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PERFORMANCE ANALYSIS OF SPEECH ENHANCEMENT USING DIFFERENT ADAPTIVE FILTERS WITH DIFFERENT MODULATION TECHNIQUES

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ABSTRACT
Adaptive filtering has become a vast area of researchers since last few decades in the field of electronics and communication. Adaptive noise cancellation is a method used for noise reduction in the speech signal. In this paper, deals with cancellation of noise on the speech signal using two old algorithms i.e. Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and one new Unbiased and Normalized Adaptation Noise Reduction (UNANR) algorithm. The UNANR algorithm model does not contain any bias unit, and the coefficients are adaptively updated by using the steepest-descent algorithm. The Amplitude Modulation (AM) and Frequency Modulation (FM) are used separately in combination with Additive White Gaussian Noise (AWGN) channel. The channel signal quality parameter Peak Signal to Noise Ratio (PSNR) and Root Mean-Square Error (RMSE) are measured and compared. The simulations result of LMS, NLMS and UNANR are compared and shows that the performance of the UNANR based algorithm is superior to that of the LMS algorithm in noise reduction.

KEYWORDS: Adaptive filtering, LMS, NLMS, UNANR, PSNR, RMSE

INTRODUCTION
Speech is the vocalized form of communication in daily life. Speech is most natural form of human communication [1]. It existed since human civilizations began and even till now [2]. The perception of speech signal is generally measured in terms of its quality and intelligibility. The quality is a subjective measure that indicates the pleasantness or naturalness of the perceived speech [3]. Intelligibility is an objective measure which predicts the percentage of words that can be correctly identified by listeners [4], [5]. Enhancement means the improvement in the value or quality of something. In this paper, the speech enhancement of voice signal using various adaptive filters. The speech signal is first mixed with a noise signal then it is modulated with two kind of the analog modulation techniques i.e. Amplitude Modulation (AM) and Frequency Modulation (FM); one at a time [6]. Then Additive White Gaussian Noise (AWGN) channel is chosen as a communication channel in configuration with one of the modulation technique [7]. Then at the receiver side demodulation is performed and filtered with adaptive filters. The same process is also done with the Rician fading channel [8]. The different kinds of filters which are used i.e. Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Unbiased and Normalized Adaptive Noise Reduction (UNANR) [9], [10].

SPEECH ENHANCEMENT
The main objective of speech enhancement technique is to improve the quality and minimize the loss in intelligibility of the signal and listener fatigue [11], [12]. The basic overview of speech enhancement is as shown in Fig. 1. The speech signal is first mixed with a noise signal then it is modulated with two kind of the analog modulation techniques i.e. Amplitude Modulation (AM) and Frequency Modulation (FM); one at a time [1], [2]. The speech enhancement of voice signal using various adaptive filters which are used i.e. Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Unbiased and Normalized Adaptive Noise Reduction (UNANR) [13], [14].

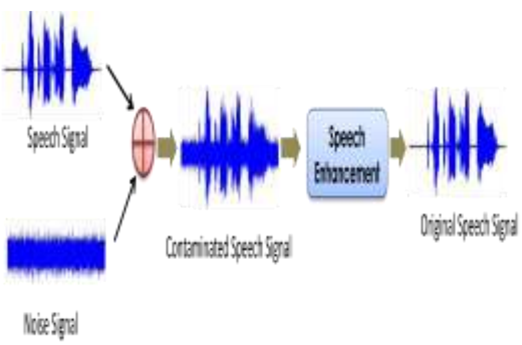


Fig. 1: Basic System of Speech Enhancement

MODULATION TECHNIQUES

Modulation is the process of varying some parameter of a periodic waveform in order to use that signal to convey a message [2]. Normally a high-frequency sinusoidal waveform is used as carrier signal [3]. For this purpose, if the variation in the parameter of the carrier is continuous in accordance to the input analog signal the modulation technique is termed as analog modulation scheme [4].

Therefore, the types of modulation techniques along with their mathematical expressions and waveforms are tabulated in Table 2.

