Unit - III

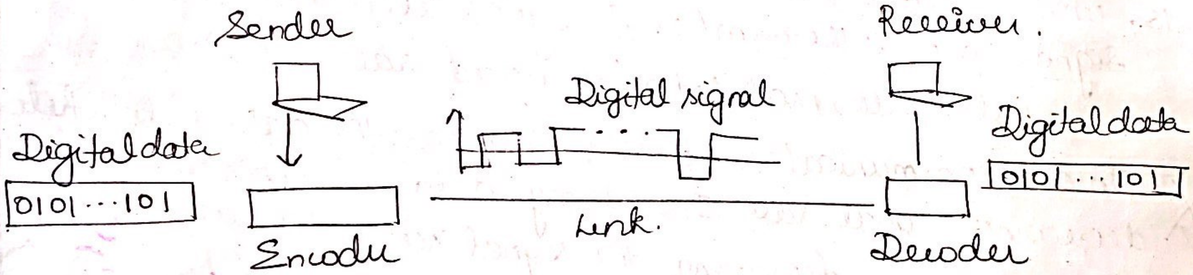
Line coding

Line coding is the process of converting digital data to digital signals. Any data in the form of text, numbers, graphical images, audio or video are stored in computer memory as sequence of bits.

Sender digital data are encoded into digital signals

Receiver digital data are recreated by decoding digital signal.

Line coding and decoding



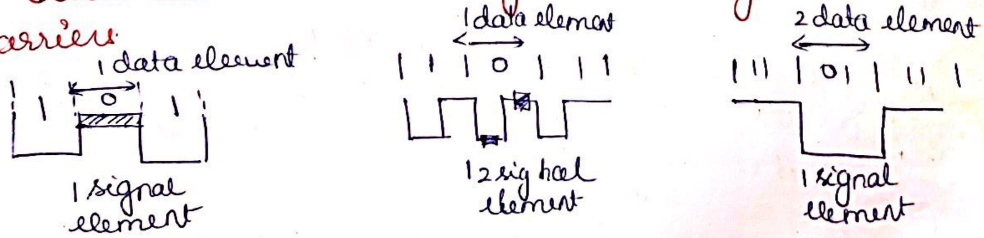
Characteristics

· Signal Element verses Data Element

Data element is the smallest entity that represent a piece of information ie bit. It is what we need to send.

Signal element is the shortest unit of a digital signal. It is what we can send.

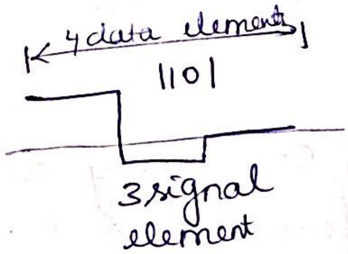
Data elements are being carried, signal elements are carriers.



a) One data element per one signal element (r=1)

b) One data element/ 2 signal element (r=1/2)

c) 2 data elements/ one signal element (r=2)



Each data element is a person who needs to be carried from one place to another. Signal element is a vehicle that can carry people. If r=1, it means each person is driving a vehicle. If r>1, ie more than one person is travelling in a vehicle.

2) Data Rate versus Signal Rate

Data rate defines the number of data elements (bits) sent in 1s. unit is bits per second (bps). It is called as bit rate.

Signal rate is the number of signal elements sent in 1s.

pulse rate, modulation or baud rate.

In data communication, it is to increase the data rate while decreasing signal rate. Increasing data rate increases speed of transmission; decreasing the signal rate decreases bandwidth requirement.

$$\boxed{S = N/r}$$

S = signal rate r = one signal element.
N = data rate.

3 cases worst, best and average.

worst case when we need maximum signal rate.
best case when we need minimum.
average case – $S_{ave} = \left(C \times N \times \frac{1}{r}\right)$ C = Case factor.

eg Prob 1: A signal is carrying data in which one data element is encoded as one signal element (r=1). If the bit rate is 100 kps - what is avg value of baud rate if C is between 0 and 1?

Assume $C = 1/2$ $S_{av} = C \times N \times \frac{1}{r} = \frac{1}{2} \times 100 \times 10^3 \times \left(\frac{1}{1}\right) = 50k$ baud

③

3) Bandwidth

Actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.

Baud rate determines the required bandwidth for a digital signal.

Bandwidth reflects the range of frequencies. Bandwidth is defined as a range of frequencies. Bandwidth is $\propto$ proportional to the signal rate (baud rate)

Min bandwidth $\boxed{B_{min} = C \times N \times \frac{1}{r}}$

Max data rate $\boxed{N_{max} = \frac{1}{C} \times B \times r}$

4) Baseline wandering

In decoding a digital signal, the receiver calculates a running average of the received signal power. This average is called baseline. The incoming signal power is evaluated to determine the value of the data element.

5) DC component

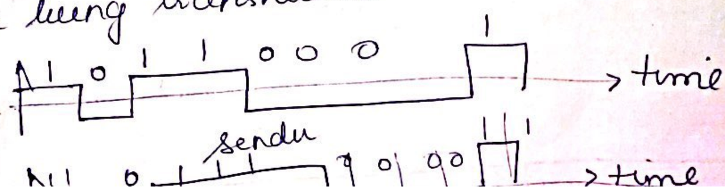
When the voltage level in a digital signal is constant for a while, spectrum creates very low frequencies. These frequencies around zero called DC (direct current) components. that cannot pass low frequencies

eg A telephone line cannot pass frequencies below 200Hz. Long distance link use one or more transformers to isolate different parts of line electrically. — we use a scheme with no DC component.

eg

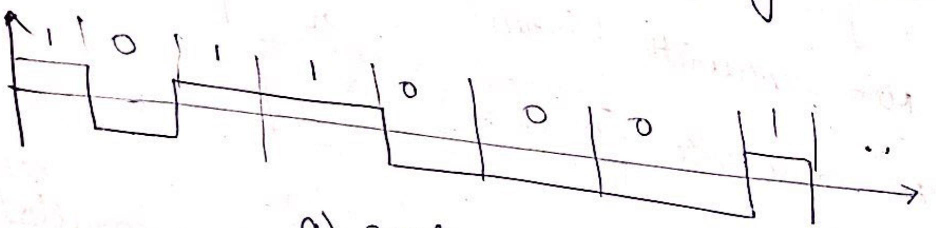
6) Self synchronisation

Self synchronising signal includes timing info in the data being transmitted

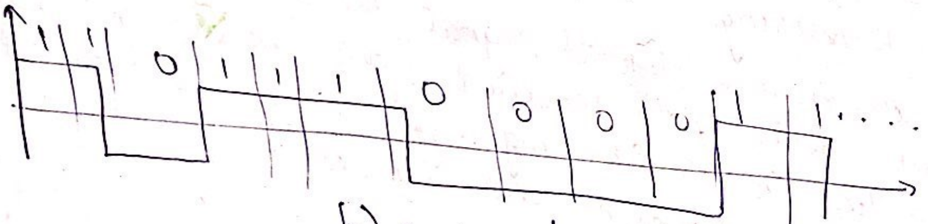


If there are transitions in the signal that alert the receiver to the beginning, middle or end of the pulse. If the receiver's clock is out of synchronisation, points can reset the clock.

Effect of lack of lack of synchronisation



a) sent.



b) received.

3) In a digital transmission, the receiver clock is 0.1% faster than the sender clock, How many extra bits/sec does the receiver receive if the data rate is 1Kbps? How many if the data rate is 1Mbps?

At 1 Kbps, the receiver 1001 bps instead of 1000 bps.

1000 bits sent ⟶ 1001 bits received ⟶ 1 extra bits

At 1 Mbps, the receiver receives 1001,000 bps instead of 1,000,000 bps

1000,000 bits sent ⟶ 1001 000 bits received ⟶ 1000 extra bits

7) Built in Error detection

Built in error detecting capability in the generated code detect some or all of the errors that occurred during transmission

8) Immunity to Noise and Interference

It is desirable to have characteristic in code that is immune to noise and other interference.

9) Complexity

Complex system is costly to implement than a simpler. eg a scheme with 4 signal levels is more difficult to interpret that one that uses only 2 levels.

Line coding schemes

Divide line coding schemes into 5 broad categories

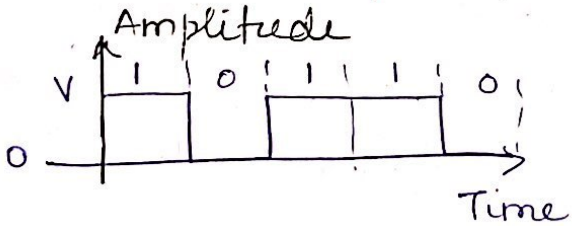
a) Unipolar Scheme

* all the signal levels are on one side of the time axis either above or below

Linecoding
- Unipolar — NRZ
- Polar — NRZ, RZ and biphase (Manchester and diff Manchester)
- Bipolar — AMI and pseudoternary
- Multilevel
- Multitransition

NRZ (Non-Return to Zero) positive voltage defines bit 1 and zero voltage defines bit 0.

