Digital Signal Processing

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Presented By

MD Aminul Haque
ID 1838520119
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Application of Digital Signal Processing

Digital Signal Processing is used in multiple areas such as:

- **Audio signal Processing:**

  Audio signals are electronic representations of sound waves. Analog processors operate directly on the electrical signal, while digital processors operate mathematically on its digital representation. The digital processor decodes data, DAC converter converts the signal to analog for human hearing.

- **Audio data compression:**

  An audio code is an algorithm or a device that compresses the original digital audio signal to a lower bit rate and decompresses the coded data (bit stream) to produce a perceptually similar version of the original audio signal.

- ### Digital image processing:

DSP can improve the quality of images taken under extremely unfavorable conditions in several ways: brightness and contrast adjustment, edge detection, noise reduction, focus adjustment, motion blur reduction, etc.

- ### Speech Recognition:

A basic speech recognition algorithm can be split into three states: listening, processing, and matching. In the listening phase, the DSP analyses the present audio signal to determine if speech is present. When speech is detected, the DSP starts to process the information to describe the speech in a compact way.

- ### Digital Transmission in Computer:

Data or information can be stored in two ways, analog and digital. For a computer to use the data, it must be in discrete digital form.signals can also be in analog and digital form. To transmit data digitally, it needs to be first converted to digital form.

- ## Weather forecasting:

In a typical weather-forecasting system, recently collected data are fed into a computer model in a process called assimilation. This ensures that the computer model holds the current weather conditions as correctly as possible before using it to predict how the weather may change over the next few days.

- ## Computer Graphics:

The DSP is used to speed-up the geometric processing subsystem, Application control selects, position objects and light sources for each scene.

Other appliances where it is used are Mp3,Radar,Sonar, CT scans, MRI etc.

Adaptive Filters

2016:8(b) : Define Adaptive filter.

The adaptive filter is computational device that attempts to model the relationship between two signals (input and output) in real time and in an iterative manner.

Adaptive filters are digital filters whose coefficients change with an objective to make the filter converge to an optimal state. The optimization standard a cost function, which is most commonly the mean square of the error signal between the output of the adaptive filter and the desired signal.

2019 8(b) : Define Adaptive filter. Draw the block diagram and explain the function of Adaptive filter.

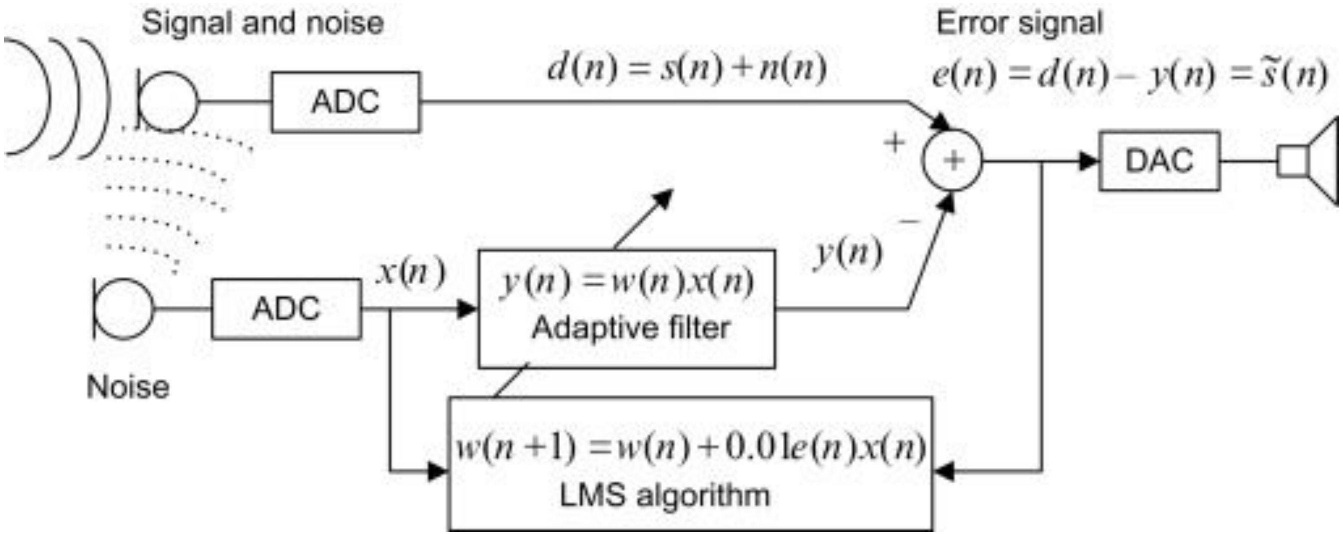

Figure: Block diagram of Adaptive filter.

- The adaptive filter has two input signals, an desired signal s(n), which is accompanied by a interruption n(n), and a reference signal, x(n), similar in characteristics to the interruption.

- The FIR filter operates on the reference signal, and it adapts its impulse response such that its output resembles the disturbance accompanying the desired signal.