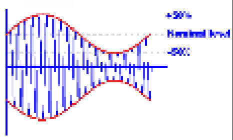
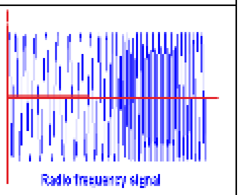
TABLE-1: TYPE OF MODULATION TECHNIQUES

Sr. No.	Type of Analog Modulation Techniques	Short Form
01	Amplitude Modulation	A.M.
02	Frequency Modulation	F.M.
03	Phase Modulation	P.M.

ADAPTIVE FILTERS STRUCTURE

The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unaltered [15]. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics [16]. The Adaptive noise cancellation (ANC) efficiently attenuates low frequency noise for which passive techniques are ineffective [17]. The basic adaptive filter structure is shown in figure 2.

TABLE-2: TYPE OF MODULATION TECHNIQUES

Sr. No.	Types of Modulation Techniques	Equation	Waveform
	Amplitude Modulation	y (t) = [A + M cos (ω_m t + φ)] . cos(ω_c t)	
	Frequency Modulation	y (t) = A cos [$\omega_c t$ + m_f sin (φ)	

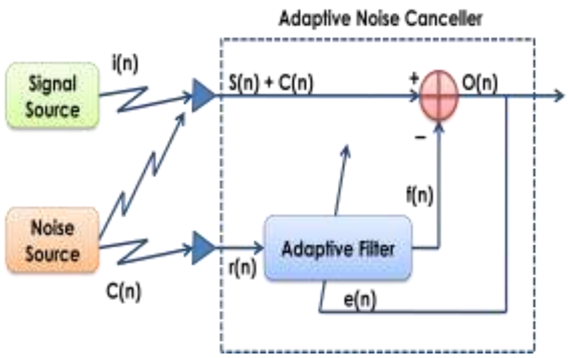


Fig. 2: Adaptive Filter Structure

Let us consider an adaptive filter with a primary input *i(n)*, that is noisy speech signal *S(n)* with additive noise *C(n)*. While the reference input is noise *r(n)*, which is correlated in some way with *C(n)* [11]. If the filter output is *f(n)*, the output of the summer *O(n)* is nothing but the error signal and it is written as, filter error given as

$$e = \{S(n) + C(n)\} - f(n), \quad (1)$$

then, mean square error (MSE) is obtain as

$$e^2 = \{S(n) + C(n)\}^2 - 2f(n)\{S(n) + C(n)\} + f(n)^2 \quad (2)$$

$$= \{C(n) - f(n)\}^2 + S(n)^2 + 2S(n)\, C(n) - 2f(n)\, S(n) \quad (3)$$

Since the signal and noise are uncorrelated, the mean-squared error (MSE) is

$$E[e^2] = E[\{C(n) - f(n)\}^2] + E[S(n)^2 \quad (4)$$

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Minimizing the MSE results in a filter error output that is the best least-squares estimate of the signal *S(n)* . The adaptive filter extracts the signal, or eliminates the noise, by iteratively minimizing the MSE between the primary and the reference inputs [13], [14]. Minimizing the MSE results in a filter error output *f(n)* that is the best least-squares estimate of the signal *S(n)* [15], [16].

Therefore, different kinds of Adaptive filter Algorithms with mathematical equations are listed in Table-3.3 [12], [17].

Table 3: Adaptive filter Algorithms

Sr. No.	Algorithms	Adaptive filter	Equation
	Least Mean Square	LMS	$w(n+1) = w(n) + \mu\, i(n) e(n)$
	Normalized Least Mean Square	NLMS	$w(n+1) = w(n) + \mu(n)\, i(n)\, e(n)$
	Unbiased and Normalized Adaptive Noise Reduction	UNANR	$\hat{w}_k(n+1) = \frac{w_k(n+1)}{\sum_{k=1}^{M} w_k(n+1)}$

