ELEMENTS OF COMMUNICATION SYSTEM

Multiple Choice Type Questions

1. The positive RF peaks of an AM voltage rise to a maximum value of 12V and drop to a minimum value of 4V. The modulation index assuming single tone modulation is [WBUT 2013, 2014, 2017]

a) 3 b) $\frac{1}{3}$ c) $\frac{1}{4}$ d) $\frac{1}{2}$

Answer: (d)

2. If the modulation index of an AM wave is changed from 0 to 1, the transmitted power is [WBUT 2013, 2014, 2017]

a) unchanged b) halved
c) doubled d) increased by 50%

Answer: (d)

3. The maximum frequency deviation in commercial FM is [WBUT 2013, 2014, 2017]

a) 88 MHz b) 108 MHz c) 75 kHz d) 15 kHz

Answer: (c)

4. In commercial FM broadcasting, there is a channel 98.3. Here '98.3' represents [WBUT 2013, 2014, 2017]

a) modulating frequency in MHz b) carrier frequency in MHz
c) image frequency in kHz d) frequency deviation in kHz

Answer: (b)

5. PWM signal can be generated by [WBUT 2013, 2014, 2017]

a) a monostable multi-vibrator b) a astable multi-vibrator
c) integrating the PPM signal d) differentiating the PPM signal

Answer: (a)

6. The major practical problem of FDM system is [WBUT 2013, 2014, 2017]

a) amplitude distortion b) fading c) cross-talk d) shot noise

Answer: (c)

7. The spectral density of white noise is [WBUT 2015]

a) exponential b) uniform c) poisson d) Gaussian

Answer: (d)

8. The maximum transmission efficiency of AM is [WBUT 2016]

a) 33.33% b) 50% c) 66.6% d) 100%

Answer: (a)

9. What is the frequency swing of an FM broadcast transmitter when the modulation is 60% [WBUT 2016, 2018]

a) 20 kHz b) 45 kHz c) 30 kHz d) 10 kHz

Answer: (b)

10. What is the BW of AM wave message BW is Wm? [WBUT 2016]

a) Wm b) 2Wm c) 3Wm d) 4Wm

Answer: (b)

11. Varactor diode used for [WBUT 2016]

a) FM generation b) AM generation
c) PM generation d) all of these

Answer: (a)

12. The amplitude modulator works on the principle of [WBUT 2018]

a) Multiplication b) Addition c) Subtraction d) Division

Answer: (a)

13. Which of this frequency components contain the message information? [WBUT 2018]

a) Upper and lower sideband b) Upper sideband
c) Lower sideband d) None of these

Answer: (a)

14. A carrier of 100W is modulated to a depth of 50%. The total transmitted power is [WBUT 2018]

a) 112.5 W b) 125 W c) 150 W d) 100 W

Answer: (b)

15. If the deviation is 75 kHz and maximum modulating frequency is 5 kHz, what is the bandwidth of an FM wave? [WBUT 2018]

a) 80 kHz b) 160 kHz c) 40 kHz d) 320 kHz

Answer: (a)

16. The frequency deviation in wideband FM system is [WBUT 2018]

a) 25 kHz b) 75 kHz c) 5 kHz d) 10 kHz

Answer: (b)

17. The modulation index of an FM is given by [MODEL QUESTION]

a) δ / f_m b) f_m / δ c) f_m d) $\delta \cdot f_m$

Answer: (a)

18. Time division multiplexing requires [MODEL QUESTION]

a) constant data transmission
b) transmission of data samples
c) transmission of data at random
d) transmission of data of only one measurand

Answer: (b)

19. A superheterodyne receiver with an IF of 450 kHz is tuned to a signal at 1200 kHz. The image frequency is [MODEL QUESTION]
   a) 750 kHz   b) 900 kHz   c) 1650 kHz   d) 2100 kHz

Answer: (d)

20. PAM signal can be demodulated by [MODEL QUESTION]
   a) a low pass filter   b) a band pass filter
   c) a high pass filter   d) none of these

Answer: (b)

21. The biggest disadvantage of PCM is [MODEL QUESTION]
   a) its inability to handle analog signals
   b) the high error rate which its quantizing noise introduces
   c) its incompatibility with TDM
   d) the large bandwidth that is required for it

Answer: (b)

22. PCM is preferred to PAM because of the [MODEL QUESTION]
   a) resistance to quantizing error   b) simplicity
   c) lower cost   d) superior noise immunity

Answer: (c)

23. A PAM signal can be demodulated using a [MODEL QUESTION]
   a) High pass filter   b) Low pass filter
   c) Band pass filter   d) None of these

Answer: (b)

24. The main advantage of TDM over FDM is that it [MODEL QUESTION]
   a) Needs less power   b) Needs less bandwidth
   c) Needs simple circuitry   d) Gives better S/N ratio

Answer: (c)

25. In time division multiplexing, each multiplexed signal occupies [MODEL QUESTION]
   a) The entire transmission bandwidth
   b) A fraction of the transmission bandwidth
   c) A bandwidth equal to the bandwidth of each input signal
   d) None of these

Answer: (a)

26. To separate channels in an FDM receiver, it is necessary to use [MODEL QUESTION]
   a) AND gates   b) Band pass filters
   c) Differentiation   d) Integration

Answer: (b)

27. In TDM system, the signals are separated from one another [MODEL QUESTION]

a) In time
b) In frequency
c) In time as well as frequency
d) In amplitude

Answer: (a)

28. N messages each band-limited to w are to be transmitted over a common channel. It is possible to achieve this objective using [MODEL QUESTION]

a) FDM
b) TDM
c) FDM as well as TDM
d) None of the above

Answer: (c)

29. In commercial FM broadcasting the maximum frequency deviation is normally [MODEL QUESTION]

a) 5kHz
b) 15 kHz
c) 75 kHz
d) 180 kHz

Answer: (c)

Short Answer Type Questions

1. State and prove sampling theorem for band limited signals. What is aliasing? [WBUT 2013]

OR,

State and explain Sampling theorem. [WBUT 2013]

Answer:

1st Part:

This theorem states that the sampling rate will be at least twice of the signal frequency.

Let, m(t) be a signal whose highest frequency is f_M can be reconstructed exactly (with no overlapping) from it's samples taken uniformly at a rate $R \geq 2f_M$ H_z (samples per second) i.e., minimum sampling frequency is $f_s = 2f_M$

Proof

The theorem requires that the sampling rate be rapid enough so that at least two samples are taken of highest frequency component f_M.

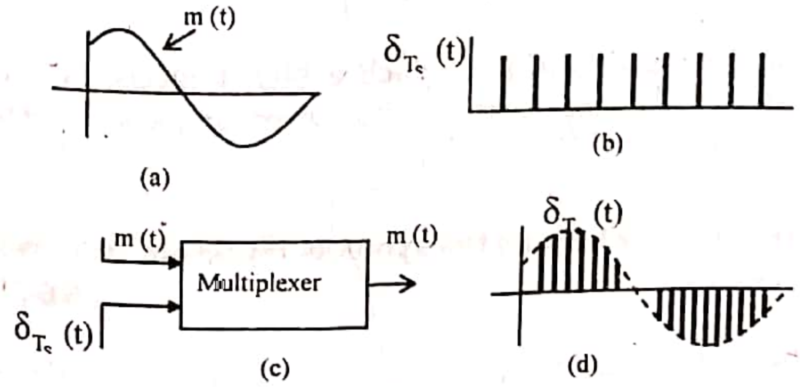


Fig: (a) Signal m(t) (b) Sampling impulse functions
(c) Sampling by a multiplier (d) Samples of m(t)

Let, T_s is the sampling time.

Fourier expression of impulse function train can be expressed as

$$\delta_{T_S}(t)=\frac{1}{T_S}.[1+2\cos\omega_s t+2\cos 2\omega_s t+2\cos 3\omega_s t+......]\text{where, }\omega_s=\frac{2\pi}{T_s}=2\pi f_s$$

The output of the multiplier yields sampled signal is,

$$\delta_{T_S}(t)\times m(t)=\frac{1}{T_s}[m(t)+2m(t)\cos\omega_s+2m(t)\cos 2\omega_s+2m(t)\cos 3\omega_s+......]$$

In the expression 1st term indicates a constant factor with a message signal, but 2nd term is a product of $m(t)$ and a sinusoid of frequency $2f_M$ which gives rise a DSB – SC signal, similarly, succeeding terms yields DSB – SC signals with carrier frequencies 4 f_m, 6 f_m, 8 f_m, etc.

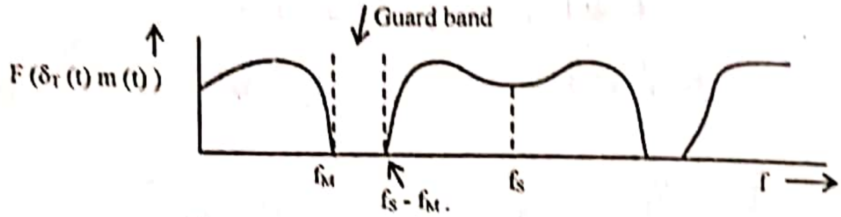


Fig: (a) When $f_S > 2\ f_M$ a guard band appears

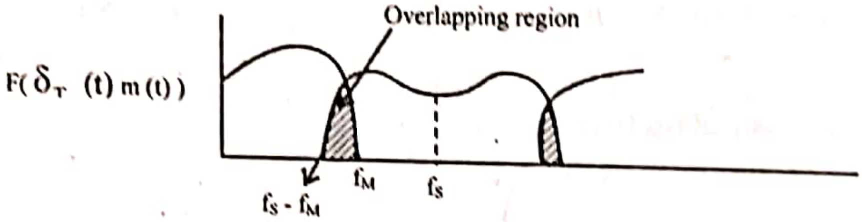


Fig: (b) Overlapping Spectra when, $f_S < 2\ f_M$.

The above figure shows that when $f_s < 2f_M$ overlapping occurs, when, $f_s = 2\ f_M$ then there overlapping is just avoided but when $f_s > 2f_M$ there is no overlapping at the same time a guard band separation is created which avoids the Inter Symbol Interference effects (ISI effect) which is a essential requirement for digital communication so, we can conclude that minimum sampling rate is $2f_M$. This minimum sampling rate is known as Nyquist rate.

2nd Part:

Aliasing is defined as the phenomenon in which a high frequency component in the frequency spectrum of signal takes identity of a lower frequency component in the spectrum of the sampled signal.

2. Prove that $P_t = P_c(1+m^2/2)$ where all the symbols represent their usual meaning.

[WBUT 2013, 2017]

Answer:

AM wave contains three terms: carrier term, LSB and USB. So total carrier power,

P = Carrier Power + Upper Side band power + Lower Side band power

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

$$P_c=\left(\frac{V_c}{\sqrt{2}}\right)^2=\frac{V_c^2}{2}$$

$$P_{USB}=\left(\frac{m_aV_c}{2\sqrt{2}}\right)^2=\frac{m_a^2V_c^2}{8}$$

$$P_{LSB}=\left(\frac{m_aV_c}{2\sqrt{2}}\right)^2=\frac{m_a^2V_c^2}{8}$$

So,

$$P=\frac{V_c^2}{2}+\frac{m_a^2V_c^2}{8}+\frac{m_a^2V_c^2}{8}=\frac{V_c^2}{2}+\frac{V_c^2m_a^2}{4}=\frac{V_c^2}{2}\left(1+\frac{m_a^2}{2}\right)=P_c\left(1+\frac{m_a^2}{2}\right)$$

Where $P_c=\frac{V_c^2}{2}$

3. Define the Carson's rule for FM bandwidth. **[WBUT 2014]**

OR,

Write short note on Carson's rule. **[WBUT 2016]**

An FM wave modulated to a depth of 8, generates a signal of BW of 180 kHz. Find the frequency deviation. **[WBUT 2014]**

Answer:

1st Part:

Carson's Rule

The rule of thumb (Carson's rule) states that the bandwidth frequency to pass an FM wave is twice the sum of the deviation and the highest modulating frequency.

This rule is only an approximation and gives accurate results if modulation index f_m is in excess of 6.

2nd Part:

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

or, $BW=2(m_f f_m+f_m)$ $(\because \Delta f=m_f f_m)$

or, $BW=2f_m(1+m_f)$

From Carson's Rule,

Bandwidth $=2(m_f+1)f_m=2(8+1)f_m=180\,KHz$

or, $f_m=\frac{180\,KHz}{18}=10\,KHz$

$\Delta f=$ frequency deviation $=m_f f_m=(8\times10)\,KHz=80\,KHz$

4. Why is double-sideband suppressed-carrier modulation called so? What is the bandwidth of DSB-SC? **[WBUT 2014]**

Answer:

1st Part:

The AM signal as derived in the previous section is given by

$$v_{AM} = V_c \sin\omega_c t + \frac{mV_c}{2}\cos(\omega_c - \omega_m)t - \frac{mV_c}{2}\cos(\omega_c + \omega_m)t$$

Thus the AM signal has three components, namely, unmodulated carrier, LSB and USB. The message to be transmitted is present only in LSB and USB. Further, if we consider the power relation given by

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

Therefore, the power required for the carrier component is given by

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

Let the modulation index be unity, i.e., $m = 1$.

$$P_c = \frac{2}{3}P_{AM}$$

Thus two-third of total AM power is utilized for the transmission of carrier component, which does not bear any message. A significant saving in power requirement can be achieved by suppressing the carrier before transmission. This thought process led to the first variant of basic AM termed as double sideband suppressed carrier (DSBSC) technique.

2nd Part:

The two additional terms produced are the two sidebands outlined. The frequency of the lower sideband (LSB) is $f_c - f_m$ and the frequency of the upper sideband (USB) is $f_c + f_m$. The very important conclusion to be made at this stage is that the bandwidth required for amplitude modulation is twice the frequency of the modulating signal.

Thus, the equation of DSBSC wave contains two terms, namely, LSB and USB. The bandwidth required for DSBSC is twice the frequency of the modulating signal, as in the case of AM. That is,

$$B_{DSBSC} = (f_c + f_m) - (f_c - f_m) = 2f_m$$

5. a) What is pilot carrier in AM transmission? **[WBUT 2015]**
b) What do you mean by white noise?
c) What is Carson's Rule?

Answer:

a) Pilot carrier, an attenuated carrier which is reinserted into the SSB signal, to facilitate receiver tuning and demodulation. This system is also referred to as pilot carrier system.

b) White noise is not the noise source. It is the classification of noise. The noise which has Gaussian distribution and have flat spectral density over a wide range of frequencies is called white noise.

c) *Refer to Question No. 3 of Short Answer Type Questions.*

4. Justify how FM can be obtained from PM and vice versa. **[WBUT 2015]**

Answer:

In FM, we vary frequency of the carrier, according to e_m (t) so that instantaneous frequency is given by

$$\underset{(FM)}{\omega_i} = \frac{d\theta}{dt} = \omega_C + k_f\, e_m\,(t) \qquad \text{... (1)}$$

And the phase of the FM signal is therefore given as

$$\theta(t) = \int \omega_i dt \quad = \int [\omega_c + k_f e_m(t)].dt$$

$$\underset{(FM)}{\theta\,(t)} = \omega_C t + k_f \int e_m(t).dt \qquad \text{... (2)}$$

In PM, however we vary phase of the carrier according to the instantaneous value of $e_m(t)$.

i.e.

