# MODEL ANSWER

**SUMMER– 19 EXAMINATION**

**Subject Title:** Digital Communication Subject Code: 17535

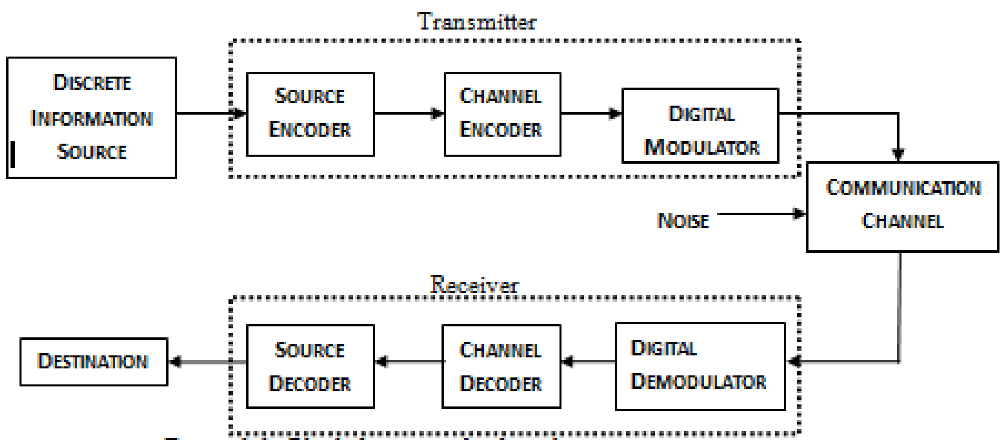
Important Instructions to examiners:

1) The answers should be examined by key words and not as word-to-word as given in the model answer scheme.
2) The model answer and the answer written by candidate may vary but the examiner may try to assess the understanding level of the candidate.
3) The language errors such as grammatical, spelling errors should not be given more Importance (Not applicable for subject English and Communication Skills.
4) While assessing figures, examiner may give credit for principal components indicated in the figure. The figures drawn by candidate and model answer may vary. The examiner may give credit for anyequivalent figure drawn.
5) Credits may be given step wise for numerical problems. In some cases, the assumed constant values may vary and there may be some difference in the candidate's answers and model answer.
6) In case of some questions credit may be given by judgement on part of examiner of relevant answer based on candidate's understanding.
7) For programming language papers, credit may be given to any other program based on equivalent concept.

| Q. No. | Sub Q.N. | Answer | Marking Scheme |
|---|---|---|---|
| Q.1 | | **Attempt any THREE of the following :** | 12-Total Marks |
| | i) | **Define channel capacity with mathematical expression.** | 4M |
| | Ans: | **(Definition - 2M, Mathematical expression -2M)**<br>**Channel capacity:-**<br>It is defined as the maximum rate at which data can be transferred over the channel with an arbitrary small probability of error. The unit of channel capacity is *bits/sec*.<br><br>**Mathematical expression:-**<br>Channel capacity of a channel with bandwidth B and band limited additive Gaussian white noise is given by:-<br>$$C = B \log 2 [1 + S/N] \text{ bits/sec}$$<br>where B – channel bandwidth (Hz)<br>S – Average signal power<br>N – Average noise power within the channel bandwidth<br>C – Channel capacity bits/sec<br>S/N – ratio of total signal power to total random noise power at the input of the receiver Within the frequency limits of this channel i.e. over the bandwidth. | |
| | ii) | **State sampling theorem and list its types.** | 4M |

| | | | |
|---|---|---|---|
| | Ans: | **(Statement - 3M, Types - 1M)**<br><br>**SAMPLING THEOREM:**<br>Sampling theorem states that a band-limited signal of finite energy having the highest frequency component *fm* Hz can be represented and recovered completely from a set of Samples taken at a rate of *fs* samples per second provided that **fs ≥ 2fm**. Here fs is the sampling<br>Frequency.<br>**Types of Sampling:-**<br>**1.** Natural sampling<br>**2.** Flat top Sampling | |
| | iii) | **Describe the need of multiplexing.** | 4M |
| | Ans: | **Need of Multiplexing:-**<br>• Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a signals data link.<br>• When many signals or channels are to be transmitted, then from transmitter's side that sends simultaneously i.e. multiplexer converts many into one, so that at the receiving end also all input we get simultaneously.<br>• Sending many signals separately is expensive and requires more wires to send. So there is a need of multiplexing.<br>• Efficient utilization of channel capacity.<br>• For example in cable T.V distributor sends many channels through single wire. | 4M |
| | iv) | **List four applications of spread specturm modulation.** | 4M |
| | Ans: | **(Each application - 1M)**<br>**Applications Of Spread Specturm Modulation:-**<br>The Spread Specturm Communications are widely used today for Military, Industrial, Avionics, Scientific, and Civil uses. The applications include the following:<br>1. Jam-resistant communication systems<br>2. CDMA radios<br>3. High Resolution Ranging: Spread Specturm Communications is often used in high resolution ranging. It is possible to locate an object with good accuracy using SS techniques. for example where it could be used is Global Positioning System (GPS).<br>4. WLAN: Wireless LAN (Local Area Networks) widely use spread spectrum communications.<br>i. Infrared (IR) Communications<br>ii. Direct Sequence Spread Spectrum Communications<br>iii. Frequency Hopping Spread Spectrum Communications.<br>5. Cordless Phones<br>6. Long-range wireless phones for home and industry<br>7. Cellular base stations interconnection.<br>8. Bluetooth.<br>9. Satellite communication | 4M |

| b | | Attempt any one of the following : | 6- Total Marks |
|---|---|---|---|
| | (i) | Draw the block diagram of basic digital communication system and explain its working. | 6M |
| | Ans: | **Block diagram of basic digital communication system:** | 2M |




## Explanation:- (4M)

**DISCRETE INFORMATION SOURCE:**

- The information to be transmitted originates here. These information/messages may be available in digital form or it may be available in an analog form.
- If it is analog it is sampled and digitized using an A/D converter to make the final source output to be digital in form.

**SOURCE ENCODER :**

- The bit stream at the source output will have considerable redundancy and so will not be efficient representation of the message or information given by the source, from the point of view of the number of digits used.
- A fewer number of digits might be sufficient to convey the information. The source encoder therefore reduces the redundancy by performing a one to one mapping of its input bit stream in to another bit stream at its output, but with fewer digits.
- Thus in a way it performs data compression.

**CHANNEL ENCODER:**

- The channel encoder is intended to introduce controlled redundancy into the bit stream at its input in order to provide some amount of error- correction capability to the data being transmitted.
- The data gets corrupted by the additive noise on the channel and this gives rise to the possibility of the channel decoder committing mistakes in the decoding of the data received from the channel.
- Redundancy helps in detecting erroneously decoded bits and makes it possible to correct the errors before passing on the data to the source decoder.

**DIGITAL MODULATOR:**

- The physical channels are basically analog in nature; the digital modulator takes each digital binary digit at its input and maps it, in a one –to – one fashion, into a continuous waveform.

**PHYSICAL CHANNEL:**

- The digitally modulated signal is passed on to the physical channel, which is nothing but the physical medium through which the signals are transmitted.
- It may take a variety of forms- a pair of twisted wires, coaxial cable, a wave guide, a microwave radio, or an optical fiber.
- During its passage through the channel, the signal gets corrupted by noise. This noise may be thermal noise originating from electronic circuits or atmospheric noise, or manmade noise, or as is generally the case, a combination of most or all of them.

**THE DIGITAL DEMODULATOR:**

- The digital demodulator of the receiver receives the noise corrupted sequence of waveforms from the channel and by inverse mapping tries to give at its output, an estimate of the sequence of the binary digits that were available at the input of the digital modulator at the transmitting end.

**THE CHANNEL DECODER:**

- The output sequences of digits from the digital demodulator are fed to the channel decoder. Using its knowledge of the type of coding performed by the channel encoder at the transmitting end and using the redundancy introduced by the channel encoder, it produces as its output, the output of the source coder of the transmitter with as few errors as possible.

**THE SOURCE DECODER:**

- Using its knowledge of the type of encoding performed by the source encoder of the transmitter, the source decoder of the receiver tries to reproduce at its output, a replica of the output of the digital source at the transmitting end.

| | | |
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| **ii)** | **Explain Hamming distance (dmin.). How many errors can be corrected and detected for the given minimum distance.** | **6M** |
| **Ans:** | **(Hamming distance with example - 3M, Errors can be detected and corrected with example - 3M)** | **6M** |

The Hamming distance between two words is the number of differences between corresponding bits. The minimum Hamming distance is the smallest Hamming distance between all possible pairs in a set of words.

