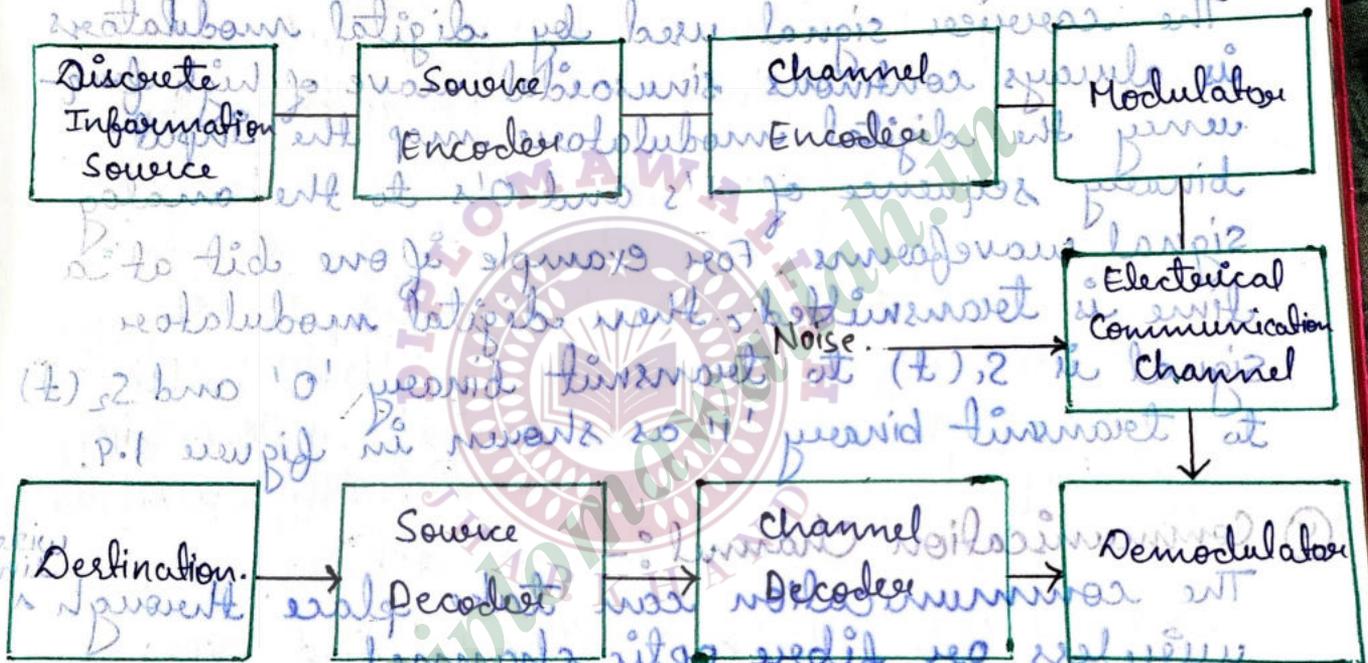


Unit:-09

Digital Communication

Block diagram



① Discrete Information source :-

Information source may be classified into two categories based upon the nature of their output i.e analog information sources and discrete information sources. In case of analog communication, the information source is analog information sources such as microphone. It produces a message signal which is more continuous amplitude signals.

In case of digital communication, the information source produces a message signal which is not continuously varying with time.



② Channel Encoder and Decoder:-
 After converting the message or information signal in the form of binary sequence by the source encoder the signal is transmitted through the channel. The communication channel adds noise and interference to the signal being transmitted.

③ Digital Modulator and Demodulator:-
 • The carrier signal used by digital modulators is always continuous sinusoidal wave of high frequency. The digital modulator map the input binary sequence of 1's and 0's to the analog signal waveforms. For example if one bit at a time is transmitted, then digital modulator signal is $S_1(t)$ to transmit binary '0' and $S_2(t)$ to transmit binary '1' as shown in figure 1.9.

