Pulse Analog Modulation:-

The basis of a pulse modulation system is the sampling process, whereby a continious time signal (analog signal) can be conver-ted into a corresponding sequences of samples that may or may not be uniformly spaced in time. The converted signal is called discrete-time-sampling signal. Generally samples are taken to be uniformly spaced in time. This is called uniform sampling which leads to simpler design and simpler algorithm for discrete-time signals. So, a continious curve can be adequately described by sampling points. So, we can transmit only the sample values as they occur instead of sending the signal continiously. This is the importance of sampling that is used in pulse modulation.

The difference b/w pulse modulation and continious modulation is that in continious time modulation, some parameters of a carrier wave is varied continiously in accordance with the modulating signal. On the other hand in, pulse modulation, some parameters of each pulse is varied by a particular sample value of the message.

There are two type of pulse modulation techniques-

i) Pulse analog modulation
ii) Pulse digital modulation (Pulse-code-modulation)

Pulse analog modulation is discrete in time due to sampling. but some

characteristic feature of each pulse (amplitude-duration and or position) is varied in continious manner in accordance with the sample value of the msg.
In PCM, a discrete-time or discrete-amplitude representation is used for signal. This is achieved through sampling, quantizing and coding of an analog signal.

Advantages of pulse modulation over continious wave modulation-

i) Pulse duration is short, → power saving
ii) Off time can be utilized for sending another message on time sharing basis.. Such multiplexing is known as Time-Division-Multiplexing (TDM).

Practical A signal can be reconstructed from the sampled values only if the samples are taken at a specified rate. known as Nyquist rate.

Practical Sampling-- Ideal sampling can not be achieved in practice. There are some differences b/w ideal and practical sampling.

i) practically sampled signals contain finite amplitude and duration rather than pulses
ii) Practical reconstruction filters are never ideal filters.

There are two types of sampling. Natural sampling and flat-top sampling.

Natural Sampling— In natural sampling: the sampled signal consists of a sequence of pulse of varying amplitude whose tops are not flat but follow the msg waveform to be sampled.

Let $g(t) \rightarrow$ analog signal
$s(t) \rightarrow$ sampling signal of amp. A & duration T.
Time period of pulses $\rightarrow T_s$.

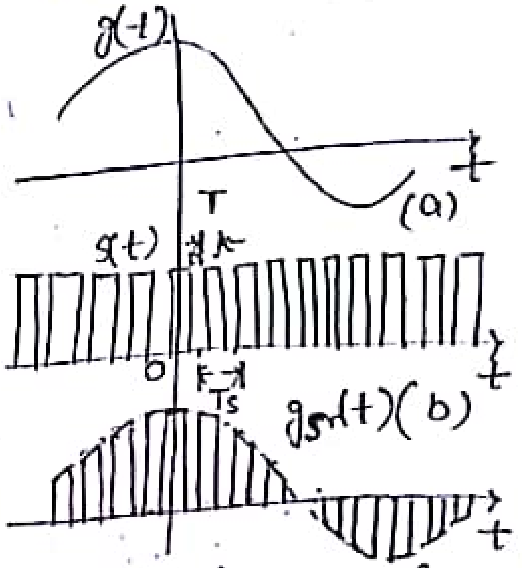

Fig: (a) The signal
(b) The sampling sig.
(c) Sampled v signal v/s time.

sampled signal can be written as $g_{sn}(t) = g(t)\, s(t)$
sampled signal consists of pulses with varying amplitude, whose tops are not flat but follow the analog signal $g(t)$.

With natural sampling, as with instantaneous sampling, a signal sampled at the Nyquist rate may be reconstructed exactly by passing the signal through ideal low-pass filter, with cut-off freqn at W. where W, is highest freqn component present in $g(t)$.

The natural sampled signal $g_{ns}(t)$ given by-
$$g_{sn}(t) = g(t)\cdot s(t) \quad \text{------} \quad (1)$$

$s(t)$ is a periodic signal and may be expressed in the form of natural sampling complex Fourier Series as-
$$s(t) = \frac{AT}{T_s} \sum_{n=-\infty}^{\infty} \operatorname{sinc}\left(\frac{nT}{T_s}\right) \exp\left(\frac{j2\pi n t}{T_s}\right) \quad \text{---} \quad (2)$$

from eqn(1) + (2) we get,

$$g_{sn}(t) = \frac{AT}{T_s} \sum_{n=-\infty}^{\infty} \operatorname{sinc}\left(\frac{nT}{T_s}\right) \exp\left(\frac{j2\pi nt}{T_s}\right) g(t) \quad \text{---(3)}$$

Taking fourier transform both sides we get

$$G_{sn}(f) = \frac{AT}{T_s} \sum_{n=-\infty}^{\infty} \operatorname{sinc}\left(\frac{nT}{T_s}\right) G\left(f - \frac{n}{T_s}\right) \quad \text{---(4)}$$

where $G_{sn}(f) = F[g_{sn}(t)]$ and $G(f) = F[f(t)]$

Flat-top sampling :-

In flat-top sampling pulses of the sampled signals have constt. amplitudes. This constant value of amp. is established by the sample value of the signal at some point within the interval. The signal $g(t)$ is sampled instantaneously by ideal sampling at a rate $\frac{1}{T_s}$, but the duration of each sample is lengthed for a time T.

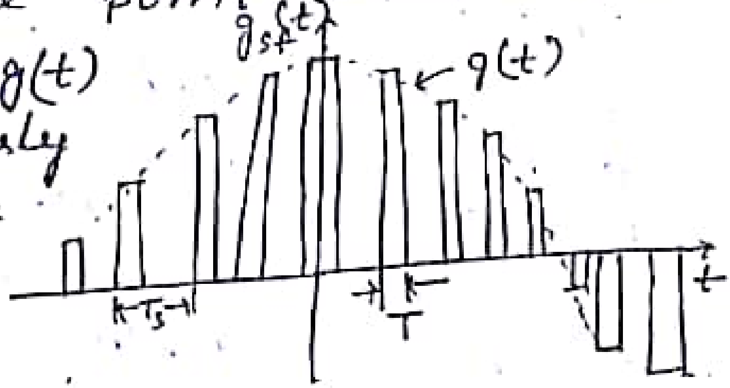

Fig: Flat-top sampling

BW of transmission is inversely proportional to pulse duration, this type of sampling will reduce the bandwidth requirement for transmission.

$$g_{sf} = \sum_{n=-\infty}^{\infty} g(nT_s)\, h(t - nT_s) \quad \text{-----(5)}$$

where $h(t)$ is rectangular pulse of unit amplitude, given by

$$h(t) = 1, \quad 0 < t < T \quad \text{----(6)}$$
$$= 0, \quad \text{otherwise}$$

