Analog Communication

1.	**What are the basic components of any communication system? Draw and explain the block diagram of the typical communication system.**	

The elements of communication system are as follows:

i) Information source
ii) Transmitter
iii) Communication channel or medium
iv) Noise
v) Receiver

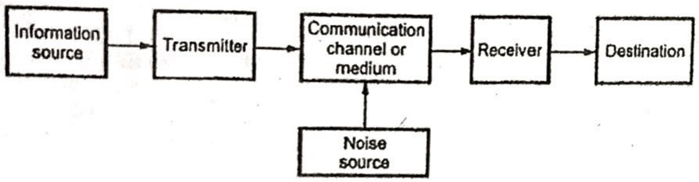

Fig:- Block Diagram of Communication System

Information: The communication system communicates messages. The messages come from the information sources. It may contain human voice, picture, code, data, music and their combination. //

Transmitter: The transmitter is a collection of electronic circuits designed to convert the information into a signal suitable for transmission over a given communication medium or channel. //

Communication channel: The communication channel is the medium by which the electronic signal is transmitted from one place to another. There are basically two types of communication media and accordingly there are two types of communications (a) Wire communication or line communication, and (b) wireless communication or radio communication. Line communication is a wireless communication, requiring no physical wires between transmitter and receiver to carry the signal. Radio communication makes possible communication over very very long distances, even from Earth to Moon!

Noise: The noise is an unwanted signal, produced by the atmosphere and added to the message in the communication channel, degraded the transmitted information. Noise is one of the serious problems of electronic communications. It cannot be completely eliminated. However, there are many ways to deal with noise, and reduce the possibility of degradation of signal due to noise.

Receiver: A receiver is a collection of electronic circuits designed to convert the signal back to the original information.

2.	**What are the typical channels uses in line communication? What are the advantages of optical fiber over co-axial cable?**	

The typical channels used in line communication are given below:

i) Twisted pair cable (i)
ii) Co-axial cable (or copper cable) (ii)
iii) Fiber optic cable (iii) Fiber optic cable.

The advantages of optical fiber over co-axial cable are as follows:

i) Fiber optics are significantly faster than co-axial cable.
ii) Fiber optics are more eco-friendly than co-axial cable.
iii) Fiber optics are the technology of the future.
iv) Fiber optics offer more secure communication than co-axial cable.

3	**(a) Justify-The baseband signal cannot be transmitted through space by radio.** **Or, Justify that Modulation is suitable for distance transmission.** **(b) If a baseband signal of frequency 15 kHz is to be transmitted through space by radio, determine the size or height of antenna for this transmission.** **(c) If we consider a modulated signal with 1 MHz frequency in the broadcast band, what would be the size or height of the antenna?**	**24th 31st & 34th BCS**

The message signal that is to be transmitted is referred to as 'baseband signal'. In a communication system, when the original information signals (baseband signals) may be transmitted over the medium directly, it is referred to as "baseband transmission". One example of baseband transmission is telephony, especially for the local calls. In some computer networks, the digital signals are applied directly to co-axial cables for transmission to another computer.

But there are many instances when the baseband signals are incompatible for direct transmission over the medium. For example, voice signals cannot travel longer distances in air, the signal gets attenuated rapidly. Hence for transmission of baseband signals by radio, modulation technique has to be used.

* The height of the antenna required for the transmission and reception of radio waves in radio transmission is a function of wavelength of the frequency used. The minimum height of the antenna is give as $\frac{\lambda}{4}$. The wavelength $\lambda = \frac{c}{f}$; where c is the velocity of light and f is the frequency.

$\frac{\lambda}{4}$; $\lambda = \frac{c}{f}$

For the above equation, it can be easily noticed that at low frequencies wavelength is very high and hence the antenna height is also very high.

For example, consider the baseband signal with f = 1 kHz. Then f=1kHz

Height of antenna $= \frac{\lambda}{4} = \frac{c}{4\times f} = \frac{3\times10^8}{4\times15\times10^3} = 5000\ meters$

This 5000 meters height of a vertical antenna is unthinkable and unpracticable.

On the other hand, if we consider a modulated signal with 1 MHz frequency in the broadcast band, the height of antenna $= \frac{3\times10^8}{4\times1\times10^6} = 75$ meters.

$\frac{\lambda}{4}$ This height of antenna is practical and such antenna can be installed.

4	**A radio station transmits signa in 1 MHz frequency band. What is the required antenna length for efficient radion of electromagnetic energy.**	**PGCB 2016**

Wavelength, $\lambda = \frac{c}{f} = \frac{3\times10^8}{1\times10^6} = 300\ m$ $c = f\lambda$; $\lambda = \frac{c}{f}$

Length of the transmitting antenna $= \frac{\lambda}{4} = \frac{300}{4} = 75\ m$

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Length of the transmitting antenna for efficient transmission $= \frac{\lambda}{10} = \frac{300}{10} = 30m$ $\frac{\lambda}{4}$

5	**What is the need of modulation?**	

The advantages of using modulation technique are given below:

i) It reduces the height of antenna.
ii) It avoids mixing of signals.
iii) It increases the range of communication.
iv) It allows multiplexing of signals.
v) It allows adjustments in the bandwidth.
vi) It improves the quality of reception.

6	**What are the different types of modulation?**	

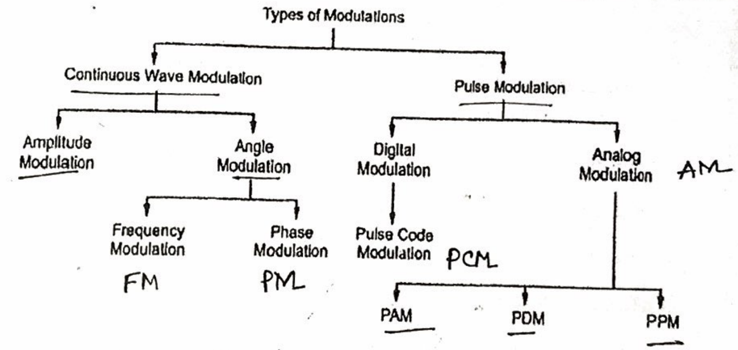


Fig: Different types of Modulations

7	**What do you mean by bandwidth and how it is measured? Give the bandwidth requirement for any four transmitting signals. Or, What is the bandwidth requirement of Telegraph signals? Or, what is the bandwidth requirement of voice / speech signal? What is the bandwidth requirement of music signal? Or what is the bandwidth requirement of television signal? Or, What is the bandwidth requirement of digital data transmission?**	**32nd BCS**

Answer:

Bandwidth is the portion of frequency range occupied by a signal. More specifically, it is the difference between the upper limit and lower limit of the signal.

Telegraph signal: The shortest time element or time duration of telegraph signal is 20 milliseconds. Therefore the bandwidth required is $B_w = f_{highest} = \frac{1}{20\times10^{-3}} = 50\ Hz.$ 50 Hz

Voice or speech signal: Voice signal ranges from 300 Hz to 3000 Hz. So the bandwidth required is $B_W = upper\ limit - lower\ limit = 3000 - 300 = 2700\ Hz.$ 2700 Hz

Music signals: For high quality music transmission, the frequency ranges from 30 Hz to 15 kHz. So, the bandwidth, $B_w = 15000 - 30 = 14970\ Hz$ 14970 kHz

Television signal: For television signals, bandwidth required is 6 MHz. 6MHz

Digital data transmission: Since many digital data transmission utilize telephone channels, the bandwidth of the telephone is an appropriate consideration. The internationally accepted standard telephone channel occupies the frequency range of 300 to 3400 Hz. So bandwidth is 3100 Hz. 300 – 3400 Hz 3100 Hz

8	**Describe Amplitude Modulation. What is Modulation Index? Draw the frequency spectrum of the Modulated Signal.**	[25th BCS]

Amplitude varied about a mean value linearly with the baseband/message signal.

Amplitude Modulation (AM) is defined as a process in which the amplitude of the carrier wave is varied about a mean value, linearly with the baseband or message signal.

Consider a sinusoidal carrier wave c(t) defined by

$c(t) = A_c\cos(2\pi f_c t)$

where A_c is the carrier amplitude and f_c is the carrier frequency.

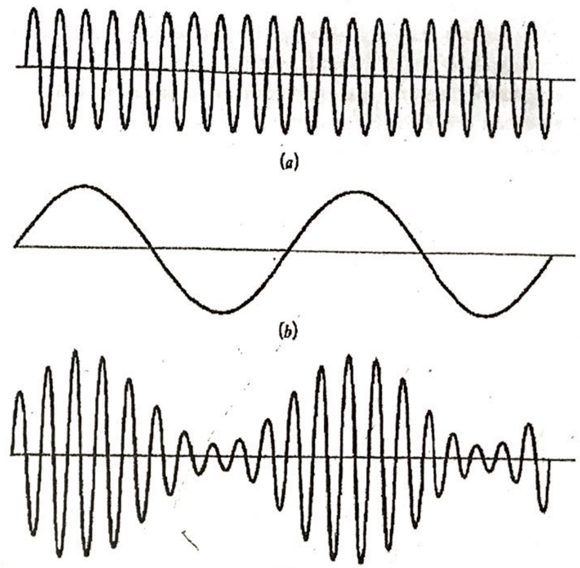
Let, m(t) denote the baseband signal which carries the specification of the message.

An amplitude modulated (AM) wave may thus be described as a function of time in the form:

$s(t) = A_c[1+k_a m(t)]\cos(2\pi f_c t)$

where k_a is a constant called the amplitude sensitivity of the modulator.

$|k_a m(t)|$ is also called modulation index.



Amplitude Modulation

Modulation index:

The modulation index (or modulation depth) of a modulation scheme describes how much the modulated variable of the carrier signal varies around its un-modulated level. It is defined differently in each modulation scheme.

un-modulated carrier

The amplitude modulation, AM, modulation index can be defined as the measure of extent of amplitude variation about an un-modulated carrier. It can be expressed as:

Modulation index, $m = \frac{A_m}{A_c}$ $m = \frac{A_m}{A_c}$; Also denoted by μ.

Where A_m is the amplitude of m(t) and A_c is the amplitude of carrier. Modulation index is also denoted by μ.

There are two requirements of amplitude modulation. They are:

i) The amplitude of $k_a m(t)$ is always less than unity, that is,

$$|k_a m(t)| < 1, for\ all\ t$$

It ensures that the function $1+k_a m(t)$ is always positive and so we may express the envelope of the AM wave s(t) as $A_c[1+k_a m(t)]$. When $|k_a m(t)| > 1\ for\ any\ t$, the carrier wave becomes over modulated, resulting in carrier phase reversals when the factor $1+k_a m(t)$ crosses zero. The modulated wave then exhibits envelope distortion. The absolute maximum value of $k_a m(t)$ multiplied by 100 is referred to as the percentage modulation.

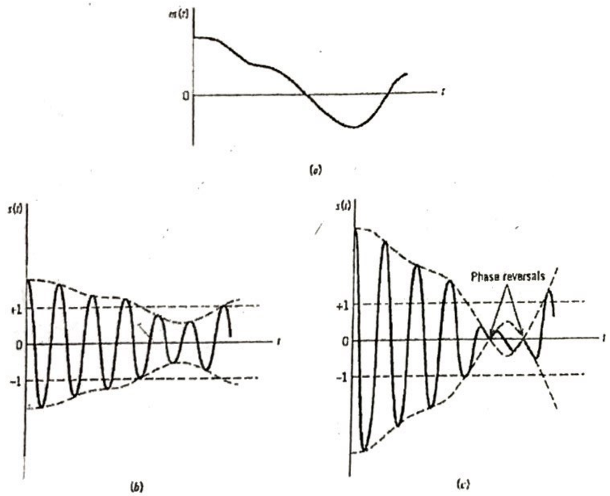


FIGURE 2.3 Illustrating the amplitude modulation process. (a) Baseband signal $m(t)$. (b) AM wave for $|k_a m(t)| < 1$ for all t. (c) AM wave for $|k_a m(t)| > 1$ for some t.

ii) The carrier frequency f_c is much greater than the highest frequency component W of the message signal m(t), that is

$$f_c \gg W$$ $f_e >> W$

We call W the message bandwidth.

Now, the modulated wave, $s(t) = A_c[1+k_a m(t)]\cos(2\pi f_c t)$

We find that the Fourier transform of the AM wave s(t) is given by

$$S(f) = \frac{A^c}{2}[\delta(f-f_c) + \delta(f+f_c) + \frac{k_a A_c}{2}[M(f-f_c) + M(f+f_c)$$

Now the frequency spectrum:

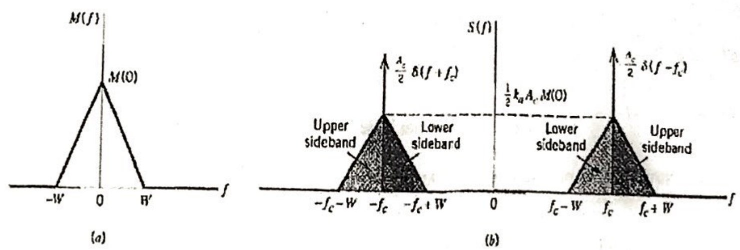


FIGURE (a) Spectrum of baseband signal. (b) Spectrum of AM wave.

9	**Derive the modulated signal for a single tone message signal, draw the frequency spectrum and also derive the power equations.**	

If the message signal contains single frequency component, then the resulting modulated signal is known as a single tone AM.

Assume,

$$m(t) = A_m \cos(2\pi f_m t)$$

And, $c(t) = A_c \cos(2\pi f_c t)$

We know that,

$$S_{AM}(t) = A_c[1 + k_a m(t)]\cos(2\pi f_c t)$$

$$= A_c[1 + k_a A_m \cos(2\pi f_m t)]\cos(2\pi f_c t)$$

Where, $k_a A_m = \mu \rightarrow$ modulation index of AM

When, $\mu<1$: under modulation

When, $\mu=1$: critical Modulation

When, $\mu>1$: over modulation

Now,

$$S_{AM}(t) = A_c\cos(2\pi f_c t) + k_a A_c A_m \cos(2\pi f_c t)\cos(2\pi f_m t)$$

$$= A_c\cos(2\pi f_c t) + A_c\mu\cos(2\pi f_c t)\cos(2\pi f_m t)$$

$$= A_c\cos(2\pi f_c t) + \frac{A_c\mu}{2} \times 2\cos(2\pi f_c t)\cos(2\pi f_m t)$$

$$= A_c\cos(2\pi f_c t) + \frac{A_c\mu}{2}[\cos 2\pi(f_c + f_m)t + \cos 2\pi(f_c - f_m)t]$$

$$= A_c\cos(2\pi f_c t) + \frac{A_c\mu}{2}\cos 2\pi(f_c+f_m)t + \frac{A_c\mu}{2}\cos 2\pi(f_c-f_m)t]$$

⇓ ⇓ ⇓

Carrier Upper Side Band Lower Side Band

Now, the Fourier transform of $S_{AM}(t)$ is

$$S_{AM}(t) = \frac{1}{2}A_c[\delta(f-f_c)+\delta(f+f_c)] + \frac{1}{4}\mu A_c[\delta(f-f_c-f_m)+\delta(f+f_c+f_m)]$$

$$+ \quad \frac{1}{4}\mu A_c[\delta(f-f_c+f_m)+\delta(f+f_c-f_m)]$$

Thus, the spectrum of an AM wave, for the special case of sinusoidal modulation, consists of delta function at $\pm f_c,\ f_c \pm f_m,\ and -f_c \pm f_m$

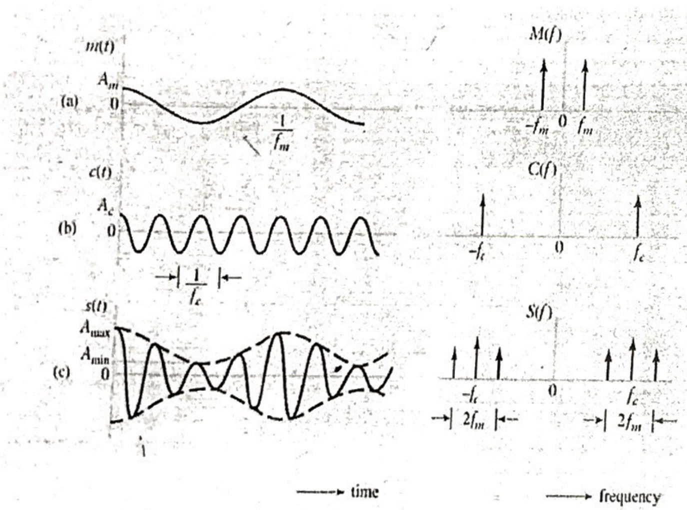


FIGURE Illustration of the time-domain (on the left) and frequency-domain (on the right) characteristics of amplitude modulation produced by a single tone. (a) Modulating wave. (b) Carrier wave. (c) AM wave.

Power of AM:

Let, P_t=carrier power after modulation — P_t = [total] carrier power after modulatio

P_c=carrier power before modulation — P_c = " " before modulatior

P_{USB} =Power of Upper side band .

P_{LSB} = Power of Lower side band .

Now, $P_t = P_c + P_{UBB} + P_{LSB}$ — $P_t = P_c + P_{USB} + P_{LSB}$.

আমরাজানি, Power ,$P = \frac{V_{rms}^2}{R} = \frac{\left(\frac{V_m}{\sqrt{2}}\right)^2}{R} = \frac{V_m^2}{2R} = \frac{(Amplitude\ )^2}{2R}$; *Where R is the resistance*

$P = \frac{(Amplitude)^2}{2R}$

So, $\boxed{P_c = \frac{A_c^2}{2R};}$ — carrier power

$$P_{USB} = \frac{\left(\frac{A_c\mu}{2}\right)^2}{2R} = \frac{A_c^2\mu^2}{8R};$$

$P_c = \frac{A_c^2}{2R}$;

$$P_{LSB} = P_{USB} = \frac{A_c^2\mu^2}{8R}$$

$= \frac{A_c^2\mu^2}{8R}$

$$P_{SB} = P_{LSB} + P_{USB} = \frac{A_c^2\mu^2}{8R} + \frac{A_c^2\mu^2}{8R} = \frac{A_c^2\mu^2}{4R};$$

$$Again, P_{SB} = \frac{A_c^2\mu^2}{4R} = \frac{A_c^2}{2R} \times \frac{\mu^2}{2} = P_c \times \frac{\mu^2}{2};$$

$$So, P_t = \frac{A_c^2}{2R} + \frac{A_c^2\mu^2}{4R}$$

$$or, P_t = \frac{A_c^2}{2R}\left(1+\frac{\mu^2}{2}\right)$$

$$\boxed{or, P_t = P_c\left(1+\frac{\mu^2}{2}\right)}$$

$P_t = P_c\left(1+\frac{\mu^2}{2}\right)$

For, μ = 0 (No modulation), $P_t = P_c$; — μ = 0; $P_t = P_c$

For, μ = 1 (100% modulation), P_t=1.5 P_c; — μ = 1; P_t = 1.5 P_c

Hence, μ increases from 0 to 1, total AM power is increased by 50 %.

μ 0 → 1, 50%.

Modulation Efficiency: It specifies share of sideband power in total power.

So, $\eta = \frac{side\ band\ power}{total\ power}$

$$\boxed{\eta = \frac{\text{side band power}}{\text{total power}}}$$

$$= \frac{P_{SB}}{P_t}$$

$$= \frac{\frac{P_c\mu^2}{2}}{P_c\left[1+\frac{\mu^2}{2}\right]}$$

$$So, \eta = \frac{\mu^2}{2+\mu^2};$$

$P_c = 100\%$ of P_t

Now, if $\mu = 0$ i.e. no modulation, then, $\eta = 0$, i.e. P_{SB}= 0% of P_t and P_c=100% of P_t.

If, $\mu = 0.5$, then, $\eta = 0.11$, i.e. P_{SB} = 11% of P_t and P_c=89% of P_t.

If $\mu = 1$, then $\eta = 0.33$, i.e. P_{SB} = 33.33% of P_t and P_c=66.6% of P_t.

উপরেরআলোচনা থেকে এটা স্পষ্ট যে, Amplitude Modulation (AM) এর ক্ষেত্রে অনেক বেশী পাওয়ার Carrier এর জন্য নষ্ট হয়ে যায়।

Voltage and current of the transmitter of AM:

We know, transmitter power, $P_t = P_c(1+\frac{\mu^2}{2})$ $\quad P_t = P_c\left(1+\frac{\mu^2}{2}\right)$

Again, we, power, $P = \frac{(voltage\ in\ rms)^2}{R} = \frac{V^2}{R} = \frac{(Voltage\ amplitude)^2}{2R}$

So, putting this value in power equation;

$$P = \frac{(V_{rms})^2}{R} = \frac{V^2}{R} = \frac{(\text{Voltage amplitude})^2}{2R}$$

$$\frac{V_t^2}{R} = \frac{V_c^2}{R}\left(1+\frac{\mu^2}{2}\right)$$

$$So, \boxed{V_t = V_c\sqrt{1+\frac{\mu^2}{2}}} \quad V_t = V_c\sqrt{1+\frac{\mu^2}{2}}$$

Similarly for current putting, $P = I^2R$, we get,

$$\boxed{I_t = I_c\sqrt{1+\frac{\mu^2}{2}}} \quad I_t = I_c\sqrt{1+\frac{\mu^2}{2}}$$

10	**Draw the block diagram of square law modulator and switching modulator.**	

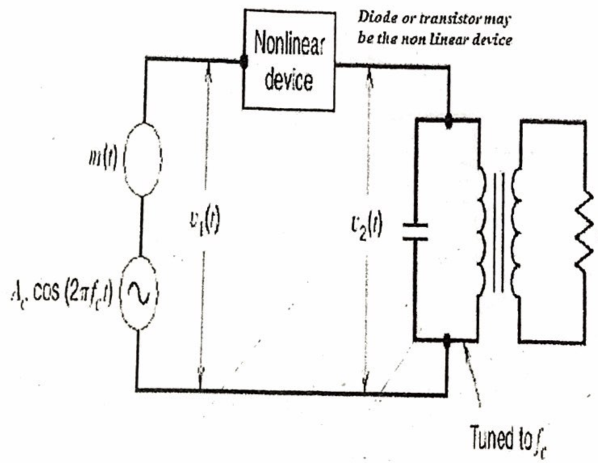


Figure : Square law modulator

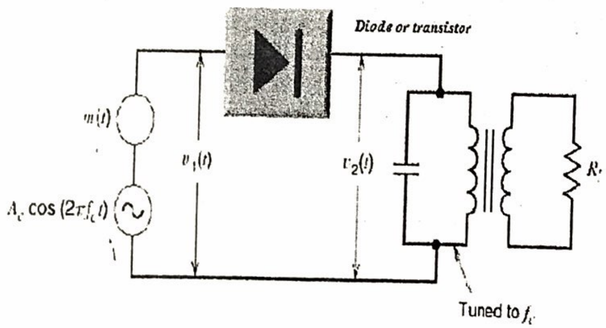
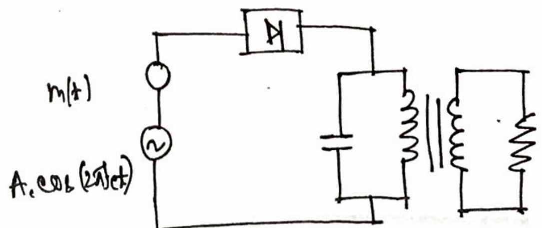


Figure Switching modulator



11	**Draw the complete envelope detector circuit for detecting message signal.**	

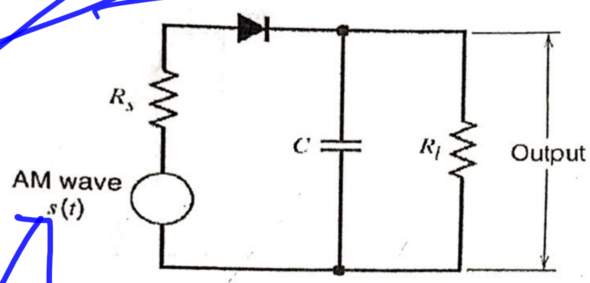


Figure: Envelope detector

AM only

It is a simple and highly effective system. This method is used in most of the commercial AM radio receivers.

During positive half cycles of the input signals, the diode D is forward biased and the capacitor C charges up rapidly to the peak of the input signal. When the input signal falls below this value, the diode becomes reverse biased and the capacitor discharges slowly through the load resistor R_l. The discharging process continues until the next positive half cycle. When the input signal becomes greater than the voltage across the capacitor, the diode conducts again and the process repeated.

We assume that the diode is ideal and also assume that the Am wave applied to the envelop detector is supplied by a voltage source of internal impedance R_s. The charging time constant must be short compared with the carrier period $\frac{1}{f_c}$, that is, $R_s \ll \frac{1}{f_c}$; so that the capacitor charges rapidly and thereby follows the applied voltage up to the positive peak when the diode is conducting.

On the other hand, discharging time constant R_lC must be long enough to ensure that the capacitor discharges slowly through the load resistor R_l between positive peaks of the carrier wave, but not so long that the capacitor voltage will not discharge at the maximum rate of change of the modulating wave, that is, $1/f_c \ll R_l \ll \frac{1}{W}$, where W is the message bandwidth. The result is that the capacitor voltage or detector output is very nearly the same as the envelope of the AM wave.

✓12	**What are the advantages and disadvantages of AM?**	

Advantages of AM:

- Generation and demodulation of AM wave are easy.
- AM systems are cost effective and easy to build

Disadvantages of AM:

- AM contains unwanted carrier component, hence it requires more transmission power.
- The transmission bandwidth is high. The transmission bandwidth is equal to twice the message bandwidth.

13	**How can the limitations of AM be overcome? Or, Write short note on: DSBSC, SSB and VSB.**	

Answer:

As AM is required high power and high bandwidth, to overcome these limitations, the conventional AM system is modified at the cost of increased system complexity. Therefore, three types of modified AM systems are discussed. increased system complexity

DSBSC (Double Side Band Suppressed Carrier) Modulation:

In DSBSC modulation, the modulated wave consists of only the upper and lower side bands. Transmitted power is saved through the suppression of the carrier wave, but the channel bandwidth requirement is the same as before.

If m(t) is the message signal and $A_c\cos(2\pi f_c t)$ is the carrier signal then, DSBSC wave is

$$s(t) = A_c \cos(2\pi f_c t)\, m(t)$$

The Fourier transform of above equation is

$$s(f) = \frac{1}{2}A_c[M(f-f_c) + M(f+f_c)]$$

power – unwanted carrier.
high BW

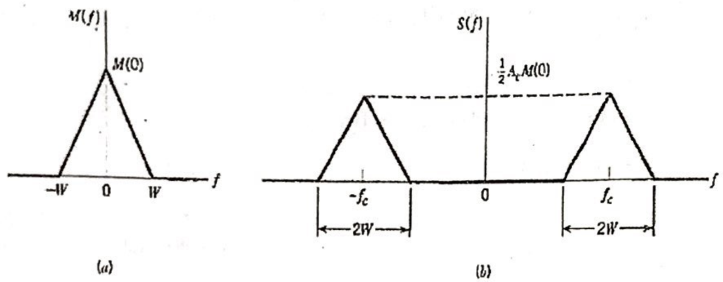


FIGURE (*a*) Spectrum of baseband signal. (*b*) Spectrum of DSB-SC modulated wave.

SSBSC (Single Side Band Suppressed Carrier) Modulation: The SSBSC modulated wave consists of only the upper side band or lower side band. SSBSC is suited for transmission of voice signals. It is an optimum form of modulation in that it requires the minimum transmission power and minimum channel bandwidth. Its disadvantage is that cost and complexity are increased.

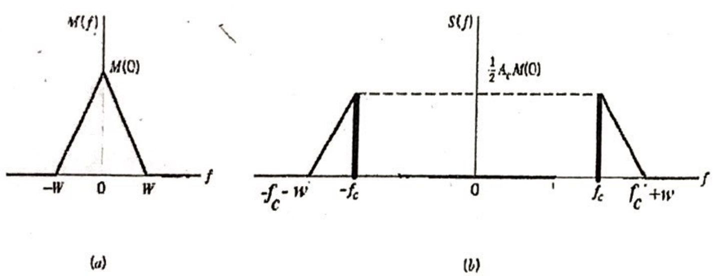


FIGURE (*a*) Spectrum of baseband signal. (*b*) Spectrum of USB modulated wave.

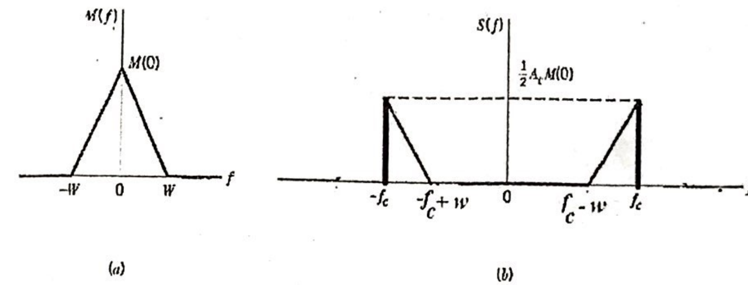
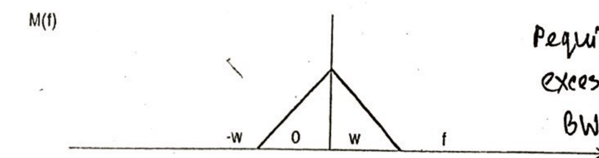


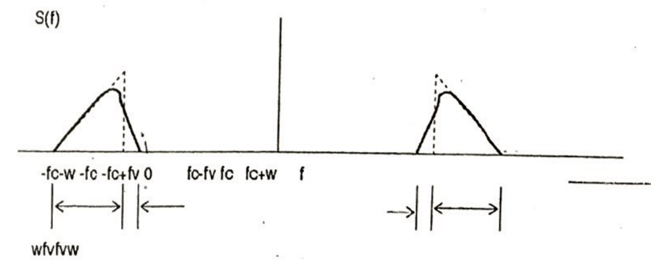
FIGURE (*a*) Spectrum of baseband signal. (*b*) Spectrum of LSB modulated wave.

VSB (Vestigial Side Band) Modulation: In VSB, one side band is completely passed and just a trace of vestige of other side band is retained. The required channel bandwidth is therefore in excess of the message bandwidth by an amount equal to the width of the vestigial side band. This method is suitable for the transmission of wide band signals.



Fig(a) Spectrum of message signal

Required channe BW is excess of the message BW by an amount equal to the vestige.



Fig(b) Spectrum of VSB wave containing vestige of the Lower side band

14	**A carrier wave of frequency 10 MHz and peak value 10 V is amplitude modulated by a 5 kHz sine wave of amplitude 6V. Determine the modulation index, side frequencies and amplitude side frequencies.**	

Solution: Modulation index is denoted by m or μ . So, don't be confused!!

Modulation index, $m = \frac{A_m}{A_c} = \frac{6}{10} = 0.6$.

Side frequencies are,

Upper side band: $f_c + f_m = 10MHz + 5kHz = 10.005\ MHz$

Lower side band: $f_c - f_m = 10MHz - 5kHz = 9.995\ MHz$

The Amplitude of side frequencies is given by $\frac{mA_c}{2}$ or $\frac{\mu A_c}{2} = \frac{0.6 \times 10}{2} = 3\ volts$

15	**The tuned circuit of the oscillator in a simple AM transmitter employs a 40μ H coil and 12 nanofarad (nF) capacitor. If the oscillator output is modulated by audio frequency of 5 kHz, what are the lower and upper side band frequencies and the bandwidth required to transmit this amplitude modulated wave?**	

Solution: Here the carrier frequency is the frequency of the oscillator.

So carrier frequency, $f_c = \frac{1}{2\pi\sqrt{LC}} = 230\ kHz.$

$\therefore f_{USB} = 230 + 5 = 235\ kHz$

And $f_{LSB} = 230 - 5 = 225\ kHz$

Bandwidth, $W = f_{USB} - f_{LSB} = 235 - 225 = 10\ kHz.$

16	**An audio frequency signal $10sin2\pi500t$ is used to amplitude modulate a carrier of $50sin2\pi10^5t$. Calculate:** i) Modulation index ii) Sideband frequencies iii) Amplitude of each sideband frequencies iv) Bandwidth required v) Total power delivered to the load of 600 Ω. vi) Transmission efficiency	

Solution:

i) Modulation index, $\mu = \frac{A_m}{A_c} = \frac{10}{50} = 0.2$

And percent modulation = 0.2 × 100 = 20%

ii) Sideband frequencies:

Given, $\omega_m = 2\pi \times 500; so\ f_m = 500\ Hz$

And, $\omega_c = 2\pi \times 10^5; so, f_c = 100\ kHz.$

So, $f_{USB} = f_c + f_m = 100\ kHz + 500\ Hz = 100.5\ kHz$

And $f_{LSB} = f_c - f_m = 100\ kHz - 500\ Hz = 99.5\ kHz.$

iii) Amplitude of each frequencies:

Amplitude of upper or lower side bands $= \frac{\mu A_c}{2} = \frac{0.2 \times 50}{2} = 5\ V$

iv) Bandwidth required, $B_w = f_{USB} - f_{LSB} = 100.5 - 99.5 = 1\ kHz.$

v) Total power delivered into a load of 600 Ω :

We know that, total power, $P_t = P_c\left(1+\frac{\mu^2}{2}\right) = \frac{A_c^2}{2R}\left(1+\frac{\mu^2}{2}\right) = \frac{(50)^2}{2\times600}\left(1+\frac{(0.2)^2}{2}\right) = 2.125\ W.$

vi) Transmission efficiency, $\eta = \frac{\mu^2}{2+\mu^2} \times 100\% = \frac{0.2^2}{2+0.2^2} \times 100 = 1.96\%$

17	**A broadcast transmitter radiates 20 kW when the modulation percentage is 75. How much of this is carrier power? Also calculate the power of each sideband.**	

Solution:

we know that, $P_t = P_c(1+\frac{\mu^2}{2})$

So, $P_c = \frac{P_t}{1+\frac{\mu^2}{2}} = \frac{20}{1+\frac{0.75^2}{2}} = 15.6\ kW$

18	**The total antenna current of an AM transmitter is 5A. If the modulation index is 0.6, calculate the antenna current when only the carrier is sent.**	

Solution:

We know that: $I_t = I_c\sqrt{1+\frac{\mu^2}{2}}$

So, $I_c = \frac{I_t}{\sqrt{1+\frac{\mu^2}{2}}} = \frac{5}{\sqrt{1+\frac{(0.6)^2}{2}}} = 4.6\ A$

19	**The rms antenna current of an AM transmitter increases by 15% over its un-modulated value, when sinusoidal modulation by 1 kHz signal is applied. Determine the Modulation index.**	

Solution:

Given that: $I_t = I_c + \left(I_c \times \frac{15}{100}\right) = 1.15\, I_c$

Now, we know that: $I_t = I_c\sqrt{1+\frac{\mu^2}{2}}$ $I_t = I_c\sqrt{1+\mu^2/2}$

So, $\mu^2 = 2\left(\frac{I_t^2}{I_c^2}-1\right) = 2\times((1.15)^2-1) = 0.645$

$\therefore \mu = \sqrt{0.645} = 0.8$

20	**Draw the frequency spectrum of $s(t) = A_{m1}\sin(2\pi f_{m1}t) + A_{m2}\sin(2\pi f_{m2}t)$. Also determine the total Power.**	

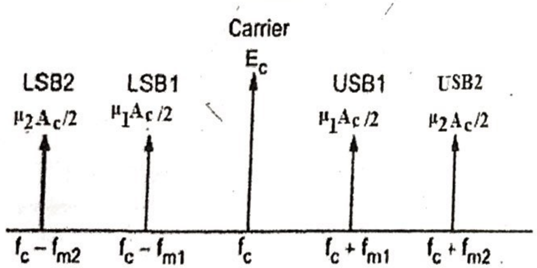


Fig:- frequency spectrum of AM wave with two modulating signal

(Only positive side is shown)

Total Power in the AM:

$$P_t = P_c + P_{USB1} + P_{USB2} + P_{LSB1} + P_{LSB2}$$

$$= \frac{A_c^2}{2R} + \frac{\left(\frac{\mu_1 A_c}{2}\right)^2}{2R} + \frac{\left(\frac{\mu_2 A_c}{2}\right)^2}{2R} + \frac{\left(\frac{\mu_1 A_c}{2}\right)^2}{2R} + \frac{\left(\frac{\mu_2 A_c}{2}\right)^2}{2R}$$

$$= \frac{A_c^2}{2R} + \frac{\mu_1^2 A_c^2}{8R} + \frac{\mu_2^2 A_c^2}{8R} + \frac{\mu_1^2 A_c^2}{8R} + \frac{\mu_2^2 A_c^2}{8R}$$

$$= \frac{A_c^2}{2R} + \frac{\mu_1^2 A_c^2}{4R} + \frac{\mu_2^2 A_c^2}{4R}$$

$$= \frac{A_c^2}{2R}\left(1 + \frac{\mu_1^2}{2} + \frac{\mu_2^2}{2}\right)$$

$$= P_c\left(1 + \frac{\mu_1^2}{2} + \frac{\mu_2^2}{2}\right)$$ $\mu = \sqrt{\mu_1^2 + \mu_2^2}$

Here, we can see that total modulation index, $\mu = \sqrt{\mu_1^2 + \mu_2^2}$

21	**Calculate the total modulation index if the carrier wave is amplitude modulated by three modulating signals with modulation indices 0.6, 0.3 and 0.4 respectively.**	

Solution:

$\mu_t = \sqrt{\mu_1^2 + \mu_2^2 + \mu_3^2}$

we know, total modulation index, $\mu_t = \sqrt{\mu_1^2 + \mu_2^2 + \mu_3^2} = 0.781$

22	**A certain AM transmitter radiates 10 kW with the carrier un-modulated, and 11.8 kW when the carrier is sinusoidally modulated. Calculate the modulation index. If another sine wave, corresponding to 30 percent modulation, is transmitted simultaneously, determine the radiated power.**	

We know that, $P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$ $P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$

So, $\mu^2 = 2\left(\frac{P_t}{P_c} - 1\right) = 2\left(\frac{11.8}{10} - 1\right) = 0.36$

$\therefore \mu = \sqrt{0.36} = 0.6$ $\mu = 0.6$

With 30% modulation of another modulating signal, we get,

$\mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.6^2 + 0.3^2} = 0.67$ $\mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.6^2 + 0.3^2}$

We know that, $P_{total} = P_c\left(1 + \frac{\mu_t^2}{2}\right) = 10\left(1 + \frac{(0.67)^2}{2}\right) = 12.24\, kW$ $= 0.67.$

23	**A 1000 kHz carrier is simultaneously modulated with 300 Hz, 800 Hz, 1 kHz audio sine waves. What will be frequency present in the output?**	

Solution:

$f_{USB1} = 1000\, kHz + 300\, Hz = 1000.3\, kHz$

$f_{LSB1} = 1000\, kHz - 300\, Hz = 999.7\, kHz.$

$f_{USB2} = 1000\, kHz + 800\, Hz = 1000.8\, kHz$

$f_{LSB2} = 1000\, kHz - 800\, Hz = 999.2\, kHz$

$f_{USB3} = 1000\, kHz + 1\, kHz = 1001\, kHz$

$f_{LSB3} = 1000\, kHz - 1\, kHz = 999\, kHz.$

24	A 360 W carrier is simultaneously modulated by two audio waves with modulation percentages of 55 and 65, respectively. What is the total sideband power radiated?

Solution:

$\mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.55^2 + 0.65^2} = 0.85$ $\mu_t =$

So, $P_{SB} = P_c\left(\frac{\mu_t^2}{2}\right) = 300 \times \frac{0.85^2}{2} = 108.375\ W$ $\frac{\mu_t^2}{2} \times P_c$

total power

$P_t = P_{SB} + P_{carrier}$

25	When a broadcast AM transmitter is 50 percent modulated, its antenna current is 12A. What will the current be when the modulation depth is increased to 0.9?

Solution:

First, we need to find the current when only carrier is transmitted.

$I_c = \frac{I_t}{\sqrt{(1+\mu^2)}} = \frac{12}{\sqrt{1+\frac{0.5^2}{2}}} = 11.31\ A$ $I_t = I_c\sqrt{1+\frac{\mu^2}{2}}$

Now, we will calculate antenna current with modulation index 0.9

$I_t = I_c\sqrt{1+\frac{\mu^2}{2}} = 11.31 \times \sqrt{1+\frac{0.9^2}{2}} = 13.41\ A.$

$0.6 = \mu$

26	The output current of 60 percent modulated AM generator is 1.5 A. To this what value will this current rise if the generator is modulated additionally by another audio wave, whose modulation index is 0.7? What will be the percentage power saving if the carrier and one of the sidebands are now suppressed?

Solution:

First we will need to calculate the antenna current when only the carrier is transmitted.

So, $I_c = \frac{I_t}{\sqrt{1+\frac{\mu^2}{2}}} = \frac{1.5}{\sqrt{1+\frac{0.6^2}{2}}} = 1.38A$

Now, total modulation index, $\mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.6^2 + 0.7^2} = 0.922$

Now, the transmitter current with modulation index 0.922

$I_t = I_c\sqrt{1+\frac{\mu_t^2}{2}} = 1.38 \times \sqrt{1+\frac{0.992^2}{2}} = 1.647A$

Now, for the second question, we know: $I_t = I_c\sqrt{1+\frac{\mu_t^2}{2}}$

Total power (including all sidebands), $P_t = P_c + P_{USB} + P_{LSB} = P_c + P_c\left(\frac{\mu^2}{4}\right) + P_c\left(\frac{\mu^2}{4}\right)$

And, if carrier and one of the sideband is suppressed, then Power required, $P_t = P_c\left(\frac{\mu^2}{4}\right)$

Therefore power saved, $P_c + P_c\left(\frac{\mu^2}{4}\right) + P_c\left(\frac{\mu^2}{4}\right) - P_c\left(\frac{\mu^2}{4}\right) = P_c + P_c\left(\frac{\mu^2}{4}\right)$

So, percentage power saving $= \frac{P_c + P_c\left(\frac{\mu^2}{4}\right)}{P_c\left(1+\frac{\mu^2}{4}+\frac{\mu^2}{4}\right)} = \frac{\left(1+\frac{\mu^2}{4}\right)}{1+\frac{\mu^2}{2}} = \frac{1+\frac{0.922^2}{4}}{1+\frac{0.922^2}{2}} = \frac{1.213}{1+0.425} = 0.851 = 85.1\%$

27	The output of a transmitter is given by $400(1+0.4\sin 6280t)\sin 3.14\times 10^7 t$. This voltage is fed to a load of 600 Ω resistance. Determine a) carrier frequency b) Modulation frequency c)carrier power d)total power output e) Peak power output

Solution:

Given, $s(t) = 400(1+0.4\sin 6280t)\sin 3.14\times 10^7 t$

single tone

Compare this equation with the standard equation of AM modulation for single tone. That is

$s(t) = A_c[1+\mu\cos(2\pi f_m t)]\cos(2\pi f_c t)$. $= A_c[1+\mu\cos(2\pi f_m t)]\cos(2\pi f$

It can also be expressed as cosine. That would be $s(t) = A_c[1+\mu\sin(2\pi f_m t)]\sin(2\pi f_c t)$

$S(t) = A_c[1+\mu\sin(2\pi f_m t)]\sin(2\pi f_c t)$

a) Carrier frequency, $f_c = \frac{3.14\times10^7}{2\pi} = 5\ MHz$

b) Modulating frequency, $f_m = \frac{6280}{2\pi} = 1000\ Hz$

c) Carrier power, $P_c = \frac{A_c^2}{2R} = \frac{(400)^2}{2\times600} = 133.33\ W$ $P_t = P_c\left(1+\frac{\mu^2}{2}\right)$

d) Total output power, $P_t = P_c\left(1+\frac{\mu^2}{2}\right) = 133.33\left(1+\frac{0.4^2}{2}\right) = 144\ W$

e) Peak output voltage, $= A_c + A_m = A_c + \mu A_c = A_c(1+\mu) = 400(1+0.4) = 560$

So, peak power output $= \frac{(peak\ output\ voltage)^2}{2R} = \frac{(560)^2}{2\times600} = 261.33\ W$ $= \frac{(peak\ o/p\ v)^2}{2R} =$

28	A carrier wave with amplitude 12 V and frequency 10 MHz is amplitude modulated to 50% level with a modulating frequency 1 kHz. Write down equation of the above wave and sketch the waveform in frequency domain.

