

- Sampling Theory
 - process \Rightarrow continuous time \rightarrow discrete time
 - no. of samples depends on max freq component of signal

Sampling Theorem

- A band limited signal of finite energy which has no freq component higher than $f_m(H_3)$ is completely by its sample values at uniform interval less than or equal to $\frac{1}{2f_m}$

$$T_s \leq \frac{1}{2f_m}$$

or

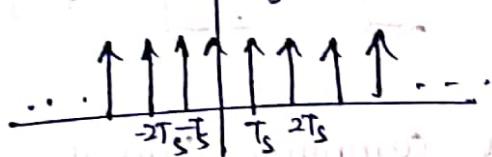
$$f_s \geq 2f_m$$

Note: if signal is band limited to f_m then,

$$x(w) = 0 \text{ for } w > w_m$$



$$\delta_{T_s}(t) = \frac{1}{T_s} (1 + 2\cos w_s t + 2\cos 2w_s t + \dots)$$

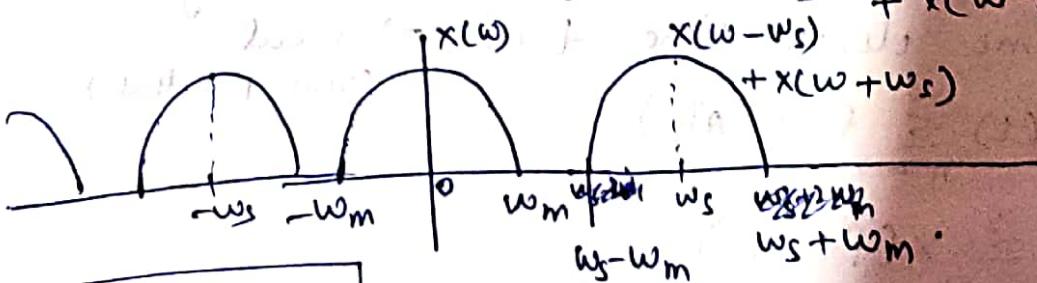


$$x(t) \quad \text{MULTIPLIER} \quad g(t) = x(t) \delta_{T_s}(t)$$

$$g(t) \triangleq$$

$$g(t) = x(t) \left[\frac{1}{T_s} (1 + 2\cos w_s t + 2\cos 2w_s t \dots) \right]$$

$$g(w) = \frac{1}{T_s} (x(w) + x(w-w_s) + x(w+w_s) + x(w-2w_s) + x(w+2w_s) + \dots)$$



$$w_s \geq 2w_m$$

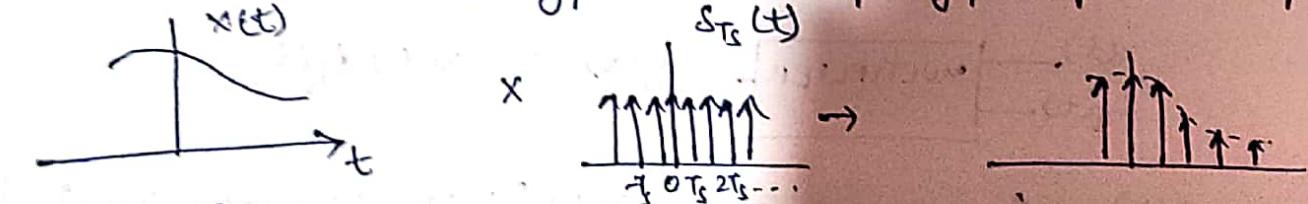
- * if $f_s > 2f_m \rightarrow$ no overlapping and periodically repetition.
- * At receiver end we place low pass filter of frequency ω_m so we can extract original information.
- $f_s > 2f_m \rightarrow$ no overlapping
- $f_s = 2f_m \rightarrow$ touch (Nyquist rate)
- $f_s < 2f_m \rightarrow$ overlapping (distortion)
- $T_s = \frac{1}{2f_m} \rightarrow$ Nyquist time interval.

Effect of undersampling \rightarrow Aliasing

If $f_s < 2f_m$ when we apply LPF we won't be able to extract og sample bcos of aliasing (overlapping).

- 1) If we go for sampling, we pass original signal through LPF. this is even referred to as pre alias filter other name is band limit filter. Avoid aliasing
- or $f_s > 2f_m$

Instantaneous sampling / Ideal sampling / Impulse sampling



$$s_{T_s}(t) = \sum_{n=-\infty}^{\infty} s(t - nT_s)$$

\hookrightarrow To generate ideal samples we use switching sampler

\hookrightarrow If we assume closing time $t \rightarrow 0 \Rightarrow$ ideal (not practical)

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

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

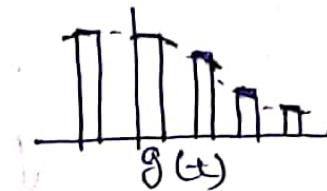
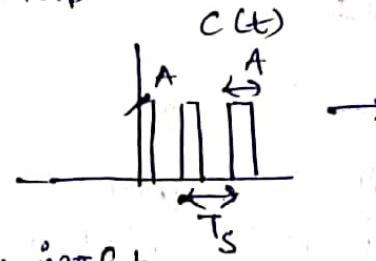
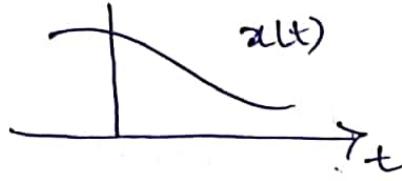
demonit

\hookrightarrow not practical

\hookrightarrow high noise interference. (bcos signal energy is low)

Natural Sampling

↳ It uses chopping principle



$$c(t) = \frac{2A}{T_s} \sum_{n=-\infty}^{\infty} \sin(f_n t) e^{j2\pi f_n t}$$

$$g(t) = x(t) * c(t)$$

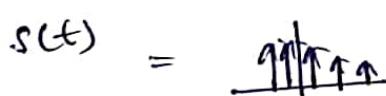
$$g(t) = \frac{2A}{T_s} \sum_{n=-\infty}^{\infty} x(t) \sin(f_n t) e^{j2\pi f_n t}$$

$$G(\omega) = \frac{2A}{T_s} \sum_{n=-\infty}^{\infty} \sin((nf_s - \omega)t) \times (f - nf_s)$$

This method is used practically
noise interference is less.

flat top sampling

- ↳ uses sample and hold circuit
- ↳ flat top is easier than natural.
- ↳ has high noise interference.

