ECE 531 - SOFTWARE DEFINED RADIO

LAB 1

Written by: SINA MALEK

SUBMISSION: 02/07/2022

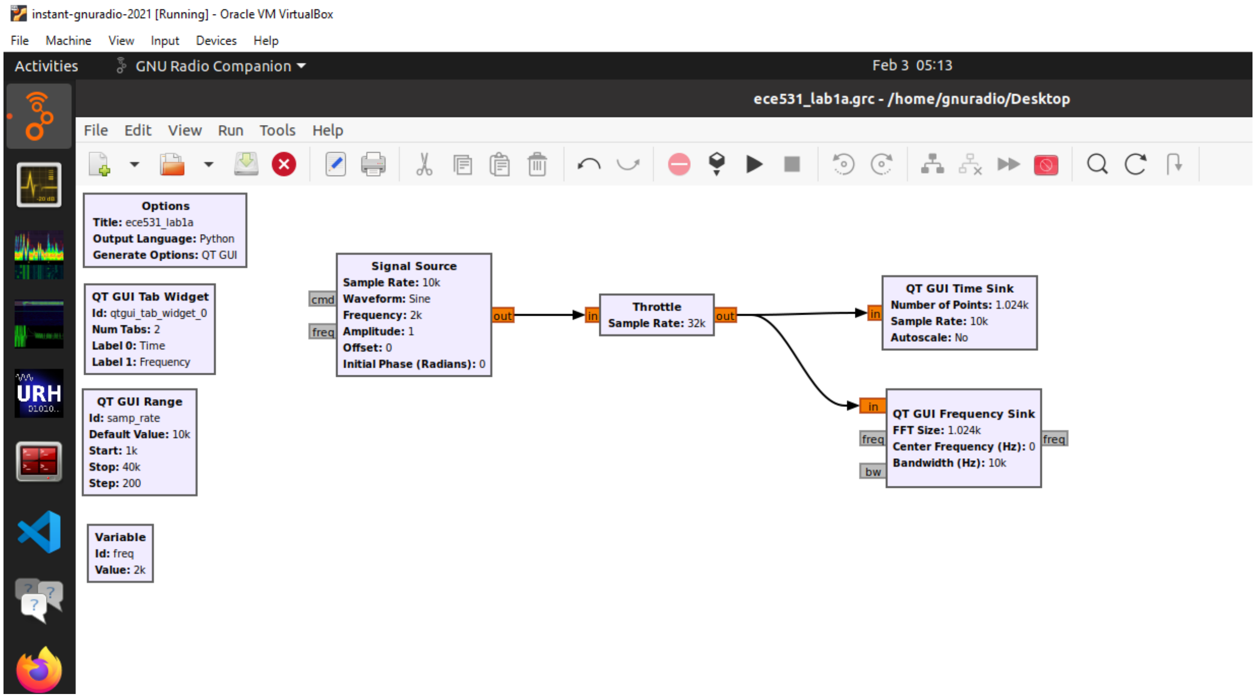
GitHub: https://github.com/maleksina

Abstract

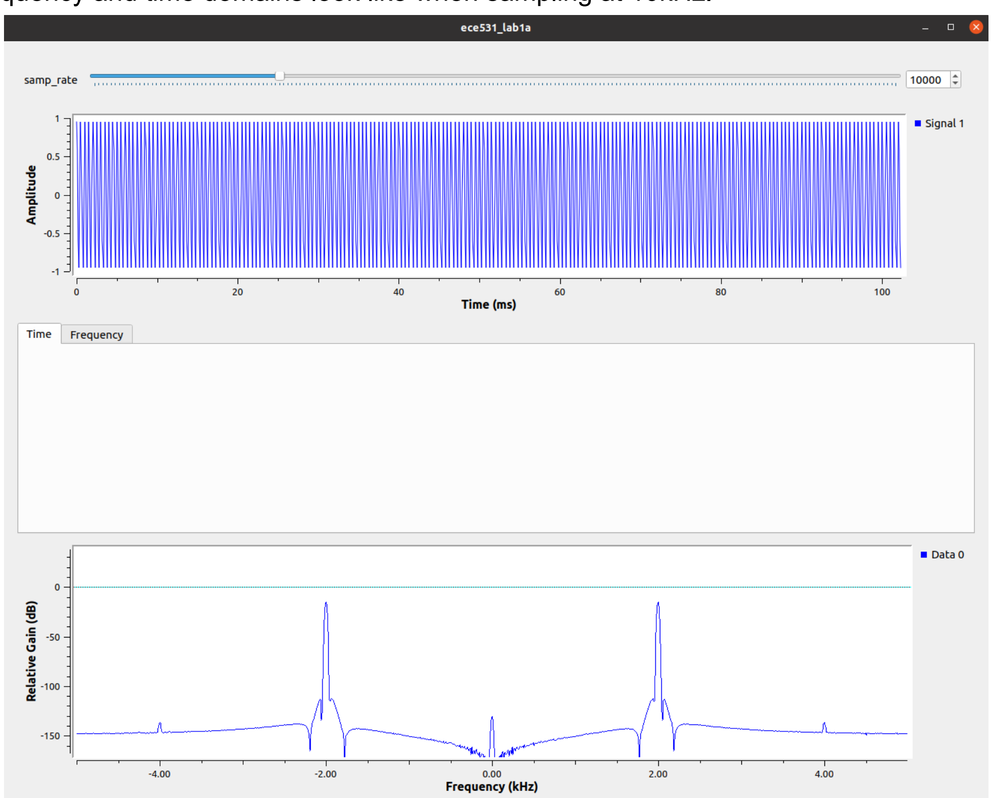
This lab consisted of utilizing GNURadio to create simulations of different scenarios to understand certain fundamentals of SDRs (Software Defined Radio). The software that was used was VirtualBox, an OVA file of the GNURadio. Another software that was installed but will be used later, is MATLAB with Toolboxes. The source code that was used in the lab has been uploaded as well onto my GitHub: https://github.com/maleksina.

3.2 SAMPLING RATES

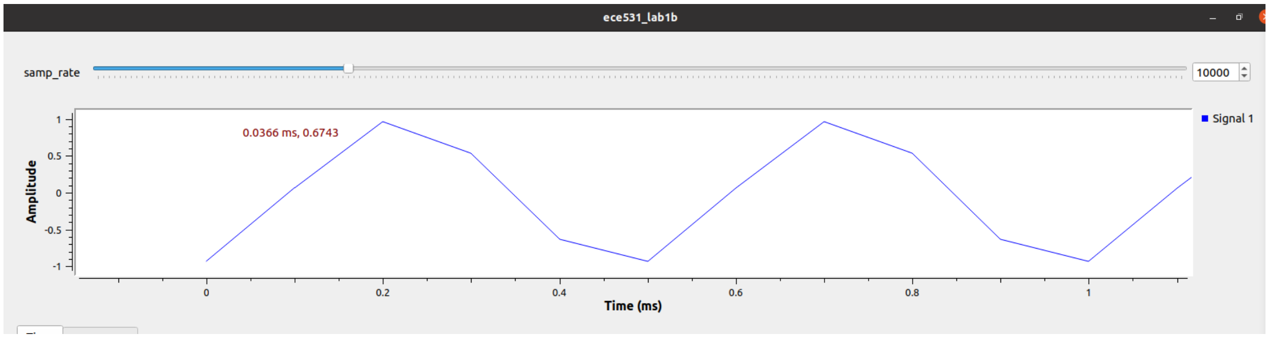
Using the diagram created below:



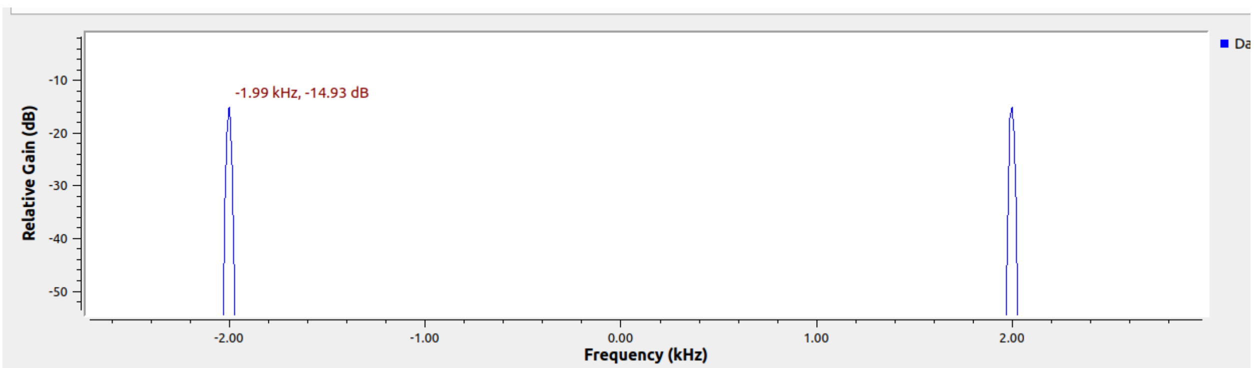
This is what the frequency and time domains look like when sampling at 10kHz.

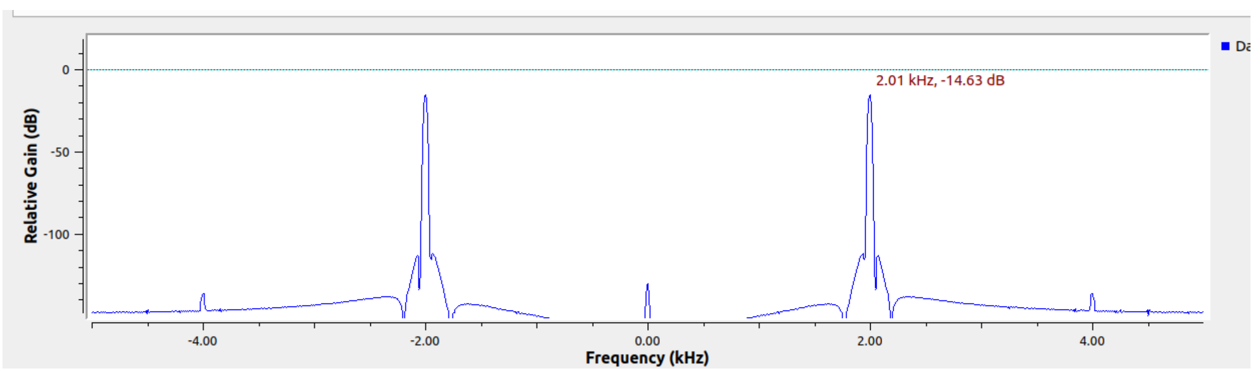


This is the time domain when the sampling rate is 10kHz.

