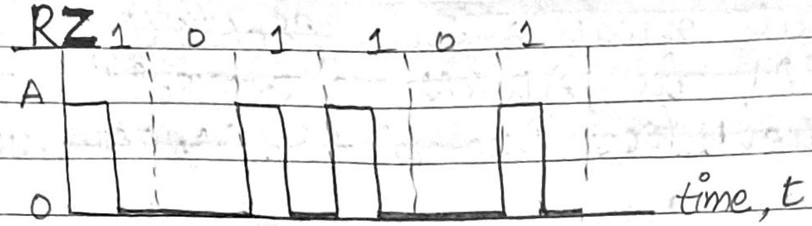
Line Codes

- Unipolar
  - RZ
  - NRZ
- Polar
  - RZ
  - NRZ
  - Manchester
  - Differential Manchester
- Bipolar
  - AMI

Unipolar

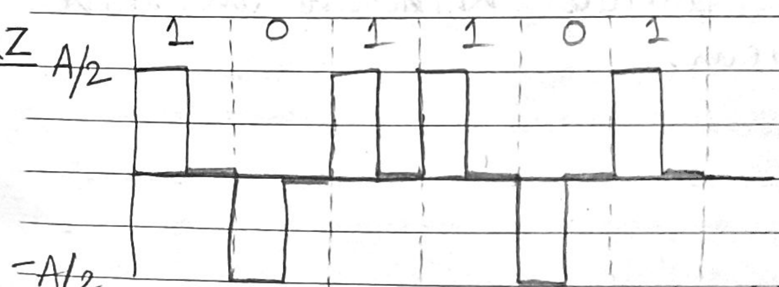
RZ



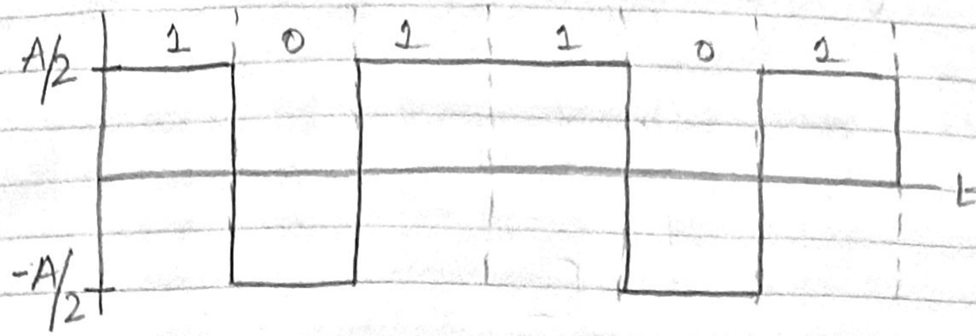
NRZ

Polar

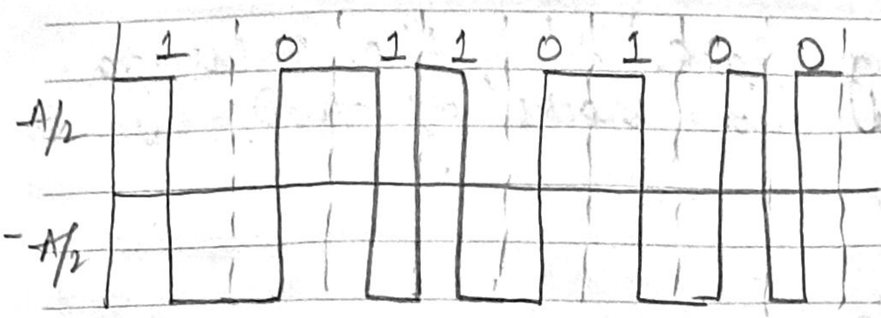
RZ



NRZ

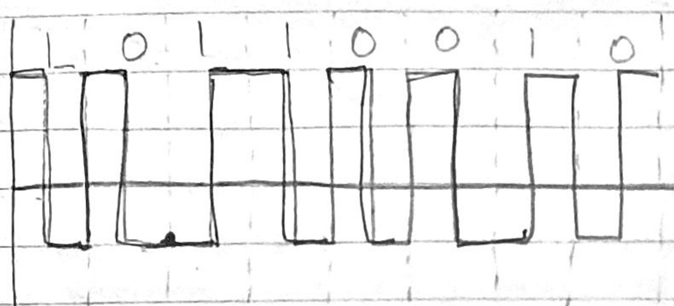


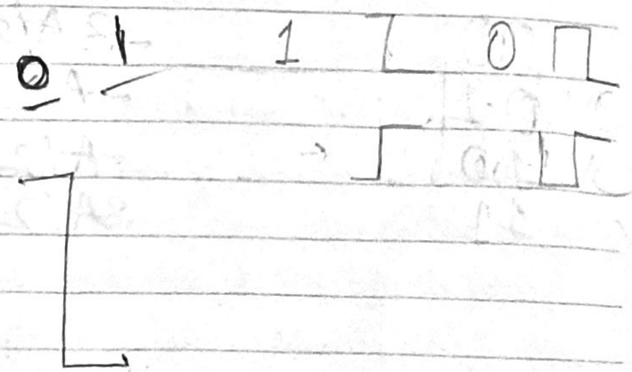
Split phase Mancester Format



1 → +ve to -ve
0 → -ve to +ve

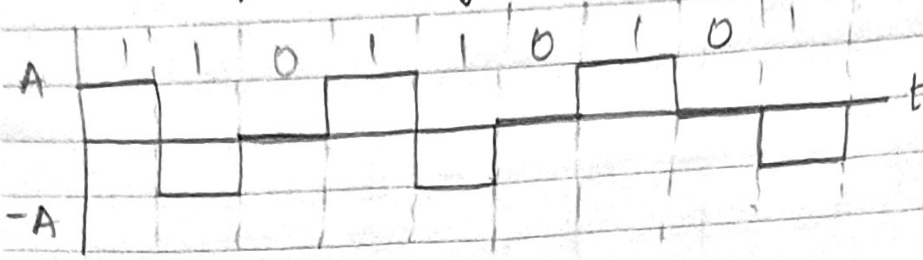
Diff Manchester





Bipolar [Alternate Mark Inversion (AMI)]

