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PERFORMANCE ANALYSIS OF ADAPTIVE FILTERS FOR SPEECH SIGNAL

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Abstract

Adaptive filtering has become a spacious area of researcher since last few decades in the field of communication. Adaptive noise cancellation is an approach used for noise reduction in speech signal. The received speech signal at the receiver easily gets corrupted by background and channel noise where both speech signal and noise signal changes continuously with time, then to separate them adaptive filtering is needed. This paper deals with cancellation of noise on speech signal using two old (LMS and NLMS) and one new (UNANR) algorithm. The UNANR (Unbiased and Normalized Adaptive Noise Rejection) model does not contain a bias unit, and the coefficients are adaptively updated by using the steepest-descent algorithm. Two modulation techniques, AM and FM are applied separately in combination with two communication channels i.e. AWGN and Rician. Signal quality parameter PSNR with respect to SNR measured and compared. The results show 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.

I. INTRODUCTION

Acoustic noise problems like industrial equipment noise, contending speakers, background noise, car engine noise, room reverberation, low-quality microphones are subjected on speech signals in real time surroundings. Speech signals are corrupted by these several forms of noise and also they are subjected to distortion caused by communication channels. Traditionally, acoustic noise cancellation used to apply passive techniques such as enclosures, barriers and silencers to remove the unwanted noise signal [1,2]. Silencers were important for noise cancellation over broad frequency range but expensive and not efficient at low frequencies. Mechanical vibration is a type of noise that creates problems in all areas of communication and electronic appliances. In all such situations extraction of high quality signals is an ideal task. There are two types of unvoiced speech signals. One type of unvoiced signal contains a huge number of low-frequency components and mass speech energy and another type is with high frequency and low energy, which is very similar to that of the noise. Thus, a separation of voiced and unvoiced signal is proposed in this paper in presence of channel noise also.

Solution of this problem can be given by filtering only. Mainly filtering techniques are broadly divided in non-adaptive and adaptive filtering techniques. Practically, the statistical nature of all speech signals is non-stationary; as a result non-adaptive filtering may not be suitable. Extracting or enhancing the desired information from a mixture of conflicting information is a simplest form of signal processing. Signal processing gives operations like extracting, enhancing, storing, and transmitting useful information. In conventional filter design techniques, adaptive filters do not have constant filter coefficients and no priori information is known. 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. In this paper, noise is defined as any kind of undesirable signal, whether it is suffered by electrical, acoustic, vibration or any other kind of media like sounds produced by machinery in mechanical industries. In this paper, adaptive algorithms are applied to both background noise and channel noise.

II. ADAPTIVE FILTERS

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 unchanged. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics. Adaptive noise cancellation (ANC) efficiently attenuates low frequency noise for which passive methods are ineffective.

A. Basic Adaptive Filter Structure

Suppose 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 $C(n)$. 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 $e = \{S(n) + C(n)\} - f(n)$, then

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

$$= \{C(n) - f(n)\}^2 + S(n)^2 + 2\,S(n)\,C(n) - 2f(n)S(n) \qquad (1)$$

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

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

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. Minimizing the MSE results in a filter error output *f(n)* that is the best least-squares estimate of the signal *S(n)*.

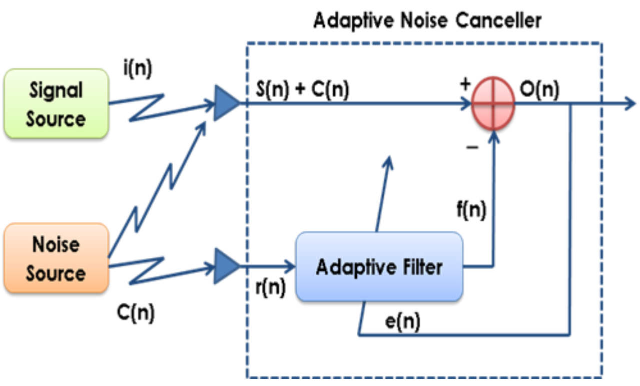


Fig. 1: Adaptive Filter Structure

B. Conventional LMS Algorithms

The LMS algorithm is a method to estimate gradient vector with instantaneous value. It changes the filter tap weights so that e(n) is minimized in the mean-square sense. The conventional LMS algorithm is a stochastic implementation of the steepest descent algorithm. It simply replaces the cost function $\xi(n) = E[e^2(n)]$ by its instantaneous coarse estimate.
Coefficient updating equation for LMS is given by,

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \mu\, \boldsymbol{i}(n)\, e(n) \qquad (3)$$

Where μ is an appropriate step size to be chosen as $0 < \mu < \frac{2}{trR}$ for the convergence of the algorithm.

