Task03: Signal Processing II

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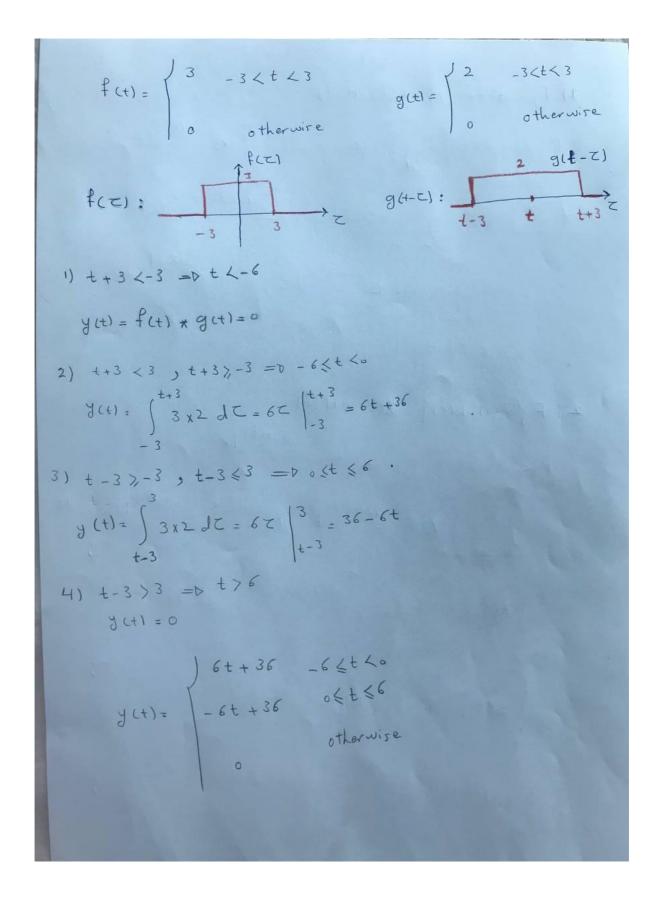
Problem 1: Convolution

$$f(t) = \begin{cases} A, & a > |t| \\ 0, & otherwise \end{cases} \qquad g(t) = \begin{cases} B, & b > |t| \\ 0, & otherwise \end{cases}$$

where A = 3, a = 3, and B = 2, b = 3

(a) Compute an analytic y(t) which is the convolution of f(t) and g(t):

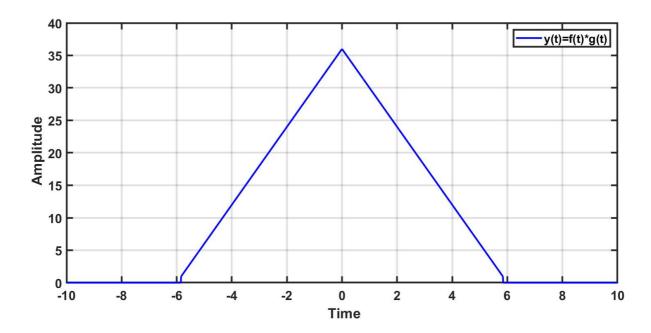
$$y(t) = f(t) * g(t)$$



(b) Write a code to numerically compute y(t) and plot y(t). Please use for-loop and do not use conv.

```
clear; clc ; close all;
syms t

A = 3; a = 3;
f = @(t) A*(abs(t)<=a);</pre>
```



(c) Write a code to numerically compute y(t) and plot y(t). Please use conv.

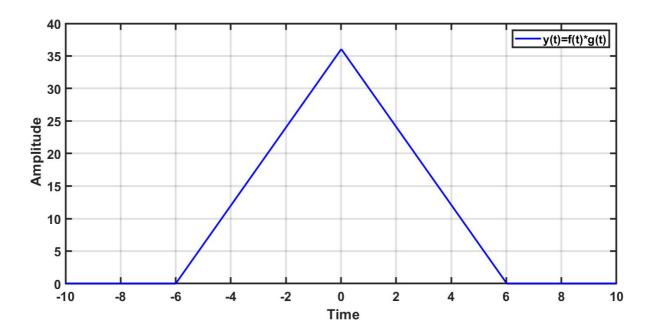
```
clear; clc ; close all;
syms t

A = 3; a = 3;
f = @(t) A*(abs(t) <= a);

B = 2; b = 3;
g = @(t) B*(abs(t) <= b);</pre>
```

```
t = -10:0.01:10;
fig1 = figure(1);
set(fig1,'Position', [100 100 1100 500]);

y = conv(f(t), g(t))*0.01; plot(t,y((floor(numel(y)/4)):floor(numel(y)/4)*3),'-b', 'linewidth', 2);hold on;
legend('y(t)=f(t)*g(t)'); axis tight;grid on;
ylabel('\bf Amplitude'); ylim([0 40])
xlabel('\bf Time');xlim([-10 10])
set(gca,'fontsize',15,'linewidth',2,'fontweight','bold');
```



