Design and Evaluation of Modified Adaptive Block Normalized Algorithm for Acoustic Echo Cancellation in Hands-Free Communications

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Abstract

The aim of this paper is to design and simulate an AEC in order to enhance the quality of speech disturbed by echo phenomenon. Therefore, in order to design the normalized adaptive AEC, we have used digital signal processing techniques, especially Simulink embedded functions. Effectiveness of the suggested AEC using adaptive normalized algorithm was verified using Matlab/Simulink software. Finally, our AEC has been tested by using ERLE criteria and the analysis results show more efficiency according to ITU-T recommendation G.168.

Keywords: Digital Filters, Modified Block Normalized Adaptive Algorithm, Acoustic Echo Cancellation (AEC), Hands-free communication, Simulink\Matlab, Embedded functions, Mean Square Error (MSE), Echo Return Loss Enhancements (ERLE).

1. Introduction

Acoustic Echo Cancellation (AEC) is a technique that significantly enhances the audio quality of a hands-free communications system. Hence, it renders call progress as smooth as possible and offers more comfort to the callers by suppressing or lessening echo phenomenon [1].

Adaptive filters have emerged as an important technology in digital signal processing and real time application, especially acoustic echo cancellation systems. With its ability to remove the echo from original speech, the adaptive filters become a key solution of an AEC.

Different works of adaptive algorithms for echo cancellation are presented [2-3-4]. In [5], we modeled and designed an AEC by using a pre-defined model block adaptive filter to cancel acoustic echo in hands-free communications.

The goal of this work is to design and verify the performance of a software AEC using linear adaptive filter and digital signal processing techniques. In this step, we propose a design of modified block efficient Normalized Adaptive filter based embedded functions. It's used to enhance the quality of speech in spite of echo phenomenon.

The paper is structured as follows: section 2 presents echo cancellation problem formulation, section 3 presents model design and simulation results, and Section 4 concludes this paper.

2. Echo Cancellation Problem Formulation

2.1 Adaptive Filters for modeling AEC

The outcome of this modeling technique is to help design an AEC that can be carried out by using adaptive filters while keeping in mind that these filters might be Finite Impulse Reponses (FIR) filters or Infinite Impulse Response (IIR) filters.

All the way through this work, we have exclusively implemented FIR filters to achieve the modeling of Acoustic echo cancellers based on normalized adaptive algorithms in time domain. Figs. 1 and 2 respectively illustrate system's diagram block.
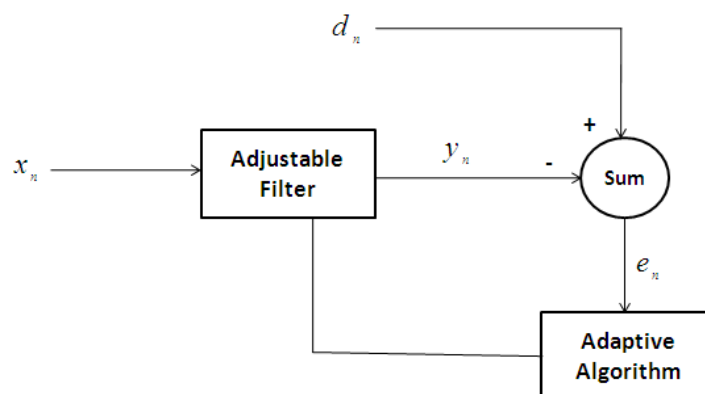

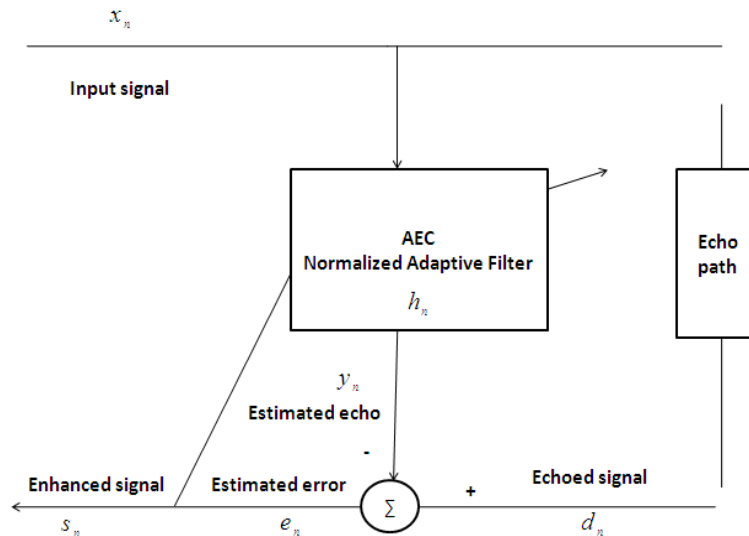Figure 1. Normalized Adaptative filter structure

Figure 2. Acoustic echo canceller based on normalized algorithm Block diagram

This algorithm involves two steps:

-estimate the echo signal by using the following equation:

$$y_n = \sum_{k=0}^{N-1} h_k x_n(k) \tag{1}$$

Where $h_n(k)$ are the filter coefficients and N is the filter length. Therefore y_n is the convolution (inner product) of the two vectors h_n and x_n.

