Basics on Signal Processing

Matlab Exercises

- Convolution
- DFT
- Filters

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Patras, 2011

CONVOLUTION

Exercise 1a

The digital signal x(n)={3,4,5,2} is applied to a LTI system the impulse response of which is h(n)={1,2,3}. Calculate the output of the system. Compare it to the output of another system the input of which is h(n) and the impulse response is x(n). Plot on the same graph all digital signals.

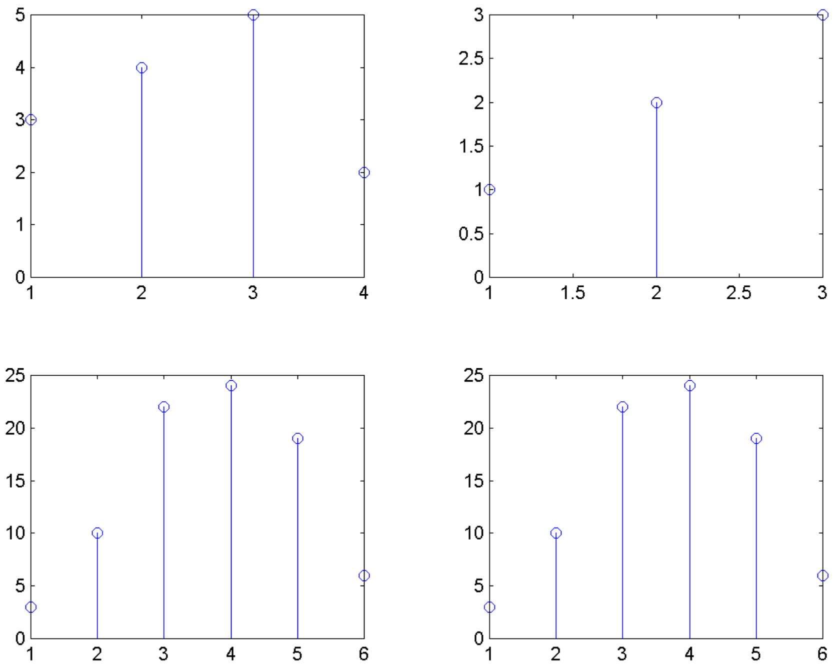
(The aim of this exercise is to understand that convolution operation is commutative).

```
x=[3 4 5 2];
h=[1 2 3];

y1=conv(x,h);
y2=conv(h,x);

dif=y1-y2

subplot(2,2,1), stem(x);
subplot(2,2,2), stem(h);
subplot(2,2,3), stem(y1);
subplot(2,2,4), stem(y2);
```



Exercise 1b

The digital signal x(n)={3,4,5,2} is applied to a LTI system the impulse response of which is h(n)={1,2,3}. The output of the system is then applied to another LTI system the impulse response of which is g(n)={3,2,1}. Calculate the output of the second system. Compare it to the output of a system the input of which is x(n) and the impulse response is h(n)*g(n). Plot on the same graph all digital signals.

(The aim of this exercise is to understand the cascade connection of systems).

