Intersymbol Interference:

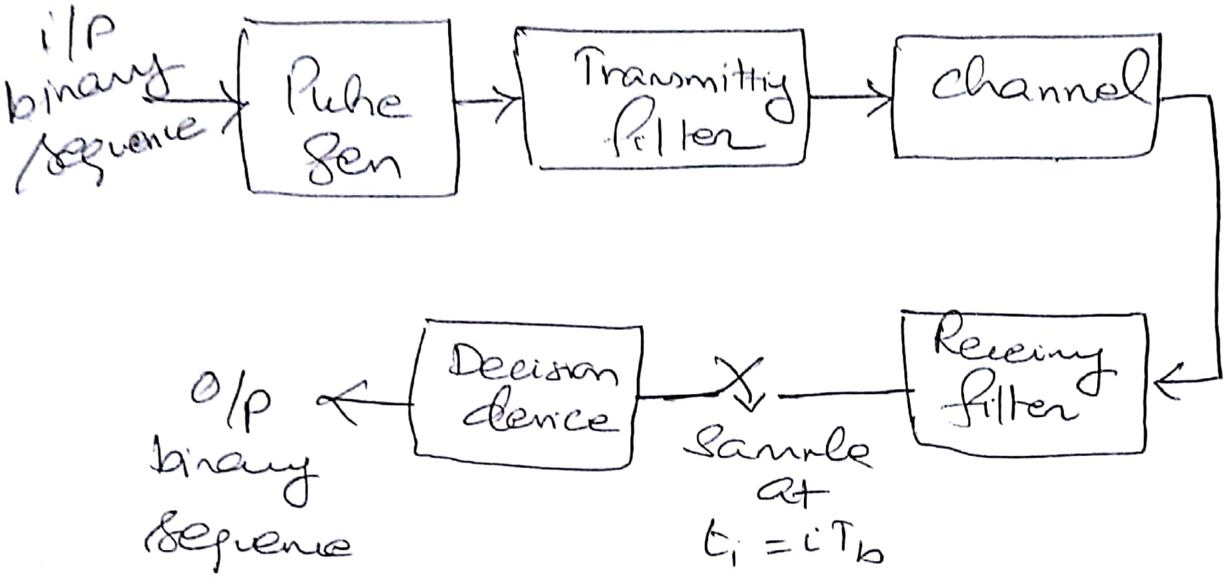
When a short pulse of duration T_b sec is transmitted through a bandlimited system, the frequency components of the input pulse are differentially attenuated and hence differenticly delayed by the system.

Consequently the pulse appearing at the o/p of the system is dispersed over an interval longer than T_b secs and interfere with each other. The overlapping of pulses is referred as inter symbol interference (ISI)

ISI arises due to imperfections in the overall frequency response of the system.

The presence of ISI increases the errors in the decision device at the receiver o/p.

Baseband binary PAM System

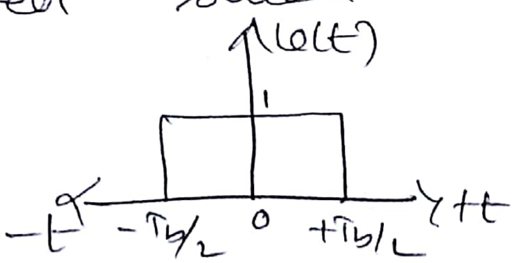


The i/p consists of a binary data sequence $\{b_k\}$ with a bit duration of T_b secs.

This sequence is applied to a pulse generator, producing the discrete PAM signal given by

$$x(t) = \sum_{k=-\infty}^{+\infty} a_k\, v(t - kT_b) \rightarrow (1)$$

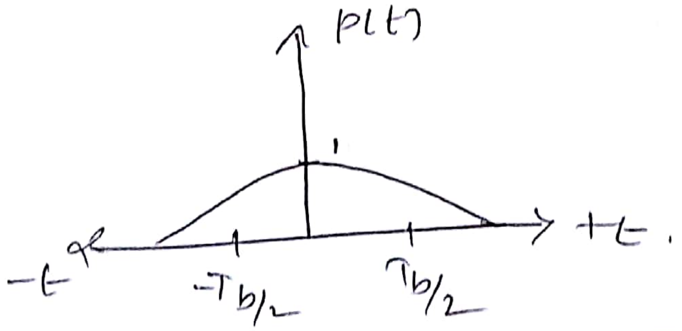
Where $v(t)$ denotes the basic rectangular pulse normalized such that $v(0) = 1$



coefficient a_k depends on
the o/p data and type of
format used
Signal $x(t)$ passed through
a transmitting filter $H_T(f)$
the resulting filter o/p is the
transmitted signal which is
modified as a result of
transmission through the
channel of transfer function
$H_C(f)$. The channel may
be coaxial cable or optical
fiber where the major
source of degradation is
dispersion the spreading
of pulse is called dispersion.
The channel o/p is passed
through a receiving filter
of transfer function $H_R(f)$
the filter o/p is sampled
periodically. when the
sample value ~~is~~ exceeds
the threshold value, the

decision is made in favor of Symbol 1 & if the Sample value is less than the threshold, then the decision is made in favor of Symbol 0. If the Sample value is equal to the threshold value exactly, then the decision is made in favor of Sym - 0 & Sym - 1.

Let $P(t)$ be the received Pulse shape such that $P(0) = 1$



Then the received Signal is Given by

$$y(t) = \mu \sum_{K=-\infty}^{+\infty} a_K P(t - KT_b) \rightarrow (2)$$

μ - Scaling factor depends on the channel

The o/p from the Sample is given by

$$y(t_i) = \mu \sum_{k=-\infty}^{+\infty} a_k \, p(t_i - kT_b)$$

$$= \mu \sum_{k=-\infty}^{+\infty} a_k \, p(iT_b - kT_b) \rightarrow (3)$$

Sub. $i = k$, in (3) we get

$$y(t_i) = \mu a_i \, p(0) + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{+\infty} p(iT_b - kT_b)$$

when $p(0) = 1$, then

$$y(t_i) = \mu a_i + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{+\infty} p(iT_b - kT_b) \rightarrow (4)$$

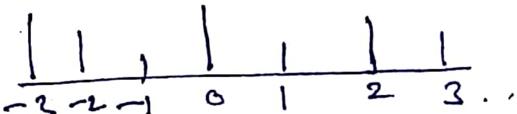
$y(t_i) = \mu a_i$ implies zero ISI and this is the sample value produced by the ith transmitted bit. The second term is due to the residual effect of all other transmitted bits in the decoding of the ith transmitted bit.

∴ The condition for the perfect reception in the absence of noise is given by

$$p(iT_b - kT_b) = \begin{cases} 1 & i = k \\ 0 & i \neq k \end{cases} \quad (5)$$

This condition is referred as Nyquist criterion for distortionless baseband binary transmission.

Let us consider a sequence of samples $\{p(nT_b)\}$ where $n = 0, \pm 1, \pm 2, \pm 3 \ldots$



According to sampling theorem sampling in the time domain produces periodicity in the frequency domain.

$\boxed{T_s = T_b}$

$$\therefore P_\delta(f) = R_b \sum_{n=-\infty}^{+\infty} P(f - nR_b) \rightarrow (6)$$

where $R_b = 1/T_b$

where $P_\delta(f)$ is the Fourier transform of an infinite periodic sequence of delta function of period T_b whose strengths are weighted by the respective samples of $p(t)$ given by

$$\sum_{m=-\infty}^{+\infty} p(mT_b)\,\delta(t-mT_b)$$

Taking Fourier transform we get.

$$P_\delta(f) = \int_{-\infty}^{+\infty} \left(\sum_{m=-\infty}^{+\infty} p(mT_b)\,\delta(t-mT_b) \right) e^{-j\omega t}\,dt \quad \rightarrow (7)$$

In order to impose the condition for zero ISI, Replace ~~m by~~ $i-k$ by m in (5)

we get

$$p(mT_b) = \begin{cases} 1 & m = 0 \\ 0 & m \neq 0 \end{cases} \quad \rightarrow (8)$$

Sub. $m = 0$ we get

$$P_\delta(f) = \int_{-\infty}^{+\infty} p(0)\,\delta(t)\,e^{-j\omega t}\,dt$$

$$= p(0) \int_{-\infty}^{+\infty} \delta(t)\,e^{-j\omega t}\,dt$$

$$= 1 \rightarrow (8)$$

Equating Eq. (6) & (9) we get

$$R_b \sum_{n=-\infty}^{+\infty} P(f - nR_b) = 1$$

$$\sum_{n=-\infty}^{+\infty} P(f - nR_b) = 1/R_b = T_b \rightarrow (10)$$

This is the requirement for zero ISI in the frequency domain.

