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ID: 12671   Programe: BE (E)

Q1

Sol:-

(a)

(a) The amplitude ~~seep~~ spectrum is thus

$|c_n| = |0.25 \operatorname{sinc}(\frac{n}{4})|$.

We get the power spectrum as follows.

$$P(f) = \sum_{n=-\infty}^{\infty} \left(\frac{1}{10}\right) \operatorname{sinc}^2\left(\frac{n}{4}\right) \delta (f - nf_0)$$

(b) Using table the total average power is than as follow

$$P = \left(\frac{1}{T_0}\right) \int_{-T_0/2}^{T_0/2} |g(t)|^2 dt = \int_{-1/8}^{1/8} 4 dt = 1$$

The power content in terms of the number of power components. These are nine power components in the frequency range [-4, 4] whose sum is that:-

$$\sum_{n=-4}^{4} \left(1/10\right) \operatorname{sinc}^2\left(n/4\right) = 0.904$$ (i-e, over 90% total power is take).

By considering a wider range than the frequency range [-4, 4], The portion

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of power can be beyond 90% To this end, we now needed to solve

$$\sum_{n=-w}^{w} \left(\frac{1}{16}\right) \operatorname{sinc}^2\left(\frac{n}{4}\right) \geq 0.96$$

for w where w is an integer. for $w = 13$ Hz, more than 96% of the total power lies in the frequency range [-13, 13]

[i-e ..., a total of 27 power components.

x================x

Q1 part (b)

* f(x) absolutely integrable over a period.
* f(x) must have finite number of exterma in any given interval i-e, there must be a finite number of maxima and manima in the interval.
* f(x) must have a finite number of discontinues in any given interval

however the discontinuity cannot be finite.

⊛ f(x) must be bounded

The last three conditions are satisfied if f is a function of bounded variation over a period.

Q#2 Part (A) Fourier Transform for Periodic Signals.

The Fourier Transform of a periodic signals in a strict mathematical sense does not exist as periodic signals are not energy signals. In a limiting sense, fourier Transform can be defined ~~as~~ for complex exponentials.

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~~g(t)~~

Consider the periodic signals $g(t)$ with period T_0, we can then define the ~~peri~~ periodic signals $g(t)$ using the generating function $p(t)$ where $p(t)$ equals $g(t)$ over one signal period and is zero elsewhere as shown

$$g(t) = \sum_{m=-\infty}^{\infty} p(t - mt_0)$$

Complex exponential fourier series.

$$g(t) = \sum_{m=-\infty}^{\infty} c_n \exp\left(\frac{j2\pi nt}{T_0}\right)$$

$$c_n = \left(\frac{1}{T_0}\right)\int_{-T_0/2}^{T_0/2} g(t)\exp\left(-\frac{j2\pi nt}{T_0}\right)dt = f_0$$

$$\int_{-\infty}^{\infty} p(t)\exp(-j2\pi n f_0 (t)dt = f_0 P(n T_0).$$

where c_n is the complex coefficient $P(t)$ is the fourier Transform of $p(t)$ and

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$f_0 = 1/T_0$ As the RHs indicates.

X ———— X

Q#2 Part (B)

Determine the fourier Transform of $g(t) = \text{sinc}(t)$

Sol:-

We know

$$\left(u\left(t+\frac{1}{2}\right)-u\left(t-\frac{1}{2}\right)\right) \leftrightarrow \text{sinc}(f)$$

By using duality property of the Fourier Transform.

we get

$$\text{sinc}(t) \leftrightarrow \left(u\left(-f+\frac{1}{2}\right)-u\left(-f-\frac{1}{2}\right)\right)$$

Since $\left(u\left(-f+\frac{1}{2}\right)-u\left(-f-\frac{1}{2}\right)\right)$ is an even function. we than have following.

$$\text{sinc}(t) \leftrightarrow \left(u\left(f+\frac{1}{2}\right)-u\left(f-\frac{1}{2}\right)\right) \quad \text{Ans.}$$

X ———— X

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Q#3 Part (A)

Absolute Bandwidth:→

~~Ab~~ Absolute bandwidth provides a theoretical definition. Assuming the spectrum is zero beyond $f_2 \geq f_1 \geq 0$, ~~where~~ we have $B = f_2 - f_1$ absolute bandwidth can be applied to frequency-limited Signals and ideal lowpass and bandpass filters. No absolute bandwidth can be defined for high-pass filter, as $f_2 \to \infty$ and consequently $B \to \infty$. In essence for all realizable Signals and filters. the absolute bandwidth is infinite.

