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Adaptive Noise Canceller

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Abstract- Adaptive Noise Cancellation is an alternative technique of estimating signals corrupted by additive noise or interference. Its advantages lies in that, with no prior estimates of signal or noise, levels of noise rejection are attainable that would be difficult or impossible to achieve by other signal processing methods of removing noise. In this paper we describe the noise cancellation using the Recursive Least Square (RLS) and Least Mean Square (LMS) algorithms to remove noise from an input signal.
Keywords – Noise Signal, Adaptive Filter, RLS Algorithm, LMS Algorithm

I. INTRODUCTION

Linear filtering is required in a variety of applications. A filter will be optimal only if it is designed with some knowledge about the input data. If this information is unknown, adaptive filters are used. The adjustable parameters in the filter are assigned with values based on the estimated nature of the signals. So, these filters are adaptable to the changing environment. These filters have advantage over the linear filter that its frequency response is adjustable to improve the performance in accordance with some criterion. Because of their self adjusting performance and in-build flexibility, adaptive filters are used in various applications like adaptive noise cancelling, line enhancing, frequency tracking, channel equalization etc.

II. ADAPTIVE NOISE CANCELLATION-PRINCIPLE

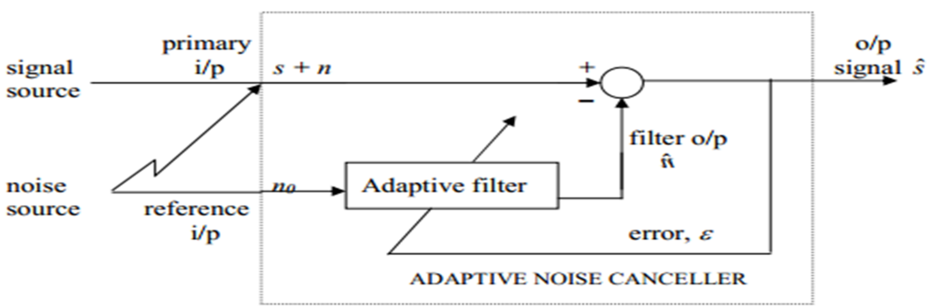

Fig. 1 Adaptive Noise Canceller

As shown in fig., an adaptive noise canceller(ANC) has two inputs-primary and reference. The primary input receives a signal 's' from the signal source that is corrupted by the presence of noise 'n' uncorrelated with the signal. The reference input receives a noise 'n0' uncorrelated with the signal but correlated in some way with the noise 'n'. The noise 'n0' passes through a filter to produce an output 'n^' that is a close estimate of primary input noise. This noise estimate is subtracted from the corrupted signal to produce an estimate of the signal at 's^', the ANC system output.

The commonly used adaptive algorithms which have found widespread application are Least Mean Square(LMS) and the Recursive Least Square(RLS).

III. RECURSIVE LEAST SQUARE (RLS) ALGORITHM

The Recursive Least Square (RLS) adaptive filter is an algorithm which recursively finds the filter coefficients that minimize a weighted linear squares cost function relating to the input signals. The RLS algorithms are well known for their excellent performance when working in time varying environments but at the cost of an increased computational complexity and some stability problems. In this algorithm the filter tap weight vector is updated using Eq.

w(n) = wT (n-1) + k(n) en-1(n) ..….. (1)
k(n) = u(n) / (λ + XT (n) u(n)) …….(2)
u(n) = wλ-1(n-1) X(n) ....(3)
Eq. (2) and (3) are intermediate gain vector used to compute tap weights.

Where λ is a small positive constant very close to, but smaller than 1. The filter output is calculated using the tap weights of above iteration and the current inout vector as in eq.(4)
yn-1 (n) = w*T* (n-1) X(n) …….(4)
en-1 (n) = d(n) – yn-1(n) …….(5)
In the RLS algorithm [1] the estimate of previous samples of output signal, error signal and filter weight is required that leads to higher memory requirements.

IV. LEAST MEAN SQUARE (LMS) ALGORITHM

It is class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.
The basic idea behind LMS filter is to approach the optimum filter weights (R-1 P), by updating the filter weights in a manner to converge to the optimum filter weight. The algorithms starts by assuming a small weights (zero in most cases), and at each step, by finding the gradient of the mean square error, the weights are updated. That is, if the MSE-gradient is positive, it implies the error would keep increasing positively, if the same weight is used for further iteration, which means we need to the weights. In the same way, if the gradient is negative, we need to increase the weights. So, the basic weight update eq. is
w*n*+1 = w*n* – μΔε [n]
where , **ε** represents the mean-square error. The negative sign indicates that, we need to change the weights in a direction opposite to that of the gradient slope.

