Govt. Engineering College, Jhalawar

Model Question paper

Branch:- IVth sem (CS & IT) subject:- Principle of communication systems

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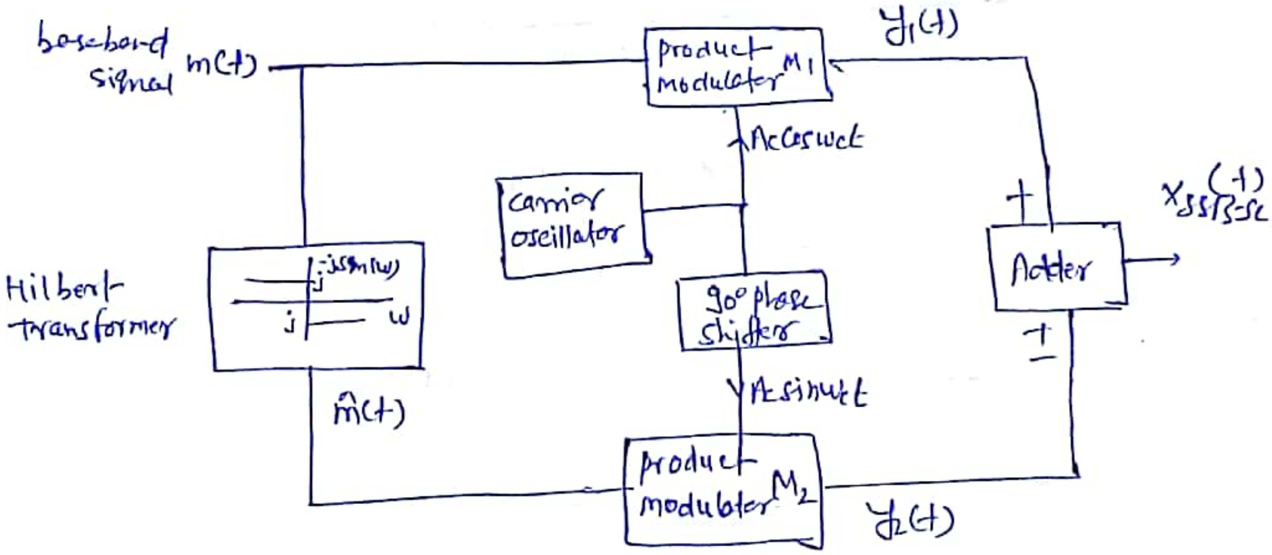
Q.1 Explain the phase Discrimination method for SSB-SC generation & Draw the frequency spectrum.

Ans:-



→ This block diagram shows the method of SSB-SC generation. This system uses Two balanced modulator M_1 & M_2 & Two 90° phase shifter Network.

Working:- The baseband signal m(t) applied directly ~~to~~ to the product modulator M_1 & through Hilbert transformer to the product modulator M_2. Hence we get the Hilbert transform $\hat{m}(t)$ at the o/p of Hilbert transformer. The output of carrier oscillator is applied ~~to~~ as it is to modulator M_1 where as it passed through a -90° phase shifter and applied to modulator M_2.

$$\text{output of } M_1 = m(t) \cdot A_c \cos w_c t = y_1(t)$$

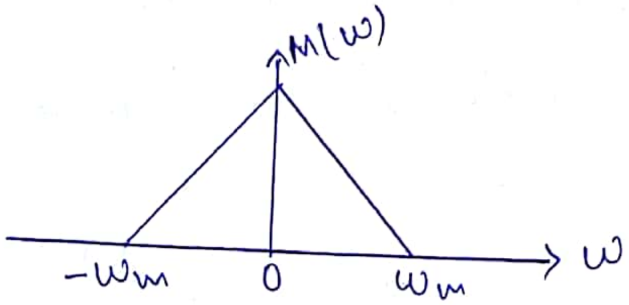
$$\text{output of } M_2 = \hat{m}(t) \cdot A_c \sin w_c t = y_2(t)$$

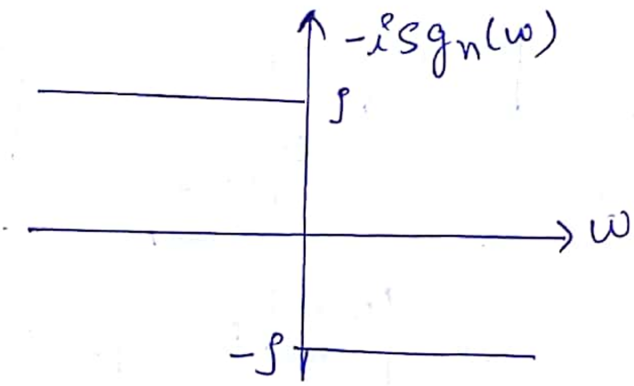
The output of M_1 & M_2 are applied to an adder. so the o/p of adder is $X_{SSB-SC}(t) = m(t) \cos w_c t \pm \hat{m}(t) \sin w_c t$