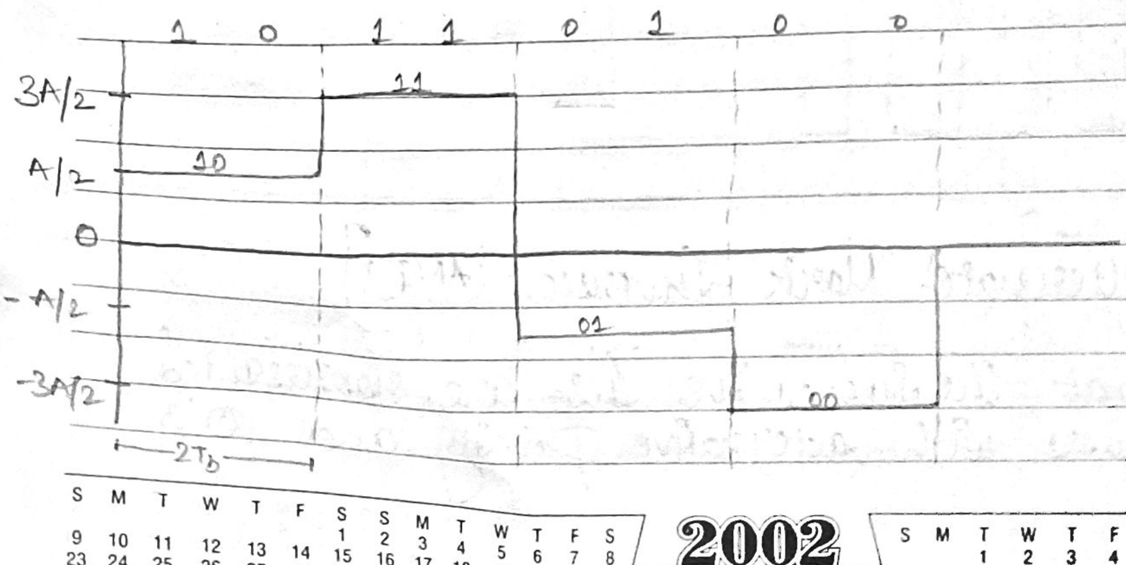
In this format, the Successive 1's are represented by pulses with alternative Polarity and 0's are represented by no pulse



* Polar Quaternary NRZ format.

- To reduce Signalling rate 'r', the message bits are grouped in blocks of two.

Message Com.	$x(t)$
00	$-3A/2$
01	$-A/2$
10	$A/2$
11	$3A/2$



$T_S = 2T_b$

signalling rate

$$r = \frac{1}{2T_b}$$

High density Bipolar (HDB) Signalling.

In NRZ or AMI, the transmitted signal is equal to zero when 0 is to be transmitted. The absence of signal can cause syncronisation problem at the receiver, if long sequence of 0's is transmitted

Sol^n This can be solved by adding pulse when long strings of 0's exceeding a no. n are being transmitted It is HDBN, here $N = 1, 2, 3..$ (Generally 3)

If max no. 0's occur they are replaced by binary sequence of (N+1) length. These sequence contain some binary 1's which are necessary for syncronisation

→ The (N+1) long special sequence for HDB3 are 000V or B00V where B, V are binary 1

Message bit: 1 0 1 1 0 0 0 0 1 1 0 0 0 0 0 0 0 0 0 0 0 1

HDB code: 1 0 1 1 0 0 0 V 1 1 1 0 0 V 1 0 0 V 0 0 1

Bipolar NRZ equivalent

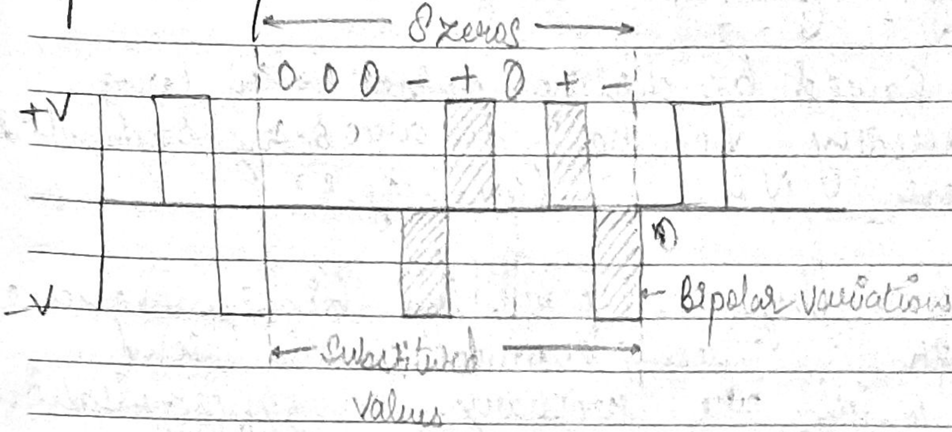
BNZS (Bipolar N-zeros Substitution/Suppression)

- In order to have syncronisation between transmitter and reciever, the line code needs to cross the zero line frequency

- As per U.S. T_1 standard, not more than 15 0's can be sent in succession to ensure proper syncronisation.

Sol → B8ZS (Binary 8 zero suppression)

whenever 8 successive 0's are detected, the implementation of this code will inser 8-bit special sequence



Sampling Theorem

Sampling is a process of converting analog continous signal to the discrete signal.

