Modulation and Coding - Report

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Contents

Chapter 1

Exercise 1

1.1 Question 1

For the first question, I use this Matlab script to sample all signals:

```
Fe = 8000; % Sampling frequency
Te = 1/Fe; % Sampling period
L = 8; % Number of samples
t = (0:L-1)/Fe; % Chosen times for samples
f = 0:Fe/L:Fe-Fe/L; % Chosen frequency for samples
x = 1+0*t; % Expression of the signal
y_fft = fft(x); % Calculate the DFT


subplot(311);
stem(f,abs(y_fft)); % Plot the spectrum of the signal
xlabel('Frenquency');
ylabel('Amplitude');

subplot(312);
stem(f,real(y_fft)); % Plot the real part of the DFT
xlabel('Frenquency');
ylabel('Amplitude');

subplot(313);
stem(f,imag(y_fft)); % Plot the imaginary part of the DFT
xlabel('Frenquency');
ylabel('Amplitude');
```

For each experiment, **x** is changing. **L** is set to 8 by default, but can be redefined. In this case, it will be specified.

- **x=1, signal length 8 samples**

```
x = 1+0*t; % To make the array easily
```

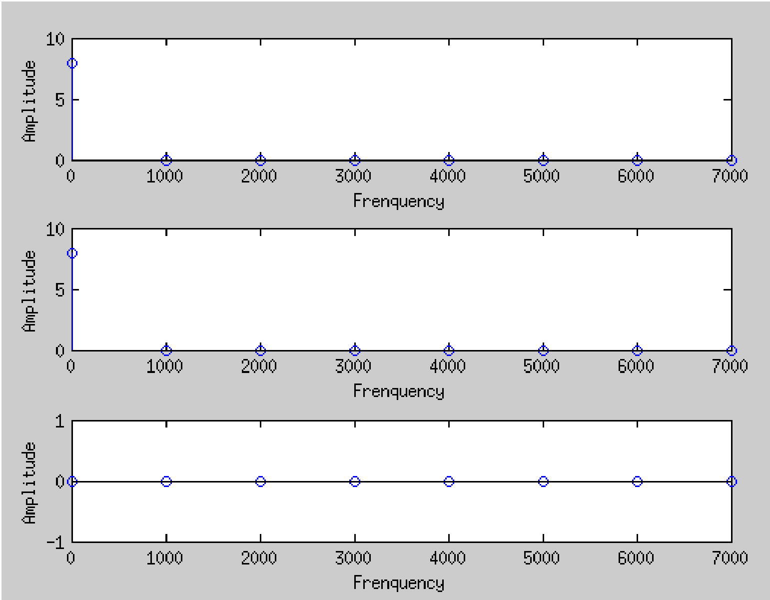

Figure 1.1: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*1000*t), signal length 8 samples**

```
x = sin(2*pi*1000*t);
```

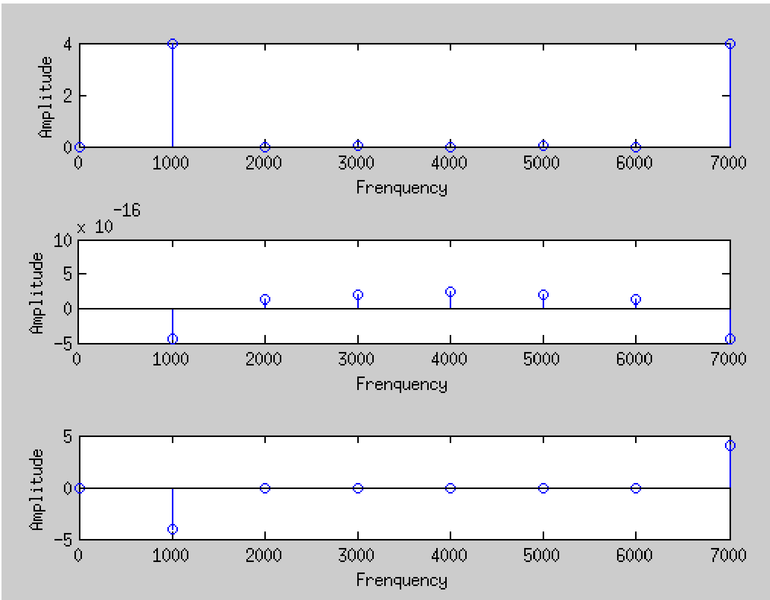

Figure 1.2: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*2000*t), signal length 8 samples**

```
x = sin(2*pi*2000*t);
```

