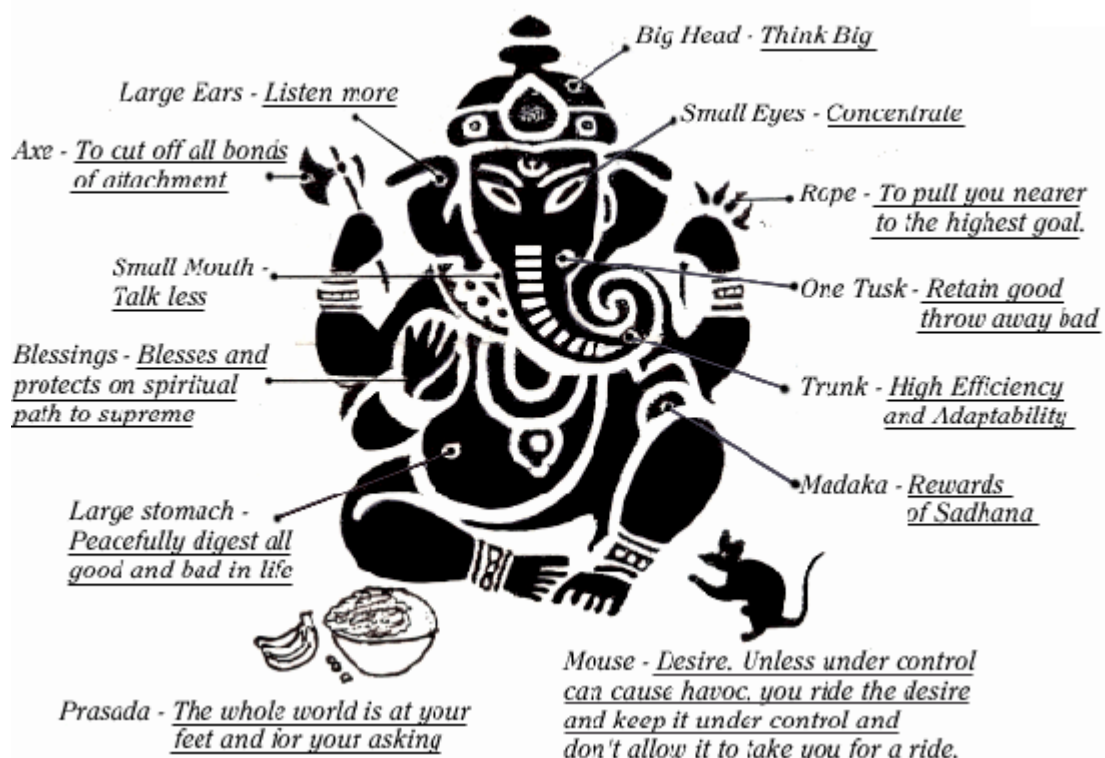
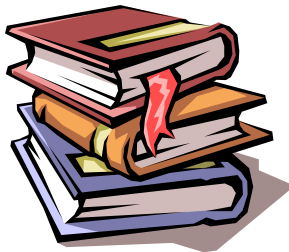




1. DT Signal-----
2. DFT + FFT-----
3. Analysis of DT System-----
4. Digital IIR Filter-----
5. Digital FIR Filter-----
6. Multirate Signal Processing-----
7. Telecommunication Applications--

Inspiration from LORD GANESH





REFERENCE BOOKS

SUBJECT : D. S. P.



1. **Proakis Manolakis,**
(Digital Signal Processing : Principles, Algorithms and Applications)
3rd Edition Prentice Hall of India (1992)

An excellent text with the comprehensive discussion on a wide range of topics. It is a very well written book with rich set of exercise problems. This is an USEFUL text.

2. **S. Salivahanan, A. Vallavaraj, C. Gnanapriya,**
(Digital Signal Processing)
Tata Mcgraw Hill Publication First edition (2001).

An excellent text with systematic, simple and easy to understand explanation for every topic on continuous Time and Discrete Time System. This is an USEFUL text.

3. **Oppenheim and Saffer,**
(Discrete Time Signal Processing)
Prentice Hall of India (1992)

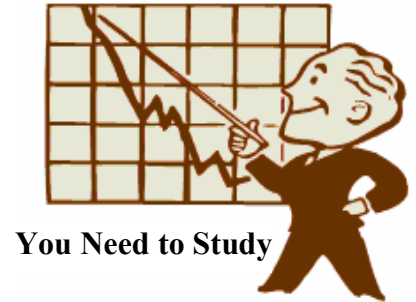
This is a good reference text for Z-transform,DT system analysis,DFT and FFT. It has a rich set of exercise problems.