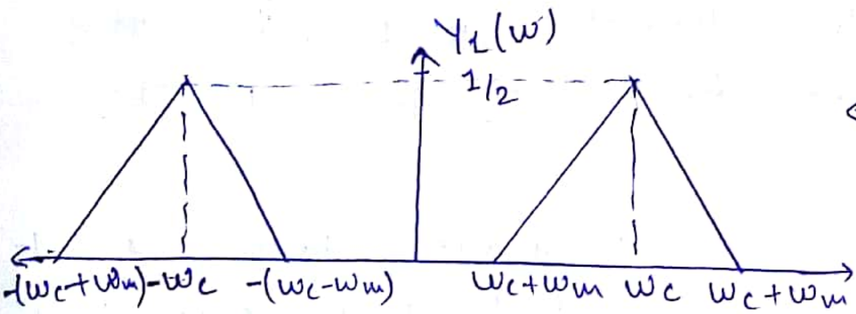
Frequency spectrum of SSB-SC signal:—

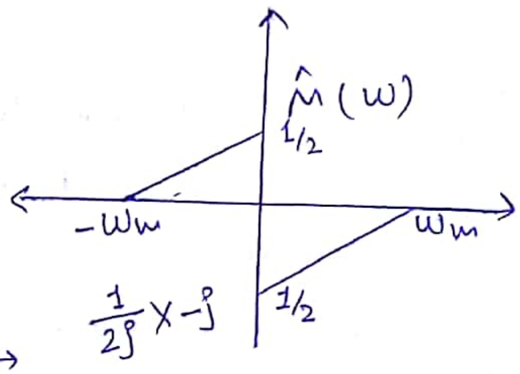
$$y_1(t) = m(t)\cos\omega_c t \xleftrightarrow{F.T.} \frac{1}{2}[M(\omega+\omega_c)+M(\omega-\omega_c)]$$

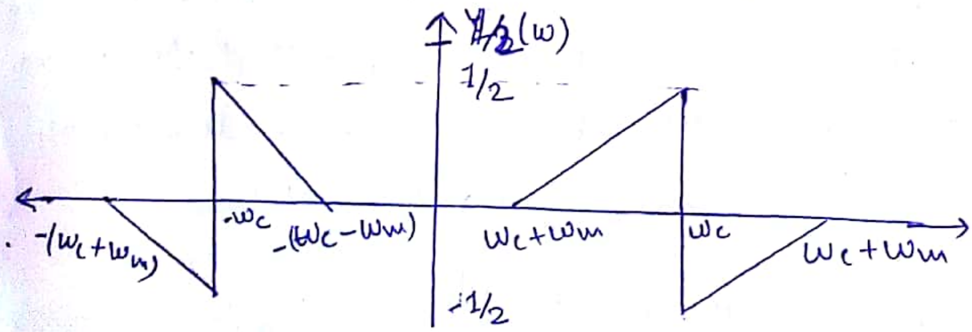
$$y_2(t) = \hat{m}(t)\sin\omega_c t \xleftrightarrow{F.T.} \frac{1}{2j}[\hat{M}(\omega-\omega_c)-\hat{M}(\omega+\omega_c)]$$

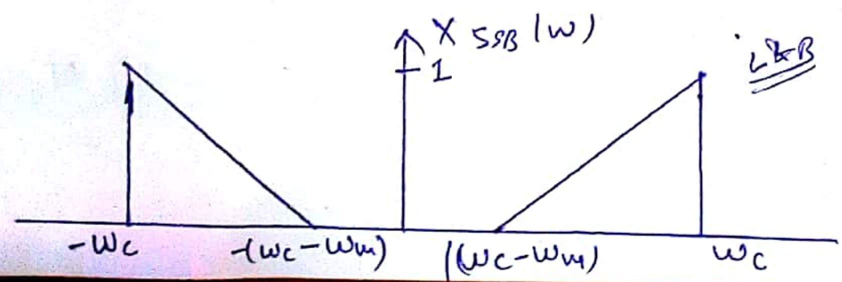












Q.2:- An angle modulated signal is given by $x_c(t) = 6\cos[2\pi 10^7 t + 0.2\sin 10^4 \pi t]$

(i) If $x_c(t)$ is phase modulated signal with $k_p = 5$ r/V

(ii) if $x_c(t)$ is frequency modulated signal with $k_f = 5\times10^2$ Hz/V

then determine the baseband signal m(t).

