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UNIT

Introduction

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1-2 B (EC-Sem-4)

Introduction

PART-1

Review of Signals and Systems, Frequency Domain Representation of Signals.

CONCEPT OUTLINE

Fourier transform is a transformation technique which transforms signal from continuous-time domain to the corresponding frequency-domain and vice-versa, which applies for both periodic as well as aperiodic signals.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 1.1. Define signal. Give the classification of signals.

OR

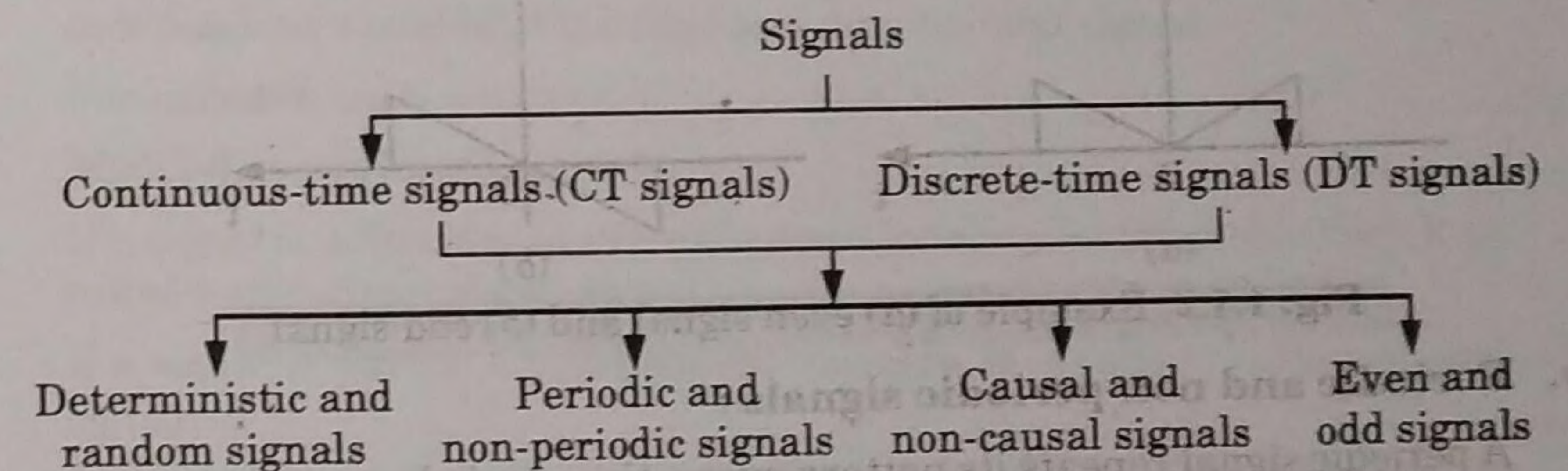
Differentiate between the following:

- Continuous-time signals and discrete-time signals.
- Periodic and aperiodic signals.
- Deterministic and random signals. **AKTU 2016-17, Marks 10**

Answer

Signal : It is a physical quantity that varies with time, space or any other independent variable or variables. It gives the information about behaviour or nature of the phenomenon.

Classification of signals : The signals are broadly classified as :



- Continuous-time and discrete-time signals :** A signal $x(t)$ is a continuous-time signal if time t , is a continuous variable. If time t , is a discrete variable, then signal is called as discrete-time signal and denoted by $x[n]$.

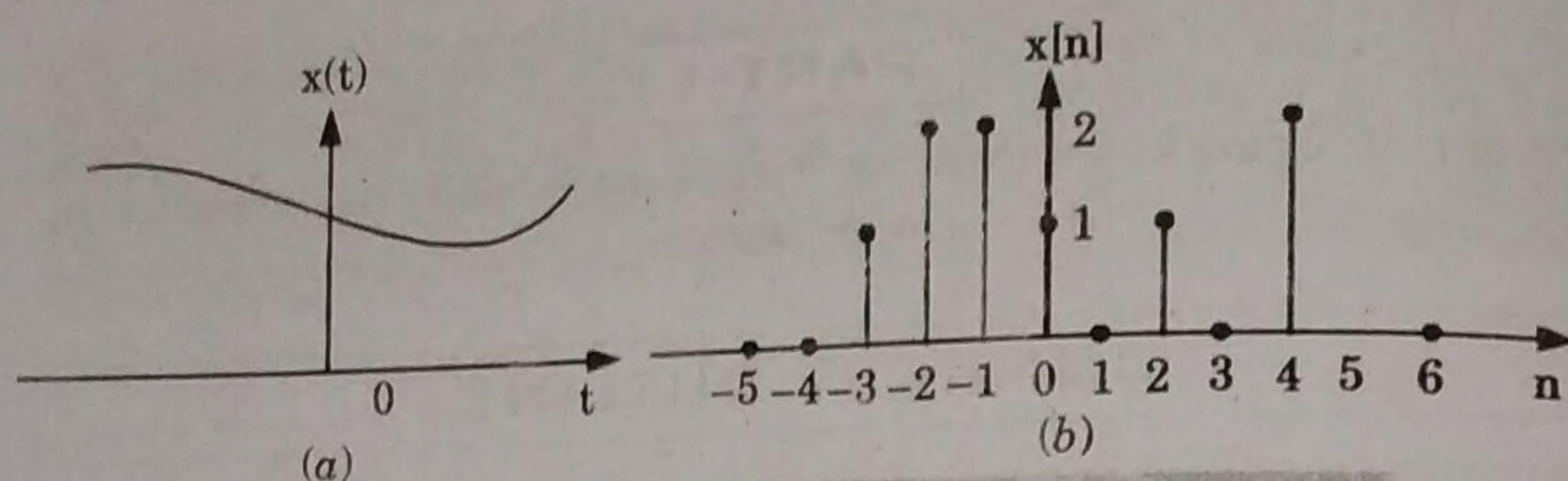


Fig. 1.1.1. (a) Continuous-time (b) Discrete-time signals.

The sequence shown in Fig. 1.1.1(b) can be written as

$$x[n] = \{1, 2, 2, 1, 0, 1, 0, 2\}$$

↑

ii. Deterministic and random signals:

1. Deterministic signals can be completely specified in time. The pattern of this signal is regular and can be characterized mathematically. So, its future value at any instant can be determined easily.

For example : $y(t) = b \sin(\pi t)$.

2. Random signals take random values at any given time. The pattern of this signal is irregular and it cannot be defined mathematically. So, future value of these signals at any instant cannot be predicted.

For example : noise.

- iii. **Even and odd signals:** A signal $x(t)$ or $x[n]$ is referred to as symmetric or even signal if

$$x(-t) = x(t) \text{ or } x[-n] = x[n]$$

and referred to as non-symmetric or odd signal if

$$x(-t) = -x(t) \text{ or } x[-n] = -x[n]$$

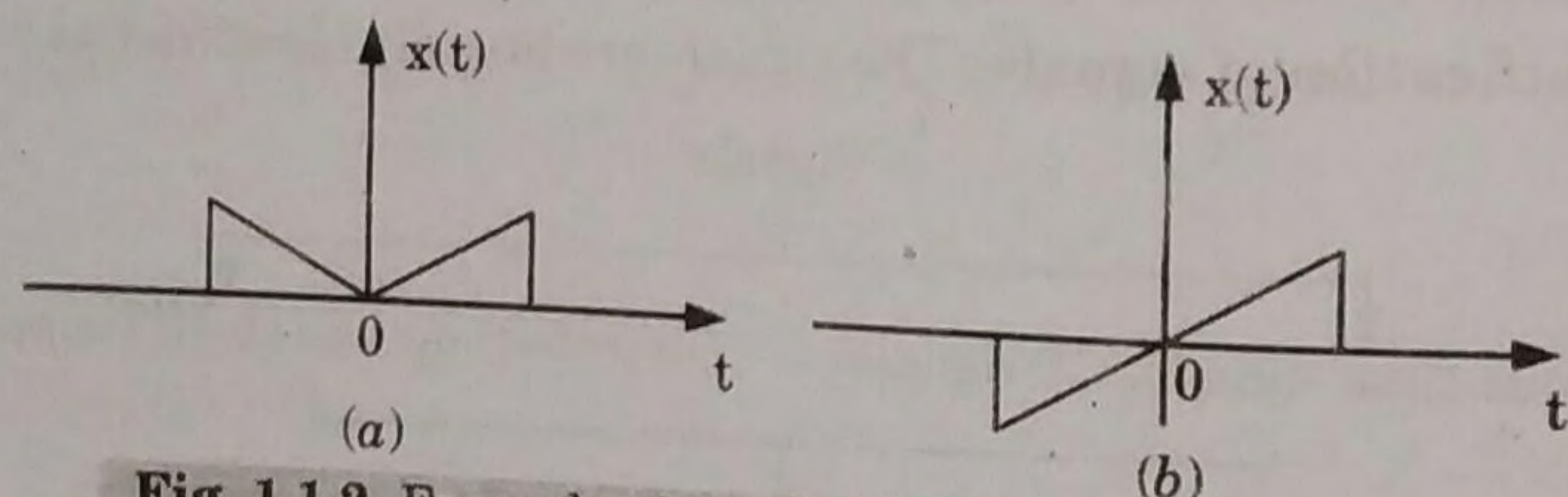


Fig. 1.1.2. Example of (a) even signal and (b) odd signal.

iv. Periodic and non-periodic signals:

1. A periodic signal repeats its pattern over a time period. For continuous-time, $x(t + T) = x(t)$ where, T is period of $x(t)$. For discrete-time, $x[n + N] = x[n]$ where, N is period of $x[n]$.

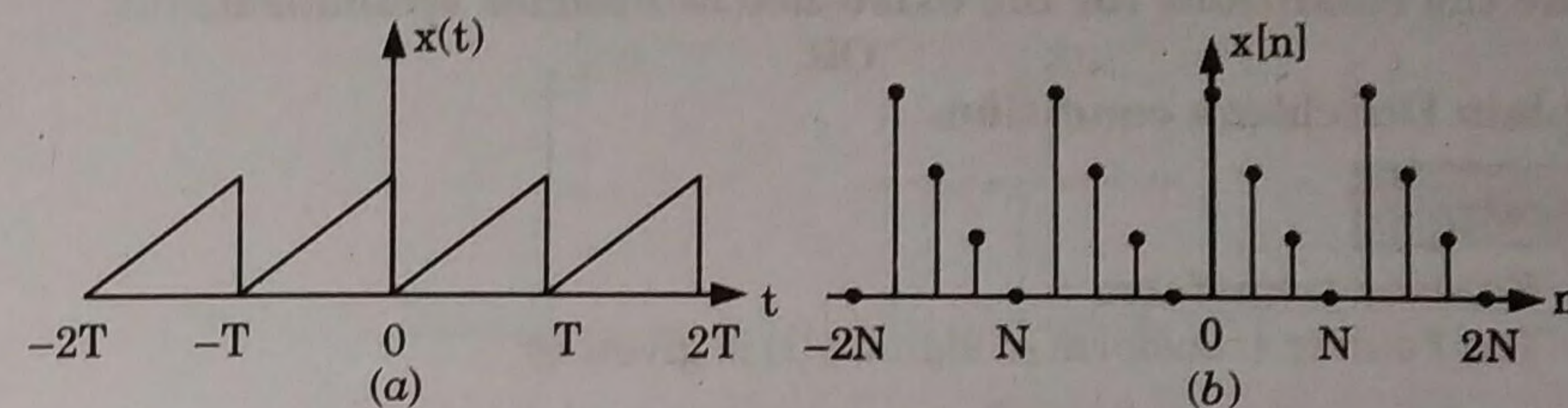


Fig. 1.1.3. Examples of periodic signals.

2. A signal which does not satisfy the periodicity condition is referred to non-periodic signal.

v. Energy and power signals:

Energy signal: The energy signal is one that has a finite energy and zero average power i.e., $0 < E < \infty$ and $P = 0$.

$$\text{For continuous-time signal, } x(t); E = \int_{-\infty}^{\infty} |x(t)|^2 dt$$

$$\text{For discrete-time signal, } x[n]; E = \sum_{n=-\infty}^{\infty} |x[n]|^2$$

Power signal: A power signal is defined as a signal having finite average power and infinite energy i.e., $0 < P < \infty$ and $E = \infty$

$$\text{For continuous-time signal, } x(t); P = \lim_{T \rightarrow \infty} \frac{1}{2T} \int_{-T}^T |x(t)|^2 dt$$

$$\text{For discrete-time signal, } x[n]; P = \lim_{N \rightarrow \infty} \frac{1}{2N+1} \sum_{n=-N}^N |x[n]|^2$$

vi. One-dimensional and multi-dimensional signals:

One-dimensional signal: If the signal is a function of single independent variable, it is called one-dimensional signal.

For example : $x(t) = a \cos \omega_0 t$.

Multi-dimensional signal:

1. If a signal is a function of more than one independent variable then it is called multi-dimensional signal.
2. If a signal is having n independent variable, it is called n -dimensional signal. For example : A picture is an example of a two dimensional signal since the intensity $I(x, y)$ at each point is a function of two independent variables.

Que 1.2. Define Fourier transform (FT) and inverse Fourier transform (IFT).

OR

State the conditions for the existence of Fourier transform.

OR

Explain Dirichlet's condition.

Answer**Fourier transform :**The Fourier transform of signal $x(t)$ is given by,

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

Inverse Fourier transform :

The inverse Fourier transform is given by,

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega ; \text{ for } -\infty < t < \infty$$

Condition :

- The Fourier transform does not exist for all aperiodic functions. The conditions for a function $x(t)$ to have Fourier transform, called Dirichlet's conditions which are as follows :
 - $x(t)$ is absolutely integrable over the interval $-\infty$ to ∞ , i.e.,

$$\int_{-\infty}^{\infty} |x(t)| dt < \infty$$
 - $x(t)$ has a finite number of discontinuities in every finite time interval. Further, each of these discontinuities must be finite.
 - $x(t)$ has a finite number of maxima and minima in every finite time interval.
- Dirichlet's condition is a sufficient condition but not necessary condition for the signal to be Fourier transformable.

Que 1.3. Find the Fourier transform of the following signal :

- $\delta(t)$
- $\cos(\omega_0 t)$
- $\sin(\omega_0 t)$

Answeri. $\delta(t)$:

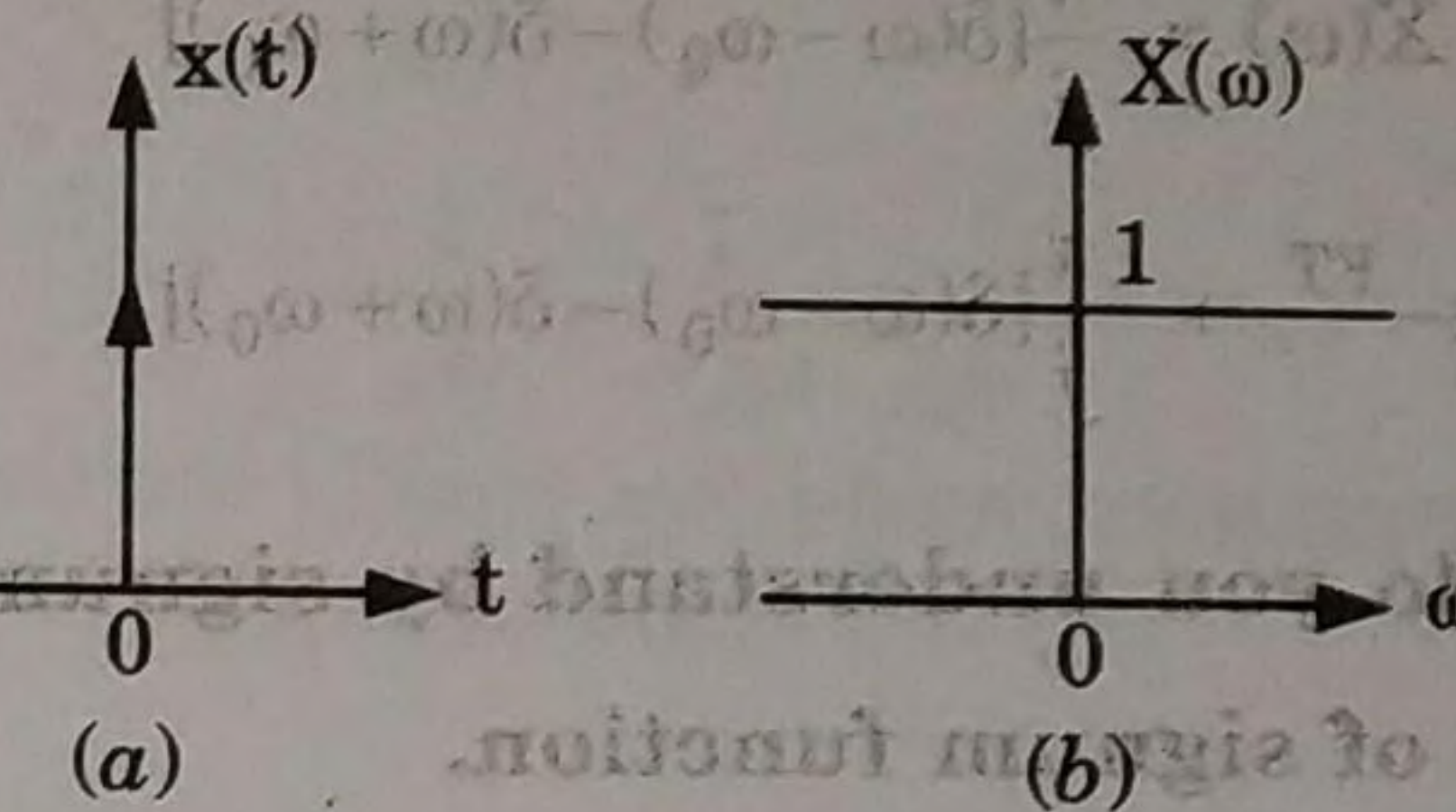
By definition

$$x(t) = \delta(t)$$

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt = \int_{-\infty}^{\infty} \delta(t) e^{-j\omega t} dt = e^{-j\omega t} \Big|_{t=0} = 1$$

$$\delta(t) \longleftrightarrow 1$$

That is, the unit impulse has a Fourier transform consisting of equal contributions at all frequencies.

**Fig. 1.3.1.** (a) Unit impulse function and (b) its Fourier transform.ii. $\cos(\omega_0 t)$:

$$x(t) = \cos(\omega_0 t) = \frac{e^{j\omega_0 t} + e^{-j\omega_0 t}}{2}$$

Taking the Fourier transform, we have

$$F[x(t)] = X(\omega) = F\left[\frac{e^{j\omega_0 t} + e^{-j\omega_0 t}}{2}\right] = \frac{1}{2} (F[e^{j\omega_0 t}] + F[e^{-j\omega_0 t}])$$

$$\therefore e^{j\omega_0 t} \xrightarrow{\text{FT}} 2\pi\delta(\omega - \omega_0) \quad \dots(1.3.1)$$

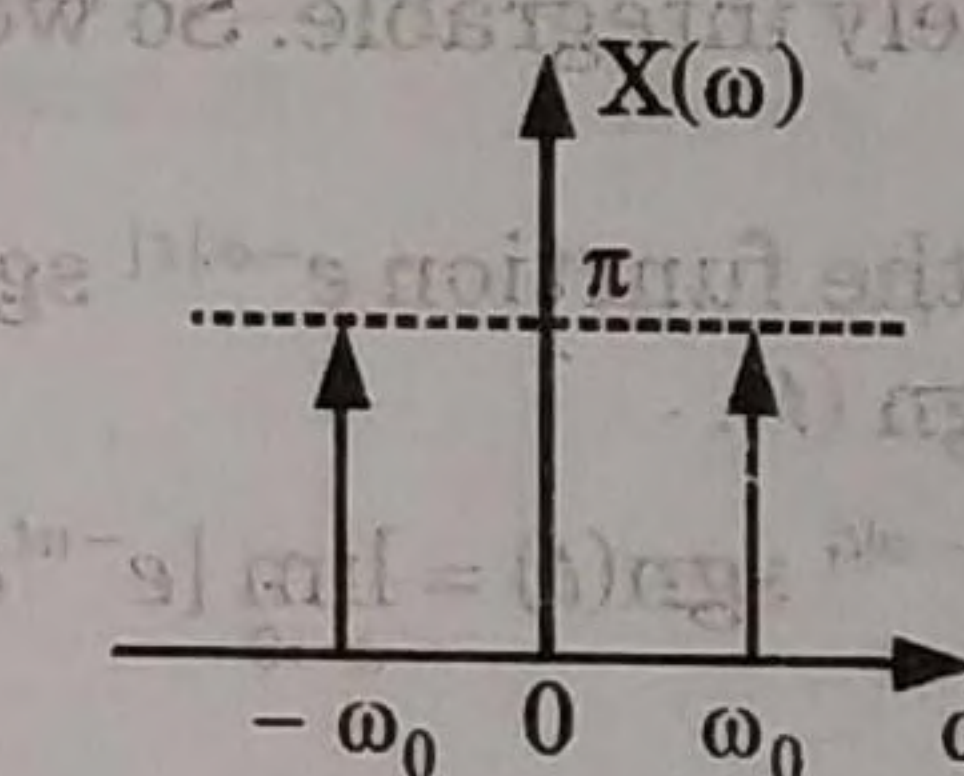
$$e^{-j\omega_0 t} \xrightarrow{\text{FT}} 2\pi\delta(\omega + \omega_0) \quad \dots(1.3.2)$$

$$X(\omega) = \frac{1}{2} [2\pi\delta(\omega - \omega_0) + 2\pi\delta(\omega + \omega_0)]$$

$$X(\omega) = \pi[\delta(\omega - \omega_0) + \delta(\omega + \omega_0)]$$

$$\cos(\omega_0 t) \xrightarrow{\text{FT}} \pi[\delta(\omega - \omega_0) + \delta(\omega + \omega_0)]$$

The Fourier transform of $\cos(\omega_0 t)$ consists of two impulses at ω_0 and $-\omega_0$.

**Fig. 1.3.2.**iii. $\sin(\omega_0 t)$:

$$x(t) = \sin(\omega_0 t) = \frac{e^{j\omega_0 t} - e^{-j\omega_0 t}}{2j}$$

By taking the Fourier transform, we have

$$F[x(t)] = X(\omega) = F\left[\frac{e^{j\omega_0 t} - e^{-j\omega_0 t}}{2j}\right] = \frac{1}{2j} (F[e^{j\omega_0 t}] - F[e^{-j\omega_0 t}])$$

Using eq. (1.3.1) and (1.3.2), we have

$$X(\omega) = \frac{1}{2j} [2\pi\delta(\omega - \omega_0) - 2\pi\delta(\omega + \omega_0)]$$

$$X(\omega) = \frac{\pi}{j} [\delta(\omega - \omega_0) - \delta(\omega + \omega_0)]$$

$$\sin(\omega_0 t) \xrightarrow{\text{FT}} \frac{\pi}{j} [\delta(\omega - \omega_0) - \delta(\omega + \omega_0)]$$

Que 1.4. What do you understand by signum function? Find Fourier transform of signum function.

Answer

Signum function :

1. It is represented by $\text{sgn}(t)$ and is given as :

$$\text{sgn}(t) = \begin{cases} -1 & \text{for } t < 0 \\ 0 & \text{for } t = 0 \\ 1 & \text{for } t > 0 \end{cases}$$

or $\text{sgn}(t) = u(t) - u(-t) = 2u(t) - 1$

2. Graphically signum function can be given as

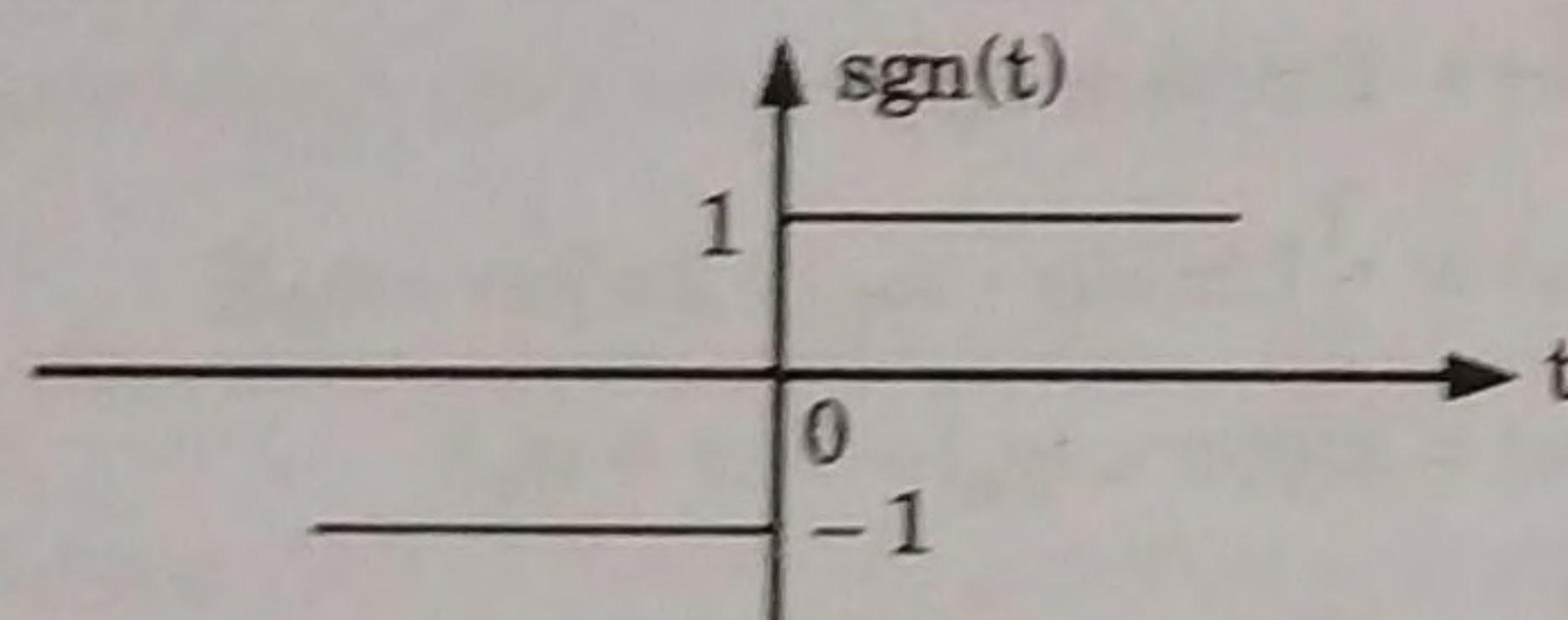


Fig. 1.4.1.

Fourier transform of signum function :

- This function is not absolutely integrable. So we cannot directly find its Fourier transform.
- Therefore, let us consider the function $e^{-a|t|} \text{sgn}(t)$ and substitute the limit $a \rightarrow 0$ to obtain the $\text{sgn}(t)$.
- Given $x(t) = \text{sgn}(t) = \lim_{a \rightarrow 0} e^{-a|t|} \text{sgn}(t) = \lim_{a \rightarrow 0} [e^{-at} u(t) - e^{at} u(-t)]$

$$\therefore X(\omega) = F[\text{sgn}(t)] = \int_{-\infty}^{\infty} \left[\lim_{a \rightarrow 0} [e^{-at} u(t) - e^{at} u(-t)] \right] e^{-j\omega t} dt$$

$$= \lim_{a \rightarrow 0} \left[\int_{-\infty}^{\infty} e^{-at} e^{-j\omega t} u(t) dt - \int_{-\infty}^{\infty} e^{at} e^{-j\omega t} u(-t) dt \right]$$

$$= \lim_{a \rightarrow 0} \left[\int_0^{\infty} e^{-(a+j\omega)t} dt - \int_{-\infty}^0 e^{(a-j\omega)t} dt \right]$$

$$= \lim_{a \rightarrow 0} \left[\int_0^{\infty} e^{-(a+j\omega)t} dt - \int_0^{\infty} e^{-(a-j\omega)t} dt \right]$$

$$= \lim_{a \rightarrow 0} \left[\left[\frac{e^{-(a+j\omega)t}}{-(a+j\omega)} \right]_0^{\infty} - \left[\frac{e^{-(a-j\omega)t}}{-(a-j\omega)} \right]_0^{\infty} \right]$$

$$= \lim_{a \rightarrow 0} \left[\frac{e^{-\infty} - e^0}{-(a+j\omega)} - \frac{e^{-\infty} - e^0}{-(a-j\omega)} \right]$$

$$= \lim_{a \rightarrow 0} \left[\frac{1}{a+j\omega} - \frac{1}{a-j\omega} \right] = \frac{1}{j\omega} - \frac{1}{-j\omega} = \frac{2}{j\omega}$$

4. The amplitude spectrum $|X(\omega)|$ and phase spectrum $\angle X(\omega)$ are shown in Fig. 1.4.2(a) and (b) respectively.

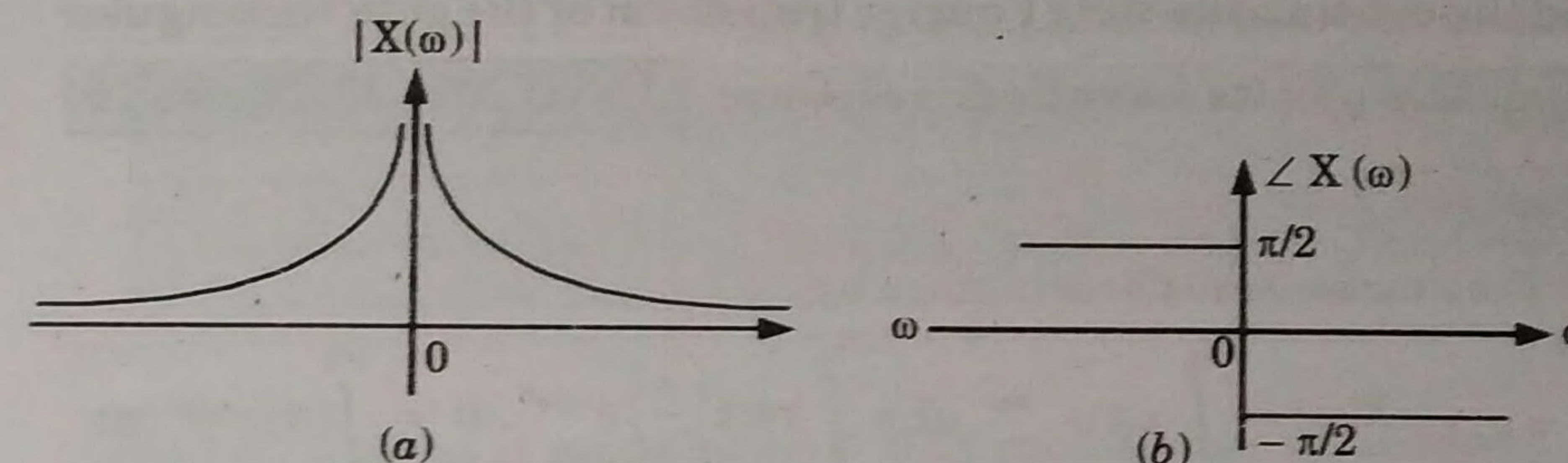


Fig. 1.4.2. (a) Magnitude spectrum (b) Phase spectrum.

Que 1.5. Find the Fourier transform of the following and sketch the magnitude and phase spectrum of $x(t) = e^{-2t} u(t)$.

Answer

1. The Fourier transform

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt = \int_0^{\infty} e^{-2t} e^{-j\omega t} dt = \int_0^{\infty} e^{-(2+j\omega)t} dt$$

$$= -\frac{1}{(2+j\omega)} [e^{-(2+j\omega)t}]_0^{\infty} = \frac{1}{(2+j\omega)}$$

2. Now magnitude and phase are as follows :

$$|X(\omega)| = \frac{1}{\sqrt{4 + \omega^2}}$$

$$\angle X(\omega) = -\tan^{-1} \frac{\omega}{2}$$

3. The amplitude spectrum $|X(j\omega)|$ and phase spectrum $\angle X(j\omega)$ are shown in Fig. 1.5.1 (a) and (b) respectively.

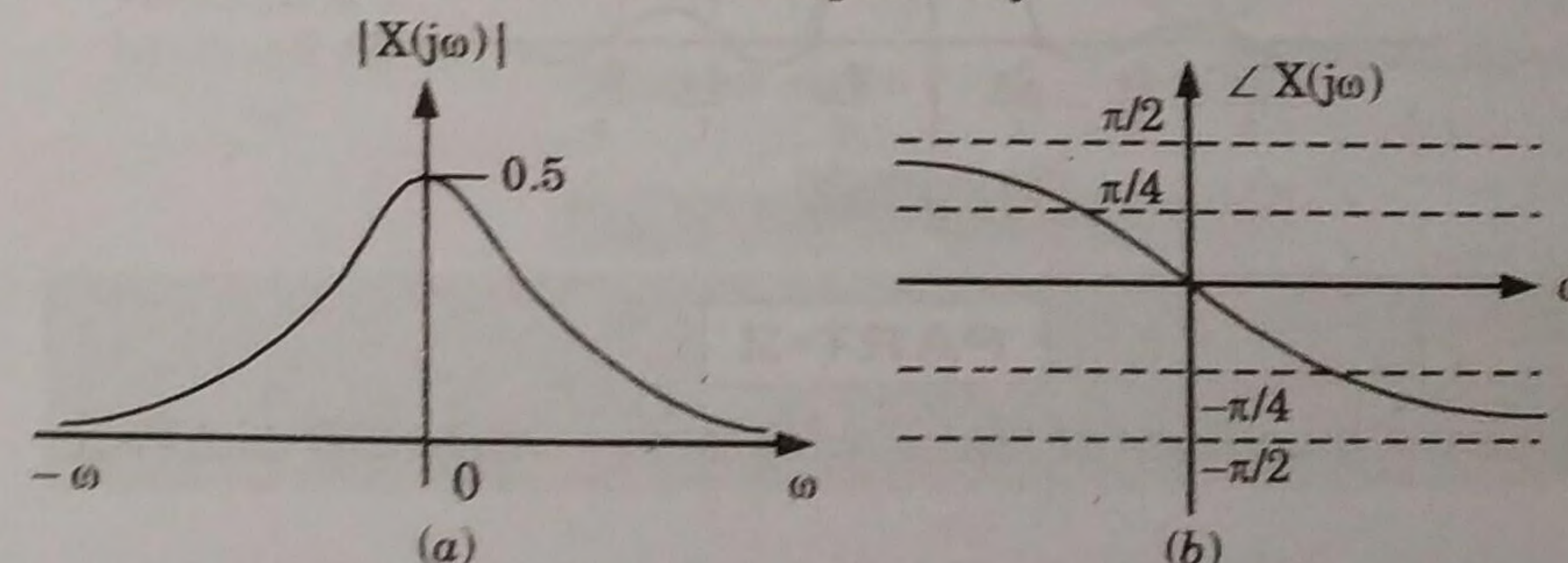


Fig. 1.5.1. (a) Magnitude spectrum (b) Phase spectrum.

Que 1.6. Determine the Fourier transform of the rectangular pulse :

$$x(t) = \text{rect}\left(\frac{t}{\tau}\right) = \begin{cases} 1 & \text{for } -\frac{\tau}{2} < t < \frac{\tau}{2} \\ 0 & \text{otherwise} \end{cases}$$

OR

Find the continuous-time Fourier transform of the gate rectangular signal. Also plot its magnitude response. **AKTU 2015-16, Marks 10**

Answer

1. The Fourier transform is given by,

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt = \int_{-\tau/2}^{\tau/2} \text{rect}\left(\frac{t}{\tau}\right) e^{-j\omega t} dt = \int_{-\tau/2}^{\tau/2} 1 \cdot e^{-j\omega t} dt$$

$$X(\omega) = \left[\frac{e^{-j\omega t}}{-j\omega} \right]_{-\tau/2}^{\tau/2} = \left[\frac{e^{-j\omega \tau/2} - e^{+j\omega \tau/2}}{-j\omega} \right]$$

$$= \left[\frac{-2j \sin\left(\frac{\omega \tau}{2}\right)}{-j\omega} \right] = \frac{2 \sin\left(\frac{\omega \tau}{2}\right)}{\omega} = \frac{\sin\left(\frac{\omega \tau}{2}\right)}{\omega/2} = \tau \cdot \frac{\sin\left(\frac{\omega \tau}{2}\right)}{(\omega \tau/2)}$$

$$X(\omega) = \tau \text{sinc}\left(\frac{\omega \tau}{2}\right)$$

2. Since, $\text{sinc}(x) = 0$ for $x = \pm n\pi$

Therefore, $\text{sinc}(\omega \tau/2) = 0$ for $(\omega \tau/2) = \pm n\pi$

or $\omega = \frac{\pm 2n\pi}{\tau} \quad n = 1, 2, 3, \dots$

or $\omega = \frac{\pm 2\pi}{\tau}, \frac{\pm 4\pi}{\tau}, \frac{\pm 6\pi}{\tau}, \frac{\pm 8\pi}{\tau}, \dots$

3. The plot of the $X(\omega)$ is given in Fig. 1.6.1.

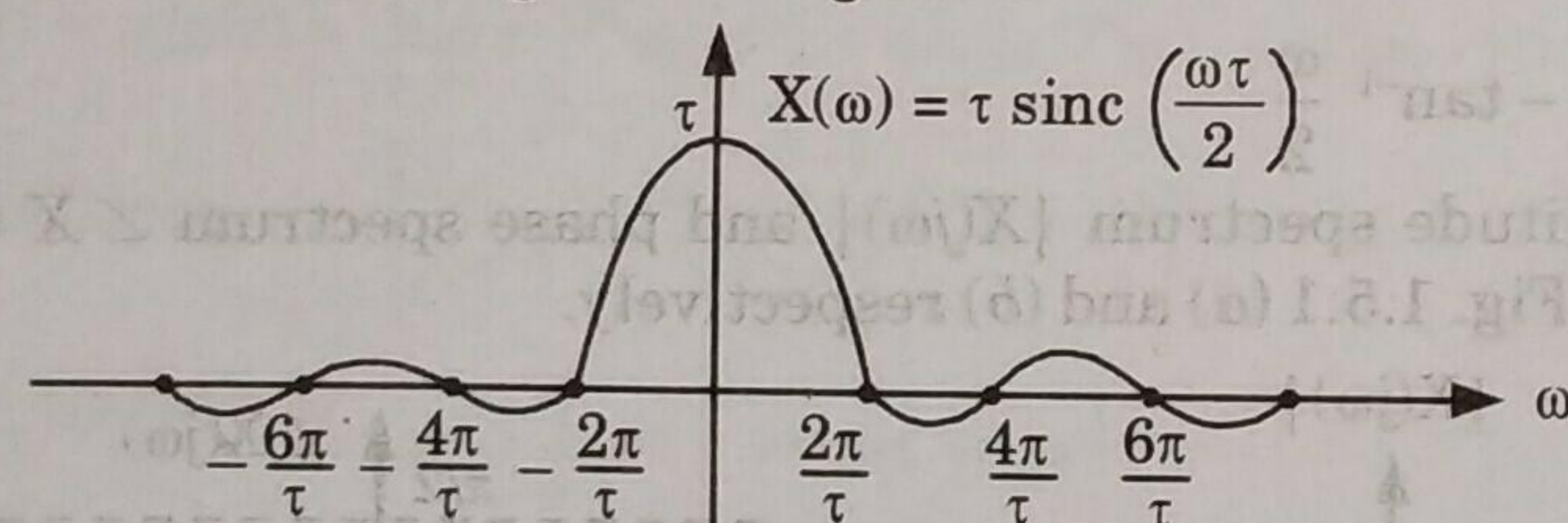


Fig. 1.6.1.

PART-2

Principles of Amplitude Modulation Systems - DSB, SSB and VSB.

CONCEPT OUTLINE

- In amplitude modulation, the amplitude of carrier signal is varied in accordance with the modulating signal.
- The advantage of SSB over DSB-SC is that the required channel bandwidth is reduced by a factor of 1/2.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 1.7. What is modulation ? Why modulation is needed ?

OR

Discuss the basic needs of modulation for the communication over a channel. **AKTU 2014-15, Marks 05**

Answer

Modulation :

1. Modulation is defined as the process by which some characteristic of a signal called carrier is varied in accordance with the instantaneous value of baseband or modulating signal.
2. The signal resulting from the process of modulation is called modulated signal.

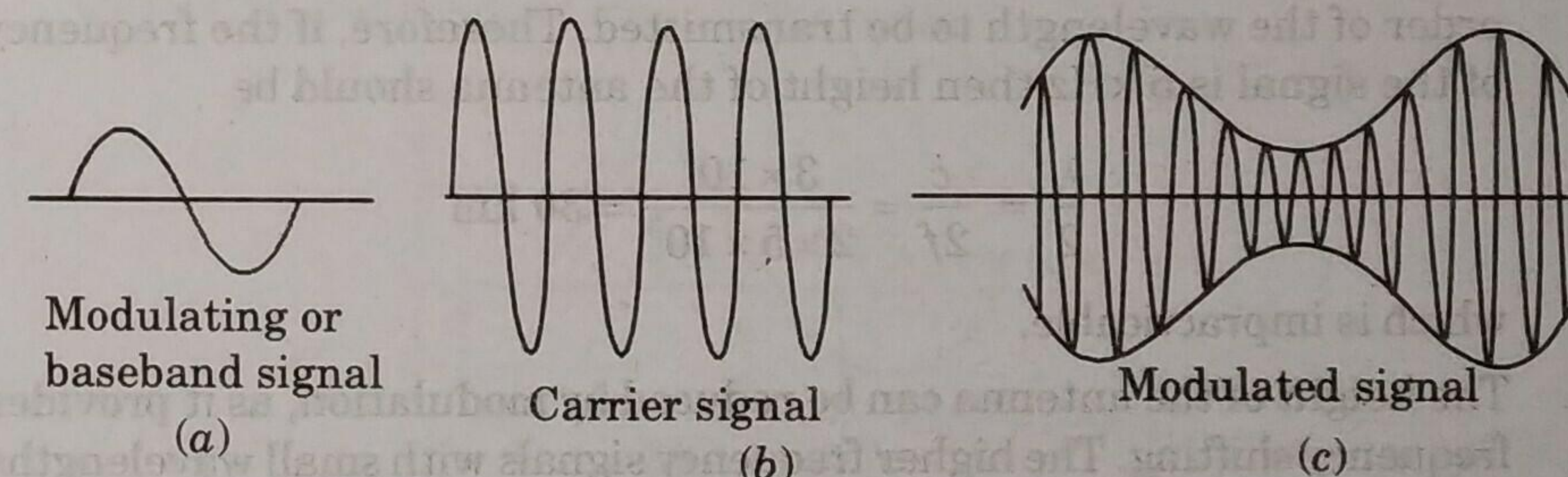


Fig. 1.7.1. Process of modulation.

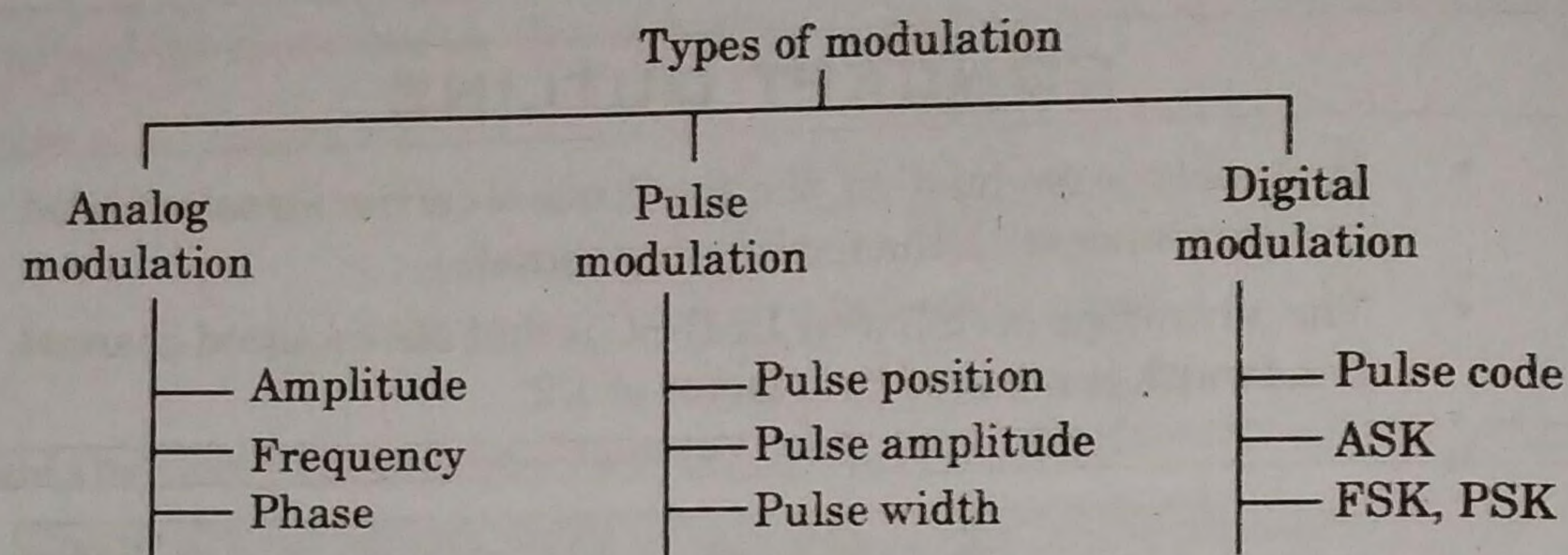


Fig. 1.7.2.

Demodulation :

1. The receiver re-creates the original message signal from a degraded version of transmitted signal after propagation through the channel.
2. This re-creation is accomplished by using a process called demodulation.

Need of modulation :**1. Multiplexing :**

- i. Sending multiple messages over a channel is called multiplexing. If the signals are transmitted without modulation, they will interfere with each other because their baseband (spectrum) are identical.
- ii. However, the signals can be send without interfering using multiplexing techniques, which are of two types frequency division multiplexing and time division multiplexing.

2. Practicability of antenna :

- i. The message signals are sent through an antenna at the transmitter when free space is used as the communication medium.
- ii. For affective transmission the height of the antenna should be of the order of the wavelength to be transmitted. Therefore, if the frequency of the signal is 5 kHz then height of the antenna should be

$$\frac{\lambda}{2} = \frac{c}{2f} = \frac{3 \times 10^8}{2 \times 5 \times 10^3} = 30 \text{ km}$$

which is impracticable.

- iii. The height of the antenna can be reduced by modulation, as it provides frequency shifting. The higher frequency signals with small wavelengths are used for modulation.
- iv. Therefore, if an audio frequency is translated to radio frequency of 1 MHz, then height of the antenna is :

$$\frac{\lambda}{2} = \frac{3 \times 10^8}{2 \times 10^6} = 150 \text{ meter}$$

which is practically achieved.

3. Narrowbanding :

- i. Even if we assume that the height of the antenna is practicable, the audio signals transmitted extend from 50 kHz to 10 kHz, the ratio of highest to lowest wavelength is 200.
- ii. So a single antenna cannot receive all the signal. Now if the audio signals are translated to radio range frequency of 1 MHz, then ratio becomes $\frac{10^6 + 50}{10^6 + 10^4} \approx \frac{1}{1.01}$, which is approximately unity. So a single antenna can receive all the signals.

Que 1.8. Explain the amplitude modulation and deduce equation for this wave. Also derive the expression of modulation index in terms of maximum and minimum voltage.

Answer**Amplitude Modulation :**

1. Amplitude Modulation (AM) is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating voltage keeping frequency of the carrier constant.
2. Baseband or modulating signal is generally low frequency audio signal and carrier is high frequency RF-signal.
3. Let the carrier voltage (v_c) and modulating voltage (v_m) respectively given as :

$$v_c = A_c \cos \omega_c t \quad \dots(1.8.1)$$

$$v_m = A_m \cos \omega_m t \quad \dots(1.8.2)$$

where A_c is the maximum amplitude of carrier wave and A_m is the maximum amplitude of modulating wave.

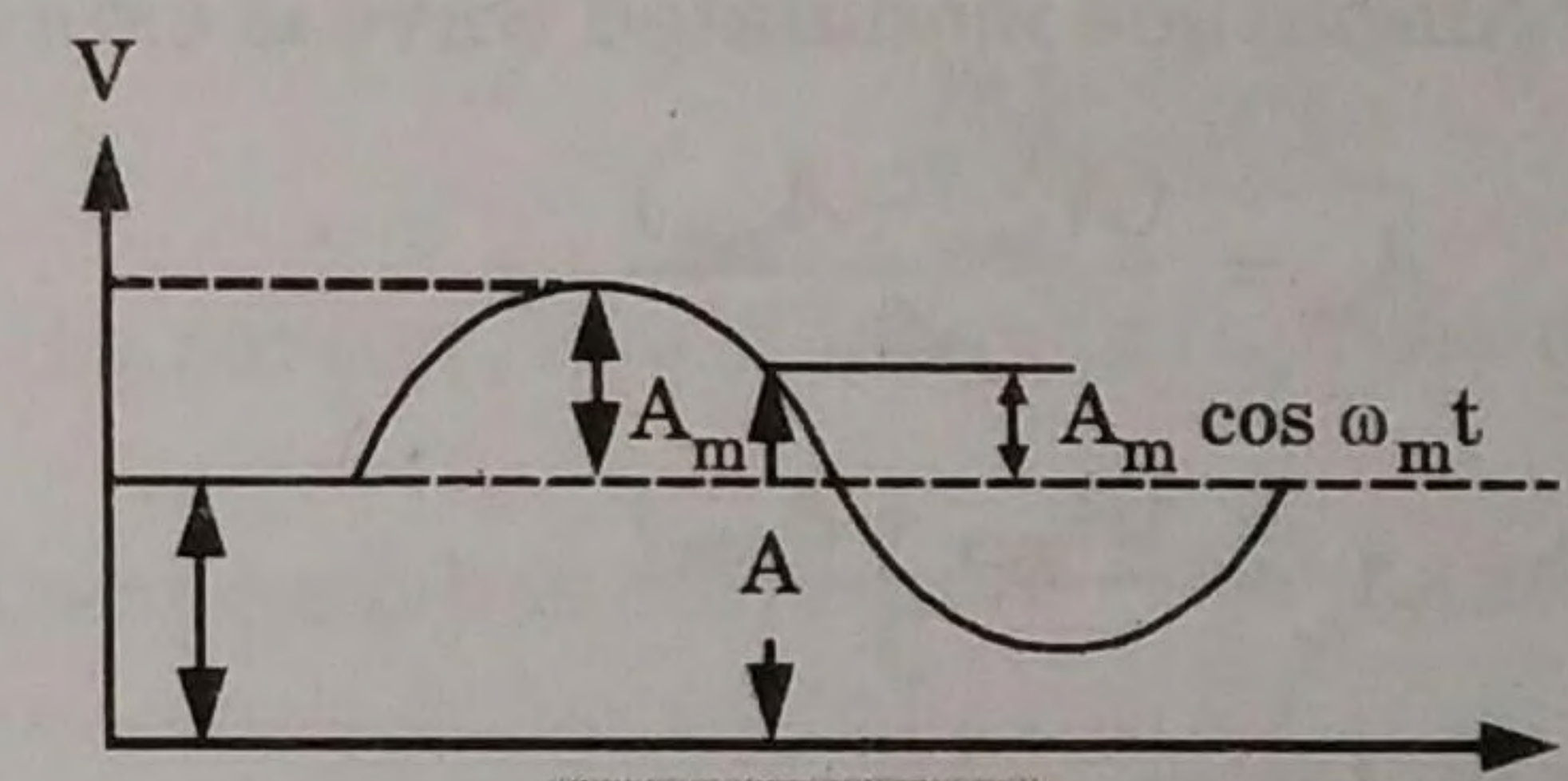


Fig. 1.8.1.

4. From the Fig. 1.8.1, it is clear that amplitude of AM wave A is given as :

$$A = A_c + v_m \quad \dots(1.8.3)$$

From eq. (1.8.2) we have

$$A = A_c + A_m \cos \omega_m t$$

$$A = A_c \left(1 + \frac{A_m}{A_c} \cos \omega_m t \right) \quad \dots(1.8.4)$$

5. The ratio A_m/A_c is known as modulation index (m_a). The modulation index is a number lying between 0 and 1, and it is often expressed as a percentage and called the percentage modulation.

6. So, from eq. (1.8.4) we have amplitude of modulated wave,

$$A = A_c (1 + m_a \cos \omega_m t) \quad \dots(1.8.5)$$

7. So the instantaneous voltage of amplitude modulated wave is given as:

$$V_{AM} = A \cos \omega_c t \quad (\text{since frequency is constant})$$

8. From eq. (1.8.5) we have,

$$V_{AM} = A_c (1 + m_a \cos \omega_m t) \cos \omega_c t \quad \dots(1.8.6)$$

This is the complete equation of AM-wave.

9. Now expanding eq. (1.8.6) we have

$$V_{AM} = A_c \cos \omega_c t + \frac{m_a A_c}{2} (2 \cos \omega_m t \cos \omega_c t)$$

$$V_{AM} = \underbrace{A_c \cos \omega_c t}_{\text{Carrier}} + \underbrace{\frac{m_a A_c}{2} \cos (\omega_c - \omega_m) t}_{\text{LSB}} + \underbrace{\frac{m_a A_c}{2} \cos (\omega_c + \omega_m) t}_{\text{USB}}$$

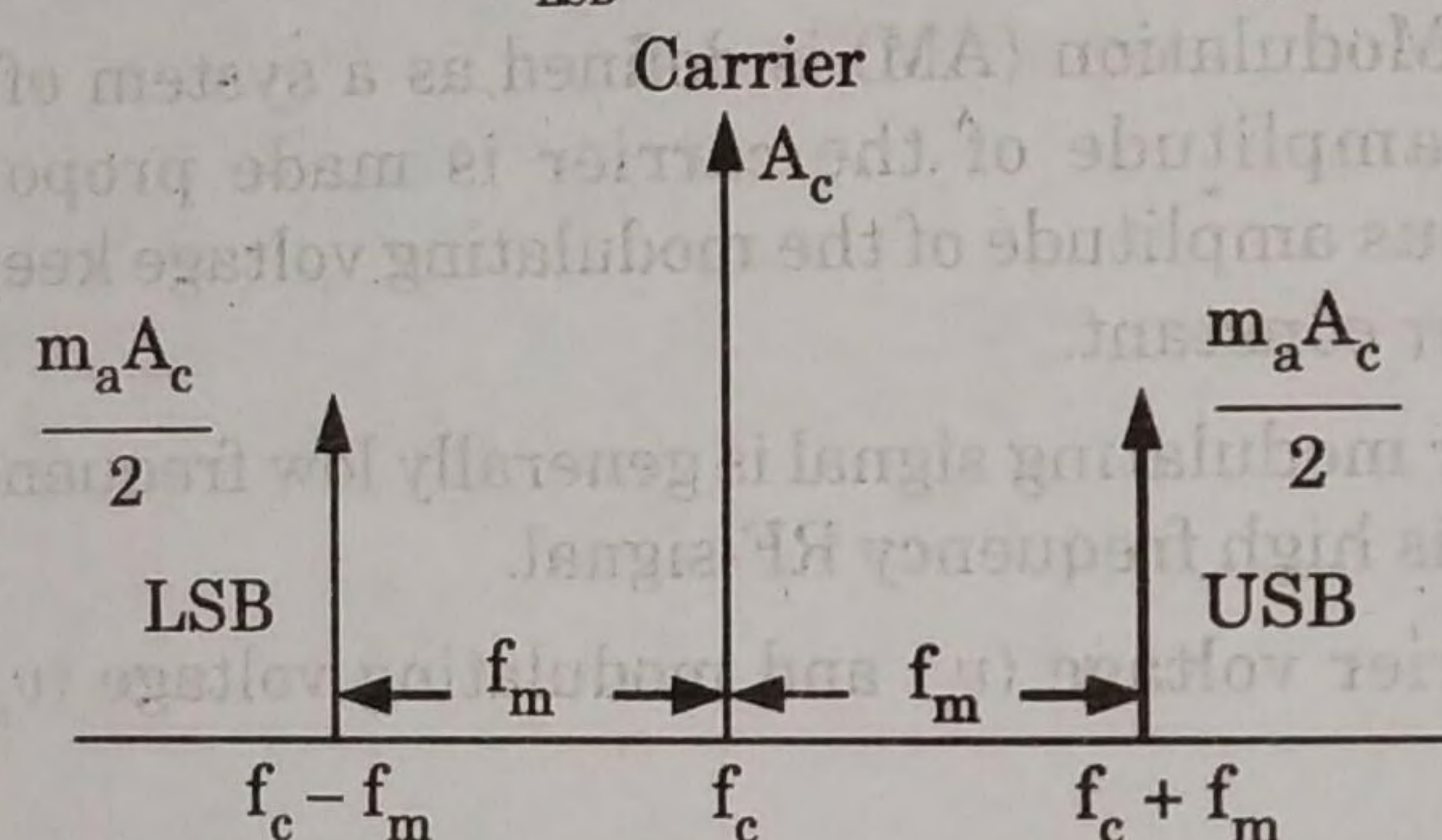


Fig. 1.8.2. Frequency spectrum of AM-wave.

10. Amplitude Modulated wave consists of unmodulated carrier and two sidebands LSB and USB.

11. The envelope of Amplitude modulated wave is expressed in Fig. 1.8.3.

$$A_m = \frac{(A_{\max} - A_{\min})}{2}$$

and

$$A_c = \frac{(A_{\max} + A_{\min})}{2}$$

12. We know that modulation index,

$$m_a = \frac{A_m}{A_c} = \frac{(A_{\max} - A_{\min})}{(A_{\max} + A_{\min})}$$

$$m_a = \frac{(A_{\max} - A_{\min})}{(A_{\max} + A_{\min})}$$

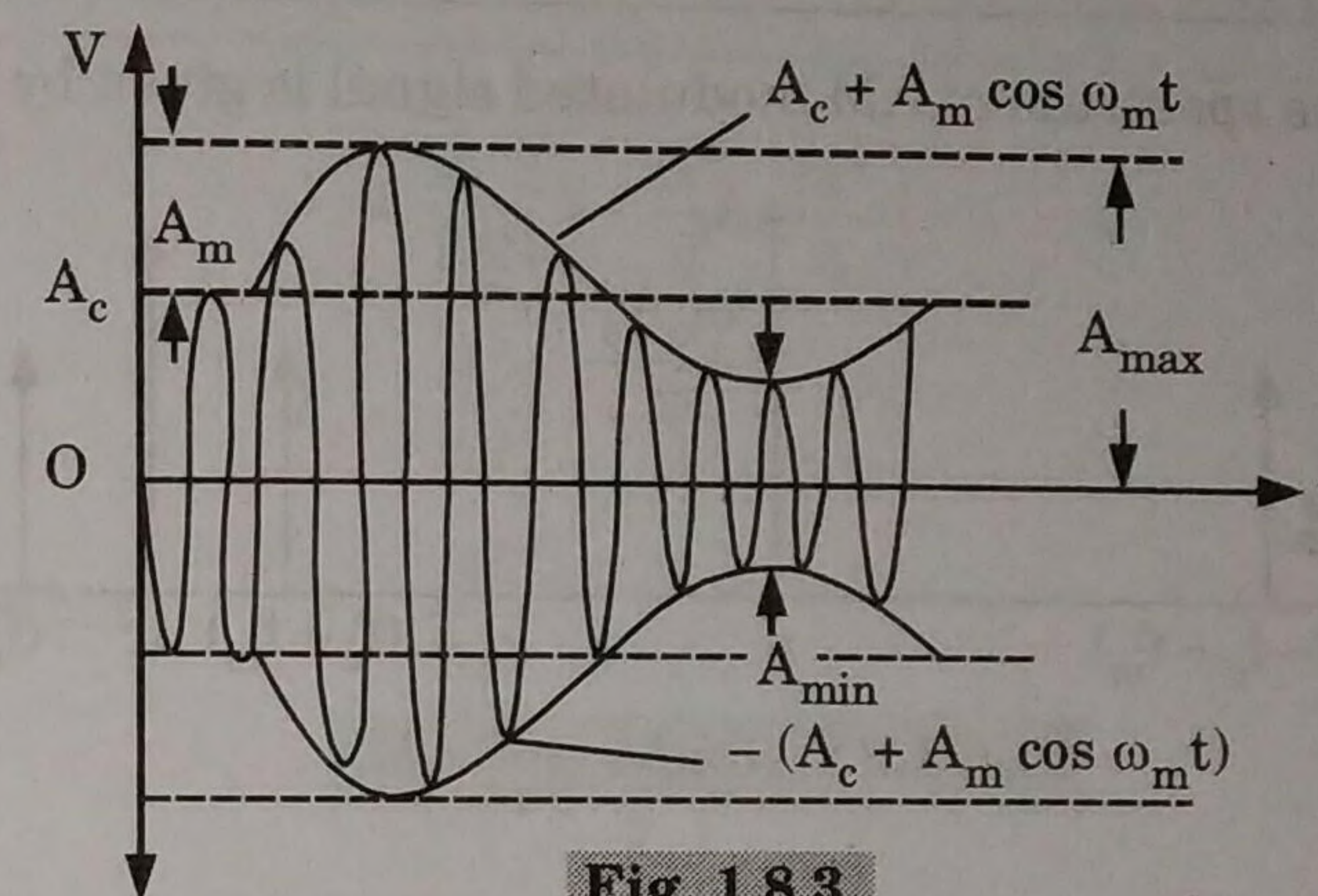


Fig. 1.8.3.

Que 1.9. Draw the phasor diagram of AM modulated signal and also draw the spectrum of AM modulated signals.

AKTU 2014-15, Marks 05

Answer

Phasor diagram of AM modulated signal :

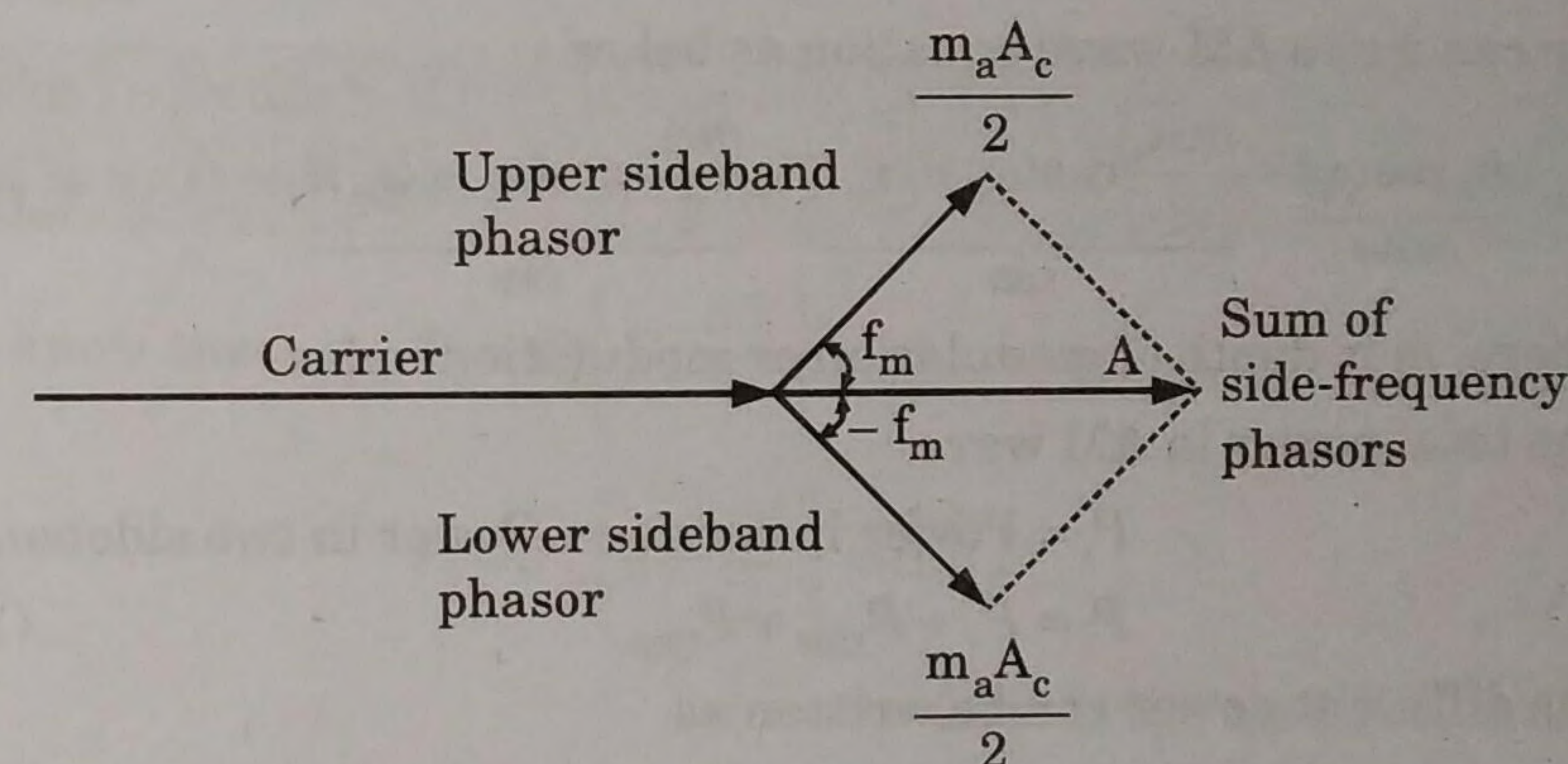


Fig. 1.9.1.

1. Here, there is a carrier, two sidebands represented by upper and lower sideband phasor each has amplitude $\frac{m_a A_c}{2}$, and resultant A shows the AM signal.

Spectrum of AM modulated signal :

1. The AM signal is given as

$$V_{AM} = A_c \cos (2\pi f_c t) + \frac{m_a A_c}{2} \cos 2\pi (f_c + f_m) t + \frac{m_a A_c}{2} \cos 2\pi (f_c - f_m) t$$

2. Hence, the spectrum of AM modulated signal is given by

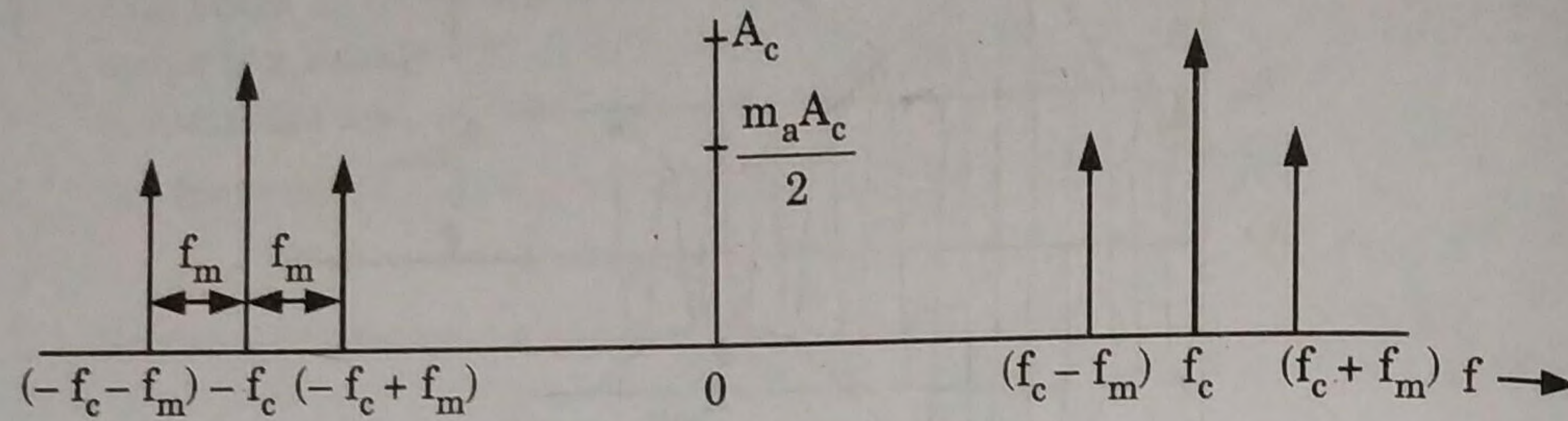


Fig. 1.9.2. Single tone AM.

Que 1.10. Derive the relation between the output power of an AM transmitter and the depth of modulation.

OR

Derive the power calculation of AM signal.

AKTU 2016-17, Marks 10

Answer

1. We can write AM-wave equation as below :

$$v = \underbrace{A_c \cos \omega_c t}_{\text{Carrier}} + \underbrace{\frac{mA_c}{2} \cos(\omega_c - \omega_m)t}_{\text{LSB}} + \underbrace{\frac{mA_c}{2} \cos(\omega_c + \omega_m)t}_{\text{USB}} \quad \dots(1.10.1)$$

where, m is depth of modulation or modulation index.

