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SAI VIDYA INSTITUTE OF TECHNOLOGY

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PRINCIPLES OF COMMUNICATION SYSTEMS (15EC45)

IV SEMESTER ECE

MODULE 5 : DIGITAL REPRESENTATION OF ANALOG SIGNALS

SYLLABUS: Introduction, Why Digitize Analog Sources?, The Sampling process, Pulse Amplitude Modulation, Time Division Multiplexing, Pulse-Position Modulation, Generation of PPM Waves, Detection of PPM Waves, The Quantization Process, Quantization Noise, Pulse–Code Modulation: Sampling, Quantization, Encoding, Regeneration, Decoding, Filtering, Multiplexing, Application to Vocoder.

MODULE 5

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Introduction, Why Digitize Analog sources ?, The sampling Process, Pulse Amplitude Modulation, Time Division Multiplexing, Pulse position modulation, Generation of PPM waves, Detection of PPM waves, The Quantization Process, Quantization Noise, Pulse Code Modulation : sampling, Quantization, Encoding, Regeneration, Decoding, Filtering, multiplexing, Application to Vocoder.

* INTRODUCTION :

* The evolution from analog to digital transmission is the conversion of common information sources such as voice and music, which are inherently analog to digital representation.
* In the first step from analog to digital, an analog source is sampled at discrete times. The resulting analog samples are then transmitted by means of analog pulse modulation.
* In the second step from analog to digital, an analog source is not only sampled at discrete times but the samples themselves are also quantized to discrete levels.

* WHY DIGITIZE ANALOG SOURCES :

There are many advantages that the transmission of digital information has over analog.

1) Digital systems are less sensitive to noise than analog.

2) With digital systems, it is easier to integrate different services. For example, video and the accompanying sound track, into the same transmission scheme.

3) The transmission scheme can be relatively independent of the source. For example, a digital transmission scheme that transmits voice at 10kbps could also be used to transmit computer data at 10kbps.

4) Circuitry for handling digital signals is easier to repeat and digital circuits are less sensitive to physical effects such as vibration and temperature.

5) Digital signals are simpler to characterize interms of bits 1 and 0 and do not have variability as analog signals. This makes the associated hardware easier to design.

6) Various media sharing stratergies known as multiplexing techniques are more easily implemented with digital transmission stratergies.

7) Digital techniques make it easier to specify complex standards that may be shared on a worldwide basis.

8) The techniques such as equalization, especially adaptive versions, are easier to implement with digital transmission techniques.

* COMPARISON BETWEEN ANALOG & DIGITAL COMM SYSTEMS:

S.NO	Parameter	Analog system	Digital System
1)	Bandwidth	Less	More
2)	Error correction and Detection	Not possible	Possible
3)	Immune to Noise	Less	More
4)	System Complexity	Less	More
5)	System Cost	More	Less
6)	Quality of reconstruction	Good	Very Good
7)	Synchronisation	Not required	required
8)	Privacy and security to data	Not possible	possible
9)	Flexibility and Liability	Less	More
10)	Power required	More	Less
11)	Implementation	Difficult	Easy
12)	Programming	Not possible	Possible

* SAMPLING PROCESS :

Statement: Sampling theorem states that any continuous time signal can be completely represented in its samples and recovered back if the sampling frequency is greater than or equal to twice the highest frequency component of base band signal.

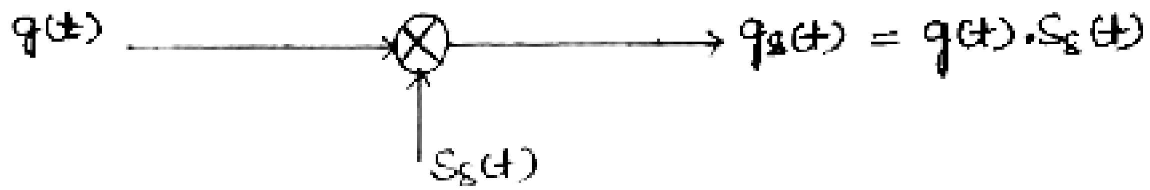
That is Sampling frequency, $f_s \geq 2W$ **.**

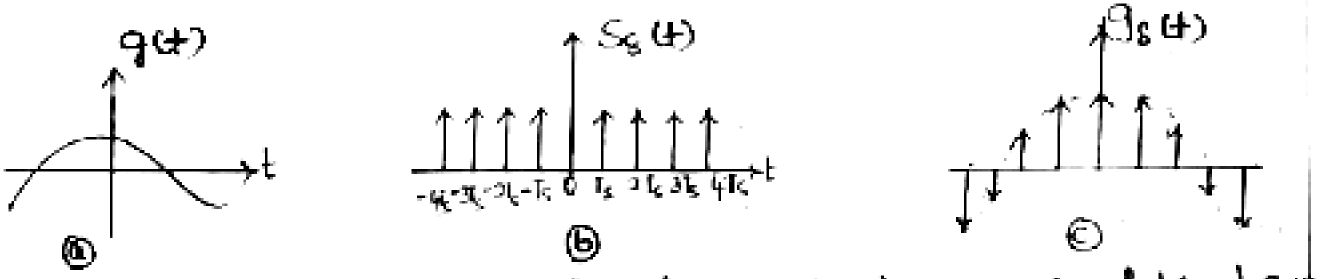
Where W= Highest frequency in base band continuous time signal.

This condition is also called Nyquist condition for sampling process.

Explanation and Proof:

* Consider an arbitrary signal g(t) of finite energy which is specified for all time. A segment of the signal g(t) is shown in fig(1)(a). Suppose that we sample the signal g(t) instantaneously and at a uniform rate, once every 'Ts' seconds. Consequently, we obtain an infinite sequence of samples spaced 'Ts' seconds apart and denoted by $\{g(nT_s)\}$, where 'n' takes on all possible integer values. We refer to 'Ts' as the sampling period, and to its reciprocal $f_s = 1/T_s$ as the sampling rate. This ideal form of sampling is called instantaneous sampling.





Fig(1): (a) analog signal (b) Periodic signal ($\delta_\delta(t)$) (c) Sampled signal $g_\delta(t)$

* Let '$g_\delta(t)$' denote the signal obtained by individually weighting the elements of a periodic sequence spaced T_s seconds. Therefore, sampled output $g_\delta(t)$ is given by,

$$g_\delta(t) = g(t) \cdot \delta_\delta(t) \qquad (1)$$

* Let $\delta_\delta(t)$ denotes the periodic impulse train and is represented as,

$$\delta_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \qquad (2)$$

substituting Eq-(2) in Eq-(1) we get

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

using shifting property of impulse function

WKT, $g(t) \cdot \delta(t - nT_s) = g(nT_s)\, \delta(t - nT_s)$

$$\therefore \boxed{g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\, \delta(t - nT_s)} \qquad (3)$$

For frequency domain consider,

$$g_\delta(t) = g(t) \cdot \delta_\delta(t)$$

Taking Fourier Transform on both sides, we get

$$G_\delta(f) = G(f) * \delta_\delta(f) \qquad (4)$$

where,

$$S_\delta(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \quad \text{———} (5)$$

substituting Eqⁿ (5) in Eqⁿ (4) we get.

$$G_\delta(f) = G(f) * f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

From convolution property of impulse function

wkt, $G(f) * \delta(f - nf_s) = G(f - nf_s)$

$$\therefore \quad G_\delta(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s) \quad \text{———} (6)$$

Eqⁿ (6) can be rewritten as,

$$G_\delta(f) = f_s\, G(f) + f_s \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} G(f - nf_s) \quad \text{———} (7)$$

When the spectrum of $G_\delta(f)$ is passed through an LPF, then the 2nd term in RHS of Eqⁿ (7) is eliminated resulting in

$$G_\delta(f) = f_s \cdot G(f)$$

$$\therefore \quad G(f) = \frac{1}{f_s} \cdot G_\delta(f) \quad \text{———} (8)$$

where $f_s = 2W$

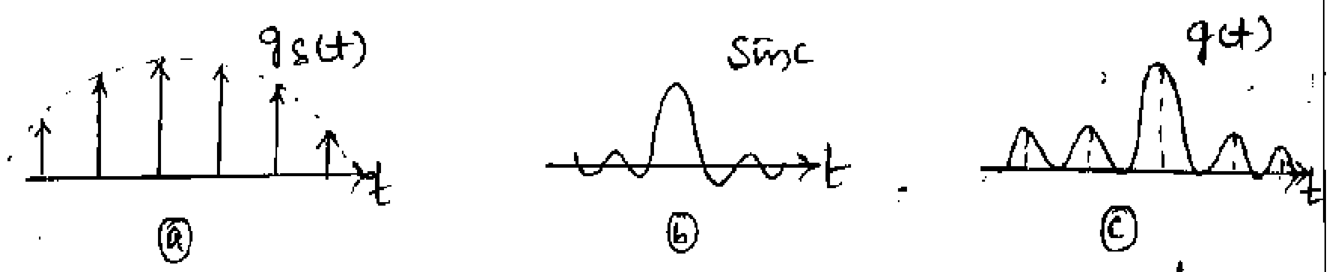


Fig : Recovering g(t) signal from sequence of samples g(nTs).

