PAPER-1

BAPATLA ENGINEERING COLLEGE:BAPATLA

III/IV B.TECH DEGREE EXAMINATION

EI DEPARTMENT

ANALOG AND DIGITAL COMMUNICATION(18EID11)

FEB,2021,Fifth semester

1.Answer all questions. 1*10=10M

1.a.Why we need modulation?
1.b.Define Sensitivity.
1.c.Mention the Efficiency of DSB-SC.
1.d.Mention the significance of Carson's Rule.
1.e.Define companding.
1.f.Define bit rate .
1.g.Define Quadrature Amplitude modulation (QAM).
1.i.Define MSC.
1.j. What are the links in satellite communication?

Answer any Four questions. 4*10=40M

2.a.When the modulation percentage is 75,an AM transmitter produces 10KW.How much of this is carrier power?What would be the percentage power saving if the carrier and one of the sidebands were suppressed before transmission took place? 4M

2.b.Define Amplitude modulation and modulation index.Derive the relation between the output power of AM transmitter and depth of modulation. 6M

3.a.Explain about NBFM and show that it is similar to Amplitude Modulation except the sign of the lower sideband.(proof with bessel or mathematical derivation 4m +phasor diagrams -2m)

3.b.Explain about pre emphasis and de-emphasis circuits and explain why pre-emphasis and de-emphasis circuits are used. 4M

4.a.Explain in detail about the operation of PCM transmitter and receiver. 6M

4.b.Explain the generation of PWM and PPM. 4M

5.a.Describe FSK transmitter and Receiver. 7M

5.b. Determine the :

(i)peak frequency deviation (ii) minimum bandwidth (iii) baud for FSK signal with a mark frequency of 49 kHz, space frequency of 51 kHz, and input bit rate of 2 kbps. 3M

6.a. Describe with neat diagram, the operation of a QPSK modulator. Draw its phasor and constellation diagram. 6M

6.b. sketch the waveforms of in-phase and quadrature components of the QPSK signal in response to the input binary sequence 10011100. 4M

7. a.Write shanno-fano code algorithm with example. 5M

7.b. A DMS X has four symbols x1,x2,x3 and x4 with P(x1)=1/2, P(x2)=1/4, P(x3)= P(x4)=1/8.

Construct a shannon-fano code for X.Show that this code has the optimum property that $n_i=I(X_i)$ and that the code efficiency is 100 percent. 5M

8.Explain the concept of GSM. 10M

9.Define Handoff and explain its types in detail. 10M

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FEB,2021,Fifth semester

SCHEME AND SOLUTION

1.Answer all questions. 1*10=10M

1.a.Why we need modulation?

A:
1.Low frequency signal their own cannot travel long distances.
2.Simultaneous transmission message signal on a single channel is not possible.
3.For direct transmission of message signals,an antenna height of 1000km is needed which is not possible practically.

1.b.Define Sensitivity.
A:It is defined as the minimum RF signal level that can be detected at the input to the receiver and still produce a usable demodulated information signal.

1.c.Mention the Efficiency of DSB-SC.
A:Efficiency=50%

1.d.Mention the significance of Carson's Rule.
A:Maximum bandwidth using Carson's rule
B=2[delta f max+fm(max)]hz

1.e.Define companding.
A:companding is a process of compressing and then expanding.
With companding systems,the higher amplitude analog singals are compressed prior to transmission and then expanded in the receiver.
companding is a means of improving the dynamic range of a communication system.

1.f.Define bit rate .
A: Bit rate is defined as number of bits transmitted during one second between the transmitter and receiver.

1.g.Define Quadrature Amplitude modulation (QAM).
A: QAM is a form of digital modulation similar to PSK except the digital information is contained in both the amplitude and phase of the transmitted carrier.

(or)
QAM is defined as changing the amplitude as well as the frequency of the carrier signal with respect to the binary information or digital signal.

1.i.Define MSC.
A: **Mobile services switching center (MSC)**—The MSC performs the telephony switching functions of the system. It controls calls to and from other telephone and data systems. It also performs such functions as toll ticketing, network interfacing, common channel signaling, and others.

