

It may be noted that error detection is possible if received code vector is not equal to zero otherwise code vector, this shows that the (minimum) transmission error in the received code vector must be less than minimum distance ( $d_{min}$ ).

Table shows requirement error capability of code.

S.NO	Name of Error detected/Corrected	Distance requirement
01.	detect upto 's' Errors per word.	$d_{min} \geq s+1$
02	Correct upto 't' errors per word	$d_{min} \geq 2t+1$
03	Correct upto 's' and detect 's+t' errors per word.	$d_{min} \geq t+s+1$

$$d_{min} = \text{No. of zero code vector.}$$

$\Leftrightarrow$  Weight of code vector (No. of ones)

(8) Code Efficiency:- Expressed as ratio of message bits in block to the transmitted bit for that particular block by the Encoder.  $\eta = \frac{k}{n}$ .



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NON Zero Elements in the transmitted code vector,  $w(x) \rightarrow$  code vector.

0  $\rightarrow$  Even  
1  $\rightarrow$  odd flip

$$x = 0110101$$

$$w(x) = 5$$

ERROR CODING

(1) Parity Coding:-  
1. Odd Parity.  
2. Even Parity.

- 1) Vertical Redundancy check
- 2) Longitudinal redundancy check (LRC)
- 3) Linear Block Codes
- 4) Hamming Codes,
- 5) Cyclic Codes
- 6) Convolutional Codes.

PARITY Coding:

Even:- Parity of binary word is known as Even P/P containing even no. of ones

Odd:- 011011010  $\rightarrow$  61's b8  $\rightarrow$  if odd then 1  
b8  $\rightarrow$  1  $\rightarrow$  then even  
even .. 0 } b8 b7.

Odd:- if P contains odd no. of ones.

(2) Vertical Redundancy Check:-

ASCII ASCII

b1 b2 b3 b4 b5 b6 b7 b8

Character  
K | 1 0 1 0 0 1 0 . Even

0 | 1 1 1 0 0 1 1 . odd:  
1 | 0 0 1 0 0 1 1 .

### ③ Longitudinal Redundancy Check :- (LRC)

In

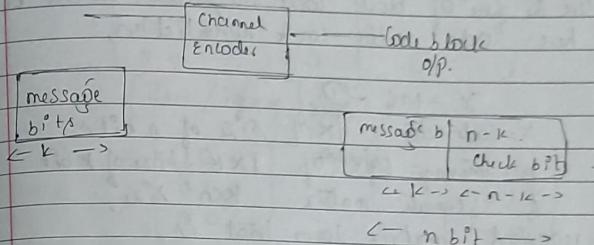
	Six	Inform. block	KTB	End of Block	BCC
start of character		many characters	Block chkd char.		
message					
transmitted					
Block					
Chkd. char.					
path					
Char parity					
row parity					
	b1	0 0 1 1 0 1 1 1 1 1			
	b2	1 1 0 0 0 0 1 1 1 1			
	b3	0 1 0 0 1 0 0 1 1 1			
	b4	0 1 0 1 0 [ ] 0 , 0 0			
	b5	0 0 0 0 1 0 1 0 1 1			
	b6	0 0 0 0 0 0 0 0 0 0			
	b7	0 0 1 1 0 1 1 1 0 0			
	b8	1 0 0 1 1 1 0 1 0 0	x		

### ④ Linear Block

For a Block of  $M$  message bits an  $(n-k)$  parity bits are called. Check bits are added, this means that the total bits of o/p encoder are ' $n$ ', such types of codes

are known as  $(n,k)$  block codes.

### Systematic code



A code is known as linear if the sum of any 2 codewords produce another codeword.

Steps for determination of all code words for a linear block code:

① Let us consider that, the particular cod. vector consists of  $m_1, m_2, \dots, m_k$  message bit and  $c_1, c_2, \dots, c_{n-k}$  check bits. Then the codewords may be written as:

$$m_1, m_2, \dots, m_k$$

$$c_1, c_2, \dots, c_{n-k}$$

$$x = (m_1, m_2, \dots, m_k, c_1, c_2, \dots, c_{n-k})$$

$$q = n - k$$

② This means that  $q$  are the no. of redundant bits added by the encoder.

The above code can be written as:

$$x = (M/C)$$

$M$  =  $k$  bit message vector  
 $C$  =  $n$  bit code vector

③ Here check we play the role of error detection and correction (ECC)

④ Here the form of block code to generate check bit  
here code vector can be represented as,

$$X = MG \quad \text{①}$$

$X$ : codeword  $1 \times n$  size of  $n$  bits.

$M$ : message bits matrix of  $1 \times k$ . size of  $k$  bits

$G$ : generator matrix  $k \times n$  size of  $n$  bits.

⑤ represent matrix form like this

$$(x)_{n \times 1} = [M]_{1 \times k} [G]_{k \times n}$$

$G$  matrix depends on linear block code used  
(7,4)  $\rightarrow$   $(n, k)$ .

$$n = (n - k) + 3 \quad m = 4$$

⑥ Generate  $P_k$  represent as  $G = [I_k \oplus P_{k \times q}]_{k \times n}$

here  $I_k = k \times k$  identity matrix

$P = k \times q$  submatrix

NOW code vector maybe obtained as

$$C = MXP$$

$$G = C_1 C_2 \dots C_q \quad P = \begin{bmatrix} P_{11} & P_{12} & \dots & P_{1q} \\ \vdots & & & \\ P_{k1} & P_{k2} & \dots & P_{kq} \end{bmatrix}_{k \times q}$$

$D_1 = 1$   
 $D_2 = 1$   
 $D_3 = 0$   
 $D_4 = 0$

$$\text{On solving the matrix Eq :-}$$

$$C_1 = M_1 P_{11} \oplus M_2 P_{12} \oplus M_3 P_{13}$$

$$C_2 = M_1 P_{21} \oplus M_2 P_{22} \oplus M_3 P_{23}$$

$$C_3 = M_1 P_{31} \oplus M_2 P_{32} \oplus M_3 P_{33}$$

$$G = [I_k \oplus P_{k \times q}]_{k \times n}$$

$$C = MP.$$

① Generator Matrix for (6,3) matrix block code  $P$  shown below obtain all code words for this code.

$$\rightarrow G = \begin{bmatrix} 1 & 0 & 0 & 1 & 0 & 1 \\ 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 & 1 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 \end{bmatrix}_{3 \times 6}$$

$$I_k = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix} \quad P_{k \times q} = \begin{bmatrix} 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \end{bmatrix}$$

$$C = MP. \quad n = 6 \\ M = 3 = M_1 M_2 M_3 \quad k = 3$$

$$\rightarrow (C_1, C_2, C_3) = [M_1 \ M_2 \ M_3] \begin{bmatrix} 0 & 1 & 1 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \end{bmatrix}$$

$$C_1 = M_2 \oplus M_3$$

$$C_2 = M_1 \oplus M_3$$

$$C_3 = M_1 \oplus M_2$$

(b) code vector of (6,3) linear block code

Sl No	Bit of message vector in one block			$C_1$	$C_2$	$C_3$	Complex vector MINUS GCR
	$M_1$	$M_2$	$M_3$	$M_2 \oplus M_3$	$M_1 \oplus M_3$	$M_1 \oplus M_2$	
1	0	0	0	0	0	0	000 000
2	0	0	1	1	0	0	001 110
3	0	1	0	1	0	1	010 101
4	0	1	1	0	1	1	011 011
5	1	0	0	0	1	1	100 011
6	1	0	1	1	0	1	101 101
7	1	0	10	1	1	1	110 111
8	1	1	1	0	0	0	111 000

### Hamming Codes:-

Are defined as  $(n, k)$  linear block codes, these codes satisfy the following condition.