* It is called NRZ because the signal does not return to zero at the middle of the bit.



Normalised power

$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

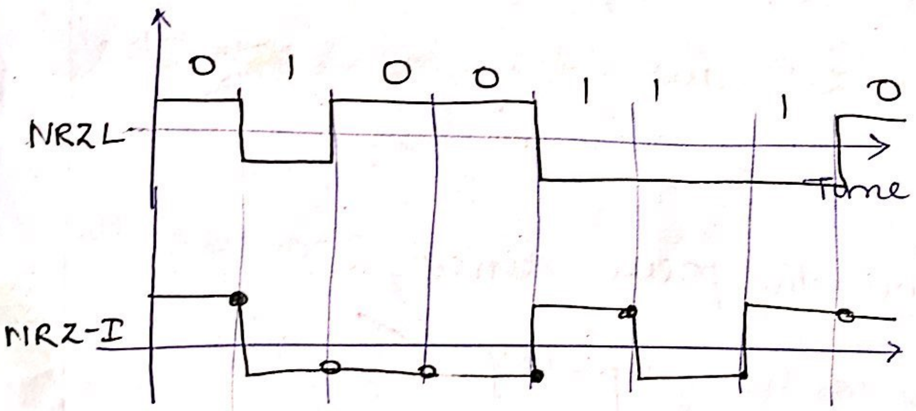
* It is very costly, the normalised power (the power needed to send 1 bit / unit line resistance) is double that for polar NRZ — not used in data communications

b) **Polar Scheme** - voltages are on both sides of the time axis. eg voltage level for 0 can be positive and voltage level for 1 can be negative.

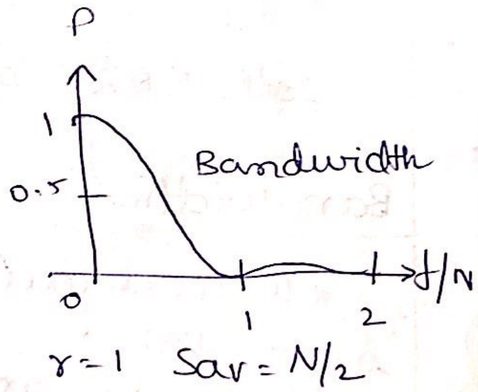
Non-return to zero (NRZ)

1) In polar NRZ, 2 levels of voltage amp are used.
2) 2 version of polar NRZ: NRZ-L [NRZ-level]
NRZ-I - [NRZ - Invert]

NRZ-L level of the voltage determines the value of the bit
NRZ-I - the change or lack of change in the level of the voltage determines the values of the bit.
If there is no change, bit is 0.
If there is change bit is 1



o No inversion : Next bit is 0
• Inversion: Next bit is 1



Normalized bandwith

Analysis

1) Baseline wandering is problem for both variations, it is twice as severe in NRZ-L
2) If there is a long sequence of 0s and 1s in NRZ-L average signal power becomes skewed.

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3) The receiver has difficulty discerning the bit value in case of NRZ-I problem occurs only for long sequence of 0s.

If long sequence of 0s are eliminated - base line wandering is avoided.

4) Synchronization problem exists in both the cases, It is more serious in NRZ-L than in NRZ-I.
With long sequence of 0s can cause a problem in both schemes, long sequence of 1s affect only NRZ-L

5) Problem with NRZ-L occurs when there is a sudden change of polarity in the system. eg if twisted pair is the medium, change in polarity of wire results in all 0's interpreted as 1s and all 1s interpreted as 0's.

Both NRZ-L and NRZ-I have average signal rate of N/2 Bd

Bandwidth

* vertical axis shows the power density (power of 1Hz of bandwidth)
* Horizontal axis shows the frequency.
* value of the power density is very high around frequencies close to zero. ie DC components carry a high level of energy. ie energy is concentrated in frequencies between 0 and N/2 ie the average of the signal rate is N/2, energy is not distributed between 2 halves. NRZ-L and NRZ-I both have a DC component problem.

4) System is using NRZ-I to transfer 10Mbps data. What are the average signal rate and minimum bandwidth?

Average signal rate is $S = N/2 = 500$ K baud.
Minimum bandwidth for this average baud rate $B_{min} = S = 500$ KHz

b) Return to Zero (RZ)

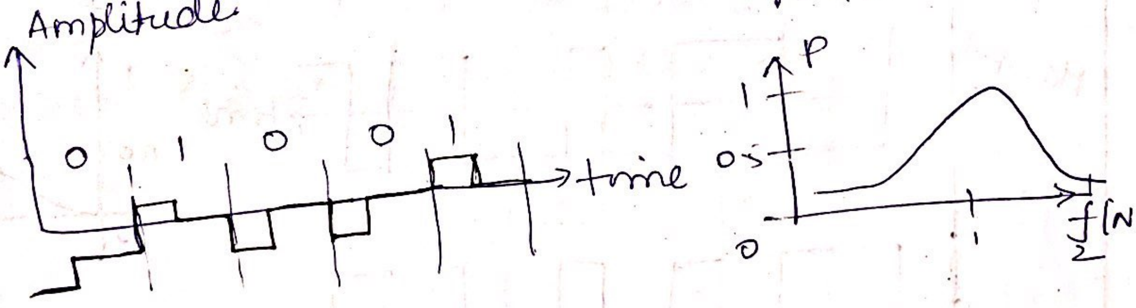
Problem of NRZ occurs when sender and receiver clocks are not synchronised i.e it doesnot know when one bit has ended and the next bit starting.

Solution: 1) uses 3 values - positive, negative and zero
2) RZ the signal changes not between bits but during the bit.

in the middle of each bit

For a 0 the signal goes to 0 in the middle of each bit. It remains until the beginning of next bit.

$r = 1/2$ $S_{av} = N$



Disadvantages

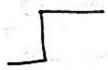
1) Requires 2 signal changes to encode a bit and occupies greater bandwidth.
2) A sudden change of polarity resulting in all 0s interpreted as 1s and all 1s interpreted as 0s still exists
3) NO DC component problem

4) Complexity problem. – uses 3 level of voltage – more complex to create and discern.

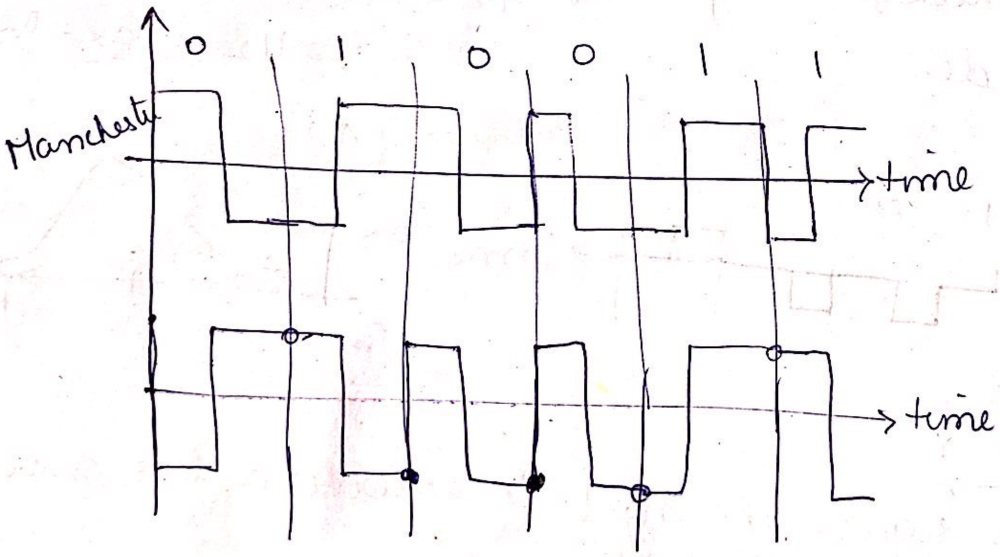
Biphase: Manchester and Defferential Manchester

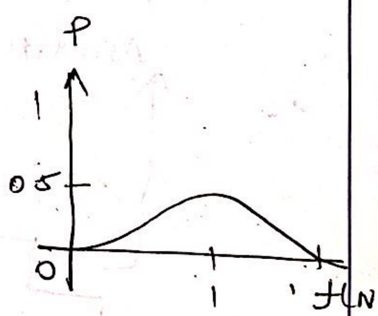
Manchester: 1) idea of RZ (transition at the middle of the bit and idea of NRZ-L are combined
2) Duration of bit is divided into 2 halves. voltage remains at one level during first half and moves to the other level in second half.
3) Transition in middle provides synchronisation.

Bipolar Polar biphase Manchester and differential Manchester

0 is    1 is 

$r = 1/2$ $S_{av} = N$





o No inversion: Next bit is 1   • Inversion: Next bit is 0

Differential Manchester combines ideas of RZ and NRZ-I

- always a transition at middle of the bit but bit values are determined at the beginning of the bit.
- if the next bit is 0 there is transition, if the next bit is 1, there is none.

Manchester scheme overcomes problems associated with NRZ-L.

