# Cmpe 362

# HOMEWORK2

REPORT

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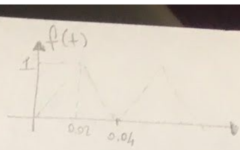
# Question 1:
# Part A:

1

$f = \frac{1}{T}$

$w = \frac{2\pi}{T}$  $T = 0.04$




$$f(t) = \begin{cases} 50t & 0 < t < 0.02 \\ 2 - 50t & 0.02 < t < 0.04 \end{cases}$$

Even function

$$f(t) = a_0 + \sum_{n=1}^{\infty} a_n \left(\cos(nwt) + j\sin(nwt)\right)$$

Since even multiply with 2

$$a_0 = \frac{2}{0.04} \int_0^{0.02} f(t).dt = 50.\int_0^{0.02} 50t.dt$$

$$a_0 = 50^2 \left.\frac{t^2}{2}\right|_0^{0.02} = \frac{50^2.(0.02)^2}{2} = \frac{1}{2}$$

$$a_n = \frac{4}{0.04} \int_0^{0.02} 50t.\cos(nwt).dt$$

$j\sin(nwt).dt \rightarrow 2.$ part (imaginary)

Integration by parts

$u = t \quad du = dt$

$dv = \cos(nwt)\,dt \quad v = \frac{1}{nw}\sin(nwt)$

$dv = \sin(nwt).dt \quad v = \frac{1}{nw}(-\cos(nwt))$

$w = \frac{2\pi}{T} = \frac{2\pi}{0.04} = 50\pi$

$$a_n = 5.10^3 \left( \left.\frac{t}{nw}\sin(nwt)\right|_0^{0.02} - \int_0^{0.02} \frac{1}{nw}\sin(nwt).dt \right)$$

$$a_n = 5.10^3 \left( \underbrace{\left.\frac{t}{50\pi n}\overset{(-\cos(50\pi n t))}{\sin(50\pi n.t)}\right|_0^{0.02}}_{\sin(n\pi)=0 \;\rightarrow 0} - \underbrace{\int_0^{0.02} \frac{1}{50\pi n}.\overset{(-\cos(50\pi n t))}{\sin(50\pi n t).dt}}_{0 \text{ for imaginary part}} \right)$$

$$a_n = 5.10^3 \left( \left.\frac{\cos(50\pi n t)}{(50\pi n)^2}\right|_0^{0.02} \right) = \frac{5.10^3}{5^2.10^2.\pi^2.n^2}\left(\cos(n\pi) - \cos 0\right)$$

$$a_n = \frac{2}{\pi^2 \cdot n^2}\left(\cos(n\pi) - 1\right) - \frac{2j}{\pi n}\cos(n\pi)$$ Blue : $b_k$'s

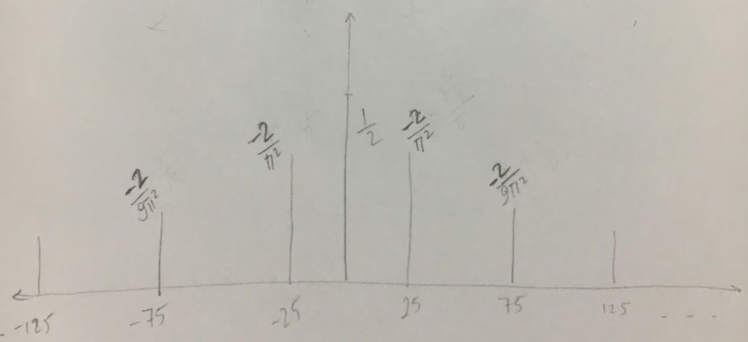
depends on $n$ is even or odd.

$$a_n = \begin{cases} 0 & \text{if } n \text{ is even} \quad \frac{-2j}{\pi n} \\ \frac{-2}{\pi^2 \cdot n^2} & \text{if } n \text{ is odd} \quad \frac{+2j}{\pi n} \\ \frac{1}{2} & \text{if } n = 0 \end{cases}$$

