LAKIREDDY BALI REDDY COLLEGE OF ENGINEERING
(AUTONOMOUS)
Accredited by NAAC & NBA (CSE, IT, ECE, EEE & ME)
Approved by AICTE, New Delhi and Affiliated to JNTUK, Kakinada
L.B.Reddy Nagar, Mylavaram-521230, Krishna Dist, Andhra Pradesh, India

B.Tech V Semester

20EC09 – DIGITAL COMMUNICATIONS

Course Educational Objective: This course provides the knowledge on different digital modulation techniques. The course provides different concepts on information theory, block codes and convolution codes. It gives the methods of optimum receivers for digital communication systems and performance of probability of error for digital modulation techniques.

Course Outcomes (COs): **At the end of the course, students will be able to**

CO1: Understand the concepts of digital communication system (Understand – L2)
CO2: Analyze the Baseband and Pass band digital modulation techniques (Analyze – L4)
CO3 :Examine the optimum reception and probability of error of digital modulation(Apply – L3)
CO4:Apply source coding and error control coding techniques in digital communication process(Apply – L3)

UNIT – I

Pulse Digital Modulation: Elements of a Digital Communication System, Sampling and Quantization of signals- Quantization noise, Pulse Digital Modulation Systems: Pulse Code Modulation(PCM) System, Encoding, Regenerative repeaters, Decoding, Reconstruction, effect of noise in PCM-Calculation of output SNR in PCM; Need for non-uniform quantization-Companding- μ-law, A-law; Differential Pulse Code Modulation; Delta Modulation; Adaptive Delta Modulation.

- Elements of a Digital Communication System,
- Sampling

- Quantization of Digital signals
  - Introduction– Uniform Quantization
  - Quantization noise

- Pulse Code Modulation(PCM) System
  - Block diagram of Tx-Rx
  - SNR in PCM system(output)

- Non -uniform quantization
  - Need for non-uniform quantization
  - Companding- μ-law, A-law
- Differential Pulse Code Modulation
  - Delta Modulation- Drawbacks of DM
  - Adaptive Delta Modulation

why digital :-

* to reduce The Size of The Data to be Transmitted.
* To Perform error detection & Error Correction.
* To enable Security.
* To Provide Mixing of Signals & Data.

The Necessity of Digitization

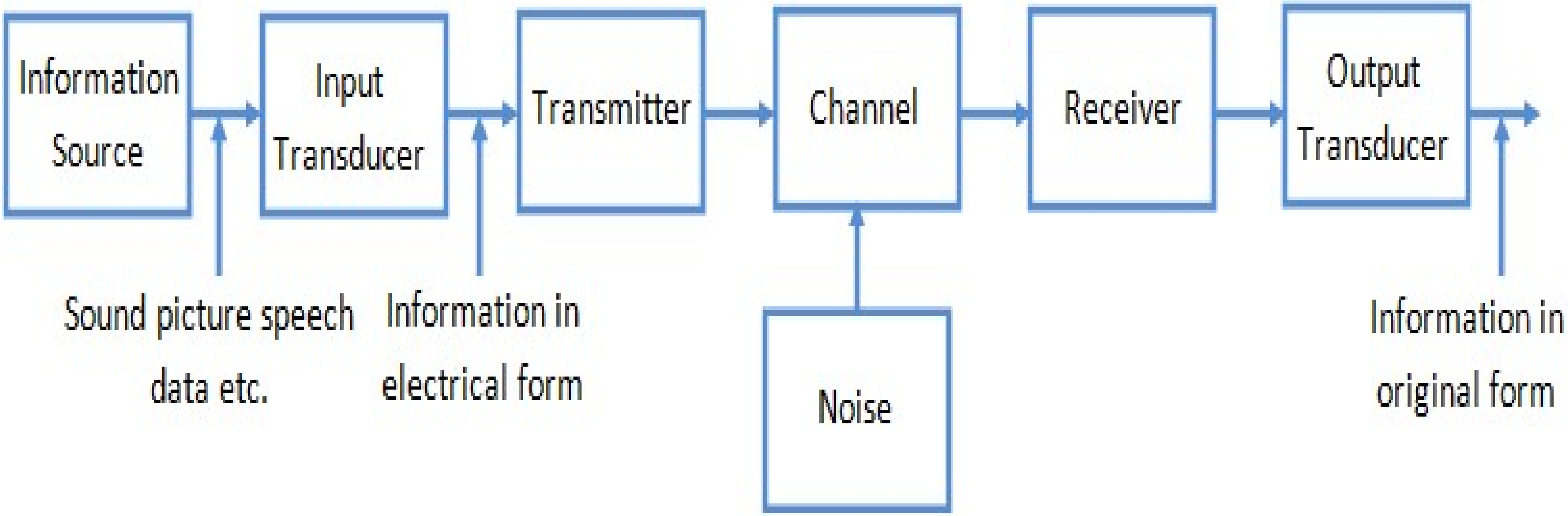
The conventional methods of communication used analog signals for long distance communications, which suffer from many losses such as distortion, interference, and other losses including security breach. In order to overcome these problems, the signals are digitized using different techniques. The digitized signals allow the communication to be more clear and accurate without losses.

