T.Y. Diploma : Sem. V
[ET/EN/EJ/EX/ED/EI]

Digital Communication

Time: 3 Hrs.] Prelim Question Paper Solution [**Marks : 100**

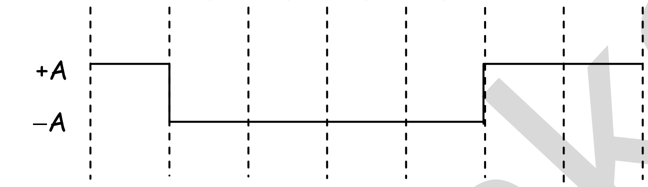
Q.1(a) Attempt any THREE of the following : [12]

Q.1(a) (i) List and explain different types of errors in data communication [4]

(A) Types of error :

i) Single bit error or one bit error
ii) Burst error

Single bit error : if only 1 bit in transmitted bit sequence is changed because of noise then it is called as single bit error.



Received bit seq. 1 0 0 1 0 1 1 $\neq$ code of C

Burst error : If 2 or more than 2 bits are changed because of noise then it is called as Burst error.

Q.1(a) (ii) List various properties required for line codes. [4]

(A) Properties of line codes

i) DC content
ii) Signal power

i) For a good line code DC content should be as low as possible. Split phase Manchester consist of very low DC component where as NRZ – L and unipolar NRZ consists of large DC contents. If there is channel is less noisy then we use split phase Manchester.
ii) For a good line code signal power should be as high as possible so that noise required to corrupt the large. NRZ – L and unipolar NRZ has high signal power than split phase Manchester and RZ.

Q.1(a) (iii) Write bandwidth requirement for DPSK, QAM, QPSK, BPSK. [4]

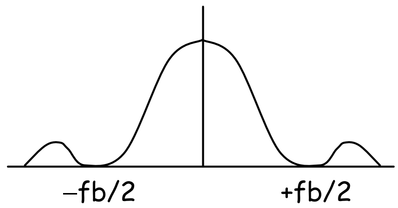
(A) Bandwidth requirement of

(i) DPSK

Symbol duration = Ts = 2Tb

$$\frac{1}{Fs} = \frac{2}{fb}$$

$$fs = \frac{fb}{2}$$



$$\therefore \text{ Bandwidth} = \left(\frac{fb}{2} - \left(-\frac{fb}{2}\right)\right) = fb$$

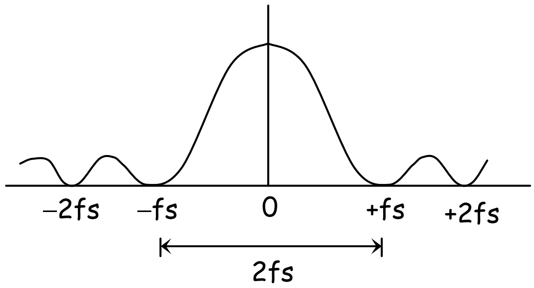
Bandwidth = fb

(ii) QAM

Bandwidth = fs – (–fs) = 2fs

$$= \frac{2}{Ts}$$

$$= \frac{2}{N.Tb}$$

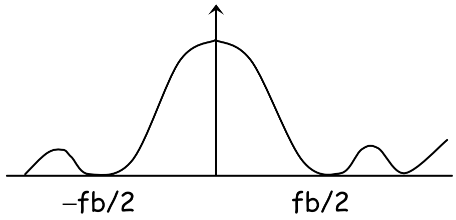


$\therefore$ Bandwidth = $\frac{2Fb}{N}$

(iii) QPSK

Bandwidth = fb/2 – (–fb/2)

bandwidth = fb

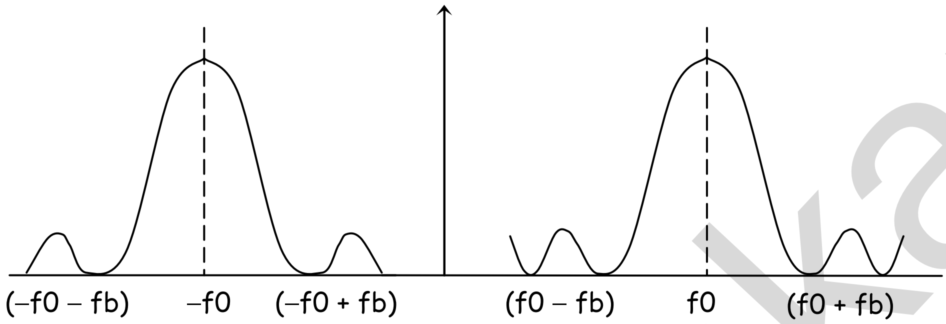


(iv) BPSK

Bandwidth = (f0 + fb) – (f0 – fb)

= f0 + fb – f0 + fb

Babdwidth = 2fb



Q.1(a) (iv) List and explain different types of frequency hopping. **[4]**

(A) Types of frequency hopping:-

1) Slow frequency hopping
2) Fast frequency hopping

Slow frequency hopping:-In slow frequency hopping the symbol rate R_s of the MFSK signal is an integer multiple of the hop rate R_n that means several symbols are transmitted corresponding to each frequency hop.

Each frequency hop:- several symbols

Here frequency hopping takes place slowly and thus

Hop rate R_h > Symbol rate R_s

Fast frequency hopping:- In fast frequency hopping, multiple frequencies or hops are used to transmit one symbol. That is each symbol-> several hops. So several frequencies changes for one symbol such that Symbol rate R_s > Hop rate R_h.

Q.1(b) Attempt any ONE of the following : **[6]**

Q.1(b) (i) List the different error detecting methods. Describe checksum method with suitable example. **[6]**

(A) Different error detection schemes

- Repetition codes
- Parity bits
- Checksums
- Cyclic redundancy checks (CRCs)
- Cryptographic hash functions

Checksums

Most of error detection techniques uses a process known as checksum to generate an error-detection character. The character results from summing all the bytes of a messageM together, discarding and carry over from the addition. Again, the process is repeated at the receiver and the two checksums are compared. A match between receiver checksum and transmitted checksum indicates good data. A mismatch indicates an error has occurred. This method, like CRC, is capable of detecting single or multiple errors in the message. The major advantage of checksum is that it is simple to implement in either hardware or software. The drawback to checksum is that, unless you use a fairly large checksum (16- or 32-bit instead of 8-bit), there are several data-bit patterns that could produce the same checksum result, thereby decreasing its effectiveness. It is possible that if enough errors occur in a message that a checksum could be produced that would be the same as a good message. This is why both checksum and CRC error-detection methods do not catch 100% of the errors that *could* occur, they both come pretty close.

