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Design & Implementation of Adaptive Filtering Algorithm using NLMS having Diffferent Targets

Sunita Rani, Suman Rani

***Abstract*— this paper presents a review of adaptive algorithms that is LMS (Least mean square) Algorithm and NLMS (Normalised least mean square) algorithm. The adaptive filters NLMS (Normalized Least Mean Square) filter, is the most widely used and simplest to implement. NLMS algorithm has low computational complexity, with good convergence speed which makes this algorithm good for echo cancellation. It has minimum steady state error. Recently, adaptive filtering Algorithms have a trade-off between complexity and the convergence speed. In this, it presents an NLMS filter with different target filters such as FIR and IIR. Also effects of parameters like step size, frequency will also find out. Three performance criteria are used in the study of these algorithms; the minimum mean square error, convergence rate and complexity. Comparison of LMS and NLMS filter will also be proposed. We will use MATLAB for simulation of Adaptive filters.**

***Index Terms*—Adaptive Filters, NLMS filters, LMS filters, Different targets in adaptive filters.**

I. INTRODUCTION

DSP (Digital signal processing) is one of the technical fields that demands high speed and low power digital filters. To meet these requirements, the order of digital filter should be as small as possible .DSP is used in many applications such as image processing, speech processing, biomedical and military application, Echo cancellation, Robotics, Cellular telephone and power line monitors. Digital filter is very important class of linear time invariant system that is used to remove unwanted signal such as noise or echo signal. Digital filter is used because it has advantages over analog filter such as easier storage and maintenance, higher flexibility and minimum effect of interference noise [1].

Digital filter are classified in two types such as FIR filter and IIR filter. FIR Filter has finite impulse response and no feedback is required. IIR filter has infinite impulse response and depend upon present and past input values. These filters have some advantages and disadvantages. FIR filters are more stable and require more storage for high filter order coefficient. Whereas IIR filters are low filter order and become unstable [2]. Linear filtering is required in a variety of applications. A filter will be optimal only if it is designed with some knowledge of input data. If this knowledge of information is not known, then filter is called as adaptive filter.

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Sunita Rani, M.Tech Student, ECE Department, Prannath Parnami institute of tech. and Science, Hisar, India
Suman Rani, HOD, ECE Department, Prannath Parnami institute of tech. and Science, Hisar, India.

Adaptive signal processing is more popular due to high flexibility and accuracy in the communication fieldMost popular algorithm is LMS and NLMS algorithm. NLMS algorithm has many advantages over LMS like low computational complexity, good convergence speed and minimum steady state error.

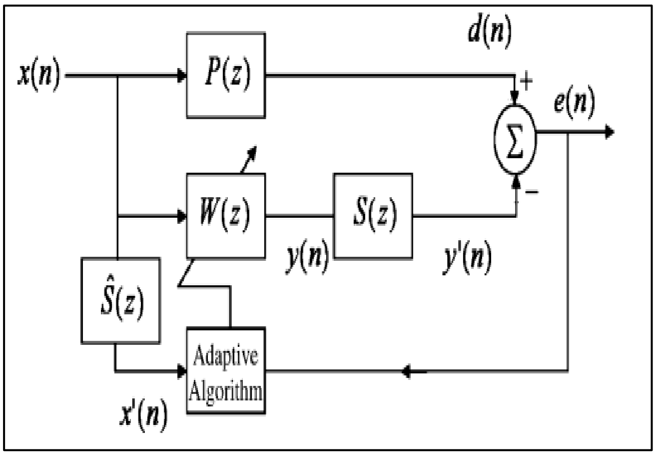

Figure 1: Feed-Forward NLMS Algorithm [11]

The designing of digital filter requires the approved specification with fixed coefficients. If this description is time changing or not accessible then this problem can be manipulated by digital filter with adaptive quantities, which is known as adaptive filter. The adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Because of the difficulty of the optimization procedures, most adaptive filters are digital filters. Adaptive signal processing has been introduced and its growth to the advanced related fields of digital computing, DSP and high speed joined circuit technology has been made rapidly.
The designing of digital filter requires the approved specification with fixed coefficients. If this specification is time varying or not accessible then this problem can be manipulated by digital filter with adaptive coefficients, which is known as adaptive filter. To design Adaptive filters, LMS, NLMS and RLS algorithm is used. Many other algorithms have been developed based on Linear Programming (LP), Quadratic Programming and Heuristic methods in Artificial Intelligence (AI). Remez Exchange Algorithm (to design equiripple filter) and linear Programming (to design adaptive filter) are optimum in the sense that these methods achieve both a given discrimination and a specified selectivity with a minimum length of the filter impulse response.
This paper is organised as follows. In section 2nd, we discuss related work with filters. In section 3rd it describes the adaptive filter. In section 4th, it describes the adaptive filter

algorithm for analysis the Digital filter and measure performance in terms of parameters. Finally conclusion is given in section 5th

