

Adaptive Noise Elimination

Pranavi Mhatre*, Vivek Ramakrishnan**

**(Department of Electronics & Telecommunications, SIES GST Nerul, Mumbai University, Navi Mumbai-400706)*

***(Department of Electronics & Telecommunications, SIES GST Nerul, Mumbai University, Navi Mumbai-400706)*

Corresponding Author: Pranavi Mhatre

ABSTRACT

An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Because of the complexity of the optimization algorithms, almost all adaptive filters are digital filters. Adaptive filters are required for some applications because some parameters of the desired processing operation (for instance, the locations of reflective surfaces in a reverberant space) are not known in advance or are changing. The closed loop adaptive filter uses feedback in the form of an error signal to refine its transfer function. Generally speaking, the closed loop adaptive process involves the use of a cost function, which is a criterion for optimum performance of the filter, to feed an algorithm, which determines how to modify filter transfer function to minimize the cost on the next iteration. The most common cost function is the mean square of the error signal. As the power of digital signal processors has increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, camcorders and digital cameras, and medical monitoring equipment. The adaptive filter is being implemented using MATLAB and the noise is eradicated successfully.

Keywords - Adaptive Filters, Closed loop adaptive filter, Digital Filters, MATLAB, Simulink

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I. INTRODUCTION

Adaptive filters are digital filters capable of self-adjustment. These filters can change with accordance to their input signals. An adaptive filter is used in applications that require differing filter characteristics in response to variable signal conditions. Adaptive filters are typically used when noise occurs in the same band as the signal or when the noise band is unknown or varies over time. The adaptive filter requires two inputs: the input signal and a noise or reference input. An adaptive filter has the ability to update its coefficients. New coefficients are sent to the filter from a coefficient generator. The coefficient generator is an adaptive algorithm that modifies the coefficients in response to an incoming signal. In most applications the goal of the coefficient generator is to match the filter coefficients to the noise so that the adaptive filter can subtract the noise from the signal. Since the noise signal varies, the coefficients must vary to match it, hence the name adaptive filter [1]. The digital filter is typically a special type of finite impulse response (FIR) filter, but it can be an infinite impulse response (IIR) or other type of filter. Adaptive filters have uses in a number of applications including noise cancellation, linear prediction, adaptive signal enhancement, and adaptive control which are variedly applied in

Acoustic environment modelling for Sonar applications [2]. Noise cancellation is also an important field where adaptive filtering is used. Adaptive modelling plays a significant role in control systems and signal processing. For signal processing, impulsive noise suppression is carried out using nonlinear M-filters, namely, median and myriad filters. And the algorithm used is called as a new Adaptive algorithm that is applicable for noise cancellation.

II. MOTIVATION

Adaptive filters are used in various real-time applications such as echo cancellation, noise cancellation, system identification and prediction. Field-programmable gate arrays (FPGAs) are also used most widely for applications where timing requirements are strict. Thus implementation of filter in real-time is needed. The objective of this paper is to design and implement an Adaptive filter which is robust to impulsive noise. The term adaptive filter implies changing the characteristic of a filter in some automated fashion to obtain the best possible signal quality in spite of changing signal/system conditions. Adaptive filters are usually associated with the broader topic of statistical signal processing. The operation of signal filtering by definition implies

extracting something desired from a signal containing both desired and undesired components. With linear FIR and IIR filters the filter output is obtained as a linear function of the observation (signal applied) to the input [1]. An optimum linear filter in the minimum mean square sense can be designed to extract a signal from noise by minimizing the error signal formed by subtracting the filtered signal from the desired signal. For noisy signals with time varying statistics, this minimization process is often done using an adaptive filter.

III. LITERATURE REVIEW

Basics of Adaptive Filter Theory, and what drawbacks it can overcome which are faced in other types of filters is explained in [1]. Adaptive filters are usually associated with the broader topic of statistical signal processing. For noisy signals with time varying statistics, minimization process is often done using an adaptive filter. Applications of adaptive filter in Sonar Applications is explained in [2]. These applications are variedly used in Acoustic environment modelling. Other applications were also cross referred. Significance of Adaptive Filters in Noise cancellation is explained in [3]. Also the significant role in control systems and signal processing of adaptive modelling. The methodology was studied in [4]. The entire system architecture is based on the LMS algorithm mentioned in the book.

