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INHIBITION OF ACOUSTIC NOISE USING AN ADAPTIVE LMS FILTER

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***Abstract*—In an audio speech signal, acoustic noise is a common problem while the speech is processed. Here, we are going to create color noise and add with an audio signal, after that a model are introduced to eliminate that noise. This paper elaborates a new approach for noise cancellation in speech enhancement using an Adaptive LMS (Least Mean Square) filter and with the help of MATLAB Simulink we get the correct speech signal. This filter is used to remove the acoustic noise due to its simplicity in computation & robust behaviour when implemented in finite-precision hardware. It provides better communication by suppressing the acoustic noise to a larger extent, since it provides a better balance between complexity & convergence speed. In spite of various methods, the results obtained in this way of noise cancellation are optimistic.**

Keywords- Adaptive filter, LMS filter, enhancement, Acoustic noise, noise cancellation.

I. INTRODUCTION

The basic way of communication in humans to convey information or message is speech. This communication helps the humans to fulfill their basic needs with a bandwidth [5] of 4 KHz. Noise is an unwanted signal is the major drawback affects the speech signal. There are various sources of noise which affects the speech such as electrical, acoustic, vibration or any other noise components. Among that, acoustic noise plays a major role since it becomes more noticeable as the number of commercial equipment increases.

Acoustics is a branch related to physics. This noise introduces a masking problem to reduce the distinct nature of speech which affects the communication [1]. During the elimination of this acoustic noise, the change of signal characteristics becomes more common. There are number of adaptive algorithms to remove the acoustic noise but we use Least Mean Square (LMS) filter to overcome this problem. The reason behind this is when compared with traditional filter design style, adaptive filter do not have fixed co-efficient. The co-efficient can be adjusted to increase the accuracy by reducing noise. There are two types of adaptive filter namely linearand non-linear. For the adaptive filtering [3], we can use both FIR and IIR filter. Here, we use an FIR filter due to its adjustable zeros and free stability. Whereas in IIR filter we have both adjustable zeros and poles.

In this paper, we develop a model which contains an audio speech signal as input which willintroduce an acoustic noise while the speech is processed. A color noise is added and using adaptive LMS filter we suppress the acoustic noise leaving the speech signal unchanged. This method improves the efficiency to a greater extent. The various forms are included [7] in colored noise such as pink, red, gray. Once the acoustic noise is removed from the signal, using the above said method the original input signal is obtained. In section I we give a brief introduction about adaptive filter and acoustic noise. In section II and III we explained the flowchart and block diagram. Further in section IV the methodology used for removing the noise is discussed. Section V and VI include the implementation and experimental results. In the results the difference between the input signal and filtered signal is shown. Finally in section VI we have given the conclusion.

II. BLOCK DIAGRAM

In this paper the model for the acoustic noise canceller using LMS filter is shown. The generated noise is added to the original audio signal which is in .Wav format file. The noise added signal is filtered [2] by fir filter in order to eliminate the white noise which is presented in noise signal that is generated from the random noise generator and to get the remaining colored noise from the signal. This noise is further filtered by using an adaptive LMS filter that we designed and created in order to cancel the acoustic noise completely.

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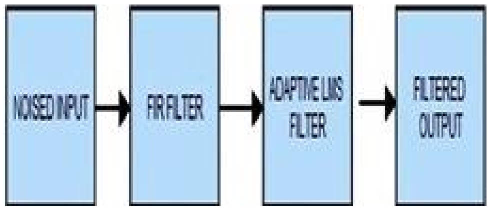

Fig.1. Block diagram

III. FLOW CHART

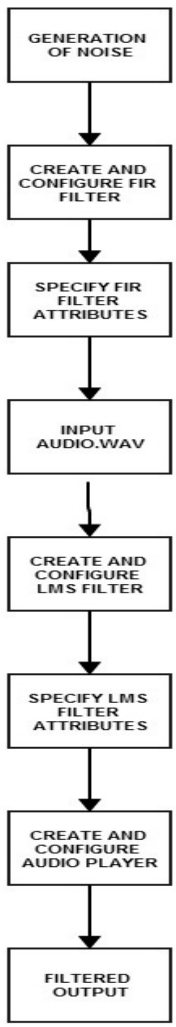

Fig. 2. Flow chart

IV. METHODOLOGY

Here, we use a method called Least Mean Square algorithm which is used to suppress the acoustic noise by using Simulink in MATLAB11a software. In MATLAB 11a Simulink has a Data Acquisition Toolbox helps to cancel the acoustic noise from the original signal. For suppressing the acoustic noise from the original signal, we are using the Block LMS filter.

