Echo Cancellation Using NLMS Adaptive Algorithm

Tejendra Pratap Singh
EC Department, NITM, Gwalior
rickyrajj@gmail.com

Brijendra Mishra
EC Department, NITM, Gwalior
Brijendra.mishra07@gmail.com

Abstract:-This paper presents the Cancellation of an acoustic echo. Such echo canceller is essential to control the echoes generated by the acoustic coupling between the microphone and loudspeaker in hands free telephones or teleconferencing applications. Conventional acoustic echo canceller encounters problems like slow convergence rate (especially for speech signal) and high computational complexity as the identification of the echo path requires filter with more than a thousand taps. In short the NLMS technique allows high quality acoustic echo cancellation whilst maintaining reasonable computational burden for low cost applications. This is especially the case when the acoustic echo path increases in length.

Keywords:- Speech Processing, NLMS, MSE, Echo Cancellation.

I. Introduction

In a teleconferencing environment, the far end microphone and being sent back to him as echo often capture speech by the near end speaker [1]. For acoustic echo cancellation, the initial speech transmitted to the far end is adaptively filtered to follow the echo of the speech retransmitted from the far end. The difference of the two signals (i.e. the error of the adaptive filter) is transmitted to the near end. The adaptive filter in adapting its filter parameters uses this error signal [6], [7]. Fig. 1 shows such an acoustic echo-cancelling set-up.

To implement a low-pipelined Least Mean Block (LMS) as well as a pipelined delayed-LMS (DLMS) flexible digital Finite Impulse Response (FIR) filters On-Field Programmable Gate Array (FPGA) chips for common noisecancellation purposes and compare the behavior of nonpipelined and pipelined adaptive calculations when it comes to FPGA source, speed and location [8], [14]. We identify flexible noise-cancellation and show it using real-time speech signals. Adaptive filters find app for their vibrant character and so they work with the theory of detrimental interference [9], [15].Furthermore, it generally does not require a previous familiarity with the transmission prices to make certain stability [10]. NLMS is the positive choice for many of the industries due less computational difficulty and good amount of sound reduction. [16]. In the existing reports the writers took the feedback signal as sinusoidal signal etc. so that you can measure the effectiveness phase measurement will be the main aspect for your convergence pace and mean square error [17]. Speech enhancement aims to boost presentation quality by using different calculations. It could appear Basic, but quality suggests a-one of these [11], [18].Additional calculations of presentation improvement for noise reduction on foundation of method might be grouped into three basic classes: filter tactics, spectral recovery, and model-based practices [13].

In an acoustic echo cancellation, a model of the room impulse response is identified. Since the condition in the room may vary continuously, the model needs to be updated continuously. This is done by means of adaptive filtering algorithms [12]. The well-known NLMS algorithm has generally been chosen for practical implementation. However, this algorithm performs poorly in the acoustic echo cancellation context. Recursive- Least-Squares algorithms are known to exhibit better performances, but suffer from complexity and instability.

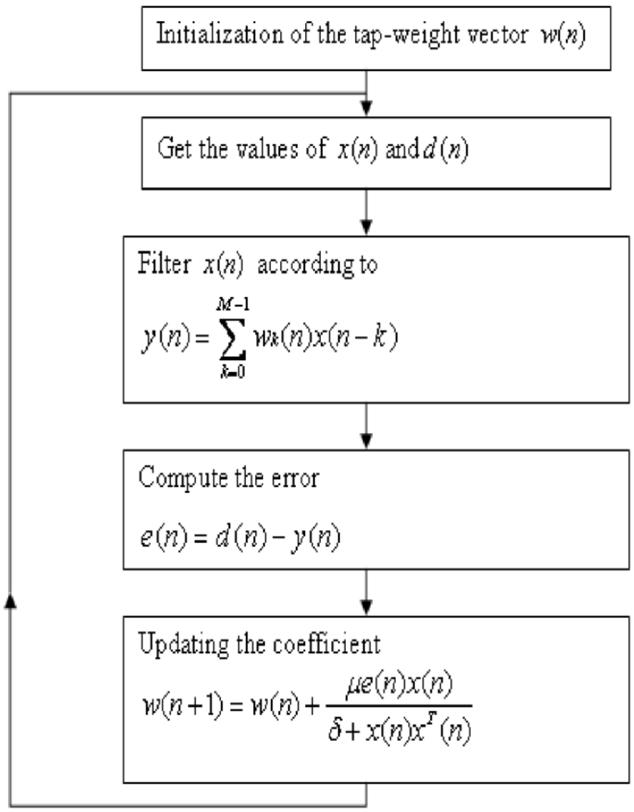


Figure.1 Basic Diagram of the NLMS Algorithm let x(n) be the input signal (from the far end speaker) travelling to the near end speaker through the loudspeaker and d(n) is the signal picked up by the microphone which in this case is the desired input signal (echoes) [2]. The adaptive filter is used to model the transfer function of the room in which the loudspeaker and microphone are in to generate a replica of the echo, y(n) following that, the estimated echo is subtracted from the desired input signal d(n) yielding the estimation error signal,

e n d n y n( )□ ( )□ ( )

