4TL4: Digital Signal Processing
Lab #2

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C01
L02

Group 1

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As a future member of the engineering profession, the student is responsible for performing the required work in an honest manner, without plagiarism and cheating. Submitting this work with my name and student number is a statement and understanding that this work is my own and adheres to the Academic Integrity Policy of McMaster University and the Code of Conduct of the Professional Engineers of Ontario. Submitted by **[Zhengda Li, li939, 400324486]**
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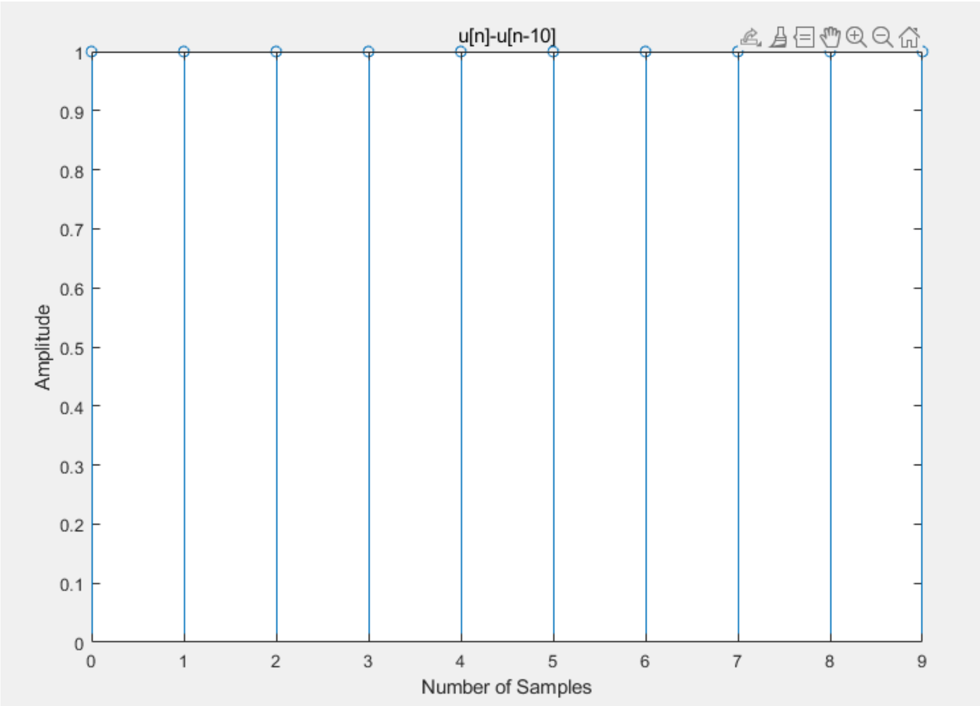
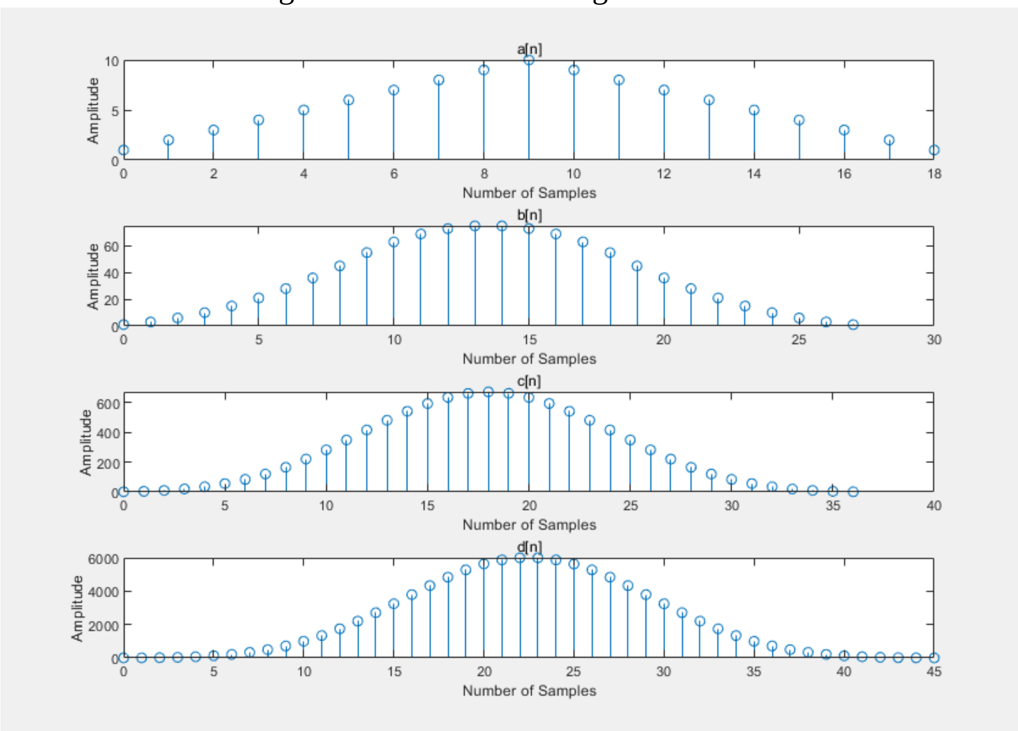
Experiment 1: Introduction to Convolution

First, we created the discrete-time sequence $x[n] = u[n] - u[n-10]$ using MATLAB. The x[n] we got is shown in the graph below:



Note that we removed all the zero values because we want to make the convolution graph cleaner. Convolved signals are shown in the figure below:

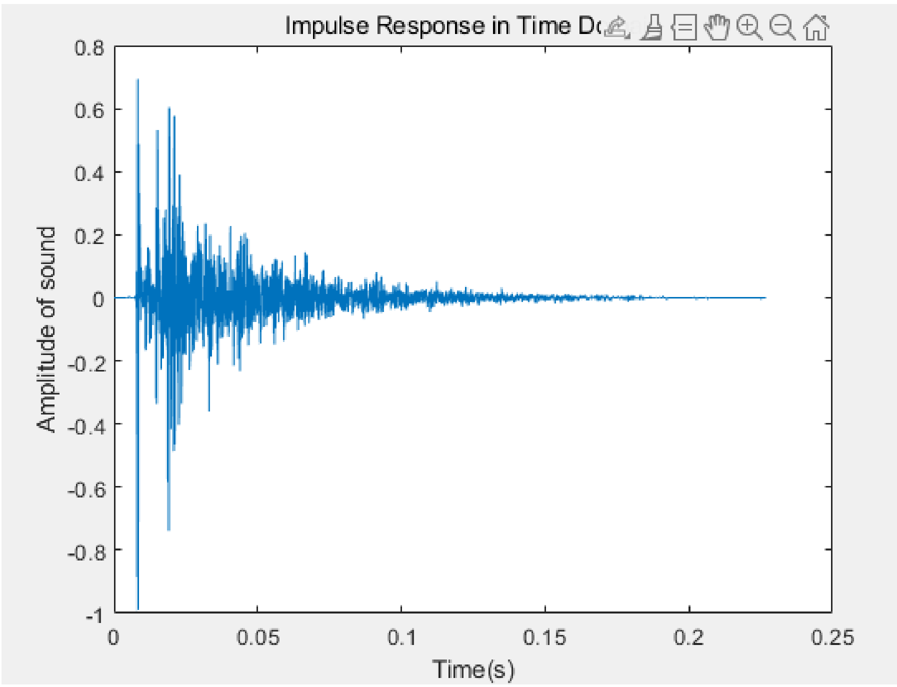


We are expecting the wave peak to become more and more obvious due to the convolution. The output graph matches the expectation.

