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Studying Contemporary and Efficient Noise Filtering Methods

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ABSTRACT:
The present system of speech enhancement is developing by adaptive filtering approach in digital filters. The adaptive filter utilizes the Normalized Least Mean Square algorithm for noise removal, but in practical application of NLMS algorithm, a key parameter is the step size. As it is known, if the step size is large, the convergence rate of NLMS algorithm will be rapid, but the steady-state mean square error (MSE) will increase. That means speech enhancement has some limitations in SNR improvement and rate of convergence.
In this project an optimal estimation of adaptive filtering using Unbiased and normalized adaptation noise reduction (UNANR) algorithm has been implemented for the noisy speech. The aim of this project is to implement various adaptive noise cancellers for speech enhancement based on gradient steepest descent approach.
The project gives the concept of speech enhancement using Unbiased and normalized adaptation noise reduction (UNANR) algorithm. The speech enhancement method aimed at suppressing the background noise is evolving more slowly than the speech. This UNANR algorithm gives better performance measurement of parameters like signal to noise ratio, rate of convergence and steady-state mean square error. The removal of noise from speech signals has applications ranging from cellular communications to front ends for speech recognition systems.
From this thesis, we can say that the signal to noise improvement in the input signal after UNANR filtering is much higher and it is also simple to implementation compared to that of NLMS filter algorithm. Therefore we conclude that the Unbiased and Normalized adaptation noise reduction (UNANR) algorithm is an efficient adaptive filtering algorithm than Normalized Least Mean Square (NLMS) algorithm.
KEYWORDS: *UNANR FILTER, NLMS FILTER, ADAPTIE FILTER PROCESS, OVERVIEW OF ADAPTIVE NOISE-REDUCTION SYSTEM, GENERAL SETUP OF ADAPTIVE FILTER*

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I. INTRODUCTION

1.7.1 Adaptive Filtering

Adaptive filtering can be considered as a process in which the parameters used for the processing of signals changes according to some criterion. Usually the criterion is the estimated mean squared error or the correlation (K. & M., 2003) (H., M., & A, 2010). The adaptive filters are time-varying since their parameters are continually changing in order to meet a performance requirement. In this sense, an adaptive filter can be interpreted as a filter that performs the approximation step on-line. Usually the definition of the performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design.

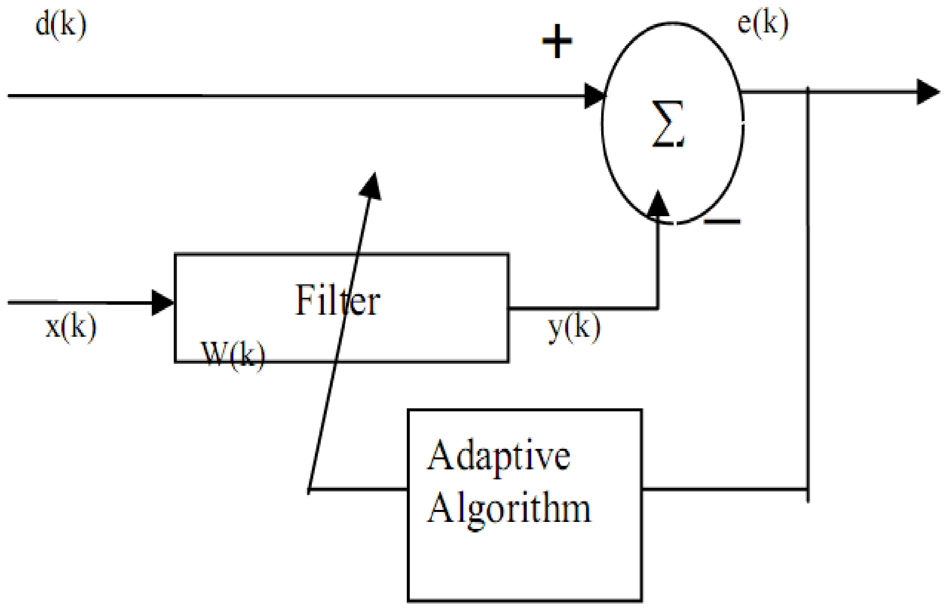

Fig 1.3 General setup of adaptive filter

The general set up of adaptive filtering environment (Sayed A. Hadei, April 2010) (Sambur M. I., 1998) is shown in Fig 1.3, where k is the iteration number, x(k) denotes the input signal, y(k) is the adaptive filter output, and d(k) defines the desired signal. The error signal e(k) is calculated as d(k)-y(k). The error is then used to form a performance function or objective function that is required by the adaptation algorithm in order to determine the appropriate updating of the filter coefficients. The minimization of the objective function implies that the adaptive filter output signal is matching the desired signal in some sense.

