DECODE®

A Guide For Engineering Students

PRINCIPLES OF COMMUNICATION SYSTEMS

(For END SEM Exam - 70 Marks)

SUBJECT CODE : 204193

S.E. (E&Tc / Elex.) Semester - IV

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SYLLABUS

Principles of Communication Systems - (204193)

Credit	Examination Scheme :
03	End - Sem (Theory) : 70 Marks

Unit III FM Transmission & Reception for Single Tone
Phase Modulation (PM) and Frequency Modulation (FM), Relationship between Phase and Frequency Modulation, Modulation Index, Spectrum of FM (single tone) : Feature of Bessel Coefficient, Power of FM signal, Bandwidth of tone modulated FM signal, modulation index : AM vs. FM, Spectrum of constant Bandwidth FM, Narrowband and Wideband FM.
FM Modulators and Demodulators : FM generation by Armstrong's Indirect method, frequency multiplication and application to FM, FM demodulator. **(Chapter - 3)**

Unit IV Pulse Modulation
Need of analog to digital conversion, sampling theorem for low pass signal in time domain, and Nyquist criteria, Types of sampling - natural and flat top. Pulse amplitude modulation & concept of TDM : Channel bandwidth for PAM, equalization, Signal Recovery through holding. Pulse Width Modulation (PWM) and Pulse Position Modulation (PPM) : Generation & Detection. **(Chapter - 4)**

Unit V Digital Representation of Analog Signals
Quantization of Signals : Quantization error, Uniform & Non-Uniform types of Quantization, Mid-rise & Mid-tread Quantizer.
Companding : A-law & μ-law.
Pulse Code Modulation system : Generation & Reconstruction, Differential Pulse code modulation, Delta Modulation, Adaptive Delta Modulation. **(Chapter - 5)**

Unit VI Baseband Digital Transmission
Line codes : Properties and spectrum.
Digital Multiplexing and hierarchies : T1, AT&T, E1, CCITT, Scrambling & Unscrambling.
Synchronization : Carrier Synchronization, Bit Synchronization and Frame Synchronization. Intersymbol Interference, Equalization. **(Chapter - 6)**

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IMPORTANT FORMULAE

Chapter - 3 : FM Transmission and Reception for Single Tone

1. Modulation Index (FM) : $m_f = \frac{\Delta f}{f_m}$
2. Deviation ratio = $\frac{\text{Maximum Frequency deviation}}{f_m \text{ (max)}}$
3. Deviation constant = $\frac{\Delta f}{V_m}$
4. Bandwidth (f_m) : $2 [\Delta f + f_m]$

Chapter - 4 : Pulse Modulation

1. **Sampling of signals :** Nyquist rate = 2 W

   Ideally sampled signal : $X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$

   Naturally sampled signal : $S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc } e(nf_s\tau) X(f - nf_s)$

   Flat top sampled signal : $S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) H(f)$

2. Sampling rate, $f_s \geq 2W$
3. Nyquist rate = $2W$ and Nyquist interval = $\frac{1}{2W}$
4. **Pulse modulation :**

   BW of PAM, $B_T \geq f_{max}$ or $B_T \geq \frac{1}{2t_\tau}$

   BW of PPM and PWM, $B_T \geq \frac{1}{2t_r}$

Chapter - 5 : Digital Representation of Analog Signals

1. PCM signaling rate, $r = \nu f_s$

   PCM transmission BW, $B_T = \frac{1}{2} r$, or $\frac{1}{2} \nu f_s$ or νW

   PCM signal to noise ratio, $\frac{S}{N} = 4.8 + 6\nu$ dB

   For sinusoidal signal, $\frac{S}{N} = 1.8 + 6\nu$ dB

   Non uniform quantization, $\frac{S}{N} = \frac{3q^2}{[ln(1+\mu)]^2}$ or $\frac{3q^2 A^2}{[1 + ln\, A]^2}$

2. **Nonuniform quantization :**

   A-law, $z(x) = \begin{cases} \frac{A|x|}{1 + lnA} & for\ 0 \leq |x| \leq \frac{1}{A} \\ \frac{1 + ln(A|x|)}{1 + ln\, A} & for\ \frac{1}{A} \leq |x| < 1 \end{cases}$

   μ − law $z(x) = \text{sgn}(x) \frac{ln(1 + \mu|x|)}{ln(1+\mu)}$, $|x| < 1$

   $\left(\frac{S}{N}\right) = \frac{3q^2}{[ln(1+\mu)]^2}$

3. DM, slope overload distortion will not occur if $A_m \leq \frac{\delta}{2\pi f_m T_s}$

   DM, signal to noise ratio, $\frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$

   ADM and DM number of bits/sample = 1

   ADM and DM signaling rate = f_s

Chapter - 6 : Baseband Digital Transmission

1. Bit rate in T_1 Frame, $R_b = 1.544$ Mbps

END...

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Unit III

3 FM Transmission and Reception for Single Tone

3.1 : Introduction to Phase and Frequency Modulation

Q.1 Define FM and PM. [SPPU : Dec.-01, Marks 7]

Ans. :

- **Frequency Modulation :** When frequency of the carrier varies as per amplitude variations of modulating signal, then it is called Frequency Modulation (FM). Amplitude of the modulated carrier remains constant.
- **Phase Modulation :** When phase of the carrier varies as per amplitude variations of modulating signal, then it is called Phase Modulation (PM). Amplitude of the modulated carrier remains constant.

Q.2 Define instantaneous phase and instantaneous frequency with the help of equation and explain exponential modulation. [SPPU : May-10, Marks 8]

Ans. :

- The basic difference between FM and PM lies in which property of the carrier is directly varied by modulating signal.
- Note that when frequency of the carrier varies, phase of the carrier also varies and viceversa. But if frequency is varied directly, then it is called FM. And if phase is varied directly, then it is called PM.
- The instantaneous phase deviation is denoted by $\theta(t)$. It is the instantaneous change in phase of the carrier with respect to reference phase.
- The instantaneous phase of the carrier is precise phase of the carrier at a given instant. It is mathematically expressed as,

$$\text{instantaneous phase} = \omega_c t + \theta(t) \text{ rad} \quad \text{... (Q.2.1)}$$

Here $\theta(t)$ is instantaneous phase deviation and

ω_c is carrier frequency.

Now let us define instantaneous frequency deviation. i.e.,

$$\text{Instantaneous frequency deviation} = \frac{d}{dt}\theta(t) = \theta'(t) \text{ Hz} \quad \text{... (Q.2.2)}$$

- Here note that instantaneous frequency deviation is the instantaneous change in carrier frequency. It is equal to the rate at which instantaneous phase deviation takes place.
- Similarly instantaneous frequency is defined as the frequency of the carrier at a given instant of time. It is given as,

$$\text{Instantaneous frequency} = \omega_i(t) = \frac{d}{dt}[\omega_c t + \theta(t)] \quad \text{... (Q.2.3)}$$

$$= \omega_c + \theta'(t) \text{ rad/sec} \quad \text{... (Q.2.4)}$$

3.2 : Frequency Modulation

Q.3 Derive an expression for frequency modulated wave.

[SPPU : Dec.-92, 99, 01, 07, 13, May-13, Dec.-13] Dec.-17, Marks 6, Dec.-16, Marks 3]

Ans. : The instantaneous frequency f of the frequency modulated wave is given by,

$$f_i = f_c(1 + kV_m \cos\omega_m t) \quad \text{... (Q.3.1)}$$

where f_c = Unmodulated carrier frequency

k = Proportionality constant

$V_m \cos\omega_m t$ = Instantaneous modulating voltage

Cosine term has maximum value, ± 1, therefore, the maximum frequency deviation occurs when $\cos\omega_m t = \pm 1$. Thus, under these conditions, the instantaneous frequency is given by,

$$f_i = f_c(1 \pm kV_m) = f_c \pm f_c kV_m \quad \text{... (Q.3.2)}$$

$K = K_f$ = Frequency deviation constant

The above equation gives maximum frequency deviation

$$\Delta f = f_c k_f V_m \quad \text{... (Q.3.3)}$$

$$v_{Fm} = V_c \sin\theta$$

The instantaneous amplitude of the FM signal is given by,

$$v_{Fm} = V_c \sin \theta \quad \text{... (Q.3.4)}$$

We denote ω_i as instantaneous angular velocity. It is given by,

$$\omega_i = 2\pi f_i = 2\pi f_c (1 + k_f V_m \cos \omega_m t) \text{ (refer equation Q.3.1)}$$
$$= \omega_c (1 + k_f V_m \cos \omega_m t)$$

In order to find θ, ω_i must be integrated with respect to time. Thus

$$\theta = \int \omega_i \, dt = \int \omega_c (1 + k_f V_m \cos \omega_m t)$$
$$= \omega_c \int (1 + k_f V_m \cos \omega_m t) \, dt$$
$$= \omega_c \left(t + \frac{k_f V_m \sin \omega_m t}{\omega_m} \right) = \omega_c t + \frac{k_f V_m \omega_c \sin \omega_m t}{\omega_m}$$
$$= \omega_c t + \frac{2\pi f_c k_f V_m \sin \omega_m t}{2\pi f_m} \quad \because \int \cos nx \, dx = \frac{\sin nx}{n}$$
$$= \omega_c t + \frac{\Delta f \sin \omega_m t}{f_m} \quad \because \Delta f = f_c k_f V_m \text{ ... (Q.3.5)}$$

- Substituting the value of θ from equation (Q.3.5) in equation (Q.3.4) we have,

$$v_{Fm} = V_c \sin \left(\omega_c t + \frac{\Delta f}{f_m} \sin \omega_m t \right) \quad \text{... (Q.3.6)}$$

The above equation gives the instantaneous value of the FM voltage.

Q.4 State clearly the difference between modulation index and deviation ratio. ☞ **[SPPU : May-06, 10, June-22, Marks 4]**

Ans. : Modulation Index

Modulation in FM is generally expressed in terms of the modulation index. The modulation index is the ratio of the frequency deviation to the modulating frequency.

$$m_f = \text{Modulation index} = \frac{\text{Frequency deviation}}{\text{Modulating frequency}}$$

$$\therefore \quad m_f = \frac{\Delta f}{f_m}$$

Thus the FM signal in general can be expressed as

$$v_{Fm} = V_c \sin(\omega_c t + m_f \sin \omega_m t) \quad \text{... (refer equation Q.3.6)}$$

Deviation Ratio (DR)

- The deviation ratio is the ratio of maximum frequency deviation to maximum modulating signal frequency. i.e.,

$$\text{Deviation Ratio (DR)} = \frac{\text{Maximum frequency deviation}}{f_{m(max)}} \quad \text{... (Q.4.1)}$$

- Thus the deviation ratio is basically the modulation index corresponding to maximum modulating frequency.

Q.5 Justify FM is called constant BW system. State Carson's rule ☞ **[SPPU : Dec.-05, 08, May-99, 08, Dec.-11]**

Ans. :

- To find the bandwidth occupied by FM wave, the useful rule is that frequency modulated wave contains sideband components of importance on either side of the carrier wave over a frequency interval approximating the sum of the frequency deviation and the modulating frequency.
- The total bandwidth in which most of the energy of the wave is contained is then twice this value.

$$\text{Thus, } BW_{FM} = 2[m_f + 1] f_m = 2\left[\frac{\text{Frequency deviation}}{\text{Modulating frequency}} + 1\right] f_m$$
$$= 2\left[\frac{\Delta f}{f_m} + 1\right] f_m$$
$$\therefore \quad BW_{FM} = 2[\Delta f + f_m]$$

- To illustrate the use of the above equation, let us consider the following examples :

1) $\Delta f = 75 \text{ kHz} \quad f_m = 50 \text{ Hz}$

$$BW_{FM} = 2\left[75 + \frac{50}{1000}\right] \text{kHz} = 150.1 \text{ kHz}$$

2) $\Delta f = 75 \text{ kHz} \quad f_m = 500 \text{ Hz}$

$$BW_{FM} = 2\left[75 + \frac{500}{1000}\right] \text{kHz} = 151 \text{ kHz}$$

3) $\Delta f = 75 \text{ kHz} \quad f_m = 5 \text{ kHz}$

$$BW_{FM} = 2[75 + 5] = 160 \text{ kHz}$$

- Thus, although the modulation frequency changes from 50 Hz to kHz, or by a factor of 1:100, the bandwidth occupied by the spectrum alters very little, from 150.1 kHz to 160 kHz. These examples explain why frequency modulation is sometimes referred to as a **constant-bandwidth system.**
- We have,

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

$$\therefore \quad BW = 2(\Delta f + f_{m\,max}) \qquad \text{...Carson's rule}$$

- In commercial FM broadcasting,

Maximum frequency deviation permissible = 75 kHz

and Maximum modulating frequency = 15 kHz,

then

$$\text{Max } BW_{FM} = 2\ [75 + 15] = 180 \text{ kHz.}$$

- The above equation for calculating maximum bandwidth required to transmit the FM wave is known as **Carson's rule.**

Q.6 "FM is superior to AM". Justify.

[SPPU : Dec.-10, May-12, Marks 4, May-17, Marks 6]

Ans. :

Sr. No.	FM	AM
1.	The equation for FM wave is $v = E_c \sin[\omega_c t + m_f \sin \omega_m t]$.	The equation for AM wave is $v = E_c[1 + m \sin \omega_m t] \sin \omega_c t$.
2.	The modulation index can have value either less than one or more than one.	The value of modulation index is always between zero and one.
3.	Since in FM, amplitude of the carrier is constant, the transmitted power is constant, independent of the modulation index.	Transmitted power is dependent upon modulation index $P_T = P_c\left[1 + \frac{m^2}{2}\right]$.
4.	The modulation index determines the number of significant pairs of sidebands in an FM signal.	In an AM signal, only two sidebands are produced, for any value of modulation index.

5.	The amplitudes of the carrier and side bands vary with the modulation index and can be calculated with Bessel functions.	The amplitudes of the sidebands is dependent on the modulation index, and is always less than the amplitude of carrier.
6.	The carrier or sideband amplitudes are zero at some modulation indices.	The sideband amplitude is never zero for any value of modulation index greater than zero.
7.	The bandwidth of an FM signal is proportional to the modulation index.	The bandwidth of an AM signal is twice the highest modulating frequency.
8.	For FM, the % of modulation is the ratio of the actual frequency deviation and the maximum permissible frequency deviation multiplied by 100.	For AM, % modulation is the ratio of amplitude of modulating voltage to the amplitude of the carrier multiplied by 100.
9.	The main advantage of FM over AM is its noise immunity, as limiter stage in FM receiver clips off noise signals.	The AM system is more susceptible to noise and more affected by noise than FM.
10.	The capture effect in FM allows the strongest signal on a frequency to dominate without interference from the other signal.	When two AM signals occupy the same frequency, both signals will generally be heard regardless of their relative signal strength.
11.	In FM, greater transmitter efficiency can be realized using class-C amplifiers, as amplitude of FM signal is constant.	The efficiency of AM is less than that of FM due to use of class-B amplifier.
12.	The bandwidth of FM signal is much under than the bandwidth of AM. The bandwidth of a typical FM channel is 200 kHz.	The bandwidth required to transmit AM signal is much less than that of FM typically 10 kHz in AM broadcasting.
13.	The circuits to produce and demodulate FM are usually more complex and expensive than AM circuits.	The demodulation of AM signal is very easy practically by use of a diode, which is very simple in operation and cheap cost-wise.

Q.7 In an F.M. system, the frequency deviation constant is 5 kHz/V. If a sinusoidal modulating signal of 15 V, 10 kHz is applied, calculate the peak frequency deviation and the modulating index.

[SPPU : May-94, 11, Dec.-10, Dec.-15, Marks 6]

Ans. : Given : $f_m = 10$ kHz

Frequency deviation constant (K) = 5 kHz/V

AF voltage = 15 V

$\therefore$ Frequency deviation = 15×5 kHz = **75 kHz**

$$\text{Modulation index} = \frac{\Delta f}{f_m} = \frac{75 \text{ kHz}}{10 \text{ kHz}} = 7.5$$

Q.8 An angle modulated signal has the form : $y(t) = 10 \cos(2p f_c t + 4 \sin 2\pi f_m t)$ where $f_c = 10$ MHz and $f_m = 1000$ Hz i) Assuming that this is FM signal determine the modulation index and the transmitted signal bandwidth. ii) Repeat (i) if f_m is doubled.

[SPPU : May-09, 13, Marks 10, Dec.-16, Marks 6]

Ans. : i) Given : $m_f = 4$, $f_m = 1000$ Hz

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

$$\therefore \quad \Delta f = m_f \times f_m = 4 \times 1000 = 4 \text{ kHz}$$

$$BW_{FM} = 2(\Delta f + f_m) = 2(4 \text{ kHz} + 1 \text{ kHz}) = 10 \text{ kHz}$$

ii) Now, $f_m = 2000$ Hz

$$\therefore \quad \Delta f = 4 \times 2000 = 8 \text{ kHz}$$

$$BW_{FM} = 2(8 \text{ kHz} + 2 \text{ kHz}) = 20 \text{ kHz}$$

Q.9 An angle-modulated signal with carrier frequency $\omega_c = 2\pi \times 10^6$ is described by the equation $\psi_{EM}(t) = 10 \cos(\omega_c t + 0.1 \sin 2000 \pi t)$.
i) Find the power of the modulated signal
ii) Find the frequency deviation Δf iii) Find the phase deviation $\Delta\phi$
iv) Estimate the bandwidth. **[SPPU : May-05, 11, Marks 8]**

Ans. : The signal frequency $f_m = \frac{2000\pi}{2\pi} = 1000$ Hz.

i) The carrier amplitude is 10, and therefore the power modulating signal is,

$$P = \frac{(E_c/\sqrt{2})^2}{R} \; 10^2/2 = \frac{50}{R}$$

ii) To find the frequency deviation Δf, we have to find the instantaneous frequency ω_i, given by,

$$\omega_i = \frac{d}{dt}\theta(t) = \omega_c + 200\pi \cos 2000\pi t$$

The carrier deviation is $200\pi \cos 2000\pi t$.

$$\therefore \quad \Delta\omega = 200\pi$$

we have $\Delta f = \frac{\Delta\omega}{2\pi} = \frac{200\pi}{2\pi} = 100$ **Hz**

iii) The angle $\theta(t) = \omega t + (0.1 \sin 2000 \pi t)$. The phase deviation is the maximum value of the angle inside the parenthesis, and is given by,

$$\Delta\phi = 0.1 \text{ rad}$$

iv) $BW_{FM} = 2(\Delta f + f_m) = 2(100 + 1000)$

$= 2.2$ **kHz**

Q.10 An angle modulated signal with carrier frequency $\omega_c = 2\pi \times 10^5$ is described by the equation $\phi_{FM}(t) = 10 \cos(\omega_c t + 5 \sin 2000 t + 10 \sin 3000 \pi t)$.
Find :
i) The power of the modulated signal ii) Frequency deviation Δf
iii) Phase deviation $\phi\Delta$ iv) Bandwidth of $\phi_{FM}(t)$.

[SPPU : May-08, Dec.-13, Marks 8]

Ans. : Given :

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

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

In this case, $f_m = 3000\pi / 2\pi = 1500$ Hz

i) Carrier amplitude is 10 and power of modulating signal is,

$$P = \frac{(10)^2}{2R} = \frac{100}{2R} \qquad \text{Assuming } R = 1$$

$$P = 50 \text{ W}$$

ii) Instantaneous frequency ω_i is given by :

$$\omega_i = \frac{d}{dt}\theta(t)$$

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

Carrier deviation is $10000\,\pi \cos 2000\,\pi t + 30000\,\pi \cos 3000\,\pi t$

Two sinusoids will add in phase at some point and maximum value of this is $10000\,\pi + 30000\,\pi$. This is maximum carrier deviation $\Delta\omega$.

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{10000\pi + 30000\pi}{2\pi} = \mathbf{20000\ Hz}$$

iii) The angle $\theta(t) = \omega_c t + (5 \sin 2000\,\pi t + 10 \sin 3000\,\pi t)$

Phase deviation is the maximum value of angle inside the parenthesis and is given by :

$$\Delta\phi = 5 + 10 = \mathbf{15\ rad.}$$

iv) $$B_{FM} = 2\,(\Delta f + f_m)$$

$$= 2\,(20000 + 1500)$$

$$= 2\,(21500) = \mathbf{43000\ Hz}$$

Q.11 A carrier is frequency modulated with a sinusoidal signal of 2 kHz resulting in frequency deviation of 5 kHz :

i) Find bandwidth of modulated signal.

ii) The amplitude of modulating sinusoid is increased by a factor of 3 and its frequency is halved. Find the maximum frequency deviation and bandwidth of new modulated signal.

☞ [SPPU : Dec.-11, Marks 8, May-18, June-22, Marks 6]

Ans. : Given : $f_m = 2$ kHz, $\Delta f = 5$ kHz

i) Bandwidth $= 2\,(\Delta f + f_m) = 2\,(5 + 2) = \mathbf{14\ kHz}$

ii) Now $f_m = \frac{2\text{ kHz}}{2} = 1$ kHz

Δf = Deviation constant × Amplitude of modulating signal

Thus, when the amplitude of modulating signal is increased by factor 3, Δf is also increased by factor of 3

$$\therefore \quad \Delta f = 5\text{ kHz} \times 3 = 15\text{ kHz}$$

$\therefore$ New bandwidth $= 2\,(\Delta f + f_m) = 2\,(15 + 1) = \mathbf{32\ kHz}$

Q.12 FM wave is represented by the following equation, $V = 10 \sin[5 \times 10^8 t + 4 \sin 1250\,t]$. Calculate :

i) Carrier and modulating frequency

ii) Modulation index and maximum deviation

iii) Power dissipated by FM wave in 5 Ω resistor.

☞[SPPU : May-18, June-22, Dec.-22, Marks 6]

Ans. : Comparing the given equation with standard form,

$$V = A \sin[\omega_c t + m_f \sin(\omega_m t)]$$

We get, $A = 10$, $\omega_c = 5 \times 10^8$, $m_f = 4$, $\omega_m = 1250$

i) $$f_c = \frac{\omega_c}{2\pi} = \frac{5 \times 10^8}{2\pi} = \mathbf{79.577\ MHz}$$

$$f_m = \frac{\omega_m}{2\pi} = \frac{1250}{2\pi} = \mathbf{198.94\ Hz}$$

ii) Modulation index,

$$m_f = 4$$

Maximum deviation $\Delta_f = m_f \times f_m = 4 \times 198.94 = \mathbf{795.77\ Hz}$

iii) $$P = \frac{(V_{rms})^2}{R} = \left(\frac{A}{\sqrt{2}}\right)^2 \times \frac{1}{5} = \left(\frac{10}{\sqrt{2}}\right)^2 \times \frac{1}{5} = \mathbf{10\ W}$$

3.3 : Narrowband FM

Q.13 What is narrowband FM. **☞ [SPPU : Dec.-07, May-09]**

OR

Derive equation for bandwidth of NBFM. Give block diagram of generation of narrowband FM. **☞ [SPPU : Dec.-12]**

Ans. :

- When the modulation index is less than 1, it is called narrowband FM. The FM equation given by equation (Q.3.6) can also be expressed as,

$$v_{FM} = V_c \cos\left[\omega_c t + \frac{\Delta f}{f_m} \sin \omega_m t\right]$$

Here $\frac{\Delta f}{f_m} = m_f$, i.e modulation index. Then above equation becomes,

$$v_{Fm} = V_c \cos\left[2\pi f_c t + m_f \sin 2\pi f_m t\right] \quad \text{... (Q.13.1)}$$

- Expanding above relation,

$$v_{Fm} = V_c \cos(2\pi f_c t) \cos[m_f \sin(2\pi f_m t)] - V_c \sin(2\pi f_c t) \sin[m_f \sin(2\pi f_m t)] \quad \text{... (Q.13.2)}$$

- For narrowband FM, the modulation index, m_f is very small therefore following approximations can be considered.

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

and $\sin[m_f \sin(2\pi f_m t)] \approx m_f \sin(2\pi f_m t)$

- Therefore equation (Q.13.2) becomes,

$$v_{Fm} = V_c \cos(2\pi f_c t) - m_f V_c \sin(2\pi f_c t) \sin(2\pi f_m t) \quad \text{... (Q.13.3)}$$

- Above equation can be further expanded as,

$$v_{Fm} = V_c \cos(2\pi f_c t) + \frac{1}{2} m_f V_c \{\cos 2\pi(f_c + f_m)t - \cos 2\pi(f_c - f_m)t\} \quad \text{... (Q.13.4)}$$

This equation gives the spectrum of narrowband FM. Observe that there is carrier frequency (f_c), upper sideband $(f_c + f_m)$ and lower sideband $(f_c - f_m)$ and hence bandwidth $2 f_m$.

Generation of Narrowband FM

- Fig. Q.13.1 shows the block diagram of narrowband FM signal generator.

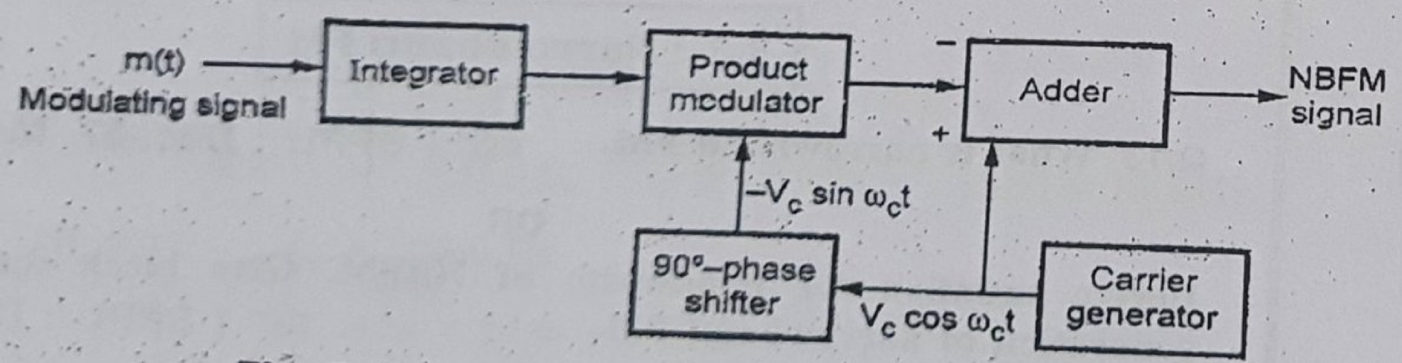

Fig. Q.13.1 Block diagram of NBFM generator

- One of the input of product modulator is 90° phase shifted carrier signal, $V_c \cos \omega_c t$, where $\omega_c = 2\pi f_c$. The other input of the product modulator is integrated modulating signal $\int_0^t m(t)\,dt$.

- The output of product modulator is added to original carrier signal $V_c \cos \omega_c t$. This gives the narrowband FM signal as per equation (Q.13.3).

3.4 : Wideband FM

Q.14 What is wideband FM.

☞ [SPPU : Dec.-07, Marks 8, Dec.-19, Marks 6]

OR Compare and contrast wideband FM and narrow band FM.

☞ [SPPU : Dec.-99,02,03,06,09,11, May-05, 14, Marks 8, Dec.-22, Marks 6]

Ans. : If the modulation index is higher than 10, then it is called wideband FM.

Sr. No.	Parameter	Narrowband FM	Wideband FM
1.	Modulation Index	Less than 1	Greater than 1
2.	Equation	$v_{FM}(t) = V_c \cos(2\pi f_c t) - m_f E_c \sin(2\pi f_c t) \sin(2\pi f_m t)$	$v_{FM}(t) = V_c \sum_{n=-\infty}^{\infty} J_n(m_f) \cos[2\pi(f_c + n f_m)t]$
3.	Spectrum	Contains two sidebands and carrier.	Contains infinite number of sidebands and carrier.
4.	Bandwidth	$2 f_m$	$2(\Delta f + f_{m(max)})$
5.	Noise suppression	Poor	Better
6.	Maximum modulation index	5 to 2500	Just greater than unity
7.	Range of modulating frequency	30 Hz to 3 kHz	30 Hz to 15 kHz

8.	Maximum deviation	5 kHz	75 kHz
9.	Transmission quality	Low	High
10.	Applications	Used for mobile communication such as police wireless, ambulances, short range ship to shore communication etc.	Used for entertainment broadcasting

Table Q.14.1 Comparison between narrowband and wideband FM

3.5 : Phase Modulation

Q.15 Derive the equation for PM signal.

[SPPU : Dec.-99, 04, 13, 16, Marks 3]

Ans. : • Consider the sinusoidal carrier signal

$$v_c = V_C \sin(\omega_c t + \phi_c) \quad ...(Q.15.1)$$

- If the phase ϕ_c is varied such that its magnitude is made proportional to the instantaneous amplitude of the modulating signal, the resulting wave is phase modulated.
- Phase of phase modulated signal is proportional to instantaneous modulating signal which can be given as,

$$\phi_{pm}(t) = \phi_c + Kv_m = \phi_c + KV_m \sin \omega_m t \quad ...(Q.15.2)$$

where K is phase deviation constant.

$$\therefore \quad K = Kp$$

- Usually ϕ_c can be dropped from equation (Q.15.2) since it is a constant that does not affect the modulation.
- Now equation for phase modulated wave becomes

$$v_{pm} = V_c \sin(\omega_c t + KpV_m \sin\omega_m t) \quad ...(Q.15.3)$$

- In order to emphasize the similarity with sinusoidal frequency modulation we replace KpV_m by m_p, where m_p is phase modulation index

$$\text{Then} \quad v_{pm} = V_c \sin(\omega_c t + m_p \sin\omega_m t) \quad ...(Q.15.4)$$

Q.16 "Phase and frequency modulation are inseparable'. Explain.

[SPPU : May-05, 10, Dec.-13, Marks 8]

Ans. :

- In phase modulation, the greater the amplitude of the modulating voltage, the greater the phase shift.
- Let us assume that positive half cycle of modulating sinusoidal signal produces a lagging phase shift and negative half cycle produces a leading phase shift.
- Then as the modulating signal goes positive amount of phase lag increases with the amplitude of the modulating signal. This means the carrier output is time delayed, as phase shift is equivalent to time delay.
- The time delay increases with the amplitude of the modulating signal, resulting in lowering of frequency.
- When the modulating signal goes negative, phase shift is leading phase shift. The result is carrier frequency increase. This is shown in the Fig. Q.16.1.

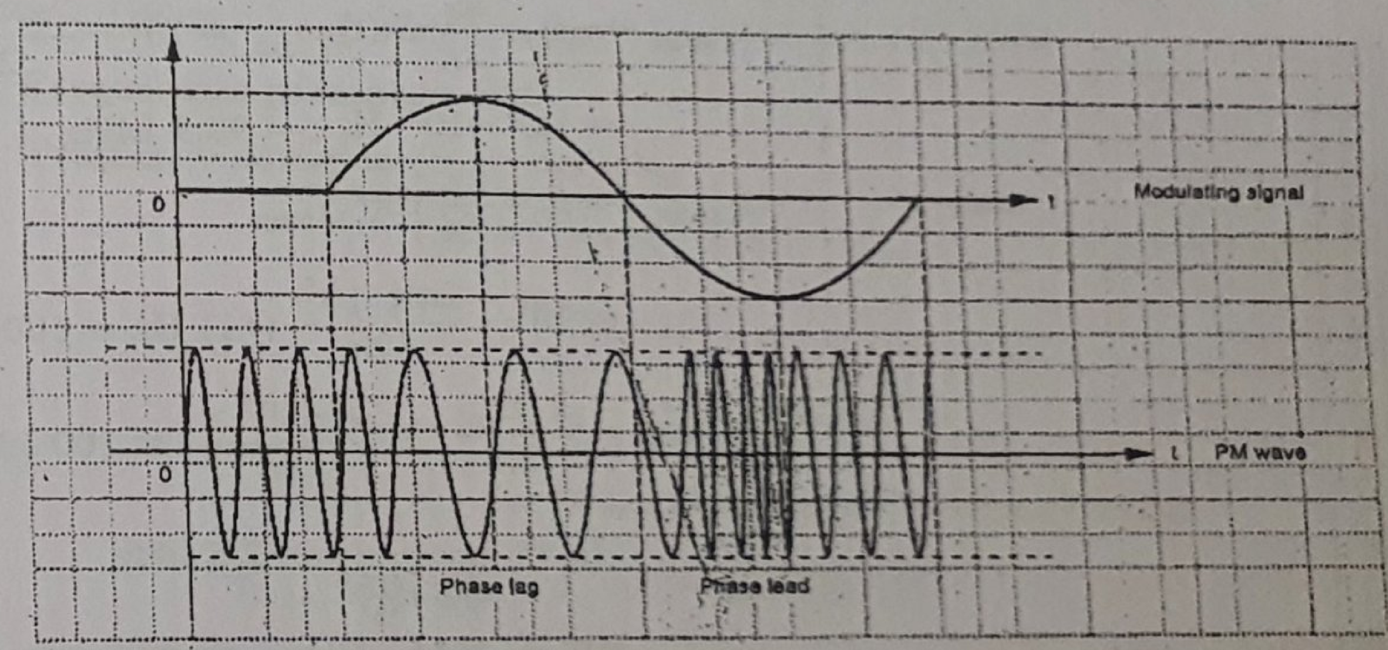

Fig. Q.16.1

- In this way, phase modulation produces frequency modulation. Since the amount of phase shift is variable, the effect is as if the carrier frequency is varied.

- Thus phase modulation gives rise to frequency modulation ; therefore many times phase modulation is often referred to as indirect method of frequency modulation.

Q.17 Estimate bandwidth of FM and PM wave when the modulating signal m(t) is bipolar triangular wave with peak magnitude as 1 V. The period of the signal is 4×10^{-4} s. Also sketch FM and PM waves indicating maximum and minimum frequencies, given that f_c = 100 MHz. Assume bandwidth of m(t) to be the frequency of its third harmonic. Given that $k_f = 2\pi\times10^5$ and $k_p = 5\pi$.

[SPPU : Dec.-12,13, Marks 10]

Ans. : For FM : Given : $f_c = 100\text{ MHz} = 10^8\text{ Hz},\ k_f = 2\pi\times10^5$

$$m(t)_{min} = -1\text{ V},\quad m(t)_{max} = +1\text{ V}$$

$$f_{i\,min} = f_c + \frac{k_f}{2\pi}m(t)_{min} = 10^8 + \frac{2\pi\times10^5}{2\pi}(-1)\text{ Hz}$$

$$= 10^8 - 10^5\text{ Hz} = 99.9\text{ MHz}$$

and

$$f_{i\,max} = f_c + \frac{k_f}{2\pi}m(t)_{max}$$

$$= 10^8 + \frac{2\pi\times10^5}{2\pi}(+1)\text{ Hz} = 100.1\text{ MHz}$$

$$\Delta f = \frac{k_f E_m}{2\pi} = \frac{2\pi\times10^5\times1}{2\pi} = 100\text{ kHz}$$

$$f_m = \frac{1}{4\times10^{-4}} = 2.5\text{ kHz}$$

$\therefore$ $$BW = 2(\Delta f + f_m) = 2(100 + 2.5) = 205\text{ kHz}$$

For PM :

$$f_i = f_c + \frac{k_p}{2\pi}\frac{d}{dt}[m(t)]$$

$\therefore$ $$f_i = f_c + \frac{8\pi}{2\pi}\frac{d}{dt}[m(t)] = f_c + 4\frac{d}{dt}[m(t)]$$

$\Rightarrow$ $$f_{i\,min} = f_c + 4\left[\frac{d}{dt}m(t)\right]_{min}$$

$$f_i = 10^8 + 4[-32000] = 99.872\text{ MHz}$$

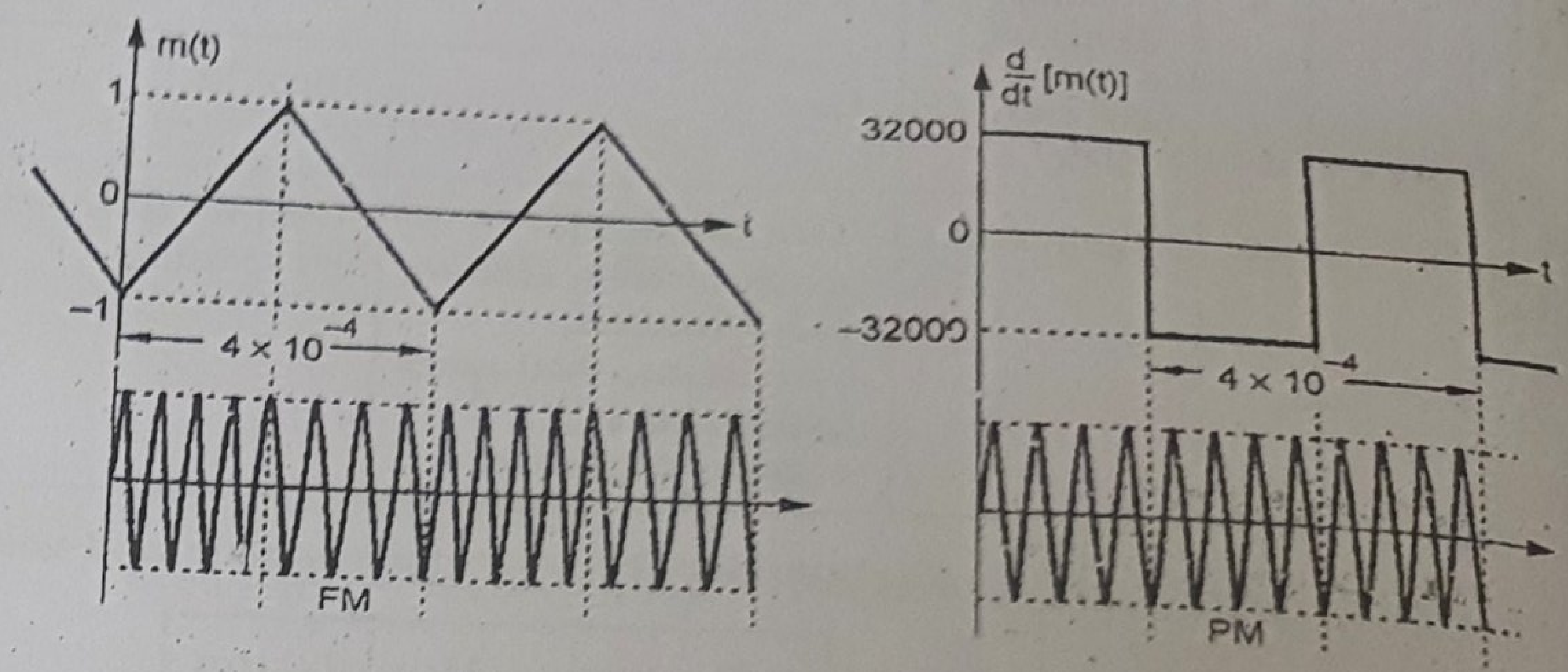

Fig. Q.17.1 FM and PM waveform

$$f_{i\,max} = 10^8 + 4[32000] = 100.128\text{ MHz}$$

$$\Delta f = \frac{K_p E_m}{2\pi} = \frac{5\pi\times32000}{2\pi} = 80\text{ kHz}$$

$$BW = 2(\Delta f + f_m) = 2(80 + 2.5) = 165\text{ kHz}$$

Q.18 Given $m(t) = \sin 2000\,\pi t$, $k_f = 200000\,\pi$ rad / V and $k_p = 10\,\pi$
i) Estimate the bandwidth of FM and PM.
ii) Repeat part (i) if the message signal amplitude is doubled.
iii) Repeat part (i) if the message signal frequency is doubled.

[SPPU : May-06, 11, Marks 10, May-16, Marks 6]

Ans. : Given : $m(t) = \sin 2000\,\pi t$, $k_f = 200000\,\pi$ rad/V and $k_p = 10$.

The peak amplitude of m(t) is unity. Hence, $E_m = 1$

i) $$\Delta f = \frac{k_f E_m}{2\pi} = \frac{200000\pi}{2\pi}\times1 = 100\text{ kHz}$$

$$f_m = \frac{2000\pi}{2\pi} = 1000\text{ Hz}$$

$$m_p = k_p E_m = 10\times1 = 10\text{ rad}$$

BW for FM $= 2(\Delta f + f_m) = 2(100\text{ kHz} + 1\text{ kHz}) = 202$ kHz

$$m'(t) = 2000\,\pi\,\cos 2000\pi t$$

Peak amplitude of $m'(t)$ is 2000π. Hence, $E'_m = 2000\,\pi$

$$\Delta f = \frac{1}{2\pi} k_p E_m = \frac{10 \times 2000\pi}{2\pi}$$

$$= 10 \text{ kHz}$$

$$\text{BW for PM} = 2(\Delta f + f_m) = 2(10 \text{ kHz} + 1 \text{ kHz}) = 22 \text{ kHz}$$

ii) If the message signal amplitude is doubled.

For FM

$$\Delta f = \frac{k_f}{2\pi} E_m = \frac{20000\,\pi \times 2}{2\pi} = \mathbf{200 \text{ kHz}}$$

$\therefore$ **Bandwidth for FM** $= 2\ (200 \text{ kHz} + 1 \text{ kHz}) =$ **402 kHz**

For PM : Doubling m(t) doubles its derivative, so that now $E'_m = 4000\,\pi$

$$\therefore \qquad \Delta f = \frac{10 \times 4000\pi}{2\pi} = 20 \text{ kHz}$$

$\therefore$ **Bandwidth for PM** $= 2\ (20 \text{ kHz} + 1 \text{ kHz})\ 1000 \text{ Hz} =$ **42 kHz**

iii) If the message signal frequency is doubled.

For FM :

$$f_m = \frac{2000\pi}{2\pi} \times 2 = 2000 \text{ Hz}$$

$\therefore$ **BW for FM** $= 2\ (\Delta f + f_m) = 2\ (100 \text{ kHz} + 1 \text{ kHz}) =$ **204 kHz**

For PM

Now $m(t) = \sin 4000\pi t$, hence $m'(t) = 4000\,\pi \cos 4000\pi t$

Hence $E'_m = 4000\,\pi$

$$\Delta f = \frac{k_p E'_m}{2\pi} = \frac{10 \times 4000\,\pi}{2\,\pi} = 20 \text{ kHz}$$

$\therefore$ **BW for PM** $= 2\ (20 \text{ kHz} + 2 \text{ kHz})$

$= $ **44 kHz**

Q.19 Sketch Frequency Modulation (FM) and Phase Modulation (PM) waveform for the digital modulation signal m(t), the signal given in Fig. Q.19.1.

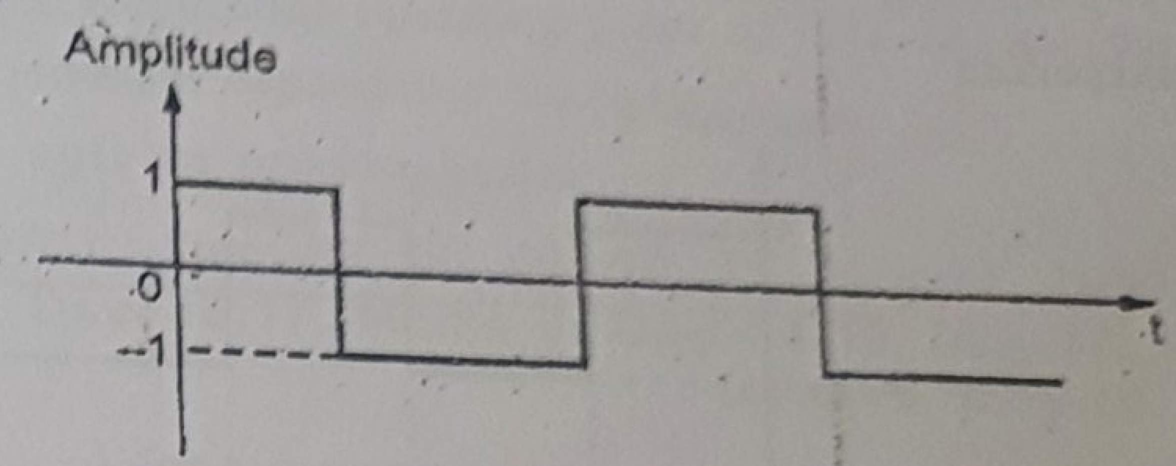

Fig. Q.19.1

The constants k_f and k_p are $2\pi \times 10^5$ and $\pi / 2$ respectively and $f_c = 100$ MHz. Calculate the frequencies present in the FM and PM waves. What is the limitation on the product kpm(t).

[SPPU : May-16, Marks 6]

Ans. : Given : $k_f = 2\pi \times 10^5$, $k_p = \pi/2$ and $f_c = 100$ MHz.

1. Frequencies present in FM

Frequency deviation $(\delta) = k_f \times V_m$

$$(\delta) = 2\pi \times 10^5 \times (\pm 1) \text{ Volt} \qquad \because V_m = \pm 1 \text{Volt}$$

$$\therefore \qquad = \pm 2\pi \times 10^5$$

$$\therefore \qquad f_{\min\,(V_m = -1)} = f_c - \delta = f_c - 2\pi \times 10^5$$

$$= 100 \times 10^6 - (2\pi \times 10^5) = \mathbf{99.37 \text{ MHz}}$$

$$\therefore \qquad f_{\max\,(V_m = +1)} = f_c + \delta = f_c + 2\pi \times 10^5$$

$$= 100 \times 10^6 + (2\pi \times 10^5) = \mathbf{100.63 \text{ MHz}}$$

2. Frequencies present in PM

Phase deviation $\theta = k_p \cdot V_m = \pi / 2 \cdot \pm 1 \qquad \because V_m = \pm 1$

3. Waveforms

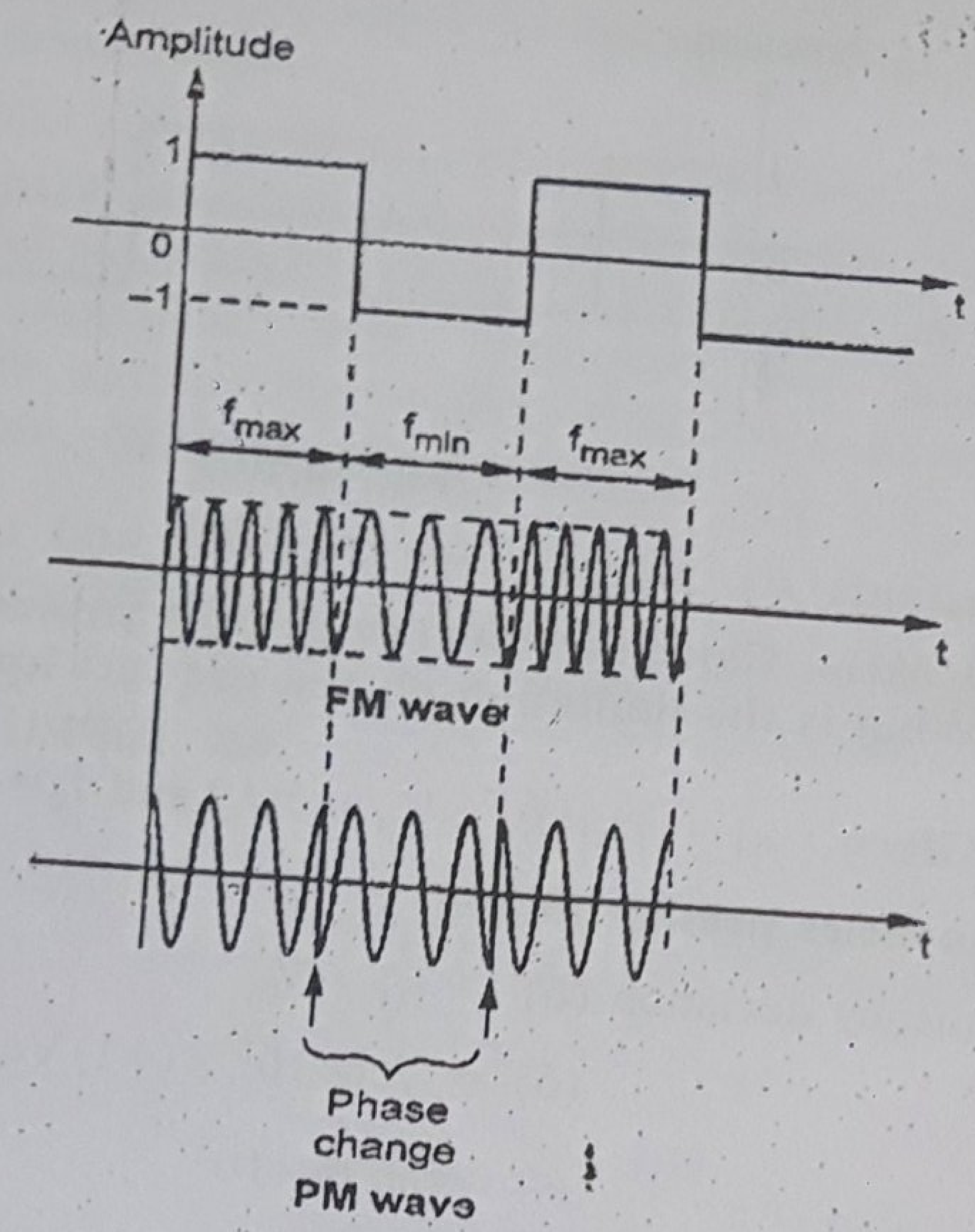

Fig. Q.19.1

3.6 : Generation of FM

Q.20 Explain the direct method for FM generation with block diagram.

[SPPU : May-13, Dec.-13, Marks 8, Dec.-17, Marks 6, Dec.-19, Marks 2]

Ans. : • In the direct method of FM generation, the instantaneous frequency of the carrier wave is varied according to the modulating signal by means of a device called a **voltage-controlled oscillator.**

- To produce FM, we use a circuit that converts a modulating voltage into a corresponding change in capacitance or inductance to the oscillator tank circuit.
- When modulating signal is not applied, the oscillator frequency is equal to the desired carrier frequency then FM will be produced when the reactance of the device is varied by the modulating voltage.

- The larger the amplitude of the modulating voltage, the larger the reactance variation and therefore the frequency deviation.
- The direct method for the generation of FM wave has the disadvantage that the carrier frequency is not obtained from a highly stable oscillator.
- In practical circuits, this problem is solved by providing some auxiliary means by which a very stable frequency generated by a crystal will be able to control the carrier frequency.
- Here, the output of the FM generator is applied to a mixer together with the output of a crystal-controlled oscillator.

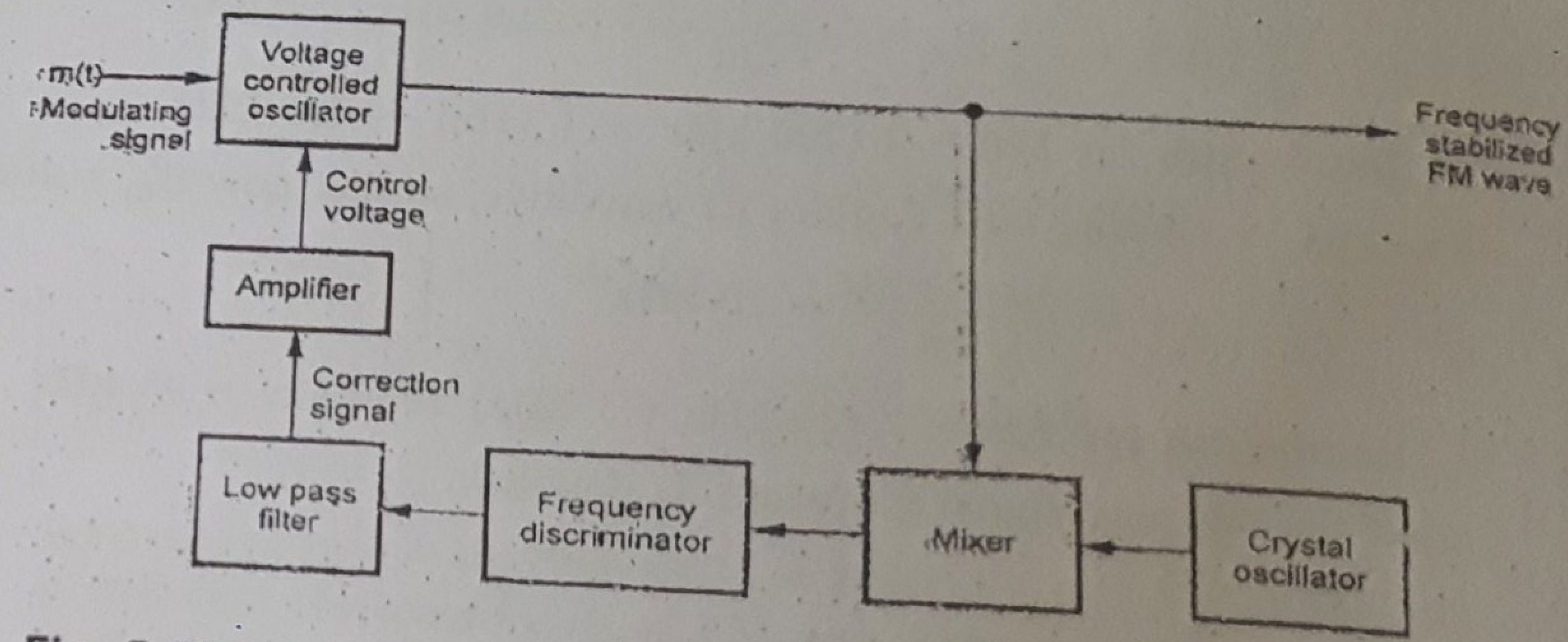

Fig. Q.20.1 Feedback scheme for generation of frequency stabilized FM wave

- The difference frequency term from the output of mixer is applied to a frequency discriminator. A frequency discriminator is a device whose output is an instantaneous voltage amplitude that is proportional to the instantaneous frequency of the FM wave applied to its input.
- The output of frequency discriminator is applied to low-pass filter.
- For correct carrier frequency the output of low-pass filter is zero.
- However, it gives dc output voltage proportional to the carrier frequency deviation from the assigned value of the carrier frequency.
- This dc voltage, after suitable amplification is applied to voltage control oscillator to modify the frequency of oscillator in direction that tends to restore the carrier frequency to its assigned value.