- By subtracting the output of the filter from the desired signal plus disturbance, a good estimation of the desired signal is obtained (e(n) = ŝ(n))

- The most important aspect of the adaptive filter is that it removes the disturbance from the desired signal without affecting the desired signal itself. This is because the FIR filter of the adaptive filter operates on the reference signal only.

- When a desired signal is accompanied by a sinusoidal disturbance, the disturbance can be removed by a simple band-stop filter. However, the filter removes the stop band frequencies from the desired signal as well.

- With adaptive filter, the sinusoidal disturbance is removed with no effect on the desired signal

More About Adaptive Filters

❖ **Finite impulse response:**

A finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time.

A finite impulse response (FIR) filter structure can be used to implement almost any sort of frequency response digitally. An FIR filter is usually implemented by using a series of delays, multipliers, and adders to create the filter's output.

❖ **Non linear Adaptive filter:**

The nonlinear filters is used to overcome limitation of linear models. There are some commonly used approaches: Volterra LMS, Kernel adaptive filter, Spline Adaptive Filter and Urysohn Adaptive Filter.

Applications of Adaptive Filters

- **System Identifications:**

  Channel identification, Echo cancellation.

- **Inverse system identifications:**

  Digital communication equalisation.

- **Noise cancellations:**

  Active Noise cancellations, interface cancellations for CDMA.

- **Periodic signal extraction:**

  Speech coders, CDMA interfaces suppression.

- **Smart antenna Systems.**
- **Adaptive feedback cancellations.**

Code Division Multiple Access (CDMA)

Frequency Analysis a signal using DFT

To compute the spectrum of either a continuous-time or discrete-time signal, the values of the signal for all time are required. The signal to be analyzed is an analog signal, we would first pass it through an anti analysing filter and then sample it at a rate,

Fs, ≥ 2B, where **B** is the band width of the filtered signal. Thus the highest frequency that is contained in the sample signal is **Fs/2**

For practical purposes, we limit the duration of the signal to the time interval **T = LT,** where **L is the number of samples** and **T is the sample interval.** As we will observe in the following discussion, the finite observation interval for the signal places a limit on the frequency resolution, that is, it limits ability to distinguish two frequency components that are separated by less than **$1/T_0$ = 1/LT** in frequency.

Let **{x(n)}** denote the sequence to be analyzed. Limiting the duration of the sequence to **L** samples, in the interval **$0 \le n \le L - 1$,** is equivalent to multiplying **(x(n))** by a rectangular **window ω(n) of length L**. That is,

$$\hat{x}(n) = x(n)\, \omega(n)$$

$$\text{Where } \omega(n) = \begin{cases} 1, & 0 \le n \le L - 1 \\ 0, & \text{otherwise} \end{cases}$$

Now suppose that the sequence **x (n)** consists of a single sinusoid, that is,

$$x(n) = \cos \omega_0 (n)$$

then the Fourier transform of the finite-duration sequence **x(n)** can be expressed as ,

$$\hat{x}(\omega) = 1/2\, [W(\omega - \omega_0) + W(\omega + \omega_0)]$$

where **W (ω)** is the Fourier transform of the window sequence, which is (for the rectangular window)

$$W(\omega) = \frac{\sin\left(\frac{\omega L}{2}\right)}{\sin\left(\frac{\omega}{2}\right)} e^{-j\,\omega\,(L-1)/2}$$

To compute **X^** (ω) we use the DFT. By padding the sequence **x^(n)** with N-L zeros, we compute the DFT of the truncated points) sequence {x^(n)}. So The magnitude spectrum

$$|x\hat{}(k)| = |X\hat{}(\omega_k)| \text{ for } \omega_k = 2\pi k/N,\ k = 0,1,2....N$$

So We note that the windowed spectrum **(ω)** not localized a single frequency, but instead out over the whole frequency range. Thus power of the original signal sequence **{x(n)}** was concentrated at a single frequency has been spread the window into entire frequency range. that the power "leaked the entire frequency range. Consequently characteristic of windowing the signal is called leakage.

So, for general signal frequence **{x(n)}** the frequency domain relationship between the window sequence **x^(n)** and the original sequence **x(n)** is given by the convolution formula.

$$X^\wedge(\omega) = \frac{1}{2\Pi}\int_{-\pi}^{\pi} x(\theta)\, W(\omega - \theta)d\theta$$

The DFT of the window sequence **x^(n)** is the sampled version of the spectrum **x(ω)**
Thus we have,

$$x^\wedge(k) \equiv x^\wedge(\omega)_{\omega = 2\pi/N}$$

$$= \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\theta) w\left(\frac{2\pi k}{N} - \theta \right) d\theta \quad k=0,1,2...N-1$$

Just as in the case of the sinusoidal sequence, if the spectrum of the window is relatively narrow in width compared to the spectrum **X (ω)** of the signal, the window function has only a small (smoothing) effect on the spectrum **X (ω).** On the other hand, if the window function has a wide spectrum compared to the width of **X (ω),** as would be the case when the number of samples **L** is small, the window spectrum masks the signal spectrum and, consequently, the DFT of the data reflects the spectral characteristics of the window function. This situation should be avoided.

Thank You