<u>MID SEM EVALUATION : ECS 330</u>

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Order of Solving questions: Question 2, Question 1, Question 5

Question 2:

A Butterworth filter is a type of signal processing filter designed to have a frequency response as flat as possible in the passband (maximally flat magnitude filter).

Here is the code for the filter design. To demonstrate the output of the filter we change the value of F_s and monitor the waveforms.

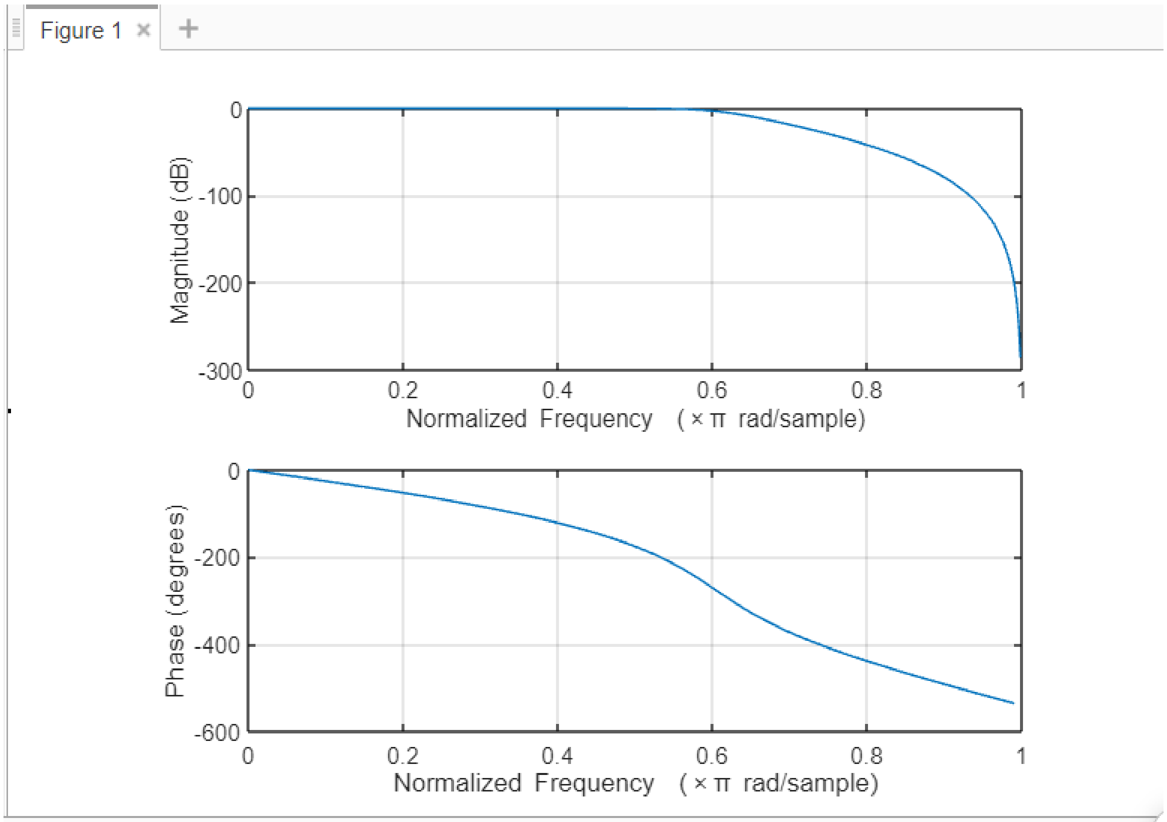
```
% Butterworth LowPass filter designed using FDESIGN.LOWPASS.

% All frequency values are in Hz.
Fs = 10000;  % Sampling Frequency

N  = 6;     % Order
Fc = 3000;  % Cutoff Frequency

% Construct an FDESIGN object and call its BUTTER method.
h  = fdesign.lowpass('N,F3dB', N, Fc, Fs);
Hd = design(h, 'butter');
```

For 10000 Hz as the sampling frequency, the output is as follows :