- No of check bits  $Q \geq 3$
- Block length  $k$  &  $n = 2^Q - 1$
- No of message bits  $m = n - Q$
- Min Distance  $D_{\min} = 3$

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Date \_\_\_\_\_  
Page \_\_\_\_\_

Error detection and correction of Hamming code.

$D_{\min} = 3$  it can be used to detect double errors and/or correct single error.  
 $d_{men} \geq s+1 \Rightarrow d \geq 2t+1$

① The Parity check matrix of a particular  $(7, 4)$  linear block code is expressed as:

$$H = \begin{bmatrix} 1 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 1 & 0 \\ 1 & 0 & 1 & 1 & 0 & 1 & 0 \\ 1 & 0 & 1 & 1 & 1 & 0 & 1 \end{bmatrix}$$

Obtain the generator matrix by first all the code vectors.

- what will be the min distance b/w code vectors
- How many errors can be detected, corrected.

→ Given that  $n = 7$ ,  $k = 4$ . And no of check bits  $Q = n - k = 7 - 4 = 3$ ,  
 $\therefore D_{\min} = 2^{\lceil \frac{Q}{2} \rceil} - 1 = 2^{\lceil \frac{3}{2} \rceil} - 1 = 4$ .

This indicates that given code is hamming code.  
Parity check matrix is  $Q \times n$  size, and is given by the eqn.

Also  $H = [P^T : I_Q]_{Q \times n}$

$$P = \begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 0 \\ 1 & 0 & 1 \\ 0 & 1 & 1 \end{bmatrix}$$

Further general matrix is triangular  $(I_k : P_{k \times n})_{k \times n}$



$$\rightarrow \underline{q} = \begin{bmatrix} I_{4 \times 1} & P_{4 \times 2} \end{bmatrix}_{4 \times n}$$

$$= \begin{bmatrix} I_4 & P_{4 \times 2} \end{bmatrix}_{4 \times 2}$$

$$\underline{q} = \begin{bmatrix} 1000 & 111 \\ 0100 & 110 \\ 0010 & 101 \\ 0001 & 011 \end{bmatrix}_{4 \times 2}$$

$$G_p = M_p P, \quad C_3 = M_3 P$$

$$[G_1 \ G_2 \ G_3] = [M_1 \ M_2 \ M_3 \ M_4] \begin{bmatrix} 1 & 1 & 1 \\ 1 & 0 \\ 0 & 1 \\ 0 & 1 \end{bmatrix}$$

$$G_1 = M_1 \oplus M_2 \oplus M_3$$

$$G_2 = M_2 \oplus M_3 \oplus M_4$$

$$G_3 = M_1 \oplus M_3 \oplus M_4$$

(b) min distance = 3

$$d_{min} \geq s+1$$

$s=2$  Error check-

$$d_{min} \geq 2+1$$

$t=1 \rightarrow$  corrected

$$3 = 2 \times 1 + 1$$



S1	Message bit	C <sub>1</sub>	C <sub>2</sub>	C <sub>3</sub>
	M <sub>1</sub> M <sub>2</sub> M <sub>3</sub> M <sub>4</sub>	M <sub>1</sub> , M <sub>2</sub> , M <sub>3</sub>	M <sub>1</sub> , M <sub>2</sub> , M <sub>4</sub>	M <sub>1</sub> , M <sub>3</sub> , M <sub>4</sub>

0 0 0 0	0	0	0	00000000
0 0 0 1	0	1	1	0001001
0 0 1 0	1	0	0	00100100
0 0 1 1	1	0	1	0011001
0 1 0 0	1	0	0	0100101
0 1 0 1	1	1	0	0101110
0 1 1 0	0	1	1	0110011
0 1 1 1	0	0	0	0111000
1 0 0 0	1	1	1	1000111
1 0 0 1	1	0	0	1001100
1 0 1 0	0	0	1	1010001
1 0 1 1	0	1	0	1011010
1 1 0 0	0	1	0	1100010
1 1 0 1	0	0	1	1101001
1 1 1 0	1	0	0	1110100
1 1 1 1	1	1	1	1111011

(+) min distance

Cyclic Code :- Type may be described as the sub class of linear block code. They have the property that cyclic shift of one code word produces another code word.

As an example, consider an n bit code vector

$$x = \{x_{n-1}, x_{n-2}, \dots, x_1, x_0\}$$

z<sub>n-1</sub>, z<sub>n-2</sub>, ..., z<sub>1</sub>, z<sub>0</sub> are individual bits of code vector

x. If we shift above code word cyclically then we get another code vector i.e.

$$x^t = \{x_{n-1}, x_{n-2}, \dots, x_1, x_0, x_{n-1}\}$$

Cyclic code: A linear code is known as cyclic code if every cyclic shift of the code vector produces some other valid code vector.

① The codewords may be defined with the help of polynomial of degree less than  $(n-1)$ . Thus we can define code vector in terms of polynomial as:-

$$x(p) = x_{n-1}p^{n-1} + x_{n-2}p^{n-2} + \dots + x_1p + x_0$$

where  $p$  is an arbitrary variable and power of  $p$  represents the position of codeword. So here  $p^0$  represents the position of codeword.

- a) types → a. systematic cyclic code
- b. non systematic cyclic code.

A generator polynomial of a (7,4) cyclic code is  $g(p) = p^3 + p + 1$ . obtain all the code vector for the code in non systematic and systematic form

ANSWER

$$\Rightarrow n = 7; k = 4; q = n - k + 3$$

$$\text{Message vector } 2^k = 2^4 = 16(\text{Mod})$$

Let us consider any message vector as,  $m = (M_3, M_2, M_1, M_0)$

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$$m = (M_3, M_2, M_1, M_0) = (0, 1, 0, 1)$$

then the message polynomial will be  $M(p)$ .

$$M(p) = M_3 p^3 + M_2 p^2 + M_1 p + M_0$$

$$= 0 \quad 0 \quad 1 \quad 0$$

$$M(p) = p^3 + 1$$

$$\text{GCRH } g(p) = p^3 + p + 1$$

To obtain non systematic code vector,

$$\begin{aligned} X(p) &= M p g(p) \\ &= (p^2 + 1)(p^3 + p + 1) \\ &= p^5 + p^3 + p^2 + p^3 + p + 1 \\ &= p^5 + p^5 + p^5 + 2p^3 + p^2 + p + 1 \\ &= p^5 + p^3 (X(0)) + p^2 + p + 1 \end{aligned}$$

$$X(p) = p^5 + p^2 + p + 1 \quad (q^6 + q^5 + q^3 + q^2 + q + 1)$$

Note that the degree of above polynomial is  $(n-1) = 7$  ( $6-1$ )

The code vector corresponding to above polynomial is

$$X = \{x_6, x_5, x_4, x_3, x_2, x_1, x_0\}$$

$$X = \{0, 1, 0, 0, 1, 1, 1\}$$

in this  $\underline{\underline{0}}$  is the code vector for message vector (0101).

This code is called non systematic cyclic code.

### (b) Systematic code value :-

For SCC we will first find check bits. These check bits are given by check bit polynomial  $C(p)$ . From the div.  $C(p)$  is ob expressed as

$$C(p) = \text{rem} \left[ \begin{matrix} p^3 M(p) \\ g(p) \end{matrix} \right]$$

Since  $Q=3$ ,  $p^3 M(p)$  will be = 0101

$$\begin{aligned} p^3 M(p) &= p^3(p^2 + 1) \\ &= p^5 + p^3 + p^3 + 1 \\ &= p^5 + p^3 + p^2 + p + 1 \end{aligned}$$

$$\begin{aligned} g(p) &= p^3 + p + 1 \\ &= p^3 + p^2 + p + 1 \end{aligned}$$

$$\begin{aligned} &= \text{rem} \left[ \begin{matrix} p^5 + p^3 + p^2 + p + 0 \\ p^3 + p^2 + p + 1 \\ p^2 + 0 \cdot 10 \end{matrix} \right] \end{aligned}$$

$$(p^3 + p^2 + p + 1) \text{ } p^5 + p^4 + p^3 + p^2 + p + 0$$

$$p^5 + p^4 + p^3 + p^2$$

$$0 \cdot 0 - 0 - 0 - 0$$

$$p^2 + op + 0$$

remainder.

