ELG4177 - DIGITAL SIGNAL PROCESSING
Lab3

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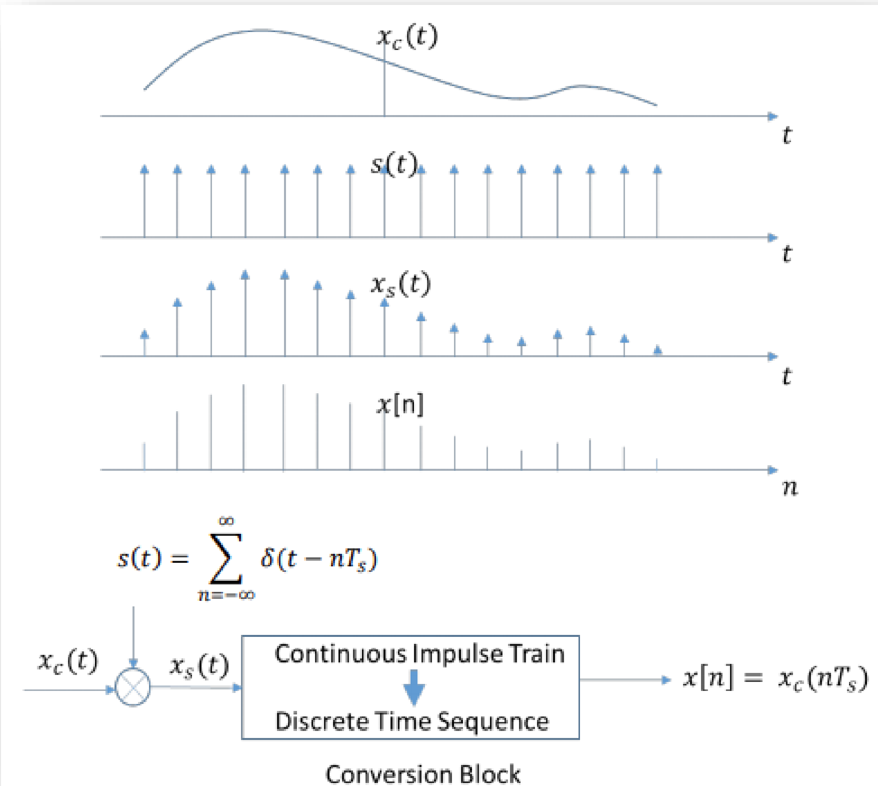
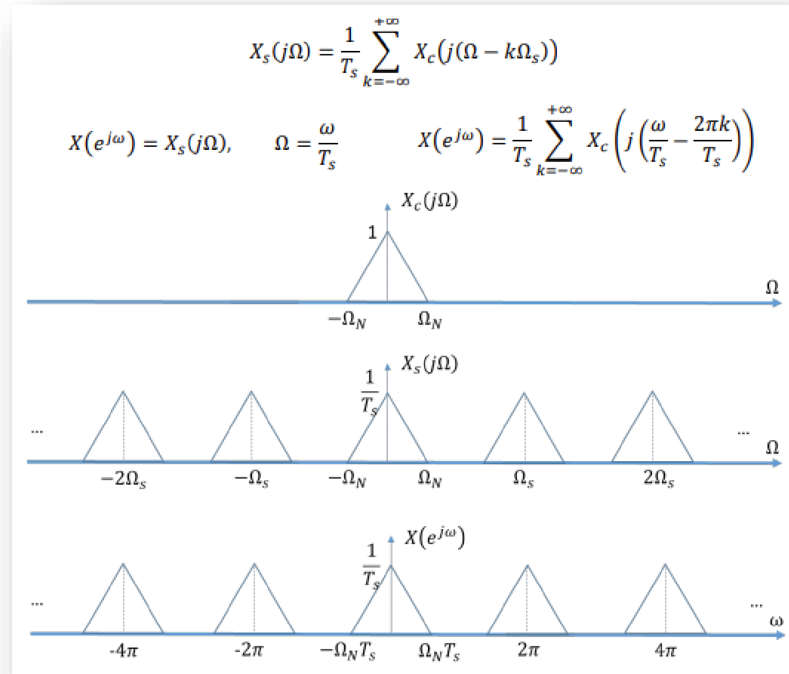
Assignment 03

