G. PULLAIAH COLLEGE OF ENGINEERING AND TECHNOLOGY

Accredited by NAAC with 'A' Grade of UGC, Approved by AICTE, New Delhi

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Department of Electronics and Communication Engineering

Bridge Course
On
Digital Communication Systems

1.SIGNALS & SYSTEMS

Signal:

Signal is a time varying physical phenomenon which is intended to convey information.

OR

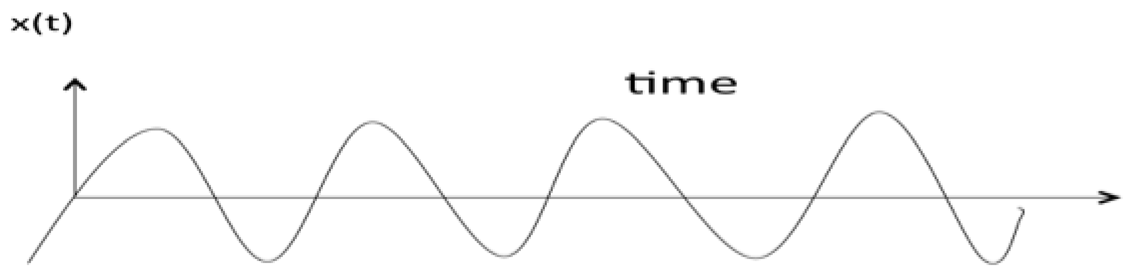
Signal is a function of time.

OR

Signal is a function of one or more independent variables, which contain some information.

Example: voice signal, video signal, signals on telephone wires etc.

Note: Noise is also a signal, but the information conveyed by noise is unwanted hence it is considered as undesirable.



System:

System is a device or combination of devices, which can operate on signals and produces corresponding response. Input to a system is called as excitation and output from it is called as response.

For one or more inputs, the system can have one or more outputs.

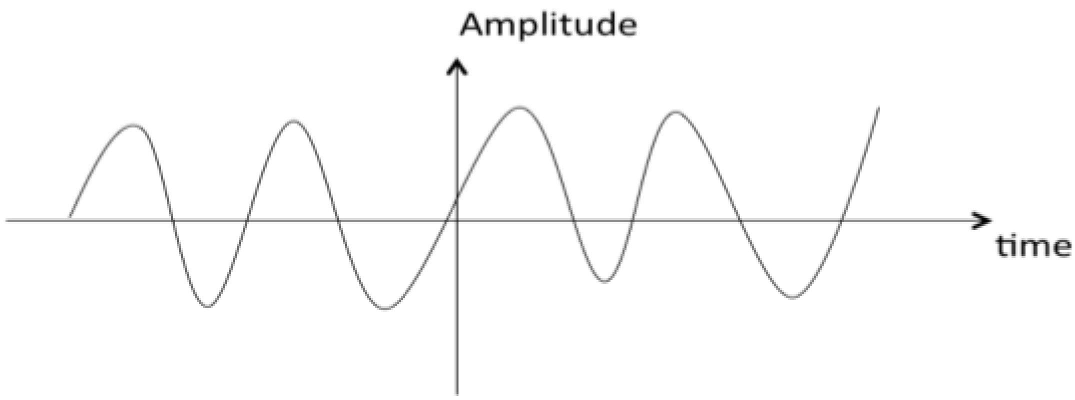
Example: Communication System

Signals are classified into the following categories

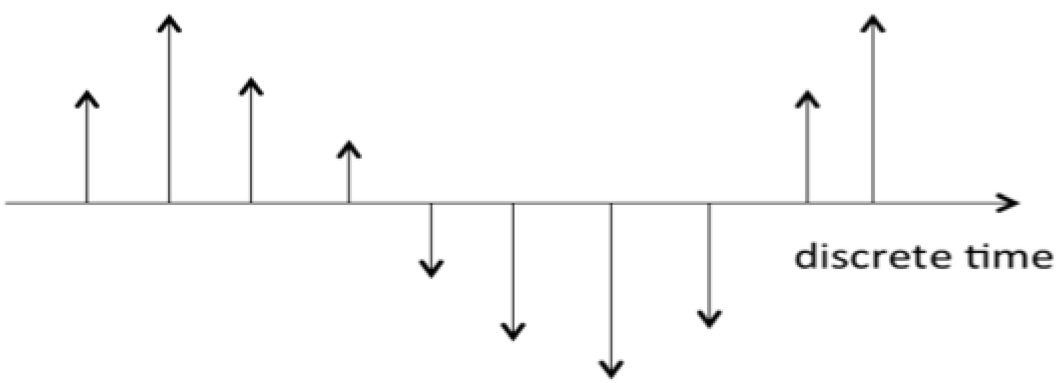
- Continuous Time and Discrete Time Signals
- Deterministic and Non-deterministic Signals
- Even and Odd Signals
- Periodic and Aperiodic Signals
- Energy and Power Signals
- Real and Imaginary Signals

Continuous Time and Discrete Time Signals

A signal is said to be continuous when it is defined for all instants of time.

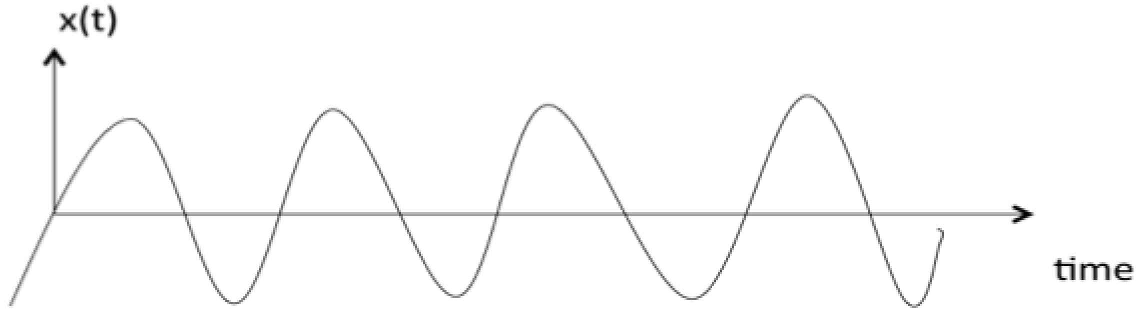


A signal is said to be discrete when it is defined at only discrete instants of time/