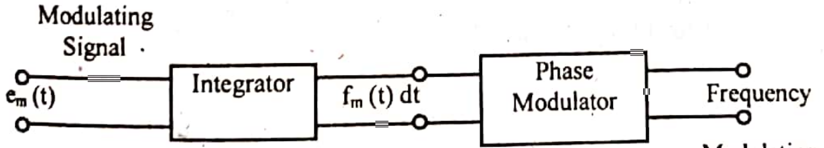
$$\underset{(PM)}{\theta\,(t)} = \omega_c t + \theta_0 + k_p.e_m(t) \qquad \text{... (3)}$$

Hence, the instantaneous frequency for PM becomes

$$\underset{(PM)}{\omega_i} = \frac{d\theta}{dt} = \omega_C + k_P.\frac{m(t)}{dt} \qquad \text{... (4)}$$

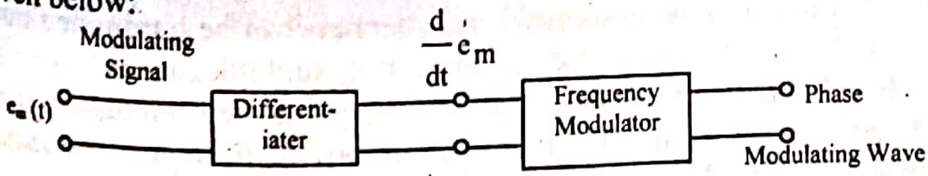
FM from PM

From equations (2) and (3) it can be seen that if we integrate the modulating signal $e_m(t)$ and then allow to phase – modulate it, then we get FM. This is shown in the block diagram given below:



PM from FM

Similarly from (1) and (4), it can be seen that if we define the modulating signal $e_m(t)$ and then allow to frequency modulating it, then we get PM. This is shown in the block diagram given below:



7. What are the advantages of digital communication over analog communication? **[WBUT 2015]**

Answer:

1. Because of the advances in digital IC technologies and high speed computers, digital communication systems are simpler and cheaper compared to analog systems.
2. Using data encryption, only permitted receivers can be allowed to detect the transmitted data. This is very useful in military applications.
3. Wide dynamic range is possible since the data is converted to the digital form.
4. Using multiplexing, the speech, video and other data can be merged and transmitted over common channel.
5. Since the transmission is digital and channel encoding is used, the noise does not accumulate from repeater to repeater in long distance communication.
6. Since the transmitted signal is digital, a large amount of noise interference can be tolerated.
7. Since channel coding is used, the errors can be detected and corrected in the receivers.

8. What do you mean by FDM? **[WBUT 2015]**

Answer:

Frequency Division Multiplexing (FDM)

This is a technique of separating the signals from different channels in frequency. The diagram below shows the principle of FDM.

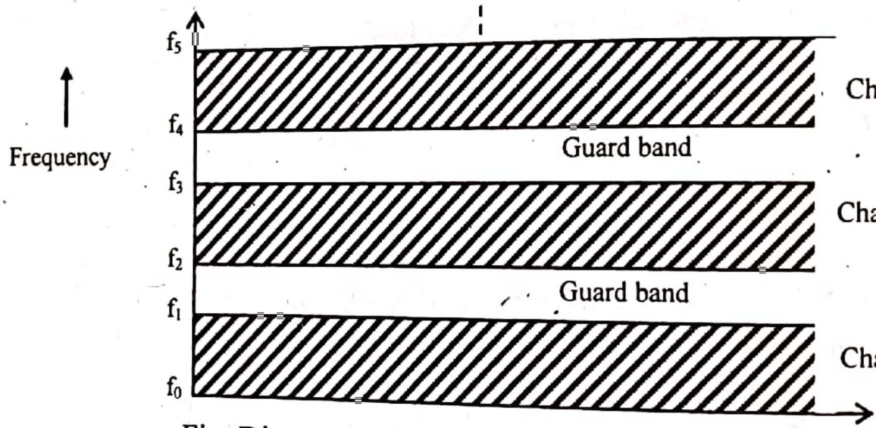


Fig: Diagrammatic representation of FDM

Channel 1 is assigned a frequency band from f_0 to f_1. Similarly channel 2 is assigned a frequency band from f_2 to f_3 and channel 3 from f_4 to f_5 and so on. The spectral regions between assignments such as frequency bands from f_1 to f_2 and from f_3 to f_4 are called guard bands which act as buffer zones to reduce interference between adjacent frequency channels.

Thus with FDM, many relatively narrowband channels can be transmitted over a single wideband transmission system. FDM is an analog multiplexing scheme. The inputs signals to an FDM system are analog and they remain analog throughout transmission. AM and FM radio broadcasting, TV broadcasting and high volume telecommunication system are examples of FDM.

9. What is thermal noise? [WBUT 2015]

Answer:

Thermal Noise: Thermal noise occurs in all transmission media and all communication equipment, including passive devices. It arises from random electron motion and is characterized by a uniform distribution of energy over the frequency spectrum with a Gaussian distribution of levels.

Every equipment element and the transmission medium proper contribute thermal noise to a communication system if the temperature of that element or medium is above absolute zero. Thermal noise is the factor that sets the lower limit of sensitivity of a receiving system and is often expressed as a temperature, usually given in units referred to absolute zero. These units are kelvins.

Thermal noise is a general expression referring to noise based on thermal agitations. The term "white noise" refers to the average uniform spectral distribution of noise energy with respect to frequency. Thermal noise is directly proportional to bandwidth and temperature. The amount of thermal noise to be found in 1 Hz of bandwidth in an actual device is where k is Boltzmann's constant, equal to 1.3803×10^{-23} J/K, and T is the absolute temperature (K) of the circuit (device). At room temperature, $T = 17°C$.

$$P_n = kT(W/H_z) \qquad \cdots(1)$$

10. a) Briefly explain the elements of communication system. [WBUT 2016, 2018]
b) Compare DSBFC, DSBSC and SSBSC. [WBUT 2016]

Answer:

a) Basic Communicating System

The word communication means exchange or sharing of ideas or thoughts with one another. The physical medium between transmitter and receiver is called communication channel. Channel is divided into two categories namely wired channel and wireless channel.

Signal is a physical variable which is a function of time. If the signal is a continuous time dependent variable, signal is *Analog* and if the signal is discrete time dependent then it is called Digital or Discrete signal.

The communication linked with analog signal is called Analog Communication and the signal, linked with Digital signal is called Digital Communication. The essential components in a basic communication system are shown in the following block diagram.

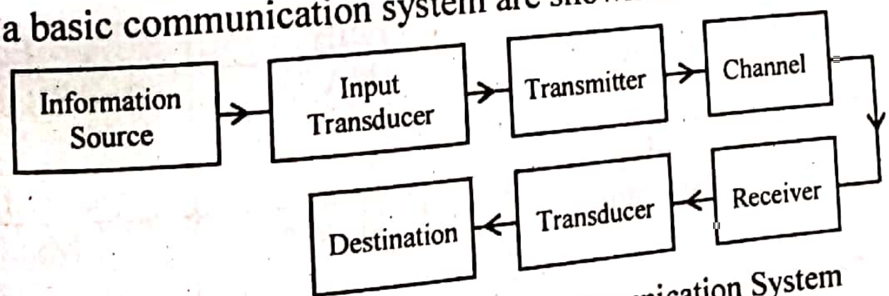


Fig: Block Diagram of communication System

Source

The message or information is generated in the information source. The information may be voice or related to it.

Input Transducer
The input transducer converts the non-electrical signal to electrical, which is suitable for transmission. The electrical signal is called baseband or message signal.

Transmitter
The transmitter modifies the baseband or message signal for efficient transmission. The function of transmitter is to modulate the signal for long distance communication. The transmitter consists of a Low pass filter, sampler, coder, and modulator.

Channel
The channel is that medium, through which the baseband signal propagates from transmitter to receiver, i.e., the channel provides a physical connection between the transmitter and the receiver. The channels may be wire, co-axial cable wave guide, optical fiber or radio link etc.

Receiver
The receiver recovers the original message signal through demodulation. Receiver consists of demodulator decoder, filter and de-emphasiser.

Output Transducer
Output transducer converts electrical signal to the original message.

Noise
Noises are unwanted signal which interfere the original signal. A noise is always random in nature. The art of communication engineering is to suppress the noise through advanced digital signal processing technique. The function of information source is to generate signal. Input transducer converts the physical signal into electrical signal. The function of transmitter is to transmit the electrical signal into space through which the signal is being transmitted. The function of receiver is to receive the electrical signal. The transducer, connected at the receiving end converts electrical signal again into physical signal which is being collected at the receiver. In electronics and communication systems, noise is a random fluctuation or variation of an electromagnetic analog signal such as a voltage or a current.

b)

Parameter of Comparison	DSBFC	DSBSC	SSBSC
Carrier suppression	NA	Fully	Fully
Sideband suppression	NA	NA	One SB completely
Bandwidth	$2f_m$	$2f_m$	f_m
Transmission efficiency	Minimum	Moderate	Maximum
Number of modulating inputs	1	1	1
Applications	Radio broadcasting	Radio broadcasting	Point to point mobile communication

11. Derive the mathematical expression for Single Tone Frequency Modulation. **[WBUT 2016]**

Answer:

Single-Tone FM

We start with the single-tone modulating signal:

$$m(t) = A_m \cos(\omega_m t)$$

Frequency modulation is the process where the instantaneous frequency of the modulated signal varies about the carrier frequency in proportion to $m(t)$.

That is:

$$\omega_i(t) = \omega_c + k_f m(t)$$

For single-tone FM, we have:

$$\omega_i(t) = \omega_c + k_f A_m \cos(\omega_m t)$$

Substituting $\Delta\omega$ for $k_f A_m$ in the above expression, we have:

$$\omega_i(t) = \omega_c + \Delta\omega cis(\omega_m t)$$

Thus, the instantaneous frequency of a single-tone FM wave varies about ω_c as shown in Fig. 1.

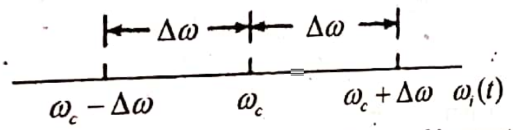


Fig: 1 Figure to illustrate the variation of instantaneous frequency in single-tone FM

Since the limits of $\cos(\omega_m t)$ are ± 1, we note that $\Delta\omega = k_f A_m$ is the maximum variation in instantaneous frequency on either side of the carrier frequency ω_c, for single-tone FM. We refer to $\Delta\omega$ as the maximum frequency deviation.

$$\theta(t) = \int_0^t \omega_i(\tau)\,d\tau = \int_0^t \left[\omega_c + \Delta\omega \cos(\omega_m \tau)\right] d\tau = \omega_c t + \left(\frac{\Delta\omega}{\omega_m}\right)\sin(\omega_m t)$$

The ratio $\left(\frac{\Delta\omega}{\omega_m}\right)$ is an important parameter in FM and is called the modulation index of the single-tone FM signal. It is denoted by the symbol β. Thus:

$$\beta = \left(\frac{\Delta\omega}{\omega_m}\right)$$

where $\Delta\omega = k_f A_m$. The frequency modulation index is the ratio of the maximum frequency deviation to the modulating signal frequency.

Hence, the expression for a single-tone FM wave is given by:

$$s_{FM}(t) = A_c \cos\left[\omega_c t + \beta \sin(\omega_m t)\right]$$

It is clear from the above analysis that, both the frequency deviation as well as the modulation index of a single-tone FM wave increase with increasing modulating signal frequency, the modulation index decreases with increasing modulating signal frequency.

12. a) Prove that the transmission efficiency for single tone sinusoidal modulation is 33%. **[WBUT 2016]**

OR,

Determine transmission efficiency for single tone sinusoidal modulation. **[WBUT 2018]**

b) What do you mean by modulation index? **[WBUT 2016, 2018]**

Answer:

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

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

$$P_t = P_c + \frac{P_c}{2}$$

$$P = P_{SB1} + P_{SB2} = \frac{P_c}{4} + \frac{P_c}{4}$$

each sideband carry same information.
Power efficiency,

$$\eta = \frac{\text{useful power}}{\text{total power}} = \frac{P_{sB}}{P_c + P_{sB}} = \frac{\frac{m^2 P_c}{2}}{P_c + \frac{m^2 P_c}{2}} = \frac{m^2}{2+m^2}$$

As m = 1, $\eta = \frac{m^2}{2+m^2} = \frac{1}{2+1} = \frac{1}{3}$

So the efficiency in % is given by $\eta = \frac{1}{3} * 100 = 33\%$

So only 1/3 power is used by sidebands to carry information. It is proved that in case of AM with modulation index equal to 1, only 33.33% of the transmitted power is used to carry information.

b) Modulation Index

Modulation index indicates how much modulated variable of the carrier signal varies around its unmodulated level.

In AM modulation, this quantity also called modulation depth indicates by how much the modulated variable varies around its original level.

Mathematical calculation

Mathematically modulation index, m_a is defined by, $m_a = \frac{KV_m}{V_c}$

where, K = proportionality constant;
V_m = amplitude of modulating signal;
V_c = amplitude of carrier signal.
We know that,
A = amplitude of modulated signal
$= V_c(1 + m_a \sin \omega_m t)$

So, $A_{max} = V_c(1 + m_a)$ and $A_{min} = V_c(1 - m_a)$

Finally, modulation index, $m_a = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$

13. What do you mean by multiplexing? What is the difference between TDM and FDM? **[WBUT 2016]**

Answer:

1st Part:

Multiplexing is defined as the process of transmitting several message signals simultaneously over a single channel.

2nd Part:

Time Division Multiplexing (TDM) is a multiplexing technique in which a number of signals are made to pass through a common channel at different time slots.

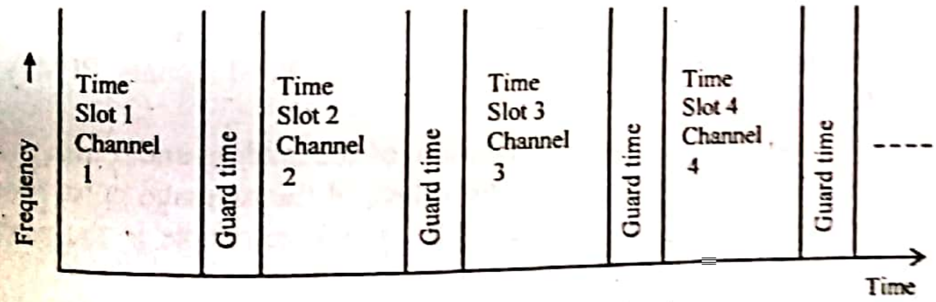


Fig: Time Division Multiplexing

Thus each channel is assigned a short duration of time called a time slot. Different channels occupy different time slots. All channels, however, occupy the entire frequency band. The unused time regions between two adjacent time slots are called guard times. These guard times allow for some time uncertainty between two adjacent channels and thus reduce interference between channels.