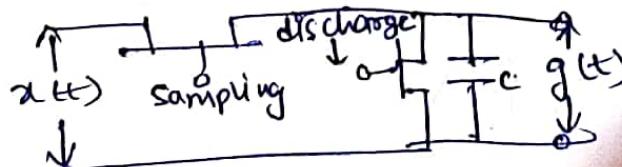


$$g(t) = s(t) * h(t)$$

$$= \sum_{n=-\infty}^{\infty} x(t) h(t - nT_s)$$

$$G(\omega) = \sum_{n=-\infty}^{\infty} x(f - n f_s) H(f)$$

Same as PAM (pulse amplitude modulation).



time domain \rightarrow frequency domain
using fourier transform

FT exist for \rightarrow Energy signal

Power signal

impulse related signal

using
property

integrable

$$x(t) \Leftrightarrow X(j\omega) \text{ or } X(f)$$

rad/s

Hz

$$X(j\omega) = \int_{-\infty}^{\infty} x(t) \cdot e^{-j\omega t} dt$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(j\omega) e^{j\omega t} d\omega$$

properties

1) linearity

$$x_1(t) \Leftrightarrow X_1(j\omega) \quad x_2(t) \Leftrightarrow X_2(j\omega)$$

$$ax_1(t) + bx_2(t) \Leftrightarrow aX_1(j\omega) + bX_2(j\omega)$$

2) conjugation

$$x(t) \Leftrightarrow X(j\omega)$$

$$x^*(t) \Leftrightarrow X^*(j\omega)$$

$$3) \text{ Area under } x(t) = X(j\omega)|_{\omega=0}$$

$$4) \text{ Area under } X(j\omega) = 2\pi(x(t))|_{t=0}$$

5) time reversal

$$x(t) \Leftrightarrow X(j\omega)$$

$$x(-t) \Leftrightarrow X(-j\omega)$$

6) time scaling

$$x(at) \Leftrightarrow \frac{1}{|a|} X(j\omega/a)$$

7) time shifting

$$x(t \pm t_0) \Leftrightarrow X(j\omega) e^{\pm j\omega t_0}$$

8) frequency shifting.

$$X[j(\omega + \omega_0)] \Leftrightarrow e^{\pm j\omega_0 t} x(t)$$

9) convolution in time

$$x_1(t) \Leftrightarrow X_1(j\omega)$$

$$x_2(t) \Leftrightarrow X_2(j\omega)$$

$$x_1(t) * x_2(t) \Leftrightarrow X_1(j\omega) \cdot X_2(j\omega)$$

$$x_1(t) * x_2(t) = \int_{-\infty}^{\infty} x_1(z) \cdot x_2(t-z) dz$$

10) multiplication in time / convolution in freq

$$x_1(t) \cdot x_2(t) \Leftrightarrow \underbrace{[X_1(j\omega) * X_2(j\omega)]}_{j\pi}$$

11) differentiation in time

$$\frac{d(x(t))}{dt} = (j\omega) X(j\omega)$$

$$\frac{d^n(x(t))}{dt^n} = (j\omega)^n X(j\omega)$$

12) integration in time

$$x(t) \Leftrightarrow X(j\omega)$$

$$\int_{-\infty}^z x(z) dz \Leftrightarrow \frac{X(j\omega)}{j\omega} + \pi X(0) \delta(\omega)$$

13) differentiation in frequency

$$t \cdot x(t) = j \frac{dX(j\omega)}{d\omega}$$

$$t^n x(t) = j^n \frac{d^n X(j\omega)}{d\omega^n}$$

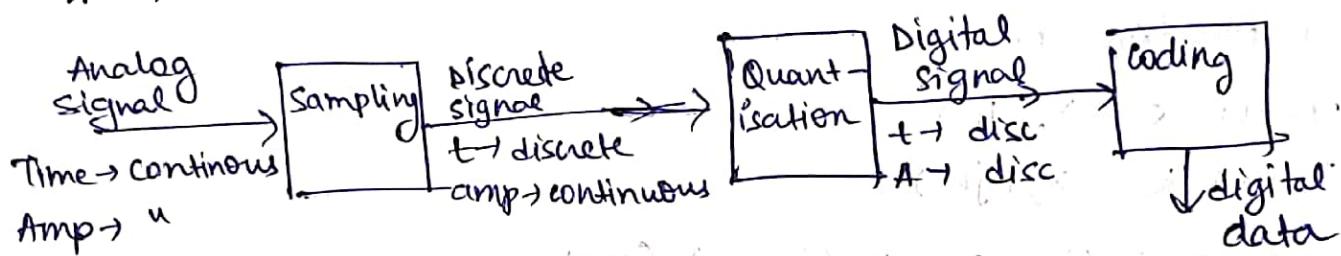
14) Modulation

$$x(t) \rightleftharpoons X(\omega) \text{ or } X(j\omega)$$

$$\text{i)} x(t) \cdot \cos \omega_0 t \rightleftharpoons \frac{1}{2} [X(\omega + \omega_0) + X(\omega - \omega_0)]$$

$$\text{ii)} x(t) \cdot \sin \omega_0 t \rightleftharpoons \frac{j}{2} [X(\omega + \omega_0) - X(\omega - \omega_0)]$$

Quantisation

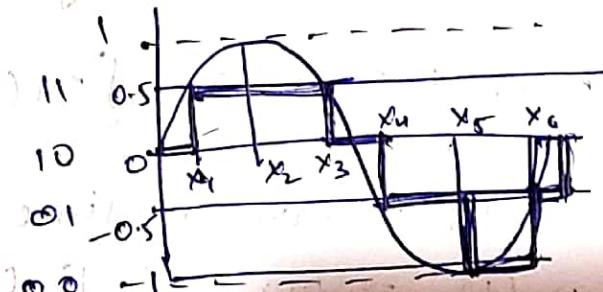


$$\text{no. of bits} = n = 2 \text{ (say)}$$

$$\text{u " levels} = 2^n \Rightarrow 4$$

$$\text{Step size } \Delta = \frac{A_{\max} - A_{\min}}{L}$$

$$= 0.5$$



points	x_1	x_2	x_3	x_4	x_5	x_6	x_7
sampling	0.5	0.8	0.6	-0.6	-0.9	-0.5	
quantisation	0.5	0.5	0.5	0.5	1	0.5	
quantisation error	0.07	0.3	0.1	(0.1)	0.1	0	

features of quantisation:

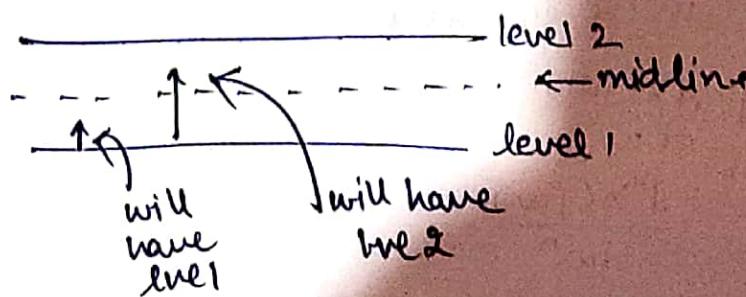
1) continuous → finite set

2) infinite precision → finite precision

3) Rounding off samples to nearest quantised level.

to choose a level of quantification divide the 2 levels and chose the level according to region

Ex:



dynamic range of quantisation

↳ ratio of largest to smallest measurable amplitude.

