Analog Communication Technique

[PULSE MODULATION & DIGITAL TRANSMISSION OF ANALOGY SIGNAL]

[Pulse Modulation & Digital Transmission of Analogy Signal]

$e_o(t)$ is the signal i.e. obtained after passing through Low pass filter.

vi. Let the transfer function of the low pass filter is $h(t)$.

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PULSE MODULATION & DIGITAL TRANSMISSION OF ANALOG SIGNAL :

i. We consider a basic problem associated with the transmission of a signal over a noisy communication channel.

ii. If the signal is transmitted by radio, then when signals arrives at its destination, it will be greately attenuated & also combined with noise due to thermal noise present in all the receivers.

iii. As a result, the received signal may not be separated against its background of noise.

iv. One attempt to solve this problem is simplify as to increase its signal level at the transmitting end to so high level that inspite of the attenuation, the received signal substantially overwrites the noise.

v. But such a solution is hardly possible because the signal power & consequently voltage levels beyond the range of amplifiers to generate & cables to handle it is quite difficult.

vi. An amplifier at the receiver will not help the above situation since at this point both the signal & noise levels will be increased together.

vii. But suppose that a repeater (A repeater is the term used for an amplifier in a communication channel) is located at the middle point of the long communication path.

viii. This repeater will increase the signal level, in addition it will increase the level of the noise introduced in the 1st half of the communication path.

ix. Hence, such a mid way repeater has the advantage of improving the received signal to noise ratio.

x. If we now were to transmit a digital signal over the same channel, we could find that significantly less signal power would be needed in order to obtain the same performance at the receiver.

xi. In practice, we find that SNR (signal to Noise Ratio) of 40-60' are required for analog signal while 10-12 dB are required for digital signal.

SAMPLING THEOREM:

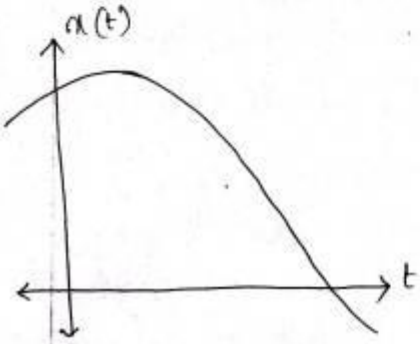
The sampling thm may be stated as follows:

A continous time signal may be completely represented in its samples & recovered back if the sampling frequency i.e., $f_s \geq 2 f_m$.

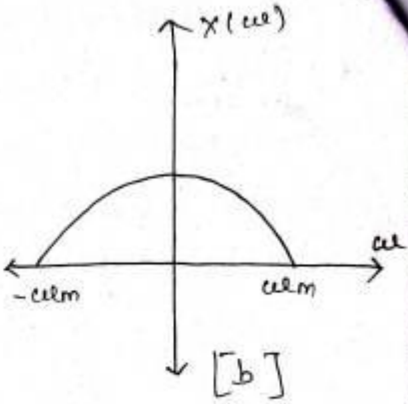
Here, f_s = sampling frequency

f_m = maximum frequency present in the signal

PROOF:



[a]



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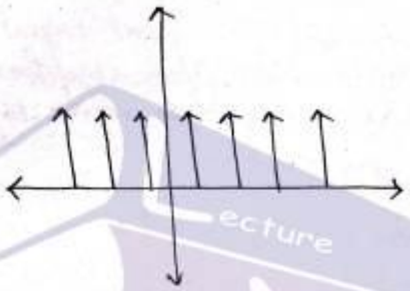

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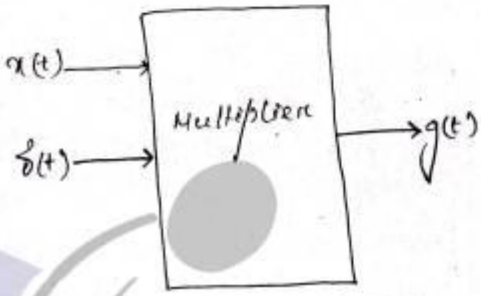


[d]