Q.21 Write a note on reactance modulator.

☞ [SPPU : May-99, Dec.-95, 08, 09]

OR Explain basic reactance modulator for FM generation.

☞ [SPPU : May-12, Marks 8]

Ans. :

- The Fig. Q.21.1 the basic circuit of a FET reactance modulator, which behaves as a reactance that may be connected across the tuned circuit of the oscillator for producing frequency modulation.

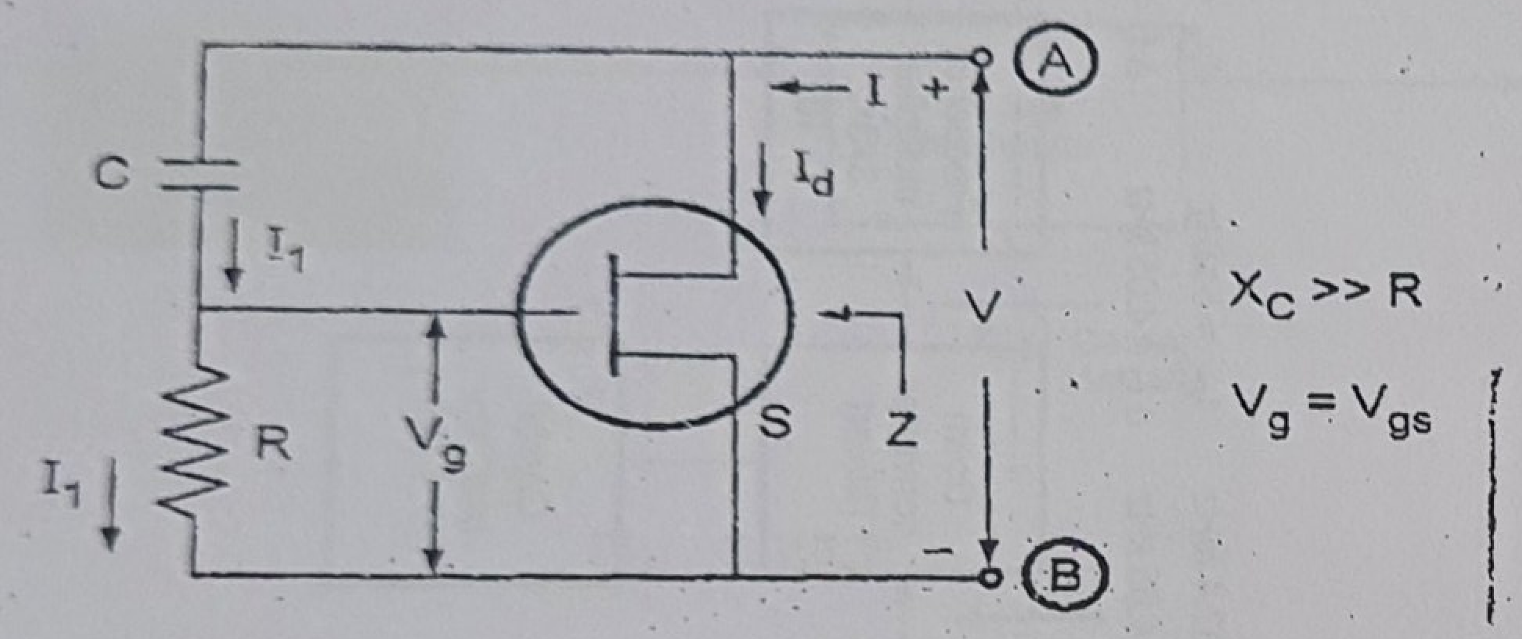

Fig. Q.21.1 FET reactance modulator

- Neglecting gate current, let the current through C and R be I_1. At the carrier frequency, the reactance of C is much larger than R. This is achieved by properly selecting the value of C.

Then,
$$I_1 = \frac{V}{R + \frac{1}{j\omega C}}$$
$$\approx \frac{V}{\frac{1}{j\omega C}} = j\omega C V$$
$$V_g = I_1 R = j\omega C R V$$
$$I_d = g_m V_{gs} = g_m V_g = j\omega C R g_m V$$

- Then impedance of FET is,

$$Z = \frac{V}{I_d} = \frac{1}{j\omega[g_m CR]} = \frac{1}{j\omega[C_{eq}]}$$

- Thus impedance is seen to be capacitive reactance with ,

$$C_{eq} = g_m CR \qquad \text{... (Q.21.1)}$$

- By modulating voltage the operating point of FET i.e. g_m can be varied and hence equivalent capacitance changes.

Q.22 Write a note on pre-emphasis.

☞ [SPPU : Dec.-99, 08, 09, 10, 11, May-97, 2000, 04, 07 Dec.-15,17, Marks 3, Dec.-18,19, May-19, Marks 6 Dec.-16, May-16, Marks 4

OR Explain need for preemphasis.

☞ [SPPU : May-05, 08, Dec.-08, 10, 11 June-22, Dec.-22, Marks 6]

Ans. : • An important advantage of FM is that it can be used to transmit a wide range of audio frequencies upto 15 kHz. The frequencies above 5 kHz contain mainly the higher harmonics of the fundamental frequency [lower frequencies] in voice or music.

- These higher harmonics are of smaller amplitude but if they are reproduced at the receiver they give the unusually fine quality that is characteristic of F.M.
- The extent to which noise is suppressed in FM, is greater the larger the frequency - deviation of the signal. Then the signal to noise ratio is better.
- However the higher harmonics being smaller in amplitude, produce smaller frequency deviations and then the less complete is the suppression of noise at these higher frequencies.

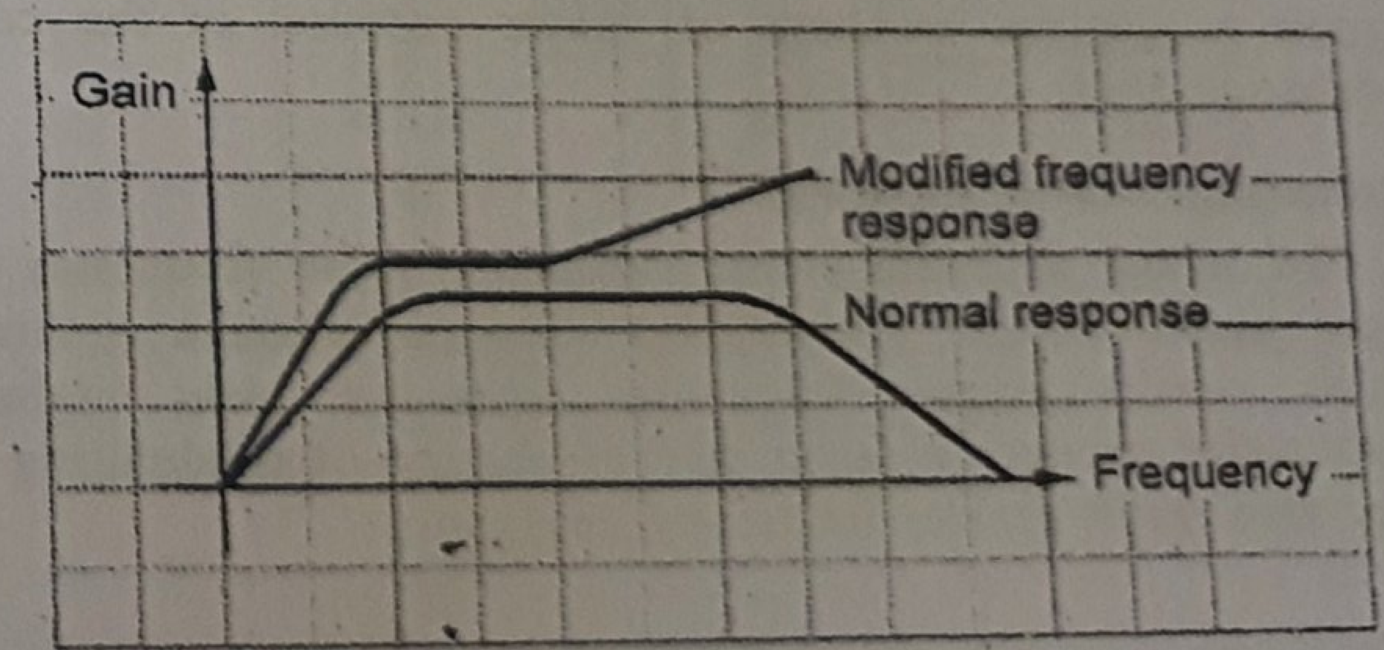

Fig. Q.22.1 Modified frequency response

- As a result, it is desirable to increase the amplitude of the higher modulating frequencies before modulation. Termed "pre-emphasis', this improves the noise suppression at the receiver by making modulation index at high frequencies greater than would otherwise be the case.
- In pre-emphasis the frequency response characteristic of the audio amplifier is modified as illustrated in the Fig. Q.22.1.

Q.23 Explain with block diagram the Armstrong method of FM generation. ☞ **[SPPU Dec.-98, 99, 01, 05, 06, 07, 09, 10, 11, May-2000, 04, 05, Marks 10, May-17,19, Dec.-18, June-22, Marks 6, Dec.-19, Marks 4]**

Ans. : • FM generation using Armstrong method is achieved by performing following steps :

1. Generation of PM wave
2. Generation of NBFM from PM wave
3. Generation of WBFM from NBFM

- Fig. Q.23.1 shows block diagram of FM generation using Armstrong method. (See Fig. Q.23.1 on next page)
- As shown in the Fig. Q.23.1 crystal oscillator is used to generate a stable unmodulated carrier which is applied to the 90° phase shifter and the summing circuit.
- The 90° phase shifted carrier is applied to the balanced modulator alongwith the modulating signal.
- Thus 90° phase shifted carrier is DSBSC modulated in the balanced modulator giving us only two sidebands with their resultant in phase with the 90° shifted carrier.
- The two sidebands and the unshifted carrier are applied to a summing circuit to get the resultant of vector addition of the carrier and two sidebands.

Generating NBFM from PM

- In PM along with the phase variation, some frequency variation also takes place. Higher modulating voltages produce greater phase shift which results greater frequency deviation. And higher modulating frequencies produce a faster rate of change of modulating voltage hence they also result greater frequency deviation.

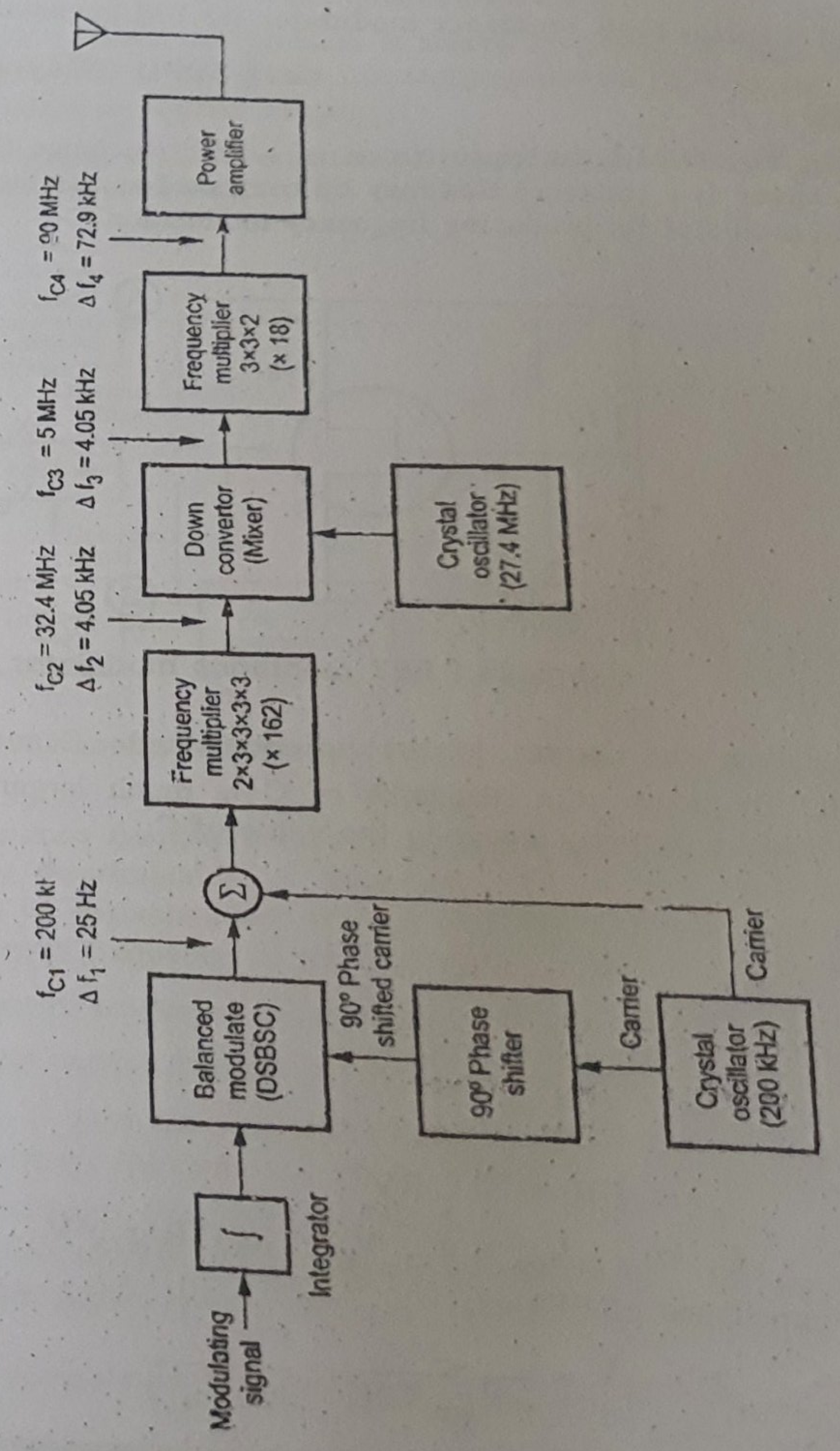

Fig. Q.23.1 Armstrong indirect FM transmitter

- Thus, in PM the carrier frequency deviation is proportional to,
  - Modulating voltage and
  - Modulating frequency
- However, in FM the frequency deviation is only proportional to the modulating voltage regardless of its frequency.
- Thus, to suppress high frequency modulating signals and to make frequency deviation independent of modulating frequency the modulating signal is passed through a low pass RC filter (integrator) as shown in Fig. Q.23.2.

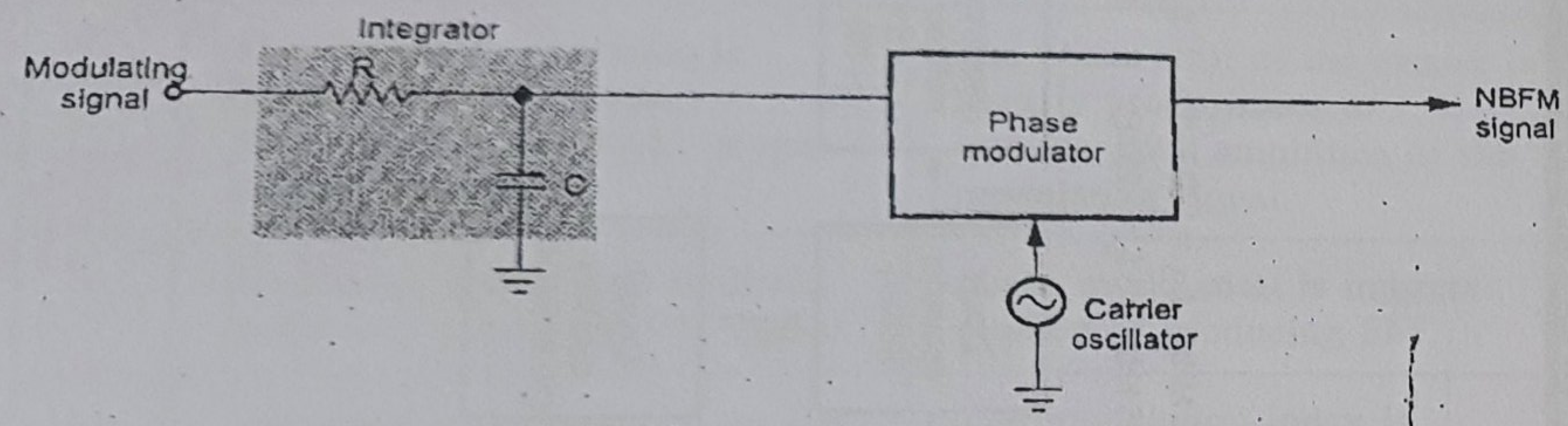

Fig. Q.23.2

- As a result, the high frequency modulating signals are attenuated but there is no change in the amplitudes of low frequency modulating signals.
- The output of integrator is then applied to the phase modulator.
- With this input the frequency deviation at the output of the phase modulator will be effectively proportional only to the modulating voltage and hence we obtain FM wave at the output of phase modulator.
- The NBFM is converted to WBFM by using frequency multipliers.

Q.24 Design an Armstrong indirect FM modulator to generate an FM carrier with a carrier frequency of 96 MHz and $\Delta f = 20$ kHz. A narrowband FM generator with $f_c = 200$ kHz and adjustable Δf in the range of 9 to 10 Hz is available. There is an oscillator with the range of 9 to 10 MHz. There is a adjustable frequency in the range of 9 to 10 MHz. There is a bandpass filter with any centre frequency and only frequency doublers are available. (only block diagram is expected).

[SPPU : May-08, Dec.-12, Marks 8]

Ans. : Given Δf = 20 kHz, Δf available = 9 to 10 Hz.

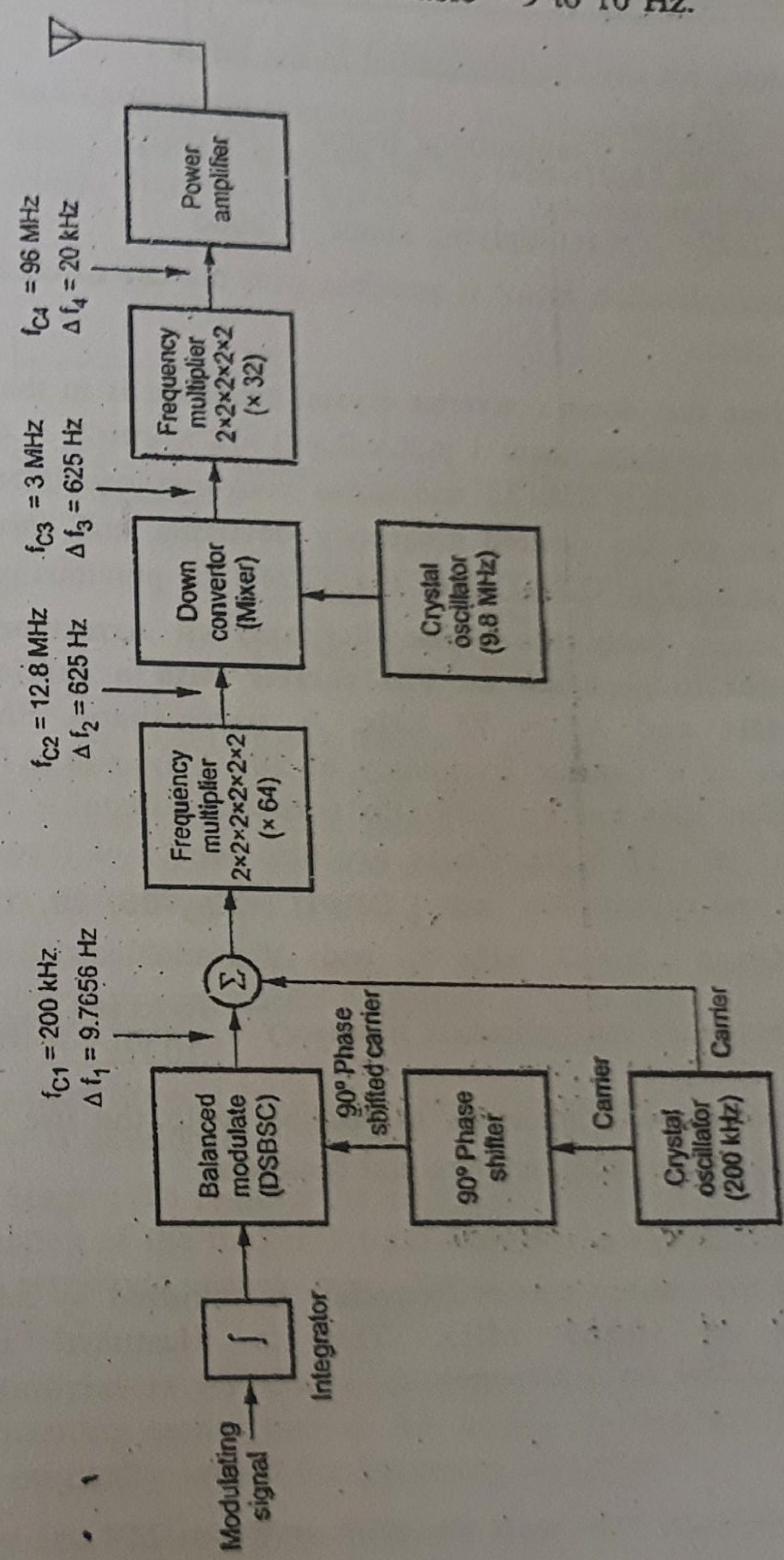

Fig. Q.24.1

Therefore, we need multiplication in the range :

$$\frac{20\text{ kHz}}{9} < \text{multiplying factor} < \frac{20\text{ kHz}}{10}$$

$$= 2222 \quad < \text{multiplying factor} < 2000$$

This multiplication factor is possible with the use of chain of 11 doublers. ($2^{11} = 2048$).

It is given that down converter crystal oscillator is in the range of 9 to 10 MHz. By grouping chain 1 = $2 \times 2 \times 2 \times 2 \times 2 \times 2 = 64$, and chain 2 = $2 \times 2 \times 2 \times 2 \times 2 = 32$ and down converter oscillator frequency = 9.8 MHz we get the desired frequency deviation and carrier frequency as shown in the Fig. Q.24.1. (See Fig. Q.24.1 on previous page).

Q.25 Design (only the block diagram) an Armstrong indirect FM modulator to generate an FM carrier with a carrier frequency of 98.3 MHz and Δf = 75 kHz. A narrowband FM generator is available at a carrier frequency of 100 kHz and Δf = 10 Hz. The stock room also has an oscillator with an adjustable frequency in the range of 10 - 11 MHz. There are also plenty of frequency doublers, triplers and quintuples. ☞ **[SPPU : May-06, 10, 13, Marks 10]**

Ans. : Given : $\Delta f = 75$ kHz and Δf available = 10 Hz

$\therefore$ The frequency multiplication necessary = $\frac{75\text{ kHz}}{10\text{ Hz}} = 7500$

This frequency multiplication is possible with the use of the following chain of the quintuples, triplers and doublers.

5, 5, 5, 5, 3, 2, 2.

However, the desired carrier frequency is achieved by adjusting oscillator frequency to 10.865 MHz. This is illustrated in Fig. Q.25.1 (See Fig. Q.25.1 on next page).

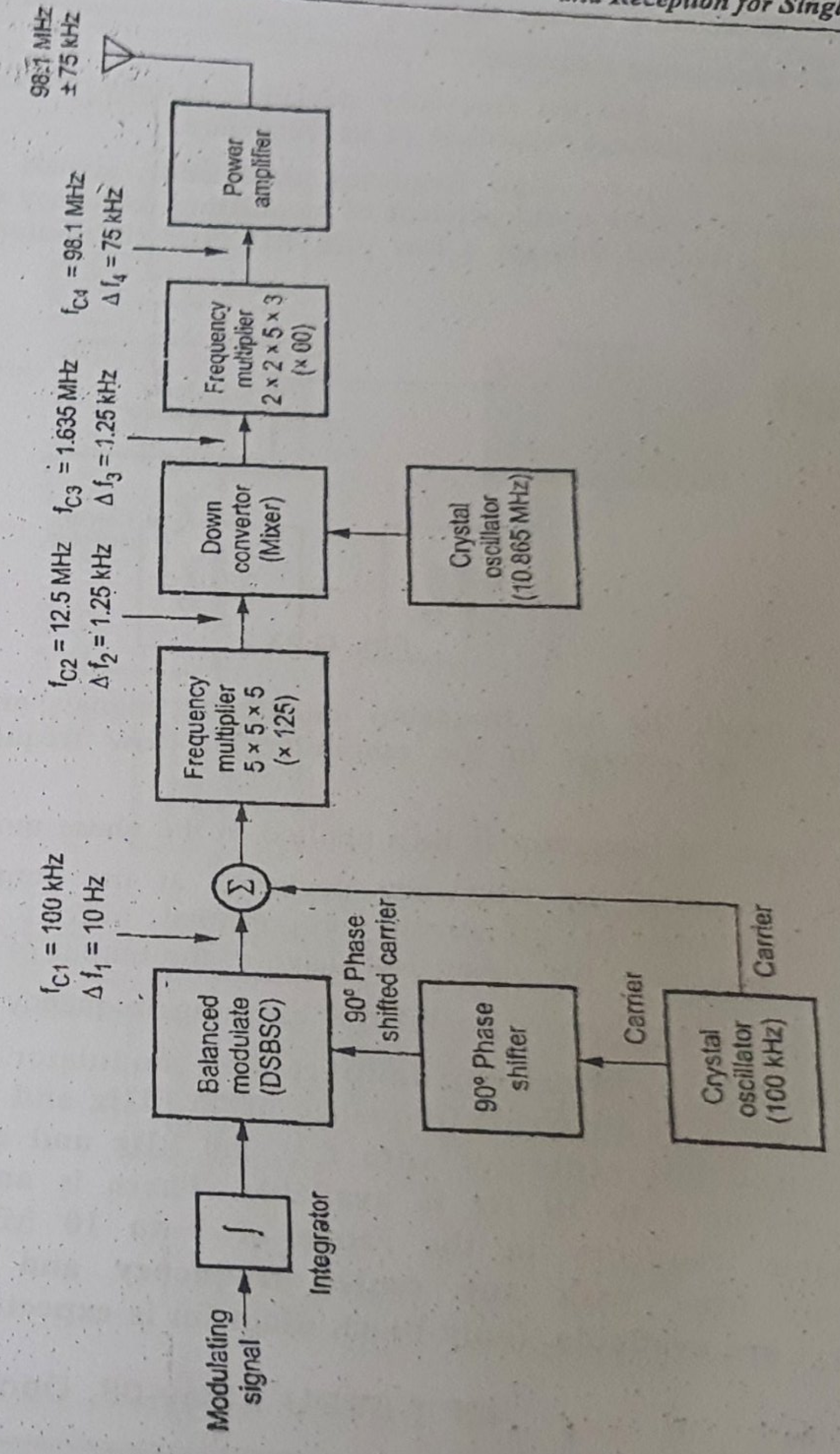

Fig. Q.25.1

Q.26 Differentiate between frequency modulation and phase modulation.

[SPPU : Dec.-14,15, June-22, Marks 6]

Ans. :

Sr. No.	FM	PM
1.	The equation for FM wave is : $v = A \sin [\omega_c t + m_f \sin \omega_m t]$	The equation for PM wave is $v = A \sin [\omega_c t + m_p \omega_m t]$
2.	The frequency deviation is linearly proportional to instantaneous amplitude of the modulating signal.	The phase shift of the carrier is linearly proportional to instantaneous amplitude of the modulating signal.
3.	Frequency modulation is direct method of producing FM signal.	Phase modulation is indirect method of producing FM.
4.	The modulation index of an FM signal is the ratio of the frequency deviation to the modulating frequency.	The modulation index is proportional to the maximum amplitude of the modulating signal.
5.	To have better quality of transmission and reception of higher audio frequencies, pre-emphasis and de-emphasis circuits are used.	The amount of frequency shift produced by a phase modulator increases with the modulating frequency. Hence an audio equalizer is required to compensate this.
6.	Amplitude of the FM wave is constant.	Amplitude of the PM wave is constant.
7.	Noise is better suppressed in FM systems as compared to PM system.	Noise immunity is inferior to that of FM.
8.	FM is mainly used for FM broadcasting used for entertainment purposes.	PM is used in mobile communication system.

3.7 : FM Receivers

Q.27 Draw and explain superheterodyne FM receiver.

[SPPU : May-06,12,13, Dec.-10,11,22, Marks 6]

Ans. : The block diagram of typical F.M. receiver is shown in the Fig. Q.27.1.

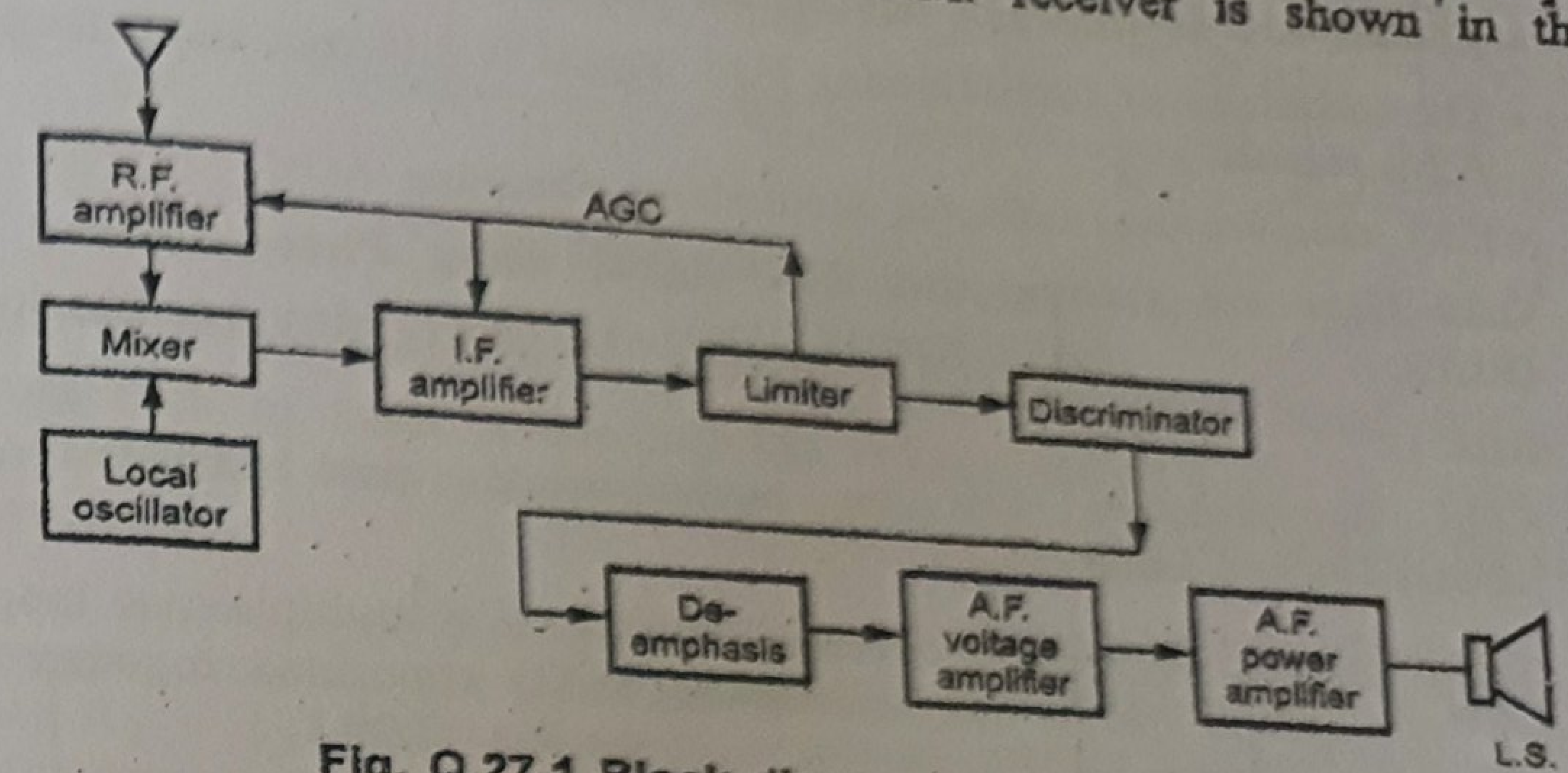

Fig. Q.27.1 Block diagram of F.M. receiver

- **R.F. Amplifier Stage :** Since F.M. signal has a larger bandwidth it is likely to encounter more noise. Hence to reduce the noise figure of the receiver, an RF amplifier stage is used. The RF amplifier stage matches the antenna to the receiver.
- **Mixer Stage :** With the help of local oscillator, this stage down converts the incoming carrier frequency to I.F., which is 10.7 MHz for F.M. receiver.
- **I.F. Amplifier Stage :** In the I.F. amplifier stages, the most of the gain of receiver is developed.
- **Limiter Stage :** To remove the amplitude variations of the signal is the main function of the limiter. At the output of the limiter stage, we get a constant amplitude signal, even though the amplitude of input signal may be varying.
- **FM Demodulators :** FM demodulators, change the frequency deviation of the incoming carrier into an AF amplitude variation (identical to the one that originally caused the frequency variation).

Q.28 How are FM receivers different from AM receiver ?

[SPPU : Dec.-07]

Ans. : FM receivers differs from A.M. receiver with respect to following points :

- A.M. receiver operates in MW and SW bands, while F.M. receiver operates at much higher frequencies viz. 88 MHz to 108 MHz.
- Limiter and de-emphasis circuits are required only in FM receiver.
- The technique of demodulating F.M. signal is different from detection of A.M. signal.
- F.M. receiver uses different methods of obtaining AGC.

Q.29 How we recover the FM signal using Phase Locked Loop (PLL). ☞ [SPPU : Dec.-10,13,17, May-14, Marks 6]

Ans. :

- FM demodulation can be accomplished quite simply with a Phase-Locked Loop (PLL).
- Basically, the phase-locked loop consists of a multiplier, a loop-filter, and a Voltage-Controlled Oscillator (VCO) connected together in the form of feedback system, as shown in the Fig. Q.29.1.

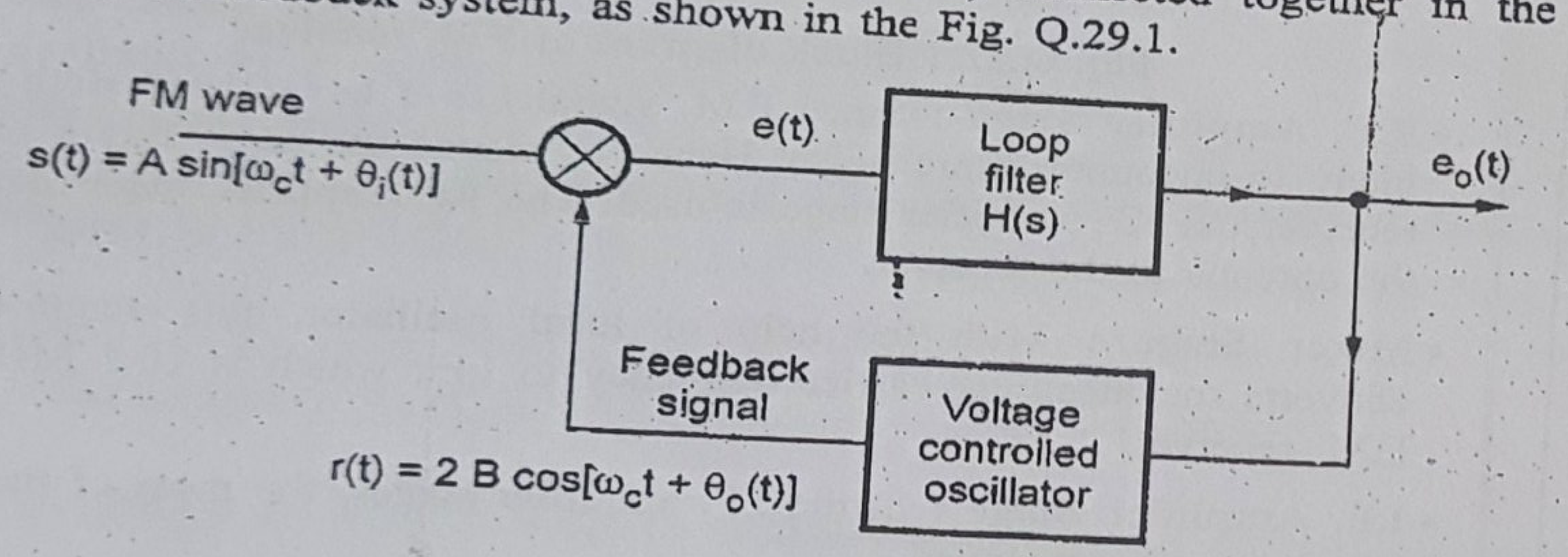

Fig. Q.29.1 Phase-locked loop

- The VCO is a sinusoidal generator whose frequency is determined by a voltage applied to it from an external source. The incoming FM signal is used to control the frequency of the VCO.
- As the incoming frequency varies, the PLL generates a control voltage or error voltage $e_o(t)$ to change the VCO frequency, which follows that of the incoming signal.
- This control voltage varies at the same rate as the frequency of the incoming signal and it is proportional to the modulating signal.

Q.30 State different methods of FM detection. ☞ [SPPU : Dec.-19, Marks 2]

Ans. : Different methods of FM detection are :

1. Slope detection
2. Balanced slope detection
3. Phase shift discriminator
4. Ratio detector
5. Quadrature detector.

Q.31 With the help of neat phasor diagram explain balanced slope detector in FM. ☞ [SPPU : May-18, Dec.-22, Marks 6]

Ans. : • Fig. Q.31.1 shows the balanced slope detector. It consists of two slope detector circuits. As shown in the Fig. Q.31.1 these two circuits are connected back to back, to the opposite ends of a center-tapped transformer.

- Due to this connection the input fed to these circuits is 180° out of phase. The primary circuit is tuned to the IF (f_c). The top secondary circuit (T_1) is tuned for the highest input frequency ($f_c + \delta f$) whereas the bottom secondary circuit (T_2) is tuned for the lowest input frequency ($f_c - \delta f$).
- The secondary tuned circuits are connected to a diode detector with an RC load. Therefore, the total output of the balanced slope detector is the sum of the individual outputs of the diode detectors.

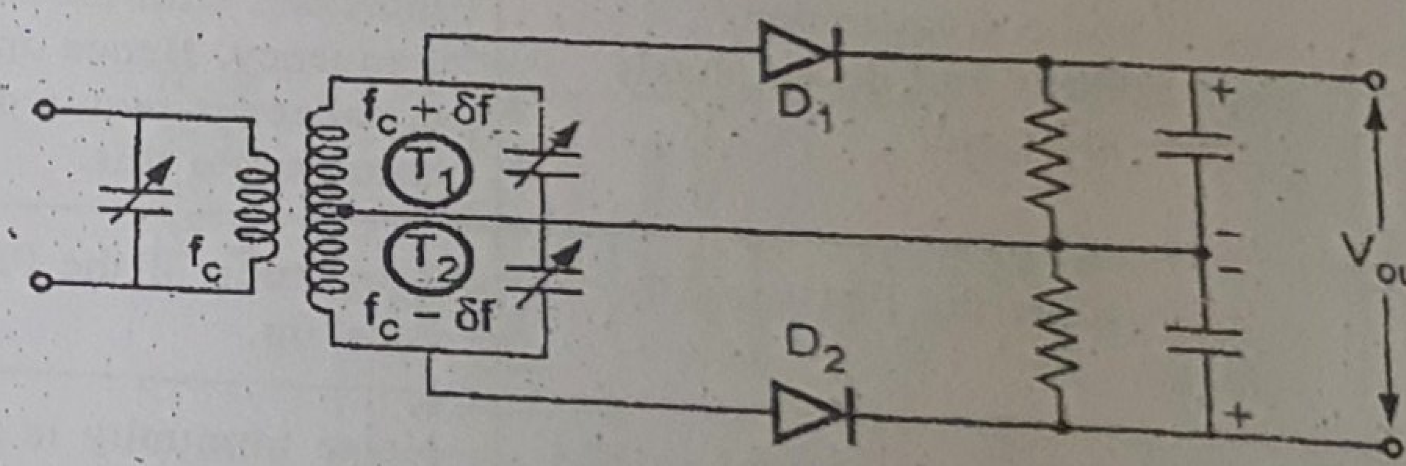

Fig. Q.31.1 Balanced slope detector

- Fig. Q.31.2 shows the characteristic of the balanced slope detector. When the input frequency is equal to f_c, the voltage applied to the two diodes will be identical. The d.c. output voltages will also be identical, and thus the detector output will be zero.

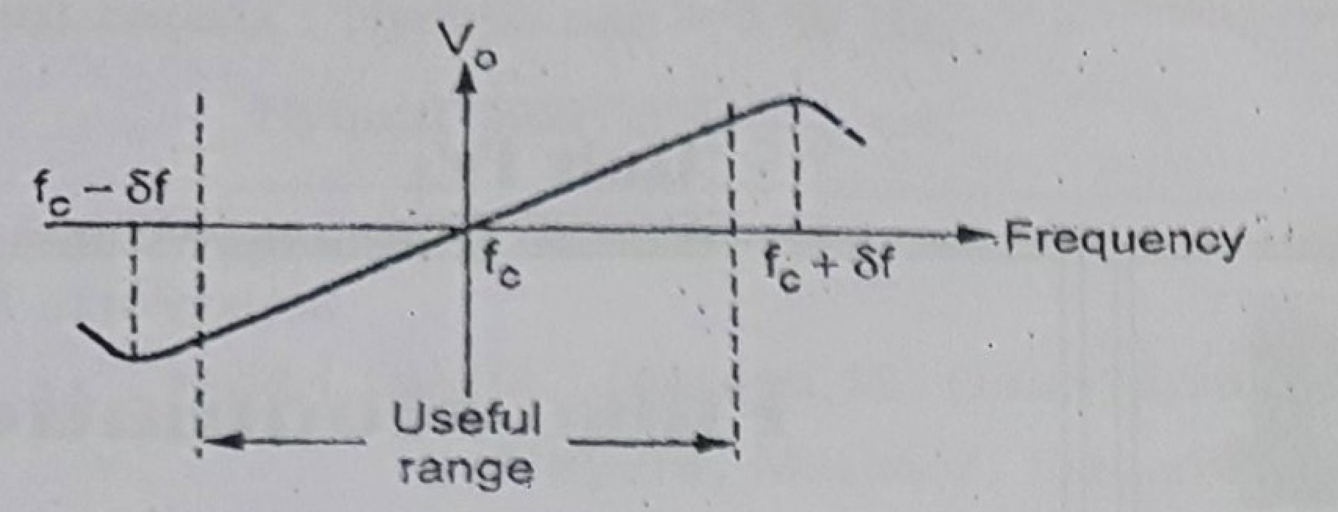

Fig. Q.31.2 Balanced slope detector characteristic

- Now consider the instantaneous frequency to be equal to $f_c + \delta f$. Since T_1 is tuned to this frequency, the output of D_1 will be quite large. On the other hand, the output of D_2 will be very small, since the frequency $f_c + \delta f$ is quite a long way from $f_c - \delta f$.
- Similarly, when the input frequency is equal to $f_c - \delta f$, the output of D_2 will be a large negative voltage, and that of D_1 a small positive voltage. When the instantaneous frequency is between these two extremes, the output will have some intermediate value. It will then be positive or negative, depending on which side of f_c the input frequency happens to lie. The required S-shaped characteristics is thus obtained.
- The balanced slope detector circuit is considerably more efficient than the previous one. There are now 3 different frequencies to which the various tuned circuits of the transformer must be adjusted, therefore for this circuit the tuned circuit alignment is even more difficult.
- Amplitude limiting is still not provided and the linearity, although better than that of the single slope detector, is still not good enough.

Q.32 Discuss principle working of FM detection. Briefly explain any one FM detector method. ☞ **[SPPU : May-17, Marks 6]**

Ans. : • The primary and secondary circuits are tuned to carrier frequency. C_3 is the coupling capacitor which is almost short at signal frequency, while L_3 is RFC. This part of the circuit develops two output voltages, applied to the two diodes, D_1 and D_2. These voltages are frequency dependent.

- The capacitor C_9 is selected to be large in value, say, typically 10 µF. It charges to the peak value of voltage across L_2, and because of large time constant, it holds this voltage.

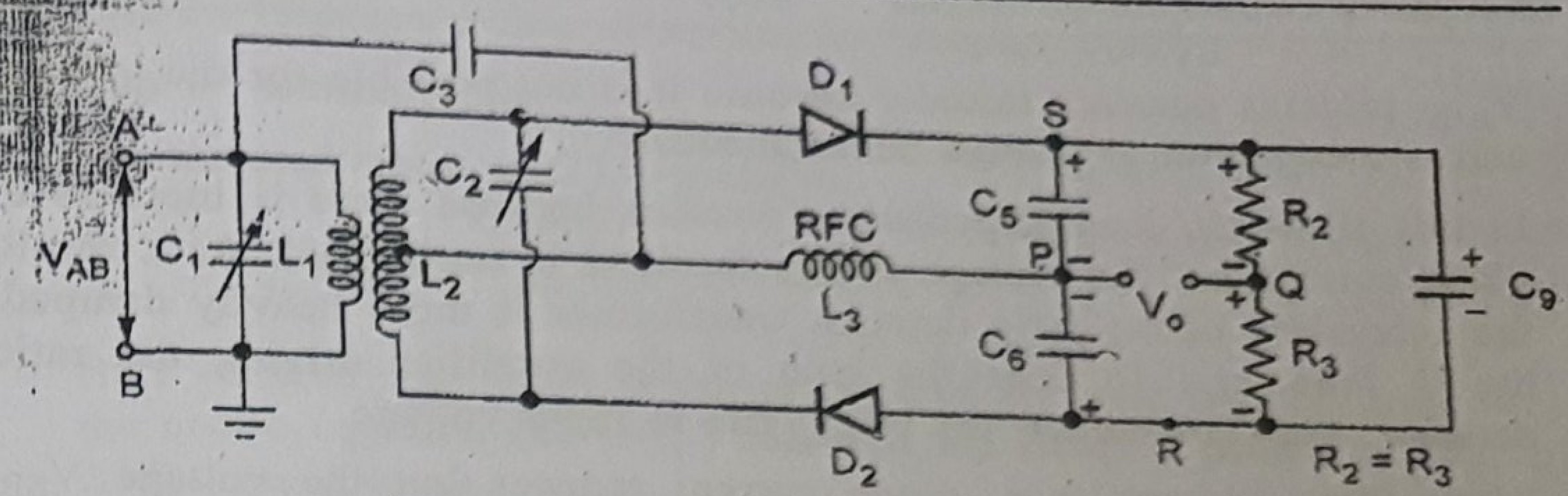

Fig. Q.32.1 Ratio detector

- Since the two resistances R_2 and R_3 are chosen to be equal, voltage across each of them is one-half the voltage across C_9; i.e. voltage between Q and R is half the voltage across C_9, which is constant for all frequencies.
- At the carrier frequency, since both diodes conduct equally, the voltages across C_5 and C_6 are equal (at one-half the voltage across C_9). Also, since $R_2 = R_3$, their voltage drops are equal. This makes point P and Q at the same potential, and the output voltage is zero.
- As the input frequency changes, so that D_2 conducts more than D_1; E_{c6} increases but E_{c5} decreases, but in such a way that the sum of these voltages is always constant. The output voltage available is positive.
- Conversely, when D_1 conducts more heavily than D_2 ; V_{C5} exceeds V_{C6}, but their sum remains constant. This makes point (P) negative with respect to (Q). Thus the output voltage swings negative. Thus, in the circuit, the sum of the voltages V_{C5} and V_{C6} remains constant, but their ratio changes, depending on signal frequency. Hence the circuit is called ratio detector.

Q.33 Justify "ratio detector acts as detector as well as limiter" ? ☞ **[SPPU : May-17,19, Marks 6)**

Ans. : • In ratio detector circuit (shown in Fig. Q.33.1), the value of capacitor C_9 is large. Because of large value it takes a long time to charge and discharge. Due to this noise pulse or minor variations in amplitude are totally eliminated and constant voltage is obtained.

- When input voltage V_{AB} tries to rise, extra diode current flows, but this excess current flows into the capacitor C_9, charging it. The voltage

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$V_{a'b'}$ remains constant initially because it is not possible for the voltage across a capacitor to change instantaneously.

- In this situation, load impedance decreases because there is increase in diode current but the voltage across the load is unchanged. As a result, the secondary of the ratio detector transformer is more heavily damped, the Q falls, and so does the gain of the amplifier driving the ratio detector. This counteracts the initial rise in input voltage.
- When input voltage falls, diode current reduces but the voltage V_{SR} remains constant. This results increase in load impedance. Increase in load impedance reduces damping and increase the gain of the driving amplifier. This time it counteracts an initial fall in the input voltage. Thus, we can say that ratio detector provides what is known as **diode variable damping** of its tuned circuit and maintains constant output voltage despite changes in the amplitude of the input.
- In this way ratio detector provides amplitude limitation and we can say that ratio detector acts as well as limiter.

END...

Unit IV

4 Pulse Modulation

4.1 : Sampling Theorem and Nyquist Criteria

Important Points to Remember

- Sampling theorem provides minimum sampling frequency so that the original signal can be sampled and recovered back from its samples.
- Sampling theorem provides the effects of under sampling and ways to avoid them.
- Sampling theorem provides the filtering requirements to reduce distortion.
- Time limited signal $x(t) = 0$ for $t \leq t_1$ and $t > t_2$
- Criteria for sampling Theorem $f_s \geq 2W$

  For sampling theorem in time domain.
- $X_\delta(f) = \dots f_s X(f-2f_s) + f_s X(f-f_s) + f_s X(f) + f_s X(f+f_s) + f_s X(f+2f_s) + \dots$
- $$x(t) = \text{IFT}\left\{\frac{1}{f_s}\sum_{n=-\infty}^{\infty} x(nT_s)\, e^{-j2\pi f n T_s}\right\} \quad \dots(4.1)$$
- Reconstruction of x(t)

  $x(t) = \dots + x(-2T_s)\,\text{sinc}(2Wt+2) + x(-T_s)\,\text{sinc}(2Wt+1) + x(0)\,\text{sinc}(2Wt) + x(T_s)\,\text{sinc}(2Wt-1) + \dots\dots$
- To avoide aliasing

  i) Sampling rate $f_s \geq 2W$

  ii) Strictly band limit the signal to "W"

- Nyquist criteria : Nyquist rate = 2 W Hz ...(4.2)

  Nyquist interval = $\frac{1}{2W}$ sec. ...(4.3)

Q.1 State and prove sampling theorem with suitable waveforms and mathematical expression.

☞ [SPPU : May-15,18, Dec.-15,18, Marks 6, May-16, Marks 7, Dec.-19, Marks 3]

OR

State sampling theorem. **☞ [SPPU : May-14, Marks 3, Dec.-11,12, Marks 2, May-17, Marks 7]**

OR

State and prove Sampling theorem in time domain.

☞ [SPPU : Dec.-10,13, June-22, Marks 6]

OR

Explain sampling theorem showing the help of sampled signal in time domain. Also explain how to reconstruct the signal.

☞ [SPPU : May-12, 10, Marks 8]

OR

State and prove low pass sampling Thorem for band limited signal.

☞ [SPPU : May-10,15, Dec.-15, Marks 6, Dec.-14, Marks 2]

Ans. : Sampling theorem for Low Pass (LP) signals.

