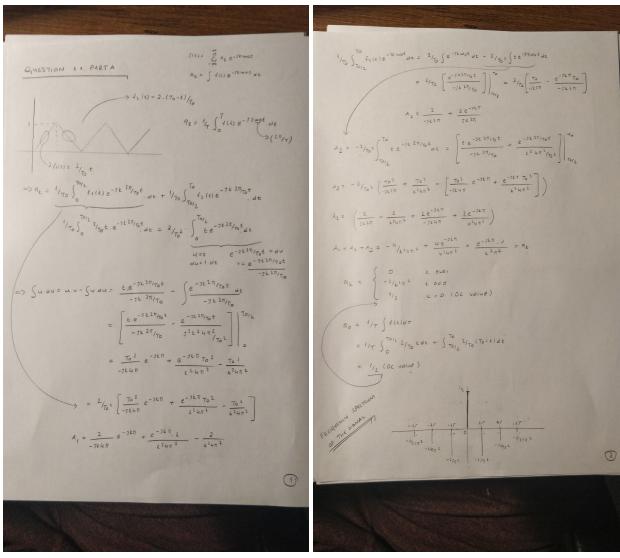
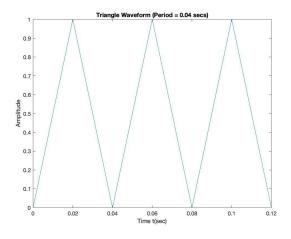
# CMPE 362: Project 1 — Due:March 8st 23:59

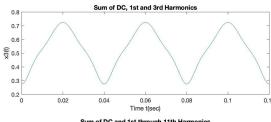
Note: Prepare a report (pdf file) includes your code, explanations and comments of your code for each question. You will compress everything into a zip file. Name it as YourNumber-CmpE362-HW2.zip and submit it via Moodle.

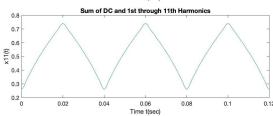
### **Question 1.PART A:**

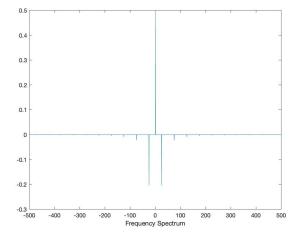


# **Question 1.PART B:**









#### Code:

```
close all;
%% First part
figure('Name','Triangle Wave','NumberTitle','off');
T = 1/25; % period
t = 0:0.001:3*T; % time variable with 0.001 step size
x = sawtooth(2*pi*1/T*t,0.5)/2 + 0.5; % signal
plot(t,x);
title('Triangle Waveform (Period = 0.04 secs)');
xlabel('Time t(sec)'); ylabel('Amplitude');
%% Second part
figure('Name','Harmonic Summation','NumberTitle','off');
% DC & 1st harmonic & 3rd Harmonic
wZero = 50 * pi; % wZero value
harmonicOne = -(2 / (pi ^ 2)) * exp(wZero * t * 1j); % 1st harmonics
harmonicThree = -(2 / (3 ^ 2 * pi ^ 2)) * exp(3 * wZero * t * 1j); % 3rd harmonics
x3 = harmonicOne + harmonicThree + 0.5; % Sum of DC & 1st harmonic & 3rd Harmonic
subplot(2,1,1);
plot(t, abs(x3));
title('Sum of DC, 1st and 3rd Harmonics');
xlabel('Time t(sec)'); ylabel('x3(t)');
% DC & 1st through 11th Harmonic
harmonicSeven = -{2 / (7 ^ 2 * pi ^ 2)) * exp(7 * wZero * t *1j); % 7th harmonics
harmonicEleven= -(2 / (11 ^ 2 * pi ^ 2)) * exp(11 * wZero * t *1j); % 11th harmonics
x11 = harmonicOne + harmonicThree + harmonicFive + harmonicSeven + ...
 harmonicNine + harmonicEleven + 0.5; % Sum of DC & 1st through 11th Harmonic
subplot(2,1,2);
plot(t,abs(x11));
title('Sum of DC and 1st through 11th Harmonics');
xlabel('Time t(sec)'); ylabel('x11(t)');
%% Third part
t = 0:0.001:T*50-0.001; % time variable with 0.001 step size
x = sawtooth(2*pi*25*t,0.5)/2 + 0.5; % signal
<u>figure</u>
y = fft(x);
                  % fft of x
n = length(x); % number of samples
y0 = fftshift(y/n); % shift y values
f0 = (-n/2:n/2-1)*(1000/n);
plot(f0,y0)
xlabel('Frequency Spectrum')
```