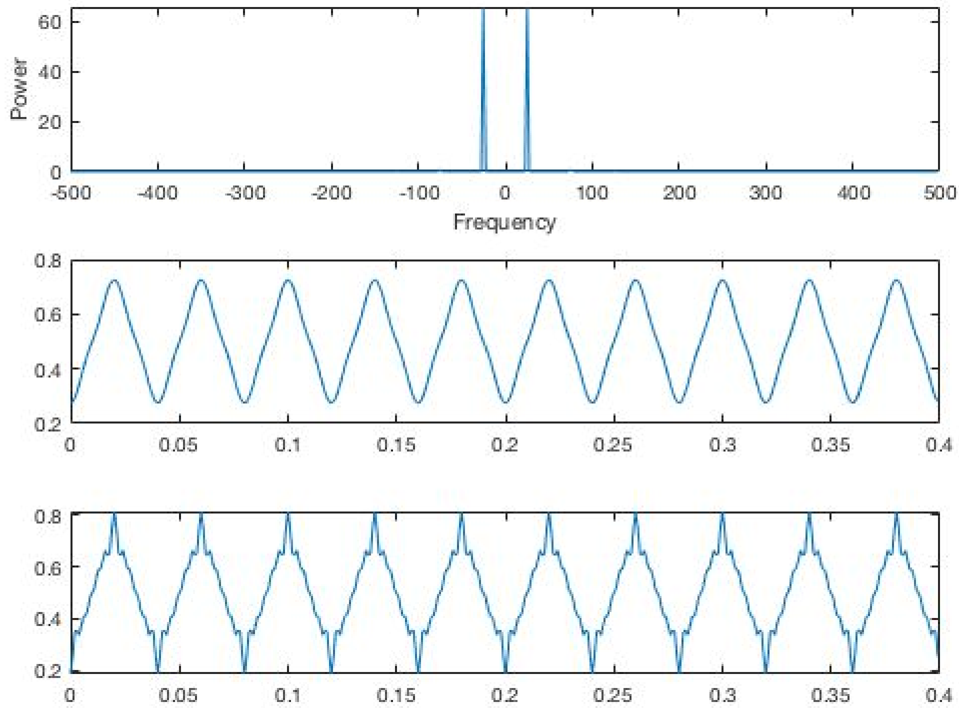
$b_k$

imaginary part

$$f(t) = \frac{1}{2} + \sum_{n=1}^{\infty} \frac{-2}{\pi^2 n^2}\cos(nwt) + \frac{2j}{\pi n} \cdot \sin(nwt)$$



# Part B:




1: Freq-power 2: Sum DC+First and Third Harmonics 3: Sum DC+First through Eleventh Harmonics

Code:

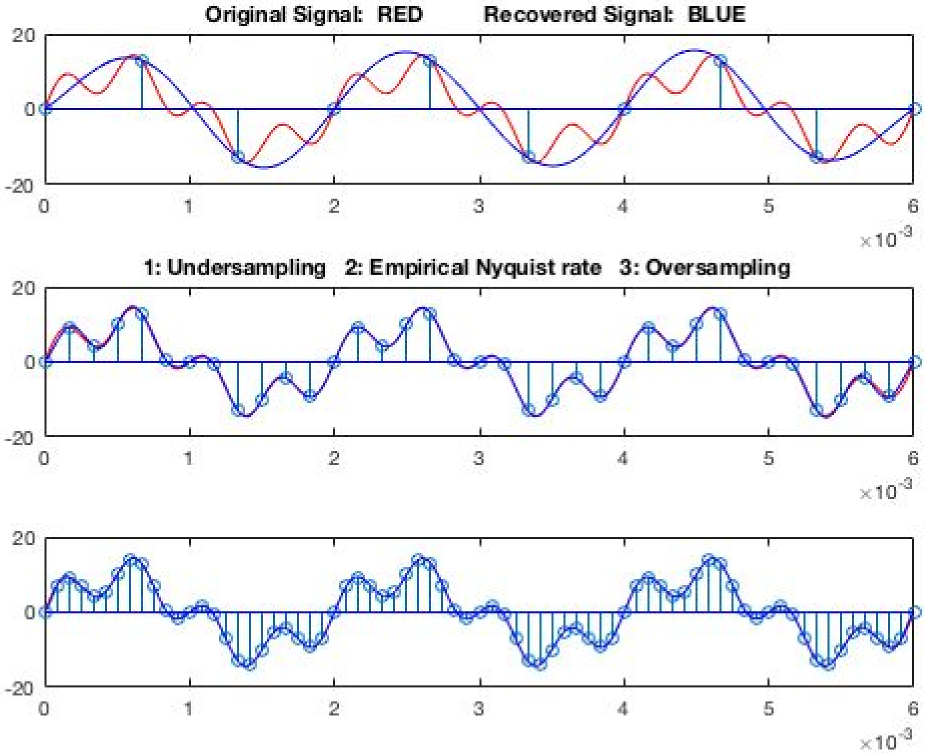
```
T = 10*(1/25);
fs = 1000; %sample freq
t = 0:1/fs:T-1/fs;
x = sawtooth(2*pi*25*t,1/2); %sawtooth func
y = fft(x);
n = length(x);          % number of samples
y0 = fftshift(y);         % shift y values
f0 = (-n/2:n/2-1)*(fs/n); % 0-centered frequency range
power0 = abs(y0).^2/n;    % 0-centered power
subplot(3, 1, 1);
plot(f0,power0);
xlabel('Frequency');
ylabel('Power');
ffund=25 % fund freq.
%1 to 11 harmonics
j = sqrt(-1);
x1=(-2/(pi^2)).*cos(2.*pi.*ffund.*t)+(2/(pi*1)).*j*sin(2.*pi.*ffund.*t);
x3=(-2/(pi^2*9)).*cos(2.*pi.*ffund*3.*t)+(2/(pi*3)).*j*sin(2.*pi.*ffund*3.*t);
x5=(-2/(pi^2*9)).*cos(2.*pi.*ffund*5.*t)+(2/(pi*5)).*j*sin(2.*pi.*ffund*5.*t);
x7=(-2/(pi^2*9)).*cos(2.*pi.*ffund*7.*t)+(2/(pi*7)).*j*sin(2.*pi.*ffund*7.*t);
```

```
x9=(-2/(pi^2*9)).*cos(2.*pi.*ffund*9.*t)+(2/(pi*9)).*j*sin(2.*pi.*ffund*9.*t);
x11=(-2/(pi^2*9)).*cos(2.*pi.*ffund*11.*t)+(2/(pi*11)).*j*sin(2.*pi.*ffund*11.*t);
first=0.5+x1+x3; %first part
second=0.5+x1+x3+x5+x7+x9+x11; %second part
%plots
subplot(3, 1, 2);
plot(t,first);
subplot(3, 1, 3);
plot(t,second);
```

# Question 2:

f1=2000,f2=500 Hz
Signal :    5*sin(2*pi*f1*t) +   10*sin(2*pi*f2*t)

sample frequencies:
fs1=1500; **Undersampling**
fs2=6000; **Empirical Nyquist Sampling Rate**
fs3=12000; **Oversampling**




