Problem Formulation of LMS Filter

K Samhitha[1] G Anusha[2]

[1]EE16BTECH11019

[2]EE16BTECH11011

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What is Least Mean Squares Algorithm ?

Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square 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.

Problem Formulation

The signal-noise waveform d(n) contains a human voice along with an instrument sound in the background. This sound is captured in noise waveform X(n). The goal is to suppress X(n) in d_n.

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

Where e(n) is the desired signal

Cont..

We want an estimate of W(n) from X(n). This can be done by considering

$$y(n) = W^T(n)X(n)$$

Where

$$X(n) = \begin{pmatrix} X(n) \\ X(n-1) \\ X(n-2) \\ .. \\ .. \\ X(n-M+1) \end{pmatrix}_{MX1}$$

Cont..

$$W(n) = \begin{pmatrix} w_1(n) \\ w_2(n) \\ w_3(n) \\ .. \\ .. \\ w_{n-M+1}(n) \end{pmatrix}_{MX1}$$

And estimating W(n).
The human voice can be characterized as

$$e(n) = d(n) - W^T(n)X(n)$$

Cont..

The goal is to find W(n) that will allow $W^T(n)X(n)$
to mimic the instrument sound in d(n). This is possible if e(n) is minimum. This problem can be expressed as

$$\min_{W(n)} e^2(n)$$

And last

Thank You