Advantages of Digital Communication

There are many advantages of digital communication over analog communication

1. The effect of distortion, noise, and interference is much less in digital signals as they are less affected.
2. Digital circuits are more reliable & easy to design and cheaper than analog circuits.
3. The hardware implementation in digital circuits, is more flexible than analog.
4. Signal processing functions such as encryption and compression are employed in digital circuits to maintain the secrecy of the information.
5. The probability of error occurrence is reduced by employing error detecting and error correcting codes
6. Spread spectrum technique is used to avoid signal jamming.

MODEL OF COMMUNICATION SYSTEM

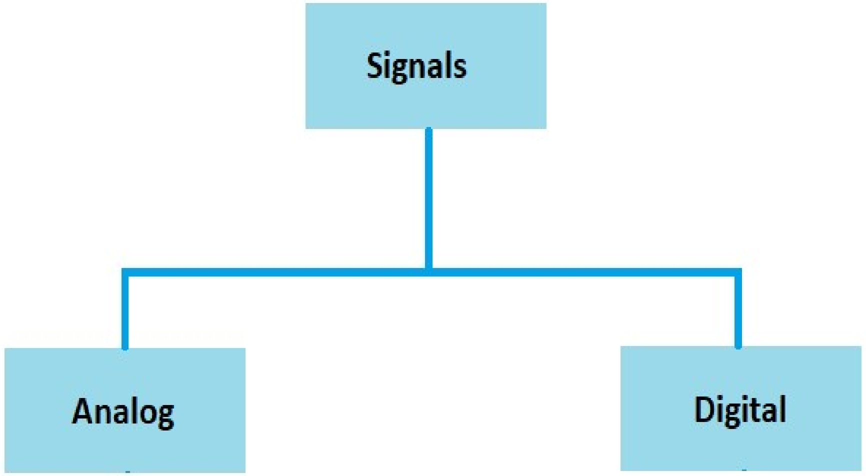


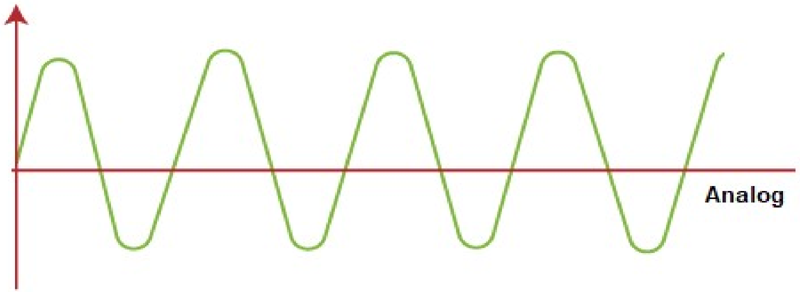
The transmission of information is called communication.

