

# Performance Evaluation of LMS, DLMS and TVLMS Digital Adaptive FIR filters by using MATLAB

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**Abstract** — This paper describes the comparison between adaptive filtering algorithms that is least mean square (LMS), Delay least mean square (DLMS), Time varying least mean square (TVLMS). Implementation aspects of these algorithms and Signal to Noise ratio are examined. Here, the adaptive behavior of the algorithms is analyzed. MATLAB/ Simulink is used to design. Noise Cancellation is a technique used for reducing undesired noise signal. Analysis of LMS, DLMS and Time-Varying Least Mean Square algorithm and comparison on the basis of various performance indices like MSE, SNR. The experimental results reveal that the LMS algorithm outputs are the best.

**Keywords** — Adaptive digital filters, LMS algorithms, DLMS algorithms, TVLMS algorithms, FIR filters.

## 1. Introduction

The most popular adaptive algorithms are the least mean square (LMS) algorithm. The least mean square algorithm is well – known for engineers involved in active noise control (ANC). It can be applied for the adaptation of the controller as well as for off line or on-line estimation of the relevant acoustic transfer functions. Because of it is proved to be a robust algorithm for adaptation of transversal digital filters used for different purposes in ANC system. But the LMS adaptive algorithm is practically used due to its simplicity and demonstrated efficient performance [5]. Now several algorithms are known FIR finding weights of linear functions in order to minimize E error and such among them is the linear adaptive filters

### 1.1 Least Means Square (LMS)

The least-mean-square (LMS) algorithm was developed by Widrow [1][2]. It is still used in adaptive signal processing for its simplicity, less computation, ease of implementation and good convergence property. The LMS algorithm is described by the equation:

$$e(n) = d(n) - XT(n)W(n) \text{-----(1)}$$

$$w(n+1) = w(n) + 2\mu(n)e(n)X(n) \text{-----(2)}$$

Where  $w(k)$  is filter coefficient at time  $k$ ,  $\mu(k)$  is step size,  $e(k)$  is adaptation error and  $x(k)$  are the filter input respectively. Equation (1) shows that LMS algorithm uses an adaptation error and a variable step size to update the filter coefficients. In this algorithm, the step size parameter  $\mu(n)$  is constant. The choice of this parameter  $\mu$  is very important to the convergence and stability. As in [2], the stability of the convergence of LMS algorithm requires the step size parameter  $\mu$  satisfy the condition

$$0 \leq \mu \leq \frac{1}{\text{tr}(R)} \leq 1 \lambda_{\max} \text{-----(3)}$$

Where  $\text{tr}(R)$  is the trace of the autocorrelation matrix of input  $x$  and  $\lambda_{\max}$  is the maximum eigen value of  $R$ . In

general a smaller step size leads to a small steady state misadjustment (SSM) but a slower convergence rate. While a larger step size gives faster convergence but a large SSM so, suitably select  $\mu$  based on equation (3). But it is still far from optimum trade-off between SSM and convergence rate. Then TVLMS algorithm is proposed to solve this problem. The TVLMS algorithm uses a suitable step size in the initial stage to speed up the convergence and the step size is adjusted to a smaller value gradually during the convergence.

In section 2, the TVLMS is discussed and their performance is introduced. In section 3, the new TVLMS algorithm is proposed. In section 4, the proposed algorithm is applied to noise cancellation system and simulation results are presented. Finally draws the conclusion.

The delay least mean squares (DLMS) algorithm is used to achieve lower adaption delay. It does not support the pipelined implementation. The digital signal processors (DSPs) are used for implementing different schemes of digital FIR filters. MATLAB [6] is a powerful language for technical computing the name MATLAB is known as “Matrix”. Laboratory, because its basic data element in a matrix (array) generally it is used in math computations modeling and simulations, data analysis and processing, visualization and graphics and algorithm development.

## 2. Adaptive Filter

Adaptive filters self learn. As the signal into the filter continues, the adaptive filter coefficients adjust themselves to achieve the desired result, such as identifying an unknown filter or canceling noise in the input signal. Figure 3.1 shows block diagram that defines the Inputs and Output of a Generic RLS Adaptive Filter. An adaptive filter designs itself based on the characteristics of the input signal to the filter and a signal that represents the desired behavior of the filter on its input. Designing the filter does not require any other

frequency response information or specification. To define the self-learning process the filter uses, The selection of the adaptive algorithm used to reduce the error between the output signal  $y(k)$  and the desired signal  $d(k)$  is done. When the LMS performance criterion for  $e(k)$  has achieved its minimum value through the iterations of the adapting algorithm, the adaptive filter has finished its work and its coefficients have converged to a solution. Now the output from the adaptive filter matches closely the desired signal  $d(k)$ . When there is a change in the input data characteristic, sometimes called the filter environment, the filter adapts to the new environment by generating a new set of coefficients for the new data. Notice that when  $e(k)$  goes to zero and remains there you achieve perfect adaptation, the ideal result but not likely in the real world.