Problem 2: Convolution Theorem

(a) Proof the convolution theorem and explain the meaning of these relationships in your words.

$$F\{x(t) * h(t)\} = X(f).H(f)$$

$$F\{x(t).h(t)\} = X(f) * H(f)$$

1)
$$F_{YX}(t) + h(t)^{\frac{1}{2}} = X(t) \cdot H(t)$$
 $F_{YX}(t) + h(t)^{\frac{1}{2}} = X(t) \cdot h(t) \cdot dt^{\frac{1}{2}} e^{-i2\pi t} dt$
 $V = t - \overline{Z}$
 $= \int_{-\infty}^{\infty} X(t) \cdot h(t) \cdot e^{-i2\pi t} dt$
 $= \int_{-\infty}^{\infty} X(t) \cdot h(t) \cdot e^{-i2\pi t} dt$
 $= (2\pi t) \cdot h(t)^{\frac{1}{2}} = X(t) \cdot h(t) \cdot e^{-i2\pi t} dt$
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From the above equations, it is concluded that the Fourier transform of the convolution of two time signals can be obtained by the multiplication of these signals in frequency domain. Moreover, the Fourier transform of the multiplication of two time signals is equal to the convolution of these two signals while they are in frequency domain.

(b) Compute a Fourier transform of the triangular function in both analytic and numeric ways (Note that this function is not periodic):

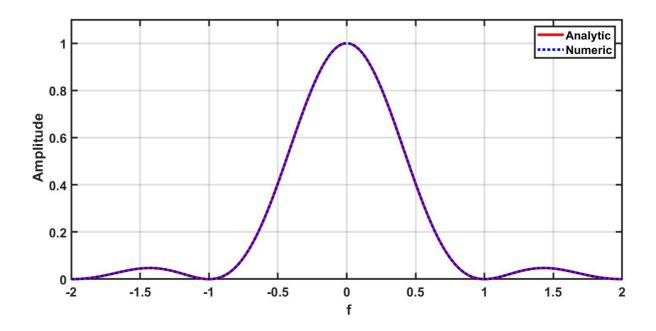
$$x(t) = \begin{cases} 1 - |t| & |t| < 1 \\ 0 & |t| > 1 \end{cases}$$

$$x(t) = \int_{0}^{\infty} |-1t| \qquad |+| < 1$$

$$x(t) = \int_{0}^{\infty} x(t) e^{-i2nt} dt = \int_{0}^{\infty} (1+t) e^{-i2nt} dt + \int_{0}^{\infty} (1-t) e^{-i2nt} dt$$

$$= \int_{0}^{\infty} e^{-i2nt} dt + \int_{0}^{\infty} t e^{-i2nt} dt - \int_{0}^{\infty} t e^{-i2nt} dt + \int_{0}^{\infty} (-i2nt) dt + \int_{0}^{\infty} (-$$

```
clear; clc ; close all;
t = -1:0.01:1;
f = -2:0.01:2;
x = Q(t) (1-abs(t))*(abs(t)<=1);
% Analytic
x_{ana} = @(f) (sin(pi*f)./(pi*f)).^2;
% Numeric
Integ = @(t) x(t)*exp(-i*2*pi*f*t);
x_num = @(f) integral(@(t) Integ(t), -inf, inf, 'ArrayValued', true);
fig1 = figure(1); set(fig1, 'Position', [100 100 1100 500]);
plot(f,abs(x_ana(f)),'-r', 'linewidth', 3); hold on;
plot(f,abs(x_num(f)),':b', 'linewidth', 3); hold off;
legend('Analytic', 'Numeric');
axis tight;grid on;
ylabel('\bf Amplitude'); ylim([0 1.1])
xlabel('\bf f'); xlim([-2 2])
set(gca,'fontsize',15,'linewidth',2,'fontweight','bold');
```



(c) Please explain the result in (b) using your answers for Problem 1.