1.j. What are the links in satellite communication?
A: i) Uplink ii) Downlink iii) Crosslink

Answer any Four questions. **4*10=40M**

2.a.When the modulation percentage is 75,an AM transmitter produces 10KW.How much of this is carrier power?What would be the percentage power saving if the carrier and one of the sidebands were suppressed before transmission took place? 4M

A:M=0.75,Pt=10kw,Pt=Pc(1+0.75*0.75) -2m
Pc=10/1.5625=6.4KW -1m
Percentage power saving=single sideband power/power transmitted
=$P_c m^2/4$ / $P_c m^2/4$
=100% -1m

2.b.Define Amplitude modulation and modulation index.Derive the relation between the output power of AM transmitter and depth of modulation. 6M

A: We know that AM signal has three components : Unmodulated carrier, sideband and upper sideband. Hence total power of AM wave is the sum of power P_c and powers in the two sidebands P_{USB} and P_{LSB}. i.e.,

$$P_{Total} = P_c + P_{USB} + P_{LSB}$$
$$= \frac{E_{carr}^2}{R} + \frac{E_{LSB}^2}{R} + \frac{E_{USB}^2}{R} \quad \ldots$$

Here all the three voltages are rms values and R is characteristic impedence of antenna in which the power is dissipated. The carrier power is,

$$P_c = \frac{E_{carr}^2}{R} = \frac{\left(E_c / \sqrt{2}\right)^2}{R}$$

$$= \frac{E_c^2}{2R} \quad \text{... (1.2.16)}$$

The power of upper and lower sidebands is same. i.e.,

$$P_{LSB} = P_{USB} = \frac{E_{SB}^2}{R} \qquad \text{Here } E_{SB} \text{ is rms voltage of sidebands.}$$

From equation (1.2.13) we know that the peak amplitude of both the sidebands is $\frac{m E_c}{2}$. Hence,

$$E_{SB} = \frac{m E_c / 2}{\sqrt{2}}$$

$$\therefore \quad P_{LSB} = P_{USB} = \left(\frac{m E_c / 2}{\sqrt{2}}\right)^2 \times \frac{1}{R}$$

$$= \frac{m^2 E_c^2}{8R} \quad \text{... (1.2.17)}$$

Hence the total power (equation 1.2.15) becomes,

$$P_{Total} = \frac{E_c^2}{2R} + \frac{m^2 E_c^2}{8R} + \frac{m^2 E_c^2}{8R}$$

$$= \frac{E_c^2}{2R}\left[1 + \frac{m^2}{4} + \frac{m^2}{4}\right]$$

$$\therefore \quad \boxed{P_{Total} = P_c\left(1 + \frac{m^2}{2}\right)} \quad \text{... (1.2.19)}$$

$$\frac{P_{Total}}{P_c} = 1 + \frac{m^2}{2} \quad \text{... (1.2.20)}$$

This equation relates total power of AM wave to carrier power. Maximum value of modulation index, $m = 1$ to avoid distortion. At this value of modulation index, $P_{Total} = 1.5\, P_c$. From above equation we have,

$$\frac{m^2}{2} = \frac{P_{Total}}{P_c} - 1$$

$$\therefore \quad m = \sqrt{2\left(\frac{P_{total}}{P_c} - 1\right)}$$

3.a.Explain about NBFM and show that it is similar to Amplitude Modulation except the sign of the lower sideband.(proof with bessel or mathematical derivation 4m +phasor diagrams - 2m)

a: A standard FM signal is represented by

, where $\beta_f = \frac{k_f A_m}{f_m} = \frac{\Delta f}{f_m}$ is called frequency modulation index, where again $\Delta f = k_f A_m$ is known as frequency deviation. Then by expanding, we get

$$s_{FM}(t) = A_c \cos\left[\omega_c t + \beta_f \sin 2\pi f_m t\right]$$
$$= A_c\left[\cos\omega_c t \cdot \cos(\beta_f \sin 2\pi f_m t) - \sin\omega_c t \cdot \sin(\beta_f \sin 2\pi f_m t)\right]$$

For $\beta_f << \pi/2$, $\cos(\beta_f \sin 2\pi f_m t) = 1$ and $\sin(\beta_f \sin 2\pi f_m t) = \beta_f \sin 2\pi f_m t$.