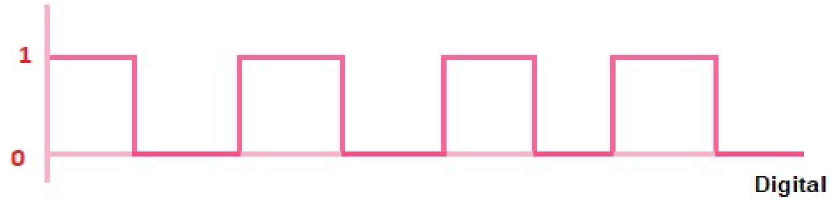
- Every communication has three essential elements : transmitter, channel and receiver The purpose of transmitter is to convert into suitable form of signal that can transmitted through the channel.

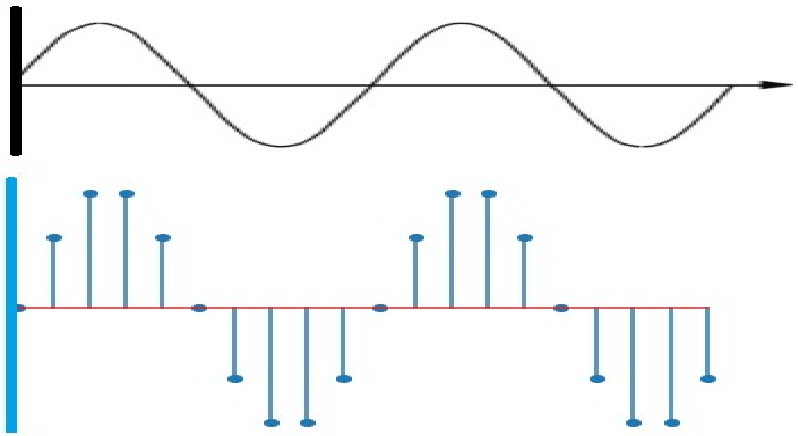
- The channel is central to operation of a communication System. The information - carrying capacity of a communication system is proportional to the channel bandwidth.

- If the o/p of the information source is a non electric signal then a transducer convert it into electric form before it pass through the channel. Moreover, noise is introduced in channel so receiver reconstruct it and send the information to user for.