2. The total power in AM wave

$$P_t = \text{Power in carrier} + \text{Power in two sidebands}$$

$$P_t = P_c + P_{\text{LSB}} + P_{\text{USB}} \quad \dots(1.10.2)$$

3. The different power can be written as

$$P_c = \frac{\left(\frac{A_c}{\sqrt{2}}\right)^2}{R} = \frac{A_c^2}{2R} \quad \dots(1.10.3)$$

$$P_{\text{LSB}} = P_{\text{USB}} = \frac{A_{\text{SB}}^2}{R} = \frac{\left(\frac{mA_c}{2\sqrt{2}}\right)^2}{R} = \frac{m^2 A_c^2}{8R}$$

$$P_{\text{LSB}} = P_{\text{USB}} = \frac{m^2}{4} \left(\frac{A_c^2}{2R}\right) \quad \dots(1.10.4)$$

where R is resistance.

4. Now from eq. (1.10.2), (1.10.3) and (1.10.4)

$$P_t = \left(\frac{A_c^2}{2R}\right) + \frac{m^2}{4} \left(\frac{A_c^2}{2R}\right) + \frac{m^2}{4} \left(\frac{A_c^2}{2R}\right)$$

$$P_t = \frac{A_c^2}{2R} \left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right)$$

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

5. We know that $P_c = \frac{A_c^2}{2R}$

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

where P_c is unmodulated carrier power.

Que 1.11. Calculate the percentage power saving when the carrier and one of the sidebands are suppressed in AM wave modulated to a depth of

i. 100 %

ii. 75 %.

AKTU 2017-18, Marks 10

Answer

Given : i. Depth = 100 %; ii. Depth = 75 %.

To Find : Percentage power saving.

i. We know that, $P_t = P_c \left[1 + \frac{m^2}{2}\right]$

$$P_t = P_c \left[1 + \frac{(1)^2}{2}\right] = 1.5 P_c$$

($\because m = 1$ for 100 % modulation depth)

Power for single sideband only is :

$$P_{\text{SB}} = P_c \frac{m^2}{4} = P_c \times \frac{(1)^2}{4} = 0.25 P_c$$

$$\text{Power saving} = \frac{1.5P_c - 0.25P_c}{1.5P_c} = \frac{1.25}{1.5} = 0.833 = 83.3\%$$

ii.

$$P_t = P_c \left[1 + \frac{(0.75)^2}{2}\right]$$

$$P_t = P_c \left[1 + \frac{0.5625}{2}\right] = 1.28 P_c$$

$$P_{\text{SB}} = P_c \frac{m^2}{4} = P_c \frac{(0.75)^2}{4} = 0.140 P_c$$

$$\text{Power saving} = \frac{1.28P_c - 0.140P_c}{1.28P_c} = 0.890 = 89\%$$

Thus power saving increases as depth of modulation is reduced.

Que 1.12. The efficiency η of ordinary AM is defined as the percentage of the total power carried by the sidebands, that is $\eta = P_s/P_t$, where P_s is the power carried by the sidebands and P_t is the total power of the AM signal.

- Find η for $m_a = 0.5$ (50 % modulation)
- Show that for a single tone AM η max is 33 % at $m_a = 1$.

AKTU 2017-18, Marks 10

Answer

Given : $m_a = 0.5 = 50\%$

To Find : η for $m_a = 0.5 = 50\%$

To Show : $\eta_{\max} = 33\%$ at $m_a = 1$

- We know that,

$$\eta = \frac{P_s}{P_t}$$

- Power of sidebands, $P_s = P_c \frac{m_a^2}{2}$

- Total power, $P_t = P_c \left[1 + \frac{m_a^2}{2} \right]$

where, $m_a = \text{Modulation index}$

- Efficiency,
$$\eta = \frac{P_s}{P_t} = \frac{P_c \frac{m_a^2}{2}}{P_c \left[1 + \frac{m_a^2}{2} \right]} = \frac{m_a^2}{2 + m_a^2}$$

$$= \frac{(0.5)^2}{(0.5)^2 + 2} = 0.111 = 11.1\%$$

- For a single tone AM, η at $m_a = 1$, is given by,

$$\eta = \frac{m_a^2}{2 + m_a^2} = \frac{1}{2 + 1} \times 100 = 33.3\%$$

Que 1.13. An amplitude modulated signal is given by

$$\Psi_{AM}(t) = 10 \cos(2\pi \times 10^6 t) + 5 \cos(2\pi \times 10^6 t) \cos(2\pi \times 10^3 t) + 2 \cos(2\pi \times 10^6 t) \cos(4\pi \times 10^3 t) \text{ volts.}$$

- Find various frequency components present and the corresponding modulation indices.
- Calculate power contained in the signal and efficiency of the signal.

AKTU 2014-15, Marks 05

Answer

- We have,

$$\begin{aligned} \Psi_{AM}(t) &= 10 \cos(2\pi \times 10^6 t) + 5 \cos(2\pi \times 10^6 t) \cos(2\pi \times 10^3 t) \\ &\quad + 2 \cos(2\pi \times 10^6 t) \cos(4\pi \times 10^3 t) \text{ volts} \\ &= [10 + 5 \cos(2\pi \times 10^3 t) + 2 \cos(4\pi \times 10^3 t)] \cos(2\pi \times 10^6 t) \\ &= 10 [1 + 1/2 \cos(2\pi \times 10^3 t) + 1/5 \cos(4\pi \times 10^3 t)] \cos(2\pi \times 10^6 t) \end{aligned}$$

- On comparing with

$$\Psi_{AM} = A_c [1 + m_{a1} \cos(2\pi f_1 t) + m_{a2} \cos(2\pi f_2 t)] \cos(2\pi f_c t)$$

Here carrier frequency = 10^6 Hz = 1 MHz and modulating signal are of two frequencies $f_1 = 10^3$ Hz, $f_2 = 2 \times 10^3$ Hz.

- Modulating indices are $m_{a1} = 1/2$ and $m_{a2} = 1/5$ and overall modulation index can be found by,

$$m_a = \sqrt{m_{a1}^2 + m_{a2}^2} = \sqrt{\left(\frac{1}{2}\right)^2 + \left(\frac{1}{5}\right)^2} = 0.54$$

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

$$= \left(1 + \frac{(0.54)^2}{2}\right) \frac{100}{2} = 57.29 \quad \left(\because P_c = \frac{A_c^2}{2}\right)$$

- Efficiency,
$$\eta = \frac{m_a^2}{2 + m_a^2}$$

$$= \frac{(0.54)^2}{2 + (0.54)^2} = 0.1272 = 12.72\%$$

Que 1.14. Explain DSB-C modulator and DSB-C demodulator.

Answer

DSB-C Modulator :

- A DSB-C modulator can be generated by using a mixer that generates the sidebands as well as carrier signal and the baseband message signal.
- For DSB-C modulator only the baseband message is to be filtered out. Let us consider a simple switching modulator circuits with only one diode.

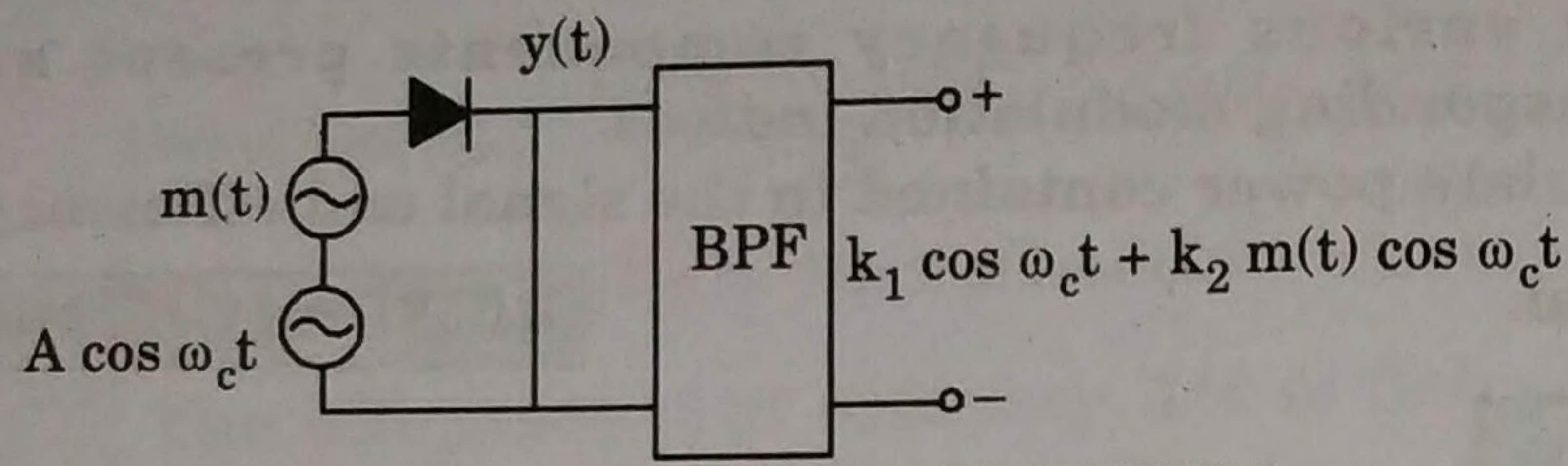


Fig. 1.14.1. An AM DSB-C modulator.

- The BPF passes frequency $\omega_c \pm \omega_m$ where ω_m is maximum frequency of message signal, carrier is represented by $A \cos \omega_c t$.
 - The switching action can be approximated by a pulse train as
- $$s(t) = \frac{1}{2} + \frac{2}{\pi} \left[\cos \omega_c t - \frac{1}{3} \cos 3\omega_c t + \frac{1}{5} \cos 5\omega_c t - \frac{1}{7} \cos 7\omega_c t + \dots \right] \quad \dots(1.14.1)$$
- Now the combined signal message $m(t) + A \cos \omega_c t$ will appear at the output when the diode is switched ON otherwise not.

$$y(t) = [m(t) + A \cos \omega_c t] s(t)$$

$$= [m(t) + A \cos \omega_c t] \times$$

$$\left[\frac{1}{2} + \frac{2}{\pi} \left(\cos \omega_c t - \frac{1}{3} \cos 3\omega_c t + \frac{1}{5} \cos 5\omega_c t - \frac{1}{7} \cos 7\omega_c t + \dots \right) \right]$$

$$y(t) = \frac{m(t)}{2} + \frac{A}{2} \cos \omega_c t + \frac{2}{\pi} m(t) \cos \omega_c t + \frac{2A}{\pi} \cos \omega_c^2 t + \dots$$

DSB-C Demodulator :

- Envelope Detector :** This detector uses linear characteristics of diode.

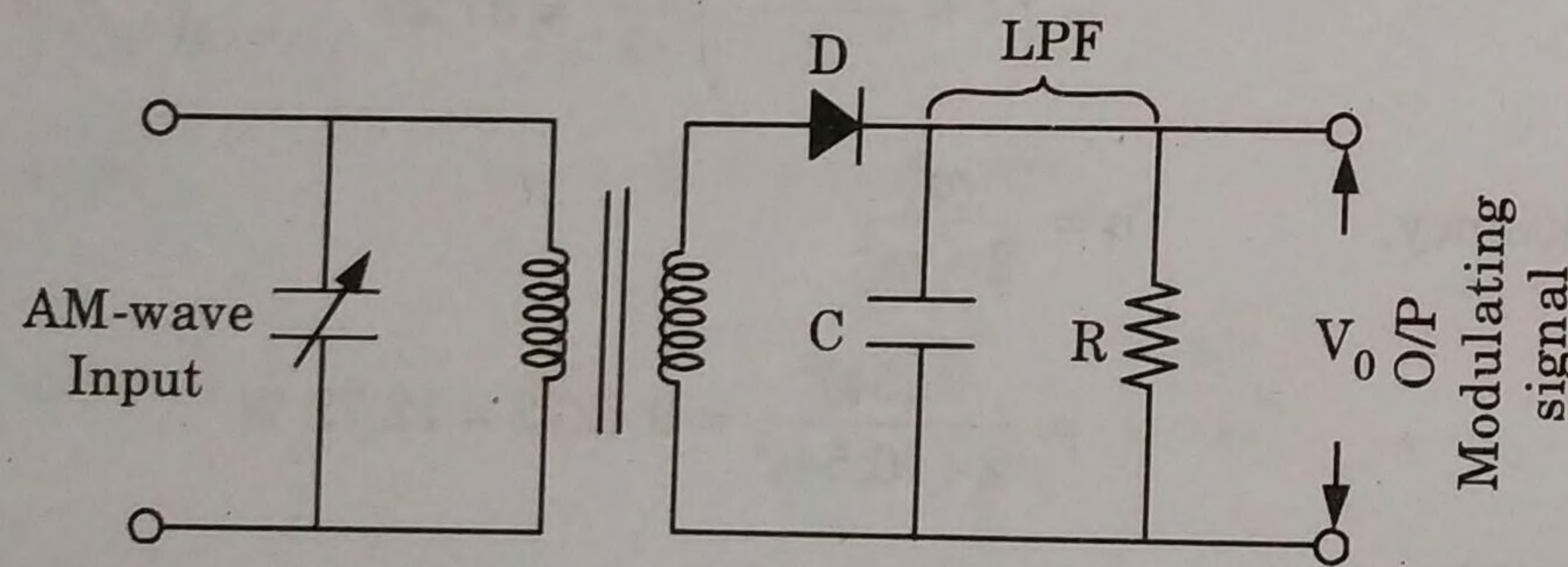


Fig. 1.14.2.

In the circuit, C is a small capacitance and R is large resistance.

- The parallel combination of R and C is the load across which the rectified output voltage V_0 is developed.
- At each positive peak of the RF cycle, C charges up to a potential almost equal to the peak signal voltage.
- The output voltage developed across R is replica of modulating signal with some ripples.

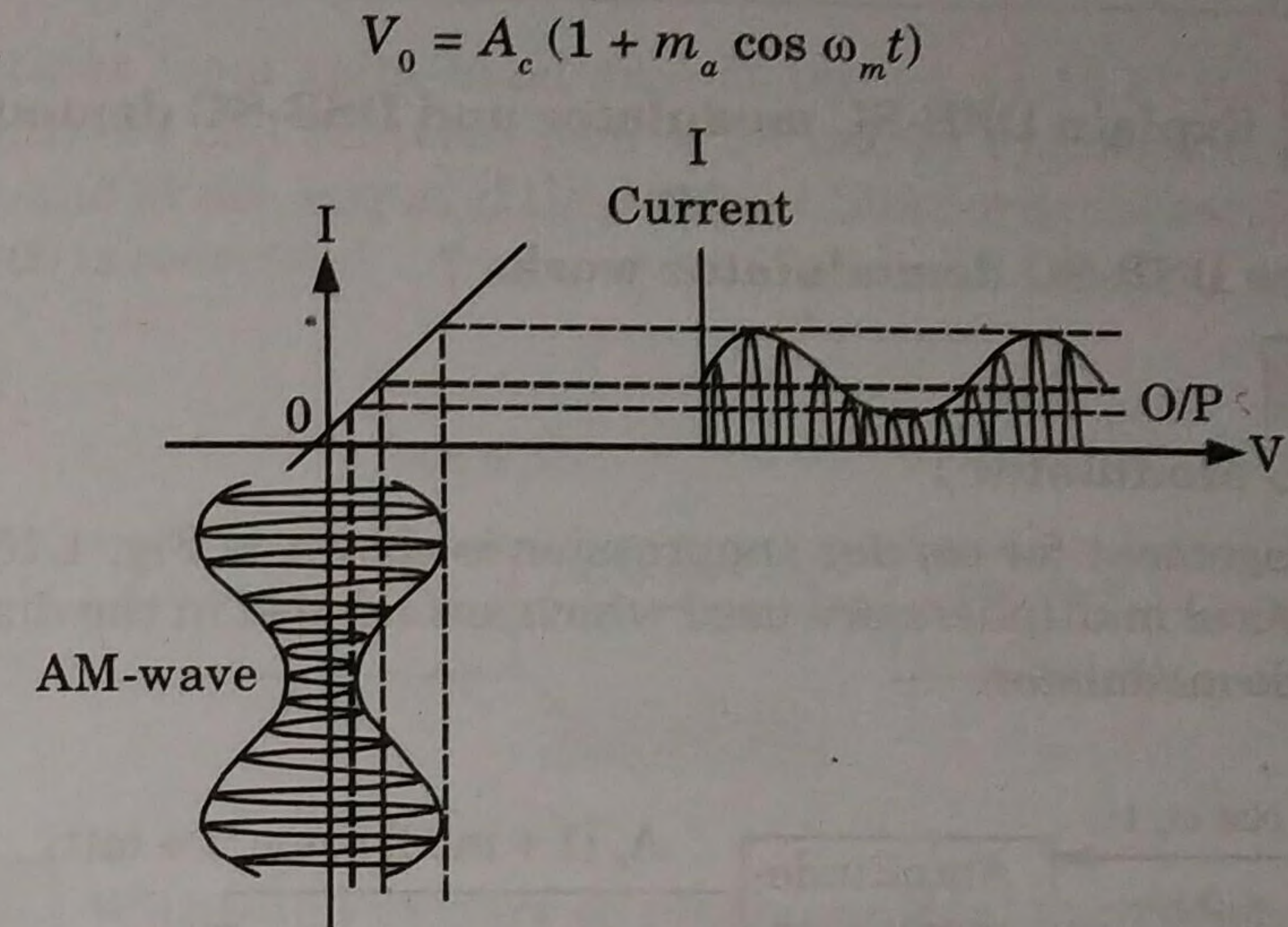


Fig. 1.14.3. Waveform representation of detection of AM wave.

- The capacitor voltage at any instant t is given as,

$$e_{cap} = e_0 \cdot e^{-\frac{(t-t_0)}{RC}}$$

$$-\frac{de_{cap}}{dt} = +\frac{e_0}{RC} \cdot e^{-\frac{(t-t_0)}{RC}} \quad [\text{where } e_0 = V_0]$$

$$\left(-\frac{de_{cap}}{dt} \right)_{t=t_0} = \frac{e_0}{RC} = \frac{A_c(1+m_a \cos \omega_m t)}{RC}$$

[For satisfactory detection]

$$-\left(\frac{de_{cap}}{dt} \right)_{t=t_0} \geq -\frac{de_0}{dt} \quad [\text{For satisfactory detection}]$$

$$\frac{A_c(1+m_a \cos \omega_m t)}{RC} \geq A_c m_a \omega_m \sin \omega_m t$$

$$\frac{1}{RC} \geq \left(\frac{m_a \omega_m \sin \omega_m t}{1+m_a \cos \omega_m t} \right)$$

- For maxima, $\cos \omega_m t = -m_a$ and $\sin \omega_m t = \sqrt{1-m_a^2}$

$$\frac{1}{RC} \geq \frac{m_a \omega_m}{\sqrt{1-m_a^2}}$$

If $m_a \ll 1$ then,

$$\frac{1}{RC} \geq m_a \omega_m$$

- This is the condition for satisfactory detection of AM wave.

Que 1.15. Explain DSB-SC modulator and DSB-SC demodulator.

OR

Explain how DSB-SC demodulator works ?

Answer

DSB-SC Modulator :

1. An arrangement for carrier suppression is shown in Fig. 1.15.1. Here two physical multipliers are used which are labeled in the diagram as amplitude modulator.

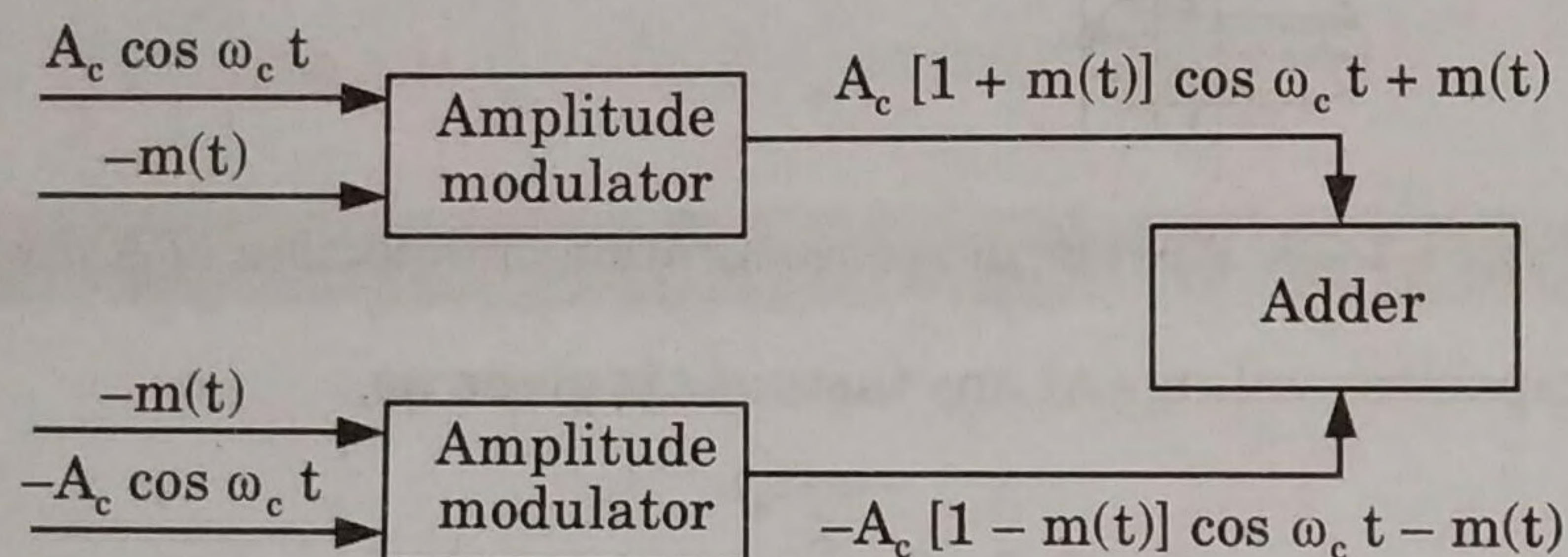


Fig. 1.15.1. The outputs of two amplitude modulators are combined to produce a double sideband suppressed carrier output.

2. The carrier inputs to the two modulators are of reverse polarity as are the modulating signals.
3. The modulator outputs are added with consequent suppression of the carrier.
4. The product terms of the two modulators will be reinforced. The arrangement of Fig. 1.15.1 is called a balanced modulator.

DSB-SC Demodulator :

1. A method for detecting the AM-SC signal shown in Fig. 1.15.2 is used at the receiver end for recovery of the message signal.
2. The method of retranslation is similar to that of translation. Here modulated signal $m(t) \cos \omega_c t$ is multiplied with $\cos \omega_c t$ (locally generated carrier), and passed through a low pass filter.
3. The signal $m(t) \cos \omega_c t$ when multiplied with $\cos \omega_c t$, yields

$$\begin{aligned} m(t) \cos^2 \omega_c t &= \frac{1}{2} m(t) [1 + \cos 2\omega_c t] \\ &= \frac{1}{2} m(t) + \frac{1}{2} m(t) \cos 2\omega_c t \end{aligned} \quad \dots(1.15.1)$$

4. It is clear from right-hand side of the eq. (1.15.1) that the term $\frac{1}{2} m(t) \cos 2\omega_c t$ centered near $\pm 2\omega_c$ can be bypassed by a low pass filter, and at the output of the low pass filter original baseband signal $\frac{1}{2} m(t)$ is recovered.

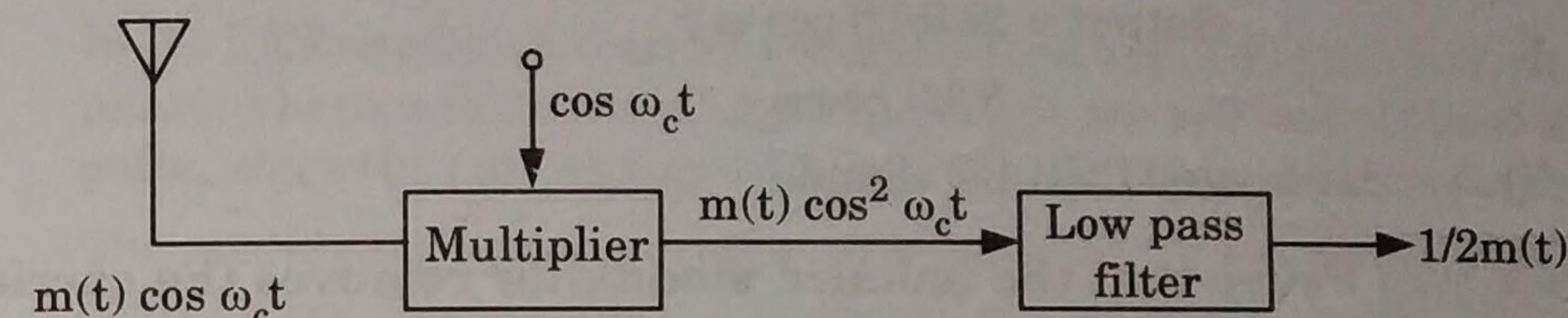


Fig. 1.15.2.

Que 1.16. With the support of mathematical expressions explain the working of balanced modulator.

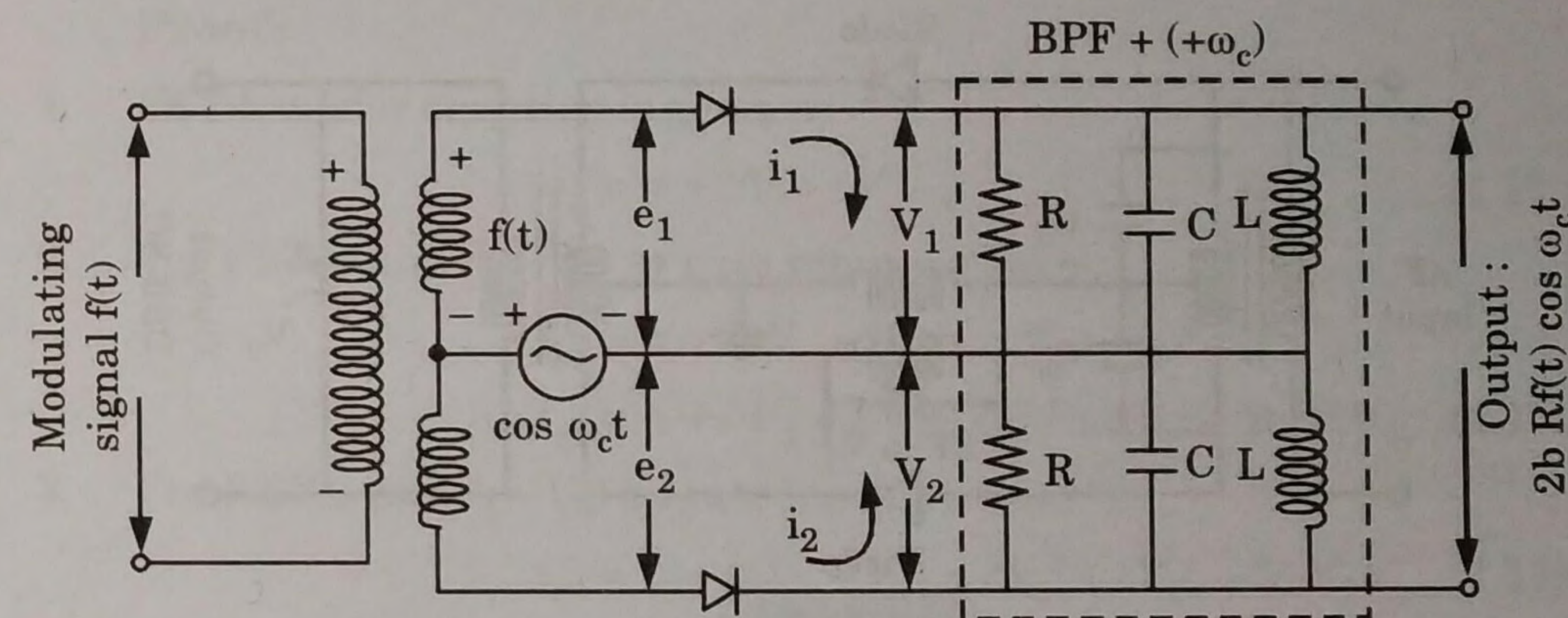


Fig. 1.16.1.

Answer

1. In a balanced modulator, two non-linear devices are connected in the balanced mode, so as to suppress the carrier wave.
2. In the Fig. 1.16.1, voltages e_1 and e_2 are given as :

$$e_1 = \cos \omega_c t + f(t)$$

$$e_2 = \cos \omega_c t - f(t)$$

3. Currents i_1 and i_2 are :

$$\begin{aligned} i_1 &= ae_1 + be_1^2 \\ &= a[\cos \omega_c t + f(t)] + b[\cos \omega_c t + f(t)]^2 \\ i_2 &= ae_2 + be_2^2 \\ &= a[\cos \omega_c t - f(t)] + b[\cos \omega_c t - f(t)]^2 \end{aligned}$$

4. The voltage at the output of the bandpass filter (BPF) is given by :

$$v_o = v_1 - v_2 = i_1 R - i_2 R$$

$$v_o = 2R[af(t) + 2bf(t) \cos \omega_c t]$$

5. The output of BPF centered around $\pm \omega_c$ is given by :

$$\text{Output} = 2bRf(t) \cos \omega_c t$$

$$= Kf(t) \cos \omega_c t$$

This is the desired DSB-SC signal.

Que 1.17. Prove that the balance modulator removes the carrier and gives the two sidebands only.

Answer

Balance modulator :

1. Using diode :

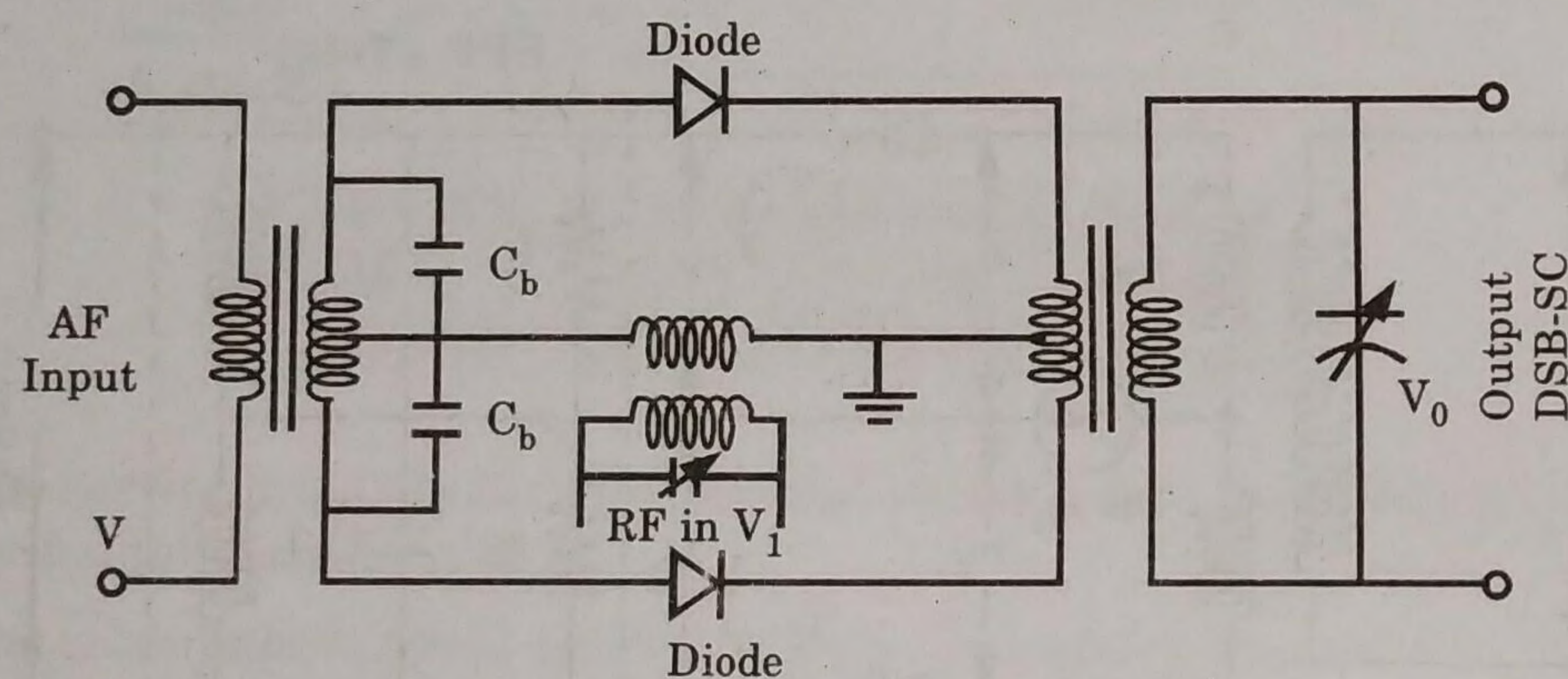


Fig. 1.17.1.

2. Using FET :

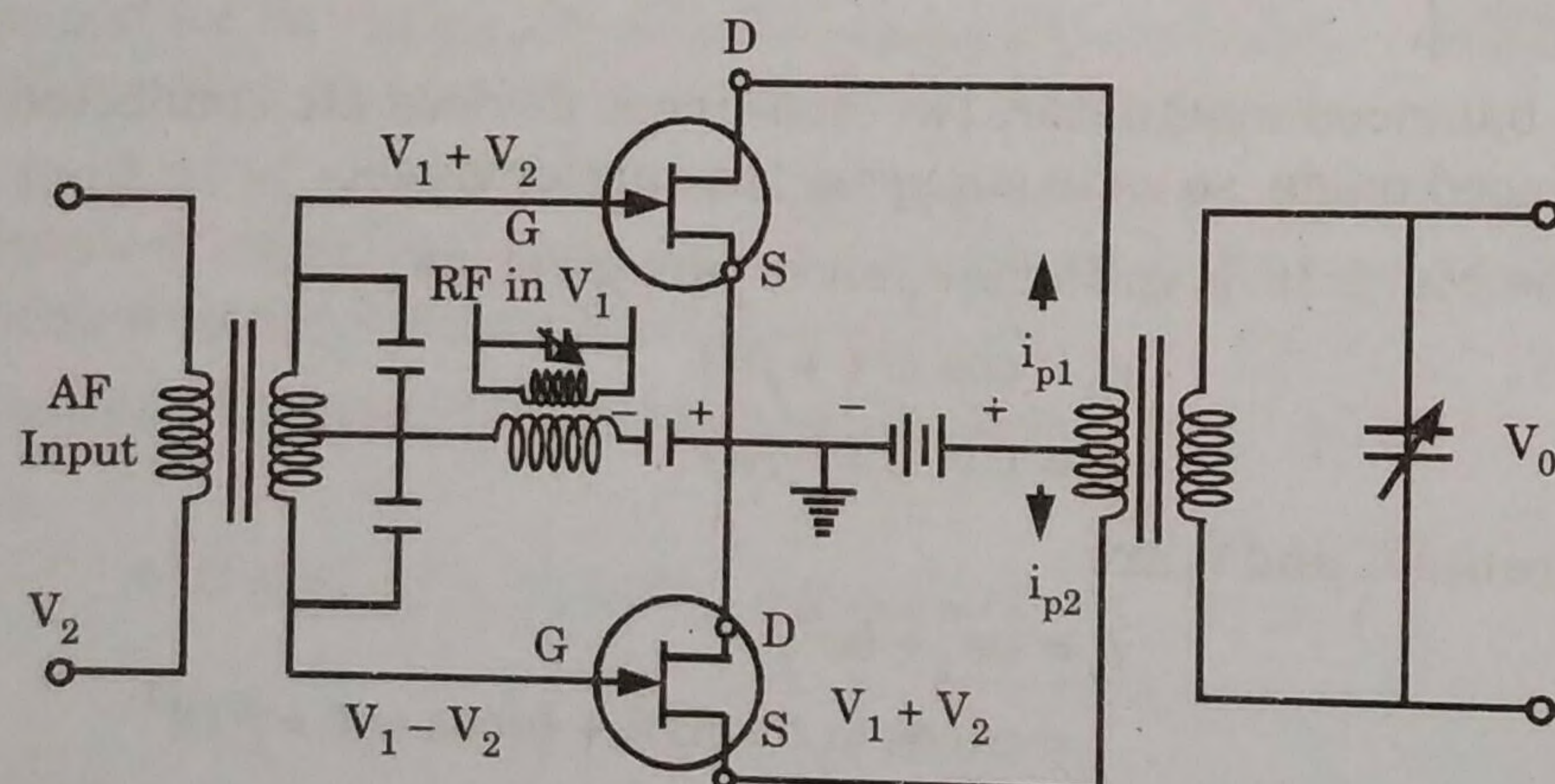


Fig. 1.17.2. Circuit diagram of balance modulator using diode and FET.

- Two circuits of the balanced modulator are shown in Fig. 1.17.1 and Fig. 1.17.2. Each utilizes the non-linearity principle.
- The modulating voltage V_2 is fed in push-pull and the carrier voltage V_1 in parallel, to a pair of identical diodes or class A (transistor or FET) amplifiers.
- In the FET circuit, the carrier voltage is thus applied to the two gates in phase; whereas the modulating voltage appears 180° out of phase at the gates, since they are at opposite ends of a center-tapped transformer.
- The modulated output currents of the two FETs are combined in the center tapped primary of the push-pull output transformer.
- If this system is made completely symmetrical, the carrier frequency will be completely cancelled.
- It is not possible to have completely symmetrical system in practical, so only carriers are suppressed and the generated output is known as Double Side Band-Suppressed Carrier (DSB-SC).

Proof :

1. FET transfer equation is given as

$$i_d = a + bV + cV^2 \quad \dots(1.17.1)$$

2. For FET with $(V_1 + V_2)$ as gate input we have

$$i_{d1} = a + b(V_1 + V_2) + c(V_1 + V_2)^2$$

$$i_{d1} = a + bV_1 + bV_2 + cV_1^2 + cV_2^2 + 2cV_1V_2 \quad \dots(1.17.2)$$

3. For FET with $(V_1 - V_2)$ as gate input we have

$$i_{d2} = a + b(V_1 - V_2) + cV_1^2 + cV_2^2 - 2cV_1V_2 \quad \dots(1.17.3)$$

4. Since two currents are out of phase.

$$i_{\text{resultant}} = i_{d1} - i_{d2}$$

$$i_1 = 2bV_2 + 4cV_1V_2 \quad \dots(1.17.4)$$

5. We may take carrier voltage V_1 by $V_c \sin \omega_c t$ and the modulating voltage V_2 by $V_m \sin \omega_m t$.

Hence from eq. (1.17.4) we have,

$$i_1 = 2bV_m \sin \omega_m t + 4cV_m V_c \sin \omega_c t \sin \omega_m t$$

$$i_1 = 2bV_m \sin \omega_m t + 2cV_m V_c [\cos (\omega_c - \omega_m)t - \cos (\omega_c + \omega_m)t] \quad \dots(1.17.5)$$

6. The output voltage V_0 is proportional to this primary current (i_1). Let the constant of proportionality be α .

Then, $V_0 = \alpha i_1$

From eq. (1.17.5) we have

$$V_0 = 2\alpha b V_m \sin \omega_m t + 2\alpha V_m V_c \cos(\omega_c - \omega_m)t - 2\alpha V_c V_m \cos(\omega_c + \omega_m)t$$

Now let $P = 2\alpha b V_m$, $Q = 2\alpha V_m V_c$

7. Then output voltage of modulator

$$V_0 = \underbrace{P \sin \omega_m t}_{\text{Modulating Frequency}} + \underbrace{Q \cos(\omega_c - \omega_m)t}_{\text{Lower sideband}} - \underbrace{Q \cos(\omega_c + \omega_m)t}_{\text{Upper sideband}} \quad \dots(1.17.6)$$

Eq. (1.17.6) shows that the carrier has been cancelled out leaving only the two sidebands and the modulating frequency.

8. The tuning of the output transformer will remove modulating frequency and the output is DSB-SC.

Que 1.18. Describe the working principle of ring modulator to generate DSB-SC amplitude modulated signal.

AKTU 2014-15, Marks 05

Answer

1. The Chopper type balanced modulator (Ring modulator) consists of four diode connected in ring form; hence the circuit is also known as ring modulator.
2. The baseband signal $m(t)$ can be chopped by a switching or chopper circuit. One such chopper circuit is shown in Fig. 1.18.1.

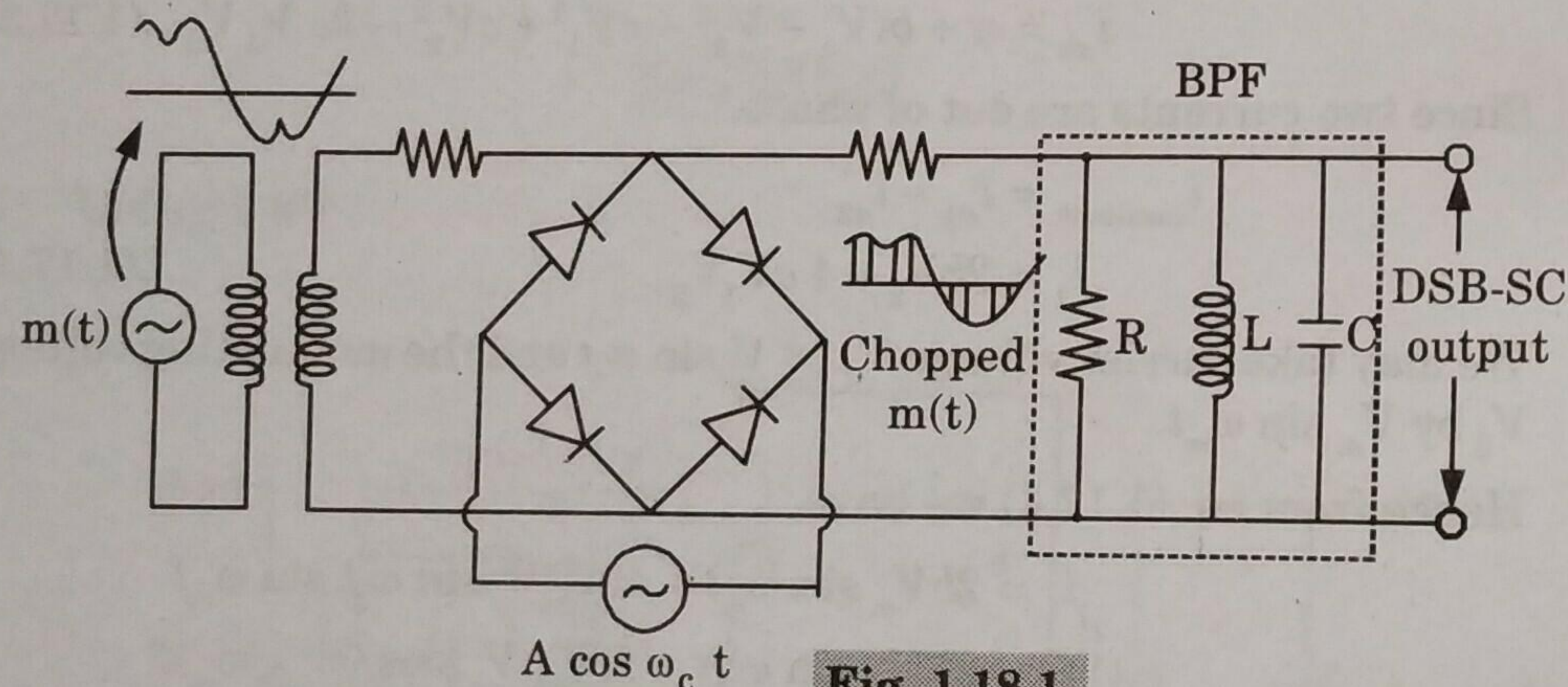


Fig. 1.18.1.

3. The diodes are used as a switching device. The carrier signal is such that its amplitude $A > |m(t)|_{\max}$ and $\omega_c > \omega_m$, where ω_m is the maximum frequency component present in the baseband signal $m(t)$.
4. For a positive cycle of the carrier signal, all the four diodes are forward biased.

5. The four conducting diodes provide a short circuit to the baseband signal $m(t)$ and the output is zero for the period of positive half cycles.
6. On the other hand, for negative cycles of the carrier, all the four diodes are reverse biased, and provide a path for the signal $m(t)$ to reach at the output.
7. Hence, for a positive half cycle of the carrier, no output appears, whereas, for the negative half cycle, $m(t)$ appears at the output, and thus, a chopped version of $m(t)$ is obtained at the output of the chopper circuit.
8. The parallel RLC network tuned to ω_c acts as a bandpass filter and at the output of this filter the DSB-SC signal is achieved.
9. The circuit is referred to as a double-balanced modulator as it is balanced with respect to the baseband signal as well as the carrier.

Que 1.19. Draw the block diagram of phase cancellation SSB generation and explain how the carrier and the unwanted sideband are suppressed.

OR

Draw the block diagram of generating SSB signals using balance modulator and explain each block.

AKTU 2014-15, Marks 05

OR

Draw and explain the block diagram of phase shift method for generating the SSB signal.

AKTU 2017-18, Marks 10

Answer

SSB modulation by phase shift or phase discrimination method :

1. Phase shift method of SSB generation is shown in Fig. 1.19.1.
2. Let baseband signal $m(t) = \cos \omega_m t$
After passing through 90° phase shift network
$$\hat{m}(t) = \cos(\omega_m t + 90^\circ) = \sin \omega_m t$$
3. Similarly, let the carrier be $\cos \omega_c t$ at one modulator and $\sin \omega_c t$ at the other.
4. The outputs of the balanced modulators (multipliers) are

$$\cos \omega_m t \cdot \cos \omega_c t = \frac{1}{2} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t] \quad \dots(1.19.1)$$

$$\sin \omega_m t \cdot \sin \omega_c t = \frac{1}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t] \quad \dots(1.19.2)$$

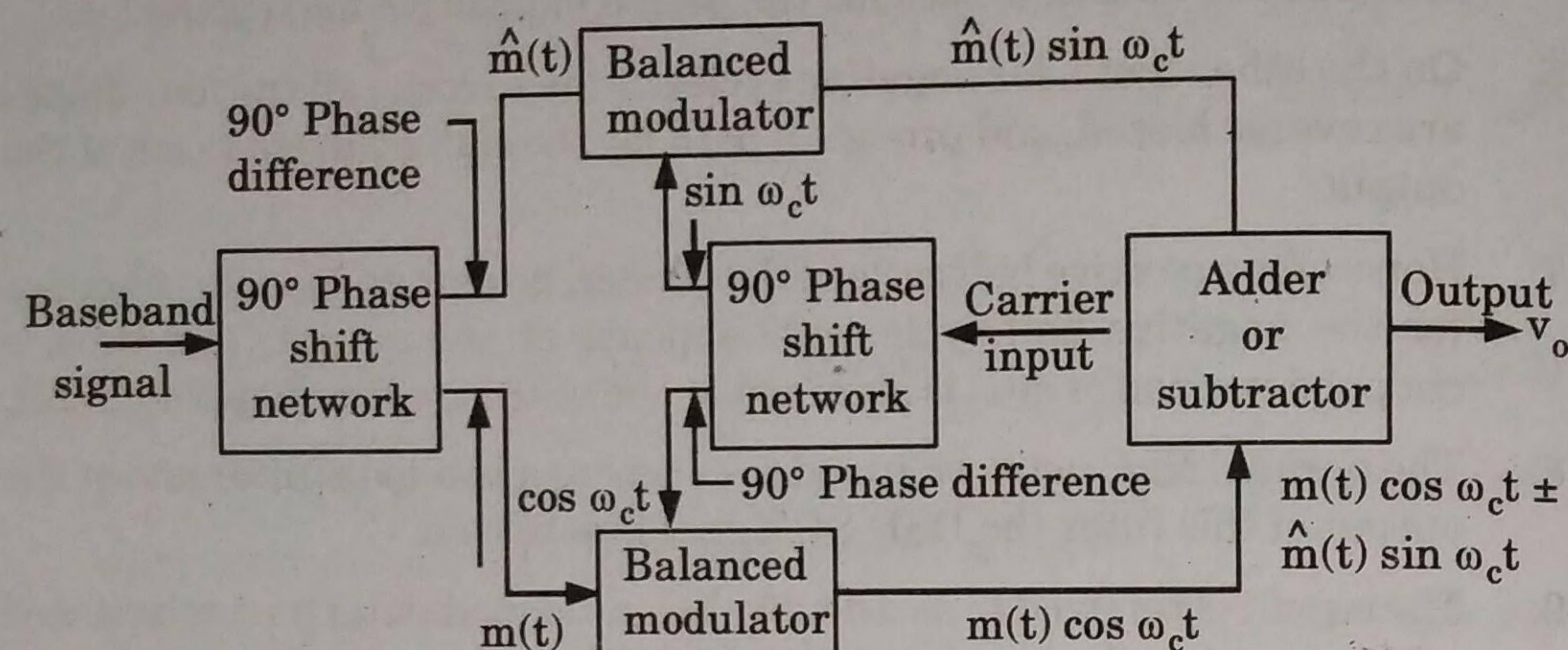


Fig. 1.19.1.

5. If eq. (1.19.1) and (1.19.2) are added, the lower sideband results, i.e.,

$$v_o = \cos(\omega_c - \omega_m)t$$

6. If eq. (1.19.1) and (1.19.2) are subtracted, the upper sideband appears at the output, i.e.,

$$v_o = \cos(\omega_c + \omega_m)t$$

Que 1.20. Explain the concept AM-SSB modulation and demodulation with the help of neat diagram and mathematical analysis for coherent detection.

AKTU 2016-17, Marks 15

Answer

SSB modulation : Refer Q. 1.19, Page 1-26B, Unit-1.

Coherent detection :

1. To demodulate $m(t)$ from single sideband signal $m_{SSB}(t)$, we can use coherent detection as shown in Fig. 1.20.1. It means we can get $m(t)$ by multiplying $m_{SSB}(t)$ with coherent carrier and passing it through a low pass filter.

3. As we know,

$$m_{SSB}(t) = A m(t) \cos \omega_c t \pm A m_h(t) \sin \omega_c t$$

where $m_h(t)$ is $m(t)$ delayed by $\pi/2$.

$$\begin{aligned} \therefore y(t) &= m_{SSB}(t) \cos \omega_c t \\ &= A m(t) \cos^2 \omega_c t \pm A m_h(t) \sin \omega_c t \cos \omega_c t \\ &= 0.5A m(t)[1 + \cos 2\omega_c t] \pm 0.5 m_h(t) \sin 2\omega_c t \end{aligned}$$

4. This signal when passed through a LPF of bandwidth that of $m(t)$ suppresses components around $2\omega_c$ and recovers the first term, i.e., the message signal.

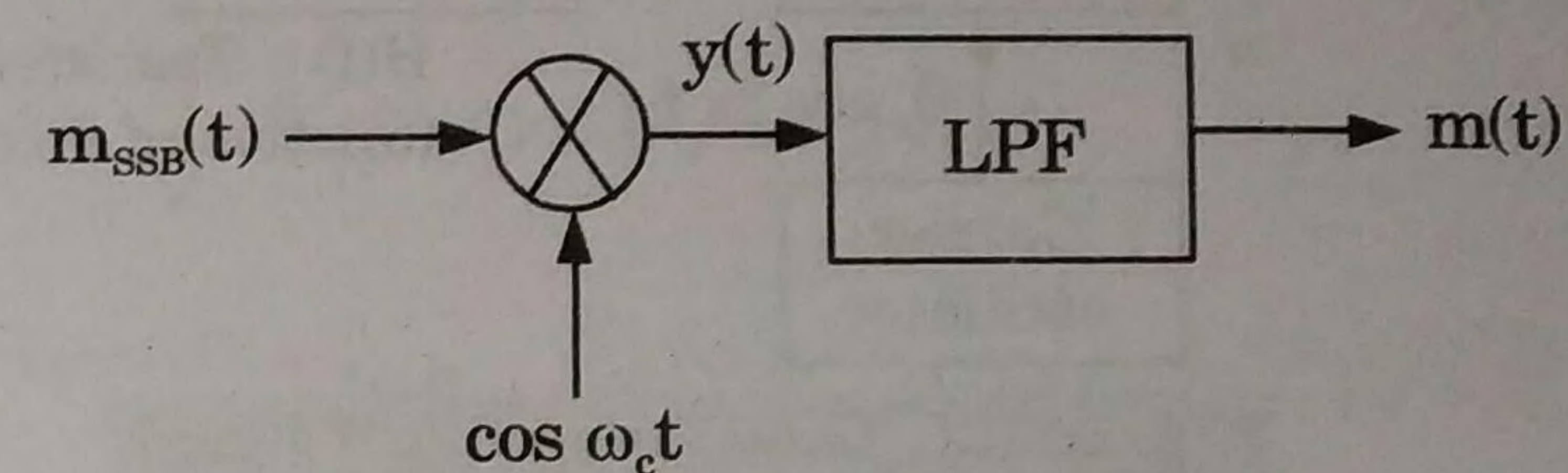


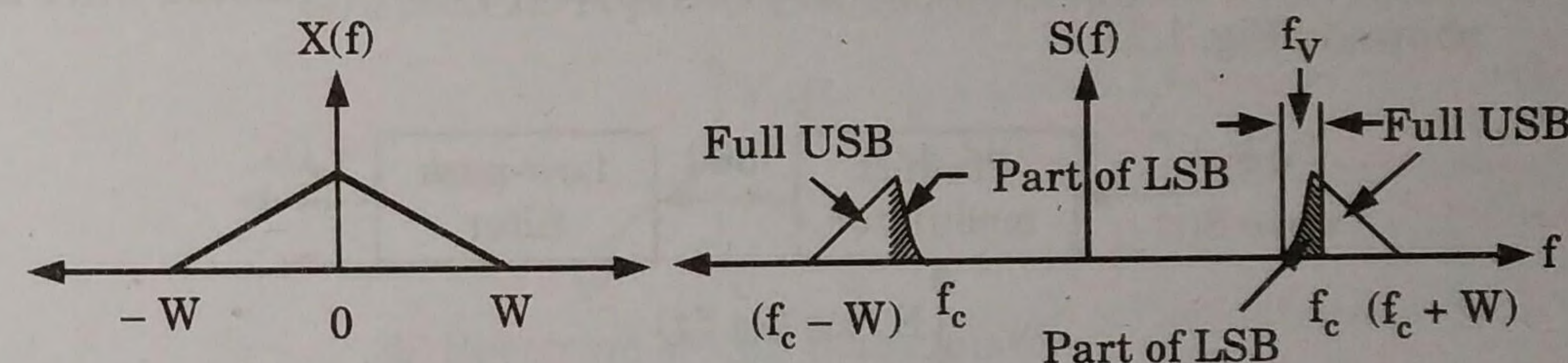
Fig. 1.20.1. SSB demodulation.

Que 1.21. What is VSB modulation ? What is the application of VSB modulation ?

Answer

Vestigial sideband modulation :

- In VSB modulation, one sideband is passed almost completely, whereas just a trace, or vestige, of the other side band is retained.
- The spectrum of VSB is shown in Fig. 1.21.1.



(a) Spectrum of message signal

(b) Spectrum of VSB signal

Fig. 1.21.1.

3. The transmission bandwidth of the VSB modulated wave is given by,

$$B = (W + f_V) \text{ Hz}$$

where

W = Message bandwidth

f_V = Width of the vestigial sideband

Generation of VSB modulated wave :

1. The block diagram of a VSB modulator is shown in Fig. 1.21.2.

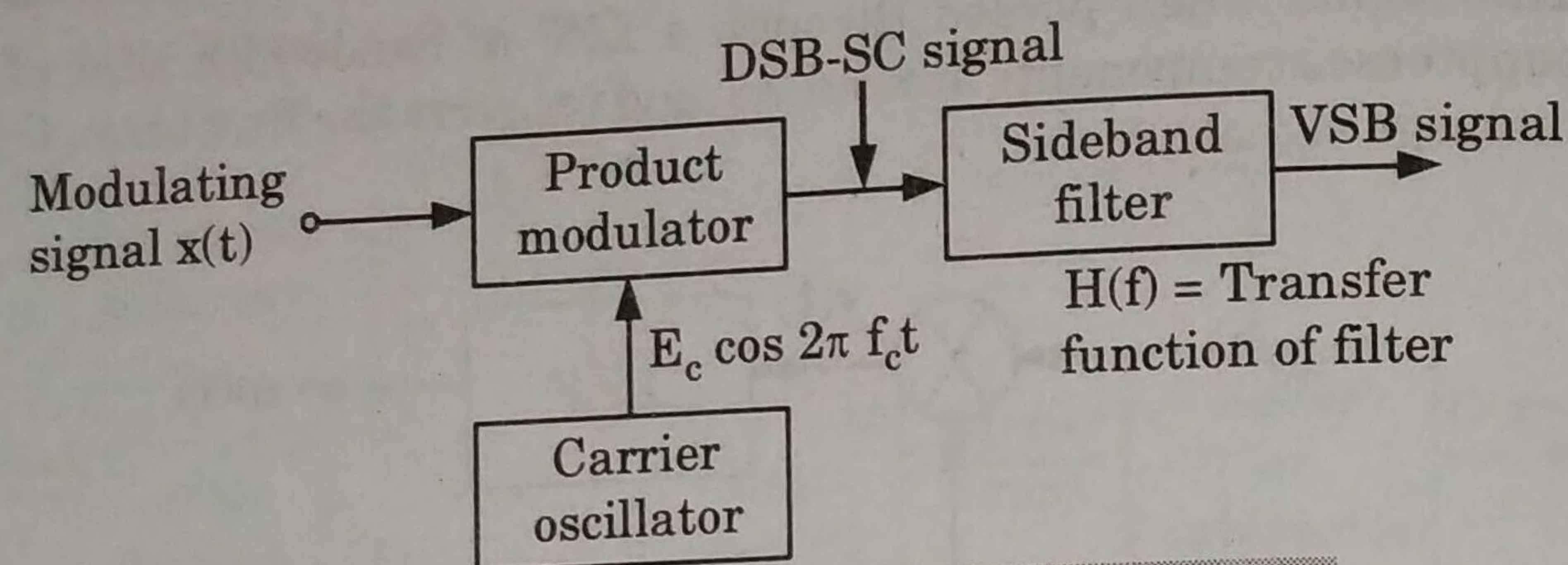


Fig. 1.21.2. Generation of VSB signal.

- The output of product modulator is

$$m(t) = x(t) \cdot E_c \cos (2\pi f_c t)$$

This represents a DSB-SC modulated wave.

- This DSB-SC signal is then applied to a sideband shaping filter.
- This filter will pass the wanted sideband as it is and the vestige of the unwanted sideband.
- Hence, the spectrum of the VSB modulated is given by,

$$S(f) = \frac{E_c}{2} [X(f - f_c) + X(f + f_c)] H(f)$$

Detection of VSB wave :

The synchronous detector for the detection of VSB modulated wave is shown in Fig. 1.21.3.

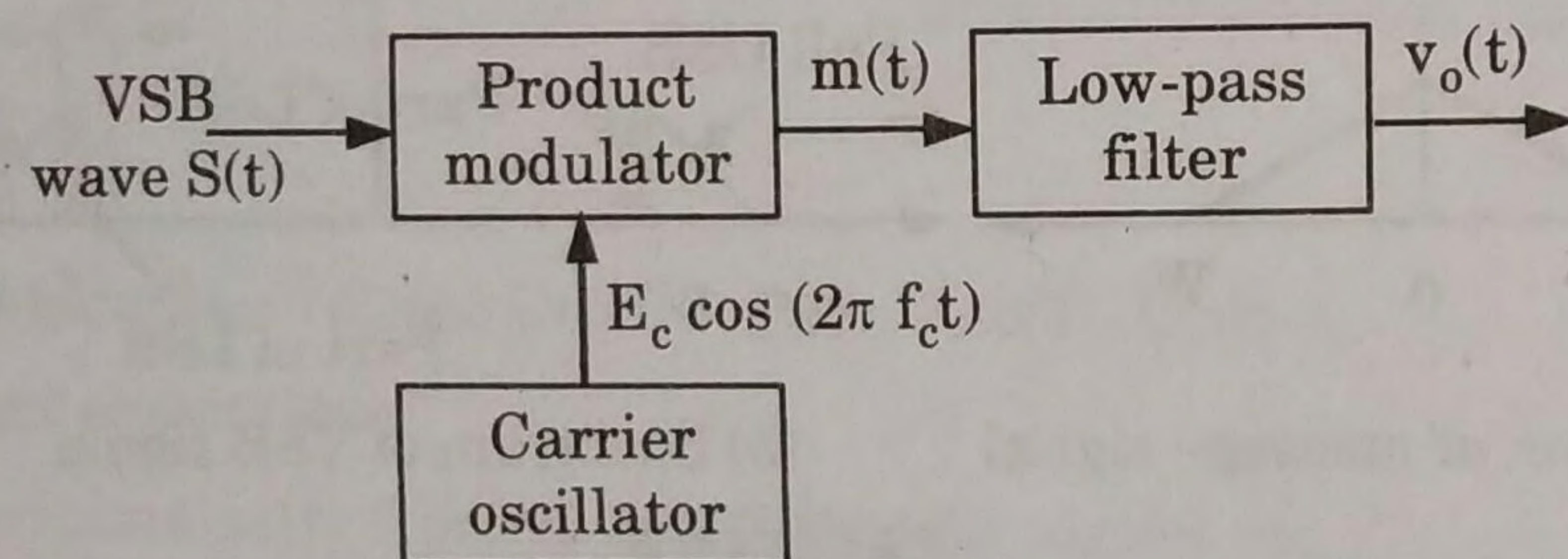


Fig. 1.21.3. Synchronous detector for VSB demodulation.

Operation :

- The VSB modulated wave is passed through a product modulator where it is multiplied with the locally generated synchronous carrier.
- Hence, the output of the product modulator is given by,

$$m(t) = S(t) \cos (2\pi f_c t) \quad \dots(1.21.1)$$

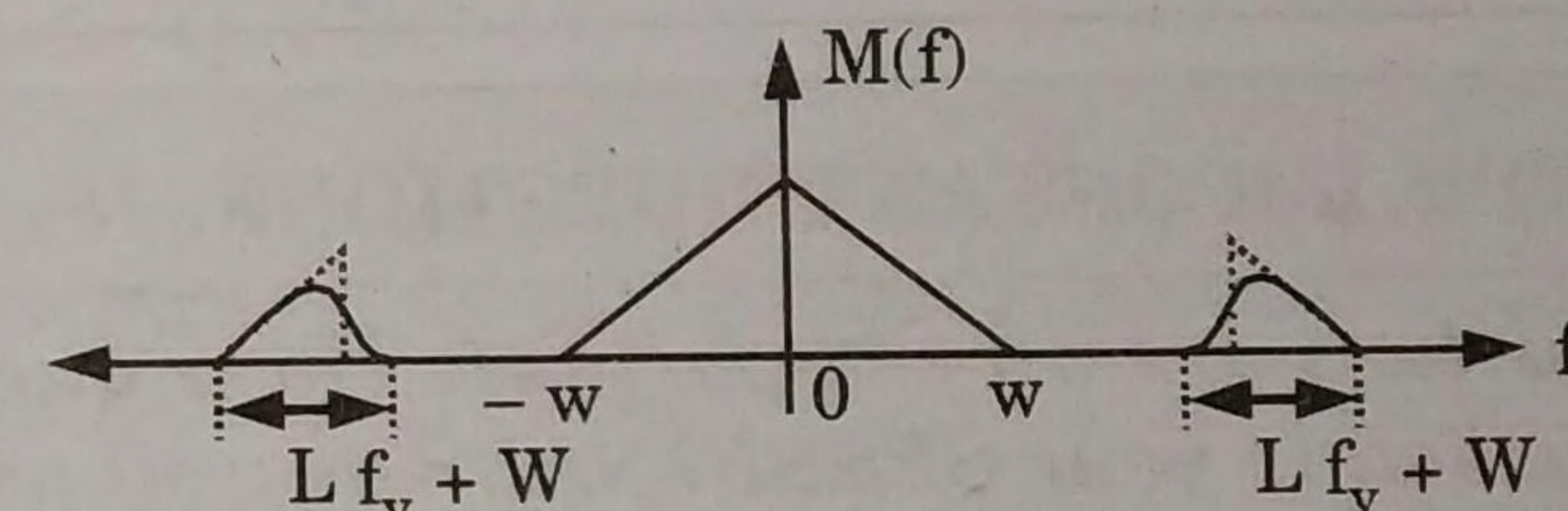
- Take the Fourier transform of both sides of eq. (1.21.1) to get,

$$M(f) = S(f) * \left[\frac{1}{2} \delta(f + f_c) + \frac{1}{2} \delta(f - f_c) \right] = \frac{1}{2} S(f + f_c) + \frac{1}{2} S(f - f_c) \dots(1.21.2)$$

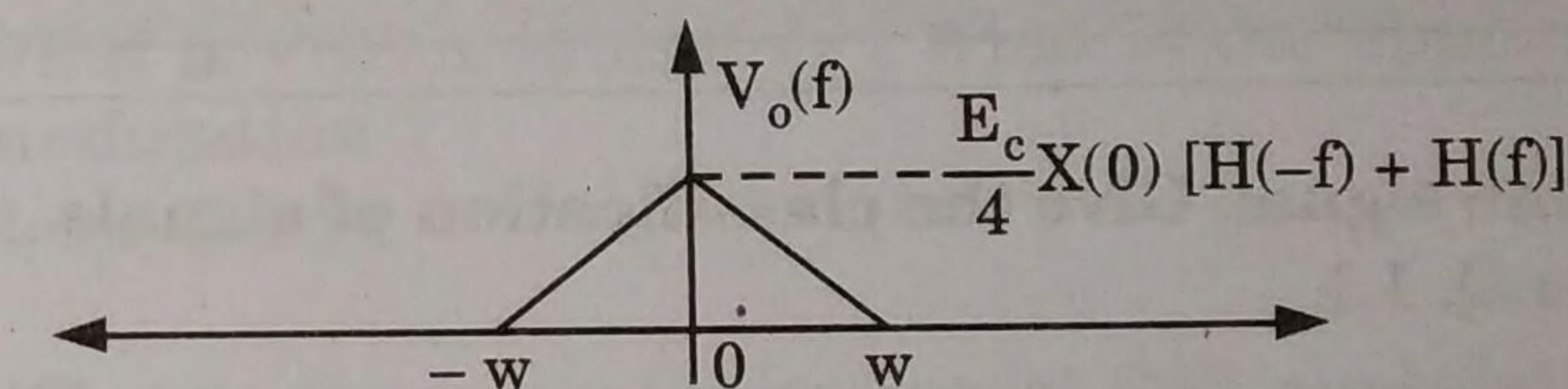
$$\text{But } S(f) = \frac{E_c}{2} [X(f - f_c) + X(f + f_c)] H(f)$$

$$\begin{aligned} \therefore M(f) &= \frac{E_c}{4} [X(f - 2f_c) H(f - f_c) + X(f + 2f_c) H(f + f_c)] \\ &+ \frac{E_c}{4} X(f) [H(f - f_c) + H(f + f_c)] \quad \dots(1.21.3) \end{aligned}$$

- The spectrum $M(f)$ is illustrated in Fig. 1.21.4(a). The first term in the eq. (1.21.3) represents the VSB modulated wave, corresponding to a carrier frequency of $2f_c$. This term will be eliminated by the filter to produce output $v_o(t)$.



(a) Spectrum of product modulator output



(b) Spectrum of VSB demodulator

Fig. 1.21.4.

- The second term in eq. (1.21.3) of $M(f)$ represents the spectrum of demodulated VSB output.

$$\therefore V_o(f) = \frac{E_c}{4} X(f) [H(f - f_c) + H(f + f_c)] \quad \dots(1.21.4)$$

This spectrum is illustrated in Fig. 1.21.4(b).

Applications of VSB : VSB modulation has become standard for the transmission of television signals. Because the video signals need a large transmission bandwidth if transmitted using DSB-SC techniques.

Que 1.22. What is the basic limitation of SSB modulation scheme ?
How it is eliminated by vestigial side band modulation ?

AKTU 2015-16, Marks 10

Answer

1. SSB signals are relatively difficult to generate due to the difficulty in isolating the desired sideband.
2. The required filter must have a very sharp cut-off characteristic, particularly when the baseband signal contains extremely low frequencies (e.g., television and telegraphic signals).
3. This is the basic limitation of SSB modulation scheme. This difficulty is overcome by a scheme known as vestigial sideband modulation.

Vestigial side band modulation : Refer Q. 1.21, Page 1-28B, Unit-1.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Define signal. Give the classification of signals.

Ans. Refer Q. 1.1.

Q. 2. Find the Fourier transform of the following signal :

- i. $\delta(t)$
- ii. $\cos(\omega_0 t)$
- iii. $\sin(\omega_0 t)$

Ans. Refer Q. 1.3.

Q. 3. Explain the amplitude modulation and deduce equation for this wave. Also derive the expression of modulation index in terms of maximum and minimum voltage.

Ans. Refer Q. 1.8.

Q. 4. Derive the relation between the output power of an AM transmitter and the depth of modulation.

Ans. Refer Q. 1.10.

Q. 5. Calculate the percentage power saving when the carrier and one of the sidebands are suppressed in AM wave modulated to a depth of

- i. 100 %
- ii. 75 %.

Ans. Refer Q. 1.11.

Q. 6. An amplitude modulated signal is given by

$$\Psi_{AM}(t) = 10 \cos(2\pi \times 10^6 t) + 5 \cos(2\pi \times 10^6 t) \cos(2\pi \times 10^3 t) + 2 \cos(2\pi \times 10^6 t) \cos(4\pi \times 10^3 t) \text{ volts.}$$

- i. Find various frequency components present and the corresponding modulation indices.
- ii. Calculate power contained in the signal and efficiency of the signal.