4. **Ashok Ambardkar,**
(Analog and Digital Signal Processing)
Brooks/ Cole Publishing company (1999)

This is a good reference book for continuous time and discrete time signals, its application and for analog and discrete topics in DSP. A very well written book. This is an USEFUL text.



CHAPTER-1



DT Signal Basic Concepts

Sr No	TOPIC	PAGE
1	Basic Concept ✓	
2	Analog and Digital filters ⌚	
3	DSP System ⌚	
4	Difference between Microprocessor and DSP Processor ⌚	
5	Discrete Time Signal ✓	
6	Classification of DT Signals ✓	
7	Sampling 🔒	
8	Reconstruction 🔒	
9	Convolution Algorithm 🔒	

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1. BASIC CONCEPTS

1.1. Signal Processing

Signals commonly need to be processed in a variety of ways. For example, the output signal from a transducer may well be contaminated with unwanted electrical "noise". The electrodes attached to a patient's chest when an ECG is taken measure tiny electrical voltage changes due to the activity of the heart and other muscles. The signal is often strongly affected by "mains pickup" due to electrical interference from the mains supply. Processing the signal using a filter circuit can remove or at least reduce the unwanted part of the signal. Increasingly nowadays the filtering of signals to improve signal quality or to extract important information is done by DSP techniques rather than by analog electronics.

1.2. What is DSP?

DSP, or Digital Signal Processing, as the term suggests, is the processing of signals by digital means. A *signal* in this context can mean a number of different things. Historically the origins of signal processing are in electrical engineering, and a signal here means an electrical signal carried by a wire or telephone line, or perhaps by a radio wave. More generally, however, a signal is a stream of information representing anything from stock prices to data from a remote-sensing satellite.



1.3. Analog and Digital Signals



In many cases, the signal is initially in the form of an analog electrical voltage or current, produced for example by a microphone or some other type of transducer. In some situations the data is already in digital form - such as the output from the readout system of a CD (compact disc) player. An analog signal must be converted into digital (i.e. numerical) form before DSP techniques can be applied. An analog electrical voltage signal, for example, can be digitized using an integrated electronic circuit (IC) device called an analog-to-digital converter or ADC. This generates a digital output in the form of a binary number whose value represents the electrical voltage input to the device.



2 Analog and Digital filters



In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range.

The following block diagram illustrates the basic idea.



There are two main kinds of filter, *analog* and *digital*. They are quite different in their physical makeup and in how they work.

An analog filter uses analog electronic circuits made up from components such as resistors, capacitors and op amps to produce the required filtering effect. Such filter circuits are widely used in such applications as noise reduction, video signal enhancement, graphic equalizers in hi-fi systems, and many other areas.

There are well-established standard techniques for designing an analog filter circuit for a given requirement. At all stages, the signal being filtered is an electrical voltage or current which is the direct analogue of the physical quantity (e.g. a sound or video signal or transducer output) involved.

A digital filter uses a digital processor to perform numerical calculations on sampled values of the signal. The processor may be a general-purpose computer such as a PC, or a specialized DSP chip.

The analog input signal must first be sampled and digitized using an ADC (analog to digital converter). The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output through a DAC (digital to analog converter) to convert the signal back to analog form.

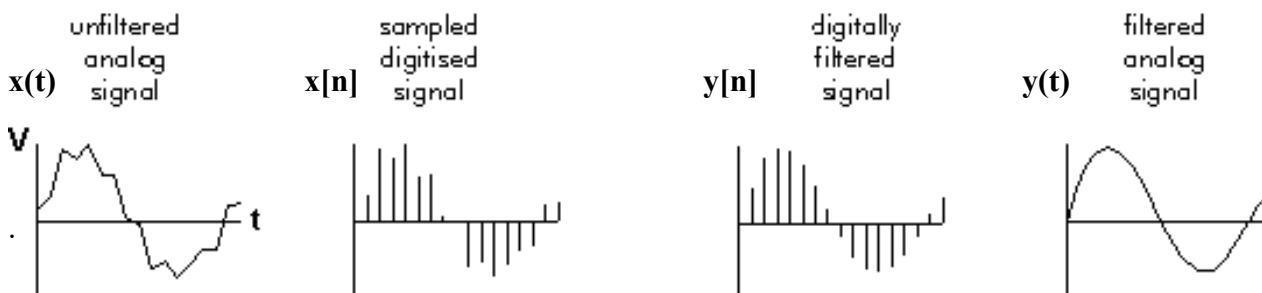
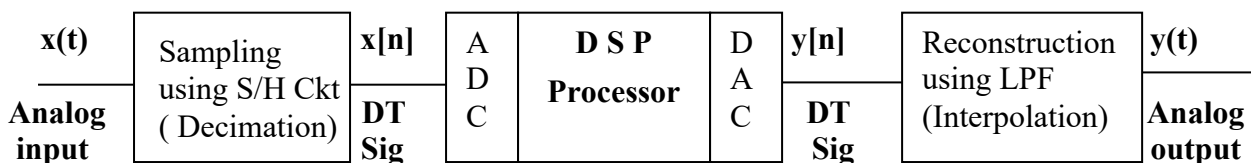
Note that in a digital filter, the signal is represented by a sequence of numbers, rather than a voltage or current.