A similar program of Linear Phase FIR Filter was studied and LMS algorithm program was developed accordingly and Execution was studied in [6]. A similar program of Linear Phase FIR Filter was studied and LMS algorithm program was developed accordingly. The MATLAB language was used to implement the program in MATLAB. The language fundamentals were studied in MATLAB Help options. This text offers a comprehensive treatment of MATLAB and its applications to the design and simulation of real, industry-standard circuits. It focuses on the use of Simulink rather than solely on the MATLAB language, showing why and how certain types of circuits are inferred from the language constructs and how any of the four simulation categories can be implemented. The text reviews fundamental concepts of digital electronics and design and includes a series of appendixes that offer tutorials on important design tools [1]. In Simulink, systems are drawn on screen as block diagrams. Many elements of block diagrams are available, such as transfer functions, summing junctions, etc., as well as virtual input and output devices such as function generators and oscilloscopes. Simulink is integrated with MATLAB and data can be easily transferred between the programs. In these tutorials, we will apply Simulink to the examples from the MATLAB tutorials to model the systems, build controllers, and simulate the systems. Simulink

is supported on Unix, Macintosh, and Windows environments; and is included in the student version of MATLAB for personal computers.

IV. OBJECTIVES

The main aim or objective of adaptive filter is to find and track the optimum filter quickly and accurately, corresponding to the same signal operating parameters. Also, the adaptive filter must satisfy all its characteristics such that the accuracy of output signal is good. The parameters which are needed to cancel in the system are noise, echo etc. and eliminate the problem of system modeling and adaptive line enhancement. Measuring the errors and neglecting them is one of the main objective of adaptive filters.

4.1 Advantages of Adaptive Filter

Adaptive filters have uses in a number of applications including noise cancellation, linear prediction, adaptive signal enhancement, and adaptive control which are variedly applied in Acoustic environment modelling for Sonar applications [2]. Noise cancellation is also an important field where adaptive filtering is used [3]. Adaptive modelling plays a significant role in control systems and signal processing. It is widely used in control tasks, especially in cases when system structure is known, but its parameters are poorly defined. Impulsive noise suppression in speech is very important for clear voice communications. For signal processing, impulsive noise suppression is carried out using nonlinear M-filters, namely, median and myriad filters. And the algorithm used here is a LMS algorithm that is applicable for noise cancellation.

4.2 Advantages of MATLAB.

MATLAB is used for the implementation of the Adaptive filter algorithm. A very large (and growing) database of built-in algorithms for image processing and computer vision applications. MATLAB allows you to test algorithms immediately without recompilation. You can type something at the command line or execute a section in the editor and immediately see the results, greatly facilitating algorithm development. The MATLAB Desktop environment, which allows you to work interactively with your data, helps you to keep track of files and variables, and simplifies common programming/debugging tasks. The ability to read in a wide variety of both common and domain-specific image formats. The ability to call external libraries, such as OpenCV. Clearly written documentation with many examples, as well as online resources such as web seminars ("webinars"). Bi-annual updates with new algorithms, features, and performance enhancements. If you are already using MATLAB for other purposes, such as simulation, optimization,

statistics, or data analysis, then there is a very quick learning curve for using it in image processing. The ability to process both still images and video. Technical support from a well-staffed, professional organization. A large user community with lots of free code and knowledge sharing. The ability to auto-generate C code, using MATLAB Coder, for a large (and growing) subset of image processing and mathematical functions, which you could then use in other environments, such as embedded systems or as a component in other software.

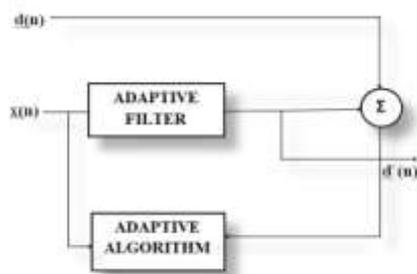
MATLAB doesn't require compiler to execute like C, C++. It just executes each sentence as it is written in code. This increase productivity and coding efficiency. It is higher level language. Using MATLAB Coder the codes written in MATLAB can be converted to C++, Java, Python, .Net etc. This makes this language more versatile. So, scientific theories can be implemented in other languages also. And those library files, or dll's can be directly implemented in other languages.

