Name: **JONATHAN IFEGUNNI**

Section:

Laboratory Exercise 5

DIGITAL PROCESSING OF CONTINUOUS-TIME SIGNALS

5.1 THE SAMPLING PROCESS IN THE TIME-DOMAIN

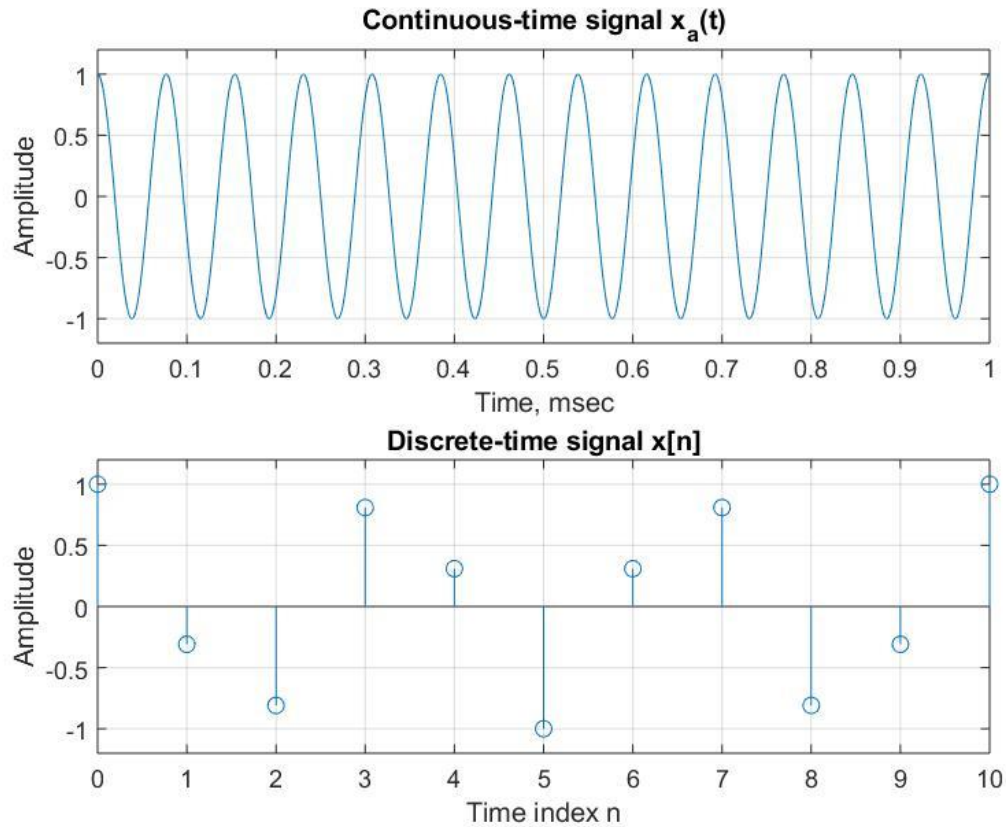
Project 5.1 Sampling of a Sinusoidal Signal

A copy of Program P5_1 is given below:

```
% Program P5_1
% Illustration of the Sampling Process
% in the Time-Domain
clf;
t = 0:0.0005:1;
f = 13;
xa = cos(2*pi*f*t);
subplot(2,1,1)
plot(t,xa);grid
xlabel('Time, msec');ylabel('Amplitude');
title('Continuous-time signal x_{a}(t)');
axis([0 1 -1.2 1.2])
subplot(2,1,2);
T = 0.1;
n = 0:T:1;
xs = cos(2*pi*f*n);
k = 0:length(n)-1;
stem(k,xs);grid;
xlabel('Time index n');ylabel('Amplitude');
title('Discrete-time signal x[n]');
axis([0 (length(n)-1) -1.2 1.2])
```

Answers:

Q5.1 The plots of the continuous-time signal and its sampled version generated by running Program P5_1 are shown below:



Q5.2 The frequency of the sinusoidal signal in Hz is – 13kHz

The sampling period in seconds is - 0.1 msec

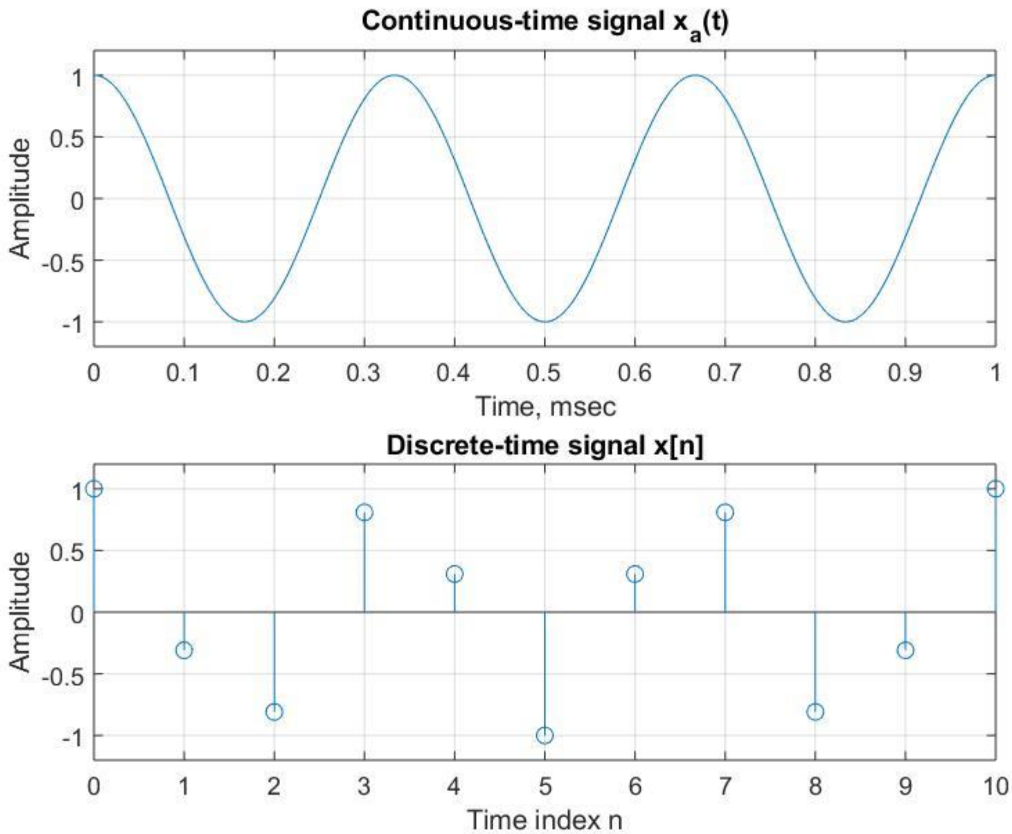
Q5.3 The effects of the two `axis` commands are –

Q5.4 The plots of the continuous-time signal and its sampled version generated by running Program P5_1 for the following four values of the sampling period are shown below:

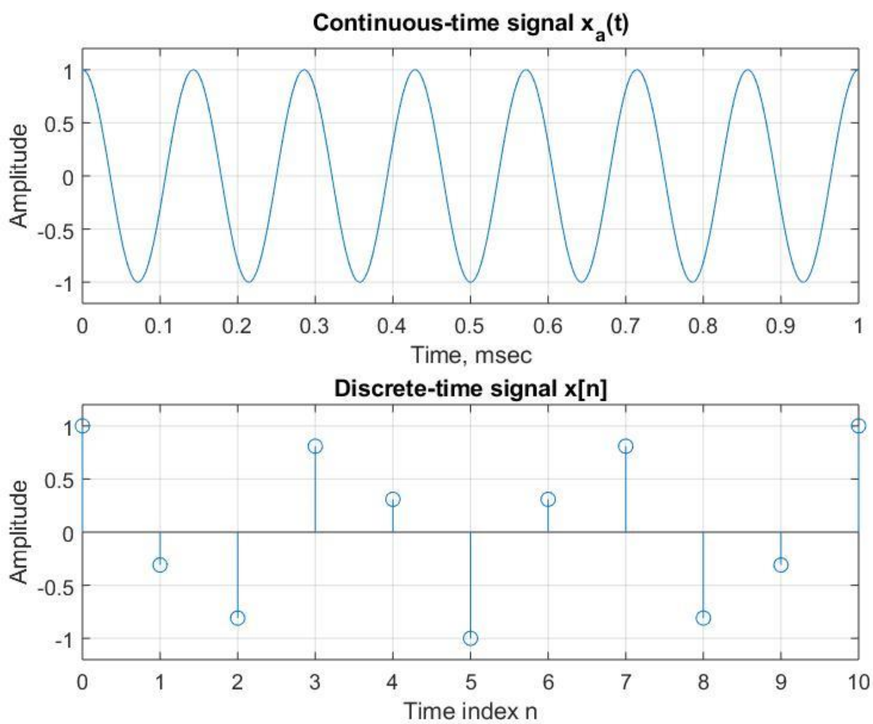
< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Based on these results we make the following observations -

Q5.5 The plots of the continuous-time sinusoidal signal of frequency 3 Hz and its sampled version generated by running a modified Program P5_1 are shown below:



The plots of the continuous-time sinusoidal signal of frequency 7 Hz and its sampled version generated by running a modified Program P5_1 are shown below:



Based on these results we make the following observations –In all cases the samples x[n] are exactly the same

Project 5.2 Aliasing Effect in the Time-Domain

A copy of Program P5_2 is given below:

Answers:

Q5.6 The plots of the discrete-time signal and its continuous-time equivalent obtained by running Program P5_2 are shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Q5.7 The range of `t` in the Program is -

The value of the time increment is -

The range of `t` in the plot is -

The plot generated by running Program P5_2 again with the range of `t` changed so as to display the full range of $y_a(t)$ is shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Based on these results we make the following observations -

Q5.8 The plots of the discrete-time signal and its continuous-time equivalent obtained by running Program P5_2 with the original display range restored and with the frequency of the sinusoidal signal changed to 3 Hz are shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

The plots of the discrete-time signal and its continuous-time equivalent obtained by running Program P5_2 with the original display range restored and with the frequency of the sinusoidal signal changed to 7 Hz are shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Based on these results we make the following observations –

These results can be explained as follows –

5.2 EFFECT OF SAMPLING IN THE FREQUENCY-DOMAIN

Project 5.3 Aliasing Effect in the Frequency-Domain

A copy of Program P5_3 is given below:

```
% Program P5_3
% Illustration of the Aliasing Effect
% in the Frequency-Domain
clf;
t = 0:0.005:10;
xa = 2*t.*exp(-t);
subplot(2,2,1)
plot(t,xa);grid
xlabel('Time, msec');ylabel('Amplitude');
title('Continuous-time signal x_{a}(t)');
subplot(2,2,2)
wa = 0:10/511:10;
ha = freqs(2,[1 2 1],wa);
plot(wa/(2*pi),abs(ha));grid;
xlabel('Frequency, kHz');ylabel('Amplitude');
title('|X_{a}(j\Omega)|');
axis([0 5/pi 0 2]);
subplot(2,2,3)
T = 1;
n = 0:T:10;
xs = 2*n.*exp(-n);
k = 0:length(n)-1;
stem(k,xs);grid;
xlabel('Time index n');ylabel('Amplitude');
title('Discrete-time signal x[n]');
subplot(2,2,4)
wd = 0:pi/255:pi;
hd = freqz(xs,1,wd);
% The following line has to be corrected from
Mitra's version
% by adding "2" to the denominator of wd
plot(wd/(T*2*pi), T*abs(hd));grid;
xlabel('Frequency, kHz');ylabel('Amplitude');
title('|X(e^{j\omega})|');
% The following line also has to be fixed to
properly range axes
axis([0 0.5/T 0 2])
```

Answers:

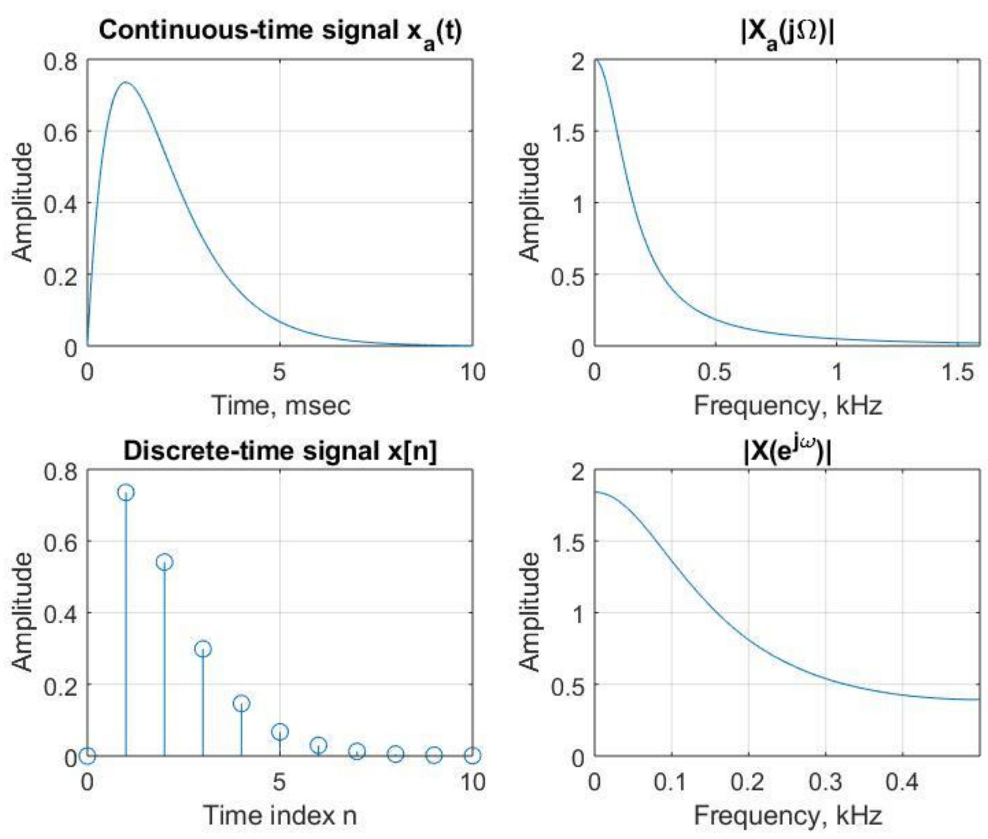
Q5.9 The continuous-time function `xa(t)` in Program P5_3 is -