Now, we may state the sampling theorem for strictly band-limited signals of finite energy into two equivalent parts :

1) A bandlimited signal of finite energy, which only has frequency components less than "W" Hertz, is completely described by specifying the values of the signal at instants of time separated by $\frac{1}{2W}$ seconds.

2) A bandlimited signal of finite energy, which only has frequency components less than "W" Hertz, may be completely recovered from a knowledge of its samples taken at the rate of 2W samples per second.

The sampling rate of 2W samples per second, for a signal bandwidth of 'W' Hertz is called the Nyquist rate; its reciprocal $1/2W$ (measured in seconds) is called the Nyquist interval.

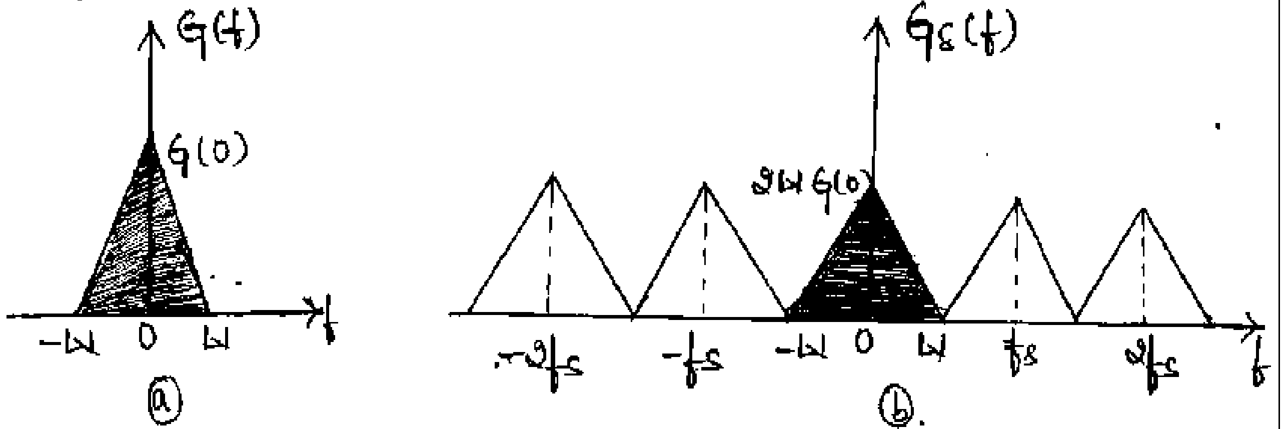
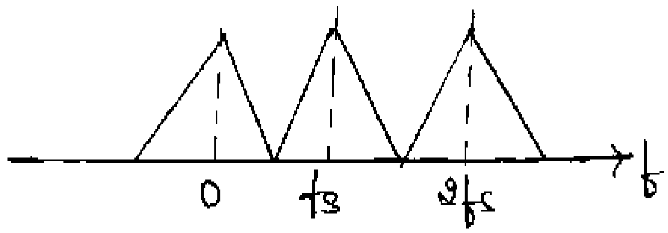


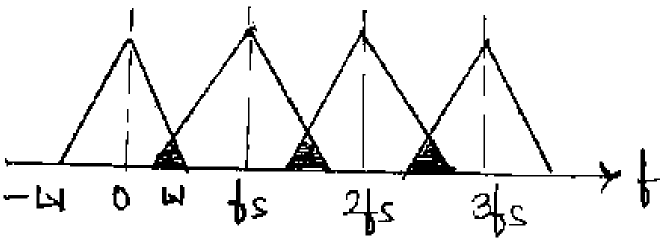
Fig : (a) Spectrum of a strictly band limited signal g(t).
(b) Spectrum of a sampled version of g(t) for $T_s = \frac{1}{2W}$.

NOTE : The concept of undersampling and over sampling is explained below.

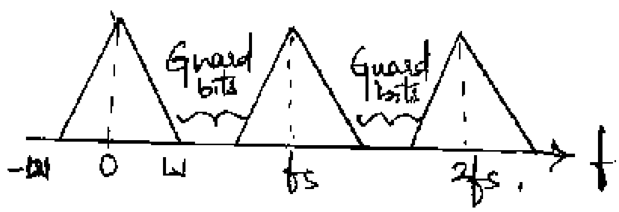
1) When sampling frequency $f_s = 2W$ then this type of sampling is called correct sampling and here there is no aliasing effect seen in this mechanism. i.e when $f_s = 2W$.



2) When $f_s < 2W$ then it is undersampling and there will be aliasing effect induced here.



3) When $f_s > 2W$ then it is over sampling and there will no aliasing effect.



* PROBLEMS :

(1) An analog signal is expressed by the equation $x(t) = 3\cos 50\pi t + 10\sin 300\pi t + \cos 100\pi t$. Calculate the Nyquist rate and Nyquist Interval for this signal.

Soln: Given

$$x(t) = 3\cos 50\pi t + 10\sin 300\pi t + \cos 100\pi t \quad \text{———} (1)$$

Comparing eqn (1) with std equation

$$x(t) = 3\cos \omega_1 t + 10\sin \omega_2 t + \cos \omega_3 t \quad \text{———} (2)$$

computing ω_1, ω_2 and ω_3 then

$\omega_1 = 50\pi \cancel{t}$ $\qquad \omega_2 = 300\pi$ $\qquad \omega_3 = 100\pi$

$\cancel{2}\pi f_1 = \cancel{50}\pi$ $\qquad \cancel{2}\pi f_2 = \cancel{300}^{150}\pi$ $\qquad \cancel{2\pi} f_3 = \cancel{100}\pi$

$\therefore \boxed{f_1 = 25\text{ Hz}}$ $\qquad \therefore \boxed{f_2 = 150\text{ Hz}}$ $\qquad \boxed{f_3 = 50\text{ Hz}}$

$\therefore \; f_m = \max(f_1, f_2, f_3)$

$\therefore \boxed{f_m = 150\text{ Hz}}$

$\therefore$ Nyquist rate $f_s = 2f_m = 2 \times 150$

$\therefore \boxed{f_s = 300\text{Hz}}$

Nyquist Interval, $T_s = 1/f_s = 1/300\text{ Hz}$

$\therefore \boxed{T_s = 3.3\text{ ms}}$

(2) An analog signal is expressed by the equation $x(t) = \frac{1}{2\pi} \cos(4000\pi t)\cos(1000\pi t)$. Calculate the Nyquist rate and Nyquist Interval for this signal.

Soln: Given

$$x(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t) \quad \text{———} (1)$$

WKT $\cos A . \cos B = \frac{1}{2}\left[\cos(A-B) + \cos(A+B)\right]$

$$\therefore \quad x(t) = \frac{1}{4\pi}\left[\cos(4000\pi - 1000\pi)t + \cos(4000\pi + 1000\pi)t\right]$$

i.e $x(t) = \frac{1}{4\pi}\left[\cos \omega_1 t + \cos \omega_2 (t)\right]$

$\therefore \quad \omega_1 = 3000\pi \qquad \omega_2 = 5000\pi$

$\cancel{2\pi} f_1 = 3\cancel{000}\pi \qquad \cancel{2\pi} f_2 = \cancel{5000}\pi$

$\boxed{f_1 = 1500\,Hz} \qquad \boxed{f_2 = 2500\,Hz}$

$\therefore \quad f_m = \max(f_1, f_2)$

$= \max(1500Hz, 2500Hz)$

$\therefore \boxed{f_m = 2500\,Hz}$

$\therefore$ Nyquist Rate $f_s = 2 \times f_m = 2 \times 2500$

$\therefore \boxed{f_s = 5000\,Hz}$

$\therefore$ Nyquist Interval $T_s = 1/f_s = 1/5000 \quad \therefore \boxed{T_s = 0.2\,ms}$

* PULSE AMPLITUDE MODULATION :

* It is an analog pulse Modulation scheme in which the amplitudes of a train of rectangular carrier pulses are varied in accordance with the sample values of the modulating signal.

* In PAM, the top of each modulated rectangular pulse is maintained flat. So PAM is same as flat-top sampling.

