Lecturer's Note

Analog and Digital Communication (ADC)

MODULE - III

Semester : 5th (ETC)

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Module - III

PULSE MODULATION SYSTEM :

→ Here the carrier wave comes in a pulse type i.e it consists of a pulsetrain.

Pulse Modulation

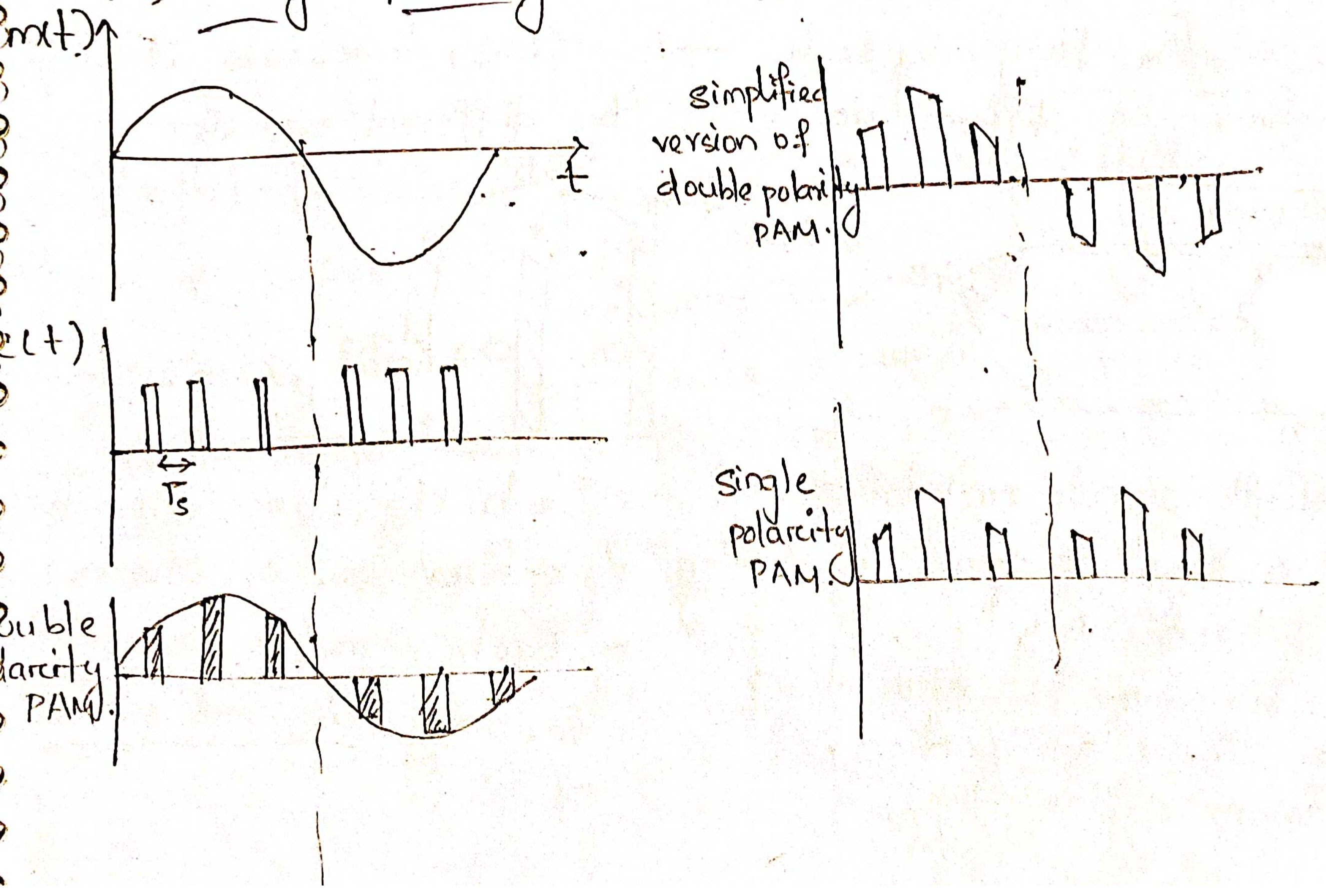
- Pulse Amplitude Modulation (PAM)
- Pulse-Time Modulation (PTM)
  - PWM → Pulse width modulation
    or
    PDM → Pulse duration modulation
    or
    PLM → Pulse length modulation
  - Pulse position modulation (PPM)

PAM :

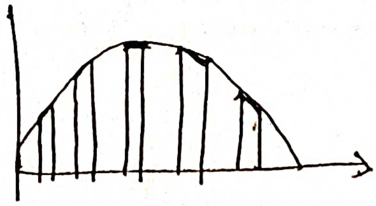
PAM is of 2 types.

(i) Double polarity PAM (Bi-polar PAM)

(ii) Single polarity PAM (Unipolar PAM)



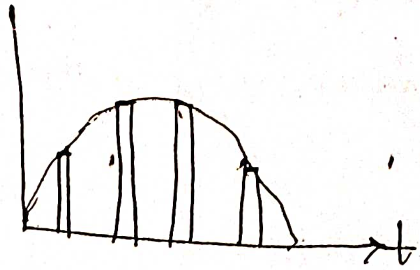
Natural Sampling



→ Here the tops arenot flat and follow the natural waveform of modulating signal.

→ Complicated electronic ckts are needed to maintain natural sampling.

Flat-top sampling



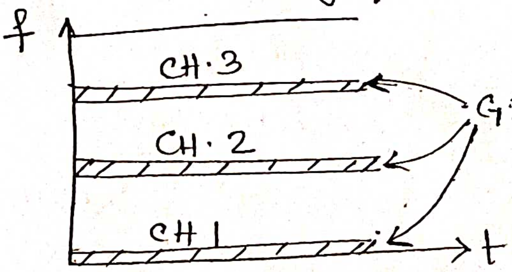
→ Here the tops of the samples are flat.

→ complicacy is reduced by flat-top sampling. So it simplifies the design of ckt.

→ But in each case sampling rate should be high to avoid distortion and for easy reconstruction of the original signal. So it follows the sampling theorem.

FDM

→ In FDM, the frequency scale is shared by different signals.

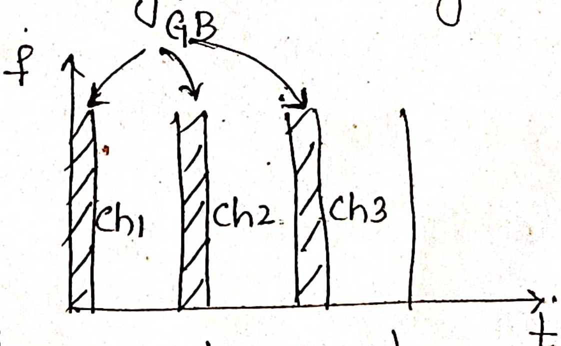


→ Multiple signals are transmitted over a single channel by sharing the channel B.W.

→ This is done by allocating each signal a portion of spectrum within that B.W.

TDM

→ In TDM, time scale is shared by different signals.



→ Each signal occupies the entire B.W of channel.

→ Each signal is transmitted for only limited period of time.

FDM	TDM
→ Ckt arrangement of FDM is complex.	→ For TDM it is simple.
→ Non-linearity & distortion is more.	→ I-I is very less.
→ FDM is used in analog communication system.	→ TDM is generally used in digital communication system.

B.W of PAM :—

→ 'n' no. of available signals each signal bandlimited to f_m Hz.

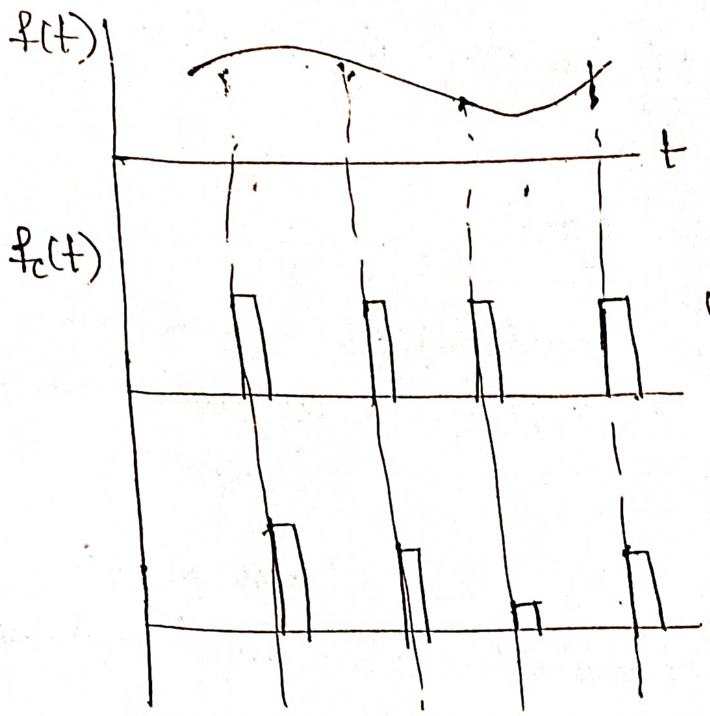
→ For each signal, we take $2f_m$ samples/sec.

" n signals, = $2nf_m$ samples/sec.

So, $\boxed{BW_{PAM} = n f_m \text{ Hz}}$

PAM :	PWM	PPM
→ Amplitude Varies.	→ Width of the pulse varies.	→ Position of the pulse varies.
→ B.W is less.	→ large B.W	→ Large B.W
→ Noise is more.	→ Noise is less.	→ Noise is less.
→ It is widely used having large application.	→ It is less efficient.	→ It is more effective.

