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Conclusion

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Introduction: The purpose of this lab was to realize the properties of DFT on Matlab, and use filters to remove noise from a sound file. It was dicovered in the lab that DFT has hermitian symetry, and noise from music can be removed using filters through convolution, and well as multiplication when both signals are in frequency domain. The spectogram was used here.

```
%Materials: PC, Matlab, headphones
```

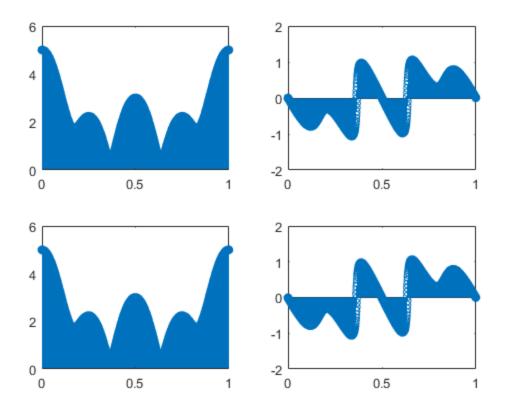
Q1

1a

```
clear all;
close all;
x=[2 1 1 0 1];
N=length(x);
M=1000;
X=fft(x,M);
Xmr=fliplr(X);
Xcmr=conj(Xmr);
```

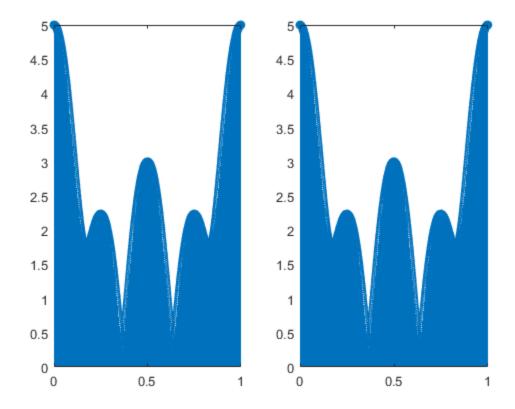
```
f=(0:1/nlen:1-1/nlen);
figure
subplot(2,2,1);
stem(f,abs(X));
%*X[M-r]
subplot(2,2,3);
stem(f,abs(Xcmr));
subplot(2,2,2);
stem(f,angle(X));
%X*[M-r]
subplot(2,2,4);
stem(f,angle(Xcmr));
% The frequncy response for both signals are identical. THe DFT was
% conducted with 1000 points to increase the resolution.
%X[r] for the 2-11th point [4.99958549434325
 - 0.0439792792922490i,4.99834219009313
 - 0.0879404535064596i,4.99627072528201
 - 0.131865427913275i,4.99337216287062
 - 0.175736128474322i,4.98964799010848
 - 0.219534512171746i,4.98510011763912
 - 0.263242577318624i,4.97973087835050
 - 0.306842373843884i,4.97354302597157
 - 0.350316013545391i,4.96653973341556
 - 0.393645680304887i,4.95872459087138 -
0.436813640258472i,4.95010160364396 - 0.479802251916369i]
%X*[M-r] for the first 10 points [4.99958549434325
 - 0.0439792792922490i,4.99834219009313
 - 0.0879404535064596i,4.99627072528201
 - 0.131865427913275i,4.99337216287062
 - 0.175736128474322i,4.98964799010848
 - 0.219534512171746i,4.98510011763912
 - 0.263242577318624i,4.97973087835050
 - 0.306842373843884i,4.97354302597157
 - 0.350316013545391i,4.96653973341556
 -0.393645680304887i, 4.95872459087138
 0.436813640258472i, 4.95010160364396 - 0.479802251916369i
%X[r]=X*[m-r]
```