Statement of sampling theorem

1) *A band limited signal of finite energy, which has no frequency components higher than W hertz, is completely described by specifying the values of the signal at instants of time separated by* $\frac{1}{2W}$ *seconds and*

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

Proof of sampling theorem

Part I : Representation of $x(t)$ in its samples $x(nT_s)$

Step 1 : Define $x_\delta(t)$

The sampled signal $x_\delta(t)$ is given as,

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(t)\delta(t-nT_s) \quad \text{... (Q.1.1)}$$

Here observe that $x_\delta(t)$ is the product of $x(t)$ and impulse train $\delta(t)$. In the above equation $\delta(t-nT_s)$ indicates the samples placed at $\pm T_s$, $\pm 2T_s$, $\pm 3T_s$ ... and so on.

Step 2 : FT of $x_\delta(t)$ i.e. $X_\delta(f)$

The Fourier transform of $X_\delta(t)$ is given as,

$$X_\delta(f) = \cdots f_s X(f-2f_s) + f_s X(f-f_s) + f_s X(f) + f_s X(f-f_s) + f_s X(f-2f_s) + \cdots$$

Step 3 : Relation between $X(f)$ and $X_\delta(f)$

Important assumption : Let us assume that $f_s = 2W$, then

$$X_\delta(f) = f_s X(f) \text{ for } -W \le f \le W \text{ and } f_s = 2W$$

or

$$X(f) = \frac{1}{f_s} X_\delta(f) \quad \text{... (Q.1.2)}$$

Fig. Q.1.1 shows the spetrums of $X(f)$ and $X_\delta(f)$

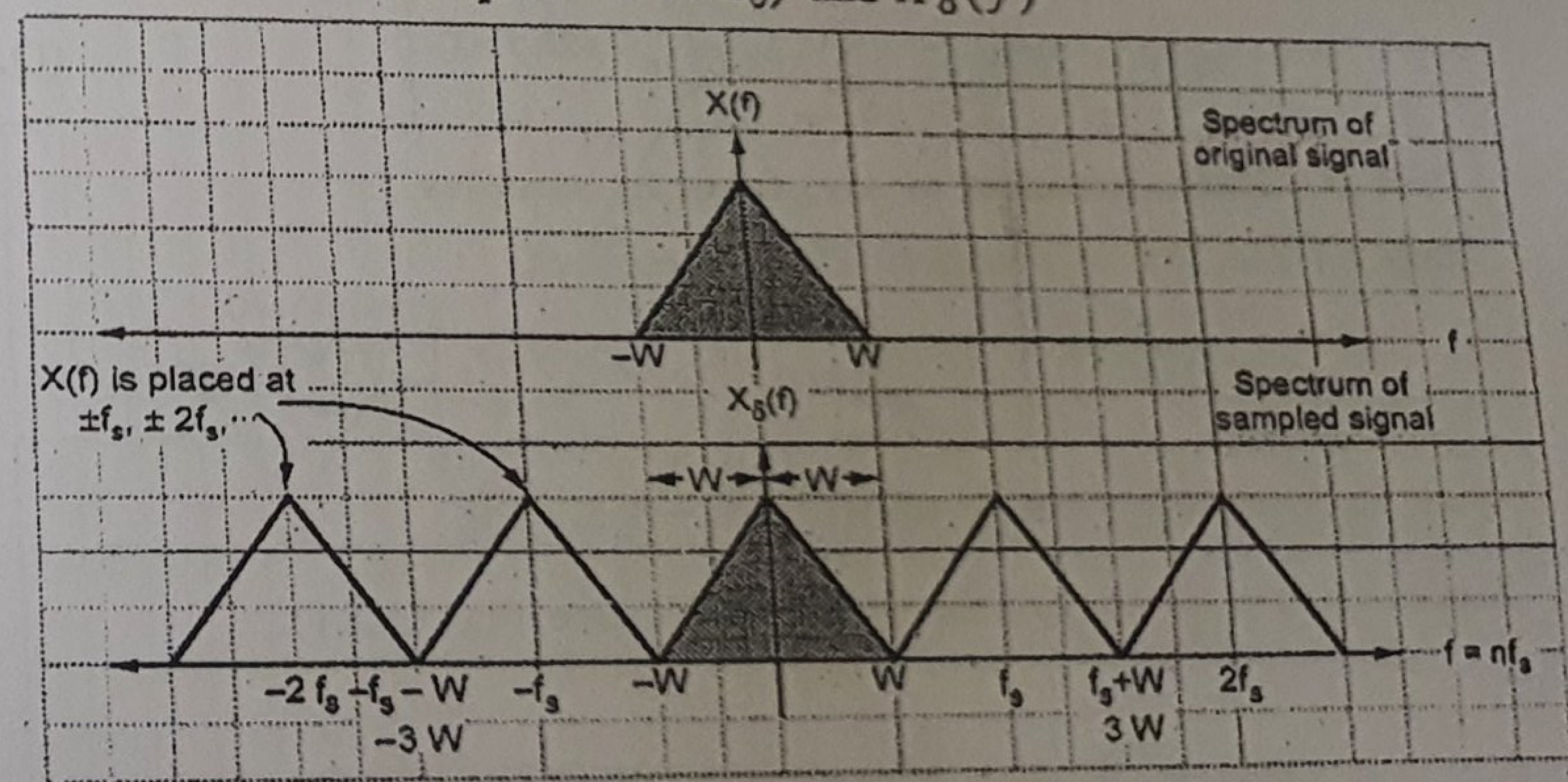


Fig. Q.1.1 Spectrum of original signal and sampled signal (f_s = 2W)

Step 4 : Relation between $x(t)$ and $x(nT_s)$

DTFT is, $$X(\Omega) = \sum_{n=-\infty}^{\infty} x(n) e^{-j\Omega n}$$

$$\therefore \quad X(f) = \sum_{n=-\infty}^{\infty} x(n) e^{-j2\pi f n}$$

$$\therefore \quad X_\delta(f) = \sum_{n=-\infty}^{\infty} x(n) e^{-j2\pi \frac{f}{f_s} n} \qquad \ldots (Q.1.3)$$

In above equation 'f' is frequency of CT signal. And $\frac{f}{f_s}$ = Frequency of DT signal in equation (Q.1.3). Since $x(n) = x(nT_s)$, i.e. samples of $x(t)$, then

$$X_\delta(f) = \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \quad \text{since } \frac{1}{f_s} = T_s$$

Putting above expression in equation (Q.1.2),

$$X(f) = \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s}$$

Inverse Fourier Transform (IFT) of above equation gives $x(t)$ i.e.,

$$x(t) = IFT\left\{ \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \right\} \qquad \ldots (Q.1.4)$$

Part II : Reconstruction of $x(t)$ from its samples

Step 1 : The IFT of equation (Q.1.4) becomes,

$$x(t) = \int_{-\infty}^{\infty} \left\{ \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \right\} e^{j2\pi f t}\, df$$

Here the integration can be taken from $-W \le f \le W$.

Since $X(f) = \frac{1}{f_s} X_\delta(f)$ for $-W \le f \le W$. (See Fig. Q.2.1).

$$\therefore \quad x(t) = \int_{-W}^{W} \frac{1}{f_s} \sum_{n=-\infty}^{\infty} x(nT_s) e^{-j2\pi f n T_s} \cdot e^{j2\pi f t}\, df$$

Interchanging the order of summation and integration,

$$x(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \int_{-W}^{W} e^{j2\pi f (t - nT_s)}\, df$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \cdot \left[\frac{e^{j2\pi f (t-nT_s)}}{j2\pi(t-nT_s)} \right]_{-W}^{W}$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \left\{ \frac{e^{j2\pi W (t-nT_s)} - e^{-j2\pi W (t-nT_s)}}{j2\pi(t-nT_s)} \right\}$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \frac{1}{f_s} \cdot \frac{\sin 2\pi W (t-nT_s)}{\pi(t-nT_s)}$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \frac{\sin \pi (2Wt - 2W nT_s)}{\pi (f_s t - f_s nT_s)}$$

Here $f_s = 2\,W$, hence $T_s = \frac{1}{f_s} = \frac{1}{2W}$. Simplifying above equation,

$$x(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \frac{\sin \pi (2Wt - n)}{\pi (2Wt - n)}$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \operatorname{sinc}(2Wt - n) \qquad \text{since } \frac{\sin \pi\theta}{\pi\theta} = \operatorname{sinc} \theta \qquad \ldots (Q.1.5)$$

Step 2 : Let us interpret the above equation. Expanding we get,

$$x(t) = \cdots + x(-2T_s)\operatorname{sinc}(2Wt+2) + x(-T_s)\operatorname{sinc}(2Wt+1) + x(0)\operatorname{sinc}(2Wt) + x(T_s)\operatorname{sinc}(2Wt-1) + \cdots$$

Step 3 : Reconstruction of $x(t)$ by lowpass filter

When the interpolated signal of equation (Q.1.5) is passed through the lowpass filter of bandwidth $-W \le f \le W$, then the reconstructed waveform shown in below Fig. Q.1.2 (b) is obtained. The individual sinc functions are interpolated to get smooth $x(t)$.

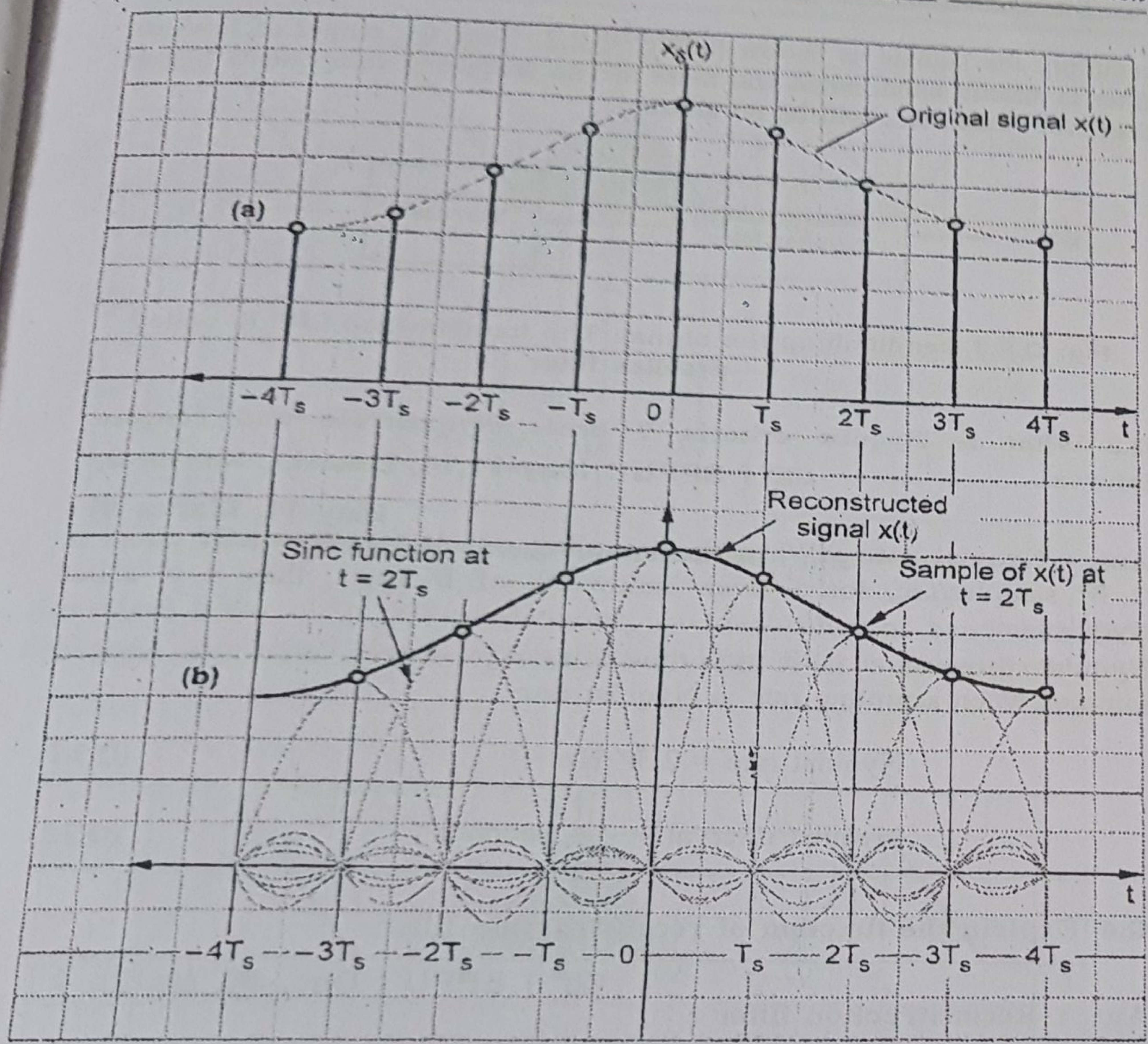


Fig. Q.1.2

(a) Sampled version of signal $x(t)$

(b) Reconstruction of $x(t)$ from its samples

Q.2 What are effects of under sampling ? Explain with help of frequency spectrum.

☞ [SPPU : Dec-14, Marks 4, Dec.-18, Marks 2]

OR

Draw the spectrum showing alising and guard band.

☞ [SPPU : May-14, Marks 7, Dec-11, Marks 6]

OR

Explain aliasing and different ways to avoid aliasing.

☞ [SPPU : May-10, Dec-10, 15,16, Marks 3, May-13, Marks 8, May-11, Marks 4, May-15, Marks 2, Dec.-22, Marks 6, June-22, Dec.-16, 17, Marks 7]

OR

State explain the sampling theorem in detail when $f_s > 2f_m$, $f_s = 2f_m$, $f_s < 2f_m$.

☞ [SPPU : Dec.-22, Marks 5]

Ans. : Effects of undersampling (Aliasing)

While proving sampling theorem we considered that $f_s = 2W$. Consider the case of $f_s < 2W$. Then the spectrum of $X_\delta(f)$ shown in Fig. Q.2.1 will be modified as follows :

i) The spectrums located at $X(f), X(f-f_s), X(f-2f_s), \ldots$ overlap on each other.

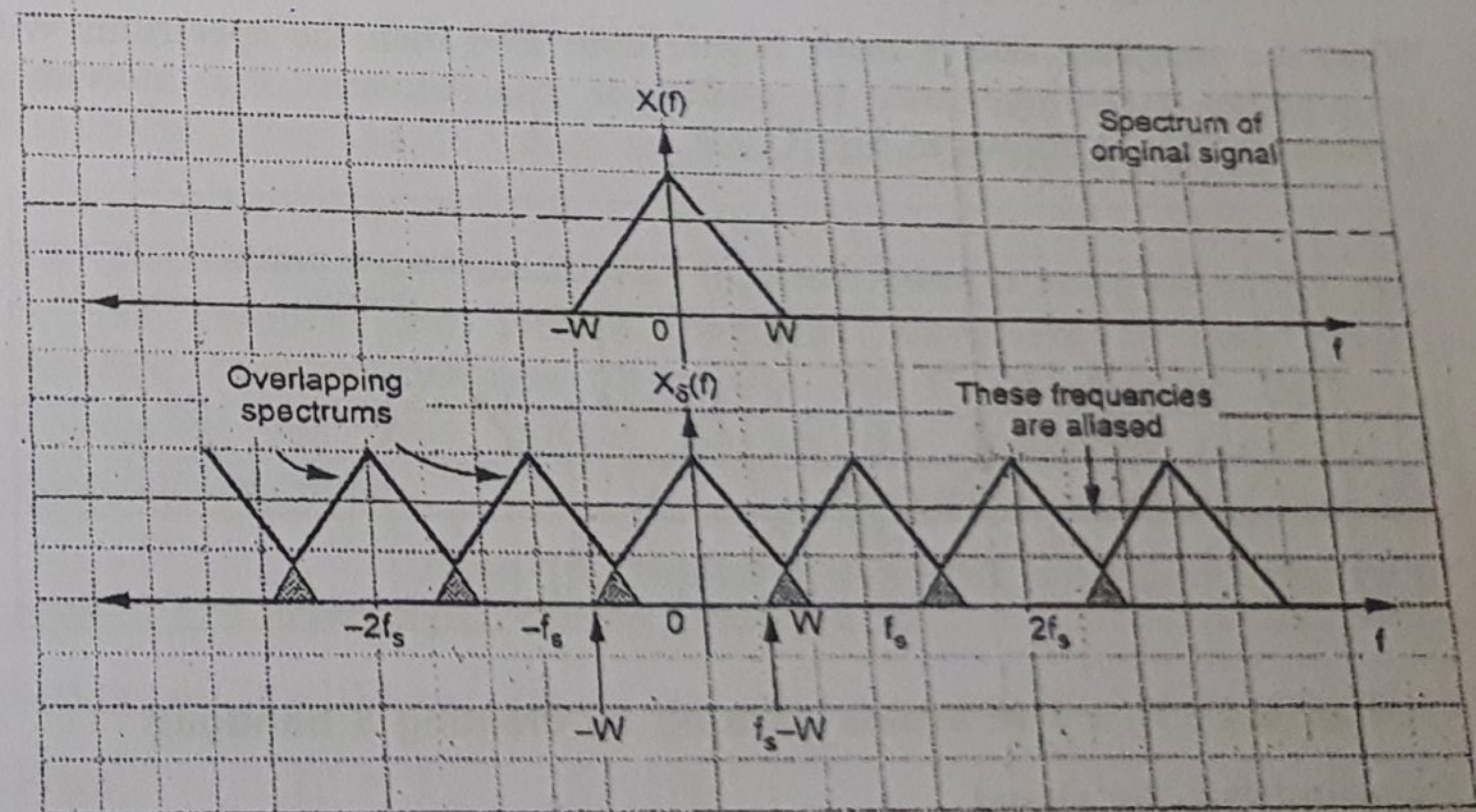


Fig. Q.2.1 Effects of undersampling or aliasing

iii) Consider the spectrums of $X(f)$ and $X(f-f_s)$ shown in above figure. The frequencies from $(f_s - W)$ to W are overlapping in these spectrums.

iii) The high frequencies near 'W' in $X(f-f_s)$ overlap with low frequencies $(f_s - W)$ in $X(f)$.

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

Effects of aliasing : i) Since high and low frequencies interfere with each other, distortion is generated.

ii) The data is lost and it cannot be recovered.

Different ways to avoid aliasing

Aliasing can be avoided by two methods :

i) **Sampling rate $f_s \geq 2W$.**

ii) **Strictly bandlimit the signal to 'W'.**

i) **Sampling rate $f_s \geq 2W$**

When the sampling rate is made higher than 2W, then the spectrums will not overlap and there will be sufficient gap between the individual spectrums. This is shown in Fig. Q.2.2.

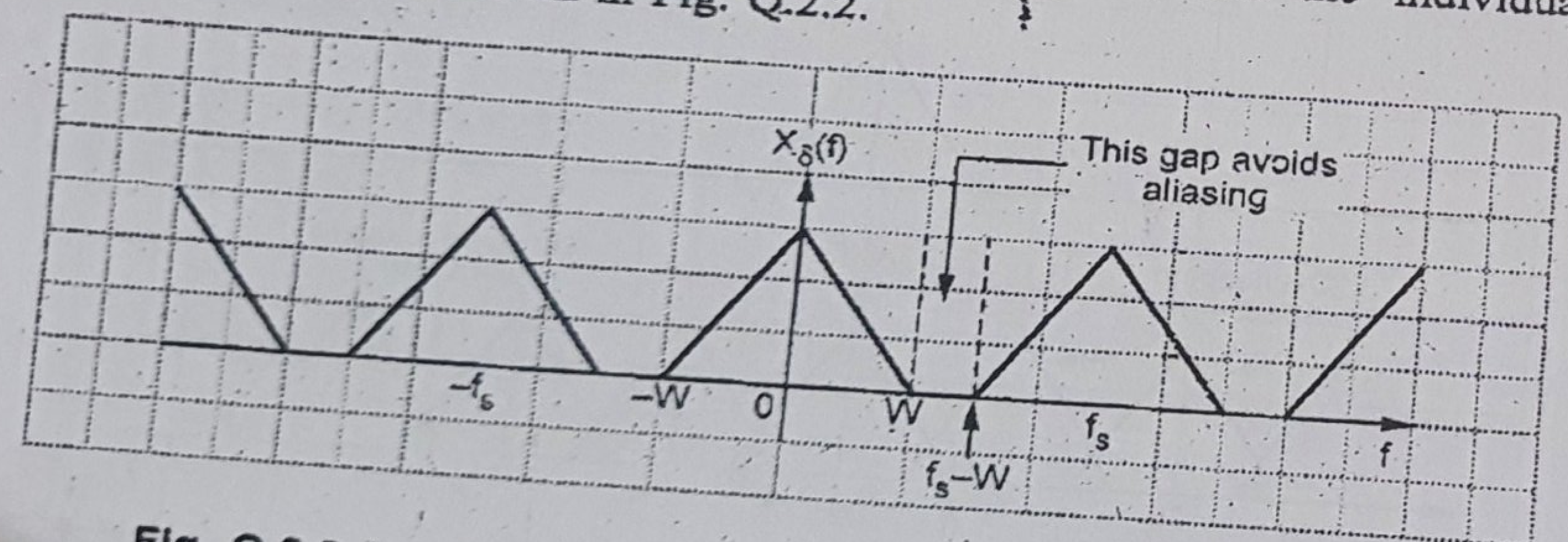


Fig. Q.2.2 $f_s \geq 2W$ avoids aliasing by creating a bandgap

ii) Bandlimiting the signal

The sampling rate is, $f_s = 2W$. Ideally speaking there should be no aliasing. But there can be few components higher than 2W. These components create aliasing. Hence a low pass filter is used before

sampling the signals as shown in Fig. Q.2.3. Thus the output of lowpass filter is strictly bandlimited and there are no frequency components higher than 'W'. Then there will be no aliasing.

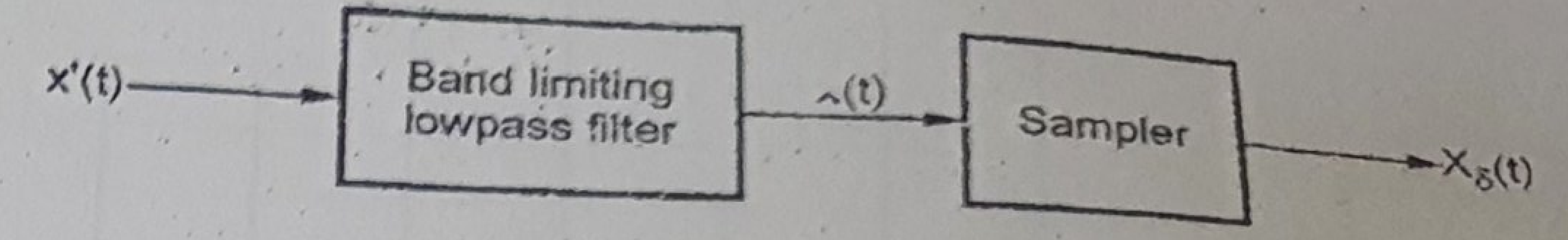


Fig. Q.2.3 Bandlimiting the signal. The bandlimiting LPF is called prealias filter

Q.3 What is Nyquist criteria ? State Nyquistrate and Nyquist interval.

[SPPU : May-11,14, Dec.-11, Marks 3, May-11, Marks 4]

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

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

$$\text{Nyquist rate} = 2\,W\ \text{Hz} \quad \text{... (Q.3.1)}$$

$$\text{Nyquist interval} = \frac{1}{2W}\ \text{seconds} \quad \text{... (Q.3.2)}$$

Q.4 Explain the function of reconstruction filler.

[SPPU : Dec.-14, Marks 4]

Ans. : Reconstruction filter :

Definition : The reconstructed signal is the succession of sinc pulses weighted by $x(nT_s)$. These pulses are interpolated with the help of a lowpass filter. It is also called *reconstruction* filter or *interpolation* filter.

Ideal filter : Fig. Q.4.1 shows the spectrum of sampled signal and frequency response of required filter. When the sampling frequency is exactly 2W, then the spectrums just touch each other as shown in Fig. Q.4.1. The spectrum of original signal, $X(f)$ can be filtered by an ideal filter having passband from $-W \leq f \leq W$.

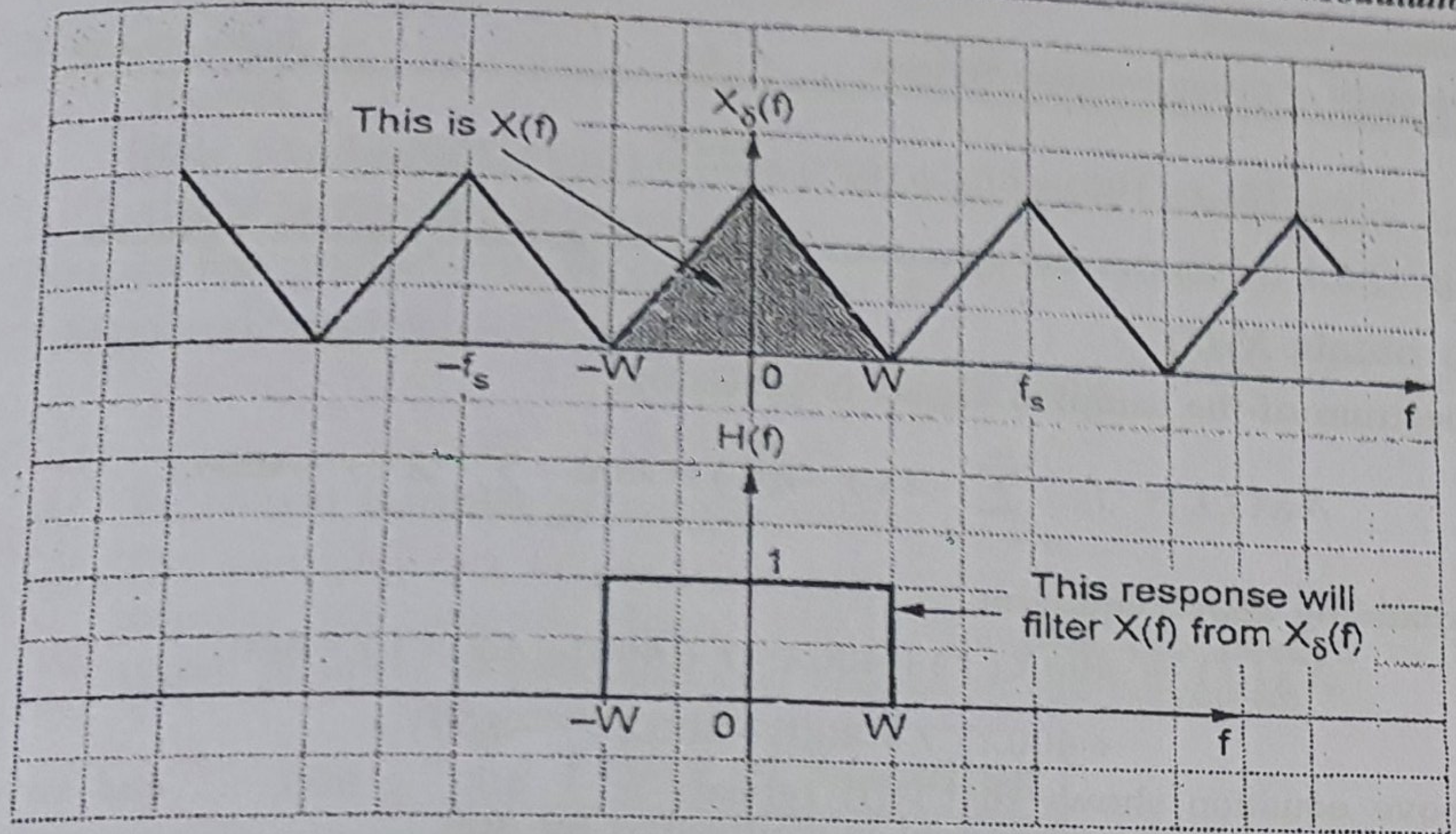


Fig. Q.4.1 Ideal reconstruction filter

Non-ideal filter : As discussed above, an ideal filter of bandwidth 'W' filters out an original signal. But practically ideal filter is not realizable. It

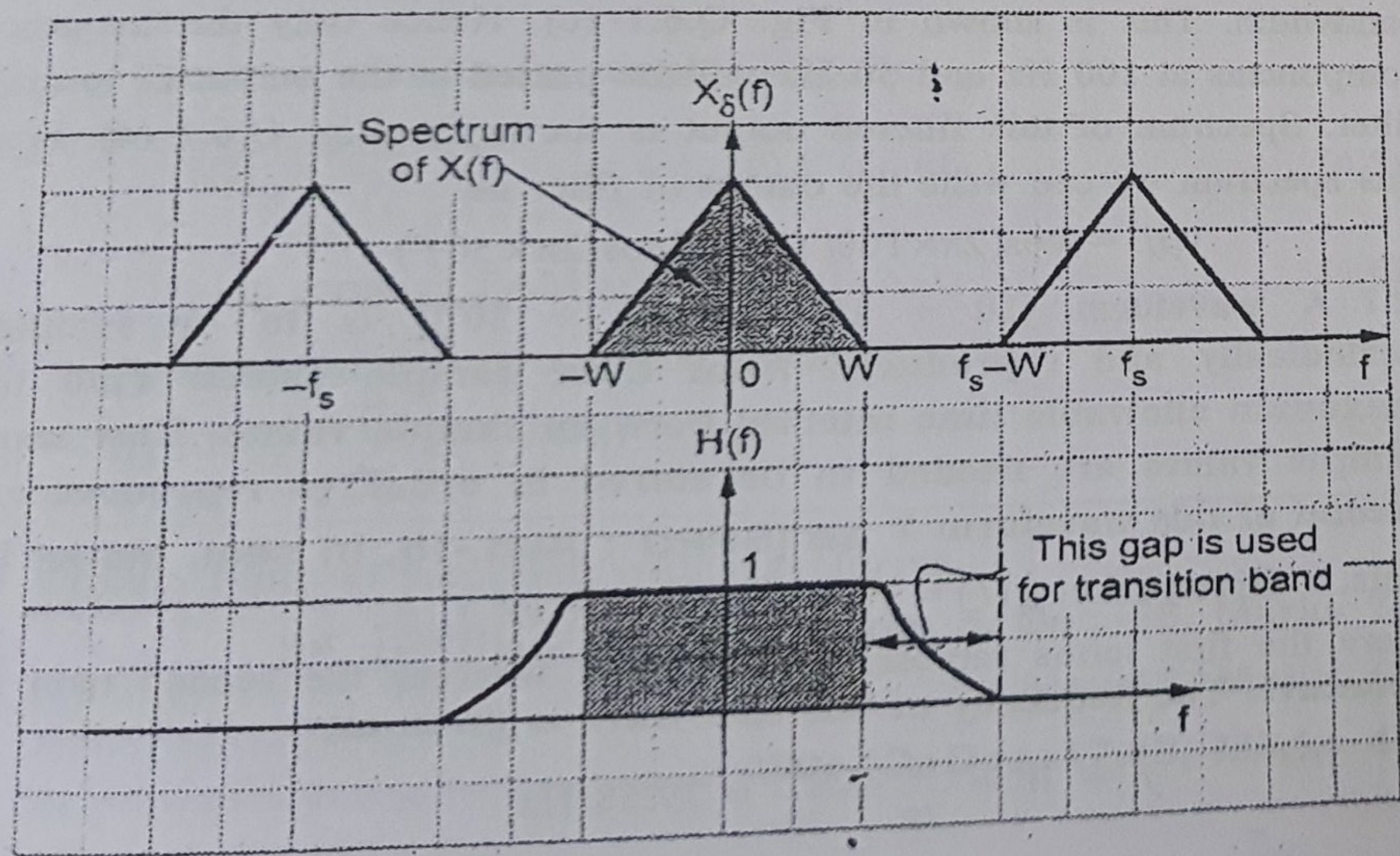


Fig. Q.4.2 Practical reconstruction filter

requires some transition band. Hence f_s must be greater than $2W$. It creates the gap between adjacent spectrums of $X_\delta(f)$. This gap can be used for the transition band of the reconstruction filter. The spectrum $X(f)$ is then properly filtered out from $X_\delta(f)$. Hence the sampling frequency must be greater than '2 W' to ensure sufficient gap for transition band.

Q.5 Find a signal g(t) which is band limited to 1Hz and its samples are $g(0) = 1$, $g(\pm 0.5) = g(\pm 1) = g(\pm 1.5) = \text{-----} = 0$

☞[SPPU : Aug.-17, Marks 4]

Ans. : Here W = 1 Hz. Hence Nyquist sampling rate will be

$$f_s = 2\,W = 2 \times 1 = 2 \text{ Hz}$$

$$\therefore \quad T_s = \frac{1}{f_s} = \frac{1}{2} = 0.5 \text{ sec.}$$

Sampled signal, $g(n) = g(t)|_{t=nT_s=0.5n} = g(0.5n)$

$\therefore$ for $n = 0,\ g(0) = 1$

$n = \pm 1,\ g(\pm 0.5) = 0$

$n = \pm 2,\ g(\pm 1) = 0$

$n = \pm 3,\ g(\pm 1.5) = 0$ ... and so on.

Thus the signal has non-zero value only at n = 0 and its amplitude is 1. Rest of all other sample values are zero. Hence the signal $g(t)$ is,

$$g(t) = 1 \text{ at } t = 0.$$

Q.6 The signal $x(t) = \cos 200\pi t + 0.25 \cos 700\pi t$ is sampled at the rate of 400 samples per second. Sampled waveform is then passed through an ideal lowpass filter with 200 Hz bandwidth. Write an expression for filter output. Sketch the frequency spectrum of sampled waveform.

☞[SPPU : May-96; May-07, June-10, Marks 8, May 15, end sem, Marks 8]

Ans. : The given signal can be expressed as,

$$x(t) = \cos(2\pi\times 100t) + 0.25\cos(2\pi\times 350t) = A_1\cos(2\pi f_1 t) + A_2\cos(2\pi f_2 t)$$

Here $A_1 = 1, \quad f_1 = 100$

$A_2 = 0.25, \quad f_2 = 350$

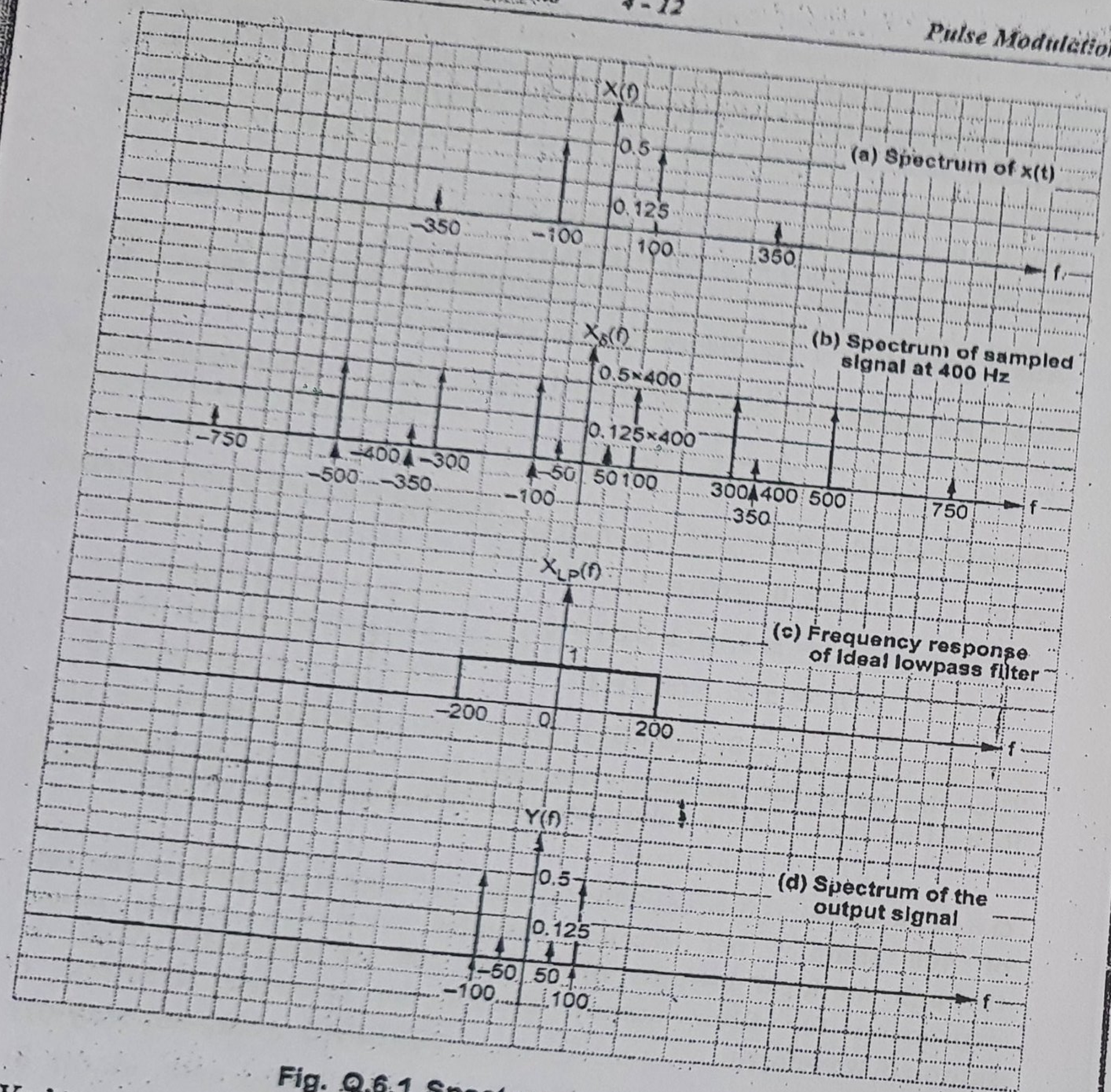


Fig. Q.6.1 Spectrums of signals

We know that spectrum of cosine signal is given as

$$FT[\cos(2\pi f_c t)] = \frac{1}{2}[\delta(f-f_c)+\delta(f+f_c)]$$

Hence spectrum of given signal becomes,

$$X(f) = \frac{A_1}{2}[\delta(f-f_1)+\delta(f+f_1)] + \frac{A_2}{2}[\delta(f-f_2)+\delta(f+f_2)]$$

$$= \frac{1}{2}[\delta(f-100)+\delta(f+100)] + \frac{0.25}{2}[\delta(f-350)+\delta(f+350)]$$

Fig. Q.6.1 (a) shows the spectrum of $x(t)$, as given by above equation.

To obtain $X_\delta(f)$

Spectrum of the sampled signal is given by,

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s) = 400 \sum_{n=-\infty}^{\infty} X(f-400n)$$

Expanding above equation,

$$X_\delta(f) = 400X(f)+400X(f+400)+400X(f-400)$$
$$+400X(f+800)+400X(f-800)+\ldots$$

Above equation shows that $X(f)$ repeats at ± 400, ± 800, ... and so on. Amplitude of each component is multiplied by 400.

Fig. Q.6.1 (b) shows the spectrum $X_\delta(f)$.

To determine output

The sampled signal is passed through ideal lowpass filter of 200 Hz bandwidth. This is shown in Fig. Q.6.1 (c). Hence only the frequency components at 100 Hz and 50 Hz will be passed at the output of lowpass filter. Spectrum of this filtered output is shown in Fig. Q.6.1 (d). From this spectrum we can write the output of filter as,

$$y(t) = \cos(2\pi\times 100t)+0.25\cos(2\pi\times 50 f)$$

Q.7 A waveform [10 + 10 sin(500t + 30°)] is to be sampled periodically and reproduced from these sample values. Find the maximum allowable time interval between sample values. How many sample values are needed to be stored in order to reproduce one second of this waveform ? ☞**[SPPU : Aug.-15, In sem, Marks 5]**

Ans. : Given : Let $x(t) = 10+10\sin(500t+30°)$

Here the first terms represents a dc shift whereas the second term is a sinewave. The frequency of this sine wave is given as,

$$f = W = \frac{\omega}{2\pi} = \frac{500}{2\pi} = 79.58 \text{ Hz}$$

i) To obtain maximum allowable time interval between the sample :

The maximum allowable time interval takes place when the signal is sampled at Nyquist rate.

$\therefore$ Nyquist rate $= 2W = 2 \times 79.58 = 159.16$ Hz

$\therefore$ Maximum time interval $= \frac{1}{\text{Nyquist rate}} = \frac{1}{159.16} = 6.28 \times 10^{-3}$ sec

$= 6.28$ msec

ii) To obtain number of sample values :

The sampling rate as per (i) is 159.16 Hz. This means 159.16 i.e. samples per second. Hence 159.16 samples are needed to be stored in order to produce one second of the waveform.

4.2 : Types of Sampling

Important Points to Remember

1) Spectrum of ideally sampled signal :

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \quad \dots (4.4)$$

2) Spectrum of naturally sampled signal :

$$S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} \text{sinc}(nf_s \tau) X(f - nf_s) \quad \dots (4.5)$$

3) Flat-top sampled PAM $s(t) = x_\delta(t) * h(t)$ ... (4.6)

4) Spectrum of flat-top sampled signal :

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) H(f) \quad \dots (4.7)$$

Q.8 Explain types of sampling with waveforms.

☞ [SPPU : Dec.-18, Marks 5, Dec.-16, Marks 7, May-15, Marks 6, May-19, Dec.-17,19, Marks 4]

OR Explain in detail types of sampling.

☞ [SPPU : Dec-13, Marks 8, Dec.-10, Marks 3]

OR Why is ideal sampling not used for practical applications ? Draw circuit diagram for flat top sampling method and explain with waveforms.

☞ [SPPU : May16, Dec-11,22, Marks 6, Dac.-15, Marks 3]

OR Draw and explain a circuit for flat top sampling.

☞ [SPPU : Dec.-10, 11, 12, 15, Marks 3, Dec.-14, Marks 8]

Ans.: Types of sampling

Depending upon the shape of the sampled pulse there are three types of sampling :

i) Ideally or instantaneously sampled PAM.

ii) Naturally sampled PAM.

iii) Flat top sampled PAM.

1. Ideal sampling or Instantaneous sampling or Impulse sampling

Basic principle : Ideal sampling is same as instantaneous sampling. Fig. Q.8.1 (a) shows the switching sampler. If closing time 'τ' of the switch approaches zero the output $x_\delta(t)$ gives only instantaneous value. The waveforms are shown in Fig. Q.8.1 (b). Since the width of the pulse approaches zero, the instantaneous sampling gives train of impulses in $x_\delta(t)$. The area of each impulse in the sampled version is equal to instantaneous value of input signal $x(t)$.

Explanation : We know that the train of impulses can be represented mathematically as,

$$s_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad \dots (Q.8.1)$$

- This is called sampling function. The sampled signal $x_\delta(t)$ is given by multiplication of $x(t)$ and $s_\delta(t)$.

Therefore, $x_\delta(t) = x(t) s_\delta(t)$

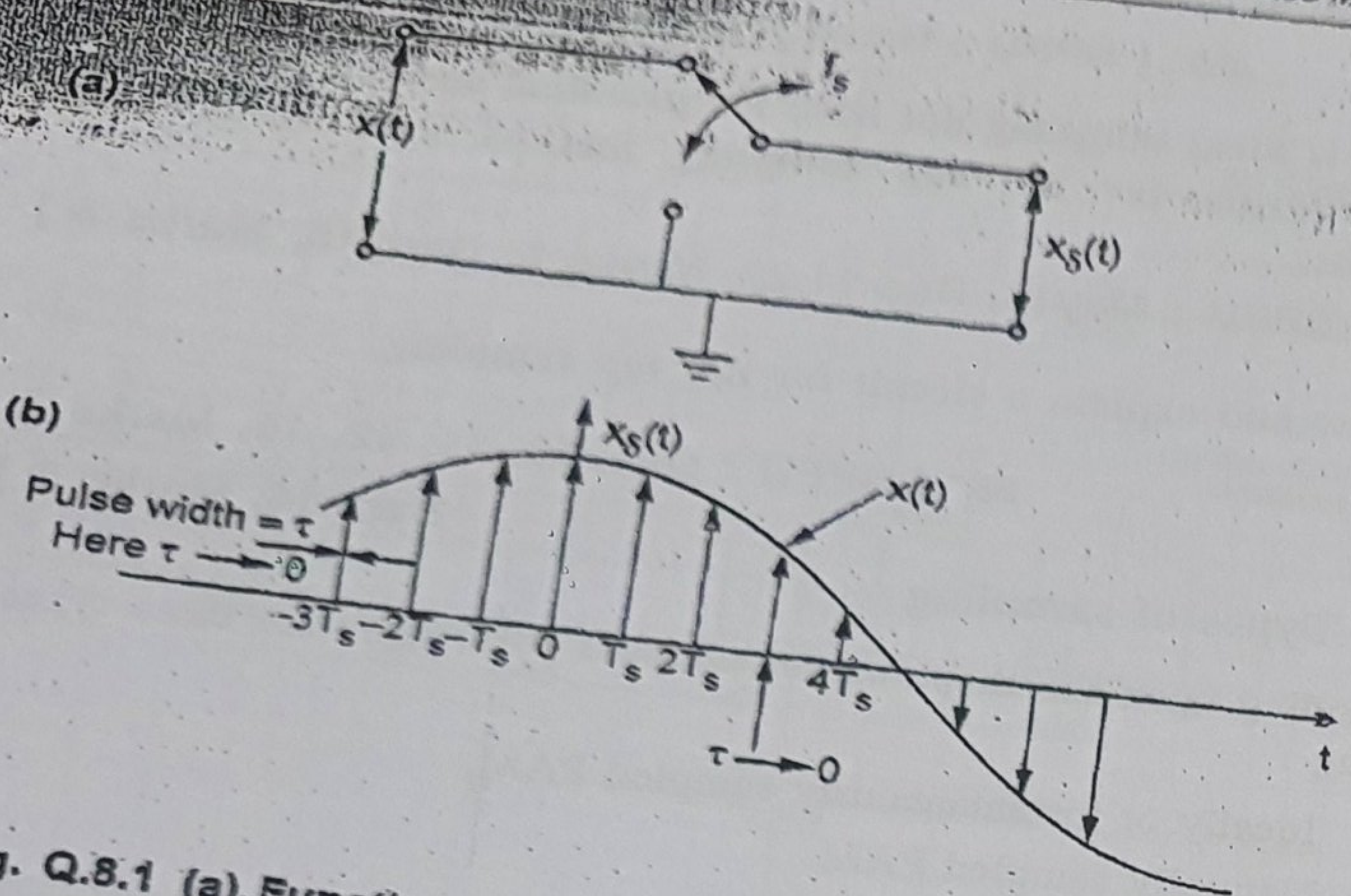


Fig. Q.8.1 (a) Functional diagram of a switching sampler (b) Waveforms of $x(t)$ and its sampled version

$$= x(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s) \qquad \text{... (Q.8.2)}$$

The Fourier transform of the ideally sampled signal given by above equation can be written as,

$$\delta \;=\; \sum_{=-\infty}^{\infty} \;- \qquad \text{... (Q.8.3)}$$

$X_\delta(f)$ is periodic in f_s and weighed by f_s.

Instantaneous sampling not suitable for practical applications.

Instantaneous sampling is possible only in theory because it is not possible to have a pulse whose width approaches zero.

2. Natural sampling or chopper sampling

- **Basic principle :** In natural sampling the pulse has a finite width τ. Natural sampling is some times called chopper sampling because the waveform of the sampled signal appears to be chopped off from the original signal waveform.

Explanation : A sampled signal $s(t)$ is obtained by multiplication of a sampling function and signal $x(t)$. Sampling function $c(t)$ is a train of periodic pulses of width τ and frequency equal to f_s Hz. Fig. Q.8.2 shows a functional diagram of natural sampler. When $c(t)$ goes high, a switch 's' is closed. Therefore,

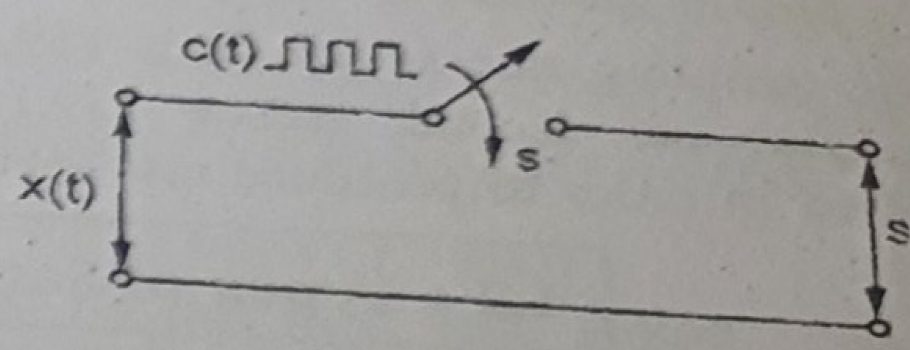


Fig. Q.8.2 Natural sampler

$$s(t) = x(t) \quad \text{when} \quad c(t) = A$$

$$s(t) = 0 \quad \text{when} \quad c(t) = 0$$

Here A is amplitude of $c(t)$.

- The waveforms of $x(t)$, $c(t)$ and $s(t)$ are shown in Fig. Q.8.3 (a), Q.8.3 (b) and Q.8.3 (c) respectively. Signal $s(t)$ can also be defined mathematically as,

$$s(t) = c(t) \cdot x(t) \qquad \text{... (Q.8.4)}$$

Here, $c(t)$ is the periodic train of pulses of width τ and frequency f_s.

The spectrum of naturally sampled signal is given as,

$$S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} sinc\,(nf_s\tau)\, X(f - nf_s) \qquad \text{... (Q.8.5)}$$

Comments : $X(f)$ are periodic in f_s and are weighed by the sinc function. Fig. Q.8.4 (a) shows some arbitrary spectra for $x(t)$ and corresponding spectrum $S(f)$ is shown in Fig. Q.8.4 (b).