Ideal Solution:

- Narrowest band is obtained by permitting only one non-zero components in Eq. (10).
- f lies within the band $-R_b/2$ to $+R_b/2$.

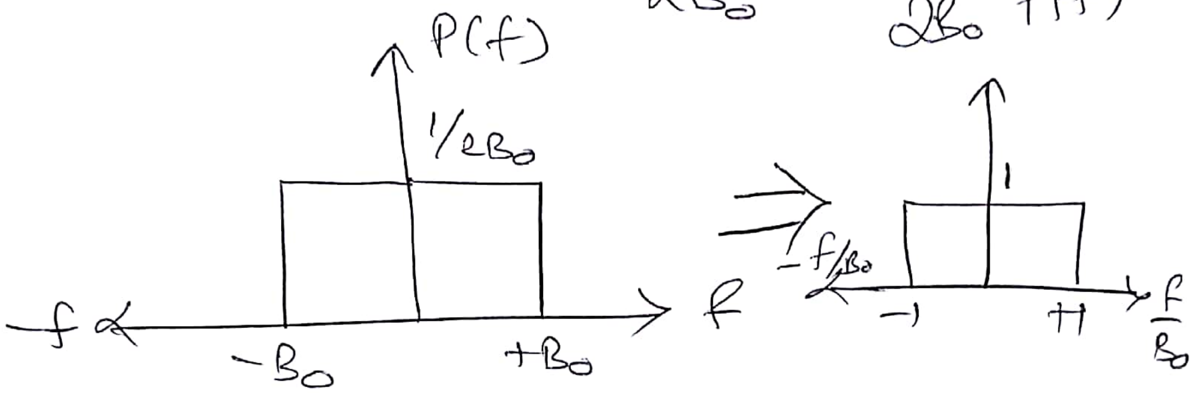
Ideal Solution:

Let $B_o = R_b/2$ then

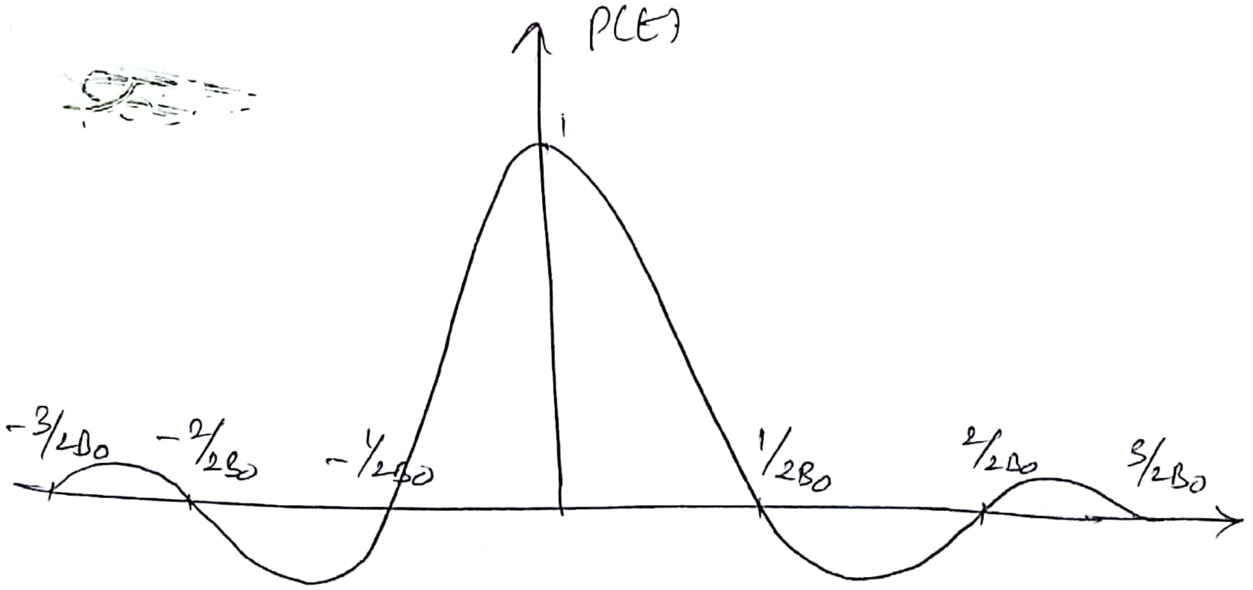
eq (10) can be written as

$$\sum_{n=-\infty}^{+\infty} P(f - n2B_o) = \frac{1}{2B_o}$$

when $n = 0$, $P(f) = \frac{1}{2B_o}$



$$P(f) = \frac{1}{2B_o} \text{rect}\left(f/2B_o \right)$$



⇒ Minimum Bandwidth

$B = B_o = R_b/2$.

Advantages of Sinc Pulse:

Requires min. Bandwidth

$$B = B_0 = R_b/2 = \frac{1}{2T_b}$$

Disadvantages:

1. The amplitude characteristics of P(f) should be flat from $-B_0$ to B_0 and zero elsewhere. This is not physically realisable.

2. Slow rate of decay of the sinc pulse till infinity.

Raised Cosine Spectrum:

- Extending the Bandwidth from B_0 to a value between B_0 and $2B_0$.
- 3 components are permitted in the series

$$P(f) + P(f - 2B_0) + P(f + 2B_0) = \frac{1}{2B_0}$$

$$-B_0 \leq f \leq B_0$$

- $P(f)$ is Satisfied the Raised Cosine Spectrum.

- Frequency characteristics of Raised Cosine Spectrum has a flat portion and a Rolloff Sinusoidal Portion that has a form.

$$P(f) = \begin{cases} \frac{1}{2B_0} & |f| < f_1 \\ \frac{1}{4B_0}\left\{1 + \cos\left[\frac{\pi(|f| - f_1)}{2B_0 - 2f_1}\right]\right\}, & f_1 \leq |f| < 2B_0 - f_1 \\ 0 & |f| \geq 2B_0 - f_1 \end{cases}$$

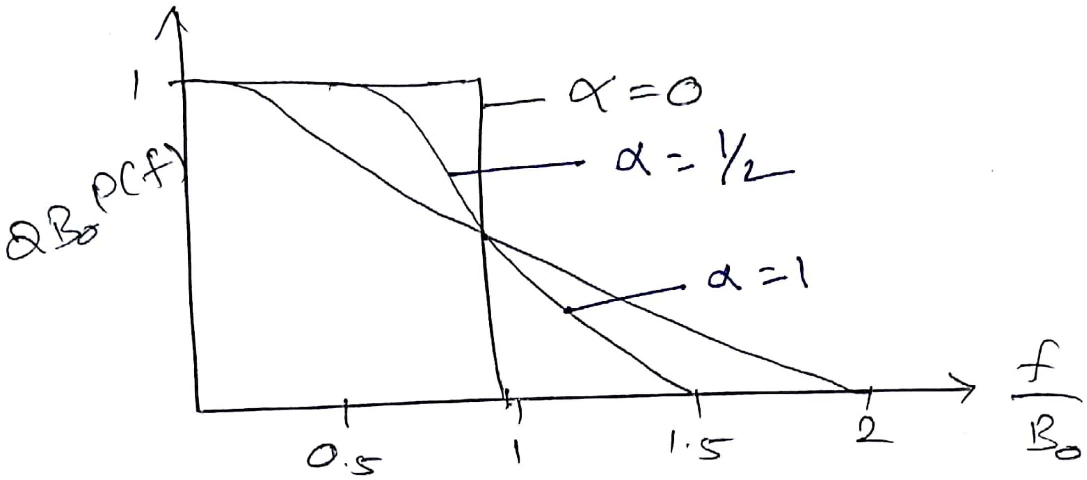
Frequency f_1 and bandwidth B_0 are related by