Ans. Refer Q. 1.13.

Q. 7. Describe the working principle of ring modulator to generate DSB-SC amplitude modulated signal.

Ans. Refer Q. 1.18.

Q. 8. Explain the concept AM-SSB modulation and demodulation with the help of neat diagram and mathematical analysis for coherent detection.

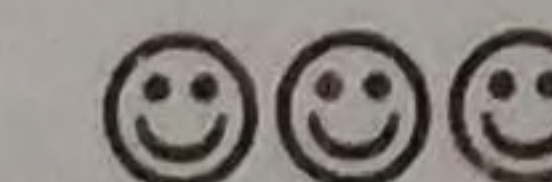
Ans. Refer Q. 1.20.

Q. 9. What is VSB modulation ? What is the application of VSB modulation ?

Ans. Refer Q. 1.21.

Q. 10. What is the basic limitation of SSB modulation scheme ? How it is eliminated by vestigial side band modulation ?

Ans. Refer Q. 1.22.



2

UNIT

Angle Modulation

CONTENTS

- Part-1 :** Angle Modulation, **2-2B to 2-18B**
Representation of
FM and PM Signals
- Part-2 :** Spectral Characteristics of **2-19B to 2-20B**
Angle Modulated Signal

PART-1

Angle Modulation, Representation of FM and PM Signals.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 2.1. Write a short note on angle modulation.

Answer

1. Angle modulation is one of the significant processes of modulation used for message transmission.
2. It is the process of varying the total phase angle of a carrier wave in accordance with the instantaneous value of the modulating signal, keeping amplitude of the carrier constant.
3. Let us consider an unmodulated carrier wave given by

$$\phi(t) = A \cos(\omega_c t + \theta_0) \quad \dots(2.1.1)$$

$$\phi(t) = A \cos \psi \quad \dots(2.1.2)$$

where $\psi = (\omega_c t + \theta_0)$ is the total phase angle of carrier wave.

Eq. (2.1.2) may be considered as the real part of the rotating phasor $Ae^{j\psi}$. Let it be denoted by $\tilde{\phi}$

$$\text{i.e.,} \quad \tilde{\phi} = Ae^{j\psi} \quad \dots(2.1.3)$$

$$\begin{aligned} \text{and,} \quad \phi(t) &= \text{Re}[Ae^{j\psi}] \\ &= A \text{Re}[\cos \psi + j \sin \psi] \\ &= A \cos \psi \quad \dots(2.1.4) \end{aligned}$$

4. The phasor $\tilde{\phi}$ rotates at a constant angular velocity ω_c provided θ_0 is independent of time.
5. The constant angular velocity ω_c of the phasor $\tilde{\phi}$ is related to its total phase angle $\psi(t)$ as
6. Differentiating eq. (2.1.5), we get

$$\psi = \omega_c t + \theta_0 \quad \dots(2.1.5)$$

$$\frac{d\psi}{dt} = \omega_c \quad \dots(2.1.6)$$

This derivative $\frac{d\psi}{dt}$ is constant with time for an unmodulated carrier.

7. Accordingly, angular velocity of the phasor $\tilde{\phi}$ will also change with time. This time dependent angular velocity is called instantaneous angular velocity, and it is denoted by ω_i .
8. Thus, eq. (2.1.6) will become

$$\frac{d\psi}{dt} = \omega_i$$

$$\psi = \int \omega_i dt$$

where ω_i is time dependent.

Que 2.2. Explain angle modulation with its types.

Answer

Angle modulation : Refer Q. 2.1, Page 2-2B, Unit-2.

Types :

i. Phase Modulation :

1. In this type of modulation, the phase angle $\psi(t)$ is varied linearly with a modulating signal $f(t)$ about an unmodulated phase angle $\omega_c t$.
2. In this the instantaneous value of phase angle ψ_i is equal to the phase of an unmodulated carrier ($\omega_c t$) plus a time varying component proportional to $f(t)$.

Mathematically,

$$\psi_i(t) = \omega_c t + k_p f(t) \quad \dots(2.2.1)$$

The proportional constant k_p is known as phase sensitivity expressed in radians/volts.

3. The carrier wave after phase modulation is given by

$$\begin{aligned} \phi_{PM} &= A \cos \psi_i(t) \\ &= A \cos[\omega_c t + k_p f(t)] \end{aligned} \quad \dots(2.2.2)$$

ii. Frequency Modulation :

1. In this type of angle modulation, the instantaneous frequency ω_i is varied linearly with a modulating signal $f(t)$ about an unmodulated frequency ω_c .
2. In this the instantaneous value of the angular frequency ω_i is equal to the frequency ω_c of the unmodulated carrier plus a time varying component proportional to $f(t)$.

Mathematically, $\omega_i = \omega_c + k_f f(t)$...(2.2.3)

The term k_f represent the frequency sensitivity of the modulator expressed in Hz/volt.

$$\omega_i = \frac{d\psi_i}{dt} = \omega_c + k_f f(t) \quad \dots(2.2.4)$$

3. The total phase angle of the FM wave can be obtained by integrating eq. (2.2.3)

$$\psi_i = \int \omega_i dt = \int [\omega_c + k_f f(t)] dt \quad \dots(2.2.5)$$

$$= \omega_c t + k_f \int f(t) dt$$

The corresponding FM wave is given by

$$\phi_{FM}(t) = A \cos \psi_i = A \cos \left(\omega_c t + k_f \int f(t) dt \right) \quad \dots(2.2.6)$$

4. If the phase angle of carrier at $t = 0$ is zero, then the limit of integration will be 0 to t .

$$\phi_{FM}(t) = A \cos \left(\omega_c t + k_f \int_0^t f(t) dt \right) \quad \dots(2.2.7)$$

Que 2.3. Relate Phase and Frequency Modulation.

OR

Show the generation of narrow band FM using phase modulator.

AKTU 2017-18, Marks 10

Answer

1. PM and FM are closely related in sense that the net effect of both is variation in total phase angle.
2. In PM, phase angle varies linearly with $f(t)$ whereas in FM phase angle varies linearly with the integral of $f(t)$.
3. We can get FM by using PM provided that at first the modulating signal is integrated and then applied to phase modulator.

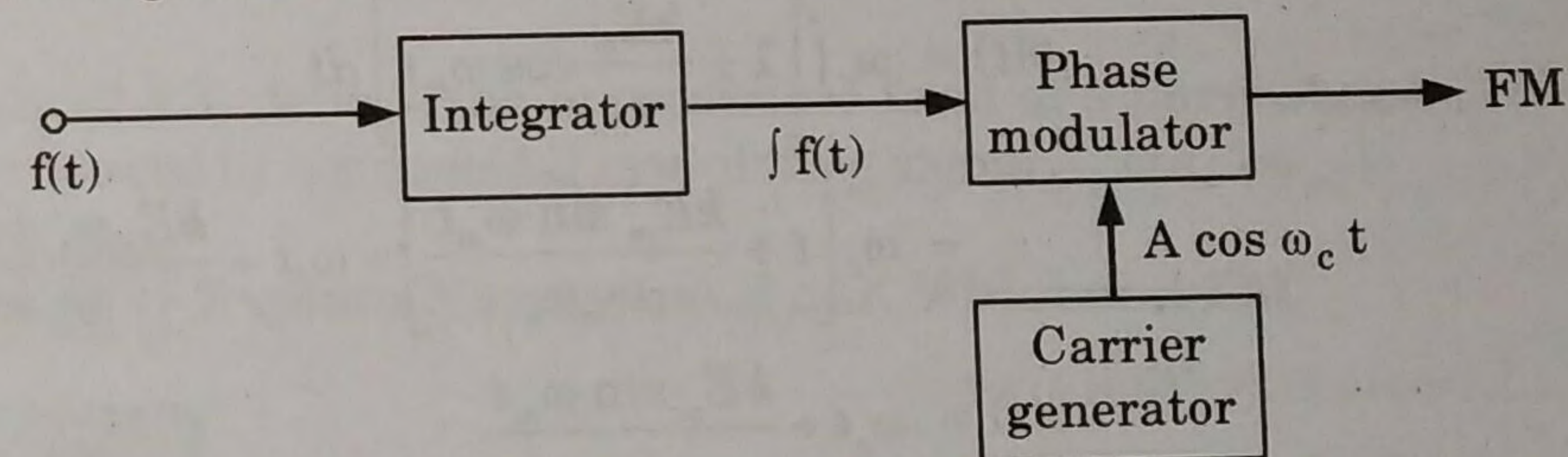


Fig. 2.3.1. FM generation using phase modulator.

4. Similarly we can generate a PM wave using frequency modulator provided that $f(t)$ is first differentiated and then applied to frequency modulator.

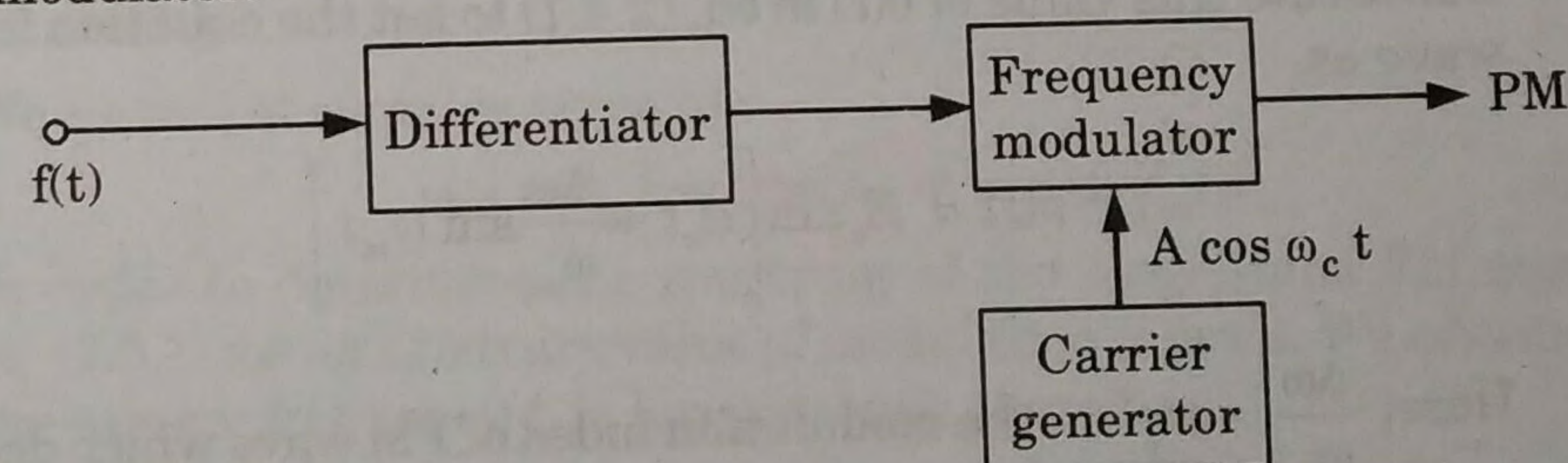


Fig. 2.3.2. PM generation using frequency modulator.

Que 2.4. Derive the expression for the instantaneous value of an FM voltage and define the modulation index.

OR

Explain and derive the expression for single tone frequency modulated wave.

Answer

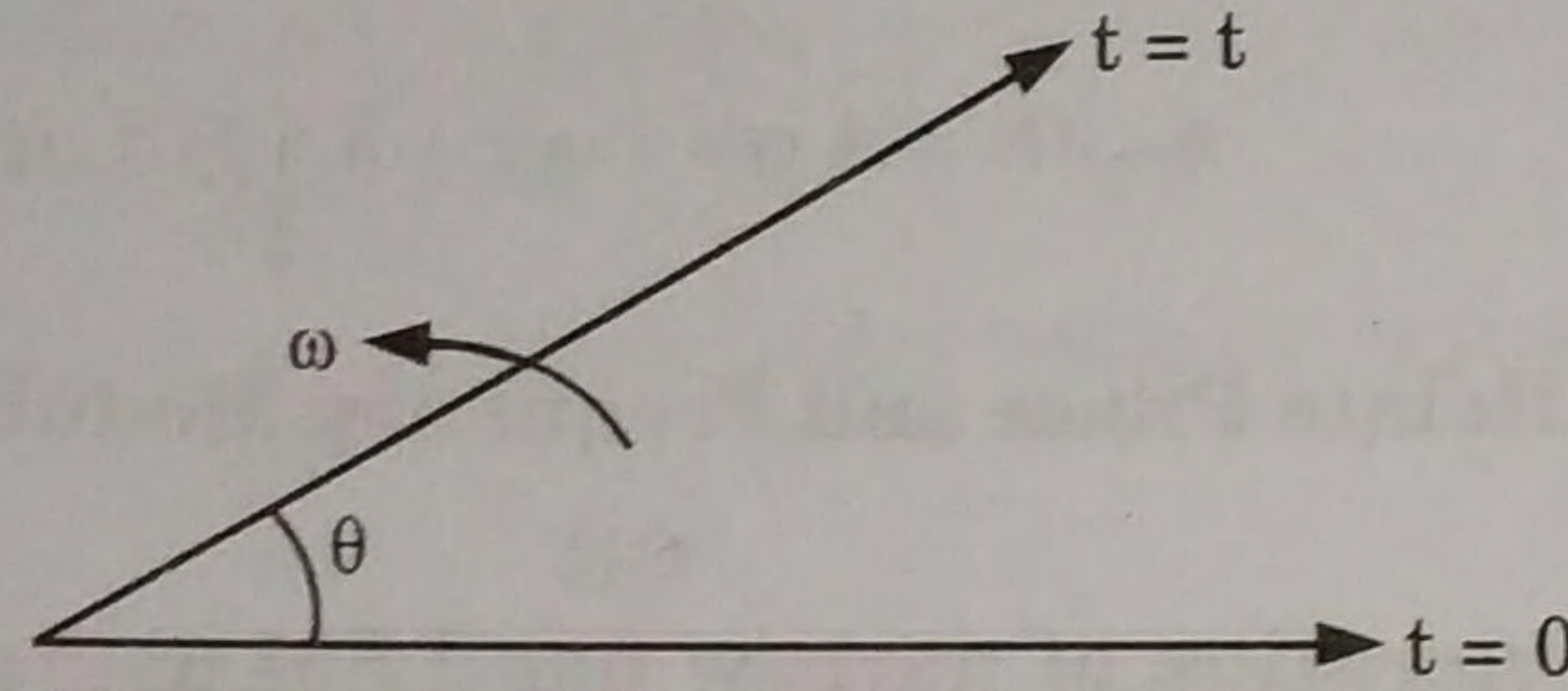


Fig. 2.4.1. Frequency modulated vector.

1. The FM wave is represented by

$$s(t) = E_c \sin \theta(t) \quad \dots(2.4.1)$$

2. If E_c is rotating at a constant velocity ω then $\theta(t) = \omega t$. The angular velocity of FM wave is given as,

$$\omega = [\omega_c + k E_m \cos \omega_m t] \quad \dots(2.4.2)$$

3. Hence

$$\begin{aligned} \theta(t) &= \int \omega dt = \int [\omega_c + k E_m \cos \omega_m t] dt \\ \theta(t) &= \omega_c \int \left[1 + \frac{k E_m}{\omega_c} \cos \omega_m t \right] dt \\ &= \omega_c \left[t + \frac{k E_m \sin \omega_m t}{\omega_m \omega_c} \right] = \omega_c t + \frac{k E_m \sin \omega_m t}{\omega_m} \\ \therefore \theta(t) &= \omega_c t + \frac{k E_m \sin \omega_m t}{\omega_m} \quad \dots(2.4.3) \end{aligned}$$

4. As per the definition, $\Delta\omega = k E_m$

$$\therefore \theta(t) = \omega_c t + \frac{\Delta\omega \sin \omega_m t}{\omega_m} \quad \dots(2.4.4)$$

5. Substitute this value of $\theta(t)$ in eq. (2.4.1) to get the equation for the FM wave as,

$$s(t) = E_c \sin \left[\omega_c t + \frac{\Delta\omega}{\omega_m} \sin \omega_m t \right] \quad \dots(2.4.5)$$

6. Here, $\frac{\Delta\omega}{\omega_m} = m_f$ i.e., the modulation index of FM wave which decides the bandwidth of FM wave.

7. Hence the equation of FM wave is given as,

$$s(t) = E_c \sin[\omega_c t + m_f \sin \omega_m t] \quad \dots(2.4.6)$$

This is the expression for a FM wave.

Que 2.5. Write a short note on Narrowband FM.

Answer

1. When k_f i.e., frequency sensitivity is small, the bandwidth of FM is narrow.

2. We know FM signal is given by

$$s(t) = A_c \cos [2\pi f_c t + \beta \sin (2\pi f_m t)] \quad \dots(2.5.1)$$

where β is modulation index.

3. Expanding the relation of eq. (2.5.1)

$$\begin{aligned} s(t) &= A_c \cos (2\pi f_c t) \cos [\beta \sin (2\pi f_m t)] \\ &\quad - A_c \sin (2\pi f_c t) \sin [\beta \sin (2\pi f_m t)] \quad \dots(2.5.2) \end{aligned}$$

4. Assuming that the modulation index β is small as compared to 1 radian then the approximations

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

$$\text{and, } \sin [\beta \sin (2\pi f_m t)] \approx \beta \sin (2\pi f_m t)$$

5. Eq. (2.5.2) simplifies to

$$s(t) \approx A_c \cos (2\pi f_c t) - \beta A_c \sin (2\pi f_c t) \sin (2\pi f_m t) \quad \dots(2.5.3)$$

6. Eq. (2.5.3) defines the approximate form of a narrowband FM signal produced by a sinusoidal modulating signal $A_m \cos (2\pi f_m t)$.

Que 2.6. Explain Narrowband and Wideband FM.

Answer

Narrowband FM : Refer Q. 2.5, Page 2-6B, Unit-2.

Wideband FM :

1. When k_f has an appreciable value, then the FM signal has a wide bandwidth.

2. We know FM signal is given by

$$s(t) = A_c \cos [2\pi f_c t + \beta \sin (2\pi f_m t)] \quad \dots(2.6.1)$$

3. In order to determine the spectrum of the single-tone FM signal of eq. (2.6.1) for an arbitrary value of modulation index β . We assume that the carrier frequency f_c is large enough therefore,

$$\begin{aligned} s(t) &= \text{Re}[A_c \exp (j2\pi f_c t + j\beta \sin (2\pi f_m t))] \\ &= \text{Re} [\tilde{s}(t) \exp (j2\pi f_c t)] \quad \dots(2.6.2) \end{aligned}$$

where $\tilde{s}(t)$ is a complex envelope of the FM signal $s(t)$.

$$\tilde{s}(t) = A_c \exp[j\beta \sin(2\pi f_m t)]$$

4. Now, we expand $\tilde{s}(t)$ in the form of complex Fourier series then we get

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j2\pi n f_m t) \quad \dots(2.6.3)$$

where c_n is complex Fourier coefficient.

$$\begin{aligned} c_n &= f_m \int_{-1/2f_m}^{1/2f_m} \tilde{s}(t) \exp(-j2\pi n f_m t) dt \\ &= f_m A_c \int_{-1/2f_m}^{1/2f_m} \exp[j\beta \sin(2\pi f_m t) - j2\pi n f_m t] dt \end{aligned} \quad \dots(2.6.4)$$

5. Defining a new variable

$$x = 2\pi f_m t \quad \dots(2.6.5)$$

Hence,
$$c_n = \frac{A_c}{2\pi} \int_{-\pi}^{\pi} \exp[j(\beta \sin x - nx)] dx \quad \dots(2.6.6)$$

6. The integral on the right hand side is recognized as the n^{th} order Bessel function of the first kind and argument β . This function is commonly denoted by $J_n(\beta)$.

$$J_n(\beta) = \frac{1}{2\pi} \int_{-\pi}^{\pi} \exp[j(\beta \sin x - nx)] dx$$

7. Therefore eq. (2.6.4) can be reduced to

$$c_n = A_c J_n(\beta)$$

8. Therefore,

$$\tilde{s}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j2\pi n f_m t) \quad \dots(2.6.7)$$

9. Substituting the value of $\tilde{s}(t)$ in eq. (2.6.2)

$$s(t) = A_c \operatorname{Re} \left[\sum_{n=-\infty}^{\infty} J_n(\beta) \exp[j2\pi(f_c + n f_m)t] \right] \quad \dots(2.6.8)$$

Eq. (2.6.8) can be written as

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[2\pi(f_c + n f_m)t]$$

10. Thus, the discrete spectrum of $s(t)$ is obtained by taking Fourier transform of both the sides

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta) [\delta(f - f_c - n f_m) + \delta(f + f_c + n f_m)] \quad \dots(2.6.9)$$

Que 2.7. Discuss the transmission bandwidth of FM signals and explain Carson's rule.

Answer

Bandwidth of FM signals :

1. The bandwidth of FM signal is given by

$$BW = 2n \omega_m \text{ radians/s}$$

where, n = Number of significant sidebands produced.

2. Since

$$n = m_f$$

$$BW = 2m_f \omega_m$$

3. As

$$m_f = \frac{\Delta\omega}{\omega_m}$$

Hence,

$$BW = 2 \frac{\Delta\omega}{\omega_m} \cdot \omega_m = 2(\Delta\omega) \text{ radians}$$

$$BW = 2(\Delta f) \text{ Hz}$$

4. Thus, the bandwidth of FM system is given as twice the frequency deviation.

Carson's rule :

1. An empirical formula for the bandwidth of a single-tone wideband FM is given by Carson's rule.

2. According to this rule, the FM bandwidth is given by

$$BW = 2(\Delta\omega + \omega_m) = 2\Delta\omega + 2\omega_m$$

As

$$\omega_m = \frac{\Delta\omega}{m_f}$$

where, m_f = FM modulation index

$$BW = 2(\Delta\omega) + \frac{2\Delta\omega}{m_f}$$

$$= 2\Delta\omega \left[1 + \frac{1}{m_f} \right] \text{ radians} = 2\Delta f \left[1 + \frac{1}{m_f} \right] \text{ Hertz}$$

Special case :

1. When $\Delta\omega \ll \omega_m$ (narrowband FM); $m_f \ll 1$

$$BW = 2\omega_m$$

2. When $\Delta\omega \gg \omega_m$ (wideband FM); $m_f \gg 1$

$$BW = 2(\Delta\omega)$$

Que 2.8. Compare Narrowband FM (NBFM) versus Wideband FM (WBFM).

Answer

S.No.	NBFM	WBFM
1.	Modulation index is less than one.	Modulation index is greater than one.
2.	Frequency deviation is less.	Frequency deviation is more.
3.	Bandwidth = $2f_m$.	Bandwidth is 15 times of NBFM bandwidth.
4.	Less suppressing of noise.	Noise is more suppressed.
5.	Spectrum contains two sidebands and carrier.	Spectrum contains infinite number of sidebands and carrier.

Que 2.9. What are the advantages of frequency modulation? The carrier frequency of an FM broadcast transmitter is 100 MHz and maximum frequency deviation is 75 kHz. If the highest audio frequency modulating the carrier is 15 kHz, what is the approximate bandwidth of the signal?

Answer**Advantages of frequency modulation :**

1. Improved noise immunity.
2. Low power is required to be transmitted to obtain the same quality of received signal at the receiver.
3. Covers a larger area with the same amount of transmitted power.
4. Transmitted power remains constant.
5. All the transmitted power is useful.

Numerical :

Given : $f_c = 100$ MHz, $\Delta f = 75$ kHz; Highest audio frequency, $f_m = 15$ kHz

To Find : Bandwidth.

1.
$$\beta = \frac{\Delta f}{f_m} = \frac{75}{15} = 5$$
2.
$$\text{Bandwidth} = 2(\beta + 1)f_m = 2(5 + 1)15 = 180 \text{ kHz}$$

Que 2.10. A carrier wave of frequency 100 MHz is frequency modulated by a sinusoidal wave of amplitude 20 V and frequency

100 kHz. The frequency sensitivity of the modulator is 25 kHz per volt. Determine approximate bandwidth of FM signal.

Answer

Given : $f_c = 100$ MHz; $E_m = 20$ V; $f_m = 100$ kHz; $k_f = 25$ kHz/V

To Find : Bandwidth.

1. Frequency deviation,

$$\Delta f = k_f E_m$$

$$= 25 \times 10^3 \times 20 = 5 \times 10^5 \text{ Hz}$$
2. So the bandwidth,

$$\text{BW} = 2(5 \times 10^5 + 10^5) \text{ Hz}$$

$$\text{BW} = 1.2 \text{ MHz}$$

Que 2.11. Write a short note on performance comparison of frequency modulation and amplitude modulation.

AKTU 2017-18, Marks 10

Answer

S.No.	Frequency modulation	Amplitude modulation
1.	Bandwidth is higher and depends on modulation index.	Bandwidth is lower compared to FM but independent of modulation index.
2.	Since bandwidth is large, fidelity is better.	AM has poor fidelity due to narrow bandwidth.
3.	Noise interference is minimum.	Noise interference is more.
4.	It is possible to decrease noise by increasing deviation.	This feature is absent in AM.
5.	All transmitted power is useful.	Carrier power and one sideband power is useless.
6.	Transmitted power remains constant irrespective of modulation index.	Transmitted power varies according to modulation index.

Que 2.12. Consider a frequency modulated signal is given by

$V_{FM}(t) = 20 \cos[2\pi \cdot 10^6 t + 0.1 \sin(10^4 \cdot \pi t)]$. Given $k_f = 10 \pi$, derive the expression for the modulating signal. **AKTU 2014-15, Marks 05**

Answer

Given : $V_{FM}(t) = 20 \cos [2\pi \cdot 10^6 t + 0.1 \sin (10^4 \cdot \pi t)]$; $k_f = 10 \pi$

To Derive : Expression for modulating signal.

1. Comparing with standard equation of $V_{FM}(t)$

$$V_{FM}(t) = A_c \cos \left[2\pi f_c t + \frac{k_f A}{f_m} \sin 2\pi f_m t \right]$$

$$\therefore A_c = 20, \frac{k_f A}{f_m} = 0.1, 2\pi f_m = 10^4 \pi$$

$$k_f A = 0.1 f_m = 0.1 \times \frac{10^4 \pi}{2\pi}$$

$$k_f A = 500$$

$$A = \frac{500}{10\pi} = \frac{50}{\pi}$$

2. Modulating signal

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

$$= \frac{50}{\pi} \cos (10^4 \cdot \pi t)$$

Que 2.13. In an FM system a 7 kHz modulating signal modulates 107.6 MHz carrier wave so that the frequency deviation is 50 kHz. Determine

- Carrier swing in the FM signal and modulation index.
- The highest and lowest frequency attained by the FM signal.

AKTU 2017-18, Marks 10

Answer

Given : $f_m = 7 \text{ kHz}$; $f_c = 107.6 \text{ MHz}$; $\Delta f = 50 \text{ kHz}$

To Find : Carrier swing; Modulation index, m_f ;

Highest frequency, f_H ; Lowest frequency f_L .

- Carrier swing of FM signal = $2 \times \Delta f = 2 \times 50 = 100 \text{ kHz}$
Modulation index,

$$m_f = \frac{\Delta \omega}{\omega_m} = \frac{\Delta f}{f_m}$$

$$m_f = \frac{50 \text{ kHz}}{7 \text{ kHz}} = 7.143$$

- Highest frequency,

$$\begin{aligned} f_H &= f_c + \Delta f \\ &= 107.6 \text{ MHz} + 50 \text{ kHz} \\ &= 107.65 \text{ MHz} \end{aligned}$$

Lowest frequency, $f_L = f_c - \Delta f$

$$= 107.6 \text{ MHz} - 50 \text{ kHz} = 107.55 \text{ MHz}$$

Que 2.14. Explain generation of FM signals and define its methods.

OR

Illustrate the principle of Armstrong method of generating FM.

Answer

- There are two methods of generating FM signals, namely, direct FM and indirect FM.
- In direct method the carrier frequency is directly varied in accordance with the input baseband signal whereas in indirect method, the modulating signal is first used to produce a narrowband FM signal and frequency multiplication is next used to increase the frequency deviation to the desired level.
- The FM modulator circuits used for generating FM signal can be grouped into two categories :

a. Parameter Variation Method :

- In this method, the modulating signal directly modulates the carrier. The generator which produces the carrier of an FM waveform is a tuned

circuit oscillator. The frequency of oscillation is given by $f = \frac{1}{2\pi\sqrt{LC_0}}$ in

which L is the inductance and C_0 is the capacitance.

- The capacitor consists here of fixed capacitor C_0 , which is shunted by a voltage variable capacitor C_v .
- A voltage variable capacitor is commonly known as varicap. In this circuit, the modulating signal varies the voltage across C_v .

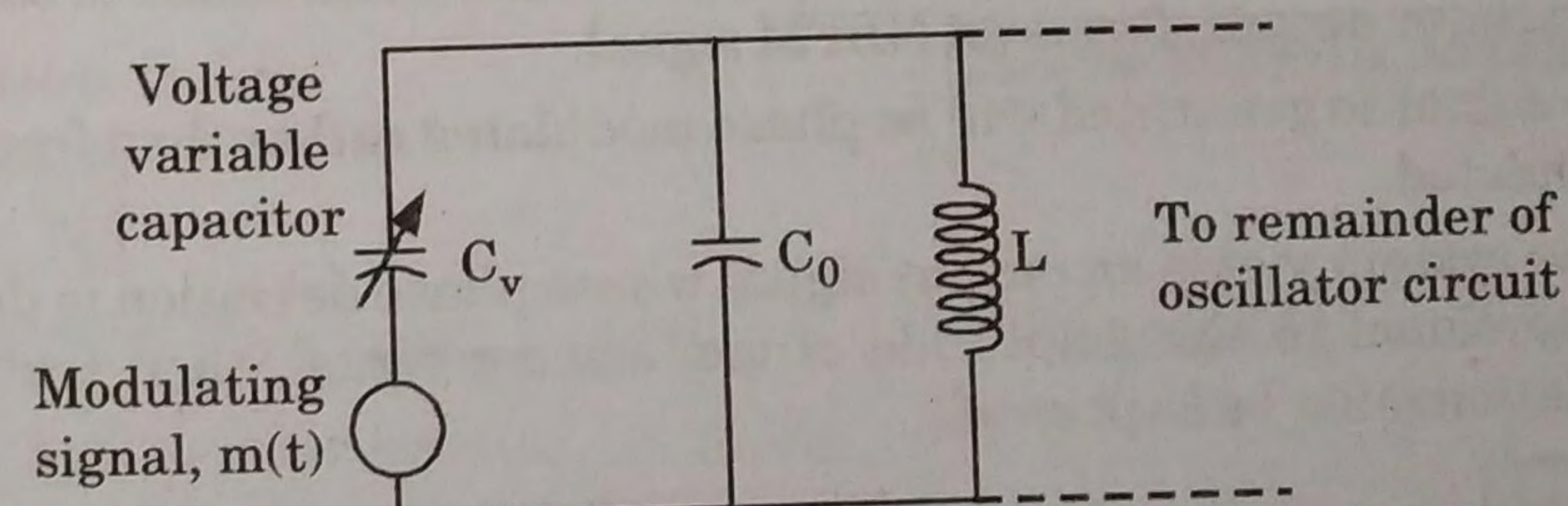


Fig. 2.14.1. A voltage variable capacitor is used to frequently modulate an LC oscillator.

4. Therefore the capacitance of C_v changes and causes a corresponding change in the oscillator frequency.
5. An oscillator whose frequency is controlled by the modulating signal voltage is called a voltage-controlled oscillator.
6. A principal difficulty with the parameter variation method of frequency modulation is the difficulty it entails when we require that the carrier frequency be maintained constant to a higher order of precision over extended period of time.
7. There is a certain measure of inconsistency in requiring a device that have long time frequency stability and yet be able to respond readily to a modulating signal.

b. FM Generation by Armstrong's Indirect Method :

1. A phase-modulated waveform in which the modulating waveform $m(t)$ is written as $\cos[\omega_c t + m(t)]$. If modulation is narrowband [$|m(t)| \ll 1$], then

$$\cos[\omega_c t + m(t)] \cong \cos \omega_c t - m(t) \sin \omega_c t$$

Here, $m(t)$ is modulating waveform and $\sin \omega_c t$ the carrier.

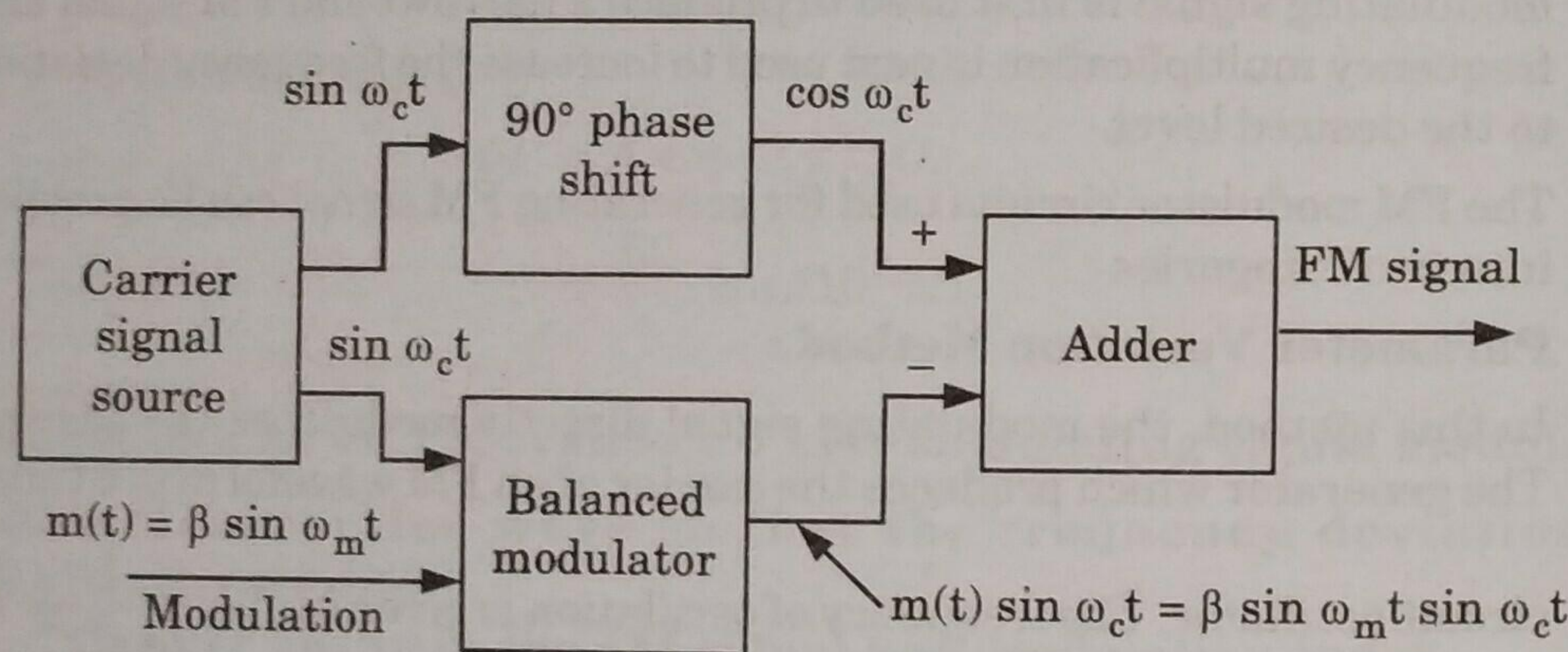


Fig. 2.14.2. Illustrating the principle of the Armstrong system of generating a PM signal.

2. In this, a balanced modulator is employed to generate the DSB-SC signal using $\sin \omega_c t$ as the carrier of the modulator.
3. This carrier is then shifted in phase by 90° and, when added to balanced modulator output, forms an NBFM signal.
4. The signal so generated will be phase modulated rather than frequency modulated.
5. If the system yields an output signal whose phase deviation is directly proportional to the amplitude of modulating signal, then the phase deviation must be kept small.

Que 2.15. An angle-modulated signal with carrier frequency $\omega_c = 2\pi \times 10^5$ is described by the equation $\Psi_{FM}(t) = 10 \cos(\omega_c t + 5 \sin 3000\pi t + 10 \sin 2000\pi t)$

- i. Find the power of the modulated signal.
- ii. Find the frequency deviation Δf .
- iii. Estimate bandwidth of the FM signals.

AKTU 2014-15, Marks 05

Answer

Given : $\omega_c = 2\pi \times 10^5$

$$\Psi_{FM}(t) = 10 \cos(\omega_c t + 5 \sin 3000\pi t + 10 \sin 2000\pi t)$$

- To Find :**
- i. Power of modulated signal
 - ii. Frequency deviation
 - iii. Bandwidth.

- i. Power of modulated signal

$$P_c = \frac{A_c^2}{2R} = \frac{(10)^2}{2 \times 1} \quad (\because R = 1)$$

$$= 50 \text{ W}$$

- ii. Frequency deviation

$$\omega_i = \frac{d}{dt} Q(t) = \frac{d}{dt} [\omega_c t + 5 \sin 2000\pi t + 10 \sin 3000\pi t]$$

$$= \omega_c + 10000\pi \cos 2000\pi t + 30000\pi \cos 3000\pi t$$

$$\therefore \text{maximum values of carrier deviation} = 10000\pi + 30000\pi$$

$$\therefore \text{frequency deviation, } \Delta f = \frac{10000\pi + 30000\pi}{2\pi} = 20 \text{ kHz.}$$

- iii. Bandwidth of FM signal

$$BW = 2[\Delta f + f_{m(\max)}]$$

$$= 2[20000 + 1500] = 43 \text{ kHz.}$$

Que 2.16. A PM modulator has a phase deviation sensitivity of 2.5 radians/V, and a modulating signal of $m(t) = 2 \cos(2\pi \times 2000t)$. Determine the peak phase deviation and phase-modulation index.

AKTU 2014-15, Marks 05

Answer

Given : $k_p = 2.5$ radians/V; $m(t) = 2 \cos(2\pi \times 2000t)$

- To Find :**
- Peak phase deviation;
 - Phase modulation index, m_p .

1. Phase modulation index

$$m_p = k_p E_m = 2.5 \times 2 = 5$$

2. Peak phase deviation $= k_p m(t)$

$$= 2.5 [2 \cos (2\pi \times 2000 t)]$$

$$= 5 \cos (2\pi \times 2000 t)$$

$\therefore$ Peak value of phase deviation = 5.

Que 2.17. With the help of block diagram explain working of FM demodulator.

Answer

The FM demodulator performs the extraction in two steps :

1. It converts the FM signal into corresponding AM signal by using frequency dependent circuits, called frequency discriminators.

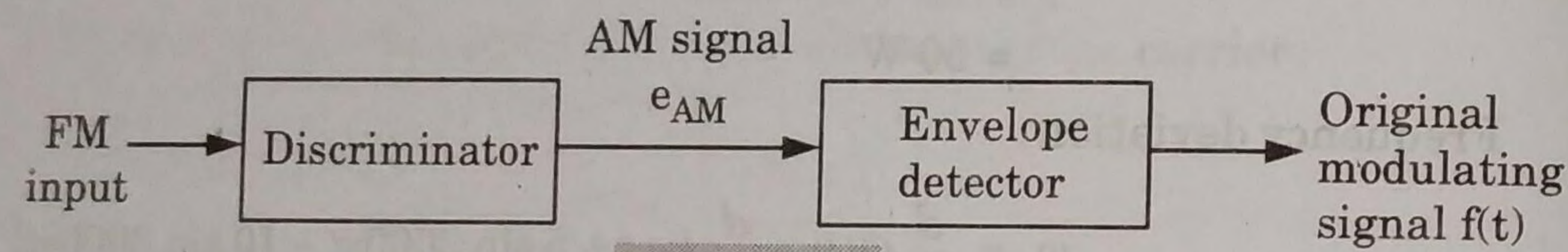


Fig. 2.17.1.

2. The original modulating signal $f(t)$ is recovered from this AM signal using linear diode envelope detector.
3. The discriminator circuit consists of a tuned circuit which is slightly detuned from the carrier frequency ω_c .
4. The circuit converts the FM signal into an AM signal.
5. A small change in the frequency ($\Delta\omega$) of the input signal produces a change in the amplitude of e_{AM} by an amount $e_{AM} = \alpha(\Delta\omega)$.
6. Thus, frequency variations at the input produce corresponding variations at the output.
7. In this way, the FM signal is converted to AM signal, which is detected to recover the modulating signal $f(t)$.

Que 2.18. Explain the working of a balanced frequency discriminator with the help of circuit diagram.

OR

Draw the block diagram of balanced frequency discriminator and explain it for demodulation of FM signal.

Answer

Frequency discriminator :

1. It converts the FM signal into a corresponding AM signal by using frequency dependent circuits, i.e., circuits whose output voltage depends on input frequency (f_i). Such circuits are called as frequency discriminators.

Balanced frequency discriminator :

1. The circuit diagram consists of two LC circuits as shown in Fig. 2.18.1.
2. The two tuned circuits are in the stagger-tuned mode, i.e., one is tuned above the carrier frequency f_c and other is tuned below f_c .

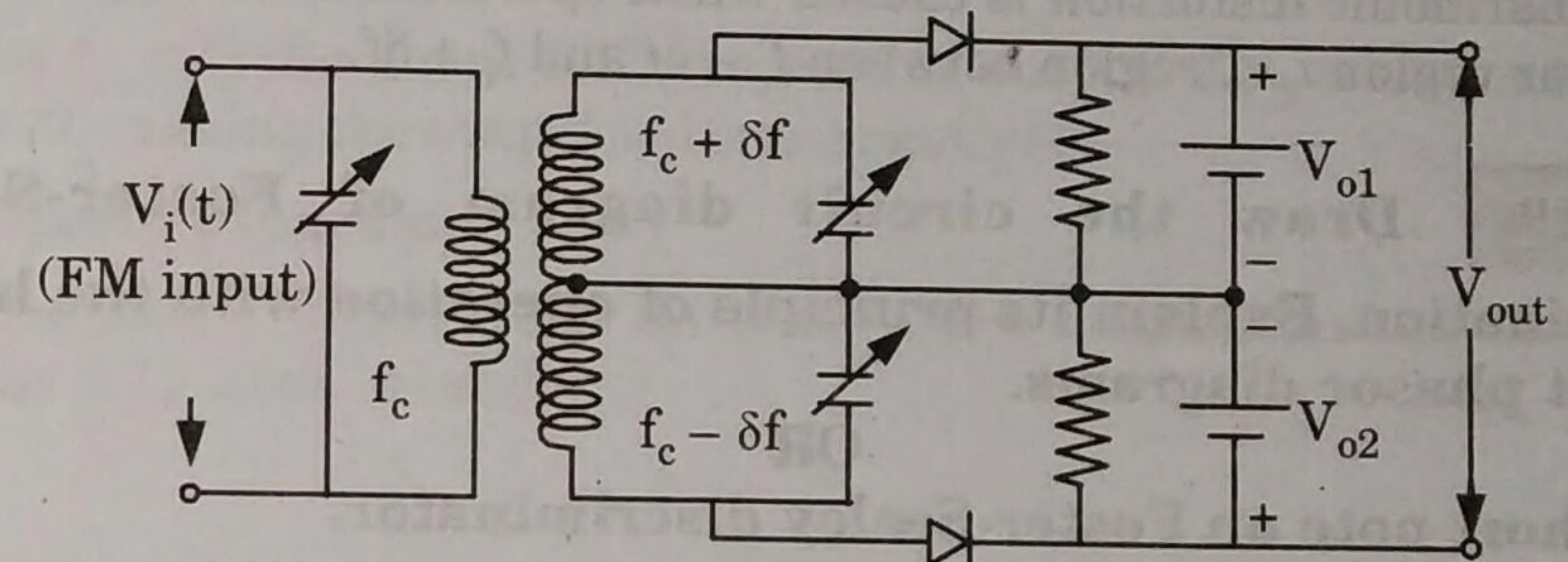


Fig. 2.18.1.

3. Fig. 2.18.2 shows the characteristic of the balanced slope detector (frequency discriminator). It shows V_{out} with respect to input frequency.

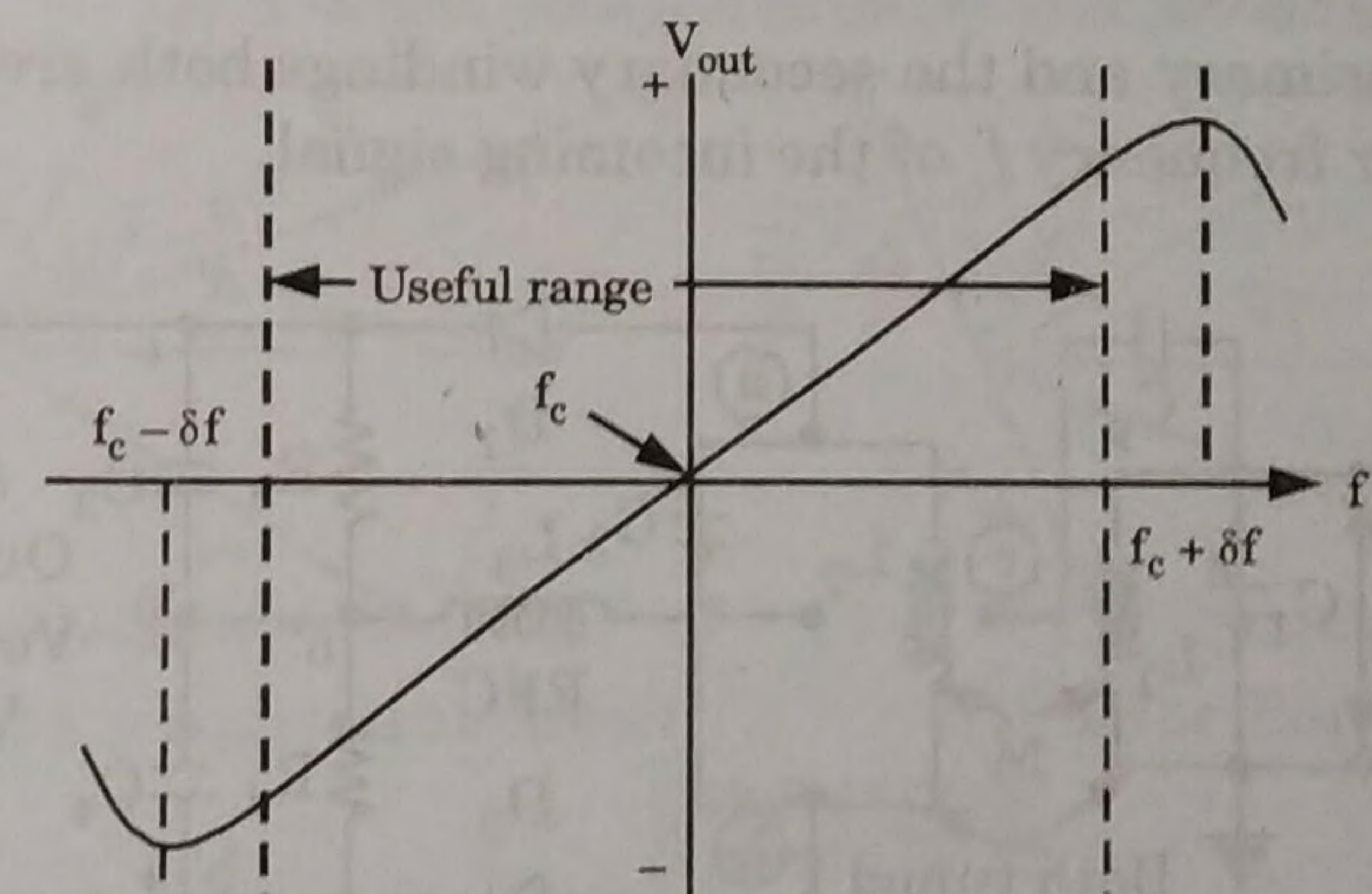


Fig. 2.18.2. Characteristic of balanced slope detector, or 'S'-curve.

4. When $f_i = f_c$
Then, $V_{o1} = V_{o2}$
 $\therefore V_{out} = V_{o1} - V_{o2} = 0$

5. When $f_i = f_c + \delta f$
 $V_{o1} > V_{o2}$
 Hence $V_{out} = V_{o1} - V_{o2}$ is maximum positive for $f_c + \delta f$.
6. When $f_i = f_c - \delta f$
 $V_{o2} > V_{o1}$
 Hence $V_{out} = V_{o1} - V_{o2}$ is maximum negative for $f_c - \delta f$.
7. For other frequencies of input, the output (V_{out}) is produced according to the characteristics shown in Fig. 2.18.2.
 For example if input frequency (f_i) tries to increase above f_c then V_{o1} will be greater than V_{o2} and net output will be positive.
8. No harmonic distortion is caused when operation is restricted to the linear region i.e., region between $f_c - \delta f$ and $f_c + \delta f$.

Que 2.19. Draw the circuit diagram of Foster-Seeley discrimination. Explain its principle of operation with the help of relevant phasor diagrams.

OR

Write short note on Foster-Seeley discriminator.

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Answer

- The circuit diagram of phase discriminator or Foster-Seeley discriminator is as shown in Fig. 2.19.1.
- Here, the primary and the secondary windings both are tuned to the same center frequency f_c of the incoming signal.

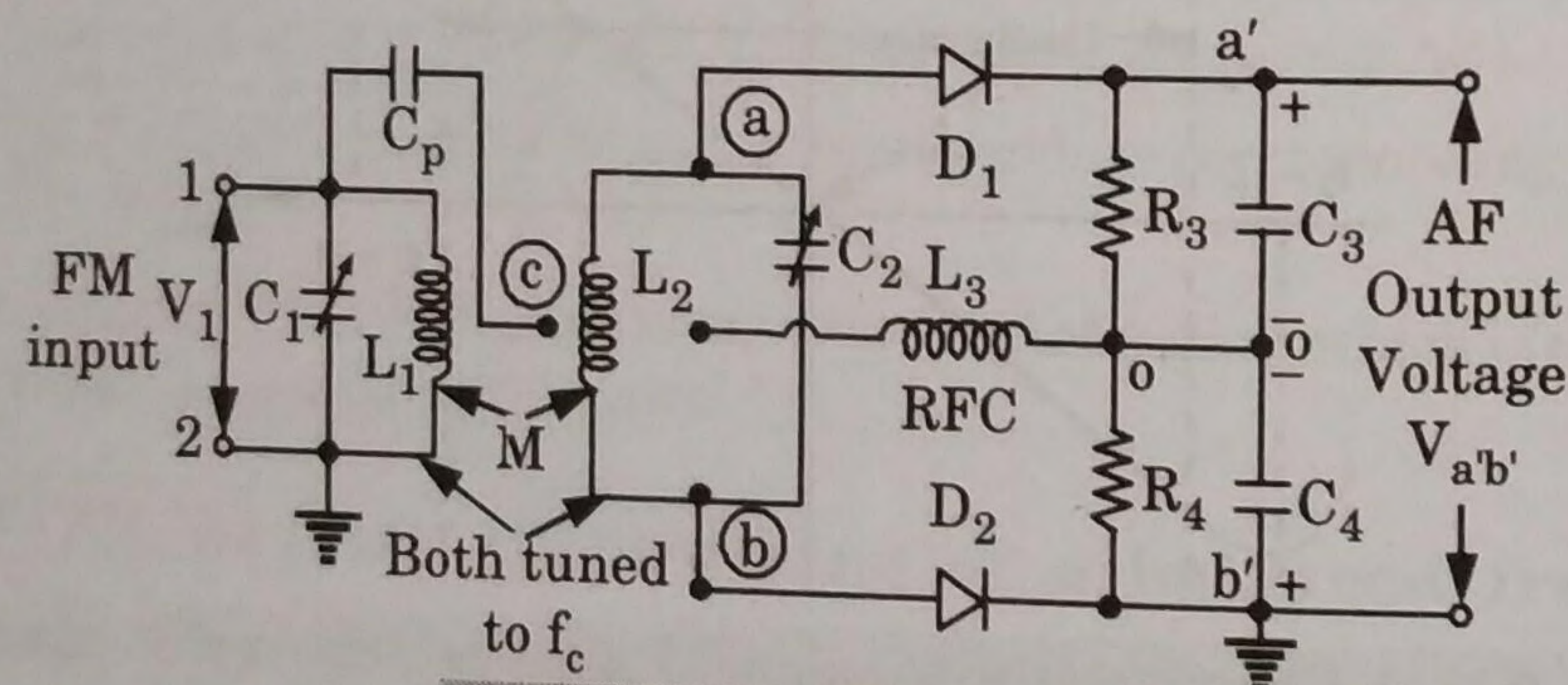


Fig. 2.19.1. Phase discriminator.

Principle of operation :

- Even though the primary and secondary tuned circuits are tuned to the same center frequency, the voltage applied to the two diodes D_1 and D_2 are not constant.
- They vary depending on the frequency of the input signal. This is due to the change in phase shift between the primary and secondary windings depending on the input frequency.
 - At $f_{in} = f_c$, the individual output voltage of the two diodes will be equal and opposite. The output voltage is zero as,

$$V_o = V_{o1} - V_{o2}$$
 - For $f_{in} > f_c$, the phase shift between the primary and secondary winding is such that the output of D_1 is higher than D_2 . Hence, the output voltage will be positive.
 - For $f_{in} < f_c$, the phase shift is such that output of D_2 is higher than that of D_1 making the output voltage negative.

Phasor diagrams :

- The phasor diagrams at different input frequencies are as shown in Fig. 2.19.2 (a), (b) and (c).

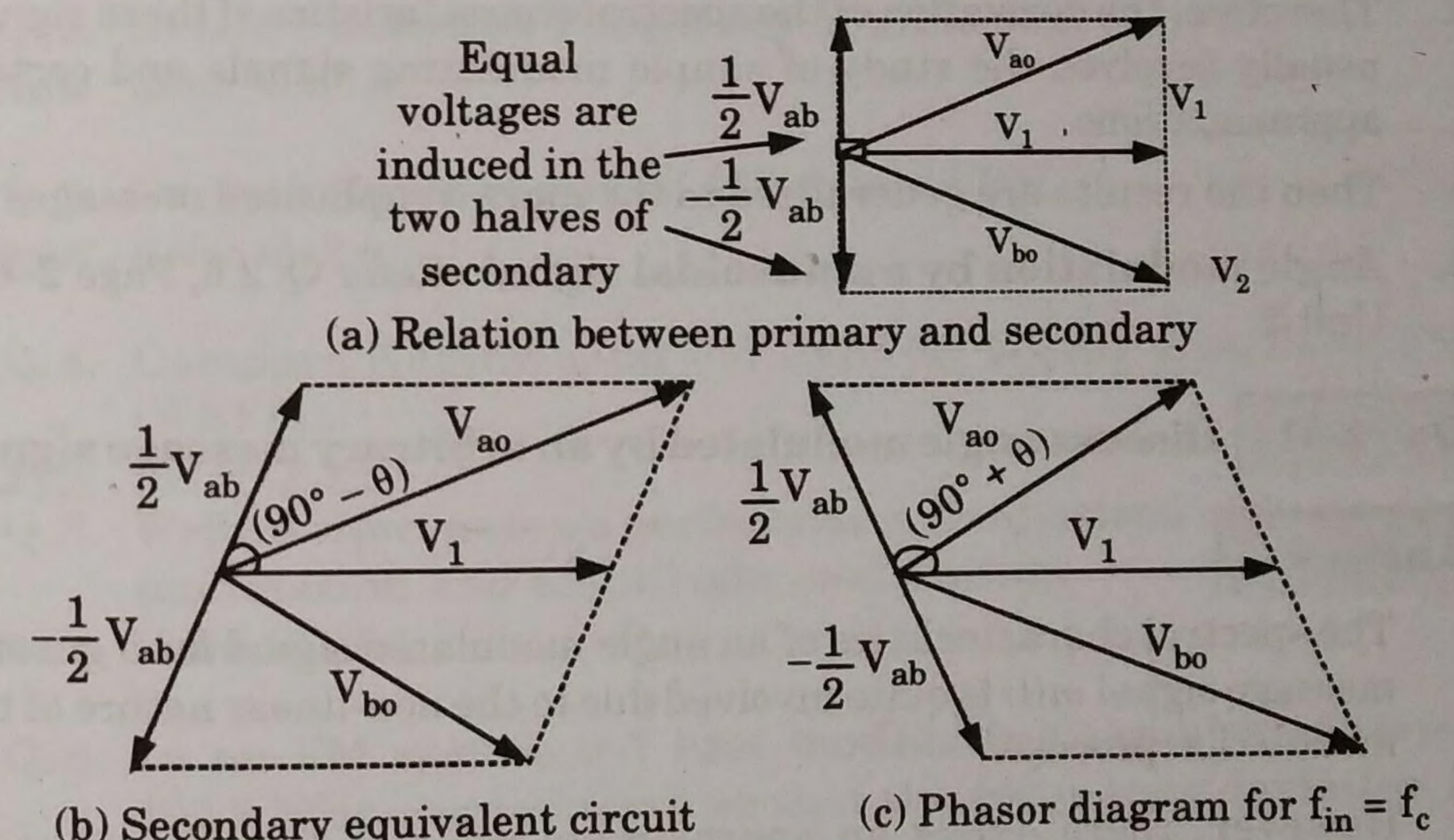


Fig. 2.19.2.

- Because the output is dependent on the primary-secondary phase relationship, this circuit is called as phase discriminator.

PART-2*Spectral Characteristics of Angle Modulated Signal.***Questions-Answers****Long Answer Type and Medium Answer Type Questions**

Que 2.20. Write a short note on spectral characteristics of angle modulated signal. Explain angle modulated by sinusoidal message signal.

Answer**A. Spectral characteristics of angle modulated signals :**

1. Due to the inherent non-linearity of angle modulation systems, the precise characterization of their spectral properties, even for simple message signals, is mathematically intractable.
2. Therefore, the derivation of the spectral characteristics of these signals usually involves the study of simple modulating signals and certain approximations.
3. Then the results are generalized to the more complicated messages.

B. Angle modulation by a sinusoidal signal : Refer Q. 2.6, Page 2-6B, Unit-2.

Que 2.21. Discuss angle modulated by an arbitrary message signal.

Answer

1. The spectral characteristics of an angle modulated signal for a general message signal $m(t)$ is quite involved due to the non-linear nature of the modulation process.
2. However, there exists an approximate relation for the effective bandwidth of the modulated signal. This is known as Carson's rule and is given by

$$\beta_c = 2(\beta + 1) W \quad \dots(2.21.1)$$

where β is the modulation index defined as

$$\beta = \begin{cases} k_p \max [m(t)|], & \text{PM} \\ \frac{k_f \max [|m(t)|]}{W}, & \text{FM} \end{cases} \quad \dots(2.21.2)$$

and W is the bandwidth of the message signal $m(t)$.

3. Since wideband FM has a β with a value that is usually around 5 or more, the bandwidth of an angle modulated signal is much greater than the bandwidth of various amplitude-modulation schemes.
4. The bandwidth of SSB is W and the bandwidth of DSB is $2W$.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Explain angle modulation with its types.

Ans. Refer Q. 2.2.

Q. 2. Relate Phase and Frequency Modulation.

Ans. Refer Q. 2.3.

Q. 3. Explain Narrowband and Wideband FM.

Ans. Refer Q. 2.6.

Q. 4. Compare Narrowband FM (NBFM) versus Wideband FM (WBFM).

Ans. Refer Q. 2.8.

Q. 5. Write a short note on performance comparison of frequency modulation and amplitude modulation.

Ans. Refer Q. 2.11.

Q. 6. In an FM system a 7 kHz modulating signal modulates 107.6 MHz carrier wave so that the frequency deviation is 50 kHz. Determine

- i. Carrier swing in the FM signal and modulation index.
- ii. The highest and lowest frequency attained by the FM signal.

Ans. Refer Q. 2.13.

Q. 7. An angle-modulated signal with carrier frequency $\omega_c = 2\pi \times 10^5$ is described by the equation $\Psi_{FM}(t) = 10 \cos(\omega_c t + 5 \sin 3000\pi t + 10 \sin 2000\pi t)$

- i. Find the power of the modulated signal.

- ii. Find the frequency deviation Δf .
 iii. Estimate bandwidth of the FM signals.

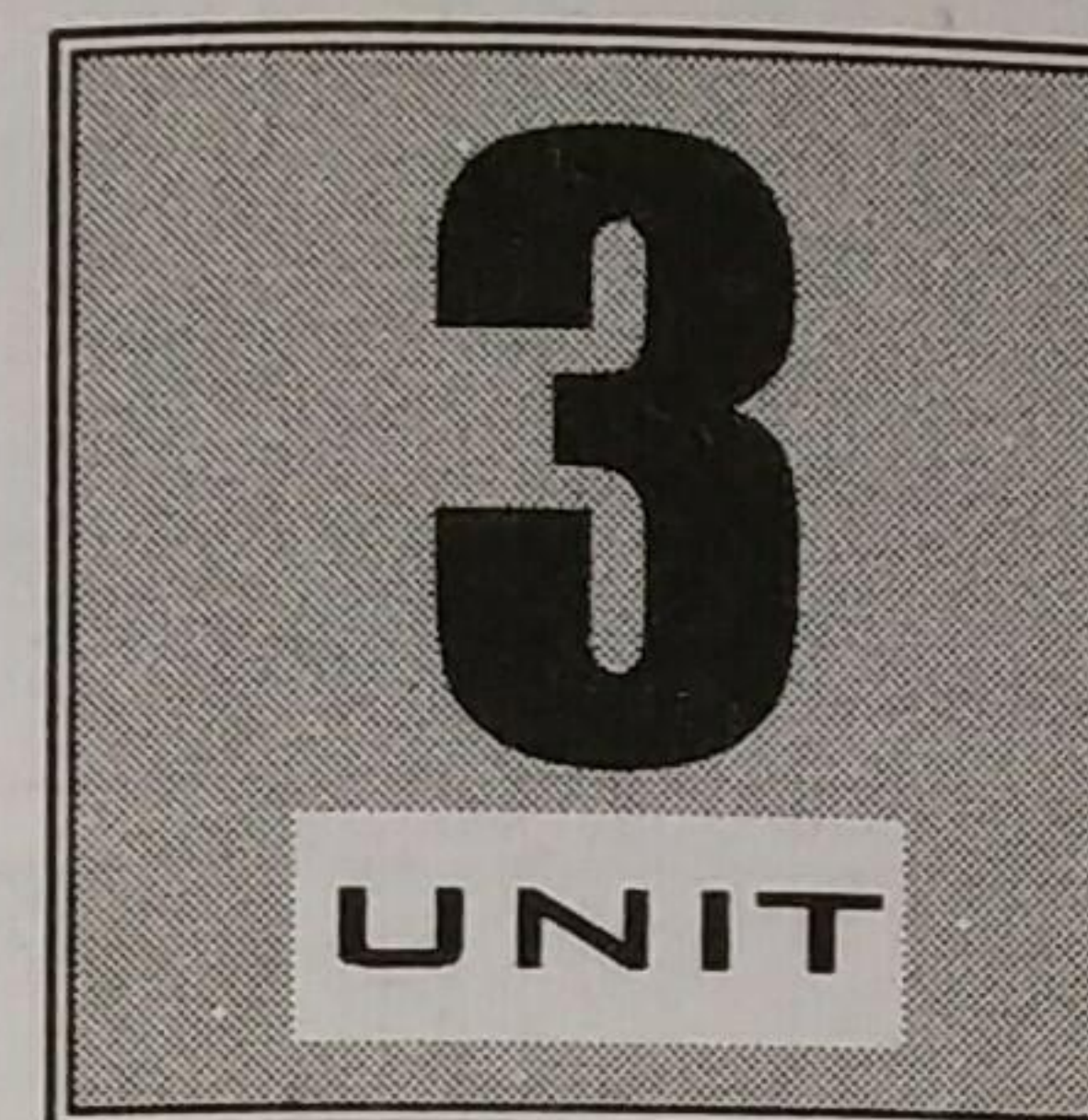
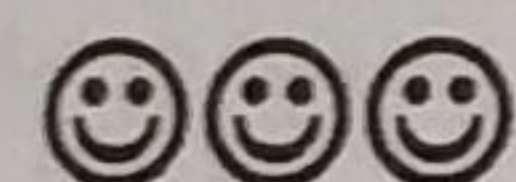
Ans. Refer Q. 2.15.

Q. 8. Draw the circuit diagram of Foster-Seeley discrimination. Explain its principle of operation with the help of relevant phasor diagrams.

Ans. Refer Q. 2.19.

Q. 9. Write a short note on spectral characteristics of angle modulated signal. Explain angle modulated by sinusoidal message signal.

Ans. Refer Q. 2.20.



Noise

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PART-1*Review of Probability and Random Process.***Questions-Answers****Long Answer Type and Medium Answer Type Questions****Que 3.1.** Explain the concept of probability.**Answer****i. Experiment :**

1. The term experiment is used in probability theory to describe a process whose outcome cannot be fully predicted because the condition under which it is performed cannot be predetermined with sufficient accuracy and completeness.

2. Tossing a coin, rolling a die are some of the examples of experiment.

ii. Event :

1. An event is a subset of outcomes that share some of the common characteristics.
2. In the experiment of rolling a die, the event "odd number on a throw" can result from any one of three outcomes (1, 3 and 5).

- iii. **Sample space :** The sample space S can be defined as the collection of all the possible and separately identifiable outcome of an experiment.

Example :

1. Let us consider an example, in an experiment of rolling a die, the sample space consist of six elements represented by six sample points (a_1, a_2, a_3, a_4, a_5 and a_6) where a_i represents the outcome "a number i is thrown".
2. The event, on the other hand, is a subset of S . The event "an odd number is thrown" is denoted by A_o and the event "an even number is thrown" is denoted by A_e .

Therefore, $A_o = (a_1, a_3, a_5)$

$$A_e = (a_2, a_4, a_6)$$

3. Let us denote the event "a number equal to or less than 4 is thrown" as B therefore

$$B = (a_1, a_2, a_3, a_4)$$

4. The complement of B i.e., B^c can be defined as an event containing all the points not present in B .

Thus for an event B , $B^c = (a_5, a_6)$ Similarly, $A_o^c = A_e$ or $A_e^c = A_o$

5. An event that does not contain sample points is known as null event.
6. The union of events A and B , denoted by $A \cup B$, is the event that contains all points in A and B .

Thus, $A_o \cup B = (a_1, a_3, a_5, a_2, a_4)$

$$A_e \cup B = (a_2, a_4, a_6, a_1, a_3)$$

7. In same way, the intersection of events A and B is denoted by $A \cap B$. It is the event that contains point's common to A and B . It can be also denoted by AB .

Therefore, $A_e B = (a_2, a_4)$ and $A_o B = (a_1, a_3)$ **Que 3.2.** Explain conditional probability and independent events.**Answer****A. Conditional probability :**

1. The Probability of one event if another event is already occurred is called a conditional probability.
2. Let there be N trials of an experiment, in which event A occurs n_1 times. Of these n_1 trials, event B occurs n_2 times, n_2 is the number of times that the joint event $A \cap B$ occurs.

$$\text{i.e., } P(A \cap B) = \lim_{N \rightarrow \infty} \left(\frac{n_2}{N} \right) = \lim_{N \rightarrow \infty} \left(\frac{n_1}{N} \right) \left(\frac{n_2}{n_1} \right)$$

$$\text{Here } \lim_{N \rightarrow \infty} \left(\frac{n_1}{N} \right) = P(A)$$

$$\text{and } \lim_{N \rightarrow \infty} \left(\frac{n_2}{n_1} \right) = P(B|A)$$

3. Because B occur n_2 of the n_1 times that A occurred.

Therefore,

$$P(A \cap B) = P(A) P(B|A) \quad \dots(3.2.1)$$

$$P(B|A) = \frac{P(A \cap B)}{P(A)} \text{ provided } P(A) > 0 \quad \dots(3.2.2)$$

$$\text{Similarly } P(A|B) = \frac{P(A \cap B)}{P(B)} \text{ provided } P(B) > 0 \quad \dots(3.2.3)$$

4. Eq. (3.2.2) and (3.2.3) can be written as

$$P(A|B) = \frac{P(A)P(B|A)}{P(B)} \quad \dots(3.2.4)$$

$$P(B|A) = \frac{P(B)P(A|B)}{P(A)}$$

...(3.2.5)

5. Eq. (3.2.4) and eq. (3.2.5) are called Bayes' rule.

B. Independent events :

1. When the occurrence of one event in no way influences the occurrence of the other event i.e., they are entirely independent. Such types of events are known as independent events.

2. The event B is said to be independent of event A if and only if

$$P(A \cap B) = P(A)P(B)$$

...(3.2.6)

3. Then from eq. (3.2.2) we get

$$P(B|A) = P(B)$$

...(3.2.7)

4. The eq. (3.2.7) state that if B is independent of A , then its probability is not affected by event A . It is natural that if event B is independent of event A , then event A is also independent of B .

$$\text{i.e., } P(A|B) = P(A)$$

...(3.2.8)

5. If A and B are mutually exclusive, then A and B cannot occur at same time. This means that they are independent events.

Que 3.3. A binary communication channel, the receiver detects binary pulses with an error probability P_e . What is the probability that out of 100 received digits, no more than four digits are in error?

