UNIT-I
Chapter-2

2-1 **Define the following terms as they relate to electrical signals: *amplitude, frequency, period, phase, periodic signal, electrical noise, frequency spectrum,* and *bandwidth.***

ANSWER:

Amplitude:- Amplitude is analogous to magnitude or displacement. The amplitude of a signal is the magnitude of the signal at any point on the waveform. Signal amplitude is generally represented on the vertical axis of a waveform graph. The maximum voltage of a signal in respect to its average value (i.e., the vertical center of the waveform) is called its *peak amplitude* or *peak voltage (V).*

Frequency:- A ***cycle*** is one complete variation in the signal, and the *period* is the time the waveform takes to complete one cycle (T). One cycle constitutes 360 degrees. Frequency is technically measured in cycles per second and expressed in hertz.

Periodic Signal:- The signal which repeats itself at regular intervals is called a periodic signal.

Phase:- The phase of a signal is measured in degrees or radians (360 degrees or 2π radians) with respect to a reference point. A phase of 45 degrees means the waveform is shifted one-eighth of a cycle from the reference. A phase shift of 180 degrees corresponds to a shift of half a cycle (i.e., a complete inversion of a sine wave), and a phase shift of 360 degrees corresponds to a shift of one complete cycle. Phase, like time, is shown on the horizontal axis of a signal graph.

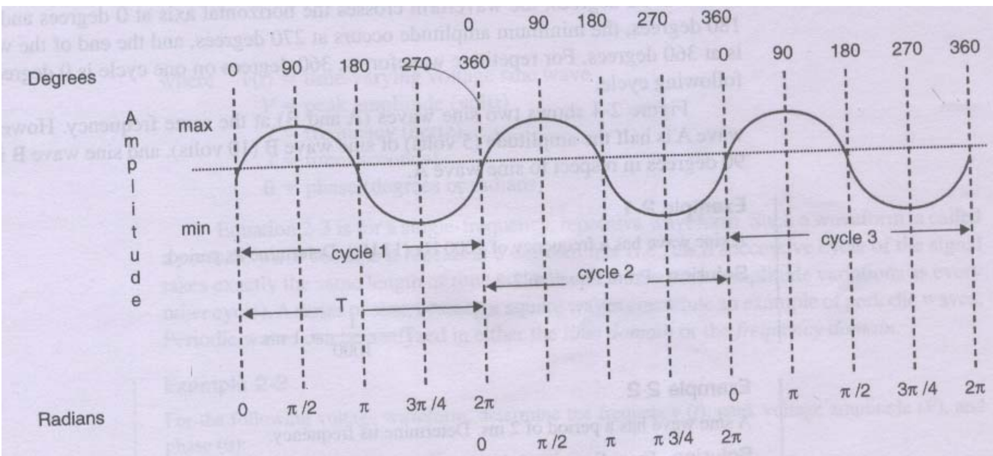


Figure 2-3 Three Sine waves showing amplitude, frequency and phase

Electrical Noise:- *Electrical noise* is defined as any undesirable electrical energy that falls within the pass- band of the signal.

Frequency Spectrum:- The ***frequency spectrum*** of a waveform Consists of all the frequencies contained in the waveform and their respective amplitudes plotted in the frequency domain.

Bandwidth:- The term *bandwidth* can be used in several ways. The bandwidth of a frequency spectrum is the range of frequencies contained in the spectrum. The bandwidth of an information signal is simply the difference between the highest and lowest frequencies contained in the information, and the bandwidth of a communications channel is the difference between the highest and lowest frequencies that the channel will allow to pass through it (i.e., its *pass band).* The bandwidth of a communications channel must be sufficiently large (wide) to pass all significant information frequencies. In other words, the bandwidth of a communications channel must be equal to or greater than the bandwidth of the information signal.

2-2 **Describe the following wave symmetries: even, odd, and half-wave.**

ANSWER:

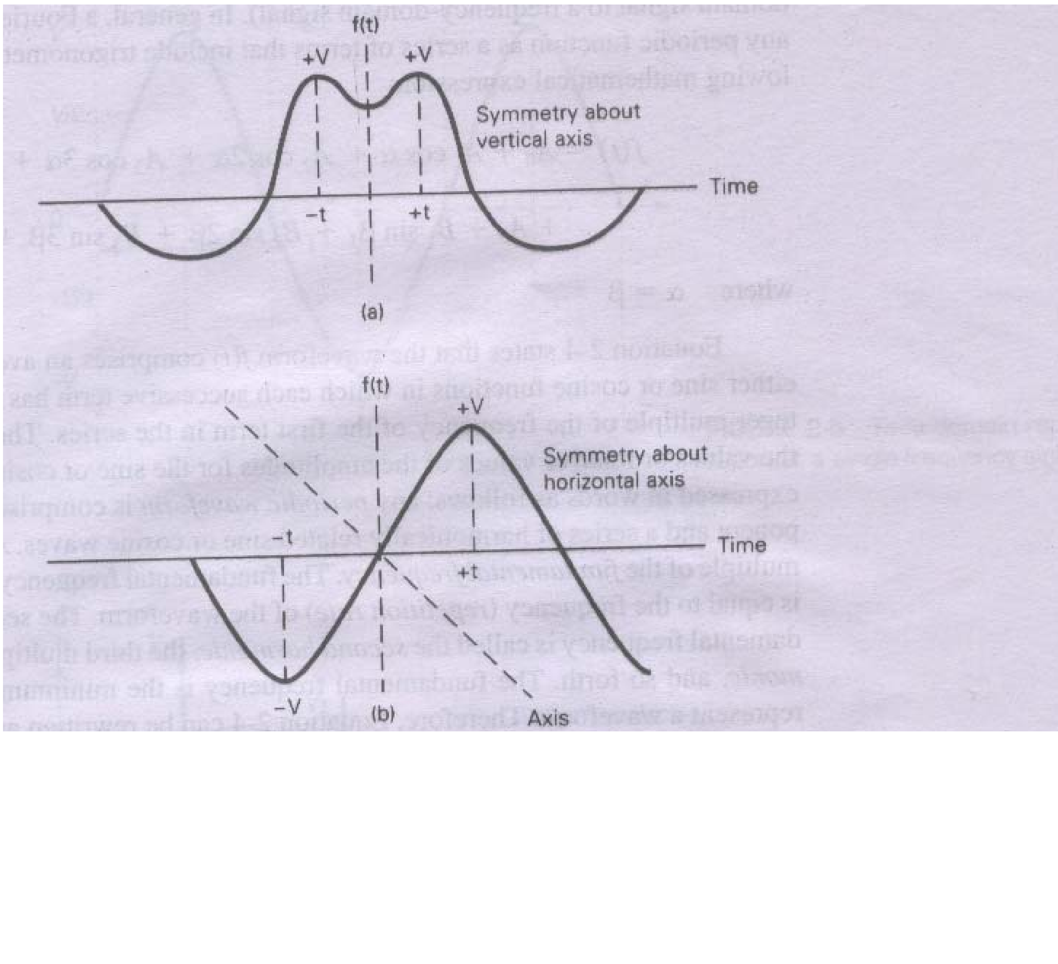
Wave symmetry:- wave symmetry describes the symmetry of a waveform in the time domain, that is, its relative position with respect to the horizontal (time) and vertical (amplitude) axes.