We first find all the Hamming distances.

$$d(00000, 01011) = 3 \qquad d(00000, 10101) = 3 \qquad d(00000, 11110) = 4$$
$$d(01011, 10101) = 4 \qquad d(01011, 11110) = 3 \qquad d(10101, 11110) = 3$$

The dmin in this case is 3.

| | | | |
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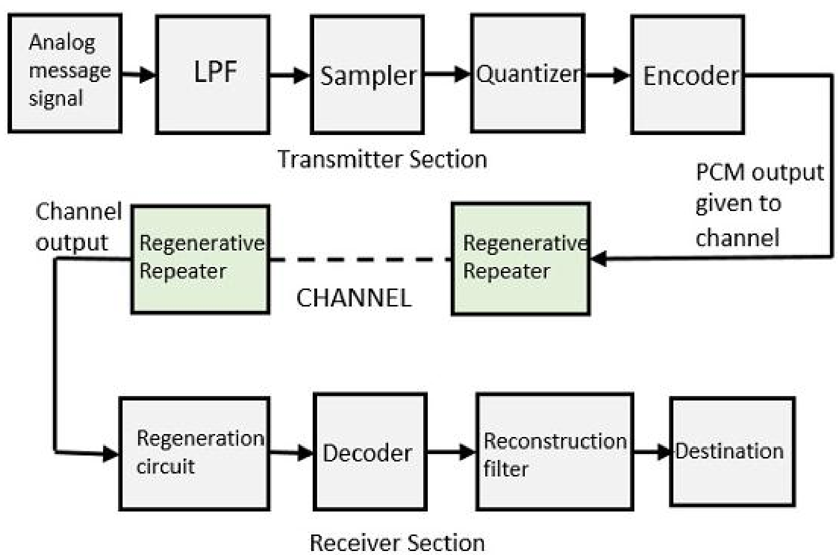
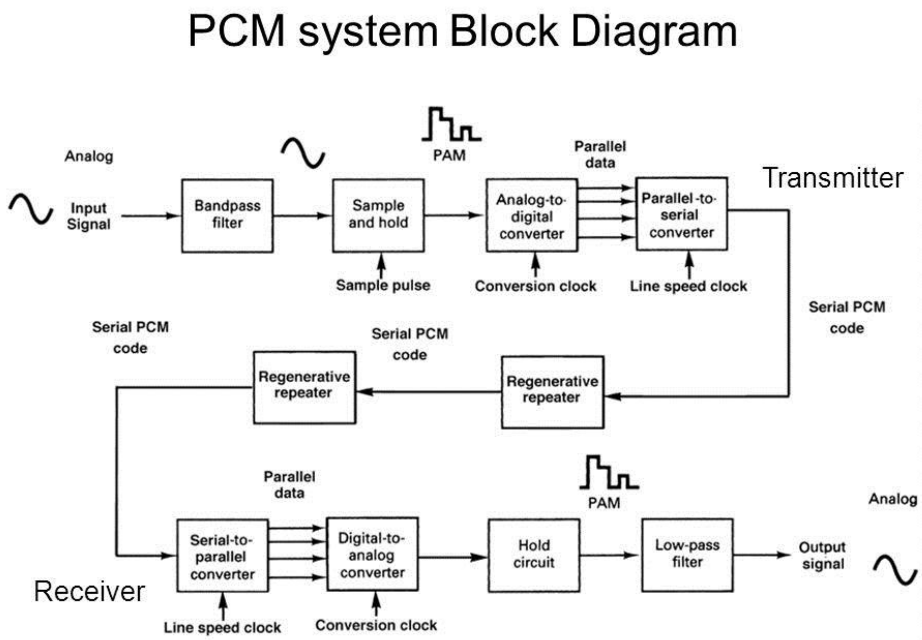
| | | **Errors can be detected and corrected with example**<br>To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be dmin = s+1.<br>For dmin=3, no. of bits error can be detected=dmin-1=3-1=2<br>To guarantee correction of up to t errors in all cases, the minimum Hamming distance in a block code must be dmin = 2t + 1.<br>For dmin=3, no. of bits error can be corrected=(dmin-1)/2=(3-1)/2=1 | |
| **Q.2** | | **Attempt any Two of the following :** | **16** |
| | **a)** | **Draw the block diagram of PCM transmitter and Receiver system and explain function of each block.** | **8M** |
| | **Ans:** | **(PCM Transmitter & Receiver diagram = 4M, Explanation = 4M)**<br><br><br>**OR** | |




**Block diagram of PCM transmitter:-**

- The analog signal *x (t)* is passed through a LPF (anti-aliasing filter). The LPF band-limits the signal to $f_m$ band-limiting is necessary to avoid the aliasing effect in the sampling process.

- The pulse generator generates a train of pulses at a frequency of $f_s$ such that $f_s > 2\ f_m$. Thus, the *Nyquist criterion* is satisfied.

- The sampler block carries out flat-top sampling process on the modulating signal at adequately high frequency. Then these samples are subjected to the operation called Quantization in the Quantizer.

- The quantization process is the process of approximation of the sampled signal. It assigns a particular level to which the sampled value is near to.

- The quantized PAM pulses are applied to an encoder. The encoder converts each quantized level into an *N*-bit digital word (binary pattern) such that $\boldsymbol{Q = 2^N}$ where *Q* is the total number of quantization levels.

- The combination of the Quantizer and the Encoder is called as an Analog-to-Digital Converter (A/D Converter). Thus, the signal transmitted over the communication channel is a digitally-encoded signal.

**PCM RECEIVER:**

**Block diagram of PCM Receiver**

- When the digitally-encoded signal is received at the receiver, the receiver makes a considered decision as to whether what has been received during that time slot is a ‘1’ or a '0' i.e. a 'pulse' or 'no pulse'.

- Decision is made during each time slot, preferably at the center of the time slot, compare the sample amplitude with a fixed pre-set threshold and declare it as a '1' if it exceeds the threshold and declare it as a '0' during that time slot, if it is less than the threshold.
- Once it is decided that it is a 1 in a time slot, local pulse generator in the receiver is triggered and it gives a clean, noise free rectangular pulse. Although the received pulses were distorted and corrupted by additive noise a clean pulse is locally generated this process is known as 'regeneration'.
- This sequence of clean 'pulse' and 'no-pulse', obtained through the process of regeneration are then fed to a serial to parallel converter then to a decoder which converts each code word into the corresponding quantized sample value.

These samples are then passed through a low pass reconstruction filter, which reconstructs an analog signal from these samples. This analog signal will be an approximation to the original analog message signal that was transmitted.

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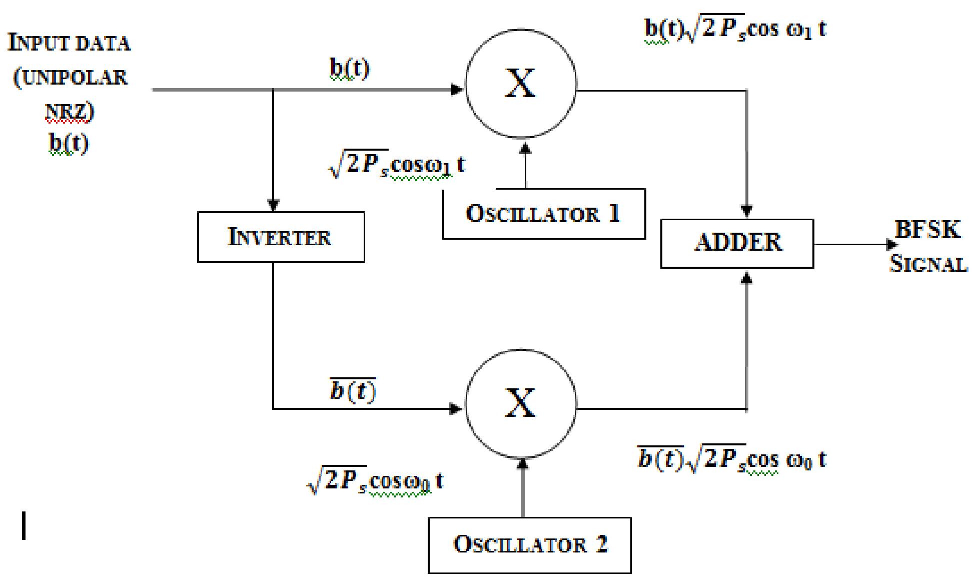
| **b)** | **Illustrate BFSK signal generation with block diagram and waveforms. State bandwidth requirement and draw its frequency spectrum.** | **8M** |
| **Ans:** | **(Block diagram = 2M, Explanation= 2M,Waveform =1M, Bandwidth = 1M, Frequency Spectrum = 2M )** | |

**Block diagram of BFSK signal Generation:-**




**Explanation:-**

- In FSK, the frequency of the carrier is changed with respect to the input bits 1 & 0.
- In case of binary data, two carrier frequencies are used. The carrier frequency corresponding to logic 0 or binary 0 is called as space frequency and the carrier frequency corresponding to binary 1 is called as mark frequency.

**FSK TRANSMITTER:**

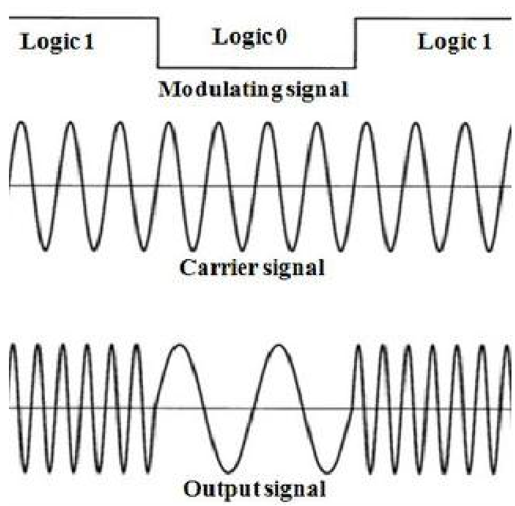
- In general, for binary FSK, the carriers can be represented by:

*For binary 0,* $V_0(t) = \sqrt{2P_s}\cos 2\pi f_0 t = \sqrt{2P_s}\cos \omega_0 t$

