Modeling and Analysis of an Adaptive Filter for a DSP Based Programmable Hearing Aid Using Normalize Least Mean Square Algorithm

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***Abstract*— The application of adaptive filters in the design and implementation of digital hearing aid brought flexibility to its adaptation in both stationary and non-stationary environment. Several existed adaptive algorithms have added one or two significant changes to effective operation of digital hearing aid device. Apart from minimum mean square error reduction, the convergence rate of Least Mean Square (LMS) algorithm in the implementation of the previous digital hearing aid had posed a challenge. In this work, a Simulink block model for Normalize Least Mean Square algorithm (NLMS) was modelled and simulated for determining the rate of convergence of NLMS algorithm. The simulation results proved that NLMS algorithm exhibited a high rate convergence feature when the step size value was varied progressively within the restricted bound of $0 < \mu < 2$ for minimum instantaneous square error reduction.**

Keywords*— Adaptive filter, Convergence rate, hearing aid, NLMS algorithm.

I. INTRODUCTION

The increased cases of hearing loss in our society, community, country and around the world at this present age, has called for serious sensitization over the issue of hearing impairment, its causes and possible solutions to remedy the effect. Recent statistics from World Health Organization (WHO) in 1995, shows that one-third of the elderly people with in the bracket of 70 years and above were reported to be at highest risk of hearing deafness [1]. Recent statistics from world Health organization (WHO), reported that out of 50% of the world's population, 360 million estimated people are living with hearing disability (328 million adults and 32 million children). In 2000, the WHO's report showed that more than 250 million people world wide are living with hearing loss in which two-third of this population lives in developing countries [2]. The outcome of the research showed that hearing loss is prevalent in adult and is largely caused by diseases, age and noise. The increased numbers of hearing impaired were as a result of some factors mentioned earlier and many more which could be of natural or artificial cause. As a result, the field has attracted huge investment and extensive research towards improving the existing digital hearing aid in the market. Hence getting rid of noise at the filter output is the major concern of this work. Several algorithms have been deployed in the process such as the LMS algorithm, which has equally shown some significant successes in terms of ease in computation of the filter coefficient, hardware implementation and noise reduction. Therefore, this paper chooses to look into the convergence rate of NLMS algorithm towards mean square error reduction as a better choice for improvement in the performance of hearing aid. The modelling of the adaptive filter was adopted and developed in Simulink using the in-built block sets provided by its library and thereafter simulated. The result obtained showed that NLMS algorithm possesses higher speed of convergence simply because of its variable step size. Hence the algorithm can reduce instantaneous noise faster and efficiently without the users experiencing the effect when put to use.

II. ADAPTIVE FILTERING

An adaptive filter is a filter with an associated adaptive algorithm for updating filter coefficients so that the filter can be operated in an unknown and changing environment [3]. In this way, an adaptive filter can be said to have a self modifying property which allows it to change the output response as the filter input characteristics changes. The number of different applications in which adaptive techniques have been successfully used has increased enormously during the last two decades. Some of these applications include system identification, adaptive beam forming, echo cancellation, channel equalization, signal enhancement, noise cancellation and control [4]. A finite Impulse response (FIR) filter therefore is a digital filter used for digital signal processing. In other words, it can be stated, that what makes a digital filter adaptive is the adaptive algorithm which updates the in-coming signal by a mere adjustment of the filter parameters such as the filter length, step size, adaptive weight etc. This automatic update makes the adaptive filter to adapt to both known and unknown environment usually in a time varying environment using a feedback mechanism as shown in fig1. This processing of providing automatic adaptation to changes in background is referred to as tracking. Therefore, the principle of adaptive filter is aimed at minimizing the mean square error (MSE) between the filter output and the desired signal [5].

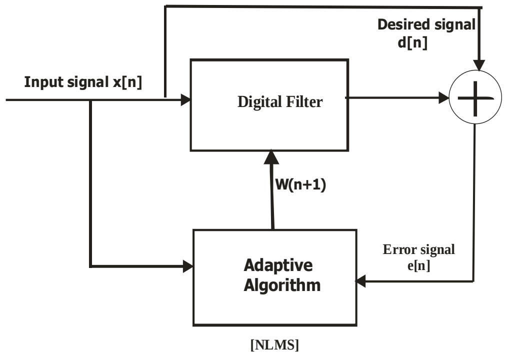

Fig 1: Adaptive filter structure using FIR filter for tracking of the filter coefficient.