Flat-top Sampling

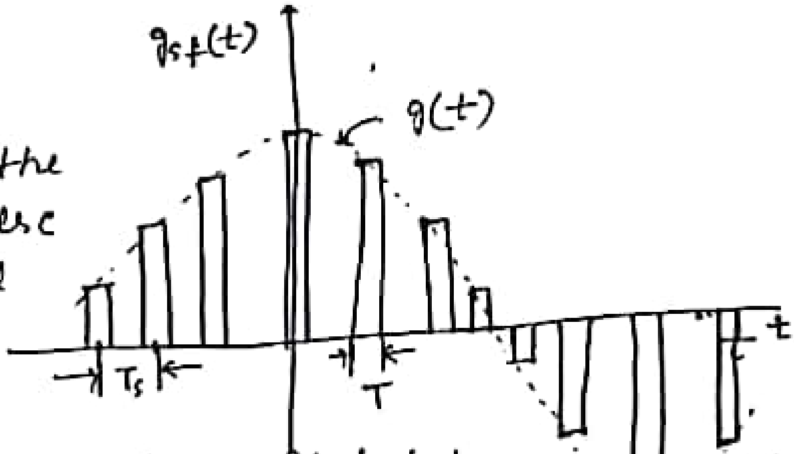

fig 1 : flat-top sampling

In a flat-top sampling, the amplitude of each pulse in the sampled signal is kept constant over the duration T. This sampling can be best understood from ideal sampling. Consider the situation in which the signal g(t) is sampled instantaneously for a time T. (fig.1). Since the bandwidth of transmission is inversely proportional to pulse duration, this type of sampling will reduce the bandwidth requirement for trn.

The flat-top sampled signal can be written as

$$g_{sf}(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\, h(t-nT_s) \quad \cdots\cdots (i)$$

where $h(t)$ is a rectangular pulse of unit amplitude, given by

$$h(t) = 1, \quad 0 \le t \le T \quad \cdots\cdots\cdots (ii)$$
$$= 0 \quad , \text{ otherwise.}$$

Further, the ideal sampled version of $g(t)$ is

$$g_s(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(t-nT_s) \quad \cdots\cdots (iii)$$

convolving $g_s(t)$ with $h(t)$ defined in eqn (ii), we get

$$g_s(t) * h(t) = \int_{-\infty}^{\infty} g_s(\tau)\, h(t-\tau)\, d\tau$$

$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(\tau-nT_s)\, h(t-\tau)\, d\tau$$

$$= \sum_{n=-\infty}^{\infty} g(nT_s) \int_{-\infty}^{\infty} \delta(\tau-nT_s)\, h(t-\tau)\, d\tau \quad \cdots (iv)$$

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Using shifting property of delta-function, we obtain from eqn (iv).

$$g_s(t) * h(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\, h(t-nT_s) \quad \text{------ (v)}$$

comparing eqn (i) with eqn (v), we get that the flat-top sampled signal is equivalent to the convolution of ideal sampled signal $g(t)$ and the pulse signal $h(t)$. Thus

$$g_{sf}(t) = g_s(t) * h(t) \quad \text{......... (vi)}$$

Taking fourier transforms on both sides, we get

$$G_{sf}(f) = G_s(f) H(f) \quad \text{----- (vii)}$$

where $G_{sf}(f)$, $G_s(f)$ and $H(f)$ are the fourier transforms of the flat-top sampled signal ~~g_{sf}~~, $g_{sf}(t)$, ideal sampled signal signal $g_s(t)$ and the pulse signal $h(t)$ respectively.

since fourier transform of an ideal-sampled signal is given by

$$G_s(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} G\left(f - \frac{n}{T_s}\right) \quad \text{---- (viii)}$$

Substituting this eqn in eqn (vii), we get

$$G_{sf}(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} G\left(f - \frac{n}{T_s}\right) H(f) \quad \text{---- (ix)}$$

suppose that $g(t)$ is strictly band-limited signal and the sampling rate $\frac{1}{T_s}$ is greater than Nyquist rate. If the sampled signal $g_{sf}(t)$ is passed

(9)

... reconstruction ... through a low-pass reconstruction filter the spectrum of the resulting filter o/p will be $G(f)H(f)$. This is equivalent to passing original signal through a low-pass filter of transfer function $H(f)$.

from eqn (ii), we have

$$H(f) = T\,\mathrm{sinc}(fT)e^{-j\pi fT} \quad \cdots (X)$$

Hence we find that by using flat-top sampling we have introduced amplitude distortion as well as delay of $\frac{T}{2}$. This distortion caused by lengthening the sample is called aperture effect.

The amplitude and delay distortion can be corrected by an equilizer in cascade with low-pass reconstruction filter. Ideally the amplitude response of the equilizer is given by

$$\frac{1}{|H_{eq}(f)|} = \frac{1}{T\sin(fT)} \quad \cdots (xi)$$

The ideal sampled version of $g(t)$ is

$$g_s(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t-nT_s) \quad \text{---} \quad (7)$$
$$= g(t)\sum_{n=-\infty}^{\infty}\delta(t-nT_s)$$

Convolving $g_s(t)$ by $h(t)$, defined in eqn(6), we get

$$g_s(t) * h(t) = \int_{-\infty}^{\infty} g_s(\tau)\, h(t-\tau)\, d\tau$$
$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(\tau-nT_s)\, h(t-\tau)\, d\tau$$
$$= \sum_{n=-\infty}^{\infty} g(nT_s) \int_{-\infty}^{\infty} \delta(\tau-nT_s)\, h(t-\tau)\, d\tau \quad \text{---} \quad (8)$$

Using the shift property of delta function, we get from eqn (8)

$$g_s(t) * h(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\, h(t-nT_s) \quad \text{---} \quad (9)$$

Thus $g_{sf} := g_s(t) * h(t)$ --- (10)

Taking Fourier transform, we get

$$G_{sf}(f) = G_s(f)\, H(f) \quad \text{----} \quad (11)$$

↓ FT of ideal sampled signal.

$$G_s(f) = G(f) * \frac{1}{T_s}\sum_{n=-\infty}^{\infty}\delta\left(f-\frac{n}{T_s}\right) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} G(f) * \delta\left(f-\frac{n}{T_s}\right)$$

$$G_s(f) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} G\left(f-\frac{n}{T_s}\right) \quad \text{---} \quad (12)$$

$$G_{sf}(f) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} G\left(f-\frac{n}{T_s}\right) H(f) \quad \text{---} \quad (13)$$

$$H(f) = T\,\mathrm{sinc}(fT)\, e(-j\pi f T) \quad \text{---} \quad (14)$$

By using flat-top sampling we have introduced amplitude distortion as well as a delay of $T/2$. This distortion caused by lengthening the samples. called "aperture effect" and can be corrected by an equilizer in cascade with the low-pass reconstruction filter

