PRINCIPLES OF COMMUNICATION (BEC-28)

UNIT-3
NOISE

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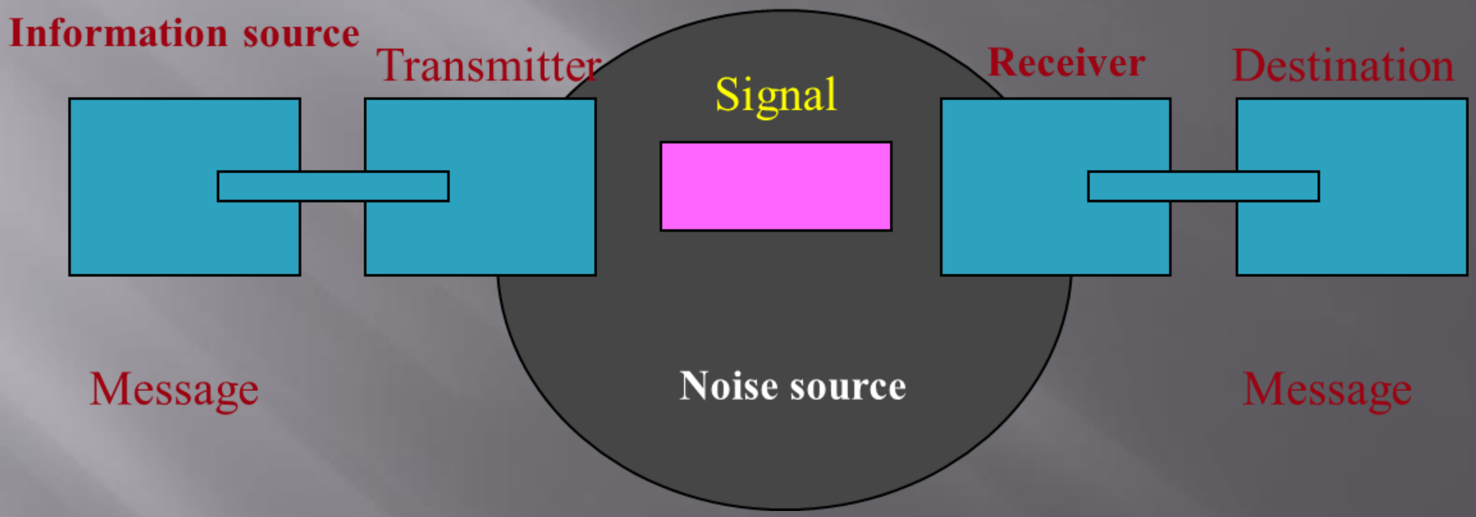
Content of Unit-3

- **Noise:** Source of Noise, Frequency domain, Representation of noise, Linear Filtering of noise, Noise in Amplitude modulation system, Noise in SSB-SC,DSB and DSB-C, Noise Ratio, Noise Comparison of FM and AM, Pre-emphasis and De-emphasis, Figure of Merit.

Noise in Communication Systems

- Claude E. Shannon conceptualized the communication theory model in the late 1940s.
- It remains central to communication study today.
- Noise is random signal that exists in communication systems



- Channel is the main source of noise in communication systems
- Transmitter or Receiver may also induce noise in the system

There are mainly 2-types of noise sources

- Internal noise source (are mainly internal to the communication system)
- External noise source

Noise Effects

Noise is an inconvenient feature which affects the system performance. Following are the effects of noise

- Degrade system performance for both analog and digital systems
- Noise limits the operating range of the systems

  Noise indirectly places a limit on the weakest signal that can be amplified by an amplifier. The oscillator in the mixer circuit may limit its frequency because of noise. A system's operation depends on the operation of its circuits. Noise limits the smallest signal that a receiver is capable of processing
- The receiver can not understand the sender
- the receiver can not function as it should be.
- Noise affects the sensitivity of receivers:

  Sensitivity is the minimum amount of input signal necessary to obtain the specified quality output. Noise affects the sensitivity of a receiver system, which eventually affects the output.
- Reduce the efficiency of communication system.

Noise Vs Interference

- Noise is a general term which is used to describe an unwanted signal which affects a wanted signal.

- Interference arises for example, from other communication systems (cross talk), 50 Hz supplies (hum) and harmonics, ignition (car spark plugs) motors … etc.

Types of Noise

The classification of noise is done depending on the type of the source, the effect it shows or the relation it has with the receiver, etc.

There are two main ways in which noise is produced. One is through some **external source** while the other is created by an **internal source**, within the receiver section.

External Noise

This noise is produced by the external sources which may occur in the medium or channel of communication, usually. This noise cannot be completely eliminated. The best way is to avoid the noise from affecting the signal.

Examples

Most common examples of this type of noise are −

- Atmospheric noise (due to irregularities in the atmosphere).
- Extra-terrestrial noise, such as solar noise and cosmic noise.
- Industrial noise.

Internal Noise

This noise is produced by the receiver components while functioning. The components in the circuits, due to continuous functioning, may produce few types of noise. This noise is quantifiable. A proper receiver design may lower the effect of this internal noise.

Examples

Most common examples of this type of noise are −

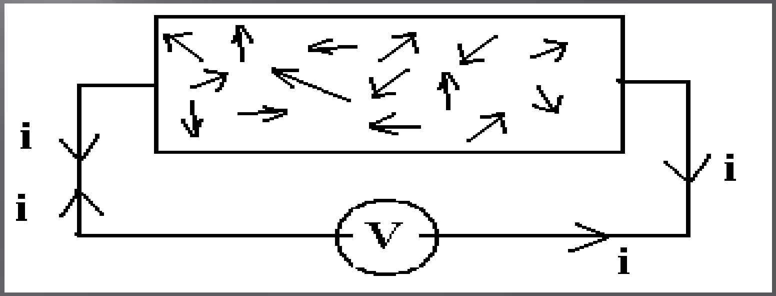
- Thermal agitation noise (Johnson noise or Electrical noise).
- Shot noise (due to the random movement of electrons and holes).
- Transit-time noise (during transition).
- Miscellaneous noise is another type of noise which includes flicker, resistance effect and mixer generated noise, etc.

Thermal Noise (Johnson Noise)

This type of noise is generated by all resistances (e.g. a resistor, semiconductor, the resistance of a resonant circuit, etc.).

Experimental results (by Johnson) and theoretical studies (by Nyquist) give the mean square noise voltage as

$$\overline{V}^{2} = 4kTBR\ (volt^{2})$$



Where k = Boltzmann's constant = 1.38 x 10^{-23} Joules per K
T = absolute temperature
B = noise bandwidth measured in (Hz)
R = resistance (ohms)

Example

1. An amplifier operating over the frequency range from 18 to 20 M Hz has a 10K ohm input register. Calculate the rms noise voltage at the input to this amplifier if the ambient temperature is 270 degree Centigrade

Solution: The rms noise voltage is given by the expression

$$\bar{V} = \sqrt{4kTBR}\ (volt)$$

Given that R=10K ohm
T=273+27=300 K
B=20-18=2 MHz
k=1.38× 10^{-23} Jule/deg-K

$$\bar{V} = \sqrt{4 \times 10 \times 10^3 \times 1.38 \times 10^{-23} \times 300 \times 2 \times 10^6}\ (volt)$$

$$\bar{V}=1.82\times 10^{-5}\ volt$$

2. Two register of 20 K ohm and 50 K ohm are at room temperature of 15 degree C or 290 K for the given bandwidth of 100 KHz . Determine the thermal noise voltage generated by

A) Each Register
B) Two register in parallel
C) two register in series

Thank you

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Signal to Noise Ratio (SNR)

Signal-to-Noise Ratio (SNR) is the **ratio of the signal power to the noise power**. The higher the value of SNR, the greater will be the quality of the received output

the received signal $Y(t)$ is the sum of the transmitted signal $X(t)$ and the noise $N(t)$, i.e.

$$\boxed{Y(t) = X(t) + N(t).}$$

Since $X(t)$ and $N(t)$ are uncorrelated, we have superposition of signal powers, i.e.

$$R_Y(0) = R_X(0) + R_N(0) \quad \text{or equivalently}$$

$$\mathsf{E}\left[|Y(t)|^2\right] = \mathsf{E}\left[|X(t)|^2\right] + \mathsf{E}\left[|N(t)|^2\right].$$

Define the *signal power* and the *noise power* at the receiver as

$$\boxed{S = \mathsf{E}\left[|X(t)|^2\right] \quad \text{and} \quad N = \mathsf{E}\left[|N(t)|^2\right].}$$

In addition, the *signal-to-noise ratio (SNR)* is defined as

$$\boxed{\mathrm{SNR} = S/N}$$

Examples

1. A video signal of having BW of 100 MHz power of 1MHz is transmitted through a channel, power loss in the channel is given by 40dB. Noise PSD is given by 10^{-20} Watts/Hz, Find SNR at the input of the receiver.

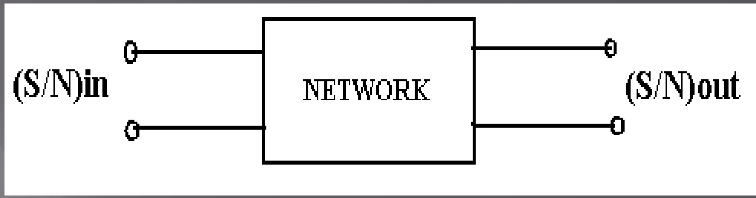
Solution:

Noise Factor or Figure

- The ratio of output SNR to the input SNR can be termed as the Figure of merit (F). It is denoted by F. It describes the performance of a device.

- The amount of noise added by the network is embodied in the Noise Factor F.

$$\text{Noise factor } F = \frac{\left(S/N\right)_{IN}}{\left(S/N\right)_{OUT}}$$



- Noise figure is a measure of the degradation in signal to noise ratio and it can be used in association with radio receiver sensitivity. Noise figure is a number by which the noise performance of an amplifier or a radio receiver can be specified. The lower the value of the noise figure, the better the performance.

Noise Performance of Various Modulation Schemes

Noise in DSB-SC

The receiver model for coherent detection of DSB-SC signals is shown in Fig. 1 The DSB-SC signal is, *s (t) = Ac m(t) cos (ωc t) . We assume m(t) to* be sample function of a WSS process *M(t) with the power spectral density, SM (f) , limited to ± W Hz.*

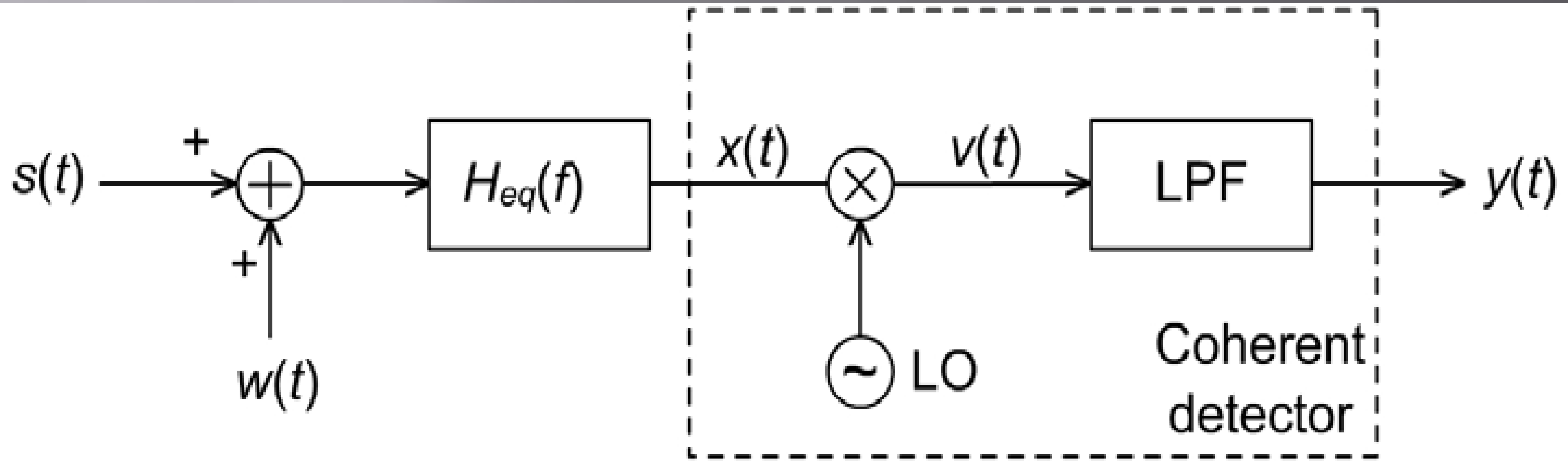


Fig. 1 Coherent Detection of DSB-SC.

the random phase added to the carrier term, *Rs (τ) , the autocorrelation function of the process S(t)* (of which *s (t) is a sample function), is given by,*

$$R_s(\tau) = \frac{A_c^2}{2} R_M(\tau)\cos(\omega_c \tau)$$

where *RM (τ) is the autocorrelation function of the message process.*

Cont….

Fourier transform of *Rs (τ) yields Ss (f) given by,*

$$S_s(f) = \frac{A_c^2}{4}\left[S_M(f - f_c) + S_M(f + f_c)\right]$$

Let *PM denote the message power, where*

$$P_M = \int_{-\infty}^{\infty} S_M(f)\,df = \int_{-W}^{W} S_M(f)\,df$$

$$\text{Then,}\quad \int_{-\infty}^{\infty} S_s(f)\,df = 2\frac{A_c^2}{4}\int_{f_c - W}^{f_c + W} S_M(f - f_c)\,df = \frac{A_c^2 P_M}{2}.$$

the average noise power in the message bandwidth $2W$ *is* $2W^*N_0/2 = W^*N_0$ .Hence,

$$\left[(SNR)_r\right]_{DSB-SC} = \frac{A_c^2 P_M}{2\,W\,N_0}$$

To arrive at the *FOM , we require (SNR)0 . The input to the detector*
x (t) = s (t) + n(t) , where n(t) is a narrowband noise quantity. Expressing n(t)
in terms of its in-phase and quadrature components, we have

$$x(t) = A_c\,m(t)\cos(\omega_c t) + n_c(t)\cos(\omega_c t) - n_s\sin(\omega_c t)$$

Assuming that the local oscillator output is cos(ω*c t) , the output v (t) of the* multiplier in the detector is given by

$$v(t) = \frac{1}{2} A_c\, m(t) + \frac{1}{2} n_c(t) + \frac{1}{2}\left[A_c\, m(t) + n_c(t)\right] \cos(2\omega_c t) - \frac{1}{2} A_c\, n_s(t) \sin(2\omega_c t)$$

As the LPF rejects the spectral components centered around 2 *fc , we have*

$$y(t) = \frac{1}{2} A_c\, m(t) + \frac{1}{2} n_c(t)$$

So, the message component at the output is (1/2)**Ac m(t).*
The average message power at the output is $(A_C)^2/2 * P_M$
As the spectral density of the in-phase noise component is *N0 for f ≤ W ,*
the average noise power at the receiver output is $2W * N_0/4 = (W * N_0)/2$ Therefore,

$$\left[(SNR)_0\right]_{DSB-SC} = \frac{\left(A_c^2/4\right) P_M}{(W N_0)/2} = \frac{A_c^2\, P_M}{2 W N_0}$$

So,

$$[FOM]_{DSB-SC} = \frac{(SNR)_0}{(SNR)_r} = 1$$

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Noise in SSB

Assuming that LSB has been transmitted, we can write *s (t) as follows:*

$$s(t) = \frac{A_c}{2} m(t)\cos(\omega_c t) + \frac{A_c}{2}\widehat{m}(t)\sin(\omega_c t)$$

where *m□ (t) is the Hilbert transform of m(t) . Generalizing,*

$$S(t) = \frac{A_c}{2} M(t)\cos(\omega_c t) + \frac{A_c}{2}\widehat{M}(t)\sin(\omega_c t).$$

We can show that the autocorrelation function of *S(t), Rs (τ) is given by*

$$R_s(\tau) = \frac{A_c^2}{4}\left[R_M(\tau)\cos(\omega_c \tau) + \hat{R}_M(\tau)\sin(\omega_c \tau)\right]$$

where *R□ M (t) is the Hilbert transform of RM (t) . Hence the average signal* power, $R_S(0) = (A_C)^2/4 * P_M$

and

$$(SNR)_r = \frac{A_c^2 P_M}{4 W N_0}$$

Noise in SSB Cont..

$$\text{Let } n(t) = n_c(t)\cos(\omega_c t) - n_s(t)\sin(\omega_c t)$$

(Note that with respect to *fc , n(t) does not have a locally symmetric spectrum).*

$$y(t) = \frac{1}{4} A_c\, m(t) + \frac{1}{2} n_c(t)$$

Hence, the output signal power is $(A_C)^2 * P_M / 16$ and the output noise power as $(W * N_0)/4$.

Thus, we obtain,

$$(SNR)_{0,SSB} = \frac{A_c^2 P_M}{16} \times \frac{4}{W N_0} = \frac{A_c^2 P_M}{4 W N_0}$$

So,

$$(FOM)_{SSB} = 1$$

Noise in DSB-C or AM

DSB-LC or AM signals are normally envelope detected, though coherent detection can also be used for message recovery. This is mainly because envelope detection is simpler to implement as compared to coherent detection.

The transmitted signal *s (t) is given by*

$$s(t) = A_c\left[1 + g_m m(t)\right]\cos(\omega_c t)$$

Then the average signal power in

$$s(t) = \frac{A_c^2\left[1 + g_m^2 P_M\right]}{2}.$$

Hence

$$(SNR)_{r,\,DSB-LC} = \frac{A_c^2\left(1 + g_m^2 P_M\right)}{2W N_0}$$

Using the in-phase and quadrature component description of the narrowband noise, the quantity at the envelope detector input, *x (t) , can be written as*

$$\begin{aligned} x(t) &= s(t) + n_c(t)\cos(\omega_c t) - n_s(t)\sin(\omega_c t) \\ &= \left[A_c + A_c g_m m(t) + n_c(t)\right]\cos(\omega_c t) - n_s(t)\sin(\omega_c t) \end{aligned}$$

Cont…

The receiver output *y (t) is the envelope of the input quantity x (t) . That is,*

$$y(t) = \left\{\left[A_c + A_c g_m m(t) + n_c(t)\right]^2 + n_s^2(t)\right\}^{\frac{1}{2}}$$

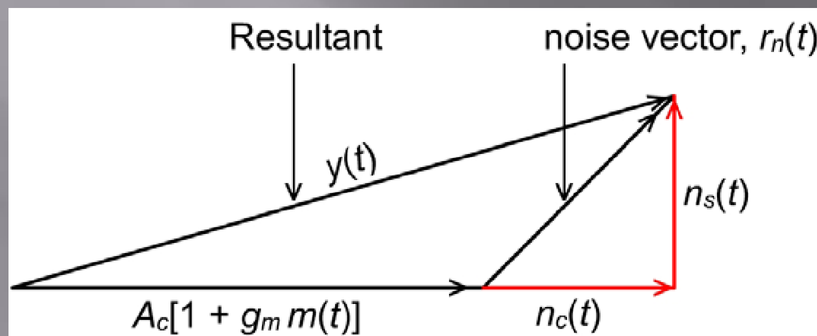

Fig. 7.5: Phasor diagram to analyze the envelope detector

The output noise power being equal to 2*W**N$_0$ *we have,*

$$\left[(SNR)_0\right]_{AM} \approx \frac{A_c^2\, g_m^2\, P_M}{2W\, N_0}$$

It is to be noted that the signal and noise are additive at the detector output and power spectral density of the output noise is flat over the message bandwidth.

$$(FOM)_{AM} = \frac{g_m^2 P_M}{1 + g_m^2 P_m}$$

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Noise in PM

For PM, ϕ(*t*) = *kp*m(t) . For convenience, let kp*kd* = *1. Then*

$$v(t) \approx m(t) + \frac{k_d\, r_n(t)}{A_c} \sin\left[\psi(t) - \varphi(t)\right]$$