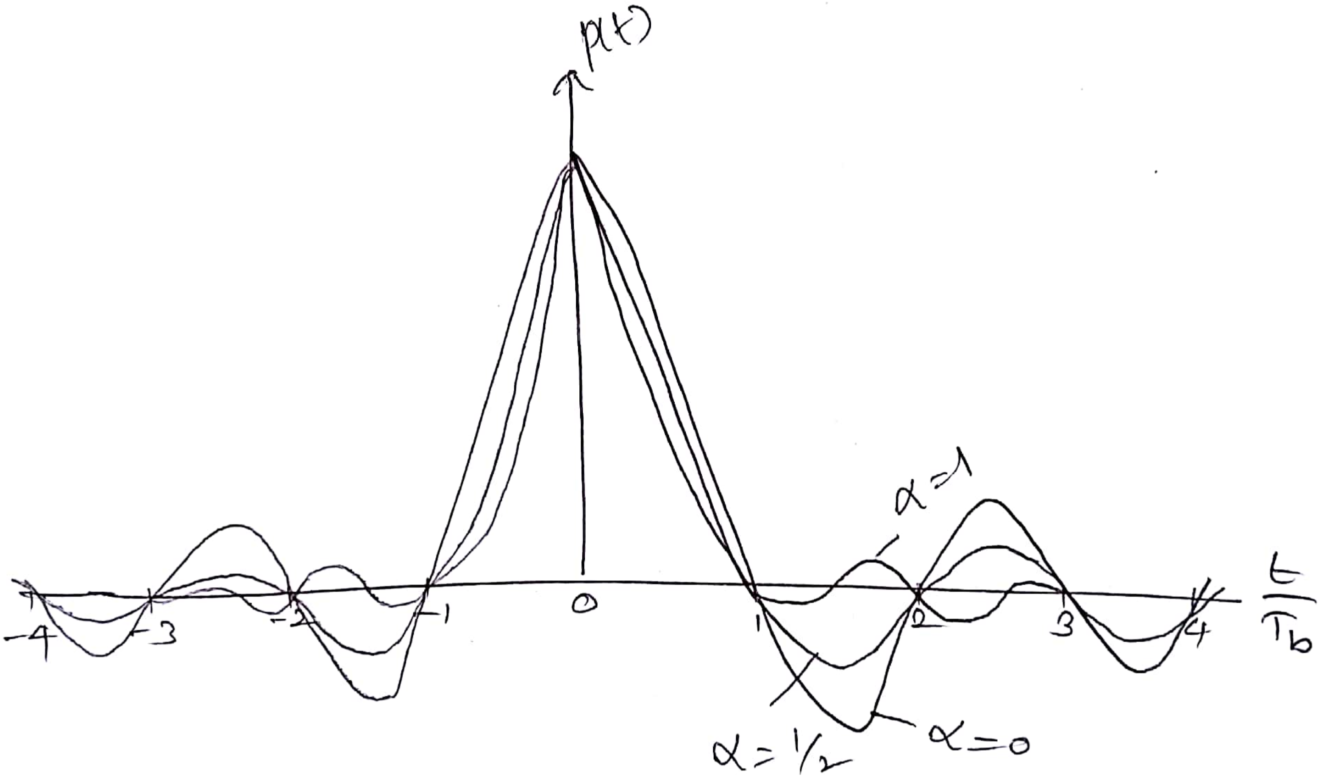
$$\alpha = 1 - \frac{f_1}{B_0}$$

α - Rolloff factor

when $\alpha = 0$, $f_1 = B_0 \Rightarrow$ Min. Bandwidth Solution.

Normalized $P(f)$





Inverse fourier transform $P(f)$ is given by

$$p(t) = \text{Sinc}(2B_0 t) \cdot \frac{\cos(2\pi \alpha B_0 t)}{1 - 16\alpha^2 B_0^2 t^2}$$

$\text{Sinc}(2B_0 t) \rightarrow$ Associated with ideal LPF.

$\frac{1}{|t|^2}$ → reduces the tails of the pulse considerably below that obtained from the ideal LPF.

- when α increases from 0 to unity, the amount of ISI decreases.

- Zero crossings at $t = \pm 3T_b/2$, $\pm 5T_b/2$, in addition to usual zero crossings at the sampling instants $t = \pm T_b, \pm 2T_b \ldots$

Bandwidth $B = 2B_0 - f_1$

where $f_1 = B_0(1-\alpha)$

Correlative coding:

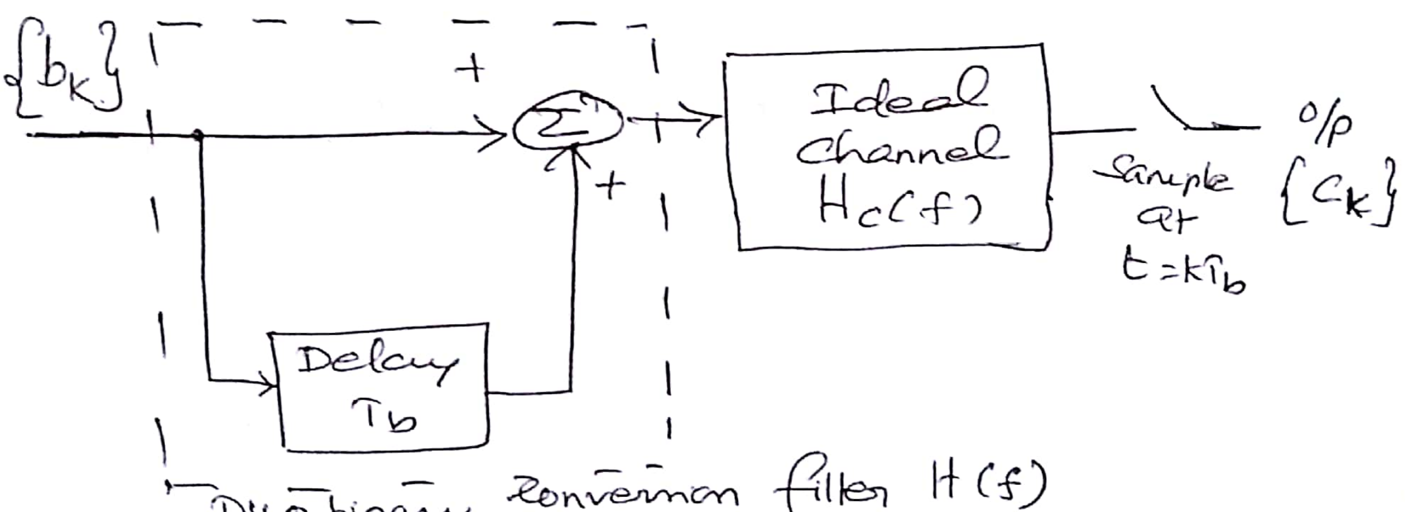
49

- Practical means of achieving the theoretical maximum signalling rate of $2B_0$ bits/sec in a bandwidth of B_0 Hz is called correlative coding.

Duo binary Signalling:

- Duo implies doubling the transmission capacity of a straight binary system.

Duo binary Signalling Scheme.



$b_k \rightarrow$ I/p binary sequence

Let b_k be the i/p binary sequence consists of uncorrelated binary digits each having a duration of T_b sec and represented in Polar format as

Sym - 1 - +1 V

Sym - 0 - -1 V

When this sequence is applied to duobinary conversion filter it is converted into a three level o/p namely −2, 0 & +2 V

Sequence is passed through a simple filter consists of delay element.

For every unit impulse applied to the i/p of this filter, we get two unit impulses spaced at T_b secs. apart at the filter at o/p.

The o/p from the duobinary conversion filter is given by

$$c_k = b_k + b_{k-1}$$

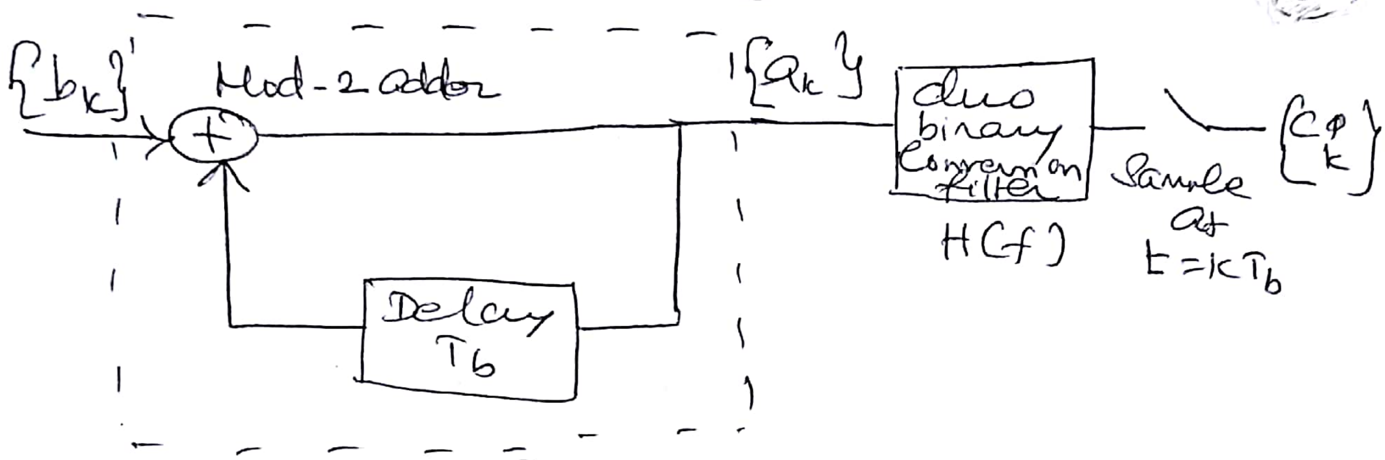
Then the o/p from the receiver is given by

$$b_k = c_k - b_{k-1}$$

Drawbacks:

If one bit is erroneous, the remaining bits become erroneous.