④ Communication Channel:-
 The communication can take place through wireless or fibre optic channel.

ENTROPY

In a practical communication system we usually transmit long sequences of symbols from an information source. Thus we are more interested in the average information that source produces than the information content of a single symbol.

1) The source is stationary so that the probabilities may remain constant with time.



ii) The successive symbols are statistically independent and come from the source at a average rate of symbol per second

The mean value of $I(x_i)$ over the alphabet of source X with m different symbols is given by

$$H(x) = E[I(x_i)] = \sum_{i=1}^m P(x_i) I(x_i)$$

$$= - \sum_{i=1}^m P(x_i) \log_2 P(x_i) \text{ b/symbol}$$

The quantity $H(x)$ is called the entropy

The Sampling Theorem :-

A continuous time signal is first converted to discrete - time signal by sampling process. The statement of sampling theorem can be given in two parts as:-

i) A band-limited signal of finite energy which has no frequency component higher than F_m Hz is completely described by its sample values at uniform interval less than or equal to $\frac{1}{2F_m}$ second apart.

ii) A band limited signal of finite energy which has no frequency components higher than F_m Hz, may be completely recovered from the knowledge of its samples taken at the rate of $2F_m$ samples per second.

NYQUIST rate and NYQUIST interval

when the sampling rate becomes exactly equal to $2F_m$ samples per second the it is called Nyquist rate. Nyquist rate. Nyquist rate is also called the minimum sampling rate.

$$F_s = 2F_m$$

Similarly, maximum sampling interval is called Nyquist interval. It is given by -

$$\text{Nyquist interval } T_s = \frac{1}{2F_m} \text{ seconds.}$$

when the continuous-time band-limited signal is sampled at Nyquist rate ($F_s = 2F_m$), the sampled spectrum $G(\omega)$ contains non-overlapping $G(\omega)$ repeating periodically. But the successive cycles of $G(\omega)$ touch each other.

Therefore the original spectrum $X(\omega)$ can be recovered from the sampled spectrum by using a low pass filter a cut-off frequency ω_m .

Therefore the original spectrum $X(\omega)$ can be recovered from the sampled spectrum by using a low pass filter a cut-off frequency ω_m .

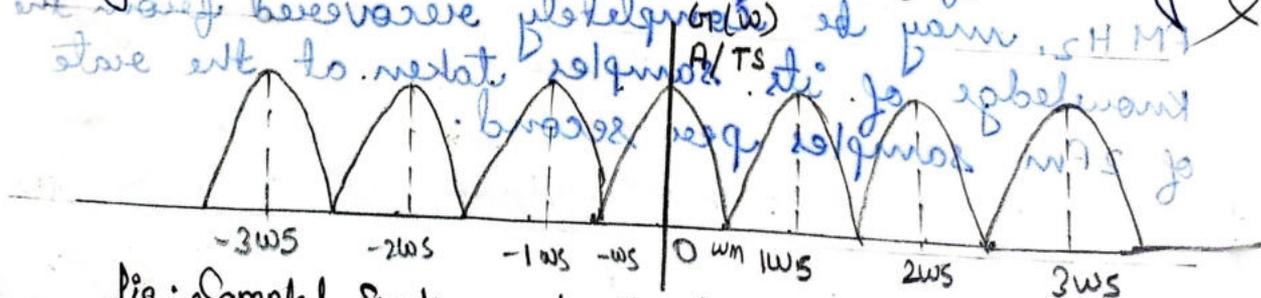


Fig: Sampled Spectrum at Nyquist rate

Effect of under Sample \rightarrow Aliasing

When a continuous - Time - band - limited signal is sampled at a rate lower than Nyquist rate $F_s < 2F_m$ then the successive cycles of the spectrum $G(\omega)$ of the sampled signal $g(t)$ overlap with each other as shown in fig.