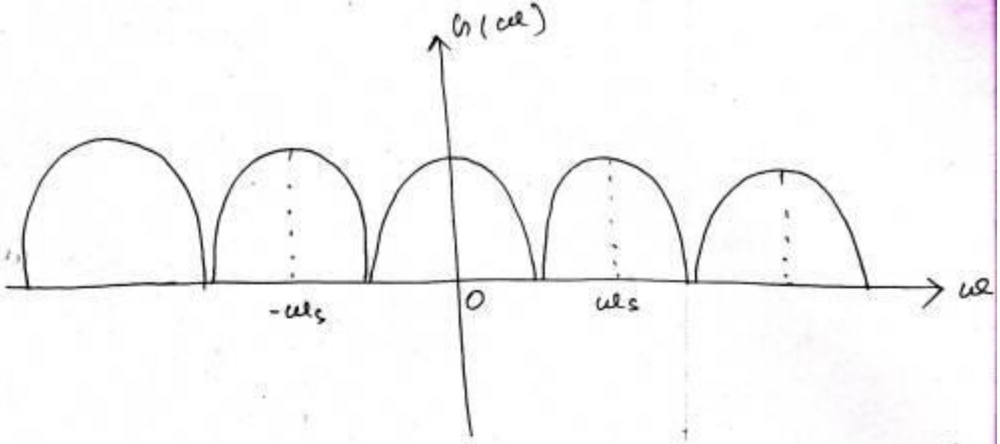
Fourier series representation of train of pulses:

$$\delta(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \cdots\right]$$

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

$$= \frac{1}{T_s}\left[x(t) + 2x(t)\cdot\cos\omega_s t + 2x(t)\cdot\cos 2\omega_s t + \cdots\right]$$

$$\Rightarrow G(\omega) = \frac{1}{T_s}\left[X(\omega) + X(\omega - \omega_s) + X(\omega + \omega_s) + X(\omega - 2\omega_s) + X(\omega + 2\omega_s) + \cdots\right]$$



The value of ωs should be $\geq 2\omega_s$ is used for non-overlapping of systems in order to avoid Inter-symbol Interference effect or ALLIASING EFFECT. This is known as NYQUIST CRITERIA.

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Q. Find out the Nyquist rate of the signal which is given as

$$x(t) = \frac{1}{2\pi} \cos(4000\pi t) \cdot \cos(1000\pi t).$$

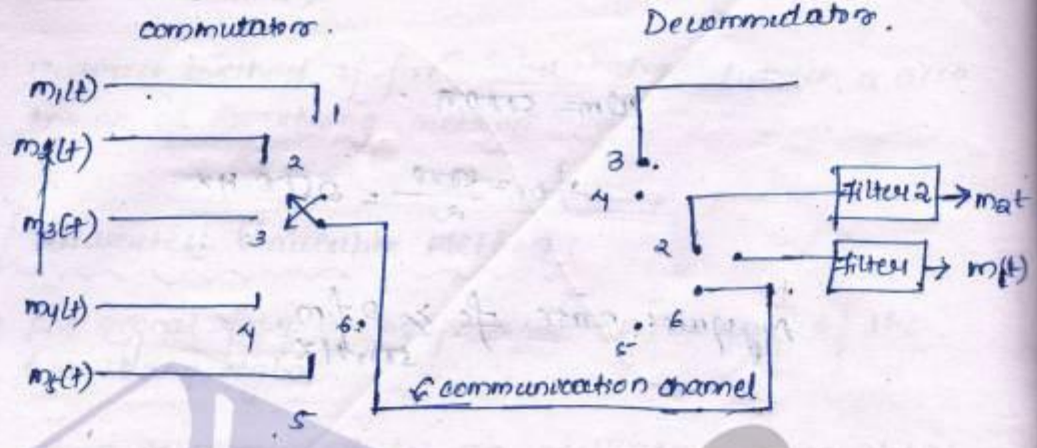
A. $$x(t) = \frac{1}{4\pi} \cdot 2\cos(4000\pi t) \cdot \cos(1000\pi t)$$

$$= \frac{1}{4\pi} \left[\cos(5000\pi t) + \cos(3000\pi t) \right]$$

$$\omega_m = 5000\pi$$

Pulse AM and concept of TDM:

Block Diagram Representation of TDM.



A technique by which we may take the advantage of sampling principle for purpose of TDM is described above.

At the transmitting end a no. of band limited signals are connected to the contact point of a Rotary switch. We assume that the signals are similarly band limited

Ex: They are all the voice signals which are limited to 3.3 KHz.

As the rotary arm of the switch swings around it samples each signal sequentially

The Rotary switch at the recieving end is in synchronism with the switch at the sending end. The 2 switches may contact simultaneously at similar no. of contacts. At the recieving end the recieved signal is passed through corresponding low pass filter at the end at the O/p the corresponding original signal is reconstructed.

In switching mechanism corresponding to the switch at the left which samples the signals is called as Commutator. The switching mechanism which performs the function of the switch in the right hand side of B.Diagram is Decommutator.

The Commutator samples and combines the sampled signal while the decommutator separates the samples belonging to the individual signals so that this signals may be reconstructed.