1.3. DSP SYSTEM

The following diagram shows the basic setup of DSP system.



Digital filter is a Discrete Time System which produces a discrete time output sequence $y[n]$ for the discrete time input sequence $x[n]$. Digital filter is nothing but mathematical algorithm implemented in hardware or software.

Real time digital filter consist of processing of real time signal using digital device called digital processor.

As shown in figure, analog input signal is first sampled and DT signal thus obtained is converted into digital signal using ADC. Digital processor, perform the operation depending upon the algorithm programmed in digital processor.

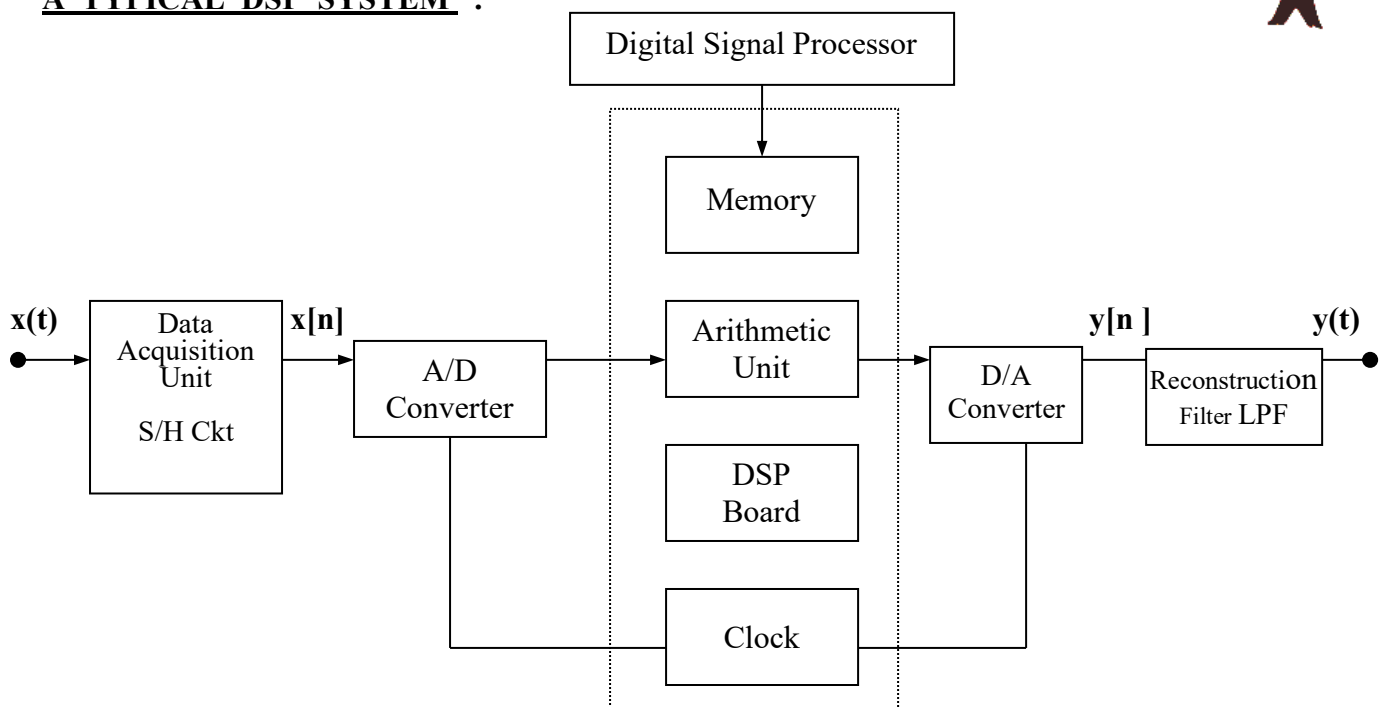
The output of the digital processor is converted into analog signal using DAC. Reconstruction filter is used to obtain the corresponding analog signal from the output DT signal.