Example : What is the checksum value for the extended ASCII message "Help!"?

Solution :
The checksum value is found by adding up the bytes representing the Help! characters:

01001000 H
01100101 e
01101100 l
01110000 p
00100001 !
00010000 Checksum

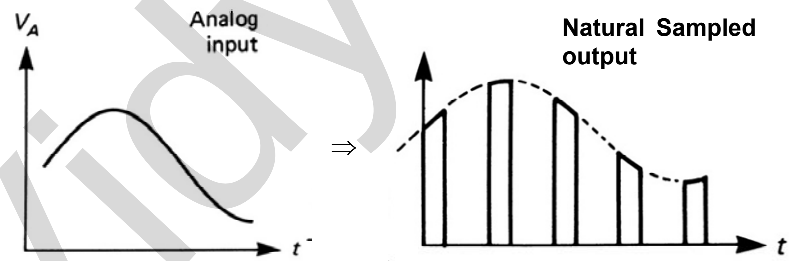
checksum error-detection process that uses the sum of the data stream in bytes. The hardware solution relies once more on exclusive OR gates, which perform binary-bit addition. Each 8-bits of data are exclusive ORed with the accumulated total of all previous 8-bit groups. The final accumulated total is the checksum character.

Q.1(b) (ii) Define sampling theorem. List types of sampling techniques. Draw the naturally sampled signal. [6]

(A) Sampling Theorem

"A bandlimited signal can be reconstructed exactly if it is sampled at a rate atleast twice the maximum frequency component in it."

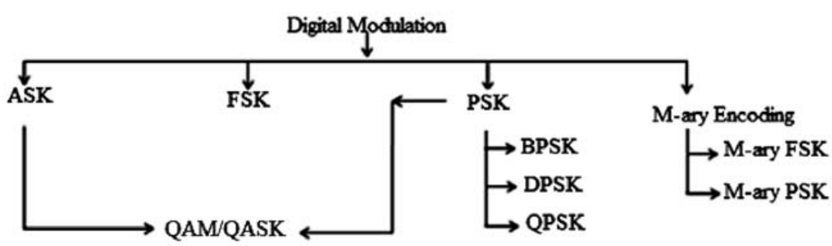
There are two types of sampling technique :
(1) Natural sampling
(2) Flat top sampling



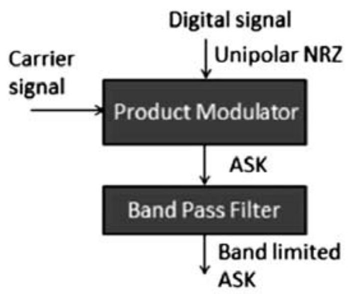
Q.2 Attempt any TWO of the following : [16]

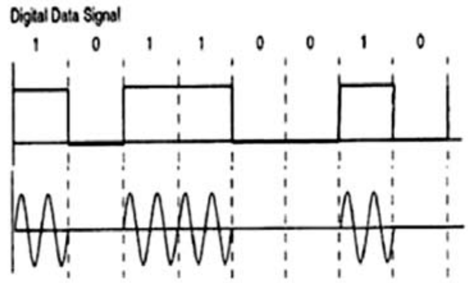
Q.2(a) List different modulation techniques and explain Amplitude shift keying in detail. [8]

(A)



Amplitude Shift Keying (ASK) is the digital modulation technique in which the amplitude of the sinusoidal carrier will take one of the two predetermined values in response to 0 or 1 value of the digital input modulating message signal.





$V_{ASK}(t) = d \sin(2\pi f_c t)$

Explanation

- Amplitude shift keying is the simplest form of digital modulation. Here the carrier is a sine wave of frequency (f_c).
- The carrier signal can be mathematically expressed as e c = sin (2nfct)
- The digital signal from the information source is a unipolar NRZ signal which acts as the modulating signal. The ASK modulator is nothing but a multiplier followed by a band pass filter as shown in above figure.
- Due to multiplication, the ASK output will be present only when a binary '1' is to be transmitted and when the digital input is '0' then we get zero output as shown in the waveform above.
- From the waveform analysis we can conclude that when a binary '1' is to be sent the carrier is transmitted and when binary '0' is to be sent then the carrier is not transmitted.
- Mathematically ASK signal can be expressed as;

  $V_{ASK}(t) = d \sin(2\pi f_c t)$

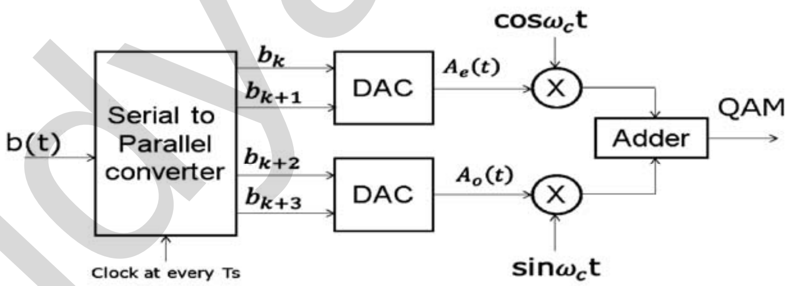
  where d = data bit which can take values 1 or 0

  Therefore, $V_{ASK}(t) = \sin(2\pi f_c t)$ When d = 1

  $V_{ASK}(t) = 0$ When d = 0

Q.2(b) Draw the block diagram of QAM generation system and explain with waveforms. [8]

(A)



$V_{QAM}(t) = A_e(t)\cos\omega_c t + A_o(t)\sin\omega_c t$

Explanation

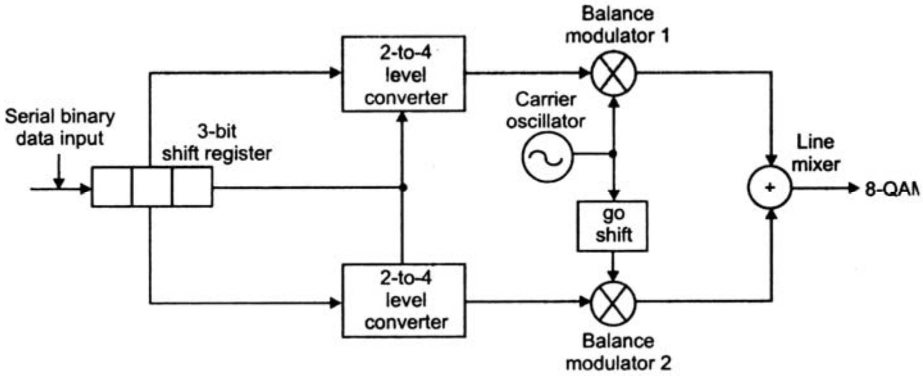
Figure shows transmitter for 4 bit QAM system. The input bit stream is applied to a serial to parallel converter. Four successive bits are applied to the digital to analog converter. These bits are applied after every Ts second. Ts is the symbol period & Ts = 4Tb. Bits B_k & B_{k+1} are applied to upper digital to analog converter. & B_{k+2}, B_{k+3} are applied to lower D to A converter. Depending upon the two input bits, the output of D to A converter takes four output levels. Thus Ae (t) & Ao (t) takes 4 levels depending upon the combination of two input bits. Ae (t) modulates the carrier cos ($2\pi f_c t$) and Ao (t) modulates sin ($2\pi f_c t$).

