

Amoo Oluwaferanmi

18/Eng04/077

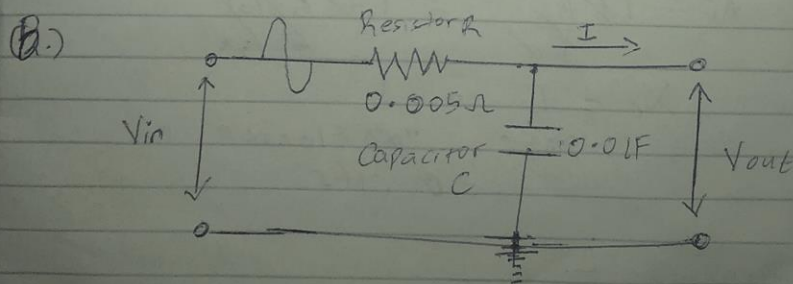
Electrical electronics Engineering

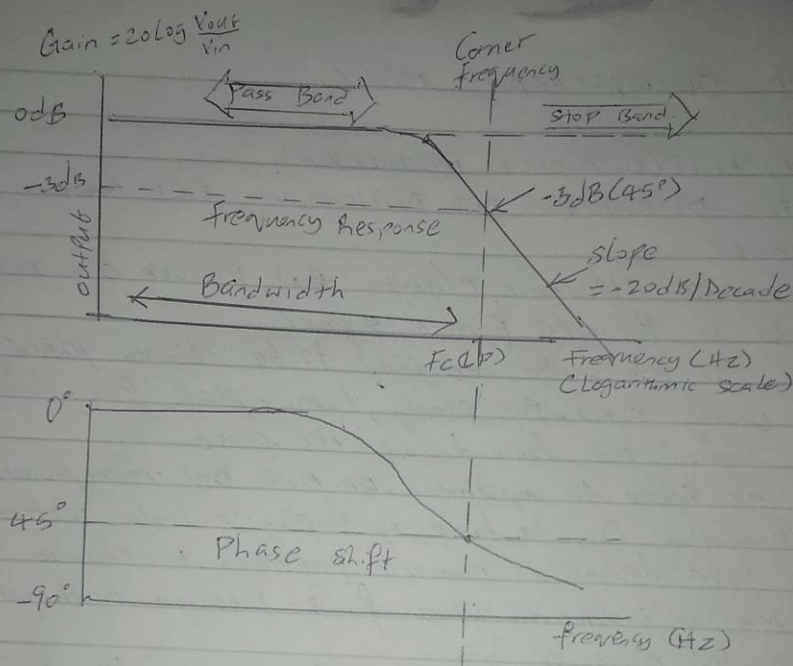
Software Development and Applications Classwork

AMOO OLUWAFERANMI MARIS
18/EN04/097

ELECTRICAL/ELECTRONICS ENGINEERING
SOFTWARE DEVELOPMENT AND APPLICATION EEE 342
CLASSWORK

- (A) Filters are electronic circuits that Remove any unwanted Components or features from a signal
- (a) They can easily be designed to be "Linear phase" (and usually are). Put simply, linear-phase filters delay the input signal but don't distort its phase
 - (b) They are simple to implement. On most DSP microprocessors, the FIR calculation can be done by looping a single instruction
 - (c) They have desirable numeric properties. In practice all filters must be implemented using finite-precision arithmetic, that is, a limited number of bits.
 - (d) They can be implemented using fractional arithmetic. Unlike HTK it is always possible to implement a FIR filter using coefficients with magnitude less than 1.0
 - (e) They are suited to multi-rate applications. By multi-rate we mean either "decimation" (reducing the sampling rate), "interpolation" (increasing the sampling rate), or both.
 - (f) They are economical or cost-effective
 - (g) Unlike passive filter circuits, Active filter circuits require power supply.





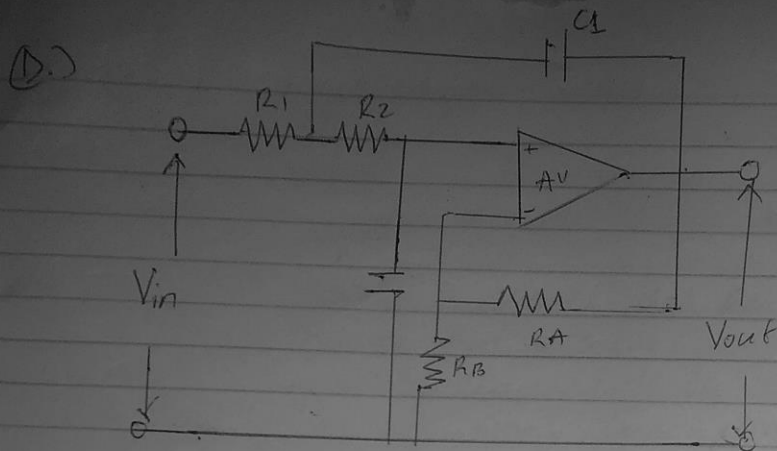
The Bode plot shows the Frequency response of the filter to be nearly flat for low frequencies and all of the input signal is passed directly to the output, resulting in a gain of nearly 1, called unity, until it reaches its cut-off frequency point (f_c). This is because the reactance of the Capacitor is high at low frequencies and blocks any current flow through the Capacitor.

$$\text{Amplitude } A_v = 1 + \frac{R_x}{R_s} = 1 + \frac{5 \times 10^3}{2 \times 10^3}$$

$$X_c = \frac{1}{2\pi f_c} = \frac{1}{2\pi \times 1000 \times 0.01} = 0.016 \Omega$$

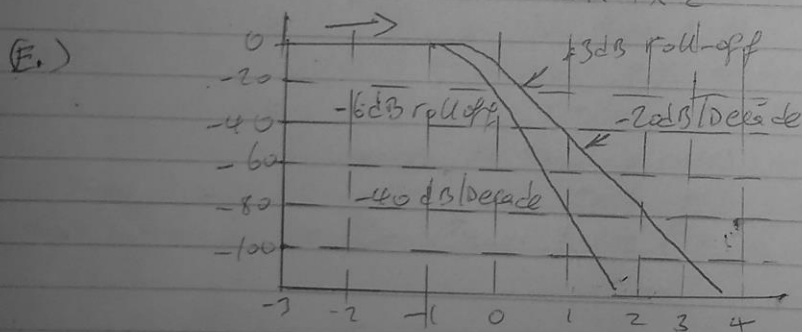
(c.)

$$\begin{aligned} \text{Cut off frequency } f_c &= \frac{1}{2\pi RC} \\ &= \frac{1}{2\pi \times 0.005 \times 0.01} = 3183.1 \text{ Hz} \end{aligned}$$



This second order low Pass Filter Circuit has two RC networks, $R_1 - C_1$ and $R_2 - C_2$ which give the filter its frequency response properties. The filter design around a non-inverting op-amp configuration so the filter's gain, A will always be greater than 1.

$$V_{out} = V_{in} \times \frac{X_c}{\sqrt{R^2 + X_c^2}} = V_{in} \frac{X_c}{Z}$$



The frequency response plot above, is basically the same as for a 1st-order filter. The difference this time is the steepness of the roll-off which is -40 dB/decade in the stop band. However, second order filters can exhibit a variety of responses depending upon the circuit voltage magnification factor, Q at the cut-off frequency point.