Experiment 2: Convolution of Signals and System Impulse Response

(a) and (b)

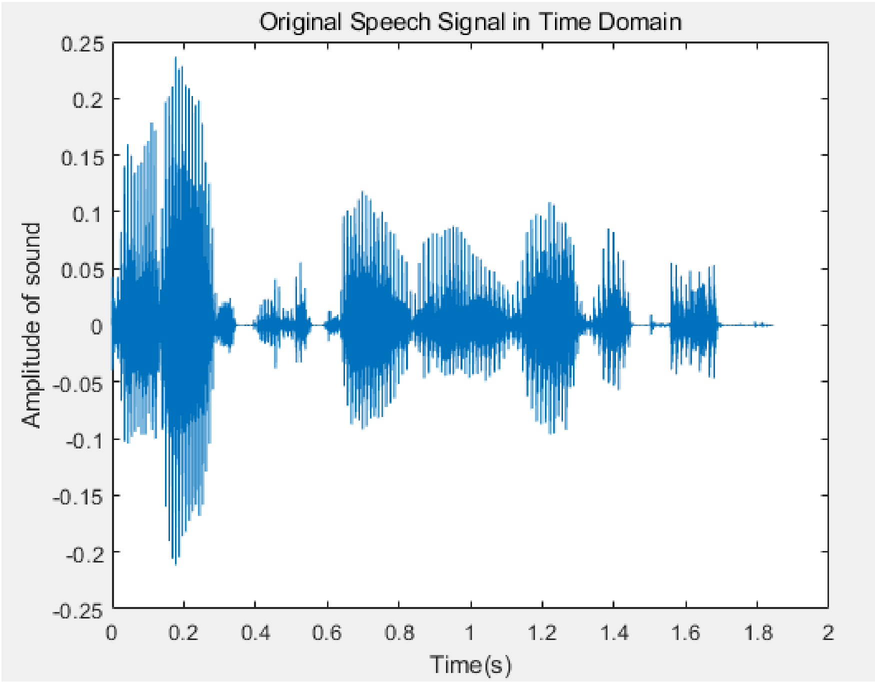
The impulse response of impr.wav signal is in the figure below:



As we can observe in the figure, the impulse response shows a quick and short spike. And we heard something like clapping hands if we play the sound file.

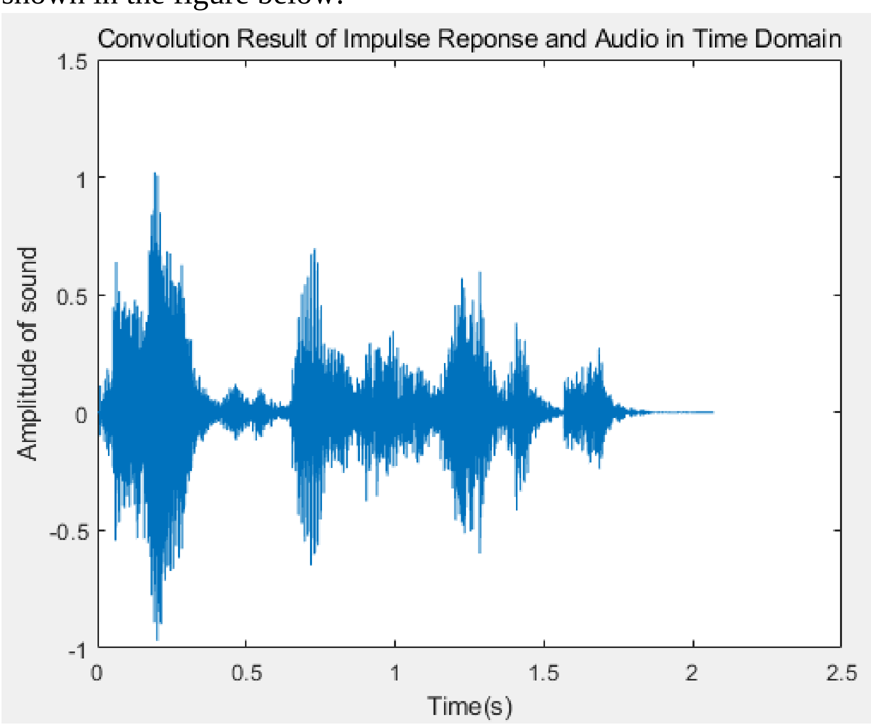
(c)

Then we will load the speech file and plot the signal using MATLAB.



(d)

Lastly, we convolve the speech signal with the impulse response and the convolved signal is shown in the figure below:



We could see that the signal is longer in the length due to convolution. When doing convolution, we are using the shape of one signal to modify the shape of another signal. As a result, we can observe that the convolved speech signal has similar quick spike as the impulse response. And after we play the audio, we could hear the clap sound and echo with the speech.

Experiment 3: The Discrete-Time Fourier Transform (DTFT)

(a) The DTFT function is shown as below.

```
function output_dtft = calculate_dtft(x,w)
N=-1000:1000;%set sample rate to 100Hz
L = N-N(1);%shift of N
W = (w(2)-w(1))/(length(N)-1);%step
Y = zeros(1,length(N));
for a = 1:length(N)
    for b = 1:length(x)
        Y(a)=Y(a)+x(b)*exp(-1i*(w(1)+W*L(a))*(b-1));
    end
end
output_dtft = Y;
```

(b) Calculation by hand and Matlab function is shown as below.

$$h_1[n] = \frac{1}{4}\delta[n] + \frac{1}{2}\delta[n-1] + \frac{1}{4}\delta[n-2]$$

$$F\{h_1[n]\} = \frac{1}{4}F\{\delta[n]\} + \frac{1}{2}F\{\delta[n-1]\} + \frac{1}{4}F\{\delta[n-2]\}$$

$$= \frac{1}{4} + \frac{1}{2}e^{jw} + \frac{1}{4}e^{2jw}$$

$$= \frac{\frac{1}{2}\cos\theta + \frac{1}{2}}{e^{jw}}$$

$$h_2[n] = -\frac{1}{4}\delta[n] + \frac{1}{2}\delta[n-1] - \frac{1}{4}\delta[n-2]$$

$$F\{h_1[n]\} = \frac{1}{4}F\{\delta[n]\} + \frac{1}{2}F\{\delta[n-1]\} + \frac{1}{4}F\{\delta[n-2]\}$$

$$= -\frac{1}{4} + \frac{1}{2}e^{jw} - \frac{1}{4}e^{2jw}$$

$$= \frac{-\frac{1}{2}\cos\theta + \frac{1}{2}}{e^{jw}}$$

X1

1x2001 complex double

	1	2	3	4	5	6	7	8	9	10	11	12
1	0.0000e+0...	-2.2040e-0...	-8.6170e-0...	-1.8650e-0...	-3.1345e-0...	-4.5412e-0...	-5.9282e-0...	-7.1161e-0...	-0.0008 + ...	-0.0008 + ...	-0.0008 + ...	-0.0006 + ...

