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COMPARISON OF ADAPTIVE FILTERS ALGORITHMS FOR SPEECH ENHANCEMENT WITH DIFFERENT CHANNELS

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ABSTRACT

In many application of noise cancellation, the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly. Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Unbiased and Normalized Adaptive Noise Reduction UNANR adaptive filters have been used in a wide range of signal processing application because of its simplicity in computation and implementation. The UNANR algorithm has established itself as the "ultimate" adaptive filtering algorithm in the sense that it is the adaptive filter exhibiting the best convergence behavior. Various Adaptive filter algorithms have been derived such as LMS, NLMS and UNANR to solve the dilemma of fast convergence rate or low excess root mean-square (RMS) error in the past two decades. This paper presented a new, easy to implement, LMS, NLMS and UNANR algorithm with various channels such as AWGN and Rician channel using the MATLAB R2013a that employs the RMS and the PSNR estimated system noise power to control the quality of online speech signal. Simulation experiments show that the NLMS and UNANR algorithm performs very well than LMS.

KEYWORDS: Adaptive filters algorithms, LMS, NLMS, UNANR, AM, AWGN

INTRODUCTION

Continuous improvement of communication and multimedia systems has led to the widespread use of processing devices and speech recording, e.g., mobile phones, speech recognition tools [1]. In most practical situations, these devices are being used in environments where undesirable background noise exists [2]. Degraded speech can cause problems for both speech recognition systems and mobile communication. Now a day, most of the people use the communication devices almost as a primary good: telephones, mobiles, internet and the customers demand a high quality and coverage [3].

A second source of noise is channel noise which affects both digital and analogue transmissions and therefore degrades the resulting speech at the receiver end.

Acoustic noise problems like industrial contending speakers, background noise, equipment noise, car engine noise, low-quality microphones, and room reverberation are subjected on speech signals in real time surroundings [4]. The Speech signals are corrupted by these several forms of noise and also they are subjected to distortion caused by communication channels [5], [6].

This thesis is all about 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 AM. Then 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. The different kinds of filters which are used i.e. LMS, Normalized Least Mean Squares NLMS and Unbiased and Normalized Adaptive Noise Reduction UNANR [11].

SPEECH ENHANCEMENT

The flowchart for simulation of this work as shown in Figure 1, what is the flow of input speech signal during speech enhancement process [12]. This flowchart is consist of seven stages which comprises recording of 1-speech signal, 2-background noise addition, 3-modulation, 4-channel noise addition, 5-demodulation, 6-adaptive filtering and 7-measuring predefined parameters.

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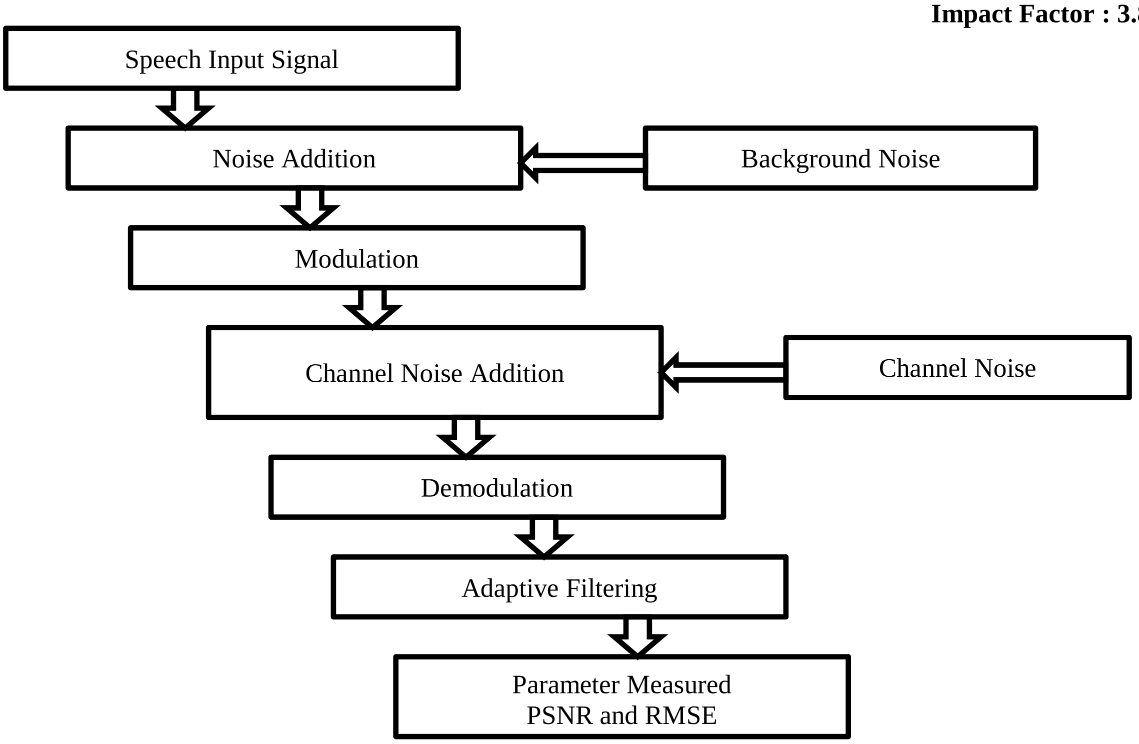