The adder combines two signals to give QAM signal. It is given as,

S (t) = Ae (t) cos ($2\pi f_c t$) + Ao (t) sin ($2\pi f_c t$)

(OR)

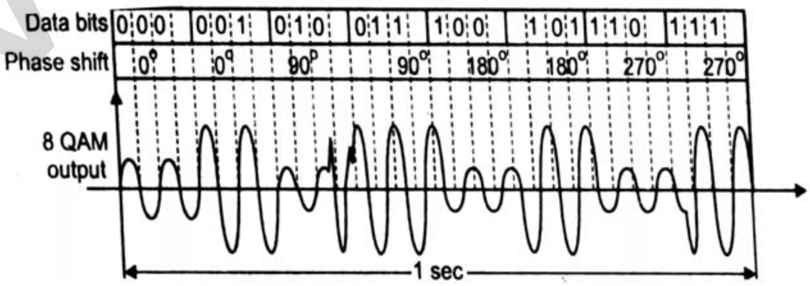
Diagram



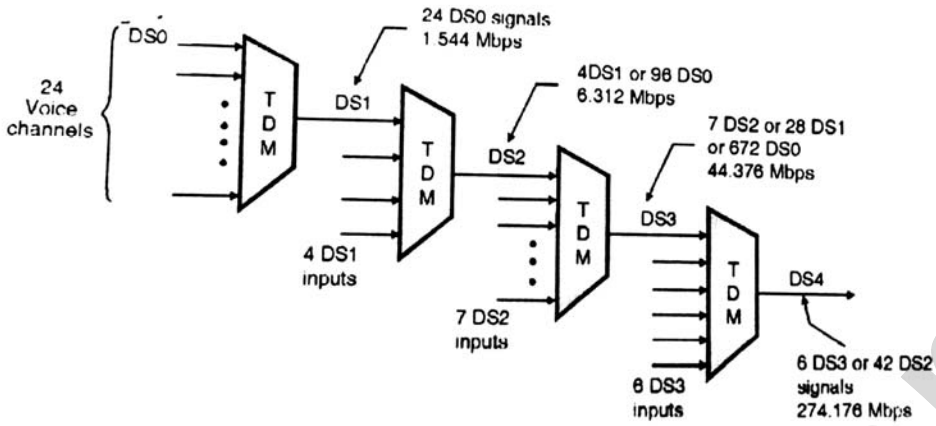
Explanation

- One of the most popular modulation techniques used in modems for encoding more bits per band in QAM.
- It uses both AM and PM of a carrier.
- In addition to producing different phase shifts the amplitude of the carrier is also varied.
- A popular version of QAM is known as 8-QAM.
- This system uses four different phase shifts as in QPSK and two carrier amplitudes.
- With four possible phase shift and two different carrier amplitudes, a total of eight different states can be transmitted.
- With right states, 3 bit can be encoded for each baud or symbol transmitted.
- Each 3 bit binary word transmitted uses different PM/AM combination.
- **3-bit Shift Register:**
  - The binary data to be transmitted is shifted serially into 3-bit shift register.
- **2-to-4 level converter:**
  - 2-to-4 level converter is a circuit that translates a pair of binary inputs into one of four possible d. c. output voltage levels.
  - A 2-to-4 level converter is basically a simple D/A converter. The result is four equally spaced voltage levels.
- **Balance Modulator:**
  - The voltage levels are applied to the two balance modulators fed by the carrier and 90° phase shifter as in a QPSK modulator
  - Each balanced modulator produces four different output PM/AM combinations.
- **Linear Mixer:**
  - When the outputs of balance modulator are combined in the linear mixer, eight different PM/AM combinations are produced.
  - This result is 8-QAM.

Waveform



Q.2(c) Describe the North American Digital Multiplexing Hierarchy with neat diagram. [8]
(A)



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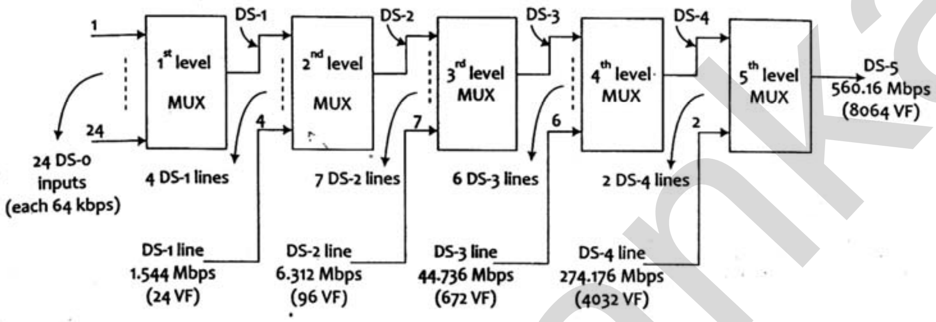
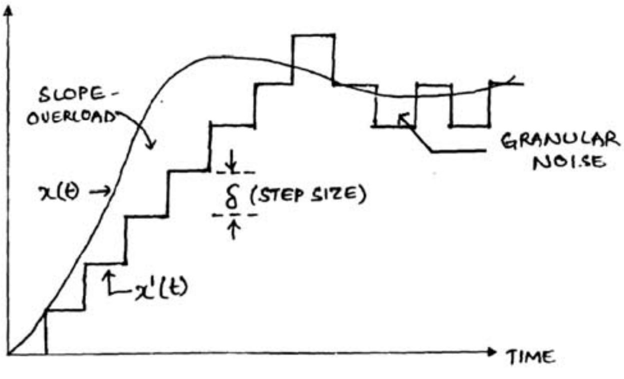


Explanation

The first digital signal in true sense is the PCM voice signal. A PCM voice signal represents 64kbits/sec. i.e. 8000 sample /second* 8 bits per samples. Such a signal is called as digital signal at level zero (DS0). It is also called as T1 signal. Due to 8000 sample/second, sampling rate, the time duration between adjacent samples will be 125 μsec. But practically DS0 signal is not transmitted because most of the telephone lines are analog. Hence in telephone central office, the subscriber analog line is passed through an anti-aliasing filter. The band limited signal is applied to a codec, which convert it into DS0 signal. 24 DS0 lines are multiplexed into a DS1. The telephone companies implement TDM through the hierarchy of digital signals. This is called as digital signal service. Multiplexed signal is converted into a frame at the DS1 or T1 level.