-update the of adaptive filter weights by minimizing a cost function (Mean Square Erreur (MSE):

$$MSE = \frac{1}{2} \sum_k (e_n)^2 \tag{2}$$

Where,

$$e_n = d_n - y_n \tag{3}$$

$$h_n = h_{n+1} + \frac{\lambda}{\beta + \|x_n\|^2} x_n e_n \tag{4}$$

2.2 Modified N LMS Block Algorithm

In this subsection, we designed a modified block in the order to evaluate the complexity of the echo path by means of a normalized adaptive algorithm. This Block may be designed

by using digital signal processing techniques and embedded functions in Matlab Simulink environment [6].
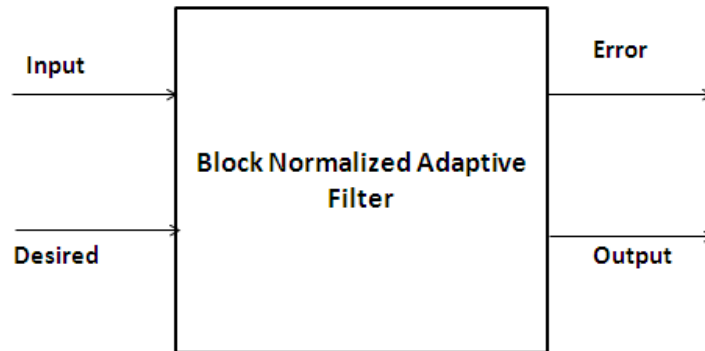

Figure 3. Modified NLMS filter block.

The block estimates the filter weights h_n needed to minimize the error, between output y_n and d_n.

Where y_n and d_n respectively designate the estimated echo and echoed signal, while e_n represent the estimated error.

Widrow & al [7] formulated the LMS algorithm for obtaining the minimum output power.

On the other hand, a periodic echoed signal has a very long correlation time [8].

Hence, we define the primary input signal to be a delayed version of the measurement y_n and the reference signal to be the measurement itself.

We then apply the following modified normalized algorithm which tracks the desired solution.

The variables of normalized filter can be summarized as follows.

Table 1. Variable of normalized filter

Variable	Description
n	the present step of the algorithm
x_n	the input signal at step n
h_n	The adaptive filter coefficients at step n
y_n	the estimated echo at step n
d_n	the echoed signal at step n
e_n	the estimated error at step n
s_n	the enhanced signal at step n
λ	the step to adjust. where $0 < \lambda < 1$
β	small number inserted in the denominator to avoid division by zero.

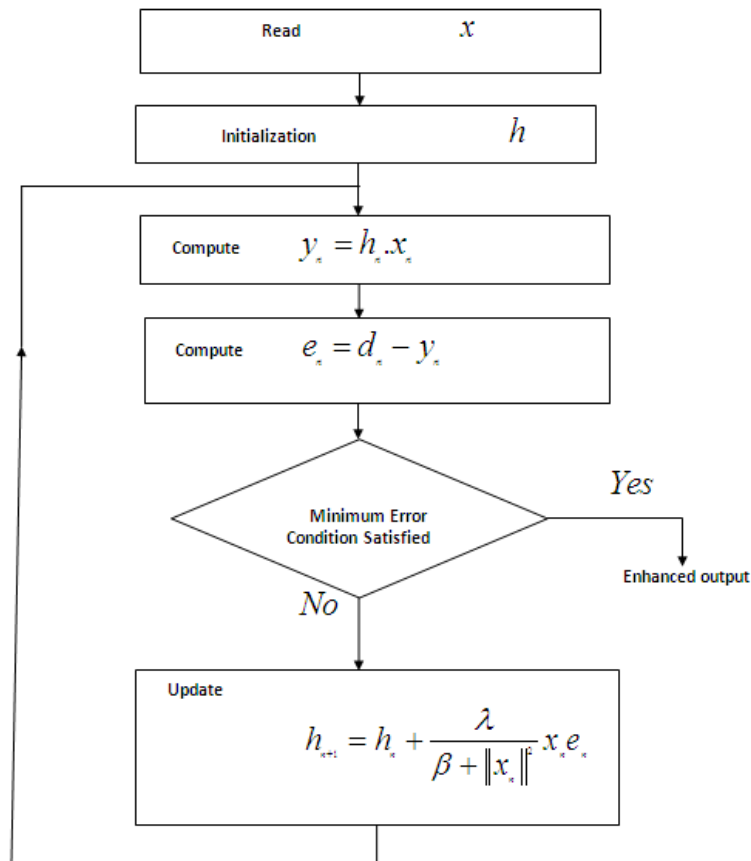Summary Flowchart in Fig. 4 describes the flow of the adaptive algorithm in time domain.