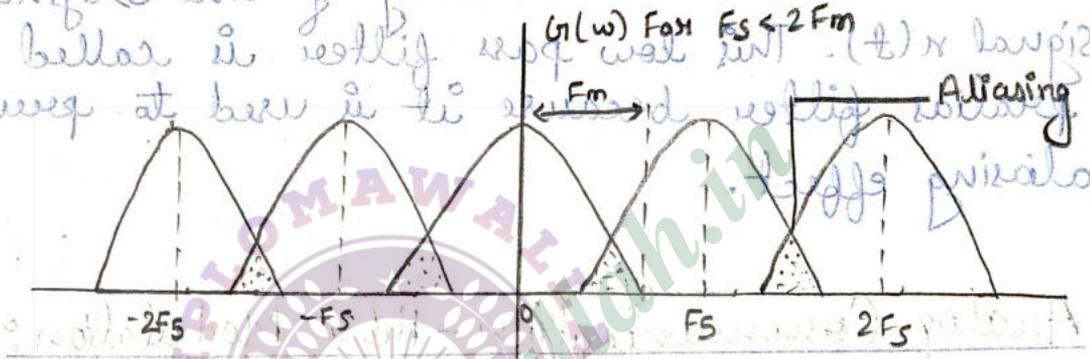


Fig: Sampl Spectrum of the sampled signal for the case $F_s < 2F_m$
 Hence, the signal is under sampled in this case ($F_s < 2F_m$) and some amount of aliasing is produced in this under sampling process. In fact, aliasing is the phenomenon in which a high frequency component in the frequency spectrum of the signal takes identity of a lower frequency component in the spectrum of the sampled signal.

It is clear that because of the overlap due to aliasing phenomenon it is not possible to recover original signal $x(t)$ from sampled signal $g(t)$ by low pass filtering.

Since any information signal contains a large number of frequencies so to decide a sampling frequency is always a problem. Therefore, a signal is first passed through a low pass filter. This low pass filter blocks all the frequencies which are above F_m Hz. This process is known as band limiting of the original signal $x(t)$. This low pass filter is called pre-filter because it is used to prevent aliasing effect.

Analog Communication - Pulse Modulation:-

After continuous wave modulation. In the next division is pulse modulation. In this chapter, let us discuss the following analog pulse modulation techniques

- Pulse Modulation
- Pulse width Modulation
- Pulse Position modulation.

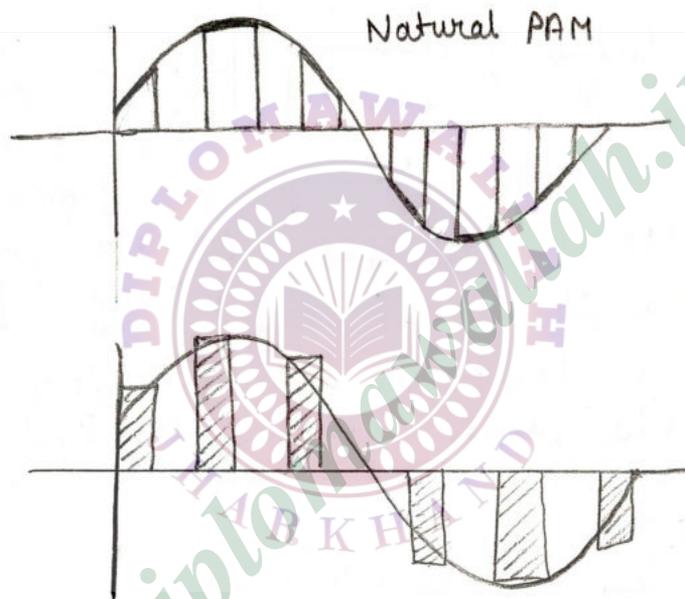
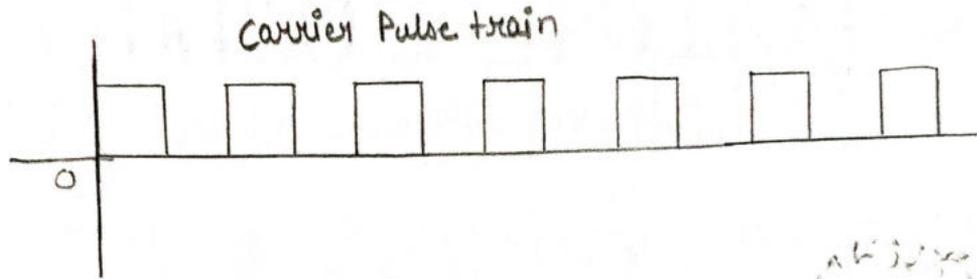
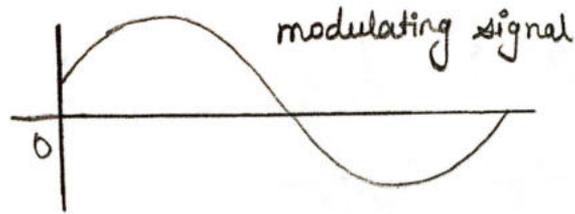
* Pulse Amplitude Modulation:-