Q.3 Attempt any FOUR of the following : [16]

Q.3(a) Explain slope Overload and granular noise with respect to delta modulation [4]
(A)



The delta modulation has two major drawbacks as under:

(i) Slope overload distortion : This distortion is arises because of large dynamic range of the input signal. As can be observed from figure the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it, the steep size 'A' becomes too small for staircase signal x'(t)to follow the step segment of x(t). Hence, there is a large error between the staircase approximated signal and the original input signal x (t). This error or noise is known as slope overload distortion. To reduce this error, the step size must be increased when slope of signal x (t) is high. Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is also known as Linear Delta Modulator (LDM).

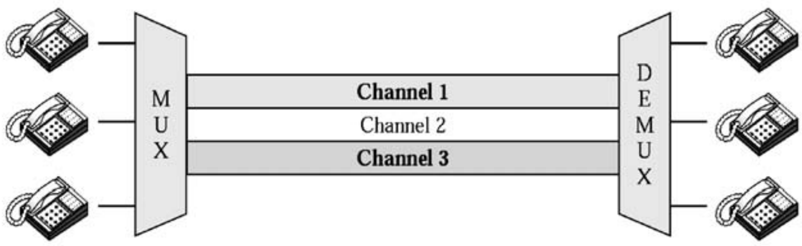
(ii) Granular or Idle Noise : Granular or Idle noise occurs when the step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount (A) because of large step size figure shows that when the input signal is almost flat, the staircase signal x'(t) keeps on oscillating by ±A around the signal. The error between the input and approximated signal is called granular noise. The solution to this problem is to make step size small.

Therefore, a large step size is required to accommodate wide dynamic range of the input signal (to reduce slope overload distortion) and small steps are required to reduce granular noise. In fact, Adaptive delta modulation is the modification to overcome these errors.

Q.3(b) Compare ASK with FSK **[4]**
(A)

Parameters	ASK	FSK
Variable characteristics	Amplitude	Frequency
Bandwidth (Hz)	f_b	$\|f1- f0\|+f_b$
Noise immunity	Low	High
Error probability	High	Low
Performance in presence of noise	Poor	Better than ASK
Complexity	Simple	Moderately complex
Bit rate	Suitable up to 1OObits /sec	Suitable up to about 1200 bits/sec
Detection method	Envelope	Envelope

Q.3(c) Explain principle of frequency division multiplexing and compare FDM and CDM techniques. **[4]**
(A) Principle of frequency division multiplexing



FDM means total range of frequency is divided into number of frequency slots. Each slots of frequency is allotted to each channel. Various channels of different frequencies combined, transmitted through single wire and separated at receiver with the help of De-multiplexer.

FDM can be applied when the bandwidth of the link is greater than the combined BW of the signal to be transmitted. These modulated signals are than combined into a single composite signal that can be transported by the links. Carrier frequency is separated by sufficient BW to accommodate the modulated signal .These BW range are the channel through which the various signals travels. Channels must be separated by guard bands to prevent signals from overlapping.

Comparison of FDM & CDM techniques

	FDM	CDM
1)	Synchronization is not required.	Synchronization is not necessary.
2)	Overall bandwidth is shared among many stations.	Sharing of bandwidth and time takes place.
3)	Due to non-linearity of devices inter modulation products are generated due to interference between adjacent channels.	Both type of interference will be present.
4)	Code word is not required.	Code words are required.
5)	Guard bands between adjacent channels are necessary.	Guard bands and guard times both are necessary.

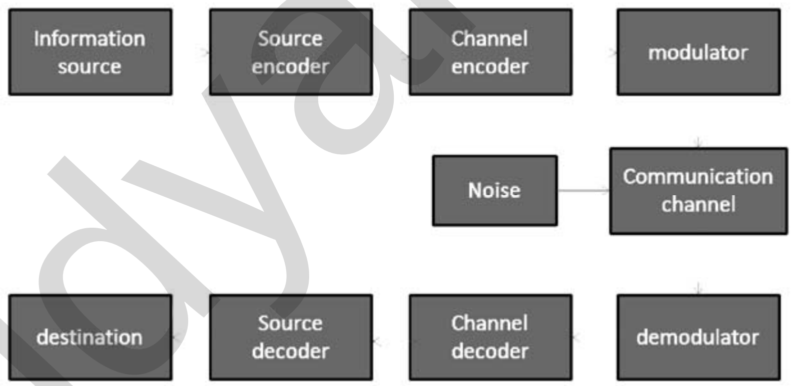
Q.3(d) List different advantages of PSK modulation. **[4]**

(A) Advantages of PSK:

i) BPSK has a bandwidth which is lower than that of a BFSK signal.
ii) BPSK has the best performance of all the systems in presence of noise.
iii) It gives the minimum possibility of error.
iv) BPSK has very good noise immunity.
v) It gives less error of probability.
vi) Used for high bit rate than 1800 bits/sec.

Q.3(e) Draw the Block diagram of digital communication system and explain in detail. **[4]**

(A)



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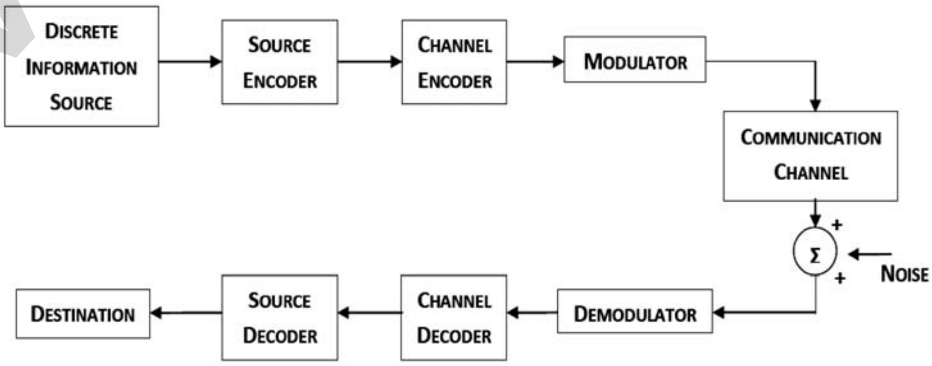


1) Information source - it may be in 332analog forming. Output of microphone gives analog signal. And if source is computer data then it is a digital form.