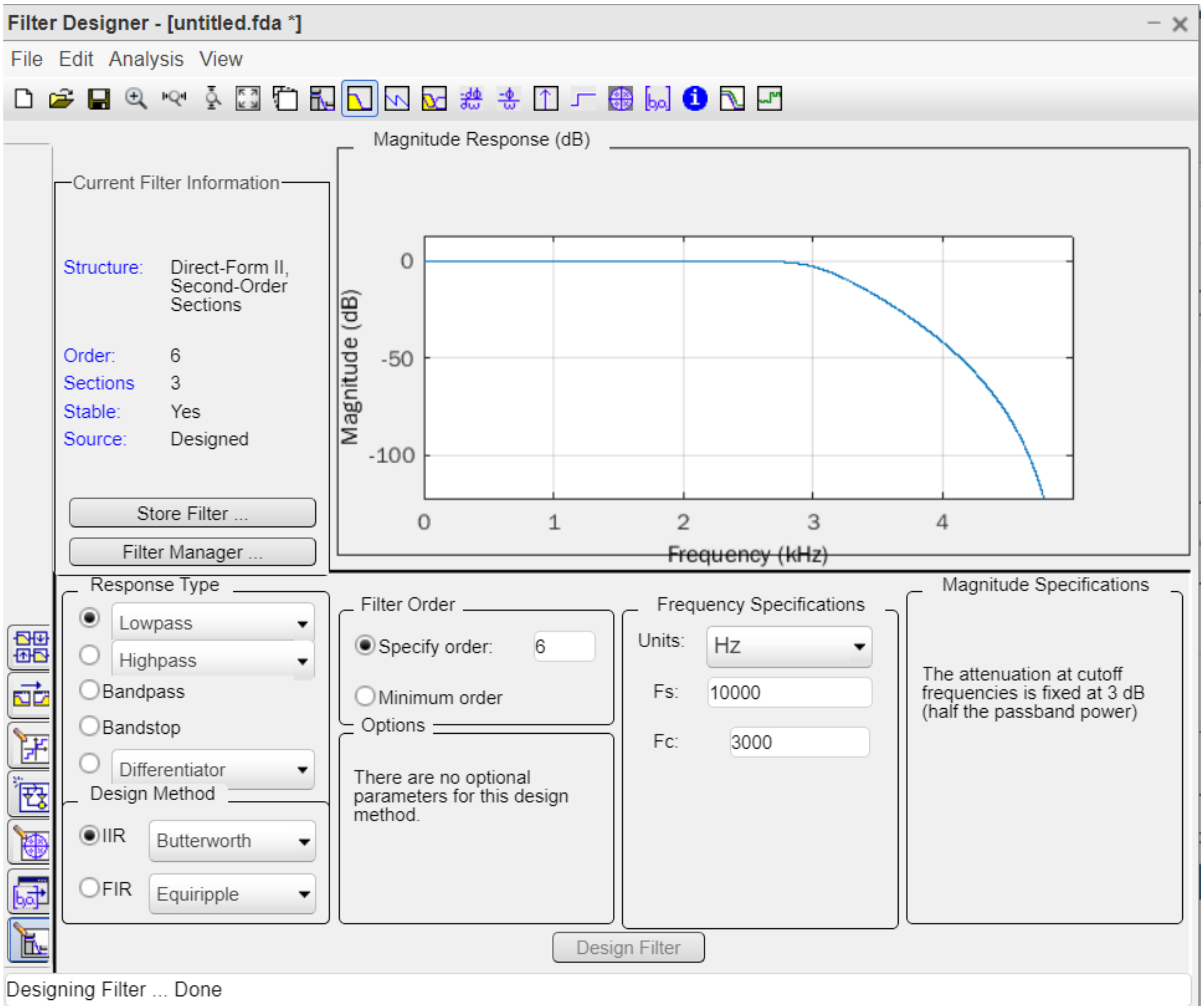


If the frequency is more than the cut-off frequency, it will roll-off towards zero with the rate of -20 dB/decade for the first-order filter. Here, we have considered it to be a 6th order filter which will be -120dB/decade

