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SRI KRISHNA COLLEGE OF TECHNOLOGY

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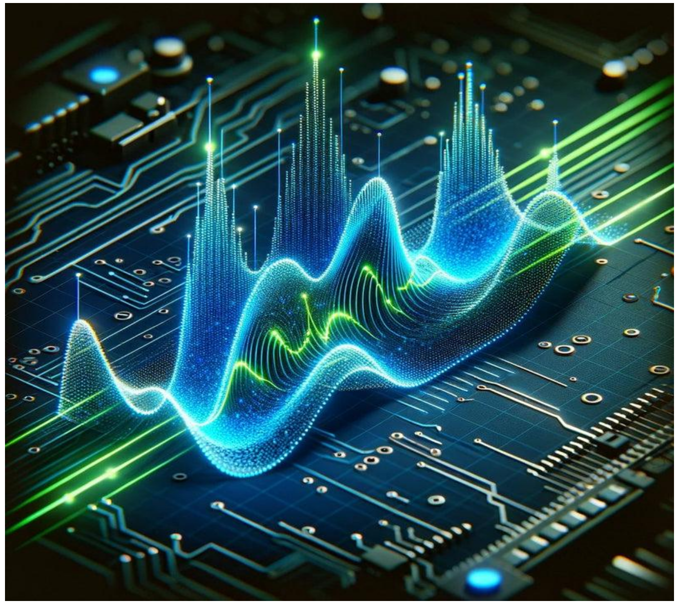
22EE603 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING



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22EE603 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING - CO’s

CO1	Illustrate the basics of discrete time signals and systems. [U]
CO2	Interpret the concepts of Discrete and Fast Fourier transform [U]
CO3	Comprehend the architecture of advanced processors. [U]
CO4	Apply the concept of transformation techniques in Discrete Time systems. [AP]
CO5	Design different types of filters using various filter design techniques. [AP]

22EE603 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING - Modules

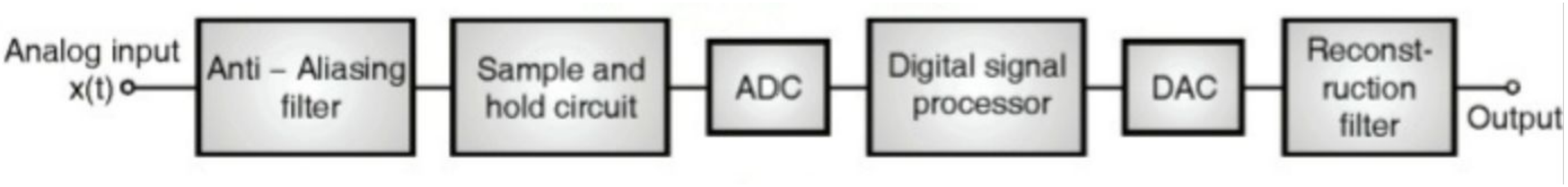
1	Signals and Systems
2	Discrete Fourier Transform and Fast Fourier Transform
3	FIR Filters, IIR Filters and Digital Signal Processors

MODULE – I : Signals and Systems

1.1	Generation and Representation of Discrete Time signals
1.2	Classification of signals
1.3	Classification of systems
1.4	Sampling of continuous time signals and aliasing effect

1.4 Sampling of continuous time signals and aliasing effect

Block Diagram of DSP:



1. **Input signal :** It is the signal generated from some transducer or from some communication system. It may be biomedical signal like ECG or EEG. Generally input signal is analog in nature. It is denoted by x(t).

2. **Anti-aliasing filter**

Anti aliasing filter is basically a low pass filter. It is used for the following purposes :

(a) It removes the high frequency noise contain in input signal.

(b) As the name indicates; it avoids aliasing effect. That means it is used to band limit the signal.

1.4 Sampling of continuous time signals and aliasing effect

3. Sample and hold circuit : As the name indicates; this block takes the samples of input signal. It keeps the voltage level of input signal relatively constant which is the requirement of ADC. Sometimes amplifiers are used to bring the voltage level of input signal upto the required voltage level of ADC.

4. Analog to digital converter (ADC) : As the name indicates; this block is used to convert analog signal into digital form. This is required because digital signal processor accepts the signal which is digital in nature.

5. Digital signal processor : It processes input signal digitally. In a simple languages processing of input signal making modifying the signal as per requirement. For this purpose DSP processors like ADSP 2100 or TMS 320 can be used.

1.4 Sampling of continuous time signals and aliasing effect

6. Digital to analog converter (DAC) : The output of digital signal processor is digital in nature. But the required final output is analog in nature. So to convert digital signal into analog signal DAC is used.