X————X

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Q# 3 part B

Sol:-

$$x(t) = u(t) - u(t-3)$$

After substituting x(t) in the signals output as then as follows.

$$y(t) = \int_{-\infty}^{\infty} x(t)\, h(t - t)\, dt$$

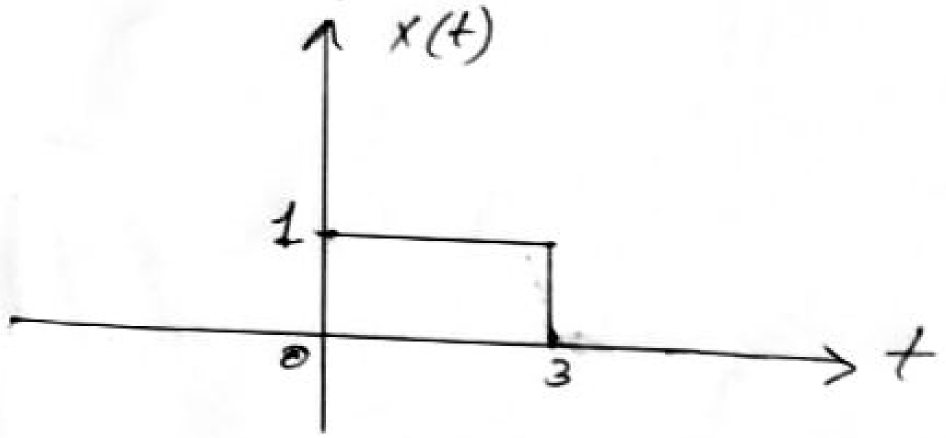
$$= \int_{-\infty}^{\infty} u(t)\, u(t-t)\, dt - \int_{-\infty}^{\infty} u(t)\, u(t-2-t)\, dt$$

$$- \int_{-\infty}^{\infty} u(t-3)\, dt - r)\, dt + \int_{-\infty}^{\infty} u(t-3\, (t-2-r)\, dr$$

$$= t u(t) - (t-2) u(t-2) - (t-3) u(t-3) + (t-5)\, u(t-5).$$

x(t), h(t) and y(t)

(a)



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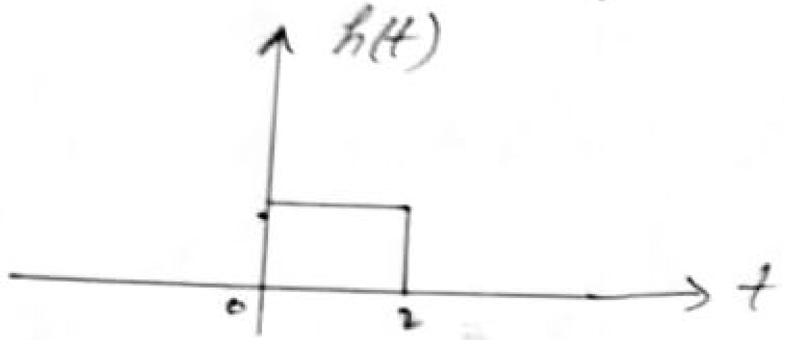
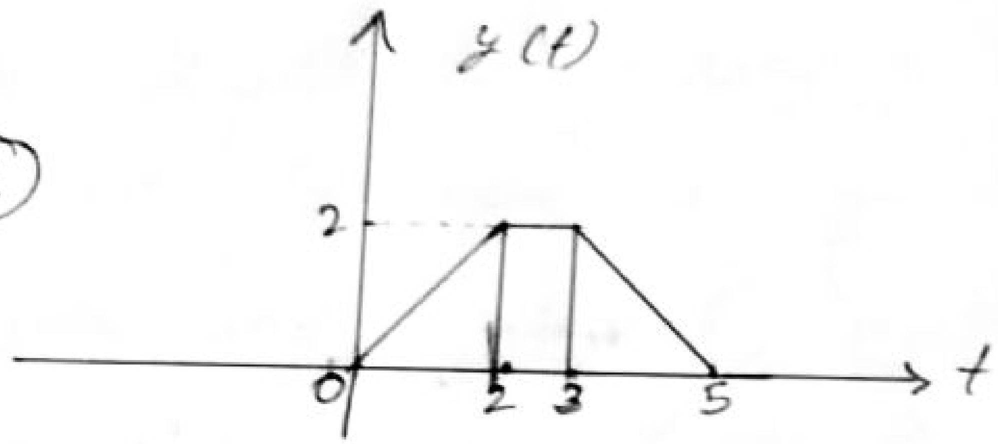
(8)

