Signals_LAB4_Sonmez_Isil

0

20

40

60

```
%% Task 1
n=10; % n-number of letters in your name
s=6; % s-number of letters in your surname
T=10; % total length of time
fs=200; % sampling frequency
dt=1/fs; % time step
t=0:dt:T-dt; % time vector
s=2*sin(2*pi*0.7*n*t)+cos(2*pi*7.1*s*t)+0.3*cos(2*pi*(s+n)*t+0.5*pi);
S=fft(s);
N=length(s); % number of samples
df=fs/N; % frequency step
f_{\text{vec}} = (0:(N-1))*df; % frequency step
S_amp = abs(S); % Amplitude
S_phase = atan2(imag(S),real(S)); % phase
S_amp_norm = abs(S)/N*2; S_amp_norm(1)=S_amp_norm(1)/2;
S_phase_norm = S_phase; S_phase_norm(S_amp_norm<0.01)=0;</pre>
figure (1), \ subplot (2,1,1), \ plot (f\_vec, \ S\_amp) \ \% \ plot \ amplitude \ spectrum
xlabel('Frequency [Hz]'), ylabel('Magnitude')
subplot(2,1,2), plot(f_vec, S_phase)
xlabel('Frequency [Hz]'), ylabel('Phase')
 Figure 1
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                   ==
                           ▲ 월目 ७ 및 및 ☆
      2000
      1500
   Magnitude
      1000
       500
         0
                20
                                                120
                                                                          200
          0
                       40
                             60
                                   80
                                          100
                                                       140
                                                             160
                                                                    180
                                    Frequency [Hz]
         4
         2
```

The code's results depends on looking at the signal's amplitude and phase. They show what frequencies are in the signal and how strong they are. The total time of the signal affects how clear these frequencies are. A longer time gives better detail, helping separate close frequencies. But with a shorter time, it's harder to see each frequency clearly and they might overlap. So, choosing the right time helps us see the signal's frequencies more accurately.

120

140

160

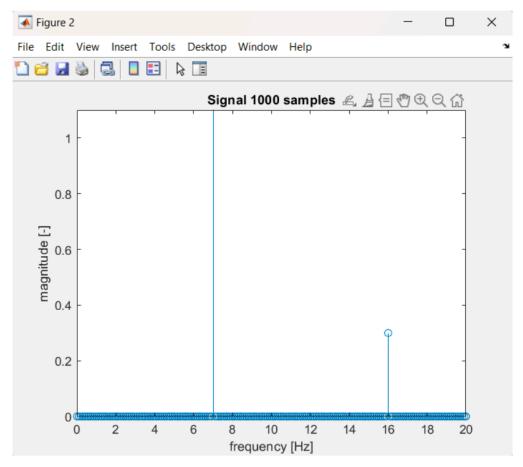
180

200

100

Frequency [Hz]

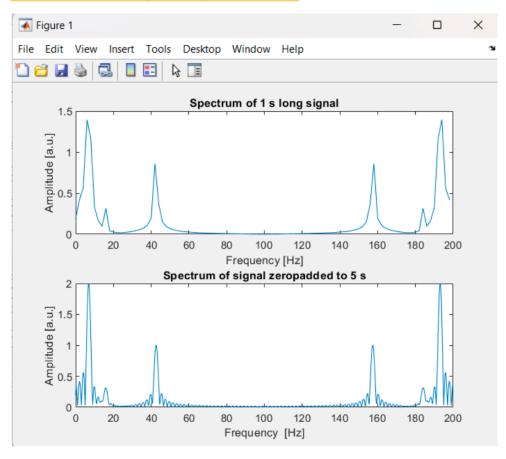
```
figure(2), subplot(1,1,1), stem(f_vec,abs(fft(s))/N*2),
xlim([0 20]), ylim([0 1.1]), ylabel('magnitude [-]')
xlabel('frequency [Hz]'); title('Signal 1000 samples'),
```