II. RELATED WORK

In literature, it proposed FIR filter design method which used the NLMS (Normalised least mean square) adaptive algorithms system identification ability. In this method, any filters including FIR, IIR and analog filters can copy its response to design the desired FIR filters .The FIR Filter generated by this method can have exact amplitude and phase response. With a target FIR filter which has a smaller or equal length or obtain same amplitude response as an IIR or analog filter. When target filter is FIR with a smaller or equal length than the NLMS adaptive filter with a step variable u, then the adaptive filter can be converted to have both exact amplitude and phase response of the target filter.

The comparison between adaptive filtering algorithms that is least mean square (LMS) normalized least mean square (NLMS) Recursive least square(RLS) according to computational complexity and signal to Noise ratio(SNR). The result of discussion is that NLMS is more stable than LMS but less stable than RLS. But NLMS had improved SNR [4].

Interference and noise can be cancelling out by using adaptive algorithm. The aim of interference cancellation is to obtain estimation of interfering signal and subtract from the corrupted signal and hence obtain a noise free signal. The effect of filter length and step size has been analysed. LMS and NLMS algorithm is analysed for equalization in terms of symbol error rate. A noise robust optimal step size frequency domain LMS algorithm is used for estimating the equalizer co-efficient. The step size ambiguity of the LMS algorithm is solved by NLMS algorithms [5].

Author had discussed the simulation of Low Pass FIR Adaptive filter using least mean square (LMS) algorithm and least P_{TH} norm 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 optimal wiener solution whereas Least P_{TH} does not need to adapt the weighting function involved and no constraints are imposed during the course of optimization [6].

Some had discussed the comparison between adaptive filtering algorithms that is least mean square (LMS), Normalized least mean square (NLMS), Recursive least square (RLS). Execution aspects of these algorithms, their computational complexity and Signal to Noise ratio are examined. Here, the adaptive behaviour of these algorithms is analysed. Recently, adaptive filtering algorithms have a nice trade-off between the complexity and the convergence speed. The study of all these algorithms cover three performance criteria: the minimum mean square error, the algorithm execution time and the required filter order [7].

Efficient utilization of limited radio frequency spectrum is only possible to use smart antenna system. LMS and NLMS algorithms are beam forming algorithms. Smart antenna uses these algorithms in coded form which calculates complex weights according to the signal environment. The efficiency is calculated on the basic of normalized assay factor and mean square error (MSE) [8].

III. ADAPTIVE FILTERS

The filter design process consists of two parts:

- the approximation problem
- the realization problem

The approximation problem deals with the choice of parameters or coefficients in the filter's transfer function. The realization part of the design problem deals with choosing a structure to implement the transfer function [9].

The approximation stage can be divided into 4 steps:

- A desired or ideal response is chosen (usually in the frequency domain)
- A class of filters is chosen (for example, FIR or IIR)
- A design criteria is chosen (least square or minimax method)
- An algorithm is selected to design the transfer function

The realization stage can also be divided into 4 steps:

- A set of structures is chosen
- A criteria for comparing different implementations is chosen
- The best structure is chosen, and its parameters are calculated from the transfer function
- The structure is implemented in hardware or software [4].

Adaptive filter use the algorithm by which itself adjust the transfer function. It enables the filter to produce an output which is same as the output of an unknown system. It removes the problem of weiner filter. It is totally based on stochastic approach. Adaptive filters works on the principle of minimizing the mean square difference that is, error between the filter output and designed signal. The error signal can be generated by the output of the programmable variable coefficient digital filter subtracted from a reference signal [10].

Adaptive filters are made up of FIR and IIR filters. FIR adaptive filters are mostly used due to the stability for any set of fixed coefficient. The algorithms for adjusting the coefficient of FIR filter are simpler in general than those for adjusting the coefficients of IIR filter.

A. *Filter Designing Steps*

There are mainly five steps involved in designing of a digital filter:

(1) Filter Specification: This may involve defining of filter type e.g. low pass, desired amplitude, phase response, sampling frequency, word length of input data.