(b)



(c)



X ———— X

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Q#4 part (A)

Sol:-

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

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

where denotes the convolution operation. Equation is called the convolution integral, and Shows that $y(t)$, which is the response to $x(t)$, is the convolution of the Input $x(t)$ and the Impulse response $h(t)$. Note that $y(t)$ is nonzero for the interval that ~~the~~ is the sum of the interval during which $x(t)$ and $h(t)$ are nonzero. In other words, if $x(t)$ is limited to the time intervals $[a, b]$ and $h(t)$ is limited to the time interval

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[c, d], y(t) is then limited to the time interval [a+c+b+d] Equation reflects that fact that the present value of the output Signal is a weighted integral over the past history of the input Signal. Impulse response h(t) acts as a memory function for the system for a causal LTI system there can be no output prior to the time t= 0. Therefore, the lower limit Integration in can be changed to zero For an LTI system the impulse h(t) contains all the information needed and thus completely characterizes the system.

X———————X

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Q# 4 part (B)

Sol:→

using the fourier Transform of the input Signal is as fallows.

$$X(f) = \frac{1}{1+j2\pi f}$$

we then determine the fourier Transform of the output Signal.

$$Y(f) = X(f)\,H(f) = \left(\frac{1}{2+j2\pi f}\right)\left(\frac{1}{1+j2\pi f}\right)$$

$$= \frac{1}{(2+j2\pi f)(1+j2\pi f)}$$

$$= \frac{1}{1+j2\pi f} - \frac{1}{2+j2\pi f}$$

Using the inverse fourier Transform

$$y(t) = e^{-1}u(t) - e^{-2t}u(t) = (e^{-t} - e^{-2t})\,u(t).$$

X————X

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Q#5 Part (A)

Distortionless Transmission:-

It is of paramount interest that in a communication channel the output Signal be an exact replica of the ~~output~~ Input Signal, after all, that is the ultimate goal in signal Transmission. therefore importent to determine the characteristics of a communication system that allows no distortion. In a distortionless Transmission, the iput and output Signals in the time domain have Identical shapes, except for a possible change of amplitude and a constant delay.

$$y(t) = kx(t - t_d).$$

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Nonlinear Distortions:-

A nonlinear system cannot be described by a Transfer function, as a change in the input signal may not directly produce a corresponding change in the output Signals. We assume here the system is memoryless in the sense that the output $y(t)$ depends only on the Input $x(t)$ at time t. To evalute the nonlinear distortion the common procedure is to approximate the Input-output relation, also known as the transfer characteristic by a power series of the ~~output~~ input $x(t)$.

$$y(t) = a_1 x(t) + a_2 x^2(t) + a_3 x^3(t) + \text{-----}$$

Assume $x(f)$ is the fouirer Transform of $x(t)$ the Fourier Transform of becomes.

$$y(f) = a_1 X(f) + a_2 X(f) * X(f) + a_3 X(f) * X(f) * X(f) \text{---}$$

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Q#5 prt B

In Ideal filter passes Signals at certain sets of frequency and completly reflects the rest. the most common types of filter are bellows.

1. low pass filter (LPF)
2. High pass filter (HPF)
3. Band pass filter (BPF)
4. Band ~~pass~~ stop filter (BSF)

which passes low high intermediate and all but intermediate frequency figure shows. the magnitude & phase responses of low pass filters high pass filter, Band pass filter & Band stop filters. Most ~~common~~ Communication filters are low pass filter & Band pass filter.