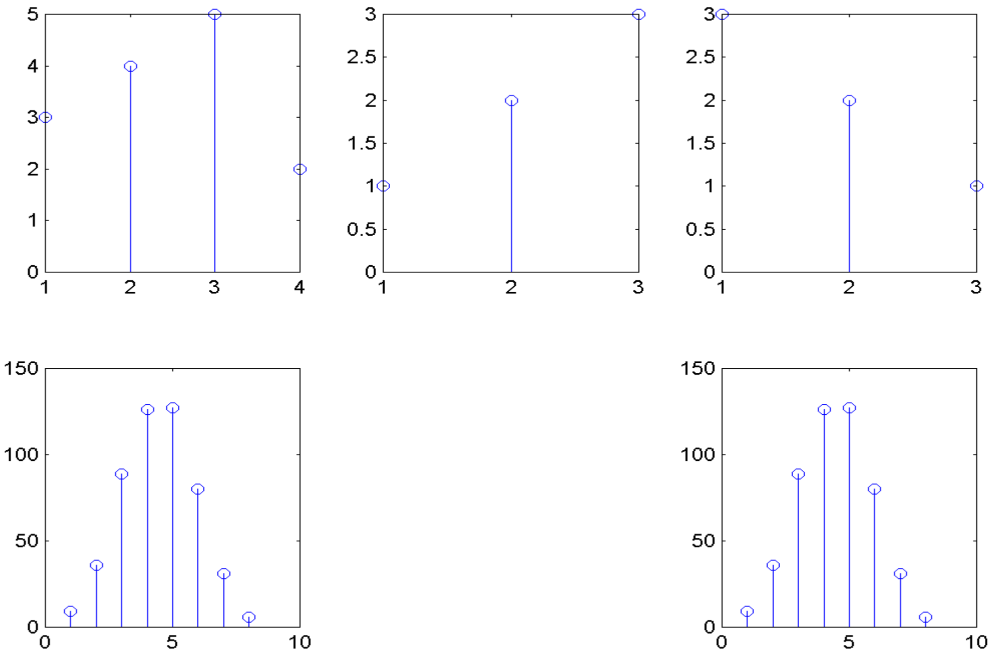
```
x=[3 4 5 2];
h=[1 2 3];
g=[3 2 1];

r=conv(x,h);
y1=conv(r,g);

hg=conv(h,g);
y2=conv(x,hg);

dif=y1-y2

subplot(2,3,1), stem(x);
subplot(2,3,2), stem(h);
subplot(2,3,3), stem(g);
subplot(2,3,4), stem(y1);
subplot(2,3,6), stem(y2);
```



Exercise 1c

The digital signal x(n)={3,4,5,2} is applied to two LTI systems the impulse responses of which are h(n)={1,2,3} and g(n)={3,2,1}. The two outputs are summed. Compare this signal to the output of a system the input of which is x(n) and the impulse response is h(n)+g(n). Plot on the same graph all digital signals.

(The aim of this exercise is to understand the parallel connection of systems).

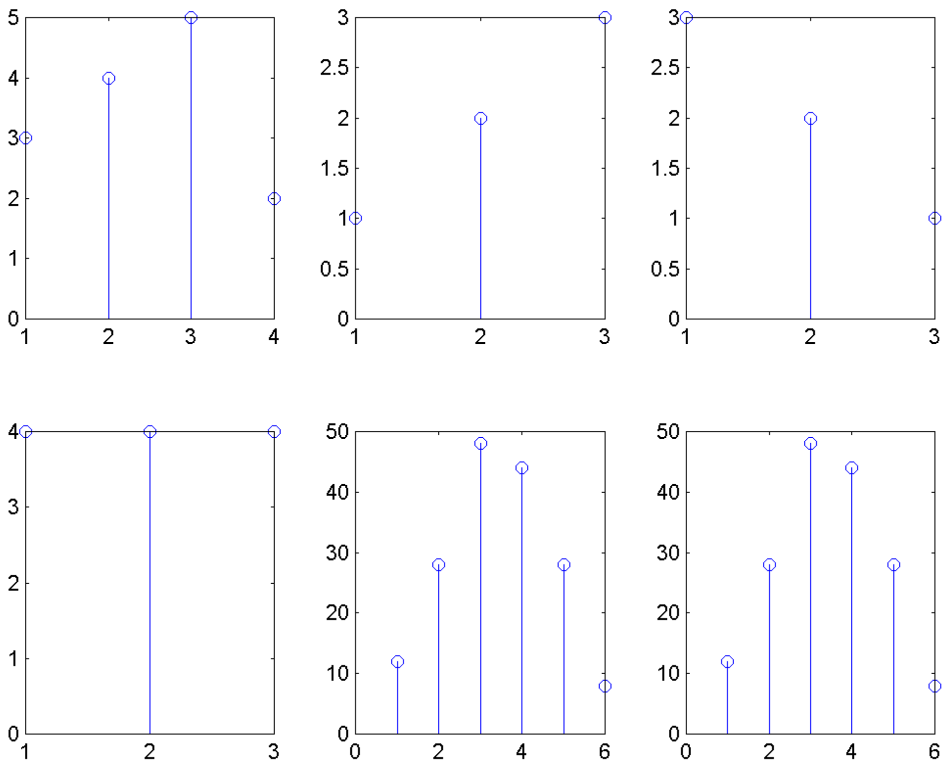
```
x=[3 4 5 2];
h=[1 2 3];
g=[3 2 1];

y1=conv(x,h);
y2=conv(x,g);
yy1=y1+y2;

hg=h+g;
yy2=conv(x,hg);

dif=yy1-yy2

subplot(2,3,1), stem(x);
subplot(2,3,2), stem(h);
subplot(2,3,3), stem(g);
subplot(2,3,4), stem(hg);
subplot(2,3,5), stem(yy1);
subplot(2,3,6), stem(yy2);
```