- A continuous time signal may be completly represent in its sample and recovered back if the sampling frequency is $f_s \geq 2f_m$

f_s = sampling freq. f_m = max. frequency present in signal

Nyquist Rate and Nyquist Interval

Nyquist Rate is called the max sampling frequency

$$f_s = 2f_m$$

Nyquist Interval T_s,

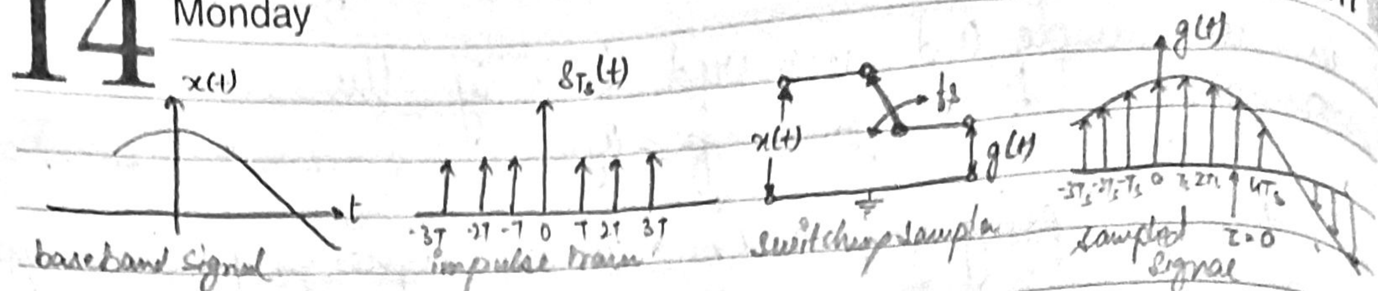
$$T_s = \frac{1}{2f_m} \text{ seconds.}$$

Sampling Tecniques

1. Ideal Sampling
2. Natural Sampling.
3. Flat top Sampling

Ideal Sampling In this the sampling fun. is a train of ~~sampling~~ impulses.

It consist of a circuit to produce instantaneous or ideal sampling



The circuit consist of a switch. If we assume that closing time t of the switch app zero, then o/p of this circuit will contain only instantaneous value of the i/p signal.

$$g(t) = x(t) \cdot \delta_{T_s}(t)$$

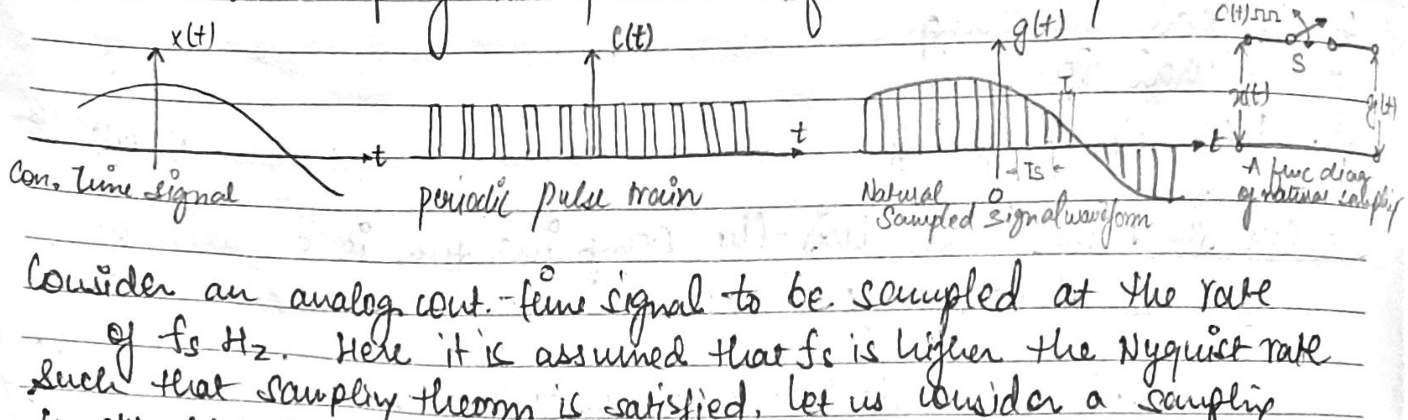
$$g(t) = x(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \Rightarrow \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

The fourier transform of the ideally sampled given by above eqn

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

Natural Sampling

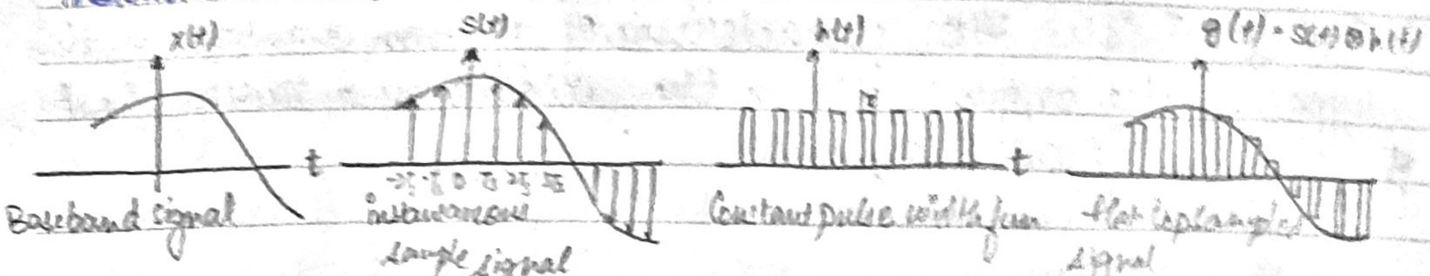
In natural sampling the pulse has finite width equal to τ.