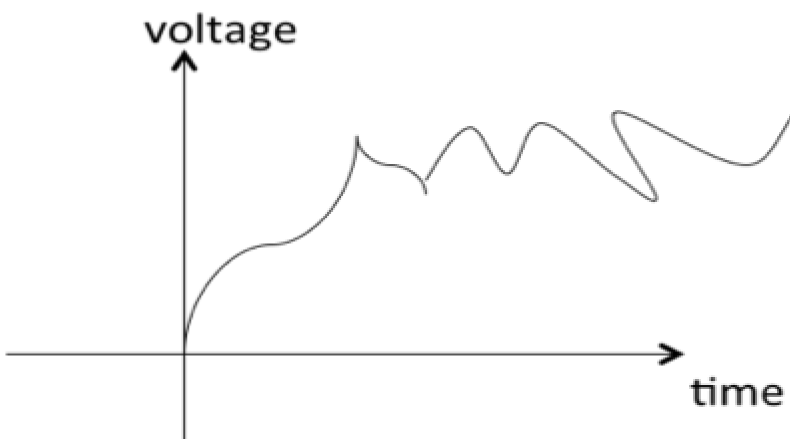


Deterministic and Non-deterministic Signals

A signal is said to be deterministic if there is no uncertainty with respect to its value at any instant of time. Or, signals which can be defined exactly by a mathematical formula are known as deterministic signals.



A signal is said to be non-deterministic if there is uncertainty with respect to its value at some instant of time. Non-deterministic signals are random in nature hence they are called random signals. Random signals cannot be described by a mathematical equation. They are modelled in probabilistic terms.



Even and Odd Signals

A signal is said to be even when it satisfies the condition $x(t) = x(-t)$

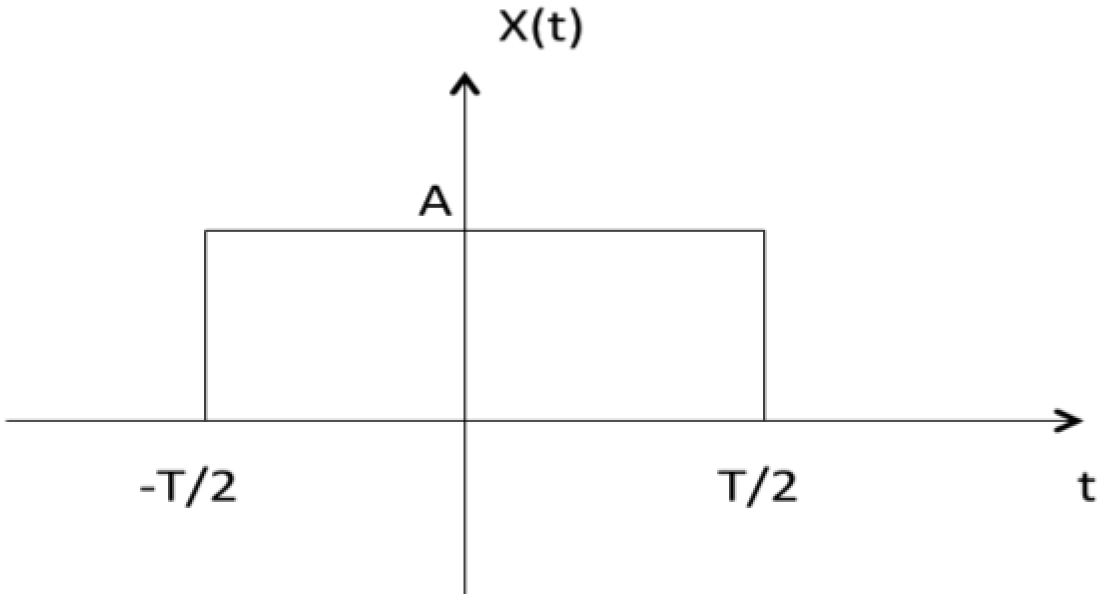
Example 1: t^2, t^4… cost etc.

Let $x(t) = t^2$

$x(-t) = (-t)^2 = t^2 = x(t)$

$\therefore t^2$ is even function

Example 2: As shown in the following diagram, rectangle function $x(t) = x(-t)$ so it is also even function.



A signal is said to be odd when it satisfies the condition x(t) = -x(-t)

Example: t, t^3 ... and sin t

Let x(t) = sin t

x(-t) = sin(-t) = -sin t = -x(t)

∴sin t is odd function.

Any function f(t) can be expressed as the sum of its even function f_e(t) and odd function f_o(t).

$f(t) = f_e(t) + f_0(t)$

where

$f_e(t) = ½[f(t) + f(-t)]$ and $f_0(t) = ½[f(t) - f(-t)]$

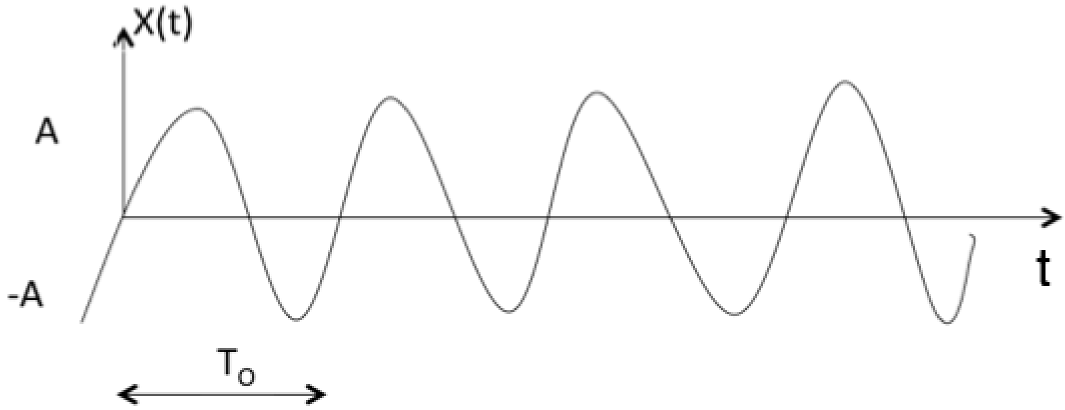
Periodic and Aperiodic Signals

A signal is said to be periodic if it satisfies the condition x(t) = x(t + T) or x(n) = x(n + N).