A. *The LMS Algorithm*

The algorithm which was introduced by Widrow and Hoff is an LMS adaptive algorithm in the year 1959.The most widely used algorithm is LMS algorithm [9]. When it can be compared to other algorithms, it is easy and stable .The algorithm which does not requires any correlation function calculation is Least Mean Square algorithm but it requires matrix inversion. It is one of the simplest algorithms to understand and the Least Mean Square algorithm also needs a less implementation process compared to other type of algorithm. The only disadvantage in LMS algorithm is weak convergences. However, the merging speed can be affected due to Eigenvalues of the input matrix. The iterative procedure can also be included in the Least Mean Square algorithm. The iterative procedure in the least mean Square algorithm yields Minimum Mean Square which corrects the weight vector in the negative direction of the gradient vector.

Similarly, this algorithm needs two inputs for processing the audio signal. One is the error signal that has been calculated already. And the other is a mentioned noise which is nothing but the sound that can be related to the already existing original signal. The Least Mean square algorithm [10] alters the filter coefficient to minimize the cost function. The LMS algorithm mainly contains the filtering process to destroy the noise from the original signal.

B. *FIR Filtering*

The FIR (Finite Impulse Response) adaptive filtering can be implemented by using the LMS adaptive filtering. The FIR adaptive filtering contains the Normalized Least Mean Square (NLMS) adaptive filtering. The Wiener filter is Finite Impulse Response and the linear phase is the major property of the FIR filter .While the signal received from the present band, does not give any dispersion [8].This appears to give distinct frequency component of the signal have an atypical delay in the system. Mostly, the adaptive filter uses the FIR

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filter for the filtering process and further the coefficient values can also be modified by using the adaptive algorithm.

The efficiency [4] of the adaptive algorithm can be calculated by using three factors. Initially, it measures the calculus complexity. At each step, the amount of calculus will be executed. Secondly, the speed associated with the adaptive filter which results to the wiener solution. Finally, the solution obtained from the adaptive algorithm deviates from the present wiener solution whose difference is estimated as an error.

To cancel the adaptive noise is the major role of adaptive filters. The LMS algorithm contains the following process first, the various types of noises which will affect the original speech signal can be recorded and follow some different steps to designing the final model. It contains various follow of blocks can be combined together to design the model. The first, input to the adaptive filter is white noise .Here, FIR filter can be used to remove the white noise. The LMS filter uses two signals one is the reference signal and another is desired signal which helps to automatically match the filter response. As it meets the correct filter model. After completing the filtration process the Block LMS filter produces the rectified voice signal as the output which is very much similar to the initial speech signal and the noise can be removed. *Christo Ananth et al.* [6] discussed about Improved Particle Swarm Optimization. The fuzzy filter based on particle swarm optimization is used to remove the high density image impulse noise, which occur during the transmission, data acquisition and processing. The proposed system has a fuzzy filter which has the parallel fuzzy inference mechanism, fuzzy mean process, and a fuzzy composition process. In particular, by using no-reference Q metric, the particle swarm optimization learning is sufficient to optimize the parameter necessitated by the particle swarm optimization based fuzzy filter, therefore the proposed fuzzy filter can cope with particle situation where the assumption of existence of “ground-truth” reference does not hold. The merging of the particle swarm optimization with the fuzzy filter helps to build an auto tuning mechanism for the fuzzy filter without any prior knowledge regarding the noise and the true image. Thus the reference measures are not need for removing the noise and in restoring the image. The final output image (Restored image) confirm that the fuzzy filter based on particle swarm optimization attain the excellent quality of restored images in term of peak signal-to-noise ratio, mean absolute error and mean square error even when the noise rate is above 0.5 and without having any reference measures.

The block LMS algorithm will helps to renew the constant of adaptive filter block by block. The block size is same as that of the filter length. The filtered noise can be subtracted and error signal should contain only the original signal .The white noise can also be removed then the audio signal has a .Wav file and a desired signal are composed of color noise. Mainly the LMS filtering method is used to vanish the color noise and get the original input signal in the audio file format. The adaptive LMS filter gives the original signal, it removes the color noise and the output signal can also be filtered. The filtered output contains the original audio signal.

V. IMPLEMENTATION

We want to bring into existence so we have created a one model to just remove the noise from the audio file by using LMS. This can be implemented by using MATLAB Simulink. Here we followed a fewer number of steps to remove the color noise from the audio file. Flow chart in figure clearly outlined the proposed system.

