Q. 2 a. Explain the need of modulation in communication system. (8)

Answer: Modulation is an important step of communication system. Modulation is defined as the process whereby some characteristic (line amplitude, frequency, phase of a high frequency signal wave (carrier wave) is varied in accordance with instantaneous value intensity of low frequency signal wave (modulating wave.)

<u>Need for modulation</u> : –

(i) **To separate signal from different transmitters :-**Audio frequencies are within the range of 20 Hz to 20 kHz. Without modulation all signals at same frequencies from different transmitters would be mixed up. There by giving impossible situation to tune to any one of them. In order to separate the various signals, radio stations must broadcast at different frequencies.

Each radio station must be given its own frequency band. This is achieved by frequency translation as a result of modulation process.

(ii) **Size of the antenna : –**For efficient transmission the transmitting antennas should have length at least equal to a quarter of the wavelength of the signal to be transmitted. For an electromagnetic wave of frequency 15 kHz, the wavelength λ is 20 km and one-quarter of this will be equal to 5 km. Obviously, a vertical antenna of this size is impractible. On the other hand, for a frequency of 1 MHz, this height is reduced to 75m.

Also, the power radiated by an antenna of length *l* is proportional to $(l/\lambda)^2$. This shows that for the same antenna length, power radiated is large for shorter wavelength. Thus, our signal which is of low frequency must be translated to the high frequency spectrum of the electromagnetic wave. This is achieved by the process of modulation.

---

b. What is Shot noise? Describe the variables on which Shot noise depends. (8)

Answer: Shot noise or Poisson noise is a type of electronic noise which can be modeled by a Poisson process. In electronics shot noise originates from the discrete nature of electric charge. Shot noise also occurs in photon counting in optical devices, where shot noise is associated with the particle nature of light.

Q. 3 a. Describe briefly amplitude modulation. Develop a mathematical expression for Amplitude Modulation Index and what happens if this index exceeds 1? (8)

Answer: Amplitude Modulation (AM) is the process in which the amplitude of the Carrier signal is varied in accordance with the information Signal. If the un-modulated carrier signal is represented by

A sin2pifc t

and the modulating signal is represented by

B sin2pifa t

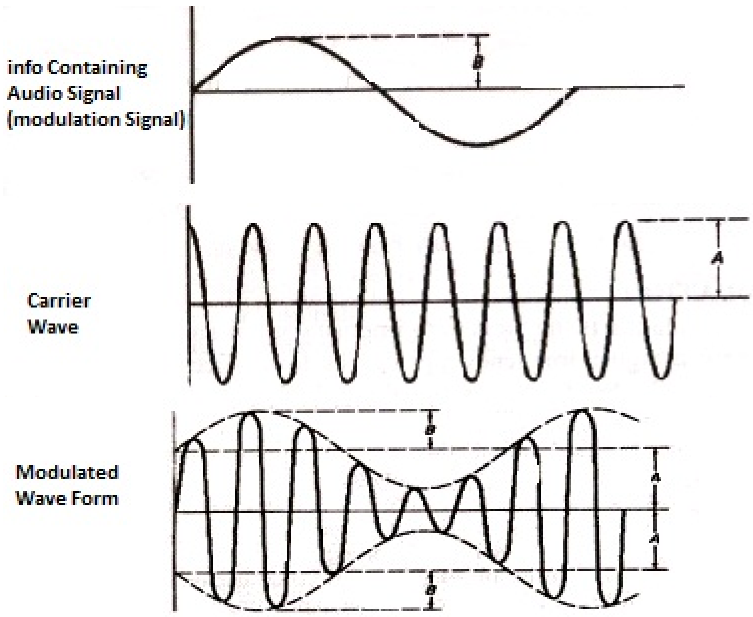
Where,

fc=Carrier frequency
A= Maximum value of un-modulated signal
fa= Frequency of the modulating signal
B= Maximum value of the un-modulated signal.

The modulated signal can be represented mathematically as

A+B (sin 2 pi fc t)(sin 2 pi fa t)

The carrier signal, modulating signal and the modulated signal is shown in the below fig.



Derivation

A carrier is described by

$$v = V_c \sin(\omega_c t + \theta)$$

To amplitude modulate the carrier its amplitude is changed in accordance with the level of the audio signal, which is described by

$$v = V_m \sin(\omega_m t)$$

The amplitude of the carrier varies sinusoidally about a mean of V_c. When the carrier is modulated its amplitude is varied with the instantaneous value of the modulating signal. The amplitude of the variation of the carrier amplitude is V_m and the angular frequency of the rate at which the amplitude varies is ω_m. The amplitude of the carrier is then:

Carrier amplitude = $V_c + V_m \sin(\omega_m t)$

and the instantaneous value (value at any instant in time) is

$$v = \{V_c + V_m \sin(\omega_m t)\} * \sin(\omega_c t) \qquad \text{Eqn. 1}$$

$$= V_c \sin(\omega_c t) + V_m \sin(\omega_m t) * \sin(\omega_c t)$$

Using Sin A * Sin B = ½ Cos (A - B) - ½ Cos (A + B) this becomes

$$v = V_c \sin(\omega_c t) + \tfrac{1}{2} V_m \cos((\omega_c - \omega_m) t) - \tfrac{1}{2} V_m \cos((\omega_c + \omega_m)t) \qquad \text{Eqn. 2}$$

This is a signal made up of 3 signal components

- carrier at ω_c (rad/s) Frequency is $f_c = \omega_c/2\pi$ Hz
- upper side frequency $\omega_c + \omega_m$ (rad/s) Frequency is $(\omega_c + \omega_m)/2\pi = f_m + f_c$ Hz

- lower side frequency $\omega_c - \omega_m$ (rad/s) Frequency is $(\omega_c - \omega_m)/2\pi = f_m - f_c$ Hz

The bandwidth (the difference between the highest and the lowest frequency) is

$BW = (\omega_c + \omega_m) - (\omega_c - \omega_m) = 2 * \omega_m$ Rad/s ($= \omega_m/\pi$ Hz)

Modulation index(m) is basically the ratio of peak voltage of modulating signal and peak voltage of carrier signal and amplitude modulation could be thought of as superimposition of modulations signal on the carrier. So when m is greater than one, over-modulation occurs and the modulating signal being of greater amplitude,part of its information is lost in the process of modulation which is undesirable.

---

b. Calculate the percentage saving in power, if only one side band transmission is transmitted for: **(8)**
(i) 80% modulation
(ii) 50% modulation

Answer:

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

i) When modulation depth is 80%.

$$P_t = P_c\left(1+\frac{(.8)^2}{2}\right) = 1.32P_c$$

$$P_{SB} = P_c\left(\frac{m^2}{4}\right) = P_c\frac{.8^2}{4} = .16P_c$$

$$\text{Saving} = \frac{1.32-0.16}{1.32} = \frac{1.16}{1.32}\times 100 \quad \text{Ans}$$

ii) Solution is similar to above.

when modulation depth is 50%

$$P_t = P_c\left(1+\frac{(0.5)^2}{2}\right) = 1.125P_c$$

$$P_{SB} = P_c\left(\frac{m^2}{4}\right) = P_c\frac{(0.5)^2}{4} = 0.062P_c$$

$$\text{Saving} = \frac{1.125-0.062}{1.125} = 0.95\times 100$$

$$= 95\%$$

MODERATION-I

Q. 4 a. What are the advantages and disadvantages of frequency modulation in comparison to amplitude modulation? (8)

Answer: Advantages of FM over AM systems are:

- The amplitude of an FM wave remains constant. This provides the system designers an opportunity to remove the noise from the received signal. This is done in FM receivers by employing an amplitude limiter circuit so that the noise above the limiting amplitude is suppressed. Thus, the FM system is considered a noise immune system. This is not possible in AM systems because the baseband signal is carried by the amplitude variations it self and the envelope of the AM signal cannot be altered.

- Most of the power in an FM signal is carried by the side bands. For higher values of the modulation index, mc, the major portion of the total power is contained is side bands, and the carrier signal contains less power. In contrast, in an AM system, only one third of the total power is carried by the side bands and two thirds of the total power is lost in the form of carrier power.
- In FM systems, the power of the transmitted signal depends on the amplitude of the unmodulated carrier signal, and hence it is constant. In contrast, in AM systems, the power depends on the modulation index ma. The maximum allowable power in AM systems is 100 percent when ma is unity. Such restriction is not applicable in case of FM systems. This is because the total power in an FM system is independent of the modulation index, mf and frequency deviation fd. Therefore, the power usage is optimum in an FM system.
- In an AM system, the only method of reducing noise is to increase the transmitted power of the signal. This operation increases the cost of the AM system. In an FM system, you can increase the frequency deviation in the carrier signal to reduce the noise. if the frequency deviation is high, then the corresponding variation in amplitude of the baseband signal can be easily retrieved. if the frequency deviation is small, noise 'can overshadow this variation and the frequency deviation cannot be translated into its corresponding amplitude variation. Thus, by increasing frequency deviations in the FM signal, the noise effect can he reduced. There is no provision in AM system to reduce the noise effect by any method, other than increasing itss transmitted power.
- In an FM signal, the adjacent FM channels are separated by guard bands. In an FM system there is no signal transmission through the spectrum space or the guard band. Therefore, there is hardly any interference of adjacent FM channels. However, in an AM system, there is no guard band provided between the two adjacent channels. Therefore, there is always interference of AM radio stations unless the received signal is strong enough to suppress the signal of the adjacent channel.