Precoded duobinary scheme:



Precoder

Let $\{b_k\}$ be the i/p to the Precoder and the o/p from the Precoder is given by

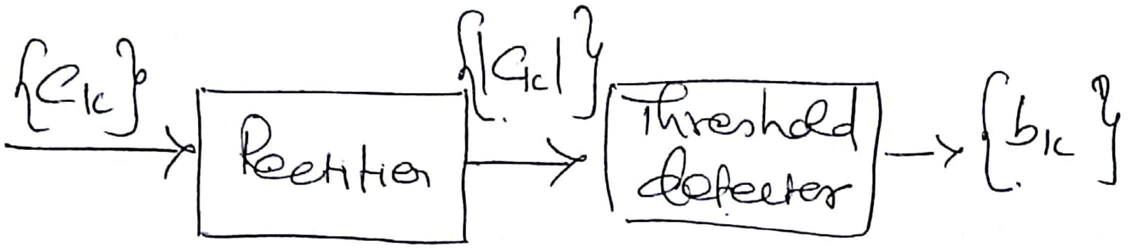
$$a_k = b_k \oplus a_{k-1} \quad \text{Mod-2}$$

The resulting Precoder o/p is $\{a_k\}$ is next applied to the duobinary coder. Then, the o/p from the duobinary coder is given by

Polar form

$$c_k = a_k + a_{k-1}$$

At the receiver side, the received sequence $\{c_k\}$ is applied to a rectifier followed by Threshold detector.

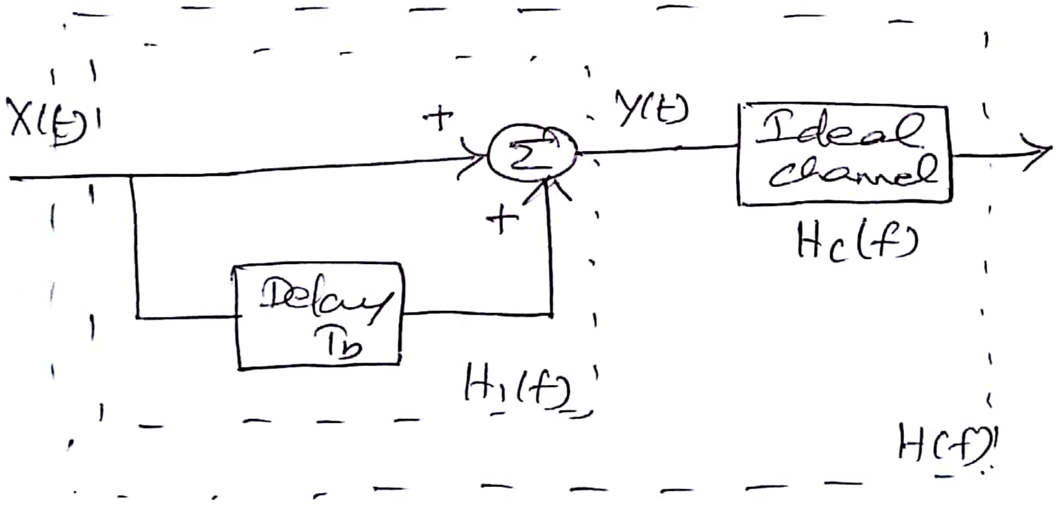


decision rule:

$$c_k = \begin{cases} \pm 2 \text{ volts} & \text{if } b_k \text{ represents Sym-0} \\ 0 \text{ volt} & \text{if } b_k \text{ represents Sym 1} \end{cases}$$

$$b_k = \begin{cases} \text{Sym } 0 & \text{if } |c_k| > 1\,V \\ \text{Sym } 1 & \text{if } |c_k| < 1\,V \end{cases}$$

Impulse response of duo binary conversion filter:



The duo binary conversion filter consists of an ideal delay element. Let X(t) be the input to the duo binary conversion filter and the o/p from the duo binary conversion filter is given by

$$Y(t) = X(t) + X(t - T_b) \rightarrow (1)$$

Taking fourier transform of eq (1) we get

$$Y(f) = X(f) + X(f) e^{-j\omega T_b}$$

$$= X(f) \left(1 + e^{-j\omega T_b}\right)$$

$$\frac{Y(f)}{X(f)} = 1 + e^{-j\omega T_b} = H_1(f) \rightarrow (2)$$

where $H_1(f)$ be the transfer function of the duo binary conversion filter. and T_b be the bit duration.

For an ideal channel of bandwidth B_0 which is equal to $R_b/2$, the transfer function of the ideal channel is given by

$$H_c(f) = \begin{cases} 1 & |f| \leq R_b/2 \\ 0 & |f| > R_b/2 \end{cases} \rightarrow (3)$$

The overall transfer function of the System is given by

$$H(f) = H_1(f) \cdot H_c(f)$$

$$= \left(1 + e^{-j2\pi f T_b}\right) H_c(f) \rightarrow 4$$

Sub eq. 3 in 4 we get

$$H(f) = \begin{cases} 1 + e^{-j2\pi f T_b} & |f| \leq R_b/2 \\ 0 & \text{Otherwise} \end{cases} \rightarrow 5$$

Impulse Response can be obtained by taking the inverse fourier transform of H(f) as given by

$$h(t) = \int_{-\infty}^{+\infty} H(f)\, e^{+j2\pi f t}\, dt$$

$$= \int_{-R_b/2}^{+R_b/2} \left(1 + e^{-j2\pi f T_b}\right) e^{+j2\pi f t}\, df$$

$$= \int_{-R_b/2}^{+R_b/2} e^{+j2\pi f t}\, df + \int_{-R_b/2}^{+R_b/2} e^{+j2\pi f (t - T_b)}\, df$$

$$= \left[\frac{e^{j2\pi f t}}{j2\pi t}\right]_{-R_b/2}^{+R_b/2} + \left[\frac{e^{j2\pi f(t-T_b)}}{j2\pi(t-T_b)}\right]_{-R_b/2}^{R_b/2}$$

$$= \frac{e^{j\cancel{2}\pi \frac{R_b}{\cancel{2}} t} - e^{-j\cancel{2}\pi \frac{R_b}{\cancel{2}} t}}{j2\pi t} + \frac{e^{j\cancel{2}\pi \frac{R_b}{\cancel{2}}(t-T_b)} - e^{-j\cancel{2}\pi \frac{R_b}{\cancel{2}}(t-T_b)}}{j2\pi(t-T_b)}$$

$$= \frac{\cancel{j2}\sin(\pi R_b t)}{\cancel{j2}\pi t} + \frac{\cancel{j2}\sin \pi R_b(t-T_b)}{\cancel{j2}\pi(t-T_b)}$$