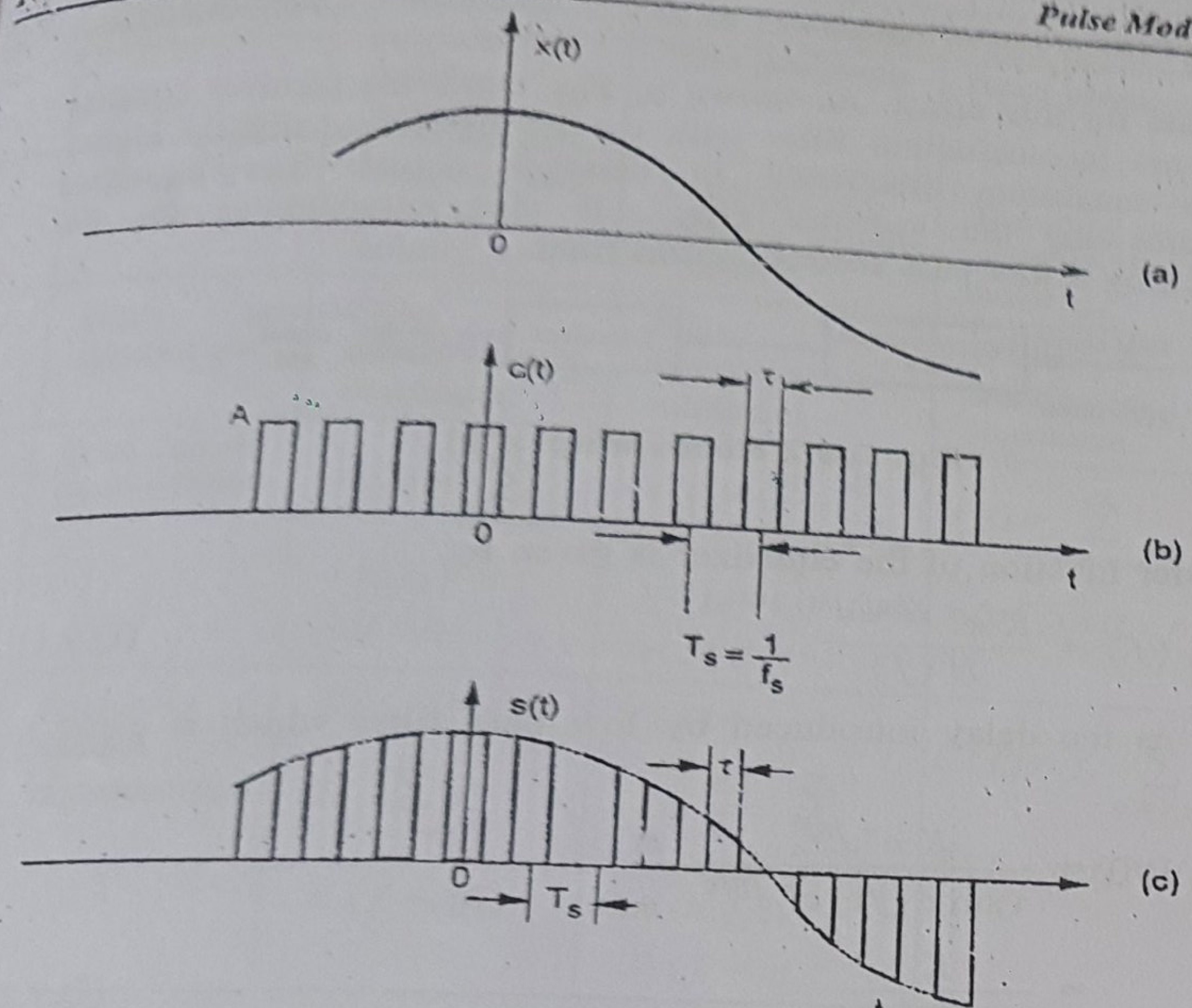


Fig. Q.8.3 (a) Continuous time signal $x(t)$
(b) Sampling function waveform i.e. periodic pulse train
(c) Naturally sampled signal waveform $s(t)$

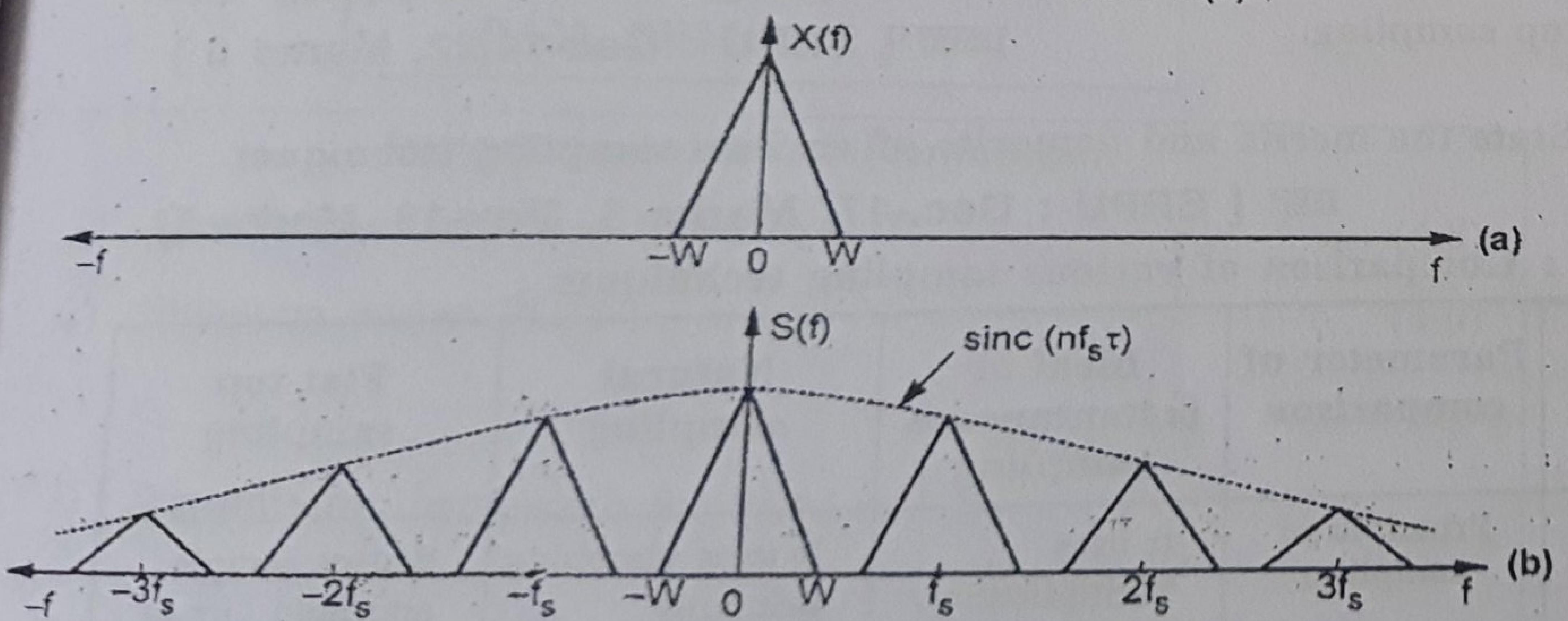
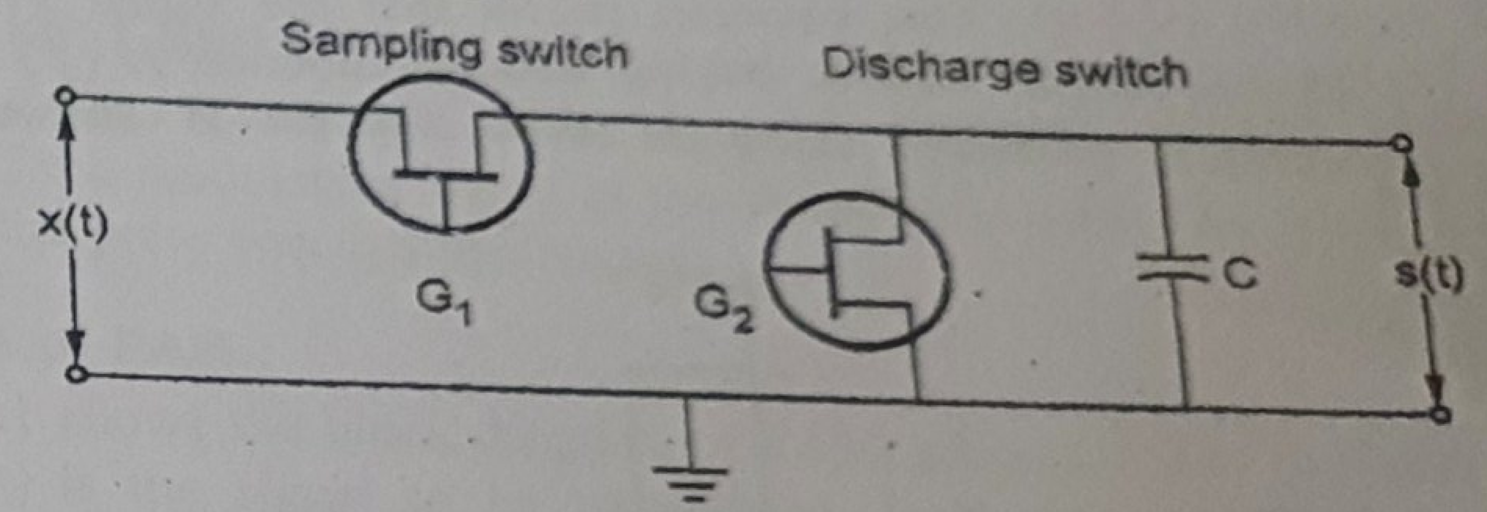


Fig. Q.8.4 (a) Spectrum of continuous time signal $x(t)$
(b) Spectrum of naturally sampled signal

Flat top sampling or rectangular pulse sampling

Basic principle : The top of the samples remains constant and equal to instantaneous value of baseband signal $x(t)$ at the start of sampling. The duration of each sample is τ and sampling rate is equal to $f_s = \frac{1}{T_s}$.

Generation of flat top samples : Fig. Q.8.5 (a) shows the functional diagram of sample and hold circuit generating flat top samples and Fig. Q.8.5 (b) shows waveforms.



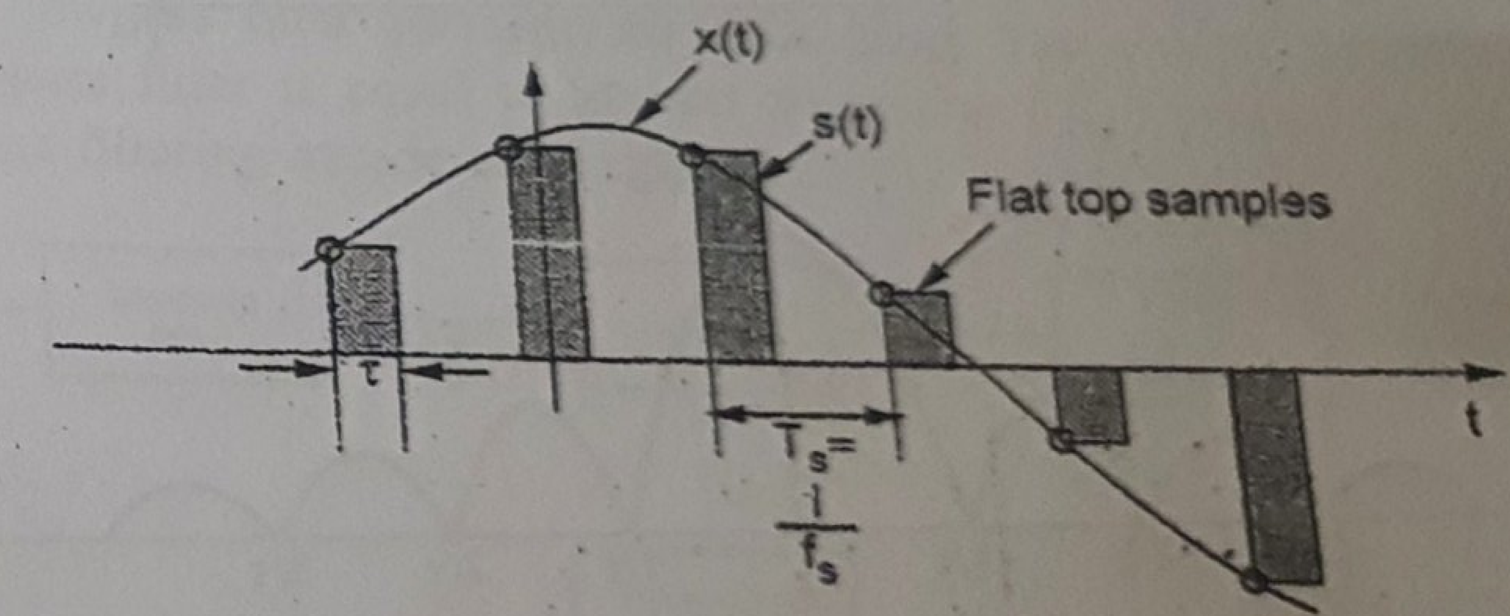


Fig. Q.8.5 (a) Sample and hold circuit generating flat top sampling
(b) Waveforms of flat top sampling

Normally the width of the pulse in flat top sampling and natural sampling is reduced as far as possible to reduce the transmission bandwidth.

The spectrum of flat top sampled signal is given as,

$$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) H(f) \quad \text{... (Q.8.6)}$$

Q.9 Explain aperture effect in detail with spectral diagrams.

☞ [SPPU : Dec.-12, 10, Marks 5]

OR What is meant by 'aperature effect' ? How can it be reduced ?

☞ [SPPU : May-17, Marks 7, May-19, Marks 6]

Ans. : Aperture effect : The spectrum of flat top sampled signal is given by equation (Q.8.6). This equation shows that the signal $s(t)$ is obtained by passing through a filter having transfer function $H(f)$. The corresponding impulse response $h(t)$ in time domain is shown in Fig. Q.9.1 (a).

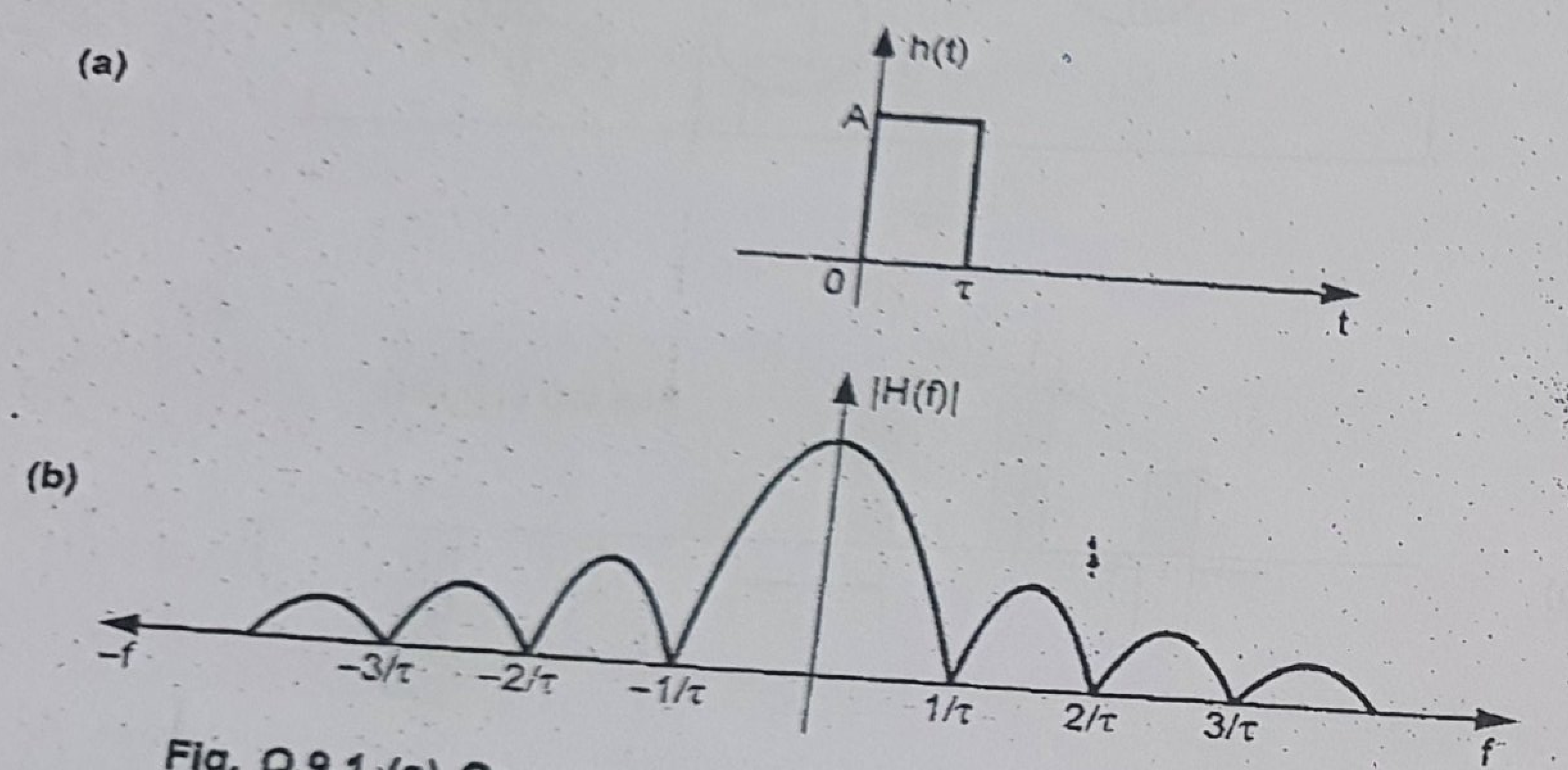


Fig. Q.9.1 (a) One pulse of rectangular pulse train (b) Spectrum of the pulse of Fig. Q.9.1 (a)

- The spectrum of a rectangular pulse is given as,

$$H(f) = \tau \, sinc(f\tau) e^{-j\pi f\tau} \quad \because A = 1 \qquad \dots (Q.9.1)$$

Thus we can see from Fig. Q.9.1 (b) that by using flat top samples an amplitude distortion is introduced in reconstructed signal $x(t)$ from $s(t)$. The high frequency rolloff of $H(f)$ acts like a low-pass filter and attenuates upper portion of message spectrum. These high frequencies of $x(t)$ are affected. This effect is called *aperture effect.*

- As the duration 'τ' of the pulse increases, aperture effect is more prominent. Therefore during reconstruction an equalizer is required to

compensate for this effect. As shown in Fig. Q.9.2 the receiver consists of low pass reconstruction filter with cut-off frequency slightly higher than the maximum frequency in message signal. The equalizer compensates for the aperture effect. It also compensates for the attenuation by a low pass reconstruction filter.

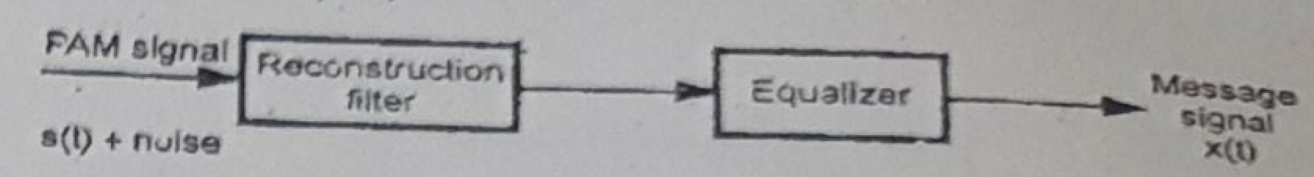


Fig. Q.9.2 Recovering $x(t)$

- The transfer function of the equalizer is given by,

$$H_{eq}(f) = \frac{K e^{-j2\pi f t_d}}{H(f)} \qquad \dots (Q.9.1)$$

- Here 't_d' is the delay introduced by low pass filter which is equal to $\tau/2$

$$\therefore \quad H_{eq}(f) = \frac{K e^{-j\pi f\tau}}{\tau \sin c(f\tau) e^{-j\pi f\tau}}$$

$$= \frac{K}{\tau \, sinc(f\tau)} \qquad \dots (Q.9.2)$$

This is the transfer function of an equalizer.

Q.10 Distinguish between ideal sampling, natural sampling and flat-top sampling.

☞ [SPPU : Dec-14,22, Marks 6]

OR State the merits and demerits of various sampling techniques.

☞ [SPPU : Dec.-17, Marks 3, May-19, Marks 5]

Ans. : Comparison of various sampling techniques :

Sr. No.	Parameter of comparison	Ideal or instantaneous sampling	Natural sampling	Flat top sampling
1.	Principle of sampling	It uses multiplication by an impulse function	It uses chopping principle	It uses sample and hold circuit

2	Realizability	This is not practically possible method	This method is used practically	This method is used practically
3	Sampling rate	Sampling rate tends to infinity	Sampling rate satisfies Nyquist criteria	Sampling rate satisfies Nyquist criteria
4	Noise interference	Noise interference is maximum	Noise interference is minimum	Noise interference is maximum
5	Time domain representation	$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\delta(t-nT_s)$	$s(t) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} x(t)\, sinc(n f_s \tau)\, e^{j 2\pi n f_s t}$	$s(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t-nT_s)$
6	Frequency domain representation	$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)$	$S(f) = \frac{\tau A}{T_s} \sum_{n=-\infty}^{\infty} sinc(n f_s \tau)\, X(f-nf_s)$	$S(f) = f_s \sum_{n=-\infty}^{\infty} X(f-nf_s)\, H(f)$

Table Q.10.1

4.3 Analog Pulse Modulation : PAM, PPM and PWM

Important Points to Remember

1) Quantization Levels in PCM $q = 2^v$ ... (4.8)

2) Signaling rate in PCM : $r = vf_s$ ... (4.9)

3) Transmission bandwidth for PCM = $\begin{cases} B_T \geq \frac{1}{2} r \\ B_T \geq 1/2\ vf_s \\ B_T \geq vW \end{cases}$

4) for μ – law $Z(x) = (Sgnx)\dfrac{ln(1+\mu|x|)}{ln(1+\mu)}$ $|x| \leq 1$. ... (4.10)

5) A-law $Z(x) = \begin{cases} \dfrac{A|x|}{1+lnA} & \text{for } 0 \leq x \leq \frac{1}{A} \\ \dfrac{1+ln(A|x|)}{1+lnA} & \text{for } \frac{1}{A} \leq x \leq 1 \end{cases}$ ... (4.11)

Q.11 Draw and explain a PAM signal. How it is generated and demodulated ?

☞ [SPPU : May-12, Marks 8, Dec.-15,19, Marks 7]

Ans. : Principle : The amplitude of the pulse is directly proportional to amplitude of the modulating signal at the sampling instant. The width and position of the pulse remains unchanged.

Generation of PAM :

- Fig. Q.11.1 shows the block diagram of PAM generator. The modulating signal $x(t)$ is the signal to be transmitted. It is given to the lowpass filter. The lowpass filter performs band limiting. The cut-off frequency of the lowpass filter is equal to highest frequency (f_m) present in $x(t)$. The lowpass filtering avoids aliasing.

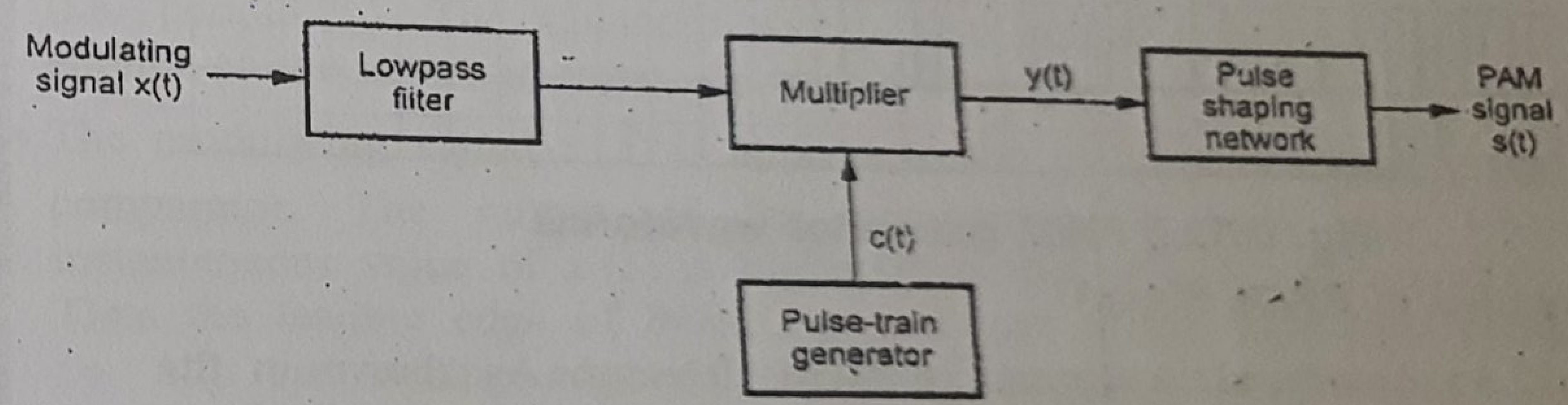


Fig. Q.11.1 Generation of PAM

- The multiplier samples $x(t)$ with the help of a pulse-train generator. The pulse train generator produces the pulse train $c(t)$ as shown in Fig. Q.11.2 (b). The multiplication of $x(t)$ and $c(t)$ produces the PAM signal $y(t)$ as shown in Fig. Q.11.2 (c). Note that the tops of the pulses are varied according to amplitude of $x(t)$.
- The pulse shaping network produces the flat top pulses as shwon in Fig. Q.11.2 (d). This is the required PAM signal.

level. As a result, time period required to charge the capacitor up to threshold voltage level changes, giving pulse modulated signal at the output, as shown in the Fig. Q.12.2.

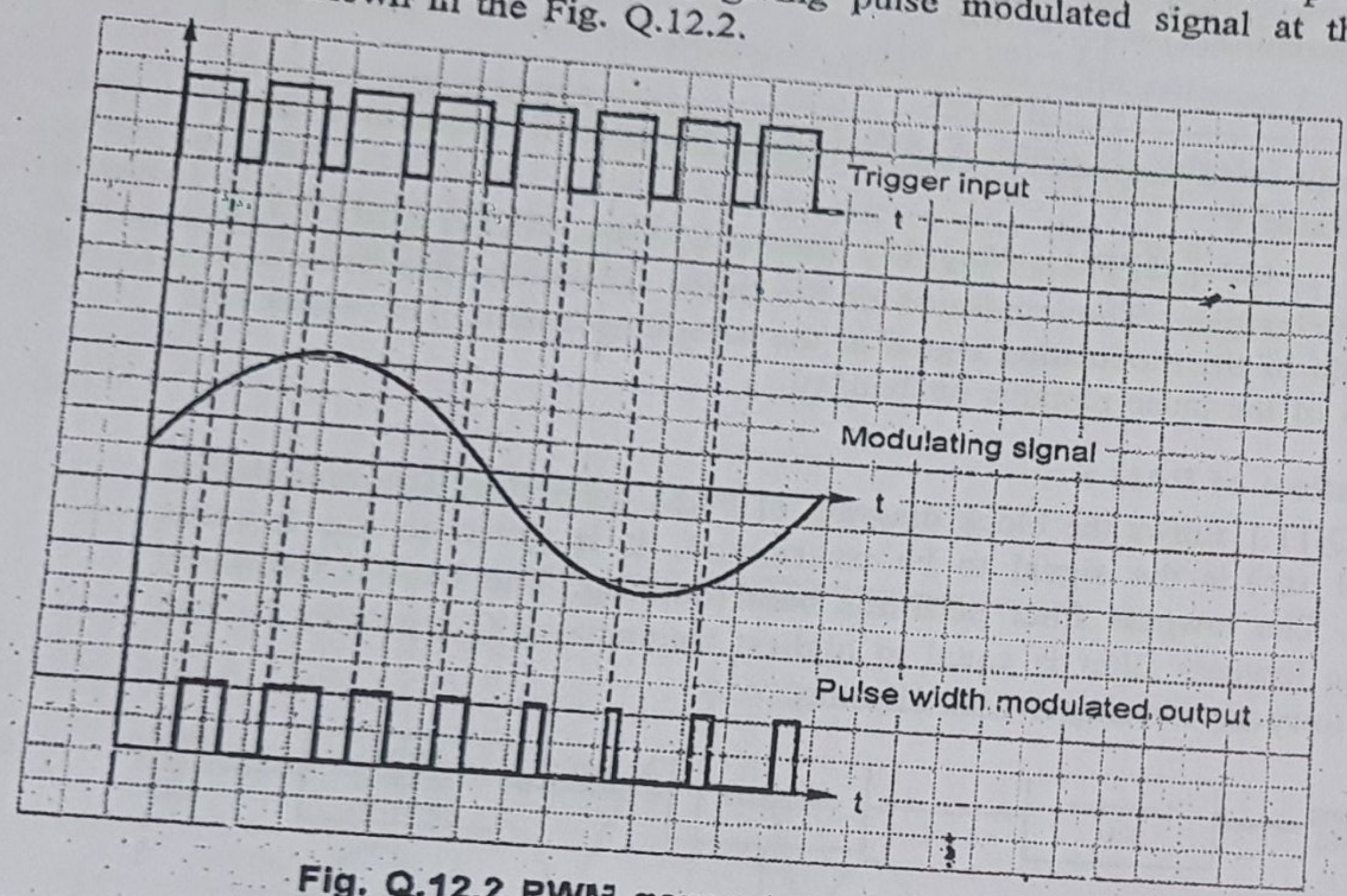


Fig. Q.12.2 PWM generator waveforms

Demodulation of PWM Signal

- Fig. Q.12.3 shows the block diagram of PWM detector. As shown in the Fig. Q.12.3, the received PWM signal is applied to the schmitt trigger circuit. The schmitt trigger circuit removes the noise in the PWM waveform. The regenerated PWM is then applied to the ramp generator and the synchronization pulse generator.
- The ramp generator produces ramps for the duration of pulses such that height of ramps are proportional to the widths of PWM pulses. The maximum ramp voltage is retained till the next pulse.
- On the other hand synchronous pulse generator produces reference pulses with constant amplitude and pulse width. These pulses are delayed by specific amount of delay.

- The delayed reference pulses and the output of ramp generator is added with the help of adder. The output of adder is given to the level shifter. Here, negative offset waveform is clipped by rectifier.
- Finally, the output of rectifier is passed through low-pass filter to recover the modulating signal.

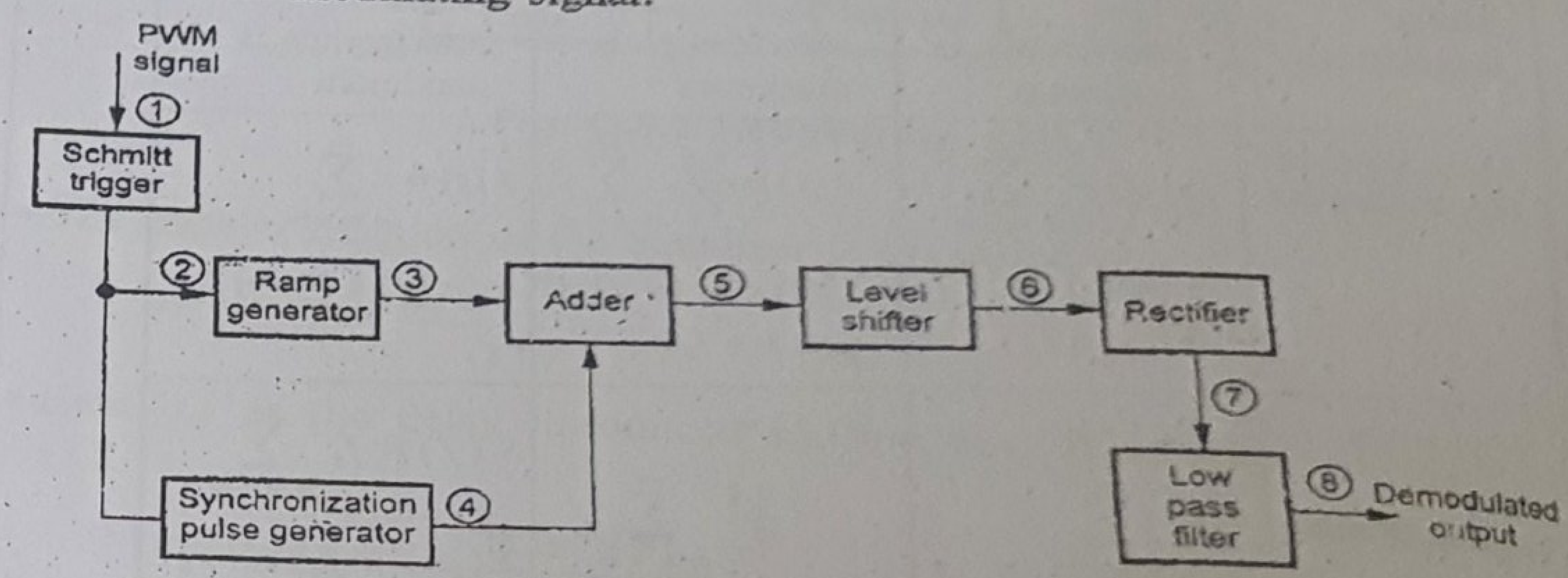


Fig. Q.12.3 PWM detector

Q.13 State advantages, disadvantages and applications of PWM.

☞ [SPPU : Dec.-14, Marks 3]

Ans.: Advantages, disadvantages and applications of PWM

Advantages

1. Unlike, PAM, noise is less, since in PWM, amplitude is held constant.
2. Signal and noise separation is very easy.
3. PWM communication does not require synchronization between transmitter and receiver.

Disadvantages

1. In PWM, pulses are varying in width and therefore their power contents are variable. This requires that the transmitter must be able to handle the power contents of the pulse having maximum pulse width.
2. Large bandwidth is required for the PWM communication as compared to PAM.

Applications

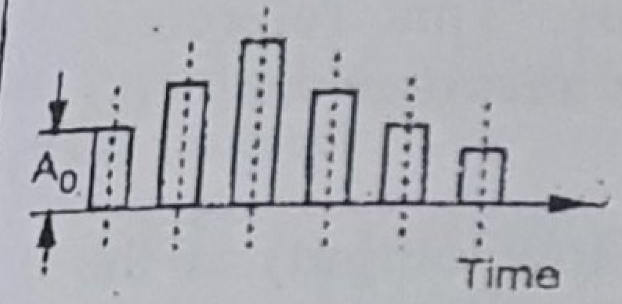
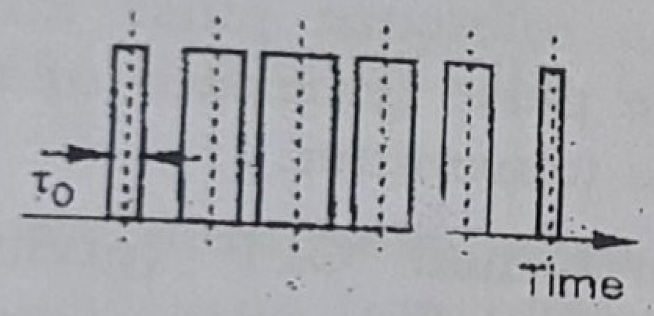
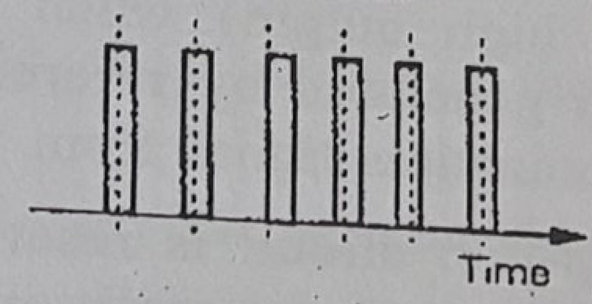
1. PWM is used for asynchronous transmission over noisy channel.
2. PWM is used to generate PPM.
3. Motor control.

Q.14 Compare between PAM, PWM and PPM.

☞ [SPPU : May-14,17,19, Dec.-12,19, June-22, Marks 6
Dec.-22, Marks 5]

Ans. : Comparison between various pulse modulation methods

Table Q.14.1 shows the comparison among various pulse modulation techniques.

Sr. No.	Pulse amplitude modulation	Pulse width/duration modulation	Pulse position modulation
1.	Waveform 	Waveform 	Waveform 
2.	Amplitude of the pulse is proportional to amplitude of modulating signal.	Width of the pulse is proportional to amplitude of modulating signal.	The relative position of the pulse is proprotional to the amplitude of modulating signal.
3.	The bandwidth of the transmission channel depends on width of the pulse.	Bandwidth of transmission channel depends on rise time of the pulse.	Bandwidth of transmission channel depends on rising time of the pulse.
4.	The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter varies.	The instantaneous power of the transmitter remains constant.
5.	Noise interference is high.	Noise interference is minimum.	Noise interference is minimum.

6.	System is complex.	Simple to implement.	Simple to implement.
7.	Similar to amplitude modulation.	Similar to frequency modulation.	Similar to phase modulation.

Table Q.14.1 Comparison of PAM, PWM, PDM

Q.15 Describe with the help of neat sketches of waveforms methods of generation of PDM / PWM. and PPM.

☞ [SPPU : Dec.-15, 18, Marks 7, May-16, Marks 6]

Ans. : Principle : The amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the modulating signal.

Generation of PWM / PDM and PPM

- The block diagram of Fig. Q.15.1 (a) shows the scheme to generate PDM and PPM. The corresponding waveforms are shown in Fig. Q.15.1 (b). The scheme of Fig. Q.15.1(a) combines both sampling and modulation operation.
- The sawtooth generator generates the sawtooth signal of frequency f_s (i.e. period T_s). The sawtooth signal, also called sampling signal is applied to the inverting input of comparator.
- The modulating signal $x(t)$ is applied to the noninverting input of the comparator. The output of the comparator is high only when instantaneous value of $x(t)$ is higher than that of sawtooth waveform. Thus the leading edge of PDM signal occurs at the fixed time period i.e. kT_s, the trailing edge of output of comparator depends on the amplitude of signal $x(t)$. When sawtooth waveform voltage is greater than voltage of $x(t)$ at that instant, the output of comparator remains zero. The trailing edge of the output of comparator (PWM) is modulated by the signal $x(t)$.
- If the sawtooth waveform is reversed, then trailing edge will be fixed and leading edge will be modulated. If sawtooth waveform is replaced by triangular waveform, then both leading and trailing edges will be modulated.
- The pulse duration modulation (PDM) or PWM signal is nothing but output of the comparator. The amplitude of this PDM or PWM signal

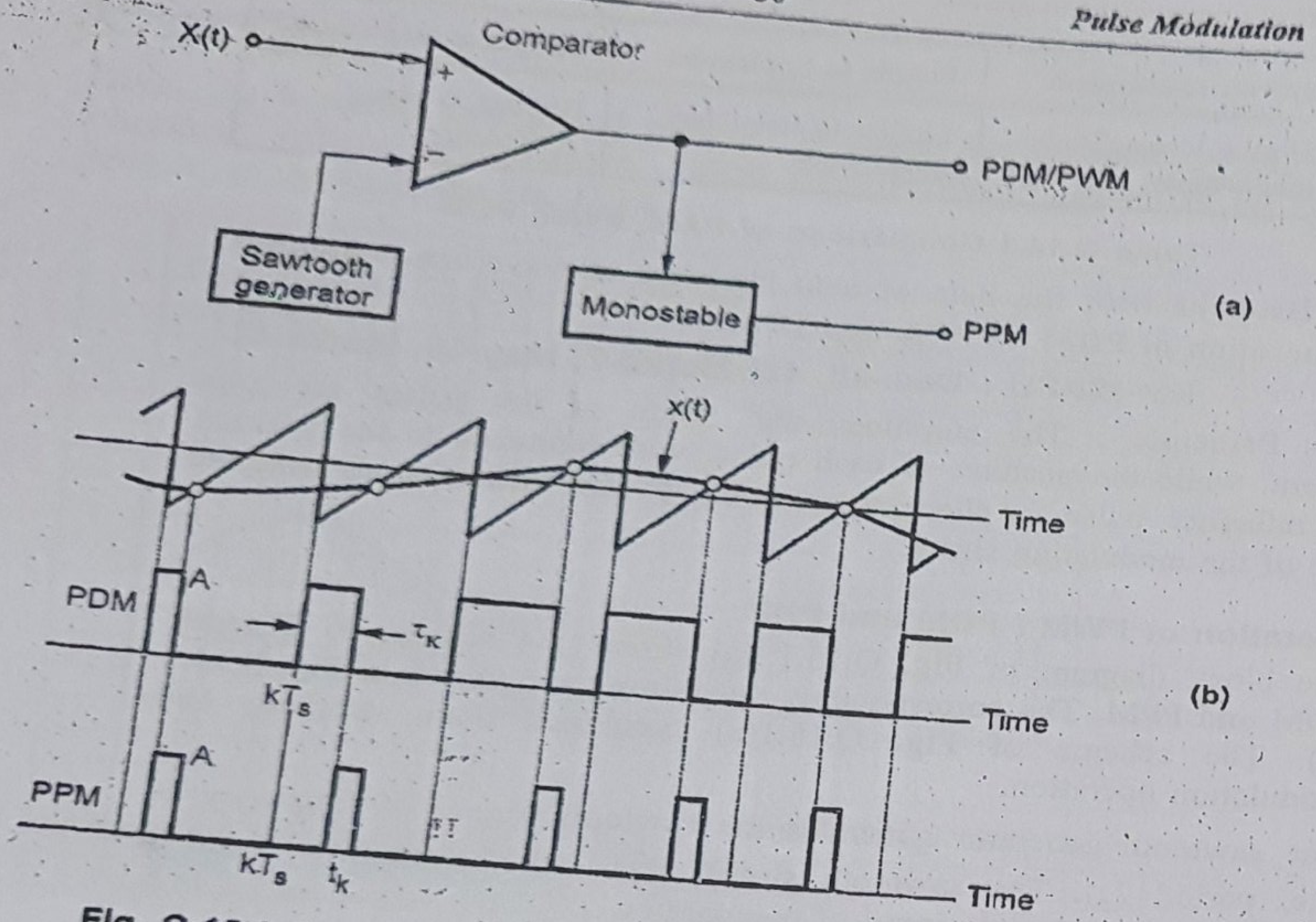


Fig. Q.15.1 Generator of PPM and PWM (a) Block diagram (b) Waveforms

will be positive saturation of the comparator, which is shown as 'A' in the waveforms. The amplitude is same for all pulses.

- To generate pulse position modulation (PPM), PDM signal is used as the trigger input to one monostable multivibrator. The monostable output remains zero untill it is triggered. The monostable is triggered on the falling (trailing) edge of PDM. The output of monostable then switches to positive saturation level 'A'. This voltage remains high for the fixed period then goes low. The width of the pulse can be determined by monostable. The pulse is this delayed from sampling time kT_s depending on the amplitude of signal $x(t)$ at kT_s.

Q.16 Explain the demodulation of ppm signal with the help of block diagram.
☞ **[SPPU : June-22, Marks 5]**

Ans. : • In the case of pulse-position modulation, it is customary to convert the received pulses that vary in position to pulses that vary in length. One way to achieve this is illustrated in Fig. Q.16.1.

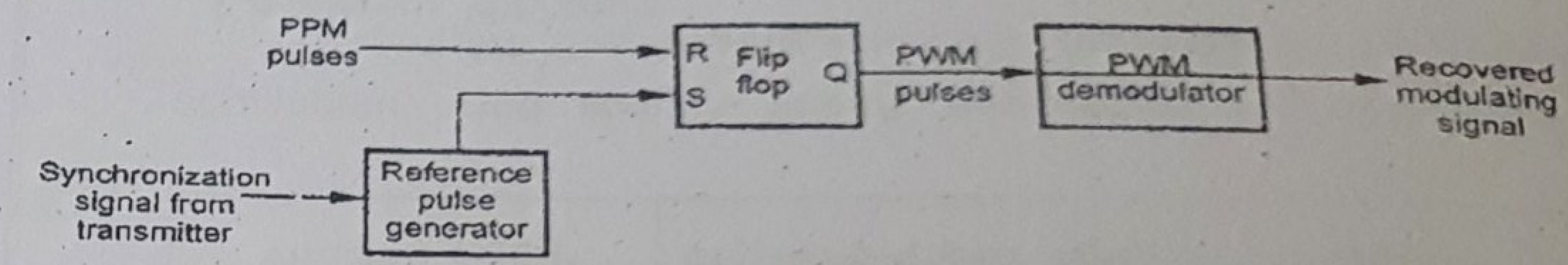


Fig. Q.16.1 PPM demodulator

- As shown in the Fig. Q.16.1 flip-flop circuit is set or turned 'ON' (giving high output) when the reference pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter.
- The flip-flop circuit is reset or turned 'OFF' (giving low output) at the leading edge of the position modulated pulse. This repeats and we get PWM pulses at the output of the flip-flop.
- The PWM pulses are then demodulated by PWM demodulator to get original modulating signal.

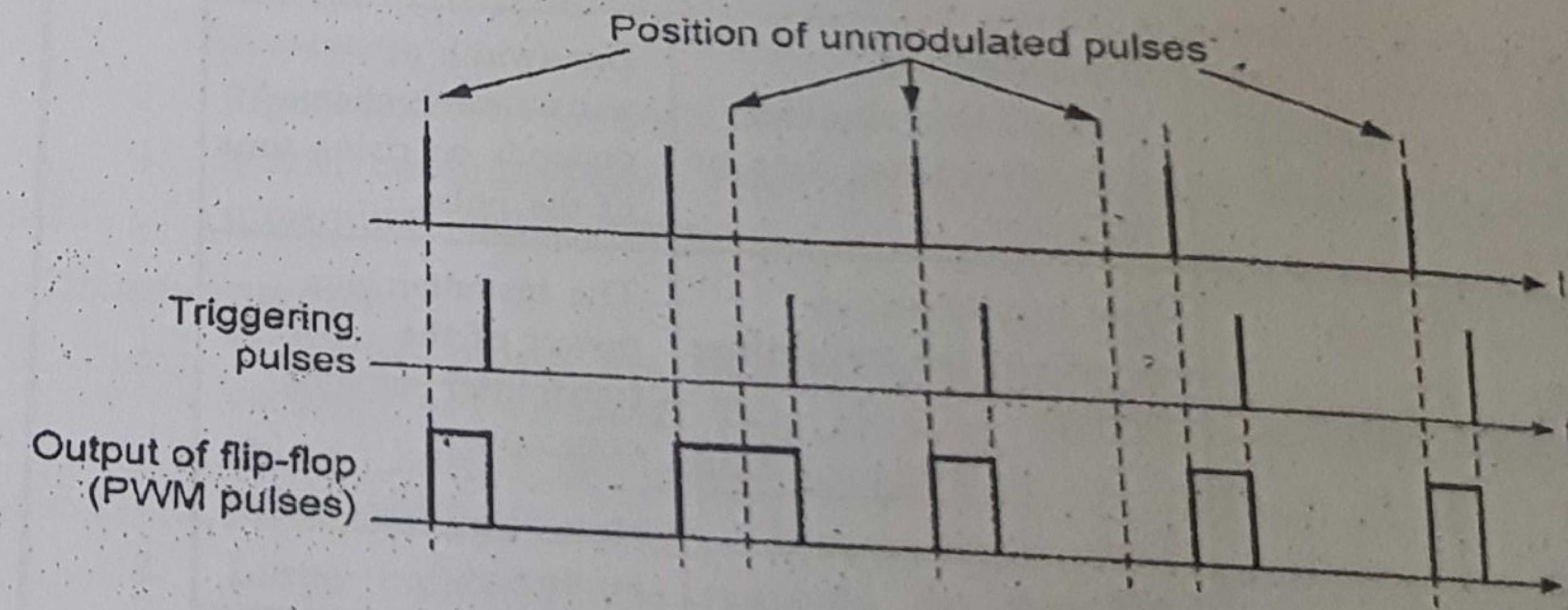


Fig. Q.16.2 Demodulation waveforms for PPM

4.4 : TDM : Channel Bandwidth and Equalization

Q.17 Explain the concept of Time Division Multiplexing (TDM) with help of block diagram. **[SPPU : June-22, Marsk 5]**

Ans. : Fig. Q.17.1 shows the Pulse Amplitude Modulated or PAM signal.

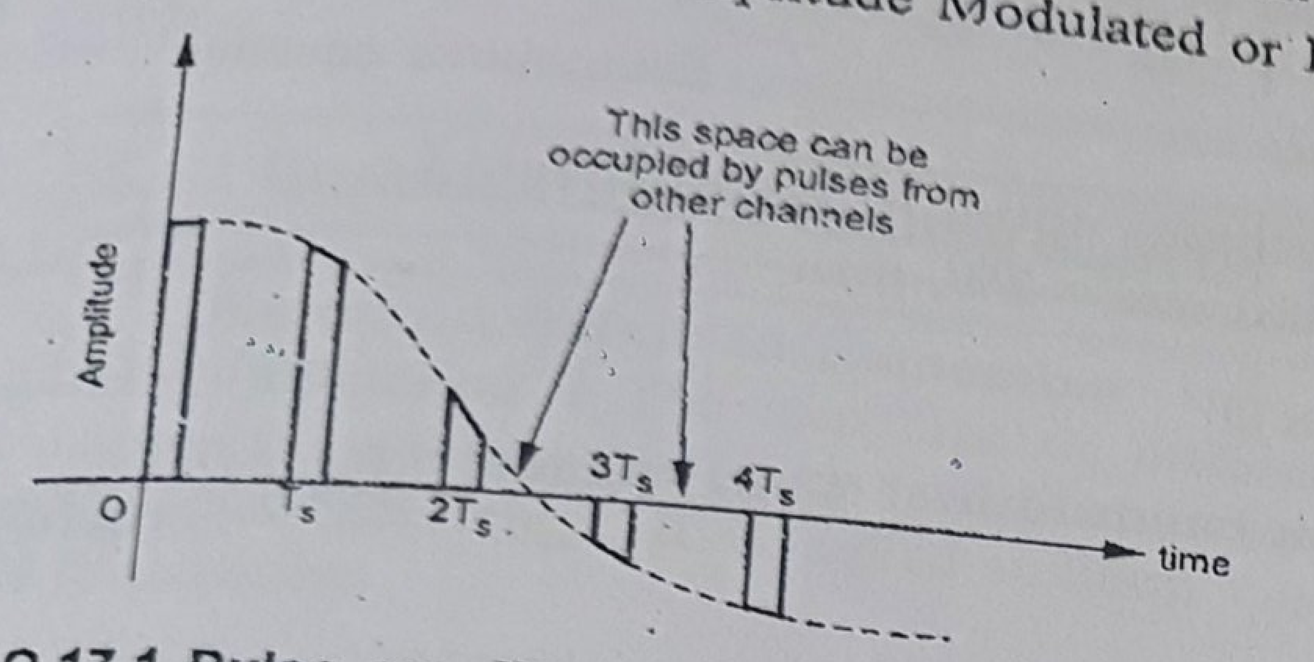


Fig. Q.17.1 Pulse amplitude modulated (PAM) signal

When such PAM signals are transmitted, the large space between the pulses go unused.

It is possible to trnsmit pulses from other channels in this space. This results in time multiplexing of pulses from different channels. It is also called Time Division Multiplexing (TDM).

The Time Division Multiplexing makes maximum utilization of the transmissions channel. The TDM can be analog or digital.

PAM/TDM System :

- Fig. Q.17.2 (a) shows the block diagram of PAM/TDM system. Fig. Q.17.2(b) shows the waveforms of the system. The system shows multiplexing of 'N' PAM channels. Each channel to be transmitted is passed through the low pass filter. (See Fig. Q.17.2 on next page)
- The outputs of the lowpass filter are connected to the rotating sampling switch or commutator. It takes the sample from each channel per revolution and rotates at the rate of f_s. Thus the sampling frequency becomes f_s.
- The single signal composed due to multiplexing of input channels is given to the transmission channel. At the receiver the decommutator separates (decodes) the time multiplexed input channels. These channel signals are then passed through low-pass reconstruction filters.

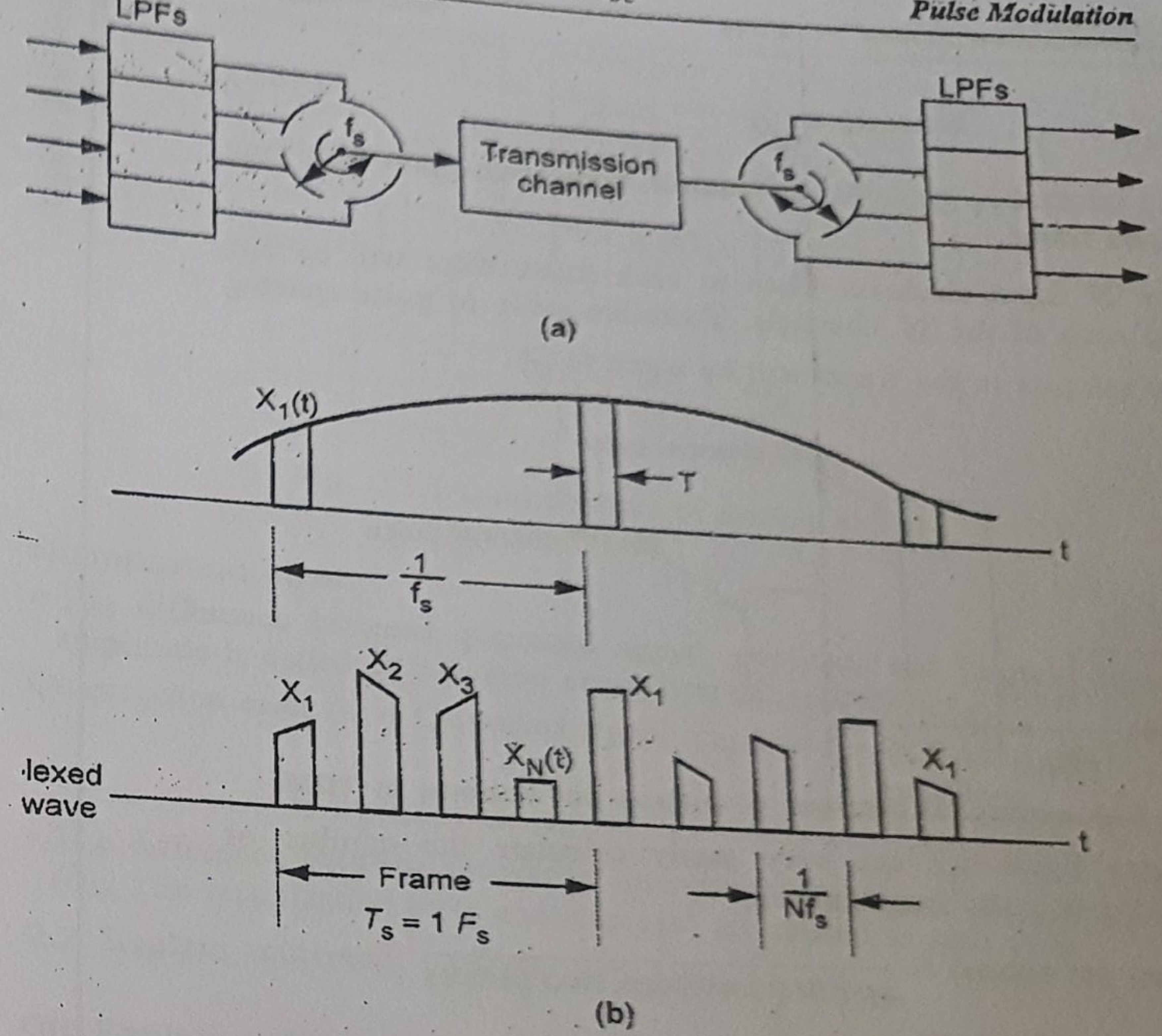


Fig. Q.17.2 TDM system (PAM/TDM system)
(a) Block diagram
(b) Waveforms

Q.18 'N' signals with highest signal frequency 'w' each area multiplexed in PAM / TDM system. Derive signaling rate and transmission bandwidth of TDM channel.

