"Design of Adaptive Filter to Improve SNR for Speech Signal Processing"

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ABSTRACT:

This paper is based design of Adaptive filter to improve SNR for speech signal processing using different parameter of adaptive filter. The adaptive filtering is one of the most useful digital filter in signal processing applications. This paper is based on improvement of the SNR for speech signal processing by using adaptive filter algorithm. Also in this paper, LMS and RLS algorithm for Adaptive Filter is explained. Adaptive algorithm can dynamically change parameter vary outside the signal characteristic and maintain the optimal filtering. Main focus of this paper is about compare the performance of the LMS and RLS algorithm according to change in filter length, step size, Mean Square Error and Signal to Noise Ratio. Also in LMS algorithm minimize the step size which could reduce the cause of input noise to step factor, improve the SNR performance of algorithm. For improve SNR used another Recursive Least Square (RLS) algorithm. RLS algorithm has fast response because it update filter coefficient are automatically. The RLS algorithm recursively solve the least square problem.

Keywords: *Adaptive Filter, Least Mean Square algorithm, Recursive Least Square algorithm, SNR.*

1. INTRODUCTION:

The speech signal processing has high requirement to the quality of voice signal. Thus speech enhancement is important for obtaining pure voice signal. The work described here mainly focuses on the enhancement of the speech signal by eliminating noise from the signal. In this paper we discussed adaptive filters and its algorithm.

1.1. ADAPTIVE FILTERS:

Adaptive filters are filters with the ability of adaptation to an unknown environment. A filter that self-adjusts its transfer function according to an algorithm driven by an error signal is called an adaptive filter. The ability of operating in an unknown environment added to the capability of tracking time variations of input statistics makes the adaptive filter a powerful device for signal-processing. An input signal is received for the adaptive filter and compared with a desired response, generating an error. That error is then used to modify the adjustable coefficients of the filter, generally called weight, in order to minimize the error[2] .This family of filters has been widely applied because of its versatility (capable of operating in an unknown system) and low cost (hardware cost of implementation, compared with the non-adaptive filters, acting in the same system).

1.2. LEAST MEAN SQUARE (LMS) ALGORITHM:

The adaptive algorithm is Least Mean Square (LMS) algorithm is a kind of method of simple implementation for the gradient search, using the instantaneous error square of the gradient instead of the one of the mean square error. Based on the adaptive filter new variable step least mean square (LMS) algorithm which could reduce the effect of input noise to step factor, improve the performance of the algorithm. This algorithm put forward a method to update step factor so that it has less influence on the input noise [1].

1.3. RECURSIVE LEAST SQUARE (RLS) ALGORITHM:

The recursive least square (RLS) algorithm uses Newton adaption method to minimize the mean square surface. So, it computes recursively, the inverse mean square function. RLS, an adaptive algorithm that finds the coefficients of filter recursively and minimizes the least square weight cost function. Here, in each iteration, new samples are iterated in recursive form. The RLS algorithms have faster convergence speed. This algorithm works good in time-varying environment[7].

2. PROPOSED WORK:

It is basic idea is to choose a controlled way to combine the step parameter with error function, using the value of error to adjust step factor [1]. If step size is increases the SNR is decreases and hence noise cancelation process will not be effective for reduced step size.

- To design of Adaptive filter structures with LMS and RLS algorithm with minimum step size
- To reduce the coefficients according to reduction in step size
- At the lower step size, to increase the SNR for both algorithms and achieve the good performance of adaptive filter for noise cancellation.

Adaptive filters track the dynamic nature of a system and allow you to eliminate time varying signals. The DSP System Toolbox libraries contain blocks that implement least mean square (LMS) and recursive least squares (RLS) adaptive filter algorithms. These filters minimize the difference between the output signal and the desired signal by altering their filter coefficients. Over time, the adaptive filter's output signal more closely approximates the signal you want to reproduce.