Therefore, the narrowband FM is described by

$$s_{NBFM}(t) = A_c\left[\cos\omega_c t - \sin\omega_c t \cdot \beta_f \sin 2\pi f_m t\right] = A_c \cos\omega_c t - A_c\beta_f \sin\omega_c t \cdot \sin 2\pi f_m t$$
$$= A_c \cos\omega_c t - \frac{A_c\beta_f}{2}\left[\cos(\omega_c - \omega_m) - \cos(\omega_c + \omega_m)\right]$$
$$= A_c \cos\omega_c t - \frac{A_c\beta_f}{2}\cos(\omega_c - \omega_m) + \frac{A_c\beta_f}{2}\cos(\omega_c + \omega_m)$$

The single-tone *Amplitude Modulation* equation is given by

$$s_{AM}(t) = A_c \cos\omega_c t + \frac{\mu A_c}{2}\cos(\omega_c - \omega_m)t + \frac{\mu A_c}{2}\cos(\omega_c + \omega_m)t$$

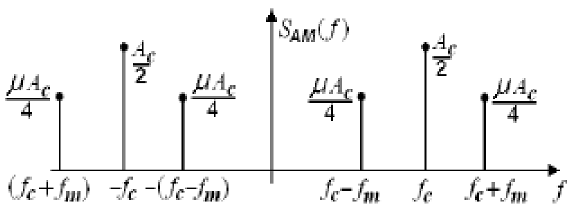
The equation $s_{NBFM}(t)$ resembles the AM ($s_{AM}(t)$) except that in narrowband FM, the phase of LSB signal reversed and the resultant sideband vector sum is always in-phase quadrature with the carrier.

Thus the FM gives rise to phase variations with very small amplitude change ($\beta_f << \pi/2$), while AM gives amplitude variations with no phase deviation.

The frequency spectrum of *Narrow Band Frequency Modulation* is represented by

$$S_{FM}(f) = \frac{A_c}{2}\left[\delta(f - f_c) + \delta(f + f_c)\right] - \frac{A_c\beta_f}{4}\left[\delta(f - f_c + f_m) + \delta(f + f_c - f_m)\right]$$
$$+ \frac{A_c\beta_f}{4}\left[\delta(f - f_c - f_m) + \delta(f + f_c + f_m)\right]$$

The spectrum of this narrowband FM wave is shown in Fig 4. For comparison AM spectrum is also shown in figure.



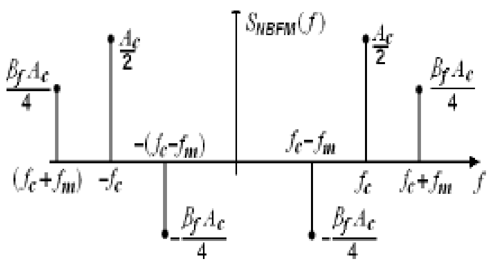
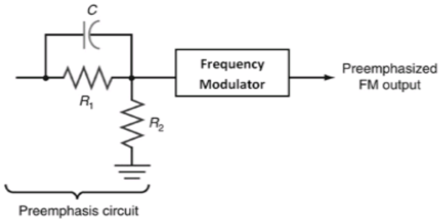


Fig Q6a .i.conventinal AM Fig Q6a.ii.Narrowband FM

3.b.Explain about preemphasis and de-emphasis circuits and explain why pre-emphasis and de-emphasis circuits are used. (5M)

A:**Pre-emphasis:** The noise suppression ability of FM decreases with the increase in the frequencies. Thus increasing the relative strength or amplitude of the high frequency components of the message signal before modulation is termed as Pre-emphasis. The Fig3 below shows the circuit of pre-emphasis.



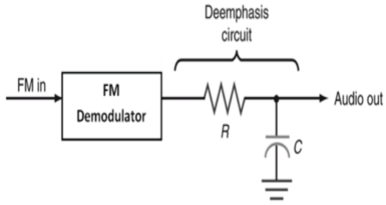
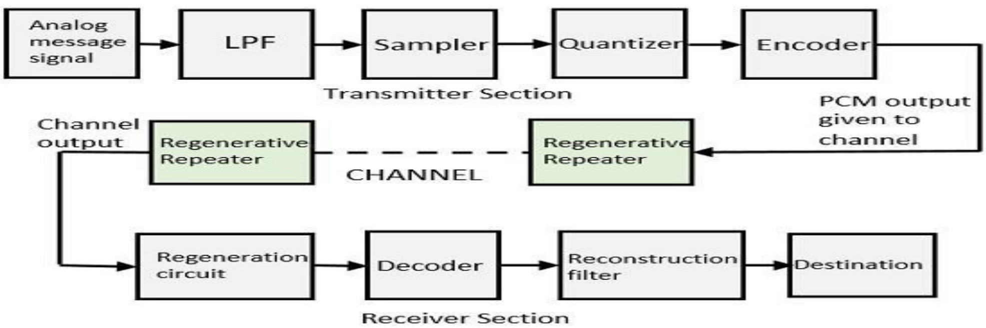


Fig 3.b.i.Preemphasis circuit Fig 3.b.ii.Demphasis circuit

- **De-emphasis:** In the de-emphasis circuit, by reducing the amplitude level of the received high frequency signal by the same amount as the increase in pre-emphasis is termed as De-emphasis. The Fig4. below shows the circuit of de-emphasis.
- The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side.
- Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation. To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed. Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant.
- The de-emphasis process ensures an modulation index and frequency deviations that the high frequencies are returned to their original relative level before amplification.
- Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator which allows only low frequencies to pass.