$$x(t) = A\cos[\omega_c t + \beta \sin \omega_m t]$$

$$\beta = \begin{cases} k_p V_m & \text{for PM} \\ \frac{k_f V_m}{\omega_m} & \text{for Am.} \end{cases}$$

$$\omega_m = 10^4 \pi = 2\pi f_m$$

$$k_p V_m = .2$$

given $k_p = 5$

$$V_m = \frac{.2}{5} = \frac{2}{50} = \frac{1}{25} = .04$$

so $m(t) = V_m \sin \omega_m t$

$$= .04 \sin(10^4)\pi t \qquad \text{Ans. (i)}$$

$$\because \beta = \frac{k_f V_m}{\omega_m}$$

$$= \frac{5\times10^2 \times V_m}{10^4 \pi} = .2$$

$$V_m = \frac{.2\times10^4 \pi}{5\times10^2} = \frac{20}{5}\pi = 4\pi$$

$m(t) = V_m \cos \omega_m t$ for Fm

$$m(t) = 4\pi \cos(10^4)\pi t$$

Ans (ii)

Q.3 Explain the sampling theorem, Compare the Ideal, Natural and Flat Top sampling.

Ans:- <u>Sampling Theorem</u> :— Statement of sampling theorem can be given in two parts as:—

(i) A band Limited signal of finite energy, which has no frequency component higher than f_m Hz, is completely described by it's sample-Values at uniform intervals less than or equal to $\frac{1}{2f_m}$ second apart.

(ii) A band Limited signal of finite energy, which has no frequency component higher than f_m Hz, may be completely recovered from knowledge of it's samples taken at the rate of $2f_m$ samples per second

<u>Proof of sampling Theorem</u>:—

Let a continuous time signal $x(t)$ whose spectrum is band Limited to f_m Hz. this means that signal has no freq. component beyond f_m Hz. So the $X(\omega)$ is zero for $|\omega| > \omega_m$

$$X(\omega) = 0 \quad \text{for } |\omega| > \omega_m$$

$$\omega_m = 2\pi f_m$$

Figure ① show continuous time signal $x(t)$. Let $X(\omega)$ its fourier transform in figure ②. sampling of $x(t)$ at a rate of f_s Hz may be achieved by multiplying $x(t)$ by an impulse train $\delta_{T_s}(t)$. this impulse train consists of unit impulses ~~seap~~ repeating periodically every T_s, second, where $T_s = 1/f_s$. figure ③ show the impulse train. The multiplication results in the sampled signal $g(t)$. in figure ④.

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

$\delta_{T_s}(t)$ is a periodic signal with period T_s, it can expressed as fourier series. So

$$\delta_{T_s}(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t + 2\cos 3\omega_s t + \cdots\right]$$

$$\omega_s = \frac{2\pi}{T_s} = 2\pi f_s$$

so

$$g(t) = \frac{1}{T_s}\left[x(t) + 2x(t)\cos\omega_s t + 2x(t)\cos 2\omega_s t + 2x(t)\cos 3\omega_s t + \cdots\right]$$

To obtain $G(\omega)$, the fourier transform of $g(t)$, we will have to take fourier transform of right hand side.

fourier transform of $x(t)$ is $X(\omega)$.

fourier transform of $2x(t)\cos\omega_s t$ is $[X(\omega-\omega_s) + X(\omega+\omega_s)]$

fourier transform of $2x(t)\cos 2\omega_s t$ is $[X(\omega-2\omega_s) + X(\omega+2\omega_s)]$ and so on.

So the

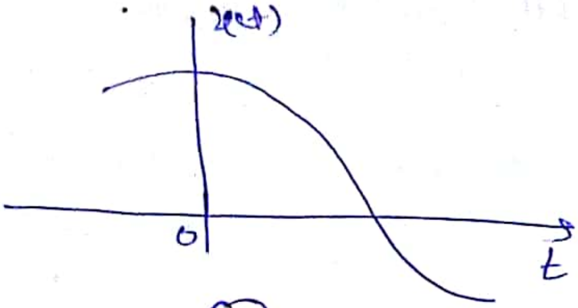
$$G(\omega) = \frac{1}{T_s}\Big[X(\omega) + X(\omega-\omega_s) + X(\omega+\omega_s) + X(\omega-2\omega_s) + X(\omega+2\omega_s)$$
$$+ X(\omega-3\omega_s) + X(\omega+3\omega_s) + \cdots\Big]$$

$$G(\omega) = \frac{1}{T_s}\sum_{n=\infty}^{\infty} X(\omega - n\omega_s)$$

So it is clear that the spectrum $G(\omega)$ consists of $X(\omega)$ repeating periodically with period $\omega_s = \frac{2\pi}{T_s}$ rad/sec or $f_s = 1/T_s$ Hz.