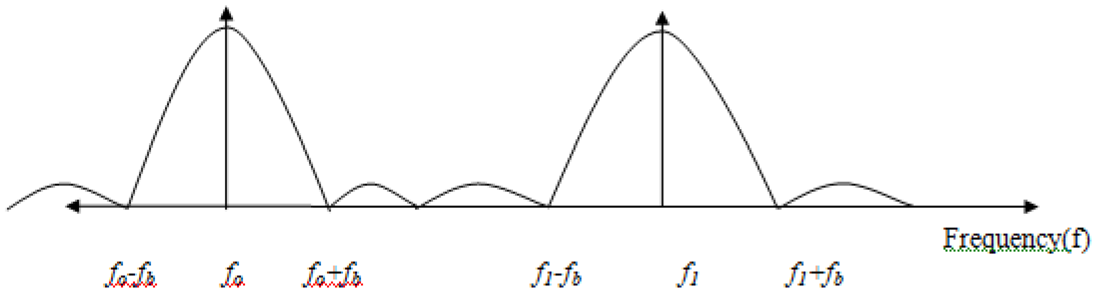
*For binary 1,* $V_1(t) = \sqrt{2P_s}\cos 2\pi f_1 t = \sqrt{2P_s}\cos \omega_1 t$

- Assume that high frequency is transmitted for 1 and low frequency is transmitted for 0
- Low frequency = $f_0 = f_c - \delta f$
- High frequency = $f_1 = f_c + \delta f$
- The two carriers may be generated from separate oscillators independent of one another as shown in the above figure.
- As shown in figure the input binary data is given directly to the multiplier and is inverted and given to second multiplier.
- Two different carriers have different frequency generated by the two oscillators and applied to the multipliers.
- The output of both the multipliers is an ASK signal which is added by the summer. Thus, the output of the adder is the FSK wave.
- Since there are two different oscillators used for the carrier signal the combined signal at the summer therefore has discontinuous amplitude and phase which is undesirable in FSK. Hence, we need a common carrier for FSK modulation to avoid these discontinuities in amplitude and phase.

**Waveform:-**




**Frequency Spectrum:-**




**Bandwidth Requirement:-**

- Minimum bandwidth necessary for FSK is given by:

$$B = (f_1 + f_b) - (f_0 - f_b)$$

i.e. $= (f_1 - f_0) + 2f_b$

Hence, $B = 2f_b + 2f_b$

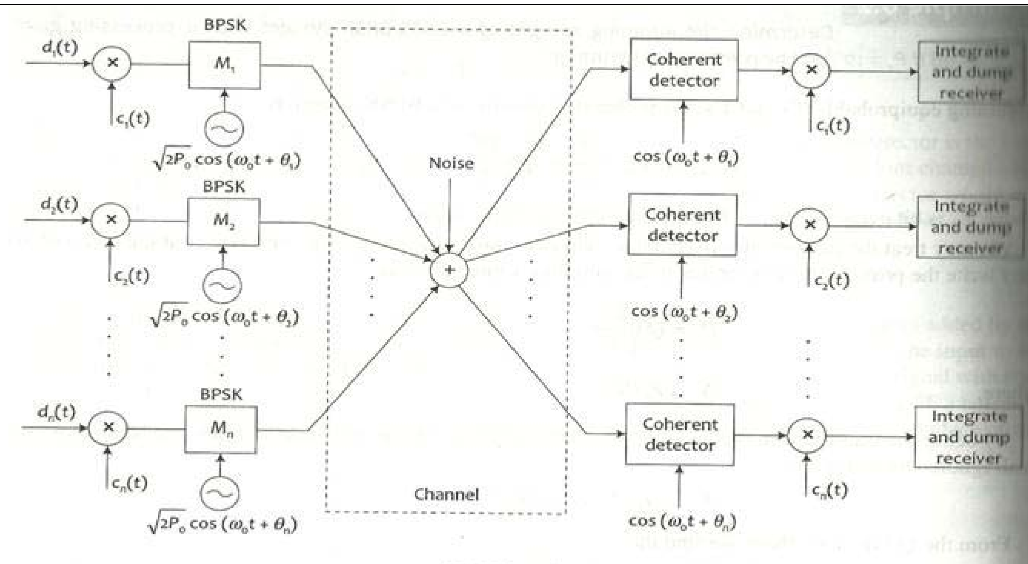
$= 4\, f_b$

| | | |
|---|---|---|
| c) | **Explain working of CDMA. State its advantages and disadvantages.** | **8M** |

**Ans:** **(Diagram = 2M , Explanation =2M, Advantages = 2M, Disadvantages =2M)**

- CDMA is used along with spread spectrum modulation technique, which neither uses frequency channels nor time slots.
- In SSM (spread spectrum modulation) spreading is achieved by PN code. If such unique code is assigned to each individual user, the demodulation will be possible only if the code matches at the receiver end. This enables the multiple access.
- CDMA system uses same frequency band and transmit simultaneously. They can use the whole available bandwidth for all the time. The transmitted signal is recovered by co-relating the received signal with the PN code used by the transmitter.
- CDMA property
  1. Non-interference with existing system
  2. Anti-jam and interference rejection
  3. Information security
  4. Accurate ranging
  5. Multipath tolerance
  - CDMA allows all the users to occupy all channels at the same time. Transmitted signal is spread over the whole band and each voice or data call is assigned a unique code to differentiate it from other calls carried over the space spectrum.
  - All the users in CDMA use same carrier and may transmit simultaneously. Each user has its own pseudorandom code word which is orthogonal to all other code words. For detection of message signal the receiver needs to know the code word use by transmitter. Each user operates independently with no. of knowledge of other users.

**CDMA MULTIPLE ACCESS USING DS-SPREAD SPECTRUM**

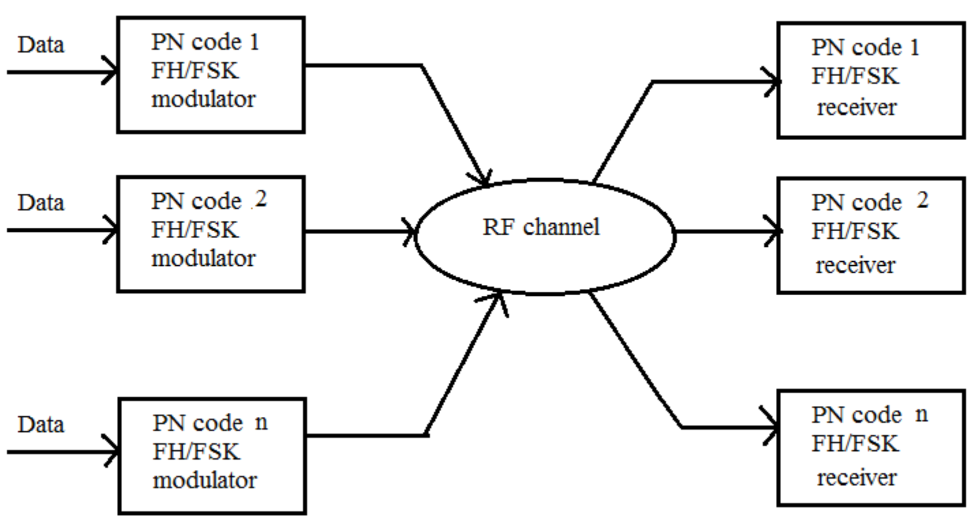



A DSSS-based CDMA system

**OR**

**CDMA with FHSS:-**

- As in DS spread spectrum multiple access is achieved in FHSS also by assigning a unique PN code to each user, which in this case controls the frequency-hopping pattern.
- These codes that are assigned must be so chosen that collision do not occur.
- The frequency produced by the frequency synthesizer during a chip period depends on the PN sequence value during that chip period.
- So sometimes it may so happen that two or more users have at a given time, the same PN sequence values produced by their respective PN sequence generators. In that case a collision is said to have occurred in the spectrum.
- Whether it is slow hopping or fast hopping FHSS, when a collision occurs it results in considerable increase in detection errors.




**Advantages:- (Any 2)**

- The CDMA does not require any synchronization.

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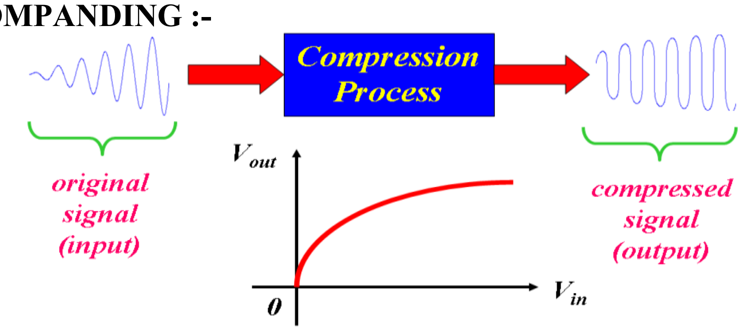
| | | ○ It has more number of users can share the same bandwidth.<br>○ It is well-matched with other cellular technologies.<br>○ Due to code word allocated to each user, interference is reduced.<br>○ Efficient practical utilization of fixed frequency spectrum.<br>**Disadvantages:- (Any 2)**<br>○ The system is more complicated.<br>○ Guard band and guard time both are required to be provided.<br>○ As the number of users increases, the overall quality of services decreases. | |
| **Q.3** | **A)** | **Attempt any FOUR of the following :** | **16** |
| | **a)** | **Explain Companding with neat diagram.** | **4M** |
| | **Ans:** | **(Diagram = 2M , Explanation =2M)**<br>**Diagram Of COMPANDING :-** | **2M** |




**Explanation:-** **2M**

The combination of compressor and expander is known as compander which performs the Companding process.