The CTFT of `xa(t)` is being computed by –

$$X_a(S) = \frac{2}{(1+s)^2} = \frac{2}{s^2+2s+1}$$

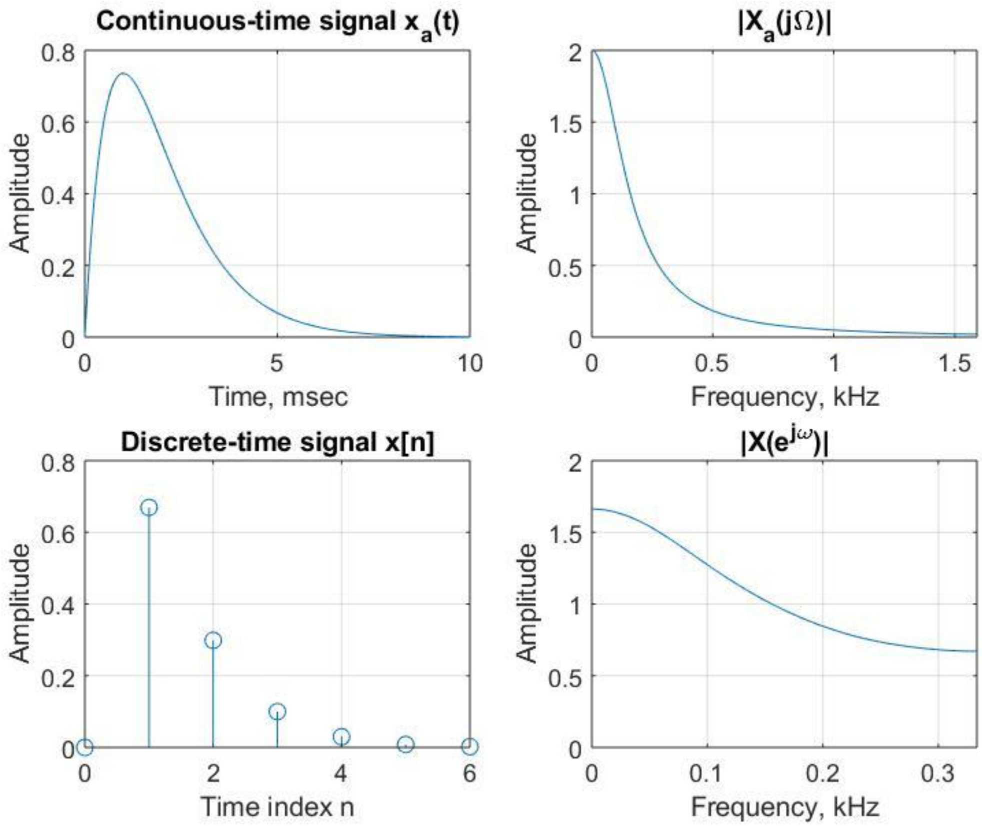
After finding the Laplace transform of the above, The vectors [2] and [1 2 1] which are the coefficient vectors were used as arguments to frequency to obtain complex-valued samples of the frequency response ( ) Xa Ω at the frequencies given in the vector wa.

Q5.10 The plots generated by running Program P5_3 are shown below:



Based on these results we make the following observations - There is aliasing because the analog signal is not bandlimited

Q5.11 The plots generated by running Program P5_3 with sampling period increased to 1.5 are shown below:



Based on these results we make the following observations – The Aliasing is much more severe with because of the reduction in sampling frequency

Q5.12 The modified Program P5_3 for the case of $x_a(t) = e^{-\pi t^2}$ is given below:

< Insert program code here. Copy from m-file(s) and paste. >

The plots generated by running the modified Program P5_3 are shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Based on these results we make the following observations -

The plots generated by running the modified Program P5_3 with sampling period increased to 1.5 are shown below:

< Insert MATLAB figure(s) here. Copy from figure window(s) and paste. >

Based on these results we make the following observations -

5.3 DESIGN OF ANALOG LOWPASS FILTERS

Project 5.4 Design of Analog Lowpass Filters

A copy of Program P5_4 is given below:

```
% Program P5_4
% Design of Analog Lowpass Filter
clf;
Fp = 3500;Fs = 4500;
Wp = 2*pi*Fp; Ws = 2*pi*Fs;
[N, Wn] = buttord(Wp, Ws, 0.5, 30,'s');
[b,a] = butter(N, Wn, 's');
wa = 0:(3*Ws)/511:3*Ws;
h = freqs(b,a,wa);
plot(wa/(2*pi), 20*log10(abs(h)));grid
xlabel('Frequency, Hz');ylabel('Gain, dB');
title('Gain response');
axis([0 3*Fs -60 5]);
```

Answers:

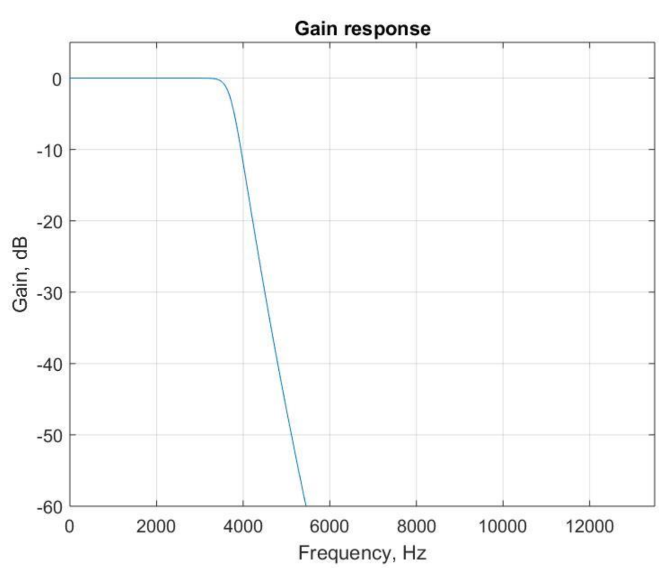
Q5.13 The passband ripple R_p in dB is – 0.5 dB

The minimum stopband attenuation R_s in dB is - 30 db

The passband edge frequency in Hz is – 3.5 kHz

The stopband edge frequency in Hz is - 4.5 kHz

Q5.14 The gain response obtained by running Program P5_4 is shown below:



Based on this plot we conclude that the filter designed __MEETS_____ the given specifications.

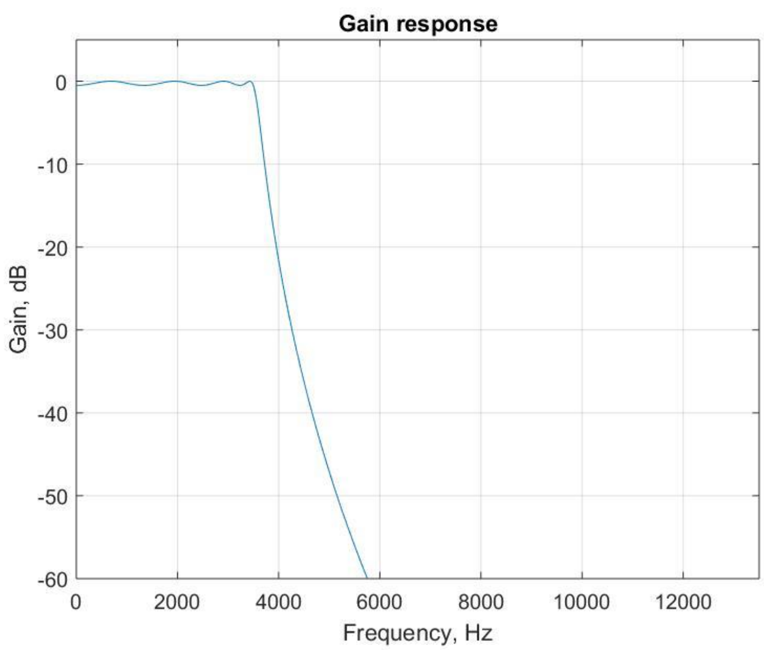
The filter order N is - 18

The 3-dB cutoff frequency in Hz of the filter is – 3.7144kHz

Q5.15 The required modifications to Program P5_4 to design a Type 1 Chebyshev lowpass filter meeting the same specifications are given below:

```
% Program Q5_15
% Design of Analog Lowpass Filter
clf;
Fp = 3500;Fs = 4500;
Wp = 2*pi*Fp; Ws = 2*pi*Fs;
[N, Wn] = cheb1ord(Wp, Ws, 0.5, 30,'s');
[b,a] = cheby1(N, 0.5, Wn, 's');
wa = 0:(3*Ws)/511:3*Ws;
h = freqs(b,a,wa);
plot(wa/(2*pi), 20*log10(abs(h)));grid
xlabel('Frequency, Hz');ylabel('Gain, dB');
title('Gain response');
axis([0 3*Fs -60 5]);
N
```

The gain response obtained by running the modified Program P5_4 is shown below:



Based on this plot we conclude that the filter designed __MEETS_____ the given specifications.