From problem 1, we can understand that a triangular function is made by the convolution of two rectangular functions. Also, we already know that the Fourier transform of a rectangular time signal is equal to sinc(f). Therefore, using the first equation of problem 2- (a), the Fourier transform of the triangular signal in (b) becomes sinc(f).sinc(f).

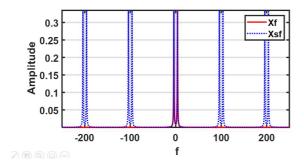
Problem 3: Discrete Fourier Transform 1

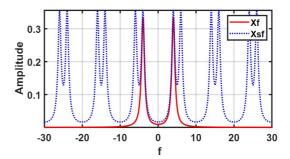
(a) Please explain the difference between the following three notations in the lecture slide:

$$X(f), X_s(f), X(k)$$

X(f) is the Fourier transform of the original time signal. However, X(f) is the Fourier transform of the sampled time signal which is continuous in frequency. Moreover, X(k) is the discrete Fourier transform of a discrete finite sequence. In fact, X(k) is X(f) evaluated at $f = k/(N\Delta)$ where k is an integer. It is worth mentioning that X(K) is discrete.

(b) What do these two graphs explain in the lecture slides?





In the left graph, the Nyquist frequency is higher than the frequency of the signal, so there is no aliasing. That is why the Fourier transforms of the original signal and the sampled one are the same at the frequency domain of the original signal. However, Xs(f) which is the Fourier transform of the sampled signal is periodic and continuous while the original signal is discrete (this is obvious in the graph). On the other hand, in the right graph, the Nyquist frequency is less than the frequency of the signal. Therefore, aliasing happens and that is why X(f) and Xs(f) are different for the frequencies higher than fN. The Fourier transform of this sampled signal is periodic either.

(c) What is the meaning of the following relationship in the lecture slide? Please explain it.

$$X_{s}(f+r/\Delta) = \sum_{n=-\infty}^{\infty} x(n\Delta)e^{-i2\pi(f+r/\Delta)n\Delta} = \sum_{n=-\infty}^{\infty} x(n\Delta)e^{-i2\pi fn\Delta - i2\pi rn} = \sum_{n=-\infty}^{\infty} x(n\Delta)e^{-i2\pi fn\Delta} = X_{s}(f)$$

This equation means that although the signal is discrete, the Fourier transform of the sampled signal is periodic; and its period is $1/\Delta$ which is equal to the sampling frequency.

(d) What is the meaning of the following relationship in the lecture slide? What issues are introduced by Fourier transform a discrete sequence? Please answer this question using this graph.

$$X_s(f) = I(f) * X(f) = \int_{-\infty}^{\infty} I(g)X(f-g)dg = \int_{-\infty}^{\infty} \frac{1}{\Delta} \sum_{n=-\infty}^{\infty} \delta\left(g - \frac{n}{\Delta}\right)X(f-g)dg = \frac{1}{\Delta} \sum_{n=-\infty}^{\infty} \int_{-\infty}^{\infty} \delta\left(g - \frac{n}{\Delta}\right)X(f-g)dg$$

$$=\frac{1}{\Delta}\sum_{n=-\infty}^{\infty}X\left(f-\frac{n}{\Delta}\right)=\frac{1}{\Delta}\left(\dots+X\left(f-\frac{2}{\Delta}\right)+X\left(f-\frac{1}{\Delta}\right)+X(f)+X\left(f+\frac{1}{\Delta}\right)+\dots\right)$$

The above relationship means that the sampled signal in frequency domain (Xs(f)) can be obtained from the summation of the repeated original signal in frequency domain (X(f)), in every sampling frequency (fs=1/ Δ). However, in each shifted signal, for frequencies above the Nyquist frequency (fs/2=1/2 Δ) aliasing occurs and that is why the frequencies above fN cannot be measured accurately by sampling. As it is obvious in the graph, the frequencies around fN are wrapped around because of aliasing.