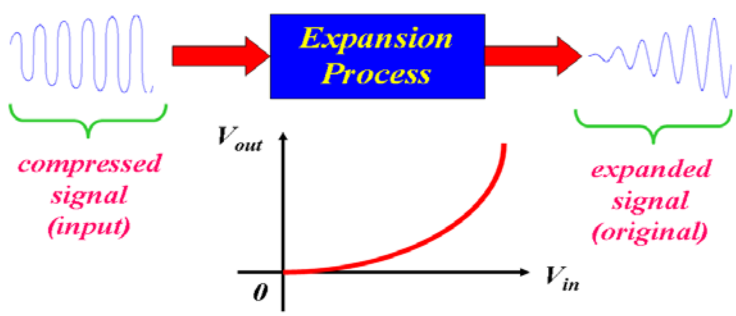
It is used to increase the signal to quantization error ratio for weak signal.

Compression process provides high gain to weak signals & less gain to strong signals. Practically implemented using a device of logarithmic (log) characteristics.

Here weak signals are artificially boosted to improve signal to quantization noise ratio.

This process also attenuates strong signals, keeping the value of $S/N_{Q\ almost}$ constant.

It takes place before the actual quantization process, in both the 1st & 2nd quadrants.




Expanding process provides a higher gain to strong signals & less gain to weaker signals.
Practically implemented using a device of exponential (antilog) characteristics.
This is because weak signals were artificially boosted & hence need to be restored back.
This process also magnifies strong signals, keeping the value of $S/N_{Q\ almost}$ constant.
It takes place after the actual quantization process, towards the receiver side.
Compression at the transmitter side.
Expansion at the Receiver side.
At the transmitter end the information signal is passed through compressor where the

| | | | |
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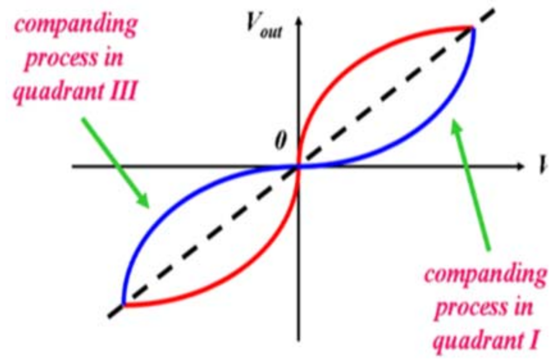
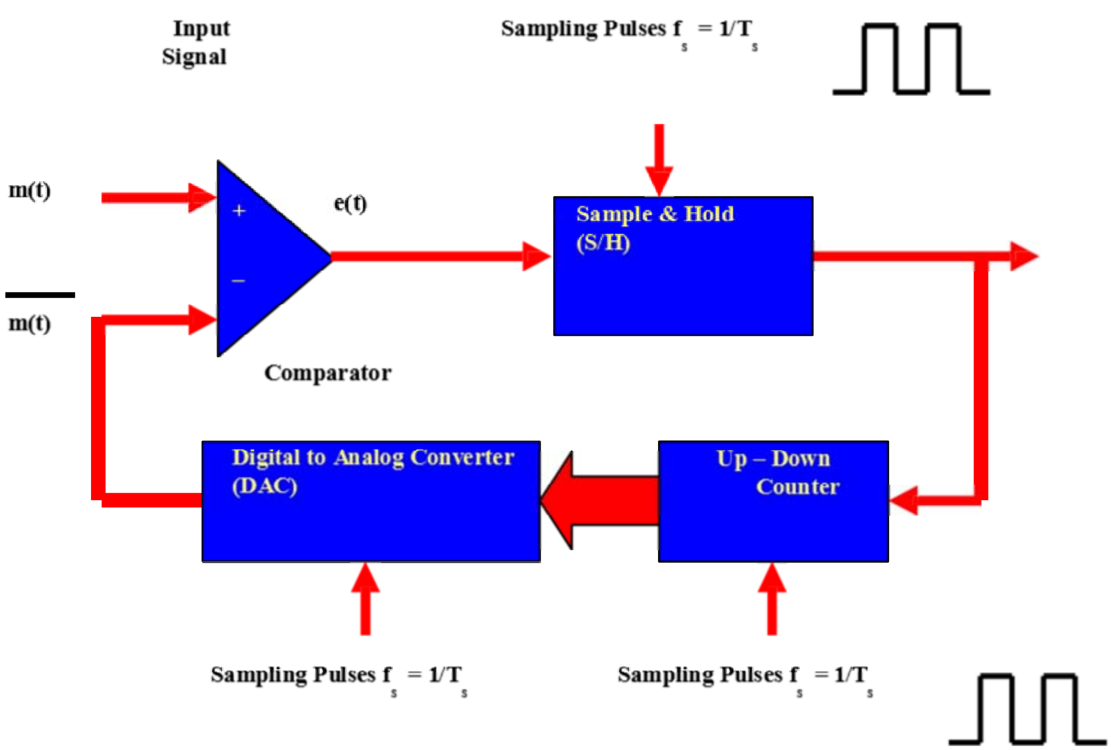
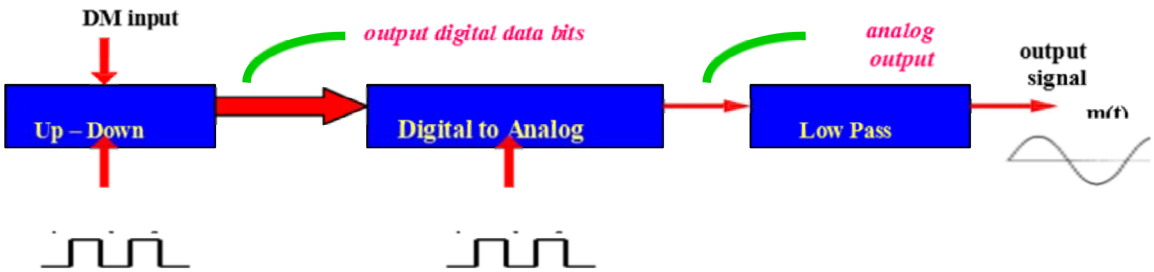
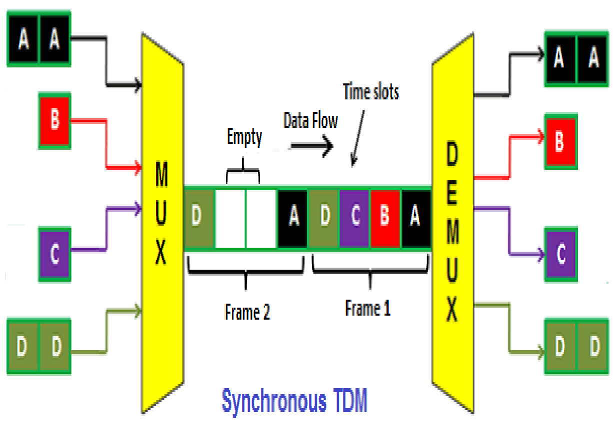
| | | signal is amplified more at low amplitude than at high amplitude.<br>At the receiver side, an inverse operation is performed to recover the original information signal. This is achieved by an expander.<br>**The Companding Characteristics:-**<br><br><br>. | |
| | **b)** | **Draw the block schematic of DM transmitter and Receiver.** | **4M** |
| | **Ans:** | **DM Transmitter Block Diagram:-**<br><br><br>**DM Receiver Block Diagram:-** | **2M**<br><br>**2M** |




**c) Describe synchronous TDM with block diagram transmitter.** **4M**

**Ans: (Diagram = 2M , Explanation =2M)** **2M**

**Diagram Of Synchronous TDM Transmitter:-**




**2M**

**Explanation:-**

In synchronous time division multiplexing, each device (transmitter) is allotted with a fixed time slot, regardless of the fact that the device (transmitter) has any data to transmit or not. The device has to transmit data within this time slot. If the device (transmitter) does not have any data to send then its time slot remains empty.

As shown in the below figure, the various time slots are arranged into frames and each frame consists of one or more time slots dedicated to each device (transmitter). For example, if there are 3 devices, there will be 3 slots in each frame. Similarly, if there are 5 devices, there will be 5 slots in each frame.

The above figure shows 4 devices (transmitter A, transmitter B, transmitter C, and transmitter D) that have 4 dedicated time slots (time slot A, time slot B, time slot C and time slot D).

The transmitter A data is sent at time slot A, transmitter B data is sent at time slot B, transmitter C data is sent at time slot C and transmitter D data is sent at time slot D.

In the time frame 2, the transmitter B and C does not have any data to send so the time slot B and C remains empty.

The main drawback of synchronous time division multiplexing is that the channel capacity is not fully utilized. Hence, the bandwidth goes wasted.

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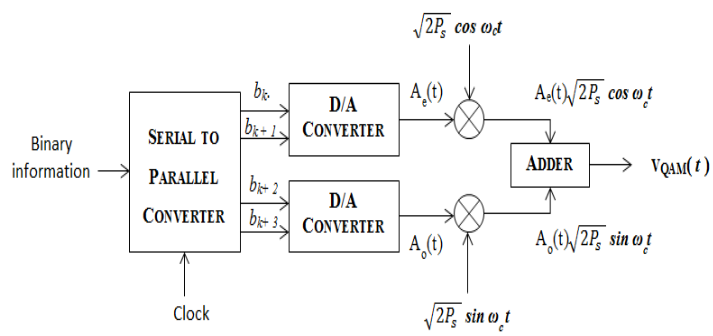
| d) | **Compare QPSK QASK (any four points).** | | **4M** |
| **Ans:** | | | **Each Point 1M** |

| Parameter | QPSK | QASK |
|---|---|---|
| 1.Information is transmitted by change in | phase | Amplitude and phase |
| | | |
| 2. Number of bits/symbol | N=2 | N=3 Or 4 Or 5 & so on |
| 3.Number of possible symbols | Four | $M=2^N$ |
| 4.Detection method | coherent | coherent |
| 5.minimum bandwidth | fb | 2fb/N |
| 6.symbol duration | 2Tb | NTb |
| 7.Noise immunity | Comparatively less than QAM | Better than QPSK |
| 8.System complexity | Less complex than QAM | More complex than QPSK |
| 9.Probability of error | More than QAM | Less than QPSK |
| 10.Performance of system | Less than QAM | Better than QPSK |

**e)** **Describe QAM transmitter with block diagram.** **4M**

**Ans:** **QAM transmitter block diagram:-** **2M**




**QAM transmitter Explanation:-** **2M**

The bi stream $b_{(t)}$) is applied to the serial to parallel converter, operating on a clock which has period of $T_{s,}$ Which is the symbol duration. The bits b($^t$) are stored by the converter and then presented in the parallel form.

