Performance Analysis of Speech Enhancement Techniques Using Adaptive Algorithms

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Abstract- *Speech Enhancement is one of the techniques used to provide good quality of a speech signal. Noise is one of the major factors that effect any kind of speech signal. To eliminate the noise present in the speech signal different adaptive algorithms are used. In this paper speech enhancement performance is estimated by using adaptive algorithms like LMS, NLMS and PNLMS respectively. A Two stage adaptive filtering is observed for airport noise for frequencies such as 0dB, 5dB and 10dB.These algorithms are used to analyze the speech signal quality. In adaptive filters, to identify an unknown system or cancel noise in the input signal, the filter coefficients adjust themselves to achieve the desired result. The effectiveness of these algorithms are evaluated by using Matlab 2017a software tool. Based on the formulae the evaluation is done to calculate and obtain better SNR values of every speech signal.*

Keywords- Speech enhancement, Noise, adaptive algorithm, LMS, NLMS, PNLMS, SNR

I. INTRODUCTION

Speech enhancement is an application of speech communication. Speech signals are degraded in ways that limit their effectiveness for communication. In such cases speech enhancement techniques can be applied to improve the speech quality. In real time environment the speech signals are corrupted by man made noises. Adaptive filtering methods are used in cancelling such noises. An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm.

The objective of speech enhancement is to empower speech quality by using several algorithms. It is one of the significant topics to enhance the performance of the systems of noisy in speech signal processing. It has many applications like cellular environments, telecommunication signal enhancement, frontends for speech recognition system etc. Various techniques are modeled for this purpose to improve the speech signal to noise ratio (SNR) and the performances depend on quality and intelligibility of the processed speech signal. Here we use three different adaptive algorithms such as LMS, NLMS and PNLMS.

II. ADAPTIVE FILTERING

An adaptive filter is a computational device that attempts to model the relationship between two signals in real time in an iterative manner. An adaptive filter is defined by four aspects.

1. The signal being processed by the filter
2. The structure that defines how the output signal of the filter is computed from its input signal
3. The parameters within this structure that can be iteratively changed to alter the filters input–output relationship
4. The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next.

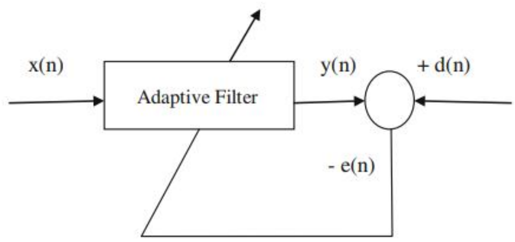


Fig.1 Adaptive filter

III. ADAPTIVE ALGORITHMS

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. In this section we discuss about three adaptive algorithms such as:

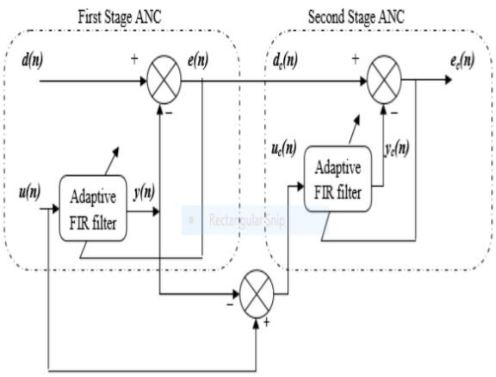


Fig.2 Proposed model

3.1 Least Mean Square

LMS is an adaptive filtering algorithm which is used in many applications for its low computational complexity, high convergence and implementation simplicity. LMS algorithm is also used in noise cancellation applications. The Least Mean Square (LMS) algorithm computes an error and a filter output in each and every cycle which is equal to the difference between desired response and the current filter output. It was invented in 1960 by Stanford university professor Bernard Widrow and his first Ph.D student Ted Hoff.

The basic weight update equation is given as:

$$w(n+1) = w(n) + \mu\, x(n) e(n)$$

Where e(n) is the error signal,
x(n) is the input signal,
w(n) is the weight vector.

3.2 Normalized Least Mean Square

The Normalized Least Mean Square (NLMS) algorithm is a variant of the LMS algorithm. In the earlier LMS algorithms, the step size μ is fixed based on the statistics of the input signal which causes slow convergence. Generally in the noisy environment, the statistics of the input signal are unknown. NLMS filter is structurally same as that of LMS, but it differs in the way that the taps weights are updated. In the LMS algorithm the weight adjustment is directly proportional to the tap input vector, x(n).

In this method, the step size μ is normalized and it is expressed as

$$w(n+1) = w(n) + \mu \frac{x(n)}{x^T(n) x(n)} e(n)$$

Where e(n) is the error signal,
x(n) is the input signal,
w(n) is the weight vector.

