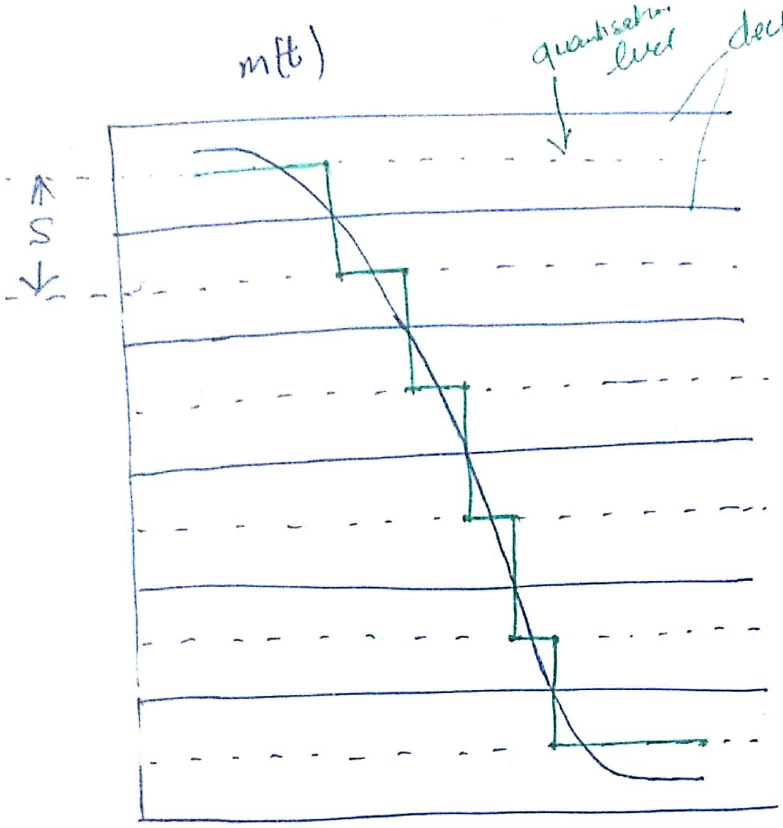
to counter balance the distortion produced by pulse stretcher network

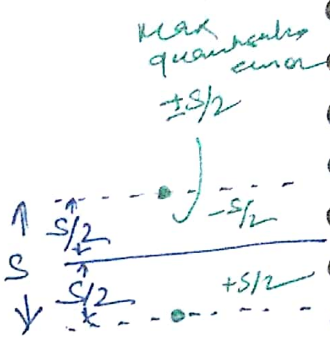
Quantisation

→ it is the process which makes a digital communication system to have super noise performance then analog communication system.

→ in the process of quantisation, a sig $m_q(t)$ which is an approx. to $m(t)$ is generated.

consider a baseband s/g $m(t)$ varing b/w lower limit V_l and upper limit V_H. Now the Baseband s/g is applied to a quantiser which gives a staircase waveform $m_q(t)$. The $m_q(t)$ is having an advantage that it is free from additive noise the baseband s/g $m(t)$ is divided into M no. of quantisation levels and seperation b/w two successive levels is step size S.





total s/g range = $V_H - V_L$
No of quantisation levels M
stepsize = S

$$\boxed{MS = V_H - V_L}$$

$$\boxed{M = \frac{V_H - V_L}{S}}$$

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Quantisation error

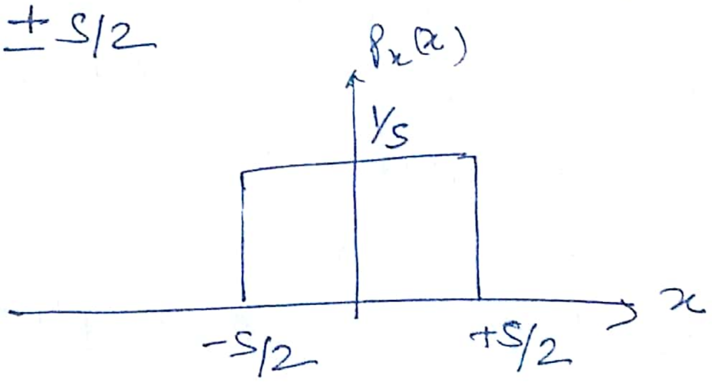
from the quantiser responce, it is clear that at any instance of time, $m(t)$ and $m_q(t)$ are not exactly equal.

the difference b/w $m(t)$ & $m_q(t)$ is called as quantisation error

at the max. quantisation error can be $\pm S/2$

quantisation noise Power

quantisation noise ~~is~~ uniformly distributed b/w $\pm S/2$



$$P_{nq} = E[e_q^2(t)] = \int_{-\infty}^{\infty} x^2 P_x(x)\, dx$$

$$= \frac{1}{S}\int_{-S/2}^{S/2} x^2\, dx$$

$$= \frac{1}{S}\left(\frac{x^3}{3}\right|_{-S/2}^{S/2} = \frac{1}{3S}\left[\frac{S^3}{8} + \frac{S^3}{8}\right]$$

$$= \frac{2S^2}{24}$$

$$\boxed{P_{nq} = \frac{S^2}{12}}$$

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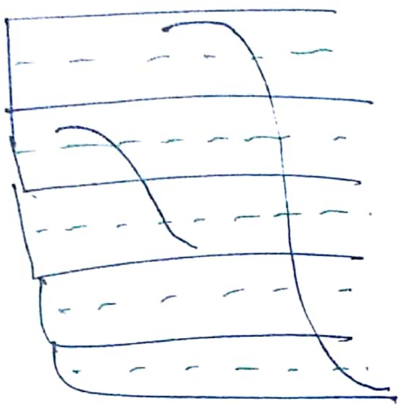
Companding



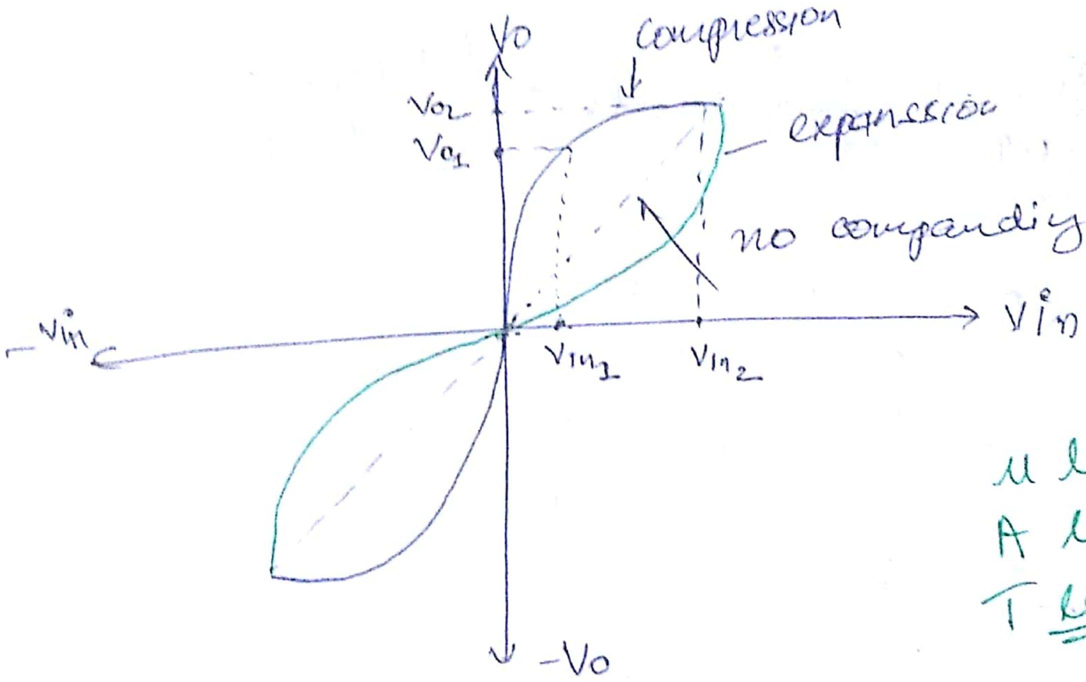
→ the quantisation error is dependent on step size. considering s/g to quantisation noise ration (SNq_r) when uniform quantiser is employed, lower amplitude s/gs will have a poorer SNQ_r when compared to larger amplitude s/g