The four bit symbols are $b_k$ +3, bk +1, and bk.

Out of these four bits the first two bits are applied to a D/A converter and the other two bits are applied to the second D/A converter.

The Output of the first converter is $A_e$(t),which is modulated by the carrier√2Ps cos∞

T whereas is the output of the second D/A converter $A_o$ is modulated by the carrier √2Ps sin ∞t in the balanced modulators.

$A_e$(t) $A_o$($_t$) are voltages levels generated by the converter -3 -1, +1,+3 volts

The balanced modulator outputs are added together to get the QASK output signal which is expressed as:

$$v_{QASK}(t) = A_e(t)\sqrt{2P_s}cos\omega_c t + A_o(t)\sqrt{2P_s}sin\omega_c t$$

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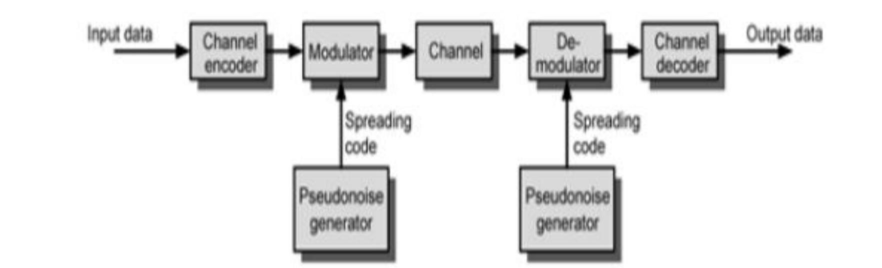
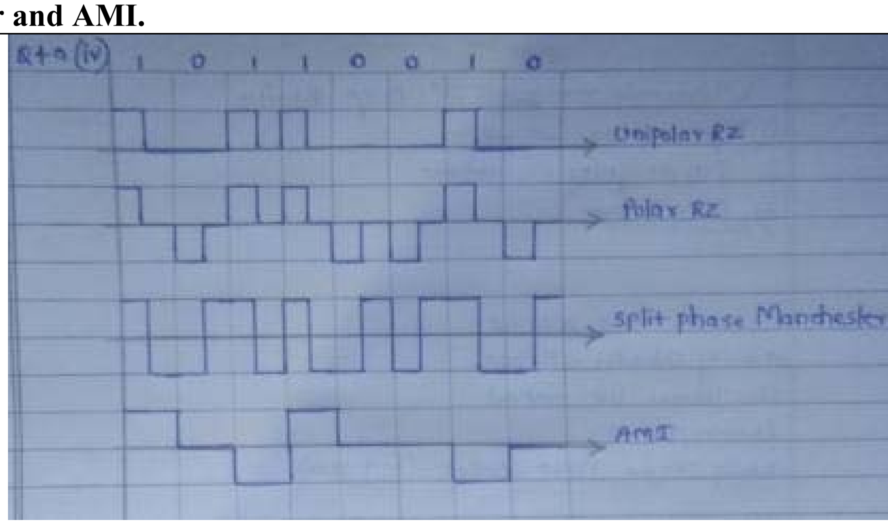
| **Q.4** | **a)** | **Attempt any THREE of the following:** | **12** |
| | **(i)** | **State advantages and disadvantages of digital communication system.** | **4M** |
| | **Ans:** | **(Any 4 advantages = 2M , Any 2 disadvantages =2M)**<br>**Advantages of digital communication System:**<br>**1.** Digital systems are simple and easy to build.<br>**2.** Insensitive to variations in atmospheric conditions like temperature, humidity etc. so highly robust.<br>**3.** Storage and retrieval of voice, data or video is easy& inexpensive.<br>**4.** It offers considerable flexibility as voice, data, video can all be multiplexed using TDM, signal and image processing operations like compression of voice and image signal can be easily used etc<br>**5.** Cost of digital communication systems are coming down because of improvement of VLSI technology and available of IC's at ever decreasing price.<br>**6.** Error correction codes ensure fairly good protection against noise and interferences.<br>**7.** Powerful encryption and decryption algorithms are available for digital data so as to maintain high level of secrecy of communication.<br>**8.** In digital communication system, repeaters are used as regenerators so signal reaching at the destination can be almost error free. So this system can be used for long-haul communication.<br>**9.** Multiple data can be send simultaneously using multiplexing.<br><br>**Disadvantages of Digital Communication System:**<br>**1.** The transmission of digitally encoded analog signals requires significantly more bandwidth.<br>**2.** Digital transmission requires precise time synchronization between the clocks in the transmitter and receiver.<br>**3.** Digital transmission systems are incompatible with older analog transmission systems.<br>**4.** Digital communication system requires greater bandwidth | **2M**<br><br><br><br><br><br><br><br><br><br><br><br><br><br><br><br><br>**2M** |
| | **(ii)** | **Explain the need of using adaptive delta modulation.** | **4M** |
| | **Ans:** | **Necessity of adaptive delta modulation technique:-**<br>In delta modulation, the step size is constant so that its slope overload distortion and granular noise both cannot be controlled. These drawbacks can be controlled by using adaptive delta modulation wherein the step size is variable.<br>Thus with adaptive delta modulation the following are the advantages-<br>1. Slope overload distortion and granular noise problem in is reduced.<br>2. Improved signal to noise ratio.<br>3. Wide dynamic range is achieved with variable step size.<br>4.Better bandwidth utilization than delta modulation | **4M** |
| | **(iii)** | **Draw and explain spread spectrum modulation system.** | **4M** |
| | **Ans:** | **(Diagram = 2M , Explanation =2M**<br><br>**Model Of a Spread Specturm modulation System:-** | **2M** |

| | | | |
|---|---|---|---|
| | | <br>General Model of a Spread Spectrum system<br><br>**Explanation Of Spread Spectrum Modulation System:-**<br>**1.** Basic elements of a spread spectrum signal modulation system is shown below.<br>**2.** Channel encoder adds extra bits to the information binary sequence for error detection & correction purpose.<br>**3.** PN sequence generation at the transmitter & receiver generates identical PN binary valued sequence<br>**4.** PN sequence is impressed on the information signal at the modulator (Tx) and remove from the received signal at the Demodulator.<br>**5.** Synchronization of the PN sequence generator at the receiver with the PN sequence contained in the incoming received signal is required in order to demodulate the received signal.<br>**6.** Prior to the transmission of information Synchronization may be achieved by transmitting a fixed PN sequence pattern which the receiver will recognize in the presence of interference with high probability. | **2M** |
| | **(iv)** | **For the binary data system 10110010 draw Unipolar RZ, Polar RZ, split phase Manchester and AMI.** | **4M** |
| | **Ans:** |  | **1M each wave-form** |
| | **b)** | **Attempt any ONE of the following** | **6M** |
| | **(i)** | **Explain working of CRC generator and checker.** | **6M** |

| | | |
|---|---|---|
| Ans: | **(Diagram = 2M , Explanation =2M and Example = 2 M)** | **4M** |

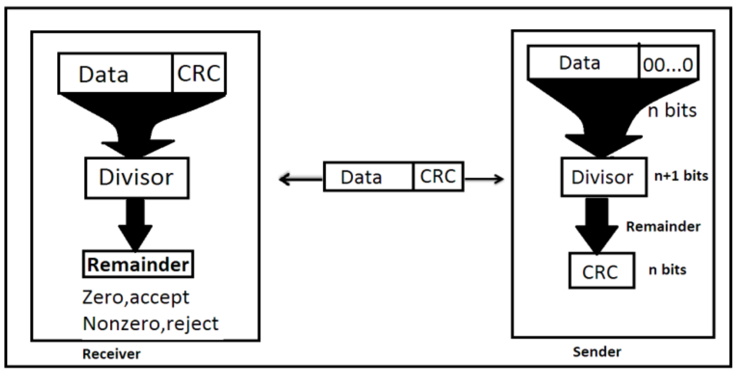
**<u>Explanation Of Cyclic Redundancy Check (CRC):-</u>**

With CRC the entire data stream is treated as long continuous binary number. In this method, a sequence of redundant bits, called the CRC or the CRC remainder, is appended to the end of the unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number.

At its destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit assume to be correct and is accepted, otherwise it indicate that data unit has been damaged in transmission and therefore must be rejected

The redundancies bits are used by CRC are derived by dividing the data unit by a predetermined divisor. The remainder is the CRC.