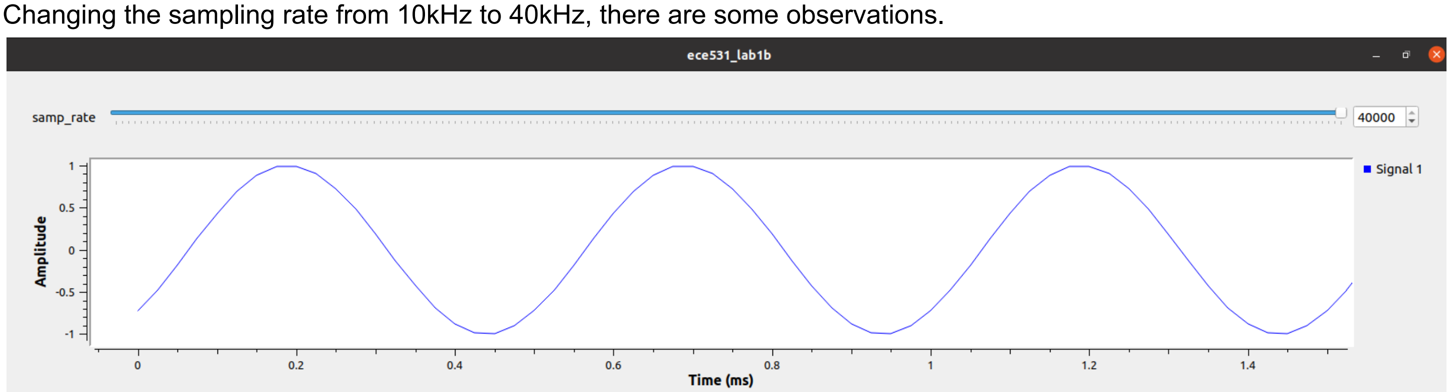


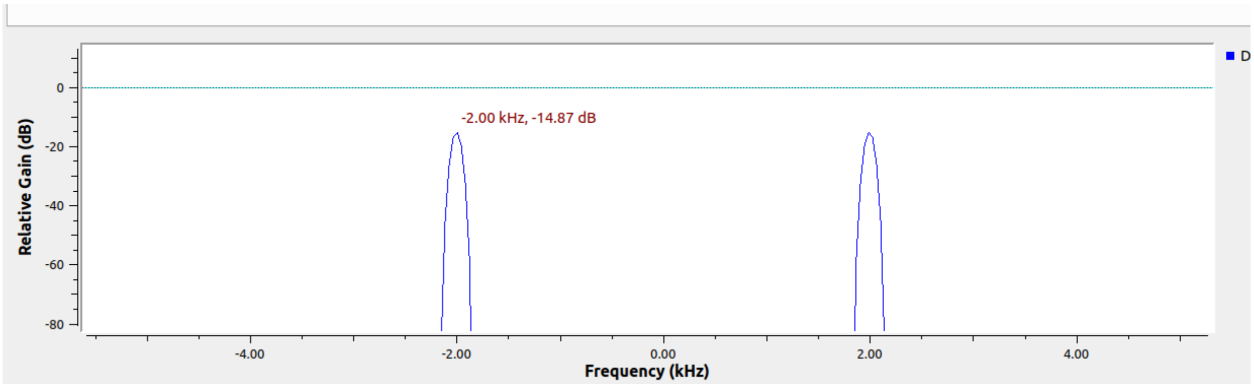
The current sample rate is 10 samples/1ms, because of the small sample rate we do not receive an accurate image of the 2k sine wave, we will need to increase this to have a more accurate and detailed sample. Also the bandwidth would be (½)sampling rate, which is 5000 Hz = 5kHz, which is the range displayed in the frequency tab.

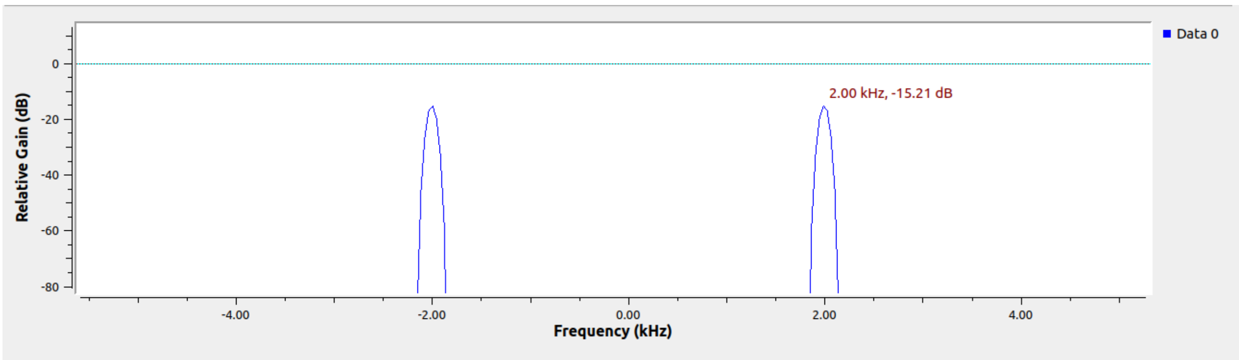


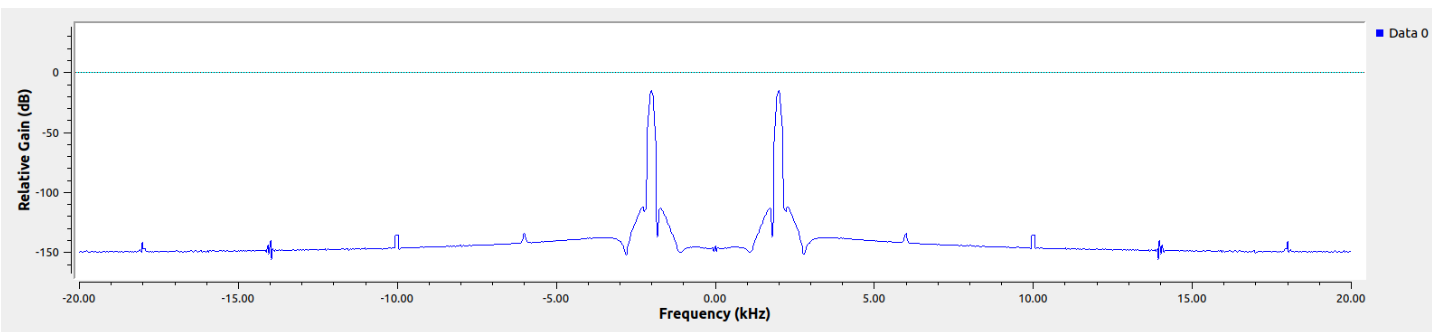


Changing the sampling rate from 10kHz to 40kHz, there are some observations.