7. Reconstruction filter : Output signal of DAC is analog, that means it is a continuous signal. But it may contain high frequency components. Such high frequency components are unwanted. To remove these components; reconstruction filter is used.

1.4 Sampling of continuous time signals and aliasing effect

Sampling Techniques:

Analog to Digital Conversion:

- ❖ Generally signals are analog in nature (eg: speech, weather signals).
- ❖ To process the analog signal by digital means, it is essential to convert them to discrete-time signal , and then convert them to a sequence of numbers.
- ❖ The process of converting an analog to digital signal is 'Analog-to-Digital Conversion'.
- ❖ The ADC involves three steps which are:

  1)Sampling

  2)Quantization

  3)coding

1.4 Sampling of continuous time signals and aliasing effect

TYPES OF SIGNALS

❖ Analog signals: continuous in time and amplitude

Example: voltage, current, temperature,…

❖ Digital signals: discrete both in time and amplitude

Example: attendance of this class, digitizes analog signals,…

❖ Discrete-time signal: discrete in time, continuous in amplitude

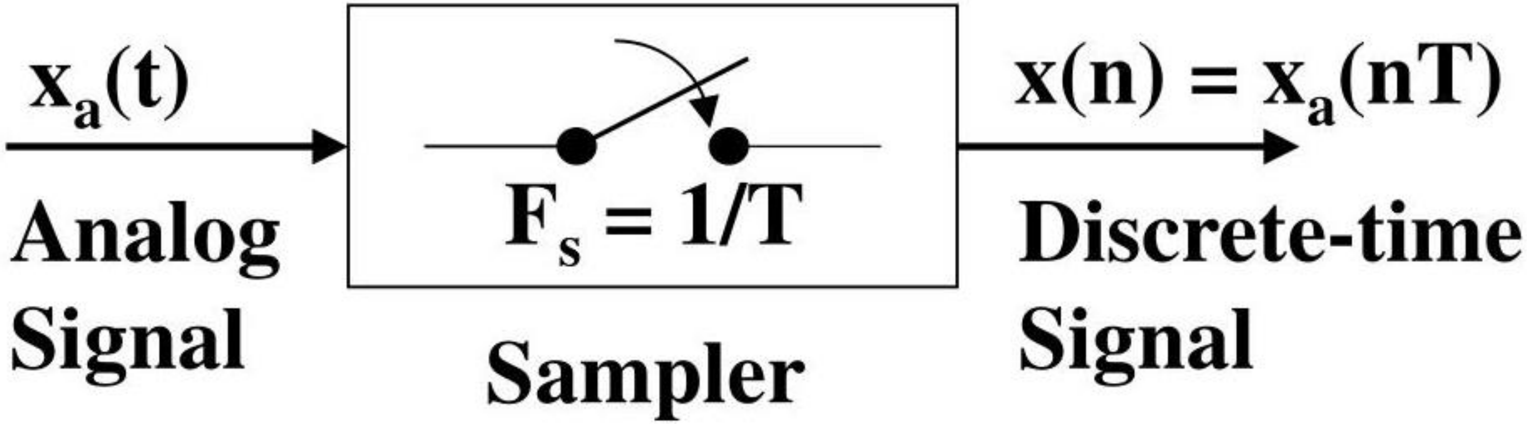
Example: hourly change of temperature in Austin

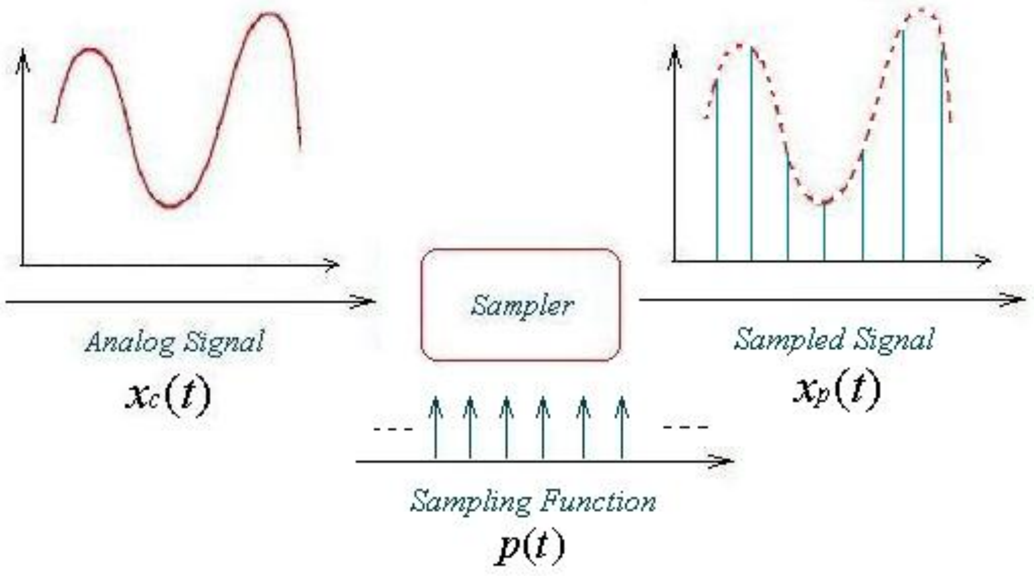
1.4 Sampling of continuous time signals and aliasing effect

