MAE/ECE-5320 LAB 02: DIGITAL AND DISCRETE SYSTEM

Section 1. Sampling and quantization

<u>There would be three main tasks to be finished in this section.</u>

<u>First Task: sampling rate and aliasing effect</u>

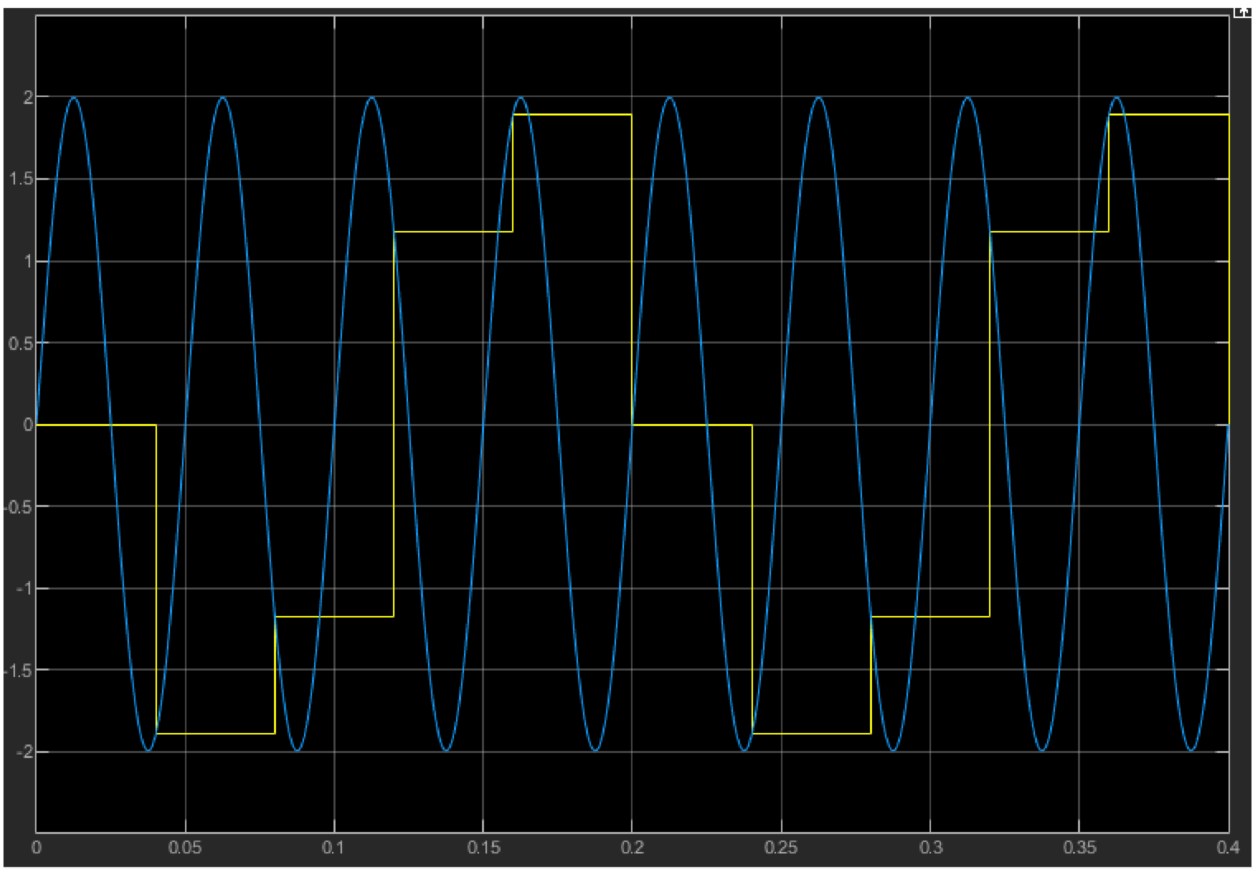
Edit the parameters in Sine Wave block as below to generate signal $2sin\,(40\pi t)$. What is the frequency (in unit Hz) of this signal, write down your answer.

A: The signal frequency is $20Hz$.

Step 1: Edit the parameters in Quantizer block as below, where 4 is the range of the signal from negative to positive, ***8 bits represent*** 2^8 ***resolutions, and the number of intervals is*** $2^8 - 1$

Step 2: Edit the parameters in Zero-Order Hold block with sample time of 0.04 second. Write down the sample frequency (in Hz unit) in the short form and show the simulation results in scope.

A: The sample frequency is $25Hz$.



