

EEE 3208

Communication Theory Lab

Experiment No: 04

Experiment Name: Study of Signal Sampling and Reconstructions

Submitted by,

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Objective:

To understand the sampling process and reconstruction of the signal from sampling and effects of various order filters (fourth & second) during the recovery process.

Answer — 1

1. Duty Cycle Selector

This block provides us to select the duty cycle of the sampling signal, which is the time the sampling signal stays "on" during one full cycle. The bigger the duty cycle, the wider the pulse width is; the smaller the value of the duty cycle, the smaller the width is.

2. Sampling Frequency Selector

It gives the provision for selecting the sampling frequency amongst 10 kHz, 20 kHz, 40 kHz, 80 kHz, 180 kHz, and 380 kHz. It defines the rate with which the continuous signal will be sampled.

Implications on Sampling and Reconstruction:

The highest frequency, say 380 kHz, gives an exact description of the high-frequency components of the original but results in too much redundant data.

The lowest frequency, say 10 kHz, might cause aliasing if it is below twice the largest frequency component of the signal-what is known as the Nyquist rate.

3. Low Pass Filter (LPF-1 kHz)

It allows only frequencies below 1 kHz to pass while attenuating the higher frequencies. This circuit conditions the signal before being sampled or reconstructed.

4. Sampling Control (Internal/External)

It is a block that allow the user to take control of the sampling through either:

Internal control: The generation of the sampling clock signal may be generated internally within the system.

External control: The sampling clock is provided from an external source.

5. Sample Output

This displays the signal after it has been sampled. The output signal obtained is a discrete-time signal made up of pulses that represent the amplitudes of the signal at sample points.

6. Sample/Hold Output

This block shows an output signal whose value remains constant at that of the sample taken up to the instant when the subsequent sample is captured. It is done by a comparator device. When the switch is closed sampling process will come into the picture and when the switch is opened holding effect will be there. The capacitor connected to the 2nd operational amplifier is known as holding capacitor.

7. 2nd Order Low-Pass Filter

A second-order LPF attenuates high-frequency components more than a first-order filter, with a roll-off rate of 12 dB/octave.

The resulting staircase waveform due to the sample/hold is smoothened. High-frequency noise is considerably reduced and approximates the original continuous signal.

8. 4th Order Low-Pass Filter

The fourth-order LPF further increases the roll-off of the unwanted high frequency, at 24 dB/octave.

Reconstructs the signal with more accuracy.

Gets rid of residual artifacts due to sampling; hence a much clearer and smoother continuous time signal results.

