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A Noise Reduction Method Based on Modified Least Mean Square Algorithm of Real Time Speech Signals With The Help of Wiener Filter

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Abstract - *Real-time voice denoising employs an adaptive filtering technique with variable length filters that tracks the noise characteristics and selects the filter equations based on those features. The LMS algorithm's primary benefits are its low computational complexity and evidence of convergence in stationary environments. This research proposes a modified LMS technique for real-time speech signal denoise. The suggested approach increases the capabilities of adaptive filtering by fusing the general LMS algorithm with the diffusion least mean-square algorithm. The suggested algorithm is successful in reducing speech signal noise, according to the calculation of the performance parameter. For replications and additional research applications, a complete MATLAB programming method is given.*

***Key Words*: Speech Enhancement, LMS, MATLAB, Modified LMS Algorithm, Segmental SNR, LLR, ISD, Cepstrum, Weiner Filter.**

1.INTRODUCTION

Voice transmissions may experience interference from various noise components while being transported via transmission lines before they reach their destinations. If these noise components are not eliminated, the voice signals may degrade to the point where their receiving ends suffer a partial or complete loss of the information content. The elimination of these undesirable components has been addressed by numerous researchers using various adaptive filtering techniques.

In order to minimise noise in speech signals, the authors showed how well the recursive least square (RLS) algorithm performed. They got three noise components machine gun, F16, and speech noises from the NOISE-92 database in addition to a clean voice signal from the Hindi speech database. The sampling frequency and resolution for both the noise and the clean signal components are 16 KHz and 16 bits, respectively. Twelve separate noisy speech signals were created by successively adding each of the three noises to a clean speech signal at signal to noise ratios (SNR) of -5dB, 0dB, 5dB, and 10dB levels. Six specially created filters were each fed a set of noisy signals in order to simulate them. The non-variable forgetting factor RLS (NVFFRLS), often known as the RLS algorithm, powers two of the filters, one of order 5 and the other of order 10, while the other two filters, both of order 2, are powered by variable forgetting factor RLS (VFFRLS) algorithm isused to drive two of the orders, one of order 5, and two of order 10. When it comes to RLS, the forgetting factor has a value of =0.99 while it has a value of min=0.95 when it comes to VFFRLS.

This is due to the fact that the VFFRLS algorithm can monitor changes in the noisy signal more precisely than the RLS method. The performance of the RLS algorithm with a variable forgetting factor for non-stationary processes has improved, which is consistent with the researchers' findings. The researchers created a brand-new adaptive filtering method called the modified adaptive filtering with averaging (MAFA) algorithm, which is utilized to remove white Gaussian noise from voice samples.

1.1 BLOCK DIAGRAM AND FLOWCHART

In this approach, the algorithm's parameters are improved by adding a weiner filter to the already-existing algorithm. The parameters of the original method and the suggested approach are compared after the Weiner filter has been added. The NOIZEUS sound database is used to source the noise signals. Signals with different strengths, ranging from 0 dB to 15 dB,

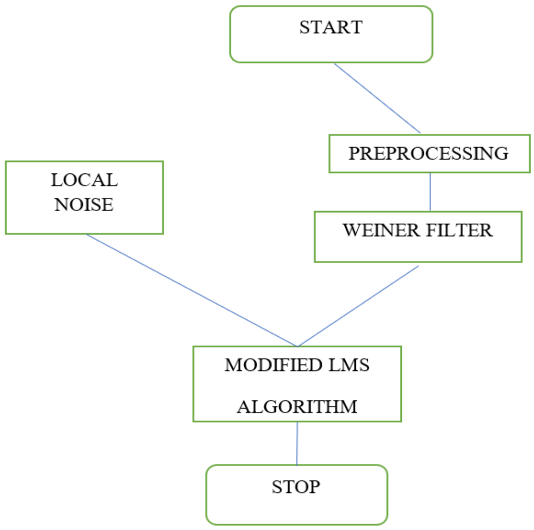

Fig.1 Proposed Noise Algorithm

Are compared. Speech is analysed frame-by-frame under the assumption that it is quasistationary. We assume that the clean speech signal xt(n) is deteriorated by additive noise vc(n) at each occurrence of time n, and the noisy signal is obtained as yc(n).