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Answer

1. Probability (no more than 4 error)

$$= P(\text{no error}) + P(1 \text{ error}) + P(2 \text{ error}) + P(3 \text{ error})$$

$$= (1 - P_e)^{100} + {}^{100}C_1 P_e (1 - P_e)^{99} + {}^{100}C_2 P_e^2 (1 - P_e)^{98} + {}^{100}C_3 P_e^3 (1 - P_e)^{97} + {}^{100}C_4 P_e^4 (1 - P_e)^{96}$$

$$= (1 - 100P_e) + 100P_e(1 - 99P_e) + 4950P_e^2(1 - 98P_e)$$

$$+ 161700P_e^3(1 - 97P_e) + 3921225P_e^4(1 - 96P_e)$$

Que 3.4. In an experiment, a trial consists of four successive tosses of a coin. If we define a random variable x as number of head appearing in a trial, determine $P_X(x)$ and $F_X(x)$.

Answer

1. For a given experiment, the sample space S and the random variable x are shown as :

S	X
HHHH	$x_1 = 4$
HHHT	$x_2 = 3$
HHTH	$x_3 = 3$
HTHH	$x_4 = 3$
THHH	$x_5 = 3$
HHTT	$x_6 = 2$
HTTH	$x_7 = 2$
TTHH	$x_8 = 2$
HTHT	$x_9 = 2$
THTH	$x_{10} = 2$
THHT	$x_{11} = 2$
TTTH	$x_{12} = 1$
TTHT	$x_{13} = 1$
THTT	$x_{14} = 1$
HTTT	$x_{15} = 1$
TTTT	$x_{16} = 0$

2. The probability of each of the 16 outcomes is $\frac{1}{16}$. Then,

$$P(X = 0) = P(x_{16}) = \frac{1}{16}$$

$$P(X = 1) = P(x_{12}) + P(x_{13}) + P(x_{14}) + P(x_{15})$$

$$= \frac{1}{16} + \frac{1}{16} + \frac{1}{16} + \frac{1}{16} = \frac{4}{16} = \frac{1}{4}$$

$$P(X = 2) = P(x_6) + P(x_7) + P(x_8) + P(x_9) + P(x_{10}) + P(x_{11})$$

$$= 6 \times \frac{1}{16} = \frac{3}{8}$$

$$P(X = 3) = P(x_2) + P(x_3) + P(x_4) + P(x_5)$$

$$= 4 \times \frac{1}{16} = \frac{1}{4}$$

$$P(X = 4) = P(x_1) = \frac{1}{16}$$

Now, $F_X(x) = P(X < x) = \begin{cases} 0, & -\infty \leq x < 0 \\ 1/16, & 0 \leq x < 1 \\ 5/16, & 1 \leq x < 2 \\ 11/16, & 2 \leq x < 3 \\ 15/16, & 3 \leq x < 4 \\ 1, & 4 \leq x < \infty \end{cases}$

Que 3.5. Write short note on random process.

Answer

Random process :

1. If the random variable is defined for a time interval $t \in [t_a, t_b]$, then $x(t)$ is a function of time and is random for every instant $t \in [t_a, t_b]$.
2. A random variable that is a function of time is called random process or stochastic process.
3. Thus, a random process is a collection of an infinite number of random variables. Communication signals as well as noises, typically random and varying with time, are characterized by random process.

A. Classification of random process :

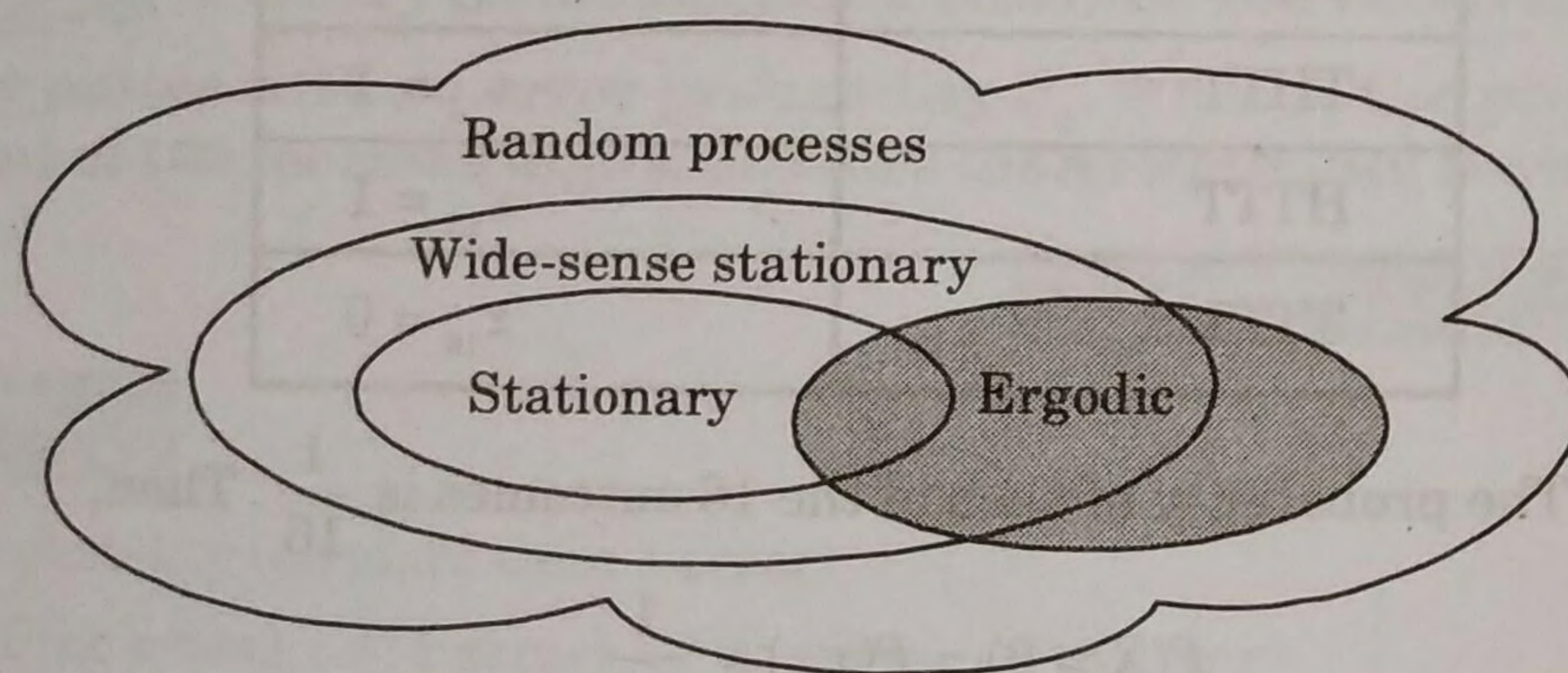


Fig. 3.5.1. Classification of random processes.

a. Stationary random process :

1. A random process whose statistical characteristics do not change with time is known as stationary random process.
2. For a stationary process, we can say that shift of time origin will be impossible to detect, but the process will appear to be the same.
3. Let us determine $p_X(x_j; t_1)$, then shift the origin by t_0 , and then again determine $p_X(x_j; t_1)$.
4. The instant t_1 in the new frame of reference is $t_2 = t_1 + t_0$ in the old frame of reference. Therefore, the PDFs of X at t_1 and $t_2 = t_1 + t_0$ must be same, i.e., $p_X(x; t_1)$ and $p_X(x; t_2)$ must be identical for this process.

5. Therefore, the first order density of a stationary random process can be expressed as

$$p_X(x; t) = p_X(x)$$

6. For a stationary random process, the autocorrelation function $R_X(t_1, t_2)$ must depend on t_1 and t_2 only through the difference $t_2 - t_1$.

Thus, for a real stationary process,

$$R_X(t_1, t_2) = R_X(t_2 - t_1)$$

7. Therefore, $R_X(\tau) = \overline{X(t)X(t+\tau)}$... (3.5.1)

b. Wide-sense (or weakly) stationary process :

1. A process that is not stationary may have a mean value and an autocorrelation function that are independent of the shift of time origin.
2. This means

$$\overline{X(t)} = \text{constant}$$

$$\text{and } R_X(t_1, t_2) = R_X(\tau)$$

$$\text{i.e., } \tau = t_2 - t_1$$

3. Such a process is known as a wide-sense stationary, or weakly stationary. Stationary condition is stronger than wide-sense.

c. Ergodic wide-sense stationary process :

1. Let $\overline{X(t)}$ is the ensemble average of sample function amplitudes at t , and $R_X(t_1, t_2) = \overline{X_1 X_2}$ is the ensemble average of the product of sample function amplitudes $X(t_1)$ and $X(t_2)$.

$$\text{Therefore } \overline{x(t)} = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} x(t) dt$$

2. Similarly, the time autocorrelation function $R_X(\tau)$ can be defined as

$$R_X(\tau) = \overline{x(t)x(t+\tau)}$$

$$= \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} x(t)x(t+\tau) dt$$

3. Thus, for ergodic (wide-sense) stationary process ensemble averages are equal to the time averages of any sample function.
4. For an ergodic process $X(t)$,

$$\overline{X(t)} = \overline{x(t)}$$

$$R_X(\tau) = R_x(\tau)$$

5. In order to understand the ergodic process, all possible ensemble averages are equal to the corresponding time averages of one of its sample function.

Que 3.6. Write short note on power spectral density for wide-sense stationary random process. **AKTU 2017-18, Marks 05**

Answer

1. Let define the PSD $S_X(f)$ of a random process $x(t)$ as the ensemble average of the PSDs of all sample functions. Thus

$$S_X(f) = \lim_{T \rightarrow \infty} \left[\frac{|X_T(f)|^2}{T} \right] \text{ W/Hz} \quad \dots(3.6.1)$$

where $X_T(f)$ is the Fourier transform of the time-truncated random process and bar at top represents ensemble average.

$$x_T(t) = x(t)\Pi(t/T)$$

2. The PSD as defined in eq. (3.6.1) is the Fourier transform of the autocorrelation function $R_X(\tau)$ of the process $x(t)$

$$\text{i.e.,} \quad R_X(\tau) = S_X(f)$$

3. This can be proved as follows,

$$X_T(f) = \int_{-\infty}^{\infty} x_T(t) e^{-j2\pi ft} dt = \int_{-T/2}^{T/2} x(t) e^{-j2\pi ft} dt$$

4. For real $x(t)$,

$$\begin{aligned} |X_T(f)|^2 &= X_T(-f) X_T(f) \\ &= \int_{-T/2}^{T/2} x(t_1) e^{j2\pi ft_1} dt_1 \int_{-T/2}^{T/2} x(t_2) e^{-j2\pi ft_2} dt_2 \\ &= \int_{-T/2}^{T/2} \int_{-T/2}^{T/2} x(t_1) x(t_2) e^{-j2\pi f(t_2 - t_1)} dt_1 dt_2 \end{aligned}$$

and,

$$S_X(f) = \lim_{T \rightarrow \infty} \left[\frac{|X_T(f)|^2}{T} \right]$$

$$= \lim_{T \rightarrow \infty} \left[\frac{1}{T} \int_{-T/2}^{T/2} \int_{-T/2}^{T/2} x(t_1) x(t_2) e^{-j2\pi f(t_2 - t_1)} dt_1 dt_2 \right]$$

5. Interchanging the operation of integration and ensemble averaging,

$$\begin{aligned} S_X(f) &= \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} \int_{-T/2}^{T/2} \overline{x(t_1) x(t_2)} e^{-j2\pi f(t_2 - t_1)} dt_1 dt_2 \\ &= \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} \int_{-T/2}^{T/2} R_X(t_2 - t_1) e^{-j2\pi f(t_2 - t_1)} dt_1 dt_2 \quad \dots(3.6.2) \end{aligned}$$

6. Here, in eq. (3.6.2) we have assume that the process $X(t)$ is at least wide-sense stationary,

so

$$\overline{x(t_1) x(t_2)} = R_X(t_2 - t_1)$$

$$\text{Let } R_X(t_2 - t_1) e^{-j2\pi f(t_2 - t_1)} = \phi(t_2 - t_1) \quad \dots(3.6.3)$$

$$\text{Then,} \quad S_X(f) = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} \int_{-T/2}^{T/2} \phi(t_2 - t_1) dt_1 dt_2 \quad \dots(3.6.4)$$

7. The integral on the right-hand side is a double integral over the range $(-T/2, T/2)$ for each of the variables t_1 and t_2 .
8. The square region of integration in the $t_1 - t_2$ plane is shown in Fig. 3.6.1.

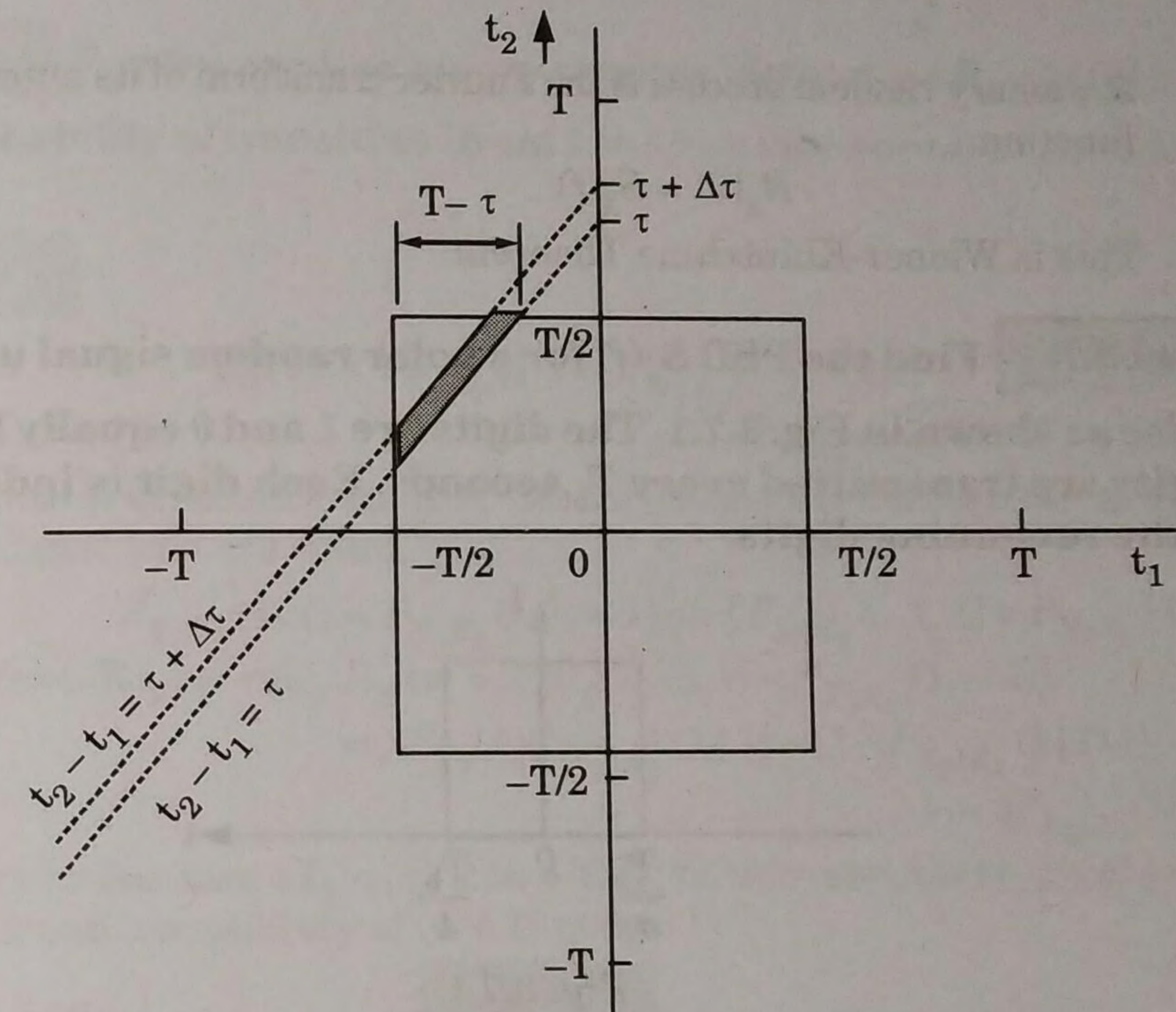


Fig. 3.6.1.

9. The integral in eq. (3.6.4) is a volume under the surface $\phi(t_2 - t_1)$ over the square region.
10. The double integral can be converted to single integral by observing the $\phi(t_2 - t_1)$ is constant along any line $t_2 - t_1 = \tau$ in the $t_1 - t_2$ plane.
11. Let us consider two lines, such that $t_2 - t_1 = \tau$ and $t_2 - t_1 = \tau + \Delta\tau$. If $\Delta\tau \rightarrow 0$, $\phi(t_2 - t_1) \simeq \phi(\tau)$ over the shaded region whose area is $(T - \tau)\Delta\tau$.
12. Hence, the volume under the surface $\phi(t_2 - t_1)$ over the shaded region is $\phi(\tau) (T - \tau) \Delta\tau$. In case τ is negative, the volume would be $\phi(\tau) (T + \tau) \Delta\tau$.
13. Thus, the volume over the shaded region is $\phi(\tau) (T - \tau) \Delta\tau$.

$$\text{Therefore,} \quad S_X(f) = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T}^T \phi(\tau) (T - |\tau|) d\tau$$

$$= \lim_{T \rightarrow \infty} \int_{-T/2}^{T/2} \phi(\tau) \left(1 - \frac{|\tau|}{T} \right) d\tau = \int_{-\infty}^{\infty} \phi(\tau) d\tau$$

14. Provided $\int_{-\infty}^{\infty} |\tau| |\phi(\tau)| d\tau$ is bounded. Substituting eq. (3.6.4) into this equation, we have

$$S_X(f) = \int_{-\infty}^{\infty} R_X(\tau) e^{-j2\pi f\tau} d\tau$$

15. Provided $\int_{-\infty}^{\infty} |\tau| R_X(\tau) e^{-j2\pi f\tau} d\tau$ is bounded. Thus, the PSD of a wide-sense

stationary random process is the Fourier transform of its autocorrelation function.

$$R_X(\tau) = S_X(f)$$

This is Wiener-Khintchine Theorem.

Que 3.7. Find the PSD $S_y(f)$ for a polar random signal use a basic pulse as shown in Fig. 3.7.1. The digits are 1 and 0 equally likely and digits are transmitted every T_b seconds. Each digit is independent of the remaining digits.

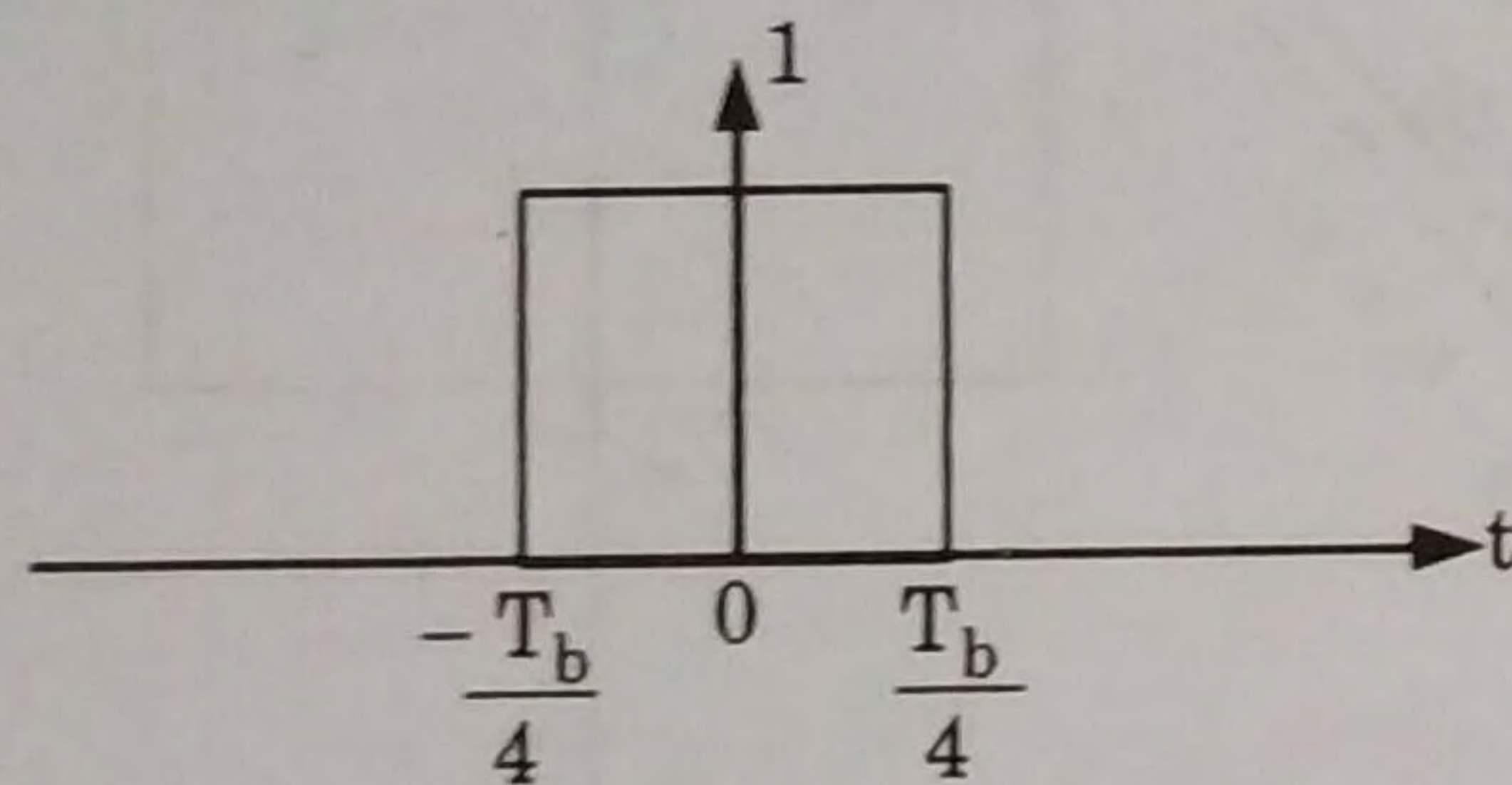


Fig. 3.7.1

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Answer

1. In this case, a_k can take on values 1 and -1 with probability 1/2 each. Hence,

$$\begin{aligned} \overline{a_k} &= \sum_{k=1, -1} a_k P(a_k) = (1)P_{ak}(1) + (-1)P_{ak}(-1) \\ &= \frac{1}{2} - \frac{1}{2} = 0 \end{aligned}$$

$$\begin{aligned} R_0 = \overline{a_k^2} &= \sum_{k=1, -1} a_k^2 P(a_k) = (1)^2 P_{ak}(1) + (-1)^2 P_{ak}(-1) \\ &= \frac{1}{2}(1)^2 + \frac{1}{2}(-1)^2 = 1 \end{aligned}$$

2. Because each digit is independent of the remaining digits,

$$R_n = \overline{a_k a_{k+n}} = \overline{a_k} \overline{a_{k+n}} = 0; \quad n \geq 1$$

3. As, we know that the PSD of $y(t)$, which is the FT of $R_y(\tau)$, is given by,

$$S_y(f) = \frac{|P(f)|^2}{T_b} \sum_{n=-\infty}^{\infty} R_n e^{-jn2\pi f T_b}$$

$$S_y(f) = \frac{|P(f)|^2}{T_b}$$

Que 3.8. For the random binary process, determine $R_X(\tau)$ and $S_X(f)$ if the probability of transition (from 1 to -1 or vice versa) at each node is 0.6.

Answer

1.
$$R_X(\tau) = P_{X_1 X_2}(1, 1) + P_{X_1 X_2}(-1, -1) - P_{X_1 X_2}(-1, 1) - P_{X_1 X_2}(1, -1)$$
2. Because of symmetry the first two terms and the last two terms on the right hand side are equal,

$$\begin{aligned} P_{X_1 X_2}(1, 1) &= P_{X_1 X_2}(-1, -1) \text{ and } P_{X_1 X_2}(-1, 1) = P_{X_1 X_2}(1, -1) \\ \text{and from Bayes' rule, } R_X(\tau) &= 2 [P_{X_1 X_2}(1, 1) - P_{X_1 X_2}(1, -1)] \\ &= 2P_{X_1}(1) [P_{X_2|X_1}(1|1) - (1 - P_{X_2|X_1}(1|1))] \\ &\quad - 2P_{X_2|X_1}(1|1) - 1 \end{aligned}$$

3. Consider the case $nT_b < |\tau| < (n+1)T_b$. In this case, there are at least n nodes and a possibility of $(n+1)$ nodes

$$\text{Prob}[(n+1) \text{ nodes}] = \frac{\tau - nT_b}{T_b} = \frac{\tau}{T_b} - n$$

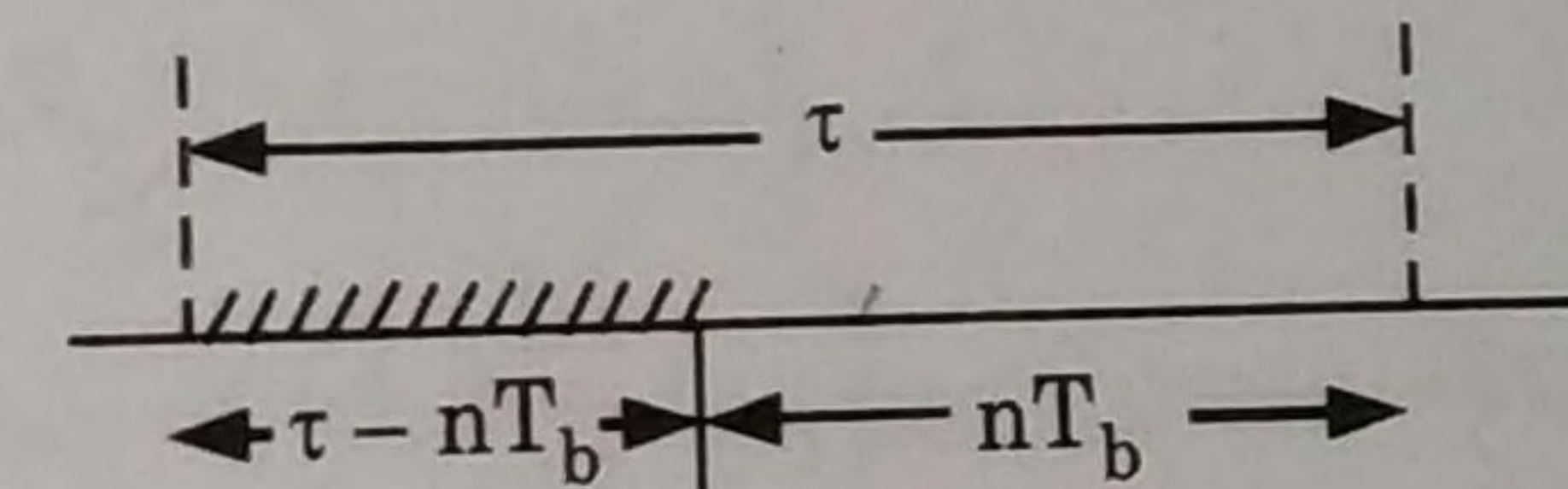


Fig. 3.8.1.

4. $\text{Prob}(n \text{ nodes}) = 1 - \text{Prob}[(n+1) \text{ nodes}] = (n+1) - \frac{\tau}{T_b}$
5. The event $(X_2 = 1 | X_1 = 1)$ can occur if there are N nodes and no state change at any node or state change at only 2 nodes or state change at only 4 nodes, etc.
6. Hence, $P_{X_2|X_1}(1|1) = \text{Prob}[(n+1) \text{ nodes}] \text{ Prob}(\text{state change at even number of nodes}) + \text{Prob}(n \text{ nodes}) \text{ Prob}(\text{state changes at even number of nodes})$
7. The number of ways in which changes at K nodes out of N nodes occur is ${}^N C_K$.

8. Hence, $P_{X_2|X_1}(1|1) = \left[{}^{n+1}C_0(0.6)^0(0.4)^{n+1} + {}^{n+1}C_2(0.6)^2(0.4)^{n-1} + \dots \right]$
 $\left(\frac{|\tau|}{T_b} - n \right) + \left[{}^nC_0(0.6)^0(0.4)^n + {}^nC_2(0.6)^2(0.4)^{n-2} + \dots \right] \left(n+1 - \frac{|\tau|}{T_b} \right)$

and

$$R_X(\tau) = 2P_{X_1|X_2}(1|1) - 1$$

$$R_X(\tau) = 1 - 1.2 \frac{|\tau|}{T_b} \quad ; |\tau| < T_b \quad (n=0)$$

$$= -0.44 + 0.24 \frac{|\tau|}{T_b} \quad ; T_b < |\tau| < 2T_b \quad (n=1)$$

$$= 0.136 - 0.048 \frac{|\tau|}{T_b} \quad ; 2T_b < |\tau| < 3T_b \quad (n=2)$$

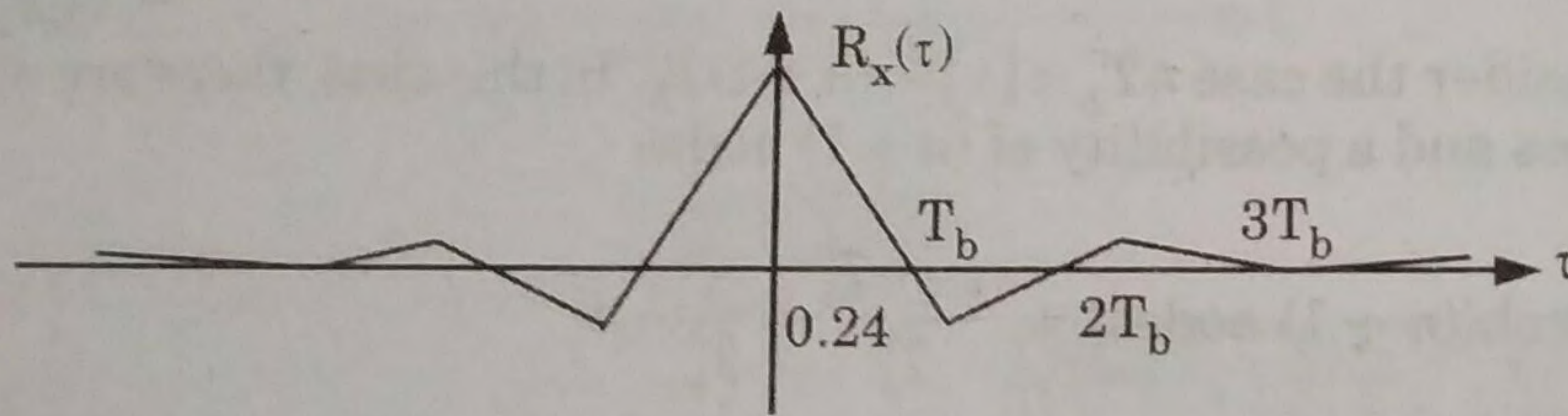
and so on.

9. The PSD can be found by differentiating $R_X(\tau)$ twice. The second derivative $\frac{d^2 R_X(\tau)}{d\tau^2}$ is a sequence of impulses as shown in Fig. 3.8.1.

10. From the time-differentiation property,

$$\frac{d^2}{d\tau^2} R_X(\tau) \leftrightarrow (j\omega)^2 S_X(\omega) = -\omega^2 S_X(\omega).$$

11. Hence, recalling that $\delta(\tau - T) \leftrightarrow e^{-j\omega T}$, we have



(a)

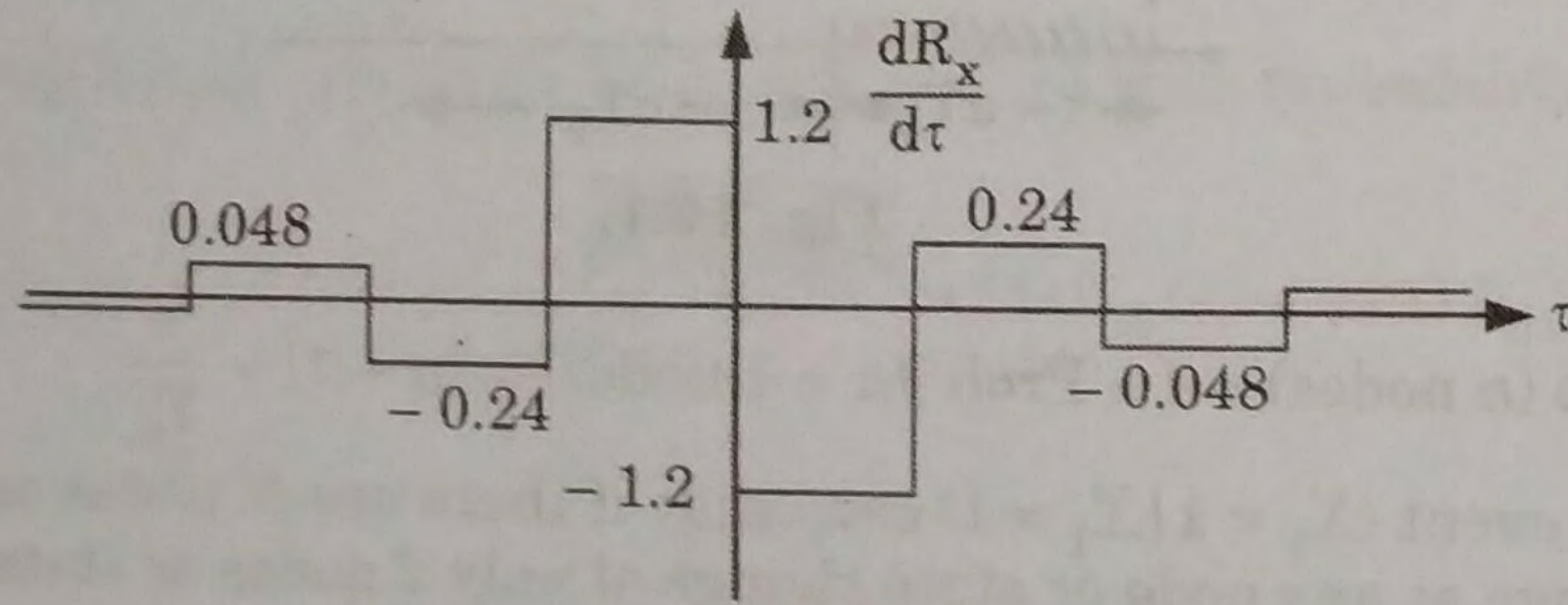
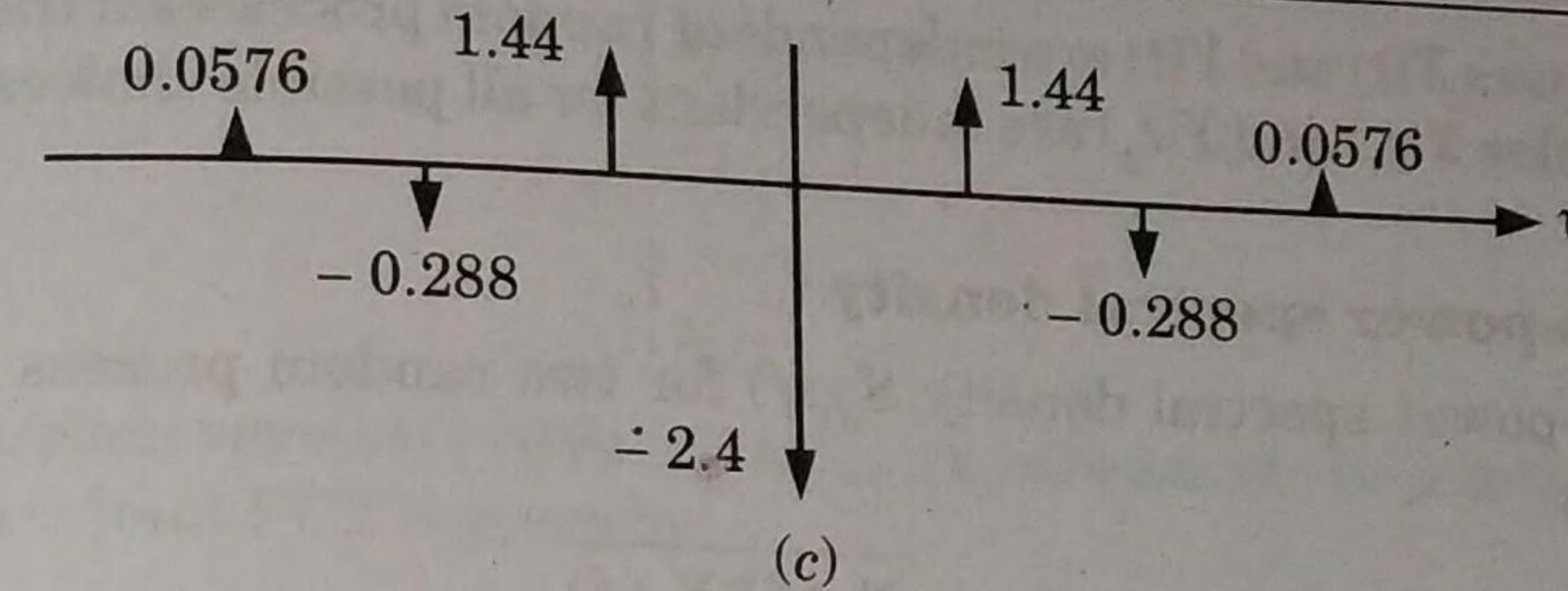
(b) $\frac{d^2 R_X}{d\tau^2}$ 

Fig. 3.8.2.

$$-\omega^2 S_X(\omega) = \frac{1}{T_b} [-2.4 + 1.44(e^{j\omega T_b} + e^{-j\omega T_b}) - 0.288(e^{j2\omega T_b} + e^{-j2\omega T_b}) + \dots]$$

$$S_X(\omega) = \frac{1}{T_b \omega^2} \left[2.4 - \left(2.88 \cos \omega T_b - \frac{1}{5} \cos 2\omega T_b + \frac{1}{25} \cos 3\omega T_b - \frac{1}{125} \cos 4\omega T_b + \dots \right) \right]$$

Que 3.9. Discuss multiple random processes.

Answer

1. For two real random processes $X(t)$ and $Y(t)$, we define the cross-correlation function.

$R_{XY}(t_1, t_2)$ as

$$R_{XY}(t_1, t_2) = \overline{x(t_1)y(t_2)}$$

2. The two processes are said to be jointly stationary if each of the processes is individually wide-sense stationary and if

$$R_{XY}(t_1, t_2) = R_{XY}(t_2 - t_1) \\ = R_{XY}(\tau)$$

i. **Uncorrelated, orthogonal (incoherent), and independent processes:**

1. Two processes $X(t)$ and $Y(t)$ are said to be uncorrelated if their cross-correlation function is equal to the product of their means;

i.e.,

$$R_{XY}(\tau) = \overline{X(t)Y(t+\tau)} = \overline{X} \overline{Y}$$

2. This implies that random variable $X(t)$ and $Y(t+\tau)$ are uncorrelated for all t and τ .

3. Processes $X(t)$ and $Y(t)$ are said to be incoherent or orthogonal,

if $R_{XY}(\tau) = 0$

4. Processes $X(t)$ and $Y(t)$ are independent random processes if the random variables $X(t_1)$ and $Y(t_2)$ are independent for all possible choices of t_1 and t_2 .

ii. **Cross-power spectral density :**

1. Cross-power spectral density $S_{XY}(f)$ for two random process $X(t)$ and $Y(t)$ as

$$S_{XY}(f) = \lim_{T \rightarrow \infty} \frac{\overline{X_T(f)^* Y_T(f)}}{T}$$

where $X_T(f)$ and $Y_T(f)$ are the Fourier transforms of the truncated processes $X(t)\Pi(t/T)$ and $Y(t)\Pi(t/T)$ respectively.

2. Thus, we can say that

$$R_{XY}(\tau) = S_{XY}(f)$$

3. Thus, for real random processes $X(t)$ and $Y(t)$

$$R_{XY}(\tau) = R_{YX}(-\tau)$$

4. Therefore, $S_{XY}(f) = S_{YX}(-f)$

PART-2

Gaussian and White Noise Characteristics.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.10. Give the mathematical expression of white gaussian noise. Explain gaussian random variables.

Answer

1. If the channel noise $n_w(t)$ is white and gaussian, then the expansion coefficients,

$$n_k = \int_0^{T_0} n_w(t) \phi_k(t) dt \quad \dots(3.10.1)$$

are also gaussian. Indeed, $(n_1, n_2, \dots, n_k, \dots)$ are jointly gaussian.

2. Here, we shall provide some fundamentals on gaussian random variables. First, we define a column vector of n random variables as,

$$x = \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_n \end{bmatrix}$$

3. Random variables (RVs) $x_1, x_2, \dots, x_n$ are said to be jointly gaussian if their joint PDF is given by,

$$p_{x_1 x_2 \dots x_n}(x_1, x_2, \dots, x_n) = \frac{1}{(2\pi)^{n/2} \sqrt{\det(K_x)}} \exp \left[-\frac{1}{2} (x - \bar{x})^T K_x^{-1} (x - \bar{x}) \right]$$

where x^T denotes the transpose of x and $\bar{x}$ denotes the mean of x .

...(3.10.2)

where K_x is the $n \times n$ covariance matrix

$$K_x = \overline{(x - \bar{x})(x - \bar{x})^T} = \begin{bmatrix} \sigma_{11} & \sigma_{12} & \dots & \sigma_{1n} \\ \sigma_{21} & \sigma_{22} & \dots & \sigma_{2n} \\ \vdots & \vdots & \ddots & \vdots \\ \sigma_{n1} & \sigma_{n2} & \dots & \sigma_{nn} \end{bmatrix} \quad \dots(3.10.3)$$

and the covariance of x_i and x_j is

$$\sigma_{ij} = \overline{(x_i - \bar{x}_i)(x_j - \bar{x}_j)} \quad \dots(3.10.4)$$

Here, $\det(K_x)$ and K_x^{-1} to denote the determinant and the inverse of matrix K_x , respectively.

4. Gaussian variables are important not only because they are frequently observed, but also because they have certain properties that simplify many mathematical operations that are otherwise impossible or very difficult.

Que 3.11. List the properties of gaussian noise.

Answer

1. The gaussian density is completely specified by only the first and second order statistics $\bar{x}$ and K_x .

$$p_{x_1 x_2 \dots x_n}(x_1, x_2, \dots, x_n) = \frac{1}{(2\pi)^{n/2} \sqrt{\det(K_x)}} \exp \left[-\frac{1}{2} (x - \bar{x})^T K_x^{-1} (x - \bar{x}) \right] \quad \dots(3.11.1)$$

2. If n jointly gaussian variables $x_1, x_2, \dots, x_n$ are uncorrelated, then they are independent. If the n variables are uncorrelated, $\sigma_{ij} = 0$ ($i \neq j$), and K_x reduces to a diagonal matrix. Thus eq. (3.11.1) becomes

$$p_{x_1 x_2 \dots x_n}(x_1, x_2, \dots, x_n) = \prod_{i=1}^n \frac{1}{\sqrt{2\pi\sigma_i^2}} \exp\left[-\frac{(x_i - \bar{x}_i)^2}{2\sigma_i^2}\right] \quad \dots(3.11.2)$$

$$= p_{x_1}(x_1) p_{x_2}(x_2) \dots p_{x_n}(x_n) \quad \dots(3.11.3)$$

As we observed earlier, independent variables are always uncorrelated, but uncorrelated variables are not necessarily independent. For the case of jointly gaussian RVs, however, uncorrelatedness implies independence.

- When $x_1, x_2, \dots, x_n$ are jointly gaussian, all the marginal densities, such as $p_{x_i}(x_i)$, and all the conditional densities, such as $p_{x_i | x_j \dots x_p}(x_i, x_j | x_k, x_l, \dots, x_p)$, are gaussian.
- Linear combinations of jointly gaussian variables are also jointly gaussian. Thus, if we form m variables $y_1, y_2, \dots, y_m$ ($m \leq n$) obtained from

$$y_i = \sum_{k=1}^n a_{ik} x_k \quad \dots(3.11.4)$$

Then, $y_1, y_2, \dots, y_m$ are also jointly gaussian variables.

PART-3

Noise in Amplitude Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.12. Define Noise. Where is it most likely to affect the signals?

Answer

- Noise may be defined as any unwanted introduction of energy tending to interfere with proper reception and reproduction of transmitted signals.
- In other way, noise refers to unwanted signal that tends to disturb the transmission and processing of information signals in a communication system.
- Noise mostly affects the transmission of signal through the channel.
- Almost all noise components are added to message in the channel.
- The source of noise may be internal or external.

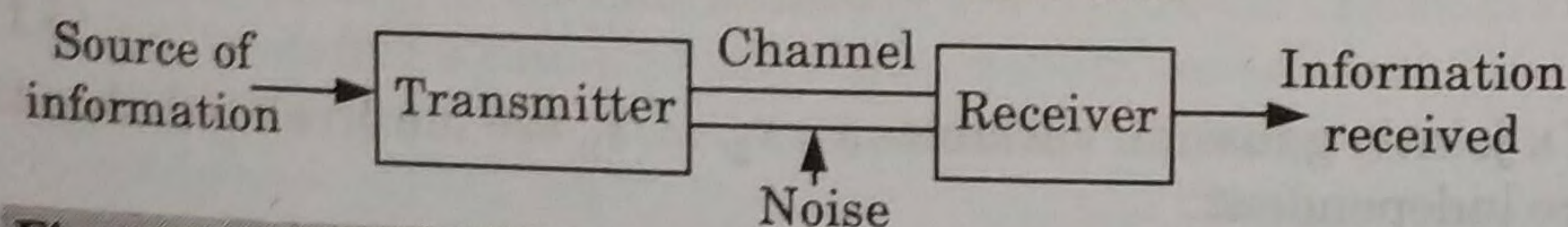


Fig. 3.12.1. Block diagram of communication system showing effect of noise on signal.

Que 3.13. Explain signal-to-noise ratio and its importance in brief.

Answer

Signal to noise ratio :

It is defined as the ratio of signal power to noise power at the same point. Therefore,

$$\frac{S}{N} = \frac{X_s}{X_n} = \frac{V_s^2 / R}{V_n^2 / R} = \left(\frac{V_s}{V_n}\right)^2$$

S = Signal power

N = Noise power

Importance : The comparison of S/N ratio at the input and output of a two-port network provides the noisiness indication of the network.

Que 3.14. Calculate signal to noise ratio for single side band suppressed carrier (SSB-SC).

Answer

- In SSB-SC system, the demodulator is multiplier.
- Here the carrier is $A \cos 2\pi f_c t$.
- For synchronous demodulation, the demodulator must be furnished with a synchronous locally generated carrier $\cos 2\pi f_c t$.
- We assume that the upper sideband is being used, hence the carrier filter has a bandpass that extends from f_c to $f_c + f_M$, where f_M is the baseband bandwidth.

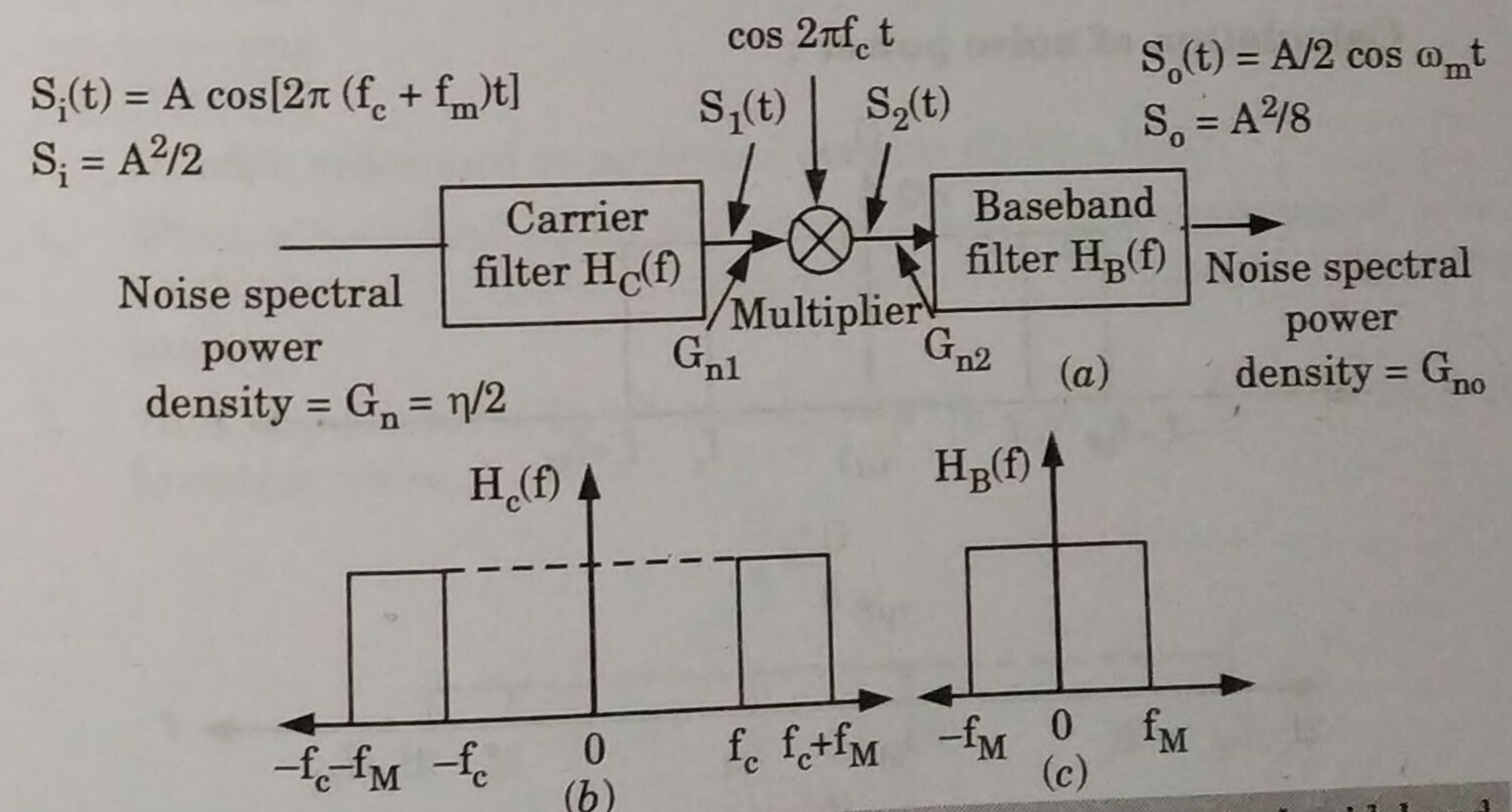


Fig. 3.14.1. (a) A synchronous demodulator operating on a single-sideband single-tone signal, (b) The bandpass range of the carrier filter, (c) The passband of the lowpass baseband filter.

5. The bandwidth of the baseband filter extends from zero to f_M .
6. Let us assume that the baseband signal is a sinusoid of angular frequency f_m ($f_m \leq f_M$). The carrier frequency is f_c , and, since we have assumed that the upper sideband is being used, the received signal is

$$S_i(t) = A \cos [2\pi(f_c + f_m)t] \quad \dots(3.14.1)$$

7. The output of the multiplier is

$$\begin{aligned} S_2(t) &= S_1(t) \cos \omega_c t \\ &= \frac{A}{2} \cos [2\pi(2f_c + f_m)t] + \frac{A}{2} \cos 2\pi f_m t \quad \dots(3.14.2) \end{aligned}$$

Only the difference-frequency term will pass through the baseband filter. Therefore, the output signal is

$$S_o(t) = \frac{A}{2} \cos 2\pi f_m t$$

which is the modulating signal amplified by 1/2.

8. The input signal power is,

$$S_i = \frac{A^2}{2}$$

while the output signal power is,

$$S_o = \frac{1}{2} \left(\frac{A}{2} \right)^2 = \frac{A^2}{8} = \frac{S_i}{4}$$

Thus, $\frac{S_o}{S_i} = \frac{1}{4}$

Calculation of noise power :

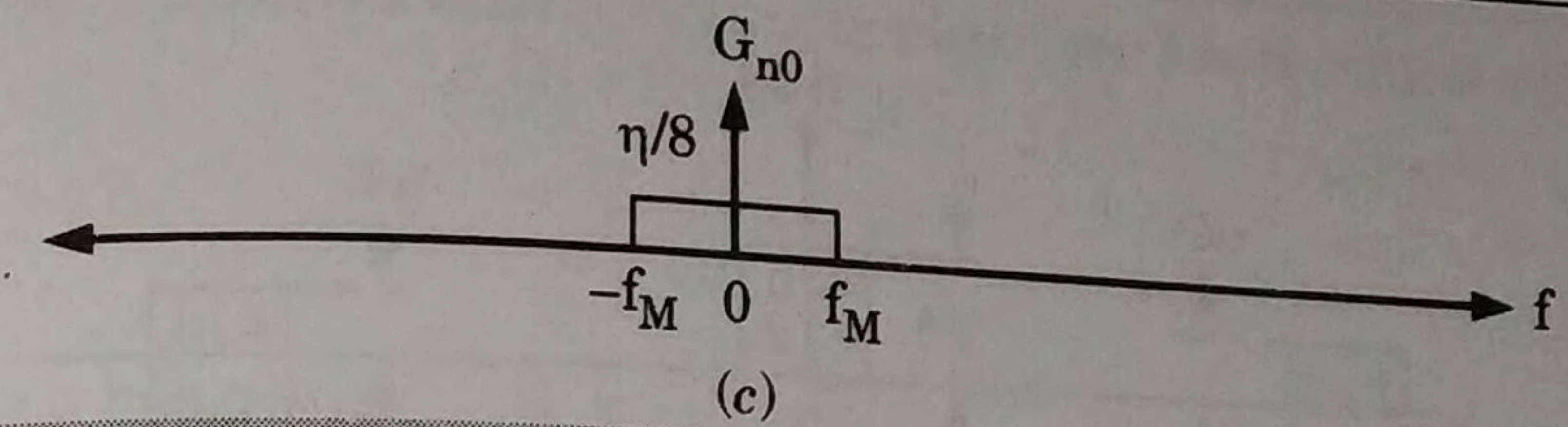
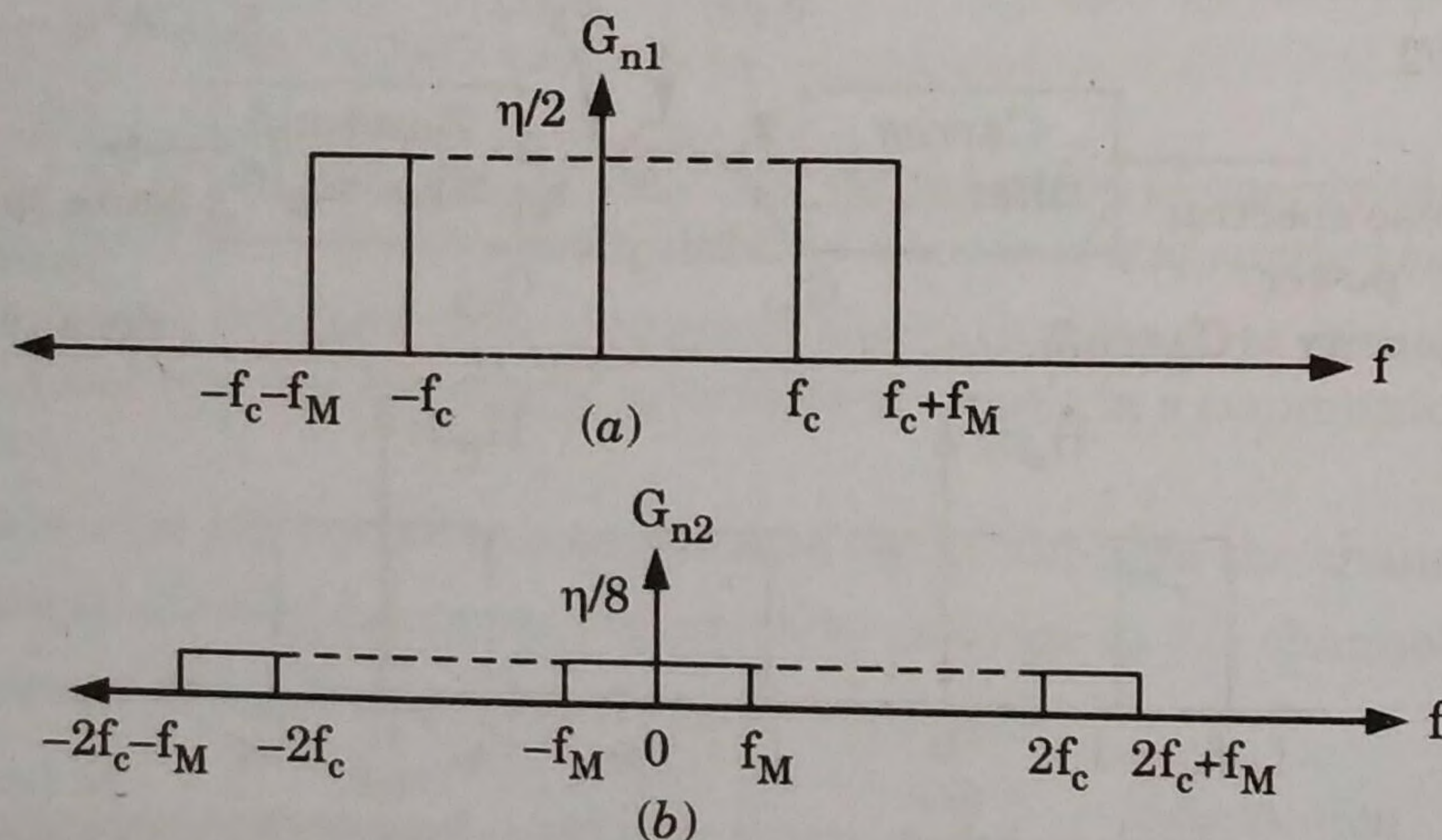


Fig. 3.14.2. Spectral densities of noises in SSB demodulator : (a) Density G_{n1} of noise input to multiplier, (b) Density G_{n2} of noise output of multiplier, (c) Density G_{n0} of noise output of baseband filter.

- When a noise spectral component at a frequency f is multiplied by $\cos 2\pi f_c t$, the original noise component is replaced by two components, one at frequency $f_c + f$ and one at frequency $f_c - f$, each new component having one fourth power of the original.
- The input noise is white and of spectral density $\eta/2$.
- The noise input to multiplier has spectral density G_{n1} . The density of noise after multiplication by $\cos 2\pi f_c t$ is G_{n2} . Finally, the noise transmitted by the baseband filter is of density G_{n0} .

$$N_o = 2f_M \frac{\eta}{8} = \frac{\eta f_M}{4} \quad \dots(3.14.3)$$

4. Thus, signal to noise ratio at the output,

$$\frac{S_o}{N_o} = \frac{S_i / 4}{\eta f_M / 4} = \frac{S_i}{\eta f_M} \quad \dots(3.14.4)$$

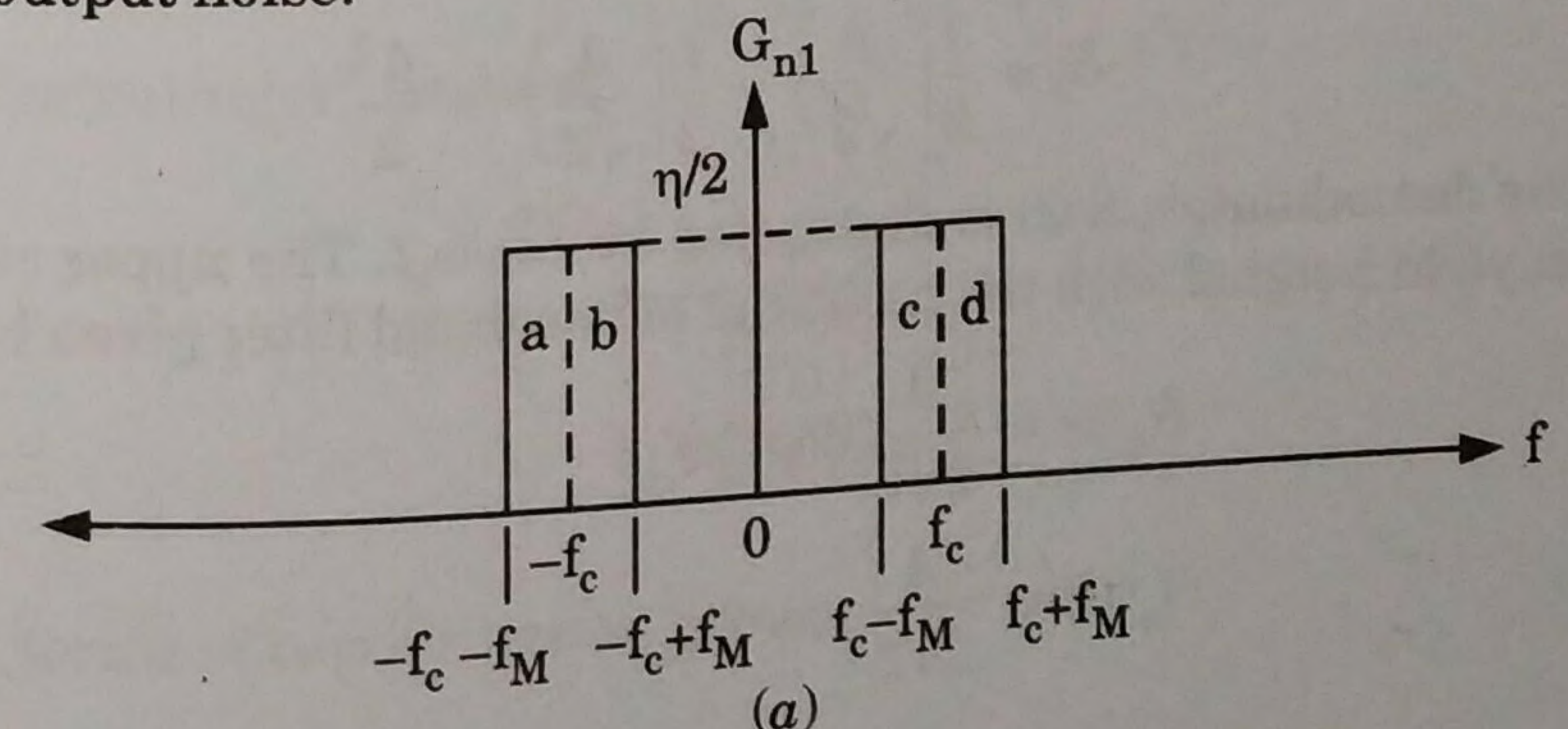
5. The importance of S_o/N_o is that it serves as a figure of merit of the performance of communication system.

Que 3.15. How to calculate signal to noise ratio for DSB-SC.

Answer

Double sideband suppressed carrier (DSB - SC) :

- When a baseband signal of frequency range f_M is transmitted over a DSB-SC system, the bandwidth of the carrier filter must be $2f_M$ rather than f_M .
- Thus, input noise in the frequency range $f_c - f_M$ to $f_c + f_M$ will contribute to output noise.



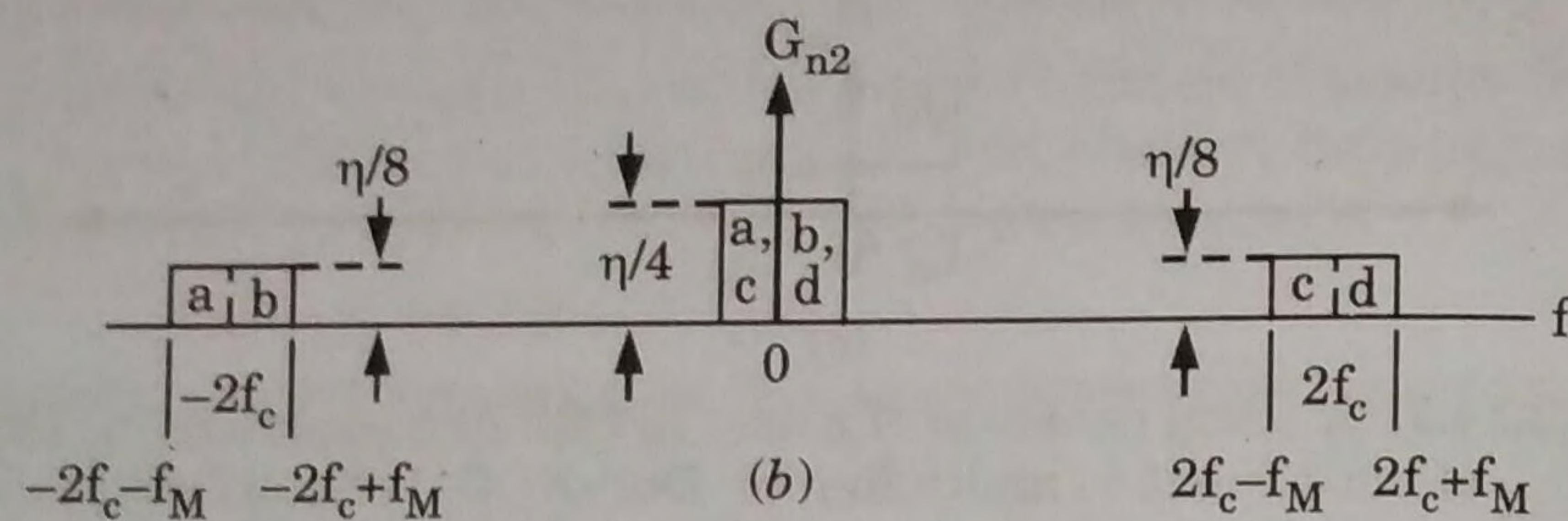


Fig. 3.15.3. Spectral densities of noise in DSB demodulation, (a) Density G_{n1} of noise at output of IF filter, (b) Density G_{n2} of noise output of baseband filter.

Calculation of noise power :

1. It shows the spectral density $G_{n1}(f)$ of the white input noise after the IF filter.
2. The noise is multiplied by $\cos \omega_c t$. The multiplication results in a frequency shift by $\pm f_c$ and the reduction of power in the power spectral density of the noise by a factor of 4.
3. The noise power spectral density in the region between $-f_M$ and $+f_M$ is $\eta/4$.
4. Hence, the output noise power is twice as large as the output noise power for SSB.

$$N_o = \frac{\eta}{4}(2f_M) = \frac{\eta f_M}{2} \quad \dots(3.15.5)$$

Calculation of signal power :

1. Let us assume a sinusoidal baseband signal of frequency $f_m \leq f_M$. To keep the received power the same as in SSB case i.e., $S_i = \frac{A^2}{2}$, we write

$$\begin{aligned} S_i(t) &= \sqrt{2} A \cos 2\pi f_m t \cos 2\pi f_c t \\ &= \frac{A}{\sqrt{2}} \cos [2\pi(f_c + f_m)t] + \frac{A}{\sqrt{2}} \cos [2\pi(f_c - f_m)t] \end{aligned}$$

2. The received power is,

$$S_i = \frac{1}{2} \left(\frac{A}{\sqrt{2}} \right)^2 + \frac{1}{2} \left(\frac{A}{\sqrt{2}} \right)^2 = \frac{A^2}{2}$$

3. In the demodulator, $S_i(t)$ is multiplied by $\cos \omega_c t$. The upper sideband term yield a signal with the passband of baseband filter given by

$$S_o'(t) = \frac{A}{2\sqrt{2}} \cos 2\pi f_m t \quad \dots(3.15.6)$$

$$S_o''(t) = \frac{A}{2\sqrt{2}} \cos 2\pi f_m t \quad \dots(3.15.7)$$

4. Observe that $S_o'(t)$ and $S_o''(t)$ are in phase and hence output signal is

$$\begin{aligned} S_o(t) &= S_o'(t) + S_o''(t) \\ &= \frac{A}{\sqrt{2}} \cos 2\pi f_m t \end{aligned}$$

which has power, $S_o = \frac{A^2}{4} = \frac{S_i}{2}$

5. Therefore, signal to noise ratio for DSB-SC will be

$$\frac{S_o}{N_o} = \frac{S_i}{\eta f_M}$$

Que 3.16. Derive the expression of signal to noise ratio for DSB-C.

Answer

1. Let us consider the case where a carrier accompanies the double sideband signal. Demodulation is achieved synchronously as in SSB-SC and DSB-SC.
2. The carrier is used as a transmitted reference to obtain reference signal $\cos \omega_c t$. Here we replace S_i by $S_i^{(SB)}$, where $S_i^{(SB)}$ is the power in the sideband alone. Then,

$$\frac{S_o}{N_o} = \frac{S_i^{(SB)}}{\eta f_M}$$

3. Suppose that the received signal is

$$\begin{aligned} S_i(t) &= A[1 + m(t)] \cos 2\pi f_c t \\ &= A \cos 2\pi f_c t + A m(t) \cos 2\pi f_c t \end{aligned}$$

where, $m(t)$ is the baseband signal which amplitude-modulates the carrier $A \cos 2\pi f_c t$. The carrier power is $(A^2/2)$. The sidebands are contained in the term $A m(t) \cos 2\pi f_c t$.