2



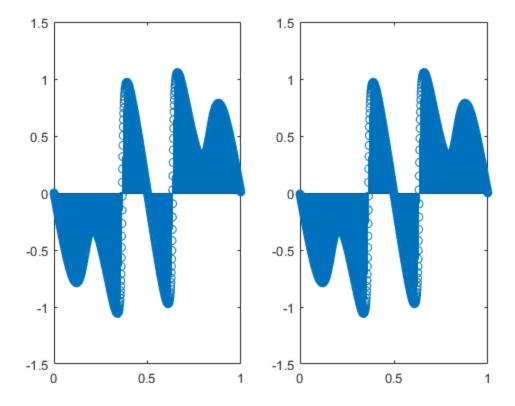
1b

```
figure
subplot(1,2,1);
stem(f,abs(X));
subplot(1,2,2);
stem(f,abs(Xmr));
% The magnitude response for both signals are identical. THe DFT was
% conducted with 1000 points to increase the resolution.
```



1c

```
figure
subplot(1,2,1);
stem(f,angle(X));
subplot(1,2,2);
stem(f,-angle(Xmr));
% The phase response for both signals are identical. THe DFT was
% conducted with 1000 points to increase the resolution.
```



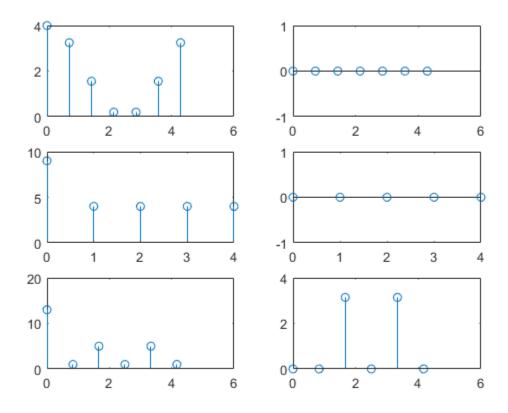
Q2

```
clear all;
close all;
x1=[2 1 0 0 0 0 1];
x2=[5 1 1 1 1];
x3=[1 \ 3 \ 3 \ 0 \ 3 \ 3];
N1=length(x1);
N2=length(x2);
N3=length(x3);
f1=(0:1/N1:1-1/N1)*5;
f2=(0:1/N2:1-1/N2)*5;
f3=(0:1/N3:1-1/N3)*5;
X1=fft(x1,N1);
X2=fft(x2,N2);
X3=fft(x3,N3);
figure
subplot(3,2,1);
stem(f1,abs(X1));
subplot(3,2,2);
stem(f1,angle(X1));
```

```
subplot(3,2,3);
stem(f2,abs(X2));
subplot(3,2,4);
stem(f2,angle(X2));

subplot(3,2,5);
stem(f3,abs(X3));
subplot(3,2,6);
stem(f3,angle(X3));

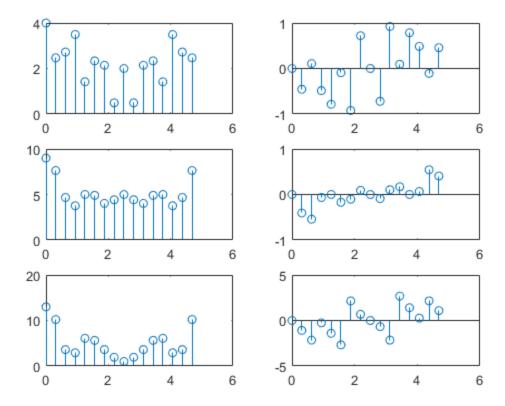
%It was realized that the phase response for two of the signals was zero.
%There are not enough samples to give a true representation of the acutal
%phase response.
```