Fig.1 shows the input signal that have undergone Analog-to-Digital (A/D) conversion process, the digital version of this signal is then passed through the adaptive FIR filter resident in digital signal processor (DSP) as shown in fig.2. The output signal, y(n), is thereafter compared with the desired signal, d(n). If there is any difference signal detected, the difference signal, e(n) regarded as the noise signal triggers the adaptive algorithm to adjust the filter coefficient of the next iterated signal. The whole process is repeated until the output signal, y(n) is approximately equal to the desired signal, d(n) before advancing for Digital-to-Analog (DAC) conversion and post amplification as the case may be.

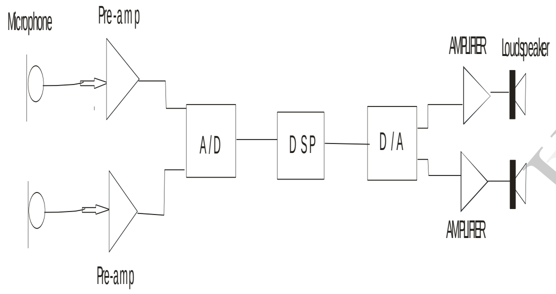

Fig 2: The structure of a digital hearing aid.

Fig.2 simply shows the components of hearing aid as containing the microphone, pre-amplifier, Analog-to-Digital converter, DSP chip, Digital-to-Analog converter, post amplifier and the loud speaker.

*B. **NLMS Algorithm***

LMS algorithm is a type of adaptive filter known as stochastic gradient-based algorithms as it utilizes the gradient vector of the filter tap weights to converge on the optimum wiener solution [6]. It is well known and widely used due to its computational simplicity. There are three main reasons why LMS algorithm has been accrued to be popular. Firstly, LMS algorithm has ease of implementation in both software and hardware due to its computational simplicity and efficient use of memory. Secondly, it's robust performance in the presence of numerical errors caused by finite-precision arithmetic. Thirdly, its behavior has been analytically characterized to the point where a user can easily set up the system to obtain adequate performance with only limited knowledge about the input and desired response signals. Nevertheless, LMS has been disadvantaged by having a fixed step size parameter, μ, for each of the iteration. Consequently, NLMS algorithm as an extension of LMS algorithm has been extensively used in eliminating the limitations of LMS algorithm. Simply because normalized least mean square is found to be more stable, robust and adaptable in both stationary and non-stationary environment due to the variable nature of its step size. It is well known that the LMS type algorithms can only minimize the current estimate error to some extent [7]. The critical issue associated with all algorithms is the choice of the step-size parameter that is the trade-off between the steady-state mis-adjustment and the speed of adaptation [8]. However, this trade-off between the convergence rate and the steady-state mis-adjustment in LMS algorithm has been addressed using variable step-size in NLMS algorithm [9,10].

III. SYSTEM MODELS AND ANALYSIS

A. Derivation of Normalize Least Mean Square algorithm.

In NLMS algorithms, practical implementation is very similar to that of the LMS algorithm. Each of the iterations of the NLMS algorithm requires these steps in the following order.

1. The output of the adaptive filter is calculated.

$$y(n) = \sum_{k=0}^{N-1} w_k(n)x(n-k) = wH\,xn \qquad (1)$$

Where

$x(n) = [\,x(n), x(n-1), \ldots\ldots, x(n-k+1)]^H$ an input signal vector (2)

2. An error signal is calculated as the difference between the desired signal and the filter output.

e (n) = d(n) –y(n) *% error signal* *(3)*

$\xi = e\{d[n] - y[n]^2\}$ % w[n] is selected to minimize the MSE *(4)*

3. The filter tap weights are updated in preparation for the next iteration.

w[n+1] =w[n] + μe[n]x[n] *(5)*

Normalization of equation 5 changes to

$$w(n+1) = w(n) + \frac{\mu_n x(n)e(n)}{\gamma + x^H(n)x(n)} \qquad \%$$

Update equation of NLMS algorithm

γ is a constant term for normalization and is always less than 1. The inclusion of the parameter, γ, in the coefficient update is to avoid large step size when $x^H x(n)$ becomes small or to avoid division by zero in normalization operation.