4. The power associated with term is $(A^2/2) \overline{m^2(t)}$, where $\overline{m^2(t)}$ is the time average of the square of the modulating waveform.
5. Thus, the total input power S_i is given by

$$S_i = \frac{A^2}{2} + S_i^{(SB)} = \frac{A^2}{2} [1 + \overline{m^2(t)}]$$

Eliminating A^2 , we have

$$S_i^{(SB)} = \frac{\overline{m^2(t)}}{1 + \overline{m^2(t)}} S_i$$

$$\frac{S_o}{N_o} = \frac{\overline{m^2(t)}}{1 + \overline{m^2(t)}} \frac{S_i}{\eta f_M}$$

In terms of carrier power, $P_c = \frac{A^2}{2}$ we get,

$$\frac{S_o}{N_o} = \overline{m^2(t)} \frac{P_c}{\eta f_M}$$

6. If the modulation is sinusoidal with

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

$$S_i(t) = A(1 + m \cos 2\pi f_m t) \cos 2\pi f_c t$$

In this case, $\overline{m^2(t)} = \frac{m^2}{2}$

and $\frac{S_o}{N_o} = \frac{m^2}{2 + m^2} \frac{S_i}{\eta f_M}$

PART-4

Noise in Frequency Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.17. Calculate signal to noise ratio for an FM system.

OR

Define and explain signal to noise ratio. Describe methods to calculate noise in AM and FM systems. **AKTU 2018-19, Marks 10**

Answer

Signal to noise ratio : Refer Q. 3.13, Page 3-17B, Unit-3.

Noise in AM system : Refer Q. 3.16, Page 3-21B, Unit-3.

Noise in FM system :

1. Let us consider that input signal to the IF carrier filter

$$S_i(t) = A \cos \left[\omega_c t + k \int_{-\infty}^t m(\lambda) d\lambda \right] \quad \dots(3.17.1)$$

where, $m(t)$ is the frequency modulating baseband waveform.

2. We assume that the signal is embedded in additive white gaussian noise of power-spectral density $\eta/2$.
3. The IF carrier filter has a bandwidth $B = 2\Delta f + 2f_M$.
4. This filter passes the signal with negligible distortion and eliminates all noise outside the bandwidth B .
5. When the signal-to-noise ratio is high, the noise does not affect the output signal power.

6. In calculating the output-signal power we shall therefore ignore the noise.
7. When the signal $S_i(t)$ arrives at the output of limiter then,

$$S_2(t) = A_L \cos \left[\omega_c t + k \int_{-\infty}^t m(\lambda) d\lambda \right] \quad \dots(3.17.2)$$

Thus, $\phi(t) = k \int_{-\infty}^t m(\lambda) d\lambda$

8. For the output of the discriminator

$$S_4(t) = \alpha \omega_c + \alpha k m(t)$$

Thus, the output signal $S_0(t) = \alpha k m(t)$ and output-signal power is

$$S_o = \alpha^2 k^2 \overline{m^2(t)} \quad \dots(3.17.3)$$

Output noise power :

1. Let us now calculate the noise output of the FM discriminator which results from the presence at the input of white noise having a power spectral density $\eta/2$. Let we set $m(t) = 0$ i.e., the noise output is approximately independent of $m(t)$.
2. At the limiter input,

$$V_i(t) = A \cos \omega_c t + n_c(t) \cos \omega_c t - n_s(t) \sin \omega_c t$$

$$= [A + n_c(t)] \cos \omega_c t - n_s(t) \sin \omega_c t \quad \dots(3.17.4)$$

3. Note that the phasor representing $n_c(t) \cos \omega_c t$ is in phase with the carrier phasor $A \cos \omega_c t$. The phasor representing $n_s(t) \sin \omega_c t$ has an amplitude $n_s(t)$ and is in phase quadrature with other two terms.

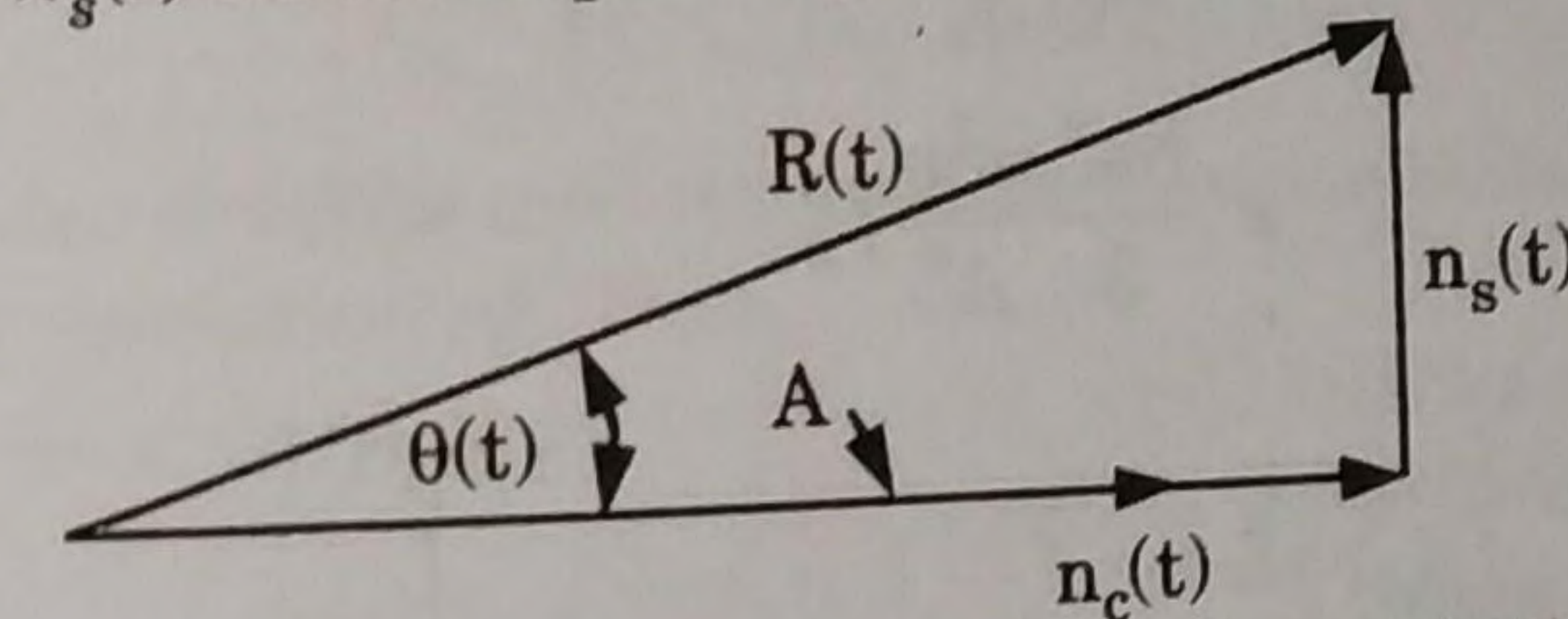


Fig. 3.17.1. A phasor diagram of the term in eq. (3.17.5).

4. The envelope $R(t)$ is easily computed i.e.,

$$R(t) = \sqrt{[A + n_c(t)]^2 + [n_s(t)]^2} \quad \dots(3.17.5)$$

5. Similarly, the phase $\theta(t)$ is

$$\theta(t) = \tan^{-1} \frac{n_s(t)}{A + n_c(t)} \quad \dots(3.17.6)$$

6. Thus, signal and noise forming $V_i(t)$ is

$$V_i(t) = R(t) \cos [\omega_c t + \theta(t)] \quad \dots(3.17.7)$$

7. Thus, the output of the limiter-bandpass filter is therefore

$$V_2(t) = A_L \cos [\omega_c t + \theta(t)]$$

where, A_L is determined by limiter and is constant.

8. Let us assume that,

$$\theta(t) \approx \frac{n_s(t)}{A}$$

Then,

$$V_2(t) = A_L \cos \left[\omega_c t + \frac{n_s(t)}{A} \right] \quad \dots(3.17.8)$$

9. Compare eq. (3.17.8) with,

$$V_4(t) = \alpha \omega_c + \alpha \frac{d}{dt} \phi(t)$$

10. The resulting equation will be

$$V_4(t) = \alpha \left[\omega_c + \frac{1}{A} \frac{d}{dt} n_s(t) \right]$$

Avoiding DC term

$$V_4(t) = \frac{\alpha}{A} \frac{d}{dt} n_s(t)$$

11. Therefore, the spectral density of $n_4(t)$ is $G_{n4}(f)$ given by

$$G_{n4}(f) = \frac{\alpha^2 \omega^2}{A^2} \eta, \quad |f| \leq \frac{B}{2}$$

12. Thus, the output noise power N_o

$$N_o = \int_{-f_M}^{f_M} G_{n4}(f) df \quad \dots(3.17.9)$$

$$= \frac{\alpha^2 \eta}{A^2} \int_{-f_M}^{f_M} 4\pi^2 f^2 df$$

$$= \frac{8\pi^2}{3} \frac{\alpha^2 \eta}{A^2} f_M^3 \quad \dots(3.17.10)$$

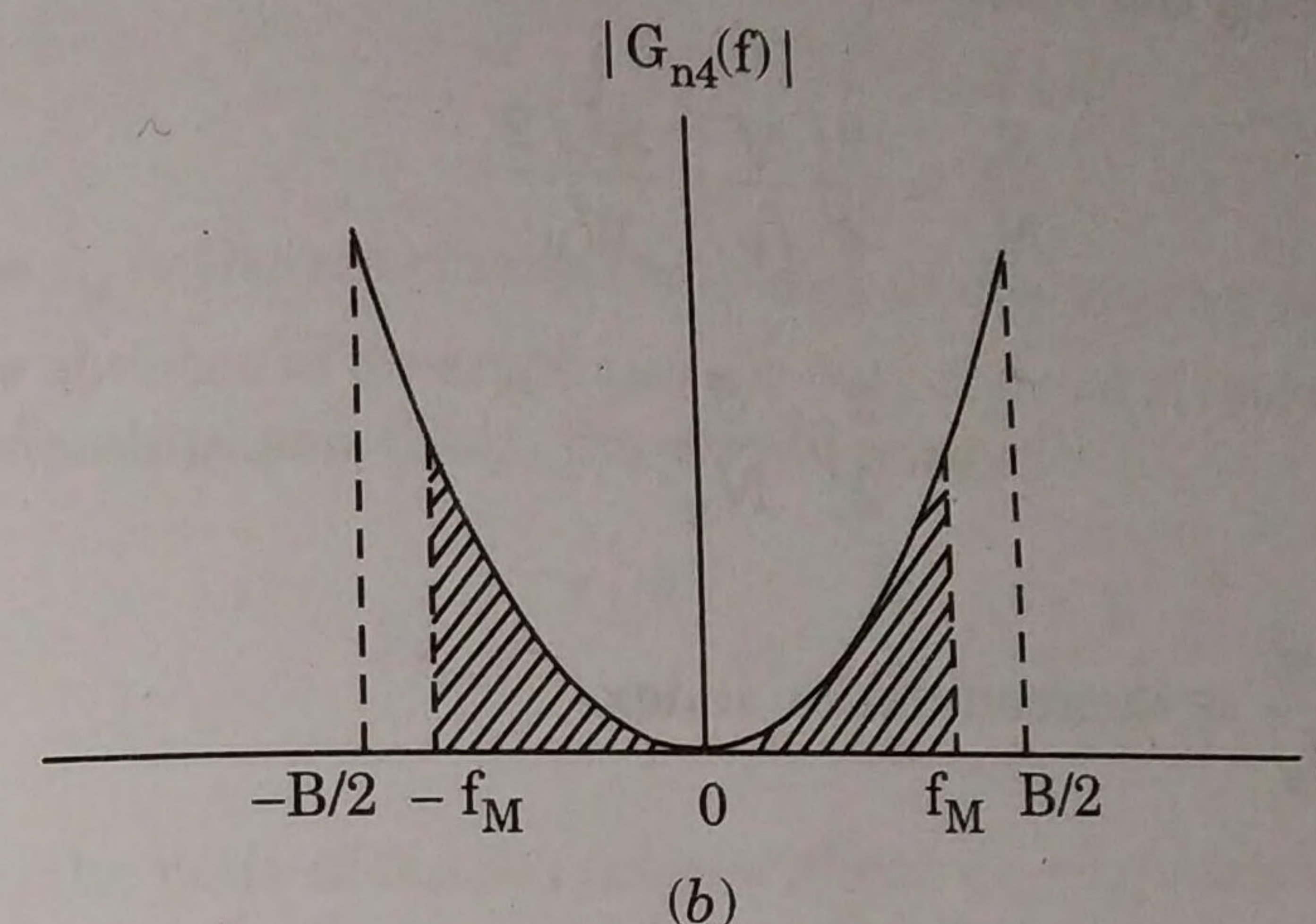
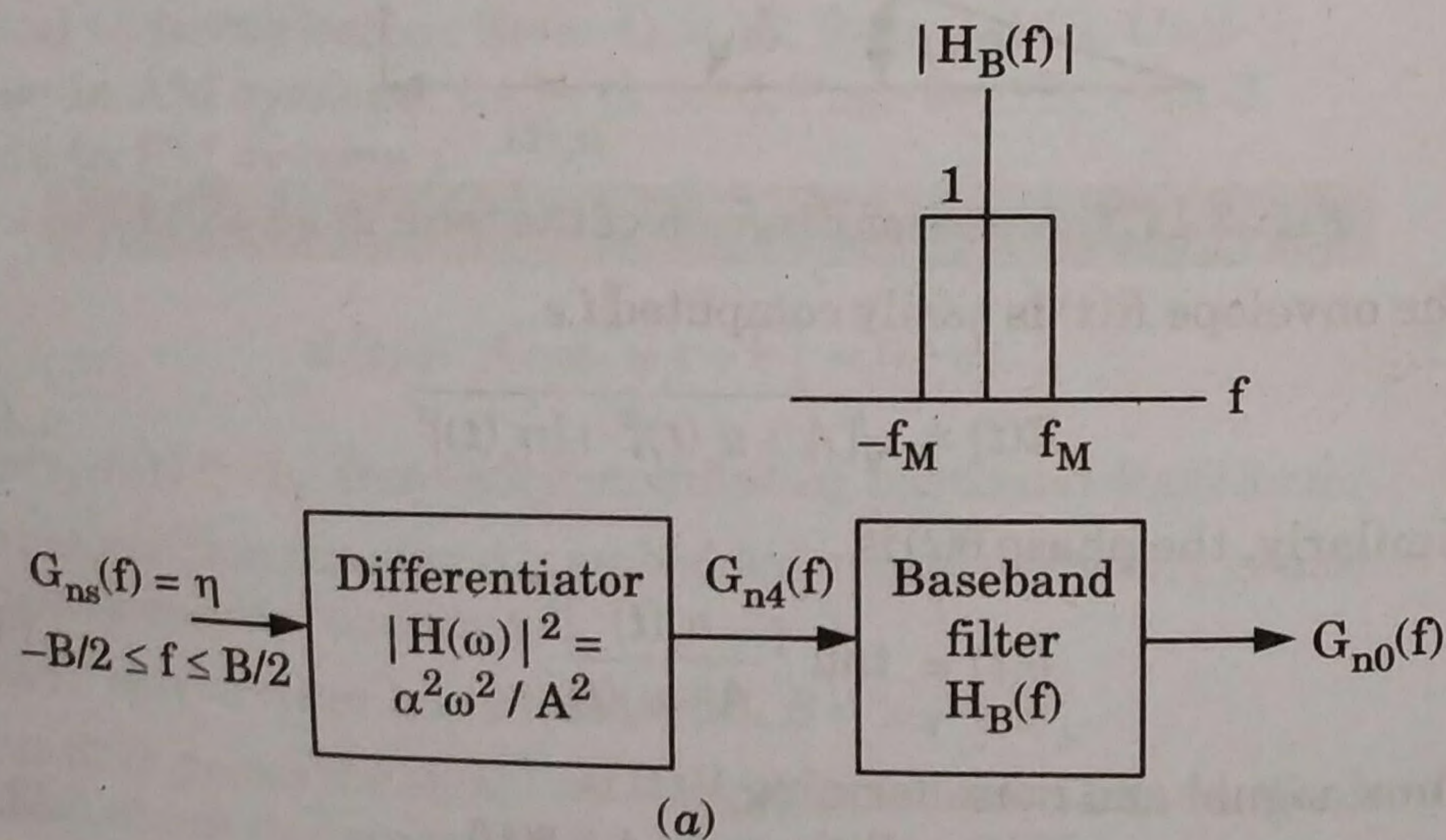


Fig. 3.17.2.

Output SNR :

1. The output signal-to-noise ratio can be computed as

$$\begin{aligned} \frac{S_o}{N_o} &= \frac{\alpha^2 k^2 \overline{m^2(t)}}{\left(\frac{8\pi^2}{3} \right) \left(\frac{\alpha^2 \eta}{A^2} \right) f_M^3} \\ &= \frac{3}{4\pi^2} \frac{k^2 \overline{m^2(t)} A^2 / 2}{f_M^2 \eta f_M} \quad \dots(3.17.11) \end{aligned}$$

2. Let us consider that the modulating signal $m(t)$ is sinusoidal and produces a frequency deviation Δf . Thus,

$$S_i(t) = A \cos \left[\omega_c t + \frac{\Delta f}{f_m} \sin 2\pi f_m t \right] \quad \dots(3.17.12)$$

where, f_m is the modulating frequency.

3. Comparing equation $S_i(t) = A \cos \left[\omega_c t + k \int_{-\infty}^t m(\lambda) d\lambda \right]$ with eq. (3.17.12),

we get differentiating argument

$$\text{i.e.,} \quad km(t) = 2\pi \Delta f \cos 2\pi f_m t$$

$$k^2 \overline{m^2(t)} = \frac{4\pi^2 (\Delta f)^2}{2} = 2\pi^2 (\Delta f)^2$$

4. Then putting the value of $k^2 \overline{m^2(t)}$ in eq. (3.17.11)

$$\frac{S_o}{N_o} = \frac{3}{2} \left(\frac{\Delta f}{f_M} \right)^2 \frac{A^2 / 2}{\eta f_M}$$

$$= \frac{3}{2} \beta^2 \frac{S_i}{N_M}$$

Here $\beta = \frac{\Delta f}{f_M}$ is the modulation index.

$$\gamma_{FM} = \frac{\frac{S_o}{N_o}}{\frac{S_i}{N_M}} = \frac{3}{2} \beta^2$$

PART-5

Pre-emphasis and De-emphasis.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.18. Discuss SNR improvement using pre-emphasis.

Answer

1. We know that the bandwidth occupied by the output of an FM modulator is fixed if normalized power of modulating signal is kept fixed.
2. The normalized power of the base band signal $m(t)$ must be the same as the normalized power of the pre-emphasized signal $m_p(t)$.
3. If $G_m(f)$ is the power spectral density of $m_p(t)$, then the power spectral density of $m_p(t)$ is $|H_p(f)|^2 G_m(f)$ and thus

$$P_m = \int_{-f_M}^{f_M} G_m(f) df$$

$$= \int_{-f_M}^{f_M} |H_p(f)|^2 G_m(f) df$$

where f_M is the maximum frequency of modulating signal.

4. In the absence of de-emphasis the output noise is represented by N_o and with de-emphasis filter, the output noise is

$$N_{od} = \left(\frac{\alpha}{A} \right)^2 4\pi^2 \eta \int_{-f_M}^{f_M} f^2 \left| \frac{1}{H_p(f)} \right|^2 df$$

5. Then, the ratio of output noise without de-emphasis to the output noise with de-emphasis.

$$\mathcal{R} = \frac{\left(\frac{\alpha}{A} \right)^2 (4\pi^2 \eta) \int_{-f_M}^{f_M} f^2 df}{\left(\frac{\alpha}{A} \right)^2 (4\pi^2 \eta) \int_{-f_M}^{f_M} f^2 |H_p(f)|^2 df}$$

$$\mathcal{R} = \frac{f_M^3 / 3}{\int_0^{f_M} f^2 df / |H_p(f)|^2}$$

6. This quantity $\mathcal{R}$ is the ratio by which pre-emphasis, de-emphasis improves the signal to noise ratio.

Que 3.19. Write the short note on pre-emphasis and de-emphasis.

AKTU 2014-15, Marks 05

OR

What is pre-emphasis and de-emphasis and how SNR improves by using pre-emphasis and de-emphasis? Find out the figure of merit in SSB-SC System.

AKTU 2017-18, Marks 10

Answer

1. Pre-emphasis and de-emphasis circuits are used for suppression of the unwanted noise.
2. The noise has a greater effect on the higher modulating frequencies than on the lower ones.

3. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal to noise ratio.
4. This boosting of the higher modulating frequencies is termed as pre-emphasis, and the compensation at the receiver is called de-emphasis.
5. The circuits used for each function is shown in Fig. 3.19.1.

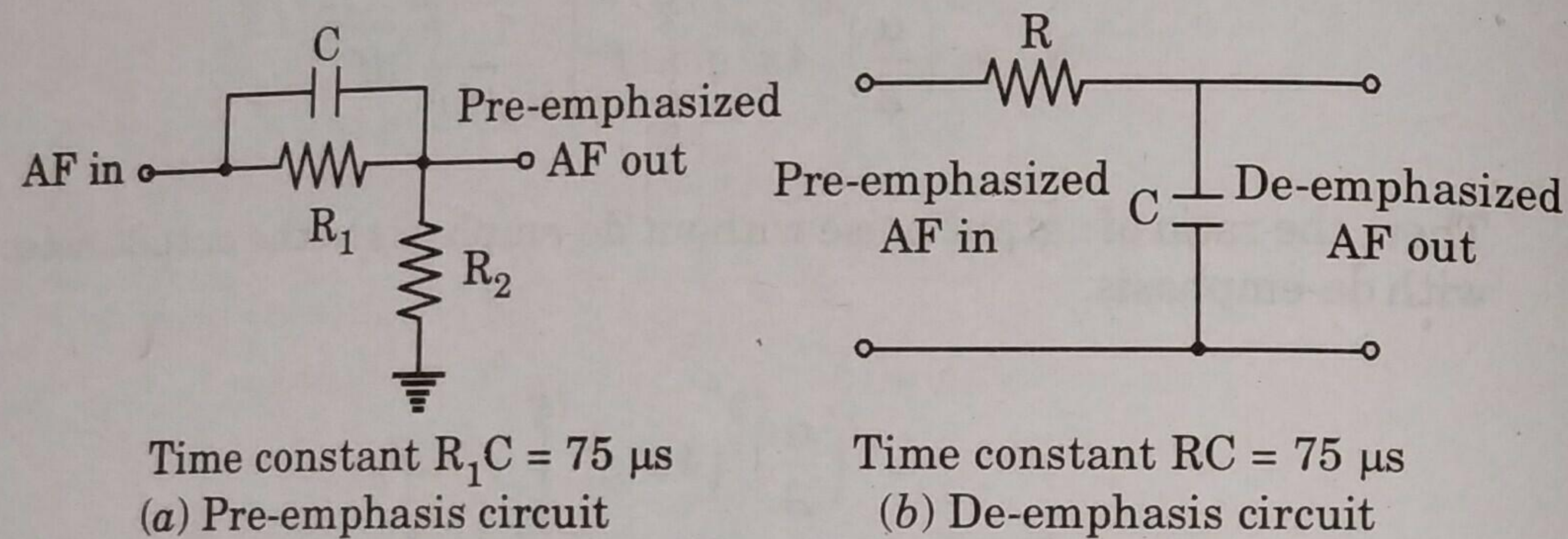
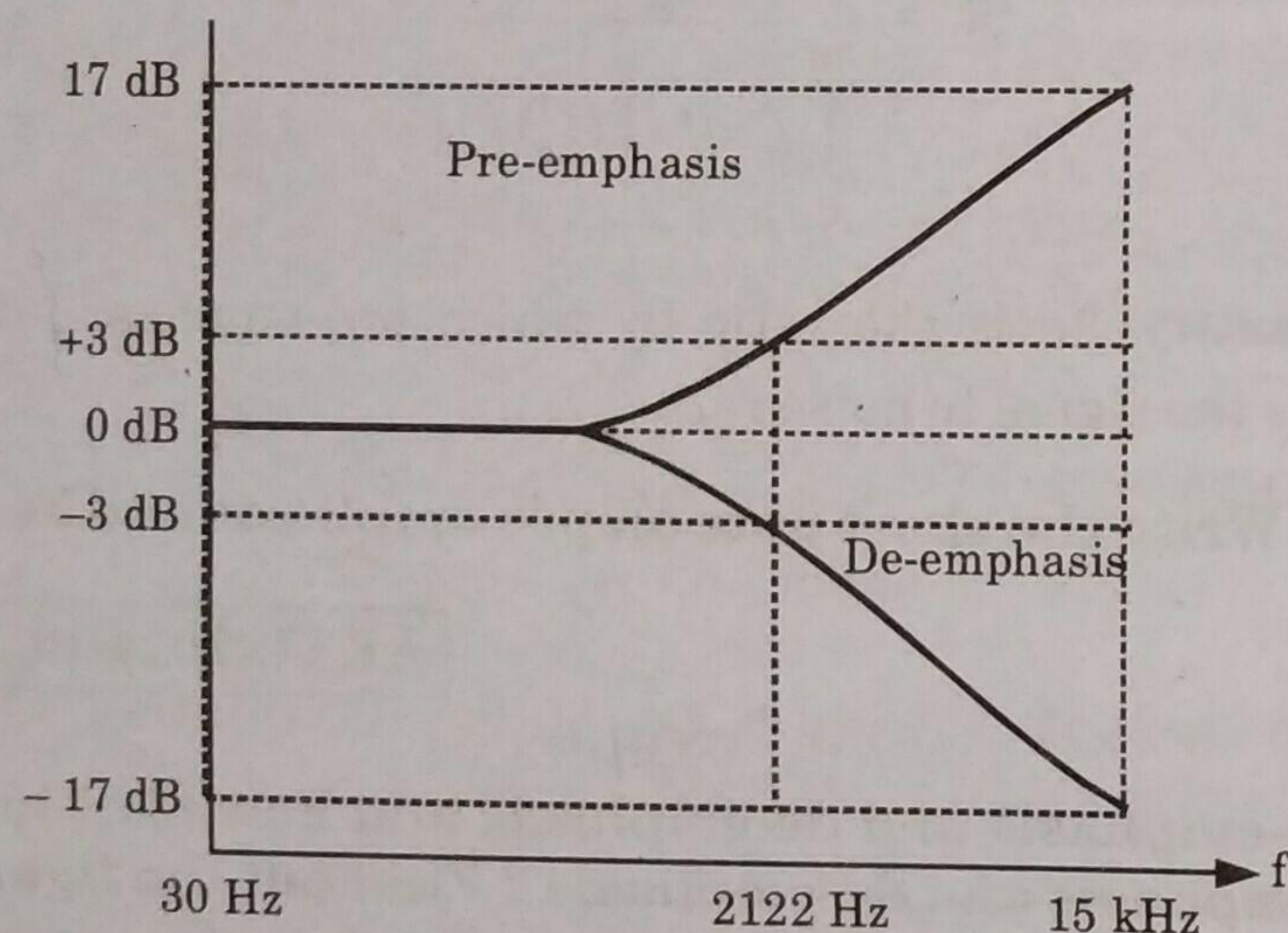


Fig. 3.19.1.

6. Normally pre-emphasis and de-emphasis is carried out according to standard curves. Fig. 3.19.2 shows the $75 \mu s$ emphasis curves.

Fig. 3.19.2. $75 \mu s$ emphasis curves.

7. A $75 \mu s$ pre-emphasis corresponds to a frequency response curve which is 3 dB up at a frequency whose time constant RC is $75 \mu s$ i.e., with $RC = 75 \mu sec$.

$$f = \frac{1}{2\pi RC} = \frac{1}{2\pi \times 75 \times 10^{-6}} = 2122 \text{ Hz}$$

8. In Fig. 3.19.2 observe that the emphasis curves have 3 dB frequency of 2122 Hz.

SNR improvement using pre-emphasis : Refer Q. 3.18, Page 3-26B, Unit-3.

PART-6

Threshold Effect in Angle Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 3.20. Explain threshold effect in angle modulation.

Answer

1. The threshold effect is more in FM as compare to AM, because in this, the signal to noise ratio (S_i/N_i) at the input of a detector, at which threshold effect starts, is higher.
2. Lower the threshold level (S_i/N_i), better is the system, because threshold can be avoided at a comparatively lower (S_i/N_i) ratio, and a small signal is needed to avoid threshold for an equivalent noise power.
3. Hence, it is desirable to lower the threshold level in the FM receivers. The process of lowering the threshold level is known as threshold improvement, or threshold reduction.
4. There are two methods used for the improvement of the threshold.
 - i. Pre-emphasis and De-emphasis circuits, and
 - ii. FMFB (Frequency modulation with feedback).

Que 3.21. Write short note on frequency modulation with feedback (FMFB).

Answer

1. In this method, a feedback path is used for the threshold improvement, as shown in the block diagram of Fig. 3.21.1. The feedback path comprises of a voltage controlled oscillator (VCO).

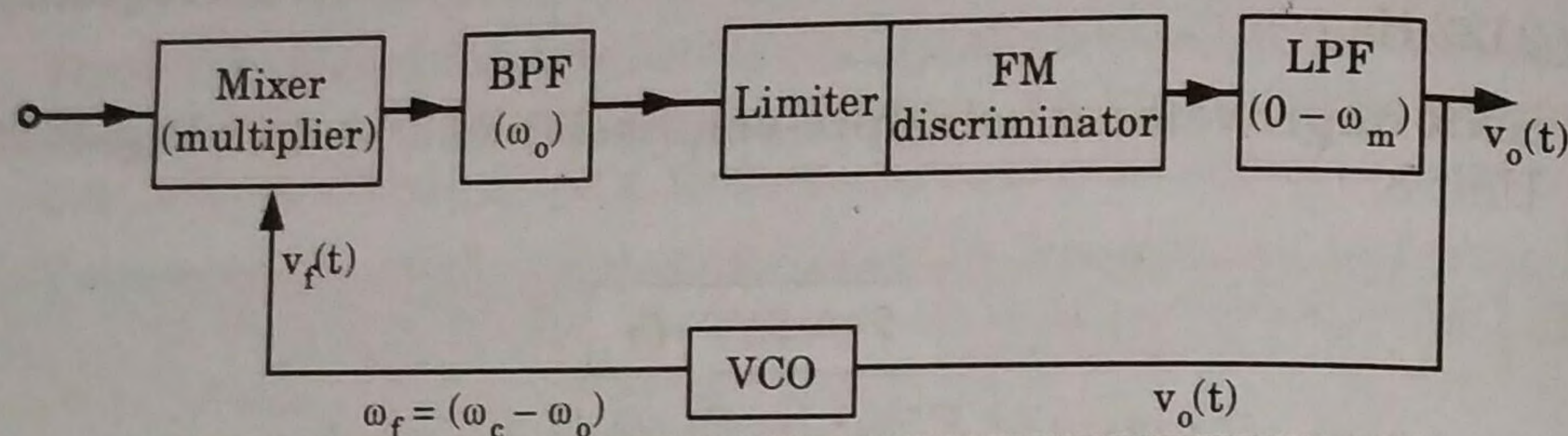


Fig. 3.21.1. Frequency modulation with feedback (FMFB).

2. The mixer circuit is a voltage multiplier circuit which generates a signal at its output which is a multiplication of the two inputs of the mixer

$$\text{i.e., Mixer output} = v_i(t) v_f(t) \quad \dots(3.21.1)$$

where v_i is the FM signal mixed with noise and $v_f(t)$ is the feedback signal coming from the VCO output.

3. The FM signal $v_i(t)$ is given by,

$$v_i(t) = A(t) \cos [\omega_c t + \Psi_n(t) + \Psi_s(t)] \quad \dots(3.21.2)$$

where, $A(t)$ is the envelope of the carrier plus noise signal $\Psi_n(t)$, and $\Psi_s(t)$, respectively are the phase deviations produced in the FM wave by the noise and message signals respectively.

Que 3.22. Discuss voltage controlled oscillator in brief.

Answer

1. Voltage controlled oscillator (VCO) is an electronic sine wave oscillator. Its frequency is controlled by an externally applied voltage. Thus, VCO is a kind of frequency modulator.
2. A VCO can be adjusted to generate a sine wave of frequency ω_c in the absence of any external control voltage. Now, when an external control voltage is applied, the frequency of VCO departs from ω_c .
3. The amount of frequency change per unit input voltage is defined as the sensitivity of the VCO, denoted by G_o ,

$$\text{i.e., } G_o = \frac{d\omega_i}{dv} \text{ rad/v} \quad \dots(3.22.1)$$

where $d\omega_i$ is the change in the instantaneous frequency for a small change dv in the external input voltage.

4. Therefore, the frequency change for an external control voltage $v_o(t)$ will be $G_o v_o(t)$, and the instantaneous frequency of VCO output becomes

$$\omega_i = \omega_c + G_o v_o(t) \quad \dots(3.22.2)$$

5. The corresponding phase angle of the VCO output voltage is

$$\phi(t) = \int \omega_i dt = \omega_c t + G_o \int v_o(t) dt \quad \dots(3.22.3)$$

6. Hence, the VCO output voltage is given as,

$$V_{VCO} = C \cos \left[\omega_c t + G_o \int_{-\infty}^t v_o(t) dt \right] \quad \dots(3.22.4)$$

where C is amplitude of VCO output voltage.

Que 3.23. What is the output of the discriminator with and without feedback?

Answer

1. A demodulation process can be achieved with feedback (FMFB). The demodulated output voltage is similar to an open-loop (without feedback) discriminator output, except that the output of FMFB is reduced.
2. In the absence of a control voltage, the VCO is offset from ω_c by an amount ω_o i.e., the VCO is adjusted to have a frequency $(\omega_c - \omega_o)$.
3. The output of the VCO then is given by,

$$v_f(t) = C \cos \left[(\omega_c - \omega_o) t + G_o \int_{-\infty}^t v_o(t) dt \right] \quad \dots(3.23.1)$$

4. The mixer output produces the sum and difference frequency of its input signals $v_i(t)$, and $v_f(t)$. The bandpass filter, tuned to ω_o , picks up only the difference frequency, and rejects the sum frequency.
5. Thus, the output of the bandpass filter is given by,

$$v_m(t) = D \cos \left[\omega_o t + \Psi_s(t) + \Psi_n(t) - G_o \int_{-\infty}^t v_o(t) dt \right] \quad \dots(3.23.2)$$

6. The amplitude D has been arbitrarily chosen because it has no significance in FM demodulation. The output $v_o(t)$ of FMFB is given by,

$$v_o(t) = K \left[\frac{d\Psi_s(t)}{dt} + \frac{d\Psi_n(t)}{dt} - G_o v_o(t) \right] \quad \dots(3.23.3)$$

where K is a constant of proportionality.

7. Solution of eq. (3.23.3) yields

$$v_o(t) = \frac{K}{1 + KG_o} \frac{d}{dt} (\Psi_s + \Psi_n) \quad \dots(3.23.4)$$

8. The output of conventional discriminator (without feedback) is proportional to a change in the instantaneous frequency of the input signal $v_i(t)$. Thus, in an open loop discriminator,

$$v_o(t) = K \frac{d}{dt} [\Psi_s(t) + \Psi_n(t)] \quad \dots(3.23.5)$$

9. This output is similar to the output voltage of FMFB given in eq. (3.23.4) except that the amplitude of the FMFB output is reduced by a factor $(1 + KG_o)$.
10. Thus, the demodulation process is identical in both the cases. Since the amplitude reduction is identical for the noise and message signals, S_o/N_o ratio remains same in both cases.

Que 3.24. Derive the relation between B_p and B_c .

Answer

1. The bandwidth B_p of the FM signal at the input of the FMFB discriminator is less, as compared to the bandwidth B_c of conventional open loop FM discriminator

$$i.e., \quad B_p < B_c \quad \dots(3.24.1)$$

2. In FMFB, the input to the discriminator is received from a bandpass filter tuned to ω_o and having a bandwidth B_p . The output of the bandpass filter is given by

$$v_m(t) = D \cos \left[\omega_o t + \frac{1}{1 + KG_o} (\Psi_s + \Psi_n) \right] \quad \dots(3.24.2)$$

3. While comparing it with the conventional FM discriminator input as given by eq. (3.24.2), we find that the frequency deviation $(\Delta\omega)_p$ in FMFB is reduced by a factor $(1 + KG_o)$, as against the deviation $(\Delta\omega)$ of the conventional discriminator

$$i.e., \quad (\Delta\omega)_p = \frac{1}{1 + KG_o} (\Delta\omega) \quad \dots(3.24.3)$$

4. Thus, the frequency deviation in the FMFB is reduced by a factor $(1 + KG_o)$. By Carson's rule the bandwidth of the FM signal is determined by its frequency deviation.

5. Therefore, the bandwidth B_p of the FMFB is less than that conventional FM case.

6. A frequency deviation of 75 kHz, if we take a modulating signal of the bandwidth 15 kHz, the bandwidth B_c (conventional FM) is given as,

$$B_c = 2(\Delta f + f_m) = 2(75 + 15) = 180 \text{ kHz}$$

7. Whereas, the bandwidth B_p for FMFB is given as,

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

8. Assuming $KG_o = 9$ we get

$$B_p = 2 \left(\frac{75}{10} + 15 \right) = 45 \text{ kHz}$$

9. Thus, the FMFB bandwidth B_p is reduced by a factor of 4

$$i.e., \quad B_p = \frac{1}{4} B_c$$

Que 3.25. Prove that

$$B_p = \frac{[m_f / (1 + KG_o)] + 1}{m_f + 1} B_c$$

Answer

1. By Carson's rule, the bandwidth of an FM signal at the input of a conventional discriminator is,

$$B_c = 2(\Delta\omega + \omega_m) = 2\omega_m \left(\frac{\Delta\omega}{\omega} + 1 \right) = 2\omega_m (m_f + 1) \text{ radian} \quad \dots(3.25.1)$$

$$\text{or} \quad B_c = 2f_m (m_f + 1) \text{ Hz}$$

2. Similarly, for the case of FMFB,

$$B_p = 2 \left(\frac{\Delta\omega}{1 + KG_o} + \omega_m \right) = 2\omega_m \left(\frac{m_f}{1 + KG_o} + 1 \right) \text{ radians}$$

$$\text{or} \quad B_p = 2 \left(\frac{m_f}{1 + KG_o} + 1 \right) f_m \text{ Hz} \quad \dots(3.25.2)$$

3. From eq. (3.25.1),

$$f_m = \frac{B_c}{2(m_f + 1)} \text{ Hz}$$

4. Substituting the value of f_m in eq. (3.25.2), then we get

$$B_p = 2 \left(\frac{m_f}{1 + KG_o} + 1 \right) \times \frac{B_c}{2(m_f + 1)}$$

$$= \frac{[m_f / (1 + KG_o)] + 1}{m_f + 1} \cdot B_c$$

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. Explain the concept of probability.

Ans. Refer Q. 3.1.

Q. 2. A binary communication channel, the receiver detects binary pulses with an error probability P_e . What is the probability that out of 100 received digits, no more than four digits are in error?

Ans. Refer Q. 3.3.

Q. 3. In an experiment, a trial consists of four successive tosses of a coin. If we define a random variable x as number of head appearing in a trial, determine $P_x(x)$ and $F_x(x)$.

Ans. Refer Q. 3.4.

Q. 4. Write short note on power spectral density for wide-sense stationary random process.

Ans. Refer Q. 3.6.

Q. 5. Find the PSD $S_y(f)$ for a polar random signal use a basic pulse as shown in Fig. 3.7.1. The digits are 1 and 0 equally likely and digits are transmitted every T_b seconds. Each digit is independent of the remaining digits.

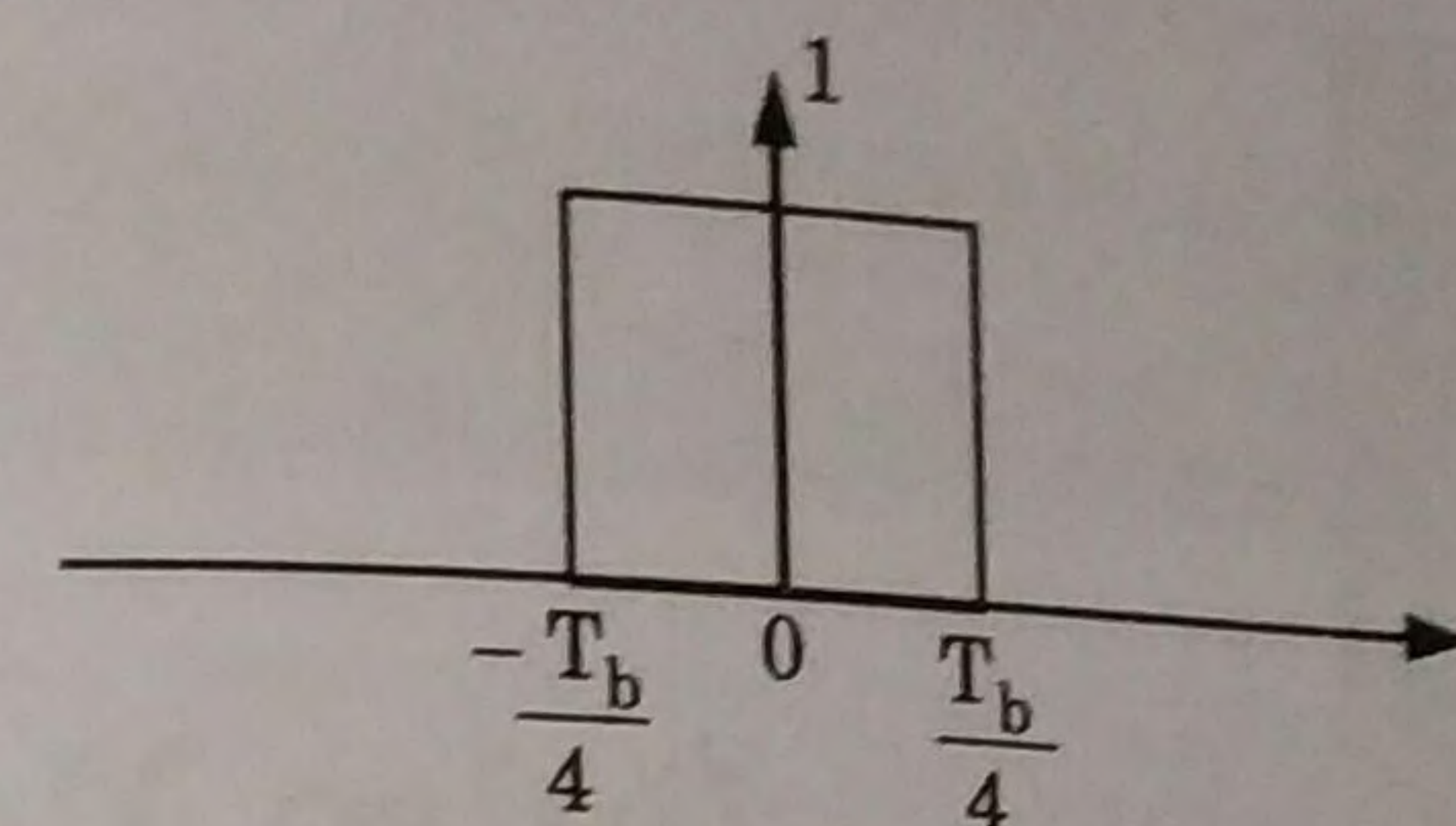


Fig. 3.7.1

Ans. Refer Q. 3.7.

Q. 6. List the properties of gaussian noise.

Ans. Refer Q. 3.11.

Q. 7. Calculate signal to noise ratio for single side band suppressed carrier (SSB-SC).

Ans. Refer Q. 3.14.

Q. 8. Define and explain signal to noise ratio. Describe methods to calculate noise in AM and FM systems.

Ans. Refer Q. 3.17.

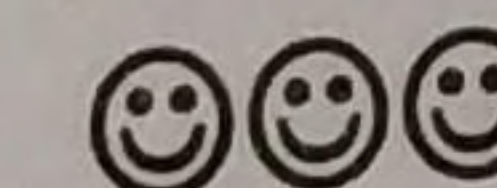
Q. 9. What is pre-emphasis and de-emphasis and how SNR improves by using pre-emphasis and de-emphasis? Find out the figure of merit in SSB-SC System.

Ans. Refer Q. 3.19.

Q. 10. Prove that

$$B_p = \frac{[m_f / (1 + KG_o)] + 1}{m_f + 1} B_c$$

Ans. Refer Q. 3.25.



4

UNIT

Pulse Modulation

CONTENTS

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PART-1

Pulse Modulation, Sampling Process.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.1. What is pulse modulation and what are their different types ? Also give advantage of pulse modulation.

Answer

Pulse modulation : In pulse modulation, the carrier consists of a pulse train, some parameters of which are varied in accordance with the instantaneous value of the modulating signal.

Types : Pulse modulation can be classified as follows :

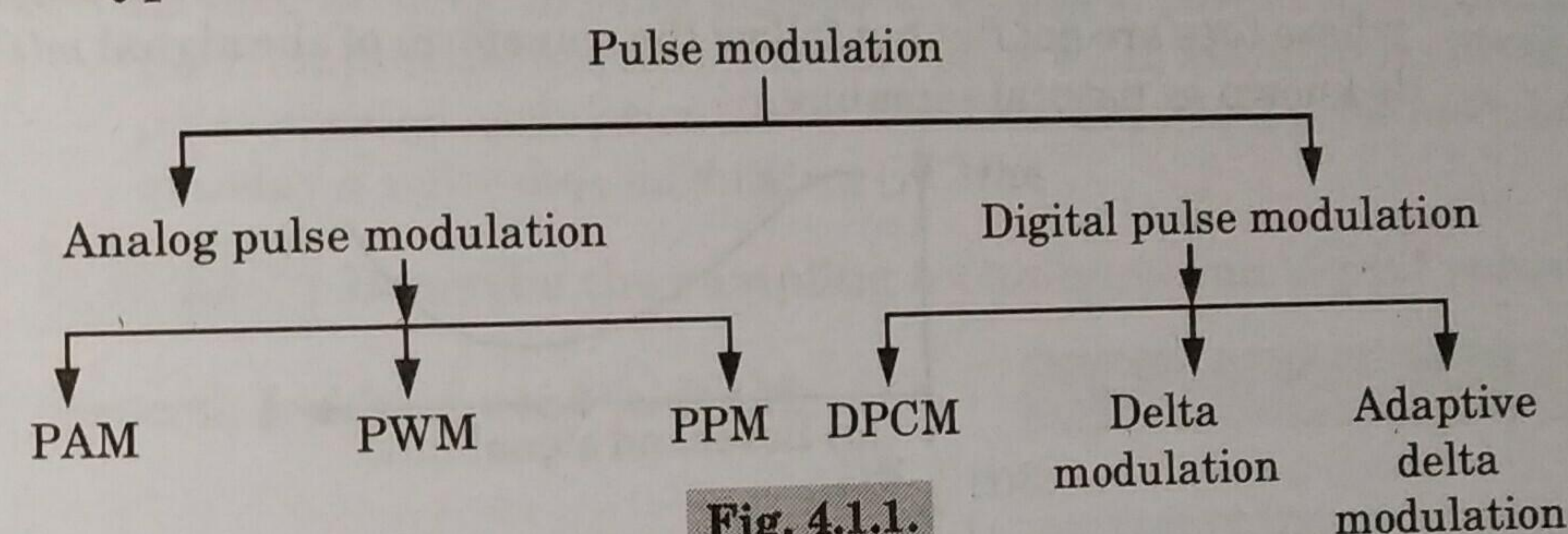


Fig. 4.1.1.

Difference between pulse analog and pulse digital modulation : Basic difference between pulse analog and pulse digital modulation is that the transmission of the analog is in the form of pulses and for digital transmission is in the form of encoded digital data.

The pulse being quite short compared to the time between them, hence pulse-modulated wave is off most of the time.

Advantage of pulse modulation :

1. The transmitted power can be concentrated into short burst rather than being delivered continuously. This gives added choice in equipment selection, since certain devices, such as high power microwave tubes and lasers, are operable only on a pulsed basis.
2. The time intervals between pulses can be filled with sample values from other messages, thereby permitting the transmission of many messages on communication system.

Que 4.2. State sampling theorem and give its application.

OR

Summarize the sampling process of a signal with mathematical expressions.

AKTU 2016-17, Marks 10

Answer

Sampling theorem :

1. Let $m(t)$ be a signal which is bandlimited such that its highest frequency spectral component is f_M . Let the values of $m(t)$ be determined at regular intervals separated by times $T_s \leq 1/2f_M$, that is, the signal is periodically sampled every T_s second.
2. Then these samples $m(nT_s)$, where n is an integer, uniquely determine the signal, and the signal may be reconstructed from these samples with no distortion.

Sampling process :

1. Natural sampling is shown in Fig. 4.2.1. Here, the sampling waveform $S(t)$ consists of a train of pulses having duration τ and separated by the sampling time T_s .
2. The sampled signal consists of a sequence of pulses of varying amplitude whose tops are not flat but follow the waveform of the signal $m(t)$, this is known as natural sampling.

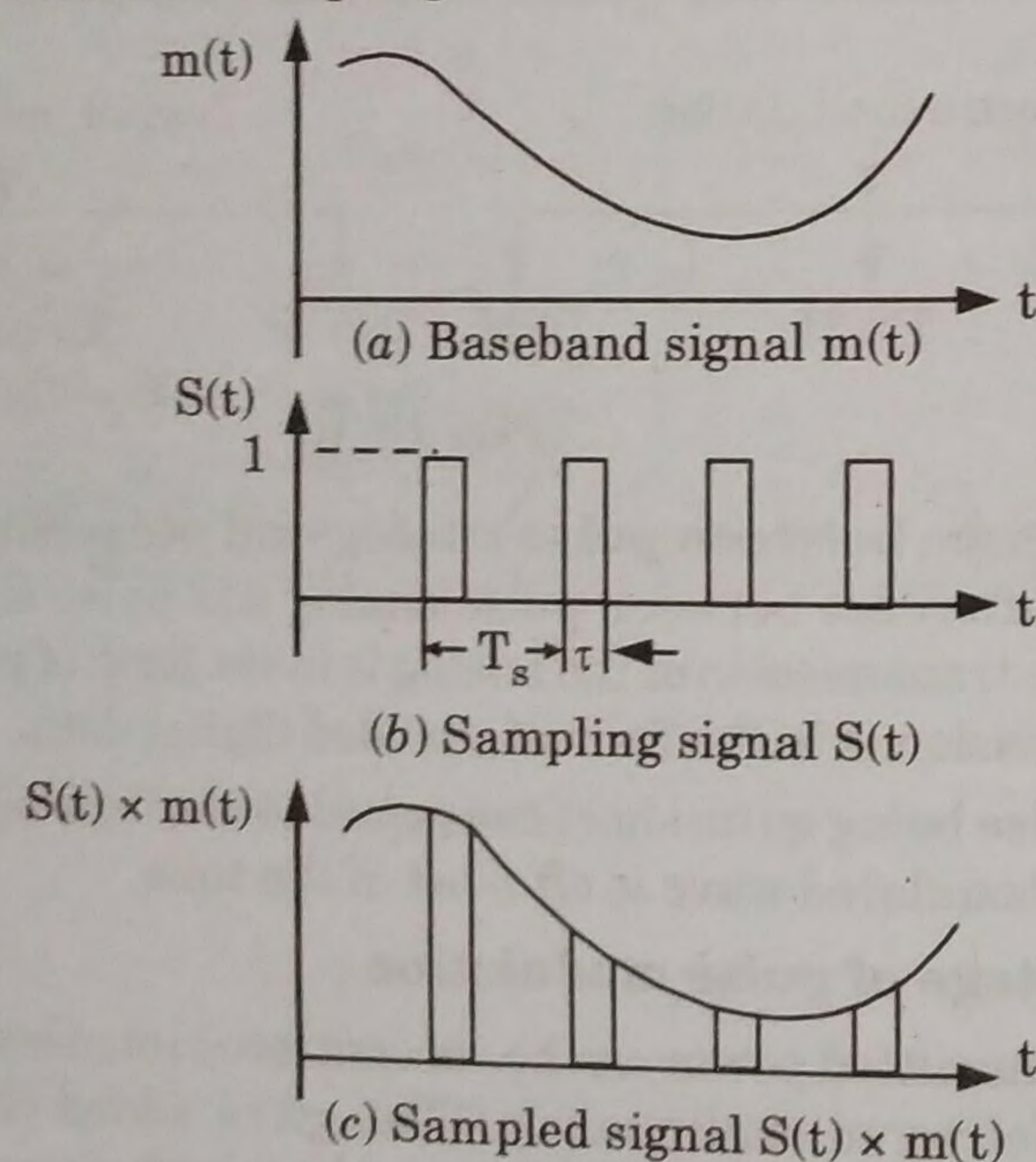


Fig. 4.2.1.

3. The sampling waveform $S(t)$ is given by

$$S(t) = \frac{\tau}{T_s} + \frac{2\tau}{T_s} \left(C_1 \cos 2\pi \frac{t}{T_s} + C_2 \cos 2 \times 2\pi \frac{t}{T_s} + \dots \right)$$

with the constant C_n given by

$$C_n = \frac{\sin(n\pi\tau/T_s)}{n\pi\tau/T_s}$$

4. The sampled baseband $S(t)m(t)$ is, for $T_s = 1/2f_M$,

$$S(t)m(t) = \frac{\tau}{T_s} m(t) + \frac{2\tau}{T_s} [m(t)C_1 \cos 2\pi(2f_M)t + m(t)C_2 \cos 2\pi(4f_M)t + \dots]$$

5. Therefore, as in instantaneous sampling, a low-pass filter with cut-off at f_M will deliver an output signal $s_0(t)$ given by

$$s_0(t) = \frac{\tau}{T_s} m(t)$$

Applications :

1. The sampling theorem is very important in signal analysis, processing and transmission because it allows us to replace a continuous time signal by a discrete sequence of numbers.
2. This leads us directly into the area of digital filtering. This opens doors to many new techniques of communicating continuous time signals by pulse train.
3. In this, we may vary the amplitude, widths, or positions. We can have pulse amplitude modulation (PAM), pulse width modulation (PWM), or pulse position modulation (PPM). The most important pulse modulation of today is pulse code modulation (PCM).

Que 4.3. Describe the sampling techniques and signal recovery

through holding used in PAM.

AKTU 2016-17, Marks 15

OR

Explain flat top sampling in detail.

AKTU 2017-18, Marks 10

Answer

Sampling techniques :

A. **Natural sampling :** Refer Q. 4.2, Page 4-2B, Unit-4.

B. **Flat top sampling :**

1. In the flat top sampling technique, the analog signal $x(t)$ is sampled instantaneously at the rate $f_s = \frac{1}{T_s}$ and the duration of each sample is lengthened to a duration " τ " as shown in Fig. 4.3.1(b).
2. Thus the amplitudes of these pulses are constant and equal to the corresponding sampled values.
3. The flat top pulses can be obtained by using the sample and hold circuit shown in Fig. 4.3.1(a).

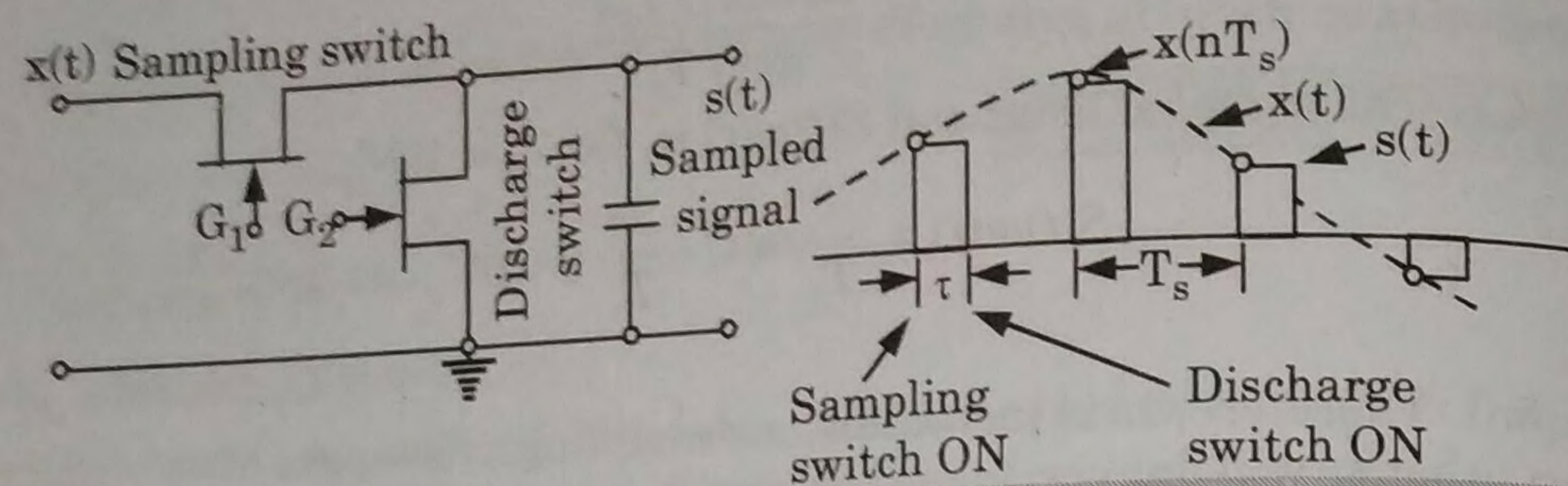


Fig. 4.3.1. (a) Sample and hold circuit to obtain the flat topped samples.
(b) Flat-top sampled signal.

Operation of the sample and hold circuit :

1. The sample and hold circuit consists of two FET switches and a capacitor as shown in Fig. 4.3.1(a).
2. The analog signal $x(t)$ is applied at the input of this circuit and the sampled signal $s(t)$ is obtained across the capacitor.
3. A gate pulse will be applied to gate G_1 at the instant of sampling, for a very short time.
4. The sampling switch will turn ON and the capacitor charges through it to the sample value $x(nT_s)$.
5. The sampling switch is then turned OFF. Both the FETs will remain OFF for a duration of " τ " seconds and the capacitor will hold the voltage across it constant for this period. Thus the pulse is stretched to " τ " seconds.
6. At the end of the pulse interval (τ), a pulse is applied to G_2 i.e. gate terminal of discharge FET.
7. This will turn ON the discharge FET and short circuit the capacitor. The output voltage then reduces to zero. This is as shown in Fig. 4.3.1(b).

PART-2

Pulse Amplitude and Pulse Code Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.4. Explain the term pulse amplitude modulation and its respective channel bandwidth.

Answer

1. Pulse amplitude modulation is the simplest and most basic form of analog pulse modulation. In PAM, the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample value of a continuous message signal. PAM is shown in Fig. 4.4.1 using flat top sampling.

Channel bandwidth for PAM :

1. Let us assume that we have to transmit n signal each band limited to f_m Hz.
2. Then for each signal we have to take $2f_m$ samples per second. Thus, in all, we have to transmit $2nf_m$ samples per second.
3. Now, according to the sampling theorem, a continuous signal band limited to W Hz can be transmitted by $2W$ samples per second conversely, it can be stated that $2W$ samples per second define a continuous signal bandlimited to W Hz.
4. Therefore, the bandwidth of the PAM system of n signals, each bandlimited to f_m Hz, will be nf_m Hz because we are transmitting $2nf_m$ samples per second.

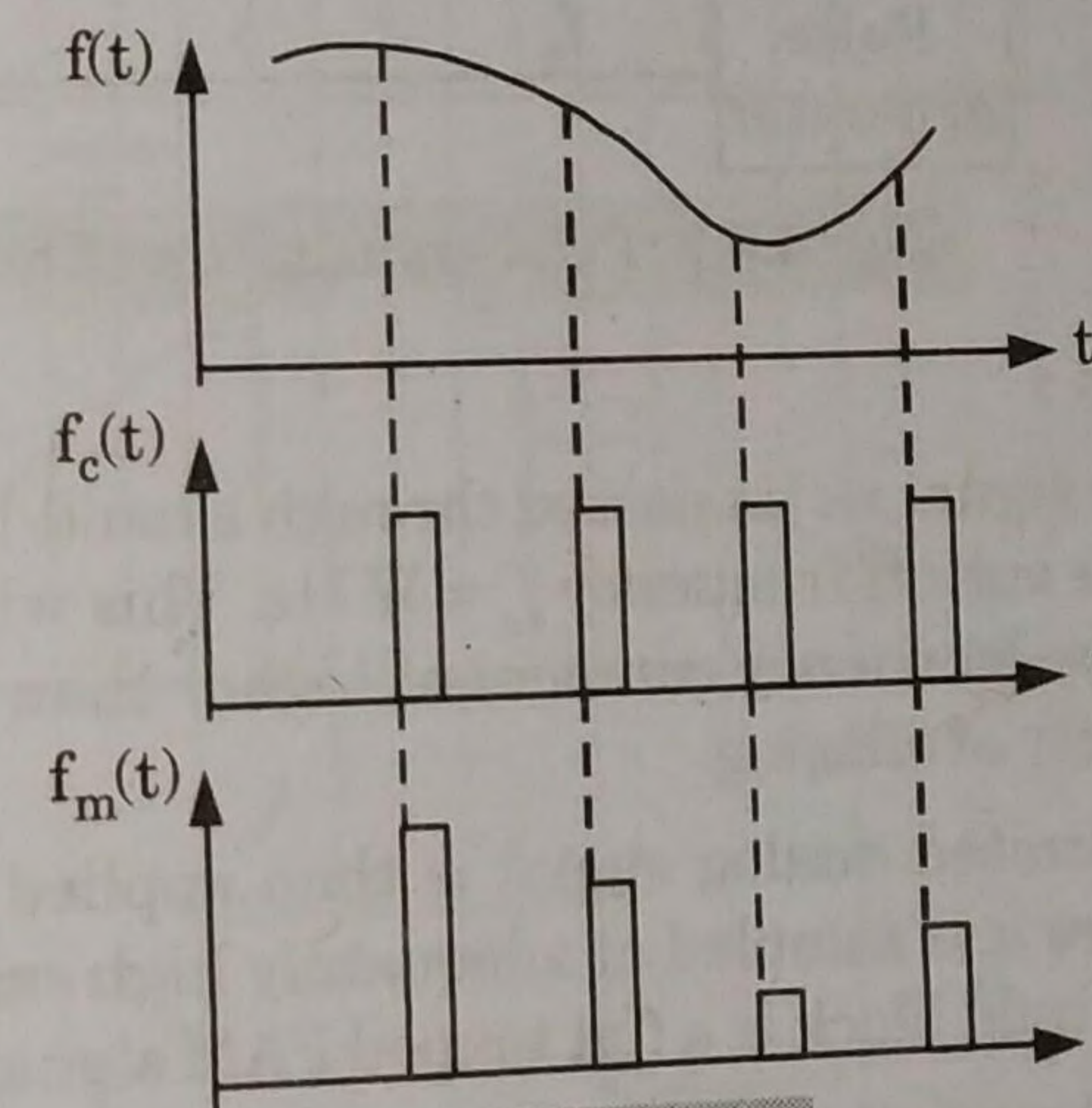


Fig. 4.4.1.

Que 4.5. What do you understand by PCM (pulse code modulation) ? Explain basic elements and operation of the PCM system with its respective block diagram.

Answer

PCM (pulse code modulation) :

1. In pulse code modulation (PCM), a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude.

- The essential processes of PCM are sampling, quantizing, and encoding, as shown in Fig. 4.5.1. Sampling is the process in which a continuous-time signal is sampled by measuring its amplitude at discrete instant.
- Representing the sampled values of the amplitude by a finite set of levels is called quantizing.
- Designating each quantized level by a code is called encoding.

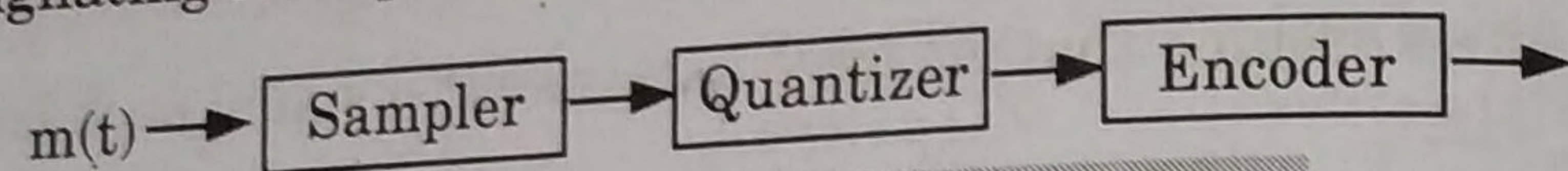


Fig. 4.5.1. Pulse code modulation.

PCM transmitter (Encoder) :

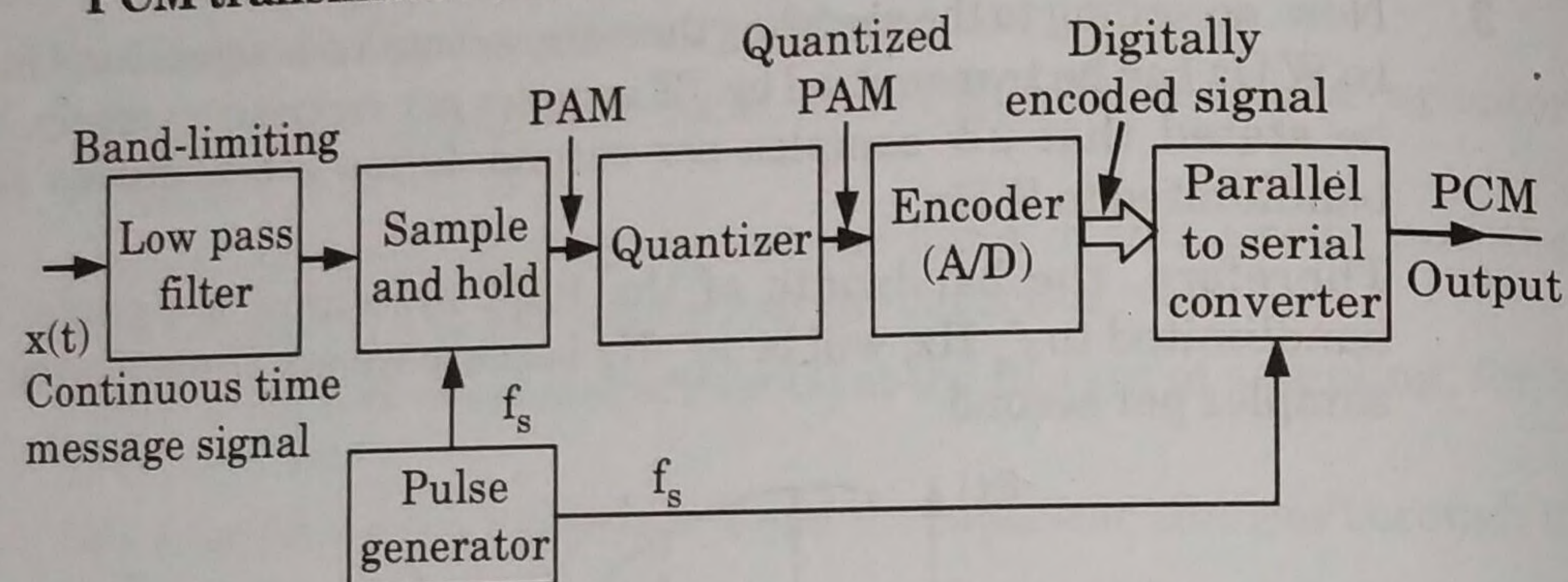


Fig. 4.5.2. PCM transmitter (Encoder).

Operation :

- The analog signal $x(t)$ is passed through a band-limiting low pass filter, which has a cut-off frequency $f_c = W$ Hz. This will ensure that $x(t)$ will not have any frequency component higher than W . This will eliminate the possibility of aliasing.
- The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal.
- These samples are then subjected to the operation called quantization in the quantizer. Quantization process is the process of approximation. The quantization is used to reduce the effect of noise.
- The combined effect of sampling and quantization produces the quantized PAM at the quantizer output.
- The quantized PAM pulses are applied to an encoder which is basically an A to D converter. Each quantized level is converted into an N -bit digital word by the A to D converter. The value of N can be 8, 16, 32, 64 etc.
- The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses.

- A pulse generator produces a train of rectangular pulses with each pulse of duration τ seconds. The frequency of this signal is f_s Hz. This signal acts as sampling signal for the sample and hold block.
- The same signal acts as clock signal for the parallel to serial converter. The frequency f_s is adjusted to satisfy the Nyquist criteria.

Waveforms : The waveforms at various points in the PCM transmitter are as shown in Fig. 4.5.3.

