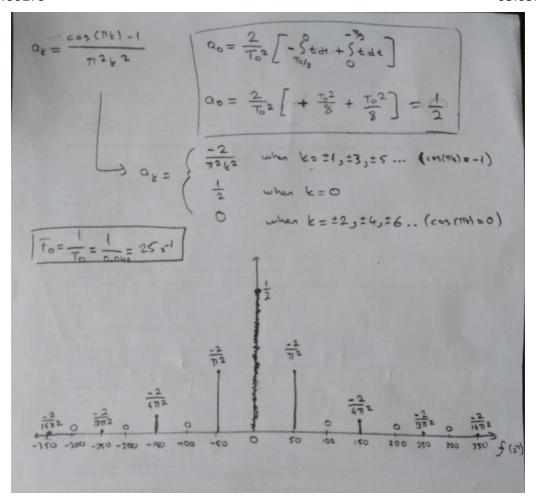
CMPE 362 Homework 2

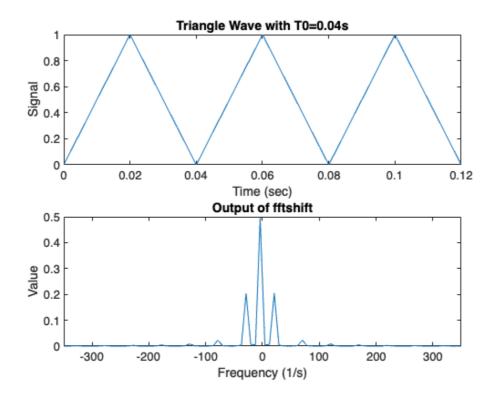
1.a.-

•
$$\times (t) = \begin{cases} \frac{-c}{100} & \text{for } 0.02 \le t < 0.02 \end{cases}$$

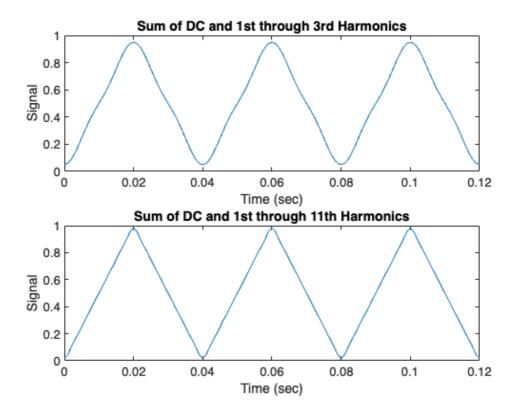
• Fourer Analysis Integral $\frac{1}{2} = \frac{1}{100} = \frac{1}{100}$



1.b.-



My calculations to find Fourier coefficients of given triangle waveform are consistent with the results obtained using discrete fast Fourier function of MATLAB. Reconstruction of the signal with the coefficients I found also proves that they are correct. Reconstructed signal is below.



