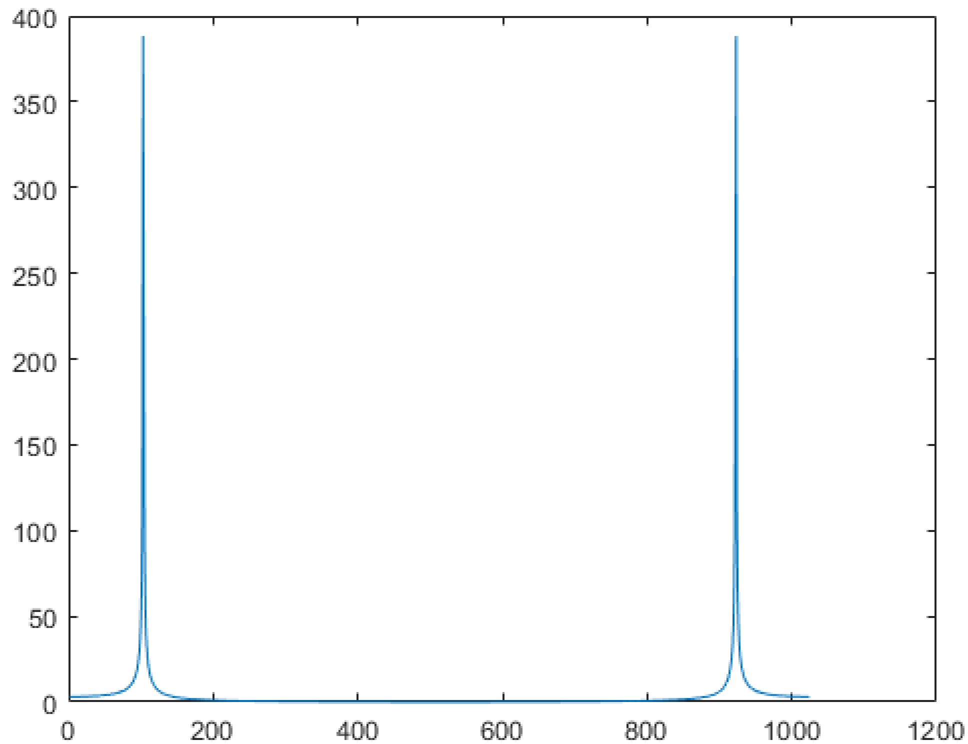
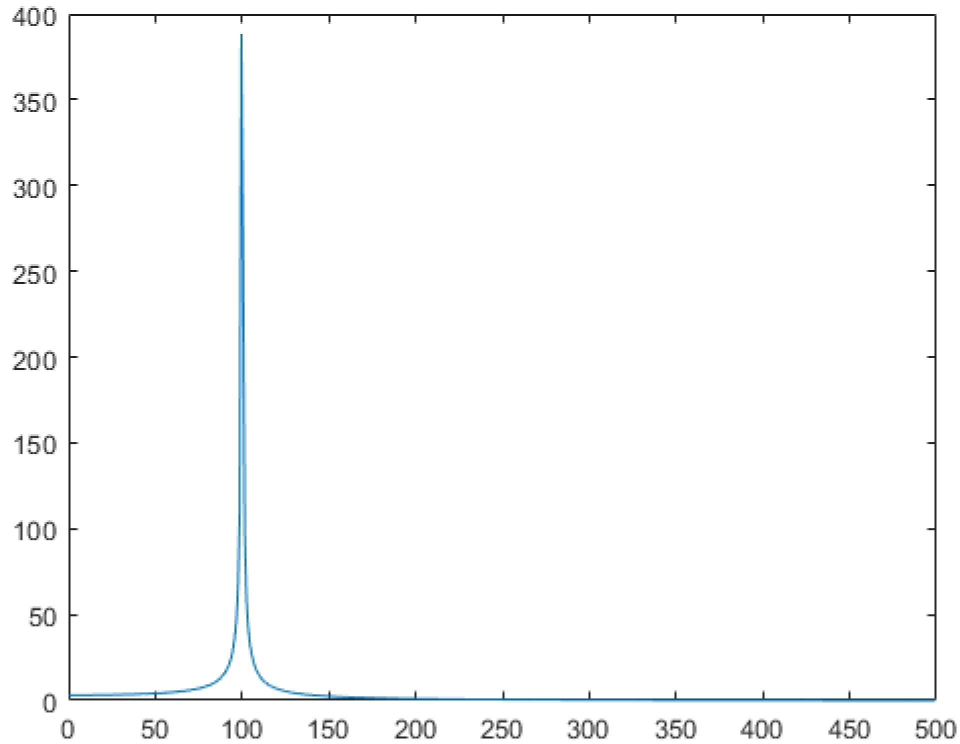
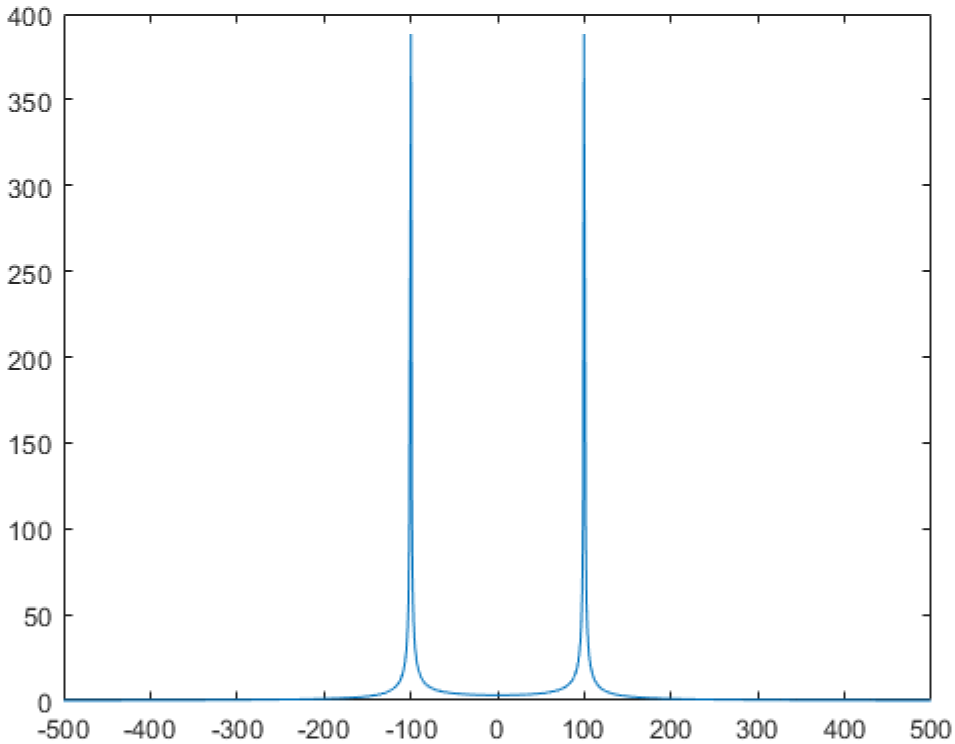
```
%%Listing 7.1. specsin0.m naive and deceptive spectrum of a sine wave
 via the FFT
f =100; Ts=1/1000; time=5.0;    % freq, sampling interval, time
t=Ts : Ts : time ;             % define a time vector
w=sin (2* pi* f *t ) ;         % define the sinusoid
N=2^10;                        % size of analysis window
fw=abs(fft(w(1:N)));           % find magnitude of DFT/FFT
plot(fw)                       % plot the waveform
```




```
%%Listing 7.2. specsin1.m spectrum of a sine wave via the FFT/DFT
f =100; Ts=1/1000; time =5.0; % freq , sampling interval, time
t=Ts:Ts:time ; % define a time vector
w=sin(2*pi*f*t ) ; % define the s inuso id
N=2^10; % size of analysis window
ssf =(0:N/2-1)/(Ts*N) ; % frequency vector
fw=abs(fft(w(1:N))) ; % find magnitude of DFT/FFT
plot(ssf, fw(1:N/2)) % plot for positive freq only
```