Pulse Amplitude Modulation –



(a) (Baseband signal f(t))

(b) (Carrier pulse train $f_c(t)$)

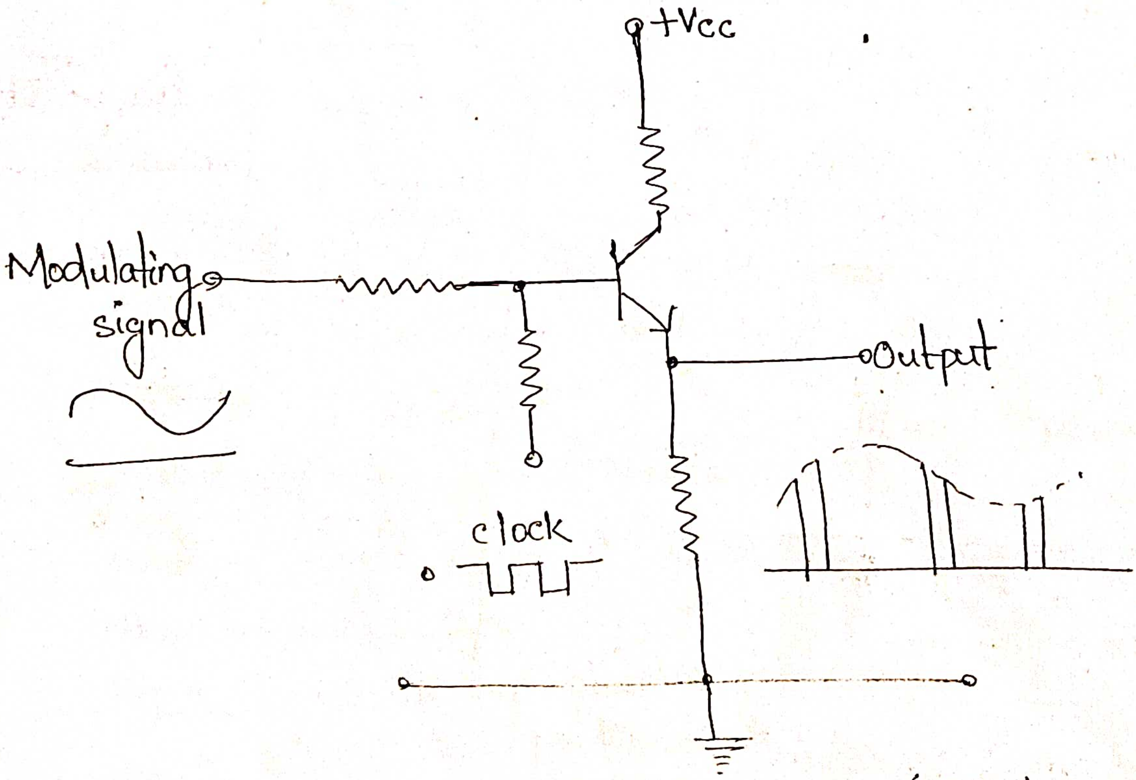
(c) (PAM signal)

→ Two methods of getting the pulse amplitude modulated waveform.

(1) Natural sampling or shaped to sampling

(2) Flat top sampling.

PAM Modulator Circuit :–



(PAM modulator)

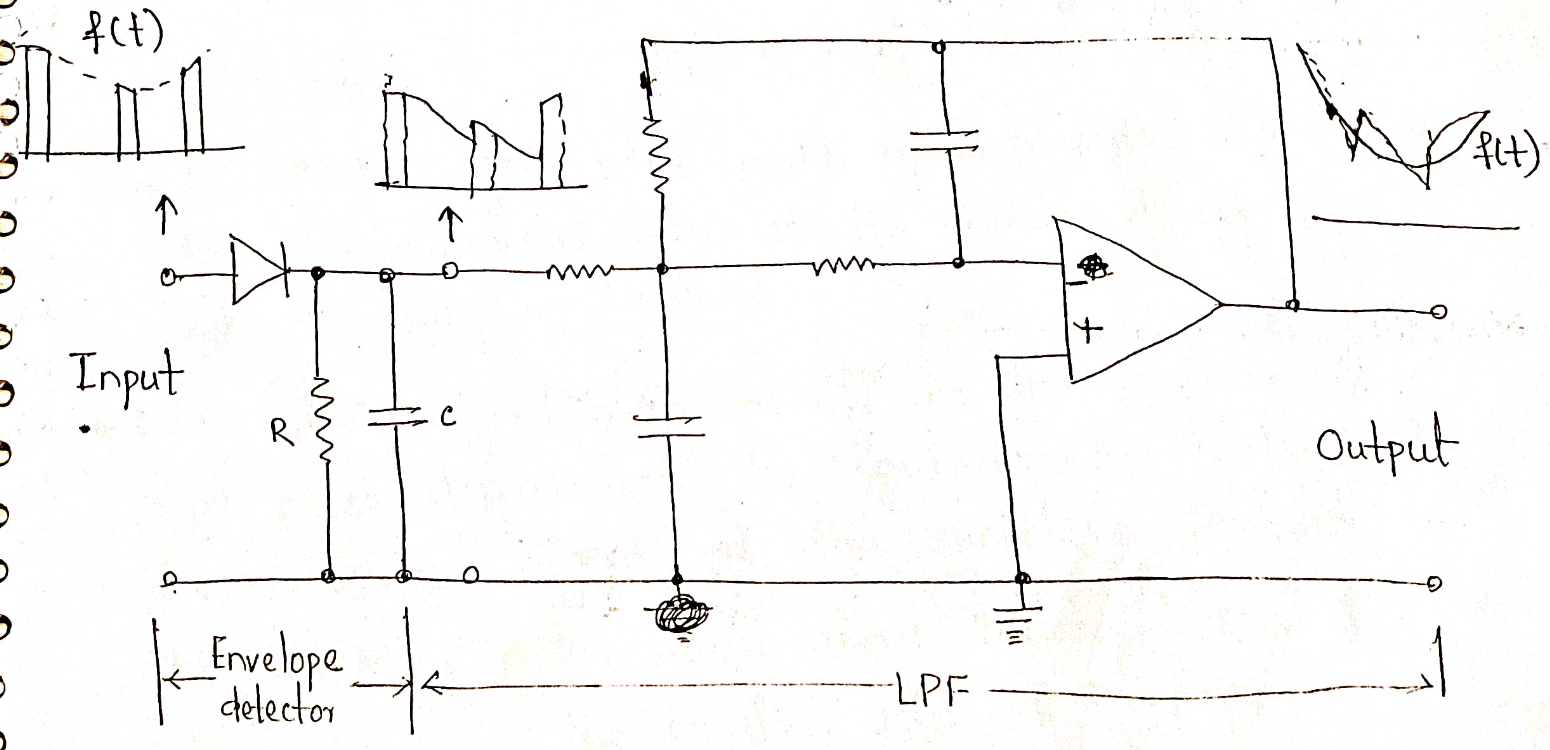
→ The given is a simple emitter follower ckt.
i.e in the absence of the clock signal, the output follows the input.

→ Here, I/P = 'modulating' signal

I/p to the transistor base = clock signal
desired carrier pulse train frequency.

→ When I/p clock is high, the ckt behaves as an emitter follower and the O/p follows the input modulating signal.
When clock I/p = Low, the transistor is cut-off and the O/p = 0 (zero).

→ PAM Demodulator Circuit :—



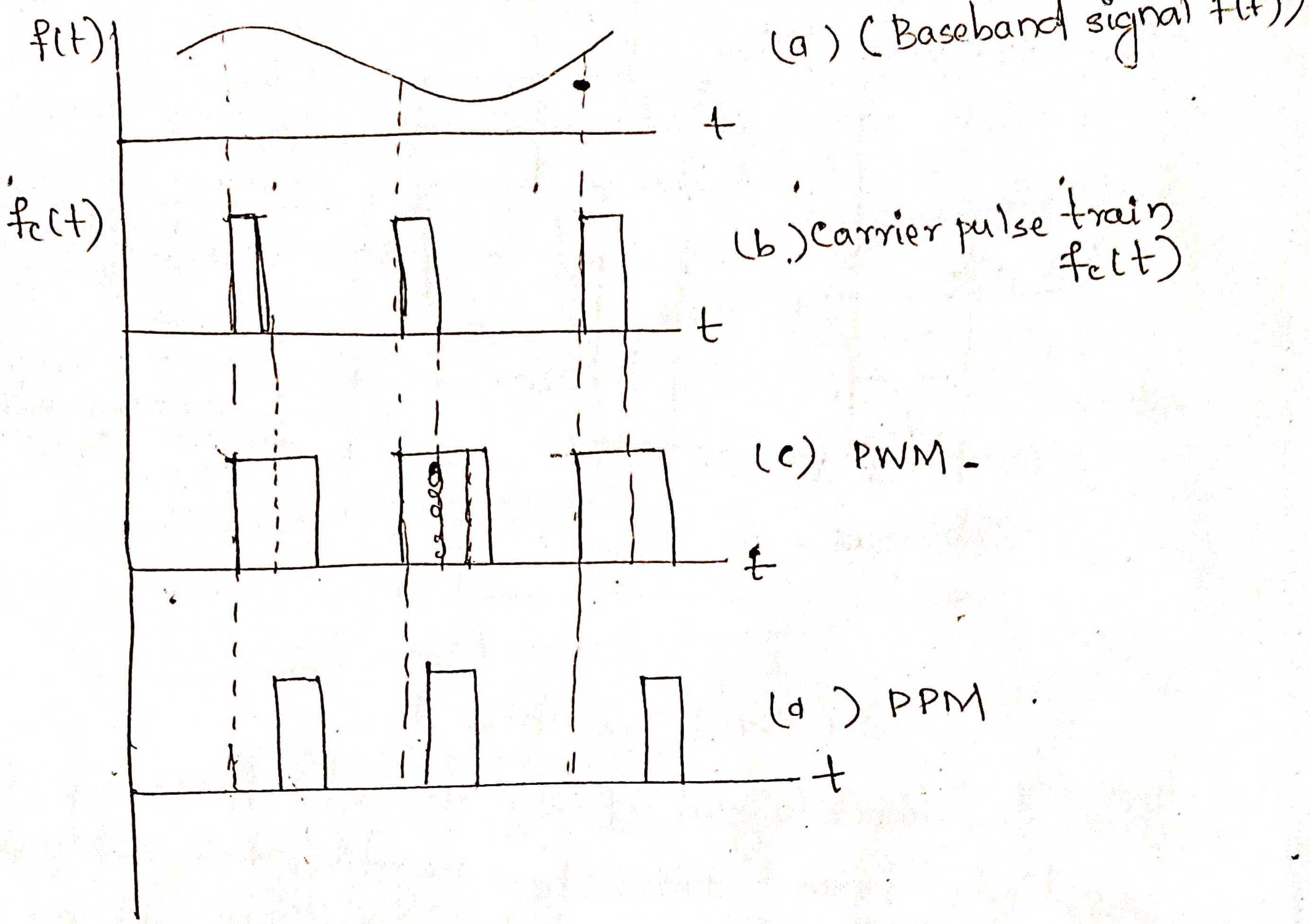
(PAM Demodulator)

→ It operates like an envelope detector followed by a low-pass filter.