Q2ii

```
clear all;
x1=[2 1 0 0 0 0 1];
x2=[5 1 1 1 1];
x3=[1 3 3 0 3 3];
N1=16;
```

```
N2=16;
N3 = 16;
f1=(0:1/N1:1-1/N1)*5;
f2=(0:1/N2:1-1/N2)*5;
f3=(0:1/N3:1-1/N3)*5;
X1=fft(x1,N1);
X2=fft(x2,N2);
X3=fft(x3,N3);
figure
subplot(3,2,1);
stem(f1,abs(X1));
subplot(3,2,2);
stem(f1,angle(X1));
subplot(3,2,3);
stem(f2,abs(X2));
subplot(3,2,4);
stem(f2,angle(X2));
subplot(3,2,5);
stem(f3,abs(X3));
subplot(3,2,6);
stem(f3,angle(X3));
% The resolution of the frequency axis was greatly increased; what has
% happened was zeros were padded at the end of the x, increasing the
% resolution of the frequncy response for thier respective DFTs.
```



Q3

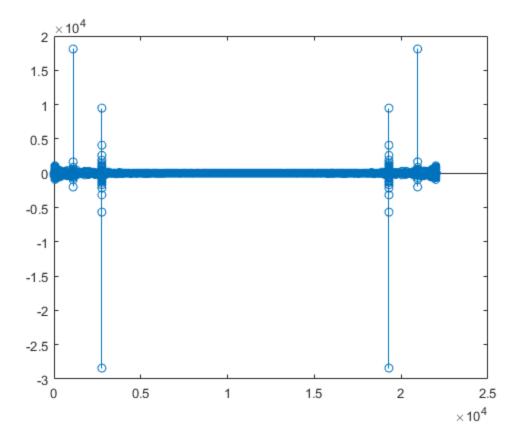
3.1

```
clear all;
[y,FS]=audioread('music_noisy.wav');
sound(y,FS);
%The file is of music playing but with 2 distinct tones in the background
%interfering with the music.
```

