# Combinational Adaptive LMS Algorithms for Speech Enhancement- A Review

**Mr.P.C.Bhaskar[1] Ms.P.S.Gosavi[2]**
[1]Department of Electronics [2]Department of Technology
[1,2]Shivaji University, Kolhapur, India (MS)

*Abstract*— Speech enhancement is required for many applications in which clean speech signal is important for further processing. In this paper review has been taken for speech enhancement algorithms. When using LMS algorithm for speech enhancement, there is problem of performance. So Combinational approach will provide an interesting way to enhance adaptive filter performance.

***Key words:*** Adaptive filter, Least Mean Square(LMS), Speech Enhancement, Convex Combination

## I. INTRODUCTION

The Digital signal processing are having a great number of various applications, which uses advance adaptive algorithm techniques to enhance the performance. It could be applied in different fields such as telecommunications, radar, sonar, video and audio signal processing, and noise reduction. The design technique used and the adaptation algorithm will decide the performance of filter.

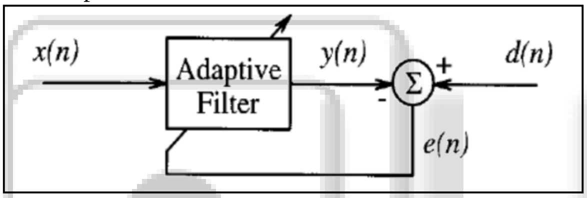


Fig. 1: Functional diagram of adaptive filter

The adaptive filter may be convex combination of two filters, such as single-input or multi-input filters, linear or nonlinear, and Finite-Impulse-Response FIR or Infinite-Impulse-Response (IIR) filters. The adaptation of the filter parameters is based on minimizing the mean square error in between desired signal and the filtered output. The most common adaptation algorithms are, Recursive-least-square (RLS), and the Least-mean-square (LMS), where RLS algorithm gives a more convergence speed compared to the LMS algorithm, when considering a computation complexity, the LMS algorithm keep maintaining its advantage. Because of easiness in computation, the LMS algorithm is commonly used in the design and implementation of integrated adaptive filters. The LMS digital algorithm is based on the gradient search according to the following equation:

$$w_{n+1} = w_n + \mu * e(n).x(n) \qquad - (1)$$

Where,

$w_n$= weights vector in moment n,

$w_{n+1}$ = weights vector in n+1,

x(n) = input signal simple vector, stored in the filter delayed line,

e(n)=corresponding filter error, which is the difference between the desired signal and the output filter's signal,

μ= filter's convergence factor.

The convergence factor μ, which directly proportional to the convergence speed and indirectly proportional to minimum error. The application depends on the adaptive filter configuration used. The differences between the configurations are given by the way the input, the desired and the output signals are used.

## II. A REVIEW ON SPEECH ENHANCEMENT ALGORITHMS

Rapid Advances in the digital communications, digital signal processing and VLSI technology has brought more attention to the adaptive least squares (LS) methods [1]. Linear & non-linear adaptive filtering algorithms are studied along with its application. A Least Mean Square (LMS) algorithm was first proposed by Windrow and Hoff [13].The Least Mean Square is computationally sensitive to Eigen value spread of input data.

Combinational approach is a useful way to enhance the performance of adaptive filters. Jerónimo [8] analyzed the behavior of one such approach, showing its universality in the sense that it performs best. Furthermore, when the component filters satisfy certain conditions, combination gives best result.

A fixed LMS algorithm produces minimum convergence rate and fixed steady state errors. So Balaram et.al[1] simulated adaptive FIR filter, based on variations in LMS, to generate better convergence rate and minimum steady state error compare to fixed LMS, and also obtains de noised signal at output, and calculate SNR values of Adaptive Filter with LMS algorithms and comparison is made among the LMS algorithms.

For combining the two NLMS filter outputs for each n, the combination need to select the scalar mixing parameter λ(n). This mixing parameter is decided using parametric sigmoidal function which having range between (0,1). The mean square analysis of this scheme has been studied in for two LMS adaptive filters By Bermudez et.al [8].

Guan Gui et.al[15] proved that combination of two filters is better than one filter, which deal with convergence speed and steady state MSE .That is more convergence speed leads to higher steady-state MSE and vice versa. Due to fixed step size, LMS filter unable the tradeoff between them. To overcome such problem, convex combination LMS filters, is more useful way. The first adaptive filter requires larger step size compared to second adaptive filter so that the combinational filter can achieve a good/fair trade-off between convergence speed and steady-state MSE.

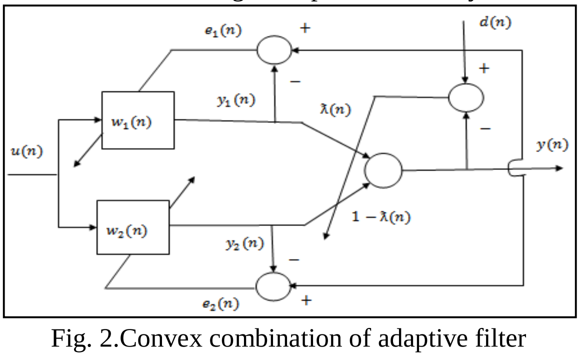


Fig. 2.Convex combination of adaptive filter

For speech enhancement an algorithm for automatic training of adaptive filters has been proposed by Sivaranjan Goswami et.al.[3] The algorithm has been implemented for NLMS adaptive filter. From there experimental work it seen that for the point at minimum Mean Square Error, the Signal to Noise Ratio is found to be increased by near about 1.5 times to the input SNR. The algorithm has been implemented in a software environment,

In certain applications, the tap length of the optimal filter may be unknown or even variable. So two modifications have been proposed by Zhang et.al.[9] to improve the performance of the convex combination algorithm. One of the modifications is given to take advantage of a fast filter to speed up the convergence of the slow one and second is to improve the convergence of the parameter..

For speech enhancement convex combination of two transversal filters is best way, was studied by A. Abdollahy et.al [7]. Researcher simulated for both fixed and variable step size cases, and simulation shows the variable step size algorithm is more effective interms of convergence rate and steady-state error.

Wu Caiyun [6] studied the speech enhancement technology. Initially researcher designed the WSLMS algorithm which can update weight value according to the variable input power. Then he combined two adaptive filters based on WSLMS algorithm, and presented the CC-WSLMS algorithm which can get an overall output of improved quality by mixing the output of several filters together.

All above research has simulated on various software. But there implementation on hardware is not gone so further in research. As the FPGAs can embed more functionality on a small silicon area without yielding any of the performance parameters like speed, accuracy. FPGAs can be viewed as a structural decomposition of an array of configurable logic blocks integrated together to form any complex digital systems. The programming procedure is same as its predecessors like microprocessor/DSP families but the dynamic-reusability and reconfigurability of its own individual hardware elements makes it most suitable for the embedded systems applications.

## III. Conclusion

This paper has reviewed the combinational adaptive LMS algorithm for speech enhancement. There are number of adaptive algorithms present in literature and every algorithm has its own properties. A review of adaptive filters shows that the LMS algorithm is famous for its high speed capability, high convergence rate and stable performance, but goal of every algorithm is to achieve minimum mean square error at a higher rate of convergence with lesser complexity. So that the combinational adaptive LMS algorithm is used to achieve minimum mean square error at a higher rate of convergence with lesser complexity that is less computational cost and ease of implementation & robust and reliable. The FPGA platform is well suited for the speech signal processing. The high-speed capability and register rich architecture of the FPGA is ideal for implementation of such convex combinational adaptive LMS algorithm.

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