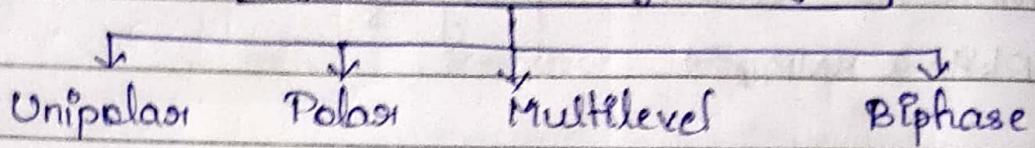


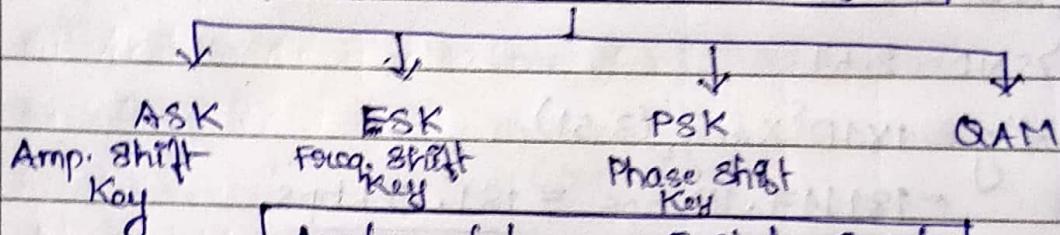
MODULE - 3

Signal Encoding Techniques

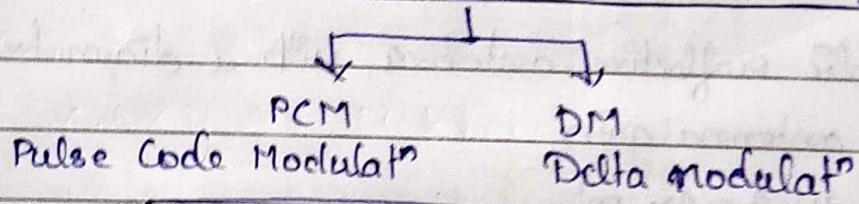
Digital data → Digital signal



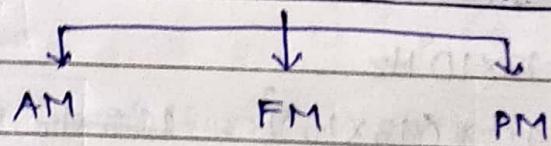
Digital data → Analog signal



Analog data → Digital signals



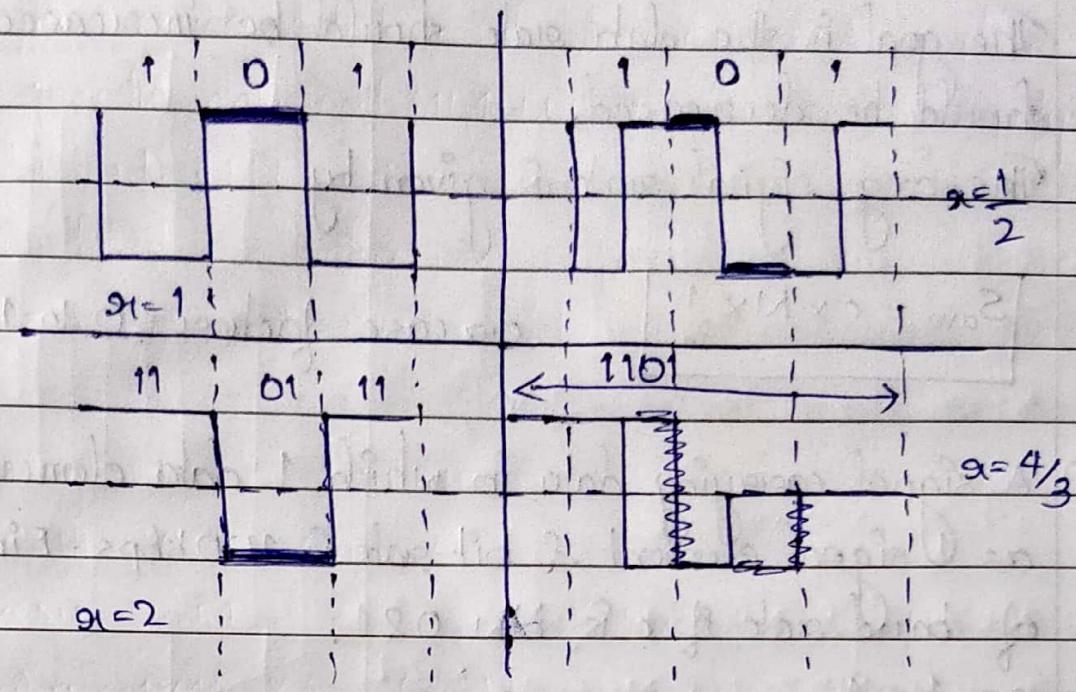
Analog data → Analog signals



Digital data → Digital signal

Line encoding

Block coding
Scrambling



Data rate, $s_1 = \frac{\text{Data Element}}{\text{Signal Element}}$

Data Element & Signal Element

It is the smallest entity of info & signal element is the shortest unit of digital signals. Data elements are carried by signal elements. The ratio s_1 specifies the no. of data elements carried by each signal element. Data rate is the no. of data elements sent in 1s, also known as bit rate (bps)

Signal rate is the no. of signal elements sent in 1s given in baud. It is also known as (pulse rate / modulation rate / baud rate)

The relationship b/w data rate (N) & signal rate (S) is given as

$$S = \frac{N}{s_1}$$

The goal is the data rate should be increased & signal rate should be decreased.

The avg. signal rate is given by

$$S_{\text{avg}} = c \times N \times \frac{1}{g_1}$$

$c \rightarrow$ case factor = 0 to 1

- ① A signal carrying data in which 1 data element is encoded as 1 signal element & bit rate is 100 kbps. Find avg value of baud rate if c is bin 021.

ans $g_1 = \frac{1}{1} = 1$

$$N = 100 \text{ kbps} = 1 \times 10^5 \text{ bps}$$

$$c = 0.5$$

$$\underline{S_{\text{avg}} = 0.5 \times 10^5 \times 1 = 50 \text{ K baud}}$$

$$B_{\min} = c \times N \times \frac{1}{g_1} \rightarrow \text{Min. bandwidth}$$

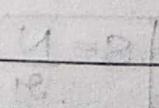
$$N_{\max} = B \times g_1 \times \frac{1}{c} \rightarrow \text{Max. data rate}$$

Baseline wandering (Problem)

DC Component (Problem)

Self-Synchronization ("")

Error correct



Baseline Wandering

When decoding a digital signal, the receiver calculates a running avg of the received signal power which is given as baseline. The incoming signal power is evaluated against the baseline to determine the value of data element. A long string of 0's or 1's cause a drift in baseline called as baseline wandering.

DC Component Problem

The digital signal having const. voltage level present problems that cannot pass the low frequencies & those freq.s around 0 are called DC components.

e.g. Telephone lines less than 200Hz cannot pass the freq.s
Hence avoid DC component problem.

Self - Synchronization

To correctly interpret the sender's bit intervals to receiver's bit intervals, a self-synchronizing signal can be used. It includes timing info in the signal that is being transmitted.

Built-in Error Detectn & Correctn

They can be used.

Line Coding

Converting digital data into digital signal

(i) Unipolar NRZ

It is called as NRZ because it does not return to 0 at the middle of the bit.

+ve. voltage $\rightarrow 1$, ~~0~~ voltage $\rightarrow 0$.

Disadvantages

- (i) Costly
- (ii) DC component problem
- (iii) Self-sync problem
- (iv) Baseline wandering

$$\text{Normalized Power} = \frac{1}{2} \times V^2$$

hence $V \rightarrow \text{voltage}$

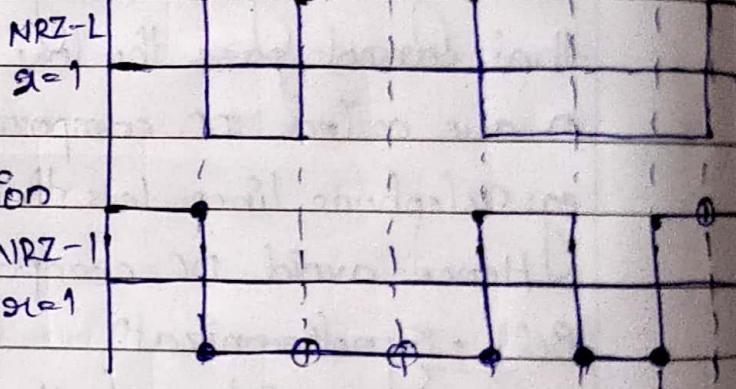
- (v) NRZ-L & NRZ-I

Polar NRZ-L & NRZ-I
NRZ-I ↓
NRZ-L Level of voltage

If next bit is 0 \rightarrow No inversion

" " " 1 \rightarrow Inversion NRZ-I

$$S_{ave} = \frac{N}{2}$$

Disadvantages

- (i) Baseline wandering problem is more in NRZ-L than in NRZ-I
- (ii) Synchronization problem
- (iii) DC component "
- (iv) Change in polarity causes errors

NRZ-L \rightarrow Level of voltage determines value of bit

$$S_{ave} = \frac{N}{2} \quad g=1$$

NRZ-I \rightarrow Change or lack of change determines value of bit

No inversion if next bit is 0.

(iii) RZ

Return to Zero (Polar)

3 levels \rightarrow 0, +ve & -ve

Save = N



Signal goes to 0 in the middle of the bit.

Advantages

(i) No DC component problem

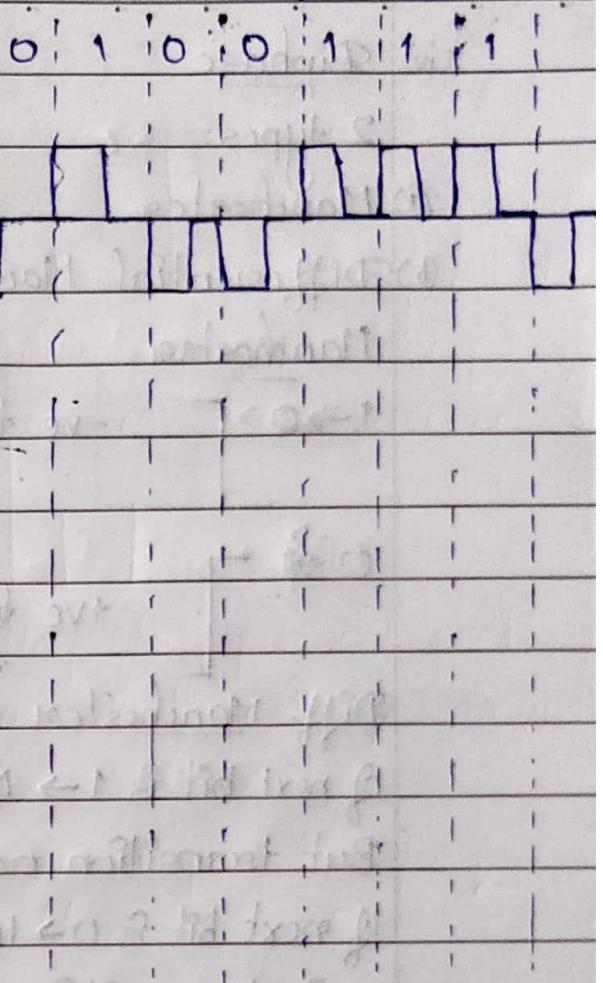
(ii) " synchronization "

Disadvantages

(i) Sudden change in polarity may cause 1 to be read as 0 or vice-versa.