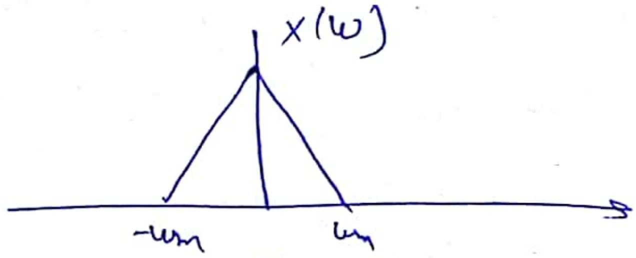
Now if we have to reconstruct $x(t)$ from $g(t)$, we must be able to recover $X(\omega)$ from $G(\omega)$. This is possible if there is no overlap between successive cycles of $G(\omega)$. so this is require that $f_s \geq 2f_m$. But the sampling interval $T_s = 1/f_s$

so $T_s < 1/2f_m$



1

Continous time signal x(t)



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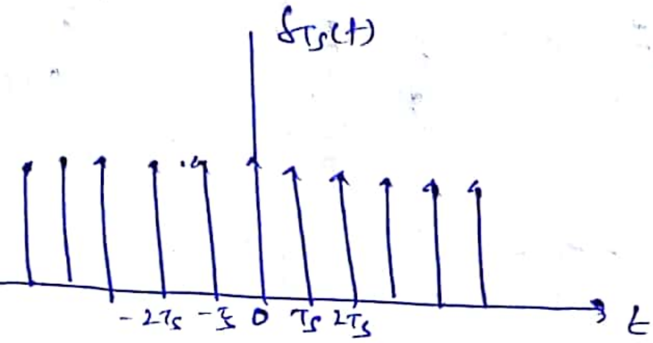
Spectrum of x(t)



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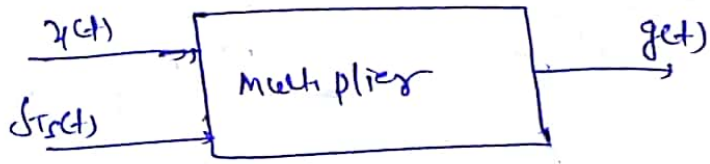
Impulse train



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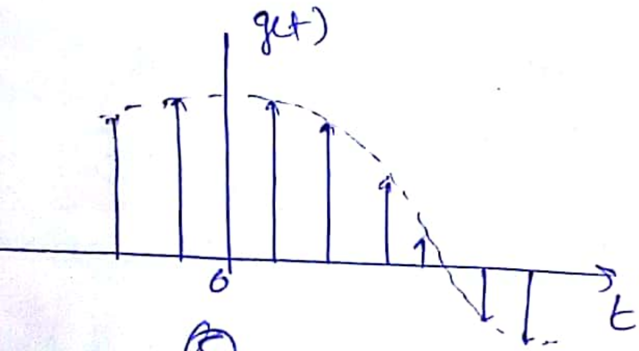

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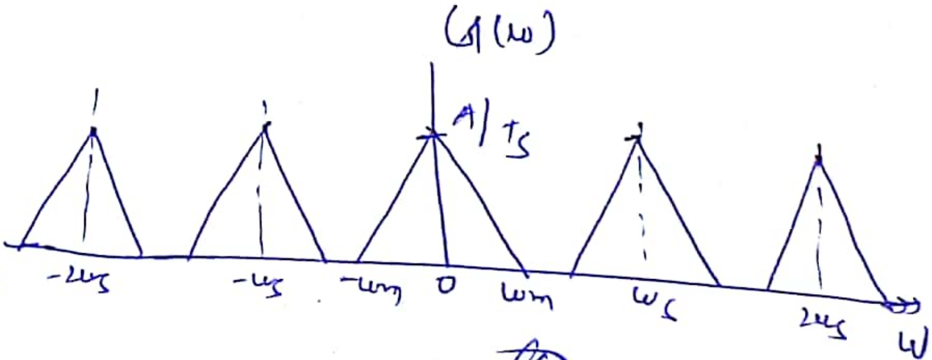
Sampled signal



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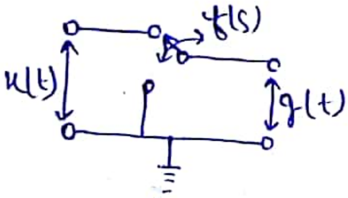
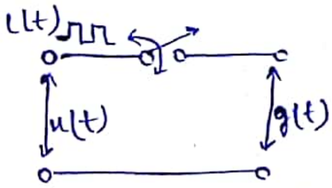
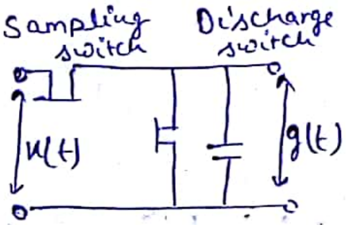
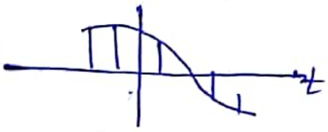
Spectrum of sampled signal