SIMULATION RESULTS DISCUSSION

Firstly speech enhancement is done with stored voice data. There can be total four possible configurations for this experiment because we have two types of modulation techniques i.e. AM and FM and here AWGN is used as communication channels. Consider the case of stored input speech signal first. In this case AM and FM are selected as modulation technique one by one to transmit the whole speech signal after addition of background noise at the transmitter side. AWGN channel is considered as channel for transferring the speech signal. During the speech signal propagation through AWGN channel, channel noise gets added to the speech signal. At the receiver side, first AM and FM demodulation is performed one by one then speech signal is passed through one of the adaptive filter i. e. LMS, NLMS and UNANR filter is selected and the values of signal parameters i.e. PSNR and RMSE are recorded at different values of SNR in dB.
The Performance of LMS, NLMS and UNANR are shown for AM and FM modulation technique with AWGN channel and graph is plotted between PSNR v/s SNR as shown in Fig. 3 & 4 and tabulated in Table 4 & 5.

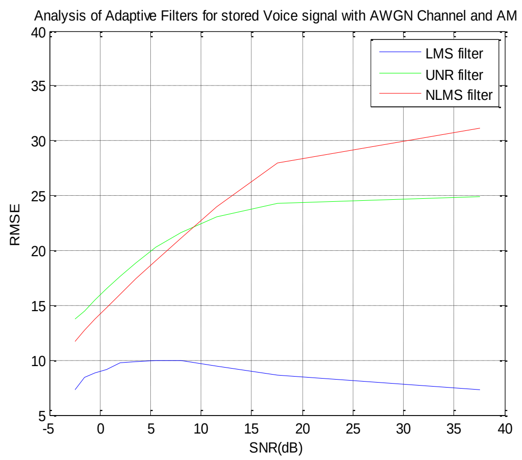

Fig. 3: Adaptive filtering for AM with AWGN channel for stored voice

It is clear from Fig. 3 and Table 4.2 that NLMS and UNANR perform much better performance than the LMS filter. Though performance of NLMS and UNANR are in same pattern but UNANR gives best results.

Table 4: Measurement of Parameters of Adaptive Filters between PSNR Vs SNR

Sr. No.	SNR (in dB)	PSNR		
		LMS	NLMS	UNANR
1	-2.47	7.342	11.728	13.714
2	-1.53	8.3781	12.732	14.48
3	-0.53	8.861	13.77	15.52
4	0.63	9.190	14.80	16.56
5	1.97	9.80	16.00	17.65
6	3.55	9.89	17.39	18.82
7	5.49	9.92	19.05	20.28
8	7.99	9.927	21.12	21.60
9	11.51	9.44	23.99	23.035
10	17.53	8.68	28.00	24.34
11	37.53	7.306	31.11	24.86

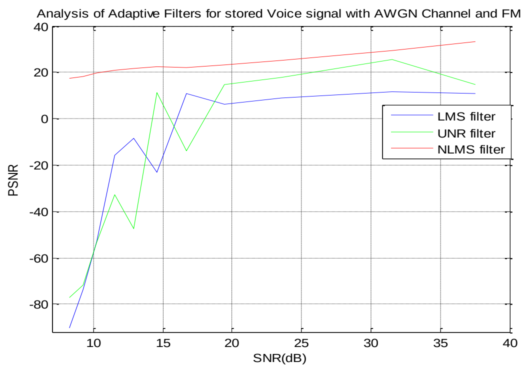

Fig. 4: Adaptive filtering for FM with AWGN channel for stored voice

Table 5: Measurement of Parameters of Adaptive Filters between PSNR Vs SNR

Sr. No.	**SNR (in dB)**	**PSNR**		
		LMS	**NLMS**	**UNANR**
1	10.23	- 52.99	19.75	-53.32
2	11.51	- 15.59	20.79	-32.79
3	12.92	-8.43	21.64	-47.41
4	14.60	- 23.04	22.46	11.08
5	16.70	10.82	21.97	-14.02
6	19.47	6.24	23.17	14.56
7	23.55	8.90	25.04	17.62
8	31.51	11.75	29.31	25.66
9	37.53	10.98	33.11	14.79

The performance of LMS, NLMS and UNANR for AM and FM modulation technique with AWGN channel are plotted between SNR and RMSE v/s SNR as shown in Fig. 5 & 6 and parameters recoded in Table 6 & 7.