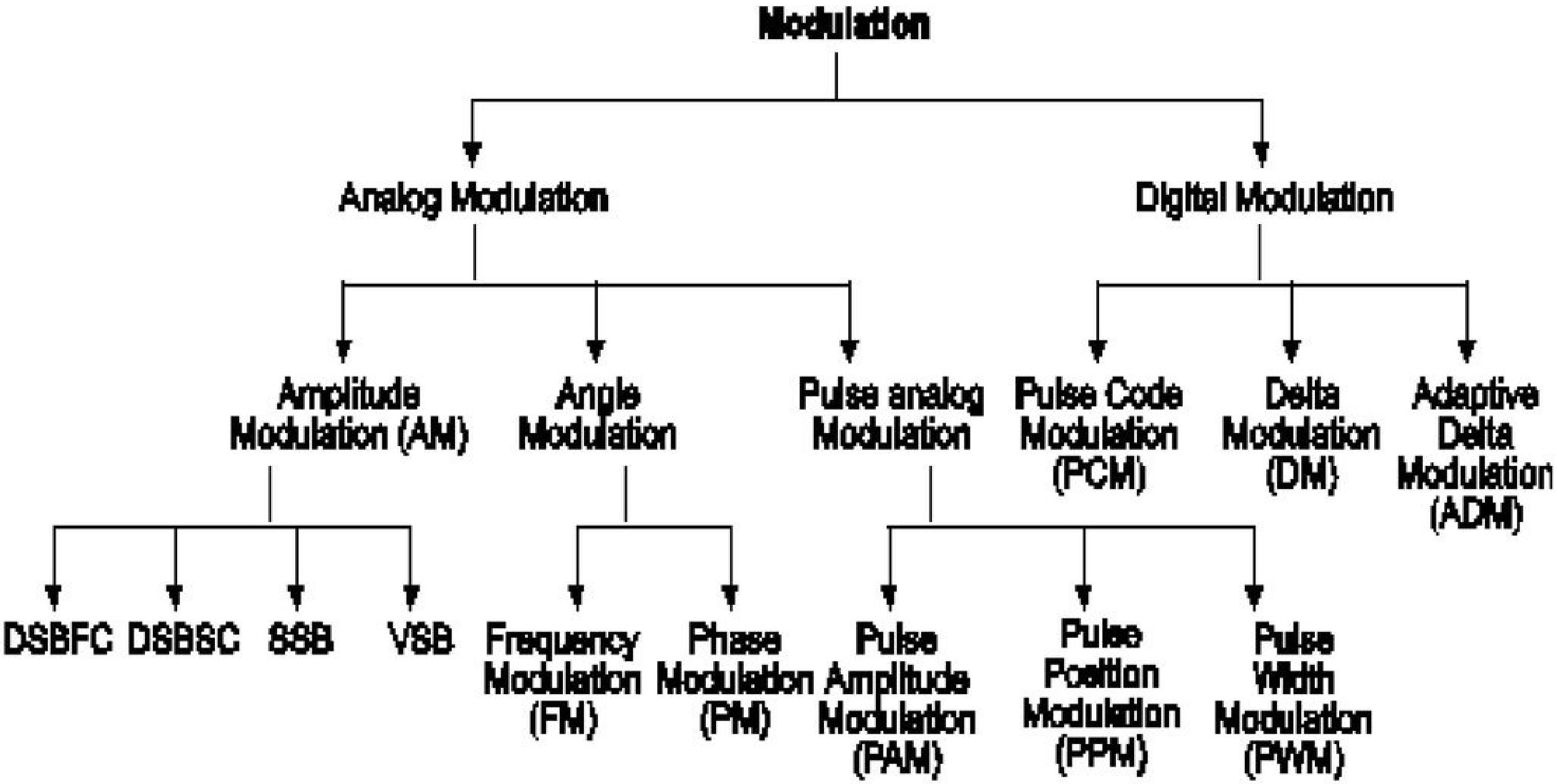






Discrete

Classification of Modulation



Sr. No.	Modulation Techniques	Type	Notation
01	Analog Modulation Techniques	(i) Amplitude Modulation (ii) Frequency Modulation (iii) Phase Modulation	A.M. F.M. P.M.
02	Digital Modulation Techniques	(i) Amplitude Shift Keying (ii) Frequency Shift Keying (iii)Phase Shift Keying	A.S.K. F.S.K. P.S.K.