Example of DSP System is Digital Telephone System.





A TYPICAL DSP SYSTEM :



Working of DSP System:

A typical DSP system is illustrated in figure. It consists of a Data Acquisition Stage that is receiving a continuous signal $x(t)$. The Data Acquisition unit is connected to an analog to digital converter under the control of synchronization pulses from a clock generator to maintain the required sampling rate. Upon application of a synchronizing pulse, the analog, to digital converter delivers a signal representing the instantaneous value of $x(t)$ as a binary number, corresponding to a pulse train (in sequential code) or a set of voltage levels (in parallel code).

The signal is delivered to a digital signal processor consisting of a DSP board with DSP chips and associated circuitry, an arithmetic unit and a memory, for performing the necessary operation. For efficient Data Processing, the DSP hardware should have high resolution in time or frequency domain, series/parallel input and output and interrupt facility. The arithmetic unit performs various operations on the input numbers such as addition, multiplication and time shift through a certain number of sampling intervals.





A Digital system consists of multipliers and delay elements connected to known signal sources. A multiplier is a digital element having an output proportional to the input, whereas a delay element is a digital element in which the output equals the input delay by 1 unit of time. Further, it may be recollected that whereas a multiplier does not have a memory, a delay element is a memory device because its output at a given time is dependent on the last input.

The memory unit of a computer is used for storing sample value of the input and output signals for use in signal processing. Regarding the overall function of the digital signal processor, it handles the incoming binary number sequence as desired by the processing algorithm decided and to yield the obtained binary number sequence representing the output. In case the user wants data in analog form, a digital to analog converter is placed at the output. Otherwise, the data may be fed straight for further digital processing. Advances in microelectronics have given birth to fabulous VLSI based memory chips and microprocessors chips which are being increasingly utilized in microcomputer digital signal processing.



1.4 Difference between Microprocessor and DSP Processor:



Sr. No.	Parameter	DSP Processor	Micro-processor
1	Instruction Cycle	Instructions are executed in single cycle of the clock	Multiple clock cycles are required for execution of one instruction
2	Instruction execution	Parallel execution is possible	Execution of instruction
3	Operand Fetch and memory	Multiple operands are fetched simultaneously	Operands are fetched sequentially
4	Memories	Separate data and program memories	Normally no such memories are present
5	On chip/off chip memories	Program and data memories are present on chip and extendable off chip	Normally on chip cache memory present. Main memory is off chip
6	Program flow control	Program sequencer and instruction register takes care of program flow	Program counter takes care of flow of execution
7	Queuing/pipelining	Queuing is implicate through instruction register and instruction cache	Queuing is performed explicate by one queue register for pipelining of instructions
8	Address generation	Address are generated combine by DAG's and program sequencer	Program counter is incremented sequentially to generate address
9	Address/ data bus multiplexing	Address and data bus are not multiplexed. They are separate on chip as well as off chip	Address / data bus may be separate on chip but are multiplexed off chip
10	Computational units	Three separate computational unit: ALU, MAC and Shifter	Only one main unit: ALU
11	On chip address and data buses	Separate address and data buses for program and data memory i.e. DMA, DMD, PMD, PMA and R bus	Address and data bus are the two bus on the chip
12	Addressing modes	Direct and indirect addressing modes	Direct, indirect, register, register indirect, immediate etc
13	Suitable for	Array processing operations	General purpose processing



1.5 Discrete Time Signal



DT signal is obtained by sampling CT signal at regular intervals of time.

$$x(t)|_{t=nT_s} = x[nT_s] = x[n]$$

➤ Linear Shifting of NON-Periodic DT Signals

$$1) \quad x[n] = \left\{ \begin{array}{cccc} 1 & 2 & 3 & 4 \end{array} \right\}$$

$$2) \quad x[n-1] = \left\{ \begin{array}{cccc} 0 & 1 & 2 & 3 & 4 \end{array} \right\}$$

$$3) \quad x[n+1] = \left\{ \begin{array}{cccc} 1 & 2 & 3 & 4 \end{array} \right\}$$

$$4) \quad x[-n] = \left\{ \begin{array}{cccc} 4 & 3 & 2 & 1 \end{array} \right\}$$

$$5) \quad x[-n+1] = \left\{ \begin{array}{cccc} & 4 & 3 & 2 & 1 \end{array} \right\}$$

$$6) \quad x[-n-1] = \left\{ \begin{array}{cccc} 4 & 3 & 2 & 1 & 0 \end{array} \right\}$$