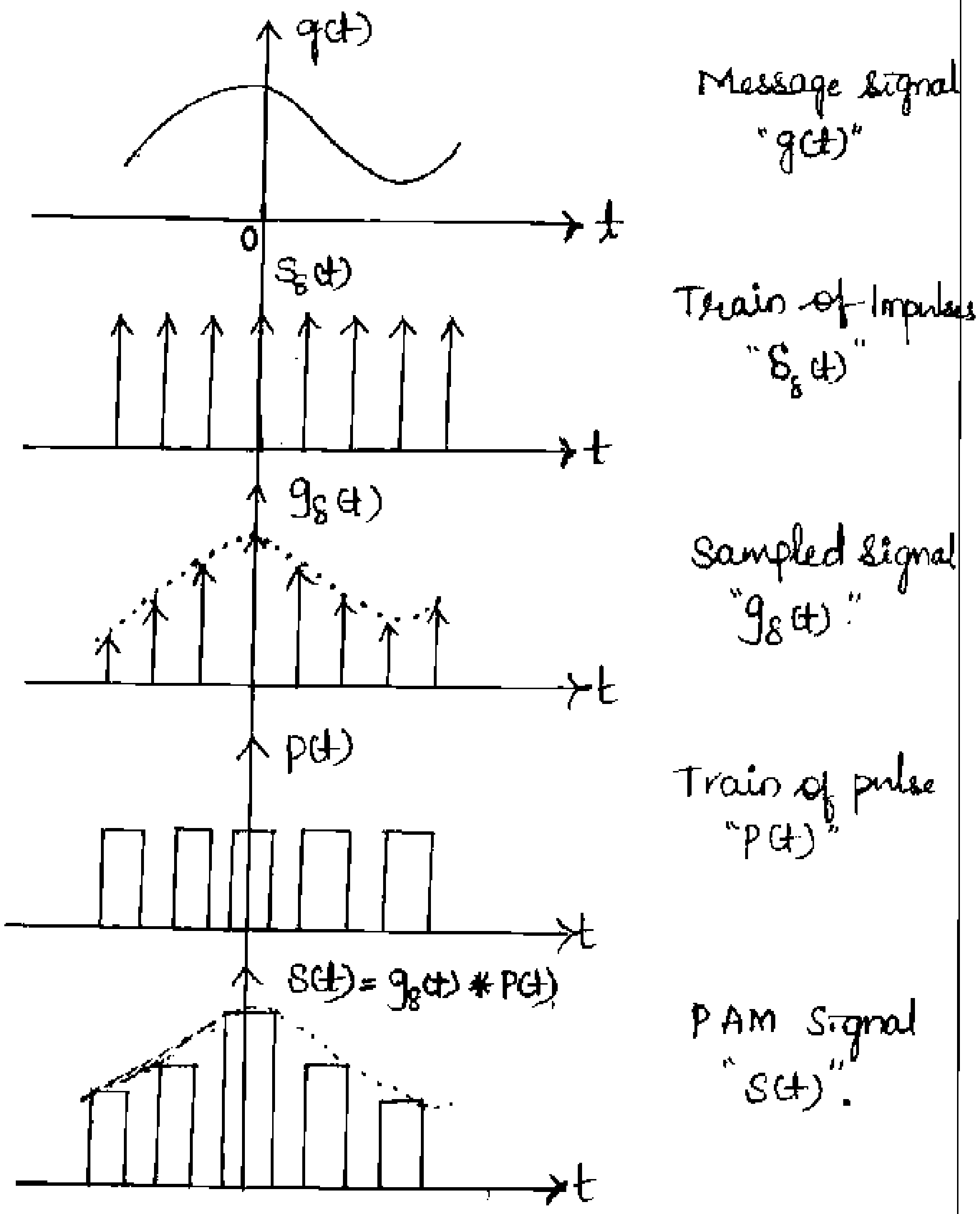


Fig4: Pulse Amplitude Modulation

The waveform of a PAM signal is illustrated in fig(4).

* Let $s(t)$ denote the sequence of flat-top samples or PAM signal, and it is expressed as

$$s(t) = \sum_{n=-\infty}^{\infty} q(nT_s)\, p(T - nT_s) \qquad (1)$$

where,

$q(nT_s)$ is the sample value of $q(t)$ obtained at time $t = nT_s$.

T_s is sampling period.

$p(t)$ is standard rectangular pulse train of duration T.

Advantages of PAM :

It is a base for all the digital modulation technique.

Disadvantages of PAM :

1) Due to Nyquist criteria, it requires high bandwidth for transmission.

2) Since, amplitude keeps varying, so there is noise associated with it.

* Detection of PAM signal

The original message signal $m(t)$ is obtained by passing PAM signal to the reconstruction filter followed by equilizer.

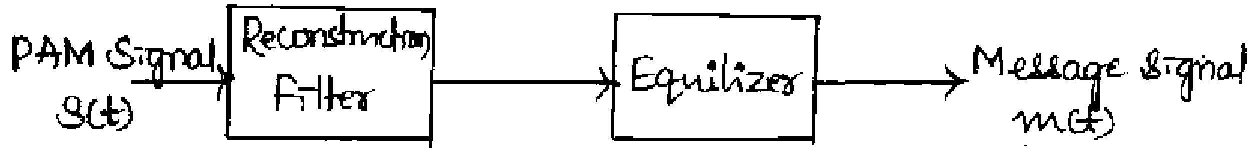


Fig4(b) : Recovering m(t) from PAM signal

* TIME DIVISION MULTIPLEXING : [TDM]

Time Division Multiplexing is a method of transmitting and receiving independent signals over a common channel by means of synchronised switches at each end of transmission line so that each signal appears on the line only a fraction of time in an alternating pattern.

* Fig(5) shows the block diagram of TDM system.

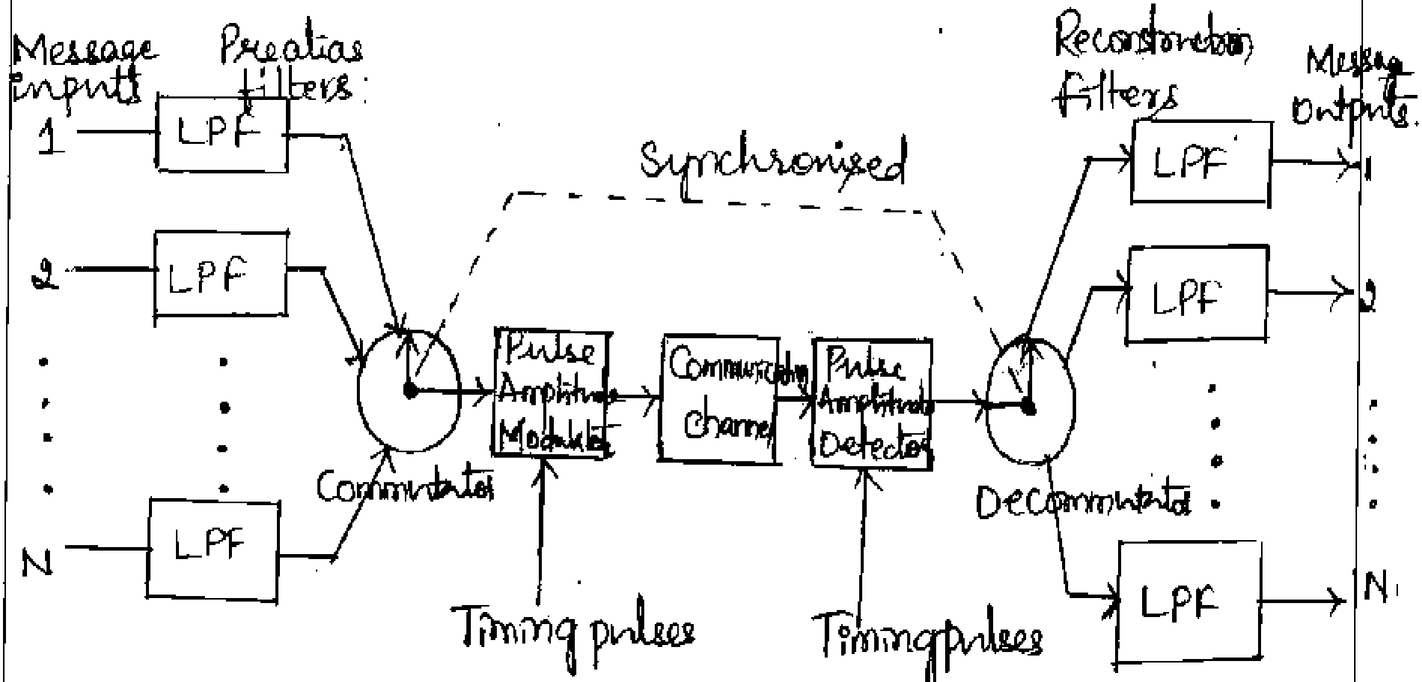


Fig 5 : Block Diagram of TDM system.