4.a.Explain in detail about the operation of PCM transmitter and receiver. 6M

(FIG2M+EXPLANATION 4M)



a: The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals,**

decoding, and **reconstruction** of the quantized pulse train. Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.

Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W**of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections (LPF, Sampler, and Quantizer) will act as an analog to digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal. Hence, the Pulse Code Modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

4.b.Explain the generation of PWM and PPM. 4M

A: Generation of PWM:PWM signal can be generated by using a comparator, where modulating signal and sawtooth signal form the input of the comparator. It is the simplest method for PWM generation.

The PWM generation is explained with the help of the Fig.4.b given below.

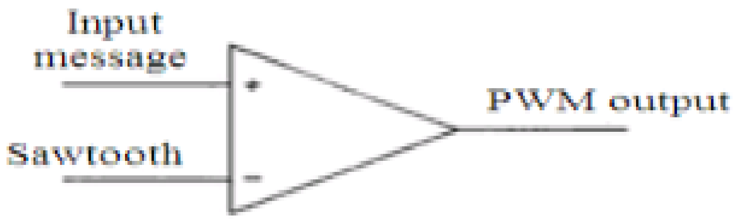


Fig4.b.i PWM generation by a comparator

As shown in the figure, one input of the comparator is fed by the input message or

modulating signal and the other input by a sawtooth signal which operates at carrier frequency.
Considering both ±ve sides, the maximum of the input signal should be less than that of sawtooth signal.

- The comparator will compare the two signals together to generate the PWM signal at its output as shown in the third waveform of Fig8.b.ii.
- The rising edges of the PWM signal coincides with the falling edge of the sawtooth signal.
- When the sawtooth signal is at the minimum value which is less than the minimum of the input signal, then the positive input of the comparator is at higher potential which gives the comparator output as positive.
- When the sawtooth signal rises and is at the maximum value, the negative input of the comparator is at higher potential, which will produce the comparator output to be negative.
- Thus the input signal magnitude determines the comparator output and its potential, which then decides the width of the pulse generated at the output.
- In other words we can say that the width of the pulse generated signal is directly