Multiply & Divide by R_b we get

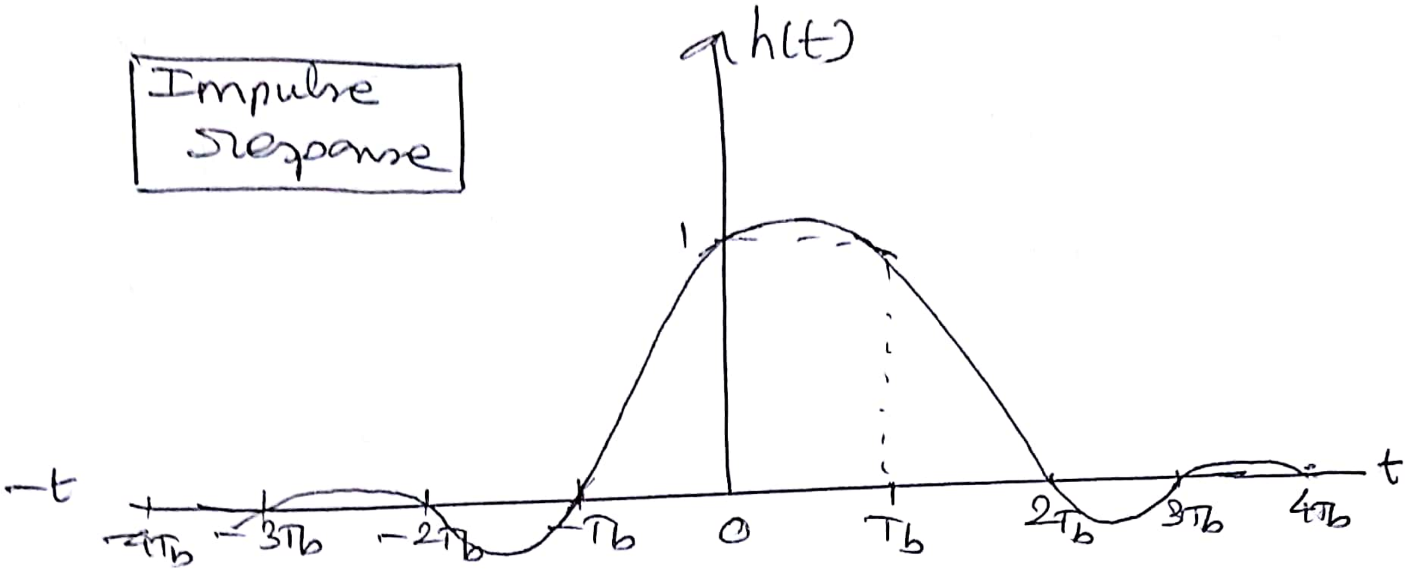
$$H(t) = R_b \frac{\sin(\pi R_b t)}{(\pi R_b t)} + R_b.\frac{\sin \pi R_b(t-T_b)}{\pi R_b(t-T_b)}$$

$$H(t) = R_b.\,\mathrm{Sinc}(R_b t) + R_b\,\mathrm{Sinc}\,R_b(t-T_b)$$

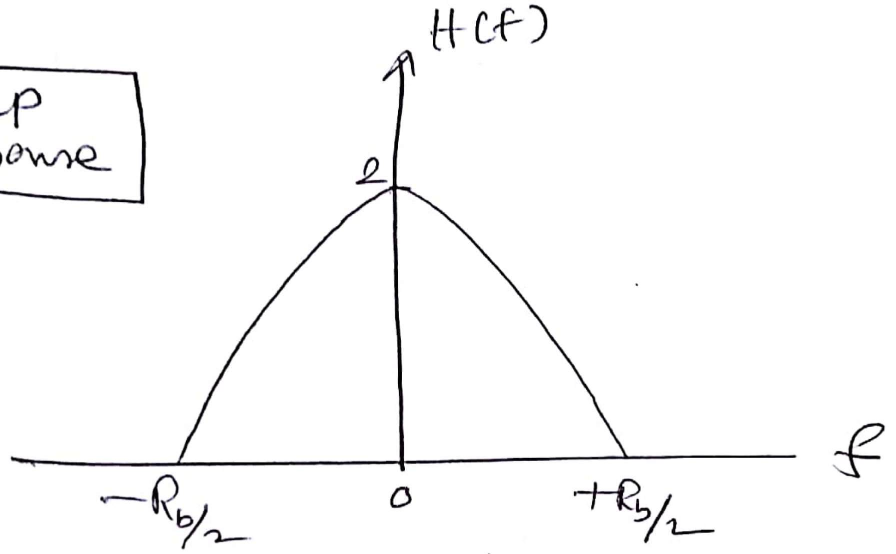
Hence for ~~eq.~~ every unit impulse applied to the input of this filter, two unit impulses spaced T_b secs apart are produced at the filter o/p.

Scanned by CamScanner

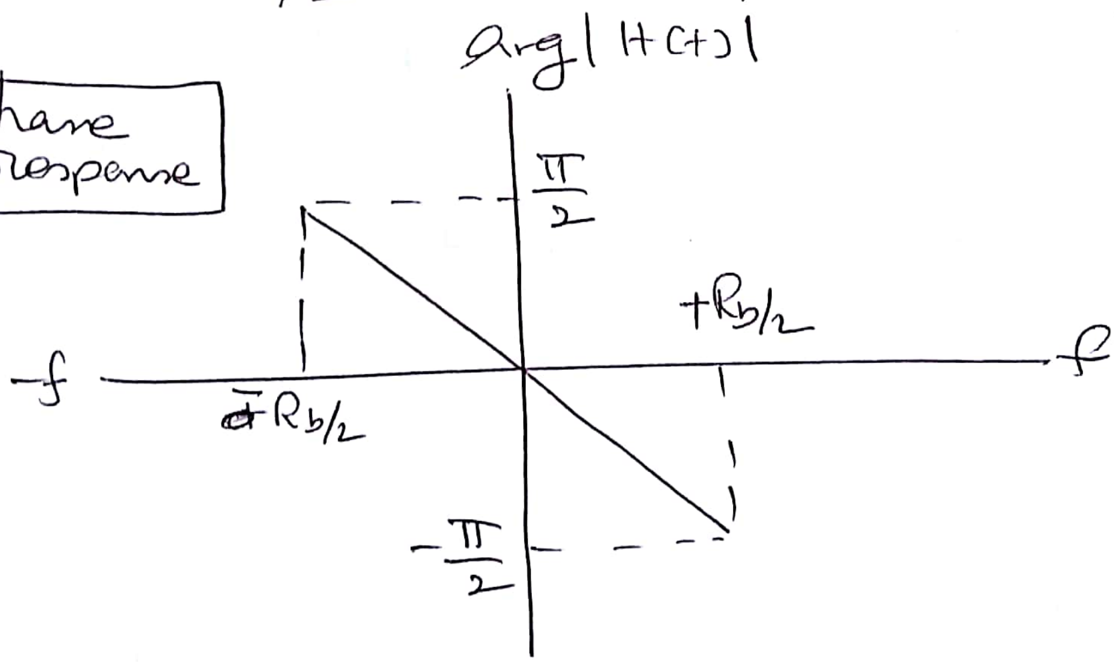
Impulse response



Amp response



Phase response



The binary data 0010110 is applied to the i/p of a duo binary system without a precoder. Construct the duo binary coder o/p and the corresponding receiver o/p without a precoder.

b_k	0	0	1	0	1	1	0
b_k in Polar form	-1	-1	+1	-1	+1	+1	-1
$c_k = b_k + b_{k-1}$ (+1)	0	-2	0	0	0	+2	0
$b_k = c_k - b_{k-1}$ (+1)	-1	-1	+1	-1	+1	+1	-1
	0	0	1	0	1	1	0

The binary data 001101001 is applied to the i/p of a duobinary system. (a) Construct the duobinary coder o/p & the corresponding receiver o/p without a precoder. (b) Suppose that due to error during transmission the level at the receiver i/p produced by the second digit is reduced to zero. Construct the new receiver o/p.

b_k	0	0	1	1	0	1	0	0	1
b_k in Polar form	-1	-1	$+1$	$+1$	-1	$+1$	-1	-1	$+1$
$c_k = b_k + b_{k-1}$ (+1)	0	-2	0	$+2$	0	0	0	-2	0
(+1) $b_k = c_k - b_{k-1}$	-1	-1	$+1$	$+1$	-1	$+1$	-1	-1	$+1$
	0	0	1	1	0	1	0	0	1

(b)

c_k	0	0	0	$+2$	0	0	0	-2	0
$b_k = c_k - b_{k-1}$ (+1)	-1	$+1$	-1	$+3$	-3	$+3$	-3	$+1$	-1
	0	1	0	1	0	1	0	1	0

The binary data 001101001 is applied to the i/p of a duobinary system.