Problem 4: Discrete Fourier Transform 2 - Use FFT

$$y_1(t) = e^{-a|t|}(b.\cos 2\pi f_1 t + c.\cos 2\pi f_2)$$

where a = 2, b= 2, c=6, f1=3, and f2=6

$$y_2(t) = e^{-a|t|}(b.\cos 2\pi f_1 t + c.\cos 2\pi f_2)$$

where a = 0.3, b= 10, c=3, f1=5, and f2=8

(a) z1 and z2 are discrete signals, which are obtained by digitizing (sampling) y1(t) and y2(t) with a sampling rate of 50 Hz and collecting them for 5 seconds, respectively. Please plot z1 and z2 in the time domain (include a proper time axis).

```
clear; clc; close all;

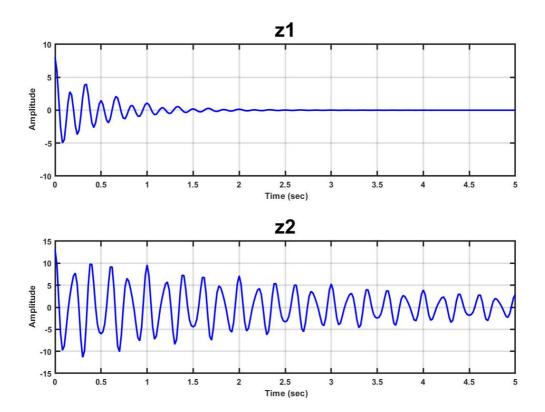
syms t

a1 = 2;
b1 = 2;
c1 = 6;
f_11 = 3;
f_12 = 6;

a2 = 0.3;
b2 = 10;
c2 = 3;
f_21 = 5;
f_22 = 8;

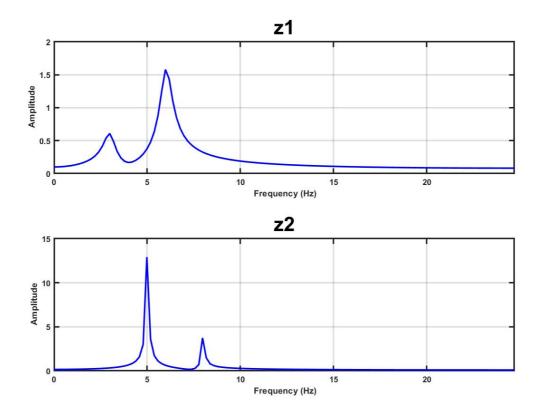
y1 = exp(-1*a1*abs(t)).*(b1*cos(2*pi*f_11*t)+c1*cos(2*pi*f_12*t));
y2 = exp(-1*a2*abs(t)).*(b2*cos(2*pi*f_21*t)+c2*cos(2*pi*f_22*t));
```

```
T = 5;
fig1 = figure(1);
set(fig1,'Position', [100 100 1000 700]);
Fsd = 50; % # of samples per a second
td = 0:1/Fsd:T;
yd1 = subs(y1, t, td);
yd2 = subs(y2, t, td);
subplot(211);
plot(td,yd1,'-b','linewidth',2);
axis tight;grid on;
ylabel('\bf Amplitude'); ylim([-10 10]);
xlabel('\bf Time (sec)');
title('z1','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
subplot(212);
plot(td,yd2,'-b','linewidth',2);
axis tight;grid on;
ylabel('\bf Amplitude'); ylim([-15 15]);
xlabel('\bf Time (sec)');
title('z2','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
```