The disadvantages of FM systems over AM systems are:

- There are an infinite number of side bands in an FM signal and therefore the theoretical bandwidth of an FM system is infinite. The bandwidth of an FM system is limited by Carson's rule, but is still much higher, especially in WBFM. In AM systems, the bandwidth is only twice the modulation frequency, which is much less than that of WBFN. This makes FM systems costlier than AM systems.
- The equipment of FM system is more complex than AM systems because of the complex circuitry of FM systems; this is another reason that FM systems are costlier AM systems.

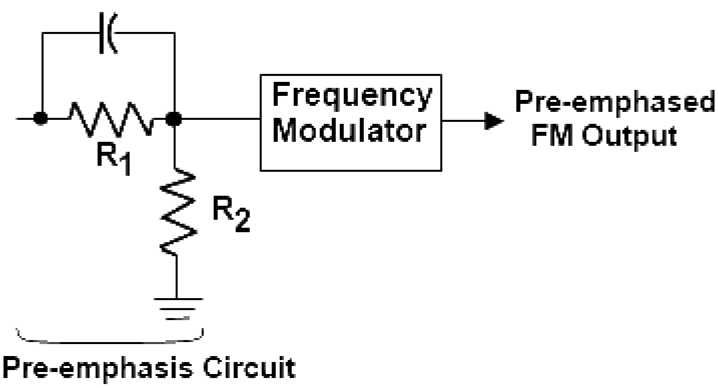
- The receiving area of an FM system is smaller than an AM system consequently FM channels are restricted to metropolitan areas while AM radio stations can be received anywhere in the world. An FM system transmits signals through line of sight propagation, in which the distance between the transmitting and receiving antenna should not be much in an AM system signals of short wave band stations are transmitted through atmospheric layers that reflect the radio waves over a wider area.

b. Describe the concept of pre-emphasis and de-emphasis with the help of circuit diagram. (8)

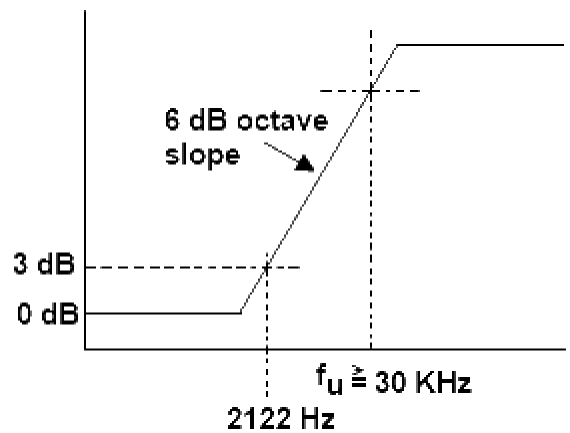
Answer: Pre-emphasis refers to boosting the relative amplitudes of the modulating voltage for higher audio frequencies from 2 to approximately 15 KHz.

Pre-emphasis circuit

At the transmitter, the modulating signal is passed through a simple network which amplifies the high frequency, components more than the low-frequency components. The simplest form of such a circuit is a simple high pass filter of the type shown in fig (a). Specification dictate a time constant of 75 microseconds (µs) where t = RC. Any combination of resistor and capacitor (or resistor and inductor) giving this time constant will be satisfactory. Such a circuit has a cutoff frequency fco of 2122 Hz. This means that frequencies higher than 2122 Hz will he linearly enhanced. The output amplitude increases with frequency at a rate of 6 dB per octave. The pre-emphasis curve is shown in Fig (b). This pre-emphasis circuit increases the energy content of the higher-frequency signals so that they will tend to become stronger than the high frequency noise components. This improves the signal to noise ratio and increases intelligibility and fidelity.



(a) Pre-emphasis Circuit



(b) Pre-emphasis Curve

The pre-emphasis circuit also has an upper break frequency fu where the signal enhancement flattens out.

See Fig (b). This upper break frequency is computed with the expression.

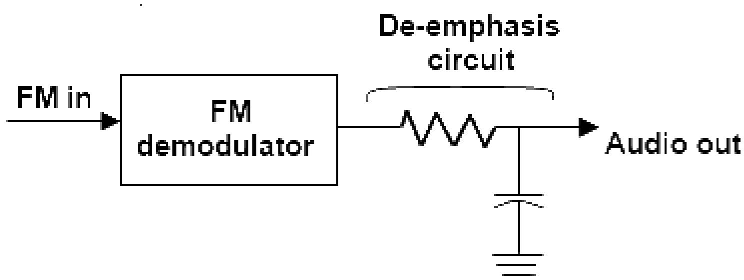
$fu = R1 + (R2/2\pi R1R1C)$

It is usually set at some very high value beyond the audio range. An fu of greater than 30KHz is typical.

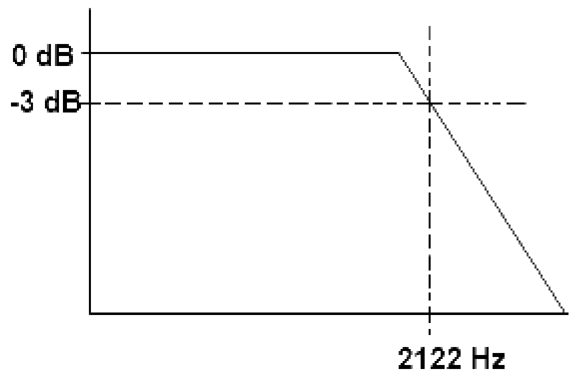
De-emphasis means attenuating those frequencies by the amount by which they are boosted.

However pre-emphasis is done at the transmitter and the de-emphasis is done in the receiver. The purpose is to improve the signal-to-noise ratio for FM reception. A time constant of 75μs is specified in the RC or L/Z network for pre-emphasis and de-emphasis.

De-emphasis Circuit

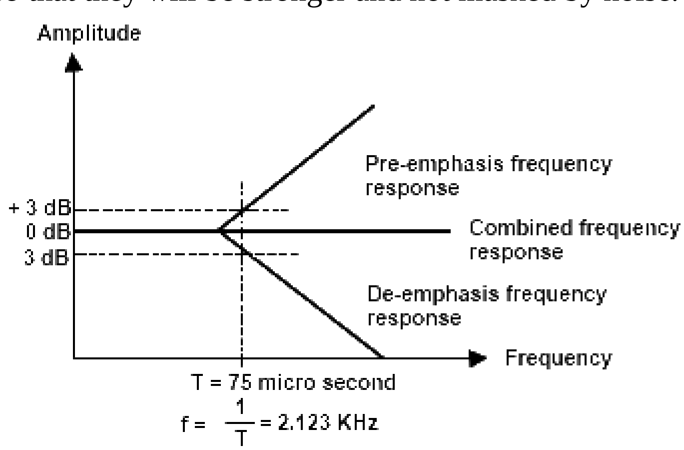


(c) De-emphasis circuit



(d) De-emphasis Curve

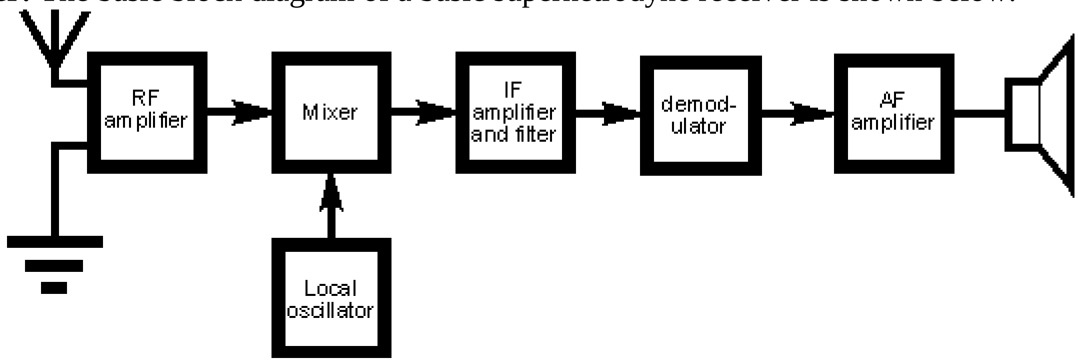
To return the frequency response to its normal level, a de-emphasis circuit is used at the receiver. This is a simple low-pass filter with a constant of 75 πs. See figure (c). It features a cutoff of 2122 Hz and causes signals above this frequency to be attenuated at the rate of 6bB per octave. The response curve is shown in Fig (d). As a result, the pre-emphasis at the transmitter is exactly offset by the de-emphasis circuit in the receiver, providing a normal frequency response. The combined effect of pre-emphasis and de-emphasis is to increase the high-frequency components during transmission so that they will be stronger and not masked by noise.



Combined frequency response

Q. 5 a. Draw the Block Diagram of basic super heterodyne receiver and briefly explain it's working. Give its uses. (8)

Answer: The basic block diagram of a basic superhetrodyne receiver is shown below.



Block diagram of a basic superheterodyne radio receiver

The way in which the receiver works can be seen by following the signal as is passes through the receiver.