→ The diode and R-C combination work as the envelope detector. 'It is followed by a 'second order OP-AMP low pass filter to have a good filtering characteristic.

→ So, for the received pulse amplitude modulated signal as the i/p signal, the baseband signal f(t) is produced as the o/p.

Pulse Time Modulation :—



→ PTM system is of two types.

1. PWM (Pulse width modulation)
2. PPM (Pulse position modulation)

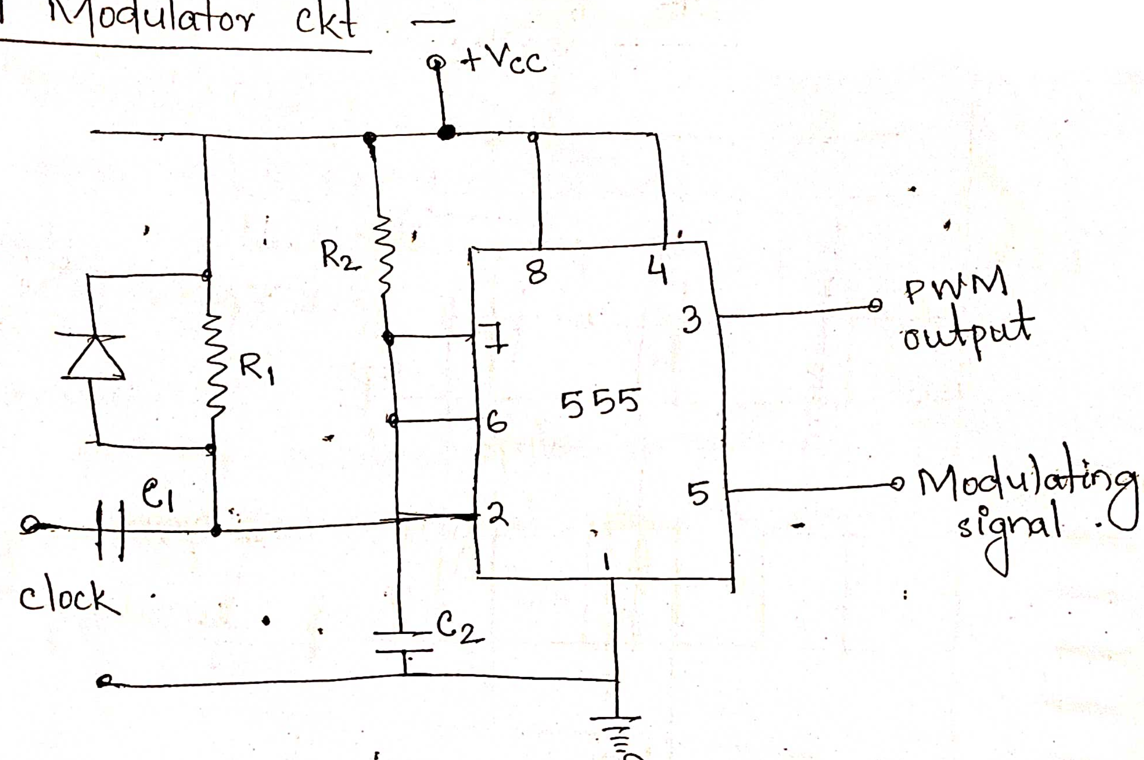
→ In PWM signal, the width of each pulse depends on the instantaneous value of the baseband signal at the sampling instant.

→ In PPM signal, the shift in the position of each pulse depends on the instantaneous value of the baseband signal at the sampling instant.

→ In PWM, information about baseband lies in the trailing edge of the pulse.

PPM, it lies in both the edges of the pulse.

PWM Modulator ckt —



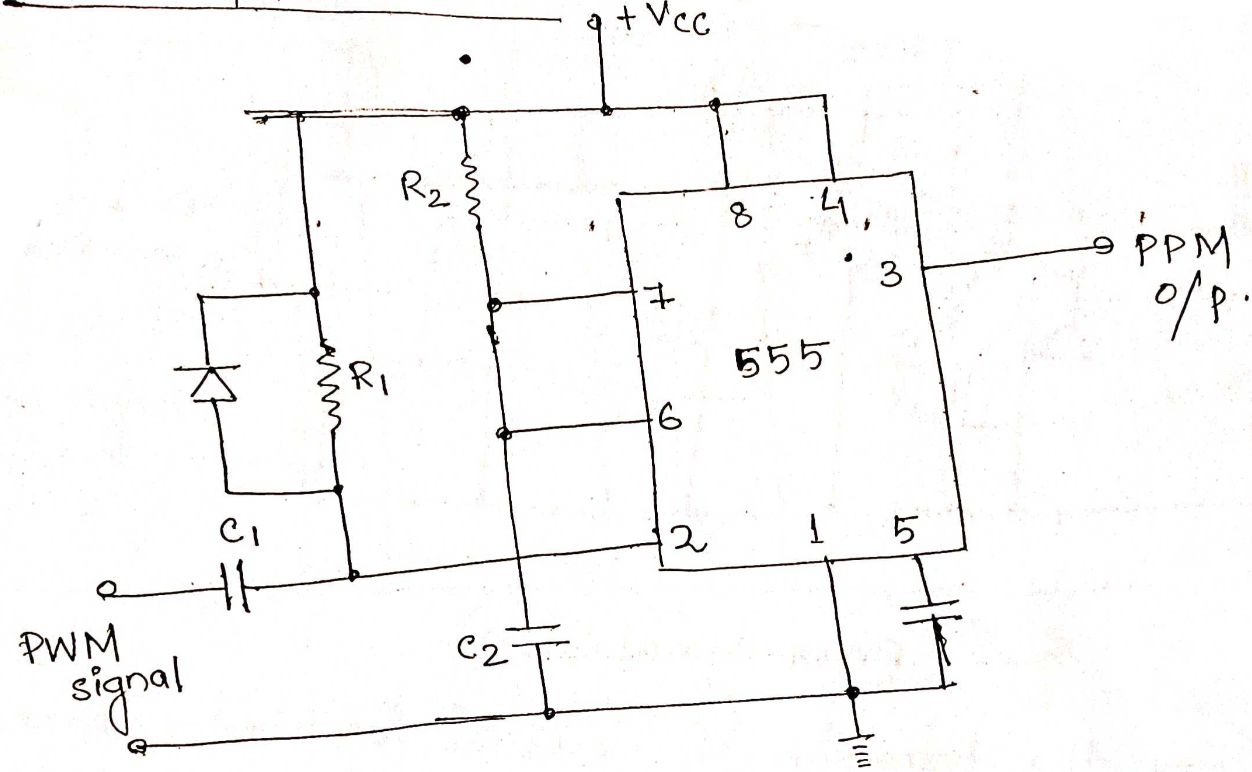
(PWM modulator)

→ Here the clock signal of the desired frequency is applied, from which the negative trigger pulses are derived with the help of a diode and an $R_1 - C_1$ combination which works as a differentiator.

→ The negative trigger pulses are applied to the pin no 2 of 555 timer which is working in monostable mode. They decide the starting time of PWM pulses.

→ The end of the pulses depends on an $R_2 - C_2$ combination and on the signal at pin no. 5 to which the modulating signal is applied. So 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 ckt :—



(PPM Modulator)

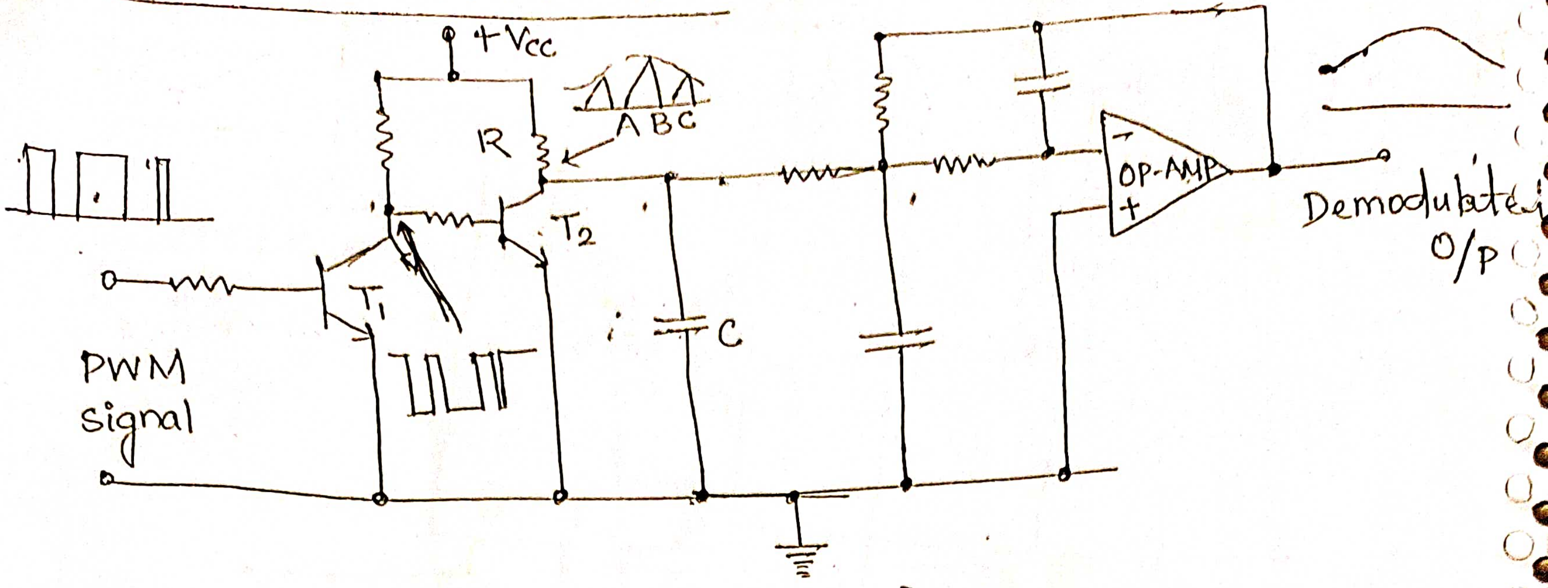
→ The PWM signal is applied to pin no. 2 through the diode and $R_1 - C_1$ combination.