Therefore "remainder polynomial" in division of  $p^3 M(p)$  by  $g(p)$  is  $p^2 + op + 0$ .



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Page \_\_\_\_\_

$$\Rightarrow C(p) = \text{rem} \left[ \begin{matrix} p^3 M(p) \\ g(p) \end{matrix} \right]$$

$$(p) = p^2 + op + b$$

$$C(p) = C_2 p^2 + C_1 p + C_0$$

$$C_2 p^2 + C_1 p + C_0 = p^2 + op + 0$$

check bits

$$C = (C_1, C_0) = (1, 0)$$

Indirectly,

$$X = \{M_3, M_2, M_1, M_0, C_1, C_0\}$$

$$X = \{0101 : 100\}$$

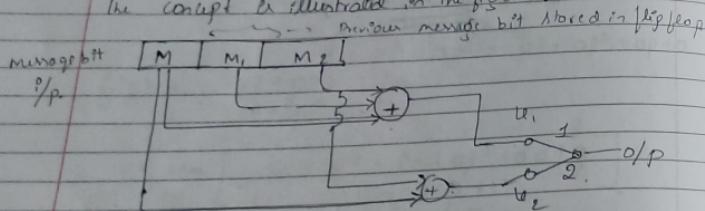
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Page \_\_\_\_\_UNIT 02WAVEFORM CODINGConvolutional Codes:-

A convolutional coding is done by combining the fixed number of input bits. The input bits are stored in fixed length shift register and they are combined with the help of modulo adder.

- This operation is equivalent to binary convolution, hence it is called convolutional coding.

The concept is illustrated in the figure.

Convolutional Encoder with  $K=3, N=2$ 

$(M, M_1)$  represents the state of shift register.

$$U_1 = M \oplus M_1 \oplus M_2$$

$$U_2 = M \oplus M_2$$

Output

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 $V_1 V_2 V_1 V_2 \dots$ 

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$$\text{Code rate } (r_o) = \frac{k}{n} = \frac{1}{2}$$

This type of encoder is called  $\frac{1}{2}$  Encoder.  
3 message bit  $\frac{p}{p}$  is only 2 have  $\times$  1 bits

Operation :-

- whenever the message bit is shifted in position m, the new value of  $v_i$ , i.e., are generated depending upon  $M, M_1, M_2$ .

$M, M_1, M_2$  store the previous 3 message bit, the current bit is present in  $M$ , hence we can write  $M \oplus M_1 \oplus M_2 = v_i$

- The o/p of encoder switch first sample  $v_i$  then  $v_2$ .
- The shift register can shift its content  $(M, M_1, M_2)$ .

- Next if p bit is taken and stored in  $M$ .

Again  $v_1, v_2$  are generated according to this new combination of  $M, M_1, M_2$ .

- Hence o/p switch samples  $v_1$  then  $v_2$ .

Hence the o/p bit stream for successive p bits will be  $x = v_1, v_2, v_1, v_2, \dots, n$ .

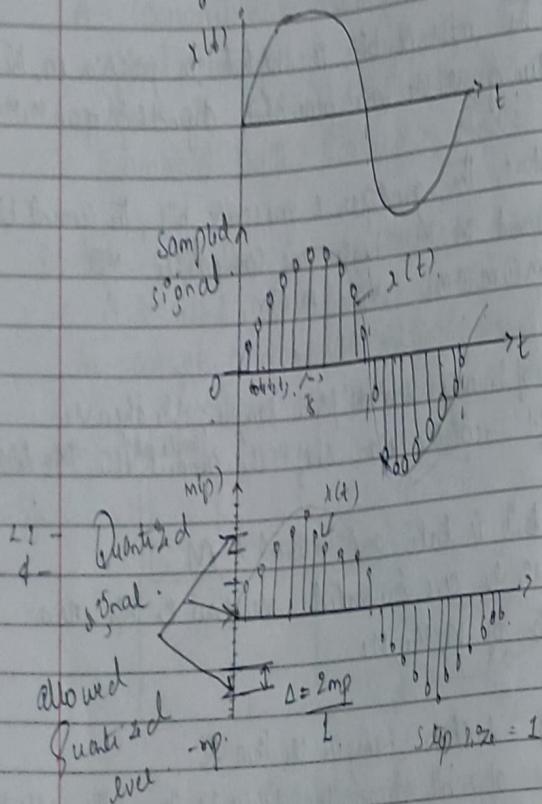
- Here it may be noted that for having message bit, we have 2 encoded o/p bits  $v_1$  and  $v_2$  are transmitted. In other words for a single message bit 1/2 encoded code word is 2 bits.

Therefore the code rate of this channel encoder is.

$$Y = \frac{K}{n} = 1 - \left\{ \begin{array}{l} \text{if odd bits} \\ \text{if } p = k \end{array} \right.$$

→ Q&:

### Concept of Quantization:



③ As shown, amplitude of signal  $x(t)$ , lies in range  $(-m_p, m_p)$  which is partitioned into  $n$  intervals,  $\Delta p$  which is partitioned into  $n$  intervals, each of magnitude  $\Delta q$

$\Delta p = \frac{2m_p}{n}$  Now each sample is approximated as

single

④ Each sample is now approximated to an off the number therefore, the information is digitized.

⑤ In quantized & approximated original one.

we can improve accuracy of quantized signal to any

desired degree, simply by increasing the No of bits L.