%% Task 2

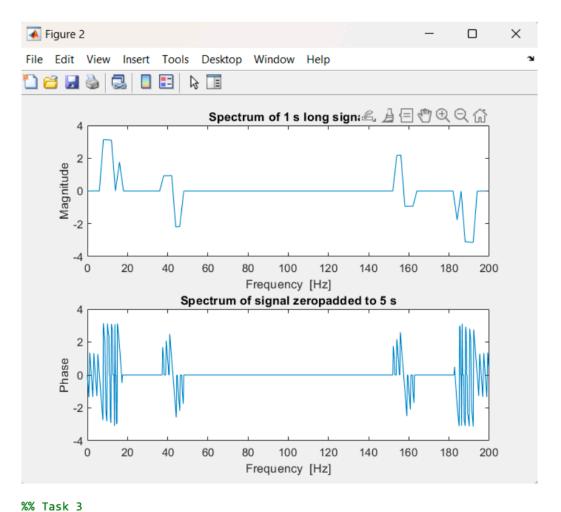
```
n=140; % n-number of letters in your name
s=6; % s-number of letters in your surname
T=0.5; % total length of time
fs=200; % sampling frequency
dt=1/fs; % time step
t=0:dt:T-dt; % time vector
x=2*sin(2*pi*0.7*n*t)+cos(2*pi*7.1*s*t)+0.3*cos(2*pi*(s+n)*t+0.5*pi);
N=length(x); % number of samples
df=fs/N; %frequency vector
fv = (0:(N-1))*df; % frequency step
y=fft(x);
amp=2*abs(y)/N;
amp(1)=amp(1)/2;
ph=atan2(imag(y),real(y)); ph(amp<0.1)=0;</pre>
N2=4*fs;
df2=fs/N2; fv2=(0:N2-1)*df2;
y=fft(x,N2);
amp2=2*abs(y)/N;
amp2(1)=amp2(1)/2;
ph2=atan2(imag(y),real(y)); ph2(amp2<0.1)=0;</pre>
```

```
figure(1),
subplot(2,1,1), plot(fv,amp), xlim([0 fs/2]);
xlabel('Frequency [Hz]'), ylabel('Amplitude [a.u.]');xlim([0 200]);
title('Spectrum of 1 s long signal')
subplot(2,1,2), plot(fv2,amp2), xlim([0 fs/2]);
xlabel('Frequency [Hz]'), ylabel('Amplitude [a.u.]');xlim([0 200]);
title('Spectrum of signal zeropadded to 5 s')
```