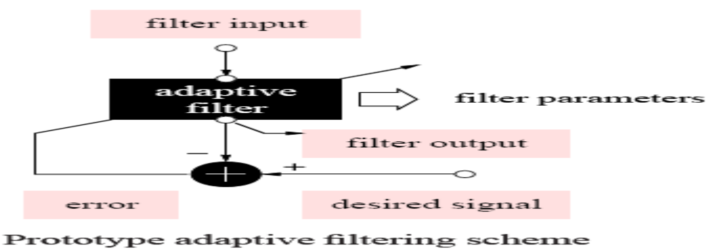


Fig. 2.1: Adaptive filter [1]

In this case first design the adaptive filter using LMS algorithm and then RLS algorithm for the same input signal. In the first case a new variable step least square algorithm. This algorithm put forward a method to update step factor so that it has less influence on the input noise. In initial stage of learning, the algorithm can get bigger step factor. When it tends to be stable, we can get a small step factor with smooth change. At the same time, we smooth it in time domain and have error signal autocorrelation being replaced. It decreases the mutation and the effect on the performance of the algorithm convergence arising from input noise The algorithm smoothes the step factor in time domain when they were in the process of iteration so that we should eliminate the influence on the overall trend when updating the step [1]. Simulink Model of LMS for Speech Signal Processing is shown in figure.

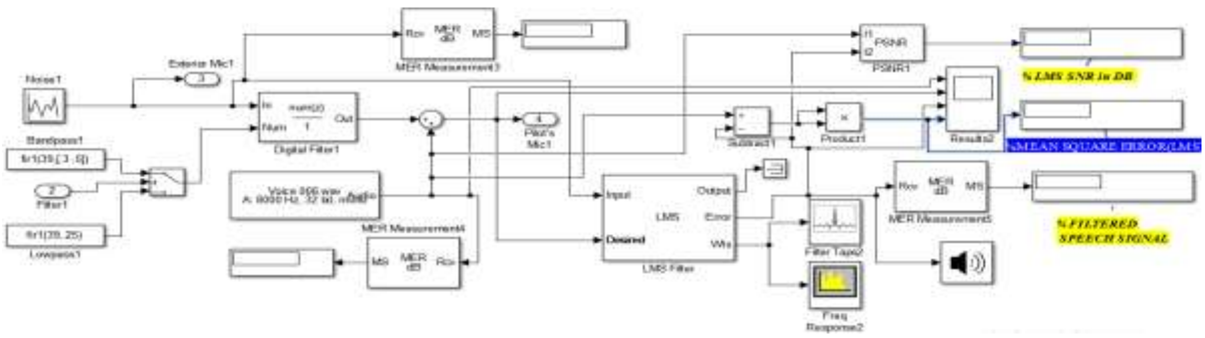


Fig. 2.2 : Simulink Model of LMS for Speech Signal Processing

It is basic idea is to choose a controlled way to combine the step parameter with error function, using the value of error to adjust step factor in real time. But when the SNR is too small, the result may not have obvious superiority. So we are trying to increase the SNR factor at the same time. The requirement for the adaptive algorithms which converge rapidly is essential in many applications where the changes in signal characteristics could be quite fast. From this point of view, the best choice is the Recursive Least Squares (RLS) algorithm [7]. The RLS algorithm recursively finds the adaptive filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. RLS algorithm has the potential to automatically adjust the coefficients of a filter, even though the statistical measures of the input signals are not present. Simulink Model of RLS for Speech Signal Processing is shown in figure.