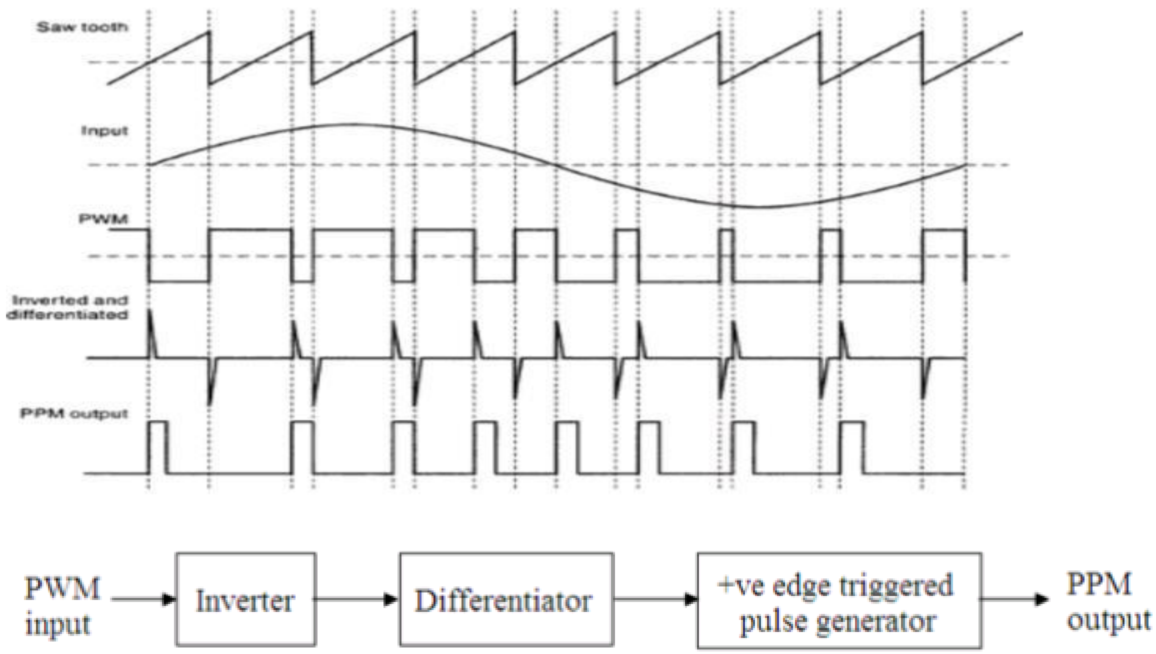


Fig .4.b.ii & iii PWM and PPM signal generation

5.a.Describe FSK transmitter and Receiver. 7M

A:

FSK TRANSMITTER:
Figure 2-6 shows a simplified binary FSK modulator, which is very similar to a conventional FM modulator and is very often a voltage-controlled oscillator (VCO).The center frequency (fc) is chosen such that it falls halfway between the mark and space frequencies.

A logic 1 input shifts the VCO output to the mark frequency, and a logic 0 input shifts the VCO output to the space frequency. Consequently, as the binary input signal changes back and forth between logic 1 and logic 0 conditions, the VCO output shifts or deviates back and forth between the mark and space frequencies.

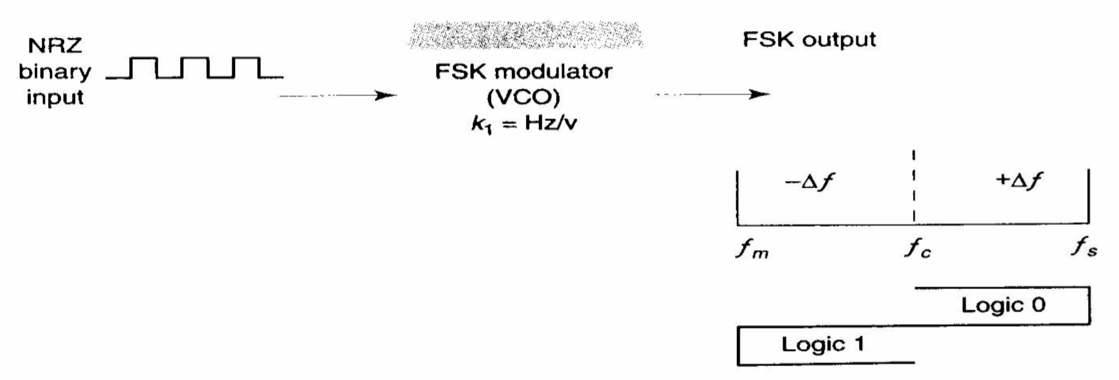


FIGURE a.FSK modulator

A VCO-FSK modulator can be operated in the sweep mode where the peak frequency deviation is simply the product of the binary input voltage and the deviation sensitivity of the VCO.

With the sweep mode of modulation, the frequency deviation is expressed mathematically as

$$f = v_m(t)k_1 \ HZ$$

FSK Receiver

FSK demodulation is quite simple with a circuit such as the one shown in Figure 2-7.

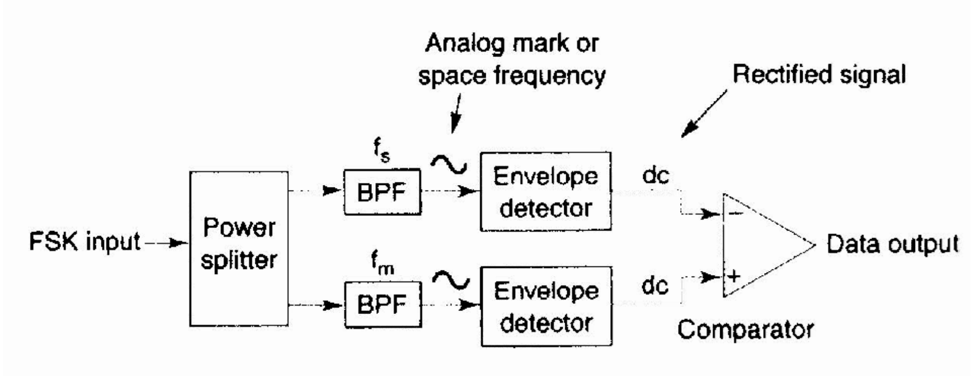


FIGURE b. Noncoherent FSK demodulator

The FSK input signal is simultaneously applied to the inputs of both bandpass filters (BPFs) through a power splitter.The respective filter passes only the mark or only the space frequency on to its respective envelope detector.The envelope detectors, in turn, indicate the total power in each passband, and the comparator responds to the largest of the two powers.This type of FSK detection is referred to as noncoherent detection.