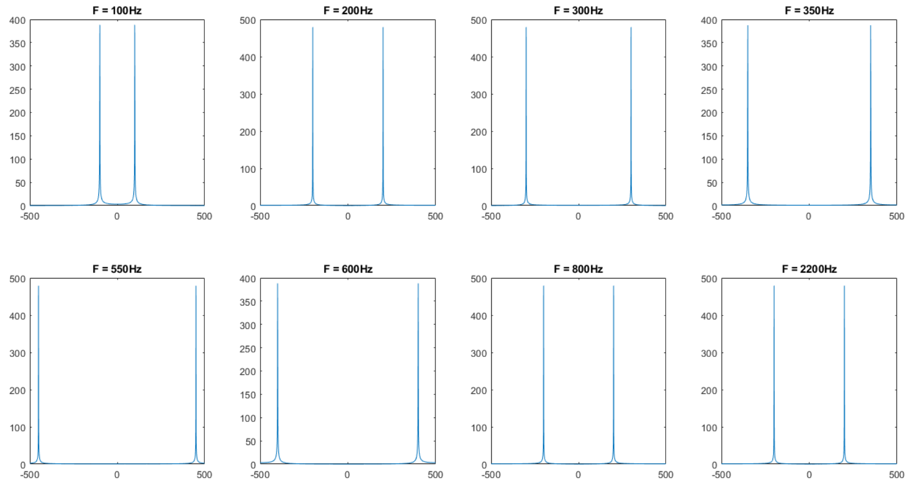



```
%%Listing 7.3. specsin2.m spectrum of a sine wave via the FFT/DFT
f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
t=Ts : Ts : time ; % define a time vector
w=sin (2* pi* f *t ) ; % define the sinusoid
N=2^10; % size of analyis window
ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
fw=fft(w( 1 :N) ) ; % do DFT/FFT
fws=fftshift( fw ) ; % shiftit for plotting
plot ( ssf, abs( fws )) % plot magnitude spectrum
```



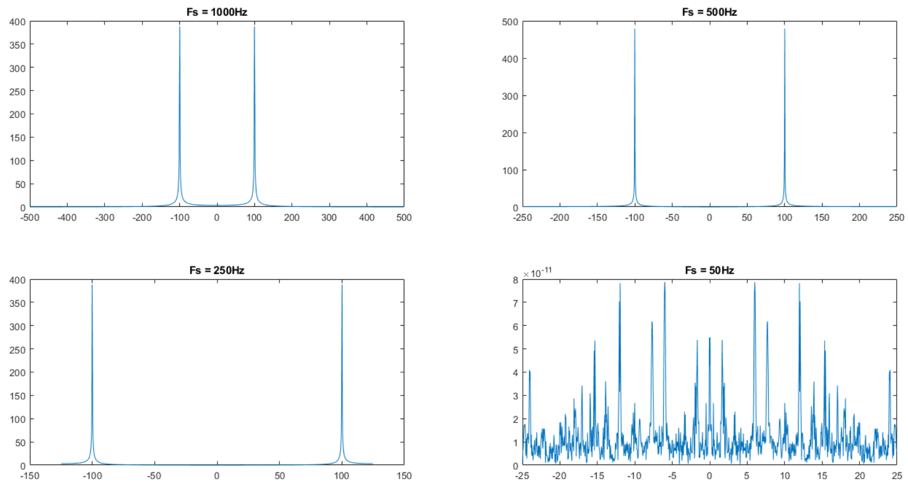

```
%%Exercise 7.5.
% Explore the limits of the FFT/DFT technique by choosing
% extreme values. What happens in the following cases?
% a. f becomes too large. Try f = 200, 300, 450, 550, 600, 800, 2200
 Hz. Comment
% on the relationship between f and Ts.
freqs = [100, 200,300,350,550,600,800,2200];
figure(1)
for i = 1:8
    subplot(2,4,i)
    f=freqs(i); Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin (2* pi* f *t ) ; % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plotting
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("F = %iHz",f))
end

% We can observe how the correct frequency is no longer identified in
 the
% plot after the sinusoid's frequency passes half of the sampling
 frequency
```

```
% (1000 Hz.) In these cases, aliasing has occured, and the frequency
 peak has
% "wrapped around," causing the frequency to be shown as 1000Hz - f.
```



b. Ts becomes too large. Try Ts = 1/500, 1/250, 1/50. Comment on the relationship between f and Ts. (You may have to increase time in order to have enough samples to operate on.)

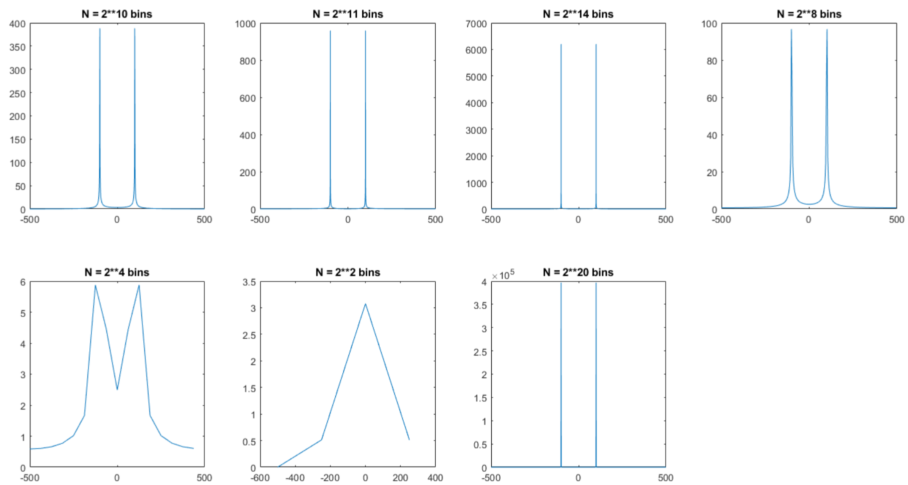
```
Ts_s = [1/1000, 1/500, 1/250, 1/50];
figure(1)
for i = 1:4
    subplot(2,2,i)
    f=100; Ts=Ts_s(i); time=100.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin (2* pi* f *t ) ; % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plotting
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("Fs = %iHz",1/Ts))
end

% This case is very similar to what happens when the frequency is
 increased
% past the Nyquist sampling rate of half the sampling frequency. In
 the
% last case, the signal's frequency is a multiple of the sampling
% frequency, so the signal is sampled at the same place every cycle,
% resulting in a sampled signal with no frequency componants. This
 explains
% why the only information the frequency spectrum plot has to show is
 very
% low amplitude noise.
```



c. N becomes too large or too small. What happens to the location in the peak of the magnitude spectrum when N = 2^11, 2^14, 2^8, 2^4, 2^2, 2^20? What happens to the width of the peak in each of these cases? (You may have to increase time in order to have enough samples to operate on.)