→ Thus the I/p to pin no. 2 is the negative trigger pulses which correspond to the trailing edges of the PWM waveform.

→ The 555 timer is working in a monostable mode and the width of the pulse is constant.

→ The negative trigger pulses decide the starting time of the O/P pulse and thus the O/p at pin no. 3 is the desired PPM signal.

PWM Demodulator ckt :—



(PWM Demodulator)

→ Here the transistor T_1 works as an inverter. Hence during the time interval A - B, when the PWM signal is high, the input to the transistor T_2 is low.

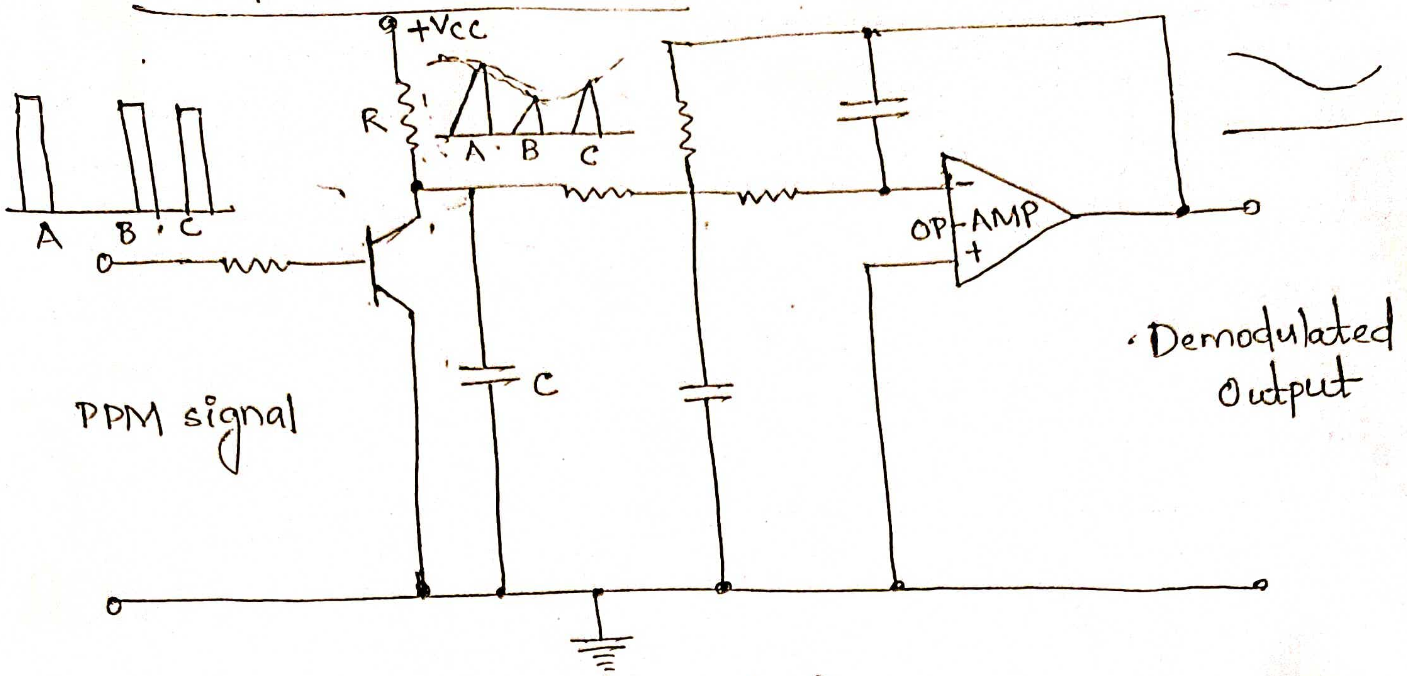
→ So during this time interval, the transistor T_2 is cut-off and the capacitor c gets charged through an R-C combination.

→ During the time interval B-C when the PWM signal is low, the I/p to the transistor T_2 is high and it gets saturated.

→ The capacitor c then discharges very rapidly through T_2. The collector voltage of T_2 during the interval B-C is then low.

→ Thus the waveform at the collector of T_2 is more or less a saw-tooth waveform whose envelope is the modulating signal. When this is passed through a second-order OP-AMP low pass filter, we get the desired demodulated O/P.

PPM Demodulator Ckt :—



(PPM Demodulator)

→ It utilizes the fact that the gaps between the pulses of a PPM signal contain the information regarding the modulating signal.

→ During the gap A - B between the pulses, the transmitter is cut-off and the capacitor C gets charged through the R-C combination.

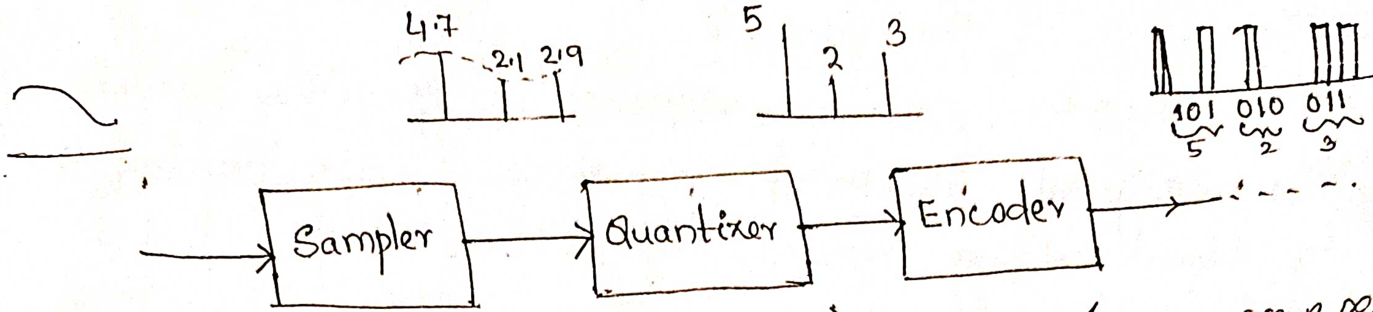
→ During the pulse duration B-C the capacitor discharges through the transistor and the collector voltage becomes low.

→ Thus the waveform at the collector is approximately a saw-tooth waveform whose envelope is the modulating signal.

→ When this is passed through a second order OP-AMP low pass filter, we get the desired demodulated o/p.

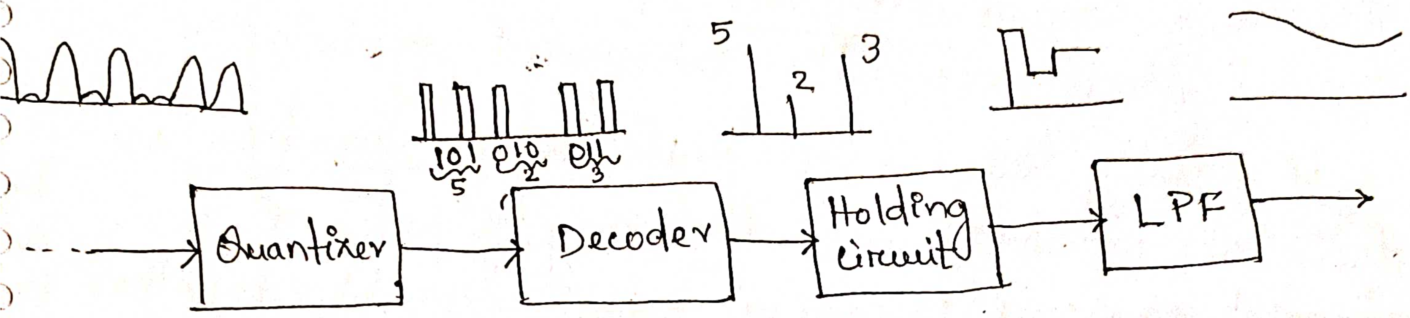
(7)

PCM system : — Pulse Code Modulation



(a) (Transmitter)

is a lossy compression technique.



(b) (Receiver)

[Block diagram of PCM system]

→ Transmitter :—

→ Here the baseband signal is sampled at Nyquist rate by the sampler. The sampled pulses are then quantized in the quantizer.

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

→ Receiver :—

→ Here the quantizer is different from the transmitter quantizer because it has to take a decision about the presence or absence of a pulse.

→ Thus there are only two quantization levels. The o/p of the quantizer goes to the decoder which is a

D/A converter that performs the inverse operation of the encoder.

→ The decoder o/p is a sequence of quantized pulses. The original baseband signal is reconstructed in the holding circuit and LPF.

→ Intersymbol Interference (ISI) :—

→ ISI is a form of distortion of signal in which one symbol interferes with subsequent symbols.

→ It is usually caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to blur together.

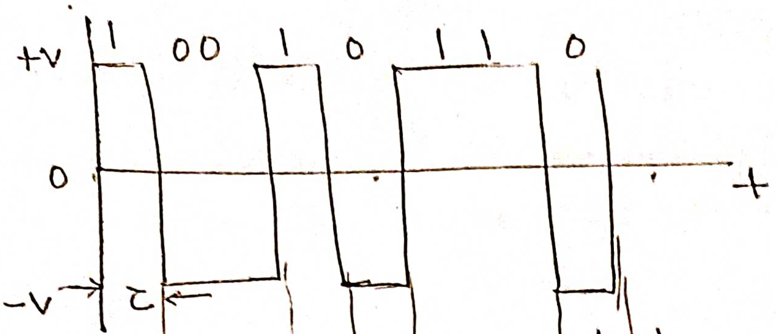
→ This is an unwanted phenomenon as the previous symbols have similar effect as the presence of noise makes the communication less reliable.