5.5 Operation of the Normalised least-mean-square (NLMS) algorithm

The Normalised least-mean-square (NLMS) algorithm consists of two basic processes (Sambur M. I., 1998) (Francesco Beritelli, 1978):

1] A filtering process, which involves (a) computing the output of a transversal filter produce by a set of tap input, and (b) generating on estimation error by computing this output to a desired response.

2] An Adaptive process, which involves the automatic adjustment of the tap weight of the filter in accordance with the estimation error. Thus the combination of these two processes working together constitutes a feedback loop around the NLMS algorithm, as illustrated in fig 5.2. First we have a transversal filter, around which the NLMS algorithm bur is built; this component is responsible for performing the filtering process. Second we have a mechanism for performing the adaptive control process on the tap weight of the transversal filter, hence the designation “adaptive weight control mechanism” in fig 5.1.

We have used a hot over the symbols for the tap-weight vector to distinguish it from the value obtained by using the steepest descent algorithm. Equivalently, we may write the result in the form of three basic relations as follows (Francesco Beritelli, 1978):

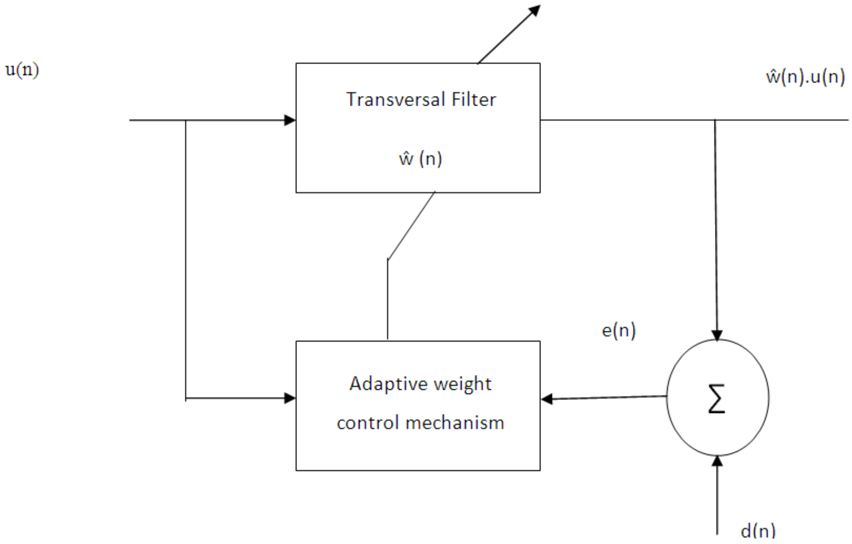

Fig 5.2 Basic block diagram of Adaptive Processing

1] Filter output:

$$Y(n) = \hat{w}(n).u(n) \quad \text{------------------------------- (5.6)}$$

2] Estimation error:

$$E(n) = d(n) - y(n) \quad \text{----------------------------- (5.7)}$$

3] Tap-weight adaptation:

$$\hat{W}(n+1) = \hat{w}(n) + \mu\, u(n)\, e^*(n) \quad \text{--------------------- (5.8)}$$

Equation (5.6) & (5.7) define the estimation error *e(n)* the computation of which is based on the current estimate of the tap-weight vector $\hat{w}(n)$. Note that the second term $\mu u(n)\, e^*(n)$, on the right-hand side of equation (5.8) represents the correlation that is applied to the current estimate of the tap-weight vector, $\hat{w}(n)$. The interactive procedure is started with an initial guess $\hat{w}(0)$.