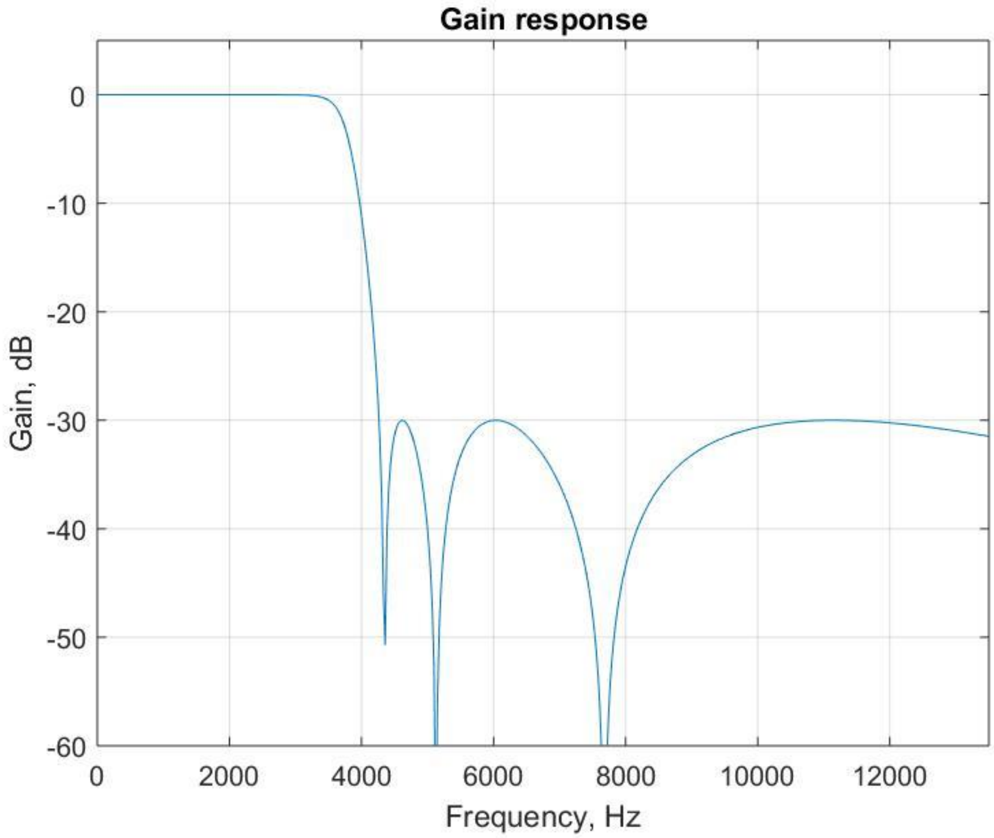
The filter order N is - 8

The passband edge frequency in Hz of the filter is – 3.5kHz

Q5.16 The required modifications to Program P5_4 to design a Type 2 Chebyshev lowpass filter meeting the same specifications are given below:

```
% Program Q5_16
% Design of Analog Lowpass Filter
clf;
Fp = 3500;Fs = 4500;
Wp = 2*pi*Fp; Ws = 2*pi*Fs;
[N, Wn] = cheb2ord(Wp, Ws, 0.5, 30,'s');
[b,a] = cheby2(N, 30, Wn, 's');
wa = 0:(3*Ws)/511:3*Ws;
h = freqs(b,a,wa);
plot(wa/(2*pi), 20*log10(abs(h)));grid
xlabel('Frequency, Hz');ylabel('Gain, dB');
title('Gain response');
axis([0 3*Fs -60 5]);
N
Wn
```

The gain response obtained by running the modified Program P5_4 is shown below:



Based on this plot we conclude that the filter designed _MEETS_ the given specifications.