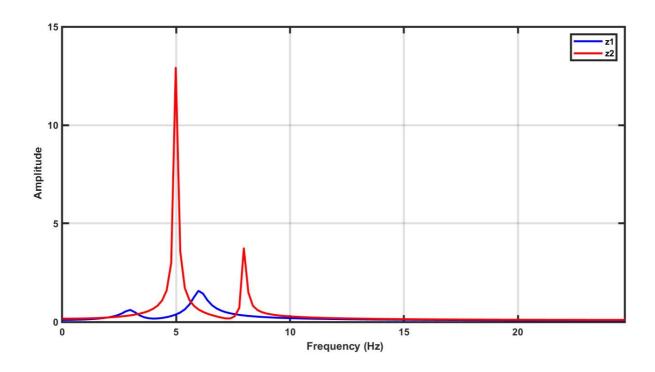


(b) Perform the discrete Fourier transform of z1 and z2, and plot your graphs in the frequency domain (include a proper frequency axis). Plot only positive frequency signals with absolute values.

```
clear; clc ; close all;
syms t
a1 = 2;
b1 = 2;
c1 = 6;
f_{11} = 3;
f_{12} = 6;
a2 = 0.3;
b2 = 10;
c2 = 3;
f_21 = 5;
f_22 = 8;
y1 = exp(-1*a1*abs(t)).*(b1*cos(2*pi*f_11*t)+c1*cos(2*pi*f_12*t));
y2 = \exp(-1*a2*abs(t)).*(b2*cos(2*pi*f_21*t)+c2*cos(2*pi*f_22*t));
T = 5;
fig1 = figure(1);
set(fig1, 'Position', [100 100 1000 700]);
Fsd = 50; % # of samples per a second
delta = 1/Fsd;
td = 0:delta:T;
yd1 = double(subs(y1, t, td));
yd2 = double(subs(y2, t, td));
% use of a 'fft' function
nfft = numel(yd1);
yk1 = fft(yd1, nfft);
yk2 = fft(yd2, nfft);
f = 1/(nfft*delta) * (0:nfft/2-1);
subplot(211);
plot(f,1/Fsd*abs(yk1(1:floor(nfft/2))),'-b','linewidth',2);
axis tight;grid on;
ylabel('\bf Amplitude'); ylim([0 2]);
xlabel('\bf Frequency (Hz)');
title('z1','fontsize',25);
set(gca, 'linewidth', 2, 'fontweight', 'bold');
subplot(212);
plot(f,1/Fsd*abs(yk2(1:floor(nfft/2))),'-b','linewidth',2);
axis tight; grid on;
ylabel('\bf Amplitude'); ylim([0 15]);
xlabel('\bf Frequency (Hz)');
title('z2','fontsize',25);
```



(c) Please compare the shape of the frequency curves of z1 and z2. Which frequency curve is thinner (more narrow)? For example, compare the frequency curve at f1 in both graphs. Which one is thinner? Please explain your answer. What makes the difference?



From the above graph, it is observed that the frequency curve of z2 is thinner than that of z1. This can be explained by Inverse spreading property of the Fourier transform. z2 is wider than z1 in time domain; therefore, it is narrower in frequency domain.

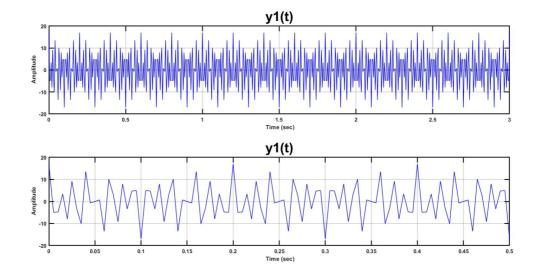
Problem 5: Discrete Fourier Transform 3 - Use FFT

$$y(t) = A1\cos 2\pi(25)t + A2\cos 2\pi(45)t + A3\cos 2\pi(75)t$$

where A1=2, A2=5, and A3=10.

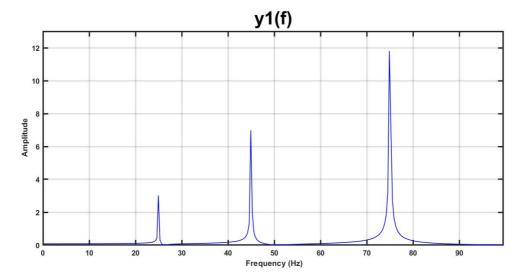
(a) y1 is a discrete signal, which is obtained by digitizing y(t) with a sampling rate of 200 Hz for 3 seconds. Please plot y1 in the time domain (include a proper time axis). Please connect the sampled points.

```
clear; clc ; close all;
A1=2;
A2=5;
A3=10;
f1 = 25;
f2 = 45;
f3 = 75;
T = 3;
fs = 200; % # of samples per a second
t = 0:1/fs:T;
y1 = A1*cos(2*pi*f1*t)+A2*cos(2*pi*f2*t)+A3*cos(2*pi*f3*t);
fig1 = figure(1);
set(fig1, 'Position', [100 100 1500 700]);
subplot(211);
plot(t,y1,'b', 'linewidth', 1);
axis tight; grid on;
ylabel('\bf Amplitude' ); ylim([-20 20]);
xlabel('\bf Time (sec)');
title('y1(t)','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
subplot(212);
plot(t,y1,'b', 'linewidth', 1);
axis tight;grid on;
ylabel('\bf Amplitude' ); ylim([-20 20]);
xlabel('\bf Time (sec)'); xlim([0 0.5]);
title('y1(t)','fontsize',25);
set(gca, 'linewidth', 2, 'fontweight', 'bold');
```



(b) Perform the discrete Fourier transform of y1 and plot your graph in the frequency domain (include a proper frequency axis). Plot only a positive frequency signal. Can we measure all three frequencies?