Differential Manchester overcomes several problem associated with NRZ-I

Problems are overcome

1) No baseline wandering
2) no dc component - each bit has positive and negative voltage contribution.
3) signal rate is double for NRZ because there is transition at the middle of the bit and may be one transition at the end of each bit.

Minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

Bipolar scheme [Multi level binary]

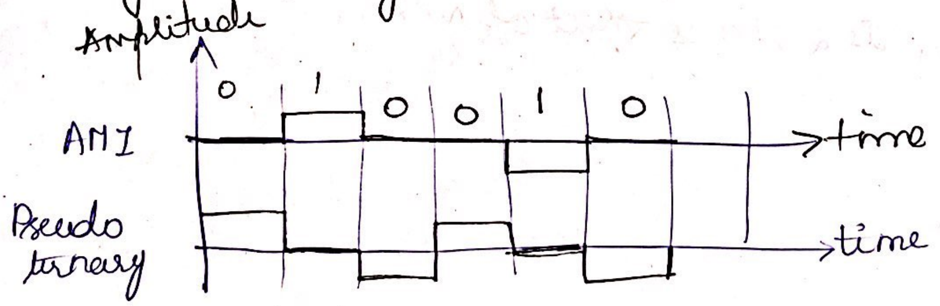
* 3 voltage levels ; positive, negative and zero.
* voltage level of one data element is at zero, while voltage level of other element alternates between positive and negative.

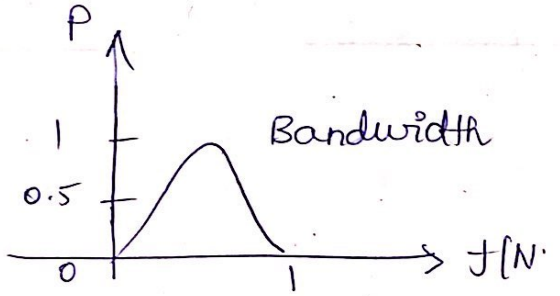
AMI [Alternate mark inversion]

Mark means 1, AMI means alternate 1 inversion.
A neutral 0 voltage represents binary 0.
Binary 1s are represented by alternating positive and negative voltages

Pseudo ternary variation of AMI encoding is called pseudo ternary in which 1 bit is encoded as a zero voltage

0 bit is encoded as alternating positive and negative voltage.





$S_{ave} = 1/2 N$

$r = 1$

1) Bipolar scheme is alternate to NRZ has datarate seniees
2) NRZ scheme has its energy concentrated near NRZ 0 freq, which makes it ~~unstable~~ unsuitable for transmission over channels with poor performance around this freq

Concentration of the energy in bipolar encoding is around frequency N/2.

Reason for a DC component

1) For a long sequence of 1's, voltage level alternates between positive and negative, it is not constant - no dc component
2) For a long sequence of 0's, voltage remains constant but its amplitude is 0. i.e it doesnt have DC component

i.e a sequence that creates a constant zero voltage doesnot have a DC component.

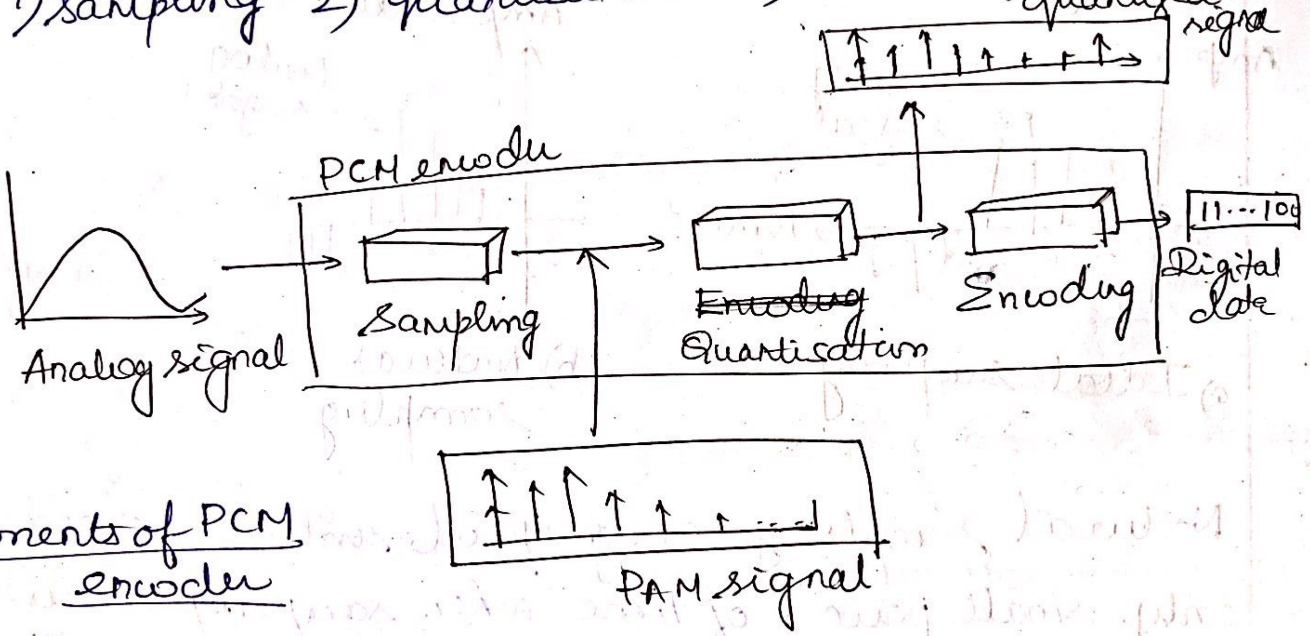
Disadv

AMI when used for long distance communication, has synchronisation problem

Pulse code Modulation (PCM).

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). It has 3 processes.

1) sampling 2) quantisation 3) encoding.



Components of PCM encoder

1) The analog signal is sampled.
2) The sampled signal is quantised
3) The quantised values are encoded as stream of bits

Sampling

1) First step in PCM is sampling
2) The analog signal is sampled every T_s sec,

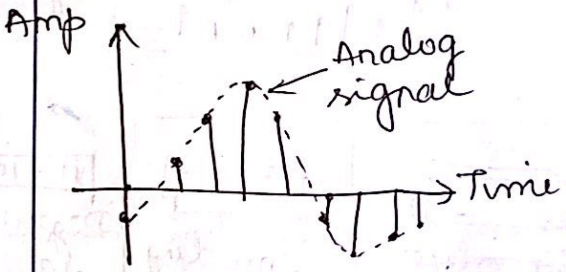
T_s – sampling rate

sampling frequency $= f_s = 1/T_s$

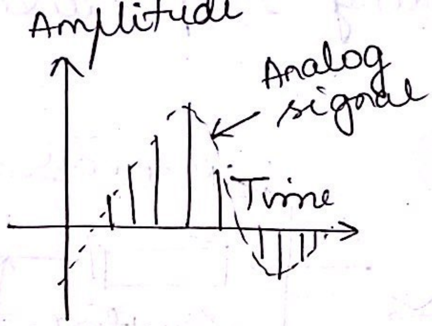
3 sampling methods
1) ideal
2) natural
3) flat top

Ideal sampling

1) pulses from analog signal are sampled.
2) This is Ideal sampling method and cannot be easily implemented.



a) Ideal sampling

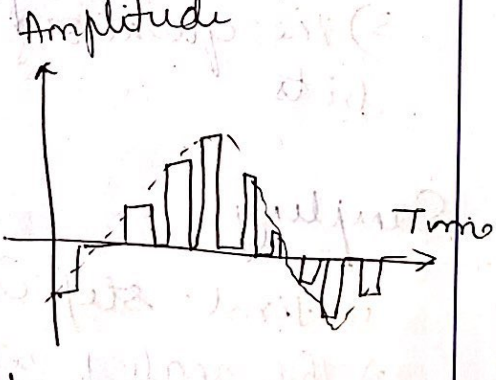


b) Natural sampling

Natural sampling 1) high speed switch is turned on for only small period of time when sampling occurs.
2) Result of is a sequence of samples that retains the shape of analog signal.

Flattop sampling

1) uses sample and hold to create flat top samples by a circuit
2) the most common sampling method.



1) Sampling process is referred to as pulse amplitude modulation [PAM]
2) Result is still an analog signal with non integral values.

Sampling Rate

According to Nyquist theorem, to produce the original analog signal, condition is that the sampling rate must be atleast twice the highest frequency of the original signal.

Characteristics of Nyquist

i) A signal can be sampled if and only if the signal is band-limited

ii) A signal with an infinite bandwidth cannot be sampled

2) Sampling rate must be atleast 2 times the highest freq, not bandwidth.

a) If analog signal is low pass, bandwidth and highest freq are the same value.

b) if analog signal is bandpass, bandwidth value is lower than the value of max freq

ex 1) Telephone companies digitize voice by a max freq = 4000Hz what is the sampling rate?