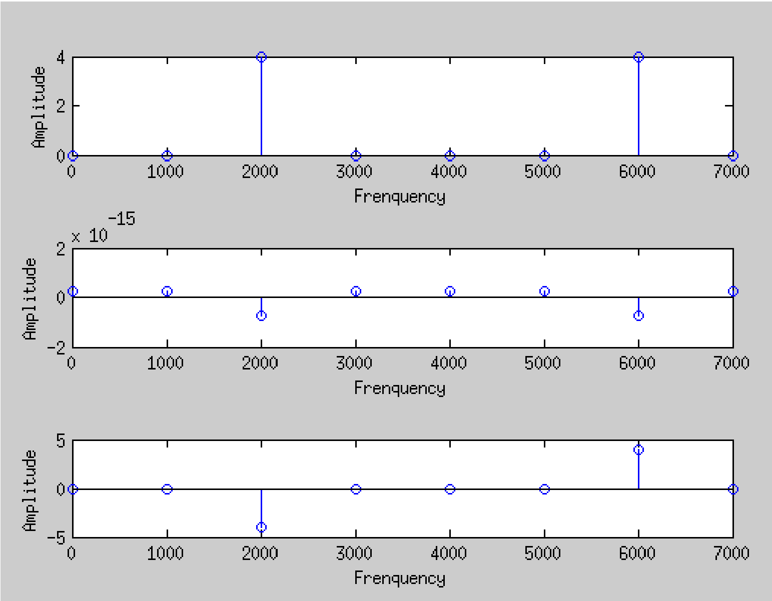

Figure 1.3: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*3000*t), signal length 8 samples**

```
x = sin(2*pi*3000*t);
```

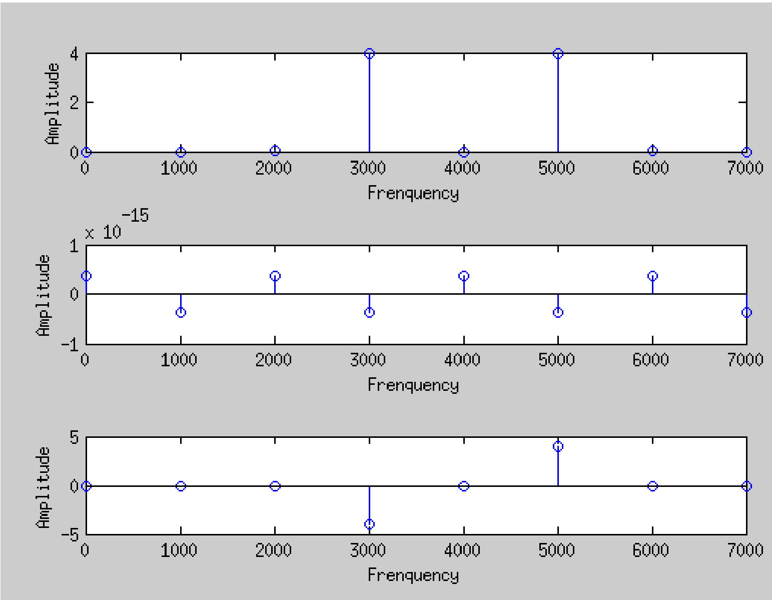

Figure 1.4: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*4000*t), signal length 8 samples**

```
x = sin(2*pi*4000*t);
```