The aim is to cancel the desired input signal d(n) (echoes) and that is by making sure the error signal e(n) is kept to the best minimum value possible. From Figure 1, it is also noted that past values of the estimation error signal e(n) is fed back to the adaptive filter. The purpose of the feedback is to effectively adjust the structure of the adaptive system, thus altering its response characteristics to the optimum possible. Simply, the adaptive filter is self adjusting hence the name 'adaptive'.

II. NLMS Algorithm

The Normalized Least Mean Square (NLMS) [3] algorithm is one of the most popular adaptation algorithms for error cancellation due to its computational simplicity.
Referring figure 1, x(n) is the vector of tap inputs at time n, d(n) is denoted as the desired response, y(n) is the estimate of the desired response (output of the filter) and e(n) is the estimation error.

e(n)=d(n) – y(n)

y(n)=$w^T(n)x(n)$

Where w(n) is the weight vector and the subscript T denotes the transpose of the w(n).

w(n)=$[w_0(n)w_1(n) \ldots \ldots \ldots \ldots \ldots w_{M-1}(n)]^T$

And x(n)x(n)^{T}is tap input vector

Table 1Condition in Simulation Experiment

Sample Rate of speech Signal	8KHz
No of Filter Taps	256
No of Taps for Acoustic Echo Path	400
NLMS Parameter	μ=0.9,□=0.001
Noise Variable	0.001

$(n) = [(n)x(n-1) \ldots \ldots x(n-M+1)]^T$
NLMS algorithm updates its weight according to
w(n+1)=w(n)+μ (n)e(n)x(n)

Where μ (n) = ______ $x(n)x^{\mu}(n)_{T+\delta}$

Where x(n)x(n)^T is tap input power and is given by
x(n)x(n)$^T = x(n)^2 + x(n-1)^2 \ldots \ldots x(n-M+1)^2$

And δ is a small positive constant used to avoid that denominator results in 0 when input signal is small. μ is a step size constant that regulates convergence rate. The value of the step size constant μ for the NLMS [4] algorithm to be convergent in the mean square should be
0 < μ < 2.
In order to achieve a fast initial convergence speed and to retain a fast tracking ability in the steady state, large value

DOI- 10.18486/ijcsnt.2016.5.3.01
ISSN-2053-6283

for step size is chosen. On the other hand, large step size will result in large steady –state misadjustment error.

III. Simulation Results Analysis of NLMS

To perform simulation of NLMS algorithm for acoustic echo cancellation the far end speech signal and its real echo were used at sample rate 8 kHz [5]. For real speech input, both the speech input and the desired input signal consisted of 80,000

sample points and the echo path was assumed to have known impulse response, $h(n)$ of 400 points long .
The filter length for real speech case was set to 256 taps. The step size parameter μ was set to be 0.9 and the constant □ was 0.001 for the simulations. Also, it was assumed that the near end speaker was silent and noise free.

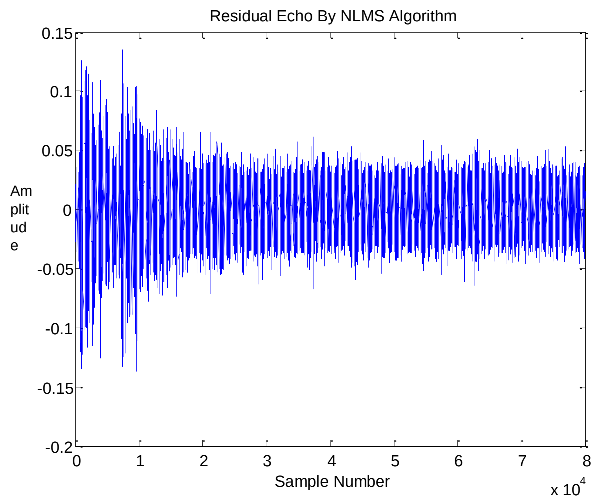


Figure 5 Residual Echo.