Quantization of Signal -

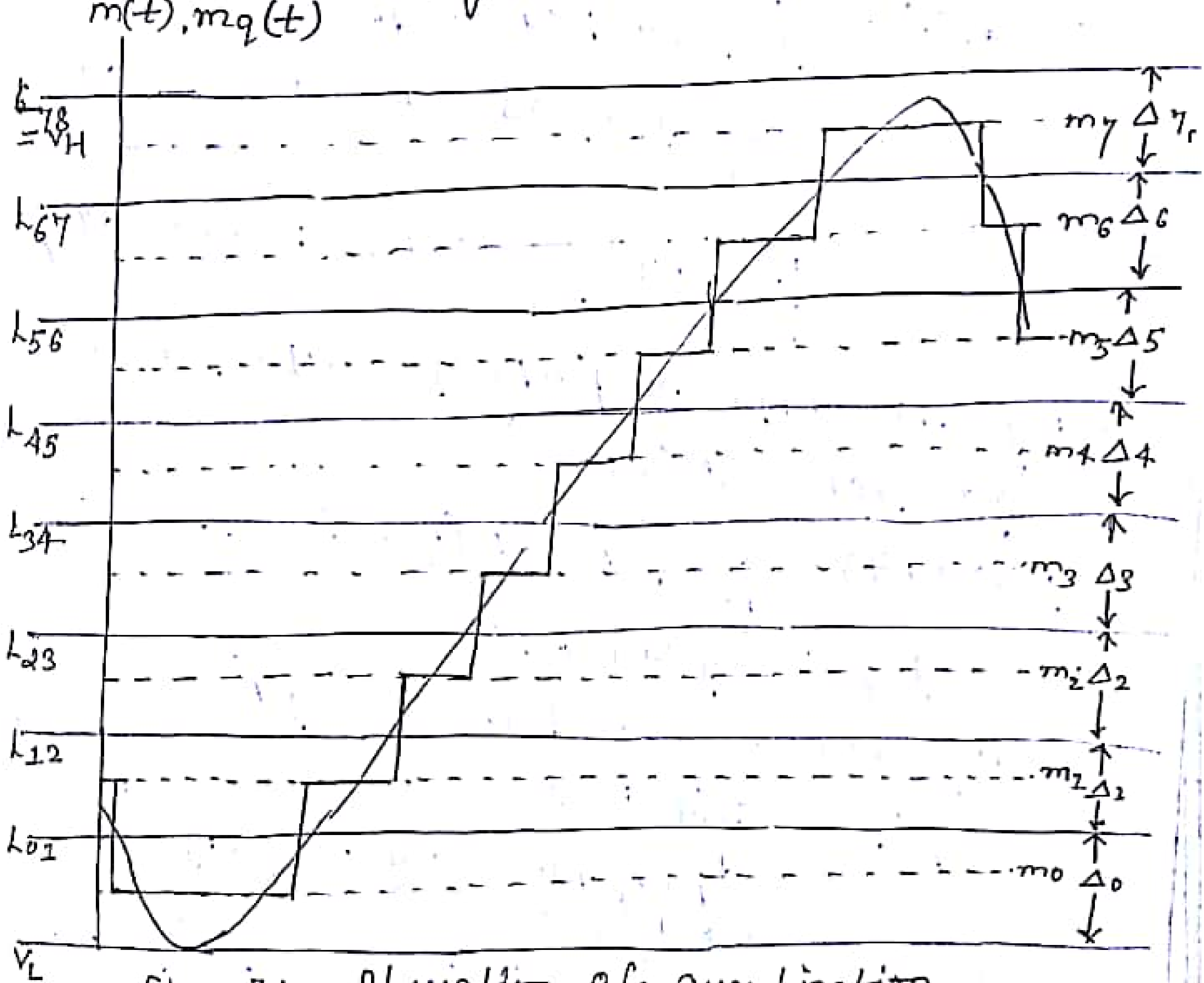


Fig: The operation of quantization -

Let sampled and quantised signal be mq(t), which is an approximation of to $\hat{m}(t)$. mq(t). mq(t) has a great merit that it is, in large measure separable from additive noise.

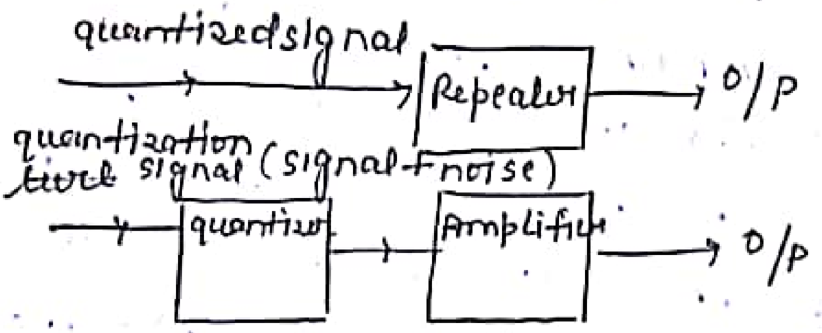
The operation of quantization is shown in above fig.; signal m(t) is confined to to range from V_L to V_H. We have divided this total range into M equal intervals

(6)

having step size $S = \frac{V_H - V_L}{M}$. In this example we have taken $M = 8$. The centre of each of these steps we can locate quantization levels $m_0, m_1, \ldots, m_7$. The quantized signal $m_q(t)$ is generated in following way—

Whenever $m(t)$ is in the range Δ_0, the signal $m_q(t)$ maintains the constt level m_0. Similarly for Δ_1 and so on. Thus the signal $m_q(t)$ will at all times be found at one of the levels, $m_0, \ldots, m_7$. The transition in $m_q(t)$ from $m_q(t) = m_0$ to $m_q(t) = m_1$ is made abruptly, when $m(t)$ passes the transition level L_{01}, which is midway between m_0 & m_1 and so on. We can also that at every instant of time, $m_q(t)$, has the value of quantization level to which $m(t)$ is closest. Thus the signal $m_q(t)$ does not change at all with time or it makes a "quantum" jump to step size S. Each quantization levels between V_H to V_L is separated by S, but the separation of V_H & V_L, from its nearest quantization level is only $S/2$. Also, at every instant of time, the quantization error $(m(t) - m_q(t))$ has a magnitude which is equal to or less than $S/2$.

So, the quantized signal is approximation of original signal & quality improve(7) when number of levels increased. & step size decreases. 256 levels can be used to obtain the quality of commercial color TV, while 64 levels give only fairly good color TV performance.