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Step 3: What is the minimum sampling rate required to sample the sinusoidal signal $2sin\ (40\pi t)$? From the results in step 2, is the sampling rate is enough so that sampled point can represent the original signal ?

A: Minimum sampling rate is double the frequency of original signal, 2*20 Hz = 40 Hz. The sample frequency is lower than Nyquist sample rate (40 Hz) so the sampled signal cannot represent the continuous signal.

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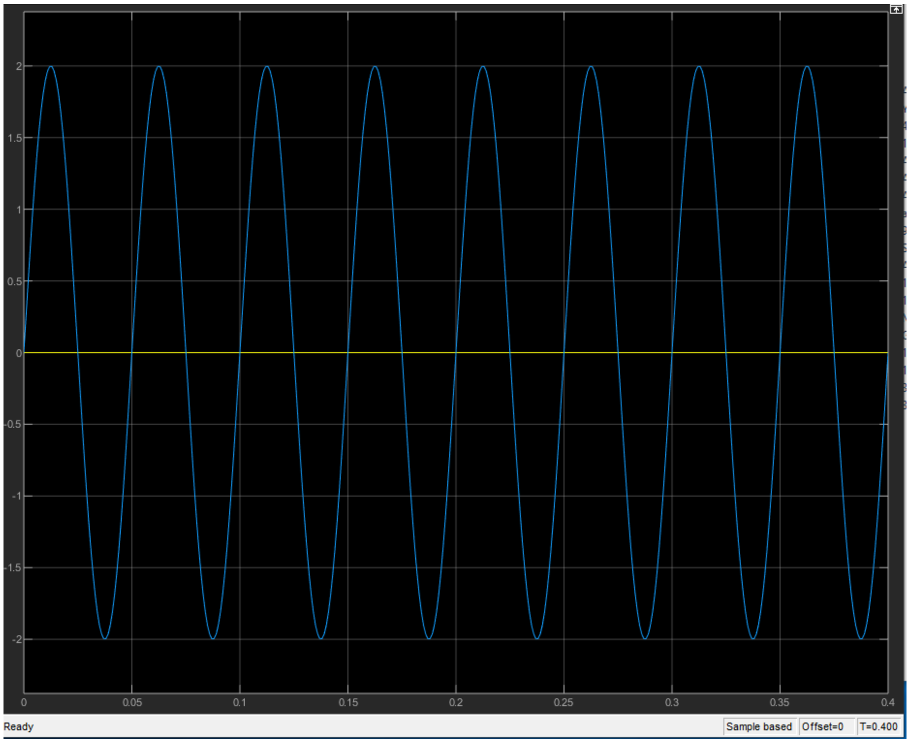
Second Task: sampling rate influences on sampled results

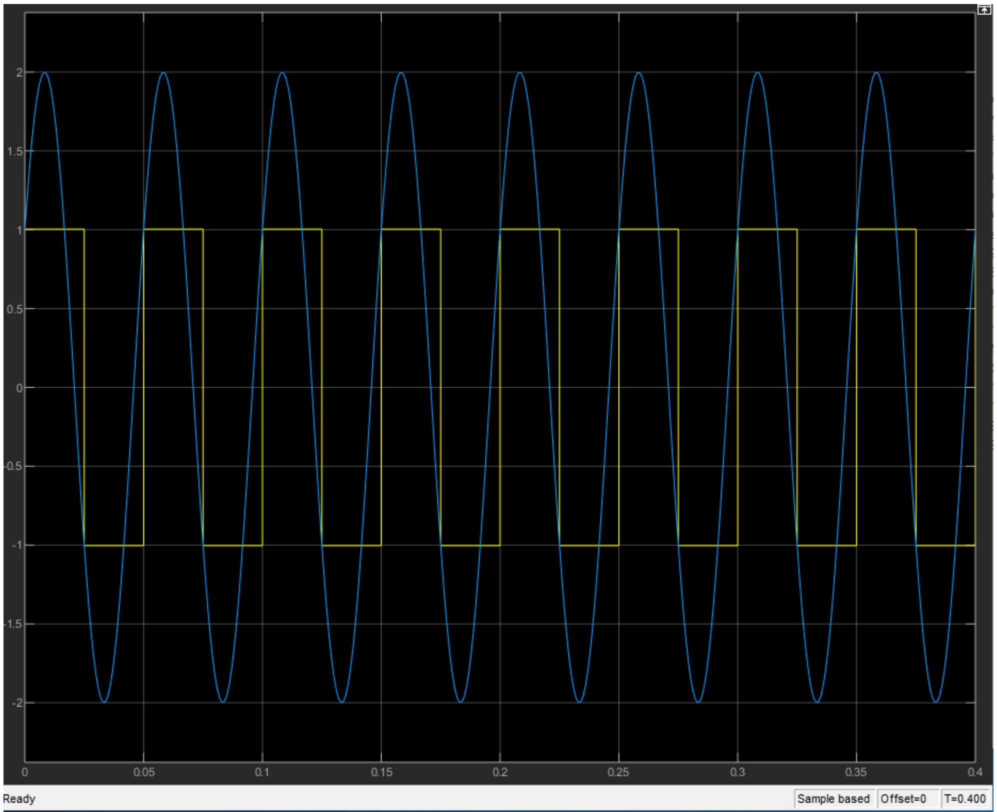
Step 1: Keep the same configuration for Sine signal and quantizer.

