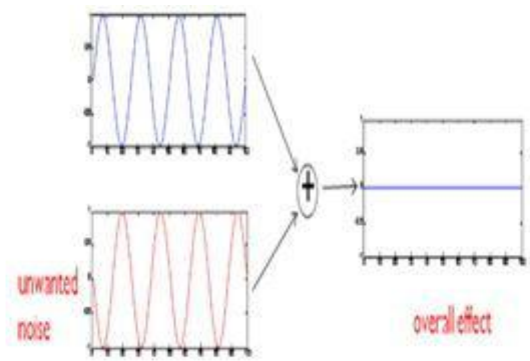

Figure 1: Basic Noise Concept

Noise is generally unwanted sound. Let's Take an example for understanding its concern with human being. Suppose if anything falls at the floor, does it make a sound if no one is there? No, noise is defined as unwanted signal, and if there is no observing the sound can't be unwanted. Therefore can be no noise.

In the diagram there is input signal with up and down curves. When the noise signal is superimposed with the input signal the output of signal become straight line means the input signal get distorted.

III ADAPTIVE FILTER

In this practical world the unknown frequency response of signal requires filtering and for this purpose adaptive filters are used. Now adaptive filter is used because it automatically design itself by providing feedback loop and can detect variations in time according to an algorithm which is derived at the occurrence of the error signal. The algorithm that we use in this paper is LMS.

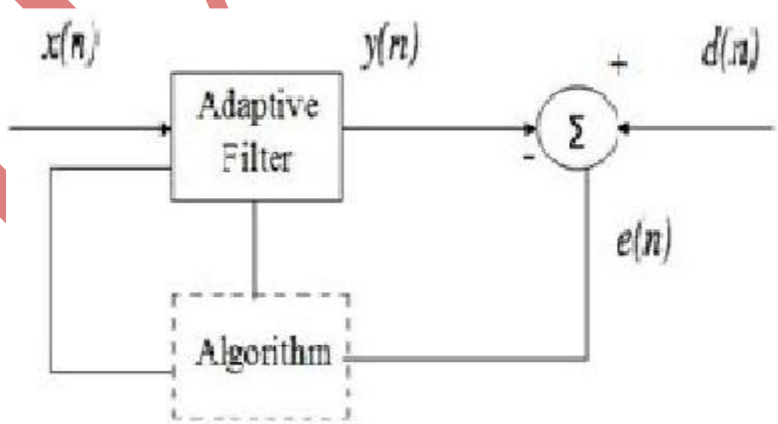

Figure 2: Block diagram of Adaptive filter

x (n)=input signal y (n)=output signal

d (n)=desire reference response e (n)=error signal

e(n)=y(n)-d(n)

y(n) is concluded by the convolution of the filter weight vector and input signal.

ie y(n)=w(n)*x(n)

Here w is the filter weight vector

The dotted lines show the algorithm is being implemented here in this circuit. The basic elements of adaption are filter structure, performance criterion and adaption rule. Adaptive filter are used in many application such as echo cancellation, adaptive equalization, speech coding.

Echo cancellation: when the characteristics of medium changes in which wave is propagating, there occur a repetition of waveform due to reflection, this is called Echo. In communication echo can degrade the quality of service. Hence to remove these adaptive filters are used. Echo cancellation is basically used in telephony to improve quality.

Speech coding: it means compression of data. it is a method of reducing the information that is used to represent the signal. The reduction is done to send the efficient data over a link. For this purpose adaptive filters are used.

Adaptive equalization: when the signal is transmitted over a channel, it may get distorted, to compensate the distortion adaptive equalization is being processed with the use of the adaptive filter.

IV LEAST MEAN SQUARE ALGORITHM

LMS algorithm is a noise cancellation algorithm and was developed by Bernard Widrow, a professor at Stanford University in 1960.His first PhD student Ted Hoff gave equal contribution in his work. It uses continuous adaption process to remove the noise efficiently and precisely from the system by using updated weight

Vector (W) in all iteration till error signal becomes zero.

4.1 Noise Cancellation Concept

Noise canceller is used to recover the original signal from the noisy signal. The original signal is obtained by doing number of iteration of LMS algorithm on the noisy signal.

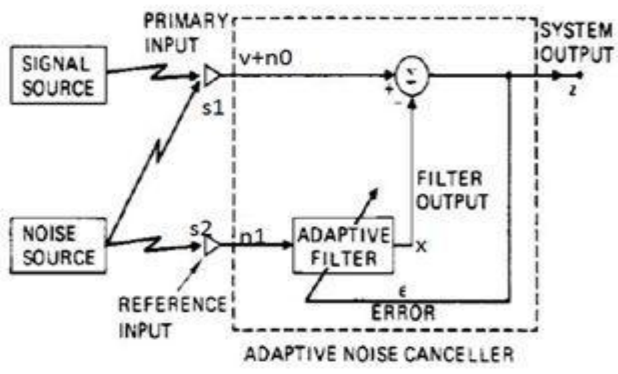

Figure 3: Adaptive Noise Cancellation Concept

Basic equation of LMS algorithm:

W(n+1)=W(n)+µu(n)e*(n) Where µ=step size parameter u(n)e*(n)=adjustment applied to the vector
In the first iteration weight vector is zero. Because there is no feedback that is W(0)=0
Second iteration is W(1)=W(0)+µu(n)*error signal. The iteration continuously goes on till error equals to zero.