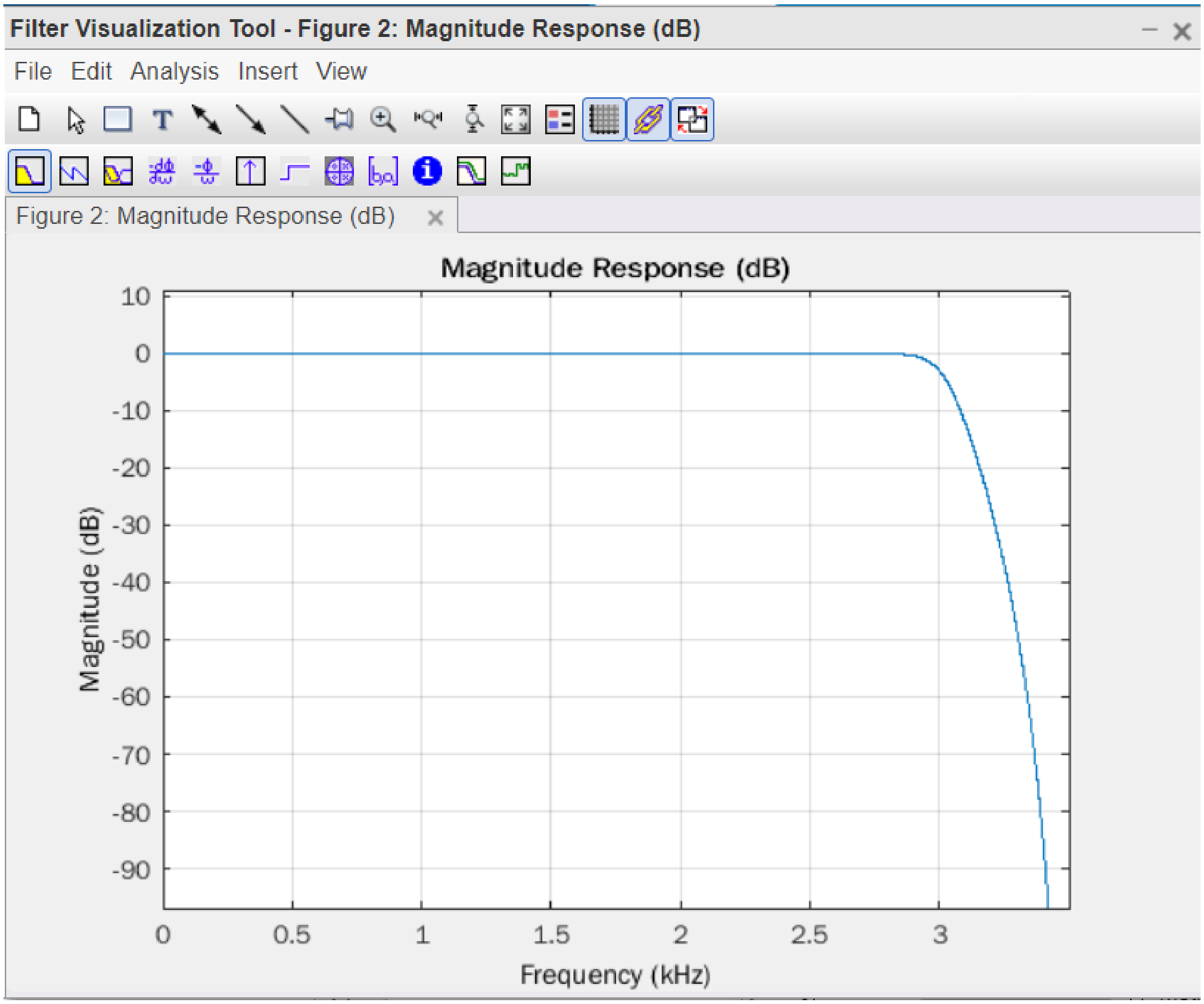
If one increases the order of the filter, the rate of a roll-off period is also increased. The quality factor for the Butterworth filter is 0.707.