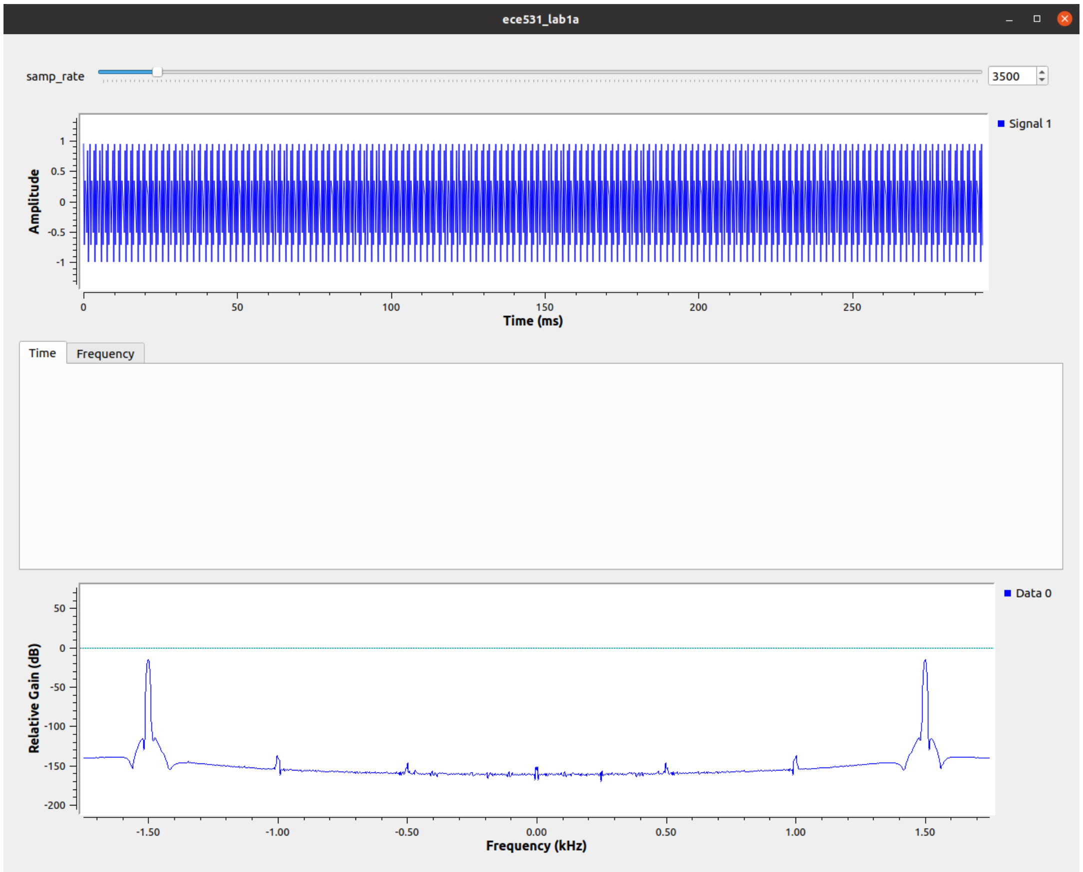


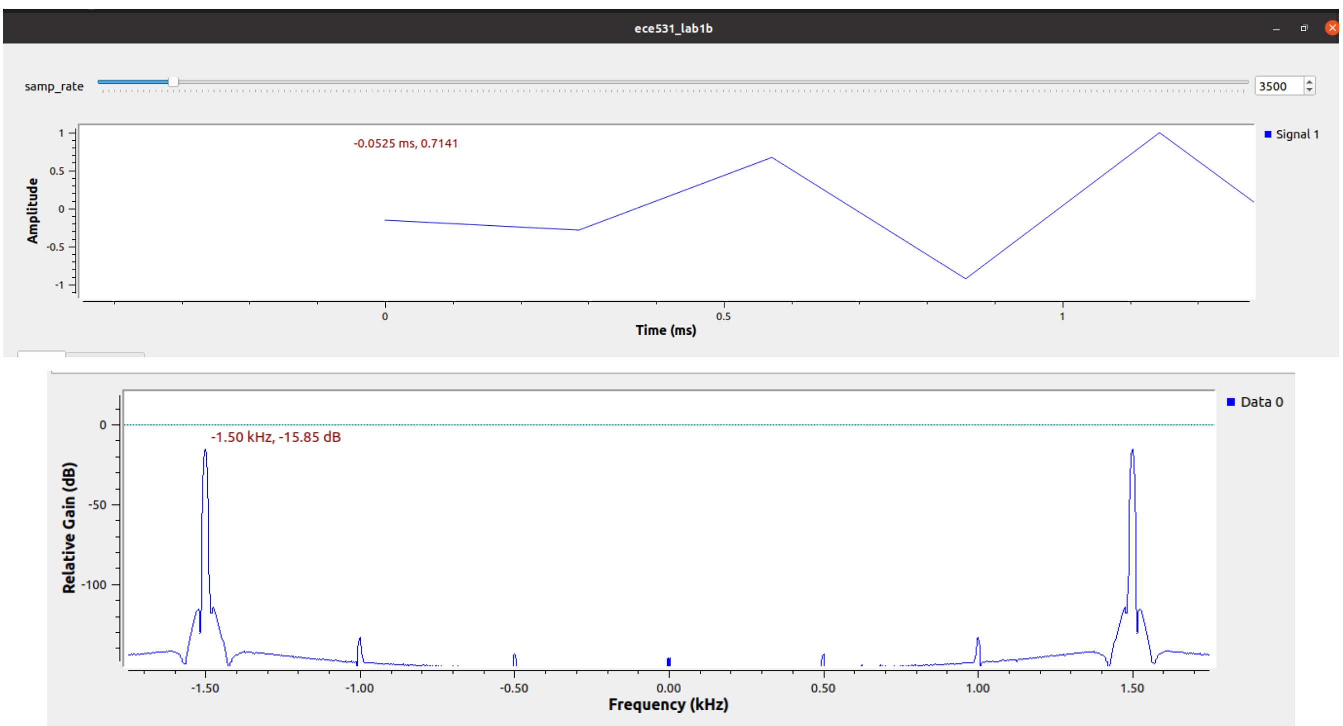


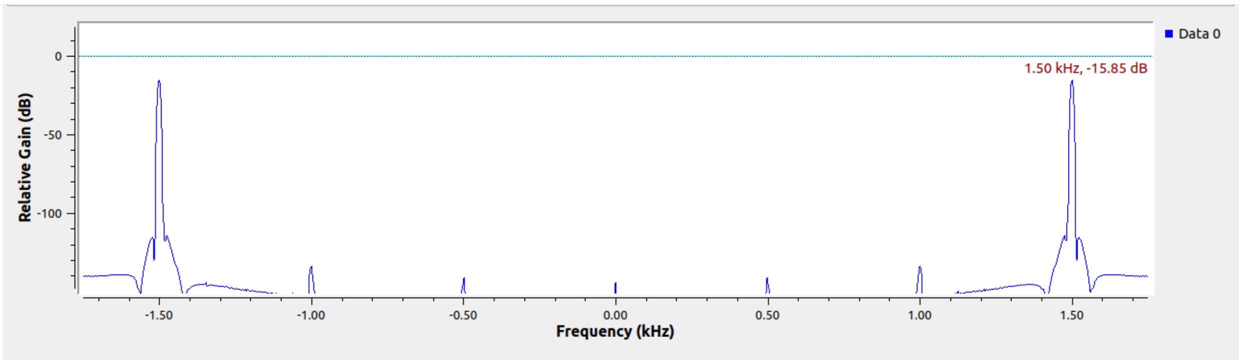


Since the sampling rate is 40 samples/1ms, and the bandwidth is (½)sampling rate, (½)40000 Hz = 20000 Hz = 20kHz
The time domain when the sampling rate is at 40kHz, looks like a more accurate sampling of the 2k sine wave. The current sample rate is now 40 samples/1ms, producing a detailed sample of the sine wave.

Changing the sampling rate from 40kHz to 3.5kHz, there are some observations.


ece531_lab1a
samp_rate
3500
Signal 1
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)


ece531_lab1b
samp_rate
3500
Signal 1
-0.0525 ms, 0.7141
Amplitude
Time (ms)
Data 0
-1.50 kHz, -15.85 dB
Relative Gain (dB)
Frequency (kHz)

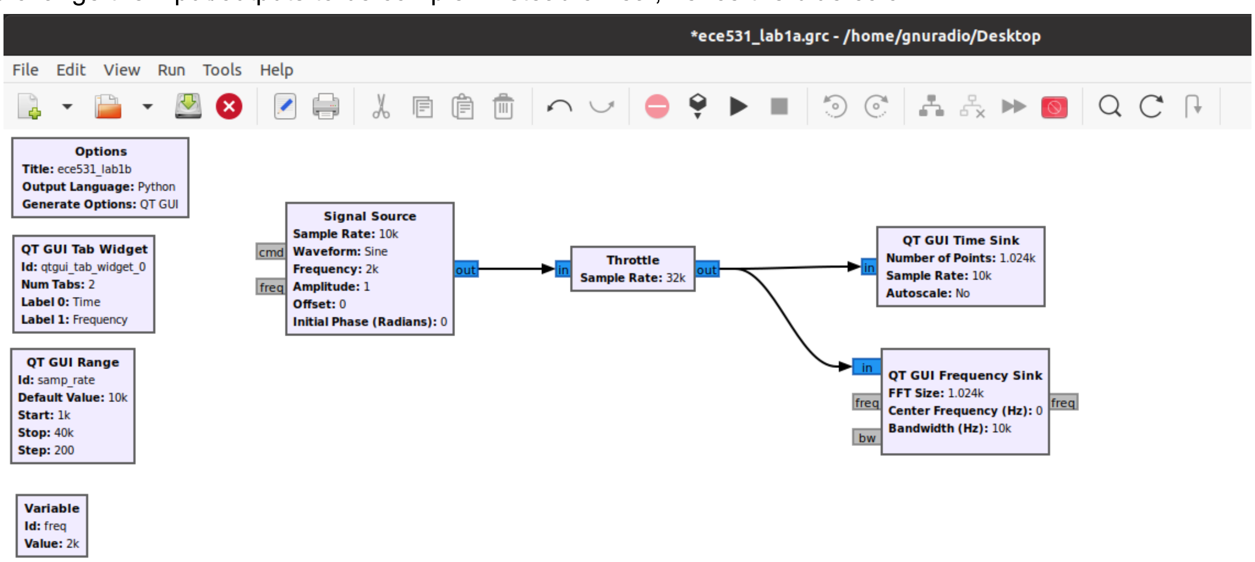


The sample rate is now 3.5 samples/1ms, this is an extremely low sampling rate and will not produce an accurate result of the time domain. The measured frequency and bandwidth is (½)sampling rate = (½)3500 Hz = 1500 Hz = 1.5kHz.

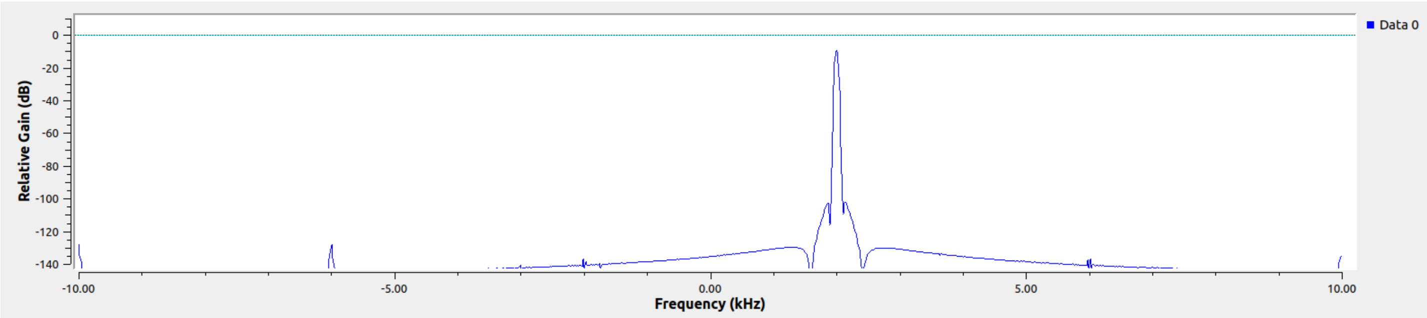
Then save the diagram and the flowgraph with a unique name.

3.3 COMPLEX SAMPLING

Here we change the input/outputs to be complex instead of real, hence the blue color.



Visualizing the signal generated in the frequency domain.



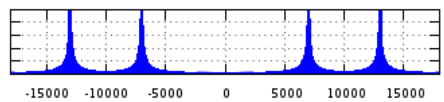
From the textbook Software Defined Radio for Engineers: “Frequency domain plot provides the spectrum of the phasor but only communicates magnitude and loses phase information. This happens because we are only plotting the real component of the spectrum”. This means we are only viewing the real values, not the imaginary ones, hence the [0,fs/2].

Mixing and multiplying signals

Using real signals or IQ Signals gives different results when you multiply them. This is because using only the real component it's not possible to uniquely determine the phase angle of the signal, hence impossible to distinguish a positive frequency from a negative.

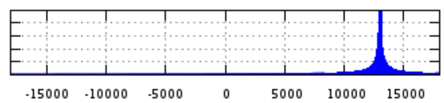
Multiplying two signals f1 and f2 in the real domain:

$$\pm f1 \otimes \pm f2 = (\pm) f1 \pm f2$$



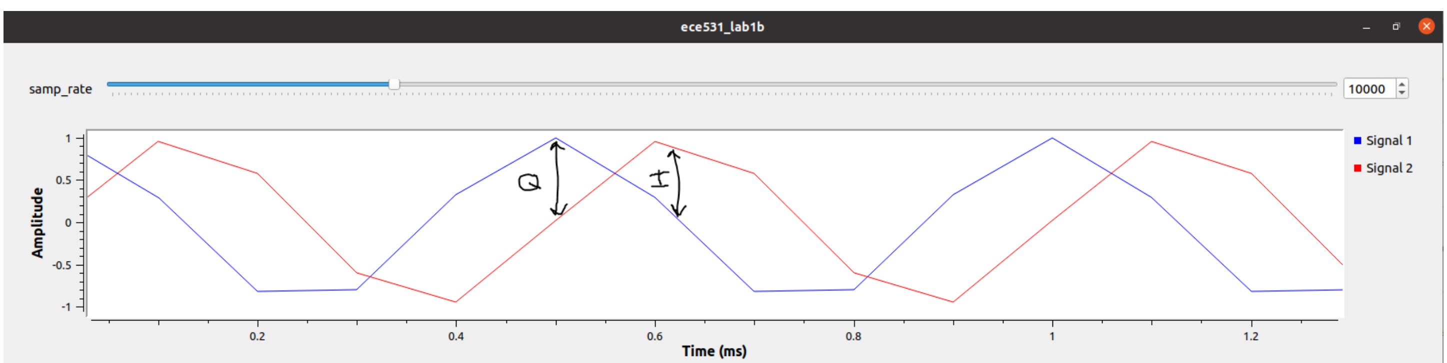
Using IQ Data the signs are now given, and the result is unambiguous:

$$f1 \otimes f2 = f1 + f2$$



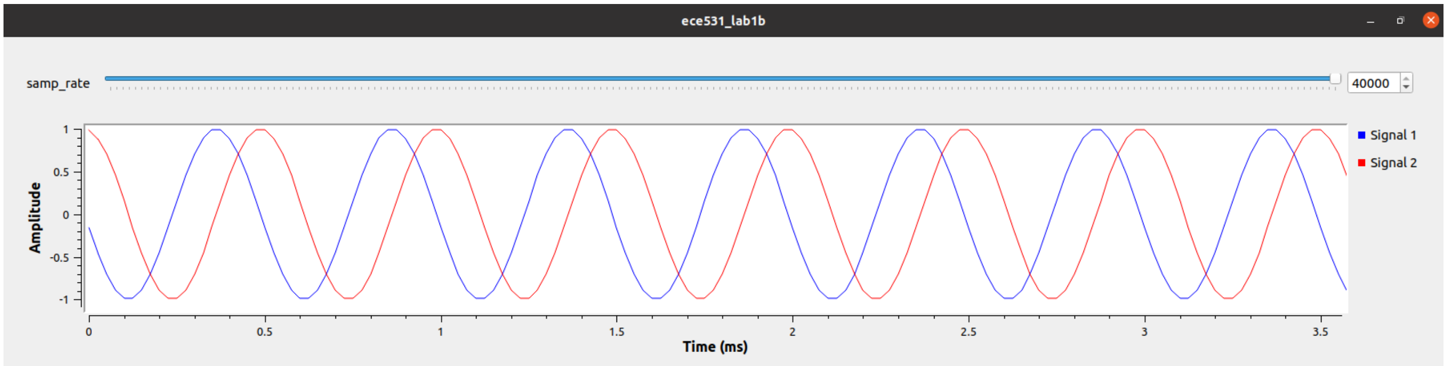
A frequency spectrum in the real domain usually never show the negative side, since it always must be symmetric around zero due to the uncertainty of the sign of the frequency of the real signal -- hence the parentheses around the sign of f1 in the first formula mixing the real signals. I've included the negative side here for illustrative purposes, despite of its redundancy.

3.

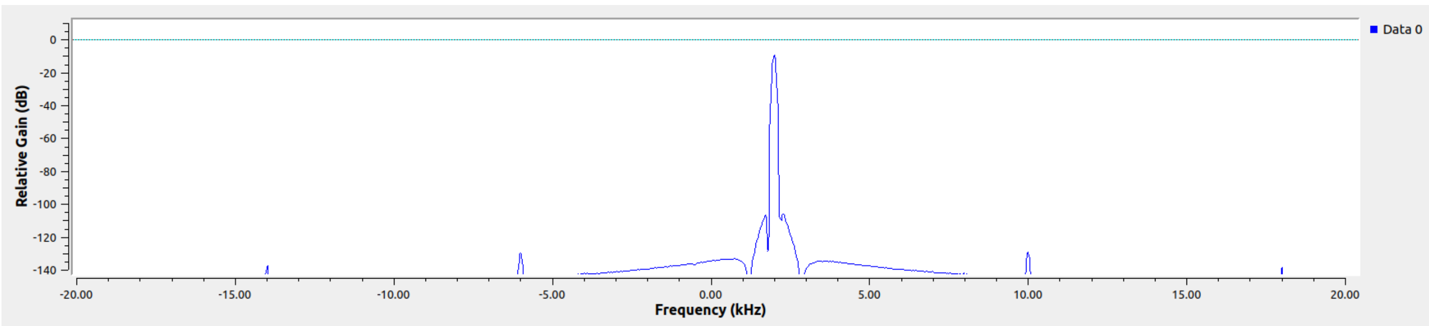


Signal 1 is -sin(f0) and Signal 2 is cos(f0). The phase relationship is that -sin(f0) is shifted 90 degrees of pi/2 of cos(f0), hence cos(f0+pi/2) = -sin(f0).

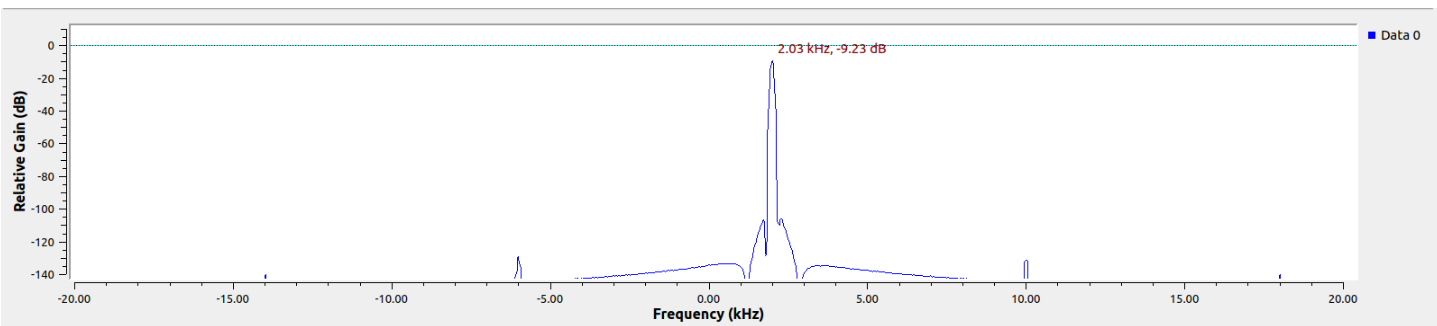
4.



The amplitude of the signal is not affected by the change of the sample rate when increasing it, only the clarity of the sample itself, the amplitudes are closer to each other I = Q.

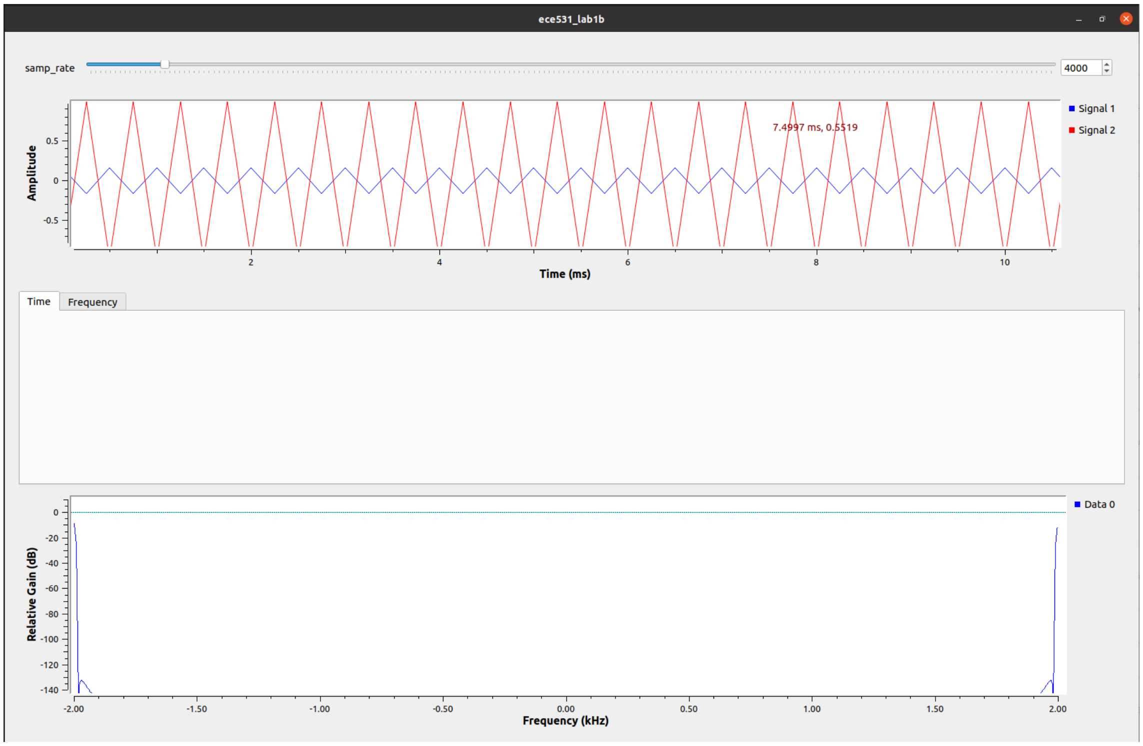


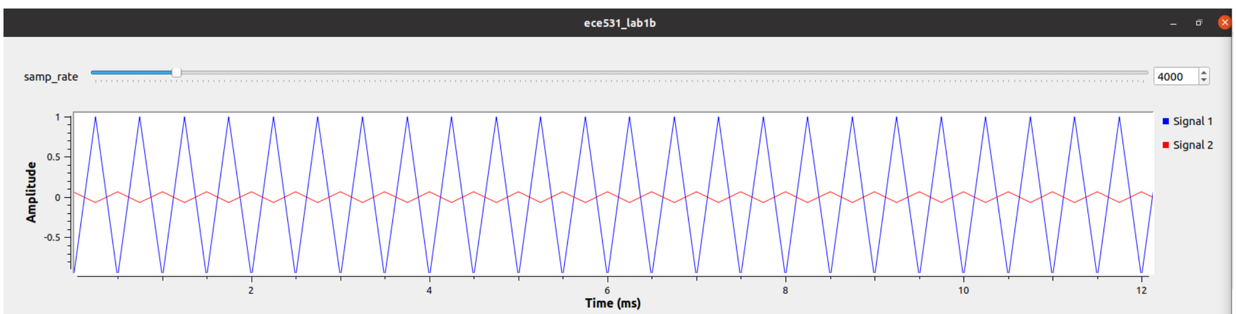
5.



Since the sampling rate is 40 samples/1ms, and the bandwidth is (½)sampling rate, (½)40000 Hz = 20000 Hz = 20kHz

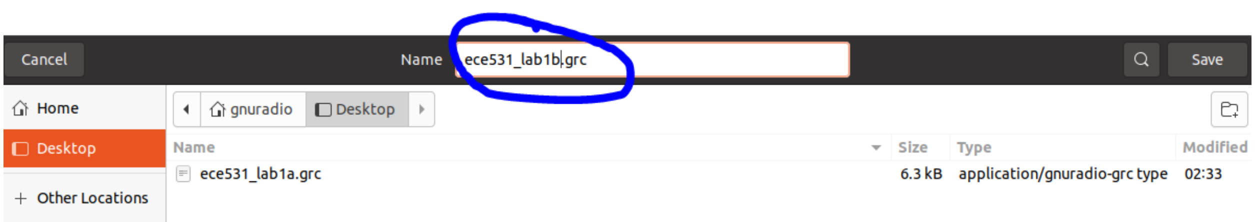
6.