In a model with additive noise,

yc(n)=xc(n)+vc(n)

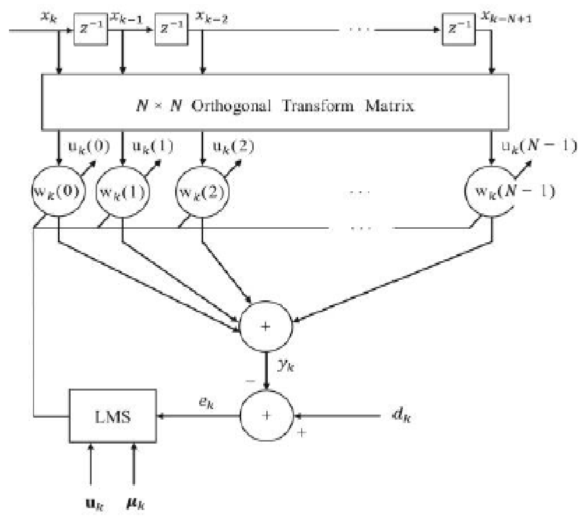

Fig.2 Detailed Algorithm for LMS Denoising

1.2 IMPLEMENTATION

Speech is analysed frame-by-frame under the assumption that it is quasistationary. We assume that the clean speech signal xt(n) is deteriorated by additive noise vt(n) at each occurrence of time n, and the noisy signal is obtained as yt(n).

□c(□)=□c(□)+□c(□)**Increase in Segmental SNR**: In any voice signal, energy change erratically and are not stationary in nature. As a result, each frame segment is computed independently and then added together to generate segmental SNR in order to obtain an accurate SNR value.

Thus, where N is the frame length, the equation for segmented SNR is provided.

Original loud speech is represented by x(n). The signal of the processed speech is x(n).

Log Likelihood Ratio (LLR): LLR determines the amount of distortion added during processing by comparing the spectrum of clean audio with processed speech. Equation (4) displays the formula for calculating LLR as DLLR

(ae,ac)=log10(aeRcaeT/acRcacT) (4)

where ac is the clean speech signal's LPC vector.

The LPC vector of the improved or processed speech signal is called ae. The clean voice signal's autocorrelation matrix is denoted by Rc.

Itakura-Saito Spectral Distance (ISD): This term refers to the difference between enhanced and clean voice signals in terms of the associated spectral envelope. ISD's standard value is never greater than 100.

Cepstrum: For linear separation, homomorphic signals incorporating convolution (such as a source and filter) are converted into the sums of their cepstra using the ceptrum representation. A common feature vector for characterising the human voice and musical sounds is the power cepstrum. The spectrum is typically initially converted using the mel scale for these applications.

Database: The noise signals have been taken from NOIZEUS database. 24 Noise signals have been taken from the database (with the exception of train signals). Each sentence is distorted by eight different types of real-world noises (Restaurant, Station, Airport,babble, Car, Street). The extracted noises segments are artificially added to the filleted clean speech signal in order to reach the desired SNR levels.

The noise signals from the NOIZEUS database are being passed through the matlab algorithm and the parameters have been verified. These parameters are compared to the existing algorithm and the output signal has been compared to the previous datasets.

SEGNAMT NO	PARAMENETS	VALUES
1	TOTAL NUMBER OF SPEECH SIGNAL	24
2	TOTAL NUMBER OF NOISY SIGNAL	8
3	SNR RANGE	0 dB , 5db ,10 dB and 15dB
4	SAMPLING SIGNALS FREQUENCY	28064 Hz
5	DOWN SAMPLING FREQUENCY	8000 Hz

Table. 1 Description of Speech Database

The results of the current experiment showed that the suggested technique outperformed the LMS Algorithm in terms of increasing the quality of the speech signal. This was supported by the three objective measurements of segmental SNR, LLR, Cepstrum, and ISD. Different levels of noise (0, 5, 10, and 15 dB) are required for proper analysis.

2. RESULTS AND DISCUSSIONS

The experimental findings of the current investigation are presented in this section. To evaluate the effectiveness of the suggested speech enhancement method and compare the denoise speech signal, all experiments were run using the NOIZEUS speech corpus database. Many researchers have used this database for applications involving voice improvement, and its gender-matched database has 30 IEEE sentences. For analysis purposes, one of the voice samples with a 10 dB SNR is left out of this work(Train).All of the voice samples in the database have this windowing applied to them. Later, an additive noise with various SNRs and the same original speech signal window length is introduced to the speech samples. The STFT is then used to derive the magnitude and phase spectrum values from the noisy voice inputs.

Finally, a two-stage method is used to reconstruct the speech signals in order to obtain the time domain speech signals. The original (clean) speech signal and the augmented speech signal are then calculated into four different objective measures for performance evaluation over each frame. The original (clean) and improved speech signals are shown in a time series plot over four different SNRs. The signal fluctuations over the course of the 28064 samples are shown in this plot for various SNRs.