SAMPLING, A/D CONVERSION AND D/A CONVERSION

Sampling

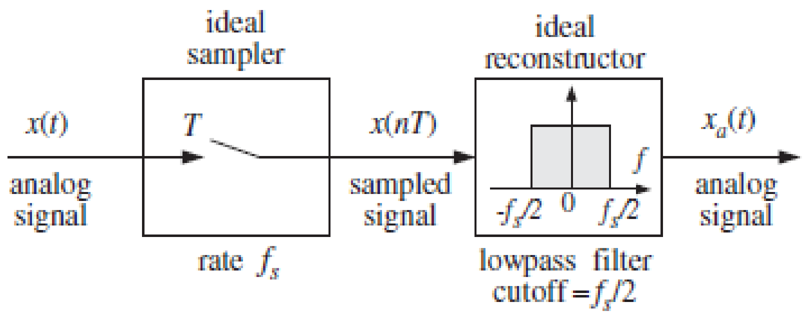




where $\Omega_s = \frac{2\pi}{T_s}$ is the sampling frequency.

(1) Time domain representation of sampling and aliasing

Sampling Process

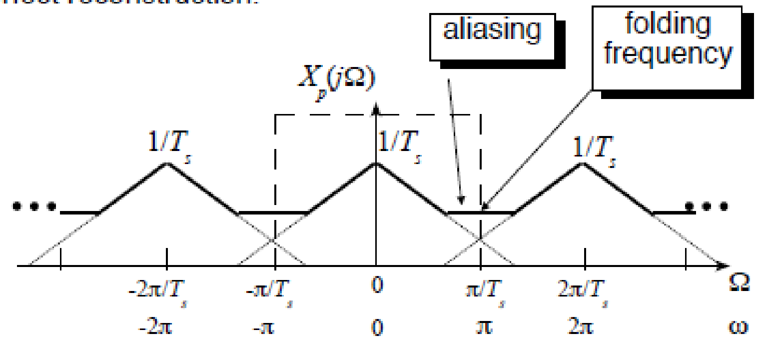


Among the frequencies, there is a unique one that lies within the Nyquist interval. It is obtained by reducing the original f modulo-fs, that is, adding to or subtracting from f enough multiples of fs until it lies within the symmetric Nyquist interval [−fs/2, fs/2].

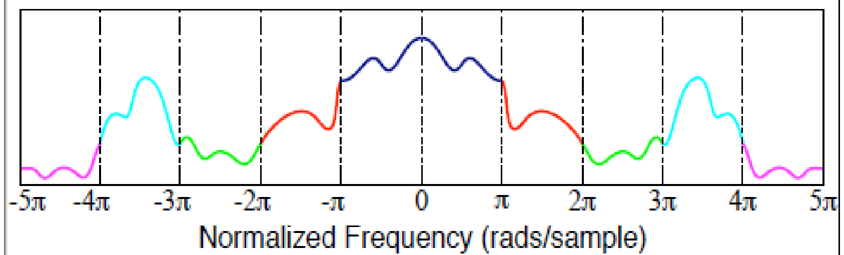
Aliasing

Aliasing occurs when the 2π periodic extensions of the bandlimited $x(t)$ overlap at $\Omega=\pi/T_s$, or similarly $\omega=\pi$.

It is not possible to "separate" the overlapping bandwidth for perfect reconstruction.

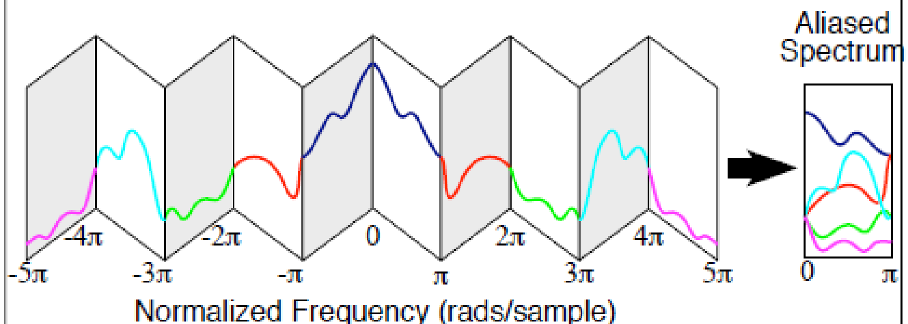


Consider the following magnitude spectrum of a signal.
We can visualize *aliasing* using the "Fan Folding" method.