Post detection LPF passes only those spectral components that are within (−*W*, *W*) .
Hence the output noise power resulting in,

$$(SNR)_{0,PM} = \frac{P_M}{2W N_0 \left(\frac{k_d}{A_c}\right)^2}$$

$$= \frac{A_c^2}{2W N_0} k_p^2 P_M$$

As, $$(SNR)_{r,PM} = \frac{\left(A_c^2\right)/2}{N_0 W}$$

We have

$$(FOM)_{PM} = \frac{(SNR)_0}{(SNR)_r} = k_p^2 P_M$$

Noise in FM

$$v(t) = \frac{k_d}{2\pi}\frac{d\theta(t)}{dt}$$

$$= k_f k_d m(t) + \frac{k_d}{2\pi A_c}\frac{d n_s(t)}{dt}$$

Again, letting *kf*kd = 1, we have*

$$v(t) = m(t) + \frac{k_d}{2\pi A_c}\frac{d n_s(t)}{dt}$$

output signal power = *PM*

Let $\quad n_F(t) = \frac{k_d}{2\pi A_c}\frac{d n_s(t)}{dt}$

Then, $\quad S_{N_F}(f) = \left(\frac{k_d}{2\pi A_c}\right)^2 |j2\pi f|^2 S_{N_S}(f)$

We can be obtained by passing *ns (t) through a differentiator with the transfer function j 2πf.*

Cont…

$$S_{N_F}(f) = \frac{k_d^2 f^2}{A_c^2} S_{N_S}(f)$$

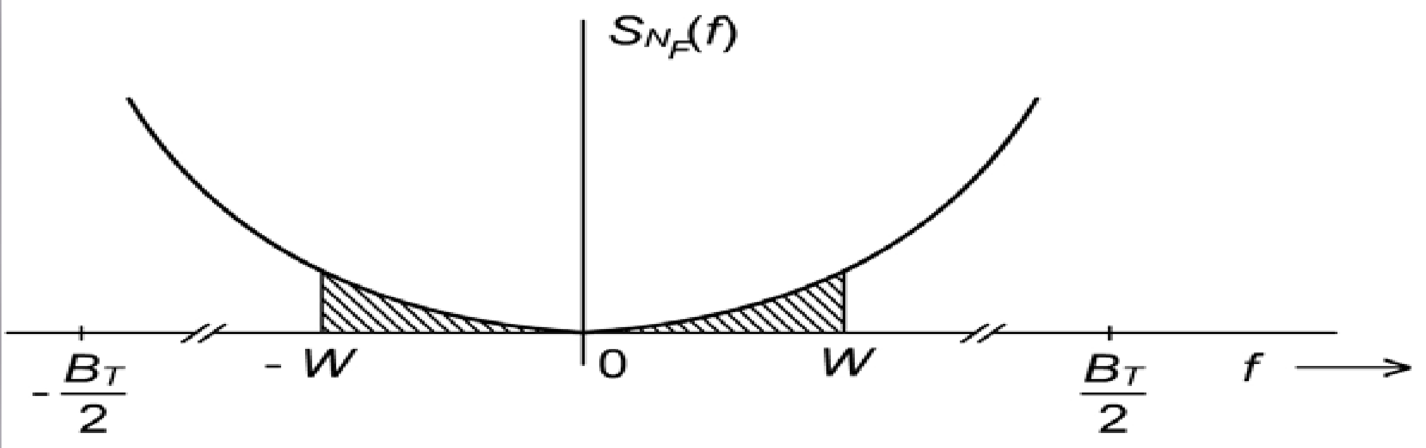


Fig. 7.13: Noise spectra at the FM discriminator output

$$\text{The output noise power} = \int_{-W}^{W} \frac{k_d^2 f^2 N_0}{A_c^2} \, df$$

$$= \frac{k_d^2 N_0}{A_c^2} \left(\frac{2}{3}\right) W^3$$

Examples

1. **For an FM, given**

$$(S/N)_{o/p} = 30\ dB$$

$$(S/N)_{i/p} = 20\ dB$$

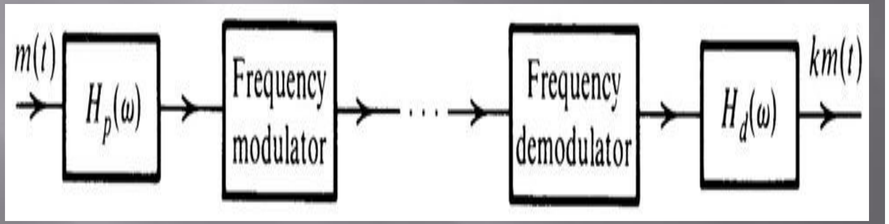
Find the value of β

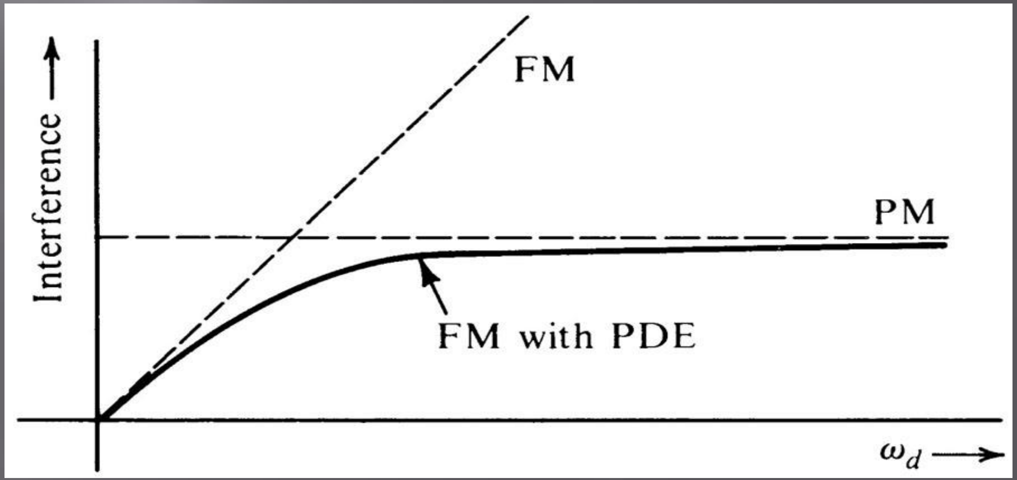
Solution:

- **2.** Compare the *FOM* of PM and FM when $m(t) = \cos(2\pi \times 5 \times 103)t$. The frequency deviation produced in both cases is 50 kHz.

Pre-emphasis & de-emphasis

- Pre-emphasis is needed in FM to maintain good signal to noise ratio.
- The characteristics of the pre-emphasis and de-emphasis filters depend largely on the PSD of the message process.
- The net effect of these filters should be a flat frequency response since the noise component before filtering has a parabolic PSD



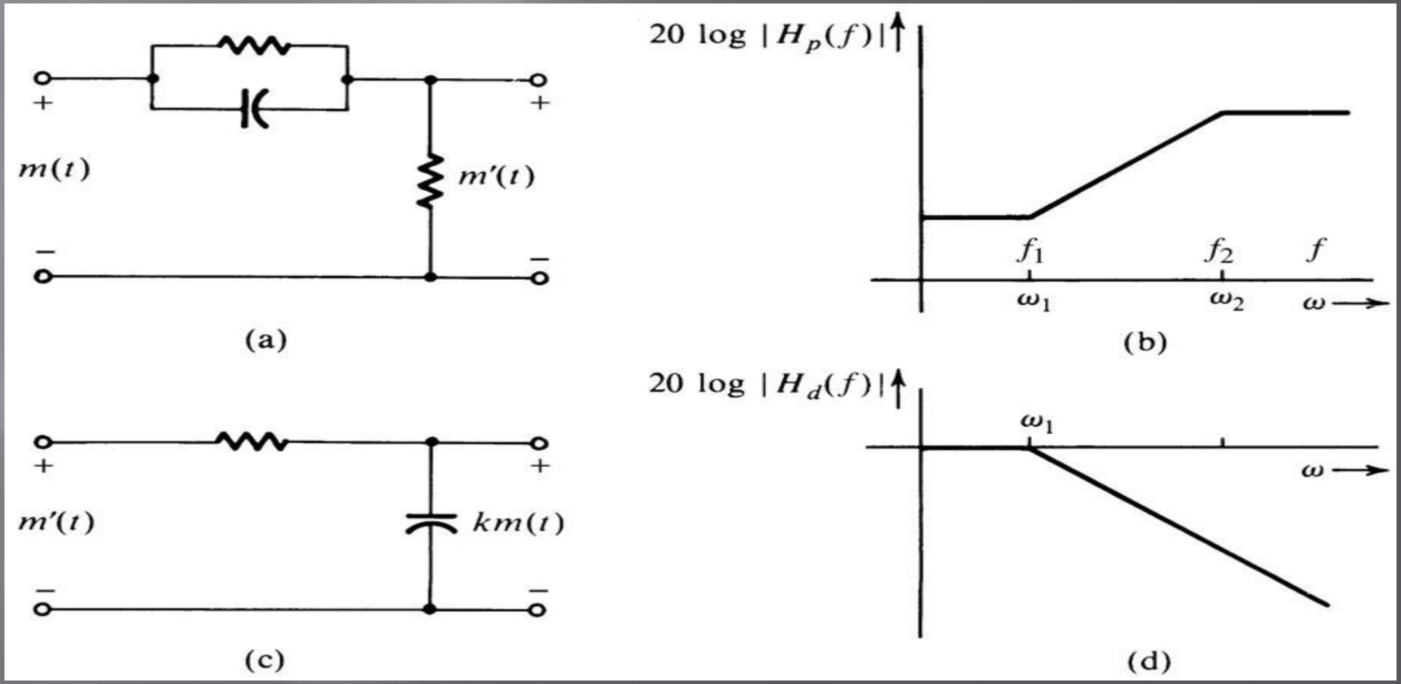


(a) Preemphasis-deemphasis in an FM system

(b) Interference Vs Pre-emphasis & De-emphasis

Pre-emphasis & de-emphasis

- In commercial FM broadcasting of music and voice, 1st order lowpass and high pass RC filters with a time constant of 75 μs are employed.
- $f_o = 1/(2\pi \times 75 \times 10^{-6}) \approx 2100$ Hz is the 3 dB frequency of the filter



Thank you

Principles of Communication (BEC-28)

Unit-4

Pulse Modulation and Digital Transmission of Analog Signal

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Content of Unit-IV

Pulse Modulation and Digital Transmission of Analog Signal: Sampling Theorem and its applications, Concept of Pulse Amplitude Modulation, Pulse width modulation and pulse position modulation, PCM, Pulse Time Modulation, TDM and FDM. Line Coding, Quantizer, Quantization Noise, Compounding multiplexer.

Pulse Modulation

Analog Pulse Modulation

- Pulse Amplitude Modulation (PAM)
- Pulse Width Modulation (PWM)
- Pulse Position Modulation (PPM)

Digital Pulse Modulation

- Pulse Code Modulation (PCM)
- Delta Modulation (DM)

Advantage of Pulse modulation:

(i) Transmitted power is no longer continuous as in CW Modulation, but pulsed in nature

(ii) Vacant time between pulse occurrence filled by interleaving/multiplexing pulse waveforms of some other Message (TDM)

Sampling Theorem

This provides a mechanism for representing a **continuous time signal** by a **discrete time signal** , taking **sufficient number of samples of signal** so that **original signal is represented in its samples completely.** It can be stated as:

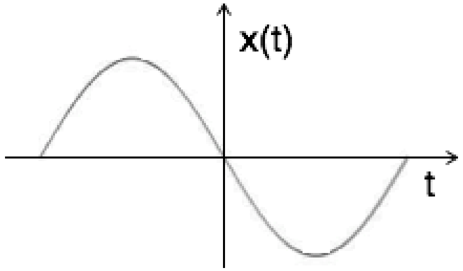
(i) A band-limited signal of finite energy with no frequency component higher **than f_m Hz**, is completely described by **its sample values** which are at uniform intervals **less than or equal to $1/2f_m$** seconds apart. [$\mathbf{T_s = \frac{1}{2fm}}$]where T_s is sampling time.

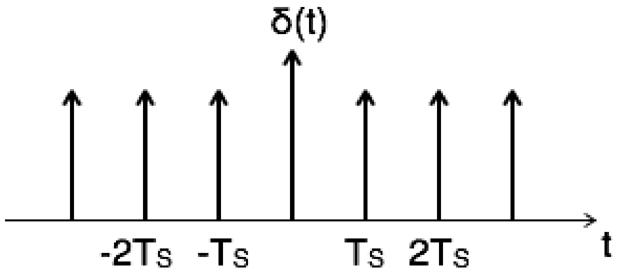
(ii) **Sampling frequency** must be **equal to or higher than $2f_m$** Hz. **[$fs \geq 2fm$]**

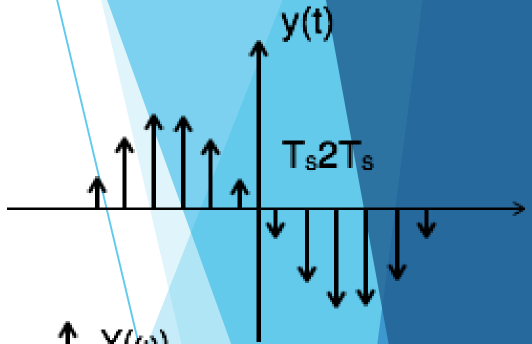
A continuous time signal may be completely represented in samples and recovered back, if $f_s \geq 2f_m$, where f_s is sampling frequency and f_m is maximum frequency component of message signal

Proof of sampling theorem

- **Sampling** of input signal **x(t)** can be obtained by **multiplying x(t)** with an **impulse train δ(t)** of period T_s.
- The output of multiplier is a discrete signal called **sampled signal** which is represented with y(t) in the diagrams,
- **y(t)=x(t).δ(t)……(1)**

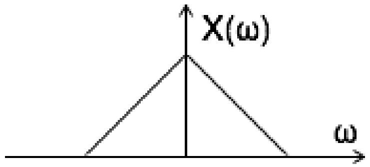


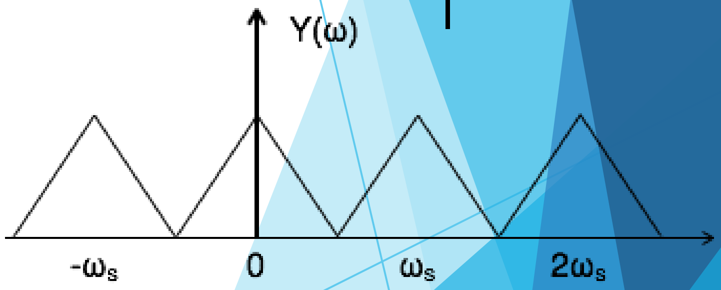




The Fourier series representation of δ(t) :

- $\boldsymbol{\delta(t)=a_0+\sum_{n=1}^{\infty}(a_n\cos n\omega_s t + b_n \sin n\omega_s t)}$ **……(2)**





where $a_0=\frac{1}{T_s}\int_{-T/2}^{T/2}\delta(t)\;dt \;=\frac{1}{Ts}\delta(0)=\frac{1}{T_s}$

$a_n=\frac{2}{T_s}\int_{-T/2}^{T/2}\delta(t)\cos n\omega_s dt \;=\frac{2}{T_s}\delta(0)\cos n\omega_s 0=\frac{2}{T_s}$

$b_n=\frac{2}{}\int\delta(t)\sin n\omega_s t\;dt \;=\frac{2}{}\delta(0)\sin n\omega_s 0=0$

$$\delta(t)=\frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)$$

$$\therefore\delta(t)=\frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)$$

Substitute δ(t) in equation 1.

$\rightarrow y(t)=x(t).\delta(t)$

$$=x(t)\left[\frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)\right]$$

$$=\frac{1}{T_s}\left[x(t)+2\sum_{n=1}^{n=\infty}(\cos n\omega_s t)x(t)\right]$$

$y(t)=\frac{1}{T_s}[x(t)+2\cos\omega_s t.x(t)+2\cos 2\omega_s t.x(t)+2\cos 3\omega_s t.x(t)......]$

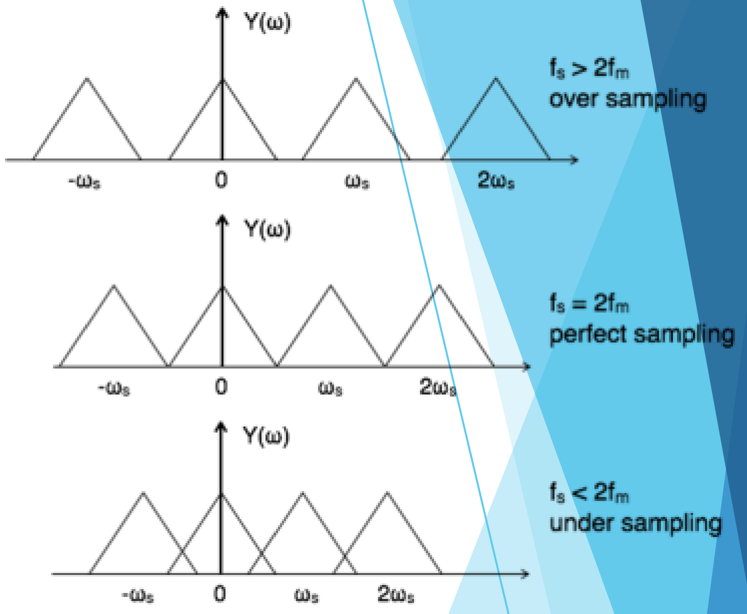
Take Fourier transform on both sides.

$$Y(\omega)=\frac{1}{T_s}[X(\omega)+X(\omega-\omega_s)+X(\omega+\omega_s)+X(\omega-2\omega_s)+X(\omega+2\omega_s)+X(\omega+3\omega_s)+..........]$$

$$Y(\omega)=\frac{1}{T_s}\sum_{-\infty}^{+\infty} X(\omega - n\omega_s)$$

To reconstruct x(t), one has to recover input signal spectrum X(ω) from sampled signal spectrum Y(ω), which is possible when there **is no overlapping between the cycles of Y(ω) which is possible if**

$$f_s \geq 2f_m$$



For **$f_s=2f_m$,** is known as **Nyquist rate.**

$T_s = \frac{1}{2fm}$ **is known as Nyquist interval**

<u>Aliasing Effect</u>

The overlapped region in case of **under sampling**

represents <u>**Aliasing effect**</u>. It can be termed as "**the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a lower-frequency component in the spectrum of its sampled version.**

This effect can be removed by considering

(i) fs >$2f_m$ or

(ii) by using anti aliasing filters which are low pass filters and eliminate high frequency components


Thank You

Principles of Communication (BEC-28)

Unit-4

Pulse Modulation and Digital Transmission of Analog Signal

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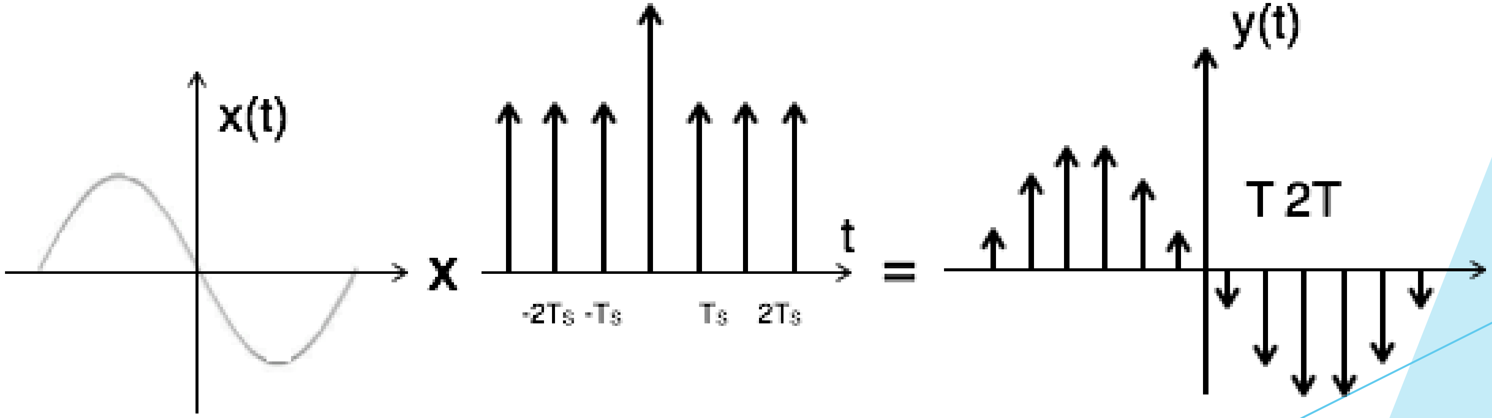
Content of Unit-IV

Pulse Modulation and Digital Transmission of Analog Signal: Sampling Theorem and its applications, Concept of Pulse Amplitude Modulation, Pulse width modulation and pulse position modulation, PCM, Pulse Time Modulation, TDM and FDM. Line Coding, Quantizer, Quantization Noise, Compounding multiplexer.