Sampling:

- Sampling is the process of converting a continuous-time signal is converted into discrete -time signals by taking samples of continuous-time signal at discrete time intervals.
- The time interval between successive samples is T seconds and sampling frequency is given by Fs=1/T Hz.
- Here relation between analog signal and sampled discrete time and frequency domain is shown in figure.

1.4 Sampling of continuous time signals and aliasing effect





1.4 Sampling of continuous time signals and aliasing effect

Aliasing effect:

- ❖ When we produce the sequence x(n) by sampling $x_a(t)$, we want to ensure that all the information in the original signal is retained in the samples.
- ❖ There will be no information loss if we can exactly recover the continuous-time signal from the samples.

If we define sampling frequency F=1/T we have

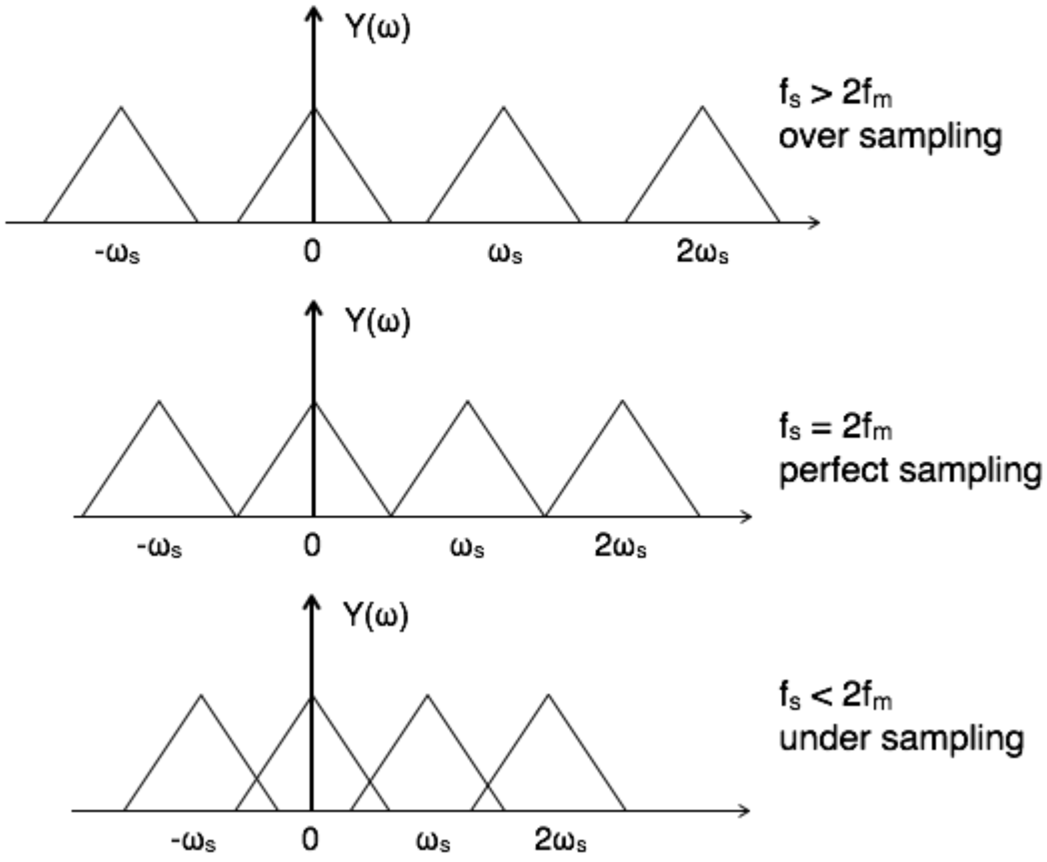
$$\Omega_m \leq \pi \mathrm{F} \leq \frac{\Omega_m}{2}$$

$$2\pi f_m \leq \pi F$$

$$F \geq 2f_m$$

From the equation we find that to avoid aliasing the sampling frequency must be greater than twice the highest frequency present in the signal.