→ this is due to the fact that denominator of SNQ_r remains same for lower as well as for larger amplitude s/g

→ but the numerator is low for low amplitude s/g. so to have an uniform SNQ_r, the stepsize is to be adjusted in accordance with s/g that is step size should be small for small amplitude and large for large amplitude

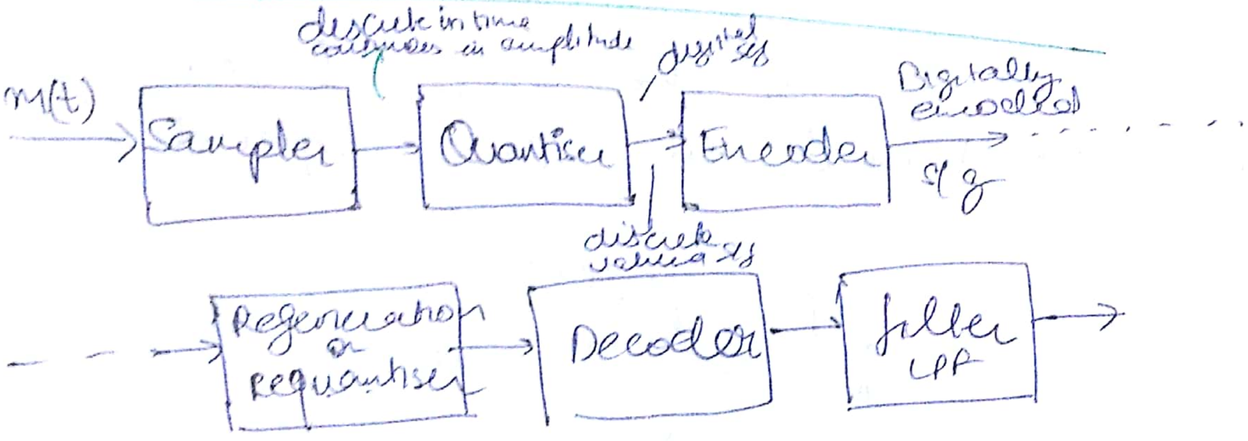
To perform this the sig is passed through a non linear n/w called compressor

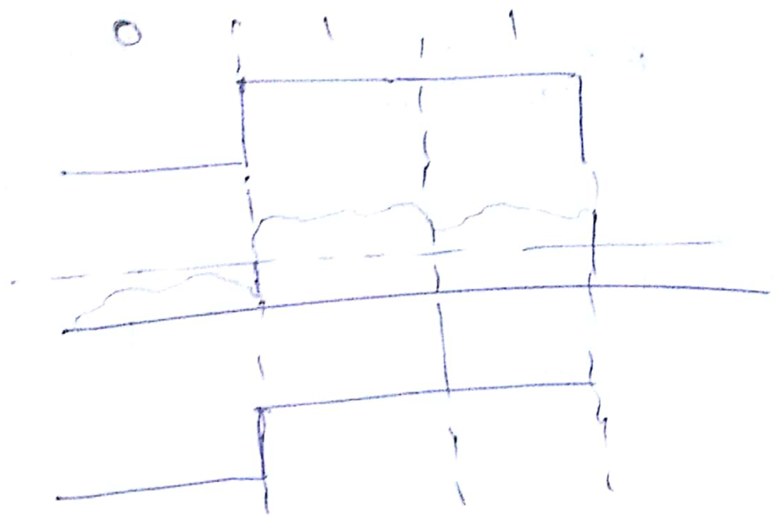
At the RXing end the original level of s/g is maintained using the expander



μ law
A law
T law by Indians

Pulse code Modulation





Working

→ the modulating s/g m(t) is first passed through a sampler to convert the sig in discrete in time and continuous in valued s/g. The quantiser converts it into discrete in time and discrete in valued s/g (digital s/g).

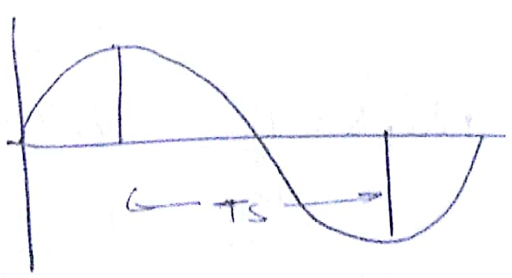
→ The encoder assigns N no. of bits to each sample and finally digitally encoded PCM s/g is transmitted over channel.

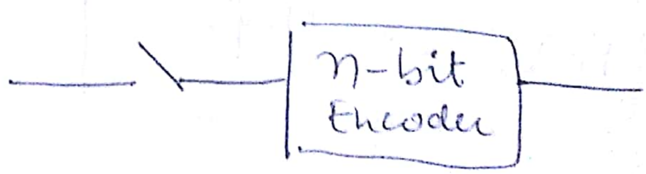
→ at the receiving end, the s/g is passed through Regenerator or Requantiser which removes the additive noise present in s/g and generates fresh digital pulses. The decoder converts the digitally encoded s/g into corresponding analog value and LPF is used to reconstruct the original s/g.

Band width of PCM s/g

M - no of levels

$$2^n \geq M$$





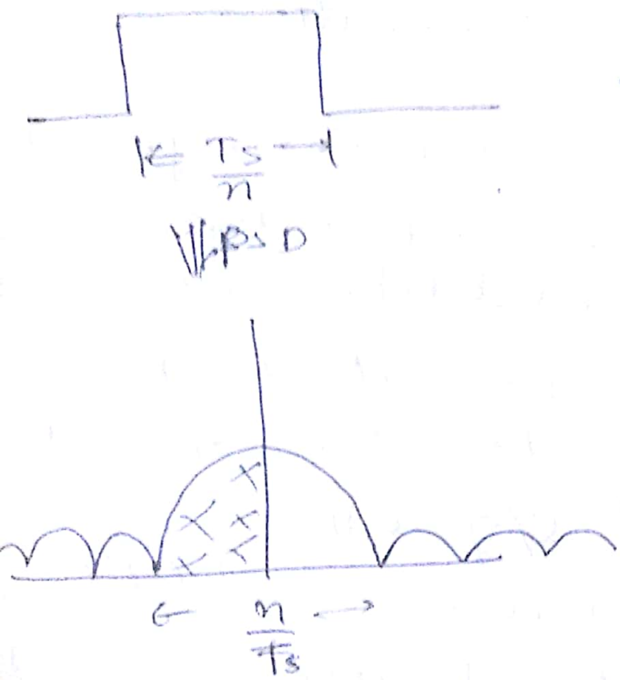
T sec ⟷ n bits

1 sec ⟷ $\frac{n}{T_s}$ bits/sec

$R_b = n f_s$ bits/sec

Bit rate / Data rate / Tx Rate / Pulse rate n bits $\longrightarrow$ Ts sec

Max Pulse width = $\frac{T_s}{n}$ sec



$$(BW)_{PCM} \geq \frac{n}{2T_s}$$

$$(BW)_{PCM} \geq \frac{n f_s}{2}$$

$$BW_{min} = \frac{n f_s}{2}$$

$$\boxed{BW_{min} = n \cdot f_m}$$

Signal to Quantisation noise Ratio in PCM

$\rightarrow$ Consider a sinusoidal modulating s/g $m(t)$ varying b/w $\pm A$

$$P_s = \frac{A^2}{2}$$

$$V_H - V_L = 2A$$

$$MS = V_H - V_L$$

$$S = \frac{2A}{M}$$

$$P_{nq} = \frac{S^2}{12} = \frac{\left(\frac{2A}{M}\right)^2}{12} = \frac{4A^2}{12M^2}$$