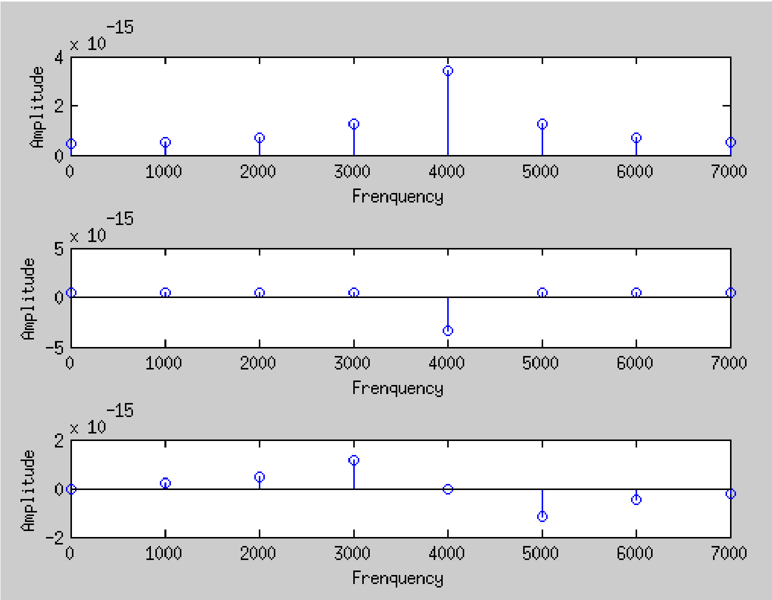

Figure 1.5: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*5000*t), signal length 8 samples**

```
x = sin(2*pi*5000*t);
```