Step 2: Edit the parameters in Zero-Order Hold block with sample time of 0.025s. Write down the sample frequency (Hz unit) in the short form and show the simulation results.

A: The sample frequency is $40Hz$.



Step 3: Edit the parameters in Zero-Order Hold block with sample time of 0.025s. Add $\frac{\pi}{6} rad$ phase delay into the original sin wave which is still a 20Hz frequency signal. Show the final simulation results. Has the frequency and amplitude of the signal been recovered?



A: The sampling start point has been changed, so the frequency of original signal can be recovered which follows the Nyquist rate theorem. However, the amplitude of sampled signal is still distorted.

Step 4: Edit the parameters in Zero-Order Hold block with sample time of 0.005s. Write down the sample frequency (Hz unit) in the short form and show the simulation results.

A: The sample frequency is $200Hz$.

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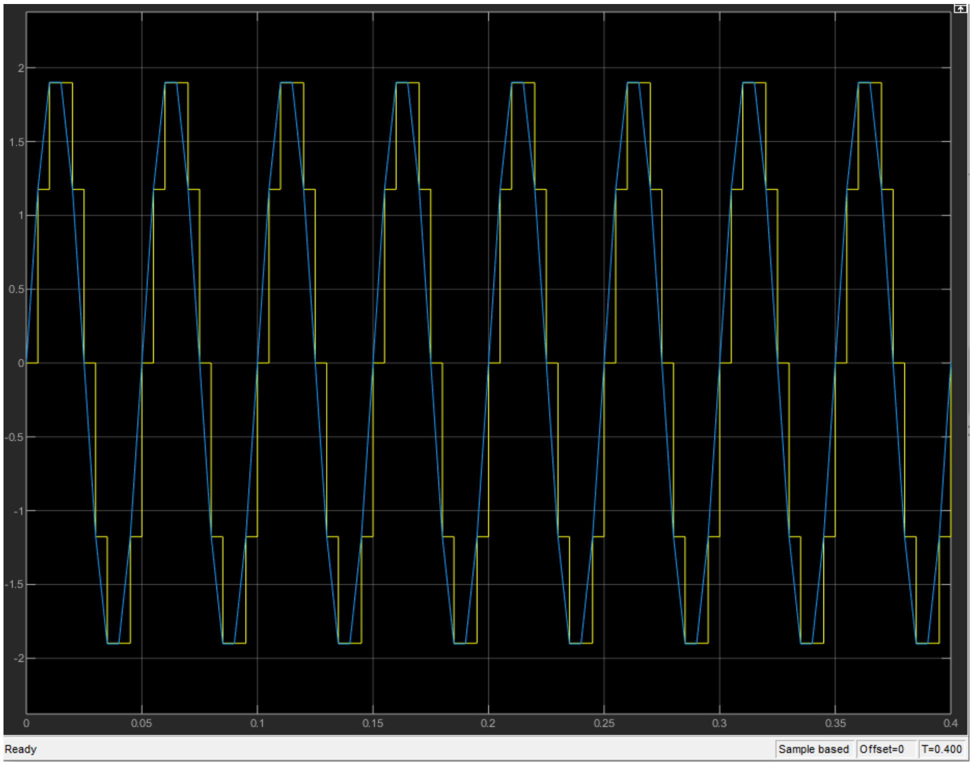
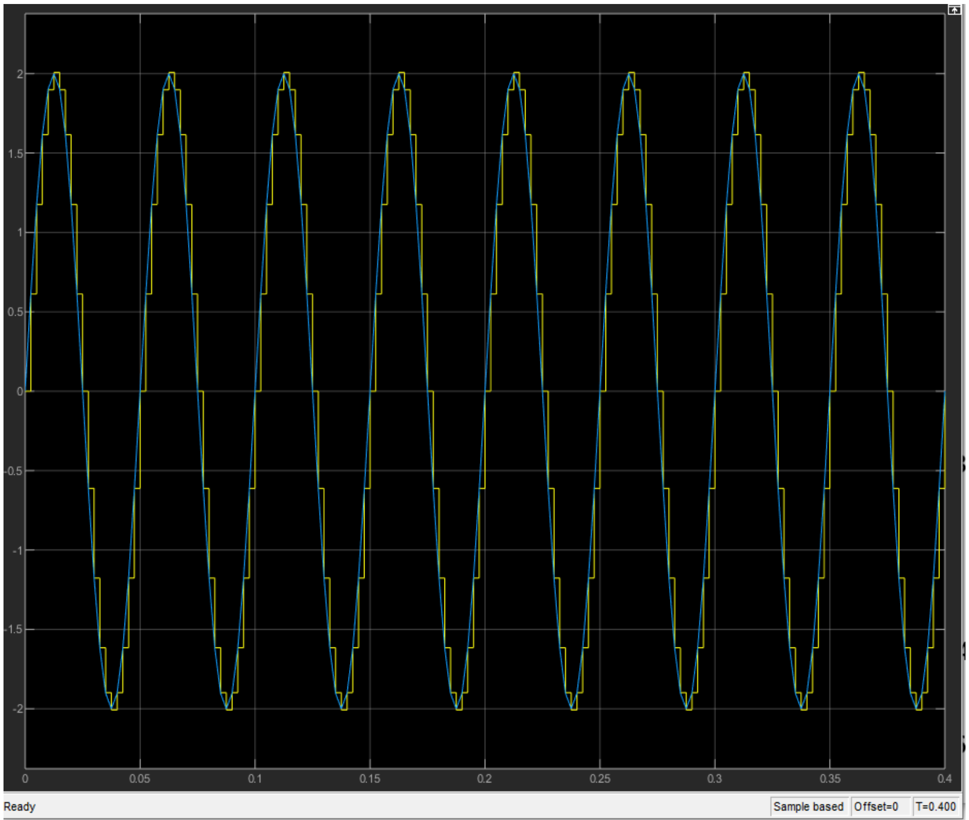