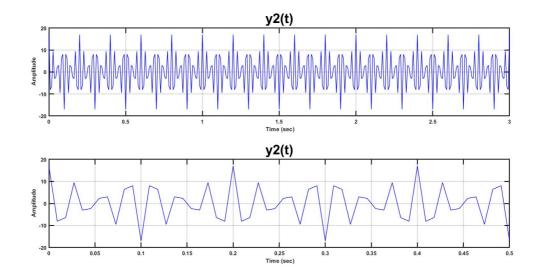
```
clear; clc ; close all;
A1=2;
A2=5;
A3=10;
f1 = 25;
f2 = 45;
f3 = 75;
T = 3:
fs = 200; % # of samples per a second
t = 0:1/fs:T;
y1 = A1*cos(2*pi*f1*t)+A2*cos(2*pi*f2*t)+A3*cos(2*pi*f3*t);
% use of a 'fft' function
nfft = numel(t);
yk1 = fft(y1, nfft);
delta = 1/fs;
f = 1/(nfft*delta) * (0:nfft/2-1);
fig1 = figure(1);
set(fig1,'Position', [100 100 2000 500]);
plot(f,1/fs*abs(yk1(1:(nfft/2))),'b', 'linewidth', 1);
axis tight;grid on;
ylabel('\bf Amplitude' ); ylim([0 13]);
xlabel('\bf Frequency (Hz)');
title('y1(f)','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
```



Based on the above graph, we are able to measure all three frequencies (25, 45, and 75 Hz) with 200 Hz sampling rate because all the aforementioned frequencies are less than the Nyquist frequency (100 Hz).

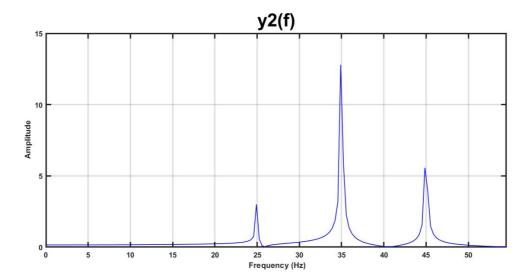
(c) y2 is a discrete signal, which is obtained by digitizing y(t) with a sampling rate of 110 Hz for 3 seconds. Please plot y2 in the time domain (include a proper time axis). Please connect the sampled points.

```
clear; clc ; close all;
A1=2;
A2=5;
A3=10;
f1 = 25;
f2 = 45;
f3 = 75;
T = 3;
fs = 110; % # of samples per a second
t = 0:1/fs:T;
y2 = A1*cos(2*pi*f1*t)+A2*cos(2*pi*f2*t)+A3*cos(2*pi*f3*t);
fig1 = figure(1);
set(fig1, 'Position', [100 100 1500 700]);
subplot(211);
plot(t,y2,'b', 'linewidth', 1);
axis tight;grid on;
ylabel('\bf Amplitude' ); ylim([-20 20]);
xlabel('\bf Time (sec)');
title('y2(t)','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
subplot(212);
plot(t,y2,'b', 'linewidth', 1);
axis tight;grid on;
ylabel('\bf Amplitude' ); ylim([-20 20]);
xlabel('\bf Time (sec)'); xlim([0 0.5]);
title('y2(t)','fontsize',25);
set(gca, 'linewidth', 2, 'fontweight', 'bold');
```