Answer — 2

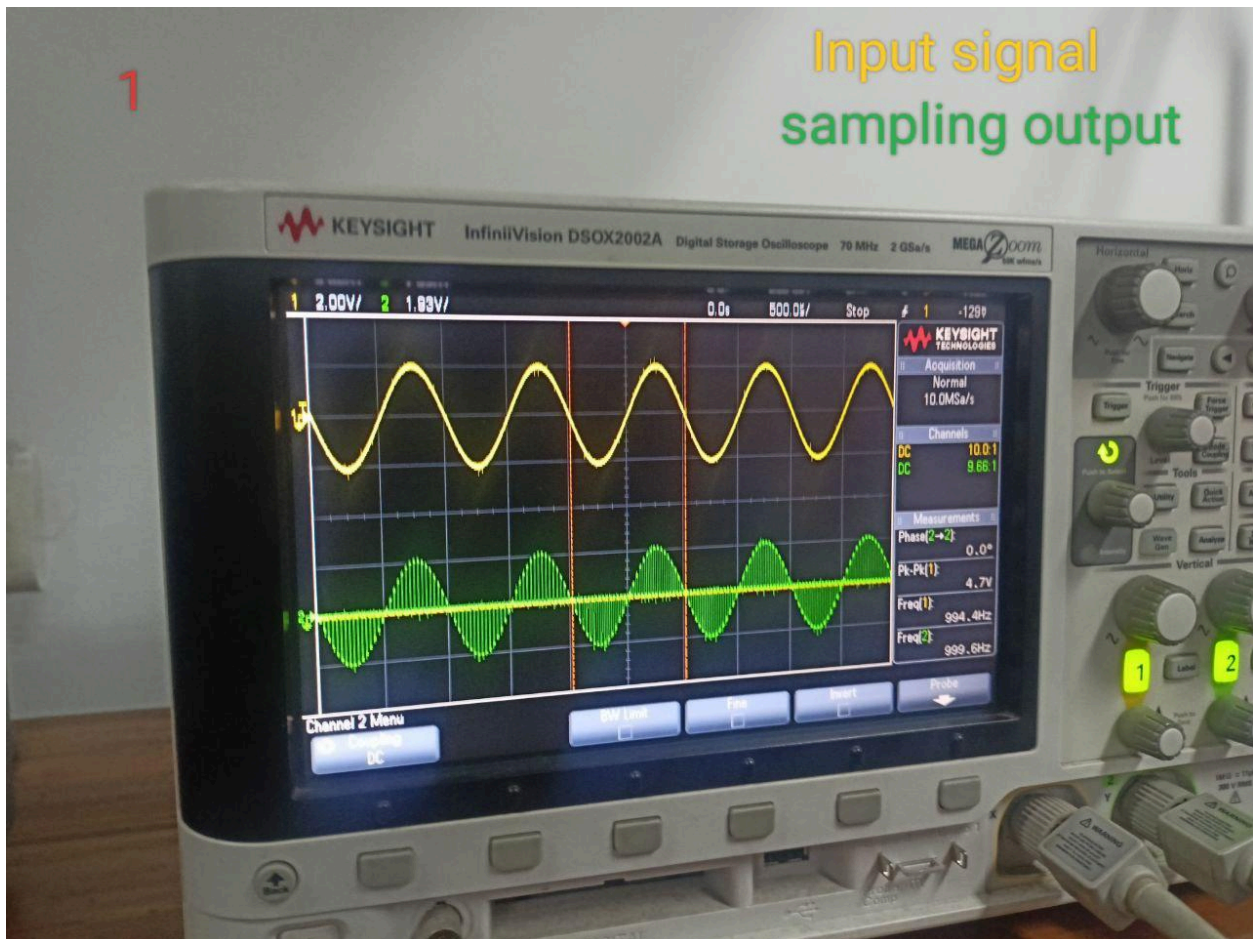


Fig-1: Input signal wave 994Hz 4.7V (yellow) and 1kHz Sampling O/p (green)



Fig 2: Sampled signal wave (yellow) and Sample and Hold Signal (Green)



Fig 3: 1kHz input signal (yellow) and Filter output - 4th order Filter (green)

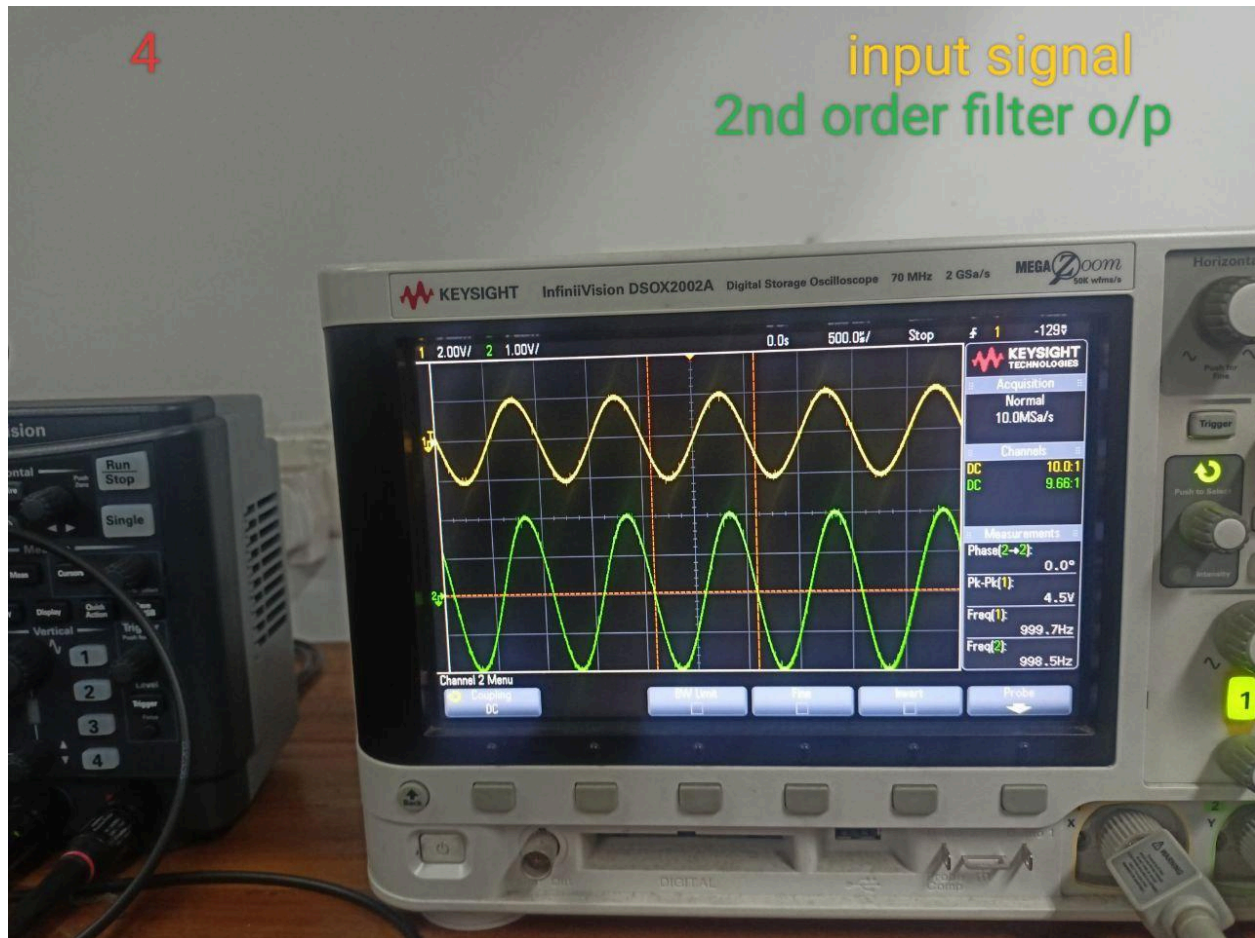


Fig 4: Input signal (yellow) vs Filter output- 2nd order filter (green)