	13	14	15	16	17	18	19	20	21	22	23	24	25
1	-0.0003 + ...	0.0001 + 0...	0.0006 + 0...	0.0013 + 0...	0.0022 + 0...	0.0031 + 0...	0.0043 + 0...	0.0055 + 0...	0.0068 + 0...	0.0082 + 0...	0.0097 + 0...	0.0111 + 0...	0.0125 + 0

X2

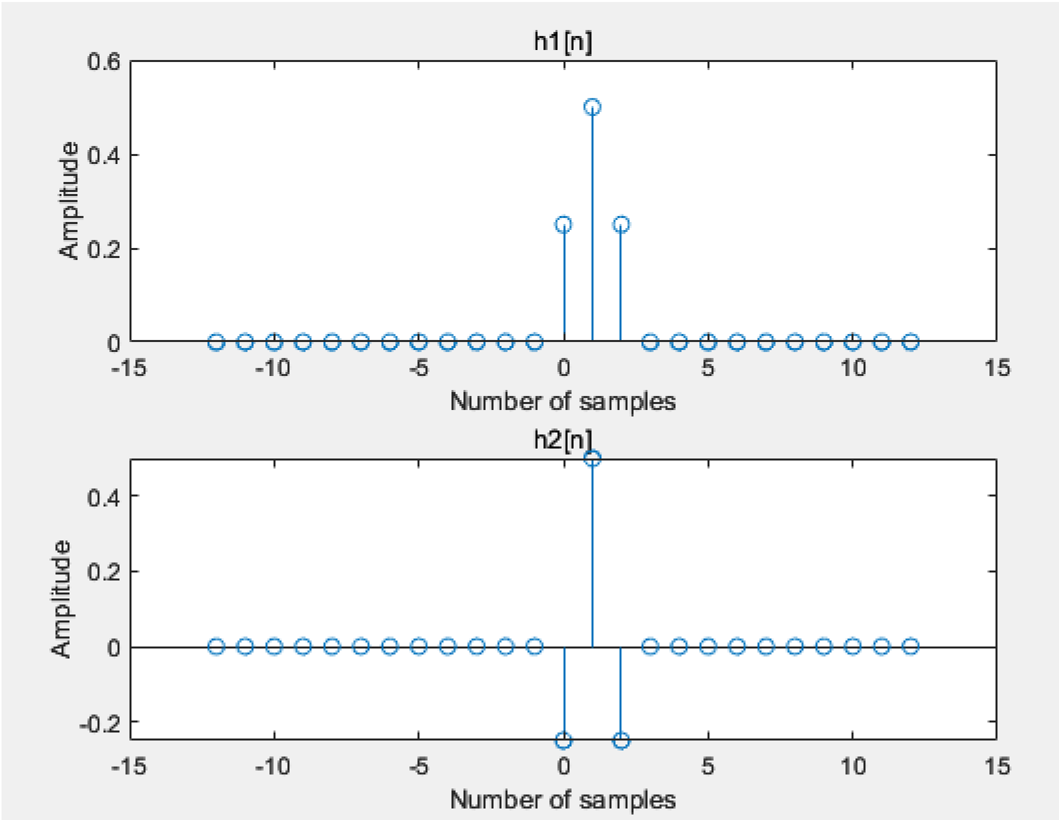
1x2001 complex double

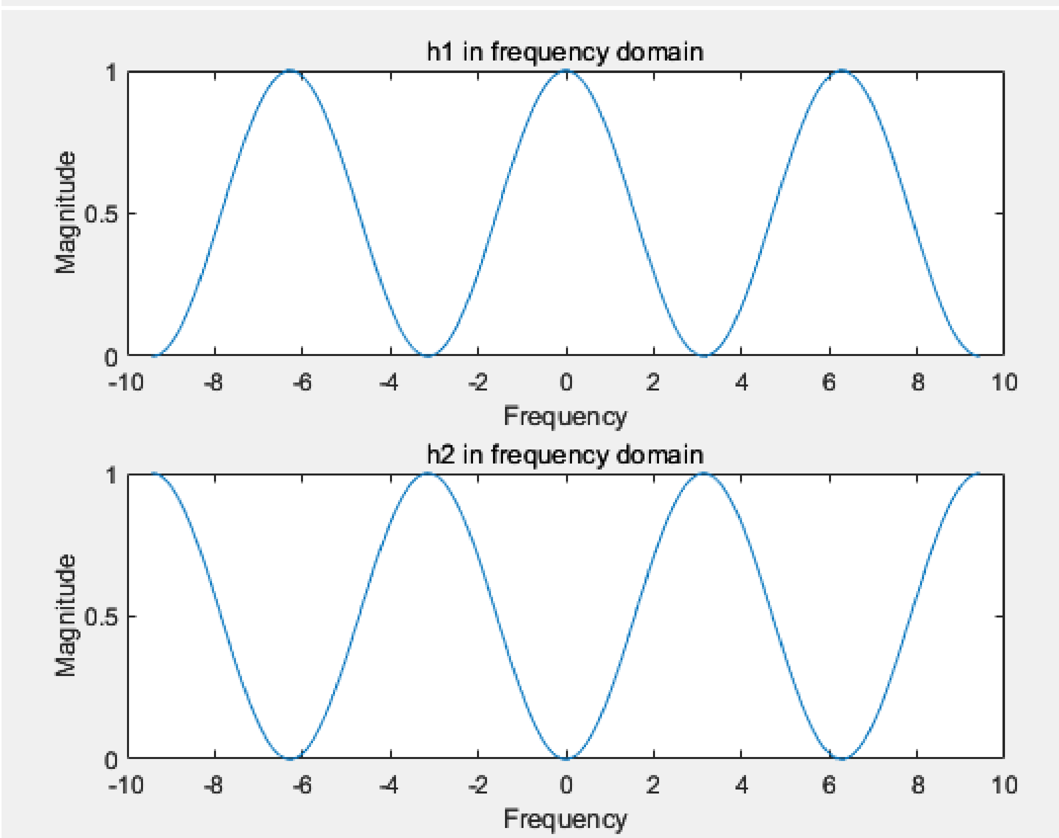
	1	2	3	4	5	6	7	8	9	10	11	12
1	-1.0000 + ...	-0.9925 + ...	-0.9700 + ...	-0.9330 + ...	-0.8820 + ...	-0.8177 + ...	-0.7411 + ...	-0.6535 + ...	-0.5561 + ...	-0.4504 + ...	-0.3380 + ...	-0.2206 + ...

