A New Improved Scheme for Combined Reduction of Acoustic Echo and Background Noise using NLMS Filters

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***Abstract* — In this paper, the basic emphasis is on implementation of a new scheme of noise and echo cancellation from speech signals using adaptive filtering techniques. The proposed model can be divided into two major parts: acoustic echo canceller and background noise canceller; both parts are kept independent of each other so that time-variance in background noise canceller does not affect the echo canceller. Echo cancellation is done by correlating far-end echo with input signal and the core of noise cancellation of this scheme lies in cascading technique for reconstruction of noise.**

***Index Terms*— Acoustic Echo Cancellation, Linear Prediction Error Filter, Adaptive Noise Estimation Filter, LMS, ALE, NLMS.**

I. Introduction

With ever increasing power and falling cost of digital signal processors and the availability of cheap memory chips, the use of speech processing systems for voice communication and recognition systems is becoming more and more common. As the presence of noise significantly degrades the perceptual aspects and quality of desired speech signal the main objective of the proposed scheme is to improve the intelligibility of processed speech signal by estimating the background noise and cancelling it from the corrupted speech signal along with the undesired echo version of original speech.

The results of the proposed scheme have shown significant improvement in the overall quality and intelligibility of speech when compared with traditional available LMS interference cancelling schemes in [9] and [1] where only the background noise is estimated and its wideband and narrowband component estimated by ALE filters are suppressed without taking care of acoustic echo which severely effects the intelligibility of desired speech signal.

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Other recently proposed schemes [3-5] have exploited the blind and psychoacoustic approach towards suppression of acoustic echo and background noise. The scheme in [3] uses post-filter with conventional acoustic echo canceller in the reference path which does not yield enough results although computational complexity is kept low. In [4] & [5], multiple-microphone schemes have been proposed which obviously will raise the computational complexity. The scheme in [5] has relatively lower computational complexity but it requires priori information while in [4] blind signal separation has been employed which bypasses need for priori information.

The proposed scheme in this paper, Fig. 4 keeps echo canceller and noise canceller independent and parallel to each other yielding faster output. Acoustic Echo Cancellation (AEC) block estimates far-end echo from reference path while in parallel noise cancellation is done by a two-stage process; first Linear Prediction Error Filter (LPEF) is applied to the recorded input for whitening of noisy signals and then the output of LPEF is fed to Adaptive Noise Estimation Filter (ANEF) which estimates the noise from input signal to reconstruct it. This reconstructed noise is then subtracted from speech reference path (from which echo has already been cancelled) leaving pure speech in the reference path behind.

II. ADAPTIVE FILTER ALGORITHM FOR PROPOSED SCHEME

A. Normalized Least Mean Square Algorithm

NLMS algorithm is a non linear adaptive filtering algorithm and the most common platform for supervised adaptive filters. It is a continuation of LMS algorithm developed by Widrow and Hoff [6] and the difference of these two algorithms is the method used by these algorithms for updating filter coefficients.

In NLMS algorithm, first of all the algorithm computes the output of filter using the initial filter coefficients convoluted with the input,

$$y(n) = \sum_{k=0}^{M-1} h_o(k)x(n-k) \quad (1)$$

where: $y(n)$ – filter output, $h_o(n)$ – filter coefficients vector, $x(n)$ – filter input, M – filter length.

Then the error is computed using,

$$e(n) = d(n) - y(n) \quad (2)$$

where: $e(n)$ – error, $d(n)$ – desired response.

Then the filter coefficients are modified using the computed error and a new set of filter coefficients is obtained to process the input signal. This procedure similarly goes on for every iteration until the error $\boldsymbol{e(n)}$ is minimized. The equation for updating coefficients is,

$$h_{n+1}(k) = h_n(k) - \frac{\mu e(n) x(n-k)}{x_n {x_n}^T} \quad (3)$$

where: μ – step-size.