MATLAB has inbuilt rich library of Neural Network, Fuzzy Logic, Simulink, Power System, Hydraulics, Electrical, Communication, Electromagnetics etc. Thus developing any scientific simulation is easy to dousingsuchrichlibrary.

Disadvantage is its cost of License. It's very costly user has to buy each and every module and pay for it. Disadvantage is during cross compiling or converting MATLAB to other language code is very difficult. It's very difficult or requires deep MATLAB knowledge to deal with all errors. MATLAB is not suggested to make any product. Because, MATLAB doesn't create application deployment (installation) like task (like setup files and other executable which copies during installation).

V. METHODOLOGY

5.1 System Architecture



BLOCK DIAGRAM OF ADAPTIVE FILTER.

$d(n)$ - Filter Desired Output.
 $d'(n)$ - Filter Output.
 $e(n)$ - Filter Error.
 $e(n) = d(n) - d'(n)$

Fig 1. block diagram of adaptive filter

There are two input signals to the adaptive filter: 'd(k)' and 'x(k)' which are sometimes called the primary input and the reference input respectively. $d(n)$ - which includes the desired signal plus undesired interference and $x(n)$ - which includes the signals that are correlated to some of the undesired interference in $d(n)$. n - represents the discrete sample number.

5.2 LMS Algorithm

Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time. The idea behind LMS filters is to use steepest descent to find filter weights which minimize a cost function [4].

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The idea behind a closed loop adaptive filter is that a variable filter is adjusted until the error (the difference between the filter output and the desired signal) is minimized.

We have shown that adaptation equation (SD p, R) can be written in an equivalent form as (see also the Figure with the implementation of SD algorithm)

$$w(n+1) = w(n) + [E e(n)u(n)] \quad (1)$$

In order to simplify the algorithm, instead the true gradient of the criterion

$$w(n)J(n) = 2Eu(n)e(n) \quad (2)$$

LMS algorithm will use an immediately available approximation

$$w(n)J(n) = 2u(n)e(n) \quad (3)$$

Using the noisy gradient, the adaptation will carry on the equation

$$w(n+1) = w(n) + 2\mu w(n)J(n) = w(n) + u(n)e(n) \quad (4)$$

In order to gain new information at each time instant about the gradient estimate, the procedure will go through all data set $(d(1), u(1)), (d(2), u(2)), \dots$, many times if needed.

VI. MODELLING AND DESIGNING IN MATLAB

LMS algorithm does not require the autocorrelation matrix of the filter input and the cross correlation between the filter input and its desired

signal. It does not use the expectation operation that is present in the steepest descent method. Implementation of the algorithm is simple and it does not require matrix inversion. It includes a step size parameter which controls the stability and convergence of the algorithm. It is a stable and robust algorithm

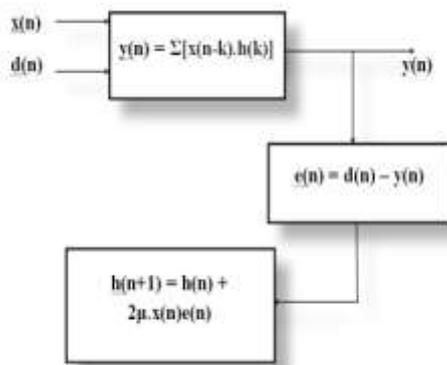


Fig 2: block diagram of lms algorithm

LMS algorithm does not use the exact values of the expectations, the weights would never reach the optimal weights in the absolute sense, but a convergence is possible in mean. That is, even though the weights may change by small amounts, it changes about the optimal weights. However, if the variance with which the weights change, is large, convergence in mean would be misleading. This problem may occur, if the value of step-size is not chosen properly [4].