Solution:

We know the standard equation of AM, $s(t) = A_c[1+\mu\cos(2\pi f_m t)]\cos(2\pi f_c t)$

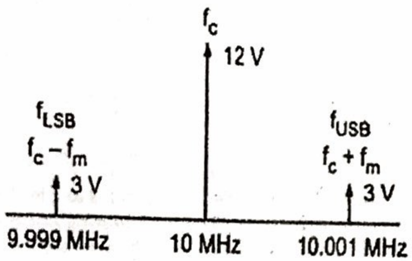
So, $s_{AM}(t) = 12[1+0.5\cos(2\pi 1000t)]\cos(2\pi 10\times10^6 t)$

Or, $s_{AM}(t) = 12[1+0.5\sin(2\pi 1000t)]\sin(2\pi 10\times10^6 t)$

$f_{USB} = f_c + f_m = 10\ MHz + 1\ kHz = 10.001\ MHz$

$f_{LSB} = f_c - f_m = 10\ MHz - 1\ kHz = 9.999\ MHz$

Amplitude of the sidebands $= \frac{\mu A_c}{2} = \frac{0.5\times12}{2} = 3\ V$



(only positive sides are shown)

29	**The positive peaks of an AM voltage wave rise to a maximum value of 15 V and drop to a maximum value of 5V. Determine the modulation index and the un-modulated carrier amplitude, assuming sinusoidal modulation.**

Solution:

Maximum value of AM wave, $A_c + A_m = 15$

And Minimum value of AM wave, $A_c - A_m = 5$

Solving, we get, $A_c = 10\ V$ and $A_m = 5\ V$

So, modulation index, $\mu = \frac{A_m}{A_c} = \frac{5}{10} = 0.5$

30	**In the trapezoidal pattern displaying modulation, the length of the long vertical side is 5 cm, and of the short vertical side, 2 cm. Determine the modulation depth.**

Solution:

For trapezoidal pattern, $\mu = \frac{L_1-L_2}{(L_1+L_2)} = \frac{5-2}{5+2} = 0.43$ $\mu = \frac{L_1 - L_2}{L_1 + L_2}$

31	**Calculate the percentage power saving when carrier and one of the sidebands are suppressed in an AM wave modulated to a depth of 100 and 50 percent.**

For modulation depth 100 percent:

Power required for double sideband with full carrier (DSBFC) for μ =1

$P_{DSBFC} = P_c\left(1+\frac{\mu^2}{2}\right) = P_c\left(1+\frac{1}{2}\right) = 1.5\ P_c$

Power required for single side band suppressed carrier for μ =1;

$P_{SSB} = P_c\left(\frac{\mu^2}{4}\right) = P_c\left(\frac{1}{4}\right) = 0.24\ P_c$

Hence, percentage power saving when μ = 1

$$\%\,powe\ savings = \frac{P_{DSBFC} - P_{SSB}}{P_{DSBFC}} \times 100\%$$

$$= \frac{1.5P_c - 0.25P_c}{1.5P_c} \times 100 = 83.33\%$$

For the modulation depth of μ = 0.5

$P_{DSBFC} = P_c\left(1+\frac{\mu^2}{2}\right) = 1.125\ P_c$

$P_{SSB} = P_c\left(\frac{\mu^2}{4}\right) = 0.0624\ P_c$

$\%\,power\ savings = \frac{P_{DSBFC} - P_{SSB}}{P_{DSBFC}} \times 100 = 94.44\%$

32	**A carrier wave $V_c = 4\sin(2\pi\times500\times10^3\times t)$ is amplitude modulated by an audio wave $V_m = 0.2\sin 3(2\pi\times500\times t) + 0.1\sin 5(2\pi\times500\times t)$. Determine the upper and lower sidebands and sketch the complete spectrum of the modulated wave. Estimate total power in sidebands.**

A carrier wave is given by $V_c = 4\sin(2\pi\times500\times10^3\times t)$ and a modulating audio wave is given by $V_m = 0.2\sin 3(2\pi\times500\times t) + 0.1\sin 5(2\pi\times500\times t)$

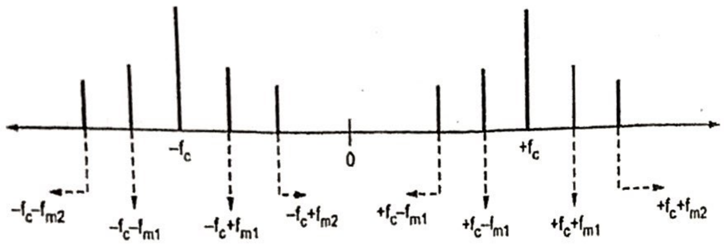
So, carrier frequency, $f_c = 500\ kHz$

There are two modulating frequencies: $f_{m1} = 1500\ Hz$ and $f_{m2} = 2500\ Hz$

Upper side band frequency components are: 501.5 kHz and 502.5 kHz

Lower side band frequency components are: 498.5 kHz and 497 kHz

The complete frequency spectrum is given below:



Now, modulation indices for the two modulating signal are:

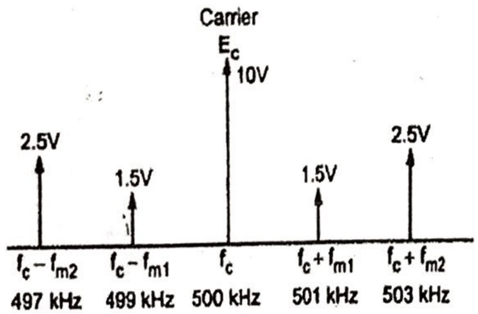
$\mu_1 = \frac{A_{m1}}{A_c} = \frac{0.2}{4} = 0.05$ and $\mu_2 = \frac{A_{m2}}{A_c} = \frac{0.1}{4} = 0.025$

$P_{sb} = P_c\left(\frac{\mu_t^2}{2}\right)$

Total modulation index, $\mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.05^2 + 0.025^2} = 0.056$

So, the total sideband power is given by $P_{Sb} = P_c\left(\frac{\mu_t^2}{2}\right) = 0.028\,P_c$; where P_c is the carrier power.

33	**A complex modulating waveform consisting of a sine wave of amplitude 3 V and frequency 1000 Hz plus a cosine wave of amplitude 5V and frequency 3000 Hz amplitude modulates a carrier of 500 kHz with 10 V peak. Plot the spectrum of modulated wave and determine average power when the modulated wave is fed into a 50 Ω load.**	



Now, $\mu_1 = \frac{A_{m1}}{A_c} = \frac{3}{10} = 0.3$ and $\mu_2 = \frac{A_{m2}}{A_c} = \frac{5}{10} = 0.5$ $\quad \mu_1, \mu_2$

The carrier power, $P_c = \frac{A_c^2}{2R} = \frac{10^2}{2\times50} = 1\,W$ $\quad P_c = \frac{A_c^2}{2R} = \frac{(10)^2}{2\times 50} = 1W$

So, total power, $P_t = P_c\left(1 + \frac{\mu_1^2}{2} + \frac{\mu_2^2}{2}\right) = 1 \times \left(1 + \frac{0.3^2}{2} + \frac{0.5^2}{2}\right) = 1.17\,W$.

1.17W.

34	**A broadcast radio transmitter radiates 10 kW, when the modulation percentage is 60. How much of this is carrier power?**	

Answer: $\quad \mu = 0.6$

we know, total power for AM, $P_t = P_c(1 + \frac{\mu^2}{2})$ $\quad P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$

So, $P_c = \frac{P_t}{1+\frac{\mu^2}{2}} = \frac{10}{1+\frac{0.6^2}{2}} = 8.47\,kW$ $\quad P_c = \frac{P_t}{(1+\mu^2/2)}$

35	**A radio transmitter radiates 10 kW and carrier power is 8.5 kW. Calculate modulation index.**	

Solution:

We know, total power of AM: $P_t = P_c(1 + \frac{\mu^2}{2})$ $\quad P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$

So, $\mu^2 = 2\left(\frac{P_t}{P_c} - 1\right) = 2\left(\frac{20}{8.5} - 1\right) = 0.353$ $\quad \mu = \sqrt{\left(\frac{P_t}{P_c} - 1\right)2}$

$\therefore \mu = \sqrt{0.353} = 0.59$

36	**A 400 W carrier is modulated to a depth of 7.5%. calculate total power in the modulated wave.**	

Solution:

$P_t = P_c\left(1 + \frac{\mu^2}{2}\right) = 401.125\,W$

37	**The antenna current of an AM transmitter is 8 Amps, when only the carrier is sent, but it increases to 8.93A, when the carrier is modulated by a single sine wave. Find percentage modulation. Determine the antenna current when the percent modulation changes to 0.8.**	

We know, $I_t = I_c\sqrt{1 + \frac{\mu^2}{2}}$ $\quad 8.93 = 8\sqrt{\left(1 + \frac{\mu^2}{2}\right)}$

So, $\mu^2 = 2\left(\frac{I_t^2}{I_c^2} - 1\right) = 2\left(\frac{8.93^2}{8^2} - 1\right) = 0.492$ $\quad \mu = 0.70$

$\therefore \mu = 0.70$

So, the percent modulation is 70%.

Now, for percent modulation 80% i.e. 0.8

Antenna current, $I_t = I_c\sqrt{1 + \frac{\mu^2}{2}} = 8 \times \sqrt{1 + \frac{0.8^2}{2}} = 9.19\,A$

38	Consider the message signal, m(t) = 20 cos (2πt) volts and the carrier wave, c(t) = 50cos(100πt) Volts. Now, (a) Sketch (to scale) the resulting wave for 75 % modulation. (b) Find the power developed across a load of 10 ohms due to this AM wave.	

Answer:

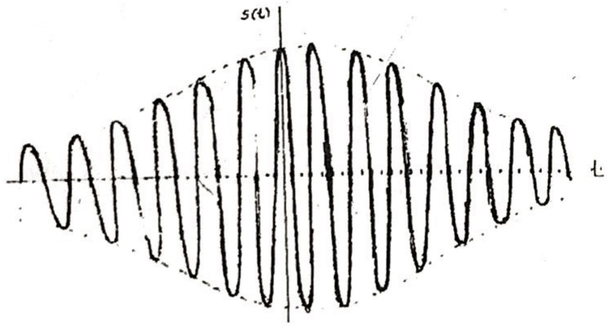
$s(t) = [1 + k_a m(t)]\, c(t)$

(a) An AM wave is defined by $s(t) = [1 + k_a m(t)]c(t)$

Now, for a percentage modulation of 75%, the corresponding AM wave is

$$s(t) = 50[1 + 0.75\cos(2\pi t)]\cos(100\pi t)$$

Hence, s(t) has the waveform:



Now the expression of s(t) can be found:

$s(t) = 50[1 + 0.75\cos(2\pi t)]\cos(100\pi t)$

$or, s(t) = 50\cos(100\pi t) + 0.75 \times 50\cos(100\pi t)\cos(2\pi t)$

$or, s(t) = 50\cos(100\pi t) + 37.5\cos(100\pi t)\cos(2\pi t)$

$or, s(t) = 50\cos(100\pi t) + \frac{37.5}{2} \times 2\cos(100\pi t)\cos(2\pi t)$

$or, s(t) = 50\cos(100\pi t) + 18.75\cos(102\pi t) + 18.75\cos(98\pi t)$

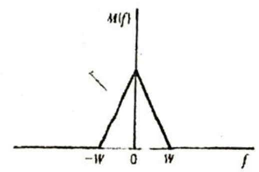
Hence, the average power of s(t) is

$$P_t = P_c + P_{USB} + P_{LSB}$$

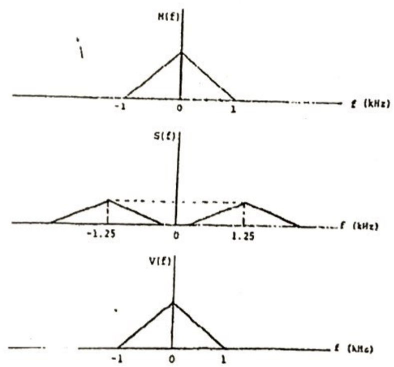
$$or, P_t = \frac{50^2}{2 \times 10} + \frac{18.75^2}{2 \times 10} + \frac{18.75^2}{2 \times 10}$$

$$P_t = 125 + 17.57 + 17.57$$

$$P_t = 160.16\ watts$$

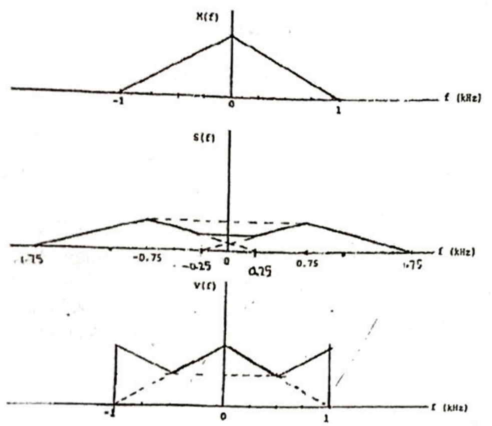
39	Consider a message signal m(t) with the spectrum shown in figure below. The message bandwidth W=1kHz. This signal is applied to a product modulator, together with a carrier wave $A_c\cos(2\pi f_c t)$, producing the DSBSC modulated wave s(t). This modulated wave is next applied to a coherent detector. Assuming perfect synchronism between the carrier waves in the modulator and detector, determine the spectrum of the detector output when: (a) the carrier frequency f_c=1.25 kHz and (b) the carrier frequency f_c=0.75 kHz. What is the lowest carrier frequency for which each component of the modulated wave s(t) is uniquely determined by m(t)? 	

For f_c = 1.25 kHz, the spectra of the message signal m(t), the product modulator output s(t), and the coherent detector output v(t) are as follows, respectively.



coherent detecton .

(c) For the case when f_c=0.75, the respective spectra are as follows:



$f_c \geq 1$ kHz.

To avoid side band overlap, the carrier frequency f_c must be greater than or equal to 1 kHz. The lowest carrier frequency is therefore 1 kHz for each side band of the modulated wave s(t) to be uniquely determined by m(t).

40	**Suppose we have carrier frequency ω = 2π× 10^5 rad/sec; Find the frequency component of Amplitude Modulated (AM) signal s(t) of the message signal given below:** **(a) m(t) = $A_0\cos(2\pi\times 10^3 t)$** **(b) m(t) = $A_0\cos(2\pi\times 10^3 t) + A_0\cos(4\pi\times 10^3 t)$** **(c) m(t) = $A_0\cos(2\pi\times 10^3 t)\sin(4\pi\times 10^3 t)$** **(d) m(t) = $A_0\cos^2(2\pi\times 10^3 t)$** **(e) m(t) = $\cos^2(2\pi\times 10^3 t) + \sin^2(4\pi\times 10^3 t)$** **(f) m(t) = $A_0\cos^3(2\pi\times 10^3 t)$**	

The AM signal is defined by ,$s(t) = A_c(1 + k_a m(t))\cos(\omega_c t)$

Where $A_c\cos(\omega_c t)$ is the carrier and k_a is a constant.

We are given, ω_c=2π× 10^5 rad/sec; so, f_c=100 kHz.

(a) m(t)=$A_0\cos(2\pi\times 10^3 t)$

so, $\omega_0 = 2\pi \times 10^3 t\ rad/sec$

$$\therefore f_0 = 1\ kHz$$

The frequency components of s(t) for positive frequencies are:

$$f_c = 100\ kHz$$

$$f_c + f_0 = 100 + 1 = 101\ kHz$$

$$f_c - f_0 = 100 - 1 = 99\ kHz$$

(b) m(t) = $A_0\cos(2\pi\times 10^3 t) + A_0\cos(4\pi\times 10^3 t)$

This message signal consists of two sinusoidal components with frequencies f_0=1 kHz and f_1 = 2 kHz. Hence, the frequency components of s(t) for positive frequencies are:

$$f_c = 100\ kHz$$

$$f_c + f_0 = 100 + 1 = 101\ kHz$$

$$f_c - f_0 = 100 - 1 = 99\ kHz$$

$$f_c + f_1 = 100 + 2 = 102\ kHz$$

$$f_c - f_1 = 100 - 2 = 98\ kHz$$

(c) m(t) = $A_0\cos(2\pi \times 10^3 t)\sin(4\pi \times 10^3 t)$

First, we note that, $2\cos A\sin B = \sin(A + B) + \sin(A - B)$

$$\therefore \cos A \sin B = \frac{1}{2}\sin(A + B) + \frac{1}{2}\sin(A - B)$$

Hence, $m(t) = A_0\cos(2\pi \times 10^3 t)\sin(4\pi \times 10^3 t)$

$$\therefore m(t) = \frac{A_0}{2}\sin(6\pi \times 10^3 t) + \frac{A_0}{2}\sin(2\pi \times 10^3 t)$$

Which consists of two sinusoidal components with frequencies $f_0 = 3\,kHz$ and $f_1 = 1kHz$.

The frequency components of s(t) for positive frequencies are:

$$f_c = 100kHz$$

$$f_c + f_0 = 100 + 3 = 103\ kHz$$

$$f_c - f_0 = 100 - 3 = 97\ kHz$$

$$f_c + f_1 = 100 + 1 = 101\ kHz$$

$$f_c - f_1 = 100 - 1 = 99\ kHz$$

(d) $m(t) = A_0\cos^2(2\pi \times 10^3 t)$

we know, $(1 + cos2\theta) = 2\cos^2\theta$

$$so, \cos^2\theta = \frac{1}{2}(1 + cos2\theta)$$

Hence, $m(t) = A_0 \cos^2(2\pi \times 10^3 t)$

$$\therefore m(t) = \frac{A_0}{2}[1 + \cos(4\pi \times 10^3 t)]$$

Which consists of dc components and sinusoidal component of frequency $f_0 = 2\ kHz$.

The frequency components of s(t) for positive frequencies are therefore:

$$f_c = 100\ kHz$$

$$f_c + f_0 = 100 + 2 = 102\ kHz$$

$$f_c - f_0 = 100 - 2 = 98\ kHz$$

(e) $m(t) = \cos^2(2\pi \times 10^3 t) + \sin^2(4\pi \times 10^3 t)$

we know, $(1 + cos2\theta) = 2\cos^2\theta$; $\quad \therefore \cos^2\theta = \frac{1}{2}(1 + cos2\theta)$

and, $(1 - cos2\theta) = 2\sin^2\theta$; $\quad \therefore \sin^2\theta = \frac{1}{2}(1 - cos2\theta)$

Hence, $m(t) = \frac{1}{2}[1 + \cos(4\pi \times 10^3 t)] + \frac{1}{2}[1 - \cos(8\pi \times 10^3 t)]$

$$\therefore m(t) = 1 + \frac{1}{2}\cos(4\pi \times 10^3 t) - \frac{1}{2}\cos(8\pi \times 10^3 t)$$

Which consists of dc component, and two sinusoidal components with $f_0 = 2\ kHz$ and $f_1 = 4\ kHz$. The frequency components of s(t) for positive frequencies are:

$$f_c = 100\ kHz$$

$$f_c + f_0 = 100 + 2 = 102\ kHz$$

$$f_c - f_0 = 100 - 2 = 98\ kHz$$

$$f_c + f_1 = 100 + 4 = 104\ kHz$$

$$f_c - f_1 = 100 - 4 = 96\ kHz$$

(f) $m(t) = A_0\cos^3(2\pi \times 10^3 t)$

we know, $\cos^3\theta = cos\theta.\frac{1}{2}[1 + cos2\theta]$

$$= \frac{1}{2}cos\theta + \frac{1}{2}cos\theta\cos(2\theta)$$

$$= \frac{1}{2}cos\theta + \frac{1}{2}[\frac{1}{2}\cos(\theta + 2\theta) + \frac{1}{2}\cos(2\theta - \theta)$$

$$= \frac{1}{4}\cos(3\theta) + \frac{3}{4}cos\theta$$

Hence, $m(t) = A_0 \cos^2(2\pi \times 10^3 t)$

$$\therefore m(t) = \frac{A_0}{4}\cos(2\pi \times 10^3 t) + \frac{3A_0}{4}\cos(2\pi \times 10^3 t)$$

Which consists two sinusoidal components with frequencies $f_0 = 1kHz$ and $f_1 = 3\ kHz$. The frequency components of s(t) are therefore:

$$f_c = 100kHz$$

$$f_c + f_0 = 100 + 1 = 101\ kHz$$

$$f_c - f_0 = 100 - 1 = 99\ kHz$$

$$f_c + f_1 = 100 + 3 = 103\ kHz$$

$$f_c - f_1 = 100 - 3 = 97\ kHz$$

Note: For negative frequencies, the frequency components of s(t) are the negative of those for positive frequencies.

41	**Suppose that on an AM signal, the $V_{max(p-p)}$ value read from the graticule on the oscilloscope screen is 5.9 divisions and $V_{min(p-p)}$ is 1.2 divisions.** **a. What is the modulation index?** **b. Calculate V_c, V_m and m if the vertical scale is 2 V per division.**	

Answer:

(a) modulation index, $m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{5.9-1.2}{5.9+1.2} = 0.662$

$$M = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

(b) $V_c = \left(\frac{V_{max} + V_{min}}{2}\right) = \frac{5.9+1.2}{2} = 3.55$; but given that 2V per divistion

So, V_c=3.55× 2=7.1V

$$V_m = \frac{V_{max} - V_{min}}{2} = \frac{5.9 - 1.2}{2} = 2.35; but\ given\ 2v\ per\ division$$

So, V_m=2.35× 2=4.7 V

Now, $m = \frac{V_m}{V_c} = \frac{4.7}{7.1} = 0.662$

42	**A standard AM broadcast station is allowed to transmit modulating frequencies up to 5 kHz. If the AM station is transmitting on a frequency of 980 kHz, compute the maximum and minimum upper and lower sidebands and the total bandwidth occupied by the AM station.**	

$$f_{USB} = 980 + 5 = 985\ kHz$$

$$f_{LSB} = 980 - 5 = 975\ kHz$$

$$BW = f_{USB} - f_{LSB} = 985 - 975 = 10\ kHz$$

or,

$$BW = 2(5\ kHz) = 10\ kHz$$

43	**Frequency Modulation বর্ণনা করুন। What is modulation index for FM Modulation?**	25th BCS

There is another way of modulation of a sinusoidal carrier wave, namely, angle modulation in which the angle of the carrier wave is varied according to the baseband signal. That is, Angle modulation is a method of analog modulation in which either the phase or the frequency of the carrier wave is varied according to the message signal. In this method of modulation the amplitude of the carrier wave is maintained constant. An important feature of angle modulation is that it can provide better discrimination against noise and interference than Amplitude modulation. The improvement in performance is achieved at the expense of increased transmission bandwidth; that is, angle modulation provides us with a a practical means of exchanging transmission bandwidth for improved noise performance. Such a tradeoff is not possible with angle modulation.

In general form, an angle modulation signal can be represented as:

$$s(t) = A_c \cos[\theta(t)]$$

Where A_c is the amplitude of the carrier wave and θ(t) is the angle of the modulated carrier and also the function of the message signal. The instantaneous frequency of the angle modulated signal, s(t) is given by:

$$f_i(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt}$$

There are two forms of angle modulation. They are: Phase modulation and Frequency Modulation.

Phase modulation (PM) is that form of angle modulation in which the angle $\theta_i(t)$ is varied linearly with the baseband signal m(t), as shown by

$$\theta_i(t) = 2\pi f_c t + k_p m(t)$$

The term $2\pi f_c t$ represents the angle of the unmodulated carrier, and the constant k_p represents the phase sensitivity of the modulator, and $m(t)$ is a voltage waveform.

Now the phase modulated wave $s(t)$ is thus described in the time domain by:

$$s(t) = A_c \cos[2\pi f_c t + k_p m(t)]$$

Frequency Modulation (FM) is that form of angle modulation in which the instantaneous frequency $f_i(t)$ is varied linearly with the baseband signal $m(t)$ as shown by:

$$f_i(t) = f_c + k_f m(t)$$

The term f_c represents the frequency of the unmodulated carrier, and the constant k_f represents the frequency sensitivity of the modulator, and $m(t)$ is a voltage waveform.

Now the frequency modulated wave is represented by:

$$s(t) = A_c \cos[2\pi f_c t + 2\pi k_f \int_0^t m(t)dt]$$

Where, the instantaneous frequency of the resulting FM wave equals

$$f_i(t) = f_c + k_f m(t)$$

And the angle $\theta_i(t)$ is represents by

$$\theta_i(t) = 2\pi \int_0^t f_i(t)dt$$

$$or, \theta_i(t) = 2\pi f_c t + 2\pi k_f \int_0^t m(t)dt$$

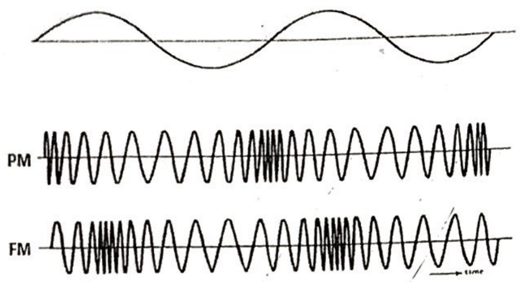


Fig: . – PM and FM Waveforms with a message signal

Frequency Deviation: The frequency deviation represents the maximum departure of the instantaneous frequency of the FM wave from the carrier frequency, f_c. In other word, frequency deviation is the multiplication of frequency sensitivity of modulator k_f and the amplitude of the message signal, A_m. Frequency deviation is represented by Δf.

$$now, \Delta f = k_f A_m \ \ ; where\ A_m\ is\ the\ Amplitude\ of\ message\ singal.$$

So, it is clear from the above equation that frequency deviation Δf is proportional to the amplitude of the modulating wave, and is independent of the modulating frequency.

Modulation index of FM: The ratio of the frequency deviation Δf to the modulating frequency f_m is commonly called the modulation index of the FM wave. We denote it by β, so that we may write

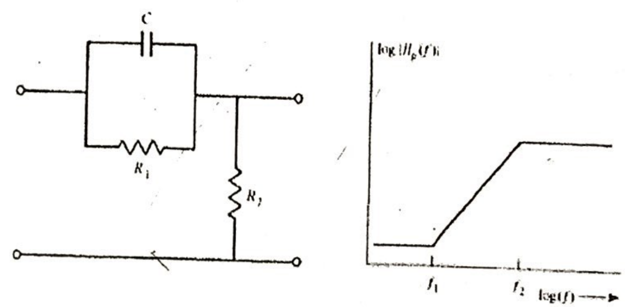
$$\beta = \frac{\Delta f}{f_m}$$

44	**What is the function of Pre-emphasis and De-emphasis circuits?**	BCS

Emphasis: Emphasis is the process of boosting the Amplitude vs Frequency characteristics of a signal to reduce the effects caused by noise while transmission or reception of message signal over the channel. The noises that may occur include both single frequency interference and thermal noise. We know that the signal with higher modulation frequencies have low SNR

(signal to noise ratio). So, Noise is inherently greater in amplitude at higher frequencies. By emphasis, approximately a 12 dB of improvement in noise can be achieved in FM.

Pre-emphasis: Signals with higher modulation frequencies have lower SNR. In order to compensate this, the high frequency signals are emphasized or boosted in amplitude at the transmission section of a communication system prior to the modulation process. That is, the pre-emphasis network allows the high frequency modulating signal to modulate the carrier at higher level, this causes more frequency deviation.



(b) Preemphasis Filter (a high pass filter)

(c) Bode Plot of Preemphasis Frequency Response

Fig: Pre-emphasis filter circuit with bode plot

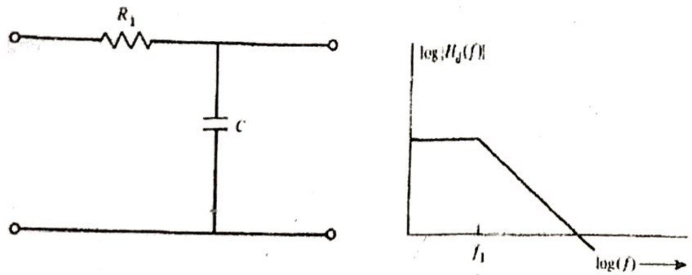
The figure shows an active pre-emphasis network, which consisting of a transistor, resistor and an inductor. It is basically a high pass filter. A pre-emphasis circuit produces a constant increase in the amplitude of the modulating signal with an increase in frequency. The Break Frequency is determined by the RC or L/R times constant of the network. Normally, the break frequency occurs at the frequency where XC or XL equals R.

Note: Break Frequency is the frequency where Pre-emphasis or De-emphasis just begins.

$$Break\ Frequency = \frac{1}{2\pi RC}\ \ or\ Break\ Frequency = \frac{1}{2\pi\left(\frac{L}{R}\right)}$$

By the use of an active pre-emphasis network, we can reduce the signal loss and distortion with the increase of SNR. Also the output amplitude of the network increases with frequencies above Break Frequency.

De-emphasis: De-emphasis is the inverse process of pre-emphasis, used to attenuate the high frequency signal is boosted at the transmitter section. The de-emphasis network at the receiver section restores the original amplitude vs frequency characteristics of the information signal, after the demodulation process.



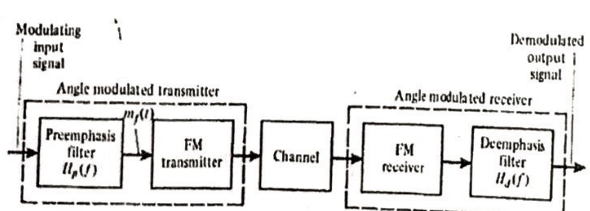
(d) Deemphasis Filter (e) Bode Plot of Deemphasis Characteristic

(a low pass filter)

Fig: De-emphasis circuit with bode plot

The above circuit shows a passive de-emphasis network consisting of a resistor and a capacitor. It is basically a low pass filter or an integrator.

The pre-emphasis network in front of the FM modulator and a de-emphasis network at the output of the FM demodulator improves the Signal to Noise Ratio for higher modulating signal frequencies, thus producing a more uniform SNR at the output of demodulator.



(a) Overall Block Diagram

Fig: Block Diagram of pre-emphasis and de-emphasis

45	**How is FM signal generated?**

There are two basic methods of generating frequency modulated signals. They are: i) Direct Method and ii) Indirect Method.

Direct FM:

$$f_i = f_c = k_f m(t)$$

In a direct FM system the instantaneous frequency is directly varied with the information signal. That is, the carrier frequency is directly varied by the input signal. In this method, the voltage controlled oscillator or VCO varies carrier with the baseband amplitude. It can be accomplished by a Voltage Controlled Oscillator (VCO), whose out frequency is proportional to the voltage of the input signal.

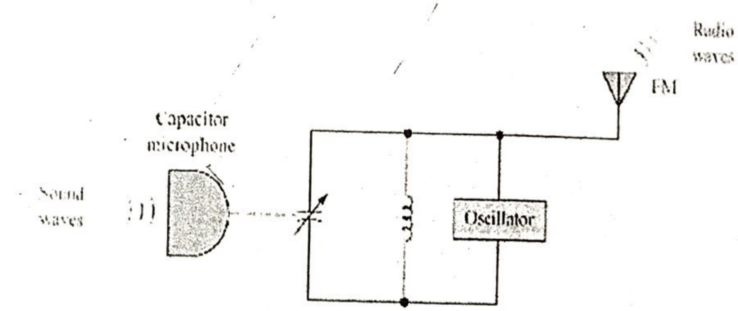


Fig: Block diagram of FM signal generation with direct method

There are some problems of direct FM generator. Crystal oscillator cannot be used in direct FM because its frequency is too stable and is difficult to change. The frequency deviation with direct FM is only about 5 KHz which is too small for wideband FM. It should be noted that the maximum frequency deviation in commercial FM radio is 75 kHz.

Indirect Method: Armstrong Modulator:

In indirect method, a NBFM wave is generated first and frequency multiplication is next used to increase the frequency deviation to the desired level.

In this method, we first obtain Narrow Band Frequency Modulation (NBFM) via a NBFM circuit with crystal oscillator. Then, apply frequency multiplier to increase both the carrier frequency and frequency deviation. Indirect FM is preferred when the stability of carrier frequency is of major concern (e.g. in commercial FM broadcasting). Actually, indirect method is wide band FM.

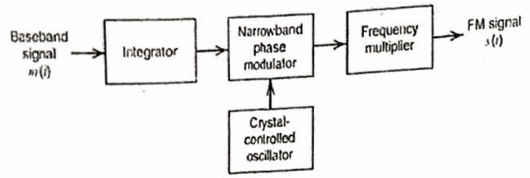


Fig: FM signal generation with indirect method

46	**Describe the generation of NBFM**	

A narrow band FM is the FM wave with a small bandwidth. The modulation index of narrow band FM is small as compared to one radian.

A frequency modulated wave is defined as:

$$s(t) = A_c \cos[2\pi f_c t + \phi(t)]$$

Where $\phi(t) = 2\pi k_f \int_0^t m(t)dt$

Now, we know, $\cos(A + B) = cosAcosB - sinAsinB$

So, $s(t) = A_c \cos(2\pi f_c t)\cos[\phi(t)] - A_c \sin(2\pi f_c t)\sin[\phi(t)]$

Assuming $\phi(t)$ is small, then using $\cos[\phi(t)] = 1$ and $\sin[\phi(t)] = \phi(t)$.

So, $s(t) = A_c \cos(2\pi f_c t) - A_c \sin(2\pi f_c t)\,[\phi(t)]$

Or, $s(t) = A_c \cos(2\pi f_c t) - 2\pi k_f A_c \sin(2\pi f_c t) . \int_0^t m(t)dt$

The above equation defines a narrow band FM wave. The generation scheme of such a narrow band FM wave is shown in the figure below.

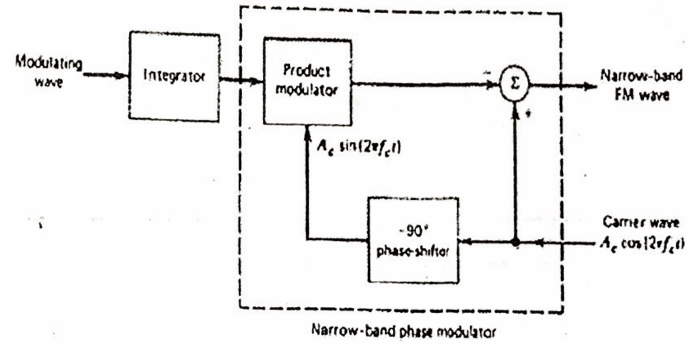


Fig: – Scheme to generate a NBFM Waveform.

The narrow band FM wave, thus generated will have some higher order harmonic distortion. This distortion can be limited to negligible levels by restricting the modulation index to $\beta < 0.5$ radians.

47	**Derive the equation for single tone frequency modulation.**	

Consider a sinusoidal modulating wave defined by

$$m(t) = A_m \cos(2\pi f_m t)$$

The instantaneous frequency of the resulting FM wave equals

$$f_i(t) = f_c + k_f A_m \cos(2\pi f_m t)$$

$$= f_c + \Delta f cos(2\pi f_m t)$$

Where the quantity $\Delta f_m = k_f A_m$ is called the frequency deviation

Now, we know, the frequency of angle modulated wave is $f_i(t) = \frac{1}{2\pi}\frac{d\theta_i(t)}{dt}$

so, the instantaneous angle, $\theta_i(t) = 2\pi \int_0^t f_i(t)dt$

$$= 2\pi \int_0^t [f_c + \Delta f cos(2\pi f_m t)]$$

$$= 2\pi f_c t + 2\pi\, \Delta f \frac{\sin(2\pi f_m t)}{2\pi f_m}$$

$$= 2\pi f_c t + \frac{\Delta f}{f_m}\sin(2\pi f_m t)$$

$$= 2\pi f_c t + \beta \sin(2\pi f_m t)$$

Where, $\beta = \frac{\Delta f}{f_m}$; *modulation index*

The resultant FM signal is $s(t) = A_c \cos[2\pi f_c t + \beta \sin (2\pi f_m t)]$

$$s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$$

The frequency deviation factor indicates the amount of frequency change in the FM signal from the carrier frequency f_c on either side of it. Thus FM signal will have the frequency components between $(f_c - \Delta f)$ to $(f_c + \Delta f)$. The modulation index, β represents the phase deviation of the FM signal and is measured in radians. Depending on the value of β, FM signal can be classified into two types:

$-\Delta f)$

$+\Delta f)$

1. Narrow band FM ($\beta \ll 1$) and
2. Wide band FM ($\beta \gg 1$)

10V f^{cy} 10 kHz

48	A sinusoidal wave of amplitude 10 volts and frequency of 10 kHz is applied to an FM generator that has a frequency sensitivity constant of 40 Hz/volt. Determine the frequency deviation and modulation index.	

40 Hz/volt.

Message signal Amplitude, A_m=10 volts, Frequency $f_m = 1000\ Hz$ and the frequency sensitivity, $k_f = 40\ Hz/volt$

Frequency deviation, $\Delta f = k_f A_m = 40 \times 10 = 400\ Hz$ $\quad \Delta f = k_f \cdot A_m$

Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{400}{1000} = 0.4$, *(indicates a narrow band FM)* $\quad \beta = \frac{\Delta f}{f_m}$

49	**Derive the equation of Narrow Band FM. Draw the frequency spectrum of NBFM.**	

We know the FM signal, $s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$

Now, $\cos(A + B) = cosAcosB - sinAsinB$

$\therefore s(t) = A_c \cos(2\pi f_c t) \cos[\beta \sin(2\pi f_m t)] - A_c \sin(2\pi f_c t) \sin[\beta \sin(2\pi f_m t)]$

Now, For NBFM, ($\beta \ll 1$), *we can approximate,*

$\cos[\beta \sin(2\pi f_m t)] \approx 1$ and $\sin[\beta \sin(2\pi f_m t)] \approx \beta \sin(2\pi f_m t)$

*

Note: if θ is very very small, than, $\cos\theta \approx 1$ and $\sin\theta \approx \theta$

Hence, $s(t) = A_c \cos(2\pi f_c t) - A_c\beta \sin(2\pi f_c t) \sin(2\pi f_m t)$

Now, we know, $2sinAsinB = \cos(A - B) - \cos(A + B)$

so, $s(t) = A_c \cos(2\pi f_c t) - A_c\beta \cdot \frac{1}{2}[\cos\{2\pi(f_c - f_m)t\} - \cos\{2\pi(f_c + f_m)t\}]$

or, $s(t) = A_c \cos(2\pi f_c t) - \frac{A_c\beta}{2}\cos\{2\pi(f_c - f_m)t\} + \frac{A_c\beta}{2}\cos\{2\pi(f_c + f_m)t\}$.

The above equation represents the NBFM signal. This representation is similar to an AM signal, except that the lower side band frequency has negative sign. The magnitude spectrum of NBFM signal is shown below, which is similar to AM signal spectrum. The Bandwidth of NBFM signal is $2f_m$, which is same as AM signal.

$2f_m$

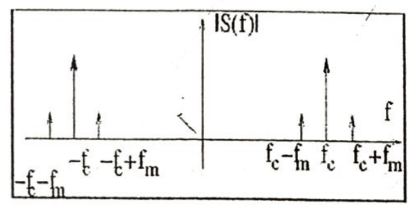


Fig: - Magnitude Spectrum of NBFM Waveform.

50	**Find the instantaneous frequency of the following waveforms:** (a) $s_1(t) = A_c\cos[100\pi t + 0.25\pi]$ (b) $s_2(t) = A_c\cos[100\pi t + \sin(20\pi t)]$ (c) $s_3(t) = A_c\cos[100\pi t + (\pi t^2)]$	

Answer:

The instantaneous frequency of the angle modulated signal, s(t) is given by:

$$f_i(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt} \qquad * \ f_i(t) = \frac{1}{2\pi} \cdot \frac{d\theta(t)}{dt}$$

(a) $f_1 = \frac{1}{2\pi}\frac{d}{dt}[100\pi t + 0.25\pi] = \frac{1}{2\pi} \times 100\pi = 50\ Hz.$

So instantaneous frequency in this case is constant.

(b) $f_2 = \frac{1}{2\pi}\frac{d}{dt}[100\pi t + \sin(20\pi t)]$

$$= \frac{1}{2\pi}\frac{d}{dt}[100\pi t] + \frac{1}{2\pi}\frac{d}{dt}[\sin(20\pi t)]$$

$$= \frac{1}{2\pi} \times 100\pi + \frac{1}{2\pi}\cos(20\pi t)\frac{d}{dt}(20\pi t)$$

$$= 50 + \frac{1}{2\pi} \times 20\pi \times \cos(20\pi t)$$

$$= 50 + 10\cos(20\pi t)$$

এখন, আমরা জানি, cos এর সর্বোচ্চ মান +1 আর সর্বনিম্ন মান -1 ।

So, the maximum value of the frequency is $(50 + 10) = 60\,Hz$

And, the minimum value of the frequency is $(50 - 10) = 40$ Hz.

Hence, the instantaneous frequency oscillates between 60 Hz and 40 Hz. 60Hz & 40 Hz

(c) $f_3 = \frac{1}{2\pi}\frac{d}{dt}[100\pi t + (\pi t^2)]$

$$= \frac{1}{2\pi}\frac{d}{dt}[100\pi t] + \frac{1}{2\pi}\frac{d}{dt}(\pi t^2)$$

$$\boxed{= 50 + t}$$

t = 0 $f_3 = 50$ Hz

So, the instantaneous frequency depends on t. When, t=0, instantaneous frequency is 50 Hz and varies linearly at 1 Hz / second.//

51	What is the relation between Frequency Modulation and Phase Modulation?	

Answer:

A frequency modulated signal can be generated using a phase modulator by first integrating m(t) and using it as an input to a phase modulator. This is possible by considering FM signal as phase modulated signal in which the modulating wave is integral of m(t) in place of m(t).

Similarly, a PM signal can be generated by first differentiating m(t) and then using the resultant signal as the input to a FM modulator. //

integrat m(t) → phase modulator → FM signal

differentiate m(t) → FM modulator → PM signal,

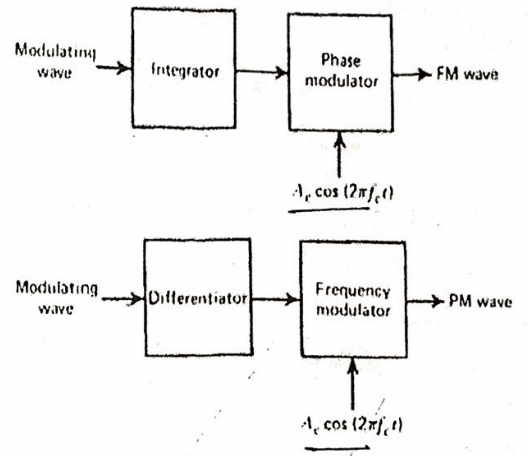


Fig: – Scheme for generation of FM and PM Waveforms //

52	Describe the transmission bandwidth of FM. //	

In theory, an FM contains an infinite number of side band frequencies so that the bandwidth is theoretically infinite. But, in practice, the FM wave is effectively limited to a finite number of side band frequencies compatible with a small amount of distortion. There are many ways to find the bandwidth of FM wave.