→ The presence of ISI in the system introduces errors in the decision device at the receiver o/p.

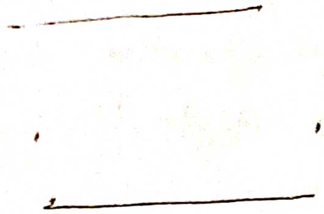
→ Therefore in the design of the transmitting and receiving filters, the objective is to minimize the effect of ISI and there by deliver the digital data to its destination with smallest possible error rate.

→ ISI can be minimised by using adaptive equalisation and errorcorrecting codes.

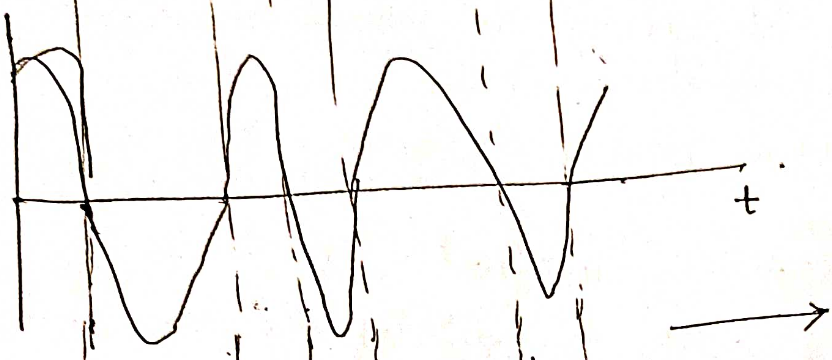
EYE PATTERNS :—



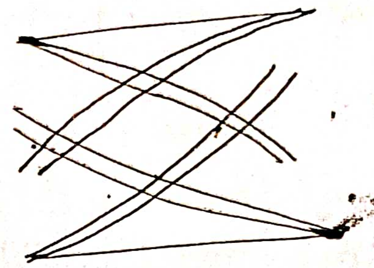
(Received ideal bit stream)



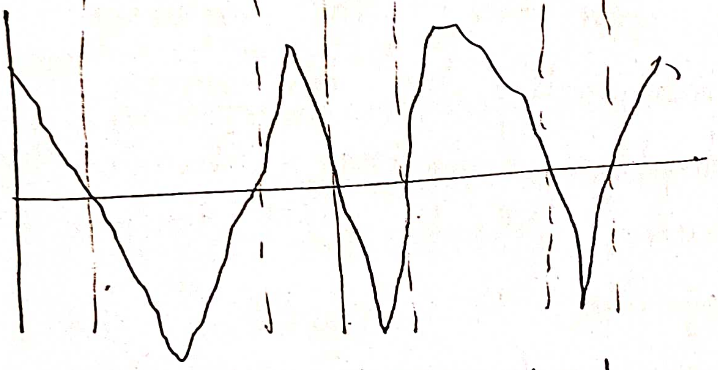
(CRO Pattern)



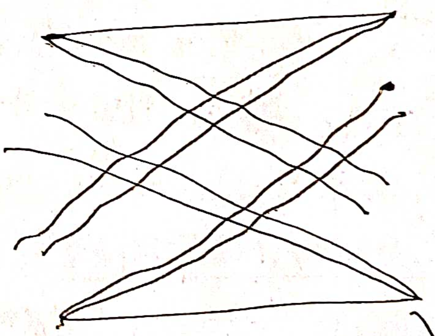
(Less distortion Received Bit stream)



(CRO Pattern)
(similar to eye)



(More distorted Received Bit stream).



(CRO Pattern)
(here eye is further closed)

→ Both ISI and noise cause errors to occur.

A CRO can be used to give an indication of the performance of a PCM system.

→ Here, the received bit stream is applied to the vertical deflection plates; and the time base frequency is made equal to the bit rate so that a sweep lasts one time slot duration.

→ From the waveform it is seen that, the more is the opening of the eye, the less is the distortion and vice versa.

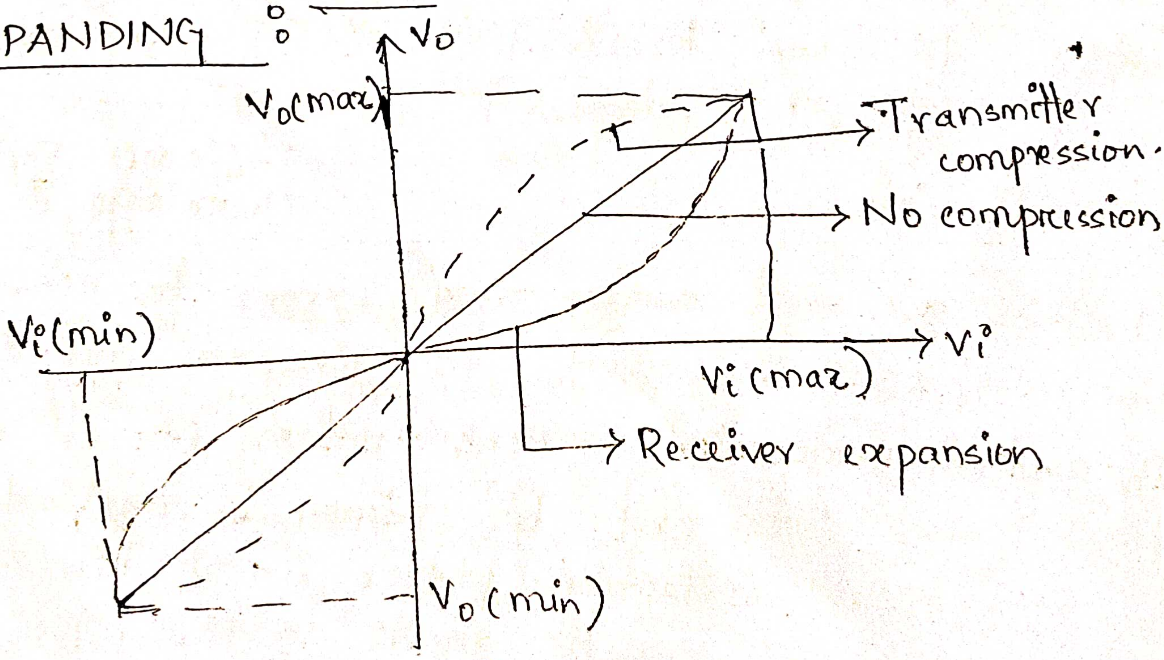
EQUALIZATION :-

→ The ISI causes distortion. These distortions can be reduced by designing a proper equalizer.

→ If $H_c(\omega)$ is the known frequency response of the channel then an equalizer is designed whose frequency response $H_e(\omega)$ is the inverse of $H_c(\omega)$.

→ Equalizing filters are inserted between the receiving filter and the A/D converter. The equalizer filter is adjusted manually by observing the eye pattern. In adaptive equalizer, this is done automatically by using feedback technique.

COMPANDING :-



(Companding)

→ The effect of an adaptive step size may be achieved by distorting the signal before quantization.

→ An inverse distortion has to be introduced at the receiver to make the overall transmission distortionless.

→ Here, the o/p is enhanced more at low amplitudes than at high amplitudes. This o/p is then applied to the quantizer. Thus the low amplitude signal will carry more quantization levels than the undistorted signal (i.e the solid line).

∴ → A signal transmitted through a non-linear network with the characteristic shown by the dashed curve will have its extremities compressed. Hence this network is known as a compressor.

→ At the receiver, an inverse operation is to be performed to recover the original signal. This is done by an expander connected between the decoder and holding ckt, whose characteristic is shown by dotted curve.

→ so the combination of <u>compressor</u> and <u>expander</u> is known as <u>compander</u>, which performs the companding operation.

→ <u>Time-Division Multiplexing of DCM signals</u> —

→ In PCM system, various signals can share the time scale giving rise to TDM. Time division multiplexing

→ 2 types of multiplexing (i) synchronous TDM

(ii) Asynchronous TDM.

Synchronous TDM :—

→ Here each sample is coded into several bits. The multiplexing is possible in two ways.

(i) Bits are taken one by one from each channel sample code.

→ After the 1st bits from all channel samples are taken, the commutator takes the 2nd bits from all channel samples and so on. This is called "bit interleaving".

(ii) All code bits of the 1st channel samples are taken followed by all code bits of the second channel samples and so on. Here, the desired commutator speed is less than that required in the first method. This is called "word interleaving".

→ A synchronizing bit is added at the end of each frame of synchronization between the commutator and decommutator.

→ Thus for n channels and N bits per sample, then the size of a frame is nN bits.

→ The signal that is to be time division multiplexed is bandlimited to the same frequency, resulting in the same sampling frequency for all channels and hence the name synchronous time division multiplexing.

→ Asynchronous TDM :—
(Pulse stuffing)

→ When the signals to be time division multiplexed are bandlimited to different frequency, their sampling frequency is also different.

→ Such asynchronously sampled signals are multiplexed by a technique called pulse stuffing.

→ For multiplexing the asynchronous signals, a storage device is used called as '<u>elastic store</u>' to store and reproduce data at different speeds.
e.g tape recorder.

→ In Asynchronous TDM, different signals are sampled at different sampling frequencies (as they are bandlimited to different frequencies).

→ The samples are stored on different storage devies like tape recorder. The recording rate of each storage device is different due to the different sampling frequencies.

→ While transmitting these signals, each storage device is played back at different speeds in such a way that the o/p sample rate of each device is the same.

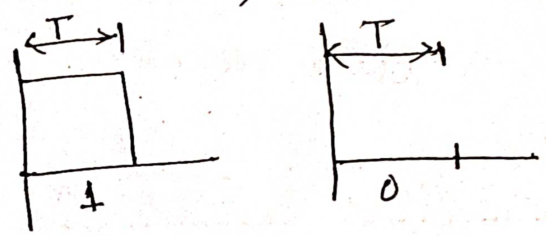
→ So these signals can now be synchronously time division multiplexed and the transmitted. At the receiver this process is reversed to recover each signal.