Ans. : • If the highest signal frequency present in all the channels is 'W', then by sampling theorem the sampling frequency f_s should be,

$$f_s \geq 2W \quad \text{... (Q.18.1)}$$

- Therefore the time space between successive samples from any one input will be

$$T_s = \frac{1}{f_s}, \quad \text{hence } T_s \leq \frac{1}{2W} \quad \dots (Q.18.2)$$

Thus the time interval T_s contains one sample from each input. This time interval is called frame.

- Let there be 'N' input channels. Then in each frame there will be one sample from each of the 'N' channels. Therefore pulse to pulse spacing between two samples in the frame will be equal to $\frac{T_s}{N}$.

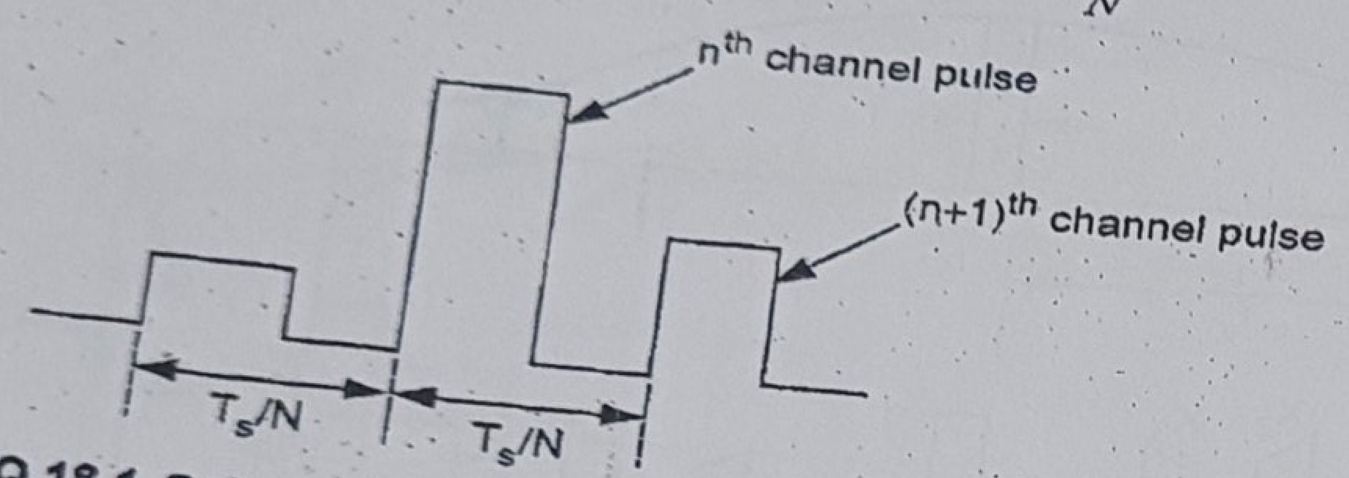


Fig. Q.18.1 Calculation of number of pulses per second in TDM

- From the above figure we can very easily calculate the number of pulses per second or pulse frequency as,

$$\text{Number of pulses per second} = \frac{1}{\text{Spacing between two pulses}}$$

$$= \frac{1}{T_s / N} = \frac{N}{T_s} = N f_s, \quad \text{since } T_s = \frac{1}{f_s} \quad \dots (Q.18.3)$$

These number of pulses per second is also called signaling rate of TDM signal and is denoted by 'r' i.e.,

$$\text{Signaling rate} = r = N f_s$$

Since $f_s \geq 2W$, then signaling rate becomes, ... (Q.18.4)

Signaling rate in PAM/TDM system : $r \geq 2NW$ ... (Q.18.5)

- The pulsed signal in TDM is converted to baseband signal.
- Baseband signal is obtained by passing pulsed TDM signal through low-pass filter. The bandwidth of this low-pass filter is given by half of the signaling rate. i.e.,

$$B_b = \frac{1}{2} r = \frac{1}{2} N f_s \quad \dots (Q.18.6)$$

∴ Transmission bandwidth of TDM channel will be equal to bandwidth of the low-pass filter,

∴ $B_T = \frac{1}{2} N f_s$ from above equation

If sampling rate becomes equal to Nyquist rate i.e.,

f_s (min) = Nyquist rate = 2W, then

$$B_T = \frac{1}{2} N \times 2W$$

Minimum transmission bandwidth of TDM channel : $B_T = NW$ ... (Q.18.7)

END...

... (Q.18.6)

equal to bandwidth of

bove equation

e i.e.,

$B_T = NW$

... (Q.18.7)

END...

Unit V

5 Digital Representation of Analog Signals

5.1 : Quantization of Analog Signals

Important Points to Remember

1) **Uniform quantization** : The step size or difference between two successive quantization levels remains constant over the complete amplitude range. It is called uniform quantization or linear quantization.

2) **Non-uniform quantization** : The step size or difference between two successive quantization levels varies as per variations of amplitude of input signal. It is called non-uniform quantization or non linear quantization.

- Maximum quantization error in midtread and midriser linear / uniform quantization is,

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right|$$

- Maximum quantization error in biased linear quantization is,

$$\varepsilon_{max} = |\delta|$$

Q.1 Define quantization and quantization error.

Ans. : • Fig. Q.1.1 shows analog signal and it's quantized waveform in stepped wave format. There are only four levels (i.e. 0, 1, 2 and 3). These four levels are called quantization levels and they can be represented by two binary digits.

- Observe that,

for $0 \le t \le T_s$, $\quad x_q(nT_s) = 0$

for $T_s \le t \le 2T_s$, $\quad x_q(nT_s) = 1$ ...and so on.

- The stepped waveform is called quantized signal $x_q(nT_s)$.

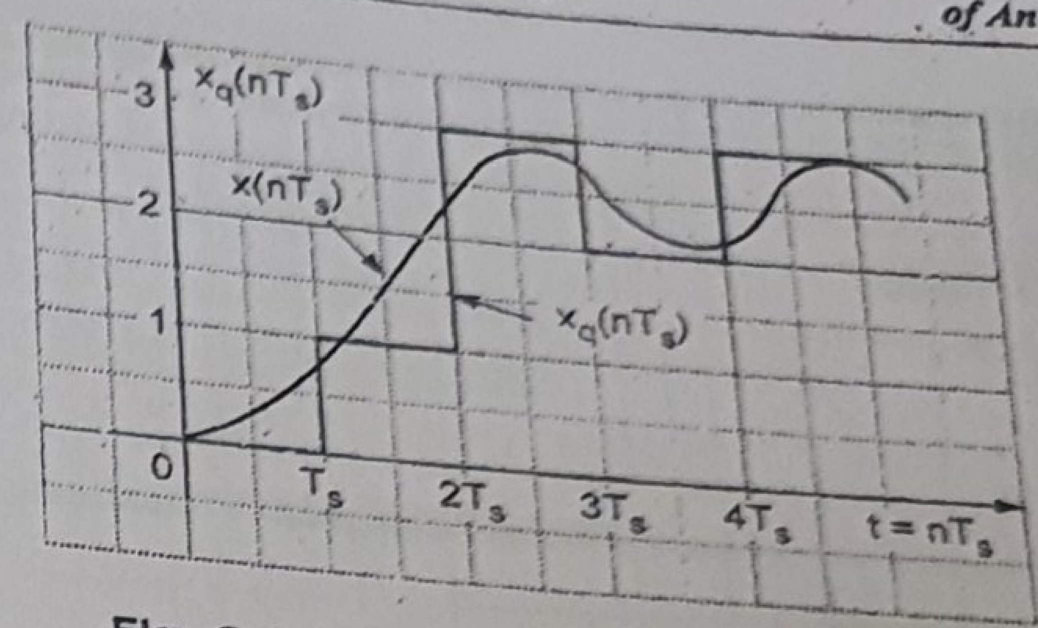


Fig. Q.1.1 Quantization of analog signal

Quantization error

- The difference between quantized signal amplitude and original signal amplitude is called quantization error. It is denoted by ε. i.e.,

Quantization error (ε) = Quantized signal amplitude – Original signal amplitude

$$= x_q(nT_s) - x(nT_s)$$

- The difference between successive quantization levels is called step size (δ). The quantization error and step size are related to each other.

Q.2 Explain midtread, midriser and bised quantizer.

OR Explain uniform quantization with neat waveform.

[SPPU : June-22, Marks 6]

Ans. : Midtread Quantizer : The transfer characteristic of the midtread quantizer is shown in Fig. Q.2.2.

As shown in Fig. Q.2.2, when an input is between $-\delta/2$ and $+\delta/2$ then the quantizer output is zero. i.e.,

For $\quad -\delta/2 \le x(nT_s) < \delta/2 ; \quad x_q(nT_s) = 0$

Here 'δ' is the step size of the quantizer.

- Similarly when input is between $\delta/2$ and $3\,\delta/2$, then quantizer output is δ i.e.,

For $\quad \delta/2 \le x(nT_s) < 3\,\delta/2 ; \quad x_q(nT_s) = \delta$

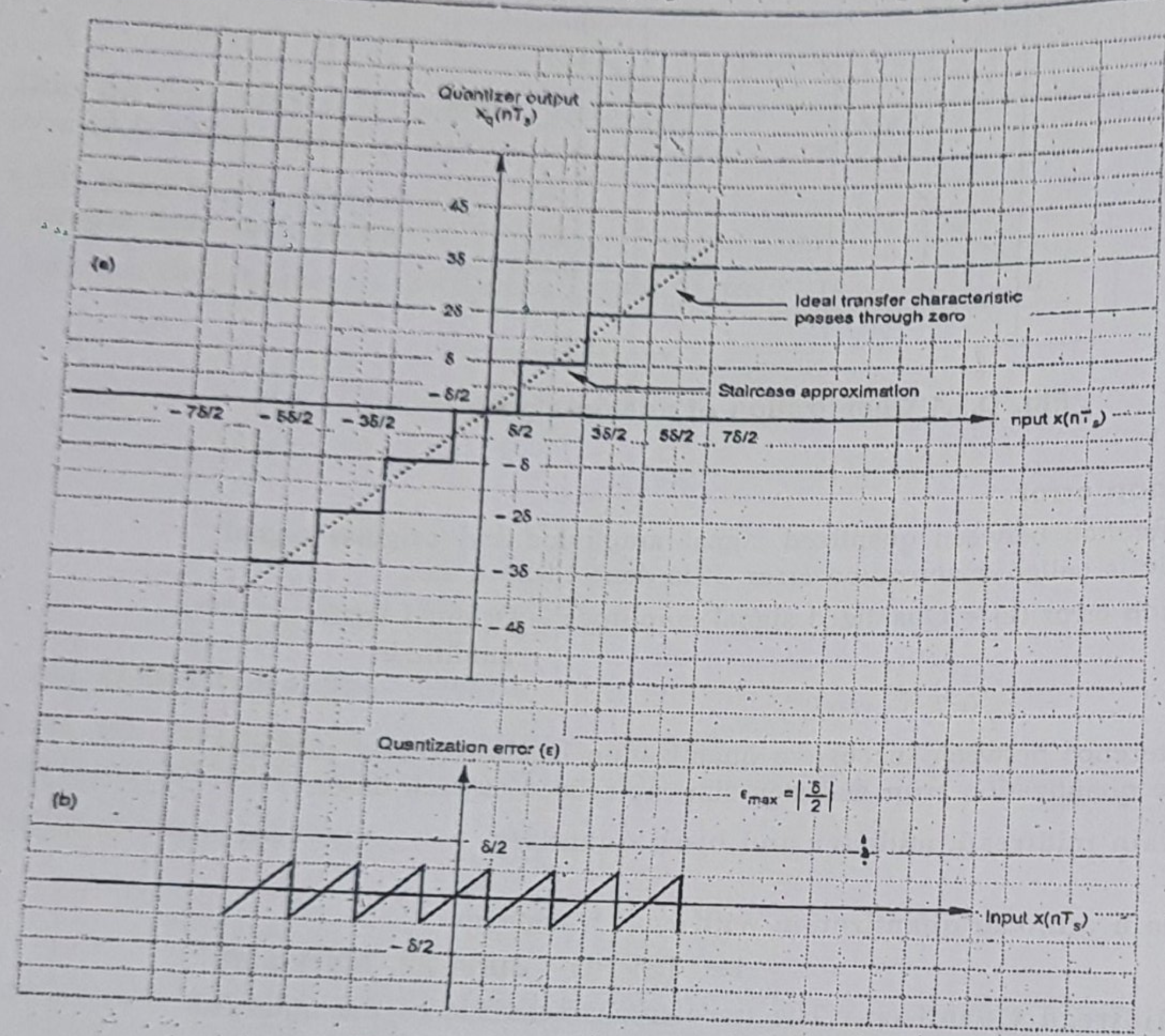


Fig. Q.2.1 Quantization characteristic of midtread quantizer

- Fig. Q.2.1 (b) shows the quantization error of midtread quantizer. Quantization error is given as,

$$\varepsilon = x_q(nT_s) - x(nT_s) \quad \text{... (Q.2.1)}$$

From Fig. Q.2.1 (b) it is clear that,

$$-\delta/2 \le \varepsilon \le \delta/2 \quad \text{... (Q.2.2)}$$

- Thus quantization error lies between $-\delta/2$ and $+\delta/2$. And maximum quantization error is, maximum quantization error, $\varepsilon_{max} = \left|\frac{\delta}{2}\right|$ ... (Q.2.3)

Midriser quantizer : The transfer characteristic of the midriser quantizer is shown in Fig. Q.2.2.

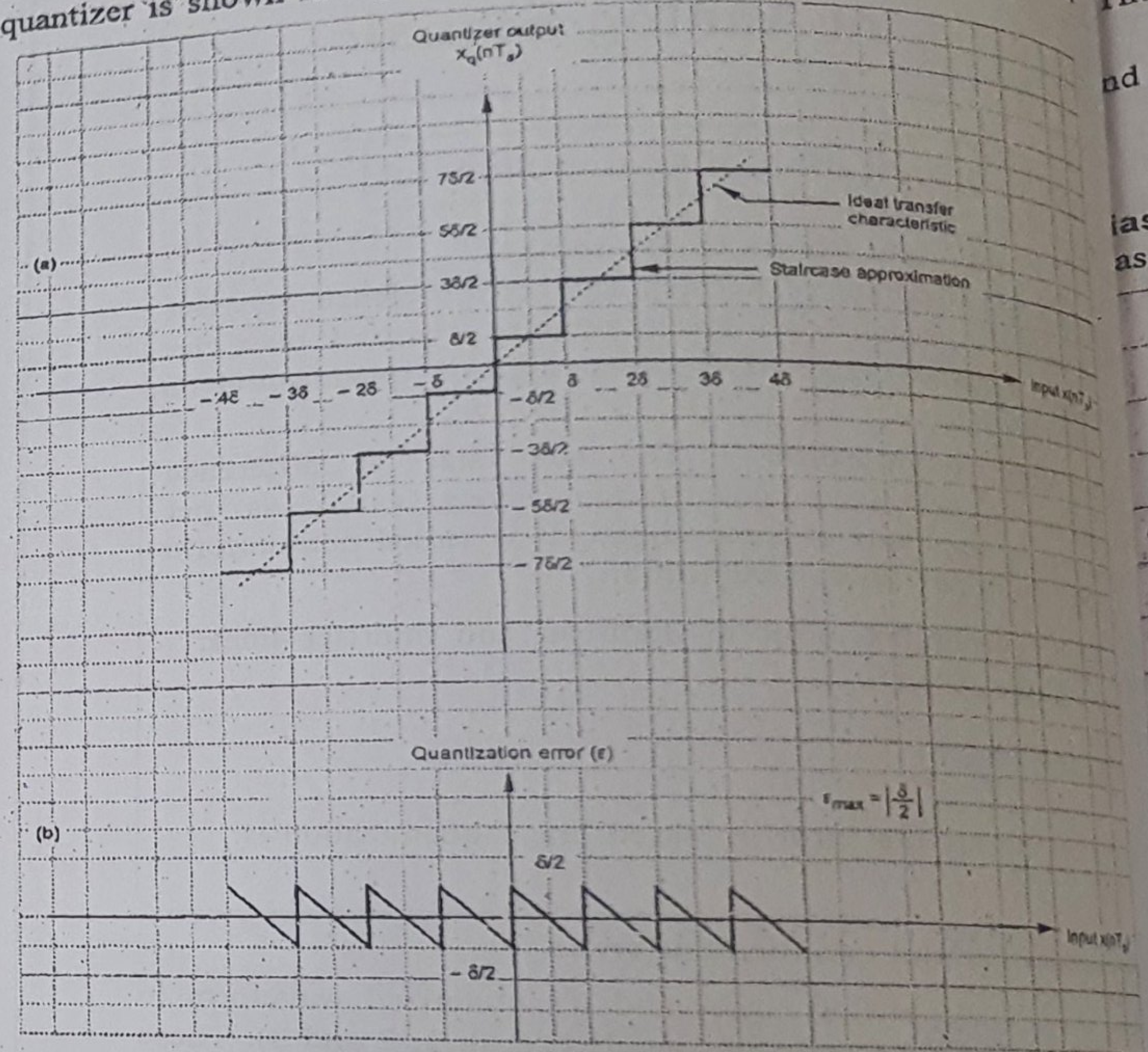


Fig. Q.2.2 (a) Transfer characteristic of midriser quantizer (b) Quantization error

In Fig. Q.2.2 observe that, when an input is between 0 and δ, the output is δ/2. Similarly when an input is between 0 and – δ, the output is – δ/2. i.e.,

For
$$0 \le x(nT_s) < \delta \;; \qquad x_q(nT_s) = \delta/2$$
$$-\delta \le x(nT_s) < 0 \;; \qquad x_q(nT_s) = -\delta/2$$

- Fig. Q.2.2 (b) shows the quantization error in midriser quantization. When input $x(nT_s) = 0$, the quantizer will assign the level of δ/2. Hence quantization error will be,

$$\varepsilon = x_q(nT_s) - x(nT_s) = \delta/2 - 0 = \delta/2$$

...hus the quantization error lies between $-\delta/2$ and $+\delta/2$, i.e.,

$$-\delta/2 \leq \varepsilon \leq \delta/2 \quad \text{... (Q.2.4)}$$

...d the maximum quantization error is,

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right| \quad \text{... (Q.2.5)}$$

...ased quantizer : • Fig. Q.2.3 shows the transfer characteristic of ...ased uniform quantizer.

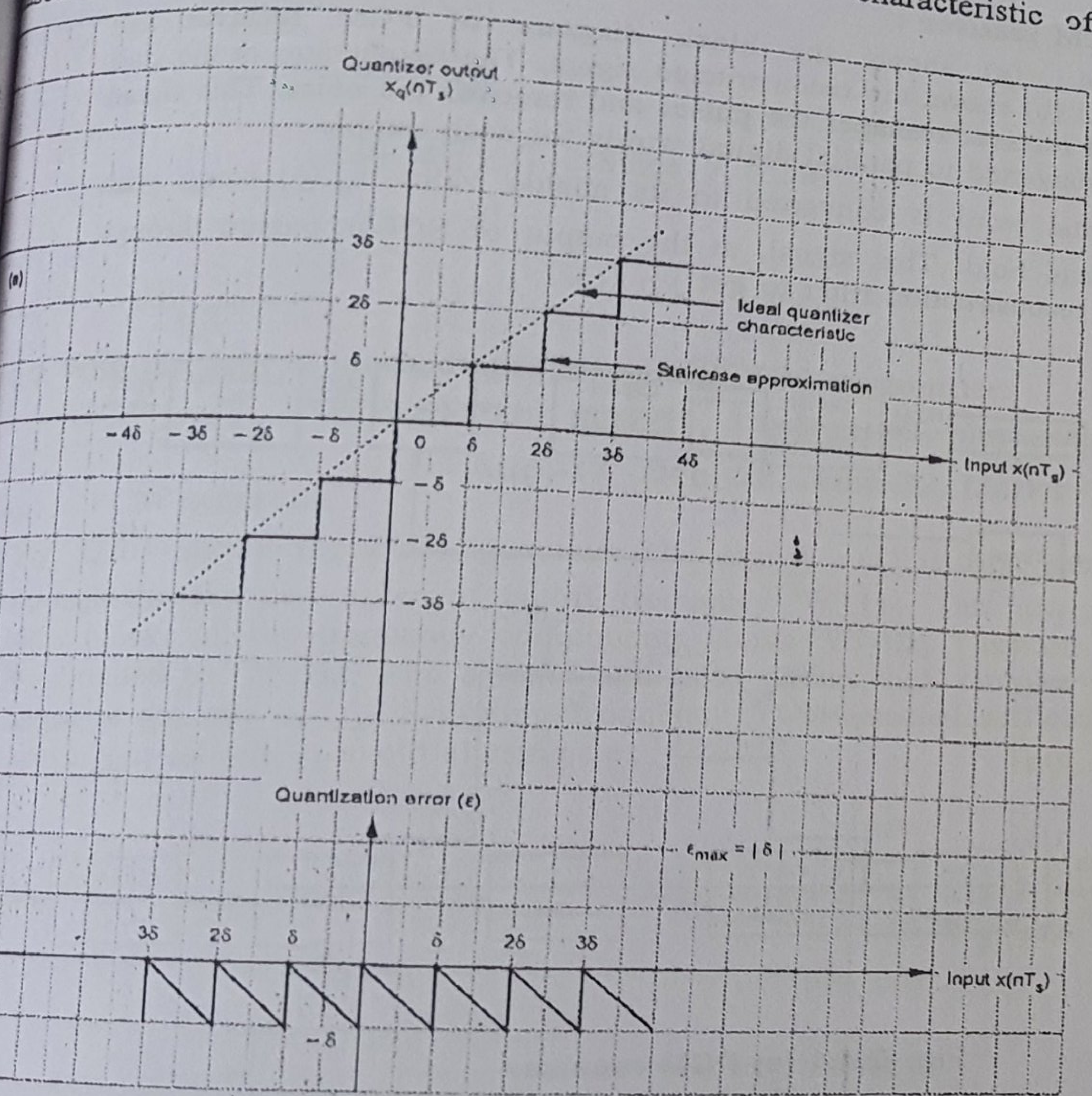


Fig. Q.2.3 (a) Biased quantizer transfer characteristic
(b) Quantization error

- The midriser and midtread quantizers are rounding quantizers. But biased quantizer is truncation quantizer. This is clear from Fig. Q.2.3. When input is between 0 and δ, the output is zero. i.e.,

for, $0 \leq x(nT_s) < \delta$; $\quad x_q(nT_s) = 0$

Similarly, for $-\delta \leq x(nT_s) < 0$; $\quad x_q(nT_s) = -\delta$

- Fig. Q.2.3 shows quantization error.

$$\varepsilon = x_q(nT_s) - x(nT_s) = 0 - \delta = -\delta$$

Thus the quantization error lies between 0 and $-\delta$. i.e.,

$$-\delta \leq \varepsilon \leq 0 \quad \text{... (Q.2.6)}$$

And the maximum quantization error is,

$$\varepsilon_{max} = |\delta| \quad \text{... (Q.2.7)}$$

5.2 : Pulse Code Modualtion System

Important Points to Remember

- Transmission bandwidth in PCM, $B_T \geq \frac{1}{2}r$ or $\frac{1}{2}v\,f_s$ or vW.
- Signalling rate in PCM, $r = v\,f_s$
- Signal to noise ratio in PCM is, $\frac{S}{N}$ = (4.8 + 6v) dB for sinusoidal signal, $\frac{S}{N}$ = (1.8 + 6v) dB.

Sr. No.	Pulse analog modulation	Pulse digital modulation
1.	Amplitude phase and width of the pulse is analog modulated.	Sampled pulse is quantized and encoded.
2.	Pulse is tranmitted without any encoding.	Pulse is not tranmitted. It's encoded digital value to transmitted.
3.	Effect of noise is more.	Effect of noise is less.
4.	Less handwidth is required.	More bandwidth is required.

Q.3 Compare analog and digital communication.

☞ [SPPU : June-22, Marks 6]

OR Explain the need of digital communication.

☞ [SPPU : Dec.-22, Marks 6]

Ans. :

Advantages of digital communication

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

Disadvantages of digital communication

- Eventhough digital communication offer many advantages as given above, it has some drawbacks also. But the advantages of digital communication outweigh disadvantages. They are as follows -

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

Q.4 Draw and explain PCM receiver and state formulae for transmission bandwidth and signaling rate.

☞ [SPPU : Dec-14, 22, Oct.-18, Marks 6]

OR With the help of block diagram explain receiver of PCM.

☞ [SPPU : May-11,14, Marks 4]

Ans. : PCM receiver :

- Fig. Q.4.1 (a) shows the block diagram of PCM receiver and Fig. Q.4.1 (b) shows the reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulses and removes the noise. This signal is then converted to parallel digital words for each sample.
- The digital word is converted to its analog value $x_q(t)$ along with sample and hold. This signal, at the output of S/H is passed through lowpass reconstruction filter to get $y_D(t)$.

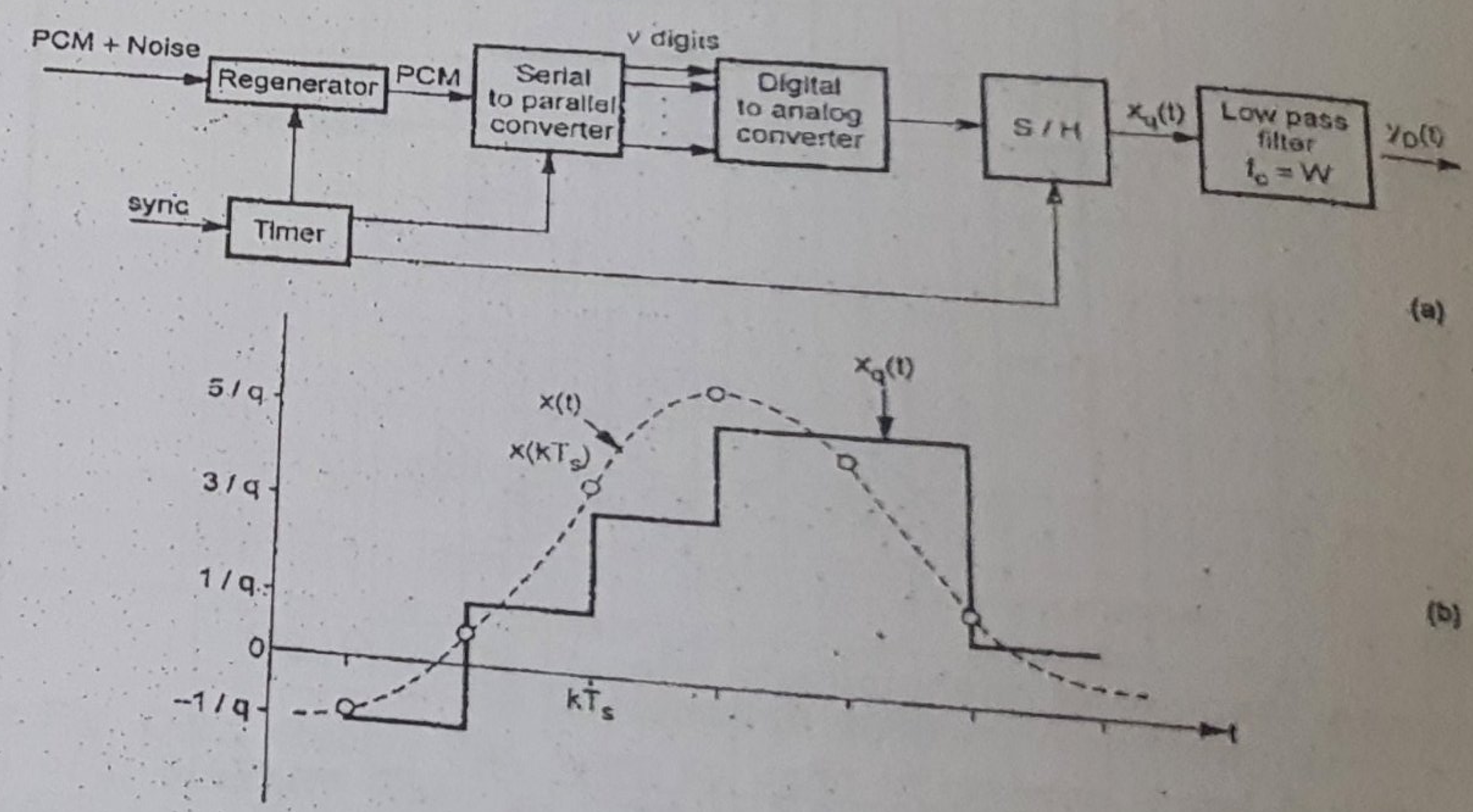


Fig. Q.4.1 (a) PCM receiver
(b) Reconstructed waveform

- As shown in reconstructed signal of Fig. Q.4.1 (b), it is impossible to reconstruct exact original signal $x(t)$ because of permanent quantization error introduced during quantization at the transmitter.

The quantization error can be reduced by increasing the binary levels. The is equivalent to increasing binary digits (bits) per sample. But increasing bits 'ν' increases the signaling rate as well as transmission bandwidth. Therefore the choice of these parameters is made, such that noise due to quantization error (called as quantization noise) is in tolerable limits.

Signaling rate in PCM : $r = \nu f_s$ Here $f_s \geq 2W$. ... (Q.4.1)

Transmission bandwidth of PCM :

$$\left\{ \begin{array}{l} B_T \geq \frac{1}{2} r \\ B_T \geq \frac{1}{2} \nu f_s \quad \text{Since } f_s \geq 2W \\ B_T \geq \nu W \end{array} \right. \quad ...(Q.4.2)$$

Q.5 With the help of block diagram explain PCM transmitter.

☞ [SPPU : May-11, 14, Dec.-19, Marks 4]

Aug.-17, Dec.-22, June-22, Marks 6]

Ans. : PCM generator :

- Fig. Q.5.1 shows the PCM generator. The signal $x(t)$ is first passed through the low-pass filter of cutoff frequency 'W' Hz. This low-pass filter blocks all the frequency components above 'W' Hz. Thus $x(t)$ is bandlimited to 'W' Hz. The sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently above Nyquist rate to avoid aliasing i.e., $f_s \geq 2W$

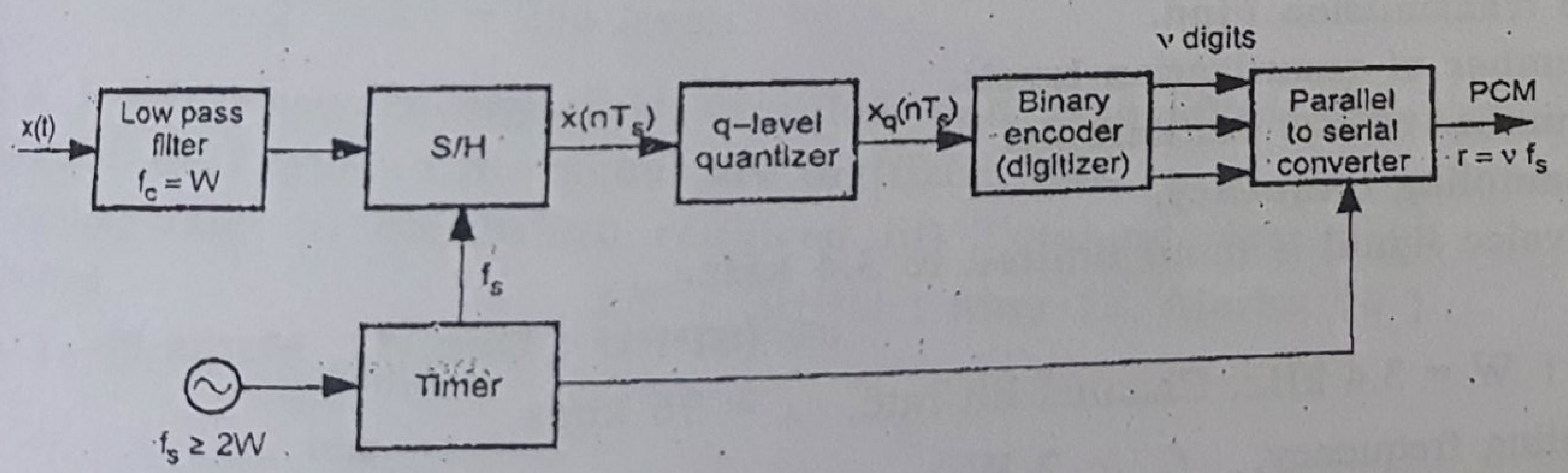


Fig. Q.5.1 PCM generator

- In Fig. Q.5.1 output of sample and hold is called $x(nT_s)$. This $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ with its fixed digital levels. Thus output of quantizer is a digital level called $x_q(nT_s)$.
- The quantized signal level $x_q(nT_s)$ is given to binary encoder. This encoder converts input signal to 'ν' digits binary word. Thus $x_q(nT_s)$ is converted to 'V' binary bits. The encoder is also called digitizer.
- It is not possible to transmit each bit of the binary word separately on transmission line. Therefore 'ν' binary digits are converted to serial bit stream to generate single baseband signal. In a parallel to serial converter, normally a shift register does this job. The output of PCM generator is thus a single baseband signal of binary bits.

An oscillator generates the clocks for sample and hold an parallel to serial converter.

Q.6 "PCM is integral part of ADC", comment ?

☞ [SPPU : Dec.-17, Marks 7]

Ans. : • In analog to digital conversion (ADC), the signal is sampled, quantized and encoded to give ν - bit equivalent.

- PCM essentiallly incorporates all these operations in addition to parallel to serial conversion at the end.
- Hence PCM is integral part of ADC. Also refer Q.5 of chapter-5 for justification.

Q.7 Enlist advantages and drawbacks of PCM system.

☞ [SPPU : May-14, Dec.-19, Marks 2]

Ans. : Advantages and disadvantages of PCM

Advantages

i) Effect of channel noise and interference is reduced.

ii) PCM permits regeneration of pulses along the transmission path. This reduces noise interference.

iii) The bandwidth and signal to noise ratio are related by exponential law.

iv) Multiplexing of various PCM signals is easily possible.

v) Encryption or decryption can be easily incorporated for security purpose.

Disadvantages

i) PCM systems are complex compared to analog pulse modulation methods.

ii) The channel bandwidth is also increased because of digital coding of analog pulses.

Q.8 Describe traditional PCM.

[SPPU : May-11,17,19, Dec.-16,18, Marks 6]

OR With the help of block diagrams, explain the working of PCM generator and receiver. **[SPPU : May-19, Marks 7]**

Ans. : Refer Q.5 for PCM transmitter and Refer Q.4 for PCM receiver.

Q.9 Derive an expression for signal to quantization noise ratio in PCM for uniform quantization. **[SPPU : May-18, Marks 6]**

OR Derive an expression for signal to noise power ratio for linear quantization. **[SPPU : Dec.-18, Marks 8, May-17, Marks 6]**

Ans. : For uniform quatization, the maximum quantization error is given as,

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right| \text{ Here '}\delta\text{' is step size.}$$

- This error is uniformly distributed over the interval $-\frac{\delta}{2} \le \varepsilon_{max} \le \frac{\delta}{2}$. The pdf of this error is given as,

$$f_\varepsilon(\varepsilon) = \frac{1}{\delta} \quad \text{for } -\frac{\delta}{2} \le \varepsilon \le \frac{\delta}{2}$$

- Mean square value of this questization error is given as,

$$E[\varepsilon^2] = \int_{-\infty}^{\infty} e^2 f_\varepsilon(\varepsilon)\, d\varepsilon$$

$$= \int_{-\delta/2}^{\delta/2} \varepsilon^2 \times \frac{1}{\delta} d\varepsilon = \frac{1}{\delta}\left[\frac{\varepsilon^3}{3}\right]_{-\delta/2}^{\delta/2}$$

$$= \frac{1}{\delta}\left[\frac{(\delta/2)^3}{3} + \frac{(\delta/2)^3}{3}\right] = \frac{\delta^2}{12}$$

- Since this quantization error has zero mean value, noise power is equal to mean square value i.e.,

$$\text{Noise power} = E[\varepsilon^2] = \frac{\delta^2}{12}$$

- Let the peak to peak normalized signal amplitude be (–1, 1). Then step size will be $\frac{(1)-(-1)}{q}$ i.e.

$$\delta = \frac{2}{q} = \frac{2}{2^v}, \text{ since number of levels } q = 2^v$$

Here 'v' are number of bits used in PCM.

$$\therefore \quad \text{Noise power} = \frac{\delta^2}{12} = \frac{(2/2^v)^2}{12} = \left(\frac{4/2^{2v}}{12}\right) = \frac{1}{3\times 2^{2v}}$$

- Let the signal power be P. Then signal to noise ratio is given as,

$$\frac{S}{N} = \frac{P}{3\times 2^{2v}} = 3\times 2^{2v} \times P$$

- For normalized power, $P = 1$, Then above equation becomes,

$$\frac{S}{N} = 3\times 2^{2v}$$

$$\left(\frac{S}{N}\right)_{dB} = 10\log_{10} 3\times 2^{2v} = (4.8 + 6v) \text{ dB}$$

This is the signal to noise ratio for linear quantization.

Q.10 A binary channel with 36 kbps bit rate is available for PCM voice transmission Find.

i) Number of quantization levels.

ii) Number of bits per sample.

iii) Sampling frequency.

The voice signal is band limited to 3.4 kHz.

[SPPU : Dec.-17, Marks 6]

Ans. : W = 3.4 kHz, Channel bit rate, r_b = 36 kbps

Sampling frequency, $f_s \ge 2$ W

$$\therefore \quad f_s \ge 2\times 3.4 \text{ kHz} = 6.8 \text{ kHz}$$

For voice transmission, 8 kHz is taken. Hence e, noise power is equal

be (−1, 1). Then step

$q = 2^{\nu}$

is given as,

comes,

lable for PCM

7, Marks 6]

$$f_s = 8 \text{ kHz, which is well above 6.8 kHz.}$$

...rate

$$r = \nu f_s$$

$$36 \times 10^3 = \nu \times 8000 \Rightarrow \nu = 4.5 \approx 5 \text{ bits}$$

...antization levels,

$$q = 2^{\nu} = 2^5 = 32.$$

...11 A binary channel with 64 kbps bit rate is available for PCM ...ice transmission find.

... Number of quantization levels

... Number of bits per sample

... Sampling frequency, the voice signal is band limited to 3.4 kHz.

☞[SPPU : Dec.-18, Marks 8]

...ns. : Given :

Bit rate = 64 kbps, W = 3.4 kHz

i) **Sampling frequency,**

$$f_s \geq 2W \geq 2 \times 3.4 \text{ kHz}$$

$$f_s \geq 6.8 \text{ kHz}$$

...r voice transmission the standard value of sampling frequency.

$$f_s = 8 \text{ kHz}$$

$$\text{Bit rate} = \nu f_s$$

$$64 \times 10^3 = \nu \times 8 \times 10^3 \Rightarrow \nu = 8 \text{ bits}$$

ii) No of bits per sample.

$$\nu = 8 \text{ bits as calculated above.}$$

iii) Now number of **quantization levels** and bits are related as,

$$q = 2^{\nu} = 2^8 = 256 \text{ levels}$$

...12 A 1 kHz sinewave signal is sampled and transmitted using ...2-bit PCM. If 25 cycle of the signal are digitized, find :

... Signaling rate ii) Bandwidth required iii) Total number of bit ...ransmitted. **☞ [SPPU : May-13, Marks 10]**

...ns. : For PCM encoding :

Given $\nu = 12$ bits

$$f_m = W = 1000 \text{ Hz}$$

i) **Signaling rate,**

$$r = \nu f_s$$
$$= \nu \times 2W \quad \text{since } f_s \geq 2W$$
$$= 12 \times 2 \times 1000$$
$$= 24000 \text{ bits/sec.}$$

ii) **Bandwidth,**

$$B_T = \nu W$$
$$= 12 \times 1000 = 12000 \text{ Hz}$$

iii) **Total number of bits transmitted**

One cycle has the period of $\frac{1}{1000} = 1 \times 10^{-3}$ sec. Hence 25 cycles will have the period of 25×10^{-3} sec.

As per the signaling rate, 24000 bits are transmitted in 1 sec. Hence number of bits transmitted in 25×10^{-3} seconds will be,

Total number of bits transmitted $= 24000 \times 25 \times 10^{-3}$

$= 600$ bits.

Q.13 The bandwidth of TV video plus audio signal is 4.5 MHz. If the signal is converted to PCM bit stream with 1024 quantization levels, determine the number of bits/sec generated by the PCM system and transmission bandwidth. Assume that the signal is sampled at the rate of 20 % above nyquist rate. If above linear PCM system is converted to companded PCM, will the output bit rate change ? Justify. **☞ [SPPU : May-12, Marks 8]**

Ans. : The given data is,

$$W = 4.5 \text{ MHz}$$

$$q = 1024 \text{ levels}$$

The Nyquist rate is,

$$\text{Nyquist rate} = 2W = 2 \times 4.5$$
$$= 9 \text{ MHz}$$

The sampling rate is 20 % above the nyquist rate. i.e.

Sampling rate, $f_s = 1.2 \times 9 = 10.8$ MHz

We know that quantization levels q and number of bits ν are related as,

$$q = 2^{\nu}$$

$$\therefore \quad 1024 = 2^{\nu}$$

$$\therefore \quad \nu = 10 \text{ bits}$$

The number of bits/sec generated by PCM system is called bit rate or signaling rate. i.e.,

Signaling rate, $r = \nu f_s$

$$= 10 \times 10.8 \times 10^6 \text{ bits/sec.}$$

$$= 108 \times 10^6 \text{ bits / sec.}$$

Bandwidth, $B_T \geq \frac{1}{2} r$ i.e. 54×10^6 Hz or 54 MHz.

The output bit rate does not change if linear PCM is converted into companded PCM. Companded PCM is used to improve the signal to noise ratio.

Q.14 A signal m(t) band-limited to 3 kHz is sampled at a rate $33\frac{1}{3}$ % higher than the Nyquist rate. The maximum acceptable error in the sample amplitude (the max. quantization error) is 1 % of peak amplitude m_p. The quantized samples are binary coded. Find the minimum bandwidth of a channel required to transmit the encoded binary signal. If 24 such signals are time-division-multiplexed, determine the minimum transmission bandwidth required to transmit the multiplexed signal. **[SPPU : Dec.-12, Marks 8]**

Ans. : i) Sampling rate for W = 3 kHz

$$\therefore f_s = \left(1+\frac{1}{3}\right),$$

$$\text{Nyquist rate} = \frac{4}{3} \times 2 \times W = \frac{4}{3} \times 2 \times 3000 = 24000 \text{ samples/sec.}$$

ii) Bits per sample

$$\varepsilon_{max} = 1\ \% = 0.01$$

$$\therefore \quad \varepsilon_{max} = \left|\frac{\delta}{2}\right| = 0.01$$

$$\therefore \quad \delta = 0.02$$

$$\delta = \frac{2}{q}, \quad \text{hence } q = \frac{2}{0.02} = 100 \text{ levels}$$

$$n = \log_2 q = \log_2 100 = \frac{\log_{10} 100}{\log_{10} 2} = 6.643 \approx 7 \text{ bits}$$

iii) Minimum bandwidth

$$B_T = \frac{1}{2} n f_s = \frac{1}{2} \times 7 \times 24000 = 84 \text{ kHz}$$

iv) Bandwidth for 24 channels

Signaling rate $r = N \times \nu \times f_s$

$$\therefore \quad B_T = \frac{1}{2} r = \frac{1}{2} \times N \times \nu \times f_s = \frac{1}{2} \times 24 \times 7 \times 24000 = 2.016 \text{ MHz}$$

Q.15 A signal m(t) bandlimited to 4 kHz is sampled at a rate 50 % higher than Nyquist rate. The maximum acceptable error in the sample amplitude is 1 % of peak amplitude. The quantized samples are binary coded. Find minimum bandwidth of a channel required to transmit the encoded binary signal. **[SPPU : Dec.-10, Marks 8]**

Ans. : To obtain sampling rate (f_s)

Here $W = 4000$ Hz and $f_s = 50$ % higher than Nyquist rate. This means,

$$f_s = 1.5 \times \text{Nyquist rate} = 1.5 \times 2\,W \text{ (since Nyquist rate} = 2\,W)$$

$$= 1.5 \times 2 \times 4000 = 12000 \text{ samples/sec.}$$

To obtain bits per sample (ν)

We know that $\delta = \frac{2}{q}$ for normalized signal. Here peak amplitude is 1 V for normalized signal. Here 1 % of 1 V is 0.01 V. Thus maximum acceptable error (ε_{max}) is 1 % or 0.01 V. For uniform quantization,

$$\varepsilon_{max} = \left|\frac{\delta}{2}\right|$$

$$0.01 = \frac{\delta}{2} \qquad \text{Here } \varepsilon_{max} = 0.01 \text{ is maximum acceptable error}$$

or $\quad \delta = 0.02 \quad$ And $\quad \delta = \frac{2}{q}$

$$\therefore \quad q = \frac{2}{\delta} = \frac{2}{0.02} = 100 \text{ levels.}$$

And $\quad q = 2^\nu$

or $\quad \nu = \log_2 q = \frac{\log_{10} q}{\log_{10} 2} = \frac{\log_{10} 100}{\log_{10} 2} = 6.643$

since ν is integer, $\nu \approx 7$ bits.

...num bandwidth

PCM, $B_T = \frac{1}{2} \nu f_s = \frac{1}{2} \times 7 \times 12000 = 42$ kHz

... The information in an analog waveform, whose maximum ...uency $f_m = 4000$ Hz is to be transmitted using a 16-level PAM ...em. The quantization distortion must not exceed ± 1 % of the ... to peak analog signal.

What is the minimum number of bits per sample that should be used in the transmission system ?

What is the minimum required sampling rate and bit rate of the system.

What is the 16-ary PAM symbol transmission rate ?

[SPPU : Dec.-11, Marks 8]

...: **a) Minimum number of bits** (ν) **:** The quantization distortion is ... For normalized signal, the amplitude is 1 V. Hence 1 % of 1 V will ...0.01 V.

...s, $\quad \varepsilon_{max} = \pm 0.01\, V$

...ce $\quad \varepsilon_{max} = \left|\frac{\delta}{2}\right|$, we have $\left|\frac{\delta}{2}\right| = |\pm 0.01|$

$$\delta = 0.02 = \frac{2}{q}$$

$$q = \frac{2}{\delta} = \frac{2}{0.02} = 100 \text{ levels.}$$

$$q = 2^{\nu}$$

$$\therefore \nu = \log_2 q = \frac{\log_{10} q}{\log_{10} 2} = \frac{\log_{10} 100}{\log_{10} 2} = 6.643$$

$$\nu = 6.643 \approx 7 \text{ bits.}$$

Sampling rate and bit rate (f_s and r)

Here $W = f_m = 4000$ Hz.

$$f_s = 2W = 2 \times 4000 = 8000 \text{ Hz.}$$

Bit rate or signaling rate, $r = \nu f_s = 7 \times 8000 = 56$ kbps

Symbol transmission rate

...e symbol or samples are taken at 8000 samples/sec.

...ence symbol rate = 8000 symbols/sec.

5.3 : Non Uniform Quantization and Companding (A-law and μ-law)

Important Points to Remember

- Companding uses μ-law and A-law for companding.
- Signal to noise ratio for μ-law and A-law companding are given as follows :

$$\left(\frac{S}{N}\right)_{\mu} = \frac{3q^2}{[(ln(1+\mu)]^2}, \quad \left(\frac{S}{N}\right)_{A} = \frac{3q^2 \cdot A^2}{[1+lnA]^2}$$

Q.17 What is non-uniform quantization and describe A - Law and μ - Law. **[SPPU : May-11, 12, Marks 8, Dec.-22, Marks 6]**

Ans. : Non-uniform quantization and companding

Definition of non-uniform quantization : The step size ' δ' is not fixed. It varies according to changes in the amplitude of message signal. It helps in reducing quantization error.

μ - Law companding for speech signals

Normally for speech and music signals a μ - law compression is used. This compression is defined by the following equation,

$$Z(x) = (\text{Sgn } x) \frac{ln(1+\mu|x|)}{ln(1+\mu)} \quad |x| \le 1 \qquad \text{... (Q.17.1)}$$

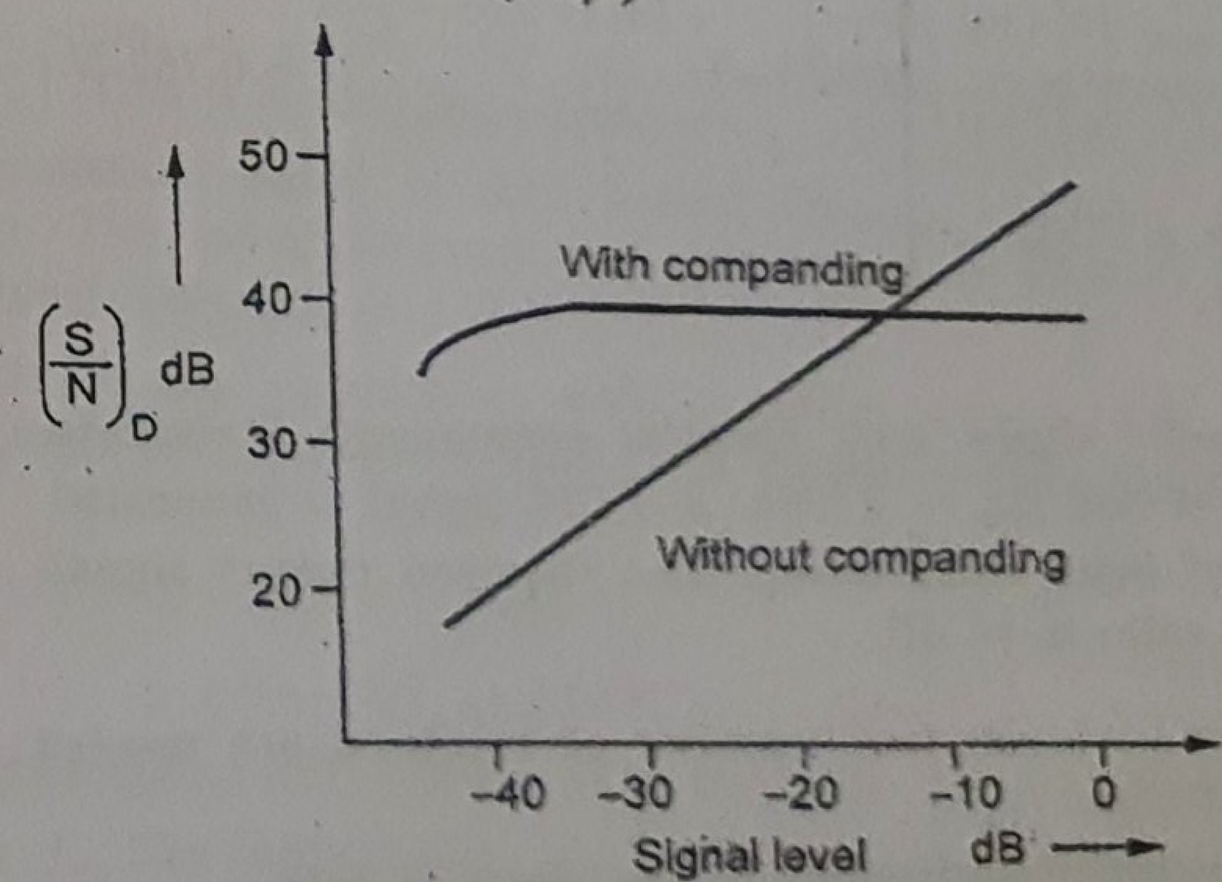


Fig. Q.17.1 PCM performance with μ-law companding

Fig. Q.17.1 shows the variation of signal to noise ratio with respect to signal level without companding and with companding.

It can be observed from above figure that signal to noise ratio of PCM remains almost constant with companding.

A-Law for companding

The A-law provides piecewise compressor characteristic. It has linear segment for low level inputs and logarithmic segment for high level inputs. It is defined as,

$$\therefore \quad Z(x) = \begin{cases} \dfrac{A|x|}{1+\ln A} & \text{for } 0 \le |x| \le \dfrac{1}{A} \\ \dfrac{1+\ln(A|x|)}{1+\ln A} & \text{for } \dfrac{1}{A} \le |x| \le 1 \end{cases} \quad \dots \text{(Q.17.2)}$$

When A = 1, we get uniform quantization. The practical value for A is 87.56. Both A-law and μ-law companding is used for PCM telephone systems.

Q.18 The input voltage of a compander with a maximum voltage range of 1 volt and μ of 255 is 0.25. What are the output voltage and gain ?