Comparison Between Ideal, Natural ~~& flat~~ Flat Top Sampling:-

Sr. No.	Parameter of Comparison	Ideal sampling	Natural sampling	Flat Top sampling
1.	Sampling principle	It uses multiplication	It uses Chopping Principal.	It uses sample and hold circuit
2.	Generation circuit			
3.	Waveform involved			
4.	Sampling rate	Ideal → Infinite	Satisfied nyquist criteria	Satisfied nyquist criteria
5.	noise interference	max.	min.	max.
6.	Time domain representation	$g(t)=\sum_{n=-\infty}^{\infty} x(nT_s)\delta(t-nT_s)$	$g(t)=\frac{\tau A}{T_s}\sum_{n=-\infty}^{\infty} x(t)\,\text{sinc}(nf_s\tau)\,e^{j2\pi nf_st}$	$g(t)=\sum_{n=-\infty}^{\infty} x(nT_s)h(t-nT_s)$
7.	Frequency domain representation	$G(f)=f_s\sum_{n=-\infty}^{\infty} X(f-nf_s)$	$G(f)=\frac{\tau A}{T_s}\sum_{n=-\infty}^{\infty}\text{sinc}(nf_s\tau)X(f-nf_s)$	$G(f)=f_s\sum_{n=-\infty}^{\infty} X(f-nf_s)H(f)$

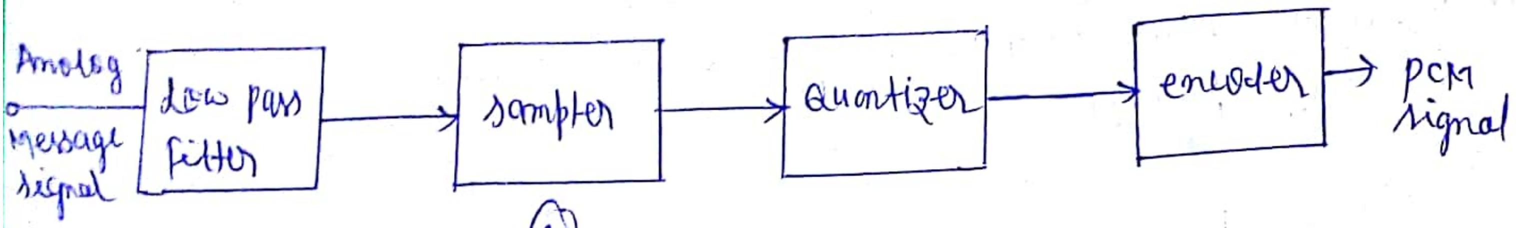
Q.4 Draw the block Diagram of PCM System and Discuss each block in Detail.

Definition

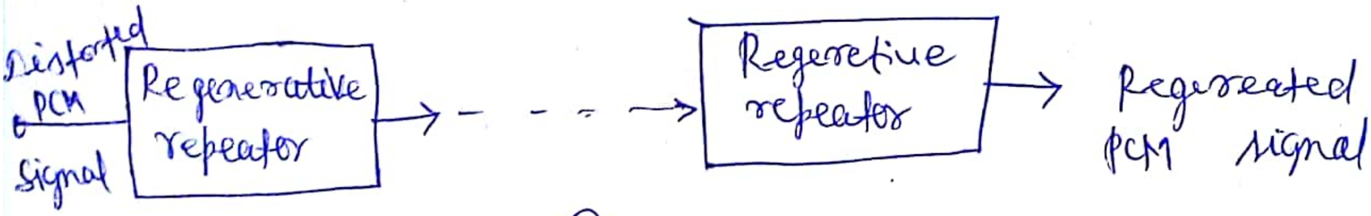
Pulse code modulation is known as digital pulse modulation technique. infact the pulse modulation code (PCM) is quite complex compared to the along pulse modulation technique. (i.e PAM, PWM and PPM) in the sense that the message signal is subjected to a great number of operations.