Where

T = fundamental time period,

1/T = f = fundamental frequency.



The above signal will repeat for every time interval T_0 hence it is periodic with period T_0.

Energy and Power Signals

A signal is said to be energy signal when it has finite energy.

A signal is said to be power signal when it has finite power.

NOTE:A signal cannot be both, energy and power simultaneously. Also, a signal may be neither energy nor power signal.

Power of energy signal = 0

Energy of power signal = ∞

Real and Imaginary Signals

A signal is said to be real when it satisfies the condition x(t) = x*(t)

A signal is said to be odd when it satisfies the condition x(t) = -x*(t)

Example:

If x(t)= 3 then x*(t)=3*=3 here x(t) is a real signal.

If x(t)= 3j then x*(t)=3j* = -3j = -x(t) hence x(t) is a odd signal.

Note: For a real signal, imaginary part should be zero. Similarly for an imaginary signal, real part should be zero.

2. SAMPLING PROCESS

Sampling: A message signal may originate from a digital or analog source. Ifthe message signal is analog in nature, then it has to be converted into digital form beforeit can transmitted by digital means. The process by which the continuous-time signal isconverted into a discrete–time signal is called Sampling.Sampling operation is performed in accordance with the sampling theorem.

Statement:- "If a band –limited signal g(t) contains no frequency components for $|f| > W$, then it is completely described by instantaneous values g(kTs) uniformly spaced in time with period $Ts \leq 1/2W$. If the sampling rate, fs is equal to the Nyquist rate or greater ($fs \geq 2W$), the signal g(t) can be exactly reconstructed.

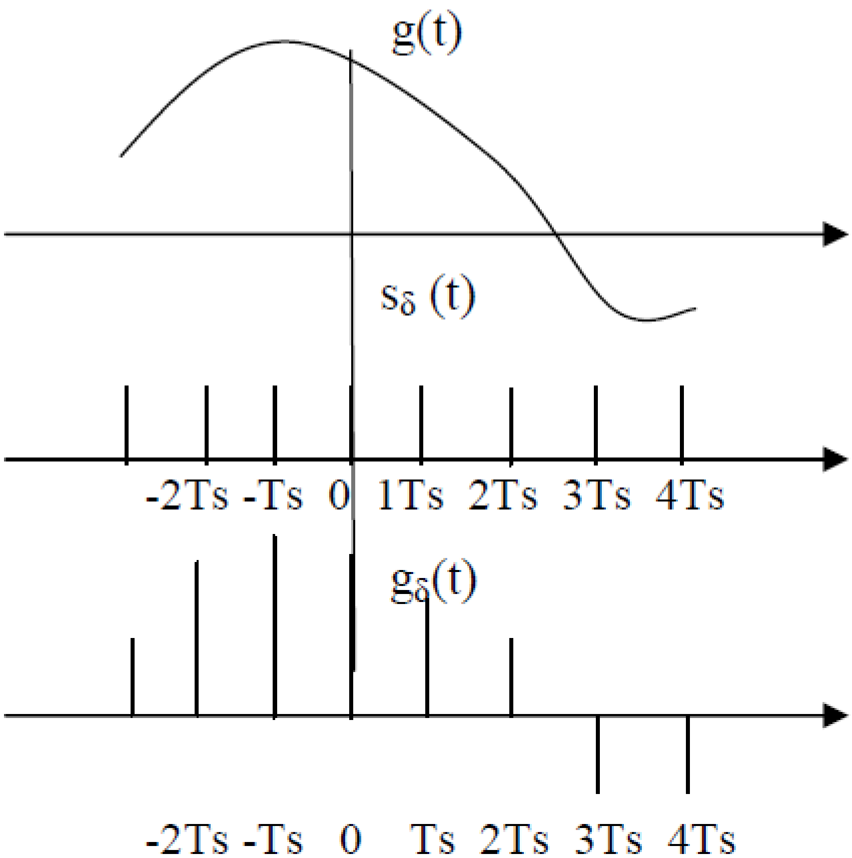


Fig 2.1: Sampling process

Proof:- Consider the signal g(t) is sampled by using a train of impulses $s_\delta(t)$.

Let $g_\delta(t)$ denote the ideally sampled signal, can be represented as

$$g_\delta(t) = g(t).s_\delta(t) \quad \text{-------------------} \quad 2.1$$

where $s_\delta(t)$ – impulse train defined by

$$s_\delta(t) = \sum_{k=-\infty}^{+\infty} \delta(t - kT_s) \quad \text{-------------------} \quad 2.2$$

Therefore

$$g_\delta(t) = g(t) \, . \sum_{k=-\infty}^{+\infty} \delta(t - kT_s)$$

$$= \sum_{k=-\infty}^{+\infty} g(kT_s).\delta(t - kT_s) \quad \text{-----------} \quad 2.3$$

The Fourier transform of an impulse train is given by

$$S_\delta(f) = F[s_\delta(t)] = f_s \sum_{n=-\infty}^{+\infty} \delta(f - nf_s) \quad \text{------------------} \quad 2.4$$

Applying F.T to equation 2.1 and using convolution in frequency domain property,

$$G_\delta(f) = G(f) * S_\delta(f)$$

Using equation 2.4, $G_\delta(f) = G(f) * f_s \sum_{n=-\infty}^{+\infty} \delta(f - nf_s)$

$$G_\delta(f) = f_s \sum_{n=-\infty}^{+\infty} G(f - nf_s) \quad \text{-----------------} \quad 2.5$$

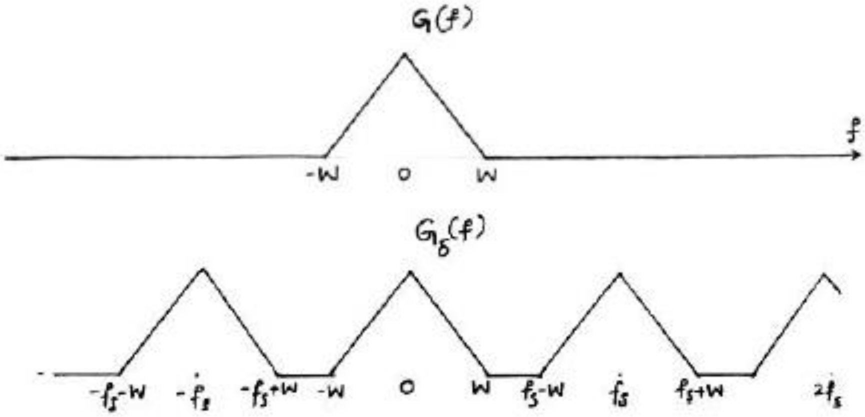


Fig. 2.2 Over Sampling ($f_s > 2W$)