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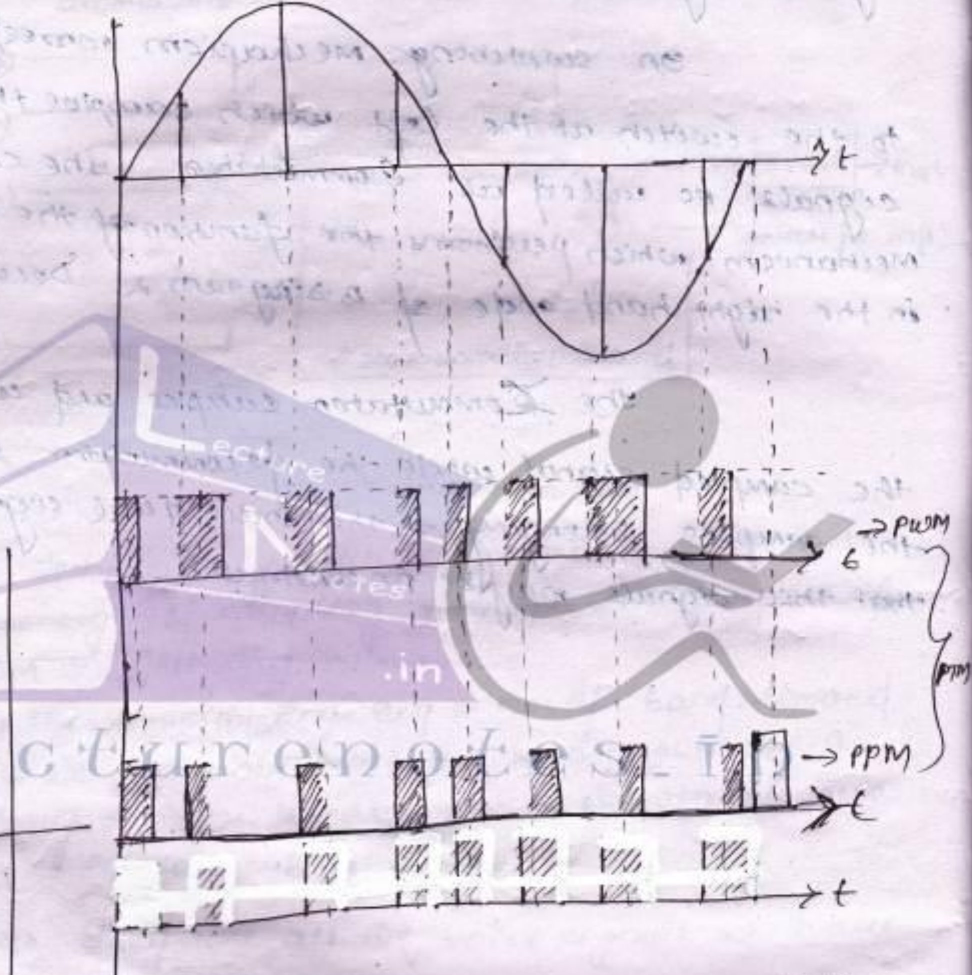
Pulse width Modulation and pulse position Modulation:



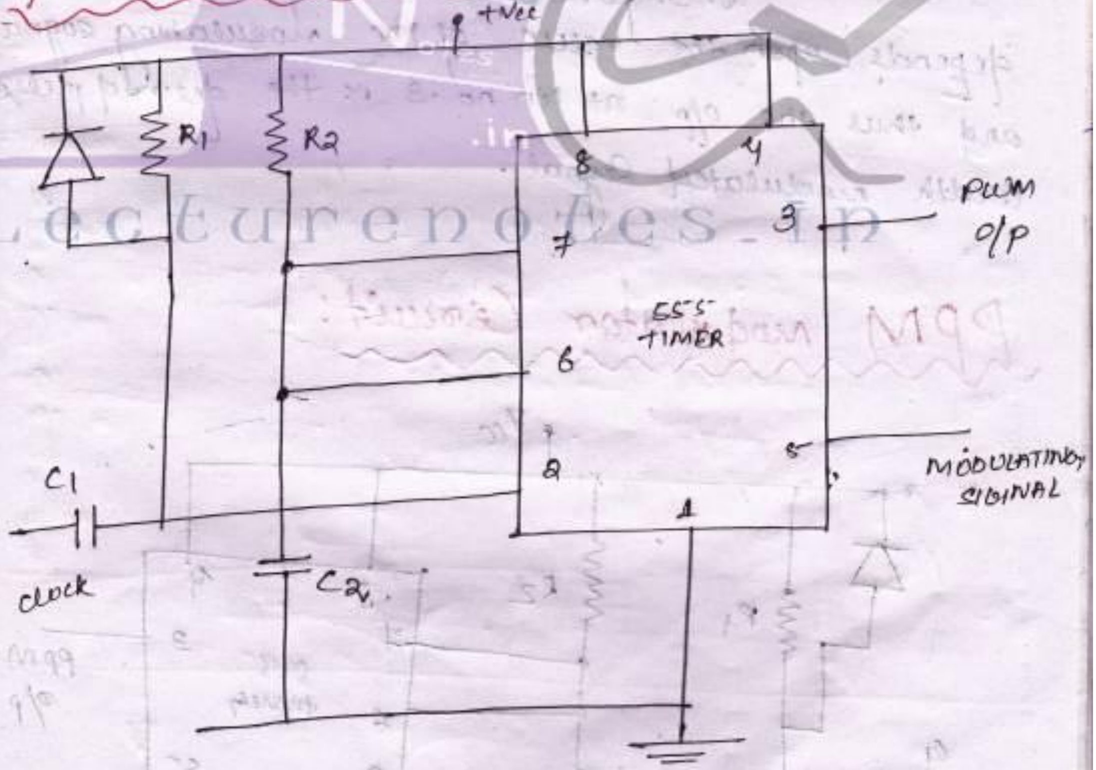
PWM · pulse location are same but their widths are change

PPM → width the same location changes.

The 2-types of PTM system namely PWM and PPM has Been Shown in the diagram. The PWM signal in which case the width of each pulse depends on the instantaneous value of Baseband signal at the sampling instance.

Similarly PPM signal is the signal where the shift in the position of each pulse depends on the instantaneous value of base band signal at the sampling instance. so this system the amplitude and width point of the pulses are kept constant while the position of each pulse is changed.

PWM Modulator circuit:



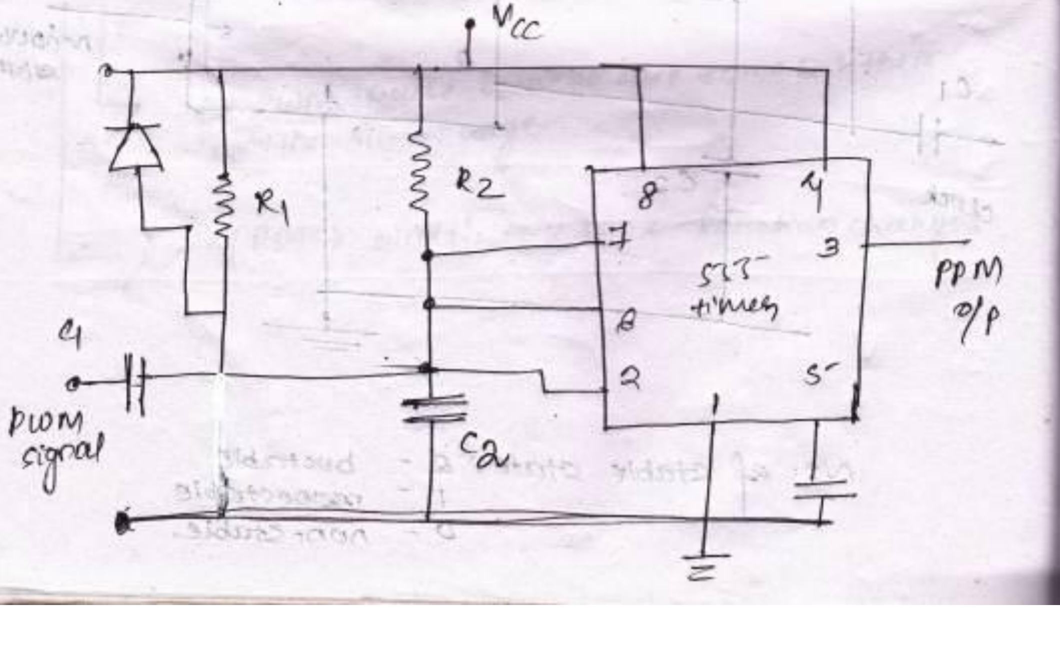
No. of stable state: 2 - bistable
1 - monostable
0 - non-stable.

In PWM modulator circuit is shown above. The clock signal of desired freq is applied as shown in the dia. from which the -ve triggered pulses are derived with the help of a diode and ~~advanced~~ C_1, R_1 and C_1 combination which works as a differentiator circuit.