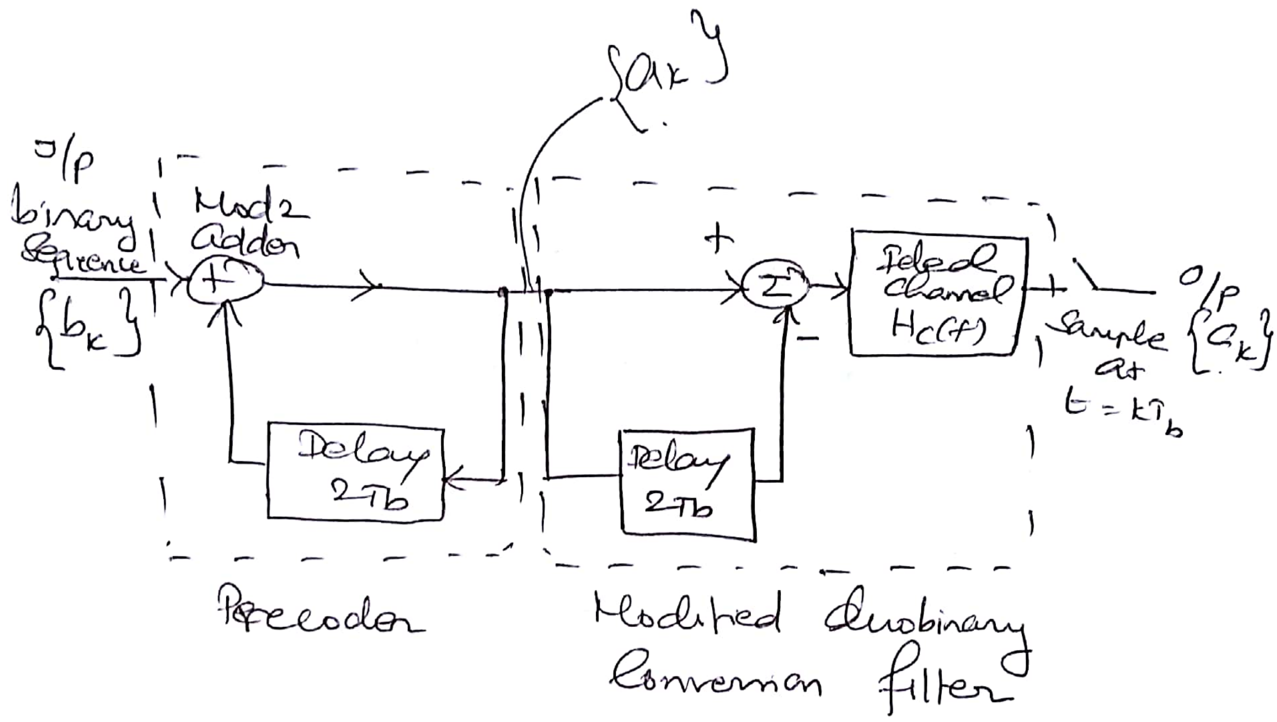
(a) Construct the duo binary coder o/p & the corresponding receiver o/p with a precoder

(b) Suppose that due to error during transmission the level at the receiver i/p produced by the second digit is reduced to zero, construct the new receiver o/p.

b_k	0	0	1	1	0	1	0	0	1
$a_k = b_k \oplus a_{k-1}$ 1	1	1	0	1	1	0	0	0	1
a_k in polar form	+1	+1	−1	+1	+1	−1	−1	−1	+1
$c_k = a_k + a_{k-1}$ +1	+2	+2	0	0	+2	0	−2	−2	0
$\lvert c_k \rvert$	2	2	0	0	2	0	2	2	0
b_k	0	0	1	1	0	1	0	0	1
(b) c_k	+2	0	0	0	+2	0	−2	−2	0
$\lvert c_k \rvert$	2	0	0	0	2	0	2	2	0
b_k	0	1	1	1	0	1	0	0	1

Modified Duobinary Technique:

- involves a correlation span of two binary digits.
- Subtracts input binary digits spaced $2T_b$ secs



Let $\{b_k\}$ be the input to the Precoder, then the o/p from the precoder is given by

$$a_k = b_k \oplus a_{k-2} \quad (\text{Mod } 2)$$

The o/p from the Modified duo binary Conversion filter is given by

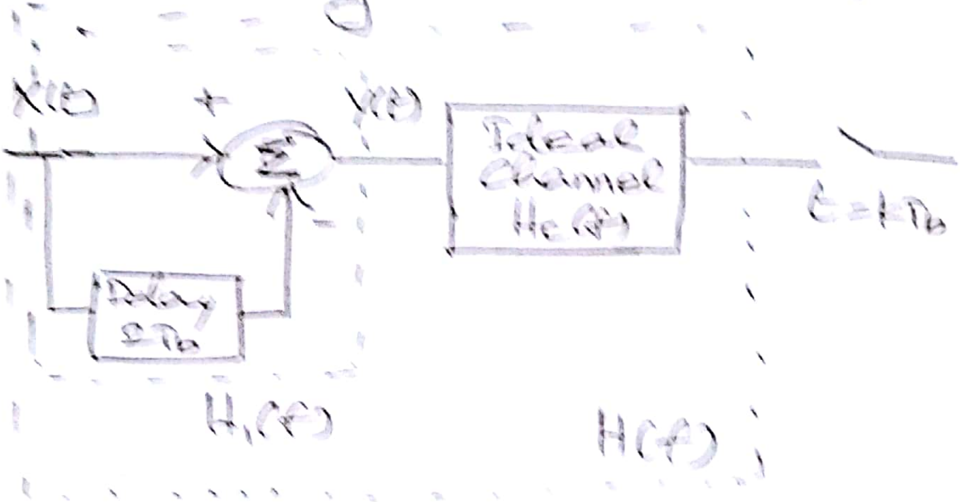
$$c_k = a_k - a_{k-2}$$

When a_k can take ± 1, then c_k can take three values as $-2V$, 0 & $+2V$

the original sequence at o/p can be detected based on the following decision rule:

$$b_k = \begin{cases} \text{Sym } 0 & \text{if } |c_k| < 1V \\ \text{Sym } 1 & \text{if } |c_k| > 1V \end{cases}$$

Impulse response of Modified duo binary conversion filter:



Let $x(t)$ be the i/p to modified duo binary conversion filter and the o/p from the filter is given by

$$y(t) = x(t) - x(t - 2T_b) \rightarrow (1)$$

Taking fourier transform of eq (1) we get

$$Y(f) = X(f)\left[1 - e^{-j2\pi f(2T_b)}\right]$$

$$H_1(f) = \frac{Y(f)}{X(f)} = 1 - e^{-j2\pi f(2T_b)} \rightarrow (2)$$

$$= 1 - e^{-j4\pi f T_b}$$

where $H_1(f)$ be the transfer function of the duobinary conversion filter and T_b be the bit duration.

For an ideal channel of bandwidth B_o which is equal to $R_b/2$, the transfer function of the ideal channel is given by

$$H_c(f) = \begin{cases} 1 & |f| \leq R_b/2 \\ 0 & |f| > R_b/2 \end{cases} \rightarrow (3)$$

The overall transfer function of the system is given by

$$H(f) = H_1(f) \cdot H_c(f)$$
$$= \left(1 - e^{-j4\pi f T_b}\right) H_c(f) \rightarrow (4)$$

Sub. (3) in (4) we get

$$H(f) = \begin{cases} 1 - e^{-j4\pi f T_b}, & |f| \leq R_b/2 \\ 0, & \text{otherwise} \end{cases} \rightarrow (5)$$

Impulse response can be obtained by taking the inverse fourier transform of $H(f)$ as given by $h(t) = \int_{-\infty}^{+\infty} H(f)\, e^{j2\pi f t}\, df$

$$h(t) = \int_{-R_b/2}^{+R_b/2} \left(1 - e^{-j4\pi f T_b}\right) e^{j2\pi f t}\, df$$

$$= \int_{-R_b/2}^{+R_b/2} e^{j2\pi f t}\, df - \int_{-R_b/2}^{+R_b/2} e^{j2\pi f (t-2T_b)}\, df$$

$$= \left[\frac{e^{j2\pi f t}}{j2\pi t}\right]_{-R_b/2}^{+R_b/2} - \left[\frac{e^{j2\pi f (t-2T_b)}}{j2\pi (t-T_b)}\right]_{-R_b/2}^{+R_b/2}$$

$$= \frac{e^{j\cancel{2}\pi \cdot \frac{R_b}{\cancel{2}} \cdot t} - e^{j\cancel{2}\pi(-R_b/\cancel{2})t}}{j2\pi t} - \left(\frac{e^{j\cancel{2}\pi \frac{R_b}{\cancel{2}}(t-2T_b)} - e^{-j\cancel{2}\pi R_b/\cancel{2}(t-2T_b)}}{j2\pi(t-T_b)}\right)$$

$$= \frac{\cancel{j2} \sin(\pi R_b t)}{\cancel{j2}\pi t} - \frac{\cancel{j2} \sin \pi R_b(t-2T_b)}{\cancel{j2}\pi(t-2T_b)}$$