- Front end amplifier and tuning block: Signals enter the front end circuitry from the antenna. This circuit block performs two main functions:
    - Tuning: Broadband tuning is applied to the RF stage. The purpose of this is to reject the signals on the image frequency and accept those on the wanted frequency. It must also be able to track the local oscillator so that as the receiver is tuned, so the RF tuning remains on the required frequency. Typically the selectivity provided at this stage is not high. Its main purpose is to reject signals on the image frequency which is at a frequency equal to twice that of the IF away from the wanted frequency. As the tuning within this block provides all the rejection for the image response, it must be at a sufficiently sharp to reduce the image to an acceptable level. However the RF tuning may also help in preventing strong off-channel signals from entering the receiver and overloading elements of the receiver, in particular the mixer or possibly even the RF amplifier.
    - Amplification: In terms of amplification, the level is carefully chosen so that it does not overload the mixer when strong signals are present, but enables the signals to be amplified sufficiently to ensure a good signal to noise ratio is achieved. The amplifier must also be a low noise design. Any noise introduced in this block will be amplified later in the receiver.
- Mixer / frequency translator block: The tuned and amplified signal then enters one port of the mixer. The local oscillator signal enters the other port. The performance of the mixer is crucial to many elements of the overall receiver performance. It should eb as linear as possible. If not, then spurious signals will be generated and these may appear as 'phantom' received signals.
- Local oscillator: The local oscillator may consist of a variable frequency oscillator that can be tuned by altering the setting on a variable capacitor. Alternatively it may be a frequency synthesizer that will enable greater levels of stability and setting accuracy.

- Intermediate frequency amplifier, IF block : Once the signals leave the mixer they enter the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters, dependent upon what is required.
- Detector / demodulator stage: Once the signals have passed through the IF stages of the superheterodyne receiver, they need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. Different demodulators used may include:

    - AM diode detector: This is the most basic form of detector and this circuit block would simple consist of a diode and possibly a small capacitor to remove any remaining RF. The detector is cheap and its performance is adequate, requiring a sufficient voltage to overcome the diode forward drop. It is also not particularly linear, and finally it is subject to the effects of selective fading that can be apparent, especially on the HF bands.
    - Synchronous AM detector: This form of AM detector block is used in where improved performance is needed. It mixes the incoming AM signal with another on the same frequency as the carrier. This second signal can be developed by passing the whole signal through a squaring amplifier. The advantages of the synchronous AM detector are that it provides a far more linear demodulation performance and it is far less subject to the problems of selective fading.
    - SSB product detector: The SSB product detector block consists of a mixer and a local oscillator, often termed a beat frequency oscillator, BFO or carrier insertion oscillator, CIO. This form of detector is used for Morse code transmissions where the BFO is used to create an audible tone in line with the on-off keying of the transmitted carrier. Without this the carrier without modulation is difficult to detect. For SSB, the CIO re-inserts the carrier to make the modulation comprehensible.
    - Basic FM detector: As an FM signal carries no amplitude variations a demodulator block that senses frequency variations is required. It should also be insensitive to amplitude variations as these could add extra noise. Simple FM detectors such as the Foster Seeley or ratio detectors can be made from discrete components although they do require the use of transformers.
    - PLL FM detector: A phase locked loop can be used to make a very good FM demodulator. The incoming FM signal can be fed into the reference input, and the VCO drive voltage used to provide the detected audio output.
    - Quadrature FM detector: This form of FM detector block is widely used within ICs. IT is simple to implement and provides a good linear output.

- Audio amplifier: The output from the demodulator is the recovered audio. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker

USES : The superheterodyne receiver offers superior sensitivity, frequency stability and selectivity. Compared with the tuned radio frequency receiver (TRF) design, superhets offer better stability because a tuneable oscillator is more easily realized than a tuneable amplifier. Operating at a lower frequency, IF filters can give narrower passbands at the same Q factor than an equivalent RF filter. A fixed IF also allows the use of a crystal filter[7] or similar technologies that cannot be tuned. Regenerative and super-regenerative receivers offered a high sensitivity, but often suffer from stability problems making them difficult to operate.

b. What factors are to be considered while choosing the value of Intermediate Frequency (IF)? (8)

Answer: The frequency spectrum has been distributed for various purposes. Otherwise, the people may use the frequencies of their choice & there will be wide range of interference. So, in order to ensure proper reception of signals, the standards have been fixed for the transmission of frequencies & also for the intermediate frequency because if the intermediate frequency is varied the overall frequency value will also vary. The intermediate frequency value should be so designed that it should not lie within the range of mixer stage. Otherwise, there is the production of noise signal due to the interference of mixer frequency & intermediate frequency. Also, the intermediate frequency should not be too high. Otherwise, it will reduce the selectivity of the receiver because of increase in bandwidth. Considering all these factors 455 khz is the most suitable intermediate frequency value for an AM receiver.

Q.6 a. Discuss the standing waves and impedances in a quarter wave and half wave length transmission lines. (8)

Answer:

Standing waves When power is applied to a transmission line by a generator, a voltage and a current appear whose values depend on the characteristic impedance and the applied power. The voltage and current waves travel to the load at a speed slightly less than v_c, depending on the velocity factor. If $Z_L = Z_0$, the load absorbs all the power, and none is reflected. The only waves then present are the voltage and current (in phase) *traveling waves* from generator to load.

If Z_L is not equal to Z_0, some power is absorbed, and the rest is reflected. We thus have one set of waves, V and I, traveling toward the load, and the reflected set traveling back to the generator. These two sets of traveling waves, going in opposite directions (180° out of phase), set up an interference pattern known as *standing waves*, i.e., beats, along the line. This is shown in Figure 7-6 for a short-circuited line. It is seen that *stationary* voltage and current minima (nodes) and maxima (antinodes) have appeared. They are separated by half the wavelength of the signal, as will be explained. Note that voltage nodes and current antinodes coincide on the line, as do current nodes and voltage antinodes.

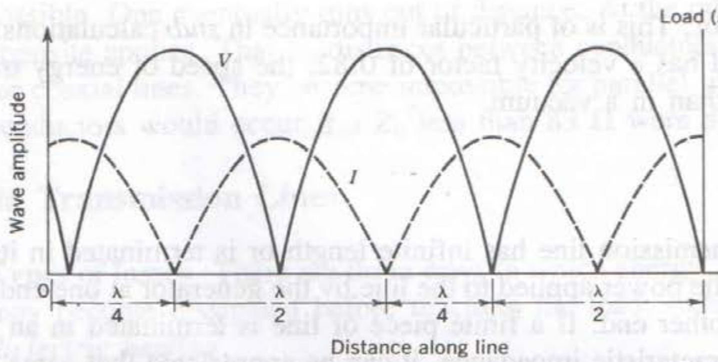


FIGURE 7-6 Lossless line terminated in a short circuit.

Consider only the forward traveling voltage and current waves for the moment. At the load, the voltage will be zero and the current a maximum because the load is a short circuit. Note that the current has a finite value since the line has an impedance. At that instant of time, the same conditions also apply at a point exactly one wavelength on the generator side of the load, and so on. The current at the load is always a maximum, although the size of this maximum varies periodically with time, since the applied wave is sinusoidal.

The reflection that takes place at the short circuit affects both voltage and current. The current now starts traveling back to the generator, unchanged in phase (series circuit theory), but *the voltage is reflected with a* 180° *phase reversal*. At a point exactly a quarter-wavelength from the load, the current is *permanently* zero (as shown in Figure 7-6). This is because the forward and reflected current waves are exactly 180° out of phase, as the reflected wave has had to travel a distance of $\lambda/4 + \lambda/4 = \lambda/2$ farther than the forward wave. The two cancel, and a current node is established. The voltage wave has also had to travel an extra distance of $\lambda/2$, but since it underwent a 180° phase reversal on reflection, its total phase change is 360°. Reinforcement will take place, resulting in a voltage antinode at precisely the same point as the current node.

A half-wavelength from the load is a point at which there will be a voltage zero and a current maximum. This arises because the forward and reverse current waves are now in phase (current has had to travel a total distance of one wavelength to return to this point). Simultaneously the voltage waves will cancel, because the 180° phase reversal on reflection must be added to the extra distance the reflected wave has to travel. All these conditions will repeat at half-wavelength distances, as shown in Figure 7-6. Every time a point is considered that is $\lambda/2$ farther from the load than some previously considered point, the reflected wave has had to travel one whole wavelength farther. Therefore it has the same relation to the forward wave as it had at the first point.

It must be emphasized that this situation is permanent for any given load and is determined by it; such waves are truly *standing* waves. All the nodes are permanently fixed, and the positions of all antinodes are constant. Many of the same conditions apply if the load is an open circuit, except that the first current minimum (and voltage maximum) is now at the load, instead of a quarter-wavelength away from it. *Since the load determines the position of the first current node, the type of load may be deduced from the knowledge of this position.*

Standing-wave ratio (SWR) *The ratio of maximum current to minimum current along a transmission line is called the standing-wave ratio, as is the ratio of maximum to minimum voltage, which is equal to the current ratio.* The SWR is a measure of the mismatch between the load and the line, and is the first and most important quantity calculated for a particular load. The SWR is equal to unity (a desirable condition) when the load is perfectly matched. When the line is terminated in a purely resistive load, the standing-wave ratio is given by

$$\text{SWR} = Z_0/R_L \quad \text{or} \quad \text{SWR} = R_L/Z_0 \quad \text{(whichever is larger)} \tag{7-8}$$

where R_L is the load resistance.