### **Question 2:**

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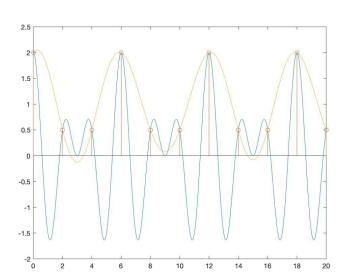
### Code:

```
clear;
clc;
close all;
t = 0:0.001:20; % time variable with 0.001 step size
c = a + b; % sum of these signals
%% UNDERSAMPLING
figure('Name','UNDERSAMPLING','NumberTitle','off');
plot(t, c); hold on;
Fs = 0.5; % sampling rate
n = 0:1/Fs:20;
x = c(1:1/Fs*1000:20*1000+1); % sampled points at fs 0.5
stem(n, x); hold on;
% sign interpolation
y = zeros(1, size(t, 2));
for i = 1:size(x,2)
 y = y + x(i) * sinc((t - ((i - 1) * 1 /Fs)) * Fs);
<u>end</u>
plot(t, y);
%% NYQUIST SAMPLING
figure('Name','NYQUIST SAMPLING','NumberTitle','off');
plot(t, c); hold on;
Fs = 1.1;
n = 0:1/Fs:20;
x = c(1:1/Fs*1000:20*1000+1); % sampled points at fs 1.1
stem(n, x); hold on;
```

% sign interpolation y = zeros(1,size(t,2));for i = 1:size(x,2)y = y + x(i) \* sinc((t - ((i - 1) \* 1 /Fs)) \* Fs);<u>end</u> plot(t, y); %% OVERSAMPLING figure('Name','OVERSAMPLING','NumberTitle','off'); plot(t, c); hold on; Fs = 5;n = 0:1/Fs:20;x = c(1:1/Fs\*1000:20\*1000+1); % sampled points at fs 5.0 stem(n, x); hold on; % sign interpolation y = zeros(1, size(t, 2));for i = 1:size(x,2)y = y + x(i) \* sinc((t - ((i - 1) \* 1 /Fs)) \* Fs);end plot(t, y);

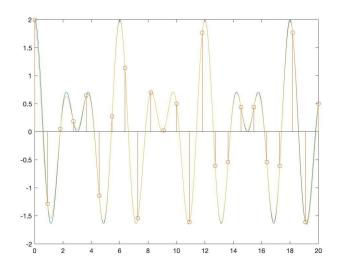
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I take frequencies respectively  $\frac{1}{2}$  and  $\frac{1}{3}$  and also found that nyquist sampling rate is 1.1.

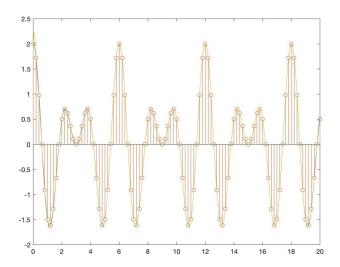


# **UNDERSAMPLING:**

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# **NYQUIST SAMPLING:**



## **OVERSAMPLING:**

#### Question 3:

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This is ideal filter construct:

I select "p232 090 clean.wav" and "p232 090 noisy.wav". Frequency spectrum for 2 signals can be seen in figure 5: Then, I find the frequency response of my ideal filter by dividing FFT(clean signal)/FFT(noisy signal). The frequency response of my filter in figure 6: Now, let's look how we could use this filter in time domain: I can find the time-response of my filter by using the functions ifftshift and ifft. There is a duality between time-domain and frequency domain. The multiplication operation in frequency domain is equivalent to the convolution operation in time domain. Therefore, I can obtain a new clean recording by taking convolution of noisy recording and time-response of your filter.

#### CODE:

```
%% IDEAL FILTER SECTION
clear;
close all;
clc;
filename = "p232_090.wav"
[ySignal, Fs] = audioread(filename);
filename = "p232_090_noise.wav";
[yNoise, Fs] = audioread(filename);
T = 1 / Fs;
% fft transform of clean
fftY = fft(y);
fftY = fftshift(fftY);
fftYNoise = fft(yNoise);
fftYNoise = fftshift(fftYNoise);
fftDivide = fftY' ./ fftYNoise';
figure,plot(abs(fftDivide));
a = ifftshift(fftDivide);
a = ifft(a);
figure, plot (a);
c = cconv(a, yNoise');
figure, plot(y);
x=size(c);
figure, plot(c);
```

### My filter code:

```
clear;
close all;
clc;
filename = "p232_090.wav";
[ySignal, Fs] = audioread(filename);
filename = "p232_090_noise.wav";
length = size(yNoise,1);
f0 = (-length/2:length/2-1)*(fs/length);
T = 1 / Fs;
Y = fft(yNoise - ySignal);
L = 189269;
P2 = abs(Y/L);
P1 = P2(1:ceil(L/2));
P1(2:end-1) = 2*P1(2:end-1);
f = Fs*(0:(L/2))/L;
figure, plot(f,P1);
Y = fft(ySignal);
P2 = abs(Y/L);
P1 = P2(1:ceil(L/2));
P1(2:end-1) = 2*P1(2:end-1);
f = Fs*(0:(L/2))/L;
figure, plot(f,P1);
zeroLele = zeros(length,1);
upper = 24000;
shift = ifftshift(zeroLele');
shift = ifft(shift);
```