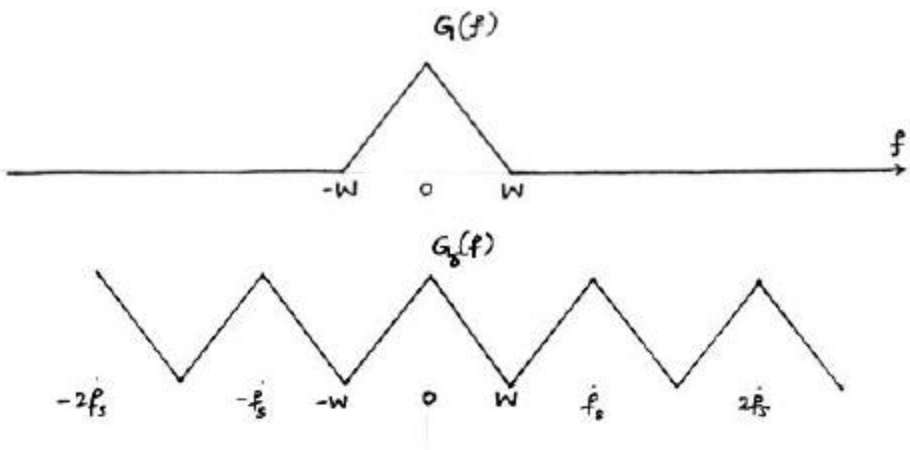


Fig. 2.3 Nyquist Rate Sampling ($f_s = 2W$)

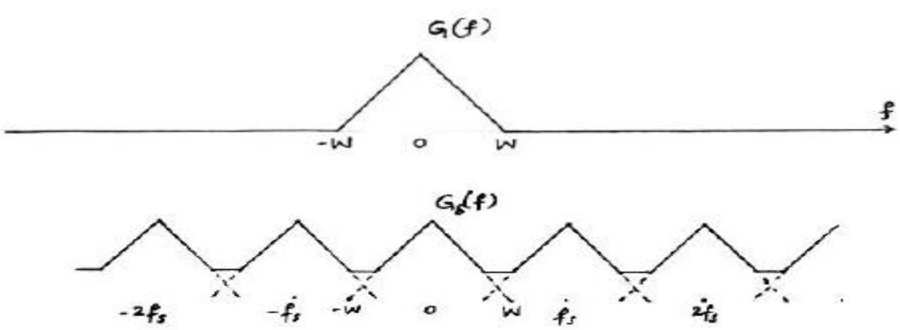
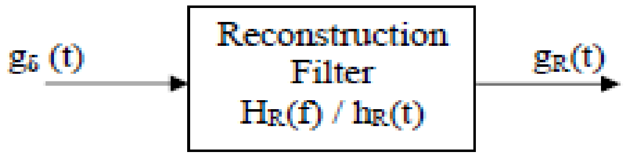


Fig. 2.4 Under Sampling ($f_s < 2W$)

Reconstruction of $g(t)$ from $g\delta(t)$:
By passing the ideally sampled signal $g\delta(t)$ through an low pass filter (called Reconstruction filter) having the transfer function HR(f) with bandwidth, B satisfying the condition $W \leq B \leq (fs - W)$, we can reconstruct the signal g(t). For an ideal reconstruction filter the bandwidth B is equal to W.



The output of LPF is, $g_R(t) = g_\delta(t) * h_R(t)$

where $h_R(t)$ is the impulse response of the filter.

In frequency domain, $G_R(f) = G_\delta(f) . H_R(f)$.

For the ideal LPF $H_R(f) = \begin{cases} K & -W \leq f \leq +W \\ 0 & \text{otherwise} \end{cases}$

then impulse response is $h_R(t) = 2WT_s . \text{Sinc}(2Wt)$

Correspondingly the reconstructed signal is

$$g_R(t) = [\, 2WT_s\, \text{Sinc}\,(2Wt)] * [g_\delta(t)]$$

$$g_R(t) = 2WT_s \sum_{K=-\infty}^{+\infty} g(kTs).Sinc(2Wt) * \delta(t - kTs)$$

$$g_R(t) = 2WT_s \sum_{K=-\infty}^{+\infty} g(kTs).Sinc[2W(t - kTs)]$$

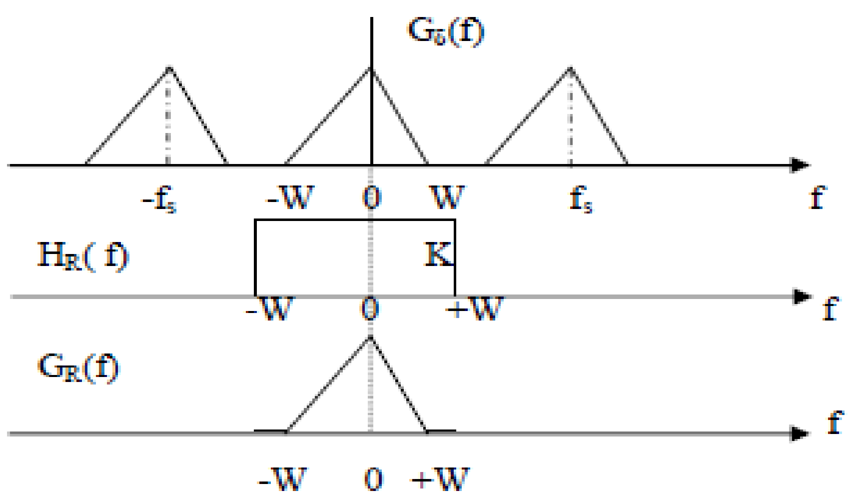
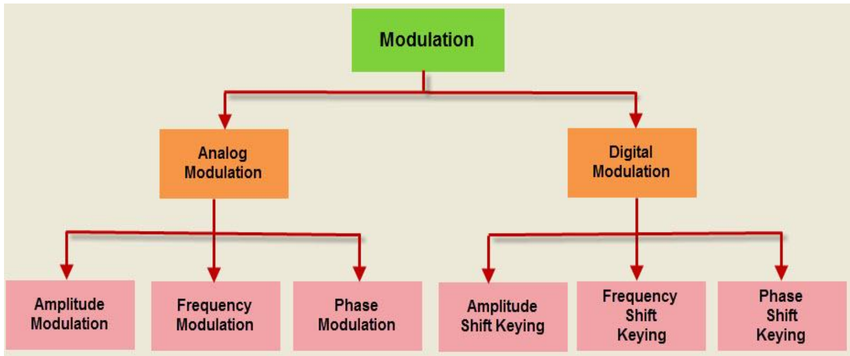