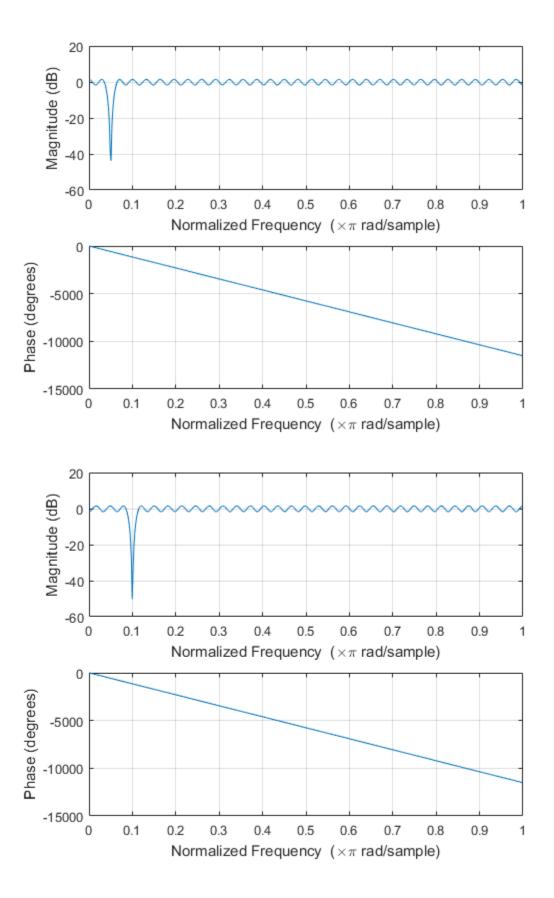
3.2

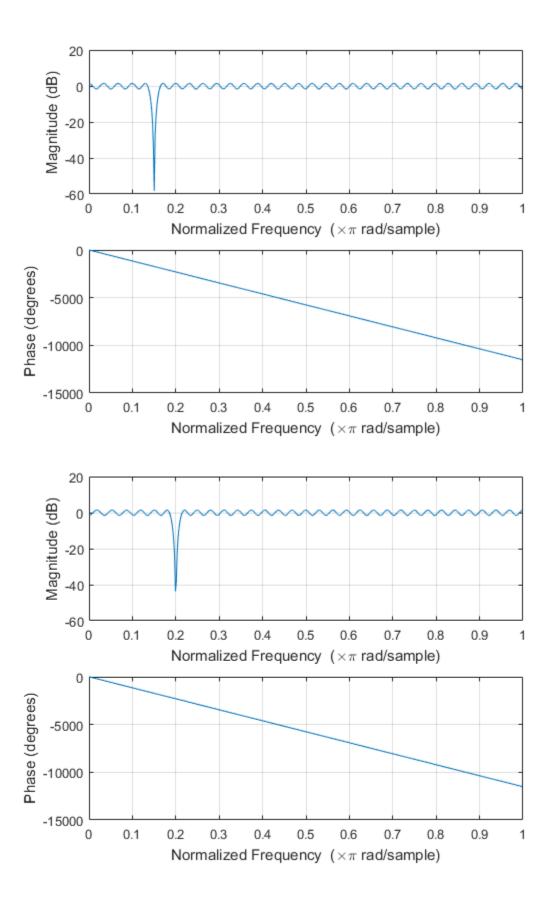
```
figure;
Y=fft(y);
nlen=length(Y);
f=(0:1/nlen:1-1/nlen)*FS;
stem(f,Y);
%THe noises are the peaks in the DFT.The two noise frequenies are 1102
2756HZ.
```

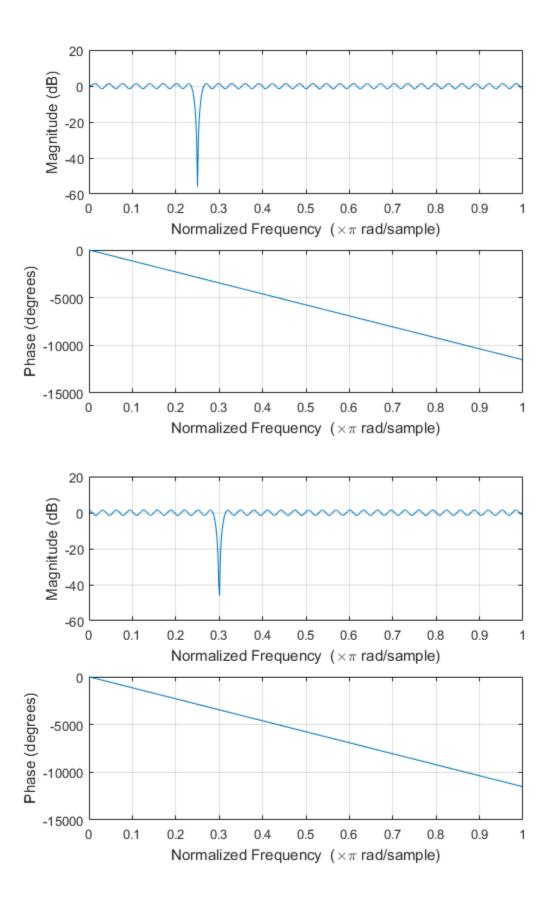
Warning: Using only the real component of complex data.