Step 5: Edit the parameters in Zero-Order Hold block with sample time of 0.0025s. Write down the sample frequency (Hz unit) in the short form and show the simulation results

A: The sample frequency is $400Hz$.



Step 6: Compare results from Steps 2, 4, and 5, and describe how sample time influence sampled results.

A: If the sampling rate f_s is *equal to or* two times *more than* the *highest* signal frequency f_h (bandwidth), the signal can be recovered. The higher the sample frequency, the more accurate the sampled signal.

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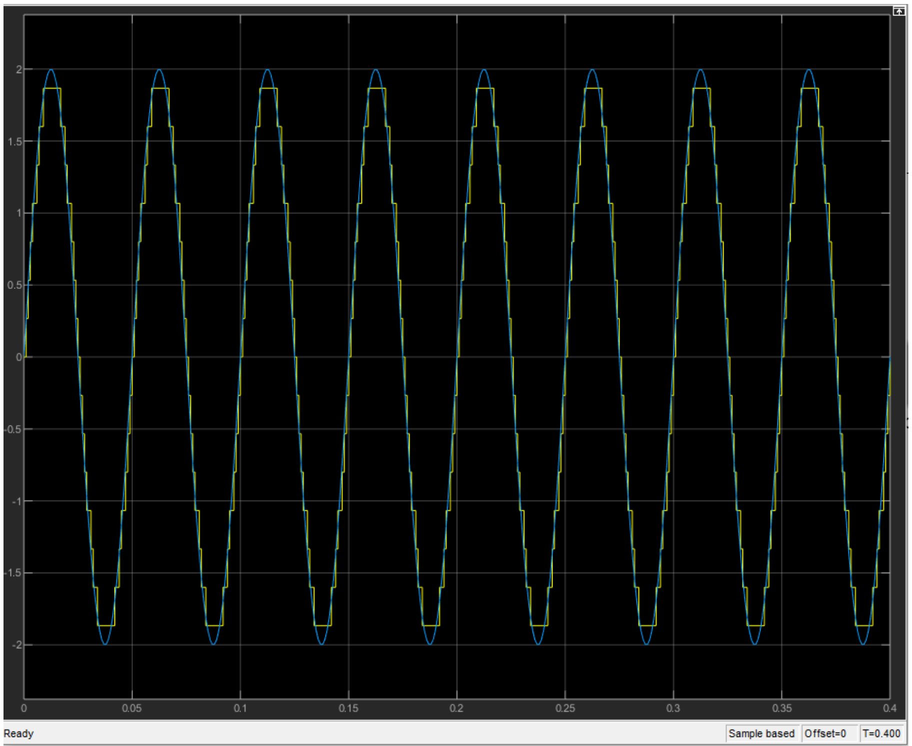
<u>Third Task: quantization resolution depends on quantization bits</u>