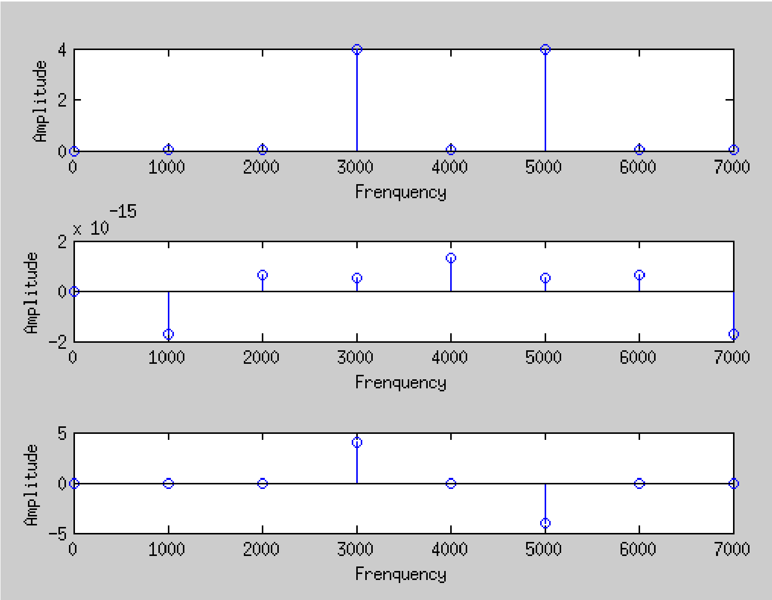

Figure 1.6: Signal's spectrum, DFT's real & imaginary parts

- **x=cos(2π*2000*t), signal length 8 samples**

```
x = cos(2*pi*2000*t);
```

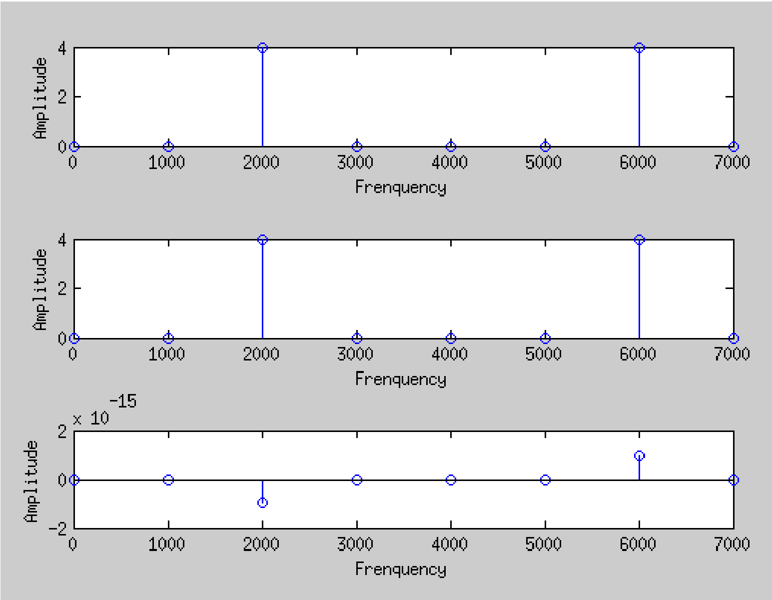

Figure 1.7: Signal's spectrum, DFT's real & imaginary parts

- **x=cos(2π*4000*t), signal length 8 samples**

```
x = cos(2*pi*4000*t);
```

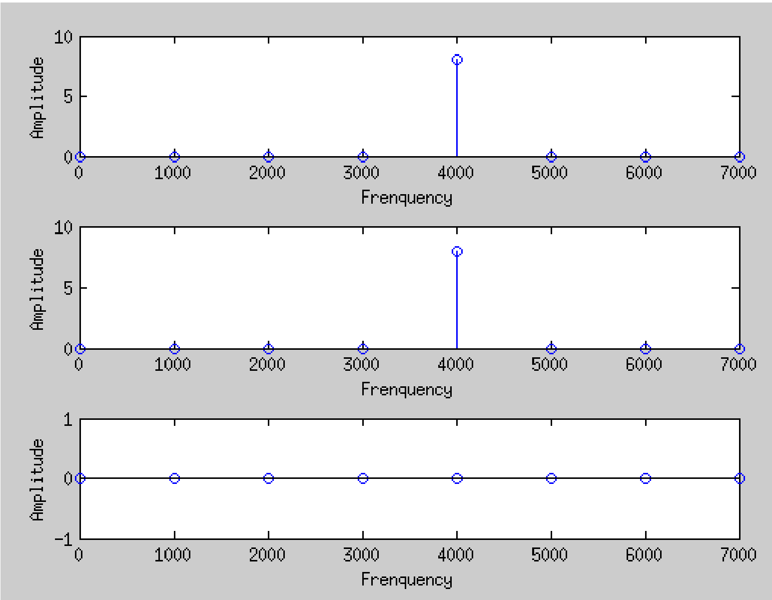

Figure 1.8: Signal's spectrum, DFT's real & imaginary parts

- **x=-1, signal length 8 samples**

```
x = −1+0∗t;
```

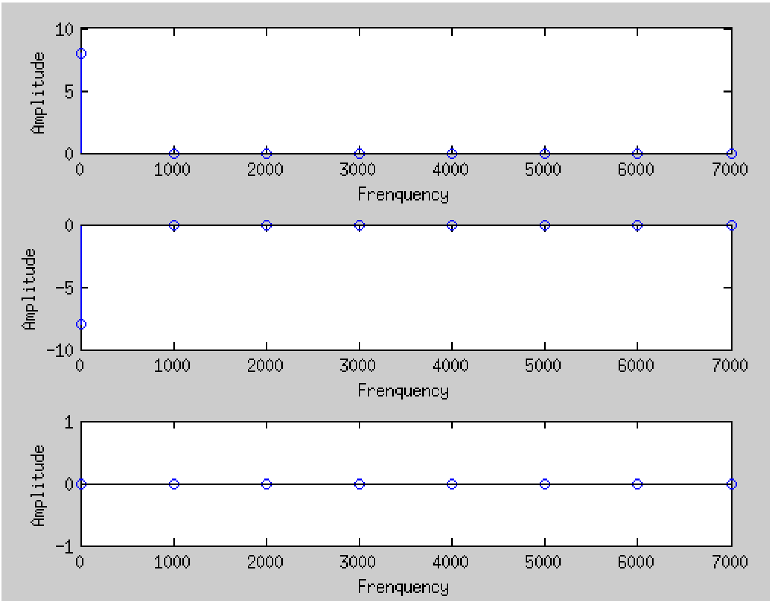

Figure 1.9: Signal's spectrum, DFT's real & imaginary parts

- **x=1, signal length 16 samples**

```
L = 16; % Only for this graph
x = 1+0∗t;
```