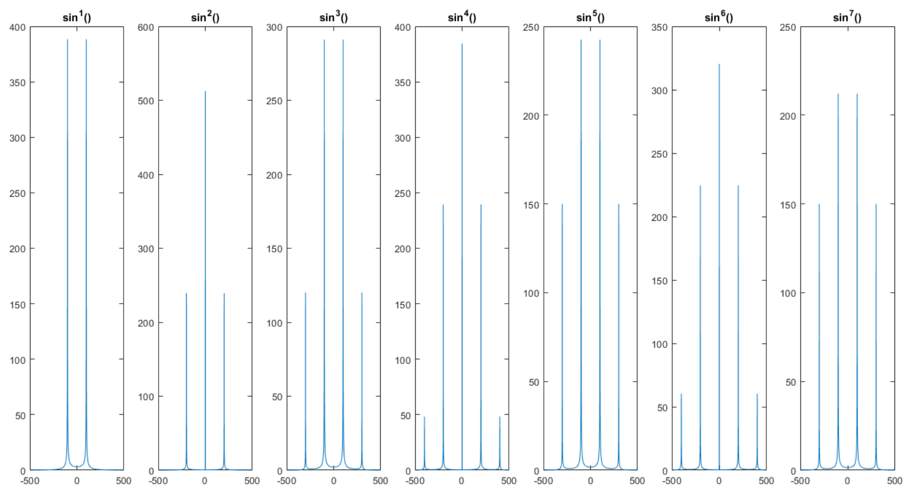
```
Ns = [2^10, 2^11, 2^14, 2^8, 2^4, 2^2, 2^20];
figure(2)
for i = 1:7
    subplot(2,4,i)
    f=100; Ts=1/1000; time=2000.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin (2* pi* f *t ) ; % define the sinusoid
    N=Ns(i); % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plotting
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("N = 2**%i bins",log(N)/log(2)))
end

%When the number of frequency bins N used in the transform is too
 small,
%the peaks get very wide and have a much smaller magnitude. This is
 because
%the frequency spectrum outputed by the discrete fourier transform is
%forced to give the average amplitude of all the signal's frequency
%componants represented by the range around each bin, and so if the
 bin is
%is wider, there will be more frequencies between successive bins
 which
%results in a wider peak, and also the magnitude of the peak will be
%reduced by the extra zero frequency components also lumped into the
 same
%bin, driving down the average.

% When the number of bins is very high, the peak's magnitude gets very
```

```
% large. I think this is because the closer the bin's width gets to
 being
% infinitely small (a continuous instead of discrete transform) the
 closer
% the frequency response gets to showing an impulse at the frequency
 which
% the sinusoid occupies. One problem with having such a large number
 of
% bins is that it requires a much longer time to sample the signal in
 order
% to operate, which may be impractical for a real-time communication
% system.
%
```



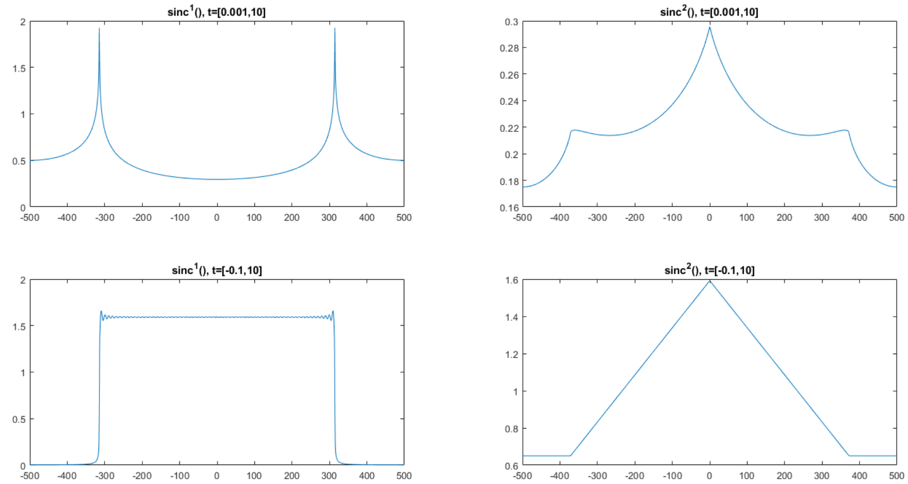

```
%%Exercise 7.6. Replace the sin function with sin^2. Use
 w=sin(2*pi*f*t).^2. What
% is the spectrum of sin^2? What is the spectrum of sin3? Consider
 sink.What is the
% largest k for which the results make sense? Explain what limitations
 there are.
figure(3)
freq_pows = [1,2,3,4,5,6,7];
for i = 1:7
    subplot(1,7,i),
    f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin(2*pi*f*t).^freq_pows(i); % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plottingplot ( ssf, abs( fws ))
 % plot magnitude spectrum
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("sin^%i()",i))
end
```

```
%in my view, the easiest way to explain the frequency spectra of
 sin^k()
%is that each multiplication by sin() is the same as convoluting the
%function's frequency response with the twin impluses of the frequency
%response of sin(). The first time this happens (for sin^2()) it has
 the
%effect of offsetting the frequency spectrum's peaks at -100 and +100
 Hz
%another 100Hz away from the origin where only one peak from each
 signal
%overlaps, as well as adding DC frequency component where the two
 identical
%signals perfectly match.
%
%Another way of looking at it is that every succesive multiplicate by
 sin()
%adds another frequency component 100Hz (or whatever the initial
 sinusoid's
%frequency was) higher than the last signal and alternates whether or
 not
%the function is even or odd. Because the frequency spectrum is always
%increasing by F, this process eventually breaks down when aliasing
 starts
%occuring, although in this case because sinusoid's frequency is a
 multiple
%of the sampling frequency, the higher frequency componants overlap
 with
%their lower harmonics.
```