(2) Coefficient Calculation: This may involve determination of coefficients of transfer function (z), which may satisfy desired specifications.

(3) Realization: This step include in converting transfer function into suitable filter network.

(4) Analysis of finite word length effects: In this step, analysis of the effects of quantizing the filter coefficients as well as effect of carrying out the filtering operation using fixed word lengths on filter performance take place.

(5) Implementation: This step involve in producing the software code and/or hardware and obtaining the actual response.

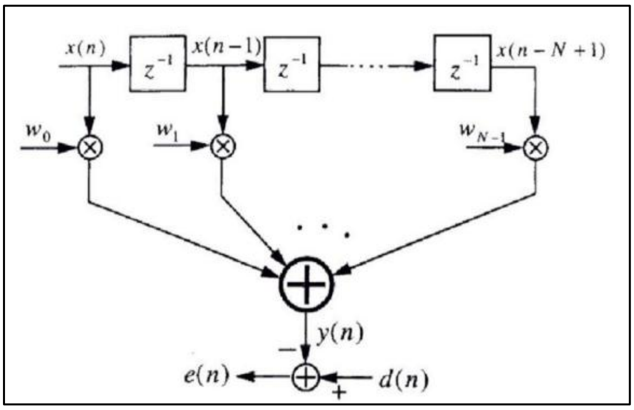

Figure 2: General Block Diagram of Adaptive Filter [11]

B. Filter Designing Methods

There are various methods to design FIR filter as follow:

- Fourier series method
- The window method
- Frequency sampling method
- Optimal filter design method

In **Fourier series method**, we first decide the continuous frequency response H(f) i.e. low pass, high pass. Then Fourier series of this frequency response found using following equation [11]

$I(k)(2/f_s)\int H(f)\cos((2^{*}\,pi^{*}k^{*}f)f_s))).\,df$ (1)

But problem of such types of filters is that these methods causes to Gibb's oscillations at cut off frequency region. So these filters are not accurate enough for practical applications. In window design technique, the desired frequency response specification $H_d(w)$, corresponding unit sample response $h_d(n)$ can be expanded in a Fourier series and given by:

$H_d(w)\sum_{n=-\infty}^{n=\infty} h_d(n)\exp(-jwn)$ (2)

$h_d(n)$ are Fourier coefficients having infinite length. So it must truncated at some points say n= M-1 to obtain a realizable FIR filter. The truncation of $h_d(n)$ can be done by multiplying $h_d(n)$ with any desired window function. Although truncated window method is simple and easy to design but not optimal i.e. minimum order cannot be achieved. Also, it is always not possible to have a closed form expression for $h_d(n)$ [2]. Different types of windows are used in DSP have their different weighting functions. These are:

1. Rectangular window :- The window function for this window is
   $W_R(n) = 1, \text{for}|n| \le \frac{M-1}{2}$
   0, otherwise (3)
2. Hamming window :- The causal function for hamming window is expressed by
   $W_H(n) = 0.54\text{-}0.46\cos\frac{2\pi n}{M-1}, 0 \le n < M\text{-}1$
   0, otherwise (4)
3. Hanning window :- The window function of causal hanning window is given by
   $W_{Hann} = 0.5\text{-}0.5\cos\frac{2\pi n}{M-1}, 0 \le n \le M\text{-}1$
   0, otherwise (5)
4. Blackman window :- The window function of this window is
   $W_B(n) = 0.42\text{-}0.5\cos\frac{2\pi n}{M-1} + 0.08\cos\frac{4\pi n}{M-1}, 0 \le n \le M\text{-}1$ (6)

III. ADAPTIVE FILTERS ALGORITHMS

An adaptive algorithm is procedures for adjusting the parameters of an adaptive filter to minimize an error function that includes the input, reference and filter output signal. Adaptive algorithm can be consist of three parts: the definition of minimizing algorithm, the definition of objective function and the definition of error signal. There are many adaptive filter algorithm used in DSP for adjusting the co-efficient of the adaptive filter like LMS (least mean square) algorithm and NLMS(normalized least mean square) algorithm.

The general form of an adaptive filtering algorithm is

$w(n+1) = w(n)+\mu(n)G(e(n),x(n),y(n))$ (7)

Where G(.) is a particular vector valued non-linear function. $\mu(n)$ is a step size parameter, e(n) and x(n) are the error signal and input signal vector respectively and y(n) is a vector of states that store at previous time instants.