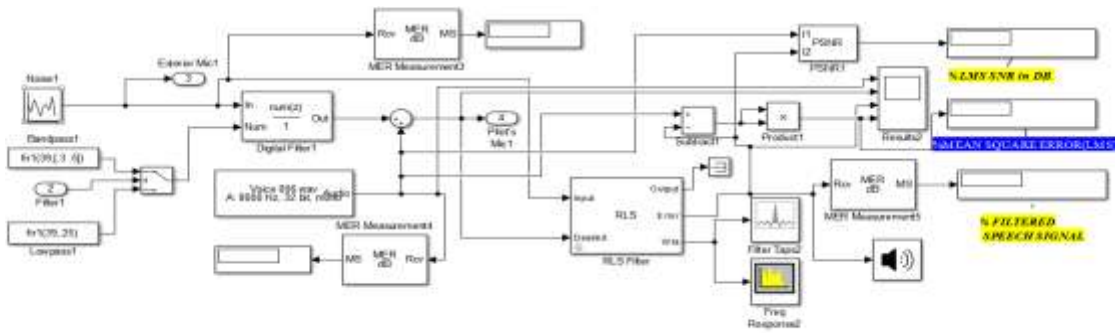


Fig. 2.3: Simulink Model of RLS for Speech Signal Processing

In second case we adapt Recursive Least square (RLS) algorithm. The Recursive Least Square (RLS) algorithm uses Newton adaption method to minimize the mean square error(MSE). So, it

computes recursively, the inverse mean square function . Improve the convergence rate of RLS and compare the performance on the basis of SNR .RLS algorithm converges faster. it has the highest noise cancellation capacity .

3 SIMULATION RESULTS :

In this section, we compare the LMS and RLS adaptive algorithms examine their performances. The speech data which sampled at 8 kHz was used to evaluate the proposed method, and the recorded time is ~5s. MATLAB Simulink designs of Adaptive filter for RLS and LMS Algorithms results are presented. The result are arranged in two Section; first section deal with the MATLAB SIMULINK of LMS adaptive Filter algorithm when audio signal is applied as input signal and second section RLS adaptive Filter algorithm when audio signal is applied as input signal. A fair performance comparison of simulation result for both algorithm is also presented in term of mean square error, percentage of noise removal , frequency response ,filter weight. Primarily filtering is done for audio signal. The effect of frequency and amplitude variation of audio signal on filter performance also investigated. finally SNR improvement comparison for LMS and RLS algorithm is presented.

3.1 LMS ALGORITHM SIMULATION RESULTS :

LMS filter is introduced for improved performance when a speech input signal is corrupted by Gaussian noise. The implemented LMS filter has filter length is 40. The following figure shows that the input signal, noisy signal (input signal + noise) and resultant error signal. The simulation of the LMS algorithm is carried out with the following specifications : Filter order N= 40, step size μ= 0.001.

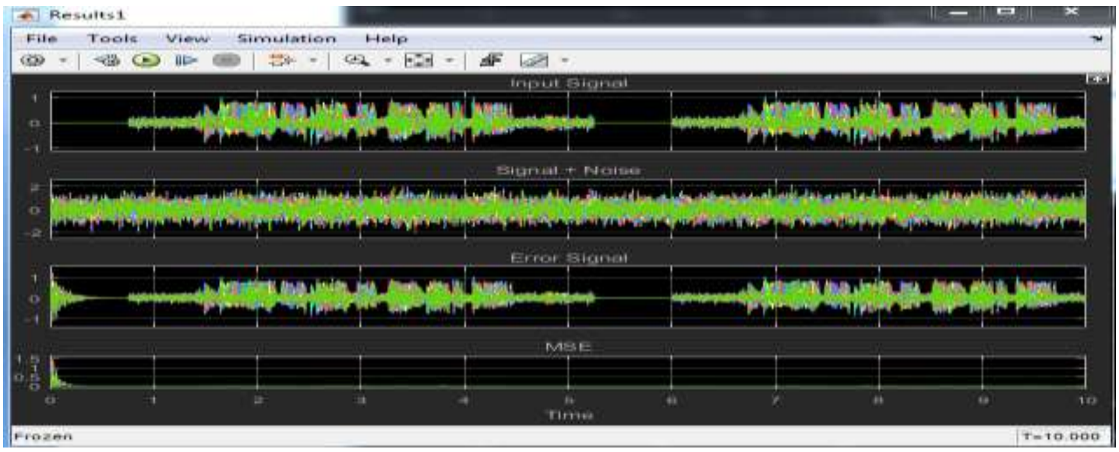


Fig. 3.1: MATLAB Simulation for LMS Algorithm

The LMS algorithm is simple and easy to implement but the convergence speed is slow, it can be seen from the simulation results derived in Fig.4.1. The LMS based filter adopts the approximate correct output

Fig.4.1, the corresponding mean squared error generated as per adaption of filter parameters. The value of MSE and SNR for LMS algorithm highly depends on the step size , which is derived by the simulation of LMS algorithm at different values of step size, varied from 0.0001 to 0.01. when step size minimum at that time SNR is high and step size is increase SNR is low. The corresponding numeric data is presented in Table 4.1.