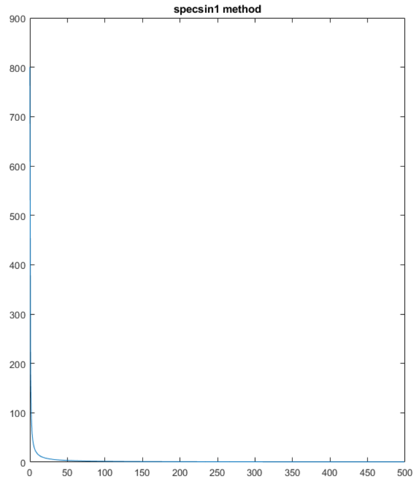


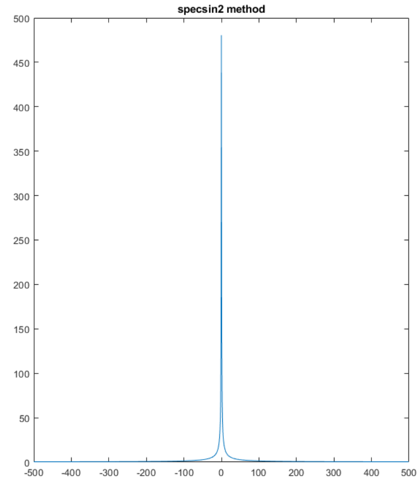

```
%%Exercise 7.7. Replace the sin function with sinc. What is the
 spectrum of the
% sinc function? What is the spectrum of sinc2?
figure(4)
freq_pows = [1,2];
for i = 1:2
    subplot(2,2,i),
```

```
    f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=(Ts: Ts : time); % define a time vector
    w_s=sinc(2*pi*f*t).^freq_pows(i); % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w_s( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shift it for plottingplot ( ssf,
 abs( fws )) % plot magnitude spectrum
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("sinc^%i(), t=[0.001,10]",i))
end
freq_pows = [1,2];
for i = 1:2
    subplot(2,2,i+2),
    f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=(-0.1: Ts : time); % define a time vector
    w_s=sinc(2*pi*f*t).^freq_pows(i); % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w_s( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shift it for plottingplot ( ssf,
 abs( fws )) % plot magnitude spectrum
    plot ( ssf, abs( fws )) % plot magnitude spectrum
    title(sprintf("sinc^%i(), t=[-0.1,10]",i))
end
%Mathematically, fourier transform of the sinc function ought to be a
%rect() function, and the transform of the sinc^2 function ought to be
 a
%triangle function. However, when I plot these functions the way the
 book's
%example code plots the sin function, I get severely distorted
 versions of
%those shapes in the frequency domain, which does not seem to be
 caused by
%the sampling rate or time over which the function is evaluated. The
 only
%way I could find to make the frequency domain plot look the right way
 was
%to start sampling the function slightly before t=0, as shown in the
 second
%row of plots
```




```
%%Exercise 7.8. Plot the spectrum of w(t) = sin(t) + je^?t. Should you
 use the
% technique of specsin1.m or of specsin2.m? Hint: think symmetry.
figure(5)
subplot(1,2,2)
f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
t=Ts : Ts : time ; % define a time vector
w=sin(t) ; % define the sinusoid
N=2^10; % size of analyis window
ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
fw=fft(w( 1 :N) ) ; % do DFT/FFT
fws=fftshift( fw ) ; % shiftit for plotting
plot ( ssf, abs( fws )) % plot magnitude spectrum
title("specsin2 method")

subplot(1,2,1)

f =100; Ts=1/1000; time =5.0; % freq , sampling interval, time
t=Ts:Ts:time ; % define a time vector
w=sin(t)+j*exp(-t) ; % define the s inuso id
N=2^10; % size of analysis window
ssf =(0:N/2-1)/(Ts*N) ; % frequency vector
fw=abs(fft(w(1:N))) ; % find magnitude of DFT/FFT
plot(ssf, fw(1:N/2)) % plot for positive freq only
title("specsin1 method")
% it seems to me that because of the imaginary component of the input
% signal, its frequency spectrum is not symmetrical about 0Hz, and
 thus it
% makes more sense to plot the spectrum using the technique shown in
% specsin2.m, since it shows both the positive and negative frequency
% components.
```

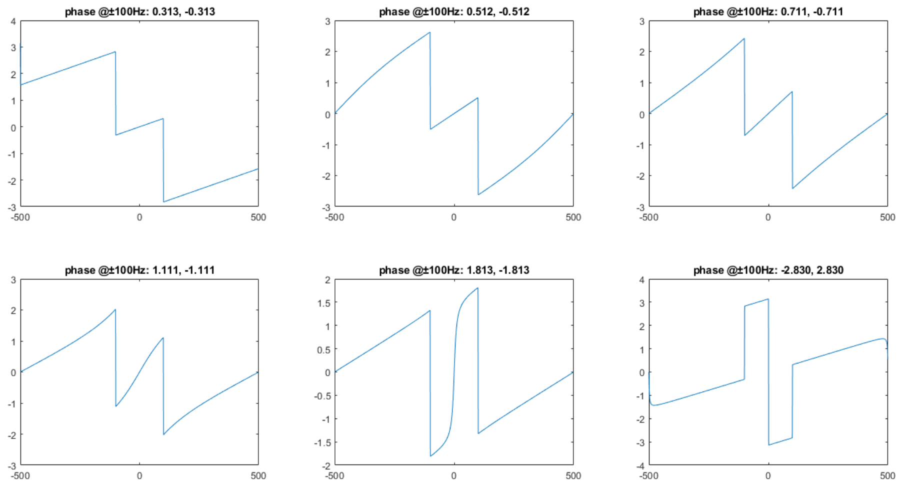





```
%%Exercise 7.9. The FFT of a real sequence is typically complex, and
 sometimes
% it is important to look at the phase (as well as the magnitude).
% a. Let w=sin(2*pi*f*t+phi). For phi = 0, 0.2, 0.4, 0.8, 1.5, 3.14,
 find the phase of
% the FFT output at the frequencies ±f.
figure(6)
phis = [0, 0.2, 0.4, 0.8, 1.5, 3.14];
for i = 1:length(phis)
    subplot(2,3,i)
    phi = phis(i);
    f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin(2*pi*f*t+phi) ; % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plotting
    plot ( ssf, angle( fws ))
    idxs = [find(abs(ssf+f) == min(abs(ssf+f))), find(abs(ssf-f) ==
 min(abs(ssf-f)))]
    angles = angle(fws(idxs));
    title(sprintf("phase @±100Hz: %0.3f, %0.3f",angles(2),angles(1)))
end
```

```
idxs =

   411   615


idxs =

   411   615
```

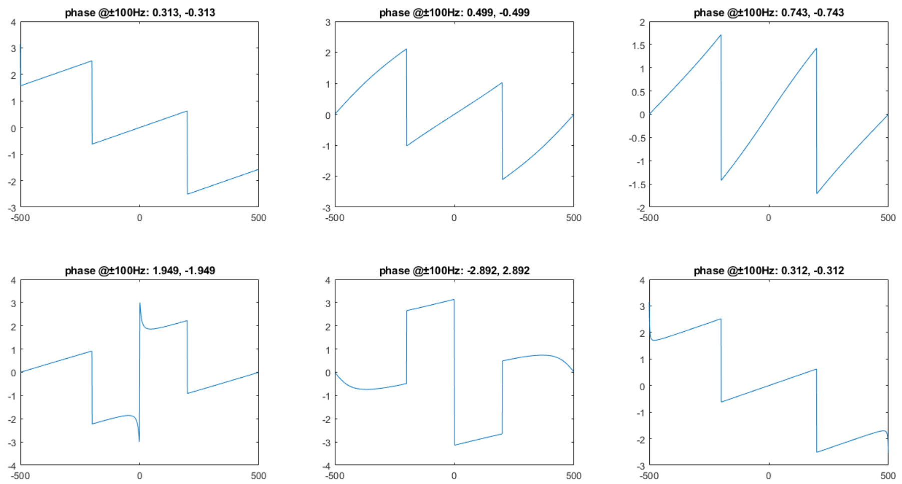
```
idxs =

   411   615


idxs =

   411   615


idxs =

   411   615


idxs =

   411   615
```