5.b. Determine the :

(i)peak frequency deviation (ii) minimum bandwidth (iii) baud for FSK signal with a mark

frequency of 49 kHz, space frequency of 51 kHz, and input bit rate of 2 kbps. 3M

Solution

a. The peak frequency deviation is determined

$\Delta f = |(f_m - f_s)|$

$\Delta f = |149\text{kHz} - 51\text{ kHz}| / 2 = 1\text{ kHz}$

b. The minimum bandwidth is determined

$B = 2(\Delta f + fb)$

$B = 2(100 + 2000) = 6\text{ kHz}$

c. For FSK, $N = 1$, and the baud is determined as

baud = 2000 / 1 = 2000

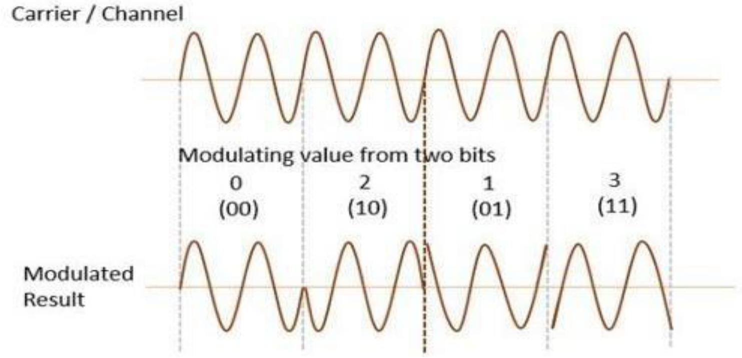
6.a. Describe with neat diagram, the operation of a QPSK modulator. Draw its phasor and

constellation diagram. 6M

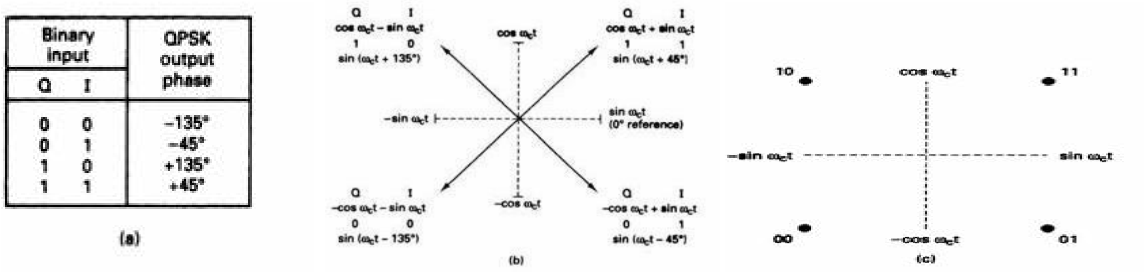
A: QUADRATURE PHASE SHIFT KEYING (QPSK):

This is the phase shift keying technique, in which the sine wave carrier takes four phase reversals such as 0°, 90°, 180°, and 270°.

The following figure represents the QPSK waveform for two bits input, which shows the modulated result for different instances of binary inputs.

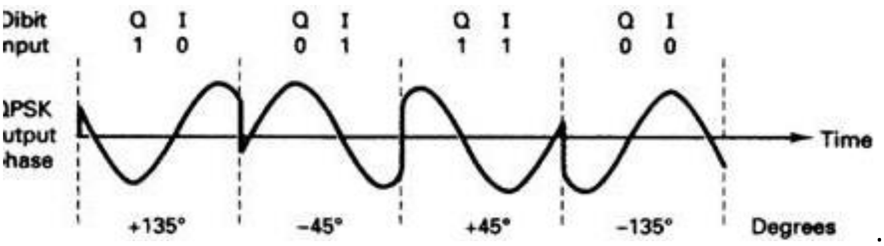


QPSK is a variation of BPSK, and it is also a DSB-SC (Double Sideband Suppressed Carrier) modulation scheme, which sends two bits of digital information at a time, called as **bigits**. Instead of the conversion of digital bits into a series of digital stream, it converts them into bit-pairs. This decreases the data bit rate to half, which allows space for the other users.