1.4 Sampling of continuous time signals and aliasing effect



1.4 Sampling of continuous time signals and aliasing effect

Sampling Theorem:

A continuous time signal can be represented in its samples and can be recovered back when sampling frequency f_s is greater than or equal to the twice the highest frequency component of message signal.

$$f_s \geq 2f_m$$

Nyquist Rate:

The minimum sampling rate is allowed by the sampling theorem is called Nyquist rate.

1.4 Sampling of continuous time signals and aliasing effect

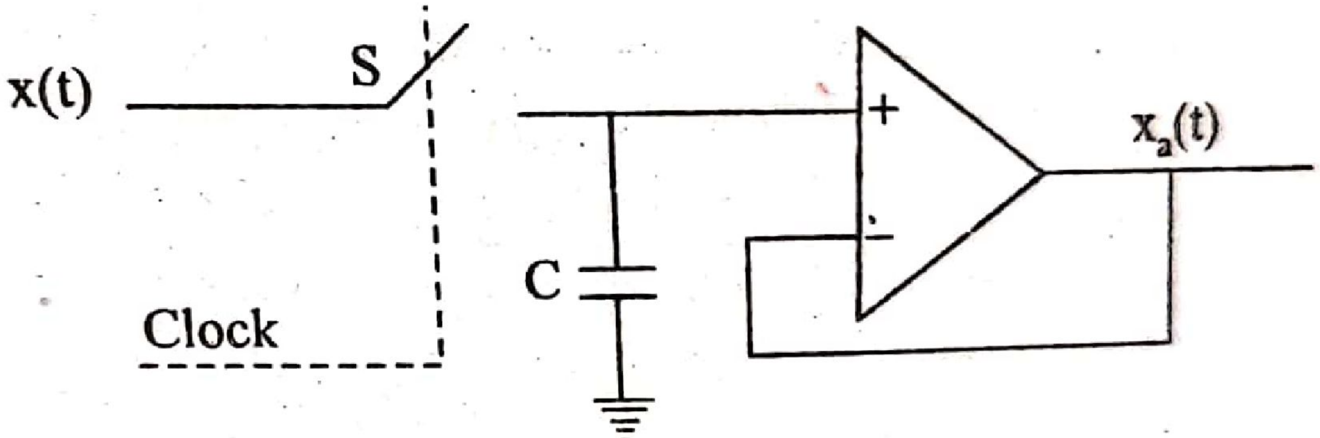
Anti- Aliasing Filter:

- In Practice, Communication signals have frequency spectra consisting of low frequency components as well as high- frequency noise components.
- If we select sampling frequency F, all signals with frequency higher than $\frac{\Omega}{2}$ appear as signals of frequencies between 0 and $\frac{\Omega_s}{2}$ due to aliasing effect.
- To avoid aliasing we can choose very high sampling frequency.
- But sampling at very high frequencies introduces numerical errors.
- Therefore, to avoid aliasing errors caused by the undesired high frequency signals, analog low pass filter, called an anti-aliasing filter is used prior to sampler to filter high frequency components before the signal is sampled.