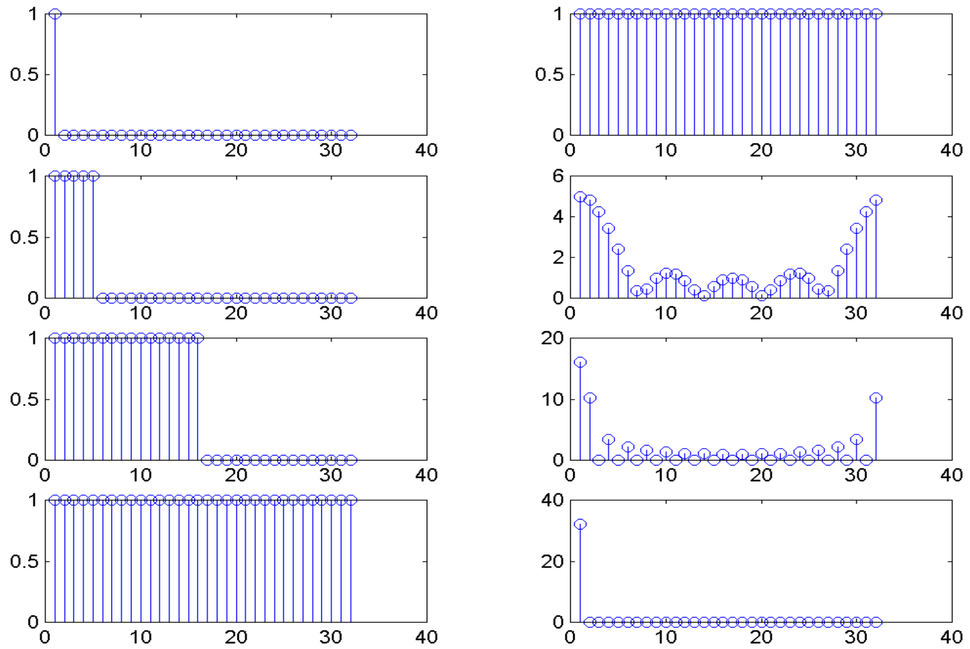


DFT

Exercise 2a

Calculate the 32-point DFT of the following sequences:
x1(n)=δ(n),
x2(n)=1 for n=1 to 5 and x2(n)=0 for n=6 to 32,
x3(n)=1 for n=1 to 16 and x3(n)=0 for n=17 to 32,
x4(n)=1 for n=1 to 32.
Plot all signals and their corresponding DFT magnitudes on the same graph.

(The aim of this exercise is to understand duality between the time and the frequency representations and the type of the DFT for the most common digital signals).




```
n=1:32;

x1=[1 zeros(1,31)];
X1=abs(fft(x1,32));
subplot(4,2,1), stem(n,x1);
subplot(4,2,2), stem(n,X1);

x2=[1 1 1 1 1 zeros(1,27)];
X2=abs(fft(x2,32));
subplot(4,2,3), stem(n,x2);
subplot(4,2,4), stem(n,X2);

x3=[ones(1,16) zeros(1,16)];
X3=abs(fft(x3,32));
subplot(4,2,5), stem(n,x3);
subplot(4,2,6), stem(n,X3);

x4=[ones(1,32)];
X4=abs(fft(x4,32));
subplot(4,2,7), stem(n,x4);
subplot(4,2,8), stem(n,X4);
```

Exercise 2b

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```

Calculate the 32-point DFT of the sequences:
x(n)=1 for n=1 to 5 and x(n)=0 for n=6 to 32,
x2(n)=x(n-4),
x3(n)=x(n-27).
Plot all signals and their corresponding DFT magnitudes on the same graph.