* The concept of TDM is illustrated in the fig(5). The Lowpass filters are used to remove high frequency components present in the message signal. The output of the pre-alias filters are then fed to a commutator which is usually implemented using electronic switching circuitry.

* The function of commutator is as follows:

1) To take a narrow sample of each of the 'N' samples of input at a rate of $f_s \geq 2W$.

2) To sequentially interleave (multiplex) these 'N' samples inside a sampling interval $T_s = 1/f_s$.

* The multiplexed signal is then applied to a pulse amplitude modulator whose purpose is to transform the multiplexed signal into a form suitable for transmission over a common channel.

* At the receiving end, the pulse amplitude demodulator performs the reverse operation of PAM and the decommutator distributes the signals to the appropriate low pass reconstruction filters. The decommutator operates in synchronisation with the commutator.

* PULSE-POSITION MODULATION :

* In pulse-duration modulation (PDM), the samples of the message signal are used to vary the duration of the individual pulses. This form of modulation is also referred to as pulse-width modulation or pulse-length modulation.

* In PPM, the position of a pulse relative to its unmodulated time of occurance is varied in accordance with the message signal as shown in fig(6)(d) for the case of sinusoidal modulation.

Let 'T_s' denote the sample duration. Using the sample $m(nT_s)$ of a message signal $m(t)$ to modulate the position of the n^{th} pulse, we obtain the PPM signal

$$s(t) = \sum_{n=-\infty}^{\infty} g(t - nT_s - k_p m(nT_s)) \quad (1)$$

where k_p is the sensitivity of the pulse-position modulator

a) Modulating signal $m(t)$

b) Pulse carrier

c) PDM wave

d) PPM wave

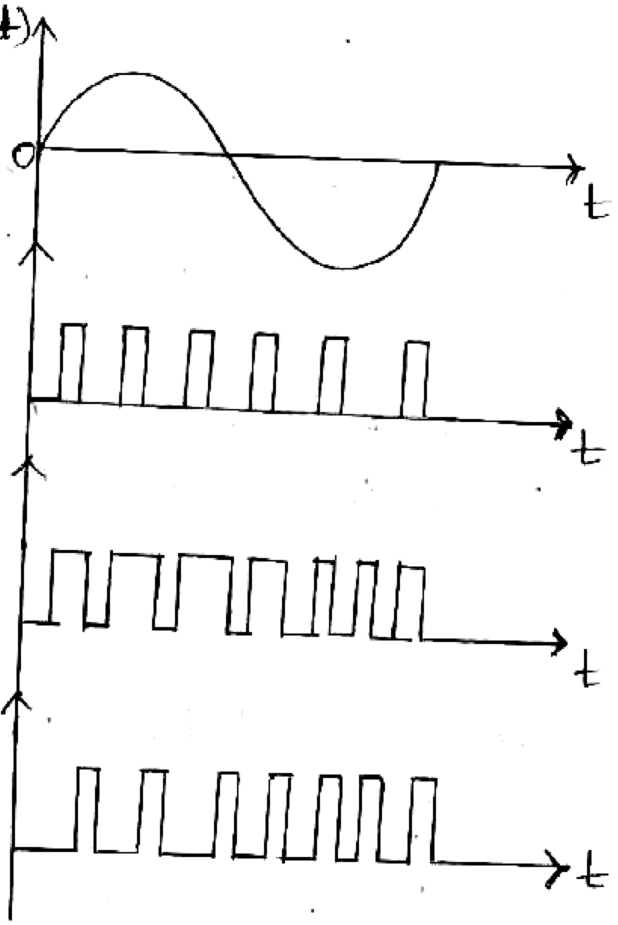


Fig (6) : Illustrating two different forms of pulse-time modulation.

GENERATION OF PPM WAVES :

The PPM signal which is generated is shown in fig(7)(a). The message signal m(t) is first converted into a PAM signal by means of a sample and Hold circuit, generating a staircase waveform u(t), which is shown in 8(b) for the message signal m(t) shown in fig 8(a).

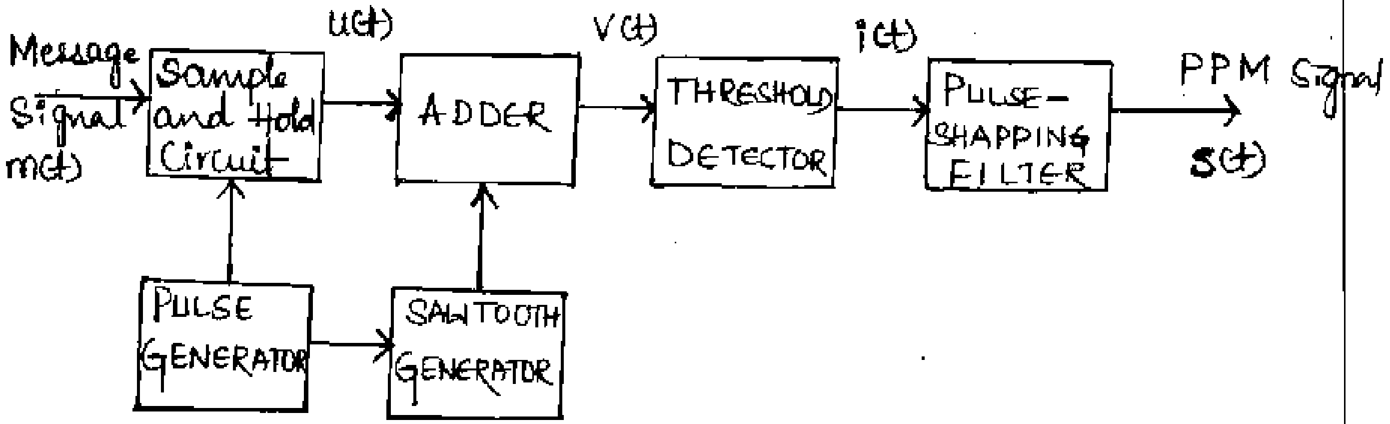
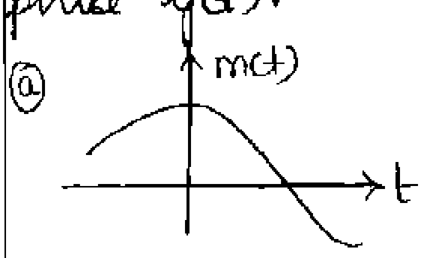
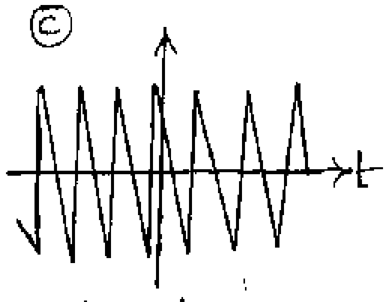
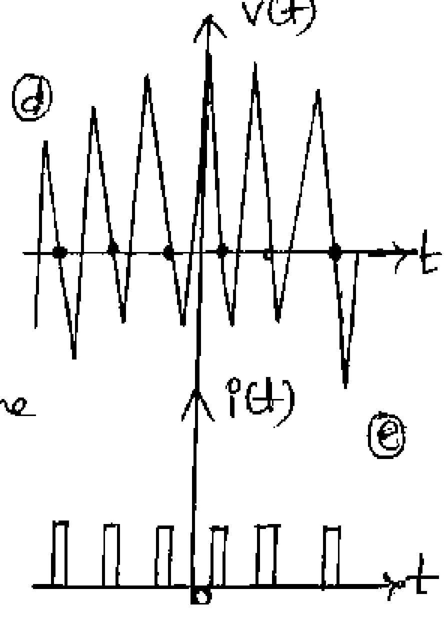


Fig 7(a) : Block diagram of PPM generator.

* Next, the signal u(t) is added to a sawtooth wave, yielding the combined signal v(t). The combined signal v(t) is applied to a threshold detector that produces a very narrow pulse each time v(t) crosses zero in the -ve going direction. The resulting sequence of "impulses" i(t) is shown in fig (8)(e). Finally, the PPM signal S(t) is generated by using this sequence of impulses to excite a filter whose impulse response is defined by the standard pulse g(t).







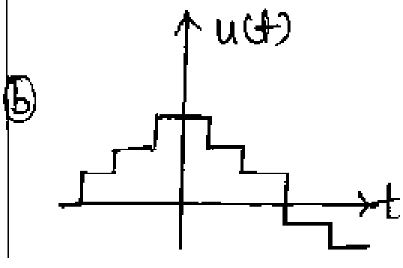


Fig 8 : Generation of PPM wave
(a) Message signal
(b) staircase signal
(c) Sawtooth wave
(e) PPM signal

* DETECTION OF PPM WAVES :

Consider a PPM wave $s(t)$ with uniform sampling, and assume that the message signal $m(t)$ is strictly band limited. The operation of one type of PPM receiver may proceed as follows :

1) Convert the received PPM wave into a PDM wave with the same modulation.