1.4 Sampling of continuous time signals and aliasing effect

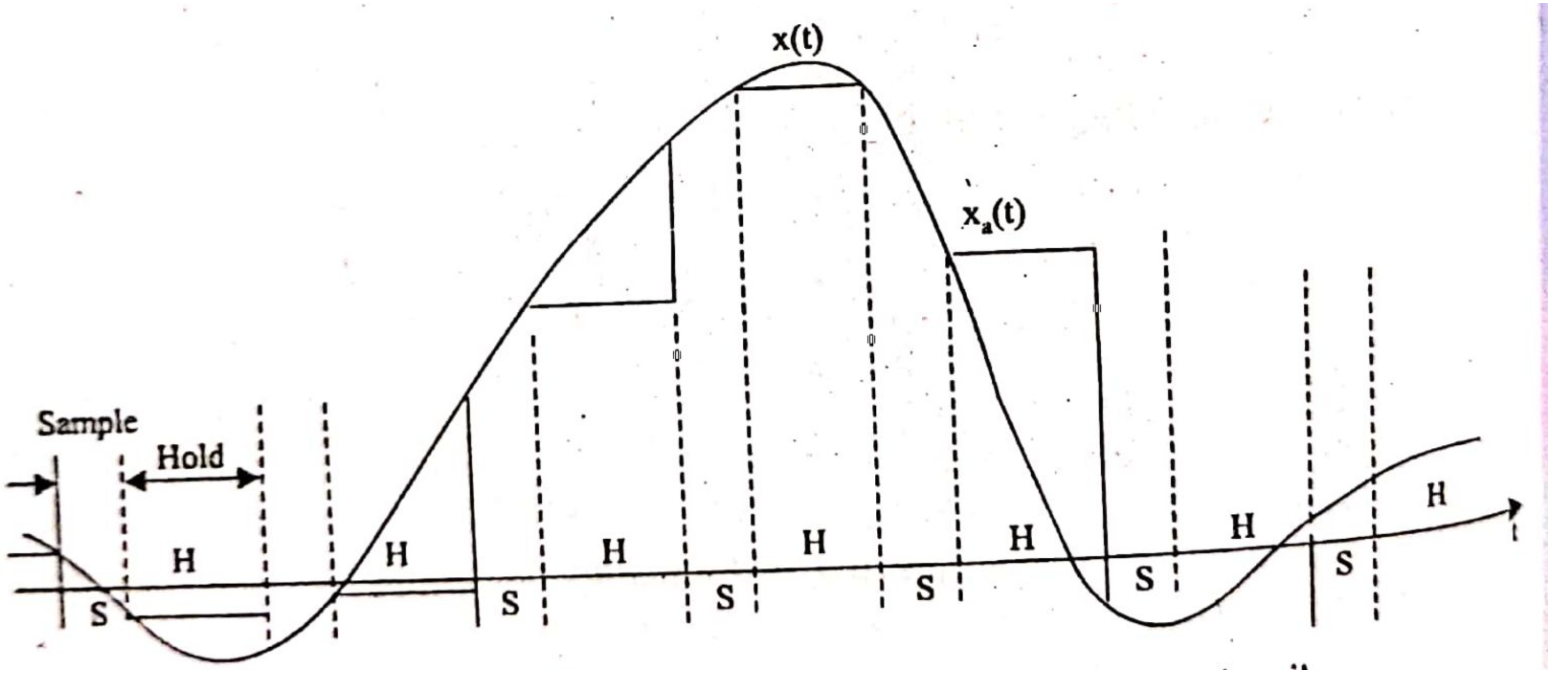
Sample and Hold Circuit:

- ❖ The output of the anti-aliasing filter is fed to a sample-and-hold (S/H) circuit.
- ❖ It samples the analog input signal at uniform intervals and holds the sampled value constant as long as the A/D converter takes time for accurate conversion.
- ❖ The use of sample-and-hold circuit allows the ADC to operate slowly.



Sample and Hold Circuit

1.4 Sampling of continuous time signals and aliasing effect



Input and Output waveforms of S/H circuit

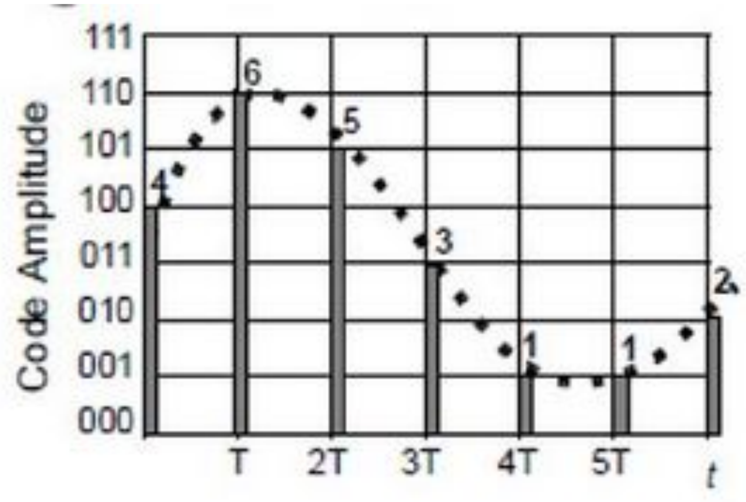
1.4 Sampling of continuous time signals and aliasing effect

- During sample mode the switch S is closed allowing the capacitor C to charge to input voltage.
- During the hold period the switch remains open, the charge on the capacitor holds the voltage across it.
- A digital clock controls the switching operation.
- The voltage follower acts as a buffer between the capacitor and the input stage of the A/D converter.

Quantization:

- The Process of converting a discrete time continuous amplitude signal x(n) into a discrete time discrete amplitude signal $x_q(n)$ is known as quantization.
- This is done by rounding off each sample in x(n) to the nearest quantization level.

1.4 Sampling of continuous time signals and aliasing effect



Quantization Error:

The process of converting x(n) to finite number of digits introduces an error known as Quantization noise.

$$e(n)= x_q(n)-x(n)$$

1.4 Sampling of continuous time signals and aliasing effect