2) Source encoder - the source encoder converts the signal produced by the information source into DataStream. If i/p signal is analog it can be converted in to digital form using A to D converter. If the i/p to the source encoder is a stream of symbols it can be converted into a stream of 1s and 0s using some coding mechanism.
3) Channel Encoder - if we have to encode the information covertly, even if errors are introduced in the medium. We need to put some additional bits in the source, so that additional information can be used to detect and coolest the errors, this process of adding bits is done by channel encoder. In channel encoding redundancy is introduced so that at the receiving end the redundancy is introduced so that at the receiving end redundant bit can be used for error detection and error correction.
4) Modulator - here the modulation is done for transferring the signal, so that the signal I can be transmitted through the medium easily.
5) Channel - it is the medium through winch the o/p of modulator along with some noise is transmitted and gives to demodulator. This channel is called discrete channel because its input as well as o/p both are in discrete nature.
6) Demodulator - it performs inverse operation than that of modulator.
7) Channel decoder- it checks the received bits and also detect and correct the errors, using additional data introduced by channel encoder.
8) Source decoder - it converts the bit stream in to actual information, here digital to analog conversion is done if the symbols are coded in to is and os at the source decoder the bits are converted in to symbols.

Q.4(a) Attempt any THREE of the following : [12]

Q.4(a) (i) What is need for delta modulation? Give its advantages and disadvantages and applications. [4]

(A) Delta Modulation (DM)

Need :

- In PCM 'N' number of bits are transmitted per quantized sample which asks for large channel Bandwidth and signaling rate.
- This disadvantage can be overcome by using DM.
- DM transmits only one bit per sample instead of 'N' therefore it extensively reduces signaling rate and channel Bandwidth.

Advantages

i) One bit codeword for output.
ii) Simplicity of design for transmitter and receiver.
iii) Low signaling rate
iv) Low channel Bandwidth

Disadvantage

i) Slope overload present
ii) Granular Noise

Application

i) Satellite transmission System
ii) Digital Communication

Q.4(a) (ii) Discuss Shannon's theorem in brief. [4]

(A) Shannon's Theorem

- Also known as source coding theorem.
- Statement – Given a discrete memoryless source of entropy H, the average code word length for any source coding is bounded as $L \geq H$.
- Explanation : For any source encoder, code efficiency is given as

$$\eta = \frac{H}{L}100\%$$

$H \rightarrow$ entropy, $H = \sum_{i=1}^{M} p_i \log_2 \frac{1}{p_i}$

$L \rightarrow$ average code word length, $L = \sum_{i=1}^{M} p_i \ell_i$

ℓ_i is length of the code in bits

As such if $L \geq H$, minimum value of L i.e. $L_{min.} = H$. Under such circumstances, efficiency will be maximum. To increase efficiency variable length coding is done.
eg. Huffman code.

Q.4(a) (iii) Compare between FHSS and DSSS (4 points) [4]

(A) Comparison between DSSS and FHSS

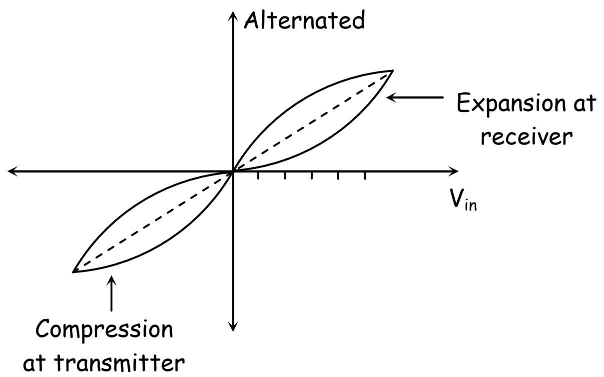
	Parameters	DSSS	FHSS
i)	Definition	PN sequence of large bandwidth is multiplied with narrow band data signal.	Data bits are transmitted in different frequency slots which are changed by PN sequence.
ii)	Chip rate	It is fixed, $R_c = \frac{1}{T_c}$	$R_c = \max(R_h, R_s)$
iii)	Modulation technique	BPSK	M-ary FSK
iv)	Processing gain	$PG = \frac{T_b}{T_c} = N$	$PG = 2^t$
v)	Error probability	$P_e = \frac{1}{2}\text{erfc}\sqrt{\frac{E_b}{JT_c}}$	$P_e = \frac{1}{2}e^{-rb} R_c / 2$
vi)	Acquisition time	Long	Short
vii)	Effect of distance	This system is distance relative.	Effect of distance is less.

Q.4(a) (iv) What is companding? Draw the companding curves for PCM system. [4]

(A) Companding

By keeping step size constant we can perform non uniform quantization. This can be achieved by amplifying low level signals and alternating high level signal is called as compression at the receiver side, the signal is alternated at low level and amplified at high level is called as expansion.

The compression of signal at transmitter and expansion at receiver is combinely called as companding.



Q.4(b) Attempt any ONE of the following : [6]

Q.4(b) (i) Explain PN sequence generation in detail [6]

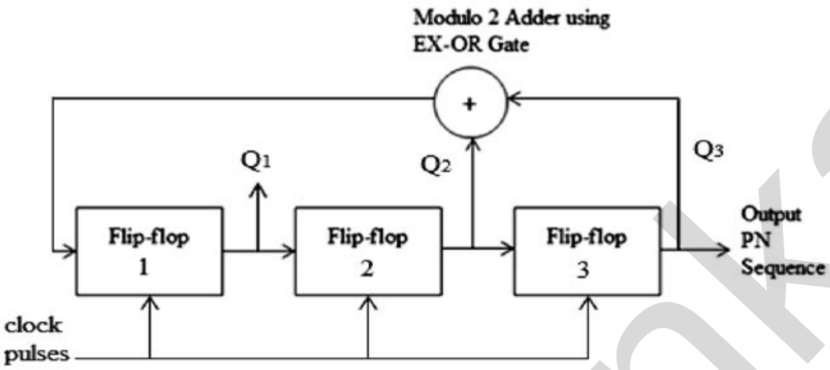
(A) Note: Diagram & generation table with four flip-flops should also be considered

A PN sequence is defined as a pseudorandom coded sequence of Is and 0s with certain auto correlation properties.