(ii) It requires greater bandwidth since 2 signals are needed to encode a bit

(iii) It is more complex since it has 3 levels of voltage.

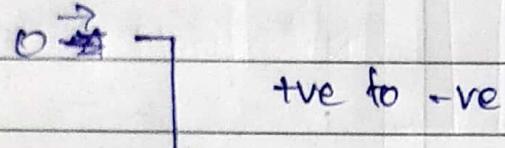
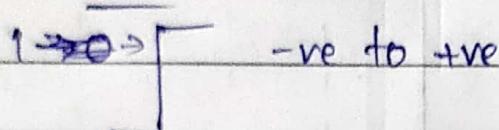


(iv) Biphase

2 types:

(a) Manchester

(b) Differential Manchester

ManchesterDiff. Manchester

If next bit is 1 → No inversion.

But transition occurs at middle

If next bit is 0 → Inversion. But
again transition occurs at middle(v) Bipolar

2 types:

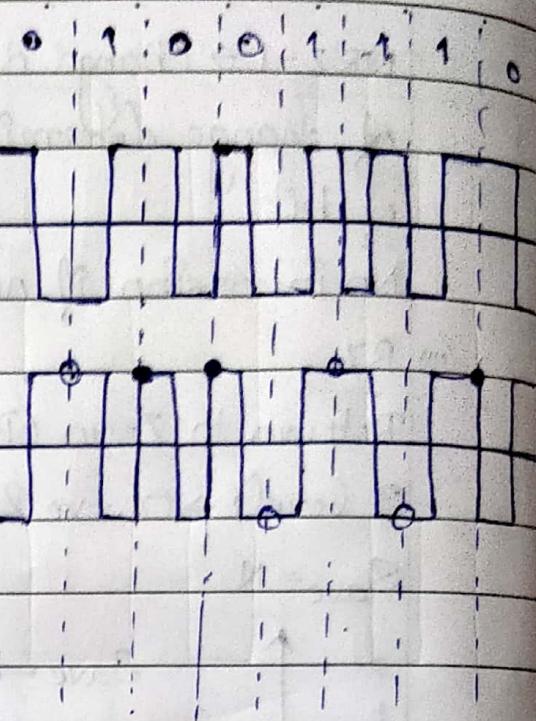
(a) Alternating Mark Inversion (AMI)

(b) Pseudo ternary AMI

AMI has 3 voltages → +ve, -ve & 0

It has 1 data element as 0 & mark denotes 1 which
alternates b/w +ve & -vePseudo ternary

Opposite of AMI

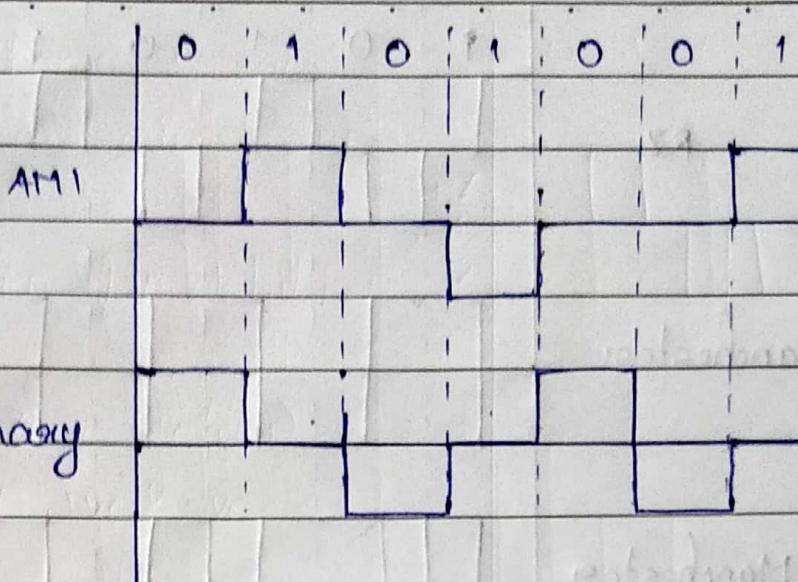
ManchesterAdv. of Manchester

No baseline wandering

& DC comp. problem

Disadv

Signal state is high



Advantage

No DC comp. problem because long sequence of 0's & 1's alternates b/w +ve & -ve

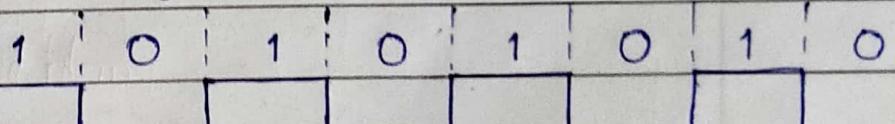
Disadvantage

Synchronization problem

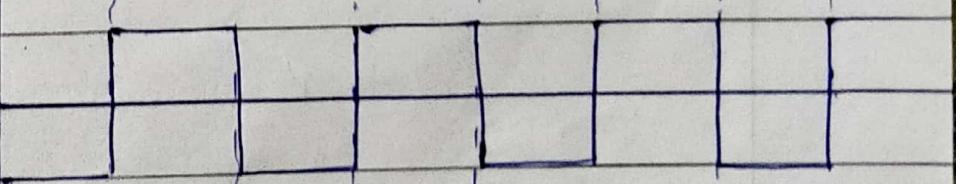
- ① Encode bit stream 10101010 into line coding schemes assuming last level has negative.

ans:

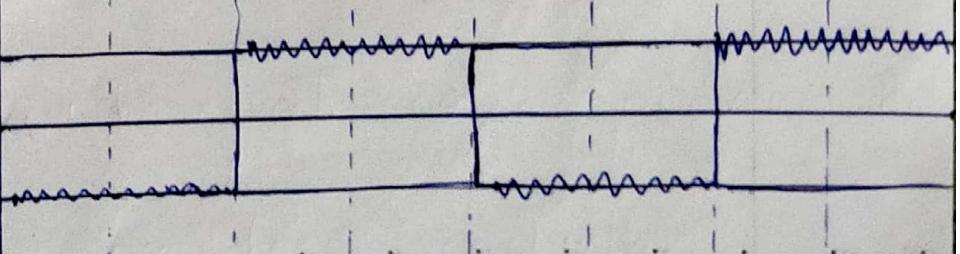
Unipolar NRZ



NRZ-L



NRZ-I



RZ

Manchester

Diff Manchester

AMI

Pseudoternary

14/10/19

Bipolar C/CS

It is a better alternative to NRZ

Multilevel Schemes

- In these schemes we ↑ the no. of data bits /symbol thereby ↑ing the bit rate
- Since it is binary data we only have 2 types of data elements 0 1 0 0 1 0
- We can combine 2 data elements into pattern of "m" elements to create " 2^m " symbols

- If we have L signal levels, we can use " n " signal elements to create L^n signal elements.

Code C/Cs

- If $2^m > L^n$ then we cannot represent data elements, we don't have enough signals.
- If $2^m = L^n$ then we have an exact mapping of 1 symbol on 1 signal.
- If $2^m < L^n$ then we have more signals than symbols & we can choose the signals that are more distinct to represent symbols & therefore have better noise immunity & error detection.

Note:- In mBnL schemes, a pattern of m data elements is encoded as a pattern of n signal elements in which $2^m \leq L^n$.

- $L=2$ binary, $L=3$ ternary, $L=4$ quaternary.
- Multilevel using Multiple channels
- In some cases, we split the signal txm up & distribute it over several links.
- The separate segments are transmitted simultaneously. This reduces signalling rate per link \rightarrow lower bandwidth.

$XD - YYYZ \rightarrow$ No. of levels
 \downarrow
 X division Type of modulatⁿ
(No. of nPACs (splts))

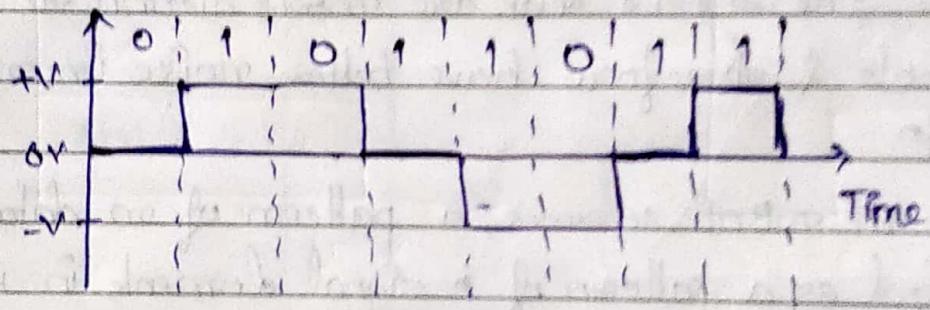
eg: 4D - PAM5
 \downarrow
4 nPACs Pulse amp.
 \uparrow mod.
5 levels

