Assignment NO: 4

* Communication System *

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Section: B

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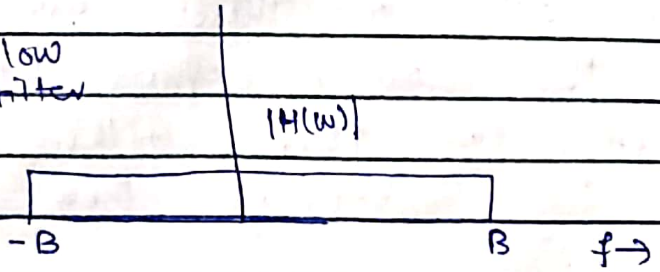
Ideal Filter:

Ideal filter allow distortion transmission of certain band of frequency and suppresses all Remaining frequencies.

Ideal low Pass Filter



$H(f) = Ke^{-j2\pi f t_d}$

$|H(f)| = K$

$\theta_h(f) = -2\pi f t_d$

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Let to above figure represent the response of an Ideal filter. Now in this case the pass band filter is distortionless. For a filter to have destortionless response in its pass band the following two condition must meet.

(i) Magnitude response must be constant.

$|H(\omega)| = K$ while K is constant.

(2) Phase Response must be linear or must be zero for $f=0$

$\theta_h(\omega) = -2\pi f t_d$

$\theta_h(\omega) = 0$ for $f=0$

Types:

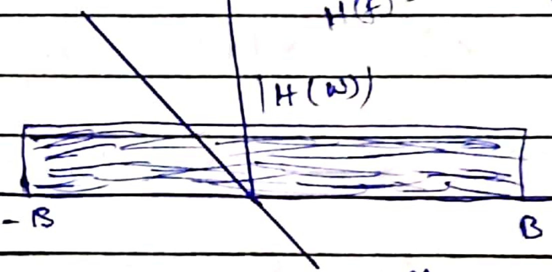
Ideal filter may be

* low pass filter
* high pass filter
* band pass filter
* Notch / Band - Reject

As all of these are Ideal in this case, therefore let discuss low pass filter (Ideal) as logic behind these are same.

The figure show on next page is Respone of Ideal low pass filter.

$$H(f) = \text{rect}\left(\frac{1}{2B}\right) e^{-j2\pi f t d}$$



(Plot A)

$$F\theta n(f) = 2\pi f t d$$

This Ideal low pass filter allow all Components below F= BHz to pass without distortion or suppress all Components above ~~BSA~~ f= BHz
BHz is the cutt off frequency.

As above figure look like gate signal of megnitude 1
Therefore it's transfer function and can be written as,

$$H(f) = 1 \cdot \text{rect}(f/2B)\, e^{-j2\pi f t d}$$

↳ Response of above figure.

<u>Impulse Response:</u>
To find its Impulse Response from the given transfer function we need Inverse Fourier transform.

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$$g(t) \xleftrightarrow{FT} G(\omega)$$

$$g(t-td) \xleftrightarrow{FT} G(\omega)\, e^{-2\pi f td}$$

$$G(\omega) \xleftrightarrow{F^{-1}} g(t)$$

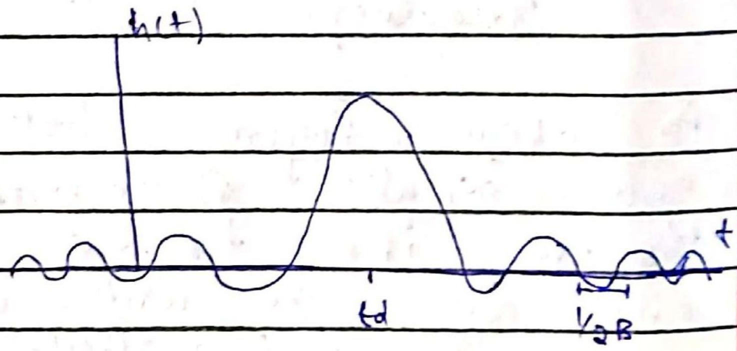
$$G(\omega) e^{-2\pi f td} \longleftrightarrow g(t-td).$$

$$h(t) = F^{-1}(H(f))$$

$$h(t) = F^{-1}(\text{rect}(f/2B)\, e^{-j2\pi f td}$$

$$h(t) = 2B\,\text{Sinc}\,(2\pi B\,(t-td)) \rightarrow (a)$$

eq (a) is the Impulse response of the plot (A) of $h(t)$ will be look like.



$h(t)$ has Infinite Samples as it goes from $-\alpha$ to $+a$

And it is not possible to Implement such a System having Infinite Sample in its Impulse response.