These -ve triggered pulses are applied to the pin no. 2 of the 555 Timer which is working in the monostable mode. They decide the starting time of the PWM pulses. The end of the pulses depends on and R_2 and C_2 combination and from the signal which is being applied at pin no. 5 to which modulating signal is applied.

Therefore the width of the pulses depends upon the value of the modulating signal and thus the o/p at pin no. 3 is the desired pulse width modulated signal.

PPM modulator Circuit:



A PPM modulator ckt is shown above. A PWM signal is applied to pin no. 2 of 555 timer which is the -ve triggred pulses. The 555 time is working in a monostable mode and the width of the pulse is constant. The -ve triggered pulses decides the starting time of the o/p pulses and thus the o/p at pin no. 3 is the desired pulse position ~~modulating~~ modulated signal.

Digital Representation of Analog Signal:-

Steps ① → Sampling.

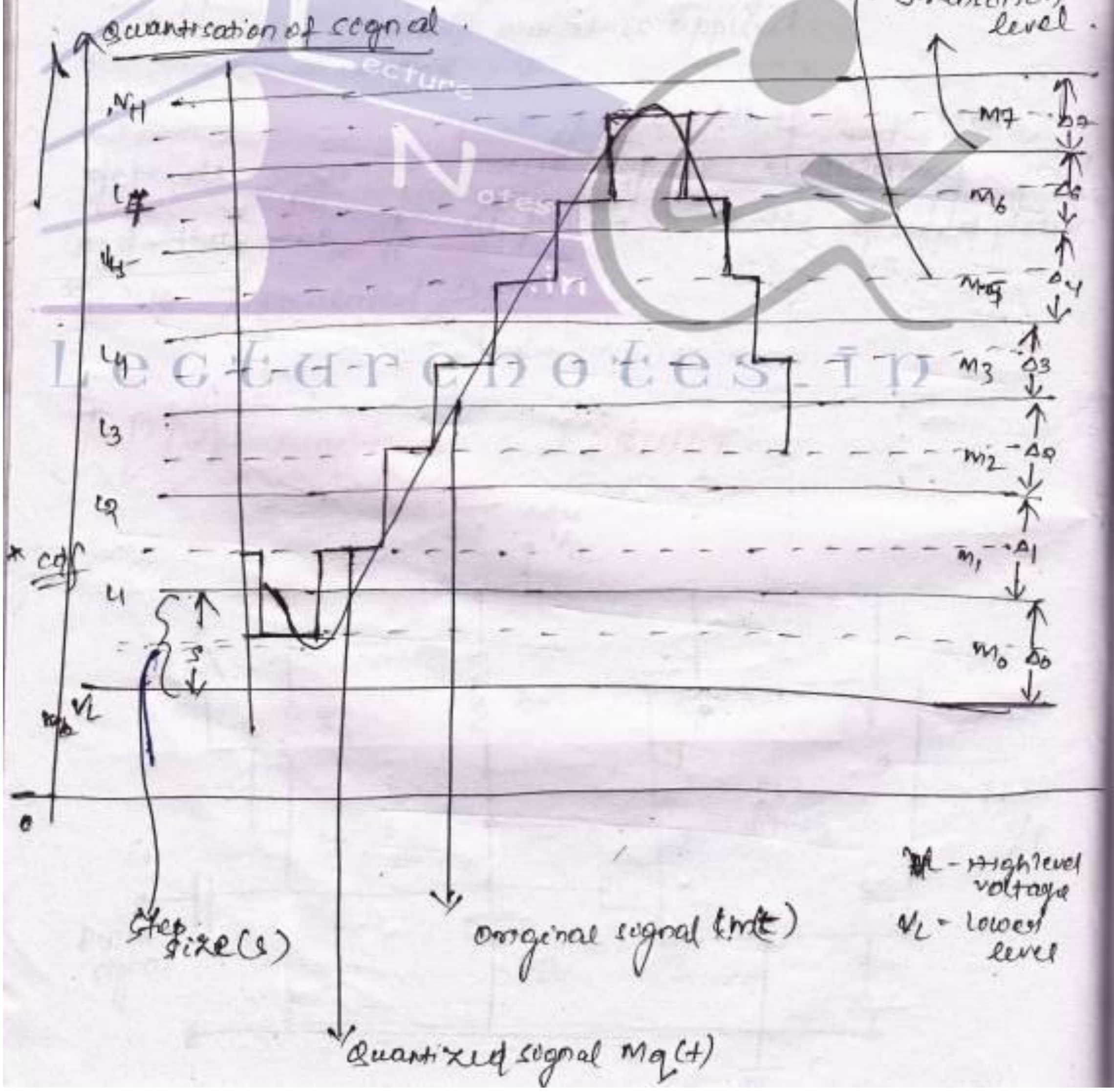
② → Quantisation

③ → Encoding.