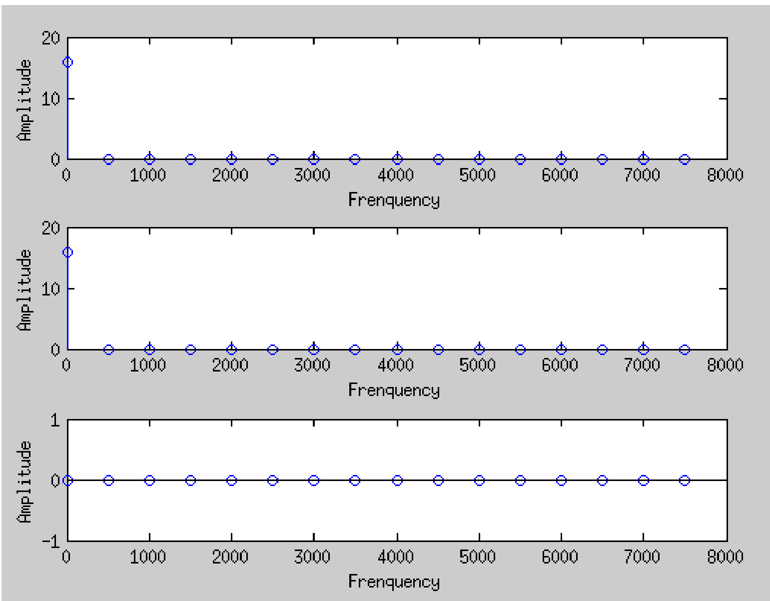

Figure 1.10: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*1000*t), signal length 16 samples**

```
L = 16; % Only for this graph
x = sin(2*pi*1000*t);
```

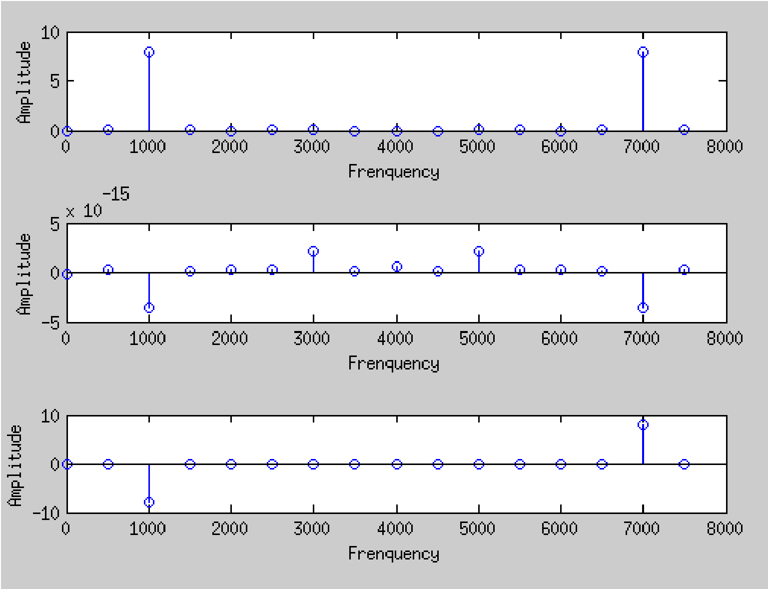

Figure 1.11: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*1000*t+0.5π), signal length 8 samples**

```
x = sin(2*pi*3000*t+0.5*pi);
```

