EECS 330 LABORATORY

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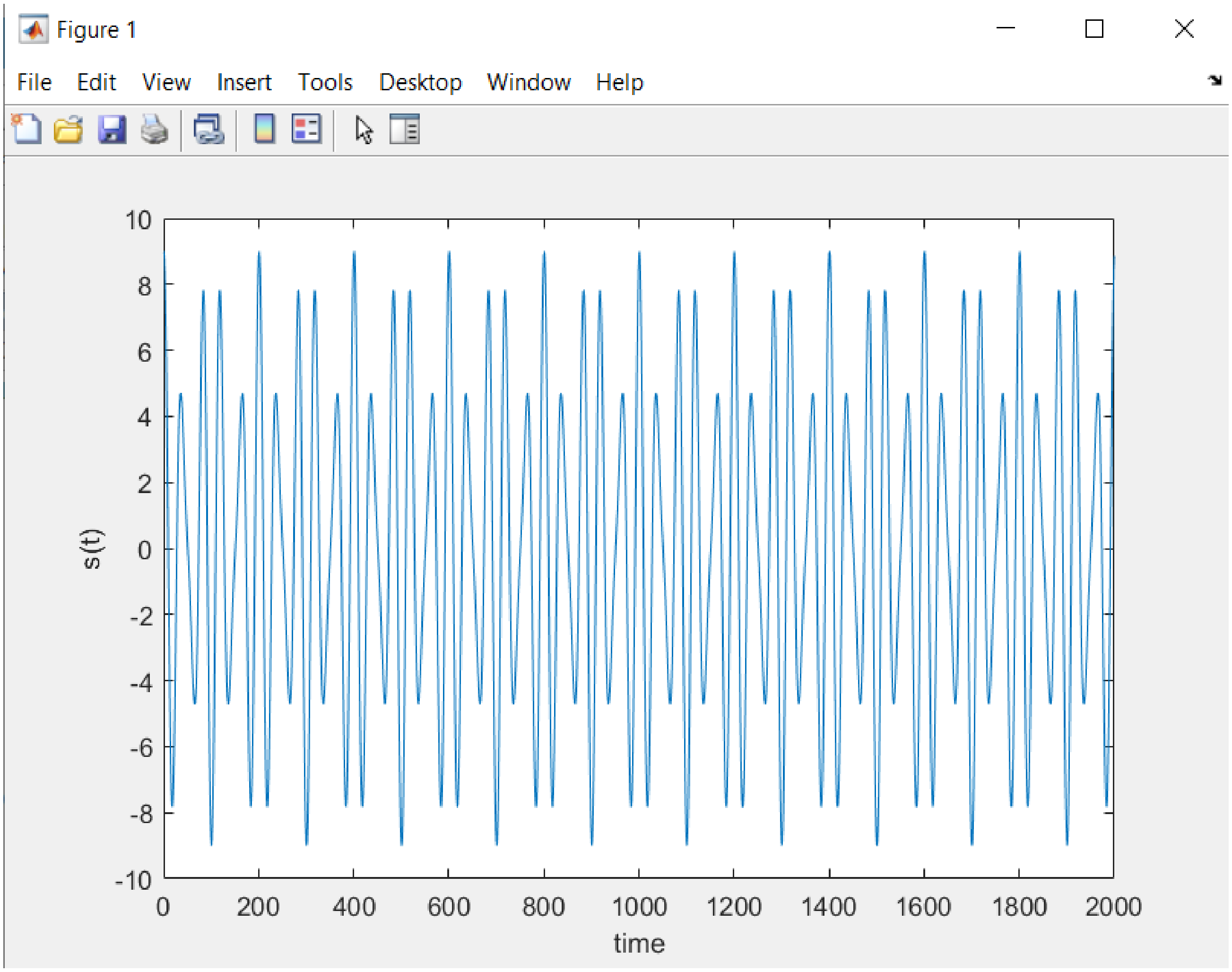
EXPERIMENT : Applications of Fourier Transform

Q1.

a)
In this question, we have assumed Sampling frequency to be equal to 2000Hz. Other parameters are shown in the code snippet used below. All the quantities are in their SI units.

```
% Sample the sinusoid s(t) = 3cos(2 pi f1 t)+ 6cos(2 pi f2 t),
%where f1 = 70 Hz,  f2=50 Hz, we assume 2000Hx to be the sampling frequency


fs = 2000; %sampling frequency
t = 0 : 1/fs : 1 - 1/fs; %specifying the range
f1 = 70;
f2 = 50;
a1 = 3;
a2 = 6;
s = a1*cos(2*pi*f1*t) + a2*cos(2*pi*f2*t) ;
plot(s)
ylim([-10,10])
ylabel('s(t)')
xlabel('time')
```

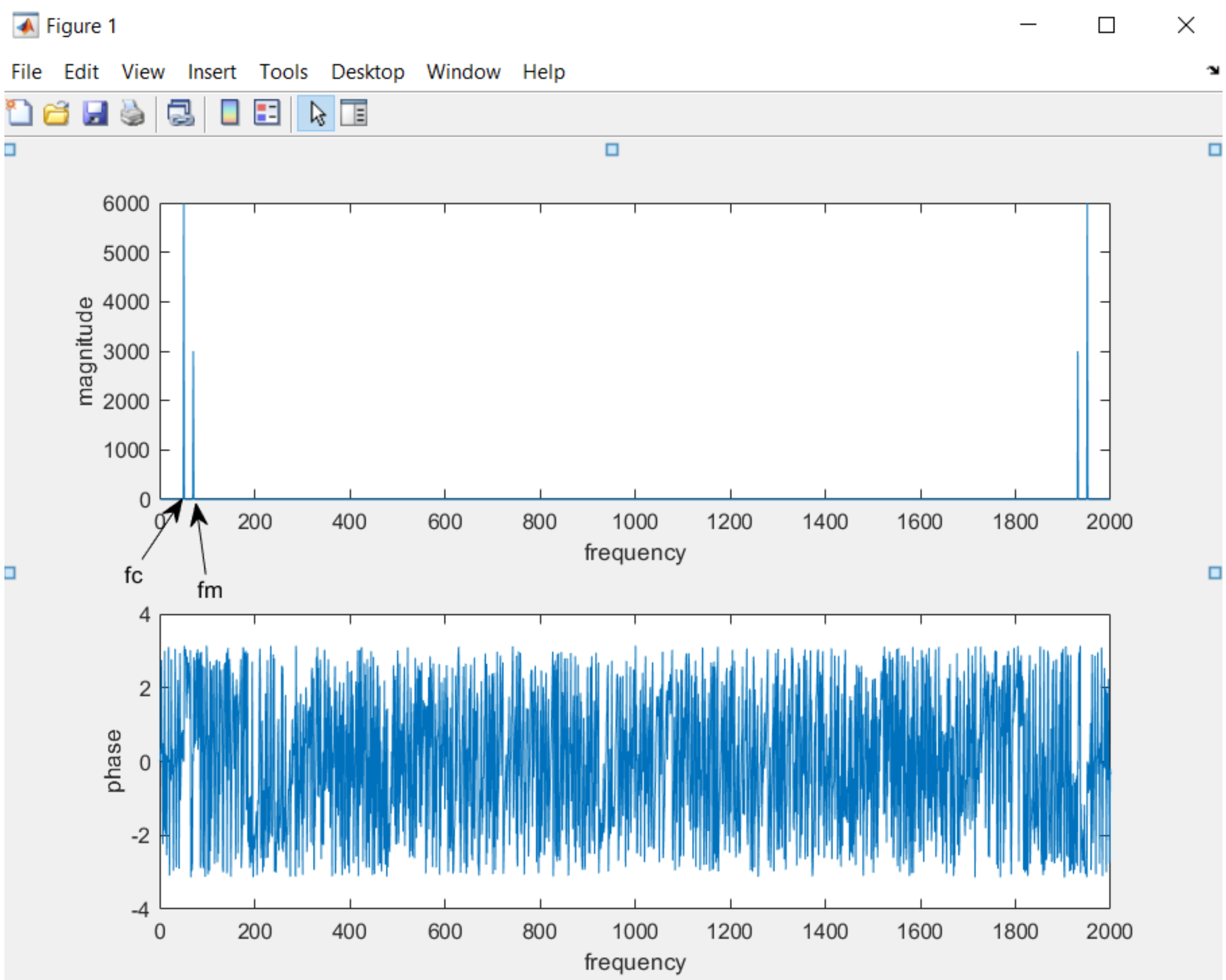
The plot is shown below :



b) We need to apply Fourier transform to this signal and plot its magnitude and phase spectrum

The following code snippet calculates the Fourier transform and plots the magnitude and phase spectrum in the plot below:

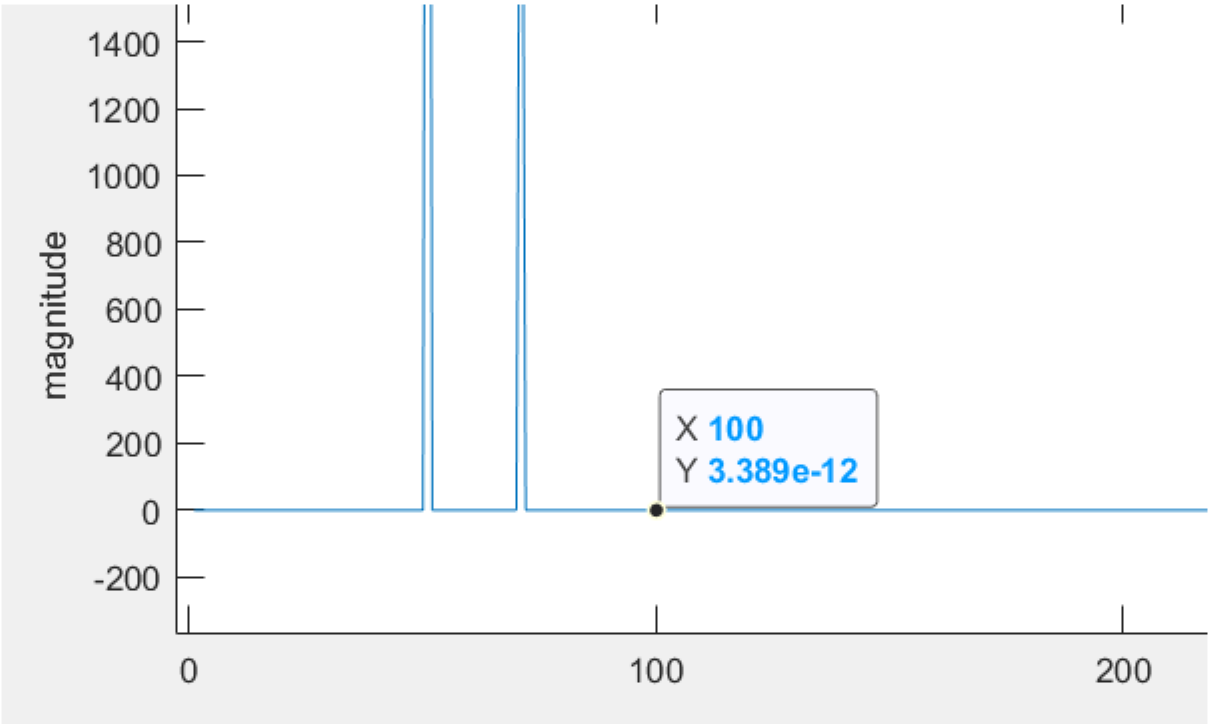
```
fs = 2000; %sampling frequency
t = 0 : 1/fs : 1 - 1/fs;
f1 = 70;
f2 = 50;
a1 = 3;
a2 = 6;
s = a1*cos(2*pi*f1*t) + a2*cos(2*pi*f2*t) ;

S = fft(s);

length(s); %2000 bins in frequency domain


subplot(2,1,1);
S_magnitude = abs(S);
plot(S_magnitude);
ylabel('magnitude')
xlabel('frequency')

subplot(2,1,2)
S_phase = angle(S);
plot(S_phase);
ylabel('phase')
xlabel('frequency')
```



The Fast Fourier Transform is basically a way of breaking down a signal into its frequency components. So if one created individual sinusoidal waves of each frequency present in the FFT, and each had an amplitude and phase shift given by the FFT for that frequency, then we could sum them up to get the original signal (which is the fourier series by definition).

For frequencies other than f1 and f2 , like for ex 100Hz, the magnitude spectrum value is shown below with the graph

```
>> S_magnitude(100)

ans =

   3.3893e-12
```



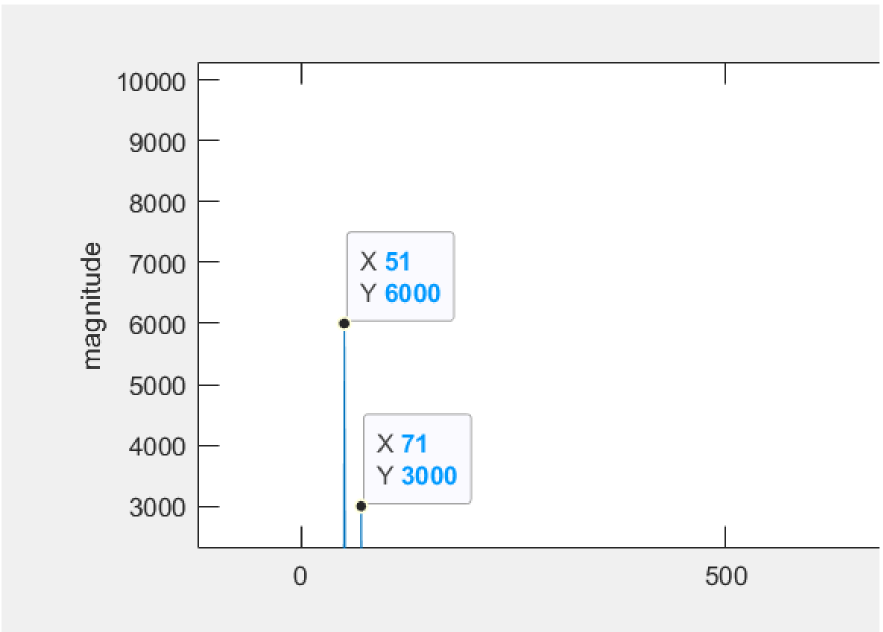
The phase spectrum for the same can be found below as :

```
>> S_phase(100)

ans =

    2.6867
```

Interpretation:

No, the phase spectrum at points other than f1 and f2 is not very reliable because the phase spectrum is noisy stemming from the fact that using the FFT function computes inverse tangents from the ration of imaginary part to real part of the FFT result, hence there is a chance that even a small decimal error made while rounding off might amplify and manifest as an incorrect phase information.

c) The following is the graph and the code snippet




```
>> S_phase(51)

ans =

   1.0275e-14
```

```
>> S_phase(71)

ans =

  -3.1859e-15
```

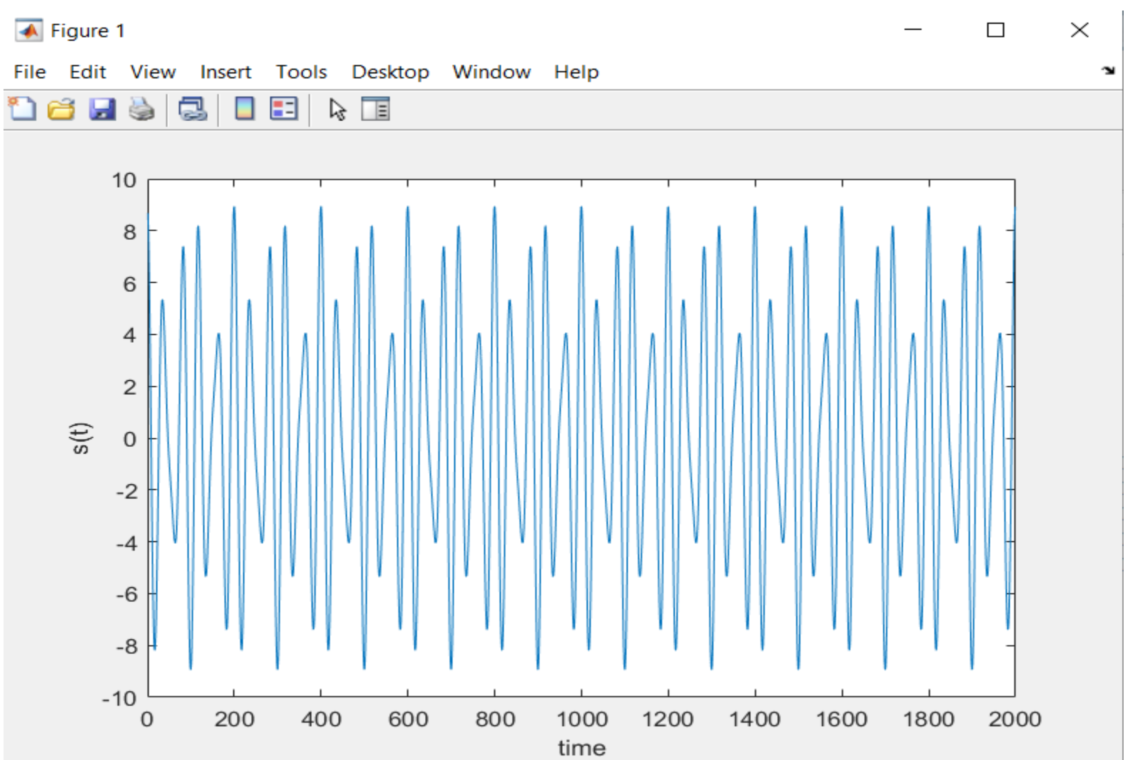
For $\theta_1 = \theta_2 = 0$, the phase is not defined:

```
>> S_phase(0)
Array indices must be positive integers or logical values.
```

d) We repeat the above steps for non zero values of θ_1 and θ_2

We have assumed $\theta_1 = 0.2$ and $\theta_2 = 0.3$

```
fs = 2000; %sampling frequency
t = 0 : 1/fs : 1 - 1/fs; %specifying the range
f1 = 70;
f2 = 50;
a1 = 3;
a2 = 6;
s = a1*cos(2*pi*f1*t + 0.2) + a2*cos(2*pi*f2*t + 0.3) ;
plot(s)
ylim([-10,10])
ylabel('s(t)')
xlabel('time')
```



```
fs = 2000; %sampling frequency
t = 0 : 1/fs : 1 - 1/fs;
f1 = 70;
f2 = 50;
a1 = 3;
a2 = 6;
s = a1*cos(2*pi*f1*t + 0.2) + a2*cos(2*pi*f2*t + 0.3) ;

S = fft(s);

length(s); %2000 bins in frequency domain


subplot(2,1,1);
S_magnitude = abs(S);
plot(S_magnitude);
ylabel('magnitude')
xlabel('frequency')

subplot(2,1,2)
S_phase = angle(S);
plot(S_phase);
ylabel('phase')
xlabel('frequency')
```