Three types of **sampling techniques**:

- **<u>Impulse sampling</u>:** Obtained by **multiplying input signal x(t) with impulse train of period 'T$_s$.**

Also called **ideal sampling**. Practically not used because pulse width cannot be zero and the generation of impulse train not possible.



<u>Natural sampling</u>

$$S(t) = \begin{cases} 1 & -\tau/2 < t < \tau/2 \\ 0 & \tau/2 < |t| < Ts/2 \end{cases} \quad \text{-------- (1)}$$

- This type of sampling similar to ideal sampling except for the fact that **instead of delta function**, now we use rectangular train of **period T_s**. i.e. multiply input signal x(t) to pulse train
- An **electronic switch** is used to periodically shift between the two contacts at a rate of **$f_s = (1/T_s)$ Hz**, staying on the input contact for C seconds and on the grounded contact for the remainder of each sampling
- The output $x_s(t)$ of the sampler consists of segments of x(t) and hence $X_s(t)$ can be considered as the product of x(t) and sampling function s(t).
- **$X_s(t) = x(t) \times s(t)$**

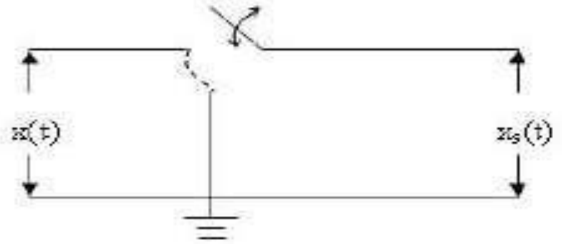


Fig: 2.11 Natural Sampling – Simple Circuit

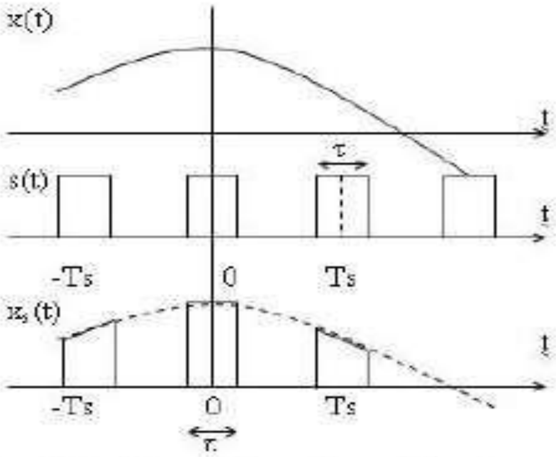
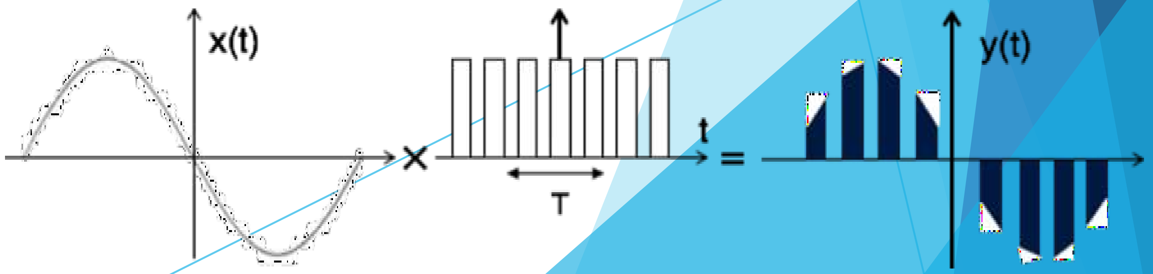


Fig: 2.12 Natural Sampling – Waveforms



Using Fourier series, we can rewrite the signal S(t) as:

$$S(t) = C_0 + \sum_{n=1}^{\infty} 2C_n\cos(n\omega_s t)$$

Where the Fourier coefficients $\mathbf{C_0 = \frac{\tau}{T_s}}$ **and** $\mathbf{C_n = f_s\tau\,sinc(nf_s\tau)}$

Therefore: $x_s(t) = x(t)[C_0 + \sum_{n=1}^{\infty} 2C_n(\cos n\omega_s t)]$

$$\mathbf{x_s(t) = C_0x(t) + 2C_1x(t)\cos(\omega_s t) + 2C_2x(t)\cos(2\omega_s t) + \ldots\ldots}$$

Applying Fourier Transform for the above equation

Using x(t)↔X(f)

$$\mathbf{x(t)\cos(2\pi f_0 t) \leftrightarrow \frac{1}{2}[X(f-f_0) + X(f+f_0)]}$$

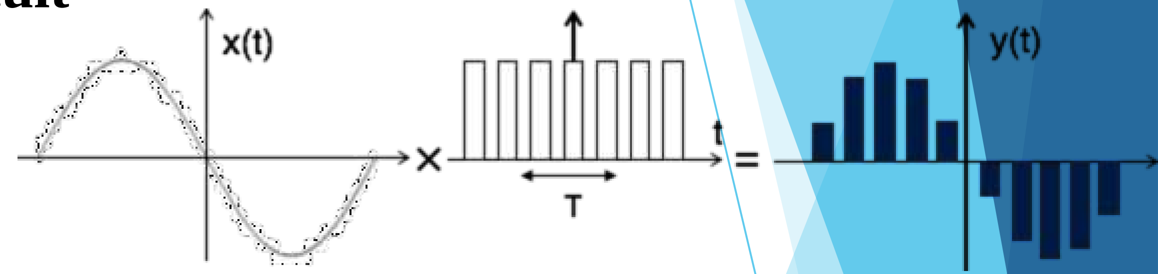
$$\mathbf{X_s(f) = C_0X(f) + C_1[X(f-f_0) + X(f+f_0)] + C_2[X(f-f_0) + X(f+f_0)] + \ldots\ldots\ldots\ldots}$$

$$\mathbf{X_s(f) = C_0X(f) + \sum_{n=-\infty}^{\infty} C_n X(f - nfs)}$$

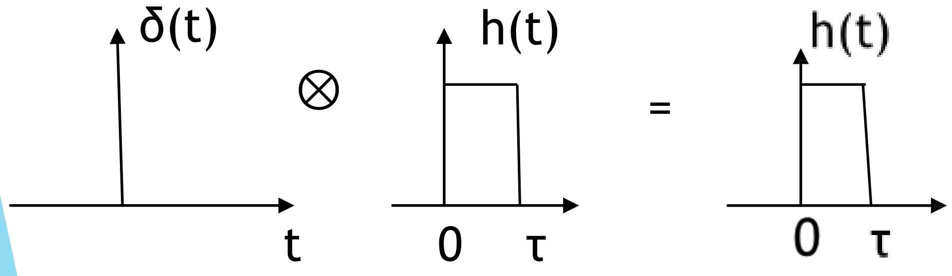
$$\mathbf{X_s(f) = A\tau/\,Ts\,.[\,\Sigma\ \sin c(n\ fs.\tau)\ X(f\text{-}n\ fs)]}$$

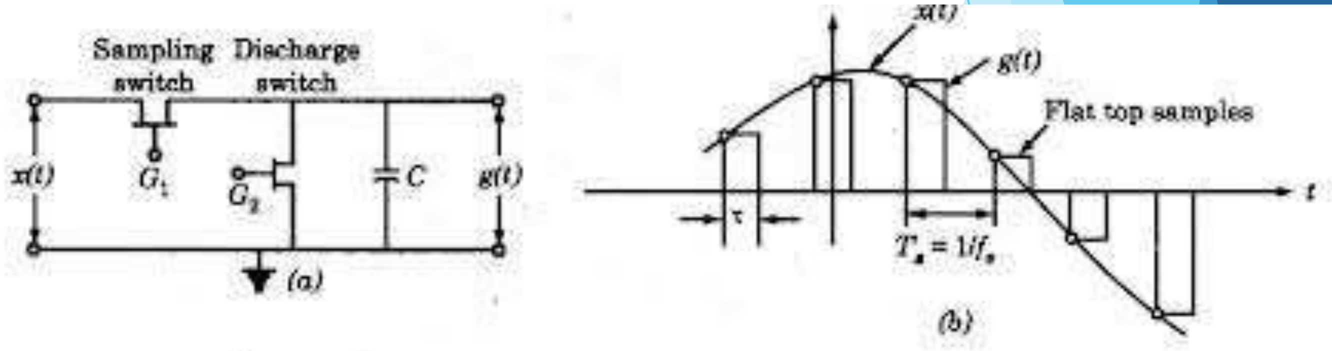
The signal $X_s(t)$ has the spectrum which consists of message spectrum and repetition of message spectrum periodically in the frequency domain with a period of fs. But **the message term is scaled by ‘Co”(** **sinc function**) which is **not the case in instantaneous sampling.**

- **<u>Flat Top sampling</u>:** During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the **pulse is in the form of flat top**.
- Here, the top of the samples are flat i.e. they have constant amplitude and is equal to the instantaneous value of the baseband signal x(t) at the start of sampling. Hence, it is called as **<u>flat top sampling or practical sampling</u>**.
- Flat top sampling makes use of **sample and hold circuit**
- Theoretically, **the sampled signal can be obtained by convolution of rectangular pulse h(t) with ideally sampled signal ,$s_\delta(t)$**



g(t)= s(t) ⊗ h(t)





f(t) ⊗ δ(t) = f(t); property of delta function
Applying a modified form; **s(t) in place of δ(t)**

On convolution of s(t) and h(t), we get a pulse whose duration is equal to h(t) only but amplitude defined by s(t).

Train of impulses given by:

$$\boldsymbol{\delta_{Ts}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nTs)}$$

Signal s(t) obtained by multiplication of message signal x(t) and $\delta_{Ts}(t)$

Thus, $\mathbf{s(t) = x(t).\ \delta_{Ts}(t)}$

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

Now sampled signal g(t) given as:

$$g(t)=s(t) \otimes h(t) \qquad\qquad G(f)=S(f)\, H(f)$$

$$= \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau \qquad\qquad S(f)=fs \sum X(f - nfs)$$

$$g(t) = \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nTs)\delta(\tau-nT_s)\, h(t-\tau)\, d\tau$$

$$\boldsymbol{g(t) = \sum_{-\infty}^{\infty} x(nTs) \int_{-\infty}^{\infty} \delta(\tau - nT_s)\, h(t-\tau)\, d\tau}$$

Using shifting property of delta function: $\int_{-\infty}^{\infty} f(t)\delta(t - to) = f(t_0)$

$$\boldsymbol{g(t) = \sum_{-\infty}^{\infty} x(nT_s)h(t - nTs)}$$

$\boldsymbol{G(f) = f_s \sum_{-\infty}^{\infty} X(f - nfs)H(f)}$ **Spectrum of flat top samples**

***Example 1:** A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?*

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

***Example 2:** A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?*

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

***Example 3:** We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?*

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Aperture Effect: Spectrum of flat topped sample is given by;

G(f)=$f_s \sum [\![X(f-nf_s)H(f)]\!]$, where **H(f)= τ.sin c(f_s.t)e^($-j\pi f\tau$)**

This equation shows that signal g(t) is obtained by passing the **signal s(t) through a filter having transfer function H(f).**

Figure(a) shows **one pulse of rectangular pulse train** and **each sample of x(t) i.e. s(t)** is **convolved** with **this pulse**

Figure (b) shows the **spectrum of this pulse**. Thus, flat top sampling introduces an **amplitude distortion** in **reconstructed signal x(t) from g(t**). There is a **high frequency roll off** making **H(f) act like a LPF**, thus **attenuating the upper portion of message signal spectrum**. This is known as <u>**aperture effect**</u>

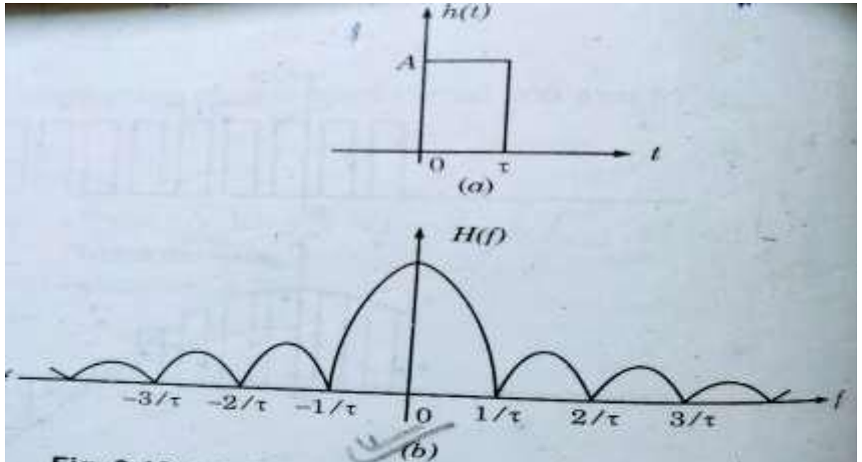


Fig. 3.18 *(a) One pulse of rectangular pulse train (b) spectrum of the pulse shown in figure (a).*

<u>How to minimize aperture effect</u>?? An **equalizer** at the receiver end is needed to compensate aperture effect. The receiver contains low pass reconstruction Filter with cut off slightly higher than f_m Hz.

PAM Signal g(t) → Reconstruction Filter → Equalizer → Message signal x(t)

Equalizer in cascade with reconstruction filter has the effect of decreasing the in band loss of reconstruction filter, **frequency increases in such away so as to compensate aperture effect.**

$$\mathbf{H_{eq}(f)} = \frac{K.e^{-j2\pi f td}}{H(f)},$$

where t_d is time delay introduced by LPF being equal to $\tau/2$

$$\mathbf{H_{eq}(f)} = \frac{K}{\tau \sin c(f\tau)}$$


Thank You

Principles of Communication (BEC-28)

Unit-4

Pulse Modulation and Digital Transmission of Analog Signal

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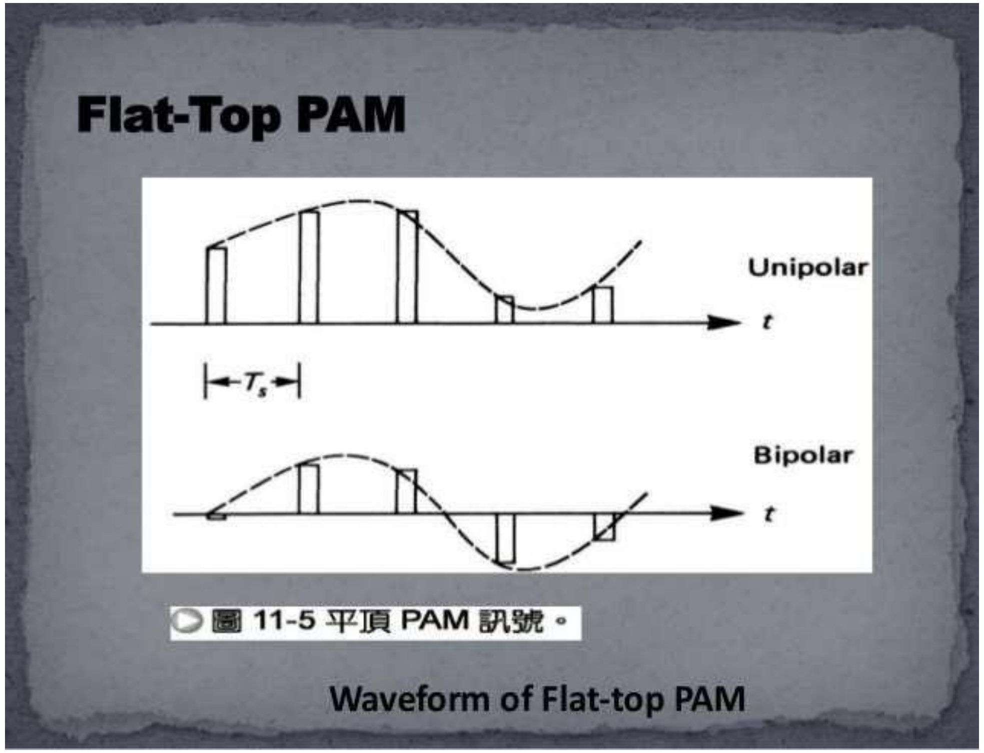
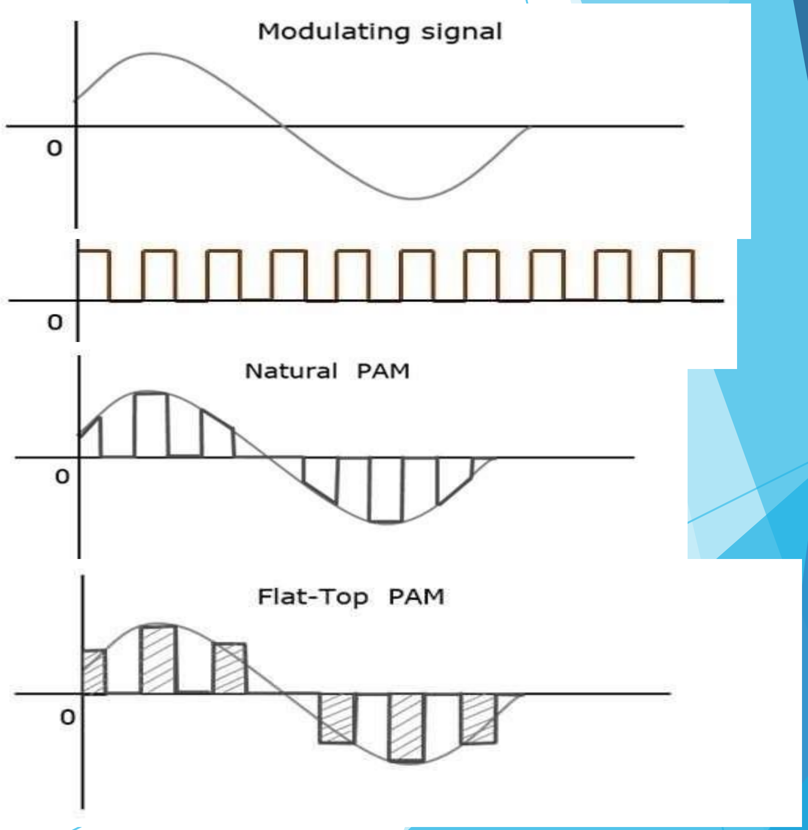
Pulse Amplitude Modulation (PAM)

- Amplitude of **the pulse carrier** varies proportional to the **instantaneous amplitude of the message signal.**
- The **width and positions** of the pulses are **constant** in this modulation.
- **PAM could be:**

(i) **Single polarity PAM**: A suitable **fixed DC bias is added to the signal** to ensure that all the pulses are positive.

(ii) **Double polarity PAM**: In this the pulses are both positive and negative.





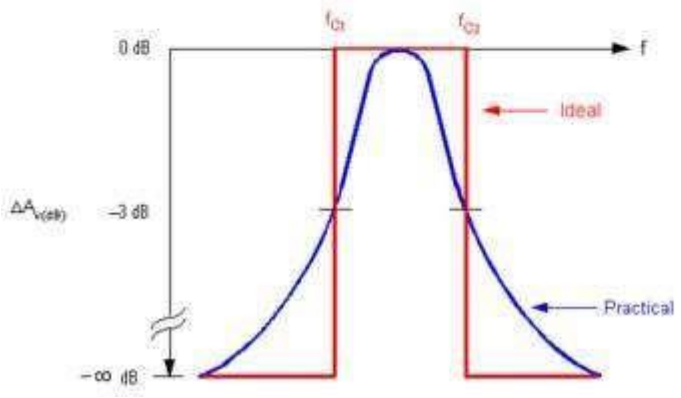
➢ Depending on type of sampling **PAM can be**:

(i) Ideal Sampling PAM, (ii) Natural sampling PAM and (iii) Flat top PAM.