5.7 Introduction to UNANR Algorithm (UNANR Filtering)

The UNANR model (K. & M., 2003) (H., M., & A, 2010) of the system performs the function of adaptive noise estimation. The UNANR model of order M, as shown in Fig 5.4, is a transversal, linear, finite impulse response (FIR) filter. The response of the filter f(n) at each time instant (sample) n can be expressed as,

$$f(n) = \sum_{m=1}^{M} wm(n)\, r(n-m+1) \quad \text{--------------------- (5.9)}$$

Where wm(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 = M), input samples. In order to provide unit gain at DC, the UNANR coefficients should be normalized such that

$$\sum_{m=1}^{M} wm(n) = 1 \quad \text{--- (5.10)}$$

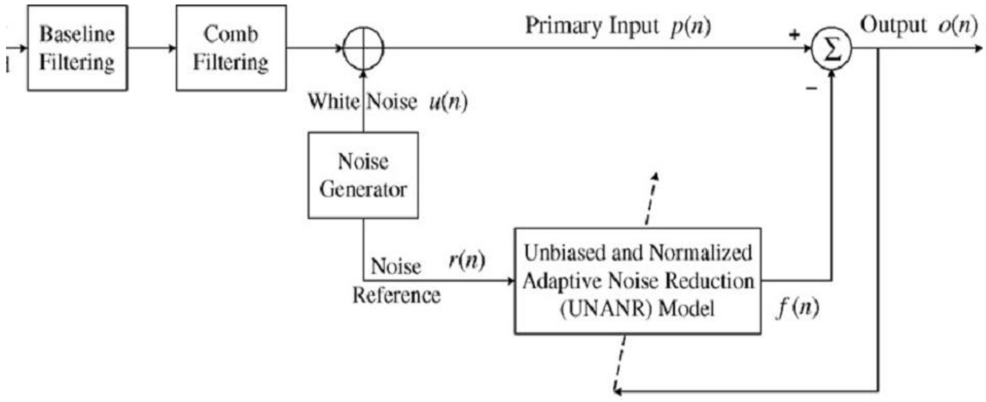

Fig 5.4 Overview of Adaptive noise-reduction system

The adaptation process of the UNANR 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 (Douglas, 1997). To provide the estimated speech signal component s(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 p(n), i.e.,

$$\hat{s}(n) = o(n) = p(n) - f(n) \quad \text{--} \quad (5.11)$$

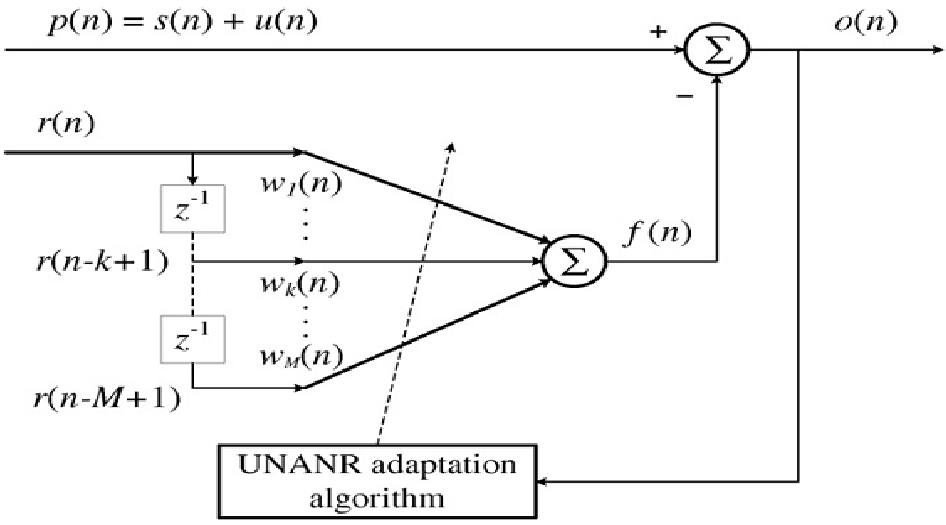

Fig 5.5 Detailed structure of the UNANR model

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

$$p(n) = s(n) + u(n) \quad \text{---} \quad (5.12)$$

Squaring on both sides of equation (5.11)

$$\hat{s}^2(n) = p^2(n) + f^2(n) - 2p(n)f(n)$$
$$= [s(n) + u(n)]^2 + f^2(n) - 2[s(n) + u(n)]f(n)$$
$$= s^2(n) + 2s(n)u(n) + u^2(n) + f^2(n) - 2[s(n) + u(n)]f(n) \quad \text{------------------} \quad (5.13)$$

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

$$\varepsilon(n) = \hat{s}^2(n) - s^2(n) = u^2(n) + 2s(n)u(n) + f^2(n) - 2[s(n) + u(n)]f(n). \quad \text{------------------------} \quad (5.14)$$

II. RESULTS:-

Table 6.2 The performance parameters using NLMS filter and UNANR algorithms with respect to noisy signal for different speech signals