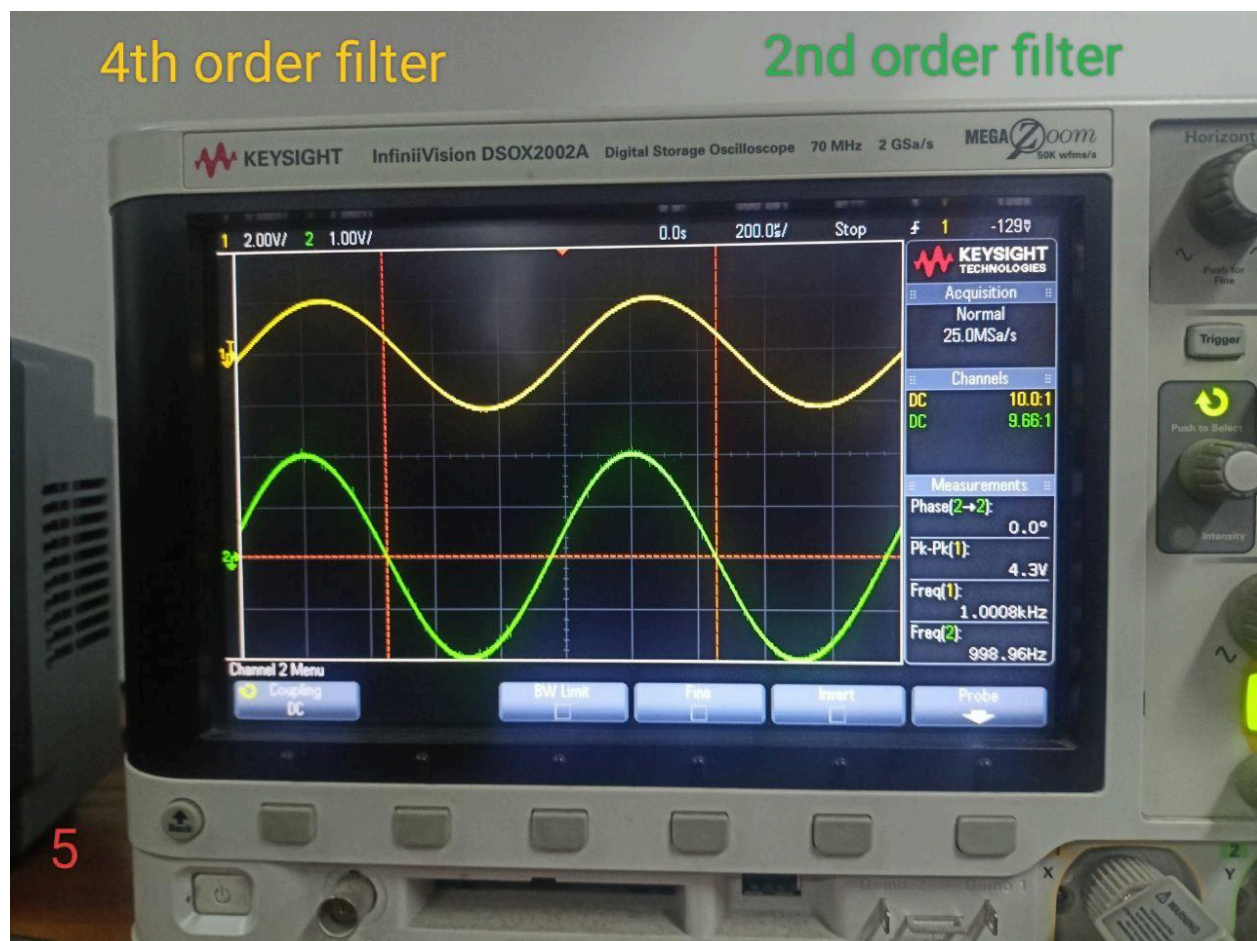


Fig 5: 4th order filter output (yellow) and 2nd order filter output (green)