Element of A PCM System :-

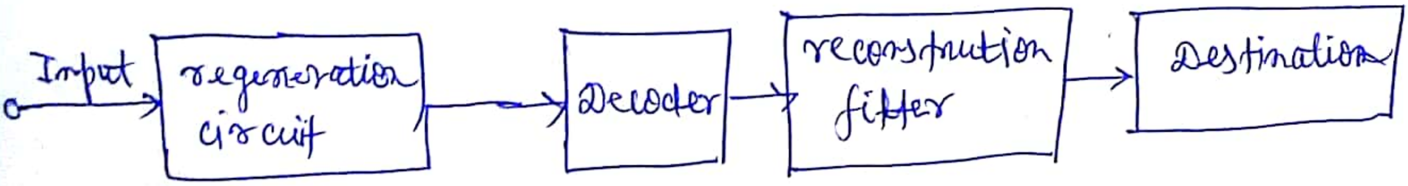
The basic element of PCM system consist of three main part i.e. transmitter, transmission path and reciever. the essential operations in the transmitter of a PCM system are sampling, quantizing and encoding as show in fig. as essential discussed earlier, sampling is the operation in which an analog (contineous time) signal is sampled according to the sampling theorem resulting in a discrete time signal. The quantizing and encoding operations are usually performed in the same circuit. which is know as an analog-to-digital converter (ADC). Also the essential operations in the reciever are regeneration of impaired signals, decoding and demodulation of the quantized samples. These operation are usually performed in the same circuit which is known as a digital-to-analog converter (DAC).



(a)



(b)



(c)

The basis element of a PCM system (a) Transmitter (b) Transmission path (c) Reciever.

reciever regenerative repeator are reconstruct the transmitted sequence of coded pulses in order to combate the accumulated effect of signal distortion and noice. as dissused in the quantization refer to the use of finit set of amplitude levels and for it. in fact this operation combined with sampling permit the use coded pulse for representing the message signal. thus it is combine use of quantize-ing and coding that distingueshes pulse code modulation from analog modulation techniques.

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Q. 5 Explain the Generation & Detection of QPSK signal with suitable diagrams.

Ans:- In PSK system, the phase of carrier signal is varied by 180°, thus giving rise to two signals with phase angles of 0° and 180°. In quadrature PSK (QPSK) system, the phase of the system is allowed to vary by 90°, thus giving rise to four signals with phase angles of 0°, 90°, 180° and 270°; i.e the signals which are in phase quadrature. The major advantange of QPSK (also known as 'Four-phase PSK) is that the bandwidth requirement is reduced to half of the PSK bandwidth requirement.

The QPSK Transmitter

Figure ~~___~~ shows a QPSK transmitter. A clock of frequency f_b and period T_b drives a toggle flip-flop which divides the frequency by 2. The true and false (complemented) outputs drive two seprate D flip-floop. The period of both these flip-floop known respectively as even clock and odd clock, is $2T_b$ and the wave-forms are complements of each other. All the three flip floops are assumed to be negative edge triggered flip-flop's.

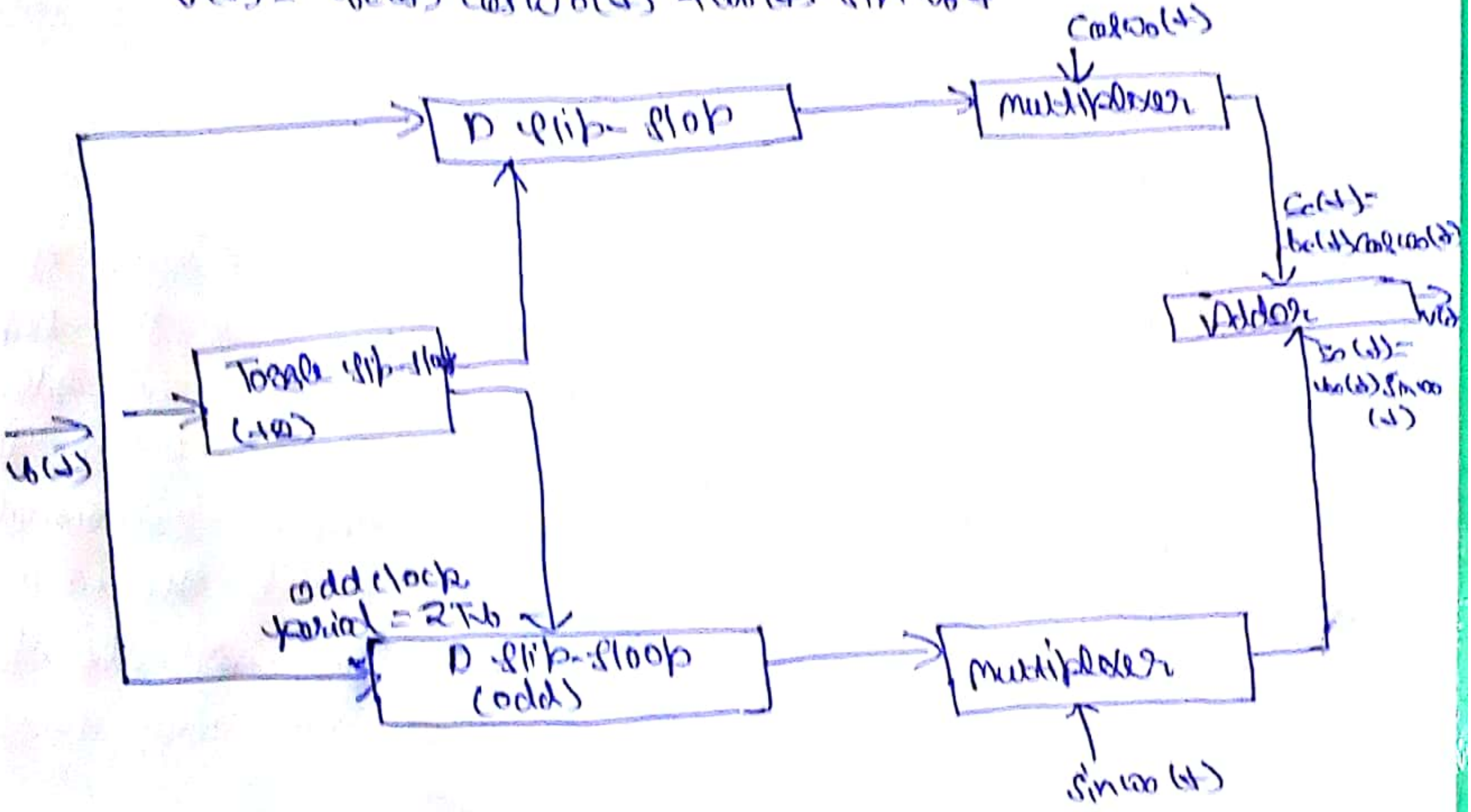
The input data stream $b(t)$ of bit duration T_b is applied as input to both the D flip floops. Since the clock period of even clock is $2T_b$ and the D flip-floop (even) is negative edge triggered, only the even numbered bits of the data stream $b(t)$ are held for next $2T_b$ duration. By same reasoning, only the odd numbered bits of the data stream $b(t)$ are held for the next $2T_b$ duration by the D flip flop (odd). The outputs of even and odd D-flip-flops are respectively $b_e(t)$ and $b_o(t)$