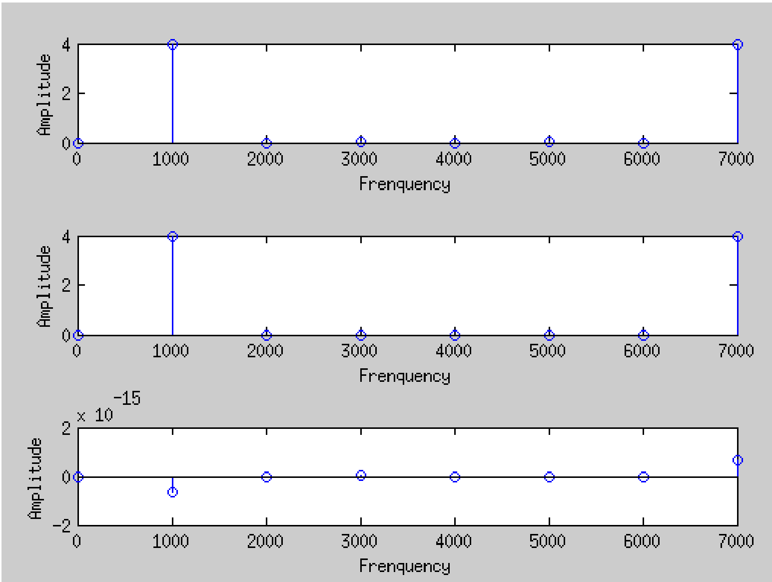

Figure 1.12: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*1000*t), signal length 18 samples**

```
L = 18; % Only for this graph
x = sin(2*pi*1000*t);
```

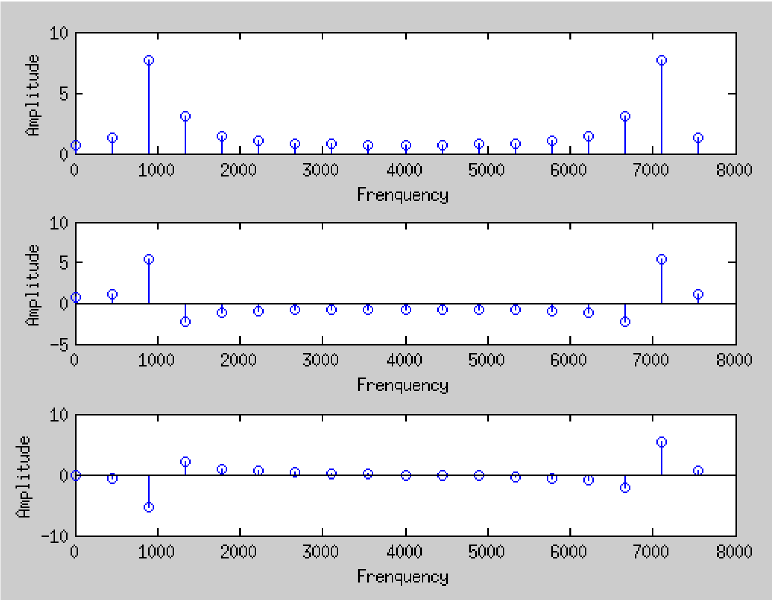

Figure 1.13: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*1000*t), signal length 20 samples**

```
L = 20; % Only for this graph
x = sin(2*pi*1000*t);
```

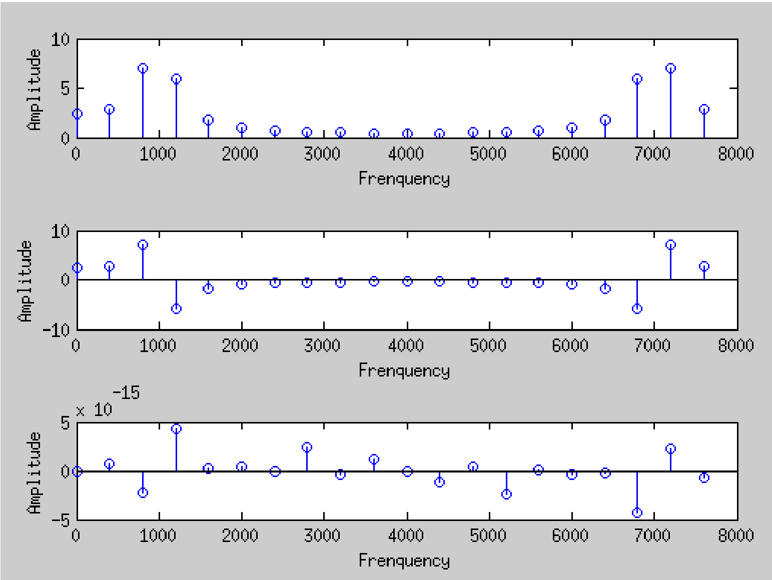

Figure 1.14: Signal's spectrum, DFT's real & imaginary parts

- **x=j*sin(2π*2000*t), signal length 8 samples**

```
x = 1i*sin(2*pi*2000*t);
```