$$= 20 \log \left[\frac{\text{largest Amp}}{\text{smallest Amp}} \right]$$

$$= 20 \log \left[\frac{2^{n-1} \Delta}{2^1 \Delta} \right] = 20 \log 2^n = 20n \log 2$$

$$= \cancel{6.02n} \quad \underline{6.02n}$$

SNR of quantisation

$$\text{SNR} = 20 \log \left[\frac{\text{signal rms voltage}}{\text{noise rms voltage}} \right]$$

$$q_{\text{e}}^2 = \frac{1}{\Delta} \int_{-\Delta/2}^{\Delta/2} q^2 dq = \frac{1}{\Delta} \left[\frac{q^3}{3} \right]_{-\Delta/2}^{\Delta/2} = \frac{\Delta^2}{12}$$

$$q_{\text{e}} = \frac{\Delta}{2\sqrt{3}} \quad \leftarrow \text{noise rms}$$

$$\text{full scale signal} = V_{\max} - V_{\min} = V_{fs}$$

$$\text{Peak Voltage} = V_f/2$$

$$\text{Rms of Peak Voltage} = \frac{V_f}{2\sqrt{2}} = \frac{2^n \Delta}{2\sqrt{2}}$$

$$\Delta = \frac{V_{\max} - V_{\min}}{2^n} \Rightarrow V_f = 2^n \Delta$$

$$\text{SNR} = 20 \log \left[\frac{\frac{2^n \Delta}{2\sqrt{2}}}{\frac{\Delta}{2\sqrt{3}}} \right] = 20 \log \left(\frac{2^n \sqrt{3}}{2} \right) = 6.02n + 1.76$$

$$\boxed{\text{SNR} = 6.02n + 1.76} \quad (\text{same for S&NR})$$

Quantisation

uniform (Step size = same)

midrise



midtread



non uniform

Quantisation noise (for uniform step size)

PDF of quantisation error

$$f_q(q) = \begin{cases} \frac{1}{\Delta} & -\frac{\Delta}{2} < q < \frac{\Delta}{2} \\ 0 & \text{else} \end{cases} \quad \left. \begin{array}{l} \text{uniform} \\ \text{distribution} \end{array} \right\}$$

mean of quantisation error

$$= \int_{-\Delta/2}^{\Delta/2} q \left(\frac{1}{\Delta}\right) dq = \frac{1}{\Delta} \left[\frac{q^2}{2} \right]_{-\Delta/2}^{\Delta/2} = 0$$

Variance of quantisation

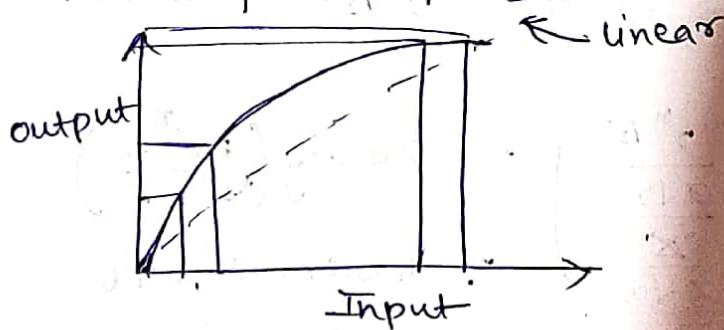
$$E(x^2) = \int_{-\infty}^{\infty} x^2 f(x) dx$$

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

(Same as SNR)

- If step size $\uparrow \rightarrow$ Quantisation noise \uparrow
- If step size $\downarrow \rightarrow$ channel noise \uparrow

- ∴ for large variance \rightarrow large step size
- for small variance \rightarrow small step size.
- ∴ non uniform step size.



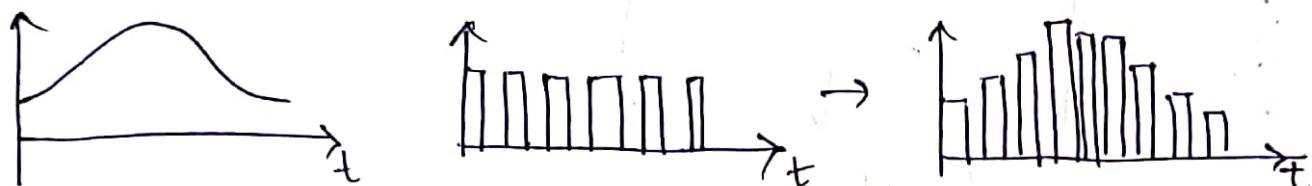
Pulse modulation

↪ Process where one parameter of carrier signal varies according to modulating signal.

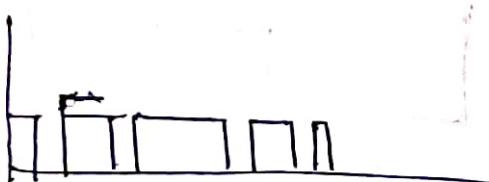
Modulating signal \rightarrow low freq which we want to transfer
carrier signal \rightarrow High freq.

Size of antenna $\rightarrow \frac{\lambda}{4}$.

1) Pulse Amp modulation (flat top)

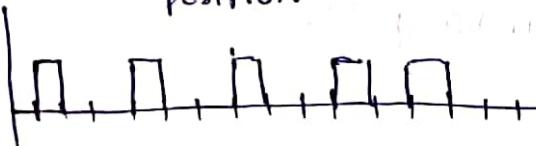


2) Pulse width modulation



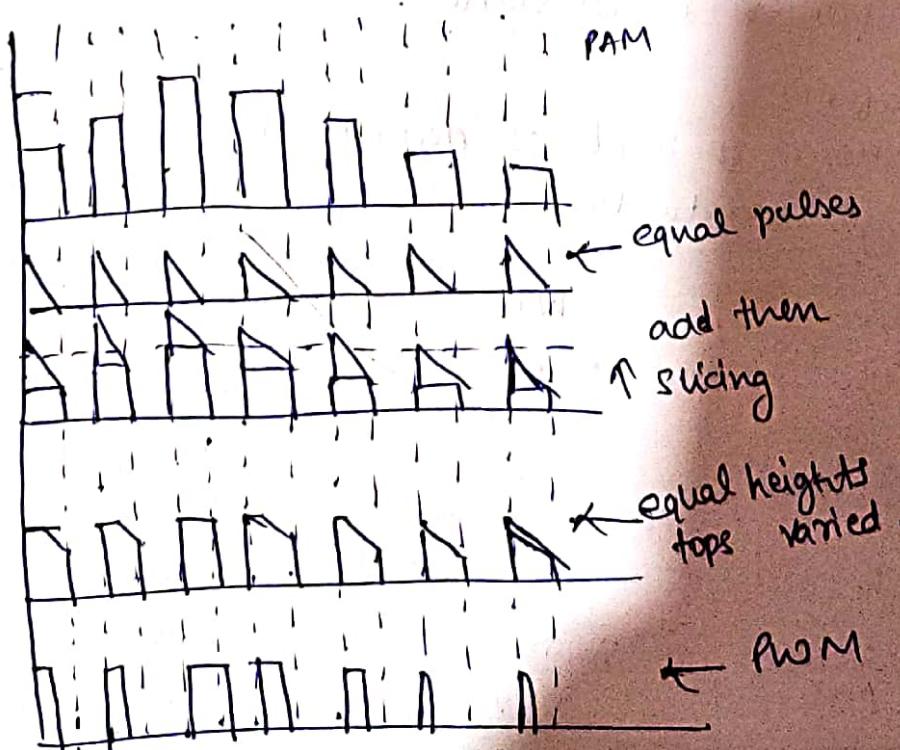
where amp is high
↪ width is high

3) Pulse phase modulation position

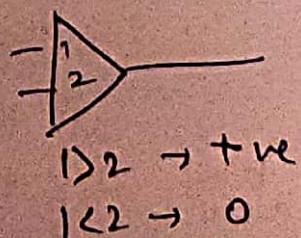


width is same
where the pulse ends in PWM
the pulse rise in PWM and
drop \approx is \approx with same width