The step-size is driven by the factor $\boldsymbol{x_n {x_n}^T}$, which is the essential difference of NLMS from LMS. In standard LMS, the filter coefficients are directly dependent on $\boldsymbol{\mu}$, because of which, stability of algorithm is not ensured [7] while in NLMS, this issue is resolved by normalizing the step-size with power of SNR of input signal [8, 9].

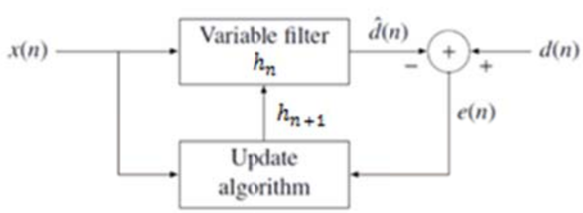

Fig.1. Block diagram representation of adaptive filter

The working of an adaptive filter is shown in Fig.1 where the input $x(n)$ is fed both to variable filter and update algorithm, the error from eq. (2) is used by update algorithm which is NLMS in our specific case. Update algorithm gives a new set of filter coefficients which helps in minimization of $e(n)$ in mean square error sense.

B. Mean Square Error Analysis

Mean square error (MSE) is the second moment of estimated value of the difference of desired outcome from the original outcome, a typical performance surface shown in Fig.2, is the plot of mean square error against the filter coefficients. The error performance surface shown in figure for illustration is a paraboloid for a two tap adaptive filter. Now for an optimal solution of minimization algorithm, the mean square error should be minimum.

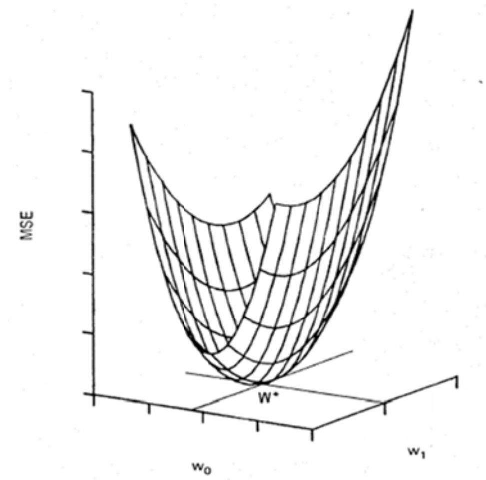

Fig.2. Error Performance surface

If the adaptive filtering algorithm does not yield an optimal solution, it will not reach the bottom of the bowl (shown in Fig.2.) and the difference of the resulting MSE with minimum MSE is called excess MSE. Another factor used for measuring performance of an adaptive algorithm is misadjustment, which is the ratio of excess MSE over minimum MSE.

These all factors are kept in mind and the point of minimum MSE can be achieved by setting appropriate step-size and filter lengths of adaptive algorithm.

C. Acoustic Echo Canceller

Acoustic echo canceller is an integral part of modern speech communication systems and its objective i.e. acoustic echo cancellation is done by employing two major methods; one is by using LMS and other is by using deconvolution.

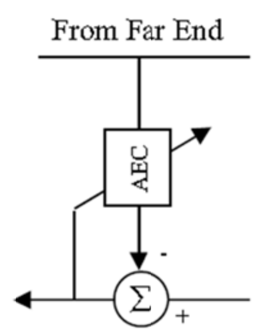

Fig.3. Acoustic echo cancellation mechanism

In the case of using LMS algorithm, a single LMS block is used with having far-end echo at input and it is cancelled from the mixed signal (having both near-end signal and far-end echo) hence passing the near-end signal as desired. When canceling acoustic echo using deconvolution, first of all impulse response of the environment has to be found on a test signal. Then, having found the impulse response, the effects of the system can be removed from the recorded signal by using the deconvolution again. This method is known as image method [10].

However in AEC using deconvolution, a more refined version of original signal can be recovered but it has not been used in the project for two reasons, one that it is a complex procedure as compared to LMS which would increase computational complexity of the scheme and other reason is that using deconvolution does not fit into complete scheme because separate testing has to be done for every environment where system is put to achieve its objective which would be redundant for noise cancellation modules. In this paper, acoustic echo cancellation using NLMS has been implemented which has no major difference from the method of cancellation of acoustic echo using LMS, details explained earlier.