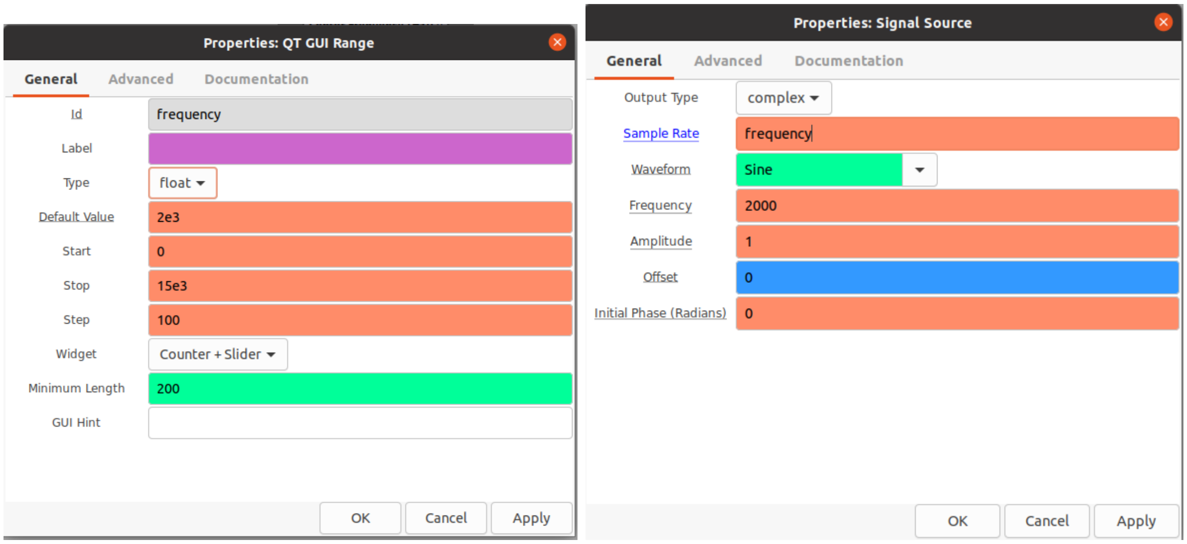
When repeatedly setting the sampling rate to be 4kHz, the amplitude of either the sine wave or cosine wave is significantly reduced. Since the bandwidth is now set to be 2kHz (4000 Hz/2 = 2000 Hz = 2kHz), the frequency domain is trying to determine which is the real number in the signal; hence we have the frequency "split" in the middle in the negative and positive of the domains.

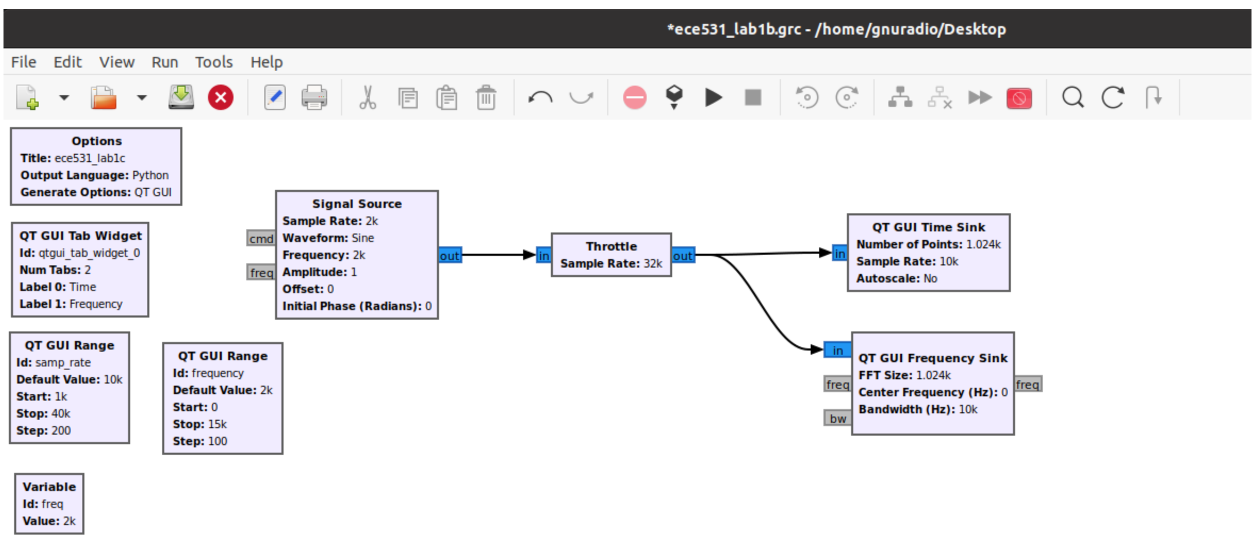
7.



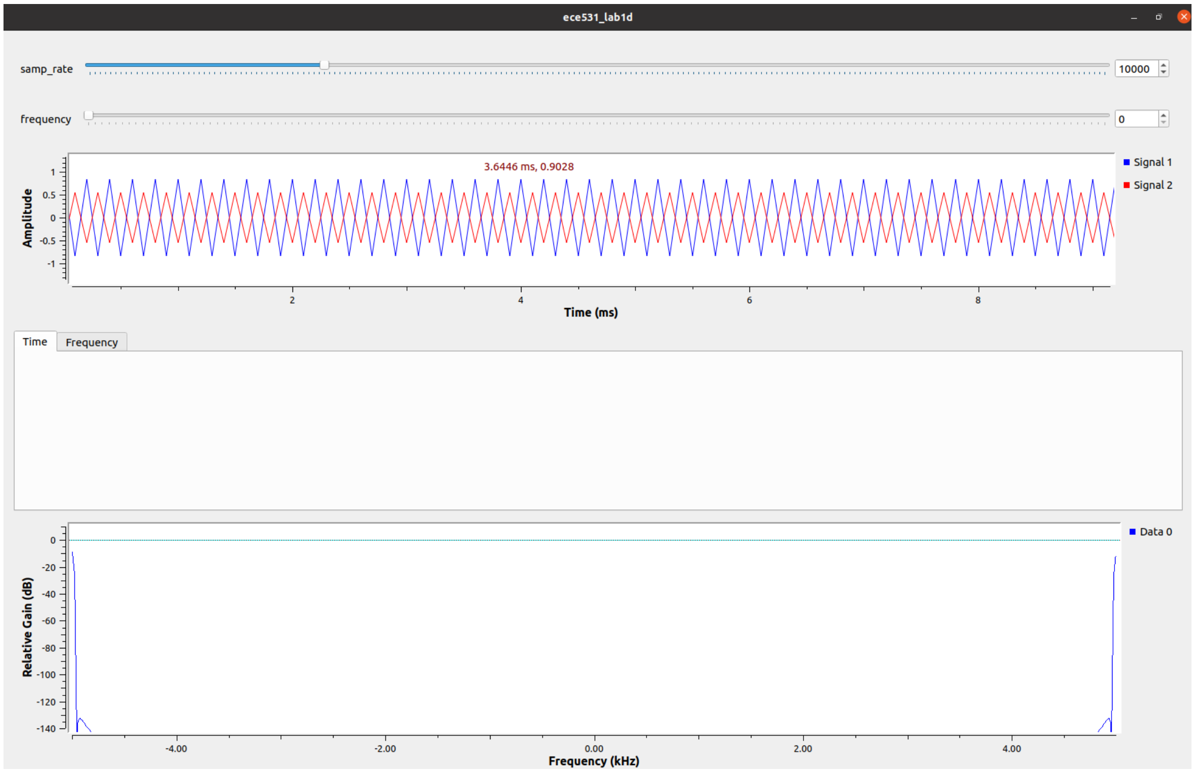
3.4.1 COMPLEX-SAMPLED FLOWGRAPH

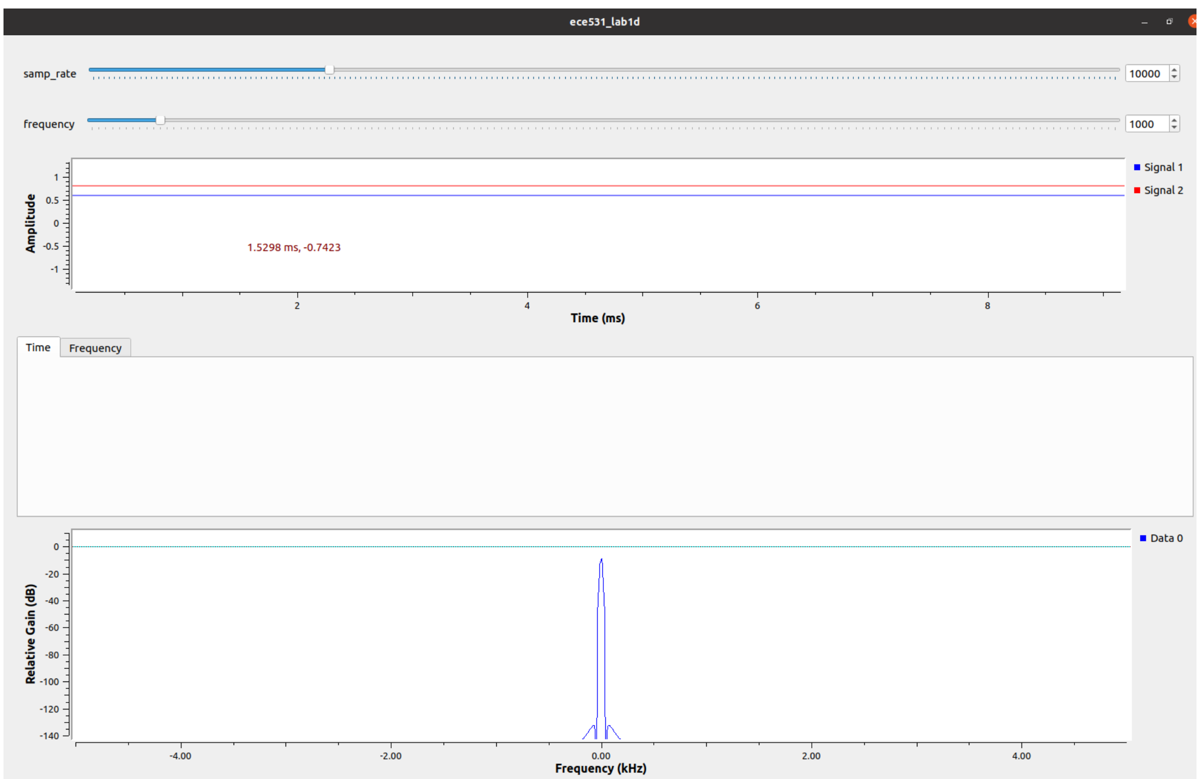
1.