6.b. sketch the waveforms of in-phase and quadrature components of the

QPSK signal in response to the input binary sequence 10011100. 4M



7. a.Write shanno-fano code algorithm with example. 5M

A:

An efficient code can be obtained by the following simple procedure, known as *Shannon-Fano algorithm:*

1. List the source symbols in order of decreasing probability.
2. Partition the set into two sets that are as close to equiprobable as possible, and assign 0 to the upper set and 1 to the lower set.

3. Continue this process, each time partitioning the sets with as nearly equal probabilities as possible until further partitioning is not possible.

7.b. A DMS X has four symbols x1,x2,x3 and x4 with P(x1)=1/2, P(x2)=1/4, P(x3)= P(x4)=1/8.

Construct a shannon-fano code for X.Show that this code has the optimum property that $n_i=I(X_i)$ and that the code efficiency is 100 percent. 5M

x_i	$P(x_i)$	Step 1	Step 2	Step 3	Code
x_1	$\frac{1}{2}$	0			0
x_2	$\frac{1}{4}$	1	0		10
x_3	$\frac{1}{8}$	1	1	0	110
x_4	$\frac{1}{8}$	1	1	1	111

$$I(x_1) = -\log_2 \frac{1}{2} = 1 = n_1 \quad I(x_2) = -\log_2 \frac{1}{4} = 2 = n_2$$

$$I(x_3) = -\log_2 \frac{1}{8} = 3 = n_3 \quad I(x_4) = -\log_2 \frac{1}{8} = 3 = n_4$$

$$H(X) = \sum_{i=1}^{4} P(x_i)I(x_i) = \frac{1}{2}(1) + \frac{1}{4}(2) + \frac{1}{8}(3) + \frac{1}{8}(3) = 1.75$$

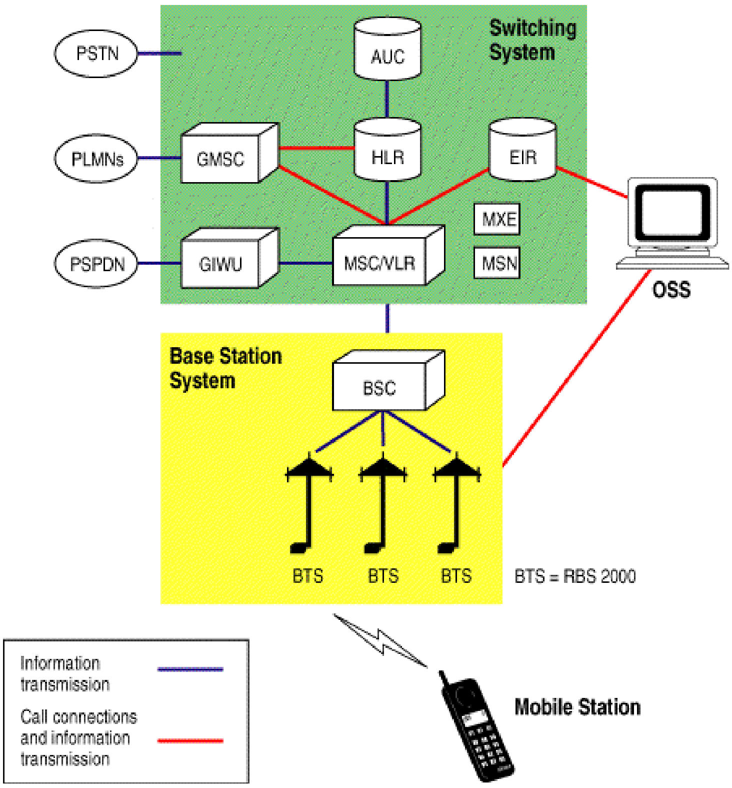
$$L = \sum_{i=1}^{4} P(x_i)n_i = \frac{1}{2}(1) + \frac{1}{4}(2) + \frac{1}{8}(3) + \frac{1}{8}(3) = 1.75$$

$$\eta = \frac{H(X)}{L} = 1 = 100\%$$

8.Explain the concept of GSM. 10M

Global system for mobile communication (GSM) is a globally accepted standardfor digital cellular communication. GSM is the name of a standardization groupestablished in 1982 to create a common European mobile telephone standardthat would formulate specifications for a pan-European mobile cellular radiosystem operating at 900 MHz. It is estimated that many countries outside ofEurope will join the GSM partnership.

Throughout the evolution of cellular telecommunications, various systems havebeen developed without the benefit of standardized specifications. This presented many problems directly related to compatibility, especially with the development of digital radio technology. The GSM standard is intended to address these problems.