Consider an analog cont.-time signal to be sampled at the rate of fs Hz. Here it is assumed that fs is higher the Nyquist rate such that sampling theorm is satisfied. Let us consider a sampling func c(t) which is a train of periodic pulse of width τ & frequency equal to fs Hz.

Flat top Sampling

In this sampling the top of the samples remain constant and is equal to the instantaneous value of the x(t) at the start of sampling. The width of each sample is τ and sampling rate is equal to $f_s = 1/T_s$. Only starting edge of the pulse represent instantaneous value.



The flat top pulse of g(t) is mathematically equivalent to convolution of instantaneous sample signal s(t) and pulse h(t).

Quantizer

The quantization process can be classified into two types as under

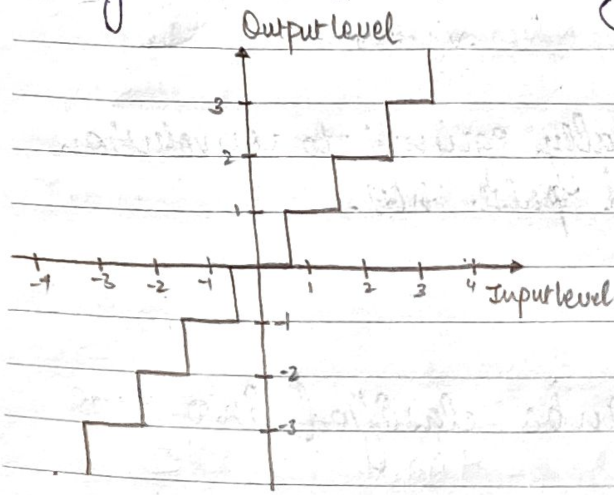
- **Uniform** : In this the step size remains same throughout the input range.
- **Non Uniform** : In this the step size varies according to the input signal values

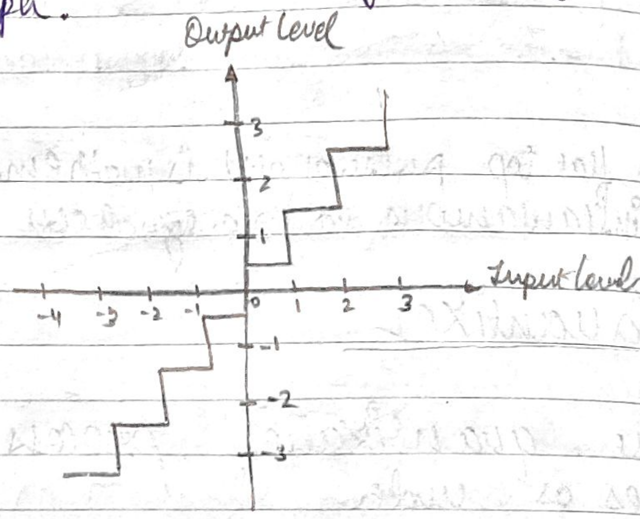
Uniform are of two types.

1. Symmetric quantizer of the midtread type
2. Symmetric quantize of the midrise type

Midtread : The I/O characteristic of uniform quantizer of this type, the origin lies in the middle of a tread of the staircase like graph.

Midrise : The I/O characteristic of uniform quantizer of this type, the origin lies in the middle of a raising part of the staircase like graph.





Non Uniform Quantization

Here the step size changes with change in the amplitude of the input for weak signal step size is small therefore the quantization noise reduces. This is achieved through COMPANDING

COMPANDING

It is non uniform quantisation. It is implement to improve S/N ratio.

$$\boxed{N_q = \frac{\Delta^2}{12}}$$ Quantisation noise

If step size is fixed, then noise power is also fixed but signal power is not constant. It is proportional to the square of signal amplitude $\therefore S \propto A^2$

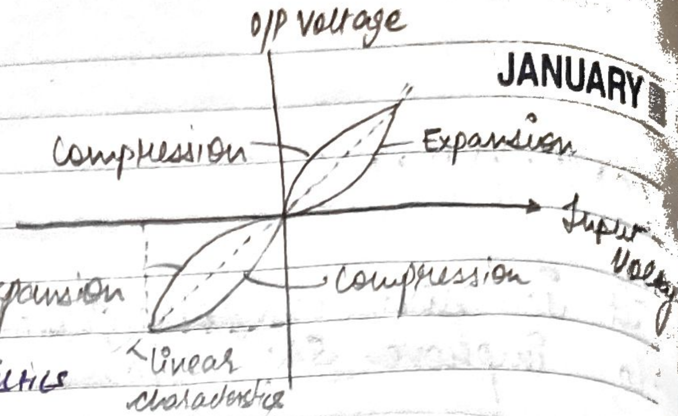
Companding = Compression + Expanding

Generally it is diff to implement non uniform quantisation because it is not known in advance the change in signal level.

Hence,

* Weak signal are amplified and strong signals are attenuated before applying them to a uniform quantiser. This process is called compression.
* At receiver exactly opp is followed which is called expanding.

Input → [Compressor] → [Uniform quantisation] → [Expander] → O/P



Compander characteristic

Due to inverse nature of compression and expansion, the overall characteristics of compander is straight line

Different types of Compressor ckt

* μ-law Companding (midtread)

→ Here the compressor characteristic is continuous. It is approximately linear for small values of I/P level & logarithmic for high I/P level.