Sampling rate = $2f_s = 2 \times 4000 = 8000$ samples/sec

2) Complex low pass signal has bandwidth of 200KHz. What is min freq rate of this signal

1) Bandwidth is between 0 and f. f - max freq.

2) sample at this $2 \times f_s = 2 \times 200$ KHz.

$= 400k$ samples/sec.

3) A complex bandpass signal has a bandwidth of 200KHz. What is the min sampling rate for the signal.

We donot know where the bandwidth starts or ends. We dont know max freq in the signal.

Quantisation

Result of sampling is a series of pulses with amplitude values between maximum and minimum amplitudes of the signal.

Set of amplitudes can be infinite with nonintegral values between the 2 limits. Values cannot be used in encoding process.

Steps involved in quantisation

1) The original analog signal has instantaneous amplitude between Vmin and Vmax

2) Divide the range into L zones, each of height Δ(delta)

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

3) We assign quantized values of 0 to L-1 to the midpoint of each zone

4) The value of the sample amplitude to the quantised values.

Sample amplitude are between -20 to +20V.

8 levels i.e. L=8 $\Delta = \frac{+20-(-20)}{8} = \frac{40}{8} = 5$

~~The~~ quantisation process selects the quantization value from the middle of each zone.

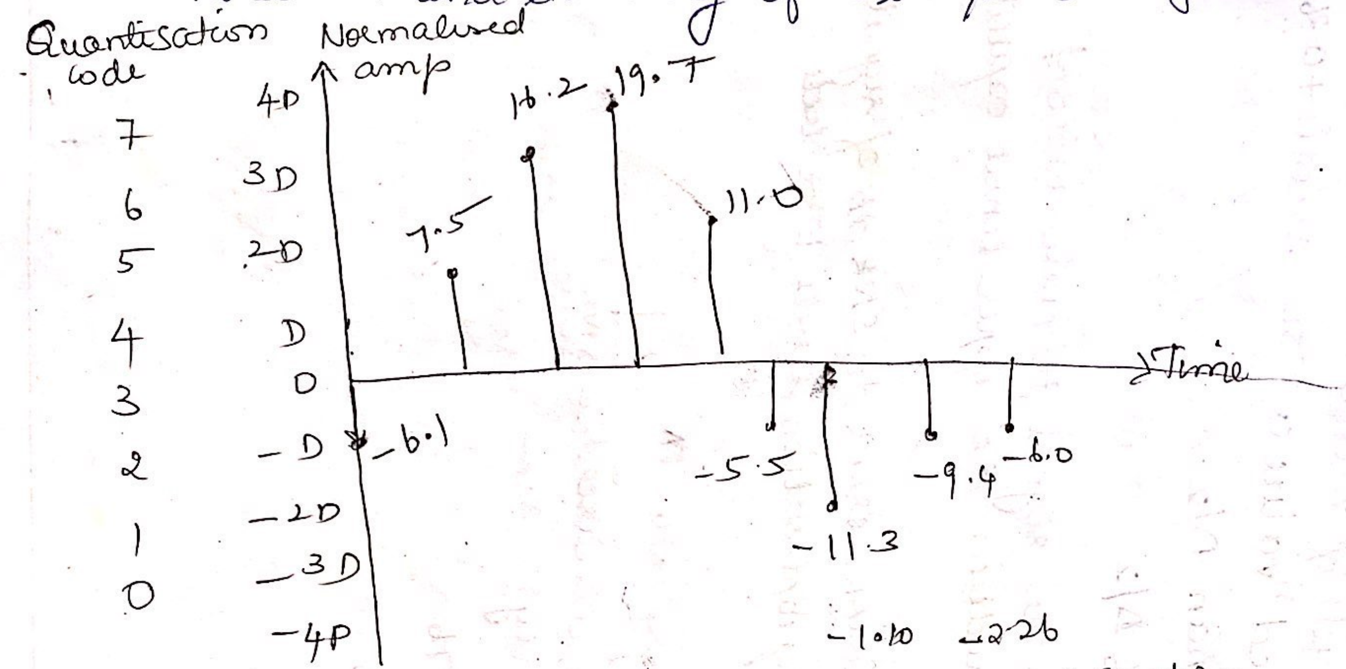
The normalised quantised values are different from the normalised amplitudes. The difference is called normalised error (third row).

Fourth row is the quantisation code for each sample based on the quantisation levels at the left of the graph

Quantisation levels

If the amplitude of a signal fluctuates between 2 values only, we need only 2 values -
Choosing lower values of L increases the quantisation error if there is a lot of fluctuation in the signal.

Quantisation and encoding of a sampled signal



Normalised PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalised quantised values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalised error	-0.28	0	0.26	-0.44	0.30	-0.40	-0.24	+0.38	-0.30
Quantisation code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

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Quantisation error

1) Quantisation is an approximation process
2) input values to the quantiser are real values.
3) output values are approximated values.
4) output values are chosen to be middle value in the zone if input value is also at the middle of the zone, no quantisation error
5) eg normalised amplitude of third sample is 3.24. and normalised quantised value is 3.50. error is +0.26, Value of error is less than $\Delta/2$.

ie $-\Delta/2 \leq \text{error} \leq \Delta/2$

1) Quantisation error changes the signal to noise ratio of the signal, which in turn reduces the upper limit capacity

2) Contribution to quantisation error to SNR db of signal depends on the number of quantisation levels L or bits per sample n_b

$$\boxed{SNR_{dB} = 6.02n_b + 1.76\ dB}$$

3) Increasing number of levels increases the SNR

1) What is the SNRdb of above eg?

$$SNR_{dB} = 6.02(n_b) + 1.76$$
$$= 6.02(3) + 1.76$$
$$= 19.82\ db.$$

2) A telephone number line must have an SNRdB above 40. What is the minimum no of bits/sample?

$$SNRdB = 6.02 n_b + 1.76 = 40$$

$\boxed{n_b = 6.35}$ Telephone assign 7 or 8 bits/sample

Uniform versus Nonuniform Quantisation

1) Distribution of instantaneous amplitude in analog is not uniform

2) Changes in amplitude occur more frequently in lower signals amplitudes than in higher ones.

3) It is better to use nonuniform eg height is not fixed. it is greater near the lower amplitudes and less near higher amplitudes.

Nonuniform quantisation is achieved by companding and expanding. signal is companded at the sender before conversion; it is expanded at the receiver after conversion.

Companding means reducing the instantaneous voltage amplitude for large values; expanding is the opposite process. Companding gives greater weight to strong signals and less weight to weak ones.

Nonuniform quantisation reduces the SNRdB of quantization,

Encoding Last step in PCM is encoding.

After each sample is quantised and number of bits per sample is decided, each sample is changed to n_b bit code word.

No of bits for each sample is determined from the number of quantisation levels. If the number of quantisation levels is L, number of bits is $\boxed{n_b = \log_2 L}$

eg if L is 8, $n_b = 3$.

$$\boxed{\text{Bit rate} = \text{sampling rate} \times \text{no of bits / sample} = f_s \times n_b}$$

(1) We want to digitise human voice. What is bit rate, assuming 8 bits/samples

Human voice contains frequencies 0 to 4000Hz.

sampling rate $= 4000 \times 2 = 8000$ samples/s.

Bit rate $= 8000 \times 8 = 64{,}000$ bps $= 64$ Kbps.

Original signal Recovery

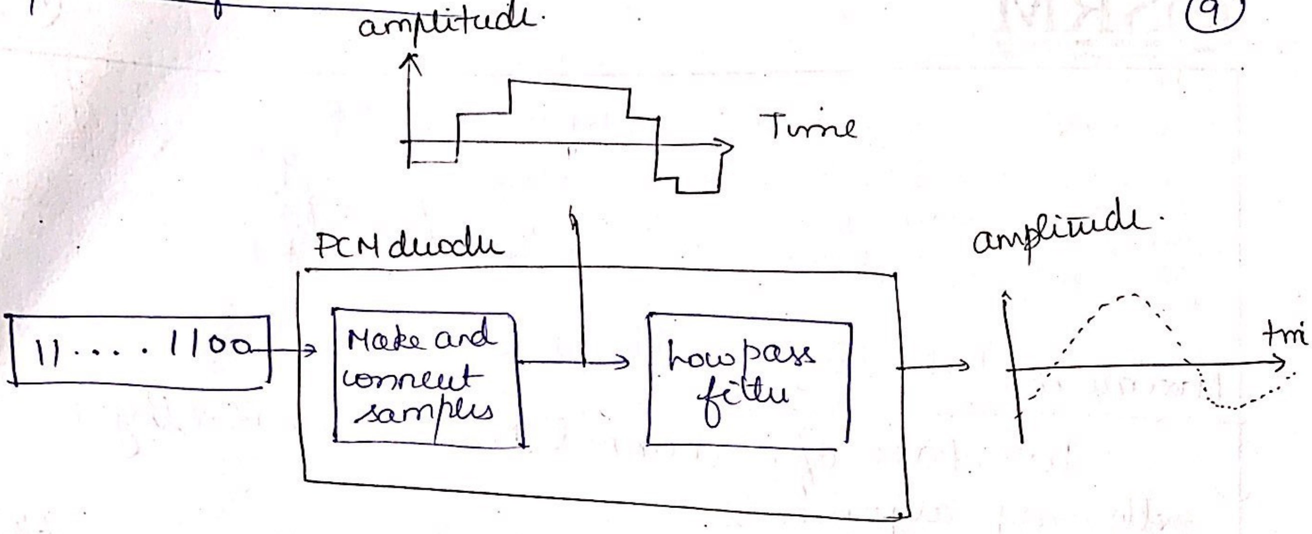
1) Recovery of original signal requires PCM decoder. PCM decoder uses circuitry to convert codewords into a pulse that holds the amplitude until the next pulse.