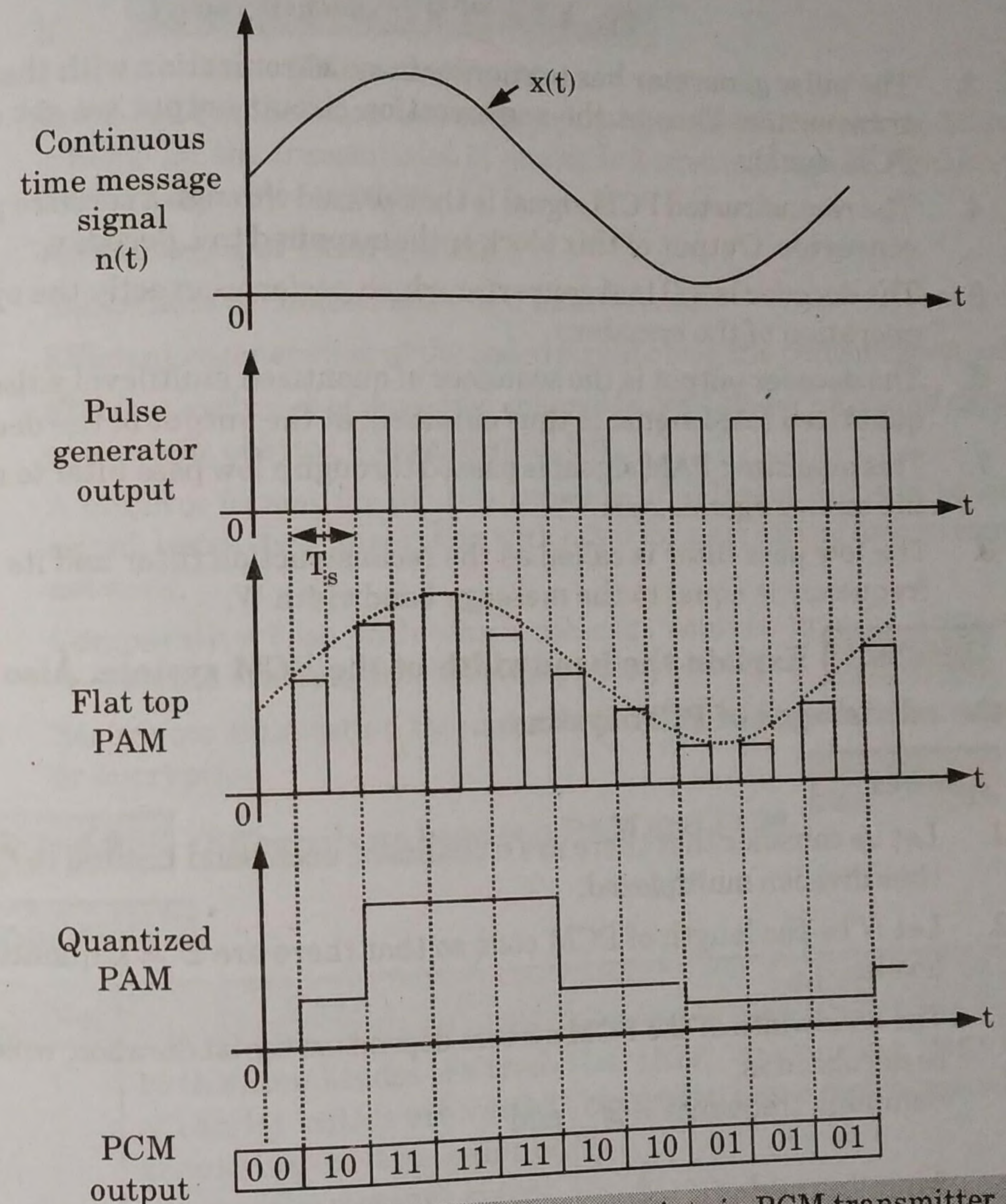


Fig. 4.5.3. Waveforms at different points in PCM transmitter.

PCM Receiver (Decoder) :

- A PCM signal contaminated with noise is available at the receiver input.
- The regeneration circuit at the receiver will separate the PCM pulses from noise and will reconstruct the original PCM signal.

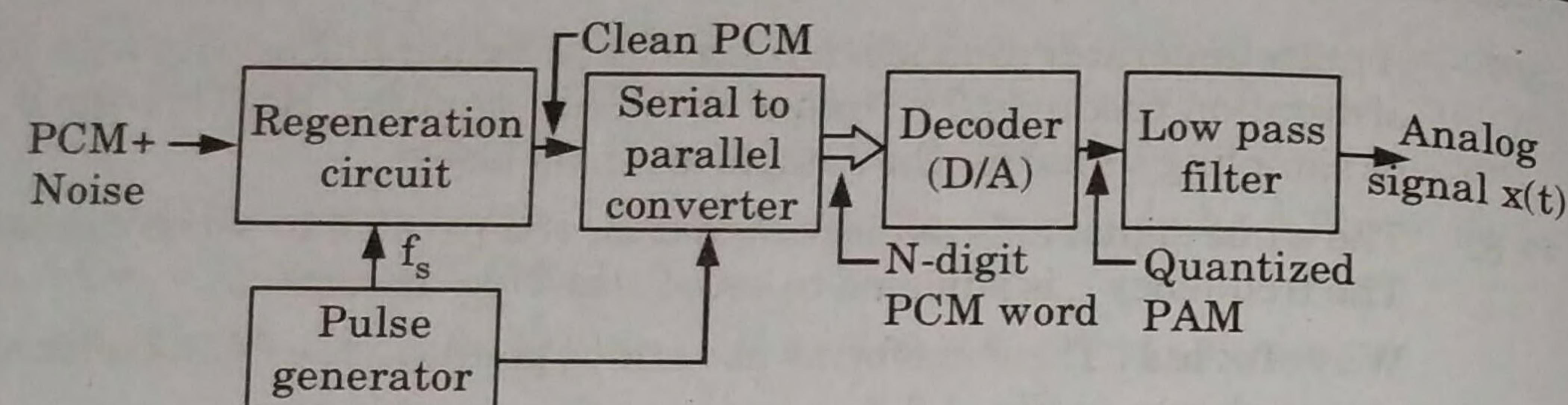


Fig. 4.5.4. PCM receiver (Decoder).

- The pulse generator has to operate in synchronization with that at the transmitter. Thus at the regeneration circuit output we get a clean PCM signal.
- The reconstructed PCM signal is then passed through a serial to parallel converter. Output of this block is then applied to a decoder.
- The decoder is a D to A converter which performs exactly the opposite operation of the encoder.
- The decoder output is the sequence of quantized multilevel pulses. The quantized PAM signal is thus obtained, at the output of the decoder.
- This quantized PAM signal is passed through a low pass filter to recover the analog signal, $x(t)$.
- The low pass filter is called as the reconstruction filter and its cut-off frequency is equal to the message bandwidth W .

Que 4.6. Explain the bandwidth of the PCM system. Also write the advantages of PCM system.

Answer

- Let us consider that there are n channels, each band limited to f_m , to be time division multiplexed.
- Let N be the length of PCM code so that there are $2^N = L$ quantization levels.
- The bandwidth of the PCM system depends on the bit duration, which may be calculated as,
Sampling frequency = $2f_m$, and
Sampling period = $\frac{1}{2f_m}$
- As there are n channels and N bits per sample, one bit separating each channel and one synchronizing bit, the total number of bits/sample
 $= nN + n + 1 = n(N + 1) + 1$
- Therefore, the bit duration
 $= \frac{\text{sampling period}}{\text{total number of bits}}$

$$T_b = \frac{1}{[n(N+1)+1]2f_m} \quad \dots(4.6.1)$$

where, T_b is bit duration.

$$6. \text{ Thus, } BW = \frac{1}{T_b} \quad \dots(4.6.2)$$

Putting the value of T_b in eq. (4.6.2) we get
 $BW = [n(N+1)+1]2f_m \text{ Hz}$

If $N \gg 1$ and $n \gg 1$

$$BW \approx 2nNf_m \text{ Hz}$$

- Pulse code modulation has emerged as the most favoured modulation scheme for the transmission of analog information bearing signals such as voice and video signals.

Advantages of PCM system :

- Robustness to channel noise and interference.
- Efficient regeneration of the coded signal along the transmission path.
- Efficient exchange of increased channel bandwidth for improved signal to noise ratio, obeying an exponential law.
- A uniform format for the transmission of different kinds of baseband signal, hence their integration with other forms of digital data in common network.
- Comparative ease with which message sources may be dropped or re-inserted in a time division multiplexing system.
- Secure communication through the use of special modulation schemes or encryption.

Que 4.7. Differentiate between PAM and PCM.

Answer

S.No.	PAM	PCM
1.	In this, amplitudes of a train of carrier pulses are varied according to the sample value of the message signal.	In this, an analog signal is converted into a digital signal.
2.	PAM falls under analog pulse modulation techniques.	PCM falls under digital communications.
3.	It is less secure.	It is more secure.
4.	It is less immune to noise.	It is more immune to noise.

PART-3**Differential Pulse Code Modulation.****Questions-Answers****Long Answer Type and Medium Answer Type Questions**

Que 4.8. What is differential pulse code modulation ? Explain working of DPCM with a proper block diagram.

AKTU 2017-18, Marks 10

Answer

1. In differential pulse code modulation (DPCM) instead of quantizing each sample, the difference between two successive samples is quantized, encoded and transmitted as in PCM.
2. The DPCM works on the principle of prediction. The value of the present sample is predicted from the past samples.
3. Fig. 4.8.1 shows the transmitter and receiver of DPCM system.

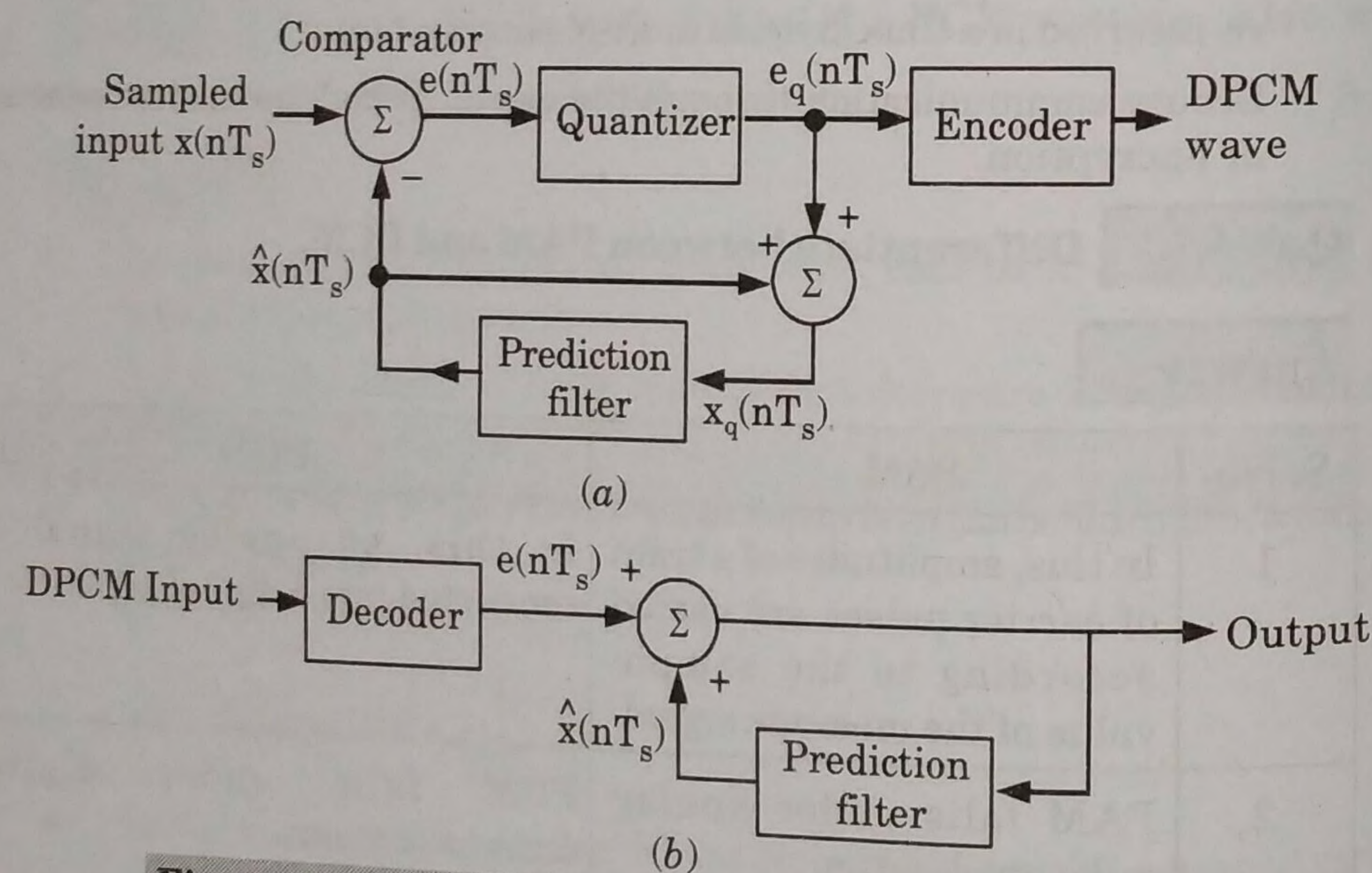


Fig. 4.8.1. DPCM system (a) Transmitter (b) Receiver.

Transmitter :

1. The input signal to the quantizer is defined by

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \quad \dots(4.8.1)$$

where,

$x(nT_s)$ is the sampled version of $x(t)$

$\hat{x}(nT_s)$ is estimate of $x(nT_s)$.

2. The difference signal $e(nT_s)$ is the prediction error, since it is the amount by which the prediction filter fails to predict the input exactly.
3. By encoding the quantizer output, we obtain a variant of PCM known as differential pulse code modulation (DPCM).
4. The quantizer output may be expressed as

$$e_q(nT_s) = e(nT_s) + q_e(nT_s) \quad \dots(4.8.2)$$

where, $e_q(nT_s)$ is the quantized version of $e(nT_s)$ and $q_e(nT_s)$ is the quantization error.

5. The input of the prediction filter is

$$= \hat{x}(nT_s) + e_q(nT_s)$$

$$= \hat{x}(nT_s) + e(nT_s) + q_e(nT_s)$$

$$= x(nT_s) + q_e(nT_s)$$

$$= x_q(nT_s)$$

where $x_q(nT_s)$ represents a quantized version of the input sample $x(nT_s)$.

Receiver :

1. The receiver consists of decoder which is used to reconstruct the quantized error signal.
2. The quantized version of the original input is reconstructed from the decoder output using the same prediction filter used in the transmitter.
3. The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal.

$$\begin{aligned} \text{Receiver output} &= e(nT_s) + \hat{x}(nT_s) \\ &= x(nT_s) \end{aligned}$$

Que 4.9. Explain processing gain of DPCM system.

Answer

1. The output signal to noise ratio of the DPCM system is given by

$$(\text{SNR})_o = \frac{\sigma_M^2}{\sigma_Q^2}$$

where σ_M^2 variance of the original input sample $m[n]$, assumed to be of zero mean and σ_Q^2 is the variance of the quantization error $q[n]$.

$$(\text{SNR})_o = \left(\frac{\sigma_M^2}{\sigma_E^2} \right) \left(\frac{\sigma_E^2}{\sigma_Q^2} \right)$$

$$= G_p (\text{SNR})_Q$$

where σ_E^2 is the variance of the prediction error.

2. Thus, the signal to quantization noise ratio is

$$(\text{SNR})_Q = \frac{\sigma_E^2}{\sigma_Q^2}$$

3. The other factor G_p is the processing gain

$$G_p = \frac{\sigma_M^2}{\sigma_E^2}$$

4. The quantity G_p when greater than unity represents a gain in signal to noise ratio that is due to the differential quantization scheme.
5. For a given baseband signal the variance σ_M^2 is fixed, so that G_p is maximized by minimizing the variance σ_E^2 of the prediction error $e[n]$.

Que 4.10. What are the advantages of differential pulse code modulation over pulse code modulation ?

Answer

Advantages of differential pulse code modulation over pulse code modulation :

- The use of DPCM provides a saving of about 8 to 16 kb/s as compared to the standard PCM.
- In DPCM instead of quantizing the samples, the prediction error is quantized and only the error signal is transmitted. Since the error is small, fewer bits are needed as compared to PCM.
- Bandwidth requirement in DPCM is less than the one required in PCM.
- In DPCM, instead of quantizing each sample, the difference between two successive samples is quantized encoded and transmitted as in PCM. This is particularly useful in voice transmission because in this case two successive samples do not differ much in amplitude. Thus, the difference signal is much less in amplitude than the actual sample and hence less number of quantization levels is needed.

PART-4

Delta Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.11. What is delta modulation ? Discuss the errors in delta modulation.

Answer

Delta modulation :

- Delta modulation (DM) is a DPCM scheme in which the difference signal $\Delta(t)$ is encoded into just a single bit.
- The single bit is used to increase or decrease the estimate $\hat{m}(t)$.
- This scheme is called linear delta modulation and is shown in Fig. 4.11.1

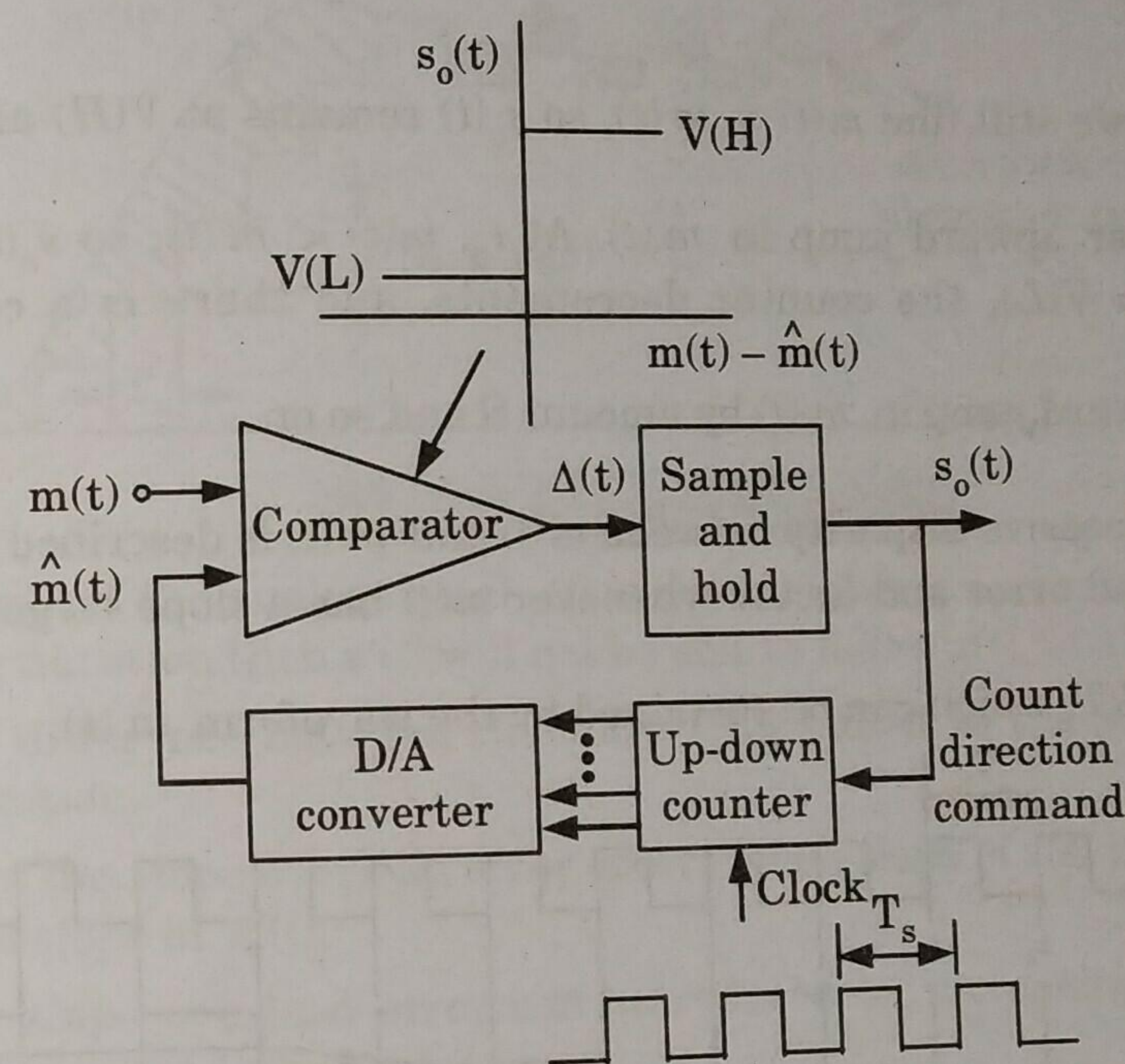


Fig. 4.11.1. A delta modulator.

- The baseband signal $m(t)$ and its quantized approximation $\hat{m}(t)$ are applied as inputs to a comparator.
- A comparator makes a comparison between inputs.
- The comparator has one fixed output $V(H)$ when $m(t) > \hat{m}(t)$ and a different output $V(L)$ when $m(t) < \hat{m}(t)$.
- The up-down counter increments or decrements its count by 1 at each active edge of the clock waveform.
- The count direction, i.e., incrementing or decrementing, is determined by the voltage levels at the "count direction command" input to the counter.

9. When this binary input, which is also the transmitted output $s_o(t)$, is at the level $V(H)$, the counter counts-up and when it is at the level $V(L)$ the counter counts down.
10. The digital output of the counter is converted to the analog quantized approximation $\hat{m}(t)$ by the D/A converter.

11. The waveforms of Fig. 4.11.2 are the waveforms for the system of Fig. 4.11.1, assuming that the active clock edge is the falling edge.

12. For a time preceding t_1 we find $m(t) > \hat{m}(t)$ so that $s_o(t) = V(H)$.
13. At t_1 , when the active clock edge appears, the counter is incremented

and immediately signal $\hat{m}(t)$ jumps up by an amount equal to the step size S .

14. At t_2 we still find $m(t) > \hat{m}(t)$, so $s_o(t)$ remains at $V(H)$ and there is another upward jump in $\hat{m}(t)$. At t_3 , $m(t) < \hat{m}(t)$, so $s_o(t)$ becomes $s_o(t) = V(L)$, the counter decrements, and there is a consequent downward jump in $\hat{m}(t)$ by amount S and so on.

15. The excessive disparity between $m(t)$ and $\hat{m}(t)$ is described as a slope-overload error and occurs whenever $m(t)$ has a slope larger than the slope S/T_s which can be sustained by the waveform $\hat{m}(t)$.

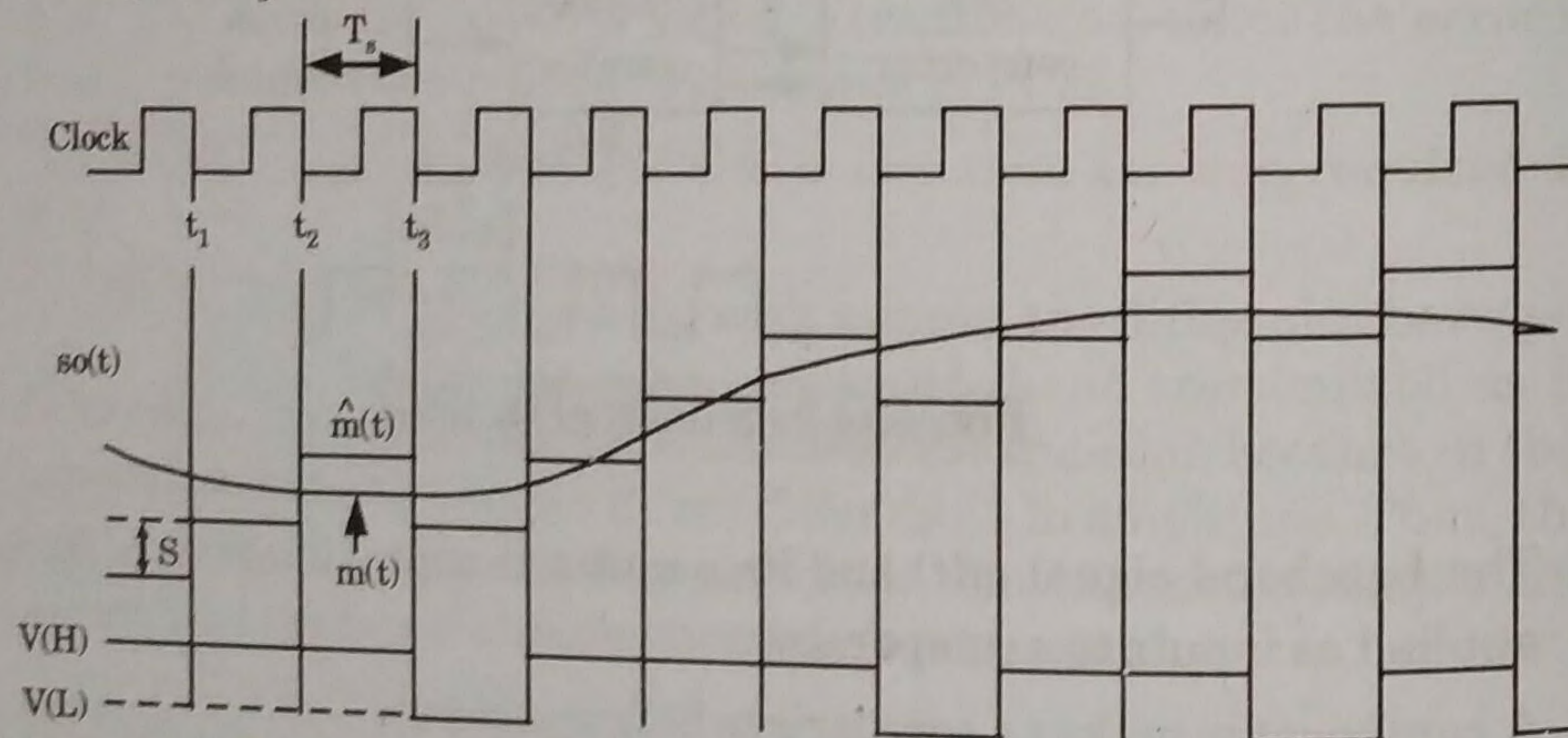


Fig. 4.11.2. The response of a delta modulator to a baseband signal $m(t)$.

16. Slope overload can be avoided if $\frac{S}{T} > \left| \frac{dm(t)}{dt} \right|_{\max}$
17. The dynamic range of amplitude of $m(t)$ is too small because of the threshold and overload effects.

Que 4.12. What is the limitation of delta modulation ?

Answer

Limitation of delta modulation :

The DM system is subjected to two types of quantization error :

a. Slope overload distortion :

1. From Fig. 4.12.1 due to small step size (Δ), the slope of the approximated signal $x'(t)$ will be small.

$$\text{The slope of } x'(t) = \frac{\Delta}{T_s} = \Delta f_s \quad \dots(4.12.1)$$

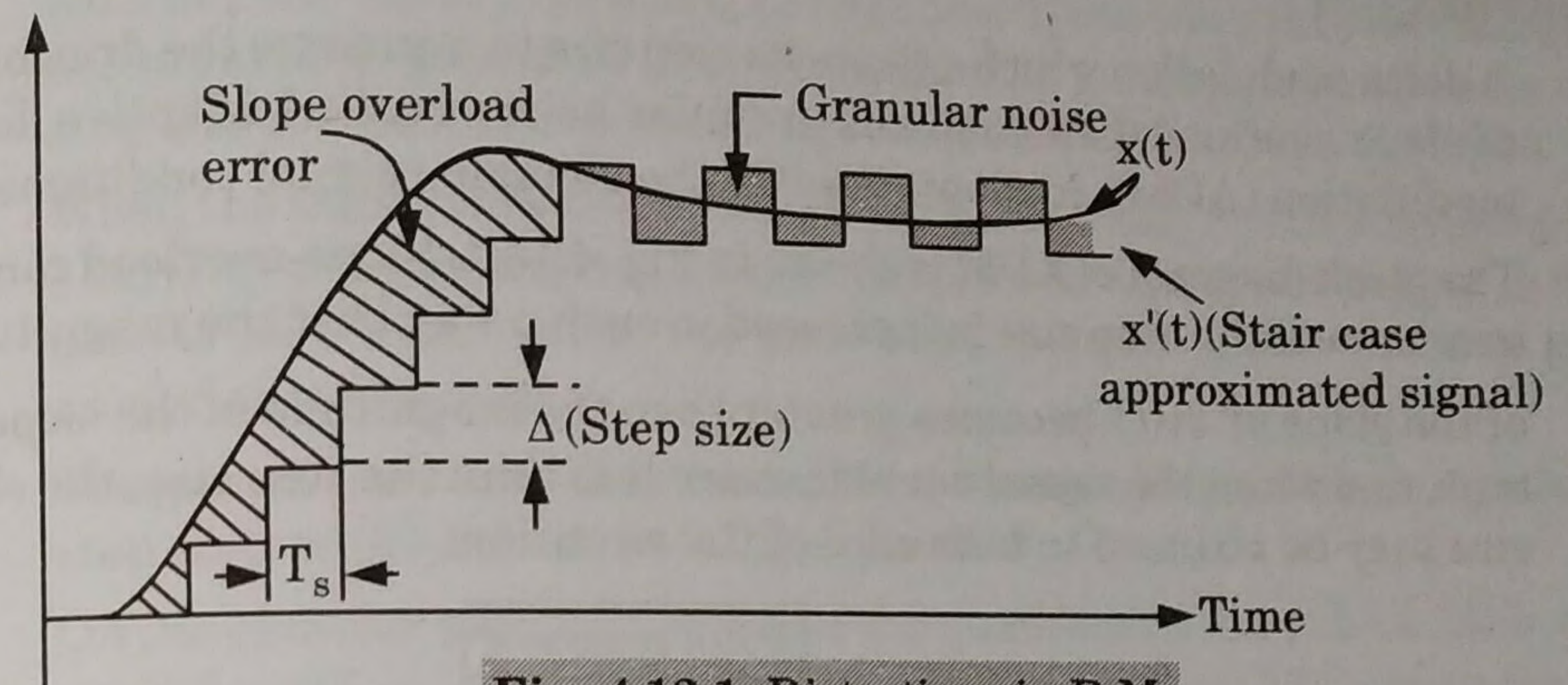


Fig. 4.12.1. Distortions in D.M.

2. If slope of the analog signal $x(t)$ is much higher than that of $x'(t)$ over a long duration then $x'(t)$ will not be able to follow $x(t)$, at all.
 3. The difference between $x(t)$ and $x'(t)$ is called as the slope overload distortion.
 4. Thus the slope overload error occurs when slope of $x(t)$ is much larger than slope of $x'(t)$.
 5. The slope overload error can be reduced by increasing slope of the approximated signal $x'(t)$.
 6. Slope of $x'(t)$ can be increased and hence the slope overload error can be reduced by either increasing the step size Δ or by increasing the sampling frequency f_s .
 7. However, with increase in Δ the granular noise increases and if f_s is increased, signaling rate and bandwidth requirements will go up.
- ##### b. Granular noise :
1. When the input signal $x(t)$ is relatively constant in amplitude, the approximated signal $x'(t)$ will hunt above and below $x(t)$ as shown in Fig. 4.12.1.
 2. It increases with increase in the step size Δ . To reduce the granular noise, the step size should be as small as possible.
 3. However, this will increase the slope overload distortion

4. In the linear delta modulator, the step size Δ is not variable. If it is made variable then the slope overload distortion and granular noise both can be controlled.
5. A system with a variable step size is known as the adaptive delta modulator (ADM).

Que 4.13. Explain working of adaptive delta modulation with a proper block diagram.

AKTU 2017-18, Marks 10

Answer

1. A delta modulation which adjusts its step size to overcome the drawback of slope overload distortion and granular noise is called adaptive delta modulation (ADM). It adopts itself to the changing signal conditions.
2. The block diagram of ADM is shown in Fig. 4.13.1. Slope-overload can be overcome if the step size is increased in such a way that the magnitude of the slope of $\bar{m}(t)$ becomes greater than the magnitude of the slope of $m(t)$, and when the signal variations are less than the step size, the step size may be reduced to take care of the situation.

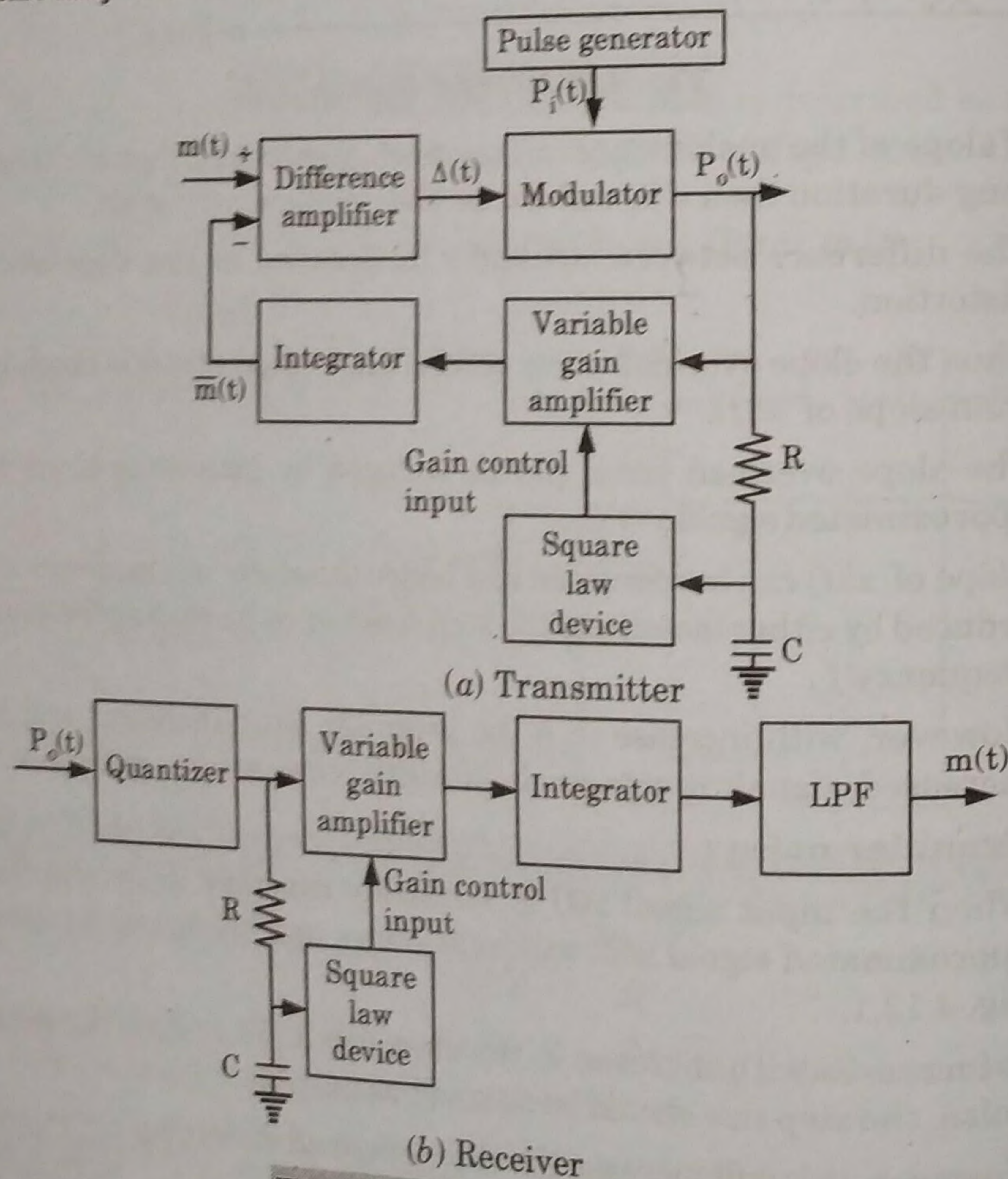


Fig. 4.13.1. ADM system.

3. On the transmitter side, a variable gain amplifier is used before the integrator with $P_o(t)$ as its input.
4. The gain of this amplifier depends on the gain control input, which is obtained by integrating $P_o(t)$ in an RC-network, and then passing the integrator output through a square law device.
5. Under the slope-overload condition, $P_o(t)$ is a long sequence of either positive or negative pulses. The RC integrator integrates these pulses. Thus, the output of this integrator is either of a large positive or negative value.
6. The square law device output is of a large positive value, irrespective of whether the input is positive, or negative. Thus, the gain control input of the variable gain amplifier is large and its gain increases. Hence the step size increases, which can then take care of the slope-overload.
7. When the signal variations are within the step size, $P_o(t)$ is a sequence of alternate positive and negative pulses. The RC integrator output in this case is zero and, hence, the gain control input of the variable gain amplifier is also zero.
8. The gain of the variable gain amplifier decreases, resulting in a reduced step size, which takes care of the situation.
9. On the receiver side, the output of the quantizer is fed to a variable gain amplifier whose gain control input is derived from an RC integrator and a square law device.
10. Thus, an adaptive adjustment of the step size is obtained at the receiver, resulting in an undistorted of the transmitted signal.

Que 4.14. Show the response of baseband signal for delta and adaptive delta modulation.

AKTU 2016-17, Marks 10

Answer

Response of baseband signal for delta modulation :

1. In delta modulation, an incoming message signal is oversampled (i.e., at a rate much higher than Nyquist rate) to increase the correlation between adjacent samples of the signal.
2. This modulation provides a staircase approximation to the oversampled version of the message signal.
3. The difference between the input and the approximation is quantized into only two levels i.e., $\pm \Delta$, corresponding to positive and negative differences.
4. Let $m(t)$ denote the input message signal and $m_q(t)$ denote its staircase approximation. Thus,

$$m[n] = m[nT_s], n = 0, \pm 1, \pm 2$$

where T_s is the sampling period and $m(nT_s)$ is the sample of the signal $m(t)$ taken at time $t = nT_s$.

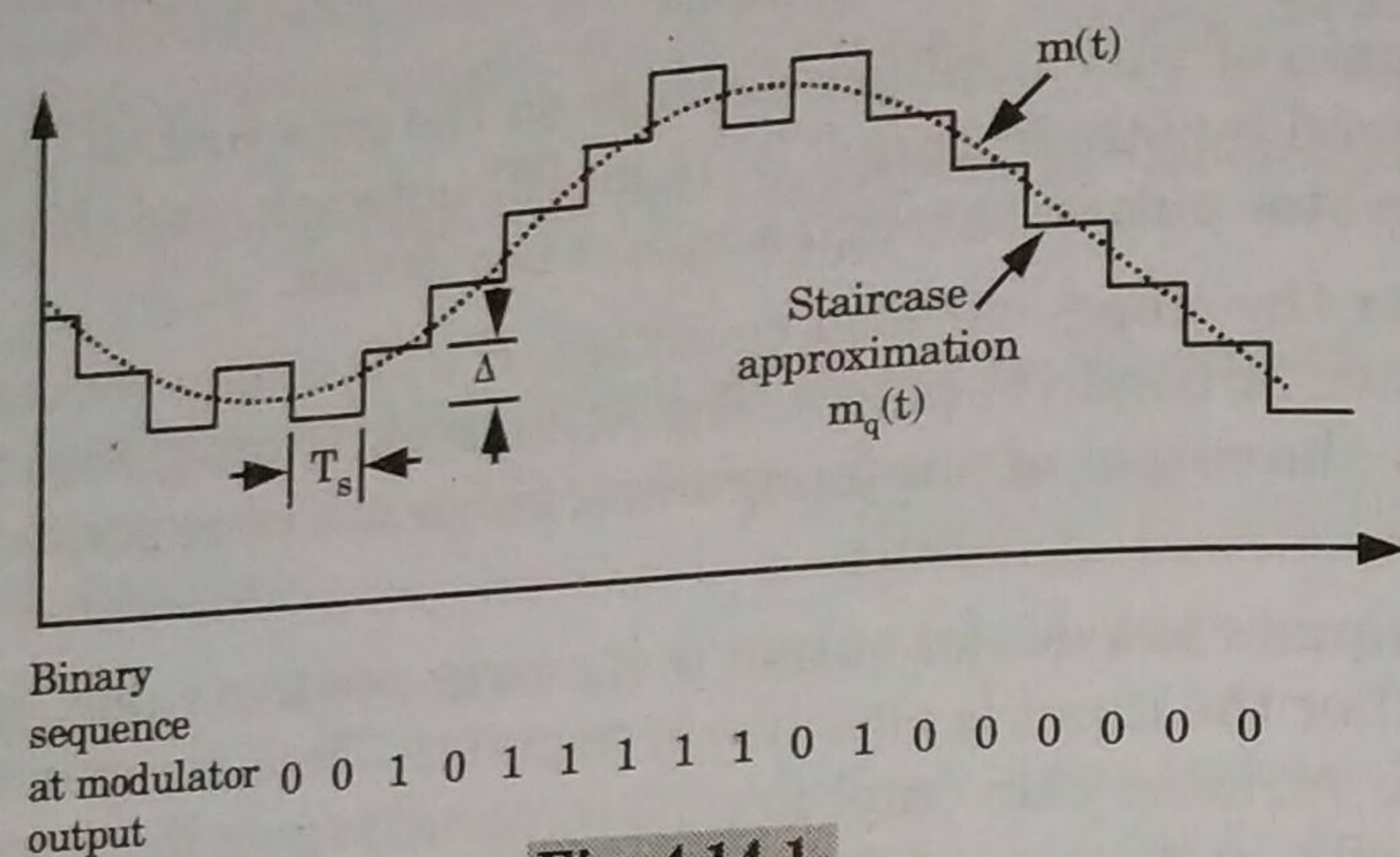


Fig. 4.14.1.

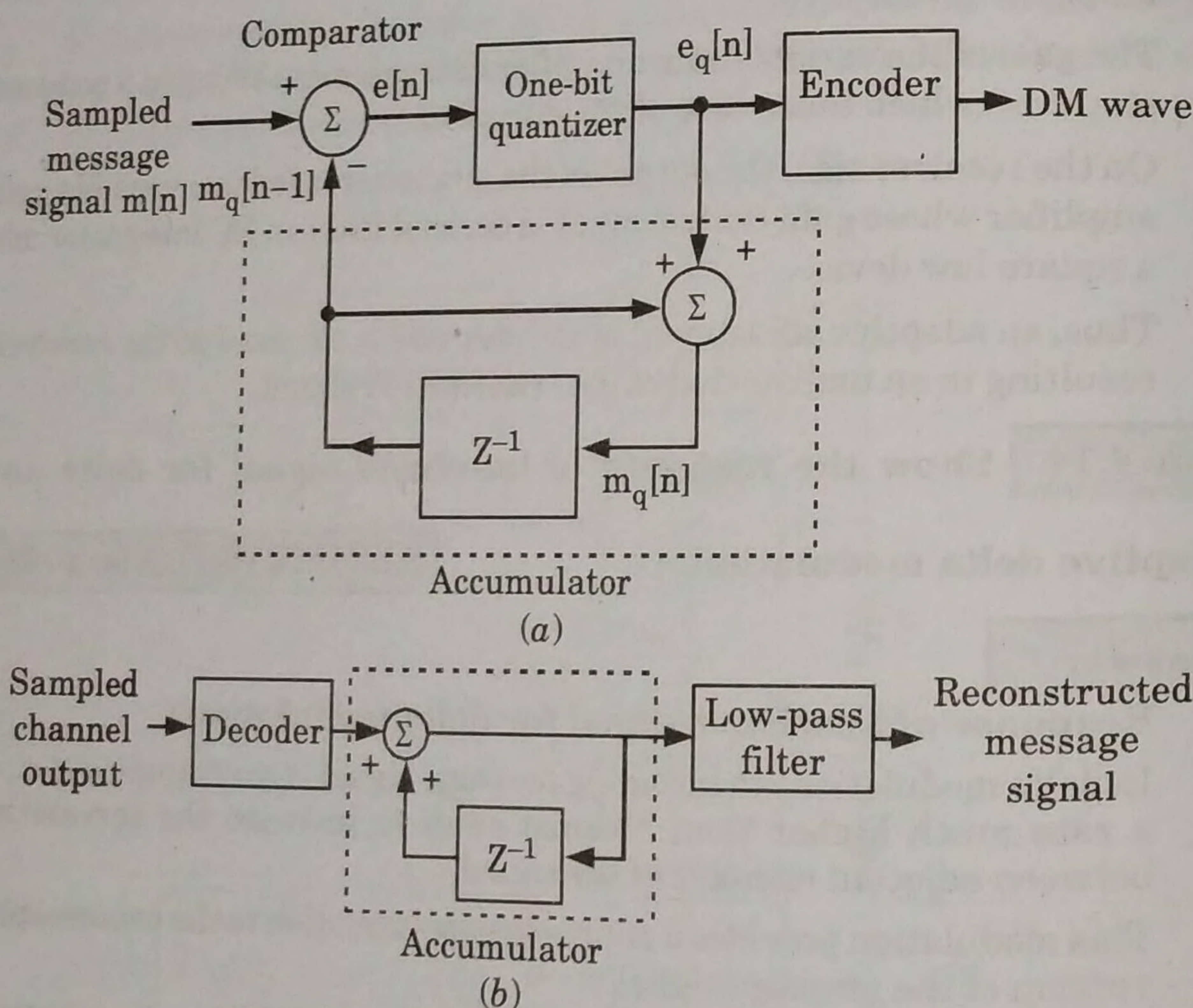


Fig. 4.14.2. DM system (a) Transmitter, (b) Receiver.

5. The basic principles of delta modulation is given as

$$e[n] = m[n] - m_q[n-1] \quad \dots(4.14.1)$$

$$e_q = \Delta \text{sgn}(e[n]) \quad \dots(4.14.2)$$

$$m_q[n] = m_q[n-1] + e_q[n] \quad \dots(4.14.3)$$

Here, $e[n]$ is an error signal representing the difference between the present sample $m[n]$ of the input signal and latest approximation $m_q[n-1]$ to it, $e_q[n]$ is the quantized version of $e[n]$, and $\text{sgn}(\cdot)$ is signum function.

6. Finally the quantizer output $m_q[n]$ is coded to produce the DM signal.
7. In delta modulation, incoming message signal is applied to modulator that involves a comparator, quantizer and accumulator interconnected.
8. The block located inside the accumulator represents a unit delay, i.e., a delay equal to one sampling period.
9. Here the comparator computes the difference between its two inputs.
10. The quantizer consists of hard-limiter with an input-output relation that is a scaled version of signum function. The quantizer output is then applied to an accumulator, producing the result.

$$m_q[n] = \Delta \sum_{i=1}^n \text{sgn}(e[i]) = \sum_{i=1}^n e_q[i]$$

11. Thus, at the sampling instant nT_s , the accumulator increments the approximation by a step Δ in a positive or negative direction, depending upon algebraic sign of the error sample $e[n]$.
12. If the input sample $m[n]$ is greater than the most recent approximation $m_q[n]$, a positive increment $+\Delta$ is applied to the approximation.
13. On the other hand, the input sample is smaller, a negative increment $-\Delta$ is applied to the approximation.
14. In the receiver shown Fig. 4.14.2(b), the staircase approximation $m_q(t)$ is reconstructed by passing the sequence of positive and negative pulses, produced at the decoder output, through an accumulator in a manner similar to that used in the transmitter.
15. The out of band quantization noise in the high-frequency staircase waveform $m_q(t)$ is rejected by passing it through a low-pass filter, as in Fig. 4.14.2(b), with a bandwidth equal to the original message bandwidth.

Response of baseband signal for adaptive delta modulation :

1. The principle underlying all ADM algorithms is two-fold :
 - i. If successive errors are of opposite polarity, then the delta modulator is operating in its granular mode; in this case, it may be advantages to reduce the step size.
 - ii. If, however, successive errors are of the same polarity, then the delta modulator is operating in its slope-overload mode; in this second case, the step size should be increased.

- Thus by varying the step-size in accordance with this principle, the delta modulator is enabled to cope with changes in the input signal.
- Fig. 4.14.3 the block diagram of an ADM is based on increasing or decreasing the step size by a factor of 50 % at each iterations of the adaptive process.
- The algorithm for adaptation of the step size is defined by

$$\Delta[n] = \begin{cases} \frac{|\Delta[n-1]|}{m_q[n]} (m_q[n] + 0.5m_q[n-1]), & \text{if } \Delta[n-1] \geq \Delta_{\min} \\ \Delta_{\min} & , \text{ if } \Delta[n-1] < \Delta_{\min} \end{cases}$$

where $\Delta[n]$ is the step size at iteration n of the algorithm, and $m_q[n]$ is 1 bit quantizer output that equals ± 1 .

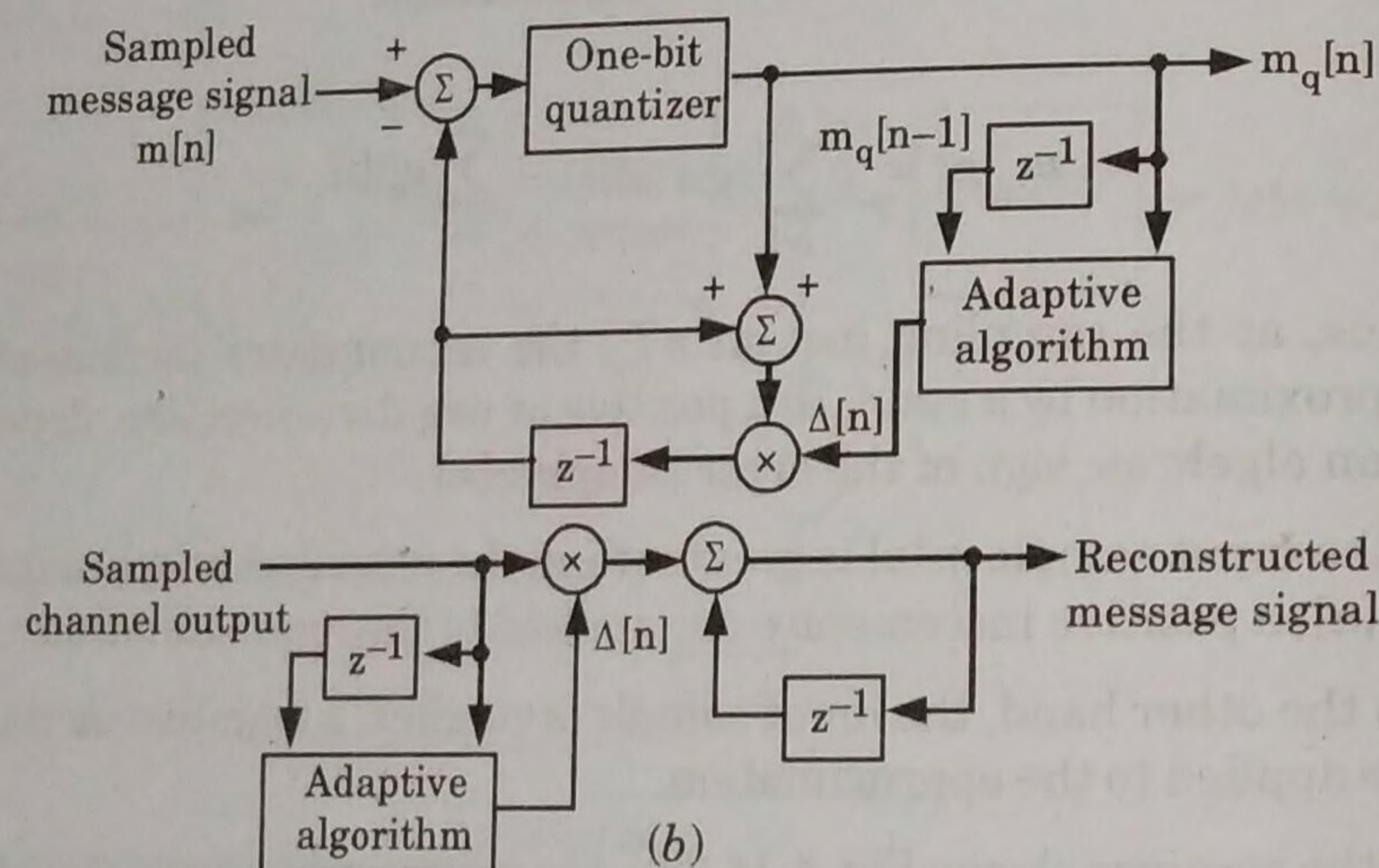


Fig. 4.14.3. Adaptive delta modulation system :
(a) Transmitter, (b) Receiver.

Que 4.15. Difference between delta modulation and adaptive delta modulation. The DM system is designed to operate at 3 times the Nyquist rate for a signal with a 3 kHz bandwidth. The quantizing step size is 250 mV.

- Determine the maximum amplitude of a 1 kHz input sinusoid for which the delta modulator does not show slope overload.
- Determine the post filtered output signal to quantizing noise ratio for the signal of part (a).

Answer

Difference :

S.No.	Data Modulation (DM)	Adaptive Delta Modulation (ADM)
1.	It uses only one bit for one sample.	Only one bit is used to encode one sample.
2.	Step size is fixed and cannot be varied.	According to the signal variation, step size varies (adapted).
3.	Slope overload distortion and granular noise is present.	Quantization error is present but other errors are absent.

Numerical :

Given : $f_m = 1 \text{ kHz}$; $\Delta = 250 \text{ mV}$; $f_s = 3 \times 2 \times 3 \times 10^3 = 18 \text{ kHz}$

To Find : a. Maximum amplitude, A_m
b. Signal to quantizing noise ratio, $(\text{SNR})_o$.

a. $m(t) = A \cos \omega_m t = A \cos 2\pi (10^3)t$

$$\left| \frac{dm(t)}{dt} \right|_{\max} = A(2\pi) (10^3)$$

The maximum allowable amplitude of the input sinusoid is

$$A_{\max} = \frac{\Delta}{\omega_m T_s} = \frac{\Delta}{\omega_m} f_s$$

$$A_{\max} = \frac{250}{2\pi(10^3)} \times 18 \times 10^3 = 716.2 \text{ mV}$$

- b. Assuming that the cut-off frequency of the low-pass filter is f_m , i.e., $f_M = f_m$, we have

$$(\text{SNR})_o = \left[\frac{S}{N_q} \right]_o$$

$$\begin{aligned} \left[\frac{S}{N_q} \right]_o &= \frac{3 f_s^3}{8 \pi^2 f_m^2 f_M} = \frac{3 f_s^3}{8 \pi^2 f_m^3} \quad (\text{Since } f_M = f_m) \\ &= \frac{3[(3)(6)(10^3)]^3}{8\pi^2(10^3)^3} = 221.6 = 23.5 \text{ dB} \end{aligned}$$

Que 4.16. Consider a DM system designed to accommodate analog message signals limited to bandwidth $W = 5$ kHz. A sinusoidal test signal of amplitude $A = 1$ volt and frequency $f_m = 1$ kHz is applied to the system. The sampling rate of the system is 50 kHz.

- Calculate the step size Δ required to minimize slope overload.
- Calculate the Signal to Noise Ratio of the system for the specified sinusoidal test signals.

Answer

Given : $A = 1$ V; $f_m = 1$ kHz; $W = 5$ kHz; $f_s = 50$ kHz

To Find : a. Step size, Δ
b. Signal to noise ratio, (SNR).

a.
$$A \leq \frac{\Delta f_s}{2\pi f_m}$$

$$\Delta \geq \frac{2\pi f_m A}{f_s}$$

$$\Delta \geq \frac{2 \times \pi \times 10^3 \times 1}{50 \times 10^3} = 0.126 \text{ V}$$

b.
$$(\text{SNR})_o = \frac{3}{8\pi^2} \frac{f_s^3}{f_m^2 W} = \frac{3}{8\pi^2} \times \frac{(50 \times 10^3)^3}{10^6 \times 5 \times 10^3} = 950$$

In decibel,

$$(\text{SNR})_o = 10 \log_{10} 950 = 29.77 \text{ dB}$$

PART-5

Noise Consideration in PCM.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.17. Calculate the quantization noise power in pulse code modulation.

AKTU 2014-15, Marks 05

Answer

- The quantising process gives rise to a random error which is referred as quantising noise.
- The mean-squared value of the quantising noise can be related to the signal power by relating the step size Δ , to the total amplitude range of the message signal.
- Let us consider a sinusoidal signal of peak amplitude E volt. Thus the amplitude range is $-E$ to E that is total $2E$.
- Let the total number of quantisation levels be Q . Then assuming $Q \gg 1$, we have,

$$\Delta = \frac{2E}{Q}$$

- The mean signal power is equal to $(E^2/2)$, while the mean quantising noise-power is $(\Delta^2/12)$. The quantising noise N_q will be

$$N_q = \frac{\Delta^2}{12} = \frac{1}{12} \left(\frac{2E}{Q} \right)^2 = \frac{E^2}{3Q^2}$$

- Therefore, the signal to quantization noise ratio will be

$$\frac{S}{N_q} = \frac{E^2/2}{E^2/3Q^2} = \frac{3}{2} Q^2$$

Thus, S/N_q can be increased by increasing the number of discrete levels Q .

Que 4.18. What is the effect of thermal noise ?

Answer

- The effect of additive thermal noise is to cause the matched filter detector to make an occasional error in determining whether a binary 1 or a binary 0 was transmitted.
- If the thermal noise is white and gaussian, the probability of such an error depends on the ratio E_b/η where E_b is the signal energy transmitted during a bit and $\eta/2$ is the two sided power spectral density of the noise.
- The error probability depends also on the type of modulation employed, i.e., direct transmission, PSK, FSK, etc.

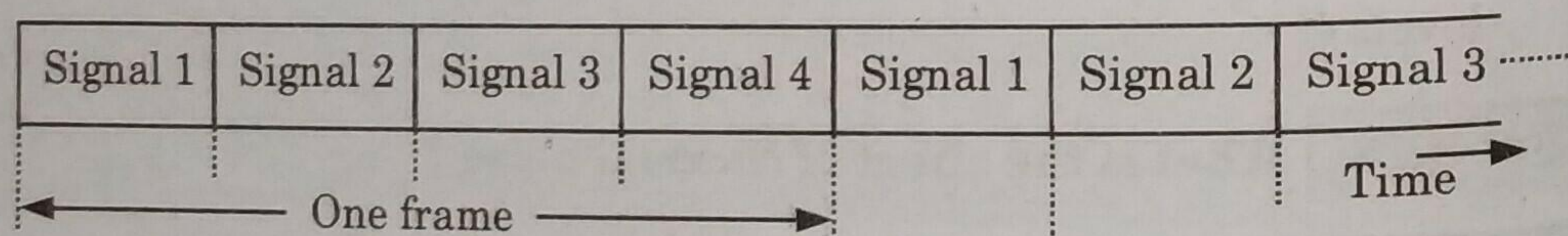
4. PCM systems operate with error probabilities which are small enough so that we may ignore the likelihood that more than a single bit error will occur within a single word.

PART-6*Time Division Multiplexing.***Questions-Answers****Long Answer Type and Medium Answer Type Questions**

Que 4.19. Explain the TDM principle. Draw a circuit diagram illustrate the flat-top sampling. Draw the block diagram of PCM scheme and explain its functionality. **AKTU 2015-16, Marks 10**

Answer**TDM (Time Division Multiplexing) :**

1. In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one.
2. Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.
3. The TDM principle is illustrated in Fig. 4.19.1.

**Fig. 4.19.1.** Principle of TDM.

4. As shown in the Fig. 4.19.1 one transmission of each channel completes one cycle of operation called as a "Frame".
5. For a TDM, the data rate of the multiplexed signal is always n times the data rate of individual sources, where n is the number of sources.

Flat top sampling : Refer Q. 4.3, Page 4-4B, Unit-4.

PCM : Refer Q. 4.5, Page 4-6B, Unit-4.

Que 4.20. Discuss the multiplexing of PCM system. Describe T1 digital system in detail.

OR

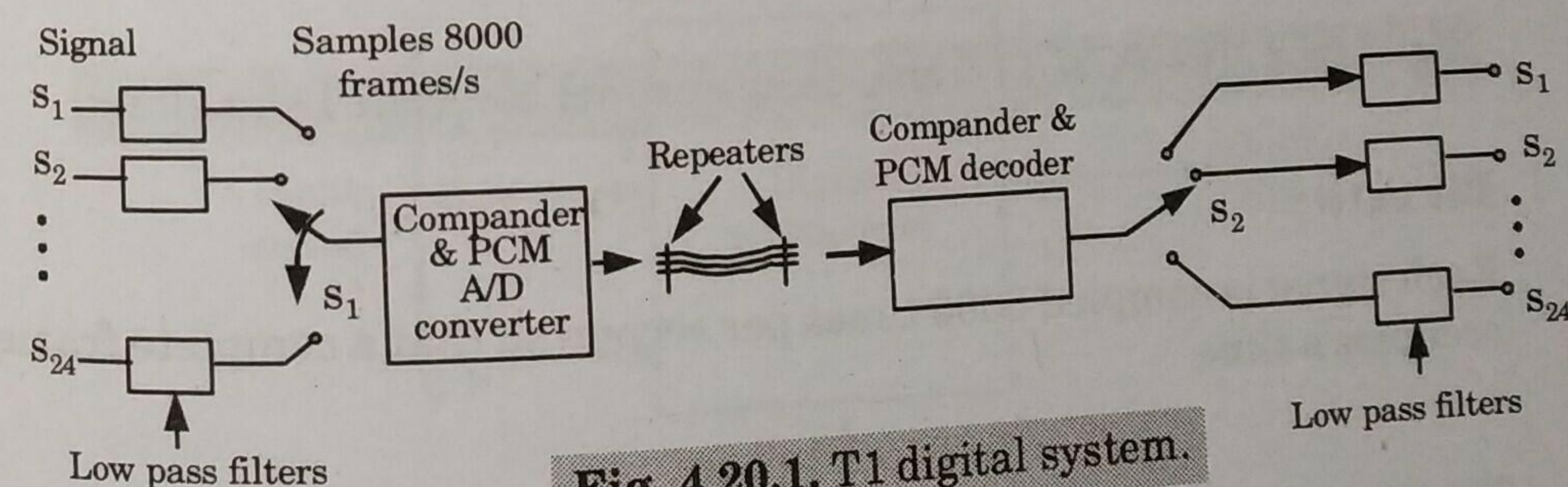
Write a technical note on TDM for T1 system, with the help of a neat diagram.

AKTU 2016-17, Marks 10**Answer**

1. In PCM, time division multiplexing is possible. In this, each sample is coded into several bits.
2. The multiplexing is possible in two ways :
 - i. Bits are taken, one by one from each channel sample code. When the first bits from all channel samples are taken, the commutator takes the second bits from all channel samples, and so on.
 - ii. All code bits of the first channel samples are taken followed by all code bits of the second channel samples, and so on. In this method, the desired commutator speed is less than that required in the first method.
3. A synchronizing bit is added at the end of each frame for synchronization between commutator and decommutator.
4. The signal that is to be time division multiplexed is band limited to the same frequency, resulting in the same sampling frequency for all the channels, and hence the name synchronous time division multiplexing.

T1 digital system :

1. T1 digital system is used to convey multiple signals over telephone lines using wideband co-axial cable.
2. It accommodates 24 analog signals which refer as S_1 through S_{24} .

**Fig. 4.20.1.** T1 digital system.

3. Each signal is band limited to approximately 3.3 kHz and is sampled at the rate of 8 kHz. Each of the time division multiplexed signal (still analog) is next A/D converted and companded.
4. The resulting digital waveform is transmitted over a co-axial cable, the cable serving to minimize signal distortion and serving to suppress signal corruption due to noise from external sources.

- The repeater retransmits a distortionless and noise-free duplicate of the signal originally sent.
- At the destination the signal is companded, decoded and demultiplexed, making available, individually the 24 original signals.

Bits/frame :

- The commutator sweeps continuously from S_1 to S_{24} and back to S_1 , etc., at the rate of 8000 revolutions per second thereby providing 8000 samples per second of each signal.
- Each sample is encoded into eight bits (corresponding to $2^8 = 256$ quantization levels).
- The digital signal generated during the course of one complete sweep of the commutator is therefore $24 \times 8 = 192$ bits.

Frame synchronization :

- To provide synchronization an extra single bit is made available immediately preceding the 192 bits that carry the encoded signals.
- The 192 bit slots assigned to the encoded signal together with the one extra timing bit, for a total of 193 bits, is called a frame.
- The time slots for the 24 signals together with the extra frame-synchronizing bit F is shown in Fig. 4.20.2.

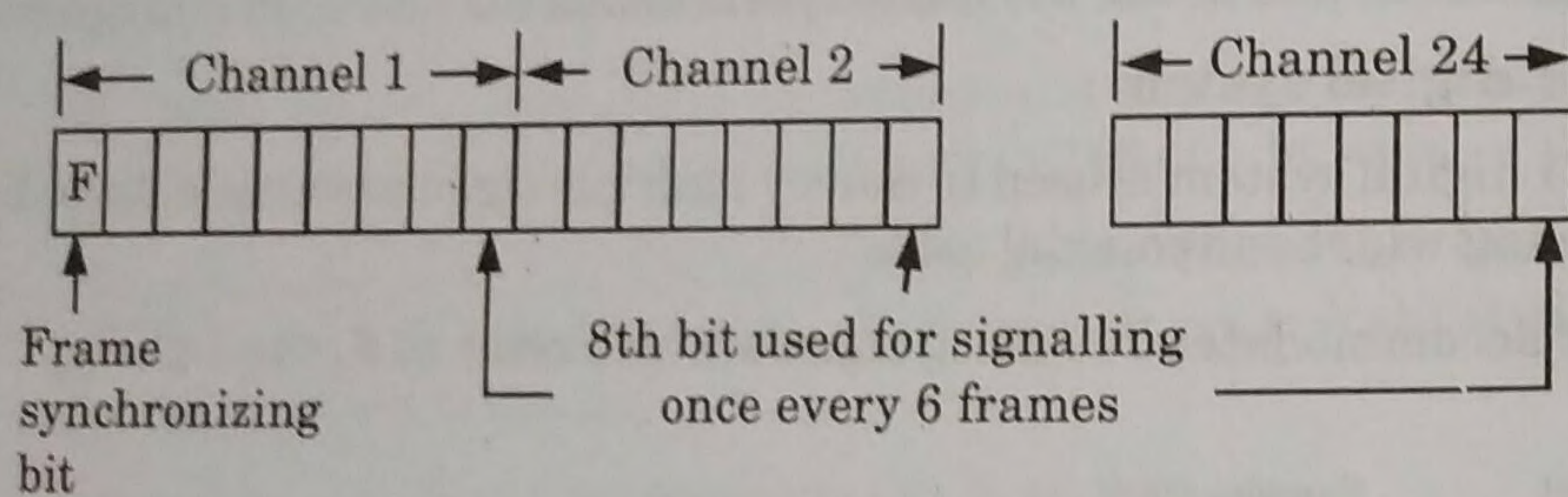


Fig. 4.20.2. The PCM T1 frame using channel associated signalling.

Bit rate :

- Each signal is sampled 8000 times per second so that a complete frame occupies a time

$$T_p = 1/8000 = 125 \mu s$$

- This time T_p accommodates 193 bits so that the bit rate on a T1 channel is

$$f_b(T1) = \frac{193}{125} \text{ Mb/s} = 1.544 \text{ Mb/s}$$

Signaling : The frequency of the bits used for signaling is $\frac{1}{6}$ th of the frame bit rate, or

$$f_b(T1)_{\text{(signaling)}} = \frac{1}{6}(8000) = 1,333 \text{ Hz}$$

This type of signaling is called channel-associated signaling.

Line coding for T1 system : The line code used for T1 system is AMI waveform.

PART-7

Digital Multiplexers.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 4.21. Explain digital multiplexing.

Answer

- A digital multiplexer has N input and only one output.
- By applying control signals any one input can be made available at the output terminal.

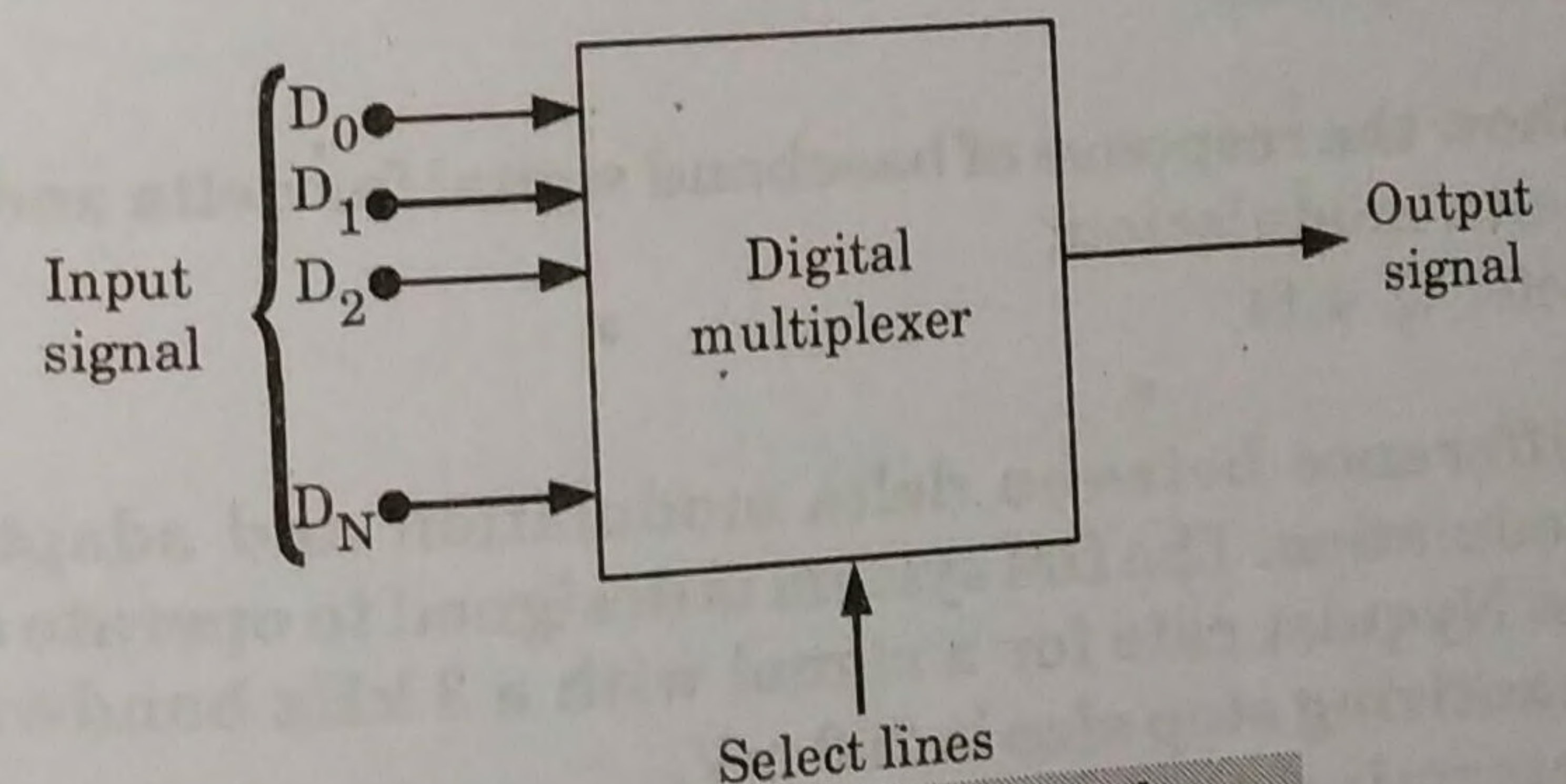


Fig. 4.21.1. Digital multiplexer.