Even symmetry:-If a periodic voltage waveform is *symmetric* about the vertical axis, it is said to have *axes,* or *mirror, symmetry* and is called an *even function.* The signal contains a dc component and the cosine terms (note that a cosine wave is itself an even function). Even functions satisfy the condition $f(t) = f(-t)$

Equation states that the magnitude and polarity of the function at ***+t*** is equal to the magnitude and polarity a-**-t.** A waveform that contains only the even functions is shown in on Figure 2-7a.

Odd symmetry:- If a periodic voltage waveform is symmetric about a line midway between the vertical axis and the negative horizontal axis (i.e.. the axes in the second and fourth quadrants) and passing through the coordinate origin, it is said to have *point,* or *skew, symmetry* and is called an *odd function.* The signal contains a dc component and the sine wave terms. Odd functions satisfy the condition $f(t) = -f(-t)$

Equation states that the magnitude of the function at *+t* is equal to the negative of the magnitude at ***t*** (i.e., equal in magnitude but opposite in sign). A periodic waveform that contains only the odd functions is shown in Figure 2-7b.



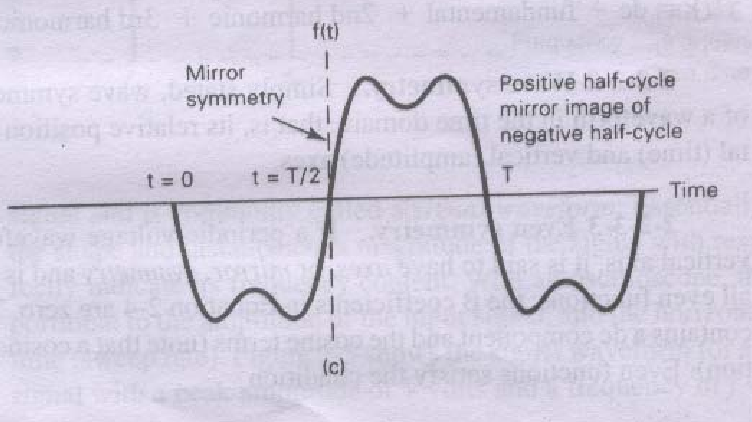


FIGURE 2-7 Wave symmetries: (a) even symmetry; (b) odd symmetry; (c) half-wave symmetry

Half-wave symmetry:- If a periodic voltage waveform is such that the waveform for the first half cycle (t = 0 to t = *T/2)* repeats itself except with the opposite sign for the second half cycle (t = *T12* to t = T), it is said to have *half-wave symmetry.* For all waveforms with half-wave symmetry, the even harmonics in the series for both the sine and cosine terms are zero. Therefore, half-wave functions satisfy the condition $f(t) = -f(T + t)/2$

***2-3* Give a brief description of the following forms of electrical noise: man-made, thermal, correlated, and impulse.**

ANSWER:

Man-made noise:- *Man-made noise* is noise that is produced by mankind. The predominant sources of man-made noise are spark-producing mechanisms, such as commutators in electric motors, automobile ignition systems, ac power-generating and switching equipment, and fluorescent lights. Man-made noise is impulsive in nature and contains a wide range of frequencies that are propagated through space in the same manner as radio waves. Man-made noise is most intense in the more densely populated metropolitan and industrial areas and is, therefore, sometimes called *industrial noise.*

Thermal noise:- *Thermal noise* is associated with the rapid and random movement of electrons within a conductor due to thermal agitation and is present in all electronic components and communications systems. Because thermal noise is uniformly distributed across the entire electromagnetic frequency spectrum, it is often referred to as *white noise* (analogous to the color white containing all colors [frequencies] of light). Thermal noise is a form of additive noise, meaning that it cannot be eliminated and that it increases in intensity with the number of devices in a circuit and with circuit length. Therefore, thermal noise sets the upper bound on the performance of a communications system.

Correlated noise:- Correlated noise is produced by *nonlinear amplification* and includes *harmonic distortion* and *inter modulation distortion,* both of which are forms of *nonlinear distortion. Harmonic distortion* occurs when unwanted *harmonics* of a signal are produced through nonlinear amplification *(nonlinear mixing).*Harmonics are integer multiples of the original signal. The original signal is the first harmonic and is called the fun*damental frequency.* Two times the original signal frequency is the second harmonic, three times is the third harmonic, and so forth. *Amplitude distortion* is another name for harmonic distortion. A more meaningful measurement is total harmonic distortion (TDH), which is the ratio of the quadratic sum of the rms values of all the higher harmonics to the rms value of the fundamental frequency.

As the figure 2-12a shows, the output spectrum contains the original input frequency plus several harmonics ($2f_1$, $3f_1$, $4f_1$) that were not part of the original signal.

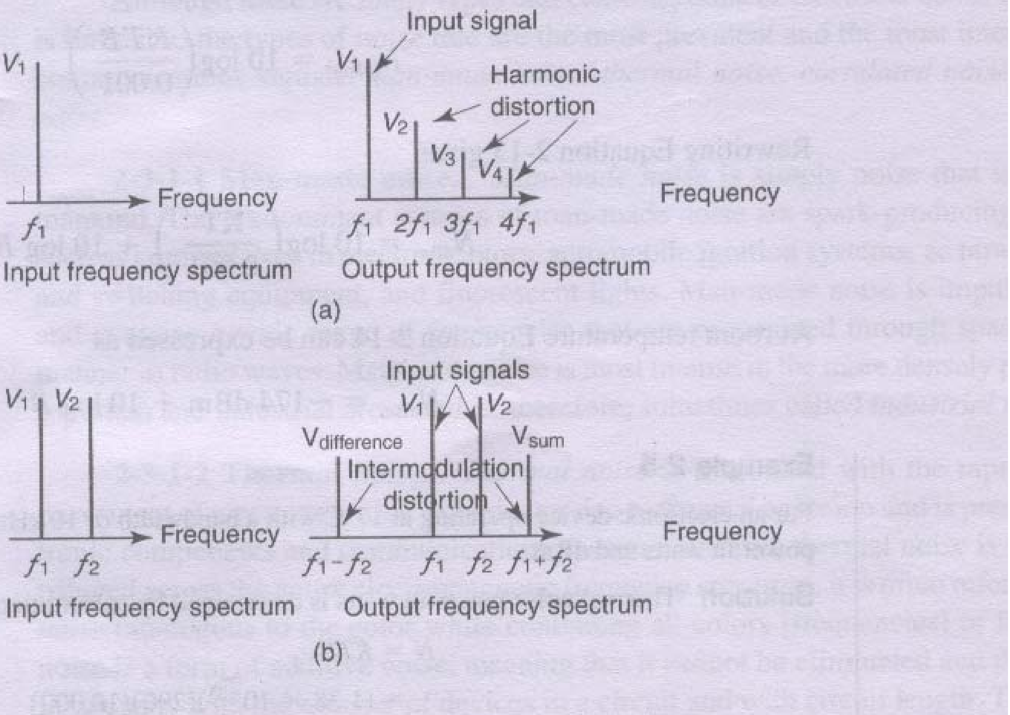


FIGURE 2-12 Correlated noise: (a) harmonic distortion; (b) intermodulation distortion

Inter modulation distortion is the generation of unwanted *sum* and *difference* frequencies produced when two or more signals are amplified in a nonlinear device. The sum and difference frequencies are called *cross products.* The emphasis here is on the word *un anted* because in communications circuits it is often desirable to produce harmonics or to mix two or more signals to produce sum and difference frequencies. Cross-product frequencies can interfere with the information signals in a circuit or with the information signals in other circuits.