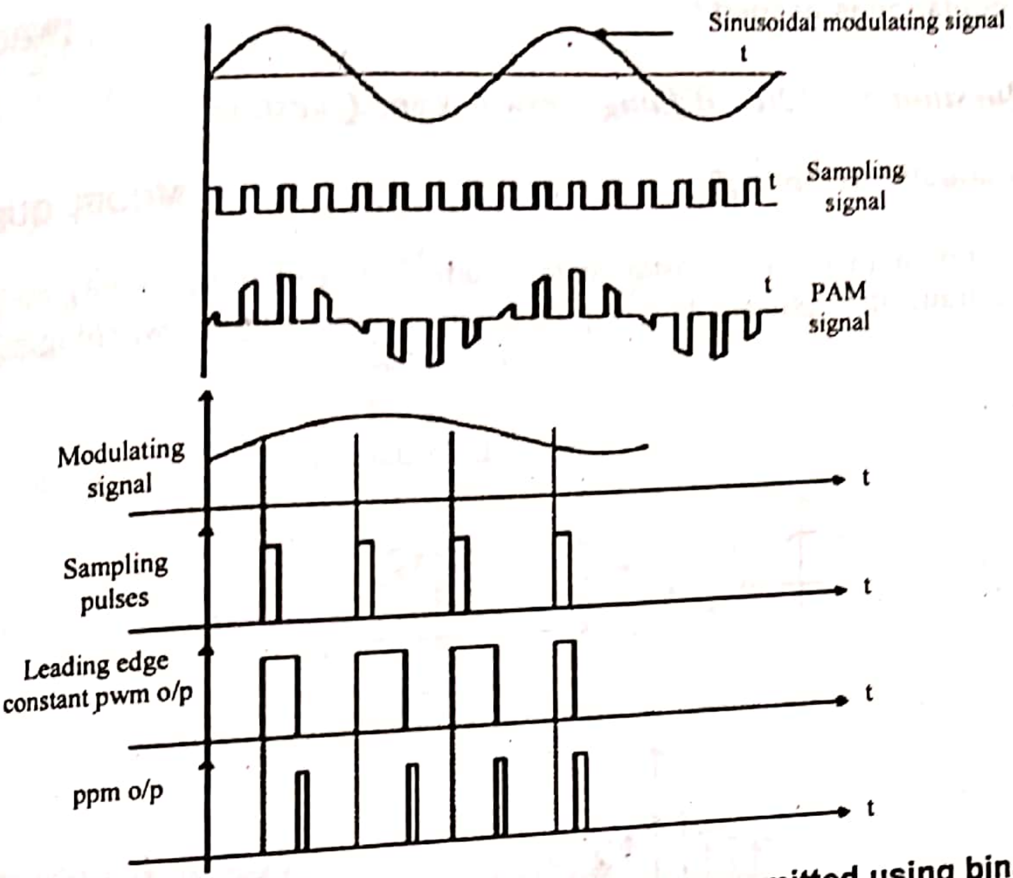
FDM: ***Refer to Question No. 8 of Short Answer Type Questions.***

14. Discuss the merits and demerits of TDM and FDM system. **[WBUT 2017]**

Answer:
The circuitry needed in TDM system is simpler than that in FDM system. In FDM, each channel occupies a different frequency band and therefore different bandpass filters are required for each channel. In a TDM system, all the channels require identical circuits. The circuits consist of switches, gates and a low pass filter. The same circuitry is used by all the channels.
The TDM system is relatively immune to interference within the channels as compared to the FDM system. The non-linearities in the various amplifiers of an FDM system produce harmonic distortion and thus they introduce interference within the channels. This is called interchannel cross-talk which is prevalent in an FDM system. In a TDM system, the signals from different channels are allotted different time slots and these signals are not applied to the system simultaneously. Thus the non-linearity requirements of a TDM system are the same as those of a single channel.
However, TDM has a disadvantage. The transmission bandwidth requirement of a TDM system is much higher than an FDM system. The bandwidth requirement of an SSB / FDM system having 24 voice channels, each channel occupying 4 KHz bandwidth is $24 \times 4 = 96$ KHz. For PCM / TDM system using T_1 carrier the transmission bandwidth would be at least 772 KHz.
TDM offers greater system flexibility and more immunity to noise as compared to FDM system. TDM is easier to implement with digital modulation whereas FDM is more convenient with analog modulation.

15. Explain how a PPM signal may be converted into a PAM signal. **[WBUT 2017]**

Answer:
Pulse Position modulation is defined as a process of varying the position of the signal pulse, by taking the reference signal, in accordance to the modulating signal variations. The amplitude and width of the pulse remains constant for PPM signals. PPM is a kind of modification of the PWM signal.
In the absence of the modulating signal the position of the leading and trailing edge of the pulse is equal to original position. The positive values of the message pulse results in the proportionate right shift and negative values of the message results in the proportionate left shift.
Pulse amplitude modulation is defined as a process of varying the amplitude of the signal pulse in accordance to the modulating signal variations. PAM is the basic form of analog pulse modulation in which the width and position characteristics of the pulse are kept constant while varying the amplitude.
There are two sampling techniques used for sampling the modulating signal in PAM, which are Flat top sampling and Natural sampling.



16. A television signal having of BW of 4.2 MHz is transmitted using binary system. Number of quantization level used is 512. Determine

i) Code word length
ii) Transmission bandwidth
iii) Output signal to quantization ratio
iv) Final bit rate. **[WBUT 2018]**

Answer:

$f_m = 4.2\,\text{MHz}$

$Q = 512$

1. Code word length (N)

$$Q = 2^N$$

$$512 = 2^N$$

$$N = 9 \text{ bits/word}$$

2. Transmission Bandwidth, $B_T = Nf_n = N(2f_m) = 9\times(4.2)\,\text{MHz} = 37.8\,\text{MHz}$

3. Final bit rate, $r = Nf_1 = 9\times 2f_m = 18\times 4.2\,\text{MHz} = 75.6\,\text{Mb/sec.}$

4. Signal to quantization noise ratio.

Since the TV signal is not a sinusoidal signal, let us use the general expression of signal to quantization ratio.

$$= 4.8 + 6N\text{ dB} = 4.8 + (6\times 9) = 58.8\,\text{dB}$$

This is the maximum signal to noise ratio that we are expected to get from this system.

17. Why modulation is needed? [WBUT 2018]

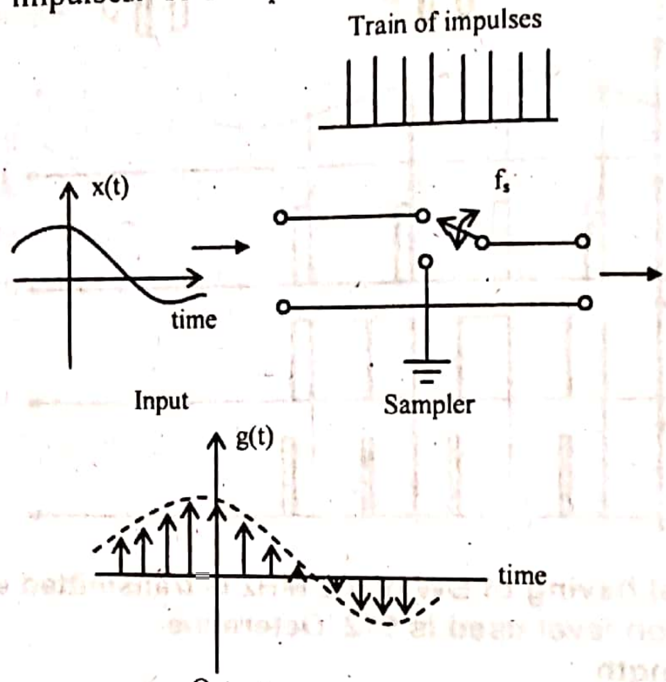
Answer:

Refer to Question No. 12(e) of Long Answer Type Questions.

18. What is impulse sampling? [MODEL QUESTION]

Answer:

In impulse sampling (also called instantaneous sampling or ideal sampling), the sampling function is a train of impulses. The impulses are produced by a switching sampler as shown below.



In impulse sampling $\tau \to 0$ and because of this, the power content in the instantaneously sampled pulse is negligible. Thus impulse sampling is an ideal case and is not suitable for transmission purpose.

19. An FM signal has a frequency deviation of 5 Hz and a modulating frequency of 1 kHz. The SNR at the input to the receiving detector is 20 dB. Calculate the approximate SNR at the detector output. [MODEL QUESTION]

Answer:

We know that, frequency deviation

$$(\Delta f) = \beta f_m = (\text{Depth of modulation} \times \text{frequency of modulation})$$

According to the problem,

$$\Delta f = 5\,\text{kHz}$$

$$f_m = 1\,\text{kHz}$$

So, $$\beta = \frac{\Delta f}{f_m} = \frac{5}{1} = 5$$

Now, given,

$$SNR_R = 20\,\text{dB}$$

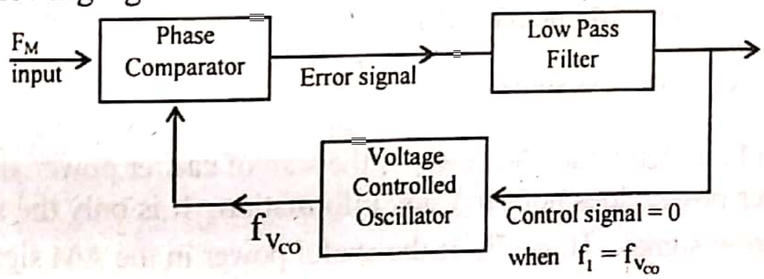
So, $$\frac{SNR_O}{SNR_R} = \frac{3}{2}\beta^2 = \frac{3}{2}(5)^2$$

or, $$SNR_O = \frac{3}{2}(5)^2 \times 20 = 750 \text{ dB}$$

20. Explain how PLL can be used as FM demodulation. **[MODEL QUESTION]**

Answer:

Demodulation of FM signals is being done by using phase locked loop. PLL is basically an electronic feedback control system. Block diagram of PLL is shown in the fig. Basically a PLL consists of phase comparator, voltage controlled oscillator and low pass filter. The input of the PLL goes to the phase comparator. Other input to the phase comparator comes from VCO with PLL. If the two input frequencies to the phase comparator are identical, phase comparator output is zero. If the two input frequencies are not identical the comparator's output when passed through low pass filter is a level which is applied to the VCO input. This level applied to VCO input charges VCO frequency and makes it exactly match with input frequency. If VCO frequency is equal to input frequency, PLL is said to achieve LOCK and control voltage remains zero as long as PLL input frequency remains constant. If PLL input is IM signal, the low pass filter output will be the modulating (intelligence signal). PLL has three possible states of operation e.g., Free running, capture and Locking or tracking. Block diagram of PLL is shown in the following figure:



Long Answer Type Questions

1. a) What do you mean by Amplitude Modulation? Explain with necessary waveform.
b) What is Transmission efficiency in AM? Derive the expression of it.
c) With neat diagram and mathematical derivation explain the principle of Operation of Square Law Diode Modulation technique.
d) A given AM broadcast station transmits a total power of 5 kW when the carrier is modulated by a sinusoidal signal with a modulation index of 0.7071. Calculate –
i) the carrier power
ii) the transmission efficiency. **[WBUT 2013]**

Answer:

a) The amplitude modulation is the amplitude of a carrier signal that varies with the modulating voltage whose frequency is much lower than that of the carrier.

Let the carrier voltage and modulating voltage be represented by

$$v_c = V_c \sin \omega_c t, \text{ and } v_m + V_m \sin \omega_m t$$

Then the amplitude modulated voltage will be expressed as

$$V_c + V_m \sin \omega_m t = V_c + mV_c \sin \omega_m t = V_c(1 + m \sin \omega_m t)$$

V_m is considered equal to mV_c where 'm' is constant and the 'm' is termed modulation index.

The instantaneous voltage of the resulting amplitude modulated wave is

$$V_c(1 + m \sin \omega_m t) \sin \omega_c t$$

Figure describes the below.

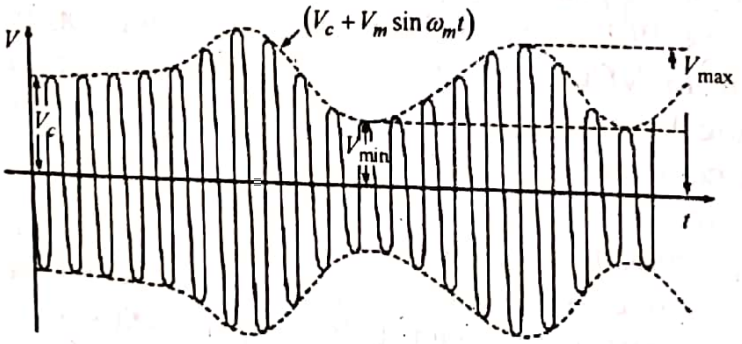


Fig: Amplitude modulation

b) Transmission efficiency of AM signal is defined as the percentage of total AM power contained in the sidebands. Mathematically,

Transmission efficiency of AM signal, $\eta_{P_{AM}} = \dfrac{P_{sb}}{P_{AM}}$

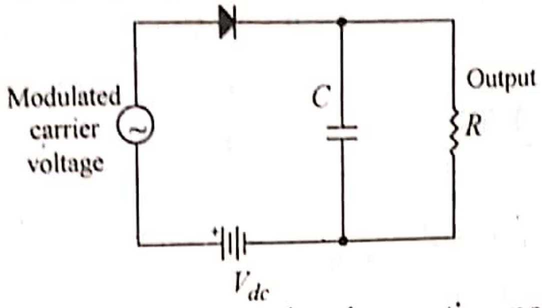
It may be recalled here that total AM power is the sum of carrier power and the sidebands power. The carrier power does not carry any information. It is only the sidebands which carry the information signal. Thus, P_{sb} is the useful power in the AM signal.

We know that $P_{sb} = \dfrac{m_a^2}{2} P_c$ and $P_{AM} = P_c\left(1 + \dfrac{m_a^2}{2}\right)$

Therefore, $$\eta_{P_{AM}} = \frac{P_{sb}}{P_{AM}} = \frac{\dfrac{m_a^2}{2} P_c}{P_c\left(1 + \dfrac{m_a^2}{2}\right)} = \frac{\dfrac{m_a^2}{2}}{\left(1 + \dfrac{m_a^2}{2}\right)}$$

Hence, $\boxed{\eta_{P_{am}} = \dfrac{m_a^2}{2 + m_a^2}}$

c) Figure shows the basic circuit of square law diode detector.



The diode is biased positively so that zero signal operating point will work in the non-linear region of dynamic current-voltage characteristic. Figure 2 shows the following:

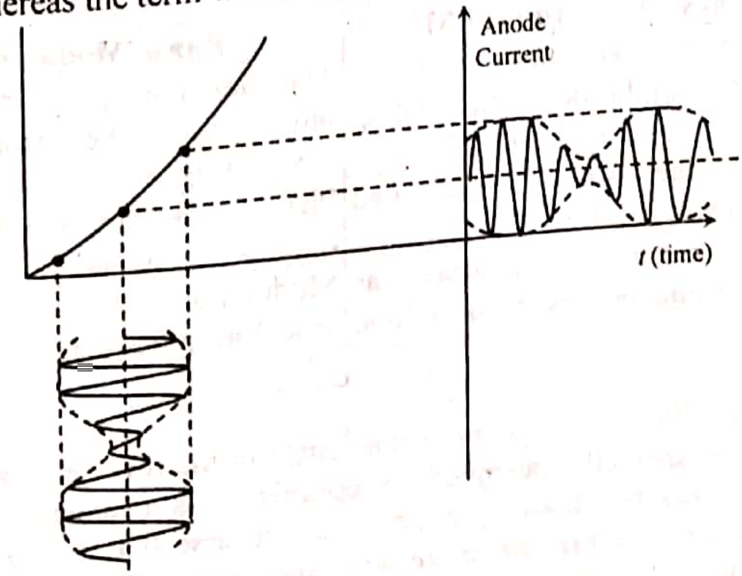
(a) superposition of modulated carrier voltage over the dynamic characteristic of diode, and

(b) output current of diode-detector with resistance load. The dynamic current and voltage characteristic of the diode follows approximately the square law relation.

$$i_{anode} = K_1 v_{\text{anode voltage}} + K_2 v^2_{\text{anode voltage}}$$

where $v_{\text{anode voltage}} = V_c[1 + m\cos\omega_m t]\cos\omega_c t$

'm' is the modulation index of amplitude modulated signal. V_c is the amplitude of the carrier voltage. From the expression of the i_{anode}, it is observed that the output current will consist of the frequencies $2\omega_c$, $2(\omega_c+\omega_m)$, $2(\omega_c-\omega_m)$, ω_m and $2\omega_m$. The radio frequency components will be bypassed through the shunt capacitor 'C' as shown in Figure 1. The components with frequency ω_m and $2\omega_m$ will appear across the load resistance R as shown in Figure 2. the components with frequencies ω_m and $2\omega_m$ will be developed across the load resistance R. The components with frequency 'ω_m' will be the desired output whereas the term with frequency '$2\omega_m$' will develop distortion.



d) $P_t = P_C\left[1+\dfrac{m_a^2}{2}\right]$ $\qquad P_t = 5\text{kW}$

$$\Rightarrow \quad 5\times10^3 = P_C\left[2+\frac{0.707^2}{2}\right] \qquad m_a = 0.707$$

$$\Rightarrow \quad P_C = \frac{5\times10^3}{1+\dfrac{0.707^2}{2}} = \frac{5\times10^3}{1.2499} = 4.00\,\text{kW}$$

Transmission efficiency, $\dfrac{P_C}{P_t} = \dfrac{4}{5} = 80\%$.