Block diagram of Digital communication Systems

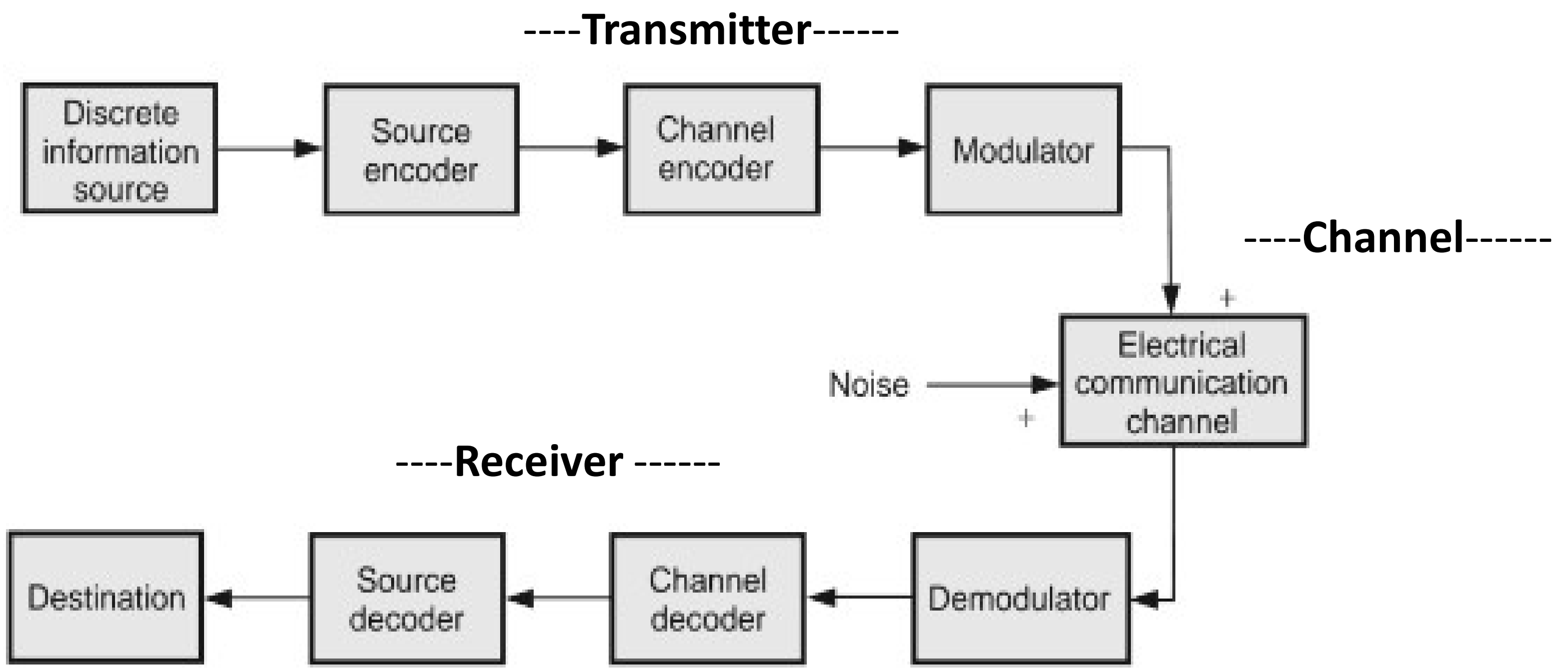


Fig. 1.1.1 Basic digital communication system

In the above **diagram 3 basic signal** processing operations

1.Source coding (source encoder & decoder)

2.Channel coding (Channel encoder & decoder)

3.Modulation

Information Source

- The information source generates the message signal to be transmitted. In case of analog communication, the information source is analog. In case of digital communication, the information source produces a message signal which is not continuously varying with time. Rather the message signal is intermittent with respect to time.
- The examples of discrete information sources are data from computers, teletype etc. Even the message containing text is also discrete.
- The analog signal can be converted to discrete signal by sampling and quantization. In sampling, the analog signal is chopped off at regular time intervals. Those chopped samples form a discrete signal.

Source Encoder and Decoder

- The symbols produced by the information source are given to the source encoder. These symbols cannot be transmitted directly. They are first converted into digital form (i.e. Binary sequence of 1's and 0's) by the source encoder. Every binary '1' and '0' is called a bit. The group of bits is called a codeword.
- The source encoder assigns codewords to the symbols. For every distinct symbol there is a unique codeword. The codeword can be of 4, 8, 16 or 32 bits length. As the number of bits are increased in each codeword, the symbols that can be represented are increased.

  For example, 8 bits will have $2^8 = 256$ distinct codewords. Therefore 8 bits can be used to represent 256 symbols, 16 bits can represent $2^{16} = 65536$ symbols and so on.
- In both of the above examples the number of bits in every codeword is same throughout. That is 8 in first case and 16 in next case respectively. This is called fixed length coding. Fixed length coding is efficient only if all the symbols occur with equal probabilities in a statistically independent sequence.

Ex:- ANJANI - Message

	Fixed length coding	Variable length coding
Message →	ANJANI	ANJANI
Source Alphabets →	{A, N, J, I}	{A, N, J, I}
No. of Symbols →	4	4
No. of bits assigned →	2 for each Symbol	More Probable Symbol → A, N.
Total No. of bits →	4×2 = 8 bits	

No. of bits assigned → A → 0
N → 1
J → 01
I → 10
6 - bits

ie more Probable of occurance - less no of bits assigned
less " " - more " "

Channel Encoder and Decoder

- To reduce the effect of channel noise

- To provide reliable communication over noisy channel

- Introducing redundancy in the channel encoding in prescribed format it can reconstruct the original signal as accurately as possible at the decoder output.