It is customary to put the larger quantity in the numerator of the fraction, so that the ratio will always be greater than 1. Regardless of whether the load resistance is half as large or twice as large as the line characteristic impedance, the ratio of a voltage maximum to a voltage minimum is 2:1, and the degree of mismatch is the same in both cases.

If the load is purely reactive, SWR will be infinity. The same condition will apply for a short-circuit or an open-circuit termination. Since in all three cases no power is absorbed, the reflected wave has the same size as the forward wave. Somewhere along the line complete cancellation will occur, giving a voltage zero, and hence SWR must be infinite. When the load is complex, SWR can still be computed, but it is much easier to determine it from a transmission-line calculator, or to measure it.

The higher the SWR, the greater the mismatch between line and load or, for that matter, between generator and line. In practical lines, power loss increases with SWR, and so a low value of standing-wave ratio is always sought, except when the transmission line is being used as a pure reactance or as a tuned circuit. This will be shown in Section 7-1.5.

Normalization of impedance It is customary to *normalize* an impedance with respect to the line to which it is connected, i.e., to divide this impedance by the characteristic impedance of the line, as

$$z_L = \frac{Z_L}{Z_0} \tag{7-9}$$

thus obtaining the normalized impedance. (Note that the normalized impedance is a dimensionless quantity, not to be measured or given in ohms.) This is very useful because the behavior of the line depends not on the absolute magnitude of the load impedance, but on its value relative to Z_0. This fact can be seen from Equation (7-8); the SWR on a line will be 2 regardless of whether $Z_0 = 75\ \Omega$ and $R_L = 150\ \Omega$ or $Z_0 = 300\ \Omega$ and $R_L = 600\ \Omega$. The normalizing of impedance opens up possibilities for transmission-line charts. It is similar to the process used to obtain the universal response curves for tuned circuits and *RC*-coupled amplifiers.

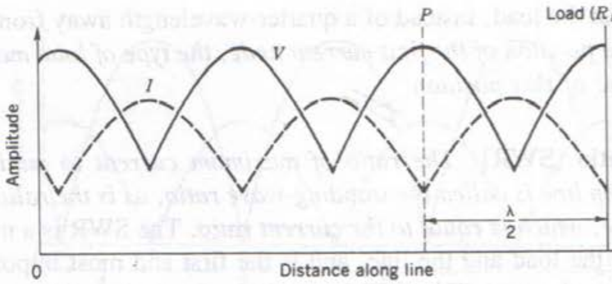


FIGURE 7-7 Lossless line terminated in a pure resistance greater than Z_0 (note that voltage SWR equals current SWR).

Consider a pure resistance connected to a transmission line, such that $R_L \neq Z_0$. Since the voltage and current vary along the line, as shown in Figure 7-7, so will the resistance or impedance. However, conditions do repeat every half-wavelength, as already outlined. The impedance at P will be equal to that of the load, if P is a half-wavelength away from the load and the line is lossless.

7-1.5 Quarter- and Half-Wavelength Lines

Sections of transmission lines that are exactly a quarter-wavelength or a half-wavelength long have important impedance-transforming properties, and are often used for this purpose at radio frequencies. Such lines will now be discussed.

Impedance inversion by quarter-wavelength lines Consider Figure 7-8, which shows a load of impedance Z_L connected to a piece of transmission line of length s and having Z_0 as its characteristic impedance. When the length s is exactly a quarter-wavelength line (or an odd number of quarter-wavelengths) and the line is lossless, then the impedance Z_s, seen when looking toward the load, is given by

$$Z_s = \frac{Z_0^2}{Z_L} \tag{7-10}$$

This relationship is sometimes called *reflective impedance;* i.e., the quarter-wavelength reflects the opposite of its load impedance. Equation (7-10) represents a very important and fundamental relation, which is somewhat too complex to derive here, but whose truth may be indicated as follows. Unless a load is resistive and equal to the characteristic impedance of the line to which it is connected, standing waves of voltage and current are set up along the line, with a node (and antinode) repetition rate

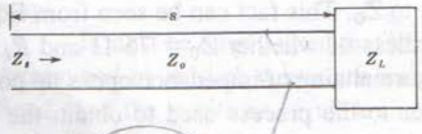


FIGURE 7-8 Loaded line.

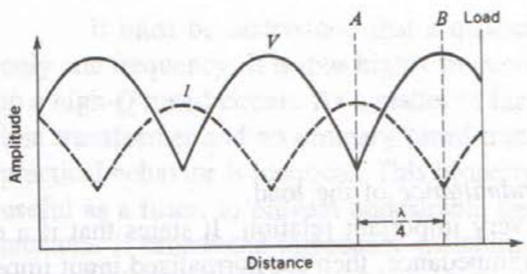


FIGURE 7-9 Standing waves along a mismatched transmission line; impedance inversion.

of $\lambda/2$. This has already been shown and is indicated again in Figure 7-9. Note that here the voltage and current minima are not zero; the load is not a short circuit, and therefore the standing-wave ratio is not infinite. Note also that the current nodes are separated from the voltage nodes by a distance of $\lambda/4$, as before.

It is obvious that at the point A (voltage node, current antinode) the line impedance is low, and at the point B (voltage antinode, current node) it is the reverse, i.e., high. In order to change the impedance at A, it would be necessary to change the SWR on the line. If the SWR were increased, the voltage minimum at A would be lower, and so would be the impedance at A. The size of the voltage maximum at B would be increased, and so would the impedance at B. Thus an increase in Z_B is accompanied by a decrease in Z_A (if A and B are $\lambda/4$ apart). This amounts to saying that *the impedance at A is inversely proportional to the impedance at B*. Equation (7-10) states this relation mathematically and also supplies the proportionality constant; this happens to be the square of the characteristic impedance of the transmission line. The relation holds just as well when the two points are not voltage nodes and antinodes, and a glance at Figure 7-9 shows that it also applies when the distance separating the points is three, five, seven and so on, quarter-wavelengths.

Another interesting property of the quarter-wave line is seen if, in Equation (7-10), the impedances are normalized with respect to Z_0. Dividing both sides by Z_0, we have

$$\frac{Z_s}{Z_0} = \frac{Z_0}{Z_L} \tag{7-11}$$

but

$$\frac{Z_s}{Z_0} = z_s$$

and

$$\frac{Z_L}{Z_0} = z_L$$

whence $Z_0/Z_L = 1/z_L$.

Substituting these results into Equation (8-11) gives

$$z_s = \frac{1}{z_L} = y_L \tag{7-12}$$

where y_L is the *normalized admittance* of the load.

Equation (7-12) is a very important relation. It states that if a quarter-wavelength line is connected to an impedance, then the normalized input impedance of this line is equal to the normalized load admittance. Both must be normalized with respect to the line. Note that there is no contradiction here, since all normalized quantities are dimensionless. Note also that this relation is quite independent of the characteristic impedance of the line, a property that is very useful in practice.

Quarter-wave transformer and impedance matching In nearly all transmission-line applications, it is required that the load be matched to the line. This involves the tuning out of the unwanted load reactance (if any) and the transformation of the resulting impedance to the value required. Ordinary RF transformers may be used up to the middle of the VHF range. Their performance is not good enough at frequencies much higher than this, owing to excessive leakage inductance and stray capacitances. The quarter-wave line provides unique opportunities for impedance transformation up to the highest frequencies and is compatible with transmission lines.

Equation (7-10) shows that the impedance at the input of a quarter-wave line depends on two quantities: these are the load impedance (which is fixed for any load at a constant frequency) and the characteristic impedance of the interconnecting transmission line. If this Z_0 can be varied, the impedance seen at the input to the $\lambda/4$ transformer will be varied accordingly, and the load may thus be matched to the characteristic impedance of the main line. This is similar to varying the turns ratio of a transformer to obtain a required value of input impedance for any given value of load impedance.

EXAMPLE 7-4 **It is required to match a 200-Ω load to a 300-Ω transmission line, to reduce the SWR along the line to 1. What must be the characteristic impedance of the quarter-wave transformer used for this purpose, if it is connected directly to the load?**

SOLUTION

Since the condition SWR = 1 is wanted along the main line, the impedance Z_s at the input to the $\lambda/4$ transformer must equal the characteristic impedance Z_0 of the main line. Let the transformer characteristic impedance be Z_0'; then, from Equation (7-10),

$$Z_s = \frac{Z_0'^2}{Z_L} = Z_0 \quad \text{(of main line)}$$

$$Z_0' = \sqrt{Z_0 Z_L} \tag{7-13}$$

$$= \sqrt{200 \times 300} = 245\Omega$$

Equation (7-13) was derived for this exercise, but it is universal in application and quite important.

It must be understood that a quarter-wave transformer has a length of $\lambda/4$ at only one frequency. It is thus highly frequency-dependent, and is in this respect similar to a high-Q tuned circuit. As a matter of fact, the difference between the transmission-line transformer and an ordinary tuned transformer is purely one of construction, the practical behavior is identical. This property of the quarter-wave transformer makes it useful as a filter, to prevent undesirable frequencies from reaching the load, often an antenna. If broadband impedance matching is required, the transformer must be constructed of high-resistance wire to lower its Q, thereby increasing bandwidth.