So, if we increase the order of the filter, the number of cascade stages with the filter is also increased. But in practice, we cannot achieve Butterworth's ideal frequency response as it produces **excessive ripples** in the passband. **Hence, we chose 6 as an optimal order for designing the filter. The sampling frequency was varied to check the outputs for various inputs as shown in the graphs below.**

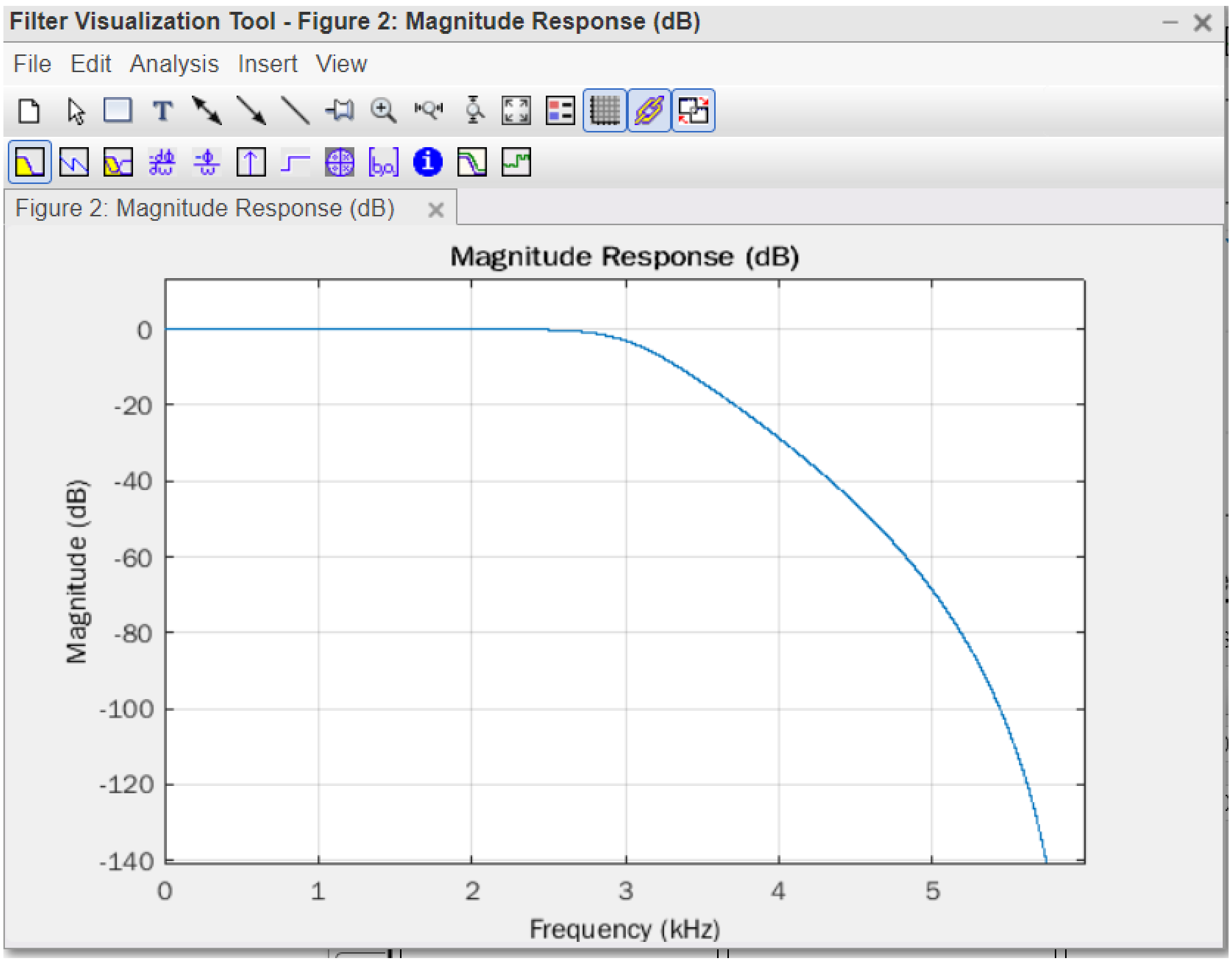
For F_s = 10000 Hz



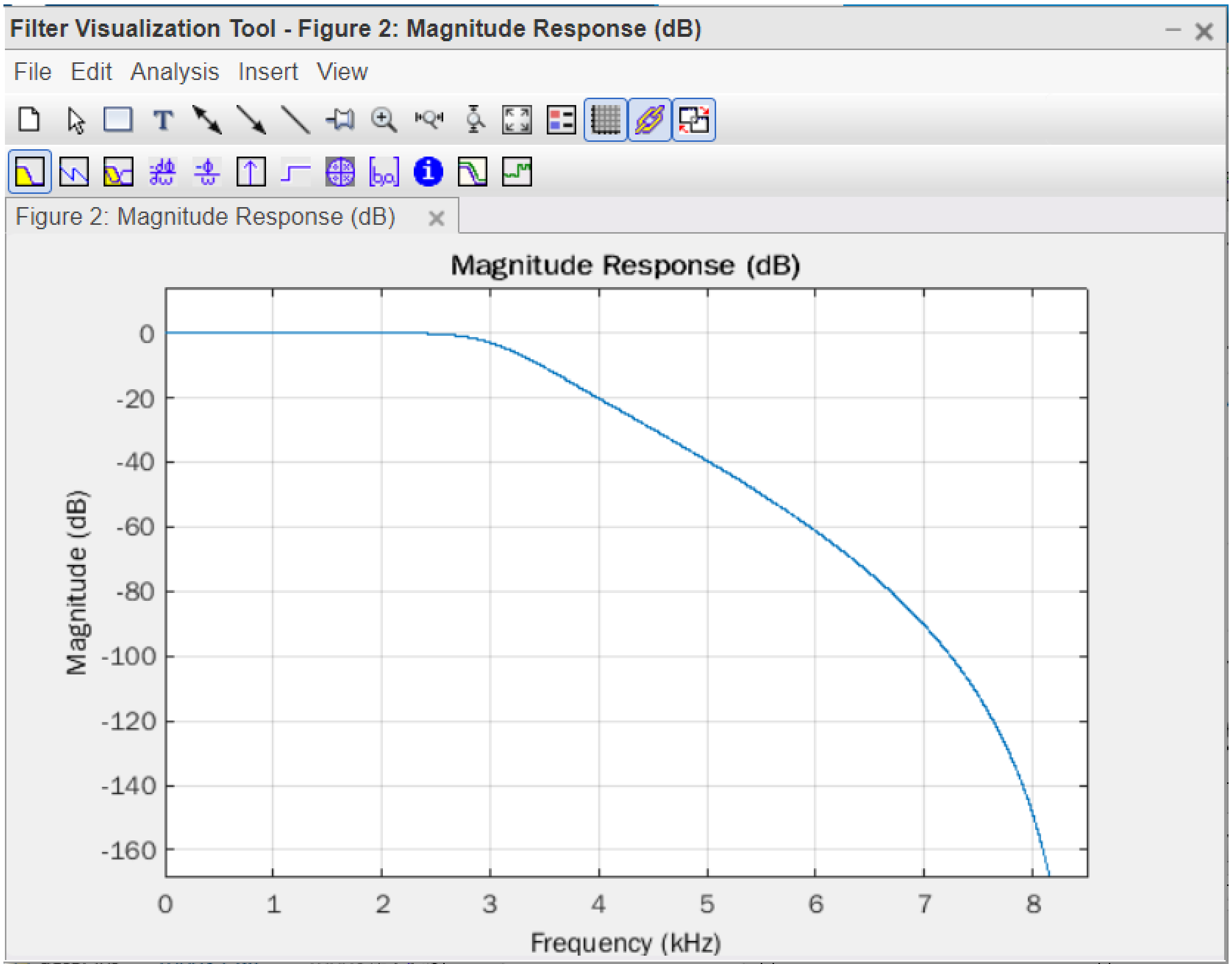