Note: Each symmetric 2π range is shown in a different colour.

For the entire example spectrum, the aliasing is as follows.



Fa=f mod(fs)

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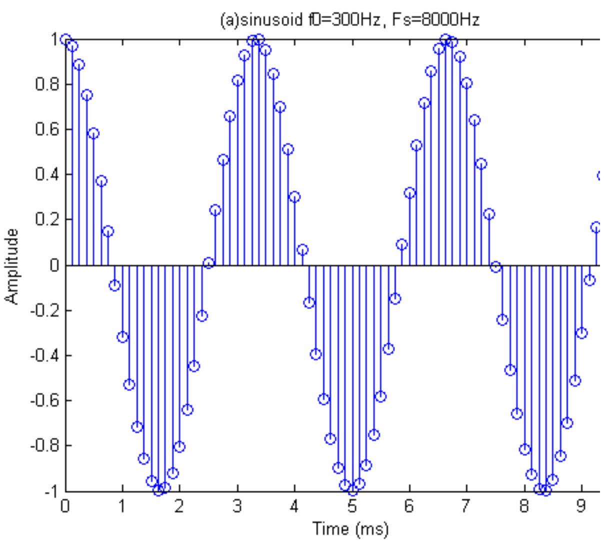
Sinusoid Signal

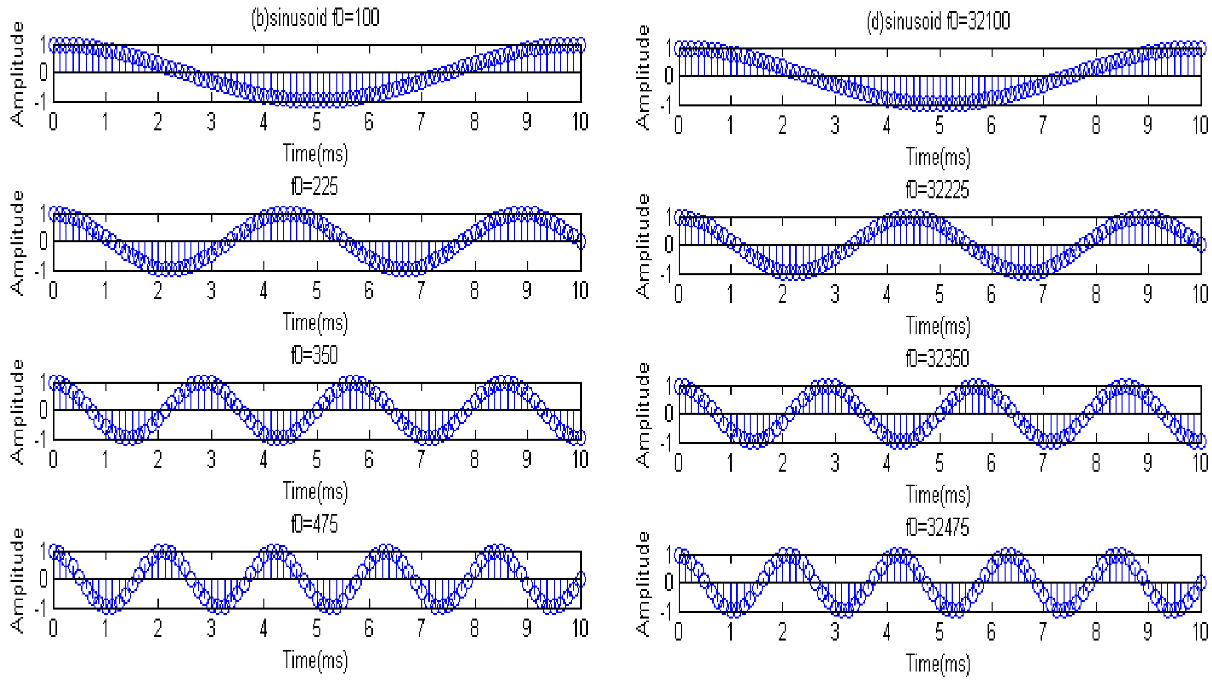
$$x[n] = \sin(2\pi \frac{f_0}{f_s} n + \phi)$$

f_s = 8 kHz

By varying the value f0

When f0=300 Hz





Can you predict in advance if the frequency will increase or decrease ? Why/How ?

