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# ADAPTIVE LMS FILTERING APPROACH FOR SPEECH ENHANCEMENT

*[1] LALIT P. PATIL, [2] AMITESHWAR BHALAVI, [3] PROF. RUPESH DUBEY*

**[1] ME(EC) , IES IPS'A, INDORE**
**[2] Asst. Professor(EC), IES IPS'A, INDORE**
**[3] HOD (EC) , IES IPS'A, INDORE**

***lalitpatil5589@rediffmail.com, amiteshwarbhalavi@gmail.com, rupeshdubey007@gmail.com***

***ABSTRACT:*** *This paper presents an efficient algorithm for high-quality speech with removal of noise. An optimal estimate of adaptive filtering using least mean algorithm has been implemented for the observed noisy speech. This Algorithm is basic adaptive algorithm has been extensively used in many applications as a consequence of its simplicity and robustness.*

***Keywords — Adaptive filtering, LMS algorithm, MSE, Noise cancellation, Speech enhancement.***

**Introduction:** Normally speech signals are corrupted by several forms of noise such as competing speakers, background noise, car noise, and also they are subjected to distortion caused by communication channels; E.g. examples are room reverberation, low-quality microphones, etc. In such Conditions extraction of high resolution signals is a done by filtering method. Basically filtering techniques are broadly classified as non-adaptive and adaptive filtering techniques. In this paper LMS adaptive filtering has become one of the effective and popular approaches for the speech enhancement. Speech enhancement improves the signal quality by suppression of noise and reduction of distortion. Speech enhancement has many applications; for example, mobile communications, robust speech recognition, low-quality audio devices, and hearing aids etc.

**Adaptive Filter:** An adaptive filter comprises of two basic components [2], these are a digital filter and an adaptive algorithm. The digital filter produces an output in response to an input signal and the adaptive algorithm is responsible for adjusting the coefficients of the digital filter.

The block of an adaptive filter is shown in Figure 2. The signal d (n) is called the desired signal. The input and the output of the filter are denoted by x (n) and y (n) respectively. The signal e (n) is called the estimation error and is defined by e (n) =d (n)-y (n). The error signal contributes some objective function and the adaptive algorithm aims to minimize those functions.

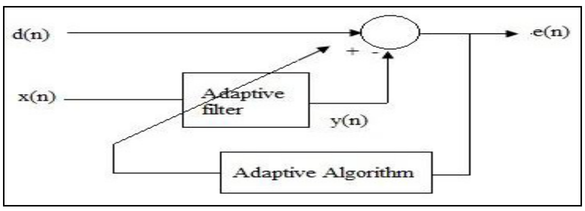


**Fig1: Adaptive filter block diagram.**

**Adaptive Noise Cancellation :** The block diagram of a dual input adaptive noise canceller is shown in Figure 3. The adaptive noise canceller mainly consists of two sensors: primary sensor, which intends to supply a desired signal along with noise and a reference sensor which is responsible for supplying noise alone.

The signal and noise at the output of the primary sensor are uncorrelated and the noise at the output of the reference sensor is correlated with the noise component of the primary-sensor output. The adaptive filter operates on the reference sensor output and thus produces an estimate of the noise and this is subtracted from the primary sensor output [2]. The adjustments applied to the tap weights in the adaptive filter are controlled with the aid of the overall output of the adaptive noise canceller. The adaptive canceller tends to minimize the mean-square error (MSE) value of the overall output, thereby causing the output to be the best estimate of the desired signal in the minimum-mean-square error sense[3].

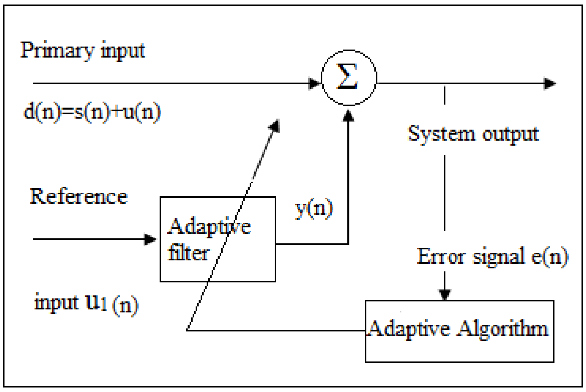


**Fig2: Adaptive Noise Canceller**

$$Y(n) = \sum_{i=0}^{N-1} Wi(n)x(n-i)$$

$$e(n) = d(n) - y(n)$$

We assume that the signals involved are real-valued. The LMS algorithm changes (adapts) the filter tap weights so that e(n) is minimized in the mean square sense. When the processes x(n) & d(n) are jointly stationary, this algorithm converges to a set of tap-weights which, on average, are equal to the Wiener-Hopf solution.

The LMS algorithm is a practical scheme for realizing Wiener filters, without explicitly solving the Wiener-Hopf equation. The conventional LMS algorithm is a stochastic implementation of the steepest descent algorithm. It simply replaces the cost function.

$$\zeta = E[e2(n)]$$

Substituting

ζ= e2(n)

For in the steepest descent recursion, we obtain

$\overline{W}(n-1) = \overline{W}(n) - \mu\nabla\theta^2(n^n)$…. W N-1(n)

Note that the I$^{th}$ element of the gradient vector

$$\nabla = \frac{\partial}{\partial w0}\ \frac{\partial}{\partial w1} \dots\dots \frac{\partial}{\partial wn-1}$$

$$\frac{\partial e^2(n)}{\partial w1} = 2e(n)\frac{\partial e^2(n)}{\partial w1}$$

$$= -2e(n)\frac{\partial y(n)}{\partial w1}$$

$$= -2e(n)x(n-1)$$

**The LMS algorithm:**

Input:

Tap -weight vector

Input vector

Desired output: d

Output:

Filter output: y

Tap -weight vector update:

1. Filtering: y
2. Error estimation.
3. Tap-weight vector adaption.

**Simulation results:**

| | Before Filtering | After Filtering |
|---|---|---|
| Speech Sample | SNR = 28.3582 dB | PSNR = 38.560 dB |
| | | RMSE = 0.0001 |
| | | Time = 2.6403 sec |
| | SNR = 18.8158 dB | PSNR = 29.238 dB |
| | | RMSE = 0.0003 |
| | | Time = 0.1337 sec |

**Conclusion:**

In this paper the problem of noise removal from speech signals using LMS adaptive filtering is presented. Chosen the steps related to the filtering are taken. The proposed treatment, however exploits the modifications in the weight update formula for all categories to its advantage and thus pushes up the speed over the respective LMS-based realizations. Our simulations, however, confirm that the ability of LMS algorithms is better than other algorithms in terms of SNR improvement and convergence rate. Hence these algorithm acceptable for all practical purposes.

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