➤ Circular Shifting of Periodic DT Signals

$$1) \quad x[n] = \{ 1, 2, 3, 4 \}$$

$$2) \quad x[n-1] = \{ 4, 1, 2, 3 \}$$

$$3) \quad x[n+1] = \{ 2, 3, 4, 1 \}$$

$$4) \quad x[-n] = \{ 1, 4, 3, 2 \}$$

$$5) \quad x[-n+1] = \{ 2, 1, 4, 3 \}$$

$$6) \quad x[-n-1] = \{ 4, 3, 2, 1 \}$$

- NOTE :
- (1) When digital frequencies are separated by multiples of $\pm 2\pi$
Then Discrete Time Signals are exactly Same..(i.e. their Sample values are identical)
 - (2) Range of Digital frequency ω is $(-\pi, \pi]$ for mathematical analysis.
 - (3) Range of Digital frequency f is $\left[\frac{-1}{2}, \frac{1}{2} \right]$ for mathematical analysis.
-





1.6 Classification of Discrete Time signals

1. Finite length / Infinite length

If the number of samples are finite then signal is finite. Similarly if number of samples are infinite then signal is infinite.

Examples :

(i) Finite Length : $x[n] = \left\{ \underset{\uparrow}{1} \quad 2 \quad 3 \quad 4 \right\}$

(ii) Infinite Length Signal : $x[n] = u[n]$

2. Causal / Anti-causal / Both-sided

If $x[n] = 0$ for all $n < 0$ then $x[n]$ is causal signal.

If $x[n] = 0$ for all $n \geq 0$ then $x[n]$ is anticausal signal.

If $x[n]$ is neither causal nor anticausal then $x[n]$ is bothsided signal.

Examples :

(i) Causal signal : $x[n] = u[n]$

(ii) Anti-causal signal : $x[n] = u[-n-1]$

(iii) Both sided signal : $x[n] = u[n] + u[-n-1]$

3. Periodic / Non-periodic

If the digital frequency of the signal is rational number then the signal is periodic. Otherwise signal is nonperiodic.

Examples :

(i) Periodic signal : $x[n] = \cos(0.6\pi n)$ where $w = 0.6\pi$ and $f = 0.3 = \frac{3}{10}$

Here f is a rational number, so $x[n]$ is periodic signal with period = 10.

(ii) Non periodic signal : $x[n] = \cos(2n)$ where $w = 2$ and $f = 1/\pi$

Here f is Not rational number, so $x[n]$ is not periodic signal.

4. Energy / Power / Neither Energy nor Power

Energy of signal is defined as, $E = \sum_{n=-\infty}^{\infty} |x[n]|^2$

If Energy of DT signal is finite ($0 < E < \infty$) then $x[n]$ is an energy signal.

If Energy is infinite then go for average power.





The average power of the $x[n]$ is given as $P = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_{n=0}^{N-1} |x[n]|^2$

If P is finite and nonzero then $x[n]$ is a power signal.

Examples :

(i) Energy signal :

$$x[n] = \left(\frac{1}{2}\right)^n u[n] \quad E = 2 \text{ (finite)}$$

$$x[n] = \left\{ \underset{\uparrow}{1} \quad 2 \quad 3 \quad 4 \right\} \quad E = 30 \text{ (finite)}$$

(ii) Power Signal : $x[n] = u[n]$

(iii) Neither Energy NOR Power Signal : $x[n] = e^{2n} u[n]$

5. Even / Odd

If $x[n] = x[-n]$ then $x[n]$ is even signal.

If $x[n] = -x[-n]$ then $x[n]$ is odd signal.

Examples :

(i) Even signal : $x[n] = \left\{ -1 \quad -2 \quad \underset{\uparrow}{3} \quad -2 \quad -1 \right\}$

(ii) Odd Signal : $x[n] = \left\{ 1 \quad 2 \quad \underset{\uparrow}{0} \quad -2 \quad -1 \right\}$

(iii) Neither Even nor Odd signal : $x[n] = \left\{ \underset{\uparrow}{1} \quad 2 \quad 3 \quad 4 \right\}$

6. Causal Symmetric / Causal Antisymmetric

If $x[n] = x[N-1-n]$ then $x[n]$ is causal symmetric.

