

$$\alpha_n = C \left(\frac{1}{1 + a_n^b} \right)$$

Where C, a and b are constants.

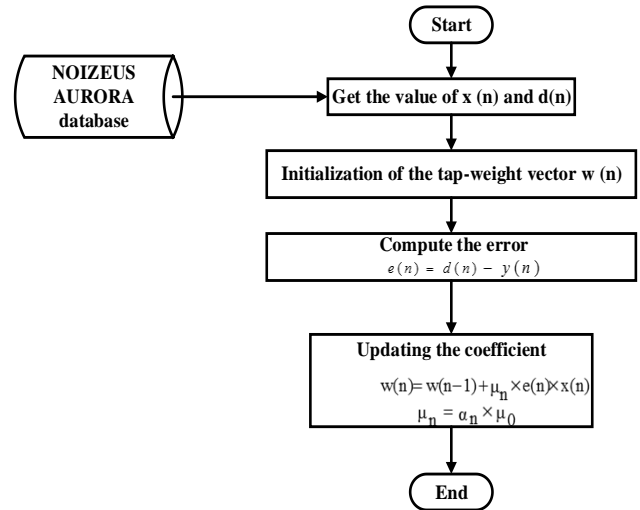


Fig. 3.9 Flow diagram for RLS algorithm

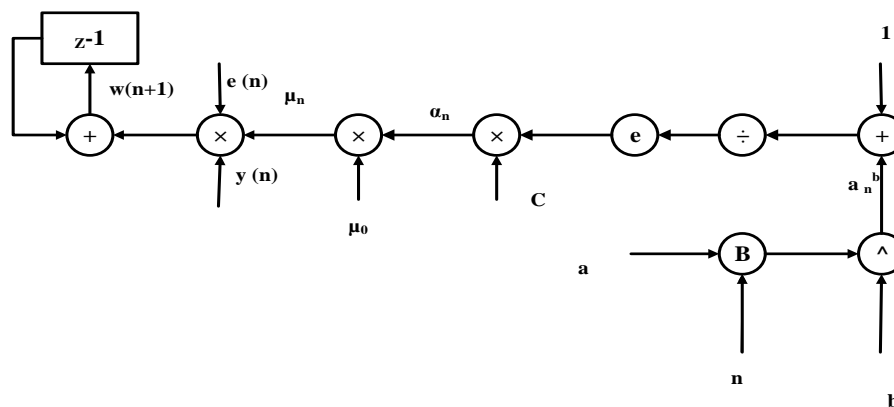


Fig. 3.10 Schematic Flow diagram for TVLMS algorithm

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 μ_n rather than a fixed convergence parameter μ . 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

3.3 Modified TVLMS Algorithm

The algorithm for Modified TVLMS is given by following equations,

$$w(n) = w(n-1) + \mu \times e(n) \times x(n) \quad (1.2)$$

Where, $\mu = \alpha_n \times \mu_0$

$$\alpha_n = C \left(\frac{1}{FsFact + a_n^b} \right) fs$$

$$C = fs^2$$

$$a = \frac{1}{fs}$$

$$b = \frac{1}{fs}$$

Where fs is sampling frequency of input signal, and FsFact is constant.

3.4 Experimental Conditions

The speech signal with different combinations of noise signal is used for experimentation. The NOIZEUS AURORA 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. These signals are collected from NOIZEUS database. The speech signal that we use was sp07 "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.

4. Results

Table 1: Performance comparison TVLMS & M-TVLMS for different noise level for SNR

SNR	0dB	5dB	10dB
TVLMS	10.5812	8.5311	7.4149
Modified TVLMS	14.1355	8.9532	12.9926

Table 2: Performance comparison TVLMS & M-TVLMS for different noise level for MSE

MSE	0dB	5dB	10dB
TVLMS	0.0002322	0.0002209	0.000217
Modified TVLMS	0.0001096	0.0002034	7.25E-05

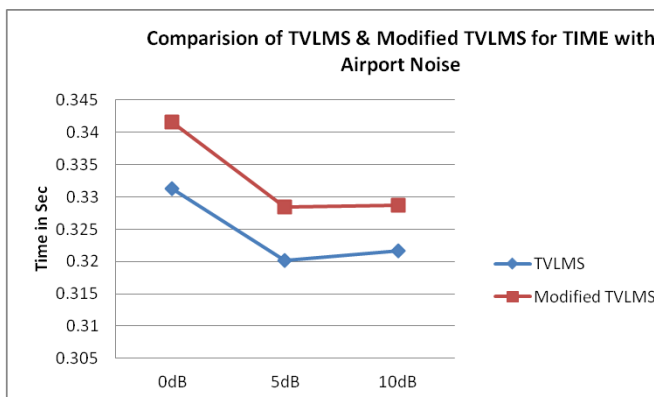
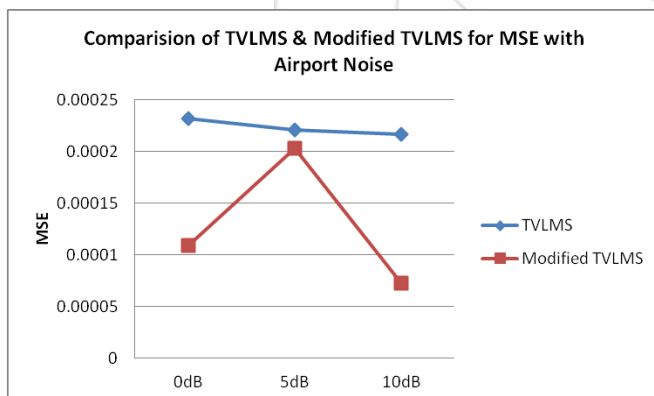
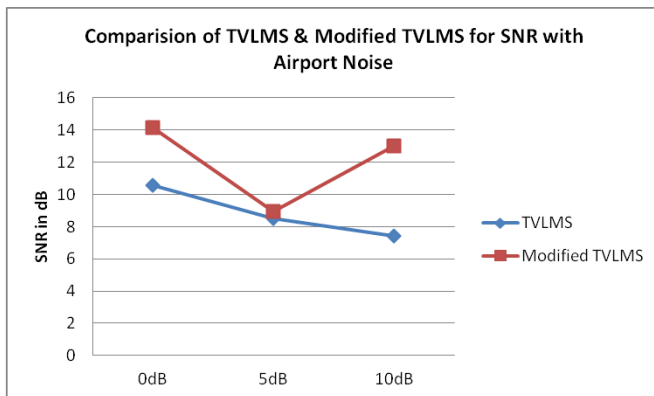
Table 3: Performance comparison TVLMS & M-TVLMS for different noise level for Execution Time

Execution Time	0dB	5dB	10dB
TVLMS	0.33124	0.32018	0.32168
Modified TVLMS	0.3416	0.32845	0.3287

provides a better SNR as compared to existing TVLMS algorithm when the speech is corrupted by airport noise. The test is performed at 0dB, 5dB and 10dB airport noise. The experimentation and validation are carried out for Mean Square Error (MSE) is very less in case of Modified TVLMS as compared with existing TVLMS for low noisy data, medium noisy and highly noisy data. The performance parameter called as execution time is also compared and the experimentation confirms the modified TVLMS algorithm converges fast. The experimentation and validation is carried out for modified TVLMS and is compared with existing methods and it is observed that modified method performs better as compared to existing methods.

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5. Conclusions

Experimental results reveal that the Modified TVLMS

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