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

Before start, you need

1. **PC installed with MATLAB.**

Section 1. Sampling and quantization

In the lecture, the following graph was introduced so that we have gained the basic idea about D/A converter and A/D convert. In this lab section, we will use ~~the~~ Simulink to simulate AD convert process.

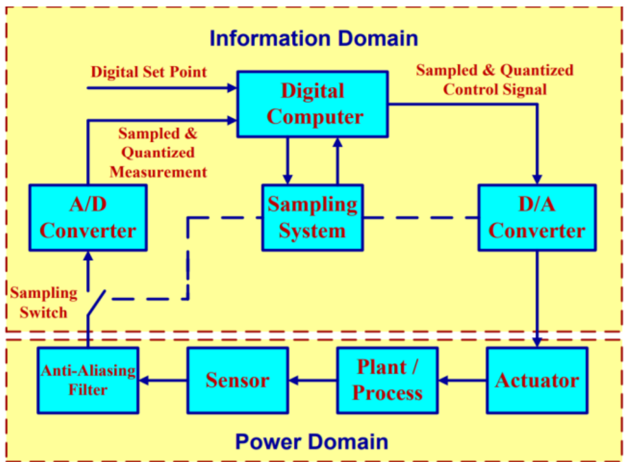

Figure 1. Matlab initial window

The A/D data sampling with zero-order hold process can be realized in ~~the~~ Simulink with block 'zero order hold' and followed with the Quantizer block for quantization. Follow the steps described below and generate corresponding results for your short form report.

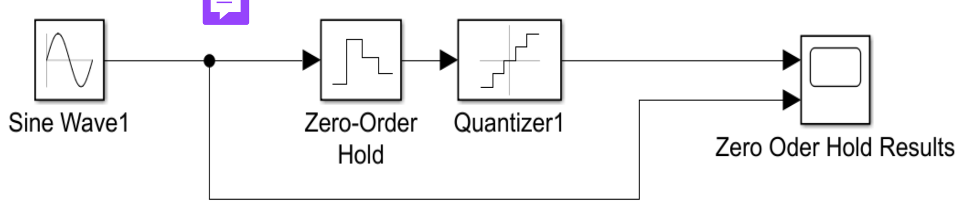

Figure 2. Reference for Zero Order Hold

A. Choose the blocks from the Simulink Library Browser and build the model as above in Figure 2. Select the simulation step time as 0.4 second.

B. Click on model settings and choose fixed-step in solver selection, then set fixed-step size to **0.0025**. *(The 'variable step' uses adaptive step to accelerate simulation, fixed step is set by the user. To obtain a smooth response curve, we usually need to set a small step size than system frequency)*

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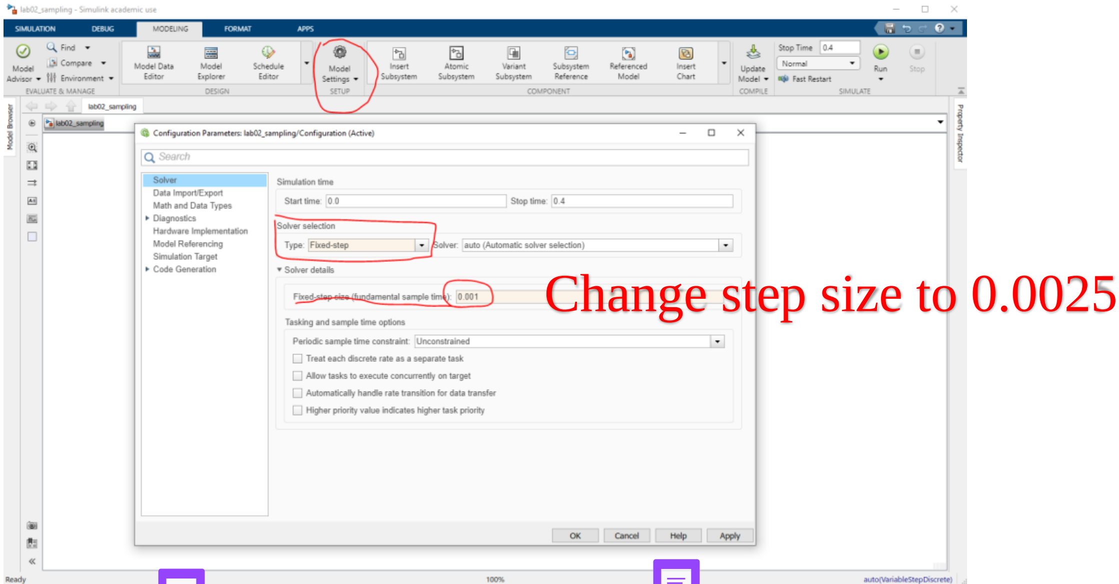

C. Edit the parameters in Sine Wave block as below to generate signal, $2\sin(40\pi t)$. What is the frequency (in unit Hz) of this signal, write down your answer.