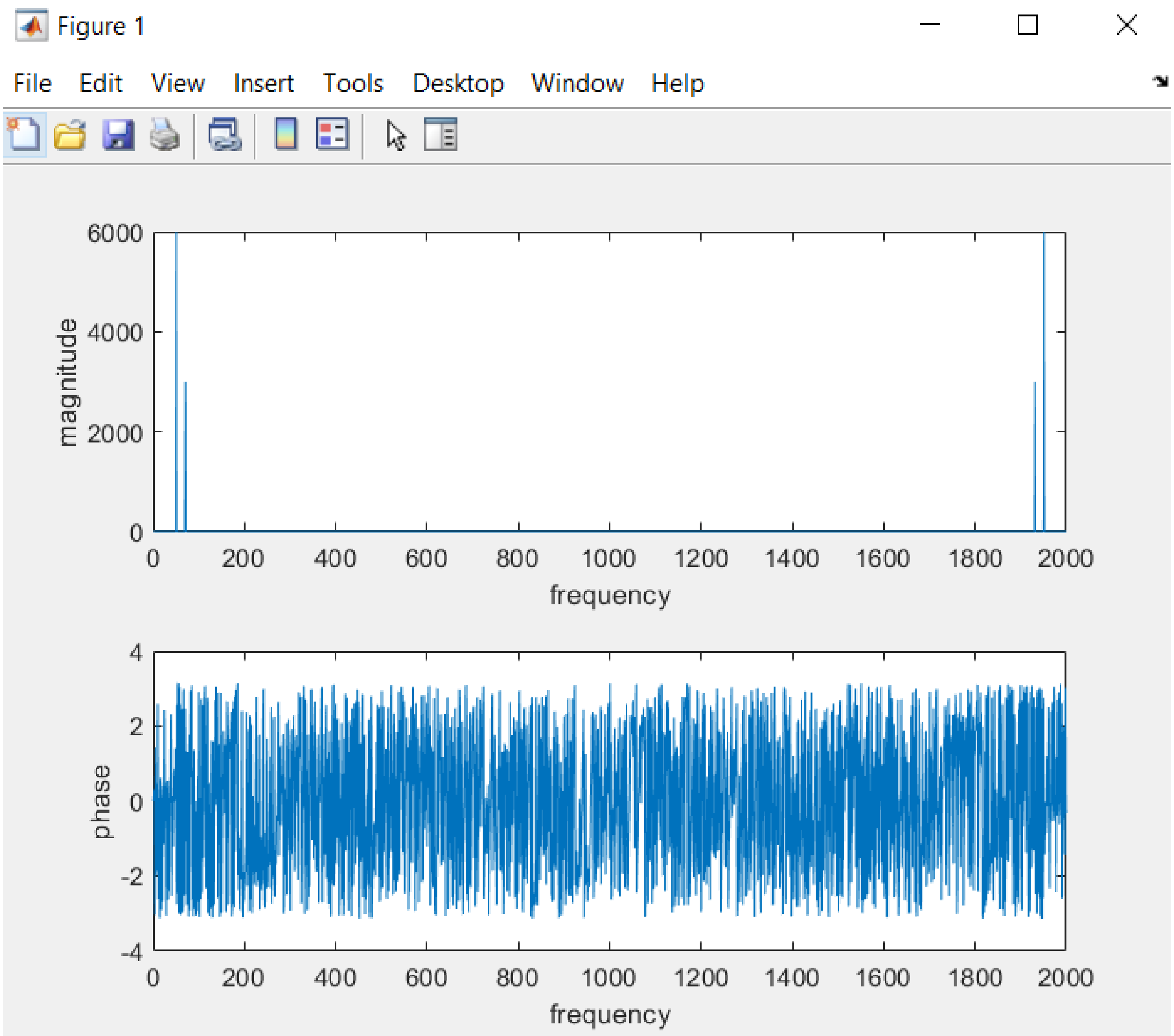
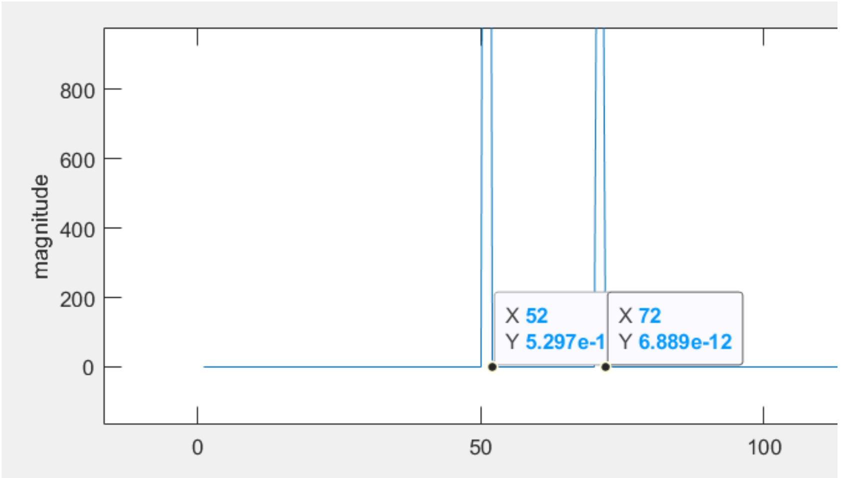


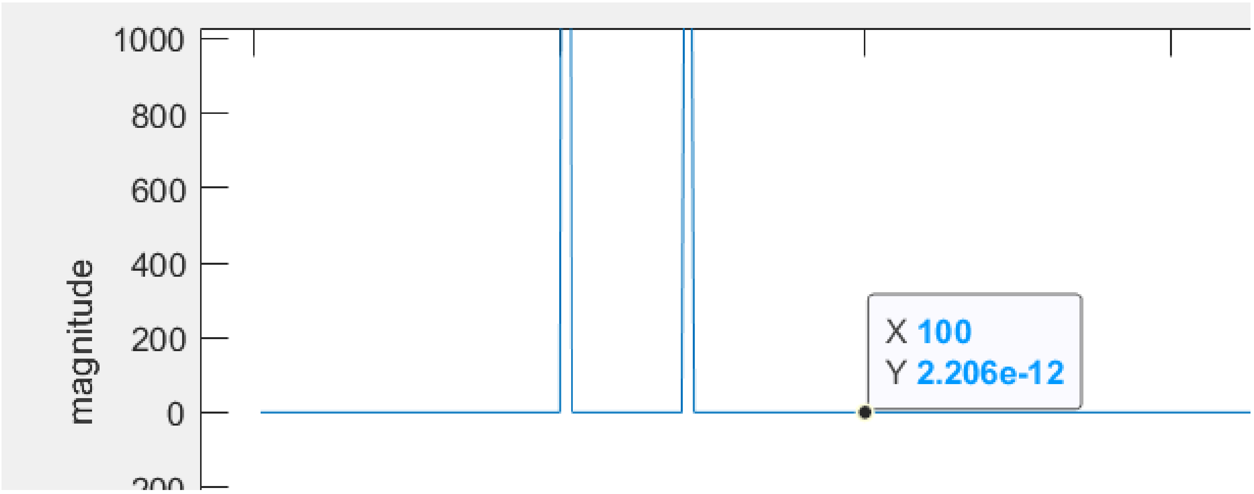
Figure 1
File Edit View Insert Tools Desktop Window Help
magnitude
6000
4000
2000
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
frequency
phase
4
2
0
-2
-4
0 200 400 600 800 1000 1200 1400 1600 1800 2000
frequency




```
>> S_phase(52)

ans =

   -2.7534
```

```
>> S_phase(72)

ans =

    1.3251
```

```
>> S_magnitude(100)

ans =

   2.2060e-12
```