Multiply & divide by R_b we get

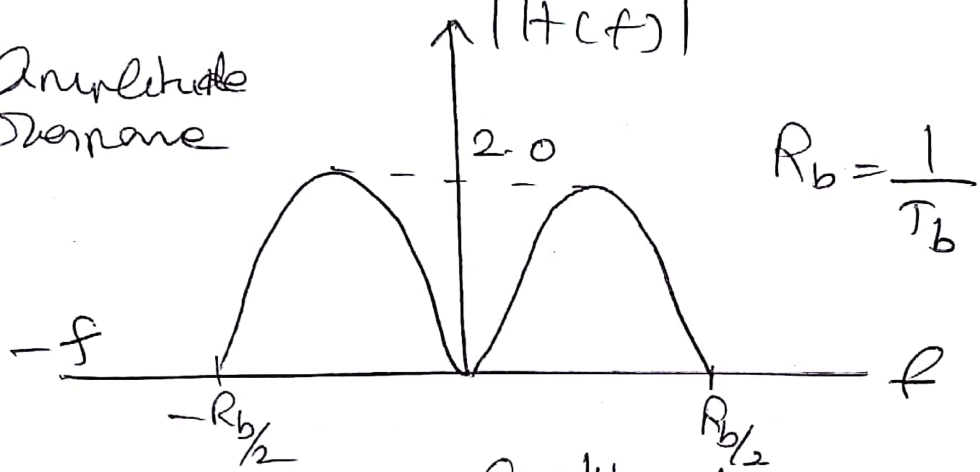
$$H(f) = R_b \frac{\sin \pi R_b t}{\pi R_b t} - R_b \frac{\sin \pi R_b (t-2T_b)}{\pi R_b (t-2T_b)}$$

$$= R_b \operatorname{sinc}(R_b t) - R_b \operatorname{sinc} R_b(t-2T_b)$$

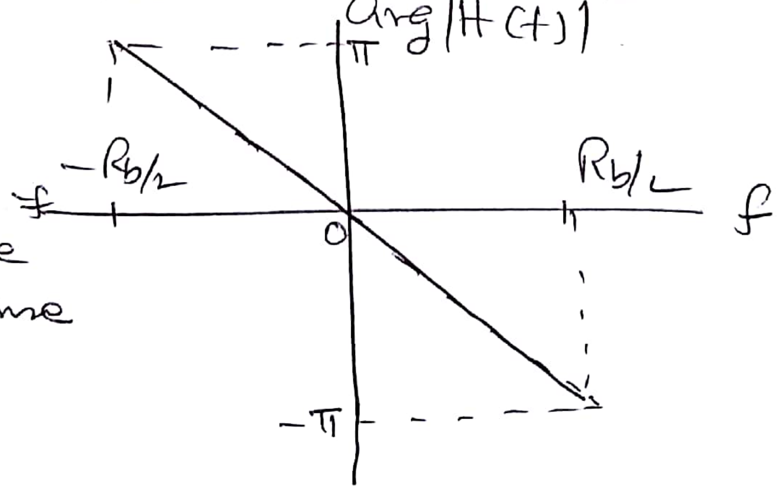
The impulse response of the Modified duo binary coder consists of two sinc pulses that are spaced by $2T_b$ secs.

Frequency response:

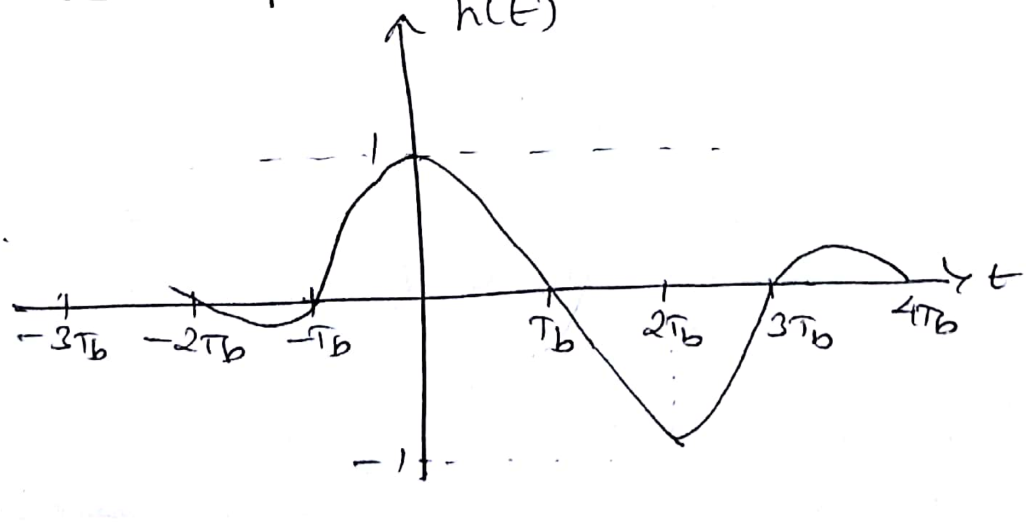
Amplitude response



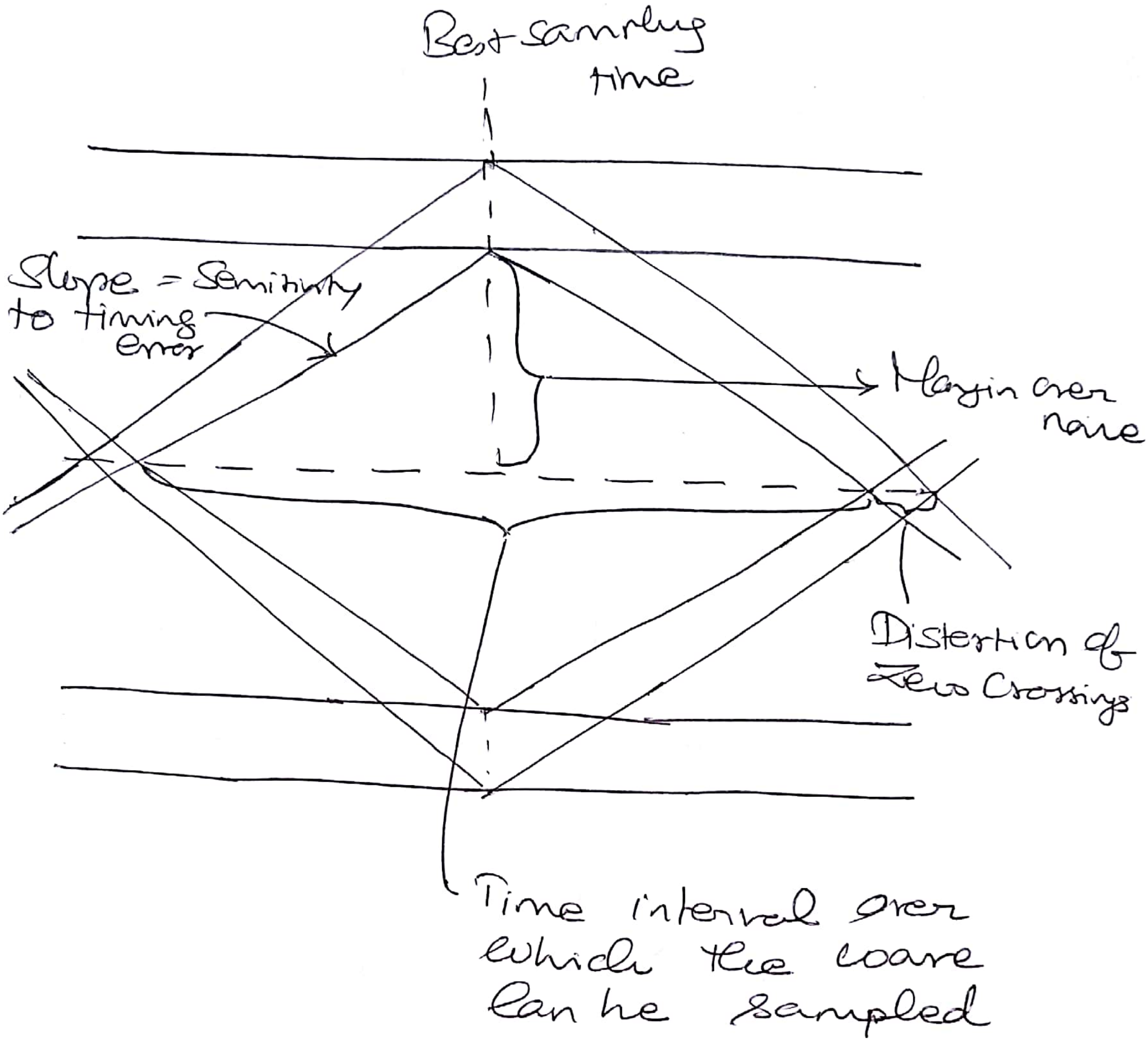
Phase response



Impulse response:



Eye Pattern:



- Used to study ISI in a PCM System
- Received wave is applied to the vertical deflection plates of an oscilloscope and a sawtooth wave applied to horizontal deflection plates at the rate of $R = 1/T$ the resulting display is called Eye Pattern

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The interior region of eye is called Eye Opening

WIDTH of the Eye Opening:
- time interval over which the received wave can be sampled without error from ISI

HEIGHT of the Eye Opening:
- defines the noise margin at a specified sampling time

RATE of closure of eye
- defines the sensitivity of the system to timing error.