	13	14	15	16	17	18	19	20	21	22	23	24	25
1	-0.1000 + ...	0.0219 + 0...	0.1434 + 0...	0.2626 + 0...	0.3776 + 0...	0.4869 + 0...	0.5886 + 0...	0.6813 + 0...	0.7637 + 0...	0.8344 + 0...	0.8925 + 0...	0.9370 + 0...	0.9674 + 0

We could find the results of DTFT for h1 and h2 are complex numbers, and the results should be complex number.

(c) The magnitudes of DTFTs are shown below.

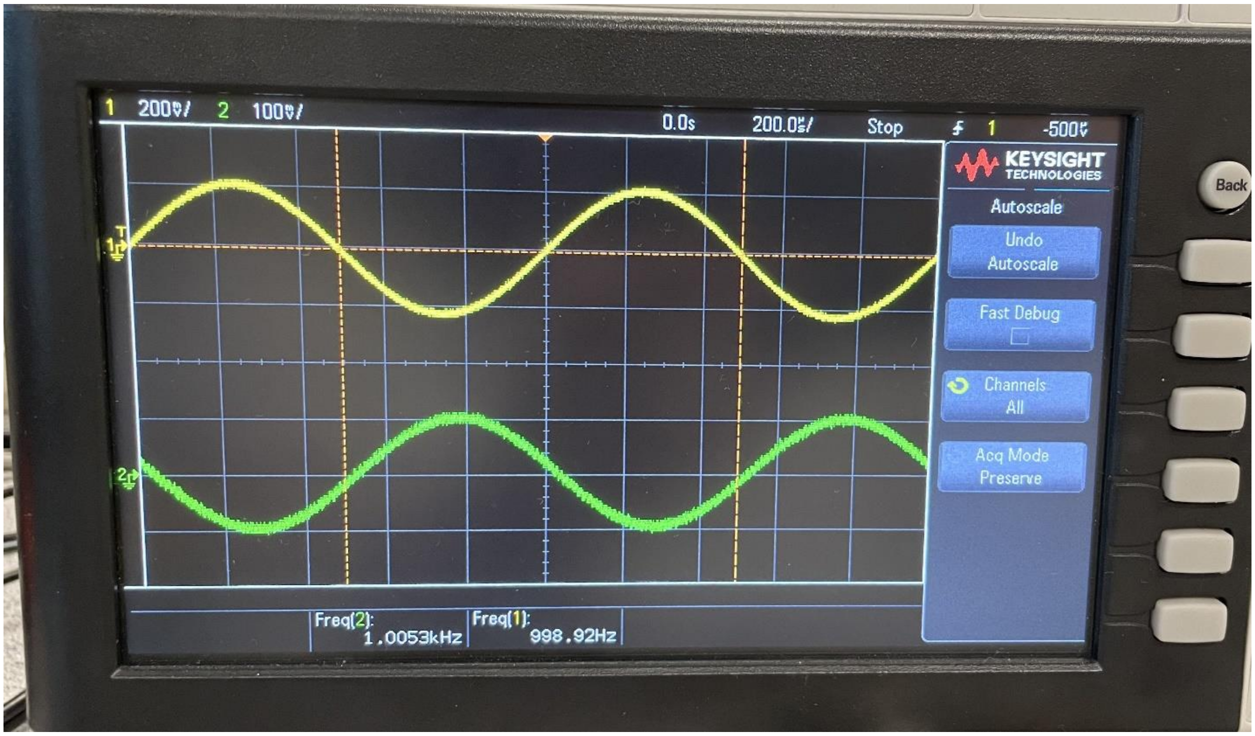


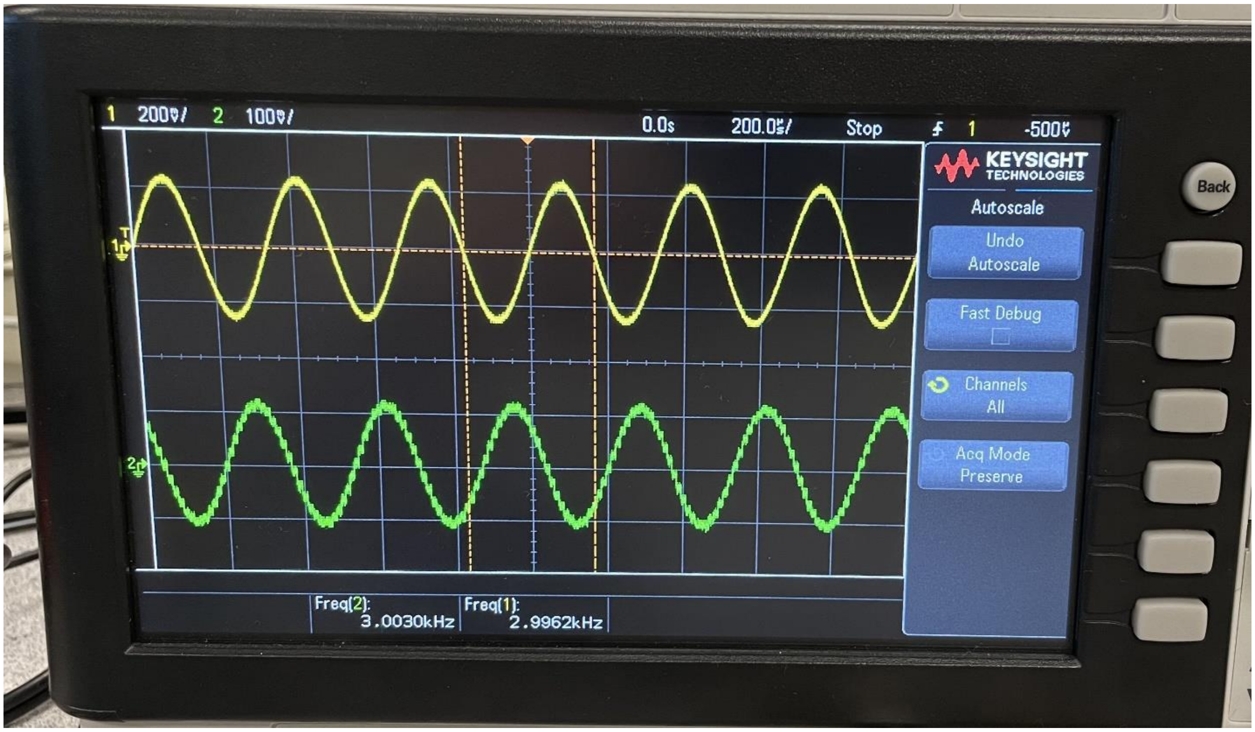


From the figure, we could find periodicity in these spectra, and the period of the results is 2π. From the calculation in part b, the magnitude of h1 is $\frac{1}{2}\cos(w) + \frac{1}{2}$, and the magnitude of h2 is $-\frac{1}{2}\cos(w) + \frac{1}{2}$, where the period of cos is 2π, so the period of DTFTs should also be 2π.