➢ The **advantage** of this modulation is the **generation and detection** is **easy** in this modulation and also **allows multiplexing.**

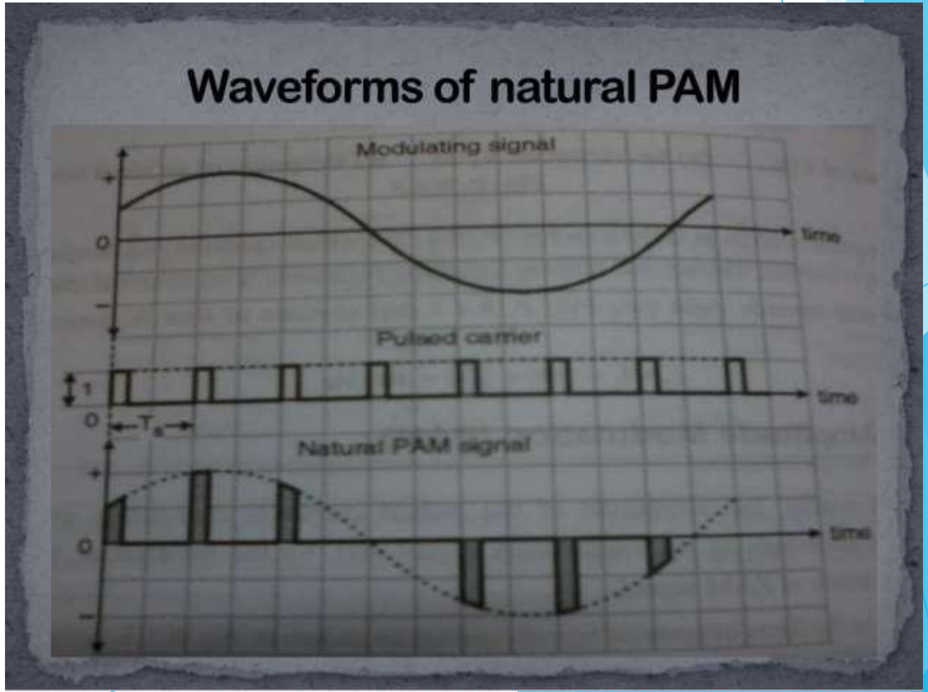
➢ The **disadvantage** is **large band width** of transmitted signal.

BPF characteristics



- For a **PAM signal** produced with **natural sampling**, the **sampled signal follows the waveform of the input signal** during the time that each sample is taken.
- A PAM signal is generated by using **a pulse train**, called the **sampling signal** (or clock signal) to operate an **electronic switch or "chopper"**. This produces **samples of the analog message signal**.
- The **switch is closed** for the duration of **each pulse**, allowing the **message signal** at that **sampling time to become part of the output.**
- The **switch is open for the remainder of each sampling period making the output zero**. This is known as **Natural PAM**.

In simplest form **PAM** can be visualized as **o/p of an AND gate** whose **two inputs** are **message signal x(t)** and **pulses at sampling rate**



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- For **flat-top sampling**, a sample-and-hold circuit is used in conjunction with the chopper to hold the amplitude of each pulse at a constant level during the sampling time,
- **Flat-top sampling**, produces **pulses whose amplitude remains fixed** during the sampling time. The **amplitude value** of the pulse depends on **the amplitude of the input signal at the time of sampling.**
- **Aperture Effect** seen in this type of **PAM. Equalizers** used at **receiver end**

<u>Transmission Bandwidth in PAM</u>

$\tau \ll T_s$

$f_s \geq 2fm\ ;\ Ts \leq \frac{1}{2fm}$

$\tau \ll Ts \leq \frac{1}{2fm}$

If on and off time of PAM pulse is same **then** $f_{max} = \frac{1}{2\tau}$

$BW \geq fmax;\ \ BW \geq \frac{1}{2\tau}$

$BW \geq \frac{1}{2\tau} \gg fm$

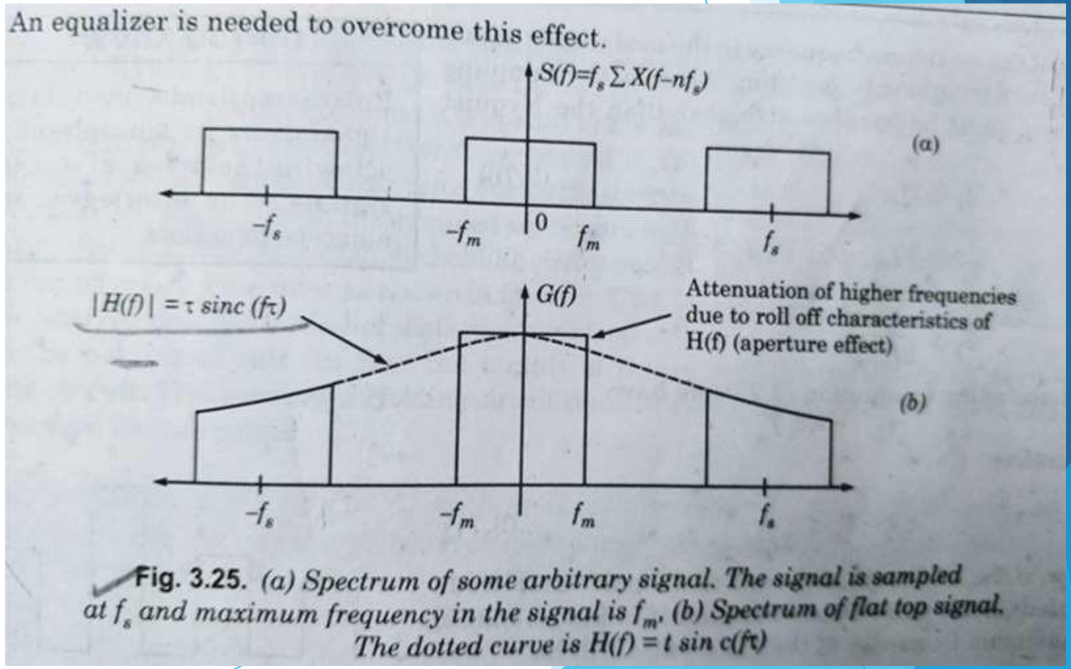


Fig. 3.25. (a) Spectrum of some arbitrary signal. The signal is sampled at f_s and maximum frequency in the signal is f_m. (b) Spectrum of flat top signal. The dotted curve is $H(f) = \tau\ sin\ c(f\tau)$

Transmission of PAM signals

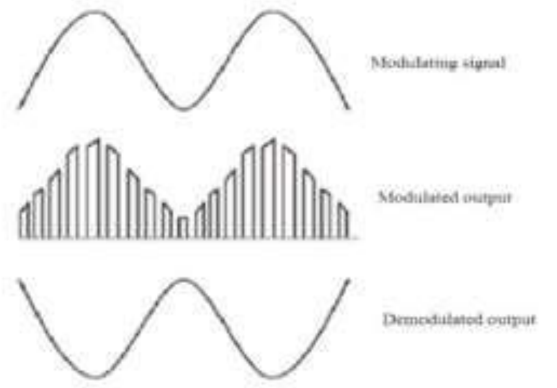
- For PAM signals to be transmitted through space using antennas, they must be **amplitude/ frequency/ phase** modulated by a **high frequency carrier** and only then they can be transmitted. Thus the **overall system is PAM-AM. PAM-FM or PAM-PM** and at receiving end, AM/ FM/PM detection is first employed to get the PAM signal and then message signal is recovered.

Drawbacks of PAM

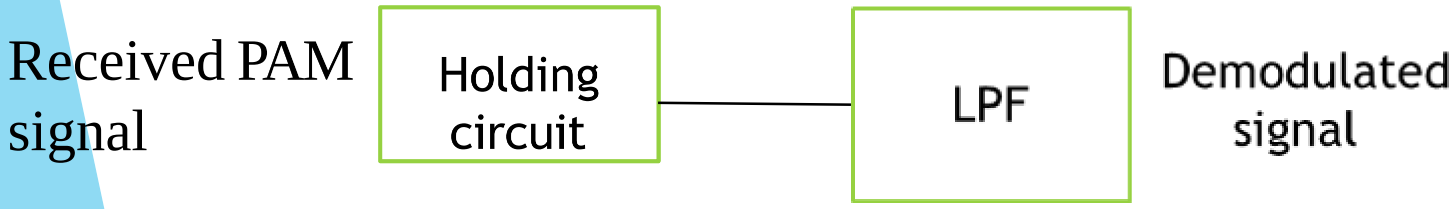
- Bandwidth required for transmission of PAM signal is very large in comparison to maximum frequency present in modulating signal.
- Since amplitude of PAM pulses varies in accordance with modulating signal so interference of noise is maximum in PAM
- Variation of the peak power required by transmitter

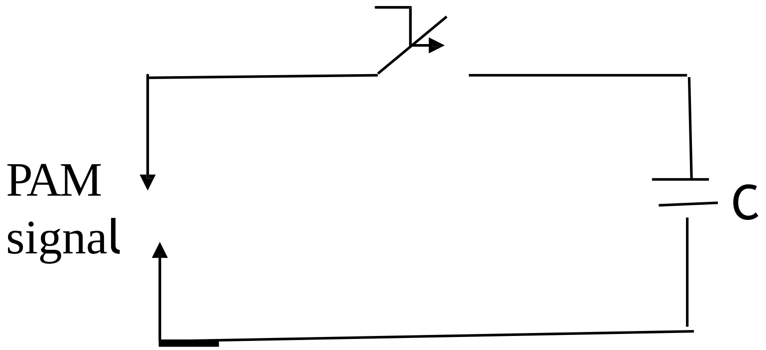
<u>Demodulation of PAM</u>

- PAM signal sampled at Nyquist rate can be **reconstructed at the receiver end** , by passing it through an efficient **Low Pass Filter (LPF)** with exact cut off frequency of **fs/2**. This is known as **<u>Reconstruction or Interpolation Filter</u>**.
- The low pass filter eliminates the high-frequency ripples and generates the demodulated signal. This signal is then applied to the inverting amplifier to amplify its signal level to have the demodulated output with almost equal amplitude with the modulating signal

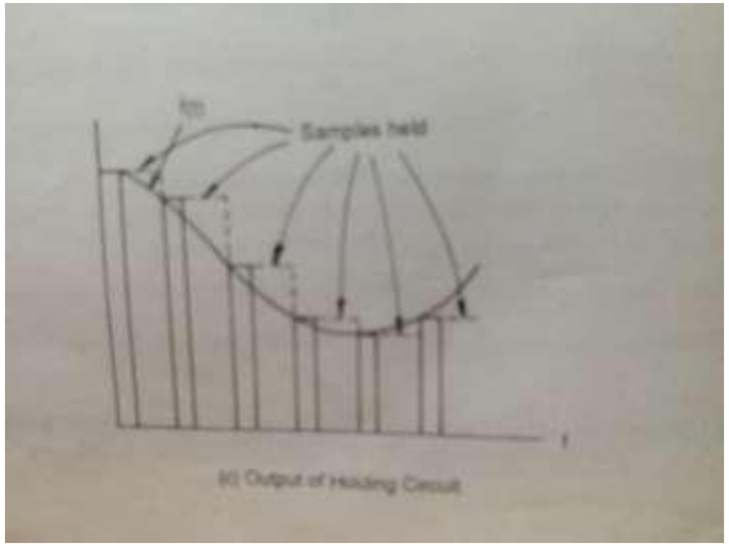


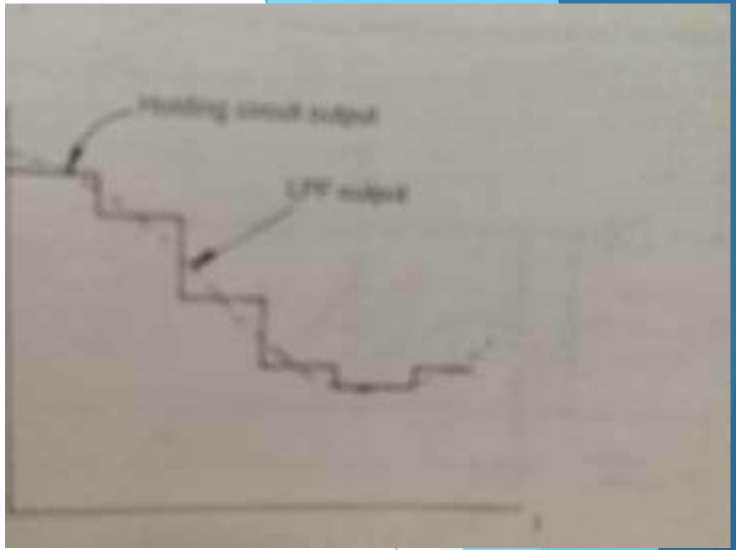
- For a **flat topped PAM**, a **holding circuit** followed by a **LPF** gives demodulated signal





Zero order Holding Circuit





- Switch S closes after the arrival of pulse and opens at the end of pulse.
- Capacitor C charges to pulse amplitude value and holds this value during interval between two pulses.
- The sampled values are shown in fig.
- Holding circuit o/p smoothened in LPF.
- Known as zero order holding circuit, which considers only the previous sample to decide value between two pulses
- First order holding circuit considers previous two samples, second order holding circuit considers previous three samples.


Thank You

Principles of Communication (BEC-28)

Unit-4

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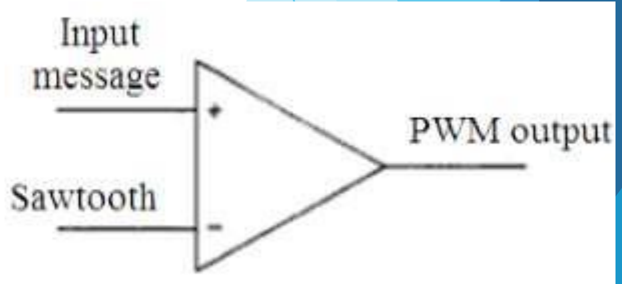
Pulse Time modulation

- In **PTM**, **amplitude of pulse is constant** while **position or width of pulse** is made proportional to the **amplitude of the signal** at the **sampling instant**.
- It can be PWM and PPM
- In both the cases amplitude constant and does not carry information so amplitude limiters can be used (like in FM) providing good noise immunity

Generation of PTM signals can be either by:

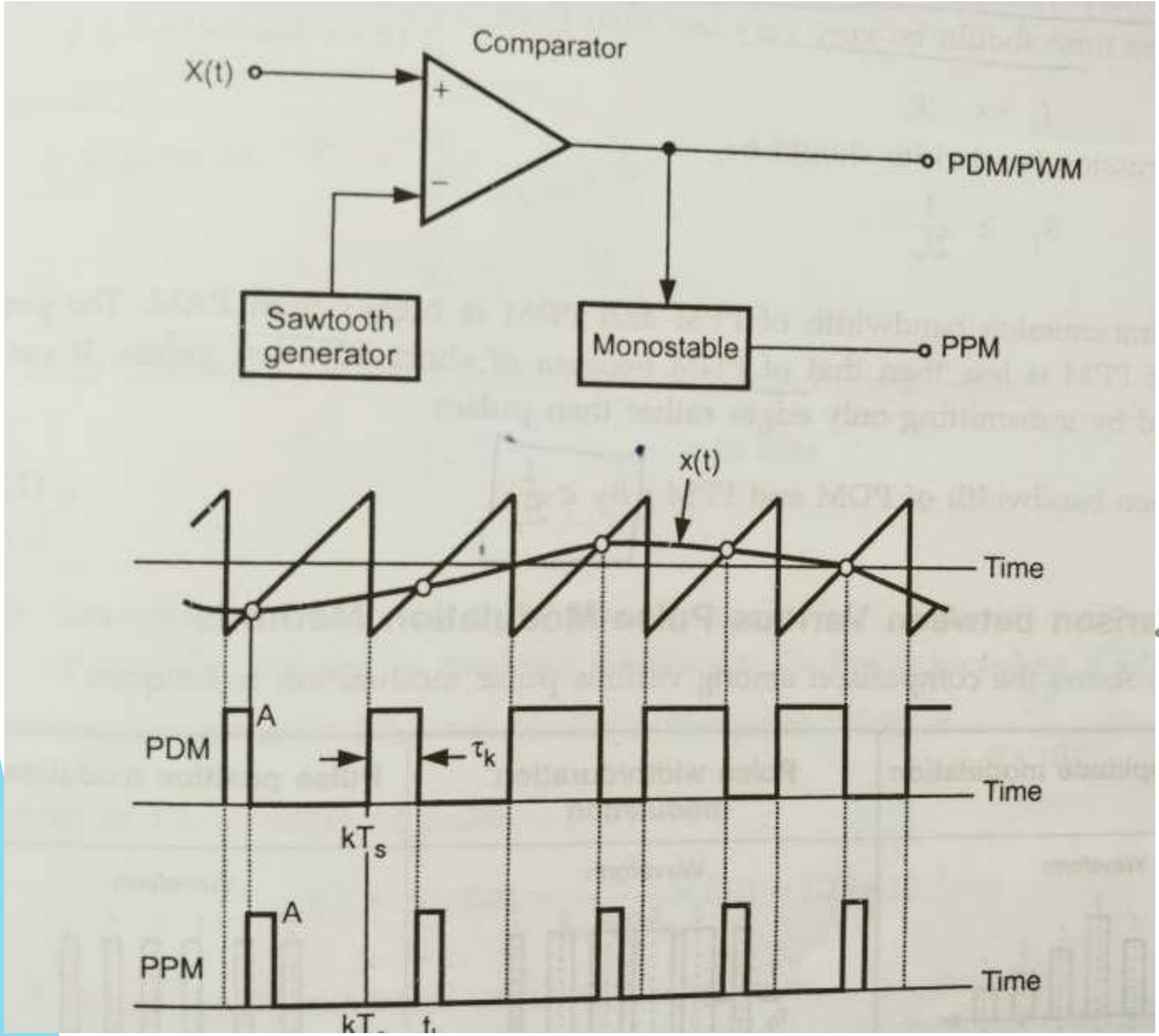
(i) **Indirect Method**: Firstly PAM signals are generated, Synchronized is generated during each pulse interval. These two signals are added and the sum is applied to a comparator whose reference level is suitably chosen. The second crossing of comparator level used for PPM

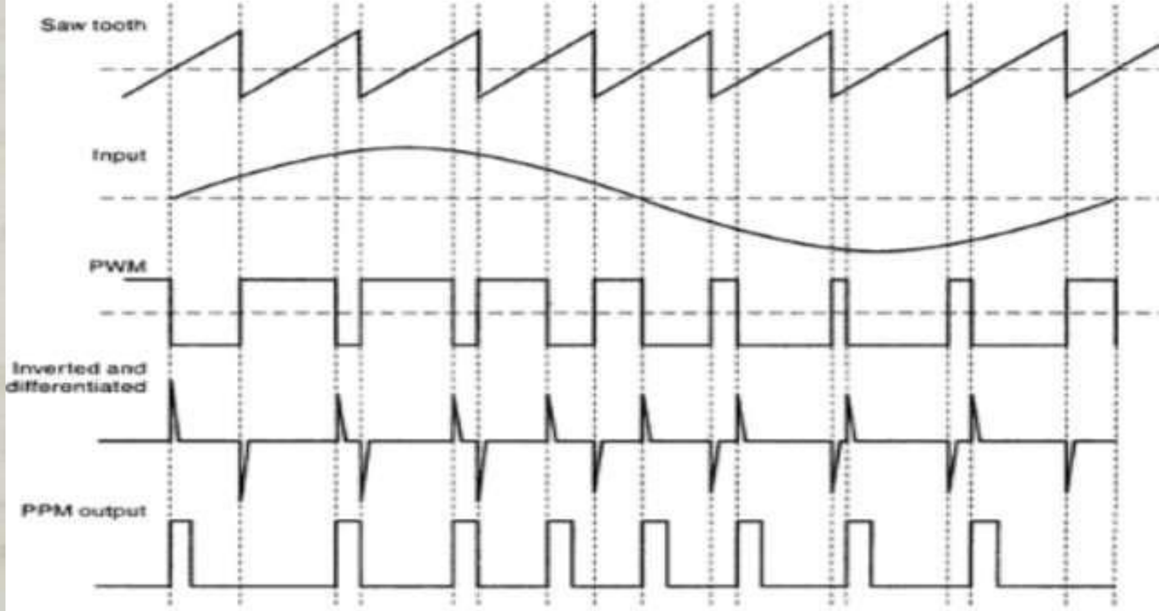
(ii) Direct method: PTM waveforms generated without using PAM waveforms



Pulse Width modulation

The pulse width modulation is the modulation of signals by varying the width of pulses. The amplitude and positions of the pulses are constant in this modulation