Key features of the system is → Coding rate, Error detection & Error Correction Capabilities, Complexity of Encoder & Decoder.

Important parameters

1) **Block size :** This gives the maximum number of distinct codewords that can be represented by the source encoder. It depends upon maximum number of bits in the codeword. For example, the block size of 8 bits source encoder will have $2^8 = 256$ codewords.

2) **Codeword length :** This is the number of bits used to represent each codeword. For example, if 8 bits are assigned to every codeword, then codeword length is 8 bits.

3) **Average data rate :** It is the output bits per second from the source encoder. The source encoder assigns multiple number of bits to every input symbol. Therefore the data rate is normally higher than the symbol rate. For example let us consider that the symbols are given to the source encoder at the rate of 10 symbols/sec and the length of codeword is 8 bits. Then the output data rate from the source encoder will be,

$$\text{Date rate} = \text{Symbol rate} \times \text{Codeword length}$$

$$= 10 \times 8 = 80 \text{ bits/sec}$$

Information rate is the minimum number of bits per second needed to convey information from source to destination as stated earlier. Therefore optimum data

Sampling theorem (or Nyquist Criterion): the sampling is performed at a proper rate, no info is lost about the original signal and it can be properly reconstructed "If a signal is sampled at a rate at least, but not exactly equal to twice the maximum frequency of the signal, then the waveform can be exactly reconstructed from the samples without any distortion"

Sampling frequency $f_s \geq f_{max}$

The minimum sampling rate of (2W) samples per second, for an analog signal bandwidth of W Hz, is called the Nyquist rate.
Suppose that a signal is band-limited with no frequency components higher than **W** Hertz. That means, W is the highest frequency. For such a signal, for effective reproduction of the original signal , the sampling rate should be twice the highest frequency. **fS = 2W**
where fS is the sampling frequency
W is the highest frequency

Sampling is the processes of converting continuous by taking the “samples” at discrete-time intervals
Sampling interval: – The time that separates sampling points (interval b/w samples),

Analog-to-digital conversion is (basically) a 2 step process: –

1.Sampling - Convert from continuous-time analog signal – 1s obtained by taking the “samples” **of xa (t)** at discrete-time intervals, **Ts**

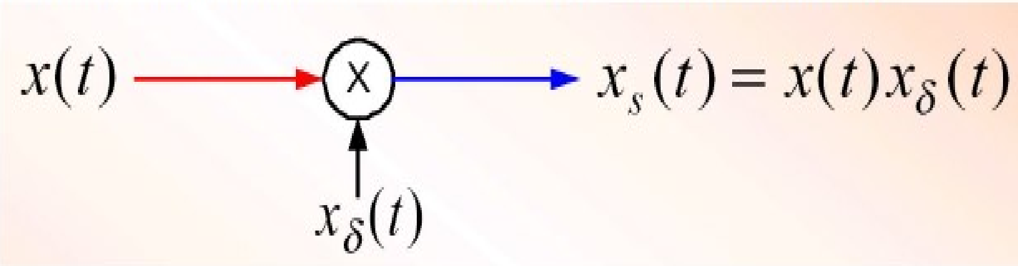
2.Quantization – Convert from discrete-time continuous valued signal

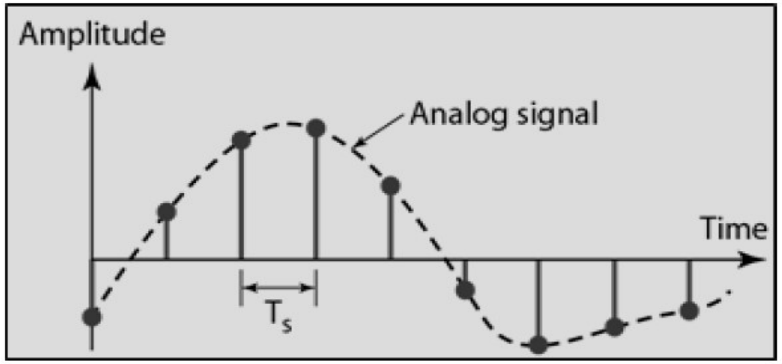
Sampling Rate (or sampling frequency fs):The rate at which the signal is sampled, expressed as the number of samples per second reciprocal of the sampling interval),
$1/T_s = f_s$
Types of Sampling :1. Natural Sampling 2. Flat top Sampling