Fig. 1: Simulation Flowchart

There are various parameters used for simulation which are presented in Tabular form in Table 1.

Table 1: Simulation Parameters

S. No.	Parameter	Type/ Value
1	Technology	Speech Enhancement
2	Speech Input signal	Online
3	Modulation Methods	AM
4	Communication Channel	AWGN, Rician
5	SNR Range	38 to -2.5
6	Adaptive Filters	LMS, NLMS, UNANR
7	Measuring Entity	PSNR, RMSE

COMMUNICATION CHANNELS

A medium through which a message or information is transmitted to its intended destination is called communication channel [2]. In an analog channel model, the transmitted message is modeled as an analog signal. The model can be a linear or non-linear, time-continuous or time-discrete (sampled), memory less or dynamic (resulting in burst errors), time-invariant or time-variant (also resulting in burst errors), baseband, pass band (RF signal model), real-valued or complex-valued signal model [4].

TYPES OF CHANNEL

There are many types of channels but two channels are used in this thesis: (AWGN) channel, a linear continuous memory less model and Rician fading, log-normal shadow fading and frequency selective (dispersive) fading [8].

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Additive White Gaussian Noise (AWGN) Channel

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It's a simple model of the imperfections that communication channel consists of. If you transmit a certain signal into space or atmosphere or copper line to be received at the other end, there are disturbances (aka noise) present in the channel (space/atmosphere/copper line) due to various reasons [3]. One such reason is the thermal noise by the virtue of electrons' movement in the electronic circuit being used for transmission and reception of the signal [9].

The Power spectral density (PSD) of Thermal noise defined by equation as:

$$P=2KRT\text{-}Volts^2/Hz \qquad (1)$$

Where K is the universal Boltzmann constant and T is the ambient temperature in degree Kelvin. Now this PDF is depend upon temperature T can be represented by a noiseless, resistor R in series with a random white-noise voltage (thermal noise) of power spectral density. Hence this Johnson noise is a White Noise [10]. The spectrum of PDF has uniform frequencies over the entire frequency range of interest. This disturbance or noise is modeled as Additive White Gaussian Noise [11].

Time Domain Representation of this Noise

Let's first see the time-domain behavior of noise:

Additive: Because the noise will get added to your transmitted signal not multiplied. So, the received signal r(t) = f(t)+n(t) , where f(t) was the original clean transmitted signal, and n(t) is the noise or disturbance in the channel.

White: The term white is used in analogy with white light, which is superposition of all visible spectral components.

