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1 Transversal Filters

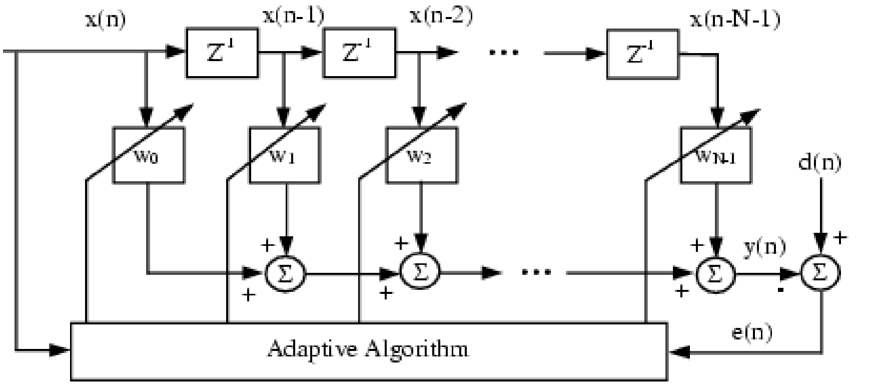

Figure 2.1: Transversal adaptive filter structure.

The most commonly used structure in implementing adaptive filters is the transversal structure shown in Fig. 2.1. The transversal adaptive filter can be split into two main parts, the filter part and the update part. The function of the former is to calculate the filter output $y(n)$, while the function of the latter is to adjust the set of N filter coefficients $w_i, i=0,1,\cdots,N-1$ (tap weights) so that the output $y(n)$ becomes as close as possible to a desired signal $d(n)$. The filter part processes a single input sample $x(n)$ and produces a single output sample $y(n)$ (assuming sample per sample implementation). The filter output is calculated as a linear combination of the input sequence $x(n-i), i=0,1,\cdots,N-1$ composed of delayed samples of $x(n)$,

$$y(n) = \sum_{i=0}^{N-1} w_i(n) \cdot x(n-i).$$

Expressing the set of N filter coefficients at time index n and the sequence of delayed input samples in vector notations such that $\underline{\mathbf{w}}(n) = [w_0(n)\ w_1(n)\ \cdots\ w_{N-1}(n)]^T$ and $\underline{\mathbf{x}}(n) = [x(n)\ x(n-1)\ \cdots\ x(n-N+1)]^T$, where $(\cdot)^T$ is the vector transpose operator, eq (2.1) can be written as

$$y(n) = \underline{\mathbf{w}}(n)^T \cdot \underline{\mathbf{x}}(n) = \underline{\mathbf{x}}(n)^T \cdot \underline{\mathbf{w}}(n).$$

The transversal filter structure is, therefore, a linear temporal filter that processes the temporal samples of its input signal $x(n)$ to produce the temporally and consequently spectrally modified (filtered) output $y(n)$. In fixed transversal filter applications, the set of filter coefficients are chosen at the system design time to achieve the required spectral filtering and remain constant during the filter operation. In adaptive filters applications, however, an adaptive algorithm is used to continuously adjust the filter coefficients so that a certain performance criterion is optimized in some sense. Regardless of the optimization method, it is usually desired to adjust the filter coefficients such that the filter output $y(n)$ resembles a desired signal $d(n)$, or equivalently, the error signal $e(n)$ must be minimized. The details of the optimization process defines the adaptive algorithm and its behavior. The adaptive signal processing toolbox contains several transversal adaptive algorithms such as the Least Mean Squares (see Section 4.9), the Normalized Least Mean Squares (see Section 4.11), the leaky Normalized Least Mean Squares (see Section 4.7), the Variable Step Size Least Mean Squares (see Sections 4.10 and 4.20), and the Recursive Least Squares (see Section 4.16). When the number of filter coefficients N, is large, it is much more efficient to perform filtering and coefficient update in the frequency domain. This requires collecting a block of samples of the input signal before the fast Fourier transform (FFT) can be calculated. For this reason, a frequency domain transversal filter is usually a block processing filter that accepts a block of B input samples and produces a block of B output samples. Several implementations of block frequency domain adaptive filters are included in the adaptive signal processing toolbox, such as the Block Frequency Domain Adaptive Filter (see Section 4.2), the Partitioned Block Frequency Domain Adaptive Filter (see Section 4.12), and the Reduced Complexity Partitioned Block Frequency Domain Adaptive Filter (see Section 4.13).

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