It should be mentioned that the procedure becomes somewhat more involved if the load is complex, rather than purely resistive as so far considered. The quarter-wave transformer can still be used, but it must now be connected at some precalculated distance from the load. It is generally connected at the nearest resistive point to the load, whose position may be found with the aid of a transmission-line calculator, such as a *Smith chart*.

Half-wavelength line As was mentioned previously, the reflected impedance is an important characteristic of the matching process; the half-wavelength line reflects its load impedance directly. A half-wave transformer has the property that the input impedance must be equal to the impedance of the load placed at the far end of the half-wave line. This property is independent of the characteristic impedance of this line, but once again it is frequency-dependent.

The advantages of this property are many. For instance, it is very often not practicable to measure the impedance of a load directly. This being the case, the impedance may be measured along a transmission line connected to the load, at a distance which is a half-wavelength (or a whole number of half-wavelengths) from the load. Again, it is sometimes necessary to short-circuit a transmission line at a point that is not physically accessible. The same results will be obtained if the short circuit is placed a half-wavelength (etc.) away from the load. Yet again, if a short-circuited half-wave transmission line is connected across the main line, the main line will be short-circuited at that point, but only at the frequency at which the shunt line is a half-wavelength. That frequency will not pass this point, but others will, especially if they are farther and farther away from the initial frequency. The short-circuited shunt half-wave line has thus become a band-stop filter. Finally, if the frequency of a signal is known, a short-circuited transmission line may be connected to the generator of this frequency, and a half-wavelength along this line may be measured very accurately. From the knowledge of frequency and wavelength, the velocity of the wave along the line can be calculated. We may determine the velocity factor, and therefore the dielectric constant of the insulation, as discussed in Section 7-1.3.

b. Explain how a smith chart can be used for the calculation of the following: (8)
(i) Admittance
(ii) Impedance
(iii) VSWR

Answer:

7-2.1 Fundamentals of the Smith Chart

Description The polar impedance diagram, or Smith chart as it is more commonly known, is illustrated in Figure 7-12. It consists of two sets of circles, or arcs of circles, which are so arranged that various important quantities connected with mismatched transmission lines may be plotted and evaluated fairly easily. The complete circles, whose centers lie on the only straight line on the chart, correspond to various values of normalized resistance ($r = R/Z_0$) along the line. The arcs of circles, to either side of the straight line, similarly correspond to various values of normalized line reactance $jx = jX/Z_0$. A careful look at the way in which the circles intersect shows them to be

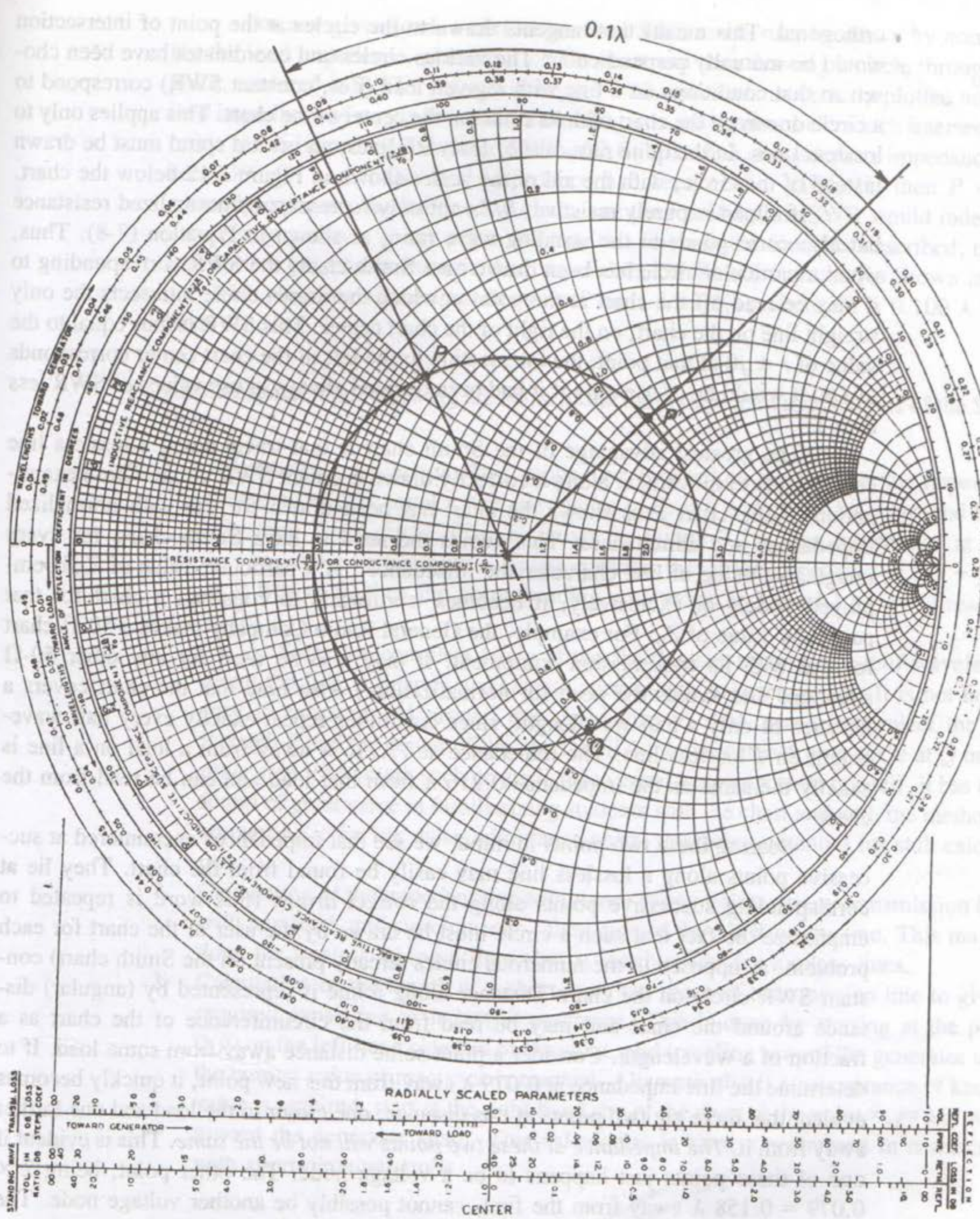


FIGURE 7-12 Smith chart.

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orthogonal. This means that tangents drawn to the circles at the point of intersection would be mutually perpendicular. The various circles and coordinates have been chosen so that conditions on a line with a given load (i.e., constant SWR) correspond to a circle drawn on the chart with its center at the center of the chart. This applies only to lossless lines. In the quite rare case of *lossy* RF lines, an inward spiral must be drawn instead of the circle, with the aid of the scales shown in Figure 7-12 below the chart.

If a load is purely resistive, R/Z_0 not only represents its normalized resistance but also corresponds to the standing-wave ratio, as shown in Equation (7-8). Thus, when a particular circle has been drawn on a Smith chart, the SWR corresponding to it may be read off the chart at the point at which the drawn circle intersects the only straight line on the chart, on the right of the chart center. This SWR is thus equal to the value of $r \pm j0$ at that point; the intersection to the left of the chart center corresponds to $1/r$. It would be of use only if it had been decided always to use values of SWR less than 1.

The greatest advantage of the Smith chart is that travel along a lossless line corresponds to movement along a correctly drawn constant SWR circle. Close examination of the chart axes shows the chart has been drawn for use with normalized impedances and admittances. This avoids the need to have Smith charts for every imaginable value of line characteristic impedance. (If a particular value of Z_0 is employed widely or exclusively, it becomes worthwhile to construct a chart for that particular value of Z_0. For example, the General Radio Company makes a 50-Ω chart for use with its transmission equipment. It may also be used for any other 50-Ω situations and avoids the need for normalization.) Also note that the chart covers a distance of only a half-wavelength, since conditions repeat exactly every half-wavelength on a lossless line. The impedance at 17.716 λ away from a load on a line is exactly the same as the impedance 0.216 λ from that load and can be read from the chart.

Bearing these two points in mind, we see that impedances encountered at successive points along a lossless line may easily be found from the chart. They lie at corresponding successive points along the correct *drawn* (this word is repeated to emphasize the fact that such a circle must be drawn by the user of the chart for each problem, as opposed to the numerous circles already present on the Smith chart) constant SWR circle on the chart. Distance along a line is represented by (angular) distance around the chart and may be read from the circumference of the chart as a fraction of a wavelength. Consider a point some distance away from some load. If to determine the line impedance at 0.079 λ away from this new point, it quickly becomes evident that there are two points at this distance, one closer to the load and one farther away from it. *The impedance at these two points will not be the same*. This is evident if one of these points just happens to be a voltage node. The other point, being 2 × 0.079 = 0.158 λ away from the first, cannot possibly be another voltage node. The same reasoning applies in all other situations. The direction of movement around a constant SWR circle is also of importance. The Smith chart has been standardized so that movement away from the load, i.e., toward the generator, corresponds to clockwise motion on the chart. Movement toward the load corresponds to counterclockwise motion; this is always marked on the rim of commercial Smith charts and is shown in Figure 7-12.