LINE CODES :—

→ The digital data (0's and 1's) are transmitted over the line by means of "Lines Codes" (or Data Transmission codes or Modulation codes'). Thus they give electrical representation of symbols 0 and 1.

→ Types of Line codes

1. UNRZ (Unipolar Non-Return to zero code) —

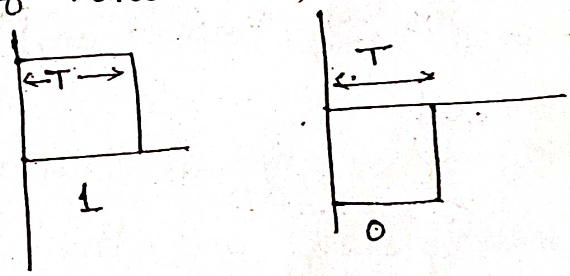


→ Here 1 = positive pulse

0 = no pulse

→ It is called as 'On-Off Code'.

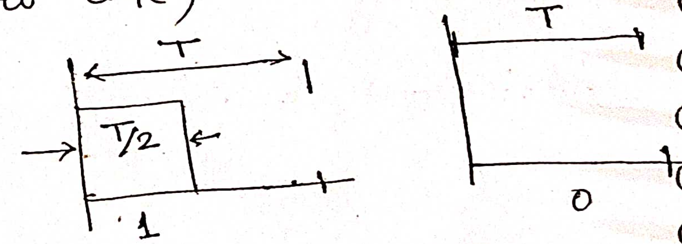
2. BNRZ (Bipolar Non-Return to zero code) —



1 = positive pulse

0 = negative pulse.

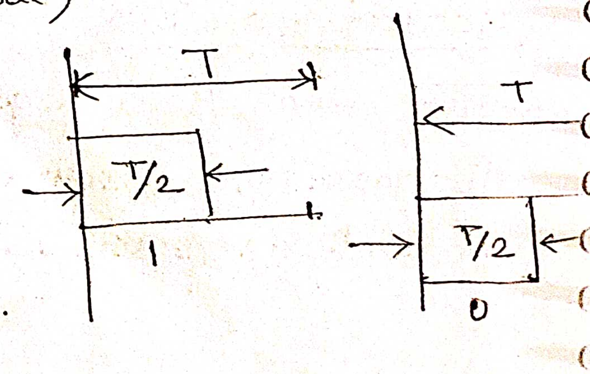
3. URZ (Unipolar Return to zero code) —



1 = positive pulse of half symbol-width

0 = nopulse.

4. BRZ (Bipolar Return to zero code) —

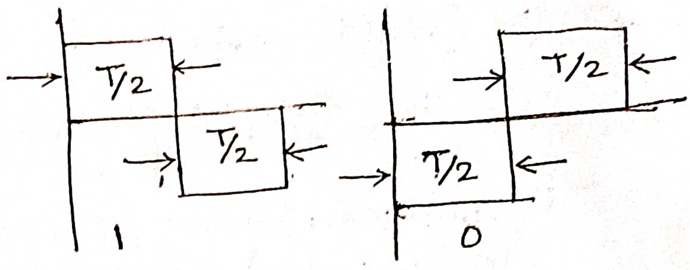


1 = positive pulse of half symbol width

0 = negative pulse of half symbol width.

5. Split-Phase Code (Manchester Code) :—

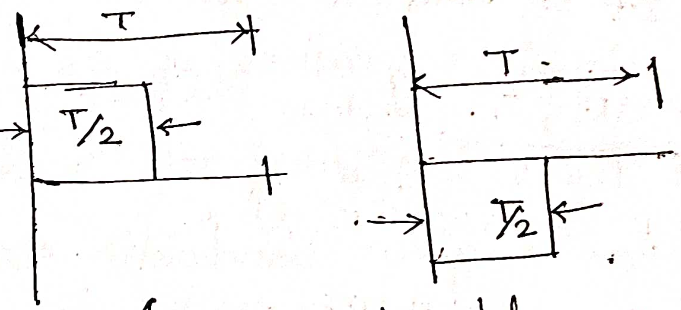
1 = positive half symbol width pulse followed by a negative half symbol width pulse.



0 = negative half symbol width pulse followed by a positive half-symbol width pulse.

6. Differential Code or BRZ-AMI (Binary Return to zero - Alternate Mark Inversion) code :—

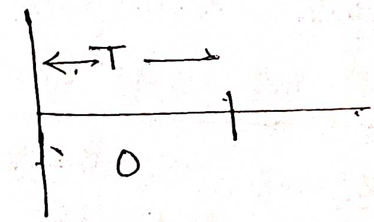
1 = positive pulse of half-width and a negative pulse of half width.



(symbols alternately occur (1))

0 = no pulse.

Mark = 1 & space = 0



→ Properties of a Line Code :—

1. Transmission B.W —

→ The min B.w required depends on the highest fundamental frequency of the waveform. It should be as small as possible.

2. Favourable Power spectral Density –

The signal spectrum should be matched to the channel frequency response. Zero dc component is preferable.

3. Timing (clock) Recovery –

It should be possible to extract timing or clock information from the signal.

4. Error Detection and correction capability –

5. Ease of Detection and Decoding –

6. Transparency –

It should be possible, to correctly transmit a digital signal regardless of 0's and 1's.

7. Power efficiency –

For a given bandwidth and a specified error probability the transmitted power for a line code should be as small as possible.

B.W of PCM system :–

→ Let there are n channels each bandlimited to f_m are to be time division multiplexed.

→ Let N = length of PCM code

$\Rightarrow 2^N = M$ = quantization levels.

→ The B.W of PCM system depends on the bit duration (bit time slot) which is calculated as

Sampling frequency $= 2f_m$

sampling period $= 1/2f_m$.

→ For n channels and N bits/sample and one synchronizing bit, the total no. of bits/sampling period (frame)

$= nN + 1$.

→ so, the bit duration $= \dfrac{\text{sampling period}}{\text{total no. of bits}}$

$$\Rightarrow T_b = \frac{1}{(nN+1)2f_m} \text{ sec.} \quad \text{——— (1)}$$

→ BW of the squarewave signal is,

$$BW = 1/T_b \quad \text{——— (2)}$$

→ Using eq^n (1) & (2) the B.W of the PCM system becomes,

$$\boxed{BW = (nN+1)\,2f_m} \text{ Hz.}$$

If $N >> 1$ and $n >> 1$,

then $\boxed{BW \approx 2nNf_m}$ Hz.

→ Noise in PCM systems :—

→ There are two major sources of noise in a PCM system,

(i) Transmission noise introduced outside the transmitter.

(ii) Quantization noise introduced in the transmitter.

→ In PCM transmitter, a quantized value of the sample is encoded instead of its actual value. Hence error occurs. It is called as the noise due to quantization.

Let,

$m(t)$ = the message signal

M = equal voltage intervals each having a magnitude of S V.

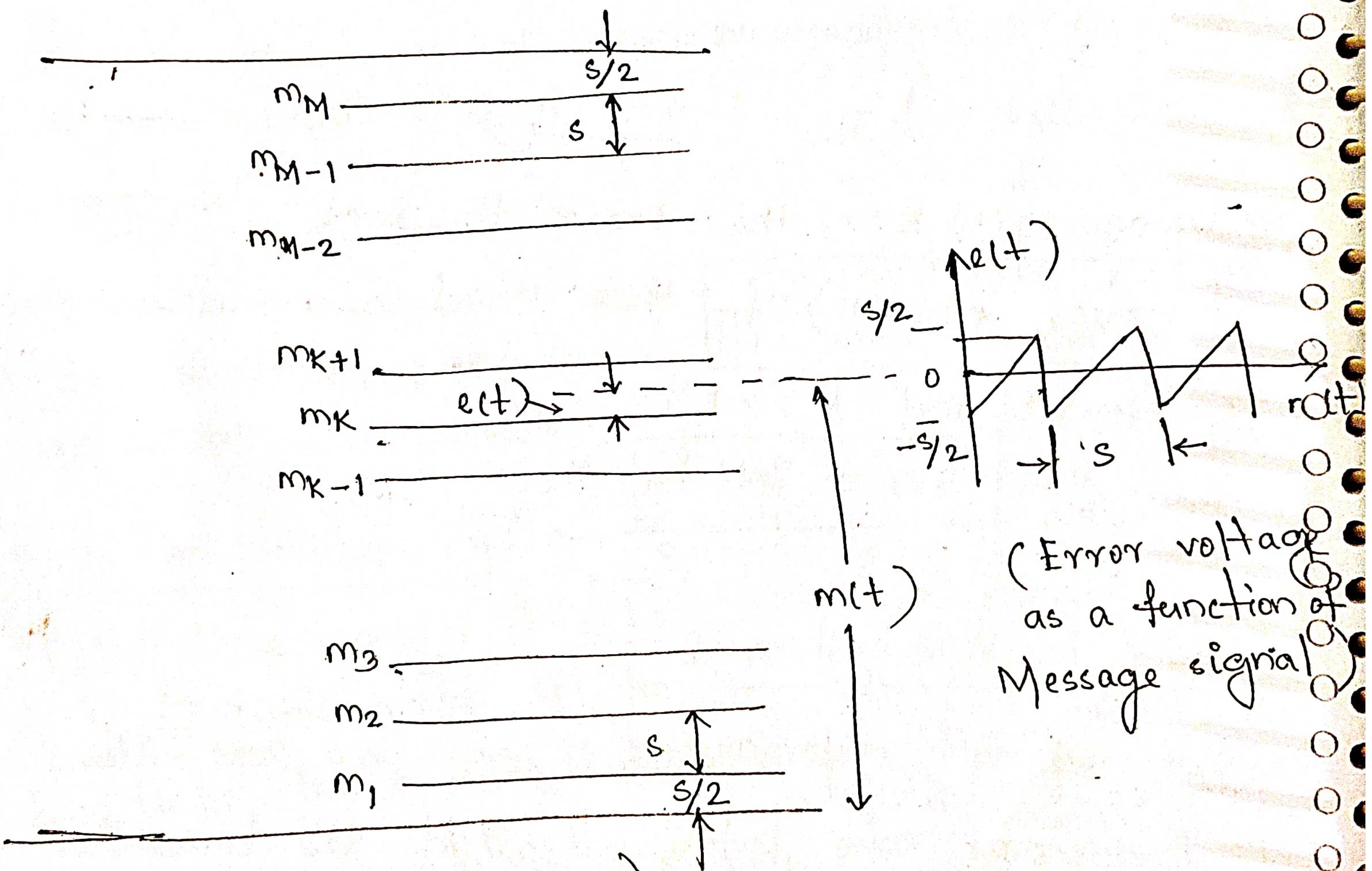
$m_1, m_2 \ldots m_M$ = quantization levels.