For a sampling frequency of 7000Hz



For a sampling frequency of 12000 Hz



For a sampling frequency of 18000 Hz

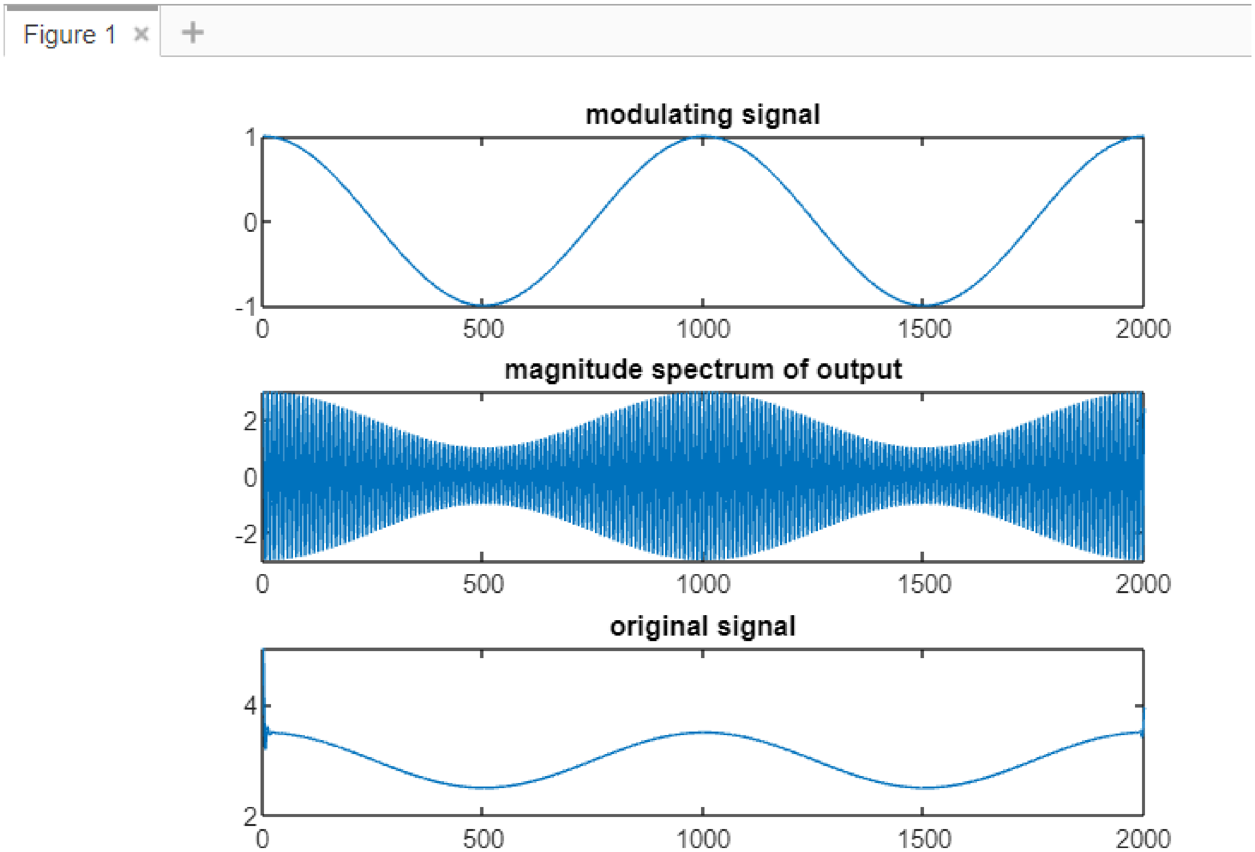


Question 1:

The following is the required code for this question :

```
clc;
clear all;
close all;
fc=100000; %carrier frequency
fs=1000000; %sampling frequency
f=1000;
m=0.5;
a=1/m;
opt=-a;
t=0:1/fs:((2/f)-(1/fs));
x=cos(2*pi*f*t);
y=modulate(x,fc,fs,'amdsb-tc',opt);
subplot(3,1,1);
plot(x);
title('modulating signal');
subplot(3,1,2);
plot(y);
title('modulated dsb-sc signal');
x_recov=demod(y,fc,fs,'amdsb-tc',opt);
subplot(3,1,3);
plot(x_recov);
title('original signal')
```

The following is the plot for the magnitude spectrum of the output from the product modulator



Question 5 :

We are given roll-off rate = -60db/decade, this implies that we require 60/20 = 3 stage Low Pass Filter.
The LT spice schematic simulated shows the three stage scenario

To have it as an active low pass filter, we need a fourth stage so as to get positive gain.