Now,

$$t_0 \rightarrow 4 \rightarrow 10$$

$$t_1 \rightarrow 2 \rightarrow 10$$

$$t_2 \rightarrow 4_3 \rightarrow 30.$$

$$t_3 \rightarrow t_4 \rightarrow 40$$

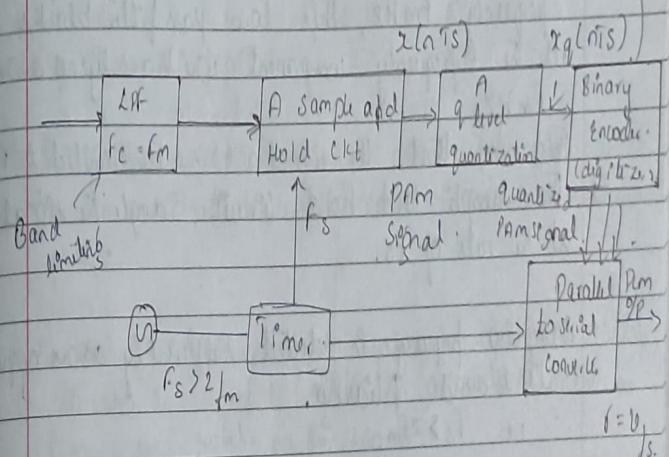
$$t_4 \rightarrow t_5 \rightarrow 40$$

$$t_5 \rightarrow t_6 \rightarrow 30.$$

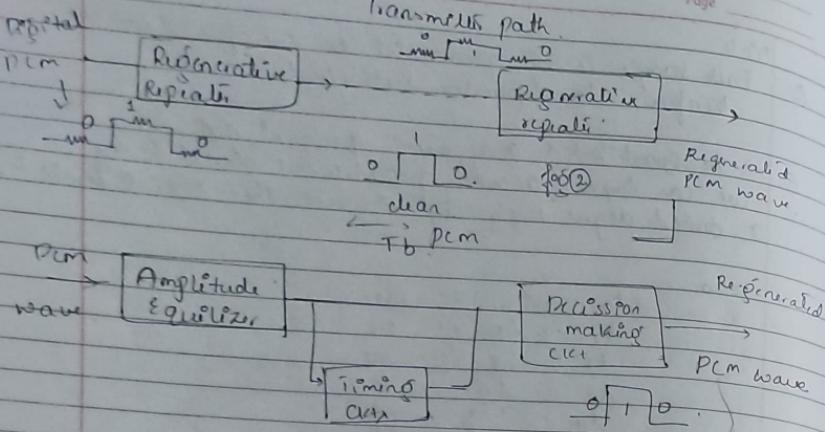
$$t_6 \rightarrow t_7 \rightarrow 20$$

Digital  
Digital modulated  
signal

### PCM (Pulse Code Modulation)



practical PCM generator



② **PCM generation / Transmitter:** In PCM generation, the signal  $x(t)$ , a continuous time NTs is first passed through the LPF of cutoff frequency  $F_m$  Hz. Then low pass filter blocks all the frequency component which are lying above  $F_m$  Hz.

③ This means that the signal  $x(t)$  is bandlimited to  $F_m$  Hz. The sample and hold circuit samples the signal at the rate of  $F_s$ .

Sampling frequency is selected sufficiently above request rate to avoid aliasing.  
i.e.  $F_s > 2F_m$ .

④ The job of sample and hold unit is denoted by  $x(NTs)$ . This signal is discrete in time and continuous in amplitude. "DEMONIC PRINCE"

$x(NTs) \rightarrow L(t)$

⑤ A D level quantizer compares the  $x(NTs)$  with PLS fixed digital level and assigns any one of the digital level to  $x(NTs)$  which results in minimum distortion or error.

⑥ This error is also called an quantization error now the quantized signal level  $x_q(NTs)$  is given to binary encoder.

This encoder converts  $x_q(NTs)$  to  $n$  digits ( $q = 2^n$ ) binary word. Thus  $x_q(NTs)$  is converted to binary bits.

⑦ This encoder is also known as digitizer.

⑧ Also the oscillator generates clocks and for sample and hold unit and parallel to serial converter.

### PCM transmission Path

As shown in  $\text{fig } 2$

⑨ The path between Transmitter and Receiver over which a PCM signal travels / travel is called PCM transmission path.

⑩ The most important feature of PCM system is lies in its ability to reduce the effect of distortion and noise. When the PCM wave travels on the channel.

This is achieved by means of using a chain of regenerator repeaters on the channel.

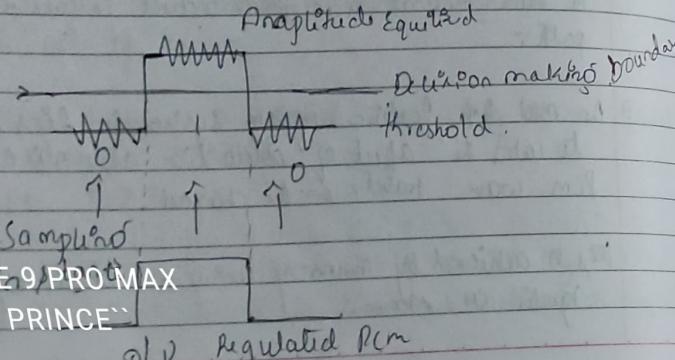


- Q) Such repeaters are spaced close enough to each other on the transmission path.

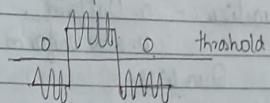
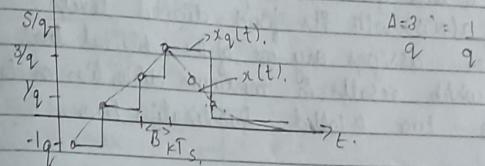
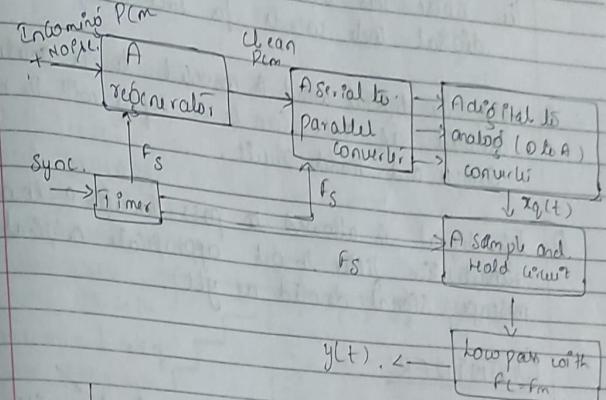
The receiver performs basic 3 operations namely  
 ① Equalization,  
 ② Timing,  
 ③ Decision making.

- Q) As shown in fig (b) Amplitude Equalizer initiates the deformed Pcm wave to compensate for the effects of Amplitude and phase distortions.

- ④ Timing CLK produces a periodic pulse train which is derived from the input Pcm pulses.  
 ⑤ This pulse train is then applied to decision making device. The decision device makes decision about whether the equalized Pcm wave has a zero value or a value at the instant of sampling. Such decision is made by comparing equalized Pcm with a reference level called threshold as shown in fig.



### PCM Receiver:-



$$\frac{1}{f_c} = T_s$$

- Q) Fig shows the block diagram of Pcm receiver and its reconstructed filter signal. The regenerator at the start of the Pcm receiver rectifies the pulse and removes

The noise. This signal is then converted to parallel digital code words for each sample.

- ⑥ Now, the digital word is converted to its analog value denoted as  $y_{\text{d}}(t)$ . Can be seen in fig with the help of Sample and hold circuit.
- ⑦ This signal is allowed to pass through a low pass reconstruction filter to get appropriate original message signal denoted as  $y(t)$ .

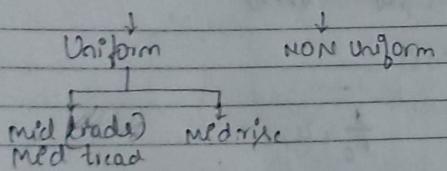
### QUANTIZER:-

- ① A Q level Quantizer compares a discrete time i/p  $x(nT_s)$  with its fixed digital levels. Then assigns any one of the digital level to  $x(nT_s)$  which results in minimum distortion or Error. This error is called Quantization Error.

Types of Quantizer:-

- Uniform.
- NON Uniform.

#### Quantizer:-

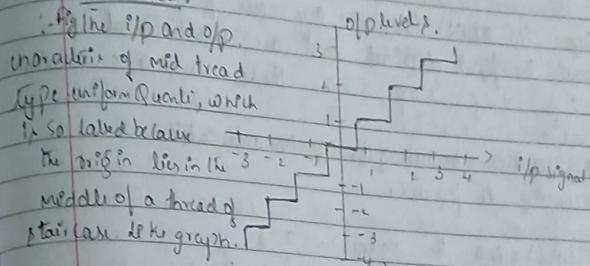


- ② Uniform Quantizer is type of Quantizer in which step size REDMI NOTE 9 PRO MAX come through out the i/p stage
- ③ "DEMONIC PRINCE"

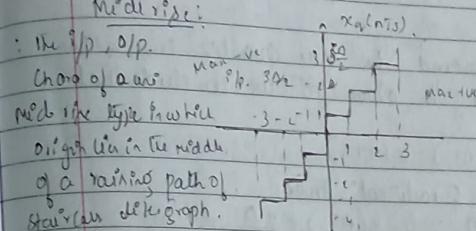


(b) NON :- in which step size varies according to the i/p range

#### Mid trapezoidal:



#### Mid treads:



#### Working principle of Quantizer:-

It may be noted that both types of uniform quantizer are symmetric about the origin.