4th and 2nd order at Duty cycle 20%

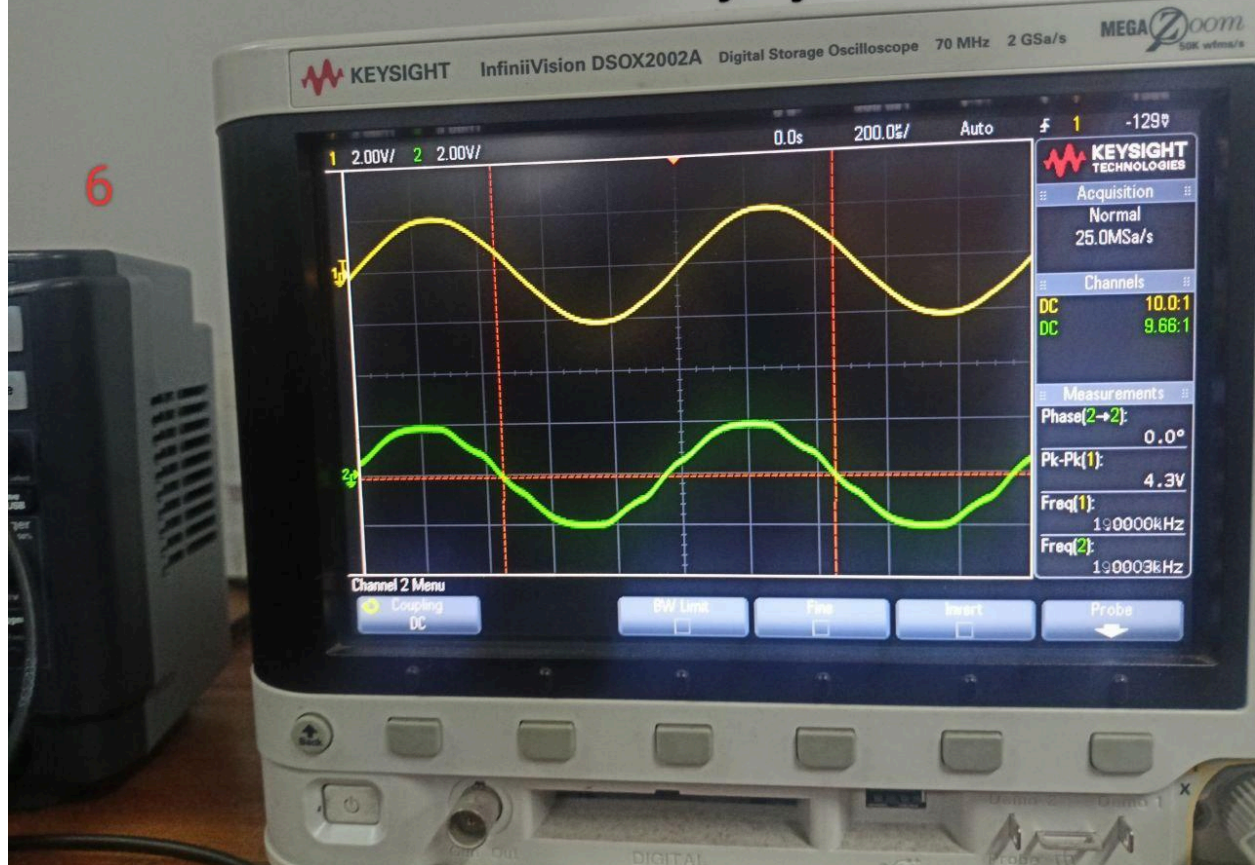


Fig 6: 4th order filter output (yellow) and 2nd order filter output (green) at Duty cycle = 2 or 20% the lower the duty cycle the clearer the effect of lower order filter is shown. The higher the filter order is the better the signal output is.