2. a) Compare and analyze between Phase and Frequency Modulation.
b) Explain with neat circuit diagram the operation of generation of FM using Varactor diode.
c) Write down the differences between Direct and Indirect method of FM generation.
d) A single tone FM is represented by the voltage equation:

$$v(t) = 12\cos(6*10^8 t + 5\sin 1250t)$$

Determine:

i) Carrier frequency
ii) Modulating frequency
iii) The modulation index
iv) Maximum deviation
v) What power will this FM wave dissipate in 10Ω resistor? **[WBUT 2013]**

Answer:

a)

No.	Frequency Modulation (FM)	Phase Modulation (PM)
1.	The maximum frequency deviation depends upon amplitude of modulating voltage and modulating frequency.	The maximum phase deviation depends only upon the amplitude of the modulating voltage.
2.	Frequency of the carrier is modulated by the modulating signal.	Phase of the carrier is modulated by the modulating signal.
3.	Modulation Index is increased as modulation frequency is reduced and vice versa.	Modulation index remains same if the modulating frequency is changed.

b) All the diodes exhibit small junction capacitance in the reverse biased condition. The varactor diodes are specially designed to optimize this characteristic. The junction capacitance of the varactor diode changes as the reverse bias across it is varied. The variations in capacitance of this diode are wide and linear. The varactor diodes provide the junction capacitance in the range of 1 to 200 pF. Fig. shows how varactor diode can be used to generate FM. L_1 and C_1 from the tank circuit of the carrier oscillator. The capacitance of the varactor diode depends upon the fixed basis set by R_1 and R_2 is made variable so that the center carrier frequency can be adjusted over a narrow range. The Radio Frequency Choke (RFC) has high reactance at the carrier frequency to prevent the carrier signal from getting into the modulating signal circuits. At positive going modulating signal adds to the reverse basis applied to the varactor diode D, which decreases its capacitance and increases the carrier frequency. A negative going modulating signal subtracts from the basis, increasing the capacitance, which decreases the carrier frequency.

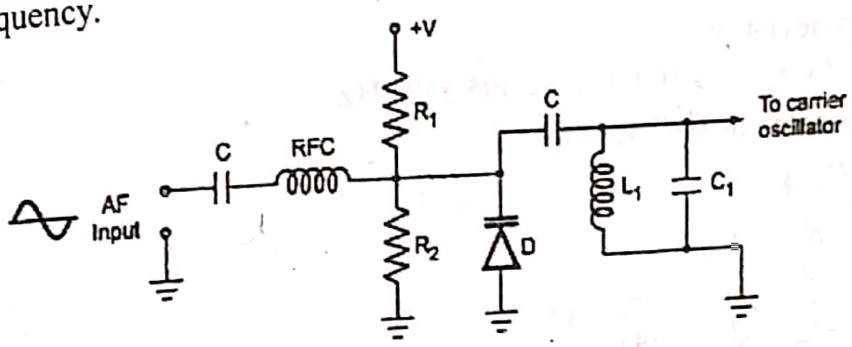


Fig. varactor diode for FM generation

The frequency of the LC oscillator changes due to temperature effects. Hence crystals are used in FM generators to provide frequency stability.

c)

Direct FM	Indirect FM
In this type of angle modulation, the frequency of the carrier is varied directly by the modulating signal. This means, an instantaneous frequency deviation is directly proportional to amplitude of the modulating signal.	In this type of angle modulation FM is obtained by phase modulation of the carrier. Instantaneous phase of the carrier is directly proportional to the amplitude of the modulating signal.

d) Single tone FM

$$v(t) = 12\cos\left(6\times10^8 t + 5\sin 1250t\right)$$

The expression for a single tone FM wave is given by

$$v_{FM}(t) = A_C \cos\left[\omega_C t + \beta \sin(\omega_m t)\right]$$

Comparing the given equation with standard form

$$A_C = 12\text{ V}$$

$$\omega_C = 6\times10^8$$

$$\beta = \text{modulation index} = 5$$

$$\omega_m = 1250$$

(i) f_C = Carrier frequency

$$= \frac{\omega_C}{2\pi} = \frac{6\times10^8}{2\pi} = \frac{600\times10^6}{2\pi}\text{Hz} = 95.54\text{ MHz}$$

(ii) f_m = modulating frequency

$$= \frac{\omega_m}{2\pi} = \frac{1250}{2\pi}\text{Hz} = 199.04\text{ Hz.}$$

(iii) Modulation index $\beta = 5$

(iv) Maximum deviation

$$\Delta f = \beta\times f_m = 5\times199.04\text{ Hz} = 995.2229\text{ Hz}$$

(v) Power dissipated in 10 Ω resistor

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

$$V_{rms} = \frac{A}{\sqrt{2}} = \frac{12}{\sqrt{2}} = \frac{12}{1.41} = 8.51$$

$$P = \frac{(8.51)^2}{10} = 7.2430\text{ W}$$ **Ans.**

3. a) Draw and explain the Block diagram of PCM system. **[WBUT 2013]**

OR,

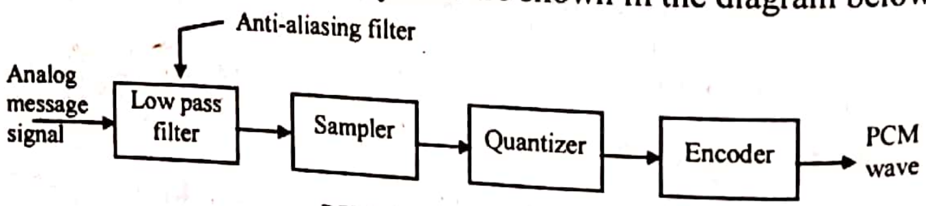
Draw and explain the block diagram of generation and detection of PCM. **[WBUT 2015]**

OR,

Explain the transmitter and receiver of Pulse Code Modulation. **[WBUT 2016]**

Answer:

Pulse code modulation is a digital pulse modulation technique. In a PCM system the message signal is subjected to a large number of operations. A PCM system basically consists of three main parts i.e. transmitter, transmission path and receiver. The essential operations done in the transmitter of a PCM system are sampling, quantization and encoding. Regeneration of the distorted PCM signal is done at intermediate points along the transmission path of the signal from the transmitter to the receiver. The regeneration is done using regenerative repeaters. The essential operations in the receiver are the regeneration of the PCM signal, decoding and reconstruction of the original analog signal. The basic elements of a PCM system are shown in the diagram below.



PCM Transmitter

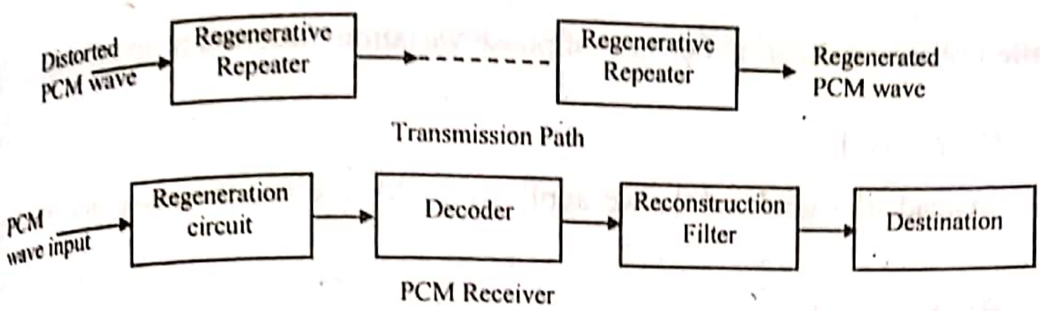


Fig: Block diagram of a PCM system

The low pass filter before sampling is used to avoid the effect of aliasing and it is called an anti-aliasing filter.

b) A television signal having BW of 10.2 MHz is transmitted using binary PCM system. Given that the number of quantization level is 512. Determine –
i) code word length
ii) transmission BW
iii) final bit Rate
iv) output signal to quantization noise ratio. **[WBUT 2013]**

Answer:

$f_m = 10.2\,\text{MHz}$

$Q = 512$

1. Code word length (N)

$$Q = 2^N$$

$$512 = 2^N$$

$$N = 9 \text{ bits/word} \quad \textbf{Ans.}$$

2. Transmission Bandwidth, $B_T = Nf_s = N(2f_m) = 9\times(10.2)\,\text{MHz} = 91.8\,\text{MHz}$ **Ans.**

3. Final bit rate, $r = Nf_1 = 9\times 2f_m = 18\times 10.2\,\text{MHz} = 183.6\,\text{Mb/sec.}$

4. Signal to quantization noise ratio.

Since the TV signal is not a sinusoidal signal, let us use the general expression of signal to quantization ratio.

$$= 4.8 + 6N \text{ dB}$$

$$= 4.8 + (6\times 9)$$

$$= 58.8 \text{ dB } \textbf{Ans.}$$

This is the maximum signal to noise ratio that we are expected to get from this system.

4. a) Explain the relation between Phase Modulation (PM) and Frequency Modulation (FM) with diagram.
b) Explain the Practical Armstrong Method for FM Generation. **[WBUT 2014, 2018]**

Answer:

a) Relation between phase modulation and frequency modulation

$$\left[f(t)\right]_{PM} = V\cos\left[\omega_c t + K_p m(t)\right]$$

It is called phase modulation because of phase variation direction proportional to message signal.

$$\theta = K_p m(t)$$

Now, if instead of signal $m(t)$, we apply to a PM system, the integration of message signal.

$$\theta = K\left[\int m(t)\,dt\right]$$

K is a new constant.
But if we study the equation is

$$\left[f(t)\right]_{PM} = V_c \cos\left[\omega_c t + K_f \int m(t)\,dt\right]$$

But converted into FM means "If a message signal is passed through a integrator and phase modulation system, the output of system will be frequency modulated signal."

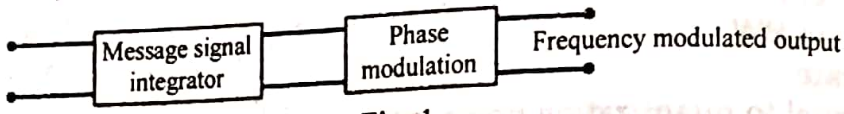


Fig: 1

Thus, this concept can be used to generate FM by PM. Because of FM is not generated directly so it is called indirect method of generation of FM.

$$\left[f(t)\right]_{FM} = V_c \cos\left[\omega_c t + K_f \int m(t)\,dt\right]$$

In this equation phase

$$\theta = K_f \int m(t)\,dt$$

Now, instead of signal we apply the derivative of signal $m(t)$ phase will be modified to

$$\theta = K_f \int \frac{d}{dt} m(t)\,dt = Km(t)$$

where K = new constant
So, equation modified to

$$\left[f(t)\right]_{FM} = V_c \cos\left[\omega_c t + Km(t)\right]$$

That is the equation of phase modulated output.
"If a signal is first differentiated and then fed to a FM system, the output will be phase modulated output."

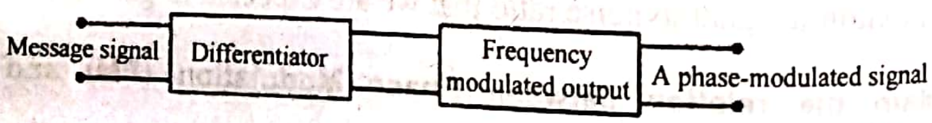
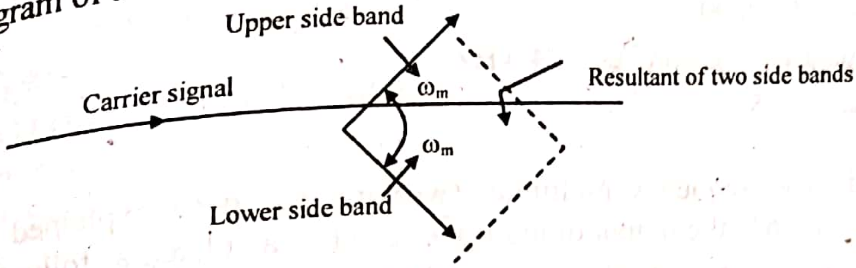


Fig: 2

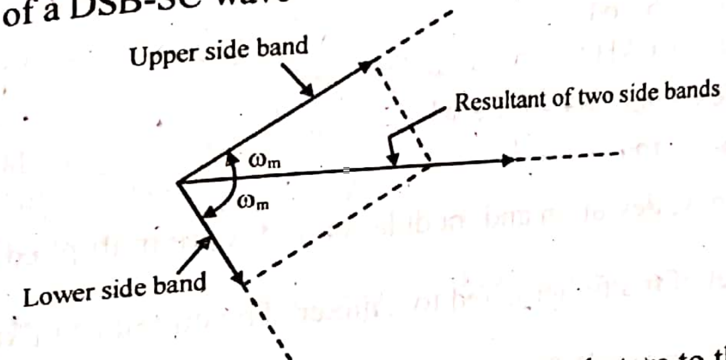
Practically NBPM may help to generate NBFM.

b) Practically to get FM (generally used FM is WBFM) frequency deviation must be increased. So we have to use several stages of multipliers and to change the carrier frequency to make it suitable for transmission mixer can also be used. The following phase diagram illustrates the real picture more clearly.

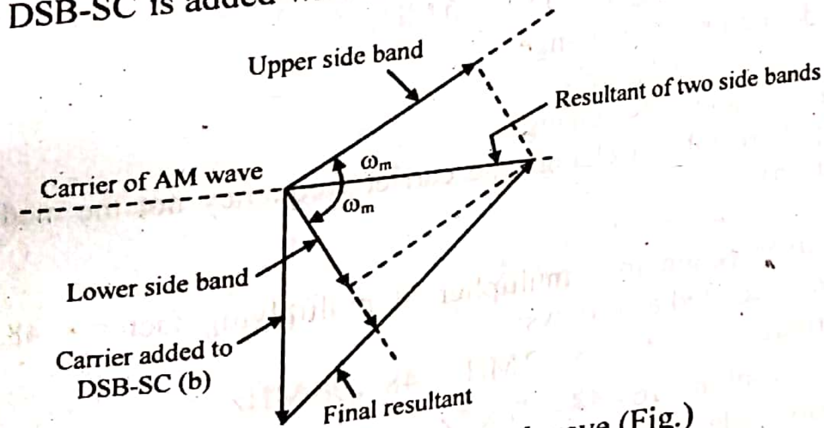
(i) Phase diagram of a AM wave



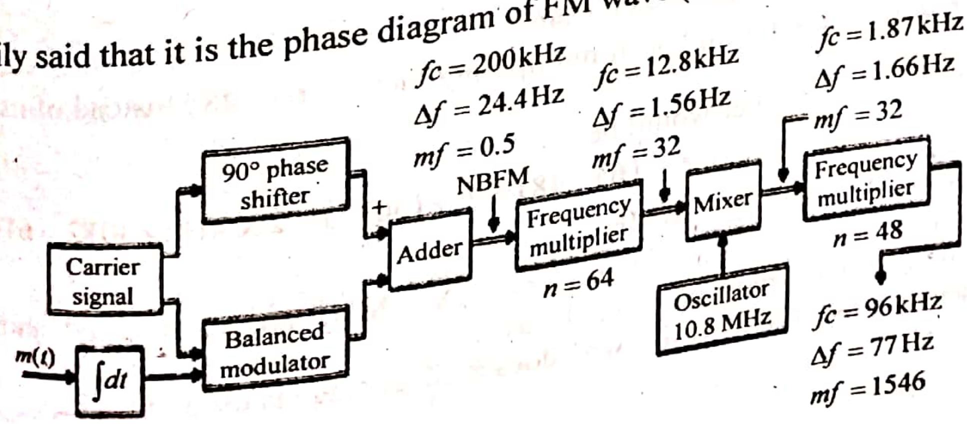
(ii) Phase diagram of a DSB-SC wave



(iii) Now, this DSB-SC is added with a carrier (that is quadrature to that of actual carrier of AM wave).



It can easily said that it is the phase diagram of FM wave (Fig.)



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(a) The output of adder is NBFM as discussed in previous section. For NBFM, the following values are prescribed by CCIR (Consultative Committee for International Radio).