$$SN_qR = \frac{P_s}{P_{nq}} = \frac{\frac{A^2}{2}}{\frac{4A^2}{12M^2}}$$

$$= \frac{3}{2} M^2$$

$$M = 2^n$$

$$SN_qR = \frac{3}{2} 2^{2n}$$

$(SN_qR)_{dB} = 10\log_{10}\frac{3}{2} + 10\log_{10}2^{2n}$

$$\boxed{(SN_qR)_{dB} = 1.761 + 6.02n \text{ dB}}$$

$n \rightarrow$ no of bits

$$\boxed{(SN_qR)_{dB} \cong 1.8 + 6n \text{ dB}}$$

if n is high

$$\boxed{(SN_qR) \cong 6n} \text{ dB}$$

$n = 6$; $SN_qR = 37.8$ dB
$n = 7$ $SN_qR = 43.8$ dB +6
$n = 8$ $SN_qR = 49.8$ dB +6

[in PCM system, for each bit increase, the SN_qR increases by 6 dB]

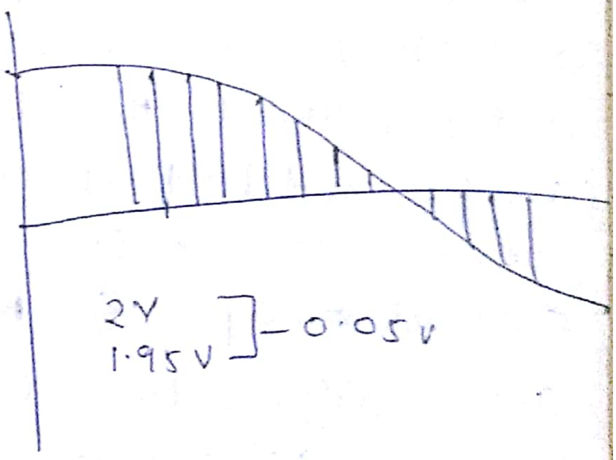
$$\boxed{SN_qR \Rightarrow \text{dynamic Range}}$$

Differential pulse CM

disadvantage of PCM

① as the no of bits increases the system performance improves but at the same time system complexity also increases

① Since BW is directly proportional to no of bits so it also increases proportionally.



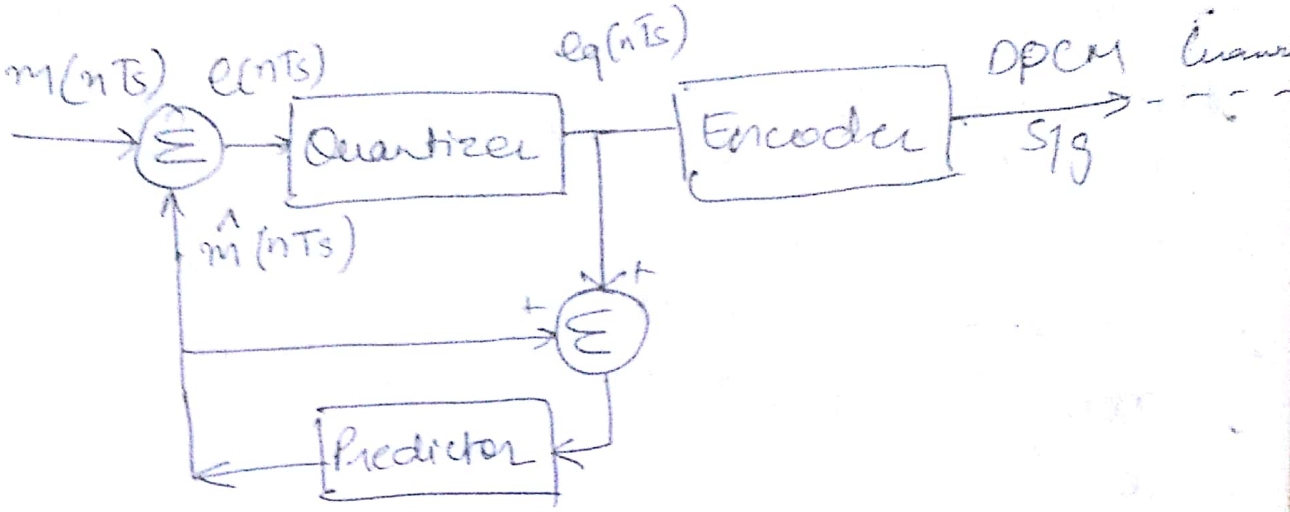
when a s/g is sampled at a rate higher than the nyquist rate, the resulting s/g is found to exhibit as high correlation b/w adjacent samples i.e the s/g does not change rapidly from one sample to next, so instead of transmitting samples at each sampling instant difference b/w sample and its predictive value is transmitted the difference can be added at the receiver to get the original ~~sample~~ s/g (m(t)) such differential scheme has the advantage when being transmitted by using PCM that the s/g formed by all these above differences will be confined to smaller voltage range then the original s/g. Hence no. of quantisation levels required will be less and fewer bits will be needed to encode the message.

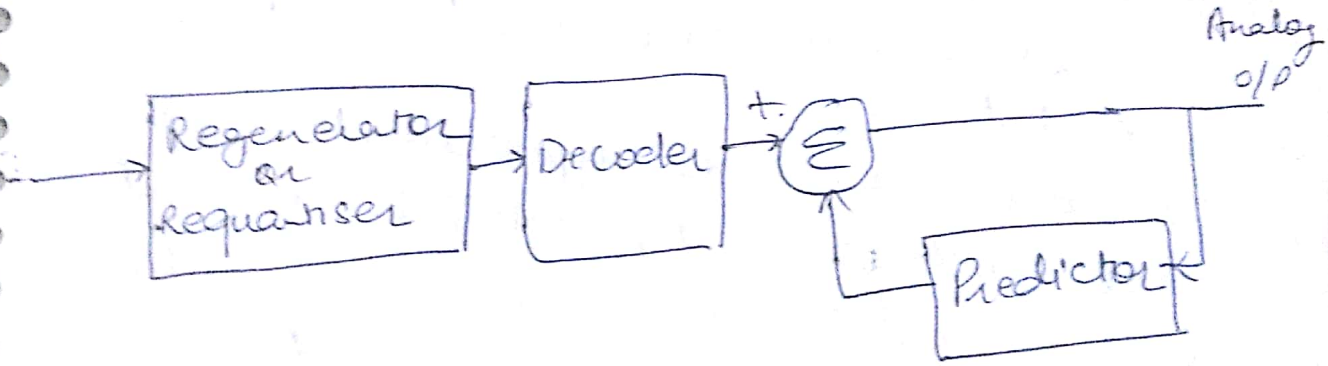
Generation and Detection of DPCM s/g

→ the sampled i/p & its predicted value is feed to an adder which generates the difference b/w actual sample value and its predictive value

→ The difference value is quantised, encoded and transmitted as DPCM s/g.

→ the quantised value is feed to an adder to which another i/p is predicted value the adder generates the sum of quantised value and predicted value





and is given as i/p to predictor, so that it can predict next sample value

→ at the Rxing end the DPCM s/g is passed through regenerator which removes the additive noise from the s/g and generates fresh pulses. The decoder ckt converts digitally encoded s/g into corresponding analog value.