7

4th and 2nd order filter at Duty cycle 80%

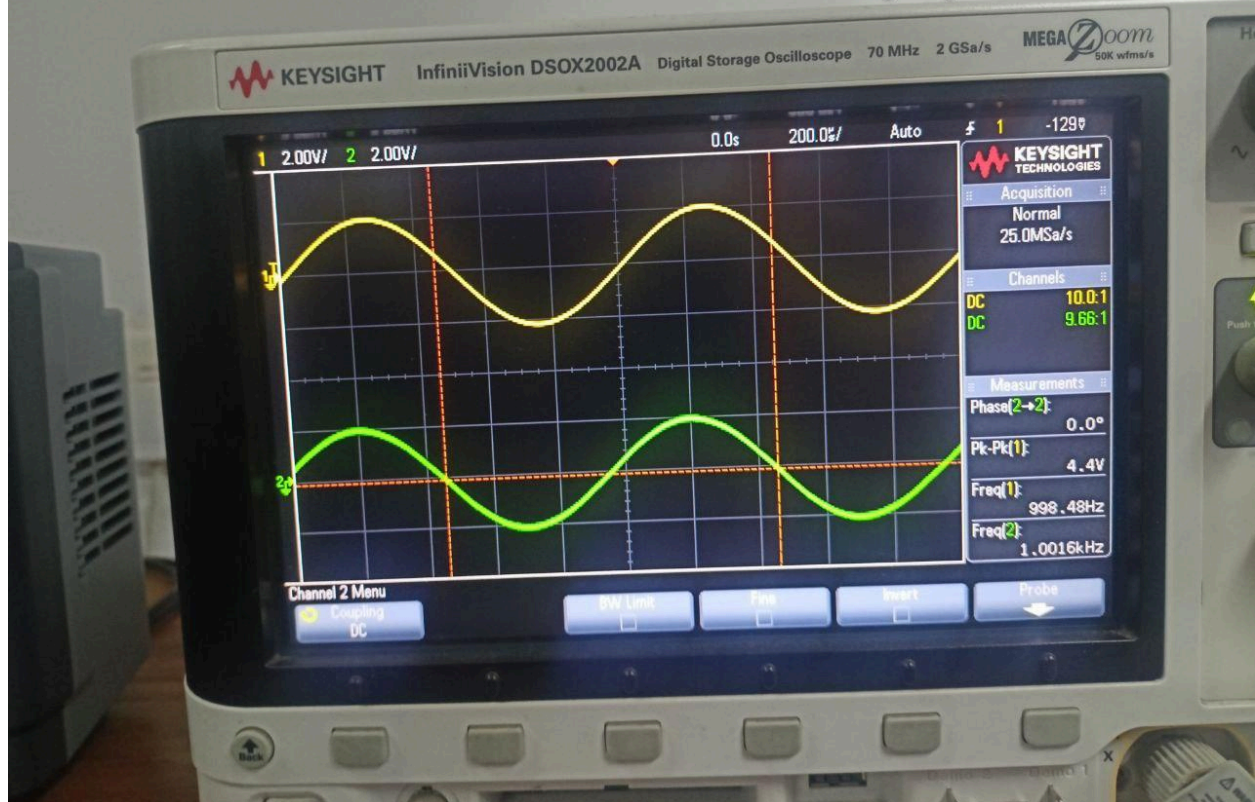


Fig 7: 4th order filter output (yellow) and 2nd order filter output (green) at Duty cycle = 8 or 80%

Duty Cycle (%)	Peak Voltage (Vp)
10	2.55
30	2.55
50	2.35
70	2.2
90	2.15

Fig 8: Duty cycle vs peak voltage comparison. The higher the duty cycle is, the lower the peak volt gets.

Answer 3

By observing Fig-1, the sampled signal looks more like the input signal for a high sampling frequency and becomes less like the input as the sampling frequency gets lower.

In case the sampling frequency is low, the sampled signal exhibits more distortion and aliasing. It can thus be said that this makes the signal less useful.

From figures, it is easy to estimate a higher sampling frequency minimizes the distortions in the reconstructed signal and improves the faithfulness.

Answer 4

From the figure-6 (duty cycle=20%) & 7(duty cycle = 80%)it was visible that lowers the duty cycle which gives steeper and more discreet sampled signals.

The peak voltage for a higher value of duty cycles as shown in fig-8 are lower; therefore, it adversely affects the amplitude and thus the clarity of the sampled signal.

Answer 5

On observing fig-4 it can be seen that output from the 2nd order LPF partial smoothened the sampled signal, yet some residual high-frequency components left which implies an even less exact reconstruction.

Fig-3 and Fig-5 represent the output of the 4th-order LPF. It is observed that the 4th-order LPF does a much better smoothing of the sampled signal since it has a higher roll-off rate. The waveform is much cleaner and closer to the input signal.

Fig-6 and Fig-7 depict how the 4th-order LPF smooths the signal in comparison with the 2nd order for varying duty cycles. And this proves that the higher order filter works much better.