☞ **[SPPU : May-11, Marks 6]**

Ans. : Here $\mu = 255$, $V_i = 0.25$ V, $V_{max} = 1$ V

Output of μ-law compander is given as,

$$V_o = \frac{V_{max} \cdot \ln\left[1+\mu \dfrac{V_i}{V_{max}}\right]}{\ln[1+\mu]} = \frac{1 \cdot \ln\left[1+255 \cdot \dfrac{0.25}{1}\right]}{\ln[1+255]} = 0.752 \text{ V}$$

$$\therefore \quad \text{Gain} = \frac{V_o}{V_i} = \frac{0.752}{0.25} = 3$$

Q.19 Consider an audio signal with spectral components limited to the frequency band of 500 Hz to 3 kHz. A PCM signal is generated with sampling rate of 8000 samples/sec. The required output signal to quantisation noise ratio is 40 dB.

i) How many number of levels and number of bits/level are needed for uniform quantisation ?

ii) Calculate the bandwidth requirement of the above system.

iii) If A-law compander is used, what will be the changes in the number of levels, number of bits/sample and bandwidth ?

☞ **[SPPU : May-12, Marks 10, Dec.-16, Marks 8]**

Ans. : Here $f_s = 8000$ Hz

and $\left(\dfrac{S}{N}\right)_{dB} = 40$ dB.

$$\therefore \quad 40 = 10 \log_{10} \frac{S}{N} \Rightarrow \frac{S}{N} = 10^4 = 10{,}000$$

i) Number of levels (q) and number of bits (v)

For audio signal, $\left(\dfrac{S}{N}\right)_{dB} = 4.8 + 6v$

$$40 = 4.8 + 6v$$

$$\therefore \quad v = 5.86 \approx 6 \text{ bits.}$$

$$\text{And} \quad q = 2^v = 64 \text{ levels.}$$

ii) Bandwidth (B_T)

$$B_T = \frac{1}{2} v f_s = \frac{1}{2} \times 6 \times 8000 = 24 \text{ kHz}$$

iii) Number of levels (q), bits (v) and bandwidth (B_T) for A-law companding

Here value of A is not specified. Hence assume $A = 87.56$ (which is commonly used).

$$\therefore \quad \left(\frac{S}{N}\right) = \frac{3q^2A^2}{(1+\ln A)^2}$$

Putting for $\dfrac{S}{N}$ and A,

$$10{,}000 = \frac{3q^2(87.56)^2}{(1+\ln 87.56)^2}$$

$$\therefore \quad q = 3.6 \approx 4 \text{ levels}$$

$$v = \log_2 q = \frac{\log_{10} q}{\log_{10} 2} = \frac{\log_{10} 4}{\log_{10} 2} = 2 \text{ bits}$$

$$B_T = \frac{1}{2} v f_s = \frac{1}{2} \times 2 \times 8000 = 8000 \text{ Hz}$$

5.4 : Delta Modulation (DM & ADM)

Important Points to Remember

- Delta modulation uses only one bit per sample. This bit indicates whether 'δ' is positive or negative. The step size (δ) must be as small as possible.
- The sampling duration T_s must be as small as possible.
- Slope overload distortion will occur if amplitude (A_m) of the signal,

$$A_m > \frac{\delta}{2\pi f_m T_s}$$

- In delta modulation, signal to quantization noise power ratio is given as,

$$\frac{S}{N} = \frac{3}{8\pi^2 W \cdot f_m^2 T_s^3}$$

- Slope overload distortion and granular noise can be eliminated by delta sigma modulation or adaptive delta modulation.

Draw the block diagram of DM transmitter and receiver and explain its working. Comment on the drawbacks of DM. Also mention its advantages. ☞ **[SPPU : Dec.15, Marks 6, Aug.-17, June-22, Oct.-18, Marks 6, May-16, Marks 5]**

Ans. : DM principle : Delta modulation transmits only one bit per sample. This one bit indicates whether the sample amplitude is to be increased or decreased. The step size is fixed.

DM transmitter : Fig. Q.20.1 shows the block diagram of DM transmitter.

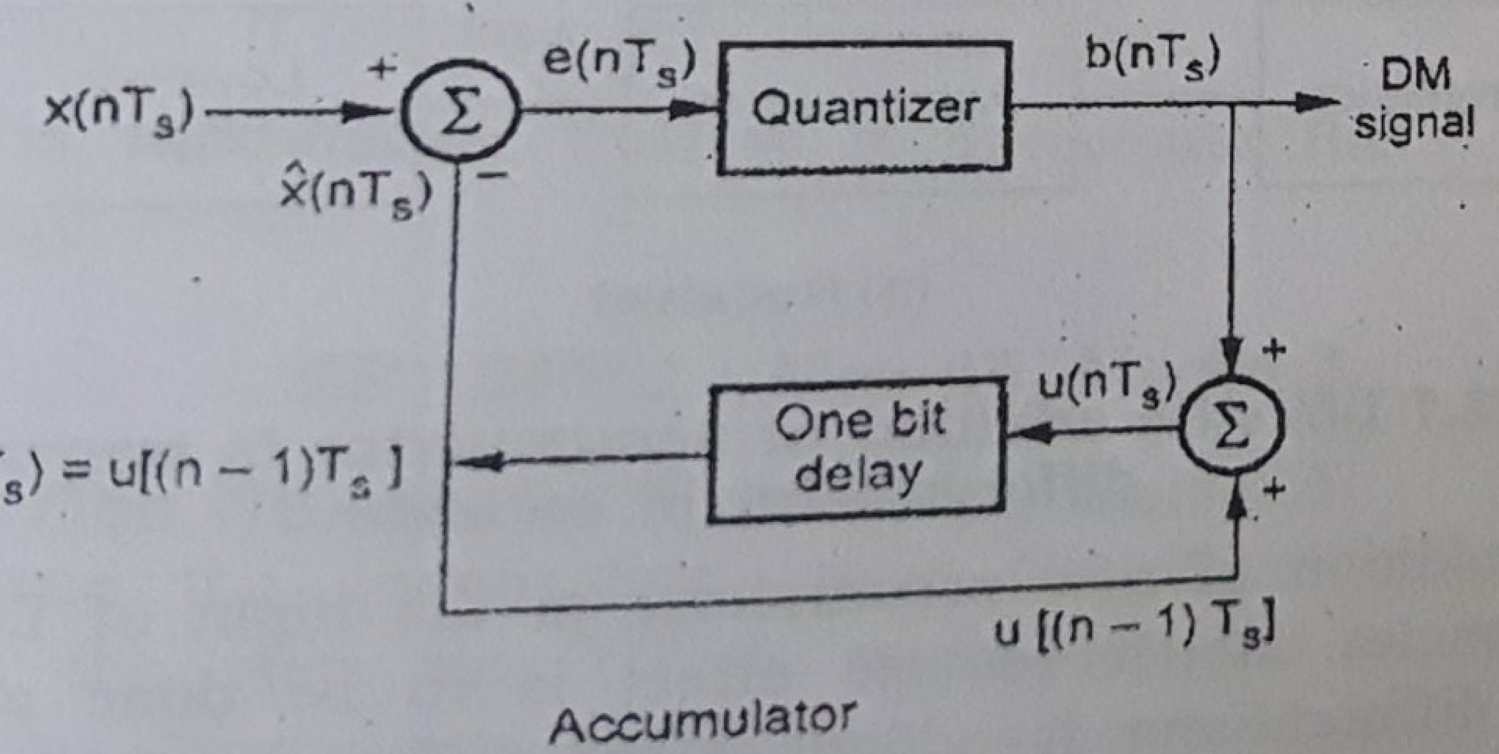


Fig. Q.20.1 DM transmitter

- $x(nT_s)$ is the input signal and $\hat{x}(nT_s)$ is the reconstructed signal.
- Error $e(nT_s)$ is the difference between input signal $x(t)$ and reconstructed signal $\hat{x}(nT_s)$.
- One bit quantizer provides $b(nT_s) = +\delta$ if the error is positive and $b(nT_s) = -\delta$ if the error is positive. This is nothing but DM signal.
- One bit delayed reconstructed signal $\hat{x}(nT_s) = u[(n-1)T_s]$ is added with the DM signal. This provides the latest reconstructed signal. This is accumulator operation.

DM receiver

- Fig. Q.20.2 shows the block diagram of DM receiver.

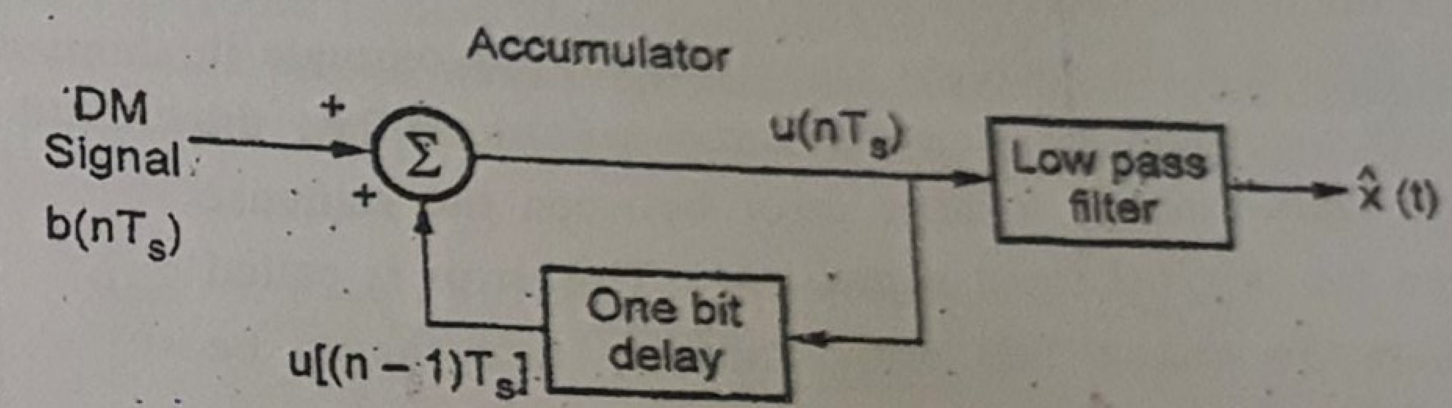


Fig. Q.20.2 DM receiver

- The DM signal is added with one bit delayed reconstructed signal $u[(n-1)T_s]$. This is accumulator operation.
- The reconstructed signal is then passed through low pass filter for smoothing. The cutoff frequency of this low pass filter is equal to highest signal frequency in $x(t)$.

The summer in the accumulator adds quantizer output $(\pm\delta)$ with the previous sample approximation. This gives present sample approximation. i.e.,

$$u(nT_s) = u(nT_s - T_s) + [\pm\delta]$$
$$= u[(n-1)T_s] + b(nT_s) \qquad \text{.... (Q.20.1)}$$

- The staircase signal $u(T_s)$ is passed through the low pass filter of cutoff frequency f_c. The reconstructed signal $\hat{x}(t)$ is close to $x(t)$.

Advantages of DM

i) Transmits only one bit per sample.

ii) Signalling rate and channel bandwidth is reduced.

iii) No A/D or D/A converters are required.

Disadvantages

i) Slope overload distortion arises because reconstructed signal could not follow actual signal.

ii) Granular noise occurs because of too large step size.

Q.21 Explain slope overload distortion and granular noise in delta modulation.

☞ [May-15, Marks 3, May-16, Marks 3, Dec.-22, Marks 6]

Ans. : Slope overload distortion (Startup error) : The rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it, the step size 'δ' becomes too small for staircase signal $u(t)$ to follow the steep segment of $x(t)$. Thus there is a large error between the staircase approximated signal and the original input signal $x(t)$. This error is called *slope overload distortion.* To reduce this error, the step size should be increased when slope of signal of $x(t)$ is high.

Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore this modulator is also called Linear Delta Modulator (LDM).

Granular noise (Hunting) : Granular noise occurs when the step size is too large compared to small variations in the input signal. That is for very small variations in the input signal, the staircase signal is changed by large amount (δ) because of large step size. When the input signal is almost flat, the staircase signal $u(t)$ keeps on oscillating by $\pm\delta$ around the signal. The error between the input and approximated signal is called *granular noise.* The solution to this problem is to make step size small. Thus large step size is required to accommodate wide dynamic range of the input signal (to reduce slope overload distortion) and small steps are required to reduce granular noise. Adaptive delta modulation is the modification to overcome these errors.

Q.22 Explain delta sigma modulator with the help of block diagrams. How it eleminates the accumulation of noise that was present in DM ?

☞[SPPU : Dec.-10, Marks 4, May-15, Marks 3, Oct.-16, Marks 5, Dec.-17, Marks 7]

Ans. :

- We know that delta modulator calculates the difference between input signal and predicted signal. This operation is equivalent to evaluating derivative of input signal. Hence following problems are present in delta modulation.
  - i) The noisy data can cause cumulative errors in demodulated signal.
  - ii) If the signal has DC component, then errors are created in demodulation.
- Above problems can be eliminated by putting an integrator or accumulator before the DM. Fig. Q.22.1 shows the DM.

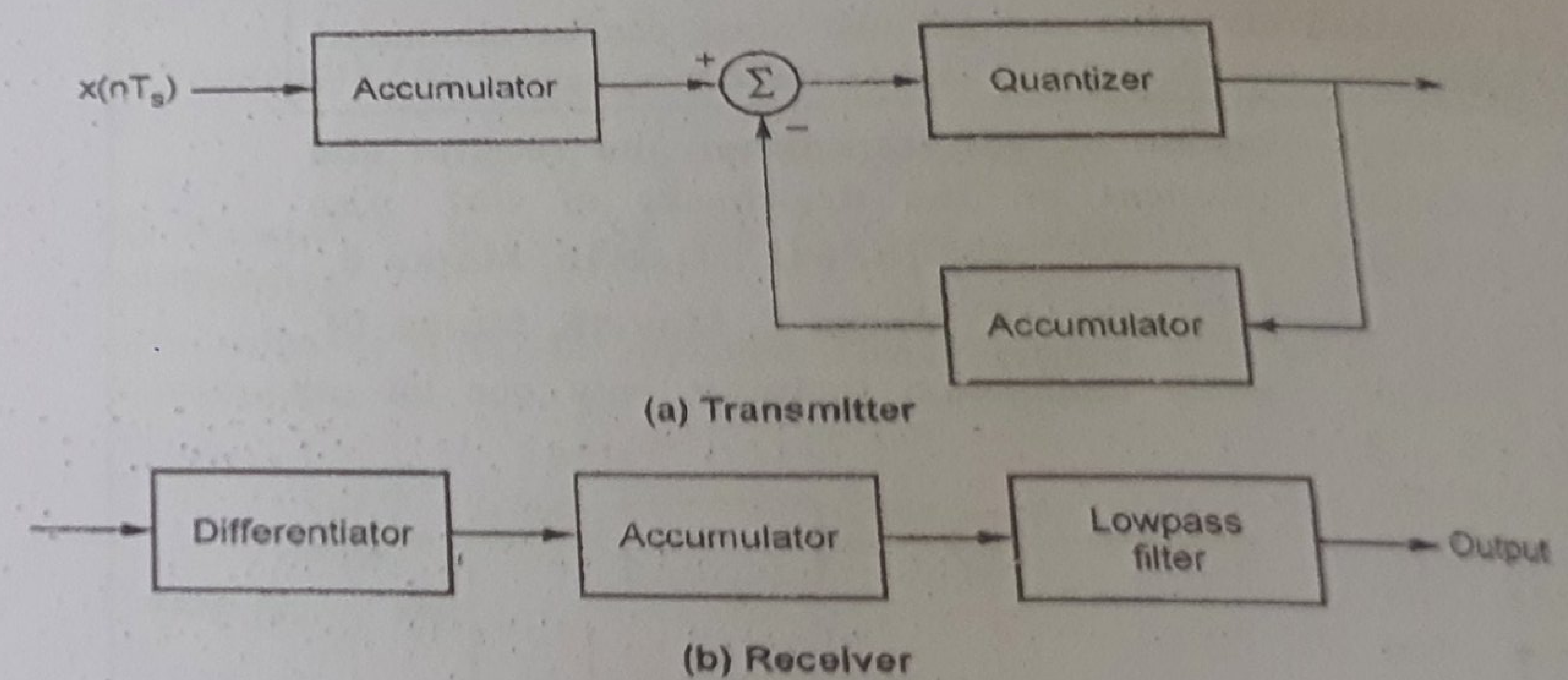


Fig. Q.22.1 DM with addition of accumulator in transmitter and differentiator in receiver

- Now the addition of new accumulator at the input of DM emphasizes low frequencies. Hence reverse effect is to be done at the receiver. Therefore differentiator is added at the receiver of DM as shown in Fig. Q.22.1 (b).

help of block diagrams.
at was present in DM ?
4, May-15, Marks 3,
5, Dec.-17, Marks 7]

difference between input equivalent to evaluating ems are present in delta

errors in demodulated

errors are created in

g an integrator or DM.

Output

nitter and

1 emphasizes the receiver. as shown in

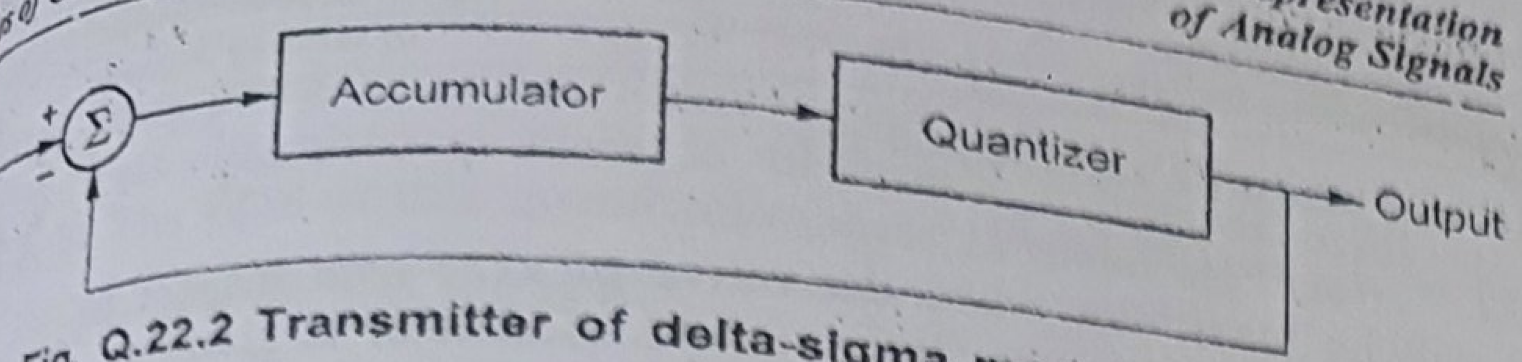


Fig. Q.22.2 Transmitter of delta-sigma modulation

Q.22.1 (a) note that both the inputs to summer are comming ugh accumulators. Hence single accumulator can be placed after mer. Fig. Q.22.2 shows this. It is the block diagram of delta-sigma ulation. Sigma stands for integration or accumulation. In the above k diagram observe that only error signal is accumulated.

consider the block diagram of receiver of Fig. Q.22.1 (b). The ations performed by differentiator and accumulator cancel each . Hence the receiver will simply contain lowpass filter as shown in Q.22.3.

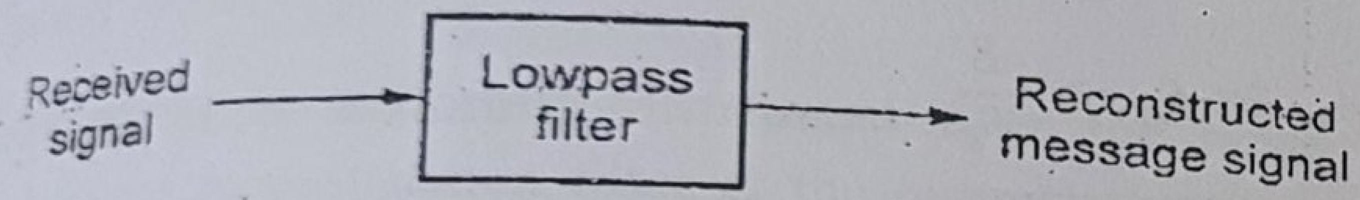


Fig. Q.22.3 Receiver of delta-sigma modulation

ntages of delta-sigma modulation :

) Due to integration, low frequencies of input signal are pre-emphasized.

) The correlation between adjacent samples is increased. This reduces the variance of error.

) The receiver is most simple. It contains only lowpass filter.

What is adaptive delta modulation ? How it overcomes the ations of delta modulation ?

☞ [SPPU : May-15, Marks 3,

☞ May-16, Marks 3, Dec.-16, Marks 8, June-22, Marks 6]

: Principle : To overcome the quantization errors due to slope load and granular noise, the step size (δ) is made adaptive to ations in the input signal $x(t)$. Particularly in the steep segment of the l $x(t)$, the step size is increased. When the input is varying slowly,

the step size is reduced. Then the method is called *Adaptive Delta Modulation (ADM)*.

Transmitter : Fig. Q.23.1 (a) shows the block diagram of ADM transmitter. Note the step size controller in the block diagram. It controls the step size according to certain rule depending upon one bit quantizer output.

For example if one bit quantizer output is high (1), then step size may be doubled for next sample. If one bit quantizer output is low, then step size may be reduced by one step. Fig. Q.23.2 shows the waveforms of adaptive delta modulator and sequence of bits transmitted.

Receiver : In the receiver of adaptive delta modulator shown in Fig. Q.23.1 (b) the first part generates the step size from each incoming bit. Exactly the same process is followed as that in transmitter. The previous input and present input decides the step size. It is then given to an accumulator which builds up staircase waveform. The lowpass filter then smoothens out the staircase waveform to reconstruct the smooth signal.

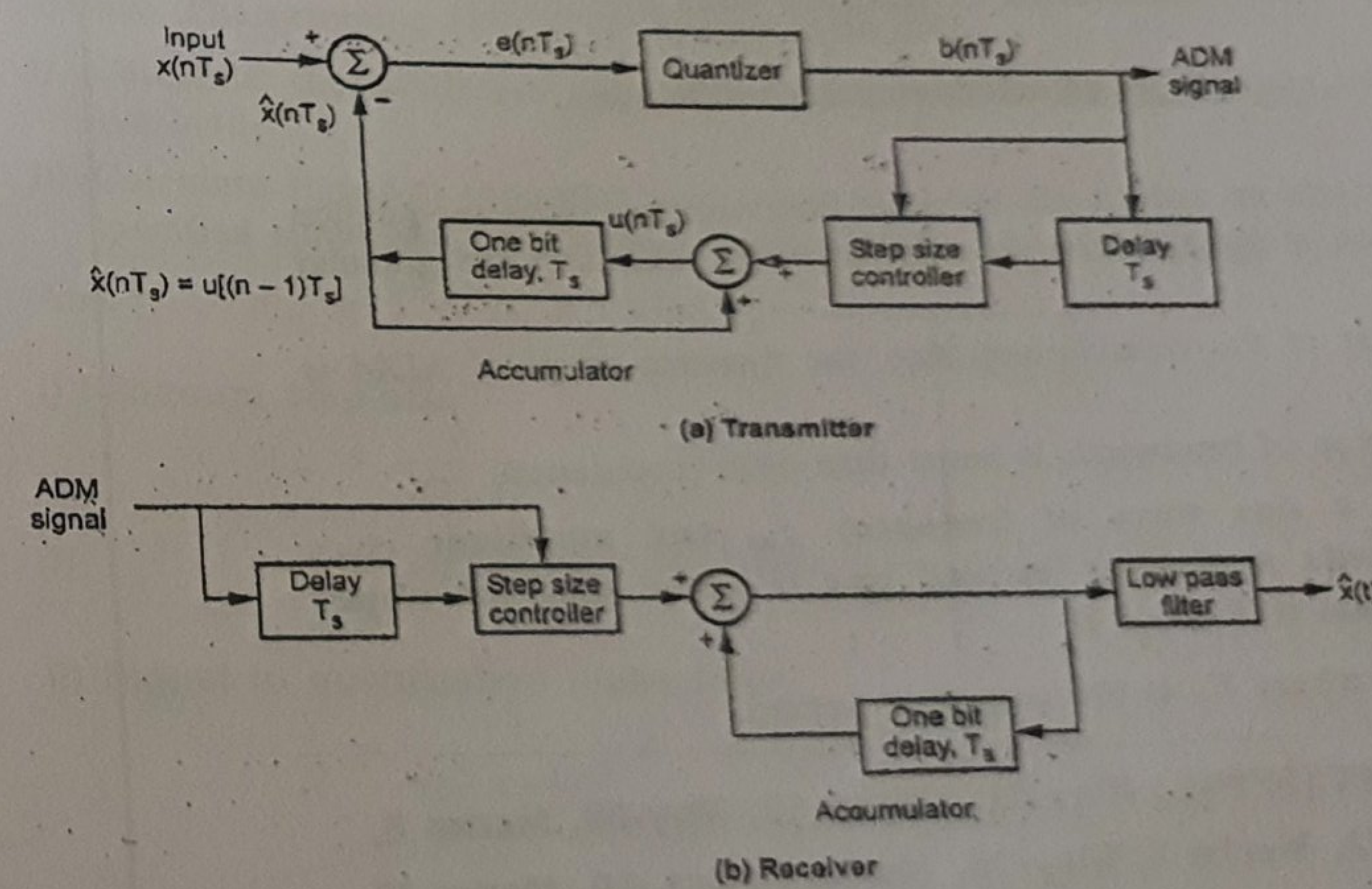


Fig. Q.23.1 Adaptive delta modulator

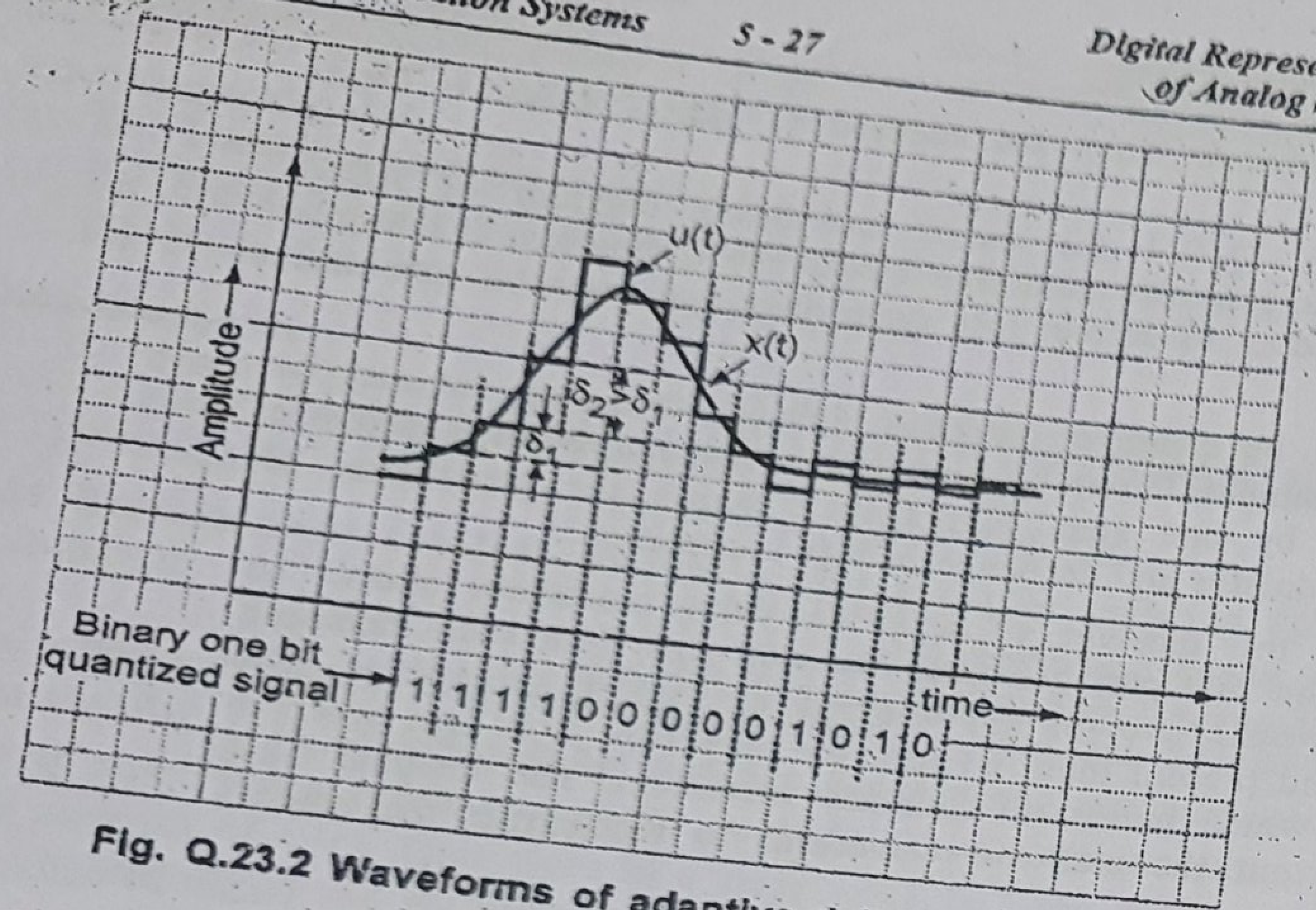


Fig. Q.23.2 Waveforms of adaptive delta modulation

Q.24 State the advantages of adaptive delta modulation.

Ans. :

1. The signal to noise ratio is better than ordinary delta modulation because of the reduction in slope overload distortion and granular noise.
2. Because of the variable step size, the dynamic range of ADM is wide.
3. Utilization of bandwidth is better than delta modulation.

Q.25 Consider a sine wave of frequency f_m and amplitude A_m applied to a delta modulator of step size δ. Show that the slope overload distortion will occur if

$A_m > \frac{\delta}{2\pi f_m T_s}$ **where T_s is the sampling period.**

☞[SPPU : May-01, Marks 10; May-09, Marks 5, Dec.-12, Marks 8, May-15, Marks 7, Oct.-16, Marks 5]

OR Derive the condition to avoid slope overload error in delta modulation.

☞[SPPU : May-12, Marks 4]

Ans. : Let the sine wave be represented as,

$$x(t) = A_m \sin(2\pi f_m t)$$

Slope of $x(t)$ will be maximum when derivative of $x(t)$ with respect to 't' will be maximum. The maximum slope of delta modulator is given as,

$$\text{Maximum slope} = \frac{\text{Step size}}{\text{Sampling period}} = \frac{\delta}{T_s} \quad \text{... (Q.25.1)}$$

Slope overload distortion will take place if slope of sine wave is greater than slope of delta modulator i.e.

$$\max\left|\frac{d}{dt} x(t)\right| > \frac{\delta}{T_s}$$

$$\max\left|\frac{d}{dt} A_m \sin(2\pi f_m t)\right| > \frac{\delta}{T_s}$$

$$\max\left|A_m\, 2\pi f_m \cos(2\pi f_m t)\right| > \frac{\delta}{T_s}$$

$$A_m\, 2\pi f_m > \frac{\delta}{T_s}$$

or

$$\boxed{A_m > \frac{\delta}{2\pi f_m T_s}} \quad \text{... (Q.25.2)}$$

Q.26 Derive an expression for signal to quantization noise power ratio for delta modulation. Assume that no slope overload distortion exists.

☞[SPPU : May-07,12,14, Dec.-09, Marks 8, May-12, Marks 5]

Ans. : i) To obtain signal power :

- Slope overload, distortion will not occur if $A_m \le \frac{\delta}{2\pi f_m T_s}$. Hence maximum signal amplitude will be $A_m = \frac{\delta}{2\pi f_m T_s}$.

Normalized signal power is given as,

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

$$P = \frac{\delta^2}{8\pi^2 f_m^2 T_s^2}, \quad \text{with } A_m = \frac{\delta}{2\pi f_m T_s} \quad \text{... (Q.26.1)}$$

This is an expression for signal power in delta modulation.

...obtain noise power : The maximum quantization error is equal to ...size δ. This quantization error is uniformly distributed over an ...al $[-\delta, \delta]$. The PDF of this quantization error is given as,

$$f_{\varepsilon}(\varepsilon) = \frac{1}{2\delta} \text{ for } -\delta \le \varepsilon \le \delta$$

...an square value of this quantization error is given as,

$$E[\varepsilon^2] = \int_{-\infty}^{\infty} \varepsilon^2 f_{\varepsilon}(\varepsilon)\, d\varepsilon = \int_{-\delta}^{\delta} \varepsilon^2 \cdot \frac{1}{2\delta}\, d\varepsilon = \frac{1}{2\delta}\left[\frac{\varepsilon^3}{3}\right]_{-\delta}^{\delta}$$

$$= \frac{1}{2\delta}\left[\frac{\delta^3}{3} + \frac{\delta^3}{3}\right] = \frac{\delta^2}{3}$$

...quantization error has zero mean value. Hence normalized noise ...er is given as,

$$\text{...ise power} = E[\varepsilon^2] = \frac{\delta^2}{3}$$

...s noise power is uniformly distributed over the range $[-f_s, f_s]$. ...re is lowpass reconstruction filter at the output. The filter has cut-off ...uency of 'W'. Hence output noise power in the range (–W, W) will

...t noise power,

$$\frac{W}{f_s} \times \text{Noise power} = \frac{W}{f_s} \times \frac{\delta^3}{3} = \frac{WT_s\delta^2}{3}, \quad \text{since } f_s = \frac{1}{T_s} \qquad \text{... (Q.26.2)}$$

...ignal to noise power ratio

...signal to noise power ratio is given as,

$$\frac{S}{N} = \frac{P}{N} = \frac{\delta^2/(8\pi^2 f_m^2 T_s)}{(WT_s\delta^2)/3} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$$

... A delta modulator system is designed to operate 5 times the ...uist rate for a signal with 3 kHz bandwidth. Determine the ...imum amplitude of 1.2 kHz input sinusoid for which a delta ...ulator does not have slope overload. Quantizing step size is ...mV. Derive the expression used. (Refer Q.26 of Chapter-5)

☞[SPPU : Dec.-14, May-17, Marks 6,

☞Aug.-17, Oct.-18, Marks 4, May-18, Marks 8]

Ans. :

Bandwidth, W = 3 kHz, f_m = 1.2 kHz

Nyquist rate = $2 \times f_m = 2 \times 1.2$ kHz = 2.4 kHz

Sampling frequency,

f_s = 5 × Nyquist rate = 5 × 2.4 kHz = 12 kHz

Step size, δ = 250 mV

$$T_s = \frac{1}{f_s} = \frac{1}{12\times10^3}$$

$$A_m \le \frac{\delta}{2\pi f_m T_s} = \frac{250\times10^{-3}}{2\pi \times 1.2\times10^3 \times \frac{1}{12\times10^3}} = 0.398 \text{ V}$$

Q.28 Consider a DM system designed to accomodate 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.

i) Calculate the minimum step size Δ required to minimize slope overload.

ii) Calculate signal-to (quantization) noise ratio of the system for the specified sinusoidal test signal. ☞[SPPU : May-13, Marks 8]

Ans. : $A_m = 1\text{ V}, f_m = 1\text{ kHz}, T_s = \frac{1}{f_s} = \frac{1}{50\times10^3}$

i) Minimum step size

$$A_m < \frac{\delta}{2\pi f_m T_s}$$

or

$$\delta \ge 2\pi f_m T_s A_m \ge 2\pi\times1000\times\frac{1}{50\times10^3}\times1 = 0.125 \text{ V}$$

ii) Signal to quantization noise ratio

$$\frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3} = \frac{3}{8\pi^2\times5000\times(1000)^2\times\left(\frac{1}{50\times10^3}\right)^3}$$

$$= 949.88 = 10\log_{10} 949.88 = 29.77 \text{ dB}$$

Q.29 A signal having bandwidth 3 kHz is to be encoded using
i) 8 bit PCM ii) DM system
If 10 cycles of the signal are digitized, state how many bits will there in digitized, output in each case if sampling frequency is 10 kHz. Also find bandwidth required in each case.

[SPPU : Dec.-10, Marks 8, Oct.-16, Marks 5]

Ans. : Given : f_s = 10,000 samples/sec
PCM transmits ν = 8 bits/sample DM transmits 1 bit/sample
Here W = 3 kHz. Hence period of one cycle will be,

$$T = \frac{1}{3000}$$

Hence time for 10 cycles will be,

$$T_{10} = 10 \times T = \frac{10}{3000} = \frac{1}{300} \text{ sec}$$

i) 8-bit PCM

Signaling rate in 8-bit PCM will be,

$$r = \nu f_s = 8 \text{ bits/sample} \times 10{,}000 \text{ samples/sec} = 80{,}000 \text{ bits/sec.}$$

$$\therefore \quad \text{Bandwidth, } B_T = \frac{1}{2} r = \frac{1}{2} \times 80{,}000 = 40 \text{ kHz.}$$

Number of bits in 10 cycles (or $T_{10} = \frac{1}{300}$ sec will be),

$$\text{Number of bits} = r \times T_{10} = 80{,}000 \text{ bits/sec} \times \frac{1}{300} \text{ sec}$$

$$= 266.67 \approx 267 \text{ bits.}$$

ii) DM system

Signaling rate in DM system will be,

$$r = f_s = 10{,}000 \text{ bits/sec}$$

$$\therefore \text{ Bandwidth, } B_T = \frac{1}{2} r = \frac{1}{2} \times 10{,}000 = 5 \text{ kHz.}$$

$$\text{Number of bits} = r \times T_{10} = 10{,}000 \text{ bits/sec} \times \frac{1}{300} \text{sec}$$

$$= 33.33 \approx 34 \text{ bits.}$$

5.5 : Differential Pulse Code Modulation (DPCM)

Important Points to Remember

i) It works on the principle of production. The value of present sample is predicted from the past samples.

ii) The prediction may not be exact, but it is very close to the actual simple value.

Q.30 Explain with neat schematic and mathematical analysis the transmitter and receiver of DPCM.

[SPPU : Dec.-14, Marks 8, June-22, Marks 6]

Ans. : DPCM Transmitter

- Fig. Q.30.1 shows the transmitter of DPCM system.

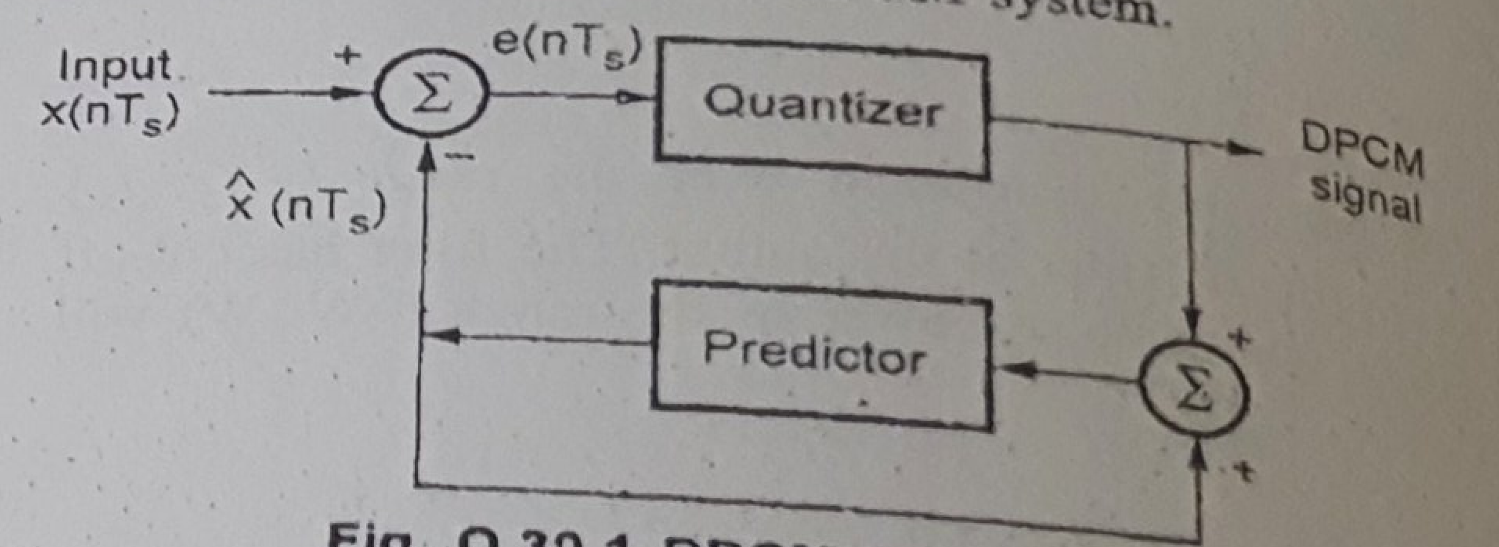


Fig. Q.30.1 DPCM transmitter

- The difference between input signal $x(nT_s)$ and reconstructed signal $\hat{x}(nT_s)$ is calculated.
- This difference $e(nT_s)$ is quantized by the quantizer.
- A predictor generator present estimate of the signal from quantized difference and previous estimates.
- Error, $e(nT_s) = x(nT_s) - \hat{x}(nT_s)$

$$\hat{x}(nT_s) + e(nT_s) = x(nT_s) \qquad \text{...(Q.30.1)}$$

- $\nu(nT_s)$ is quantized version of $e(nT_s)$. Hence,

$$\nu(nT_s) = e(nT_s) + q(nT_s) \qquad \text{...(Q.30.2)}$$

Here $q(nT_s)$ is quantization error

- Predictor input $u(nT_s)$ is given by,

$$u(nT_s) = \hat{x}(nT_s) + \nu(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$

...ation (Q.30.2)

$= x(nT_s) + q(nT_s)$ from eq (Q.30.1)

(nT_s) is quantized version of the input $x(nT_s)$.

... receiver

... Q.30.2 shows the block diagram of DPCM receiver.

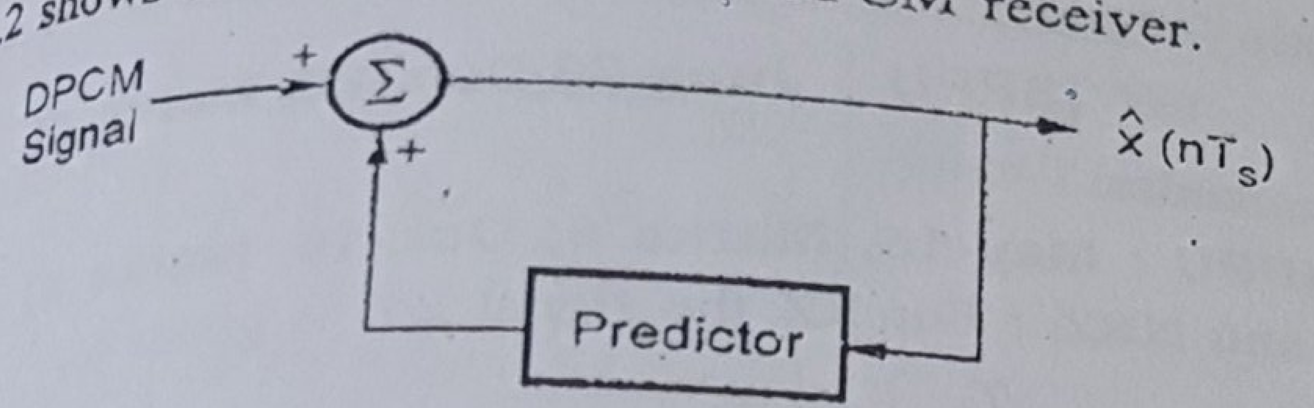


Fig. Q.30.2 DPCM receiver

... predictor generates the present reconstructed signal $\hat{x}(nT_s)$ from its ...ous values and incomming DPCM signal.

... reconstructed signal is more accurate since predictor considers ...plete history of the reconstructed signal $\hat{x}(nT_s)$.

...e receiver output,

$$e(nT_s) + \hat{x}(nT_s) = x(nT_s)$$

Compare various source coding methods.

☞ **[SPPU : May-18, Dec.-18,22, Marks 6]**

	Parameter	PCM	Delta Modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
	Number of bits	It can use 4, 8 or 16 bits per sample.	It uses only one bit for one sample.	Only one bit is used to encode one sample.	Bits can be more than one but are less than PCM.
2.	Levels, step size	The number of levels depend on number of bits. Level size is fixed.	Step size is fixed and cannot be varied.	According to the signal variation, step size varies (Adapted).	Fixed number of levels are used.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise is present.	Quantization error is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4.	Bandwidth of transmission channel	Highest bandwidth is required since number of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is lower than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6.	Complexity of notation	System is complex.	Simple.	Simple.	Simple.

END...

Unit VI

6 Baseband Digital Transmission

6.1 : Line Codes : Properties and Spectrum

Important points to remember

- A digital data is represented by different formats or waveforms. It is known as line coding.
- Various formats like unipolar Rz_0 and NRZ, polar RZ and NRZ, Bipolar NRZ, split phase manchestor, M - any coding are available.

Q.1 What are line codes ? State the desirable properties of line codes. [SPPU : Dec-14, (In sem), Dec.-13,18, Marks 4, May-18, Dec.-22, Marks 6]

Ans. : Line codes : The digital data obtained by PCM, DM, ADM, DPCM etc are represented by different waveforms or data formats. These are basically different electrical levels to represent the digital symbols. These are called line codes.

Desirable properties of line codes :

1. The line code should have adquate timing content, so that clock information can be extracted from the waveform.
2. The line code should be immune to channel noise and interference.
3. The line code should allow error detection and correction.
4. The power spectrum of line code should be matched to that of channel to reduce signal distortion.

5. The waveform of the line code should be transparent to the digital data being transmitted.

Q.2 Explain various data formats. [SPPU : June-22,Dec.-22, Marks 6]

OR Explain various line codes. [SPPU : May-15, Marks 6, Oct.-18, Marks 4]

Ans. : 1. Unipolar RZ and NRZ : For RZ the signal $x(t)$ is given as,

$$x(t) = \begin{cases} V & for\ 0 \le t \le \frac{T_b}{2} \\ 0 & for\ \frac{T_b}{2} \le t \le T_b \end{cases} \text{ when '1' is transmitted}$$

and $x(t) = 0$ for $0 \le t \le T_b$ When '0' is transmitted

For unipolar NRZ,

$$x(t) = \begin{cases} A & for\ 0 \le t \le T_b & \text{When '1' is transmitted} \\ 0 & for\ 0 \le t \le T_b & \text{When '0' is transmitted} \end{cases}$$

This format carries average DC valve

2. Polar RZ and NRZ : For polar RZ, x(t) is given as,

$$x(t) = \begin{cases} V/2 & for\ 0 \le t \le \frac{T_b}{2} \\ 0 & for\ \frac{T_b}{2} \le t \le T_b \end{cases} \text{ When '1' is transmitted}$$

and
$$x(t) = \begin{cases} -V/2 & for\ 0 \le t \le \frac{T_b}{2} \\ 0 & for\ \frac{T_b}{2} \le t \le T_b \end{cases} \text{ When '0' is transmitted}$$

For polar NRZ,

$$x(t) = \begin{cases} +V/2 & for\ 0 \le t \le T_b & \text{When '1' is transmitted} \\ -V/2 & for\ 0 \le t \le T_b & \text{When '0' is transmitted} \end{cases}$$

This format does not carry average DC value.

... NRZ (Pseudo - trinary or Alternate Mark Inversion) AMI

... 1's are represented by pulses with alternate polarity and 0's ... by no pulses.

... doesnot have DC components.

... phase manchester : This format is represented as,

$$x(t) = \begin{cases} V/2 & for\ 0 \le t \le \frac{T_b}{2} \\ -V/2 & for\ \frac{T_b}{2} \le t \le T_b \end{cases} \text{ When '1' is transmitted}$$

$$x(t) = \begin{cases} -V/2 & for\ 0 \le t \le \frac{T_b}{2} \\ V/2 & for\ \frac{T_b}{2} \le t \le T_b \end{cases} \text{ When '0' is transmitted}$$

...mat also doesnot have DC component.

... Quaternary NRZ : There are 4 voltage levels. The bits are ...d in the blocks of two. The voltage levels are $\frac{-3V}{2}$ for 00, $-\frac{V}{2}$ for ... for 10 and $\frac{3V}{2}$ for 11 combination of bits. This format reduces the ...ng rate.

...y coding : The messages are first gray coded and polar quaternary encoding is done. Table Q.2.1 shows the message bits, their gray and alloted voltage levels.

Message bits	00	01	10	11
Gray code	00	01	11	10
$x(t)$	$\frac{-3V}{2}$	$-V/2$	$3V/2$	$V/2$

Table Q.2.1

...-ary coding : In M-ary coding $k = \log_2 M$ successive bits are ...ined. Hence there are 'M' levels. These M - levels are encoded in the ...ar way as in polar qnaternary NRZ.

Q.2.1 shows the waveforms of all the line codes for message string 1 0 1 0 1 1 0.

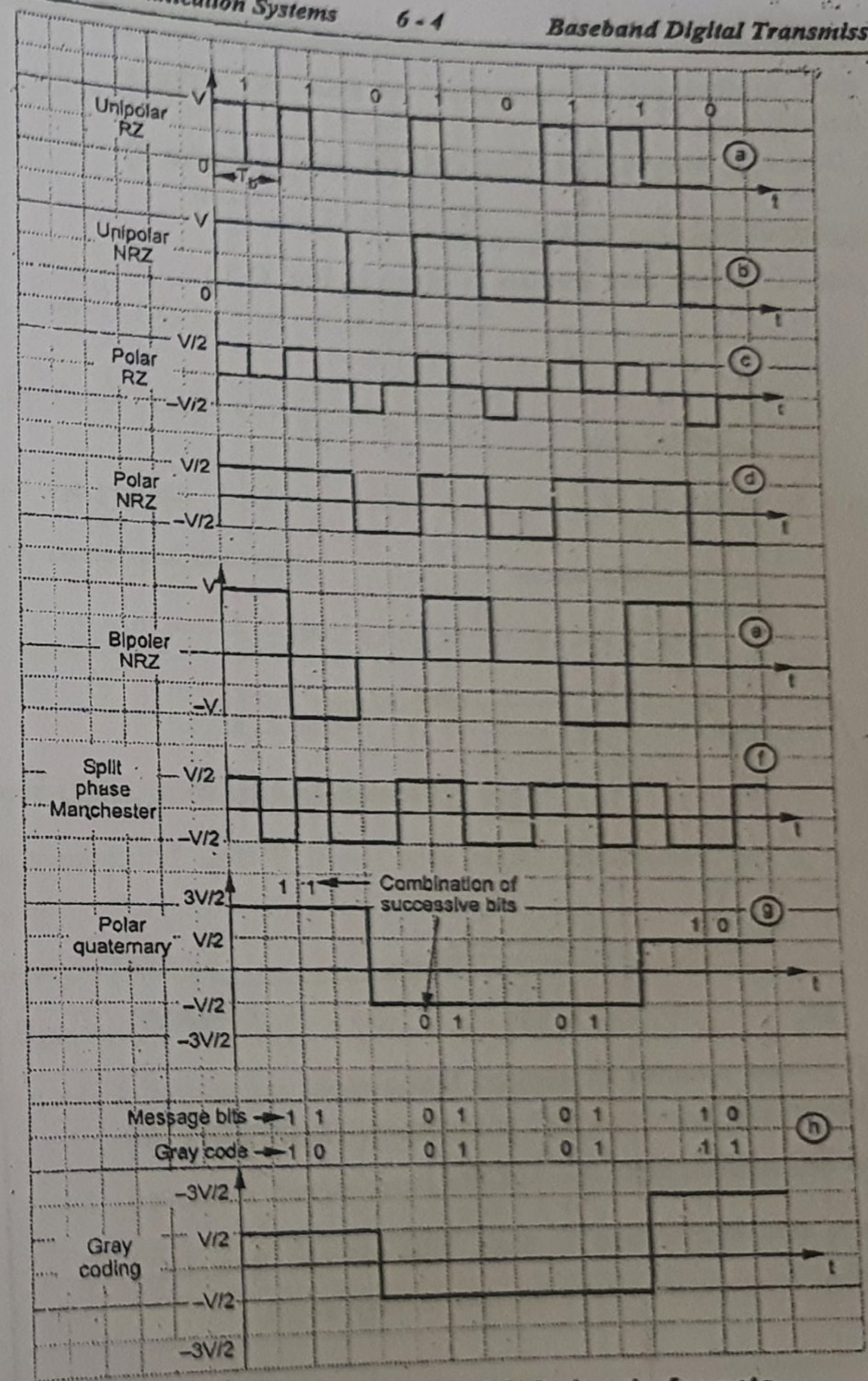


Fig. Q.2.1 Various digital PAM signals formats

(a) Unipolar RZ (b) Unipolar NRZ (c) Polar RZ (d) Polar NRZ (e) Bipolar NRZ (f) Split phase manchester (g) Polar quaternary NRZ (h) Gray coding

Q.3 Draw and give mathematical expression of PSD for unipolar NRZ, AMI and manchestar.

[SPPU : Dec.-13, May-12,14, Dec.-15, (In sem), Marks 8]

OR Explain the spectral features of line codes.