When the ISI is severe, the eye will be completely closed.

Baseband M-ary PAM Systems:

- O/p from the Pulse Generator takes on one of the M possible amplitude levels.

= Symbol duration

$$T = T_b \log_2 M$$

where M typically integer Power of 2.

- M ary PAM System can transmit information at a rate that is $\log_2 M$ faster than the corresponding binary PAM system.
- for a fixed bit rate, ~~the~~ the M-ary PAM requires less channel Bandwidth than a binary PAM. System.
- To realize the same BER, M ary PAM requires more Power than binary PAM System.

The binary data 011100101 are applied to the i/p of a modified duo binary system

(a) Construct the modified duo binary coder o/p & the corresponding receiver o/p without a precoder

(b) Suppose that due to error during transmission, the level produced by the 3rd digit is reduced to zero, construct the new receiver o/p

b_k	0	1	1	1	0	0	1	0	1
b_k in Polar form	-1	+1	+1	+1	-1	-1	+1	-1	+1
$c_k = b_k - \overset{+1\ +1}{b_{k-2}}$	-2	0	+2	0	-2	-2	+2	0	0
$\hat{b}_k = c_k + \overset{+1\ +1}{b_{k-2}}$	-1	+1	+1	+1	-1	-1	+1	-1	+1
b_k	0	1	1	1	0	0	1	0	1
c_k	-2	0	0	0	-2	-2	+2	0	0
$b_k = c_k + \overset{+1\ +1}{b_{k-2}}$	-1	+1	-1	+1	-1	-1	+1	-1	+1
	0	1	0	1	0	0	1	0	1

X

The binary data 011100101 are applied to the i/p of a modified duo binary coder

(a) Construct the modified duo binary coder o/p & the corresponding receiver o/p with a precoder

(b) Suppose that due to error during transmission, the level produced by 3rd digit is reduced to zero. Construct the new receiver o/p

b_k	0	1	1	1	0	0	1	0	1
$a_k = b_k \oplus a_{k-2}$	1	0	0	1	0	1	1	1	0
a_k in Polar form	+1	−1	−1	+1	−1	+1	+1	+1	−1
$c_k = \overset{+1}{a_k} - \overset{+1}{a_{k2}}$	0	−2	−2	+2	0	0	+2	0	−2
$\lvert c_k \rvert$	0	2	2	2	0	0	2	0	2

b_k	0	1	1	1	0	0	1	0	1
c_k	0	−2	0	+2	0	0	+2	0	−2
$\lvert c_k \rvert$	0	2	0	2	0	0	2	0	2
b_k	0	1	0	1	0	0	1	0	1

x

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Determine the minimum (optimum) transmission bandwidth and transmission bandwidth when the bit duration is 0.647 μs

Solution: $T_b = 0.647\ \mu s$

Minimum bandwidth $B = \frac{R_b}{2}$

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

$$= \frac{1}{2 \times 0.647\ \mu s}$$

$$= 772\ \text{kHz}$$

Transmission BW $B = 2B_0 - f_1$

where $f_1 = (1-\alpha) B_0$

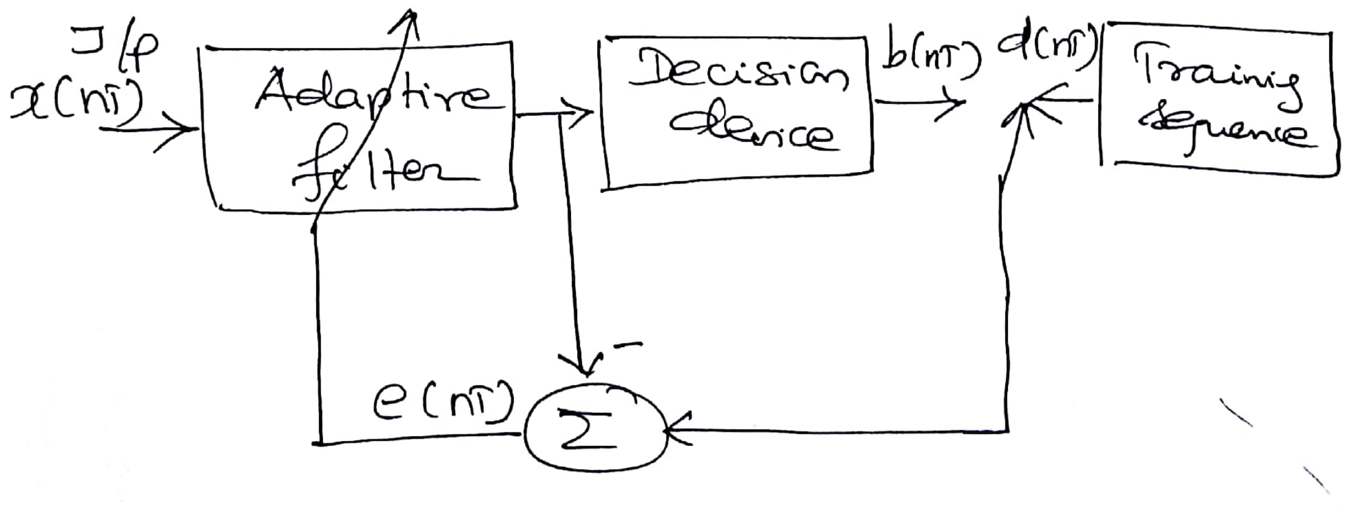
Taking $\alpha = 1$, $f_1 = 0$

$$\Rightarrow B = 2B_0 = 2 \cdot \frac{R_b}{2}$$

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

$$= \frac{1}{0.647\ \mu s} = 1.545\ \text{MHz}$$

Equalisation:



Adaptive Equalization for data transmission:

Two basic signal processing operations to acheive high speed transmission of digital data are

i) Discrete PAM - discrete ~~set of~~ possible amplitude levels to represent amplitudes of succesive pulses.

ii) Linear modulation.

The dispersive nature of the channel leads to ISI and results in distortion. In order to realize the full transmission capacity of the channel, there is a need for adaptive equalization. Equalization refers to the process of correcting channel induced distortion. This process is adaptive when it adjusts itself continiously during data transmission by operating on the input signal

Two types of Equalization:

i) Prechannel equalization:
- At the transmitter end
- Requires feedback of the channel.

ii) Postchannel Equalization:
- done at the Receiver end
- done before the data transmission.
- a training sequence is transmitted through the channel so as to adjust the filter parameters to optimum values.
- Equalizer is positioned after the receiving filter in the receiver.

Adaptive equalizer:

- Consists of tapped delay line filter whose coefficients are updated in accordance with the Least Mean Square (LMS) Algorithm.
- The filter coefficients are adjusted in a step by step fashion synchronously with the incoming sequence

Two modes of operating the adaptive equaliser:

i) Training period:

A known sequence is transmitted and a synchronized version of the ~~the~~ signal is generated in the receiver where it is applied to the adaptive equaliser as the desired response. The training sequence may be linear maximal length or pseudo noise (PN) sequence. The length of the training sequence must be equal to or greater than that of the equaliser.

ii) decision directed mode:

The error signal is given by

$$e(nT) = b(nT) - y(nT)$$

where $y(nT)$ is the equaliser o/p and $b(nT)$ is the final correct estimate of the transmitted symbol $b(nT)$. An adaptive equaliser operating in this mode is able to track relatively slow variations in channel characteristics.