If $x[n] = -x[N-1-n]$ then $x[n]$ is causal antisymmetric.

Examples :

(i) Causal Symmetric signal : $x[n] = \left\{ 1 \quad 2 \quad 3 \quad 2 \quad 1 \right\}$

(ii) Causal Anti-symmetric Signal : $x[n] = \left\{ 1 \quad 2 \quad 0 \quad -2 \quad -1 \right\}$

NOTE-1 : [1] In Linear Convolution, If input signals are Causal,
Then Resultant output signal is also causal.

[2] $x[n] * h[n] = h[n] * x[n]$

[3] Application of convolution is to find output of a Digital filter for any given input signal.





➤ PROPERTIES OF CONVOLUTION

i) Commutative

$$x[n] * h[n] = h[n] * x[n]$$

ii) Associative

$$(x[n] * h_1[n] * h_2[n]) = (x[n] * h_1[n]) * h_2[n]$$

iii) Distributive

$$x[n] * [h_1[n] + h_2[n]] = x[n] * h_1[n] + x[n] * h_2[n].$$

NOTE-2 :Application of correlation is to find degree of similarity between two signals.

For ex. Two signals are given say $x[n]$ and $p[n]$.

To find out whether they are similar signals are not :-

Calculate coefficient of correlation r using Carls' Pearson correlation formula,

If $r = 1$ Then two signals are exactly same.

If $0.95 \leq r < 1$ Then two signals are closely matched

If $0.9 \leq r < 0.95$ Then two signals are quite similar.

Depending on application Threshold value can be set.

If $r < \text{Threshold}$ Then two signals are NOT similar.

If $r \geq \text{Threshold}$ Then two signals are similar.

NOTE -3 : Any arbitrary signal $x[n]$ can be decomposed into its even part and odd part of the signal components. ie. $x[n] = x_e[n] + x_o[n]$

Where,

$$(i) x_e[n] = \left[\frac{x[n] + x[-n]}{2} \right] \quad (ii) x_o[n] = \left[\frac{x[n] - x[-n]}{2} \right]$$



◆ Frequently Asked Questions ◆



(1) What is DSP?

Ans : Digital Signal Processing is a technique that converts signals from real world sources (usually in analog form) into digital data that can then be analyzed. Analysis is performed in digital form because once a signal has been reduced to numbers, its components can be isolated, analyzed and rearranged more easily than in analog form.

Eventually, when the DSP has finished its work, the digital data can be turned back into an analog signal, with improved quality. For example, a DSP can filter noise from a signal, remove interference, amplify frequencies and suppress others, encrypt information, or analyze a complex waveform into its spectral components. This process must be handled in real-time - which is often very quickly. For instance, stereo equipment handles sound signals of up to 20 kilohertz (20,000 cycles per second), requiring a DSP to perform hundreds of millions of operations per second.

(2) What are the applications of DSP ?

Ans :

- Speech coding & Decoding
- Speech encryption and decryption
- Speech recognition
- Speech Synthesis
- Speaker identification
- Hi-fi audio encoding & decoding
- Modern algorithms
- Noise cancellation
- Audio equalization
- Audio mixing & editing
- Vision
- Image compression & decompression
- Image compositing
- Echo cancellation
- Spectral estimation

(3) What do you mean by real time signal ? Give example.

Ans : Signal is processed with the same speed it is captured. Signal is captured, sampled and processed with the same speed. Signal is not stored before processing. Entire input signal never available before processing. Processed signal can be stored.

For example, in digital telephone system, Signal is captured, Sampled, Processed , Transmitted and Made it available to the end user. Real Time Processing is Online Processing.

(4) How Discrete Time signal is obtained ?

Ans : DT signal is obtained by sampling CT signal at regular intervals of time.

$$x(t)|_{t=nT_s} = x[nT_s] = x[n]$$

In practical application sampling is implemented using S/H circuit.



(5) What do you know about Analog Signal, Digital Signal, CT signal, DT Signal ?

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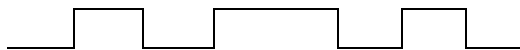
Ans : i] Analog Signal : Signal value can be anything. NO fixed signal level.