### 3. Time Varying LMS (TVLMS) Algorithm

The basic idea of TVLMS algorithm is to utilize the fact that the LMS algorithm need a large convergence parameter value to speed up the convergence of the filter coefficient to their optimal values, the convergence parameter should be small or better accuracy. In other words, the convergence parameter is set to a large value in the initial state in order to speed up the algorithm convergence. As time passes, the parameter will be adjusted to a small value so that the adaptive filter will have a smaller mean squared error. A novel approach for the least-mean-square (LMS) estimation algorithm is proposed. The approach utilizes the conventional LMS algorithm with a time-varying convergence parameter  $\mu_n$  rather than a fixed convergence parameter  $\mu$ . It is shown that the proposed time-varying LMS algorithm (TVLMS) provides reduced mean-squared error and also leads to a faster convergence as compared to the conventional fixed parameter LMS algorithm. These algorithms have been tested for noise reduction and estimation in single-tone sinusoids and nonlinear narrow-band FM signals corrupted by additive white Gaussian noise. The study shows that the TV-LMS algorithm has a computation time close to conventional LMS algorithm with the advantages of faster convergence time and reduced mean-squared error.

### 4. Experimental Conditions

The speech signal with different combinations of noise signal is used for experimentation database has used. For the implementation and analysis of algorithms, different speech signal data corrupted with three level 0dB, 5dB and 10dB of noise is considered and experimentations are carried out. The speech signal that we use was “We find joy in the simplest thing.” Different noise signals include Airport noise, Babble noise, Car Noise, Exhibition Noise, Restaurant Noise, Station Noise, Street Noise and Train Noise with 0dB, 5dB and 10dB values.

Our work has two main types to explain main working in proper order. It consists of two ways to demonstrate the main situation.

#### 4.1 Training

In this stage we first input the the sound signal .It shown in the following wave type with desired sound . We can hear the sound.

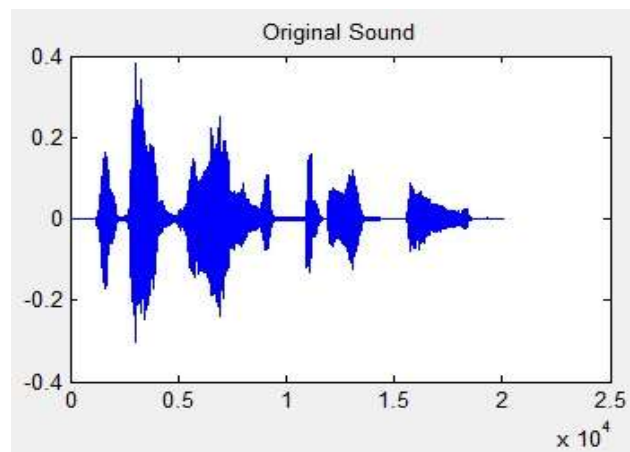


Figure 1: Original Sound wave

But during time when it mixes with noise, this sound could not hear clearly like this wave form.

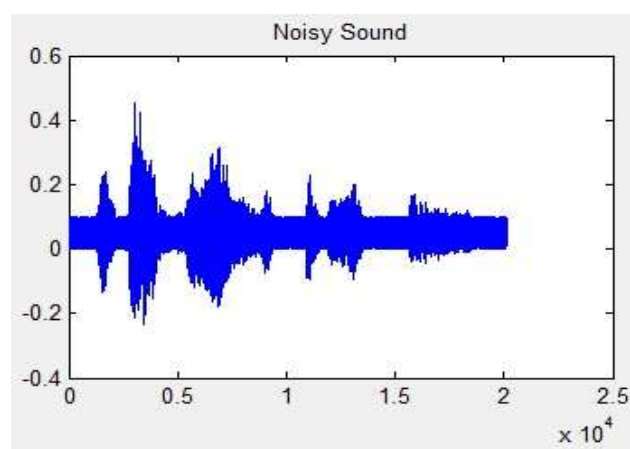


Figure 2: Noisy Wave

#### 4.2 Evaluation

At the last when we apply LMS , DLMS or TVLMS ,It shows the new sound that can be hear clearly with low noise . It can show in the following wave form.