Carrier frequency $f_c = 200$

Maximum frequency deviation $\Delta f = 24.4\,\text{Hz}$

Modulation index $= 0.5$

(b) This is fed to a frequency multiplier (working has been explained already) with Multiplying factor $= 64$, the output of multiplier can be calculated as follows:

Carrier frequency $= f_c \times n = 200\,\text{kHz} \times 64 = 12.8\,\text{MHz}$

Maximum deviation $= 25 \times 64$

$$\Delta f = 1.6\,\text{kHz}$$

So, modulation index $= m_f \times 64 = 0.5 \times 64$

$$m_f = 32$$

(The carrier frequency, deviation and modulation index are multiplied by same factor).

(c) Now, this output of multiplier is fed to a mixer. The output of mixer can be calculated as follows:

Carrier frequency = Difference of both frequencies

$= 12.8 - 10.8 = 2.0\,\text{MHz}$

Maximum deviation = No change

$= 1.6\,\text{kHz}$

Modulation index = No change

(The mixer is applied only to change the carrier frequency not the modulation index / maximum deviation).

(d) Output of mixer is fed to a multiplier of multiplying factor = 48, the output of multiplier can be calculated as follows:

Carrier frequency $= f_c \times 48 = 2\,\text{MHz} \times 48 = 96\,\text{MHz}$

Maximum deviation $= 16 \times 48 = 77\,\text{kHz}$

Modulation index $= 32 \times 48 = 1546$

If we apply only single multiplier of multiplying factor $= 64 \times 48$. Instead of using two of 64 and 48 respectively.

The output of this multiplier would be

$$= f_c \times (64 \times 48) = f_c \times (3072) = 200\,\text{kHz} \times 3072 = 614.4\,\text{MHz}$$

Now, our desired output $f_c = 96\,\text{MHz}$

So, a mixer of local oscillator frequency 518.4 MHz has to be used and the local oscillator of such high frequency must be avoided.

One another question arises again 'why does a mixer stage is used'?

For answer, just perform the calculation as given below.

The desired output $f_c = 2.0\,\text{MHz}$

From input $f_c = 12.8\,\text{MHz}$

can be achieved by a multiplier of multiplication factor $= 0.16\,\text{MHz}$

But due to this the

Output maximum deviation $= 0.256\,\text{MHz}$

Output modulation $= 5.12$

Then to achieve final output (standard), multiplier stage have to apply of having multiplication factor

$$= \frac{1546}{5.12} = 301.95 \text{ (By calculation of modulation index)}$$

Due to this frequency multiplier, the output

Carrier frequency $f_c = 301.95 \times 2.0\,\text{MHz} = 603.9\,\text{MHz}$

That is not desired standard output carrier frequency.

Thus a intelligent implementation of multiplier and mixer is required to achieve desired output.

5. a) How AM signal can be detected using envelop detector. [WBUT 2014, 2018]

Answer:

Envelope Detector

Consider the case of a narrow-band AM wave, that is, one in which the carrier frequency is large compared with the message bandwidth and for which the percentage modulation is less than 100 percent. Then the desired demodulation can be accomplished by using a simple, yet highly effective device known as envelope detector.

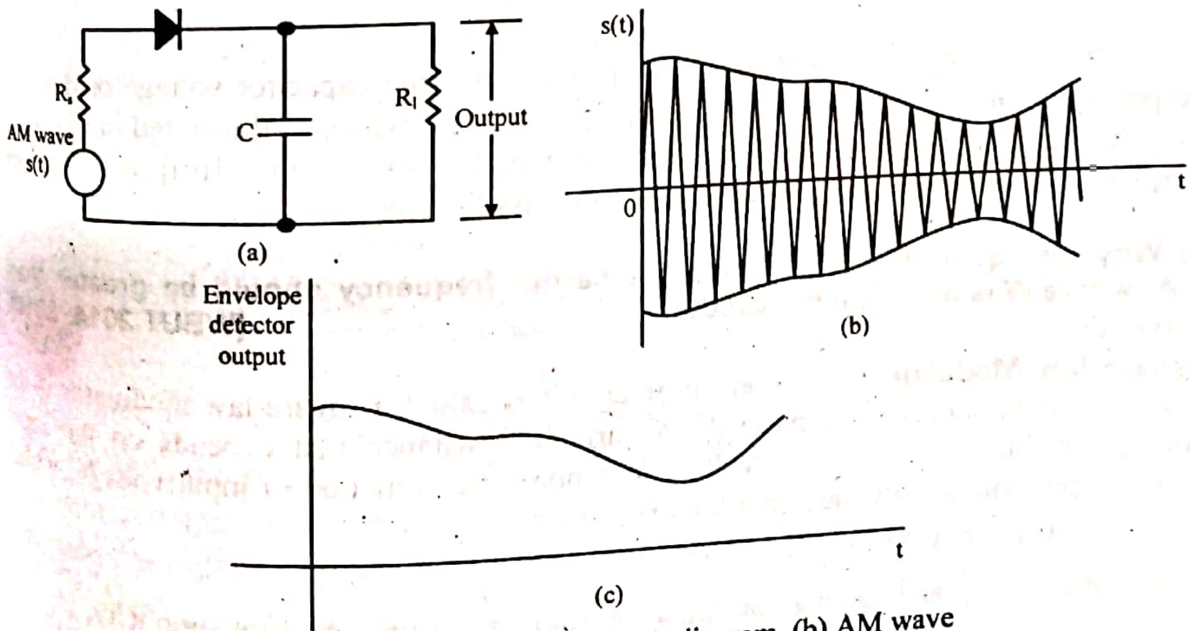

Fig: 1 Envelope detector. (a) Circuit diagram, (b) AM wave input, (c) Envelope detector output

Ideally, an envelope detector produces an output signal that follows the envelope of the input signal waveform exactly. Some version of this circuit is used in almost all commercial AM radio receivers.

An envelope detector of the series type is shown in Fig. 1(a), which consists of a diode and a resistor-capacitor filter. The operation of this envelope detector is as follows. On the positive half-cycle of the input signal, the diode is forward-biased and the capacitor C charges up rapidly to the peak value of the input signal. When the input signal falls below this value, the diode becomes reverse-biased and the capacitor C discharges slowly through the load resistor R_l. The discharging process continuous 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 is repeated. We assume that the diode is ideal, presenting zero impedance to current flow in the forward-biased region and infinite impedance in the reverse-biased region. We further assume that the AM wave applied to the envelope detector is supplied by a voltage source of internal impedance R_s. The charging time constant R_sC must be short compared with the carrier period $1/f_c$ that is,

$$R_s << \frac{1}{f_c}$$

so that the capacitor C charges rapidly and thereby follows the applied voltage up to the positive peak when the diode is conducting. On the other hand, the 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,

$$\frac{1}{f_c} << R_lC << \frac{1}{W}$$

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, as illustrated in Fig. 1(c). The detector output usually has a small ripple (not shown in Fig. 1(c)] at the carrier frequency; this ripple is easily removed by low-pass filtering.

b) Why for square-law modulation the carrier frequency should be greater than 3 W, where W is message bandwidth. **[WBUT 2014, 2018]**

Answer:

Square-law Modulator: The non-linear circuit is called a square-law modulator if the device is continuously on, but with a variable resistance that depends on the input voltage. In this case, the output voltage is a non-linear function of input voltage, which may be approximated by the quadratic expression

$$v_o(t) = av_i(t) + bv_i^2(t)$$

Substituting $v_i(t)$ and again assuming a sinusoidal message signal for simplicity,

$$v_o(t) = a\left[V_m\cos(2\pi f_m t) + V_c\cos(2\pi f_c t)\right] + b\left[V_m\cos(2\pi f_m t) + V_c\cos(2\pi f_c t)\right]^2$$

Expanding and simplifying the above equation, using the trigonometric identities, we obtain

$$v_o(t)=\frac{1}{2}b\left(V_c^2+V_m^2\right)+aV_m\cos\left(2\pi f_m t\right)+\frac{1}{2}bV_m^2\cos\left(4\pi f_m t\right)$$
$$+aV_c\cos\left(2\pi f_c t\right)+bV_cV_m\cos 2\pi\left(f_c-f_m\right)t$$
$$+bV_cV_m\cos 2\pi\left(f_c+f_m\right)t+\frac{1}{2}bV_c^2\cos\left(4\pi f_c t\right)$$

It is to be noticed that $v_o(t)$ contains the AM signal comprising the carrier f_c and sidebands $f_c \pm f_m$. However, there are also other components at DC, f_m, $2f_m$ and $2f_c$. These extra components can be excluded using a bandpass filter for the switching modulator. Because there is a component at twice the message frequency $2f_m$, which must be excluded by the filter while still passing the LSB, it means that if f_m is the maximum frequency component of the message signal, then the following condition must be satisfied:

$$f_c > 3f_m$$

6. a) Explain with proper circuit diagram how DSB-SC signal is obtained using ring modulator. **[WBUT 2015]**

Answer:

DSB-SC wave can be obtained using a product modulator. A method for generating a DSB-SC wave using a square-law modulator. Using two identical AM conventional AM wave generators, it would be possible to generate a DSB-SC signal. This is accomplished using the balanced modulator, whose block diagram is shown in Fig. below.

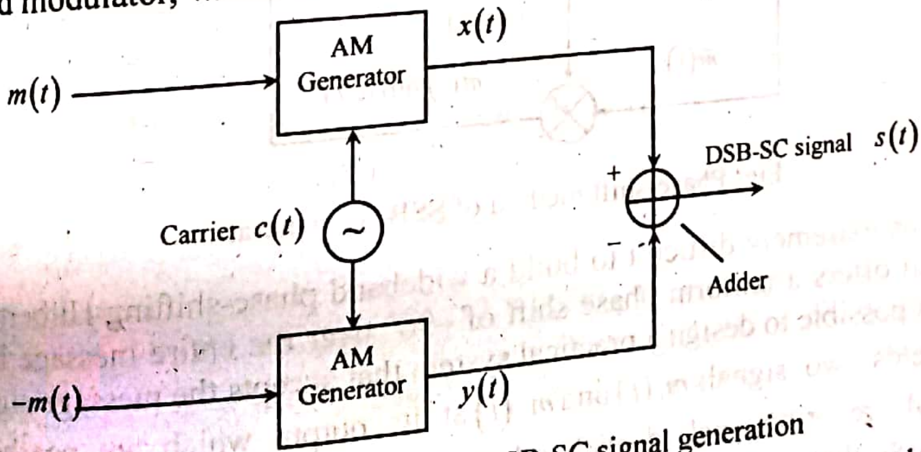

Fig: Balanced modulator for DSB-SC signal generation

The two AM generators are assumed to be perfectly identical to each other. The only difference in operation is that the modulating signal, $m(t)$ is inverted before applying to the lower branch AM generator. Accordingly, we have the following expressions:

$$x(t)=A_c\left[1+k_a m(t)\right]\cos\left(\omega_c t\right)$$
$$y(t)=A_c\left[1-k_a m(t)\right]\cos\left(\omega_c t\right)$$

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The output $s(t) = x(t) - y(t) = 2A_c k_a m(t)\cos(\omega_c t)$ which is clearly a DSB-SC signal.

b) Explain the principle of SSB-SC generation by phase shift method. [WBUT 2015]

Answer:

The phase-shift method of SSB-SC generation is a direct implementation of Eq. (1), as shown in Fig. 1. The Hilbert transformer yields $\bar{m}(t)$ as the Hilbert transform of $m(t)$. The multipliers in the top and bottom arms are balanced modulators. The top multiplier yields the first product (DSB-SC) term $m(t)\cos(\omega_c t)$. The carrier signal $\cos(\omega_c t)$ undergoes a –90° phase shift to yield $\sin(\omega_c t)$ which when multiplied by $\bar{m}(t)$ in the lower arm yields the second product term $\bar{m}(t)\sin(\omega_c t)$. This term, when added to the first product term in the output adder, yields the lower-sideband SSB-SC signal. Similarly, when the second product term is subtracted from the first product term at the output adder, we obtain the upper-sideband SSB-SC signal.

$$S_{SSB}(t) = \left[m(t)\cos(\omega_c t) \pm \bar{m}(t)\sin(\omega_c t)\right] \quad \cdots(1)$$

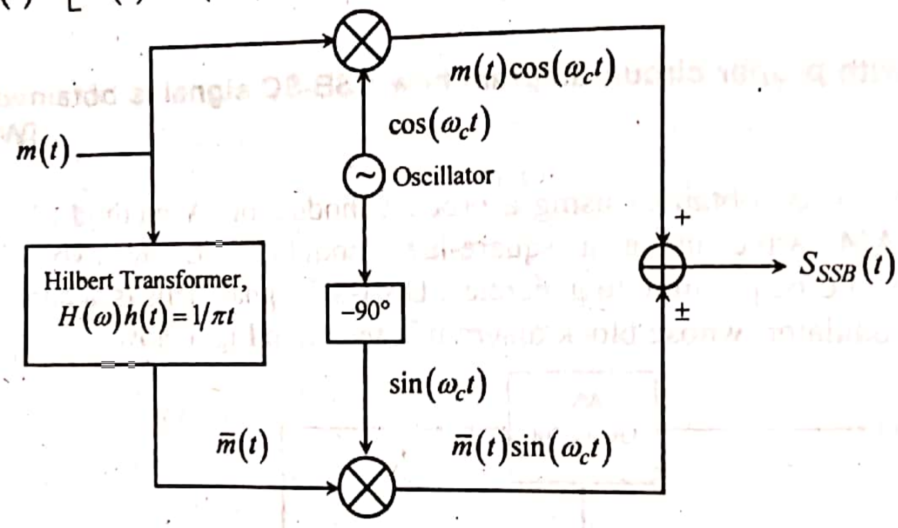


Fig: Phase-shift method of SSB-SC generation

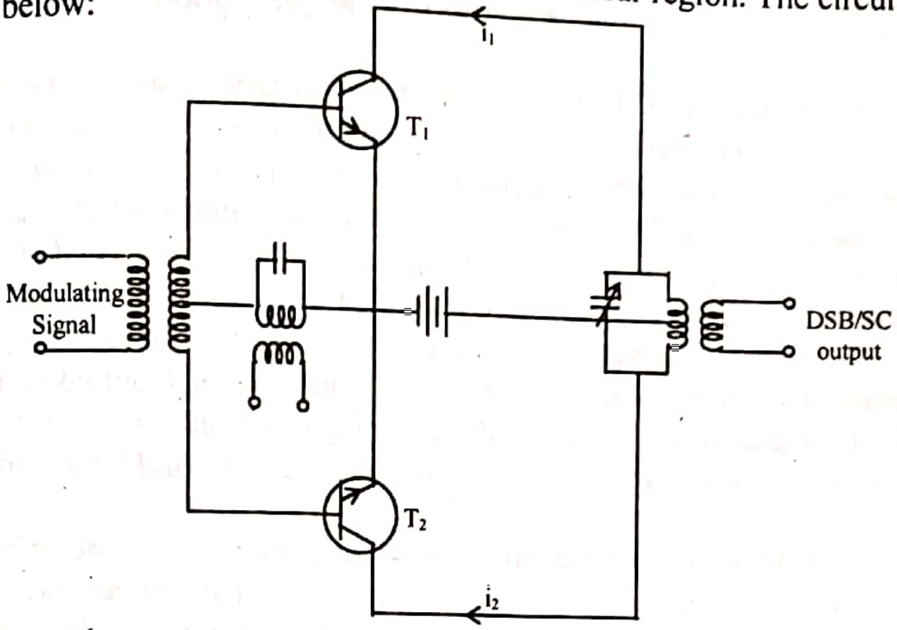
In practice, it is extremely difficult to build a wideband phase-shifting Hilbert transform network which offers a uniform phase shift of –90° over the entire message bandwidth. However, it is possible to design a practical system that accepts the message signal, as the input and yields two signals $m_1(t)$ and $m_2(t)$ at its output which are nearly equal in amplitude and are orthogonal to each other. These two signals could be fed to the multipliers. Note that, over the message bandwidth, there is a (nearly) 90° differential phase between $m_1(t)$ and $m_2(t)$. The actual phase relationship between the input and either of the outputs is not important, as long as the differential phase between the two outputs is 90°.