Gaussian: This thermal noise is random in nature, of course noise can't be deterministic otherwise you would subtract the deterministic noise from r(t) as soon as you receive r(t). So, this random thermal noise has Gaussian distribution with 0 mean and variance as the Noise power. Just leave the variance part if you don't understand it now, remember only that if variance of Gaussian is high then, it’s bad as you may need to increase the power of f(t) or be satisfied with higher probability of error. 0 mean means that the expected value r(t) during any time interval T is 0. But simply put, it also means that on an average n(t) will take 0 value. And probability of n(t) = 0 is the highest and probability rapidly decreases as you increase the magnitude of n(t) .

Frequency Domain Representation of this Noise

Looking at the frequency domain behavior of this noise:-

White: meaning same amount of all the colors. Or same power for all the frequencies. Which means that this noise is equally present with the same power at all the frequencies. So, in frequency domain, Noise level is uniform throughout at every frequency. So, this is AWGN channel.

The AWGN channel model is a simple but basic concept for modeling channel effects on electromagnetic signals in communication systems. The simplest channel model in wireless commutations is the well known (AWGN) model. The mathematical expression of the AWGN channel as follows:

The AWGN channel adds white noise $n(t)$ to the signal $f(t)$:

$$r(t) = f(t) + n(t) \qquad (2)$$

The noise has a constant spectral density and the amplitudes are normal distributed with variance $\sigma^2 = N_0/2$. N0 is the single-sided noise spectral density.

White Noise is existent in all communication systems independent of their propagation and induced by many sources like thermal noise in electronic circuits, terrestrial noise, and cosmic noise [8].

Hence the AWGN channel model is essential but not sufficient to model terrestrial propagation effects. The terrestrial propagation faces further effects like multipath, slow and deep fading, which can affect the channel severe. To consider these, other channel models have to be used additionally [9]. The Rayleigh and the Rician channel model are common representatives of these and described below.
If the average received signal power is S; signal Bandwidth W and the noise power spectral density is N_0 [W/Hz], the channel capacity of AWGN is expressed as:

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$$c_{awgn} = W \log_2\left(1 + \frac{S}{N_0 W}\right) \text{ Bit/sec} \qquad (3)$$

Where S/NoW is the received signal-to-noise ratio (SNR).When the SNR is large (SNR >> 0 dB), the capacity C≈ W $\log_2 P'/N_0W$ is logarithmic in power and approximately linear in bandwidth. This is called the bandwidth-limited regime [5].
When the SNR is small (SNR << 0 dB), the capacity C≈ W $\log_2 e$ is linear in power but insensitive to bandwidth. This is called the power-limited regime.

Rician fading Channel

Rician fading is more applicable when there is a dominant line of sight (LOS) component. Contrary to the Rayleigh channel the Rician channel assumes an additional direct path, the line of sight, between transmitter and receiver [6]. The ratio of signal energy from the direct path and the multipath contributing to the energy of the received signal is expressed by the C factor:

$$C=\frac{|E_0|^2}{\sum_{n-1}^{N}|E_n|^2} \qquad (4)$$

For $C \rightarrow 0$ the Rician Channel approaches the Rayleigh Channel. For $C \rightarrow \infty$ the channel has only the line-of sight path. The Rician distribution is:

$$F(x) = \frac{x}{\sigma^2}\exp\left(\frac{x^2+s^2}{2\sigma^2}\right)B_0 \qquad (5)$$

Where B_0 is the modified Bessel function, $s^2 = |E_0|^2$, the energy of the line-of-sight path. For s = 0 the Rician distribution is a Rayleigh distribution. In the Rayleigh channel is used to simulate mobile reception, while the Rician channel is used for the simulation of fixed receivers.

ADAPTIVE FILTERS ALGORITHMS

Such a filter with adjustable parameters is called an adaptive filter. Adaptive filter adjusts their coefficients to minimize an error signal and can be realized [4]. The most common form of adaptive filter is using least mean square (LMS) algorithm and NLMS algorithm [13].