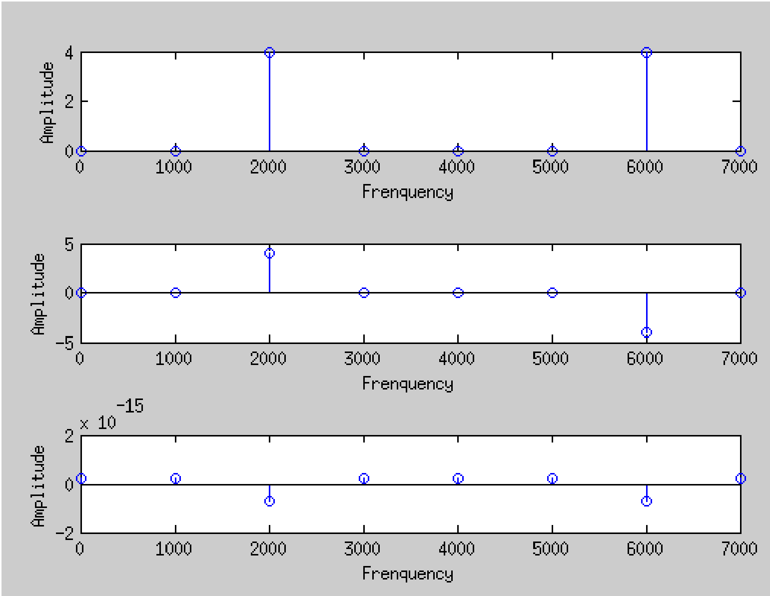

Figure 1.15: Signal's spectrum, DFT's real & imaginary parts

- **x=j*cos(2π*2000*t), signal length 8 samples**

```
x = 1i*cos(2*pi*2000*t);
```

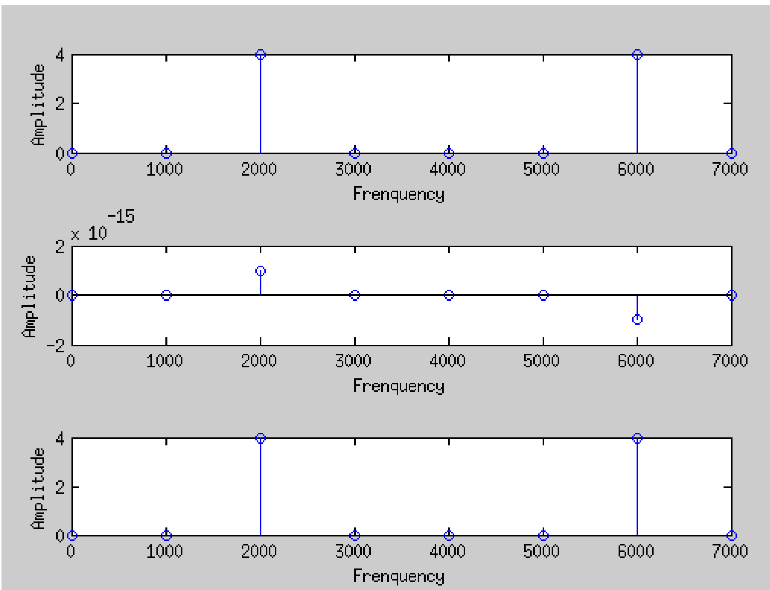

Figure 1.16: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*2000*t)+j*sin(2π*2000*t), signal length 8 samples**

```
x = sin(2*pi*2000*t)+1i*sin(2*pi*2000*t);
```