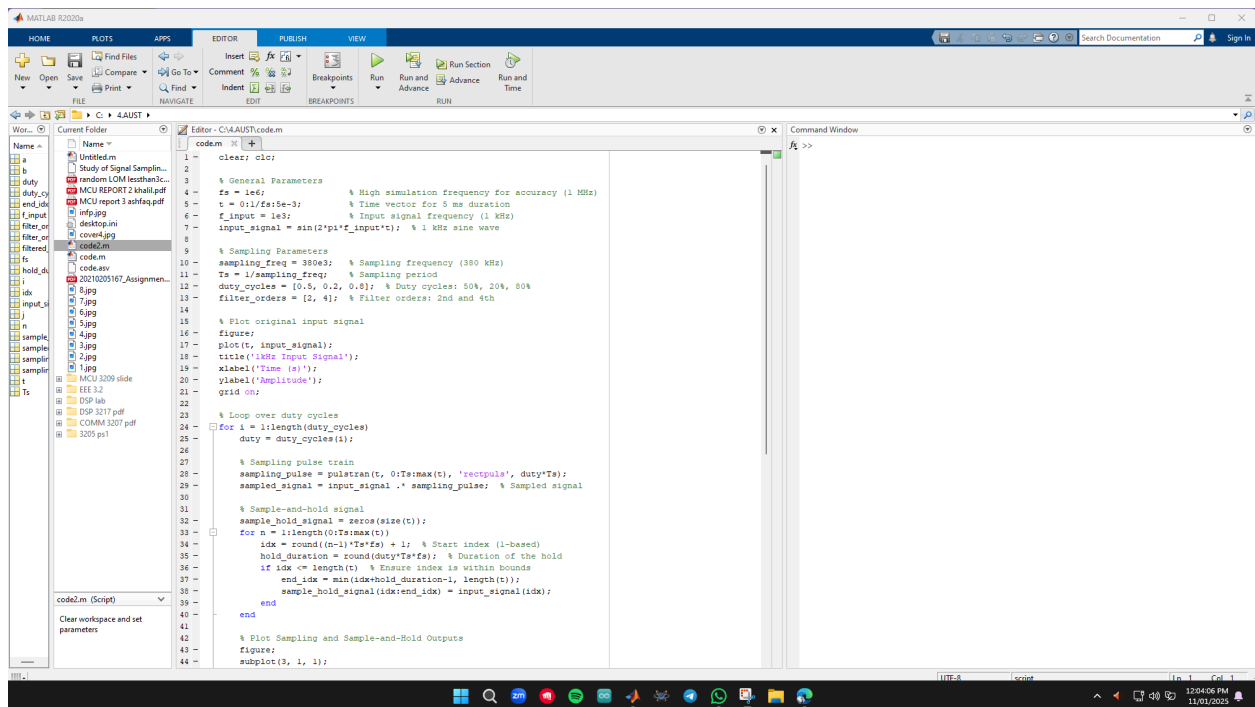
Answer 6

From Fig-2, the sample and hold block converts the sampled signal-yellow into a staircase-like waveform-green. This it does to keep the amplitude of the sampled signal constant till the next sampling

The staircase waveform from this sample and hold block lessens the rapid fluctuation and develops a smoother signal, which is quite easier to process and reconstruct in the filter stages. It is the most crucial block to stabilize the sampled data before filtering.

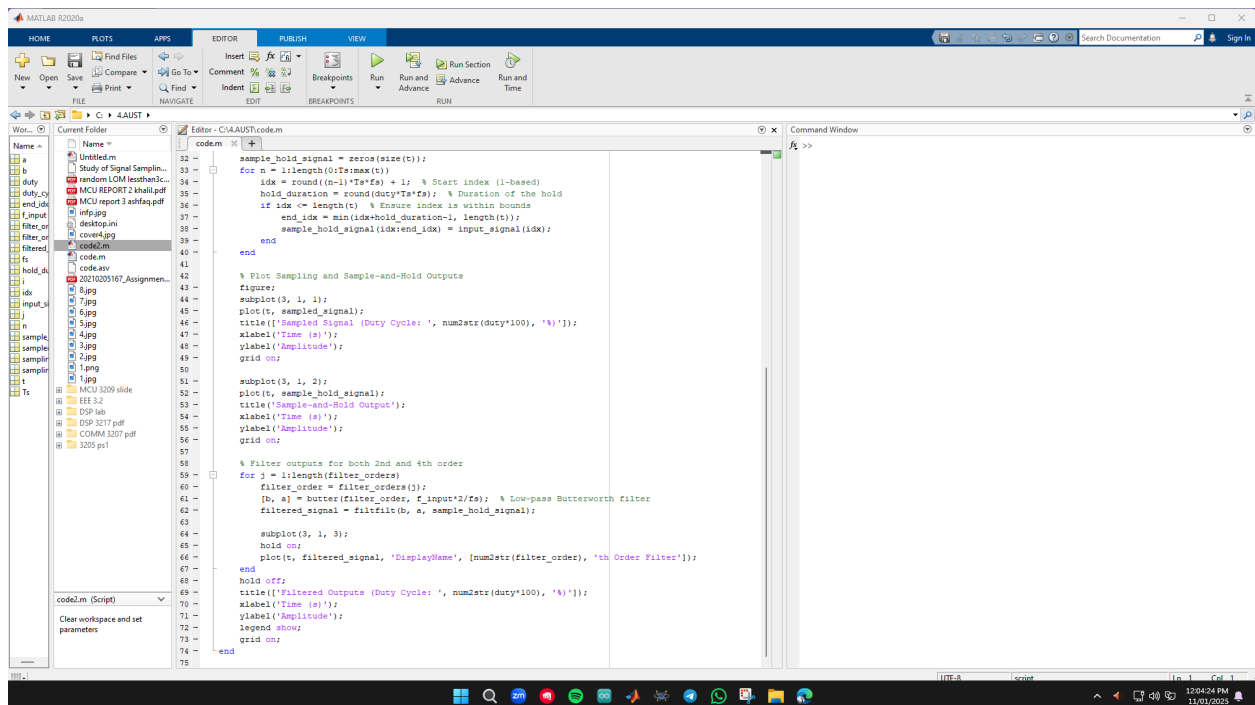
Answer- 7

Matlab codes:



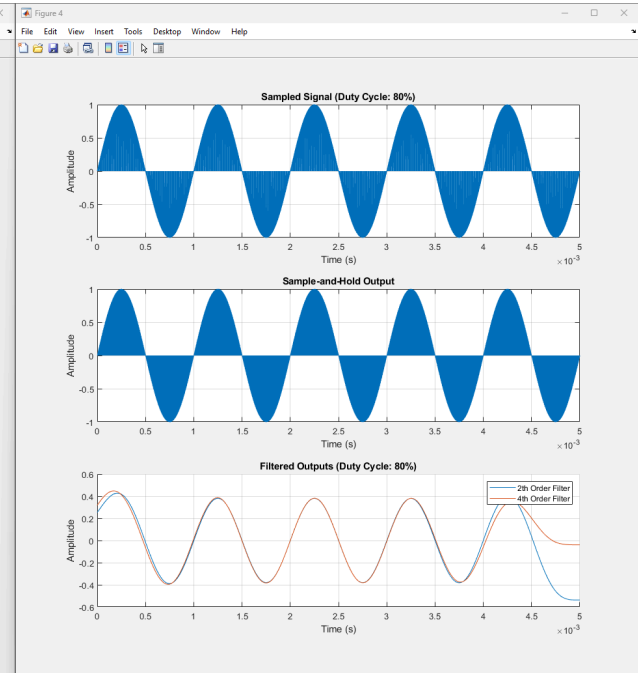
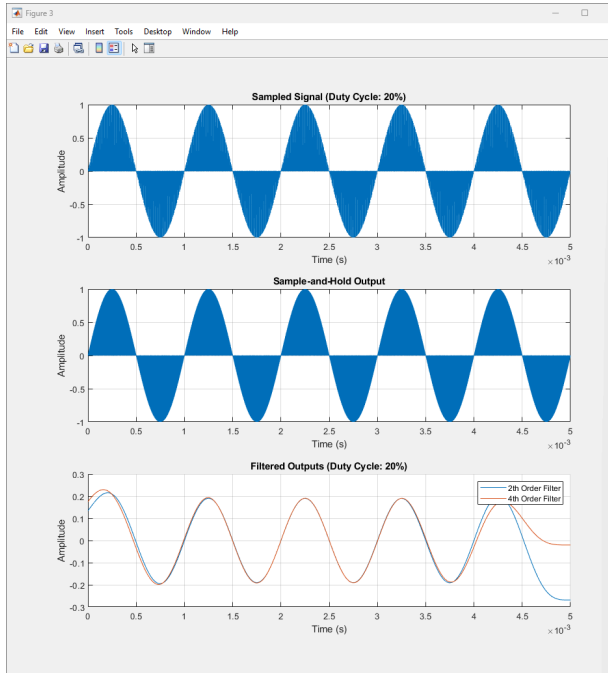
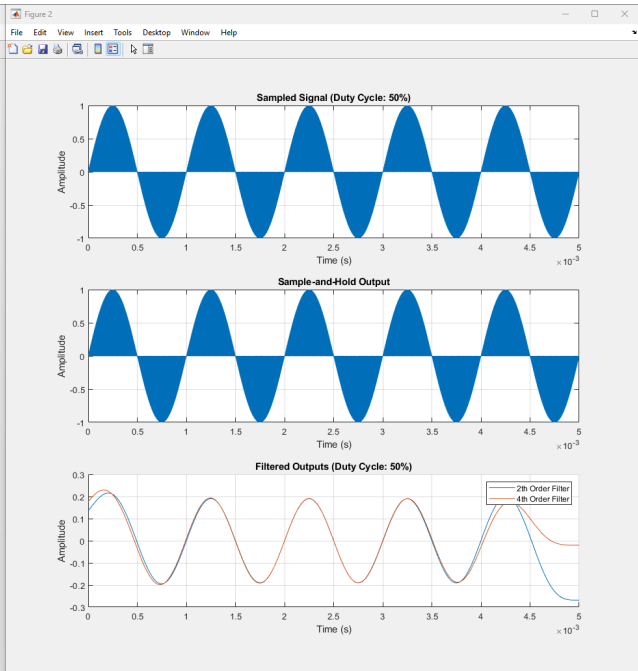
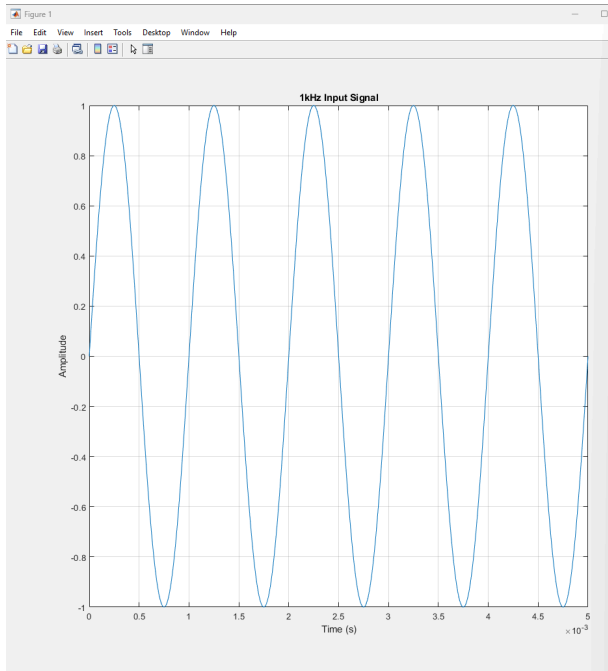
The screenshot shows the MATLAB R2020a Editor window with the following code:

```
1 clear; clc;
2
3 % General Parameters
4 fs = 1e5; % High simulation frequency for accuracy (1 MHz)
5 t = 0:1/fs:5; % Time vector for 5 ms duration
6 f_input = 1e3; % Input signal frequency (1 kHz)
7 input_signal = sin(2*pi*f_input*t); % 1 kHz sine wave
8
9 % Sampling Parameters
10 sampling_freq = 380e3; % Sampling frequency (380 kHz)
11 Ts = 1/sampling_freq; % Sampling period
12 duty_cycles = [0.5, 0.2, 0.8]; % Duty cycles: 50%, 20%, 80%
13 filter_orders = [2, 4]; % Filter orders: 2nd and 4th
14
15 % Plot original input signal
16 figure;
17 plot(t, input_signal);
18 title('1kHz Input Signal');
19 xlabel('Time (s)');
20 ylabel('Amplitude');
21 grid on;
22
23 % Loop over duty cycles
24 for i = 1:length(duty_cycles)
25     duty = duty_cycles(i);
26
27     % Sampling pulse train
28     sampling_pulse = pulstran(t, 0:Ts:(max(t)-Ts), duty*Ts);
29     sampled_signal = input_signal .* sampling_pulse; % Sampled signal
30
31     % Sample-and-hold signal
32     sample_hold_signal = zeros(size(t));
33     for n = 1:length(0:Ts:(max(t)-Ts))
34         idx = round((n-1)*Ts*fs) + 1; % Start index (1-based)
35         hold_duration = round(duty*Ts*fs); % Duration of the hold
36         if idx <= length(t) % Ensure index is within bounds
37             end_idx = min(idx+hold_duration-1, length(t));
38             sample_hold_signal(idx:end_idx) = input_signal(idx);
39         end
40     end
41
42 % Plot Sampling and Sample-and-Hold Outputs
43 figure;
44 subplot(3, 1, 1);
```