X-axis - Time domain axis

Y-axis - Amplitude.

Quantisation of signals:



(i) In order to make digital representation of analog signals. The first step is ~~sign~~ sampling the time axis then ~~the~~

(ii) discretize the amplitude axis and then represent that signal in terms of binary digits to complete the process.

We now describe how the signal is subjected to the operation of Quantization. When quantizing a signal ie $m(t)$, we create a new signal $M_q(t)$ which is approximation to the original signal $m(t)$. However the quantized signal has the greater merit is that it is separable from the additive noise.

The operation of the quantization is described above. ~~where~~

We are taking a signal whose peak to peak is confined in the range from V_L to V_H. We have divided this total range in equal intervals each of size 's' and 's' is step size. And these levels are known as transition levels.

Let 'M' no. of transition levels. Then how to calculate 's' size?

$M \rightarrow$ No. of Transition level.

Imp

$$\boxed{\text{Step size} = \frac{V_H - V_L}{M}}$$

In the centre part of each of the steps we locate the quantisation levels i.e $m_0, m_1 \ldots m_7$

The Quantized signal is generated in the following way:-

whenever $m(t)$ is the range of 'Δ_0', this signal (Quantized, $mq(t)$) maintains the constant level which is 'M_0'.

whenever $m(t)$ is in the range of 'Δ_1', at that time the $Mq(t)$, quantized signal maintains constant level M_1 and so on..

And thus the signal $Mq(t)$ is generated in this way which is present in staircase form.

From this concept at every instant of time a quantization error is generated which is given as.

$$= M(t) - Mq(t)$$

$$\boxed{\text{value of Quantization Error} = \frac{S^2}{12}}$$

S = step size.

Quantization Error:

It has been pointed out that a quantised signal and the original signal from which it was derived that they differ from one another.

This difference of error was to be viewed as a noise due to quantisation process and it is called as quantisation error.

We now calculate the mean square quantization error which is denoted by the symbol $\overline{e^2}$

e = diff. b/w the original signal and the quantised signal voltages.

Let us divide the total peak to peak range of the message signal m(t) into 'M' no. of equal voltage intervals.

$m_1, m_2 \ldots m_m$

According to the previous fig of quantised signal

$$e = m(t) - m_q(t)$$
$$= m(t) - m_k$$

$$k = 1, 3, \ldots 4$$

Let $f(m)dm$ where m= message signal be the probability that the ~~bas~~ message signal m(t) lies in the range from highest voltage level to lowest voltage level. then the meansquare Quantization error is given as.

$$\overline{e^2} = \int_{-S/2}^{S/2} f(m)(m-m_1)^2 dm + \int_{-S/2}^{+S/2} f(m)(m-m_2)^2 dm +$$

$$\ldots\ldots + \int_{-S/2}^{S/2} f(m)(m-m_k)^2 dm \quad \text{(i)}$$

If we take more and more no. of quantization level then step size value (S) is small in comparision to peak to peak range of the message signal. In this case it is reasonable to make the approximation that f(m) is constant within each quantization range. then in Eq(i) the first term $f(m)$

$$f(m) = f^1$$

2nd term $f(m) = f^2$

where 'k' lies in range from.

$$x = (m - m_k)$$

$$\Rightarrow dx = dm$$

m= message signal (variable)

m_k = Quantization level (constant)

$$\boxed{f_s^1 + f_s^2 + f_s^3 + \dots = 1}$$

$$\boxed{\overline{e^2} = (f^1 + f^2 + \dots)\int_{-s/2}^{+s/2} x^2\,dx} \quad (2)$$

$$\Rightarrow \underbrace{(f^1 + f^2 + \dots)}_{1}\frac{s^3}{12}$$

$$\Rightarrow (f_s^1 + f_s^2 + \dots)\frac{s^2}{12}$$

$$\Rightarrow \boxed{\overline{e^2} = \frac{s^2}{12}}$$

Q. 6 message signals of bandwidth 5kHz are time division multiplexed and transmitted. Determine the signaling rate and the minimum channel bandwidth of PAM/TDM channel.

Soln:

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

$$\boxed{MCB = N f_m}$$

S.Rate = $N f_m$.

$= 6 \times 5$

$= 30$ kHz.

M.C.B = $N f_c$.

$f_c = 2 f_m$

$= 2 \times 5$

$= 10$.