2) Integrate this PDM wave using a device with a finite integration time, thereby computing the area under each pulse of the PDM wave.

3) Sample the output of the integrator at a uniform rate to produce a PAM wave, whose pulse amplitudes are proportional to the signal samples $m(nT_s)$ of the original PPM wave $s(t)$.

4) Finally, demodulate the PAM wave to recover the message signal $m(t)$.

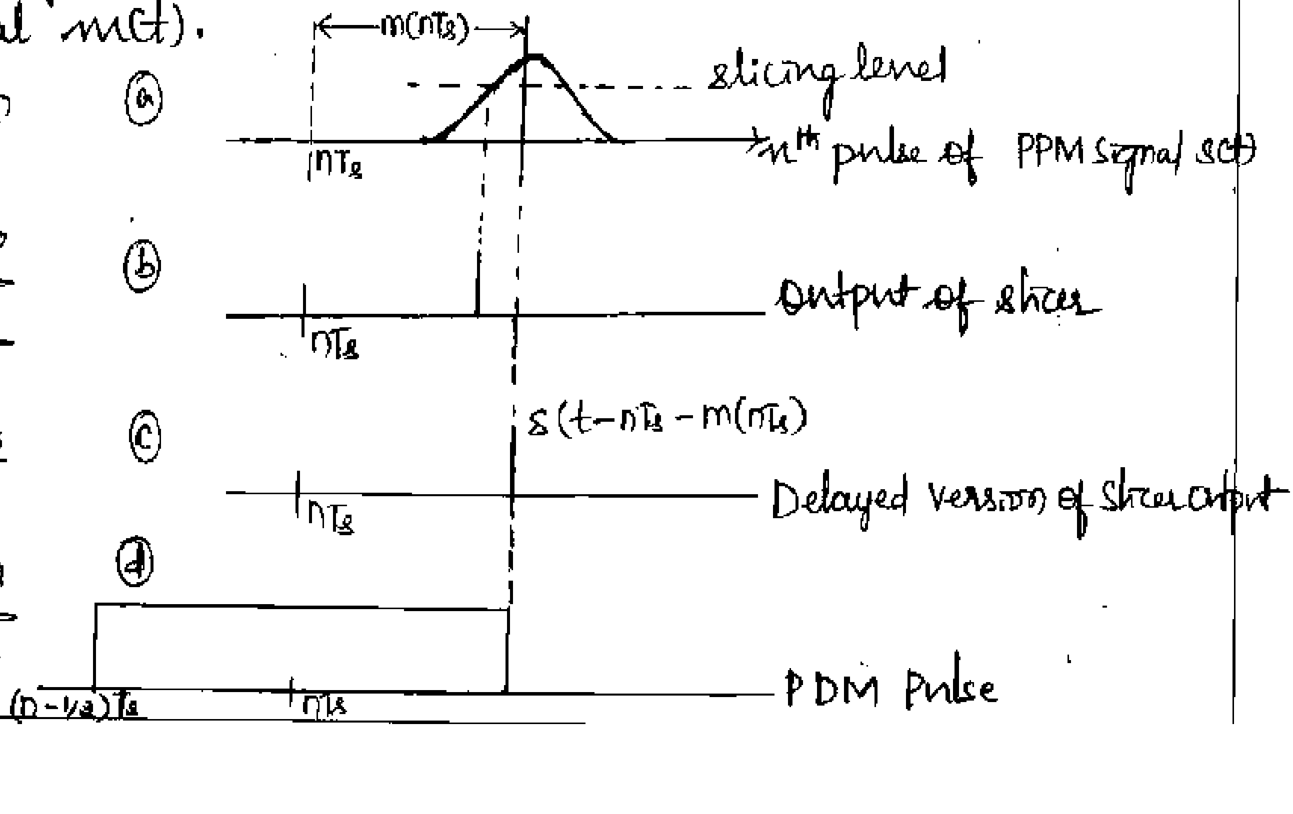


Fig : Detection of noiseless PPM signal.

* NOISE IN PULSE-POSITION MODULATION :

* In a PPM system, the transmitted information is contained in the relative positions of the modulated pulses. The presence of additive noise affects the performance of such a system.

* The output signal to noise ratio, assuming a full-load sinusoidal modulation, is therefore

$$(SNR)_o = \frac{\pi^2 B_T T_s^2 A^2}{32 N_0} \quad \text{———} \quad (1)$$

* The avg noise power in a message bandwidth 'W' is equal to WN_0. The channel signal to noise ratio is therefore,

$$(SNR)_c = \frac{3A^2}{4 T_s B_T W N_0} \quad \text{———} \quad (2)$$

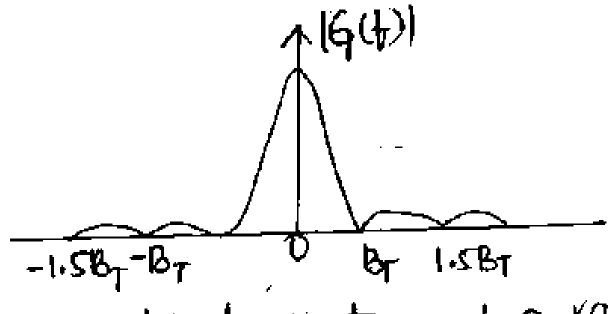


Fig : Amplitude spectrum of a raised Cosine pulse.

* Thus, the figure of merit of a PPM system using a raised cosine pulse is as follows.

$$\text{Figure of Merit} = \frac{(SNR)_o}{(SNR)_c} = \frac{\pi^2 B_T T_s^2 A^2}{32 N_0} \times \frac{4 T_s B_T W N_0}{3A^2}$$

$$\therefore \boxed{\text{Figure of Merit} = \frac{\pi^2}{24} B_T^2 T_s^3 W}$$

* THE QUANTIZATION PROCESS :

The process of transforming sampled amplitude values of a message signal into a discrete amplitude value (levels) is referred to as quantization.

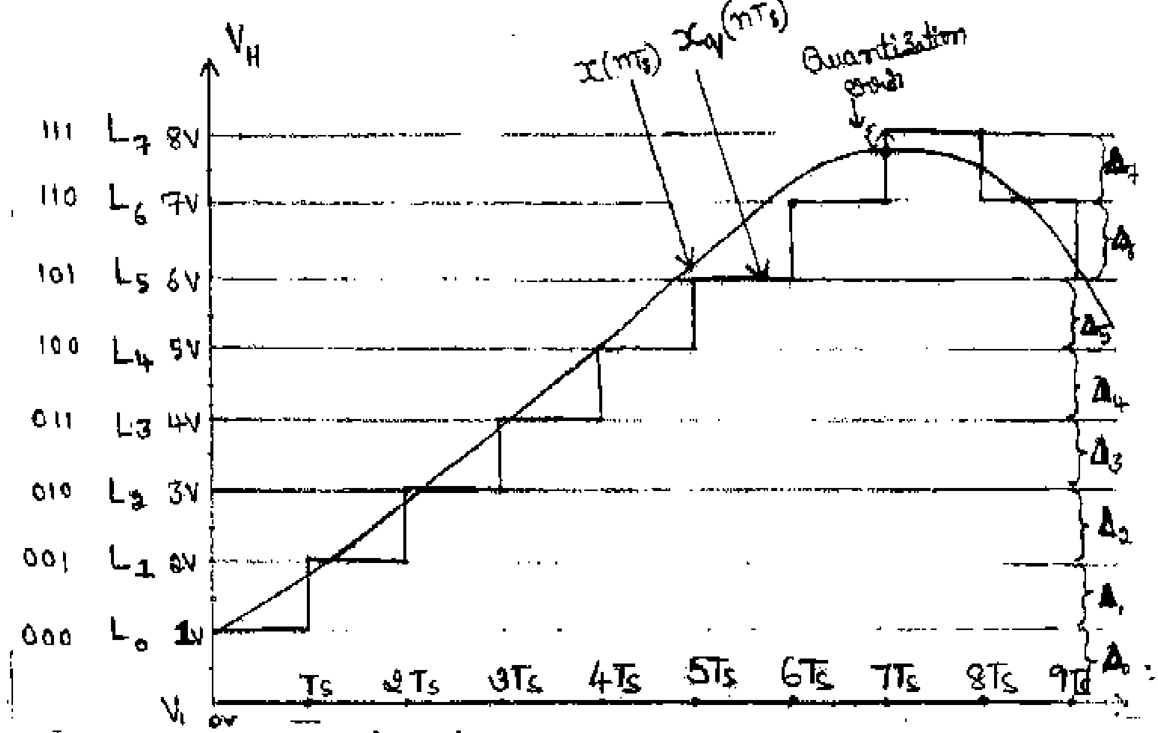


Fig : Quantization Process

* The signal $x(t)$ whose excursion is confined to the range from V_L to V_H being divided into 8-equal levels.

$$x(nT_s) \longrightarrow \boxed{\text{Quantizer}} \longrightarrow x_q(nT_s)$$

* Step size is denoted by 'Δ' and is given by

$$\Delta = \frac{V_H - V_L}{L}$$

where $L = 2^R$ and R is no. of bits.

$$\therefore \quad \Delta = \frac{V_H - V_L}{2^R}$$

If the step size 'Δ' is maintained same through the process of quantization, then it is called "uniform quantization".