In pulse amplitude modulation (PAM) technique the amplitude of the pulse carrier varies which is proportional to the instantaneous amplitude of the message signal.

The following figure explains the pulse amplitude modulation.

* Pul

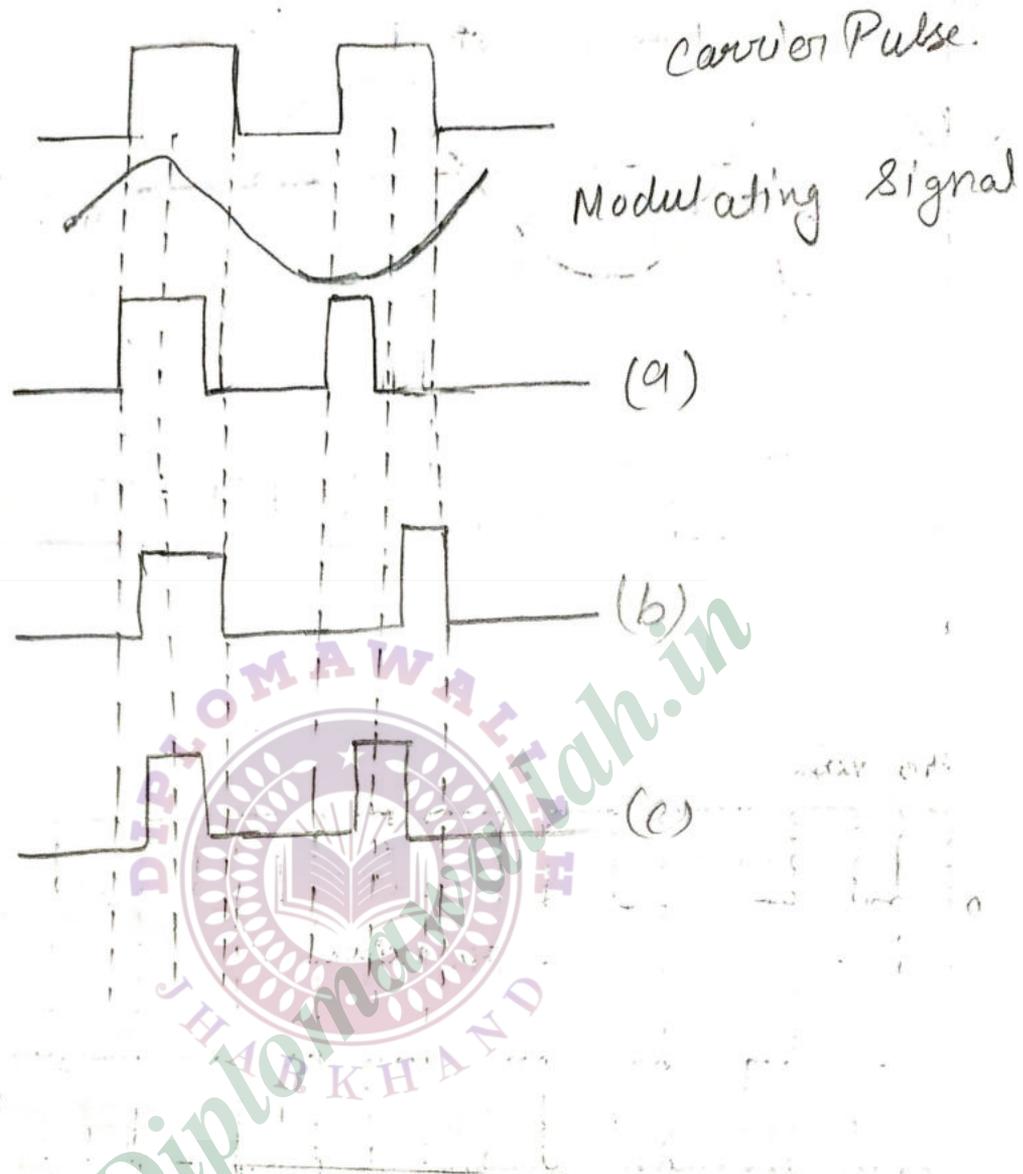
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* Pulse width Modulation is also called as Pulse duration modulation *