Figure 2-12b shows the input and output frequency spectrums for a nonlinear device with two input frequencies ($f_1$andf_2). .

Impulse noise:- *Impulse noise* is characterized by high-amplitude peaks of short duration in the total noise spectrum. As the name implies, impulse noise consists of sudden bursts of irregularly shaped pulses that generally last between a few microseconds and several milliseconds, depending on their amplitude and origin. The significance of impulse noise (hits) on voice communications is often more annoying than inhibitive, as impulse hits produce a sharp popping or crackling sound.

2-4 **Define *analog modulation* and describe the following types: amplitude, Frequency and phase.**

ANSWER:

Analog modulation:- The process of impressing relatively low-frequency information signals onto a high-frequency *carrier signal* is called *modulation. Demodulation* is the reverse process where the received signals are transformed back to their original form.

Analog modulation is used for the transmission of conventional analog signals. such as voice, music, and video, and it not particularly useful for data communications systems.

Amplitude Modulation:- *Amplitude modulation* (AM) is the process of changing the amplitude of a relatively high- frequency carrier signal in proportion to the instantaneous value of the modulating signal (information). In the modulator, the information signals act on or modulate the RF carrier, producing a *modulated wave.*

Figure 2-13 illustrates the relationship among the earner (V_c sin[$2\pi f_c t$]), the modul ating signal (V_m sin[$2\pi f_m t$]), and the modulated wave (V_{am}[t]). The figure shows how an AM waveform is produced when a single-frequency modulating signal acts on a high- frequency carrier signal. Since the output waveform contains all the frequencies that make up the AM signal and it is used to transport the information through the system, the shape of the modulated wave is called the ***AM** envelope.* Note that when there is no modulating signal, the output waveform is simply the carrier signal. However, when a modulating sign al is applied, the amplitude of the output wave varies in accordance with the modulating signal.

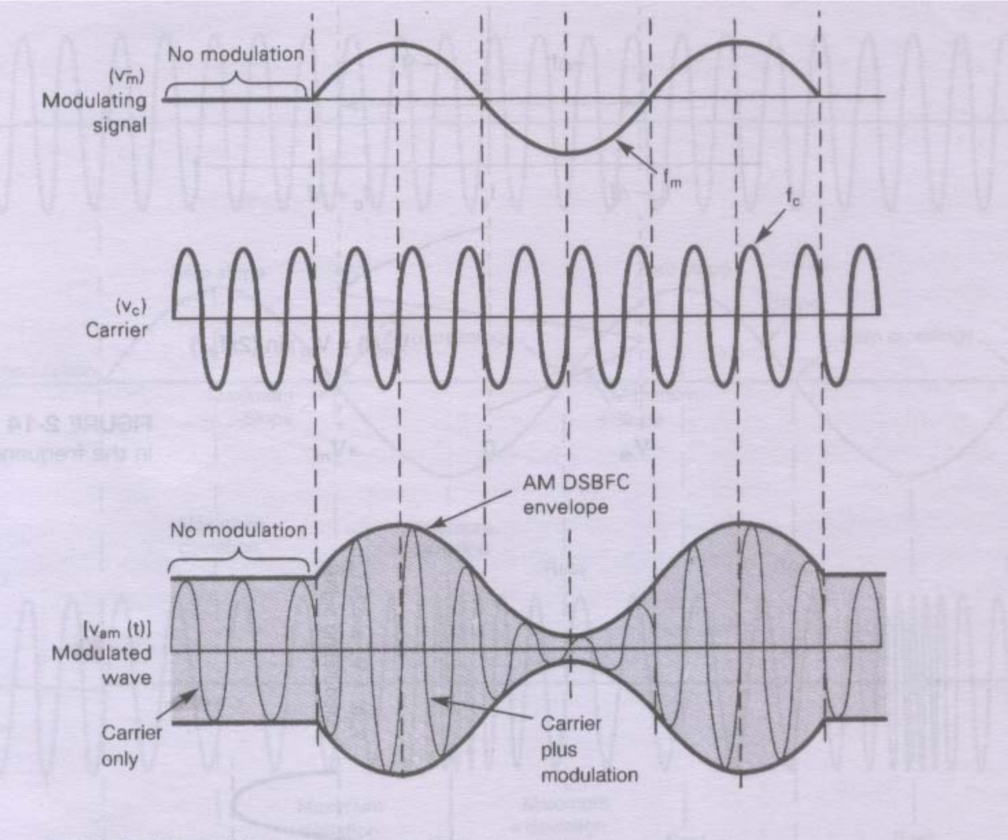


FIGURE 2-13 AM generation

Angle Modulation:- *Angle modulation* results whenever the phase angle (ɵ)of a sinusoidal signal is varied with respect to time. Angle modulation includes both frequency (FM) and phase (PM) modulation. Figure 2-16 illustrates both frequency and phase modulation of a sinusoidal carrier by a single-frequency modulating signal. It can be seen that the FM and PM waveforms are identical except for their time relationship (phase). Thus, it is impossible to distinguish an FM waveform from a PM waveform without knowing the dynamic characteristics of the modulating signal. With FM, the maximum frequency deviation (positive and negative change in the carrier frequency) occurs during the maximum positive and negative peaks of the modulating signal (i.e., the frequency deviation is proportional to the amplitude of the modulating signal). With PM, the maximum frequency deviation occurs during the zero crossings in the modulating signal (i.e., the frequency deviation is proportional to the slope of the modulating signal).