It is evident from Fig. 5 and Table 6 that NLMS and UNANR perform much better performance than the LMS filter. Though performance of NLMS and UNANR are in same pattern but UNANR gives best results.

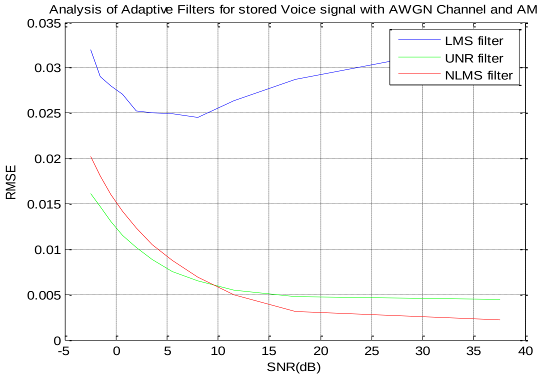

Fig 5: Adaptive filtering for AM with AWGN channel for stored voice

Table 6: Measurement of Parameters of Adaptive Filters between RMSE vs SNR

Sr. No.	SNR (in dB)	RMSE		
		LMS	NLMS	UNANR
1	-2.47	0.032	0.020247	0.0161
2	-1.53	0.029	0.018037	0.01473
3	-0.53	0.028	0.01600	0.0130
4	0.63	0.0271	0.014204	0.0115
5	1.97	0.0252	0.0123	0.0102
6	3.55	0.0250	0.0105	0.0089
7	5.49	0.0249	0.00873	0.00756
8	7.99	0.02451	0.006866	0.00649
9	11.51	0.02632	0.00493	0.0055
10	17.53	0.0287	0.003107	0.00473
11	37.53	0.033	0.002172	0.004463

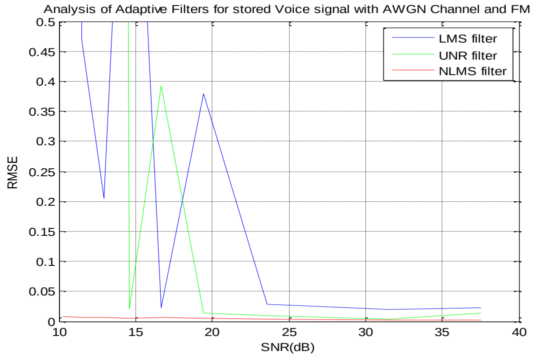

Fig. 6: Adaptive filtering for FM with AWGN channel for stored voice

Table 7: Measurement of Parameters of Adaptive Filters between RMSE vs SNR

Sr. No.	**SNR (in dB)**	**RMSE**		
		LMS	**NLMS**	**UNANR**
1	10.23	34.8812	0.0080	36.23
2	11.51	0.470	0.007	3.40
3	12.92	0.206	0.0064	18.35
4	14.60	1.109	0.0058	0.0217
5	16.70	0.022	0.0064	0.392
6	19.47	0.380	0.0054	0.0146
7	23.55	0.028	0.0043	0.0102
8	31.51	0.0201	0.0026	0.00406
9	37.53	0.0220	0.0017	0.0142

CONCLUSION

In this paper, work presented the speech enhancement through LMS, NLMS and UNANR. Performances of these filters are measured with respect to PSNR and RMSE v/s SNR when speech signal is affected by both background and channel noise. The comparison and analysis of performances of these adaptive filters have been done through plotted graphs. From all the performed experiments it is apparent that NLMS and UNANR filters have better performance than LMS. There is a fight between in NLMS and UNANR filter's performances in most of the graphs. As soon as the SNR improves performances of the filters get improved and when background noise level gets increased then performances of LMS, UNANR and NLMS filters gets degraded.

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