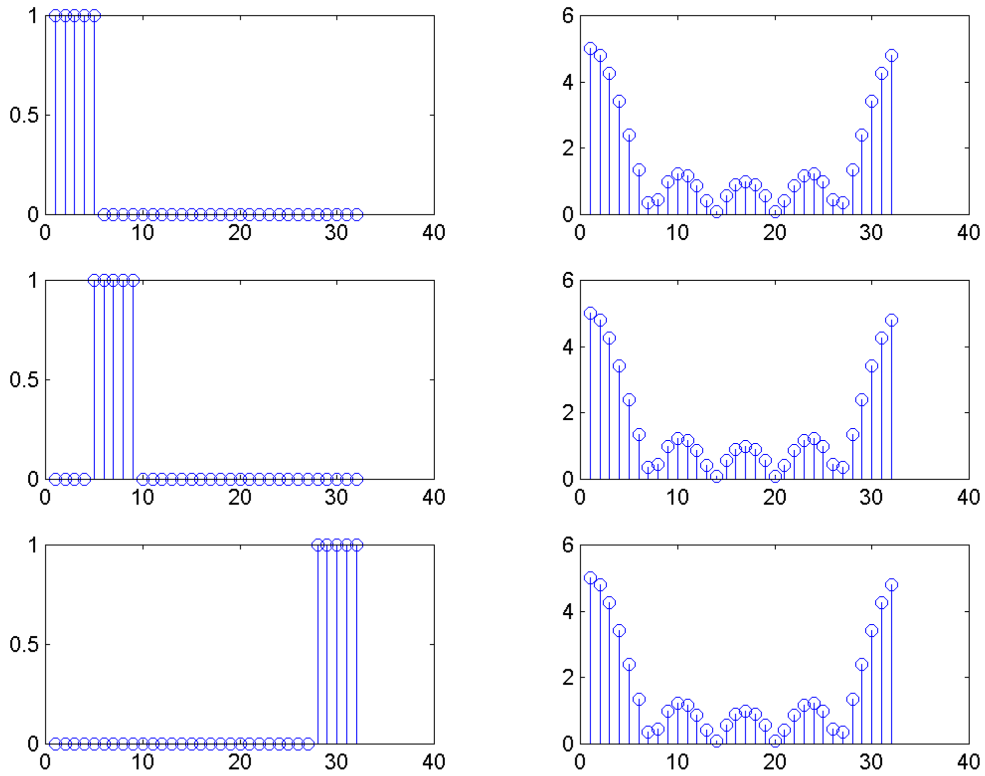
(The aim of this exercise is to understand the effect of the time-shift on the frequency domain).

```
n=1:32;

x=[1 1 1 1 1 zeros(1,27)];
X=abs(fft(x,32));
subplot(3,2,1), stem(n,x);
subplot(3,2,2), stem(n,X);

xm4=[zeros(1,4) 1 1 1 1 1
zeros(1,23)];
XM4=abs(fft(xm4,32));
subplot(3,2,3), stem(n,xm4);
subplot(3,2,4), stem(n,XM4);

xm27=[zeros(1,27) 1 1 1 1 1];
XM27=abs(fft(xm27,32));
subplot(3,2,5), stem(n,xm27);
subplot(3,2,6), stem(n,XM27);
```



Exercise 2c

Calculate the linear convolution of the sequences x(n)={3,4,5,2} and h(n)={1,2,3} by means of the DFT.