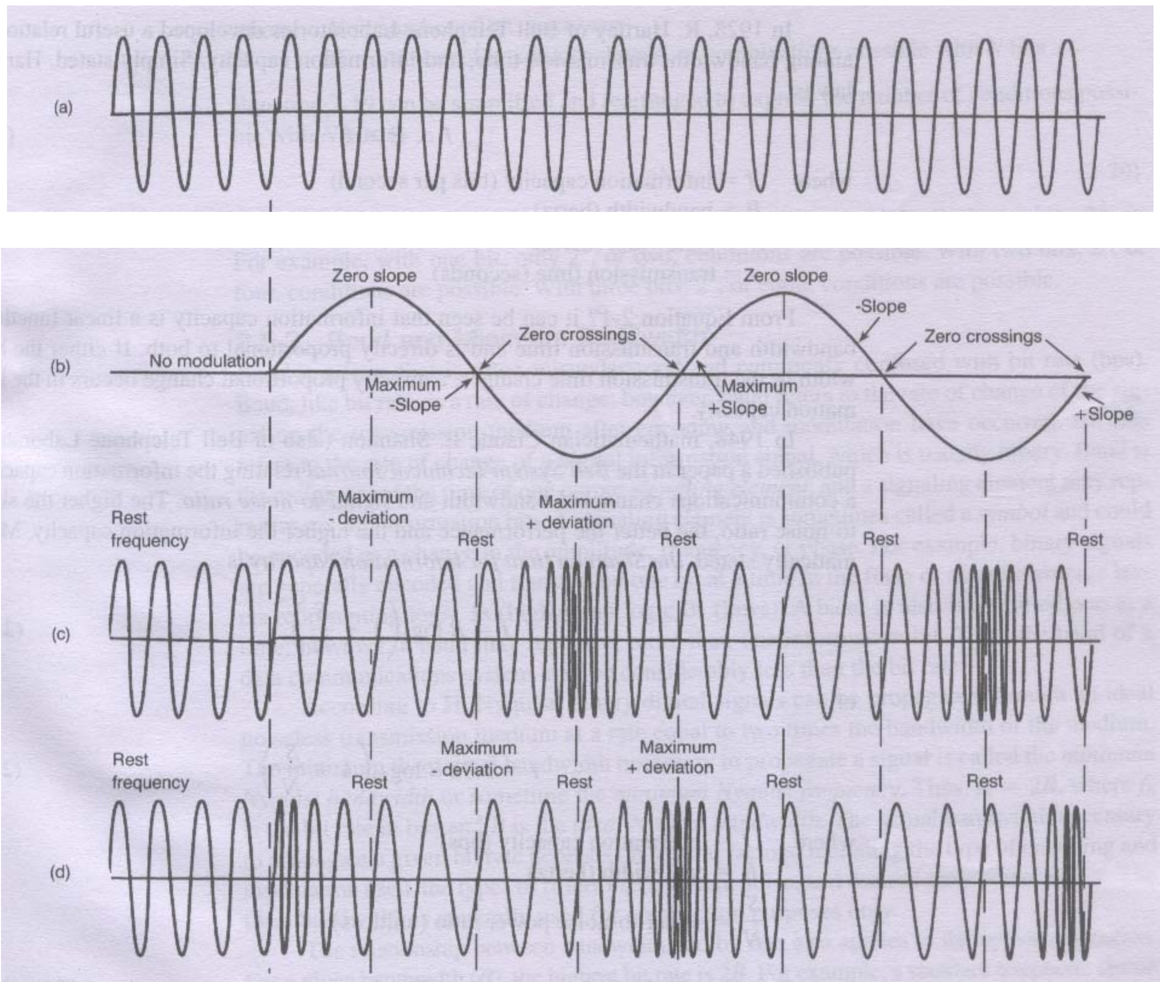


FIGURE 2-16 Phase and frequency modulation of a sine-wave carrier by a sine-wave signal: (a) unmodulated carrier; (b) modulating signal; (c) frequency-modulated wave; (d) phase-modulated wave

2-5 **Describe the following terms: *information capacity, bit, bit rate, baud* , probability of error , bit error rate and *Bandwidth efficiency.***

ANSWER:

Information Capacity:- Information capacity is a measure of how much information can be propagated through a communications system and a function of bandwidth and transmission time.

Bit:-Information capacity represents the number of independent symbols that can be carried through a system in a given unit of time. The most basic digital symbol used to represent information is the *binary digit,* or *bit.*

Bit rate :-It is often convenient to express the information capacity of a system as a *bit **rate.*** Bit rate is the number of bits transmitted during 1 second and is expressed in *bits per second* (bps).

Baud:- Baud refers to the rate of change of the signal on the transmission medium after encoding and modulation have occurred. Bit rate refers to the rate of change of a digital information signal which is usually binary.

Probability of Error and Bit Error Rate:-*Probability of error (P[e])* and *bit error rate* (BER) are often used interchangeably, although in practice they do have slightly different meanings. *P(e)* is a theoretical (mathematical) expectation of the bit error rate for a given system. BER is an empirical (historic al) record of a system's actual bit error performance. For example, if a system has a *P(e)* of $l0^{-5}$, this means that mathematically you can expect one bit error in every 100,000 bits transmitted ($1/l0^{-5}$ = 1/100,000). If a system has a BER of 10^{-5}, this means that in past performance there was one bit error for every 100,000 bits transmitted. A bit error rate is measured and then compared with the expected probability of error to evaluate a system's performance.

Bandwidth efficiency:- *Bandwidth efficiency* (sometimes called *information density* or *spectral efficiency)* is often used to compare the performance of one digital modulation technique to another. In essence, bandwidth efficiency is the ratio of the transmission bit rate to the minimum bandwidth required for a particular modulation scheme. Bandwidth efficiency is generally normalized to a 1-Hz bandwidth and, thus, indicates the number of bits that can be propagated through a transmission medium for each hertz of bandwidth. Mathematically, bandwidth efficiency is

$B\eta$ = transmission bit rate(bps)/minimum bandwidth(Hz) = bits/second/hertz = bits / second / cycles / second = bits/cycle

Where $B\eta$ = bandwidth efficiency

Bandwidth efficiency can also be given as a percentage by simply multiplying $B\eta$ by 100.

2-6 **Briefly describe the significance of the Shannon limit for information capacity.**
ANSWER: In 1928, R. Hartley of Bell Telephone Laboratories developed a useful relationship among bandwidth, transmission time, and information capacity. Simply stated, Hartley's law is

$$I \propto B \times t$$

Where I = information capacity (bits per second)
B = bandwidth (hertz)
t = transmission time (seconds)

From Equation 2-17 it can be seen that information capacity is a linear function of bandwidth and transmission time and is directly proportional to both. If either the bandwidth or the transmission time changes, a directly proportional change occurs in the information capacity.
In 1948, mathematician Claude E. Shannon (also of Bell Telephone Laboratories) published a paper in the *Bell System Technical Journal* relating the information capacity of a communications channel to bandwidth and *signal-to-noise ratio.* Mathematically stated, *the Shannon limit for information capacity* is

$$I = B\log_2(1 + S/N)$$

or

$$I = 3.32\log_{10}(1 + S/N)$$

where I= information capacity (bps)
B= bandwidth (hertz)
S/N= signal-to-noise power ratio (unit less)

For a standard telephone circuit with a signal-to-noise power ratio of 1000 (30 dB) and a bandwidth of 2.7 kHz, the Shannon limit for information capacity is

I = 26.9 kbps

Shannon's formula is often misunderstood. The results of the preceding example indicate that 26.9 kbps can be propagated through a 2.7-kHz communications channel. This may be true, but it cannot be done with a binary system. To achieve an information transmission rate of 26.9 kbps through a 2.7 KHz channel, each symbol transmitted must contain more than one bit.

2-7 **Give a brief description of amplitude-shift keying, frequency-shift keying.**

ANSWER:

Amplitude-Shift Keying:- The simplest digital modulation technique is *amplitude-shift keying* (ASK), where a binary information signal directly modulates the amplitude of an analog carrier. ASK is similar to standard amplitude modulation except there are only two output amplitudes possible. Amplitude-shift keying is sometimes called *digital amplitude modulation* (DAM). Mathematically, amplitude shift keying is

$$v_{ask}(t) = [1 + v_m(t)][\tfrac{A}{2}\cos(\omega_c t)] \qquad (2\text{-}23)$$

where $v_{ask}(t)$ = amplitude-shift keying wave
$v_m(t)$ =digital information (modulating) signal (volts)
$A/2$ = unmodulated carrier amplitude (volts)
ω_c= analog carrier radian frequency (radians per second, $2\pi f_c t$)

In Equation 2-23, the modulating signal, $v_m(t)$, is a normalized binary waveform, where +1V= logic 1 and- 1V = logic 0. Therefore, for a logic 1 input $v_m(t)$ = +1V, Equation 2-23 reduces to

$$v_{ask}(t) = [1+1)]\left[\frac{A}{2}\cos(\omega_c t)\right] = Acos(\omega_c t)$$

And for a logic 0 input, Equation 2-23 reduces to

$$v_{ask}(t) = [1-1]\left[\frac{A}{2}\cos(\omega_c t)\right] = 0$$

Thus, the modulated wave, $v_{ask}(t)$, is either $Acos(\omega_c t)$ or 0. Hence, the carrier is either "on" or *"off,"* which is why amplitude-shift keying is sometimes referred to as *on-off keying (OOK).*