Multilevel Coding

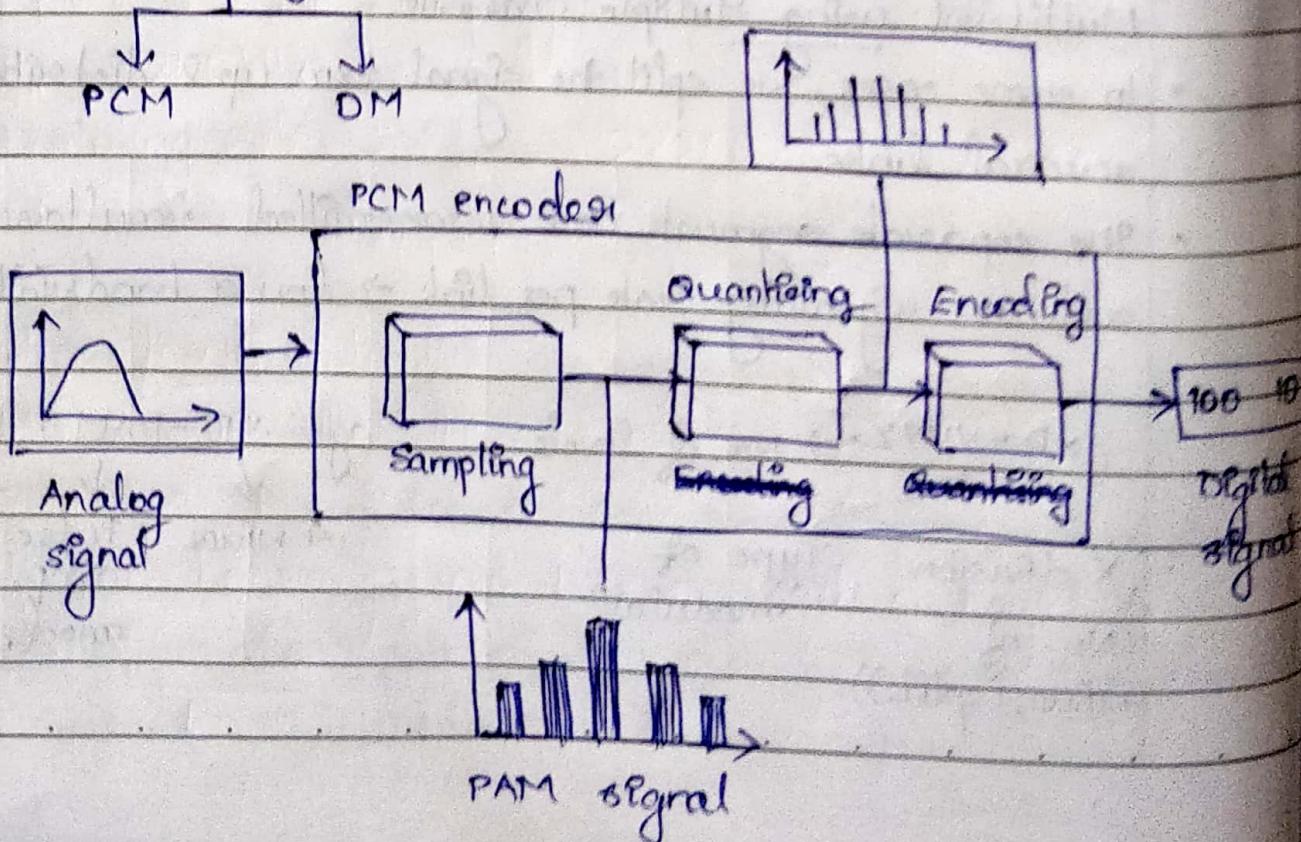
MLT-3 scheme

In multilevel transmit, rules are:

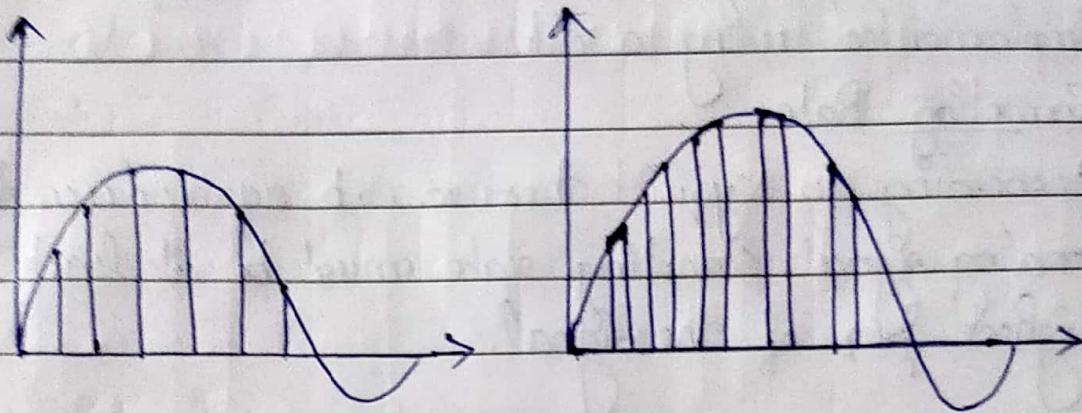
- If next bit is 0, then no transition
- " " " " 1 & current level is not 0, then next level 0
- If next bit is 1, & current level is 0, then next level is opposite of last non-zero level



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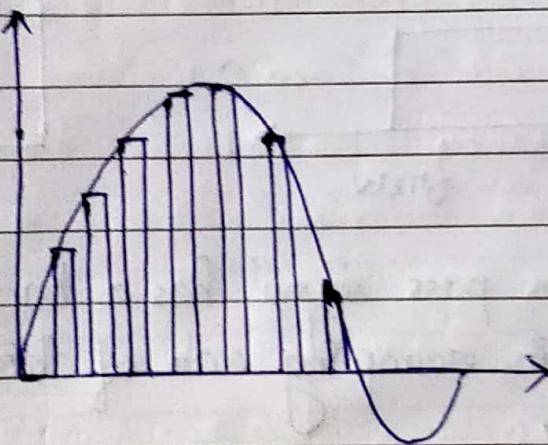
Analog to Digital ~~PCM~~

Sampling
~~ideal~~



Ideal

Natural



Flat top sampling

Analog signal is sampled at every T_s second. It is known as sampling period.

$$\text{Sampling freq. } f_s = \frac{1}{T_s}$$

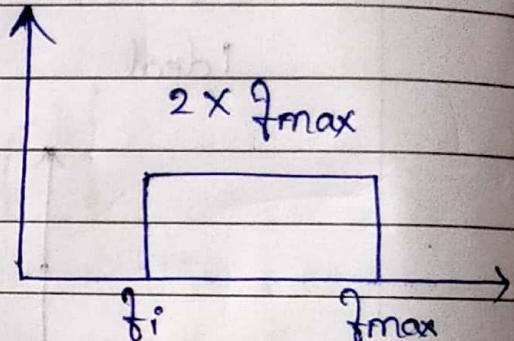
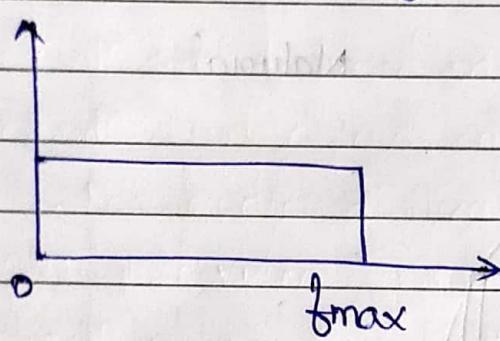
In ideal sampling, the pulses are sampled which cannot be implemented in real time. In natural sampling, the high speed switch is turned on for small period when sampling.

occurs. It retains shape of signal.

Flat top sampling is also known as sample & hold, which is implemented using a ch.

Sampling Rate

According to Nyquist theorem, to reproduce the original analog signal, sampling rate must be at least 2x the highest freq. of the signal.



- ① A complex low pass signal has a bandwidth of 200 kHz. What is the min. sampling rate of this signal?

ans: $f_{\text{max}} = 200 \text{ kHz}$

Sampling rate = $2 \times f_{\text{max}} = 2 \times 200 \times 1000 \text{ Hz} = \underline{\underline{400000}}$

= 400000 samples/sec

Steps for Quantisation

- (i) We assume that the original analog signal has amplitude b/w V_{max} & V_{min} .

- (ii) Divide the range into L zones, each of height Δ

$$\Delta = \frac{V_{\text{max}} - V_{\text{min}}}{L}$$

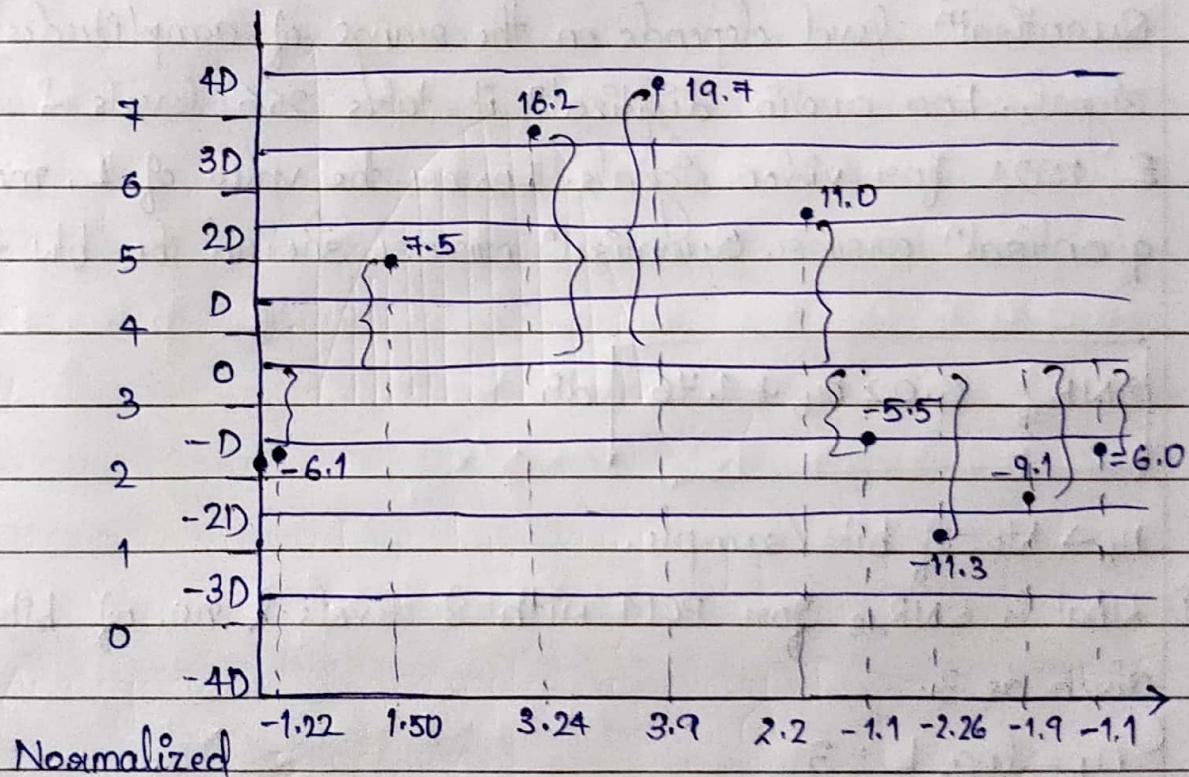
(iii) Assign quantised values of $0 \dots (L-1)$ to the midpoint of each zone

(iv) Approximate the values of sampled amplitude to quantised values.

e.g. A sampled signal has amplitude b/w $-20V$ to $+20V$ & has 8 levels

$$L=8$$

$$\Delta = \frac{20 - (-20)}{8} = \frac{40}{8} = 5$$



PAM values (distance of points above & below 0)

Normalized $-1.5 \quad 1.5 \quad 3.5 \quad 3.5 \quad 2.5 \quad -1.5 \quad -2.5 \quad -1.5 \quad -1.5$
quantized values

Normalized $-0.28 \quad 0 \quad 0.26 \quad -0.4 \quad 0.3 \quad -0.4 \quad -0.24 \quad 0.4 \quad -0.4$

Encoding 2 5 7 7 6 8 1 2 2

Digital signal \rightarrow Binary representations of quantised values

2 5 7 7 6 2 1 2 2

01010111111110010001010010

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Quantisation Errors

Quantisation process selects the quantisation values from middle of each zone.