(The aim of this exercise is to understand that (a) convolution in the time-domain is equivalent to multiplication in the frequency domain, (b) multiplication of the DFTs corresponds to circular convolution, and (c) zero-padding of the original signals is needed in order for the circular convolution to give the same results as the linear convolution).

```
x=[3 4 5 2];
h=[1 2 3];

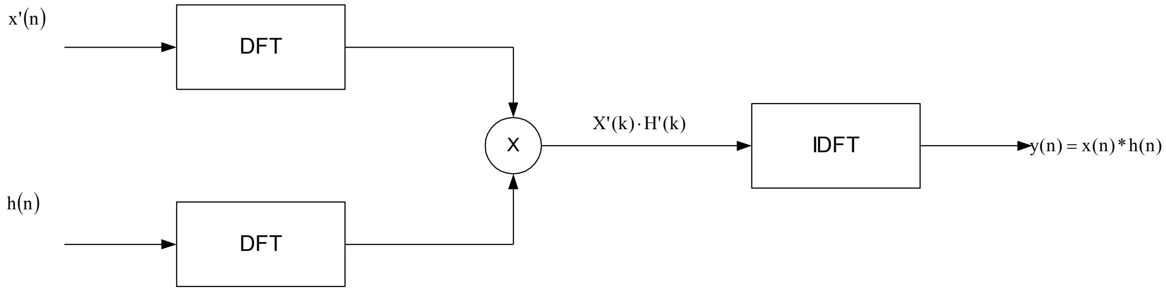
xn=[3 4 5 2 0 0];
hn=[1 2 3 0 0 0];

X=fft(xn);
H=fft(hn);

Y=H.*X;

y=ifft(Y)

yc=conv(x,h)
```



Exercise 2d

Generate 2^{20} samples of the signal x1=sin(2*pi*n1*0.0625)+sin(2*pi*n1*0.125) and calculate its FFT. Take the first 2^{20}-1000 samples of the previous signal and calculate the corresponding FFT. Plot on the same graph the first 16 samples of the signals and the DFT coefficients (magnitude only). Comment on the required time and on the graphs.

(The aim of this exercise is to understand (a) the speedup achieved by the FFT as compared to the DFT, and (b) the frequency leakage).

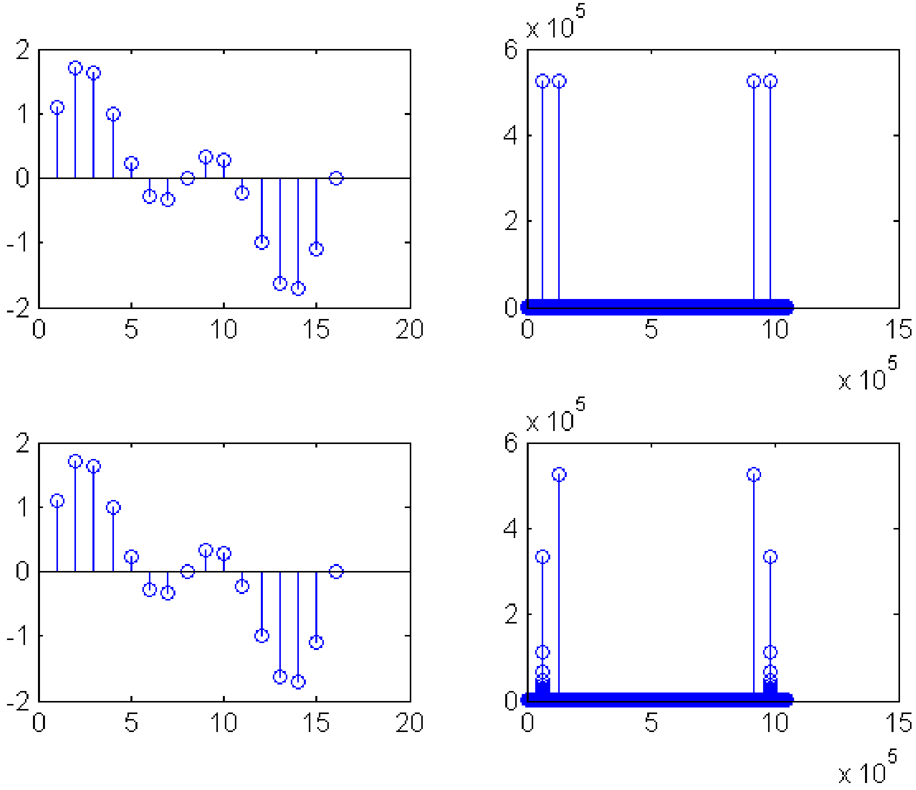
```
clear all;
close all;

N1=2^20;
n1=1:N1;
x1=sin(2*pi*n1*0.0625)+sin(2*
pi*n1*0.125);

N2=2^20-1000;
n2=1:N2;
x2=x1(1:N2);

tic, X1=abs(fft(x1,N1)); toc
tic, X2=abs(fft(x2,N2)); toc


subplot(2,2,1),
stem(x1(1:16));
subplot(2,2,2), stem(X1);
subplot(2,2,3),
stem(x2(1:16));
subplot(2,2,4), stem(X2)
```