2) After staircase signal is completed, it passes through low pass filter to smooth the staircase signal to an analog signal.

3) Filter has cutoff frequency as the original signal at the sender.

4) If the signals are sampled (at or greater than) Nyquest sampling rate and if there are enough quantisation levels, original signal will be recreated

Components of PCM decoder



PCM bandwidth

The new minimum bandwidth of the channel to pass the digitized signal is

Minimum bandwidth of a line encoded signal is

$$B_{min} = c \times N \times \frac{1}{r}$$

$$B_{min} = c \times N \times \frac{1}{r} = c \times n_b \times f_s \times \frac{1}{r}$$

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

where $\frac{1}{r} = 1$ [for NRZ or bipolar signal)

$c = \frac{1}{2}$ (avg situation)

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

This means the minimum bandwidth of digital signal is n_b times

1) A low pass analog signal of 4KHz. we send the analog signal, a channel with min bandwidth of 4KHz. If we digitize the signal and send 8 bits / sample, a channel with a minimum bandwidth of $8 \times 4KHz = 32 KHz$.

Maximum Data Rate of a channel

Data Rate of a channel $\boxed{N_{max} = 2 \times B \times \log_2 L}$

Following arguments:

1) The available channel is low pass with bandwidth B
2) the digital signal is to send L levels, each level is a signal element

$$\boxed{r = \frac{1}{\log_2 L}}$$

3) we pass the digital signal through a low pass filter to cut off the frequencies above BHz.
4) The resulting signal is a analog signal and sample it at $2 \times B$ samples/sec. and quantise it using L levels.
5) Resulting bit rate = $N = f_s \times n_b$

$$= 2 \times B \times \log_2 L$$

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

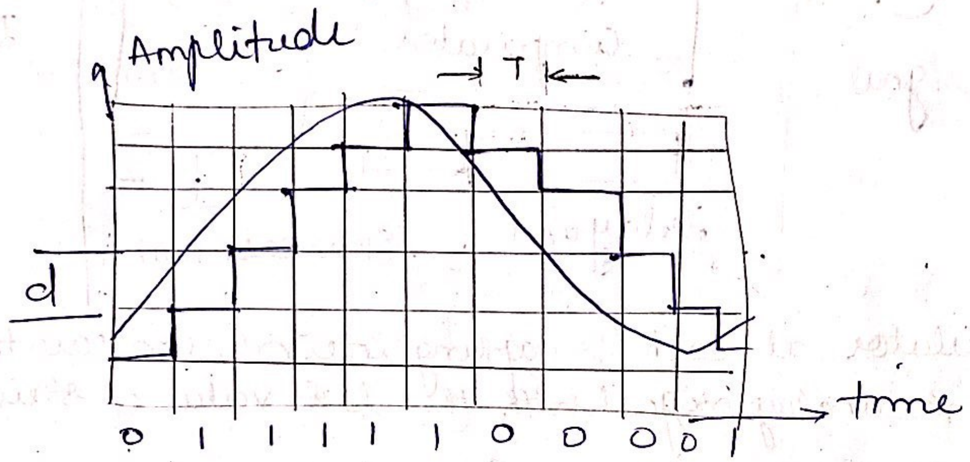
Minimum Required bandwidth

$$B_{min} = \frac{N}{(2 \times \log_2 L)} \text{ Hz}.$$

Delta Modulation (DM)

Disadv 1) PCM is very complex 2) PCM finds value of signal amp for each sample.

1) To reduce complexity DM is used.
2) DM finds the change from previous sample.
3) No code words here. 4) bits are sent one after another



The process of delta modulation

Modulator

Modulator is used at the sender site to create a stream of bits from an analog signal. The

1) Process records the small positive or negative changes called δ.
2) If delta is +ve, process records 1. if -ve, process records 0.] process needs a bar against which the analog signal is compared.
3) Modulator builds a second signal that resembles a staircase.
4) Finding the change is then reduced to comparing the input signal with the gradually made staircase signal.

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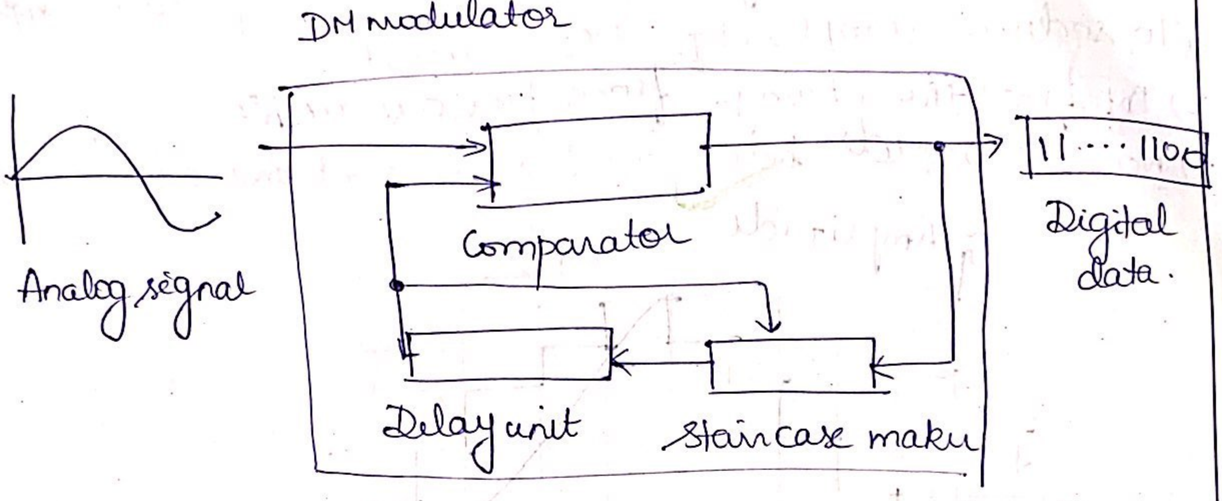


1) The modulator at each sampling interval, compares the value of the analog signal with the last value of staircase signal.

2) If amplitude of the analog signal is larger than the next bit in digital data is 1, otherwise 0.

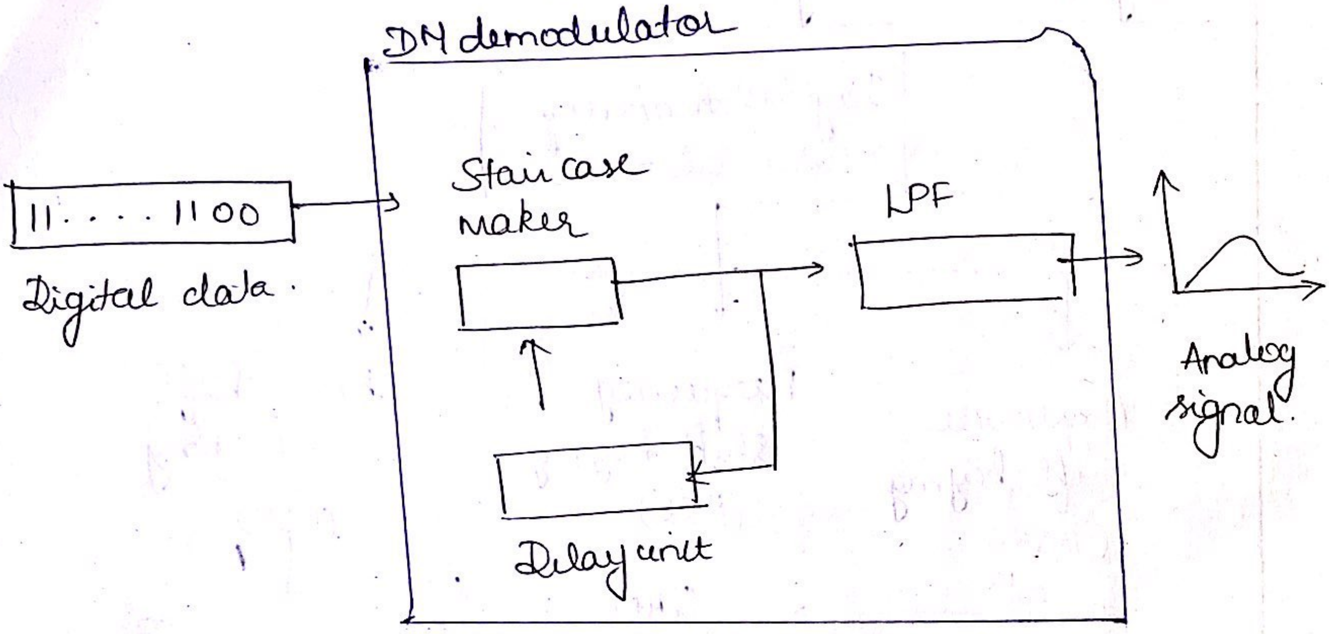
3) O/P of comparator makes the staircase itself.