* **Carson's Rule: Carson's Rule:**

In single tone modulation, for the smaller values of modulation index the bandwidth is approximately as $2f_m$. $2f_m$

For the higher values of modulation index, the bandwidth is considered as slightly greater than the total deviation $2\Delta f$. Thus the Bandwidth for sinusoidal modulation is defined as:

$$\boxed{B_T \cong 2\Delta f + 2f_m}$$

$2\Delta f$

$B_T = 2\Delta f + 2f_m$

$$or, B_T = 2\Delta f(1 + \frac{1}{\beta})$$

$\beta = \frac{\Delta f}{f_m}$

$$or, B_T = 2(1 + \beta)f_m$$ //

For non-sinusoidal modulation, a factor deviation ratio (D) is considered. The deviation ratio is defined as the ratio of maximum frequency deviation to the bandwidth of message signal.

Deviation ratio, $D = \frac{\Delta f}{W}$

Deviation ratio, $D = \left(\frac{\Delta f}{W}\right)$, where W is the bandwidth of the message signal. The corresponding bandwidth of the FM signal is,

$$B_T = 2(1 + D)W \qquad B_T = 2(1+D)W$$

Universal Curve: An accurate method of bandwidth assessment is done by retaining the maximum number of significant side frequencies with amplitudes greater than 1% of the un-modulated carrier wave. *Thus the bandwidth of an FM wave is defined as the separation between the two frequencies beyond which none of the side frequencies is greater than 1% of the carrier amplitude obtained when the modulation is removed.*

Then, Transmission Bandwidth, $BW = 2n_{max} f_m$ $\qquad BW = 2n_{max} f_m$

Where, f_m is the modulation frequency and n_{max} is the maximum number of significant frequency side band.

n_{max} is the maximum value of the integer n which satisfies the requirement $|J_m(\beta)| > 0.01$

$|J_m(\beta)| > 0.01$

Here $J_m(\beta)$ is the Bessel Function.

The value of n_{max} varies with the modulation index β and can be determined from the table given below:

Table

Number of Significant Side Frequencies of a Wide-band FM Signal for Varying Modulation Index

Modulation Index β	Number of Significant Side Frequencies $2n_{max}$
0.1	2
0.3	4
0.5	4
1.0	6
2.0	8
5.0	16
10.0	28
20.0	50
30.0	70

We should keep this remember for Bessel function Table in case of FM.

For, $\beta = 0.1$; $n_{max} = 1$

For, $\beta = 0.3$; $n_{max} = 2$

For, $\beta = 0.5$; $n_{max} = 2$

For, $\beta = 1$; $n_{max} = 3$

For, $\beta = 2$; $n_{max} = 4$

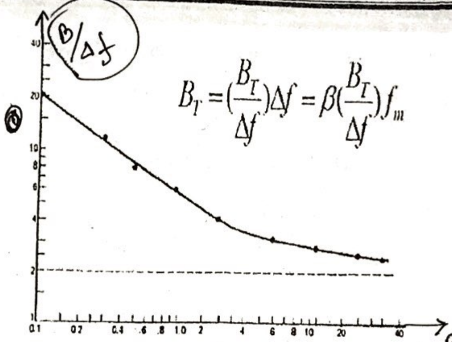
For, $\beta = 5$; $n_{max} = 8$

For, $\beta = 10$; $n_{max} = 14$

For, $\beta = 20$; $n_{max} = 25$

For, $\beta = 30$; $n_{max} = 35$

5 $\qquad$ 16



β (mod. index)

$$B_T = \left(\frac{B_T}{\Delta f}\right)\Delta f = \beta\left(\frac{B_T}{\Delta f}\right) f_m$$

For universal curve, we should keep remember:

For $\beta = 5$; $\quad \frac{B_T}{\Delta f} = 3.2$

[redacted]

Fig: ... – Universal Curve

53	**Find the bandwidth of a single tone modulated FM signal described by $S(t) = 10\cos[2\pi 10^8 t + 6\sin(2\pi 10^3 t)]$**	

$\beta = 6, \; f_m = 10^3$

Compare the given S(t) with the general equation of FM which is,

$s(t) = A_c \cos\{(2\pi f_c t + \beta \sin(2\pi f_m t)\}$

So, we get, carrier frequency, $f_c = 10^8\ Hz$

Message signal frequency, $f_m = 10^3 = 1000\ Hz$

Modulation index, $\beta = 6$

$$\begin{aligned} B_T &= 2\Delta f + 2f_m \\ &= 2\Delta f\left(1 + \frac{1}{\beta}\right) \\ &= 2f_m(1+\beta) \\ &= 2(1+D)W \end{aligned}$$

We know from Carson's rule, Transmission bandwidth, $B_T = 2(1+\beta)f_m$

$$= 2 \times (1 + 6) \times 1000$$

$$= 14\ kHz.$$

54	**A carrier wave of frequency 91 MHz is frequency modulated by a sine wave of amplitude 10 volts and 15 kHz. The frequency sensitivity of the modulator is 3 kHz / V.** **(a) Determine the approximate bandwidth of FM wave using Carson's Rule** **(b) Repeat part (a), assuming that the amplitude of the modulating wave is doubled.** **(c) Repeat part (a), assuming that the frequency of the modulating wave is doubled.**	

(a) Frequency deviation, $\Delta f = k_f A_m = 3 \times 10 = 30\ kHz$ $\Delta f = k_f \cdot A_m$

Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{30}{15} = 2$ $\beta = \frac{\Delta f}{f_m}$

By Carson's rule, Bandwidth, $B_T = 2(1+\beta)f_m = 2 \times (1+2) \times 15 = 90\ kHz$

(b) When the amplitude is doubled: $B_T = 2\Delta f(1 + 1/\beta)$

Frequency Deviation, $\Delta f = k_f A_m = 3 \times (2 \times 10) = 60\ kHz$

Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{60}{15} = 4$

Bandwidth, $B_T = 2(1+\beta)f_m = 2 \times (1+4) \times 15 = 150\ kHz.$

(c) When the frequency of the message signal, is doubled: $2W(1+D)$

Frequency Deviation, $\Delta f = k_f A_m = 3 \times 10 = 30\ kHz$

Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{30}{2 \times 15} = 1$ $B_T = 2f_m(1+\beta)$

Bandwidth, $B_T = 2(1+\beta)f_m = 2 \times (1+1) \times (15 \times 2) = 120\ kHz.$

55	**Determine the bandwidth of an FM signal, if the maximum value of the frequency deviation Δf is fixed, at 75 kHz. for commercial FM broadcasting by radio and modulation frequency is W = 15 kHz.**

Solution:

Frequency Deviation ratio, $\boxed{D = \frac{\Delta f}{W} = \frac{75}{15} = 5;}$ $B_T = 2W(1+D)$

Where Δf is the frequency deviation and W is the bandwidth of message signal.

So, the transmission bandwidth, $B_T = 2(1+D)W = 2 \times (1+5) \times 15 = 180\ kHz.$

Note: The deviation ratio D plays the same rule for non-sinusoidal modulation that the modulation index β plays for the case of sinusoidal modulation. Then, replacing β by D and replacing f_m by W, we use Carson's rule or the universal curve to obtain a value for the transmission bandwidth of the FM. From a practical view point, Carson's rule somewhat underestimates the bandwidth requirement of an FM system, whereas using the universal curve yields a somewhat conservative result. //

For example, from the universal curve, we can see, for $D = 5$; $\frac{B_T}{\Delta f} = 3.2$

$$or, B_T = 3.2 \times \Delta f$$

β by D

I luW

$$or, B_T = 3.2 \times 75 = 240\ kHz$$

Thus, Carson's rule underestimates the transmission bandwidth by $\frac{240-180}{240} \times 100 = 25\%$ compared with the result of using the universal curve.

Again, we can determine the bandwidth from the wide band significant side frequencies table. From the table, we can see, for modulation index or deviation ratio 5, the value of $2n_{max} = 16$. So, the bandwidth, $B_T = 2n_{max} f_m = 16 \times 15 = 240\ kHz.$ which is same as determined by universal curve. //

56	**Consider an FM signal obtained from a modulating signal frequency of 2000 Hz and maximum Amplitude of 5 volts. The frequency sensitivity of modulator is 2 kHz/V. Find the bandwidth of the FM signal considering only the significant side band frequencies.**

Solution: $f_m = 2000\ Hz$

Frequency Deviation, $\Delta f = k_f A_m = 2 \times 5 = 10\ kHz.$ $\Delta f = k_f \times A_m$

Modulation Index, $\beta = \frac{\Delta f}{f_m} = \frac{10 \times 1000}{2000} = 5;$ $\beta = \Delta f / f_m$

By universal curve method: We can see from table of universal curve method that

for $\beta = 5$; $2n_{max} = 16$

so, Bandwidth, $B_T = 2n_{max} f_m = 16 \times 2000 = 32\ kHz$. $\boxed{B_T = 2n_{max} f_m}$

By Carson's rule: Bandwidth, $B_T = 2(1+\beta)f_m = 2 \times 6 \times 2000 = 24\ kHz.$ = 1

Universal curve method is more accurate compared to Carson's Rule.//

57	**A carrier wave of frequency 91 MHz is frequency modulated by a sine wave of amplitude 10 volts and 15 kHz. The frequency sensitivity of the modulator is 3 kHz/V. Determine the bandwidth by transmitting only those side frequencies with amplitude that exceed 1% of the un-modulated carrier wave amplitude. User universal curve for this calculation.**

Solution:

Frequency Deviation, $\Delta f = k_f A_m = 3 \times 10 = 30\ kHz$ $\Delta f = k_f A_m$

Modulation Index, $\beta = \frac{\Delta f}{f_m} = \frac{30}{15} = 2$ $\beta = 5,\ n_{max} = 8$

From the universal curve; for $\beta = 2$; $\left(\frac{B}{\Delta f}\right) = 4.3$ $\beta = 2\ n_{max} = 4$

$$or, B = 4.3 \times \Delta f$$

$$or, B = 4.3 \times 30\ k\ Hz$$

$$\therefore B = 129\ kHz.$$

58 //	Advantages and disadvantages of FM over AM	

Advantages of FM over AM are:

less power
w distortion

1. It requires less radiated power
2. Low distortion is occurred due to improved signal to noise ratio (about 25dB) with respect to man made interference. SNR
3. Geographical interference between neighbouring stations is smaller.
4. It has well defined service areas for given transmitted power.

Disadvantages of FM:

W বেশি
nplicated
r spectrum

1. Much more Bandwidth (as much as 20 times as much) is required.
2. It requires more complicated receiver and transmitter.
3. It uses too much spectrum space. //

59	Applications of FM	

Some of the applications of the FM modulation are listed below:

i) FM radio, (88-108MHz band, 75 kHz) 88 – 108 MHz , 75 kHz
ii) TV sound broadcast, 25 kHz
iii) 2-way mobile radio, 5 kHz or 2.5 kHz.

Note: FM broadcasting is a method of radio broadcasting using frequency modulation technology. It was invented by American engineer Edwin Armstrong in 1933. It is used world wide to provide high fidelity sound over broadcast radio. FM broadcasting is capable of better sound quality than AM broadcasting. So, it is used for most music broadcasts. FM radio stations use the VHF (very high frequency) frequencies. FM radio frequency band is 88-108 MHz.

60 //	**An FM wave is defined below:** $s(t) = 12\sin(6 \times 10^8 \pi t + 5\sin 1250\pi t)$ **Find the carrier and modulating frequencies, the modulating index, and the maximum deviation of the FM wave. Also, find the band width of the FM. What power will the FM wave dissipate in a 10 ohm resistor?**	

Solution:

We know the general equation of FM wave is

$$s(t) = A_c\cos(2\pi f_c t + \beta\sin(2\pi f_m t))$$

Comparing with the given FM wave,

এখানে দেখা যাচ্ছে, প্রশ্নে আছে সাইন, কিন্তু জেনেরাল ইকোয়েশনে আছে, কস। এইটা কোনো সমস্যা না। সাইন আর কস একই। শুধু কস সাইন এর চেয়ে ৯০ ডিগ্রি ল্যাগ থাকে।অর্থাৎ,

$$sin\theta = \cos(\theta - 90°)$$

Carrier frequency $= 3 \times 10^8 = 300\ MHz$

Modulating signal frequency, $f_m = \frac{1250}{2} = 625\ Hz.$

Modulation index, $\beta = 5.$

Maximum frequency deviation, $\Delta f = \beta f_m = 5 \times 625 = 3125\ Hz.$

Using Carson's rule, Bandwidth

$$B_T = 2(1+\beta)f_m = 2(1+5)625 = 7500\ Hz.$$

Or,

$$B_T = 2(\Delta f + f_m) = 2(3125 + 625) = 7500\ Hz.$$

$$B_T = 2\Delta f\left(1+\frac{1}{\beta}\right) = 2 \times 3125 \times \left(1+\frac{1}{5}\right) = 7500\ Hz.$$

$$B_T = 2(\Delta f + f_m) = 2\Delta f\left(1+\frac{1}{\beta}\right)$$
$$= 2f_m(1+\beta)$$

Using Bessel function table,

We can see from the Bessel function table, for modulation index β =5, the number of significant sideband frequencies that satisfies |J(β)|>0.01, is $n_{max} = 8$.

So, Modulation index $\beta = 5$; $n_{max} = 8$;

So, Bandwidth, $B_T = 2n_{max} f_m = 2 \times 8 \times 625 = 10{,}000\ Hz = 10\ k$

We should keep this remember for Bessel function Table.

For, $\beta = 0.5$;	$n_{max} = 2$
For, $\beta = 1$;	$n_{max} = 3$
For, $\beta = 2$;	$n_{max} = 4$
For, $\beta = 3$;	$n_{max} = 6$
For, $\beta = 4$;	$n_{max} = 7$
For, $\beta = 5$;	$n_{max} = 8$
For, $\beta = 10$;	$n_{max} = 14$
For, $\beta = 20$;	$n_{max} = 25$

Using Universal curve, for $\beta = 5$; $we\ get \frac{B_T}{\Delta f} = 3.2$;

So, $B_T = 3.2 \times 3125 = 10{,}000\ Hz = 10\ kHz.$

Power dissipated across 10 Ω resistor = P

We know, $P = VI = I^2R = \frac{V^2}{R}$; where V=Voltage in r.m.s.

Now, $V_{rms} = \frac{V_m}{\sqrt{2}}$ $\qquad P = \frac{V^2}{R} = I^2R = VI$

So, Power, $P = \frac{V_{rms}^2}{R} = \frac{\left(\frac{V_m}{\sqrt{2}}\right)^2}{R} = \frac{V_m^2}{2R}$; $= \frac{V_m^2}{2R}$

In this case, Amplitude, $A_c = 12$;

So, Power dissipated, $P = \frac{A_c^2}{2R} = \frac{(12)^2}{2 \times 10} = 7.2\ W$ //

61	Consider an FM signal with: $\Delta f = 10\ kHz,\ \ f_m = 10\ kHz,\ A_c = 10\ V,\ \ f_c = 500\ kHz$ **Compute and draw the spectrum for FM signal.**	

Solution:

Modulation Index, $\beta = \frac{\Delta f}{f_m} = \frac{10}{10} = 1;$

Now, the spectrum of FM is defined as:

$$S(f) = \frac{A_c}{2}\sum_{n=-\infty}^{\infty} J_n(\beta)[\delta\{f-(f_c+nf_m)\} + \delta\{f+(f_c+nf_m)\}]$$

From Bessel function Table:

for $\beta = 1$; the coefficients are $J_0 = 0.77,\ J_1 = 0.44,\ J_2 = 0.11,\ J_3 = 0.02.$

The single side spectrum is shown below:

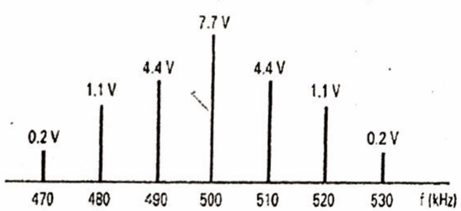


Fig: The single side spectrum for $\beta = 1$

> Here f_m=10 kHz and f_c=500 kHz.
>
> For, β =1, we can see from the Bessel table that both side of carrier frequency we find 3 sidebands. So the number of maximum sidebands n= 3. It is also shown in table of the number of significant sideband for β .
>
> For, $\beta = 1; \rightarrow n = 3;$
>
> For, $\beta = 2; \rightarrow n = 4$
>
> For, $\beta = 5; \rightarrow n = 8$

BESSEL TABLE

Modulation Index	Carrier J_0	J_1	J_2	J_3	J_4	J_5	J_6	J_7	J_8	J_9	J_{10}
		Sidebands									
0.0	1.00	—	—	—	—	—	—	—	—	—	—
0.25	0.98	0.12	—	—	—	—	—	—	—	—	—
0.5	0.94	0.24	0.03	—	—	—	—	—	—	—	—
1.0	0.77	0.44	0.11	0.02	—	—	—	—	—	—	—
1.5	0.51	0.56	0.23	0.06	0.01	—	—	—	—	—	—
2.0	0.22	0.58	0.35	0.13	0.03	—	—	—	—	—	—
2.5	-0.05	0.50	0.45	0.22	0.07	0.02	—	—	—	—	—
3.0	-0.26	0.34	0.49	0.31	0.13	0.04	0.01	—	—	—	—
4.0	-0.40	-0.07	0.36	0.43	0.28	0.13	0.05	0.02	—	—	—
5.0	-0.18	-0.33	0.05	0.36	0.39	0.26	0.13	0.06	0.02	—	—
6.0	0.15	-0.28	-0.24	0.11	0.36	0.36	0.25	0.13	0.06	0.02	—
7.0	0.30	0.00	-0.30	-0.17	0.16	0.35	0.34	0.23	0.13	0.06	0.02
8.0	0.17	0.23	-0.11	-0.29	0.10	0.19	0.34	0.32	0.22	0.13	0.06

Tabulated value for Bessel Function for the first kind of the n[th] order

> Keep Remember:
>
> For β = 1; We get
>
> J_0=0.77
>
> J_1=0.44
>
> J_2=0.11
>
> J_3=0.02

61	Draw the frequency spectrum of FM signal for $\beta = 2$ and $f_m = 20\ kHz$ and $f_c = 800\ kHz$	

Solution:

For β = 1

Now, the spectrum of FM is defined as:

$$S(f) = \frac{A_c}{2}\sum_{n=-\infty}^{\infty} J_n(\beta)[\delta\{f-(f_c+nf_m)\} + \delta\{f+(f_c+nf_m)\}]$$

For β =2; We get

$J_0 = 0.22;\ J_1 = 0.58;\ J_2 = 0.35;\ J_3 = 0.13;\ J_4 = 0.03$

> Keep Remember:
>
> For β =2; We get
>
> J_0 = 0.22
>
> J_1 = 0.58
>
> J_2 = 0.35
>
> J_3 = 0.13
>
> J_4 = 0.03

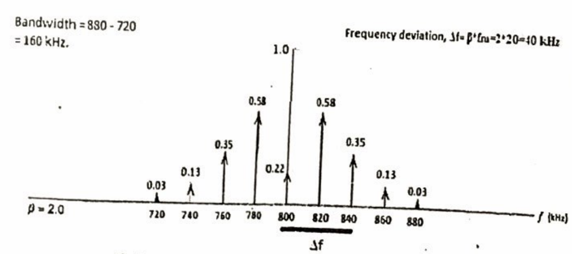


Fig: Frequency spectrum of FM signal of modulation Index = 2

63	**Draw the frequency spectrums of FM signals of modulation index of 1 and modulation index of 2.**

Solution:

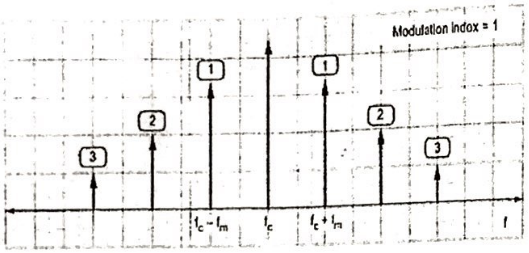


Fig: Effect of modulation index in significant number of sidebands

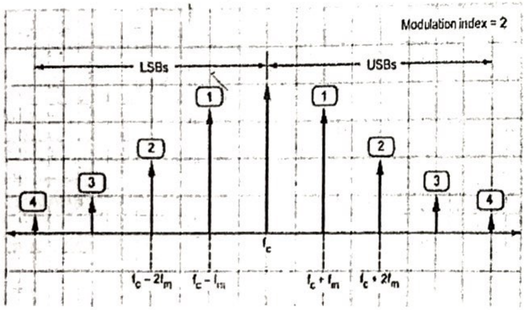


Fig: Effect of modulation index in significant number of sidebands

64	**Calculate the bandwidth occupied by a FM signal with a modulation index of 2 and a highest modulating frequency of 2.5 kHz. Determine the bandwidth with table of Bessel functions.**

Solution:

From the table, we get, for the modulation index of 2, there are 4 significant side band frequencies for FM.

So, bandwidth, $B_T = 2n_{max} f_m = 2 \times 4 \times 2.5 = 20$ kHz.

$B_T = 2n_{max} \cdot f_m$

65	**For an FM modulator with a modulation index β =1, a modulating signal $V_m(t) = V_m \sin(2\pi 1000t)$ and un-modulated carrier**

$V_m(t) = V_m \cdot \sin \cdot (2\pi\ 1000t)$

	$V_c(t) = 10\sin(2\pi 500kt)$, determine **(a) Number of sets of significant sideband** **(b) The amplitude** **(c) Then draw the frequency showing their relative amplitudes**

Solution:

Message signal frequency, $f_m = 1000\ Hz = 1kHz$

Carrier frequency, $f_c = 500\ kHz$.

(a) From the Bessel table, for β = 2, the number of maximum significant sideband $n_{max} = 3$.
(b) And also we find from the amplitude of all sideband including carrier. They are:

$J_0 = 0.77,\ J_1 = 0.44,\ J_2 = 0.11,\ J_3 = 0.02.$

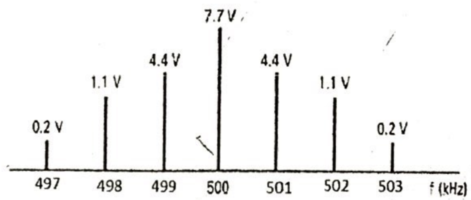


Fig: The single side spectrum for β = 1

66	**For an FM modulator with a peak frequency deviation Δ f = 10 kHz, a modulating signal frequency $f_m = 10\ kHz$. V_c = 10 V and 500 kHz carrier, determine:** **(a) Actual minimum bandwidth from the Bessel function table** **(b) Approximate minimum bandwidth using Carson's Rule** **(c) Plot the output frequency spectrum for the Bessel approximation.**

Solution:

(a) Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{10}{10} = 1$

$\beta = 1,\ n_{max} = 3$

From Bessel function table, for modulation index of β = 1, the maximum number of sideband frequencies are n_{max}=3.

So, Bandwidth, $B_T = 2n_{max} f_m = 2 \times 3 \times 10 = 60\ kHz$.

(b) Using Carson's Rule, $Bandwidth,\ B_T = 2\Delta f + 2f_m = (2 \times 10) + (2 \times 10) = 40\ kHz.$
(c) Frequency spectrum for β =1 is given below:

β=1 J= 0.77, 0.44, 0.11, .02

β=2 J= 0.22, 0.58, 0.35, 0.13, 0.03

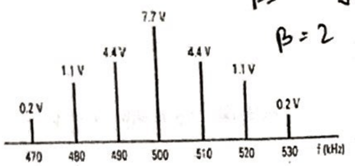


Fig: The single side spectrum for β = 1

67	A carrier wave of amplitude 5V and frequency 90 MHz is modulated by a sinusoidal voltage of amplitude 5V and frequency 15 kHz. The frequency deviation constant is 1 kHz/V. Sketch the spectrum of the modulated FM wave.	

Solution:

Frequency deviation constant

Amplitude of modulating voltage is 5V and frequency deviation constant is 1 kHz/ v. Actually frequency deviation constant is also called sensitivity of the modulator.

= Sensitivity of the modulator.

Hence the Frequency deviation, $\Delta f = k_f A_m = 1 \times 5 = 5\ kHz$.

Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{5}{15} = 0.333$

$\Delta f = k_f \cdot A_m = 1 \times 5 = 5\ kHz$.

For, β =0.333, from the table of Bessel function, we find the value of coefficient J_0, J_1 and J_2.

For, carrier: $J_0 = 0.96$

First side frequency: $J_1 = 0.18$

Second side frequency: $J_2 = 0.02$

These are the values for un-modulated carrier of 1V.

Now, given amplitude of the carrier is 5 V.

Hence: $J_0 = 0.96 \times 5 = 4.8V$

$J_1 = 0.18 \times 5 = 0.9V$

$J_2 = 0.02 \times 5 = 0.1\ V$

The frequency spectrum is shown below:

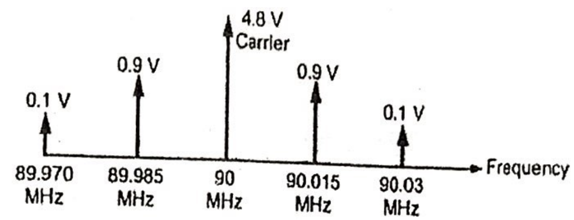


Fig: Frequency spectrum of FM for modulation index of 0.33

68	What is percent Modulation?	

The term "percent modulation" as it is used in reference to FM. It is the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

The band between 88 MHz and 108 MHz has been allocated for commercial FM broadcast. The maximum frequency deviation is 75 kHz for FM broadcast station in this band.

The sound accompanying the picture in TV broadcast is transmitted using FM technique. For this sound transmission Maximum frequency deviation is allowed 25 kHz.

Thus 100% modulation corresponds to 75 kHz for the commercial FM broadcast and 25 kHz for TV sound broadcast.

So, percent modulation, $M = \frac{\Delta f\ actual}{\Delta f\ maximum} \times 100\%$

$M = \frac{\Delta f\ actual}{\Delta f\ maximum}$

69	Determine the percent modulation of an FM wave with a frequency deviation of 15 kHz for (a) FM broadcast , (b) for TV broadcast.	

a) $M = \frac{15}{75} \times 100 = 20\%$

b) $M = \frac{15}{25} \times 100 = 60\%$

70	Determine the frequency deviation and carrier swing required to provide 80% modulation in the FM broadcast band. Repeat this for an FM signal serving as the audio portion of a TV broadcast.	

Solution:

a) FM broadcast band:

$$M = \frac{Actual\ frequency\ deviation}{Max\ frequency\ deviation} \times 100\%$$

carrier swing = 2 × frequency deviation.

$$\therefore 80 = \frac{Actual\ frequency\ deviation, \Delta f}{75\ kHz} \times 100\%$$

$$\therefore \Delta f = 60\ kHz$$

Now, carrier swing $= 2 \times frequency\ deviation = 2 \times 60 = 120\ kHz$

b) Sound in TV broadcast:

$$M = \frac{actual\ frequency\ deviation}{Maximum\ frequency\ deviation} \times 100\%$$

$$\therefore 80 = \frac{actual\ frequency\ deviaiton, \Delta f}{25\ kHz} \times 100\%$$

$$\therefore \Delta f = 20\ kHz$$

Now, carrier swing, $= 2 \times frequency\ deviation = 2 \times 20 = 40\ kHz.$

71	**In an FM system, when the audio frequency is 500 Hz and modulating voltage 2.5V, the deviation produced is 5 kHz. If the modulating voltage is now increased to 7.5V, calculate the new value of frequency deviation produced. If the AF voltage is raised to 10 V while the modulating frequency dropped to 250 Hz, what is the frequency deviation? Calculate the modulation index in each case.**	

Frequency deviation constant, R is sometimes referred as sensitivity of the modulator, k_f. So, don't be confused with that!!!

F^{ey} deviation constant $R = K_f$

We know frequency deviation, $\Delta f = k_f A_m$ or $\Delta f = RA_m$

So, frequency deviation constant, $R = \frac{\Delta f}{A_m} = \frac{5\ kHz}{2.5\ V} = 2\ kHz/V$

The modulating voltage is now 7.5V.

$\Delta f = K_f \cdot A_m$
$= 2 \times 7.5$
$= 15\ kHz.$

$\therefore$ Frequency deviation produced, $\Delta f = k_f A_m = 2 \times 7.5 = 15\ kHz$

Similarly, when modulating voltage is 10V, then

Frequency deviation, $\Delta f = 2 \times 10 = 20\ kHz$

Now, for modulating voltage 2.5 volt,

Modulation index, $\beta = \frac{frequency\ deviation}{modulation\ frequency} = \frac{\Delta f}{f_m} = \frac{5\ kHz}{500\ Hz} = 10$

For modulating voltage 7.5 V:

Modulation index, $\beta = \frac{15\ kHz}{500\ Hz} = 30$

For modulating voltage 10V:

Modulation index, $\beta = \frac{20\ kHz}{250\ Hz} = 80$

$\beta = \frac{\Delta f}{f_m} = 80.$

72	**A 93.2 MHz carrier is frequency modulated by a 5 kHz sine wave. The resultant FM signal has a frequency deviation of 40 kHz.** **a) Find the carrier swing of the FM signal.** **b) What are the highest and lowest frequencies attained by the frequency modulated signal.** **c) Calculate the modulation index for the wave.**	

Solution:

a) $carrier\ swing = 2 \times frequency\ deviation$ $= 2 \times f^{ey}$ deviation

$= 2 \times 40\ kHz = 80\ kHz$

b) Highest frequency reached = carrier frequency + frequency deviation

=93.2 MHz + 40 kHz = 93.24 MHz.

The lowest frequency reached = carrier frequency − frequency deviation = carrier f^{ey} − f^{ey} deviation

=93.2 MHz − 40 kHz = 93.16 MHz.

c) Modulation index $\beta = \frac{frequency\ deviation}{modulating\ frequency} = \frac{\Delta f}{f_m} = \frac{40}{5} = 8\ kHz$ $= \frac{\Delta f}{f_m} = \frac{40}{5} = 8\ kHz$

73	**When a 50.4 MHz carrier is frequency modulated by a sinusoidal AF modulating signal, the highest frequency reached is 50.405 MHz. Calculate:** **a) The frequency deviation produced,** **b) Carrier swing of the wave** **c) Lowest frequency reached.**	

Solution:

a) Frequency deviation = [Highest frequency reached] − [carrier frequency]

= 50.405 MHz − 50.4 MHz = 5 kHz

Δf = highest f^{ey} − carrier f^{ey}

b) Carrier swing = 2 × frequency deviation = 2 × 5 kHz = 10 kHz.

$2 \times 5\ kHz = 10\ kHz.$

c) Lowest frequency attained = [carrier frequency] − [frequency deviation]

=50.4 MHz − 5 kHz = 50.395 MHz. //

74	**The carrier swing of a frequency modulated signal is 70 kHz and the modulating signal is a 7 kHz sine wave. Determine the modulation index of the FM signal.**	

Solution:

Carrier swing = 2 × frequency deviation

Or, 70 kHz = 2 × frequency deviation

Therefore, frequency deviation, $= \frac{70\ kHz}{2} = 35\ kHz.$

Modulation index, $\beta = \frac{frequency\ \ deviation}{Modulating\ \ frequency} = \frac{35}{7} = 5$

$\beta = \frac{f^{cy}\ deviation}{modulating\ f^{cy}} = \frac{35}{7} = 5$

75	**Calculate the carrier swing, carrier frequency, frequency deviation, and modulation index for an FM signal which reaches a maximum frequency of 99.047 MHz and a minimum frequency of 99.023 MHz. The frequency of the modulating signal is 7 kHz.**	

Carrier swing = [Highest frequency reached] – [Lowest frequency reached]

= 99.047 MHz – 99.023 MHz = 24 kHz

Frequency deviation $= \frac{1}{2}\ [carrier\ swing] = \frac{1}{2} \times 24 = 12\ kHz$

Now, Frequency deviation = [Highest frequency reached] – [carrier frequency]

So, carrier frequency = [Highest frequency reached] – [Frequency deviation]

= 99.047 MHz – 12 kHz = 99.035 MHz

$f_{carrier} = f^{cy}_{highest} - \Delta f$

Or,

Carrier frequency = [frequency deviation] + [lowest frequency reached]

= 12 kHz + 99.023 MHz = 99.035 MHz

Modulation Index, $\beta = \frac{frequency\ \ deviation}{modulating\ \ frequency} = \frac{\Delta f}{f_m} = \frac{12\ kHz}{7\ kHz} = 1.7$

$\frac{f^{cy}\ deviation}{modulating\ f^{cy}}$

76	**Consider a Commercial FM system. Each station is allotted to a frequency deviation of $\pm 75\ kHz$ ($150\ kHz$ *carrier swing*) and 25 kHz of guard band added above and below the carrier frequency swing. Determine the number of maximum number of stations can be made available.**	

Solution:

Commercial FM band (108 − 88) MHz

We know, Commercial FM band, 88 MHz – 108 MHz.

So, total Bandwidth = 108 MHz – 88 MHz = 20 MHz.

BW = 108 − 88

Now, the bandwidth of each channel $= (2 \times 75) + (2 \times 25) = 200\ kHz$

So, the maximum number of stations can be made available $= \frac{20\ MHz}{200\ kHz} = 100$ (ans)

Note: This is the standard format of FM modulation band.//

77	**FM mono radio employs a peak frequency deviation of 75 kHz and a sinusoidal audio message signal bandwidth is 15 kHz. What is the bandwidth of FM?**	DPDC. ~~PGCB~~ **2016**

Bandwidth of FM $B_W = 2(\Delta f + f_m) = 2(75 + 15) = 180\ kHz$

$BW = 2(\Delta f + f_m)$

180 kHz

Digital Communication

	Important basics	

Digital Baseband Transmission:

Baseband transmission: When the signal is transmitted over the channel, without any modulation, it is called baseband transmission.

One of the major problem occurred in baseband transmission is intersymbol interference. This interference takes place due to dispersive nature of the channel.

Nyquist criterion gives a condition for distortionless baseband transmission. It is possible to reduce the effect of intersymbol interference with the help of raised cosine spectrum. The source coding techniques such as PCM, DM, ADM, DPCM are used to encode analog signal for digital baseband transmission.

78	**What are the different types of modulation?**	

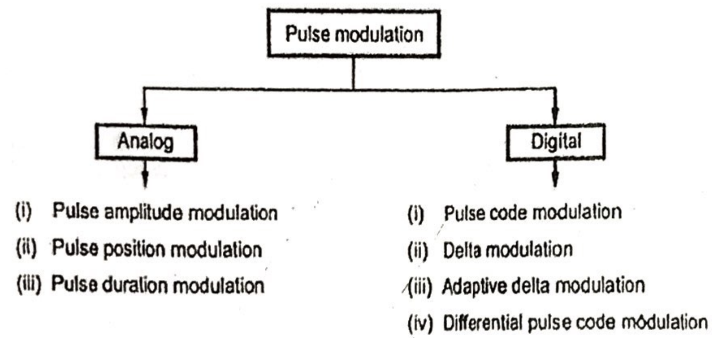
There are three types of modulation.

i) Amplitude modulation
ii) Angle modulation
iii) Pulse modulation.

Pulse modulation can be further classified as:

i) Pulse analog modulation
ii) Pulse digital modulation

The above two techniques can be further classified as,



79	**Why digital communication is so popular?** **What are the advantages and disadvantage of digital communication?** **Why digital communication is more preferable to analog communication system?**	24th 31st, 33rd & 34th BCS

Advantages are:

1. Because of the advances in digital IC technologies and high speed computers, digital communication systems are simpler and cheaper compare to analog systems.
2. Regenerative repeaters can be used at fixed distance along the link, to identify and regenerate a pulse before it is degraded to an ambiguous state.
3. Using multiplexing, the speech, video and other data can be merged and transmitted over common channel.
4. Since the transmitted signal is digital, a large amount of noise interference can be tolerated.
5. Since channel coding is used, the error can be detected and corrected in receivers.
6. Transmitted data is more secure than analog communication.
7. Digital communication is adaptive to other advanced branches of data processing such as digital signal processing, image processing and data compression etc.

Disadvantages: Even though digital communication offer many advantages as given above, it has some drawback also. But the advantages of digital communication outweigh disadvantages. The disadvantages are:

1. Because of analog to digital conversion, the data rate becomes high. Hence more transmission bandwidth is required for digital communication.
2. Digital communication needs synchronization.

80	**What are the basic elements of digital communication? What is hybrid communication?**	34th BCS

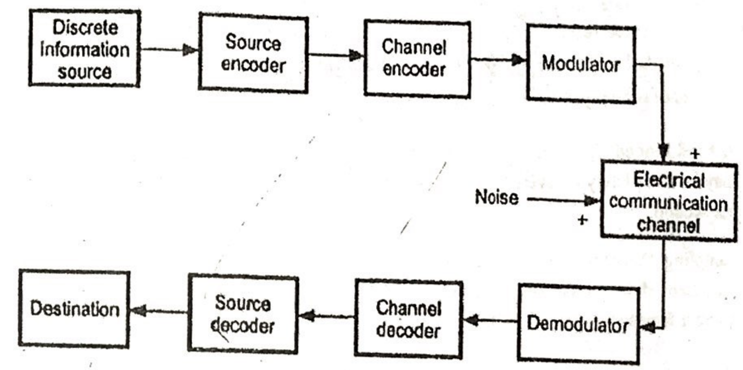


Fig. Basic digital communication system

Source Encoder / Decoder: The source encoder converts the input into a binary sequence of 0's and 1's by assigning code words to the symbols in the input sequence. On the other hand, the source decoder converts the binary output of the channel decoder into a symbol sequence.

Channel Encoder / Decoder: The channel coding operation consists of systematically adding extra bits to the output of the source coder. These extra bits do not convey any information but helps the receiver to detect and / or correct some of the errors. On the other hand, the channel decoder recovers the information bearing bits from the coded binary stream.

Modulator: The modulator converts the input bit stream into an electrical waveform suitable for transmission over the communication channel. In other word, Modulation is the process of placing the message signal over some carrier to make it suitable for transmission over long distance.

Demodulator: The extraction of the message from the information bearing waveform produced by the modulation is accomplished by the demodulator. In other word, Demodulation is the process of separating message signal from the modulated carrier signal.

Channel: The channel provides the electrical connection between the source and destination. The different channels are: Pair of wires, Co-axial cables, Optical fibre, Radio channel, Satellite channel or combination of any of these. The communication channels have only finite bandwidth. The important parameters of the channel are Signal to Noise power ratio (SNR).

Hybrid communication system: Hybrid communication system is a communication system that can accommodate both digital and analog signals.

81	**State sampling theorem.**	

Sampling theorem states that, a bandlimited signal of finite energy, which has no frequency components higher than W Hz, completely described by specifying the values of the signal at instants of time separated by $\frac{1}{2W}$ seconds and,

A band-limited signal of finite energy which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

Sampling theorem can also be stated as : A continuous time signal **can be completely represented in its samples** and **recovered back if the sampling frequency is twice of the highest frequency content of the signal**, i.e.,

$f_s \geq 2W$

Here, f_s is the sampling frequency and W is the higher frequency content.

82	**What is meant by aliasing effect?** **Or, Justify that Appropriate reconstruction of a signal from its sampled version requires the sampling frequency should be not less than twice the base band signal frequency.**	24th 31st & 34th BCS

Aliasing effect takes place when sampling frequency is less than nyquist rate. Under such condition, the spectrum of the sampled signal overlaps with itself. Hence higher frequencies take the form of lower frequencies. This interference of the frequency component is called aliasing effect.

When the high frequency interferes with low frequency and appears as low frequency, then the phenomenon is called aliasing.

Effect of aliasing: (a) since high and low frequencies interfere with each other, distortion is generated. (b) Data is lost and it cannot be recovered.

Different ways to avoid aliasing:

Aliasing can be avoided by two methods: (a) Sampling rate $f_s \geq 2W$ and (b) Strictly bandlimit the signal to 'W'

(a) Sampling rate $f_s \geq 2W$

When the sampling rate is made higher than 2W, then the spectrums will not overlap and there will be sufficient gap between the individual spectrums. This is shown in figure below:

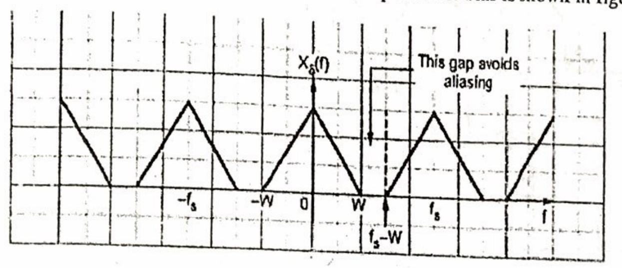
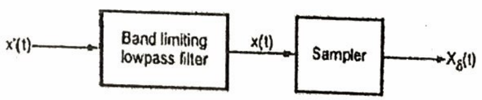


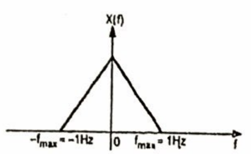
Fig. $f_s \geq 2W$ avoids aliasing by creating a bandgap

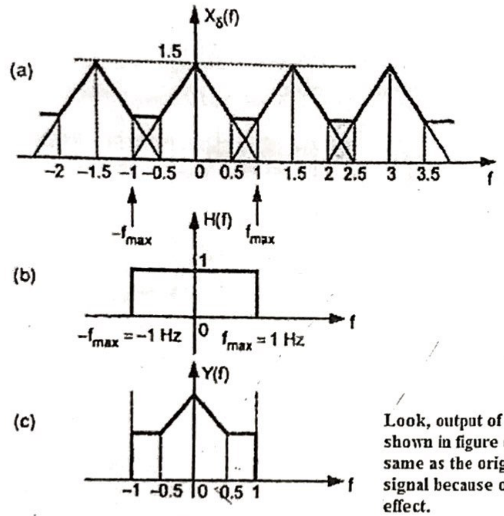
(b) Bandlimiting the signal

The sampling rate is $f_s = 2W$. Ideally speaking there should be no aliasing. But there can be few components higher than 2W. These components create aliasing. Hence a low pass filter is used before sampling the signal. Thus the output of low pass filter is strictly bandlimited and there are no frequency components higher than 'W'. Then there will be no aliasing.



Bandlimiting the signal.

83	**Figure shows the spectrum of a message signal. The signal is sampled at the rate of $f_s = 1.5f_{max}$, where $f_{max} = 1Hz$, is the maximum frequency. Sketch the spectrum of the sampled version of the signal. If the sampled signal is passed through an ideal low pass filter of bandwidth f_{max}, sketch the spectrum of the output signal from this filter.**  Fig. **Spectrum of message signal $x(f)$**	



Look, output of the filter shown in figure (c) is not same as the original input signal because of aliasing effect.

(a) Spectrum of the sampled signal at $f_s = 1.5 f_{max}$
(b) Response of ideal low-pass filter with $|H(f)| = 1$ for $-f_{max} < f < f_{max}$
(c) Spectrum of signal at the output of the lowpass filter

84	**Define Nyquist rate.**	

Let the signal be bandlimited to 'W' Hz. Then Nyquist rate is given as,

Nyquist rate = 2W samples /sec

Aliasing will not take place if sampling rate is greater than Nyquist rate.