As the NLMS is an extension of the standard LMS algorithm, its practical implementation is very similar to that of the LMS algorithm except that the NLMS algorithm has a time varying step size $\mu(n)$. This step size can improve the convergence speed of adaptive filter. Each iteration of the NLMS algorithm requires these steps in the following order [11]. The only difference with respect to LMS is the coefficient updating step (4).

A. LMS Algorithm

The LMS (least man square) algorithm was first developed by widrow and Hoff in 1960 [12].The LMS algorithm is a stochastic gradient algorithm that iterates each tap weight in the filter in the direction of the gradient of the squared amplitude of an error signal with respect to that tap weight. Three steps are involved in every iteration of LMS algorithm as:-

1. The output of the FIR filter, y(n) is calculated using equation

$y(n)= w^T(n)x(n)$ (8)

2. The value of the error estimation is calculated using equation

$e(n)=d(n)-y(n)$ (9)

3. The tap weights of the FIR vector are updated for next iteration by equation

$W(n+1)=w(n)+2\,\mu e(n)\,x(n)$ (10)

LMS algorithm is most widely used due to its computational simplicity. It has the fixed step size with upper bound and lower bound as

$0<\mu<2/\lambda_{max}$

B. NLMS Algorithm

The main limitation of LMS algorithm is that it is sensitive to scaling of its input x(n) which makes it hard to select a step size μ that makes stability of the algorithm. NLMS (normalized least mean square) algorithm also is modified form of LMS algorithm by normalising with power of input with time varying step size [1]. In each iteration of the NLMS algorithm requires three steps in the following order:-

1. The output of the adaptive filter is calculated.

$$y(n) = w^T(n)\, x(n) \quad (11)$$

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

$$e(n)= d(n)-y(n) \quad (12)$$

3. The step size value is calculated from the input vector
4. The filter tap weights an updated in preparation for the next iteration

$$w(n+1)=w(n)+2\mu e(n)x(n) \quad (13)$$

NLMS algorithm has greater stability with unknown signals. It has also good convergence speed and relative computational simplicity.

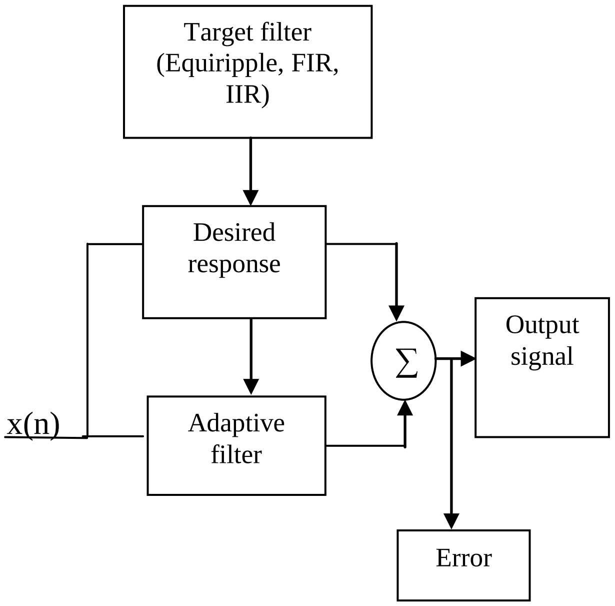

Figure 3: Block Diagram of Proposed System [1]

Block diagram of proposed system is given in fig. 3. Here we first take a sinusoidal input signal x(n) that is combined with desired response to give output d(n). This desired response is the output of different target filters used in our designing process. These target filters are equiripple filter, FIR filter, Butterworth IIR filter. The desired response d(n) will combine with the adaptive filter response h(n) to give final output. Then we will find the error in the output signal.

C. Performance Parameters

RMSE:
It is a measure of the difference between value predicated by estimation and value actually observed from the thing being estimated.
Convergence Rate:
The convergence rate is defined as the number of iterations required for the algorithm to converge to its steady state mean square difference that is error.
Complexity:
Computational complexity is the measure of the number of arithmetic calculation like Multiplications, addition and subtraction for different adaptive algorithm.

V. CONCLUSION

In this paper, it proposes the designing and implementation of adaptive filter. The adaptive filter used is NLMS. Here it will use three different target filters FIR, IIR and multiband Equiripple filter. Also covers the effects of stationary signals on the performance of adaptive filters. We will test the signal with variation in step size, filter order and sample frequency. The effects of changes in parameters will be noted within a specific filter and later a comparison between the filters will be done. All simulations will be done with the help of MATLAB tool.

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