For any given load, a correct constant SWR circle may be drawn by normalizing the load impedance, plotting it on the chart and then drawing a circle through this point, centered at 0. The point P in Figure 7-12 represents a correctly plotted normalized impedance of $z = 0.5 + j0.5$. Since it lies on the drawn circle which intersects the r axis at 2.6, it corresponds to an SWR of 2.6. If the line characteristic impedance had been 300 Ω, and if the load impedance had been $(150 + j150)$ Ω, then P would correctly represent the load on the chart, and the resulting line SWR would indeed be 2.6. The impedance at any other point on this line may be found as described, by the appropriate movement from the load around the SWR = 2.6 circle. As shown in Figure 7-12, the normalized impedance at P' is $1.4 + j1.1$, where P' is $0.100\ \lambda$ away from the load.

Applications The following are some of the more important applications of the Smith chart:

1. Admittance calculations. This application is based on the fact that the impedance measured at Q is equal to the admittance at P, if P and Q are $\lambda/4$ apart and lie on the same SWR circle. This is shown in Figure 7-12. The impedance at Q is $1 - j1$, and a very simple calculation shows that if the impedance is $0.5 + j0.5$, as it was at P, then the corresponding admittance is indeed $1 - j1$, as read off at Q.

   Since the complete circle of the Smith chart represents a half-wavelength along the line, a quarter-wavelength corresponds to a semicircle. It is not necessary to measure $\lambda/4$ around the circle from P, but merely to project the line through P and the center of the chart until it intersects the drawn circle at Q on the other side. (Although such an application is not very important in itself, it has been found of great value in familiarizing students with the chart and with the method of converting it for use as an admittance chart, this being essential for stub calculations.)
2. Calculation of the impedance or admittance at any point, on any transmission line, with any load, and simultaneous calculation of the SWR on the line. This may be done for lossless or lossy lines, but is much easier for lossless lines.
3. Calculation of the length of a short-circuited piece of transmission line to give a required capacitive or inductive reactance. This is done by starting at the point $0, j0$ on the left-hand side rim of the chart, and traveling toward the generator until the correct value of reactance is reached. Alternatively, if a susceptance of known value is required, start at the right-hand rim of the chart at the point $\infty, j\infty$ and work toward the generator again. This calculation is always performed in connection with short-circuited stubs.

Q. 7 a. What are Waveguides? Briefly describe the working principle of a Waveguide by explaining the propagation of waves in it? Explain how a section of Rectangular Waveguide depends upon the frequency of the signal? (8)

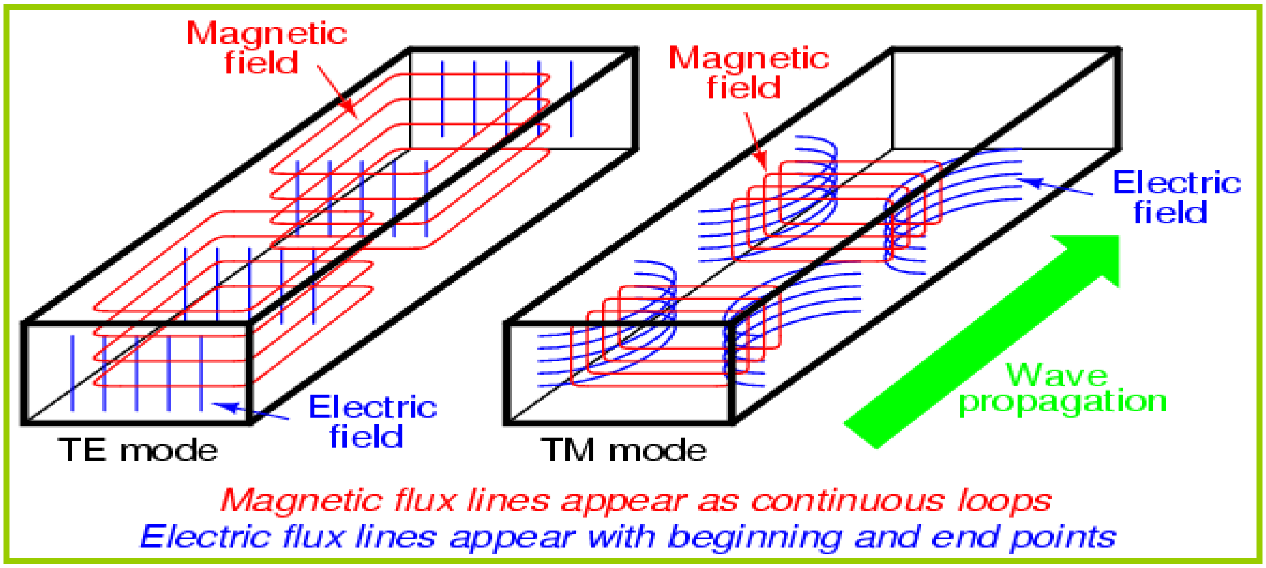
Answer: A waveguide is a structure that guides waves, such as electromagnetic waves or sound waves. There are different types of waveguides for each type of wave. The original and most common meaning is a hollow conductive metal pipe used to carry high frequency radio waves, particularly microwaves. The frequency of the transmitted wave also dictates the shape of a waveguide: an optical fiber guiding high-frequency light will not guide microwaves of a much lower frequency. As a rule of thumb, the width of a waveguide needs to be of the same order of magnitude as the wavelength of the guided wave.

Principle of operation: Waves propagate in all directions in open space as spherical waves. The power of the wave falls with the distance R from the source as the square of the distance (inverse square law). A waveguide confines the wave to propagate in one dimension, so that, under ideal conditions, the wave loses no power while propagating. The conductors generally used in waveguides have small skin depth and hence large surface conductance. Due to total reflection at the walls, waves are confined to the interior of a waveguide. The propagation inside the waveguide, hence, can be described approximately as a "zigzag" between the walls. This description is exact for electromagnetic waves in a hollow metal tube with a rectangular or circular cross-section.

Propagation of electromagnetic waves in the waveguide

If energy is fed into a waveguide, then an electric field (E- field) is formed in the center of the wider wall “a”. This electric field is strongest in the center of the waveguide and decreases in the direction of the narrower wall “b”. It has a sinusoidal shape in cross section. By the electric field also arises a magnetic field. However, the magnetic field can not stand vertically on a metallic conductor. As propagation direction remains only the direction which is passed through the waveguide.

The electric field changes over time with the frequency, and has in the longitudinal direction of the waveguide maxima and minima in the distance of half the wavelength. High frequency energy that is fed into a waveguide, generates an electromagnetic transverse wave (TEM mode) whose electric and magnetic fields are perpendicular to each other. The electric field is established between the two wider waveguide walls, the magnetic field lies between the two narrower walls. These fields do not remain in the respective states. Considered over the timeline, they change the intensity and polarity in the rhythm of the input signal. This electromagnetic wave propagates in the waveguide at near-light speed. The electric and the magnetic field change its strength and polarity permanently, but they are always perpendicular to each other locally. If the electric field is in propagation direction, it is called an E-wave or TM wave (**T**ransverse **M**agnetic). If the magnetic field is in propagation direction, it is called an H-wave or TE wave (**T**ransverse **E**lectric).



Dimensions of the waveguide which determines the operating frequency range

1.The size of the waveguide determines its operating frequency range.

2. The frequency of operation is determined by the dimension 'a'.

3. This dimension is usually made equal to one – half the wavelength at the lowest frequency of operation, this frequency is known as the waveguide *cutoff frequency.*

4. At the cutoff frequency and below, the waveguide will not transmit energy. At frequencies above the cutoff frequency, the waveguide will propagate energy.

b. Rectangular Waveguide is having inside dimensions of 5 × 2 cms. Calculate the cutoff frequency with a dominant mode of TE1,0? (8)

Answer: For a standard rectangular waveguide, the cutoff wavelength is given by,

$$\lambda_c = \frac{2}{\sqrt{\left(\frac{m}{a}\right)^2 + \left(\frac{n}{b}\right)^2}}$$

Where m and n represent the dominant modes;

a and b are the dimensions of the waveguide.

Given: a= 5cm; b=2 cm; m= 1; n=0

$$\lambda_c = \frac{2}{\sqrt{\left(\frac{m}{a}\right)^2 + \left(\frac{n}{b}\right)^2}}$$

Cut off freq= 10 GHZ

Q.8 a. Explain with a block diagram, how demodulation of PPM pulses can be achieved. List the advantages and disadvantages of PPM, over other type of systems. (8)

Answer:

9.4.2 Demodulation of PPM Signal

In the case of pulse-position modulation, it is customary to convert the received pulses that vary in position to pulses that vary in length. One way to achieve this is illustrated in Fig. 9.29.

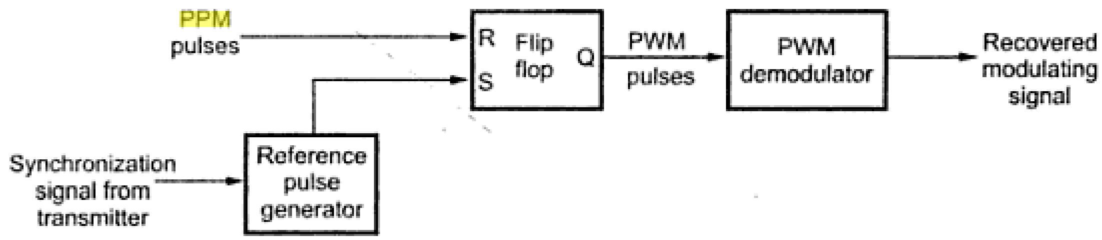


Fig. 9.29 PPM demodulator

As shown in the Fig. 9.29 flip-flop circuit is set or turned 'ON' (giving high output) when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flip-flop circuit is reset or turned 'OFF' (giving low output) at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip-flop.