* The difference between the continuous amplitude sample level and quantized signal level is known as quantization error.

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

where Quantization error varies from $+\Delta/2$ to $-\Delta/2$.

* The random errors due to quantization process produces a noise at the output of the quantizer and this noise is referred to as Quantization noise.

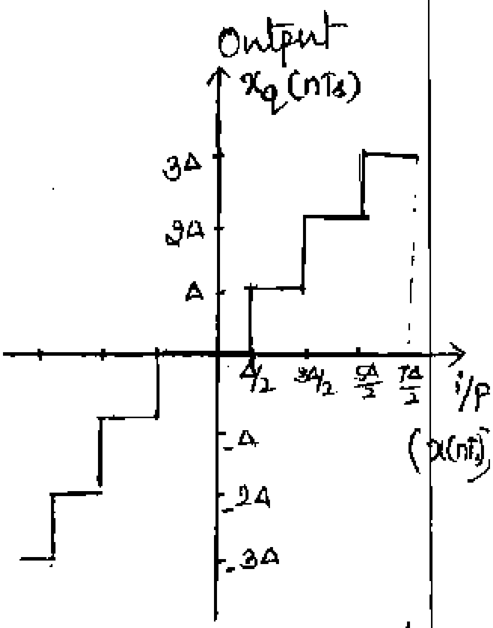
* Consider fig(A), the sampling, Quantizing and coding of an analog signal is as follows.

sampled values of an analog signal	1.7V	2.7V	3.9V	5V	6.2V	7.2V	7.7V	7.4V
Nearest Quantizer level	L_1	L_2	L_3	L_4	L_5	L_6	L_7	L_8
quantizer level voltage	2V	3V	4V	5V	6V	7V	8V	7V
Binary Code	001	010	011	100	101	110	111	110

* There are two types of quantizer they are
  - Mid-tread type quantizer
  - Mid-riser type quantizer

1) Mid-Tread type Quantizer :

* In mid-tread quantizer, the decision threshold of the quantizers are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \ldots$, and the representation levels are located at $0, \pm\Delta, \pm 2\Delta$, where Δ is the step size.

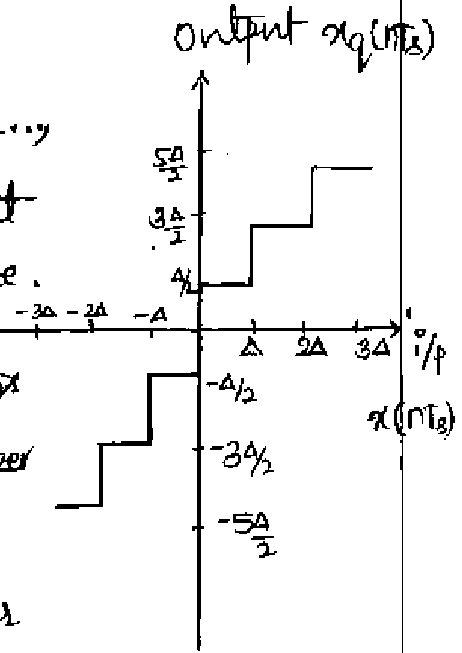


Midtread Quantizer

* A uniform quantizer characterized in this way is referred to as a symmetric quantizer of the midtread type, because the origin lies in the middle of a tread of a staircase graph. Here, quantization levels are odd number.

2) Mid-Riser type Quantizer :

* In midriser quantizer, the decision threshold of the quantizers are located at $0, \pm\Delta, \pm 2\Delta \ldots$, and the representation levels are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \pm\frac{5\Delta}{2} \ldots$, where Δ is the step size.

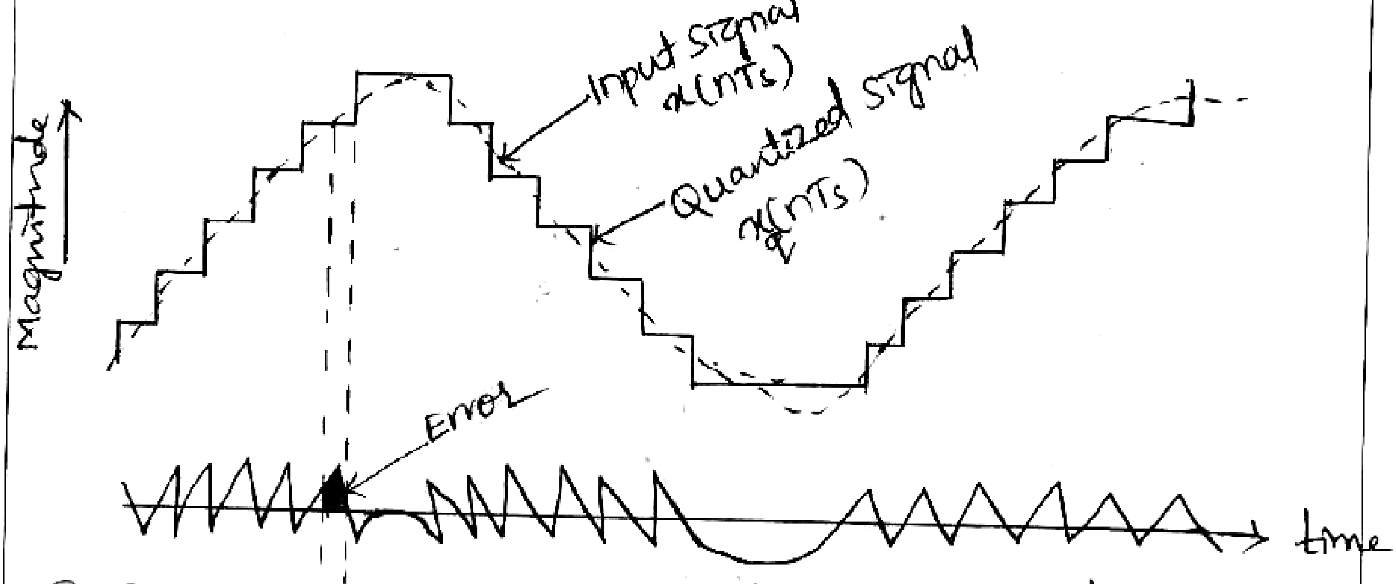


Midrise Quntizer

* A uniform quantizer characterized in this way is referred to as a symmetric quantizer of the mid riser type, because the origin lies in the middle of a riser of the stair-case graph.
* Here Quantization levels are even number.

* QUANTIZATION NOISE

* The use of quantization introduces an error defined as the difference between the input signal 'm' and the output signal 'v'. This error is called Quantization noise. Fig(B), illustrates a typical variation of the Quantization noise as a function of time, assuming the use of a uniform quantizer of the midtread type.