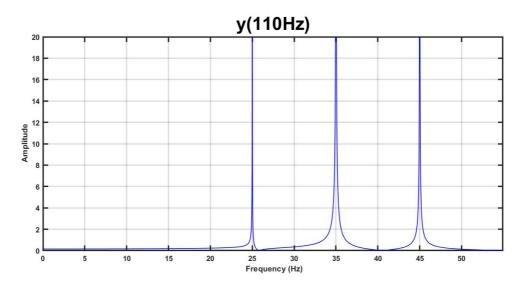
(d) Perform the discrete Fourier transform of y2 and plot your graph in the frequency domain (include a proper frequency axis). Plot only a positive frequency signal. Can we measure all three frequencies?

```
![26](C:\Users\Asus\Desktop\waterloo\smart structures\signal
processing2\26.jpg)clear; clc; close all;
A1=2;
A2=5;
A3=10;
f1 = 25;
f2 = 45;
f3 = 75;
T = 3;
fs = 110; % # of samples per a second
t = 0:1/fs:T;
y2 = A1*cos(2*pi*f1*t)+A2*cos(2*pi*f2*t)+A3*cos(2*pi*f3*t);
% use of a 'fft' function
nfft = numel(t);
yk2 = fft(y2, nfft);
delta = 1/fs;
f = 1/(nfft*delta) * (0:nfft/2-1);
fig1 = figure(1);
set(fig1, 'Position', [100 100 1100 500]);
plot(f,1/fs*abs(yk2(1:(nfft/2))),'b', 'linewidth', 1);
axis tight; grid on;
ylabel('\bf Amplitude' ); ylim([0 15]);
xlabel('\bf Frequency (Hz)');
title('y2(f)','fontsize',25);
set(gca, 'linewidth', 2, 'fontweight', 'bold');
```



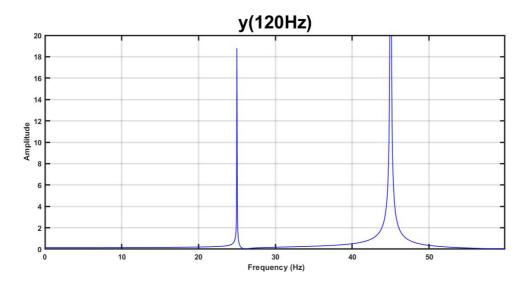
Based on the above graph, we can capture frequencies of 25 and 45 Hz but we are unable to measure frequency of 75 Hz with 110 Hz sampling rate. This is due to the fact that the Nyquist frequency (55 Hz) is less than 75 Hz; therefore, aliasing occurs for the frequency of 75 Hz. It can be seen in the graph that instead of measuring 75 Hz frequency, aliasing frequency of 35 Hz is captured.

(e) If you digitize a longer-duration signal (let's say 100 seconds) with a sampling rate of 110 Hz, can you measure and extract all three frequencies contained in the original signal, y(t)? Please explain your answer.



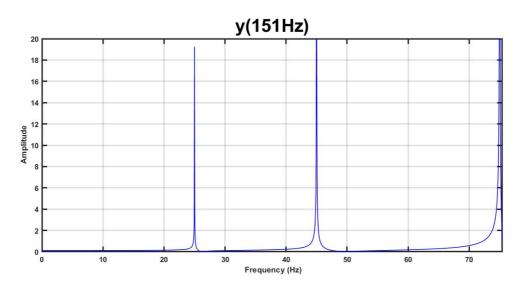
No, we cannot measure frequency of 75 Hz even after utilizing a longer time duration (Zero-Padding Effect) because the Nyquist frequency (55 Hz) is still less than 75 Hz.

(f) If you digitize the signal with a sampling rate of 120 Hz for 20 seconds, can you measure and extract all frequencies contained in the original signal, y(t)? please explain your answer.



With a sampling rate of 120 Hz, we are able to measure frequencies of 25 and 45 Hz. However, we cannot capture the frequency of 75 Hz because the Nyquist frequency (60 Hz) is still less than 75 Hz.

(g) If you digitize the signal with a sampling rate of 151 Hz for 20 seconds, can you measure and extract all frequencies contained in the original signal, y(t)? please explain your answer.



using a sampling rate of 151 Hz, we are able to measure all three frequencies (25, 45, and 75 Hz) since all these frequencies are less than the Nyquist frequency (75.5 Hz).

