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Echo Cancellation System with Dual Adaptive Filter and Effect of Multiplication Factor in LMS Algorithm Weight Equation

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Abstract:
The paper presents a Novel concept of application of dual adaptive filter in modeling of an echo cancellation system for active noise cancellation. It also investigates the robustness and practicability of the proposed model by analyzing the convergence rate, adaptability, step size requirements and closeness to desired output in comparison with conventional single filter model. The effect of adding a multiplication factor in the weightequation of the Least Mean Square (LMS) algorithm is analyzed and a new concept is proposed. Simulation and theoretical results indicate better performance and good filtering results as compared to conventional model.

Keywords: *Active noise cancellation, least mean square algorithm, Echo canceller,*Convergence Rate.

1. Introduction

Active noise cancellation (ANC) is a technique widely used to cancel and reduce unwanted noise generated by various factors in communication systems, digital signal processing (DSP) system, acoustic environments, and industry, automobile and defense systems. Many algorithms and models have been developed and tested in order to make the active noise cancellation system more robust and error less. The research in this field is still on as a wide scope of improvement still lies due to great number of parameters involved. One such factor that greatly affects the communication system is the echo generated by reflections caused by delays, deformations and unmatched terminations present in the communication network. This paper addresses the problem of echo cancellation and presents a model to counter the problem more effectively. As the echo generated in communication system is a random signal and no predefined equations or model can be defined so the cancellation technique is to be adaptive with the random changes .Thus the concept of adaptive filtering is found suitable and applicable in modeling the counter signals for cancelling the random echo signal. Various adaptive algorithms namely least mean square (LMS), Recursive Least mean square (RLS), Filtered-X Least means square (F-XLMS), Normalized least mean square (NLMS) have been analyzed and implemented in designing noise cancellation models by various researchers from time to time. The LMS algorithm has the advantage of low complexity, ease of implementation and higher stability [1] so is widely used in the given application and has been implemented in designing the model proposed in this paper. The basic blocks showing the interference of echo signal is shown fig.1 below:

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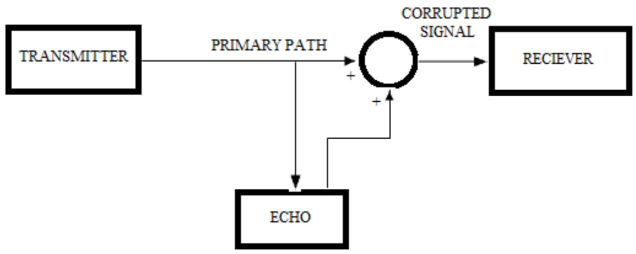


Fig 1. Basic communication system model with echo added to primary signal.

Fig. 1 above shows how an echo is added to the primary signal path and interferes with the desired signal. The challenge is to predict the echo signal using the adaptive algorithm and get it cancelled from the primary signal, for doing so the basic blocks as required are shown in fig.2 below:

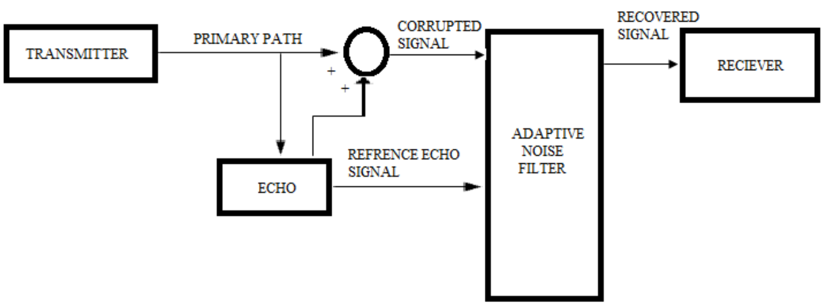


Fig.2 Basic block diagram of communication system with adaptive noise/echo cancellation.

Fig. 2 above shows the basic blocks of a communication system with adaptive echo canceller, the adaptive canceller predicts the echo as well as the noise added to the signal and recovers the signal through active cancellation techniques. The main thrust area of this paper lies in the adaptive noise cancellation block. The paper presents a novel concept of cancelling the echo and noise signal using dual adaptive filter as compared to conventional concept of single adaptive filter [1]. The paper is organized as section 2 gives a small literature review of the work done in the field and the new concept of LMS algorithm with a multiplication factor, section 3 describes the model proposed, section4 gives the analysis through simulation and results.