Speech Samples	Before Filtering		
Sample-I	SNR = 28.4973 dB	PSNR = 42.9626 dB	PSNR = 49.7032 dB
		RMSE = 0.00011	RMSE = 5.0629e-
		Time = 0.2249 sec	Time = 0.6411 sec
Sample-II	SNR = 17.5265dB	PSNR = 30.5199 dB	PSNR = 37.1324 dB
		RMSE = 0.000433	RMSE = 0.000202
		Time = 0.21238 sec	Time = 0.62172 sec
Sample-III	SNR = 23.2201 dB	PSNR = 33.8718 dB	PSNR = 41.9642 dB
		RMSE = 0.000257	RMSE = 0.000101
		Time = 0.22212 sec	Time = 0.62613 sec
Sample-IV	SNR = 10.2964 dB	PSNR = 20.8513 dB	PSNR = 27.5632 dB
		RMSE = 0.000877	RMSE = 0.000404
		Time = 0.24031 sec	Time = 0.63726 sec
		PSNR = 16.7419 dB	PSNR = 22.8644 dB

Sample-V	SNR = 1.6489 dB	RMSE = 0.002251	RMSE = 0.001112
		Time = 0.20378 sec	Time = 0.63142 sec
Sample-VI	SNR = 14.5179 dB	PSNR = 29.0258 dB	PSNR = 35.7243dB
		RMSE = 0.000547	RMSE = 0.000253
		Time = 0.18804 sec	Time = 0.63392 sec

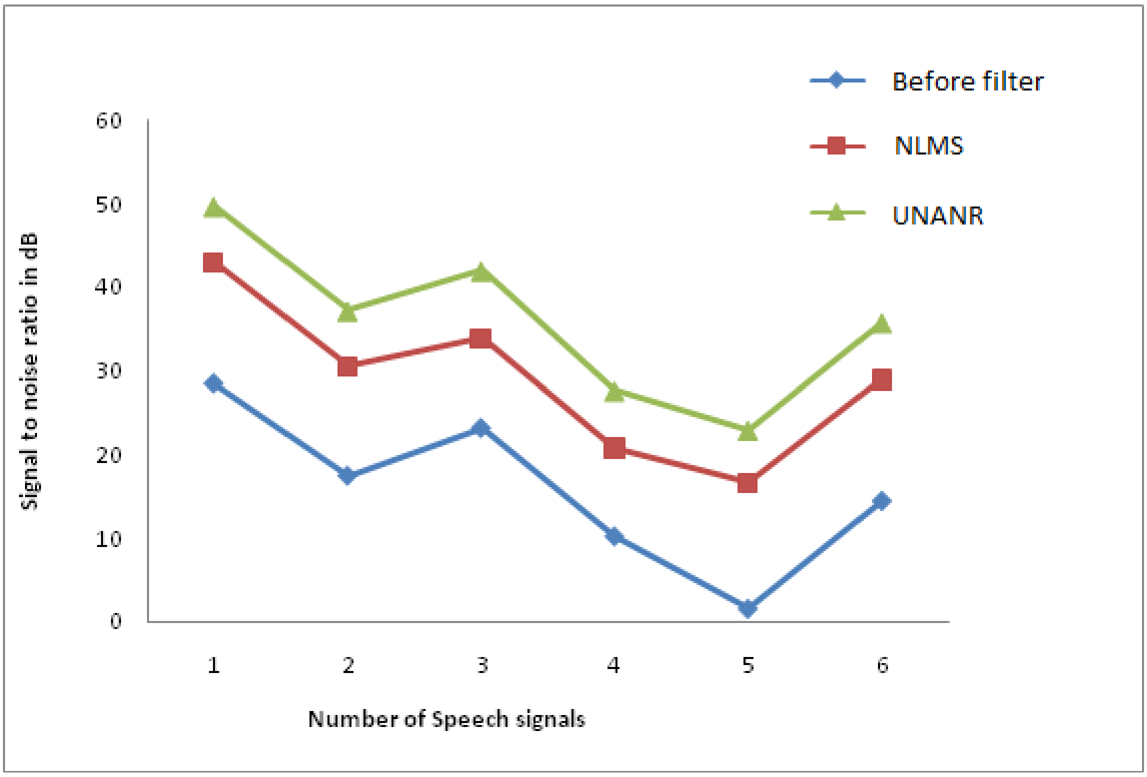

Fig 6.11 Plot of SNR (dB) before filtering process, System's improvement with respect to NLMS filter and UNANR model versus to different samples of speech signal

Here the six speech samples are considered with different input signal to noise ratio to compare the improvement in signal to noise ratio after filtering using NLMS filter and UNANR model algorithm. The speech signal samples are considered without addition of noise and with noisy signal. The reason for this could be that the noise-reduction capability of the system with the NLMS filter does not change much with the nature of primary input, and that the system performance is only determined by the learning rate parameter of the NLMS filter.