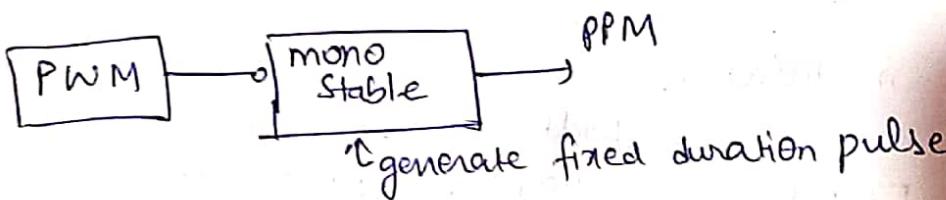
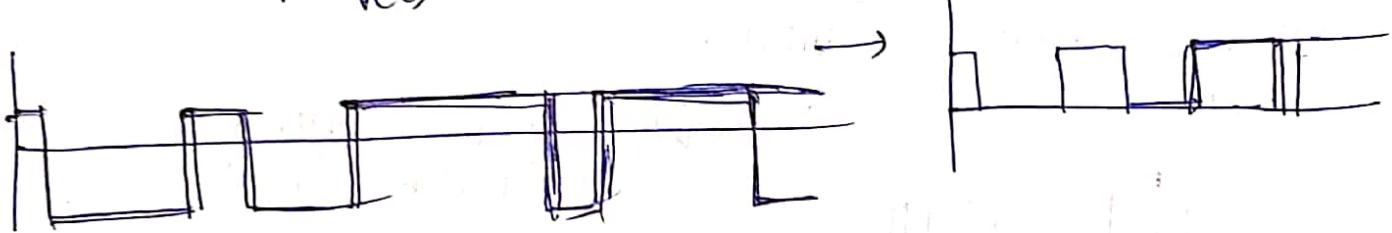
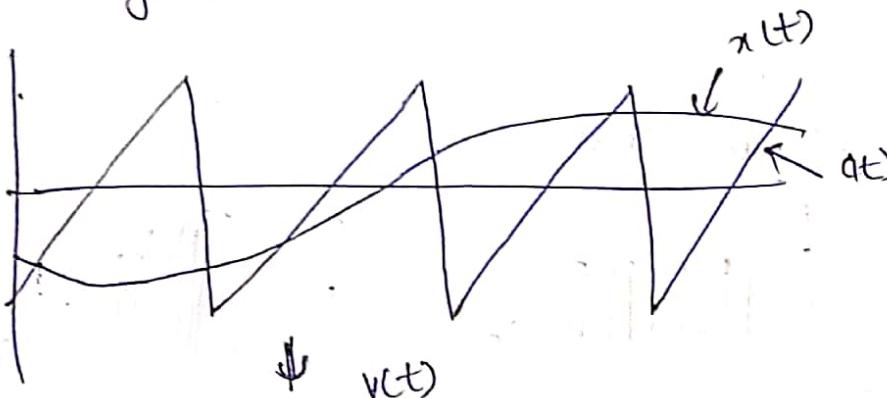
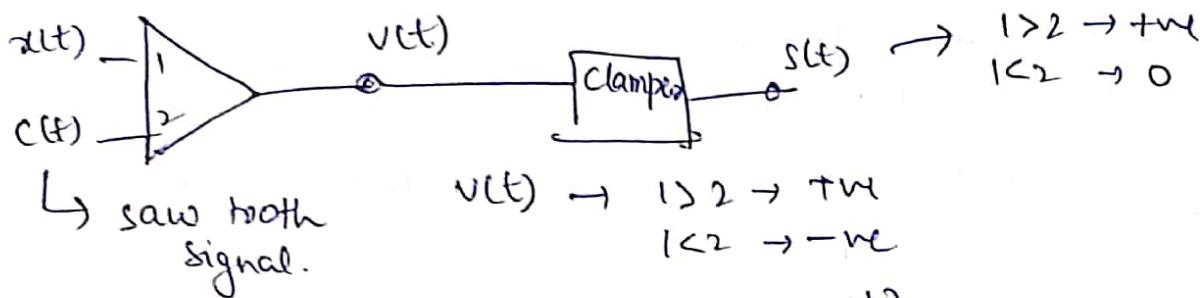
Generation



for slicing
we use
comparator



After.



advantages of PPM

- ↳ less affected by noise
- ↳ noise contaminated can be reconstructed
- ↳ transmitted power is constant.

disadvantages of PPM

- ↳ synchronisation is reqd at demodulation
- ↳ requires high bandwidth.

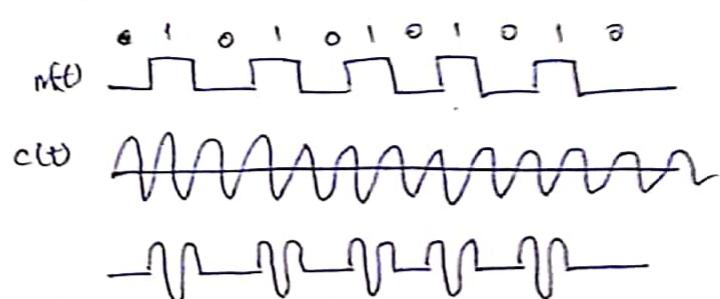
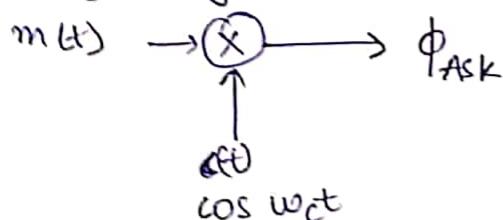
	PPM	PWM	PPM
B.W Bandwidth	width	wrt time	rise time
Transmitted power	wrt time	min	constant
noise	high similar to AM	similar to FM	low Similar to PM

Digital modulation

Digital \rightarrow Analog

1) Amplitude shift keying

Amplitude of carrier signal varies wst amplitude of msg signal.



for Binary, n=1

\hookrightarrow BASK, OOK
on-off keying

Bandwidth:

$$\text{BW} \propto r \text{ (band rate)}$$

$$\text{BW} = (1+d)r$$

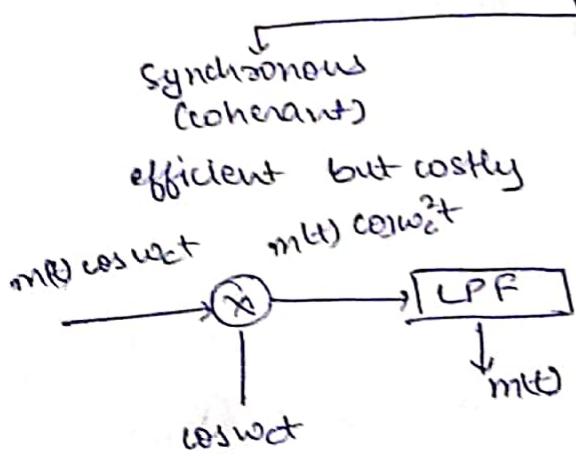
$$\text{BW} = (1+d) \frac{R}{n}$$

} R \rightarrow data rate
n \rightarrow bits reqd for ~~se~~ elements
d \rightarrow factor for modulation and filtering process
 $\in \{0, 1\}$
 \uparrow ideal \uparrow worst

o modulation of ASK (generation of ASK)

$$m(t) \times c(t) \quad m(t) \quad \begin{cases} 1 & \text{+ve} \\ 0 & \text{-ve no voltage} \end{cases}$$

o demodulation ASK



non synchronony
(non coherent)

$$m(t)\cos wct \xrightarrow{\text{envelope}} m(t)$$

low cost
poor performance with
less SNR received
signal.