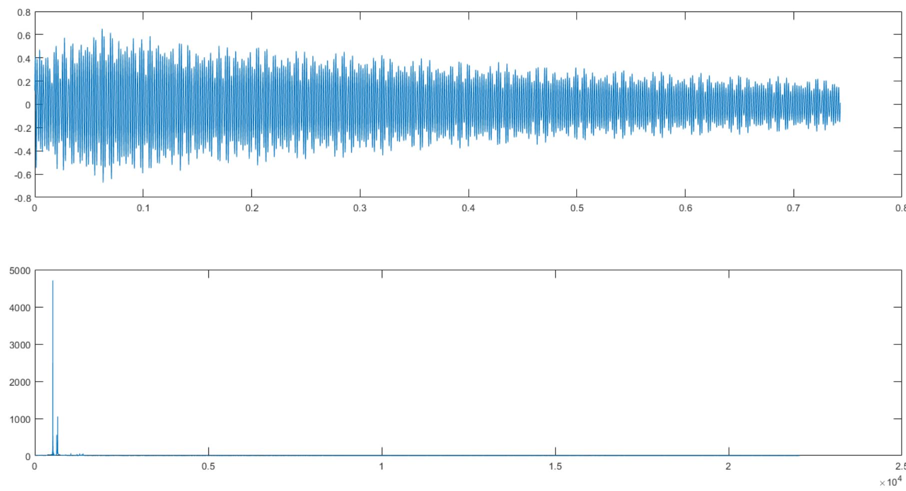

```
%%b. Find the phase of the output of the FFT when w=sin(2*pi*f*t
+phi).^2.
figure(7)
phis = [0, 0.2, 0.4, 0.8, 1.5, 3.14];
for i = 1:length(phis)
    subplot(2,3,i)
    phi = phis(i);
    f=100; Ts=1/1000; time=10.0; % freq , sampling interval, time
    t=Ts : Ts : time ; % define a time vector
    w=sin(2*pi*f*t+phi).^2 ; % define the sinusoid
    N=2^10; % size of analyis window
    ssf=(-N/2:N/2-1)/(Ts*N) ; % frequency vector
    fw=fft(w( 1 :N) ) ; % do DFT/FFT
    fws=fftshift( fw ) ; % shiftit for plotting
    plot ( ssf, angle( fws ))
```

```
    idxs = [find(abs(ssf+100) == min(abs(ssf+f))), find(abs(ssf-f) ==
 min(abs(ssf-f)))]
    angles = angle(fws(idxs));
    title(sprintf("phase @±100Hz: %0.3f, %0.3f",angles(2),angles(1)))
end
```

```
idxs =

   411   615


idxs =

   411   615


idxs =

   411   615


idxs =

   411   615


idxs =

   411   615


idxs =

   411   615
```

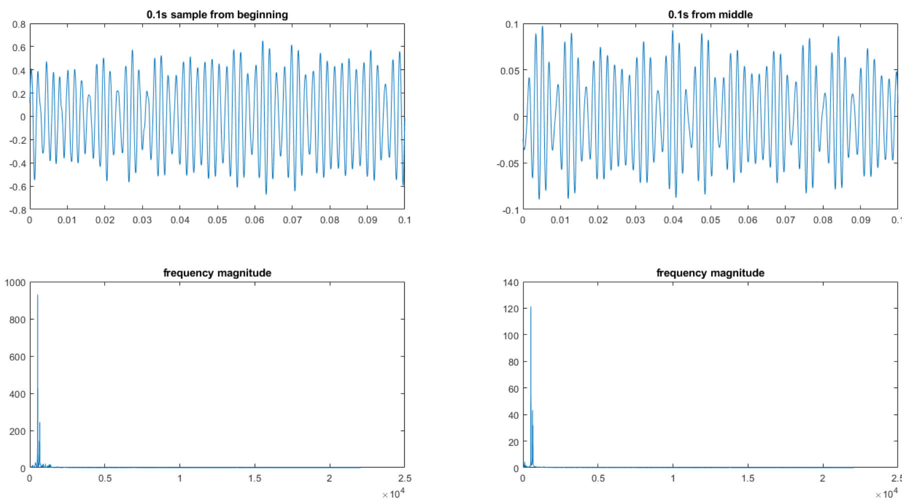


```
%%Listing 7.4. specgong.m find spectrum of the gong sound
filename= 'gong.wav' ; % name of wave file
[ x, sr ]= audioread( filename ) ; % read in wavefile
Ts=1/sr ; % sample interval & # of samples
N=2^15; x=x ( 1 :N)' ; % length for analysis
sound(x ,1/Ts ) % play sound ( if possible )
time=Ts * ( 0 : length (x) -1); % time base for plotting
subplot (2 , 1 , 1 ) , plot ( time , x) % and plot top figure
magx=abs ( fft(x ) ) ; % take FFT magnitude
ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
subplot(2,1,2), plot(ssf,magx(1:N/2)) % plot mag spectrum
```