4) if next bit is 1, staircase makes the last point of staircase signal δ up.
if next bit is 0, it moves it δ down.

5) We need a delay unit to hold the staircase function for a period between 2 comparisons.

Demodulator takes the digital data and using staircase maker and delay unit creates the analog signal. The created analog signal, needs to pass through LPF for smoothing.

Delta Demodulation



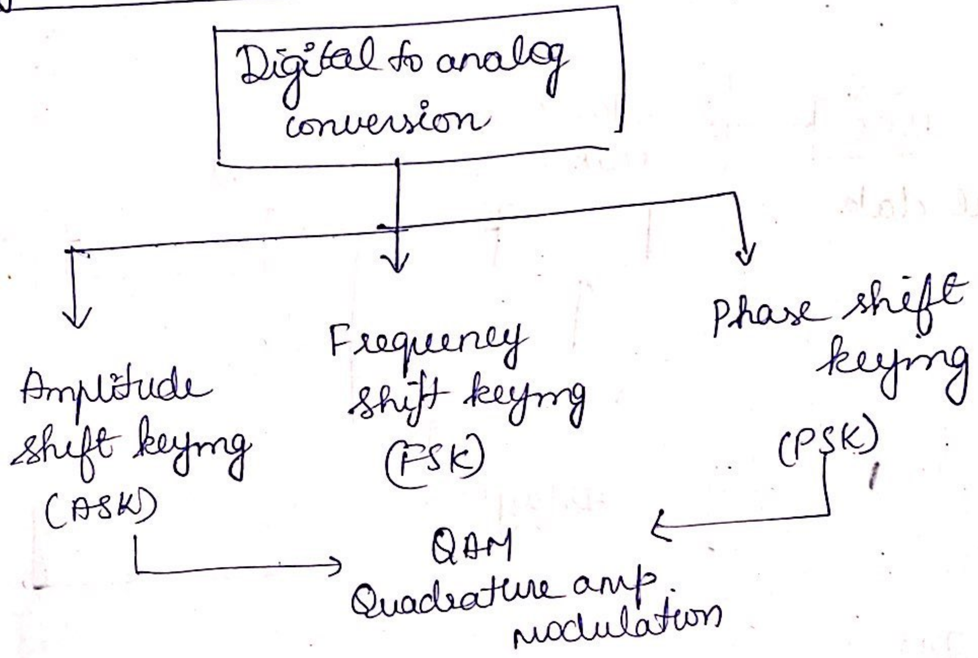
Adaptive DM

Better performance is done by not fixing δ value. In adaptive delta modulation, value of δ changes according to the amplitude of analog signal.

Quantisation error

Quantisation error in DM is less than that of ~~DM~~ PCM

Digital to analog conversion



Definitions

Data element is the smallest piece of information that is exchanged. Signal element is the smallest unit of a signal that is constant.

Data rate and signal rate $\boxed{S = N \times \frac{1}{r}}$

r is the no of data element carried in one signal element

$r = \log_2 L$.

Bit rate is the number of bits/sec. Baud rate is the number of signal elements/sec. In analog transmission, baud rate is less than or equal to bit rate

1. An analog signal carries 4 bits / signal element. If 1000 signal elements are sent / sec find the bit rate

$r = 4$, $S = 1000$ $N = ?$

$S = N \times 1/r$ $N = S \times r = 1000 \times 4 = 4000$ bps.

2. An analog signal has a bit rate of 8000 bps. and a baud rate of 1000 baud. How many data element are carried by each signal element? How many ~~data~~ signal element we need?

$S = 8000$ $N = 8000$

$S = N \times \frac{1}{r}$ $r = N/S = 8000/1000 = 8$ bits / baud.

$r = \log_2 L =$ $L = 2^r = 2^8 = 256$

Bandwidth

The required bandwidth for analog transmission of digital data is proportional to the signal rate except for FSK. in which diff between carrier signals need to be added.

Carrier Signal

Sending device produces a high frequency signal that acts as a base for information signal. This base signal is called carrier signal or carrier freq. The receiving device is tuned to the frequency of the carrier signal that is excepts from the sender.

Digital information that changes the carrier signal by modifying one or more of its characteristics (amp, frequency or phase) This kind of modification is called modulation (shift keying)

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Amplitude Shift Keying

The amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remains constant while amplitude changes.

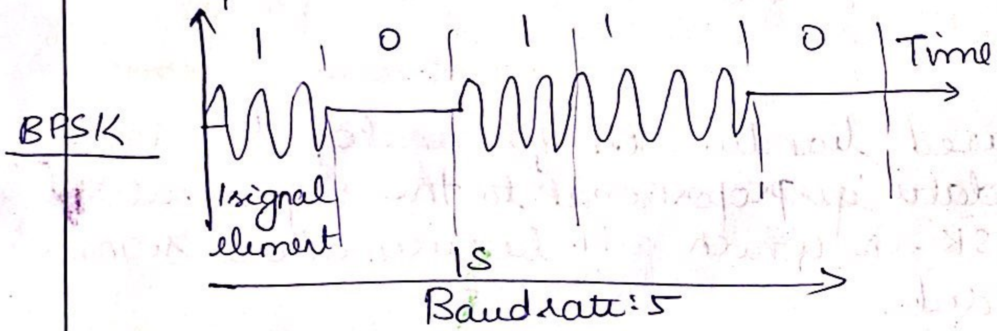
Binary ASK (BASK)

Several levels (kinds) of signal elements, each with a different amplitude.

ASK has only 2 levels - binary amplitude shift keying or on off keying (OOK)

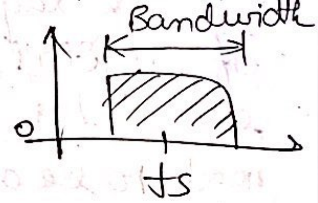
Peak amplitude of one signal level is 0; the other is same as the amplitude of carrier frequency.

Amplitude Bitrate: 5



$r=1, S=N$

$B=(1+d)S$



Bandwidth for ASK

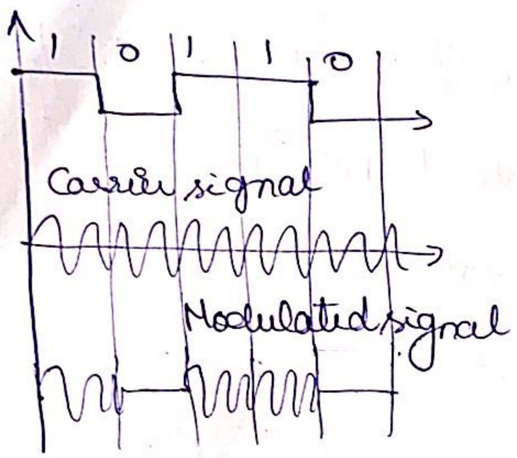
1) Bandwidth is proportional to the signal rate (baud rate)
2) d depends on the modulation and filtering process.
3) d is between 0 and 1.

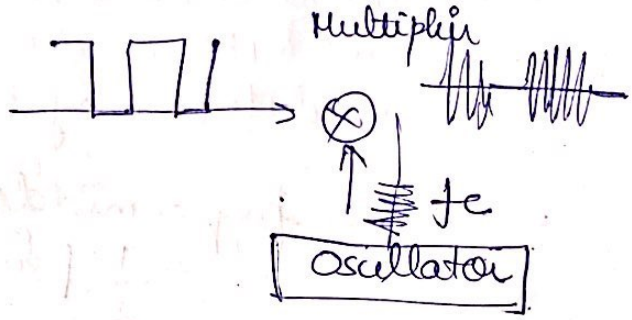
$$B=(1+d)\times S$$

1) Bandwidth max value $=2S$
   min value $=S$

2) Location of bandwidth - middle of BW $=f_c$

Implementation of ASK





1) If digital data are presented as a unipolar NRZ digital signal with a high voltage of 1V and low voltage of 0V,
2) Implementation is achieved by multiplying NRZ digital signal by multiplying NRZ digital signal by carrier signal coming from an oscillator.
3) When amplitude of NRZ is 1, amplitude of carrier freq is held, NRZ is 0 amp of carried freq is 0.

Multilevel ASK

It use 2 or more amplitude levels. ie 4, 8, 16 or more different amplitude for the signal and modulate data using 2, 3, 4 or more but at a time. ie $r=2, r=3, r=4$

Problems in ASK

(1) We have an available bandwidth of 100 KHz which spans from 200–300 KHz. What are the carrier freq and bit rate of modulated using ASK with d=1?