The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

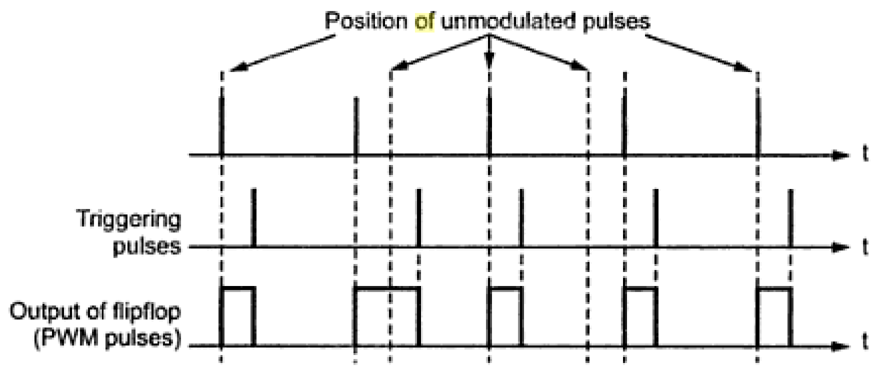


Fig. 9.30 Demodulation waveforms for PPM

9.4.3 Advantages of PPM

1. Like PWM, in PPM amplitude is held constant thus less noise interference.
2. Like PWM, signal and noise separation is very easy.
3. Due to constant pulse widths and amplitudes, transmission power for each pulse is same.

9.4.4 Disadvantages of PPM

1. Synchronization between transmitter and receiver is required.
2. Large bandwidth is required as compared to PAM.

b. What is Information Theory and Coding of Information? Briefly describe Baudot Code? (8)

Answer: Information theory is a branch of applied mathematics, electrical engineering, and computer science involving the quantification of information. Information theory was developed by Claude E. Shannon to find fundamental limits on signal processing operations such as compressing data and on reliably storing and communicating data. The main concepts of information theory are, first, that a "message" conveys a quantifiable amount of information to the recipient, and, second, that a "channel" (over which messages are transmitted) has a quantifiable information "capacity" defining the limiting rate at which it can convey information. An underlying concept is that information must be news to the recipient; in particular, telling the recipient something the recipient already knows conveys no information. The object of coding theory is to encode messages so as to send them over a channel at as high a rate as possible (within the limits of the channel capacity).

Coding theory is the study of the properties of codes and their fitness for a specific application. Codes are used for data compression, cryptography, error-correction and more recently also for network coding. Codes are studied by various scientific disciplines—such as information theory, electrical engineering, mathematics, and computer science—for the purpose of designing efficient and reliable data transmission methods. This typically involves the removal of redundancy and the correction (or detection) of errors in the transmitted data. There are four types of coding:[1]

1. Data compression (or, *source coding*)
2. Error correction (or *channel coding*)
3. Cryptographic coding
4. line coding

The **Baudot code**, invented by Émile Baudot,[1] is a character set predating EBCDIC and ASCII. Baudot invented his original code in 1870 and patented it in 1874. It was a 5-bit code, with equal on and off intervals, which allowed telegraph transmission of the Roman alphabet and punctuation and control signals.

Baudot code uses :

- 5 bits to represent information
- mostly paper tape or punch cards
- has no parity bit
- it has only 58 characters
- two characters of unique functions: Letter shift -change one type of letters to another type and Figure shift - a shift results all letters are treated as upper cases

Disadvantage:

- it has no unique representation code of all symbol
- still commonly used technique in
- telegraph, teletypwriter & telex

Q.9 a. Describe the elements of Long- Distance telephony? (8)

Answer:

Wire and Wireless Telephone Systems.—The production of sound depends upon setting the air into suitable vibration. When we speak the vocal chords or membranes located down in the throat vibrate and set the column of air in the throat into vibration and these vibrations are in turn transmitted into the surrounding air. The air waves thus formed beat upon the membrane of the ear and cause it to vibrate in the same manner as the vocal chords did to produce them. The vibration of the membrane of the ear, or the ear drum as we more commonly call it, affects the nerves of the ear and the message is in turn transmitted to the brain.

Transmission over wire or wireless systems is quite similar to the above procedure but instead of using air waves, waves of electric current are employed. Fig. 1 shows a typical telephone circuit which for the sake of simplicity is arranged to transmit only in one direction. M is a telephone transmitter or microphone as it is sometimes called. B is a battery which supplies the force necessary to send the current around the electric circuit comprising the line wires L_1 and L_2; R is the telephone receiver. The essential parts of a telephone transmitter are a thin sheet of material, usually metal, called the diaphragm, which is arranged so as to be set into motion when sound waves impinge upon it; and some type of variable resistance contact arranged to be operated by the motion of the diaphragm. The variable resistance contact usually consists of a small chamber in

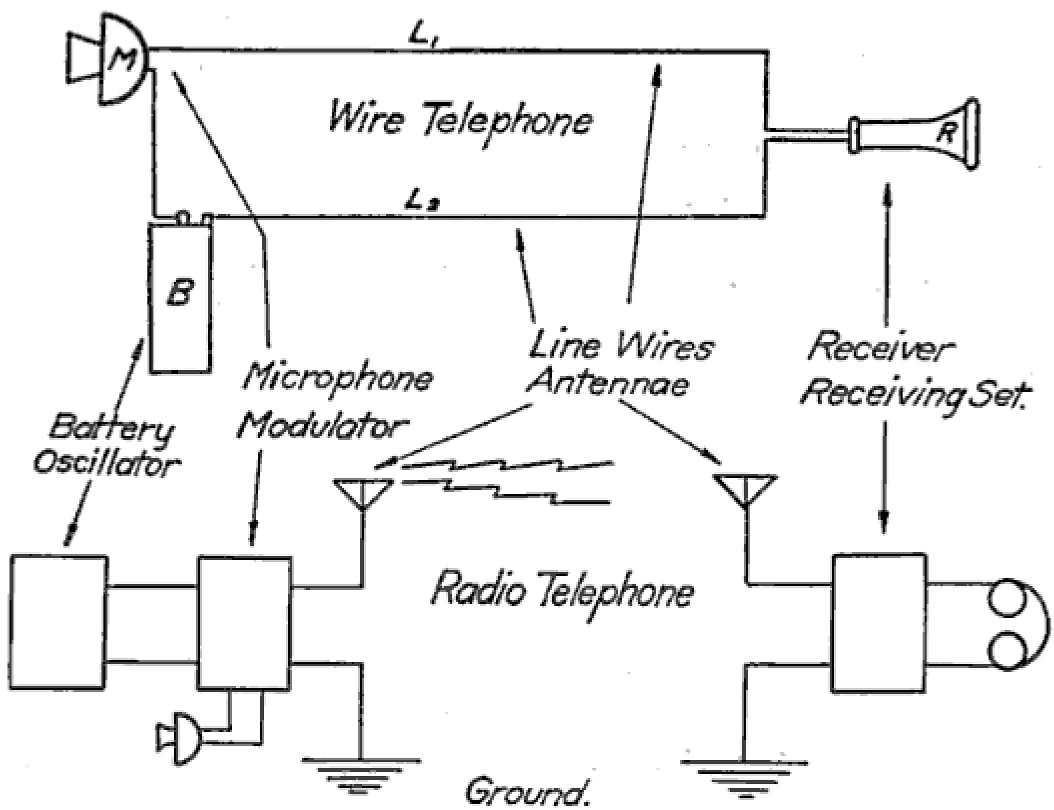


Fig. 1.—Graphical comparison of wire and radio telephone systems.

which two carbon discs are situated and insulated from one another. One disc is stationary and the other is attached to the diaphragm and free to move with it. The space between the two discs is loosely packed with carbon grains about the size of a pin head. The carbon granules have the property of varying the electrical resistance between the two carbon discs, which resistance depends upon the degree of pressure applied to the granular carbon. When the carbon granules are tightly compressed their resistance is low and this allows the battery to send a comparatively large current through the circuit. When pressure is removed the resistance increases and the current is correspondingly reduced. Thus the sound waves impressed on the transmitter diaphragm are reproduced in the electrical circuit as changes in the strength of the electric current.

The receiver consists of a thin sheet of soft iron so arranged as to be attracted by an electromagnet. Sometimes this magnet is a combination of permanent and electromagnet and sometimes the permanent magnet is omitted. The coil of wire wound around the iron core constituting the electromagnet has the property of attracting the diaphragm with varying degrees of force depending upon the current through the coil. Since the diaphragm is flexible it will move in response to these changes in pull and thus send out an air wave which resembles to a greater or less degree the current through the coil. The essential features in wire telephony are a source of power, the battery; a modulator of the power source so that the output varies according to the sound to be transmitted, in this case the transmitter or microphone; means for transmitting the modulated power output from the transmitting to the receiving station, represented by the line wires; and finally a device to change the electrical impulses back to sound waves, which is the function of the receiver.