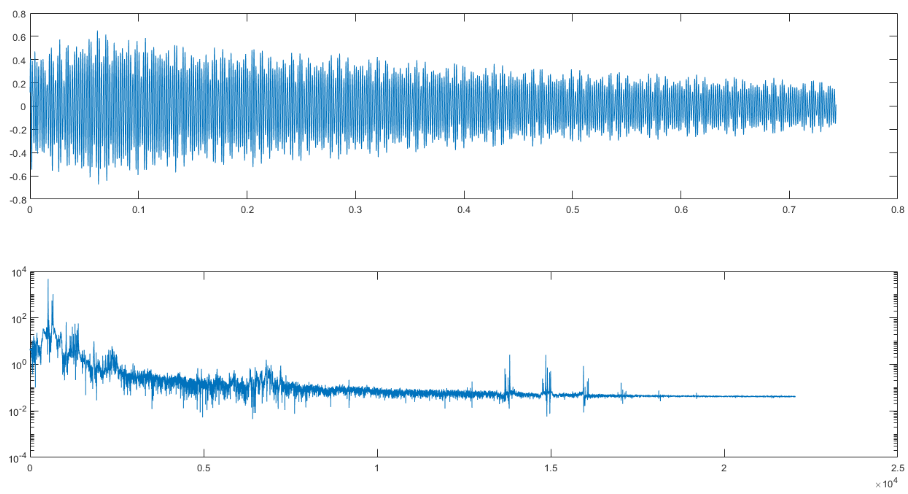

```
%%Exercise 7.10. Determine the spectrum of the gong sound during the
 first 0.1 s.
% What value of N is needed? Compare this with the spectrum of a 0.1 s
 segment
% chosen from the middle of the sound. How do they differ?
figure(8)
filename= 'gong.wav' ; % name of wave file
[ x, sr ]= audioread( filename ) ; % read in wavefile
Ts=1/sr ; % sample interval & # of samples
N=0.1/Ts; x=x ( 1 :N)' ; % length for analysis
sound(x ,1/Ts ) % play sound ( if possible )
time=Ts * ( 0 : length (x) -1); % time base for plotting
subplot (2 , 2 , 1 ) , plot ( time , x) % and plot top figure
title("0.1s sample from beginning")
magx=abs ( fft(x ) ) ; % take FFT magnitude
ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
subplot(2,2,3), plot(ssf,magx(1:N/2)) % plot mag spectrum
title("frequency magnitude")
%The number of samples N required to analyze a certain time amount of
 the
%sound is dependant on the sampling frequency, which is 1/Ts. Thus the
%value of N required for a 0.1s clip of sound is 0.1/Ts, which in this
 case
%is equal to 4410.
```

```
%In analyizing the frequency response during the sound's first 0.1s,
 it is
%interesting to note that almost all the frequencies are below about
%1400Hz. In addition, the sound is dominated by a tone at 520Hz, as
 well as
%what may be harmonics at around 630Hz and 660Hz.
filename= 'gong.wav' ; % name of wave file
[ x, sr ]= audioread( filename ) ; % read in wavefile

Ts=1/sr ; % sample interval & # of samples
N=0.1/Ts;
ranger = (1 :N)+round((length(x)/2)) ;
x=x ( ranger)' ; % length for analysis
sound(x ,1/Ts ) % play sound ( if possible )
time=Ts * ( 0 : length (x) -1); % time base for plotting
subplot (2 , 2 , 2) , plot ( time , x) % and plot top figure
title("0.1s from middle")
magx=abs ( fft(x ) ) ; % take FFT magnitude
ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
subplot(2,2,4), plot(ssf,magx(1:N/2)) % plot mag spectrum
title("frequency magnitude")

%The sample from the middle of the sound has the same frequency peaks,
%although their amplitudes are significantly lower. Inaddition, the
 peak at
%630Hz is now larger than the peak at 660Hz, which is the opposite of
 what
%was the case at the beginning of the sound.
```



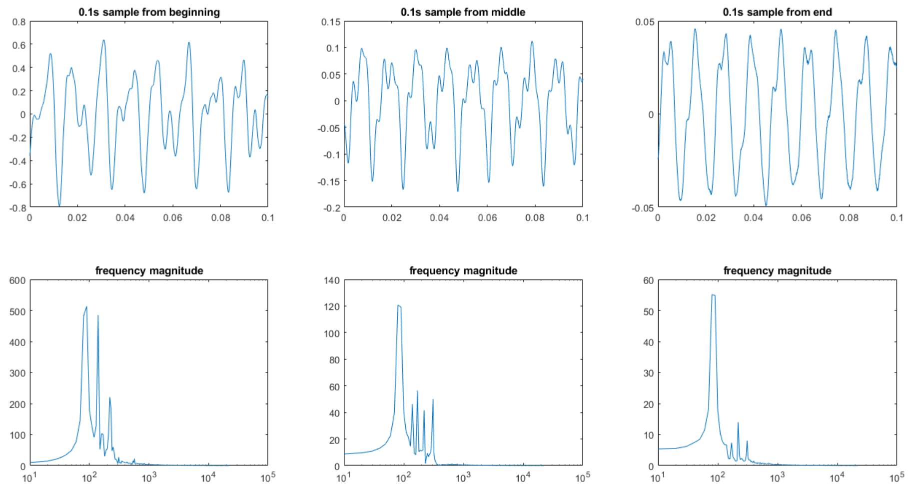

```
%%Exercise 7.11. A common practice when taking FFTs is to plot the
 magnitude
% on a log scale. This can be done in Matlab by replacing the plot
 command
% with semilogy. Try it in specgong.m. What extra details can you see?
figure(9)
filename= 'gong.wav' ; % name of wave file
```

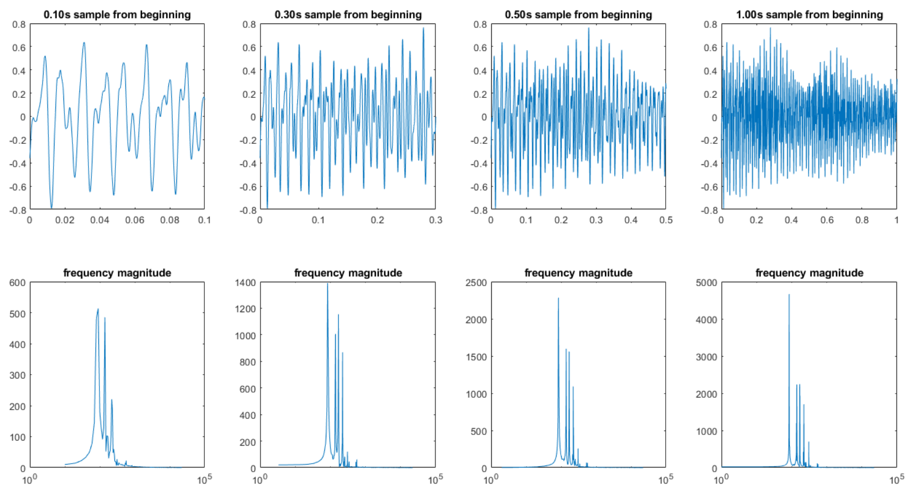
```
[ x, sr ]= audioread( filename ) ; % read in wavefile
Ts=1/sr ; % sample interval & # of samples
N=2^15; x=x ( 1 :N)' ; % length for analysis
sound(x ,1/Ts ) % play sound ( if possible )
time=Ts * ( 0 : length (x) -1); % time base for plotting
subplot (2 , 1 , 1 ) , plot ( time , x) % and plot top figure
magx=abs ( fft(x ) ) ; % take FFT magnitude
ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
subplot(2,1,2), semilogy(ssf,magx(1:N/2)) % plot mag spectrum

%plotting the frequency response with a log scale reveals repeating
 higher
%freuquency components that get lost in the noise when only a simple
%magnitude is plotted. These frequency components could be the result
 of
%aliasing, but are more likely to be harmonics of the gong's
 fundamental
%tones.
```



```
%%Exercise 7.12. The waveform of the sound produced by another, much
 larger
% gong is given in gong2.wav on the website. Conduct a thorough
 analysis of this
% sound, looking at the spectrum for a variety of analysis windows
 (values of N)
% and at a variety of times within the waveform.
figure(10)
locations = ["beginning", "middle", "end"];
for i = 1:3
    filename= 'gong2.wav' ; % name of wave file
    [ x, sr ]= audioread( filename ) ; % read in wavefile
    Ts=1/sr ; % sample interval & # of samples
    N=0.1/Ts; x=x ( (1:N)+round((length(x)/3)*(i-1)))' ; % length for
 analysis
    sound(x ,1/Ts ) % play sound ( if possible )
    time=Ts * ( 0 : length (x) -1); % time base for plotting
```