Normalised residue = Normalised quantised value - Normalised amplitude value

Quantisation code is based on quantisation levels.

Quantisation level depends on the range of amplitudes of analog signal. For audio digitization it takes 256 levels & around 1024 for video signals. Lower the value of L, more is the quantisation error. Quantisation errors should be b/w $+\frac{\Delta}{2}$ & $-\frac{\Delta}{2}$

$$\text{SNR}_{\text{dB}} = 6.02 n_b + 1.76 \quad \boxed{\text{dB}}$$

$n_b \rightarrow$ No. of bits/sample.

- ① What is SNR_{dB} for PCM with 8 levels & no. of bits/sample given as 3.

ans $n_b = \log_2 8 = 3$

$$\text{SNR}_{\text{dB}} = 6.02 \times 3 + 1.76 = \underline{\underline{19.82 \text{ dB}}}$$

Uniform vs Non-Uniform Quantisation

If the distribution of amplitude in analog signal is not uniform then the height of the signal Δ is not fixed.

Data is greater at the lowest amplitudes & lesser near the highest amplitudes.

Non-uniform quantisation can be produced by using a process called companding i.e. the signal is reduced for larger values at the ~~sender~~ & expanded at the receiver after conversion. Companding gives greater weight to strong signal & less weight to noise.

Encoding

After the sample is quantised n_b to be encoded is determined from quantisation level L .

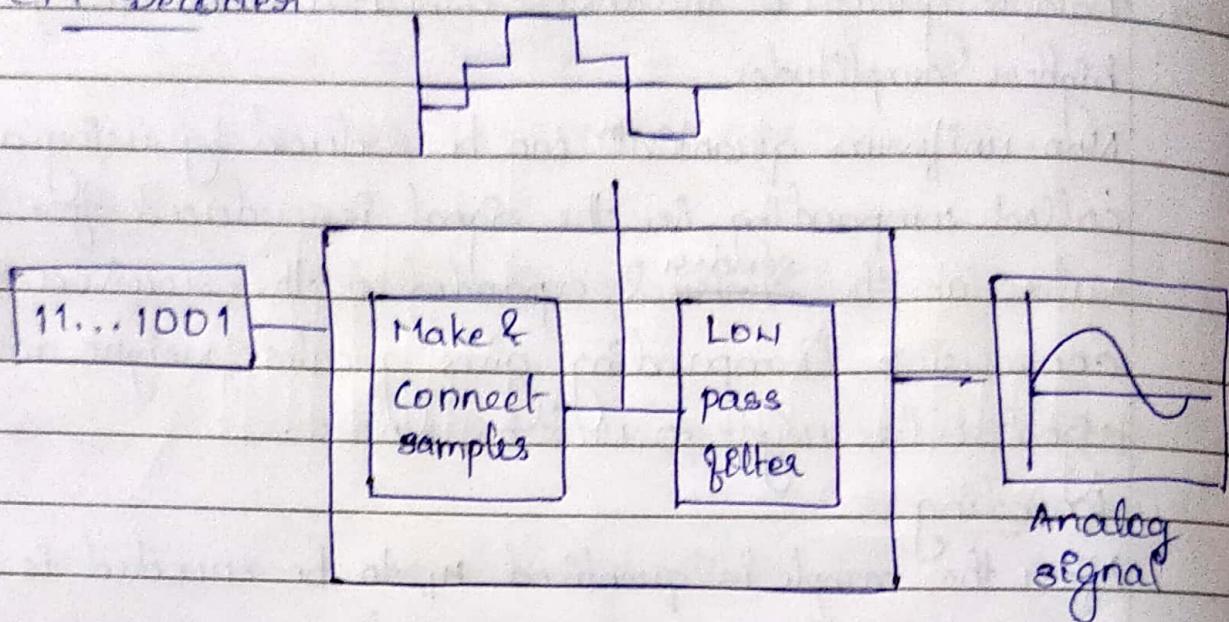
$$n_b = \log_2 L$$

$$\begin{aligned}\text{Bit rate} &= \text{Sampling rate} \times \text{No. of bits per sample} \\ &= f_s \times n_b\end{aligned}$$

① We want to digitize human voice. What is the bit rate, assuming 8 bits per sample? (f_{max} range $\rightarrow 0 - 4000 \text{ Hz}$)

ans. Sampling rate, $f_s = 2 \times f_{\text{max}} = 2 \times 4000 = 8000 \text{ Hz}$

$$\begin{aligned}\text{Bit rate} &= \text{Sampling rate} \times \text{No. of bits per sample} \\ &= 8000 \times 8 \\ &= 64000 \text{ bps} \\ &= \underline{\underline{64 \text{ kbps}}}\end{aligned}$$

PCM Decoder

Decoder uses circuitry to convert codewords to pulse by holding the amplitude until the next pulse. The resulting staircase signal is passed through low pass filter to smooth it out into the analog signal. The filter has same cut-off freq as the sender.

$$\text{Min. Bandwidth, } B_{\min} = c \times N \times \frac{1}{s_1}$$

$$= c \times n_b \times f_s \times \frac{1}{s_1}$$

$$= c \times n_b \times 2 \times B_{\text{analog}} \times \frac{1}{s_1}$$

$$\text{If } c = \frac{1}{2} \text{ & } s_1 = 1,$$

$$B_{\min} = n_b \times B_{\text{analog}}$$

① We have low pass analog signal of 4kHz. If signal is digitized with 8 bits per sample, what is the min. bandwidth required?

ans $n_b = 8 \text{ bits/sample}$

$$B_{\text{analog}} = 4 \text{ kHz} = 4000 \text{ Hz}$$

$$B_{\min} = n_b \times B_{\text{analog}}$$

$$= 8 \times 4000$$

$$= 32 \text{ kHz}$$

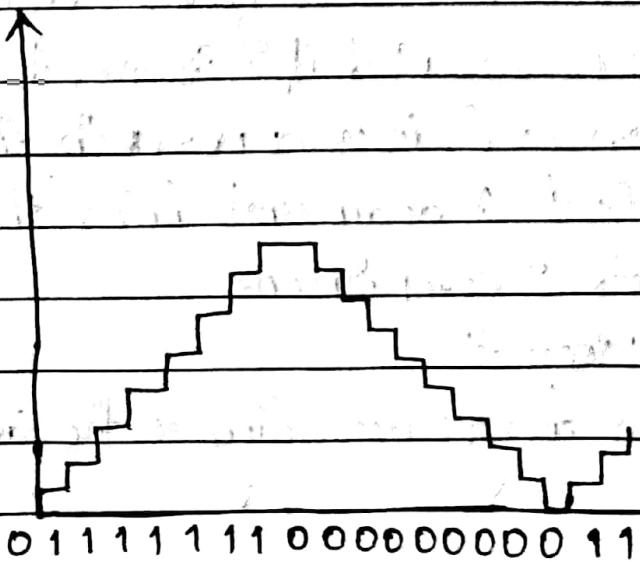
$$N_{\max} = 2 \times B \times \log_2 L \text{ bps}$$

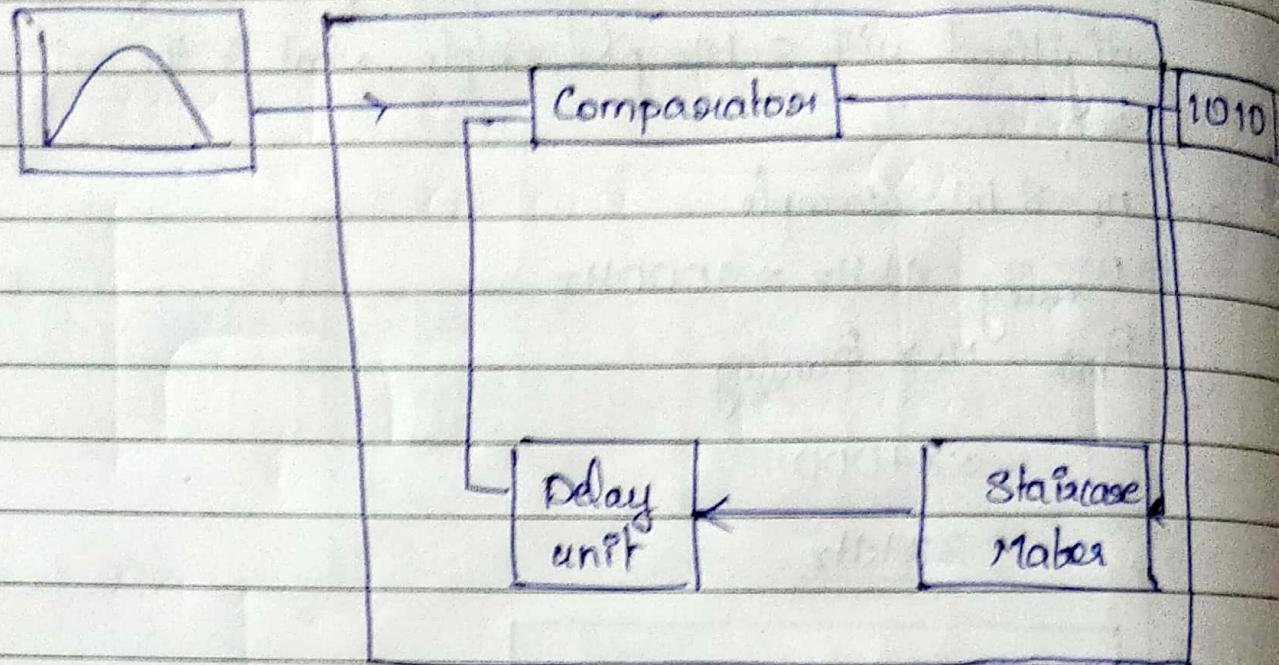
$$B_{\min} = N$$

$$2 \times \log_2 L$$

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Delta Modulation





To reduce the complexity of PCM, delta modulation is used. It finds the change from the prev. sample. There are no codewords here. The bits are sent one after the other. At the sender site, a stream of bits from an analog signal encodes the small +ve or -ve changes called as delta. If delta is +ve, it encodes a 1 & if it is -ve, it encodes a 0. At each sampling interval, it compares with the last value of the staircase signal. A delay unit holds the staircase func for a period b/w 2 comparisons.

Adaptive Delta Modulator

The value of delta changes according to the amplitude of the analog signal.