Nyquist rate: When the sampling rate becomes exactly equal to '2W' samples /sec, for a given bandwidth of W Hz, then it is called Nyquist rate.

Nyquist interval: It is the time interval between any two adjacent samples when sampling rate is Nyquist rate.

Nyquist rate = 2W Hz $= 2W$ Hz

Nyquist interval $= \frac{1}{2W}$ *seconds* $= \frac{1}{2W}$ Seconds.

85	**Find the Nyquist rate and Nyquist interval for following signals.** a. $m(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$ b. $m(t) = \frac{\sin 500\pi t}{\pi t}$	

a. $m(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$

$$= 1/2\pi\left\{\frac{1}{2}\cos(4000\pi t - 1000\pi t) + \cos(4000\pi t + 1000\pi t)\right\}$$

$$= \frac{1}{4\pi}[\cos 3000\pi t + \cos 5000\pi t]$$

$$\frac{1}{4\pi}[\cos(2\pi \times 1500t) + \cos(2\pi \times 2500t)]$$

For, $\cos 3000\pi t$, we get $f_1 = 1500\ Hz$

For, $\cos 5000\pi t$, we get $f_2 = 2500\ Hz$

Here highest frequency, $W = f_2 = 2500\ Hz$

$\therefore$ *Nyquist rate* $= 2W = 2 \times 2500 = 5000\ Hz$

$\therefore$ *Nyquist interval* $= \frac{1}{2W} = \frac{1}{2\times 5000} = 0.2\ ms$

b. $m(t) = \frac{\sin 500\pi t}{\pi t} = \frac{\sin 2\pi 250 t}{\pi t}$

So, we get, $f = 250\ Hz$

Nyquist rate $= 2W = 2 \times 250 = 500\ Hz$

Nyquist interval $= \frac{1}{2W} = \frac{1}{2\times 250} = 2\ ms$

86	**List and describe briefly steps involved in PCM for signal processing.**	33rd BCS

PCM (Pulse Code Modulation) is an important method of analog to digital conversion. In this modulation, the analog signal is converted into an electrical waveform of two or more levels.

The PCM system block diagram is shown in figure. The essential operation in the transmitter of a PCM system are sampling, Quantization and coding. The Quantization and coding operation are usually performed by the same circuits, normally referred to as analog to digital converter (ADC). The essential operations in the receiver are regeneration, decoding and demodulation of quantized samples. Regenerative repeaters are usually used to reconstruct the transmitted sequence of coded pulses in order to combat the accumulated effects of signal distortion and noise.

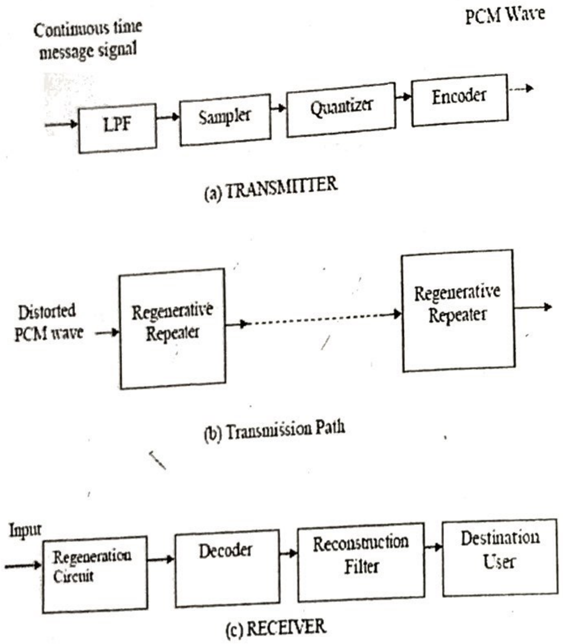


Fig: . PCM System: Basic Block Diagram

Basic Blocks:

1. Anti aliasing Filter
2. Sampler
3. Quantizer
4. Encoder

An anti-aliasing filter is basically a filter used to ensure that the input signal to sampler is free from the unwanted frequency components. For most of the applications, these are low pass filters. It removes the frequency components of the signal which are above the cutoff frequency

of the filter. The cut off frequency of the filter is chosen such it is very close to the highest frequency component of the signal.

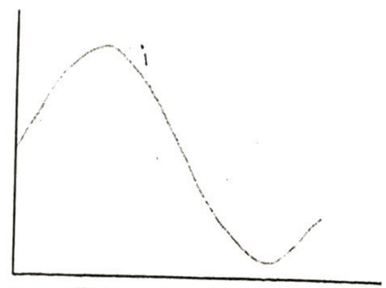
Sampler units samples the input signal and these samples are then fed to the Quantizer which outputs the quantized values for each of the samples. The Quantizer output is fed to an encoder which generates the binary code for every sample. The Quantizer and encoder together is called as analog to digital converter (ADC).

To recover the signal from the sampled data, a 'demodulator' is used. Demodulator apply the procedure of modulation in reverse. Demodulator tries to reconstruct the signal using regenerative repeaters. Because of aliasing, the signal has a significant amount of high frequency content. To remove these undesirable frequencies and get the original signal, the demodulator passes the signal through analog filter which removes the high frequency content and the original signal can be found.

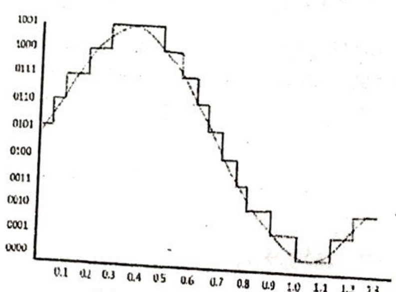
87	**What is quantization? How an analog signal is quantized? What are the types of quantization?**		**33rd BCS Similar**

The digitizing of analog signals involves the rounding off the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as Quantization.

The following figure represents an analog signal. The signal to get converted into digital has to undergo sampling and quantization.



Before Quantization



After Quantization

Quantization is representing the sampled values of the amplitude by a finite set quantization level. Both sampling and quantization result in the loss of information. The quality of a

Quantizer output depends upon the number of quantization levels used. The spacing between two adjacent representation levels is called a step size.

Types of quantization:

There are two types of quantization: a) Uniform Quantization and b) Non uniform quantization.

The type of quantization in which the quantization levels are uniformly spaced in termed as uniform quantization. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non uniform quantization.

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

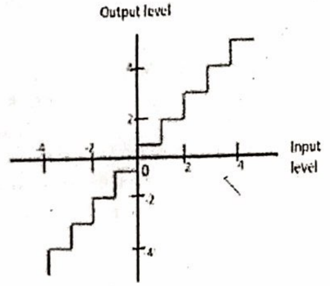


Fig 1 : Mid-Rise type Uniform Quantization

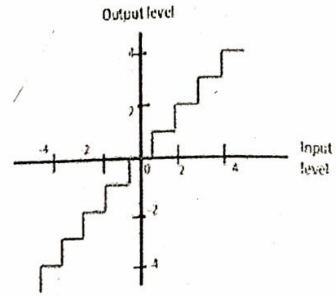


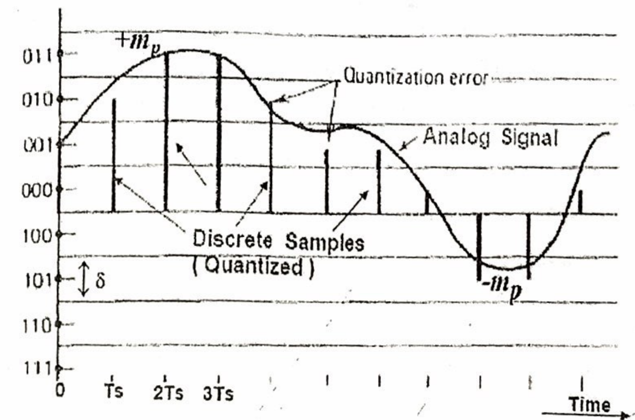
Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid rise type and figure 2 shows the mid-tread type of uniform quantization.

The Mid-Rise type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.

The Mid-Tread type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.

Some important parameter to understand PCM for MATH:



(3 bit quantizer)

Fig... Typical Quantization process.

Number of bit of quantizer: //

n = number of bits/sample used in quantizer.

n = *number of bit per sample used in quantizer*

$n = \log_2 L$ = where L is the number of the level of the quantizer.

$n = \log_2 L = L =$ num

In the above figure, it should each level is expressed by 3 bits. So n=3 here.

Quantization Level:

L = number of quantization level = 2^n

$L = 2^n$ = number of quantization level 2^n

Step size:

Step size, $\delta = \frac{total\ amplitude\ range}{number\ of\ level}$

$\delta = \frac{\text{total amplitude range}}{\text{number of level.}}$

Now, total amplitude range = Max amplitude – Min amplitude

$= +m_p - (-m_p)$

$= 2m_p$

$2m_p$ = total amplitude range.

If the number of quantization level, = L

Then, step size, $\delta = \frac{2m_p}{L}$

$\delta = \frac{2mp}{L}$

Quantization Noise or quantization error in PCM:

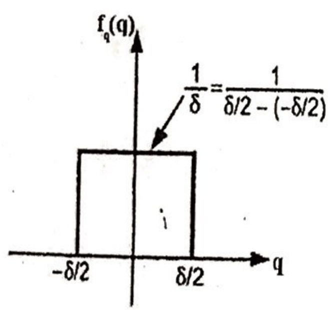
Quantization noise is produced in the transmitter end of a PCM system by rounding off sample values of an analog baseband signal to the nearest permissible representation levels of the quantizer. If δ be the step size of a quantizer, then maximum error can occur is $\pm\frac{\delta}{2}$. It can be proven that maximum quantization error, $\frac{\delta}{2} = \frac{m_p}{L}$

Probability density function (PDF) of quantization error:

If step size δ is sufficiently small, then it is reasonable to assume that the quantization error 'q' will be uniformly distributed random variable. We know that the maximum quantization error in PCM is given by: $q_{max} = \left|\frac{\delta}{2}\right|$

$$q_{max} = \pm\frac{\delta}{2}$$

Thus over the interval $\left(-\frac{\delta}{2}, \frac{\delta}{2}\right)$ quantization error is uniformly distributed random variable.



Uniform distribution for quantization error

$$f_q(q) = \begin{cases} 0 & fo\ q \le \frac{\delta}{2} \\ \frac{1}{\delta} & for\ -\frac{\delta}{2} \le q \le \frac{\delta}{2} \\ 0 & for\ q > \frac{\delta}{2} \end{cases}$$

Quantization Noise Power for PCM:

If V_{noise} = the noise voltage, then noise power, $= \frac{V_{noise}^2}{R}$

Here, V_{noise}^2 is the mean square value of noise voltage. Since noise is defined by random variable 'q' and PDF (probability density function) $f_q(q)$, its mean square value is given as $\overline{q^2}$.

We know mean square value of a random variable X is given as,

$$\bar{x}^2 = \int_{-\infty}^{\infty} x^2 f_x(x)dx$$

So, here, $\bar{q}^2 = \int_{-\infty}^{\infty} q^2 f_q(q)dq$

$$= \int_{-\frac{\delta}{2}}^{\frac{\delta}{2}} q^2 \times \frac{1}{\delta} dq = \frac{\delta^2}{12}$$

So, the mean square value of noise voltage is $V_{noise} = \frac{\delta^2}{12}$

Quantization noise power $= \frac{V_{noise}^2}{R}$

If R = 1 ohm, Then, Quantization noise power $= \frac{V_{noise}^2}{1} = \frac{\delta^2}{12}$

88	**Derive the expression for the signal to quantization noise ratio for PCM system that employs linear quantization technique. Assume that input to the PCM system is a sinusoidal signal.** **Or,** **A PCM system uses a uniform quantizer by an 'n' bit encoder. Show that rms signal to quantization noise ratio is approximately given by (1.76 + 6n) dB.**

Assume that the modulating signal be a sinusoidal voltage, having peak amplitude A_m.

The power of the signal, $P_{signal} = \frac{V_{rms}^2}{R} = \frac{\left(\frac{A_m}{\sqrt{2}}\right)^2}{R} = \frac{A_m^2}{2}$; assuming R=1 ohm

Quantization noise, $N_q = \frac{\delta^2}{12}$ $= \frac{\delta^2}{12}$

P signal $= \frac{A_m^2}{2}$

Again, we know, step size, $\delta = \frac{2m_p}{L}$

Where $m_p = maximum\ peak\ voltage$; Here, $m_p = A_m$

$2m_p = maximum\ amplitude\ range = peak\ to\ peak\ voltage$

$L = number\ of\ the\ level\ of\ the\ quantizer$

∴ Quantization noise, $N_q = \frac{\delta^2}{12} = \frac{\left(\frac{2m_p}{L}\right)^2}{12} = \left(\frac{4m_p^2}{12L^2}\right) = \frac{m_p^2}{3L^2} = \frac{A_m^2}{3L^2}$

Signal to quantization noise ratio, $\frac{S}{N} = \frac{\frac{A_m^2}{2}}{\frac{A_m^2}{3L^2}} = \frac{A_m^2}{2} \times \frac{3L^2}{A_m^2} = \frac{3}{2}L^2 = 1.5L^2$

$$SNR(dB) = 10\log_{10}\frac{S}{N} = 10\log_{10}(1.5L^2) = 10\log_{10}1.5 + 10\log_{10}L^2$$

$$= 1.76 + 20\log_{10}L$$

$$= 1.76 + 20\log_{10}2^n$$

$$= 1.76 + 20n\log_{10}2$$

$$= 1.76 + 6.02n$$

Note: if signal is not sinusoidal, then

We will normalize the value assuming that $m_p = 1$ and $P_{signal} \leq 1$

$$\frac{S}{N} = \frac{P_{signal}}{P_{noise}} = \frac{1}{\frac{m_p^2}{3L^2}} = 3L^2 = 3(2^n)^2 = 3.2^{2n}$$

$$SNR(dB) = 10\log_{10}\left(\frac{S}{N}\right) = 10\log_{10}(3.2^{2n}) = 4.8 + 6n$$

Number of bit, n ছাড়া যদি আর কিছু দেওয়া না থাকে, আর SNR বের করতে বলে তবে এই সূত্র ব্যবহার করতে হবে।

89	**A television signal with a bandwidth of 4.2 MHz is transmitted using binary PCM. The number of quantization levels is 512.** **Calculate,** **a. Code word length** **b. Transmission bandwidth** **c. Final bit rate** **d. Output signal to quantization noise ratio** PCM	

Code word length means the number of bits required per sample. That is the number of bit of the quantizer.

bit required/sample.

Given that quantization level, $L = 512$

We know, number of bit of the quantizer, $n = \log_2 L = \log_2 512 = \frac{\log_{10}512}{\log_{10}2} = 9$

To obtain transmission bandwidth: $n = \log_2 L$

We know for PCM

Transmission bandwidth, $B_T \geq \frac{R_b}{2}$; or, $B_T \geq \frac{nf_s}{2}$; or, $B_T \geq nW$

$B_T \geq nW \geq \frac{nf_s}{2}; \geq \frac{R_b}{2}$

Where W = bandwidth of the signal.

Given that W=4.2 MHz = 4.2×10^6 MHz

So, transmission bandwidth, $B_T \geq nW$ $B_T \geq nW$

$$\geq 9 \times 4.2 \times 10^{\wedge}6$$

$$\therefore B_T \geq 37.8\ MHz$$

So, minimum transmission bandwidth will be 37.8 MHz.

To obtain final bit rate:

The final bit rate will equal to signaling rate. We know for PCM

* bit rate = signaling rate

Signaling rate, $R_b = nf_s$ $R_b = nf_s$

Here, $f_s = sampling\ frequency$

From sampling theorem, we know, $f_s \geq 2W$ $f_s \geq 2W$

$$\therefore f_s \geq 2 \times 4.2\ MHz$$ $f_s \geq 8.4$ MHz

$$\therefore f_s \geq 8.4\ MHz.$$

So, the signaling rate, $R_b = nf_s = 9 \times 8.4\ MHz = 75.6\ mbps$

Note: We can also determine, transmission bandwidth from bit rate using, $B_T = \frac{R_b}{2}$ $B_T = \frac{R_b}{2}$

To obtain output signal to noise ratio:

SNR (dB) = 4.8 + 6n

SNR in dB $SNR(dB) = 4.8 + 6n = 4.8 + 6 \times 9 = 58.8\ dB$.

90	**The bandwidth of signal input to the PCM is restricted to 4 kHz. The input varies from -3.8 V to +3.8 V and has the average power of 30 mW. The required signal to noise ratio is 20 dB. The modulator produces binary output. Assume uniform quantization.** **a. Calculate the number of bits required per sample** **b. Output of 30 such PCM coders are time multiplexed. What is the maximum required bandwidth for the multiplexed signal?** **c. What would be the signaling rate?**	-3.8 to +3.8 V

Given that, SNR(dB) = 20 dB

$SNR\,(dB) = 10\log_{10}\left(\frac{S}{N}\right)$

Or, $20 = 10\log_{10}\left(\frac{S}{N}\right)$

Or, $\log_{10}\left(\frac{S}{N}\right) = \frac{20}{10} = 2$

Or, $\frac{S}{N} = 10^2 = 100$

We know, $SNR = \frac{S}{N} = \frac{P_{signal}}{P_{noise}} = \frac{P_{signal}}{\frac{\delta^2}{12}} = \frac{P_{signal}}{\left(\frac{2m_p}{L}\right)^2 \times \frac{1}{12}} = \frac{P_{signal}}{\frac{m_p^2}{3L^2}} = \frac{3L^2}{m_p^2} \times P_{signal}$

$\frac{S}{N} = \frac{3L^2}{m_p^2} \times P_{signal}$

$\therefore L^2 = \frac{\frac{S}{N} \times m_p^2}{3 \times P_{signal}} = \frac{100 \times 3.8^2}{3 \times 30 \times 10^{-3}} = 16044.44$

$\therefore L = \sqrt{16044.44} = 127$

Now, $n = \log_2 L = \frac{\log_{10} 127}{\log_{10} 2} \approx 7$; [No. of bit must be an integer] //

To obtain maximum bandwidth:

We know, transmission bandwidth of PCM, $B_T \geq nW$; where W is the highest frequency of the signal.

Here given, W = 4 kHz

Since there are 30 PCM coders which are time multiplexed, the transmission bandwidth will be

$B_T = 30 \times nW = 30 \times 7 \times 4\,kW = 840\,kHz$

Again we know, transmission bandwidth is half of the signaling rate i.e. $B_T = \frac{R_b}{2}$

So, signaling rate is two times the transmission bandwidth,

$R_b = 2 \times B_T = 2 \times 840 = 1680\,kbps$

SNR (dB) = 4.8+6n

91	**The information in analog signal voltage waveform is to be transmitted over a PCM system with an accuracy of $\pm 0.1\%$ (full scale). The analog voltage waveform has a bandwidth of 100 Hz and an amplitude range of -10 to +10 Volts.** **a. Determine the maximum sampling rate required.** **b. Determine the number of bits in each PCM word** **c. Determine minimum bit rate required in the PCM signal.** **d. Determine the minimum absolute channel bandwidth required for the transmission of the PCM signal.**	

a. Here signal Bandwidth W = 100 Hz. i.e. the maximum frequency of the signal.

So, By sampling theorem , minimum sampling frequency $f_s \geq 2W$

$$\therefore f_s \geq 2 \times 100$$

$$\therefore f_s \geq 200\,Hz$$

এখানে একটা জিনিস লক্ষ্য করেন, প্রশ্নে বলছে, ম্যাক্সিমাম স্যাম্প্লিং রেট, কিন্তু আমরা বের করলাম মিনিমাম। প্রশ্নে একটু সমস্যা আছে। আসলে খেয়াল করেন, স্যাম্প্লিং রেটের কোনো আপার লিমিট নাই। যত ইচ্ছা তত নিতে পারেন। কিন্তু এর লোয়ার লিমিট আছে। স্যাম্প্লিং থিওরেম অনুসারে আপনি স্যাম্প্লিং রেট সিগনালের ম্যাক্সিমাম ফ্রিকোয়েন্সির দ্বিগুনের চেয়ে কম নিতে পারবেন না। কারন এর কম হলে, সিগনাল ডিস্টরটেড হয়ে যাবে। এবং আপনি সিগনাল রিকনস্ট্রাকশন করার সময় পুনরায় অরিজিনাল সিগনাল পাবেন না। তাহলে বুঝা গেল, প্রশ্নে মিনিমাম স্যাম্প্লিং রেট বলা উচিত ছিল।

b. Maximum amplitude, $m_p = 10$

We know , Maximum quantization error $\pm\frac{\delta}{2}$

So, $\left|\frac{\delta}{2}\right| = \frac{0.1}{100}$

$\therefore \delta = 0.002$

Or, $\frac{2m_p}{L} = 0.002$; [We know, $\delta = \frac{2m_p}{L}$]

Or, $L = \frac{2m_p}{0.002} = \frac{2 \times 10}{0.002} = 10{,}000$

Again we know, $n = \log_2 L = \log_2 10000 = 13.23 \approx 14$

কিন্তু বিট সংখ্যা কখনো ভগ্নাংশ হতে পারবে না। আমর বিট সংখ্যা সব সময় বেশীটা নিব। এক্ষেত্রে ১৩ এর পরিবর্তে ১৪ নিব।

c. Signaling rate, $R_b = nf_s = 14 \times 200 = 2800\,bit/sec$

d. The transmission channel bandwidth of PCM is given by

$B_T \geq \frac{R_b}{2}$

Or, $B_T \geq \frac{2800}{2}$

$\therefore B_T \geq 1400\ Hz$

92	**A PCM system uses a uniform quantizer followed by a 7 –bit binary encoder. The bit rate of the system is equal to 50×10^6 bits/sec.** **a. What is the maximum message bandwidth for which the system operates satisfactorily.** **b. Determine the output signal to quantization noise ratio when a full sinusoidal modulating wave frequency 1MHz is applied to the input.**	

a. We need to find the message bandwidth W.

Now, we know, sampling frequency, $f_s \geq 2W$

> স্যাম্প্লিং ফ্রিকোয়েন্সি সিগনাল ব্যান্ডউইথের দ্বিগুন অথবা এর চেয়ে বেশী হতে হবে।

Given that the number of bit n = 7.

For PCM

> যদি $R_b = nf_s$ লিখি তাহলে শুধু মিনিমাম সিগনালিং রেট বুঝায়। তাই greater than equal দিয়ে লিখলে ভাল।

Signaling rate, $R_b = 50 \times 10^6$ bits/sec

Now, we know, $R_b \geq nf_s$

Or, $R_b \geq n \times 2W$

Or, $\frac{R_b}{2n} \geq W$

Or, $W \leq \frac{R_b}{2n}$

Or, $W \leq \frac{50\times10^6}{2\times7}$

Or, $W \leq 3.57\ MHz$

b. Now given that modulating wave is sinusoidal. We know that for sinusoidal signal, the signal to quantization noise ratio is given by

$\left(\frac{S}{N}\right)_{dB} = 1.76 + 6n = 1.76 + 6 \times 7 = 43.8\ dB$

93	**The information in an analog waveform with maximum frequency $f_m = 3\ kHz$ is to be transmitted over an M-level PCM system where the number of pulse levels is M=16. The quantization distortion is specified not to exceed 1% of peak to peak analog signal.** **a. What is the maximum number of bits per sample that should be used in this PCM system?** **b. What is the minimum sampling rate and what is the resulting bit transmission rate?**	Similar Question in BUET M.Sc 2014

a. Given that quantization level, L=M=16

We know for PCM that no. of bits / sample $n = \log_2 L = \log_2 16 = 4$

b. We know the sampling rate, $f_s \geq 2W$

So, minimum sampling rate, $f_s = 2W = 2 \times 3\ kHz = 6\ kHz$

Now, the resulting minimum bit transmission rate or signaling rate, $R_b = nf_s$

$R_b = 4 \times 6 = 24\ kbps$

3.5 kHz

94	**A signal of bandwidth 3.5 kHz is sampled quantized and coded by a PCM system. The coded signal is then transmitted over a transmission channel of supporting a transmission rate of 50 k bits /sec. Calculate the maximum signal to noise ratio that can be obtained by this system.** **The input signal has peak to peak value of 4 volts and rms value of 0.2 volt**	

Now, signal power, $P_{signal} = \frac{V_{rms}^2}{R}$;

If R = 1 ohm, then, $P_{signal} = V_{rms}^2 = 0.2^2$; [Normalizing the signal power]

Now, quantization noise power, $P_{noise} = \frac{\delta^2}{12}$; $where\ \delta = \frac{2m_p}{L}$

Given that, $2m_p = 4\ volts$

$m_p = maximum\ amplitude\ range = \frac{peak\ to\ peak\ voltage}{2}$

$2m_p = maximum\ operating\ range = peak\ to\ peak\ voltage$

$\delta = \frac{4}{L}$

So, now we have to find the value of L first. We know, $L = \log_2 n$

Again, we know, signaling rate, $R_b = nf_s$; $where\ f_s = sampling\ frequncy \geq 2W$

$f_s \geq 2W$;

Or, $f_s \geq 2 \times 3.5\ kHz$

Or, $f_s \geq 7\ kHz$

Now, given that $R_b = 50 \times 10^3$bits /sec

Or, $nf_s = 50 \times 10^3$

Or, $n = \frac{50\times10^3}{7\times10^3} = 7.142 \approx 8\ bits$

So, $L = 2^n = 2^8 = 256$

So, $\delta = \frac{2m_p}{L} = \frac{4}{256}$

So, Quantization noise power, $P_{noise} = \frac{\delta^2}{12} = \left(\frac{4}{256}\right)^2 \times \frac{1}{12} = 0.0203\ mW$

So, Signal to quantization noise power, $\frac{S}{N} = \frac{P_{signal}}{P_{noise}} = \frac{0.2^2}{0.0203\times10^{-3}} = 1970 = 32.9\ dB$

95	**Consider an audio signal comprised of the sinusoidal term $s(t) = 3cos(500\pi t)$.** **a. Find the signal to quantization noise ratio when this is quantized using 10 bit PCM.** **b. How many bits of quantization are needed to achieve a signal to quantization noise ratio at least 40 dB?**	

a. We, for sinusoidal signal, SNR in dB, $\frac{S}{N} = 1.76 + 6n$

Given that, 10 bit PCM is used. So, n = 10

So, Signal to quantization noise ratio, $\left(\frac{S}{N}\right)_{dB} = 1.76 + 6 \times 10 = 61.76\ dB$

b. Here given that SNR in dB = 40 dB. We have to find the number of bits.

So, For sinusoidal signal in PCM, $\left(\frac{S}{N}\right)_{dB} = 1.76 + 6n$

Or, $40 = 1.76 + 6n$

Or, $n = 6.37 \approx 7$

96	**A 7 bit PCM system employing uniform quantization has an overall signaling rate of 56 k bits per second. Calculate the signal to quantization noise ratio that would result when its input is a sine wave with peak to peak amplitude equal to 5. What is the theoretical maximum frequency that this system can handle?**	

Given that 7 bit PCM. So, n = 7.

For sinusoidal signal input for PCM, $\left(\frac{S}{N}\right)_{dB} = 1.76 + 6n = 1.76 + 6 \times 7 = 43.76\ dB$ (ans)

Now, we know, signaling rate, $R_b = nf_s$

Or, $f_s = \frac{R_b}{n} = \frac{56\times10^3}{7} = 8\ kHz$

Now, we know, sampling frequency, $f_s \geq 2W$

Or, $W \leq \frac{f_s}{2}$

Or, $W \leq \frac{8\ kHz}{2}$

Or, $W \leq 4\ kHz$

So, the maximum frequency that can be handled is 4 kHz.

97	**The bandwidth of TV video plus audio signal is 4.5 MHz. If the signal is converter to PCM bit stream with 1024 quantization levels, determine the number of bits/sec generated by the PCM system. Assume that the signal sampled at the rate of 20% above Nyquist rate. Determine the minimum bandwidth required to transmit the signal. If above linear PCM system is converted to companded PCM, will the output bit rate change?**	**PGCB 2011**

Given, that signal bandwidth W = 4.5 MHz.

The Nyquist rate = 2W = 2 × 4.5 = 9 MHz

$L = 1024$

$n = \log_2 L$

The sampling rate is 20% above the Nyquist rate. i.e.

Sampling rate, $f_s = 9 + 9 \times \frac{20}{100} = 10.8\ MHz$

Again, given that, quantization level, $L = 1024\ levels$.

So, the number of bit/sample, $n = \log_2 1024 = 10\ bits$

So, the signaling rate, $R_b = nf_s = 10 \times 10.8 = 108\ mbps$ (ans)

Now, we know the transmission bandwidth for PCM, $B_T \geq \frac{R_b}{2}$

$B_T \geq \frac{R_b}{2}$

Or, $B_T \geq \frac{108}{2}$

Or, $B_T \geq 56\ MHz$

So, the minimum bandwidth is 56 MHz. (ans)

does not change.

The output bit rate does not change if linear PCM is converted into companded PCM. Companded PCM is used to improve the signal to noise ratio.

improve SNR.

98	**A compact disc (CD) records audio signals digitally by using PCM. Assume the audio signal bandwidth to be 15 kHz.** **a. What is Nyquist rate?** **b. If the Nyquist samples are quantized into L = 65,536 levels and then binary coded, determine the number of binary digits required to encode a sample.**	

	c. Determine the number of binary digits per second (bits/sec) required to encode the audio signal. **d. For practical reasons, the signals are sampled at a rate well above Nyquist rate at 44100 samples per second. If L = 65536 determine number of bits per second required to encode the signal and transmission bandwidth of encoded signal.**	

a. To obtain Nyquist rate:

Given that the bandwidth of the signal, W = 15 kHz

∴Nyquist rate = 2W = 2 × 15 =30 kHz.

b. To determine the number of bits: $n = \log_2 L = \log_2 65536 = \frac{\log_{10} 65536}{\log_{10} 2} = 16$

c. To determine signaling rate:

Signaling rate for PCM, $R_b \geq nf_s$

Here, we take $f_s = f_{Nq} = 30{,}000\ Hz$

So, minimum signaling rate, $R_b = nf_s = 16 \times 30{,}000 = 480\ k\ bits/sec$

d. Now, for 65536 levels , n = 16.

Also given that, $f_s = 44100\ samples/sec$

So, signaling rate, $R_b = nf_s = 16 \times 44100 = 705.6\ kbps$

So, transmission bandwidth, $B_T \geq \frac{R_b}{2}$

Or, $B_T \geq \frac{705.6}{2}$

Or, $B_T \geq 352.8\ kHz$

So, the minimum bandwidth will be 352.8 kHz.

99	**The output signal to noise ratio of a 10 bit PCM was found to be 30 dB. The desired SNR is 42 dB. It was decided to increase the SNR to the desired value by increasing the number of quantization levels. How many bits will require for 42 dB SNR to encode the signal? Find the fractional increasing in the transmission bandwidth required for this increase in SNR.**	

To obtain number of bits for 42 dB:

First we look at the equation of signal to noise ratio for non-sinusoidal signal, when only the number of bits are given.

$$\left(\frac{S}{N}\right)_{dB} = 4.8 + 6n$$

Now, we notice, if n = 1, then $\left(\frac{S}{N}\right)_{db} = 10.8\ dB$

If n = 2, then $\left(\frac{S}{N}\right)_{dB} = 16.8\ dB$

So, if we increase the number of bit by 1, the SNR also increases by (16.8 – 10.8) = 6 dB.

Now, the desired SNR is 42 dB.

So, SNR increases = 42 – 30 =12 dB

Now, 6 dB increases for the increasing the no. of bit by 1.

So, 12 dB will increase for the increasing the no. of bit 2.

So, For 42 dB SNR, total no. of bit will be (10+2)=12 bits (ans)

Now, for 30 dB SNR: no. of bit n = 10

So, transmission bandwidth, $B_T = \frac{R_b}{2} = \frac{nf_s}{2} = \frac{10f_s}{2} = 5f_s$

For 42 dB SNR, no. of bit n = 12

So, transmission bandwidth, $B_T = \frac{R_b}{2} = \frac{nf_s}{2} = \frac{12f_s}{2} = 6f_s$

Therefore, Fractional increases in bandwidth, $= \frac{6f_s - 5f_s}{5f_s} \times 100\% = 20\ \%$

100	**A telephone signal with cut off frequency of 4 kHz is digitized into 8 bit PCM, sampled at Nyquist rate. Calculate baseband transmission bandwidth and quantization noise ratio.**	

Given that, highest signal bandwidth, W = 4 kHz and n = 8

So, transmission bandwidth, $B_T = nW = 8 \times 4 = 32\ kHz.$

Note: Actually transmission bandwidth, $B_T = \frac{R_b}{2} = \frac{nf_s}{2} = \frac{n(2W)}{2} = nW;$

Because, $f_s \geq 2W$.

Now, quantization noise ratio for PCM for non sinusoidal signal when only n is given.

$$\left(\frac{S}{N}\right)_{dB} = 4.8 + 6n = 4.8 + 6 \times 8 = 52.8\ dB.$$

101	Twenty four voice channels of 4 kHz bandwidth each sampled at Nyquist rate and encoded into 8 bit PCM are time division multiplexed with 1 bit/frame as synchronization bit. What is the bit rate at the output of multiplexers?	

Given that, no. of channels = 24

Bandwidth, W = 4 kHz

No. of bit in PCM, n = 8

Now, sampling frequency = Nyquist rate = 2W=2 × 4 =8 kHz.

So, sampling period = $\frac{1}{8\ kHz} = 125\ \mu s$

So, each channel is continuously sampled after every 125 μ s. So, time duration of each frame will be 125 μ s.

Now, the no. of bit in each frame = no of channel/frame× no. of bit/channel + no. sync bit

=24 × 8 + 1=193 bits

Now, the bit rate, $r_b = \frac{total\ no.of\ bit\ in\ each\ frame}{time\ duration\ of\ each\ frame} = \frac{193}{125\times10^{-6}} = 1.544\ mbps$

ঐকিক নিয়মের মত মনে করলে হবে। ১২৫ সে. সময় লাগে ১৯৩ বিটের। তাহলে ১ সে. এ কত বিট যাবে?

102	The T_1 carrier system used in digital telephony multiplexes 24 voice channel based on 8 bit PCM. Each voice signal is usually put through a low pass filter with cut off frequency 3.4 kHz. The filtered signal is sampled at 8 kHz. In addition a single bit is added at the end of frame for the purpose of synchronization. a. The duration of each bit b. The resultant transmission rate c. Minimum required transmission bandwidth	

Total number of bit in each frame = 24 × 8+1=193 bits

Sampling frequency = 8 kHz

So, time duration of each frame = $\frac{1}{8000} = 125\ \mu s$

So, transmission rate, $r_b = \frac{no.of\ bit\ in\ each\ frame}{time\ duration\ of\ each\ frame} = \frac{193}{125\mu s} = 1.544\ mbps$

ঐকিক নিয়মের মত মনে করলে হবে। ১২৫ সে. সময় লাগে ১৯৩ বিটের। তাহলে ১ সে. এ কত বিট যাবে?

Again, time duration of each bit, $\frac{time\ duration\ of\ each\ frame}{no.of\ bit\ in\ each\ frame} = \frac{125\mu s}{193} = 0.638\mu s$

ঐকিক নিয়মের মত মনে করলে হবে।

Transmission bandwidth in PCM must be greater than half of the bit rate. i.e.

$$B_T \geq \frac{r_b}{2}$$

$$\geq \frac{1.544\times10^6}{2}$$

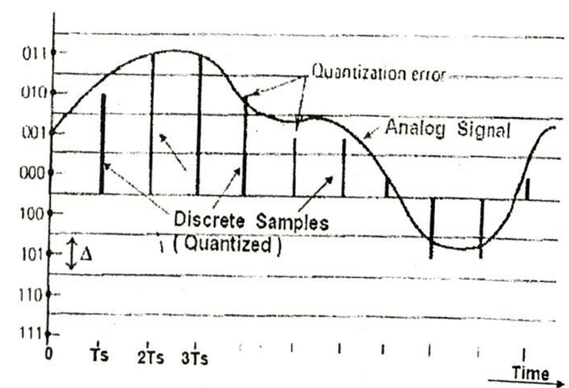
$$\therefore B_T \geq 772\ kHz.$$

103	Bandwidth Vs Quantization Error What bandwidth is needed to transmit a PCM encoded signal? Suppose that we want maximum error 0.5% m_p for a 3 kHz signal.	

We know that maximum quantization error, $\frac{\Delta}{2} = \frac{m_p}{L}$

Where,

$\Delta = step\ size\ for\ PCM\ (i.e.\ difference\ between\ two\ consecutive\ level\ of\ quantizer)$



(3 bit quantizer)

Fig. Typical Quantization process.

Quantization noise is produced in the transmitter end of a PCM system by rounding off sample values of an analog baseband signal to the nearest permissible representation levels of the quantizer. If Δ be the step size of a quantizer, then maximum error can occur is $\pm\frac{\Delta}{2}$. It can be proven that maximum quantization error, $\frac{\Delta}{2} = \frac{m_p}{L}$

$$\therefore \frac{0.5}{100} \times m_p = \frac{m_p}{L}$$

$$\therefore L = 200$$

We know, $L = 2^n$; *where n is the number of bit per sample*

$\therefore n = \log_2 200 = 7.64 \approx 8.$ [It is because we cannot take no. of bit fractional]

Now, Bandwidth of the signal, $W = 3000\ Hz$

So, Nyquist rate, $f_{NQ} = 2 \times 3000 = 6000\ Hz$

প্র্যাকটিক্যালি আমরা নাইকুইস্ট রেটের চেয়ে বেশী রেটে স্যাম্পিং করি। কিন্তু মিনিমাম নাইকুইস্ট রেটে স্যাম্পিং করতে হবে। নতুবা আমরা রিকনস্ট্রাক্ট করার সময় অরিজিনাল সিগনাল পাব না। যেহেতু এখানে নাইকুইস্ট রেটের চেয়ে কত বেশী রেটে স্যাম্পিং করতে হবে বলা নাই, তাই মনে করি নাইকুইস্ট রেটেই স্যাম্পিং করতে হবে। তাই স্যাম্পিং ফ্রিকোয়েন্সি ৬০০০ হার্জ।

So, $f_s = 6000\ Hz$

Signaling rage, $R_b = nf_s = 8 \times 6000 = 48000\ bit/s = 48kbps$

So, the bandwidth of PCM, $B_T = \frac{R_b}{2} = \frac{48000}{2} = 24000\ Hz = 24\ kHz.$

104	Explain delta modulation in detail with suitable diagram. Explain ADM and compare its performance with DM	

We have been seen in PCM that, it transmits all the bits which are used to code the sample. Hence signaling rate and transmission bandwidth are large in PCM. To overcome this problem Delta modulation is used.

Delta modulation transmits only one bit per sample. That is the present sample value is compared with the previous sample value and the indication, whether the amplitude is increased or decreased is sent. Input signal $x(t)$ is approximated to step signal by the delta modulator. This step size is fixed. The difference between the input signal $x(t)$ and staircase approximated signal confined to two levels, i.e. $+\delta$ and $-\delta$. If the difference is positive, then approximated signal is increased by one step i.e. $'\delta'$. If the difference is negative, then approximated signal is reduced by $'\delta'$. When the step is reduced, $'0'$ is transmitted and if the step is increased, $'1'$ is transmitted. Thus for each sample, only one binary bit transmitted.

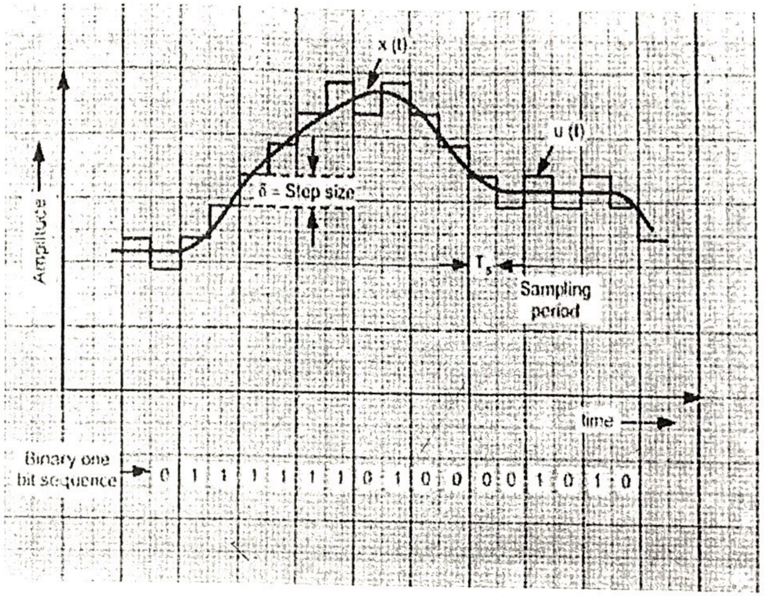
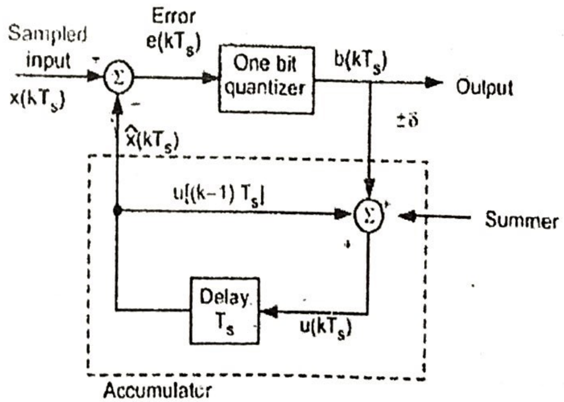


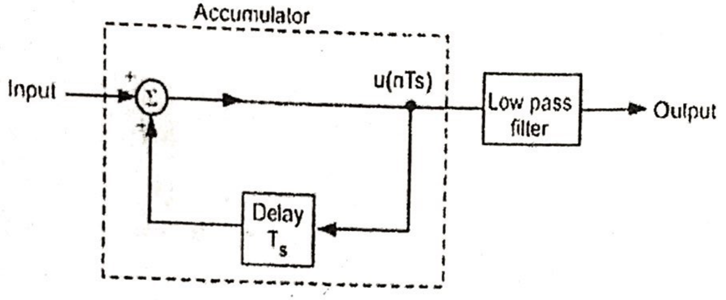
Fig:- Delta Modulation waveform

Block diagram of Delta modulation:



a. Transmitter

(a)



b. receiver

105	**What are the slope overload distortion and the granular noise in delta modulation and how it is removed in ADM?** **Or,** **Write down the advantages and disadvantages of delta modulation.**	**BPDB 2011**

Advantages of Delta Modulation over PCM:

1. Delta modulation transmits only one bit for one sample. Thus, the signaling rate and transmission channel bandwidth is quite small for delta modulation.
2. Transmitter and receiver implementation is very much simpler for delta modulation. There is no analog to digital converter involved in delta moduation.

Disadvantages of Delta modulation:

The delta modulation has two drawbacks:

1. Slope overload distortion (startup error)
2. Granular Noise (hunting)

Slope overload distortion: When the input signal is rising or falling with a slope larger than $\frac{\delta}{T_s}$; where δ is the size of the individual steps and T_s is the sampling period, we say that the sampler is suffering from Slop Overload Distortion. From the figure given below, we can see that the rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it, the step size 'δ' becomes too small for staircase signal $u(t)$ to follow the steep segment of $x(t)$. Thus, there is a large error between the staircase approximated signal and the original input signal $x(t)$. This

error is called slope overload distortion. To reduce this error, the step size should be increased when slope of the signal of $x(t)$ is high.