Transmitted signal.

$$\begin{aligned} x_1(t) &= A \cos 2\pi f_c t && \leftarrow \text{bit 1} \\ x_2(t) &= 0 && \leftarrow \text{bit 2} \end{aligned}$$

E_B = energy per bit

$$\begin{aligned} &= \int_0^{T_B} x_1^2(t) dt = \int_0^{T_B} (A \cos 2\pi f_c t)^2 dt \\ &= \frac{A^2}{2} [T_B - 0] \\ E_B &= \frac{A^2 T_B}{2} \end{aligned}$$

According to GSOP

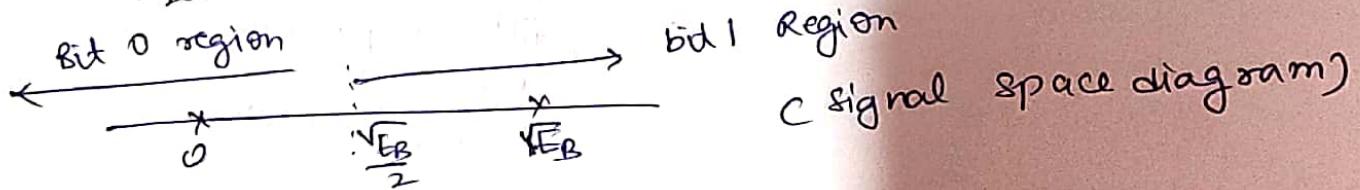
$$\phi_1(t) = \frac{x_1(t)}{\sqrt{E_B}} \Rightarrow \sqrt{\frac{2}{T_B}} \cos 2\pi f_c t$$

$$\phi_2(t) = 0$$

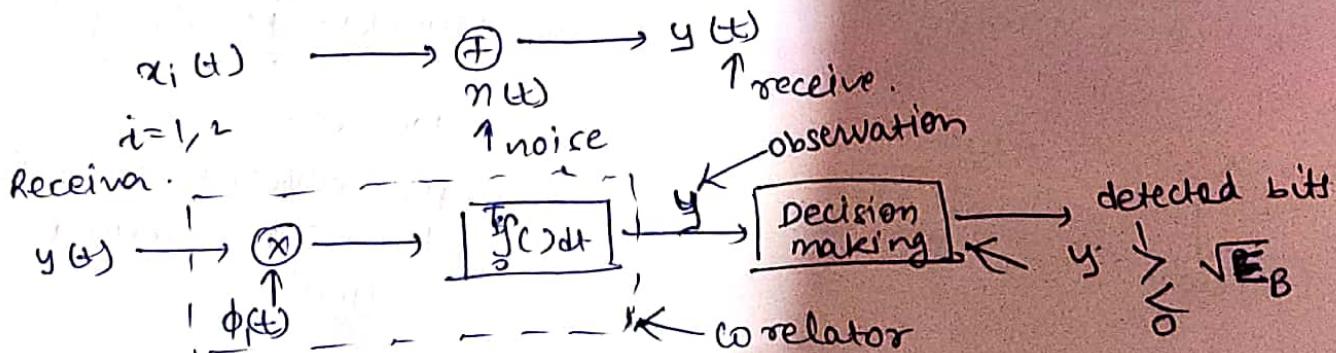
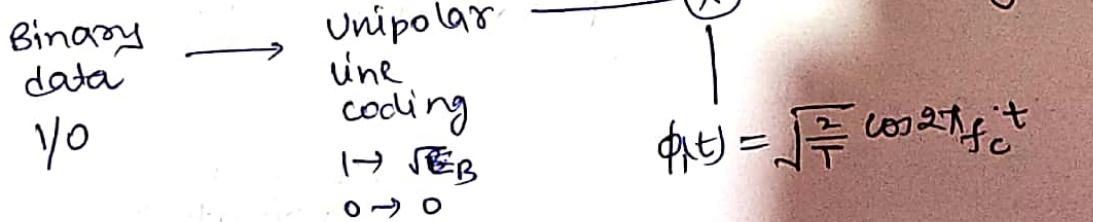
M=2 \leftarrow transmitted waveform
 N=1 \leftarrow basis fn seqd.

$$\text{as } x_1(t) = \sqrt{E_B} \phi_1(t) \quad \leftarrow \text{bit 1}$$

$$x_2(t) = 0 \phi_2(t) \quad \leftarrow \text{bit 2}$$



BASK Transmitter.



Probability of error

$$P_e \rightarrow \frac{P_e(1) + P_e(0)}{2}$$

$1 \rightarrow 0$ } error
 $0 \rightarrow 1$ }

$$y(t) \left\{ \begin{array}{ll} x(t) + n(t) & \text{bit 1} \\ 0 + n(t) & \text{bit 2} \end{array} \right.$$

correlation receiver output

$$y = \int_0^{T_B} y(t) \phi_i(t) dt$$

for bit 1

$$\begin{aligned} y &= \int_0^{T_B} (x_i(t) + n(t)) \phi_i(t) dt \\ &= \int_0^{T_B} (\sqrt{E_B} \phi_i(t) + n(t)) \phi_i(t) dt \\ &= \sqrt{E_B} \int_0^{T_B} \phi_i^2(t) dt + n \uparrow \\ &\quad \uparrow \text{orthogonal} \qquad \text{Gaussian as } n(t) \text{ is gaussian} \\ &\Rightarrow 1 \end{aligned}$$

$$\Rightarrow y = \sqrt{E_B} + n \quad \text{bit 1}$$

$$y = n \quad \text{bit 0}$$

n is Gaussian with mean 0 and variance $\frac{N_0}{2}$

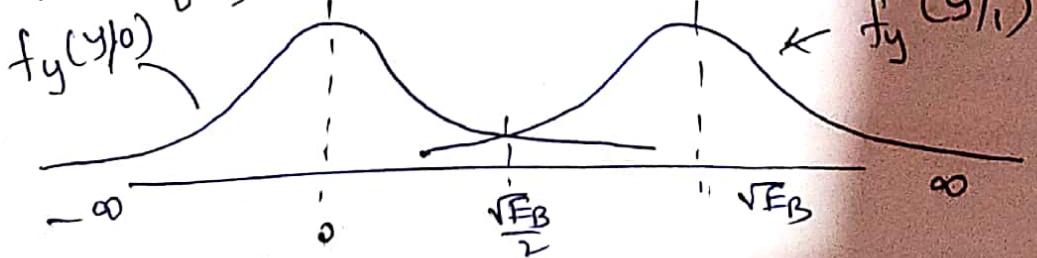
$$\text{Mean of } y = E(y) \left\{ \begin{array}{ll} \sqrt{E_B} & \text{bit 1} \\ 0 & \text{bit 0} \end{array} \right.$$