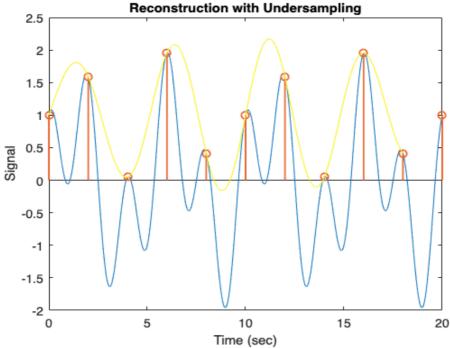
2-

$$f_1 = 0.20 s^{-1}$$

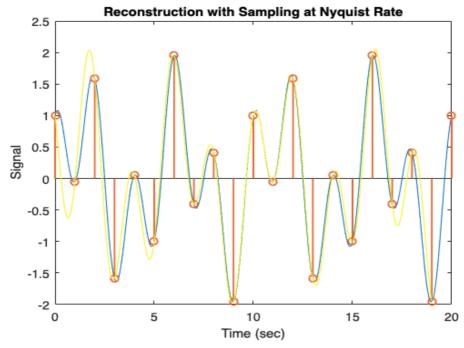
$$x_1 = \sin (2 * \pi * f_1 * t)$$

$$f_2 = 0.50 s^{-1}$$

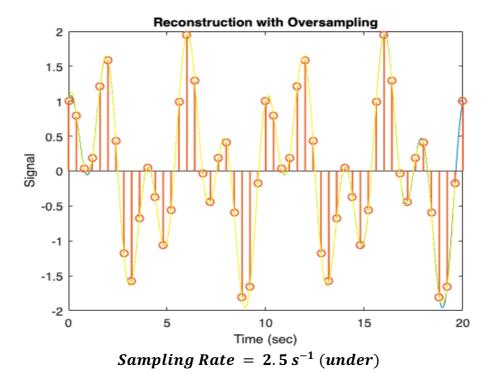
$$x_2 = \cos (2 * \pi * f_2 * t)$$



Sampling Rate = $0.50 s^{-1}$ (under)



Sampling Rate = $1 s^{-1}$ (under)



3- I decided to use a filter with windows size 5. Initially, I implemented 5-pt moving averaging filter to see if it works well or not. Unfortunately, filter output was not good enough. It was almost the same as noisy input signal.

After the first try, I decided to train a 1-dimension filter using PyTorch. I wrote a simple train code and used the given test data as my train data. (Train script is included in my submission.) After 20 epoch training with batch size 1, I obtained the trained filter coefficients and put them in my MATLAB code. Result was not perfect again, however it was better than running averaging filter, when I listened output signals. However, according to FFT output below, we can see that my trained filter could not clean the low frequency noise completely.

Trained Filter Coefficients

 $b_k = \{0.15, 0.24, 0.00, 0.35, 0.01\}$

Difference Equation

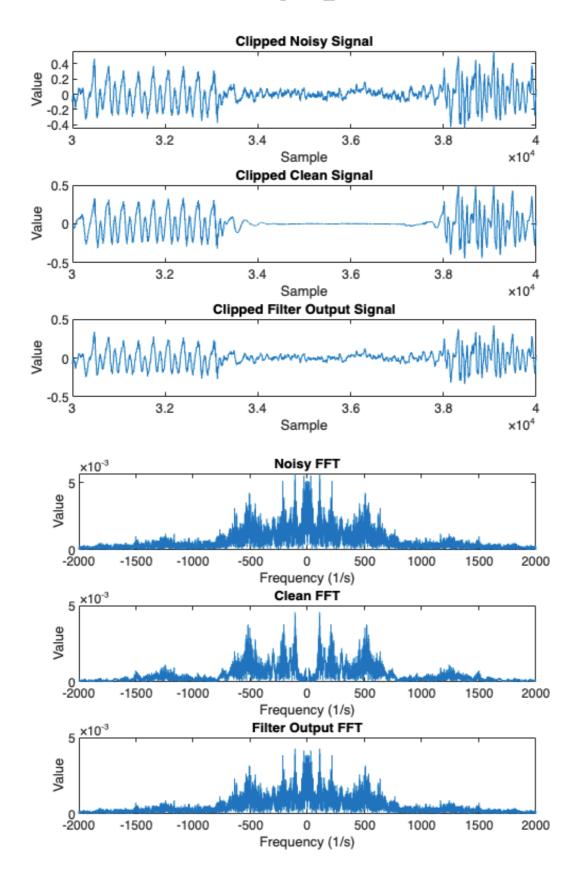
$$y[n] = 0.15 * x[n] + 0.24 * x[n-1] + 0.35 * x[n-3] + 0.01 * x[n-4]$$