Transfer function **:**

H(w) = output voltage/source voltage = V_0 / V_s = $4(1/(1 + jw/w_{c1})^3$

$w_{c1} = 1/R_fC_f$

The cut-off frequency of the overall filter is 1kHz, let that be denoted as w_{c3}

$w_{c3} = 2\pi f_c = 2\pi \times 10^3$ rad/s

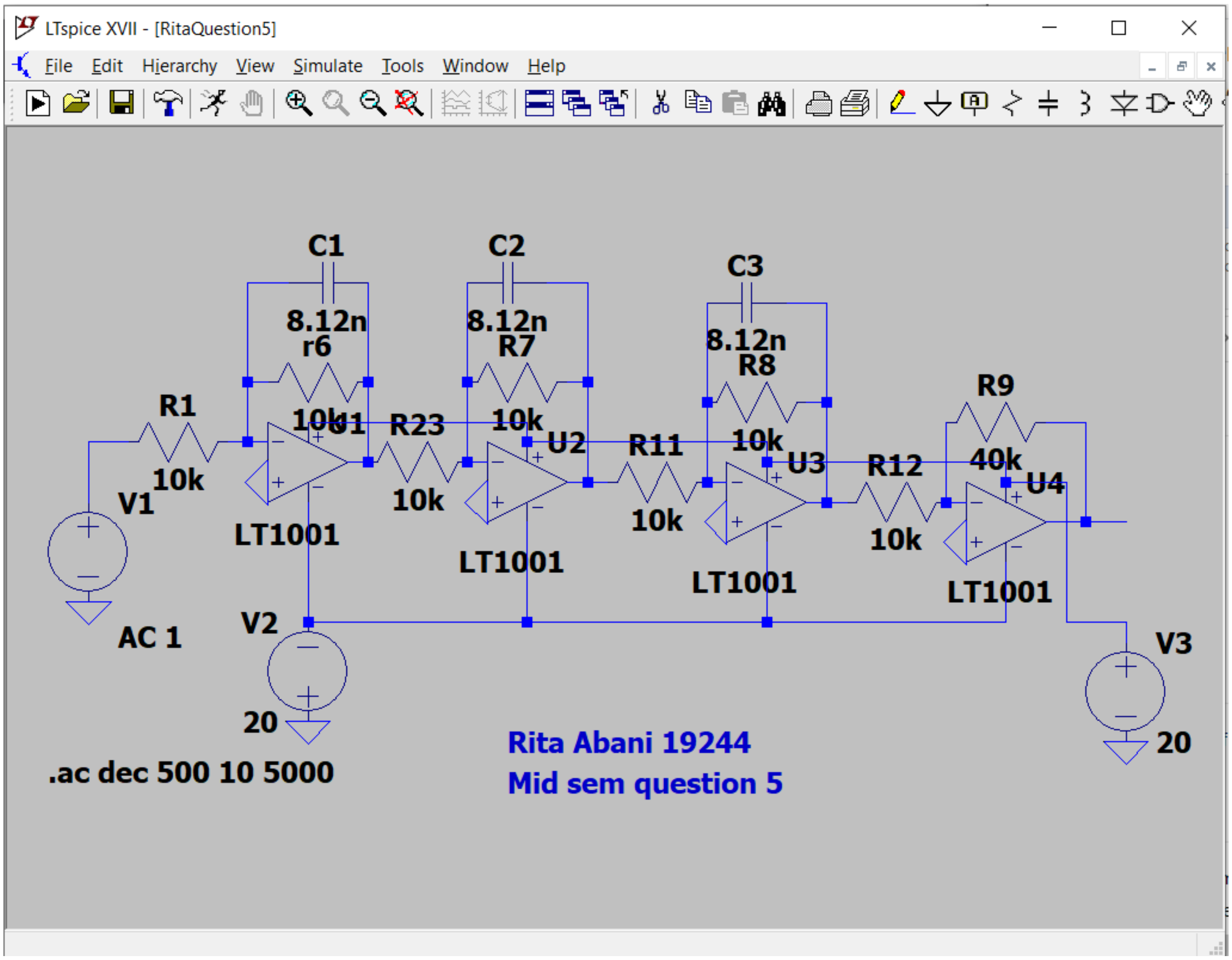
By theory, we can deduce graphically that $\mathbf{w_{c3} = 0.51\ w_{c1}}$ (which is also a standard formula)

So, $w_{c1} = w_{c3} / 0.51 = 2\pi \times 10^3 /0.51 = 12320$ rad/s

R_f is given to us to be 10k ohms

Hence, $C_f = 1/ R_fw_{c1} = 8.12$ nF

The screenshot of the LTSpice schematic is as shown below



The output waveform is shown below :