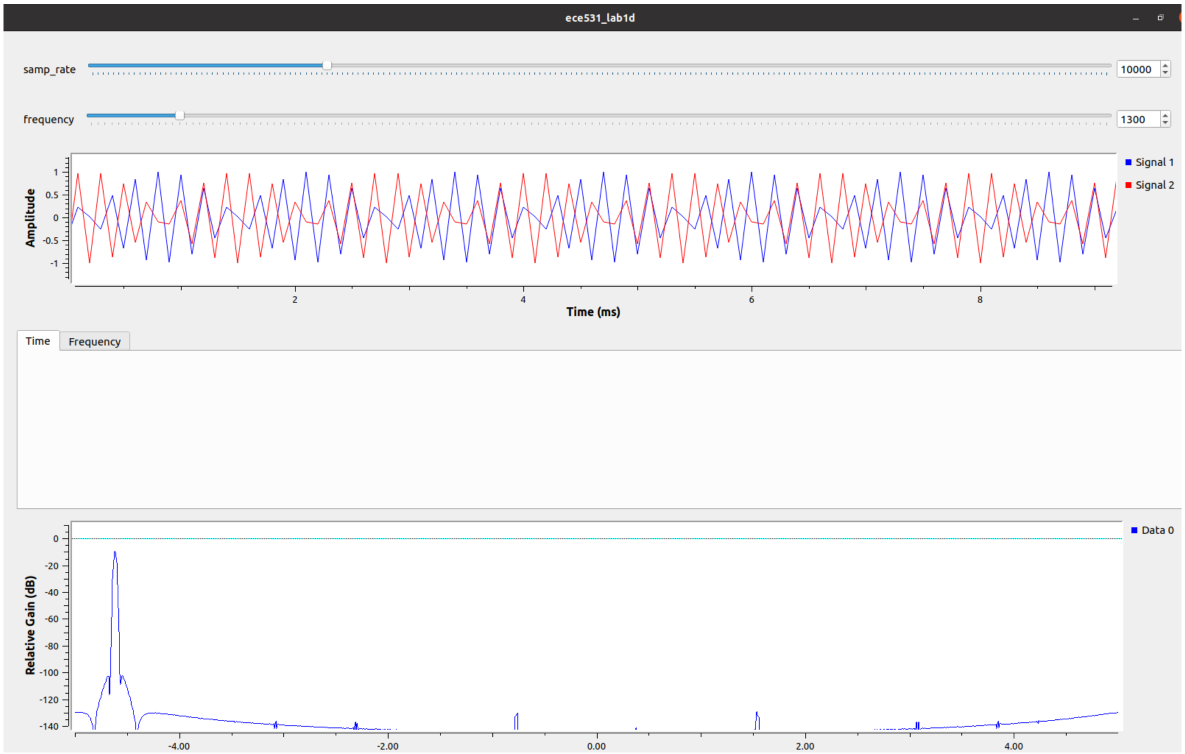


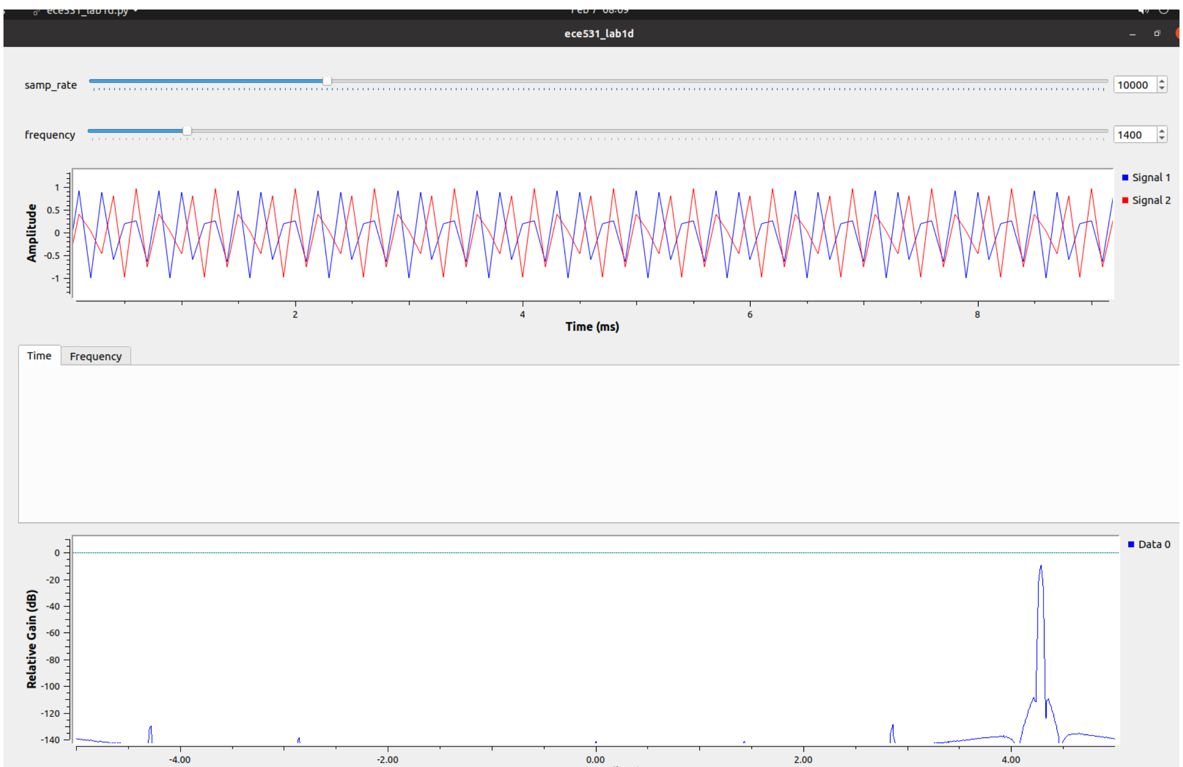


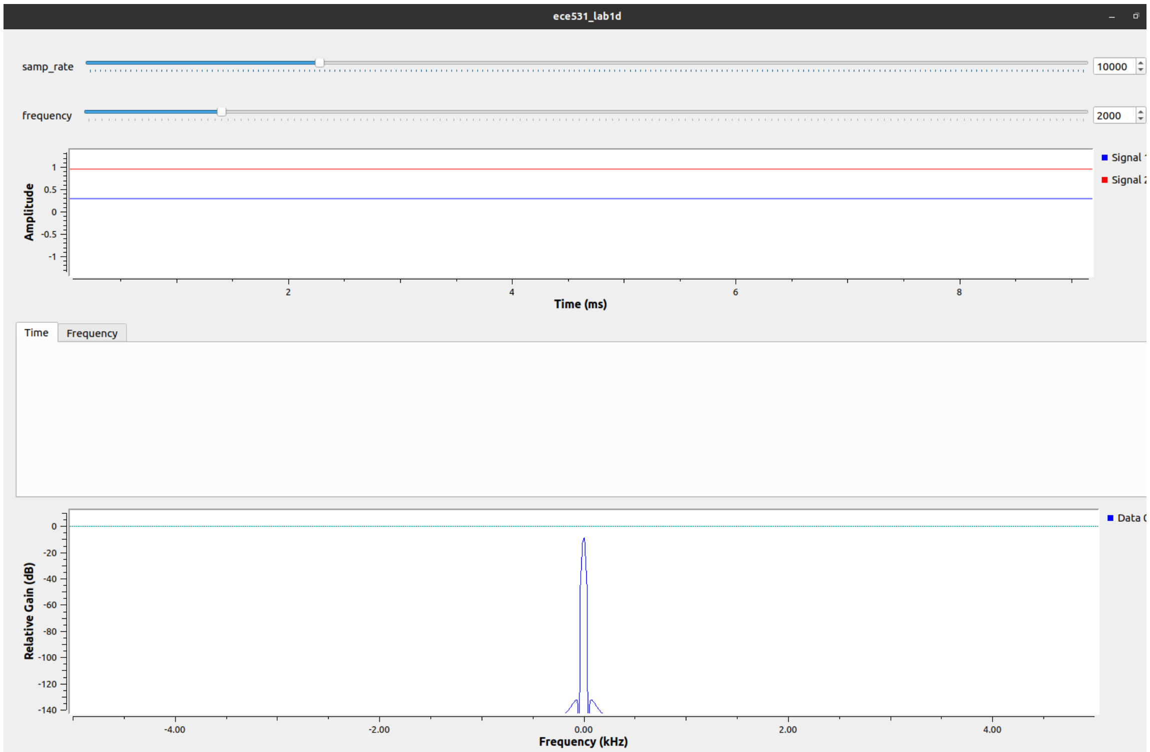
3.

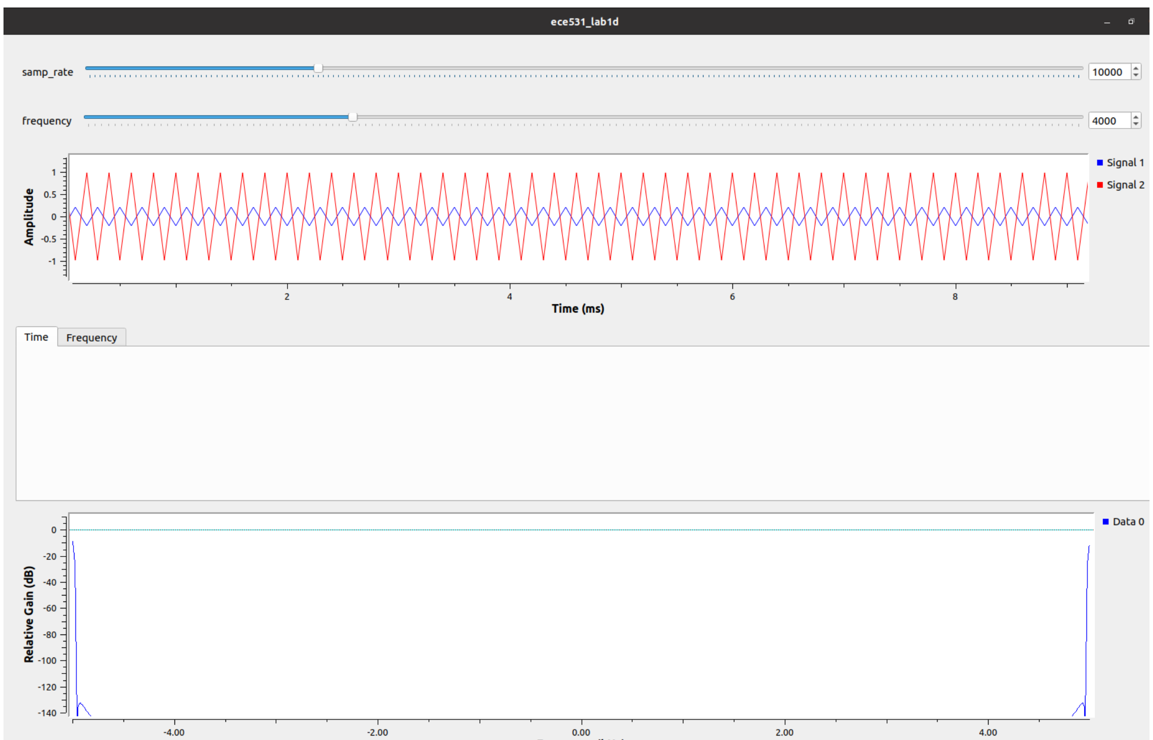

ece531_lab1d
samp_rate
10000
frequency
0
3.6446 ms, 0.9028
Signal 1
Signal 2
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)