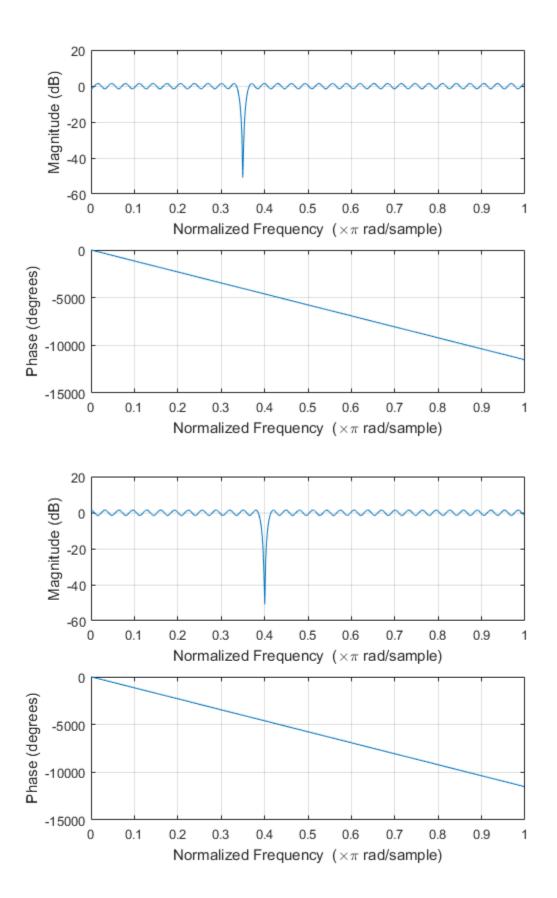


```
figure;
load filters;
freqz(h2,1);%notch frequency=0.05pirad/
sample*1cycle/2pirad*22050samples/sec=551Hz
figure;
freqz(h4,1);%notch frequency=1102.5Hz
figure;
freqz(h6,1);%notch frequency=1653.75Hz
figure;
freqz(h8,1);%notch frequency=2205Hz
figure;
freqz(h10,1);%notch freqeuncy=2756.25Hz
figure;
freqz(h12,1);%notch freqeuncy=3307.5Hz
figure;
freqz(h14,1);%notch freqeuncy=3858.75Hz
figure;
freqz(h16,1);%notch freqeuncy=4410Hz
These are bandstop fitlers since one of the frequenncies is
attenuated,
%but allows the others.
```









%h4 and h10 were chosen becuase it would attuenuate the noise.

3.5

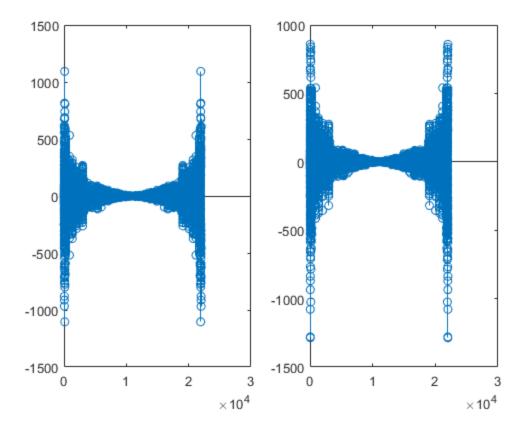
```
y1=conv(y,h4');
y1=conv(y1,h10');
sound(y1,FS);
% Method 1:Convoltuon of y and h4 and h10.

H4=fft(h4,297702);
H10=fft(h10,297702);
H4=H4';
H10=H10';

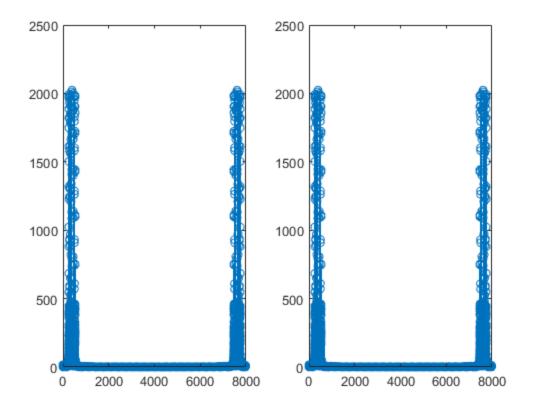
Yc=Y.*H4;
Yc=Yc.*H10;
sound(ifft(Yc),FS);
% Method 2: Multiplication of the music signal and the filter signal in
% frequncy domain; converted back to time signal inorder to play music.
```

```
figure;
Y1=fft(y1);
nleny1=length(Y1);
subplot(1,2,1);
f=(0:1/nleny1:1-1/nleny1)*FS;
stem(f,Y1);
%magnitude repsone of method 1; no more noise
nlenyc=length(Yc);
subplot(1,2,2);
f=(0:1/nlenyc:1-1/nlenyc)*FS;
stem(f,Yc);
%magnitude repsone of method 2; no more noise

Warning: Using only the real component of complex data.
Warning: Using only the real component of complex data.
```