Ans. : The PSD plot of various data formats is given in Fig. Q.3.1

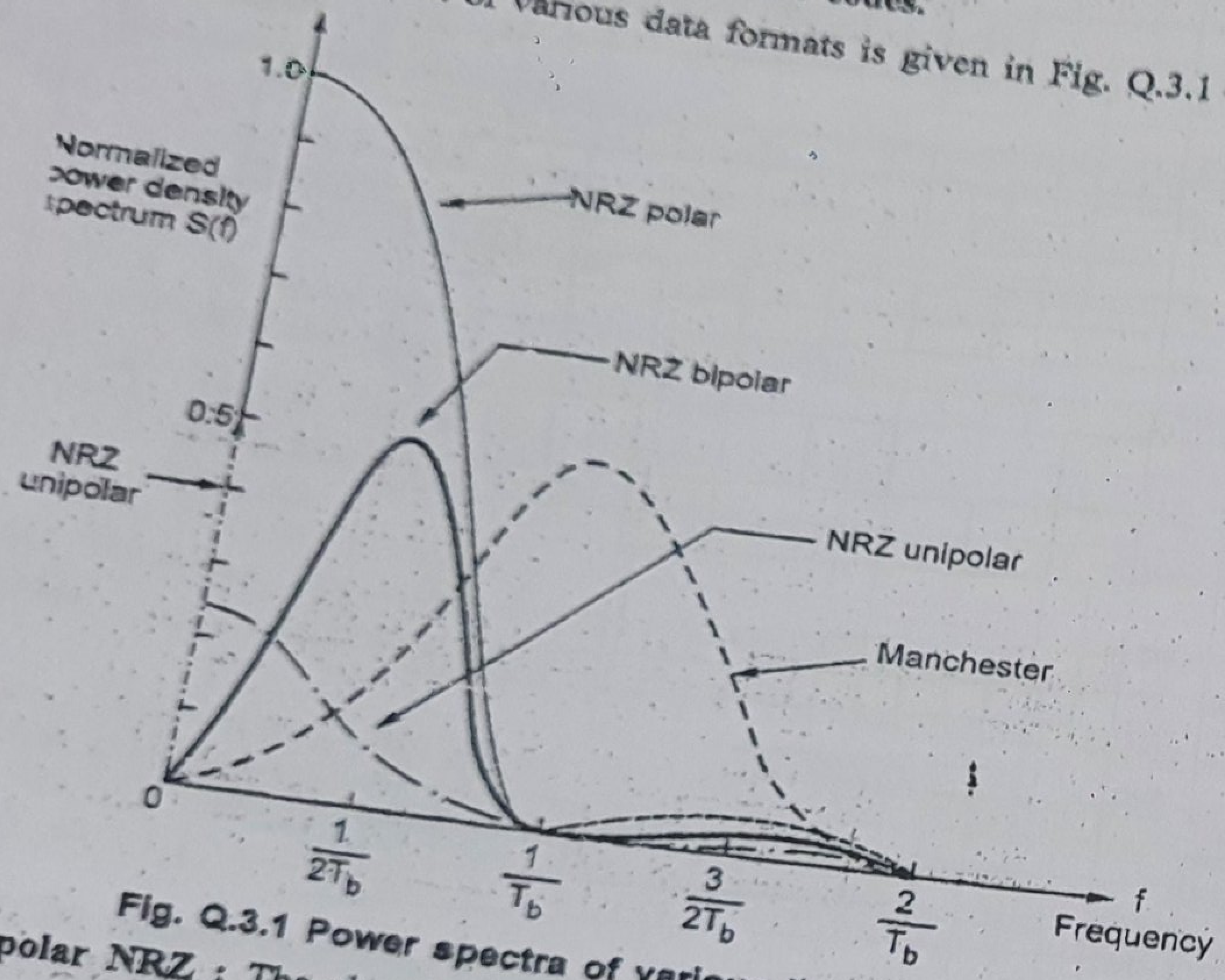


Fig. Q.3.1 Power spectra of various line codes

i) Unipolar NRZ : The signal has some DC component. Hence most of the power lies between DC and bit rate $\left(\frac{1}{T_b}\right)$ of the input signal. Observe that the power spectra is sinc shaped and its main lobe extends from DC to $\frac{1}{T_b}$. Power contained in frequencies above bit rate is very small.

Its PSD is given as,

$$S(f) = \frac{A^2 T_b}{4} \text{sinc}^2 (fT_b) + \frac{A^2}{4} \delta(f)$$

ii) NRZ polar format : The main lobe of sinc pulse contains most of the power from DC to bit rate $\left(\frac{1}{T_b}\right)$. The power contained in frequencies above bit rate is very small. It's PSD is given as,

$$S(f) = A^2 T_b \text{sinc}^2 (fT_b)$$

iii) AMI (Bipolar NRZ) : The spectra is a pulse having peak power near $\frac{1}{2T_b}$ i.e. half bit rate and negligible power at DC and bit rate. Thus power lies inside the bandwidth equal to bit rate $\left(\frac{1}{T_b}\right)$. The power content in frequencies above bit rate is very small. Its PSD is given as,

$$S(f) = A^2 T_b \text{sinc}^2 (fT_b) \text{ sinc}^2 (\pi f T_b)$$

iv) Manchester : Most of the power lies in bandwidth of twice of bit rate $\left(\frac{2}{T_b}\right)$. Negligible power is contained at DC and $\left(\frac{2}{T_b}\right)$. Peak of the spectra occur somewhere near bit rate. Its PSD is given as,

$$S(f) = A^2 T_b \text{ sinc}^2 (fT_b / 2) \sin^2 (f\pi T_b / 2)$$

v) Polar RZ : PSD of polar RZ is given by,

$$S(f) = \frac{T_b}{16} \text{sinc}^2 \left(\frac{fT_b}{2}\right)$$

Q.4 Evaluate the power spectral density of unipolar NRZ and polar RZ code. Plot the spectrum.

[SPPU : Dec.-06, Marks 8, Dec.-12, Marks 10]

Ans. : Power spectral density of unipolar NRZ

The binary sequence has equiprobable occurrence of '0' and '1'. Hence their probabilities of occurrence are $\frac{1}{2}$ each. Hence for n = 0, $E[a_k a_{k-n}]$ will be,

$$E[a_k a_{k-n}] = E[a_k^2] = (0)^2 \frac{1}{2} + a_k^2 \cdot \frac{1}{2}$$

$$= \frac{a_k^2}{2}$$

most of the frequencies

power near Thus power content in

of bit rate ne spectra

polar

s 10]

Hence $a_{k-n}]$

product $a_k a_{k-n}$ has four possible values, i.e. 0, 0, 0 and a_k^2 for ... These four values are equiprobable and statistically independent. ...e they have the probability of $\frac{1}{4}$ each. Hence $E[a_k a_{k-n}]$ for $n \neq 0$ be,

$$[a_k a_{k-n}] = 3\times 0\times\frac{1}{4}+a_k^2\times\frac{1}{4}$$

$$= \frac{a_k^2}{4}$$

...RZ unipolar format, the pulse is of unit amplitude and duration T_b. ...e its Fourier transform will be,

$$P(f) = T_b \operatorname{sinc}(fT_b)$$

...psd is given as,

$$S_x(f) = \left[\frac{1}{T_b}\sum_{n=-\infty}^{\infty} R_a(n)e^{-j2\pi fnT_b}\right]|P(f)|^2$$

...now that $R_a(n) = E[a_k a_{k-n}]$

$$R_a(0) = E[a_k^2] = \frac{a_k^2}{2} \quad and$$

$$R_a(n) = E[a_k a_{k-n}] = \frac{a_k^2}{4} \quad for \quad n \neq 0$$

...e psd will be,

$$S_x(f) = \left[\frac{1}{T_b}R_a(0)+\frac{1}{T_b}\sum_{\substack{n=-\infty\\ n\neq 0}}^{\infty} R_a(n)e^{-j2\pi fnT_b}\right]|P(f)|^2$$

$$= \left[\frac{1}{T_b}\frac{a_k^2}{2}+\frac{1}{T_b}\sum_{\substack{n=-\infty\\ n\neq 0}}^{\infty}\frac{a_k^2}{4}e^{-j2\pi fnT_b}\right]T_b^2\operatorname{sinc}^2(fT_b)$$

$$e^{-j2\pi fnT_b} = \frac{1}{T_b}\sum_{k=-\infty}^{\infty}\delta\left(f-\frac{k}{T_b}\right) \quad \text{by FT relations}$$

$$\therefore \quad S_x(f) = \left[\frac{1}{T_b}\cdot\frac{a_k^2}{2}+\frac{1}{T_b^2}\sum_{k=-\infty}^{\infty}\frac{a_k^2}{4}\delta\left(f-\frac{k}{T_b}\right)\right]T_b^2\operatorname{sinc}^2(fT_b)$$

$$= \frac{a_k^2 T_b}{2}\operatorname{sinc}^2(fT_b)+\sum_{k=-\infty}^{\infty}\frac{a_k^2}{4}\delta\left(f-\frac{k}{T_b}\right)\operatorname{sinc}^2(fT_b)$$

Here $\operatorname{sinc}^2(fT_b) = 0$ at $\pm\frac{1}{T_b}, \pm\frac{2}{T_b}, \pm\frac{3}{T_b}, \ldots$

and $\delta\left(f-\frac{k}{T_b}\right)$ exists at $\pm\frac{1}{T_b}, \pm\frac{2}{T_b}, \pm\frac{3}{T_b}, \ldots$

Hence the product $\delta\left(f-\frac{k}{T_b}\right)\operatorname{sinc}^2(fT_b) = 0$ at all values of k except $k = 0$. Hence psd becomes,

$$S_x(f) = \frac{a_k^2 T_b}{2}\operatorname{sinc}^2(fT_b)+\frac{a_k^2}{4}\delta(f)\operatorname{sinc}^2(0)$$

$$= \frac{a_k^2 T_b}{2}\operatorname{sinc}^2(fT_b)+\frac{a_k^2}{4}\delta(f)$$

Power Spectral Density of Polar RZ code

The symbols '0' and '1' occur with equal probabilities i.e. $\frac{1}{2}$ each. For n = 0.

$E[a_k a_{k-n}]$ will be,

$$E[a_k a_{k-n}] = E[a_k^2] = [-a_k]^2\times\frac{1}{2}+[a_k]^2\times\frac{1}{2} = a_k^2$$

For $n \neq 0$, following are the possible symbols.

a_k	a_{k-n}	$a_k \cdot a_{k-n}$
$-a$	$-a$	a^2
a	$-a$	$-a^2$
$-a$	a	$-a^2$
a	a	a^2

Thus two symbols occur with negative values with probability of $\frac{1}{2}$ each and two symbols occur with positive values with probability of $\frac{1}{2}$ each.

Then

$$E[a_k a_{k-n}] = 2(a_k^2)\frac{1}{2}+2(-a_k^2)\frac{1}{2} = 0$$

Since $R_a(n) = E[a_k a_{k-n}]$ we have,

$$R_a(0) = E[a_k^2] = a_k^2 \text{ and}$$

$$R_a(n) = E[a_k a_{k-n}] = 0 \text{ for } n \neq 0$$

Polar RZ pulse is of amplitude $\frac{1}{2}$ and duration $\frac{T_b}{2}$.

$$P(f) = \frac{1}{2}\times\frac{T_b}{2}\, sinc\left(\frac{fT_b}{2}\right) = \frac{T_b}{4}\, sinc\left(\frac{fT_b}{2}\right)$$

The psd is given as,

$$S_n(f) = \left[\frac{1}{T_b}\sum_{n=-\infty}^{\infty} R_a(n) e^{-j2\pi fnT_b}\right]|P(f)|^2$$

$$= \left[\frac{1}{T_b}a_k^2 + \frac{1}{T_b}\sum_{n=-\infty}^{\infty} 0\times e^{-j2\pi fnT_b}\right]|P(f)|^2$$

$$= \frac{1}{T_b}a_k^2 \times\left[\frac{T_b}{4}\, sinc\left(\frac{fT_b}{2}\right)\right]^2$$

$$= \frac{a_k^2 T_b}{16}\, sinc^2\left(\frac{fT_b}{2}\right) = \frac{T_b}{16}\, sinc^2\left(\frac{fT_b}{2}\right)$$

Here 'a_k' is already included in rectangular pulse as amplitude of $\pm\frac{1}{2}$. Hence 'a_k' can be considered unity in above equation.

Q.5 Find PSD for polar and unipolar (ON-OFF) signalling where p(t) is a full width rectangular pulse. p(t) = rect$\left(\frac{t}{T_b}\right)$ given :

$$S_x(\omega) = \frac{1}{T_b}\left(R_0 + 2\sum_{n=1}^{\infty} R_n \cos n\omega T_b\right).$$

[SPPU : May-11, Marks 6, Dec.-07, Marks 8]

Ans. : ON-Off signaling means RZ. Hence unipolar ON-OFF means unipolar RZ and polar ON-OFF means polar RZ.

i) PSD of Polar ON-OFF

Here the pulse width is T_b. Thus it is polar RZ with '1' is represented by a pulse of amplitude 1 and duration T_b. And '0' is represented by a pulse of amplitude '-1' and duration T_b. Thus the bit duration is 2 T_b instead of T_b. The PSD of polar - RZ is derived in Q.8 for a rectangular pulse amplitude of $\frac{1}{2}$ and duration $\frac{T_b}{2}$. Hence for a pulse of amplitude '1' and duration 'T_b' the Fourier transform will be,

$$P(f) = T_b \sin c\,(f\,T_b)$$

for polar signaling in Q.4 we have derived that,

$$R_a(0) = E\left[a_k^2\right] = a_k^2 \text{ and}$$

$$R_a(n) = E[a_k a_{k-n}] = 0 \text{ for } n \neq 0$$

The PSD is given as,

$$S_x(f) = \left[\frac{1}{T_b}\sum_{n=-\infty}^{\infty} R_a(n) e^{-j2\pi fnT_b}\right] |P(f)|^2$$

Since $R_a(n) = 0$ for $n \neq 0$ and $R_a(0) = a_k^2$

$$S_x(f) = \frac{1}{T_b}R_a(0)|P(f)|^2$$

...) signalling where
...ect $\left(\frac{t}{T_b}\right)$ given :

...c.-07, Marks 8]
ON-OFF means

...is represented by
...ented by a pulse
...s 2 T_b instead of
...ectangular pulse
...mplitude '1' and

$$= \frac{1}{T_b} . a_k^2 . \text{sinc}^2 (fT_b)$$

$a_k = \pm 1$, amplitude of the pulse. Hence $a_k^2 = 1$ always.

$$S_x(f) = T_b \, sinc^2 (fT_b)$$

$$S_x(\omega) = T_b \, sinc^2 \left(\frac{\omega}{\omega_b}\right) \text{ Here } f = \frac{\omega}{2\pi} \text{ and } \omega_b = \frac{2\pi}{T_b}$$

...SD of Unipolar ON-OFF

...is basically unipolar RZ pulse. It's amplitude is '1' and duration is T_b.
...PSD of unipolar RZ pulse of duration $\frac{T_b}{2}$ is given as,

$$S_y(\omega) = S_x(\omega) \, | P(\omega) |^2$$

$$= \frac{1}{4T_b} \left[1 + \omega_b \sum_{n=-\infty}^{\infty} \delta(\omega - n\omega_b) \right] | P(\omega) |^2$$

...for a pulse of duration T_b and amplitude '1' we have,

$$P(f) = T_b \, sinc \, (f \, T_b)$$

$$\text{or } P(\omega) = T_b \, \text{sinc} \left(\frac{\omega}{\omega_b}\right)$$

$$S_y(\omega) = \frac{1}{4T_b} \left[1 + \omega_b \sum_{n=-\infty}^{\infty} \delta(\omega - n\omega_b) \right] \left[\left(T_b \, sinc \frac{\omega}{\omega_b} \right) \right]^2$$

$$S_y(\omega) = \frac{T_b}{4} \, sinc^2 \left(\frac{\omega}{\omega_b}\right) \left[1 + \omega_b \sum_{n=-\infty}^{\infty} \delta(\omega - n\omega_b) \right]$$

...raw the line codes-Unipolar RZ, Polar NRZ, AMI, Manchester, ...RZ and quaternary polar for the bit stream 10110100.

☞ [SPPU : Aug.-17, Marks 6]

Ans. : Fig. Q.6.1 below shows the waveform.

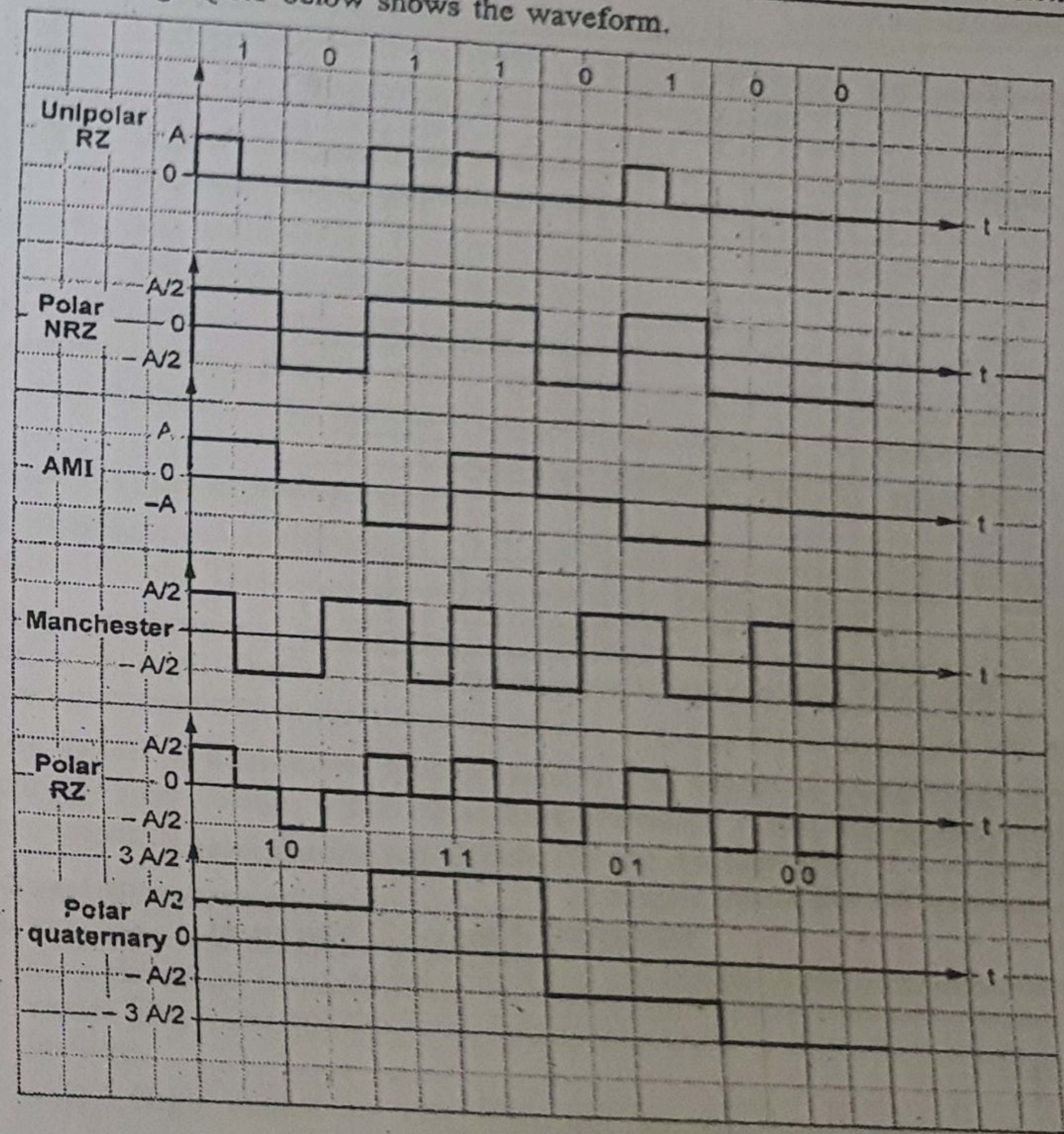


Fig. Q.6.1 Waveforms of Q.6

Q.7 For the sequence 10111001, sketch the waveform using the following data formats : i) Unipolar RZ ii) Polar NRZ iii) Alternate mark inversion iv) Split phase manchester coding. Draw the corresponding spectrum of the above formats and explain.

☞ [SPPU : Dec.-11, Marks 10, Oct.-16, Marks 5, Dec.-22, Marks 6]

Ans. : The waveforms are sketched in Fig. Q.7.1.

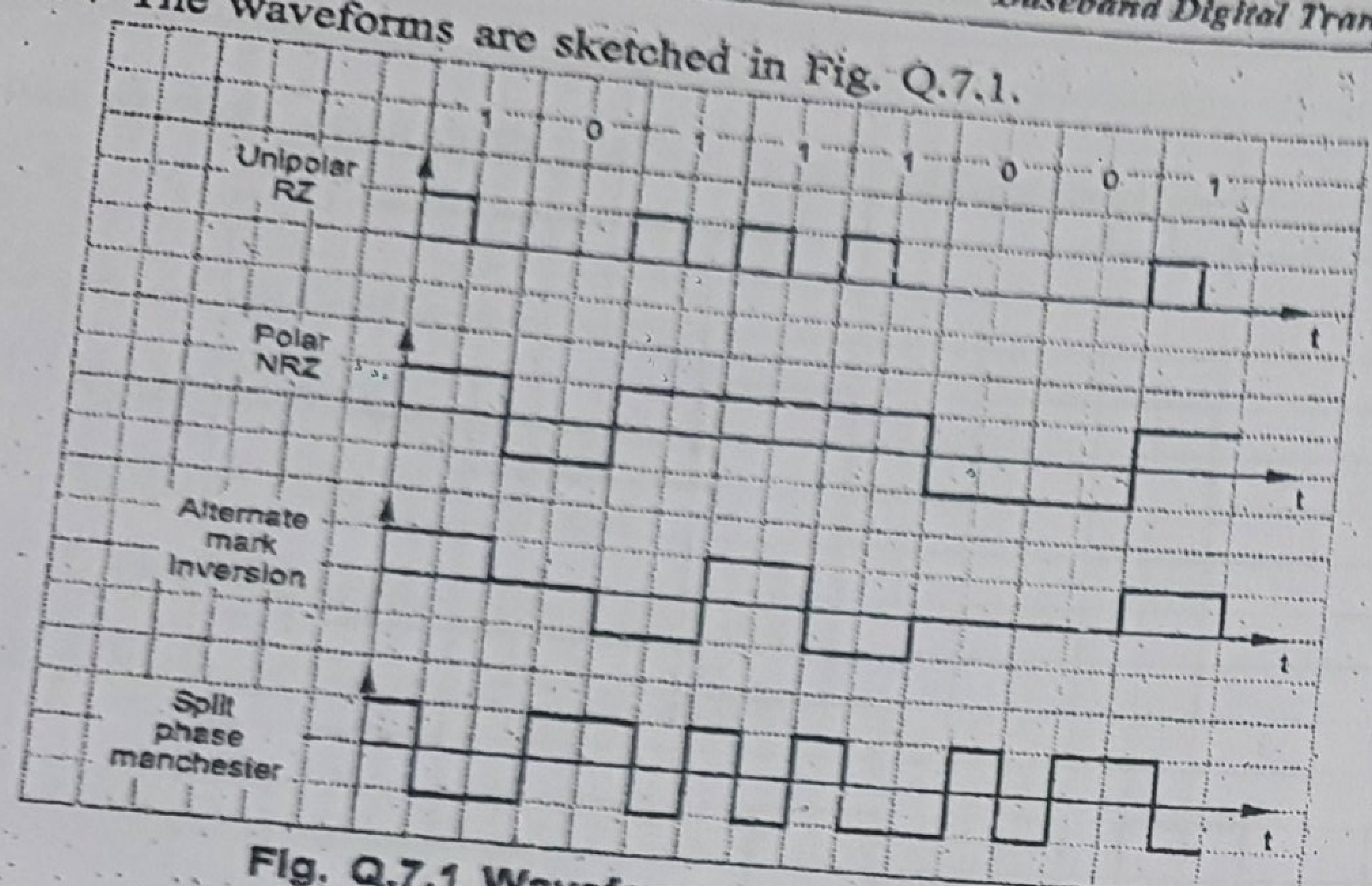


Fig. Q.7.1 Waveforms of example Q.7

Q.8 Consider the sequence of 1 s and 0 s

i) An alternate sequence of 1 s and 0 s

ii) A continuous sequence of five 1 s followed by five 0 s

iii) A sequence of two 1 s followed by ten 0 s sketch the manchester format representation for the above sequence.

☞ **[SPPU : May-12, Marks 8]**

Ans. : The split phase manchester waveforms are shown below :

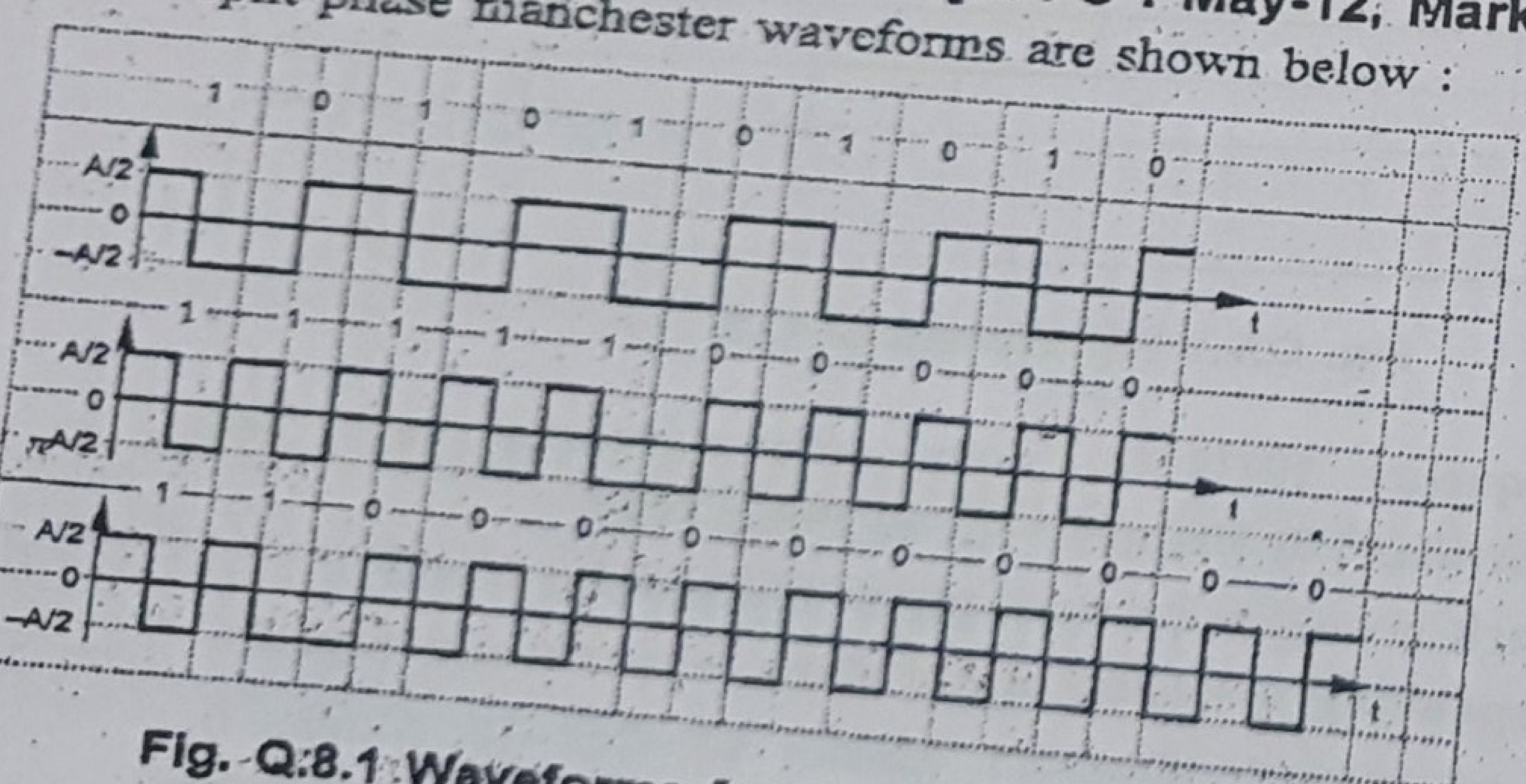


Fig. Q.8.1 Waveforms for manchester coding

Q.9 Represent the data 10011101 using following data formats.
i) Unipolar RZ ii) Split phase Manchester
iii) M-ary format for M = 4. ☞ **[SPPU : May-19, Marks 6]**

Ans. : Fig. Q.9.1 shows the various data formats.

For M-ary format with M = 4, bits can be combined as follows :

Bits combination	00	01	10	11
$x(t) = a_n$	$-\frac{3A}{2}$	$-\frac{A}{2}$	$\frac{A}{2}$	$\frac{3A}{2}$

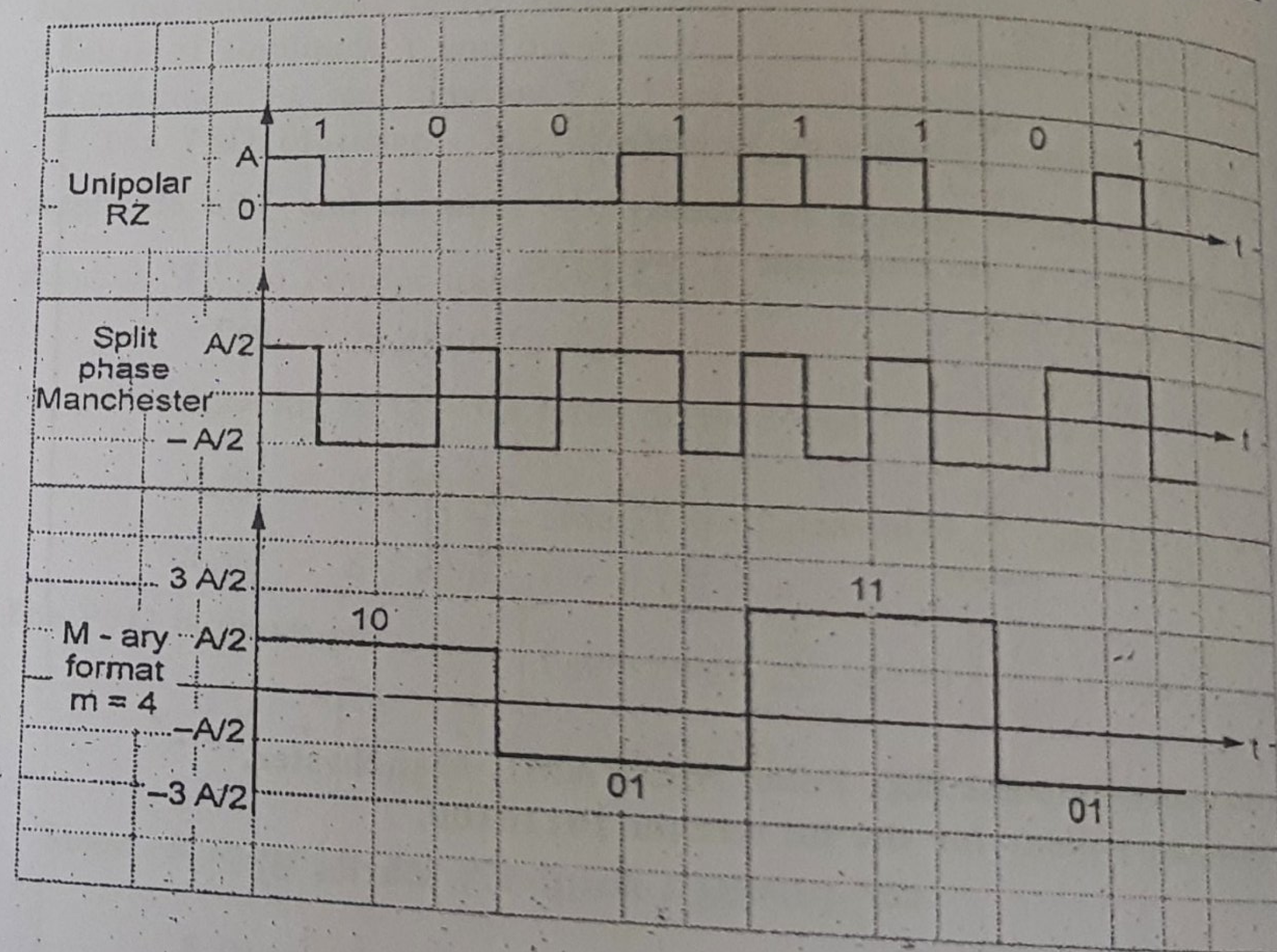


Fig. Q.9.1 Data formats

Q.10 Compare various line coding formats.

☞ **[SPPU : Oct.-18, Marks 4]**

Ans. : See Table Q.10.1 on next page.

Sr. No.	Parameter	Unipolar RZ	Unipolar NRZ	Polar RZ	Polar NRZ	Bipolar NRZ (AMI)	Manchester	Polar Quaternary
1.	DC component in the signal	Present	Present	may be present	may be present	Absent	Absent	may be present
2.	Signal frequency (Bandwidth)	f_b	$\frac{f_b}{2}$	f_b	$\frac{f_b}{2}$	$\frac{f_b}{2}$	f_b	$\frac{f_b}{4}$
3.	Peak Power requirement	high	high	low	low	high	low	very high
4.	Noise immunity	Poor	fair	fair	fair	good	good	poor
5.	Cross talk	high	high	high	moderate	low	low	low
6.	Synchronization capability (effect of strings of 0's or 1's)	better	poor	better	poor	good	good	poor

Table Q.10.1 Comparison of line coding formats

6.2 : Digital Multiplexing and Hierarchies

Important points to remember

- Digital multiplexing is used in telephone lines. It transmits data on single line by multiplexing from various sources.
- The multiplexing improves the utilization of the telephone line and allows many sources to transmit.

Q.11 What are different types of multiplexers used in digital communication? Explain quasi - sychronous multiplexer in detail with a neat sketch. ☞[SPPU : Dec.-13, 11, May-13, 15, Marks 10, Oct.-16, Marks 5, Dec.-17, Marks 3]

Ans. : Types of multiplexers

i) Synchronous multiplexers : All the digital sources transmit dat at the same bit rate. There is single master clock for all the sources.

ii) Asynchronous multiplexers : Different sources transmit data at different rates. There is no common clock for these sources. Mainly the data is transmitted in the start/stop type.

iii) Quasi synchronous multiplexers : The input bit rates from the sources may change within the specified bounds. Hence the multiplexers are arranged in such a way that they transmit synchronously.

Quasi Synchronous Multiplexing

- The quasi synchronous multiplexing is necessary when input bit rates have the same nominal value, but small variations around this nominal value. The quasi synchronous multiplexers need to have three essential features.
  - i) High output bit rate such that all maximum input rates are accommodated.
  - ii) They should have special buffers called elastic stores to store the input bits temporarily.
  - iii) They should stuff the bits to pad the output stream.
- Fig. Q.11.1 shows the block diagram of quasi synchronous multiplexer.

- As shown in the Fig. Q.11.1, there are total 'N' number of input channels. Their bit rates vary from r_{min} to r_{max} around the nominal value. The MUX has constant output rate i.e. $r_o > N\ r_{max}$.

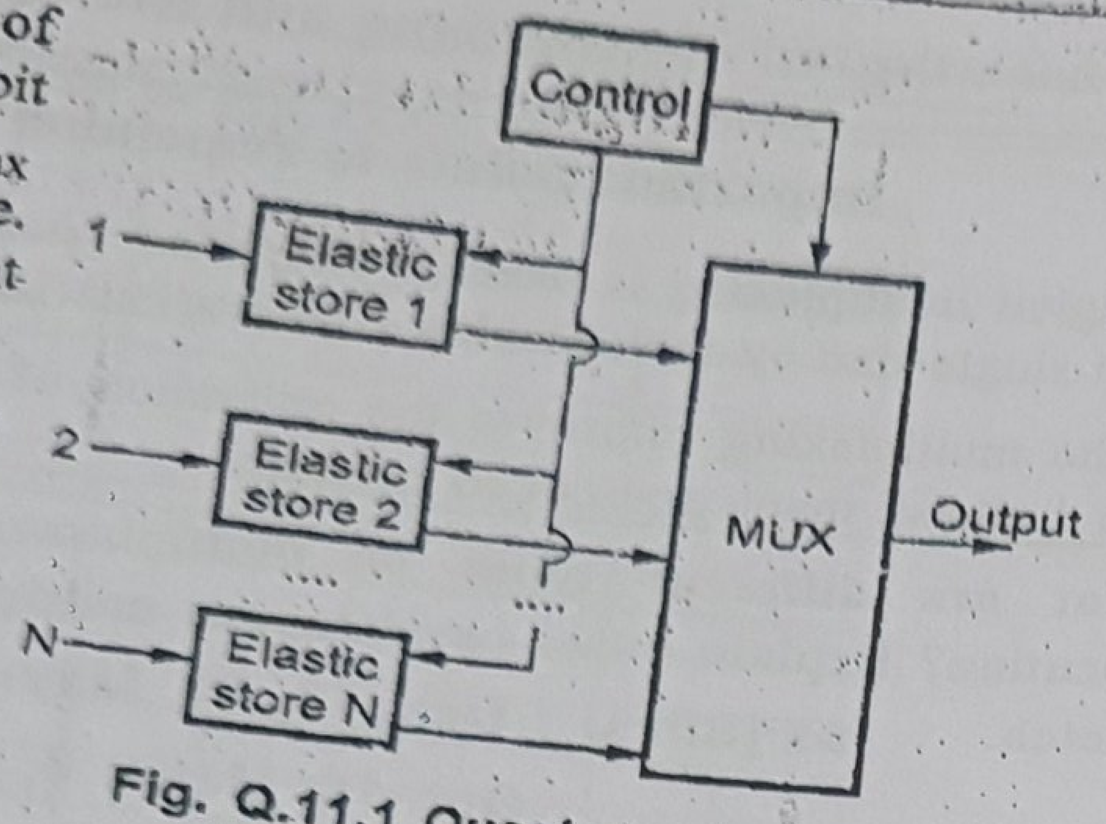


Fig. Q.11.1 Quasi synchronous multiplexer

- The control unit receives signal from the elastic stores and sends read signals to them. When this read signal is received, the elastic store sends its bit to MUX. The control unit also sends signal to MUX for stuffing extra bits in the output stream. These extra bits are shifted in the output stream when no bit is ready from the particular elastic store.

Advantages

i) Signals with variable bit rate can be transmitted synchronously.

ii) More flexible system.

Disadvantage : There is no overall master clock control.

Q.12 Explain T1 carrier system i.e. multiple channel alignment for TDM/PCM

[SPPU : May-11, Aug.-17, Marks 6, Oct.-16, Dec.-17, Marks 5, May-18, Marks 8]

OR Explain digital signal hierarchy using T1 carrier system.

[SPPU : Dec.-15, (End Sem), Marks 6, May-19, Marks 7]

Ans. : • The multiple channel alignment is very important in TDM/PCM system. Fig. Q.12.1 shows the TDM frame format of most widely used *T1* system.

- This system contains a multiframe of 12 frames. The duration of the multiframe is 1.5 msec. Each frame consists of samples from 24 channels. Thus the samples of 24 channels are Time division Multiplexed. Each channel sample is encoded into 8 bits. Thus the total bits of 24 channels will be $24 \times 8 = 192$ bits. This indicate the start of the next frame, the frame sync bit or '*S*' bit is transmitted at the

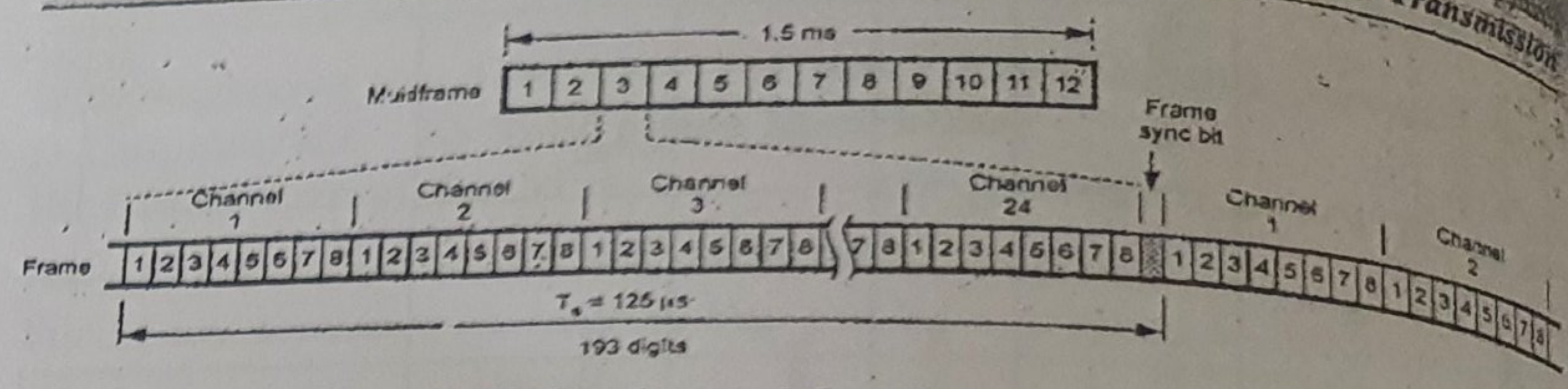


Fig. Q.12.1 Multiple channel frame alignment in T1 system

begining of each frame. Thus the total bits in one frame are $(24 \times 8) + 1 = 193$ bits.

Calculation of bit rate : • Each channel is normally sampled at 8 kHz rate. Thus the time between any two successive samples of single channel will be $\frac{1}{8000\text{ Hz}} = 125$ microseconds. In the TDM system, the samples from each channel is transmitted in each successive frame. Hence the duration of the frame will also be 125 microseconds. This is shown in Fig. Q.12.1.

$$\text{Bits per frame} = 24 \text{ channels/frame} \times 8 \text{ bits/channel} + 1 \text{ frame sync bit}$$

$$= 193 \text{ bits}$$

$$\therefore \quad \text{Bit rate } R_b = \frac{\text{Number of bits per frame}}{\text{Time of one frame}} = \frac{193 \text{ bits}}{125 \times 10^{-6} \text{ seconds}}$$

$$= 1.544 \times 10^6 \text{ bits/sec}$$

Q.13 Draw AT and T and CCIT hierarchy multiplexing system and explain.

[SPPU : Dec.-15, (Insem), Marks 8, May-15 (End sem), May-16(End Sem), Marks 7, Oct.-18, Marks 4, May-17, Marks 8, Dec.-16, 22, Marks 6, June-22, Marks 5]

Ans. : AT and T hierarchy : • Fig. Q.13.1 shows AT and T hierarchy. Observe that four T_1 lines are multiplexed in M12 multiplexer. The bit rate of each T_1 line is 1.544 Mbps. The channel bank multiplexes 24 voice PCM signals of 64 kbps bit rate into single T_1 line. It can also be used to transmit digital data, visual telephone and TV signals.

- The M23 multiplexer at third level multiplexes seven T_2 lines. Each of these lines have 6.312 Mbps bit rate.

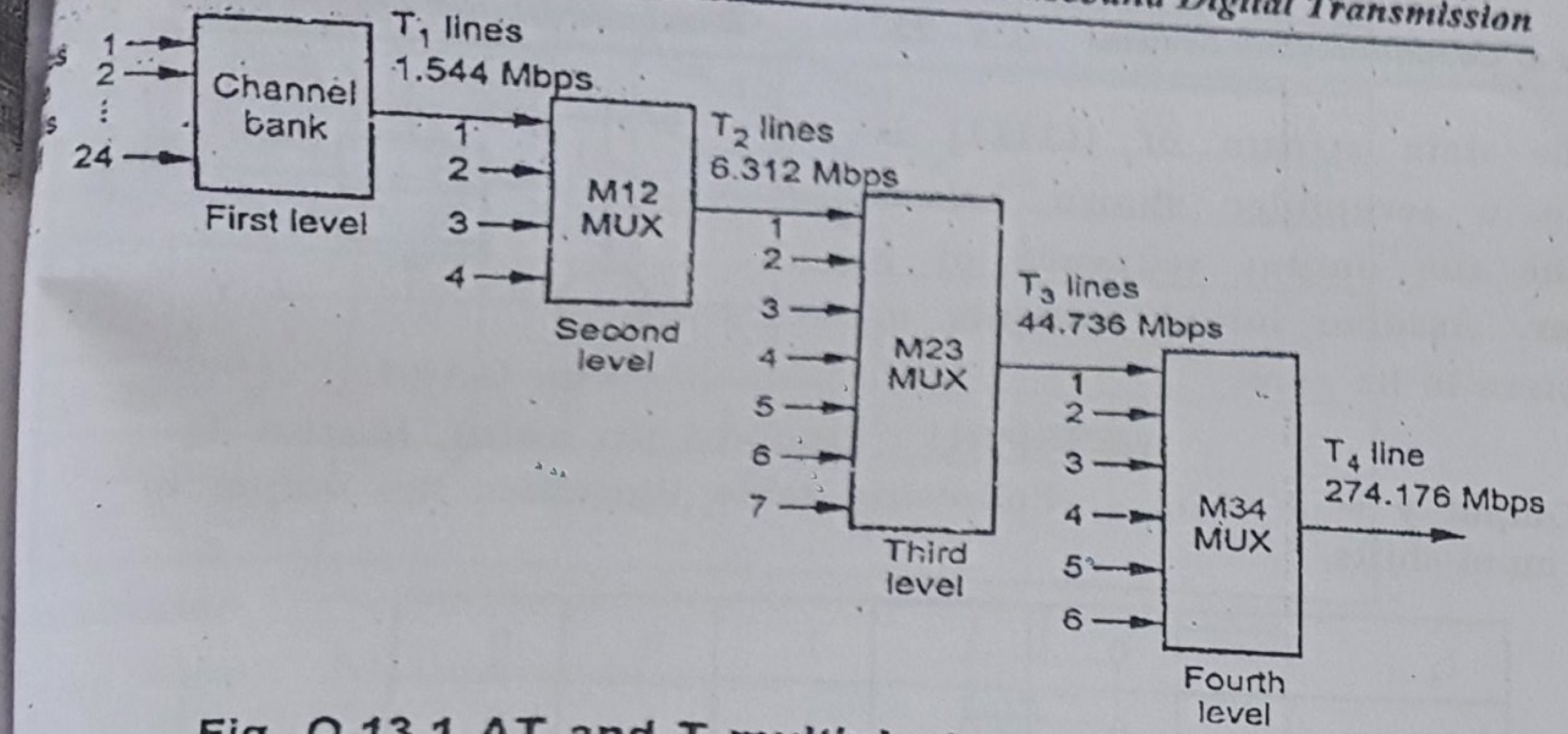


Fig. Q.13.1 AT and T multiplexing hierarchy

...the fourth level M34 multiplexer multiplexes six T_3 lines. Each of ...se lines have 44.736 Mbps bit rate.

... T_4 line have 274.176 Mbps bit rate. Thus at the end of 4^{th} level ...l voice PCM channels would be $24 \times 4 \times 7 \times 6 = 4032$. Thus T_4 ... carries 4032 voice PCM channels.

... signaling rate of T_4 line will be $r = 274.176 \times 10^6$ bps. Hence ...smission channel bandwidth will be,

$$B_T \geq \frac{1}{2} r = \frac{1}{2} \times 274.176 \times 10^6 = 137 \text{ MHz}$$

... Hierarchy

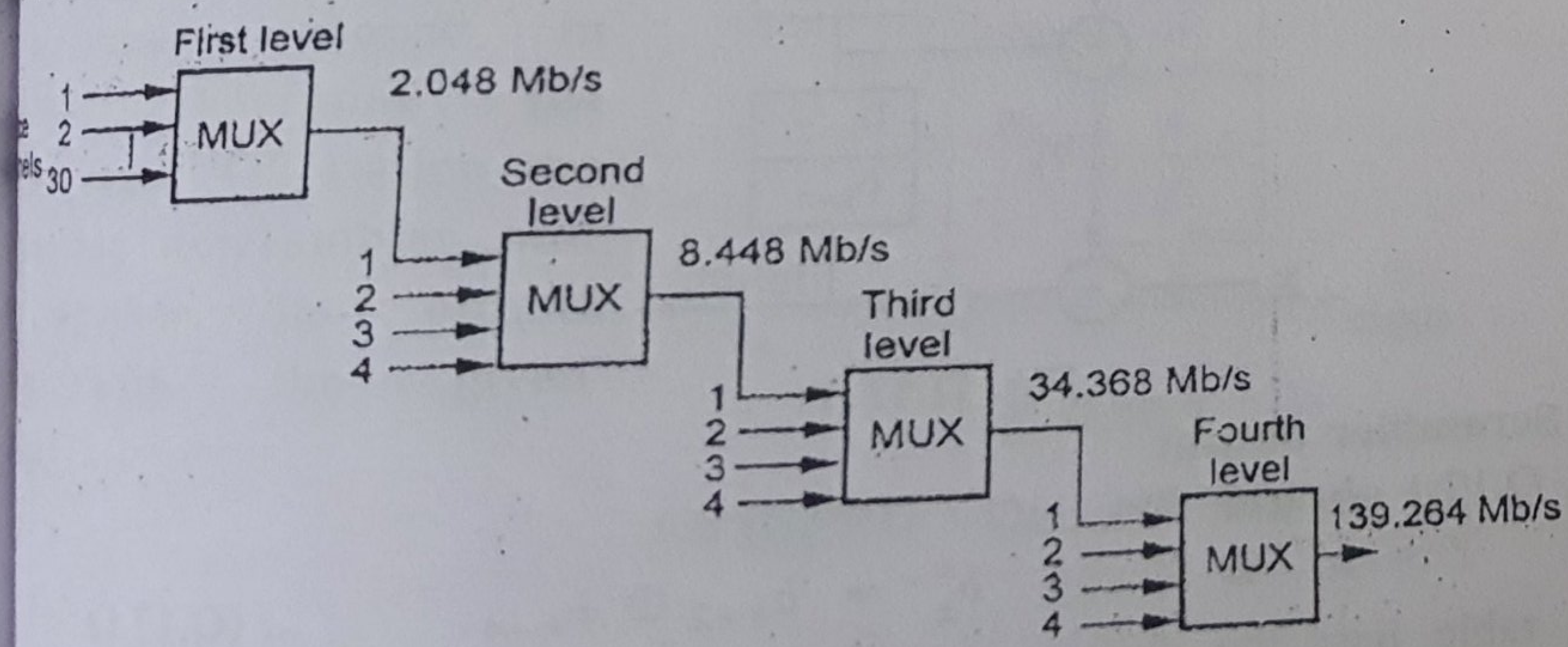


Fig. Q.13.2 CCIT hierarchy

- Fig. Q.13.2 shows the CCIT hierarchy for digital multiplexing. The CCIT hierarchy is used in Europe for digital multiplexing. In the first level it multiplexes 30 voice channels. The rest of the three levels multiplex 4 channels.

Q.14 Compare AT and T and CCIT hierarchy of multiplexing.

[SPPU : May-11, Marks 3]

Ans. :

Multiplexing level	AT and T		CCIT	
	Number of inputs	Output bit rate in Mb/s	Number of inputs	Output bit rate in Mb/s
First	24	64 kbps × 24 = 1.544	32	64 kbps × 32 = 2.048
Second	4	6.312	4	8.448
Third	7	44.736	4	34.368
Fourth	6	274.176	4	139.264

Table Q.14.1 Comparison between bit rates of AT and T and CCIT hierarchies

6.3 : Scrambling and Unscrambling

Important points to remember

- The long strings of like bits are reduced and randomized when scrambling operation is applied to the message at transmitter.
- Receiver synchronization is improved because of scrambling and unscrambling.
- To recover the original message signal, the received signal is unscrambled at the receiver.

Q.15 What is scrambling ? With the help of neat diagrams explain scrambler and unscrambler operation.

[SPPU : June-22, Dec.-22, Marks 6]

Ans. : Scrambling : The scrambling alters the input bit sequence in some fixed format and reverse operation is done at the receiver. Because of scrambling, long strings of like bits are reduced and randomized. This improves the receiver synchronization.

Scrambler Operation

Fig. Q.15.1 shows the block diagram of binary scrambler. It uses a four stage shift register with tap gains α_1 and α_2 equal to zero and $\alpha_3 = \alpha_4 = 1$. The block diagram does not show the clock signal explicitly. It is assumed to be present in the block diagram.