Carrier freq = middle of bandwidth

$= \frac{1}{2} \left[\overset{\text{Range}}{300-200}\right]$

$= \frac{1}{2}$ range

$B = (1+d)S \qquad = 250\,KHz.$

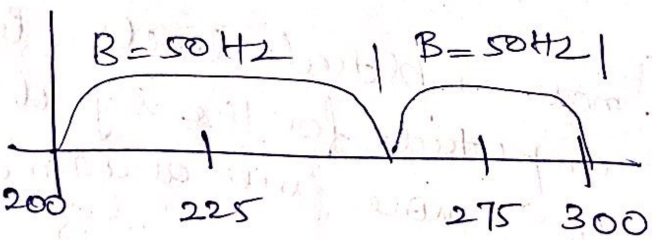
$S = \frac{1}{r} \qquad r = 1 \quad d = 1$

$B = 100\,KHz$ ie

$B = (1+d)S = 2 \times N \times \frac{1}{r} = 100\,KHz$

$\boxed{N = 50\,kbps}$

(2) If full duplex links with communication is used in both directions, determine 2 carrier frequencies and bandwidth.



$B = (1+d)S = 2 \times N \times \frac{1}{r}$

$r = \frac{1}{2}$

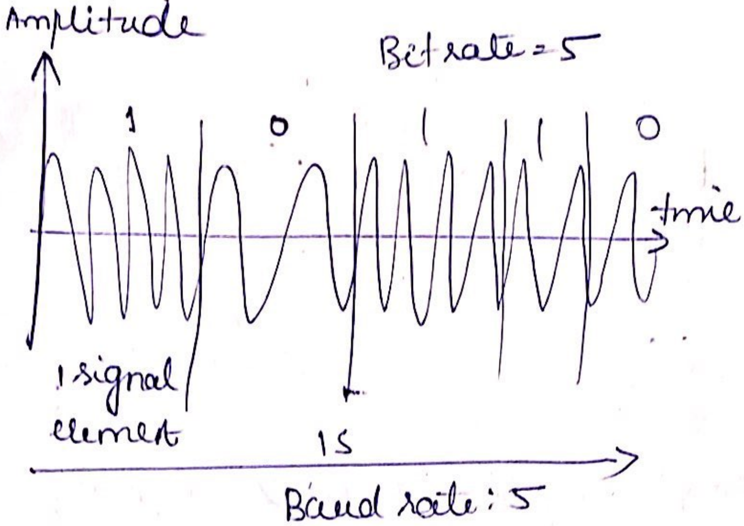
$\boxed{\therefore N = 25\,kbps}$

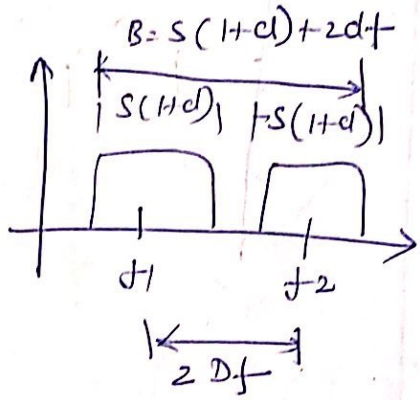
separating 2 ...

Frequency Shift keying

The frequency of the carrier signal is varied to represent data i.e. f_1 and f_2.

$r = 1 \quad S = N \quad B = (1+d)S + \Delta f$





Middle of one bandwidth is f_1 and middle of other is f_2. Both f_1 and f_2 are Δf apart from midpoint between 2 bands. Difference between 2 frequencies = $2\Delta f$.

Bandwidth

FSK is 2 ASK signal with its own carrier frequencies. If difference between 2 frequencies is $2\Delta f$.

$$B = (1+d) \times S + 2\Delta f$$

what is minimum value of $2\Delta f$.

Minimum value should be atleast S for proper modulation and demodulation.

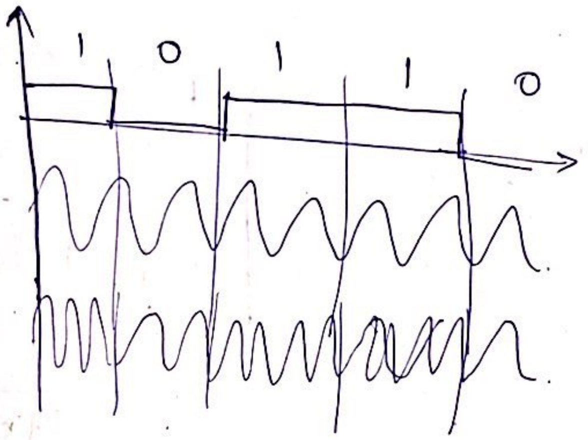
Implementation

- noncoherent – implemented by treating BFSK as 2 ASK mod with 2 carrier freq
- coherent – voltage controlled OSC (VCO)

Noncoherent BFSK, – discontinuity in phase when signal element ends and next begins

Coherent BFSK – phase continuous through boundary of 2 signal element.

VCO voltage controlled oscillator changes its frequency according to is zero.



Input to the oscillator is unipolar NRZ signal, when the amplitude of NRZ is 0, the oscillator keeps its regular frequency. amp is positive, frq is increased.

Multi level FSK

MFSK is not uncommon with FSK method. we can use 4 different frequencies f1, f2, f3 f4 to send 2 bits at a time.

to send 3 bits at a time, we use 8 frequencies.

$$B = (1+d) \times S + (L-1)2\Delta f$$

$$B = L \times S$$

MFSK uses more bandwidth used in case of noise.

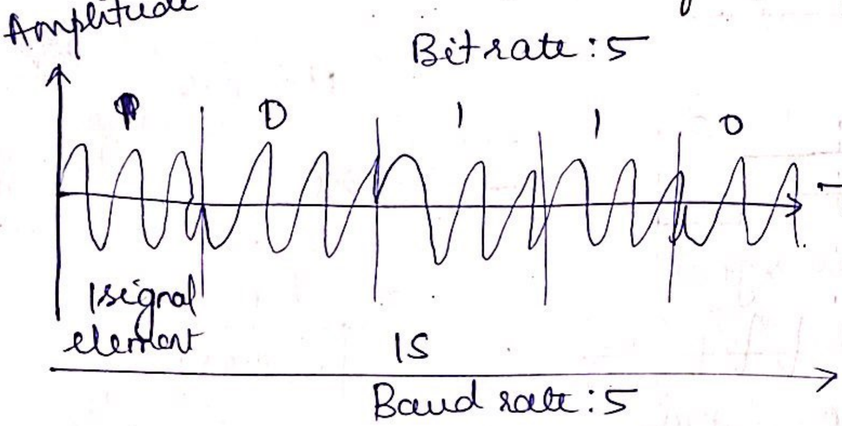
Phase shift keying

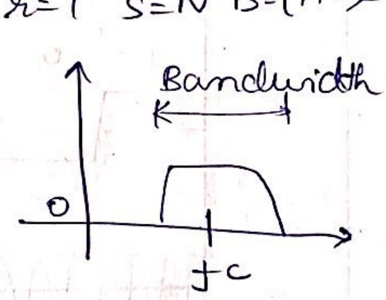
The phase of the carrier is varied to represent 2 or more different signal elements. Both peak amplitude and frequency remain constant as phase changes.

PSK is common than ASK or FSK.

Binary PSK (BPSK)

Simplest PSK is binary PSK, we have 2 signal elements one with phase of 0°, other with a phase of 180°.





Binary PSK is simple as binary ASK with adv- less susceptible to noise

ASK – bit detection is the amplitude of the signal

PSK – it is phase.

Noise can change the amplitude easier than it can change the phase.

PSK is less susceptible to noise than ASK. PSK is superior to FSK – no need of 2 carrier signals.

PSK needs more sophisticated hardware to distinguish between phases.

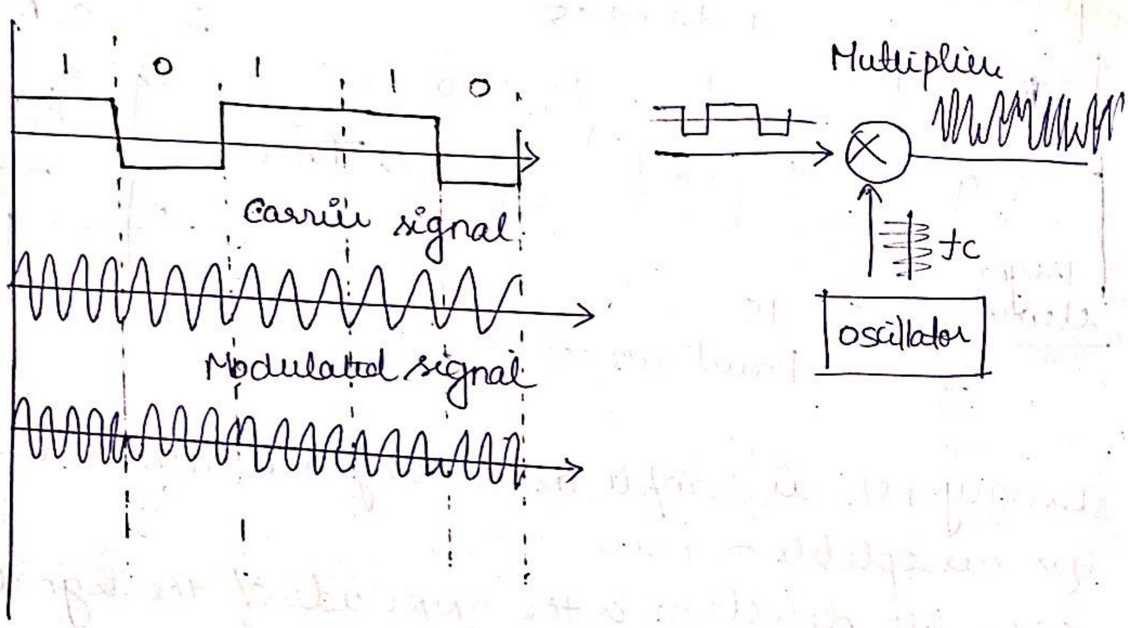
Bandwidth

Bandwidth is same as that for binary ASK, but less than that for BFSK. No bandwidth is wasted for seperating 2 carrier signals.