Fig: 2.5 Spectrum of sampled signal and reconstructed signal

3. MODULATION

Modulation is the process of changing the characteristics (amplitude, phase and frequency) of the carrier signal according to instantaneousamplitude of message signal.



Analog Modulation

Analog CW Modulation:

In analog CW modulation, analog signal (sinusoidal signal) is used as a carrier signal that modulates the message signal or data signal.

Analog Modulation

- Amplitude Modulation (AM)
- Frequency Modulation (FM)
- Phase Modulation (PM)

Amplitude Modulation:

Amplitude modulation was developed in the beginning of the 20th century. It was the earliest modulation technique used to transmit voice by radio. This type of modulation technique is used in electronic communication. In this modulation, the amplitude of the carrier signal varies in accordance with the message signal, and other factors like phase and frequency remain constant.

Frequency Modulation:

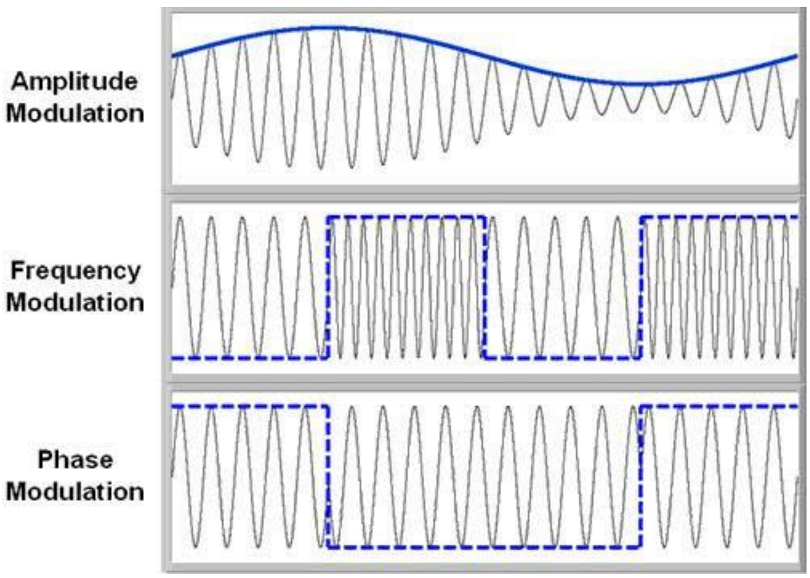
In this type of modulation, the frequency of the carrier signal varies in accordance with the message signal, and other parameters like amplitude and phase remain constant. Frequency modulation is used in different applications like radar, radio and telemetry, seismic prospecting and monitoring newborns for seizures via EEG, etc.

This type of modulation is commonly used for broadcasting music and speech, magnetic tape recording systems, two way radio systems and video transmission systems. When noise occurs naturally in radio systems, frequency modulation with sufficient bandwidth provides an advantage in cancelling the noise.

Phase Modulation:

In this type of modulation, the phase of the carrier signal varies in accordance with the message signal. When the phase of the signal is changed, then it affects the frequency. So, for this reason, this modulation is also comes under the frequency modulation.

Generally, phase modulation is used for transmitting waves. It is an essential part of many digital transmission coding schemes that underlie a wide range of technologies like GSM, WiFi, and satellite television. This type of modulation is used for signal generation in al synthesizers, such as the Yamaha DX7 to implement FM synthesis.



Analog Pulse Modulation

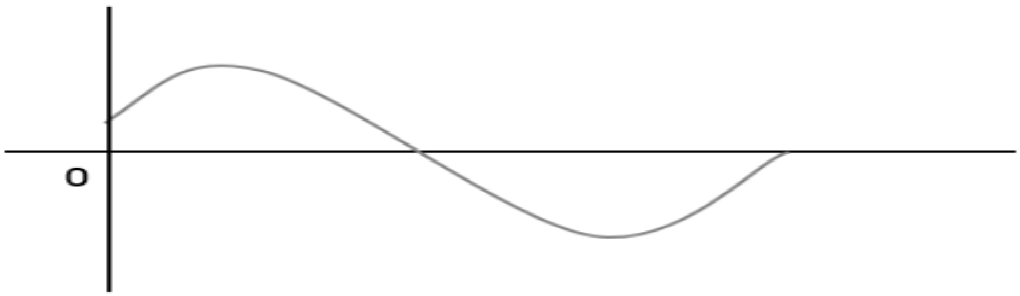
After the continuous wave modulation, the next division is Pulse modulation. Pulse modulation is further divided into analog and digital modulation. The analog modulation techniques are mainly classified into Pulse Amplitude Modulation, Pulse Duration Modulation/Pulse Width Modulation, and Pulse Position Modulation.