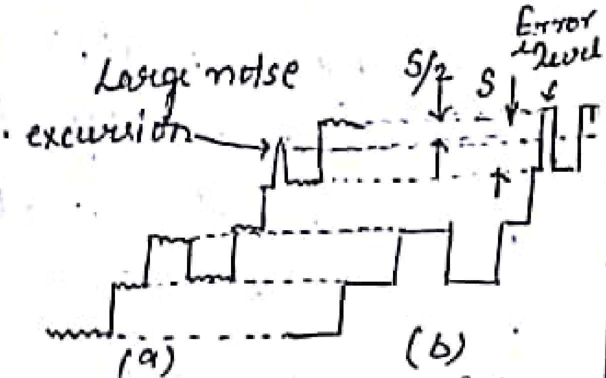


Fig: (a) A quantized signal with added noise
(b) The signal after requantization. One instance is recorded in which the noise level is so large that an error results.

As shown in adjacent figure, the quantizer o/p is the level to which the i/p is closest. Therefore as long as the noise has an instantaneous amp. less than S/2, the noise will not appear at the o/p. One instance in which noise does exceeds S/2 is indicated in the figure, and, correspondingly an error in the level occurs.

So, the method of signal quantization. the effect of additive noise can be significantly reduced.

Reducing the Probability of Error -

i) By decreasing the spacing of the repeaters
ii) By increasing the step size.

iuull amplitude ... of the ...

iii) Increasing step size results in an increased discrepancy b/w true signal and the quantized signal $m_q(t)$. The difference $m(t) - m_q(t)$ can be regarded as a noise and is called 'quantization noise'. The quantized signal is not perfect replica of original signal due to errors caused by additive noise and quantization noise.

Quantization Error:—

Mean square quantization error may be written as $\overline{e^2} = \int_{m_1 - S/2}^{m_1 + S/2} f(m)\,(m-m_1)^2\,dm$

+ ..... upto M levels.

where; $f(m)dm$ be the the ~~prob~~ probability that $m(t)$ lies in the voltage range $m - dm/2$ to $m + dm/2$

Now solving eqn of $\overline{e^2}$ we get,

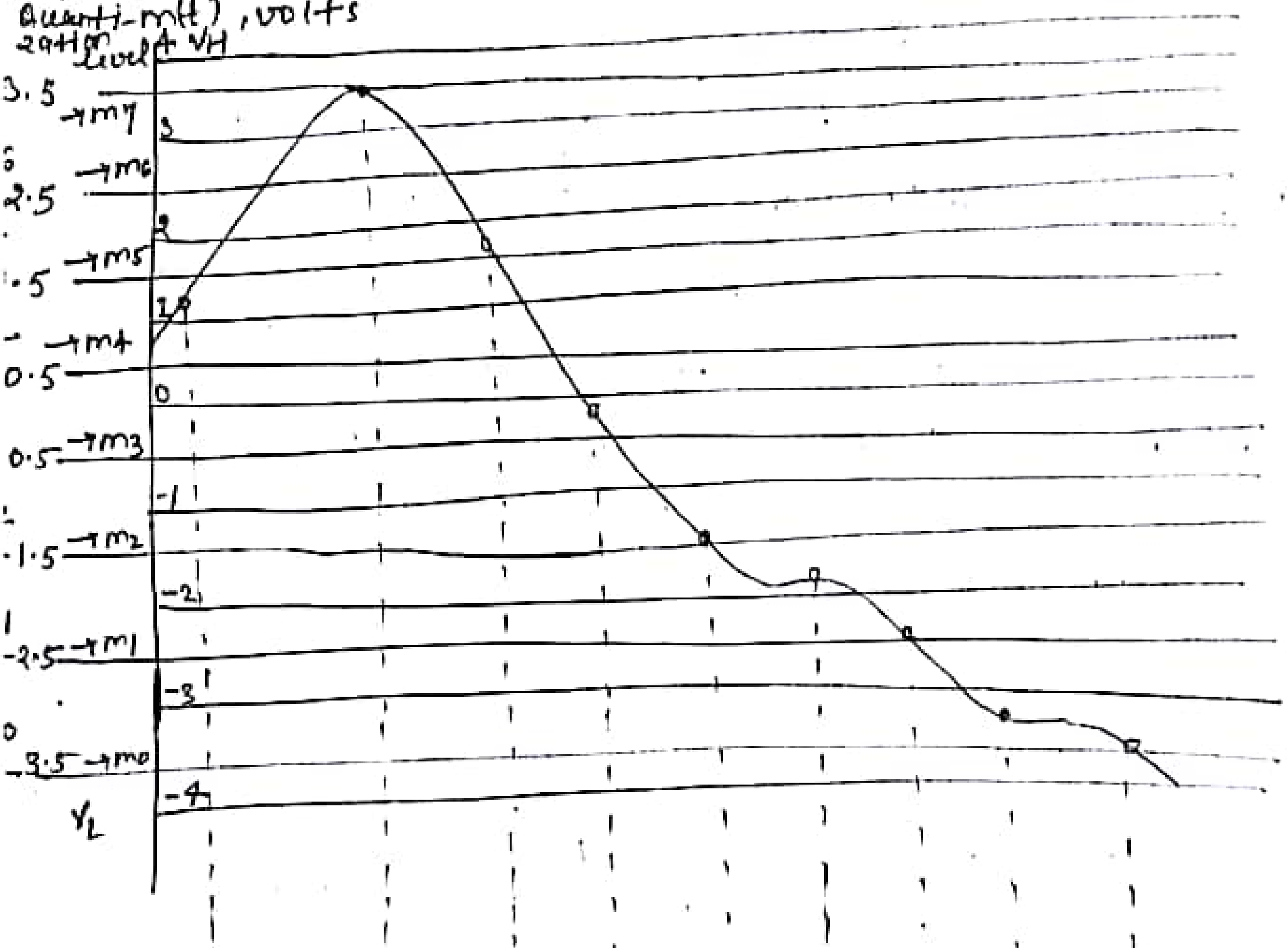
$$\boxed{\overline{e^2} = S^2/12}$$

Companding :—

... number ... ...

ode no.

Pulse-Code Modulation —



Sample value	1.3	3.6	2.3	0.3	-1.4	-1.7	-2.5	-3.2	-3.5
Nearest quantization level	1.5	3.5	2.5	0.5	-1.5	-1.5	-2.5	-3.5	-3.5
Code Number	5	7	6	4	2	2	1	0	0
Binary representation	101	111	110	100	010	010	001	000	000

Fig: A message signal is regularly sampled. Quantization levels are indicated. For each sample and quantized value is given and its binary representation is indicated.