UNANR System:

The UNANR model of the system performs the function of adaptive noise estimation. The UNANR model of order *M*, as shown in figure 3.7, is a linear, transversal and finite impulse response (FIR) filter. The response of the filter *f(n)* at each time instant (sample) *n* can be written as,

$$f(n) = \sum_{m=1}^{M} w_m(n) r(n-m+1) \qquad (6)$$

Where $w_m(n)$ represents the UNANR coefficients, and $r(n - m + 1)$ denotes the reference input noise at the present $(m = 1)$ and preceding $m - 1, (1 < m \leq M)$, input samples [14]. In order to provide unit gain at DC, the UNANR coefficients should be normalized such that

$$\sum_{m=1}^{M} w_m(n) = 1 \qquad (7)$$

The adaptation process of this model is designed to modify the coefficients that get convolved with the reference input in order to estimate the noise present in the given speech signal [13].

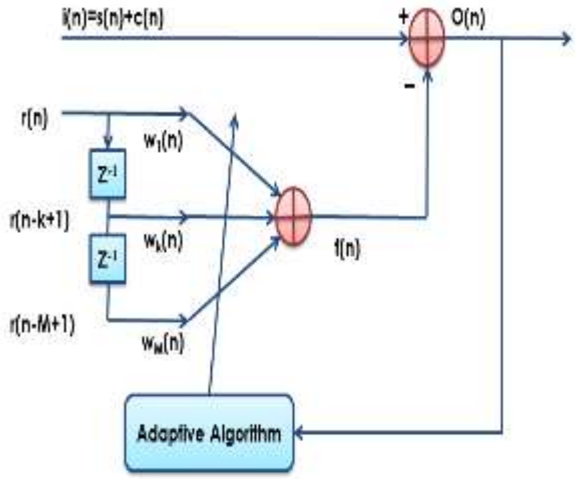

Fig. 2: UNANR Model

To provide the estimated speech signal component, ŝ(n), at the time instant n, the output of the adaptive noise-reduction system subtracts the response of the UNANR model *f(n)* from the primary input *i(n)*, i.e.,

$$\hat{s}(n) = o(n) = i(n) - f(n) \qquad (8)$$

Where the primary input includes the desired speech component and the additive white noise, i.e.

$$i(n) = s(n) + c(n) \qquad (9)$$

Squaring both sides of yields

$$\hat{s}^2(n) = i^2(n) + f^2(n) - 2i(n)f(n) \qquad (10)$$

$$= [s(n) + c(n)]^2 + f^2(n) - 2[s(n) + c(n)]f(n)$$

$$= s^2(n) + 2s(n)\,u(n) + u^2(n) + f^2(n) - 2[s(n) + c(n)]f(n) \qquad (11)$$

Different from the MMSE criterion, the goal of the UNANR coefficient adaptation process is considered to be the minimization of the instantaneous error $\varepsilon(n)$ between the estimated signal power $\hat{s}^2(n)$ and the desired signal power $s^2(n)$, i.e.,

$$\varepsilon(n) = \hat{s}^2 - s^2(n) = c^2(n) + 2s(n)c(n) + f^2(n) - 2[s(n) + c(n)]f(n) \qquad (12)$$

Such a goal can be achieved by optimizing the UNANR coefficients according to the steepest-descent algorithm [13]. The process of convergence in the multidimensional coefficient space follows a deterministic search path provided by the negative gradient direction as:

$$-\nabla w_k \varepsilon(n) = -\frac{\partial f^2(n)}{\partial w_k} + 2\frac{\partial [s(n)+c(n)]f(n)}{\partial w_k} \qquad (13)$$

$$= -2r(n-k+1)\sum_{m=1}^{M} w_m(n)r(n-m+1) - 2i(n)r(n-k+1) \qquad (14)$$

$$= -2r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) - i(n)] \qquad (15)$$

By substituting (13) and (17) into the standard steepest descent algorithm [13], we may derive the UNANR adaptation rule as

$$w_k(n+1) = w_k(n) - \eta\nabla w_k \varepsilon(n) \qquad (16)$$

$$= w_k(n) - 2\eta r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) - i(n)] \qquad (17)$$

$$= w_k(n) + 2\eta r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) \qquad (18)$$