LMS algorithm summary:
The LMS algorithm [1] for a pth order algorithm can be summarized as

Parameters: P=filer order
μ= step size
Initialization: ĥ (0) = 0
Computation: For n=0, 1, 2….
X(n)=[x(n),), x(n - 1), …, x(n – p + 1)]T
e (n) = d(n) – ĥH (n) X(n)
ĥ (n+1) = ĥ(n) + μ e*(n) X(n)

V. PERFORMANCE COMPARISON OF ADAPTIVE ALGORITHMS

Performance Comparison of Adaptive Algorithms **S. No.**	Algorithms	MSE	Complexity	Stability
1.	RLS	6.2*10^(-3)	4N^2	High Stable
2.	LMS	1.5*10^(-2)	2N+1	Less Stable

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From the table , it is clear that , the performance of RLS adaptive algorithm is high as compared to other algorithm due to the less mean-square error (MSE).

VI. CONCLUSION

Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal. The principal advantage of the method are its adaptive capability, its low output noise, its low signal distortion. The adaptive capability allows the processing of inputs whose properties are unknown and in some cases non-stationary. Output noise and signal distortion are generally lower than that achieved with conventional optimal filter configurations.

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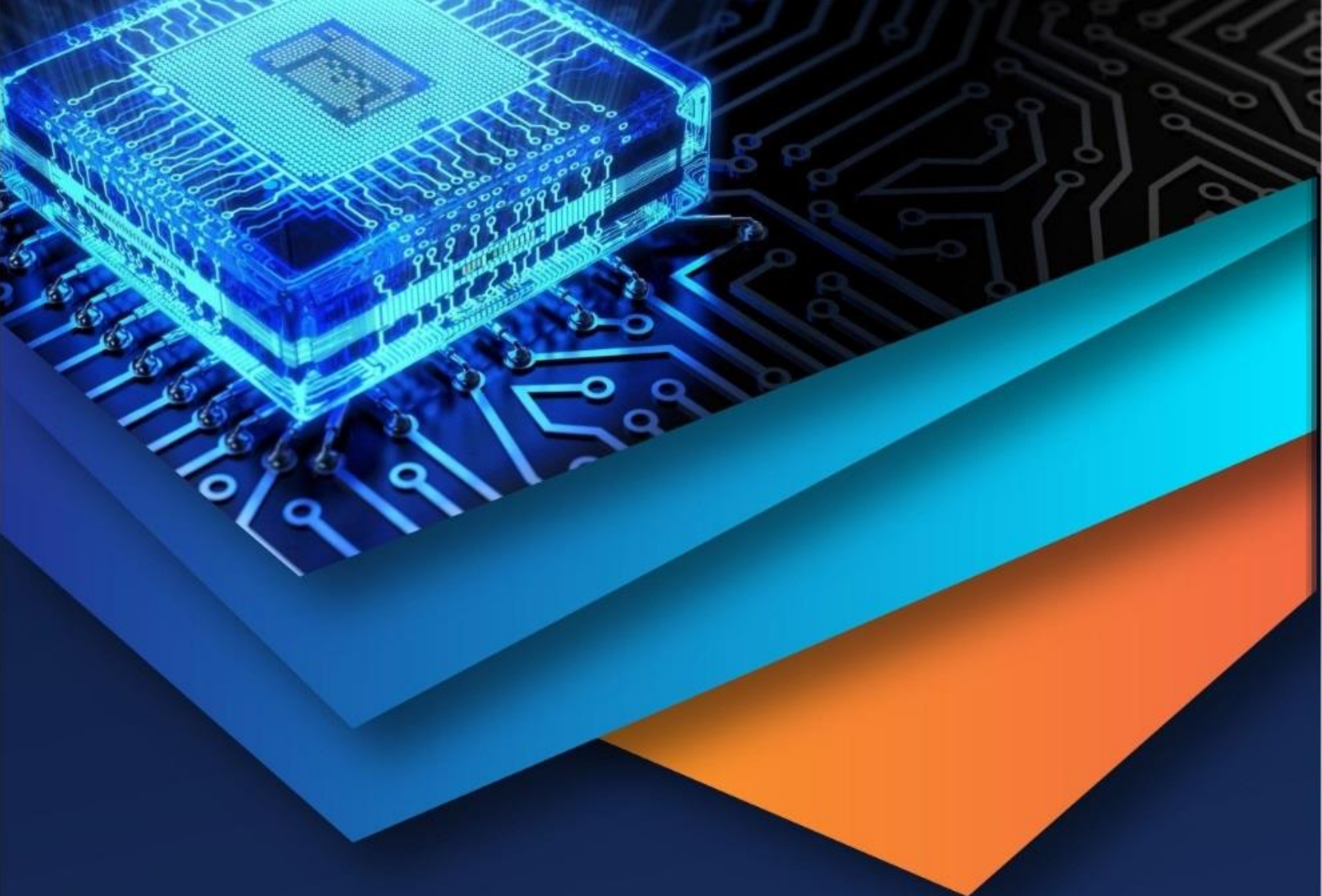

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