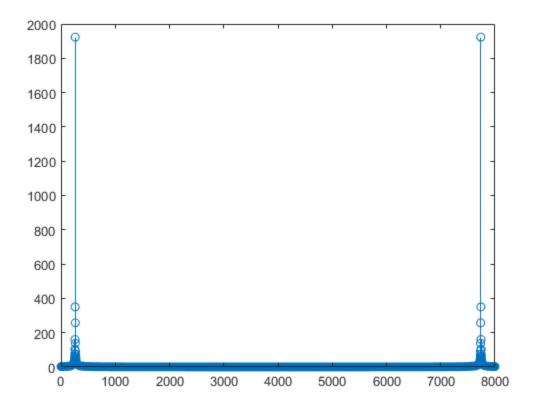


```
clear all;
close all;
[y1,FS1]=audioread('CScale.wav');
[y2,FS2]=audioread('CScaleZ.wav');
figure;
sound(y1,FS1);
sound(y2,FS2);
subplot(1,2,1);
Y1=fft(y1);
nlen1=length(Y1);
f=(0:1/nlen1:1-1/nlen1)*FS1;
stem(f,abs(Y1));
subplot(1,2,2);
Y2=fft(y2);
nlen2=length(Y2);
f=(0:1/nlen2:1-1/nlen2)*FS2;
stem(f,abs(Y2));
% The difference cant be told between the two plots.
```



There are 4002 points in ecah key.

```
window_length=4002;
x1=y1(1:4001);
x2=y1(4001:8002);
x3=y1(8002:12003);
x4=y1(12003:16004);
x5=y1(16004:20005);
x6=y1(20005:24006);
x7=y1(24006:28007);
x8=y1(28007:32008);
%magnitude response of key 1
figure;
x1fft=fft(x1);
nlen3=length(x1fft);
f=(0:1/nlen3:1-1/nlen3)*FS1;
stem(f,abs(x1fft));
%magnitude response shows only one peark of 261 Hz
```



```
x1fft=fft(x1,4002);
x2fft=fft(x2,4002);
x3fft=fft(x3,4002);
x4fft=fft(x4,4002);
x5fft=fft(x5,4002);
x6fft=fft(x6,4002);
x7fft=fft(x7,4002);
x8fft=fft(x8,4002);
```

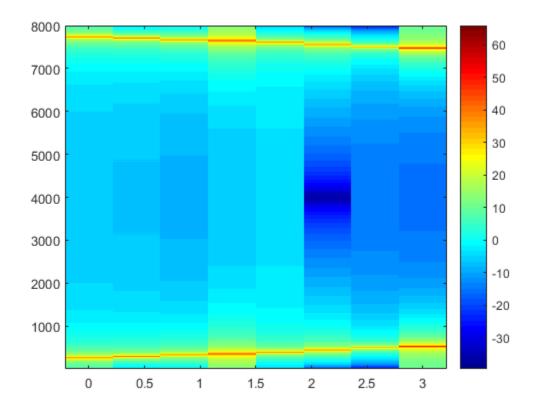
4.5 Already columns

4.6

```
spect=[x1fft,x2fft,x3fft,x4fft,x5fft,x6fft,x7fft,x8fft];
```

```
figure
spect_mag=20*log10(abs(spect));
t=(0:window_length:(length(y1)-window_length))/FS1;
f=(1:window_length)*FS1/window_length;
```

```
imagesc(t, f, spect_mag);
axis xy
colormap(jet)
colorbar
% It does match the frequncies as time goes on. The frequncies
increase
% according to Cscale.
```