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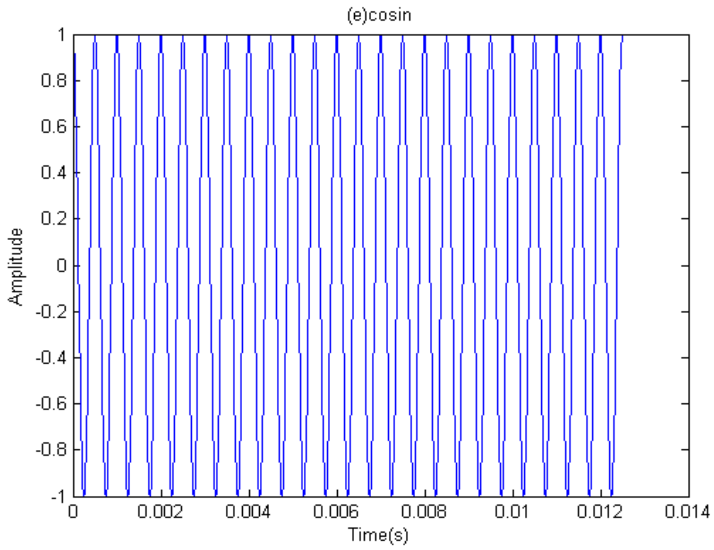
(2) Frequency domain representation of sampling and aliasing, A/D and D/A conversions

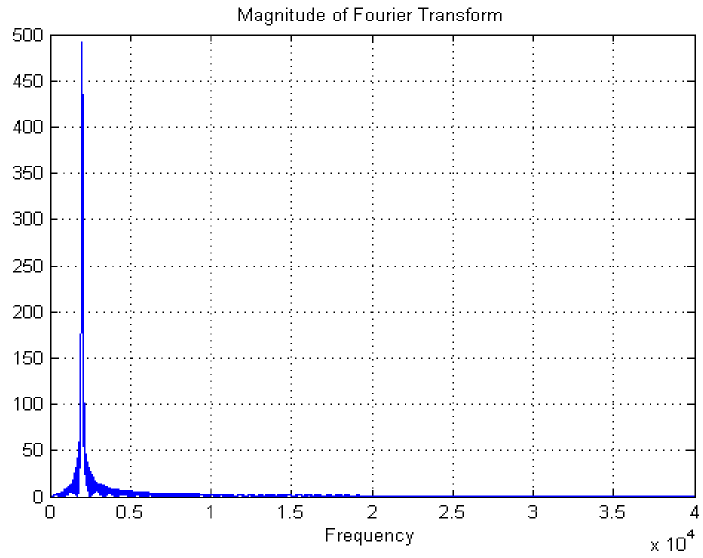
$$x(t) = \cos(2\pi f_0 t) \approx \cos(2\pi f_0\, n\, \Delta t)$$

n=0,1,2,... f_0 =2kHz

Δt =1/80000

```
function freqmagplot(x,dt)
  L=length(x);
  Nfft=round(2.^round(log2(5*L)));
  X=fft(x,Nfft);
  f=((1/dt)/Nfft)*(0:1:Nfft/2-1);
  plot(f,abs(X(1:Nfft/2)));
  title('Magnitude of Fourier Transform');
  xlabel('Frequency'),grid;
end
```






```
dt = 1/80000; n = 1:1000; f0 = 2000;
x = cos(2*pi*f0*n*dt);
```