Digital to Analog Conversion

↓ ↓ ↓
ASK FSK PSK

- ① An analog signal has a bit rate of 8000 bps & a baud rate of 1000 bauds. How many data elements are carried by each signal element? How many signal elements do we need?

ans. $S = 1000 \text{ bauds}$

$$N = 8000 \text{ bps}$$

$$g_1 = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$C = \underline{\underline{1}}$$

$$\Rightarrow g_1 = \log_2 L$$

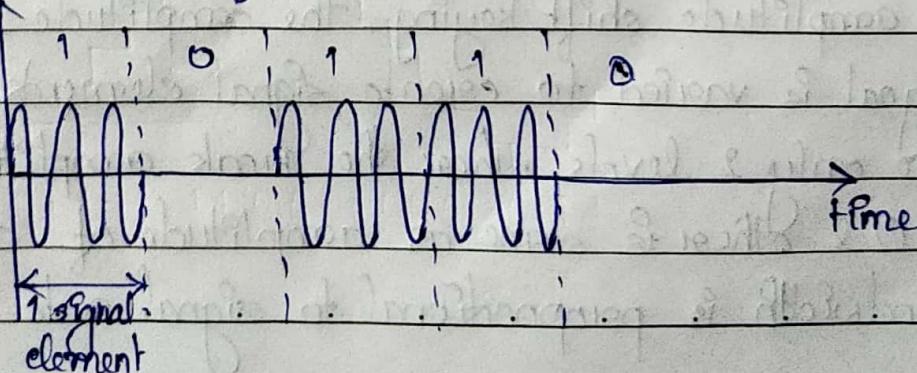
$$\Rightarrow L = 2^{g_1} = 2^8 = \underline{\underline{256}} \text{ levels}$$

Carrier Signal

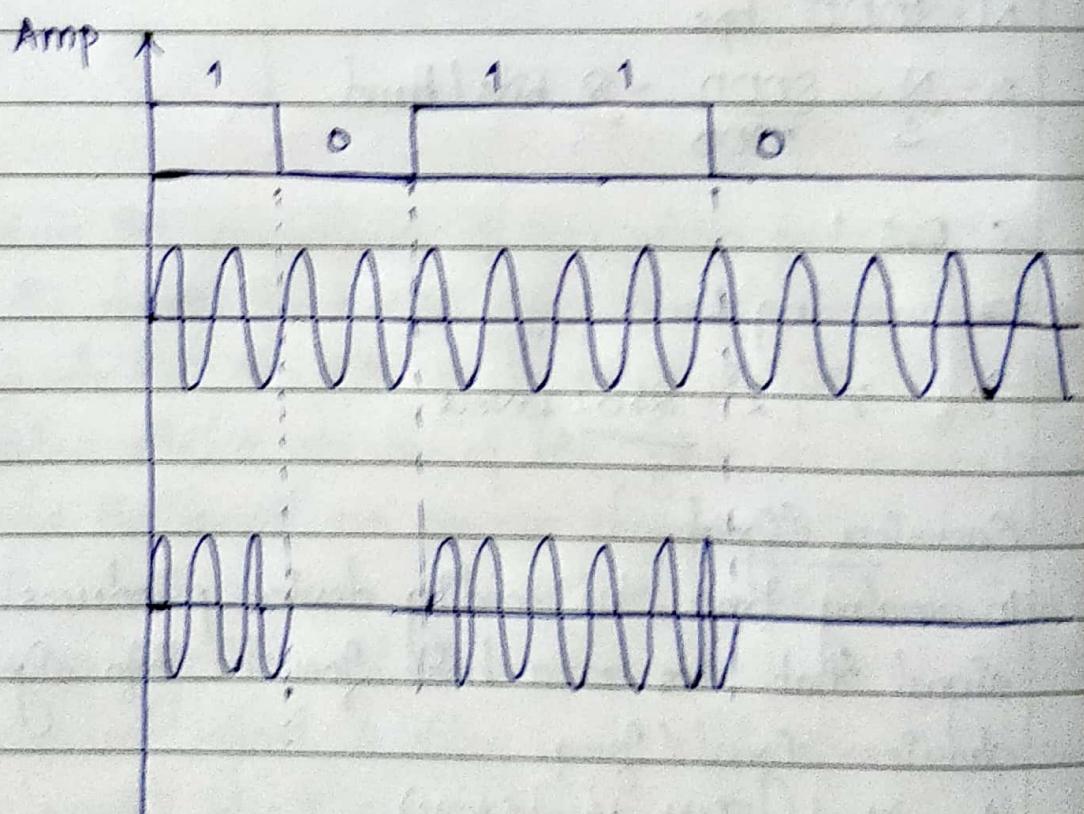
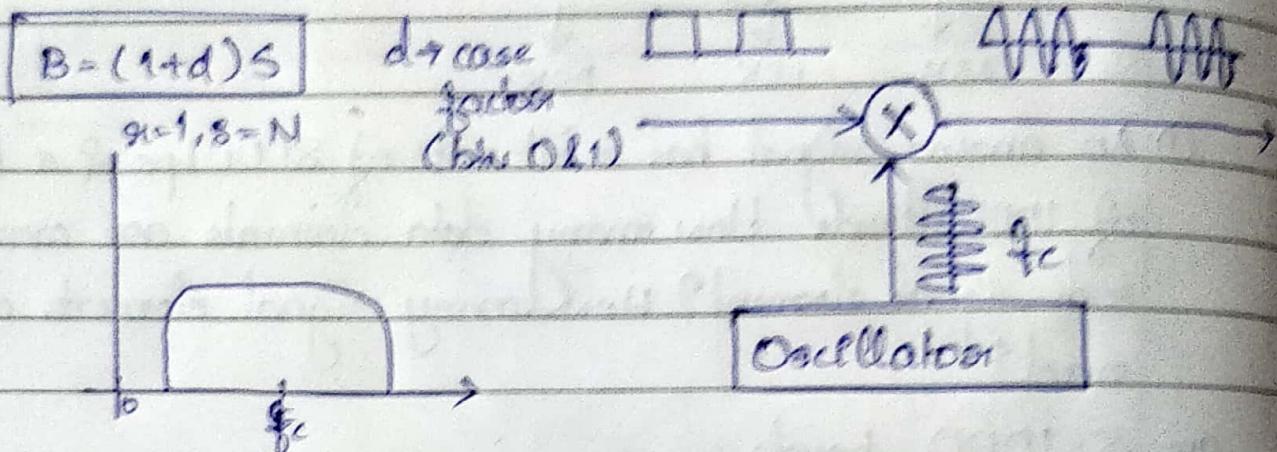
In analog form, the sending device produces a high freq. signal that acts as a base for the info signal called as carrier signal / freq.

Amplitude Shift Key (ASK)

Amp.



BASK \rightarrow Binary ASK or ON-OFF Keying



In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. In BASK there are only 2 levels where the peak amplitude of 1 signal is 0.2 other is same as amplitude of carrier freq. The bandwidth is proportional to signal rate.

$$B = (1+d)S$$

where d lies b/w 0 & 1

The middle of the bandwidth is the carrier freq.

- ① Available bandwidth of 100 kHz from 200 to 300 kHz.

What is the carrier freq. & bit rate if $d=1$ when data is modulated by ASK?

ans. $B = 100 \text{ kHz}$

$$d = 1$$

$$s_c = 1$$

~~$$1 \times 10^5 = 2 \times s$$~~

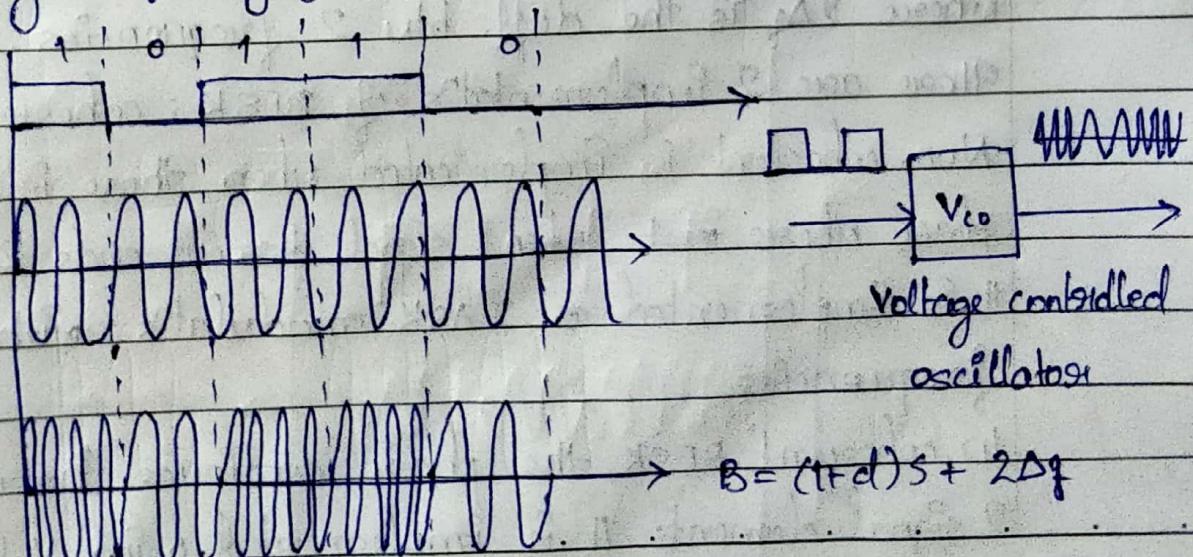
$$\Rightarrow S = 5 \times 10^4 = \underline{\underline{50 \text{ baud}}}$$

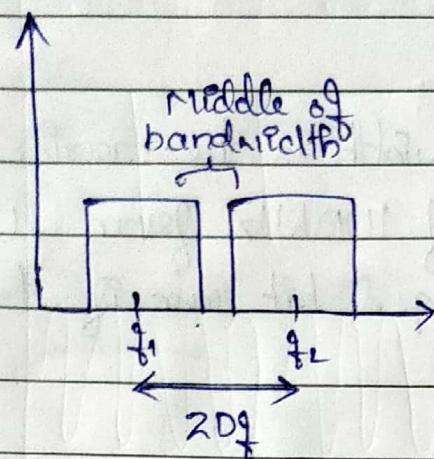
~~$$S = c \times N \times \frac{1}{T}$$~~

$$\Rightarrow N = S \times T \times \frac{1}{c} = 50 \times 1 \times \frac{1}{1} = \underline{\underline{50 \text{ kbps}}}$$

$$f_c = \underline{\underline{250 \text{ kHz}}} \text{ (Middle of bandwidth)}$$

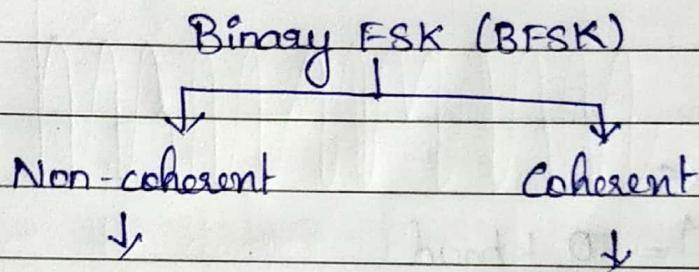
Frequency Shift Keying (FSK)





$d \rightarrow$ case factor

$\Delta f \rightarrow$ Diff b/w 2 frequencies
 $f_1 \& f_2$



2 ASK signals are combined

V_{co}

In FSK, the freq of the carrier signal is varied to represent the data. The freq of the modulated signal is const for duration of 1 signal element, but changes for the next signal element. Amp. & phase remain const. FSK bandwidth is calculated by $B = (1+d)S + 2\Delta f$

where $2\Delta f$ is the diff. b/w 2 frequencies.