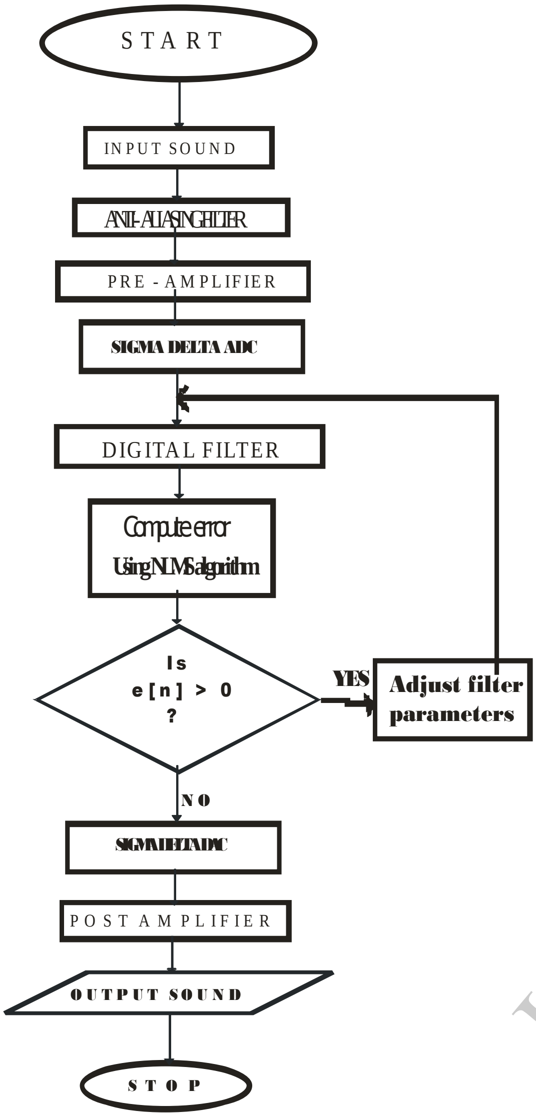

Fig.3: Flow chart diagram of hearing aid using NLMS algorithm.

Fig.3 shows the workability of hearing aid which ideally summarizes the total operation/the working principle of a hearing aid when fully implemented. The first operation of the hearing aid is to collect the human voice which is analog in nature using the microphone. The microphone changes the analog nature of this voice to electrical voltage and as well provides the first stage amplification. Since the input signal is still in analog form, the gain of this signal is amplified with noise. During sampling, the Nyquist criterion theory is therefore obeyed to prevent aliasing. After changing the analog signal to digital world using the A/D sigma delta converter, the digital signal is therefore extensively manipulated using the adaptive filter whose algorithm has been burnt on a DSP chip to perform digital signal filtering and the corresponding update. If no error is detected, the signal advances to sigma delta D/A converter for Digital - to - Analog conversion. But if error is detected (i.e e(n) > 0), then the e(n) will therefore trigger the adaptive algorithm to correct the mis-adjustment of the next iterated signal such as the gain .When the error has been minimized, the filtered signal now in its original form finally undergoes post amplification using the loud speaker to output sound. This ends the whole algorithmic process.

B. Simulink Model Block Development of NLMS Algorithm.

In this work, an analytical method was adopted in evaluating the performance of the algorithm. The model of an NLMS algorithm was designed and developed in Simulink as shown in Fig.4. The Simulink program provided an in-built block sets from its library needed for the design. The major block set called Normalized LMS block performed both adaptive filtering and estimation of the filter coefficient. The simulator provided submenu for configuring the block settings to be able to perform the required task such as variation of the step size value, the filter length and the type of algorithm. Also notice in fig. 5, that the input signal was generated using the sine wave block and the noise signal generated with a random source block which generates the Gaussian noise signal. Thereafter, the corrupted signal is then passed onto the digital filter for digital signal processing. The output signal was later compared to the desired signal for error detection and determination of rate of convergence. Any difference in the signal triggers the adaptive algorithm for automatic adjustment of its filter parameters so as to minimize error until the desired signal becomes approximately equal to the filter output. At this point, it was assumed that the minimum mean square error has been achieved. The evaluation was made possible by the time scope block which shows the input signal, input + noise signal, filter output and the error signal, while the display block shows the adaptive weight of the filter.

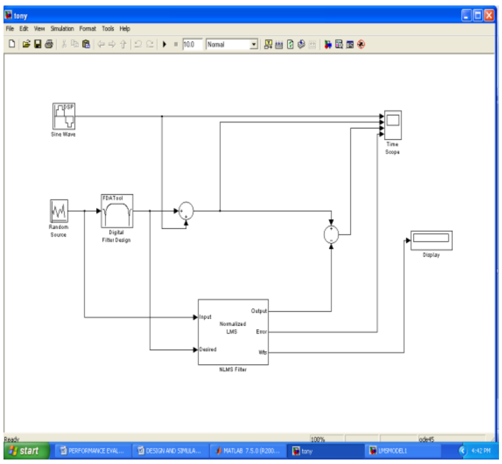

Fig.4: Simulink block model of NLMS algorithm.