On the other hand, regarding the UNANR model, the system's SNR improvement curve increases much higher than NLMS curve and input signal curve. When the input becomes more and more heavily contaminated by white noise, it can be inferred that the proposed UNANR system is better than NLMS filtering algorithm.

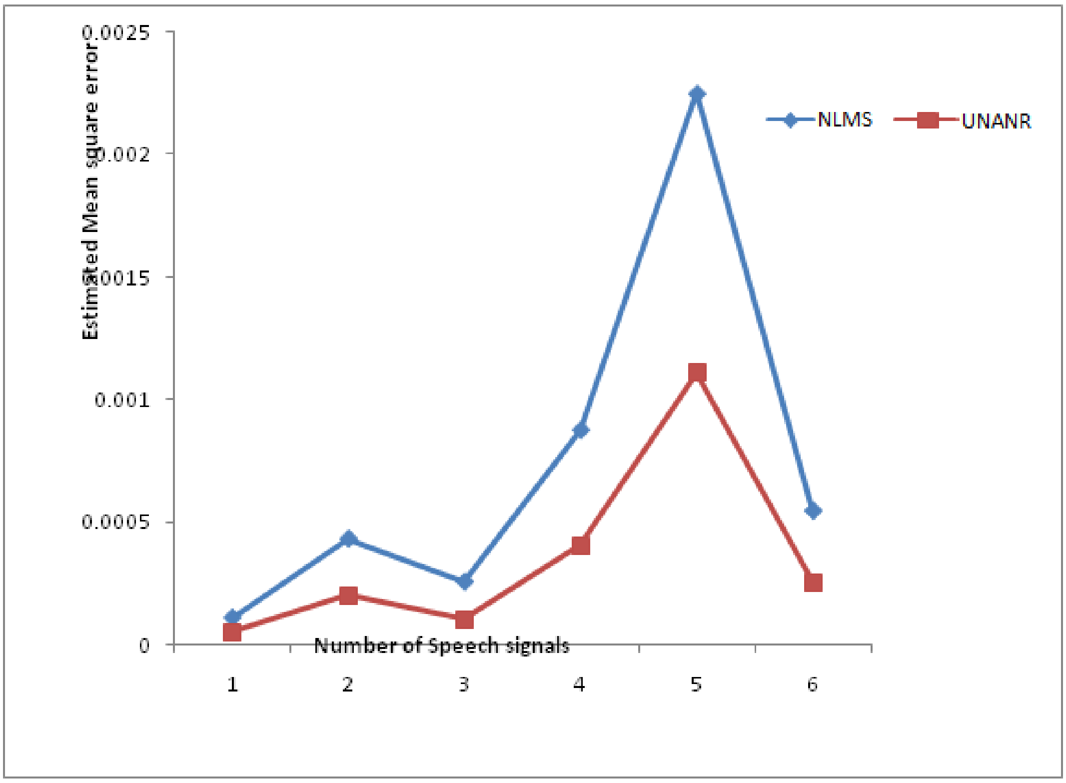

Fig 6.12 Plot of estimated root mean square error (RMSE) versus to different samples of speech signal

Here the five speech sample has shown that the characteristics of Root Mean Square Error (RMSE) variations with respect to the speech samples. The root mean square error depends on signal to noise ratio, as the signal to noise ratio increases the root mean square error decreases respectively. So the NLMS filter algorithm has smaller improvement in SNR compare to UNANR model.

Therefore the above graph shows that NLMS filter algorithm having higher value of root mean square error compared to UNANR algorithm. For different speech samples with variable signal to noise ratio of the input signal, the RMSE varies for NLMS filter algorithm and UNANR model.