Figure 4. Flowchart of normalized adaptive algorithm

3. Model design and Simulink Results

3.1 Echo Canceller design using Matlab/Simulink

The overall performance of the module is guaranteed as shown in Fig. 5. It is designed and verified by using the following parameters:

The delay time is 32 (ms) and the number of samples for each read from file length (From Wave File) is 64 samples per frame at 8000 Hz sampling rate.

AEC implementation is setup with NLMS adaptive filter having a length of 64. The step size is chosen as $\lambda = 0.00001$.
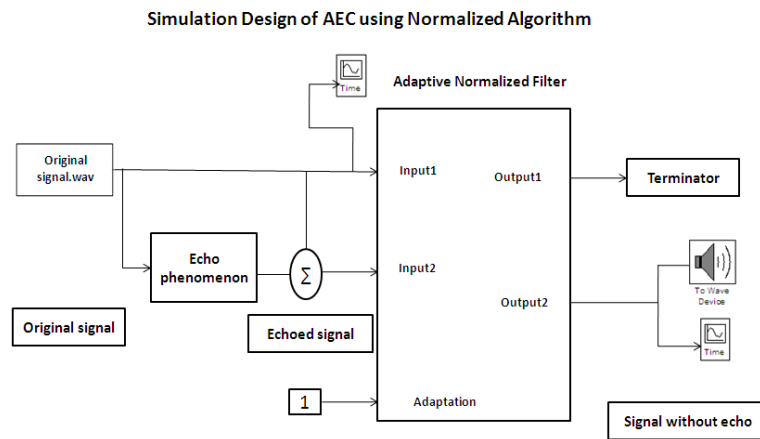
Simulation Design of AEC using Normalized Algorithm

Figure 5. Echo cancellation under Simulink using functional programming

3.2 Simulink Results

The AEC modeled throughout this work was tested using Matlab/Simulink software. The original signal and echoed signal used in this simulation are respectively reported by Figs. 6 and 7.

Fig. 8 shows system's output and reveals that speech quality has been improved.
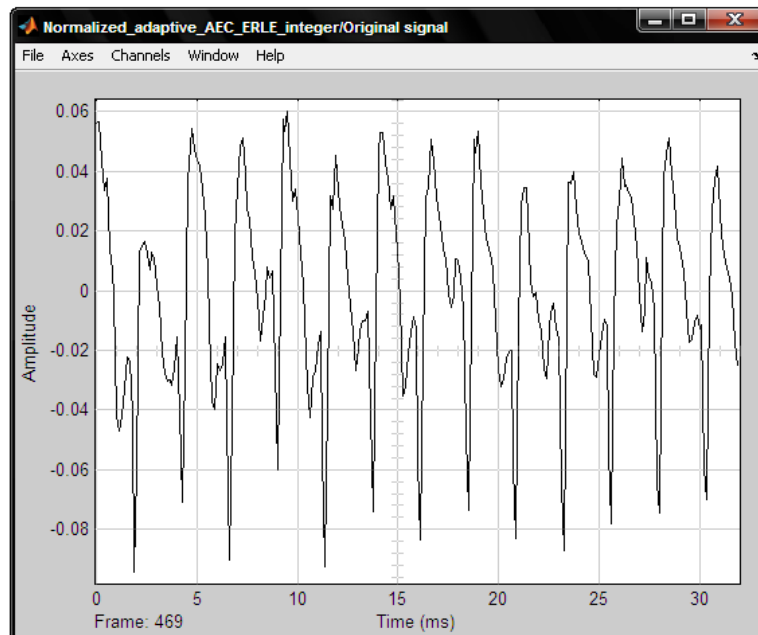