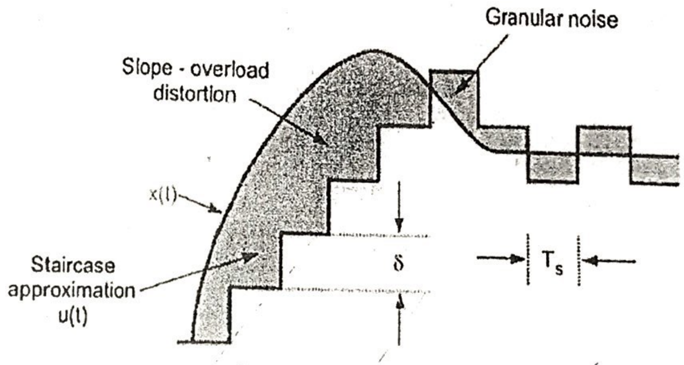


Fig: Slope overload distortion and Granular Noise

Granular Noise: Granular noise occurs when the step size is too large compared to small variations in the input signal. That is for very small variation in the input signal, the staircase signal is changed by large amount (δ) because of large step size. When the input signal is almost flat, the staircase signal keeps on oscillating by $\pm\delta$ around the signal. This error between the input and the approximated signal is called granular noise. The solution to this problem is to make step size small.

106	**Explain ADM and compare its performance with DM**	

To overcome the quantization errors due to slope overload and granular noise, the step size (δ) is made adaptive (প্রতিকূল পরিবেশের সাথে খাপ খাইয়ে নেয়ার যোগ্য) to variations in the input signal $x(t)$. Particularly in the steep segment of the signal $x(t)$, the step size is increased.. When the input is varying slowly, the step size is reduced. Then the method is called Adaptive Delta Modulation (ADM).

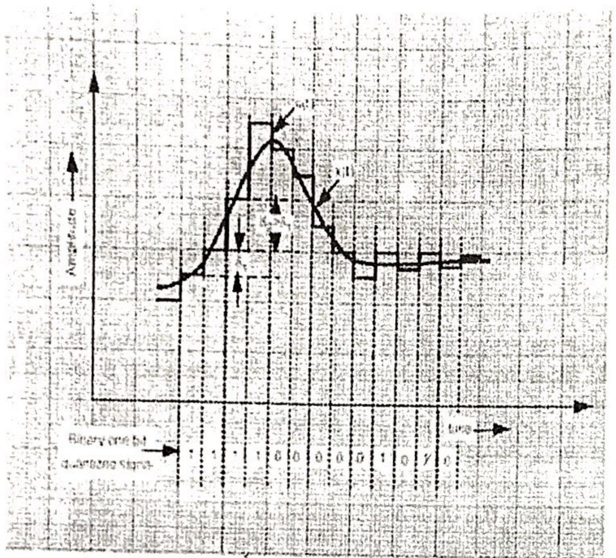


Fig:- Wave form of Adaptive Delta Modulation

Adaptive delta modulation has certain advantages over delta modulation. i.e.

1. The signal to noise ratio is better than ordinary delta modulation because of the reduction in slope overload distortion and granular noise.
2. Because of the variable step size, the dynamic range of ADM is wide.
3. Utilization of Bandwidth is better than delta modulation.

Plus other advantages of delta modulation are, only one bit per sample is required and simplicity of implementation of transmitter and receiver.

107	**Consider a sine wave of frequency f_m and amplitude A_m applied to a delta modulator of step size δ. Show that the slope over load distortion will occur if $A_m > \frac{\delta}{2\pi f_m T_s}$. Where T_s is the sampling period. Or, Derive the condition to avoid slope overload distortion.**	

Let the sine wave be represented as $x(t) = A_m \sin(2\pi f_m t)$

Slope of $x(t)$ will be maximum when derivative of $x(t)$ with respect to 't' will be maximum. আমরা জানি সাইন ওয়েব এর ম্যাক্সিমাম ভ্যালু বের করতে হলে এর ডিফারেনসিয়েশন করতে হয়।.

Again, The maximum slope of delta modulator is given as:

$$\text{Maximum slope} = \frac{step\ size}{Sampling\ perid} = \frac{\delta}{T_s}$$

আমরা জানি, ঢাল = লম্ব / ভূমি। যদি ডেল্টা মডুলেশনের গ্রাফ এ দেখি, তবে স্ট্যায়ারকেজ সিগনালে লম্ব হল স্টেপ সাইজ আর ভূমি হল স্যাম্পিং পিরিয়ড।

Slope overload distortion will take place if slope of sine wave is greater than slope of delta modulator i.e.

$$\max\left|\frac{d}{dt}x(t)\right| > \frac{\delta}{T_s}$$

$$\text{Or, } \max\left|\frac{d}{dt}(A_m \sin 2\pi f_m t)\right| > \frac{\delta}{T_s}$$

$$\text{Or, } \max|A_m 2\pi f_m \cos(2\pi f_m t)| > \frac{\delta}{T_s}$$

$$\text{Or, } A_m 2\pi f_m t > \frac{\delta}{T_s}$$

$$\therefore \quad A_m > \frac{\delta}{2\pi f_m T_s}$$

So, if $A_m > \frac{\delta}{2\pi f_m T_s}$, then slope overload will take place.

Hence, to avoid slope overload distortion $A_m \le \frac{\delta}{2\pi f_m T_s}$

108	**A delta modulator system is designed to operate at five times the Nyquist rate for a signal with 3 kHz bandwidth. Determine the maximum amplitude of a 2 kHz input sinusoid for which the delta modulator does not have slope overload. Quantizating step size is 250 mV.**	

We know that the condition to avoid slope overload for delta modulation is :

$$A_m \le \frac{\delta}{2\pi f_m T_s}$$

Here, $step\ size, \delta = 250\ mV = 0.25\ V$

Signal frequency, $f_m = 2\ kHz$

We can see from the question, that there are two frequency given i.e. 3 kHz and 2 kHz.

So, the maximum frequency in the signal, $W = 3\ kHz$

So, Nyquist rate, $f_{NQ} = 2 \times W = 2 \times 3 = 6\, kHz.$

So, the sampling frequency, $f_s = 5\, times\ Nyquist\ rate$; [It is told in the question.]

$$f_s = 5 \times 6 = 30\, kHz$$

$\therefore$ $sampling\ period, T_s = \frac{1}{f_s} = \frac{1}{30\times10^3}$

Hence, putting these values in $A_m \leq \frac{\delta}{2\pi f_m T_s}$

$$A_m \leq \frac{0.25}{2\pi\times10^3\times\left(\frac{1}{30\times10^3}\right)}$$

$$A_m \leq 0.6\, volts$$

So, the maximum amplitude is 0.6 volts.

109	**Find the signal amplitude for minimum quantization error in a delta modulation system if step size is 1 volts having repetition period 1ms. The information signal operates at 100 Hz.**	

The quantization error is minimum when slope over load distortion is absent. Then the quantization error is due to granular noise only. We know that the slope overload distortion is absent if, $A_m \leq \frac{\delta}{2\pi f_m T_s}$

Or, $A_m \leq \frac{1}{2\pi\times100\times1\times10^{-3}}$

Or, $A_m \leq 1.6\, V$

That is, the signal amplitude should be less than 1.6 volts to have minimum quantization error.

110	**Derive an expression for signal to quantization noise power ratio for delta modulation. Assume that no slope overload distortion exists.**	

To obtain signal power:

We know that slope overload distortion will not occur if $A_m \leq \frac{\delta}{2\pi f_m T_s}$. ....(1)

So, the maximum signal amplitude will be, $A_m = \frac{\delta}{2\pi f_m T_s}$ ....(2)

Signal power is given as, $P = \frac{V^2}{R}$

Here V is the rms valude of the signal. Here, $V = \frac{A_m}{\sqrt{2}}$.

Hence, $P = \frac{\left(\frac{A_m}{\sqrt{2}}\right)^2}{R}$

Normalized signal power is obtained by taking R=1.

Hence, $P_{signal} = \frac{A_m^2}{2}$ ....(3)

$$P_{signal} = \frac{\left(\frac{\delta}{2\pi f_m T_s}\right)^2}{2} = \frac{\delta^2}{8\pi^2 f_m^2 T_s^2}$$

This is an expression for signal power in delta modulation.

নোটঃ এতবড় সূত্র মনে রাখা একটু কষ্ট। তাই অংক করার সময় আমরা (১) নং বা (২) নং সমীকরনের সূত্রটি মনে রাখলেই চলবে। প্রথমে ম্যাক্সিমাম এম্প্লিচুড বের করব। তারপর তা পাওয়ার এর সূত্রতে (৩) নং সমীকরনে বসিয়ে দিব।

To obtain noise power:

We know that the maximum quantization error in delta modulation is equal to step size 'δ'

Let, the quantization error be uniformly distributed over an interval $[-\delta, \delta]$. From the figure quantization error can be expressed as:

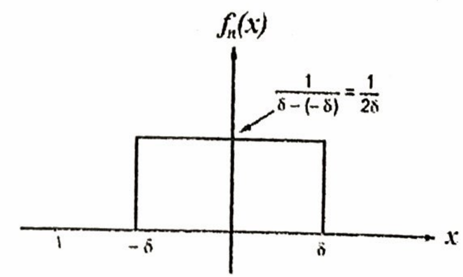


Fig. Uniform distribution of quantization error

$$f_n(x) = \begin{cases} \frac{1}{2\delta}; & -\delta < x < \delta \\ 0; & otherwise \end{cases}$$

The noise power is given as, Noise power $= \frac{V_{noise}^2}{R}$

Here V_{noise}^2 is the mean square value of noise voltage.

So, the mean square value of noise voltage $= \int_{-\infty}^{\infty} x^2 f_n(x)dx$

$$= \int_{-\delta}^{\delta} x^2 \left(\frac{1}{2\delta}\right) dx$$

$$= \frac{1}{2\delta}\int_{-\delta}^{\delta} x^2 dx$$

$$= \frac{1}{2\delta}\left[\frac{x^3}{3}\right]_{-\delta}^{\delta}$$

$$= \frac{1}{2\delta}\left[\frac{\delta^3}{3} + \frac{\delta^3}{3}\right]$$

$$= \frac{\delta^2}{3}$$

Hence, the noise power $= \frac{\frac{\delta^2}{3}}{R}$

Normalized noise power can be obtained with R = 1. Hence, noise power $= \frac{\delta^2}{3}$

This noise power is uniformly distributed over $-f_s$ to f_s range. This is shown in the figure below.

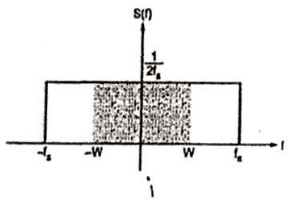


Fig. PSD of noise

Now, at the output of delta modulator receiver, there is a low pass filter whose cut off frequency is 'W'. This cut off frequency is equal to highest signal frequency. For this reason, the reconstruction filter passes part of the noise power at the output. So, the output noise power will be: এখানে যে ব্যাপারটা বুঝানো হচ্ছে, নয়েজ পুরাটা আউটপুটে পাওয়া যাবে না। কারন, যখন ফিল্টারিং করা হচ্ছে তখন নয়েজও বাদ যাচ্ছে। তাই পুরা নয়েজ পাওয়ারের পরিবর্তে এর একটা অংশ পাওয়া যাবে। ব্যাপারটা অনেকটা ঐকিক নিয়মের মত চিন্তা করলেও হয়। f_s এ নয়েজ পাওয়ার এত হলে, W তে কত!!!

Output noise power $= \frac{W}{f_s} \times noise\ power = \frac{W}{f_s} \times \frac{\delta^2}{3}$

We know that $f_s = \frac{1}{T_s}$, hence above equation can also be written as:

Output power noise $= \frac{WT_s\delta^2}{3}$

আমার কাছে Output noise power $P_{noise} = \frac{W}{f_s} \times \frac{\delta^2}{3}$ এই সূত্রটা মনে রাখাই সহজ।

To obtain signal to noise power ratio:

Signal to noise power ratio at the output of delta modulation receiver is given as,

$$\frac{S}{N} = \frac{Signal\ Power}{Noise\ Power}$$

$$\frac{S}{N} = \frac{\frac{\delta^2}{8\pi^2 f_m^2 T_s^2}}{\frac{WT_s\delta^2}{3}}$$

$$\therefore \frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$$

আমার কাছে এই সূত্রটা মনে রাখা বেশী কষ্ট কর। তাই অংক করার সময় আলাদা আলাদা ভাবে সিগনাল পাওয়ার ও নয়েজ পাওয়ার বের করে পরে সিগনাল টু নয়েজ রেশিও বের করব।

111	**A DM system is designed to operate at 3 times the nyquist rate for a signal with a 3 kHz bandwidth. The quantizing step size is 250 mV.** **1. Determine the maximum amplitude of a 1 kHz input sinusoidal for which the delta modulator does not show slope over load.** **2. Determine the post filtered output SNR for the signal of part (1)**	

Given, bandwidth $W = 3\ kHz$

Signal frequency, $f_m = 1\ kHz$

So, nyquist rate $= 2f_m = 2 \times 1kHz = 2\ kHz$

Therefore, sampling frequency, $f_c = 3 \times nyquist\ rate$

$$= 3 \times 2\ kHz = 6\ kHz$$

Step size, $\delta = 250\ mV = 0.250\ V$

Sampling period, $T_s = \frac{1}{f_s} = \frac{1}{6000}\ s$

To obtain maximum signal amplitude:

We know, slope overload distortion does not occur if,

$$A_m \leq \frac{\delta}{2\pi f_m T_s}$$

$$A_m \leq \frac{250\times10^{-3}}{2\pi\times1000\times\left(\frac{1}{6000}\right)}$$

$$A_m \leq 0.2387\ V$$

Thus maximum amplitude is 238.7 mV.

To obtain signal to noise ratio:

Signal power, $P_{signal} = \frac{A_m^2}{2} = \frac{(0.2387)^2}{2} = 0.0285\ W$

Noise power, $P_{noise} = \frac{W}{f_s} \times \frac{\delta^2}{3} = \frac{3}{6} \times \frac{0.250^2}{3} = 0.01042\ W$

So, SNR $= \frac{P_{signal}}{P_{noise}} = \frac{0.0285}{0.01042} = 2.74$

SNR (dB) $= 10\log\left(\frac{P_{signal}}{P_{noise}}\right) = 10\log 2.74 = 4.38\ dB$

112	**A DM system is tested with a 10 kHz sinusoidal signal with 1V peak to peak at the input. It is sampled at 10 times the Nyquist rate.** **1. What is the step size required to prevent slope overload?** **2. What is the corresponding SNR?**	

The condition to prevent slope overload distortion is

$$A_m \leq \frac{\delta}{2\pi f_m T_s}$$

Now, given that: signal frequency, $f_m = 10\ kHz$

Signal amplitude $A_m = \frac{1V}{2} = 0.5V$; [as it is said 1V peak to peak]

Nyquist rate $f_{NQ} = 2 \times f_m = 2 \times 10\ kHz = 20\ kHz$

So, sampling frequency, $f_s = 10 \times f_{NQ} = 10 \times 20\ kHz = 200\ kHz$

$\therefore T_s = \frac{1}{f_s} = \frac{1}{200\times10^3}\ sec$

To obtain step size:

$A_m \leq \frac{\delta}{2\pi f_m T_s}$

$or, \delta \geq 2\pi f_m T_s A_m$

$or, \delta \geq 2\pi \times 10 \times 10^3 \times \frac{1}{200\times10^3} \times 0.5$

$or, \delta \geq 0.157\ V$

Thus the step size greater than 157 mV will prevent the slope overload.

To obtain signal to noise ratio:

Signal power: $P_{signal} = \frac{A_m^2}{2} = \frac{0.5^2}{2} = 0.125\ Watt$

Noise power: $P_{noise} = \frac{W}{f_s} \times \frac{\delta^2}{3}$

But, if we notice that this is the noise power after post filtered where W is the cutoff frequency of the filter and it is equal to voice signal bandwidth i.e. it is equal to the highest frequency of the signal. But W is not given here. For this reason, we have to calculate the noise power for whole sampling frequency. In that case, noise power, $P_{noise} = \frac{\delta^2}{3} = \frac{0.157^2}{3}\ Watt$

So, signal to noise ratio, $SNR = \frac{0.125}{\frac{0.157^2}{3}} = 15.2 = 11.8\ dB$; $[SNR(dB) = 10\log_{10}(\frac{S}{N})]$

113	**In a single integration DM scheme, the voice signal is sampled at a rate of 64 kHz. The maximum signal amplitude is 1 volt, voice signal bandwidth is 3.5 kHz.** **a. Determine the minimum value of step size to avoid slope overload.** **b. Determine granular noise N_o** **c. Assuming signal to be sinusoidal, calculate signal power and signal to noise ratio.** **d. Assuming that noise signal amplitude is uniformly distributed in the range (-1,1) determine the signal power and signal to noise ratio.**	

Given that,

$f_s = 64\ kHz$ $\quad A_m = 1V;$ $\quad W = 3.5\ kHz$

$T_s = \frac{1}{f_s} = \frac{1}{64\times10^3}$

To obtain step size:

The condition to avoid slope overload is given as:

$A_m \leq \frac{\delta}{2\pi f_m T_s}$

$or, \delta \geq 2\pi f_m T_s A_m$

$or, \delta \geq 2\pi \times 3.5 \times 10^3 \times \left(\frac{1}{64\times10^3}\right) \times 1$

$or, \delta \geq 0.3436\ V$

To obtain granular noise power:

As slope overload will not occur if step size is 0.3436 V, only granular noise will occur.

Noise power, $P_{noise} = N_o = \frac{W}{f_s} \times \frac{\delta^2}{3}$

$= \frac{3.5\times10^3}{64\times10^3} \times \frac{(0.3436)^2}{3}$

$= 2.15 \times 10^{-3} V = 2.15\ mV$

To obtain signal to noise ratio:

Signal power: $P_{signal} = \frac{A_m^2}{2} = \frac{1^2}{2} = 0.5\ watt$

Signal to noise ratio, $SNR = \frac{S}{N} = \frac{P_{signal}}{P_{noise}} = \frac{0.5}{2.15\times10^{-3}} = 232.3 = 23.66\ dB$

To calculate SNR if signal is uniformly distributed over the range (-1,1):

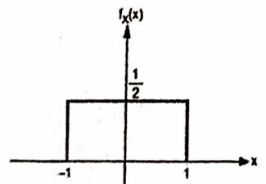


Fig. Uniformly distributed signal

Figure shows the power density function of the signal.

Here, $f_x(x) = \begin{cases} \frac{1}{2} & ; \quad -\delta < x < \delta \\ 0 & ; \quad otherwise \end{cases}$

To calculate the power, we have to find mean square of the x.

Mean square of x, $\int_{-1}^{1} x^2 f_x(x)dx = \int_{-1}^{1} x^2.\frac{1}{2}dx = \frac{1}{2}\left[\frac{x^3}{3}\right]_{-1}^{1} = \frac{1}{3}$

Signal power= $\frac{V_{rms}^2}{R}$

Here V_{rms} = *root mean square of x*

If R=1, then Normalize signal power = $V_{rms}^2 = \frac{1}{3}\ watt$

Hence signal to noise ratio becomes, $\frac{S}{N} = \frac{signal\ power}{noise\ power} = \frac{\frac{1}{3}}{2.15\times10^{3}} = 155 = 21.9\ dB$

Note: SNR (dB) = $10\log_{10}(\frac{S}{N})$

114	**Describe the transmission bandwidth in PCM.**	

Let the quantizer use 'n' number of binary digits to represent each level. Then the number of levels that can be represent by 'n' digits will be,

$$L = 2^n$$

Here, 'L' represents total number of digital levels of quantizer.

For example, if n = 3 bits, then total number of levels will be,

$L = 2^3 = 8\ levels$

Now, each sample is converted to 'n' binary bits. i.e. Number of bits per sample = n

We know that, Number of samples per second = f_s

∴ Number of bits per second is given by,

(Number of bits per second)=(Number of bits per samples) × (Number of samples per second)

$\therefore R_b = nf_s$

Where R_b = Number of bits per second = signaling rate in PCM

n = number of bits of quantizer

And f_s= samples per second; where $f_s \geq 2W$; W = highest frequency of the signal.

Now, Bandwidth needed for PCM transmission will be given by half of the signaling rate. It is because 2 bits/sec can be transmitted with 1 Hz bandwidth.

So, Transmission bandwidth of PCM $B_T \geq \frac{R_b}{2}$;

Where $R_b = nf_s$ And $f_s \geq 2W$

115	**A PCM system uses a uniform quantizer followed by a 7 bit encoder. The system bit rate is 50 Mbits/sec. Calculate the maximum bandwidth of the message signal for which this system operates satisfactorily.**

Given that: bit rate, R= 50 Mbits/sec and no. of bits of encoder, n=7

We know that bit rate, $R_b = nf_s$

$\therefore f_s = \frac{R_b}{n} = \frac{50\times10^6}{7} = 7.14\ MHz$

Maximum signal bandwidth, $BW = \frac{f_s}{2} = 3.57\ MHz.$

116	**The bandwidth of a video signal is 4.5 MHz. This signal is to be transmitted using PCM with the number of quantization levels Q=1024. The sampling rate should be 20% higher than the Nyquist rate. Calculate the system bit rate.**

Given that: Bandwidth, W = 4.5 MHz.

$\therefore Nyquist\ rate, f_{NQ} = 2 \times Bandwidth = 2W = 2 \times 4.5 = 9\, MHz$

But, sampling rate f_s should be 20 % higher than the Nyquist rate.

$\therefore f_s = f_{NQ} + f_{NQ} \times 20\% = f_{NQ} + 0.2f_{NQ} = 1.2f_{NQ} = 1.2 \times 9 = 10.8\ MHz.$

We know that, Quantization level, $L = 2^n$; where n is the number of the bit of encoder.

So, we can write, $\therefore 1024 = 2^n$

$\therefore 2^{10} = 2^n$

$\therefore n = 10$

Again, $L = 2^n$

$\therefore n = \log_2 L$

$\therefore n = \frac{\log_{10} L}{\log_{10} 2}$

The log used in calculator is 10 base log.

$\therefore$ $System\ bit\ rate, R_b = nf_s = 10 \times 10.8 = 108\ Mbps$

117	**If a voice frequency signal is sampled at the rate of 32,000 samples/sec and characterized by peak value of 2 volts, determine the value of step size to avoid overload. What is quantization noise power, N_Q and corresponding SNR? Assume the bandwidth of the signal as 4 kHz.**

Given, sampling rate, $f_s = 32{,}000\ samples/sec$

Sampling period, $T_s = \frac{1}{f_s} = \frac{1}{32000}$

So, maximum amplitude $A_m = 2V$

Signal Bandwidth, $f_m = 4\ kHz.$

Now, we know the condition to avoid slope overload in delta modulation that

$A_m \le \frac{\delta}{2\pi f_m T_s}$

Note: $f_s = \frac{1}{T_s}$

$\delta \ge 2\pi f_m T_s A_m$

Or, $\delta \ge 2\pi \times 4 \times 10^3 \times \frac{1}{32000} \times 2$

$\therefore \delta \ge 1.57\ Volt$

Quantization noise power (N_Q):

The quantization noise power for a delta modulation is given by;

$N_Q = \frac{\delta^2}{3} = \frac{1.57^2}{3} = 0.822\ W$

Signal to noise ratio:

$SNR = \frac{signal\ power}{noise\ power}$

$Signal\ power, P_s = \frac{A^2}{2} = \frac{2^2}{2} = 2\ W$

Average output noise power $P_{noise} = \frac{W}{f_s}\left(\frac{\delta^2}{3}\right) = \frac{4000}{32000}\left(\frac{1.57^2}{3}\right) = 0.1027\ W$

Here, W = cut off frequency of the filter which is equal to the highest frequency of the signal.

$\therefore SNR = \frac{2}{0.1027} = 19.47$

118	**A compact disc (CD) record audio signals digitally by PCM. Assume audio signal's bandwidth to be 15 kHz. If signals are sampled at a rate 20% above Nyquist rate practical reasons and the samples are quantized into 65,536 levels, determine bits/sec required to encode the signal and minimum bandwidth required to transmit encoded signal.**

Given, Bandwidth, $W = 15\ kHz$

Nyquist Rate, $f_{NQ} = 2 \times W = 2 \times 15 = 30\ kHz$

Sampling rate, $f_s = f_{NQ} + f_{NQ} \times 20\% = f_{NQ} + 0.2f_{NQ} = 1.2f_{NQ} = 1.2 \times 30 = 36\ kHz$

Now, signaling rate, $R_b = nf_s$

Given, quantization level, L=65536

Now, $n = \log_2 L = \log_2 65536 = 16$

$\therefore signaling\ rate, R_b = nf_s = 16 \times 36 = 576\ kbits/sec$

Now, the minimum bandwidth, $B_T = \frac{R_b}{2} = \frac{576}{2} = 288\ kHz$

Note: Maximum bandwidth $= \frac{f_s}{2}$; and minimum bandwidth $= \frac{R_b}{2}$

119	**Define information, bit and entropy.**	33rd BCS

Information: Information is that which informs. In other word, it is the answer to a question of some kind. It is thus related to data and knowledge, as data represents values attributed to parameters, and knowledge signifies understanding of real things or abstract concepts. Information can be encoded into various forms for transmission and interpretation. For example, information may be encoded into a sequence of signs or transmitted via a sequence of signals. It can also be encrypted for safe storage and communication.

Let us consider, the communication system which transmits messages $m_1, m_2, m_3 \ldots\ldots$, with probabilities of occurrence $p_1, p_2, p_3, \ldots\ldots\ldots$. The Amount of information transmitted through the message m_k with probability p_k is given :

Amount of Information: $I_k = \log_2\left(\frac{1}{p_k}\right)$

The unit of information is 'bit'. And, binary digit is represented by 'binit'.

Bit: The bit is a basic unit of information in computing and digital communication. A bit can have only one of two values. These values are mostly represented as either a 0 or 1. In information theory, one bit is typically defined as the uncertainty of a binary random variable that is 0 or 1 with equal probability.

Entropy: In information theory, systems are modeled by a transmitter, channel and a receiver. The transmitter produces messages that are sent through the channel. The channel modifies the message in some way. The receiver attempts to infer which message was sent. In this context, entropy is the average value of the information contains in each message.

Average of information or entropy, $H = \sum_{k=1}^{M} p_k \log_2\left(\frac{1}{p_k}\right)$

And

Information Rate: R = rH

Where R = information rate

H = entropy of average information (প্রতি মেসেজে কতটি ইনফরমেশন বিট থাকে)

$r = rate\ at\ which\ messages\ are\ generated$ (প্রতি সেকেন্ডে কতটি মেসেজ জেনেরেট হয়)

120	**Calculate the amount of information if binary digits (binits) occur**	

	with equal likelihood in binary PCM.	

We know that in binary PCM, there are only two binary levels, i.e. 1 or 0. Since they occur with equal likelihood, their probabilities of occurrence will be,

$$p_1('o'\ level) = p_2('1'\ level') = \frac{1}{2}$$

Hence the amount of information carried will be given as:

$$I_1 = \log_2\left(\frac{1}{p_1}\right) = \log_2\left(\frac{1}{0.5}\right) = 1 \text{ bit}$$

$$I_2 = \log_2\left(\frac{1}{p_2}\right) = \log_2\left(\frac{1}{0.5}\right) = 1 \text{ bit}$$

So, In binary PCM 1 bit of information is carried by one binary digit.

121	**In binary PCM if '0' occur with probability $\frac{1}{4}$ and '1' occur with probability $\frac{3}{4}$, then calculate amount of information conveyed by each binary digit (binit).**	

Amount of information carried by '0' binary digit, $I_1 = \log_2\left(\frac{1}{p_1}\right) = \log_2 4 = 2\ bits$

Amount of information carried by '1' binary digit, $I_2 = \log_2\left(\frac{1}{p_2}\right) = \log_2\left(\frac{4}{3}\right) = 0.415\ bits$

Notice that binary digit with lower probability carries more information.

122	**If there are M equally likely and independent messages, then prove that amount of information carried by each message will be, $I = N\ bits$; where $M = 2^N$ and N is an integer.**	

Since all the M messages are equally likely and independent, probability of occurrence of each message will be $\frac{1}{M}$; [According to rules of probability].

We know that amount of information is given as $I_k = \log_2\left(\frac{1}{p_k}\right)$

$$\text{Or, } I_k = \log_2 M;\ \left[as\ p_k = \frac{1}{M}\right].$$

$$\text{Or, } I_k = \log_2 2^N;\ [as\ M = 2^N\ given]$$

$$\therefore I_k = N\ bits$$

This shows that when the messages are equally likely and coded with equal number of binary digits (binits), then the information carried by each message (measured in bits) is numerically same as the number of binits used for each message.

123	**Prove that If receiver knows the message being transmitted, the**	

	amount of information carried is zero.	

It is stated that receiver knows the message is being transmitted, hence the probability of occurrence of this message will be $p_k = 1$.

So, amount of information $I_k = \log_2\left(\frac{1}{p_k}\right) = \log_2 1 = 0\ bits$

This proves the statement that if receiver knows message, the amount of information carried is zero.

124	**If I_1 is the information carried by message m_1 and I_2 is the information carried by message m_2, then prove that the amount of information carried compositely due to m_1 and m_2 is $I_{1,2} = I_1 + I_2$**	

Let, p_1 is the probability of occurrence of message m_1 and p_2 is the probability of occurrence of message m_2. Then probability of composite message (একত্রে ঘটার সম্ভাবতা) $p_1 p_2$.

So, the amount of information carried compositely due to m_1 and m_2 is

$$I_{1,2} = \log_2\left(\frac{1}{p_1 p_2}\right) = \log_2\left(\frac{1}{p_1}\right) + \log_2\left(\frac{1}{p_2}\right) = I_1 + I_2$$

125	**An analog signal is bandlimited B Hz and sampled at Nyquist rate. The samples are quantized into 4 levels. Each level represents one message. Thus there are 4 messages. The probability of occurrence of these 4 levels (messages) are $p_1 = p_4 = \frac{1}{8}$ and $p_2 = p_3 = \frac{3}{8}$. Find out the information rate.**	

We know information rate, $R = rH$

Now, average information bit per message or entropy, $H = \sum_{k=1}^{M} p_k \log_2\left(\frac{1}{p_k}\right)$

So, $H = p_1 \log_2\left(\frac{1}{p_1}\right) + p_2 \log_2\left(\frac{1}{p_2}\right) + p_3 \log_2\left(\frac{1}{p_3}\right) + p_4 \log_2\left(\frac{1}{p_4}\right)$

Or, $H = \frac{1}{8}\log_2 8 + \frac{3}{8}\log_2 \frac{8}{3} + \frac{3}{8}\log_2 \frac{8}{3} + \frac{1}{8}\log_2 8$

Or, $H = \frac{1}{4}\log_2 8 + \frac{3}{4}\log_2 \frac{8}{3}$

Or, $H = \frac{1}{4} \times \frac{\log_{10} 8}{\log_{10} 2} + \frac{3}{4} \times \frac{\log_{10}\left(\frac{8}{3}\right)}{\log_{10} 2}$

Or, $H = 0.75 + 1.061$

$\therefore H = 1.811\ bits/message$

Now, rate of message generate, r

The bandwidth of the signal is B Hz.

The signal is sampled at Nyquist rate. Nyquist rate = 2B samples /sec.

Since every sample generates one message signal, message per second, r = 2B message/sec

So, the information rate: $R = rH = 2B$ message /sec × 1.8 bits / message

Or, R = 3.6 B.

126	**Find the information rate of a binary PCM if all messages are equally likely. Assuming the signal is sampled at Nyquist rate and signal bandwidth is B.**	

Binary PCM means it's encoder has 2 bits. So, the number of level $= 2^n = 2^2 = 4$. So, the number of level is 4. Each level will represent one message. Now, the probability of each message will be $p_1 = p_2 = p_3 = p_4 = \frac{1}{4}$

So, average information bit per message or entropy is given as

$$H = p_1 \log_2\left(\frac{1}{p_1}\right) + p_2 \log_2\left(\frac{1}{p_2}\right) + p_3 \log_2\left(\frac{1}{p_3}\right) + p_4 \log_2\left(\frac{1}{p_4}\right)$$

Or, $H = \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4$

Or, H = 2.

Again, Now, rate of message generate, r

The bandwidth of the signal is B Hz.

The signal is sampled at Nyquist rate. Nyquist rate = 2B samples /sec.

Since every sample generates one message signal, message per second, r = 2B message/sec

So, information rate, R = r H = 2B × 2 = 4 B.

127	**We know that the entropy for equally likely levels, $H = \log_2\left(\frac{1}{p}\right)$; where p is probability of each level. Or, if M = no. of level or messages, then prove, $H = log_2 M$**	**Important for Math**

Assume there are M levels. So, probability of each level, $p = \frac{1}{M}$

$$H = p_1 \log_2\left(\frac{1}{p_1}\right) + p_2 \log_2\left(\frac{1}{p_2}\right) + \cdots \ldots \ldots + p_M \log_2\left(\frac{1}{p_M}\right)$$

Or, $H = p\log_2\left(\frac{1}{p}\right) + p\log_2\left(\frac{1}{p}\right) + \cdots \ldots \ldots + p\log_2\left(\frac{1}{p}\right)$

Or, $H = M \times p\log_2\left(\frac{1}{p}\right)$

Or, $H = \frac{1}{p} \times p\log_2\left(\frac{1}{p}\right)$

$\therefore H = \log_2\left(\frac{1}{p}\right)$

Or, $H = \log_2 M$

128	Describe Shannon's Theorem on Channel capacity.	

There are two theorem on channel capacity.

1. Channel coding Theorem (Shannon's Second Theorem)
2. Shannon Hartley Theorem for Gaussian channel

Channel coding theorem: We have seen previously that the information is transmitted through the channel with rate 'R' called information rate. Shannon's theorem says that it is possible to transmit information with an arbitrarily small probability of error provided that information rate 'R' is less than or equal to rate 'C', called channel capacity. Thus the channel capacity is the maximum information rate with which the error probability is within the tolerable limits.

Therefore, if $R \leq C$, it is possible to transmit information without any error if noise is present.

Shannon Hartley Theorem for Gaussian channel: The channel capacity of a white bandlimited Gaussian channel is : $C = Blog_2\left(1+\frac{S}{N}\right)$ $bits/sec$

Here, B is the channel Bandwidth

S is the signal power

N is the total noise power within the channel bandwidth.

Note: We know that signal power is given as:

$$\text{Power, } P = \int_{-B}^{B} power\ spectral\ density$$

Here B is bandwidth. And power spectral density of white noise is $\frac{N_0}{2}$. Hence the noise power N becomes: Noise power, $N = \int_{-B}^{B} \frac{N_0}{2} df$

$$\therefore N = N_0 B$$

Points to remember: Channel capacity depends of two factors. (a) Bandwidth (B) of the channel and (b) Signal to noise (S/N) ratio.

Noiseless channel has infinite capacity.

Infinite bandwidth channel has limited capacity.

129	**The data is to be transmitted at the rate of 10000 bits/sec over a channel having bandwidth B = 3000 Hz. Determine the signal to noise ratio required. If the bandwidth is increased to 10000 Hz, then determine the signal to noise ratio.**	

Given that the data is to be transmitted at the rate of 10,000 bits / sec. Hence channel capacity must be at least 10000 bits / sec for errorless transmission.

For bandwidth B = 3000 Hz

So, channel capacity, $C = B\log_2\left(1+\frac{S}{N}\right)$

$$\text{Or,} 10000 = 3000\log_2\left(1+\frac{S}{N}\right)$$

$$\text{Or, } \log_2\left(1+\frac{S}{N}\right) = \frac{10000}{3000} = 3.333$$

$$\text{Or, } \left(1+\frac{S}{N}\right) = 2^{3.333} = 10.08$$

$$\text{Or, } \frac{S}{N} = 10.08 - 1 = 9.08$$

For bandwidth B = 10000 Hz

So, channel capacity, $C = B\log_2\left(1+\frac{S}{N}\right)$

$$\text{Or, } 10000 = 10000\log_2\left(1+\frac{S}{N}\right)$$

$$\text{Or, } \frac{S}{N} = 1$$

Above results show that bandwidth is increased to 10000 Hz, the signal to noise ratio reduced by nine times. This means, if bandwidth is increased, the required signal power is reduced.

130	**An analog signal having 4 kHz bandwidth is sampled at 1.25 times the Nyquist rate, and each sample is quantized into one of 256 equally likely levels. Assuming the samples to be statistically independent:** **a. What is the information rate of the source?** **b. Can the output of this source transmitted without error over**	

	AWGN channel with a bandwidth of 10 kHz and S/N ratio of 20 dB? c. Find S/N ratio required for error-free transmission of part (b) d. Find the bandwidth required for an AWGN channel for an error free transmission of the output of this source if S/N ratio is 20 dB.	

a. To obtain information rate: We know information rate, $R = rH$

Here, r = rate of message generate per second

and H = entropy or average amount of information bit per message.

Now, given that Bandwidth B = 4 kHz.

Nyquist rate = 2 B = 2 × 4 kHz = 8 kHz

Hence the sampling rate, r = 1.25 × Nyquist rate = 1.25 × 8000 = 10,000 messages / sec

[Note: the unit of sampling rate kHz or samples / second or message / sec. any one of them can be used]

Again, given that the samples are quantized into 256 equally likely levels. So, there will be M = 256 samples and each sample will represent one message. Now, according to probability law, each sample will have a probability of occurrence as $\frac{1}{256}$. এইটা সম্ভাবতার সাধারন সূত্র হতে পাওয়া যায়। যেমন একটা ঝুড়ির ভিতর থেকে ১০ টা বলের ভিতর যে কোন একটা বল উঠানোর সম্ভাবতা ১/১০ বা ০.১। কারন যেকোনো বল উঠতে পারে। সবার সম্ভাবতা সমান।

We know that the entropy for equally likely levels, $H = \log_2\left(\frac{1}{p}\right)$ or $H = \log_2 M$

$$\text{Or, } H = \log_2 256$$

$$\therefore H = 8\ bit/sample$$

So, the information rate, $R = rH = 10000 \times 8 = 80{,}000\ bits/sec$

b. To check for error free transmission of B = 10 kHz and S/N = 20 dB:

The condition for error free transmission is , $R \leq C$

Now, $\left(\frac{S}{N}\right)_{dB} = 10\log_{10}\frac{S}{N}$

Or, $20 = 10\log_{10}\left(\frac{S}{N}\right)$

Now, rate of message generate, r

The bandwidth of the signal is B Hz.

The signal is sampled at Nyquist rate. Nyquist rate = 2B samples /sec.

Since every sample generates one message signal, message per second, r = 2B message/sec

So, the information rate: $R = rH = 2B$ message /sec × 1.8 bits / message

Or, R = 3.6 B.

126	**Find the information rate of a binary PCM if all messages are equally likely. Assuming the signal is sampled at Nyquist rate and signal bandwidth is B.**	

Binary PCM means it's encoder has 2 bits. So, the number of level $= 2^n = 2^2 = 4$. So, the number of level is 4. Each level will represent one message. Now, the probability of each message will be $p_1 = p_2 = p_3 = p_4 = \frac{1}{4}$

So, average information bit per message or entropy is given as

$$H = p_1\log_2\left(\frac{1}{p_1}\right) + p_2\log_2\left(\frac{1}{p_2}\right) + p_3\log_2\left(\frac{1}{p_3}\right) + p_4\log_2\left(\frac{1}{p_4}\right)$$

Or, $H = \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4 + \frac{1}{4}\log_2 4$

Or, H = 2.

Again, Now, rate of message generate, r

The bandwidth of the signal is B Hz.

The signal is sampled at Nyquist rate. Nyquist rate = 2B samples /sec.

Since every sample generates one message signal, message per second, r = 2B message/sec

So, information rate, R = r H = 2B × 2 = 4 B.

127	**We know that the entropy for equally likely levels, $H = \log_2\left(\frac{1}{p}\right)$; where p is probability of each level. Or, if M = no. of level or messages, then prove, $H = log_2 M$**	**Important for Math**

Assume there are M levels. So, probability of each level, $p = \frac{1}{M}$

$$H = p_1\log_2\left(\frac{1}{p_1}\right) + p_2\log_2\left(\frac{1}{p_2}\right) + \cdots\ldots\ldots + p_M\log_2\left(\frac{1}{p_M}\right)$$

Or, $H = p\log_2\left(\frac{1}{p}\right) + p\log_2\left(\frac{1}{p}\right) + \cdots \ldots \ldots \ldots + p\log_2\left(\frac{1}{p}\right)$

Or, $H = M \times p\log_2\left(\frac{1}{p}\right)$

Or, $H = \frac{1}{p} \times p\log_2\left(\frac{1}{p}\right)$

$\therefore H = \log_2\left(\frac{1}{p}\right)$

Or, $H = \log_2 M$

128	Describe Shannon's Theorem on Channel capacity.	

There are two theorem on channel capacity.

1. Channel coding Theorem (Shannon's Second Theorem)
2. Shannon Hartley Theorem for Gaussian channel

Channel coding theorem: We have seen previously that the information is transmitted through the channel with rate 'R' called information rate. Shannon's theorem says that it is possible to transmit information with an arbitrarily small probability of error provided that information rate 'R' is less than or equal to rate 'C', called channel capacity. Thus the channel capacity is the maximum information rate with which the error probability is within the tolerable limits.

Therefore, if $R \le C$, it is possible to transmit information without any error if noise is present.

Shannon Hartley Theorem for Gaussian channel: The channel capacity of a white bandlimited Gaussian channel is : $C = Blog_2\left(1+\frac{S}{N}\right)$ $bits\ /sec$

Here, B is the channel Bandwidth

S is the signal power

N is the total noise power within the channel bandwidth.

Note: We know that signal power is given as:

$$\text{Power, } P = \int_{-B}^{B} power\ spectral\ density$$

Here B is bandwidth. And power spectral density of white noise is $\frac{N_0}{2}$. Hence the noise power N becomes: Noise power, $N = \int_{-B}^{B} \frac{N_0}{2} df$

$$\therefore N = N_0 B$$

$\therefore \frac{S}{N} = 10^{\frac{20}{10}} = 100$

Now, the channel capacity of AWGN channel is given by

$C = B\log_2\left(1+\frac{S}{N}\right)$

$or, C = 10{,}000\log_2(1+100)$

$\therefore C = 66.582\ kbits\ /sec$

If $R \le C$, then error free transmission is possible. Here R = 80000 and C =66.582 kbits/sec. hence R>C. Therefore, error free transmission is not possible.

c. S/N ratio for error free transmission:

We have, information rate R = 80000 bit/sec and bandwidth 10000 Hz.

For error free transmission, $R \le C$

$$\text{Or, } R \le B\log_2\left(1+\frac{S}{N}\right)$$

$$\text{Or, } 80000 \le 10000\log_2\left(1+\frac{S}{N}\right)$$

$$\text{Or, } 8 \le \log_2\left(1+\frac{S}{N}\right)$$

$$\text{Or, } \log_2\left(1+\frac{S}{N}\right) \ge 8$$

$$\text{Or, } \left(1+\frac{S}{N}\right) \ge 2^8$$

$$\text{Or, } \frac{S}{N} \ge 255$$

$$\text{Or, } \left(\frac{S}{N}\right)_{dB} = 10\log_{10}255 = 24\ dB$$

d. To determine the bandwidth:

Now, given S/N ratio = 20 dB

$\left(\frac{S}{N}\right)_{dB} = 10\log_{10}\left(\frac{S}{N}\right)$

$\text{Or},20 = 10\log_{10}\left(\frac{S}{N}\right)$

$\text{Or}, \frac{S}{N} = 10^{\frac{20}{10}} = 100$

The condition for error free transmission is $R \leq C$. Hence

$R \leq C$

$\text{Or}, R \leq B\log_2\left(1+\frac{S}{N}\right)$

$\text{Or}, 80{,}000 \leq B\log_2(1+100)$

$\text{Or}, B \geq 11.974\ kHz$

131	**In an facsimile transmission of a picture, there are about 2.25 × 10^6 picture elements per frame. For good reception, twelve brighten levels are necessary. Assuming all these levels to be equiprobable, calculate the channel bandwidth required to transmit on picture in every three minutes for a signal to noise power ratio of 30 dB. In SNR requirement increases by 40 dB, calculate the new bandwidth. Explain the trade-off between bandwidth and SNR by comparing two results.**	

Given that 2.25 × 10^6 picture elements per frame and we have to transmit picture in every three minutes.