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~~Delta Modulation~~

A this analog value is added with the o/p of receiving & predictor to generate the analog o/p

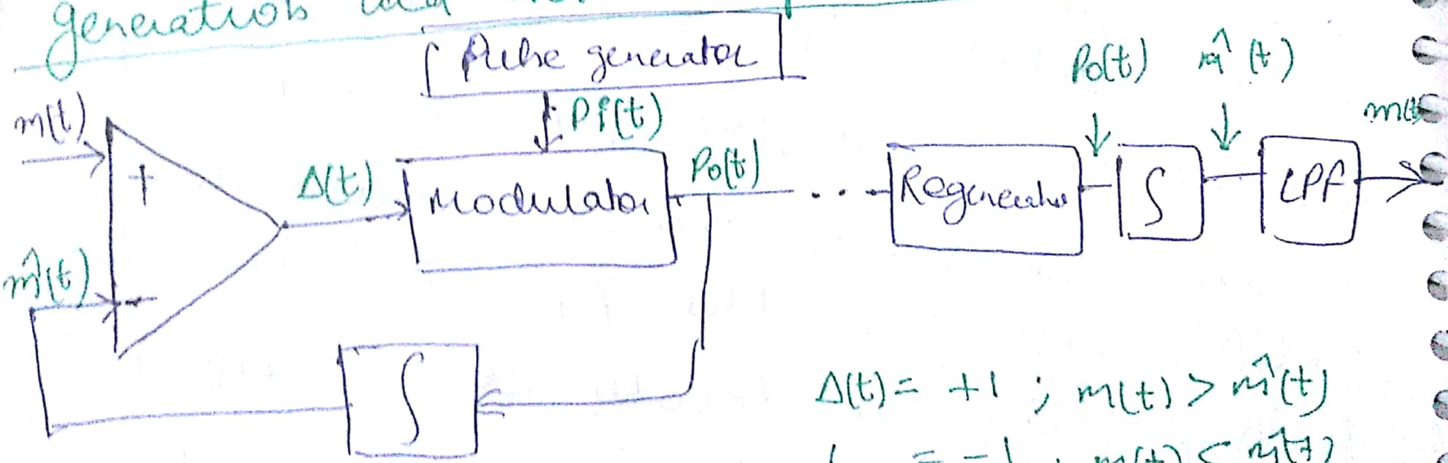
the Receiving & Predictor predicts the same as TXing and predictor if i/p to both is same

Delta Modulation

Delta Mod. is a oversampled modulation scheme in which the txed s/g is encoded in one bit

In DM, the staircase approximation of original modulating s/g m(t) is generated which is passed through a low pass filter to reconstruct the original s/g back

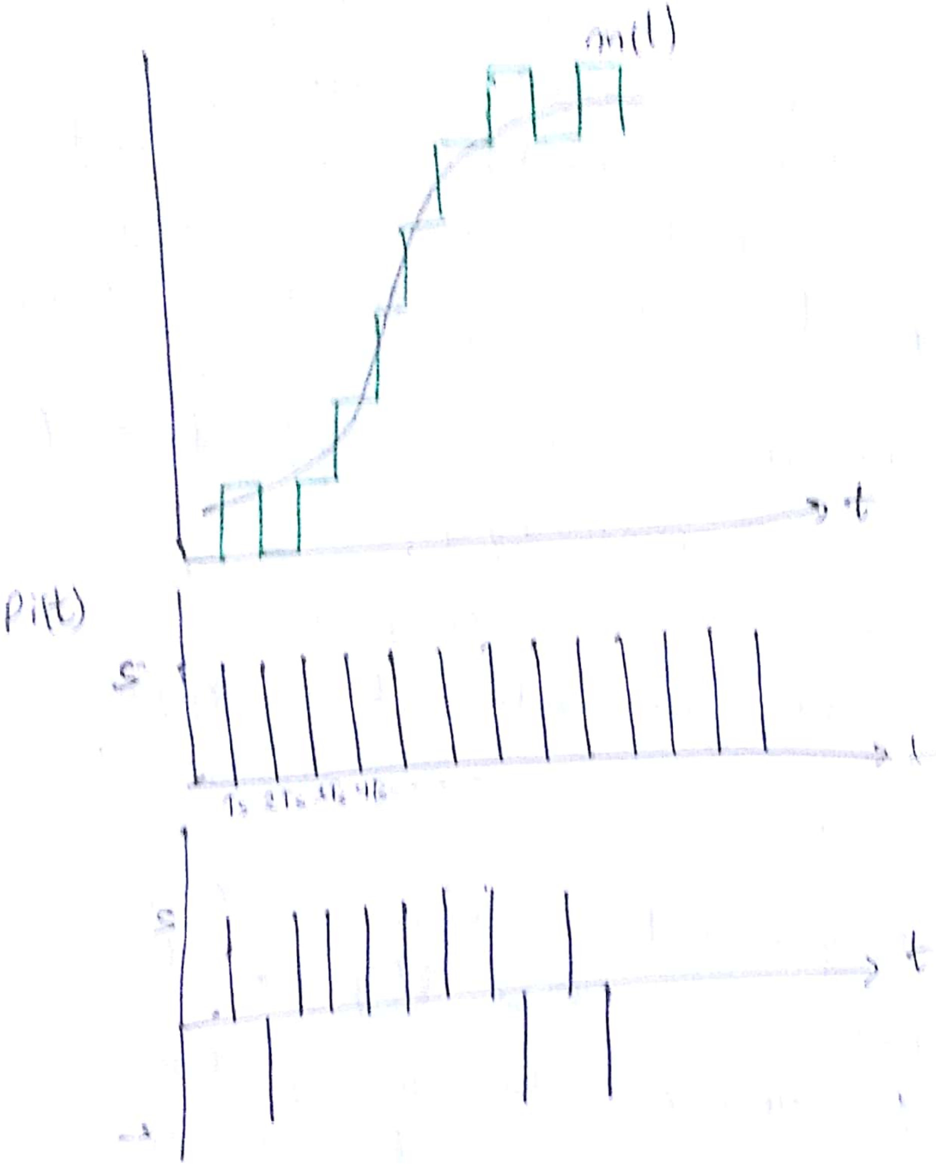
generation and detection of DM s/g



$\Delta(t) = +1 \; ; \; m(t) > \hat{m}(t)$

$\quad\;\; = -1 \; ; \; m(t) < \hat{m}(t)$

$\quad\;\; = \text{Random} \; ; \; m(t) = \hat{m}(t)$



Working

The modulating s/g $m(t)$ and its approximation value $\hat{m}(t)$ are compared in an opamp comparator which generates

$$\Delta t = \begin{cases} +1 & m(t) > \hat{m}(t) \\ -1 & m(t) < \hat{m}(t) \\ \text{Random} & m(t) = \hat{m}(t) \end{cases}$$

(2) the modulator ckt multiplies the o/p of comparator with o/p of pulse generator. $P_i(t)$ to generate o/p impuses $P_o(t)$, The polarity of $P_o(t)$ is dependent on polarity of $\Delta(t)$

(3) the $P_o(t)$ impulses are encoded and transmitted as DM s/g.
Simultaneously these impulses are integrated to generates the approx. s/g $\hat{m}(t)$

<u>At RX</u>

(4) at Rxing end, regenerator removes the additive noise from the s/g and generates fresh pulses these pulses are decoded to generate $P_o(t)$ at receiving end.

(5) $P_o(t)$ is again integrated to regenerate approx. s/g $\hat{m}(t)$ at Rxing end and finally it is passed through a LPF to convert it into a smooth s/g.