Eg $x(t) = \cos(100\pi t)$. ← continuous Sinusoidal signal

$x[n] = \{ 10.5, 4.7, 3.5, 5.7, 3.8 \}$ ← sampled signal

ii] Digital Signal : Only two levels +5v and 0. ie. Logically High and Low.

Eg. Binary data



iii] Continuous Time Signal : Signal is defined for every value of time. Signal value can be anything.

Eg. $x(t) = \cos(100\pi t)$. ← continuous Sinusoidal signal

Bilevel Signal



iv] Discrete Time Signal : Signal is defined for Discrete instant of Time. NOT for every value of time. Signal value can be anything.

Eg. $x[n] = \{ 10.5, 4.7, 3.5, 5.7, 3.8 \}$ ← sampled signal

(6) What is antialiasing filter? Can it be Digital filter ? justify.

Ans : When processing the analog signal using DSP System, it is sampled at some rate depending upon the bandwidth. The rate of sampling is decided by the Nyquist criterion. However, signals that are found in physical systems will never be strictly bandlimited. To eliminate signal content beyond the desired bandwidth, antialiasing filter is used.

The filter cannot be a digital filter. This is because antialias filtering is required to be performed in the analog domain prior to applying the signal to A/D converter where aliasing would take place.

(7) Let $x(t) = 10 \cos(100\pi t) + 20 \cos(120\pi t) - 5 \sin(50\pi t)$. If $x(t)$ is sampled with sampling frequency $F_s = 200$ Hz. What will be Discrete Time Signal $x[n]$ at $n=0$?

Ans : 30

(8) What do you mean by Causal signal, Anti-causal Signal and Both-sided signal ?

Ans : If $x[n] = 0$ for all $n < 0$

Then $x[n]$ is causal signal.

If $x[n] = 0$ for all $n \geq 0$

Then $x[n]$ is anticausal signal.

If $x[n]$ is neither causal nor anticausal

Then $x[n]$ is bothsided signal.

(9) What are the classification of signals ?

Ans : DT signal are classified as

(i) Causal Signal, Anti-causal Signal, Bothsided Signal.

(ii) Even Signal, Odd Signal

(iii) Energy Signal, Power Signal

(iv) Periodic Signal, Non periodic Signal

(v) Symmetric, Anti-symmetric

(vi) Finite Length Signal, Infinite Length Signal



(10) Give one example of Causal, Anticausal and Bothsided signal.



Ans : Examples :

- (i) Causal signal : $x_1[n] = u[n]$ $x_2[n] = (\frac{1}{2})^n u[n]$
(ii) Anti-causal signal : $x_1[n] = u[-n-1]$ $x_2[n] = (\frac{1}{2})^n u[-n-1]$
(iii) Both sided signal : $x_1[n] = u[n] + u[-n-1]$ $x_2[n] = (2)^n u[n] + (3)^n u[-n-1]$

(11) What is an energy signal ? Give example.

Ans : Energy of signal is defined as, $E = \sum_{n=0}^{N-1} |x[n]|^2$

If Energy of DT signal is finite ($0 < E < \infty$) then $x[n]$ is an energy signal.

Ex : $x[n] = (\frac{1}{2})^n u[n]$ $E = 2$ (finite)

$$x[n] = \left\{ \underset{\uparrow}{1} \quad 2 \quad 3 \quad 4 \right\} \quad E = 30 \text{ (finite)}$$

(12) Consider $x_1[n]$ is periodic with period = 4 and $x_2[n]$ is periodic with period = 6 .
Let $x[n] = x_1[n] + x_2[n]$. What will be the period of $x[n]$?

$$ANS = N = LCM\{4, 6\} = 12$$

(13) What is Power Signal ? Give example.

Ans : The Average Power of the $x[n]$ is give ZFVCn as $P = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_{n=0}^{N-1} |x[n]|^2$

If P is finite and nonzero then $x[n]$ is a power signal.

Ex $x[n] = u[n]$

(14) What is symmetric signal ? Give example.

Ans : If $x[n] = x[N-1-n]$

Then $x[n]$ is causal symmetric

Ex. Causal Symmetric signal : $x[n] = \left\{ \underset{\uparrow}{-1} \quad -2 \quad 3 \quad -2 \quad -1 \right\}$

(15) What is Anti-symmetric signal ? Give example.

Ans :

If $x[n] = -x[N-1-n]$

Then $x[n]$ is causal anti-symmetric.