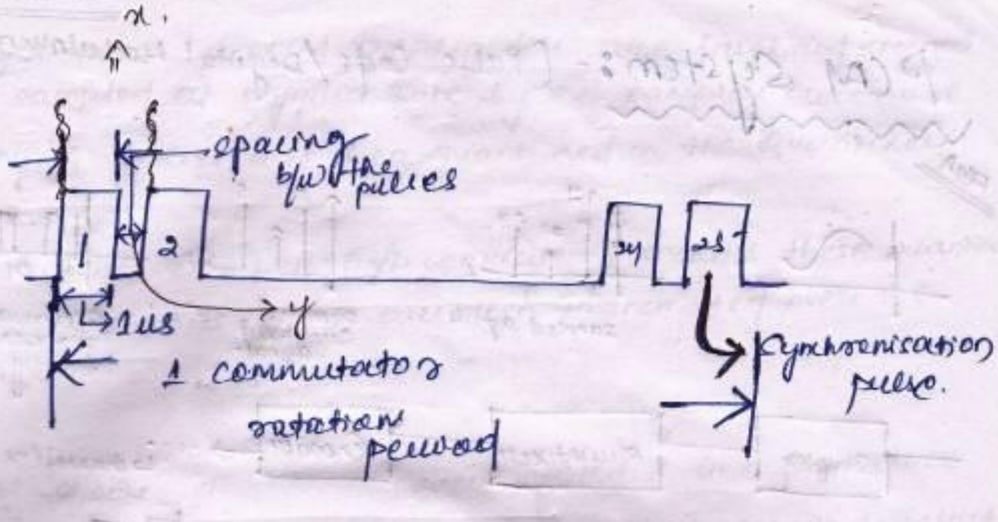
MCB = 6×10.

$= 60$ kHz

$= 60$ kbits/sec

9) 24 voice signals are sampled uniformly and then TD multiplexed. The sampling operation uses flat top samples with 1 μs duration. The multiplexing operation includes the provision for synchronization by adding an extra pulse of appropriate amplitude and 1 μs duration. The highest freq component of each voice signal is 3.4 kHz.

(i) Assuming a sampling rate of 8 kHz calculate the spacing b/w the successive pulses of the multiplexed signal.



leading of of 1 signal to the leading of 2 signal is the 1 complete rotation of the commutator.

1 commutator rotation period $= \frac{1}{f_s}$

$= 125 \mu sec.$

$x = \frac{125 us.}{25} = 5 \mu sec.$

$y = 5 - 1 = 4 \mu sec.$

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PCM System:- (Pulse Code/Digital Modulation System)

Fig A

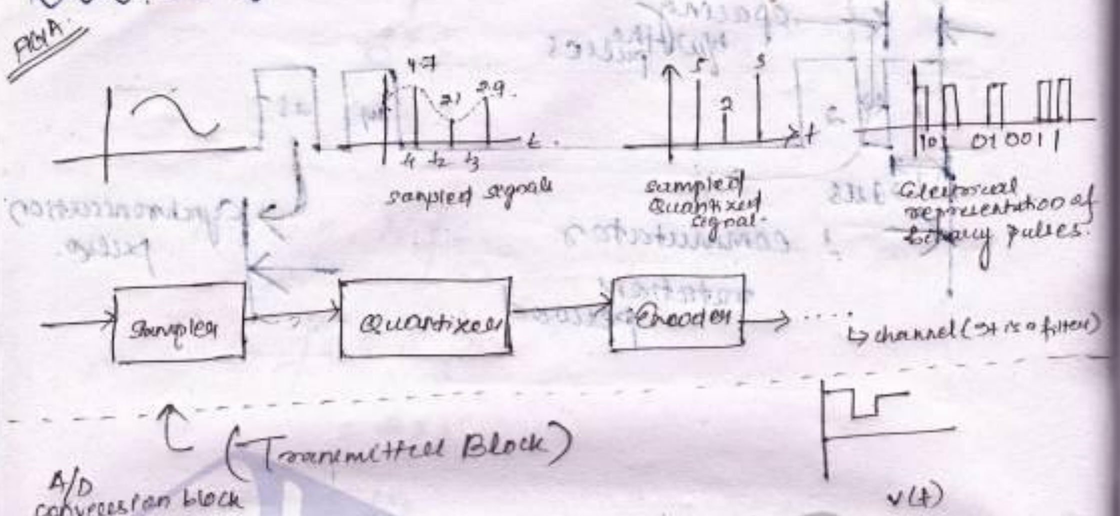
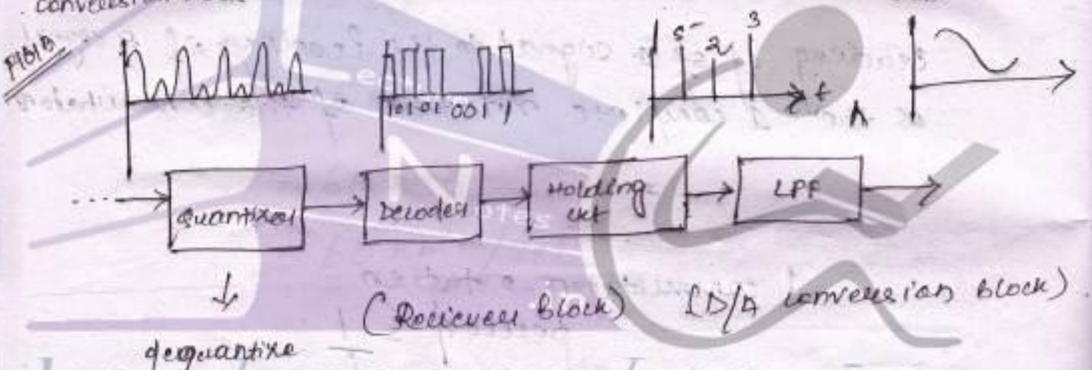