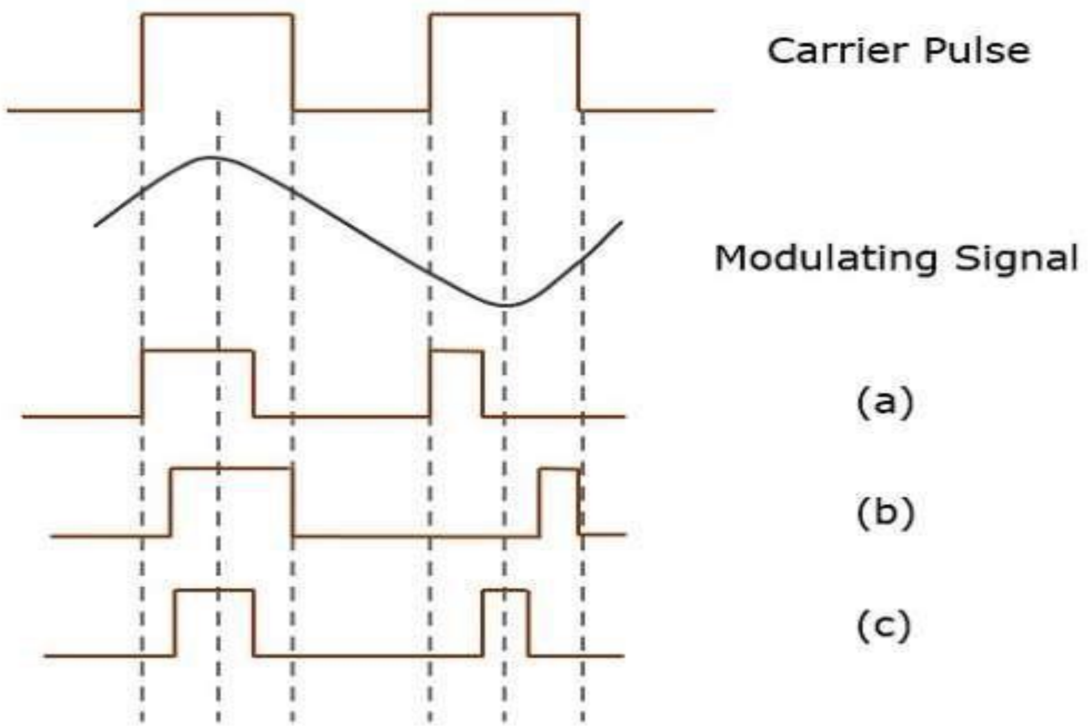


Generation of PWM and PPM by Direct Method

- **The non inverting input** of the **comparato**r is fed by the **input message or modulating signal x(t)** and the other input by a **saw-tooth signal** which operates at carrier frequency.
- The comparator compares the two signals together to generate the PWM signal at its output. Its o/p is high only when the instantaneous value of x(t) is higher than sawtooth waveform.
- The rising edges of the PWM signal occurs at the fixed time period (kT_s) while trailing edge depends on amplitude of message signal x(t).
- When saw-tooth voltage waveform greater than x(t), o/p of comparator is zero, **trailing edge is modulated**
- If **saw-tooth. waveform is reversed**, trailing edge is fixed while **leading edge is modulated**.
- Replacing saw-tooth waveform by **triangular**, both leading and trailing edge modulated. (**symmetrical PWM**)
- The amplitude of PDM/PWM will be positive saturation of the comparator shown as ‘A’, being same for all pulses,

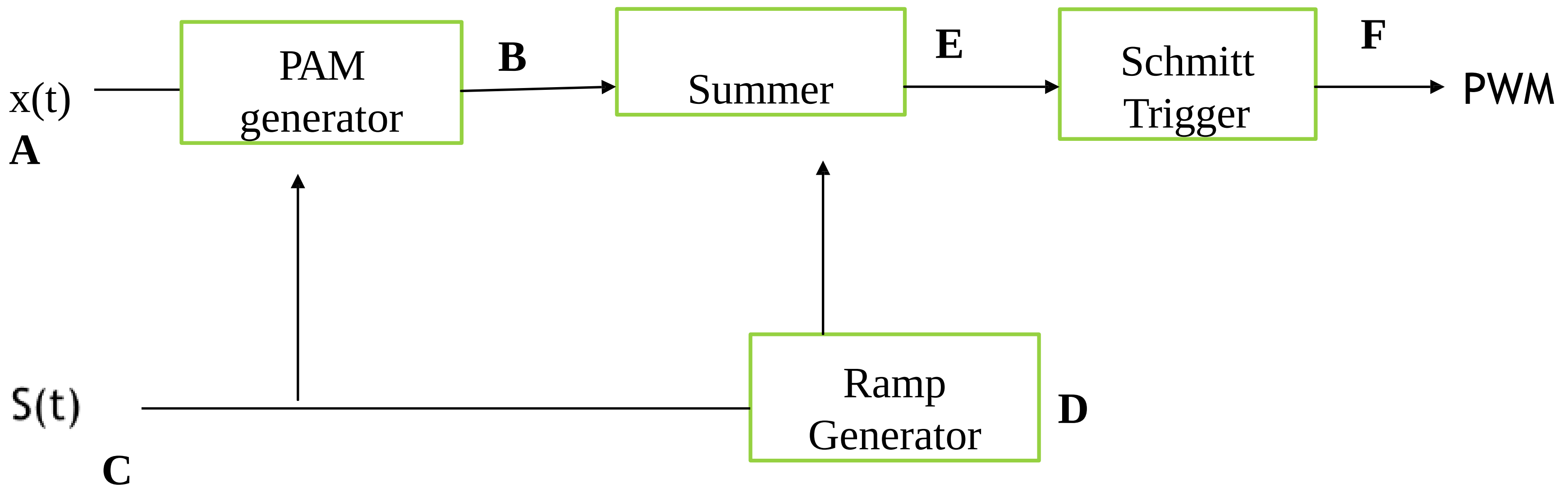
Three types of pulse-width modulation (PWM) are possible:

- The leading edge of the pulse being constant, the trailing edge varies according to the message signal.
- The trailing edge of the pulse being constant, the leading edge varies according to the message signal
- The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal (Symmetrical PWM)

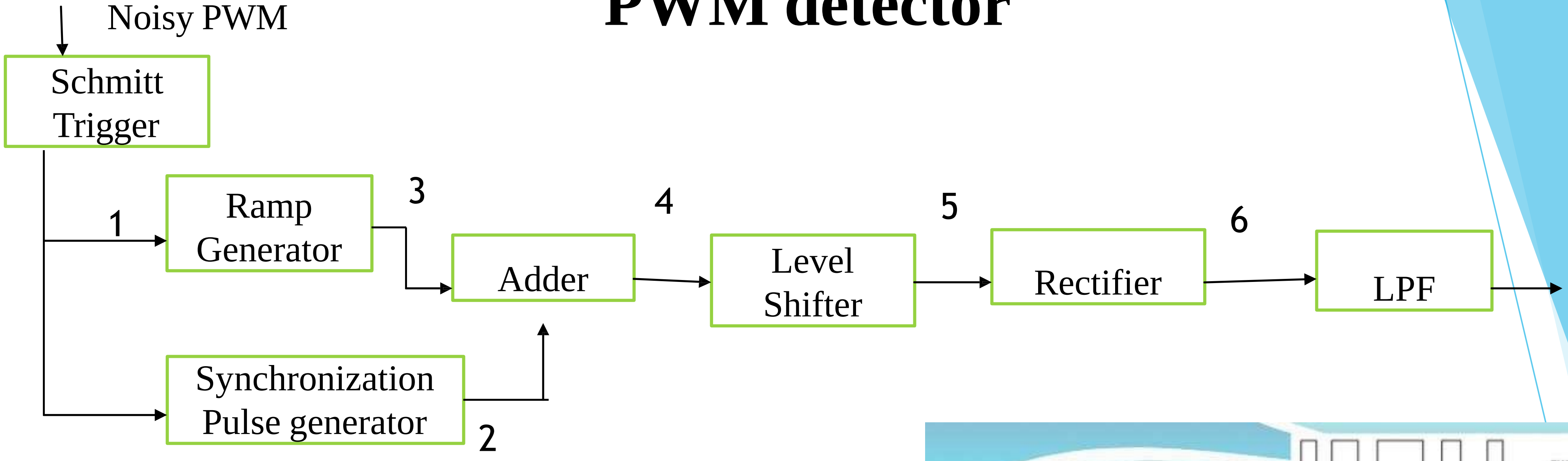


Indirect Method:

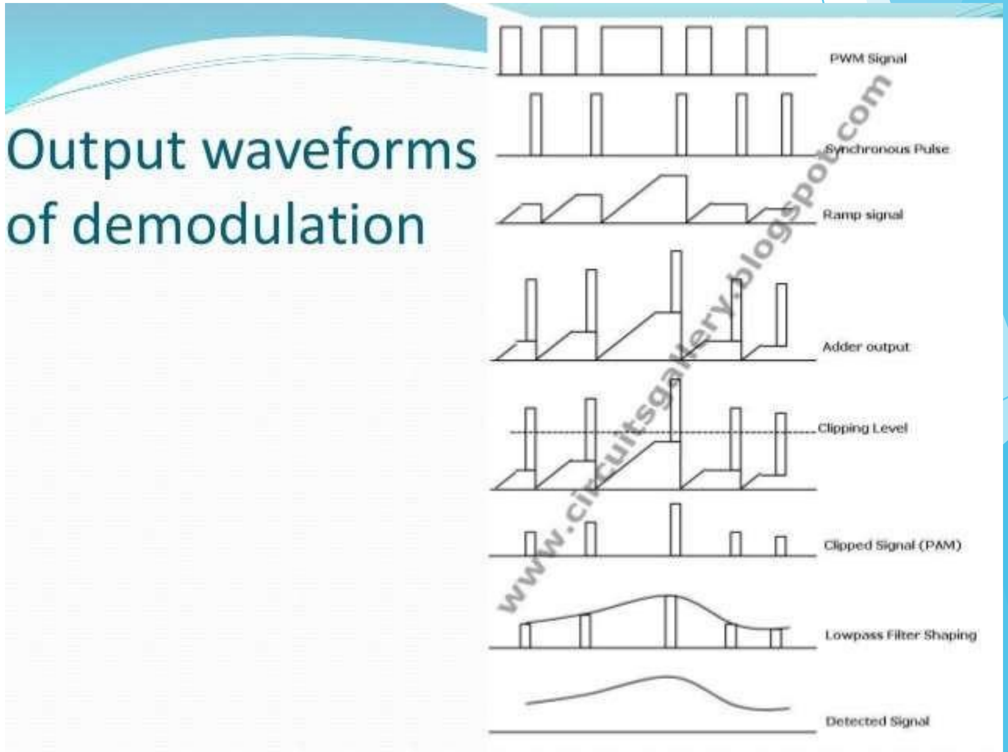
Modulating signal (A) applied to i/p of PAM circuit [s(t) pulse train] and PAM signal generated(B). S(t) also is i/p to Ramp generator(Integrator circuit), all having equal slopes, amplitude and generation(D). These ramp pulses added to PAM pulses to produce varying height samples. These varying height ramp gates a S.T ckt to generate varying width rectangular pulses of PWM.



PWM detector



- Received PWM signal applied to ST circuit to remove noise
- Regenerated PWM applied to Ramp generator and synchronization pulse.
- Heights of Ramp proportional to width of pulses.
- Pulse generator produces reference pulses with constant amplitude and width but delayed by specific amount.
- Delayed reference pulses added to o/p of ramp generator
- The o/p given to level shifter, negative offset shifts waveform. Then clipped by rectifier followed by LPF to give message signal.

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Principles of Communication (BEC-28)

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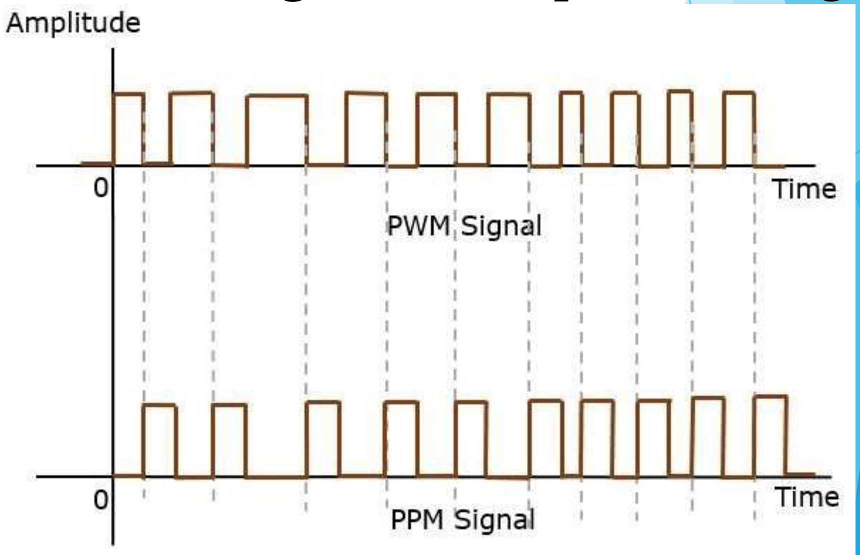
Pulse position modulation

- **(PPM)** is an analog modulating scheme in which 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 varies according to the instantaneous sampled value of the message signal.
- The transmitter has to send **synchronizing pulses** (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses.
- PPM is done in accordance with the PWM signal.
- PWM signal is used as the trigger input to a monostable multivibrator.
- Its o/p remains zero until it is triggered on the trailing edge of PWM
- O/P of monostable MV switches to positive saturation value A **and remains high for fixed period** then goes low
- Hence, the position of these pulses is proportional

  to the width of the PWM pulses.



Advantage As the amplitude and width are constant

the power handled is constant

Disadvantage: Synchronization between Transmitter and receiver is a necessity

- The **PDM is differentiated**, and then **rectified** and **shaped**.
- PPM carries exactly the same information as long as the position of the clock pulses (leading edge) is well defined in the received signal.
- PPM is superior to PDM for message transmission, since the wide pulses of PDM require more energy than PPM when transmitted
- PPM is suited for communication in the presence of noise.
- Very high peak narrow pulses can be transmitted and the pulse position can be determined even when the noise level is high,
- However, transmitting very narrow pulses requires a large band width
- When light is used as the media for transmitting analog signals, **PPM or PCM** are the most suitable types of modulation because **the maximum power output in the modulated light source, such as LED or LASER is achieved when it is pulsed at a very low duty cycle**.
- In PPM, necessary to transmit a series of sync pulses at a much lower repetition rate than the sampling pulses, to avoid interference with original signal and/or minimise the number of pulses transmitted in order to conserve transmission power

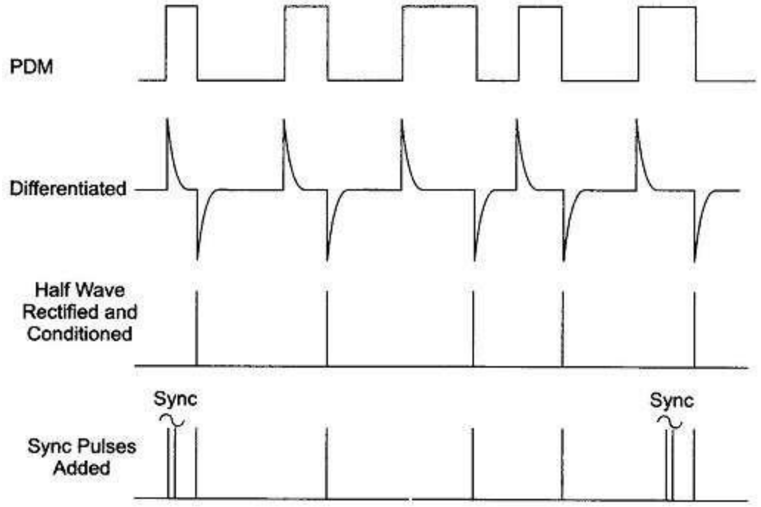


Fig. 18.14 Generation of PPM from PDM (Note: Sync Pulses are Double for Recognition)

Transmission BW of PWM and PPM

- Both PWM and PPM have DC value.
- Both need a shrp rise time and fall time to preserve the message information
- Rise time be very less than Ts i.e. $\mathbf{t_r \ll Ts}$
- **Transmission BW:** $\mathbf{B_T \geq \frac{1}{2tr}}$
- BW higher than PAM

PAM

- The amplitude of the pulse is proportional to the amplitude of modulating the signal.
- **Band width** of transmitting channel depends on the **width of the pulse**
- Instantaneous power of transmitter varies. Noise interference is high
- Complex system. Similar to A.M.

PWM

- Width of pulse is proportional to amplitude of modulating signal.
- The **Bandwidth of transmitting channel** depends on **rise time of the pulse**.
- Instantaneous power of transmitter varies. Noise interference is minimum.
- Simple to implement Similar to F.M.

PPM

- Relative position of pulse is proportional to amplitude of modulating signal.
- The bandwidth of transmitting channel depends on the rise time of the pulse.
- Instantaneous power remains constant. Noise interference is minimum.
- Simple to implement. Similar to P.M.

- Difference Between PAM, PWM, and PPM

Parameter	**PAM**	**PWM**	**PPM**
Type of Carrier:	Train of Pulses	Train of Pulses	Train of Pulses
Variable Characteristic :	Amplitude	Width	Position
Bandwidth Requirement:	Low	High	High
Noise Immunity :	Low	High	High
Information Contained in:	Amplitude Variations	Width Variations	Position Variations
Power efficiency (SNR)	Low	Moderate	High
Transmitted Power	Varies	Varies	Remains Constant
Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
Bandwidth	depends on width of the pulse	rise time of the pulse	rise time of the pulse
Transmitter power	Inst. power varies with amplitude of pulses	Instantaneous power varies with width of the pulses	Constant
Complexity of generation and detection	Complex	Easy	Complex
Similarity with other Modulation Systems	Similar to AM	Similar to FM	Similar to PM

Question 1: ***For a PAM transmission of voice signal with f_m=3kHz, calculate the transmission BW. Given that f_s=8kHz and the pulse duration τ=0.1T_s***

Soln: $Ts = \frac{1}{fs} = 125\mu s$

$\tau = 0.1Ts = 0.1 \times 125 = 12.5\mu s$

$$\mathbf{BW \geq \frac{1}{2\tau} \geq 40\ kHz}$$

Question 2: ***For the above signal if rise time is 1% of pulse width, find minimum Tx BW for PWM and PPM?*** **Soln:** $t_r = \tau \times 0{,}01 = 1.25\text{x } 10^{-7}$

$$B_T \geq \frac{1}{2\;_{r}} \geq \mathbf{4MHz}$$

Thus BW of PWM/PPM much higher than PAM


Thank You

Principles of Communication (BEC-28)

Unit-4

Pulse Modulation and Digital Transmission of Analog Signal

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Content of Unit-IV

Pulse Modulation and Digital Transmission of Analog Signal: Sampling Theorem and its applications, Concept of Pulse Amplitude Modulation, Pulse width modulation and pulse position modulation, PCM, Pulse Time Modulation, TDM and FDM. Line Coding, Quantizer, Quantization Noise, Compounding multiplexer.

Pulse Code Modulation(PCM)

- The pulse code modulator technique samples the input signal x(t) at a sampling frequency.

- This sampled variable amplitude pulse is then digitalized by the analog to digital converter. Figure.(1) shows the PCM generator.

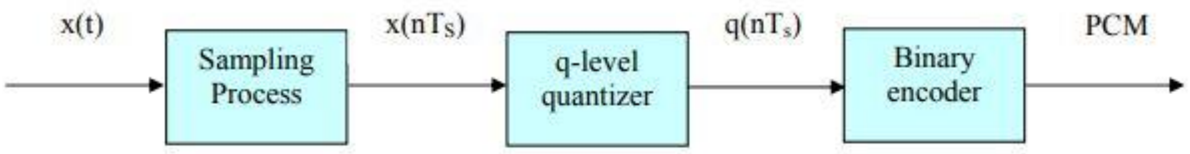


Figure.(1): PCM modulator

In the PCM generator, the signal is first passed through sampler which is sampled at a rate of (fs) where:

$$fs \geq 2fm$$

- The output of the sampler x(nTs) which is discrete in time is fed to a q -level quantizer. The quantizer compares the input x(nTs) with it's fixed levels. It assigns any one of the digital level to x(nTs) that results in minimum distortion or error.