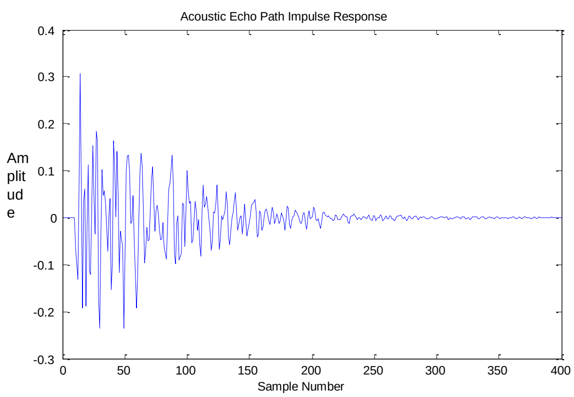
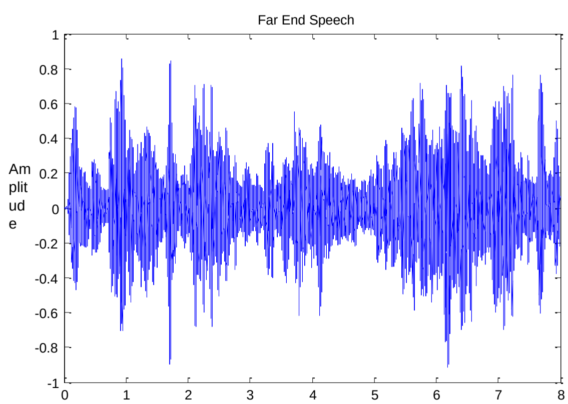


Figure 2 Room Impulse Response.



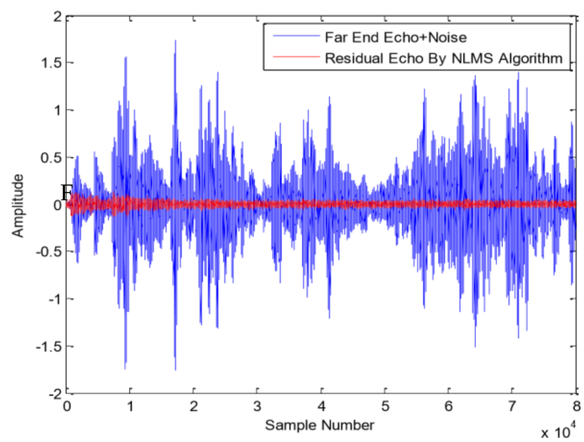
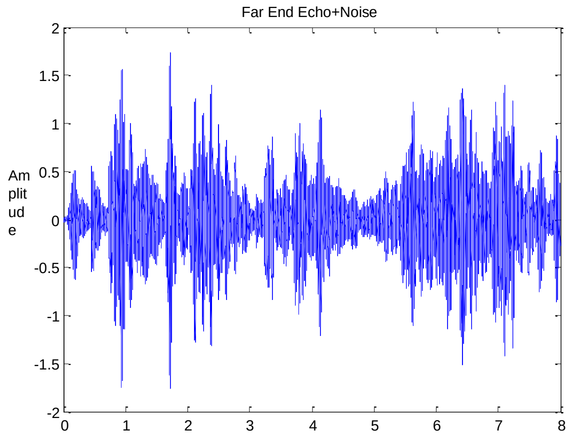


Figure 6 Acoustic Echo Cancellation Using NLMS Adaptive Filter Compare to Far End Echo.

Sample Number x 10^4



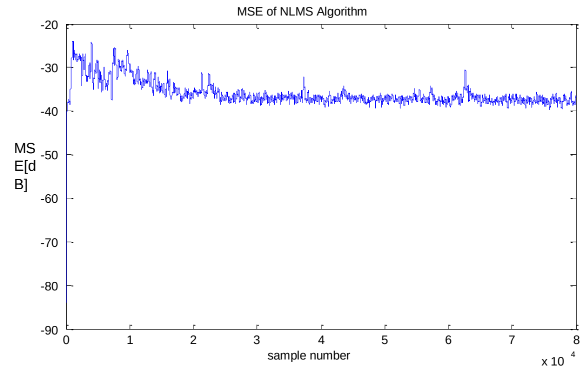


Figure 7 Mean Square Error Performance of NLMS

IV. Conclusion

Besides that, the acoustic echo canceller assumes that the near end speaker is silent. So further work can be made to consider the double talk situation. In this case both the near end speech and echoes are picked up by the microphone. So it would be in appropriate to cancel the near end speech signal as well. Thus, the incorporation of the double talk detector is recommended for the expansion of the design. Besides that, the problem of ambient noise in acoustic echo cancellation is recommended for further work. This includes a joint structure of acoustic echo cancellation and noise reduction to enhance speech quality.

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