# Code:

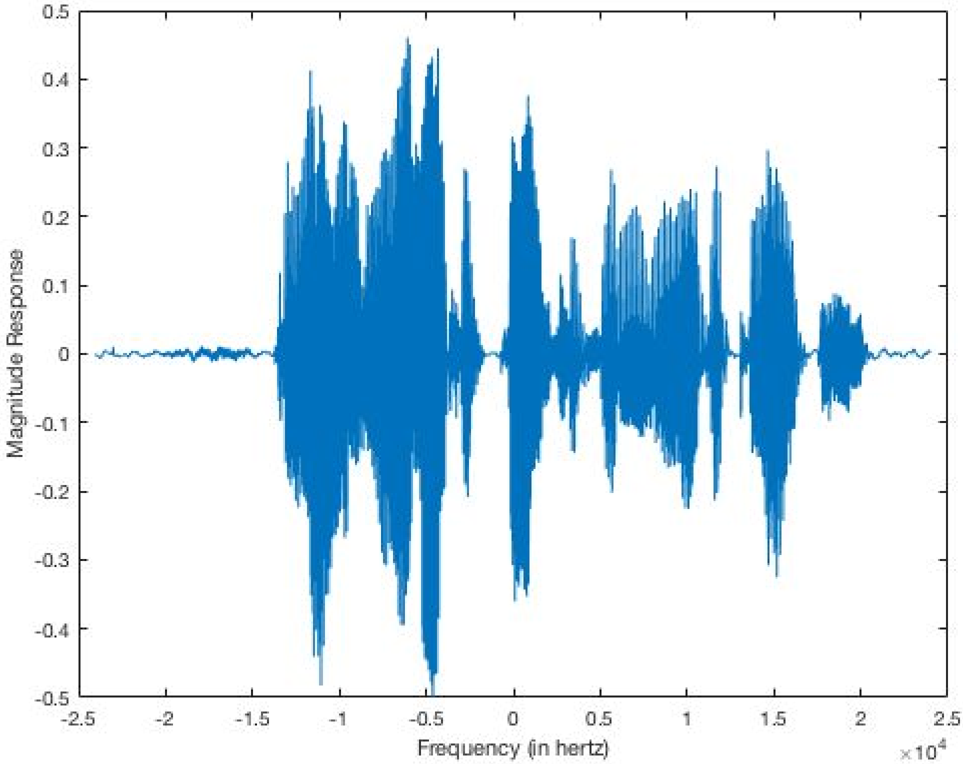
```
f1 = 2000;% input signal freq1
f2 = 500;%input signal freq2
fs1=1500;fs2=6000;fs3=12000;%sample freq.s
T = 1/f2;
startT = 0;
endT = 3*T;  %take first 3 period duration
dt = 1/(f1*100);

dt1 = 1/fs1;
dt2 = 1/fs2;
dt3 = 1/fs3;
t = startT:dt:endT;
x = 5*sin(2*pi*f1*t)+10*sin(2*pi*f2*t);  %construct addition of two signal with diffrent freq.
t1 = startT:dt1:endT;            %sample with dif sample freq. fs1
x1 = 5*sin(2*pi*f1*t1)+10*sin(2*pi*f2*t1);
t2 = startT:dt2:endT;    %sample with dif sample freq. fs2
x2 = 5*sin(2*pi*f1*t2)+10*sin(2*pi*f2*t2);
t3 = startT:dt3:endT;    %sample with dif sample freq. fs2
x3 = 5*sin(2*pi*f1*t3)+10*sin(2*pi*f2*t3);
figure()  %plots
subplot(311)
plot(t,x,'r');
title(' Original Signal:  RED         Recovered Signal:  BLUE  ')
hold on
stem(t1,x1);
plot(t,iPolate(x1,t,dt,dt1),'b');
subplot(312)
plot(t,x,'r');
```

```
title('1: Undersampling   2: Empirical Nyquist rate   3: Oversampling    ');
hold on
stem(t2,x2);
plot(t,iPolate(x2,t,dt,dt2),'b');
subplot(313)
plot(t,x,'r');
hold on
stem(t3,x3);
plot(t,iPolate(x3,t,dt,dt3),'b');
function [Y]=iPolate(x,t,dt,dt_sample) %function to interpolate
Y = zeros(length(t));
for i=1:length(t)
for j=1:length(x)
Y(i)=Y(i) + x(j)*sinc(((i-1)*dt-(j-1)*dt_sample)/dt_sample);
end
end
end
```

