LMS

In this LMS filter ,an adaptive finite impulse response (FIR) filter is implemented that converges an input signal to the desired signal using LMS algorithm. The filter adapts its weights until the error between the primary input signal and the desired signal is minimal.

LMS — solves the Wiener-Hopf equation and finds the filter coefficients for an adaptive filter.

$$f(u(n), e(n), \mu) = \mu e(n) u^*(n)$$

Variable	**Description**
n	The current time index
u(n)	The vector of buffered input samples at step n
u*(n)	The complex conjucate of the vector of buffered input samples at step n
$e(n)$	The estimated error at step n
μ	The adaptative step size

Input

Input - Input signal

Input is specified as a scalar or column vector.
When input is fixed

Desired - Desired signal

Desired signal is specified as a scalar or column vector.The desired signal must have same dimensions , data type and complexities as the Input signal

Step-size - step-size

Enter the step size as μ .For convergence of the normalised LMS equations , 0<μ<2.Input type must match the type of the Input port.

Adapt - Update filter weights

When the input to this port is greater than zero, the block continuously updates the filter weights. When the input to this port is less than or equal to zero,the filter weights remain at their current values.

Reset

Sinal to reset the value of the filter weights to their initial values,specified as a scalr. The reset signal rate must be the same rate as the data signal input.

Output

Ouput - Estimate of desired signal

Estimate of the desired signal ,retured as a scalar or a column vector.It is the same size and complexities as the input signal.

Error - Error between output and desired signals

Error between the output and desired signals,returned as a scalar or a column vector. This error is the result of subtracting the output signal from the desired signal.

Wts - Filter weights

Filter weights ,retured as scalar or a column vector .For each iteration, the block outputs the current updated filter weights from this port.

More about

LMS Filter Algorithms

When we select LMS for the Algorithm parameeter ,the block calculates the filter weights by using the least mean square (LMS) algorithm. The algorithm is defined by these equations.

$$y(n) = w^T(n-1)u(n)$$
$$e(n) = d(n) - y(n)$$
$$w(n) = \alpha w(n-1) + f(u(n), e(n), \mu)$$

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$$f(u(n), e(n), \mu) = \mu e(n)u(n)$$

Weighted Filter

Normal auditory systems can usually hear between 20 and 20,000 Hz. When we measure sound, the measurement instrument takes the incoming auditory signal and analyzes it for these different features. Weighting filters in these instruments then filter out certain frequencies and decibel levels depending on the filter.
In the feild of audio measurement ,some special units are used to indicate a weighted measurement as opposed to a basic measurement of energy level.For sound, the unit is the phon(1 kHz equivalent level).A-weighted decibels are abbreviated dB(A) or dBA. When acoustic measurement are being referred to, then the units used will be dB SPL(sound pressure level) referenced to 20 micropascals = 0 dB SPL.

Finite impulse response

In signal processing , a finite impulse response (FIR) filter is a filter whose impulse response is of finite duration because it settles to zero in finite time.For a casual discrete -time FIR filter of order N, each value of the output sequence is a weighted sum of the most recent input values

$$y[n] = b_0x[n] + b_1x[n-1] + ... + b_Nx[n-N] = \sum_{i=0}^{N} b_i \cdot x[n-i]$$

FIR filters are designed by finding the coefficient and the filter order that meet certain .specifications ,which can be in the time domain (e.g., a matched filter) or the frequency domain (most common).Matched filters perform a cross -correlation between the input signal and a known pulse shape. The FIR convolution is a cross - correlation between the input signal and a time -reversed copy of the impulse response. Therefore , the matched filter's impulse response is "designed" by saampling the known pulse-shape and using samples in reverse order as the coefficients of the filter.

LMS mean square error (MSE) method

To design FIR filter in the MSE sense ,we minimize the mean square error between the filter we obtained and the desired filter.

MSE $= f_s^{-1} \int_{-f_s/2}^{f_s/2} |H(f) - H_d(f)|^2 \, df$

Where f_s is sampling frequency, $H(f)$ is the spectrum of the filter we obtained, and $H_d(f)$ is. the spectrum of the desired filter.

Frequency response

The filter's effect on the sequence x[n] is described in the frequency domain by the convolution :$y[n] = x[n] * h[n] = \mathcal{F}^{-1}\left\{X(\omega) \cdot H(\omega)\right\}$ where operators $\mathcal{F} and \mathcal{F}^{-1}$ respectively denote the discrete-time Fourier transform(DTFT) and its inverse. Therefore ,the complex-valued,multiplicative function $H(\omega)$ is the filter's frequency response.It is defined by a Fourier series:

$$H_{2\pi}(\omega) \triangleq \sum_{n=-\infty}^{\infty} h[n] \cdot \left(e^{i\omega}\right)^{-n} = \sum_{n=0}^{N} b_n \cdot \left(e^{i\omega}\right)^{-n}$$