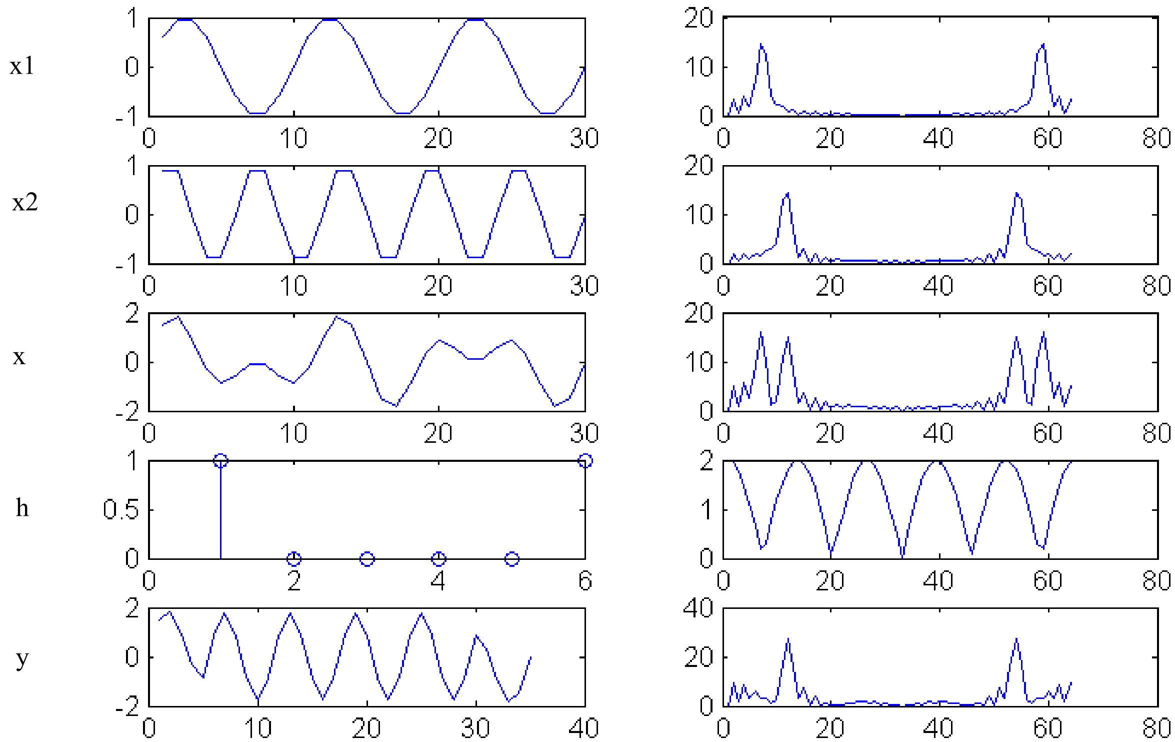


FILTERS

Exercise 3a

Generate 30 samples of two sinusoidal signals x1, x2 with frequencies F1=30 and F2=50 correspondingly (the sampling period is Fs=300) and add them. A filter with an impulse response of h={1,0,0,0,0,1} is then applied to the signal. Plot on the same graph the original signal, the 'filtered' signal as well as their corresponding FFTs. Comment on the graphs.