The experimental findings of the current investigation are presented in this section. To compare the denoise speech signal and evaluate the effectiveness of the proposed speech enhancement method, all tests were run using the NOIZEUS speech corpus database. The signal fluctuations over the course of the 28064 samples are shown in this plot for various SNRs. The spectral power distribution for a spoken signal at 0 dB, 5 dB, 10 dB, and 15 dB with airport noise.

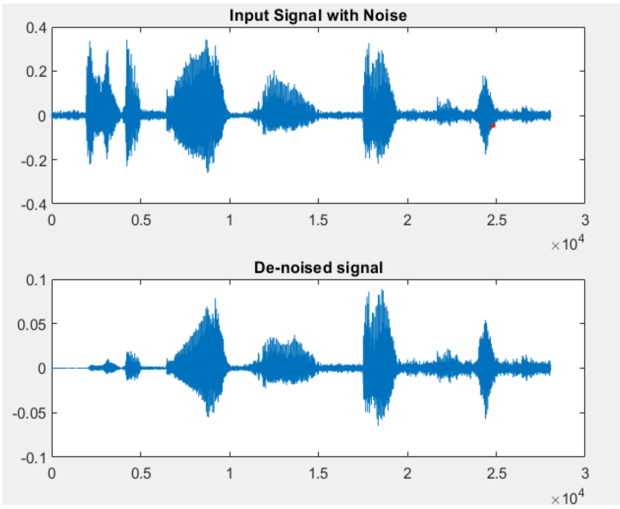

Fig 3 . Street 15dB

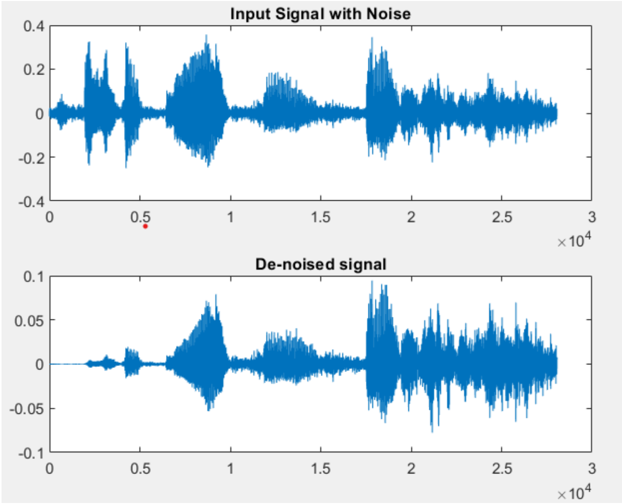

Fig 4. Airport 5 Db

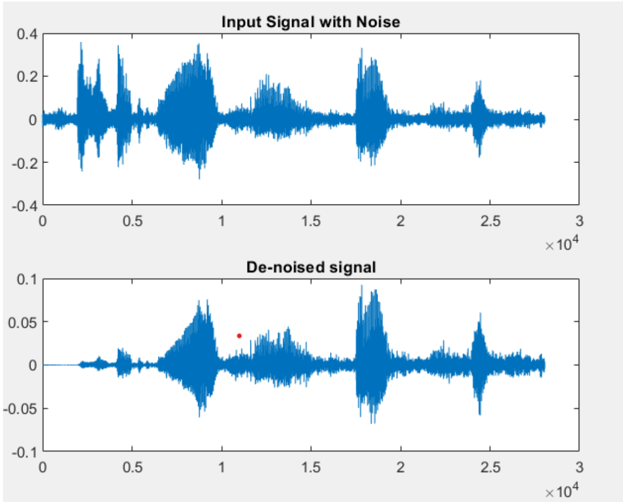

Fig 5. Babble 10 Db

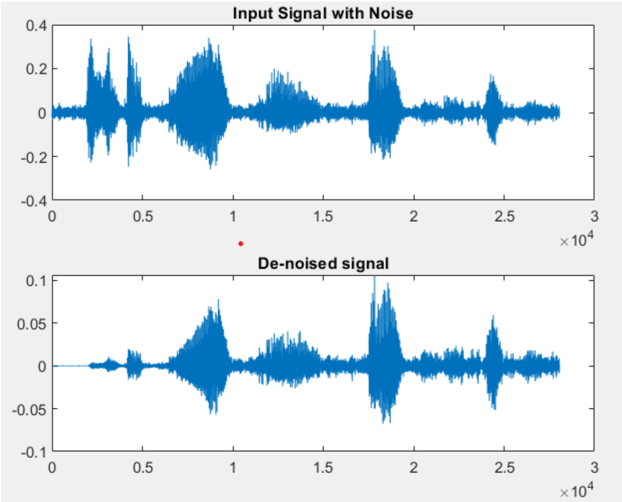

Fig 6. Restaurant 10 Db

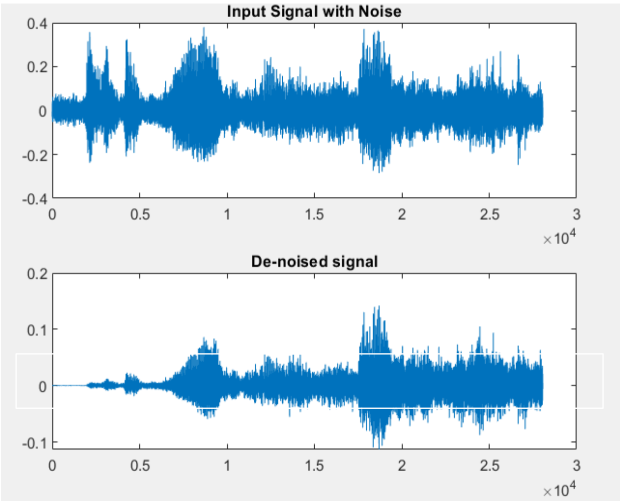

Fig 7. Car 0 Db

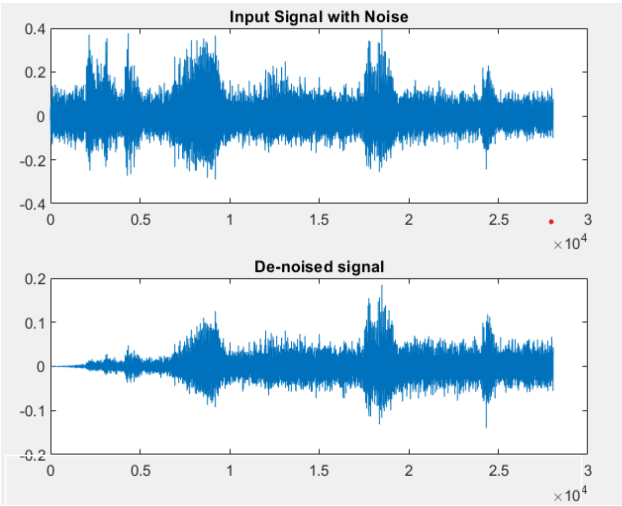

Fig 8. Station 0 Db

Table. 2 Speech Enhancement Outputs

S. No.	Noise Type	Parameters	Noise Level			
			0 dB	5 dB	10 dB	15 dB
1	Airport	Seg SNR	3.24	2.194	1.8	1.69
		Cepstrum	7.124	7.178	7.304	7.226
		LLR	1.25	1.265	1.355	1.3154
		ISD	88.06	99.83	99.99	99.92
2	Babble	Seg SNR	3.20	2.15	1.78	1.67
		Cepstrum	7.2341	7.388	7.55	7.344
		LLR	1.344	1.40	1.46	1.36
		ISD	91.41	99.66	99.89	99.98