→ Let $m(t)$ be closest to the quantization level m_K.

i.e m_K is the quantized o/p.

The quantization error is,

$$\boxed{e = m(t) - m_K}$$



(Error voltage as a function of Message signal)

(Quantization Levels)

[Noise in PCM system)

→ Here $e(t)$ is a function of the instantaneous value of the signal $m(t)$

$f(m)\,dm$ = probability that $m(t)$ lies in the voltage range $\left(m - \frac{dm}{2}\right)$ to $\left(m + \frac{dm}{2}\right)$

→ the mean square ~~error~~ quantization error or quantization noise is,

$$\overline{e^2} = N_q$$

$$= \int_{m_1 - \frac{s}{2}}^{m_1 + \frac{s}{2}} f(m)(m - m_1)^2\,dm + \int_{m_2 - \frac{s}{2}}^{m_2 + \frac{s}{2}} f(m)(m - m_2)^2\,dm + \dots + \int_{m_M - \frac{s}{2}}^{m_M + \frac{s}{2}} f(m)(m - m_M)^2\,dm \quad \text{--- (1)}$$

→ If M (no. of quantization levels) is large, f_m is constant within each quantization range as s (step size) is very small.

→ Let $f(m) = f^{(1)}$, in the 1st term of (1)

$f(m) = f^{(2)}$, " 2nd "

$f(m) = f^{(M)}$, ... Mth or last term of (1).

then eq^n (1) becomes,

$$N_q = f^{(1)} \int_{m_1 - \frac{s}{2}}^{m_1 + \frac{s}{2}} (m - m_1)^2\,dm + f^{(2)} \int_{m_2 - \frac{s}{2}}^{m_2 + \frac{s}{2}} (m - m_2)^2\,dm$$

$$+ \dots + f^{(M)} \int_{m_M - \frac{s}{2}}^{m_M + \frac{s}{2}} (m - m_M)^2\,dm \quad \text{--- (2)}$$

Let $x = m - m_K$

$\Rightarrow dx = dm$.

→ Hence eq^n (2) becomes,

$$N_q = f^{(1)} \int_{-S/2}^{S/2} x^2 dx + f^{(2)} \int_{-S/2}^{S/2} x^2 dx + \cdots + f^{(M)} \int_{-S/2}^{S/2} x^2 dx$$

$$= \left[f^{(1)} + f^{(2)} + \cdots + f^{(M)}\right] \int_{-S/2}^{+S/2} x^2 dx$$

$$= \left[f^{(1)} + f^{(2)} + \cdots + f^{(M)}\right] S^3/12$$

$$= \left[f^{(1)}S + f^{(2)}S + \cdots + f^{(M)}S\right] \frac{S^2}{12} \quad \text{——— (3)}$$

→ The probability that m lies in the entire range of signal,

$$f^{(1)}S + f^{(2)}S + \cdots + S + f^{(M)}S = 1$$

→ Eq^n (3) becomes,

$$\boxed{N_q = \frac{S^2}{12}}, \quad \text{——— (4)}$$

→ The mean square value of the o/p signal is equal to the mean square value of the quantized samples.

$$S_o = \overline{m_K^2}$$

$$= \frac{1}{M}\left[\left(\frac{S}{2}\right)^2 + \left(\frac{3S}{2}\right)^2 + \left(\frac{5S}{2}\right)^2 + \cdots + \left\{\frac{(2M-1)S}{2}\right\}^2\right]$$

$$= \frac{S^2}{4M}\left[1^2 + 3^2 + 5^2 + \cdots + (2M-1)^2\right]$$

$$\simeq \frac{S^2}{4M} \cdot \frac{4M^3}{3} \text{ for large } M. = \frac{S^2 M^2}{3} \quad \text{——— (5)}$$

→ Using eq^n (4) and (5), we get the o/p signal to quantization noise ratio as,

$$\frac{S_o}{N_q} = \frac{S_o}{N_o} = \left(\frac{s^2 M^2}{3}\right) \Big/ \left(\frac{s^2}{12}\right)$$

$$\boxed{= 4M^2} \quad \text{—— (6)}$$

Noise figure —

→ We must find the i/p signal to noise ratio S_i/N_i.

Let the mean square value of noise $= \sigma_n^2$

so, $\boxed{N_i = \sigma_n^2}$ —— 7

→ Let 0 = 0-volt level

1 = A volt level

so avg. signal power = $\boxed{s_i = A^2/2}$ (assuming an equal probability for 0 & 1)

→ Let $\boxed{A = K \times \sigma_n}$ (A should choosen in such a value)

where K = constant

$$\text{so, } \boxed{S_i = \frac{A^2}{2} = \frac{K^2 \sigma_n^2}{2}} \quad \text{—— (8)}$$

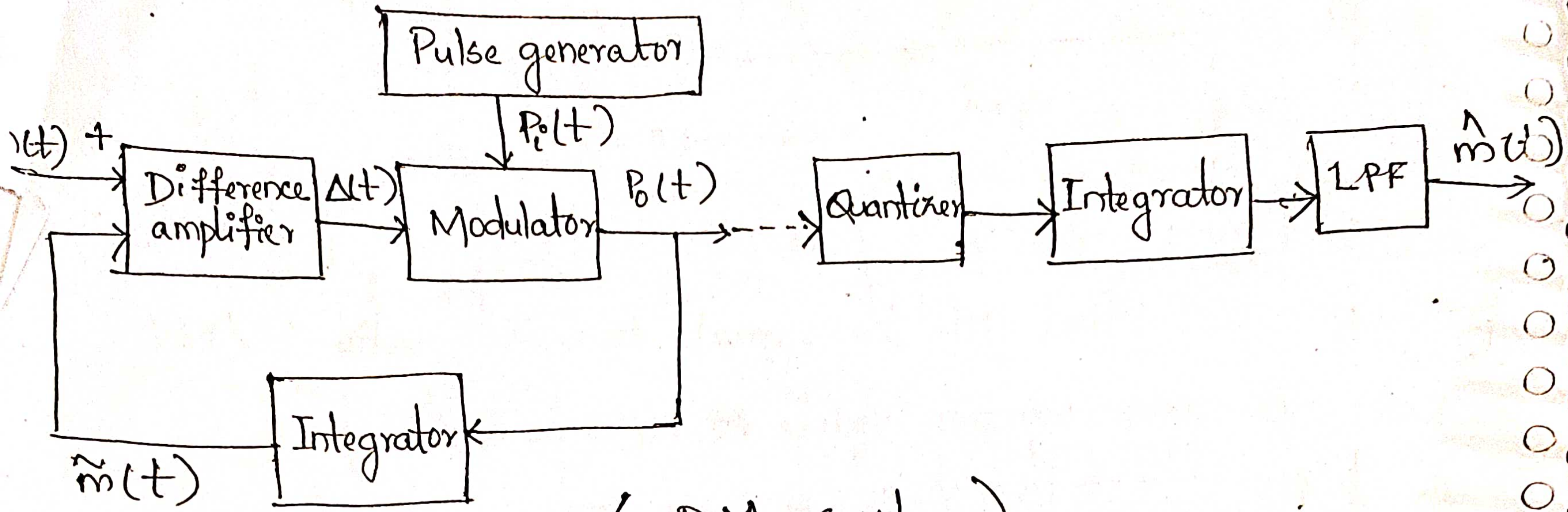
$$\text{→ So } \boxed{\frac{S_i}{N_i} = \frac{K^2 \sigma_n^2}{2\sigma_n^2} = \frac{K^2}{2}} \quad \text{—— (9)}$$

Noise figure,

$$\boxed{F = \frac{S_i/N_i}{S_o/N_o} = \frac{K^2/2}{4M^2} = \frac{K^2}{8M^2}} \quad \text{—— (10)}$$

DELTA MODULATION :—

→ Using DM, an analog signal can be encoded into bits. So DM is also PCM.



(DM system)

→ The pulse generator produces a pulse train $P_i(t)$ of positive pulses. The modulator receives $P_i(t)$ and $\Delta(t)$ the O/P of the difference amplifier.