Pulse Code Modulation(PCM)

- The error is called quantization error, thus the output of the quantizer is a digital level called q(nTs).
- The quantized signal level q(nTs) is binary encode. The encoder converts the input signal to v digits binary wor

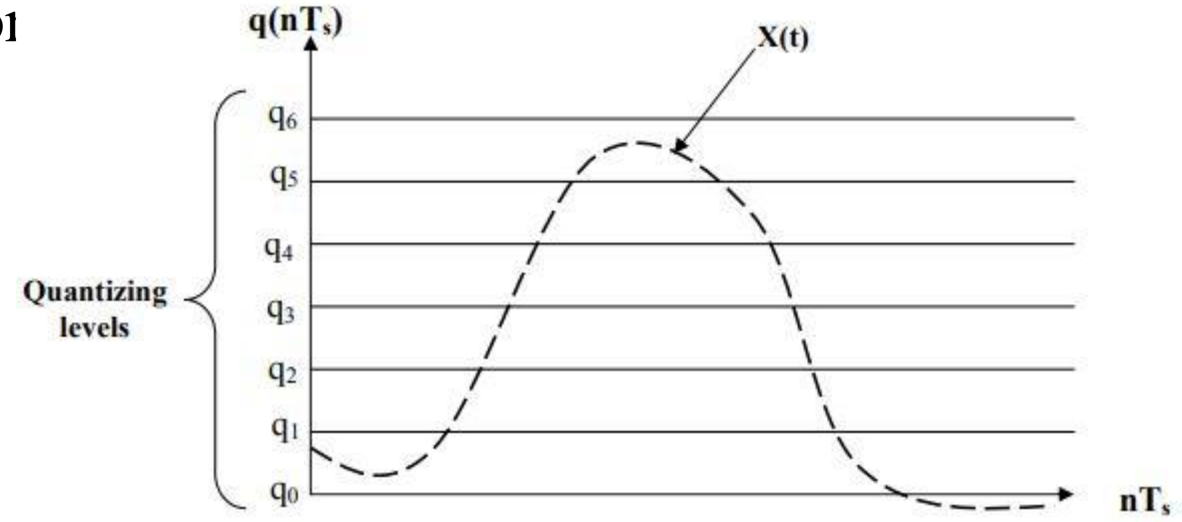


Figure.(2) A sampled signal and the quantized levels

- The receiver starts by reshaping the received pulses, removes the noise and then converts the binary bits to analog. The received samples are then filtered by a low pass filter; the cut off frequency is at fc.

i.e. , fc= fm

Pulse Code Modulation(PCM)

- It is impossible to reconstruct the original signal x(t) because of the permanent quantization error introduced during quantization at the transmitter.

- The quantization error can be reduced by the increasing quantization levels. This corresponds to the increase of bits per sample(more information).

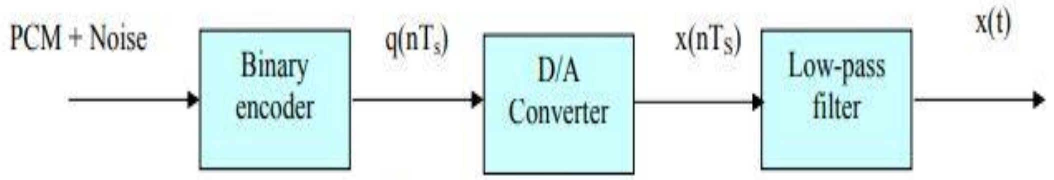


Figure.(3): PCM demodulator

- The choice of the parameter for the number of quantization levels must be acceptable with the quantization noise (quantization error).

Pulse Code Modulation(PCM)

- Figure.(4) shows the reconstructed signal.

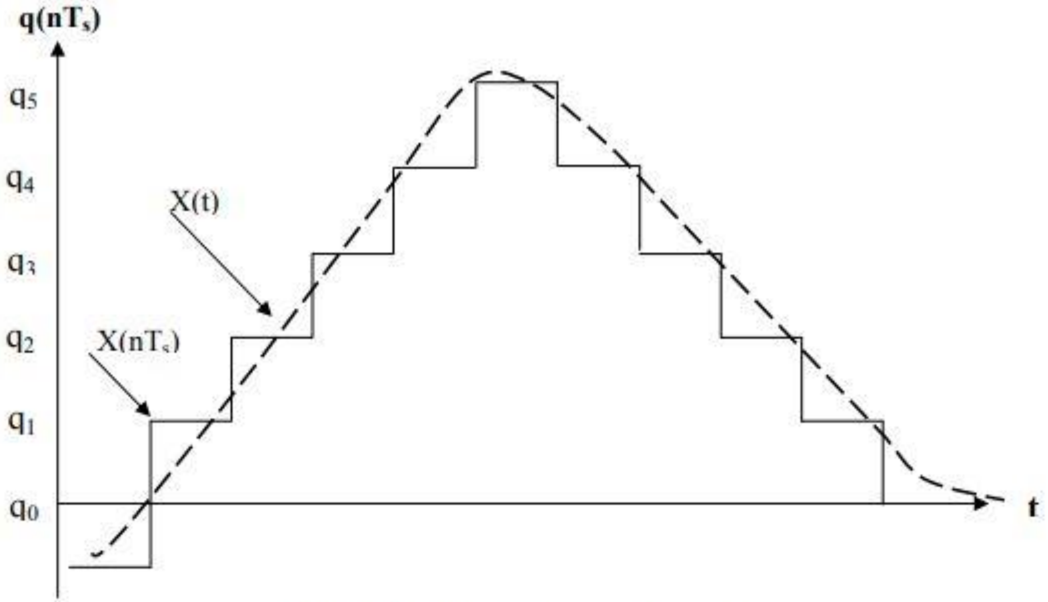


Figure.(4):The reconstructed signal

Signaling rate in PCM

Let the quantizer use 'v' number of binary digits to represent each level. Then the number of levels that can be represented by v digits will be :

$$Q=2^v$$

Pulse Code Modulation(PCM)

- The number of bits per second is also called signaling rate of PCM and is denoted by **'r'**:

Signaling rate (r)= vf_s

Example

➢ If the number of binary bits = 3 and the sampling rate is 2 sample/sec find the signaling rate, number of quantization levels?

Solution:

$$f_s=2, \quad v=3$$

$$\begin{aligned} \text{signaling rate(r)} &= v\, f_s \\ &=3*2 \\ &=6 \text{ bits/sec} \end{aligned}$$

$$\begin{aligned} \text{Number of quantization(q)} &=2^v \\ &=2^3 \\ &=8 \text{ levels} \end{aligned}$$

Multiplexing

- Multiplexing refers to the **combination of information streams from multiple sources** for **transmission over a shared medium .**
- Multiplexor is a mechanism that implements the concept. It permits hundreds or even thousands of signals to be combined and transmitted over a single medium. De-multiplexing refers to the separation of a combination, back into separate information streams .

Principle used

- Each sender communicates with a single receiver
- All pairs share a single transmission medium
- Multiplexor combines information from the senders

  for transmission in such a way that the de multiplexer

  can separate the information for receivers.
- Cost savings obtained using single channel to send

  Multiple signals.

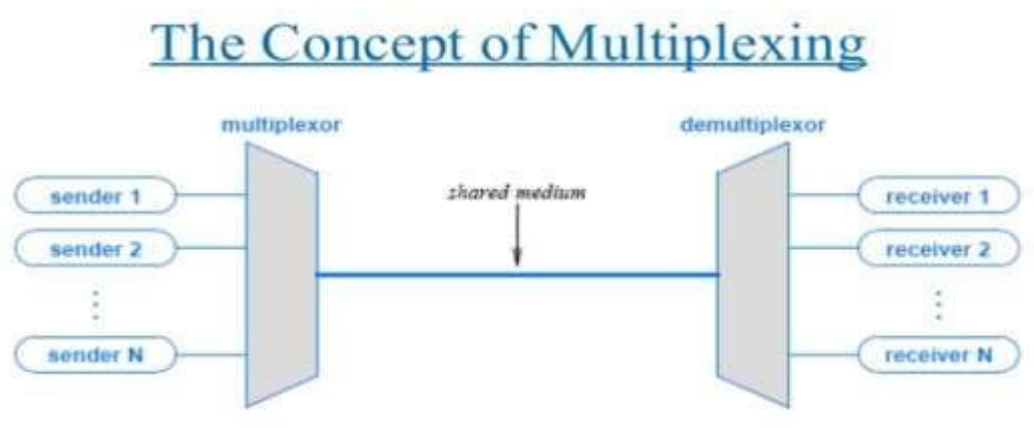
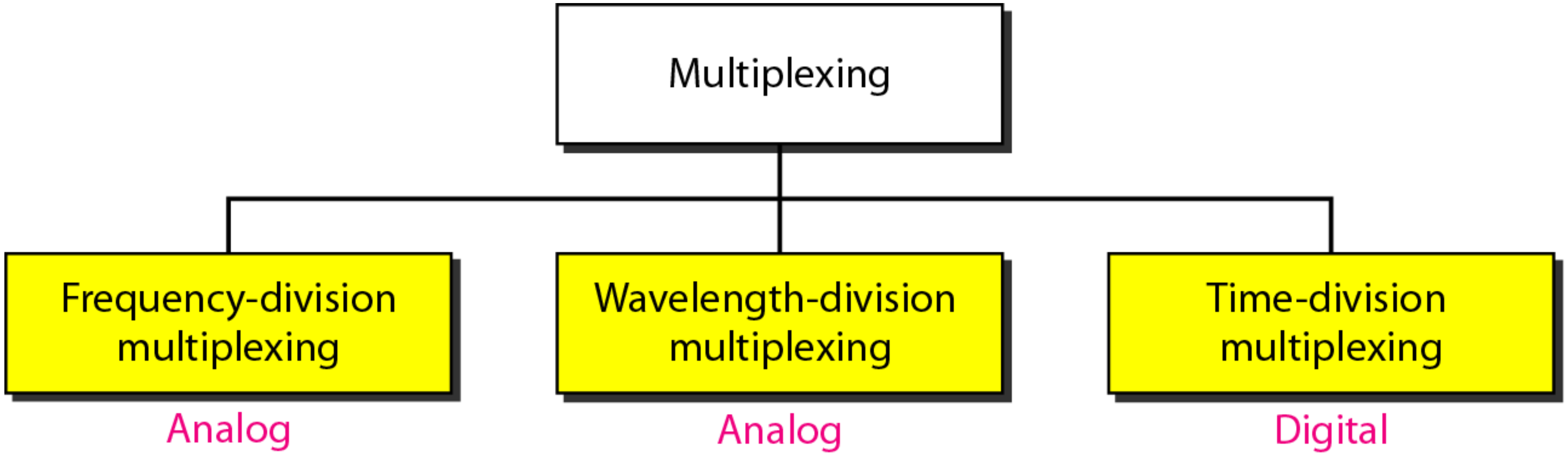


Figure 11.1 The concept of multiplexing in which independent pairs of senders and receivers share a transmission medium.

Four basic types of multiplexing

- Frequency Division Multiplexing (FDM)
- Wavelength Division Multiplexing (WDM)
- Time Division Multiplexing (TDM) •
 Code Division Multiplexing (CDM)

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Principles of Communication (BEC-28)

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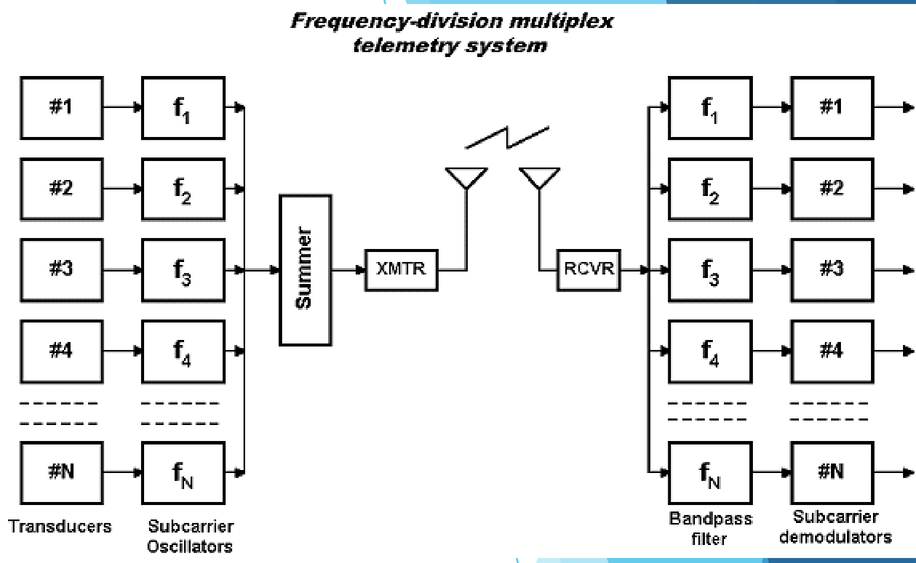
Frequency Division Multiplex (FDM): **Separation of spectrum into smaller frequency**.

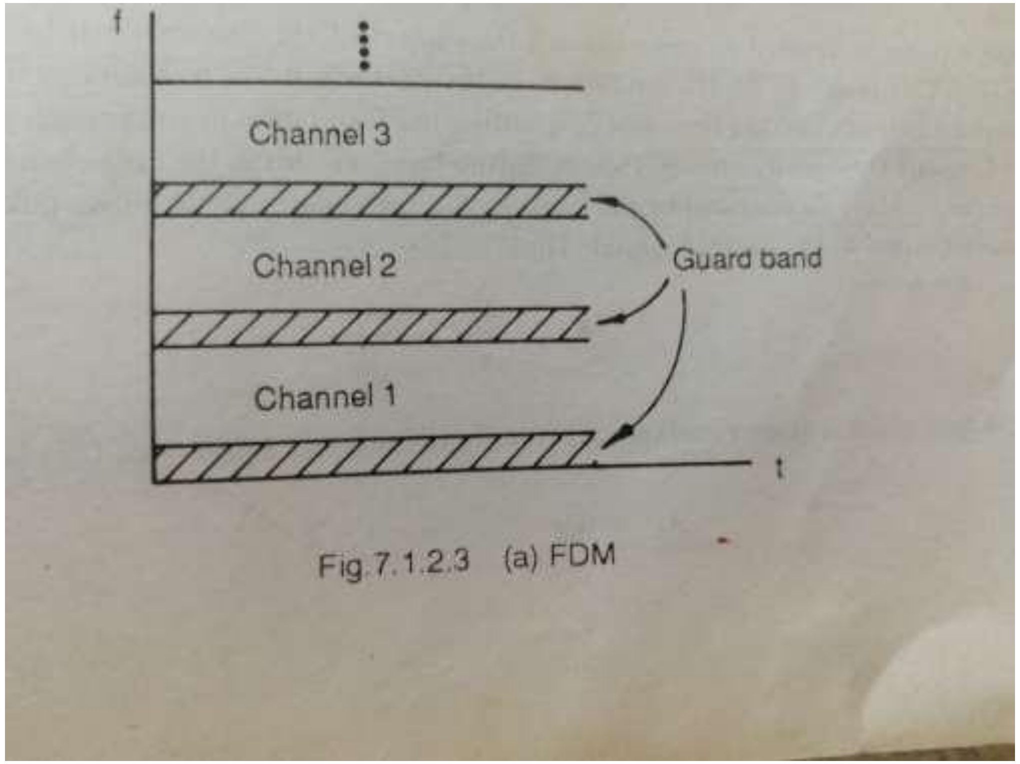
- Channel gets band of the spectrum for the whole time. Each signal allocated different frequency band i.e, Multiple carriers used
- Each message signal is limited to f_m Hz.

Example: Multiplexing of telephonic signals from n subscribers

- Telephonic message (BW=3kHz) and broadcast signal limited to 5kHz. Without multiplexing if n channels transmitted, Interference and no useful information.



- In FDM, each baseband signal translated by Analog Modulation (AM/Angle) to different carrier frequencies.
- Each carrier separated from neighbouring by at least 2fm
- Multiplexed signals can be transmitted over a common channel without interference.
- At receiver, various carrier frequencies selected using BPF tuned to appropriate carrier frequencies and demodulated by separate detector.



FDM System

Advantages: No dynamic coordination needed and works also for analog signals

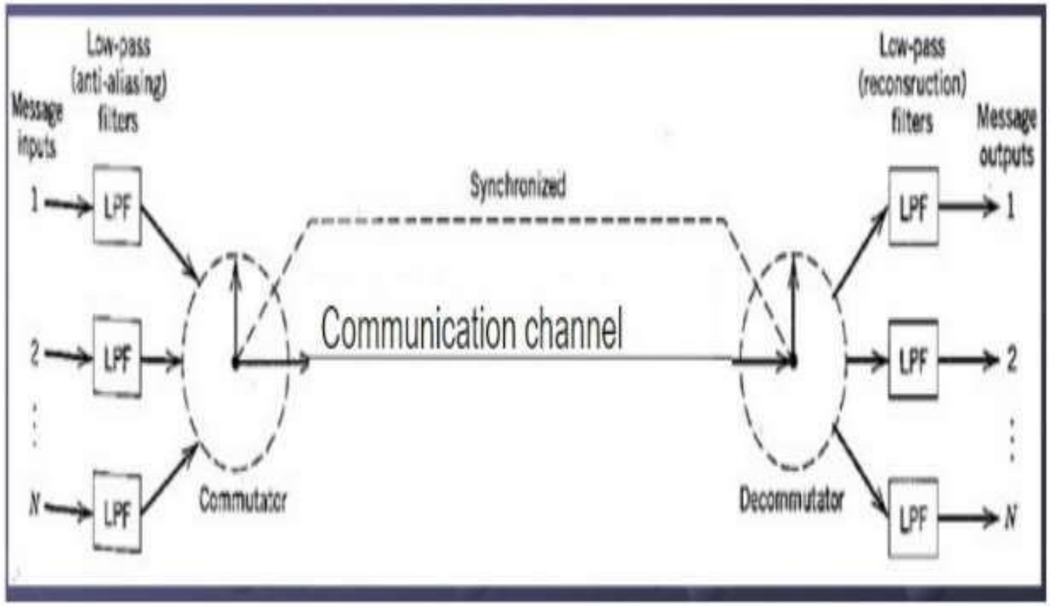
Disadvantages: Waste of bandwidth if traffic distributed uneven; inflexible;

Time Division Multiplexing

- Time Division Multiplexing (TDM) is the time interleaving of samples from several sources so that the information from these sources can be transmitted serially over a single communication channel.
- **At the Transmitter** :Simultaneous transmission of several signals on a time-sharing basis.

