Prelab 4 : 100%
Prelab 5 : 100%
LabRep 5 : 90%

Jacobs University Bremen

Signals and Systems Laboratory Course
Fall Semester 2020

Lab Experiment: Sampling

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Experiment 5 : Sampling

Prelab

Problem 1: Analog signals are usually passed through a low-pass filter prior to sampling. Why is this necessary?

- Analog signals are passed through a low pass filter to attenuate any high frequency components that are present. Especially those high frequency components that are higher than twice the sampling frequency. We prevent aliasing by doing this. Removal of higher frequencies is done because even if we are sampling a signal whose highest frequency component is at less than twice the sampling frequency, higher frequencies might have been added due to noise. It also restricts the bandwidth of the signal, which in turn reduces the sampling frequency.

What is the minimum sampling frequency for a pure sine wave input at 3KHz? Assume that the signal can be completely reconstructed.

- Minimum sampling frequency = 2x3= 6kHz. That means that the sampling frequency should be greater than 6KHz to get the reconstructed signal closer to original signal.

What is the Nyquist frequency?

- The Nyquist frequency can be defined to be a sampling frequency that is exactly twice the maximum frequency, but when sampled at this rate, the signal cannot be completely reconstructed

What are the resulting frequencies for the following input sinusoids 500Hz, 2.5KHz, 5KHz and 5.5KHz if the signals are sampled by a sampling frequency of 5KHz?

- 500Hz= Since the sampling frequency is twice the signal, there is no aliasing. Therefore, resultant signal's frequency would be 500Hz.

- 2500Hz= Since the sampling frequency is exactly twice the signal (Nyquist Theorem), the resultant signal's frequency is 2500Hz. But we might face some distortion.

- 5000Hz= Since sampling frequency < 2×signal's frequency, aliasing occurs. It is twice the Nyquist frequency; therefore resultant signal is 0Hz.

- 5500Hz= Since sampling frequency < 2×signal's frequency, aliasing occurs and the aliasing frequency is 500Hz.

Mention three frequencies of signal that alias to a 7Hz signal. The signal is sampled by a constant 30 Hz sampling frequency. Alias.f = maxfrequency +n *(sampling frequency)

- f(n=1) =7Hz +30 Hz= 37Hz

- f(n=2) =7Hz +2*30 Hz= 67Hz

- f(n=3) =7Hz +3*30 Hz= 97Hz

Problem 2: Impulse Train Sampling and Real Sampling Consider the circuit shown in figure (6.2).

The input signal x(t) is given by a sine function, with an amplitude of 5 V peak and a frequency of 50 Hz. The sampling signal p(t) is represented by a unity impulse train. Use an overall sampling rate of 100 k samples/s) for the whole problem.

1. Carry out simulations for the following cases:

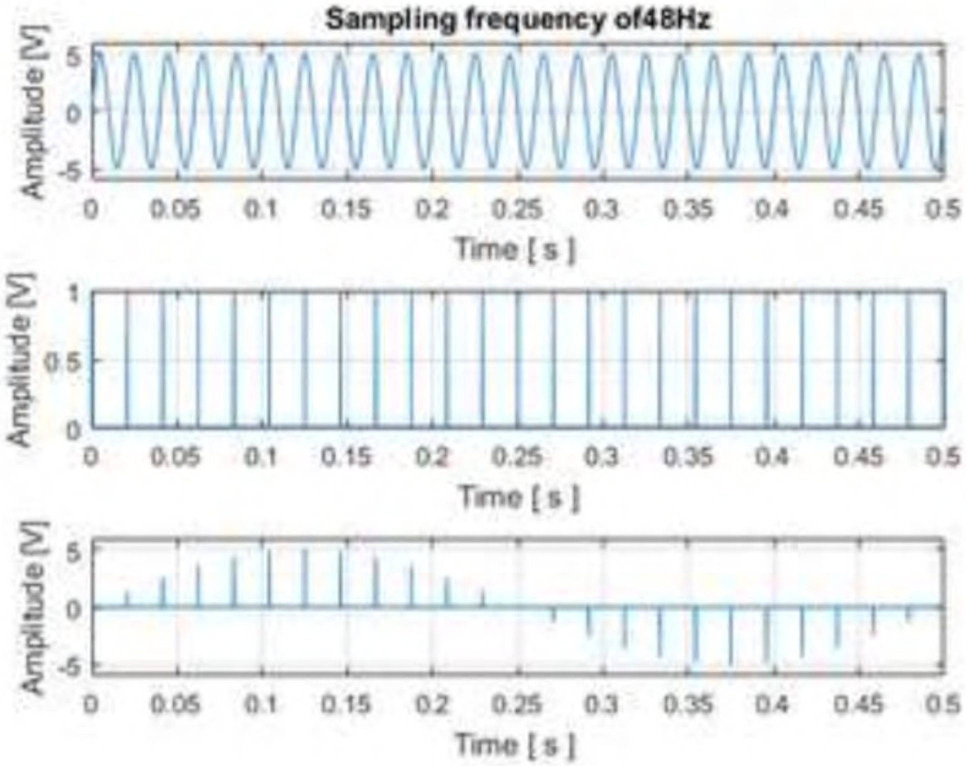
(a) Under Sampling (use 48 Hz)

(b) Nyquist Sampling

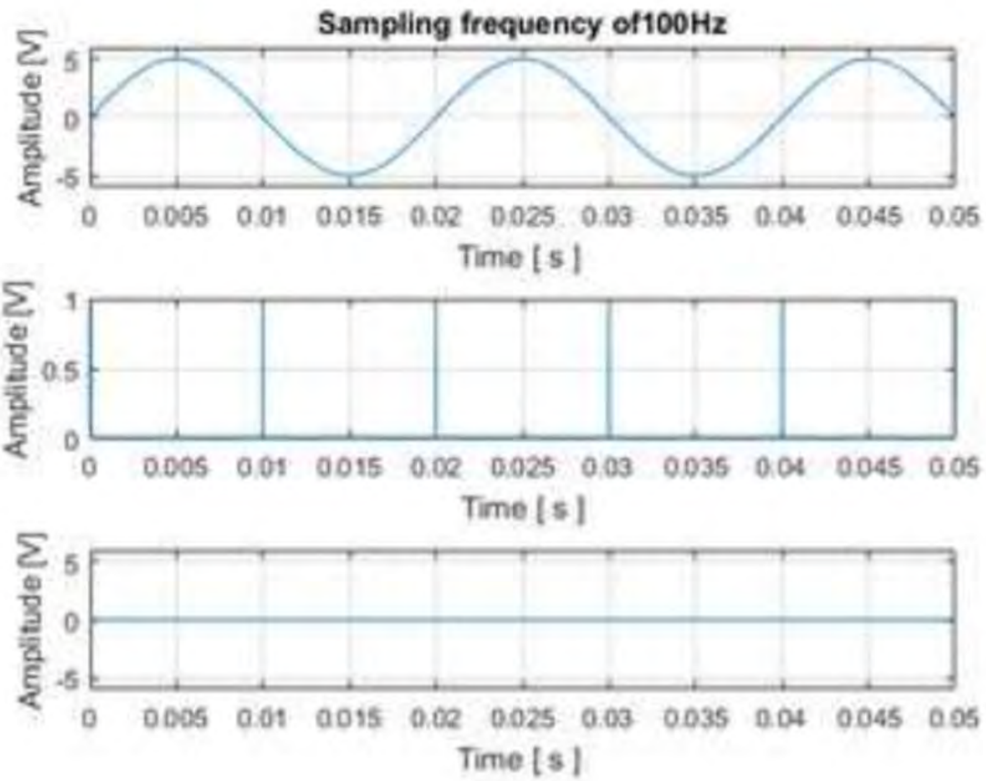
(c) Over Sampling (use 1000 Hz)

Use the command subplot to visualize the continuous signal x(t), the sampling signal p(t) and the result for each of these cases.

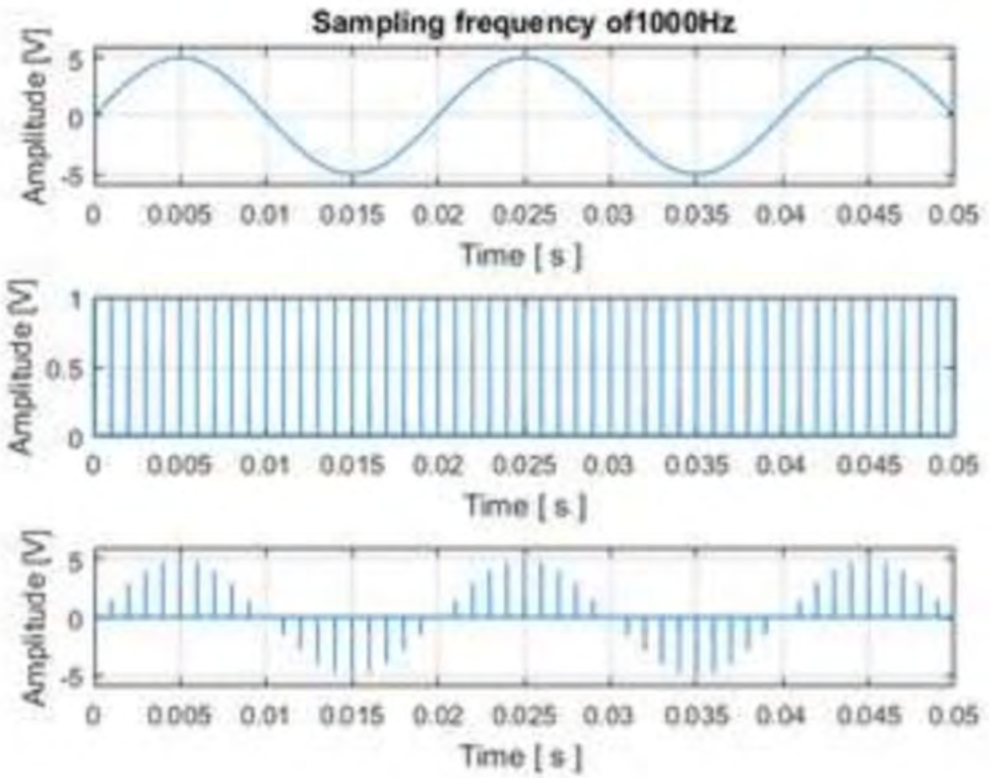
```
clc;

clear all;
f=50; %signal frequency
fs = [ 48,  2*f,  1000 ] ;

for i = 1: 3
   figure ( i )
   t_end = 0.05 ;
   if i == 1
      t_end = 0.5 ;
   end
   t = 0: 1/ 100e3 : t_end ;
   x = 5* sin ( 2* pi * f * t ) ;
   subplot( 3 , 1 , 1 )
   plot ( t , x )
   xlabel ( 'Time [ s ] ' )
   ylabel ( ' Amplitude [V] ' )
   ylim([-6 6 ] )
   grid on
   title( [ 'Sampling frequency of' , num2str( fs (i)) , 'Hz' ] )
   f_train = fs( i ) ;
   impulse_train = ( 1 + square ( 2* pi * f_train * t ,0.1 ) ) / 2 ;
   subplot ( 3 , 1 , 2 )
   plot ( t , impulse_train )
   xlabel ( 'Time [ s ]' )
   ylabel ( ' Amplitude [V] ' )
   grid on
   subplot ( 3 , 1 , 3 )
   plot ( t , impulse_train.* x ) ;
   ylim([-6 6 ] )
   xlabel ( 'Time [ s ] ' )
   ylabel( ' Amplitude [V] ' )
   grid on
end
```



Under sampling



Under sampling



Over Sampling

The signal x(t) should be sampled by a rectangular pulse train. Modify the sampling function p(t), so that the width of the sampling pulse is 50% of the sampling period.