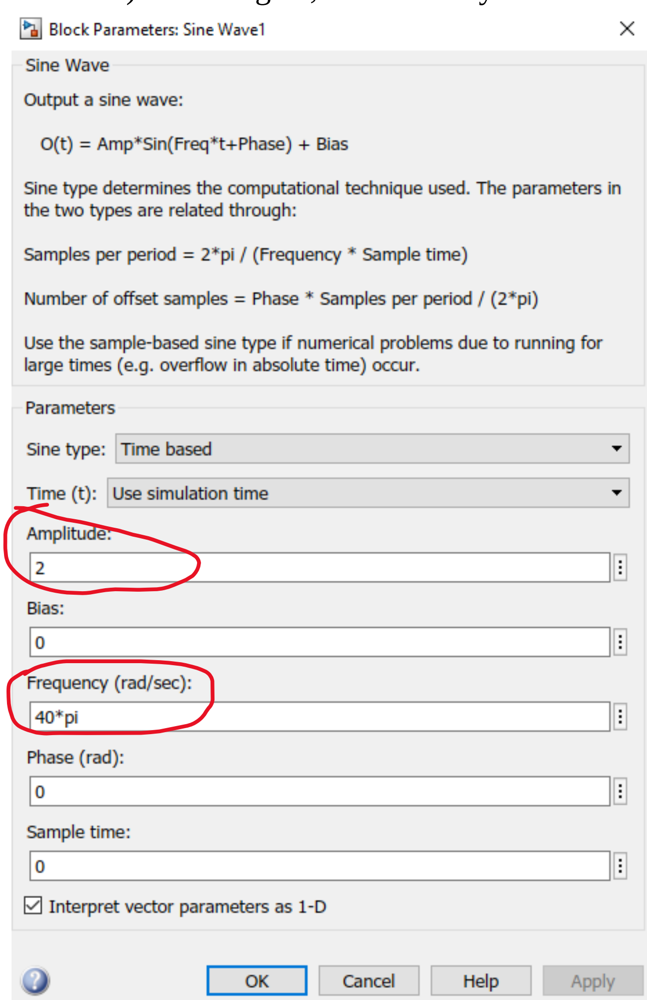

Figure 3. Parameters for Sine Wave Block

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<u>There would be three main tasks based on the model you build.</u>

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

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

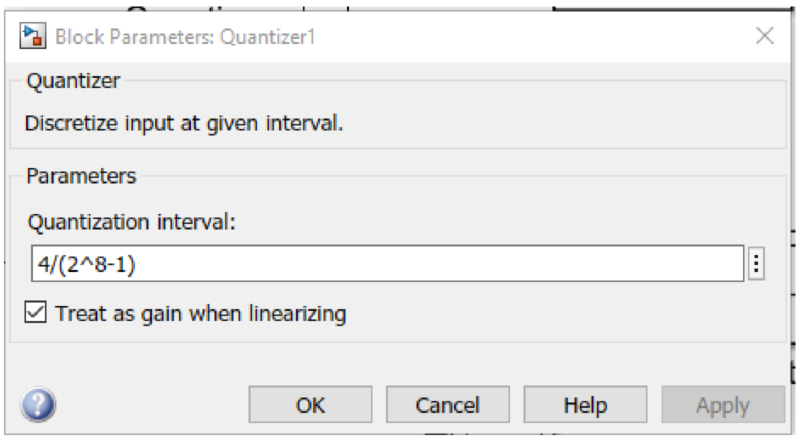

Figure 4. Parameters for Quantizer Block

Step 2: Edit the parameters in Zero-Order Hold block and set sample time to 0.04 second. Write down the sample frequency (in Hz unit) and attach the simulation results in scope.

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

<u>Second Task: sampling rate influences on sampled results</u>

Step 1: Still keep the same configuration for the quantizer.

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

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?

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

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

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

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

<u>Set the step size to 0.001 in this task.</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.

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.

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

Section 2. FFT of Signals

2.1 FFT Code for Single Sided Spectrum

Use Fourier transforms to find the frequency components of a signal.

Specify the parameters of a signal with a sampling frequency of 1 kHz and a signal duration of 1.5 seconds. Form a signal containing a 50 Hz sinusoid of amplitude 0.7 and a 120 Hz sinusoid of amplitude 1, that is, $y = 0.7\sin(2pi * 50t) + \sin(2pi * 120t)$. It is difficult to identify the frequency components by looking at the signal in time domain. Compute the two-sided spectrum P2. Then compute the single-sided spectrum P1 based on P2 and the even-valued signal length L. Define the frequency domain f and plot the single-sided amplitude spectrum P1. On average, longer signals produce better frequency approximations. The code is shown below.

```
Fs = 1000;            % Sampling frequency
T = 1/Fs;             % Sampling period
L = 1500;             % Length of signal
t = (0:L-1)*T;        % Time vector
S = 0.7*sin(2*pi*50*t) + sin(2*pi*120*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)|')
```

Figure 5. Code for FFT

2.2 Exercise

Exercise 1:

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

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

Exercise 2:

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

2.2.1 Calculate the frequency components of the new signal.

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.

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?