Implementation

Implementation of BPSK is same as that of ASK i.e the signal element with phase 180° is seen as the complement of signal element with phase 0°.

Implementation of BASK



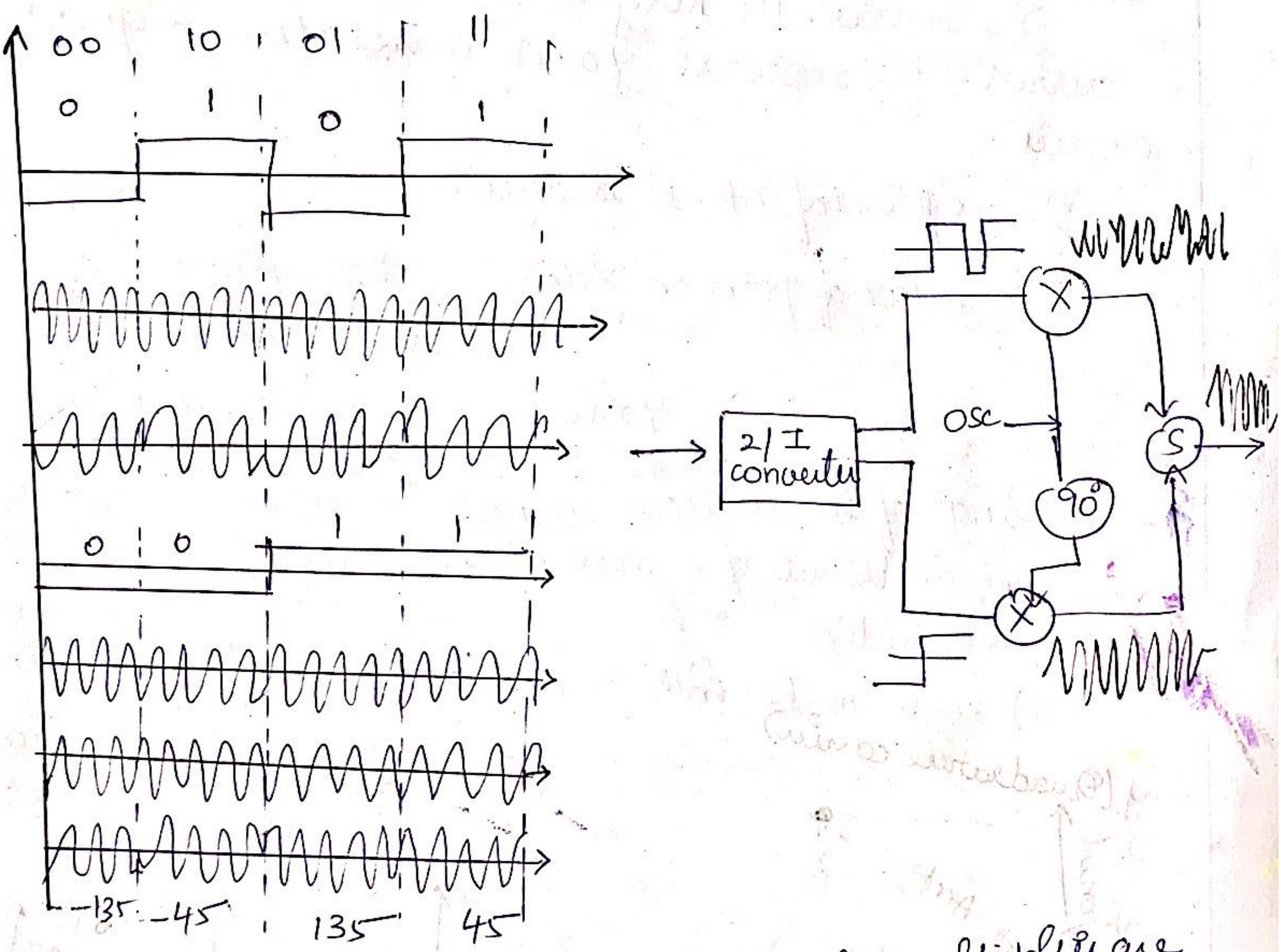
Quadrature PSK (QPSK)

Simplicity of BPSK enticed designers to use 2 bits at a time in each signal element decreasing the baud rate and eventually the required bandwidth

It is called as quadrature PSK or QPSK because it uses 2 seperate BPSK modulation: one is in phase, and other quadrature (out of phase)

The incoming bits are first passed through a serial to parallel conversion that sends one bit to one modulator and next bit to the modulator.

If the duration of each bit in the incoming signal is T, the duration of each bit sent to the corresponding BPSK signal is 2T. BPSK signal has one - half the frequency of original signal.



2 composite signals created by each multiplier are sine waves with the same frequency, but different phases when they are added, result is another sine wave, one of the possible phases : 45°, -45°, 135°, and -135°. L= 4, 2 signal element (r = 2).

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Constellation Diagram can define the amplitude and phase of a signal element when we are using 2 carrier (one in phase and one quadrature).

A signal element type is represented as a dot. The bit or combination of bits can carry is written next to it.

2 axis

1) 2 the axis. The horizontal X axis is related to in-phase carrier the vertical Y axis is related to quadrature carrier.

2) Four pieces of info is deduced.

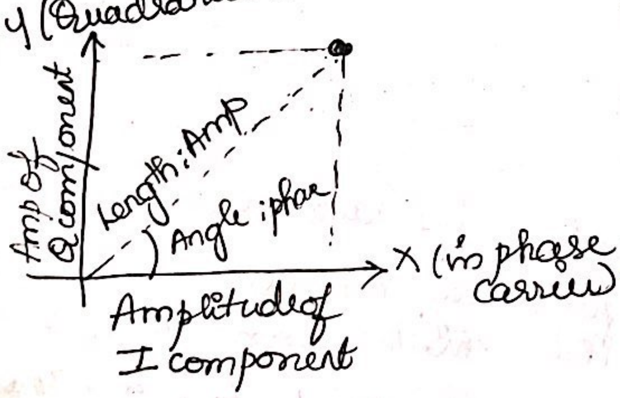
3) projection of point on X axis – peak amplitude of in phase component

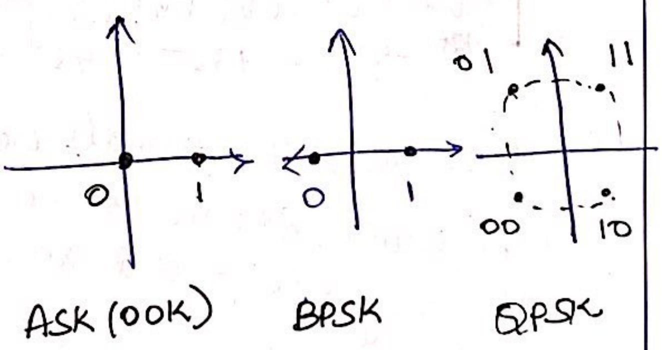
Y axis – peak amp of quadrature component.

4) length of line (vector) connects the point to origin is peak amplitude of signal element (combination of X and Y components)

5) angle makes with x axis is the phase of signal element.

3 constellation diagram





ASK, using only an in phase carrier. 2 points should be on the X axis. Binary 0 has amp of 0V
1 has amp of 1V.
Points are located at the origin and at 1V.

BPSK uses only in phase carrier. use a polar NRZ signal for modulation.
2 types of signal elements,
1) one with amp 1 and other with −1
2) It creates 2 diff signal one with amp 1V in phase and other with amp 1V and 180° out of phase

QPSK 2 carrier - one in phase and other quadrature
1) point 11 is made of 2 combined signal element, both with amp of 1V.
2) the amp of final signal element sent for third bit data element is $2^{1/2}$. and this 45°.
3) All 3 elements have an amp of $2^{1/2}$ but phases are diff (45°, 135°, −135°, & −45°).
4) Amp of carrier is 1 / ($2^{1/2}$) to make amp of 1V

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