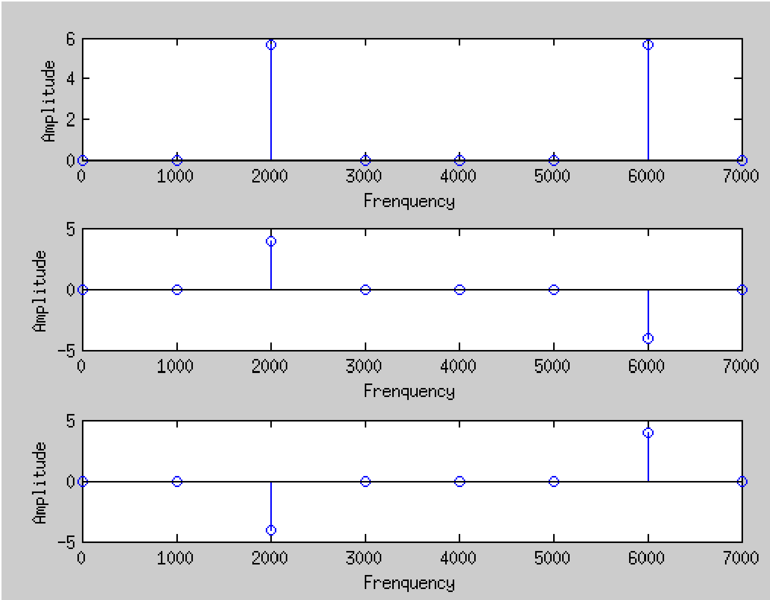

Figure 1.17: Signal's spectrum, DFT's real & imaginary parts

- **x=sin(2π*2000*t)+j*cos(2π*2000*t), signal length 8 samples**

```
x = sin(2*pi*2000*t)+1i*cos(2*pi*2000*t);
```

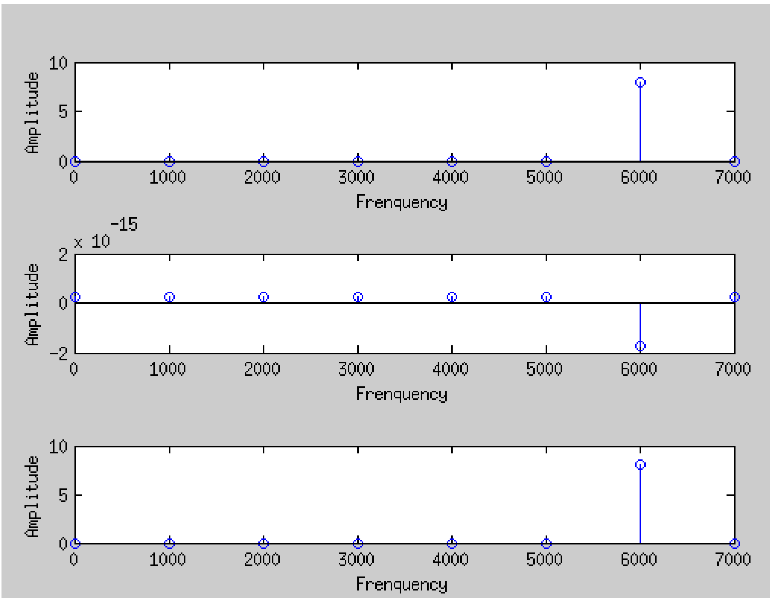

Figure 1.18: Signal's spectrum, DFT's real & imaginary parts

According to these experiments, it is possible to conclude that:

- A constant signal always has a real DFT, positive if the signal is positive, negative in the other case. There is only the first harmonic with an amplitude of: the signal's value multiplied by the number of samples.
- When the signal is out of phase, the spectrum doesn't change, but there is a change in the real and the imaginary part of the DFT (inversion, for a $\frac{1}{2}\pi$ change).
- If the Nyquist-Shannon ($Fe < 2f$) theorem is not respected, there are a lot of harmonies, and the DFT is not longer unique: it is impossible to recover the original signal. In the case of $Fe = 2f$, the first harmonic and its image are mingled.
- For a signal with a given frequency f, which respects the Nyquist-Shannon theorem, there is the first harmonic (maximum amplitude) for the frequency f, and there is an image of this harmonic for the frequency f+Fe/2.
- By adding the first harmonic amplitude and its image's one, the number of samples is retrievable. That means that the amplitude of the first harmonic is the half of the number of samples: this looks like the Fourier coefficients.

1.2 Question 2

For this question, I use the same script as for the first one, with these values:

```
L = 4;
```