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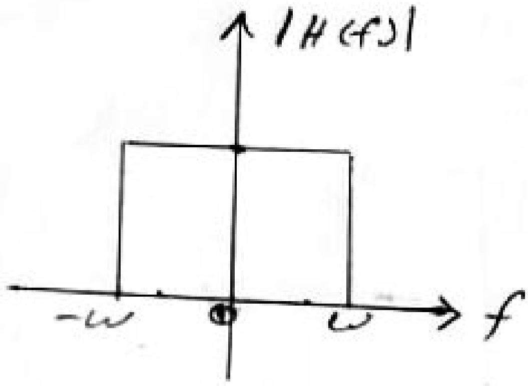
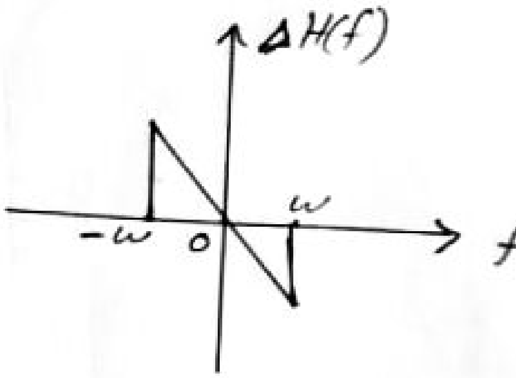
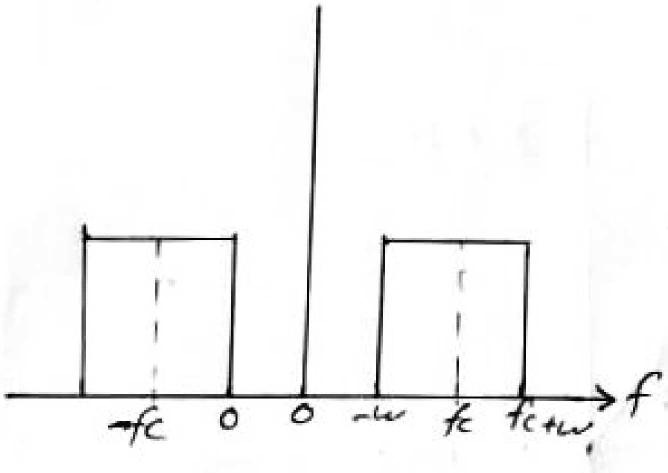
Figure Graphs.