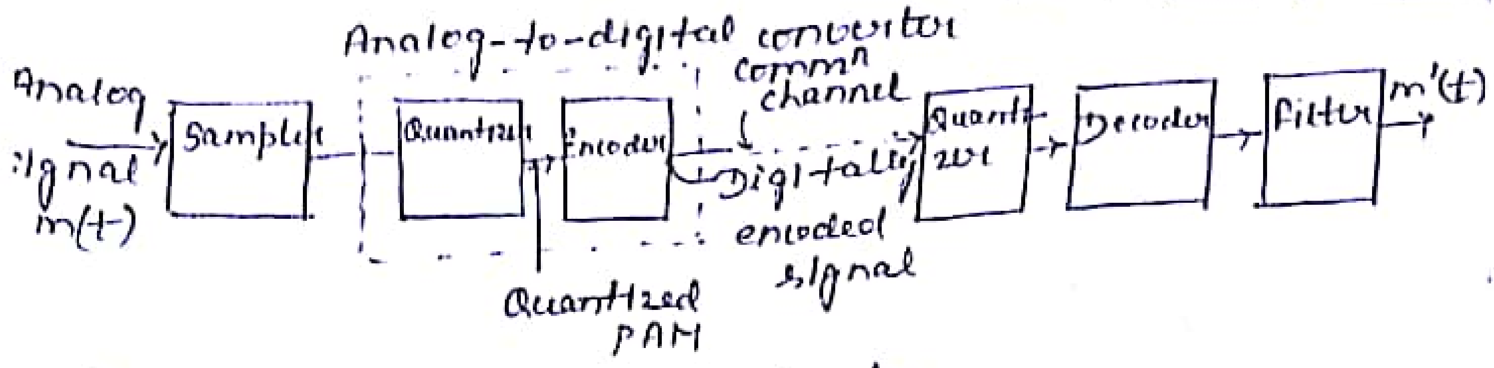


Fig: PCM communication system

Advantages of Pulse code Modulation:-

i) PCM will have low noise addition and data loss is also very low.

ii) we can repeat the exact txed signal at the Rx. This is called repeatability. And we can transmit the signal with any distortion loss also.

iii) PCM is used in music play back CD's and also used in DVD for data storing, whose sampling rate is much higher.

iv) PCM can be used in storing data.

v) Multiplexing of signal can also be done using PCM.

vi) PCM requires larger BW.

vii) PCM permits the use of pulse regeneration

Disadvantages of PCM:-

i) specialized circuitry is required for transmitting and also for quantizing the samples at same ~~...~~ quantized levels. For encoding using PCM, we need to have complex and special circuitry.

ii) PCM receivers are cost effective, compared to other receivers.

iii) Developing PCM is bit complicated and checking the transmission quality is also difficult & takes more time.

iv) Large BW is required for PCM, when compared to BW used by the normal number in ... ...

analog signals to transmit data.

v) channel BW should be more for digital encoding.

vi) PCM systems are complicated when compared to analog modⁿ methods and other systems.

vii) Decoding also needs special equipment's are they are also too complex.

Applications:-

i) In telecommunication systems, air traffic control systems etc.

ii) PCM is used in compressing the data, that is why it is used in storing data in optical disks like DVD, CDs etc. PCM is also used in database management systems.

iii) PCM is used in mobile phones, normal telephones etc.

iv) Remote controlled cars planes, trains etc use pulse code modulation.

Analog Pulse Modⁿ :- Pulse modulation consists sample of modulating signal, and these sample values directly modulate a periodic pulse train with one pulse for each sample.

Three types of analog pulse modⁿ schemes-

i) Pulse amplitude modⁿ (PAM)

ii) Pulse width modⁿ (PWM) or PDM

iii) Pulse position modⁿ (PPM)

there are some analogies b/w analog pulse modⁿ & CW modⁿ. In PAM. 1) the message information (in the form of sample values) is conveyed by the

...plitude of the PAM wave. ... The highest message is conveyed by a time parameter. In PDM and PPM, the is analogous to exponential mod° (angle modⁿ) in which instantaneous freqⁿ or phase (also time parameters) is varied in accordance with the msg signal.

Pulse modⁿ should not be considered as modⁿ in the usual sense. Some points should be considered while designing pulse modⁿ systems —

i) Rich in d.c. components and low-freqⁿ components. Direct transmission is very difficult.

ii) Overlapping of pulses should be avoided during trn.

iii) PM waves needs reconstruction of the signal through the extraction of the sample values and low-pass filtration.

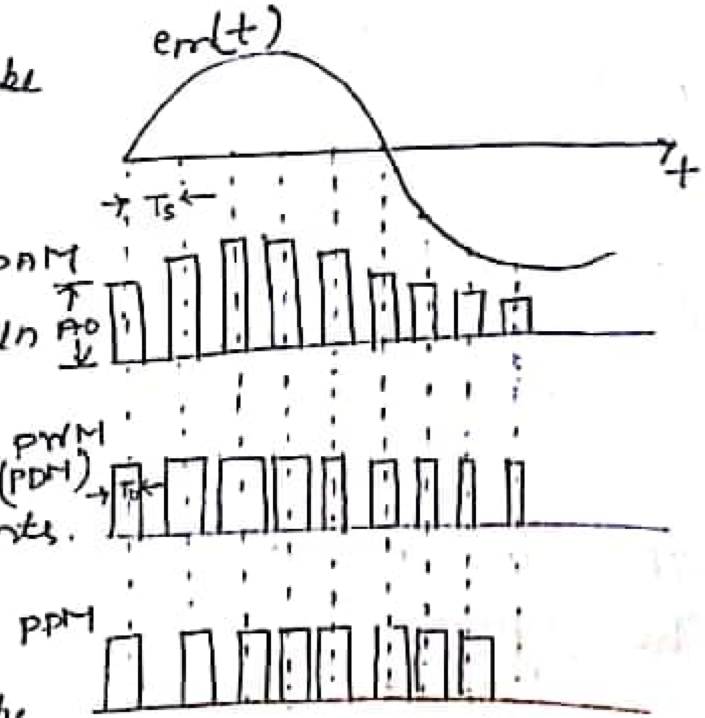


Fig: Types of pulse analog modⁿ.