Variance of y

$$E(y - \bar{y})^2 = E \left\{ (\sqrt{E_B} + n) - \sqrt{E_B} \right\}^2 = E(n^2) = \frac{N_0}{2}$$

$$E(y - \bar{y})^2 = E(n^2) = E(n^2) = \frac{N_0}{2}$$

PDF of y



$$\begin{cases} f_{y|0}(y|0) = \frac{1}{\sqrt{\pi N_0}} e^{-y^2/N_0} \\ f_{y|1}(y|1) = \frac{1}{\sqrt{\pi N_0}} e^{-(y-\sqrt{E_B})^2/N_0} \end{cases}$$

Likelihood fn.

$$P_e(0) = \int_{\frac{-\sqrt{E_B}}{2}}^{\infty} f_{y|0}(y|0) dy = \int_{\frac{-\sqrt{E_B}/2}{2}}^{\infty} \frac{1}{\sqrt{\pi N_0}} e^{-y^2/N_0} dy$$

$$\boxed{\operatorname{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-z^2} dz} = z = \frac{y}{\sqrt{N_0}}$$

$$\Rightarrow \frac{1}{\sqrt{\pi N_0}} \int_{\frac{-\sqrt{E_B}}{2\sqrt{N_0}}}^{\infty} e^{-z^2} \sqrt{N_0} dz$$

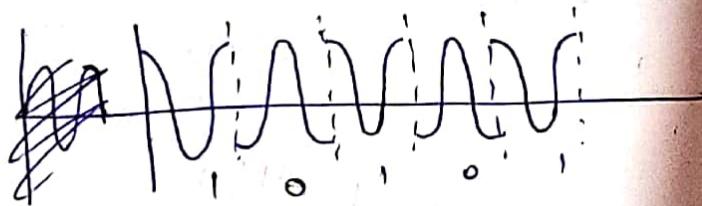
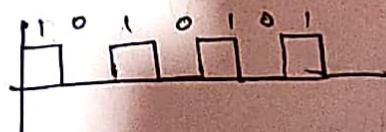
$$= \frac{1}{2} \left(\frac{2}{\sqrt{\pi}} \int_{\frac{-\sqrt{E_B}}{2\sqrt{N_0}}}^{\infty} e^{-z^2} dz \right)$$

$$\Rightarrow P_e(0) = \frac{1}{2} \operatorname{erfc}\left(\frac{1}{2} \sqrt{\frac{E_B}{N_0}}\right)$$

$$P_e(1) = \int_{-\infty}^{\frac{\sqrt{E_B}}{2}} f_{y|1}(y|1) dy = \frac{1}{2} \operatorname{erfc}\left(\frac{1}{2} \sqrt{\frac{E_B}{N_0}}\right)$$

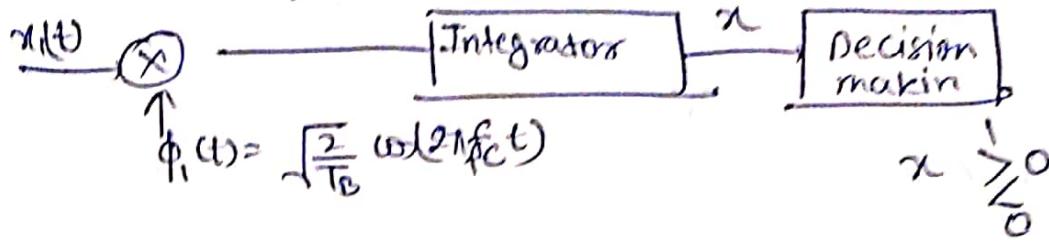
$$\Rightarrow P_e = \frac{P_e(0) + P_e(1)}{2} = \frac{1}{2} \operatorname{erfc}\left(\frac{1}{2} \sqrt{\frac{E_B}{N_0}}\right)$$

#) Binary phase shift keying

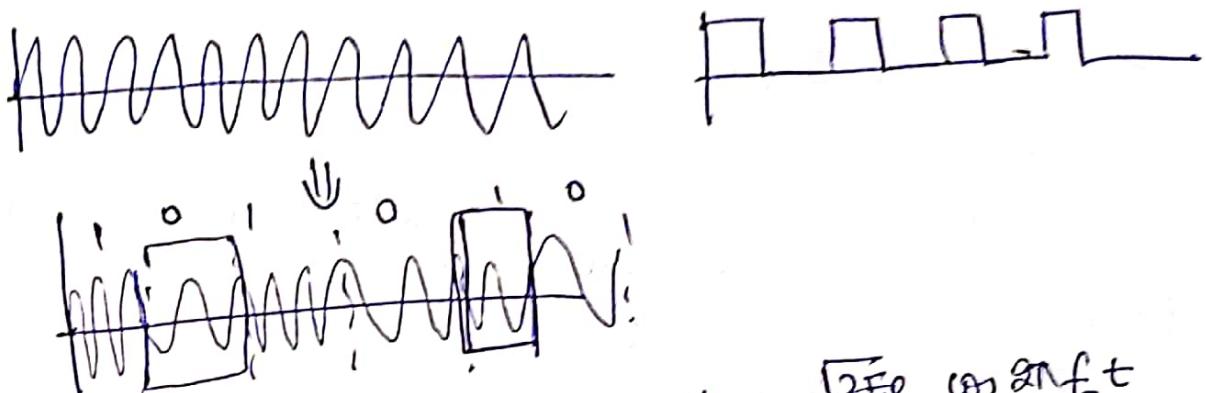


1 → phase
0 → opposite phase.

Circuit at receiver



BFSK



$$s_1(t) = \sqrt{\frac{2E_B}{T_B}} \cos 2\pi f_1 t$$

$$s_2(t) = \sqrt{\frac{2E_B}{T_B}} \cos 2\pi f_2 t$$

Basis fn

$$\phi_1(t) = \frac{s_1(t)}{\sqrt{E_B}}$$

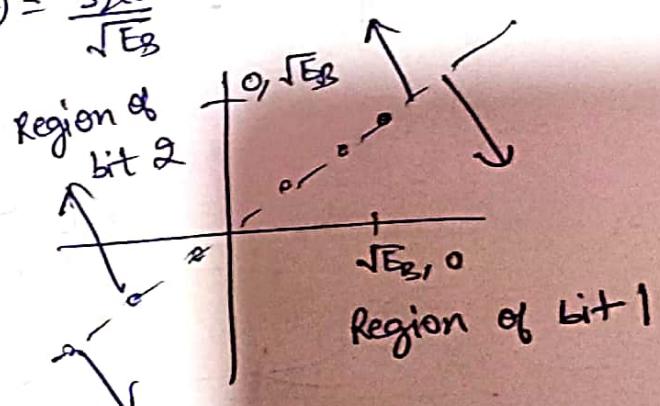
$$s_{11} = \int s_1 \phi_1 = \sqrt{E_B}$$