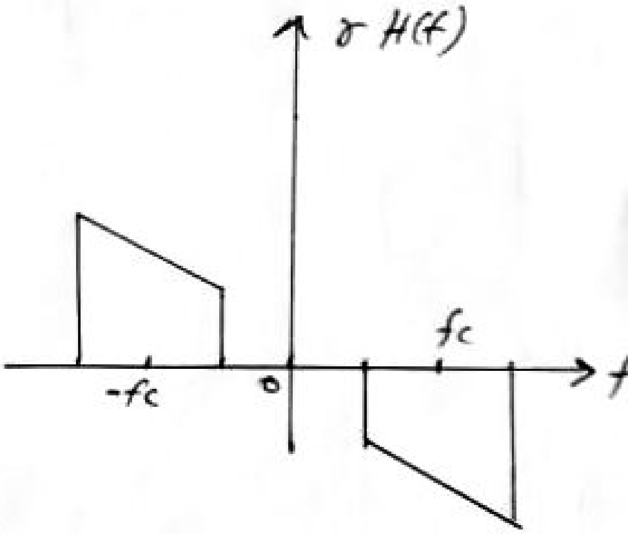
low pass filter



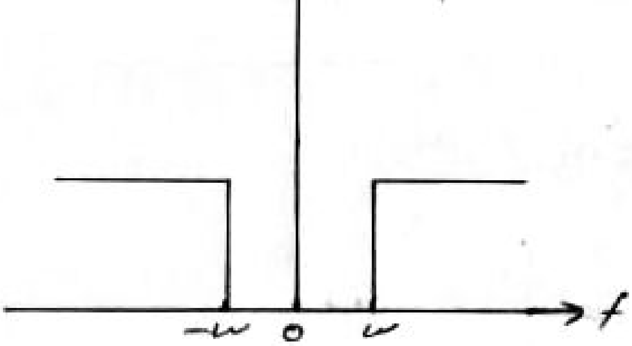


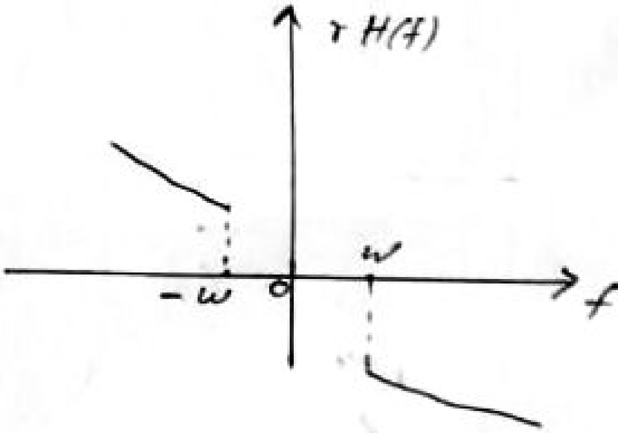
High pass filter





Band pass filter





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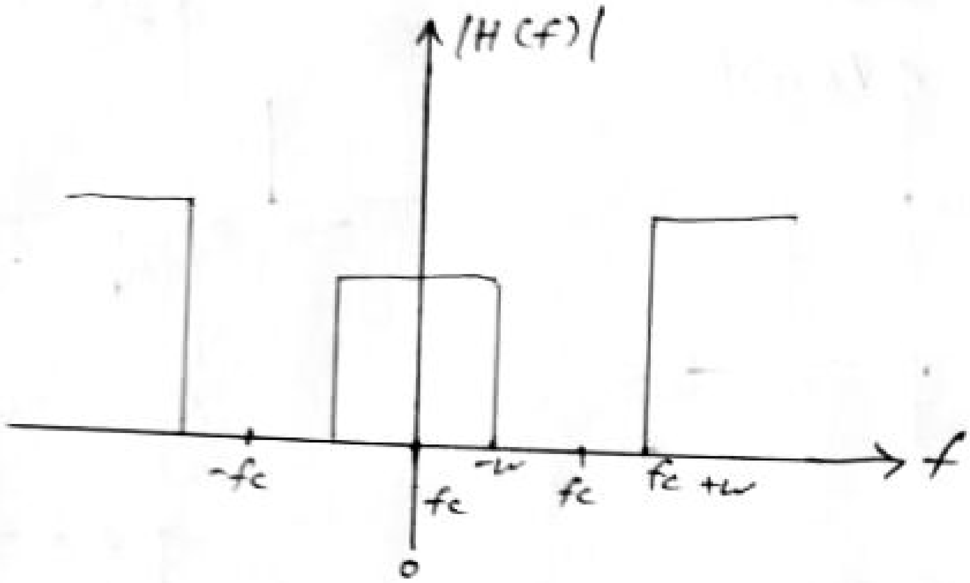
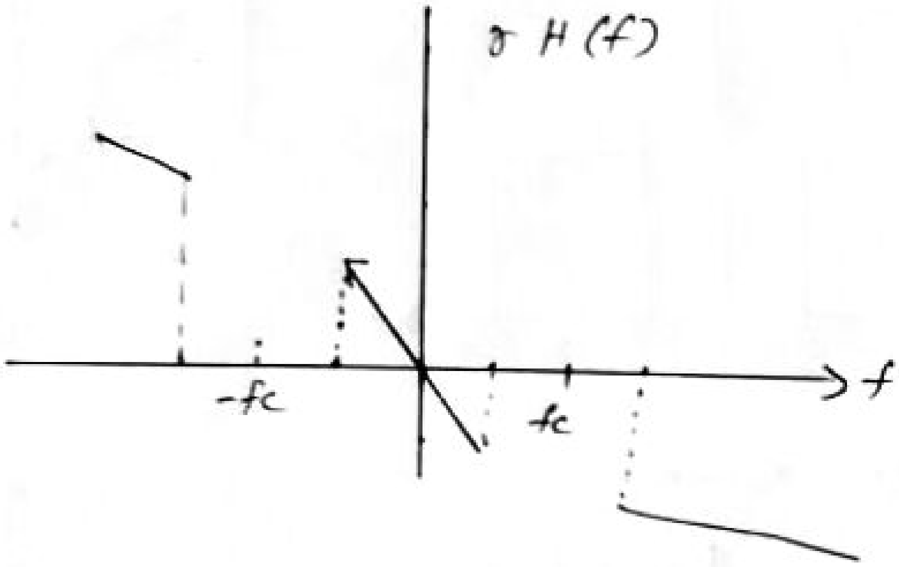
Band stop filter





The end.