These are multiplied respectively by $\cos \omega_0 t$ and $\sin \omega_0 t$

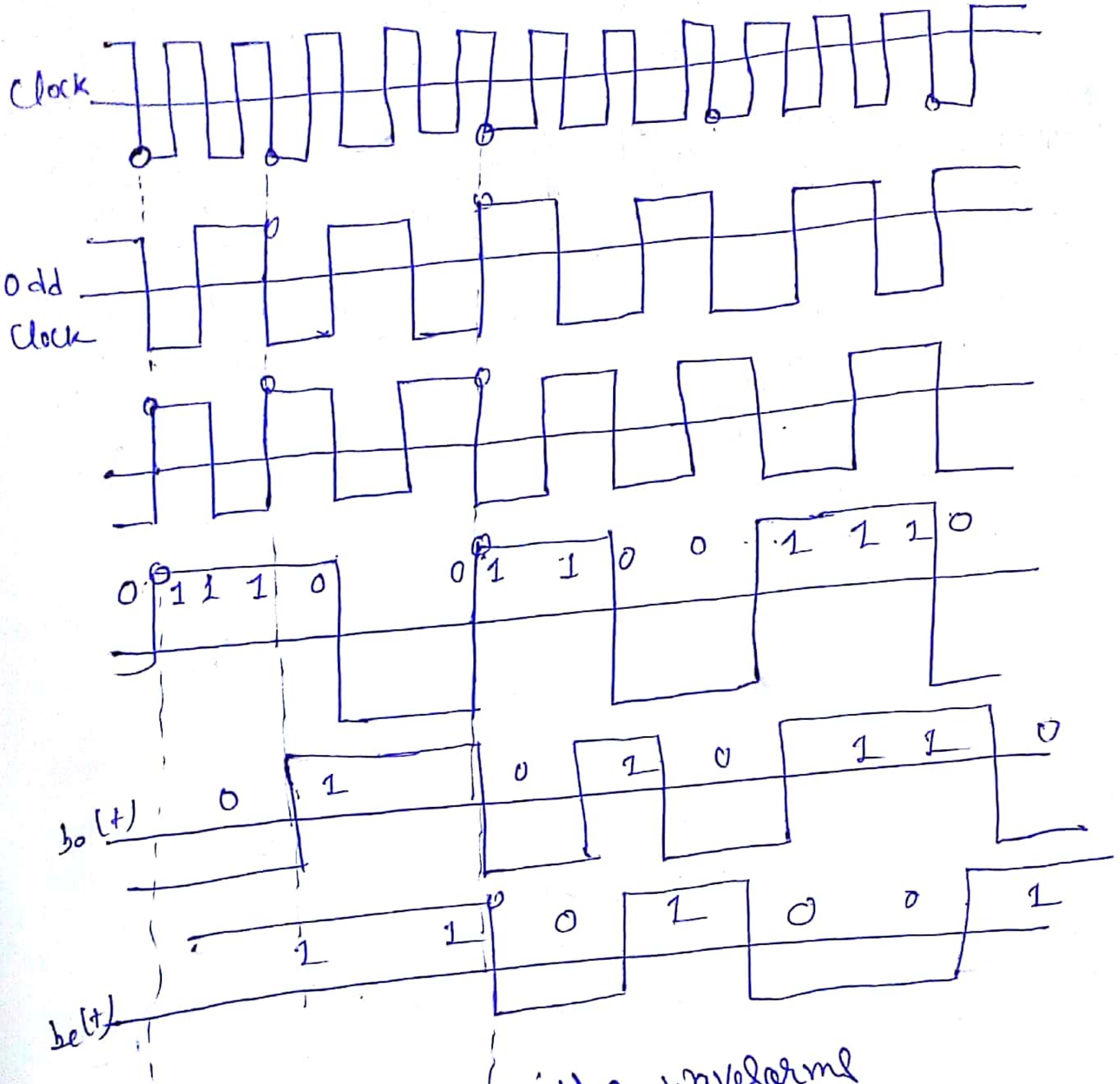
$$S_e(t) = b_e(t) \cos \omega_0 t$$
$$S_o(t) = b_o(t) \sin \omega_0 t$$