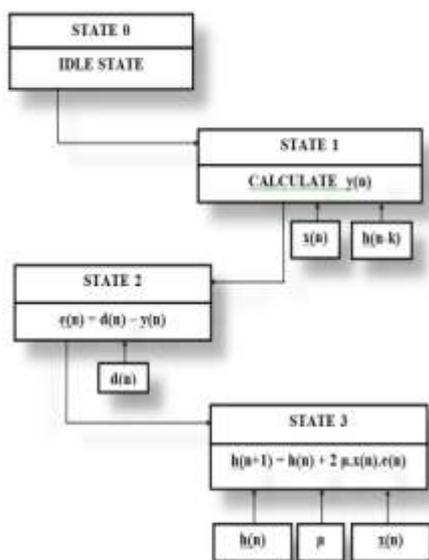


fig3: state diagram

VII. IMPLEMENTAION IN SIMULINK

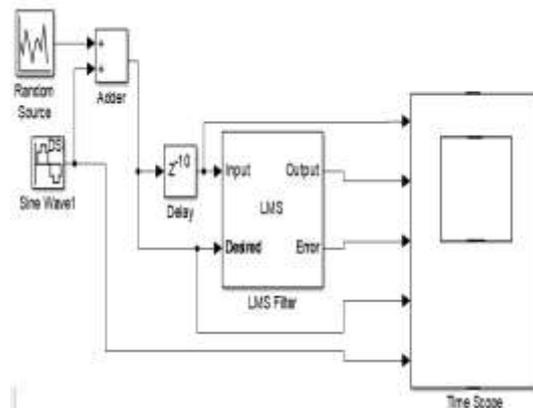


Fig 4. Simulink Model

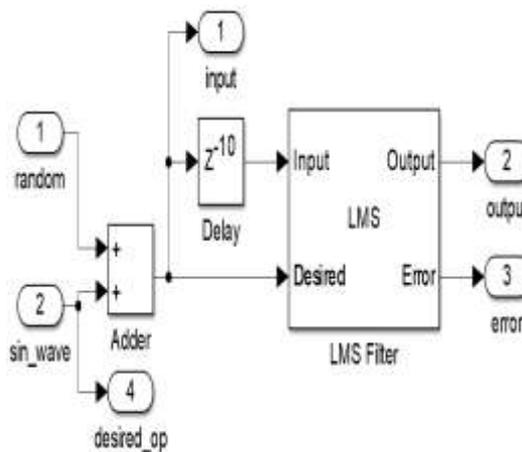


Fig 5. Matlab block in simulink

VIII. RESULTS IN SIMULINK

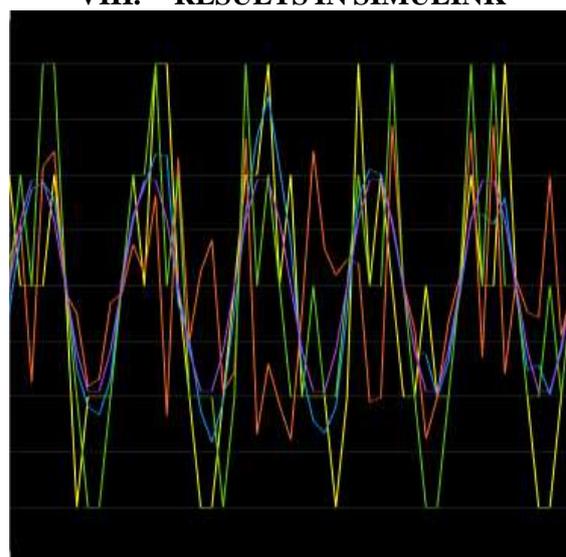


Fig 6. simulink waveform of the signals

Table 1. color code of signal

Color	Signal
Green	Input Signal + Noise
Yellow	Delayed Input Signal + Noise
Orange	Error signal
Blue	Output of LMS signal
Purple	Desired Sine wave

IX. RESULTS IN MATLAB

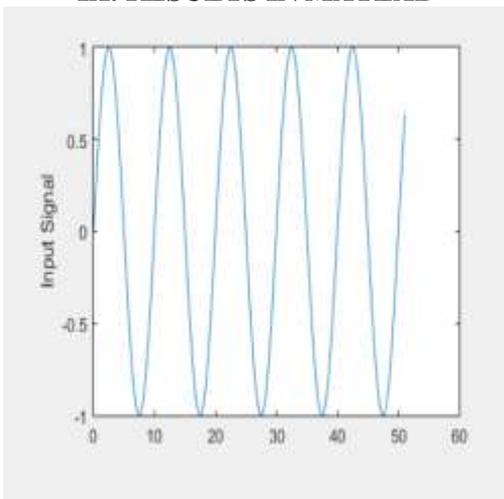


Fig.7.input waveform.

