## Lab Report

## (signal processing)

### 1.1 a)

#### **Calculation Part:**

(1)
(a) Given that

The function.

$$a(t) = 2\cos(2\pi t) + \cos(6\pi t)$$
and also
$$T = 1$$

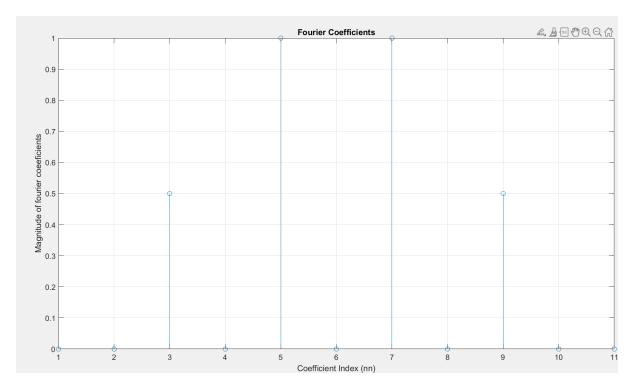
$$N = 5$$
We need to find the fourier coefficients of zet)

To find fourier coefficients
$$a_k = \frac{1}{T} | z(t)| e^{ikw} + ikw + ikw$$

by compasing the both eqns of xH)

$$a_0 = 0$$
  $a_1 = 1$   $a_2 = 0$   $a_3 = \frac{1}{2}$ 
 $a_{-1} = 1$   $a_{-2} = 0$   $a_{-3} = \frac{1}{2}$ 

### **MATLAB Plot:**



## **Observation:**

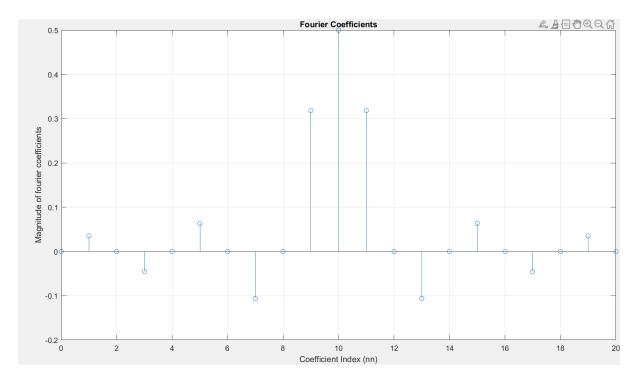
From the plot and the calculation part we are getting the same values.

## 1.1 b)

(It is a periodic square wave function)

## **Observation:**

## **MATLAB plot:**



From the plot we can say that it is a even symmetric function.

\*every coefficient index nn is satisfying the f(-x)=f(x)

So it is even function.

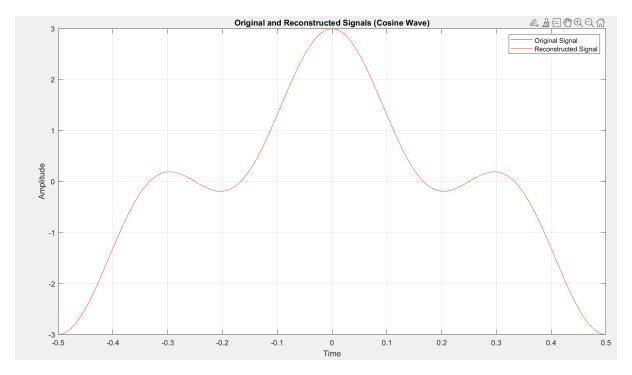
### 1.2

$$\hat{x}(t) = \sum a_k *e^{(jk\omega_0t)}$$
 (where k is running from -N to N)

It is for finding the Partial Fourier sum of order N of the signal x(t). If the N is increased to infinity then the partial sum approaches to the original signal i.e x(t).

b)

**MATLAB PLOT:** 



Here with respect to plot the original and the reconstructed signal both are overlapping.

c)

There is small error.