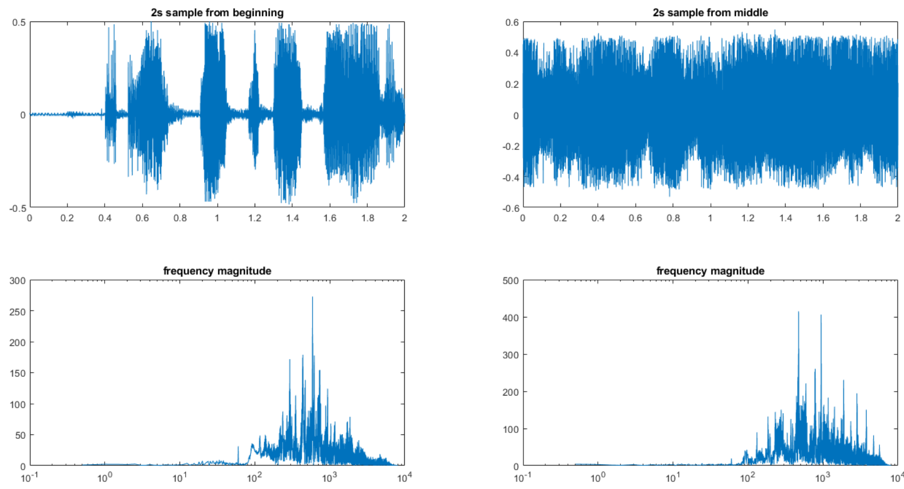
```
    subplot (2, 3 , i ) , plot ( time , x) % and plot top figure
    title(sprintf("0.1s sample from %s",locations(i)))
    magx=abs ( fft(x ) ) ; % take FFT magnitude
    ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
    subplot(2,3,3+i), semilogx(ssf,magx(1:N/2)) % plot mag spectrum
    title("frequency magnitude")
end

figure(11)
windows = [0.1,0.3,0.5,1];
for i = 1:4
    filename= 'gong2.wav' ; % name of wave file
    [ x, sr ]= audioread( filename ) ; % read in wavefile
    Ts=1/sr ; % sample interval & # of samples
    N=windows(i)/Ts; x=x ( (1:N))' ; % length for analysis
    sound(x ,1/Ts ) % play sound ( if possible )
    time=Ts * ( 0 : length (x) -1); % time base for plotting
    subplot (2, 4 , i ) , plot ( time , x) % and plot top figure
    title(sprintf("%0.2fs sample from beginning",windows(i)))
    magx=abs ( fft(x ) ) ; % take FFT magnitude
    ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
    subplot(2,4,4+i), semilogx(ssf,magx(1:N/2)) % plot mag spectrum
    title("frequency magnitude")
end

%the main difference between this sound and the previous one is that
 the
%frequency peaks are at much lower frequencies than with the smaller
 gong,
%with the fundamental tone being at 80Hz. In addition, more frequency
 peaks
%are visible. Looking at the signal at different times reveals that
 these
%frequency peaks change in relative magnitude over time, meaning that
 some
%harmonics are decaying faster than others. Finally, I observed that
%sampling over a longer time span resulted in similar changes to the
 the
%frequency peaks as sampling at different times, as well as increasing
 the
%magnitude of all peaks when the sample had a longer window, probably
 as a
%result of the increased energy transmitted over the longer window.
```






```
%%Exercise 7.13. Choose a .wav file from the website (in the Sounds
 folder) or
% download a .wav file of a song from the Internet. Conduct a FFT
 analysis of the
% first few seconds of sound, and then another analysis in the middle
 of the song.
figure(12)
locations = ["beginning", "middle", ];
for i = 1:2
    filename= 'im_a_pepper2.wav' ; % name of wave file
    [ x, sr ]= audioread( filename ) ; % read in wavefile
    Ts=1/sr ; % sample interval & # of samples
    N=2/Ts; x=x ( (1:N)+round((length(x)/2) *(i-1)))' ; % length for
 analysis
    sound(x ,1/Ts ) % play sound ( if possible )
    time=Ts * ( 0 : length (x) -1); % time base for plotting
    subplot (2, 2 , i ) , plot ( time , x) % and plot top figure
    title(sprintf("2s sample from %s",locations(i)))
```

```
    magx=abs ( fft(x ) ) ; % take FFT magnitude
    ssf =(0:N/2-1)/(Ts*N) ; % freq base for plotting
    subplot(2,2,2+i), semilogx(ssf,magx(1:N/2)) % plot mag spectrum
    title("frequency magnitude")
end
```




```
%%Listing 7.5. waystofilt.m “conv” vs. “filter” vs. “freq domain” vs.
 “time domain”
h=[1 -1 2 -2 3 -3]; % impulse response h[ k ]
x=[1 2 3 4 5 6 -5 -4 -3 -2 -1]; % input data x [ k ]
yconv=conv(h , x) % convolve x[k]* h [ k ]
yfilt=filter(h , 1 , x) % filterx[ k ] with h[ k ]
n=length (h)+length (x)-1; % pad l eng th f o r FFT
ffth=fft([h, zeros(1,n-length(h))]) ; % FFT o f h [ k ] i s H[ n ]
fftx=fft([x, zeros(1,n-length(x))]) ; % FFT of input i s X[ n ]
ffty=ffth.*fftx ; % product of H[ n ] and X[ n ]
yfreq=real ( ifft(ffty )) % IFFT of product give s y [ k ]
z=[zeros(1, length (h)-1) ,x] ; % initial filter state=0
for k=1: length (x) % time?domain method
    ytim(k)=fliplr(h)* z (k : k+length (h) -1)' ; % do f o r each x
 [ k ]
end % to directly calculate y [ k ]
```

```
yconv =

  Columns 1 through 13

     1      1      3      3      6      6     -6      6    -18      6    -30
 6      5

  Columns 14 through 16

     5      3      3
```

```
yfilt =

     1     1     3     3     6     6    -6     6   -18     6   -30


yfreq =

  Columns 1 through 7

    1.0000    1.0000    3.0000    3.0000    6.0000    6.0000   -6.0000

  Columns 8 through 14

    6.0000  -18.0000    6.0000  -30.0000    6.0000    5.0000    5.0000

  Columns 15 through 16

    3.0000    3.0000
```

```
%%Listing 7.6. waystofiltIIR.m ways to implement IIR filters
a=[1 -0. 8] ; lena=length(a)-1; % autoregressive coefficients
b=[1] ; lenb=length (b ) ; % moving average coefficients
d=randn( 1 , 20) ; % data to filter
if lena>=lenb % dimpulse needs lena>=lenb
    h=impz (b , a ) ; % impulse response of filter
    yfilt=filter(h , 1 , d) % filter x[ k ] with h[ k ]
end

yfilt2=filter(b , a , d) % filter using a and b
y=zeros (lena,1); x=zeros ( lenb , 1 ) ; % initial states in filter
for k=1: length (d)-lenb % time?domain method
    x=[d(k) ; x(1:lenb-1)] ; % past values of inputs
    ytim(k)=-a(2: lena+1)*y+b*x ; % directly calculate y[ k ]
    y=[ytim(k); y(1:lena -1)] ; % past values of outputs
end
```