TABLE 4.1: STEP SIZE VERSUS MEAN SQUARED ERROR

SR.NO	STEP SIZE(μ)	MEAN SQUARED ERROR	SIGNAL TO NOISE
1	0.0001	7.97×10^{-5}	46.78
2	0.0002	8.13×10^{-5}	47.14
3	0.0003	4.023×10^{-5}	46.78
4	0.0004	1.488×10^{-5}	45.83
5	0.0005	4.823×10^{-6}	45.01
6	0.0006	1.637×10^{-6}	44.39
7	0.0007	7.767×10^{-7}	43.90
8	0.0008	6.465×10^{-7}	43.46
9	0.0009	8.367×10^{-7}	43.04
10	0.001	1.264×10^{-6}	42.62
11	0.002	1.228×10^{-5}	39.23
12	0.003	2.184×10^{-5}	37.12
13	0.004	2.185×10^{-5}	35.57
14	0.005	1.359×10^{-5}	34.23
15	0.006	4.08×10^{-6}	32.97
16	0.007	1.121×10^{-6}	31.77
17	0.008	5.326×10^{-6}	30.65
18	0.009	2.141×10^{-5}	29.59
19	0.01	4.786×10^{-5}	28.59

With the help of filter coefficients and samples, the response of **LMS filter tap** is shown in following figure .

Fig.3.2: MATLAB Simulation for LMS Algorithm Filter coefficient

Finally the **frequency response of LMS filter** is given bellow

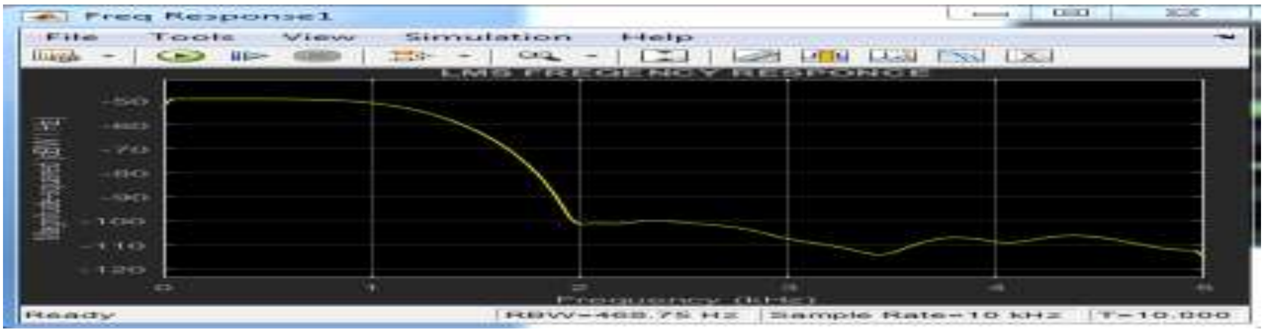


Fig. 3.3: MATLAB Simulation for LMS Algorithm Filter Frequency response

3.2 RLS ALGORITHM SIMULATION RESULTS:

The design RLS filter has filter length is 40. The following figure shows that the input signal, noisy signal (input signal + noise) and resultant error signal. The simulation of the RLS algorithm is carried out with the following specifications: Filter order N= 40, The forgetting factor λ = 1.