$$s_{12} = \int s_1 \phi_2 = 0$$

$$s_{21} = \int s_2 \phi_1 = 0$$

$$s_{22} = \int s_2 \phi_2 = \sqrt{E_B}$$

$$\phi_2(t) = \frac{s_2(t)}{\sqrt{E_B}}$$



error in BFSK is less than BPSK b/cos
distance b/w $(\sqrt{E_B}, 0)$ and $(0, \sqrt{E_B})$ is less than $2\sqrt{E_B}$
(easily understandable by graph)

if received signal is $x(t)$

$$x_1 = \int_0^{T_B} x(t) \phi_1(t) dt$$

$$x_2 = \int_0^{T_B} x(t) \phi_2(t) dt$$

$$x_1 > x_2$$

Type → Binary (single bit is modulated)
 Quadrature (2 bit) → 4 combination ($\phi = N_2$)
 m-array (general)
 no. of level ↑ → $P_e \uparrow$

$$s_1(t) = \sqrt{\frac{2E_B}{T_B}} \cos(2\pi f_c t)$$

$$s_2(t) = \sqrt{\frac{2E_B}{T_B}} \cos(2\pi f_c t + \pi) = -\sqrt{\frac{2E_B}{T_B}} \cos(2\pi f_c t)$$

orthogonal signals → signals lie in diff direction ∵ min noise interference

$$\int \phi_1 \phi_2 = 0 \quad \text{as} \quad \phi_1 \perp \phi_2$$

↑ orthonormal fn

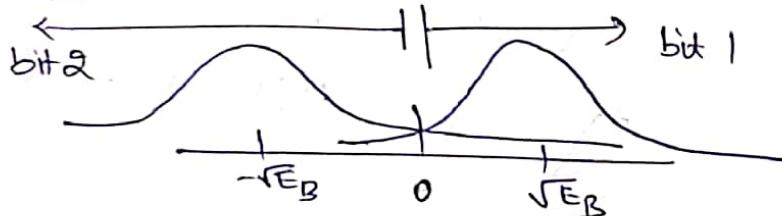
$$\phi_1 = \sqrt{\frac{E_B}{T_B}} \cos \sqrt{\frac{2}{T_B}} \cos(2\pi f_c t)$$

$$s_{11} = \int_0^{T_B} s_1(t) \phi_1(t) dt = \int_0^{T_B} \sqrt{E_B} \phi_1^2(t) dt \quad \int \phi_i \cdot \phi_1 = 1$$

$$s_{11} = \sqrt{E_B}$$

} ← mean

$$s_{21} = \int s_2 \phi_1 = -\sqrt{E_B}$$



likelihood fn when 0 is transmitted.

$$f_x(x|0) = \frac{1}{\sqrt{\pi N_0}} \exp \left[-\frac{1}{N_0} (x_1 + \sqrt{E_B})^2 \right]$$

$$P_e(0) = \int_0^{+\infty} f_x(x|0) dx = \frac{1}{2} \operatorname{erfc} \left(\sqrt{\frac{E_B}{N_0}} \right)$$

$$P_e(1) = \int_{-\infty}^0 f_x(x|1) dx = \frac{1}{2} \operatorname{erfc} \left(\sqrt{\frac{E_B}{N_0}} \right)$$

$$P_e = \frac{P_e(0) + P_e(1)}{2} = \frac{1}{2} \operatorname{erfc} \left(\sqrt{\frac{E_B}{N_0}} \right)$$

for error calculation

$$\text{error} = x_1 - x_2$$

$$\text{let } x_1 - x_2 = l$$

$$l \begin{matrix} > 0 \\ \downarrow \\ < 0 \end{matrix}$$

expectation value

$$E[l|1] = E[x_1|1] - E[x_2|1]$$

$$= \sqrt{E_B} - 0 = \sqrt{E_B}$$

expectation is
linear fn

$$E[l|0] = E[x_1|0] - E[x_2|0]$$

$$= 0 - \sqrt{E_B} = -\sqrt{E_B}$$

$$\text{Var}[l] = \text{Var}[x_1] + \text{Var}[x_2] = \frac{N_0}{2} + \frac{N_0}{2} = N_0$$

$$\rightarrow \text{mean} = 0$$

$$\text{var} = N_0$$

conditional probability

$$f_l(l|0) = \frac{1}{2\pi N_0} e^{-\frac{(l+\sqrt{E_B})^2}{2N_0}}$$

$$P(l|0) = \int_0^\infty f_l(l|0) dl$$

$$= \frac{1}{2} \operatorname{erfc}\left(\frac{\sqrt{E_B}}{2N_0}\right)$$

$$\therefore P(l|1) = \frac{1}{2} \operatorname{erfc}\left(\frac{\sqrt{E_B}}{2N_0}\right)$$

$$P_e = \frac{1}{2} \operatorname{erfc}\left(\sqrt{\frac{E_B}{2N_0}}\right)$$

clearly error is BFSK < BPSK

Error detecting codes

↳ To minimise effect of noise in channel

parity code

↳ for 1 bit only (or odd bits)

parity \rightarrow even or odd times of 1

new bit	P	data(msg)	Tx	corruption	Rx
		0 1001	01001	01101	11010
	1	1011	11011	11010	
	0	0101	00101	10101	
	0	1111	01111	11111	

if Tx has even parity and Rx has odd or vice versa
then odd no. of bits are corrupted

\Rightarrow not for even corrupted bits (draw back)

Hamming code

↳ we send data along with parity bits / redundant bits

↳ represented as (n, k)

$$\text{Parity bits} = n - k = p$$

total bits msg bits

$$\text{cond}^n \text{ for parity bit} = 2^p \geq p + k + 1$$

p=3 satisfy

linear block code

↳ A block code satisfy condⁿ that sum of any 2 code word = another code word

$$c_p = c_i + c_k$$

Property
↳ all 0 [0 ... 0] is always a code word

$$c_p = c_i + c_k$$

$$d(c_i, c_k) = \begin{matrix} w(c_p) \\ \uparrow \text{Hamming distance} \end{matrix}$$

$$\hookrightarrow d_{\min} = w_{\min}$$

Weight = Total no. of 1
~~distance~~ \Rightarrow min hamming dist = min weight.

(7,4) Hamming code = Linear block code
 ↑ satisfy the condn.

$$[c] = [i][g] \quad \leftarrow \text{matrices}$$

c = code word

i = information word

g = generator matrix

$g \rightarrow \cancel{n \times k}^{\text{matrix}}$

$$[c] = [I : P]$$

for (7,4)

systematic code

$$[g] = \begin{bmatrix} 1 & 0 & 0 & 0 & | & 1 & 0 & 1 \\ 0 & 1 & 0 & 0 & | & 1 & 1 & 1 \\ 0 & 0 & 1 & 0 & | & 1 & 1 & 0 \\ 0 & 0 & 0 & 1 & | & 0 & 1 & 1 \end{bmatrix}$$

Identity

Parity

example:

$$[i] = 1110$$

$[g] \rightarrow$ as above

$$[c] = [i][g] = [1110100]$$

in code word

$$[c] = [m : p_c]$$

tailway msg bits bcos

$$\begin{aligned} & \text{we do mod 2} \\ & \text{sum} \\ & 1+1=0 \\ & 1+0=1 \\ & 0+0=0 \end{aligned}$$

$$q = [I : P]$$

$$m \times I = m$$

for p_c we can also do

$$[p_c] = [m] [P]$$

(simple calculation)

Parity check matrix

$$[H] = [P^T : I_{n-k}]$$

n, k, P from Generator matrix.