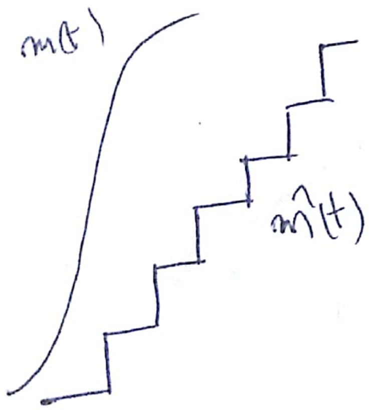
types of error

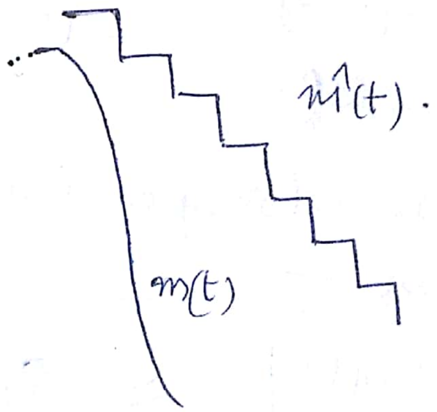
There are two types of errors present in DM system.

① slope overload error

the error produced due to steep slope of i/p sig as compared to slope of approx s/g is known as slope overload error

There are two types of slope overload error





① +ve slope overload error

the error produced due to +ve steep slope of i/p s/g is known as +ve slope over load error

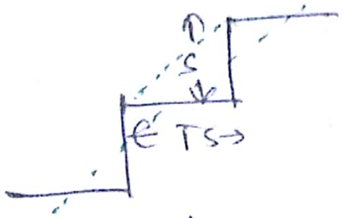
② -ve slope overload error

the error produced due to -ve steep slope of i/p s/g is known as -ve slope overload error for a sinusoidal s/g

$m(t) = A \sin \omega_m t$

$$\frac{dm(t)}{dt} = A\omega_m \cos \omega_m t$$

$$\left.\frac{dm(t)}{dt}\right|_{max} = A\omega_m$$



$$\text{slope of } \hat{m}(t) = \frac{S}{T_S}$$

$$= S f_s$$

To avoid slope overload error :-

$$\boxed{\left.\frac{dm(t)}{dt}\right|_{max} \le S f_s}$$

$$\boxed{A\omega_m \le S f_s}$$

$$A \le \frac{S f_s}{2\pi f_m}$$

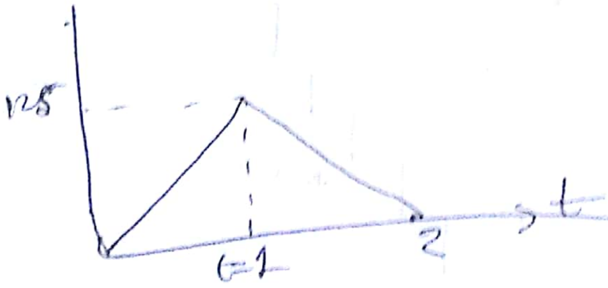
$$\boxed{S \ge \frac{A\omega_m}{f_s}}$$

Q Consider a Δ modulator with step size S and sampling freq 32K samples/sec. the i/p to the DM is

$x(t) = 125t(u(t) - u(t-1)) + (250 - 125t)(u(t-1) - u(t-2))$

calculate min step size to avoid slope overload error

$$\frac{dm(t)}{dt} \le S f_s$$



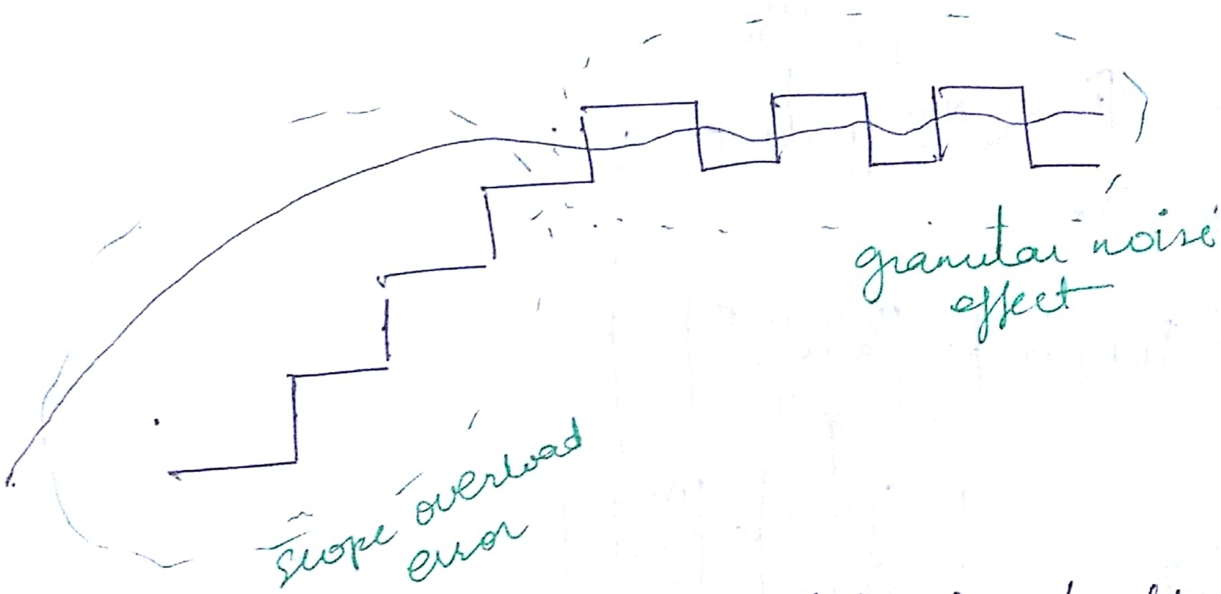
$m = 125$

$$S_{min} = \frac{125}{32 \times 10^3}$$

$$= 2^{-8}$$

(2) Granular Noise

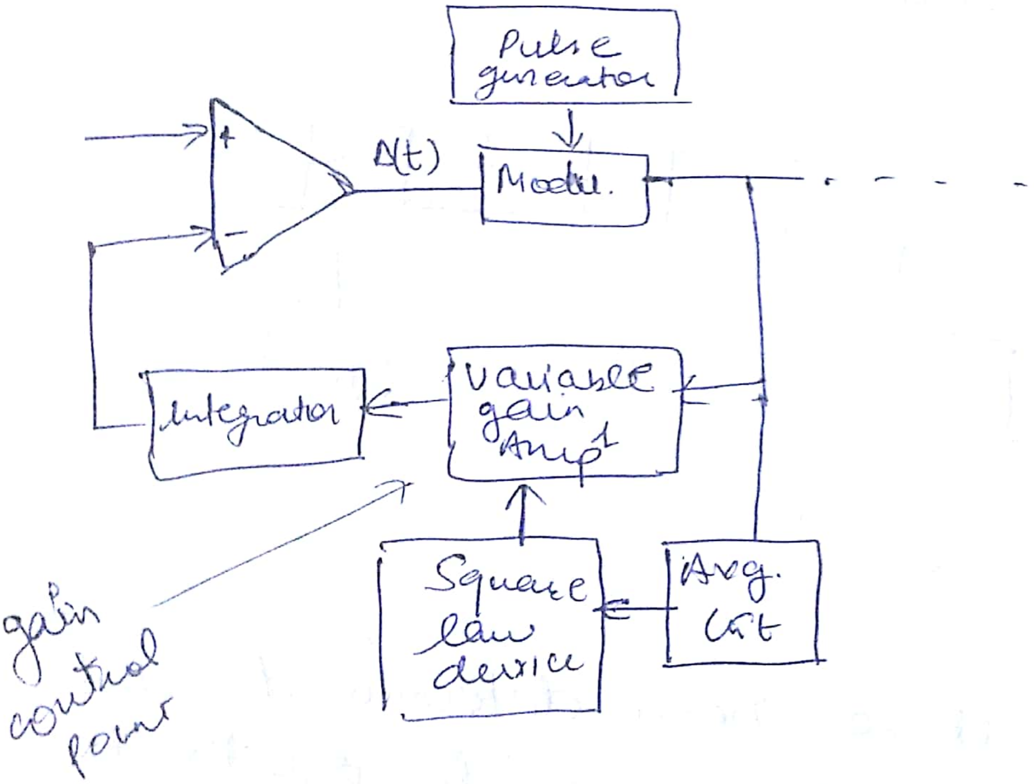
when step size S is too large relative to the local slope characteristics of i/p waveform $m(t)$ there by causing $\hat{m}(t)$ to oscillate around a relatively flat segment of input waveform. This phenomenon is known as granular noise