#### Working principle of Qu

Q1 As shown in fig: midrise type. Let us assume that if p to the Quantizer  $x_{(NTS)}$  varies from  $-4\Delta$  to  $4\Delta$ . Then mean that peak to peak value of  $x_{(NTS)}$  will be  $bw \cdot 4\Delta$  to  $4\Delta$ . ( $\Delta$  is step size).

Then  $x_{(NTS)}$  can take any value from  $-4\Delta$  to  $4\Delta$ , now the fixed digital level are available at  $\pm \frac{\Delta}{2}, \pm \frac{3\Delta}{2}, \pm \frac{5\Delta}{2}, \pm \frac{7\Delta}{2}$ . These levels are available at Quantizer because of its Q characteristic according to figure.

$$x_{(NTS)} = 4\Delta \text{ then } x_q(x_{(NTS)}) = \frac{7\Delta}{2}$$

$$\text{if } x_{(NTS)} = -4\Delta \text{ then } x_q(x_{(NTS)}) = -\frac{7\Delta}{2}$$

Q2 It may be observed, from fig. the max Quantiz' error will be  $\pm \frac{\Delta}{2}$ . From the above fig we conclude that

$$E = x_q(x_{(NTS)}) - x_{(NTS)}$$

E - represent Quantiz' Error.

$$\text{now if } x_{(NTS)} = 0, \rightarrow x_q(x_{(NTS)}) = \pm \frac{4\Delta}{2}$$

If  $x_{(NTS)}$  will assign anyone of nearest quanti' level if  $\frac{\Delta}{2}$  is approx'd then Quant' Error,  $\frac{\Delta}{2}$  i.e. then  $\pm \frac{\Delta}{2}$ .

Q3 Hence the max Quantiz' error will be  $\pm \frac{\Delta}{2}$

### Transmission BW of a PCM System

Let us assume that the Quantizer uses  $V$  no. of binary digits to represent each level. Then the no. of level that may be represented by  $V$  digits will be  $q = 2^V$  here  $q$  represents total no. of digital levels of a  $Q$  level Quantizer

If  $V = 4$  bits, then  $q = 2^4 = 16$  levels. Each sample is converted to  $V$  binary bits i.e. No. of bits/sample is  $V$ .

$$\begin{aligned} \text{WKT, } \text{No. of samples/second} &= \text{Sampling freqe}(f_s) \\ \therefore \text{No. of bits/second} &= \text{No. of bits/sample} \times \text{No. of samples/second} \\ \text{No. of bits/second} &= (\text{No. of bits/sample}) \times (f_s \text{ samples/second}) \\ &= (V \text{ bits/sample}) \times (f_s \text{ samples/second}) \end{aligned}$$

Q4 The no. of bits/second is known as Signalling rate of PCM.  
 $R = V \cdot f_s \text{ bits/sec}$   
 $f_s \gg f_m$ . Since Bandwidth needed for PCM transmission is given by the half of signalling rate.  
 $\therefore$  Trans' BW in Pm is

$$\text{BW in PCM} = \text{BW} \geq \frac{1}{2} R$$

$$\therefore B = \frac{1}{2} V \cdot f_s \approx 1.12 f_m$$

$$\boxed{B \geq V \cdot f_m}$$

10. fs U 2 fm  
10. 2x4-2

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Page / /

- ① A T1E signal having a BW of 4.2 MHz is transmitted using binary PCM system given that no of quantization level is 512. determine
- (i) code word length
  - (ii) Trans. Bandwidth
  - (iii) Final Bitrate
  - (iv) O/P Signal to Quantiz. Noise ratio

$$\Rightarrow f_m = 1.2 \text{ MHz}$$

$$Q = 512$$

$$q = 2^6 \quad 512 = 2^9; [0 = 9 \text{ bits}] \quad (\text{code word length})$$

$$BW \geq \frac{1}{2} r \quad \Rightarrow 16 \text{ fm}$$

$$BW \geq 37.8 \text{ MHz}$$

$$q = 16 \times 2 f_m = 9 \times 8.4 = 75.6 \text{ bit/sec}$$

- ② The info in analog waveform with max freq  $f_m = 8 \text{ kHz}$  is to be transmitted over an (2) M Level where the no of quantization level is 16. the quantization distortion is specified not to exceed  $\pm 1\%$  of peak to peak signal
- (i) Max no of bits/sample. That should be used in this PCM system.

What is (ii) minimum sampling rate and what is resulting bit transmission rate.

$$\Rightarrow Q = 16 = 2^4 \quad [2 = 1]$$

$$r = 16 \cdot f_s \\ = 16 \times 2 \times 8 \text{ kHz} = 256 \text{ bit/sec}$$

$$BW = \frac{1}{2} r = 12 \text{ kHz}$$

- ③ The information in an Analog signal voltage waveform is to be transmitted over a PCM system with an accuracy of  $\pm 0.1\%$  (full scale). The analog signal has an voltage of 100 mV and an amplitude of  $(-mp \text{ to } mp) = (-10 \text{ to } 10 \text{ mV})$ .
- (i) Find min sampling rate required
  - (ii) find No of bits in each PCM
  - (iii) find min. Bit rate ( $r$ )
  - (iv) find min. Absolute channel BW required for transmission

→ Here an accuracy is given as  $\pm 0.1\% \Rightarrow \epsilon_{max}$

$$\epsilon_{max} = \left| \frac{\Delta}{2} \right| = 0.001$$

$$\text{Step size } \Delta = 0.002$$

WKT the step size, no of quantiz. level and max value of the signal are related as

$$\Delta = \frac{2 \times x_{max}}{Q} = \frac{2 \times 10}{2} = 0.002 \quad x_{max} = mp \\ |x_{max}| = 10 \text{ opp.}$$

$$Q = 20 \cdot 10 \\ 0.002 \Rightarrow Q = 10 \times 10^3$$



Quantization, Noise / Error in PCM

A uniform quantizer noise is given by  $E = \Delta g(n)S - x_{(n+1)}$ .  
Let us assume that if  $x$  units to a linear uniform quantizer, has continuous amplitude  $-x_{\max}$  to  $x_{\max}$ .

This means that if  $x$  is  $\frac{1}{2}\Delta$  and off is  $\frac{1}{2}\Delta$ .

Total amplitude range becomes.

$$x_{\max} = x_{\min} + (\Delta x_{\max})$$

$$\Delta x_{\max} = \Delta x_{\min}$$

This total amplitude range is divided into Q bins.