\*\*The value of Maximum absolute error (MAE) between original signal and reconstructed signal : 4.440892098500626e-16

\*\* The Root mean squared (RMS) error between original signal and reconstructed signal :8.768389183253862e-17

(Or)

Max Absolute Error: 0.000000000000004440892099

RMS Error: 0.000000000000000876838918

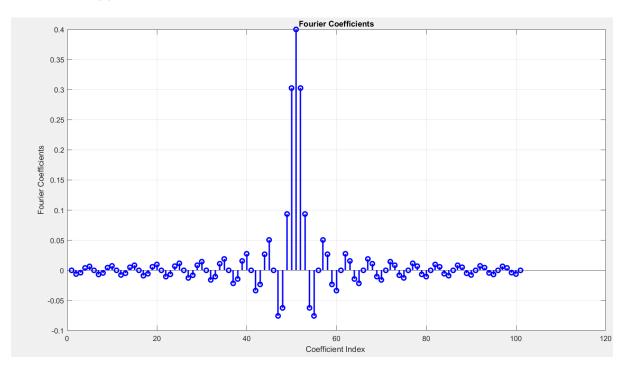
1.3)

a)

Here the Fourier series coefficients  $\{a_k\}$  for a real ,periodic square wave which have the amplitude as 1 in the interval [-T1,T1] and the period is T. (T1<T/2)

## **Observation:**

## **MATLAB Plot:**



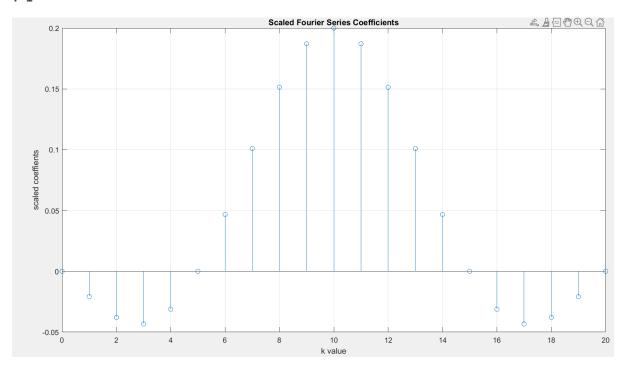
# b)

**Given T1=0.1** 

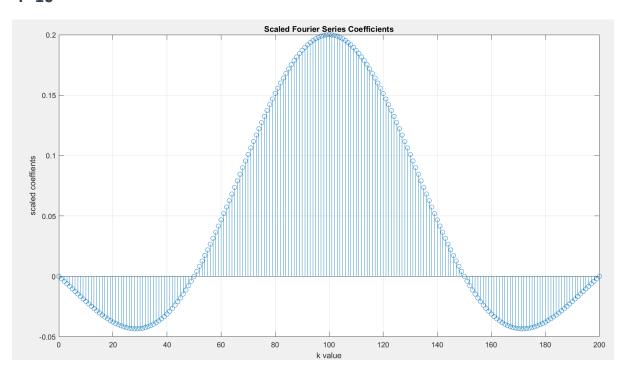
N=10\*T

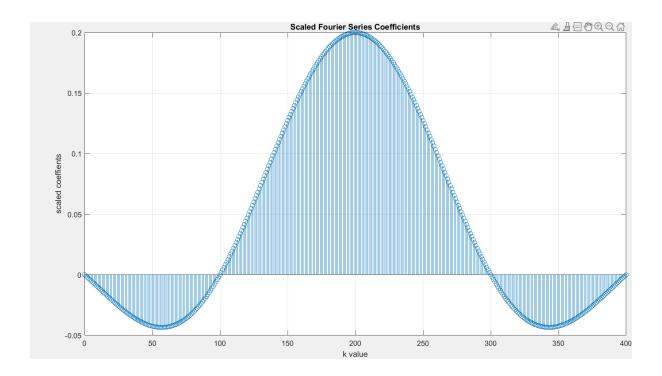
The plot of scaled coefficients  $T^*a_{\mbox{\tiny K}}$ 

T=1



# T=10





#### **Observation:**

The fourier series representation allows you to decompose a periodic signal into a sum of sinusoidal components (harmonics). The coefficients represent the amplitudes of these sinusoidal components.

- \*As you increase the period T, that means stretching the signal.
- \*As you increase T the signal becomes smoother within each period.