In pulse width modulation (PWM) or pulse duration modulation (PDM) or pulse time modulation (PTM) technique the width or the duration or the time of the pulse carrier varies, which is proportional to the instantaneous amplitude of the message signal.

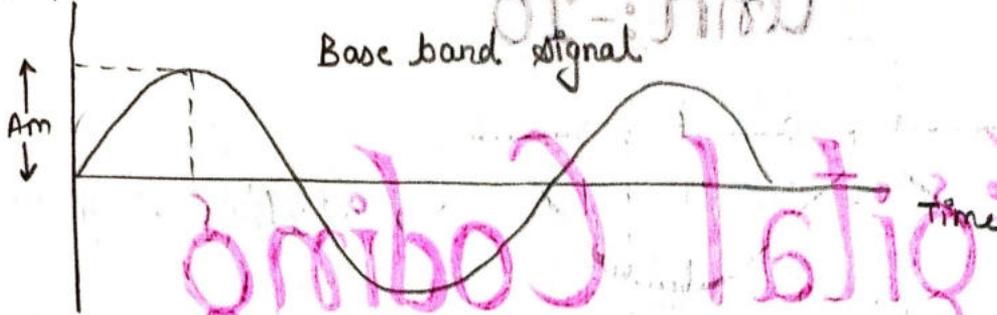
of the message signal.



* Pulse Position Modulation *

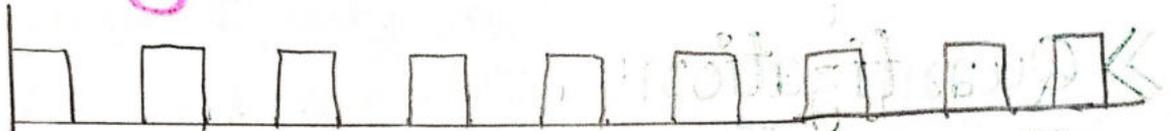
Pulse position modulation (PPM) is an analog modulation scheme in which the amplitude and the width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.

Amplitude



Digital Coding

Periodic Sequential pulse train



The digitization of analog signal involves the sampling of the signal at regular intervals and the conversion of the sampled values into a digital form. The process of converting an analog signal into a digital signal is called digitization.

Amplitude



PWM signal is a digital signal where the width of the pulse varies according to the amplitude of the original analog signal. This is called pulse width modulation.

0



PPM signal is a digital signal where the position of the pulse varies according to the amplitude of the original analog signal. This is called pulse position modulation.

Quantization