Step 1: Keep the same configuration for Sine signal but set sample time to 0.001s.

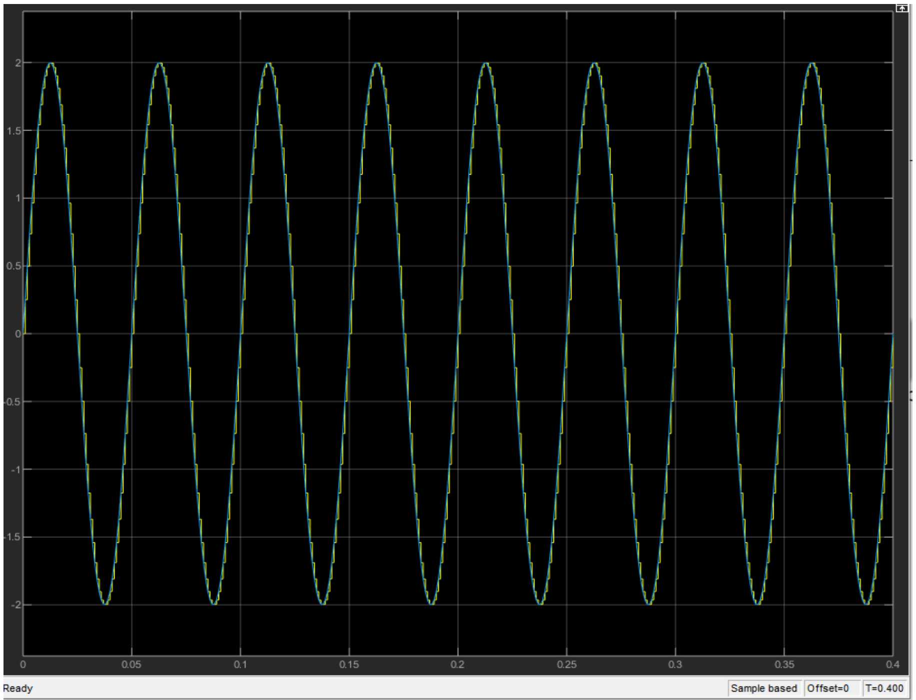
Step 2: Set the bits as 4 in Quantizer block. What value should be put in the quantization interval of the quantizer block? Write down the answer and attach the simulation results in scope.

Quantization Interval number is 4/(2^4-1).



Step 3: Now we try to set the bits as 16 in Quantizer block. What value should be put in the quantization interval of the quantizer block? Write down the answer and attach the simulation results in scope.

Quantization Interval number is 4/(2^16-1).



Step 4: Compare results from Steps 2 and 3, and describe how quantizer bits influence sampled results.

A: The accuracy of digitized signals depends on quantizer bits used. The error of quantization reduces as the quantizer bit increases.

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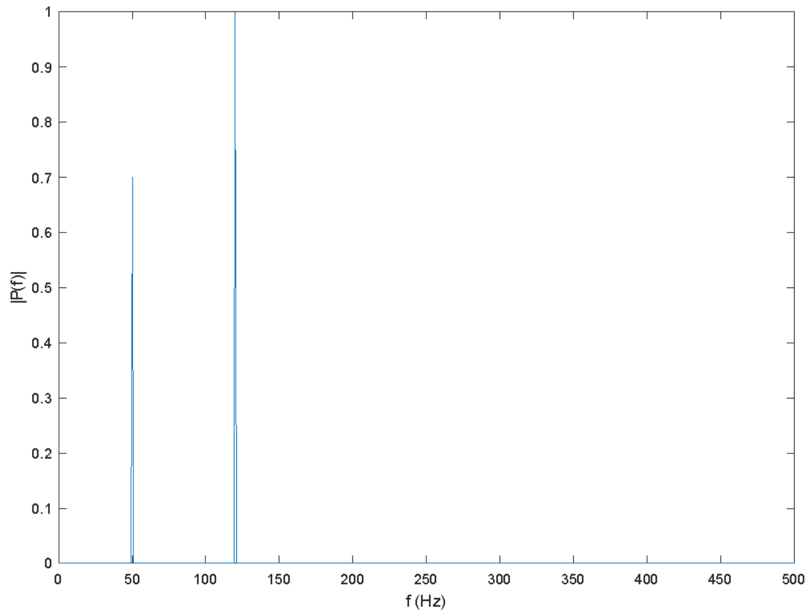
Section 2. FFT of Signals