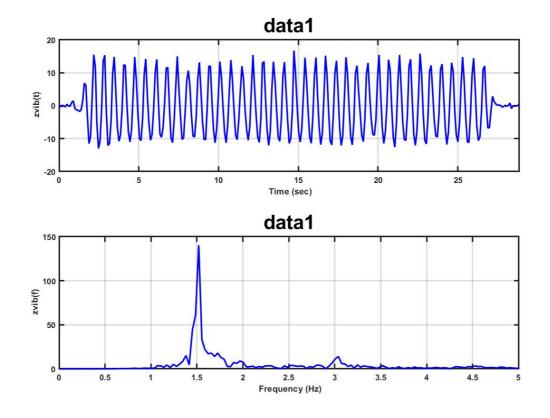
Problem 6: Frequency Analysis

Two sinusoidal accelerations are measured using an accelerometer in a smartphone. Each of the waves is stored at data1.mat and data2.mat.

(a) Load data1.mat and plot the acceleration signal in a z direction (zvib). What is the main frequency of this wave? The sampling frequency is 10.1355 Hz.

After loading data1.mat, I noticed that the sampling frequency is 10.1192 Hz not 10.1355.

```
clear; clc ; close all;
load('data1.mat')
T = 0:length(zvib)-1;
t = T/fs;
% use of a 'fft' function
nfft = numel(t);
Xk1 = fft(zvib, nfft);
delta = 1/fs;
f = 1/(nfft*delta) * (0:nfft/2-1);
fig1 = figure(1);
set(fig1, 'Position', [100 100 1000 700]);
subplot(211);
plot(t,zvib,'-b','linewidth',2);
axis tight;grid on;
ylabel('\bf zvib(t)'); ylim([-20 20]);
xlabel('\bf Time (sec)');
title('data1','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
subplot(212);
plot(f,1/fs*abs(Xk1(1:floor(nfft/2))),'-b','linewidth',2);hold on;
axis tight;grid on;
ylabel('\bf zvib(f)'); ylim([0 150]);
xlabel('\bf Frequency (Hz)');
title('data1','fontsize',25);
set(gca,'linewidth',2,'fontweight','bold');
y = [1/fs*abs(xk1(1:floor(nfft/2)))]';
[y_max,n] = max(y);
frequency_main=f(n)
```



```
frequency_main = 1.5196
```

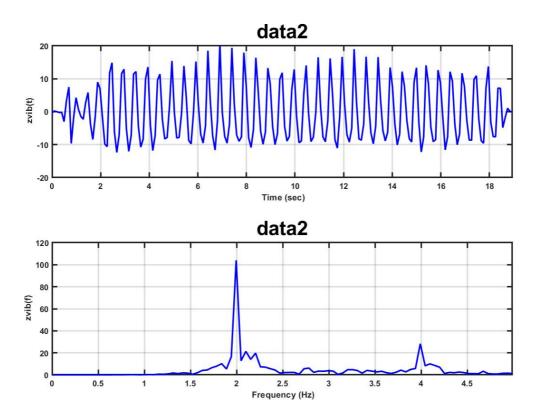
(b) Load data2.mat and plot the acceleration signal in a z direction (zvib). What is the main frequency of this wave? The sampling frequency is 10.1192 Hz.

After loading data2.mat, I noticed that the sampling frequency is 10.1355 Hz not 10.1192.

```
clear; clc ; close all;
load('data2.mat')
T = 0:length(zvib)-1;
t = T/fs;
% use of a 'fft' function
nfft = numel(t);
Xk1 = fft(zvib, nfft);
delta = 1/fs;
f = 1/(nfft*delta) * (0:nfft/2-1);
fig1 = figure(1);
set(fig1, 'Position', [100 100 1000 700]);
subplot(211);
plot(t,zvib,'-b','linewidth',2);
axis tight;grid on;
ylabel('\bf zvib(t)'); ylim([-20 20]);
xlabel('\bf Time (sec)');
title('data2','fontsize',25);
```

```
set(gca, 'linewidth', 2, 'fontweight', 'bold');
subplot(212);
plot(f, 1/fs*abs(xk1(1:floor(nfft/2))), '-b', 'linewidth', 2); hold on;
axis tight; grid on;
ylabel('\bf zvib(f)'); ylim([0 120]);
xlabel('\bf Frequency (Hz)');
title('data2', 'fontsize', 25);
set(gca, 'linewidth', 2, 'fontweight', 'bold');

y = [1/fs*abs(xk1(1:floor(nfft/2)))]';
[y_max,n] = max(y);
frequency_main=f(n)
```



 $frequency_main = 1.9956$