IV. SYSTEM IMPLEMENTATION AND RESULT ANALYSIS

The simulation of the NLMS algorithm was carried out with the following specifications: filter length, L = 11, filter order = 10, leakage factor = 1.0, Initial filter weight = 0, step size, μ = [0.005, 0.05, 0.075, and 1.5, 1.9]. The filter output and the error signal will be examined at each stage of the simulation to observe their behaviour towards MSE reduction and high speed of convergence when the step size value is either increased or decreased. The simulation was divided into two: simulation involving NLMS and secondly that of LMS algorithm. Thereafter, their results were evaluated for better performance.

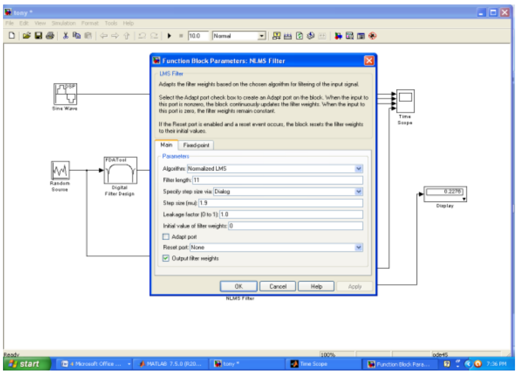

Fig.5: Function block parameter: NLMS algorithm

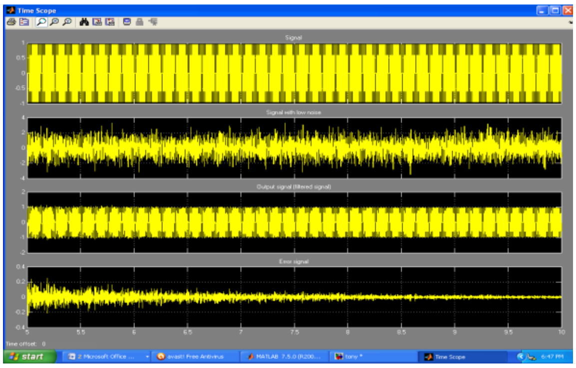

Fig.6: NLMS filtered output for μ = 0.005.

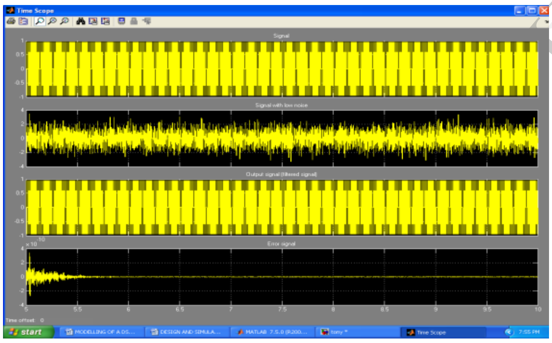

Fig.7: NLMS filtered output for μ = 0.05.

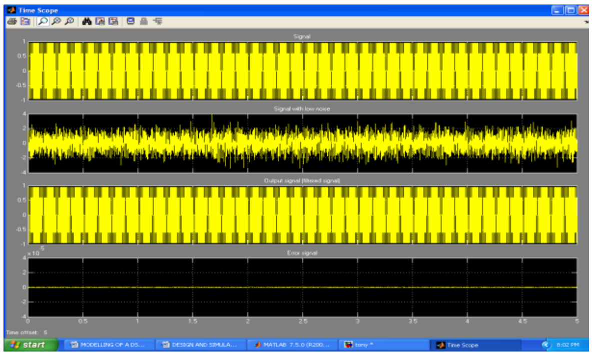

Fig.8: NLMS filtered output for μ = 0.075.

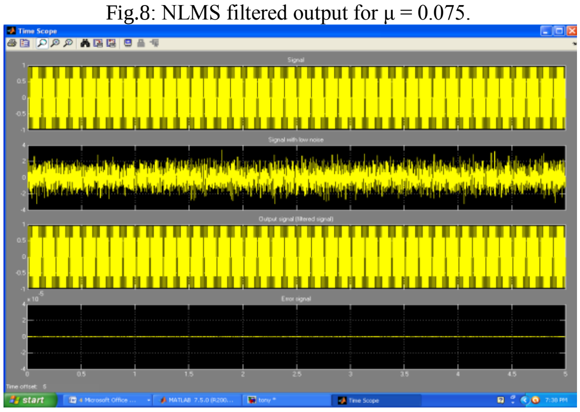

Fig.9: NLMS filtered output for μ = 1.5.