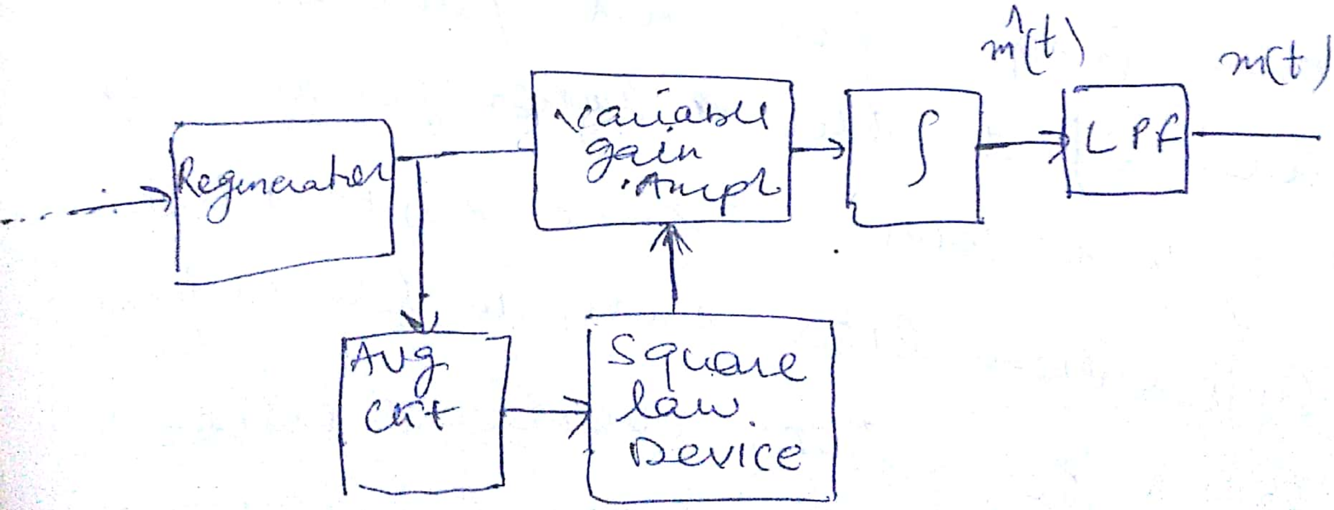


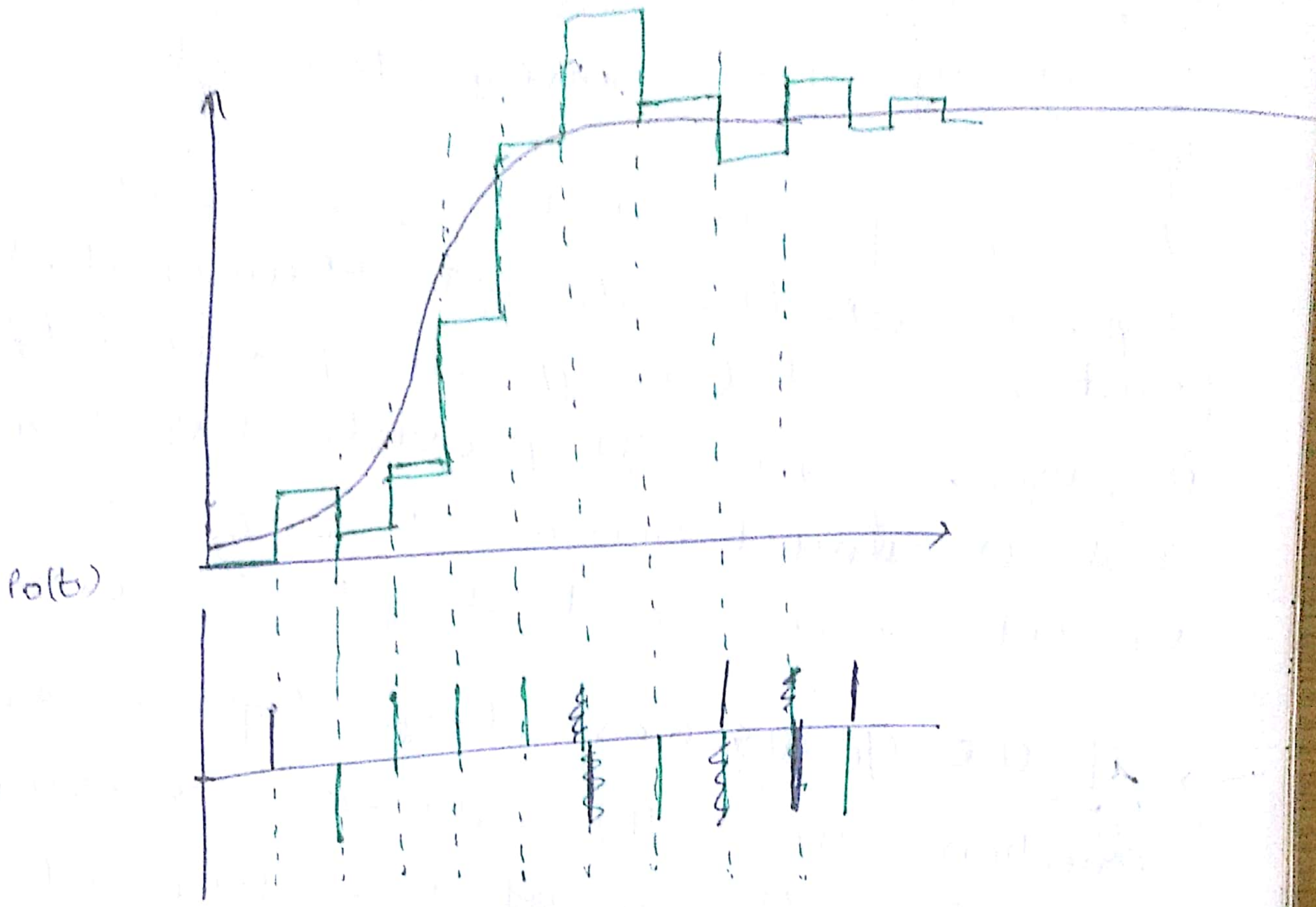
Practically it has been determined that the slope overload error mainly effects the lower freq. region and granular noise error effects the higher frequency region of sig waveform. Since most of the information lies only in lower freq. region so. slope overload error is more objectionable

Adaptive delta Modulation ADM

the Basic principle of ADM system is to very the step size in accordance with slope of input s/g i.e. when the slope of i/p s/g is steep, the step size must be increase and if i/p s/g slope is small then the step size must be reduced







Working

→ The assembly of averaging ckt, square LD. and variable gain Ampr are used to vary the step size of approximation according to the slope of input s/g $m(t)$ s/g. when the i/p to averaging ckt is either continuously +ve or continuously -ve impulse train then its o/p will be +ve & -ve high resp. If the i/p is alternately +ve & -ve impulse. Then its o/p will be zero.

→ the o/p of square law device is always +ve irrespective of i/p.
the o/p of variable gain ampr depends on the i/p at gain control point. If the i/p at gain control pt is high, ampr will provide high gain and it almost blocks the s/g if i/p at gain control pt is zero

→ if the i/p sig has steep slope in +ve direction then p(t) will be continuously +ve hence the o/p of averaging ckt will be +ve high and ampr will provide high gain. So as to minimise +ve slope overload error

→ when i/p s/g has steep slope in -ve direction then p(t) will be continuously -ve hence o/p of averaging ckt will be high in negative direction and amplifier will increase the step size in negative direction to avoid negative slope overload error.