Experiment 4: Effect of Sampling and Aliasing on the TMS320C6713 DSK

(d) fs = 8 kHz, f1 = 1 kHz, f2 = 3 kHz, and f3 = 1 kHz without downsampling:

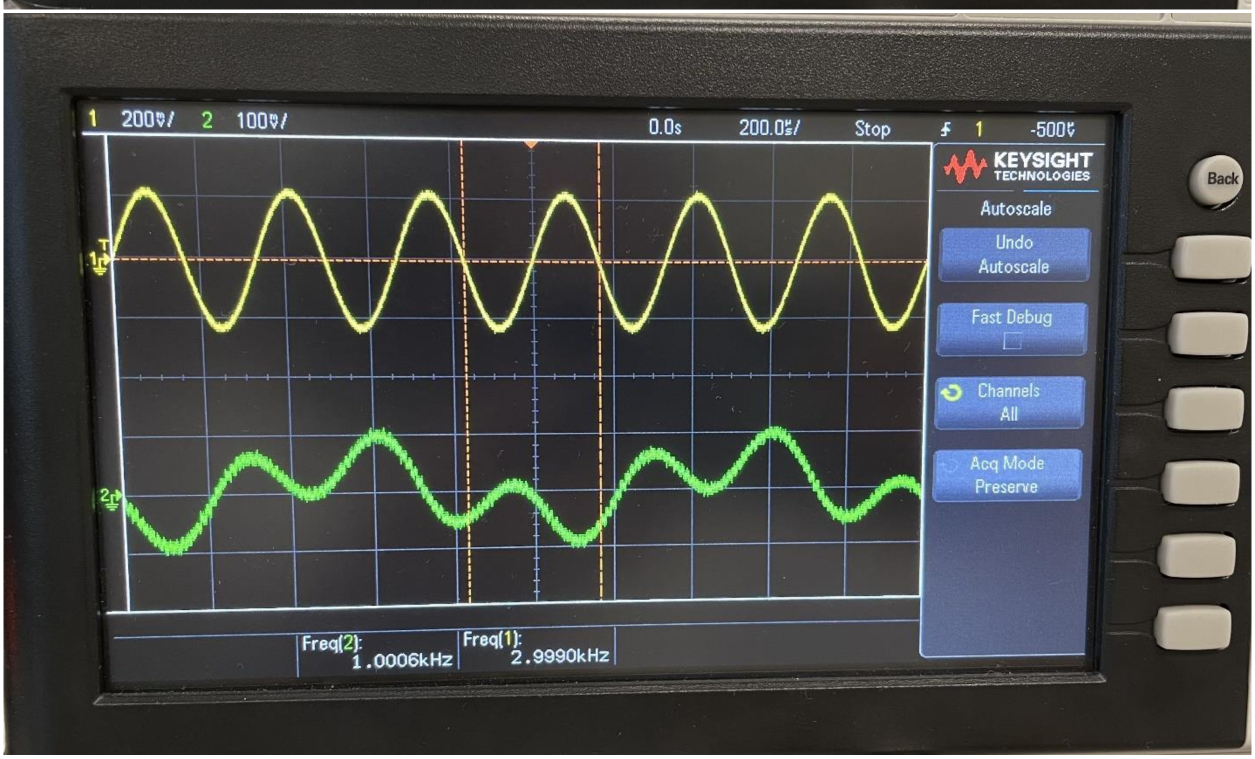


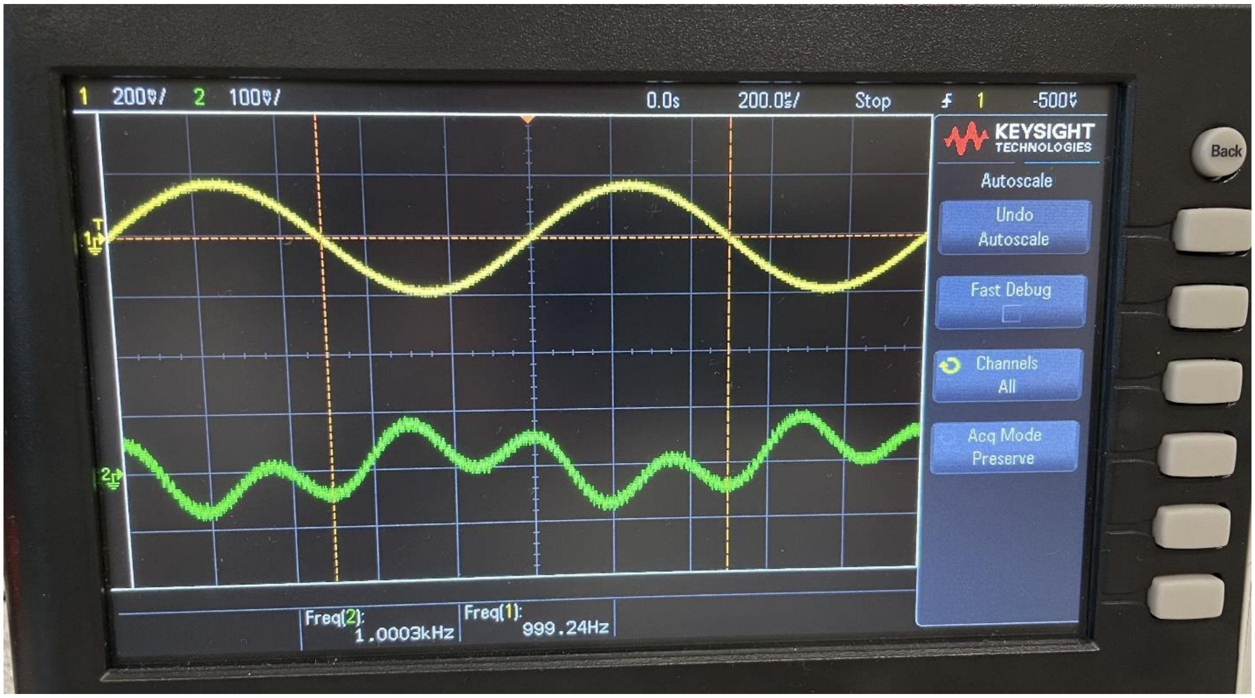


We heard three short clips and the second one has higher frequency.