There are 2 implementations of BFSK: coherent & non-coherent. Non-coherent is implemented when there is discontinuity in phase where ~~is~~ when 1 signal element ends & the next begins. It is implemented as 2 ASK modulations with 2 carrier frequencies.

In coherent BFSK, the phase continues through boundary of 2 signal elements. It is implemented using V_{co} which changes

its freq. according to s/p voltage... In case of multilevel FSK,

$$B = (1+d)S + (1-d)2\Delta_f$$

- ① Same problem ① in ASK. What should be the carrier freq. & bit rate if we modulate using FSK with $d=1$
ans. $B = 100 \text{ kHz} \cdot 1 \times 10^5 \text{ Hz}$

$$d=1$$

$$2\Delta_f = 275 - 225 = 50 \text{ kHz} \\ = 5 \times 10^4 \text{ Hz}$$

$$B = (1+d)S + 2\Delta_f$$

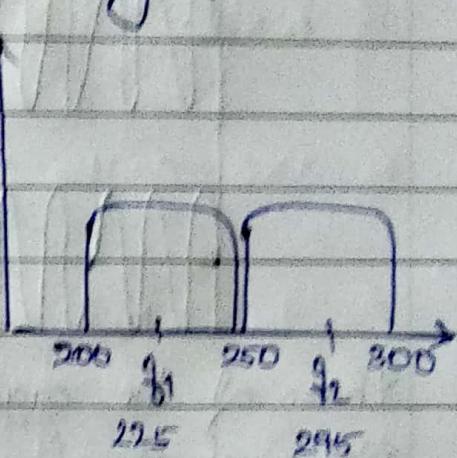
$$10^5 = 2S + 5 \times 10^4$$

$$\Rightarrow 2S = 10^5 - 5 \times 10^4$$

$$\Rightarrow 2S = 5 \times 10^4$$

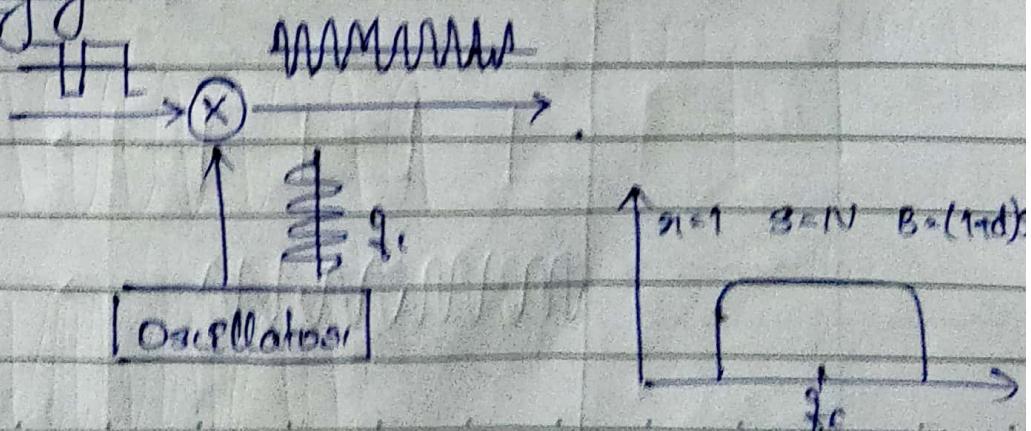
$$\Rightarrow S = 2.5 \times 10^4$$

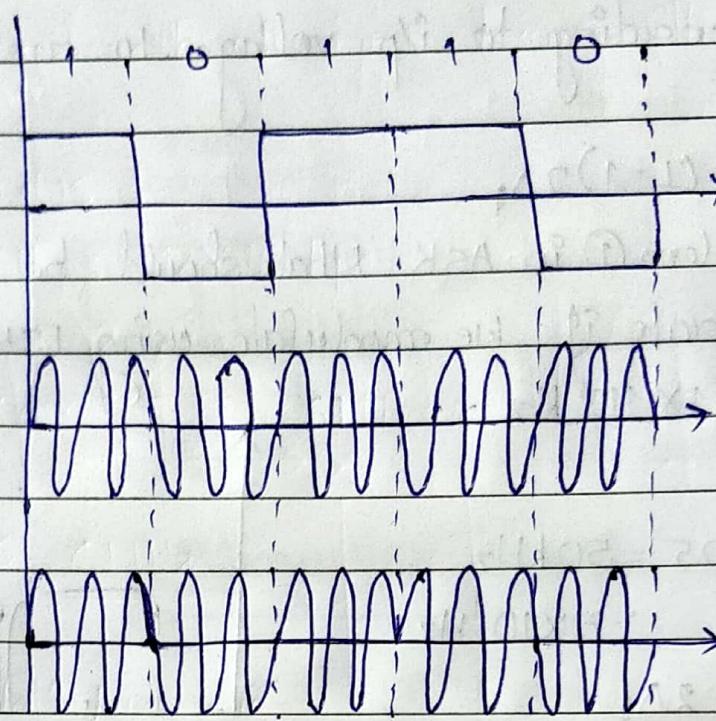
$$\Rightarrow \underline{S = 25 \text{ kbaud}}$$



~~$S = N$~~ $\Rightarrow N = s \times S = 1 \times 25 = \underline{\underline{25 \text{ kbps}}}$

Phase Shift Keying (PSK)

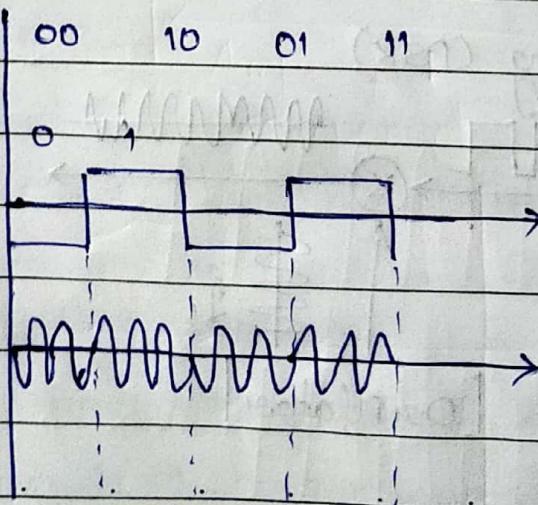


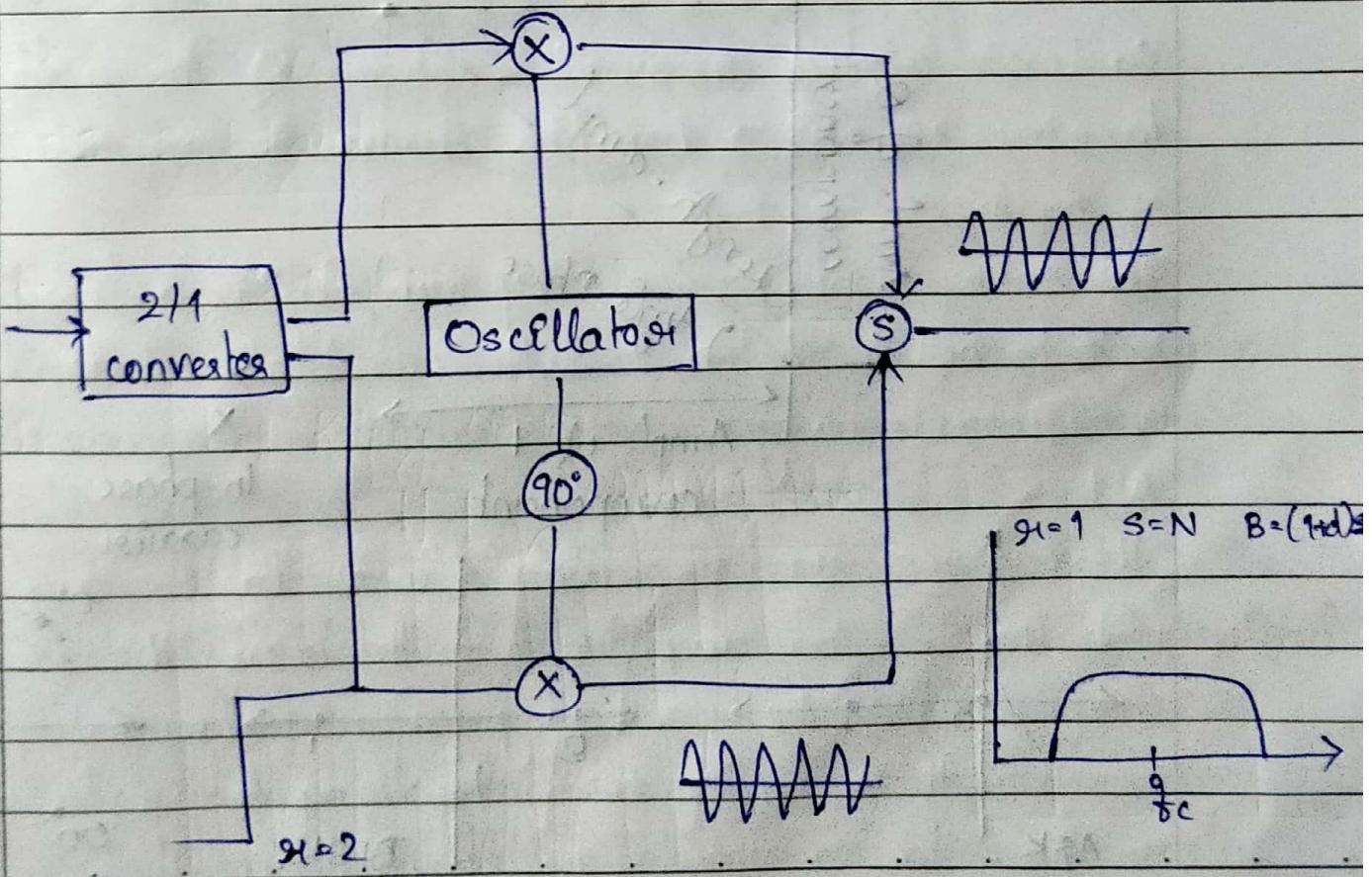
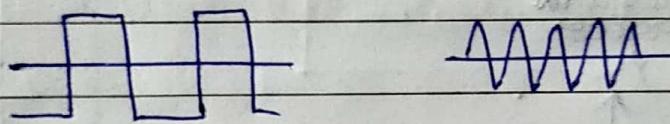
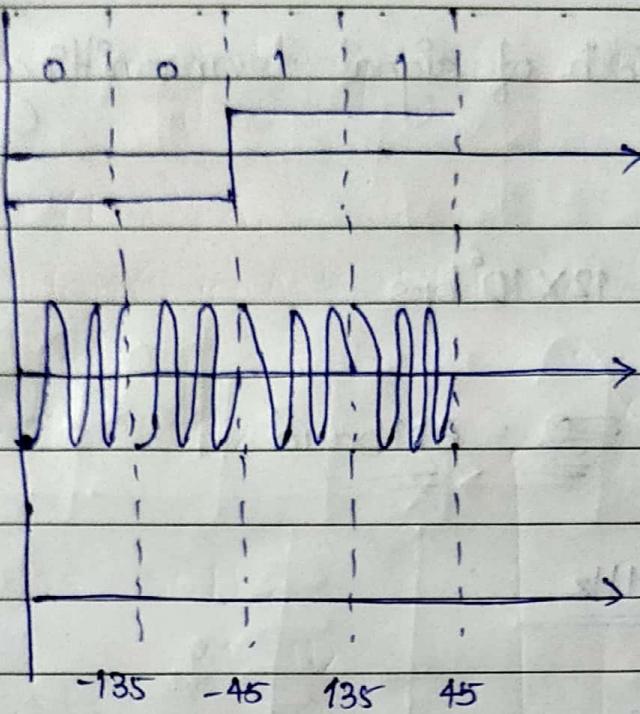