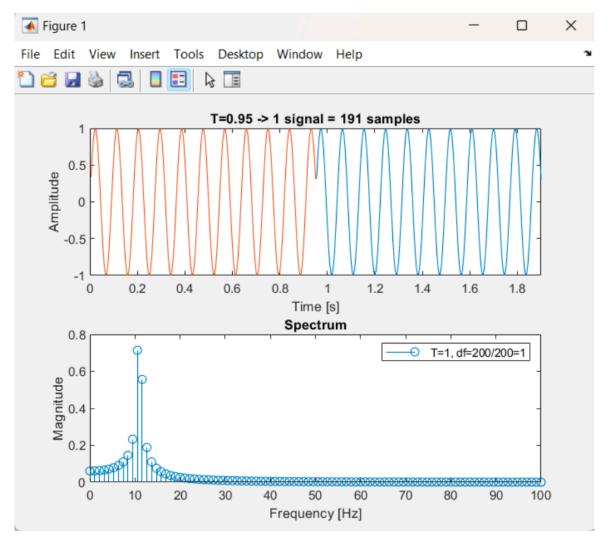


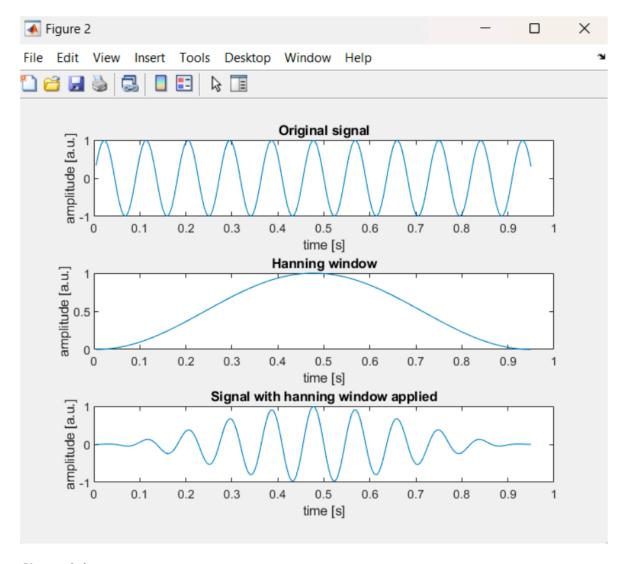
```
figure(2)
subplot(2,1,1), plot(fv,ph), xlim([0 fs/2]);
xlabel('Frequency [Hz]'), ylabel('Magnitude');xlim([0 200]);
title('Spectrum of 1 s long signal')
subplot(2,1,2), plot(fv2,ph2), xlim([0 fs/2]);
xlabel('Frequency [Hz]'), ylabel('Phase');xlim([0 200]);
title('Spectrum of signal zeropadded to 5 s')
```

The code compares how the frequency spectrum of a signal changes when we pad it with zeros to make it longer. When the signal is short (0.5 seconds), it's like hard to see the details. But when we do it to 5 seconds, it's like making the details clearer. So, adding zeros helps us see the signal's frequencies better

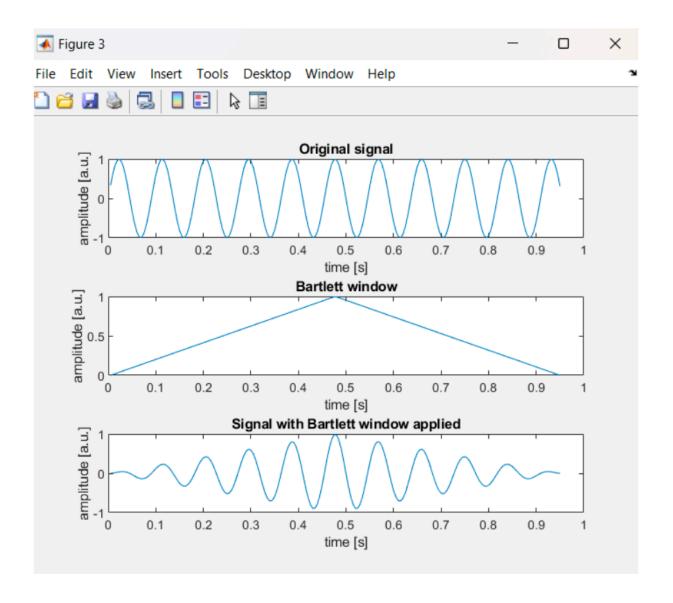


```
clc;
clear all;
close all;
fs=200;
dt=1/fs;
t=dt:dt:0.95;
x=sin(2*pi*11*t);
T=dt:dt:length(t)*2*dt;
X=[x x];
y=abs(fft(x))/length(x)*2;
f=(0:length(y)-1)*fs/length(y);
figure(1)
subplot(2,1,1), plot(T,X), hold on, plot(t,x); hold off;
xlim([0 max(T)]); xlabel('Time [s]'), ylabel('Amplitude'),
title('T=0.95 -> 1 signal = 191 samples');
subplot(2,1,2), stem(f,y); hold off;
xlim([0 fs/2]); xlabel('Frequency [Hz]'), ylabel('Magnitude'),
title('Spectrum');
legend('T=1, df=200/200=1', 'T=0.95, df=200/191=1.0471');
```