(e) fs = 8 kHz, f1 = 1 kHz, f2 = 3 kHz, and f3 = 1 kHz with downsampling:

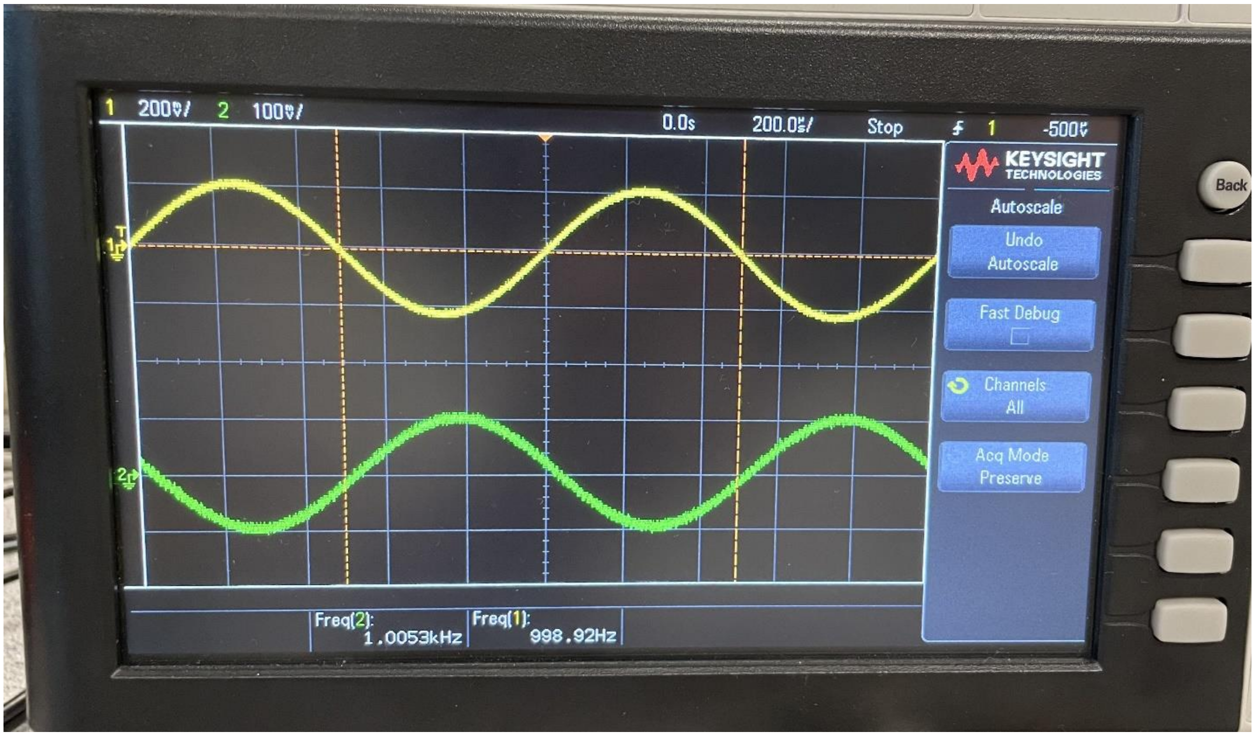

1 200mV/ 2 100mV/
0.0s 200.0µs/ Stop 1 -500mV
KEYSIGHT TECHNOLOGIES
Autoscale
Undo Autoscale
Fast Debug
Channels All
Acq Mode Preserve
Freq(2): 999.32Hz
Freq(1): 1.0003kHz
Back

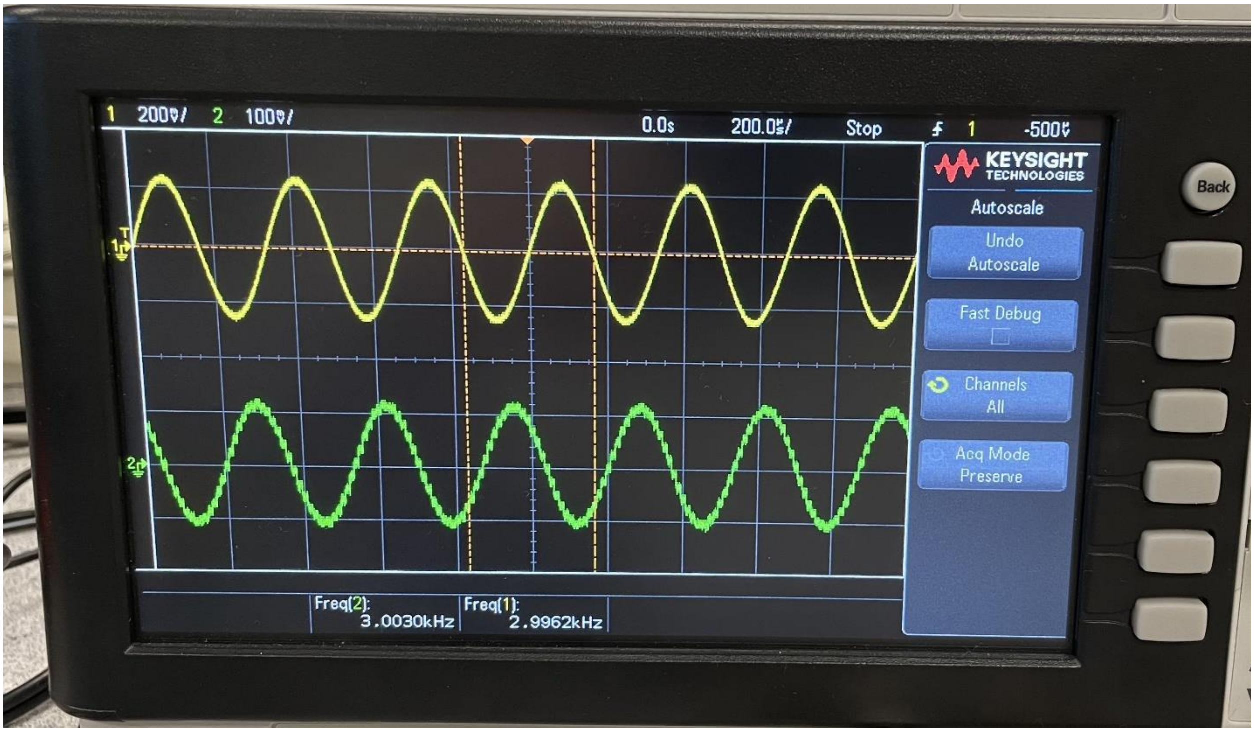

1 200mV/ 2 100mV/
0.0s 200.0µs/ Stop 1 -500mV
KEYSIGHT TECHNOLOGIES
Autoscale
Undo Autoscale
Fast Debug
Channels All
Acq Mode Preserve
Freq(2): 1.0006kHz
Freq(1): 2.9990kHz
Back



We heard three clips and they have the same frequency. We could also observe this on the oscilloscope. That is due to the anti-aliasing of downsampling.