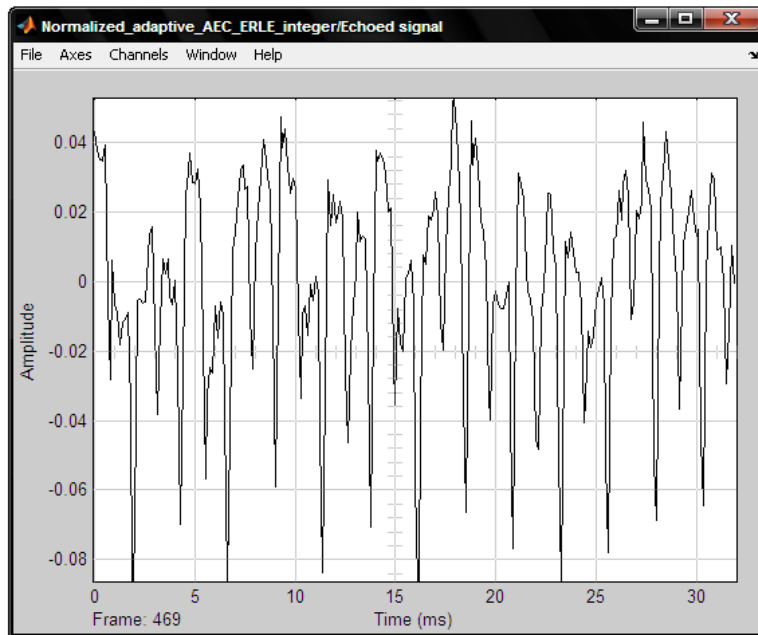
Figure.6. Original signal (Simulink)
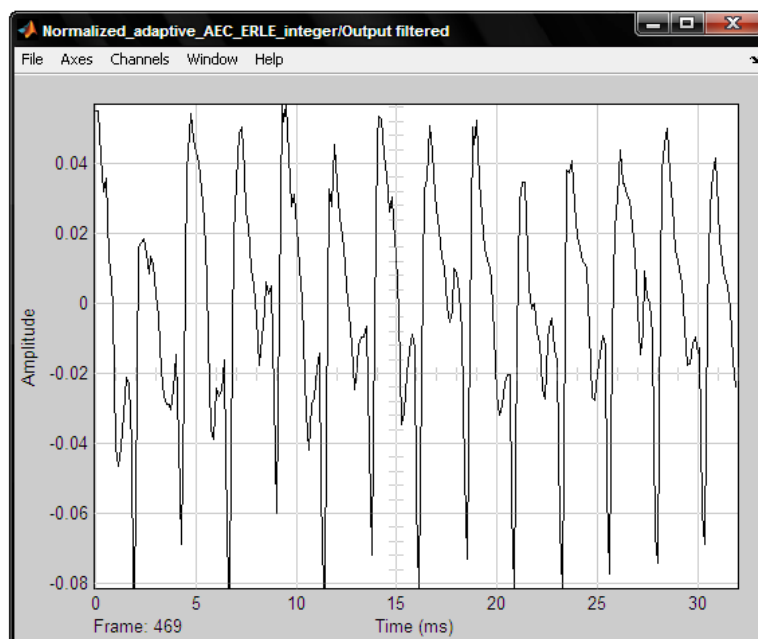
Figure.7. Echoed signal (Simulink)

Figure.8. Filtered output (Simulink result)

At this point, we expanded several scenarios to test the convergence of our AEC based on modified normalized adaptive algorithm. For each scenario, we did verify the convergence performance of the algorithm by adjusting step-size and filter length parameters and observed which step-size best values lead to optimal weights. Therefore, the designed AEC provides significant enhancement in the transmission path by reducing the strength of the echoed signal.

3.3 performance of the proposed AEC

The effectiveness of the proposed AEC is verified by using Echo Return Loss Enhancements (ERLE) criteria. It is calculated as:

$$ERLE_{dB} = 10\log_{10}(\frac{Power(Echoed\ signal)}{Power(Output\ AEC)}) \quad (5)$$

The proposed AEC based Normalized Least Square algorithm has shown an improved ERLE performance according to ITU-T recommendation G.168 [11]. In this context, the simulation results for ERLE are presented in Table 2.

Table 2. ERLE improvement of the proposed AEC (N=256, L=64, λ=0.00001)

Time (s)	ERLE of enhanced output (dB)
3	27.23
6	27.6
9	20.12
12	20.29
15	29.63
18	18.72
21	22.74
24	14.75
27	28.96
30	27.7

4. Conclusion

This paper proposed an AEC based on normalized adaptive algorithm and digital signal processing. With quantifiable results, this work seems to be an efficient method to be used to improve the quality of speech in hands-free communication.

The keypoint of this approach is to design an AEC module for both simulation and real time implementation, taking benefit from the simplicity of programming using functional simulink. The experimental results show that the ERLE values are very reasonable and in accordance with ITU-T recommendation G.168.

Finally, knowing that the rapid development of hands-free communication systems tends to generate a lot of nonlinear acoustical echo, we will therefore design and implement, in a future work, a nonlinear AEC to reject this disturbance. Additionally, we will mainly focus on the improvement and effectiveness of this nonlinear AEC.

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