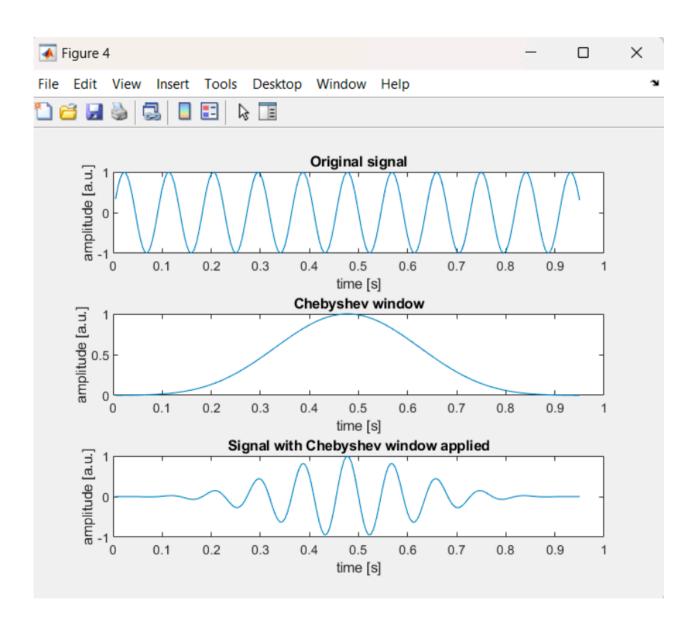




```
figure(3)
subplot(3,1,1), plot(t,x); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Original signal')
subplot(3,1,2), plot(t,window2); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Bartlett window')
subplot(3,1,3), plot(t,x2); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Signal with Bartlett window applied')
```



```
figure(4)
subplot(3,1,1), plot(t,x); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Original signal')
subplot(3,1,2), plot(t,window3); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Chebyshev window')
subplot(3,1,3), plot(t,x3); xlabel('time [s]'),
ylabel('amplitude [a.u.]'), title('Signal with Chebyshev window applied')
```



```
clc:
clear all;
close all;
fs = 200:
dt = 1/fs:
t = dt:dt:0.95;
x = \sin(2*pi*11*t);
T = dt:dt:length(t)*2*dt;
X = [x x];
y original = abs(fft(x))/length(x)*2;
f_original = (0:length(y_original)-1)*fs/length(y_original);
% Different types of windows
window_types = {@rectwin, @bartlett, @hann, @hamming, @blackman};
window_labels = {'Rectangular', 'Bartlett', 'Hann', 'Hamming', 'Blackman'};
% Additional window types
window_types_extra = {@hanning, @chebwin, @bartlett};
window_labels_extra = {'Hanning', 'Chebyshev', 'Bartlett'};
figure;
for i = 1:length(window_types_extra)
    window_func = window_types_extra{i};
    window_label = window_labels_extra{i};
    % Apply windowing function
    window_length = length(x);
    window = window_func(window_length);
    x windowed = x .* window';
    % Calculate FFT of original and windowed signals
    y_{original} = abs(fft(x))/length(x)*2;
    y_windowed = abs(fft(x_windowed))/length(x_windowed)*2;
    % Plot frequency spectra
    subplot(length(window_types_extra), 1, i);
    plot(f_original, y_original, 'b'); hold on;
    plot(f original, y windowed, 'r'); hold off;
    xlim([0 fs/2]);
    xlabel('Frequency [Hz]'), ylabel('Magnitude'),
    title(['Frequency spectrum with ' window_label ' window']);
    legend('Original Signal', 'Windowed Signal');
end
```

Conclusion

%Windowing functions reduce spectral leakage by smoothing the signal's edges. This prevents signal energy from spreading into neighboring frequency bins. Different windows offer varying levels of leakage reduction, balancing factors like main lobe width and side lobe levels. Choosing the right window function is about finding the best compromise for your signal and analysis goals.