**Diagram Of Cyclic Redundancy Check:-** **2M**




**Example:** At Tx

| | 1 1 1 1 0 1 quotient |
|---|---|
| 1 1 0 1<br>divisor | 1 0 0 1 0 0 0 0 0 (data plus extra 3 zero)<br>1 1 0 1<br>------------------<br>1 0 0 0<br>1 1 0 1<br>--------------------<br>1 0 1 0<br>1 1 0 1<br>-----------------------<br>1 1 1 0<br>1 1 0 1<br>--------------------------<br>0 1 1 0<br>0 0 0 0<br>---------------------------<br>1 1 0 0<br>1 1 0 1<br>-----------------------------<br>0 0 1 remainder |

At Rx

| | |
|---|---|
| | 1 1 1 1 0 1 quotient |
| 1 1 0 1<br>divisor | 1 0 0 1 0 0 0 0 1 (data plus CRC received)<br>1 1 0 1<br>-----------------<br>1 0 0 0<br>1 1 0 1<br>--------------------<br>1 0 1 0<br>1 1 0 1<br>-----------------------<br>1 1 1 0<br>1 1 0 1<br>--------------------------<br>0 1 1 0<br>0 0 0 0<br>----------------------------<br>1 1 0 1<br>1 1 0 1<br>------------------------------<br>0 0 0 result |

**(ii) Explain frequency hopping. Compare slow and fast frequency hopping (four points)** **6M**

**Ans: (Explanation =2M and each point of comparison =1 M)** **2M**

**FHSS:**

It combines spread spectrum modulation with MFSK.It is the process of modifying frequency of MFSK signal using frequency hope generated by bits of PN sequence.

Different types of frequency hopping:-Slow and fast frequency hopping.

**FHSS(Frequency Hoping Spread spectrum):**

Signal is broadcasting over seemingly random series of frequencies Receiver hops between frequencies in sync with transmitter. Eavesdroppers hear unintelligible blips. Jamming on one frequency affects only a few bits.

| Slow frequency hopping | Fast frequency hopping |
|---|---|
| 1.More than one symbols are transmitted per frequency hop | 1.More than one frequency hops are required to transmit one symbol |
| 2.chip rate is equal to symbol rate | 2.chip rate is equal to hop rate |
| 3.symbol rate is higher than hop rate | 3.hop rate is higher than symbol rate |
| 4.same carrier frequency is used to transmit one or more symbols | 4.one symbol is transmitted over multiple carrier in different hops |
| 5.A jammer can detect this signal if the carrier frequency in one hop is known | 5.A jammer cannot detect this signal because one symbol is transmitted using more than one carrier frequencies |

**1M each point**

**Q.5 Attempt any TWO of the following:** **16**

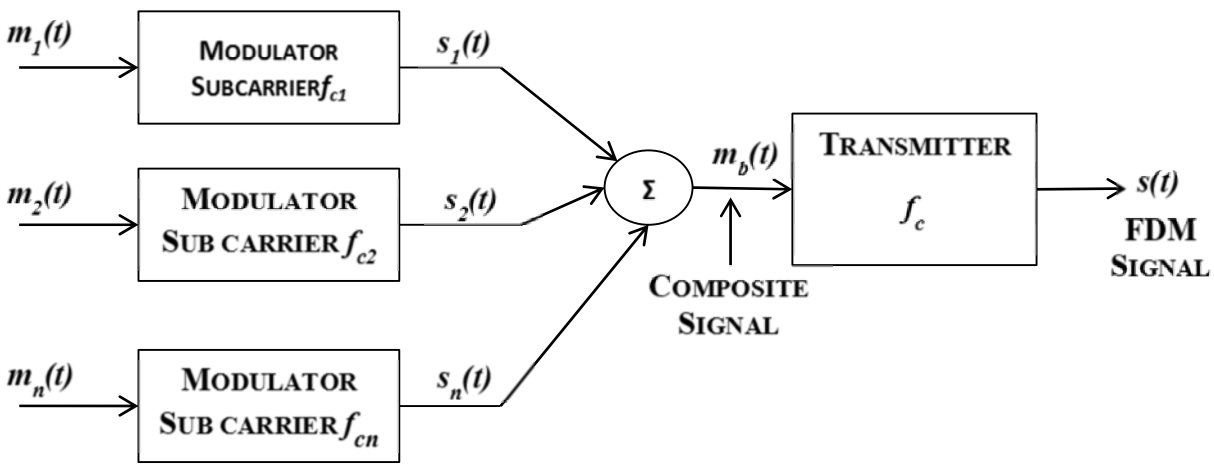
**a) Draw block diagram of FDM transmitter and Receiver and explain function of each block.** **8M**

**Ans:**

**(block diagram of Tx/Rx = 2M each , Explanation =2M each )** **2M**

**FDM TRANSMITTER:**

- In FDM, the transmission channel is shared by multiple signals, each being allotted a portion of the spectrum of the bandwidth. A generalized block diagram of FDM transmitter is depicted in Figure .

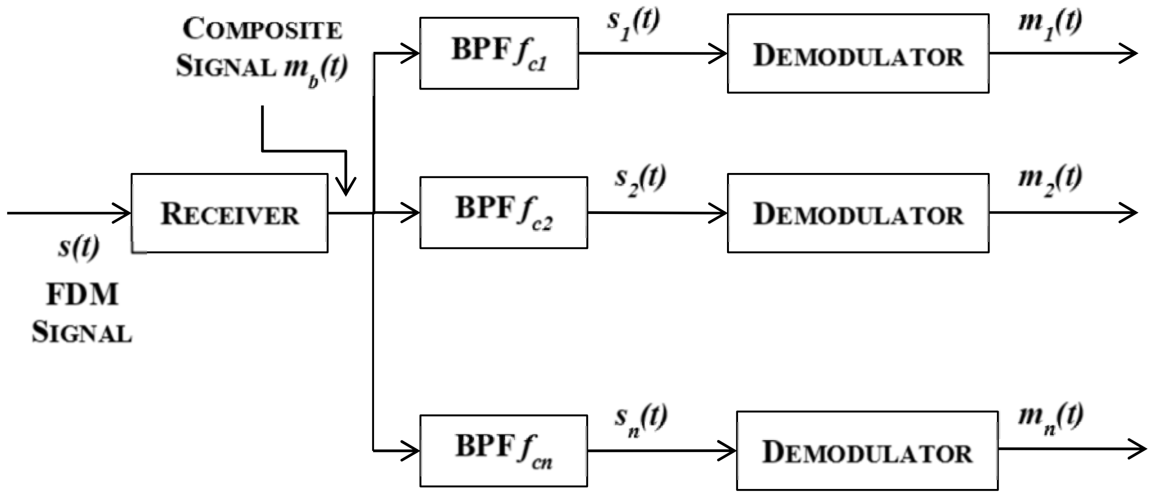



**2M**

**FDM Transmitter**

- A number of analog signals (or digital signals converted into analog) $m_i(t)$, $i$ = 1, 2, . . ., $n$ are multiplexed onto the same transmission medium. Each signal $m_i(t)$ is modulated onto a carrier $f_{ci}$.
- As multiple carriers are to be used, each is referred to as a *subcarrier*. Any type of analog modulation may be used. The resulting analog signals are summed together to produce a composite signal, $m_b(t)$. **2M**
- The composite signal may be shifted, as a whole, to another carrier frequency by an additional modulation step.
- This second modulation step need not use the same modulation technique as the first. Thus, the FDM signal generated may be transmitted over a suitable medium.

**FDM RECEIVER:**




**FDM Receiver**

2M

- The FDM receiver is shown in Figure. The FDM signal is received by the receiver and demodulated to retrieve the composite signal $m_b(t)$ which is further amplified.
- This composite signal is passed through $n$band pass filters, each filter having a center frequency equal to the subcarrier frequency $f_{ci}$.
- In this way the composite signal is split into its component signals. Each component signal is further demodulated to obtain the analog outputs that were originally transmitted. If required, these signals are stored and displayed.

| | | |
|---|---|---|
| **b)** | **Draw and explain block diagram of DPSK transmitter and receiver.** | **8M** |
| **Ans:** | **(block diagram of Tx/Rx = 2M each , Explanation =2M each )** | **2M** |

**DPSK MODULATOR:**

- The generation block diagram of DPSK signal is shown in Figure 3.16. The data stream to be transmitted, $d(t)$, is applied to one input of an exclusive-OR logic gate. To the other gate input the output of the exclusive-OR gate $b(t)$ delayed by time $T_b$ allocated to one bit is applied. This second input is then $b(t - T_b)$.

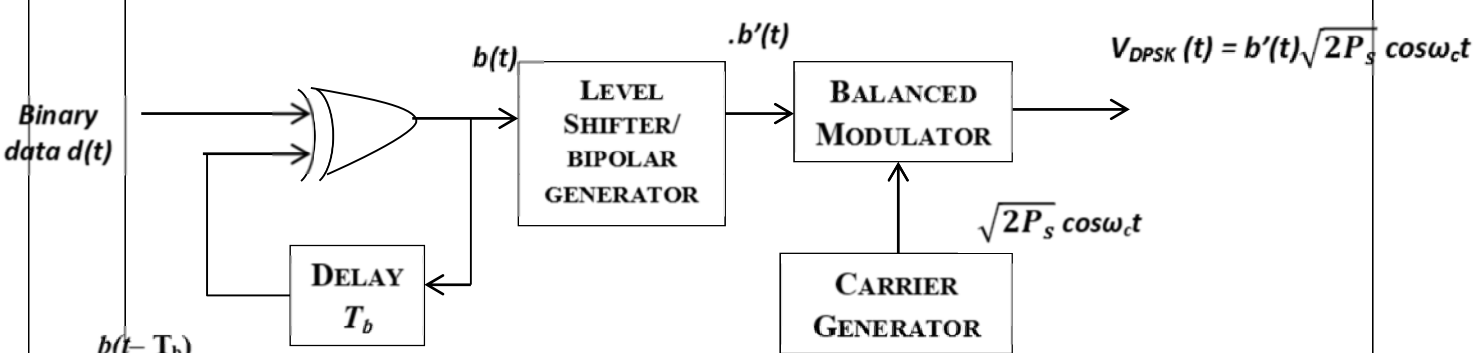



**Figure : DPSK Transmitter**

The truth table of the exclusive-OR gate is shown below: **2M**

| **d(t)** | | **b(t - $T_b$)** | | **b(t)** | |
|---|---|---|---|---|---|
| Logic Level | Voltage | Logic Level | Voltage | Logic Level | Voltage |
| 0 | -1 | 0 | -1 | 0 | -1 |
| 0 | -1 | 1 | 1 | 1 | 1 |
| 1 | 1 | 0 | -1 | 1 | 1 |
| 1 | 1 | 1 | 1 | 0 | -1 |