→ $$z(x) = (\operatorname{sgn} x) \frac{\ln(1 + \mu |x|/x_{max})}{\ln(1+\mu)}$$

where $0 \le |x|/x_{max} \le 1$

$z(x)$ = output , x = input to the compressor

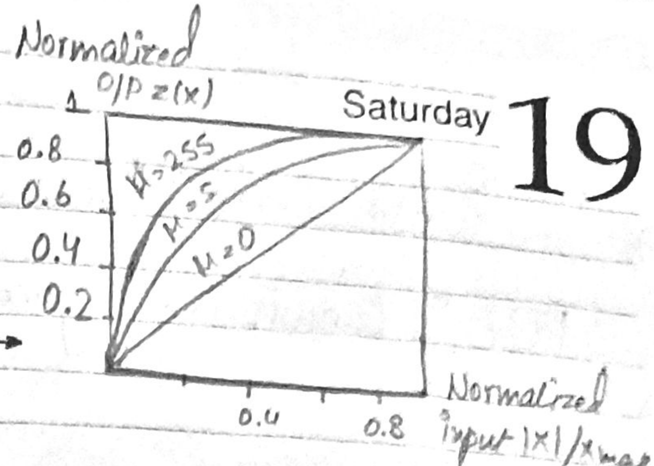
→ $|x|/x_{max}$ represent normalized value of I/P w.r.t the max value of x_{max}.

→ ($\operatorname{sgn} x$) represent ±1 (+ve and -ve value of I/O)

Practically used value for $\mu = 255$

If $\mu = 0$, uniform quantisation

Compressor characteristics of μ-law compressor →



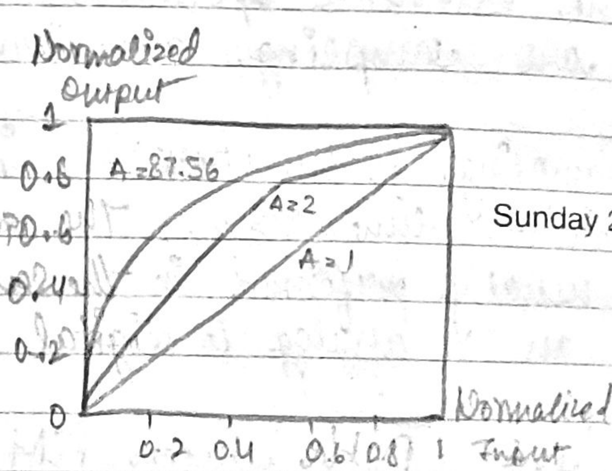
* A law Companding

→ The compressor characteristic is piecewise, made up of linear segment for low value I/P and logarithmic for high value I/P

→ $$\frac{z(x)}{x_{max}} = \begin{cases} \dfrac{A|x|/x_{max}}{1+\log_e A} & \text{for } 0 \le \dfrac{|x|}{x_{max}} \le 1 \\ \dfrac{1+\log_e[A|x|/x_{max}]}{1+\log_e A} & \text{for } \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases}$$

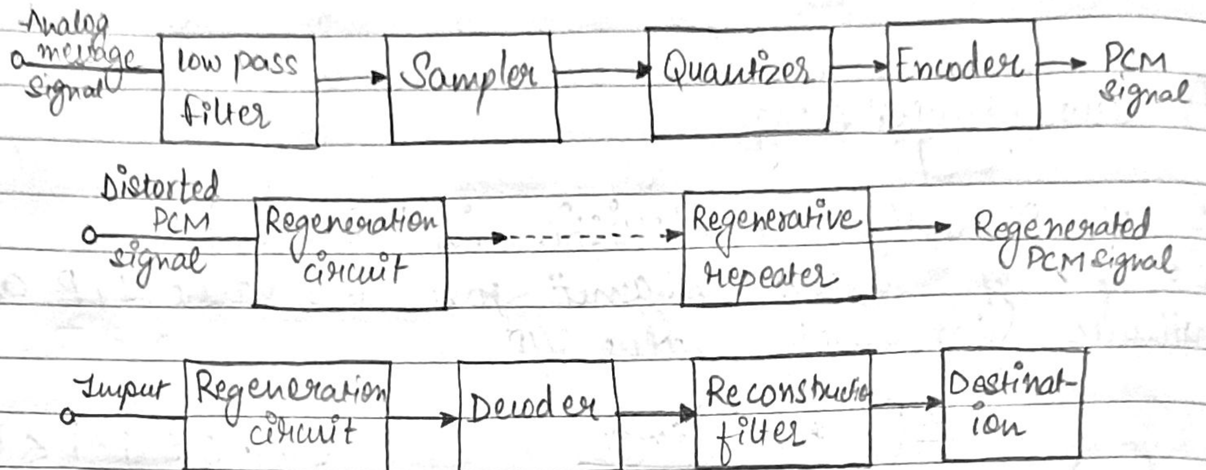
→ A = 1, characteristics is linear which correspons to uniform quantization

Compressor characteristics of A-law compressor



* Pulse Code Modulation (PCM)

PCM is known as digital pulse modulation technique



(a) Transmitter (b) Transmission path (c) Receiver

It consist of three main parts, transmitter, transmission path and receiver.

The essential operation in the transmitter of a PCM system are sampling quantizing and enconding.

Sampling is the operation in which an analog signal is in a discrete time signal. The quantizing and encoding operations are usually performed in the same circuit. Which is known as an analog-to digital conveter (ADC) sampled acc to sampling theorem to resulting

The path between PCM transmitter & PCM reciver over which the PCM diagram of PCM reciver is shown in fig.