Ideal Sampling (or Impulse Sampling)

s accomplished by the multiplication of the signal *x(t)* by the uniform train of impulses (comb unction)

:onsider the instantaneous sampling of the analog signal *x(t)*

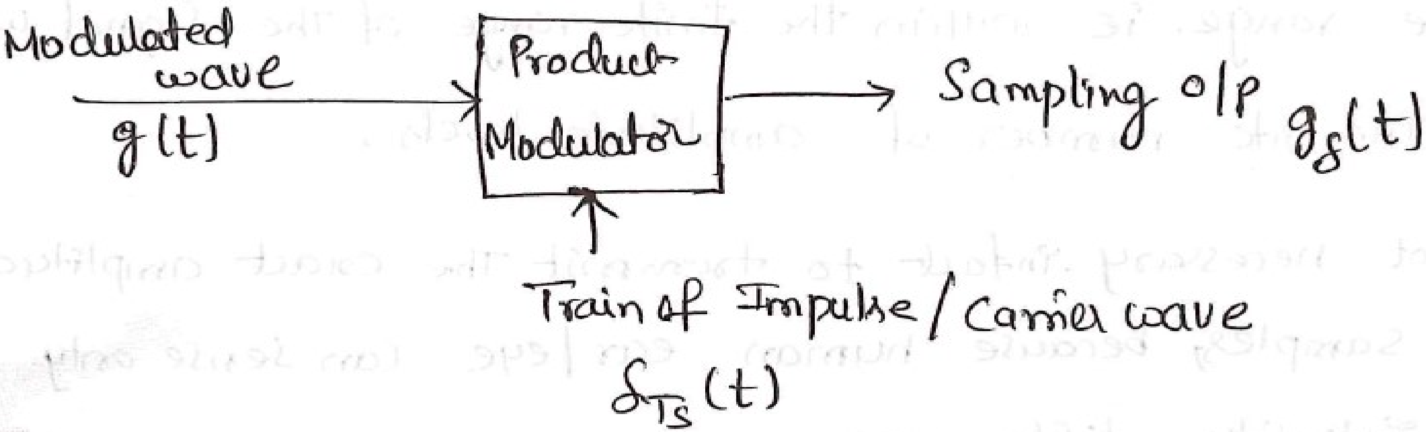




Ideal Sampling

Train of impulse functions select sample values at regular intervals

$$x_s(t) = x(t)\sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$



$$g_\delta(t) = g(t)\,\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} g(t)\,\delta(t - nT_s)$$

$$G_\delta(f) = \sum_{n=-\infty}^{\infty} f_s\, G(f - nf_s)$$

Nyquist Rate $= 2W$ Hz.

Nyquist Interval $= \frac{1}{2W}$ sec.