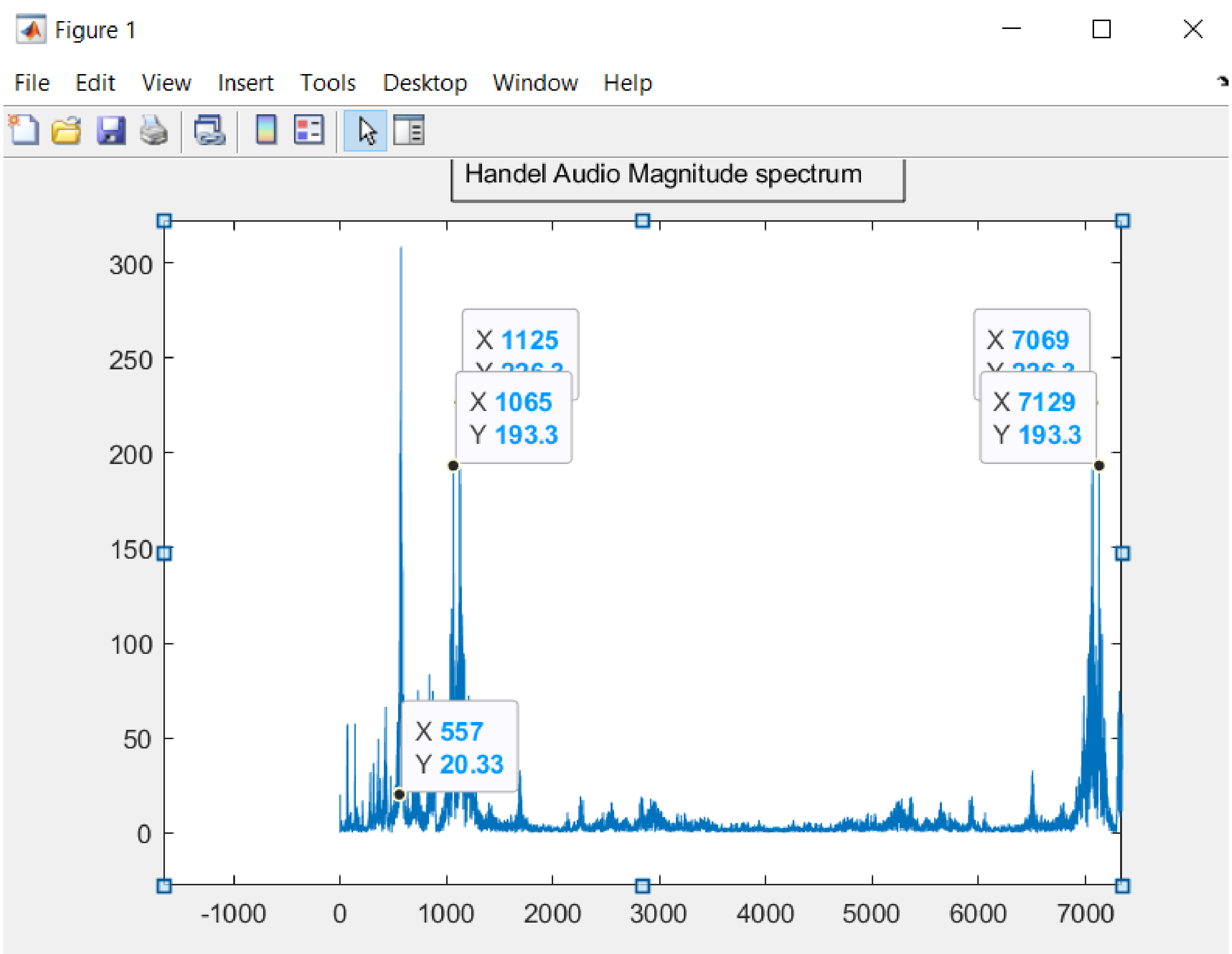
The phase cannot be found at the points 0.2 and 0.3 as they are values of θ_1 and θ_2 .

Q2)

a) The following are the codes and graphs for the reading and plotting the magnitude spectrum of three in-built audio signals in matlab, namely the handel, gong and train.

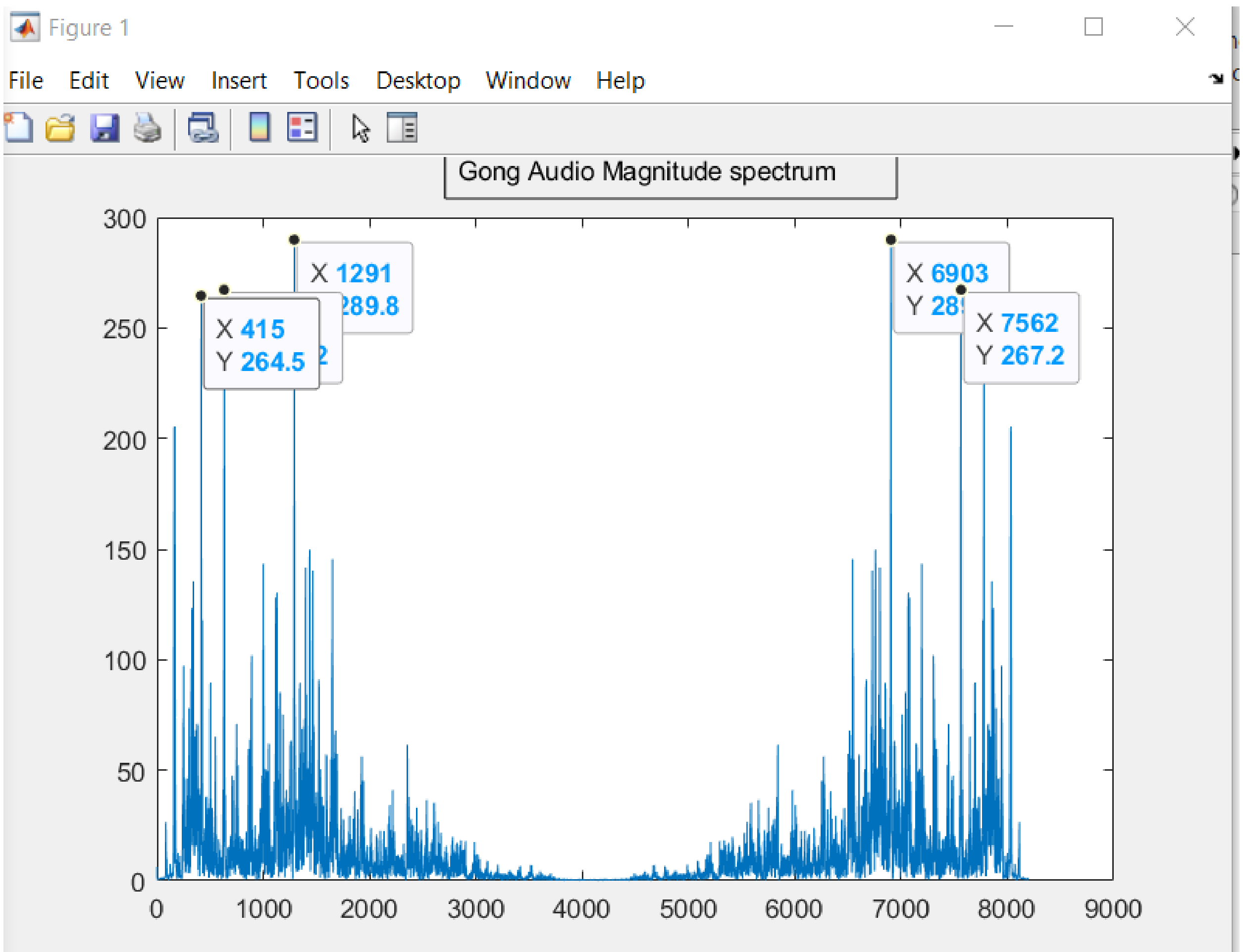
Audiosig function:

```
load handel.mat
audiowrite('audio1.wav',y,Fs);
Y=fft(y,Fs);
Y_mag = abs(Y);
plot(Y_mag)
```