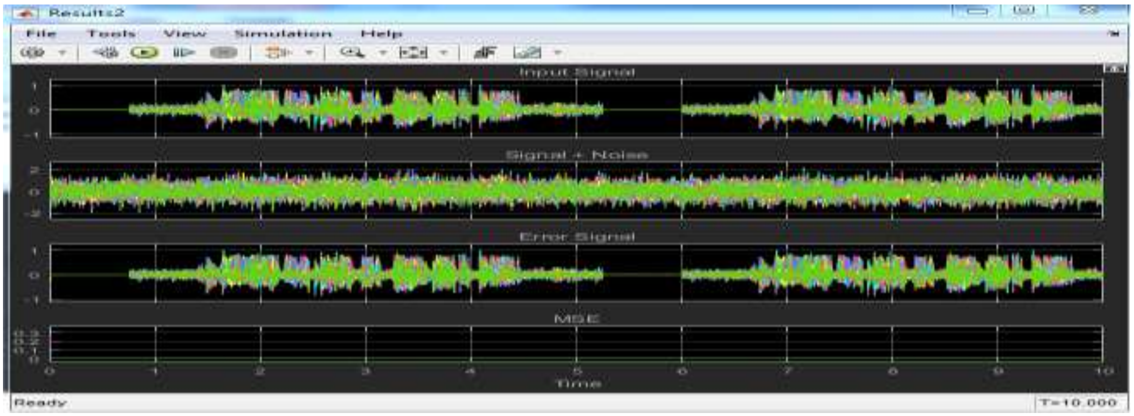


Fig. 3.4: MATLAB Simulation for RLS Algorithm

With the help of filter coefficients and samples, the response of RLS filter tap is shown in following figure.

Fig. 3.5: MATLAB Simulation for RLS Algorithm Filter coefficient

Finally the frequency response of RLS filter is given bellow

Fig.3.6: MATLAB Simulation for LMS Algorithm Filter frequency response.

The RLS algorithm is much faster and produce a minimum Mean Square Error , it is 0.00705 db for filter length N=40.

3.3 PERFORMANCE COMPARISON OF ADAPTIVE ALGORITHMS :

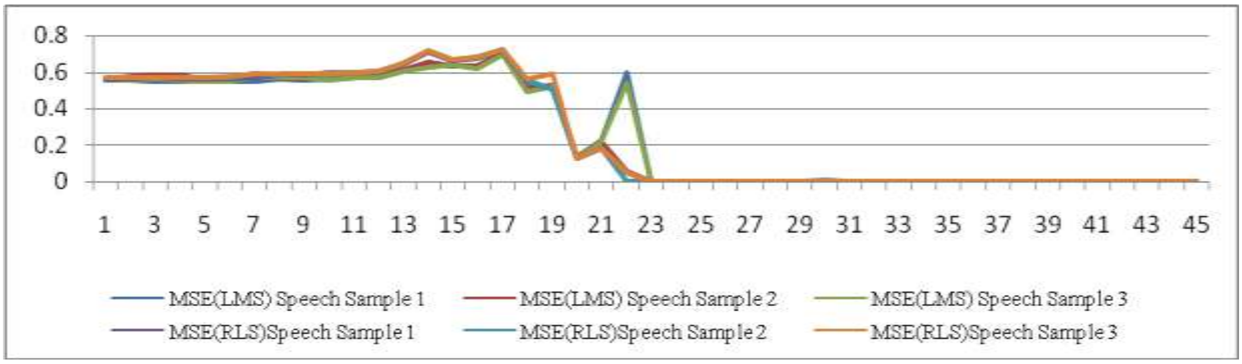
If we compare the filtered output and mean square error of both algorithms this show that RLS has faster learning rate with minimum MSE. The filter order also affect the performance reducing of noise in speech signal. The MSE change as filter order change.

Table 4.2 : MEAN SQUARED ERROR VERSUS FILTER ORDER

FILETR ORDER	MEAN SQUARED ERROR			MEAN SQUARED ERROR		
	Speech Sample 1	Speech Sample 2	Speech Sample 3	Speech Sample 1	Speech Sample 2	Speech Sample 3
1	0.5588	0.5624	0.5596	0.5683	0.5682	0.5689
2	0.5545	0.5757	0.5558	0.5683	0.5677	0.5687
3	0.5525	0.5817	0.5578	0.5689	0.5681	0.5691
4	0.5523	0.5866	0.5499	0.5665	0.5654	0.5661
5	0.5589	0.5649	0.5433	0.5675	0.5665	0.568
6	0.5489	0.5545	0.5444	0.5675	0.5706	0.5721
7	0.5496	0.5884	0.5666	0.5709	0.5887	0.5912
8	0.5624	0.5659	0.5649	0.5838	0.5842	0.5901
9	0.5611	0.565	0.5644	0.5849	0.5882	0.5868
10	0.563	0.567	0.556	0.597	0.5849	0.5860
11	0.5703	0.5783	0.5664	0.6009	0.5969	0.5988
12	0.5728	0.5825	0.5698	0.6013	0.6004	0.6026
13	0.610	0.6111	0.6002	0.6429	0.6436	0.6453
14	0.6554	0.6532	0.6207	0.7101	0.7112	0.7132
15	0.6343	0.6326	0.6403	0.6654	0.6662	0.668