3	Restaurant	Seg SNR	3.24	2.14	1.78	1.66
		Cepstrum	7.39	7.460	7.30	7.41
		LLR	1.35	1.38	1.32	1.39
		ISD	91.88	99.83	100	100
4	Station	Seg SNR	2.33	2.13	1.75	1.69
		Cepstrum	7.19	7.38	7.47	7.39
		LLR	1.30	1.36	1.41	1.38
		ISD	94.45	99.81	100	100
5	Car	Seg SNR	3.03	1.92	1.74	1.67
		Cepstrum	7.19	7.23	7.37	7.39
		LLR	1.30	1.31	1.34	1.37
		ISD	92.00	99.99	100	100
6	Street	Seg SNR	3.35	1.81	1.72	1.69
		Cepstrum	6.96	7.58	7.35	7.21
		LLR	1.17	1.47	1.35	1.30
		ISD	90.20	100	100	99.97

3. CONCLUSIONS

The modified LMS algorithm with the additional Weiner filter is suggested in this thesis. The proposed algorithm has produced satisfactory results when tested with different noise levels. This method can be utilised in a variety of settings, from the car to the airport, and is effective at reducing different types of noise levels. Through the objective measurements of Segmental SNR, LLR, ISD, and Cepstrum, this experiment demonstrated that the suggested method worked well in enhancing the quality of the speech signal over LMS Algorithm. Future work should concentrate on improving the current data and testing the algorithm with different other parameters.

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