C. 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 2, 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,

$$f(n) = \sum_{m=1}^{M} w_m(n)\, r(n - m + 1) \qquad (4)$$

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 $m - 1$, ($1 < m \le M$), input samples. In order to provide unit gain at DC, the UNANR coefficients should be normalized such that:

$$\sum_{m=1}^{M} w_m(n) = 1 \qquad (5)$$

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.

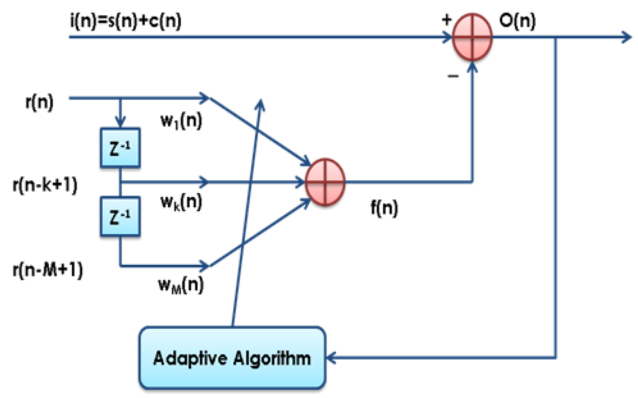


Fig. 2: UNANR Model

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.,

$$\hat{s}(n) = o(n) = i(n) - f(n) \qquad (6)$$

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

$$i(n) = s(n) + c(n). \qquad (7)$$

Squaring both sides of (7) yields

$$\hat{s}^2(n) = i^2(n) + f^2(n) - 2i(n)\, f(n) \qquad …….(10)$$
$$= [s(n) + c(n)]^2 + f^2(n) - 2[s(n) + c(n)]\, f(n)$$
$$= s^2(n) + 2s(n)u(n) + u^2(n) + f^2(n) - 2[s(n) + c(n)]f(n) \qquad (8)$$

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

$$\varepsilon(n) = \hat{s}^2(n) - s^2(n) = c^2(n) + 2s(n)c(n) + f^2(n) - 2[s(n)+c(n)]f(n) \qquad (9)$$

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:

$$-\nabla w_k \varepsilon(n) = -\frac{\partial f^2(n)}{\partial w_k} + 2\frac{\partial[s(n)+c(n)]f(n)}{\partial w_k}$$
$$= -2r(n-k+1)\sum_{m=1}^{M} w_m(n)r(n-m+1) - 2i(n)r(n-k+1)$$
$$= -2r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) - i(n)] \quad (10)$$

In above equations we may derive the UNANR adaptation rule as derive following equation (11).

$$w_k(n+1) = w_k(n) - \eta\nabla w_k \varepsilon(n)$$
$$= w_k(n) - 2\eta r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) - i(n)]$$
$$= w_k(n) + 2\eta r(n-k+1)[\sum_{m=1}^{M} w_m(n)r(n-m+1) \quad (11)$$

Where η(η > 0) represents the learning rate that indicates the search magnitude in the negative gradient direction. Before the UNANR model provides its response f (n +1) referring to (13), at each time instant (n+1), the estimated coefficients $\hat{w}_k(n+1)$ should be normalized so as to meet the requirement of (11). The UNANR coefficient normalization formulation is given by in equation (12):

$$\hat{w}_k(n+1) = \frac{w_k(n+1)}{\sum_{k=1}^{M} w_k(n+1)} \quad (12)$$

III. SIMULATION RESULTS

In the conducting tests the speech signal contaminated with a field noise is given as input to the adaptive filter and a reference signal which must be somewhat interrelated with noise in the been plotted to check the performance of the filters. Graphs are plotted between SNR v/s PSNR and SNR v/s RMSE. Adaptive input. The noise signal is given as reference signal. Graphs have filtering is done with modulation techniques viz. AM and FM for AWGN and Rician channels. The filtering results are shown in Graph 1-4.