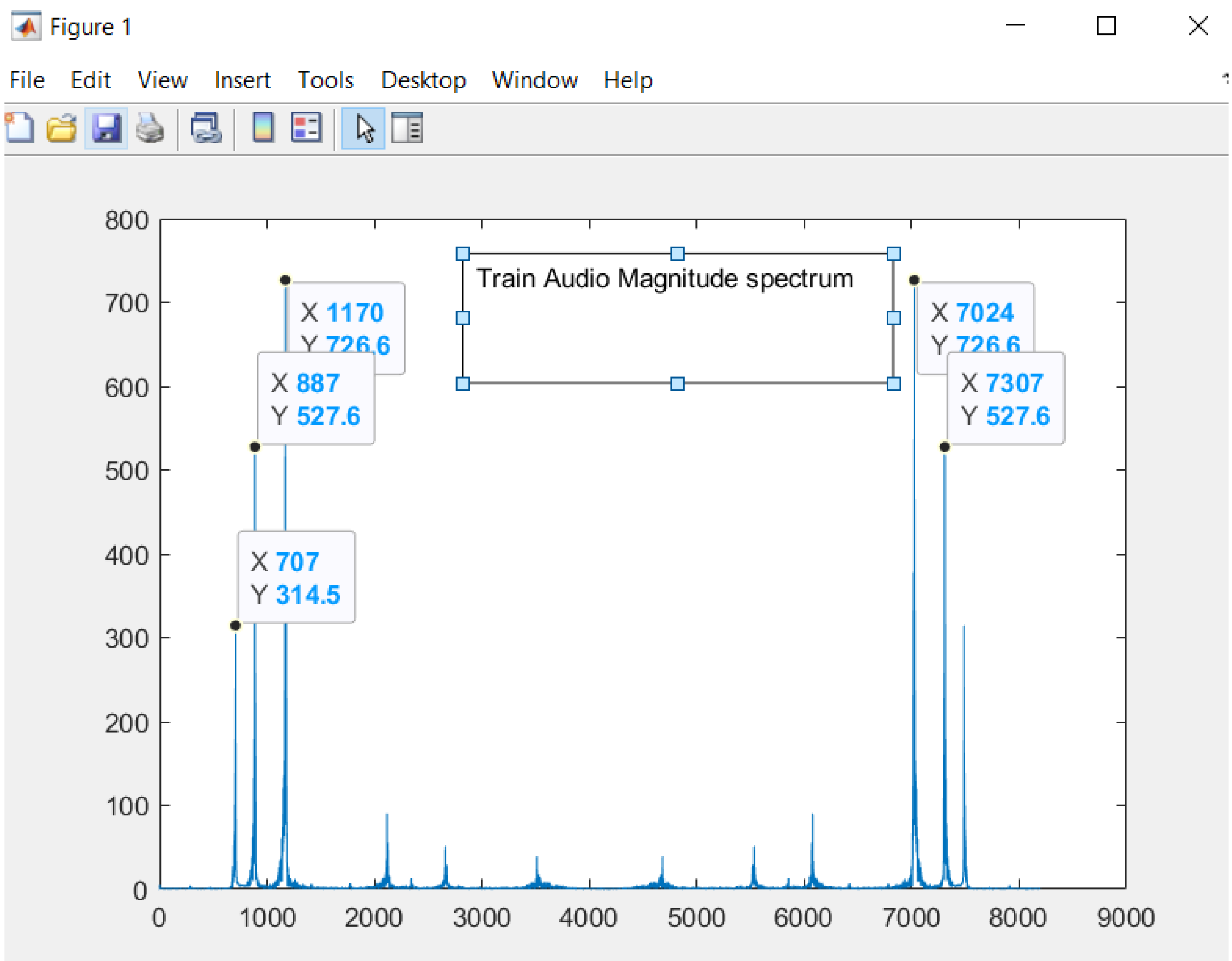
Fs	8192
y	*73113x1 double*

load gong.mat
audiowrite('audio2.wav',y,Fs);
Y=fft(y,Fs);
Y_mag = abs(Y);
plot(Y_mag)



Fs	8192
y	*42028x1 double*
Y	*8192x1 complex ...*
Y_mag	*8192x1 double*

```
load train.mat
audiowrite('audio3.wav',y,Fs);
Y=fft(y,Fs);
Y_mag = abs(Y);
plot(Y_mag)
```



Fs	8192
y	*12880x1 double*
Y	*8192x1 complex ...*
Y_mag	*8192x1 double*

b) The five most prominent frequencies in each of the above audio signals are labelled on the plots (X-coordinates). They are listed below (they correspond to the prominent peaks of the curves)

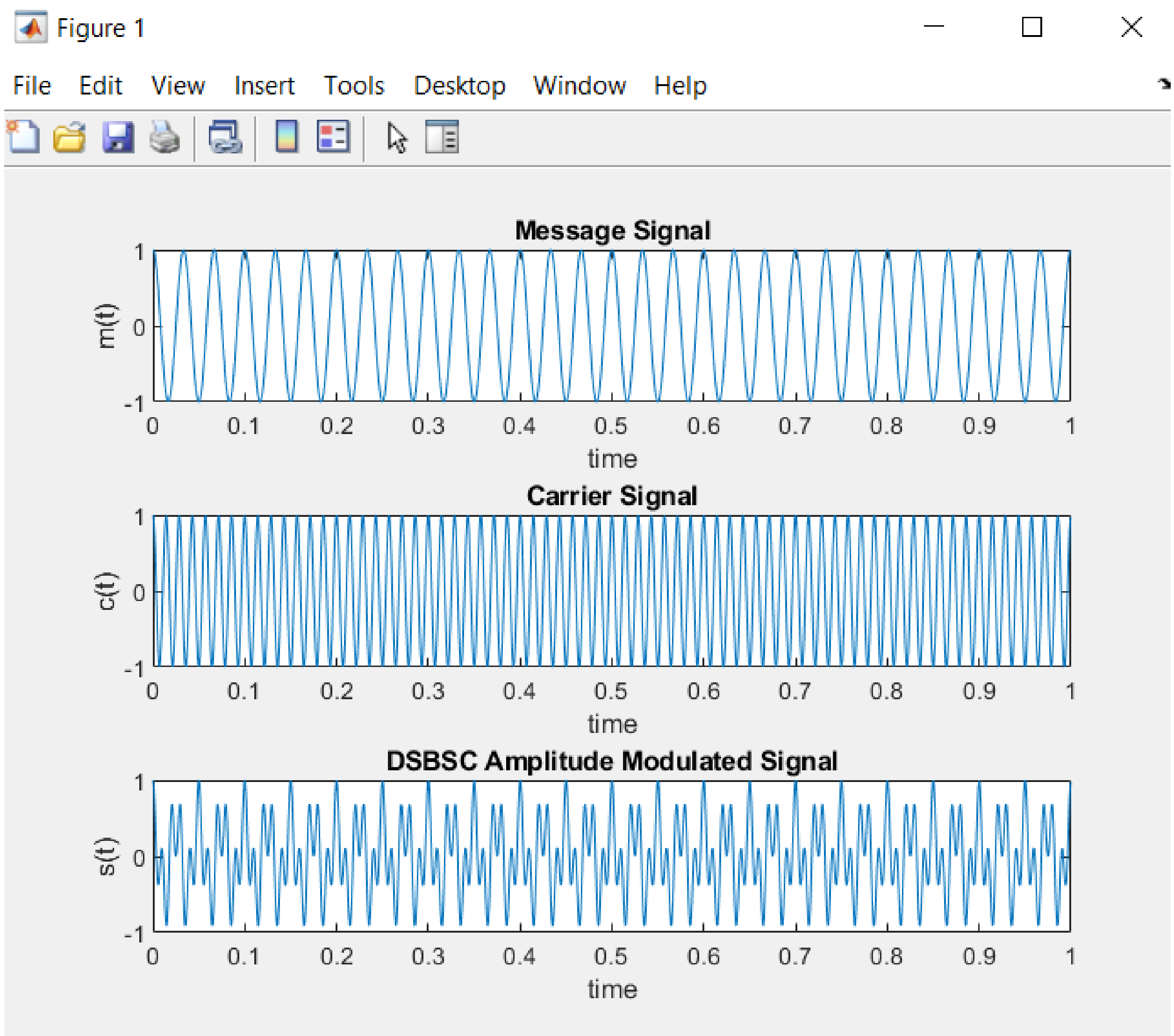
Handel: 557, 1065 , 1125 , 7069 , 7129
Gong: 415, 632 , 1291, 6903 , 7562
Train: 707, 887 , 1170, 7024, 7307

As seen from the graphs, the gong has a precise pitch, loud and resonant sound ,the sound overshoots and damps after a while. The train noise is on the contrary arbitrary, making rolling sounds in the track.

Q3.

a) The chosen parameters have been mentioned in the code snippet below.

```
fs = 2000;
fc = 70;
fm = 30;  % fm < fc

l = 1/fs;
t = 0: l : 1-l;

m = cos(2*pi*fm*t);
subplot (3,1,1);
plot (t,m);
xlabel('time');
ylabel('m(t)');
title('Message Signal');

c = cos(2*pi*fc*t);
subplot (3,1,2);
plot (t,c);
xlabel('time');
ylabel('c(t)');
title('Carrier Signal');
```



b)

```
fs = 2000;
fc = 70;
fm = 30;  % fm < fc

l = 1/fs;
```

```
t = 0: I : 1-I;

m = cos(2*pi*fm*t);
c = cos(2*pi*fc*t);
s = c.*m;

M = fft(m);
M_magnitude = abs(M);

C = fft(c);
C_magnitude = abs(C);

S = fft(s);
S_magnitude = abs(S);

subplot(3,1,1);
plot(M_magnitude);
ylabel('M(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of message signal m(t)');

subplot(3,1,2);
plot(C_magnitude);
ylabel('C(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of carrier signal c(t)');

subplot(3,1,3);
plot(S_magnitude);
ylabel('S(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of DSBSC signal s(t)');
```