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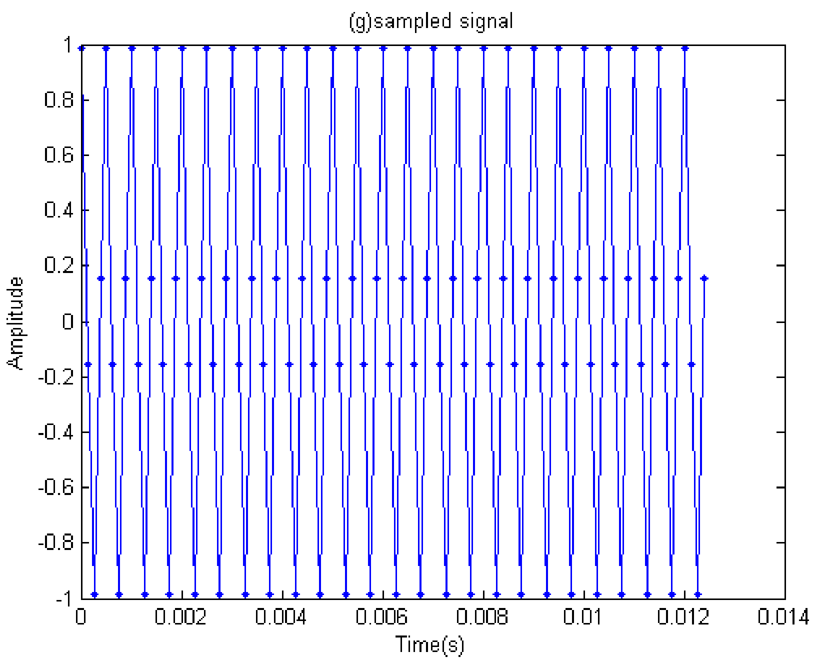
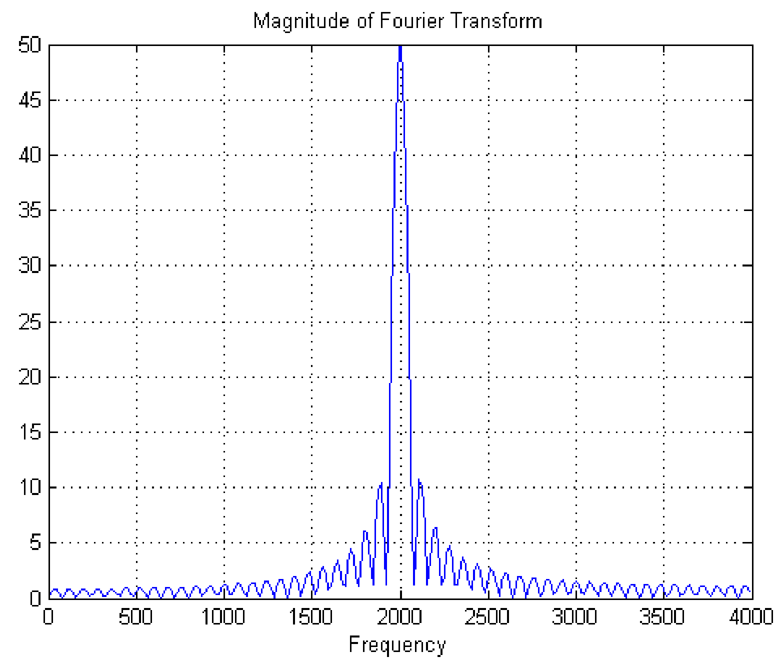
Simulate the A/D conversion

- we need to keep one sample in every 10 samples.





Plot the resulting discrete time signal and its discrete time Fourier transform (*freqmagplot* can be used, but with the appropriate value for dt !).