Pulse Amplitude Modulation:

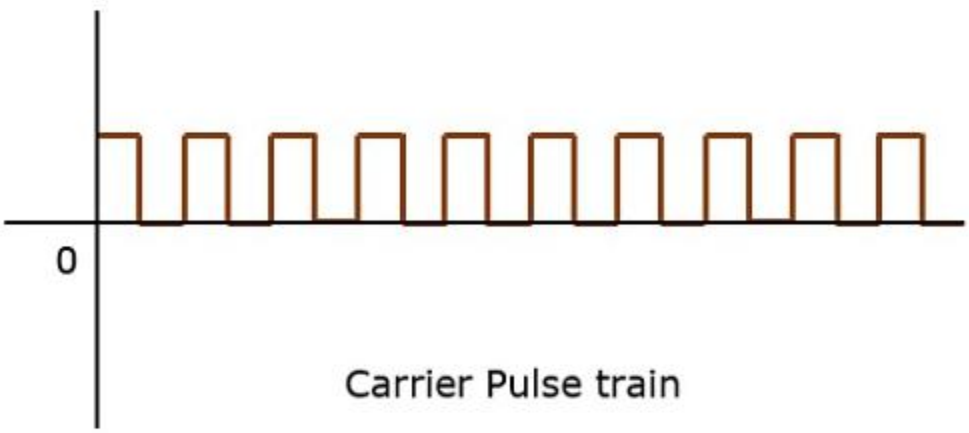
Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. In natural PAM, a signal sampled at the Nyquist rate is reconstructed, by passing it through an efficient Low Pass Frequency (LPF) with exact cutoff frequency

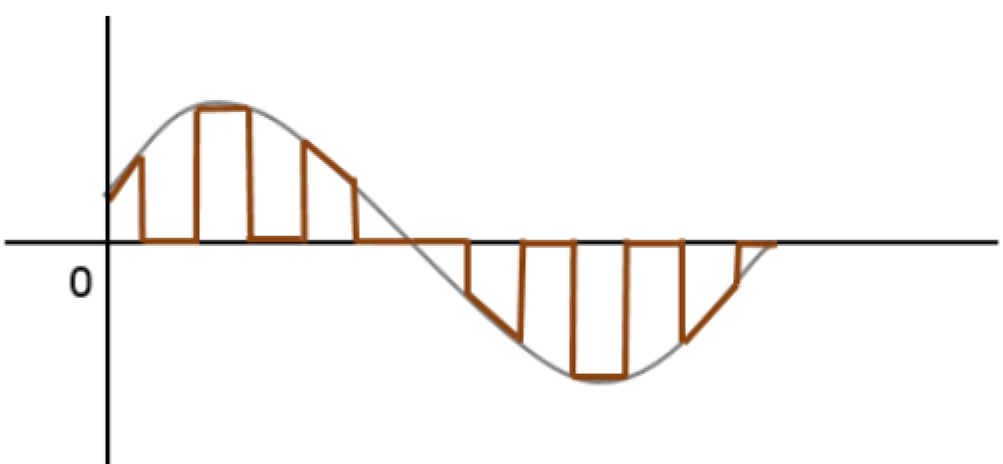
The following figures explain the Pulse Amplitude Modulation.



Modulating signal

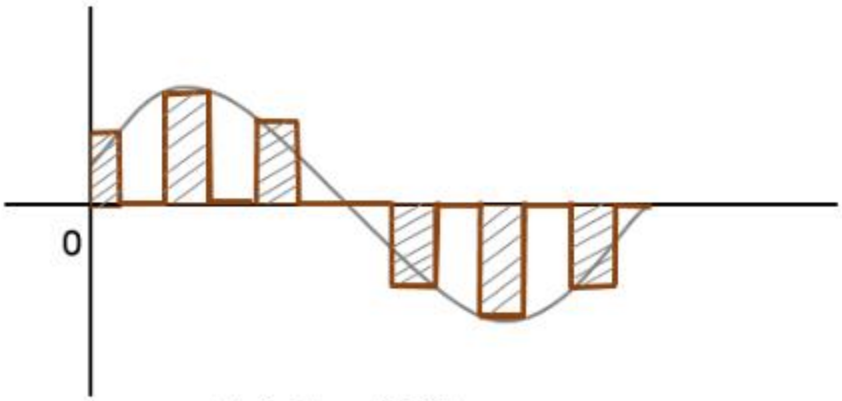


Carrier Pulse train



Natural PAM

Though the PAM signal is passed through an LPF, it cannot recover the signal without distortion. Hence to avoid this noise, flat-top sampling is done as shown in the following figure.



Flat-Top PAM

Flat-top sampling is the process in which sampled signal can be represented in pulses for which the amplitude of the signal cannot be changed with respect to the analog signal, to be sampled. The tops of amplitude remain flat. This process simplifies the circuit design.

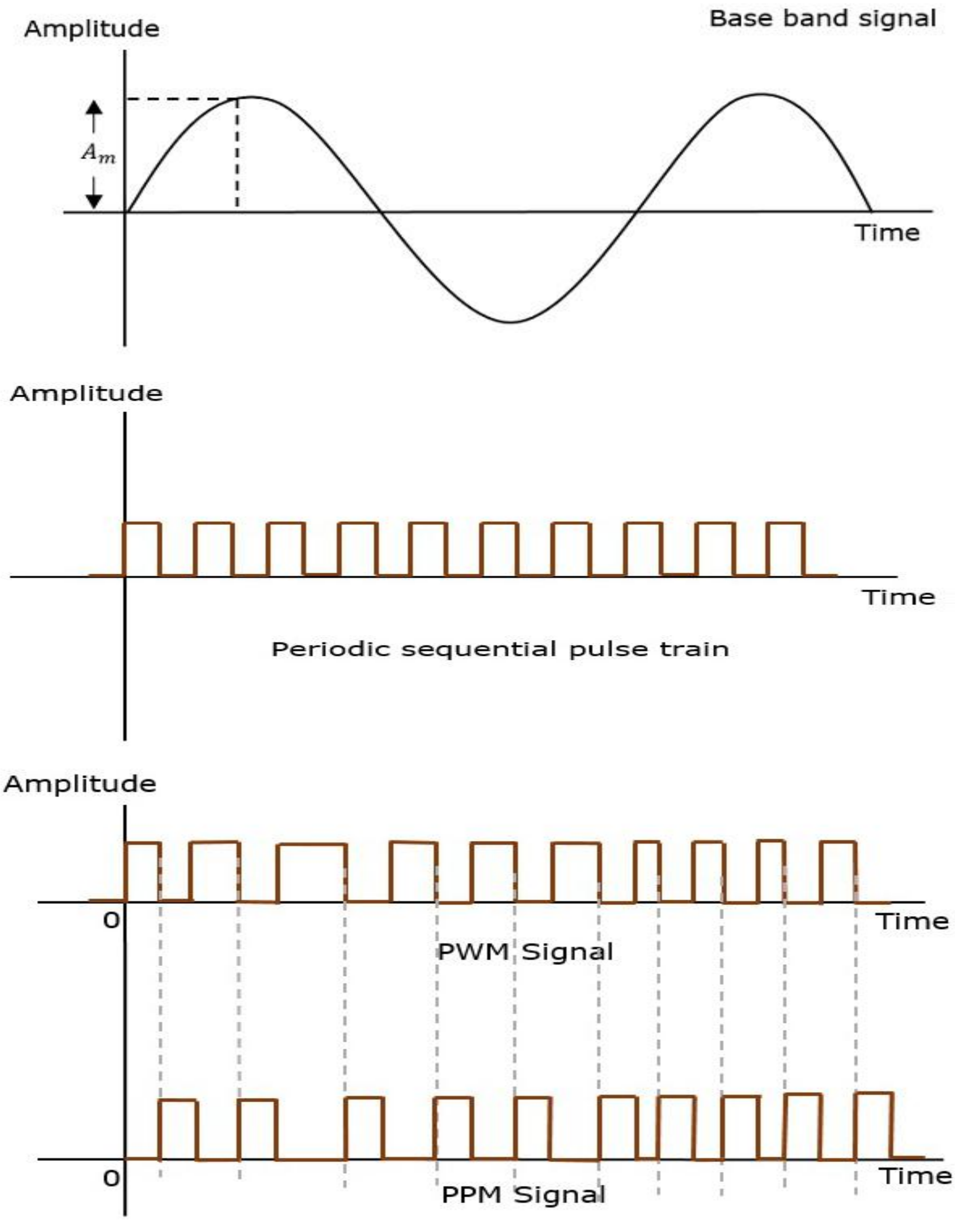
Pulse Width Modulation:

Pulse Width Modulation (PWM) or Pulse Duration Modulation (PDM) or Pulse Time Modulation (PTM) is an analog modulating scheme in which the duration or width or time of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude, to a desired level and hence the noise is limited.

Pulse Position Modulation:

Pulse Position Modulation (PPM) is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.



Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses.

Advantage:

As the amplitude and width are constant, the power handled is also constant.

Disadvantage:

The synchronization between transmitter and receiver is a must.

Comparison between PAM, PWM, and PPM

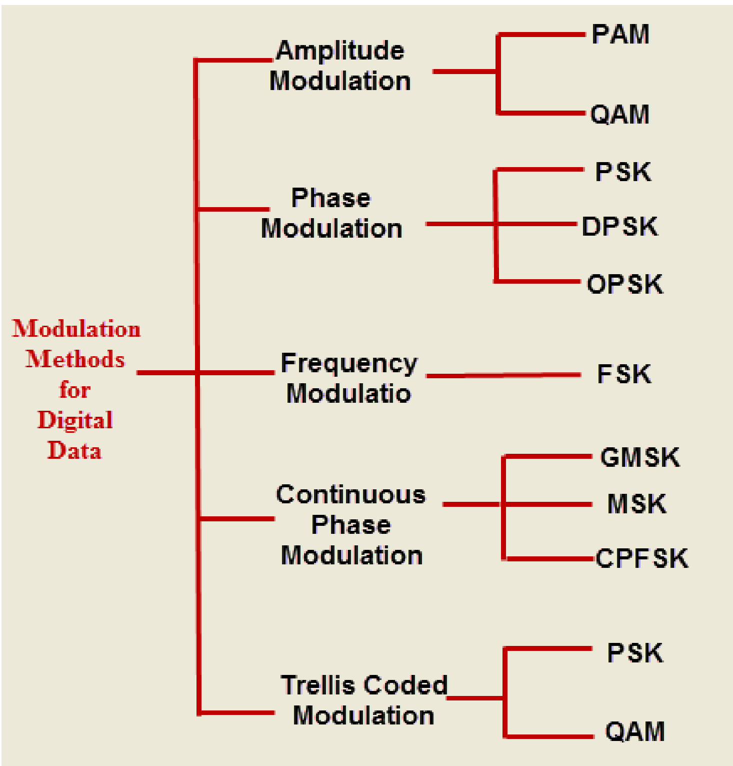
The comparison between the above modulation processes is presented in a single table.

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse	Bandwidth depends on the rise time of the pulse	Bandwidth depends on the rise time of the pulse
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses	Instantaneous transmitter power remains constant with the width of the pulses
System complexity is high	System complexity is low	System complexity is low
Noise interference is high	Noise interference is low	Noise interference is low
It is similar to amplitude modulation	It is similar to frequency modulation	It is similar to phase modulation

Digital Modulation

For a better quality and efficient communication, digital modulation technique is employed. The main advantages of the digital modulation over analog modulation include available bandwidth , high noise immunity and permissible power. In digital modulation, a message signal is converted from analog to digital message, and then modulated by using a carrier wave.

The carrier wave is switched on and off to create pulses such that the signal is modulated. Similar to the analog, in this system, the type of the digital modulation is decided by the variation of the carrier wave parameters like amplitude, phase and frequency.



Types of Digital Modulation

Difference Between Pulse Modulation And Continuous Wave Modulation

Pulse modulation	**Continuous wave modulation**
The modulated signal is in the form of pulses.	The modulated signal is in the form of continuous signals.
It is used sampling technique.	It is not used sampling technique.
It has required large bandwidth.	It has required less bandwidth.
Pulse modulation has both analog and digitalnature.	It has only analog modulation.
In pulse modulation, the train of pulses is used as a carrier.	High frequency sine wave is used as carrier.
The input signal is either analog or digital.	Input signal is analog signal only.
The example of pulse modulation is PAM, PPM, PWM, DPCM, ADM etc.	The example of continuous wave modulation is AM (amplitude modulation), FM (frequency modulation) and PM (pulse modulation).
It is used in satellite communication.	It is used in radio and TV broadcasting.

3. ELEMENTS OF DIGITAL COMMUNICATION SYSTEMS

The figure below shows the functional elements of a digital communication system.

Source of Information: 1. Analog Information Sources.2. Digital Information Sources.

Analog Information Sources → Microphone actuated by a speech, TV Camera scanning ascene, continuous amplitude signals.
Digital Information Sources → These are teletype or the numerical output of computerwhich consists of a sequence of discrete symbols or letters.
An analog information is transformed into a discrete information through theprocess of sampling and quantizing.