Phase of the carrier signal is varied to represent 2 or more different signal elements. Peak amp. & freq. remain const. In BPSK, the 2 signal elements have a phase of 0° & other 180° .

BPSK is superior to ASK since it is less susceptible to noise. It is superior to BFSK since it doesn't require 2 carrier signals. But it requires complex technique.

QPSK





Q) Find bandwidth of signal transmitting 12 Mbps for QPSK
 $d=0$

ans- $B = (1+d)S$

$$N = 12 \text{ Mbps} = 12 \times 10^6 \text{ bps}$$

$$d=0 \quad g_1=2$$

$$S = N \times \frac{1}{g_1} = 6 \text{ Mbps}$$

$$B = 1 \times 6 = \underline{\underline{6 \text{ MHz}}}$$

29/10/19

Constellat'n Diagram

Quadrature carrier

y

Amp. of Q component

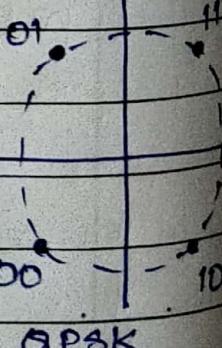
length, amp

Angle, phase

Amp. of I component

In-phase carrier

01

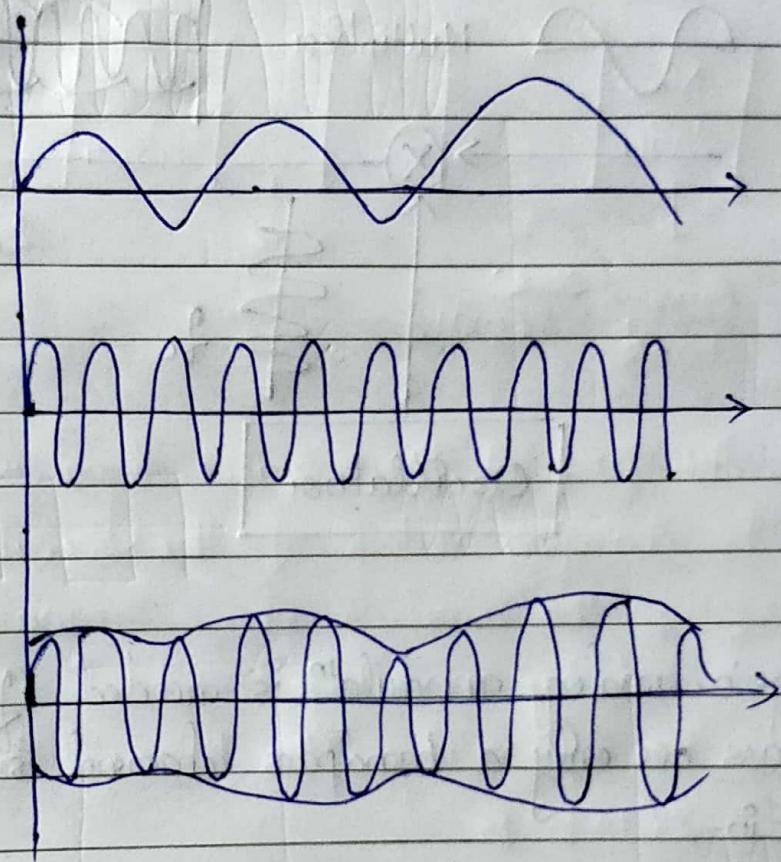
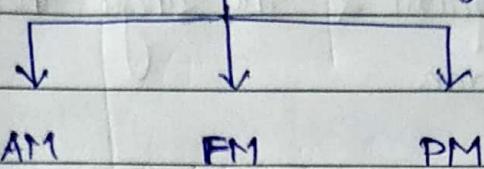


ASK

BPSK

QPSK

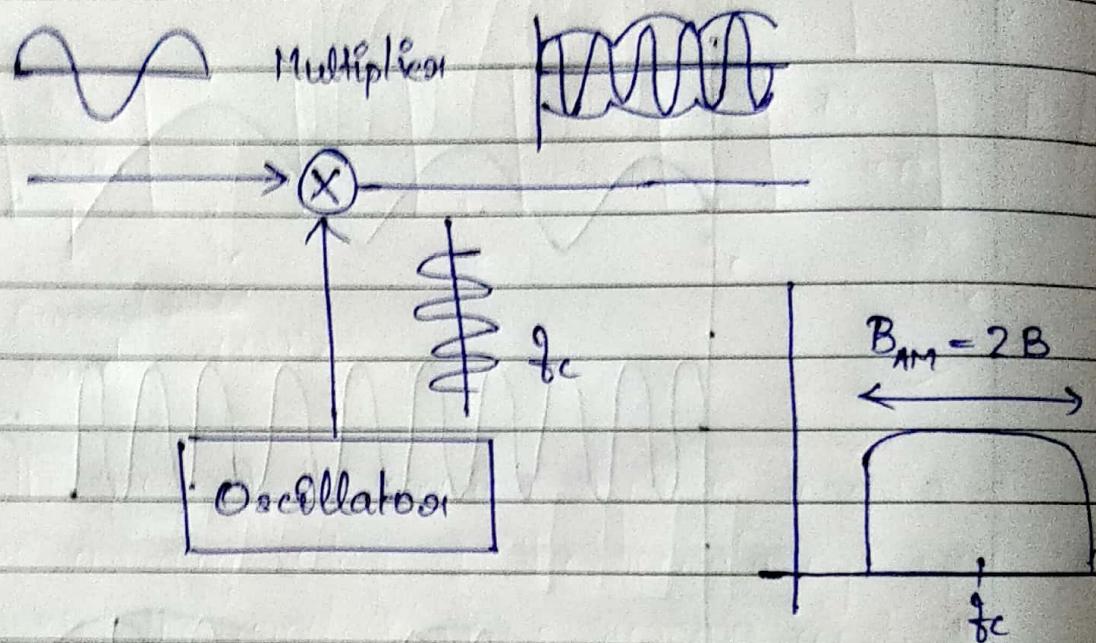
Analog Data to Analog Signal



Constellation diagram defines the ampl & phase of a signal element when 2 carriers are used. It is used for dealing with multilevel ASK or PSK or QAM.

A signal element type is represented as a dot (•). The x-axis is related to in-phase carrier & y-axis is related to quadrature carrier. The projectn of the point on x-axis defines the peak ampl. of in-phase component & projectn on y-axis defines peak ampl. of quadrature .

component. The length of the point to the origin is the ampl. of the signal element & the angle that the line makes with the x-axis is the phase.



Analog-to-analog "modulator" is needed if the medium is bandpass or only a bandpass channel is available.
e.g. Radios

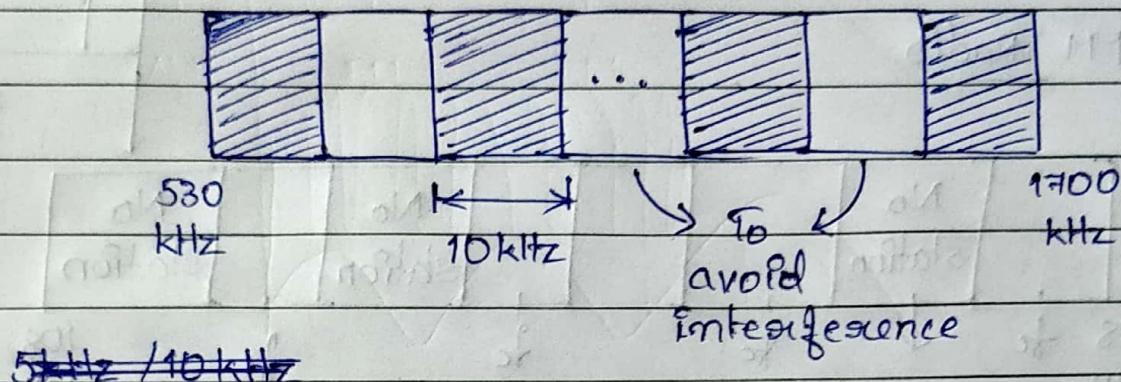
The analog signal produced by each statn is a low-pass signal & to be able to listen to different statns, it is shifted to a different range.