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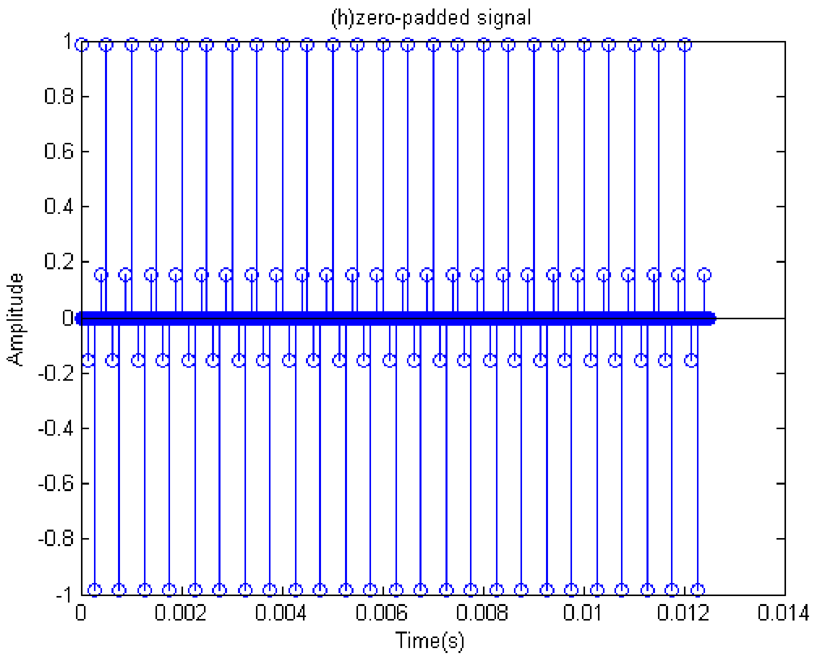
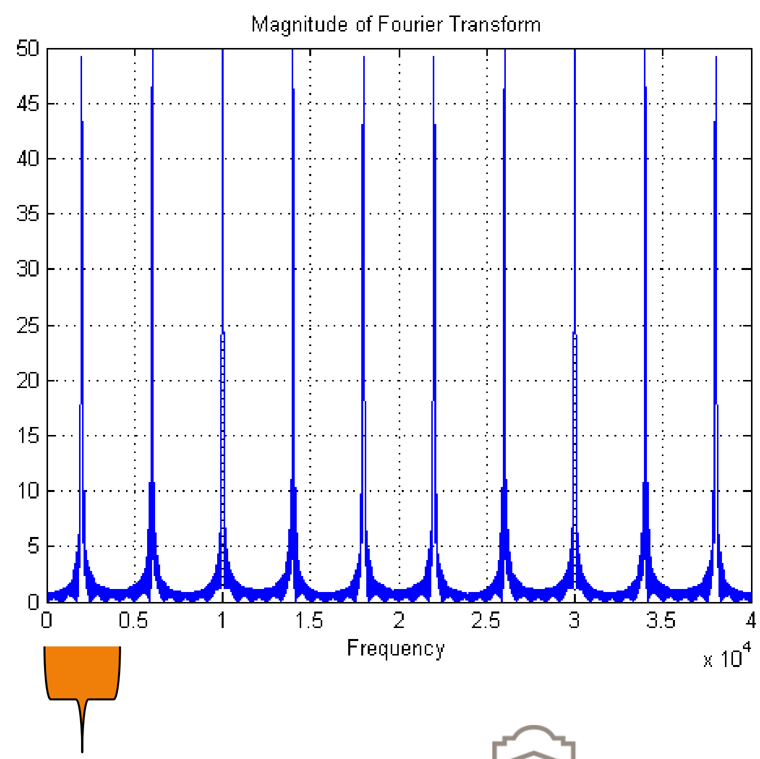
Simulate the D/A conversion

Two Steps: (1) convert it into an analog pulse signal. To simulate the analog pulse signal, 9 zeros are added between each sample of the discrete time signal. (2). filter (interpolate) the signal