From figure 3.16, $b(t)$ is given by, $b(t) = d(t) \oplus b(t - T_b)$

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| Input Data $d(t)$ | | 1 | 0 | 1 | 1 | 1 | 0 |
| Delayed input $b(t - T_b)$ | | 0 | 1 | 1 | 0 | 1 | 0 |
| XOR Output $b(t)$ | 0 | 1 | 1 | 0 | 1 | 0 | 0 |
| Output Phase | | 0° | 0° | 180° | 0° | 180° | 180° |
| DPSK input (at Receiver) | 180° | 0° | 0° | 180° | 0° | 180° | 180° |
| Recovered data stream | | 1 | 0 | 1 | 1 | 1 | 0 |

**2M**

- It is observed that when $d(t) = 0$, $b(t) = b(t - T_b)$ and when $d(t) = 1$, $b(t) = \overline{b(t - T_b)}$. As seen in Figure 3.16,$b(t)$ is applied to a level shifter which assigns a positive voltage level when $b(t) = 1$ and a negative voltage level when $b(t) = 0$. The level shifter output is then applied to a balanced modulator to which a carrier signal $\sqrt{2P_s}\cos\omega_c t$ is also applied. The modulator output, which is the transmitted signal is given by,

$$V_{DPSK}(t) = b'(t)\sqrt{2P_s}\cos\omega_c t$$
$$= (\pm 1)\sqrt{2P_s}\cos\omega_c t$$

**2M**

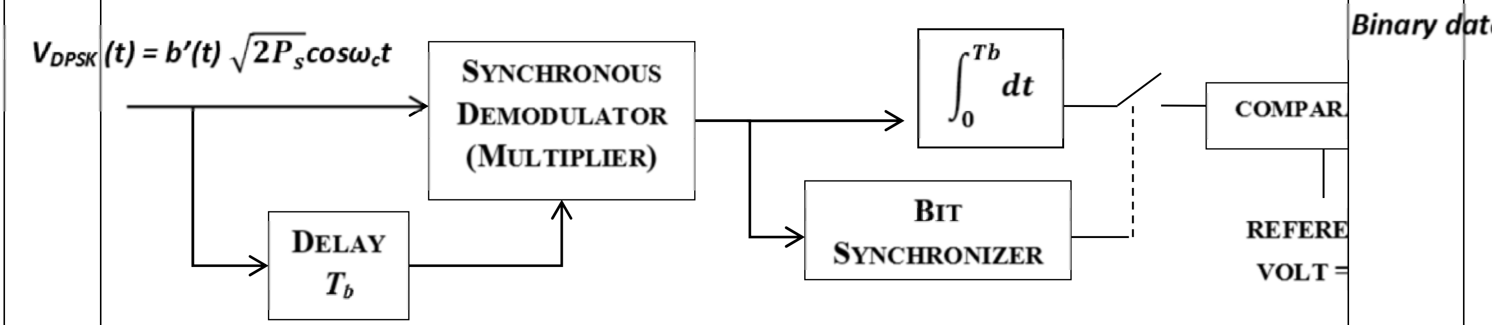
**DPSK DEMODULATOR:**

- The DPSK demodulator is shown in Figure. Here the received signal and the received signal delayed by the bit time Tb are applied to a balanced modulator. The balanced modualtor output is given by,

$$b(t)b(t - T_b)(2P_s)\cos\omega_c(t)\cos\omega_c(t - T_b)$$

- The output of the balanced modulator is applied to the integrator which suppresses the double frequency term. The first term [ $b(t)b(t - T_b)P_s\cos\omega_c T_b$]is the required signal and $\omega_c T_b$ is selected in such a way so that $\omega_c T_b = 2n\pi$ (where $n$ is an integer) so that $\cos\omega_c T_b = +1$ and the signal output will be as large as possible. Further, with this selection, the bit duration encompasses an integral number of clock cycles and the integral of the double frequency term is exactly zero.

- Output of the integrator = $b(t)\ b(t - T_b)\ P_s\ T_b$



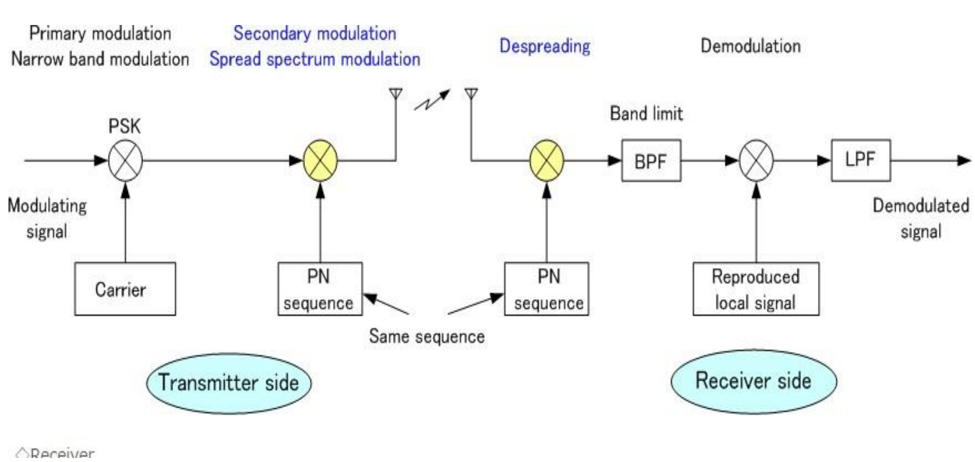

**DPSK Receiver**

- The transmitted data bit d(t) can be determined from the product $b(t)b(t - T_b)$. If there is no phase change between $b(t)$ and $b(t - T_b)$ then $d(t) = 0$.i.e. if $b(t) = b(t - T_b)$.In this case
  $b(t)\ b(t - T_b) = +1$.
  No phase change between $b(t)$ and $b(t - T_b)$ means both are 1 or both are 0, which means both are +1 or both are -1 hence the multiplication is always positive.
- If there is a phase change between them, then either $b(t)$or $b(t - T_b) = -1$.
  i.e. if $b(t) \neq b(t - T_b)$ In either case, $b(t)b(t - T_b) = -1$. Then $d(t) = 1$.
  Phase change between $b(t)$ and $b(t - T_b)$ means either is 1 and other is 0, which means one is +1 and other is -1 hence the multiplication is always negative.
- If the comparator receives +ve voltage bit received is 0 , if it receives –ve voltage bit received is 1.

**c)** **Describe DSSS transmitter and receiver working with block diagram.** **8M**

**Ans:** **(block diagram of Tx/Rx = 2M each , Working =2M each )** **4M**

**DSSS transmitter and receiver working with block diagram:-**




**Explanation:-** **4M**

The information signal undergoes primary modulation by PSK, FSK or other narrow band modulation and secondary modulation with spread spectrum modulation. Spread spectra are obtained by multiplying the primary modulated signal and the square wave, called the PN sequence. Contrariwise, as with commercial radio, there are cases where spread modulation is applied to the data first, and narrow band modulation such as PSK or FSK is applied afterwards.

The figure below is an example of spread spectrum modulation and demodulation using PSK for primary modulation.

**Receiver:**

If despreading is applied to the received diffuse wave, it returns to the PSK or FSK modulated wave resulting from primary modulation. Then, as with narrowband demodulation, if the despread wave and local signal are multiplied, and appropriate low pass processing is applied, the information signal is obtained. Despreading involves multiplying the same PN code as that used at the transmitting end for the receiving wave. At this time, it's necessary to synchronize the receiving wave and PN code.

There are two processing methods on the receiving side, demodulation of the information signal after despreading, and obtaining a positive and negative PN code by multiplying the local signal by the receiving wave and despreading using correlation detection. With the former there is process gain but the problem of synchronization remains. With the latter, the spectrum density of the receiving wave itself is low, and regeneration of the local carrier for performing synchronous detection is a problem. Commercial SS radio equipment uses the latter, but it requires considerable power and has a short communication range.

**Despreading:**

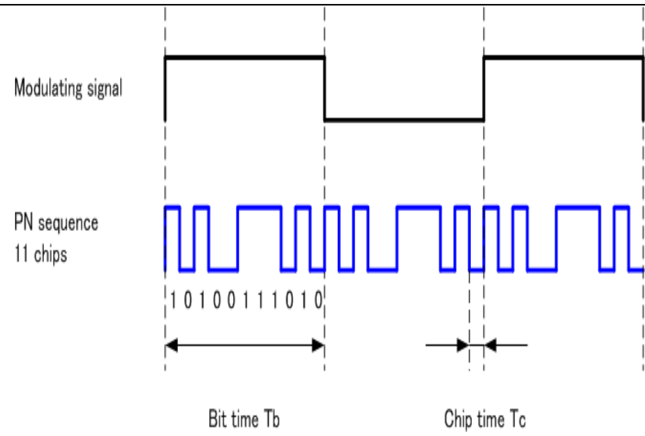
The signal that enters the antenna of the receiver includes outside interference waves and noise. If this signal is despread, the signal component returns to a narrowband modulated wave and the interference components are diffused, expanding the spectrum infinitely so that its power density falls. Therefore, by inputting the signal with frequency band restricted using a BPF, the interference component power that falls into the demodulation frequency band is reduced. The occurrence of errors is calculated using a stochastic process, so ultimately, using a spread spectrum results in fewer errors, and this is why spread spectrum communication is resistant to interference.