2. Literature review and Concept proposed

Literature review is an important part of any research paper thus a through literature review was done before reaching to the proposed concept presented in the paper. As the work is done using the adaptive filtering algorithm namely LMS algorithm ,few of the insights about the concept along with their referenced citation number is described.The basic concept behind LMS algorithm is to generate the output as the weighted sum of inputs [2]

$$y_i = x_i^T W = W^T x_i$$

$$x_i = \begin{bmatrix} x_{0i} \\ x_{1i} \\ x_{2i} \\ x_{3i} \\ - \\ x_{ni} \end{bmatrix} \text{and } w = \begin{bmatrix} w_0 \\ w_1 \\ w_2 \\ w_3 \\ - \\ w_n \end{bmatrix}$$

Where input is $x_i$and w is the weighing factor.

For the basic LMS algorithm the error between desired signal and filter output is given by:

$$e(n) = d(n) - w(n)x^T(n)$$

$$w(n+1) = w(n) + \mu e(n)X(n)$$

Where μ is the convergence factor of the equation and determine the stability of the algorithm, finding a suitable value of the convergence factor has gained a huge scope in research and many methods of optimization of μ has been formulated. In the reference [3]

$$\mu(n) = \beta\left(\frac{1}{(1 + e^{(-\alpha|e(n)|))-0.5)})}\right)$$

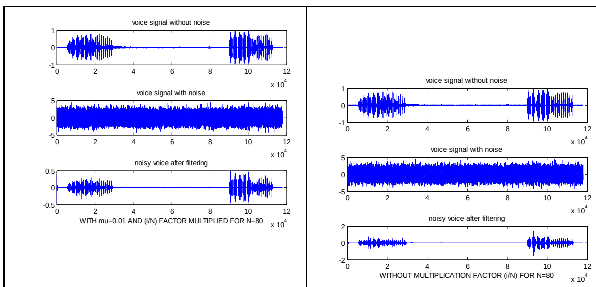
is a function of sigmoid about e(n).Using a sigmoid function overcome the mismatch between theconvergence rate and the steady state error[3].Another modification in μ is the correlation of $e(n)e(n-1)$ as shown in the equation given below.

$$\mu(n) = \beta\left(\frac{1}{(1 + e^{(-\alpha|e(n)e(n-1)|))-0.5)})}\right)$$

Correlation of error function is applied in the convergence equation to reduce the dependence of convergence factor on noise and improve the convergence rate [1]. Similarly the concept proposed in this paper is a modification in the equation of weighing factor w (n+1) given above. The approach is to add a multiplication factor (i/N) in the weight equation of LMS algorithm.The modified equation is given as under:

$$w(n+1) = w(n) + \mu\left(\frac{i}{N}\right)e(n)X(n)$$

The variable i is an integer with values ranging from 0 to N-1. N is the order of the filter. Adding (i/N) in the Widrow-Hoff LMS equation maintains the same convergence rate even at higher order as obtained by variable step size algorithmshown in[1]-[2]but with fixed step size it has the advantage of reducing the difficulty in implementation on adaptive filtering hardware and software. The simulation results comparing the output for noise cancellation in both the forms i:e with and without the multiplication factor (i/N) are given as under:



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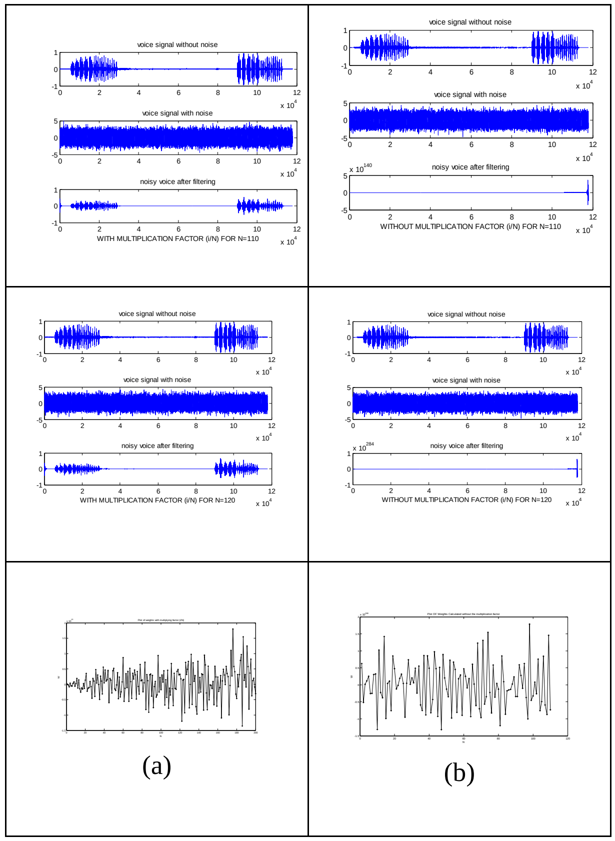


Fig.3 Plots of recovered signal and weighting parameter with and without multiplication factor.