The essential operation in the receiver and regeneration of impaired signal, decoding & demodulation of the train of quantizer samples. These opp operation are usually performed in the same circuit which is a digital -to analog converter (DAC)

* Transmission bandwidth in a PCM system.

let us assume that the quantizer use 'v' no. of binary digit.

$$q = 2^v$$

q = total no. of digital levels of a q-level quantizer

Signaling rate in PCM ⇒ $r = v f_s$

where $f_s \geq 2 f_m$

Since BW needed in PCM is 1/2 of the signaling rate,

$$BW \geq \frac{1}{2} r$$

$$r = v f_s$$

$$BW \geq \frac{1}{2} v f_s$$

$$BW = v f_m$$

Quantization Noise / Error in PCM

quantisation error

$$E = x_q(nT_s) - x(nT_s)$$

assume that the I/P $x(nT_s)$ has continuous amplitude in the range $-x_{max}$ to $+x_{max}$

So $+x_{max} = T/2\,\Delta$ $\quad -x_{max} = -T/2\,\Delta$

* Total amplitude range $= x_{max}(-(-x_{max}))$
$= 2x_{max}$

if total amplitude range is divided into q level of quantizer then step size(Δ),

$$\Delta = \frac{x_{max} - (-x_{max})}{q} = \frac{2x_{max}}{q}$$

* <u>Signal to quantization noise ratio for linear quantization.</u>

$$S/N = \frac{\text{Normalized signal power}}{\text{Normalized noise power}}$$

but $N = \dfrac{\Delta^2}{12}$ $\quad \therefore N = \text{Noise power}$

So $\dfrac{S}{N} = \dfrac{S}{\Delta^2/12}$

quantisation level

$$q = 2^{v} \quad \ldots v = \text{no. of bits}$$

Let us assume that $\dot{O}/P$ has continuous amplitude in range $-X_{max}$ to $+X_{max}$

$$\text{total amplitude} = x_{max} - (-x_{max})$$
$$= 2x_{max}$$

Step size will be

$$\Delta = \frac{2x_{max}}{q} \quad \text{or} \quad = \frac{2x_{max}}{2^{v}}$$

So,
$$\frac{S}{N} = \frac{S}{\Delta^2/12} \Rightarrow \frac{\text{Signal Power}}{\left(\frac{2x_{max}}{2^{v}}\right)^2 \cdot \frac{1}{12}}$$

normalized Signal power is denoted as 'P'

$$\frac{S}{N} = \frac{P}{\frac{4x^2_{max}}{2^{2v}} \times \frac{1}{12}} \Rightarrow \frac{3P}{x^2_{max}} \cdot 2^{2v}$$

This exp. shows that signal to noise power ratio of quantizer increases exponentially with inc. bit per sample

Case If $x_{max} = 1$ ($x(t)$ is normalised)

$$\frac{S}{N} = 3P \times 2^{2v}$$

If destination signal power 'P' is normalised

P≤1

then S/N ratio will be

$$\frac{S}{N} \le 3 \times 2^{2v}$$

S/N ratio in decibels,

$$\frac{S}{N}\,dB = 10 \log e\left(\frac{S}{N}\right) \le 10 \log_{10}\left[3 \times 2^{2v}\right] \le (4.8 + 6v)\,dB$$

$$\boxed{\frac{S}{N}\,dB \le (4.8 + 6v)\,dB}$$

Probability of error.

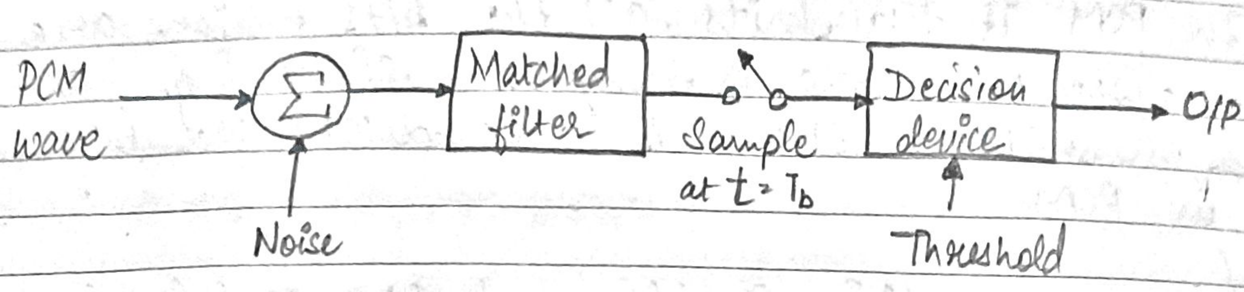
① Channel Noise / Decoding Noise

② Quantisation Noise (produced at transmitter).

Channel noise The introduction of transmission error at the receiver when the PCM signal is being reconstructed.

Due to such error, the receiver will take mistake in making the decision about whether a 0 was received or a 1 was received.

② Probability of error: It gives the probability that the symbol at receiver will make o/p diff from that transmitted.



$$P_e = \frac{1}{2}\, erfc\left[\frac{1}{2}\sqrt{\frac{E_{max}}{N_0}}\right]$$

E_{max} = peak signal energy
N_0 = Noise spectral density

Put $E_{max} = P_{max} T_b$

where P_{max} = max peak signal power
T_b = bit duration

$$P_e = \frac{1}{2}\, erfc\left[\frac{1}{2}\sqrt{\frac{P_{max} \times T_b}{N_0}}\right]$$

where N_0/T_b is avg. noise power contained in BW

* Delta Modulation

→ In PCM it transmits all the bits which are used to code a sample, signaling rate & transmission channel bandwith are large in PCM