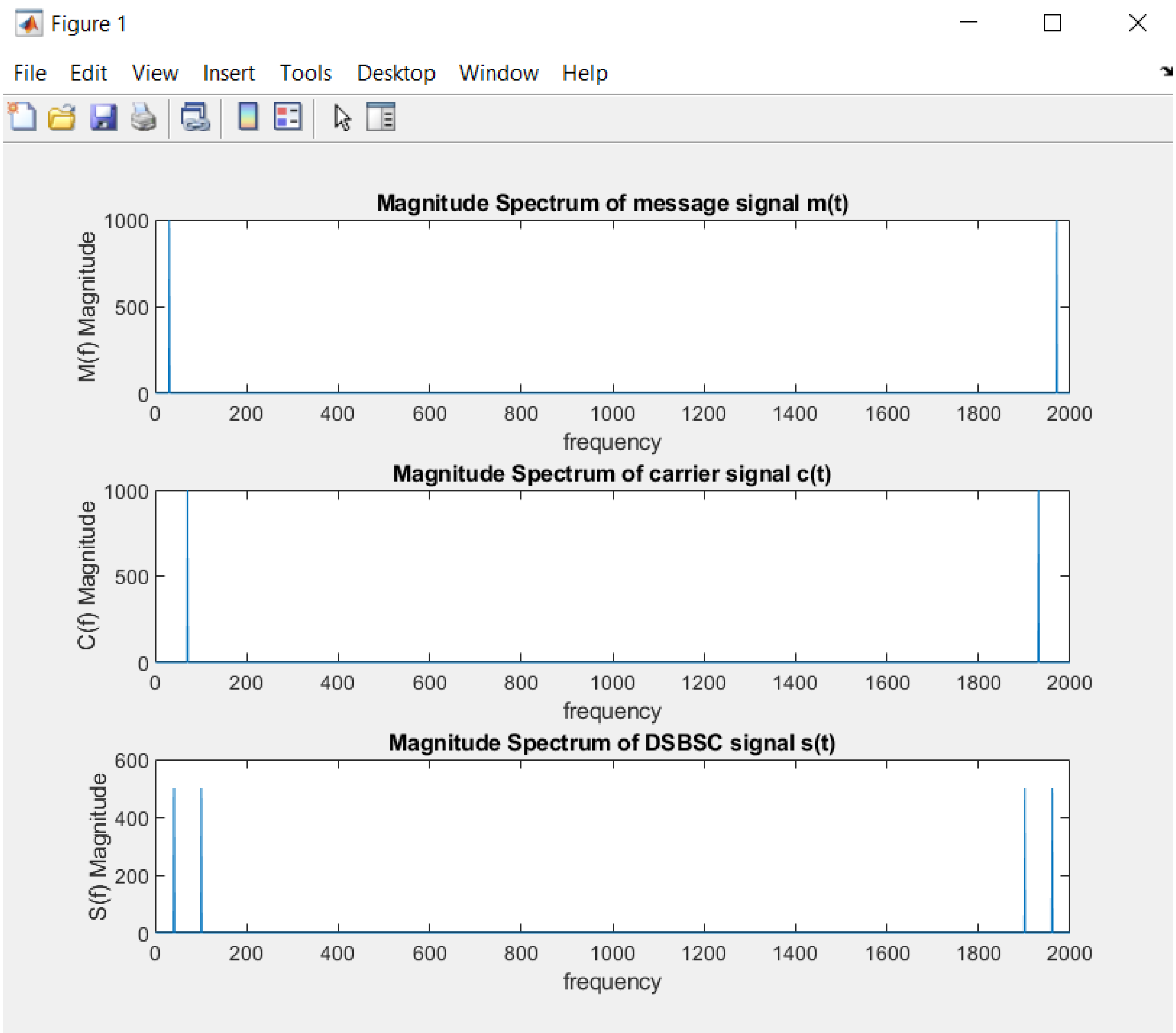


Figure 1
File Edit View Insert Tools Desktop Window Help
Magnitude Spectrum of message signal m(t)
M(f) Magnitude
frequency
Magnitude Spectrum of carrier signal c(t)
C(f) Magnitude
frequency
Magnitude Spectrum of DSBSC signal s(t)
S(f) Magnitude
frequency

c)

```
fs = 2000;
fc = 70;
fm = 30;  % fm < fc

l = 1/fs;
t = 0: l : 1-l;

m = cos(2*pi*fm*t);
c = cos(2*pi*fc*t);
s = c.*m;

coherent_output = s.*(cos(2*pi*fc*t));

COH = fft(coherent_output);
COH_magnitude = abs(COH);

subplot(2,1,1);
plot(t,coherent_output);
ylabel('Coherent Detector Output');
xlabel('time');
title('time domain');

subplot(2,1,2);
plot(COH_magnitude);
ylabel('Magnitude Spectrum of Coherent Detector Output');
xlabel('Frequency');
title('frequency domain');
```

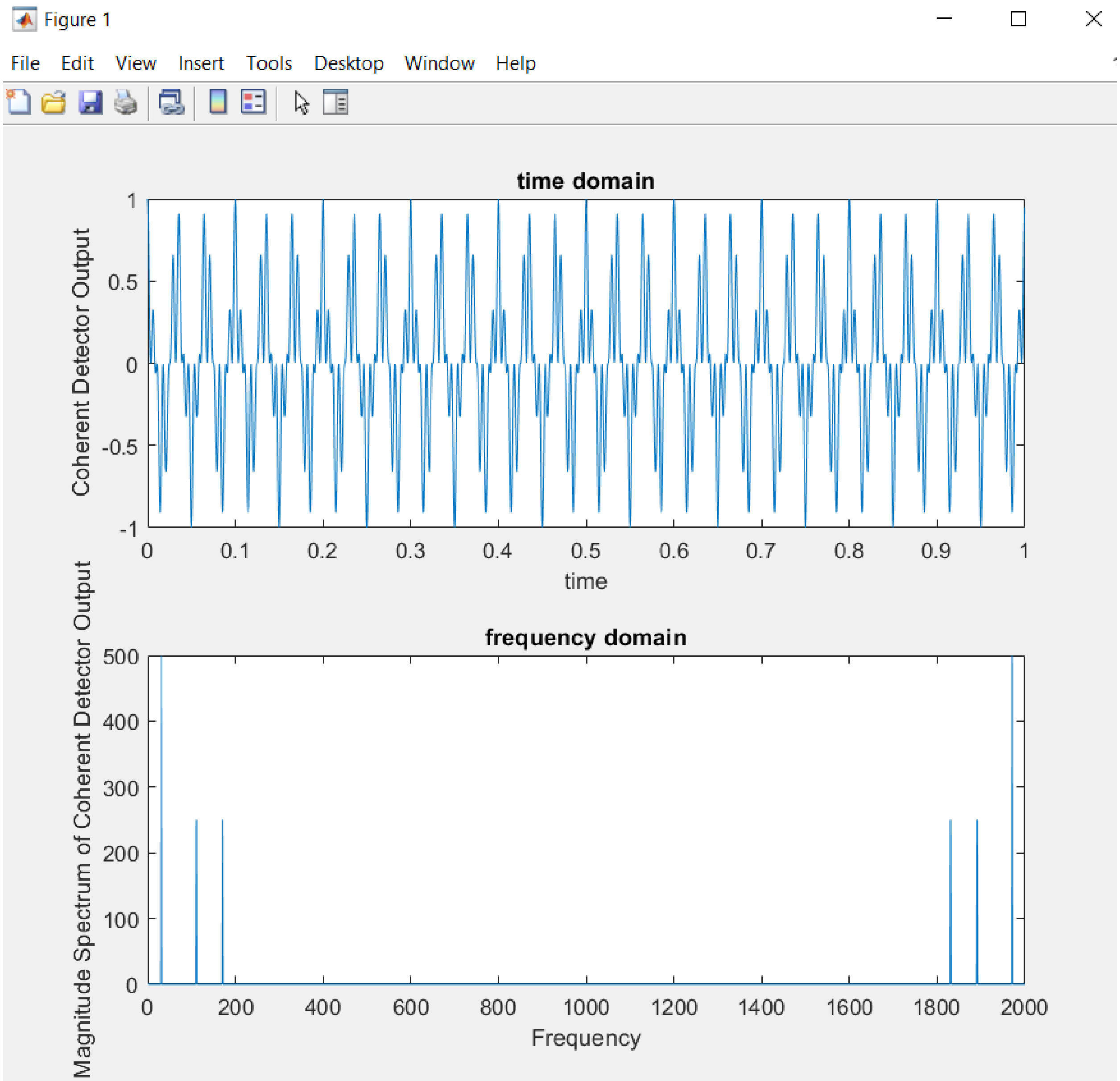


Figure 1
File Edit View Insert Tools Desktop Window Help
time domain
Coherent Detector Output
time
frequency domain
Magnitude Spectrum of Coherent Detector Output
Frequency

d)

In the case where $f_m > f_c$, **yes**, we observe a higher amount of distortion in the output of the coherent detector. This has huge repercussions in the design of antennas or other long distance communication protocols because we require carrier waves of high frequency owing to the fact that high frequency carrier waves do not require a material medium to propagate.

```
fs = 2000;
fc = 30;
fm = 70;  % fm > fc

l = 1/fs;
t = 0: l : 1-l;

m = cos(2*pi*fm*t);
subplot (3,1,1);
plot (t,m);
xlabel('time');
ylabel('m(t)');
title('Message Signal');

c = cos(2*pi*fc*t);
subplot (3,1,2);
plot (t,c);
xlabel('time');
ylabel('c(t)');
title('Carrier Signal');

s = c.*m;
subplot (3,1,3);
plot (t,s);
xlabel('time');
ylabel('s(t)');
title('DSBSC Amplitude Modulated Signal');
```