So, message rate, $r = \frac{2.25\times10^6}{3\ minutes} = \frac{2.25\times10^6}{180\ seconds} = 12500\ picture\ elements/sec$

Here each picture element is a symbol. Hence, r = 12500 symbol /sec.

Now, there 12 levels. So, M = 12. All these levels are equally probable. So, entropy or average information per message for likely probable symbol is given as

$H = \log_2 M = \log_2 12 = 3.585\ bits/symbol$

Now, the transmission rate, $R = rH = 12500 \times 3.585 = 44812\ bits/sec$

Given that SNR = 30 dB = $10\log_{10}\left(\frac{S}{N}\right)$

$\therefore \frac{S}{N} = 10^{\frac{30}{10}} = 1000$

Now, for error less transmission, $R \leq C; Where\ C = Blog_2\left(1+\frac{S}{N}\right)$

$or, R \leq Blog_2\left(1+\frac{S}{N}\right)$

$or, 44812 \leq B\log_2(1+1000)$

$or, B \geq 4495.93\ Hz\ or\ 4.5\ kHz$

To calculate the new bandwidth for S/N = 40 dB:

$\left(\frac{S}{N}\right)_{dB} = 40 = 10\log_{10}\left(\frac{S}{N}\right)$

$\text{Or}, \frac{S}{N} = 10^{\frac{40}{10}} = 10{,}000$

Now, for error free transmission $R \leq C; where\ C = B\log_2\left(1+\frac{S}{N}\right)$

$or, R \leq B\log_2\left(1+\frac{S}{N}\right)$

$or, 44812 \leq B\log_2(1+10000)$

$or, B \geq 3372.4\ Hz\ or\ 3.37\ kHz$

Note: notice that when SNR increases, bandwidth decreases.

132	**A voice grade telephone channel has a bandwidth of 3400 Hz. If the signal to noise ratio (SNR) on the channel is 30 dB, determine the capacity of the channel. If the above channel is to be used to transmit 4.8 kbps of data determine the minimum SNR required on the channel.**	

Given that channel bandwidth B = 34000 Hz

Also given, $\left(\frac{S}{N}\right)_{dB} = 30\ dB = 10\log_{10}\left(\frac{S}{N}\right)$

$\therefore \frac{S}{N} = 10^{\frac{30}{10}} = 1000$

Now, the channel capacity, $C = B\log_2\left(1+\frac{S}{N}\right)$

$= 3400\log_2(1+1000)$

$= 33.888\ kbps$

To obtain minimum SNR for 4.8 kbps data;

Here given transmission rate, R = 4.8 kbps

For, error free transmission, $R \leq C$; *where C is the channel capacity*

$$or, R \leq B \log_2 \left(1 + \frac{S}{N}\right)$$

$$or, 4.8 \times 10^3 \leq 3400 \log_2 \left(1 + \frac{S}{N}\right)$$

$$or, \log_2 \left(1 + \frac{S}{N}\right) \geq 1.412$$

$$or, 1 + \frac{S}{N} \geq 2^{1.412}$$

$$or, \frac{S}{N} \geq 2.66 - 1$$

$$\therefore \frac{S}{N} \geq 1.66$$

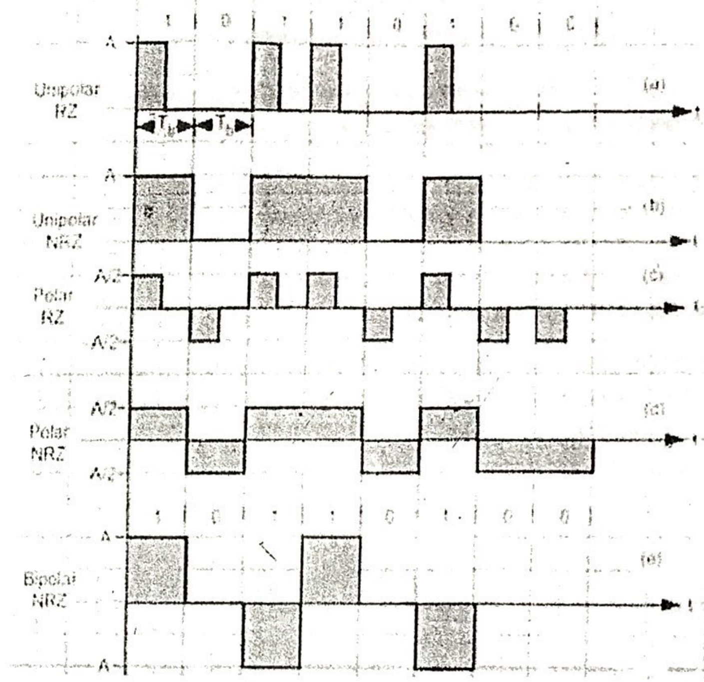
So, the minimum 1.66 SNR or 2.2 dB SNR will need to transmit date at the rate of 4.8 kbps.

Line Coding

133	**What is Line coding or Data formats**	

The analog waveforms are converted to digital signals by PCM, DM, ADM , DPCM etch techniques. This digital data can be represented by different formats or waveforms. These waveforms are commonly known as digital data formats or their representation is called line coding.

134	**Represent the bit sequence 10110100 by** **(a) Unipolar RZ and NRZ** **(b) Polar RZ and NRZ** **(c) Bipolar NRZ** **(d) Split phase Manchester.**	



(লাইন কোডিং থেকে পরীক্ষায় প্রশ্ন আসলে বর্ননা আসবে না। আসলে বিট সেকোয়েন্স দেওয়া থাকবে, সেই অনুসারে ছবি আকতে বলবে। তাই বুঝার সুবিধার জন্য বাংলায় ব্যাখ্যা করা হল যাতে সহজে মনে থাকে।)

Unipolar RZ: Unipolar বলতে একই পোলারিটি বুঝায়। এখানে আমরা শুধু মনে করব Unipolar বললে শুধু পজিটিভ পোলারিটি বুঝাবে। Unipolar এর বেলায় সব সময় ভেল্যু হয় পজিটিভ হবে, নতুবা জিরো হবে। RZ মানে রিটার্ন টু জিরো। যদি বলা হয় Unipolar RZ, তাহলে ‘1’ বিটের জন্য ‘1’ বিটের প্রথম হাফ সাইকেল হবে পজিটিভ আর বাকি হাফ সাইকেল হবে জিরো। এর জন্যই একে RZ বা Return to Zero বলা হচ্ছে। কারন, ‘1’বিটের বেলায় প্রথম হাফ সাইকেল পজিটিভ থাকার পর তা আবার অটোমেটিক জিরো তে ফিরে আসছে। পজিটিভ ভেল্যুর মান হবে এ। অন্যদিকে, ‘0’ বিটের জন্য পুরো সাইকেলের জন্যই হবে জিরো। এখানে পজিটিভ ভেল্যুর মান হবে A.

Unipolar NRZ: NRZ বলতে বুঝায় Non Return to Zero। তার মানে এইটা জিরোতে ফিরে আসবে না। কখন আসবে না? ‘1’বিটের বেলায়। ‘1’বিটের বেলায় পুরো সাইকেল জুড়েই পজিটিভ মান এ থাকবে। আর ‘0’বিটের বেলায় পুরো সাইকেল জুড়েই জিরো থাকবে।

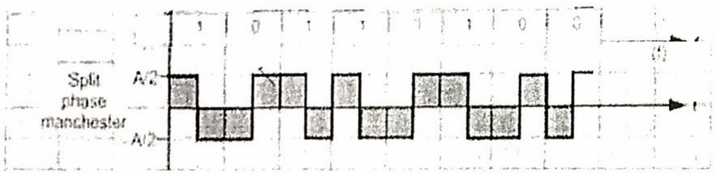
Polar RZ: পোলার এর বেলায় ভেল্যু হবে পজিটিভ, জিরো অথবা নেগেটিভ। Polar RZ এর ক্ষেত্রে, যেহেতু রিটার্ন টু জিরো বলা হচ্ছে, তাই ‘1’ বিটের বেলায়, প্রথম পজিটিভ হাফ সাইকেল যাওয়ার পর ভেল্যু জিরোতে ফিরে আসবে। পজিটিভ ভেল্যুটি

হবে $+\frac{A}{2}$। আবার, '0' বিটের বেলায়, প্রথম হাফ সাইকেল নেগেটিভ ভেল্যু থাকার পর দ্বিতীয় হাফ সাইকেলের সময় আবার জিরো তে ফিরে আসবে। এই জন্য এটিকে বলা হচ্ছে পোলার রিটার্ন টু জিরো। এক্ষেত্রে নেগেটিভ ভেল্যুর মান হবে $-\frac{A}{2}$।

Polar NRZ: এক্ষেত্রে ভেল্যু কখনই জিরো ফিরে আসবে না। '1'বিটের বেলায় পুরো সাইকেল এর জন্যই পজিটিভ মান হবে। আর '0'বিটের বেলায় নেগেটিভ মান হবে। এক্ষেত্রেও পজিটিভ ভেল্যুর মান হবে $+\frac{A}{2}$। এবং নেগেটিভ ভেল্যুর মান হবে $-\frac{A}{2}$।

Bipolar NRZ (Alternate Mark Inversion or AMI): Bipolar NRZ একটু অন্য রকম। Bipolar NRZ এর ক্ষেত্রে বিট সিকোয়েন্স খেয়াল করতে হবে। যে বিট সিকোয়েন্স দেওয়া থাকবে, তার প্রথম '1'বিটের জন্য হবে পজিটিভ ভেল্যু। এর পর দ্বিতীয় '1'বিট যখন আসবে তখন নেগেটিভ ভেল্যু হবে। তারপর, তৃতীয় '1'বিটের বেলায় আবার পজিটিভ ভেল্যু হবে। এভাবে প্রতি '1'বিটের জন্য পোলারিটি পরিবর্তন হতে থাকবে। অন্যদিকে জিরো বিট থাকলে ভেল্যু হবে জিরো। '0'বিটের সময় এক্ষেত্রে সব সময় জিরো হবে। যখনই '0' বিট আসবে তখনই জিরো হবে।

Split Phase Manchester: Split Phase Manchester এর জন্য '1' বিটের বেলায় প্রথম হাফ সাইকেল পজিটিভ ও দ্বিতীয় হাফ সাইকেল নেগেটিভ হবে। আবার '0' বিটের বেলায় প্রথম হাফ সাইকেল নেগেটিভ ও দ্বিতীয় হাফ সাইকেল পজিটিভ হবে। পজিটিভ ভেল্যুর মান হবে $+\frac{A}{2}$ আর নেগেটিভ ভেল্যুর মান হবে $-\frac{A}{2}$।



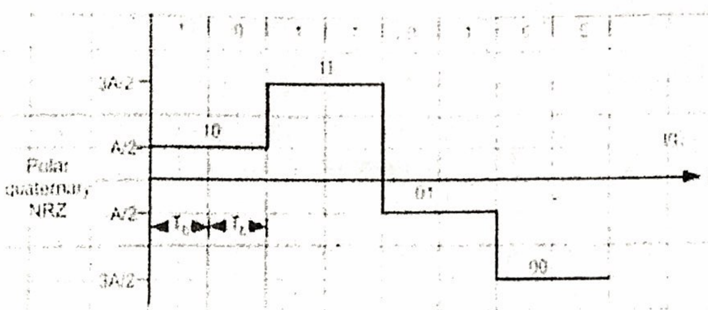
135	Represent the sequence 10110100 by polar quaternary NRZ.	

Polar Quaternary NRZ: এই পদ্ধতির ক্ষেত্রে মেসেজের বিট গুলোকে দুইটা দুইটা করে নিয়ে ব্লক তৈরী করব। যদি দুইটা দুইটা করে নিয়ে ব্লক তৈরী করি, তবে চার রকমের কম্বিনেশন পাওয়া যাবে। সেগুলো হলঃ 00, 01, 10 and 11. এই চার রকমের কম্বিনেশন বা ব্লকগুলোর প্রতিটিকে নিচের ছবিতে যেমন দেখানো হয়েছে সেরকম করে একটি একটি আলাদা চারটি অ্যাম্পিচিউড লেভেল দিতে হবে।

Message combination	$x(t) = e_n$
00	$-\frac{3A}{2}$
01	$-\frac{A}{2}$
10	$\frac{A}{2}$
11	$\frac{3A}{2}$

Table Polar quaternary NRZ : combinations of bits

এই টেবিল অনুসারে কোডিং করতে হবে যা নিচের ছবিতে দেখানো হয়েছে। এই টেবিল মুখস্ত রাখতে হবে।



পোলার কোয়াটারনারিতে শুধু দুইটি বিট নিয়ে কোডিং করা হয়েছে। দুটির বেশি বিট নিয়েও প্রতিটিকে একটি করে নির্দিষ্ট অ্যাম্প্লিচিউড এর ভ্যালু দিয়ে কোডিং করা যায়। যেমন: তিন বিট করে নিলে মোট আটটি লেভেল হবে, তখন আটটি ভ্যালু নির্দিষ্ট করে দিতে হবে। একে বলা হয় M – ary Coding.

136	Represent the sequence 10110100 by Gray coding.	

প্রথমে গ্রে কোড বুঝতে হবে। গ্রে কোড হল পর পর দুটি বাইনারি সংখ্যা ভিতরে একটি মাত্র বিটের পরিবর্তন থাকবে। আমরা যদি বাইনারি সিরিয়ালি ০০, ০১, ১০, ১১। কিন্তু গ্রে কোড অনুসারে লিখতে হলে লিখতে হবে, ০০, ০১, ১১, ১০। দেখা যাচ্ছে গ্রে কোডের ক্ষেত্রে আগের বাইনারি সংখ্যার চেয়ে পরের বাইনারি সংখ্যায় মাত্র একটি করে বিটের পরিবর্তন হয়েছে। তাহলে বলা যায়, বাইনারি ০০ এর গ্রে কোড হল ০০, বাইনারি ০১ এর গ্রে কোড হল ০১, বাইনারি ১০ এর গ্রে কোড হল ১১ এবং ১১ এর গ্রে কোড হল ১০। যদি তিনটি করে বাইনারি ডিজিটের গ্রে কোড বের করি তবে হবে এইরকম:

বাইনারি	গ্রে কোড
০০০	০০০
০০১	০০১
০১০	০১১
০১১	০১০
১০০	১১০
১০১	১১১
১১০	১০১
১১১	১০০

0 xor 0 = 0
0 xor 1 = 1
1 xor 0 = 1
1 xor 1 = 0

Binary: 1 0 0
Gray code: 1 1 0

প্রথমে বাম পাশের বিটটি যা থাকবে তাই লিখতে হবে। এখানে আছে '১'। তাই '১' ই রাখা হল। এর পর থেকে পর পর দুটি বিটের এক্স অর করতে হবে। ডান পাশে এক্স অর এর টেবিল দেওয়া হল। ১ আর ০ এর এক্স অর হল ১, তাই গ্রে কোডের দ্বিতীয় বিটও ১। ০ আর ০ এর এক্স অর ০, তাই পরের বিট ০ হয়েছে। এভাবে বাকি গুলো করতে হবে।

এখন গ্রে কোডের মাধ্যমে চ্যানেল কোডিং করার সময়, প্রথমে নিচের টেবিলটি দেখতে হবে। দুই বিট থাকলে দুই বিটের মেসেজ হতে পারে ০০, ০১, ১০, ১১। আর এই চারটির গ্রে কোড হতে পারে ০০, ০১, ১১ ও ১০। এখন এই চারটিকে একটি নির্দিষ্ট ভ্যালুর একটি অ্যাম্প্লিচিউড দিতে হবে। নিচে টেবিলে তা দেখানো হল:

বাইনারি	গ্রে কোড	অ্যাম্প্লিচিউড
০০	০০	$-\frac{3A}{2}$
০১	০১	$-\frac{A}{2}$
১০	১১	$+\frac{A}{2}$
১১	১০	$+\frac{3A}{2}$

প্রশ্নে দেওয়া আছে, ১০১১০১০০। দুইটা করে নিলেঃ ১০, ১১, ০১, ০০।

১০ গ্রে কোডের জন্য অ্যাম্প্লিচিউড হলঃ $+\frac{3A}{2}$।

১১ গ্রে কোডের জন্য অ্যাম্প্লিচিউড হলঃ $+\frac{A}{2}$।

০১ গ্রে কোডের জন্য অ্যাম্প্লিচিউড হল $-\frac{A}{2}$।

এবং ০০ গ্রে কোডের জন্য অ্যাম্প্লিচিউড হল $-\frac{3A}{2}$।

ছবিতে সেইভাবেই দেখানো হয়েছে।

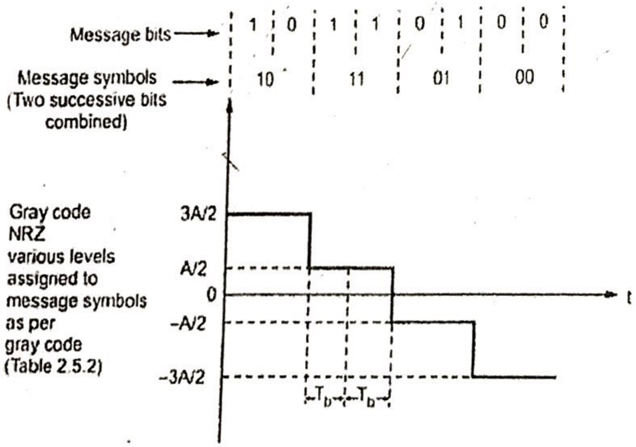


Fig. Gray code NRZ

Passband Transmission

137	What is Passband data transmission? Advantages and disadvantages.	

There are basically two types of transmission of digital signals.

a. **Baseband data transmission:** The digital data is transmitted over the channel directly. There is no carrier or any modulation. This is suitable for transmission over short distances.
b. **Passband data transmission:** The digital data modulates high frequency sinusoidal carrier. Hence it is also called digital CW modulation. It is suitable for transmission over long distances.

Types of passband modulation:

The digital data can modulate phase , frequency or amplitude of carrier. There are three basic techniques.

a. **Phase shift keying (PSK):** In this technique, the digital data modulates phase of the carrier.
b. **Frequency shift keying (FSM):** In this technique, the digital data modulates frequency of the carrier.
c. **Amplitude shift keying (ASK):** In this technique, the digital data modulates amplitude of the carrier.

Requirements of passband transmission scheme: Any passband transmission scheme should satisfy following requirements:

a. Maximum data transmission rate
b. Minimum probability of symbol error.
c. Minimum transmitted power.
d. Minimum channel bandwidth
e. Maximum resistance to interfering signals.
f. Minimum circuit complexity.

Advantages of passband transmission over baseband transmission:

1. Long distance transmission.
2. Analog channels can be used for transmission.
3. Multiplexing techniques can be used for bandwidth conservation.
4. Problems such as ISI (inter symbol interference) and cross talk are absent.
5. Passband transmission can take place over wireless channels also.
6. Large number of modulation techniques are available.

Disadvantages of Passband communication:

1. Modulation and demodulation equipments, transmitting / receiving antennas, interference problems make the system complex.
2. It is not suitable for short distance communication.

Passband transmission model:

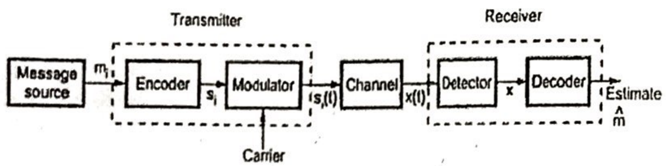


Fig. Model of passband data transmission system

138	Write a short note on PSK	30th BCS

Phase shift keying (PSK): In this technique, the digital data modulates phase of the carrier.

If only two bits, symbol '1' or '0' modulates the phase of the carrier, it is called BPSK or Binary Phase Shift Keying. Let assume, the carrier signal is $s(t) = Acos(2\pi f_0 t)$, where A is the maximum amplitude of the carrier.

In the BPSK technique, when the symbol is changed, then the phase of the carrier is changed by 180 degree (or π radians).

For example, when '1' is transmitted, the carrier would be $s_1(t) = +Acos(2\pi f_0 t)$

If the next symbol is '0' i.e. when '0' is transmitted, then the carrier would be

$$s_2(t) = Acos(2\pi f_0 t + 180°)$$

$$\boxed{cos(\theta + 180°) = -cos\theta}$$

$$or, s_2(t) = -A\ cos(2\pi f_0 t)$$

So, the general equation of BPSK can be written as

$$S(t) = b(t)Acos(2\pi f_0 t)$$

Here, b(t) = +1 when binary '1' is to be transmitted

= - 1 when binary '0' is to be transmitted

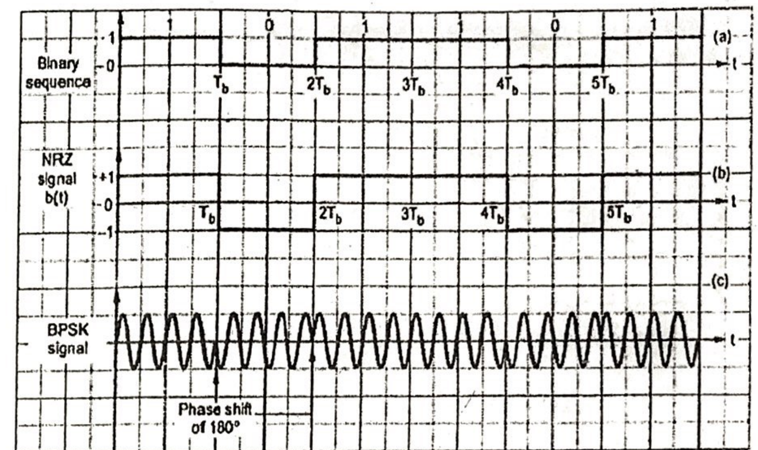


Fig. **(a) Binary sequence**
(b) Its equivalent bipolar signal *b(t)*
(c) BPSK signal

In the figure, it has been seen that the binary sequence is converted into equivalent NRZ signal, b(t). This signal directly modulates carrier $cos(2\pi f_0 t)$.

Generation of BPSK signal:

As we have been seen earlier, that the given message binary signal is converted into binary NRZ signal, b(t) using an encoder. Then, This baseband signal b(t) is applied as a modulating signal to the balanced modulator. The, applying carrier signal to the balanced modulator, the BPSK signal can be generated. The block diagram is given below:

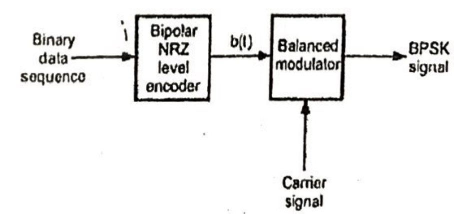


Fig. BPSK generation scheme

Reception of BPSK signal: BPSK সিগনালকে চ্যানেলের ভিতর দিয়ে পাঠানোর পর অপর প্রান্তে রিসিভার সিগনাল রিসিভ করে। তারপর একে ডিমডুলেশন করার পর অরিজিনাল মেসেজ সিগনাল এর বাইনারি সিকোয়েন্স বের করা হয়। নিচে ব্লক ডায়াগ্রাম দেওয়া হল ।

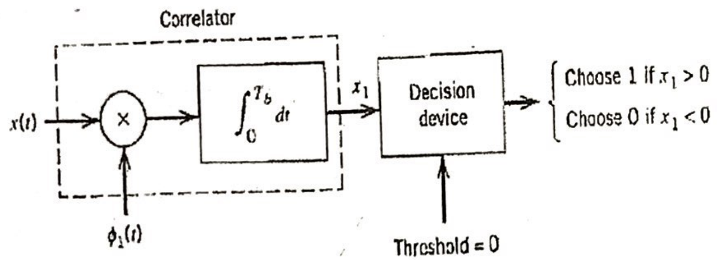
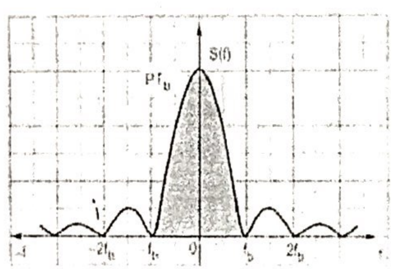


Fig: Coherent Binary PSK

To detect the original binary sequence of 1s and 0s, we apply the noisy PSK signal x(t) (at the channel output) to a correlator, which is also supplied with a locally generated coherent reference signal $\phi_1(t)$. The correlatator output, x_1, is compared with a threshold of zero volts. If $x_1>0$, the receiver decides in favor of symbol 1. On the other hand, if $x_1< 0$, it decides in favor of symbol 0. If x_1 is exactly zero, the receiver makes a random guess in favor of 0 or 1.

Spectrum of BPSK signal:



যেহেতু NRZ সিগনাল square wave, আর আমরা জানি স্কয়ার ওয়েভ এর fourier transform হচ্ছে Sinc function. তাই এর freqeuncy spectrum হবে এই রকম।

Fig. Plot of power spectral density of NRZ baseband signal

আর বিপিএসকে এর স্পেকট্রাম হবে নিচের ছবির মত।

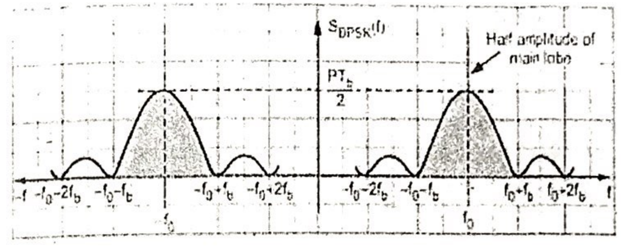


Fig. Plot of power spectral density of BPSK signal

সুতরাং দেখা যাচ্ছে বিপিএসকে সিগনালের ব্যান্ডউইথ হবে $= (f_0 + f_b) - (f_0 - f_b) = 2f_b$

where $f_b = \frac{1}{T_b}$; $T_b = bit\ duration$

Drawbacks of BPSK:

a. In the BPSK technique, there is an ambiguity in the output signal.
b. This technique suffers from ISI (inter symbol interference) and Inter channel Interference.

139	**What is interchannel interference and intersymbol interference?**	

Interchannel interference: In a given transmission channel, the interference resulting from signals in one or more other channels is called interchannel interference. Example of interchannel interference: crosstalk in telephony; cross-view and other colour in television.

Intersymbol interference: In telecommunication, Intersymbol Interference (ISI) is a form of distortion of a signal in which one symbol interfers with subsequent symbols. This is an unwanted phenomenon as the previous symbol have similar effect as noise, thus making the communication less reliable.

140	**Determine the minimum bandwidth for a BPSK modulator with a carrier frequency of 40 MHz and an input bit rate of 500 kbps.**	

Notice, $T_b = bit\ duration$, (মানে একটা বিট যেতে কত সময় লাগে)

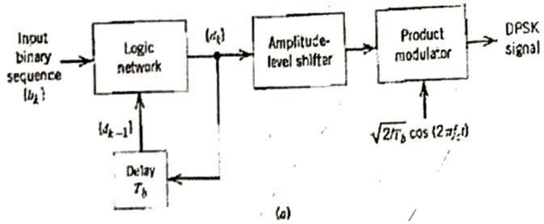
Now, $f_b = \frac{1}{T_b}$; ($\frac{1}{T_b}$ মানে হল ১ সেকেন্ডে কতটি বিট যেতে পারে। ঐকিক নিয়মের মত। তাহলে f_b হচ্ছে এখানে বিট রেট।)

So, given that , $f_b = 500\ kpbs$

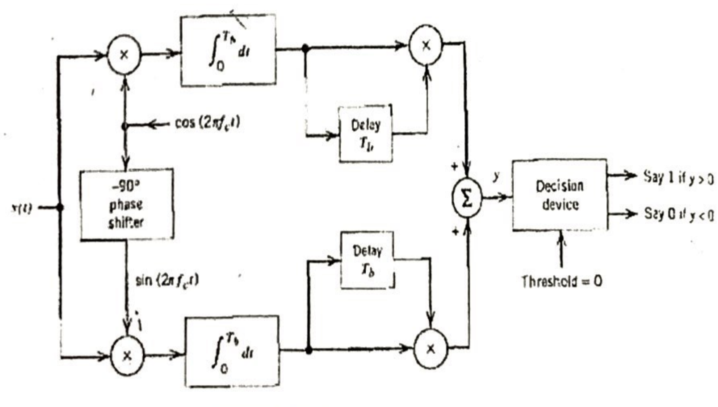
So, the bandwidth of BPSK system $BW = 2f_b = 2 \times 500 = 1000kHz = 1\ MHz$

141	Write a short note on DPSK.	

Differential phase shift keying (DPSK) is differentially coherent modulation method. DPSK does not need a synchronous carrier at the demodulator. The input sequence of binary bits is modified such that the next bit depends upon the previous bit. Therefore in the receiver the previous bits are used to detect the present bit. To do this, an XOR gate is used. The block diagram of DSK modulator is given below:



DPSK Transmitter



DPSK Receiver

In BPSK phase of the carrier changes on both the symbol '1' and '0'. Whereas in DPSK phase of the carrier changes only on symbol '1'. This is the main difference between BPSK and DPSK.

Since one previous bit is always used to define the phase shift in next bit, the symbol can be said to have two bits. Therefore, one symbol duration (T) is equivalent to two bits duration ($2T_b$).

i.e. Symbol duration $T_s = 2T_b$

So, bandwidth is given as, $BW = \frac{2}{T_s} = \frac{1}{Tb}$

$$\therefore BW = f_s$$

Thus the minimum bandwidth in DPSk is equal to f_b; i.e. maximum baseband signal frequency.

DPSK has some advantages over BPSK, but at the same time it has some drawbacks.

Advantages:

(1) DPSK does not need carrier at its receiver. Hence the complicated circuitry for generation of local carrier is avoided.

(2) The bandwidth requirements of DPSK is reduced compared to that of BPSK.

Disadvantages:

(1) The probability of error or bit error rate of DPSK is higher than that of BPSK.

(2) Noise interference in DPSK is more.

142	Write a short note on Quadrature Phase Shift Keying (QPSK)	

In communication system, we know that there are two main resources, i.e. transmission power and channel bandwidth. The channel bandwidth depends upon the bit rate or signaling rate. In digital bandpass transmission, a carrier is used for transmission. This carrier is transmitted over a channel.

If two or more bits are combined in some symbols, then the signaling rate is reduced. Therefore, the frequency of the carrier required is also reduced. This reduces the transmission channel bandwidth. Thus because of grouping of bits in symbols, the transmission channel bandwidth is reduced.

In Quadrature phase shift keying, two successive bits in the data sequence are grouped together. This reduces the bit rate of signaling rate and hence reduces the bandwidth of the channel.

In BPSK, we know that when symbol changes the level, the phase of the carrier is changed by 180°. Since there were only two symbols in BPSK, the phase shift occurs in two levels only.

In QPSK , two successive bits are combined. This combination of two bits forms four distinct symbols. When the symbol is changed to next symbol, the phase of the carrier is changed by 45°($\frac{\pi}{4}$ $radians$). Table shows these symbols and their phase shifts.

Sr.No.	Input successive bits		Symbol	Phase shift in carrier
$i=1$	1(1V)	0(-1V)	S_1	$\pi/4$
$i=2$	0(-1V)	0(-1V)	S_2	$3\pi/4$
$i=3$	0(-1V)	1(1V)	S_3	$5\pi/4$
$i=4$	1(1V)	1(1V)	S_4	$7\pi/4$

Table Symbol and corresponding phase shifts in QPSK

Thus, there are 4 symbols and the phase is shifted by $\pi/4$ for each symbol.

Generation of QPSK signal:

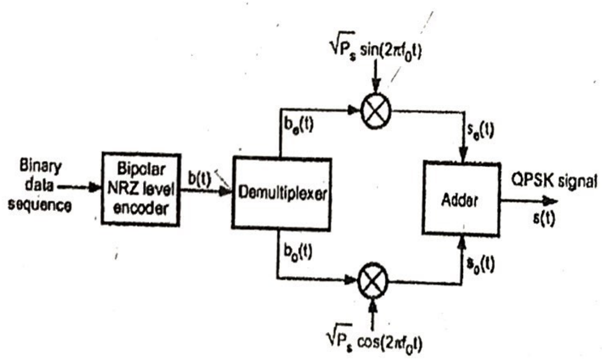


Fig. An offset QPSK transmitter

First of all, the binary bit sequences are converted into bipolar NRZ signal by an encoder. Then, they are sent to a demultiplexer. Demultiplexer separates even and odd numbered bits. The odd numbered bits are modulated by the carrier $\sqrt{p_s}\cos(2\pi f_0 t)$ and the even numbered bits are modulated by the carrier $\sqrt{p_s}\sin(2\pi f_0 t)$. Then, they are added by an adder and thus QPSK signal is generated.

The phasor diagram of QPSK signal:

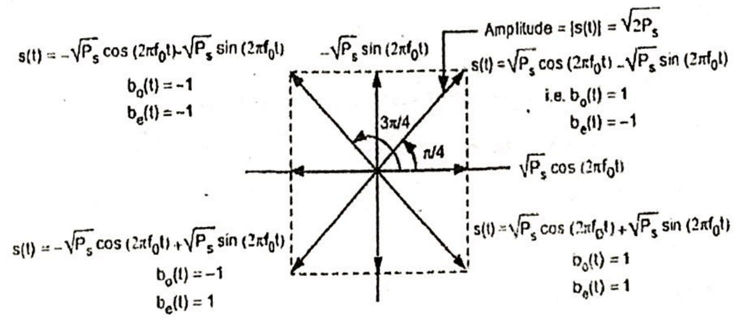


Fig. Phasor diagram of QPSK signal

Bandwidth of QPSK signal:

We have seen that the bandwidth of BPSK signal is equal of $2f_b$. Here $T_b = \frac{1}{f_b}$ is the one bit period.

In QPSK, the two waveforms, even numbered bit sequences, $b_e(t)$ and odd numbered bit sequences, $b_o(t)$ from the base band signal is used. One bit period for both of these signals is equal to $2T_b$ where T_b is the bit duration. Therefore the bandwidth of QPSK signals is

$$BW = 2 \times \frac{1}{2T_b} = \frac{1}{T_b} = f_b$$

Here, f_b is indicating the bit rate.

Thus the bandwidth of QPSK signal is half of the bandwidth of BPSK signal. But, the noise is same for the BPSK and QPSK signal.

Advantages of QPSK:

1. For the same bit error rate and noise, the bandwidth required by QPSK is to half as compared to BPSK.
2. Because of reduced bandwidth, information transmission rate of QPSK is higher.
3. Carrier power almost remains same.

143	**In a QPSK system, bit rate of NRZ stream is 10 Mbps and carrier frequenc is 1 GHz. Find the symbol rate of transmission and bandwidth requirements of the channel. Sketch the power spectral density of the QPSK signal.**	

As we know, $T_b = bit\ duration\ i.e.time\ required\ for\ one\ bit = \frac{1}{f_b}$

So, $f_b = \frac{1}{T_b} = bit\ rate\ i.e.no.of\ bits\ transmitted\ in\ one\ second$

So, given that, bit rate $f_b = 10\ Mbps$

We know that, Bandwidth of the QPSK system is equal to bit rate.

$BW = f_b = 10\ MHz$

Now, symbol duration and bit duration are related as, $T_s = 2T_b; where\ T_s\ is\ symbol\ duration$

This is because; two successive bits are combined and transmitted together.

$\therefore Symbol\ rate = \frac{1}{T_s} = \frac{1}{2T_b} = \frac{f_b}{2} = \frac{10}{2} = 5\ MHz$ (ঐকিক নিয়মের মত ভাবলে সিম্বল রেট বুঝা যাবে।)

Now, the power spectral density:

Given that carrier frequency, $f_0 = 1\ GHz$

We need to remember that the power spectral density of QPSK system is a sinc function.

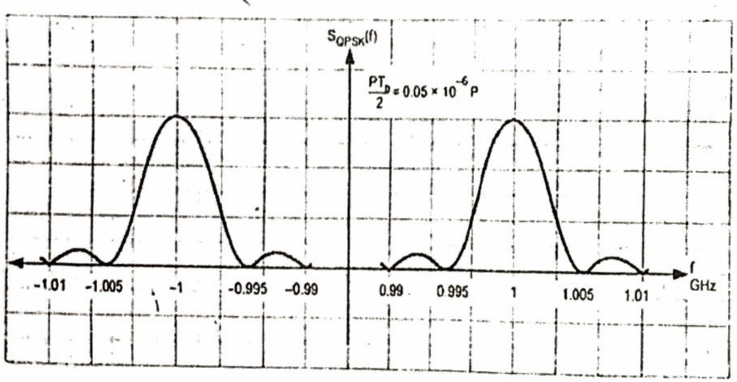


Fig. Spectral density plot of QPSK

প্রথমে $\pm f_0$ অর্থাৎ $\pm 1\ GHz$ এ দুইটা সিংক ফাংশন আঁকতে হবে। আমাদের মনে রাখতে হবে, যে আমরা এখানে ব্যান্ডউইথ পেয়েছি, ১০ মেগাহার্জ। তাহলে পজিটিভ সাইডের ক্ষেত্রে, সিংক ফাংশনের প্রধান লোব হবে (1 GHz – 5 MHz) = 0.995 GHz হতে (1 GHz + 5 MHz) = 1.005 GHz পর্যন্ত।

144	Write a short note on M-ary PSK.	

In binary PSK, only one bit ('0' or '1') to be transmitted. So, the symbol was two types. That's why it was binary PSK.

In QPSK, two bits are combined and two bits can create four symbol. That's why it was called Quadrature PSK.

Now, if N bits are combined and N bits can create $2^N = M$ symbol, then it is called M-ary PSK. QPSK is actually a special type of 4-ary PSK.

Phase shift in BPSK $= \frac{2\pi}{number\ of\ symbols} = \frac{2\pi}{2} = \pi\ or\ 180°$

Phase shift in QPSK $= \frac{2\pi}{number\ of\ symbols} = \frac{2\pi}{4} = \frac{\pi}{2}\ or\ 90°$

Phase shift in M-ary PSK $= \frac{2\pi}{number\ of\ symbols} = \frac{2\pi}{M}$

Again,

Bit duratation $= T_b$

So, bit rate $= \frac{1}{f_b}$

Now, the duration of each symbol in BPSK, $T_s = T_b$

The duration of each symbol in QPSK, $T_s = 2T_b$

The duration of each symbol in M-ary PSK, $T_s = NT_b$

Power spectral density of M-ary PSK:

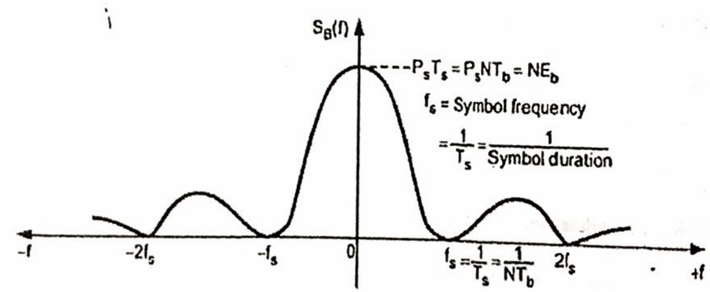


Fig. Plot of power spectral density of baseband M-ary PSK signal

So, the bandwidth of M-ary PSK, $BW = 2f_s$

$$= \frac{2}{T_s} \quad ;\left(\because f_s = \frac{1}{T_s} = \frac{1}{symbol\ period}\right)$$

$$= \frac{2}{NT_b} \quad ;(\because T_s = NT_b)$$

$$= \frac{2f_b}{N} \quad ;\left(\because f_b = \frac{1}{T_b}\right)$$

It shows that, as the number of successive bits (N) per symbol are increased, the bandwidth reduces.

145	**In digital CW communication system, the bit rate of NRZ data stream is 1 Mbps and carrier frequency is 100 MHz. Find the symbol rate of transmission and bandwidth requirement of the channel in the following cases of different techniques uses.** **a. BPSK system** **b. QPSK system** **c. 16-ary PSK system**	

Given that, bit rate, $f_b = 1\ Mbps = 1 \times 10^6\ bps$

So, the bit duration, $T_b = \frac{1}{f_b} = \frac{1}{1\times10^6} = 1 \times 10^{-6} = 1\mu s$

Carrier frequency, $f_0 = 100\ MHz$

Bandwidth in BPSK system, $BW = 2f_b = 2 \times 1 = 2\ MHz$

In BPSK, one bit is considered as one symbol.

Hence, Symbol duration, $T_s = T_b = 1 \times 10^{-6} = 1\mu s$

$\therefore symbol\ rate = \frac{1}{T_s} = \frac{1}{1\times10^{-6}} = 1 \times 10^6 symbol/sec$

Note, that, in BPSK, symbol rate or transmission rate is same as bit rate.

For QPSK system:

Bandwidth in QPSK system, $BW = f_b = 1 \times 10^6 = 1\ MHz$

In QPSK, symbol duration, $T_s = 2T_b = 2 \times 1 \times 10^{-6} = 2\ \mu s$

Hence, transmission rate or symbol rate $= \frac{1}{T_s} = \frac{1}{2\times10^{-6}} = 500 \times 10^3\ symbol/sec$

16-ary PSK system:

Bandwidth in M-ary PSK, $BW = \frac{2f_b}{N}$

Now, given, M = 16. So, $M = 2^N$

$$Or,\ N = \log_2 M$$

$$\therefore N = \log_2 16 = 4$$

Thus, 4 bits can make 16 symbol. So, the bandwidth of the 16 ary PSK

$$BW = \frac{2f_b}{N} = \frac{2\times1\times10^6}{4} = 500\ kHz$$

$Symbol\ duration\ for\ M-ary\ PSK, T_s = NT_b = 4 \times 1 \times 10^{-6} = 4\ \mu s$

$\therefore symbol\ rate = \frac{1}{T_s} = \frac{1}{4\times10^{-6}} = 250 \times 10^3 symbols/sec$

146	**Assume that you are required to transmit $f_b = 90\ Mbps$ data in an authorized bandwidth of 20 MHz. Whilch modulation would you consider? Explain why?**	

For this transmission, M-ary PSK modulation method can be used.

The minimum bandwidth of M-ary PSK is given as, $BW \leq \frac{2f_b}{N}$

Now, given, $BW = 20 \times 10^6 Hz\ and\ f_b = 90\ Mbps = 90 \times 10^6$ Hz

Here, data rate is equal to the maximum frequency f_b

Putting these values in above equation, $20 \times 10^6 \leq \frac{2\times90\times10^6}{N}$

$$or, N \leq \frac{2\times90\times10^6}{20\times10^6}$$

$$\therefore N \leq 9\ bits$$

Now, $M = 2^N = 2^9 = 512$

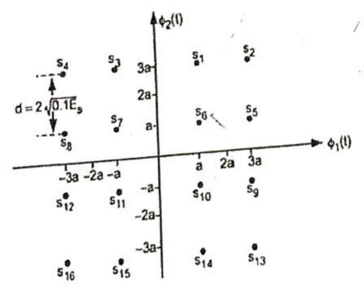
This shows that 512 ary PSK can transmit the 90 MHz signal in the bandwidth of 20 MHz.