The input signal v is transmitted over a channel and at the same channel, the sensor S1 receives noise which is mixed with the original signal. That is signal is corrupted. Suppose the combination of we an input and the noisy signal received at the sensor S1 is the primary input (v+n0) that is fed to the Canceller circuit. The second sensor S2 is introduced with noise n1 which has no concern with the original signal and this is in correlation with the noise n0. The output of S2 provides reference input to the noise canceller circuit. Then this noise is filtered by the adaptive filter. Suppose the output of the adaptive filter is x. The output of the adaptive filter is subtracted from the primary input to produce system output z.

z=v+n0-x

If there is still any error in the signal then filtering mechanism by the adjustment of the weight vector (LMS algorithm) continuously goes on till the error becomes zero.

V EXPERIMENTAL RESULT

In this section we performed the LMS algorithm for the filtration of the noise. The algorithm was implemented step by step as discussed earlier. Figure 4 shows the original signal .in this signal no noise is present. Figure 5 shows the noise introduced in the signal. Which distorted the original signal? A figure 6 show the signal where whole algorithm is applied i.e. output of a signal, which may or may not contain any noise signal. Figure 7 shows the error efficiency. Means how much error is reduced at every iteration. Figure 8 shows the convergence line. Means the graph of the weightage value added at every iteration.

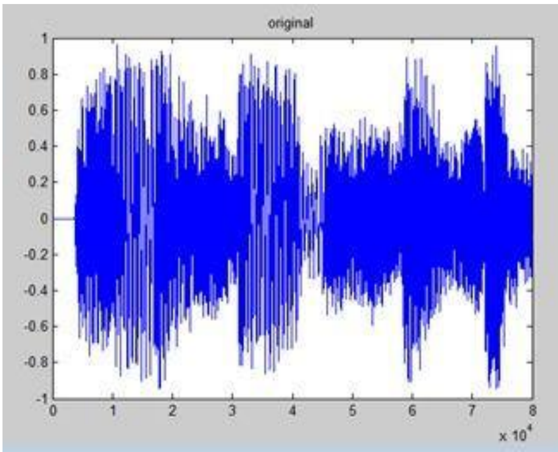

Figure 4: Original Signal

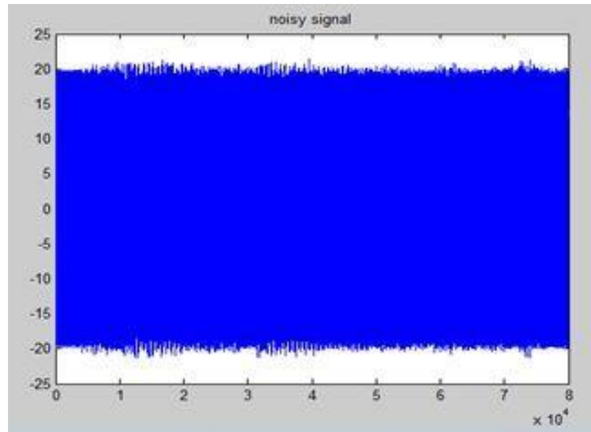

Figure 5: Noisy Signal

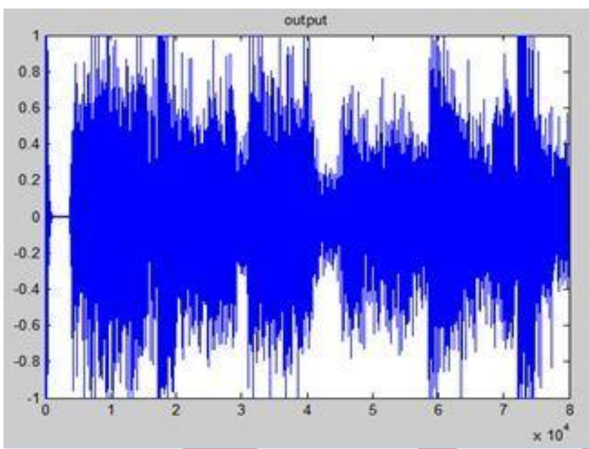

Figure 6: Filtered Output Signal

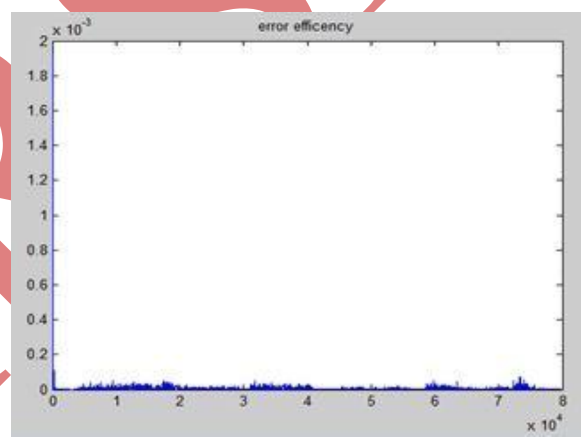

Figure 7: Error Efficiency

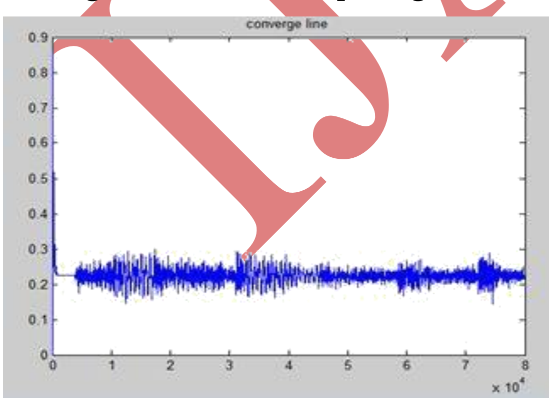

Figure 8: Converge Line

VI CONCLUSION

This paper shows a application based on noise cancellation using adaptive filter. The main aim of this paper is to use LMS algorithm as a noise cancellation algorithm. The noise free output is obtained after some LMS iterations. Therefore it gives good results in notice cancellation problem.

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