16	0.6336	0.6308	0.6182	0.6788	0.6799	0.6822
17	0.7059	0.7037	0.6932	0.7203	0.7216	0.722
18	0.5125	0.5089	0.4933	0.5543	0.5553	0.5574
19	0.5323	0.5292	0.5194	0.5055	0.5066	0.5881
20	0.134	0.1323	0.1264	0.1244	0.1245	0.1266
21	0.2235	0.2213	0.2143	0.1827	0.1829	0.1851
22	0.5969	0.05842	0.5397	0.05037	0.052	0.05162
23	0.000035	0.0006421	0.0001998	0.0006278	0.0006899	0.0005403
24	0.001087	0.0007085	0.0001549	0.000473	0.0006713	0.0004501
25	0.000041	0.000121	0.0001976	0.0004947	0.0005514	0.0004362
26	0.001815	0.001326	0.0004489	0.0001273	0.000103	0.0001704
27	0.0007292	0.00048	0.0004395	0.000011	0.000015	0.0002584
28	0.0003394	0.00021	0.000826	0.0000582	0.000080	0.000368
29	0.0000324	0.000375	0.0001804	0.0003073	0.0003402	0.0002883
30	0.007373	0.0005768	0.0001826	0.000204	0.0000008	0.000018
31	0.000059	0.0008054	0.0000180	0.000145	0.000155	0.000088
32	0.000199	0.0001566	0.0000345	0.000046	0.000058	0.000022
33	0.0001384	0.0001512	0.0001009	0.000093	0.0001234	0.000068
34	0.0003811	0.0004338	0.0000002	0.0000267	0.000095	0.000007
35	0.0003687	0.0004049	0.00000006	0.0000884	0.0000003	0.0000078
36	0.0004747	0.0004074	0.0000115	0.0000068	0.0000003	0.00000048
37	0.0003384	0.0003429	0.00000031	0.0000028	0.00000006	0.00000023
38	0.0003413	0.0003616	0.000000052	0.0000013	0.00000004	0.00000013
39	0.0003164	0.0003654	0.00000076	0.00000016	0.00000005	0.000000098
40	0.0003175	0.0003767	0.00000126	0.0000017	0.000000052	0.00000006
41	0.000613	0.0002142	0.00000092	0.0000016	0.000000032	0.00000007
42	0.0006621	0.0001678	0.00000145	0.0000009	0.000000027	0.00000011
43	0.0006853	0.0001343	0.00000146	0.00000011	0.000000021	0.00000001
44	0.0007983	0.000104	0.00000149	0.00000017	0.000000013	0.00000009
45	0.0002451	0.0000697	0.00000007	0.0000057	0.000000010	0.00000071

CHART 4.1 : MEAN SQUARE ERROR AND FILTER LENGTH



In above table when filter order is less than 18(<18) LMS has good MSE as compared to RLS. But the filter order increases greater than 18(>) it has low performance in term of MSE. It also indicate that selection of right filter is necessary to achieve the better performance, from below table 4.2 we shown that as we increases the filter order the MSE is decreases and also shown that Table 4.2 show the performance comparison of LMS and RLS in term of MSE when filtered order is changed.