147	Write a short note on QAM.

Quadrature Amplitude Modulation or QAM (also called, Quadrature Amplitude Phase shift keying or QASK) is a form of modulation which is widely used for modulation data signals onto a carrier used for radio communications. It is widely used because it offers advantages over other forms of data modulation such as PSK.

Quadrature Amplitude Modulation, QAM is a signal is which two carriers shifted in phase by 90 degrees are modulated and the resultant output consists of both amplitude and phase variations. So, it may also be considered as a mixture of amplitude and phase modulation.

Let us consider the case of 4 bit symbol. Then there will be $2^4 = 16$ possible symbols. In the QASK system, such 16 symbols are represented geometrically shown in figure.



Geometrical representation of 16 signals in QASK system

Transmitter of QASK:

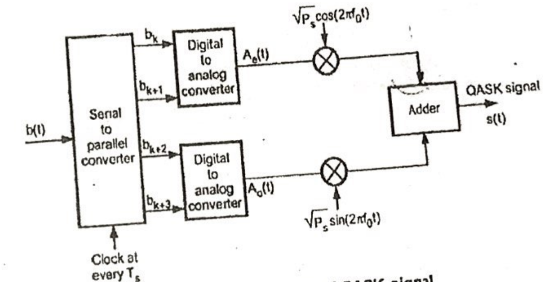


Fig. Generation of QASK signal

Power spectral density of QAM:

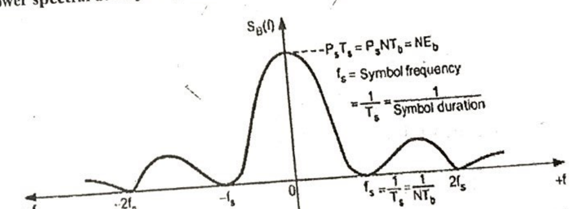


Fig. Plot of power spectral density of QAM signal

Therefore, the bandwidth will be,

$$BW = f_s - (-f_s) = 2f_s$$

$$\text{Or, } BW = \frac{2}{T_s} \quad ; since\ f_s = \frac{1}{Ts}$$

$$= \frac{2}{NT_b} \quad ; [since\ T_s = NT_b]$$

$$= \frac{2f_b}{N} \quad ; [since\ f_b = \frac{1}{T_b}]$$

Thus the power spectral and bandwidth of QAM is similar to that of M-ary PSK.

148	Write a short note on FSK.

In binary frequency shift keying, the frequency of the carrier is shifted according to the binary symbol. The phase of the carrier is unaffected. That is we have two different frequency signals according to binary symbols. Let there be a frequency by Ω. Then, we can write following equations:

If b(t) = '1'; $S_H(t) = \sqrt{2P_s}\cos(2\pi f_0 + \Omega)\,t$

If b(t) = '0'; $S_L(t) = \sqrt{2P_s}\cos(2\pi f_0 + \Omega)\,t$

Thus there is an increase of decrease in frequency by Ω. Let us use the following conversion table to combine above two FSK equations.

b (*t*) Input	*d* (*t*)	P_H (*t*)	P_L (*t*)
1	+1 V	+1 V	0 V
0	-1 V	0 V	+1 V

Table Conversion table for BFSK representation

So, we can write above two equation as:

$S(t) = \sqrt{2P_s}\cos[(2\pi f_0 + d(t)\Omega)t]$

Thus when symbol '1' is to be transmitted, the carrier frequency will be $\left(f_0 + \frac{\Omega}{2\pi}\right)$. If symbol '0' is to be transmitted, the carrier frequency will be $\left(f_0 - \frac{\Omega}{2\pi}\right)$ i.e.

$$f_H = f_0 + \frac{\Omega}{2\pi} \qquad for\ symbol\ '1'$$

$$f_L = f_0 - \frac{\Omega}{2\pi} \qquad for\ symbol\ '0'$$

Block diagram of BFSK transmitter:

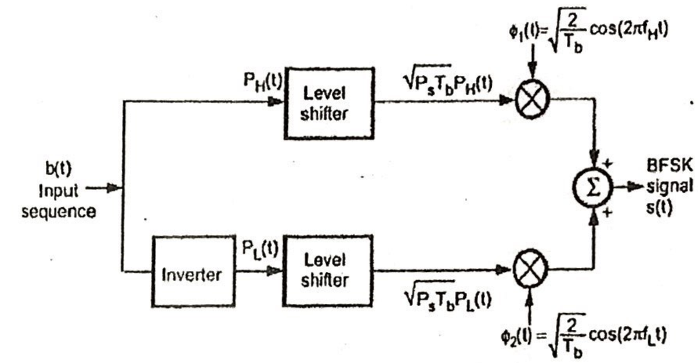


Fig. Block diagram of BFSK transmitter

The binary FSK signal is shown below:

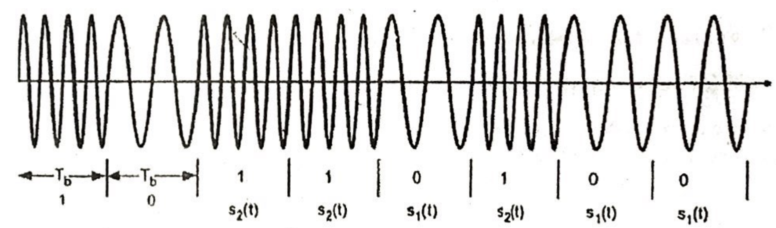


Fig. BFSK signal

Power spectral density of BFSK:

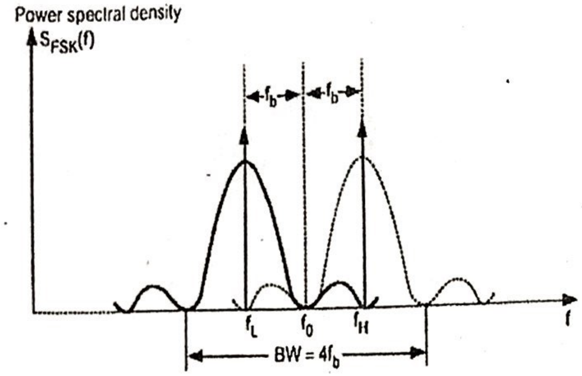


Fig. Power spectral density of BFSK signal

It is clear from the above figure that the width of one lobe is $2f_b$. The main lobes due to f_H *and* f_L are place such that the total width due to both main lobes is $4f_b$.

Bandwidth of BFSK = $2f_b + 2f_b = 4f_b$

If we compare this **bandwidth** with that of BPSK , we observe that

$$BW\ (BFSK) = 2 \times BW(BPSK)$$

So, the main disadvantage of BFSK is that its bandwidth is double the bandwidth of BPSK. Besides, the error rate of BFSK is more compared to BPSK.

Note: BPSK is for only two symbols. This principle can be extended further to 'N' successive bits. These 'N' bits form $2^N = M$ different symbols. Every symbol uses separate frequency for transmission. Such system is called M-ary FSK system.

149	Write a short note on ASK.	

Amplitude shift keying (ASK) or ON-OFF keying (OOK) is the simplest digital modulation technique. In this method, there is only one unit energy carrier and it is switched on or off depending upon the input binary sequence. The ASK wave form can be represented as,

$s(t) = \sqrt{2P_s}\cos(2\pi f_0 t)$; to transmit '1' .

To transmit symbol '0', the signal s(t) = 0. That is no signal is transmitted. S(t) contains some comple cycles of carrier frequency 'f'. Thus,

Symbol '1' $\Rightarrow$ pulse is transmitted,

Symbol '0' $\Rightarrow$ no pulse is transmitted.

Thus the ASK waveform looks like an ON-OFF of the signal. Hence it is also called ON-OFF keying (OOK).

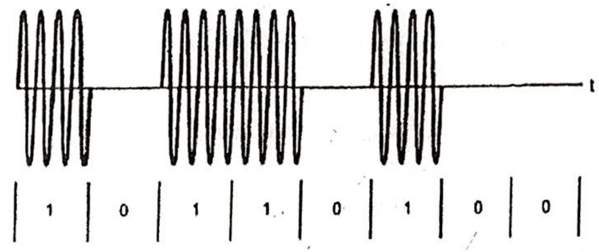
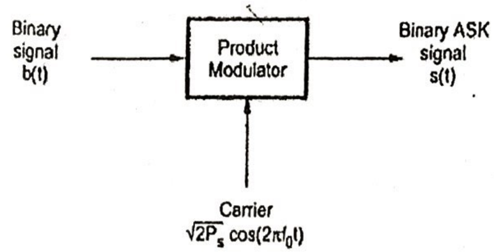


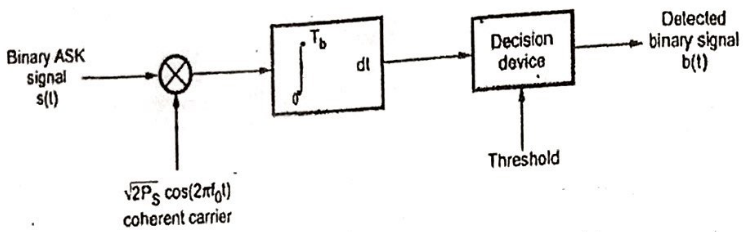
Fig. ASK waveform

ASK generator or modulator:



Block diagram of ASK generator

ASK detector:



Block diagram of coherent ASK detector

Multiplexing

150	Write a short note on FDM.	30th BCS

Frequency division multiplexing (FDM) is an **analog technique** that can be applied when the bandwidth of a link is greater than the combined bandwidth of the signals to be transmitted. In FDM, signals generated by each sending device is modulated with different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. Channels can be separated by unused bandwidth guard bands to prevent signals from overlapping. In the demodulator, the original signals are separated from the composite signal through filtering.

In addition carrier frequencies must not interfere with the original data frequencies. We consider FDM to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal before FDM is used to multiplex them.

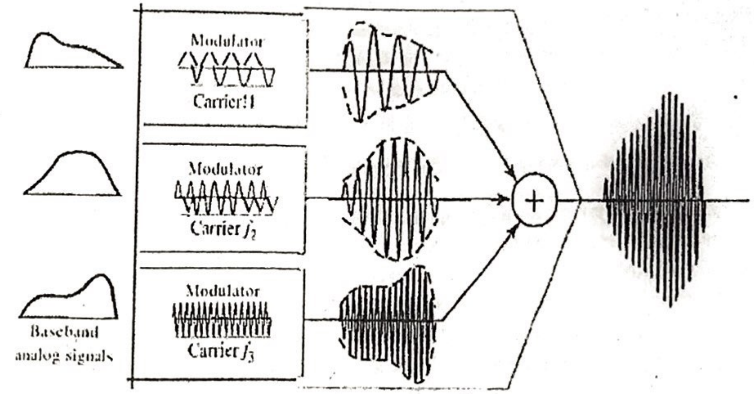


Fig:- FDM process (producing composite signal using different carriers for different signals)

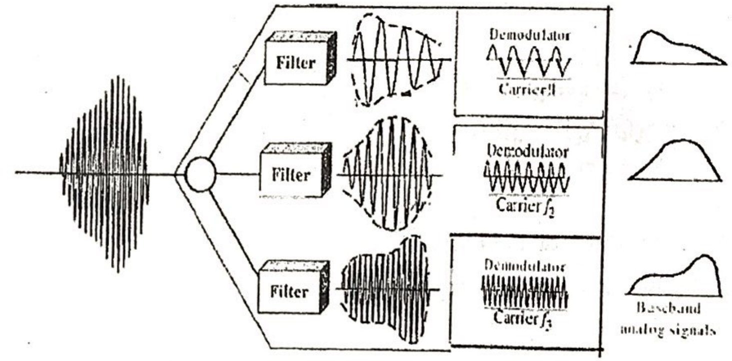


Fig:- FDM Demultiplexing (separating original signals)

Consider an example: Assume that a voice channel occupies a bandwidth of 4 kHz; we need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Assume there are no guard bands.

First we modulate each of three voice channels to a different bandwidth. We use the 20 to 24 kHz bandwidth for the first channel, the 24 to 28 kHz bandwidth for the second channel, and the 28 to 32 kHz bandwidth for the third one.

Then we combine them.

At the receiver, each channel receives the entire signal, using a filter to separate out its own signal. The first channel uses a filter that passes frequencies between 20 and 24 kHz and filters out (discards) any other frequencies. The second channel uses a filter that passes frequencies between 24 and 28 kHz, and the third channel uses a filter that passes frequencies between 28 and 32 kHz.

Each channel is then demodulated so that the original signal can be found.

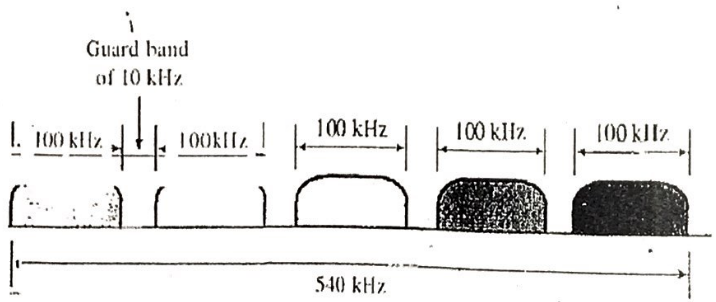
Application: A very common application of FDM is AM and FM radio broadcasting. Radio uses the air as the transmission medium. A special band from 530 to 1700 kHz is assigned to AM radio. Each AM station needs 10 kHz. However, FM has a wider band of 88 to 108 MHz because each station needs a bandwidth of 200 kHz. Besides, the first generation of cellular telephones also uses FDM. Each user is assigned two 30 kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM. Remember that an FM signal has a bandwidth 10 times that of the modulating signal, which means each channel has 30 kHz (10 × 3) of bandwidth. Therefore, each user is given, by the base station, a 60 kHz bandwidth in a range available at the time of the call.

Another common use of FDM is in television broadcasting. Each TV channel has its own bandwidth of 6 M Hz.

151	**Five channels, each with 100 kHz bandwidth, are to be Frequency division multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channel to prevent interference?**	

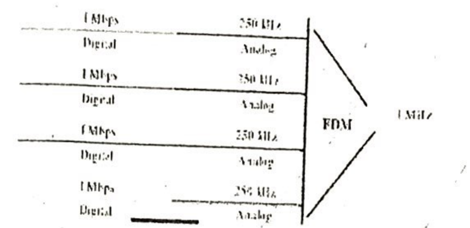
For five channels, we need at least four guard bands.

This means that the required bandwidth is at least, $= 5 \times 100 + 4 \times 10 = 540\ kHz$



152	**Four data channels (digital), each transmitting at 1 Mbps, use a satellite channel of 1 MHz. Design an appropriate configuration, using FDM.**	

The satellite channel is analog. We divide it into four channels, each channel having a 250 kHz bandwidth. Each digital channel of 1 Mbps is modulated such that each 4 bits is modulated to 1 Hz. It is found by $\frac{1Mbps}{250\ kHz} = 4$; . One solution is 16 QAM modulation.



153	**The Advanced Mobile Phone System (AMPS) uses two bands. The first band of 824 to 849 MHz is used for sending, and 869 to 894 MHz is used for receiving. Each user has a bandwidth of 30 kHz in each direction. The 3 kHz voice is modulating using FM, creating 30 kHz of modulating signal. How many people can use their cellular phone simultaneously?**	

Each band is 25 MHz (i.e. $849 - 824 = 25$ or $894 - 869 = 25$).

So, the total number of channels $= \frac{25\ MHz}{30\ kHz} = 833.33$

In reality, the band is divided into 832 channels. Of these, 42 channels are used for control. Which means only 832 – 42 =790 channels are available for cellular phone user.

154	**Write a short note on TDM.**	**30th BCS**

Time Division Multiplexing (TDM) is a digital multiplexing technique for combining several low rate channels into one high rate one.

In other word, Time division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link.

We also need to remember that TDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into a one timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

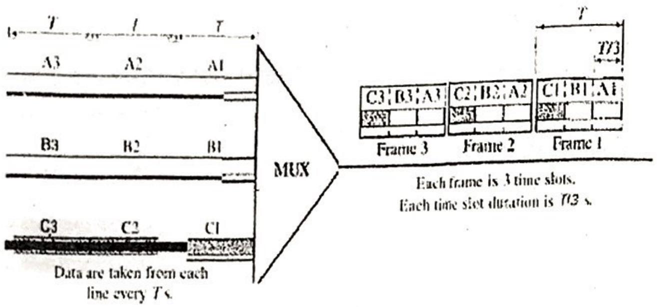


Fig: Synchronous Time Division Multiplexing

TDM Time Slot Frame n

TDM এর জন্য সিনক্রোনাইজেশন খুবই জরুরী। নয়ত ডিমাল্টিপ্লেক্সিং এর সময় এক চ্যানেলের ডাটা অন্য চ্যানেলে চলে যাবার সম্ভাবনা থাকে। তাই TDM করার সময় অবশ্যই সিনক্রোনাইজেশন করতে হবে। সিনক্রোনাইজেশন বুঝতে হলে কতগুলো টার্ম বুঝতে হবে প্রথমে।

TDM এর ইনপুটের ডাটা গুলোকে কতগুলো ইউনিটে ভাগ করা হয়। এই ইউনিট একটা বিট হতে পাওে বা একটা ক্যারেক্টার হতে পারে অথবা একটা ডাটা ব্লকও হতে পারে। এই প্রতিটা ইউনিটের যে টাইম লাগে, তাকে বলা হয় one input time slot. আবার TDM করার পরে প্রতিটা ইনপুট ইউনিট একটি আউটপুট ইউনিটে পরিনত হয়। এই আউটপুট ইউনিটের যে টাইম লাগে তাকে বলা হয়, one output time slot. এই আউটপুট টাইম স্লট ইনপুট টাইম স্লটের চেয়ে n গুন কম হয়।এখানে n হল নাম্বার অব কানেকশন। অর্থ্যাৎ ইনপুট টাইম স্লটকে কানেকশন নাম্বার দ্বারা ভাগ করলে আউটপুট টাইম স্লট পাওয়া যায়। অন্যভাবে বলা যায়, আউটপুট টাইম স্লট এর টাইম ডিউরেশন কম। মানে এটি খুব দ্রুত ট্রাভেল করে। সময় কম লাগা মানে ত দ্রুত ট্রাভেল করা।

এখন প্রতিটা ইনপুট কানেকশন থেকে একটি একটি করে ইউনিট নিয়ে একটি ফ্রেম গঠিত হয়। যদি আমাদের n সংখ্যক কানেকশন থাকে, তবে একটি ফ্রেমে n সংখ্যক টাইম স্লট থাকে এবং প্রতিটা টাইম স্লটে একটি করে ইউনিট থাকে (কারন প্রতিটা কানেকশন থেকে একটি করে ইউনিট নেওয়া হয়)। এইগুলোই হল আউটপুট টাইম স্লট। এই আউট টাইম স্লটের ডিউরেশন ইনপুট স্লটের ডিউরেশনের চেয়ে n গুন কম যেখানে n হল মোট কানেকশন সংখ্যা।

$$Duration\ of\ output\ time\ slot = \frac{duration\ of\ input\ time\ slot}{total\ number\ of\ connection} = \frac{T}{n}$$

Duration of input time slot is the same thing as the duration of input unit. Because, One unit is allocated one time slot.

আবার মনে রাখতে হবে, ফ্রেম ডিউরেশন ইনপুট ইউনিটের ডিউরেশনের সমান। অর্থ্যাৎ একটা ইনপুট ইউনিটের বা একটি টাইম স্লটের যে টাইম লাগে সেই টাইমের ভিতরে আউটপুটে একটি ফ্রেমের সেই টাইম লাগে। That is T is the duration of

frame unless a frame carries some other information. এইটা শুধু তখন যখন ফ্রেমের মধ্যে দিয়ে অন্য কোনো ইনফরমেশন আর যায় না। কারন আমরা পরে দেখব, সিনক্রোনাইজেশনের জন্য প্রতিটি ফ্রেমের শুরুতে একটি করে এক্সট্রা একটা বিট যুক্ত করা হয়।

এখন আসি ডাটা রেটের বেলায়। যেহেতু আউটপুটে প্রতি ইউনিটের টাইম ডিউরেশন n কমে যায়, তাই আউটপুটের ডাটা রেট ইনপুটের ডাটা রেটের চেয়ে n বেড়ে যাবে।

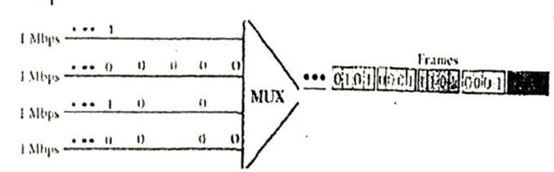
মনে রাখতে হবে, সিনক্রোনাস টিডিএম এর ক্ষেত্রে, মাল্টিপ্লেক্সিং এর পর ডাটা রেট n গুন বেড়ে যায় আর টাইম ডিউরেশন n গুন কমে যায়। আর ফ্রেম ডিউরেশন ইনপুট ডিউরেশনের সমান থাকে। একটি ফ্রেমে স্লটের সংখ্যা মোট কানেকশন সংখ্যার সমান। প্রতিটি স্লটে একটি করে ইউনিট থাকে। ইউনিট একটি বিট বা ক্যারেক্টার বা ডাটা ব্লক হতে পারে।

155	**Let 3 channel with 1 kbps data rate are to be Time Division Multiplexed (TDM). If 1 bit at a time is multiplexed (i.e. a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot and (c) each frame?**	

a. Given that data rate of each input connection is 1 kbps. মানে প্রতিটা ইনপুট চ্যানেল প্রতি সেকেন্ডে ১০০০ বিট করে ডাটা ট্রান্সফার করে। এখন প্রশ্নে বলা আছে, প্রতিটা ইউনিটে একটি করে বিট থাকে। তাহলে প্রতিটা টাইম স্লটে একটি করে বিট ট্রান্সফার হবে। তাহলে একটা বিট ট্রান্সফার হতে যে সময় লাগে সেটাই হল ইনপুট স্লট ডিউরেশন। অর্থ্যাৎ এখানে ইনপুট বিট ডিউরেশন আর ইনপুট স্লট ডিউরেশন একই। তাহলে একটু ঐকিক নিয়মের মত চিন্তা করি। প্রশ্ন মতে, ইনপুটে ১০০০ বিট ট্রান্সফার হতে সময় লাগে ১ সেকেন্ড। তাহলে একটি বিট ট্রান্সফার হতে সময় লাগে $\frac{1}{1000} = 0.001s = 1ms$ । তাহলে ইনপুট স্লট ডিউরেশন হল ১ মিলি সেকেন্ড।

b. আমরা জানি আউটপুট স্লট ডিউরেশন হবে ইনপুট স্লট ডিউরেশনের চেয়ে n গুন কম যেখানে n মোট কানেকশন সংখ্যা। এখানে মোট কানেকশন সংখ্যা হল n=3 । তাহলে আউটপুট স্লট ডিউরেশন হবে "$= \frac{1}{3}ms$

c. আবার আমরা জানি, ফ্রেম ডিউরেশন ইনপুট স্ল ট ডিউরেশনের সমান হবে। অথবা আউটপুট স্লট ডিউরেশনের চেয়ে n গুন বেশী হবে। তাহলে প্রতি ফ্রেমের ডিউরেশন = 1ms

156	**Figure shows synchronous TDM with a data stream for each input and one data stream for output. The unit of data is 1 bit. Find (a) the input bit duration (b) the output bit duration (c) the output bit rate and (d) the output frame rate.** 	

a. Given that each connection of input has data rate 1 Mbps. মানে প্রতি সেকেন্ডে ১,০০০,০০০ বিট ট্রান্সফার করতে পারে। ইনপুট বিট ডিউরেশন বের করতে বলা হয়েছে। ঐকিক নিয়মের মত করি। ১,০০০,০০০ বিট ট্রান্সফার করে ১ সেকেন্ডে, তাহলে ১ টি বিট ট্রান্সফার করে $\frac{1}{1Mbps} = 1\mu s$. অর্থ্যাৎ ইনপুট বিট ডিউরেশন হল ১ মাইক্রোসেকেন্ড।

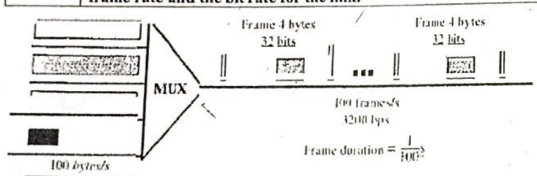
b. যেহেতু প্রতি ইউনিটে ১ টি করে বিট ধরা হয়েছে। তাই প্রতি স্লটে একটি করে বিট থাকবে। ফলে বিট ডিউরেশন আর স্লট ডিউরেশন সমান। তাই ইনপুট বিট ডিউরেশনই হল ইনপুট স্লট ডিউরেশন। আর আউটপুট বিট ডিউরেশন হবে আউটপুট স্লট ডিউরেশনের সমান। আমরা জানি আউটপুট স্লট ডিউরেশন হবে ইনপুট স্লট ডিউরেশনের চেয়ে n গুন কম যেখানে n মোট কানেকশন সংখ্যা। এখানে মোট কানেকশন সংখ্যা হল n=4 । তাহলে আউটপুট স্লট ডিউরেশন হবে "$= \frac{1}{4}\mu s$ ।

সুতরাং আউটপুট বিট ডিউরেশন "$= \frac{1}{4}\mu s$ ।

c. আমরা জানি আউটপুট বিট রেট ইনপুট বিট রেটের n গুন হবে; যেখানে n হল মোট কানেকশন সংখ্যা। এখানে $n = 4$. তাই আউটপুট বিট রেট হবে $4 \times 1\,Mbps = 4\,Mbps$ । এটাকে আবার আউটপুট বিট রেট থেকেও বের করা যায়। $\frac{1}{4}\mu s$ এ বিট ট্রান্সফার হয় ১ টি। তাহলে ১ সেকেন্ডে ট্রান্সফার হবে $4\,Mbps$

d. আবার আমরা জানি ফ্রেম রেট সব সময় ইনপুট রেটের সমান। তাহলে ফ্রেম রেট হবে ১,০০০,০০০ ফ্রেম / সেকেন্ড। কারন আমরা প্রতি ফ্রেমে ৪ টি করে বিট পাঠাচ্ছি।

157	**Four channels are multiplexed using TDM. If each channel sends 100 bytes / second and we multiplex 1 byte per channel, show the frame traveling on the link, size of the frame, the duration of the frame, the frame rate and the bit rate for the link.**	



Given that each channel sends 100 bytes per second. অর্থাৎ এখানে ইউনিট বাইটে দেওয়া আছে । আগের অংক গুলোতে ইউনিট বিটে দেওয়া ছিল। তাহলে এখানে এক ইউনিটে ৮ টি বিট আছে।

এখন টিডিএম এর নিয়ম অনুসারে, প্রতিটি চ্যানেল থেকে একটি করে ইউনিট নিয়ে একটি করে ফ্রেম গঠিত হবে। So here, each frame carries 1 byte from each channel. So, the size of each frame is equal to 4 bytes or $4 \times 8 = 32$ bits.

আমরা জানি, ফ্রেম ডিউরেশন ইনপুট ডিউরেশনের সমান। So, frame duration $= \frac{1}{100} s$ । কারন, ১০০ বাইট যেতে ১ সেকেন্ড লাগলে ১ বাইট যেতে সময় লাগবে ১/১০০ সেকেন্ড।

আবার ফ্রেম রেট হবে ইনপুট রেটের সমান। So, frame rate = 100 frames per second

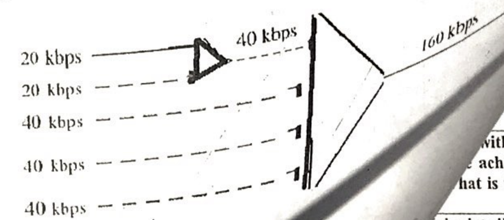
আবার সিনক্রোনাস টিডিএম এর ক্ষেত্রে, মাল্টিপ্লেক্সিং এর পর ডাটা রেট n গুন বেড়ে যায়।

Input data rate = 100 bytes per second = $100 \times 8 = 800$ bit / sec

So, output data rate = n × input data rate = $4 \times 800 = 3200$ bps

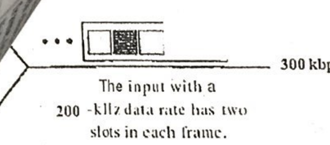
Note on data management in TDM: In all our previous discussion, we assumed that the data rates of all inputs were same. However, if data rates are not the same, there are three combination can be used: They are: (a) multilevel multiplexing, (b) multiple –slot allocation and (c) pulse stuffing.

Multilevel multiplexing: Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of other. For example, in figure we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to the last three. A second level of multiplexing can creat an output of 160 kbps.



Multiple slot allocation: Sometimes it is m…
a single input line. For example, we …
of another input. In figure, th…
output. We insert a seri…

…iple slot allocation technique.
…he second channel. Each frame



…he input rate. So, the frame rate = 100,000 frame per seconds.
…the first channel.

$\frac{1}{100,000}$ seconds per frame = 10 ms

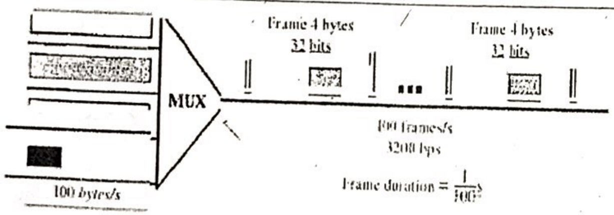
…000 = 300 kbps

b. যেহেতু প্রতি ইউনিটে ১ টি করে বিট ধরা হয়েছে। তাই প্রতি স্লটে একটি করে বিট থাকবে। ফলে বিট ডিউরেশন আর স্লট ডিউরেশন সমান। তাই ইনপুট বিট ডিউরেশনই হল ইনপুট স্লট ডিউরেশন। আর আউটপুট বিট ডিউরেশন হবে আউটপুট স্লট ডিউরেশনের সমান। আমরা জানি আউটপুট স্লট ডিউরেশন হবে ইনপুট স্লট ডিউরেশনের চেয়ে n গুন কম যেখানে n মোট কানেকশন সংখ্যা। এখানে মোট কানেকশন সংখ্যা হল n=4 । তাহলে আউটপুট স্লট ডিউরেশন হবে "$= \frac{1}{4}\mu s$ ।
সুতরাং আউটপুট বিট ডিউরেশন "$= \frac{1}{4}\mu s$ ।

c. আমরা জানি আউটপুট বিট রেট ইনপুট বিট রেটের n গুন হবে; যেখানে n হল মোট কানেকশন সংখ্যা। এখানে $n = 4$. তাই আউটপুট বিট রেট হবে $4 \times 1\, Mbps = 4\, Mbps$ । এটাকে আবার আউটপুট বিট রেট থেকেও বের করা যায়। $\frac{1}{4}\mu s$ এ বিট ট্রান্সফার হয় ১ টি। তাহলে ১ সেকেন্ডে ট্রান্সফার হবে $4\, Mbps$

d. আবার আমরা জানি ফ্রেম রেট সব সময় ইনপুট রেটের সমান। তাহলে ফ্রেম রেট হবে ১,০০০,০০০ ফ্রেম / সেকেন্ড। কারন আমরা প্রতি ফ্রেমে ৪ টি করে বিট পাঠাচ্ছি।

157	**Four channels are multiplexed using TDM. If each channel sends 100 bytes / second and we multiplex 1 byte per channel, show the frame traveling on the link, size of the frame, the duration of the frame, the frame rate and the bit rate for the link.**	



Given that each channel sends 100 bytes per second. অর্থাৎ এখানে ইউনিট বাইটে দেওয়া আছে । আগের অংক গুলোতে ইউনিট বিটে দেওয়া ছিল। তাহলে এখানে এক ইউনিটে ৮ টি বিট আছে।

এখন টিডিএম এর নিয়ম অনুসারে, প্রতিটি চ্যানেল থেকে একটি করে ইউনিট নিয়ে একটি করে ফ্রেম গঠিত হবে। So here, each frame carries 1 byte from each channel. So, the size of each frame is equal to 4 bytes or $4 \times 8 =$ 32 bits.

আমরা জানি, ফ্রেম ডিউরেশন ইনপুট ডিউরেশনের সমান। So, frame duration $= \frac{1}{100} s$ । কারন, ১০০ বাইট যেতে ১ সেকেন্ড লাগলে ১ বাইট যেতে সময় লাগবে ১/১০০ সেকেন্ড।

আবার ফ্রেম রেট হবে ইনপুট রেটের সমান। So, frame rate = 100 frames per second

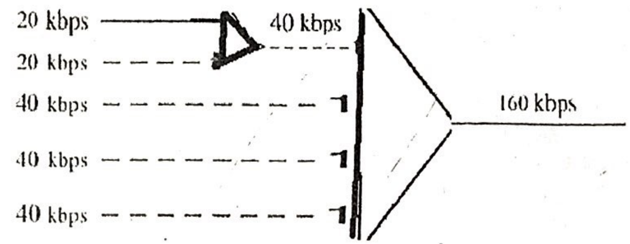
আবার সিনক্রোনাস টিডিএম এর ক্ষেত্রে, মাল্টিপ্লেক্সিং এর পর ডাটা রেট n গুন বেড়ে যায়।

Input data rate = 100 bytes per second $= 100 \times 8 = 800$ bit / sec

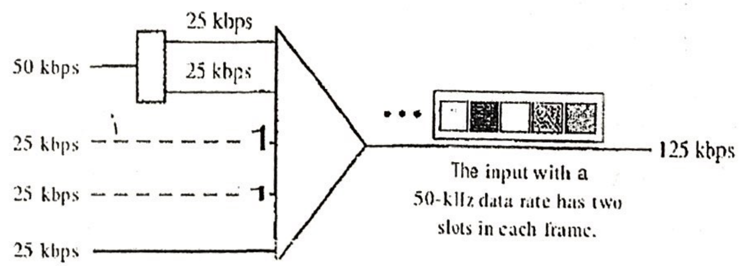
So, output data rate = n × input data rate $= 4 \times 800 = 3200$ bps

Note on data management in TDM: In all our previous discussion, we assumed that the data rates of all inputs were same. However, if data rates are not the same, there are three combination can be used: They are: (a) multilevel multiplexing, (b) multiple –slot allocation and (c) pulse stuffing.

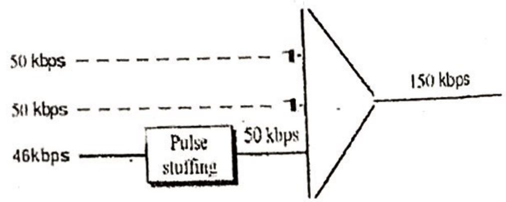
Multilevel multiplexing: Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of other. For example, in figure we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to the last three. A second level of multiplexing can creat an output of 160 kbps.



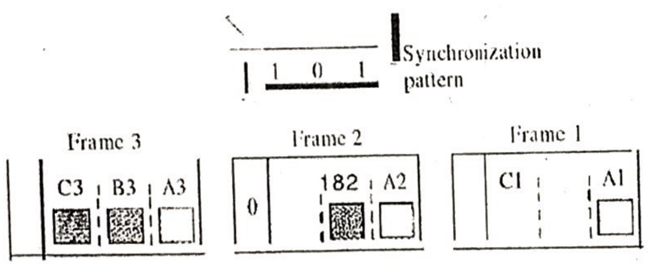
Multiple slot allocation: Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For example, we might have an input line that has a data rate that is a multiple of another input. In figure, the input line with a 50 kbps data rate can be given two slots in the output. We insert a serial to parallel converter in the line to make two input out of one.



Pulse stuffing: Sometimes the bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is to make the highest input data rate the dominant data rate and then add dummy bits to the input lines with lower rates. This will increase their rates. This technique is called pulse stuffing, bit padding or bit stuffing. In figure, the input with a data rate of 46 is pulse stuffed to increase the rate to 50 kbps.



Frame Synchronizing: The implementation of TDM is not as simple as that of FDM. Synchronization between the multiplexer and demultiplexer is a major issue. If the , multiplexer and the demultiplexer are not synchronized, a bit belonging to one channel may be received by the wrong channel. For this reason, one or more synchronization bits are usually added to the beginning of each frame. These bits, called framing bits, follow a pattern, frame to frame, that allows the demultiplexer to synchronize with the incoming stream so that it can separate the time slots accurately. In most cases, this synchronization information consists of 1 bit per frame, altering between 0 an 1.



158	**We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronization bit is added to each frame, find (a) the data rate of each source, (b) the duration each character in each source, (c) frame rate (d) the duration of each frame, (e) the number of bits in each frame and (f) the data rate of the link.**	

a. The data rate of each source = 250× 8 = 2000 bps = 2 kpbs. Because each character contains 8 bits.

b. Each source sends 250 characters per second; therefore, the duration of a character is $\frac{1}{250}s$ or 4 ms.

c. Each frame has one character from each source, which means the link needs to send 250 frames per second. ফ্রেম রেট ইনপুট রেটের সমান। প্রতি ফ্রেম গঠিত হয় প্রতি সোর্স থেকে ১ টি করে ক্যারেকটার নিয়ে। এখন, প্রতি সেকেন্ডে ২৫০ টি ক্যারেকটার পাঠানো হয়। তাহলে ফ্রেম রেটও হবে প্রতি সেকেন্ডে ২৫০ টি ফ্রেম।

d. The duration of each frame $\frac{1}{250} = 4ms$

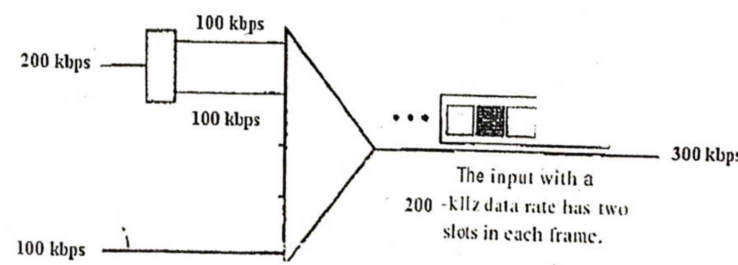
e. Each frame carries 4 characters (because number of connection is 4) and 1 extra synchronizing bit. This means that each frame has = 4× 8 + 1 =33 bits

f. আমরা জানি আউটপুট ডাটা রেট হবে ইনপুট ডাটা রেটের n গুন যেখানে n হল কানেকশন নাম্বার। কিন্তু এইটা তখনই প্রযোজ্য হবে যখন কোন সিনক্রোনাইজেশনের জন্য কোনো ফ্রেম বিট যোগ করা হবে না । যদি ফ্রেম বিট যোগ করা হয়, তবে ইনপুট রেটের ডাটা রেটের সাথে মোট ফ্রেম বিট যোগ করতে হবে। এখানে ফ্রেম আছে ২৫০ টা । So, the link data rate = 4 × (250× 8)+250 = 8250 bps

আবার এখানে ডাটা রেট অন্য ভাবেও পাওয়া যায়। এখানে ফ্রেম আছে ২৫০ টা। প্রতিটা ফ্রেমে বিট সংখ্যা ৩৩ টা। প্রতি সেকেন্ডে ২৫০ টা ফ্রেম পাঠানো হয়। তাহলে ডাটা রেট = 250 × 33 = 8250 bps

159	**Two channels, one with a bit rate of 100 kbps and another with 1 bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?**	

Here, inputs have two different data rates. We need to use here multiple slot allocation technique. We can allocate one slot to the first channel and two slots to the second channel. Each frame carries 3 bits.



Frame rate will be equal to the input rate. So, the frame rate = 100,000 frame per seconds. Because it carries 1 bit from the first channel.

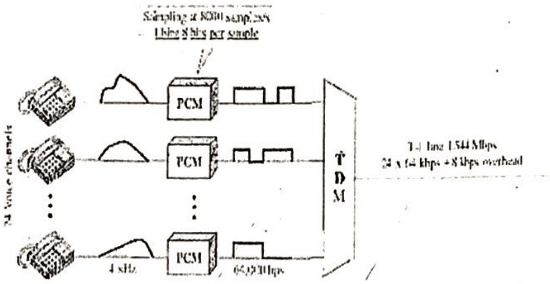
The frame duration = $\frac{1}{100,000}$ $seconds\ per\ frame = 10\ ms$

Bit rate = 3 × 100,000 = 300 kbps

160	T-1 line for multiplexing telephone lines in TDM	

T lines are digital lines designed for the transmission of digital data, audio or video. However, they also can be used for analog transmission (regular telephone connections), providing the analog signals are first sampled, then time division multiplexed.

24 separate telephone lines can be combined into one T-1 line and run only the T-1 line to the telephone exchange. Figure shows how 24 voice channels can be multiplexed onto one T-1 line.



As there are 24 voice channels, the frame on a T-1 line will use 24 time slots of 8 bits. Each slot also has an extra 1 bit for synchronization. So, total number of bit in each frame will be = (24× 8 +1=193 bits.

Now, we know that voice channel has a bandwidth 2700 Hz (300 – 3000 Hz). The standard sampling rate telephone system is 8 kHz which is higher than the Nyquist rate of voice channel.

Now each slot of the frame contains one signal segment from each channel. So, 24 segments are interleaved in one frame. The number of frame will be 8000 frames / sec as sampling rate is 8 kHz. Each frame has 193 bits. So the data rate is given as:

Data rate = 8000 frame / sec × 193 bits / frame = 1.544 Mbps.

161	Describe multiple channel frame alignment for TDM /PCM (T_1 system).	

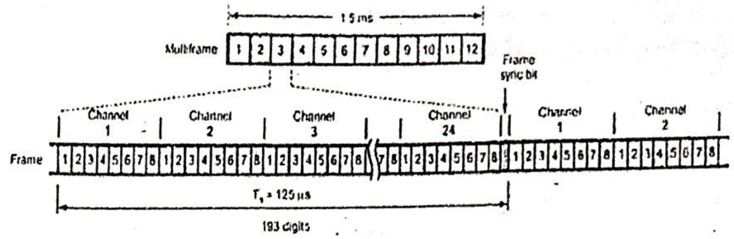


Fig. Multiple channel frame alignment in T1 system

This system contains a multi-frame of 12 frames.

The duration of the multi-frame is 1.5 mili seconds

Each frame consists of samples from 24 channels. This 24 channels are Time Division Multiplexed.

Each channel is encoded into 8 bits. Thus total bits of 24 channels will be 24 × 8 =192 bits.

There is one frame synchronous bit. This indicate the start of the next frame. It is transmitted at the beginning of each frame.

Thus the total bits in one frame are (24 × 8) + 1 =193 bits.

Each channel is normally sampled at 8 kHz rate. So, the sampling period will be $\frac{1}{8000\ Hz} = 125\ microseconds$. So, the time between any two successive samples of single channel will be 125 microseconds.

In the TDM system, the samples from each channel is transmitted in each successive frame. Hence the duration of the frame will also be 125 microseconds.

Now, Bits per frame = 24 channels/frame × 8 bits /channel +1 frame sync bit

=24 × 8+1=193 bits

Therefore, bit rate $r_b = \frac{Number\ of\ bits\ per\ frame}{Time\ of\ one\ frame} = \frac{193}{(125\times10^{-6})s} = 1.544 \times 10^6 bits/sec$

It can also be compared as $R_b = nf_s$; that we learnt earlier. Both are same. In this case, we know sampling frequency $f_s = 8\ kHz$. So, $R_b = nf_s = 193 \times 8\ kHz = 1.544 \times 10^6 bits/sec$. But in this case it was signaling rate, R_b. But we need to calculate *bit rate*, r_b. Both are not same. In the previous case, signaling rate and bit rate was equal. But, in case of TDM, it is different. Because, all bit does not carry the signal information. For example, sync bit.