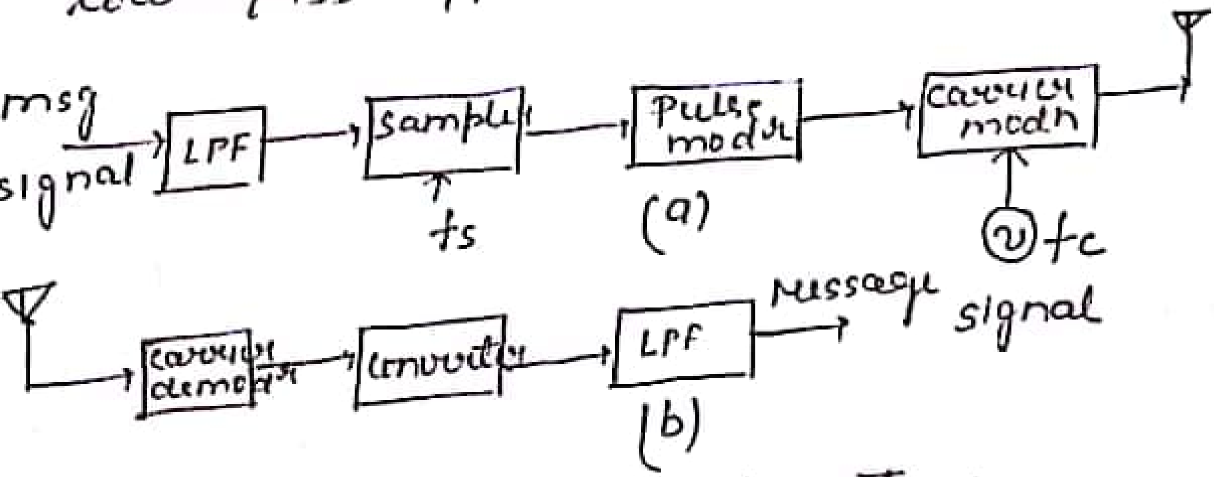


Fig: pulse modⁿ commⁿ systems
a) Tx (b) RX

Pulse - Amplitude Mod^n (PAM)

Mathematically, a PAM signal may be represented as.

$$s(t) = \sum_{n=-\infty}^{\infty} [1 + K_a m(t)]\, g(t - nT_s)$$

$m(nT_s) \rightarrow$ nth sample of msg signal $m(t)$. $T_s \rightarrow$ sampling period, $K_a \rightarrow$ amplitude sensitivity, $g(t) \rightarrow$ pulse train.

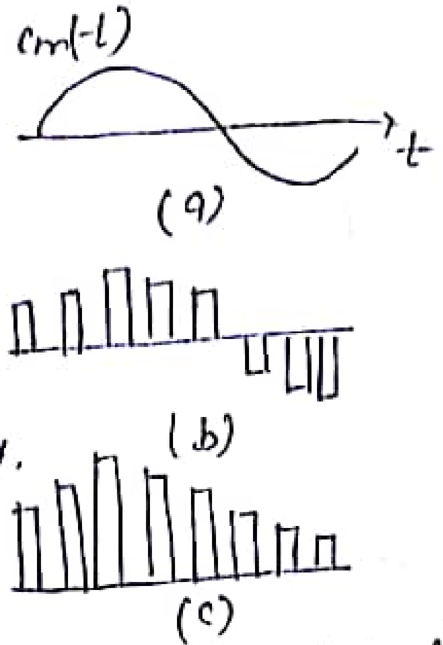


Fig: Pulse-Amp. Mod^n
a) Msg signal
b) Double polarity PAM
c) single polarity PAM.

For a single polarity PAM.

$1 + K_a m(nT_s) > 0$, for all n

The sampling rate $1/T_s$ must be equal to or greater than the highest freq^n component in the msg signal.

Generation :- The signal to be converted to PAM is applied to one input of AND gate. Pulses at the sampling freq^n are applied to the other input of AND gate to open the it during the desired time interval. The o/p of gate then consists of pulses at the sampling rate, equal in amp. to the signal voltage at each time instant. The pulses are then pass through pulse-shaping n/w to convert into flat-top pulses.

Demodulation — The PAM signal can be demodulated by a low-pass filter with a cut-off freq^n just large enough to accomodate the highest freqn component of the msg. signal $e_m(t)$. Cut-off freqn should be such so that it can sampling freqn ripples may be easily removed. The aperture effect (amplitude distortion) may be removed by using equalizer.

Noise performance of PAM is not good, in comparison to base band trn. It finds application in message processing for TDM and in the study of sophisticated pulse modn tech.

Applications-

i) Ethernet commn
ii) Used for photo biology, which is study for photosynthesis.
iii) Used as electronic driver for LED lighting.
iv) Used in many microcontrollers for genera-ting the control signals. etc.

Pulse-Time Modulation (PTM)

In PTM. first signal is sampled but the pulses indicating sample amplitude themselves all have a constant amplitude (Either width or position) the One of the timing characteristics of the pulses is ~~varying~~ varied in accordance with the sampled signal amp. at that instant of time.

Pulse-Width Modn (PWM)

This type of pulse time modn is also called pulse duration (PDM) or pulse length (PLM). In PWM, the amp of msg signal are used to vary the duration of the individual pulses. The pulse width may be varied by varying the time of occurance of leading edge, the trailing of edge or both edges of pulses in accordance with the sampled value of the modulating signal.

<u>Generation</u>:- The msg and pulse train are added and the combination is applied to a slicer, which has the property, that its output is zero, whenever the input is below the slicing level and it constant, when input exceeds the slicing level. The ~~value~~ ~~its~~ duration of each pulse is dictated by the value of the msg wave at the time of occurance of trailing edge.

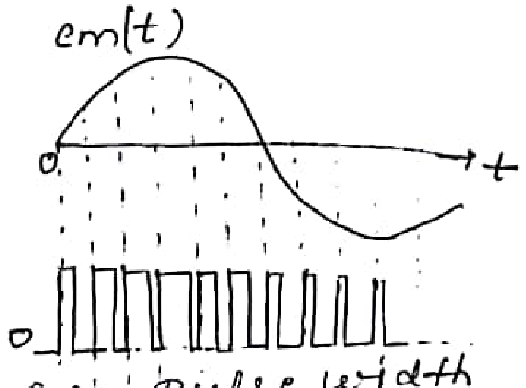


Fig:- Pulse width modn

PWM can also be generated by using emitter coupled monostable multivibrator, which is has an excellent voltage-to-time converter, since its gate width is dependent on the voltage to which the capacitor is charged. A series of rectangular pulse with varying ~~voltage~~ width can be obtained, if the applied voltage can be varied in accordance with a modulating voltage.

<u>Demodulation</u>:- From PWM, signal can be recovered using a low-pass filter. Some distortion may present after reconstruction, due to cross modulation products that fall in the signal band.

Pulse-Position Modn (PPM)

1.— It is a modified version of PWM. In PWM, long pulse expend considerable amount of power during the pulse while bearing no additional information. In PPM unused power is subtracted from the PDM, to get get more efficient pulse modn. In PPM, the position of a pulse relative to its unmodulated time of occurance is varied in accordance with the msg signal.