The phase-shift SSB modulator does not require band pass filters and is suitable even for message signals having frequency content right down to DC. However, the practical design of a wideband 90° phase-shift network is difficult.

c) What is balanced modulator? **[WBUT 2015]**

Answer:

Balanced modulator is used to suppress carrier i.e., to produce DSBSC signal. Pair of identical diodes or identical BJTS or identical FETS can be used in balanced modulator. These devices are biased so that they are used in non-linear region. The circuit diagram is shown in fig. below:



Let $v_m = V_m \sin\omega_m t$ be modulating signal and $v_c = V_c \sin\omega_c t$ be the carrier signal.

So, Input voltage at base of T_1 be $(V_c + V_m)$ and base of T_2 be $(V_c - V_m)$. If perfect symmetry is assumed, proportionality constants will be same for BJTS. Let these constants be a, band c.

Therefore, two band currents will be

$$i_1 = a + b(V_c + V_m) + c(V_c + V_m)^2$$
$$= a + bV_c + bV_m + cV_c^2 + 2cV_cV_m + cV_m^2$$

and $$i_2 = a + b(V_c - V_m) + c(V_c - V_m)^2$$
$$= a + bV_c - bV_m + cV_c^2 - 2cV_cV_m + cV_m^2$$

So, output current,

$$i = i_1 - i_2 = 2bV_m + 4cV_mV_c$$
$$= 2bV_m \sin\omega_m t + 2cV_mV_c\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right]$$

So, output voltage,

$$v_o = \alpha_i$$
$$= 2\alpha bV_m \sin\omega_m t + 2\alpha cV_mV_c\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right]$$
$$= A\sin\omega_m t - B\cos(\omega_c + \omega_m)t + B\cos(\omega_c - \omega_m)t \qquad ...(1)$$

where $A = 2\alpha\, bV_m$;

$B = 2\alpha\, cV_mV_c$

Equation (1) shows that the carrier has been cancelled out only two side bands and modulating frequencies appear at the output. Output tuned circuit is tuned to carrier frequency, signals of modulating frequencies that otherwise may appear at the output are rejected and only two side bands are present at output.

d) State the demerits of the direct method of FM generation. **[WBUT 2015]**

Answer:

(i) In the direct method it is difficult to obtain a high order stability in carrier frequency. This is because the carrier generation is directly affected by the modulating signal. The modulating signal directly controls the tank circuit of the carrier generator and, hence a stable oscillator circuit like the crystal oscillator, cannot be used (the crystal oscillator provides a stable but fixed frequency). Thus carrier generation cannot be of high stability which is an essential requirement.

A remedy to this problem is the indirect method (Armstrong method) of FM generation. In this method, the carrier oscillator is not required to respond to the modulating signal directly; rather, the carrier generation is isolated from other parts of the circuit. Hence, stable crystal oscillators can be used for generating carrier signal.

(ii) The non-linearity produces a frequency variation due to the harmonics of the modulating signal and, hence the FM signal is distorted. Proper care has to be taken for keeping this distortion minimum.

7. a) Explain with suitable block diagram the generation of FM signal using Armstrong method. **[WBUT 2015]**

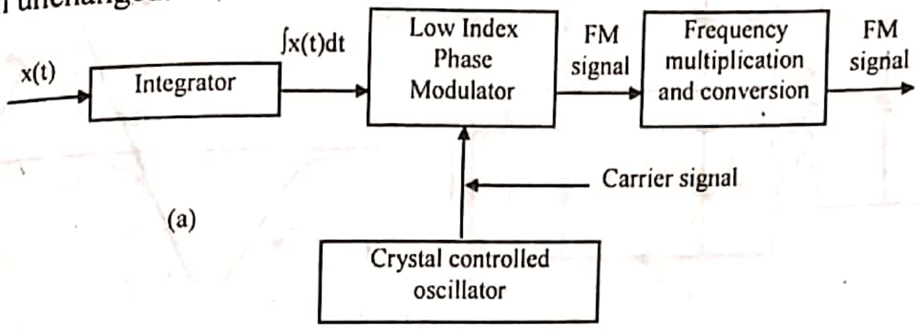
Answer:

Armstrong method of FM generation:

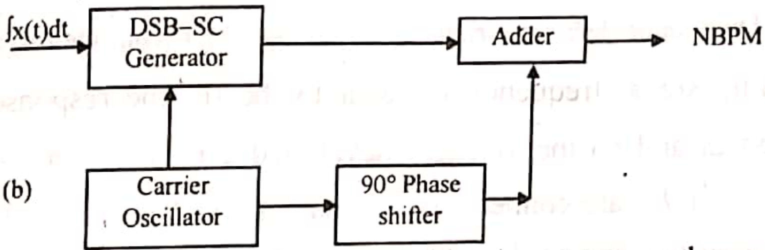
This method, of generation of FM signals, requires the generation of phase-modulated signal with a low index of modulation. The modulating signal is integrated prior to modulating the carrier, so that the output of the phase modulator becomes a frequency – modulated signal. The required phase modulated signal is generated with the help of a double-side-band suppressed carrier modulator, and hence the index of modulation, for the phase modulated signal, as also for the frequency modulated signal is limited to about 0.5 and more often to only 0.2. It may be mentioned here that if the two sidebands, of amplitude – modulated signal, are combined with its 90° phase shift carrier the resulting signal is a low modulation index phase modulated signal i.e. narrow band phase modulated (NBPM) signal. To express in different way an NBPM signal is nothing but a double-side band (DSB) signal to which a 90° phase-shifted carrier has been added.

In practice, one uses an FM signal with a large deviation. For example, the frequency deviation, used in FM broadcasting, is 75 KHz. Taking a representative frequency for the modulating signal to be 800 Hz, the required index of modulation works out to be 94, while the index with which the FM signal was generated, has not been more than say, 0.5. Thus this method of FM generation calls for an increase in the index of modulation. This can be achieved through frequency multiplication. The frequency multiplication, however, also results in multiplication of the carrier frequency of the FM signal.

Anyway, the carrier frequency can then be brought to the desired value through one or more steps of frequency conversion that leave the frequency deviation and the index modulation unchanged.



(a)



(b)

The above method of generation was suggested by Armstrong and goes after his name. It is an indirect method as it does not really modify the frequency of an oscillator but generates a PM signal to be converted subsequently into an FM signal. It is abbreviated as IFM (indirect frequency modulation).

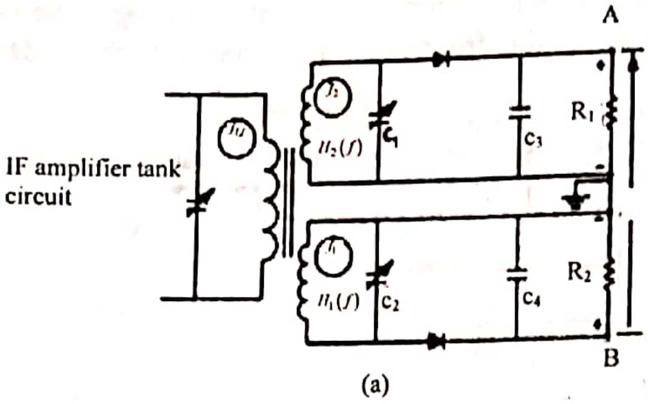
b) What are the problems in slope detectors and how is it overcome using a balanced slope detector? [WBUT 2015]

Answer:

Although it is simple and inexpensive, the slope detector suffers from one serious disadvantage, viz., non-linearity in the frequency-to-amplitude conversion. This non-linearity arises from the fact that the response curve of the resonant circuit can be considered to be linear only over a very small region.

Dual-slope Detector or Balanced Discriminator

To overcome the problem of non-linearity encountered in the simple slope detector discussed earlier, Foster and Seeley proposed the dual-slope detector. This makes use of two resonant circuits with identical responses but with slightly different resonant frequencies. The technique used in order to obtain a larger linear range is illustrated in Fig. (b).



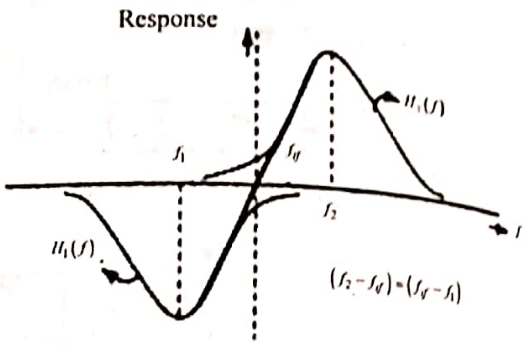


Fig: (a) Dual-slope detector circuit

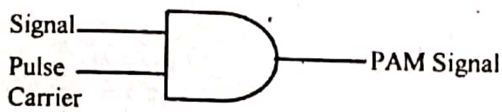
Fig: (b) Technique for larger linear range

When the incoming signal frequency is equal to the IF, the responses of $H_1(f)$ and $H_2(f)$ will be equal and so the voltages developed across R_1 and R_2 will be equal. From the way D_1 and D_2 are connected, terminals A and B will be at the same potential with respect to the ground and so E_0, the potential difference between them is zero. If the incoming signal has a frequency above the IF, the response of $H_2(f)$ will be more and that of $H_1(f)$ will be less (when compared to what it was when incoming signal frequency was IF). Hence the voltage drop across R_1 will be greater than the voltage drop across R_2. Hence, the terminal A will be at a higher potential than the terminal B with respect to ground and $E_0 \neq 0$. If the incoming signal has a frequency less than the IF, response $H_1(f)$ will be more than the response $H_2(f)$, causing B to be at a higher potential than A. Thus, the frequency variations of the incoming FM signal are converted into corresponding variations in the amplitude of E_0. Therefore, E_0 will be the modulating signal assuming the overall response [see Fig. 5.28(b)] to be perfectly linear between f_1 and f_2.

c) Discuss the method for modulation and demodulation of PAM signal.
[WBUT 2015]

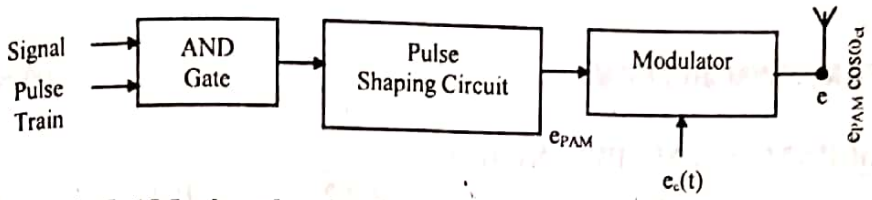
Answer:

Generation of PAM signals



In a PAM generator, the signal is fed to one input of the AND gate, pulses of the sampling frequency are applied to the other input of the AND gate to open it during the desired time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse-shaping network, which gives them flat tops. As mentioned above, the signal is then frequency or amplitude modulated, so that the system becomes PAM-FM or PAM-AM respectively.

The generation of PAM-AM signal is shown in the figure below.



Detection of PAM-AM signals: At the receiving end, the signal is demodulated or retranslated the spectrum to its original position. The output of the demodulator is the pulse train of varying amplitude. The signal $e_m(t)$ is recovered from e_{PAM} (t) by filtering through a low-pass filter as shown in figure below.

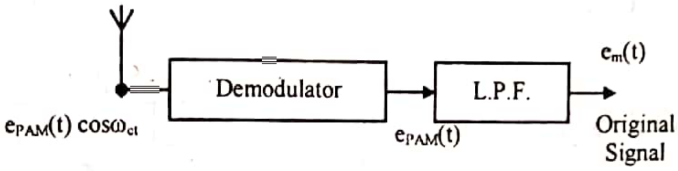


Fig: Demodulation of PAM AM Signal

So, PAM signal can be demodulated by a low pass filter with a cut off frequency just large enough to accommodate the highest frequency component of the message signal. Reconstructed signal exhibits amplitude distortion caused by aperture effect that may be removed by using equation.

d) Why flat top sampling is more preferred that natural sampling? [WBUT 2015]

Answer:

Difference between Natural and Flat-top Sampling:

i) In natural sampling, the tops of the modulated pulses follow the natural waveform of the modulating signal; whereas in flat-top sampling, the tops of the modulated pulses are flat.

ii) The electronic circuitry needed in natural sampling is more complicated as compared to that needed in flat-top sampling. The distortion in a demodulated signal of natural sampling is less than that of flat-top sampling

iii) The demodulation of a natural-sampled signal needs preservation of the pulse-top shape, which is a relatively difficult task. Therefore, normally, flat-top sampling is preferred over natural sampling.

8. a) The antenna current of an AM transmitter is 8 A, if only the carrier is sent, but it increases to 8.93A, if the carrier is modulated by a single sinusoidal wave. Determine the percentage modulation. Also find the antenna current if the per cent of modulation changes to 0.8. [WBUT 2016]

Answer:

Given: I_c =8A I_t=8.93A m=0.8

Formula: $I_t=I_c\ (1+m2/2)\ ½$

$8.93=8(1+m^2/2)\ ½$

m=0.701
I_t=8 (1+0.82/2) ½
I_t=9.1A

b) Compare PAM, PWM and PPM. **[WBUT 2016]**

Answer:

Comparison of PAM / PWM / PPM Systems

Sl. No.	Parameter	PAM	PWM	PPM
1.	Variation parameter	Amplitude	Width	Position
2.	Bandwidth required	Less	High	High
3.	Noise immunity	Less	High	High
4.	Transmitted power	Varies with amplitude of pulses	Varies with width of pulses	Remains constant
5.	Need for synchronization pulse	No	No	Yes
6.	Complexity of generation and detection	Less	More	More

9. a) What are the advantages and disadvantages of AM over DSB-SC? **[WBUT 2017]**

Answer:

AM	DSB-SC
1. AM or DSB-FC is less power efficient.	1. DSB-SC is more power efficient than AM.
2. Carrier is transmitted.	2. Carrier is not transmitted.
3. Maximum power efficiency is 33.33%	3. Maximum power efficiency is 50%
4. AM signal can use low level modulator and class C amps or a high level final stage modulator with class C carrier drive.	4. DSB-SC signal must be generated low level with a doubly balanced modulator and amplified linearity, which requires more components and reduces amplifier efficiency.
5. The de-modulation of AM is simpler.	5. The De-modulation of DSB-SC is complex.

b) The total power content of an AM signal is 1000 W. determine the power being transmitted at the carrier frequency and at each of the sidebands when the percent modulation is 100%. **[WBUT 2017]**

Answer:

$$P_t = 1000\text{ W} = 1\text{ kW}$$

$$m = 1$$

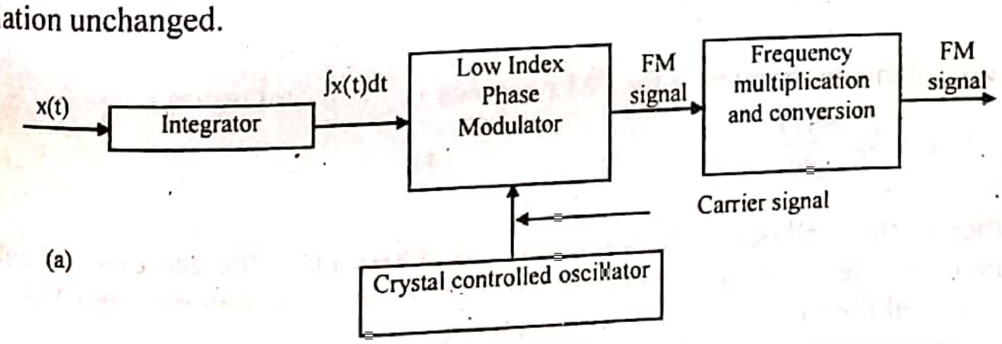
$$P_c = \frac{P_t}{\left(1+\frac{m^2}{2}\right)} = \frac{1}{1+\frac{1}{2}} = \frac{1}{\frac{3}{2}} = \frac{2}{3}\text{kW} = 666.67\text{ W}$$

So, carrier power is 666.67 W
Each sideband power,

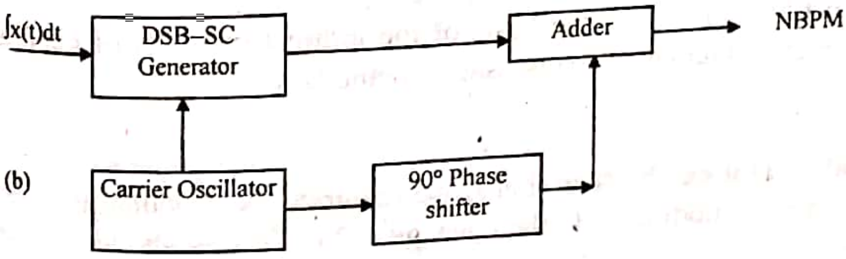
$$P_{SB} = \frac{P_t - P_c}{2} = \frac{P_c \cdot m^2}{4} \doteq 166.67 \text{ W}$$

10. a) Explain indirect method of FM generation. **[WBUT 2017]**

Answer:
This method, of generation of FM signals, requires the generation of phase-modulated signal with a low index of modulation. The modulating signal is integrated prior to modulating the carrier, so that the output of the phase modulator becomes a frequency – modulated signal. The required phase modulated signal is generated with the help of a double-side-band suppressed carrier modulator, and hence the index of modulation, for the phase modulated signal, as also for the frequency modulated signal is limited to about 0.5 and more often to only 0.2. It may be mentioned here that if the two sidebands, of amplitude – modulated signal, are combined with its 90° phase shift carrier the resulting signal is a low modulation index phase modulated signal i.e. narrow band phase modulated (NBPM) signal. To express in different way an NBPM signal is nothing but a double-side band (DSB) signal to which a 90° phase-shifted carrier has been added. In practice, one uses an FM signal with a large deviation. For example, the frequency deviation, used in FM broadcasting, is 75 KHz. Taking a representative frequency for the modulating signal to be 800 Hz, the required index of modulation works out to be 94, while the index with which the FM signal was generated, has not been more than say, 0.5. Thus this method of FM generation calls for an increase in the index of modulation. This can be achieved through frequency multiplication. The frequency multiplication, however, also results in multiplication of the carrier frequency of the FM signal. Anyway, the carrier frequency can then be brought to the desired value through one or more steps of frequency conversion that leave the frequency deviation and the index modulation unchanged.