System Function

$$H(z) = \frac{Y(z)}{X(z)} = 0.15 + 0.24 * z^{-1} + 0.35 * z^{-3} + 0.01 * z^{-4}$$

$$H(z) = \frac{0.15 * z^4 + 0.24 * z^3 + 0.35 * z^1 + 0.01}{z^4}$$

Test with p232_010.wav



Codes

1st Question

```
clear,clc,close all;
     figure;
     subplot(2,1,1);
     fs = 1000:
     t = 0:1/fs:0.12;
     x = (sawtooth(2*pi*25*t,0.5)+1)./2;
     plot(t,x);
    xlabel('Time (sec)');
ylabel('Signal');
11
     title("Triangle Wave with T0=0.04s")
    *<del>%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%</del>
12
    subplot(2,1,2);
   y = fftshift(fft(x));
    n = length(x);
     f0 = (-n/2:n/2-1)*(fs/n);
    normalized = abs(y/n);
     plot(f0,normalized);
    xlabel('Frequency (1/s)');
ylabel('Value');
    title("Output of fftshift")
    xlim([-350 350])
    figure;
    fs = 1000;
     t = 0:1/fs:0.12;
    subplot(2,1,1)
     x = t.*0;
    x = x + 0.5; % DC component
    for k = 1:3
        if mod(k,2)==1
            x = x + (-4/((pi*k)^2))*cos(2*pi*25*k*t);
        end
    end
    plot(t,x);
     xlabel('Time (sec)');
     ylabel('Signal');
     title("Sum of DC and 1st through 3rd Harmonics")
     <del>%%%%%%%%%%%%%%%%%%%%%%%%%%%%%</del>
    subplot(2,1,2)
    x = t.*0;
    x = x + 0.5; % DC component
     for k = 1:11
        if mod(k,2)==1
            x = x + (-4/((pi*k)^2))*cos(2*pi*25*k*t);
        end
    end
     plot(t,x);
     xlabel('Time (sec)');
ylabel('Signal');
     title("Sum of DC and 1st through 11th Harmonics")
```

2nd Question

```
clear, clc, close all;
      f1 = 0.20;
      f2 = 0.50;
      t = 0:0.001:20;
                           % CT time
     x1 = sin(2*pi*f1*t); % First CT signal
     x2 = cos(2*pi*f2*t); % Second CT signal
      % Undersampling
      figure;
11
      plot(t,x1+x2);
12
      hold on;
      fs\_under = 1*f2;
13
                          % Sampling Rate
14
      Ts_under = 1/fs_under;
      td_under = 0:Ts_under:20;
     x_under = sin(2*pi*f1*td_under) + cos(2*pi*f2*td_under);
     stem(td_under, x_under, "Color", [0.91 0.41 0.17], "LineWidth", 1.3);
17
     % Reconstruction
     y1 = zeros(1,length(t));
      samples = length(td_under);
      for i = 1:1:length(t)
21
          for n = 1:1:samples
              y1(i) = y1(i) + x_under(n)*sinc((t(i)-n*Ts_under)/Ts_under);
23
24
          end
     end
      hold on;
      plot(t-Ts_under,y1,"Color","y");
      xlim([0 20]);
     xlabel('Time (sec)');
29
30
     ylabel('Signal');
     title("Reconstruction with Undersampling")
     <del>%&&&&&&&&&&&&&&&&&</del>
     % Sampling at Nyquist Rate
34
     figure;
     plot(t,x1+x2);
     hold on;
     fs_nyquist = 2*f2;
                          % Sampling Rate
     Ts_nyquist = 1/fs_nyquist;
     td_nyquist = 0:Ts_nyquist:20;
     x_nyquist = sin(2*pi*f1*td_nyquist) + cos(2*pi*f2*td_nyquist);
     stem(td_nyquist, x_nyquist, "Color", [0.91 0.41 0.17], "LineWidth", 1.3);
     % Reconstruction
     y2 = zeros(1,length(t));
     samples = length(td_nyquist);
     for i = 1:1:length(t)
         for n = 1:1:samples
             y2(i) = y2(i) + x_nyquist(n)*sinc((t(i)-n*Ts_nyquist)/Ts_nyquist);
         end
     end
     hold on;
     plot(t-Ts_nyquist,y2,"Color","y");
     xlim([0 20]);
xlabel('Time (sec)');
ylabel('Signal');
     title("Reconstruction with Sampling at Nyquist Rate")
```

```
<del>%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%</del>
57
     % Oversampling
     figure;
59
     plot(t,x1+x2);
60
     hold on;
                         % Sampling Rate
     fs_over = 5*f2;
62
     Ts_over = 1/fs_over;
63
     td_over = 0:Ts_over:20;
     x_{over} = sin(2*pi*f1*td_over) + cos(2*pi*f2*td_over);
64
65
     stem(td_over, x_over, "Color", [0.91 0.41 0.17], "LineWidth", 1.3);
     % Reconstruction
     y3 = zeros(1,length(t));
67
     samples = length(td_over);
     for i = 1:1:length(t)
         for n = 1:1:samples
             y3(i) = y3(i) + x_over(n)*sinc((t(i)-n*Ts_over)/Ts_over);
71
         end
     end
74
     hold on;
     plot(t-Ts_over,y3,"Color","y");
76
     xlim([0 20]);
     xlabel('Time (sec)');
     ylabel('Signal');
     title("Reconstruction with Oversampling")
79
```