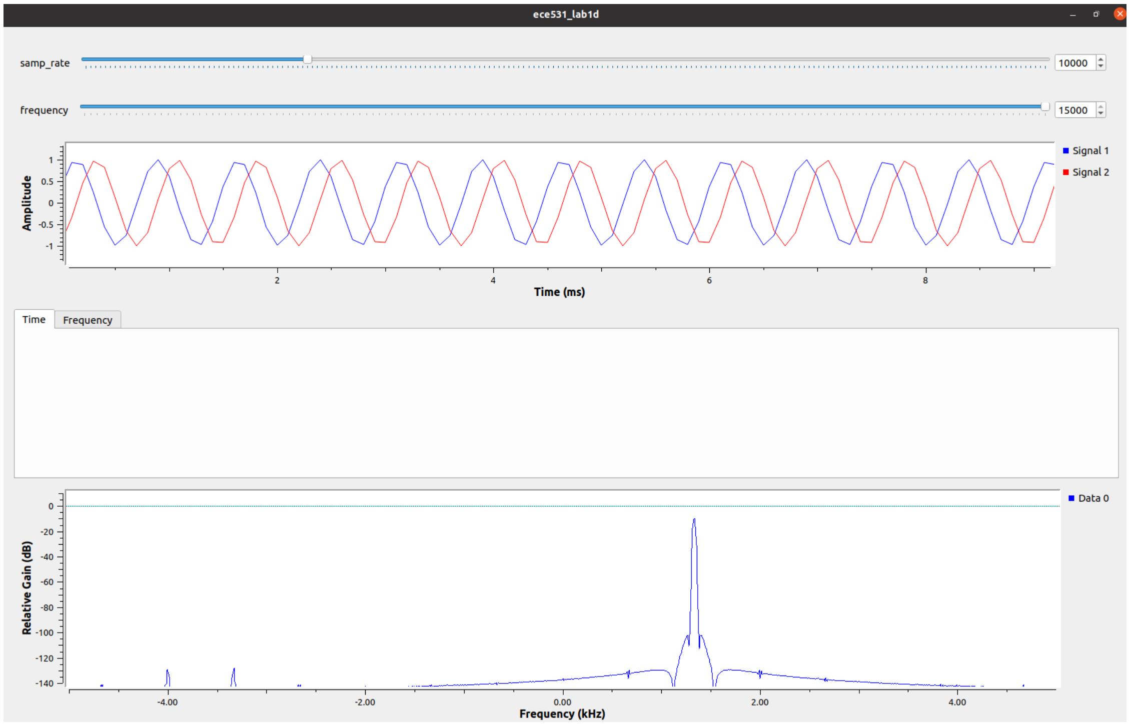

ece531_lab1d
samp_rate
10000
frequency
1000
Signal 1
Signal 2
1.5298 ms, -0.7423
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)


ece531_lab1d
samp_rate
10000
frequency
1300
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0


ece531_lab1d
samp_rate
10000
frequency
1400
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)


ece531_lab1d
samp_rate
10000
frequency
2000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)


ece531_lab1d
samp_rate
10000
frequency
4000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)

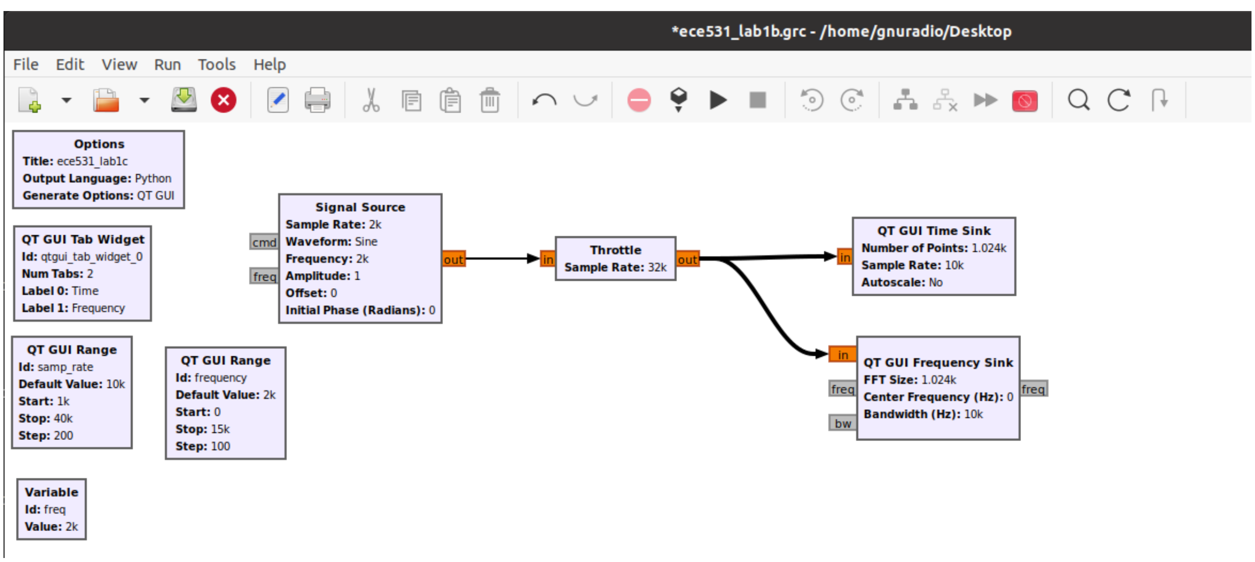


Slowly shifts left from 4000 to 15000, does not loop around as it did the previous shifting rates.

4. Looking at the time domain, we see the frequency folds over from the negative side of the bandwidth to the positive side of the bandwidth. This anomaly is called aliasing, when a signal becomes indistinguishable when being sampled [17].

3.4.2 REAL-SAMPLED FLOWGRAPH

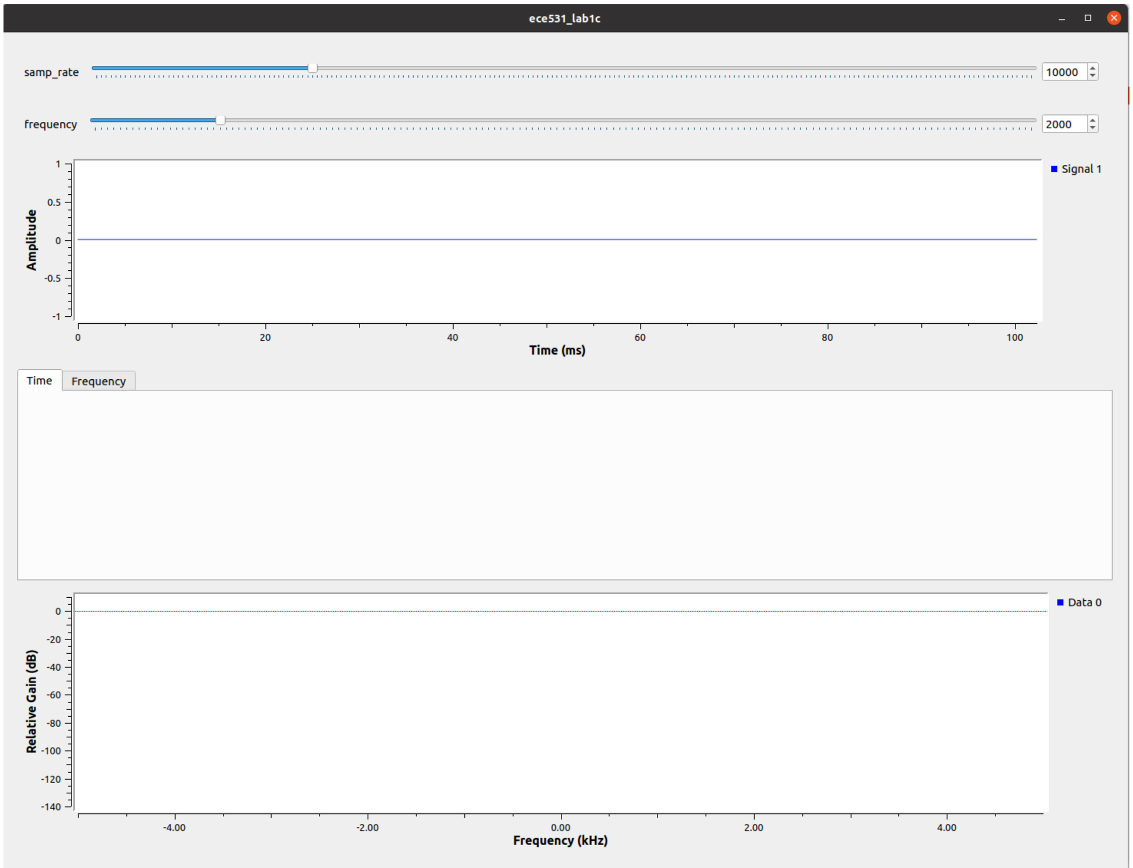
1.