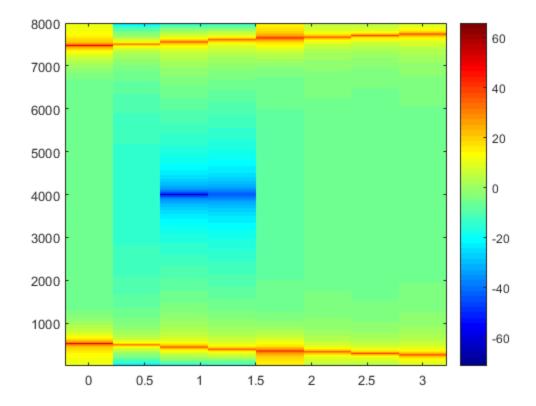


Cscalez

```
window_length=4002;
x1=y2(1:4001);
x2=y2(4001:8002);
x3=y2(8002:12003);
x4=y2(12003:16004);
x5=y2(16004:20005);
x6=y2(20005:24006);
x7=y2(24006:28007);
x8=y2(28007:32008);

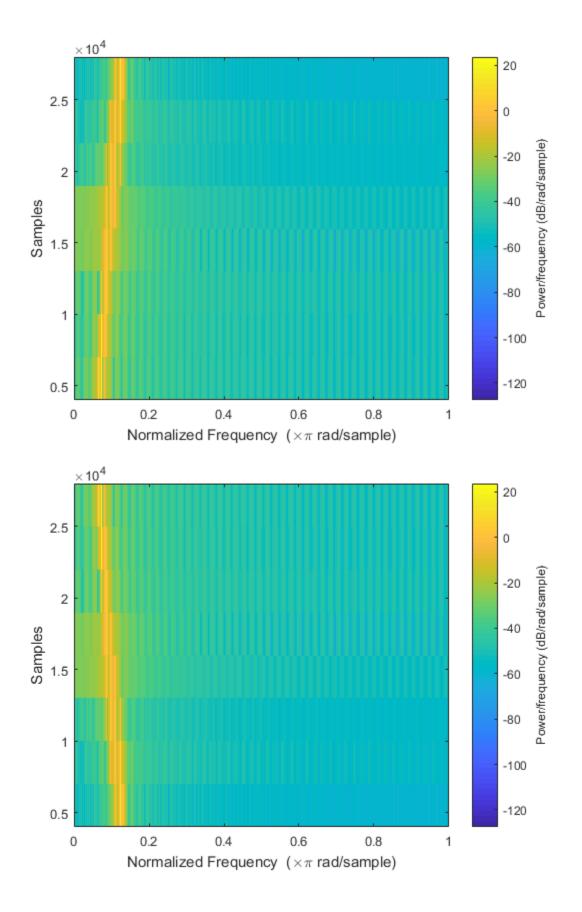
x1fft=fft(x1,4002);
x2fft=fft(x2,4002);
x3fft=fft(x4,4002);
x4fft=fft(x5,4002);
x5fft=fft(x6,4002);
```

```
x7fft=fft(x7,4002);
x8fft=fft(x8,4002);
spect=[x1fft,x2fft,x3fft,x4fft,x5fft,x6fft,x7fft,x8fft];
figure
spect_mag1=20*log10(abs(spect));
t=(0:window_length:(length(y2)-window_length))/FS2;
f=(1:window_length)*FS2/window_length;
imagesc(t, f, spect_mag1);
axis xy
colormap(jet)
colorbar
%The Cscalez looks correct, backwards of cscale.
```



5

```
figure
spectrogram(y1, [], 8000);
figure
spectrogram(y2, [], 8000);
%The x and y axis are switched. The frequncy axis is normailized.
```



Conclusion

The lab went as suspected. THe lab taught use how to normalize frequncies, the Hermitian symetry of DFT, how changing the number of points of DFT enchances the resolution of DFT, and the purpose of the spectogram.

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