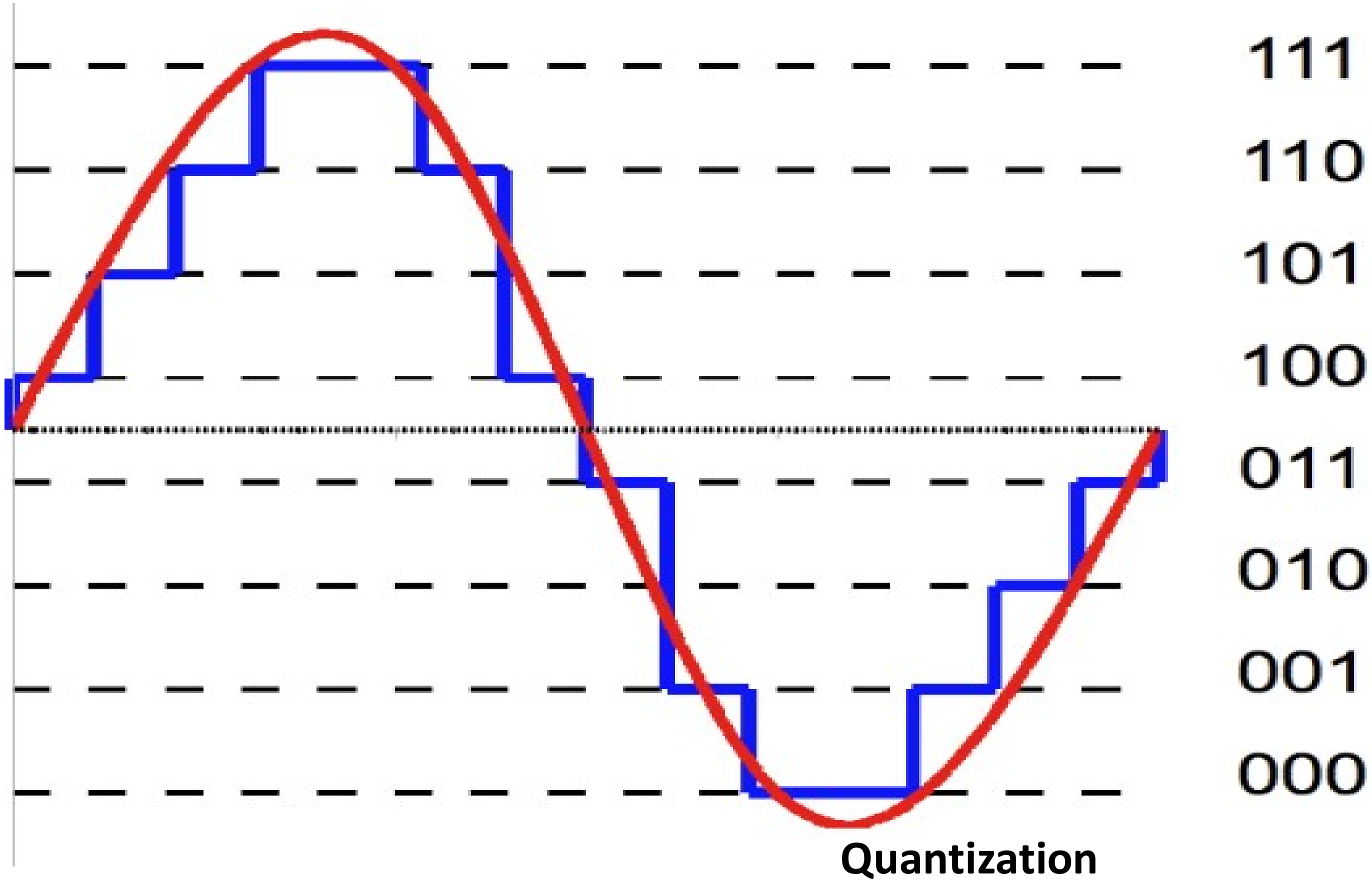
Quantization is the process of mapping continuous amplitude (analog) signal into discrete amplitude (digital) signal.
The analog signal is quantized into countable & discrete levels known as **quantization levels**. Each of these levels represents a fixed input amplitude.

Types Of Quantization

There are two types of quantization.

Uniform Quantization

The type of quantization in which the quantized levels are **uniformly spaced** is known as **uniform quantization**. In uniform quantization, each step size represents a **constant** amount of analog amplitude. it remains constant throughout the signal.
The example of **uniform quantization** is given below,



Quantization

Baseband vs. Passband Communication Systems Communication systems can be classified into two groups depending on the range of frequencies they use to transmit information. These communication systems are classified into BASEBAND or PASSBAND system

Modulation and Demodulation The process of shifting the baseband signal to passband range for transmission is known as MODULATION and the process of shifting the passband signal to baseband frequency range at the receiver is known as DEMODULATION

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