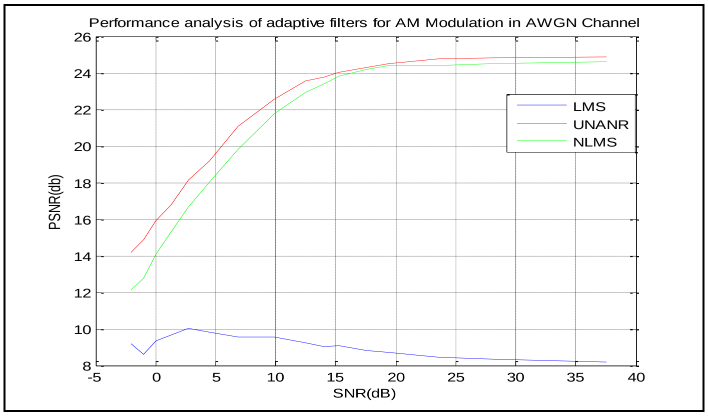


Fig.3: Adaptive filtering when AM Modulation is used for AWGN Channel

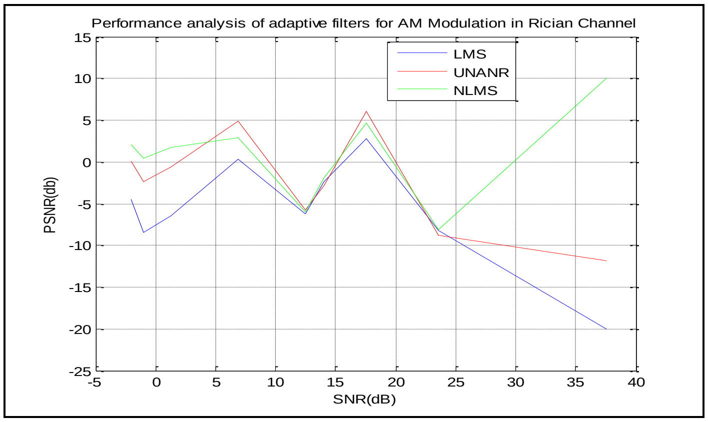


Fig.4: Adaptive filtering when AM modulation is used for Rician channel

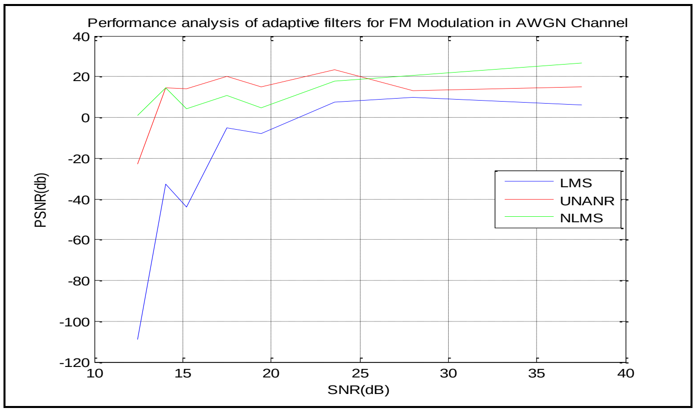
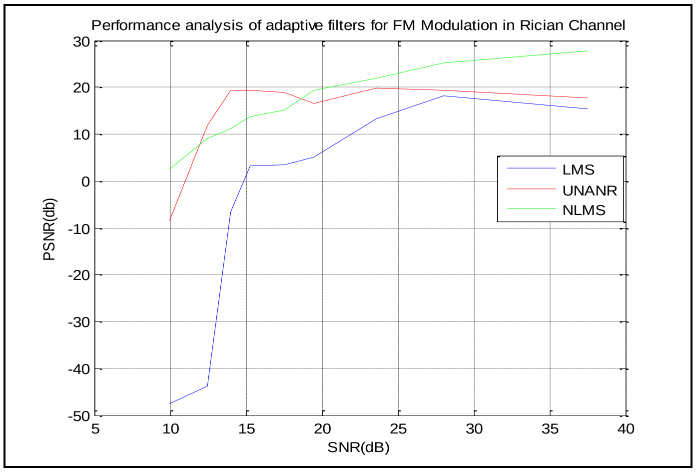


Fig.5: Adaptive filtering when FM modulation is used for AWGN channel



Graph-4: Adaptive filtering when FM modulation is used for Rician channel

IV. CONCLUSION

In this paper adaptive filtering is presented for removal of noise from speech signals in presence of channel noise. As a result, the steps are related to the filtering remains unchanged. The introduced system removes background noise as well as channel noise at very great extent when modulation is used. Our simulations, however, confirm that the ability of UNANR algorithms is better than conventional LMS and NLMS algorithms when AM is used with AWGN channel and better than LMS elsewhere. Hence these algorithms are acceptable for all practical purposes.

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