Fig(B) : Illustration of Quantization process and Noise

* Let the random variable 'Q' denotes the quantization error and 'q' its sample value.

$$q = m - v \qquad (1)$$

* Consider then an input 'm' of continuous amplitude in the range $(-m_{max}, m_{max})$ then, the step-size of the quantizer is given by,

$$\Delta = \frac{2 m_{max}}{L} \qquad (2)$$

* Now, the probability density function of the quantization error 'Q' as follows,

$$f_Q(q) = \begin{cases} \frac{1}{\Delta}, & -\frac{\Delta}{2} < q \leq \frac{\Delta}{2} \\ 0, & \text{otherwise} \end{cases} \quad \text{——— (3)}$$

* Now, the variance of Quantization error is

$$\sigma_Q^2 = \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} q^2 f_Q(q)\, dq$$

$$\sigma_Q^2 = \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} q^2 \cdot \frac{1}{\Delta}\, dq \qquad \because f_Q(q) = \frac{1}{\Delta}$$

$$= \frac{1}{\Delta}\left[\frac{q^3}{3}\right]_{-\Delta/2}^{\Delta/2} = \frac{1}{3\Delta}\left[\left(\frac{\Delta}{2}\right)^3 - \left(\frac{-\Delta}{2}\right)^3\right] = \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} - \frac{(-\Delta)^3}{8}\right]$$

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8}\right] = \frac{1}{3\cancel{\Delta}} \times \cancel{2}\left(\frac{\cancel{\Delta^3}}{\cancel{8}4}\right) = \frac{1}{3} \cdot \frac{\Delta^2}{4}$$

$$\therefore \boxed{\sigma_Q^2 = \frac{\Delta^2}{12}}$$

This is known as "Mean squared quantization error" or Normalized Noise power or Quantization error in terms of power. (A)

* Let us consider "R" which denote the number of bits per sample then the Quantized level is given by,

$$L = 2^R \quad \text{——— (4)}$$

Substituting Eq^n (4) in Eq^n (2) we get,

$$\Delta = \frac{2\, m_{max}}{2^R} \quad \text{——— (5)}$$

Now substitute Eq^n (5) in Eq^n (A) we get

$$\therefore \sigma_Q^2 = \left[\frac{\frac{2m_{max}}{2^R}}{12}\right]^2 = \frac{4\, m_{max}^2}{2^{2R}} \times \frac{1}{12_3}$$

$$\therefore \sigma_Q^2 = \frac{1}{3} m_{max}^2 2^{-2R} \qquad (6)$$

Let 'P' denote the avg power of message signal m(t). We may express the output signal to noise ratio of a uniform quantizer as,

$$(SNR)_0 = \frac{P}{\sigma_Q^2}$$

$$= \frac{P}{\frac{1}{3} m_{max}^2 2^{-2R}}$$

$$\therefore \boxed{(SNR)_0 = \left(\frac{3P}{m_{max}^2}\right) 2^{2R}}$$

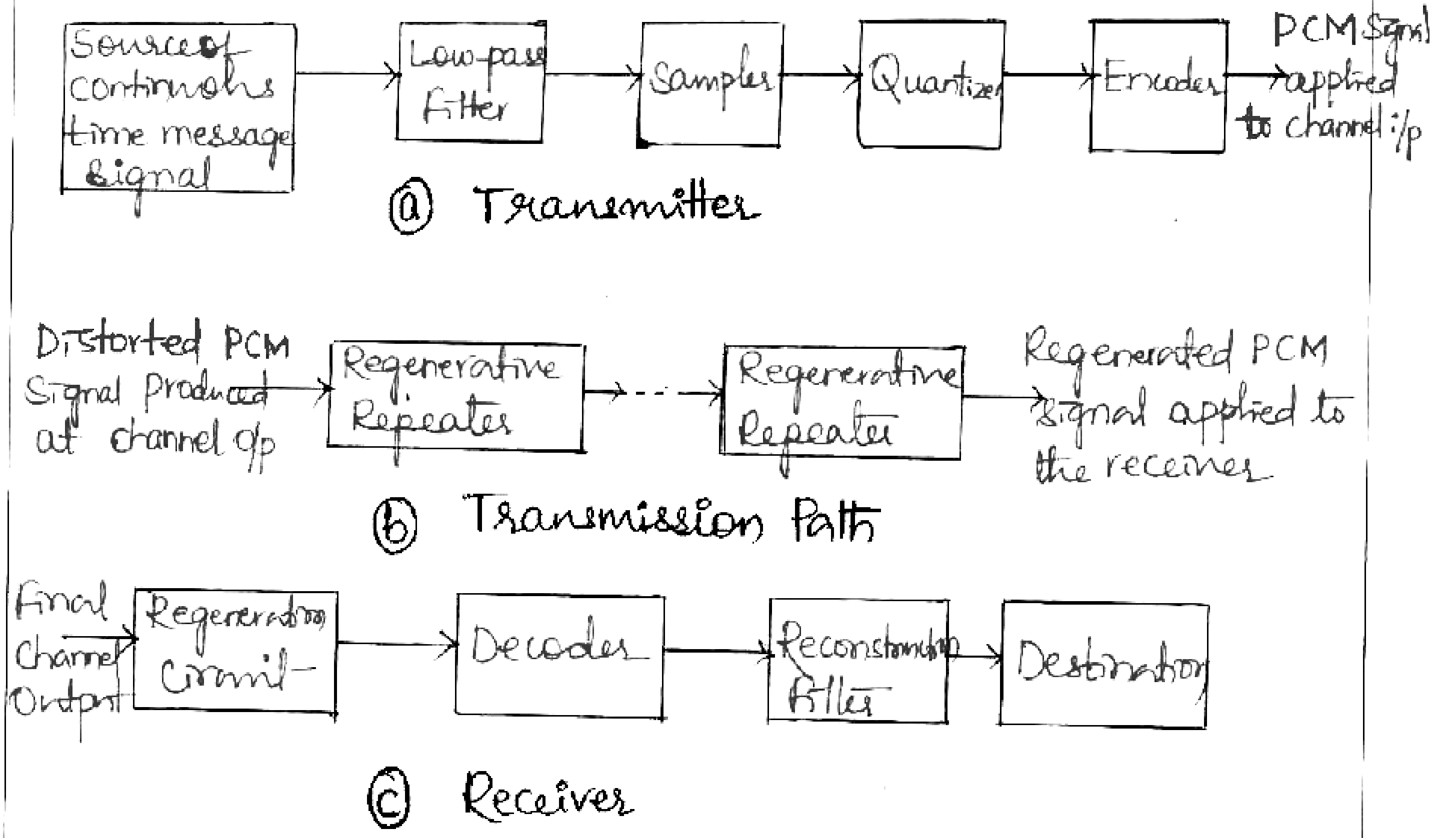
* **PULSE CODE MODULATION :**

* In pulse code Modulation (PCM), a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude.

* The basic operations performed in the transmitter of a PCM system are sampling, quantizing and encoding as shown in fig 6(a). The Lowpass filter prior to sampling

is included to prevent aliasing of the message signal. The quantizing and encoding operations are usually performed in the same circuit, which is called an analog to-digital converter.

* The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction of the train of quantized samples as shown in fig 6(c). Regeneration also occurs at intermediate points along the transmission path as necessary as indicated in fig(6)(b).



Fig(6) : The basic elements of a PCM system.

* SAMPLING :

The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process. In order to ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than or equal to the highest frequency component 'W' of the message signal in accordance with the sampling theorem.

$$f_s \geq 2W.$$

* **Quantization :**

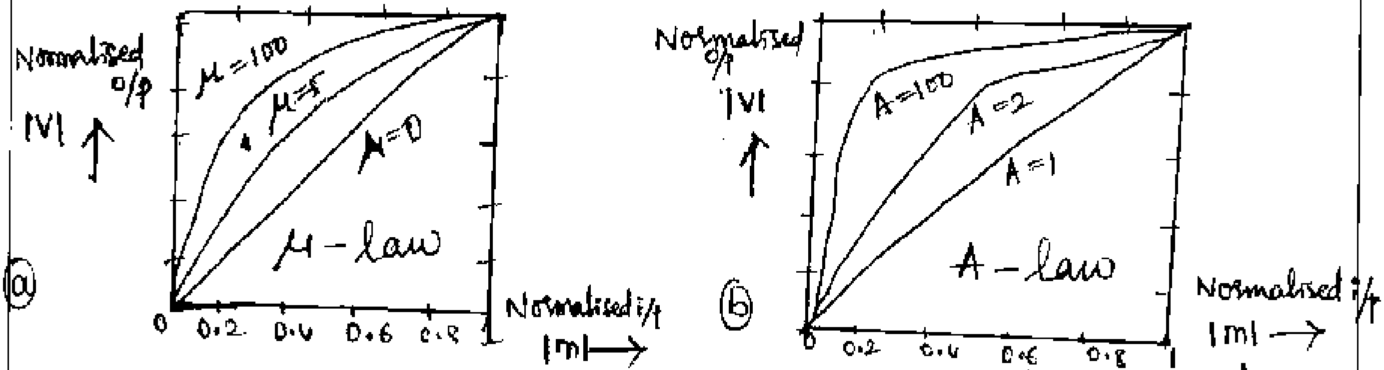
The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude.

* For uniform Quantization, we have mid-tread and mid-rise quantizer and for non-uniform quantization, we have two compression laws μ-law and A law.

* The use of a non-uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer. A particular form of compression law that is used in practice is the so called μ-law, defined by

$$|V| = \frac{\log(1+\mu|m|)}{\log(1+\mu)} \quad \text{———} (1)$$

where m and v are normalized input & output vtgs. and 'μ' is positive constant.