Where η (η > 0) denotes the learning rate that indicates the search amplitude in the negative gradient direction [14]. Before the UNANR model provides its response *f (n +1)* referring to (17), at each time instant *(n+1)*, the estimated coefficients $\hat{w}_k(n+1)$ should be normalized so as to meet the requirement of (18). The UNANR coefficient normalization formulation is given by:

$$\widehat{w}_k(n+1) = \frac{w_k(n+1)}{\sum_{k=1}^{M} w_k(n+1)} \qquad (19)$$

SIMULATION RESULTS DISCUSSION

Firstly speech enhancement is done with online voice signal. There can be total four possible configurations for this experiment because we have AM modulation technique and two types of channels i.e. AWGN and Rician channel is used as communication channels.

Consider the case of online input speech signal first. In this case AM is selected as modulation technique 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 demodulation is performed 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 modulation technique with AWGN and Rician channel and graph is plotted between PSNR v/s SNR as shown in Fig. 3 & 4 and tabulated in Table 2 & 3.

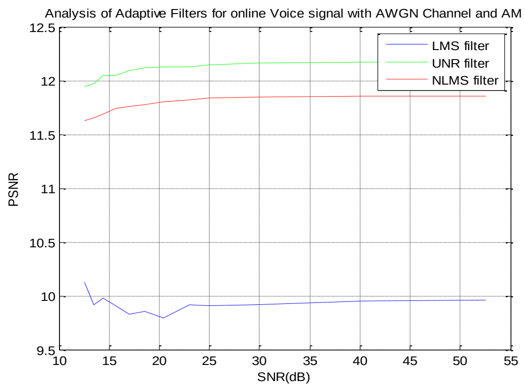

Fig. 3: Adaptive filtering for AM with AWGN channel for online voice

Table 2: Measurement of Parameters of Adaptive Filters between PSNR Vs SNR

Sr. No.	SNR (in dB)	PSNR		
		LMS	NLMS	UNANR
1	12.516	10.126	11.6271	11.9404
2	13.431	9.9156	11.6559	11.97
3	14.454	9.976	11.6887	12.044
4	15.614	9.912	11.74	12.0456
5	16.95	9.83	11.76	12.088
6	18.53	9.854	11.78	12.1188
7	20.47	9.79	11.80	12.1286
8	22.97	9.914	11.82	12.1286
9	24.91	9.910	11.8412	12.1454
10	29.59	9.919	11.8471	12.1588
11	40.47	9.95	11.8518	12.1693
12	52.516	9.96	11.8526	12.1719

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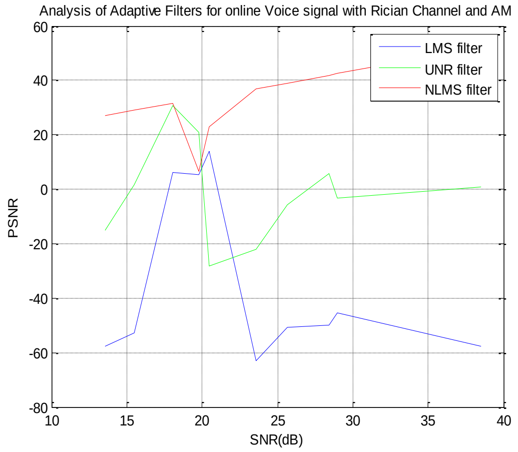

Fig. 4: Adaptive filtering for AM with Rician channel for online voice

Table 3: Measurement of Parameters of Adaptive Filters

Sr. No.	SNR (in dB)	PSNR		
		LMS	NLMS	UNANR
1	13.5282	-57.77	26.93	-14.94
2	15.49	-52.76	28.89	1.56
3	18.03	6.068	31.43	30.87
4	19.8	5.15	6.60	21.044
5	20.47	13.8	22.89	-28.20
6	23.57	-62.90	36.98	-22.20
7	25.66	-50.56	39.07	-5.59
8	28.43	-50.04	41.84	5.75
9	28.99	-45.42	42.40	-3.26
10	38.53	-57.645	51.94	0.794

The performance of LMS, NLMS and UNANR for AM modulation technique with AWGN and Rician channel is plotted between SNR and RMSE v/s SNR as shown in Fig. 5 & 6 and parameters recoded in Table 4 & 5.