Properties

$$Hg^T = gH^T = CH^T = 0$$

$$C = [I] [g]$$

$$CH^T = [I] [g][H^T] = 0$$

Error syndromes in linear block code

$$R_x = [y]$$



Error syndrome

$$[s] = [y][H^T]$$

$$[y] = [c] + [e]$$

$$[s] = ([c] + [e])[H^T]$$

$$[s] = e[H^T]$$

Syndrome

000

0000000

(1st bit corrupted)

000

1000000

(2ⁿ bit corrupted)

010

0100000

001

0010000

110

0001000

011

0000100

111

0000010

101

0000001

error pattern

Syndrome depends only on e

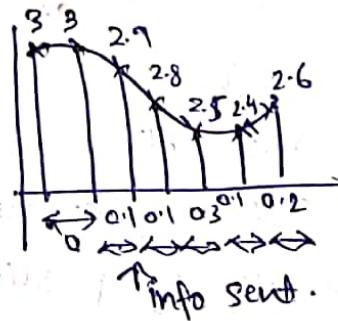
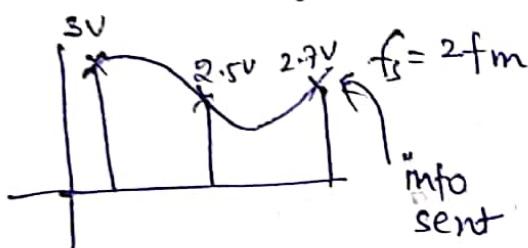
DPCM

Differential pulse code modulation

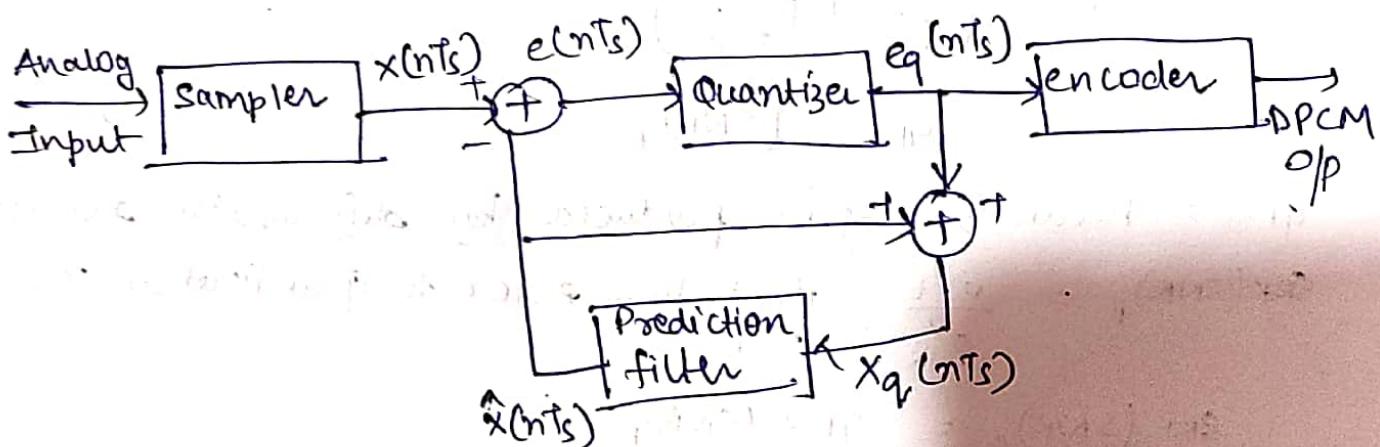
↪ when $f_s > 2f_m$ (sampling) then successive samples become more correlated.

↪ Instead of sending data for individual sample (like in PCM) we send data as successive difference in pulse voltage.

∴ no. of levels $\downarrow \rightarrow$ bit rate \downarrow



DPCM encoder



$$\begin{aligned} e(nT_s) &= \text{Error signal} \\ &= x(nT_s) - \hat{x}(nT_s) \end{aligned}$$

Quantised error signal.

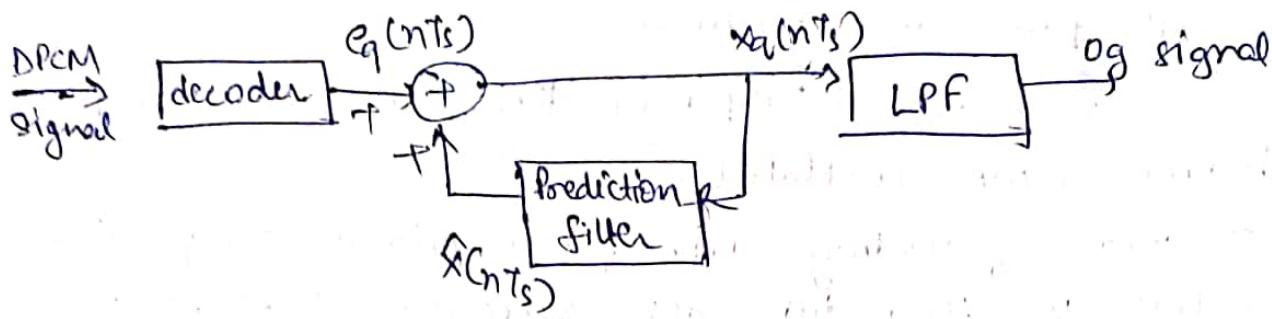
$$e_q(nT_s) = e(nT_s) + q_e(nT_s)$$

Input to prediction filter

$$\begin{aligned} x_q(nT_s) &= e_q(nT_s) + \hat{x}(nT_s) \\ &= e(nT_s) + q_e(nT_s) + \hat{x}(nT_s) \\ &= x(nT_s) - \hat{x}(nT_s) + q_e(nT_s) \end{aligned}$$

$$x_q(nT_s) = x(nT_s) + q_e(nT_s)$$

DPCM decoder.



SQNR of DPCM

$$(\text{SNR})_o = \frac{\sigma_x^2}{\sigma_q^2} \quad \text{at o/p of quantiser}$$

σ_x^2 = variance of original v_p , $x(nT_s)$

σ_q^2 = variance of quantised error $e_q(nT_s)$

σ_e^2 = variance of prediction error $e(nT_s)$

$$(\text{SNR})_o = \frac{\left(\frac{\sigma_x^2}{\sigma_e^2}\right)}{GP} \left(\frac{\sigma_e^2}{\sigma_q^2}\right) \quad (\text{SNR})_p$$

GP → Prediction gain produced by differential quantisation

~~$$(\text{SNR})_p = \frac{\sigma_e^2}{\sigma_q^2} = \text{Prediction error to quantisation noise ratio}$$~~

~~$$(\text{SNR})_o = GP \times (\text{SNR})_p$$~~

$$GP > 1 \Rightarrow \sigma_x^2 > \sigma_e^2$$

we can't change σ_x^2 but can change σ_e^2

our goal is to minimise σ_e^2 (better prediction filter)