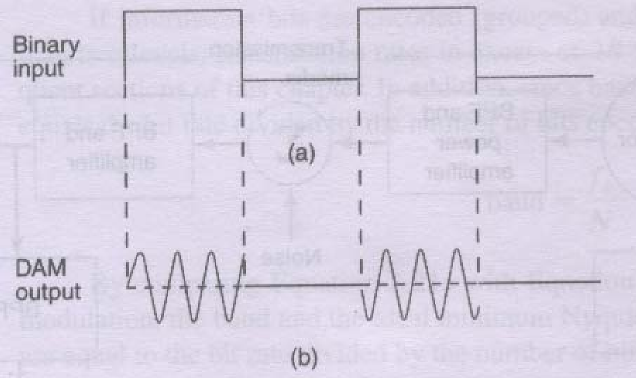


FIGURE 2-18 Digital amplitude modulation: (a) input binary; (b) output DAM waveform

Frequency Shift Keying:- FSK is a form of constant-amplitude angle modulation similar to standard frequency modulation (FM) except the modulating signal is a binary signal that varies between two discrete voltage levels rather than a continuously changing analog waveform. Consequently, FSK is sometimes called *binary FSK* (BFSK). The general expression for FSK is

$$v_{fsk}(t) = V_c\cos\{2\pi[f_c + v_m(t)\Delta f]t\} \quad (2\text{-}24)$$

where $v_{fsk}(t)$ = binary FSK wave form
V_c= peak analog carrier amplitude (volts)
f_c = analog carrier center frequency (hertz)
Δf = peak change (shift) in the analog carrier frequency (hertz)
$v_m(t)$= binary input (modulating) signal (volts)

From Equation 2-24, it can be seen that the peak shift in the carrier frequency (Δf) is proportional to the amplitude of the binary input signal $v_m(t)$ and that the direction of the shift is determined by the polarity.

The modulating signal is a normalized binary waveform where a logic 1 = +1 V and a logic 0 = +1V Thus, for a logic 1 input, $v_m(t) = +1$ and Equation 2-24 can be rewritten as

$$v_{fsk}(t) = V_c\cos\left(2\pi[f_c + \Delta f]t\right)$$

For a logic 0 input, $v_m(t) = -1$, and Equation 2-24 becomes

$$v_{fsk}(t) = V_c\cos\left(2\pi[f_c - \Delta f]t\right)$$

With binary FSK. the carrier center frequency (f_c) is shifted (deviated) up and down in the frequency domain by the binary input signal as shown in Figure 2-19. As the binary input signal changes from a logic 0 to a logic 1 and vice versa, the output frequency shifts between two frequencies: a mark, or logic 1, frequency *(f_m)* and a space, or logic 0, frequency *(f)*. The mark and space frequencies are separated from the carrier frequency by the peak frequency deviation (Δf) and from each other by $2\Delta f$.

With FSK, frequency deviation is defined as the difference between either the mark or the space frequency and the center frequency or half the difference between the mark and space frequencies.

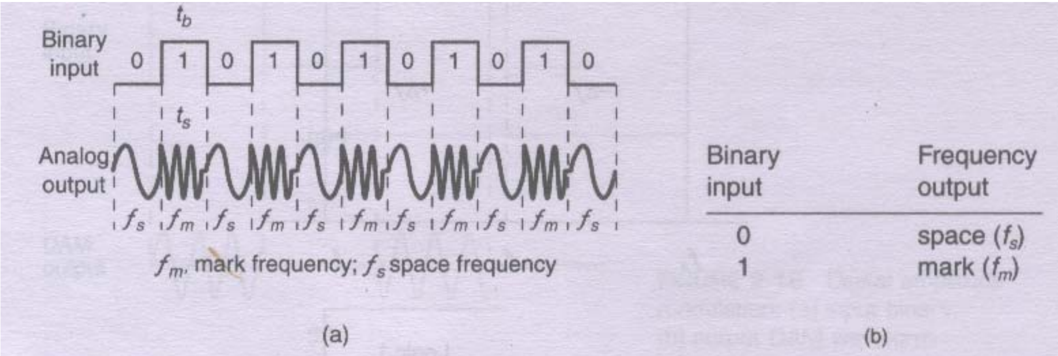


FIGURE 2-20a FSK in the time domain: (a) waveform; (b) truth table

Figure 2-20a shows in the time domain the binary input to an FSK modulator and the corresponding FSK output. As the figure shows, when the binary input *(f_b,)* changes from a logic 1 to a logic 0 and vice versa, the FSK output frequency shifts from a mark *(f_m)* to a space *(f)* frequency and vice versa.

2-8 Describe the relationship between bit rate, bandwidth, and baud for FSK.

ANSWER: In Figure 2-20a that the time of one bit *(t_b)* is the same as the time the FSK output is a mark of space frequency (t_s). Thus, the bit time equals the time of an FSK signaling element, and the bit rate equals the baud. Figure 2-20b shows the truth table for a binary FSK modulator. The truth table shows the input and output possibilities for a given digital modulation scheme.

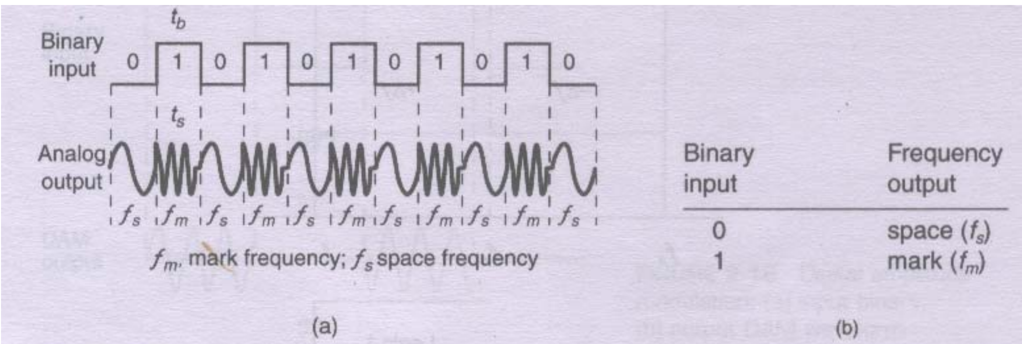


FIGURE 2-20a FSK in the time domain: (a) waveform; (b) truth table

FSK baud and bandwidth:- Baud is defined as ratio of f_b to N i.e., Baud=f_b/N where f_b is channel capacity (bps) and N is the number of bits encoded into each signaling element. The baud for binary FSK is determined by making ***N*** = 1 in the equation baud = $f_b/1=f_b$

FSK is the exception to the rule for digital modulation, as the minimum bandwidth is not determined from Equation $B=f_b/N$. where B is minimum Nyquist bandwidth(hertz). The minimum bandwidth for FSK is determined from the following formula:

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

where B = minimum Nyquist bandwidth (hertz)
Δf = frequency deviation ($|f_m - f_s|$) (hertz)
f_b = input bit rate (bps)

2-9 Give a brief description of BPSK and QPSK.