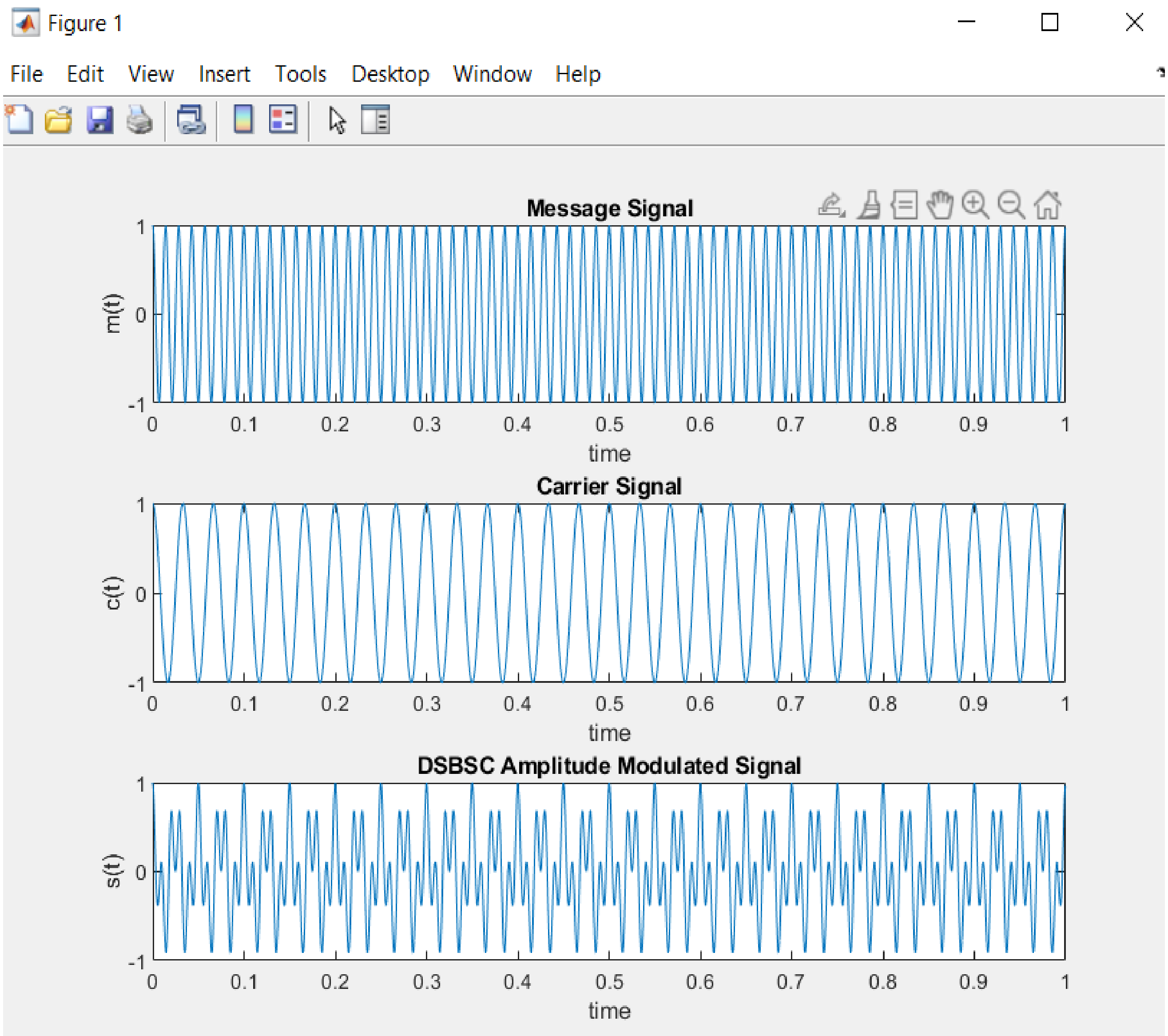


Figure 1
File Edit View Insert Tools Desktop Window Help
Message Signal
m(t)
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
time
Carrier Signal
c(t)
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
time
DSBSC Amplitude Modulated Signal
s(t)
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
time

```
fs = 2000;
fc = 30;
fm = 70;  % fm > fc

I = 1/fs;
t = 0: I : 1-I;

m = cos(2*pi*fm*t);
c = cos(2*pi*fc*t);
s = c.*m;

M = fft(m);
M_magnitude = abs(M);

C = fft(c);
C_magnitude = abs(C);

S = fft(s);
S_magnitude = abs(S);

subplot(3,1,1);
plot(M_magnitude);
ylabel('M(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of message signal m(t)');

subplot(3,1,2);
plot(C_magnitude);
ylabel('C(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of carrier signal c(t)');

subplot(3,1,3);
plot(S_magnitude);
ylabel('S(f) Magnitude');
xlabel('frequency');
title('Magnitude Spectrum of DSBSC signal s(t)');
```

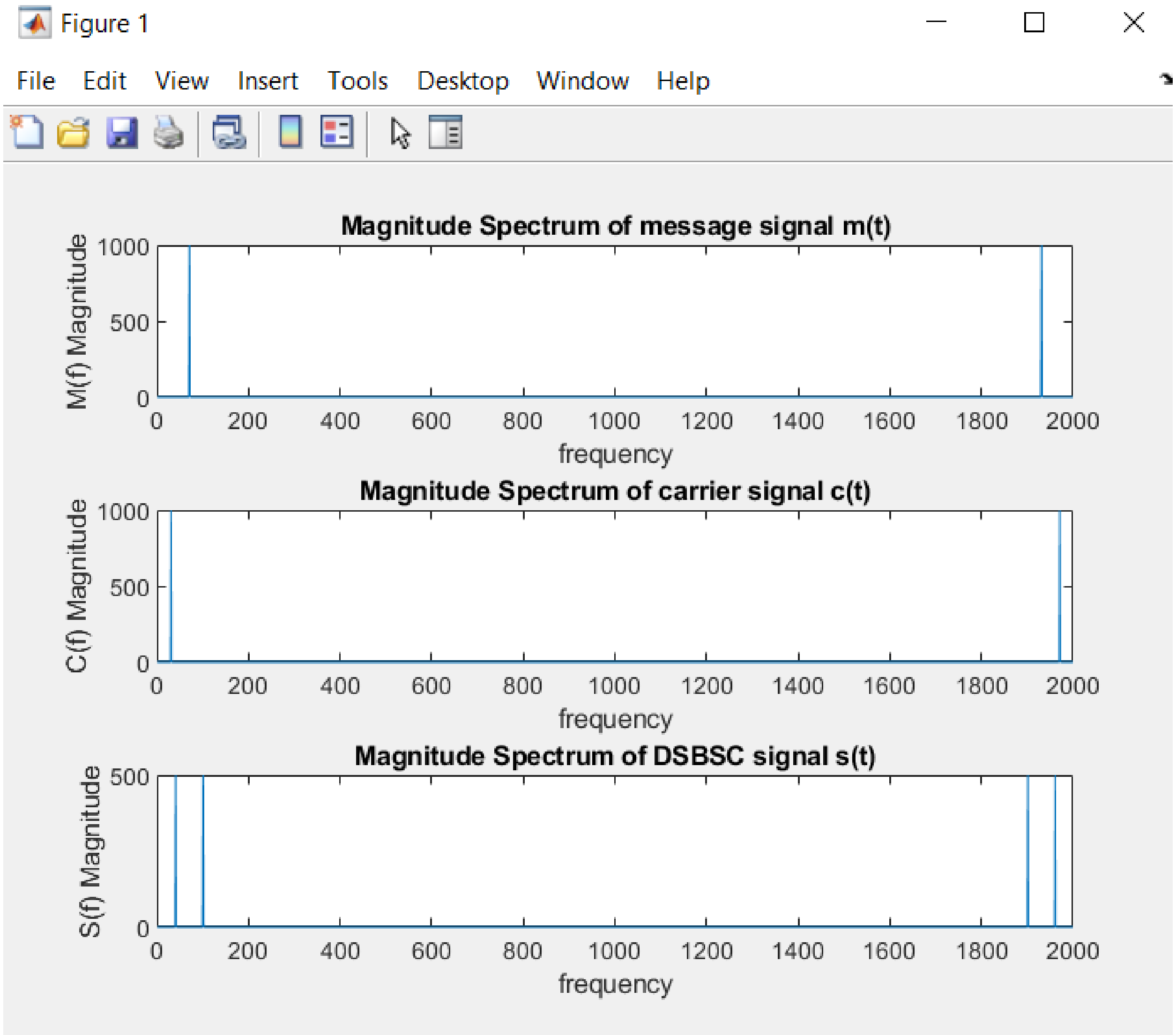


Figure 1
File Edit View Insert Tools Desktop Window Help
Magnitude Spectrum of message signal m(t)
M(f) Magnitude
1000
500
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
frequency
Magnitude Spectrum of carrier signal c(t)
C(f) Magnitude
1000
500
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
frequency
Magnitude Spectrum of DSBSC signal s(t)
S(f) Magnitude
500
0
0 200 400 600 800 1000 1200 1400 1600 1800 2000
frequency

```
fs = 1000;
fc = 30;
fm = 70;  % fm > fc

l = 1/fs;
t = 0: l : 1-l;

m = cos(2*pi*fm*t);
c = cos(2*pi*fc*t);
s = c.*m;

coherent_output = s.*(cos(2*pi*fc*t));

CoherentO = fft(coherent_output);
COH_magnitude = abs(CoherentO);

subplot(2,1,1);
plot(t,coherent_output);
ylabel('Coherent Detector Output');
xlabel('time');
title('time domain');

subplot(2,1,2);
plot(COH_magnitude);
ylabel('Magnitude Spectrum of Coherent Detector Output');
xlabel('Frequency');
title('frequency domain');
```