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→ when the slope of i/p sig is almost const. the p(t) will be alternately +ve and -ve impulses hence o/p of averaging ckt is zero and ampl will almost Block the incoming s/g thereby reducing step size

Parameter	PCM	DM	ADM	DPCM
No of bits/sample	8 bit/sample 12 bit/sample 16 bits/sample	1 bit/sample	1 bit	greater than 1 but less than that require in PCM
step size	Fixed	Fixed	Variable	fixed
Sampling Rate	Slightly greater than twice of fm	oversampled	Over sampled	Over sampled
Complexity	highly complex	simplest	Simplest	More complex than DM & ADM but less complex than PCM
Feedback	No	Yes	Yes	Yes
types of error	Quantisation error	slope overload & granular noise effect	Both slope overload error as well as granular noise but very less than DM	Mainly Quantisation error

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Multiplexing

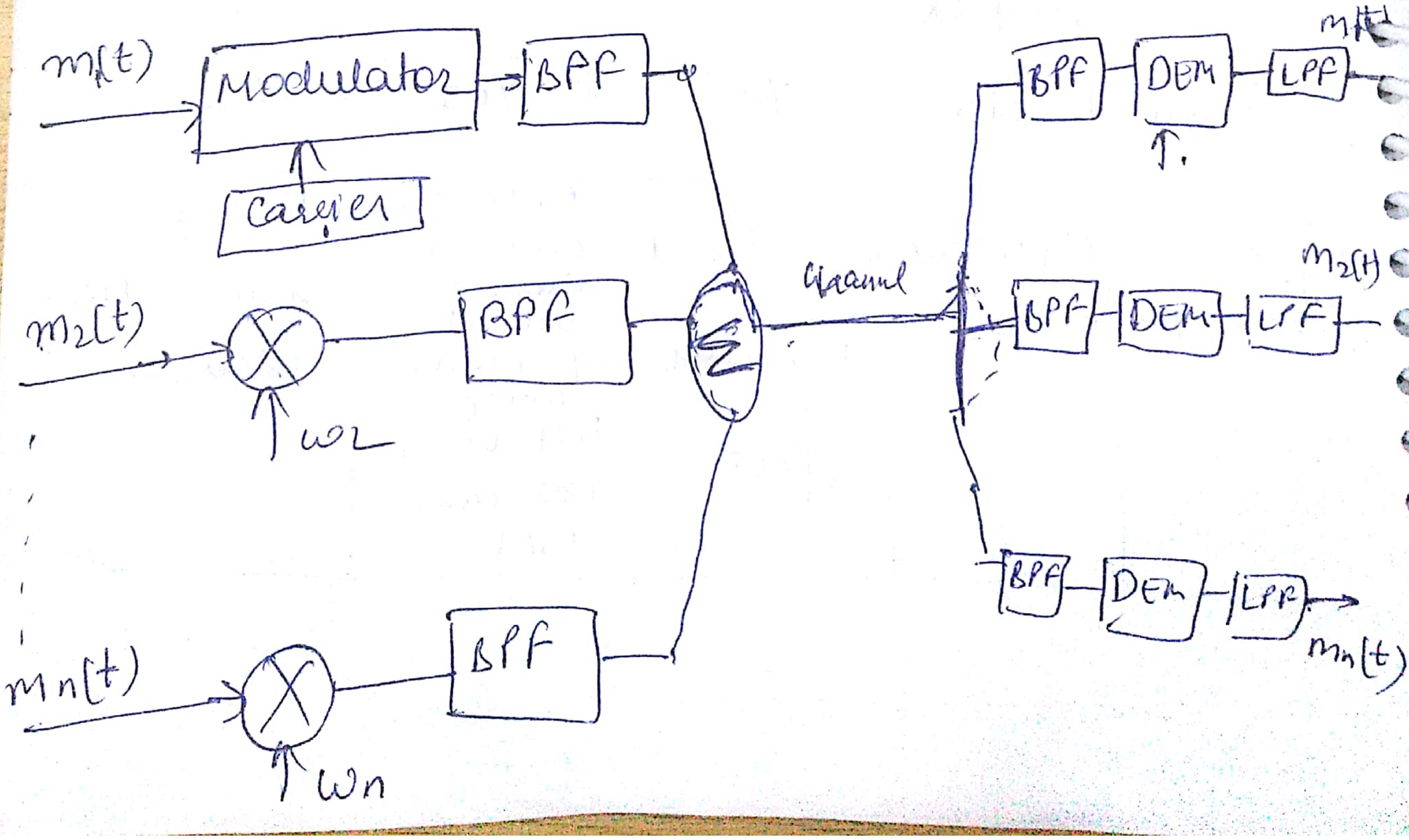
→ Transmission of more than one s/g simultaneously over same channel is known as Multiplexing

→ multiplexing requires that s/g's must be kept appart so that they donot interfare each other and hence can be seperated at the receiving end this is accomplished by seperating the s/gs either in frequency or in time

→ there are two types of multiplexing scheme

① Frequency division Multiplexing

→ the technique of seperating s/g's in frequency is known as FDM



BW of FDM sig

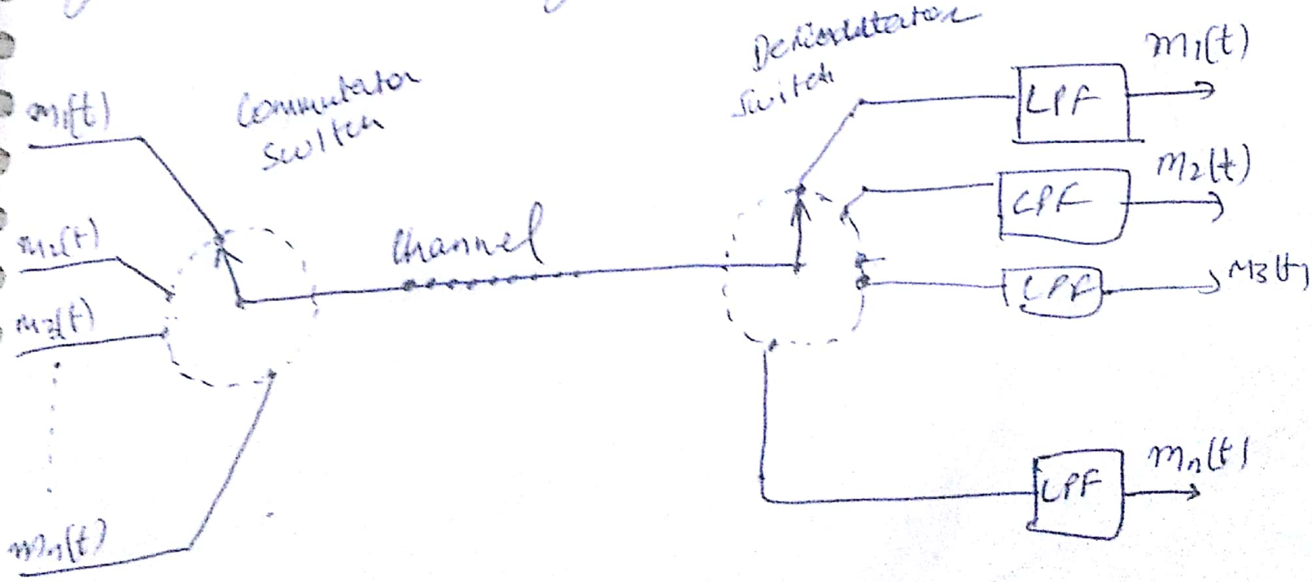
$$\boxed{BW = n f_m}$$

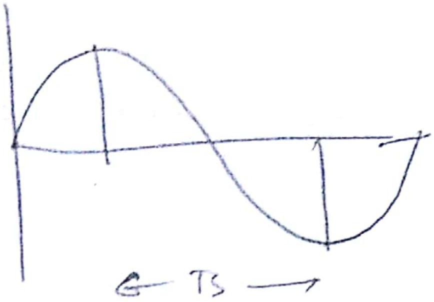
guard bands BW → f_{GB}

$$\boxed{BW = n f_m + (n-1) f_{GB}}$$

time division multiplexing

As no of messages to be transmitted increases the FDM technique presents problem. No of subcarriers needed increases and additional ckty is required both at Txing and Rxing end to handle each added channel these problems are eliminated to a great extend by using TDM