  Each signal occupies its own distinct time slot, using all frequencies, for the duration of the transmission. Slots may be permanently assigned on demand
- **At the Receiver** : Decommutator (sampler) has to be synchronized with the incoming waveform
- In Pulse modulation techniques, there is a **free space between any two consecutive pulses** of a signal. This **free space between pulses** can be occupied by **pulses from other channel.** This is **Time Division Multiplexing (TDM)** and makes **maximum utilization of transmission channel.**
- **Applications of TDM:** Digital Telephony, Data communications, Satellite Access, Cellular radio

Block diagram of TDM and PAM-TDM



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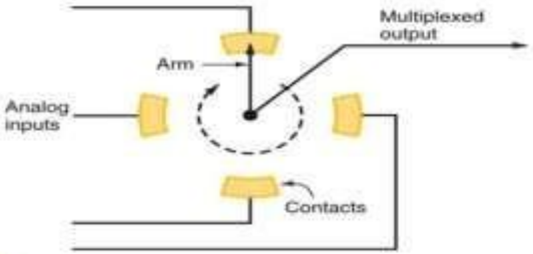
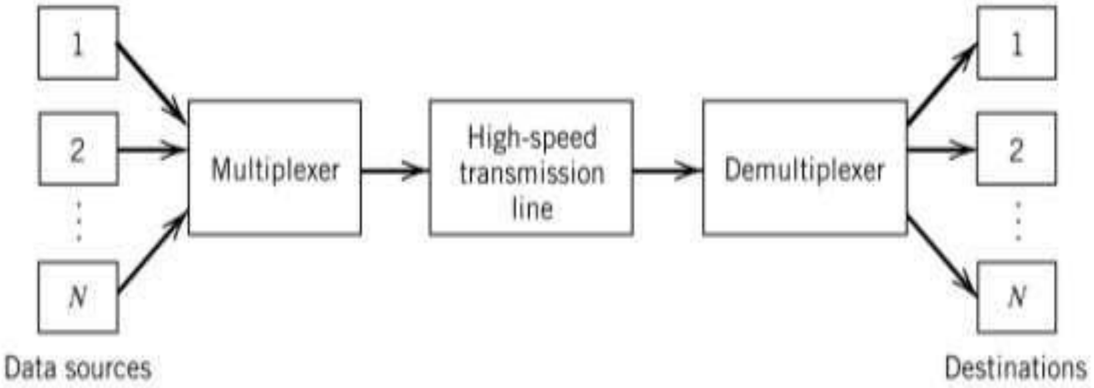
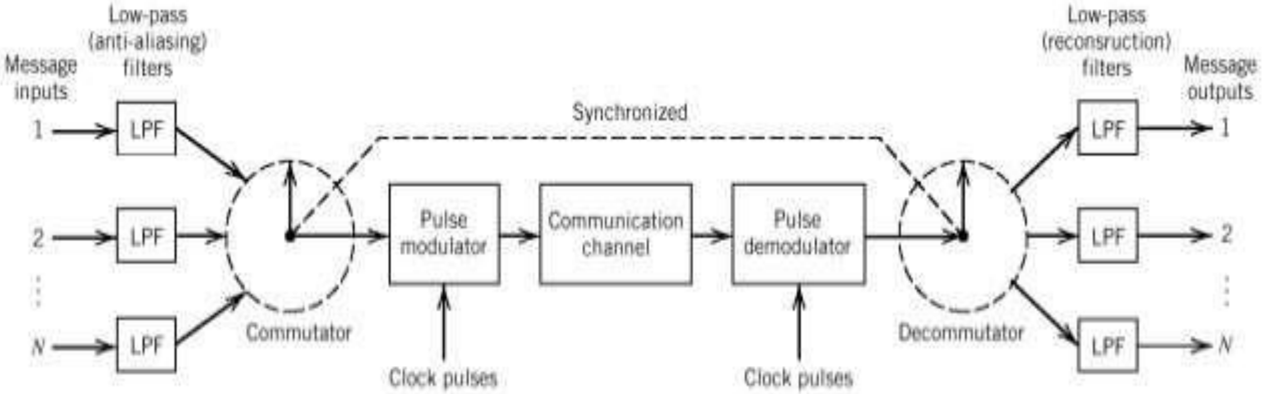
Simple rotary-switch multiplexer



Time Division Multiplexing



Conceptual diagram of multiplexing-demultiplexing.



PAM TDM System

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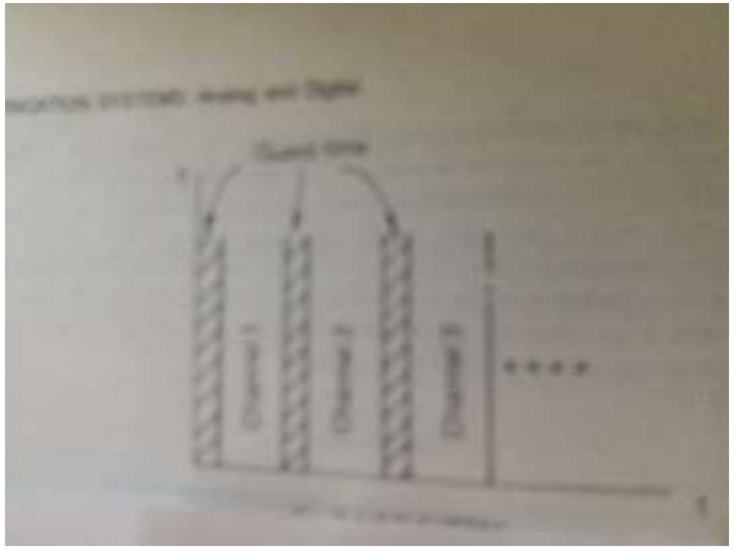
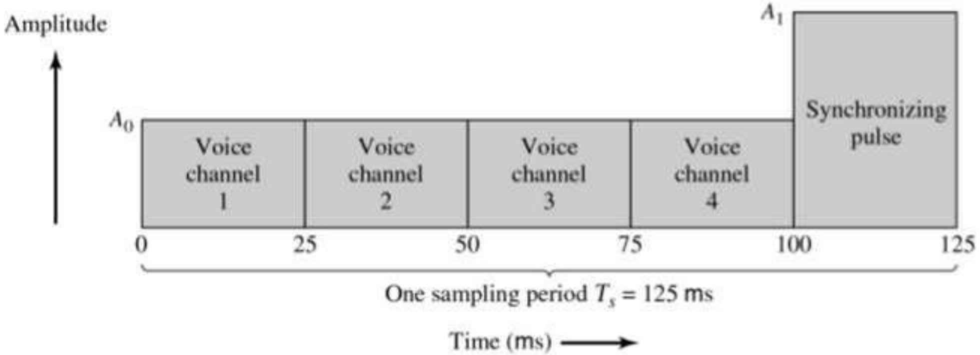


TDM

- Composition of one frame of a multiplexed PAM signal incorporating four voice-signals and a synchronizing pulse.



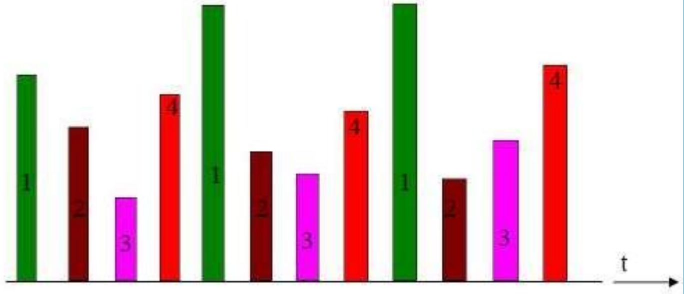
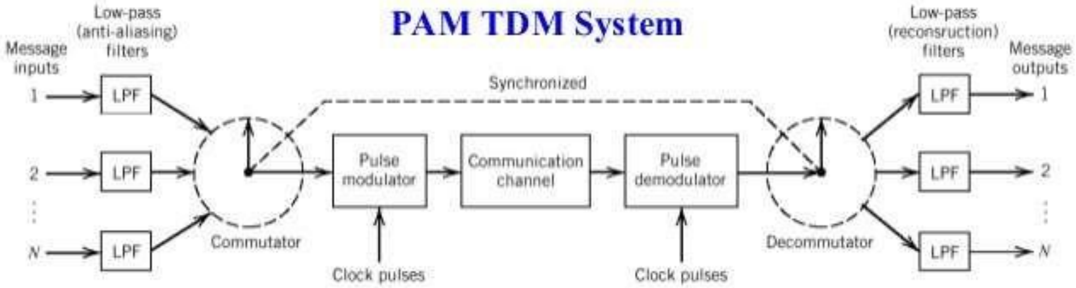
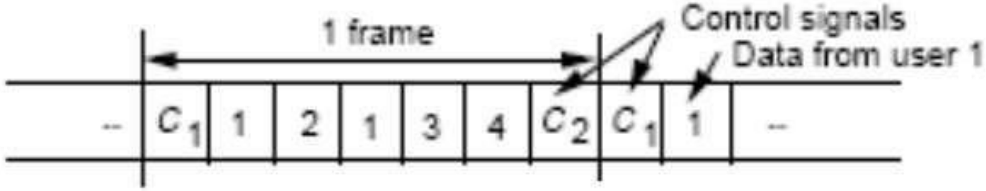


Fig: 2.23 Multiplexing of FOUR signals.

Block diagram of TDM system

PAM TDM System



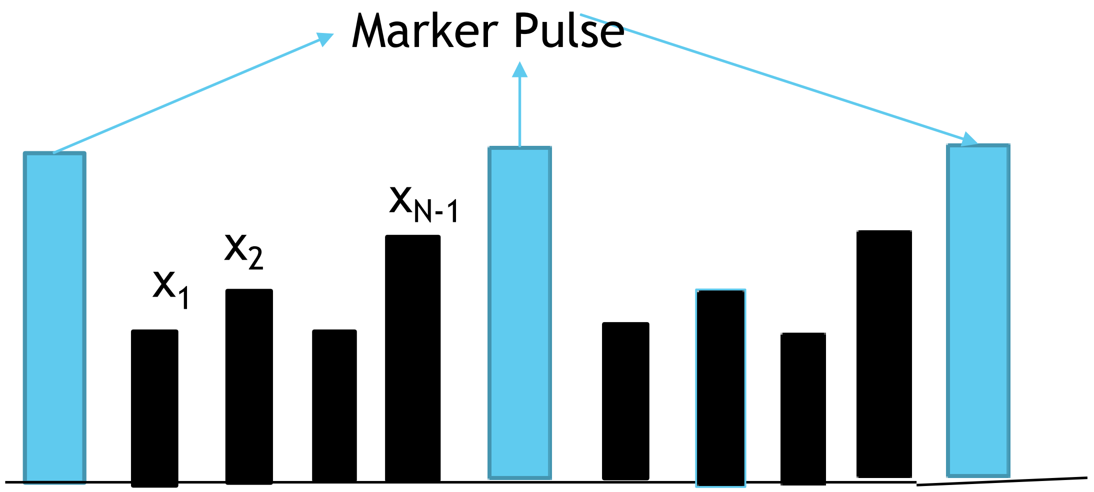


A Typical Framing Structure for TDM

- The system shows **TDM of 'N' PAM channels**.
- **Each channel** to be transmitted is passed through **LPF** to band limit its frequency to **f_m Hz (W Hz)**
- Outputs of LPF are connected to **the rotating sampling switch or commutator**.
- It takes **sample from each channel per revolution** and rotates at **the rate of fs.**
- Function of commutator is two fold**: (i) Taking narrow samples of each of N input messages at rate $1/T_s$ (ii) To sequentially interleave N samples Inside a sampling interval T_s**
- The **single signal composed due to multiplexing** of input channels given to **Transmission channel**
- If **W or f_m** is highest signal frequency in the message signal: **$fs \geq 2f_m$ or 2W**; **Ts or $1/fs \leq 1/2fm$**
- Thus time interval Ts contains one sample from each input**. It is called frame. If N input channels multiplexed**, each frame will have one sample from each of N channel's input
- Spacing between two samples: **Ts**/**N**
- No. of pulses/sec=1/ spacing between two pulses=$1/(Ts/N) = N/Ts$
- Ts=1/fs; No. of pulses per second=Nfs
- The no. of pulses transmitted per second is called **signalling rate of TDM 'r'**
- **r=Nfs**; fs ≥2fm**;** **$r \geq 2Nf_m$ or $r \geq 2NW$**
- Pulsed TDM passed through LPF to convert it to baseband signal whose BW given **by half signalling rate**
- **B.W=$r/2$ = Nf_m=NW; Minimum Transmission Bandwidth**
- **At the receiver, decommutator separates the time multiplexed input channels which then passed through reconstruction filter**

Synchronization in TDM system

- The time division multiplexing (TDM) needs synchronization between multiplexer and demultiplexer. If synchronization is not there between multiplexer and demultiplexer, a bit going to one channel may be received by the wrong channel.
- Because of this reason, one or more synchronization bits are usually added to the beginning of each frame called **Markers** (**highest amplitude**)
- These bits are called framing bits (Marker pulse), allows the demultiplexer to synchronize with the incoming stream so that that it can separate time slot accurately.
- Because of the marker pulse, no of channels to be multiplexed reduced by 1



Application of PWM

- Although PWM is also used in communications, its main purpose is **actually to control the power that is supplied to various types of electrical devices,** most especially to inertial loads such as AC/DC motors.
- Pulse-width modulation (PWM) is used for controlling the amplitude of digital signals in order to control devices and **applications requiring power or electricity**. It essentially controls the amount of power, in the perspective of the voltage component, that is given to a device by cycling the **on-and-off phases of a digital signal quickly and varying the width of the "on" phase or duty cycle**. To the device, this would appear as a steady power input with an average voltage value, which is the result of the percentage of the on time. The duty cycle is expressed as the percentage of being fully (100%) on.
- A very powerful benefit of **PWM is that power loss is very minimal**. Compared to regulating power levels using an analog potentiometer to limit the power output by essentially choking the electrical pathway, thereby resulting in power loss as heat, PWM actually turns off the power output rather than limits it. Applications range from **controlling DC motors and light dimming to heating elements.**


Thank You

Principles of Communication (BEC-28)

Unit-4

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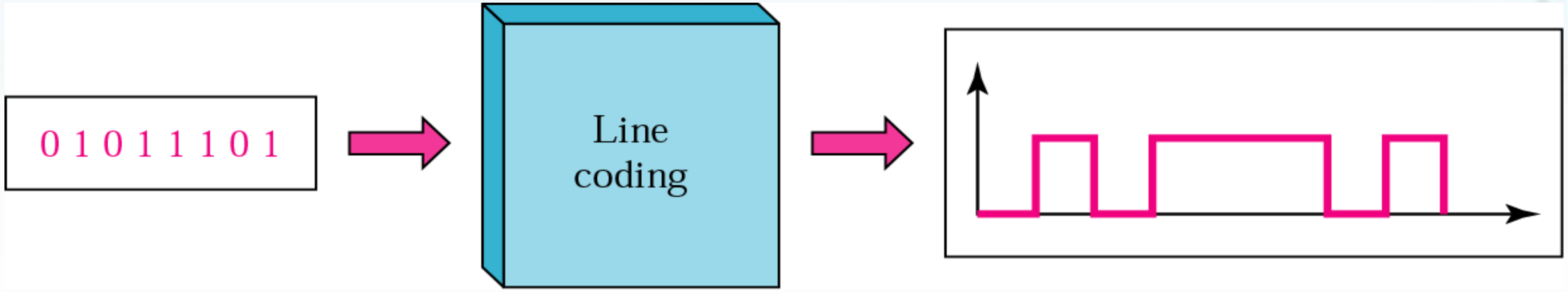
➢ What is LINE CODING ?

- *The process of converting digital data to digital signals*



Need Of Line Coding:

- Various Techniques
- Other Way: From Computers
- Information: Inherently discrete in nature
- Transmitted over band-limited channel: Signal gets Dispersed
- Causes: Overlap and Distortion
- Distortion: Inter-symbol Interference(ISI)

Properties of Line Coding

- Transmission Bandwidth: as small as possible
- Power Efficiency: As small as possible for given BW and probability of error
- Error Detection and Correction capability.
- Adequate timing content: Extract timing from pulses
- Transparency: Prevent long strings of 0s or 1s

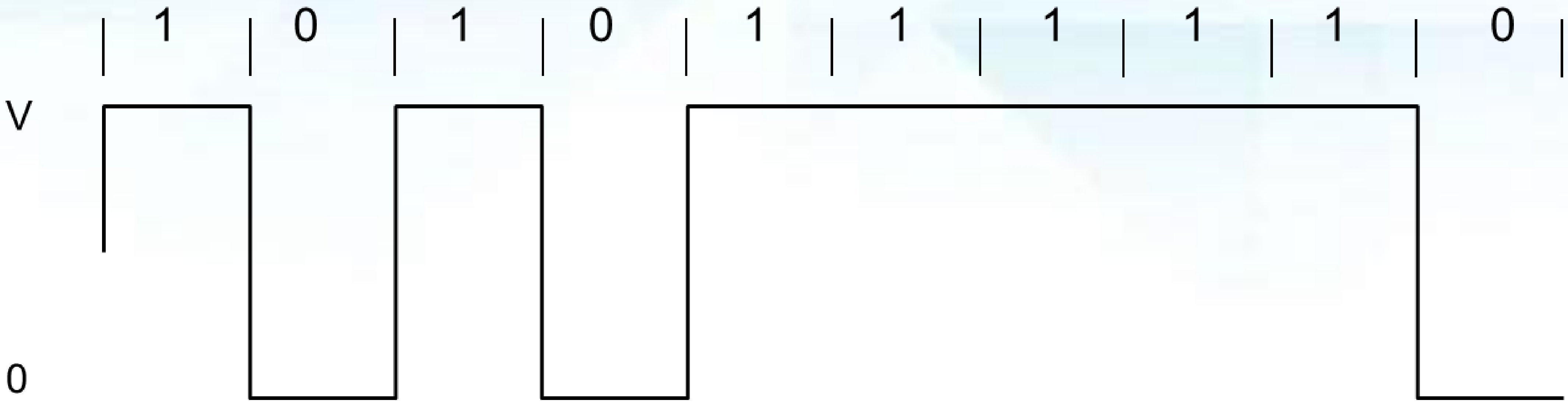
Line Coding

- Unipolar
- Polar
- Bipolar

1. Not-Return to Zero (NRZ)

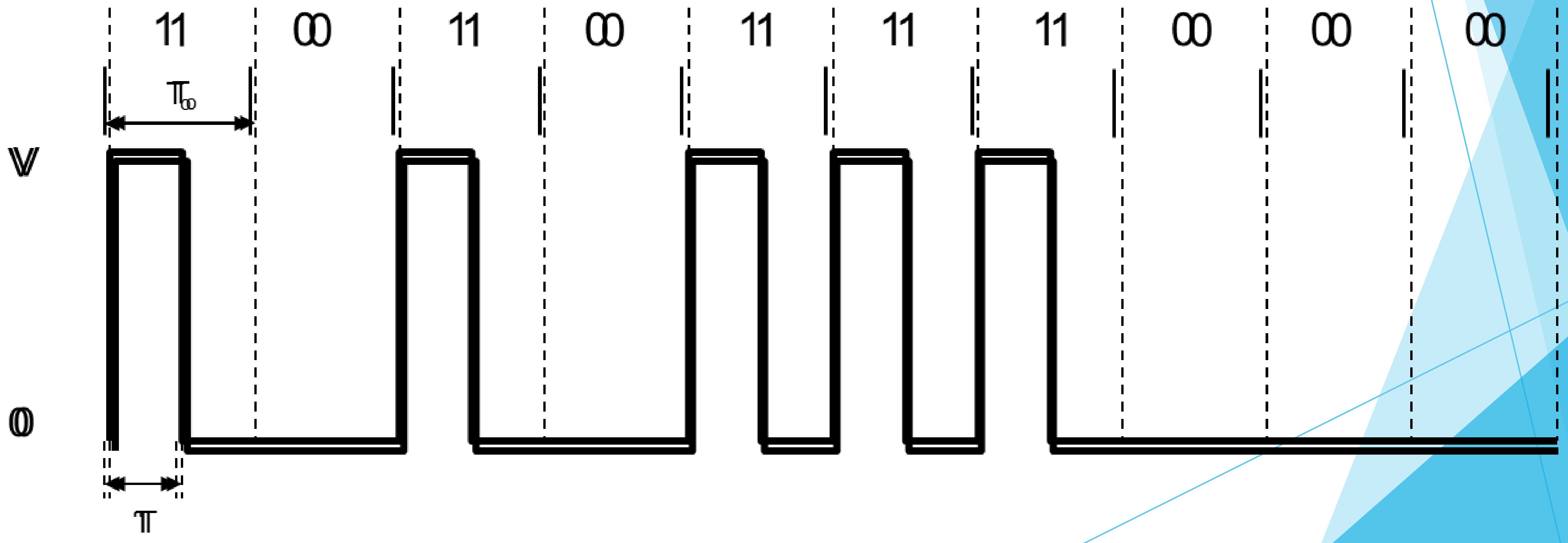
2. Return to Zero (RZ)