(The aim of this exercise is to understand that (a) the convolution of a signal with a proper filter allows some signals to pass while it blocks others (b) convolution in the time-domain is equivalent to multiplication in the frequency domain).




```
clear all
close all

Fs=300;
n=1:30;
F1=30;
F2=50;
x1=sin(2*pi*n*F1/Fs);
x2=sin(2*pi*n*F2/Fs);
x=x1+x2;

X1=abs(fft(x1,64));
X2=abs(fft(x2,64));
X=abs(fft(x,64));

h=[1 zeros(1,4) 1];
y=conv(x,h);

H=abs(fft(h,64));
Y=abs(fft(y,64));

subplot(5,2,1), plot(x1);
subplot(5,2,2), plot(X1);
subplot(5,2,3), plot(x2);
subplot(5,2,4), plot(X2);
subplot(5,2,5), plot(x);
subplot(5,2,6), plot(X);
subplot(5,2,7), stem(h);
subplot(5,2,8), plot(H);
subplot(5,2,9), plot(y);
subplot(5,2,10), plot(Y);
```

Exercise 3b

Read the image file Barbara.gif and perform down sampling, keeping one out of three pixels in each row and each column of the image. Then, resize the image to its original size and calculate the FFT for the original and the downsampled image (magnitude only). Comment on the differences between the two images.

(The aim of this exercise is to understand the aliasing effect on an image. It takes place when the sampling rate is less than the Nyquist sampling rate (twice the highest frequency of the signal)).

```
clear all;
close all;

f = imread('barbara.gif');
[ysize,xsize] = size(f);

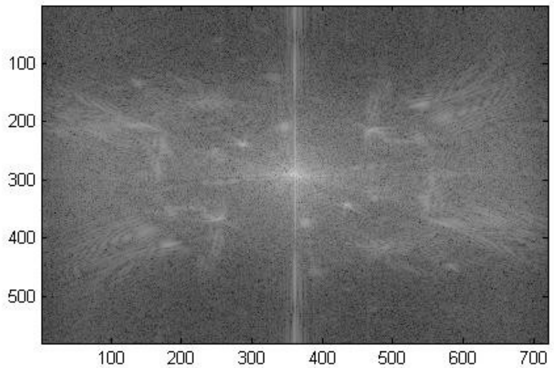
imshow(f)

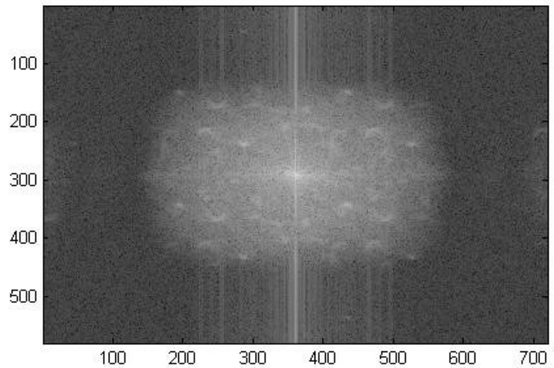
% Downsample image
fd=f(1:3:end,1:3:end);

fd=imresize(fd,[ysize
xsize]);


F=fftshift(log(1+abs(fft2(f))
));
Fd=fftshift(log(1+abs(fft2(fd
))));

subplot(2,2,1),imshow(f)
subplot(2,2,2),imshow(fd)
subplot(2,2,3),imagesc(F)
subplot(2,2,4),imagesc(Fd)
```