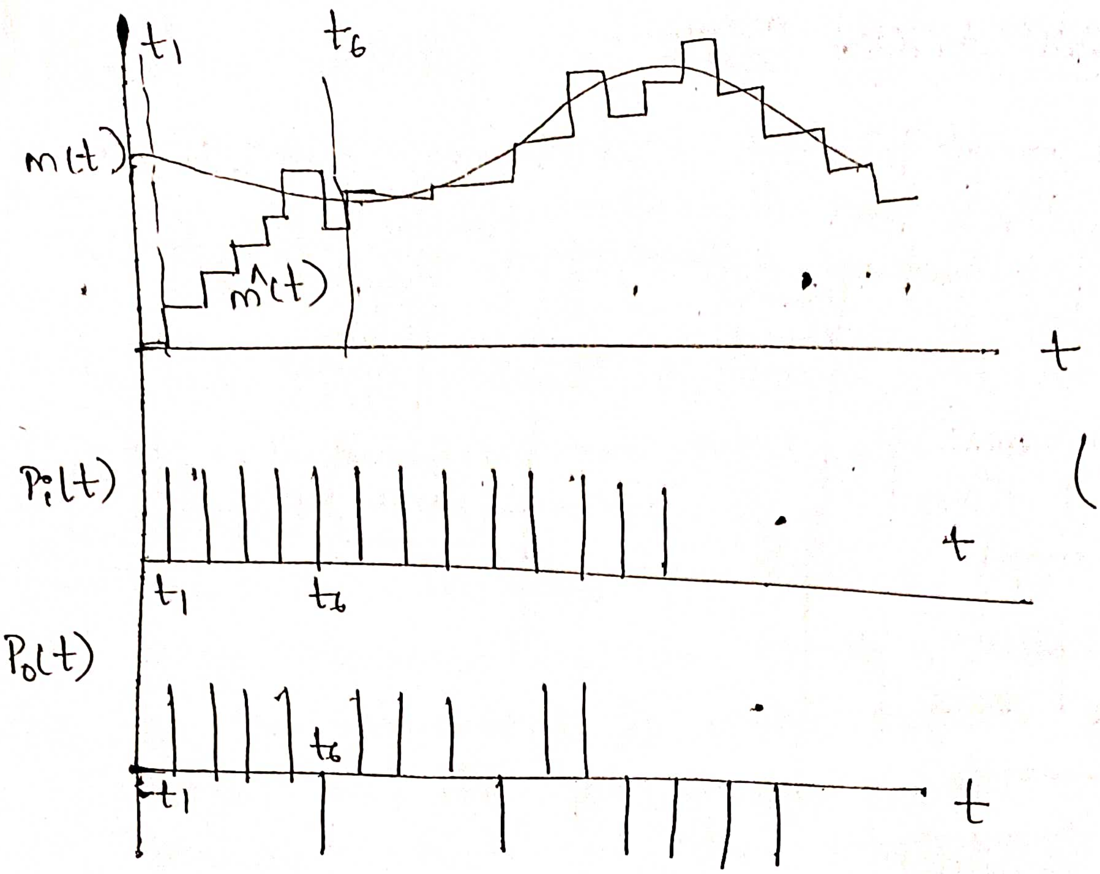
→ The modulator output $P_o(t)$ is the I/P pulse train $P_i(t)$ multiplied by +1 or −1 depending on the polarity of $\Delta(t)$.

→ $P_o(t)$ = positive pulse , if $\Delta(t)$ is +ve
= neg. pulse , if $\Delta(t)$ is −ve.

→ The O/P of the modulator $P_o(t)$ is applied to an integrator whose O/P is $\tilde{m}(t)$.

→ The I/P signal $m(t)$ and the integrator O/P $\tilde{m}(t)$ are compared in a difference amplifier whose O/P is,

$$\Delta(t) = m(t) - \tilde{m}(t)$$

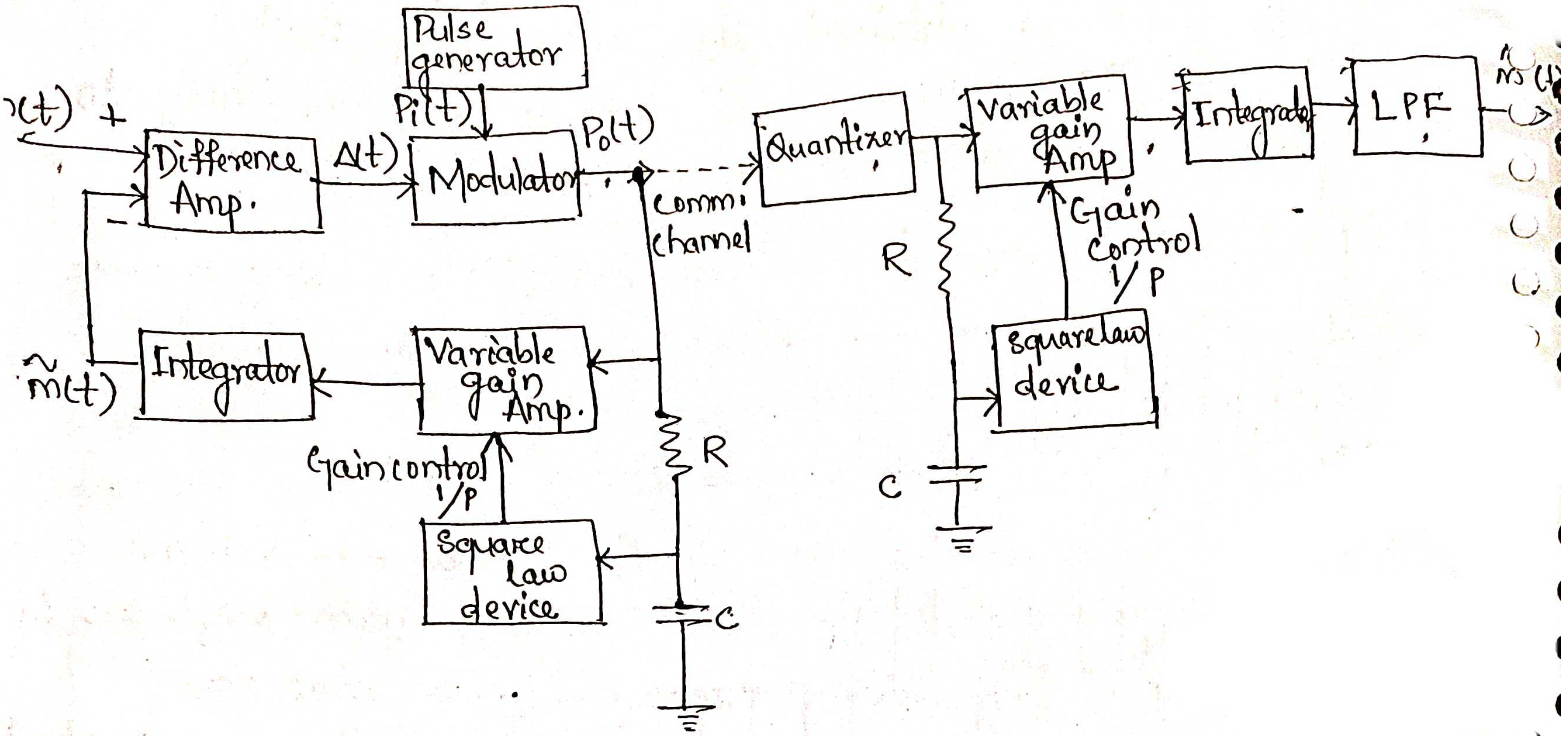


(Waveforms of DM)

→ The waveform Po(t) is transmitted. At the receiver the quantizer decides whether the received pulse is +ve or −ve. Hence assuming no error, the o/p of the quantizer is the same as the waveform Po(t) and fed to an integrator, whose o/p takes the form of the waveform $\tilde{m}(t)$. The LPF then smoothens the o/p of the integrator and gives a waveform $\hat{m}(t)$ which is similar to m(t)

→ As the difference signal $\Delta(t) = m(t) - \hat{m}(t)$ is transmitted in this method, it is known as <u>delta modulation</u>.

Adaptive Delta Modulation :



(ADM system)

→ For the +ve and -ve slopes of DM, the recovered waveform comes distorted.

→ It can be ~~avoid~~ overcomed by suitably changing the step size.

→ Slope overload can be overcome if the step size is increased in such a way that the magnitude of the slope of $\tilde{m}(t)$ becomes greater than the magnitude of the slope m(t) and when the signal variations are less than the step size, the stepsize may be reduced to take care of the situation.

→ So a DM system which adjusts its step size is known as the Adaptive Delta Modulation (ADM) system.

→ ~~Block~~ In the block diagram of ADM system, on the tx side a variable gain amplifier is used before the integrator with $P_o(t)$ as its input.

→ The amplifier gain depends on the gain control I/p which is obtained by integrating $P_o(t)$ in an RC - network and then passing the integrator O/p through a square law device.

→ Under slope overload condition, $P_o(t)$ is a long sequence of either +ve or -ve pulses. The RC integrator integrates these pulses.

→ Thus the integrator O/p is either of a large +ve or large -ve value i.e the O/p of a square law device. So the gain control I/p of the variable gain amplifier is large and its gain increases. Hence step size increases, which can then take care of the slope-overload

→ On the receiver side, the O/p of the quantizer is fed to a variable gain amplifier whose gain control I/p is derived from an RC integrator and a square law device.

→ Thus an adaptive adjustment of the stepsize is obtained at the receiver, resulting in an undistorted reception of the transmitted signal.

→ Noise in Delta Modulation :—

Quantization noise —

→ The quantization error in DM is given by,

$$\Delta(t) = \dot{m}(t) - \tilde{m}(t)$$

→ The maximum quantization error in DM is $\pm S$.
Let $\Delta(t)$ takes on all values between $-S$ and $+S$ with equal likelihood,
the the probability density of $\Delta(t)$ is,

$$f(\Delta) = \frac{1}{2S} \quad , (-S \leq \Delta \leq +S)$$

→ The normalized power of $\Delta(t)$ is,

$$\overline{[\Delta(t)]^2} = \int_{-S}^{+S} \Delta^2 f(\Delta) d\Delta$$

$$= \int_{-S}^{+S} \frac{\Delta^2}{2S} d\Delta = \frac{S^2}{3}.$$

→ Hence the O/P noise power in the baseband frequency range 0 to f_M is,

$$N_q = \frac{S^2}{3} \frac{f_M}{f_b} = \frac{S^2 f_M}{3 f_b}.$$

Output signal Power :—

Let $m(t) = A \sin \omega_M t$

where $A \rightarrow$ amplitude

$\omega_M \rightarrow 2\pi f_M$, f_M = upper limit of baseband frequency range.

→ O/P signal power is,

$$S_o = \overline{m^2(t)} = \frac{A^2}{2}.$$

→ The max. slope of $m(t) = \omega_M A$

where $\omega_M A = s f_b$

$$\Rightarrow \boxed{S_o = \frac{s^2 f_b^2}{2\omega_M^2}}$$

O/p signal to Quantization Noise Ratio (S/N_q) –

$$\frac{S_o}{N_q} = \frac{3}{8\pi^2}\left(\frac{f_b}{f_M}\right)^3$$

$$\approx \frac{3}{80}\left(\frac{f_b}{f_M}\right)^3.$$