(a)



(b)

The above method of generation was suggested by Armstrong and goes after his name. It is an indirect method as it does not really modify the frequency of an oscillator but generates a PM signal to be converted subsequently into an FM signal. It is abbreviated as IFM (indirect frequency modulation).

b) How can you produce FM using PM modulator and PM using FM modulator?
[WBUT 2017]

Answer:
In FM and PM, we are varying the angle of the carrier and in these methods, the instantaneous frequency and phase changes simultaneously. To make the thing simpler, let's consider the expression for the carrier

$e_c(t) = E_c \cos(\omega_c t + \theta_0)$

Here, ω_c = carrier frequency

and $\theta(t) = \omega_c t + \theta_0$ = phase of the carrier.

In FM, we vary frequency of the carrier, according to $e_m(t)$ so that instantaneous frequency is given by

$$\underset{(FM)}{\omega_i} = \frac{d\theta}{dt} = \omega_C + k_f\, e_m(t) \qquad \text{... (1)}$$

And the phase of the FM signal is therefore given as

$$\theta(t) = \int \omega_i dt$$
$$= \int [\omega_c + k_f e_m(t)].dt$$
$$\underset{(FM)}{\theta(t)} = \omega_C t + k_f \int e_m(t).dt \qquad \text{... (2)}$$

In PM, however we vary phase of the carrier according to the instantaneous value of $e_m(t)$ i.e.

$$\underset{(PM)}{\theta(t)} = \omega_c t + \theta_0 + k_p.e_m(t) \qquad \text{... (3)}$$

Hence, the instantaneous frequency for PM becomes

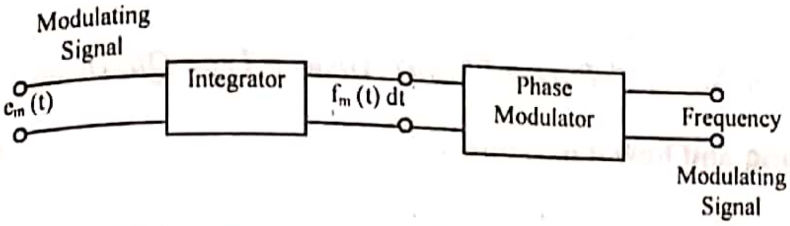
$$\underset{(PM)}{\omega_i} = \frac{d\theta}{dt} = \omega_C + k_p.\frac{m(t)}{dt} \qquad \text{... (4)}$$

Thus, in either of the methods both frequency and phases gets changed even though we tried to vary only one of these parameters. Hence, by merely watching the FM / PM wave, one can't tell the involved modulation scheme.

This relation between FM and PM, is the basis of the indirect methods of generation of FM and PM waves which is known as Armstrong's methods.

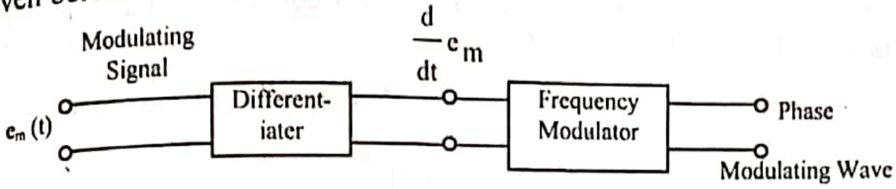
FM from PM

From equations (2) and (3) it can be seen that if we integrate the modulating signal $e_m(t)$ and then allow to phase – modulate it, then we get FM. This is shown in the block diagram given below:



PM from FM

Similarly from (1) and (4), it can be seen that if we define the modulating signal $e_m(t)$ and then allow to frequency modulating it, then we get PM. This is shown in the block diagram given below:



c) In an FM system, when the audio frequency is 400 Hz and the AF voltage 2.5V, the frequency deviation is 3.6 kHz. If the AF voltage is raised to 7.5 V, what is the new deviation? Find modulation index also. [WBUT 2017]

Answer:

Audio frequency, $f_{m_1} = 400\text{ Hz} = 0.4\text{ kHz}$

AF voltage, $E_{m_1} = 2.5\text{ V}$

Frequency deviation, $\Delta f_1 = 3.6\text{ kHz}$

We know, $\Delta f = \dfrac{kE_m}{2\pi}$

$\therefore \quad \Delta f \propto E_m$

We can write,

$$\frac{\Delta f_1}{E_{m_1}} = \frac{3.6\text{ kHz}}{2.5\text{ V}}$$

Now, the AF voltage is raised to 7.5V.

$\therefore \quad E_{m_2} = 7.5\text{ V}$

$\therefore$ New frequency deviation,

$$\Delta f_2 = \frac{\Delta f_1}{E_{m_1}} \times E_{m_2} = \frac{3.6}{5.5} \times 7.5 = 10.8\text{ kHz} \quad \textbf{(Ans.)}$$

$$\text{Modulation index, } m_{f_2} = \frac{\Delta f_2}{f_{m_2}} = \frac{10.8}{0.4} = \frac{108}{4} = 27 \quad \textbf{(Ans.)}$$

11. a) A band limited signal $x(t)$ is sampled by a train of rectangular pulses of width T and period T_S. Find an expression for the sampled signal. Determine the spectrum of the sampled signal and sketch it. [WBUT 2017]

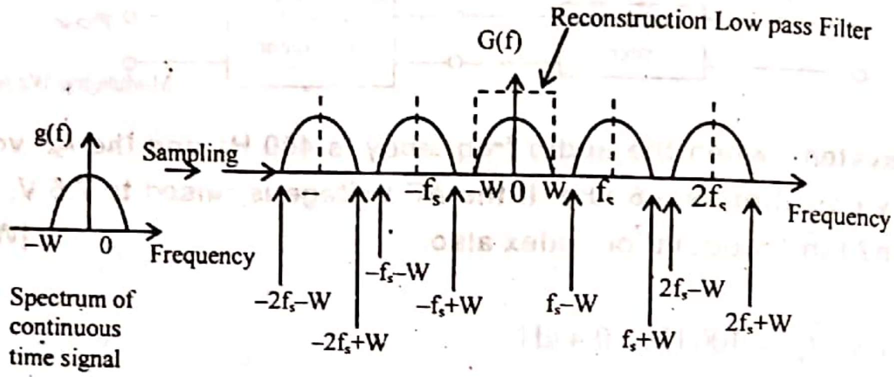
Answer:
Refer to Question No. 1(1st Part) of Short Answer Type Questions.

b) What is aliasing and how it is reduced? **[WBUT 2017]**

Answer:

1st Part:

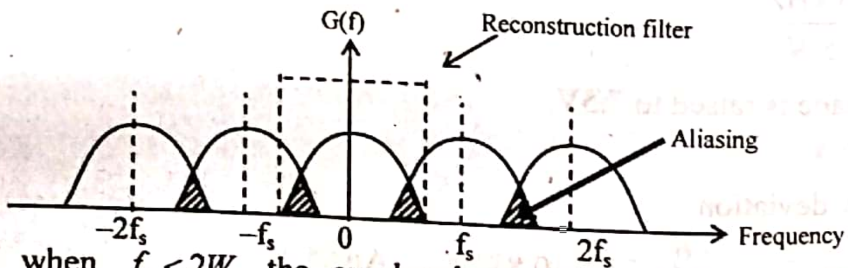
An analog signal is converted into pulse amplitude modulated (PAM) signal by the process of sampling. A PAM signal is a discrete-time signal. Fourier analysis of PAM signal gives the representation of PAM signal in the frequency domain. The spectrum of PAM signal is shown in the diagram below:



In the above diagram the sampling frequency, f_s, is greater or equal to twice the maximum frequency of the input signal, W i.e. $f_s \geq 2W$. The spectrum of the sampled signal $G(f)$ repeats periodically without overlapping.

If, however, $f_s < 2W$, the successive cycles of the sampled spectrum will overlap with each other.

This is shown in the diagram below.



In this case when $f_s < 2W$ the overlapping of the sidebands will produce beat frequencies that will interfere with the desired signal. This interference effect is called aliasing or foldover. If aliasing takes place i.e. when $f_s < 2W$, it is not possible to recover the original desired signal.

2nd Part:

To satisfy Nyquist theorem of sampling i.e. to avoid under sampling the signal must be band-limited to W Hz such that $f_s \geq 2W$. This band-limiting of the original signal is done by a low pass filter of cut off frequency W Hz. This LPF will block all frequencies *above W Hz.* Such a low pass filter used to avoid aliasing is called an anti-aliasing filter *or pre-alias filter.* An anti-aliasing filter is always used before sampling in the generation *of PCM signal.* It may be mentioned that in voice telephony speech is band-limited to *300 – 3400* Hz using pre-alias filter and speech is sampled at an 8 KHz rate.

12. Write short notes on the following:
a) Time division multiplexing **[WBUT 2014, 2015, 2016, 2018]**
b) PCM transmitter and receiver **[WBUT 2014, 2018]**
c) Envelope detector **[WBUT 2015, 2016, 2017]**
d) Anti aliasing filter **[WBUT 2015]**
e) Necessity of Modulation **[WBUT 2016]**
f) MODEM **[WBUT 2017]**
g) FDM **[WBUT 2017]**

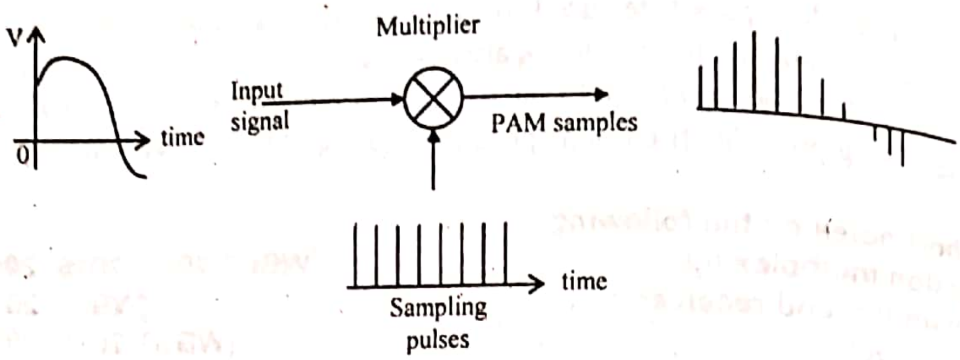
Answer:

a) Time Division Multiplexing:
Refer to Question No. 13(2nd Part) of Short Answer Type Questions.

b) PCM transmitter and receiver:
Refer to Question No. 3(a) of Long Answer Type Questions.

c) Envelope detector:
Refer to Question No. 5(a) of Long Answer Type Questions.

d) Anti aliasing filter:
When an analog signal is sampled at a rate below the Nyquist rate, the sidebands overlap producing an interference effect. This interference effect is called aliasing effect. If aliasing takes place, it is not possible to recover the original analog signal.
For successful recovery of the original signal it is necessary to avoid aliasing. An information signal may contain a large number of frequencies. Thus to satisfy Nyquist theorem of sampling i.e. to avoid under sampling the signal must be band limited to W Hz such that $f_s \geq 2W$. This band-limiting of the original signal is done by a low pass filter of cut off frequency W Hz. This LPF will block all frequencies above W Hz. Such a low pass filter used to avoid aliasing is called an anti-aliasing filter or pre-alias filter. A pre-alias filter is always used before sampling in the generation of PCM signal. As an example it may be mentioned that in voice telephony speech is band limited to 300 – 3400 Hz using pre-alias filter. In digital telephone network, speech is sampled at a 8KHz rate. Though minimum sampling frequency would have been $2 \times 3400\text{Hz} = 6.8\text{KHz}$, some over sampling is allowed by sampling at 8KHz to provide for lack of sharp cut-off characteristics for practical filters.
An anti-aliasing filter is usually used at the input of a PAM generator to avoid the effect of aliasing. PAM signal is generated by sampling the input analog signal in a sampler circuit. This is shown in the diagram below.



The sampling is done in accordance with the sampling theorem i.e. the sampling frequency f_s is kept equal to or higher than twice the maximum frequency W present in the input analog signal. If, however, $f_s < 2W$, then aliasing occurs and recovery of the original analog signal will not be possible. Since f_s is usually kept unchanged, the input analog signal is passed through a low-pass filter before sampling to band-limit the analog signal in conformity with the sampling theorem.

e) Necessity of Modulation:

The necessity of modulation serves the following purposes.

i) Frequency shifting

Basically in modulation, the frequency of modulating signal become shifted to higher frequency region. So it can be said that frequency shifting is one kind of frequency translation.

ii) Frequency multiplexing

Multiplexing means many into one. In modulation carrier signal is added to the modulating signal to make a common envelop upon the modulating signal. Hence frequency multiplexing is another purpose of modulation.

iii) Antenna size

Normally for free space communicating systems, antenna is used to transmit and receive messages. If the signal were transmitted without modulation, the height of antenna needed for an effective radiation would be half of the wavelength. To construct and install such an antenna is impracticable.

f) Modem:

The modems usually denote both the process of modulation in the transmission side and that of demodulation on the receive side. Modern data modems also include sophisticated error detection and correction devices for the reliable data transmission.