Maximum length of PN Sequence 'L' is the no. of bits in a PN sequence and it depends upon the number of flip-flops 'n' used for the PN Sequence generator and given as

$L = 2^n - 1$

The block diagram for 3 bit that is 7 bit length of PN sequence generator is as shown with feedback taps [3, 1]



Let initial value of shift register output

i.e. $Q_3\ Q_2\ Q_1$ = 0 0 1 (any initial value can be taken)

Table is as shown below–

Clock Pulse	Shift Register Outputs			EX-OR Gate Output	PN Sequence
	Q_3	Q_2	Q_1	$Q_3 \oplus Q_2$	Q_3
0	0	0	1	$0 \oplus 0 = 0$	0
1	0	1	0	$0 \oplus 1 = 1$	0
2	1	0	1	$1 \oplus 0 = 1$	1
3	0	1	1	$0 \oplus 1 = 1$	0
4	1	1	1	$1 \oplus 1 = 0$	1
5	1	1	0	$1 \oplus 1 = 0$	1
6	1	0	0	$1 \oplus 0 = 1$	1
7	0	0	1	$0 \oplus 0 = 0$	0
8	0	1	0	$0 \oplus 1 = 1$	0
9	1	0	1	$1 \oplus 0 = 1$	1
10	0	1	1	$0 \oplus 1 = 1$	0

The actual data repeats again and again as shown in above table with pseudorandom data bits in between to improve security of the data. Here the PN coded data will be of 7 bit.

The PN Sequence obtained at the output Q3 of filp–flop 3 is

'0 0 1 0 1 1 1 0 0 1 0'

Q.4(b) (ii) Generate CRC code for data word 1101010011 the divisor is 01011. [6]

(A) [Note - The divisor given is 01011, it should be taken as 1011 else it won't be solved]

To generate the CRC code as divisor is 4 bit, add three zeros to the data and get the remainder as shown below. The last three bit remainder is the required CRC send along with the data to be transmitted.

```
             1 1 1 1 0 1 1 0 1
1 0 1 1 ) 1 10 10 1 0 1 1000
          -1011
            1100
           -1 011
              1111
             -10 11
                1 000
               -10 11
                  0 1111
                  -  10 11
                      1 000
                     -1 0 1 1
                        0 1100
                         - 1 0 1 1
                             1 1 1
```

The CRC Data is 111.

The final coded data transmitted will be $\boxed{1\,1\,0\,1\,0\,1\,0\,1\,1}\boxed{1\,1\,1}$

Q.5 Attempt any TWO of the following : [16]

Q.5(a) Describe the principle of time division multiplexing with suitable sketch. [8]

(A) Definition:

- The process called multiplexing is used in order to utilize common transmission channel or medium to transmit more than one signals simultaneously.
- Two methods which are generally employed are frequency division multiplexing (FDM) and time division multiplexing (TDM).
- In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one–by–one.
- Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel. The TDM principle is illustrated in Figure 1.

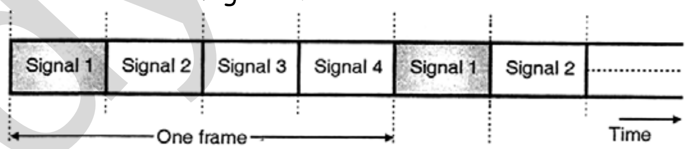


Fig. 1: Principle of TDM.

- As shown in the Figure 1 one transmission of each channel completes one cycle of operation called as a "frame".
- The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the for the digital signal multiplexing.
- The concept of TDM will be more clear if you refer to Figure 2.

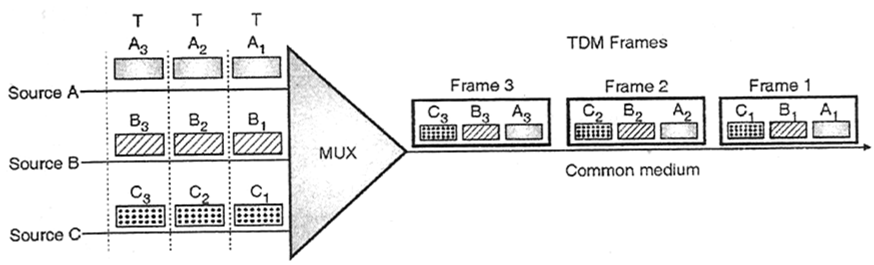


Fig. 2 : TDM system.

- The data flow of each source (A, B or C) is divided into units (says A_1, A_2 or B_1, C_1 etc.)
- Then one unit from each source is taken and combined to from one frame. The size of each unit such as A_1, B_1 etc. can be 1 bit or several bits.
- Figure 3 shows the frames of TDM signal. For 3 inputs being multiplexing, a frame of TDM will consist of 3 units i. e. one unit from each source.
- Similarly for n number of inputs, each TDM frame will consist of n units.

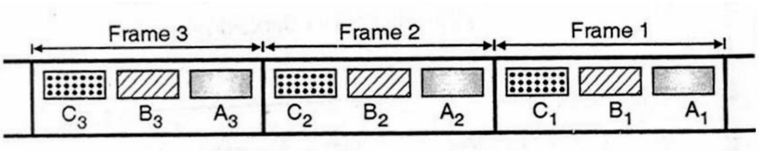


Fig. 3 : TDM frames.

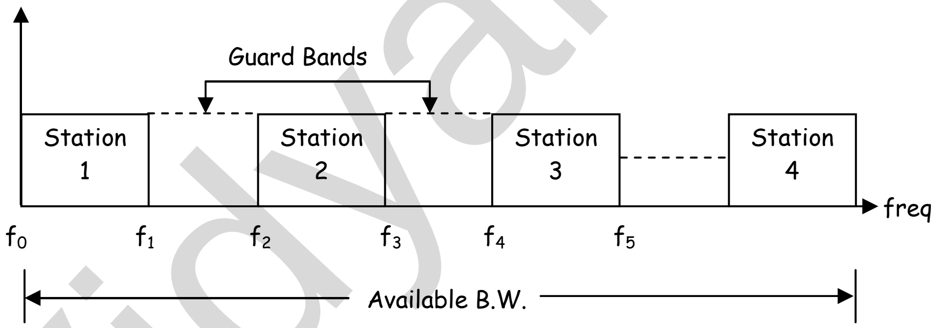
- The TDM signal in the form of frames is transmitted on the common communication medium.

Data rate :
- For a TDM, the data rate of the multiplexed signal is always n times the data rate of individual sources, where n is the number of sources.
- So, if three sources are being multiplexed, then the data rate of the TDM signal is three times higher than the individual data rate.
- Naturally the duration of every unit (A_1 or B_1 etc.) in TDM signal is n times shorter than the unit duration before multiplexing.