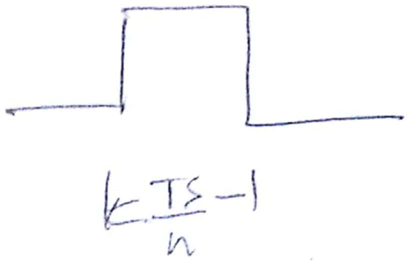


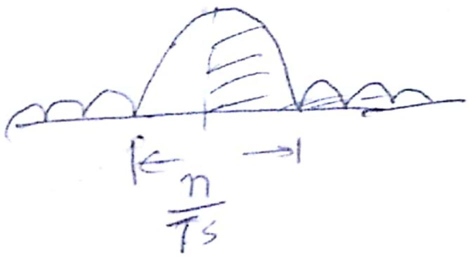
Ts sec $\longrightarrow$ n samples

1 sec $\longrightarrow$ $\frac{n}{Ts}$ samples

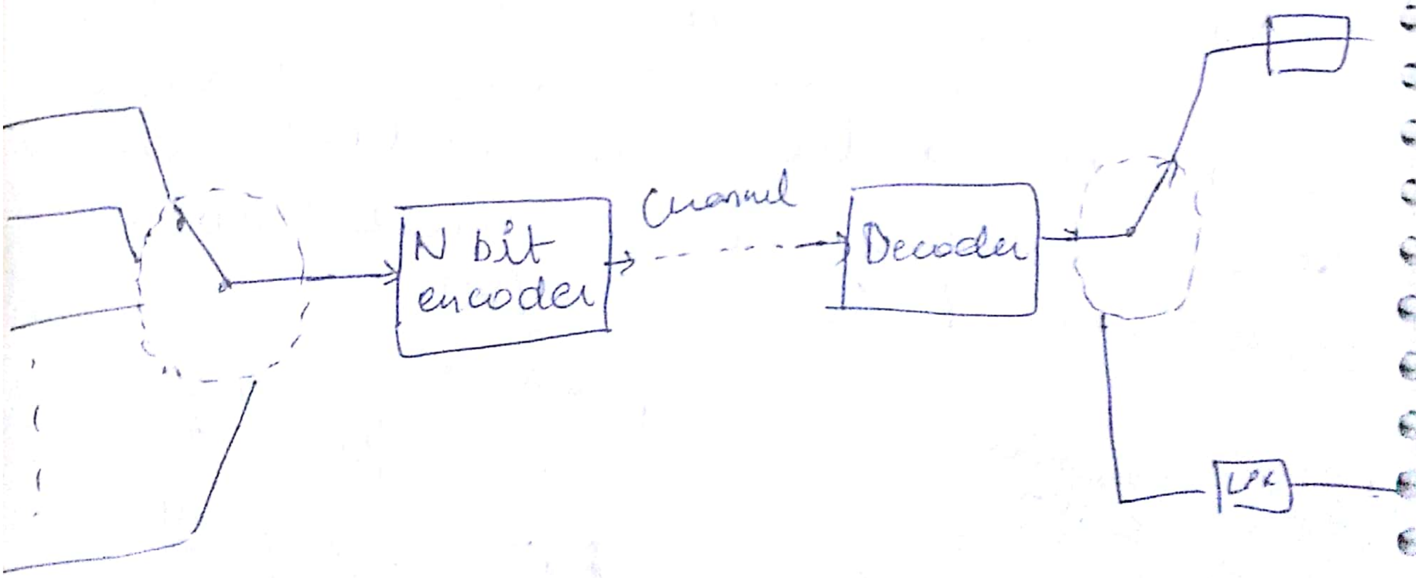
Samples Rate = nfs samples/sec

Max Sample width = $\frac{Ts}{n}$ sec





$$\boxed{BW \geq \frac{nfs}{2}}$$



1 sample $\longrightarrow$ N bit

n samples $\longrightarrow$ nN bits

T sec $\longrightarrow$ nN bits

1 sec $\longrightarrow$ $\frac{nN}{T_s}$ bits/sec

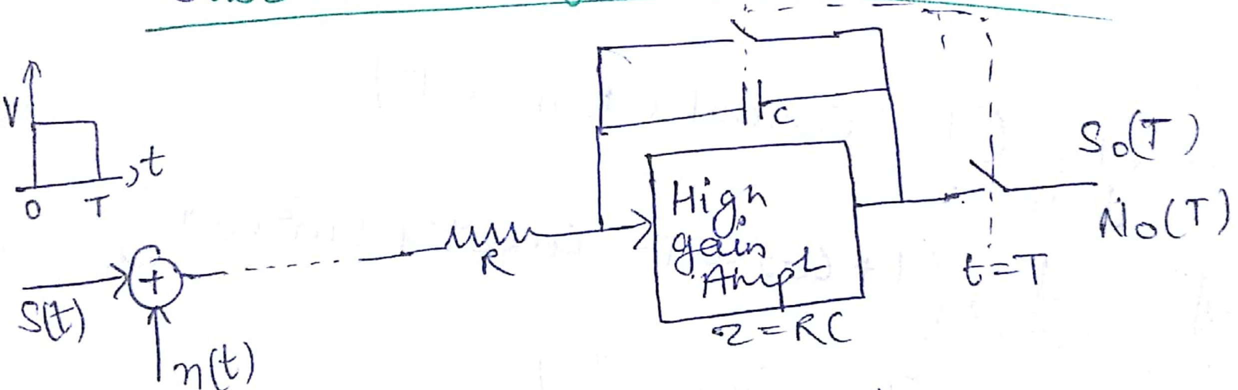
Bit rate $= nNf_s$ bits/sec

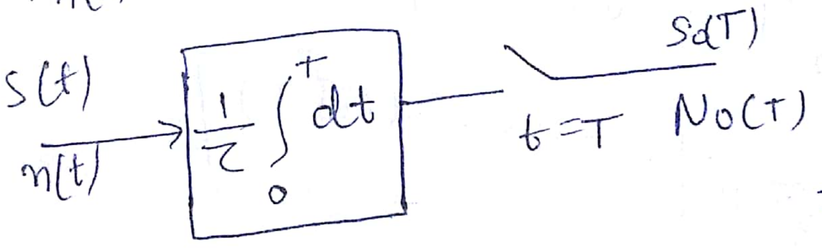
max pulse width $\longrightarrow \frac{T_s}{nN}$

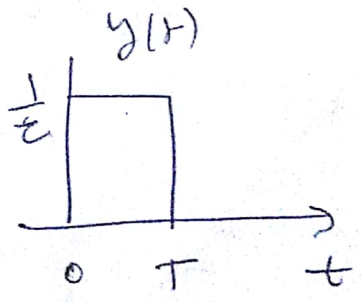
$= \frac{1}{nNf_s}$ sec

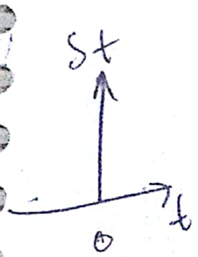
$$\boxed{BW \geq \frac{nNf_s}{2}}$$

Baseband Signal Receiver









$$S_o(T) = \frac{1}{\tau}\int_0^T V\,dt = \frac{V}{\tau}\int_0^T dt = \frac{VT}{\tau}$$