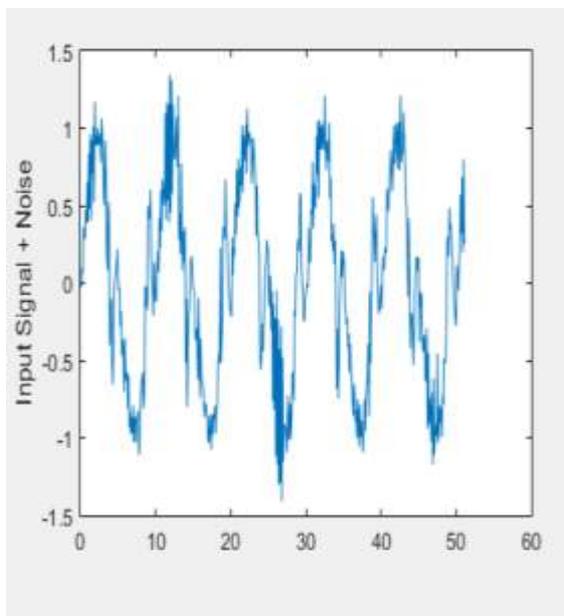


Fig.8. input waveform with noise

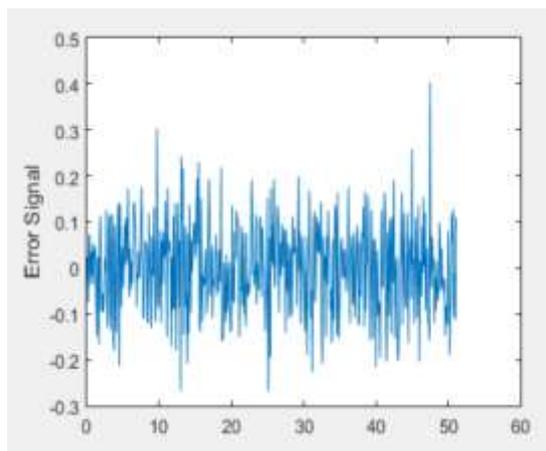


Fig. 9. error signal waveform.

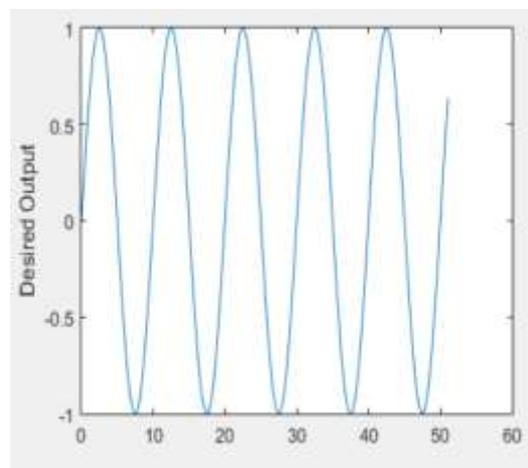


Fig.10. Desired Output Waveform.

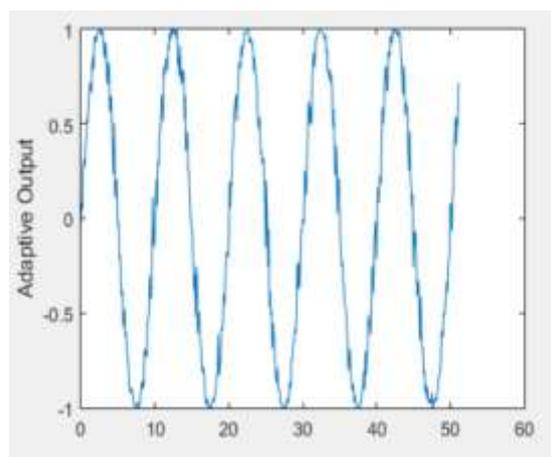


Figure 11. Adaptive noise eliminated waveform.

X. CONCLUSION

The implementation of a useful adaptive filter can be done using MATLAB. Furthermore, the MATLAB implementation is efficient and durable. Here we can reduce area consumption and also power consumption by reusing code for various stage of

computations. LMS algorithm also allows to operate on random quantities.

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