The filter order N is - 8

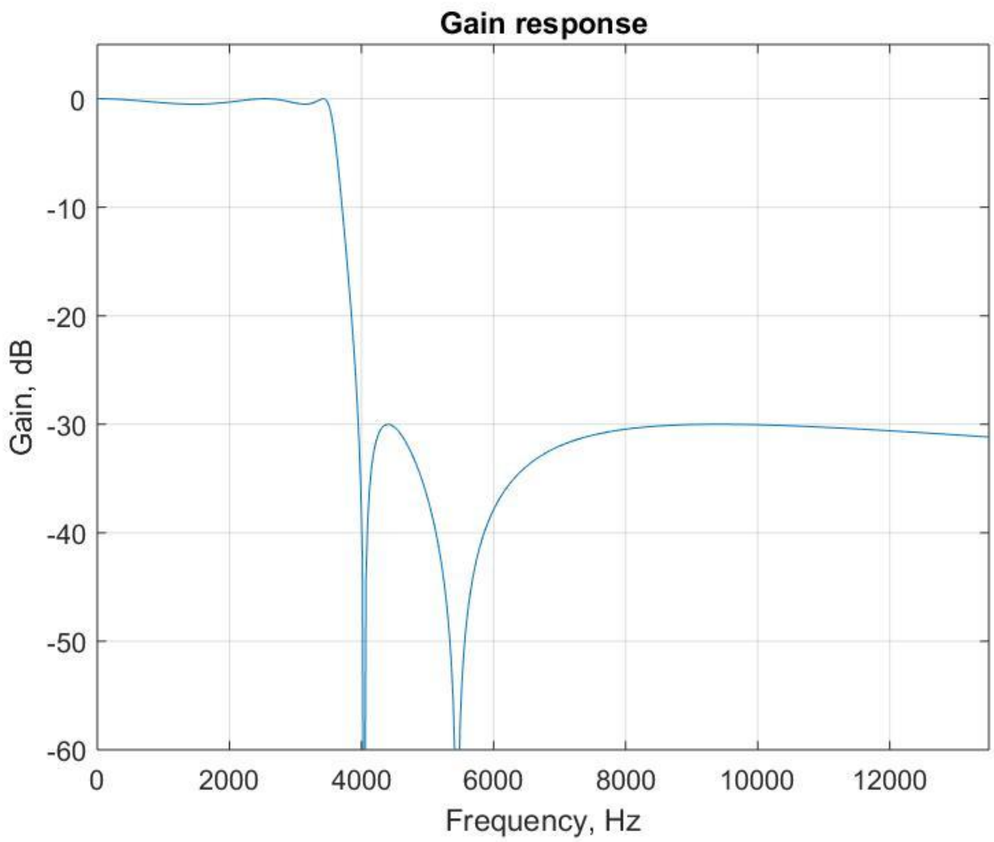
The stopband edge frequency in Hz of the filter is – 4.2653kHz

Q5.17 The required modifications to Program P5_4 to design an elliptic lowpass filter meeting the same specifications are given below:

```
% Program Q5_17
% Design of Analog Lowpass Filter
clf;
Fp = 3500;Fs = 4500;
Wp = 2*pi*Fp; Ws = 2*pi*Fs;
[N, Wn] = ellipord(Wp, Ws, 0.5, 30,'s');
[b,a] = ellip(N, 0.5, 30, Wn, 's');
wa = 0:(3*Ws)/511:3*Ws;
h = freqs(b,a,wa);
plot(wa/(2*pi), 20*log10(abs(h)));grid
xlabel('Frequency, Hz');ylabel('Gain, dB');
title('Gain response');
```

```
axis([0 3*Fs -60 5]);
N
Wn/(2*pi)
```

The gain response obtained by running the modified Program P5_4 is shown below:



Based on this plot we conclude that the filter designed __MEETS_____ the given specifications.

The filter order N is - 5

The passband edge frequency in Hz of the filter is – 3.5kHz

5.4 A/D AND D/A CONVERSIONS

Project 5.5 Binary Equivalent of a Decimal Number

Answers:

Q5.18 The function of the operator == is -

Q5.19 The binary equivalents in sign-magnitude form of the decimal fractions are:

(a)

(b)

(c)

(d)

Project 5.6 Decimal Equivalent of a Binary Number

Answer:

Q5.20 The decimal equivalents of the binary fractions along with the errors in conversion are as follows:

(a)

(b)

(c)

(d)

Project 5.7 Binary Number Representation Schemes

Answers:

Q5.21 The function of the operator `~=` is -

Q5.22 The ones'-complement representations of the binary numbers developed in Question Q5.19 are as follows:

(a)

(b)

(c)

(d)

Q5.23 The function of the operator `|` is –

The function of the operator `&` is –

Q5.24 The two's-complement representations of the binary numbers developed in Question Q5.19 are as follows:

(a)

(b)

(c)

(d)