3.3 Proportionate Normalized LMS

In order to track sparse impulse response faster Proportionate NLMS (PNLMS) was introduced from the NLMS equation. The coefficient update equation of the PNLMS slightly differs from NLMS with the extra step size update matrix X as shown below and the rest of the terms are carried over from NLMS

$$w(n+1) = w(n) + \frac{\mu X(n) x(n) e(n)}{x^T(n) X(n) x(n) + \delta_p}$$

$$Q(n-1) = \mathrm{diag}\{q_0(n), q_1(n) \ldots \ldots, q_{L-1(n-1)}\}$$

$$q_i(n) = \frac{k_i(n)}{\frac{1}{L}\sum_{i=0}^{L-1} k_i(n)}$$

$$k_i(n) = \max\{\rho . \max\{\gamma, |\hat{h}_0(n)| \ldots |\hat{h}_{L-1}(n)|\}, |\hat{h}_0(n)|\}$$

$$\rho = \frac{5}{L} \quad \gamma = 0.01$$

Where e(n) is the error signal,
x(n) is the input signal,
w(n) is the weight vector and
$\delta_p = \delta_{NLMS}/L$

IV. IMPLEMENTATION

In this paper the algorithm is implemented in two stage. Firstly the input clean speech signal is given as an input to the filter and then a noisy speech signal of different frequencies is given. These samples of speech and noise are taken from TIMIT data base. Both the signals are mixed to give a noisy output in stage one. The signal is used with the help of a Kaiser window. This is a huge advantage over other windows where the window length, ripple size and transition bandwidth have a three-way tradeoff. The stage one output is fed as an input to the second stage which is again mixed with the same noise to produce a noiseless output.
Here, the SNR values of every sample are calculated.

V. SIMULATION RESULTS

The graphs obtained are shown below and the values obtained are analyzed and shown below in tabular form:

The below shown figures are the outputs belonging to airport noise 0dB gm010 sample respectively.

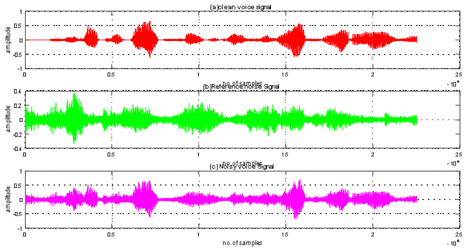
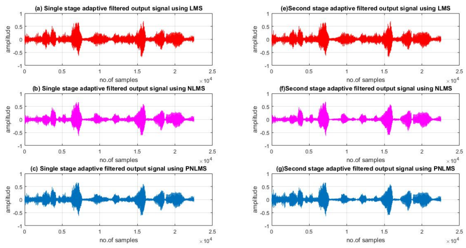


Fig (a) clean voice signal
Fig (b) reference noise signal
Fig(c) noisy voice signal



Fig(a)single stage LMS Fig(e) second stage LMS
Fig(b)single stage NLMS Fig(f) second stage NLMS
Fig(c)single stage PNLMS Fig(g)second stage PNLMS

AIRPORT		gm010	male	pr009	pr020	s1_1	average
0dB	SNRBF	4.1727	3.7246	9.2774	6.8618	0.5869	4.92468
	SNRLMS1	4.4574	4.0051	9.5441	7.1395	0.8685	5.20292
	SNRLMS2	4.6807	4.2241	9.7483	7.3562	1.0887	5.4196
	SNRNLMS1	10.7306	10.3355	15.3252	13.285	7.1726	11.36978
	SNRNLMS2	13.9834	13.5409	17.9307	16.3472	10.3962	14.43968
	SNRPNLMS1	11.861	11.5314	16.4223	14.4749	8.3566	12.52924
	SNRPNLMS2	15.7749	15.4209	19.281	18.1461	12.295	16.18358
5dB	SNRBF	6.1164	5.6684	11.2211	8.8056	2.5306	6.86842
	SNRLMS1	6.4415	5.9928	11.518	9.1235	2.8528	7.18572
	SNRLMS2	6.7005	6.2512	11.7469	9.3755	3.1089	7.4366
	SNRNLMS1	12.1852	11.9334	15.4302	14.551	8.6276	12.54548
	SNRNLMS2	15.228	15.1741	15.6031	16.9064	11.819	14.94612
	SNRPNLMS1	13.2823	13.113	16.348	15.7317	9.7784	13.65068
	SNRPNLMS2	17.1326	17.0966	16.0505	18.61	13.8	16.53794
10dB	SNRBF	6.9795	6.5314	12.0842	9.6686	3.3937	7.73148
	SNRLMS1	7.3432	6.8941	12.4209	10.0259	3.755	8.08782
	SNRLMS2	7.6418	7.1915	12.6894	10.3181	4.051	8.37836
	SNRNLMS1	12.7064	12.4709	15.752	14.4863	9.0064	12.8844
	SNRNLMS2	15.4267	15.2398	16.1733	14.633	12.2194	14.73844
	SNRPNLMS1	13.7068	13.4917	16.5786	15.443	10.1272	13.86946
	SNRPNLMS2	17.1531	17.1919	16.6003	16.2815	14.1113	16.26762

VI. CONCLUSION

In this paper, the adaptive noise reduction algorithms such as LMS, NLMS and PNLMS are proposed enhancing the speech signal. The simulation is carried out under different SNR (0dB, 5dB and 10dB) levels under airport noise for the proposed and existing algorithms. From the experimental results, it is observed that the proposed algorithms have been improved from stage 1 to stage 2 respectively. In addition noise is reduced in the enhanced signal.

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