**Demodulation:**

Demodulation is normal narrowband demodulation. The local signal is regenerated from the receiving wave and after multiplication by the receiving wave, unnecessary components are eliminated with an LPF. Primary modulation uses PSK, so synchronous detection is necessary.

**PNsequence :**

The PN sequence is switched at a far faster speed than the symbol rate of the information signal and its spectrum covers a wide band. For this reason, the spectrum of the modulated wave after primary modulation also covers a wide band. We won't go into detail here, but PN sequences must meet the conditions required for spread spectrum modulation such as the relationship of the numbers 1 and 0.




**Q.6** **Attempt any FOUR of the following:** **16**

**a) Compare analog and digital pulse modulation.** **4M**

**Ans:** (1M for each point)

| SR.NO | PARAMETER | ANALOG PULSE MODULATION | DIGITAL PULSE MODULATION |
|---|---|---|---|
| 1 | Nature of transmitted signal | Pulse with varying parameters(amplitude, width or position of the pulse) | Digital signal i.e in the form of one's and zero's |
| 2 | Noise immunity | Poor | Excellent |
| 3 | Bandwidth requirement | Lower then digital | Higher due to higher bit rate |
| 4 | Multiplexing used | FDM/TDM | TDM |
| 5 | Types | PAM,PPM,PWM | DM,ADM,PCM,DPCM |

**b) Compare FDMA and TDMA (four points)** **4M**

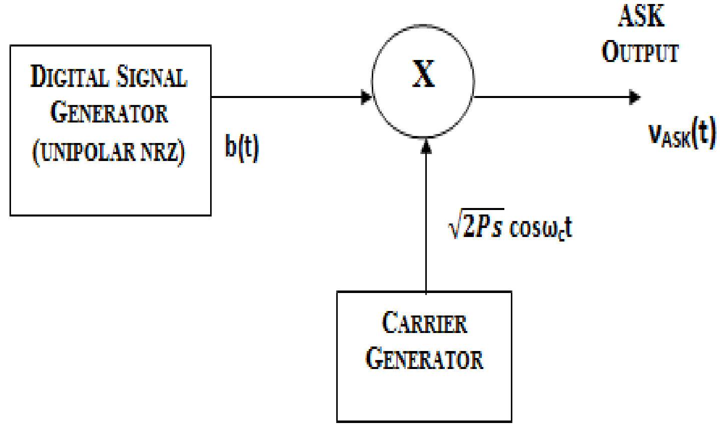
**Ans:** (1M for each point)

| Sr No | PARAMETER | FDMA | TDMA |
|---|---|---|---|
| 1 | Definition | Entire band of frequencies is divided into multiple RF channels/carriers. Each carrier is allocated to different users. | Entire bandwidth is shared among different subscribers at fixed predetermined or dynamically assigned time intervals/slots. |
| 2 | Bandwidth available | Overall bandwidth is shared among many stations. Less BW is available. | Time sharing of satellite transponder takes Place. Hence large BW is available. |
| 3 | Synchronization | Synchronization is not necessary | Synchronization is essential |
| | | Due to nonlinearity of devices Intermodulation products are generated due to interference | Due to incorrect synchronization there can be |

| | | | |
|---|---|---|---|
| 4 | Interference | between adjacent channels. | interference between the adjacent time slots. |
| 5 | Handoff | Hard handoff | Hard handoff |
| 6 | Application | GSM , PDC(pacific digital cellular), Radio, TV | Advanced mobile phone, system(AMPS), Cordless telephone |

**c)** **Draw and explain the block diagram of ASK with suitable diagram.** **4M**

**Ans:** **2M**




*ASK Transmitter*

**Explanation of ASK Transmitter** **2M**

- The ASK technique of binary modulation is illustrated in Figure where modulating signal consists of unipolar pulses.
- Because in this case the carrier is switched ON and OFF, this method is also known as *ON-OFF keying*.
- For the entire time the binary input is high, the output is a constant amplitude, constant frequency signal and for the entire time the binary input is low, the carrier is off.
- Ps is signal power given by $(\text{Amplitude})^2/2$
- ASK is given by:
- $v_{ASK}(t) = b(t)\cos\omega_c t$

**d)** **Using Shannon's theorem, compute the maximum bit rate for a channel having bandwidth 3100 Hz and signal to noise ratio 20 dB.** **4M**

**Ans:** **4M**

Solution

To calculate Maximum bit rate -

Maximum bit rate = $R_{max} = B \log_2 \left[1 + \frac{S}{N}\right]$

given data

$B$ = bandwidth = 3100 Hz

$\frac{S}{N}$ = 20 dB = 100 $\left[\because dB = 20\log(S/N)\right]$

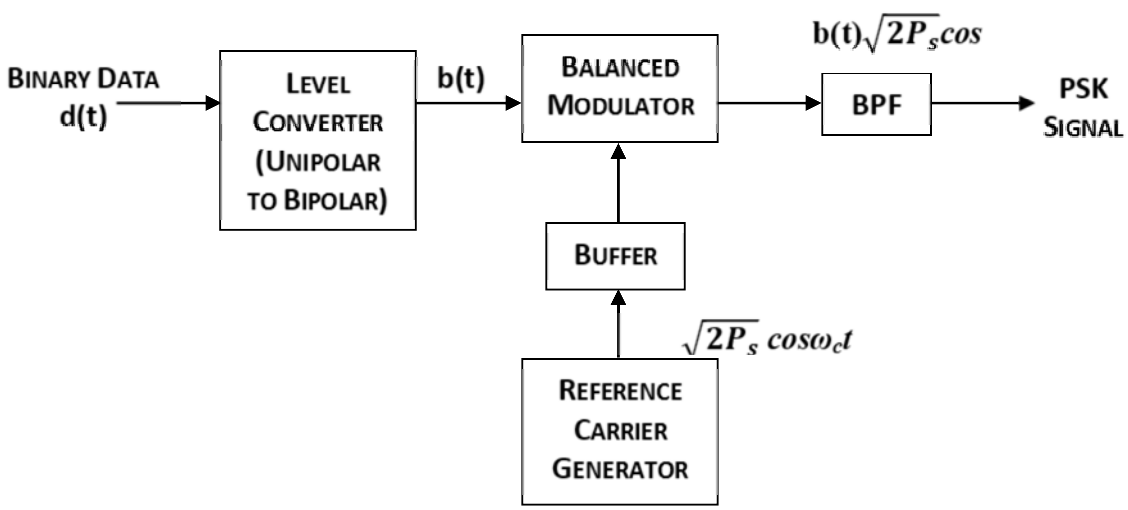
$R_{max} = 3100 \log_2 [1 + 100]$

$= 3100 \frac{\log_{10} 101}{\log_{10} 2}$

= 20640 bits/sec

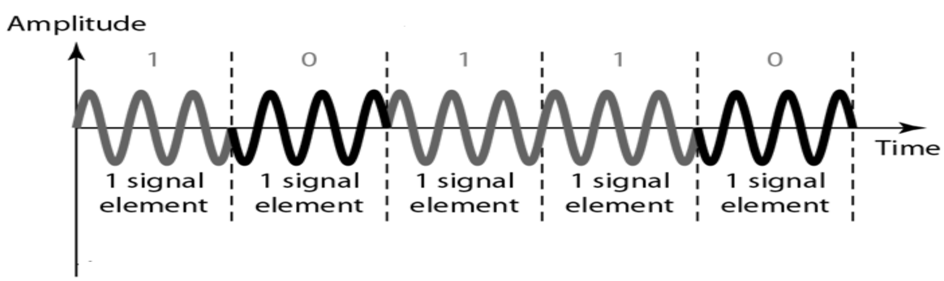
**e) Explain BPSK signal generation with block diagram and waveform.** **4M**

**Ans:** The generation block diagram of BPSK is shown below **2M**




**BPSK SIGNAL GENERATOR** **1M**

- The unipolar binary input is converted into a bipolar signal through a level converter.
- The two dc levels generated by the binary comparator (level converter) alters the flow of current through the balanced modulator resulting in the phase change of the carrier generated by the local oscillator (reference carrier generator) between 0° and 180°.

| | | | |
|---|---|---|---|
| | | • The BPF is required in order to maintain channel bandwidth restriction and to reduce *ISI (Inter-symbol Interference)*.<br>• If the data = 1, voltage level = +1, hence the o/p will be = +1x $\sqrt{2P_s} cos\omega_c t$<br>• If the data = 0, voltage level = -1, hence the o/p will be = -1x $\sqrt{2P_s} cos\omega_c t$<br>$\sqrt{2P_s}(-cos\omega_c t)=\sqrt{2P_s}\,(cos\omega_c t + 180^0)$<br>• Mathematically, BPSK is given by:<br>*For binary 1,* $V_{BPSK}(t) = \sqrt{2P_s} cos\omega_c t$<br>*For binary 0,* $V_{BPSK}(t) = \sqrt{2P_s} cos\,(\omega_c t + 180°)$<br>**BPSK WAVEFORMS**<br><br> | 1M |