2.1 FFT Code for Single Sided Spectrum

2.2 Exercise

Exercise 1:

Step 1: run the example code and attach the plot of frequency distribution.



Step 2: What are the frequencies that have peaks? are they identical to the two frequency components in the continuous signal?

50 Hz, 120 Hz. Yes, they are identical to the signal frequency components.

Exercise 2:

We still use the example code, and try to get the FFT response of the function $y = 0.7$ $sin\ sin\ (2pi * 50t)\ + sin\ sin\ (2pi * 100t)\ + 2sin\ (2 * pi * 100400 * t)$.

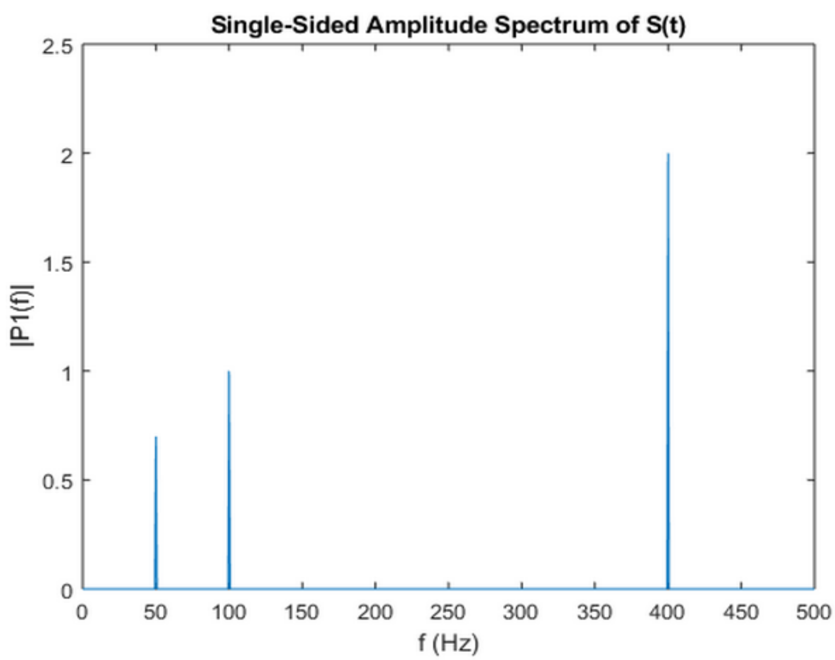
A: Reference code is listed below.

```
Fs = 1000;              % Sampling frequency
T = 1/Fs;               % Sampling period
L = 1500;               % Length of signal
t = (0:L-1)*T;          % Time vector
third_frequency=100400;
S = 0.7*sin(2*pi*50*t) + sin(2*pi*100*t)  + 2*sin(2*pi*third_frequency*t);
f = Fs*(0:(L/2))/L;
Y = fft(S);
P2 = abs(Y/L);
P1 = P2(1:L/2+1);
P1(2:end-1) = 2*P1(2:end-1);

plot(f,P1)
title('Single-Sided Amplitude Spectrum of S(t)')
xlabel('f (Hz)')
ylabel('|P1(f)|')
```

2.2.1 Calculate the frequency components of the new signal.
A: The frequency should be 50Hz, 10Hz, and 100400Hz, respectively.

2.2.2 Run the example code for the new signal, and attach the FFT plot. What are three frequency components that have peaks on FFT plot.
A: The result is shown below and they are 50Hz, 100Hz, and 400Hz, respectively.



2.2.3 Compare the results of 2.2.1 and 2.2.2. Are the peak frequencies in FFT plot identical to the signal frequency components? Try to explain why?

A: They are not the same, because the sample frequency in example code is 1000 Hz, which is too low and results in the aliasing phenomenon. The Sin wave of 100400 Hz was folded into a signal with a frequency of 400 Hz.