Quantizer bin the step size will be

$$x_{\max} = x_{\min} \quad \Delta = \Delta x_{\max}$$

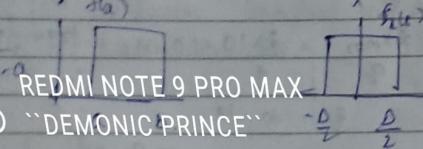
$$\frac{q}{2} \quad \frac{q}{2}$$

Now if the signal  $x(t)$  is normalized to minimum and maximum value equal  $\pm 1$ , then we have,

$$x_{\max} = 1; \quad -x_{\min} = -1; \quad \Delta = \frac{2}{q}$$

Now if the step size ' $\Delta$ ' is consider sufficiently small than it is may be assumed that quantizer error, ' $E$ ' will be uniformly distributed random variable

$$-\frac{\Delta}{2} < E_{\max} < \frac{\Delta}{2}, \text{ over the interval } (-\Delta/2, \Delta/2)$$



$$= \frac{1}{\Delta} - \frac{\Delta}{2} = \frac{1}{\Delta}$$

Probability density function,  $F$  = for quantizer error,  $E$  may be defined as.

$$F_E(E) = \begin{cases} 0 & \text{for } E \leq D_1, \\ \frac{E - D_1}{\Delta} & D_1 \leq E \leq D_2, \\ 1 & \text{for } E > D_2 \end{cases}$$

It may be observed that from fig and eqn of PDF, the Q.E has 'Q' average value function, i.e.  $\bar{x}$  = Signal to Quantizer noise ratio  $\bar{x}/\bar{E}$  the quantizer, defined as

$$\frac{\bar{x}}{N} = \frac{\text{Signal power}}{\text{Noise power}}$$

$$\text{Noise power} = \frac{V_{noise}^2}{R}$$

$V_{noise}$  is mean square value of noise voltage. The mean square value is  $= \frac{E(E^2)}{E^2} = 10^{20} \Omega^2$

$$1. \bar{x}^2 = E(E^2) = \int_{-\infty}^{\infty} E^2 F_E(E) dE$$

$$E(E^2) = \int_{-\infty}^{\infty} E^2 F_E(E) dE$$

$$= \left[ \frac{E^3}{3} \right] \left[ \frac{1}{\Delta} \right] = \frac{1}{3\Delta} \left[ \frac{2\Delta}{3} + \frac{\Delta}{3} \right] = \frac{1}{12} = \frac{1}{12}$$

Mean square noise would be.  $V_{noise}^2$  = mean square value.

$$V_{noise}^2 = \frac{1}{12}$$

Signal to Quantization noise ratio for linear Quantiz.

② Signal to noise is given by  $\frac{S}{N} = \frac{\text{Normalized Signal}}{\Delta V_{1/2}}$  - ①

③ Here the no. of bits (B) and Quantiz. level are related as  $q = 2^B$ .

④ Let us assume that i/p  $x(t)$  has continuous amplitude in the range  $\rightarrow -x_{\max}$  to  $x_{\max}$ .

Total amplitude range is  $x_{\max} - (-x_{\max}) = 2x_{\max}$ .

Now the step size  $\Delta$  will be.  $\Delta_{\max} = \frac{x_{\max}}{2}$ ,

$$q = 2^B \quad \Delta = \frac{2x_{\max}}{2^B},$$

$$\text{Now in ①} \quad \frac{S}{N} = \frac{\text{Normalized Sig}}{\Delta V_{1/2}}$$

$$= \frac{12 \text{ Norm. Sig}}{\frac{4(2x_{\max})}{2^B}} = \frac{12 \text{ Norm. Sig}}{\frac{4x_{\max}}{2^{B+2}}}$$

Let normalized signal power is denoted as  $P_{\text{sig}}$

$$\frac{S/N}{\text{in}} = \frac{3P_{\text{sig}} \times 2^{2B}}{x_{\max}^2} = 3P_{\text{sig}} \times 2^{2B}$$

required sol' for signal to Quantiz. noise ratio  
for linear Quantiz. in a PCM system

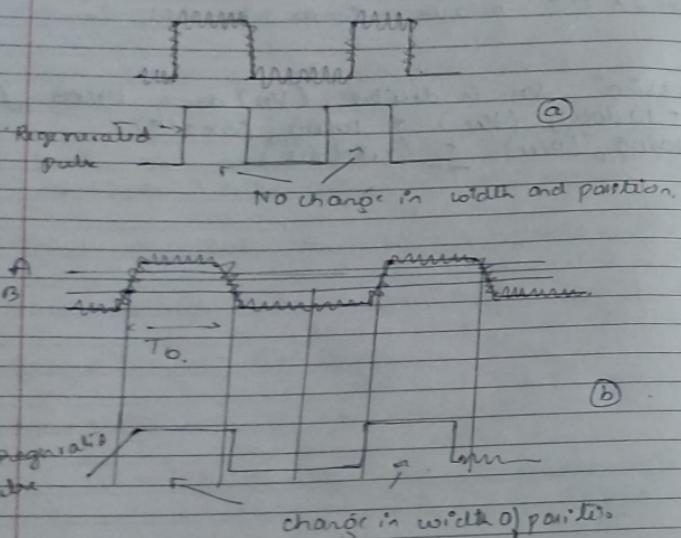
$x_{\max} = 1$ , then  $S/N$  will be  $\frac{x_{\max}}{\Delta} = \frac{1}{2^{B+2}}$  &  
also the signal power  $P$  is normalized i.e.  $P \leq 1$ .  
 $\frac{S}{N} = 3 \times 2^{2B}$  below.  $x_{\max} = 1$ ,  $P \leq 1$ .

③ Expressing  $S/N$  in decibels  $(S/N)_{\text{dB}} = 10 \log_{10} (S/N)$

$$= 10 \log_{10} (S/N) \leq 10 \log_{10} (3 \times 2^{2B})$$

$$= 10 \log_{10} (3) + 10 \log_{10} (2^{2B}) \approx (4.8 + 6B)$$

## Effect of noise on PAM System



- ② In PAM system noise is unaffected much as compared to PNRM, PWM, PPM system.
- In PAM system doesn't contain any information in its width or position of pulse hence PAM has much better noise immunity . compared to other system.

- ① Derive an expression for Signal to Quantization noise ratio for a PAM system which employs uniform quantization technique Given that input to the PAM system is a sinusoidal signal
- ② A PAM system uses a uniform quantizer followed by a 10 bit encoder Show that RMS signal to Quantization noise ratio is approximately given as  $(2.8 + 6R)$  dB

Sol:- Let us assume that modulating signal is sinusoidal voltage having amplitude equal to  $A_m$  V/m. The power of the signal will be  $V^2/R$  here  $V$  is RMS value

$$P = \frac{V^2}{R} \quad V = \left( \frac{A_m}{\sqrt{2}} \right)^2$$

$$P = \frac{A_m^2}{2R} \quad \text{--- (1)}$$

In case  $R=1$  and  $P$  is normalized in eq (1).

$$P = \frac{A_m^2}{2} \quad (R=1)$$

$$\begin{aligned} \text{HKT} \quad S/N &= \frac{A_m^2}{2} \times 2^{10} \\ N &= 3A_m^2 \times 2^{10}, \quad A_m = 1 \text{ mV} \\ &\quad 2 \times 10^{-3} \\ &= \frac{3A_m^2 \times 2^{10}}{2A_m^2} = \frac{3 \cdot 2^{10}}{2} = \underline{\underline{(1.5 \times 2^{10})}} \end{aligned}$$

Expressing in dB we get

$$\begin{aligned} \left[ \frac{S}{N} \right]_{dB} &= 10 \log_{10} \left[ \frac{S}{N} \right] = 10 \log_{10} (1.5 \times 2^{10}) \\ &= 10 \log_{10} 1.5 + 20 \log_{10} (2) \\ &= 1.7609 + 20 \times 3.010, \end{aligned}$$



$$\left(\frac{S}{N}\right) \approx 1.8 + 6 \text{ dB}$$

### NON-UNIFORM QUANTIZATION:

- If the Quantizer characteristic is NON linear and the step size is not constant instead if it is variable dependent on amplitude of S/P signal then the quantization is known as NON-uniform quantization.
- ① In this the step size is reduced with the reduction in signal level. For weak signal ( $P$  very  $p < 1$ ) the step size is small, therefore the quantization noise is reduced to improve the  $(S/N)_q$  ratio for low signals.
  - ② The step size is then varied according to signal level to the signal to noise ratio and adequately high. This is called NDQ.
  - ③ This is practically achieved through a process called Expanding (Compression + Expanding).