Exercise 3c

```
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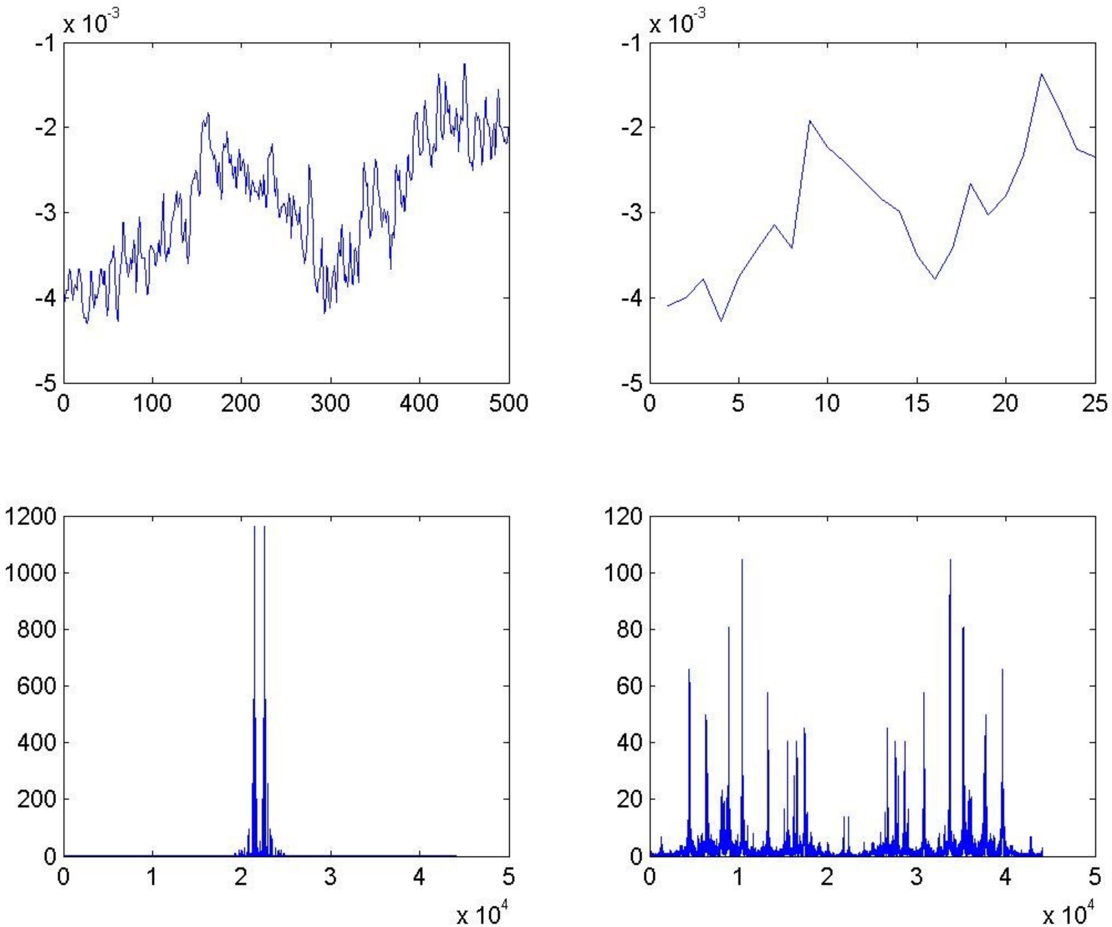
```

Open the .wav file scarlattiOrig.wav and play it. Perform down sampling, with a factor of 20. Calculate the FFT for the original and the down sampled sound file. Comment on the differences between the plots as well as the audio result.

(The aim of this exercise is to understand the aliasing effect on a sound file).




```
clear all
close all

[y,Fs,nbits]=wavread('scarlattiOrig.wav');
wavplay(y,Fs)

%Perform downsampling
yd=y(1:20:end);

Y=abs(fftshift(fft(y,Fs)));
Yd=abs(fftshift(fft(yd,Fs)));

subplot(2,2,1),plot(y(1:500))
subplot(2,2,2),plot(yd(1:500/20))
subplot(2,2,3),plot(Y)
subplot(2,2,4),plot(Yd)

wavplay(yd,Fs/20)
```