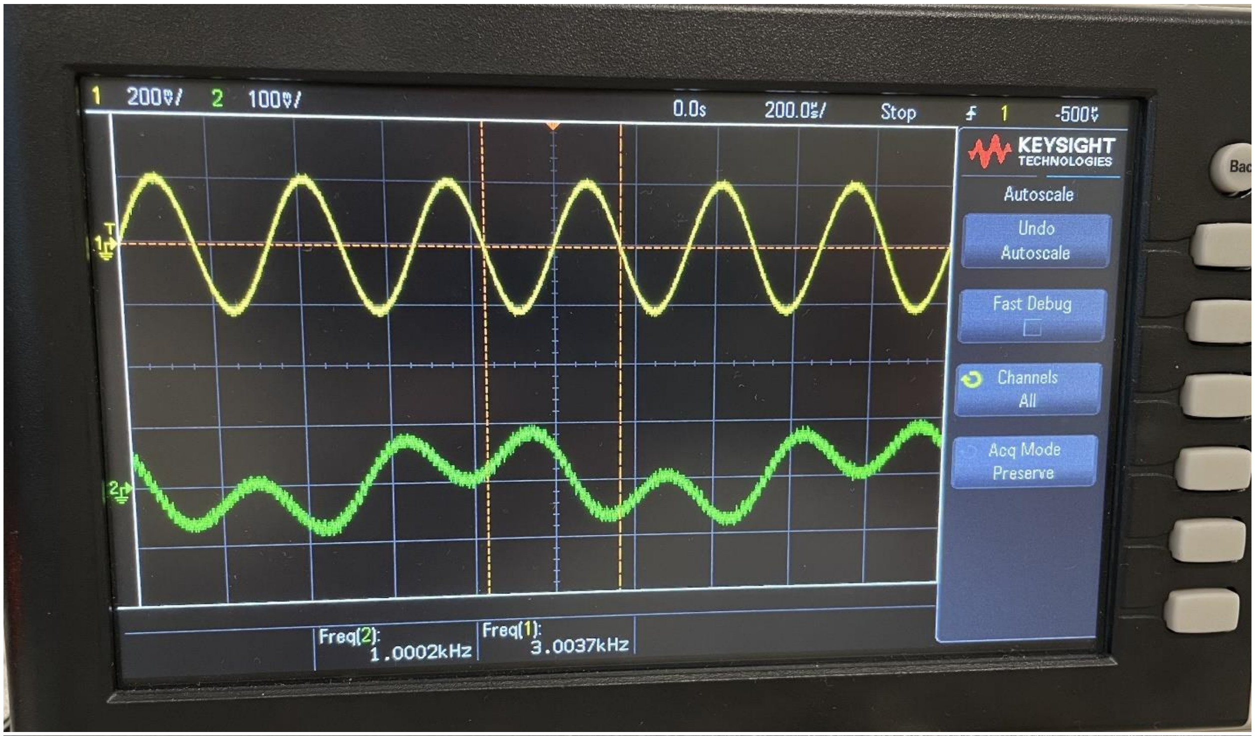
(f) fs = 8 kHz, f1 = 1 kHz, f2 = 5 kHz, and f3 = 1 kHz without downsampling:

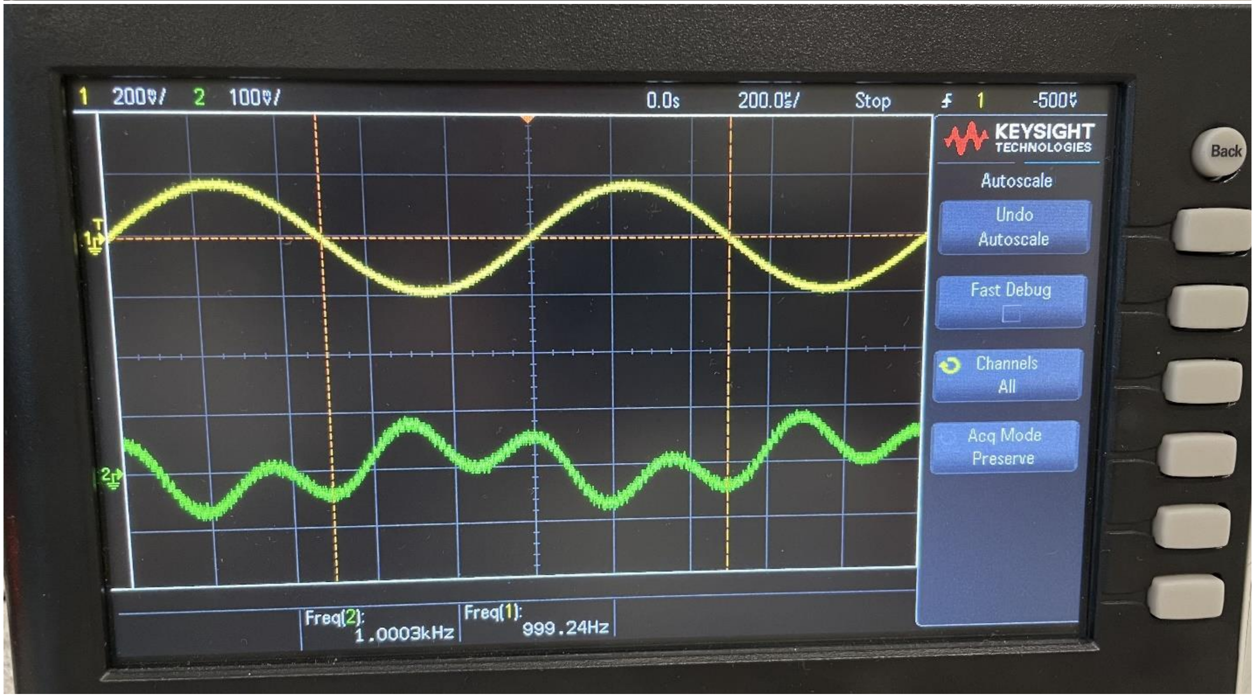




We heard three clips and the second one has higher frequency. However, it sounds the same as 3kHz signal we tested before and the signal is indeed 3kHz on the oscilloscope. This is because 8kHz of sampling rate is not high enough to sample the 5kHz signal according to Nyquist sampling theorem.

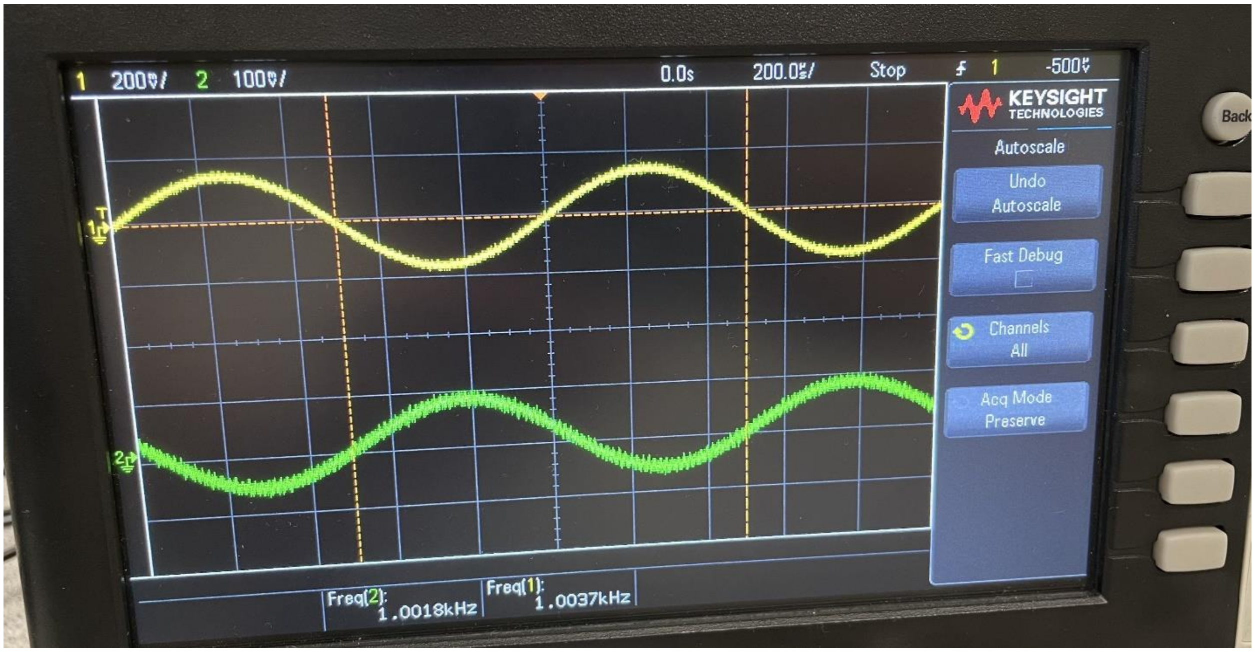
fs = 8 kHz, f1 = 1 kHz, f2 = 5 kHz, and f3 = 1 kHz with downsampling:

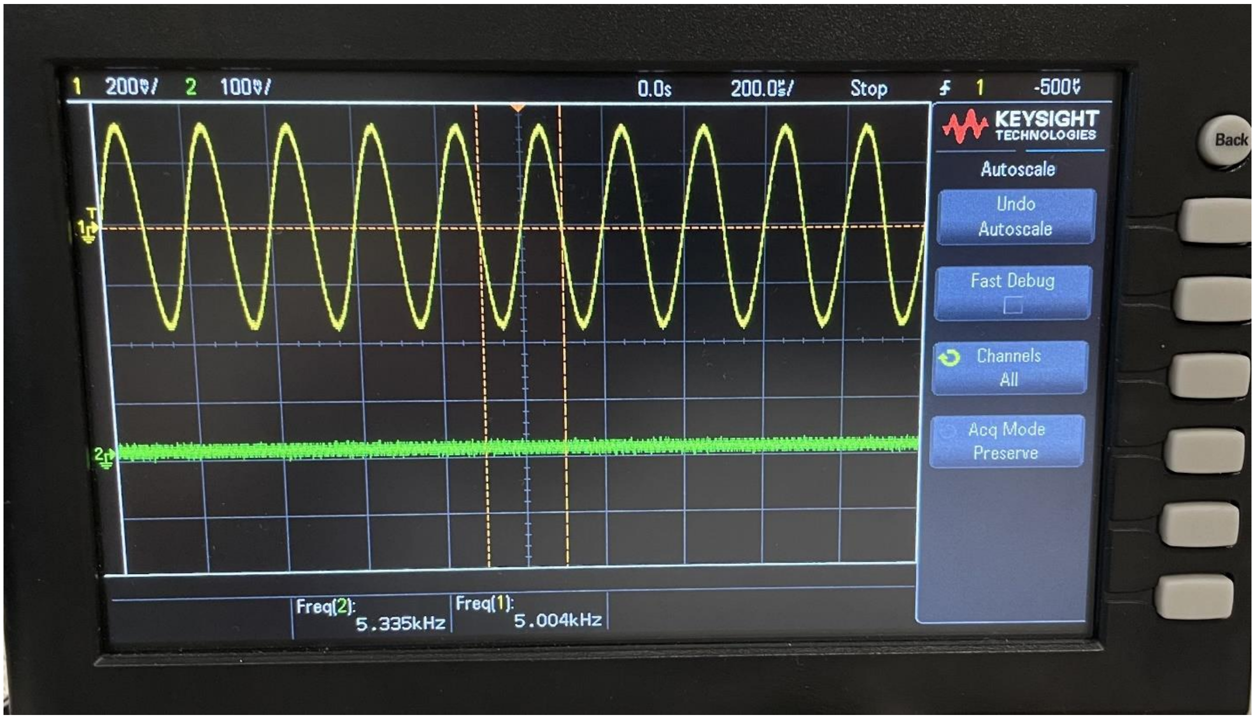


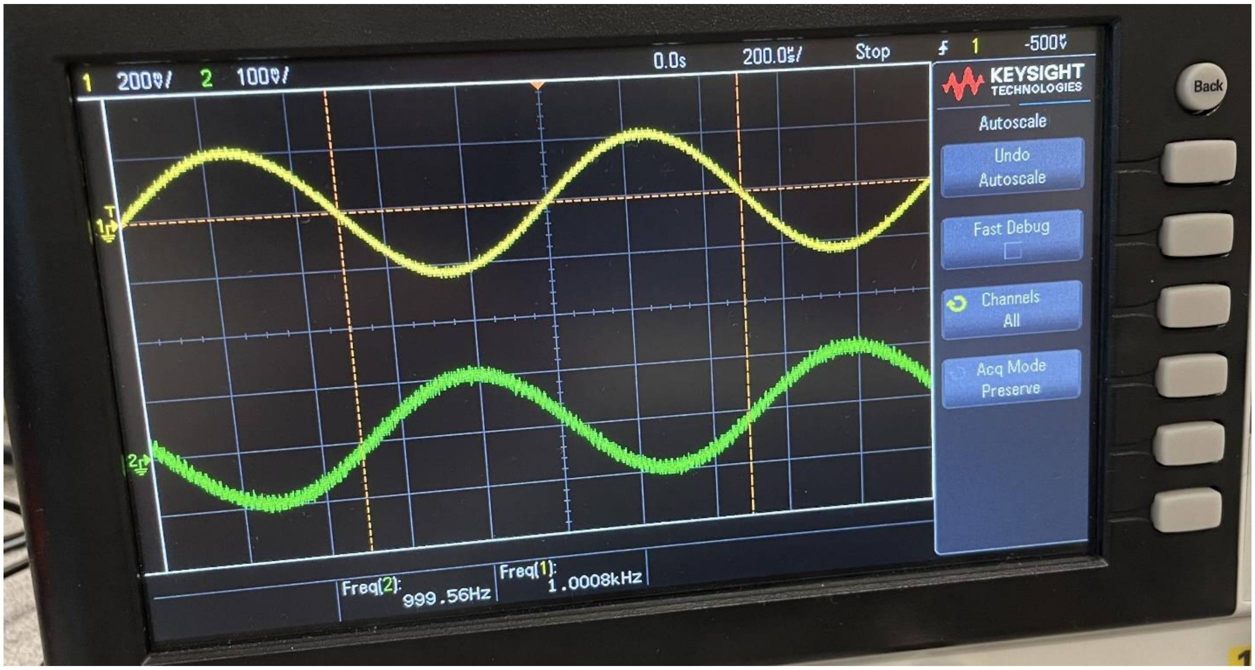


After downsampling, these three signals have same frequencies and we heard three same clips as they are all 1kHz.

(g) fs = 16 kHz, f1 = 1 kHz, f2 = 5 kHz, and f3 = 1 kHz without downsampling:







If we increase the sampling rate to 16kHz then it is high enough to sample the 5kHz signal. However, we cannot hear anything for 5kHz signal possibly because the anti-aliasing filter has filtered the 5kHz signal out. We could still hear the 1kHz signal.