AM

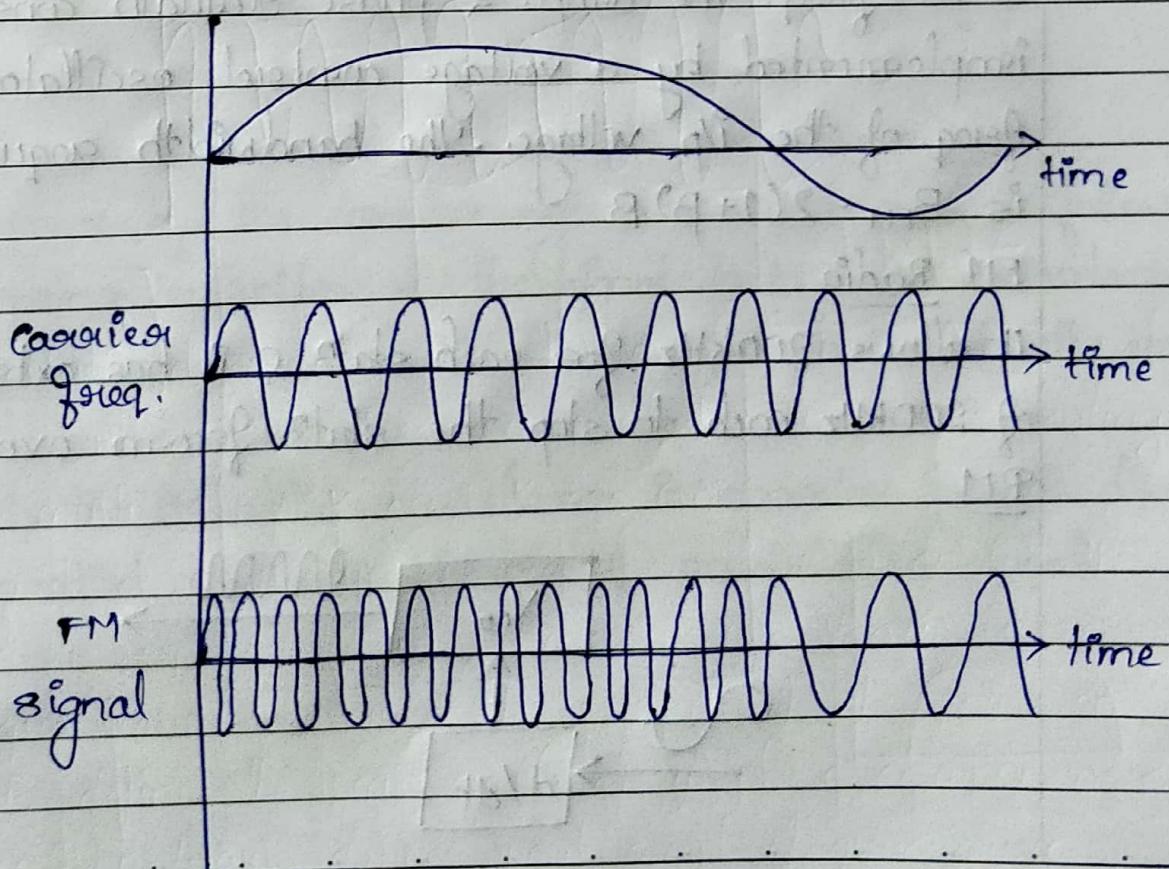
The carrier signal is modulated so that its ampl. varies with the changing amplitudes of the signal. The freq. & phase remain const. The modulating signal is the envelope of the carrier. It is implemented by using a simple multiplier so that the carrier signal changes according to the ampl.

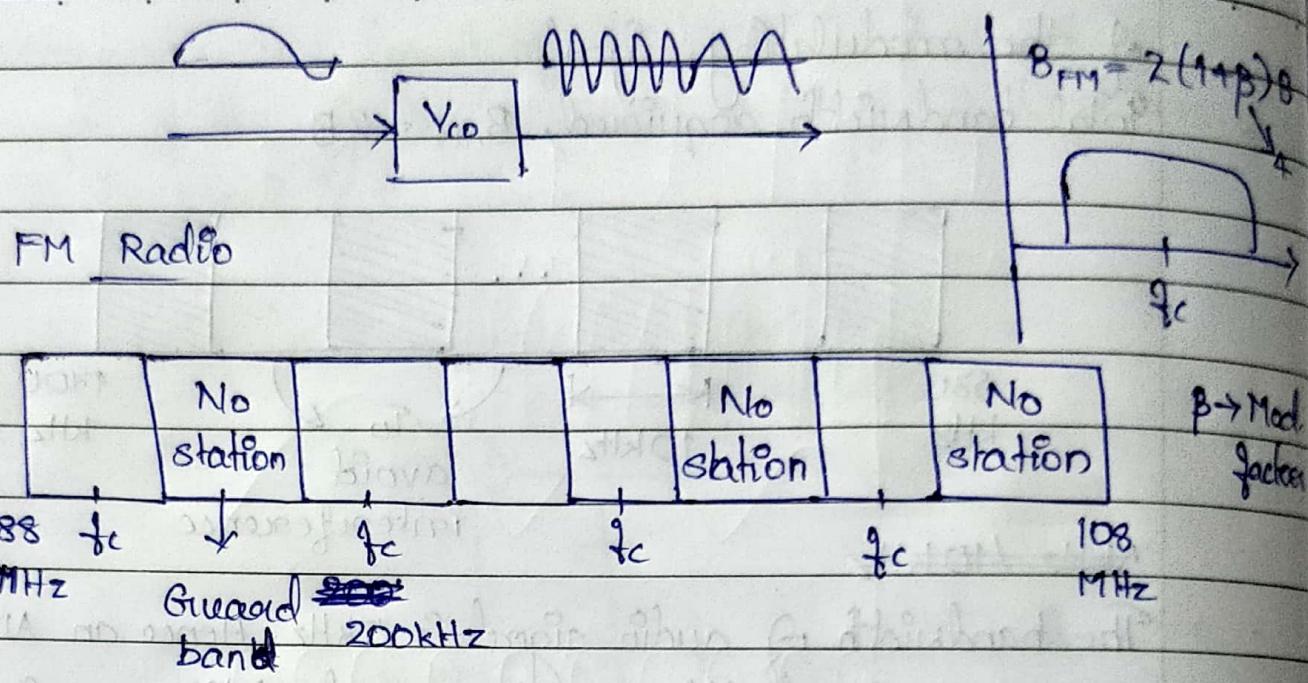
of the modulating signal.

Total bandwidth required, $B_{AM} = 2B$



The bandwidth of audio signal is 5kHz. Hence an AM radio statn has bandwidth of 10 kHz. Each statn's carrier freq is separated from its either side by 10 kHz to avoid interference.



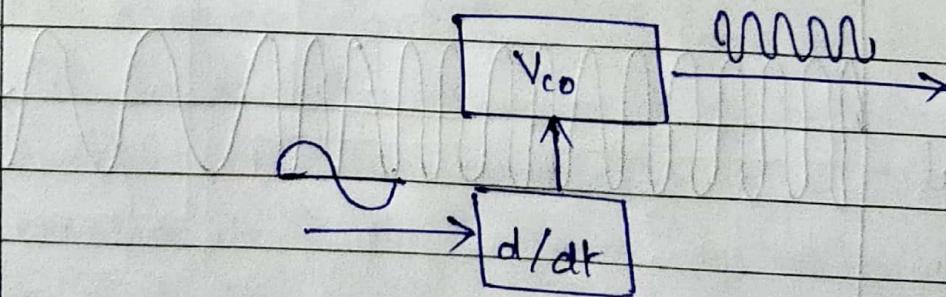


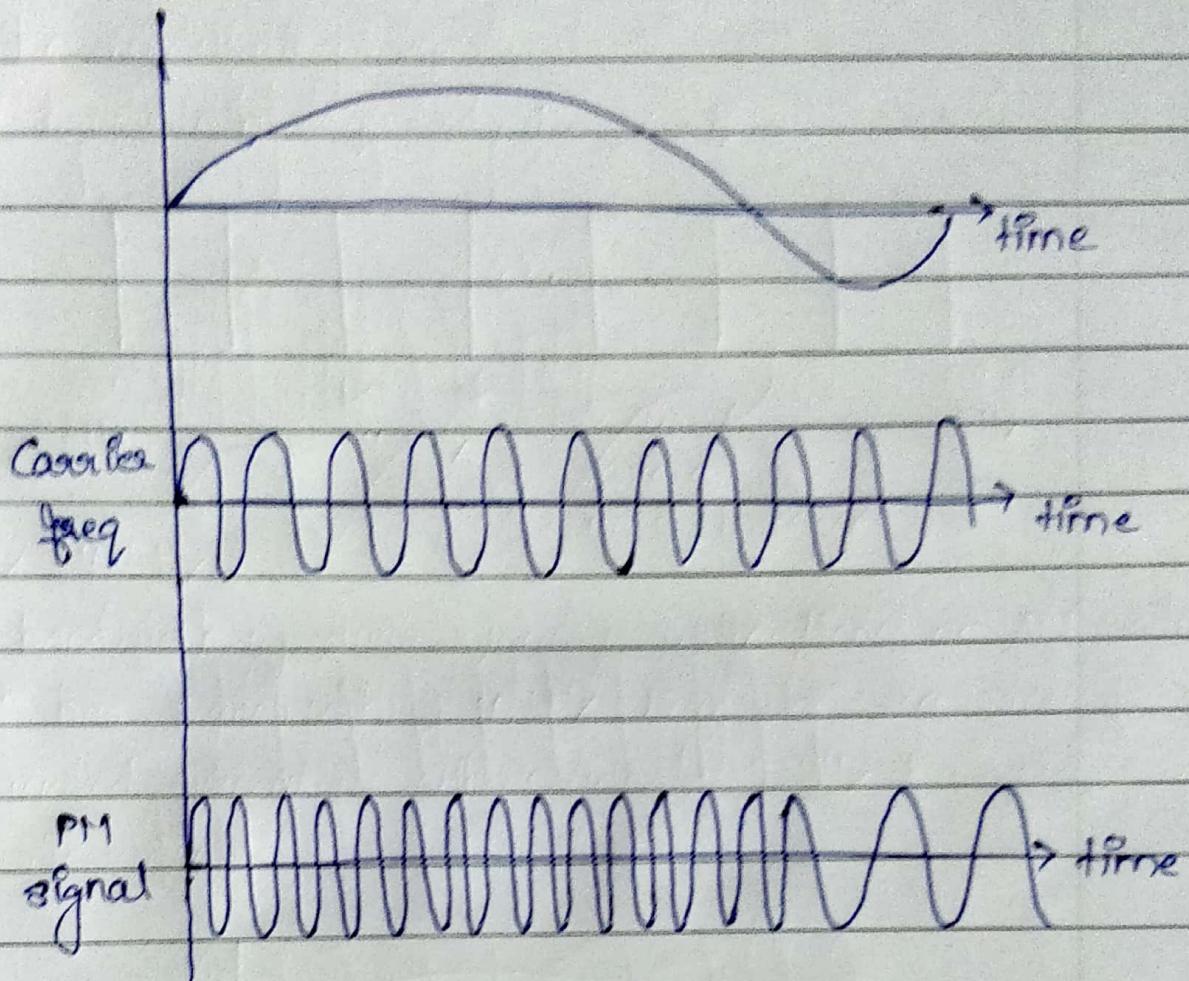
In FM, the freq. of the carrier signal is modulated to follow the changing voltage of the modulating signal based on its freq. The ampl. & phase remain const. It is implemented by a voltage control oscillator based on freq. of the i/p voltage. The bandwidth required for FM is $B_{FM} = 2(1+\beta)B$

FM Radio

It allows 200kHz for each statn & it has extra guard band of 200kHz each to stop the statns from overlapping

PM





The phase of the carrier signal is modulated to follow the change in voltage of the signal. In FM, the instantaneous change in carrier freq. is directly proportional to the ampl. of modulating signal.

In PM, the instantaneous change in carrier freq. is directly proportional to the derivative of modulating signal.

$\beta = 1 \rightarrow$ Narrow band

$\beta = 3 \rightarrow$ Wide band