→ To overcome this problem – Delta Modulation

→ DM transmit only one bit per sample

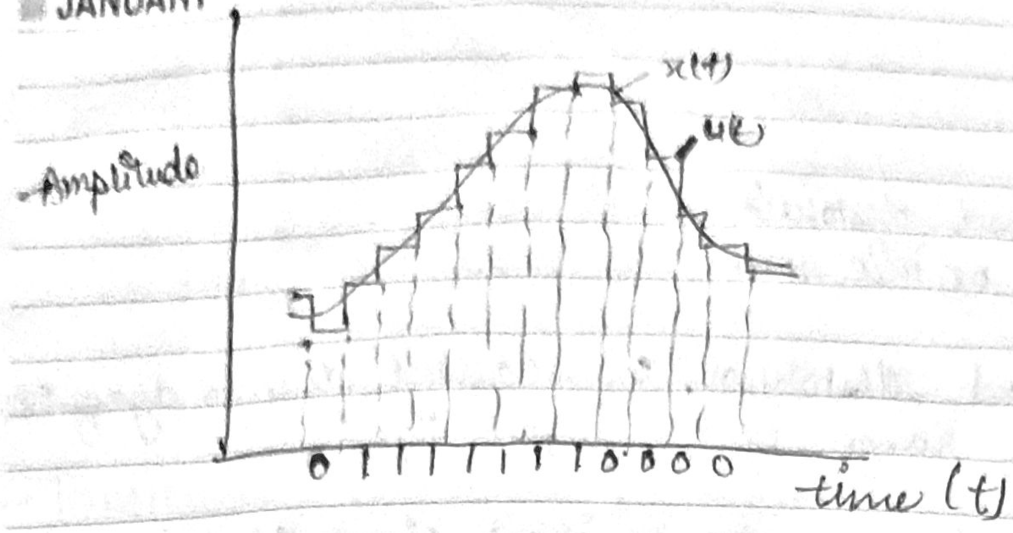
→ In this the present sample value is compared with previous sample value and its result whether the amplitude is increased or decreased is transmitted.

→ I/P signal is approximated to step signal by delta modula[tion] the step size is fixed

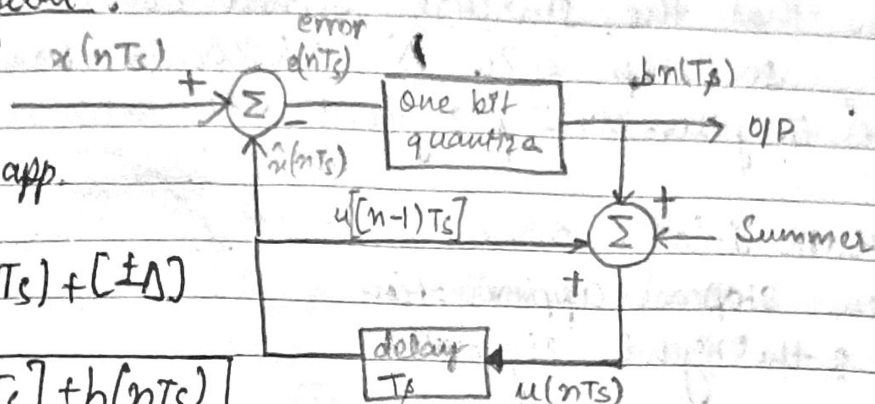
→ The step size difference between I/P signal and stai[r] approximated signal is confined to 2 level +Δ

→ If diff is +ve the approximate signal is increa[sed] by one step 'Δ'. diff -ve → decrease

→ When step size is reduced '0' is transmitted if step is increase '1' is transmitted



Transmitter part.



→ The present sample app.

$u(nT_s) = u(nT_s - T_s) + [\pm\Delta]$

$\boxed{u(nT_s) = u[(n-1)T_s] + b(nT_s)}$

→ Depending upon sign of $e(nT_s)$, Quantiser generates an O/P of $+\Delta$ or $-\Delta$
$\quad\quad\quad\quad\quad\quad\quad\quad\quad$ 1 $\quad\quad$ 0

Advantages ① As only one bit per sample is transmitted therefore signaling rate and transmission channel Bandwidth is quite small

② Transmitter and receiver implementation is very much simple

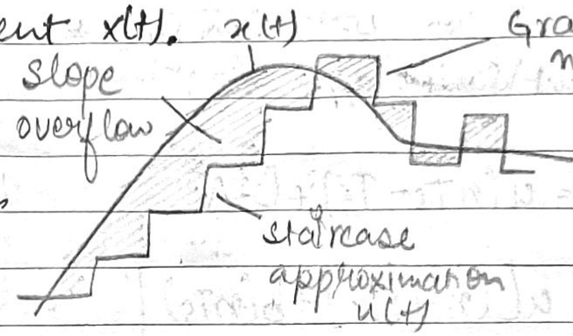
Disadvantages.

1) Slope overload destortion
2) Granular or idle noise

1) Slope overload destortion : caused due to dynamic range of the input signal.

- Sometimes the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it so step size 'Δ' becomes too small for u(t) to follow the step segment x(t).

- Hence there is large error between stepcase approximation signal & the original signal x(t)



2) Granular noise : occurs when the step size is too large compare to small variable in input signal

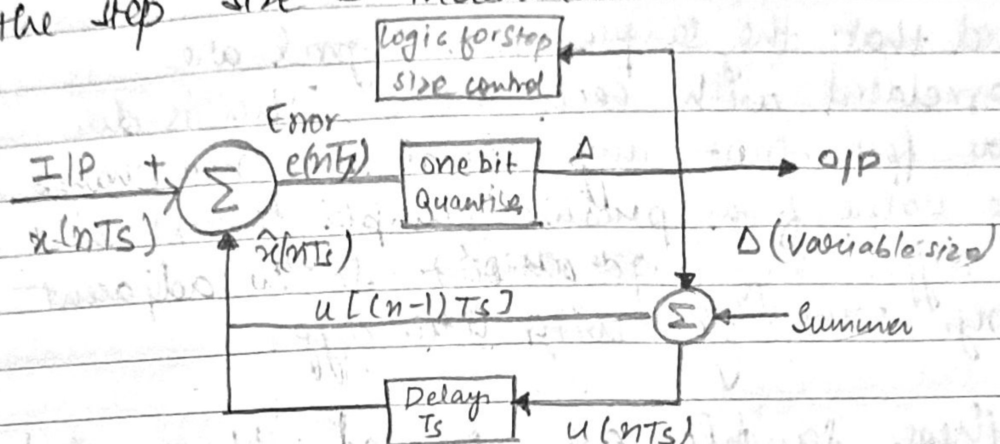
So for small variable in I/P signal the staircase signal is changed by large amount (Δ) becomes of large step size.