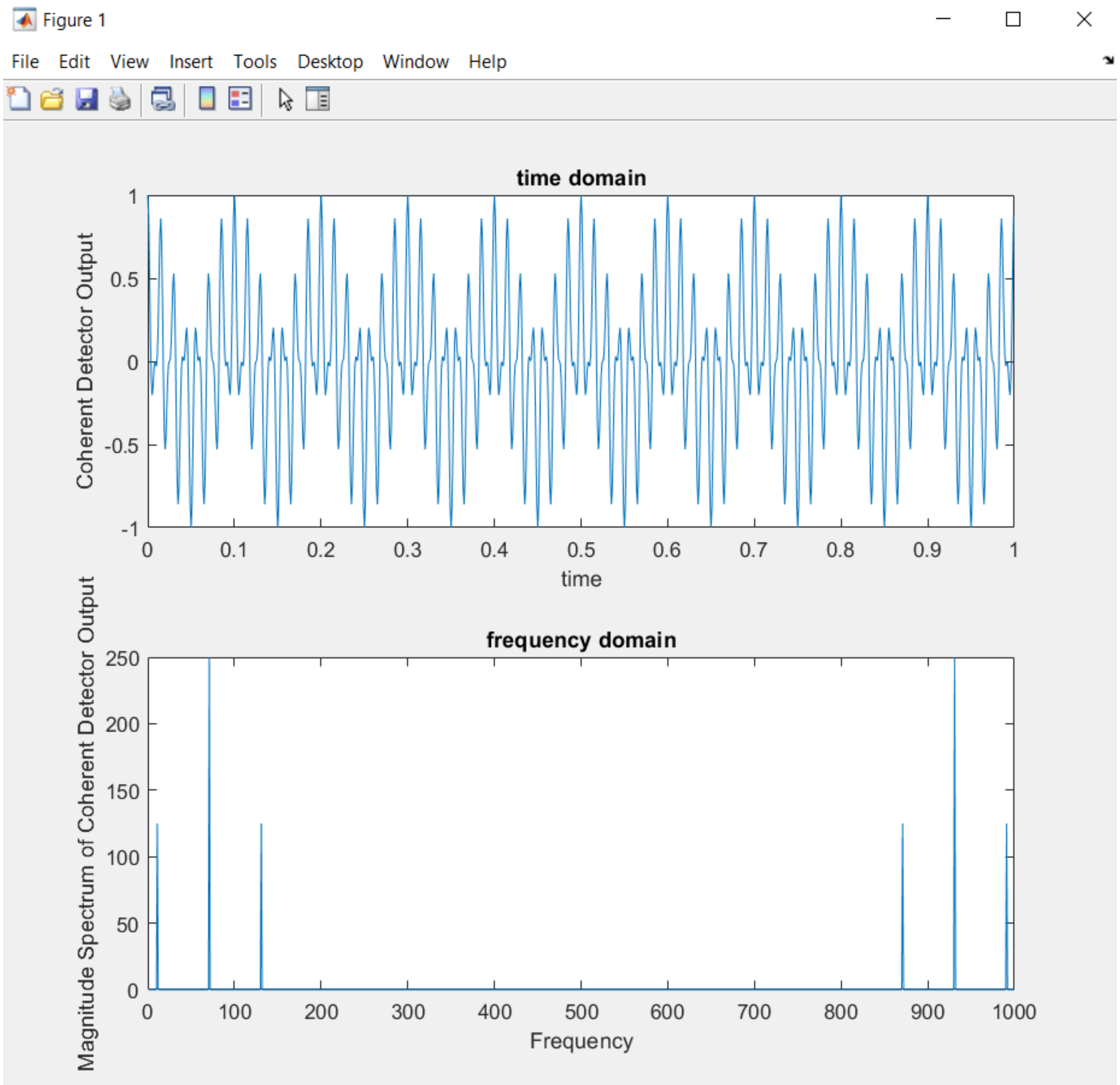
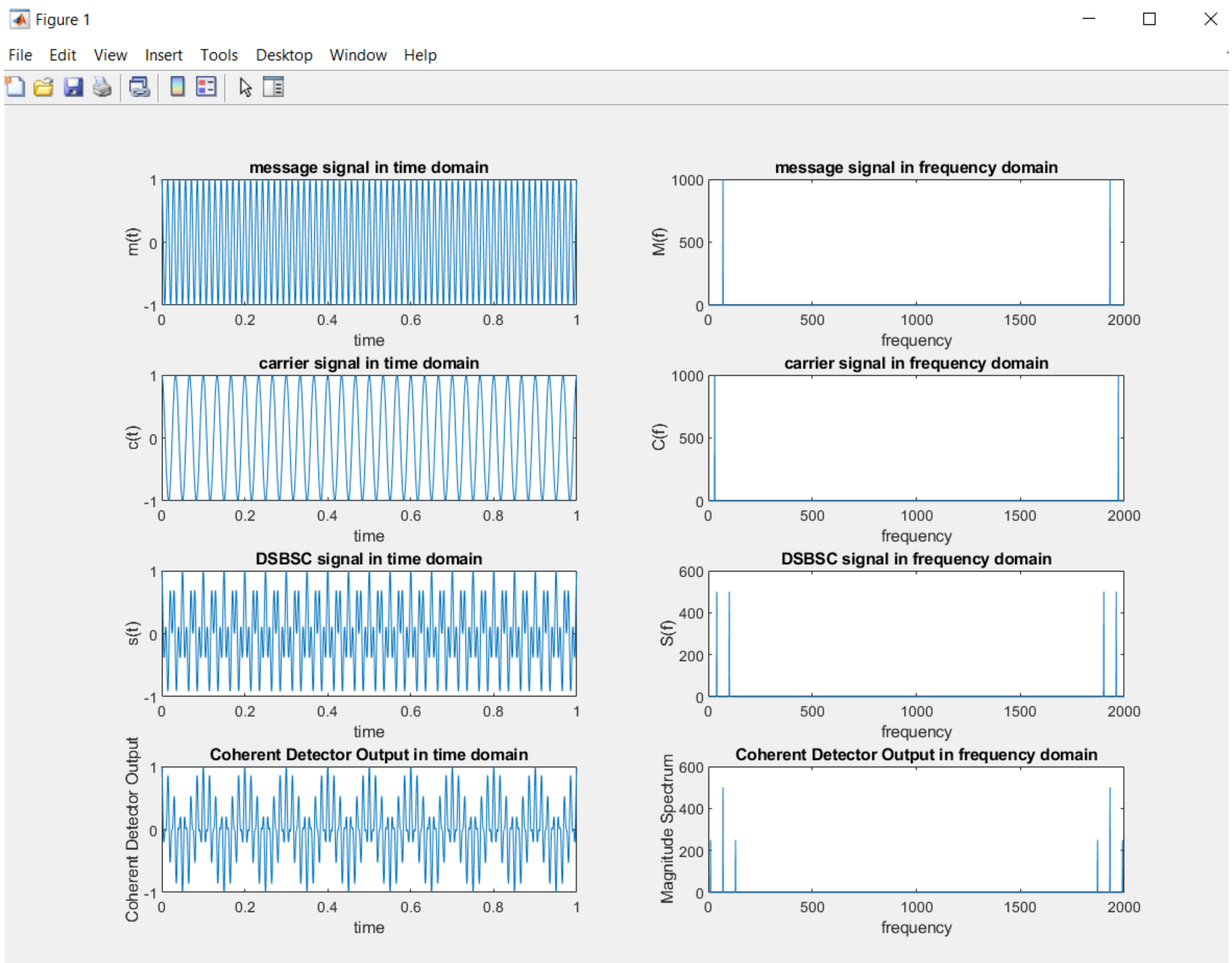


Figure 1
File Edit View Insert Tools Desktop Window Help
time domain
Coherent Detector Output
time
frequency domain
Magnitude Spectrum of Coherent Detector Output
Frequency



In the case where $f_m > f_c$, yes, we observe a higher amount of distortion in the output of the coherent detector. This has huge repercussions in the design of antennas or other long distance communication protocols because we require carrier waves of high frequency owing to the fact that high frequency carrier waves do not require a material medium to propagate.