3rd Question

```
clear,clc,close all;
    [x_noisy,fs_noisy] = audioread('noisy.wav');
    [x_clean,fs_clean] = audioread('clean.wav');
    l_noisy=length(x_noisy);
    l_clean=length(x_clean);
    % 5-pt Averaging Filter
    %pt = 5;
    %filter = ones([1 pt]).*(1/pt);
12
    % Trained filter
    filter = [0.15, 0.24, 0.00, 0.35, 0.01];
    output = cconv(filter, x_noisy, length(x_noisy));
    audiowrite("output.wav",output,fs_noisy);
    figure;
    subplot(3,1,1);
21
    X_noisy = fftshift(fft(x_noisy));
    f_noisy = (-l_noisy/2:l_noisy/2-1)*(fs_noisy/l_noisy);
23
    noisy_fourier = abs(X_noisy/l_noisy);
    plot(f_noisy,noisy_fourier);
    xlim([-2e3 2e3]);
    xlabel('Frequency (1/s)');
    ylabel('Value');
    title("Noisy FFT")
29
    subplot(3,1,2);
    X_clean = fftshift(fft(x_clean));
    f_clean = (-l_clean/2:l_clean/2-1)*(fs_clean/l_clean);
    clean_fourier = abs(X_clean/l_clean);
    plot(f_clean,clean_fourier);
    xlim([-2e3 2e3]);
    xlabel('Frequency (1/s)');
    ylabel('Value');
    title("Clean FFT")
    <del>%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%</del>
    subplot(3,1,3);
    %plot(f_noisy, clean_fourier./noisy_fourier);
    X_output = fftshift(fft(output));
    f_output = (-l_noisy/2:l_noisy/2-1)*(fs_noisy/l_noisy);
    output_fourier = abs(X_output/l_noisy);
    plot(f_output,output_fourier);
    xlim([-2e3 2e3]);
    xlabel('Frequency (1/s)');
    vlabel('Value');
    title("Filter Output FFT")
```

```
figure;
    subplot(3,1,1);
    plot(x_noisy);
54
    xlim([3e4 4e4]);
    xlabel('Sample');
    ylabel('Value');
    title("Clipped Noisy Signal")
    subplot(3,1,2);
    plot(x_clean);
xlim([3e4 4e4]);
    xlabel('Sample');
    ylabel('Value');
64
    title("Clipped Clean Signal")
    <del>%%%%%%%%%%%%%%%%%%</del>
    subplot(3,1,3);
    plot(output);
    xlim([3e4 4e4]);
    xlabel('Sample');
ylabel('Value');
    title("Clipped Filter Output Signal")
```