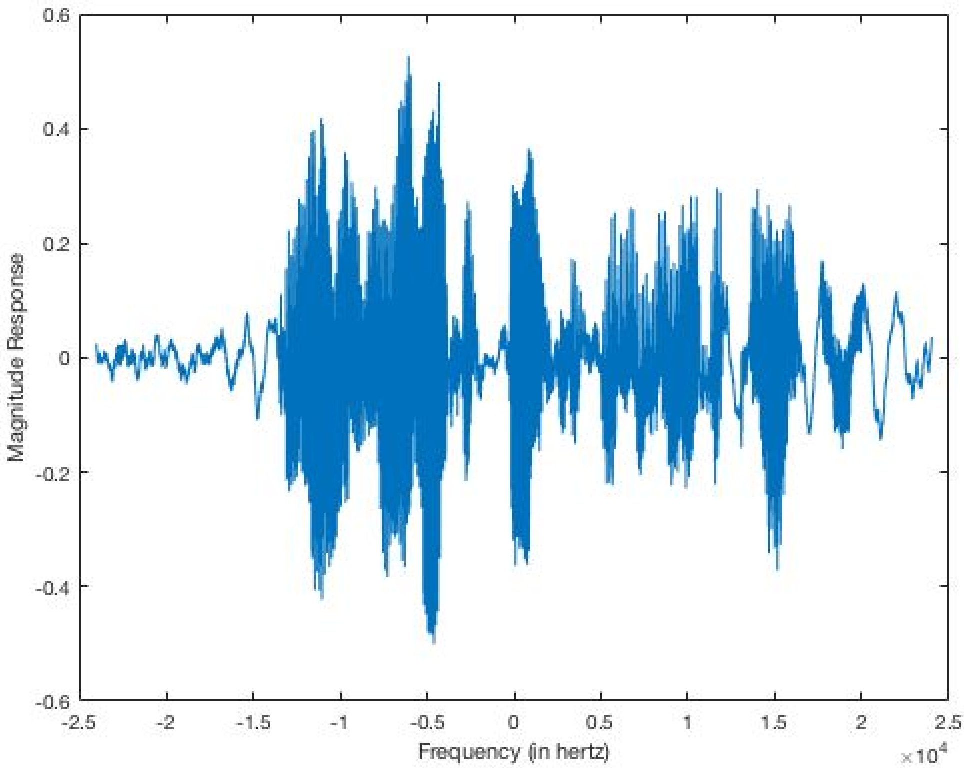
# Question 3:

I choosed this sample:
('clean_testset_wav/p232_088.wav',
'noisy_testset_wav/p232_088.wav')

I found out filter corresponding to this sample and figured out to arrange it for different length audio’s. But I was not that succesful.




clean




,
noisy

# Code:

```
%audio file to fiter
filename='noisy_testset_wav/p232_097.wav';

%Get filter from this sample
freRes=getFilter('clean_testset_wav/p232_099.wav','noisy_testset_wav/p232_099.wav');

fLength=length(freRes);
freRes=freRes(1:fLength);
tRes=ifft(freRes);
[dataNoisy, fsN] = audioread(filename);
dLength=length(dataNoisy);
minLen=min(dLength,fLength);  %find minimum of filter and data
result=filterSound(dataNoisy,tRes(1:minLen),fsN);
%sound(result,fsN);
fileResult='clean.wav';
```

```
audiowrite(fileResult,result,fsN);




function [Y]=getFilter(filename1,filename2)

[dataClean, fsC] = audioread(filename1);
[dataNoisy, fsN] = audioread(filename2);
dF = fsC/length(dataClean);                  % hertz
f = -fsC/2:dF:fsC/2-dF;
 %%Plot the spectrum:
  figure;
 plot(f,dataClean);
 xlabel('Frequency (in hertz)');
 ylabel('Magnitude Response');
 figure;
 plot(f,dataNoisy);
 xlabel('Frequency (in hertz)');
 ylabel('Magnitude Response');
data_fft_c = (fft(dataClean));
data_fft_n = (fft(dataNoisy));
Y=data_fft_c./data_fft_n;
end

function [result]=filterSound(dataNoisy,tRes)
result=cconv(dataNoisy,tRes,length(dataNoisy));
end
```