GSM network elements

GSMSERVICES

• Telephone Services
o Explanation
• Data Services
o Explanation
• Supplementary Services
o Explanation
GSM Features:
o Explanation
GSM System Architecture:
• Base Station Subsystem (BSS)
• Network and Switching Subsystems (NSS)
• Operation Support Subsystem (OSS)
GSM interface:
• Abis interface
• A – interface
GSM channel Types:
• Traffic channels
• Control channels

9.Define Handoff and explain its types in detail. 10M

A: Handoffs

• When a user/call moves to a new cell, then a new base station and new channel should be assigned (handoff)
• Handoffs should be transparent to users, while their number should be kept to minimum

• A threshold in the received power (Pr, handoff) should be determined to trigger the handoff process. This threshold value should be larger than the minimum acceptable received power (Pr, acceptable)
• Define: Δ=Pr,handoff - Pr,acceptable
– If Δ is large then too many handoffs
– If Δ is small then insufficient time to complete a handoff
• In order to correctly determine the beginning of handoff, we need to determine that a drop in the signal strength is not due to the momentary (temporary) bad channel condition, but it is due to the fact that the mobile is moving away from BS.
• Thus the BS needs to monitor the signal level for a certain period of time before initiating a handoff. The length of the time (running average measurements of signal) and handoff process depends on speed and moving pattern.
• First generation systems typical time interval to make a handoff was 10 seconds (large Δ). Second generations and after typical time interval to make a handoff is 1-2 seconds (small Δ).
• First generation systems: handoff decision was made by BS by measuring the signal strength in reverse channels.
• Second generation and after: Mobile Assisted Hand-Off (MAHO).
Mobiles measure the signal strength from different neighboring BSs. Handoff is initiated if the signal strength from a neighboring BS is higher than the current BS's signal strength.
Cell Dwell Time
• It is the time over which a call maybe maintained within a cell (without handoff).
• It depends on: propagation, interference, distance between BS and MS, speed and moving pattern (direction), etc.
• Highway moving pattern: the cell dwell time is ar.v. with distribution highly concentrated around the mean.
• Other micro-cell moving patterns mix of different user types with large variations of dwell time (around the mean).
Prioritizing Handoffs
• **Guard Channels:** Fraction of total bandwidth in a cell is reserved for exclusive use of handoff calls. Therefore, total carried traffic is reduced if fixed channel assignment is used. However, if dynamic channel assignment is used the guard channel mechanisms may offer efficient spectrum utilization.
– Number of channels to be reserved: If it is low (under-reservation) the QoS on handoff call blocking probability can not be met. If reservation is high (over-reservation) may result in waste of resources and rejection of large number of new calls.
– Static and Dynamic schemes: Advantage of static scheme is its simplicity since no communication and computation overheads are involved. However problems of under-reservation and over reservations may occur if traffic does not conform to prior knowledge. Dynamic schemes may adjust better to changing traffic conditions.
Prioritizing Handoffs
• **Queuing Handoffs:** The objective is to decrease the probability of forced determination of a call due to lack of available channels. When a handoff call (and in some schemes a new call) can not be granted the required resources at the time of its arrival, the request is put in a queue waiting for its admitting conditions to be met.
– This is achieved because there is a finite time interval between the time that the signal of a call drops below the handoff threshold, and the time that the call is terminated due to low

(unacceptable) signal level. Queuing and size of buffer depends on traffic and QoS. Queueing in wireless systems is possible because signaling is done on separate control channels (without affecting the data transmission channels).

- According to the types of calls that are queued, queuing priority schemes are classified as: handoff call queuing, new call queuing and handoff/new call queuing (handoff calls are given non-preemptive priority over new calls).

Practical Issues (Capacity/Handoff)

- To increase capacity, use more cells (add extra sites).
- Using different antenna heights and powers, we can provide “large” and “small” cells co-located at a signal location (it is used especially to handle high speed users and low speed users simultaneously.
- Reuse partitioning (use of different reuse patterns)
- Cell splitting: Change cell radius R and keep co-channel reuse ratio (D/R) unchanged. If R’=R/2 than the transmit power needs to be changed by (1/2)4 = 1/16.
- Another way is to keep cell radius R unchanged and decrease D/R ratio required (that is decrease the number of cells in a cluster). To do this it is required to decrease interference without decreasing transmit power.
- Sectoring: Use directional antennas (instead of omni-directional) and therefore you receive interference from only a fraction of the neighboring cells.
- Hard handoffs vs. soft handoffs: more than one BSs handle the call during handoff phase (used in CDMA systems).

END