It is evident from Fig. 5 and Table 4 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.

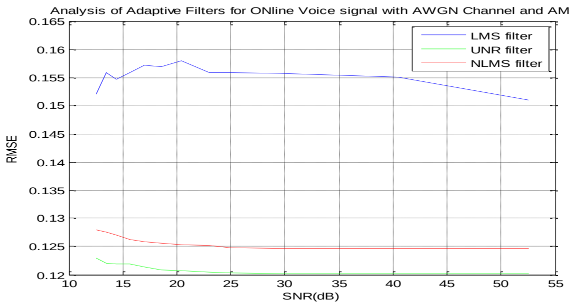

Fig 5: Adaptive filtering for AM with AWGN channel for online voice

Table 4: Measurement of Parameters of Adaptive Filters

Sr.	SNR	RMSE

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No.	(in dB)	LMS	NLMS	UNANR
1	12.516	0.1520	0.1279	0.123
2	13.431	0.1558	0.1275	0.122
3	14.454	0.1547	0.1270	0.1219
4	15.614	0.1558	0.1262	0.1219
5	16.95	0.1571	0.1258	0.1213
6	18.53	0.1569	0.1256	0.1209
7	20.47	0.1580	0.1253	0.1207
8	22.97	0.1558	0.1252	0.1205
9	24.91	0.1559	0.1248	0.1203
10	29.59	0.1557	0.1247	0.1202
11	40.47	0.1550	0.12468	0.1201
12	52.516	0.1510	0.12466	0.1201

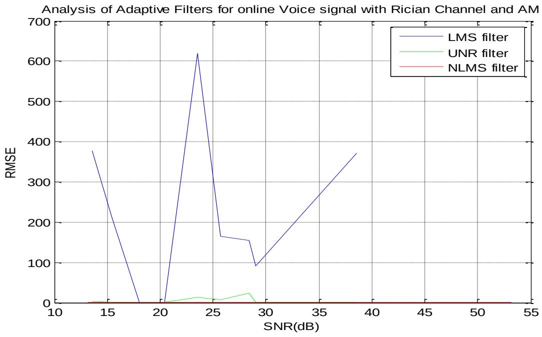

Fig. 6: Adaptive filtering for AM with Rician channel for online voice

Table 5: Measurement of Parameters of Adaptive Filters between RMSE vs SNR

Sr. No.	SNR (in dB)	RMSE LMS	NLMS	UNANR
1	13.5282	377.6	0.382	2.7525
2	15.49	212.244	0.3825	0.4076
3	18.03	0.242	0.3855	0.0139
4	19.8	0.269	0.381	0.1873
5	20.47	0.09912	0.438	0.0423
6	23.57	681.46	0.3824	12.554
7	25.66	164.67	0.3840	23.105
8	28.43	155.166	0.386	0.929
9	28.99	91.159	0.453	0.251
10	38.53	372.8	0.378	0.710

CONCLUSION

In this Paper the performances of the adaptive filters are compared with respect to the variation in SNR (dB). The used modulation technique is AM and the considered channels are AWGN and Rician fading channel. Under speech enhancement techniques, for improving quality of adaptive filters a newly emerging filter is used i.e. UNANR. This filter's performance is compared with two traditionally used adaptive filters; LMS and NLMS. The above considered technologies have been combined using the MATLAB software (Version 7.10) 2013a. Now for different cases for the performance evaluation, the selected range of SNR is 38 to -2.5. However, there is no restriction of the SNR range. But, if SNR range increases then the simulation time will increase, and the considered noise removal capability may decrease. It is necessary to evaluate the performance of the system, and PSNR and RMSE provide a base for comparing the performances of different filters. From all the performed experiments it is seen that for online voice data NLMS and UNANR filters have better performance than LMS.

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