D. Linear Prediction Error Filter

Linear prediction error filter is fed with the recorded input having near-end speech, acoustic echo and background noise components in the signal. The objective of this module is to whiten the noisy microphone which means that the output of the LPEF will be white noise only. This objective is achieved by taking benefit of the fact that real-life random signals like speech signals are stationary over a short period of time so most of the near-end speech and acoustic echo signals are predicted by LPEF in the sidelobe canceling path while if the input signals are background noise components, the filter coefficients will converge to make the output white.

E. Adaptive Noise Estimation Filter

Adaptive noise estimation filter, as clear from name, is used as an inverse filter to estimate noise from whitened signals. This is achieved by estimating the transfer function of noise generation system. Assuming that the background noise is generated by finite-order autoregressive process applied with white noise, the filter coefficients are updated in a way that the background noise is reconstructed from whitened signal.

III. THE PROPOSED SCHEME

The proposed scheme for acoustic echo and noise cancellation for single microphone input has been shown in
the figure below. This scheme has been adopted for efficacious acoustic echo and noise cancellation using the three constituent blocks discussed in the previous sections. AEC module has been placed in parallel to the other two modules which are in cascade. AEC estimates the acoustic echo in the recorded signal by cross-correlating the recorded signal with far-end echo, which is then subtracted from the reference path. On the other hand, recorded data is also fed into LPEF which yields whitened signal by predicting acoustic echo and near-end noise. This whitened signal is then used as input to the ANEF which works as inverse filter by estimating the noise, which is then subtracted from the reference path which already had acoustic echo taken out. After subtracting both the estimated echo and background noise, reference path only contains the estimated speech from the recorded input signal (which initially had components of acoustic echo, near-end speech and white background noise).

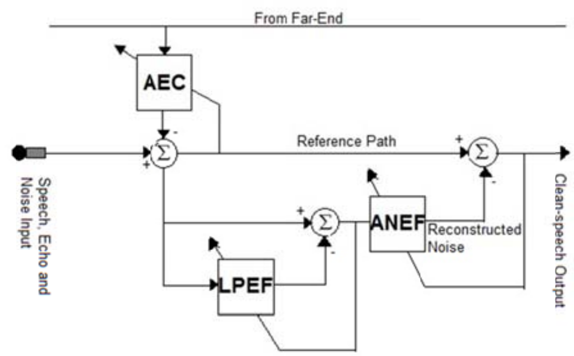

Fig.4. Block diagram representation of the proposed scheme

IV. SIMULATIONS

Simulations for the proposed scheme have been performed on Simulink utility of MATLAB R2009A. The details of the step-sizes and filter lengths been set are shown in Table 1.

Table 1. Simulation details of individual modules in proposed scheme

Module	Algorithm	Step-size	Filter length
AEC	NLMS	0.3	32
LPEF	NLMS	0.3	32
ANEF	NLMS	0.3	32

The Simulink Model of the proposed scheme has been shown in the figure below. Real-time speech signal is recorded for 5 sec through Data Acquisition Toolbox of MATLAB using computer microphones and stored in MATLAB workspace. The sampling frequency is set to 8000 samples/sec and bit resolution for quantization is 16-bits. This recorded signal is then called –up in the Simulink model. The resulting output also saves back into MATLAB workspace in vector form. The resulting output can readily be played using speakers.