Now, **the signaling information is transmitted** by replacing 8th bit (i.e. LSB) in each channel by signaling bit **in every sixth frame.**

So, signaling period = period of the signaling bit

$$= 125 \times 10^{-6} \times 6$$

$$= 750 \times 10^{-6} sec$$

$$\therefore signaling\ rate = \frac{1}{signaling\ period} = \frac{1}{750\times10^{-6}} = 1.333\ kbps$$

162	**Four E-1 Channel of 8 Mbps are multiplexed. Calculate the data bit rate.**	**BCIC 2016**

Data bit rate = 4 × 8 Mbps = 32 Mbps

163	**Suppose that, there are three messae signals to be sent using TDM. The first two signals both have frequency contents within 5 KHz, While 3rd message signal has frequency content within 10 kHz. What is the bit rate of TDM system when it operates efficiently in terms of bandwidth usage?**	**BPDB 2016**

TDM is, in principle, a digital multiplexing technique, digital data from different sources are combined into one time shared link. However, it does not mean that sources cannot produce analog data. Analog data can be sampled, changed to digital data and then multiplexed by using TDM.

In light of above discussion, we must digitized the three message signals M_1, M_2 and M_3 . Since, in question, it has been asked to use a TDM system that oerates efficiently in terms of bandwidth usuage. For this we must choose BPSK system where n = 1 (length of codeword) i.e. code can be either binary 0 or binary 1.

According to Nyquist sampling theorem, for efficient reconstruction of message signal,

$$f_s \geq 2f_m$$

Where f_m is signal bandwidth.

Then, if a signal bandwidth is B and the length of codeword is n, then, data rate after digitization = 2nB ; (because of Nyquist theorem.

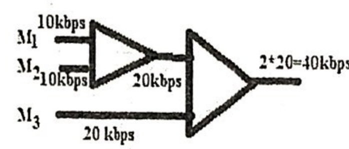
So, bit rate of M_1 = 2nB = 2× 1× 5 = 10 kbps

Bit rate of M_2 = 2 nB = 2 × 1 × 5 = 10 kbps

Bit rate of M_3 = 2 nB = 2× 1× 10 = 20 kbps

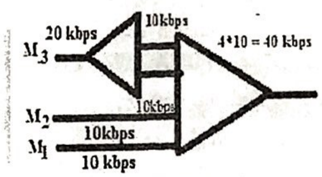
Now, if the data rates of inputs are not same for TDM, three strategies can be used. They are: (a) Multilevel multiplexing, (b) Multiple slot allocation and (c) Phase stuffing.

Using Multilevel multiplexing or Multiple slot allocation we get



using multilevel multiplexing

Bit rate for TDM system = 40 kbps



using multiple slot allocation

So, bit rate for TDM system = 2 × 20 = 40 kbps

164	**Difference between FDM and TDM**	

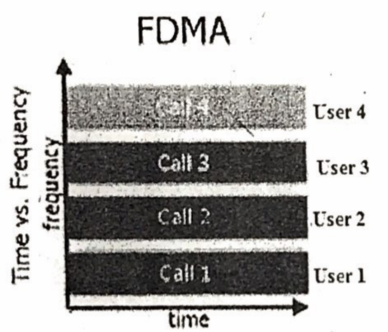
	FDM	TDM
1.	Signals from different sources are modulated with different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link.	Signals from different sources are combined into a one timeshared link.
2.	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signals which are to be multiplexed can occupy the entire bandwidth in the time domain.
3.	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signal.
4.	Synchronization is not required.	Synchronization is required.
5.	The FDM requires a complex circuitry at transmitter and receiver.	TDM circuitry is not very complex.
6.	FDM suffers from the problem of crosstalk.	In TDM, the problem of crosstalk is not severe.

Multiple Access

165	Write a short note on FDMA. What is the difference between FDMA and FDM.

Channelization is a multiple access method in which the available bandwidth of a link is shared in time, frequency or through code, between different stations. There are normally three channelization protocols: FDMA, TDMA and CDMA. (cellular)

Frequency Division Multiple Access (FDMA): In frequency division multiple access (FDMA), is available bandwidth is divided into frequency bands. Each station is allocated a band to send its data. In other words, each band is reserved for a specific station, and it belongs to the station all the time. Each station also uses a band pass filter to confine the transmitter frequencies. To prevent station interferences, the allocated bands are separated from one another by small guard bands.



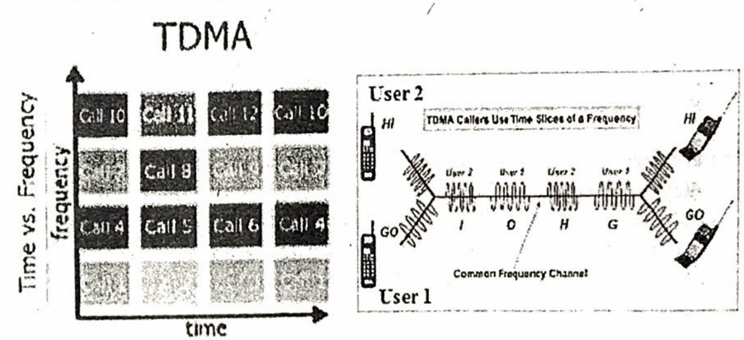
The best example of this is the cable television system. The medium of a single co-axial cable is used to broadcast hundreds of channels of video / audio programming to homes. The co-axial cable has a useful bandwidth from 4 MHz to 1 GHz. This bandwidth is divided up into 6 MHz wide channels. Initially, one TV station or channel used a single 6 MHz band. But with digital techniques, multiple TV channels may share a single band today thanks to multiplexing techniques used in each channel.

We need to emphasize that although FDMA and FDM conceptually seem similar, there are differences between them. FDM is a physical layer technique that combines the load from low-bandwidth channels and transmits them by using a high-bandwidth channel. The channels that are combined are low-pass. The multiplexer modulates the signals, combines them, and creates a band-pass signal. The bandwidth of each channel is shifted by the multiplexer.

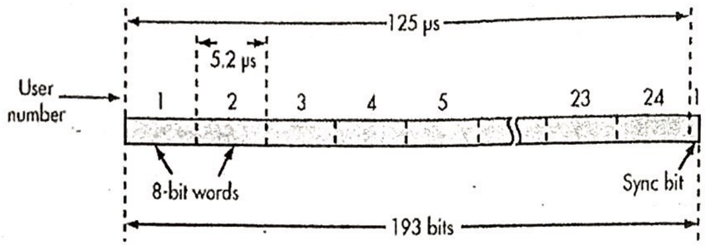
On the other hand, FDMA is an access method in the data link layer. The data link layer in each station tells its physical layer to make a band pass signal from the data passed to it. The signal must be created in the allocated band. There is no physical multiplexer at the physical layer. The signals created at each station are automatically band pass filtered. They are mixed when they are sent to the common channel.

166	Write a short note on TDMA? What is the difference between TDMA and TDM.

In time division multiple access (TDMA), the stations share the bandwidth of the channel in time. Each station is allocated a time slot during which it can send data. Each station transmits its data in is assigned time slot. The main problem with TDMA lies in achieving synchronization between the different stations. Each station needs to know the beginning of its slot and the location of its slot. This may be difficult because of propagation delays introduced in the system if the stations are spread over a large area. To compensate for the delays, we can insert guard times. Synchronization is normally accomplished by having some synchronization bits (normally referred to as preamble bits) at the beginning of each slot.



A good example is widely used T_1 transmission system, which has been used for years in the telecom industry. T_1 lines carry up to 24 individual voice telephone calls on a single line. Each voice signal usually covers 300 Hz to 3000 Hz and is digitized at an 8 kHz rate, which is just a bit more than the minimal Nyquist rate of two times the highest frequency component needed to retain all the analog content.



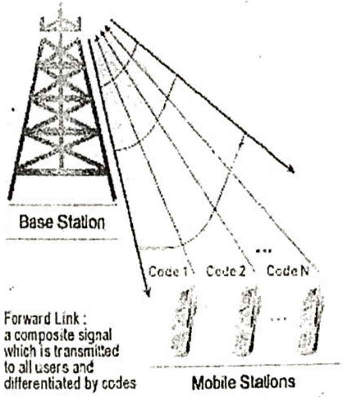
We also need to emphasize that although TDMA and TDM conceptually seem the same, there are differences between them. TDM is a physical layer technique that combines the data from slower channels and transmits them by using a faster channel. The process uses a physical multiplexer that interleaves data units from each channel.

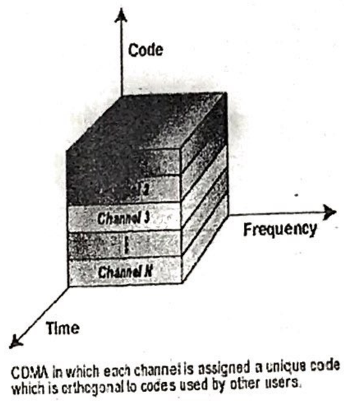
On the other hand, TDMA is an access method in the data link layer. The data link layer in each station tells its physical layer to use the allocation time slot. There is no physical multiplexer at the physical layer.

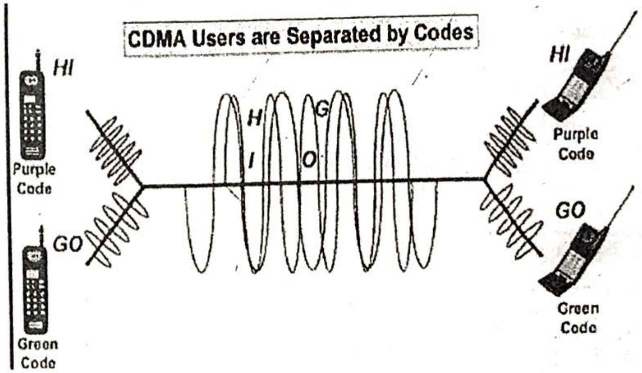
167	Write a short note on CDMA? Describe the constraints of CDMA.	30th BCS

Code division multiple access is a multiple access in which one channel carries all transmission simultaneously. CDMA differs from FDMA because only one channel occupies the entire bandwidth of the link. Again, CDMA differs from TDMA because all stations can send data simultaneously; there is no timesharing. In CDMA, each group of users is given a shared code. Many codes occupy the same channel, but only those user associated with a particular code can communicate.

To understand the concept of CDMA, assume we have a classroom of 6 students. Two of them only know Chinese and don't understand any other languages, two of them only know English and don't understand any other languages and last two students only know Bengali language. If all the students start to talk at a time, only who know the common language will be able to communicate. So, CDMA is similar to this system where language is the analogy to code here.







Advantages:

- No absolute limit on the number of users. Easy addition of more user.
- Better signal quality.
- Impossible for hacker to decipher the code sent. So, it provides secure transmission.
- Efficient practical utilization of fixed frequency spectrum.
- Flexible allocation of resources.
- CDMA is compatible with other cellular technologies; this allows for nationwide roaming.

Disadvantages:

- As the number of users increases, the overall quality of service decreases.
- CDMA is self-jamming.

- Near-far problem arises.
- Complex receivers, needs more complicated power control for senders.

168	Comparison between FDMA, TDMA and CDMA.	31st BCS

Technique	FDMA	TDMA	CDMA
Concept	Divide the frequency band into disjoint sub-bands.	Divide the time into non-overlapping time slots.	Spread the signal with orthogonal codes.
Active terminals	All terminals active on their specified frequencies.	Terminals are active in their specified time slot on same frequency.	All terminals active on same frequency.
Signal separation	Filtering in frequency	Synchronization in time.	Code separation.
Handoff	Hard handoff	Hard handoff	Soft handoff.
Advantages	Simple and robust	Flexible	Flexible
Disadvantages	Inflexible, available frequencies are fixed, requires guard bands.	Requires guard space, synchronization problem.	Complex receivers, requires special power control to avoid near-far problem.
Current Application	Radio, TV and analog cellular	GSM and PDC	2.5 G and 3G

169	What are the common multiple access technologies? Differentiate between multiplexing and multiple access technologies.	BUET M. Sc. 2012

There are three types of common multiple access methods:

a. Frequency Division Multiple Access (FDMA) : Flexible and simple
b. Time Division Multiple Access (TDMA): popular
c. Code Division Multiple Access (CDMA spread spectrum) – highly secure.

Multiplexing	Multiple Access
Multiple signals from users at **same geographical location** are combined into one signal to transmit over a single channel or media.	Users in **different geographical locations** share the use of a single channel or media to transmit over it.
Multiplexing involves transmitting multiple signals and streams simultaneously.	Multiple Access is based on schemes such as frequency division, time division or code division.
Multiplexing combines low speed signals (voice, data etc.) for transmission over a single	In multiple access technique, different users share the common high speed connection.

high speed connection.	
Multiplexing works on the physical layer (L_1) of OSI (Open System Interconnection) model.	Multiple Access works on the Data link Layer (L_2) of OSI model.
Sources of multiplexing are ordinarily not homogeneous in requirements and characteristics.	Sources of multiple access are often homogeneous in requirements and characteristics.
Multiplexing techniques are normally used with telephone, fiber optic channel, and radio channels.	Multiple Access techniques are normally used with satellite channels.
There is no possibility that a transmitted channel will damage other channels because they are transmitted from the same station.	There is always the possibility of one of the transmitted channels interfering and causing damage to other transmitted signals.
Transmitted data frames consist of information about channels and data only.	Transmitted data frame consist of detailed information about channels, earth stations, frame management and data.
User requirement are ordinarily fixed.	User requirement can changed dynamically with time.

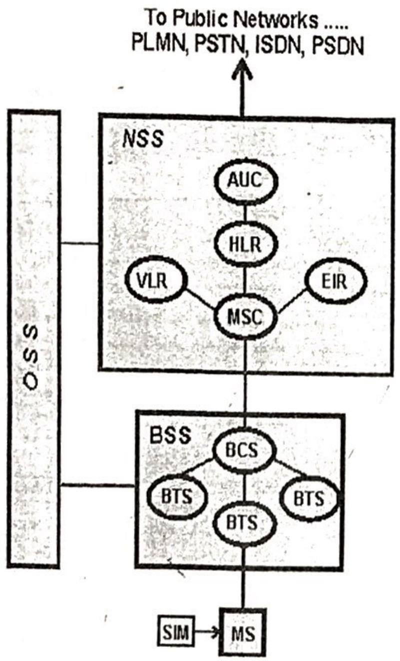
170	**Draw the architecture of a GSM cellular Mobile communication system and briefly describe the functions of each subsystem.**	29th,30th, 31st, 34th BCS

Global system for mobile communication is a second generation cellular standard developed to cater voice services and data delivery using digital modulation.

The GSM network architecture can be grouped into four main areas:

- Mobile station (MS)
- Base Station Subsystem (BSS)
- Network and Switching Sub-system (NSS)
- Operation and Support Subsystem.

A basic diagram of overall GSM architecture is given below:



Simplified GSM Network Architecture Diagram

Mobile Station: Mobile stations, known as cell or mobile, are the section of GSM cellular network that the user sees and operates. Mobile System has two main elements: the main hardware and the SIM.

The hardware contains electronics elements for signal processing. It also contains a number known as the International Mobile Equipment Identity (IMEI). This is installed by the manufacture and cannot be changed. The SIM or Subscriber Identity Module contains the information that provides the identity of the user of the network.

Base Station Subsystem (BSS): It consists of two elements:

- **Base Transceiver Station (BTS):** The BTS comprises radio transmitter receivers and their associated antennas that transmit and receive to directly communicate with the mobile.

- **Base Station Controller (BSC):** The BSC controls a group of BTSs and is often co-located with one of the BTSs in its group.

Network Switching Subsystem (NSS):

It provides the main control and interfacing for the whole mobile network. The major elements with the core network include:

- **Mobile Switching Centre (MSC):** The main element of network switching subsystem is mobile switching centre. The MSc acts like a normal switching node within a PSTN or KSDN, but also provides additional functionality to enable the requirements of a mobile user to be supported. These includes registration, authentication, call location and call routing to a mobile subscriber. It also provides an interface to the PSTN so that calls can be routed from the mobile network to a phone connected to landline.
- **Home Location Register (HLR):** This database contains all the administrative information about each subscriber along with their last known location. In this way, the GSM network is able to route calls to the relevant base station for the MS. Even when the phone is not active (but switched on), it re-registers periodically to ensure that the network is aware of its latest position.
- **Visitor Location Register (VLR):** It contains selected information from the HLR that enables the selected services for the individual subscriber to be provided.
- **Equipment Identity Register (EIR):** The EIR is the entity that decides whether a given mobile equipment may be allowed onto the network. Each mobile equipment has a number known as the International Mobile Equipment Identity. This number is installed in the equipment and is checked by the network during registration. Depending upon the information held in the EIR, the mobile may be allocated one of three states – allowed onto the network, barred access, or monitored in case its problem.
- **Authentication Cenre (AuC):** The AuC is a protected database that contains the secret key also contained in the user's SIM card. It is used for authentication and for ciphering on the radio channel.

Operation and Support Subsystem (OSS): The OSS or operation subsystem is an element within the overall GSM network architecture that is connected to components of the NSS and BSC. It is used to control and monitored the overall GSM network and it is also used to control the traffic load of the BSS.

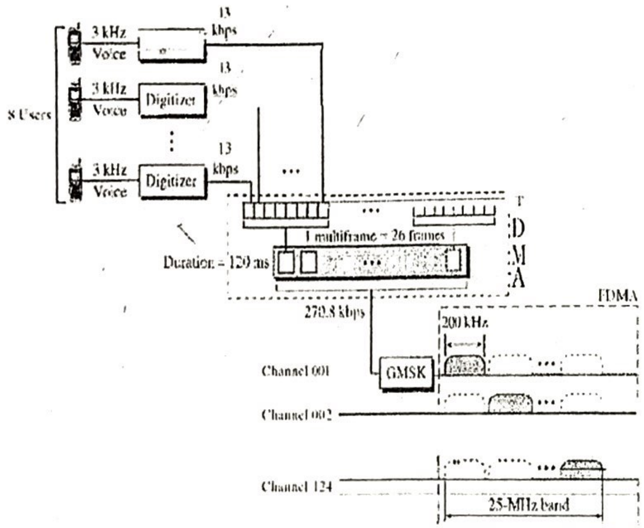
171	**Describe how TDMA-FDMA is used in GSM?**	

জিএসএম ডুপ্লেক্স কমিউনিকেশনের জন্য দুটি ব্যান্ড ইউজ করে। আপলিংকের জন্য ৮৯০ মেগা হার্জ থেকে ৯১৫ মেগা হার্জ ব্যবহার করে। আর ডাউনলিংকের জন্য ৩৩৫ মেগা হার্জ থেকে ৯৬০ মেগা হার্জ পর্যন্ত ব্যবহার করে। তাহলে প্রতিটা ব্যান্ড হল ২৫ মেগা হার্জ করে। এই ২৫ মেগা হার্জ কে ১২৫ টি চ্যানেলে ভাগ করা হয় । প্রতিটি চ্যানেলের ব্যান্ড উইথ ২০০ কিলো হার্জ করে। এই প্রতিটি ব্যান্ড আবার নির্দিষ্ট গার্ড ব্যান্ড দ্বারা সেপারেট করা থাকে।

নরমালি প্রতিটা ভয়েজ চ্যানেল কে ডিজিটাইজড করার মাধ্যমে কমপ্রেস করা হয় এবং ১৩ কেবিপিএস এর ডিজিটাল সিগনালে পরিনত করা হয়। এর পর TDMA পদ্ধতি অনুসরন করা হয়। প্রতিটা স্লটে ১৫৬.২৫ বিট থাকে। আটটি স্লট একটি ফ্রেম শেয়ার করে। মানে প্রতিটা ফ্রেমে আটটি স্লট থাকে যার প্রতিটিতে ১৫৬.২৫ বিট থকে। এই রকম ফ্রেম থকে মোট ২৬ টা। এদেরকে একত্রে হিসাব করা হয়। এদেরকে বলা হয় মাল্টিফ্রেম। টোটাল মাল্টিফ্রেম ডিউরেশন হল ১২০ মিলি সেকেন্ড। তাহলে প্রতি সেকেন্ডে $\frac{1}{120\times10^{-3}}$ টা মাল্টিফ্রেম যাবে। তাহলে $\frac{1}{120\times10^{-3}}$টি ফ্রেমে বিট সংখ্যাই হবে প্রতি সেকেন্ডে বিট রেট বা ডাটা রেট।

channel Data rate $= 156.25 \times 8 \times 26 \times \left(\frac{1}{120\times10^{-3}}\right) = 270.8$ kbps

প্রতিটা 270.8 kbps চ্যানেল GMSK মডুলেশনের মাধ্যমে ২০০ কিলোহার্জ এর এনালগ সিগনালে পরিনত করে। এইরকম ২০০ কিলো হার্জের ১২৪ টা চ্যানেল FDMA এর মাধ্যমে কমবাইন বা একত্রিত করা হয়।



172	Write the following parameters of a GSM 900 and GSM 1800 system. a. Up-link and downlink frequency range b. Channel bandwidth	31st BCS

For GSM 900:

Uplink frequency range: 890 – 915 MHz (25 MHz)

Downlink frequency range: 935 – 960 MHz (25 MHz)

Band gap between uplink and downlink: 20 MHz

Total channel: GSM 900 has 124 channels

For uplink 124 channels and for downlink 124 channels

Bandwidth of each channel: 200 kHz

For GSM 1800

Uplink frequency range: 1710 – 1785 MHz (75 MHz)

Downlink frequency range: 1805 – 1880 MHz (75 MHz)

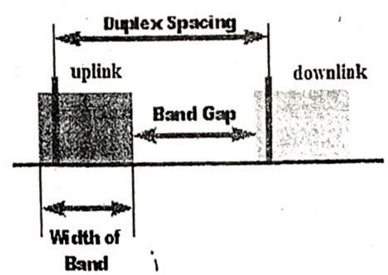
Band gap between uplink and downlink: 20 MHz

Total channel: GSM 1800 has 374 channels

Bandwidth of each channel: 200 kHz

Description:

GSM 900 : GSM 900 uses 890 – 915 MHz to send information from the mobile station (MS) to the Base Transceiver Station (BTS) (uplink) and 935 – 960 MHz for the other direction (downlink), providing 124 RF (radio frequency) channels, spread at 200 kHz. Duplex spacing of 45 MHz.



GSM – 1800: GSM – 1800 uses 1710 – 1785 MHz to send information form Mobile Station (MS) to the Base Transceiver Station (uplink) and 1805 – 1880 MHz for the other direction (downlink), providing 374 channels, spread at 200 kHz. Duplex spacing is 95 MHz.

173	**Explain the following in terms of cellular communication.** a. **Co-channel interference** b. **Frequency re-use** c. **Cell splitting**	32^{nd} & 33^{rd} BCS

Frequency re-use:

Frequency reuse is the core concept of the cellular mobile radio system. In this frequency reuse system, users in different geographical locations (different cells) may simultaneously use the same frequency channel. In other word, The same frequency band or channel used in a cell can be reused in another cell as long as the cells are far apart and the signal strengths do not interfere with each other. This, in turn, enhances the available bandwidth of each cell. Typical cluster of seven such cells and four such clusters with no overlapping area is shown in the figure.

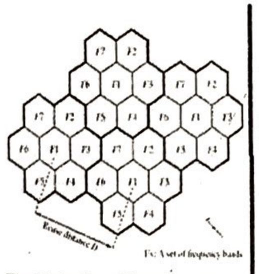


Fig: Illustration of frequency reuse

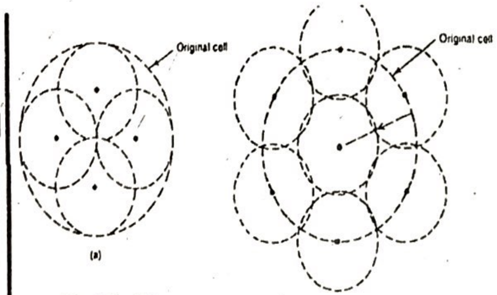


Fig: Cell splitting

The distance between the two cells using the same channel is known as the "reuse distance and represented by D. If R is the radius of each cell, and N is the number of cells in a cluster, the reuse distance is given by: $D = \sqrt{3N}\,R$

Therefore, the reuse factor, $q = \frac{D}{R} = \sqrt{3N}$

Co-channel interference: Though the frequency re-use system can drastically increase the spectrum efficiency, but if the system is not properly designed, serious interference may occur. Interference due to common use of the same channel is called co-channel interference.

Cell splitting: In cellular system, cell splitting is also used to improve the utilization of spectrum efficiency. When traffic density starts to build up the frequency channels in cell cannot provide enough mobile calls, the original cell can be split into smaller cells. There are two ways of splitting: (a) the original cell site is not used (fig: a) , (b) the original cell site is used (fig: b).

174	**In a cellular system, in a cluster a cell with 100 MSs (Mobile stations or mobile sets), if 30 requests are generated during an hour, with average holding time 360 seconds, then find the traffic load in the cell.**	

The offered traffic load is defined as: $a = \lambda T$

Where, a = average traffic load

λ = average call arrival rate (average number of MSs requesting the service) অর্থাৎ প্রতি সেকেন্ডে কতটি রিকোয়েস্ট আসে।

T = average holding time (Average length of time the MSs requiring the service) অর্থাৎ প্রতিটি মোবাইল গড়ে কত সেকেন্ড সার্বিস ইউজ করবে।

Now, the average call arrival rate $\lambda = \frac{30\ requests}{3600\ seconds}$

So, the offered traffic load, $a = \lambda T = \frac{30\ requests}{3600\ seconds} \times 360\ seconds$

$= 3$ Erlangs.

Note: A servicing channel that is kept busy for an hour is quantitatively defined as one Earlang.

175	**Write a short note on satelliete communication.**	**25^{th} 30^{th} BCS**

A communication satellite is an artificial satellite that relays and amplifies radio telecommunication signal via a transponder; it creates a communication channel between a source transmitter and a receiver at different location on Earth.

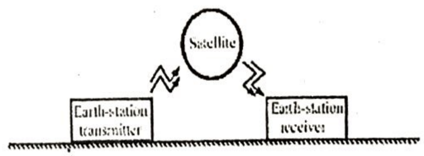


Fig. Block diagram of a typical satellite communication system.

Figure shows an artificial satellite, placed in a suitable orbit around the earth, and rotating at a fixed speed. The satellite consists of a transponder, which receives signals from an earth transmitter, amplifies them, and retransmits them back to an earth receiver. Usually microwave frequencies are used for satellite communication.

Communication satellites are used for television, telephone, radio, internet and military application. There are over 2000 communication satellites in Earth's orbit, used by both private and government organization.

Communications satellites usually have one of three primary types of orbit, while other orbital classifications are used further specify orbital details.

Geostationary Earth orbits (GEO): Geostationary satellites have geostationary orbit (GEO), which is 35,786 kilometers from Earth's surface. This orbit has the special characteristic that the apparent position of the satellite in the sky when viewed by a ground observer does not change, the satellite appears to "stand still" in the sky. This is because the satellite's orbital period is the same as the rotation rate of the Earth. The advantage of this orbit is that ground antennas do not have to tract the satellite acroos the sky, when they can be fixed to point at the location in the sky the satellite appears.

Medium Earth Orbit (MEO): Medium Earth orbit satellites are closer to Earth. Orbital altitudes ranges from 2,000 to 35 kilometres.

Low Earth Orbit (LEO): The region below medium orbits is referred to as low Earth orbit, and is about to 160 to 2,000 kilometers.

As satellites in MEO and LEO orbit the Earth faster, they do not remain visible in the sky to a fixed point on Earth continually like a geostationary satellite, but appear to a ground observer to cross the sky and "set" when they go behind the Earth. Therefore, to provide continuous communications capability with these lower orbits requires a larger number of satellites, so one will always be in the sky for transmission of communication signals. However, due to their relatively small distance to the Earth their signals are stronger.

176	**Draw a block diagram of a satellite earth station and briefly describe the operation.**	30th BCS

Transmitting Block diagram of Earth station:

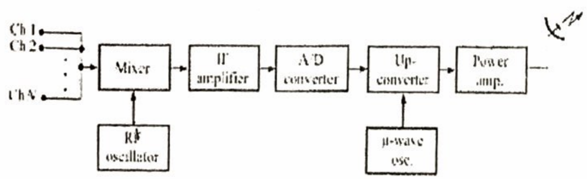


Fig. Simplified block diagram of satellite earth transmitter.

Figure shows that the earth transmitter consists of several analog channel as inputs to a mixer to which an RF carrier frequency at 70 MHz produced by an RF oscillator is also applied. Then the difference signal is IF (intermediate frequency) amplified, and analog to digital converted. The resulting digital signal from the ADC is then up-converted into microwave frequencies using a micro-wave source. The output of the up-converter is then amplified in a power amplifier and fed to the earth station antenna for up-linking with the satellite.

Block Diagram of Earth Station Receiver:

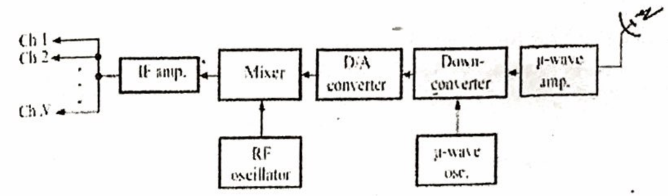


Fig. Simplified block diagram satellite earth receiver.

As shown in figure, the earth receiver receives signal from the satellite transponder antenna and feeds it to a microwave amplifier. The output of this fed to a down-converter, which converts the microwave output frequency down to a lower value. This frequency is then digital to analog converted and applied to a mixer-local oscillator configuration, whose output is IF amplified and then split into respective original channels.

177	**Advantages and disadvantages of Satellite communication**	

Advantages of satellite communication systems:

1. Since microwave frequencies are used for transmission, large bandwidth is available.
2. One microwave cable is capable of carrying 4000 audio channels or 4 video channels. In a communication satellite, there are several transponders (i.e. repeaters).
3. Hence, total number of channels available in a satellite is much more than that available in a terrestrial microwave communication link. This makes the cost per channel in satellite system lower.
4. Since satellite communication is free-space transmission, it produces minimum losses.
5. No repeater stations required.
6. No cables are required. So no expenditure on cable laying.
7. Microwave frequencies are suitable for modern digital communications.
8. Satellite systems can deliver signal to any place on the earth. It means that the remote areas where microwave stations cannot have direct access, satellites can be used for communication.

Disadvantage of satellite communication system:

1. Satellites are not easily accessible. Therefore, if any problem arises in a satellite link, it will become difficult to provide repairing activities.
2. There is an excessive time delay in communication between two subscribers using the satellite link. This is because of the large distance involved between a satellite and its earth stations, and the finite velocity of electromagnetic radiation.
3. The cost of launching a satellite is very high.
4. Addition satellites are required for reliable transmission.

178	Point out the important functions of a telephone set.	30th 33rd BCS

The telephone handset functions as a transmitter and receiver. The telephone handset is the instrument that is help up to the mouth and ear. The transmitter is the end spoken into. It changes the human voice (or acoustic sound wave) into electrical signals that are transmitted via wire to the called party. The receiver performs the inverse overation of the transmitter; that is, it changes the electrical sigal back into human voice.

The telephone set houses the electrical circuitry that performs several important functions, including equalization. This is the method by which signals coming into the telephone set are kept constant regardless of the distance traveled. The telephone box also contains the switch hook, which is the on/off switch of the telephone. By lifting the handset off the cradle, the telephone box activates a set of relays to the LEC's central office switch, incdicating that the person wishes to place a call. The central office switch, in turn, activates dial tone, letting the person know that the call can be placed. A third important function of the telephone box is the ringer, which notifies the user of an incoming call.

179	Describe briefly radio transmission techniques.	

Based on the type of channels being utilized, mobile radio transmission system may be classified as the following three categories:

Simplex system: Simplex systems utilize simplex channels, i.e. the communication is unidirectional. The first user can communicate with the second user. However, the second user cannot communicate with the first user. One example of such a system is a pager.

Half Duplex system: Half duplex radio systems that use half duplex radio channels allow for non-simultaneous bidirectional communication. The first user can communicate with the second user but the second user can communicate to the first user only after the first user has finished his conversation. At a time, the user can only transmit or receive information. A walkie talkie is an example of a half duplex system which uses 'push to talk' and release to listen' type switches.

Full duplex system: Full duplex system allow two way simultaneous communications. Both the user can communicate to each other simultaneously. This can be done by providing two simultaneous but separate channels to both the users. This is possible by one of the two following methods.

(a) Frequency Division Duplexing (FDD): FDD supports two-way radio communication by using two distinct radio channels. One frequency channel is transmitted downstream from the BS to the MS (forward channel). A second frequency is used in the upstream direction and supports transmission from the MS to the BS (reverse channel). Because of the pairing of frequencies, simultaneous transmission in both directions is possible. To

mitigate self-interference between upstream and downstream transmission, a minimum amount of frequency separation must be maintained between the frequency pair.

(b) Time Division Duplexing (TDD): TDD uses a single frequency band to transmit signals in both the downstream and upstream directions. TDD operates by toggling transmission directions over a time interval. This toggling takes place very rapidly and is imperceptible to the user.

180	Difference between 2G, 3G and 4G.	32nd BCS

Generation	Speed	Technonology	Features
2G	9.6/14.4 kbps	TDMA, CDMA	2G capabilities are achieved by allowing multiple user on a single channel via multiplexing. 2G enabled mobile phones can be used for data along with voice communication
3G	3.1 Mbps (peak) 500-700 kbps	CDMA 2000 (1XRTT, EVDO) UMTS, EDGE	3G provides amazing internet browsing speeds. Opens the door to a whole beg of oppurtunities with video calling, video streaming, etc. In 3G, universal access ad portability across different device types made possible. (Telephone & PDA's)
3.5 G	14.4 Mbps (peak) 1-3 Mbps	HSPA	3.5 G supports even higher speeds and enhances higher data needs.
4G	100-300 Mbps (peak) 3-5 Mbps	WiMAX, LTE	Speeds for 4G are increased to lightning fast in order ot keep up with data access demand used by various services. It also supports HD streaming. HD phones can be fully utilized on a 4G network.

181	What is erlang of telephone traffic?	Buet M.Sc.

Erlang is a unit of telecommunication measurement. It is used to describe the total traffic volume of one hour.

For example, if a group of user made 30 calls in one hour, and each call had an average call duration of 5 minutes, then the number of Erlangs this worked out as follows:

Minutes of traffic in the hour = number of calls × duration

$=30\times 5=150$

Hours of traffic in the hour = $\frac{150}{60}$ = 2.5 *Erlangs*

Erlang traffic measurement are made in order to help telecommunication network designers understand traffic patterns within their voice networks. This is essential if they are to successfully designed their network topology and establish the necessary trunk group sizes.

Erlang traffic measurements or estimates can be used to work out how many lines are required between a telephone system and a central office (PSTN exchange lines), or between multiple network locations.

182	**Write a short note on PLCC.**	**PDB -12**

Power line communication (PLC) or Power line carrier communication (PLCC) carries data on a conductor that is also used simultaneously for AC electric power transmission or electric power distribution to cosumers.

It is also known as power line carrier, power line digital subscriber line (PDSL), mains communication, power line telecommunications, or power line networking (PLN).

Application of PLCC:

PLCC technology can be deployed into different types of applications in order to provide economic networking solutions. Hence merging with other technologies, it proves useful in different areas. These are few key areas where PLC communication are utilized:

a. **Transmission and Distribution network:** PLCC was first adpted in the electrical transmission and distribution system to transmit information at a fast rate.
b. **Home control and Automation:** PLCC technology is used in home control and automation. This technology can reduce the resources as well as efforts for activities like power and managements, energy conservation, etc.
c. **Entertainment:** PLCC is used to distribute the multimedia contenet through out the home.
d. **Telecommunication:** Data transmission for different types of communication like telecommunication, audio, video communication can be made with the use of PLCC technology.
e. **Security system:** In monitoring houses or business through surveillance cameras, PLCC technology is far useful.
f. **Automatic Meter Reading:** Automatic Meter reading applications use the PLCC technology to send the data from home meters to Host Central Station.

183	**Modulation Scheme used by different communication systems.**	

Communication system	**Used Modulation Technique**
GSM 2G	GMSK
GPRS 2.5 G	GMSK
EDGE 2.75 G	8 PSK
CDMA 2000	QPSK in forward channel (from BTS to Mobile)
UMTS 3G	QPSK
HSDPA 3.5 G	Adaptive Modulation: QPSK, 16 QAM
WiMAX	Adaptive modulation: QPSK, 16 QAM, 64 QAM
Wi-Fi	BPSK, QPSK, 16QAM, 64 QAM

184	**Find a signal g(t) that is band limited to B Hz and whose samples are $g(0) = 1$ and $g(\pm T_s) = g(\pm 2T_s) = g(\pm 3T_s) = \cdots = 0$ Where the sampling interval T_s is the Nyquist interval for g(t), that is $T_s = \frac{1}{2B}$.**	

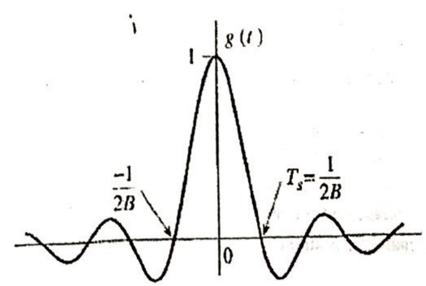
We use the interpolation formula,

$g(t) = \sum_k g(kT_s) sinc(2\pi Bt - k\pi)$ to reconstruct g(t) from its samples.

Since all but one of the Nyquist samples are zero, only one term (corresponding to k = 0) in the summation on the right hand side survives. Thus,

$$g(t) = sinc(2\pi Bt)$$

This signal is shown in the figure given below. Observe that this is only signal that has a bandwidth B Hz and sample value g(0) = 1 and $g(nT_s) = 0$ $(n \neq 0)$. No other signal satisfies these condition.



(b)

185	Write a short note on fiber optic communication.	25^{th} BCS

Fiber optic communication is a method of transmitting information from one place to another by sending pulses of light through optical fiber. The light forms an electromagnetic carrier wave that is modulated to carry information. Fiber is preferred over electrical cabling when high bandwidth, long distance, or immunity to electromagnetic interference are required.

Optical fiber is used many telecommunication companies to transmit telephone signals, internet communication, and cable television signals.

Advantages of fiber optics:

There are a number of compelling reasons that lead to the widespread adeption of fibre optic cabling for telecommunication applications:

- Much lower levels of signal attenuation.
- Fiber optic cabling provides a much higher bandwidth allowing more data to be delivered.
- Fiber optic cables are much lighter than the coaxial cables that might otherwise be used.
- Fiber optics do not suffer from stray interference pickup that occurs with coaxial cabling.

Fiber optic transmission system:

Any fiber optic data transmission system will comprise a number of different elements. There are three major elements:

- Transmitter (light source)
- Fibre optic cable
- Optical repeater
- Receiver (detector)

Fiber optic transmitter: Although the original telecommunications fiber optic system would have used large lasers, today a variety of semiconductor devices can be used. The most commonly used devices are light emitting diode (LED), and semiconductor laser diodes.

Fiber optic cable: Fiber optic cable consists of core, around which is another layer referred to as the cladding. Outside of this there is a protective outer coating. The fiber optic cables operate because their cladding has a refractive index that is slightly lower than that of the core. This means that light passing down the core undergoes total internal reflection when it reaches the core / cladding boundary, and it is thereby contained within the coreof the optical fiber.

Repeaters and amplifiers:

There is a maximum distance overwhich signals may be transmitted over fiber optic cabling. This is limited not only the attenuation of the cable, but also the distortion of the light signal along the cable. In order to overcome these effects and transmit the signals over longer distances, repeaters and amplifiers are used..

Receivers: Light travelling along a fiber optic cable needs to be converted into an electrical signal so that it can be processed and the data is carried can be extracted. The component that is at the heart of the receiver is a photo detector. This is normally a semiconductor device. Photo transistors are not used because they do not have sufficient speed.

186	Important Abbreviation:	

EVDO: Evolution Data Optimized
GPRS: General Packet Radio Service
EDGE: Enhanced Data rate for GSM Evolution
BIST: Build in Self Test
DWDM: Dense Wavelength Division Multiplexing
HSPA: High Speed Packet Access
VSAT: Very Small Aperture Terminal
WiMAX: Worldwide Interoperability for Microwave Access
WLAN: Wireless Local Area Network
ADSL: Asymmetric Digital Subscriber Line
SONET: Synchronous Optical Network
AMPS: Advanced Mobile Phone System
GSM: The Global System for Mobile Communication
GPS: Global Positioning System.
GMSK: Gaussian Minimum Shift Keying
OFDM: Orthogonal Frequency Division Multiplexing
MC-CDMA: Multi-carrier Code Division Multiple Access
LTE: Long Term Evolution
PLL: Phase Locked Loop
Modem: Mdoulator demodulator
AWGN: Additive White Gaussian Noise
3G: Third Generation (Mobile Network)
3GPP: Third Generation Partnership Project
BIOS: Basic Input Output System
CD-ROM: Compact Disk-Read Only Memory
FTP: File transfer Protocol
GPS: Global Positioning System
ISDN: Integrated Services Digital Network
LAP: Link Access Protocol

PSN: Packet Switch Network
PSTN: Public Switched Telephone Network
STDM: Statistical Time Division Multiplexing
STM: Synchronous Transfer Mode
UART: Universal Asynchronous Receiver/Transmitter
UMTS: Universal Mobile Telecommunication System.
UTRA: Universal Terrestrial Radio Access
W-CDMA: Wideband Code Division Multiple Access
BISDN: Broadbrand Integrated Services Digital AT & T Network
ASDL: Arithmatic Digital Subscriber Line

A FDM system has 24 transmitter receiver pairs. Each transmitter uses DSB-AM for transmitting its message signal. The message has a baseband spectrum over 0-4 kHz. Calculate the BW required for the FDM system assuming a guard band equal to 0.5 kHz. [ERL 2017]

Solution:

For 24 channels, we will need 23 guard band.

This means the required bandwidth is at least $= 24\times 4 + 23\times 0.5 = 107.5$ kHz.

Note:

Frequency division multiplexing (FDM) is an **analog technique** that can be applied when the bandwidth of a link is greater than the combined bandwidth of the signals to be transmitted. In FDM, signals generated by each sending device is modulated with different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. Channels can be separated by unused bandwidth guard bands to prevent signals from overlapping. In the demodulator, the original signals are separated from the composite signal through filtering.

A call centre has 25 telephone lines. 1000 calls are received in a day and average duration of each call is 15 seconds. Determine the call capacity of call center in Erlangs. [ERL 2017]

Call capacity or traffic volume in Erlangs = Average no. of calls × average duration of each call.

Note: Erlangs has no unit. So, to determine average no. of calls and average duration of each call, the unit of time should be remained same.

Now, given that

Average no. of call = 1000 calls per day

Average duration of each call = 15 minute $= \frac{15}{24\times 60} = \frac{1}{96}\ day$

So, traffic volume $= 1000 \times \frac{1}{96} = 10.416\ Erlangs$

Or,

Average no. of calls per hour $= \frac{1000}{24} = 41.67$

Average duration of each call $= 15\ minute = \frac{15}{60} = 0.25\ hours$

So, traffic volume= $41.416 \times 0.25 = 10.354$ Erlangs

Note: Number of telephone lines will not be used to determine Erlangs. Reference: http://www.had2know.com/technology/erlang-b-calculator-formula.html

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