The Quant. Noise power.  $\Delta V^2/12$  this shows that in uniform Quant. Only the step size is fixed. The Quant. Noise power remains constant however the signal power is not constant. It is proportional to the square of the signal amplitude.

Hence the signal power will be small for weak signal therefore  $S/N$  ratio for low signal is very poor this.

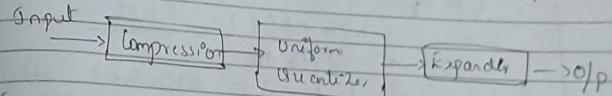
With this the quality of signal



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The remedy is to use companding

- ① Companding is a term derived from compression and expansion.



- ② In practice it is difficult to implement the non-uniform Quantizer because it is not known in advance about the change in the signal level therefore a particular method is used the weak signals are amplified and strong signal are attenuated before applying them to a uniform quantizer.

- ③ At receiver exactly opposite is followed which is called expansion. The compression of the signal at the transmitter and expansion at receiver is combined to be called companding.

- ④ A PCM system uses a uniform quantizer followed by 7 bit binary encoder. The bit-rate of the system is  $50 \times 10^6$  bits/sec. what is the maximum message signal band width for which the system operates satisfactorily calculate Q/F signal to quantized noise ratio when a full band modulating wave of frequency 1 MHz is applied to the input.

$$\text{W.L.C.T. given } R = 5 \times 10^6 \quad b = 7$$

$$T = b F_s$$

$$F_s = \frac{5 \times 10^6}{7} = 714,28 \times 10^3$$

(a) 3.87 MHz

$$\left(\frac{S}{N}\right)_1 = 1.76 + 6dB$$

$$1.76 + 6(1)$$

$$1.76 \cdot 6$$

- (i) A signal having a band width of 3.87 kHz is sample Quantized and coded by a PCM system. The coded bit signal is then transmitted over a channel at a transmission rate of 50 bits/sec. If hypothesis a transmission rate of 50 bits/sec. Determine the max signal to noise ratio that can be obtained by the system. The input signal has peak to peak value of 3V and RMS value of 0.3V.

$$W/N = \frac{S}{N} \cdot 2^B \cdot 2^{2B}$$

$$1 = 80k \text{ bits}$$

$$f_s = 2 \times 3.87$$

$$= 7.74 \text{ kHz}$$

$$P_s \cdot \frac{V^2}{R} = (0.1)^2 \cdot 0.09 \text{ W}$$

$$1 = 10, f_s$$

$$10 = \frac{80k}{7.74}$$

$$10 = 7.14 \text{ bits}$$

Take

$$3(0.04) \times 2^{27.14} \cdot 10 = 8$$

Time = 1

No. of

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• 1966.6P

• 1010g10 / 1966.6P

• 33dB

- (ii) The bandwidth of P/P signal to the sum is subdivided to 4kHz. The P/P signal varies in amplitude from -3.8V to 3.8V has the average power of 30mW. The required signal to noise ratio is given as 20dB. The PCM modulator produces binary opp assuming uniform Quantize.

- (i) Find the no of bits required for sample (ii) Given 30 bit resolution time multiplexed what would be the minimum required transmission bandwidth for the multiplexed signal.

$$B = 4kHz$$

$$\frac{30}{10} \times 10^3$$

$$T_{\max} = 3.8$$

$$P = 30 \text{ mWatt}$$

$$\frac{S}{N} = 20 \text{ dB}$$

$$100$$

$$\frac{S}{N} = \frac{3P}{2^B \cdot 2^{2B}} = 10, f_s$$

$$100 = 3(30 \times 10^{-3}) \times 2^{2B}$$

$$(3.8)^2$$

$$16044.44 = 2^{2B}$$

$$\log(16044.44) = 20, \log(2)$$

$$20 = 13.96$$

$$B = 6.9848 \approx 7 \text{ bits}/s$$

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$$Bw = Vf_m$$

$$Bw = 30 \times 10 \times f_m. \quad (30 \text{ PCM channel})$$

$$= 30 \times 10 \times 414$$

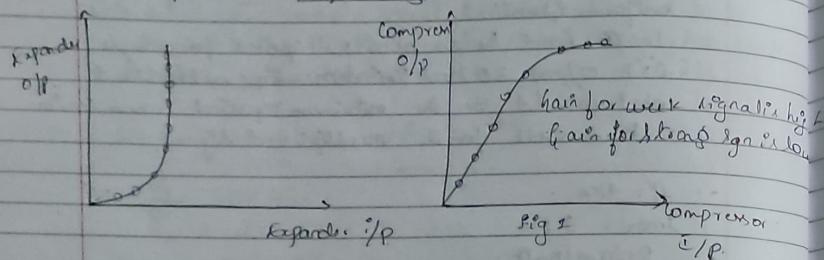
$$Bw$$

$$= 810 \text{ kHz}$$

$$r = 840 \times 2$$

$$\Rightarrow 1680 \text{ K bits/sec.}$$

### Compressor Companding :-



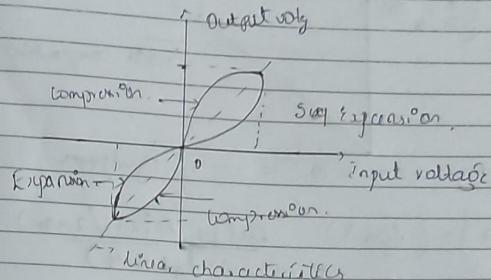
As shown in fig 1 the compressor provides higher gain to the weak signal and smaller gain to the strong input signal.

Thus weak signals are artificially boosted to improve the  $(S/N)_q$  ratio.

### Expander chara

As shown, this characteristic is exactly the inverse of compressor character. This ensures that artificially boosted

Signal by the compressor are brought back to their original amplitude at the receiver end.



Companding curve for PCM system.

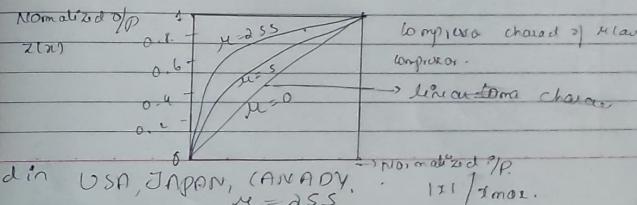
### Different types of compressor characteristics

1. M-law companding
2. A-law companding

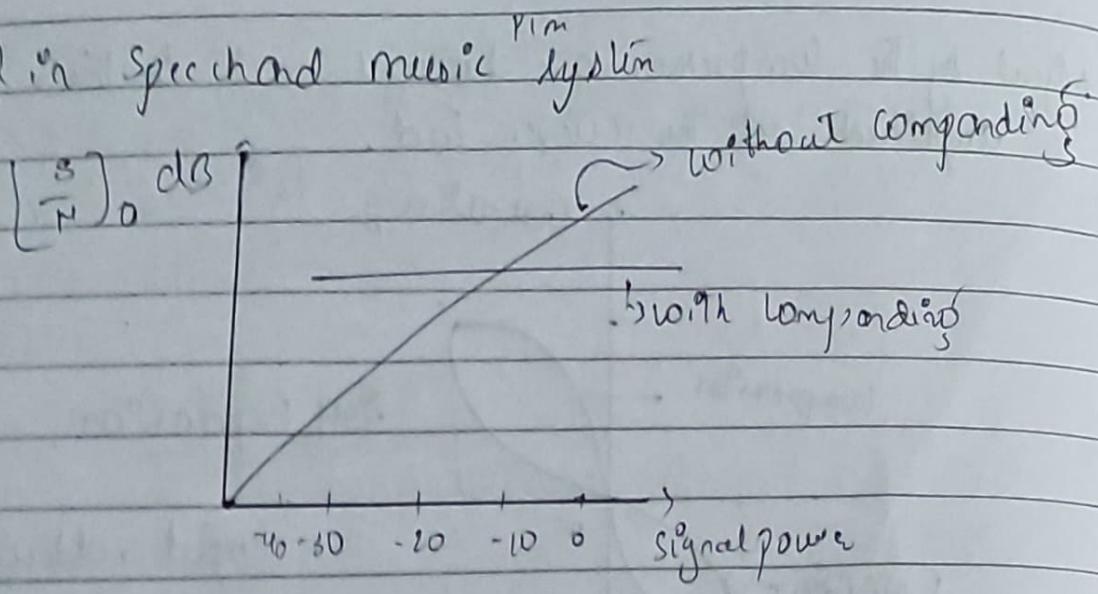
Ideally we need a linear compressor characteristic for small amplitude of the input signal and a logarithmic characteristic.