The modulation and demodulation can be done by using several techniques such as amplitude modulation, frequency modulation and phase modulation etc. In Amplitude Modulation, the base-band signal of waveform p(t) generated by digital data set is mixed with a sinusoidal carrier signal say $\cos(2\pi f_0 t)$ at the transmitting end to get a modulated signal represented by $P(t) = p(t)\cos(2\pi f_0 t)$

These frequencies at the receiving side are again mixed with the same carrier frequency of $\cos(2\pi f_0 t)$, $N(t) = P(t)\cos(2\pi f_0 t) = p(t)\cos^2(2\pi f_0 t) = \frac{p(t)}{2} + \frac{p(t)}{2}\cos(2\pi\,(2f_0)t)$

The high frequency component of $\cos(2\pi\,(2f_0)t)$ is then filtered out, leaving the demodulated signal $p(t)/2$ from which the original digital signal may be retrieved. In Amplitude Shift Keying (ASK), the strength of the carrier signal is varied to represent binary 1 or 0. Both frequency and phase remain constant while the amplitude changes. The peak amplitude of the signal during each bit duration is constant, and its value depends on the bit (0 or 1). ASK transmission is highly susceptible to noise interference Noise usually affects the amplitude; therefore, ASK is the modulation most affected by noise. A popular ASK technique is called ON/OFF Keying (OOK). In OOK, one of the bit values is represented by no voltage. The advantage is a reduction in the amount of energy required to transmit information. In Frequency Shift Keying (FSK), the frequency of the carrier signal is varied to represent binary 1 or 0. The frequency of the signal during each bit duration is constant and the value depends on the bit (0 or 1). Both peak amplitude and phase remain constant. FSK avoids most of the problems from noise. The limiting factors of FSK are the physical capabilities of the career.

In Phase Shift Keying (PSK), the phase of the carrier is varied to present binary 1 or 0. Both peak amplitude and frequency remain constant as the phase changes e.g. if we start with a phase of 0° to represent binary 0, then we can change the phase to 180° to send binary 1. The phase of the signal during each bit duration is constant and its value depends on the bit (0 or 1). PSK is not susceptible to the noise degradation that affects ASK or to the bandwidth limitation of FSK. This means that smaller variations in the signal can be detected reliably by the receiver. Therefore, instead of utilizing only two variations of a signal, each representing 1 bit, 4 variations can be used and each phase shift is represented by 2 bits. This technique is called 4-PSK or Q-PSK (Quadrature Phase Shift Keying). The pair of bits represented by each phase is called a dibit.

Q-PSK Characteristics

Dibit	Phase
00	0°
01	90°
10	180°
11	270°

Similarly for 8-PSK, the signal is varied by shift of 45°. With eight different phases, each shift can represent 3 bits (1 tribit) at a time.

The modulation techniques of combination of two or more schemes are more common to have more common to have more robustness. Under these schemes, Quadrature Amplitude Shift Keying (QAM) is more populярly used for high speed modems. QAM is a combination of ASK and PSK so that a maximum contrast between each signal unit (bit, dibit, tribit and so on) is achieved. In QAM, incoming bits are mapped into two base band signals $P_1(t)$ and $P_2(t)$. Then $P_1(t)$ is multiplied by $\cos(2\pi f_0 t)$ and $P_2(t)$ by

$\sin(2\pi f_0 t)$. The sum of these products forms the transmitted QAM signals as shown in Fig. At the receiver, the incoming waveform is separately multiplied by $\cos(2\pi f_0 t)$ and $\sin(2\pi f_0 t)$. When the multiplied waveforms are filtered, we get $P_1(t)/2$ and $P_2(t)/2$, from which the bit stream is reconstructed.

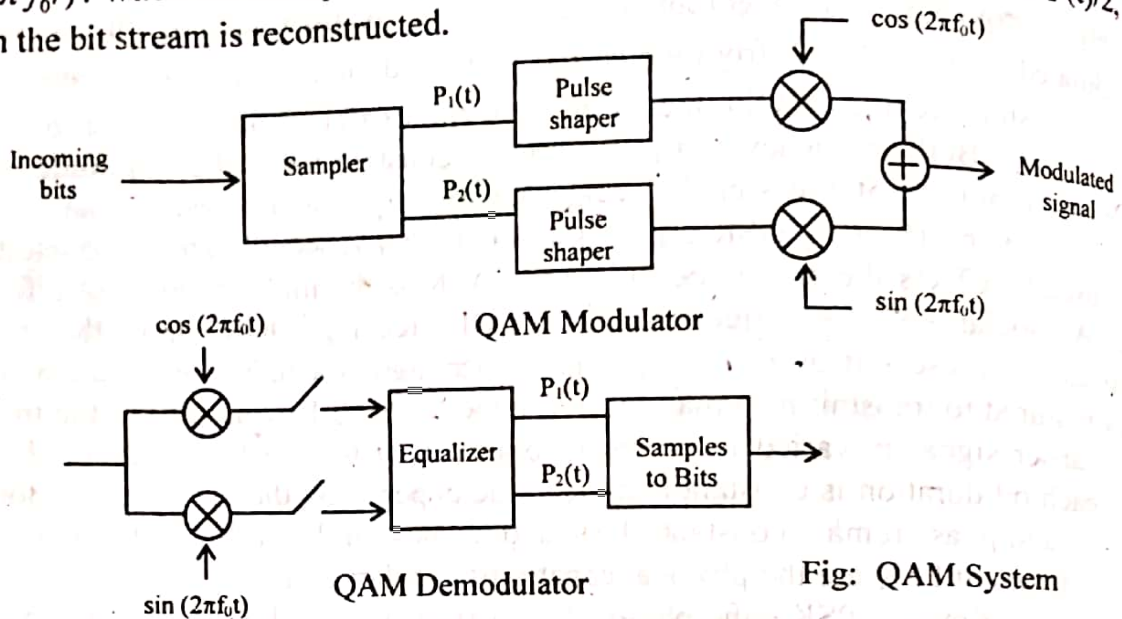


Fig: QAM System

A series of standards, known as V-series, has been defined by CCITT for interfacing DTEs to DCEs operating with Public Switched Telephone Network (PSTN). The series also defines a variety of DCEs using different modulation techniques and operating at different speeds using either a leased PSTN or dial-up line.

Some of the standards defined by CCITT known as V series are:

V.5- Standardization of data signaling rates for synchronous data transmission in PSTN.

V.24 - DTE – DCE interface and control signals.

V.28 - DTE – DCE electrical characteristics for unbalanced double- current interchange circuits.

V.53 - Limits for the maintenance of telephone type circuits used for data transmission.

Some of the important V-series modem standards are given in the following table.

Table: Some V-Series modem Standards

V-Series	Speed (bps)	Modulation	Application
V.22	1200	4-ary DPSK	FD, D-up, 2-W
V.22 bis	2400 (1200 fall back)	16- ary QAM	FD, D-up, 2-W
V.23	600/1200	FSK	HD, D-up
V.29	9600	16-ary QAM	FD, 4-W
V.32	Upto 9600	32-ary QAM	FD, D-up, 2-W
V.33	14400	128-ary AM-PM	FD, 4-W

FD ⇒ Full Duplex; D–up ⇒ Dial – up connection

2W ⇒ 2 – Wire leased circuit; HD ⇒ Half duplex

4W ⇒ 4 – wire leased circuit.

g) FDM: ***Refer to Question No. 8 of Short Answer Type Questions.***

13. Explain the generation of PWM & PPM. **[MODEL QUESTION]**

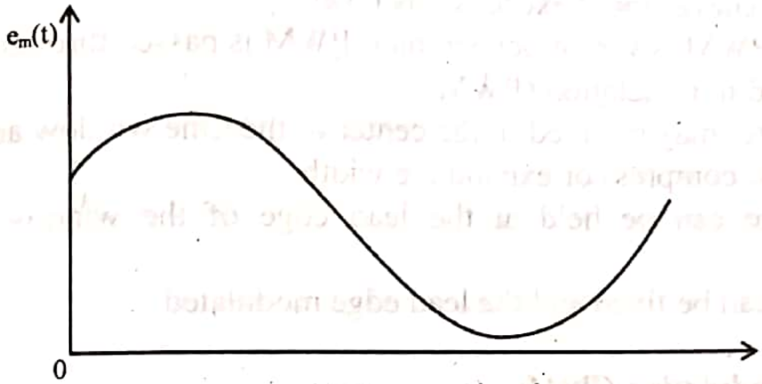
Answer:

Pulse-Width Modulation (PWM)

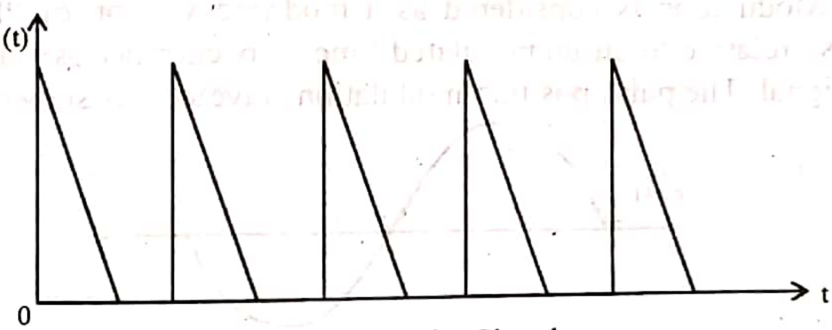
In Pulse-Width Modulation, the samples of the message signal are used to vary the duration of the individual pulses.

Pulse-width modulation (PWM) of a signal or power source involves the modulation of its duty cycle, to either convey information over a communications channel or control the amount of power sent to a load.

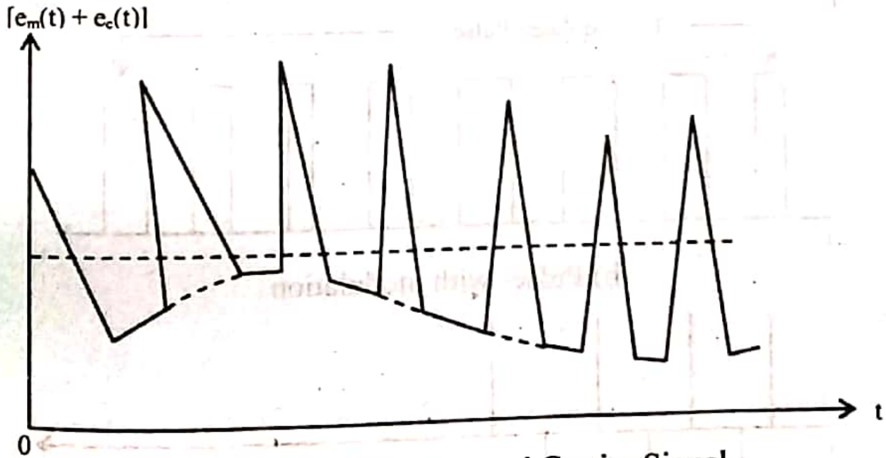
Pulse width may be varied by varying the time of occurrence of the leading edge, trailing edge or both edges of the pulse in accordance with sampled value of the modulating wave. The detailed about PWM generation is shown below.



a) Message Signal

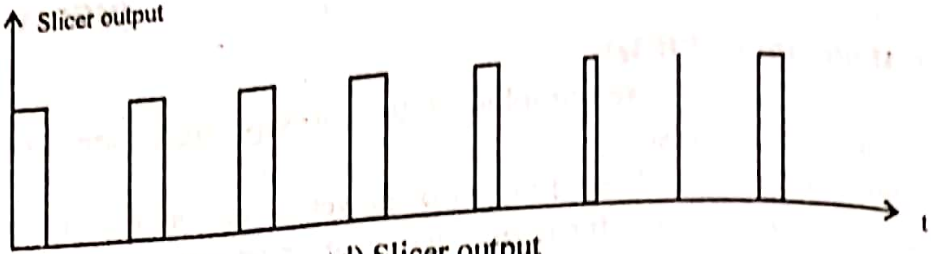


b) Saw tooth Carrier Signal



c) Combined Message and Carrier Signal

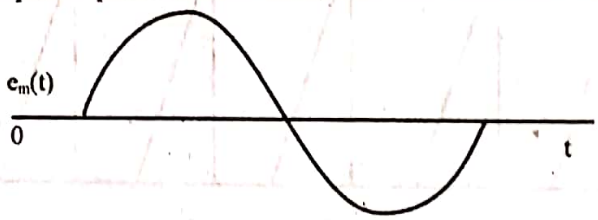
Scanned by CamScanner



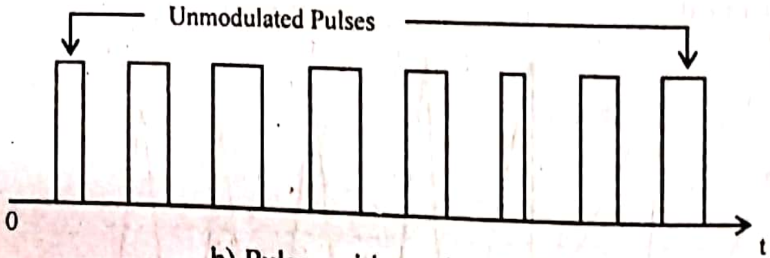
d) Slicer output

The procedure for generating PWM wave is described above. The message signal and saw-tooth wave are added and the combination is applied to a slicer.
Ideal slicer has the property that output is zero whenever the input is below slicing level and is constant whenever input exceeds this level.
Demodulation of PWM is very much simpler. PWM is passed through a low pass filter.
Types of pulse-width modulation (PWM):
1. The pulse center may be fixed in the center of the time window and both edges of the pulse moved to compress or expand the width.
2. The lead edge can be held at the lead edge of the window and the tail edge modulated.
3. The tail edge can be fixed and the lead edge modulated.

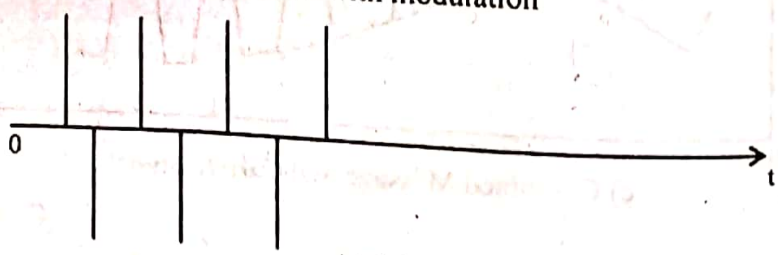
Pulse-Position Modulation (PPM)
Pulse-Position Modulation is considered as a modified version of PDM. In PPM, the position of pulse relative to an unmodulated time of occurrence is varied in accordance with message signal. The pulse position modulation waveform is shown below.



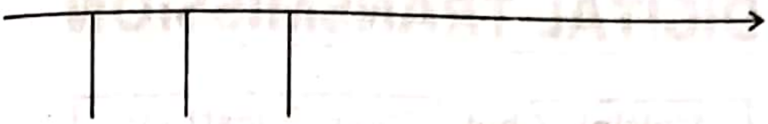
a) Modulating Signal



b) Pulse- with modulation



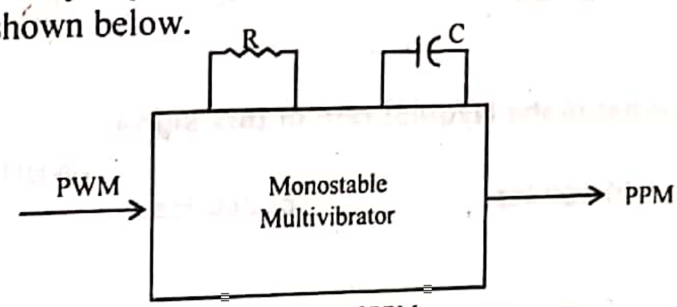
c) Differential Pulse



d) Pulse Position Modulation

Generation of PPM

The simplest method of generating PPM wave from PWM wave is to use a monostable multivibrator. The monostable multivibrator can be triggered from stable to quasi-stable state by applied external pulses. If a monostable multivibrator is so designed that it triggers at the trailing edges of a PWM signal then if PWM signal is applied at the input, output will be obviously a pulse position modulated signal. The device for converting PWM into PPM is shown below.



Generation of PPM

Demodulation of PPM

For demodulation purposes, PPM is first concerted into PWM with the help of a Flip-Flop. One of the inputs of multivibrator receives trigger pulses from a local generator which is synchronised by trigger pulses received from transmitter. PPM pulses are fed to the other base of Flip-Flop and make switch 'ON' of Flip-Flop. The period of time during which this particular stage is OFF depends on time difference between two triggers so that resulting pulse rate width that depends on time displacement of each individual PPM pulses.

Advantage

Advantage of pulse modulation system is that the increased bandwidth consumed by pulses may be used for improving noise performance.