Carry out simulations for the following cases:

(a) Under Sampling

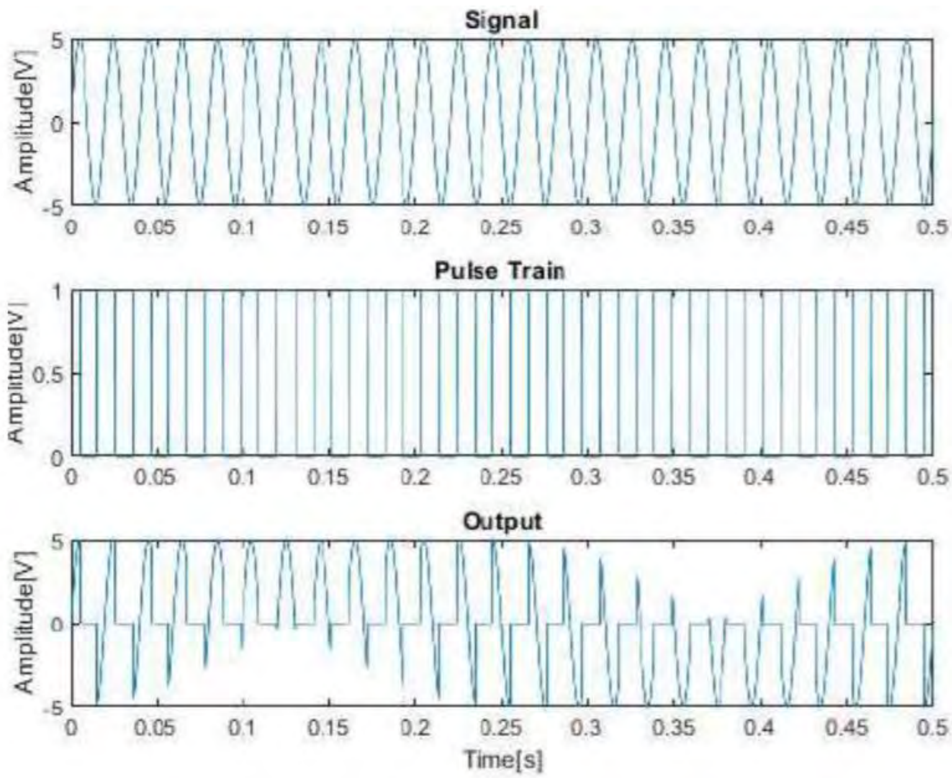
(b) Nyquist Sampling

(c) Over Sampling

Use the same sampling rates and the same plot setup as before.

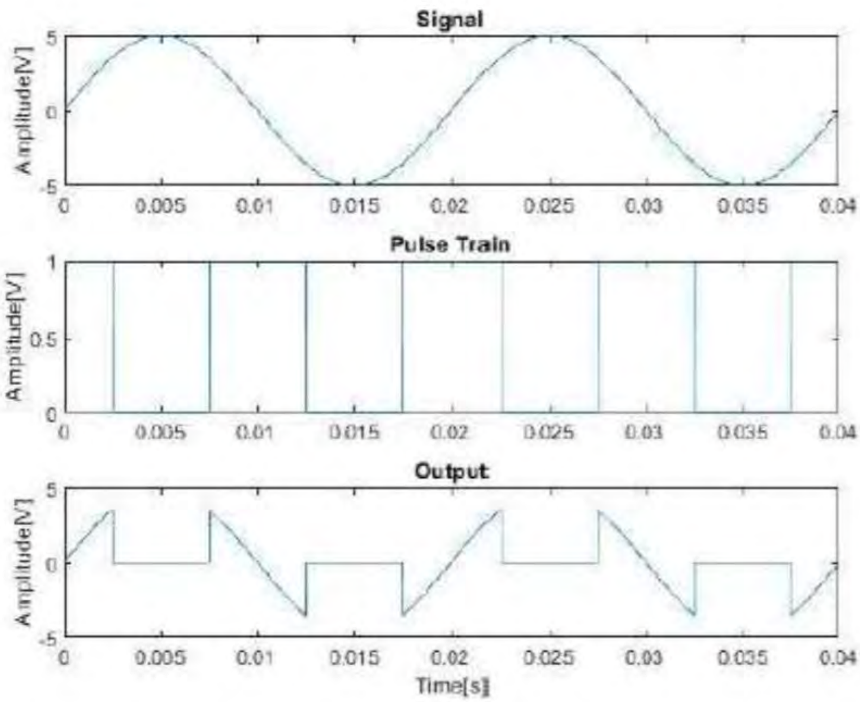
a. Under Sampling:

```
Amp=5;
f=50;
t_samp=(10^-3)/100;
t= 0: t_samp : 1-t_samp;

x=5*sin(2*pi*f*t);
figure(1);
subplot(3,1,1);
plot(t,x);
xlim([0 0.5]);
title('Signal');
ylabel('Amplitude[V]');
hold on;

fs=48;
t1= 0: 1/fs: 0.5;
pt=pulstran(t,t1,'rectpuls',0.5*1/fs);
subplot(3,1,2);
plot(t, pt);
title('Pulse Train');
ylabel('Amplitude[V]');
xlim([0 0.5]);

s=x.*pt;
subplot(3,1,3);
plot(t, s);
xlim([0 0.5]);
ylim([-5 5]);
title('Output');
ylabel('Amplitude[V]');
xlabel('Time[s]');
hold on;
```



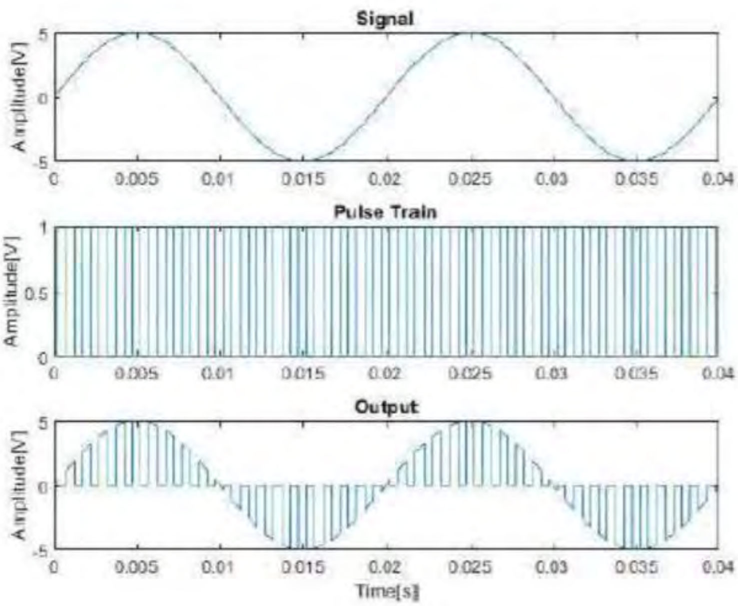
b. Nyquist Sampling:

```
Amp=5;
f=50;
t_samp=(10^-3)/100;
t= 0: t_samp : 1-t_samp;

x=5*sin(2*pi*f*t);
figure(1);
subplot(3,1,1);
plot(t,x);
xlim([0 0.04]);
title('Signal');
ylabel('Amplitude[V]');
hold on;

fs=100;
t1= -0.1: 1/fs: 0.1;
pt=pulstran(t,t1,'rectpuls',0.5*1/fs);
subplot(3,1,2);
plot(t, pt);
xlim([0 0.04]);
title('Pulse Train');
ylabel('Amplitude[V]');
hold on;

s=x.*pt;
subplot(3,1,3);
plot(t, s);
xlim([0 0.04]);
ylim([-5 5]);
title('Output');
ylabel('Amplitude[V]');
xlabel('Time[s]');
hold on;
```



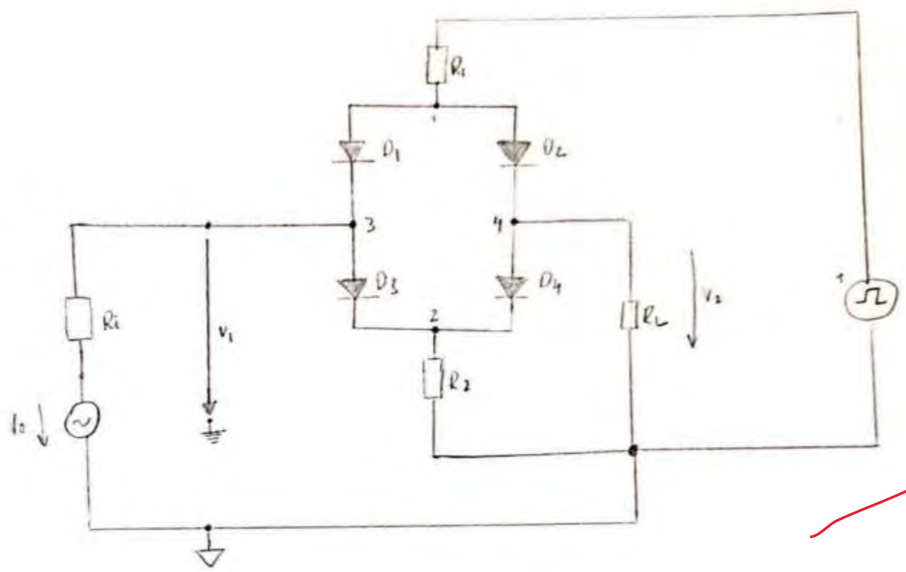
c. Over Sampling:

```
Amp=5;
f=50;
t_samp=(10^-3)/100;
t= 0: t_samp : 1-t_samp;

x=5*sin(2*pi*f*t);
figure(1);
subplot(3,1,1);
plot(t,x);
xlim([0 0.04]);
title('Signal');
ylabel('Amplitude[V]');
hold on;


fs=1000;
t1= -0.1: 1/fs: 0.1;
pt=pulstran(t,t1,'rectpuls',0.5*1/fs);
subplot(3,1,2);
plot(t, pt);
xlim([0 0.04]);
title('Pulse Train');
ylabel('Amplitude[V]');
hold on;

s=x.*pt;
subplot(3,1,3);
plot(t, s);
xlim([0 0.04]);
ylim([-5 5]);
title('Output');
ylabel('Amplitude[V]');
xlabel('Time[s]');
hold on;
```



Problem 3: Sampling using a Sampling Bridge

1)



2) Here Vs for our experiment was given via the auxiliary function generator that was provided. R1 = R2 = 10kohm , R_L = 100k ohm as were used as specified in the experiment, Our Vo (the signal we want to sample) was from the generator and Ri in this case was thus 50 ohm internal resistance of the generator. V1 is the voltage after taking the Ri internal resistance into account. Our sampling circuit is from by using the 4 diodes (D1-D4) and the resistors R1 and R2 as shown. Usually Negetive and positive sampling pulses are applied at the two nodes at the ends of R1 and R2, however in our case since we are using only one sampling pulse sources we instead have an offset at the generator input V0 of in this case 2.5V. The voltage of the sampling pulse Vs is kept high enough to ensure the diodes are forward biased. Thus, the path between node 3 and 4 starts conducting, thus there is voltage applied over the load resistor which should be the same as V1 in ideal cases. Thus we would get the samples over this R_L. Note that we need to pick our two resistors R1 and R2 such that they are not too high or too low. Too high would cause the diodes to no longer be forward biased and too low would cause V1 to pass through and be applied to R_L.