As T approaches infinity, the signal becomes a continuous waveform without any abrupt changes. In this case, many of the fourier coefficients tend to zero, the a0 component which represents the mean value of the signal. This is related to the concept of the Dirac comb function, which is the limit of a periodic train of impulses as the period becomes infinite.

#### **Conclusion:**

By increasing the period T leads to a convergence of the Fourier series coefficients towards the true continuous Fourier transform of the signal.

c)

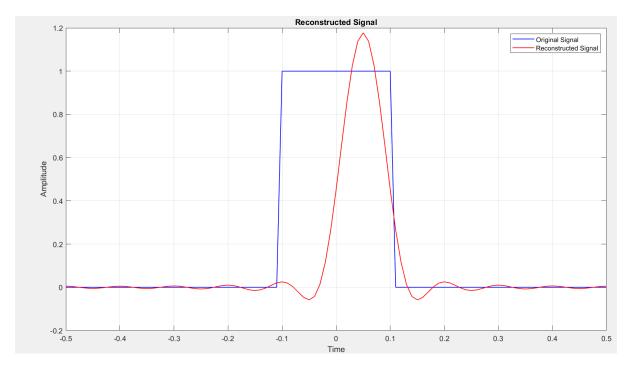
here the

N=10

T1=0.1

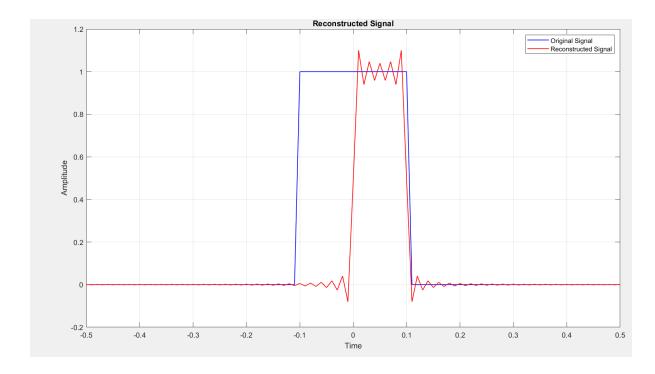
T=1

## time\_grid=-0.5:0.01:0.5

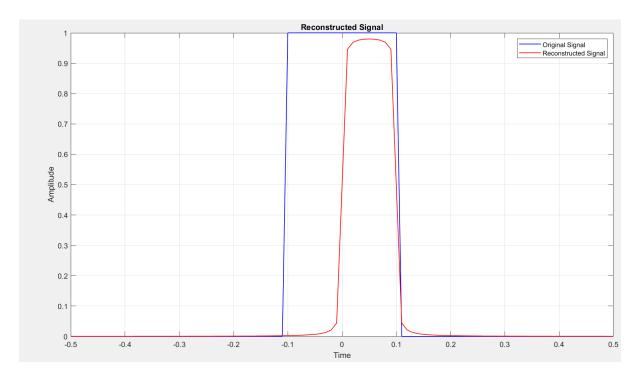


when

N=50

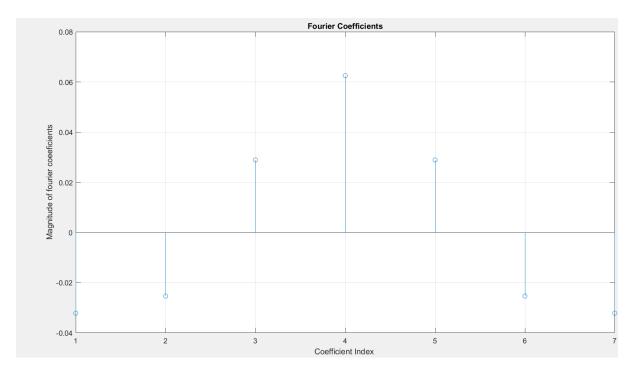


#### N=100



1.4)

a)



c)

here from the a) and b)

we can say that part a has even symmetric

i.e

f(-t)=f(t) for real part

and zero for imaginary part

in part b

we get the odd symmetry for the imaginary part

f(-t)=-f(t)

and all the values are zeros for real part.

Here we can observe the same for real and imaginary part of the both functions.