* Another compression law that is used in practice is the so called 'A'-law as shown above.

$$|v| = \begin{cases} \dfrac{A|m|}{1+\log A}, & 0 \le |m| \le \dfrac{1}{A} \\ \dfrac{1+\log(A|m|)}{1+\log A}, & \dfrac{1}{A} \le |m| \le 1 \end{cases}$$

* Encoding :

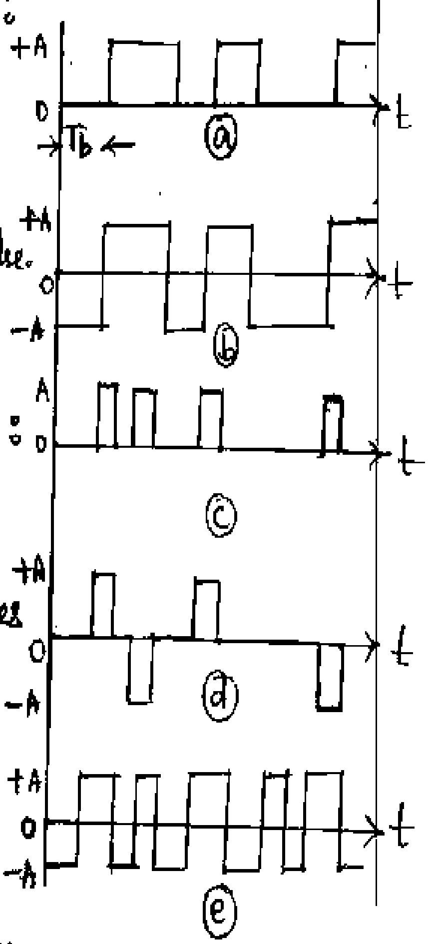
* In combining the processes of sampling and quantizing the specification of a continuous message (baseband) signal becomes limited to a discrete set of values, but not in the form best suited to transmission over a line or radio path.

* In a binary code, each symbol may be either of two distinct values or kinds, such as the presence or absence of a pulse. The two symbols of a binary code are customarily denoted as 0 and 1.

* Line code : It is a line code that a binary stream

of data takes on an electrical representation. The five line codes are illustrated in fig(7).

Binary Data: 0 1 1 0 1 0 0 1



(a) → Unipolar NRZ
(b) → Polar NRZ
(c) → Unipolar RZ
(d) → Bipolar RZ
(e) → Manchester code

1) Unipolar Nonreturn to zero (NRZ) signaling:

In this line code symbol '1' is represented by transmitting a pulse of amplitude 'A' and symbol '0' is represented by switching off the pulse.

2) Polar Non-return to zero (NRZ) signalling:

In this line code, symbol 1 and 0 are represented by transmitting pulses of amplitudes +A and −A respectively.

3) Unipolar Return to zero (RZ) signalling:

Here, symbol 1 is represented by a rectangular pulse of amplitude A and half-symbol width and symbol 0 is represented by transmitting no pulse.

4) Bipolar Return to zero (BRZ) signalling:

This line code uses three amplitude levels as shown in fig. Specifically, +ve & −ve pulses of equal amplitude are used alternatively for symbol 1. 0 for no pulse.

5) Split-phase (Manchester code)

Symbol 1 is represented a positive pulse of amplitude 'A' followed by a negative pulse of amplitude -A with both pulses being a half-symbol wide. For symbol '0', the polarities of these two pulses are reversed.

* Differential Encoding :

This method is used to encode information interms of signal transitions. In particular, a transition is used to designate symbol 0 in the incoming binary datastream, while no transition is used to designate symbol 1 as shown in fig.

ⓐ Original binary data 0 1 1 0 1 0 0 1

ⓑ Differentially encoded data "1" 0 0 0 1 1 0 1 1

ⓒ Waveform

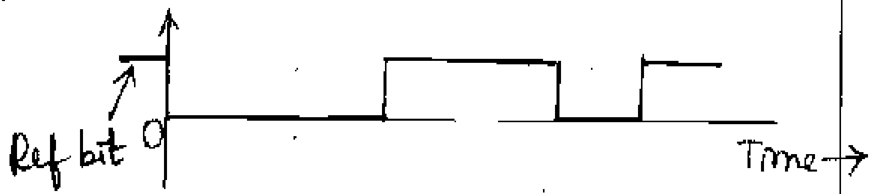


Fig : Differential encoding.

* REGENERATION :

The distorted PCM wave obtained from the transmitter is sent to the amplifier equilizer. The output of equilizer device is passed to the Decesion making device to decide the signal interms of 1 or 0 (coded o/p).

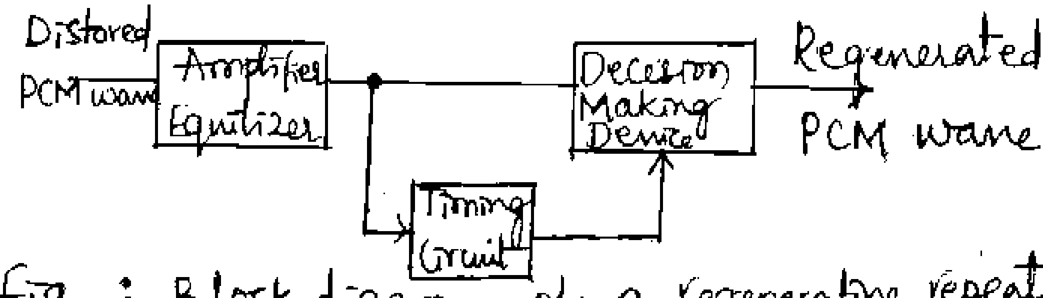


Fig : Block diagram of a regenerative repeater.

* Decoding :

The decoding process involves generating a pulse the amplitude of which is the linear sum of all the pulses in the codeword, with each pulse being weighted by its place value ($2^0, 2^1, 2^2 \ldots, 2^{R-1}$) in the code, where 'R' is the number of bits per sample.

* FILTERING :

The final operation in the receiver is to recover the message signal wave by passing the decoder output through a lowpass reconstruction filter whose cutoff frequency is equal to the message bandwidth 'W'.

* MULTIPLEXING :

In applications using PCM, it is natural to multiplex different message sources by time division whereby each source keeps its individuality throughout the journey from the transmitter to receiver.

* This individuality accounts for the comparative ease with which message sources may be dropped or reinserted in a time division multiplex system.

* APPLICATION TO VOCODERS :

* Linear Prediction coding (LPC) vocoders are model-based systems. Based on the human speech model, a

voice encoding approach can be established. Fig 8 shows analysis and synthesis of voice signals in an LPC encoder and decoder.

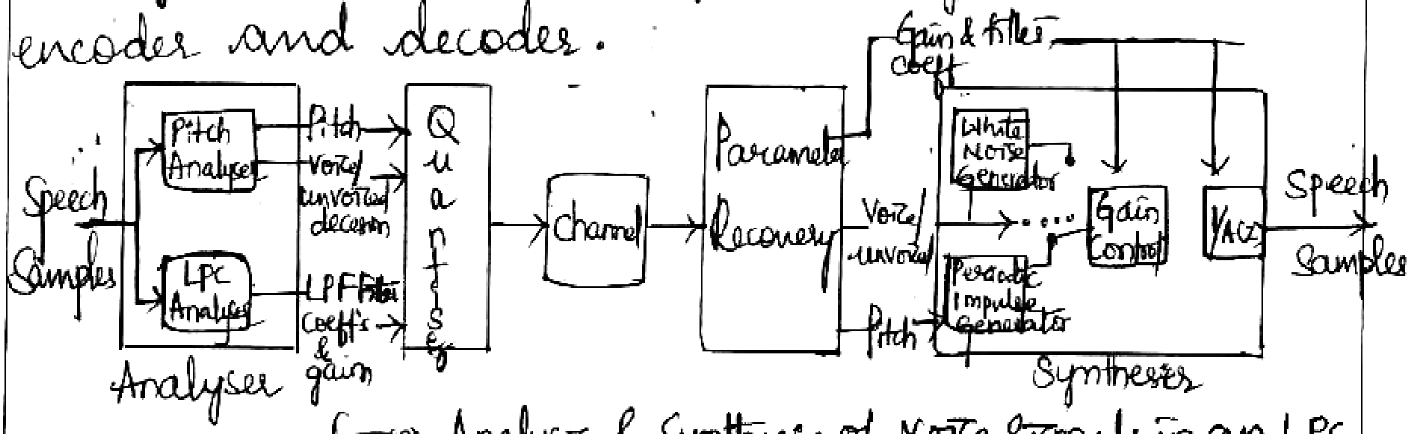


Fig 8. Analysis & Synthesis of voice signals in an LPC

* In the analysis of a sampled voice segment, the pitch analysis will first determine whether the speech is a voiced or an unvoiced piece. The LPC analyser will estimates the all-pole filter co-efficients.

* The synthesizer, which produces the speech samples at the destination followed by the Quantizer, channel, Parameter recovery and gain control.