As we told already this is one kind of model to remove the color noise from audio file by using LMS filter also. We generate the random noise named as color noise. The audio file which consists is white noise that can be removed by using FIR filter.Generally filters are used to extract the useful information such as a constituent part lying within a reserved frequency range. For example, in medical area we are using ECG is used to record the heartbeat of the human being .The heart beat signal which consists of some random noise that can be removed by using the adaptive algorithm for FIR filter. Mostly the adaptive algorithm is favored for digital filters. Hence the FIR filter is a digital type filter, adaptive algorithm can be used efficiently. In this paper the role of FIR filter is used to extract the white noise. Based on the Fir filter’s concept, the white noise can be removed from the audio file which can be implemented by using the series of multipliers, adders and delay for the filtered output. The output of the FIR filter is the linear combination of the present and previous values of the input. The figure(3) shows the input signal of the audio file .Though the white noise is extracted from the noised input signal but it consists of random noise named as color noise. Here only we introduced the LMS algorithm. The figure (4) represents the noise added signal. it shows both the type of noises. But we removed the white noise from this signal, then available noise as color noise.

Actually our aim is using of LMS filtering method to vanish the color noise and get the original signal of audio file which can be implemented in MATLAB.First why we

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took the LMS filter here, because it can easily relate the variation between the filtered signal and the noise signal. It is a steepest descent method. In that the filter is suitable by the based on the error at the actual time.The LMS filter to appear by taking too much for granted small weights and at one by one step, by verdict the gradient mean square error, the weights are updated.On condition that the gradient MSE value is optimistic, it involved, the error to makes the greatest possible value absolutely, for addition, if we use the identical weights which intends the result to bring down to a smaller weight .Similarly the gradient is negative we need to make greater the weights. In the MATLAB Simulink we need to correct noise signal by following five steps to get the output signal.

In the first step, when we give the input and the desired signal, then it returns the filtered output and the filtered error. Next we include the step size and then use the adaptive control. When the adaptation value is non zero, the LMS filter continuously updates the weights. When the adaptation value is the lowest degree the filter weights to be durable in the same state. Immediately we use the reset condition, it can reset the trigger condition. If a reset event happens, the LMS filter resets the filter weights occurring at the beginning values.

In the final step, we received the filter output, filter error and the adapted filter weights. After the removal of color noise, the LMS filter gives the output signal which looks like the original signal. Both the signals can be stored in the workspace. Finally the figure(5) given the filtered output signal.

VI. EXPERIMENTAL RESULTS

In this section we simulated the model that we designed for the acoustic noise cancellation using an adaptive filter based on LMS. The noised audio signal and the filtered signal mean without an acoustic noise signal can hear using through the audio player which is designed at simulation work space.

In our proposed method we created the digital filter module which updates its coefficients in order to remove the noise from the given noised input signal. The results are shown below.

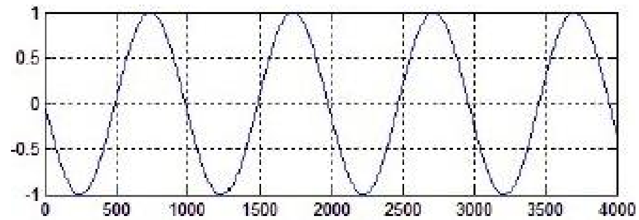

Fig. 3. Input signal

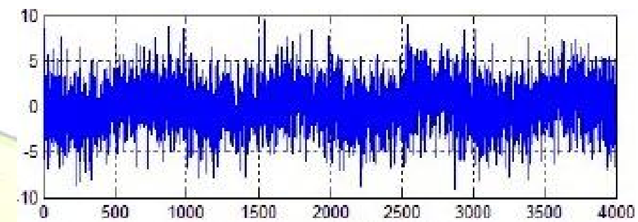

Fig. 4. Noise added signal

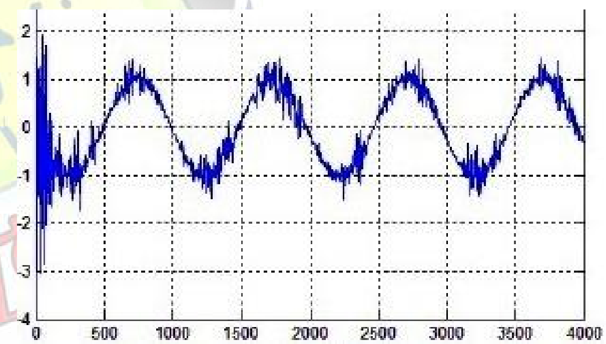

Fig. 5. Filtered output signal

VII. CONCLUSION

We designed and created a model in order to show the adaptive LMS filter works perfectly on the acoustic noise cancellation process. The results we obtained clearly shows that the ability of the adaptive LMS filters in noise cancellation and achieves the noise suppressed signal. The resulted output is almost similar to the original input audio signal.

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