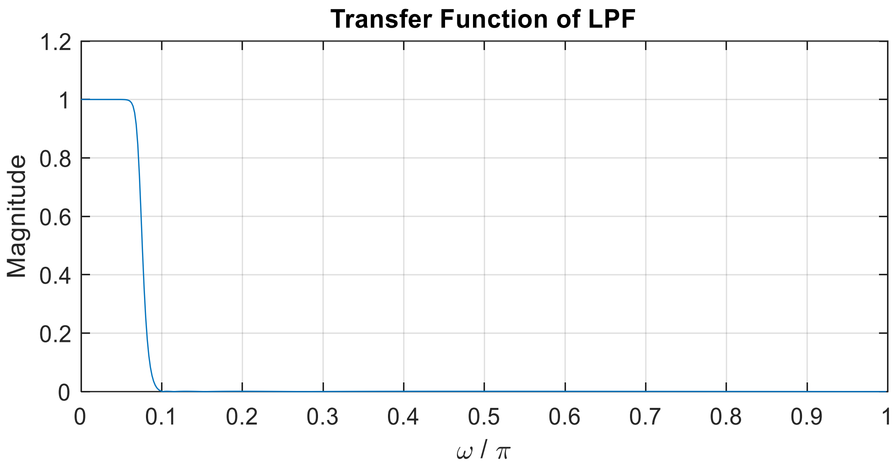
Filter this part

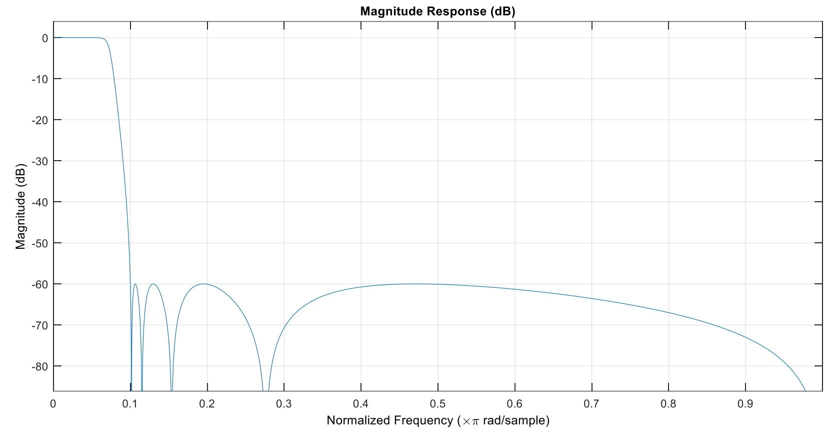
Filtering

```
[b, a] = cheby2(9, 60, fs*1/80000);
```

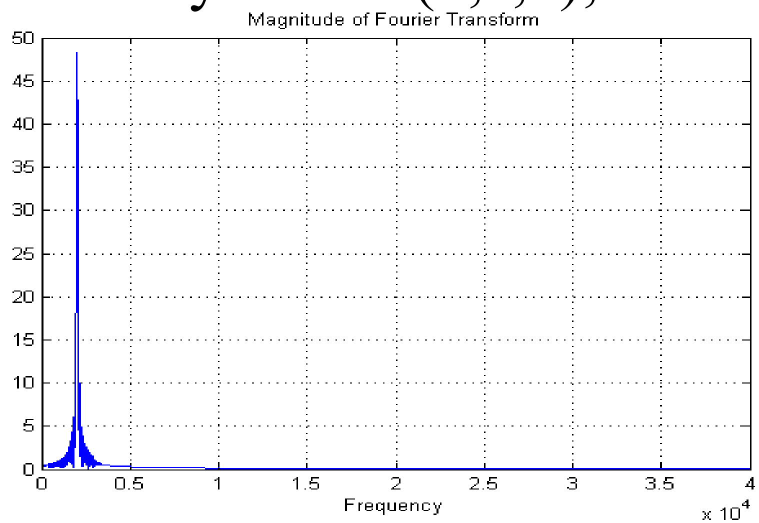
```
[H,w]=freqz(b,a);   plot(w/pi,abs(H));
```

```
fvtool(b,a)
```






```
y = filter(b,a,x);
```