$S_e(t)$ and $S_o(t)$ are added whose output $V(t)$ is transmitted. Thus

$$V(t) = b_e(t) \cos \omega_0(t) + b_o(t) \sin \omega_0 t$$



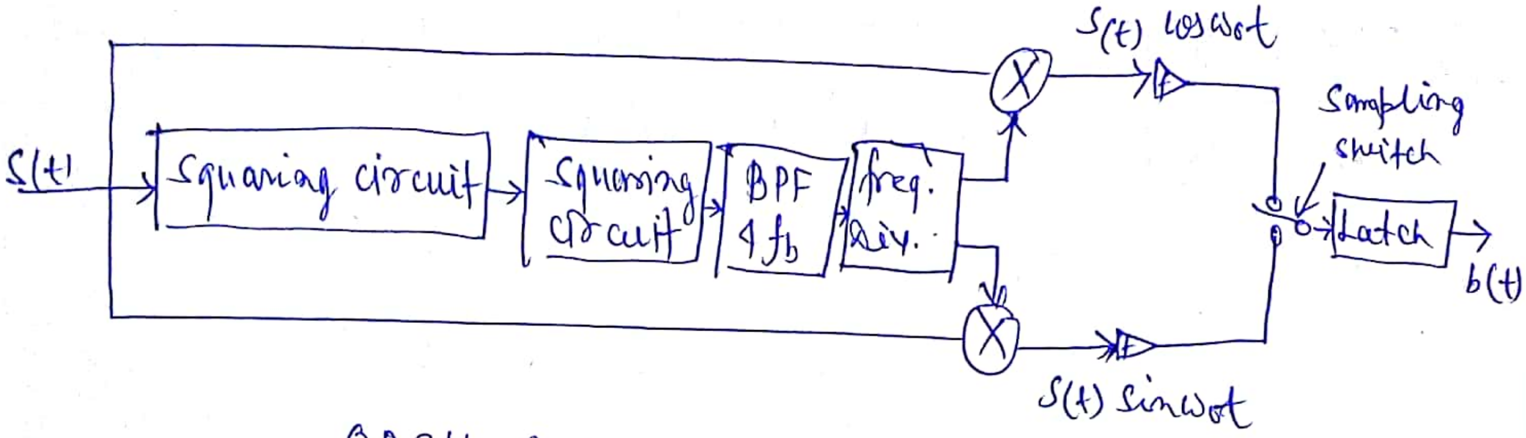
QPSK Transmitter



QPSK Transmitter waveforms

The QPSK Receiver :-

The QPSK receiver is shown in fig. As synchronous detection is required, it is necessary to locally generated two carriers cos $\omega_0 t$ and sin $\omega_0 t$. The scheme is similar to that used in PSK receiver. The only difference is that whereas PSK receiver. The waveform of twice the carrier frequency was generated using a squaring circuit the BPF of $2f_c$ and frequency divider ($\div 2$) were power by two cascaded. squaring circuits then the BPF of $4f_c$ and frequency divider($\div 4$) give the two carrier frequencies to cos $\omega_0 t$ and sin $\omega_0 t$.



QPSK Reciever