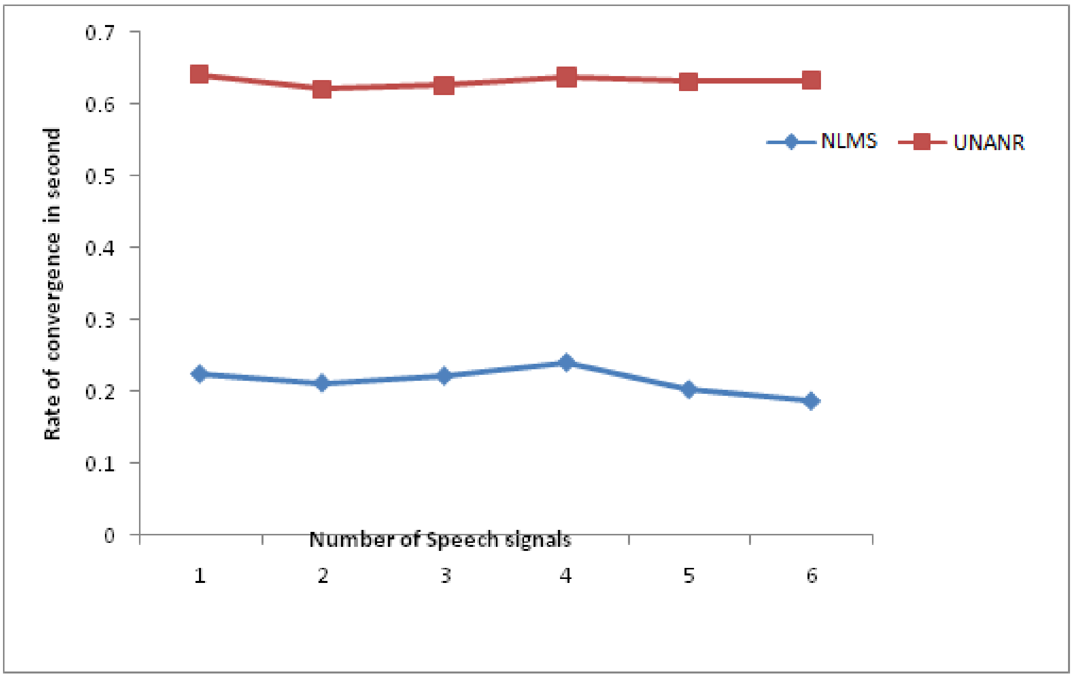

Fig 6.13 Plot of Rate of convergence versus to different samples of speech signal

Here the five speech samples have results shown in table 6.2. The rate of convergence required for both NLMS filter algorithm and UNANR model algorithm compared for all speech signals. From given graph we can see that the time required for filtering speech signals by NLMS algorithm is smaller than UNANR algorithm. The rate of convergence in adaptive filtering process also depends on signal to noise ratio improvement. As the improvement in signal to noise ratio increases the time required to convergence is also increase. But here the rate of convergence is in milliseconds, therefore the time increase will not affect that much to the system. In the case, where the signal to noise ratio improvement is considerable and rate of convergence is not that much part then this model is useful.

Therefore the comparison of different parameters considered for filtering the speech signal using NLMS filter and UNANR model algorithm is done. From this comparison we can say that the signal to noise improvement in the input signal after UNANR filtering is much higher than that NLMS filter algorithm. The UNANR model is also having simple implementation compared to that of NLMS filter algorithm.

III. CONCLUSION

In this report the concept of adaptive digital filtering is introduced by conveying an everyday application in echo cancellation in the telephone system, hands free communication in car driving etc. An introduction to digital filtering was then introduced to give some background on the basic idea of digital filters and why so much work is put into them as opposed to analogue filters. The concept of convolution is introduced, which helps to portray digital filtering as a mathematical process.

The NLMS filter algorithm and UNANR model is introduce as the main adaptive algorithm in the time domain and its operation is examined. An alternative representation of signals in the frequency domain is then introduced, which allows the convolution of two signals to be calculated in a much more efficient manner. The cost of transforming the signals to and from the frequency domain must be accounted for however and for short filter impulse responses it is too high to allow frequency domain filtering replace time domain filtering.

Therefore the comparison of different parameters considered for filtering the speech signal using NLMS filter and UNANR model algorithm is done. From this comparison we can say that the signal to noise improvement in the input signal after UNANR filtering is much higher up to 10dB (50%) than that NLMS filter algorithm and 20 dB than that of original signal. The UNANR model is also having simple implementation compared to that of NLMS filter algorithm. From this project we conclude that the convergence rate of NLMS algorithm compared to UNANR algorithm is also high. But we can also say that the UNANR model is better performance parameter compare to NLMS filter algorithm.

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