The operation of radio or wireless telephone systems requires corresponding units similar to those already enumerated and which have exactly the same functions. The principal point of difference is that in order to transmit power through the ether, which we have to employ in place of the line wires, it is necessary to use high-frequency energy, and hence the power source will have to be one capable of producing high-frequency currents and the receiving system sensitive to high-frequency currents instead of currents such as are produced in wire telephony. Thus for radio telephony the battery will be replaced by a high-frequency current generator, the microphone by a modulation system, the line wires by the two antennae and the intervening ether, and the receiver by special receiving apparatus sensitive to high-frequency currents.

b. What is Multiplexing and what were the reasons for developing it? What are its two basic forms of Multiplexing? (8)

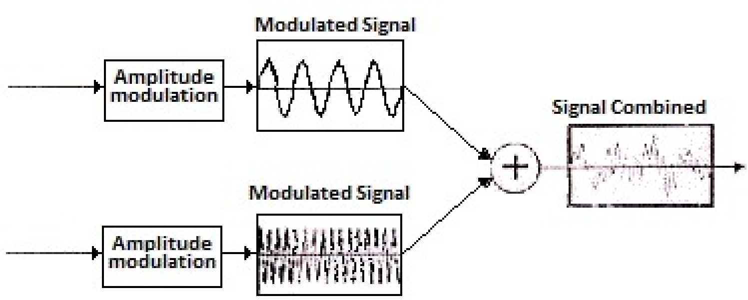
Answer:

Multiplexing is sending multiple signals or streams of information on a carrier at the same time in the form of a single, complex signal and then recovering the separate signals at the receiving end.

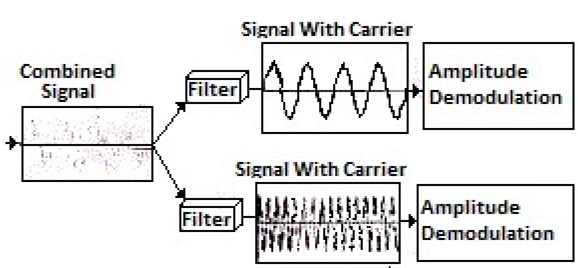
Multiplexing technique is designed to reduce the number of electrical connections or leads in the display matrix. Whereas driving signals are applied not to each pixel (picture element) individually but to a group of rows and columns at a time. Besides reducing the number of individually independent interconnections, multiplexing also simplifies the drive electronics, reduces the cost and provides direct interface with the microprocessors. There are limitations in multiplexing due to complex electro-optical response of the liquid crystal cell. However, fairly reasonable level of multiplexing can be achieved by properly choosing the multiplexing scheme, liquid crystal mixture and cell designing

FREQUENCY-DIVISION MULTIPLEXING (FDM)

In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. The carrier frequencies have to be different enough to accommodate the modulation and demodulation signals. The figure No.8.3 illustrates the FDM multiplexing process. The multiplexing process starts by applying amplitude modulation into each signal by using different carrier frequencies as/i and/j. Then both signals are combined



In demultiplexing process, we use filters to decompose the multiplexed signal into its constituent component signals. Then each signal is passed to an amplitude demodulation process to separate the carrier signal from the message signal. Then, the message signal is sent to the waiting receiver. The process of demultiplexing is shown in the figure.

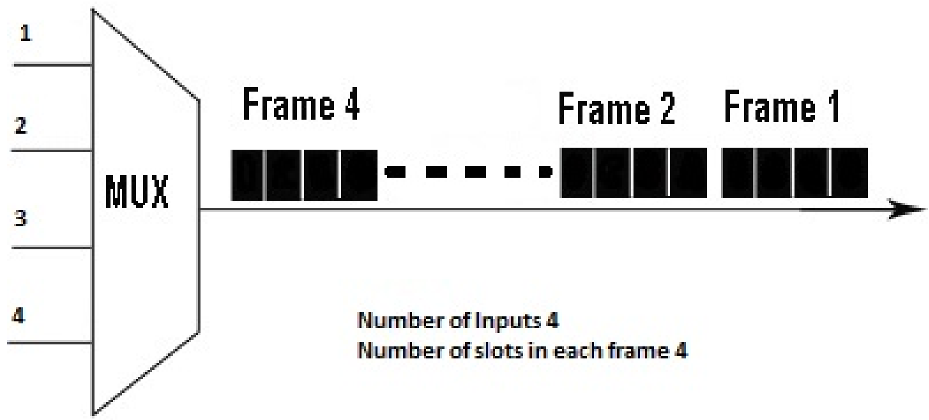


TIME-DIVISION MULTIPLEXING (TDM)

In the time-division multiplexing, multiple transmissions can occupy a single link by subdividing them and interleaving the portions. We say that TDM is a round robin use of a frequency. TDM can be implemented in two ways: synchronous TDM and asynchronous TDM.

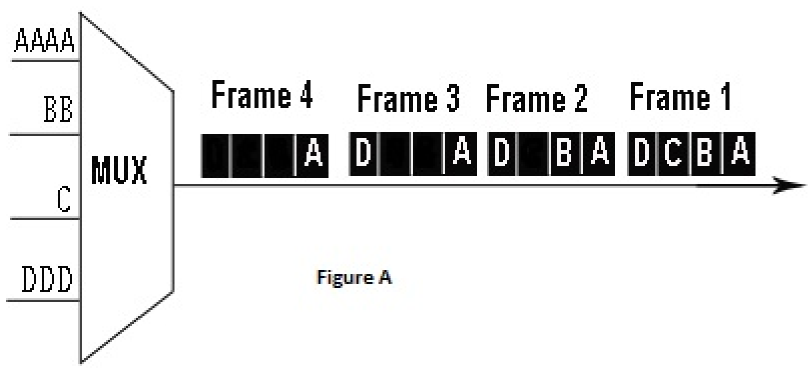
1. Synchronous TDM

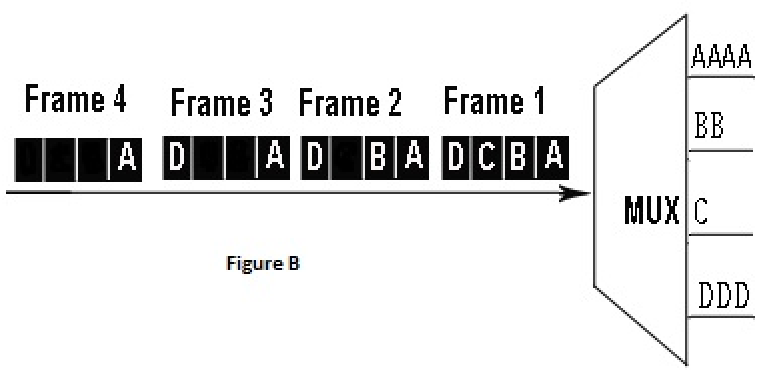
The multiplexer allocates exactly the same time slot to each device at all times, whether or not a device has anything to transmit. Time slot 1, for example, is assigned to device 1 alone and cannot be used by any other device as shown in the figure



Frames: In synchronous TDM, a frame consists of one complete cycle of time slots. Thus the number of slots in frame is equal to the number of inputs.

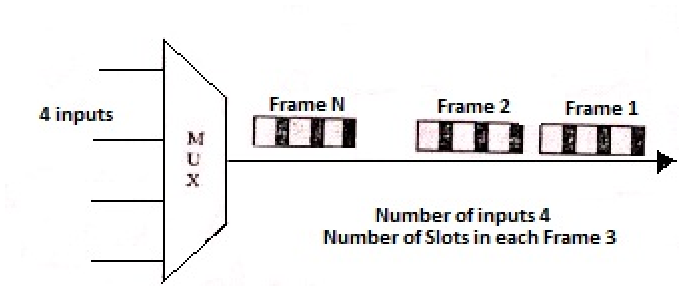
Figures (A & B) below are an example of how/the synchronous TDM works.





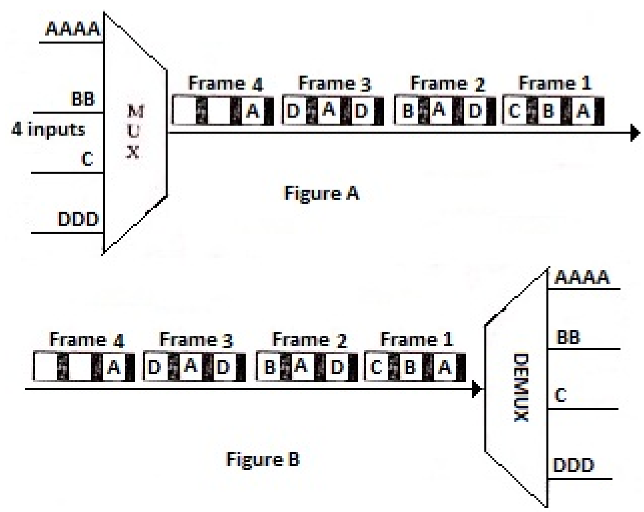
2. Asynchronous TDM

In asynchronous TDM, each slot in a frame is not dedicated to the fix device. Each slot contains an index of the device to be sent to and **a** message. Thus, the number of slots in a frame is not necessary to be equal to the number of input devices. More than one slot in a frame can be allocated for an input device Asynchronous TDM allows maximization the link. It allows a number of lower speed input lines to be multiplexed to a single higher speed line. As shown in the figure



Frames: In asynchronous TDM, a frame contains a fix number of time slots. Each slot has an index of which device to receive.

Figures (A& B) are examples of how asynchronous TDM works.



<u>**Text Book**</u>

Electronic Communication Systems, George Kennedy and Bernard Davis, Fourth Edition, Tata McGraw Hill Publishing Company Ltd.