ANSWER:

Binary phase-shift keying:- The simplest form of PSK is *binary phase- shift keying* (BPSK), where N = 1 and M = 2. Therefore, with BPSK, two phases (2^1= 2) are possible for the carrier. One phase represents a logic 1, and the other phase represents a logic 0. As the input digital signal changes state (i.e., from a 1 to a 0 or from a 0 to a 1), the phase of the output carrier shifts between two angles that are separated by 180 degrees. Hence, other names for BPSK are *phase-reversal keying* (PRK) and *biphase modulation.* BPSK is a form of square-wave modulation of a *continuous wave* (CW) signal. Figure 2-21 shows the output phase-versus-time relationship for a BPSK waveform. As the figure shows, a logic 1 input produces an analog output signal with a 0-degree phase angle, and a logic 0 input produces an analog output signal with a 180-degree phase angle. As the binary input shifts between a logic 1 and a logic 0 condition, and vice versa the phase of the BPSK waveform shifts between 0 degrees and 180 degrees, respectively. For simplicity, only one cycle of the analog carrier is shown in each signaling element, although there may be anywhere between a fraction of a cycle to several thousand cycles, depending on the relationship between the input bit rate and the analog carrier frequency. It can also be seen that the time of one BPSK signaling element *(t_s)* is equal to the time of one information bit *(t_b),* which indicates that the bit rate equals the baud.

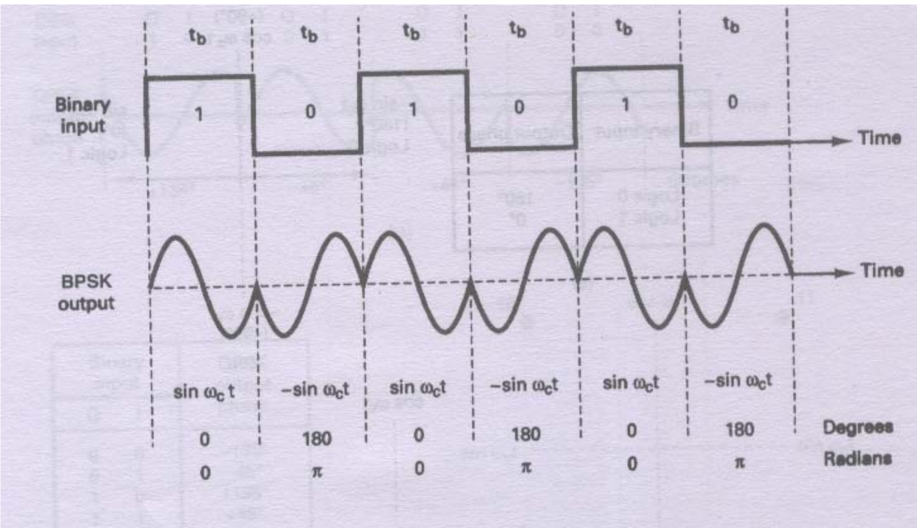


FIGURE 2-21 Output phase–versus–time relationship for a BPSK modulator

Figure 2-22 shows the *truth table, phasor diagram,* and *constellation diagram* for a BPSK signal. A constellation diagram, which is sometimes called a *signal state-space diagram,* is similar to a phasor diagram except that the entire phasor is not drawn. In a constellation diagram, only the relative positions of the peaks of the phasors are shown.

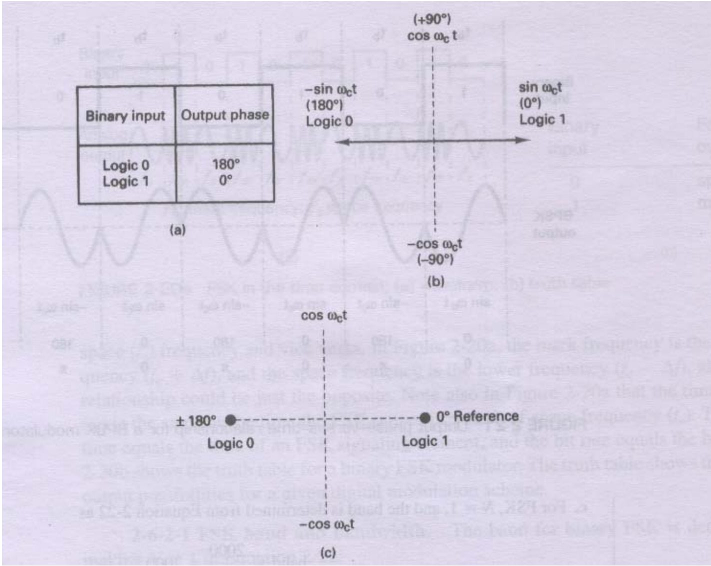


FIGURE 2-22 BPSK modulator: (a) truth table; (b) phasor diagram; (c) constellation diagram

Quaternary phase-shfi keying **(QPSK) or *quadrature PSK*** as it is sometimes called, is an *M-ary* encoding scheme where N = 2 and M= *4* (hence the name "quaternary," meaning *"4")*. With QPSK, four output phases are possible for a single carrier frequency. Because there are four output phases, there must be four different input conditions. The digital input to a QPSK modulator is a binary (base 2) signal, which requires two input bits to produce four input combinations. With two bits, there are four possible conditions: 00,01, 10, and 11. Therefore, with QPSK, the binary input data are combined into groups of two bits, called *dibits*. In the modulator, each dibit code generates one of the four possible output phases (+45, + *135,-45,* and- 135 degrees). Therefore, for each two-bit dibit clocked into the modulator, a single output change occurs, and the rate of change at the output (baud) is equal to one-half the input bit rate (i.e., two input bits produce one output phase change).

Figure 2-23 shows the output phase-versus-time relationship, truth table, and constellation diagram for QPSK.

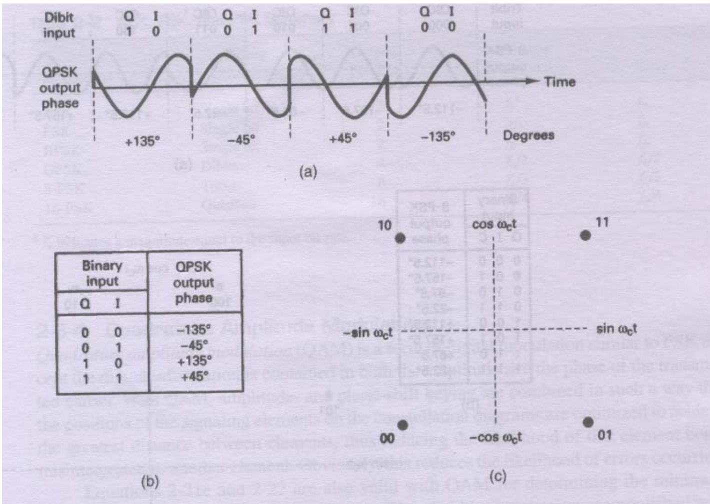
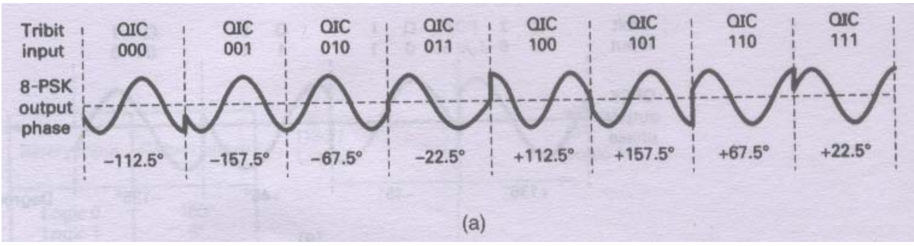


FIGURE 2-23 QPSK: (a) output phase–versus–time relationship; (b) truth table; (c) constellation

2-10 Give a brief description of 8PSK and 16PSK.