In PWM, the locations of leading edges of the pulses are kept ~~constt.~~ fixed, whereas those of trailing edges are made to vary according to modulating signal. The position of the trailing edges thus depend on pulse width which is determined by signal amp. at that instant. So, it can be said that the position of the trailing edges of PWM pulses are PPM modulated.

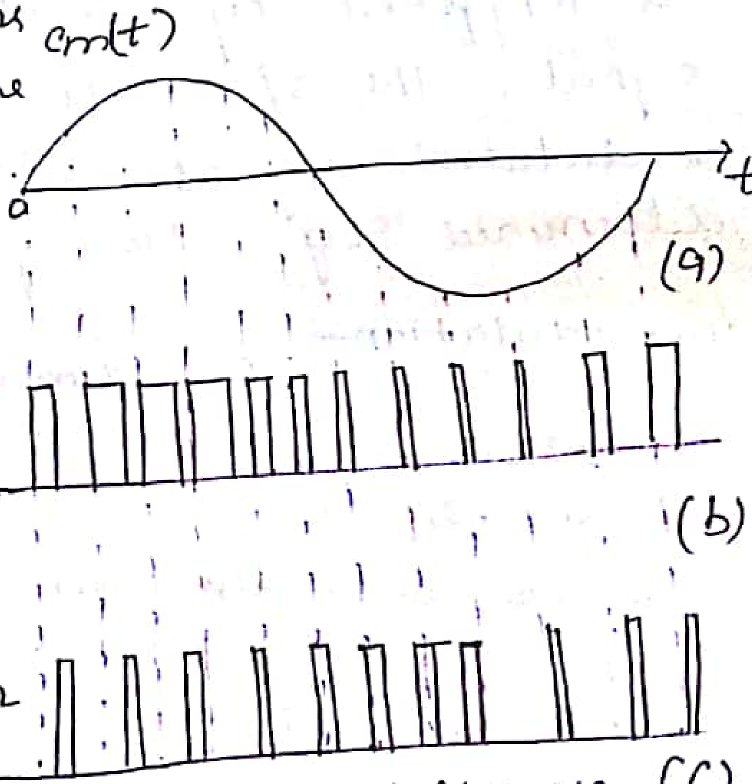


Fig: a) Modulating sig.
b) PWM c) PPM

The method of obtaining PPM from PWM is thus accomplished by getting rid of the leading edges and bodies of the PWM pulses. This pulses have time displacement proportional to the instantaneous value of msg signal.

Generation- The simplest method of generating PPM from PWM wave is to use a monostable multi-vibrator which has one stable-state and one quasi-stable state. The MSV can be trigged from from stable to quasi-stable state by externally applied pulses. The period of quasi-stable state is determined by timing circuit of the device, chosen by the designer. If any ~~MSV~~ MSV is triggered by trailing edges of PWM signal, the o/p will be a pulse position modulated signal, whose timing can be determined by timing ckt of the multivibrator.

Demodulation- For demodulation ~~PWM~~ PPM is first converted into ~~PPM~~ PWM, with the help of a flip-flop or ~~mult~~ bistable multivibrator. ~~One i/p is of PWM is tri~~ One input of MV is trigger pulses from a local generator or which is synchronised by trigger pulses from the txer. These triggers are used to switch OFF one of the stages of the FF. The PPM pulses are fed to the other base of the FF and switch that state ON. The period of time during which this particular stage is OFF depends on the time difference b/w two triggers, so that resulting pulse has a width that depends on the time displacement of each individual PPM pulses. The resulting (18)

PWM is then demodulated.

Pulse modn has major advantage that it may use the increased BW consumed by pulses to obtain an improvement in noise performance.

Thus PTM is much more superior, from this angle to PAM. The generation of PTM signals has become extremely simplified with the availablity of LIC.

Digital Modn :- Many communication system falls into three categories: BW efficient, power efficient, or cost efficient.

BW efficiency → describes the ability of a modulation scheme to accomodate the data withing a limited BW.

Power efficiency → describes the ability of the system to reliably send information at the lowest practical power level.

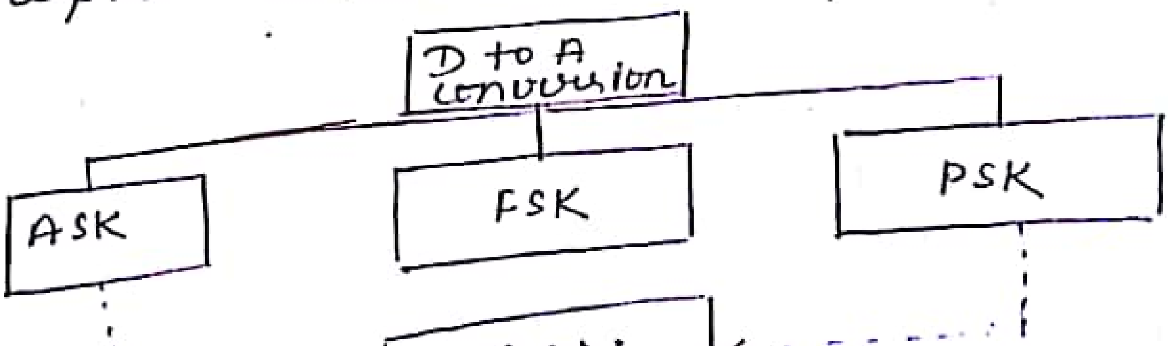
The parameter to be optimized depends on the demands of the particular system.

Why Digital Modulation? To move to Digital Modulation provides more information capacity, compatibility with digital data services, higher data security, better quality communications, and quicker system availability.

Modulation - process of converting digital data or a low-pass analog to band-pass (higher-frequency) analog signal.

Digital to Analog Modn process of changing one of the characteristic of an analog signal (typically a sine wave) based on information in a digital signal.

Sine wave is defined by 3 characteristics (amp, freqn + phase). Digital data (binary 0 + 1) can be represented by varying any one of three. This modn is used to transmit digital data over telephone wire (modem).



Types of Fig: Digital to Analog conversion (Modn)

pulse (3)

... digital communications, the modn process corresponds switching or keying the amp. freqn or phase of a sinusoidal carrier signal according to incoming digital data.