It can be seen from the simulation results of fig.3 above that the multiplication factor (i/N) has a noticeable effect on the output of the adaptive filter .The output remains near to the desirable range till double the value of the filter order N, whereas without applying the multiplication factor the filter output deviates from the desired value after N=98 and becomes totally unacceptable after N=100. It remains in acceptable range for values of N greater than 200 in the modified equation. Thusas the weights can be obtained for a larger value of N as shown in fig. 3(a) it results in increase in adaptability with fixed step size. Thus the results justify that after adding the multiplication factor same convergence rate and adaptability can be obtained with fixed convergence factor as obtained from variable convergence factor.

3. Proposed model

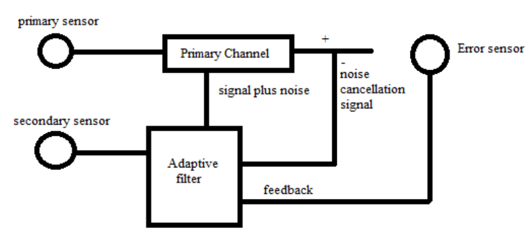


Fig.4 Basic noise cancellation model

Active noise cancellation is not only limited to the efficiency of the adaptive algorithm but its effectiveness also depends on the model in which it is implemented. The structure of the cancellation model may vary with the application but the basic model consists of the primary sensor which gives the desired signal to be transmitted through the channel, the secondary sensor records the surrounding noise which is to be cancelled .The third is the reference or

the error sensor which gives the feedback to the adaptive cancellation filter as shown in fig. 4 below.
Fig. 4 above give the very basic and conventional model for active noise cancellation but since research has advanced many models have been designed and has been published.A few ofthe new and advanced models published in various reputed publications reviewed before designing the proposed model are the echo cancellation model in which the adaptive filter simulate and cancel the echo signal generated by the deformations and discontinuities present in the communication network[1]. The hybrid analog-digital active noise control system in which an analog to digital converter (ADC) and an digital to analog converter (DAC) is used along with the active noise control system and it is shown that analog feedback reduces fluctuation in the feedback path and results in more robust system[4]. The inverse plant model in which the controller is divided into two parts the predictor and the inverse plant and it is shown that us of the inverse system reduces the computational load in the sound signal reproduction after noise cancellation[5].The model presented in this paper is based on the echo cancellation model [1] in which an echo signal is generated by delaying the primary signal.The echo signal predicted is then added along with noise to the primary signal as it gets added in real time communication network .The concept of active noise cancellation using dual adaptive filter is applied to recoverthe primary signal. The performance is analyzed in comparison to single filter model, simulation results are presented herewith. The structure of the model proposed is given in figure 5 shown on the next page:

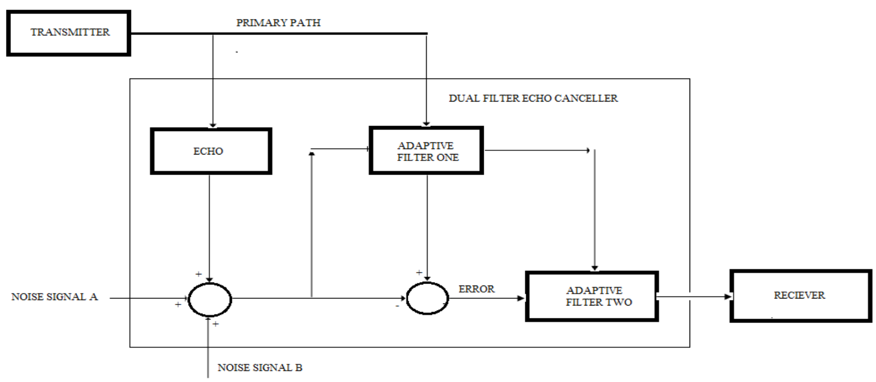


Fig.5 Proposed model using dual filter Echo Canceller

Fig. 5 given above shows the proposed model, it consists of the transmitter which generates the primary signal which is desired to be transmitted to the receiver. As it is an offline model therefore the echo signal is generated by adding a delay element to the primary path .The echo is added with two noise signals representing the noise corrupted signal during transmission. The adaptive filter one shown in the block diagram generates the error signal for the adaptive filter two .The recovered signal generated by the adaptive filter two is the desired signal to be received by the receiver.