Ex. Causal Anti-Symmetric signal : $x[n] = \left\{ \underset{\uparrow}{-1} \quad -2 \quad 0 \quad 2 \quad 1 \right\}$

(16) What is an Even signal ? Give example.

Ans :

If $x[n] = x[-n]$

Then $x[n]$ is even signal.

Ex. Even signal : $x[n] = \left\{ -1 \quad -2 \quad \underset{\uparrow}{3} \quad -2 \quad -1 \right\}$ Nonperiodic

$x[n] = \left\{ \underset{\uparrow}{-1} \quad -2 \quad 3 \quad 3 \quad -2 \right\}$ Periodic





(17) What is an odd signal ? Give example.

Ans :

$$\text{If } x[n] = -x[-n]$$

Then $x[n]$ is odd signal

$$\text{Ex. Odd signal : } x[n] = \left\{ -1 \quad -2 \quad \underset{\uparrow}{0} \quad 2 \quad 1 \right\} \quad \text{Nonperiodic}$$

$$x_p[n] = \left\{ \underset{\uparrow}{0} \quad -2 \quad 3 \quad -3 \quad 2 \right\} \quad \text{Periodic}$$

(18) What is the sum of odd signal values ?

Ans : Sum of odd signal value is 0.

$$\text{Ex } x[n] = \left\{ -1 \quad -2 \quad \underset{\uparrow}{0} \quad 2 \quad 1 \right\} \quad \text{Sum} = 0$$

$$x[n] = \left\{ \underset{\uparrow}{0} \quad -2 \quad 3 \quad -3 \quad 2 \right\} \quad \text{Sum} = 0$$

(19) How to check whether the given signal is periodic or not ?

Ans : If the digital frequency of the signal is rational number then the signal is periodic. Otherwise signal is nonperiodic.

$$\text{Ex } x[n] = \cos(0.6\pi n) \quad \text{where } \omega = 0.6\pi \text{ and } f = 0.3 = \frac{3}{10}$$

Here f is a rational number, so $x[n]$ is periodic signal with period = 10

(20) What is the concept of digital frequency f ?

Ans : Digital Frequency is ratio of Analog Frequency to Sampling frequency. $i.e. f = \frac{F}{F_s}$

(21) What is the range of ω and f ?

Ans : Range of Digital frequency ω is $(-\pi, \pi]$

$$\text{Range of Digital frequency } f \text{ is } \left(-\frac{1}{2}, \frac{1}{2} \right]$$

(22) What is the unit of digital frequency ω and f ?

Ans : Unit of digital frequency ω is radians and f is unit less quantity.

(23) Classify the following signal : Finite Length or Infinite Length :-

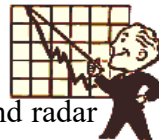
$$x[n] = u[n] + 2u[n-1] - 3u[n-5]$$

Ans : Finite length with length $N = 5$

(24) What is correlation ?

Ans : Correlation gives a measure of similarity between two data sequences. In this process, two signals are compared and the degree to which the two signals are similar is computed.





(25) What are the applications of Correlation ?

Ans : Typical applications of correlation include speech processing, image processing and radar systems.

In a radar system, the transmitted signal is correlated with the echo signal to locate the position of the target. Similarly, in speech processing systems, different waveforms are compared for voice recognition.

(26) What is the application of Convolution ?

Ans : Application of Convolution is to find output of Digital Filter for any given input signal.

Output of Digital filter $y[n]$ is linear Convolution of input signal $x[n]$ and impulse response of the filter $h[n]$.

(27) What are the properties of Convolution ?

Ans :

i) Commutative

$$x[n] * h[n] = h[n] * x[n]$$

ii) Associative

$$(x[n] * h_1[n] * h_2[n]) = (x[n] * h_1[n]) * h_2[n]$$

iii) Distributive

$$x[n] * [h_1[n] + h_2[n]] = x[n] * h_1[n] + x[n] * h_2[n].$$

(28) Consider $x_1[n]$ is periodic with period = 4 and $x_2[n]$ is periodic with period = 6 .

Let $x[n] = x_1[n] + x_2[n]$. What will be the period of $x[n]$?

Ans : Period $N = \text{LCM} \{ N_1, N_2 \} = 12$

(29) Let $x[n] = \delta[n] + 2u[n] - 2u[n-4]$. Determine which of the following classification is true for $x[n]$.

(a) Periodic, Finite length

(b) Periodic, Infinite length

(c) Non periodic, Finite length

(d) Non-periodic, Infinite length

Ans : Non periodic, finite length