- It is also called data selector. Fig. 4.21.1 shows the schematic diagram of a digital multiplexer.
- The control signals are applied to the select lines to select the desired input.

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

Q. 1. What is pulse modulation and what are their different types? Also give advantage of pulse modulation.

Ans. Refer Q. 4.1.

Q. 2. Describe the sampling techniques and signal recovery through holding used in PAM.

Ans. Refer Q. 4.3.

Q. 3. What do you understand by PCM (pulse code modulation)? Explain basic elements and operation of the PCM system with its respective block diagram.

Ans. Refer Q. 4.5.

Q. 4. Differentiate between PAM and PCM.

Ans. Refer Q. 4.7.

Q. 5. What is differential pulse code modulation? Explain working of DPCM with a proper block diagram.

Ans. Refer Q. 4.8.

Q. 6. Explain working of adaptive delta modulation with a proper block diagram.

Ans. Refer Q. 4.13.

Q. 7. Show the response of baseband signal for delta and adaptive delta modulation.

Ans. Refer Q. 4.14.

Q. 8. Difference between delta modulation and adaptive delta modulation. The DM system is designed to operate at 3 times the Nyquist rate for a signal with a 3 kHz bandwidth. The quantizing step size is 250 mV.

a. Determine the maximum amplitude of a 1 kHz input sinusoid for which the delta modulator does not show slope overload.

b. Determine the post filtered output signal to quantizing noise ratio for the signal of part (a).

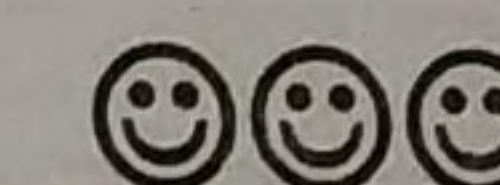
Ans. Refer Q. 4.15.

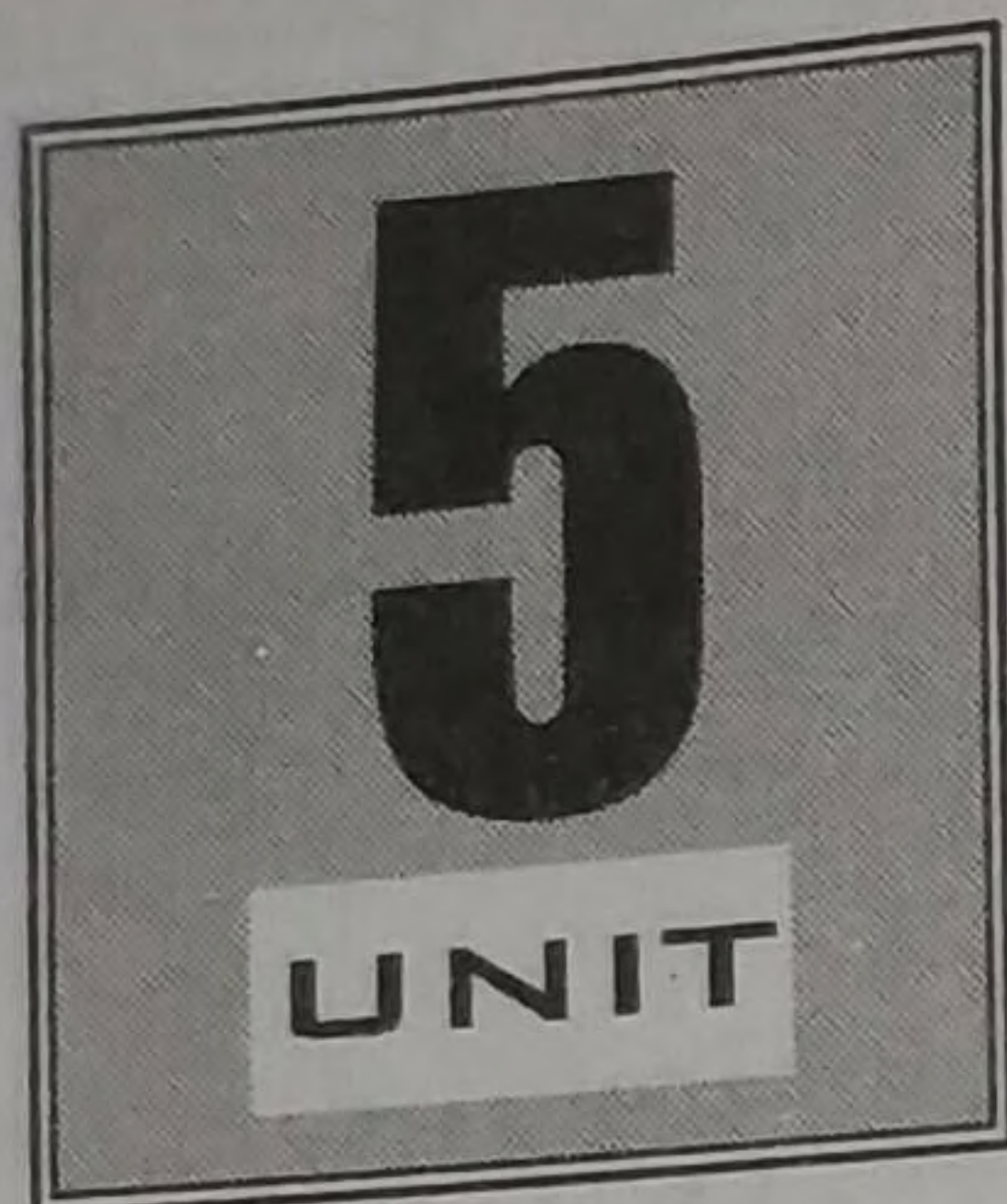
Q. 9. Calculate the quantization noise power in pulse code modulation.

Ans. Refer Q. 4.17.

Q. 10. Explain the TDM principle. Draw a circuit diagram illustrate the flat-top sampling. Draw the block diagram of PCM scheme and explain its functionality.

Ans. Refer Q. 4.19.





Digital Modulation Scheme

CONTENTS

Part-1 : Digital Modulation 5-2B to 5-13B
Schemes-Phase Shift
Keying, Frequency
Shift Keying, Quadrature
Amplitude Modulation

Part-2 : Continuous Phase 5-13B to 5-18B
Modulation and
Minimum Shift Keying

PART-1

Digital Modulation Schemes-Phase Shift Keying, Frequency Shift Keying, Quadrature Amplitude Modulation.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.1. Explain the need of digital modulation. Describe the types of digital modulation. Draw and explain the waveforms for amplitude, frequency and phase shift keying methods.

AKTU 2018-19, Marks 10

OR

Write short note on ASK, FSK and PSK.

Answer

Digital modulation :

1. Modulation is defined as the process by which some characteristic of a carrier is varied in accordance with a modulating wave.
2. In digital communication, the modulating wave consists of binary data.
3. For the carrier, it is necessary to use a sinusoidal wave.
4. With a sinusoidal carrier, the feature that is used by modulator to distinguish one signal from another is a step change in the amplitude, frequency or phase of the carrier.

Need :

1. Used in image processing for pattern recognition robotic version and image enhancement.
2. Used in digital signal processing.
3. In telephone for text messaging.

Types of digital modulation :

a. Amplitude shift keying :

1. The amplitude of the resultant output depends upon the input data whether it should be a zero level or a variation of positive and negative, depending upon the carrier frequency.
2. It represents the binary data in the form of variations in the amplitude of a signal.

3. The binary signal when ASK is modulated, gives a zero value for LOW input and gives the carrier output for HIGH input.
- b. Frequency shift keying :**
 1. The frequency of the output signal will be either high or low, depending upon the input data applied.
 2. In this the frequency of the carrier signal varies according to the discrete digital changes.
 3. The output of a FSK modulated wave is high in frequency for a binary HIGH input and is low in frequency for a binary LOW input.
- c. Phase shift keying :** The phase of the output signal gets shifted depending upon the input.

Waveforms of ASK, PSK and FSK :

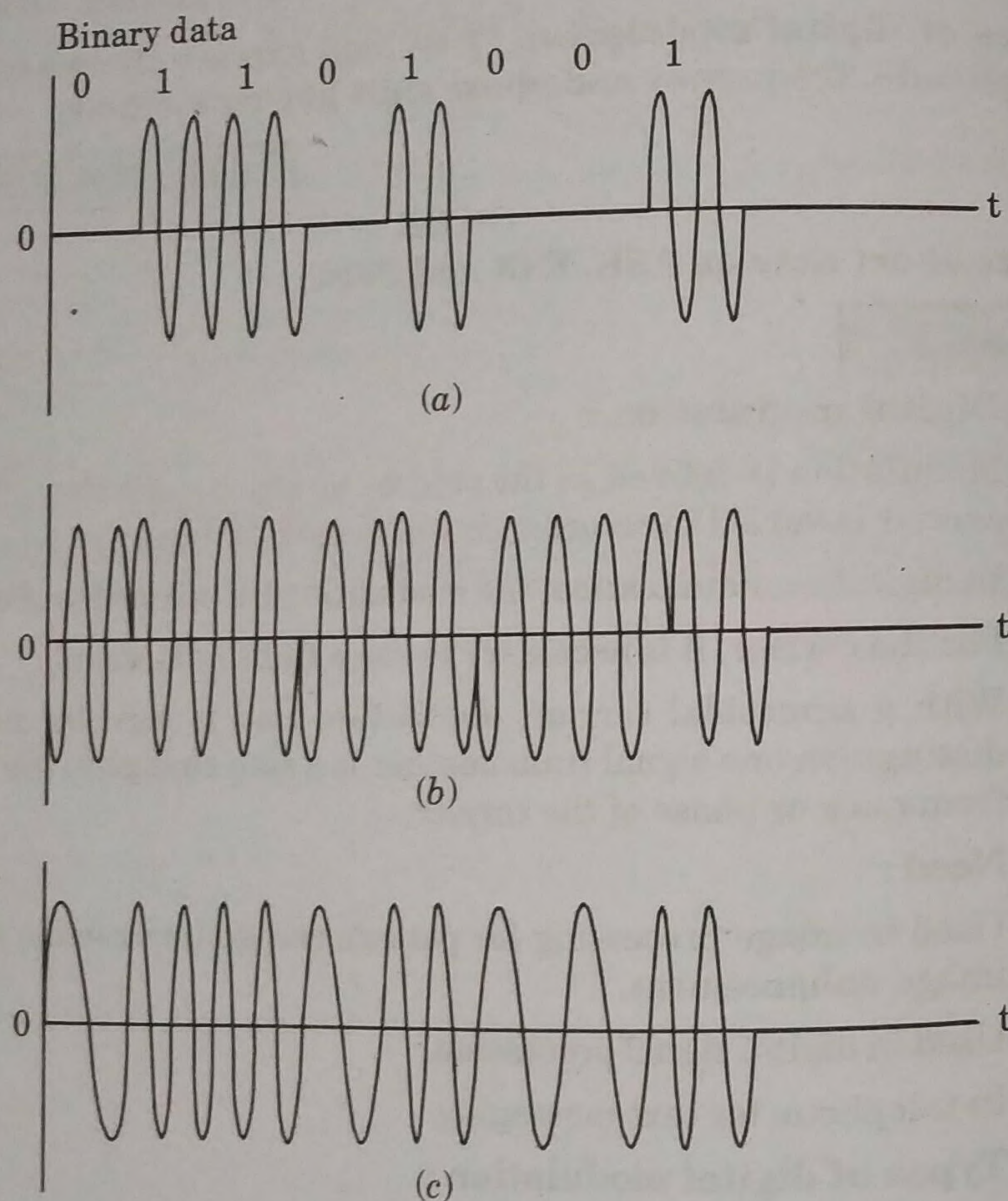


Fig. 5.1.1. Waveforms for (a) Amplitude shift keying, (b) Phase shift keying, (c) Frequency shift keying.

Que 5.2. Describe the generation and detection of phase shift keying.

OR

Explain with the help of diagram, a method of generating and demodulating phase shift keying signals.

Answer

1. In a coherent binary PSK system, the pair of signals, $S_1(t)$ and $S_2(t)$ are used to represent binary symbols 1 and 0, respectively.

$$2. \text{ It is given by } S_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t)$$

$$S_2(t) = -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t)$$

where, $0 \leq t < T_b$ and E_b is transmitted signal energy per bit.

3. In order to ensure that each transmitted bit contains an integral number of cycles of the carrier wave, the carrier frequency f_c is chosen equal to n_c/T_b for some fixed integer n_c .

Generation :

1. To generate a binary PSK wave, the input binary sequence is represented in polar form with symbols 1 and 0 represented by constant amplitude levels of $+\sqrt{E_b}$ and $-\sqrt{E_b}$.

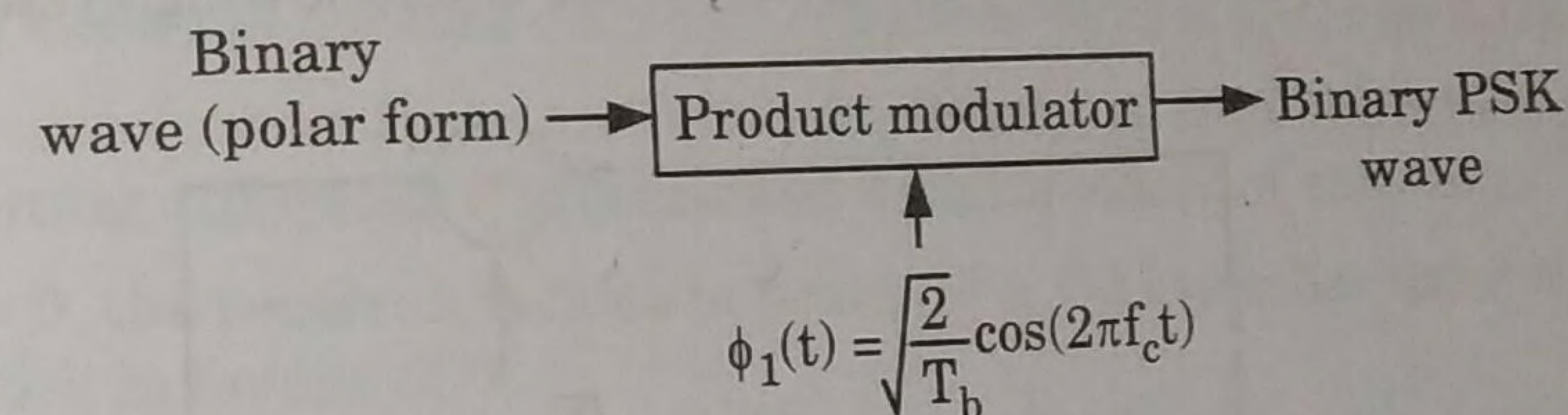


Fig. 5.2.1.

2. The binary wave and sinusoidal carrier wave $\phi_1(t)$ are applied to product modulator.
3. The carrier and the timing pulses used to generate the binary wave are usually extracted from a common master clock. The desired PSK wave is obtained at output.

Detection of coherent Binary PSK :

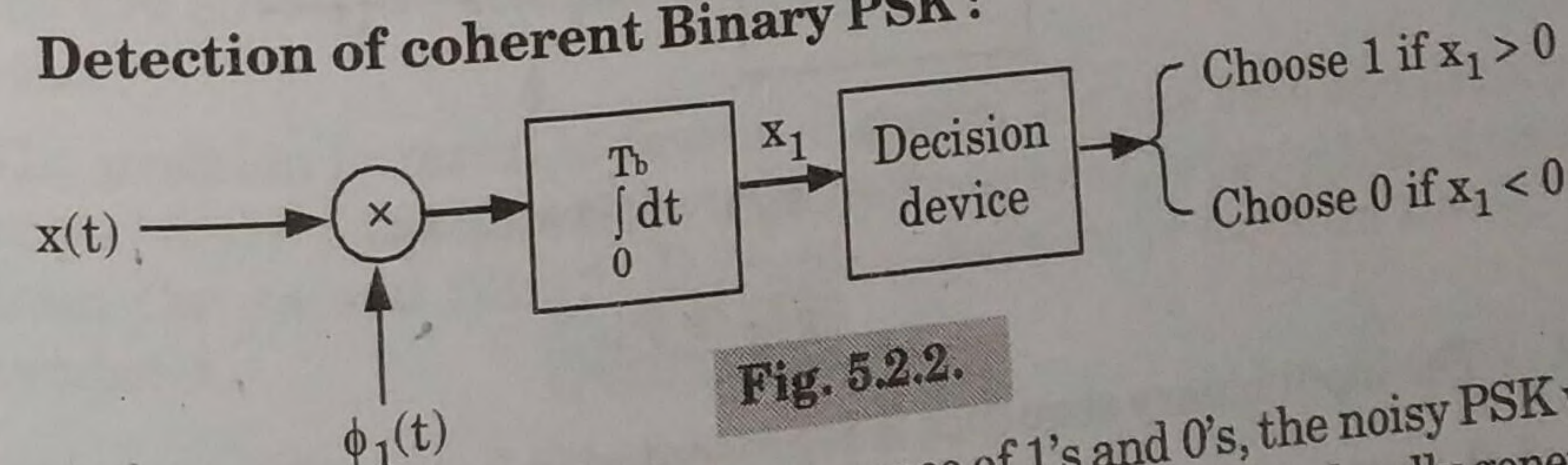


Fig. 5.2.2.

1. To detect the original binary sequence of 1's and 0's, the noisy PSK wave $x(t)$ is applied to a correlator, which is also supplied with a locally generated coherent reference signal $\phi_1(t)$.

- The correlator output x_1 , is compared with a threshold of zero volts.
- If $x_1 > 0$, the receiver decides in favour of symbol 1. On the other hand, if $x_1 < 0$, it decides a favour of symbol 0.

Que 5.3. Give and explain the method for generation and detection of BFSK (Binary Frequency Shift Keying).

OR

Explain the generation and detection of FSK.

AKTU 2016-17, Marks 7.5

Answer

In a binary FSK system, symbols 1 and 0 are distinguished from each other by transmitting one of two sinusoidal waves that differ in frequency by a fixed amount. A typical pair of sinusoidal waves is described by

$$S_i(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_i t) & 0 \leq t \leq T_b \\ 0 & \text{elsewhere} \end{cases}$$

where $i = 1, 2$ and E_b is the transmitted signal energy per bit.

Generation of BFSK :

- To generate a binary FSK signal we use the following system :

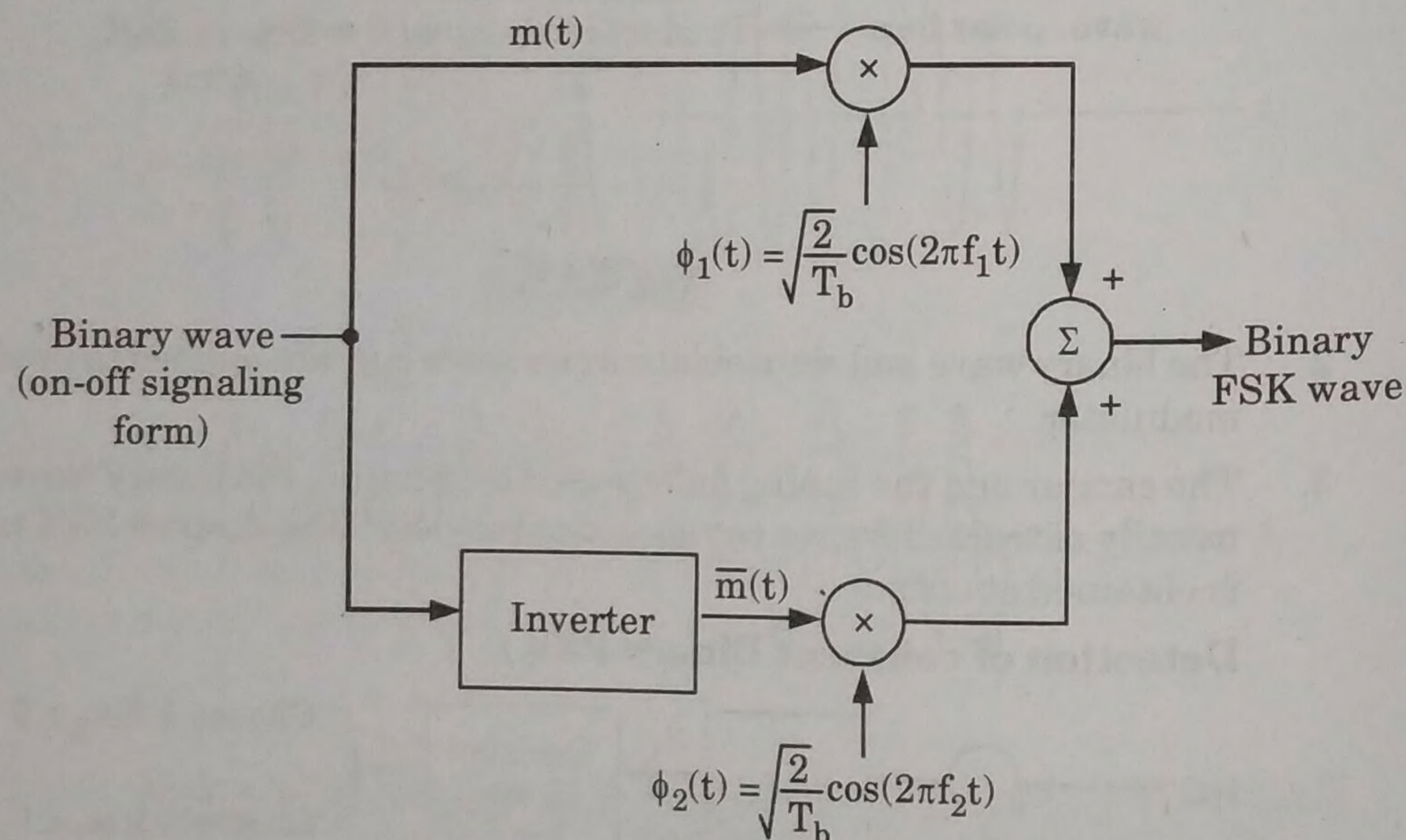


Fig. 5.3.1.

- The input binary sequence is represented in its on-off form, with symbol 1 represented by constant amplitude of $\sqrt{E_b}$ volts and symbol 0 represented by zero volts.

- By using an inverter in the lower channel, we make sure that when symbol 1 is present at the input, the oscillator with frequency f_1 in the upper channel is switched on while the oscillator with frequency f_2 in the lower channel is switched off, with the result that frequency f_1 is transmitted.
- Conversely, when symbol 0 is present at the input, the oscillator in the upper channel is switched off, and the oscillator in the lower channel is switched on, with the result that frequency f_2 is transmitted.
- The two frequencies f_1 and f_2 are chosen to equal integer multiples of the bit rate $1/T_b$.

Coherent Binary FSK detection :

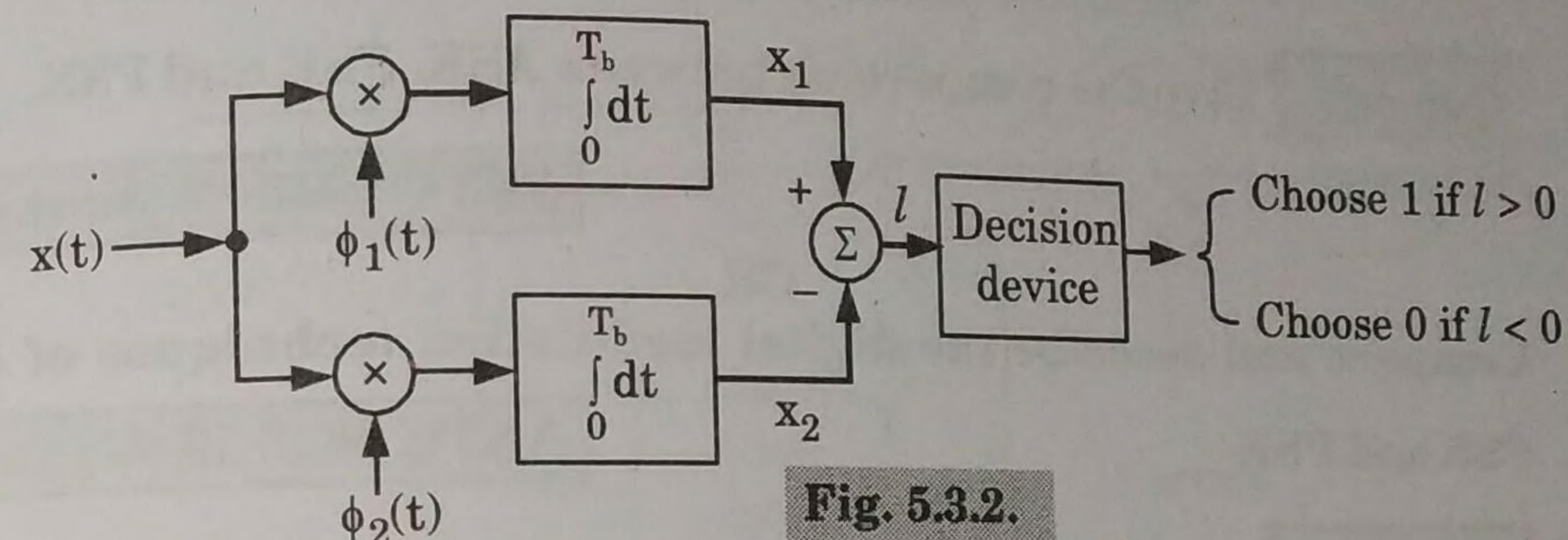


Fig. 5.3.2.

- In order to detect the original binary sequence given, the noisy received wave $x(t)$ at the receiver is shown in Fig. 5.3.2.
- It consists of two correlators with a common input, which are supplied with locally generated coherent reference signals $\phi_1(t)$ and $\phi_2(t)$.
- The correlator outputs are then subtracted, one from the other, and the resulting difference l , is compared with threshold of zero volts.
- If $l > 0$, the receiver decides in favour of 1. On the other hand if $l < 0$, it decides in favour of 0.

Non-coherent detection of BFSK :

- Binary FSK waves may be demodulated non-coherently using envelope detector.
- The received FSK signal is applied to a bank of two bandpass filters, one tuned to frequency f_1 and other tuned to frequency f_2 .
- Each filter is followed by an envelope detector. The resulting outputs of the two envelope detectors are sampled and then compared with each other.
- The decision is made in favour of symbol 1 if the envelope detector output derived from the filter tuned to frequency f_1 is larger than derived from the second filter. Otherwise a decision is made in favour of symbol 0.

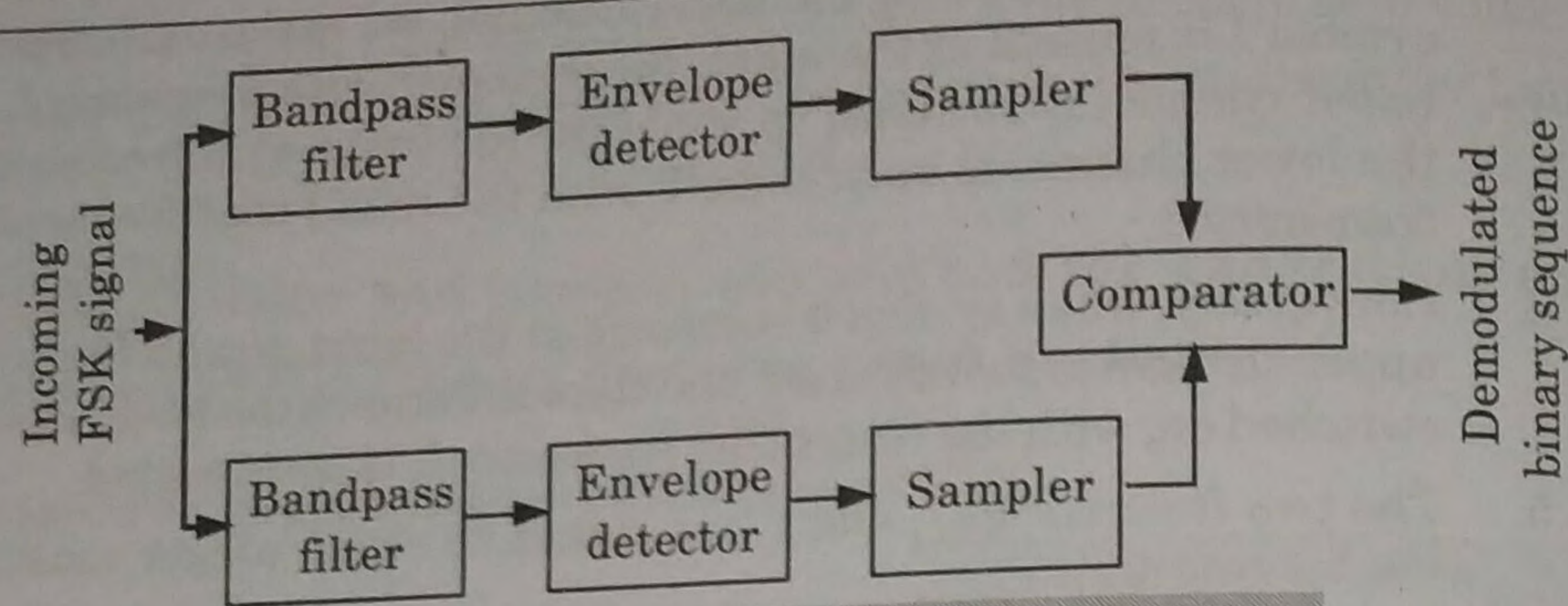


Fig. 5.3.3. Non-coherent detection of BFSK.

Que 5.4. Give the comparison between ASK, FSK and PSK.

AKTU 2016-17, Marks 7.5

OR

Compare and describe the digital modulation techniques of ASK, FSK and PSK

AKTU 2018-19, Marks 10

Answer

ASK, FSK and PSK : Refer Q. 5.1, Page 5-2B, Unit-5.

Comparison :

S. No.	ASK	FSK	PSK
1.	In this modulation technique, amplitude of carrier signal is varied while frequency and phase remain constant.	Here, the frequency of carrier signal is varied while amplitude and phase remain constant.	Here, the phase of carrier signal is varied while frequency and amplitude remain constant.
2.	This scheme is advantageous only in terms of simplicity and low implementation cost.	FSK is less susceptible to errors than ASK, as the receiver is looking for frequency changes, so noise spikes can be ignored.	PSK is extremely advantageous as it is less susceptible to errors than ASK. It is very robust in terms of noise tolerance.
3.	ASK is commonly used to transmit digital data over optical fiber and is a very popular modulation used in control application.	FSK is popularly used in data transmission over voice lines, early modem and in high frequency radio transmission.	PSK is commonly used in signaling channels and in many systems like DTV, WLAN etc.
4.	For demodulation, only the presence or absence of a sinusoid in a given time interval needs to be determined.	For demodulation, the demodulator must be able to determine which of the two possible frequencies is present at a given time.	The PSK demodulator must be able to determine the phase of the received sinusoid with respect to some reference phase.

Que 5.5. With suitable diagram explain DPSK modulator and demodulator.

Answer

A. DPSK Modulator :

- DPSK is a technique of BPSK, in which there is no reference phase signal. Here, the transmitted signal itself can be used as a reference signal.
- A DPSK Modulator is shown in Fig. 5.5.1.

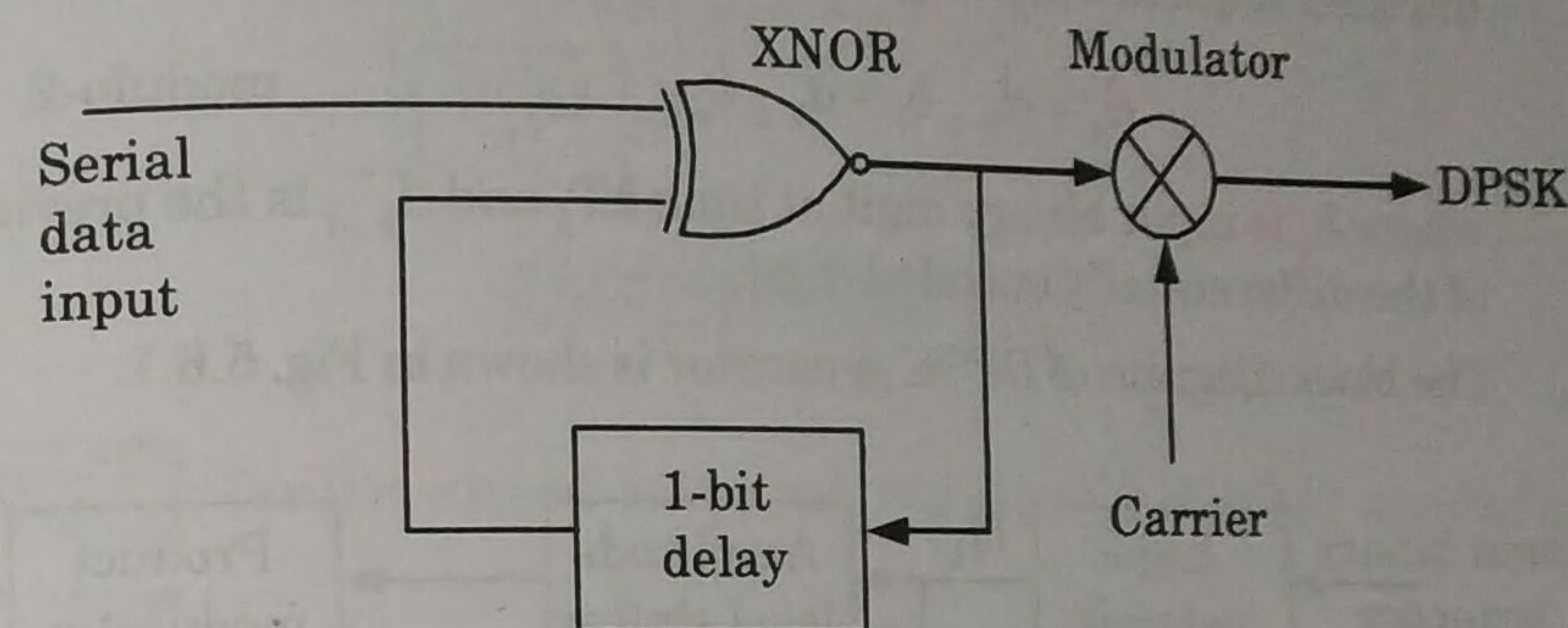


Fig. 5.5.1. DPSK modulator.

- DPSK encodes two distinct signals, i.e., the carrier and the modulating signal with 180° phase shift each. The serial data input is given to the XNOR gate and the output is again feedback to the other input through 1-bit delay.
- The output of the XNOR gate along with the carrier signal is given to the balance modulator, to produce the DPSK modulated signal.

ii. DPSK Demodulator :

- In DPSK demodulator, the phase of the reversed bit is compared with the phase of the previous bit.
- The block diagram of DPSK demodulator is shown in Fig. 5.5.2.

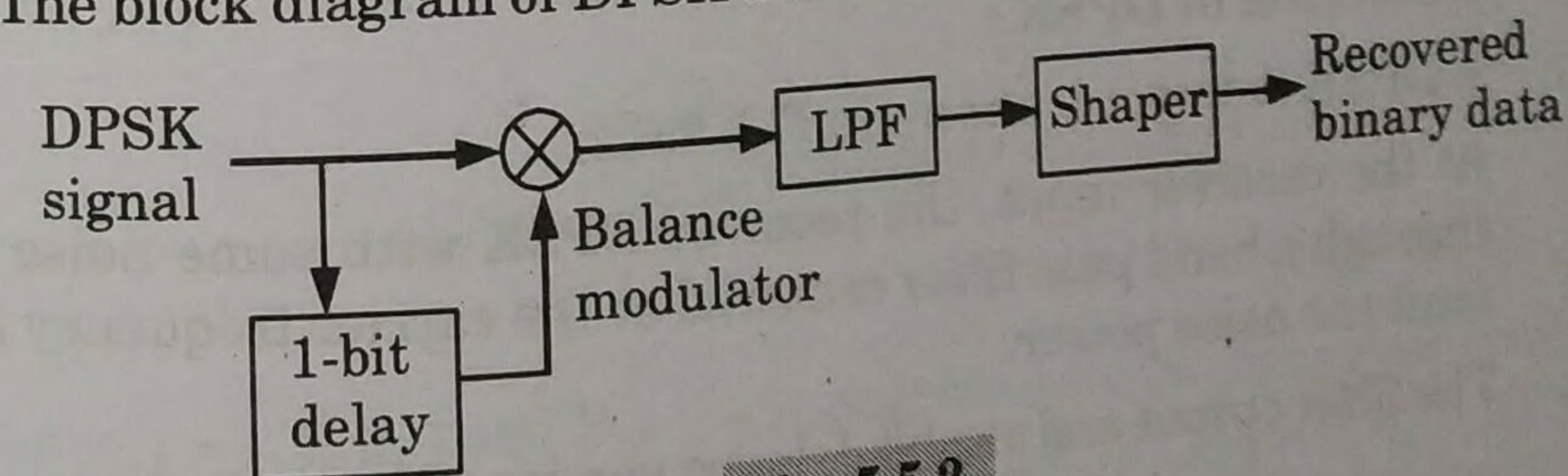


Fig. 5.5.2.

- From the Fig. 5.5.2, it is evident that the balance modulator is given the DPSK signal along with 1-bit delay input, that signal is made to confine to lower frequencies with the help of LPF.

4. Then it is passed to a shaper circuit, which is a comparator or a Schmitt trigger circuit, to recover the original binary data as the output.

Que 5.6. Explain with the help of neat diagram, the method for generation and demodulation of DPSK.

Answer

i. Generation of DPSK :

1. The differential encoding process at the transmitter input starts with an arbitrary first bit, serving as reference, and thereafter the differentially encoded sequence $\{d_k\}$ is generated by using the logical equation

$$d_k = d_{k-1} b_k + \bar{d}_{k-1} \bar{b}_k \quad \text{modulo-2} \quad \dots(5.6.1)$$

where b_k is input binary digit at time kT_b and d_{k-1} is the previous value of the differentially encoded digit.

2. The block diagram of DPSK generator is shown in Fig. 5.6.1.

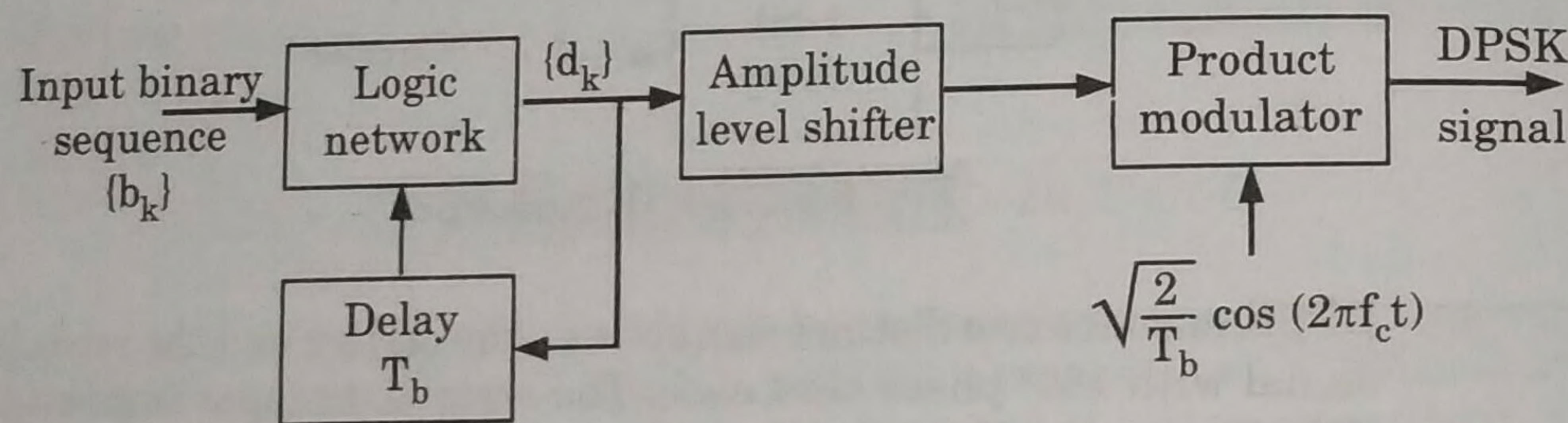


Fig. 5.6.1. DPSK transmitter.

3. It consists of a logic network and a one bit delay element inter-connected so as to convert an input binary sequence $\{b_k\}$ into a differentially encoded sequence $\{d_k\}$ in accordance with eq. (5.6.1).
4. This sequence is amplitude level shifted and then used to modulate a carrier wave of frequency f_c thereby producing the desired DPSK wave.

ii. Detection of DPSK :

1. A DPSK detector is shown in Fig. 5.6.2.
2. At the receiver input, the received DPSK with some noise is passed through a band pass filter centered at the carrier frequency f_c , so as to limit the noise power.
3. The filter output and one bit delayed version of it, applied to a correlator.
4. The resulting correlator output is proportional to the cosine of the difference between the carrier phase angles in the two correlators inputs.

5. The correlator output is finally compared with a threshold of zero volts, and a decision is thereby made in favour of symbol 1 or symbol 0. If the correlator output is positive then the receiver decides in favour of symbol 1. If the correlator output is negative, the receiver decides in favour of symbol 0.

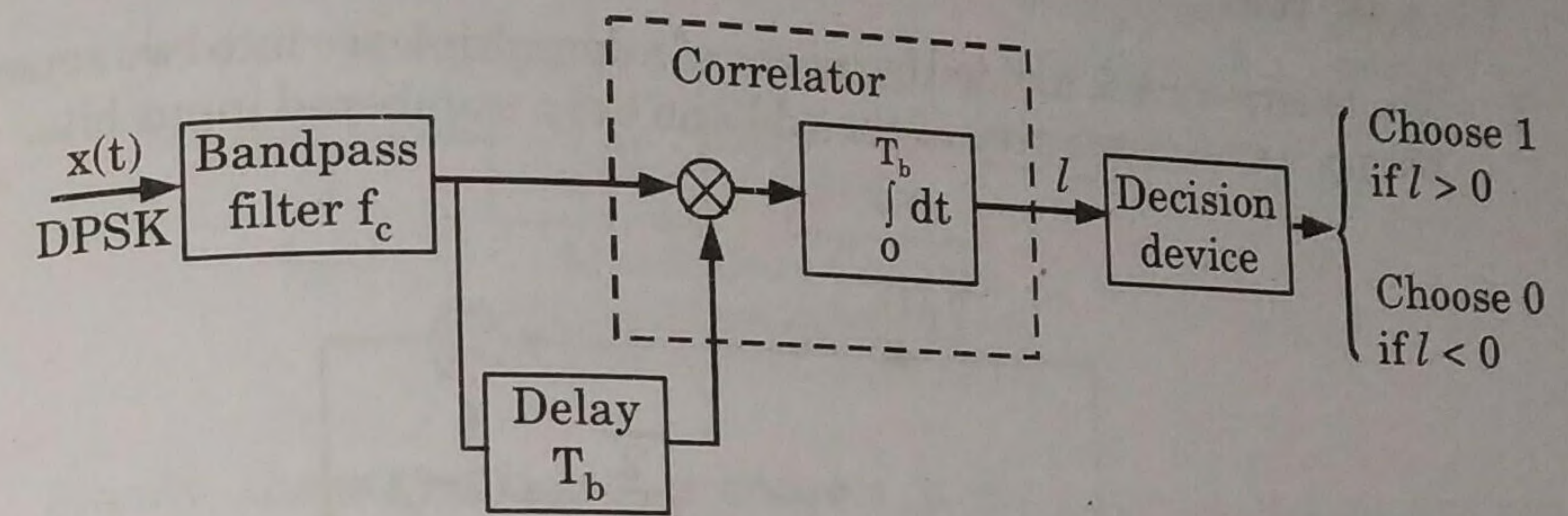


Fig. 5.6.2. DPSK receiver.

Que 5.7. Write short note on following digital modulation techniques :

- i. Differential phase shift keying (DPSK).
ii. Quadrature phase shift keying (QPSK).

OR

With the help of block diagram, explain QPSK coherent digital carrier system. Sketch the QPSK waveform for the sequence 1101010010, assuming the carrier frequency to be equal to the bit rate.

Answer

A. DPSK : Refer Q. 5.6, Page 5-9B, Unit-5.

B. QPSK :

1. In quadrature phase-shift keying (QPSK), the phase of the carrier takes on one of four equally spaced values, such as $\pi/4$, $3\pi/4$, $5\pi/4$ and $7\pi/4$ as shown by

$$S_i(t) = \begin{cases} \sqrt{\frac{2E}{T}} \cos \left[2\pi f_c t + (2i-1)\frac{\pi}{4} \right] & 0 \leq t \leq T \\ 0 & \text{elsewhere} \end{cases}$$

where, $i = 1, 2, 3, 4$ and E is the transmitted signal energy per symbol, T is the bit duration, and the carrier frequency f_c equals n_c/T for some fixed integer n_c .

2. Each possible value of the phase corresponds to a unique pair of bits called dibit.

i. QPSK transmitter :

1. In a typical QPSK transmitter, the input binary sequence $b(t)$ is represented in polar form, with symbols 1 and 0 represented by $+\sqrt{E_b}$ and $-\sqrt{E_b}$ volts.
2. The binary wave is divided by means of a demultiplexer into two separate binary waves consisting of the odd and even numbered input bits.

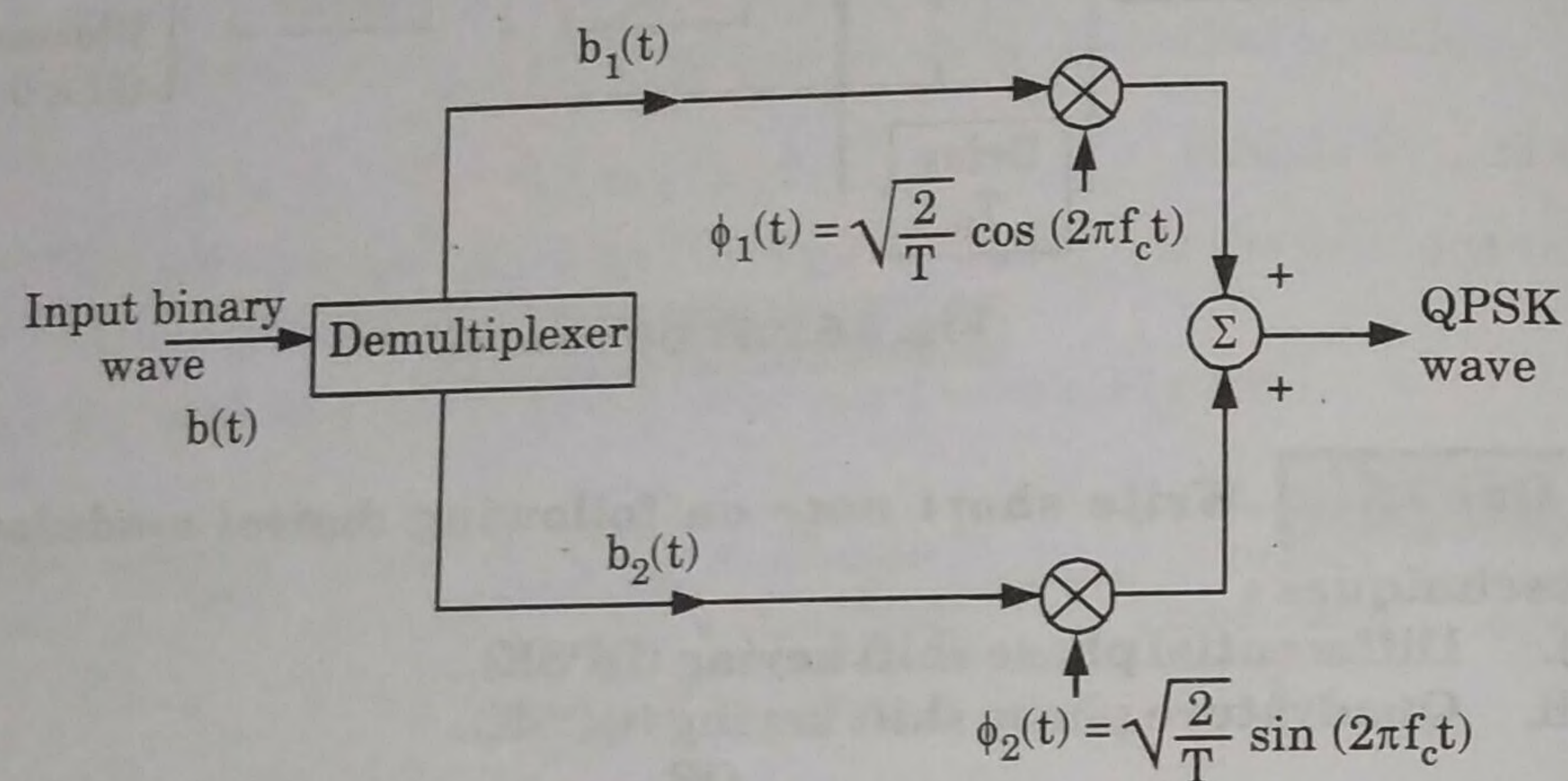


Fig. 5.7.1. QPSK generator.

3. These two binary waves are denoted by $b_1(t)$ and $b_2(t)$.
4. The two binary waves are used to modulate a pair of quadrature carriers

$$\phi_1(t) = \sqrt{\frac{2}{T}} \cos(2\pi f_c t) \text{ and } \phi_2(t) = \sqrt{\frac{2}{T}} \sin(2\pi f_c t).$$

5. Finally, the two binary PSK waves are added to produce the desired QPSK wave.

ii. QPSK receiver :

1. The QPSK receiver consists of a pair of correlators with a common input and supplied with a locally generated pair of coherent reference signal $\phi_1(t)$ and $\phi_2(t)$.
2. The corresponding output of correlators x_1 and x_2 , produced in response to the received signal $x(t)$, are each compared with a threshold of zero.
3. If $x_1 > 0$, a decision is made in favour of symbol 1 for in-phase channel output, but if $x_1 < 0$, a decision is made in favour of symbol 0.
4. Similarly if $x_2 > 0$, a decision is made in favour of symbol 1 for quadrature channel output, but if $x_2 < 0$ a decision is made in favour of symbol 0.

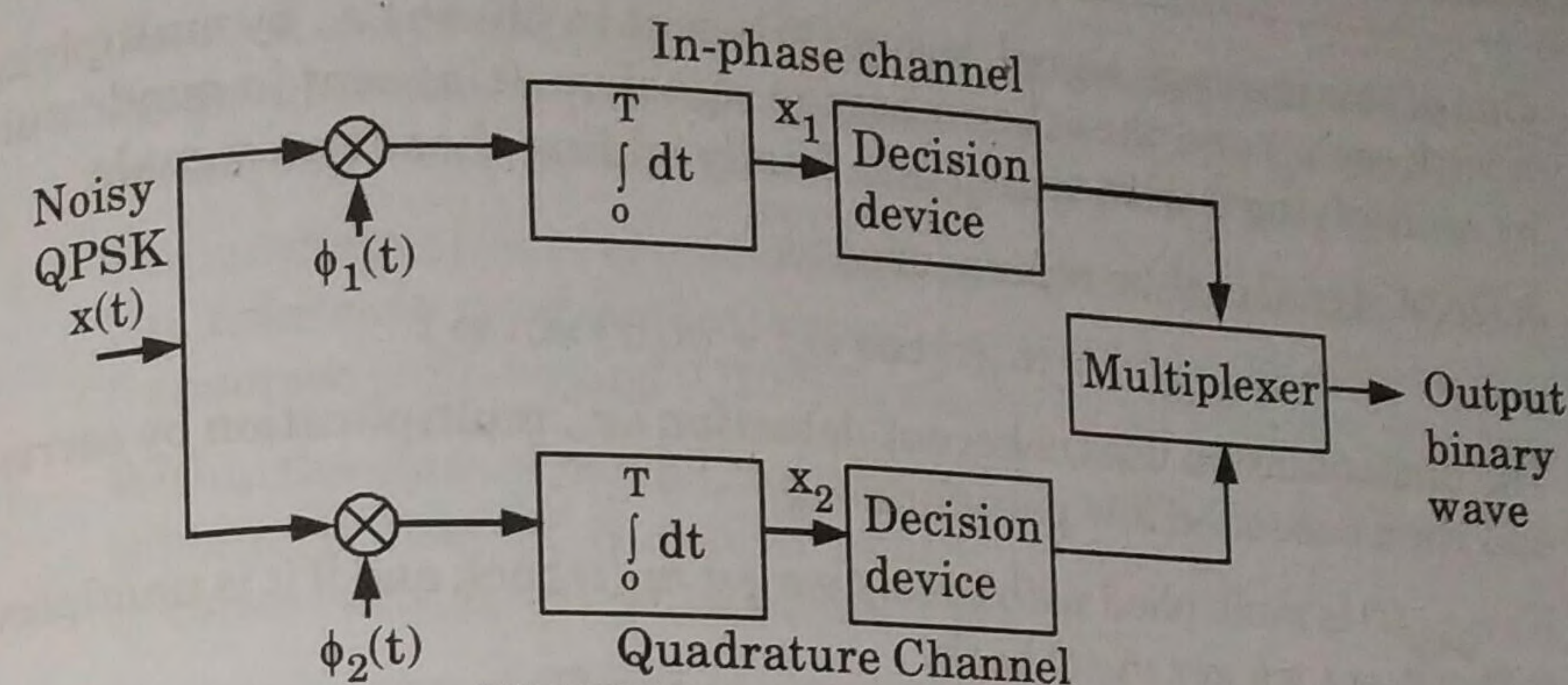


Fig. 5.7.2. QPSK detector.

5. Finally, these two binary sequences at the in-phase and quadrature channel outputs are combined in a multiplexer to reproduce the original binary sequence at the receiver output.
6. The given input binary sequence is 1101010010

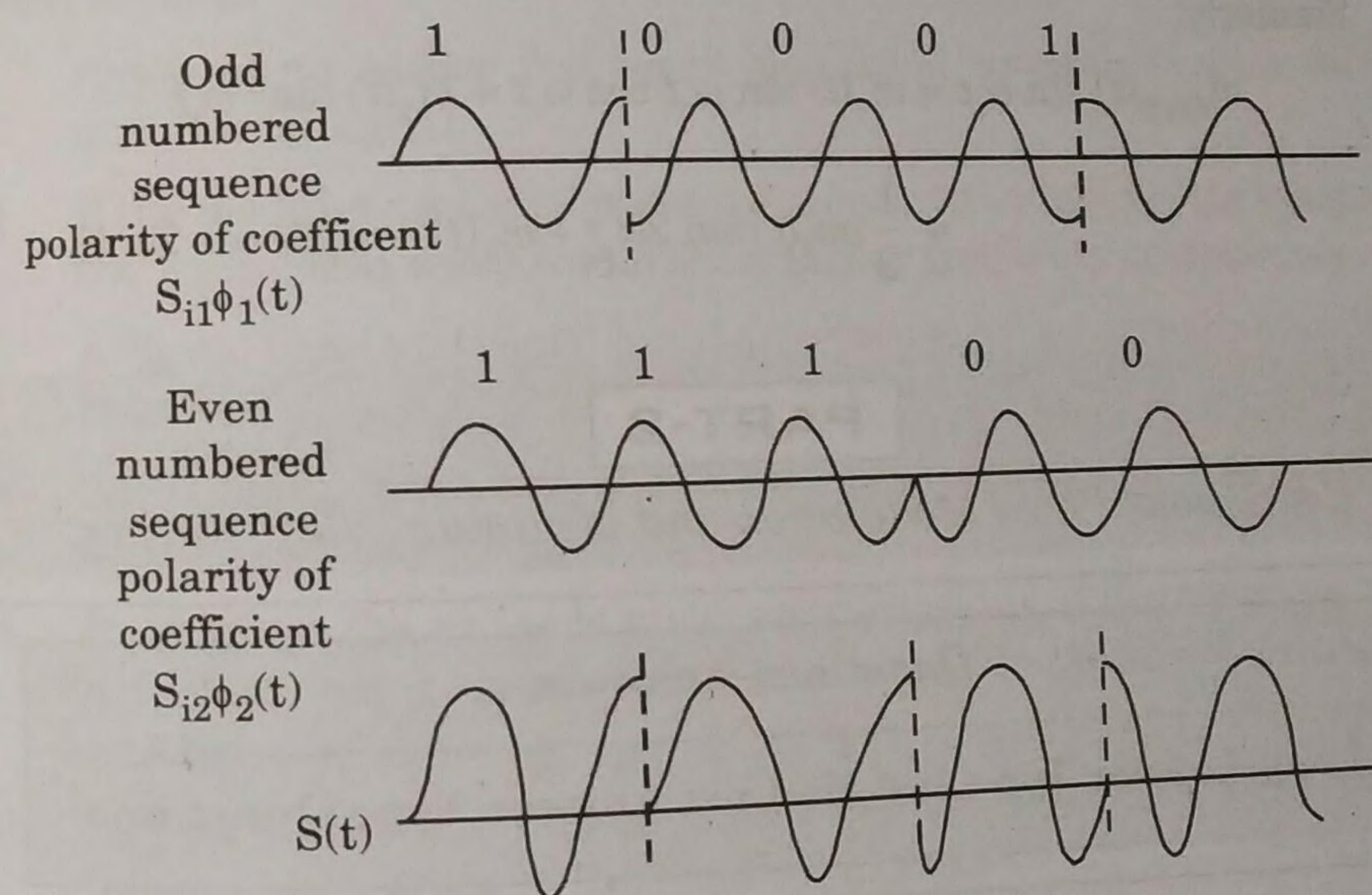


Fig. 5.7.3. QPSK detector.

Que 5.8. Describe quadrature amplitude modulation.

Answer

1. The quadrature amplitude modulation (QAM) is similar to DSB-SC but sends two message signals over the same spectrum.

2. One of this message signal, say $m_1(t)$ is sent in phase *i.e.*, by multiplying it with $\cos \omega_c t$ and the other message signal, $m_2(t)$ is sent in quadrature by multiplying it with $\sin \omega_c t$ and finally adding these two signals.

3. A QAM signal is thus represented as

$$m_{QAM}(t) = m_1(t) \cos \omega_c t + m_2(t) \sin \omega_c t$$

4. The demodulation uses coherent detection *i.e.*, multiplication by carrier and then baseband low pass filtering.

5. If $m_{QAM}(t)$ is multiplied with $\cos \omega_c t$ we get $m_1(t)$ back and if it is multiplied with $\sin \omega_c t$ we get $m_2(t)$ back.

$$m_{QAM}(t) \cos \omega_c t = m_1(t) \cos^2 \omega_c t + m_2(t) \sin \omega_c t \cos \omega_c t$$

$$= \frac{1}{2} [(1 + \cos \omega_c t) m_1(t) + m_2(t) \sin 2\omega_c t]$$

$$= \frac{1}{2} [m_1(t) + m_1(t) \cos \omega_c t + m_2(t) \sin 2\omega_c t]$$

Similarly,

$$m_{QAM}(t) \sin \omega_c t = m_1(t) \sin \omega_c t \cos \omega_c t + m_2(t) \sin^2 \omega_c t$$

$$= \frac{1}{2} [m_1(t) \sin 2\omega_c t + m_2(t) - m_2(t) \cos 2\omega_c t]$$

PART-2

Continuous Phase Modulation and Minimum Shift Keying.

Questions-Answers

Long Answer Type and Medium Answer Type Questions

Que 5.9. Write a short note on continuous phase modulation (CPM).

Answer

1. CPM schemes are classified as being full response or partial response, depending, respectively, on whether the modulating frequency pulse is of single bit duration or longer.
2. Within the class of full response CPMs, the subclass of schemes having modulation index 0.5 but arbitrary frequency pulse shape results in a form of generalized MSK.

3. Included as special cases are MSK having a rectangular frequency pulse shape and Amoroso's sinusoidal frequency-shift-keying (SFSK), possessing a sinusoidal (raised cosine) frequency pulse shape.
4. The subclass of full-response schemes with rectangular frequency pulse but arbitrary modulation index is referred to as continuous phase frequency-shift-keying (CPFSK) which, for all practical purposes.
5. Within the class of partial response CPMs, undoubtedly the most popular scheme is that of Gaussian minimum shift-keying (GMSK) which, because of its excellent bandwidth efficiency.
6. In simple terms, GMSK is a partial-response CPM scheme obtained by filtering the rectangular frequency pulses characteristic of MSK with a filter having a Gaussian impulse response prior to frequency modulation of the carrier.

Que 5.10. Explain CPFSK.

Answer

1. Phase of an analog frequency modulated waveform varies with the modulating signal.
2. If the modulating signal possess discontinuities in time, the spectrum of the modulated wave would show higher frequency components.
3. A digital baseband signal has sharp discontinuities at the data transition boundaries.
4. So, in digital frequency modulation, the problem of high spectral growth becomes severe.
5. Even when the phase is maintained constant, if the signal points are asymmetrical, transmitted power gets wasted in the transmission of impulses.
6. Moreover, the risk of non-linear distortion, after passing through the low noise amplifier in the receiver, increases.
7. Its remedy is a scheme where application of memory makes the signal set symmetric and maintains a continuous phase at the data boundaries. This scheme is CPFSK.
8. Actually CPFSK is a special case of a more general signal set called continuous phase modulation (CPM).

Que 5.11. Explain the MSK modulation scheme and its optimum receiver structure.

Answer

1. MSK signal can be represented as

$$s(t) = s_1 \phi_1(t) + s_2 \phi_2(t) \quad 0 \leq t \leq T_b$$

where the coefficients s_1 and s_2 are related to the phase states $\theta(0)$ and $\theta(T_b)$ respectively.

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos\left(\frac{\pi}{2T_b} t\right) \cos(2\pi f_c t) \quad -T_b \leq t \leq T_b$$

$$\phi_2(t) = \sqrt{\frac{2}{T_b}} \sin\left(\frac{\pi}{2T_b} t\right) \sin(2\pi f_c t) \quad 0 \leq t \leq 2T_b$$

2. To evaluate s_1 we integrate the product $s(t)\phi_1(t)$ between limits $-T_b$ and T_b , we thus obtain

$$s_1 = \int_{-T_b}^{T_b} s(t) \phi_1(t) dt = \sqrt{E_b} \cos[\theta(0)] \quad -T_b \leq t \leq T_b$$

3. Similarly,

$$s_2 = \int_0^{2T_b} s(t) \phi_2(t) dt = -\sqrt{E_b} \sin[\theta(T_b)] \quad 0 \leq t \leq 2T_b$$

i. Generation of MSK :

1. The MSK generator is shown in Fig. 5.11.1.

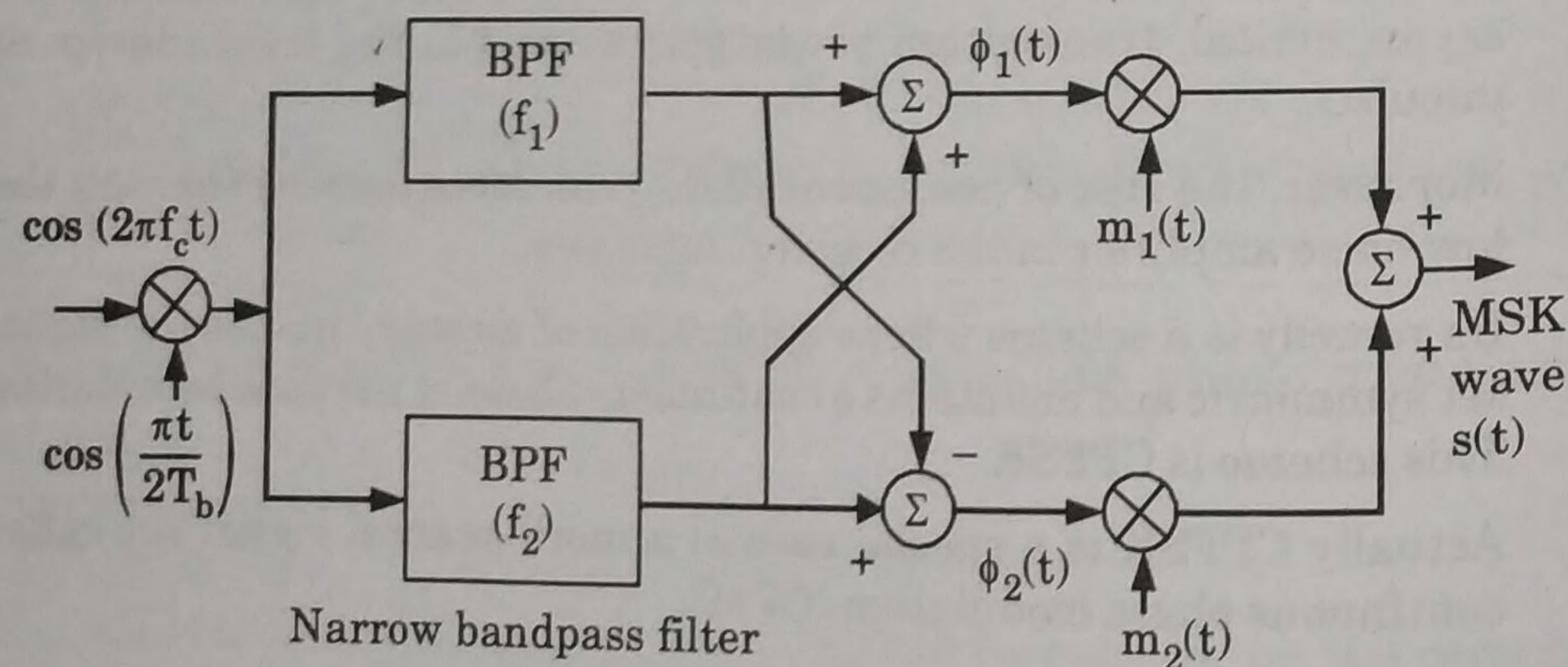


Fig. 5.11.1. MSK transmitter.

2. Two input sinusoidal waves, one of frequency $f_c = n_c / 4T_b$ for some fixed integer n_c , and the other of frequency $1/4T_b$, are first applied to a product modulator.
3. This produces two phase coherent sine waves at frequencies f_1 and f_2 , which are related to f_c and the bit rate $1/T_b$.

$$f_c = \frac{1}{2} (f_1 + f_2) \quad \dots(5.11.1)$$

$$\text{and} \quad h = \frac{1}{2} T_b (f_1 - f_2) \quad \dots(5.11.2)$$

where the parameter h refers to the deviation ratio.