ANSWER: **8-PSK**:-With 8-PSK, three bits are encoded, forming tribits and producing eight different output phases. With 8-PSK, *N* = 3, *M* = 8, and the minimum bandwidth and baud equal one-third the bit rate ($f_b/3$). Figure 2-24 shows the output phase-versus-time relationship, truth table, and constellation diagram for 8-PS K. As the figure shows, all eight points on the constellation diagram are the same distance from the intersection of the X and Y axes, and each point is separated from its two adjacent phases by 45 degrees.



Binary input Q I C	8-PSK output phase
0 0 0	−112.5°
0 0 1	−157.5°
0 1 0	−67.5°
0 1 1	−22.5°
1 0 0	+112.5°
1 0 1	+157.5°
1 1 0	+67.5°
1 1 1	+22.5°

(b)

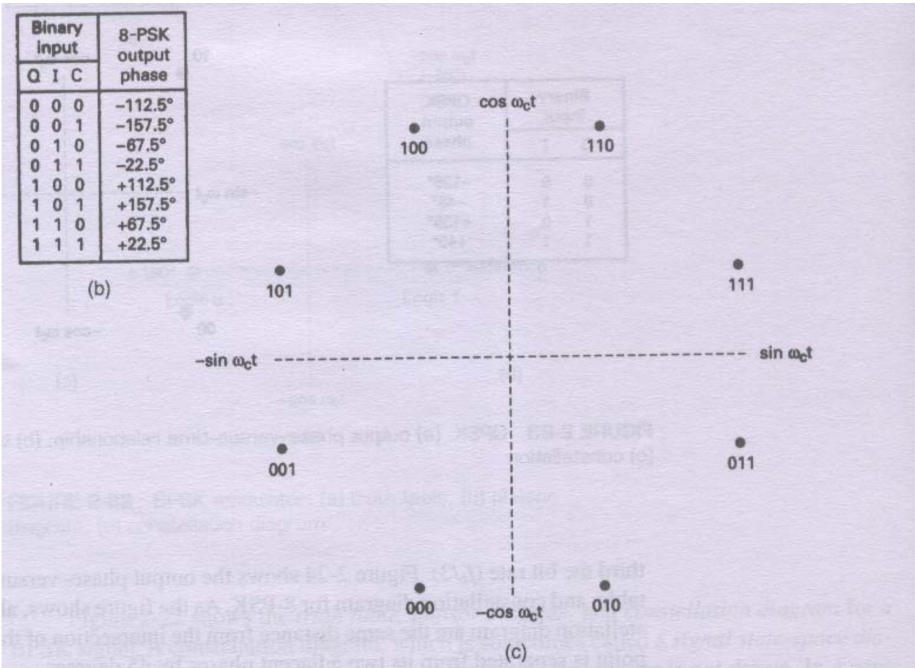


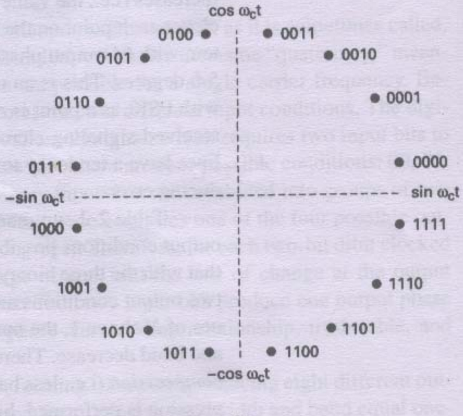
FIGURE 2-24 8-PSK: (a) output phase–versus–time relationship; (b) truth table; (c) constellation diagram

16-PSK:-With 16-PSK, four bits (called *quad bits)* are combined, producing 16 different output phases. With 16-PSK, *N* = 4, *M* 16, and the minimum bandwidth and baud equal one-fourth the bit rate *($f_b/4$).* Figure 2-25 shows the truth table and constellation diagram for 16-PS K.

Table 2-2 summarizes the relationship between number of bits encoded, number of output conditions possible, minimum bandwidth, and baud for ASK, FSK, BPSK, QPSK,8-PSK and 16-PSK.

Bit code	Phase	Bit code	Phase
0000	11.25°	1000	191.25°
0001	33.75°	1001	213.75°
0010	56.25°	1010	236.25°
0011	78.75°	1011	258.75°
0100	101.25°	1100	281.25°
0101	123.75°	1101	303.75°
0110	146.25°	1110	326.25°
0111	168.75°	1111	348.75°

(a)



(b)

FIGURE 2-25 16-PSK: (a) truth table; (b) constellation diagram

Table 2–2 ASK, FSK, and PSK Summary

Modulation	Encoding Scheme	Outputs Possible	Minimum Bandwidth	Baud
ASK	Single bit	2	f_b[a]	f_b
FSK	Single bit	2	$>f_b$	f_b
BPSK	Single bit	2	f_b	f_b
QPSK	Dibits	4	$f_b/2$	$f_b/2$
8-PSK	Tribits	8	$f_b/3$	$f_b/3$
16-PSK	Quadbits	16	$f_b/4$	$f_b/4$

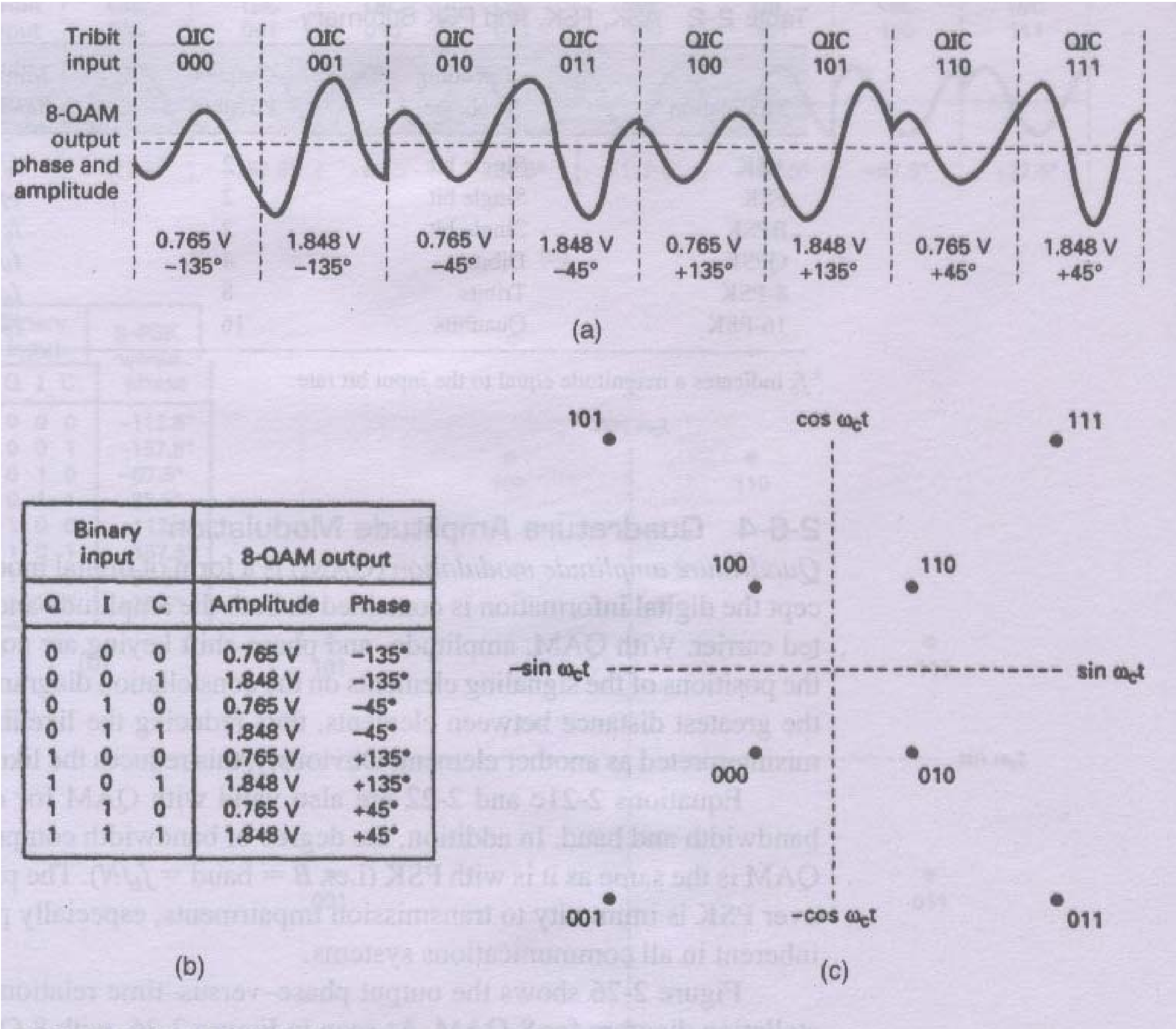
[a] f_b indicates a magnitude equal to the input bit rate.