Adaptive Delta Modulation

- To overcome quantisation error due to slope overload distortion and granular noise, the step size (Δ) is made adaptive to variations in the input signal x(t)

→ If I/P is varying slowly, the step size is reduced

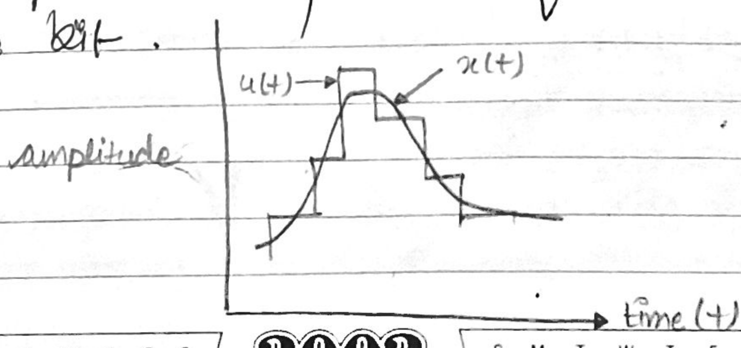
→ Particularly in the steep segment of the signal x(t) the step size is increased.



→ The logic for step size control is added in the diagram

→ The Δ will inc or dec by specified rule depending on one bit quantiser output

→ In this step size is produced from each incoming bit.



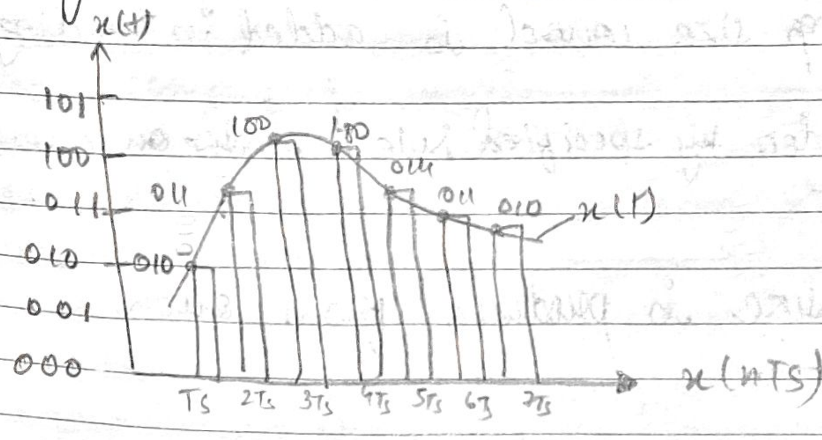
Advantage ① S/N ratio becomes better
② dynamic range of ADM is wider than simple DM
③ utilization of B/W is better than DM

Differential Pulse code Modulation

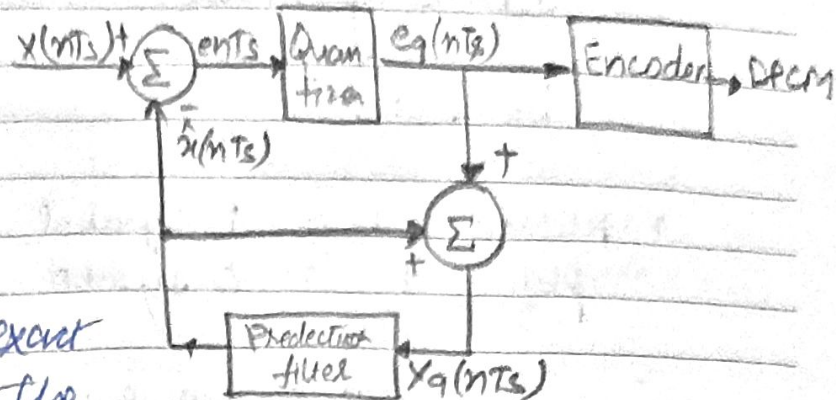
→ It is observed that the samples of a signal are highly correlated with each other. This is due to the fact that any signal does not change fast. So the value from present sample to next sample does not differ by large amount so the adjacent sample carry same info with little diff.

↓

So when these samples are encoded, the encoded signal contain some redundant info.



It works on principle of prediction. The value of present sample is predicted from the past samples.



The prediction is not exact but it is close to the actual sample value.

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

Quantiser o/p

$$e_q(nT_s) = e(nT_s) + q(nT_s)$$

$\leftarrow$ quantization error

$$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s)$$

$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$

but error

$$e(nT_s) + \hat{x}(nT_s) = x(nT_s)$$

So final value

$$\boxed{x_q(nT_s) = x(nT_s) + q(nT_s)}$$

S/N ratio for DPCM

$$SNR = \frac{\text{Mean square value of signal}}{\text{" " " of quantisation noise}}$$

$= \sigma_X^2 / \sigma_Q^2$ → ~~varing~~ variance of quantisation nois

variance of original I/P signal

$$SNR = \frac{\sigma_X^2}{\sigma_E^2} \times \frac{\sigma_E^2}{\sigma_Q^2}$$

σ_E^2 = variance of prediction error

Duo Binary Pulse

b_k = I/P

amplitude +1 = 1 symbol
amplitu −1 = 0 symbo

⇓

duo binary encoder

3 level o/p (c_k)
where c_k is sum of present binary digit b_k & it previous value

$C_k = +2 \longrightarrow b_k = b_{k-1} = 1$

$0 \rightarrow b_k \neq b_{k-1}$

$-2 = b_k = b_{k-1} = 0$