Q.5(d) Explain FDMA system with schematic diagram. Compare FDMA and TDMA. [8]

(A) FDMA (Frequency division multiple access)
- This technique is based an FDM tech
- In FDMA available bandwidth is shared by all the station
- Each station is allocated with a particular frequency band to send its data.

Example



- In above diagram frequency band f_0 to f_1 is reserved for station 1, f_2, to f_3 is reserved for s_2 etc.
- Guard bands are also provided in between the adjacent frequency slot to avoid crosstalk.

Features
- Overall channel band width is shared by multiple users therefore no of users can transmit their information simultaneously.
- Guard band are provides because :
  i) To avoid cross talk
  ii) Impossible to achieve ideal filtering to separate different users
- Power efficiency different users.
- Synchronization is not required

Advantages :
- All stations can operate continuously and simultaneously.
- Power required for transmission depends on the no of channels being transmitted.

- SNR is improved because of Fm
- Synchronization is not required.

Disadvantages :
- Each channel on earth station can used only a part of total satellite B.W.
- Inspite having guard bands adjacent channel interference is present.
- Because of the use of Fm, Bw required & therefore less no of channels can be accommodated in available Bandwidth.

Parameter	FDMA	TDMA
Technique	Sharing of overall B.W. of satellite transponder	Sharing of overall time os satellite transponder
Syndrome	Not required	Required
Code word	Not required	Not required
Power efficiency	Less	Full power n is poss
Guard time & Guard band	Guard band required	Guard time required

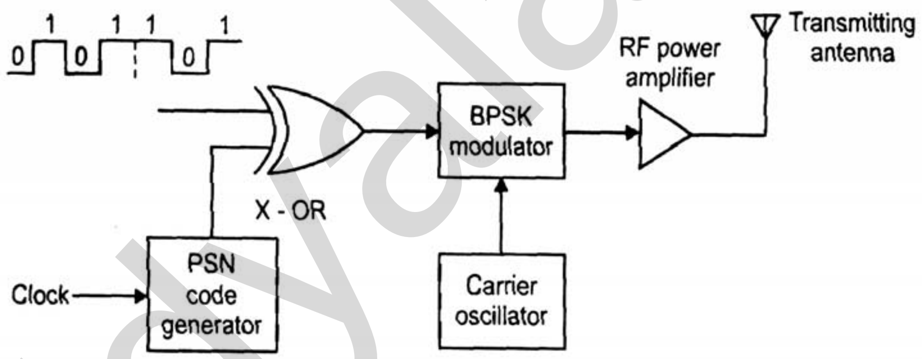
Q.5(c) Describe the direct sequence spread spectrum techniques with the help of block diagram and state its advantages [8]

(A) Note: DSSS receiver with explanation also should be considered and marks to be given

Direct sequence spread spectrum (DSSS)

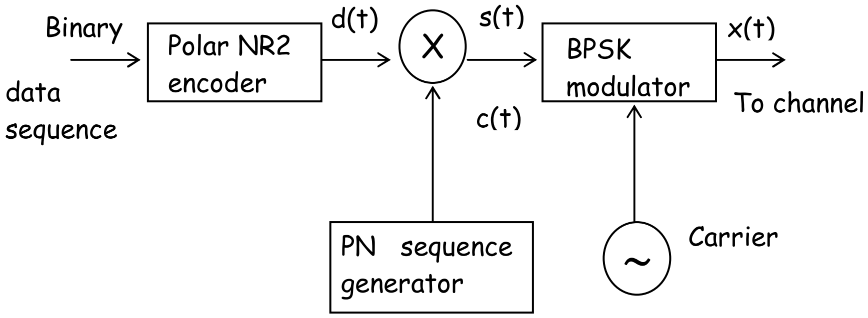
In direct sequence, the serial binary data is mixed with a higher frequency pseudorandom binary code at a faster rate and the result is used to phase-modulate a carrier.

DSSS Transmitter:



Explanation:
- A block diagram of DS transmitter is shown in figure
- The serial binary data is applied to an X-OR gate along with a serial pseudorandom code that occurs faster than binary data.
- The signal developed at the output of the X-OR gate is then applied to a BPSK modulator.
- The carrier phase is switched between 0° and 180° by the 1's and 0's of X-OR output.
- The signal phase modulating carrier, being much higher in frequency than the data signal causes the modulator to produce multiple widely spaced sidebands whose strength is such that the complete signal takes up a great deal of the spectrum. Thus the signal is spread.
- Also because of its randomness, the resulting signal is appears to be nothing more than wideband noise to a conventional narrow band receiver.
- One bit time for the pseudorandom code is called a chip and the rate of the code is called the chipping rate.
- The chipping rate is faster than the data rate.

(OR)



The averaging system reduces the interference by averaging at over a long period. The DSSS system is a averaging system. This technique can be used in practice for transmission of signal over a band pass channel (e.g. satellite channel). For such application the coherent binary phase shift (BPSK) is used in the transmitter and receiver.

The binary sequence b(t) is given to the NRZ encoder. The b (t) is converted NRZ signal d(t). The NRZ signal d(t) is used to modulate the PN sequence c(t) generated by the PN code generator.
The multiplier multiply the signal b(t) * c(t) = s(t). The s(t) signal is given to binary PSK modulator.
The modulated signal at the output of product modulator or multiplier i.e. s(t) is used modulate the carrier for BPSK modulation.

The transmitted signal x(t) is thus DSSS signal.
Product modulator output = s(t)
$s(t) = d(t) * c(t)$

The BPSK carrier signal is given by $\sqrt{2Ps} \sin 2\pi f_c t$.
The output of BPSK modulator x(t) is transmitted $x(t) = s(t) * \sqrt{2Ps}\, 2\pi f_c t$. But m (t) = ± 1

Therefore $x(t) = \pm \sqrt{2Ps} \sin 2\pi f_c t$
The phase shift of x (t) of x (t) is 0_0 to + m (t) at is 180_0 corresponding to a negative m (t).

Advantages of Spread Spectrum (SS): (any two)
1. Unauthorized listening is prevented.
2. SS signals are highly resistant to the jamming.
3. Unintentional interference occupying the same band is greatly minimized and in most cases virtually eliminated.
4. Many users can share a signal band with no interference.
5. With SS, more signals can use a band than with any other type of modulation and multiplexing.
6. Resistant to fading.
7. The pseudorandom code makes it possible to accurately determine the start and end of a transmission.
8. Superior method for radar.