ASK: strength of carrier signal is varied to represent binary 1 or 0.
- both freqn & phase remains constt, while amp. changes.
- commonly, one of the amplitudes is zero.

$$s(t) = \begin{cases} A_0 \cos(2\pi f_c t), & \text{binary } 0 \\ A_1 \cos(2\pi f_c t), & \text{binary } 1. \end{cases}$$

$s(t) = A\cos 2\pi f_c t$

$\frac{A^2}{2} = \frac{E_b}{T_b}$

$\Rightarrow A = \sqrt{\frac{2E_b}{T_b}}$

Modn

"1" $\rightarrow s_1(t) = \sqrt{\frac{2E}{T_b}} \cos 2\pi f_c t$

"0" $\rightarrow s_2(t) = 0 \quad ; 0 \le t \le T_b$

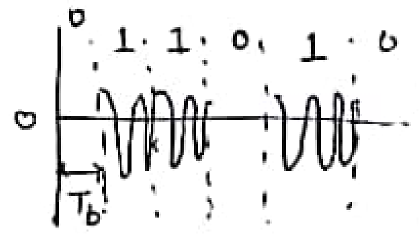
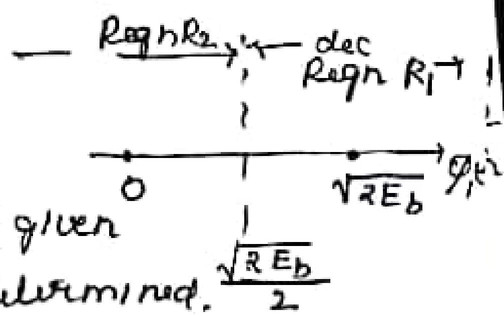


Fig: On-off signalling

Average energy per bit

$E_b = \frac{E + 0}{2}$ i.e. $E = 2E_b$.

Decision Rgn-



demodn only the presence or absence of a sinusoid in a given time interval needs to be determined.

Adv – simplicity

Disadv – very susceptible to noise interference.

appn – used to transmit digital data over optical fiber.

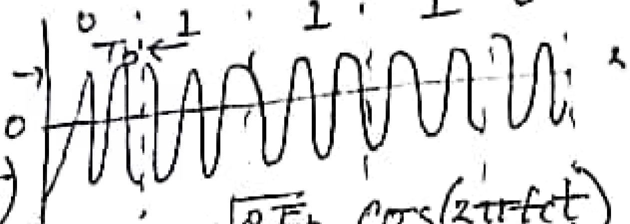
capacitor is assumed to consist

Binary Phase-Shift Keying - (BPSK)

- Mod^n

"1" $\rightarrow s_1(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t)$

"0" $\rightarrow s_2(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t+\pi) = -\sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t)$



$0 \le t < T_b$; T_b bit duration

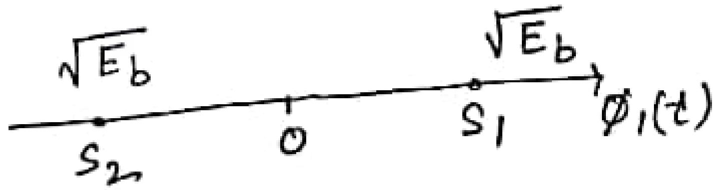
f_c : carrier freq^n chosen to be n_c/T_b for some fixed integer + n_c or, $f_c \gg 1/T_b$.

E_b: Transmitted signal energy per bit;

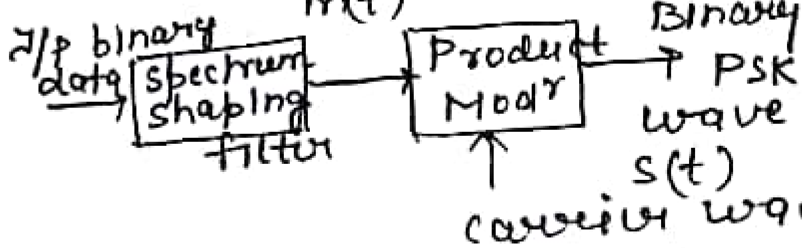
$$\int_0^{T_b} s_1^2\,dt = \int_0^{T_b} s_2^2\,dt = E_b$$

- The pair of signals differ only in a 180°- degree phaseshift.
- There is one basis fn -
  $\phi_1(t) = \sqrt{\frac{2}{T_b}}\cos(2\pi f_c t)$ as with $0 \le t < T_b$.
  Then $s_1(t) = \sqrt{E_b}\,\phi_1(t)$ and $s_2(t) = -\sqrt{E_b}\,\phi_1(t)$
- A binary PSK system is characterised by a signal space, that is one dim^n (i.e. $N=1$) and has two msg points. (i.e. $M=2$)

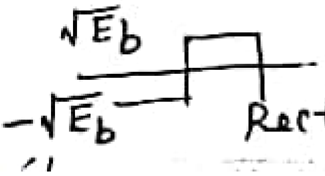
$d_{12} = 2\sqrt{E_b}$



BPSK - Tx

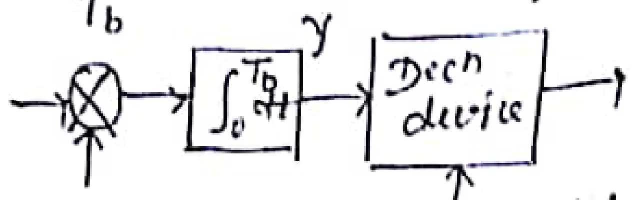


$\phi_1(t) = \sqrt{\frac{2}{T_b}}\cos(2\pi f_c t)$



BPSK-Rx

$r(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t + \theta) + n(t)$



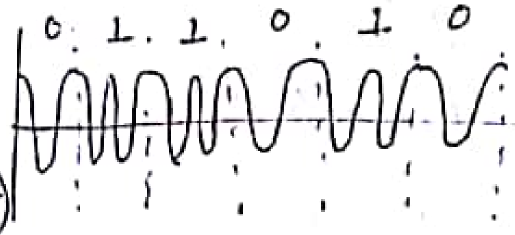
{ say 1, if threshold is exceeded
say 0, otherwise

$\underbrace{\text{demod}^r}$ $\underbrace{\text{detector}}$

- θ is carrier-phase offset, due to propagation delay or oscillators at tx and Rx, are not synchronous.
- The detection is coherent in the sense of:
  - Phase synchronization
  - Timing synchronization.

Binary FSK

• Mod^n



"1" $\rightarrow$ $s_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_1 t)$

"0" $\rightarrow$ $s_2(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_2 t)$; $0 \leq t < T_b$.

$\therefore$ E_b transmitted signal energy per bit.

f_i : txed freq^n with separation $\Delta f = f_1 - f_0$

Δf is selected so that $s_1(t)$ and $s_2(t)$ are ortogonal, i.e. $\int_0^{T_b} s_1(t) s_2(t)\, dt = 0$

Signal space for BFSK

Two ortogonal basis fns are required.

$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_1 t)$; $0 \leq t < T_b$; $s_1(t) = \sqrt{E_b}\, \phi_1(t)$

$\phi_2(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_2 t)$; $0 \leq t < T$; $s_2(t) = \sqrt{E_b}\, \phi_2(t)$

signal space representation