2-11 Give a brief description of 8-QAM and 16-QAM.

ANSWER:

Quadrature Amplitude Modulation:- *Quadrature amplitude modulation* (QAM) is a form of digital modulation similar to PSK except the digital information is contained in both the amplitude and the phase of the transmitted carrier. With QAM, amplitude- and phase-shift keying are combined in such a way that the positions of the signaling elements on the constellation diagrams are optimized to achieve the greatest distance between elements, thus reducing the likelihood of one element being misinterpreted as another element. Obviously, this reduces the likelihood of errors occurring. Equations Baud=f_b/N and B=f_b/N are also valid with QAM for determining the minimum bandwidth and baud. In addition, the degree of bandwidth compression accomplished with QAM is the same as it is with PSK (i.e., B = baud $=f1,/N)$. The primary advantage of QAM over PSK is immunity to transmission impairments, especially phase impairments that are inherent in all communications systems.

Figure 2-26 shows the output phase-versus-time relationship, truth table, and constellation diagram for 8-QAM. As seen in Figure 2-26, with 8-QAM, there are four phases and two amplitudes that are combined to produce eight different output conditions. With 8-QAM (as with 8-PSK),three bits are encoded, forming tribits and producing eight different output conditions. With 8-QAM, N = 3, M = 8, and the minimum bandwidth and baud equal one-third the bit rate ($f_b/3$).



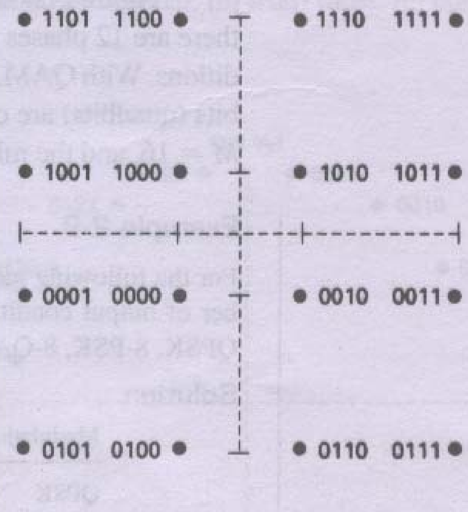
Binary input			8-QAM output	
Q	I	C	Amplitude	Phase
0	0	0	0.765 V	−135°
0	0	1	1.848 V	−135°
0	1	0	0.765 V	−45°
0	1	1	1.848 V	−45°
1	0	0	0.765 V	+135°
1	0	1	1.848 V	+135°
1	1	0	0.765 V	+45°
1	1	1	1.848 V	+45°

FIGURE 2-26 8-QAM: (a) output phase–versus-time relationship; (b) truth table; (c) constellation diagram

16-QAM:- Figure 2-27 shows the truth table and constellation diagram for 16-QAM. With 16-QAM, there are 12 phases and three amplitudes that are combined to produce 16 different output conditions. With QAM, there are always more phases possible than amplitude. With I 6-QAM, four bits (quadbits) are combined, producing 16 different output conditions. With 16-QAM, $N = 4$, $M = 16$, and the minimum bandwidth and baud equal one-fourth the bit rate ($f_b/4$).

Binary input				16-QAM output	
Q	Q'	I	I'		
0	0	0	0	0.311 V	−135°
0	0	0	1	0.850 V	−165°
0	0	1	0	0.311 V	−45°
0	0	1	1	0.850 V	−15°
0	1	0	0	0.850 V	−105°
0	1	0	1	1.161 V	−135°
0	1	1	0	0.850 V	−75°
0	1	1	1	1.161 V	−45°
1	0	0	0	0.311 V	135°
1	0	0	1	0.850 V	165°
1	0	1	0	0.311 V	45°
1	0	1	1	0.850 V	15°
1	1	0	0	0.850 V	105°
1	1	0	1	1.161 V	135°
1	1	1	0	0.850 V	75°
1	1	1	1	1.161 V	45°

(a)

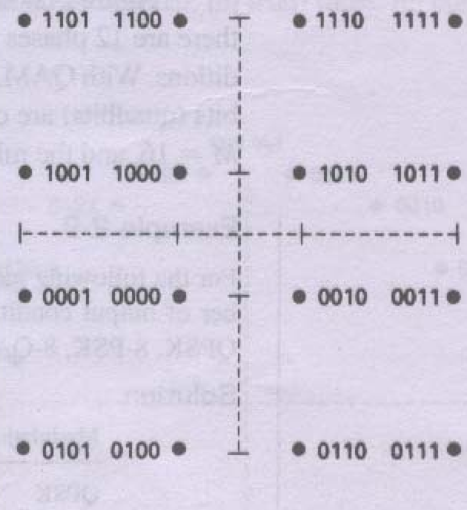


(b)

FIGURE 2-27 16-QAM modulator: (a) truth table; (b) constellation diagram

Binary input				16-QAM output	
Q	Q'	I	I'		
0	0	0	0	0.311 V	−135°
0	0	0	1	0.850 V	−165°
0	0	1	0	0.311 V	−45°
0	0	1	1	0.850 V	−15°
0	1	0	0	0.850 V	−105°
0	1	0	1	1.161 V	−135°
0	1	1	0	0.850 V	−75°
0	1	1	1	1.161 V	−45°
1	0	0	0	0.311 V	135°
1	0	0	1	0.850 V	165°
1	0	1	0	0.311 V	45°
1	0	1	1	0.850 V	15°
1	1	0	0	0.850 V	105°
1	1	0	1	1.161 V	135°
1	1	1	0	0.850 V	75°
1	1	1	1	1.161 V	45°

(a)



(b)

FIGURE 2-27 16-QAM modulator: (a) truth table; (b) constellation diagram