Execution
Experiment Setup

Devices Used:
- Oscilloscope
- BNC cable
- Signal Generator
- 270nF Capacitor
- 1N4148 diodes
- 10 mH Inductor
- Resistor (10KΩ, 100KΩ)

- Problem 1: Digital Sampling Oscilloscope

- General Overview
  In this part of the experiment, the input signal is sampled and the continuous time signal is converted into a discrete time signal. Firstly, the signal generator was set to 300 Hz frequency and an amplitude of 5 V_{PP} without offset. The signal generator was connected to oscilloscope and the input signal from the oscilloscope was sampled at 25 kS/s (i.e. with 10ms sampling rate). The vector and dot option in the oscilloscope was chosen and the output was observed. Similarly, higher frequencies including: 24900 Hz, 25000 Hz, 25020 Hz , and 25500 Hz was set in the signal generator and the alias frequency was measured, while the sampling rate was still at 25kS/s.

- Experimental Observation

1. Demonstrate that the graph on the oscilloscope screen consists of single points.
   i. **Display → Dots**

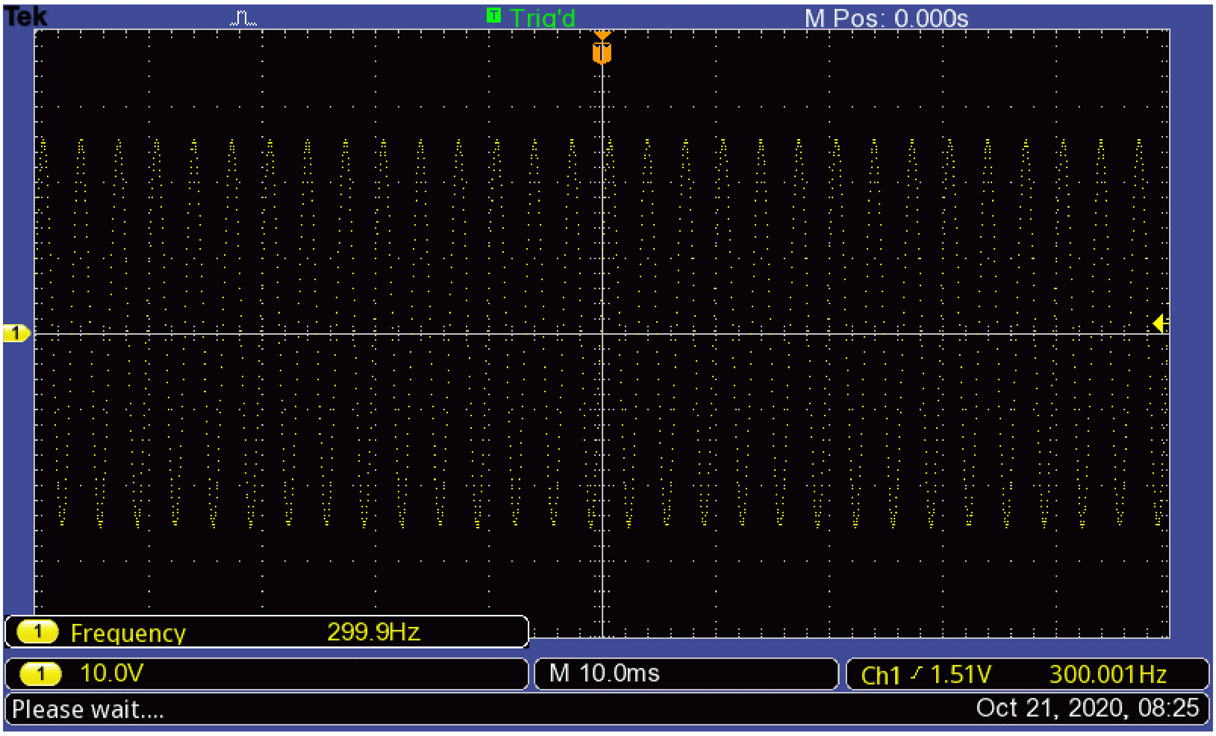


Figure 1: Hardcopy of Display → Dots

ii. **Display → Vectors**

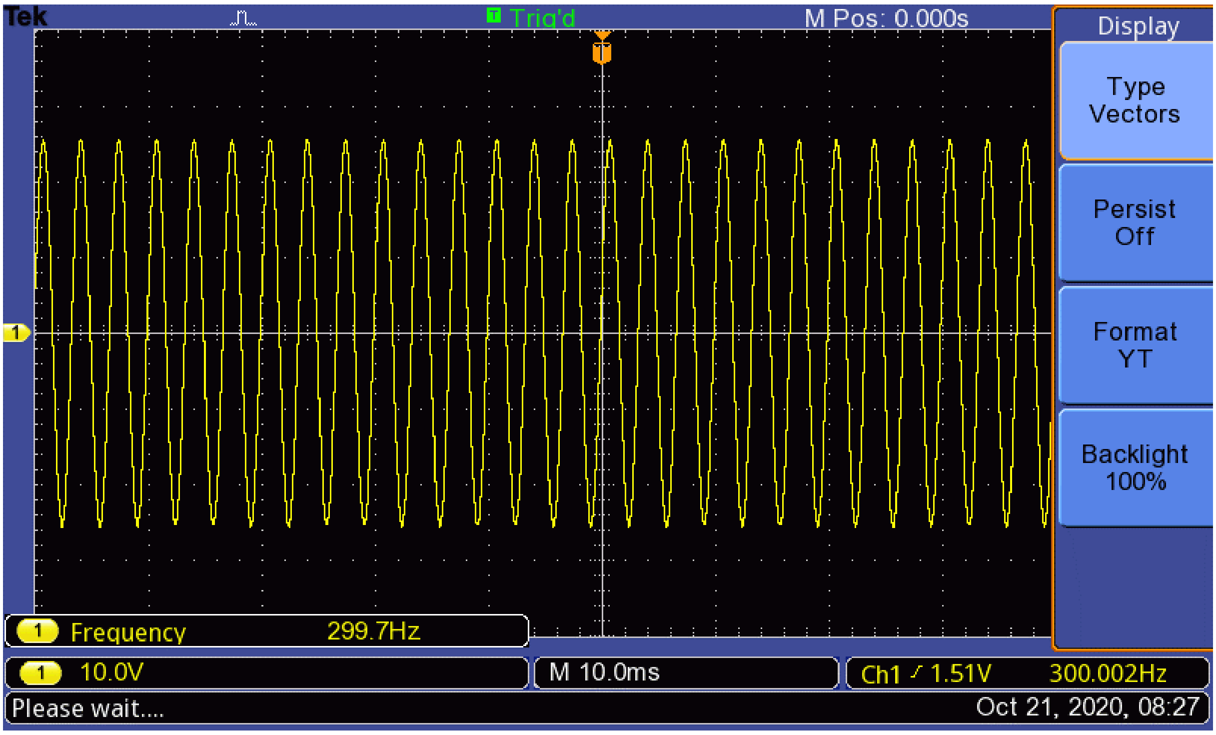


Figure 2: Hardcopy of Display → Vectors

2. What happens when the input signal exceeds the Nyquist frequency.

i. Generator Frequency = 24899.99 Hz

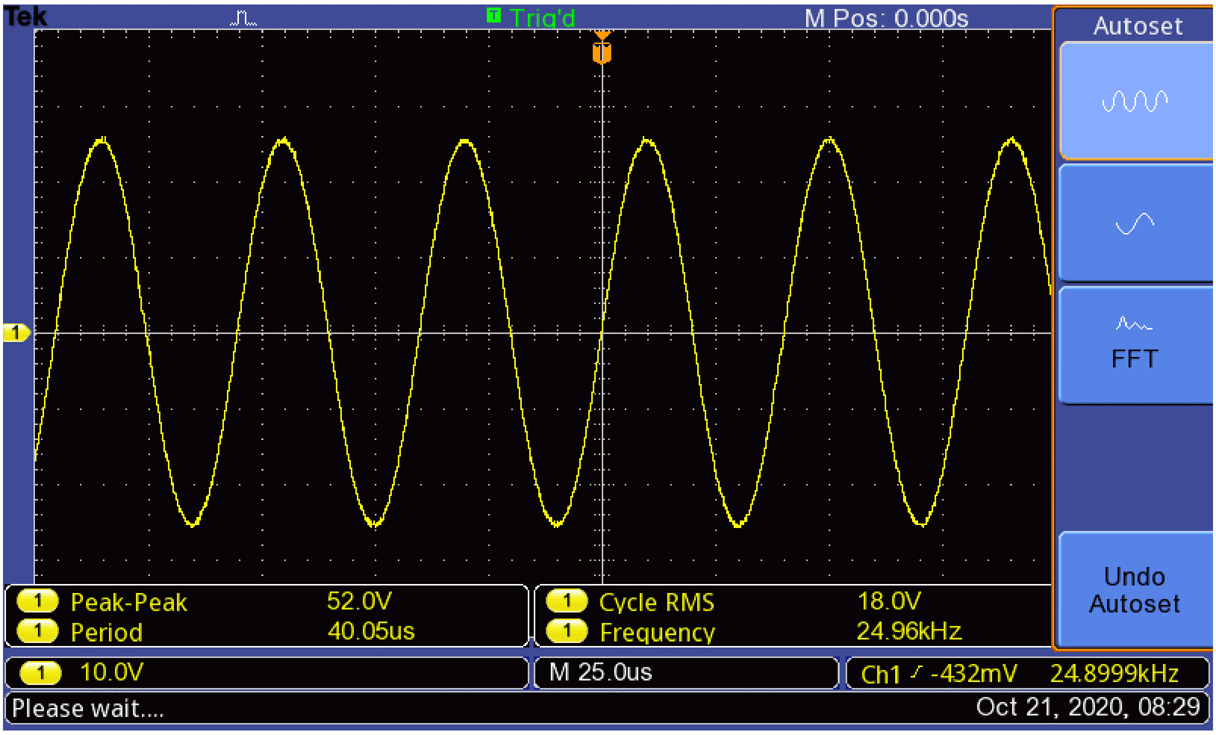


Figure 3: Hardcopy for alias frequency at 24.8999 Hz

When frequency was set to 24900 Hz, the alias frequency was measured to be:

$$f_{alias,24900\ Hz} = 100.00\ \text{Hz}$$

ii. Generator Frequency = 25000 Hz

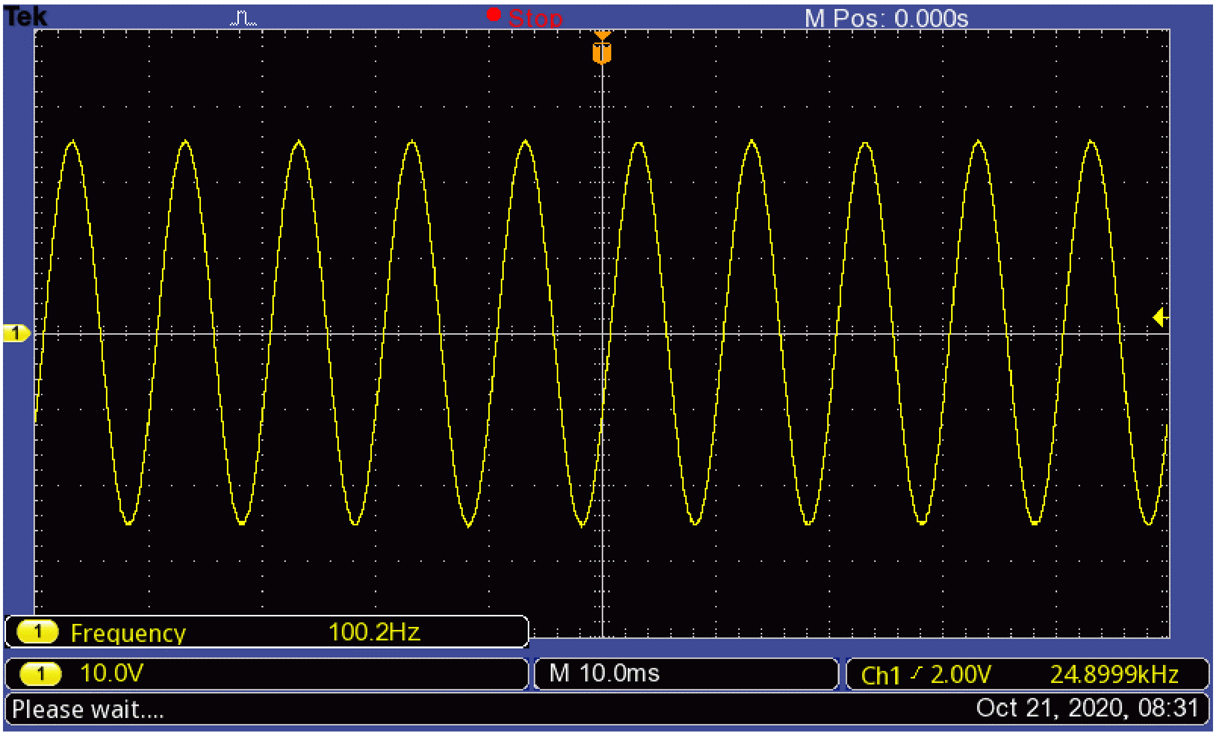
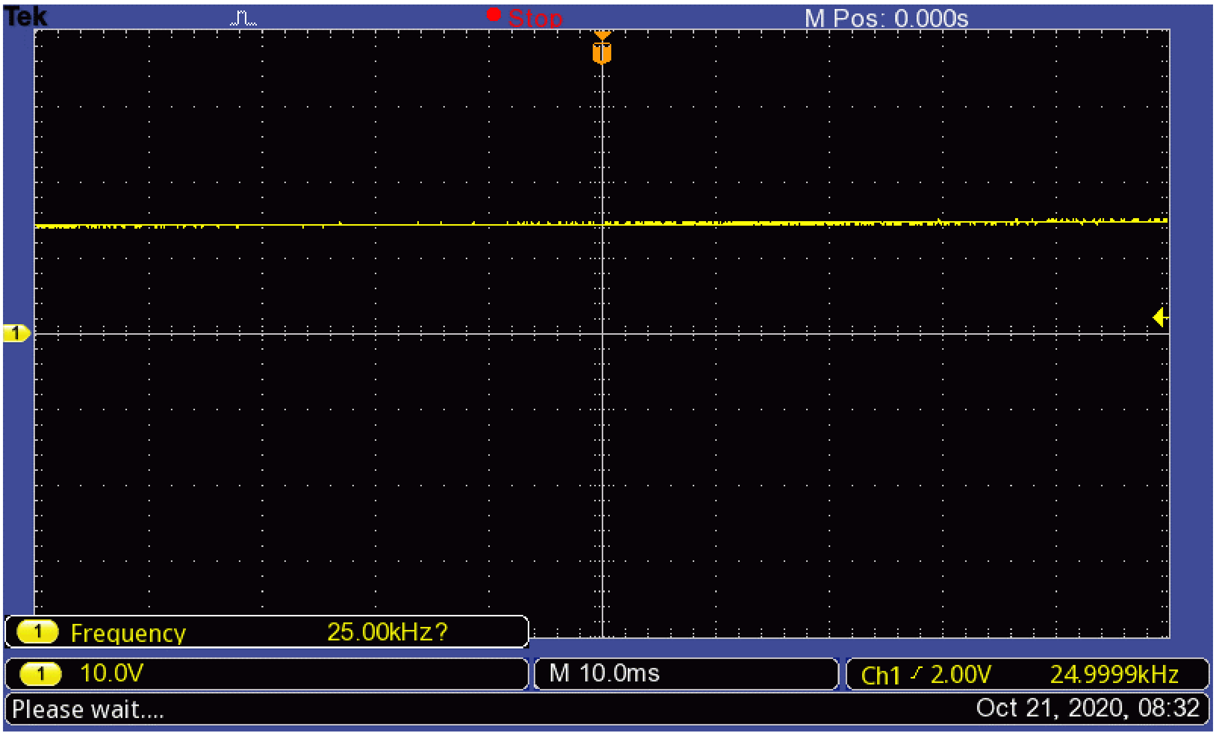


Figure 4: Hardcopy for alias frequency at 25000 Hz

When frequency was set to 25000 Hz, the alias frequency was measured to be:

$$f_{alias,25000\ Hz} = 0 \text{ Hz}$$



iii. Generator Frequency = 25020 Hz

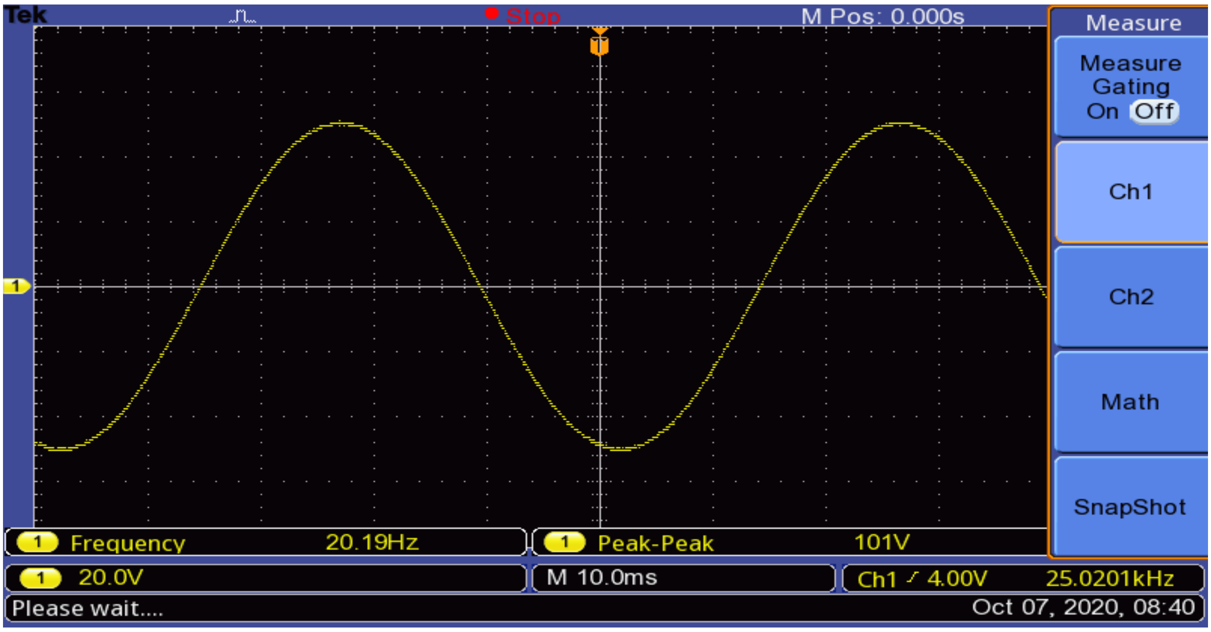


Figure 5: Hardcopy for alias frequency at 25020 Hz

When frequency was set to 25020 Hz, the alias frequency was measured to be:

$$f_{alias,25020\ Hz} = 20.20\ \text{Hz}$$

iv. Generator Frequency = 25500 Hz

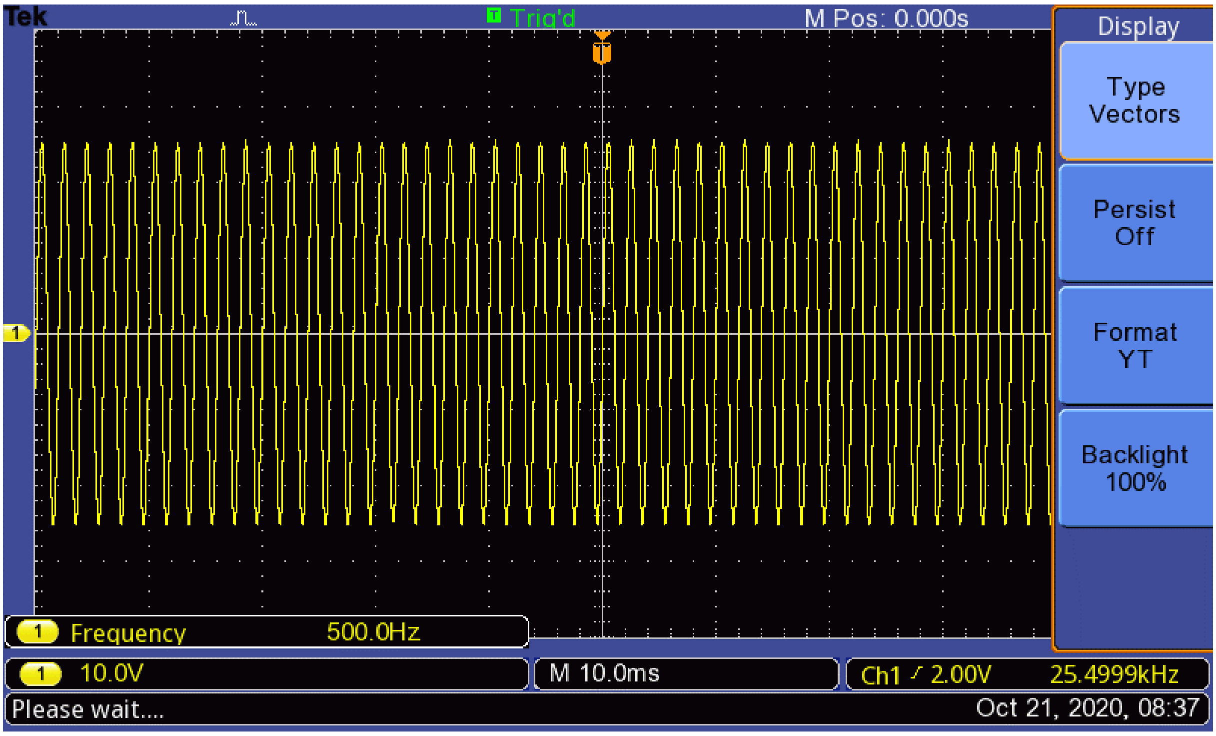


Figure 6: Hardcopy for alias frequency at 25500 Hz

When frequency was set to 25500 Hz, the alias frequency was measured to be:

$$f_{alias,25500\ Hz} = 500.0\ \text{Hz}$$

- Problem 2: Sampling using a sampling bridge

- Circuit Diagram

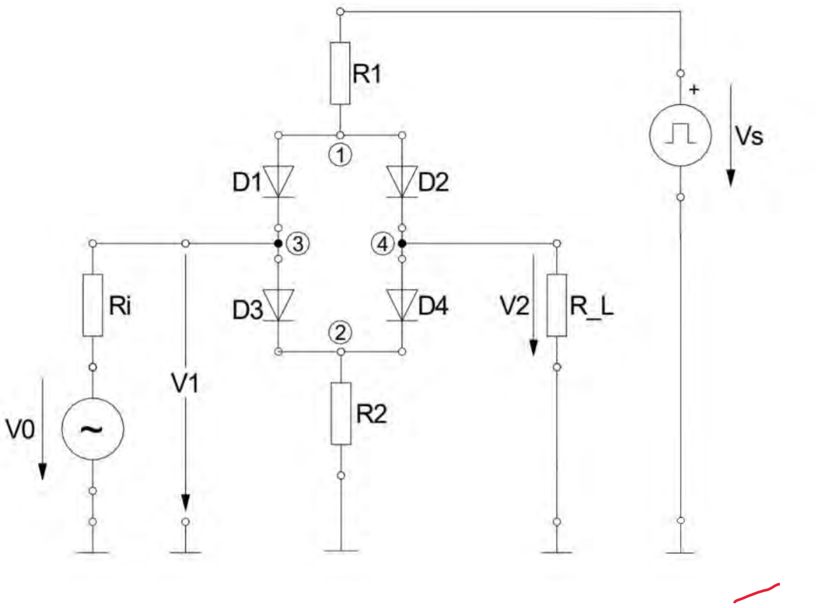


Figure 7: Circuit diagram for Sampling Bridge

- Experimental Observation

i. Input Square waveform:

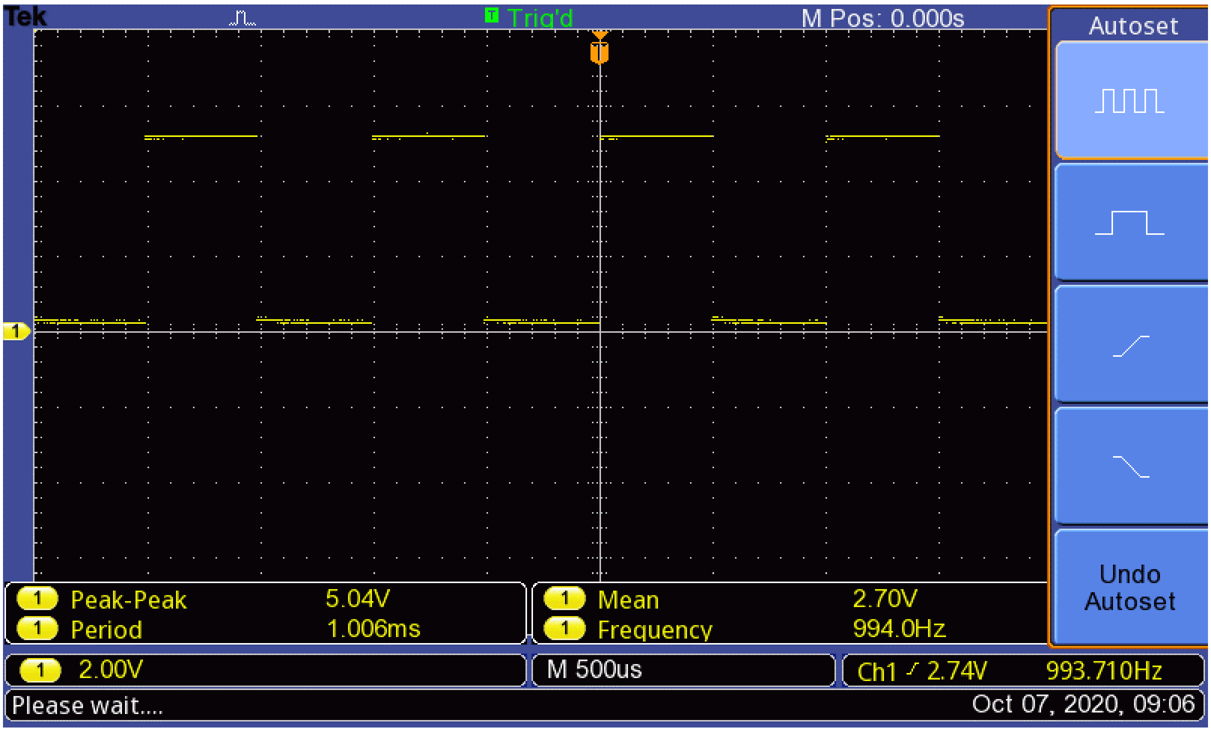


Figure 8: Hardcopy of Input Square Waveform

ii. For Sampling Frequency $f_s = 50$ Hz

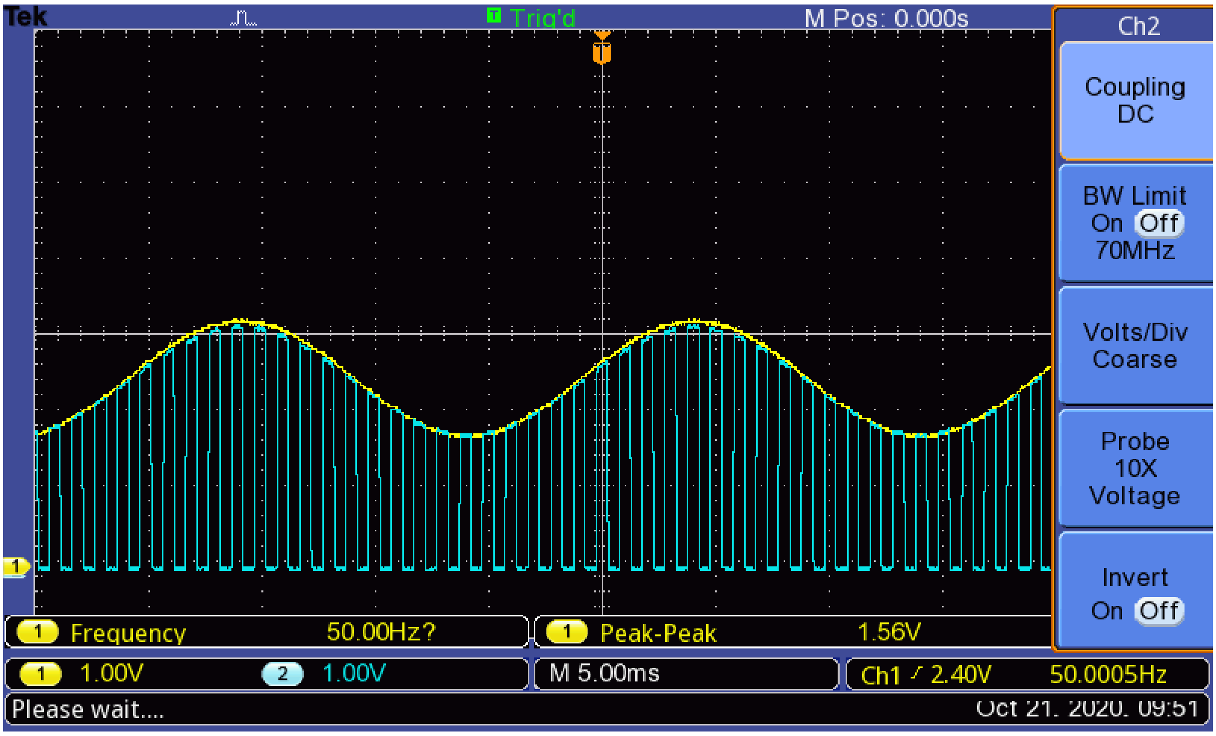


Figure 9: Hardcopy of Input and Sampled Output at 50 Hz

iii. For Sampling Frequency $f_s = 200$ Hz

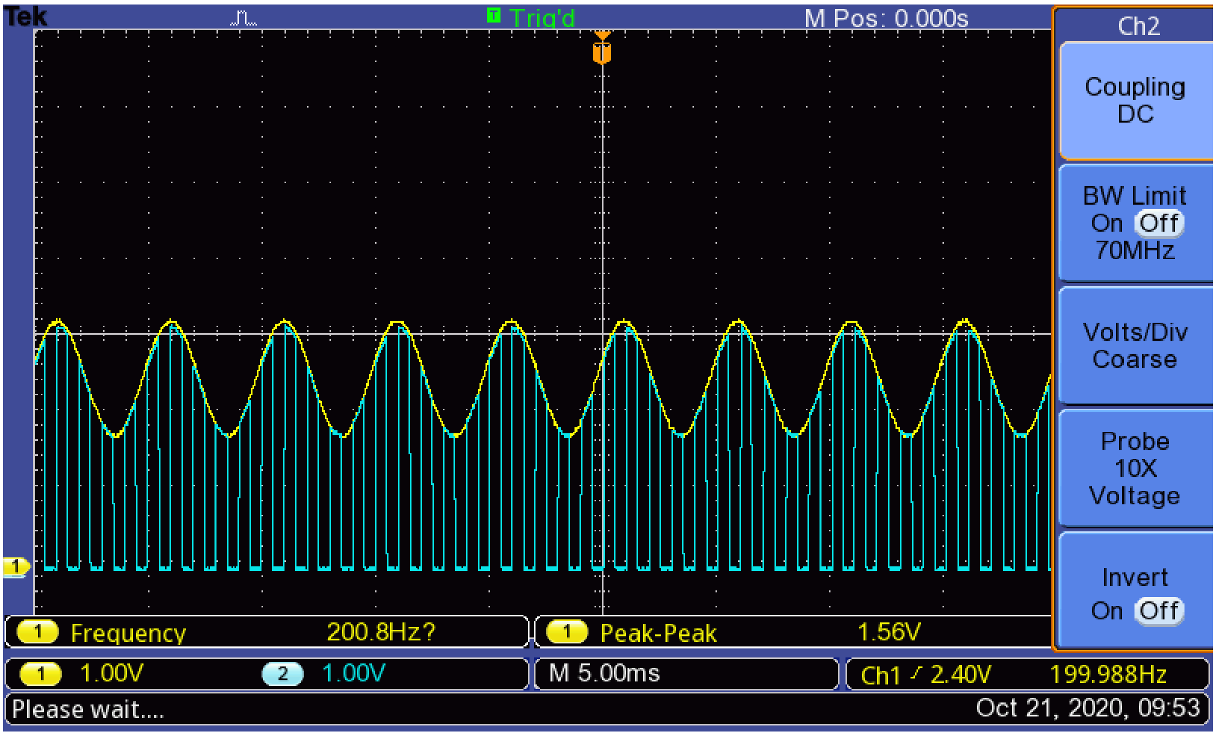


Figure 10 : Hardcopy of Input and Sampled Output at 200 Hz

Evaluation

Problem 1: Digital Sampling Oscilloscope

1. Compare the Dots and Vectors hardcopies. What would be necessary to improve the display?

   The dots hardcopy shows the discrete time sampled version of input signal. While, the Vectors hardcopy shows the continuous time sampled counterpart of the signal. In order to improve the display, more pints per unit time scale must be used within the oscilloscope, this would increase the smoothness of the plot.

2. Calculate the alias frequencies for all signals in the second step. Compare to the display!

   The alias frequency can be calculated using the formula:

   f_{alias} = | f - N· f_s |

   N=1.

   f_s = 25 KHz

   For all the input signal frequency the alias frequency was calculated and was tabulated in a table.

Input Freqeuncy / Hz	Alias Frequency (f_{alias}) Calculation / Hz	Alias Frequency (f_{alias}) Measurement / Hz
24900	100	100
25000	0	0
25020	20	20.20
25500	500	500

3. Why such a high over sampling is used?

→

A signal with a certain bandwidth can be effectively reconstructed if over sampling is used. High over sampling was used so that the signal could be reconstructed. Higher oversampling can improve resolution of the sampled signal, reduces noise and can help in avoiding aliasing and phase distortion.

➢ Problem 2: Sampling using a sampling bridge

1. Explain in detail the operation of the sampling bridge.

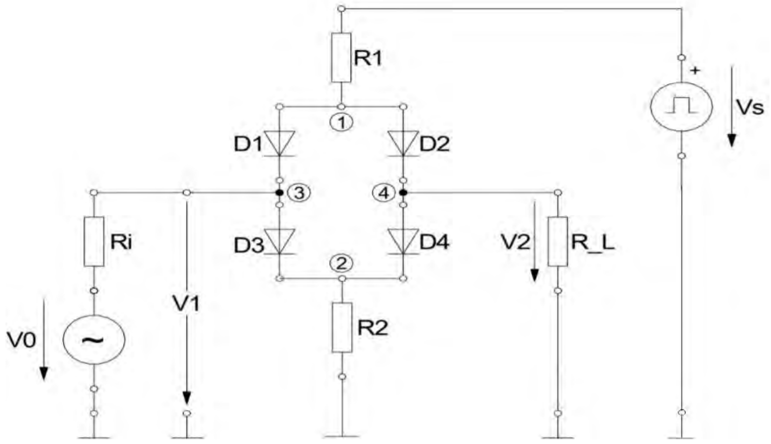


Figure 11: Circuit diagram for Sampling Bridge

The sampling circuit is made up of 4 diodes D_1 to D_4, the two resistors R_1 and R_2 and the sampling voltage V_s. The sampling bridge circuit has an sampling input voltage (V_s) which is feed from function generator. In our case, one sampling source was used. The input signal is supplied to the sampling bridge diode. The bridge has four diodes and resistors are connected across the anodes and cathodes of the diodes. V_s should be set such that it is higher than the voltage (V_1) so that the diodes are forward biased. When the sampling potential voltage (V_s) is large enough, the current path between node (3) and (4) gets conductive and the signal from the input side (V_0) is applied to the load resistor on the output side. In an ideal case, the voltage across the load resistor (V_2) is equal to the voltage (V_1). The work of the diode in this case is to help to copy sampled signal to the load without affecting its magnitude.

conclusion??

5.2 Prelab Fourier Series and fourier Transform

5.2.1

1. Given x(t) = 2 cos (2π1000t)
   a. What is the signal amplitude in Vpp?

$$V_{pp} = 2 * \text{amplitude} = 2 \times 2 = 4V$$

   b. What is the root-mean-square value of the provided signal in Vrms?

$$V_{RMS} = \frac{Amplitude}{\sqrt{2}} = \frac{2}{\sqrt{2}} = \sqrt{2} = 1.414V$$

   c. What is the amplitude of the spectral peak in dBVrms?

$$20.\log(\sqrt{2}) = 3.0103 \text{ dBVrms}$$

2. Square wave of $4V_{pp}$ and voltage level change in between -2V and 2V:
   a. What is the signal amplitudein V_{RMS}:

$$\begin{aligned} V_{RMS} &= \sqrt{\frac{1}{T}\int_0^T (x(t)^2)dt} \\ &= \sqrt{\frac{1}{T}\int_0^T 4dt} \\ &= \sqrt{\frac{4T}{T}} = 2V \end{aligned}$$

   b. What is the amplitude in dBV_{RMS}:

$$dBV_{rms} = 20\log\left(\frac{V_{out}}{V_{in}}\right) = 20\log\left(\frac{2}{1}\right) = 20\log(2) = 6.02\ dBVrms$$

5.2.2 Problem 2 : Determination of Fourier series coefficients

1. Determine the Fourier series coefficients up to the 5th harmonic of the function.

$$f(t) = \frac{2}{T}t \quad -\frac{T}{2} \leq 0 \leq \frac{T}{2}$$

a_0 and a_n will go to 0 since it is an odd function. But b_n =

$$b_n = \frac{2}{T}\int_{\frac{-T}{2}}^{\frac{T}{2}} \frac{2}{T}t.\sin(nw_o t)\,dt$$

$$= \frac{4}{T^2}\left[-\frac{t}{(nw_0)} \times \cos(nw_0 t) + \frac{1}{(nw_0)^2}\sin(nw_0 t)\right]_{-\frac{T}{2}}^{\frac{T}{2}}$$

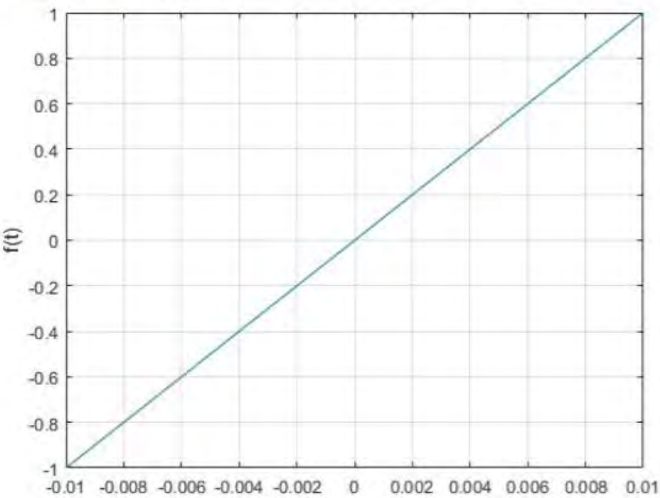
$$\textit{The sine will always be zero.}$$

$$\therefore\ b_n = \frac{-2(-1)^n}{\pi n}$$

2. Use MatLab to plot the original function.

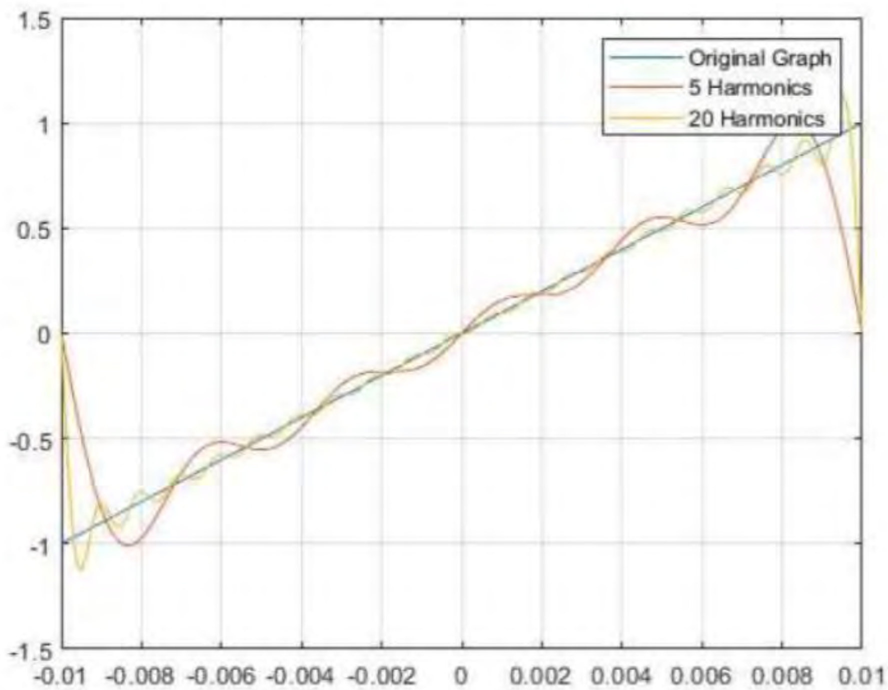
```
%%Original funtion
clear all;
T = 0.01*2;
t = [- T/2: 0.0001: T/2]; %Since 2pi/wo = 0.0001sec
y= (2/T)*t;
plot (t,y);
xlabel('time');
ylabel('f(t)');
grid on
hold on
```

Original function graph:



3. Use MatLab and plot function using the calculated coefficients.

```
%%Original graph
T=0.01*2;
t=[(-T/2):0.0001:(T/2)];
y=(2/T)*t;
plot(t,y);
grid on
hold on

%%Plotting fourier coefficient
f=0;
for n=1:5
    f=f+((-2*(-1)^n/(n*pi)))*sin(n*pi*2/T*t);
end

plot(t,f);
grid on
hold on

%%For 20 harmonics:
f=0;
for n=1:20
    f=f+((-2*(-1)^n/(n*pi)))*sin(n*pi*2/T*t);
end

plot(t,f);
grid on
hold on

legend('Original Graph','5 Harmonics','20 Harmonics');
```



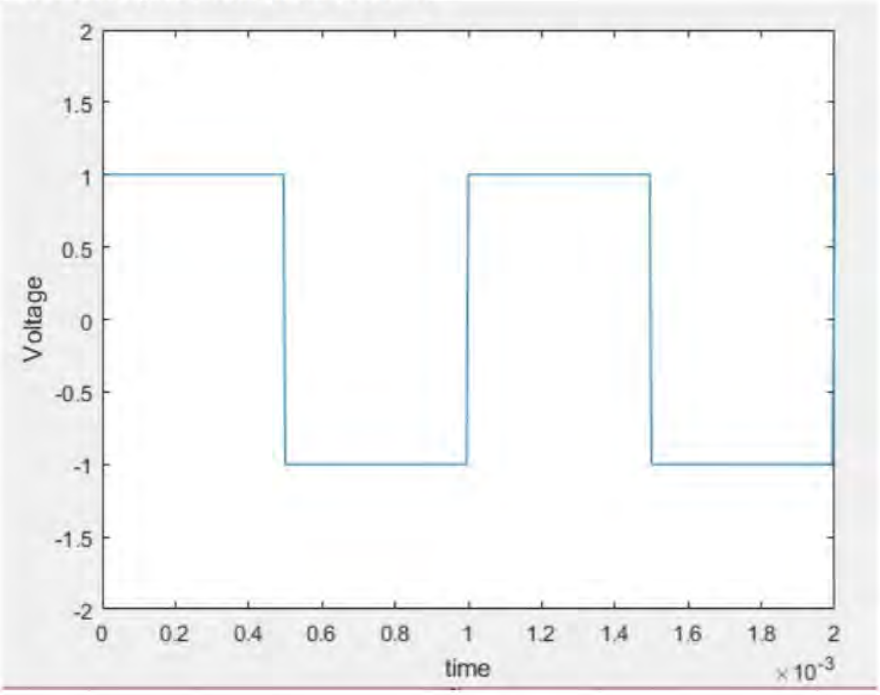
5.2.3 Problem 3 : FFT of a Square/Rectangular Wave

Write a Matlab script:

1. Generate a square wave of 1 ms period, 2 Vpp amplitude, no offset, and duty cycle 50% (hint : use 'square' function). Use 200 kHz as the sampling frequency for the problem.

```
clear all;
clc;
t= 0:0.000005: 0.002;
f_t= square(2*pi*1000*t,50);
plot(t,f_t);
axis([0 0.002 -2 2]);
ylabel('Voltage');
xlabel('time');
```

2. Plot the square wave in time domain

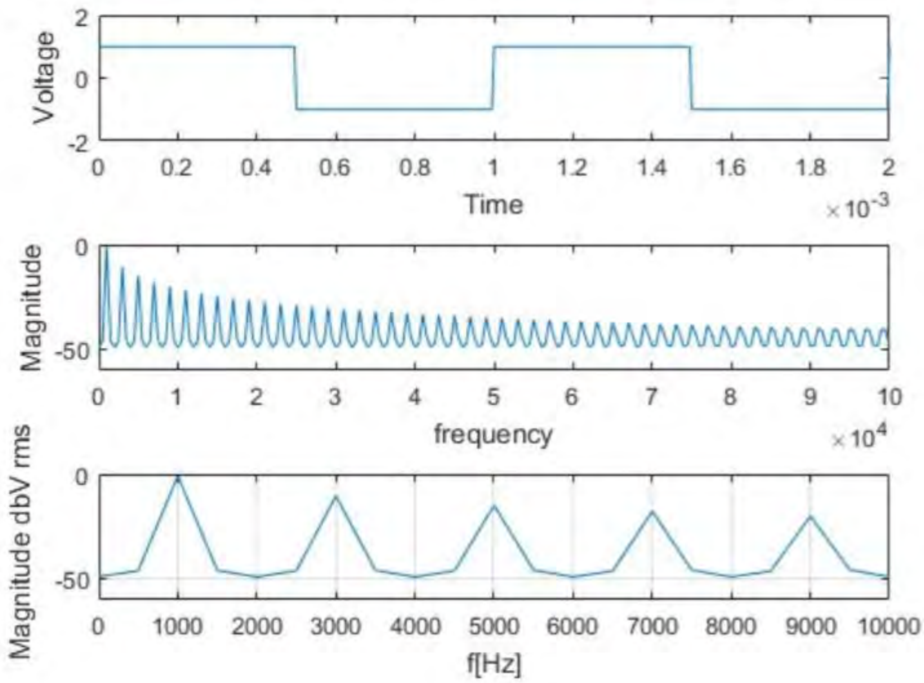


3. Obtain the FFT spectrum using Matlab FFT function. Make the FFT length to be the length of the square wave data vector. (Also for part 7)

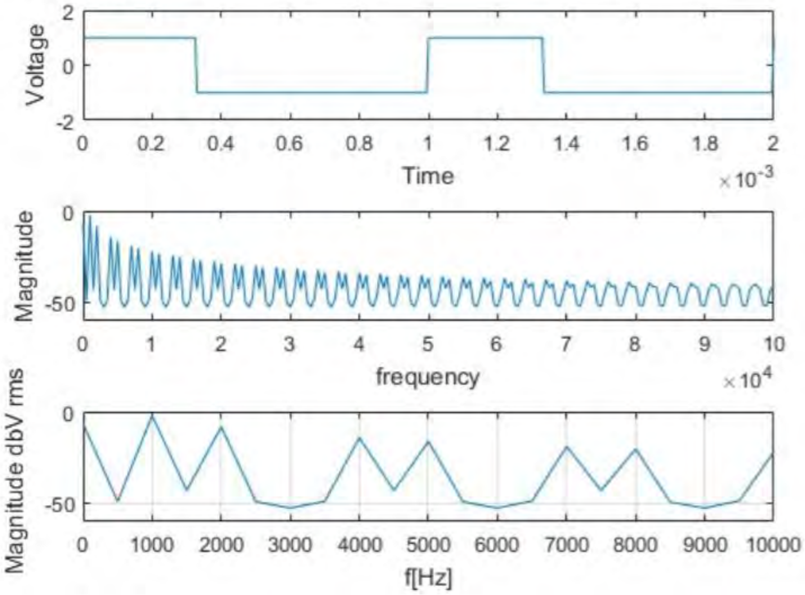
```
clear all;
clc;
t= 0:0.000005: 0.002;
%axis([0 0.002 -2 2]);
%ylabel('Voltage');
%xlabel('time');
sampf= 200000;
nyq_f= sampf/2;
duty= [50 33 20];

for i=1:3
   dcycle= duty(i);
   f_t= square(2*pi*1000*t,dcycle);
   figure(i)
   subplot(3,1,1);
   plot(t,f_t);
```

```
    xlim([0 0.002]);
    ylim([-2 2]);
    xlabel('Time');
    ylabel('Voltage');

    %%FFT of square wave
    l=length(f_t);
    fft_ft=fft(f_t)/l;
    f= sampf*(0:l-1)/(l-1);
    SSL=ceil(l/2);
    f=f(1:SSL);
    fft_ft=fft_ft(1:SSL);

    %magnitude in dBVrms
    Mag = 20*log10(2*abs(fft_ft/sqrt(2)));

    subplot(3,1,2)
    plot(f,Mag)
    ylim([-60 0])
    ylabel('Magnitude');
    xlabel('frequency');

    subplot(3,1,3)
    plot(f,Mag)
    xlabel ('f[Hz]')
    ylabel ('Magnitude dbV_ {rms }')
    ylim([-60 0])
    xlim( [0 10000] )
    grid on
end
```

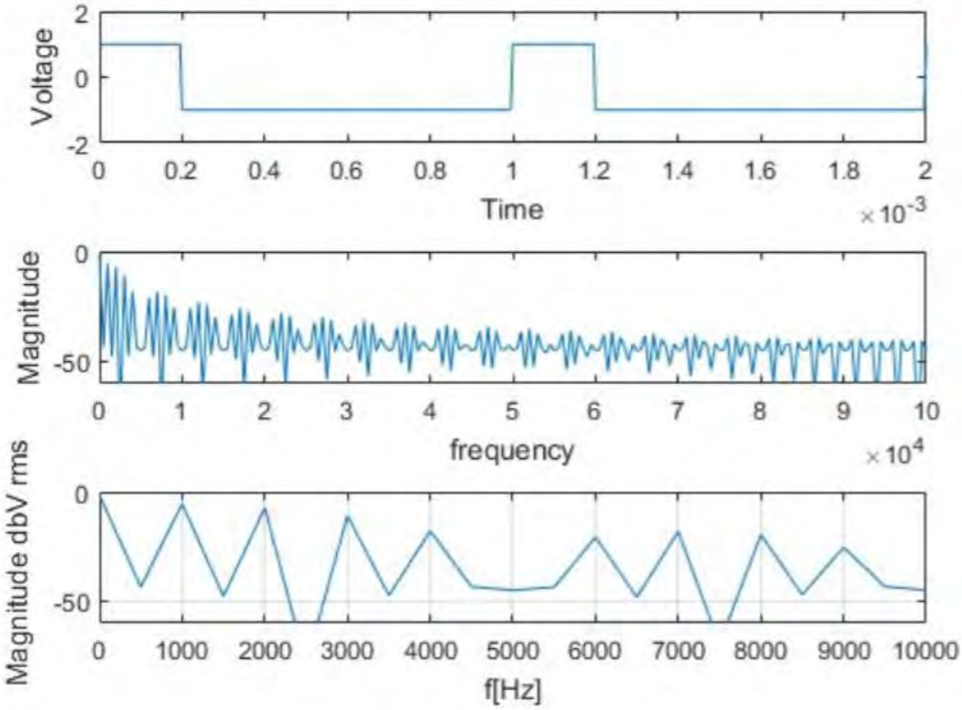
50% duty:



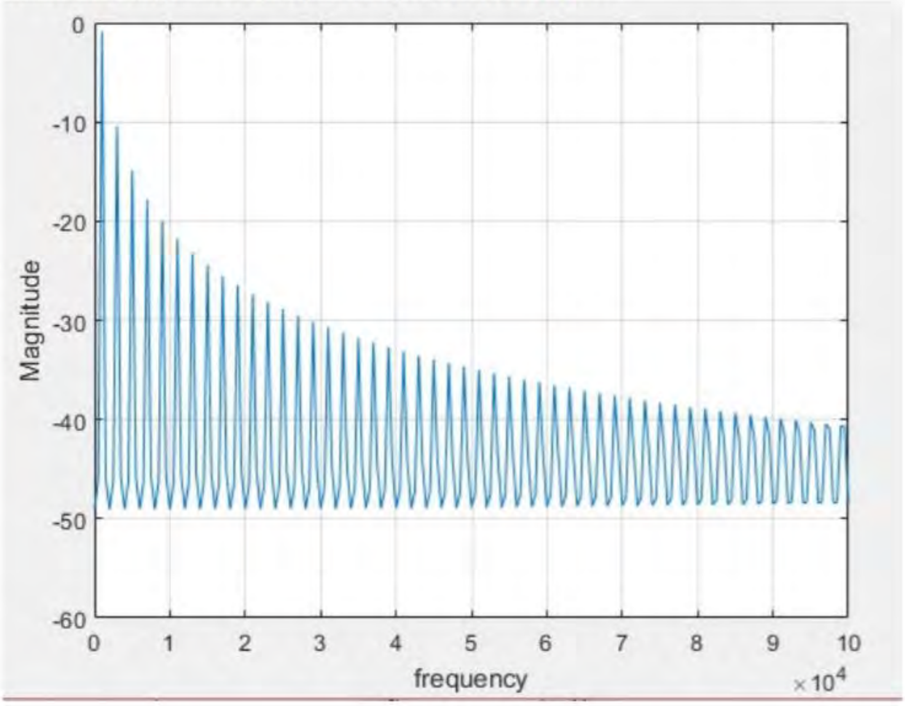
33%duty cycle:



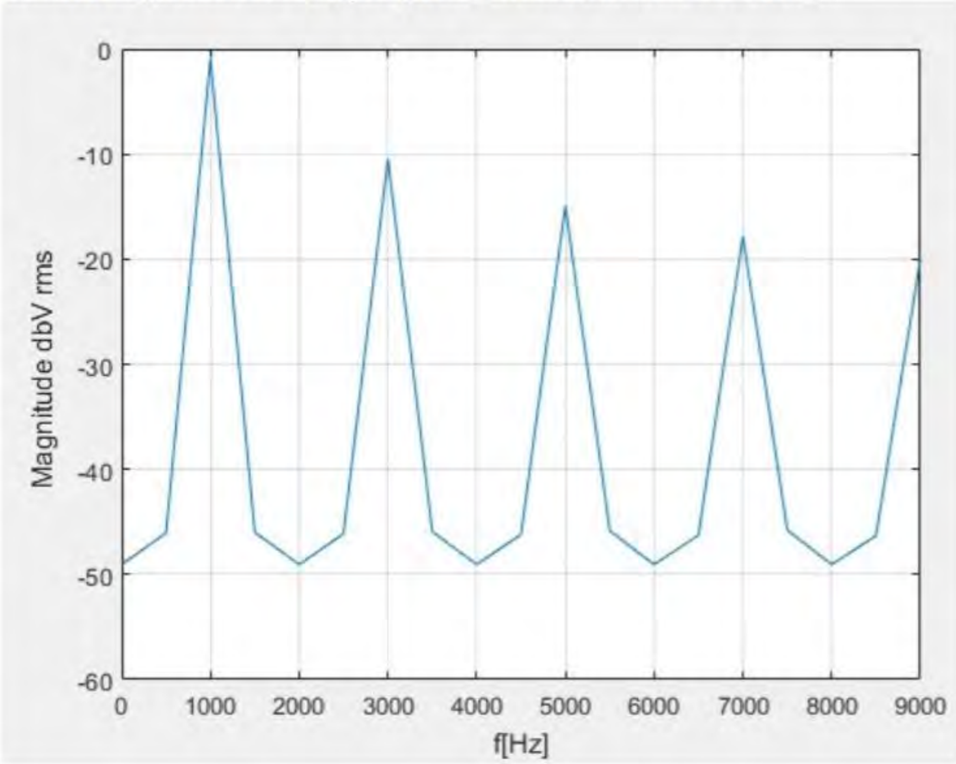
20%duty cycle:



4. Plot the single-sided amplitude spectrum in dBVrms.



5. Plot the spectrum including only the first four harmonics in dBVrms.



```
clear all;
clc;
t= 0:0.000005: 0.002;
axis([0 0.002 -2 2]);
ylabel('Voltage');
xlabel('time');
sampf= 200000;
nyq_f= sampf/2;
duty= [50 33 20];
dcycle= duty(1);
f_t= square(2*pi*1000*t,dcycle);
figure(1)
plot(t,f_t);
%%FFT of square wave
l=length(f_t);
fft_ft=fft(f_t)/l;
f= sampf*(0:l-1)/(l-1);
SSL=ceil(l/2);
f=f(1:SSL);
fft_ft=fft_ft(1:SSL);
%magnitude in dBVrms
Mag = 20*log10(2*abs(fft_ft/sqrt(2)));
figure(2)
plot(f,Mag)
ylim([-60 0])
ylabel('Magnitude');
xlabel('frequency');
grid on
figure(3)
plot(f,Mag)
xlabel ('f[Hz]')
ylabel ('Magnitude dbV_ {rms }')
ylim([-60 0])
xlim( [0 9000] )
grid on
```

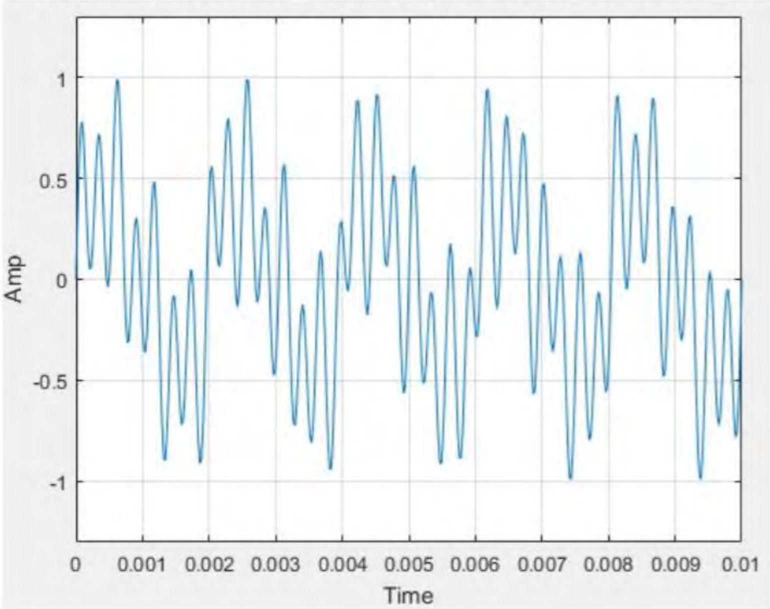
7. Discuss the changes for smaller pulse width.
From equation in the lab manual,

$$c_k = \frac{2}{k\omega_0 T}\sin(k\omega_0 T_1) = \frac{1}{k\pi}\sin(k\omega_0 T_1)$$

The magnitudes of the spectrum are different on each duty cycle. This can be scene in the pictures above. If duty cycle is decreased, magnitude spectrum reaches zero for different multiples of omega. In the equation above, T is the period of the signal and T_1 is the width of the periodic pulses. The magnitude in the impulse of the spectrum increase when the pulse width is smaller. For 20% duty cycle, the impulse spectrum is higher than 33% which is higher than 50% in FFT.

5.2.4 Problem 4 : FFT of a sound sample

2. Using Matlab, read the sound file and plot the first 10 ms of the signal in time domain.

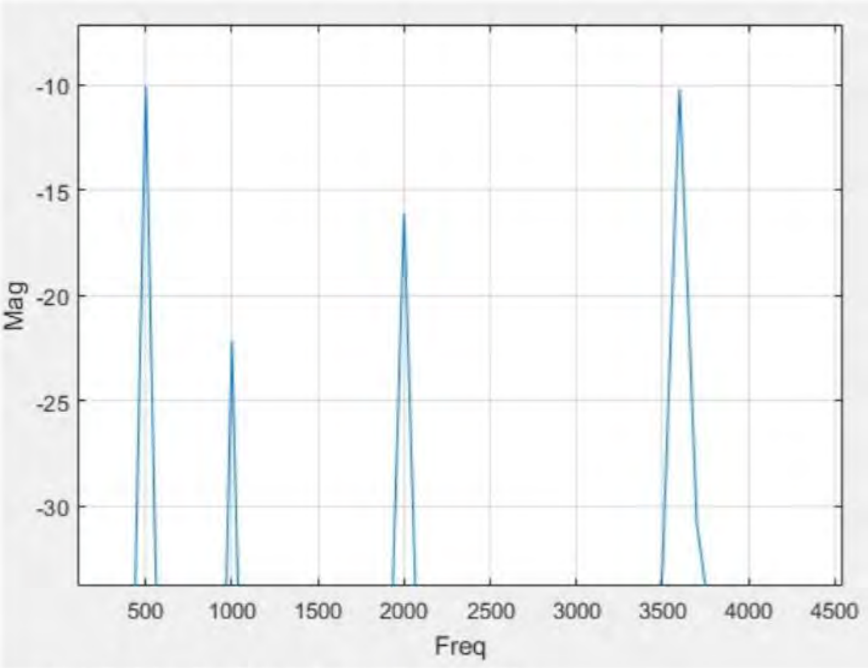


Matlab:

```
[ft, sampf] =audioread('E:\YEAR 2(CORE)\ELectronics\Signals and Syestem Lab\Experiment 2\sound_sample.wav');
delt= 1/sampf;
l=length(t);
t= 0: delt:0.01;
ft1=ft(1:l);
figure(1)
plot(t,ft1)
ylim([-1.3 1.3])
ylabel('Amp');
xlabel('Time');
grid on
hold on
```

3. Use the Matlab FFT function to compute the spectrum and plot the single-sided amplitude spectrum in dBVrms..

```
[ft, sampf]=audioread('E:\YEAR 2(CORE)\ELectronics\Signals and Syestem Lab\Experiment 2\sound_sample.wav');
delt= 1/sampf;
L1=length(t);
t= 0: delt:0.01;
ft2= ft(1:L1);
L2=length(ft2);
fft_audio=fft(ft2)/L2;
f= sampf*(0:L2-1)/(L2-1);
SSL=ceil(L2/2);
fft_audio=fft_audio(1:SSL);
f=f(1:SSL);
Mag= 20*log10(2*abs(fft_audio)/sqrt(2));
plot(f,Mag)
ylim([-60 0])
ylabel('Mag');
xlabel('Freq');
grid on
hold on
```



4. What are the tones forming this signal?
The tones forming are the values at which half of the fft spectrum appears i.e. 500Hz, 1000Hz, 2000Hz and 3600Hz.

Reference:

Lab Manual