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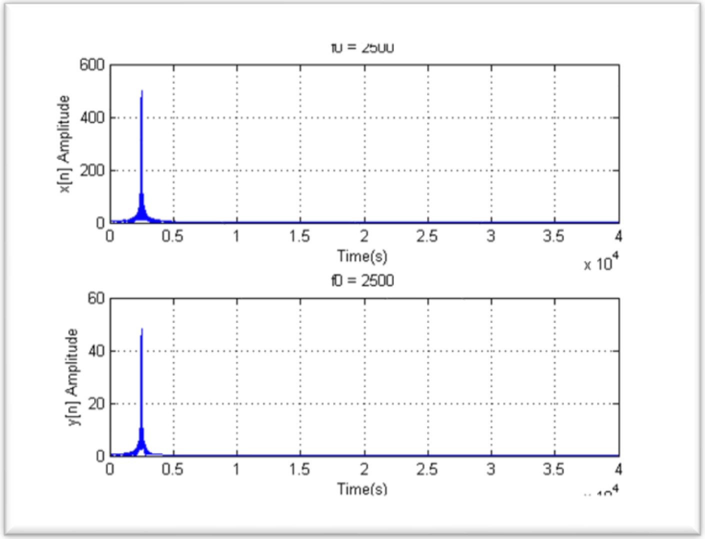
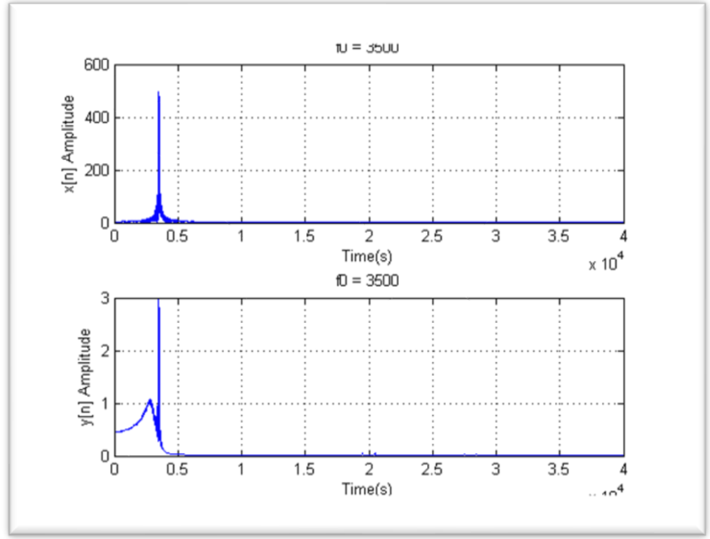
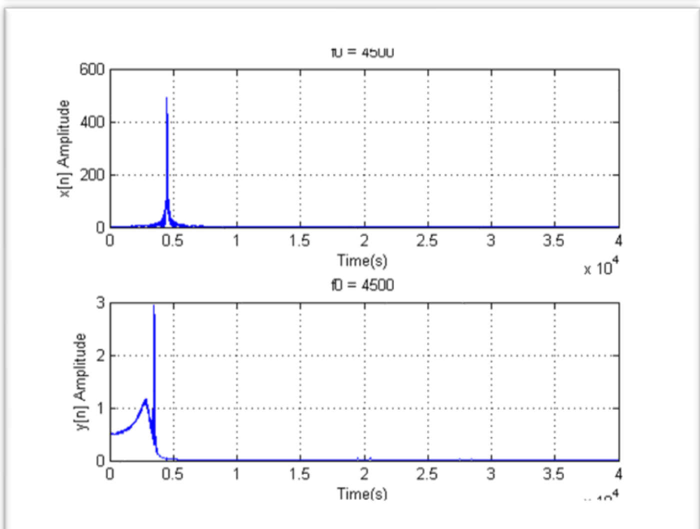
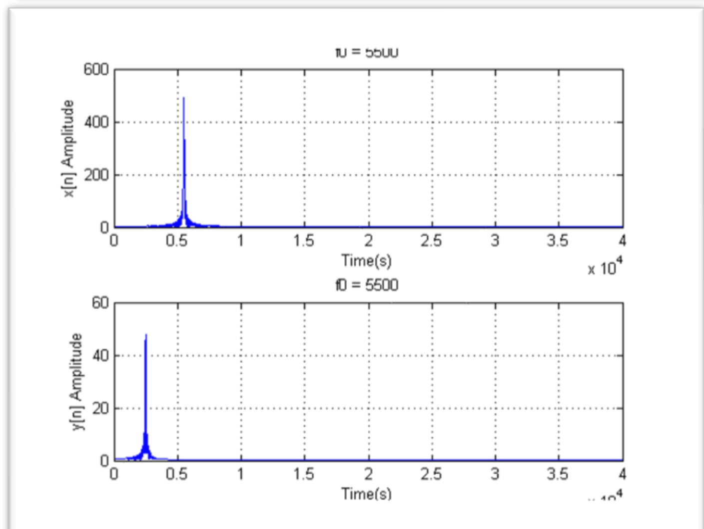
The effect of aliasing









When does aliasing occur? What is the effect of aliasing on the output signal of the D/A converter (found in i)) ?

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Complete all the assignment. Submit it the Brightspace

Thanks

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