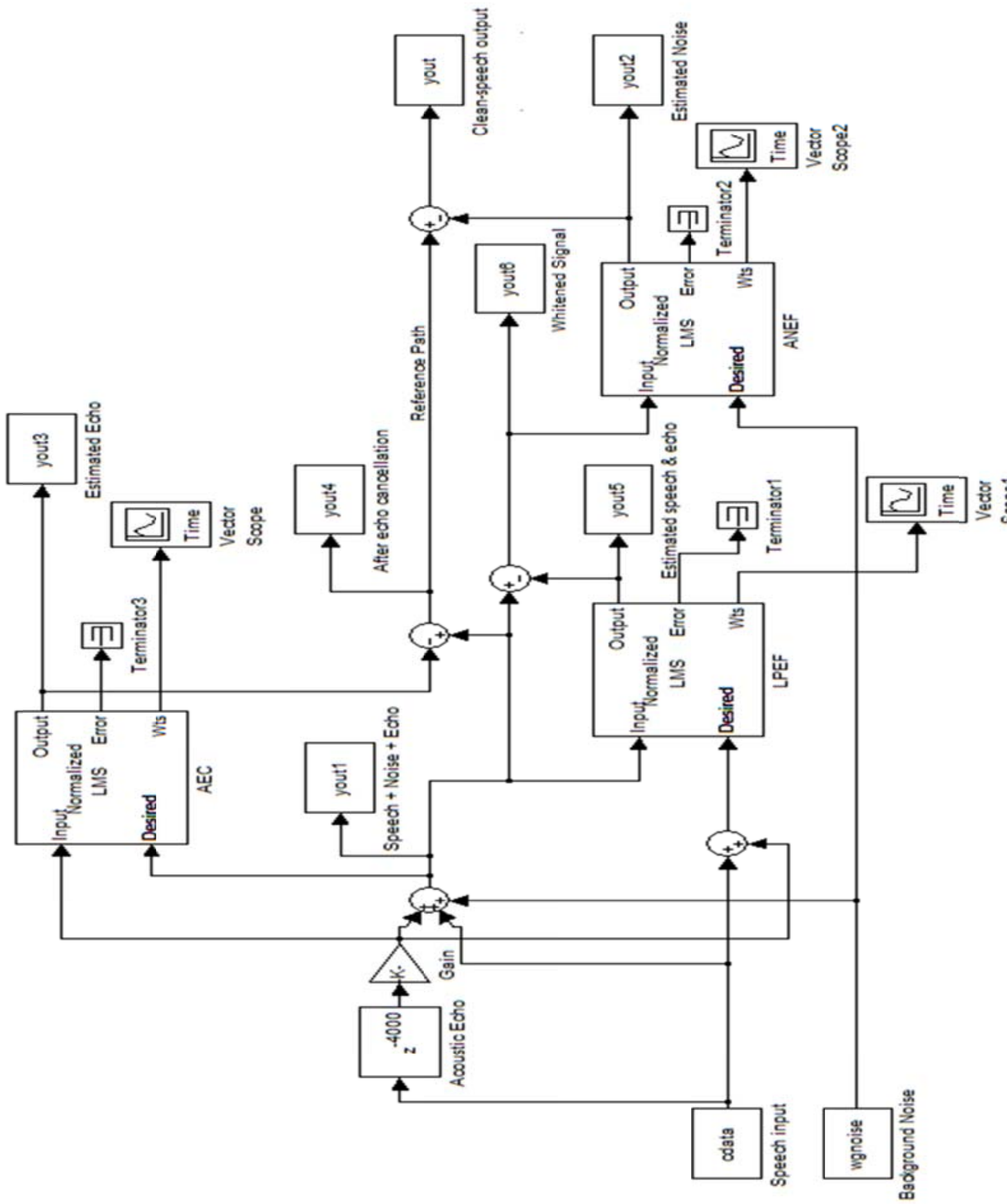

Fig.5. Simulink model of the proposed scheme

V. SIMULATION RESULTS

The recorded input signal containing the near-end speech with corrupted background noise and acoustic echo is shown below:

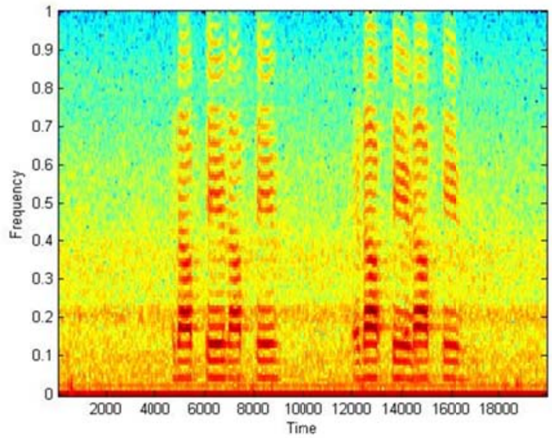

Fig.6. Microphone input containing all three components (speech, echo and noise)

The speech signal used for experimentation is shown below:

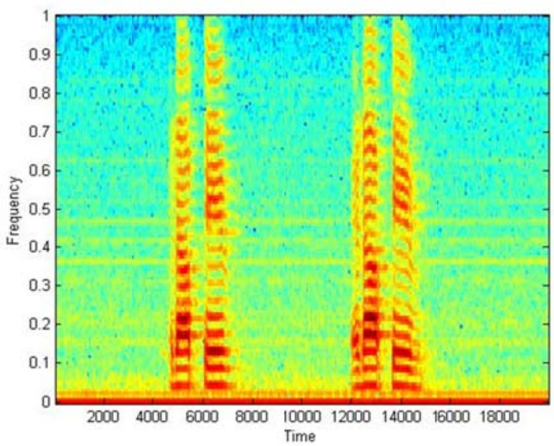

Fig.7. Desired speech signal

The figure below shows the frequency spectrum plot of reference path from which estimated acoustic echo has been subtracted; it contains background noise and near-end speech in it now:

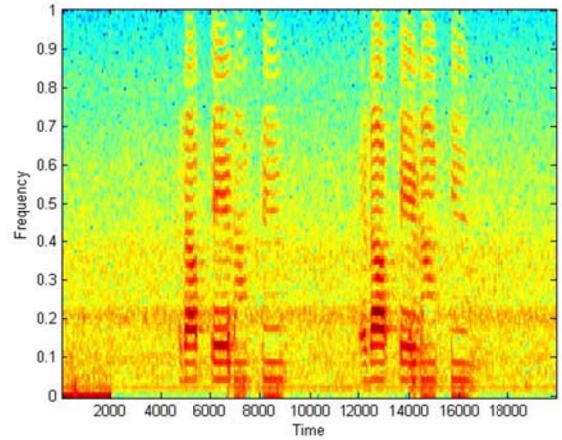

Fig.8. Reference path signal after echo cancellation

Near-end speech and acoustic echo components are predicted from corrupted recorded signal using LPEF and is shown below:

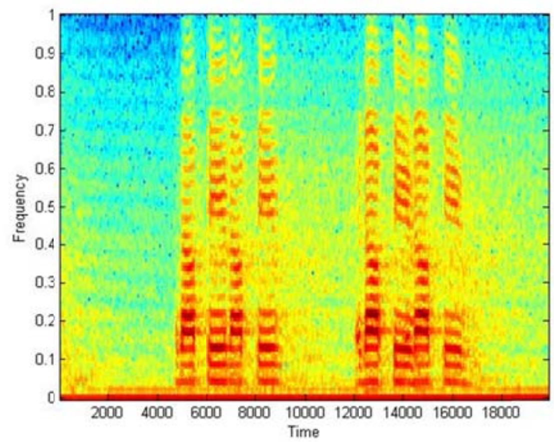

Fig.9. Estimated speech and echo in LPEF

Whitened signal which is obtained by subtracting predicted speech and echo components from corrupted signal and is then fed to the ANEF as input is shown below:

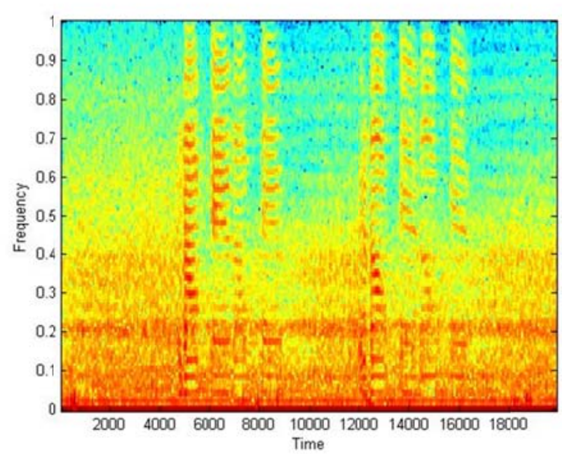

Fig.10. Whitened signal

Output of ANEF which is the reconstructed background noise is shown in the figure below:

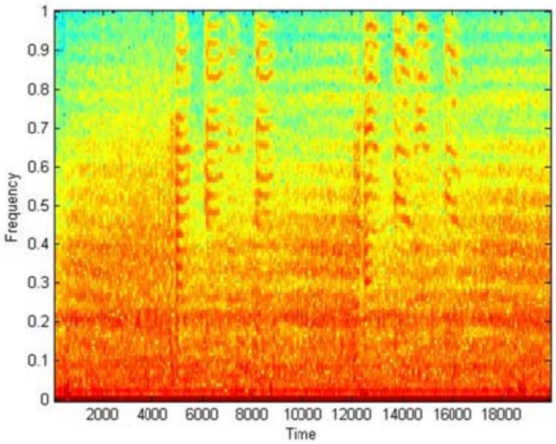

Fig.11. Estimated noise in ANEF

Output of the proposed scheme which is obtained by subtracting reconstructed noise from reference path (from which echo has already been suppressed) is shown below:

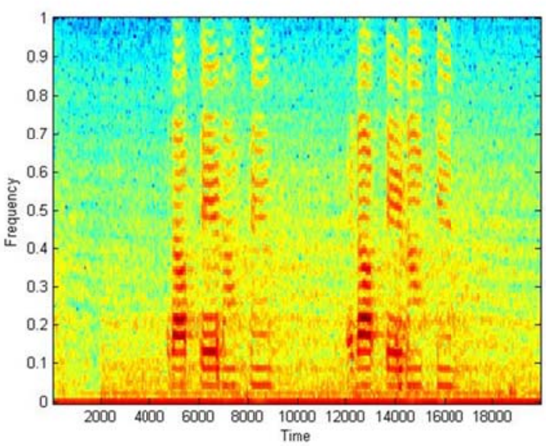

Fig.12. Actual output of the proposed scheme

Final processed speech output in Fig.12 poses a reasonable estimate of desired speech in Fig.7 in terms of its intelligibility and perceptual aspects. Section VI compares the proposed schema with other recent proposed schemes [3, 4, 5] which had jointly addressed the far end echo and background noise in corrupted speech in terms of SNR and computational time for processing corrupted speech signal.

VI. COMPARISON OF PROPOSED SCHEME WITH OTHER SCHEMES

Computation time of the schemes under consideration has been calculated using standard block-set components of Simulink Design Environment of MATLAB

SNR ratio of outputs of all schemes with same input has been listed in Table2.

	Post-Filter Scheme [3]	**BSS Scheme [4]**	**NABF Scheme [5]**	**Presented Scheme**
SNR (w.r.t. input)	0.46	0.82	0.78	0.76
Computation Time (/sample)(ms)	0.180	0.640	0.325	0.150

VII. CONCLUSION

The proposed scheme in this paper for combined reduction of acoustic echo and noise consisted of two parallel but independent stages in which AEC module cancels the far end echo from reference path leaving speech with background noise only which is then taken care of by NLMS based LPEF & Acoustic Noise estimation filters which do the estimation and subsequent cancellation of noise from corrupted speech yielding better quality speech signal.
Results of proposed scheme showed better overall quality of speech signal than proposed earlier in [3-5] in terms of optimization of minimum mean square error(cost function) as well as lower computational complexity.

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