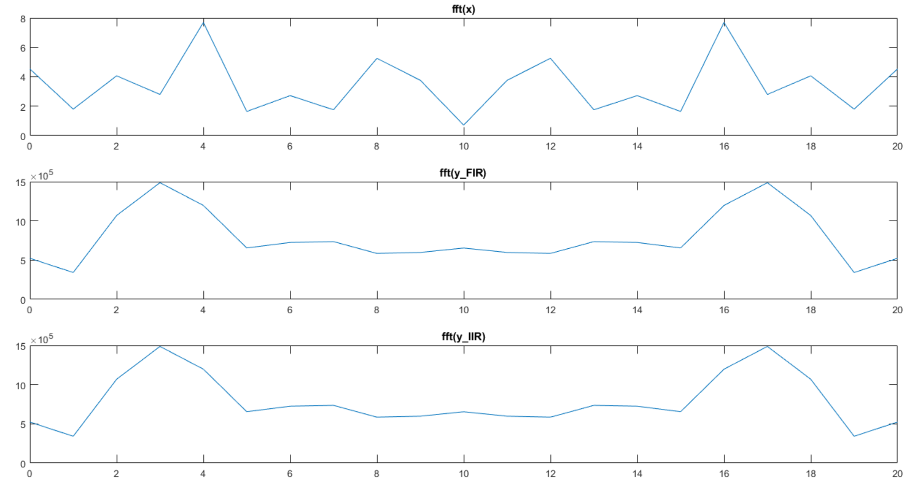
```
yfilt =

   1.0e+05 *

  Columns 1 through 7

   -0.0000   -0.0000    0.0000    0.0002   -0.0001   -0.0020    0.0009

  Columns 8 through 14

    0.0157   -0.0072   -0.1254    0.0579    1.0030   -0.4629   -8.0240

  Columns 15 through 20

   -2.5283   -3.5888   -2.5690   -1.2040   -0.7223    5.1598
```

```
yfilt2 =

   1.0e+08 *

  Columns 1 through 7

   -0.0000   -0.0000    0.0000    0.0000   -0.0000   -0.0000    0.0000

  Columns 8 through 14

    0.0000   -0.0000   -0.0001    0.0001    0.0010   -0.0005   -0.0080

  Columns 15 through 20

    0.0037    0.0642   -0.0296   -0.5135    0.2370    4.1083
```

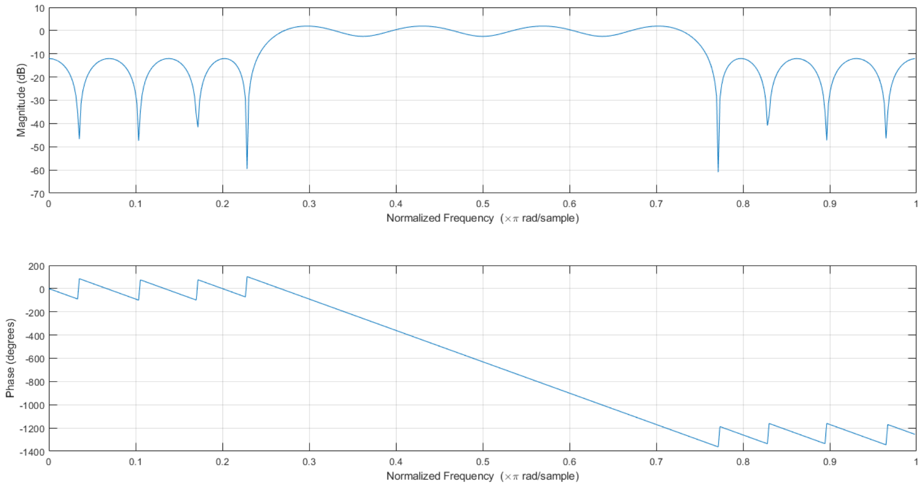
```
%%Exercise 7.15. FIR filters can be used to approximate the behavior
 of IIR filters
% by truncating the impulse response. Create a FIR filter with impulse
 response
% given by the first 10 terms of (7.9) for a = 0.9 and b = 2. Simulate
 the FIR filter
% and the IIR filter (7.8) in Matlab, using the same random input to
 both. Verify
% that the outputs are (approximately) the same.
h_FIR = [1,2,3,4,5,6,7,8,9,10];
a=0.9; b=2;
for k = 1:length(terms)
    h_FIR(k) = (a*b^k);
end
x = randn(1,21)

%FIR filter
yfilt_FIR=filter(h, 1, x) % filter x[ k ] with h[ k ]

%IIR filter
h_IIR = impz (b, a);
yfilt_IIR=filter(h, 1, x)
figure(12)
subplot(3,1,1); plot(0:length(x)-1,abs(fftshift(fft(x))));
 title("fft(x)")
subplot(3,1,2); plot(0:length(x)-1,abs(fftshift(fft(yfilt_FIR))));
 title("fft(y\_FIR)")
subplot(3,1,3); plot(0:length(x)-1,abs(fftshift(fft(yfilt_IIR))));
 title("fft(y\_IIR)")
```

```
x =

  Columns 1 through 7

    0.1205   -0.9899    1.1978   -0.5927   -0.4698    0.8864   -1.3852
```

```
  Columns 8 through 14

   -1.9568    0.4207    0.4007    0.0951    0.4967    1.0822    0.9704

  Columns 15 through 21

   -0.5686    0.8100    0.1732   -0.5055   -1.1933    0.6470   -0.3536


yfilt_FIR =

   1.0e+05 *

  Columns 1 through 7

    0.0000   -0.0000    0.0000    0.0001   -0.0000   -0.0006    0.0002

  Columns 8 through 14

    0.0046   -0.0014   -0.0368    0.0111    0.2943   -0.0885   -2.3542

  Columns 15 through 21

    3.2349   -1.9260   -0.7602    2.9792   -3.7712   -5.2449    1.1191


yfilt_IIR =

   1.0e+05 *

  Columns 1 through 7

    0.0000   -0.0000    0.0000    0.0001   -0.0000   -0.0006    0.0002

  Columns 8 through 14

    0.0046   -0.0014   -0.0368    0.0111    0.2943   -0.0885   -2.3542

  Columns 15 through 21

    3.2349   -1.9260   -0.7602    2.9792   -3.7712   -5.2449    1.1191
```




```
%%Listing 7.7. bandex.m design a bandpass filter and plot frequency
 response
fbe=[0 0.24 0.26 0.74 0.76 1 ] ; % freq band edges as a fraction of
% the Nyquist f requency
damps=[0 0 1 1 0 0]; % desired amplitudes at band edges
fl =30; % filter size
b=firpm( fl, fbe, damps ) ; % b is the designed impulse response
figure (13)
freqz(b) % plot freq response to check design
```



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