In practice this is achieved by using following method

- ① M-law companding :- n.



① On load speech and music system



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

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

$z(x) \rightarrow$  O/p of compressor

$x \rightarrow$  i/p to the compressor

$x_{\max} \rightarrow$  Normalised value of input i.e.  $|x|/x_{\max}$

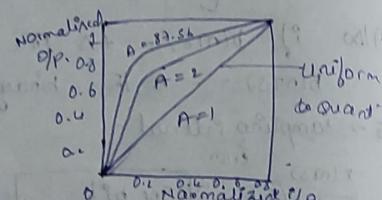
→ In the above companding compound characteristic is

piecewise made up of a linear segment for a low level

i/p and a logarithmic step for high level i/p

fig shows A-law compressor characteristics for different value of  $A$ .

$A=1$  corresponds to linear characteristic which corresponds to uniform quantization.



### Application of PCM:-

- ① It is used in telephone systems.
- ② In the space communication, space craft transmits signal to Earth, here the transmitted power is very low (10, 15W) and the distance are huge (few million km). Still due to high noise immunity only PCM system can be used in such application.

### Advantage

- ① Very high noise immunity.
- ② Due to digital nature of the signal, repeaters can be placed between transmitter and receiver.

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This is not possible in analog system.

③ Repeater further reduce the effect of noise.

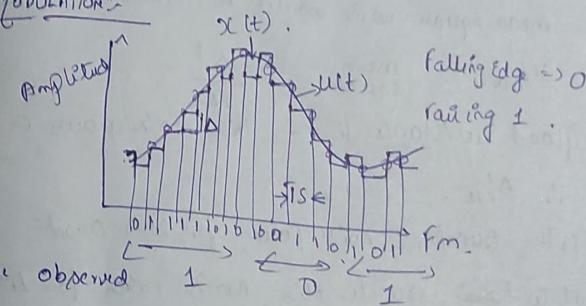
It is possible to store the PCM system can use various coding techniques so that only the desired portion can decode the received signal due to its digital nature.

### Pcm Advantages :-

① In Encoding, Decoding, Quantizing, Coding of PCM is complex.

② PCM requires large bandwidth as compared to other system.

### Delta Modulation



We have observed

in PCM that it transmits all the bits which are used to code a sample. Hence signaling rate and transmission channel bandwidth are quite large in PCM.

To overcome this problem D modulation is used.

D modulation transmits only 1 bit/sample. Here the present sample value is compared with the previous sample value and this result whether the amplitude is ↑ or ↓ in transmitted.

Q Up bignal  $x(t)$  is approximated to step signal by the Delta modulator. The step size is kept fixed difference b/w i/p signal  $x(t)$  and its linear approximated signal is confined to a level. +Δ and -Δ.

Q The difference is the approximated signal introduced by one step ( $\Delta$ ) w/o the difference is -ve. The approximated signal is reduced by  $\Delta$ .

Q When the step is reduced 0 is transmitted and if the step is +Δ it is transmitted hence for each sample, only one binary bit is transmitted.

Fig shows the analog signal  $x(t)$  and its linear approximated signal  $\hat{x}(t)$  by the delta modulator

Thus the principle of Dmod can be explained with the help of (a) The error, (b) sampled value  $x(nT_s)$  and last approximated sample is given as

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

error pronounced sample ↓ last sample approximated of  $x(t)$  sampled signal of  $x(t)$  has been waveform

If we assume  $u(nT_s)$  as the present sample approximation op. Then  $u((n-1)T_s) = \hat{x}(nT_s)$   
 Let us define a quantity  $b(nT_s)$  in such a way that  $b(nT_s) = \text{sgn}(e(nT_s))$

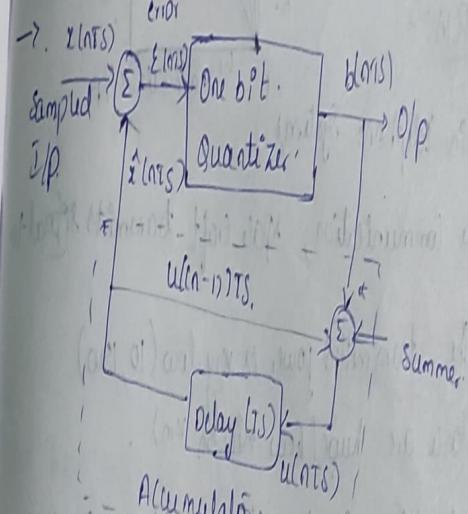
This means that depending on sign of  $e(nT_s)$ , the sign of step  $\Delta$  is decided. In other words we can write

$$b(nT_s) = \begin{cases} +\Delta & \text{if } x(nT_s) > \hat{x}(nT_s) \\ -\Delta & \text{if } x(nT_s) < \hat{x}(nT_s) \end{cases}$$

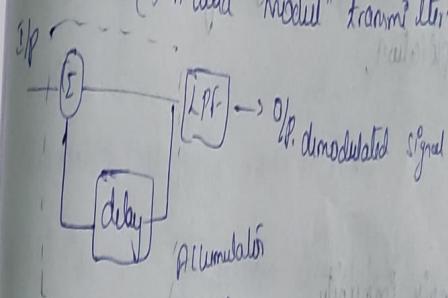
Also if  $b(nT_s) = +\Delta$  then binary 1 is transmitted.

$b(nT_s) = -\Delta$  then binary 0 is transmitted.

$T_s$  → sampling interval;

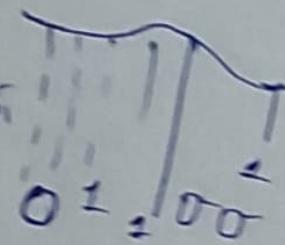


(a) A delta modulator transmitter



(b) A delta demodulator receiver

$+D \rightarrow$  one step ↑ (slope) > 1  
 binary  
digi.  
rise  $\rightarrow 11$   
fall edge  $\rightarrow 00$



④ Fig shows the it is modulated signal. The summer in the accumulator adds  $\pm D$  (adds Quantizer  $\pm \frac{1}{2}$ ) with previous sample approximation.

$$u(nTS) = u(nTS - TS) + (\pm \Delta)$$

$$u(nTS) = u(nTS) [u((n-1)TS)] + b(nTS)$$

⑤ The previous sample approx.  $u((n-1)TS)$  is restored by delaying one sample period  $TS$ . The sampled signal  $y(nTS)$  and this can approximated signal  $\hat{x}(nTS)$  are subtracted to get an error signal  $E(nTS)$ .

Then depending on sign of  $E(nTS)$  One bit quantizer generate an off of  $+D$  or  $-D$

If the step size is  $+D$  then binary one is transmitted and if the step size is  $-D$  then binary zero is transmitted.

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