4. Simulation and Results

In order to test and simulate the proposed model Simulink simulation tool has been used .The primary signal is a pre-recorded 8 bit .wav mono audio file of bird sound having a frequency of 16000Hz, it is delayed by delay element of z^{-200} to predict an echo signal, this echo signal is added to two number of noise signals of 16000Hz, 8 bit mono audio sound pre recorded as .wav multimedia file. Both the adaptive filters are of same order N=50 and convergence factor of LMS algorithm used in both the filters is μ=0.01. The simulation results is shown in fig. 6 (a-d) given below.

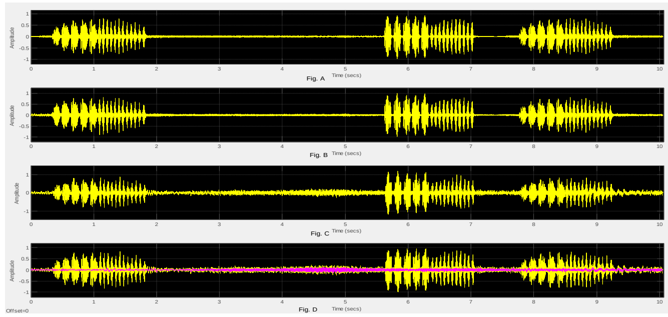
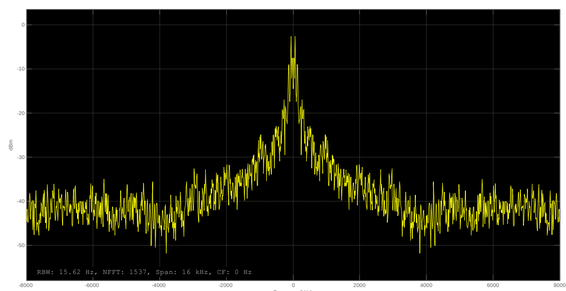
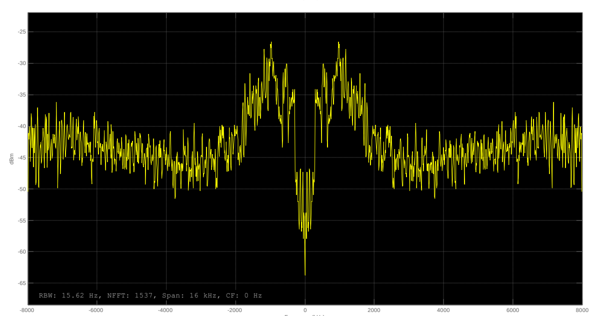


Fig. 6(a-d): Simulation waveforms.

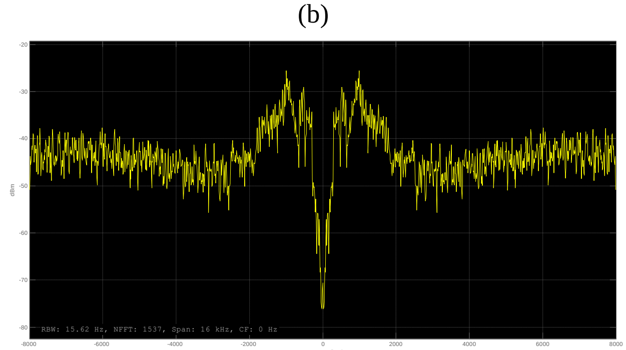
The primary signal is shown in fig. 6(a), the recovered signal after the dual filter is shown in fig.6 (b), the fig.6(c) shows the recovered signal from the conventional single filter model and fig.6(d) shows the echo signal added with the noise signal.



(a)



(b)



(c)

Fig.7 Frequency Spectrum of the original signal, recovered signal with single filter and recovered signal with proposed dual filter.

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5. Conclusion

The paper presents a dual filter echo cancellation model as compared to the single filter model and from the results shown fig.6 (a-c) it can be clearly noticed that the signal in fig 6(b) recovered after the dual filter as proposed is more close to the desired signal shown in fig 6(a) for the same convergence speed,step size and order of the filter.Fig.7 shows the similarity in frequency spectrum of recovered signal with original signal for both single filter and dual filter output. Thus the proposed model can be implemented practically in active noise cancellation system used in communication networks.

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