Fig B



Sampler ⇒ It is just the discretization in amplitude in time axis.

Quantizer ⇒ discretization in amplitude axis.

FIG A shows a PCM transmitter. The baseband signal is sampled at Nyquist Rate by the sampler block. The Sampled pulses are then quantized in the Quantizer block.

The Encoder (A/D convertor) encodes these quantized pulses into bits which are then transmitted over the channel.

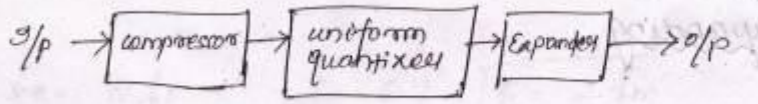
FIG B: Shows a PCM receiver system. The first block is again a Quantizer but this quantizer is different from the transmitter's Quantizer block because it has to take a decision about the presence and absence of a pulse.

The o/p of the Quantizer block goes to the decoder block which performs the inverse operation of the Encoder block. The decoder o/p is a sequence of Quantized pulses. The original baseband signal is reconstructed in the holding circuit and LPF circuit.

MODULE - 3

Companding:

It is used to improve SNR



Block diagram of Companding technique.

Companding is non-uniform quantization. It is required to be implemented inorder to improve the signal power to quantization noise power ratio of weak signals. We know that quantization noise power is given as '$S^2/12$'. This shows that '$S^2/12$' in uniform quantization, once the step size is fixed the Quantization noise power remaining constant. However the signal power is not constant. It is proportional to the square of the signal amplitude. Hence the signal power will be small for weak signals but Quantization noise power is constant.

Therefore signal to quantization noise ratio for weak signals is very poor. This will affect the quality of the signal. The remaining is to use the Companding technique.

The companding is a term derived from 2 words i.e compression and Expansion. The strong signals are attenuated and weak signals are amplified before applying to uniform Quantizer. This process is called Compression technique. and the block diagram that defines it is called Compression shown on diagram above.

At the receiver exactly the opposite is followed which is called as Expansion. The circuit used for Expansion process is called as expander.

The compression of a signal at the transmitter and expansion at receiver side is combined to be called as companding.

* A television signal having a bandwidth of 4.2 MHz transmitted using binary PCM given that the no. of Quantization levels are 512. Determine

(a) Code word length
(b) Transmission Bandwidth
(c) Final bit rate
(d) Output signal to Quantization noise Ratio

Ans: $(SNR)_{dB} = \begin{cases} (6v + 4.8)\,dB & \text{(i)} \\ (6v + 1.8)\,dB & \text{(ii)} \end{cases}$

for only sinusoidal signal.

v = no. of bits in code word.

Bandwidth = 4.2 MHz.
Quantization levels = 512.

(i) $2^{v} = 512$,
v = no. of bits in code word.

$v = 9$ bits

Transmission Bandwidth:

$$T.BW = n f_m$$
$$= 9 \times 4.2$$
$$= 37.8 \text{ MHz}$$

(iii) $S.R = N f_s$ ∴ $f_s = 8.4$. $\boxed{f_s = 2 f_m}$

$$= 9 \times 8.4$$
$$= 75.6$$

(iv) $(SNR)_{dB} = (6 \times 9 + 4.8)\,dB$

$$= 58.8\,dB$$