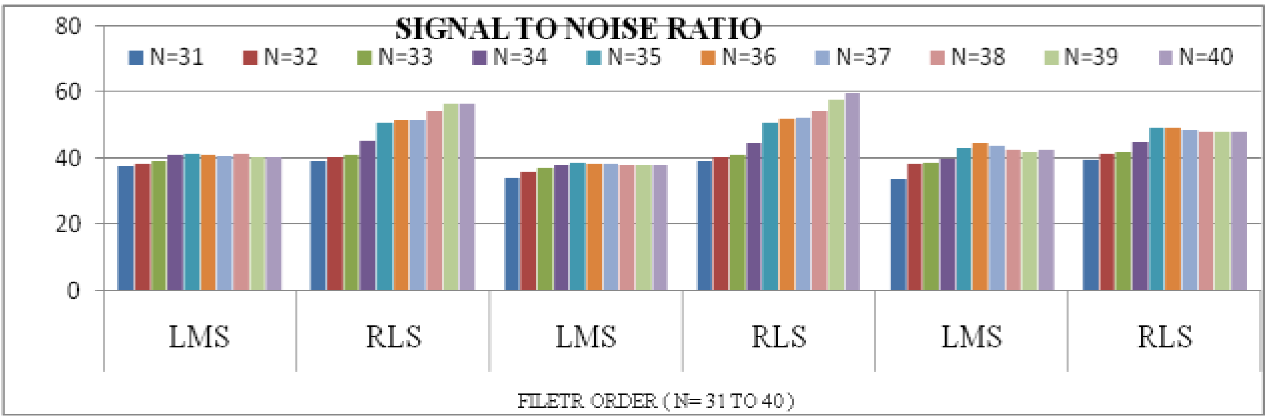
3.4 SIGNAL TO NOISE RATIO (SNR) :

Improvement of LMS and RLS for proper filtering order should be higher but as increases filter order the convergence speed get slower . As we increases the order of filter SNR get increases of both filter but SNR of RLS is better as compared to LMS algorithm. We getting different speech signal like music ,male voice , female voice and changing the filter order , from that we seen that RLS has better performance. Following table 4.3 shows the SNR improvement for LMS and RLS Adaptive Algorithms.

TABLE 4.3 SNR IMPROVEMENT

FILETR ORDER (N)	SIGNAL TO NOISE RATIO (LMS)			SIGNAL TO NOISE RATIO (RLS)		
	Speech Sample 1	Speech Sample 2	Speech Sample 3	Speech Sample 1	Speech Sample 2	Speech Sample 3
31	37.71	34.01	33.78	39.05	39.04	39.63
32	38.63	36.11	38.42	40.47	40.43	41.37
33	39.31	37.21	39	41.18	41.12	41.87
34	41	38.14	39.97	45.36	44.81	45.23
35	41.65	38.75	43.19	50.82	50.7	49.26
36	41.07	38.63	44.5	51.77	52.18	49.48
37	40.84	38.34	43.7	51.82	52.28	48.46
38	41.46	38.01	42.82	54.5	54.41	48.09
39	40.34	38.09	42.10	56.51	58	48.16
40	40.41	38.25	42.62	56.68	59.74	48.17

CHART 4.2 : FILTER LENGTH AND SNR



4. CONCLUSION:

In the present work two adaptive filter algorithms; LMS and RLS are design on MATLAB SIMULINK and the simulation results are analyzed for a speech signal. A fair performance comparison has been presented among the discussed algorithms based on the popular performance indices like Mean Squared Error (MSE), filter length, step size, convergence speed, etc. The designed system is further tested for different speech signal of different noise level for RLS and LMS algorithm.

The simulation results show that the LMS algorithm has high Mean Squared Error (MSE) and gives good results for improving SNR if step size and filter length is chosen correctly. In LMS algorithm step size is reduced the SNR is higher and step size is increases the SNR is low. In RLS algorithm filter length change from one to forty five has seen from the result, less MSE and high SNR for higher filter length. Its MSE is low for higher filter length and high SNR .

A fair performance comparison lies between LMS and RLS algorithms, On the basis of parameter filter length. A fair amount of SNR improvement is achieved in both. When the results of two algorithms were compared, it is found that the RLS algorithm has shown better performance with the advancement of 59.74 dB SNR as compare with the LMS algorithm 38.25 dB SNR, and hence it is conclude that RLS algorithm has better SNR as compared LMS algorithm

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