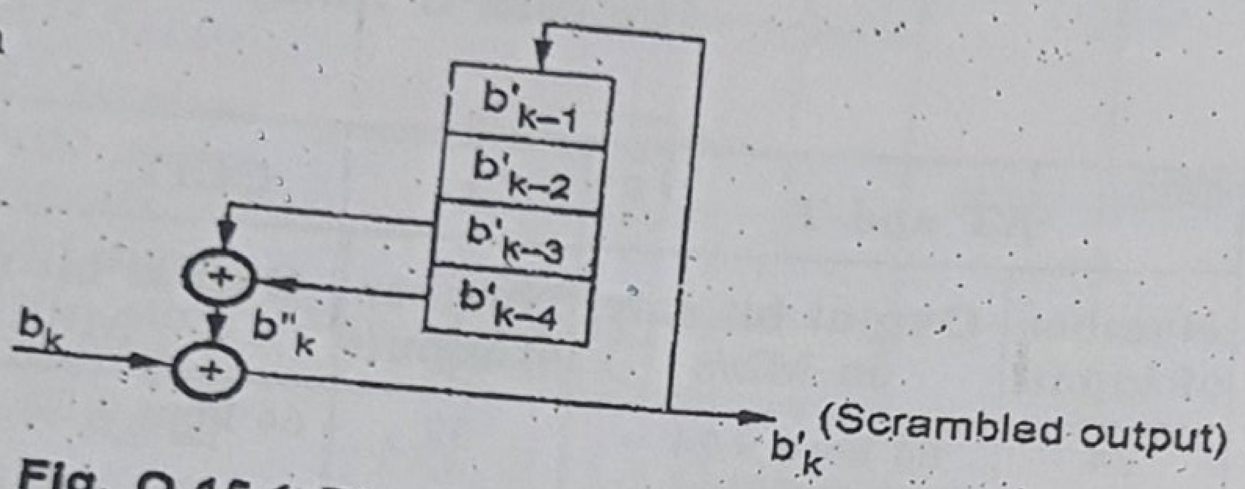


Fig. Q.15.1 Block diagram of binary scrambler

From the block diagram of Fig. Q.15.1 we can write,

$$b_k' = b_k \oplus b_k'' \qquad b_k'' = b_{k-3}' \oplus b_{k-4}' \qquad \text{... (Q.15.1)}$$

Unscrambler Operation

Fig. Q.15.2 shows the block diagram of unscrambler. Observe that the unscrambler block diagram is exactly opposite to that of scrambler. The unscrambler uses a four stage shift register.

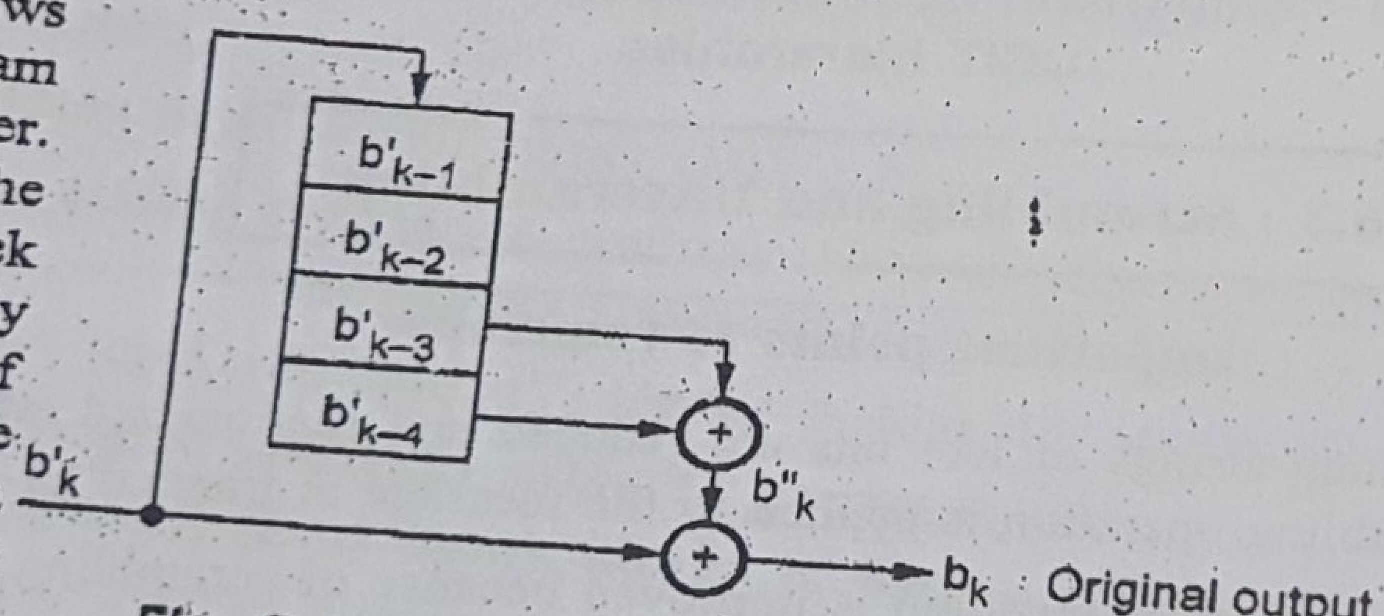


Fig. Q.15.2 Block diagram of unscrambler

Output sequence $= b_k' \oplus b_k''$ ... (Q.15.2)

putting for b_k' from equation Q.15.1.

Output sequence $= (b_k \oplus b_k'') \oplus b_k'' = b_k \oplus (b_k'' \oplus b_k'')$

$= b_k \oplus 0$

Q.16 The data stream of [11111] is given to a scrambler shown below. Determine the output sequence of a scrambler. Assume initial contents of the registers to be zero.

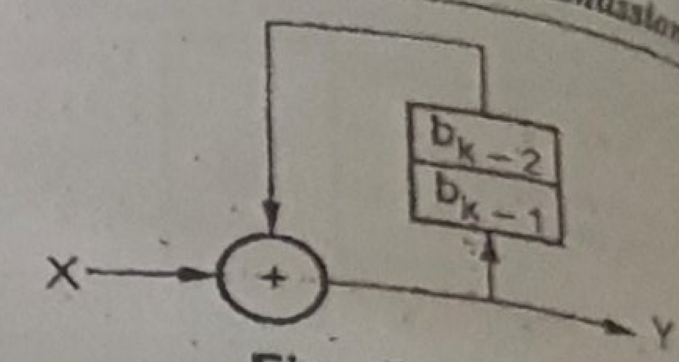


Fig. Q.16.1

[SPPU : Dec.-14 (In sem), Marks 4]

Ans. : Output $y = X \oplus b_{k-2}$. Following table illustrates the output at different input shifts.

b_{k-1}	0	1	1	0	0
b_{k-2}	0	0	1	1	0
X	1	1	1	1	1
$Y = X \oplus b_{k-2}$	1	1	0	0	1

Q.17 A scrambler is shown in Fig. Q.17.1. Design the corresponding unscrambler if a sequence is 1010110 applied to the scrambler input determine the output.

[SPPU : May-11, Marks 6]
May-13,17, Marks 8]

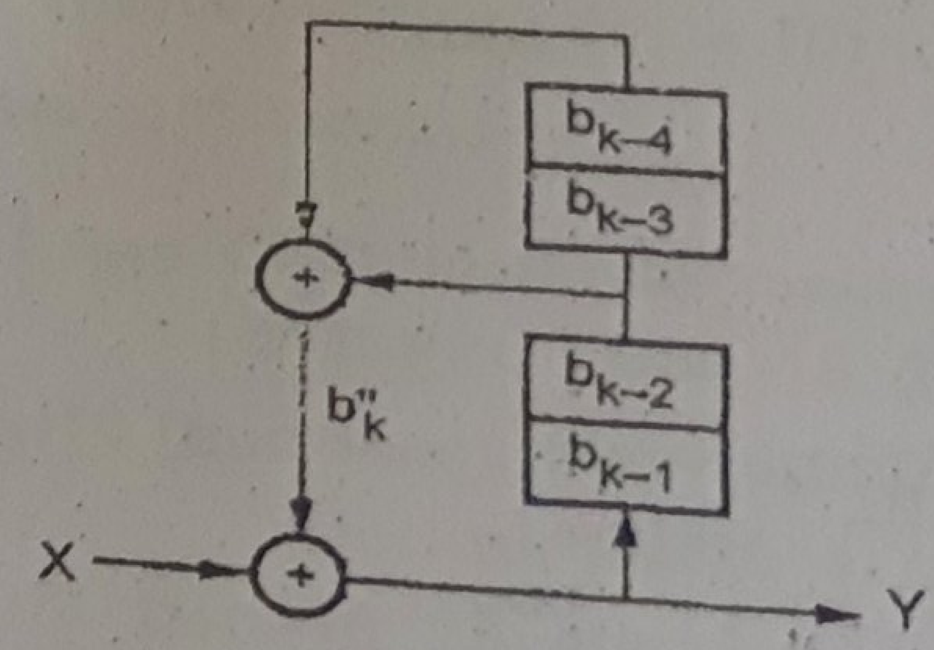


Fig. Q.17.1

Ans. : i) Scrambler output

From Fig. Q.17.1 observe that,

$$Y = b_k'' \oplus X \text{ and } b_k'' = b_{k-2} \oplus b_{k-4} \qquad \text{... (Q.17.1)}$$

Following table lists the shift register contents and output at different input shifts. The shift register contents are assumed 0 0 0 0 initially.

Shift register contents								
	b_{k-1}	0	1	0	0	0	0	1
	b_{k-2}	0	0	1	0	0	0	0
	b_{k-3}	0	0	0	1	0	0	0
	b_{k-4}	0	0	0	0	1	0	0
$b''_k = b_{k-2} \oplus b_{k-4}$		0	0	1	0	1	0	0
Input, X		1	0	1	0	1	1	0
Output, Y $= b_k'' \oplus X$		1	0	0	0	0	1	0

...scrambler

...Q.17.2 shows the unscrambler for ...bler of Fig. Q.17.1.

...this figure, $X = b''_k \oplus Y$

...know that $Y = b''_k \oplus X$

...equation (Q.17.1). Putting this in ...equation,

$$X = b''_k \oplus (b''_k \oplus X)$$
$$= (b''_k \oplus b''_k) \oplus X$$
$$= 0 \oplus X = X$$

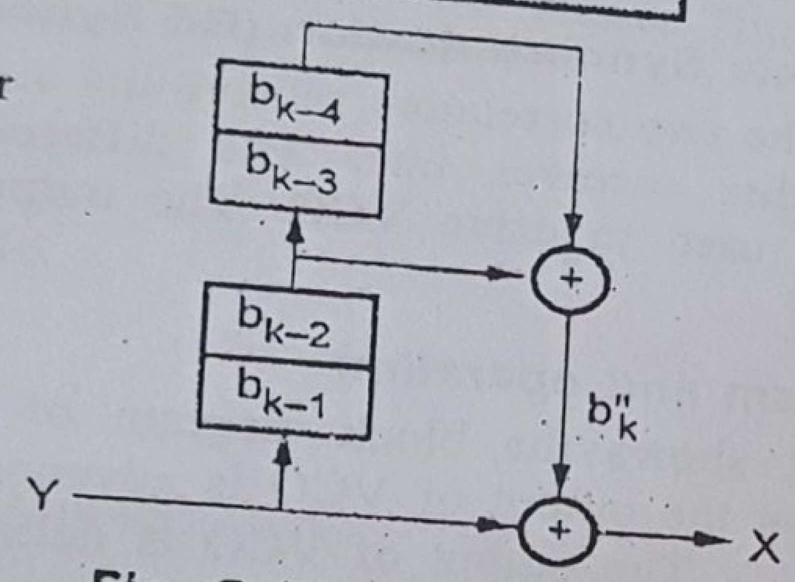


Fig. Q.17.2 Unscrambler

...he original sequence is obtained back.

... Consider a sequence ...001000000010 is applied to ... scrambler shown in ...Q.18.1. Determine the ...sequence 'O'. Design the ...ponding descrambler and ... whether its output ...es with the given ...nce.

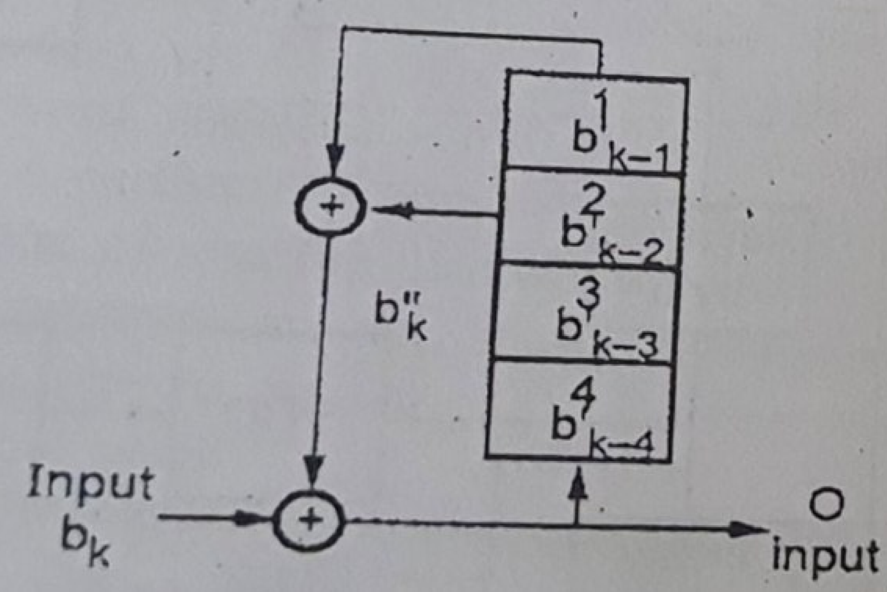


Fig. Q.18.1

☞[SPPU : Dec.-11, Marks 10]

Ans. : i) Scrambler output

From Fig Q.18.1 observe that,

Output, $\quad O = b_k \oplus b''_k$ and $b''_k = b'_{k-1} \oplus b'_{k-2}$ ... (Q.18.1)

Following table lists all the shift register contents. Initial values are assumed to be zero.

Shift register contents																		
	b'_{k-4}	0	1	0	1	1	1	1	0	1	0	1	1	1	1	0	0	1
	b'_{k-3}	0	0	1	0	1	1	1	1	0	1	0	1	1	1	1	0	0
	b'_{k-2}	0	0	0	1	0	1	1	1	1	0	1	0	1	1	1	1	0
	b'_{k-1}	0	0	0	0	1	0	1	1	1	1	0	1	0	1	1	1	1
$b''_k = b'_{k-2} \oplus b'_{k-1}$		0	0	0	1	1	1	0	0	0	1	1	1	1	0	0	0	1
Input, b_k		1	0	1	0	0	0	0	1	0	0	0	0	0	0	0	1	0
Output, $O = b_k \oplus b''_k$		1	0	1	1	1	1	0	1	0	1	1	1	1	0	0	1	1

ii) Unscramber

Fig. Q.18.2 shows the descrambler.

From this figure,

$$b_k = O \oplus b''_k$$

Putting for O from equation (Q.18.1)

$$b_k = \left(b_k \oplus b''_k\right) \oplus b''_k$$
$$= b_k \oplus \left(b''_k \oplus b''_k\right) = b_k \oplus 0 = b_k$$

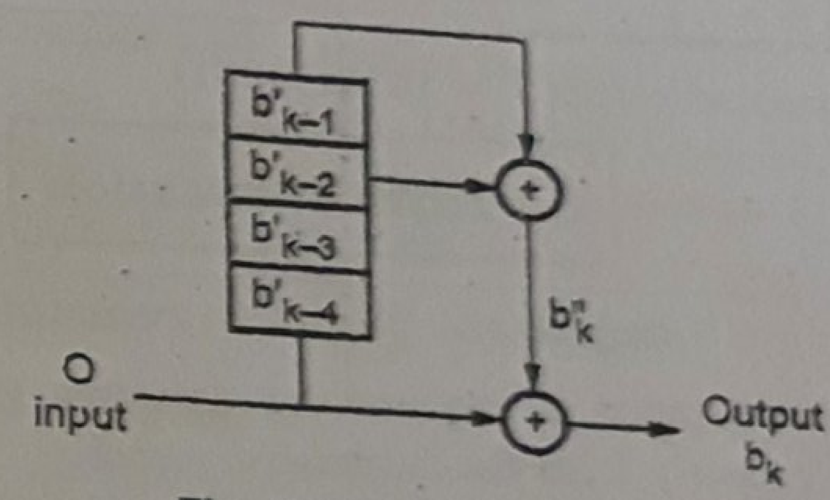


Fig. Q.18.2 Descrambler

Q.19 The data stream 10101010001 is an input to a scramble shown in Fig. Q.19.1. Obtain the scrambled output assuming initial content of all registers to be zero.

☞[SPPU : Aug.-15, Marks 5]

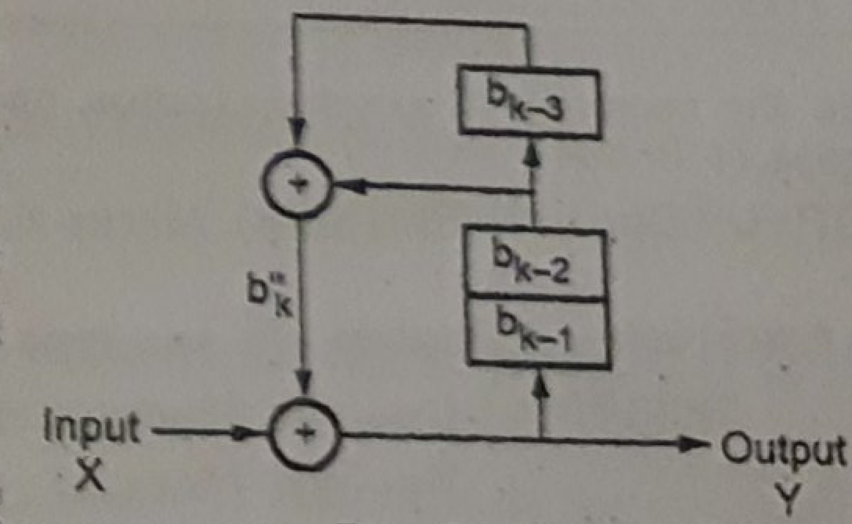


Fig. Q.19.1

Ans. : i) Scrambler output :

From Fig. Q.19.1, observe that,

$$Y = b''_k \oplus X \text{ and } b''_k = b_{k-2} \oplus b_{k-3}$$

Following table lists the shift register contents and output at different input shifts. The shift register contents are assumed 000 initially.

Shift register contents												
	b_{k-1}	0	1	0	0	1	1	1	1	0	0	1
	b_{k-2}	0	0	1	0	0	1	1	1	1	0	0
	b_{k-3}	0	0	0	1	0	0	1	1	1	1	0
$b''_k = b_{k-2} \oplus b_{k-3}$		0	0	1	1	0	1	0	0	0	1	0
Input, X		1	0	1	0	1	0	1	0	0	0	1
Output Y $= b''_k \oplus X$		1	0	0	1	1	1	1	0	0	1	1

∴ Scrambled output $Y = 10011110011$

6.4 : Synchronization

Important points to remember

- Synchronization is required for detectors to recover the digital data properly from the modulated signal
- There are three broad types of synchronization. They are :
  - i) Carrier synchronization
  - ii) Symbol and bit synchronization
  - iii) Frame synchronisation.

Q.20 Explain the necessity of synchronization and state the types of synchronization.

☞[SPPU : Dec.-14, (End sem), Marks 6, May-11, Marks 8]

OR What is synchronizer ? Explain any one type of bit synchronizer.

☞[SPPU : Dec.-12, Marks 6, May-12, Marks 8]
Dec.-17, Marks 7, Oct.-18, Marks 6,
Oct.-16, Marks 5, June-22, Marks 6, Dec.-18, Marks 8]

Ans. : Need of Synchronization : The signals from various sources are transmitted on the single channel by multiplexing. This requires synchronization between transmitter and receiver. Special synchronization bits are added in the transmitted signal for this purpose.

Synchronization is also required for detectors to recover the digital data properly from the modulated signal.

Types of Synchronization

There are three broad types of synchronization. They are,

1. Carrier Synchronization.
2. Symbol and Bit Synchronization.
3. Frame Synchronization.

Early-Late gate Synchronization (Bit Synchronization)

Principle : The two correlators receive the signal. One correlator receives early and other receives late. The difference of the output of two correlators is used to drive VCO. The output of VCO is the corrected clock signal.

Block diagram and operation

- Fig. Q.20.1 shows the block diagram of early-late gate synchronizer. Observe that the output of VCO is advanced (early) by 'δ' and given to correlator - . The output of VCO is delayed (late) by 'δ' and given to correlator - 2.

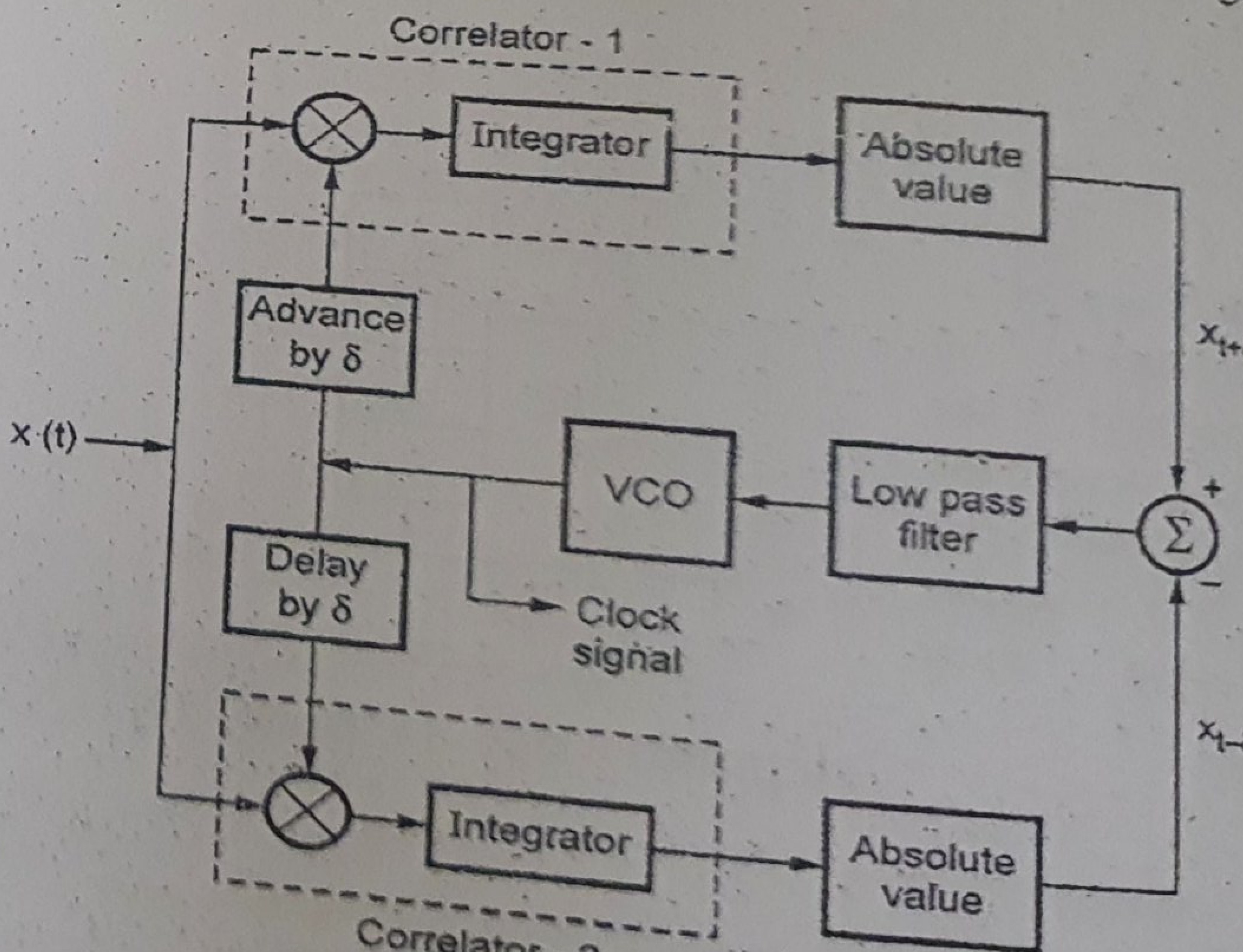


Fig. Q.20.1 Early-Late gate synchronizer

igital data

receives of two orrected

onizer. iven to iven to

e absolute values of two correletors are given to the subtractor. The ference of these absolute values of correlators is low pass filtered and becomes correction signal for VCO.

hen there is perfect synchronization, then the outputs of two relators $x_{t+\delta}$ and $x_{t-\delta}$ are same and their differnece is zero. Hence O does not change its fequency.

the clock is slightly advanced, then $x_{t+\delta} < x_{t-\delta}$. This reduces the or signal and hence delays the VCO phase. This in turn slows down clock signal.

nilarly if the clock is slightly delayed, then $x_{t+\delta} > x_{t-\delta}$. This eases the error signal and hence advances the VCO phase. This in speeds up the clock signal.

ntages :

) No zero crossings are required.

) Synchronization is more accurate.

vantages :

) Integrators are not perfect.

) Delay/Advance of clock is not accurate.

What is frame synchronization ?

☞[SPPU : Dec-12, Marks 3, Dec.-22, Marks 5]

: • In the time division multiplexing of the data, the signal samples from each input channel forms a frame. The samples of one channel led word of the frame. Depending on bits used for encoding, the length is defined. Thus each word contains some fixed number of he receiver has to know when a particular frame starts and when its dual message bits starts. This type of synchronization is called *frame onization*. For the frame synchronization, the binary transmission es special N-bit sync words as shown in Fig. Q.21.1. The initial has several repetitions of the sync word.

ord	N-bit sync word	N-bit sync word	Code word	← Actual message bits →	N-bit sync word	.....	

Prefix

Start of message

Fig. Q.21.1 Frame synchronization

- The prefix allows the time for bit sync acquisition and indicates the beginning of a frame. After the prefix, there is one more codeword. It indicates the start of message.
- The message bits follow the special codeword. To maintain the synchronization of message bits, the sync words are inserted periodically in the bit stream.

6.5 : Intersymbol Interference

Important points to remember

- Presence of outputs due to other symbols at the time of sampling the required symbol is called ISI.
- Raised cosine spectrum is used to reduce the effect of ISI.

Q.22 What is ISI ? Explain its causes and remedies to avoid it.

☞[SPPU : Dec.-15, In sem, Marks 8, Dec.-13, May-12, 14, Marks 10, Dec.-16,Oct.-18, Marks 6]

Ans. : ISI : The presence of outputs due to other bits (symbols) interfere with the output of required bit (symbol). This effect is called Intersymbol Interference (ISI)

The output of the receiver at t_i is given as,

$$y(t_i) = \underbrace{\mu A_i}_{\text{output due to } i^{th} \text{ bit}} + \underbrace{\mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} A_k \, p[(i-k)T_b]}_{ISI}, \quad i = 0, \pm 1, \pm 2, \ldots$$

Here μA_i is the contribution in output due to i^{th} transmitted bit.

- The second term in above equation represents residual effect of all other bits transmitted before and after the sampling instant 't_i'. It is ISI.

- **Causes of ISI**

i) Timing inaccuries : The ISI occurs if the rate of transmission of the transmitter is not same as the ringing and frequency of the channel.

ii) Insufficient BW : If the channel BW is reduced, then the timing error might increase in turn resulting ISI.

iii) Amplitude distortion : When the frequency characteristics of the communication channel differs from the expected one resulting pulse distortion leadings to ISI.

iv) Phase distortion : When various frequency components in the input undergo different amounts of time delay while travelling through the channel leads to phase distortion and becoming a cause for ISI

Remedies to reduce ISI

- Following are the conditions to be satisfied by the transmitted pulse to have zero ISI.

$$\rho[(i-k)T_b] = \begin{cases} 1 \text{ for } & i=k \\ 0 \text{ for } & i \neq k \end{cases}$$

or

$$\sum_{n=-\infty}^{\infty} p(f-nf_b) = T_b$$

- Above condition is satisfied by following pulse,

$$p(t) = \text{sinc}(2B_o t)$$

But this pulse is physically unrealizable.

- Raised cosine spectrum gives rise to the time domain pulse that is physically realizable. i.e.,

$$\rho(t) = \text{sinc}(2B_0 t)\frac{\cos(2\pi\alpha B_0 t)}{1-16\alpha^2 B_0{}^2 t^2}$$

This pulse becomes zero at the sampling instants of $\pm T_b$, $\pm 2T_b$ ... and so on. It eliminates ISI.

Q.23 State the Nyquist first criterion for zero ISI. Also explain raised cosine spectrum to reduce ISI.

☞[SPPU : Dec.-14, In sem, Marks 3]

Ans. : Nyquist gives criterion for zero ISI in time as well as frequency domains. It is given below :

$$\rho[(i-k)T_b] = \begin{cases} 1 \text{ for } & i=k \\ 0 \text{ for } & i \neq k \end{cases}$$

and

$$\sum_{n=-\infty}^{\infty} P(f-nf_b) = T_b$$

Above time and frequency domain equations give Nyquist pulse shaping criterion for zero ISI.

- The pulse which satisfies above criteria is,

$$p(t) = \frac{\sin(2\pi B_0 t)}{2\pi B_0 t}$$

Here B_0 is the Nyquist bandwidth. And pulse $p(t)$ provides zero ISI. Note that this pulse is not physically realizable since it extends from $-\infty \leq t \leq \infty$.

- **Raised cosine spectrum :** In raised cosine spectrum, the frequency response P(f) decreases gradually towards zero. The corresponding time domain pulse is given as,

$$p(t) = \text{sinc}(2B_0 t)\frac{\cos(2\pi\alpha B_0 t)}{1-16\alpha^2 B_0{}^2 t^2}$$

Here roll off factor, $\alpha = 1 - \frac{f_1}{B_0}$ ... (Q.23.1)

The roll of takes place from f_1 to $2B_0 - f_1$

The bandwidth required for raised cosine spectrum is given as,

$$B = 2B_0 - f_1$$

From equation (Q.23.1) $f_1 = B_0 - B_0\alpha$. Putting this value in above equation,

$$B = 2B_0 - B_0 + B_0\alpha$$

$$\therefore \quad B = B_0(1+\alpha)$$

This is the bandwidth required by raised cosine spectrum.

Q.24 A computer gives a binary data at the rate of 56 kbps and its transmitted using baseband PAM system that is designed to have a raised cosine spectrum. Determine transmission BW required for roll off rates i) α = 0.25 ii) α = 0.75 **☞[SPPU : May-14, Marks 8]**

Ans. : Data rate is, $f_b = 56$ kbps

$$\therefore \quad BW,\ B_0 = \frac{f_b}{2} = 28 \text{ kbps}$$

Now BW required using raised cosine spectrum is given by,

$$B = B_0(1+\alpha)$$

i) for $\alpha = 0.25$,

$$B = 28 \times 10^3 (1+0.25) = 35 \text{ kHz}$$

for $\alpha = 0.75$,

$$B = 28\times10^3(1+0.75) = 49 \text{ kHz}$$

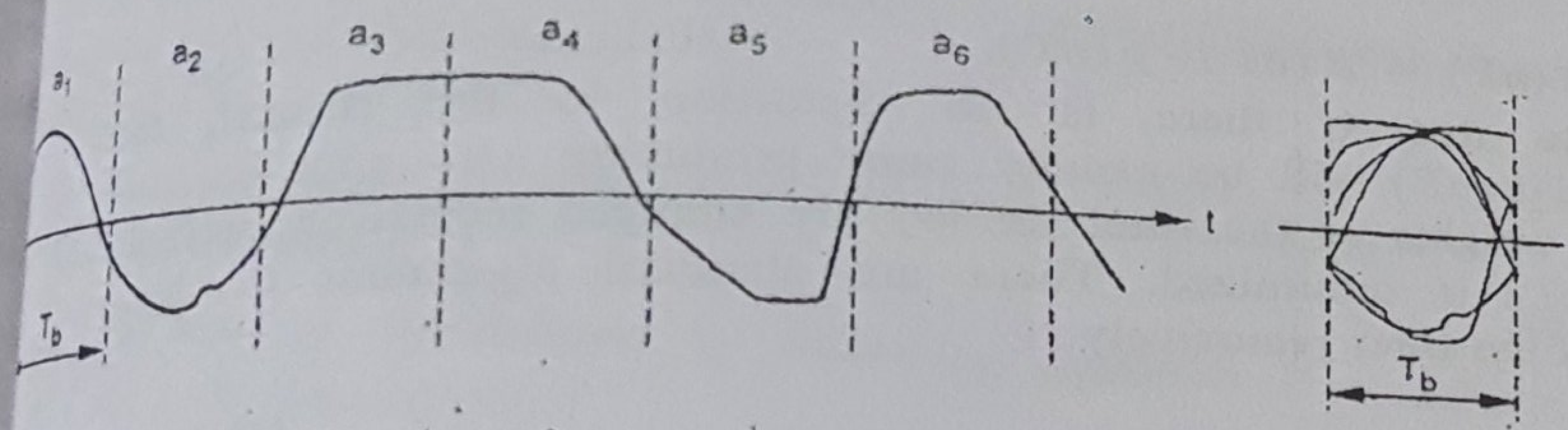
...s roll off factor increases, BW also increases

Explain the use of eye diagram to measure ISI

☞[SPPU : Dec.-13, Marks 4, June-22, Dec.-22, Marks 6]

Explain eye diagram. ☞[SPPU : May-11, Marks 4]

...: • Eye pattern is used to study the effect of ISI on baseband digital ...ission. When the signal is transmitted over the channel in the form ...ses, it gets distorted. Such signal is shown in Fig. Q.25.1 (a)



(a) Signal waveform (b) Eye diagram

Fig. Q.25.1 Eye diagram

... is the interval of one bit. a_1, a_2, a_3, a_4, a_5 and a_6 are the ...smitted bits. If we cut the waveforms of $a_1, a_2, \ldots a_6$ in their ...pective bit intervals and paste over one nother, we set the diagram shown in Fig. Q.25.1 (b).

...s diagram looks like 'eye', ...ce it is called eye diagram.

...s eye pattern indicates the ...ect of intersymbol ...rference.

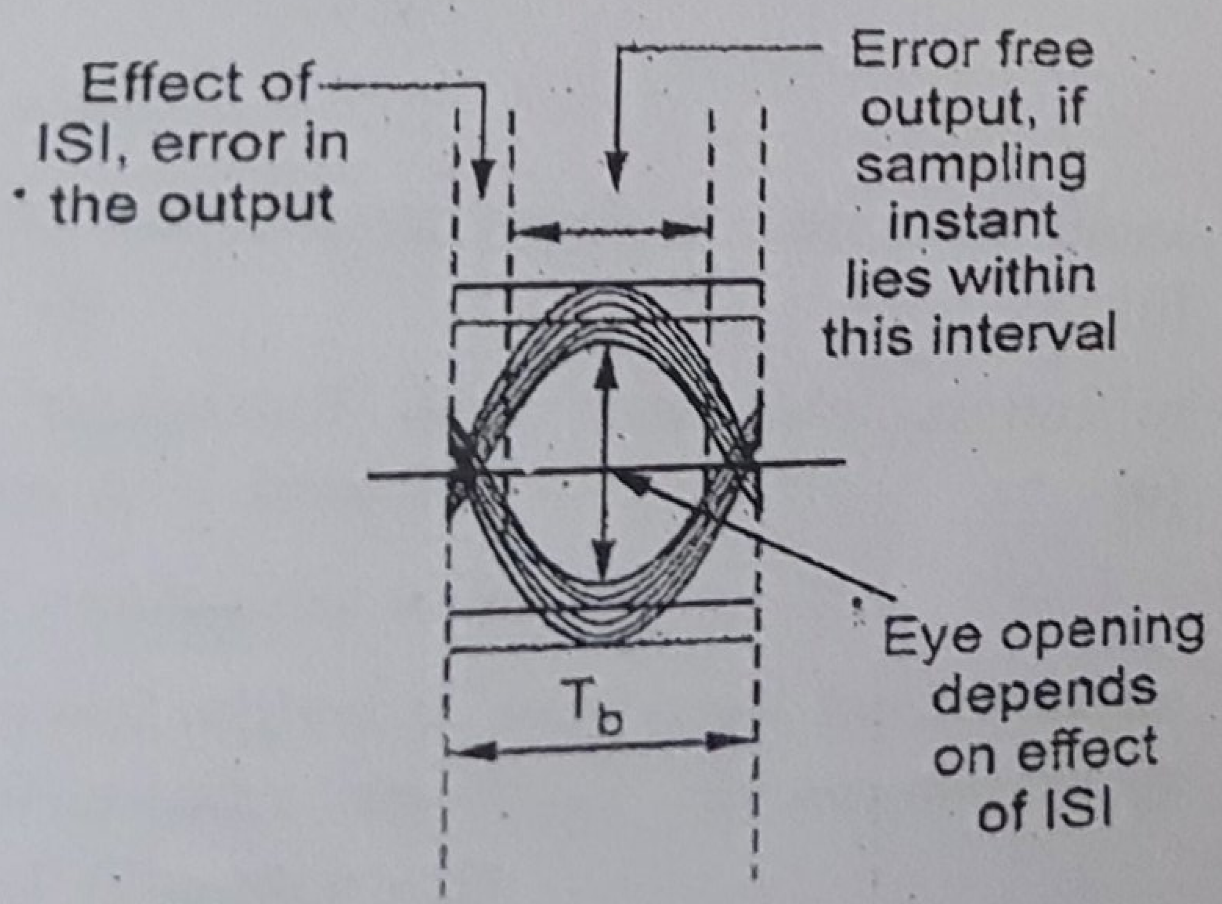


Fig. Q.25.2

...of eye diagram

...rious important conclusions ...n be derived from eye ...ttern. The interval over ...hich received waveform can ... sampled without error can ... obtained from eye pattern.

- Sensitivity of the system to timing error can be determined from eye diagram.
- The maximum clear eye opening indicates margin over noise.
- Fig. Q.25.2 indicates various parameters of eye diagram discussed above.

6.6 : Equalization

Important points to remember

- The distortion introduced along the channel is corrected with the help of specially designed filter. It is called equalization.
- Tapped delay line filter and adaptive filter are used for equalization.

Q.26 What is equalization ? What is its necessity ?

☞[SPPU : May-16, June-22, Marks 2]

Ans. : Equalization : The channel distortion such as ISI or effect of gaussian noise affects the signal being transmitted. This distortion is corrected with the help of special filters. It is called equalization.

Fig. Q.26.1 shows the block diagram of equalization.

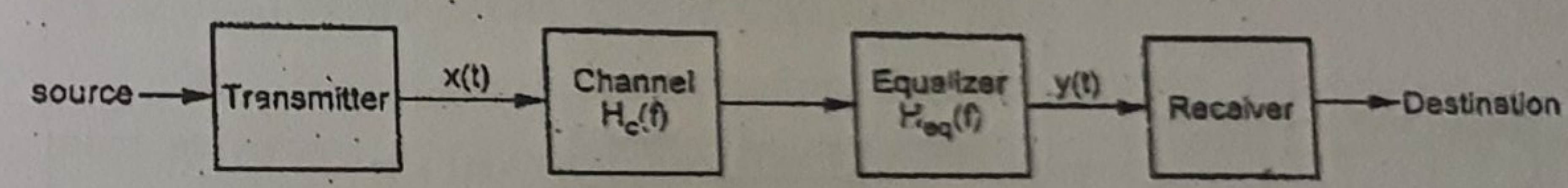


Fig. Q.26.1 Equalizer for correction of distortion introduced in the channel

The transfer function of distortionless system is given as,

$$H(f) = K e^{-j2\pi f t_0}$$

The cascade connection of channel + equalizer shown in above figure will be distortionless if,

$$H_c(f)\cdot H_{eq}(f) = K e^{-j2\pi f t_0}$$

Hence transfer function of the equalizer will be,

$$H_{eq}(f) = \frac{K e^{-j2\pi f t_0}}{H_c(f)} \quad \ldots (Q.26.1)$$

Necessity : i) Effect of ISI is corrected.

ii) Effect of channel noise is corrected.

Q.27 Explain adaptive equalization with necessary block diagram.

[SPPU : May-16, End sem, Marks 5, June-22, Marks 4]

Ans. : Necessity : Most of the channels are made up of individual links. For example, in the switched telephone network, the distortion induced depends upon

i) Transmission characteristics of individual links and

ii) Number of links in connnection.

Hence, the fixed pair of transmit and receive filters will not serve the equalization problem completely. The transmission characteristics of the channel keep on changing. Hence *adaptive equalization* is used.

Basic Principle : In adaptive equalization, the filters adapt themselves to the dispersive effects of the channel. That is the coefficients of the filters are changed continuously according to the received data. The filter coefficients are changed in such a way that the distortion in the data is reduced.

Block diagram : The adaptive equalizer shown in figure below is a tapped-delay-line filter. It consists of set of delay elements and variable

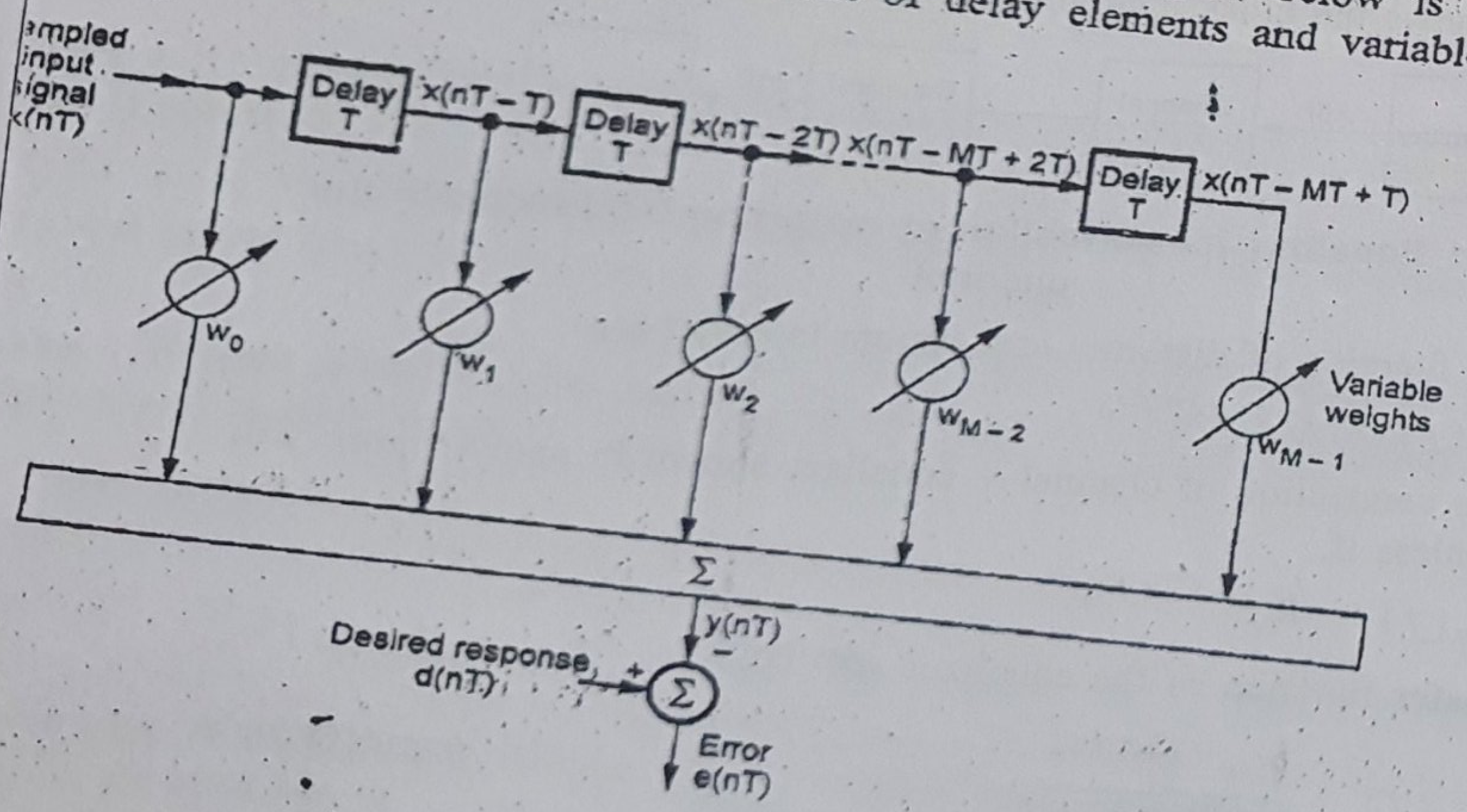


Fig. Q.27.1 Structure of adaptive equalizer

multipliers. The sequence $x(nT)$ is applied to the input of the adaptive filter. The output $y(nT)$ of the adaptive filter will be,

$$y(nT) = \sum_{i=0}^{M} w_i\, x(nT - iT) \qquad \dots (Q.27.1)$$

The weights w_i on the taps are basically adaptive filter coefficients. A known sequence $\{d(nT)\}$ is transmitted first. This sequence is known to the receiver. The response sequence $y(nT)$ is observed. As shown in Fig. Q.27.1, the error sequence between the two sequences is calculated. i.e.,

$$e(nT) = d(nT) - y(nT), \qquad n = 0, 1, \dots N-1 \qquad \dots (Q.27.2)$$

Here note that if there is no distortion in the channel, then $d(nT)$ and $y(nT)$ will be exactly same producing zero error sequence. Then the weights of the filter i.e. w_i are changed recursively such that error $e(nT)$ is minimized. There are standard algorithms to change weights of the filter recursively.

END...

ransmission

adaptive

(Q.27.1)

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Q.27.2)
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JUNE - 2022 [5869]-248 Solved Paper

Course 2019

ne : 2½ Hours] [Maximum Marks : 70

ructions to the candidates :

1) Answer Q.1 or Q.2, Q.3 or Q.4, Q.5 or Q.6, Q.7 or Q.8.

2) Neat diagrams must be drawn wherever necessary.

3) Assume suitable data, if necessary.

a) Define modulation index and deviation ratio of FM and sketch waveform for sinusoidal input.
fer Q.4 of Chapter - 3) [6]

b) Compare frequency modulation with phase modulation.
fer Q.26 of Chapter - 3) [6]

c) FM wave is represented by following equation V = 20 sin t + 4 sin 1200 t] calculate,
er Q.8 of Chapter - 3) [6]

rrier frequency

odulating frequency

odulation index and maximum deviation

ower dissipated by FM wave in 8 Ω resistor.

OR

a) Explain FM generation by Armstrong method with neat block am. (Refer Q.23 of Chapter - 3) [6]

b) A carrier is frequency modulated with sinusoidal signal of resulting in frequency deviation of 6 kHz. [6]

nd BW and modulation index of modulated wave

amplitude of modulating sinusoidal signal is increased by 2 and its ency is halved, find maximum frequency deviation and bandwidth of modulated signal. (Refer Q.11 of Chapter - 3)

c) Explain pre-emphasis in FM with circuit diagram and frequency response.
(Refer Q.42 of Chapter - 2 and Q.22 of Chapter - 3) [6]

Q.3 a) State sampling theorem in time domain. Explain sampling process with block diagram. (Refer Q.1 of Chapter - 4) [6]

b) Describe generation of pulse width modulation with diagram and waveform. (Refer Q.12 of Chapter - 4) [6]

c) Explain Aliasing effect and draw the sampled output for sampling frequency less than equal to and greater than maximum frequency of analog signal. (Refer Q.2 of Chapter - 4) [5]

OR

Q.4 a) Compare pulse amplitude modulation and pulse position modulation. (Refer Q.14 of Chapter - 4) [6]

b) Define time division multiplexing. Explain concept of TDM with neat diagram. (Refer Q.17 of Chapter - 4) [6]

c) Describe detection of PPM with block diagram.
(Refer Q.16 of Chapter - 4) [5]

Q.5 a) Draw block diagram of PCM system and describe working of PCM transmitter. (Refer Q.5 of Chapter - 5) [6]

b) State types of quantization. Explain uniform quantization with neat waveform. (Refer Q.2 of Chapter - 5) [6]

c) Discuss with neat schematic, transmitter and receiver for DPCM (Differential Pulse Code Modulation).
(Refer Q.30 of Chapter - 5) [6]

OR

Q.6 a) Compare analog and digital communication.
(Refer Q.3 of Chapter - 5) [6]

b) Draw block diagram of delta modulation system and comment on drawback of delta modulation. (Refer Q.20 of Chapter - 5) [6]

c) Explain working of adaptive delta modulation with block diagram and state advantages of ADM over DM.
(Refer Q.23 of Chapter - 5) [6]

is the maximum rate

Q.7 a) *Draw the following data formats for bit stream 10110100101*

i) Unipolar RZ ii) Unipolar NRZ iii) Polar RZ iv) Polar NRZ v) AMI (Alternative mark inversion) vi) Split phase manchester

(Refer Q.2 of Chapter - 6) [6]

b) *State different synchronization technique and explain any one in detail with neat diagram.* **(Refer Q.20 of Chapter - 6)** [6]

c) *Define equalizer. Explain adaptive equalization with block diagram and State advantages of adaptive equalization.* **(Refer Q.27 of Chapter - 6)** [5]

OR

Q.8 a) *Explain the working principle of scrambling and unscrambling with example.* **(Refer Q.15 of Chapter - 6)** [6]

b) *Describe eye pattern graphical display of inter symbol interference with diagram.* **(Refer Q.25 of Chapter - 6)** [6]

c) *Describe concept of digital multiplexer and demultiplexer with necessary diagram.* **(Refer Q.13 of Chapter - 6)** [5]

DECEMBER - 2022 [5925]-218 Solved Paper

Course 2019

Time : $2\frac{1}{2}$ Hours] **[Maximum Marks : 70**

Instructions to the candidates :

1) *Solve Q.1 or Q.2, Q.3 or Q.4, Q.5 or Q.6, Q.7 or Q.8.*
2) *Neat diagrams must be drawn wherever necessary.*
3) *Figures to the right indicate full marks.*
4) *Use of logarithmic tables slide rule, Mollier charts, electronic pocket calculator and steam tables is allowed.*
5) *Assume suitable data, if necessary.*

Q.1 a) *Explain with the help of neat block diagram Armstrong method of FM generation.* **(Refer Q.23 of Chapter - 3)** [6]

b) *Differentiate between NBFM and WBFM.* **(Refer Q.14 of Chapter - 3)** [6]

c) *Explain pre-emphasis and de-emphasis in detail.* **(Refer Q.22 of Chapter - 3 and Q.42 of Chapter - 2)** [6]

OR

Q.2 a) *With the help of block diagram explain superheterodyne FM receiver.* **(Refer Q.27 of Chapter - 3)** [6]

b) *With neat phasor diagram explain balanced slope detector in FM.* **(Refer Q.31 of Chapter - 3)** [6]

c) *A frequency modulated signal is given by,*

$x_c(t) = 10\cos\{[2\pi\times 10^8 t] + s\sin[2\pi\times 200t]\}$

Determine :

i) The carrier frequency

ii) Peak frequency deviation

iii) The modulation index. **(Refer similar Q.12 of Chapter - 3)** [6]

Q.3 a) *Discuss PWM generation and detection in detail.* **(Refer Q.12 of Chapter - 4)** [6]

b) *Distinguish between PAM, PWM and PPM.* **(Refer Q.14 of Chapter - 4)** [6]

c) *What is aliasing ? How can it be avoided.* **(Refer Q.2 of Chapter - 4)** [5]

OR

Q.4 a) *Explain flat-top sampling with waveforms.* **(Refer Q.8 of Chapter - 4)** [6]

b) *State and explain the sampling theorem in detail when $f_s > 2f_m$, $f_s = 2f_m$, $f_s < 2f_m$.* **(Refer Q.2 of Chapter - 4)** [6]

c) *Distinguish between Ideal sampling, natural sampling and flat-top sampling.* **(Refer Q.10 of Chapter - 4)** [5]

Q.5 a) *Describe with suitable block diagram pulse code modulation transmitter.* **(Refer Q.5 of Chapter - 5)** [6]

b) *Explain need of digital communication.* **(Refer Q.3 of Chapter - 5)** [6]

c) *Describe compading methods μ - law and A - law.*
(Refer Q.17 of Chapter - 5) [6]

OR

Q.6 a) *Draw and explain PCM receiver.*
(Refer Q.4 of Chapter - 5) [6]

b) *Distinguish between DM and ADM.*
(Refer Q.31 of Chapter - 5) [6]

c) *Explain in detail distortion in delta modulation.*
(Refer Q.26 of Chapter - 5) [6]

Q.7 a) *Draw and explain CCITT hierarchy of multiplexing.*
(Refer Q.13 of Chapter - 6) [6]

b) *Draw line code formats for 10110100.*
i) RZ unipolar ii) NRZ polar
iii) RZ polar iv) Alternate mark inversion.
(Refer Q.7 of Chapter - 6) [6]

c) *Draw and explain frame synchronizer.*
(Refer Q.21 of Chapter - 6) [5]

OR

Q.8 a) *What is an eye diagram ? Explain the use of eye diagram to measure ISI.* (Refer Q.25 of Chapter - 6) [6]

b) *Explain scrambling and unscrambling with diagram in detail.*
(Refer Q.15 of Chapter - 6) [6]

c) *Discuss the properties of line codes.*
(Refer Q.1 of Chapter - 6) [5]

END...

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