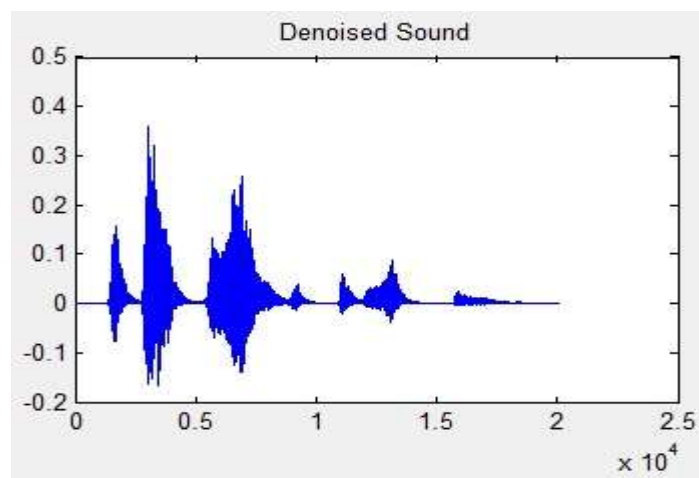


Figure 3: Denoised Sound

From this work we can understand the sound could get noisy due to the mixing of unwanted sound in it and could get noisy that could not hear clearly.

## 5. Simulation Results

It shows the following wave form.

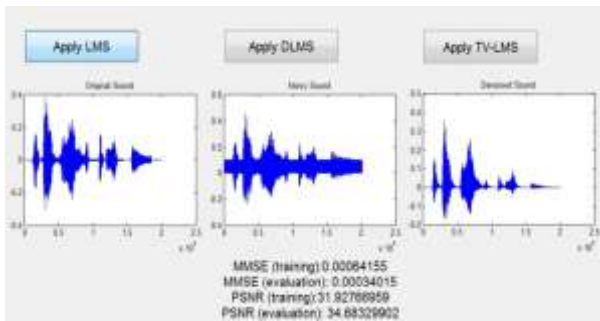


Figure 4: Result of LMS adaptive FIR filter

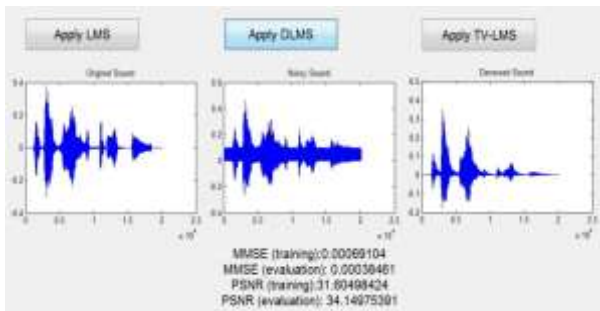


Figure 5: Result of DLMS adaptive FIR filters

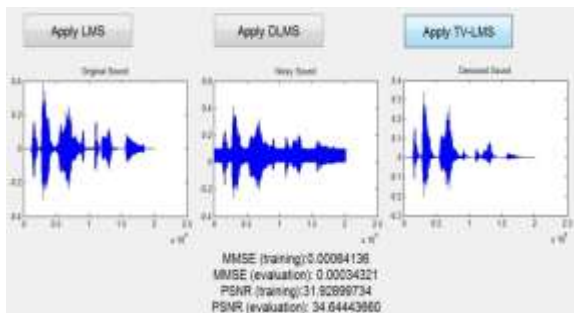


Figure 6: Result of TVLMS adaptive FIR filters

Table 1: Performance comparison of LMS, DLMS & TVLMS for different noise level for SNR during training

SNR	0db	5db	10db
LMS	28.47	30.44	31.26
DLMS	28.37	30.21	30.95
TV LMS	28.39	30.42	31.18

Table 2: Performance comparison LMS, DLMS & TVLMS for different noise level for SNR during evaluation

MSE	0db	5db	10db
LMS	31.21	32.97	33.64
DLMS	31.10	32.72	33.29
TV LMS	31.16	32.96	33.63

Table 3: Showing comparative performances of filters

Sound wave	Sound 1	Sound 2	Sound 3	Sound 4	Sound 5	
LMS	Training	31.92	28.45	30.44	31.24	31.75
	Evaluation	34.68	31.19	32.98	33.68	34.34
DLMS	Training	31.61	28.32	30.27	30.92	31.38
	Evaluation	34.14	31.11	32.74	33.29	33.83
TVLMS	Training	31.91	28.39	30.42	31.16	31.65
	Evaluation	34.64	31.19	32.97	33.62	34.26

From the above table it can be prove that the LMS algorithm is better than DLMS and TVLMS using MATLAB.

## 6. Conclusion

In this paper we presented a comprehensive comparative study between the LMS, DLMS and time-varying LMS (TV-LMS) algorithm. In a stationary white Gaussian noise environment with filter order  $M$  is set at a larger value (e.g.,  $M = 100$ ), simulations showed that the LMS algorithm provides the best MSE than the conventional DLMS and the TVLMS algorithms. However, when we choose to use a smaller filter order (e.g.,  $M = 10$ ), the LMS algorithm provides the best MSE performance as compared to TV-LMS and the conventional DLMS algorithms. Both the TV-LMS and the conventional LMS algorithms provide less computational time than the DLMS

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