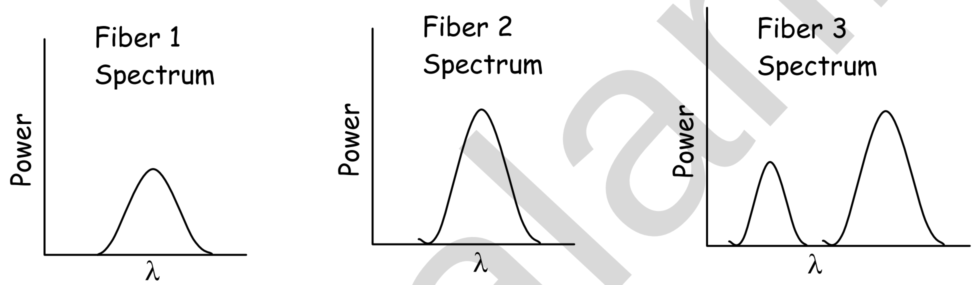
Q.6Attempt any FOUR of the following : [16]
Q.6(a) Give the advantages and disadvantages of digital communication. [4]
(A) Advantages of digital communication
i) Noise immunity is more than analog communication.
ii) Digital communication supports error detection and correction techniques.
iii) It is easy to regenerate the digital signal than analog.
iv) Digital signals are easy to store and manipulate.
v) Digital communication is computable with advance data processing technique like digital technique process image processing.
vi) Cost of digital communication system is low.

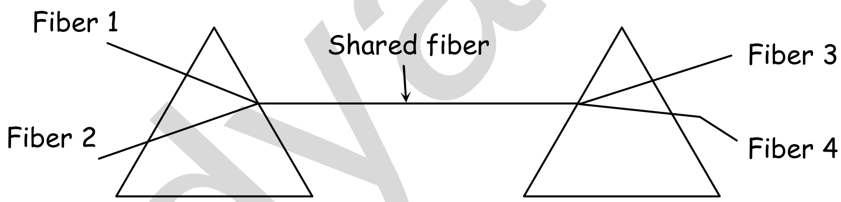
Disadvantages
i) The data rate of digital communication is very high.
ii) Loss of information.

Q.6(b) Compare QPSK and QASK (4 points) **[4]**
(A) Comparison between QPSK and QASK

	Parameters	QPSK	QASK
i)	Type of Modulation	Quadrature Phase Modulation	Quadrature Phase and Amplitude Modulation
ii)	Location of signal points	On the circumference of circle	Equally spaced and placed symmetrical about origin.
iii)	Distance between the signal points	$d = 2\sqrt{E_b}$ for N = 2	$d = 2\sqrt{0.4E_b}$ for N = 4 or N = 16
iv)	Noise immunity	Better than QASK	More than QPSK
v)	Probability of error	Less than QASK	More than QPSK
vi)	Type of Demodulation	Synchronous	Synchronous
vii)	System complexity	Less complex than QASK	More complex than QPSK

Q.6(c) Describe WDM in details **[4]**
(A) Wavelength Division Multiplexing (WDM)



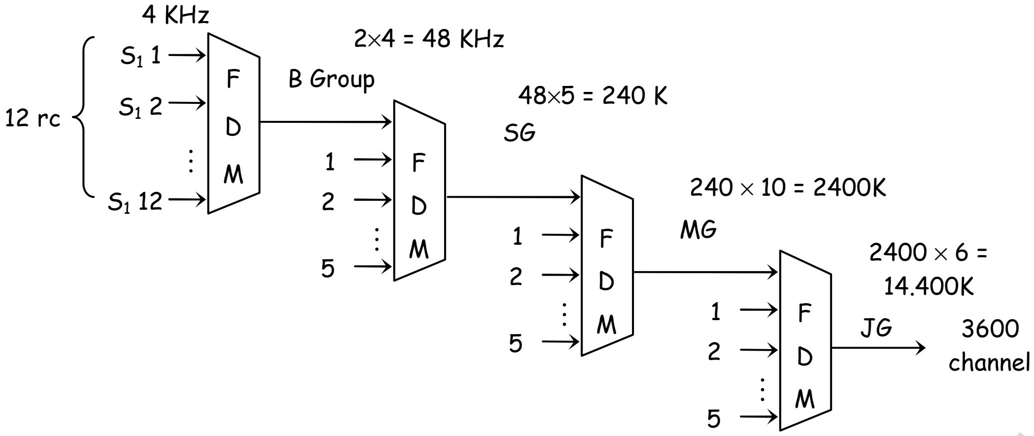


(i) Channels having different frequency ranges can be multiplexed on a long fiber.
(ii) The only difference with electrical FDM is that optical system is completely passive.
(iii) Reason WDM is popular is that the energy on a signal factor is a few gigahertz width because it is impossible to convert.
(iv) Hence wavelength division multiplexing is explained.

Q.6(d) Explain multiplexing hierarchy (AT & T) for? FDM system **[4]**
(A) Multiplexing Hierarchy in FDM

- Consider example at telephony in which each voice channel is having range of $300H_z$ to 3.4 KHz.
- Here we need to multiplex such 'n' no of voice channel by modulating it with different subcarriers.
- Multiplexing hierarchy goes as follows.

Level 1 : Basic Group : [12 voice channels multiplied together]
Level 2 : Super Group : [Upto 5 B.G mux together i.e.: upto 12×5 = 60 channels]
Level 3 : Master Group : [Upto 10 S.Cr mux together i.e. upto 600 V.C.]
Level 4 : Jumbo Group : [Upto 6 M.G. maximum together i.e. upto 3600 V.C.]

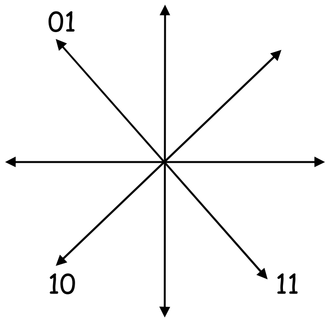
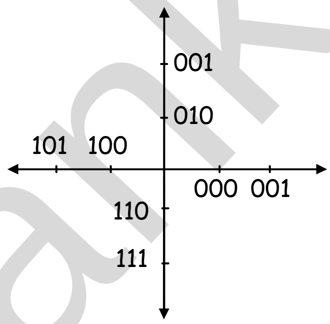


Q.6(e) Draw constellation diagram of **[4]**

(i) 4 QAM

(ii) 8 QAM

(A)

4 QAM	**8 QAM**
	
1 amp = 4 phases Here all the symbols have same amplitudes but different phases.	2 amp = 4 phases There are 2 amp levels and 4 phases.

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