- Unipolar NRZ:
  - Pulse 0: Absence of pulse
  - Pulse 1 : Presence of pulse



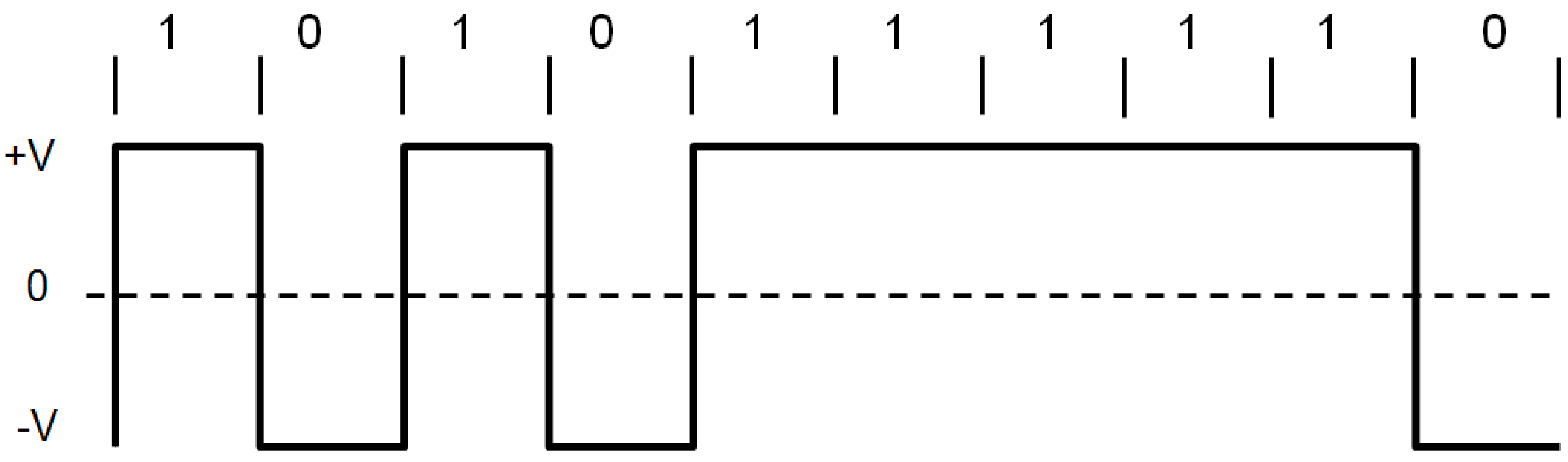
▪ Unipolar RZ:

- Pulse 0: Absence of pulse
- Pulse 1 : Presence of pulse



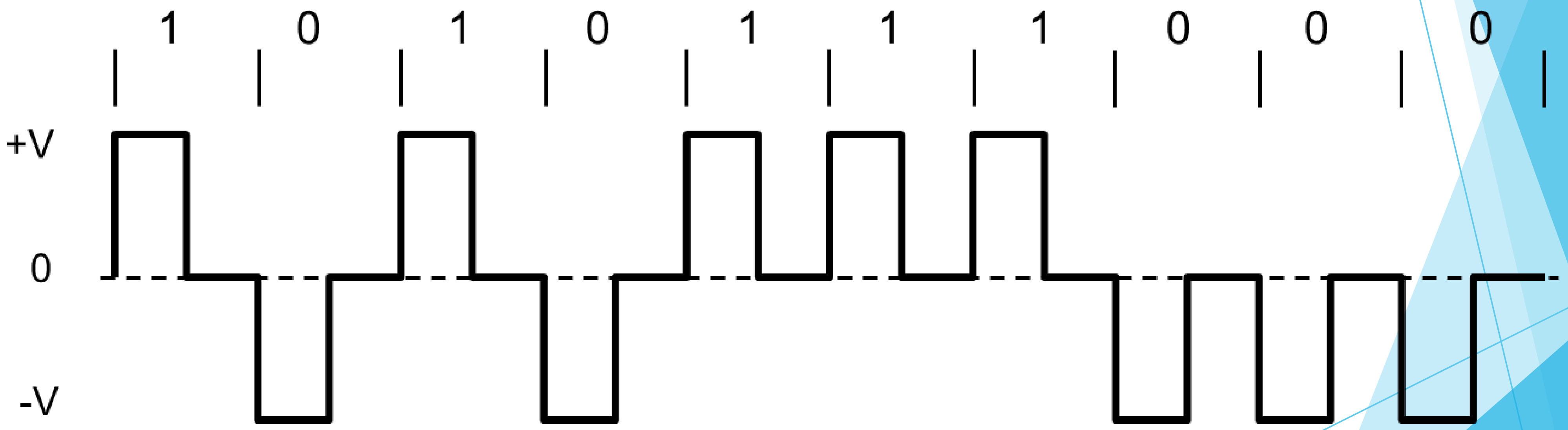
- Polar NRZ:

  - Pulse 1 : Presence of pulse
  - Pulse 0: Opposite of pulse



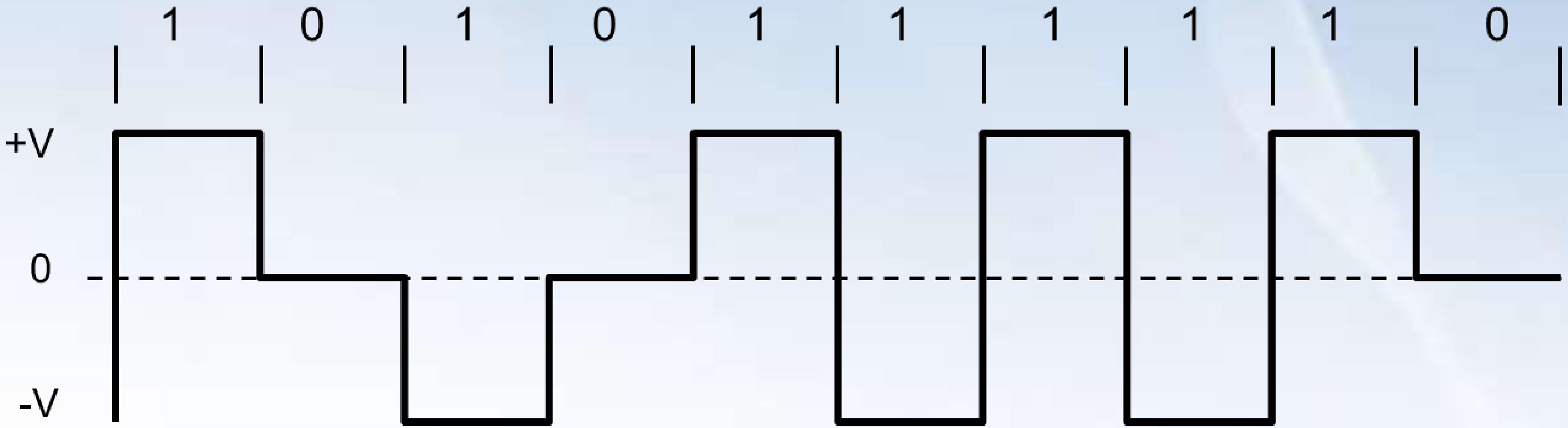
- Polar RZ:

•Pulse 1 : Presence of pulse
•Pulse 0: Opposite of pulse



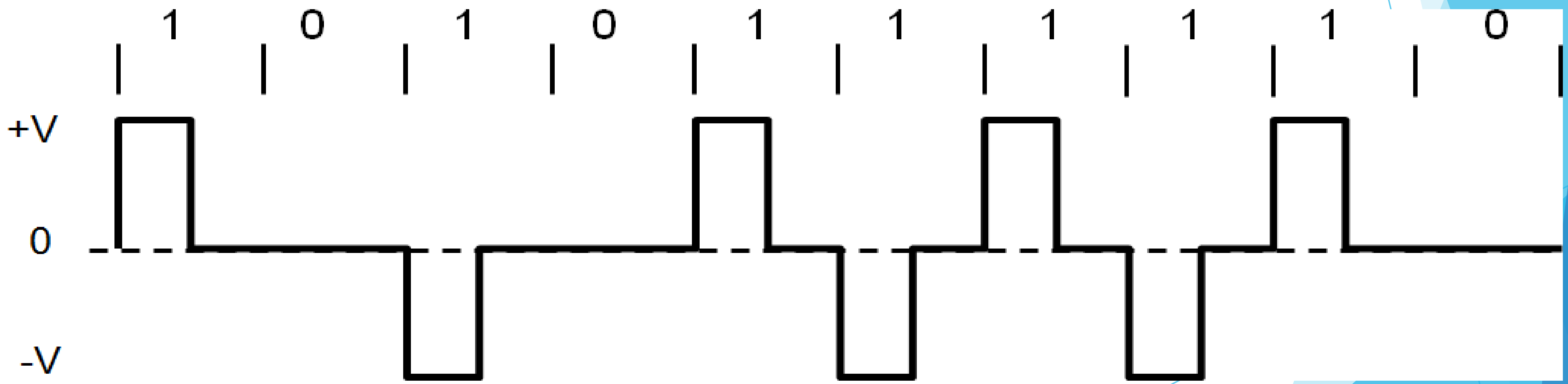
- Bipolar NRZ:

•Pulse 1 : Alternating voltage levels
•Pulse 0: Absence of pulse

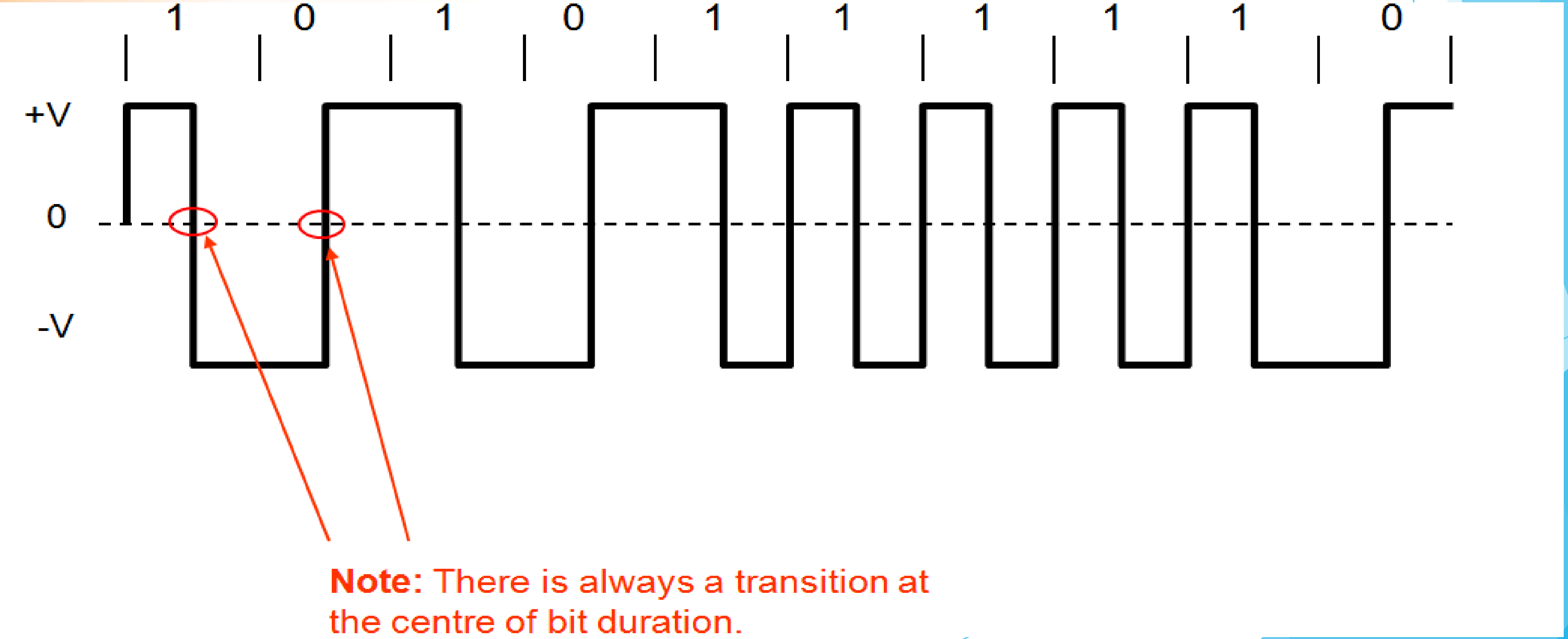


- Bipolar RZ:

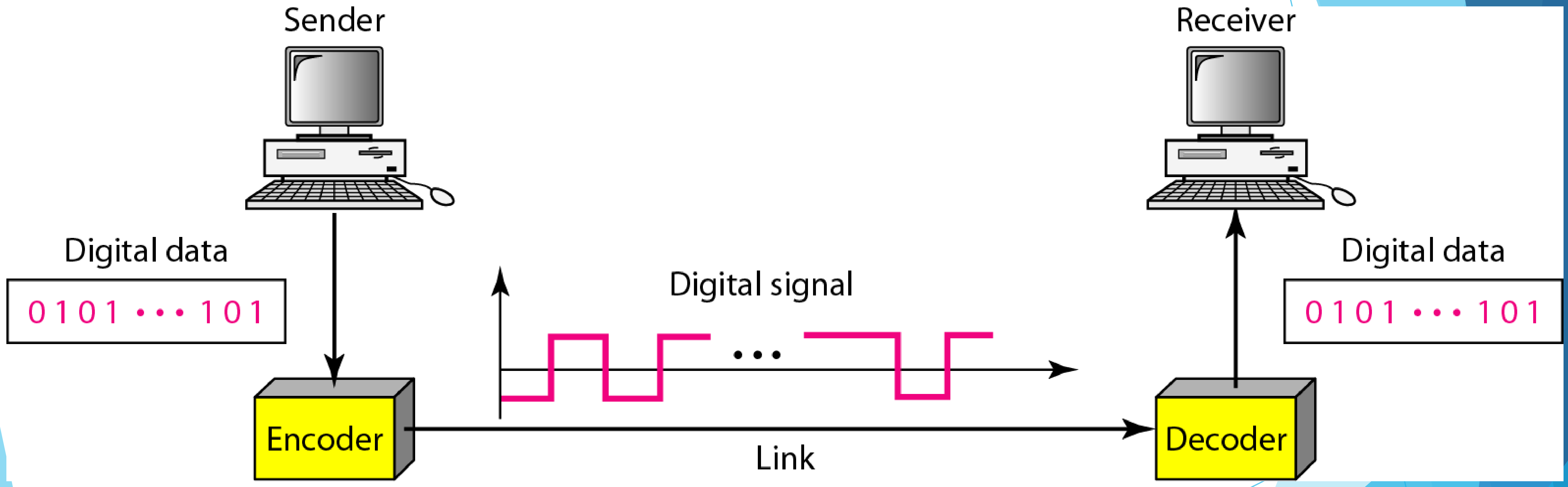
•Pulse 1 : Alternating voltage levels
•Pulse 0: Absence of pulse



- Manchester Coding:
  - Pulse 1 : +ve in 1st half and –ve in 2nd half
  - Pulse 0: -ve in 1st half and +ve in 2nd half



➤ Line coding and decoding




Thank You

Principles of Communication (BEC-28)

Unit-4

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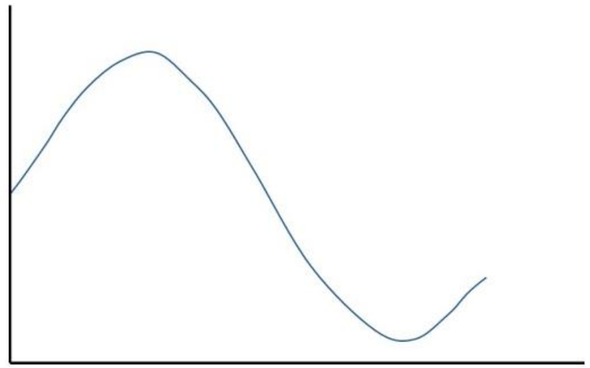
Quantization

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

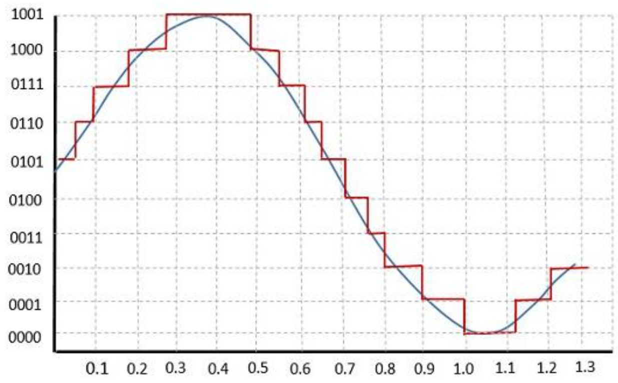
Quantizing an Analog Signal

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.



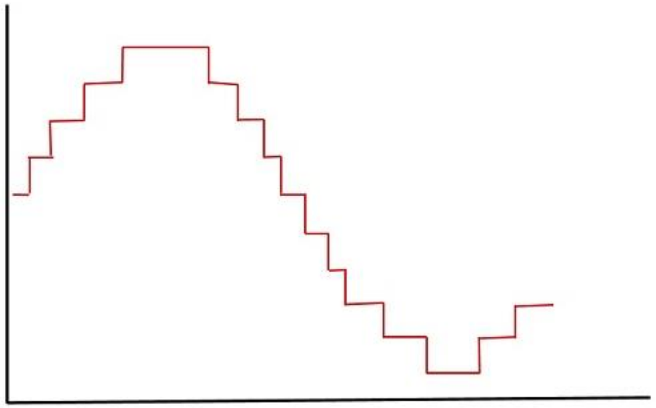
The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. **Quantization** is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.



Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as **representation levels** or **reconstruction levels**. The spacing between the two adjacent representation levels is called a **quantum** or **step-size**.

The following figure shows the resultant quantized signal which is the digital form for the given analog signal.



This is also called as **Stair-case** waveform, in accordance with its shape.

Types of Quantization

There are two types of Quantization - **Uniform Quantization and Non-uniform Quantization**.

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

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

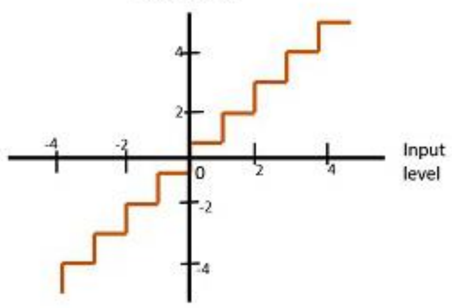


Fig 1 : Mid-Rise type Uniform Quantization

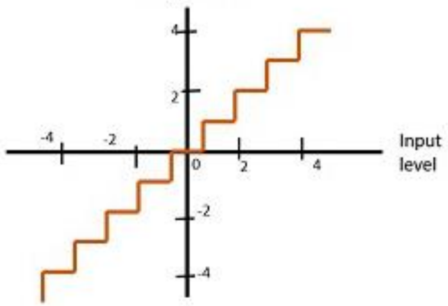


Fig 2 : Mid-Tread type Uniform Quantization

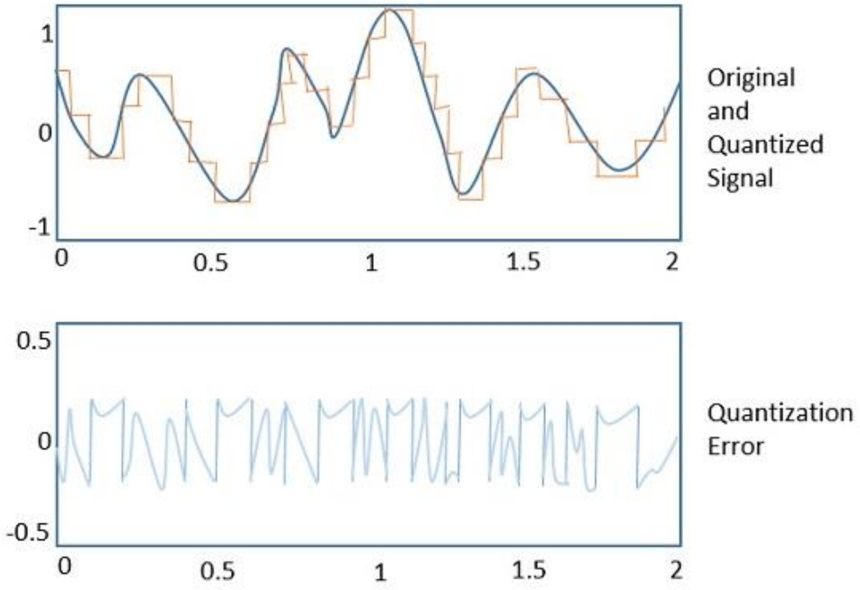
Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

Quantization Error:
For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

The difference between an input value and its quantized value is called a **Quantization Error**. A **Quantizer** is a logarithmic function that performs Quantization rounding the value

An analog-to-digital converter (**ADC**) works as a quantizer.
The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.



Quantization Noise
It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as **Quantization Noise**.

Companding in PCM

The word **Companding** is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques. They are −

A-law Companding Technique

- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear, and no compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.

μ-law Companding Technique

- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear, and no compression is done.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.

μ-law is used in North America and Japan


Thank You