4. These two sinusoidal waves are separated from each other by two narrow band filters, one centered at f_1 and other at f_2 .
5. The resulting filter outputs are next summed to produce the pair of quadrature carriers $\phi_1(t)$ and $\phi_2(t)$.
6. Finally, $\phi_1(t)$ and $\phi_2(t)$ are multiplied with two binary waves $m_1(t)$ and $m_2(t)$, both of which have a bit rate equal to $1/2T_b$.
7. These two binary waves are extracted from the incoming binary wave $b(t)$. At the output of summer we have MSK wave $s(t)$.

ii. Detection of MSK :

1. Fig. 5.11.2 shows the block diagram of MSK receiver.

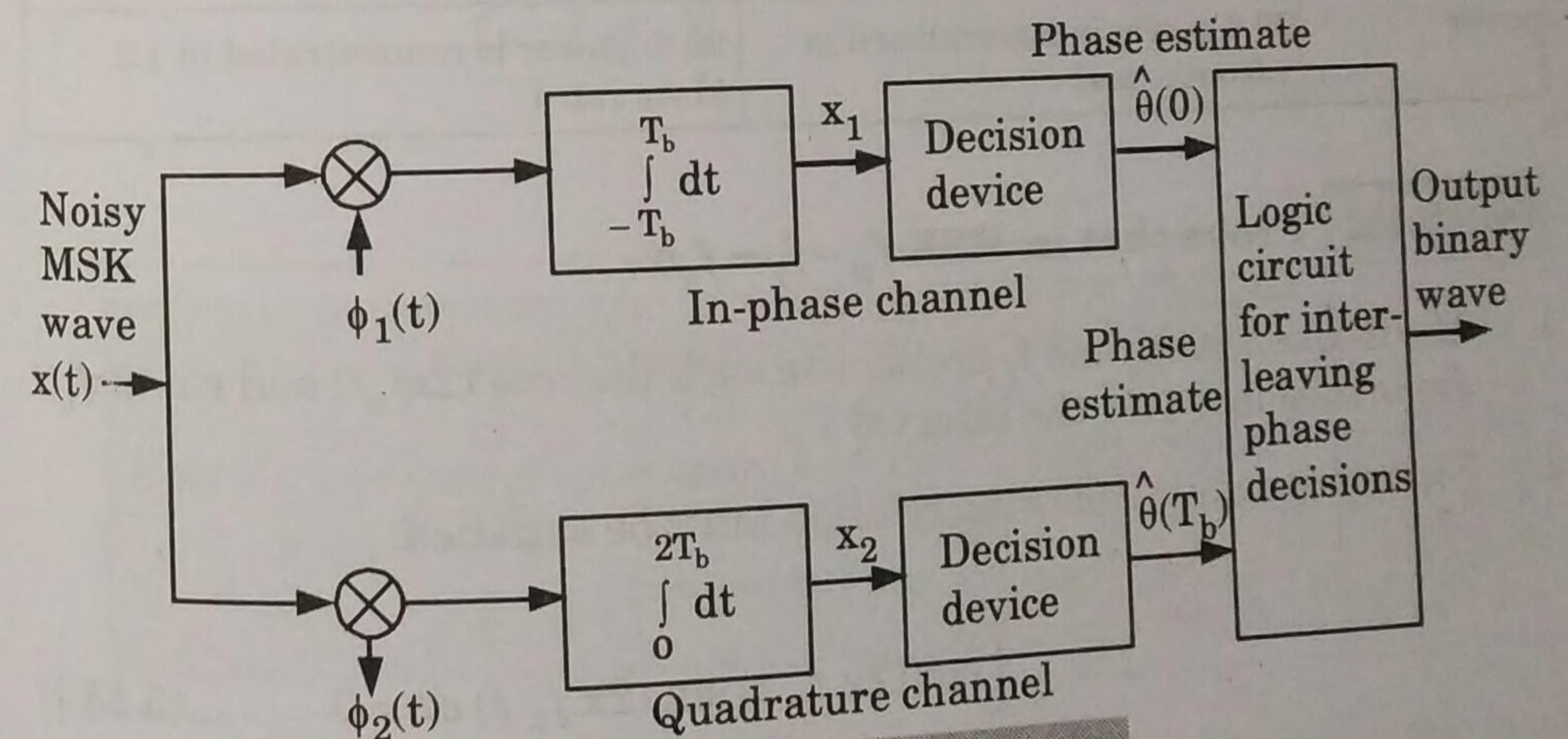


Fig. 5.11.2. MSK detector.

2. The received signal $x(t)$ is correlated with locally generated replicas of the coherent reference signals $\phi_1(t)$ and $\phi_2(t)$.

3. In both cases the integration interval is $2T_b$ and that the integration in the quadrature channel is delayed by T_b with respect to that in the in-phase channel.
4. The resulting two channels outputs x_1 and x_2 are next compared with a threshold of zero volts, and estimates of the phase $\theta(0)$ and $\theta(T_b)$ are derived. Finally these phase decisions are interleaved so as to construct the original input binary wave $b(t)$.

Que 5.12. What is Minimum Shift Keying modulation (MSK) technique? Write down the difference between QPSK and MSK.

AKTU 2017-18, Marks 10

Answer

- A. Minimum Shift Keying Modulation : Refer Q. 5.11, Page 5-14B, Unit-5.
- B. Difference between QPSK and MSK :

Specification	QPSK modulation	MSK modulation
Maximum phase change	$\pm 90^\circ, \pm 180^\circ$	Phase change of $\pm 90^\circ$ smoothly over course of a bit period
RF amplifier requirement	requires linear amplifier	Phase change is linear and hence allows use of nonlinear amplifier
Null bandwidth	equal to 1.0 times data rate	equal to 1.5 times data rate
power	99 % power is concentrated in 1.0(data rate)	99 % power is concentrated in 1.2 (data rate)

Que 5.13. Prove that in MSK $f_H - f_L = f_b/2$.

1. The frequencies f_H and f_L are chosen such that $\cos(2\pi f_H t)$ and $\sin(2\pi f_L t)$ are orthogonal over the interval T_b .
2. For orthogonality following relation must be satisfied,

$$\int_0^{T_b} \cos(2\pi f_H t) \sin(2\pi f_L t) dt = 0 \quad \dots(5.13.1)$$

3. This relation will be satisfied if we have integers m and n such that,

$$2\pi(f_H - f_L)T_b = n\pi \quad \dots(5.13.2)$$

$$2\pi(f_H + f_L)T_b = m\pi \quad \dots(5.13.3)$$

4. Let us substitute values of $f_H = f_c + f_b/4$ and $f_L = f_c - f_b/4$ in eq. (5.13.2), we have

$$2\pi \left[\left(f_c + \frac{f_b}{4} \right) - \left(f_c - \frac{f_b}{4} \right) \right] T_b = n\pi$$

$$2 \times \frac{f_b}{2} T_b = n \Rightarrow n = 1 \quad \left[\text{where } f_b = \frac{1}{T_b} \right]$$

and from eq. (5.13.3), we have

$$2\pi \left(f_c + \frac{f_b}{4} + f_c - \frac{f_b}{4} \right) T_b = m\pi$$

$$f_c = \frac{m}{4} f_b$$

5. Putting $n = 1$ in eq. (5.13.2), we get

$$2\pi(f_H - f_L)T_b = 1 \times \pi$$

$$f_H - f_L = \frac{1 \times \pi}{2\pi T_b} = \frac{f_b}{2}$$

$$f_H - f_L = \frac{f_b}{2}$$

VERY IMPORTANT QUESTIONS

Following questions are very important. These questions may be asked in your SESSIONALS as well as UNIVERSITY EXAMINATION.

- Q. 1. Explain the need of digital modulation. Describe the types of digital modulation. Draw and explain the waveforms for amplitude, frequency and phase shift keying methods.

Ans. Refer Q. 5.1.

- Q. 2. Explain the generation and detection of FSK.

Ans. Refer Q. 5.3.

Q. 3. Give the comparison between ASK, FSK and PSK.

Ans. Refer Q. 5.4.

Q. 4. With the help of block diagram, explain QPSK coherent digital carrier system. Sketch the QPSK waveform for the sequence 1101010010, assuming the carrier frequency to be equal to the bit rate.

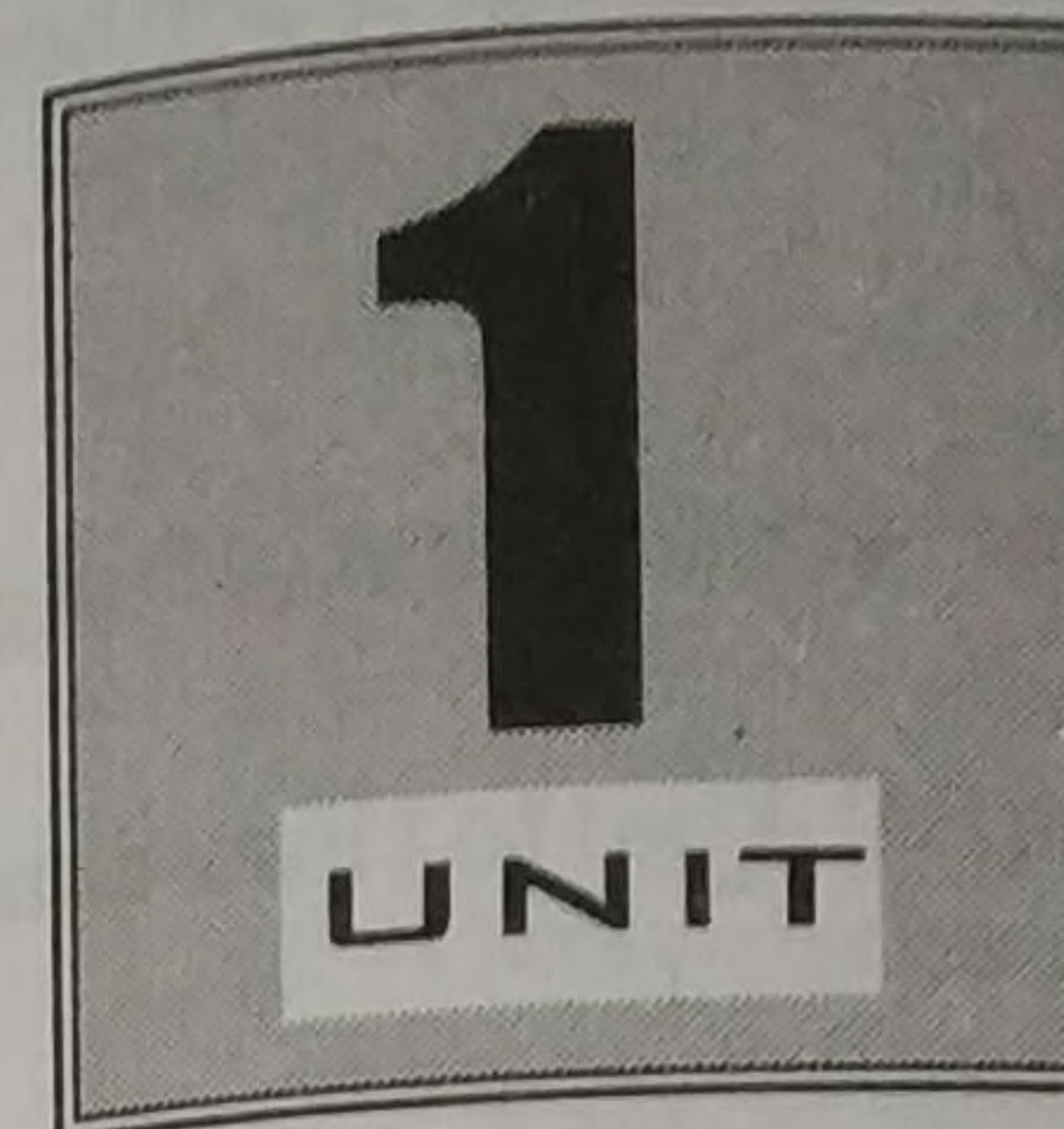
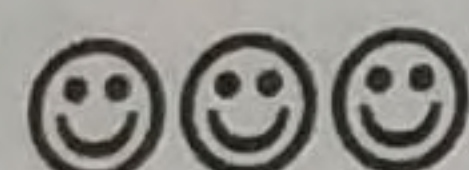
Ans. Refer Q. 5.7.

Q. 5. Explain the MSK modulation scheme and its optimum receiver structure.

Ans. Refer Q. 5.11.

Q. 6. What is Minimum Shift Keying modulation (MSK) technique? Write down the difference between QPSK and MSK.

Ans. Refer Q. 5.12.



Introduction (2 Marks Questions)

1.1. What is signal ?

Ans. It is a physical quantity that varies with time, space or any other independent variable or variables. It gives the information about behaviour or nature of the phenomenon.

1.2. Give the classification of signals.

Ans. Classification of signals :

- Continuous-time signals and discrete-time signals.
- Periodic signals and non-periodic signals.
- Even and odd signals.
- Energy signals and power signals.
- Deterministic signals and random signals.
- One-dimensional signals and multi-dimensional signals.

1.3. Give the definition of energy signal and write its formula.

Ans. **Definition :** A signal is said to be an energy signal if and only if its total energy E is finite (i.e., $0 < E < \infty$). For an energy signal average power $P = 0$.

Expression : The total energy of a continuous-time real-valued signal $x(t)$ may be written as,

$$E = \int_{-\infty}^{\infty} |x(t)|^2 \cdot dt$$

1.4. Define power signal.

Ans. A signal is said to be a power signal if and only if its average power P is finite ($0 < P < \infty$). For a power signal, total energy $E = \infty$. The average power may be written as,

$$P_{av} = \lim_{T \rightarrow \infty} \frac{1}{2T} \int_{-T}^T x^2(t) \cdot dt$$

1.5. Write expression for Fourier transform of signal $x(t)$.

Ans. For a continuous-time signal $x(t)$, Fourier transform is defined as,

$$F[x(t)] = X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

1.6. Write expression for inverse Fourier transform of signal $X(\omega)$.

Ans. It is used to convert frequency-domain signal $X(\omega)$ to time-domain signal $x(t)$. It is given as,

$$x(t) = F^{-1}[X(\omega)] = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

1.7. Give the major difference between Fourier series and Fourier transform.

Ans. Fourier series can only be applied to the periodic continuous-time signals, but Fourier transform can be applied to periodic and non-periodic both functions.

1.8. Prove the time differentiation property of Fourier transform.

Ans. We know, $x(t) = F^{-1}[X(f)] = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df$

Taking the time derivative

$$\begin{aligned} \frac{d}{dt} x(t) &= \frac{d}{dt} \left[\int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df \right] \\ &= \int_{-\infty}^{\infty} \{ j2\pi f X(f) \} e^{j2\pi ft} df \end{aligned}$$

Taking Fourier transform

$$F \left[\frac{d}{dt} x(t) \right] = j2\pi f X(f)$$

The n th time derivation can be calculated as

$$F \left[\frac{d^n}{dt^n} x(t) \right] = (j2\pi f)^n X(f)$$

1.9. State the Parseval's theorem of Fourier transform.

Ans. Energy content of the signal $x(t)$ is given by Parseval's theorem.

$$E = \int_{-\infty}^{\infty} |x(t)|^2 dt = \frac{1}{2\pi} \int_{-\infty}^{\infty} |X(\omega)|^2 d\omega$$

1.10. What is the envelope of AM wave ?

Ans. Envelope of AM wave is the waveform observed as time vs. amplitude, when the carrier and message signal are modulated.

1.11. Define modulation index for AM wave in AM system.

Ans. The modulation index is defined as the ratio of change in amplitude of carrier wave to amplitude of carrier wave. It is denoted by ' μ '.

$$\therefore \mu = \frac{A_m}{A_c} = \frac{A_{\max} - A_{\min}}{A_{\max} + A_{\min}}$$

where, $A_{\max}$ = Maximum amplitude of AM wave
 $A_{\min}$ = Minimum amplitude of AM wave

1.12. Draw phasor diagram for AM system along with its applications.

Ans. Phasor diagram of AM modulated signal :

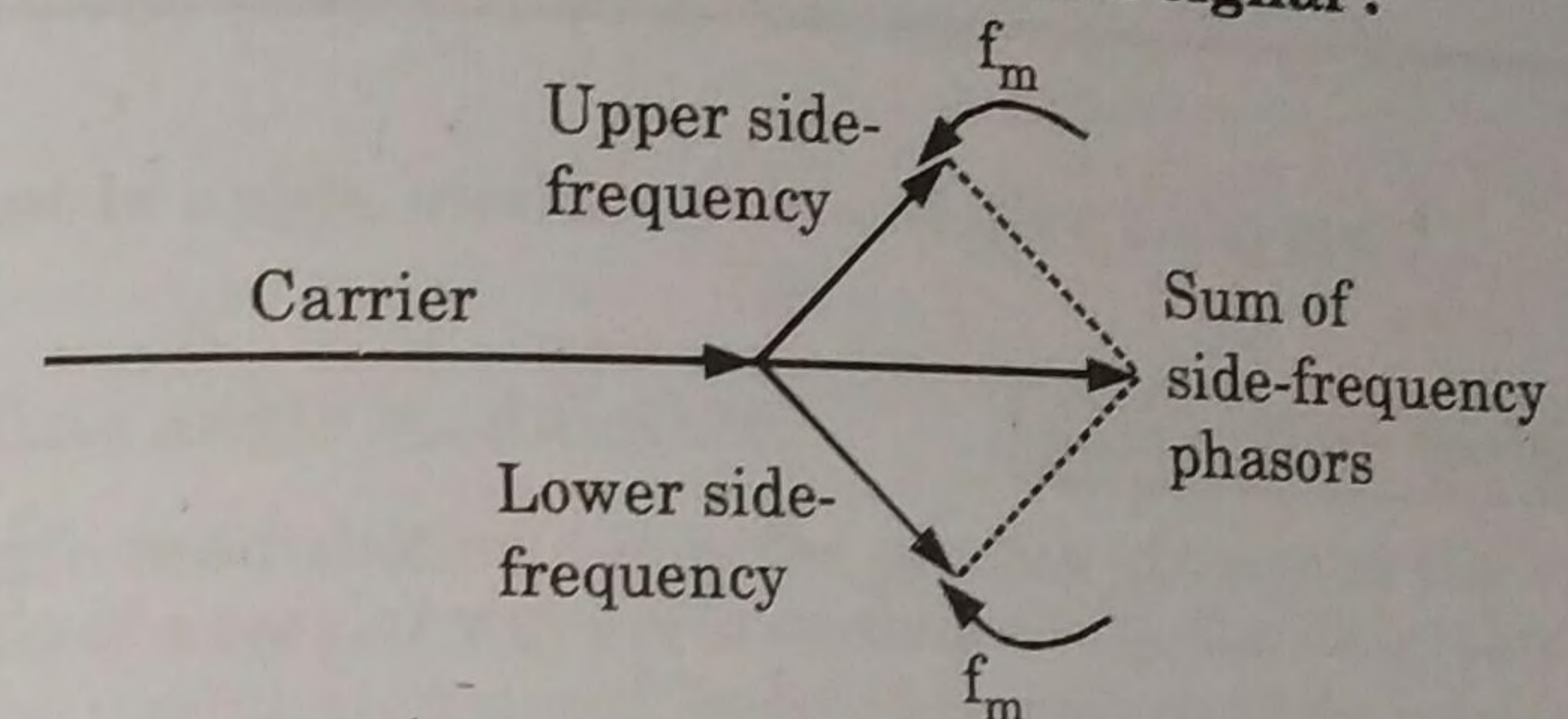


Fig. 1.

Applications :

- 1. Broadcast transmission:** AM is widely used for broadcasting either long or medium or short wave bands.
- 2. Air band radio:** The use of AM in the aerospace industry is widespread. The VHF (Very High Frequency) transmissions made by the airborne equipment use AM.

1.13. Giving the drawback of DSB-SC, explain the need of SSB-SC.

Ans. Drawbacks of DSB-SC : Complex detection and larger bandwidth.
Need of SSB-SC :

- Better management of the frequency spectrum.
- Low power consumption.

1.14. What is VSB signal ? How it is used for reduction of bandwidth ?

Ans. In vestigial side band signal, one of the sideband is partially suppressed and vestige (portion) of the other sideband is transmitted. This vestige (portion) compensates the suppression of the sideband, thereby reduced the bandwidth of the signal.

1.15. Write the limitations of amplitude modulation.

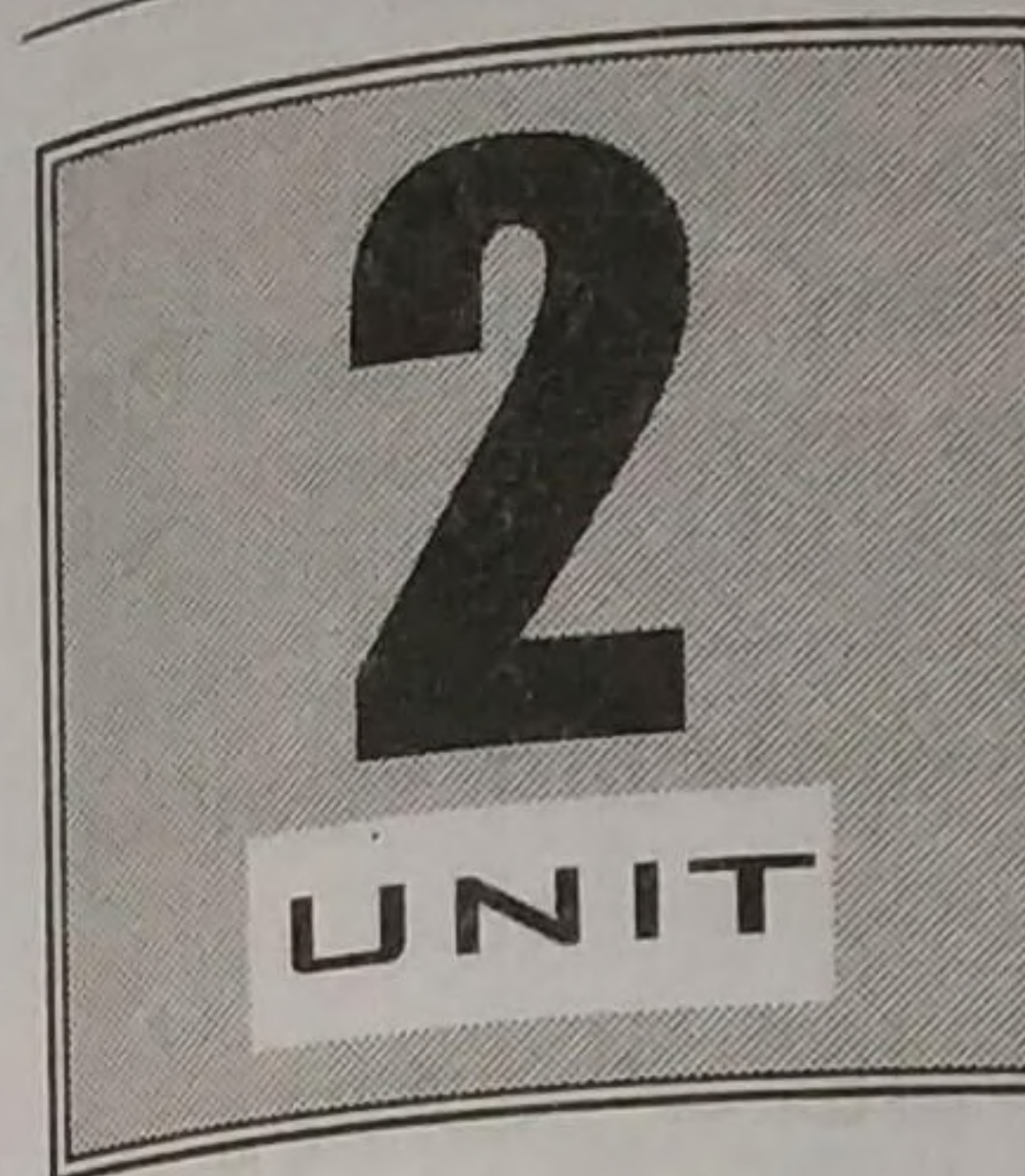
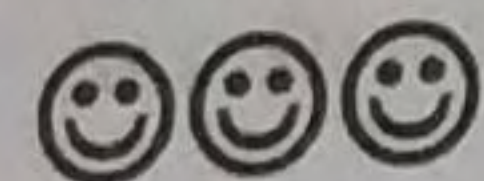
Ans. Limitations of amplitude modulation :

- It is wasteful of power.
- It is wasteful of bandwidth.

1.16. Explain the advantages of SSB-SC over DSB-SC.

Ans. Advantages of SSB-SC over DSB-SC :

- It reduces the bandwidth requirement to half as compared to DSB-SC.
- Because of narrow bandwidth of SSB, the effect of noise at the receiver circuits is reduced.



Angle Modulation (2 Marks Questions)

2.1. What is angle modulation and give its types ?

OR

Define angle modulation.

AKTU 2018-19, Marks 02

Ans. **Angle modulation :** It is the process of varying the total phase angle of a carrier wave in accordance with the instantaneous value of the modulating signal, keeping amplitude of the carrier constant.

Types :

- Phase modulation (PM)
- Frequency modulation (FM)

2.2. What is percent modulation for an FM ?

AKTU 2017-18, Marks 02

Ans. The ratio of actual frequency deviation to the maximum allowable frequency deviation is called percentage of modulation in FM.

$$\% \text{ Modulation} = \frac{\Delta f_{\text{actual}}}{\Delta f_{\text{max}}}$$

2.3. Define phase modulation.

Ans. In this type of angle modulation, the phase angle $\psi(t)$ is varied linearly with a modulating signal $f(t)$ about an unmodulated phase angle $\omega_c(t)$.

$$\psi_i(t) = \omega_c t + K_p f(t).$$

2.4. Explain the term frequency deviation.

Ans.

1. The instantaneous frequency of FM signal varies with time. The maximum change in instantaneous frequency from the average i.e., ω_c , is called frequency deviation.
2. The frequency deviation can be determined with the help of equation

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

The maximum change in ω_i from the average value ω_c will depend on the magnitude and sign of $k_f f(t)$.

3. The frequency deviation is denoted by $\Delta\omega$ and it can be either positive or negative depending on the sign of $k_f f(t)$.

$$\Delta\omega = |k_f f(t)|_{\max}$$

4. The frequency deviation is the useful parameter for determining the bandwidth of the FM signals.

2.5. Explain frequency modulation.

Ans. In this type of angle modulation, the instantaneous frequency ω_i is varied linearly with a modulating signal $f(t)$ about an unmodulated frequency ω_c .

2.6. What are the advantages of frequency modulation ?

Ans. Advantages of frequency modulation :

1. Frequency modulation is less susceptible to noise.
2. Signal to noise ratio doubles on doubling the bandwidth.
3. Bandwidth required is more, so the system performance would increase.

2.7. What is Carson's rule ?

AKTU 2017-18, Marks 02

OR

What is frequency deviation and explain the Carson's rule ?

AKTU 2016-17, Marks 02

Ans. Carson's rule : An empirical formula for the bandwidth of a single-tone wideband FM is given by Carson's rule. According to this rule, the FM bandwidth is given by

$$BW = 2(\Delta\omega + \omega_m) = 2\Delta\omega + 2\omega_m$$

Frequency deviation : Refer Q. 2.4, Page SQ-1B, Unit-2, 2 Marks Questions.

2.8. Differentiate between frequency and phase modulation.

Ans.

S.No.	Phase modulation	Frequency modulation
1.	Phase modulation is the type of angle modulation in which the angular argument $\psi(t)$ is changed in linear proportion with the instantaneous magnitude of the message signal $f(t)$.	Frequency modulation is a type of angle modulation in which the instantaneous frequency ω_i is varied in linear proportion with the instantaneous magnitude of the message signal $f(t)$.
2.	For PM, $\psi(t) = 2\pi f_c t + k_p f(t)$	For FM, $\omega_i = \omega_c + k_f f(t)$

2.9. List comparison between narrowband and wideband FM.

Ans.

S.No.	Narrowband FM	Wideband FM
1.	Modulation index < 1 .	Modulation index > 1 .
2.	Spectrum contains only two sidebands and carrier.	Spectrum contains infinite number of sidebands and carrier.
3.	$BW = 2f_m$	$BW = 2(\Delta f + f_m)$

2.10. Mention three direct methods of FM generation.

Ans. Direct methods of FM generation,

- i. Using varactor diode.
- ii. Reactance tube modulator.
- iii. Miller capacitance frequency modulator.

2.11. Write properties of Bessel's function $J_n(\beta)$.

Ans.

- i. $J_n(m_f) = J_{-n}(m_f)$; n even
- ii. $J_n(m_f) = -J_{-n}(m_f)$; n odd

$$\text{iii. } \sum_{n=-\infty}^{\infty} J_n^2(m_f) = 1$$

2.12. An angle modulated signal.

$$x_c(t) = 10 \cos [2\pi 10^6 t + 0.1 \sin (10^3) \pi t]$$

Considering $x_c(t)$ as a PM signal with $K_p = 10$. Find $m(t)$.

Ans.

$$\begin{aligned} x_{PM}(t) &= A \cos [\omega_c t + K_p m(t)] \\ &= 10 \cos [2\pi 10^6 t + 10m(t)] \\ &= 10 \cos [2\pi 10^6 t + 0.1 \sin (10^3) \pi t] \end{aligned}$$

Thus, $m(t) = 0.01 \sin (10^3) \pi t$.

2.13. A modulating signal $5 \cos (2\pi \times 15 \times 10^3 t)$ angle modulates a carrier $A \cos \omega_c t$. Find modulation index and bandwidth for the PM system. Assume $k_p = 15 \times 10^3$.

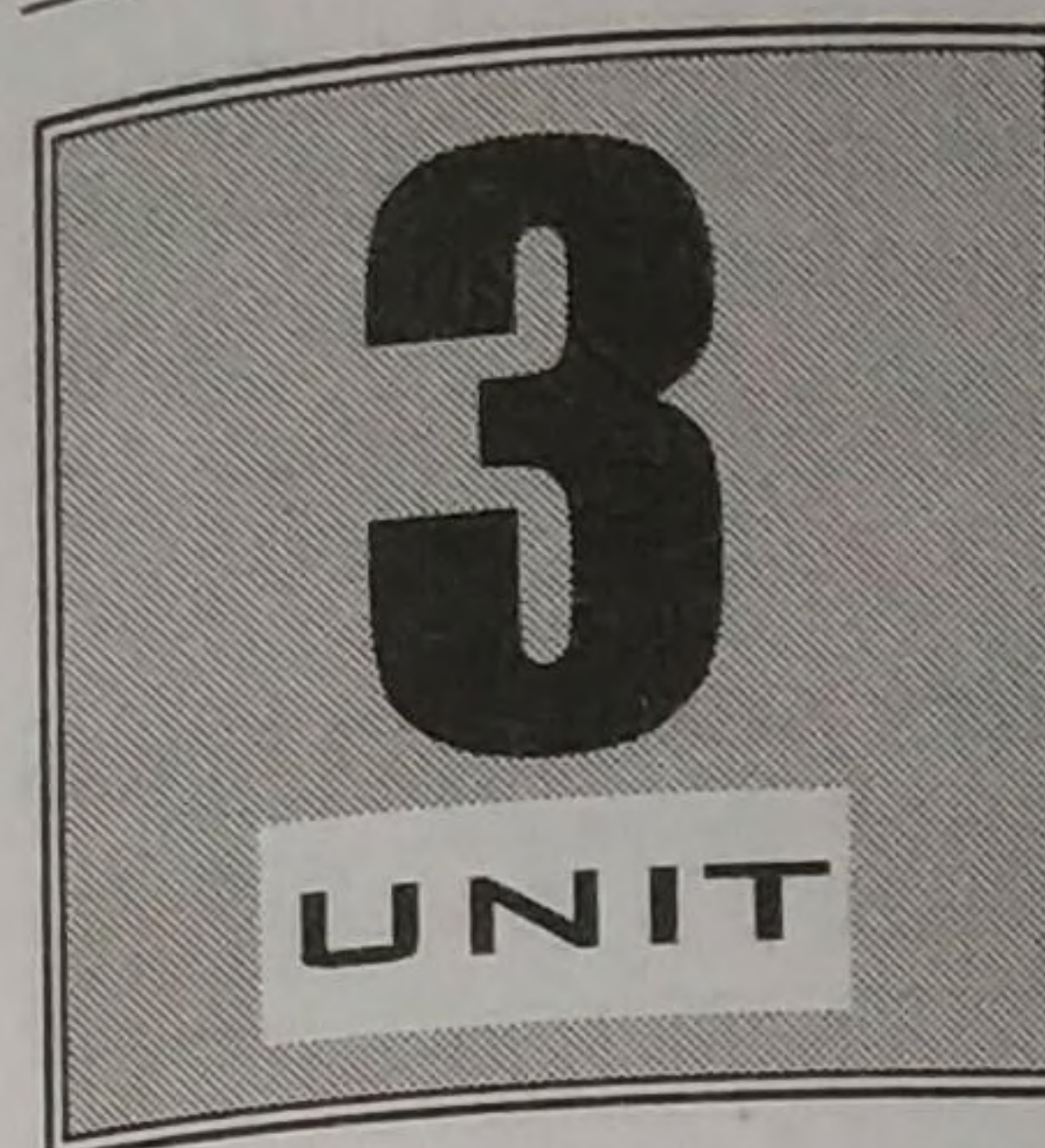
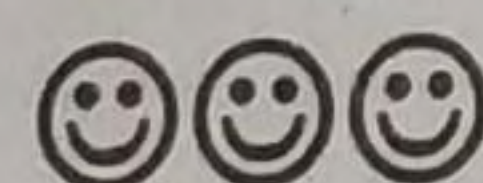
Ans. Frequency deviation,

$$\Delta f = k_p E_m f_m = 15 \times 10^3 \times 5 \times 15 \times 10^3 = 1125 \text{ MHz}$$

$$\text{BW} = 2(\Delta f + f_m) \approx 2\Delta f = 2250 \text{ MHz}$$

and

$$m_p = k_p E_m = 5 \times 15 \times 10^3 = 75000$$



Noise (2 Marks Questions)

3.1. What do you understand by probability of an event ?

Ans. The probability of an event has the same means as the relative frequency of that event. Hence, we estimate the probability of each event as the relative frequency of that event and is given by

$$P(A) = \lim_{N \rightarrow \infty} \frac{N(A)}{N}$$

3.2. What is conditional probability ?

Ans. It often happens that the probability of one event is influenced by the outcome of another event. The conditional probability $P(B/A)$ denotes the probability of event B when it is known that event A has occurred and is given by

$$P(B/A) = \frac{P(A \cap B)}{P(A)} \text{ provided } P(A) > 0$$

$$\text{Also, } P(A/B) = \frac{P(A \cap B)}{P(B)} \text{ provided } P(B) > 0$$

3.3. Explain the Bay's rule of probability.

AKTU 2016-17, Marks 02

Ans. In probability theory, Bay's theorem (alternatively Bay's law or Bay's rule) describes the probability of an event, based on prior knowledge of conditions that might be related to the event.

3.4. Explain the term random variable.

Ans. A random variable is a real valued function defined over the sample space of random experiment. It is also known as stochastic variable, a random function or stochastic function. When the outcome of the experiment is S , the random variable is denoted as $X(S)$ or x .

3.5. Differentiate between wide sense stationary and strict sense stationary random process. AKTU 2017-18, Marks 02

Ans.

S.No.	Wide sense stationary (WSS)	Strict sense stationary (SSS)
1.	A random process $\{X(t)\}$ is said to be wide sense stationary (WSS) if its mean and autocorrelation functions are time invariant.	A random process $\{X(t)\}$ is said to be SSS (or just stationary) if all its finite order distributions are time invariant.
2.	WSS does not imply SSS.	SSS implies WSS.

3.6. Define noise.

Ans. Unwanted signals that tend to disturb the transmission and processing of signals in communication systems is called as noise.

3.7. Calculate the autocorrelation for white noise.

AKTU 2016-17, Marks 02

Ans. For white noise, $PSD = \frac{\eta}{2}$

Hence, autocorrelation = Inverse Fourier transform of PSD

$$R(\tau) = \frac{\eta}{2} \delta(\tau)$$

3.8. What is thermal agitation noise ?

Ans. The noise generated in a resistance or the resistive component is random and is referred as thermal agitation, white or Johnson noise. It is due to the rapid and random motion of the molecules inside the component itself.

3.9 Discuss signal to noise ratio.

Ans. It is defined as the ratio of signal power to noise power at the same point. Therefore,

$$\frac{S}{N} = \frac{X_s}{X_n} = \frac{V_s^2 / R}{V_n^2 / R} = \left(\frac{V_s}{V_n} \right)^2$$

S = Signal power

N = Noise power

3.10. What is white noise ? Draw its spectral density curve.

Ans. White noise : White noise is a stationary random process with a uniform power spectral density over the entire frequency range. The term white is used in analogy with white light which is a superposition of all visible spectral components.

Spectral density curve : The power spectral density curve of white noise is shown in Fig. 2.

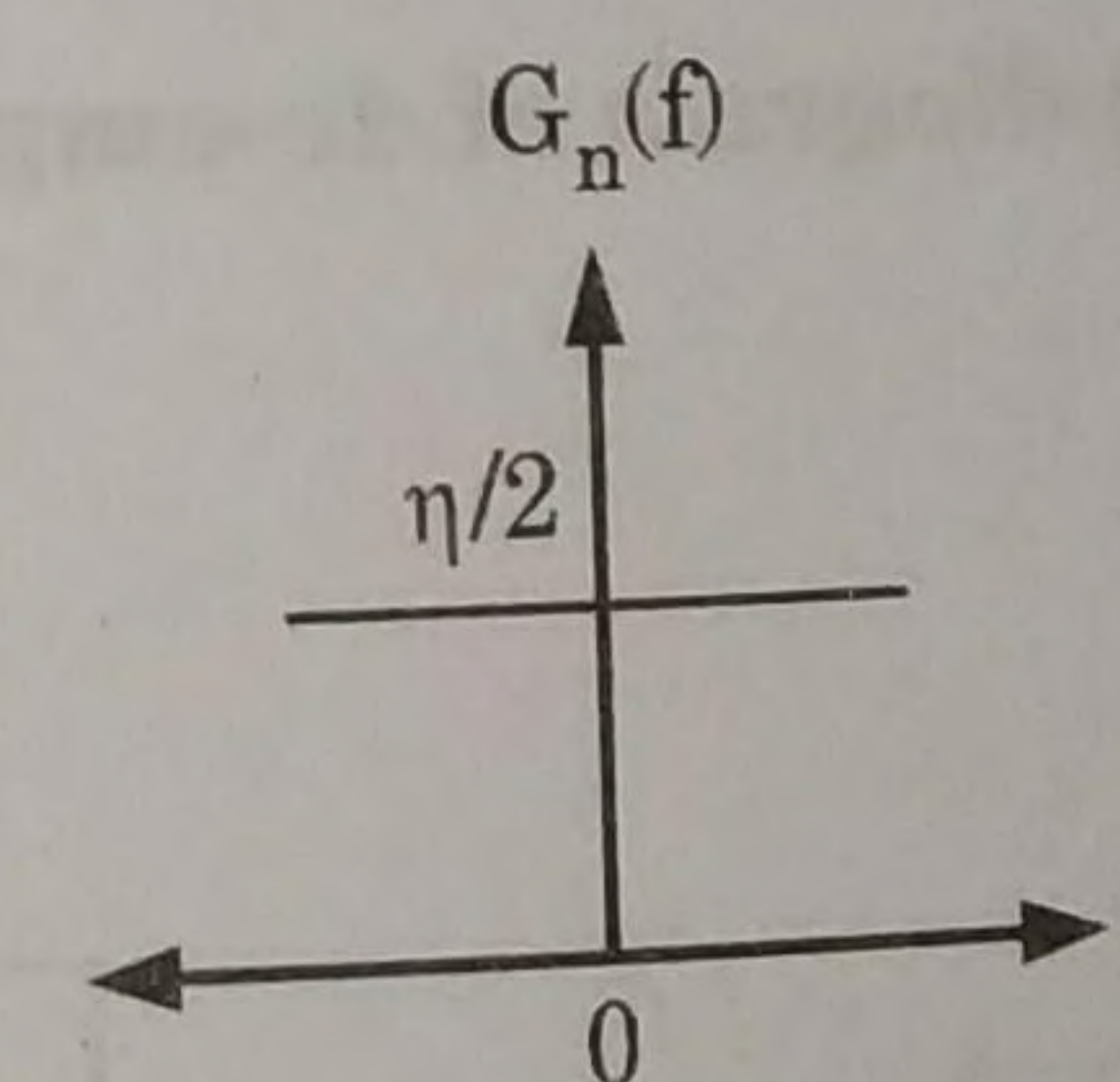


Fig. 1.

3.11. What is noise figure ? List sources of noise.

Ans. Noise figure : Noise figure is defined as the ratio of input SNR to the output SNR.

$$\text{i.e., Noise figure} = \frac{(SNR)_i}{(SNR)_o}$$

Source of noise :

The external noise is created outside the circuit, and includes :

- i. Erratic natural disturbances
- ii. Man-made noise

Internal noises are :

- i. Shot noise
- ii. Thermal noise.

3.12. Find the modulation index β that produces $(S/N)_0 = 30$ dB at the threshold.

Ans. $\left(\frac{S}{N}\right)_0 = 30\beta^2(\beta + 1) = 10^3 (= 30 \text{ dB})$

$$\beta^2(\beta + 1) = \frac{100}{3}$$

$$\beta^3 + \beta^2 = \frac{100}{3}$$

$$\beta = 2.92.$$

3.13. Draw the circuit diagram of de-emphasis and pre-emphasis network.

Ans.

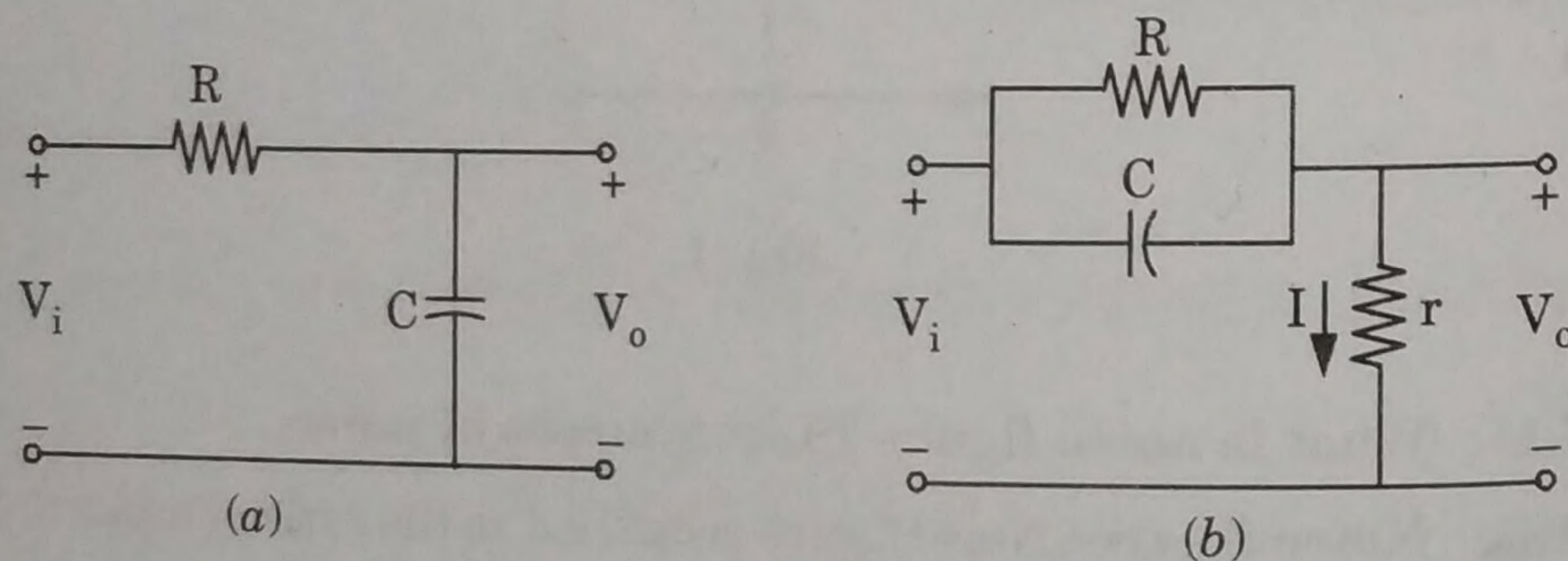


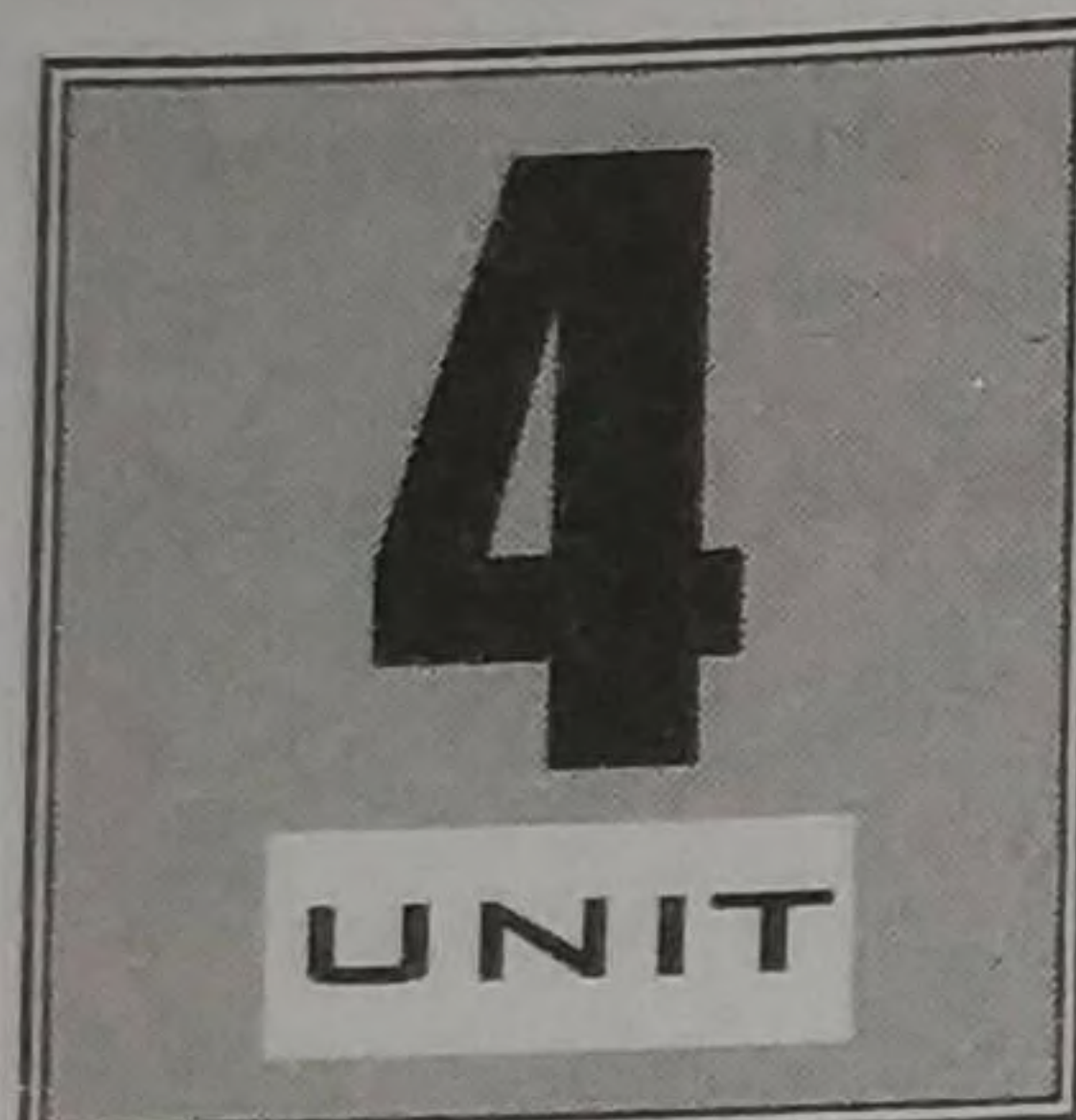
Fig. 2. (a) De-emphasis network (b) Pre-emphasis network.

3.14. What are the methods used for the improvement of threshold ?

Ans. There are two methods used for the improvement of threshold :

- i. Pre-emphasis and De-emphasis.
- ii. FMFB (frequency modulation with feedback).





Pulse Modulation (2 Marks Questions)

4.1. What is pulse modulation ?

Ans. In pulse modulation, the carrier consists of a pulse train, some parameters of which are varied in accordance with the instantaneous value of the modulating signal.

4.2. Name the two methods of getting the pulse amplitude modulated waveform.

Ans.

1. Natural sampling theorem
2. Flat top sampling.

4.3. Define Nyquist rate and Nyquist interval.

OR

What do you mean by Nyquist rate ?

AKTU 2017-18, Marks 02

Ans. The sampling rate of $2W$ samples per second, for a signal bandwidth of W hertz, is called the Nyquist rate and its reciprocal $1/2W$ (measured in seconds) is called the Nyquist interval.

4.4. Compare digital modulation and pulse modulation.

AKTU 2018-19, Marks 02

Ans.

S.No.	Digital modulation	Pulse modulation
1.	In this modulation, one or more parameters of a carrier wave are varied as a function of two or more finite and discrete states of a signal.	In this modulation, pulses are varied in some respect, such as width or amplitude, to represent the amplitude of a signal.

4.5. What is sampling theorem ?

AKTU 2017-18, Marks 02

Ans. The sampling theorem for strictly bandlimited signals of finite energy in two equivalent parts, which apply to the transmitter and receiver of a pulse modulation system, respectively :

- i. A bandlimited signal of finite energy, which has no frequency component higher than W hertz, is completely described by specifying the values of the signal at instants of time separated by $1/2W$ seconds.
- ii. A bandlimited signal of finite energy which has no frequency components higher than W hertz may be completely recovered from knowledge of its samples taken at the rate of $2W$ samples per second.

4.6. What are the applications of sampling theorem ?

Ans. The sampling theorem is very important in signal analysis, processing and transmission because it allows us to replace a continuous time signal by a discrete sequence of numbers.

4.7. What are the two major advantages of pulse modulation ?

Ans.

- i. The transmitter power can be concentrated into short burst rather than being delivered continuously. This is advantageous in case of light power microwave tubes and lasers.
- ii. The time-intervals between pulses can be filled with sample values from other messages, thereby permitting the transmission of many messages on communication channel.

4.8. Discuss delta modulation.

Ans. In delta modulation, an incoming message signal is oversampled (i.e., at a rate much higher than Nyquist rate) to purposely increase the correlation between adjacent samples of the signal. This modulation provides a staircase approximation to the oversampled version of the message signal.

4.9. What do you mean by TDM ?

Ans.

1. In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one.
2. Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.

4.10. Name the two types of multiplexing techniques.

Ans. Two types of multiplexing techniques are :

- i. Frequency division multiplexing.
- ii. Time division multiplexing.

4.11. Explain the advantages of TDM.

Ans.

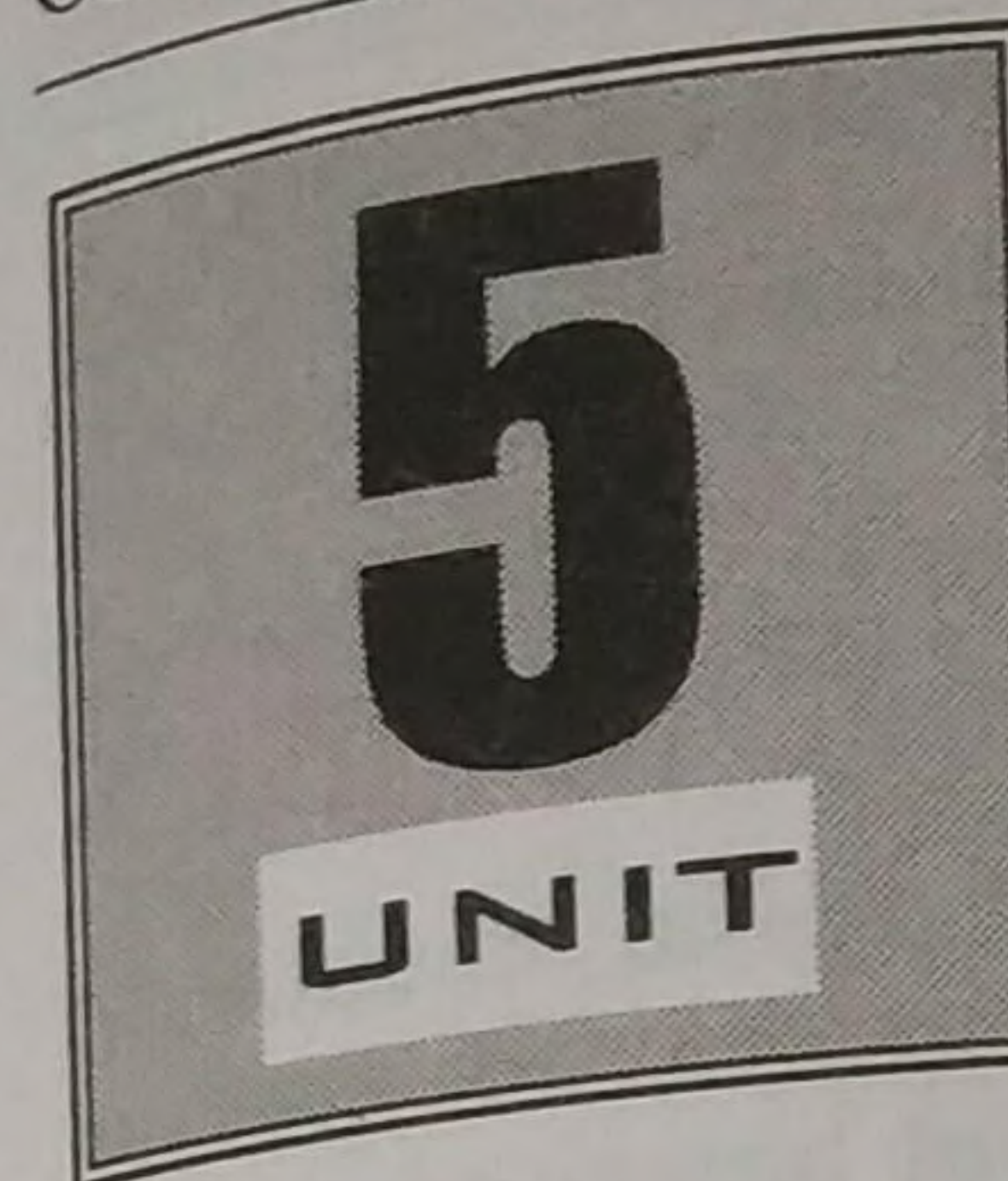
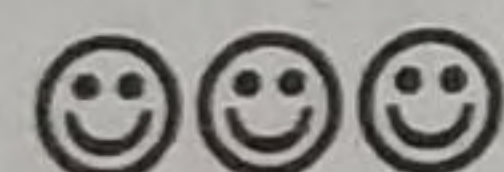
1. Full available channel bandwidth can be utilized for each channel.
2. TDM circuitry is not very complex.
3. Problem of cross-talk is not severe.
4. Only one carrier in the medium at any time.

4.12. Explain the difference between TDM and FDM.

AKTU 2016-17, Marks 02

Ans.

S.No	TDM	FDM
1.	It divides the channel by allocating a time period for each channel.	It divides the channel into multiple, but smaller frequency ranges to accommodate more users.
2.	Much better flexibility compared to FDM.	Less flexibility.
3.	Less latency.	It provides much better latency compared to TDM.



Digital Modulation Scheme

(2 Marks Questions)

5.1. Write the types of digital modulation schemes.

Ans.

- i. Amplitude shift keying (ASK)
- ii. Frequency shift keying
- iii. Phase shift keying.

5.2. Define amplitude shift keying.

Ans.

When the carrier amplitude is varied in proportion to the message signal, the modulated signal is still an on-off signal. This modulation scheme of transmitting binary data is known as amplitude shift keying (ASK).

5.3. Define frequency shift keying.

Ans.

When data are transmitted by varying the frequency, it is called frequency shift keying (FSK).

5.4. Explain DPSK.

Ans.

1. Differential phase shift keying (DPSK) can be viewed as the non-coherent version of the PSK. It eliminates the need for a coherent reference signal at the receiver by combining two basic operations at the transmitter :
 - i. Differential encoding of the input binary wave
 - ii. Phase shift keying
2. Hence, it is named as differential phase-shift keying.

5.5. What do you mean by phase shift keying ?

Ans. In coherent binary PSK system, the pair of signals $S_1(t)$ and $S_2(t)$ used to represent binary symbols '1' and '0' respectively are defined by :

$$S_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t); \text{ for symbol '1'}$$

$$S_2(t) = -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t); \text{ for symbol '0'}$$

where E_b is the transmitted energy per bit.

5.6. What is ASK ?

Ans. Amplitude shift keying (ASK) is the simplest digital modulation technique. In this method there is only one energy carrier and it is switched ON or OFF depending upon the input binary sequence. The ASK waveform may be represented as

$$S(t) = \begin{cases} \sqrt{2P_s} \cos(2\pi f_c t) & ; \text{ for symbol '1'} \\ 0 & ; \text{ for symbol '0'} \end{cases}$$

5.7. Compare the spectrum occupancy of PSK and FSK schemes.

Ans.

S. No.	PSK	FSK
1.	It requires larger bandwidth.	It requires comparatively lower bandwidth.
2.	Spectrum occupancy is higher.	Spectral occupancy is lower.

5.8. Draw the block diagram of MSK transmitter.

Ans. The block diagram of MSK transmitter is,

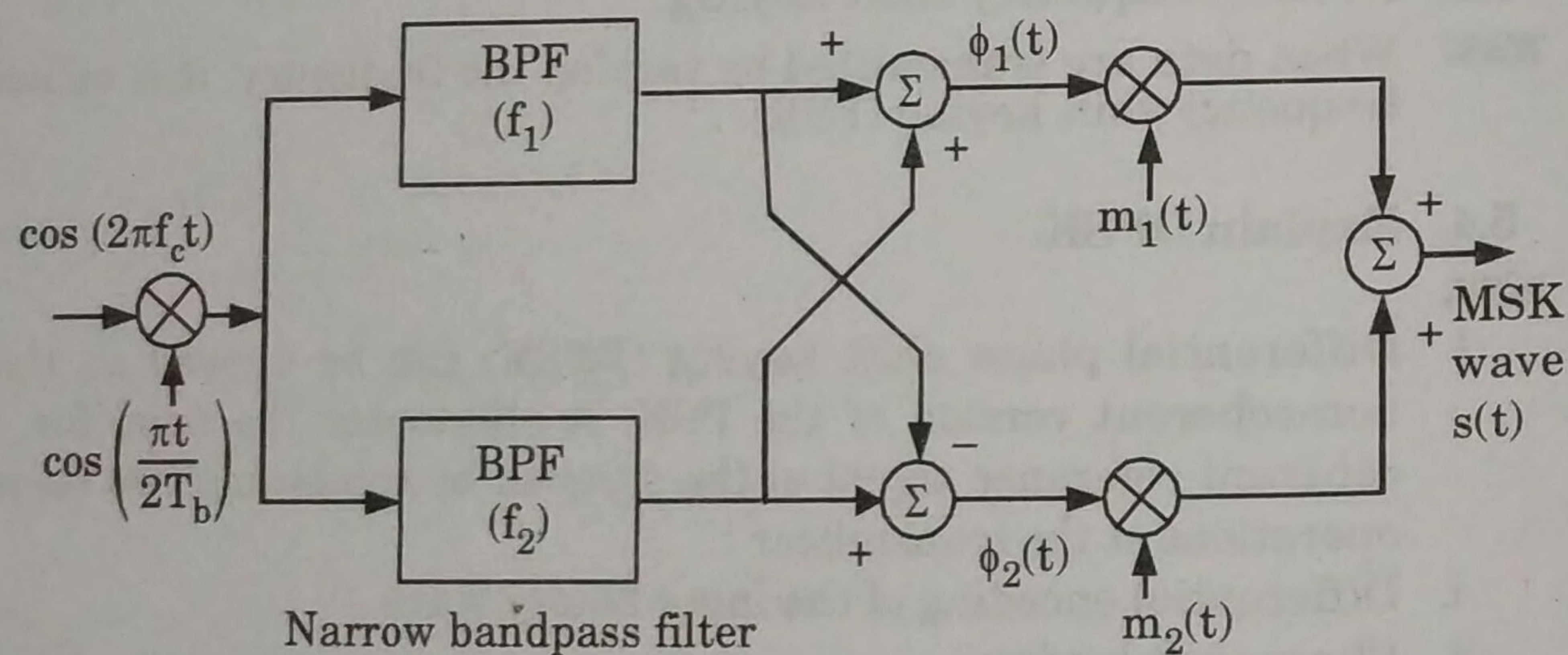


Fig. 1. MSK transmitter.

5.9. Draw the various waveforms for amplitude, frequency and phase shift keying.

AKTU 2017-18, Marks 02

Ans. Waveforms of ASK, PSK and FSK :

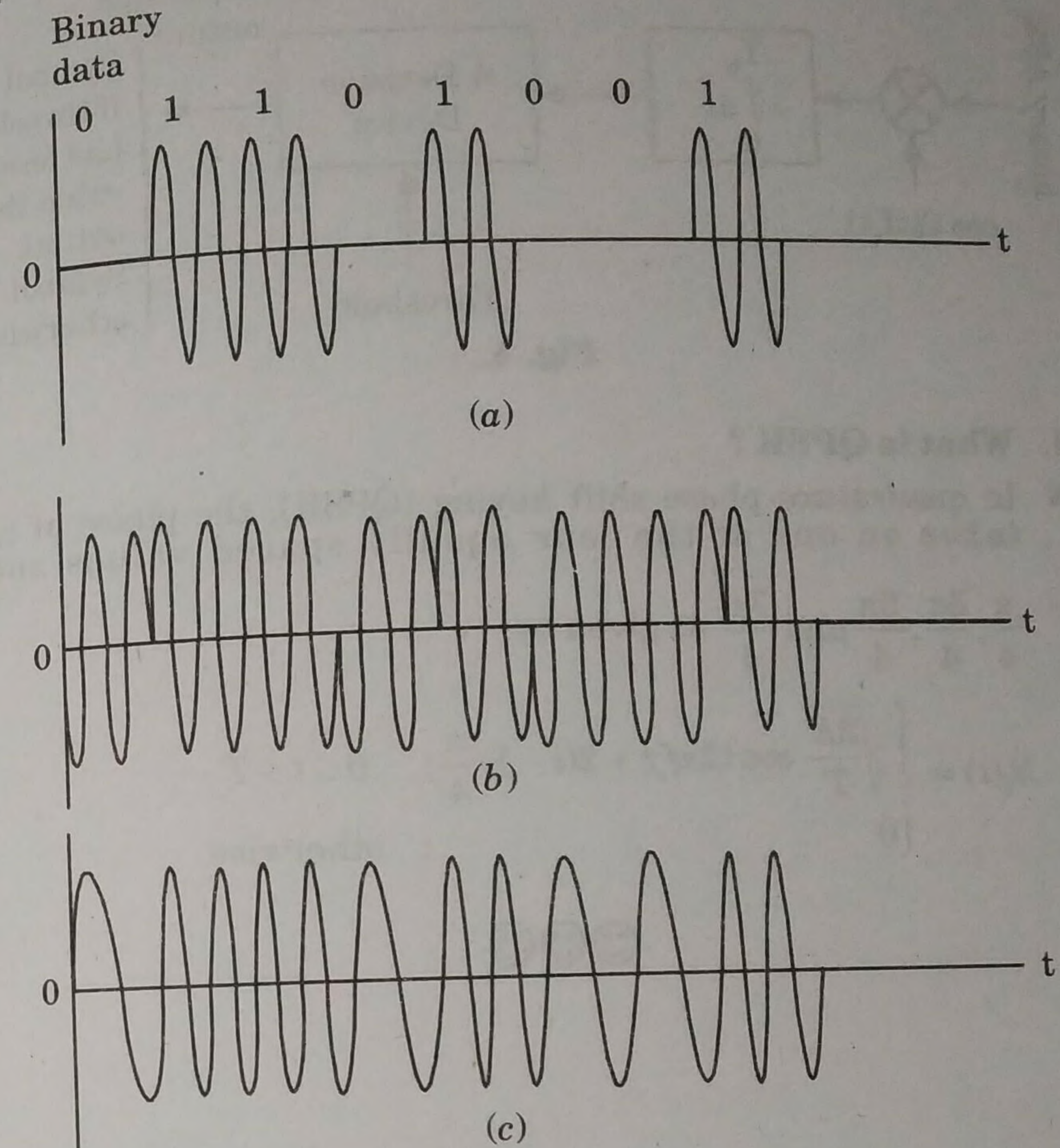


Fig. 2. Waveforms for (a) Amplitude shift keying, (b) Phase shift keying, (c) Frequency shift keying.

5.10. Give the block diagram of generation and coherent detection of ASK signal.

Ans. Generation of ASK signal is shown in Fig. 3.

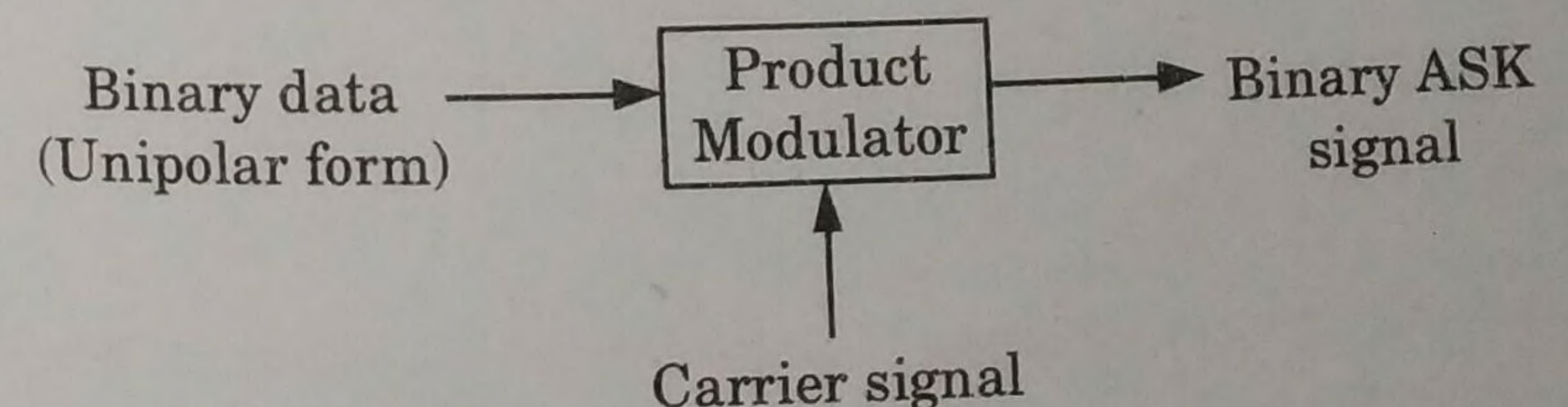


Fig. 3.

Coherent detection of ASK signal is shown in Fig. 4.

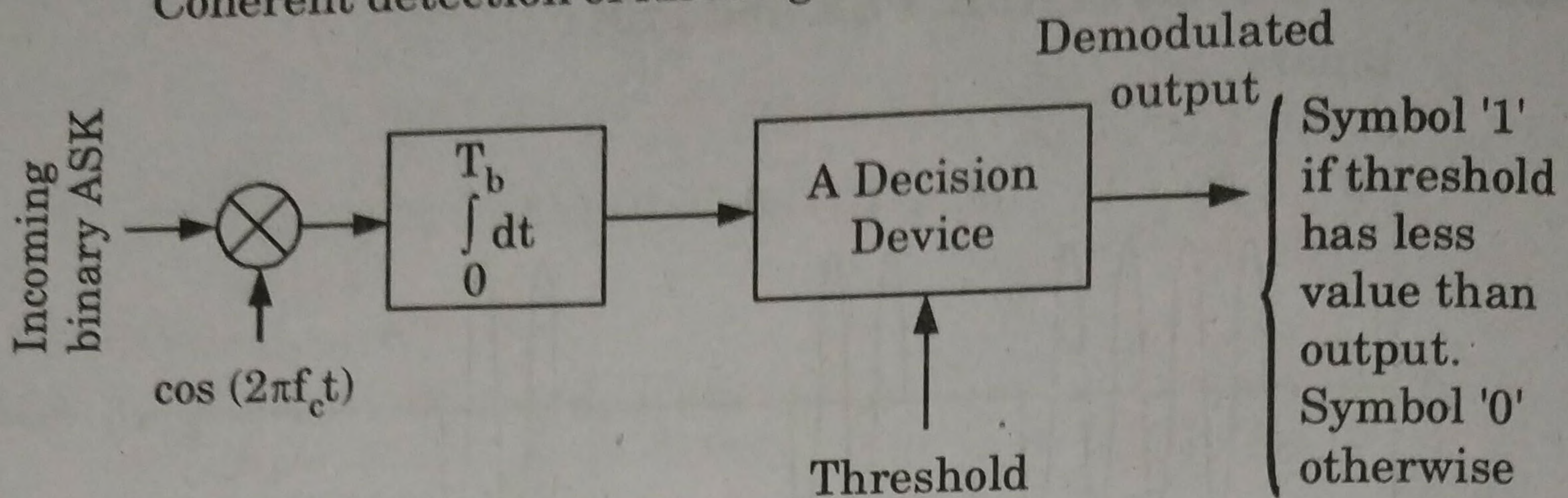


Fig. 4.

5.11. What is QPSK ?

Ans. In quadrature phase shift keying (QPSK), the phase of carrier takes on one of the four equally spaced values such as

$\frac{\pi}{4}, \frac{3\pi}{4}, \frac{5\pi}{4}$ and $\frac{7\pi}{4}$ as given by,

$$S_i(t) = \begin{cases} \sqrt{\frac{2E}{T}} \cos(2\pi f_c t + 2(i-1)\frac{\pi}{4}); & 0 \leq t \leq T \\ 0 & ; \text{ otherwise} \end{cases}$$