The screenshot shows the continuation of the MATLAB code in the Editor window:

```
32 sample_hold_signal = zeros(size(t));
33 for n = 1:length(0:Ts:(max(t)-Ts))
34     idx = round((n-1)*Ts*fs) + 1; % Start index (1-based)
35     hold_duration = round(duty*Ts*fs); % Duration of the hold
36     if idx <= length(t) % Ensure index is within bounds
37         end_idx = min(idx+hold_duration-1, length(t));
38         sample_hold_signal(idx:end_idx) = input_signal(idx);
39     end
40 end
41
42 % Plot Sampling and Sample-and-Hold Outputs
43 figure;
44 subplot(3, 1, 1);
45 plot(t, sampled_signal);
46 title(['Sampled Signal (Duty Cycle: ', num2str(duty*100), '%)']);
47 xlabel('Time (s)');
48 ylabel('Amplitude');
49 grid on;
50
51 subplot(3, 1, 2);
52 plot(t, sample_hold_signal);
53 title(['Sample-and-Hold Output']);
54 xlabel('Time (s)');
55 ylabel('Amplitude');
56 grid on;
57
58 % Filter outputs for both 2nd and 4th order
59 for j = 1:length(filter_orders)
60     filter_order = filter_orders(j);
61     [b, a] = butter(filter_order, f_input/2/fs); % Low-pass Butterworth filter
62     filtered_signal = filter(b, a, sample_hold_signal);
63
64     subplot(3, 1, 3);
65     hold on;
66     plot(t, filtered_signal, 'DisplayName', [num2str(filter_order), 'th Order Filter']);
67 end
68 hold off;
69 title(['Filtered Outputs (Duty Cycle: ', num2str(duty*100), '%)']);
70 xlabel('Time (s)');
71 ylabel('Amplitude');
72 legend show;
73 grid on;
74 end
75
```



Discussion:

Sampling and reconstructing depends on Filter order, sampling rate, frequency selection and duty cycle. The most challenging part of sampling is the time interval. The quality of reconstructed signal $M_r(t)$ depends on the quality of sampled signal $S(t)$. Here $S(t)$ sampled signal is actually convolution of message signal $M(t)$ and train of impulse $C(t)$. Calculating in frequency domain is easier than time domain. One of the most important thing to notice is to make sure the signal follow Nyquist Criteria. After sampling, $(W_s - W_m) \geq W_m$ is desired because the low pass filter can filter out the desired signal. Otherwise there will be no gap between signals and it will cause a Noise/Aliasing effect if the signal is $(W_s - W_m) < W_m$. We can use Guard Band in this case to keep a minimum distance between 2 individual signals. Otherwise it will not be possible to reconstruct the signal.

We used sample/hold because it gives uninterrupted continuous values of the sampled signal. A capacitor helps to hold the signal. Filter is a must-used component for sampling and reconstruction. We used a low pass filter in this case. The higher the filter order is the better and steeper and higher roll off to ideal Brick Wall Response will be. But the selectivity of a filter is more in lower order filter.

Usually we use 50% duty cycle. 100% duty cycle is practically not possible. Also if we use 10% time delay, there will be huge time delay and most of the info will be lost.

The more the sampling frequency we provide, the better the sampling will be. If we reduce the frequency there will be more noise, attenuation and ripple in the signal.