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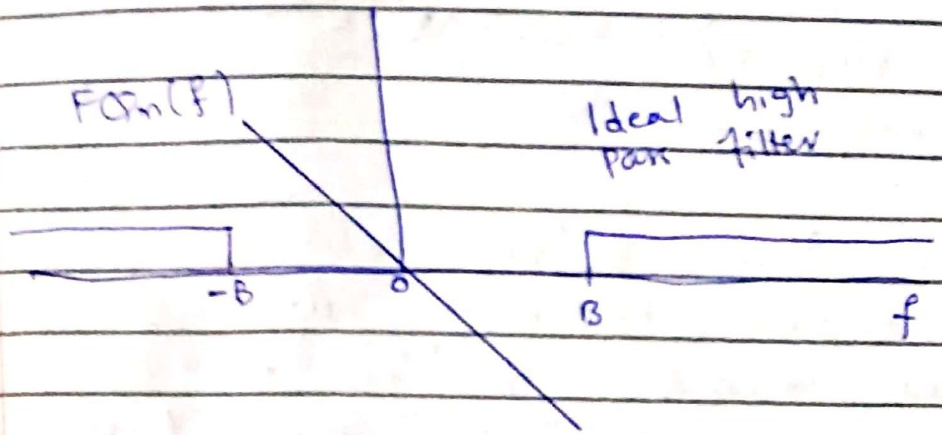
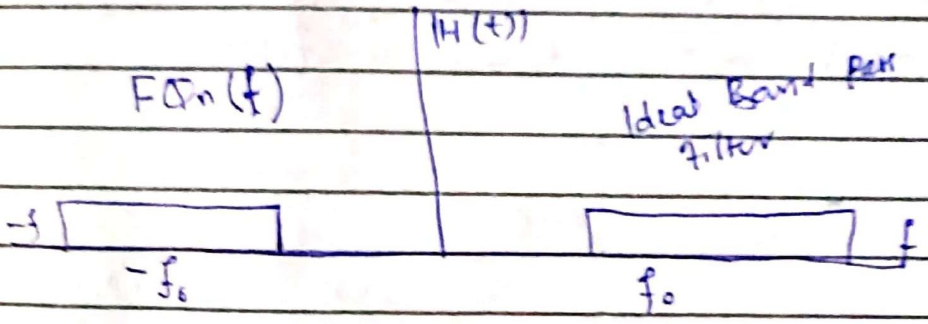
Examples:





Conclusion:

Ideal filter are not realizable since their unit response are everlasting (Think of the Sinc function)

Purpose of Ideal filter:

we often use the Ideal filter in our which as sharp stop band. In frequency and accurate bandwidth

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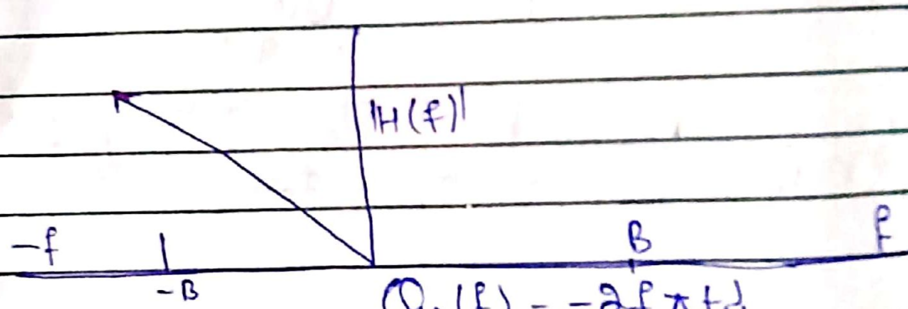
Suppose:

$$H(f) = \begin{cases} (1+k\cos 2\pi f t)\, e^{-2\pi f t_d} & |f|<B \\ 0 & |f|>B \end{cases}$$

In This $H(f)$

$|H(f)| = 1+k\cos 2\pi f t$ which is the function of "f" not a Constant These for it is not an Ideal filter

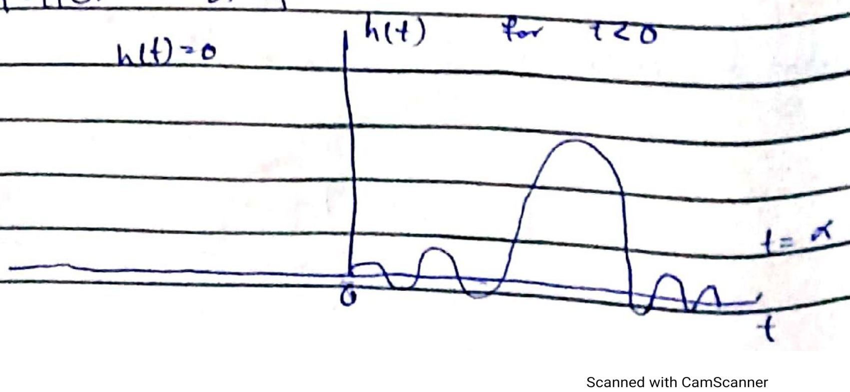


$\theta_h(f) = -2f\pi t_d$

Practical Filter:

Practical filters have long tail Complex Impulse response nonfixed bandwidth,

* Complex transfer function expression Impulse response for the physically realizable filter is,



As $h(t) = 0$ for $t < 0$ Therefore we can only obtain approximated version of the Ideal low-pass filter, high-pass + band-pass filters.

it is because that in Ideal ~~that~~ filter the no of sample are from $-\alpha$ to $+\alpha$ while in practical filter the no of Samples start from $t=0$ to $t = t_m$. t_m depend on filter designer

during practical filter implementation using large no of ~~filter~~ samples, filter response will be good, else will not be good b/c in that (for small no of samples we will get transition region.

Transition Region:

Transition region is region where the curve of Amplitude vs frequency continuosly change b/w high level and low level

it is also called transition band and can be define as the

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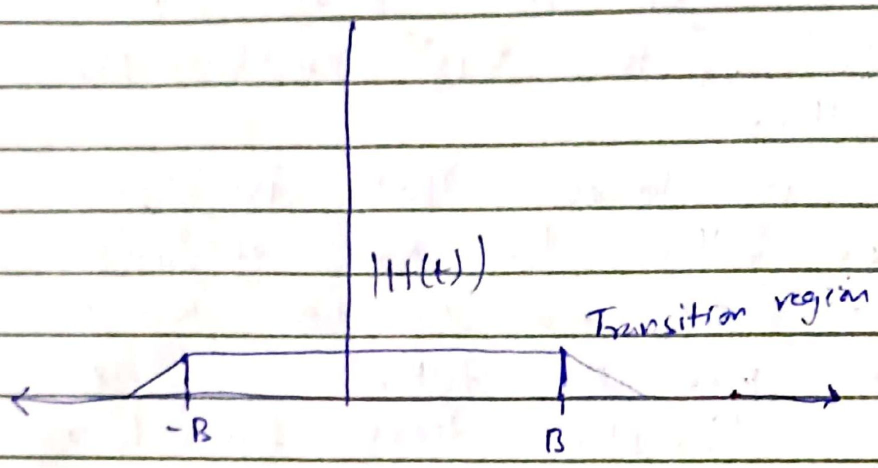
The interval b/w a pass band and stop band.



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Program:

```
clc;
clear all;
alpha p= 2;
wp =[0.2*pi, 0.4*pi];
ws =[0.1 * pi, 0.5*pi];
% to find cutt off frequency
[n,w,n]= butterd(wp/pi, ws/pi, alphap, alphas]
[b,a] = butter (n,Wn)
w = 0: 0.01 : pi;
[h,ph] = freqz (b,a,w);

m= 20*log10 (abs(h));
an = angle(h);
Subplot (2,1,1);
plot (ph/pi, m)
grid on;
xlabel ('normalized frequency');
ylabel ('gain in dB')
Subplot (2,1,2)
plot (ph/pi ,ah)
grid on;
```

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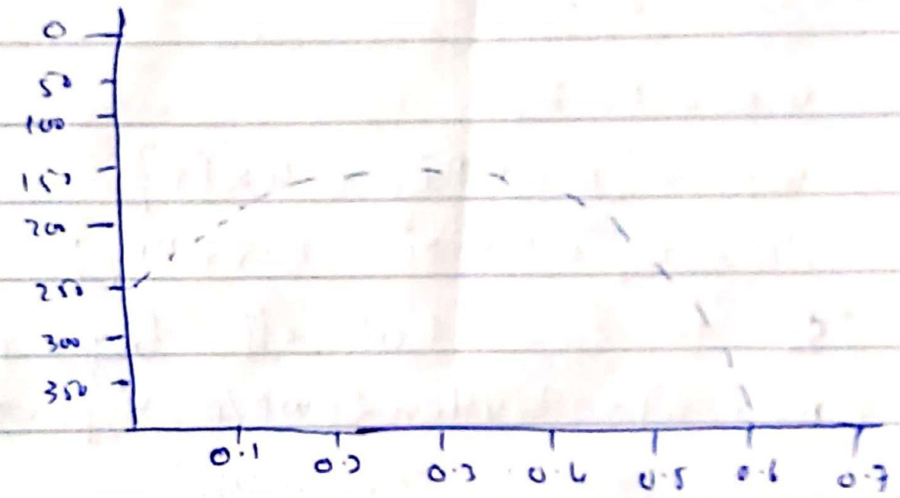
xlabel ('normalized frequency')
ylabel (' phase in radian')



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