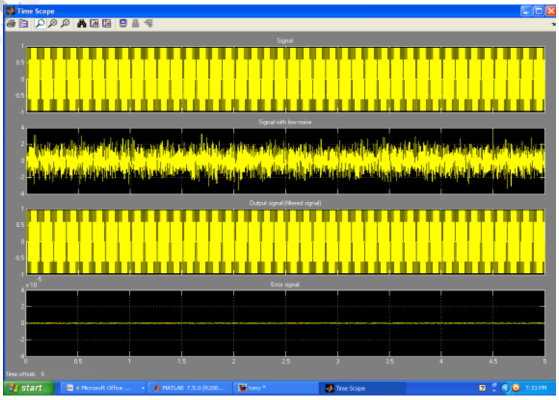

Fig.10: NLMS output for μ = 1.9.

Table1: Summary of Simulation Result

Step size (μ)	Time (ms)
0.005	8.5
0.05	5.5
0.075	No error
1.0	No error
1.5	No error
1.9	No error

Table 2: Step size, (μ) vs Adaptive weight, w (n)

Adaptive weight w(n)	μ = 0.005	μ = 0.01	μ = 0.075	μ = 1.5	μ = 1.9
W(0)	0.000168	$9.065e^{-006}$	$-1.274e^{-016}$	-1.717e^-16	6.036e^-17
W(1)	-0.08725	-0.0882	-0.08822	-0.08822	-0.08822
W(2)	-0.000623	$4.5e^{-006}$	$3.942e^{-017}$	5.993e^-17	2.156e^-16
W(3)	-0.291	-0.294	-0.294	-0.294	-0.294
W(4)	0.582	0.5883	0.5884	5.756e^-17	-6.97e^-17
W(5)	0.0004023	$-7.818e^{-006}$	$1.37e^{-017}$	0.5884	0.5884
W(6)	-0.291	-0.294	-0.294	6.412e^-17	-4.645e^-17
W(7)	-0006431	$5.323e^{-006}$	$2.336e^{-017}$	-0.294	-0.294
W(8)	-0.08723	-0.0882	-0.08822	3.551e^-17	-7.718e^-19
W(9)	$9.704e^{-0.05}$	$5.918e^{-006}$	$-0.1.017e^{-016}$	-0.08822	-0.08822
W(10)	$9.704e^{-0.05}$	$5.918e^{-006}$	$-0.1.017e^{-016}$	-3.558e^-17	5.2e^-17

Table1 simply indicates the reading obtained from the graph of the simulation. The readings showed the time taken for the output signal to converge in milliseconds. Table 2 simply shows that adaptive weight of the filter at a given step size. These readings were obtained from the display block. Observe also that there was gradual reduction of the instantaneous square error as a result of the convergence property possessed by the NLMS algorithm as the step size approaches 1.0. Although the speed of convergence was reluctantly slow initially but experienced later significantly improved when the variable step size increased progressively from 0.005 to 1.9 respectively. At the step size value of 0.075, the faster speed of convergence was obvious since the algorithm had already converge before approaching 0secs as demonstrated in Fig.8 – 10 respectively. At this point, the desired signal d(n) is assumed to be approximately equal to y(n) since e(n) is zero. This tends to prove the faster speed of convergence of NLMS algorithm at an increased value of the step size. Note that the w(n) value can never be the same in all environment as observed during simulation. This is so because the filter keeps track of the filter adaptive weight at any instant in time. As result of the tracking nature of the adaptive filter it can be applied efficiently in the implementation of digital hearing aid. This is one of the limitations of LMS algorithm.

V. CONCLUSION

The modelling of an adaptive filter using NLMS algorithm for implementation of a DSP based hearing aid was successively designed and developed in Matlab for evaluation of the speed of convergence of the algorithm for instantaneous square error reduction. The simulation results showed a high of speed of convergence of the normalized least mean square algorithm; even though the speed of convergence was slow initially. The results tend to prove the efficient use of NLMS algorithm in the implementation of digital hearing aid towards noise reduction as the user changes from one environment to another.

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