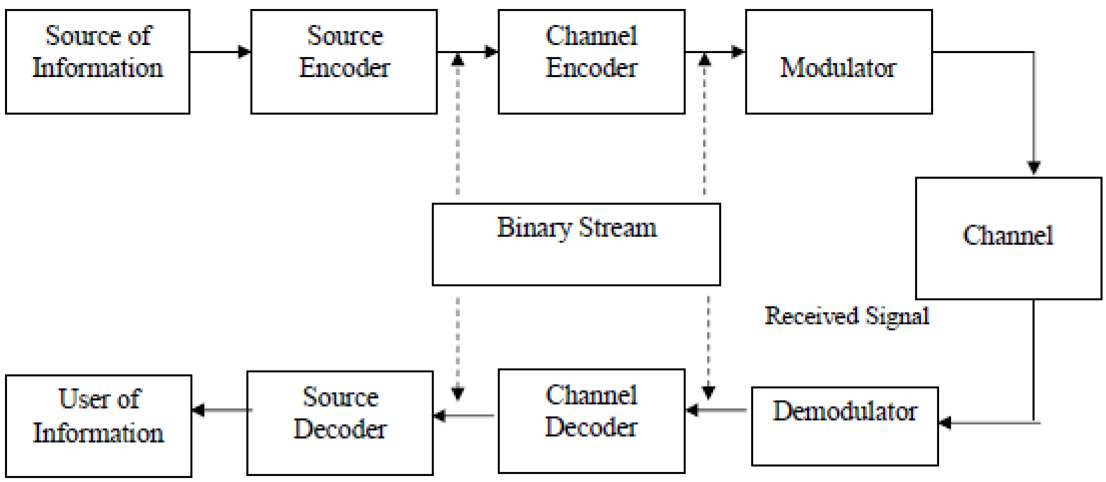


Fig 1.2: Block Diagram of a Digital Communication System

SOURCE ENCODER / DECODER:

The Source encoder (or Source coder) converts the input i.e. symbol sequenceinto a binary sequence of 0's and 1's by assigning code words to the symbols in the inputsequence. For eg. :-If a source set is having hundred symbols, then the number of bitsused to represent each symbol will be 7 because 27=128 unique combinations areavailable. The important parameters of a source encoder are block size, code wordlengths, average data rate and the efficiency of the coder (i.e. actual output data ratecompared to the minimum achievable rate)

At the receiver, the source decoder converts the binary output of the channeldecoder into a symbol sequence. The decoder for a system using fixed – length codewords is quite simple, but the decoder for a system using variable – length code wordswill be very complex.Aim of the source coding is to remove the redundancy in the transmittinginformation, so that bandwidth required for transmission is minimized. Based on theprobability of the symbol code word is assigned. Higher the probability, shorter is the
codeword.
Ex: Huffman coding.

CHANNEL ENCODER / DECODER:
Error control is accomplished by the channel coding operation that consists ofsystematically adding extra bits to the output of the source coder. These extra bits do notconvey any information but helps the receiver to detect and / or correct some of the errorsin the information bearing bits.
There are two methods of channel coding:
1. Block Coding: The encoder takes a block of ‘k’ information bits from the source encoder and adds ‘r’ error control bits, where ‘r’ is dependent on ‘k’ and error control capabilities desired.
2. Convolution Coding: The information bearing message stream is encoded in acontinuous fashion by continuously interleaving information bits and error controlbits.
The Channel decoder recovers the information bearing bits from the coded binary stream.Error detection and possible correction is also performed by the channel decoder.The important parameters of coder / decoder are: Method of coding, efficiency, errorcontrol capabilities and complexity of the circuit.

MODULATOR:
The Modulator converts the input bit stream into an electrical waveform suitablefor transmission over the communication channel. Modulator can be effectively used tominimize the effects of channel noise,to match the frequency spectrum of transmittedsignal with channel characteristics, to provide the capability to multiplex many signals.

DEMODULATOR:
The extraction of the message from the information bearing waveform producedby the modulation is accomplished by the demodulator. The output of the demodulator isbit stream. The important parameter is the method of demodulation.

CHANNEL:
The Channel provides the electrical connection between the source anddestination. The different channels are: Pair of wires, Coaxial cable, Optical fibre, Radiochannel, Satellite channel or combination of any of these.The communication channels have only finite Bandwidth, non-ideal frequencyresponse, the signal often suffers amplitude and phase distortion as it travels over the channel. Also, the signal power decreases due to the attenuation of the channel. Thesignal is corrupted by unwanted, unpredictable electrical signals referred to as noise.The important parameters of the channel are Signal to Noise power Ratio (SNR),usable bandwidth, amplitude and phase response and the statistical properties of noise.

Advantages Of Digital Communication

1. The effect of distortion, noise and interference is less in a digital communication system. This is because the disturbance must be large enough to change the pulse from one state to the other.

2. Regenerative repeaters can be used at fixed distance along the link, to identify and regenerate a pulse before it is degraded to an ambiguous state.

3. Digital circuits are more reliable and cheaper compared to analog circuits.

4. The Hardware implementation is more flexible than analog hardware because of the use of microprocessors, VLSI chips etc.

5. Signal processing functions like encryption, compression can be employed to maintain the secrecy of the information.

6. Error detecting and Error correcting codes improve the system performance by reducing the probability of error.

7. Combining digital signals using TDM is simpler than combining analog signals using FDM. The different types of signals such as data, telephone, TV can be treated as identical signals in transmission and switching in a digital communication system.

8. We can avoid signal jamming using spread spectrum technique.

Disadvantages Of Digital Communication

1. Large System Bandwidth:- Digital transmission requires a large system bandwidth to communicate the same information in a digital format as compared to analog format.

2. System Synchronization:- Digital detection requires system synchronization whereas the analog signals generally have no such requirement.

Application Of Digital Communication Systems

There are some important applications of digital communication systems are given below,

- It is used in military application for secure communication and missile guidance.
- It is used in image processing for pattern recognition, robotic vision and image enhancement.
- It is used in digital signal processing.
- The digital communication systems used in telephony for text messaging, etc
- It is used in space communication where a spacecraft transmits signals to earth.
- It is used in video compression.
- It is used in speech processing.
- It is used in digital audio transmission.
- It is used in data compression.