Project 5.8 D/A Converter Droop Compensation

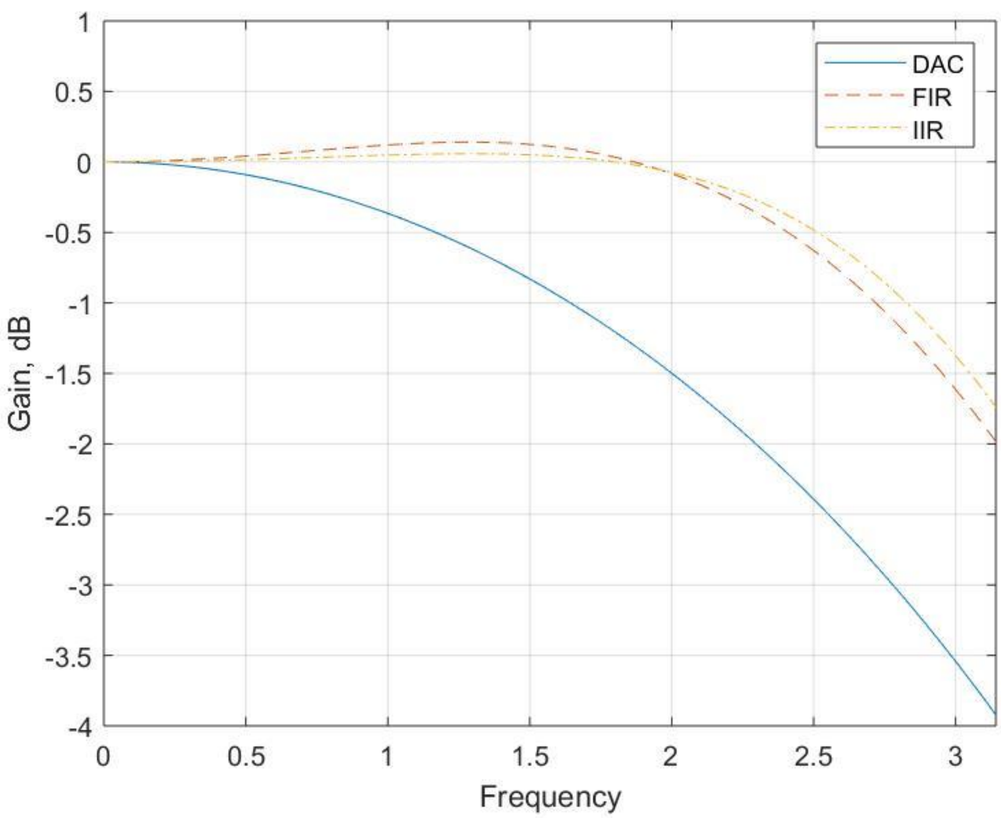
Answer:

Q5.25 The MATLAB program to determine and plot the magnitude responses of the uncompensated and the droop-compensated D/A converters is given below:

```
% Program Q5_25.m
% For a non-ideal sample-and-hold D/A converter,
% compute the magnitude response of
% 1. the uncompensated D/A
% 2. D/A with FIR droop compensation
% 3. D/A with IIR droop compensation
% all on a single graph.
T = 1.0; % sampling interval
wa = 0:(pi/T)/511:(pi/T); % analog freq vector
Lwa = length(wa);
Hra = T * ones(1,length(wa)); % ideal
reconstruction filter
% Frequency response of uncompensated zero-order
hold
Hza = zeros(1,Lwa);
Hza(1) = 1; % follows from L'Hopital's Rule
wa2 = wa(2:Lwa);
Hza(2:Lwa) = exp(-i*wa2*T*0.5) .* sin(0.5*T*wa2)
./ (0.5*T*wa2);
MagHza = abs(Hza);
GainHza = 20*log10(MagHza);
% Frequency response of FIR compensated D/A
Hfira = (1/16)*(-1 + 18*exp(-i*wa*T) - exp(-
i*2*T*wa));
HfiraComp = Hfira .* Hza;
MagHfiraComp = abs(HfiraComp);
GainHfiraComp = 20*log10(MagHfiraComp);
% Frequency response of IIR compensated D/A
Hiira = 9*ones(1,Lwa) ./ (8 + exp(-i*wa*T));
HiiraComp = Hiira .* Hza;
MagHiiraComp = abs(HiiraComp);
GainHiiraComp = 20*log10(MagHiiraComp);
% plot
```

```
plot(wa,GainHza,wa,GainHfiraComp,'--
',wa,GainHiiraComp,'-.');
grid;
xlabel('Frequency');
ylabel('Gain, dB');
axis([0 pi/T -4 1]);
legend('DAC','FIR','IIR');
```

The plot of the magnitude responses is shown below:



From this plot we make the following observations: Also, the IIR has a spectral magnitude that is way more flat and drops less that the FIR compensated DACS. The Drop of the uncompensated DAC is improved by both the FIR and the IIR compensated DACS.

Date: 11/26/2018 **Signature**: Jonathan Ifegunni