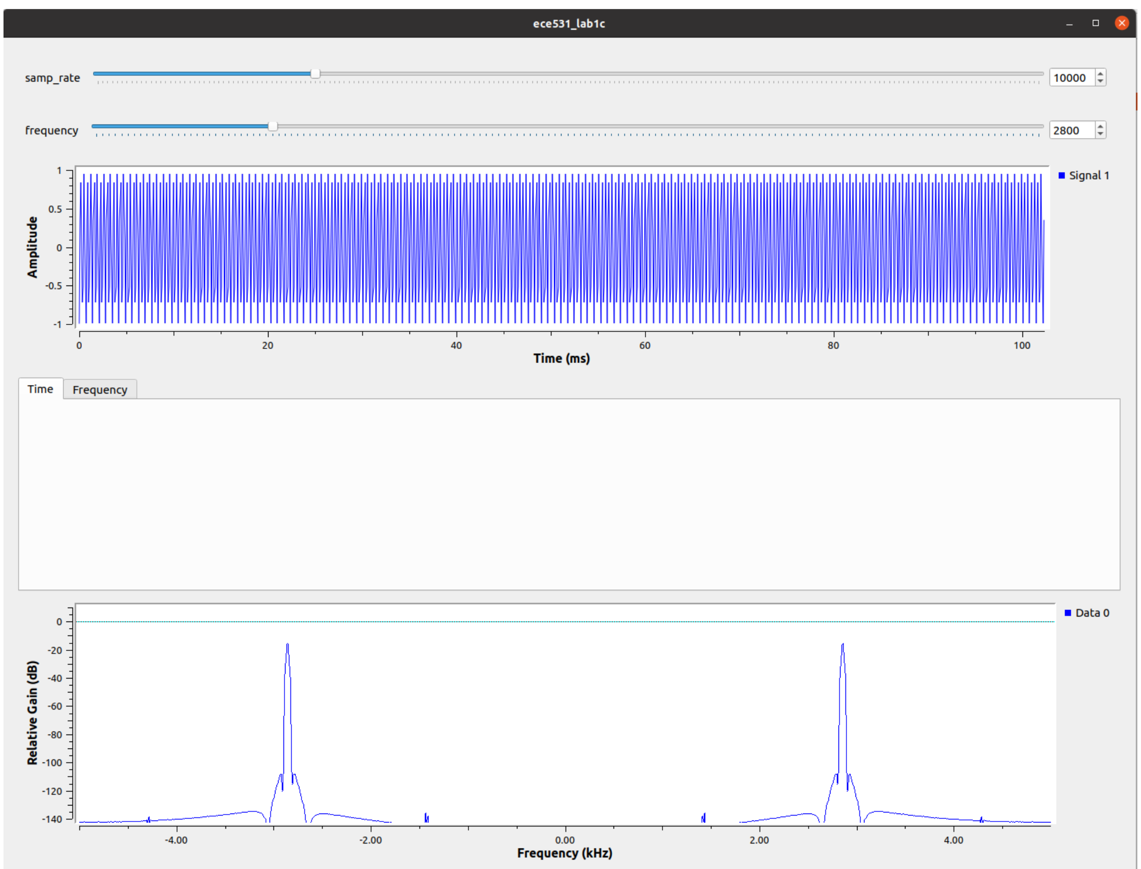


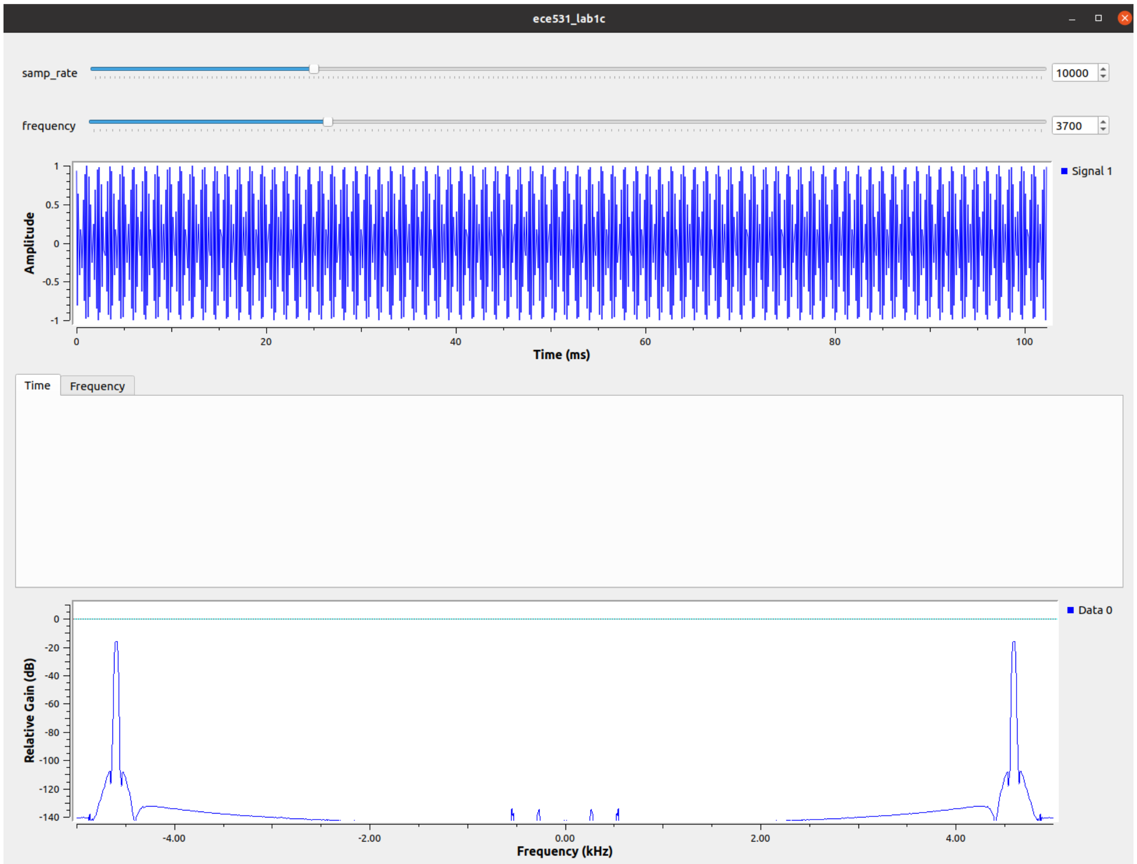
2.

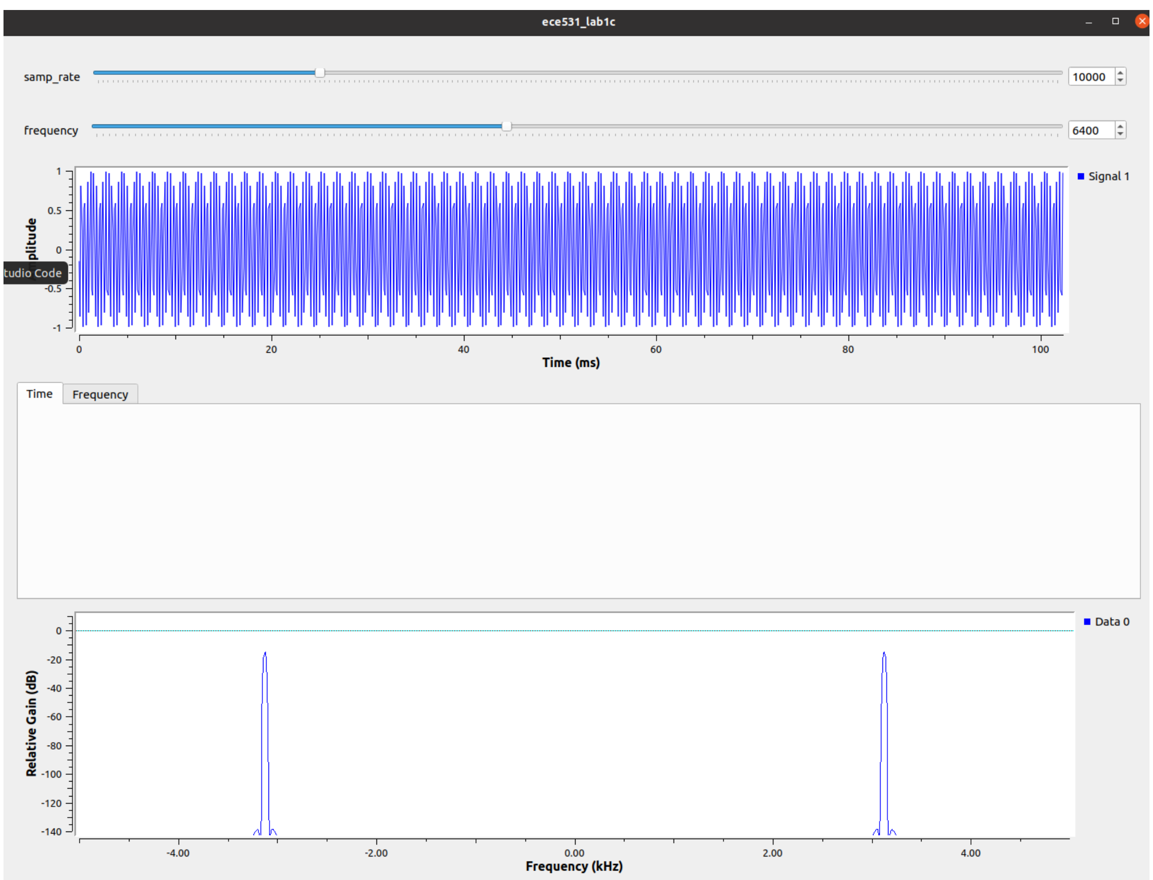
Same as above in 3.4.1 question 2.

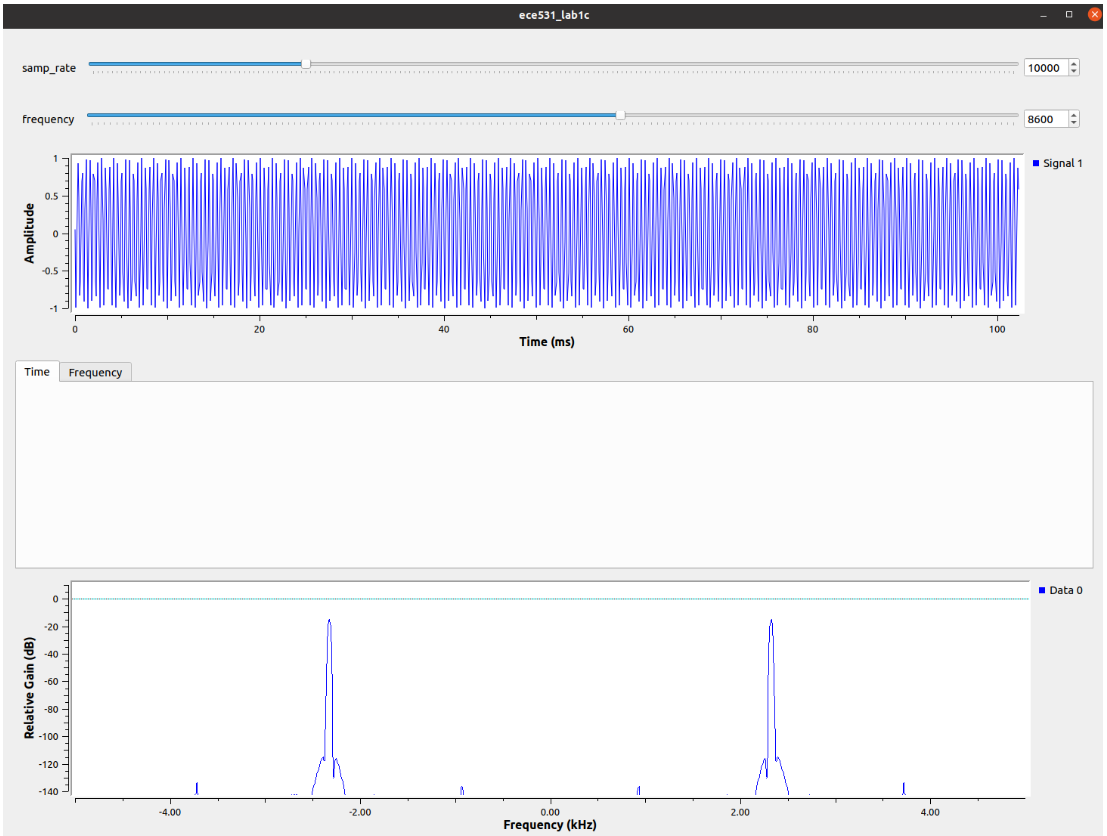
3.

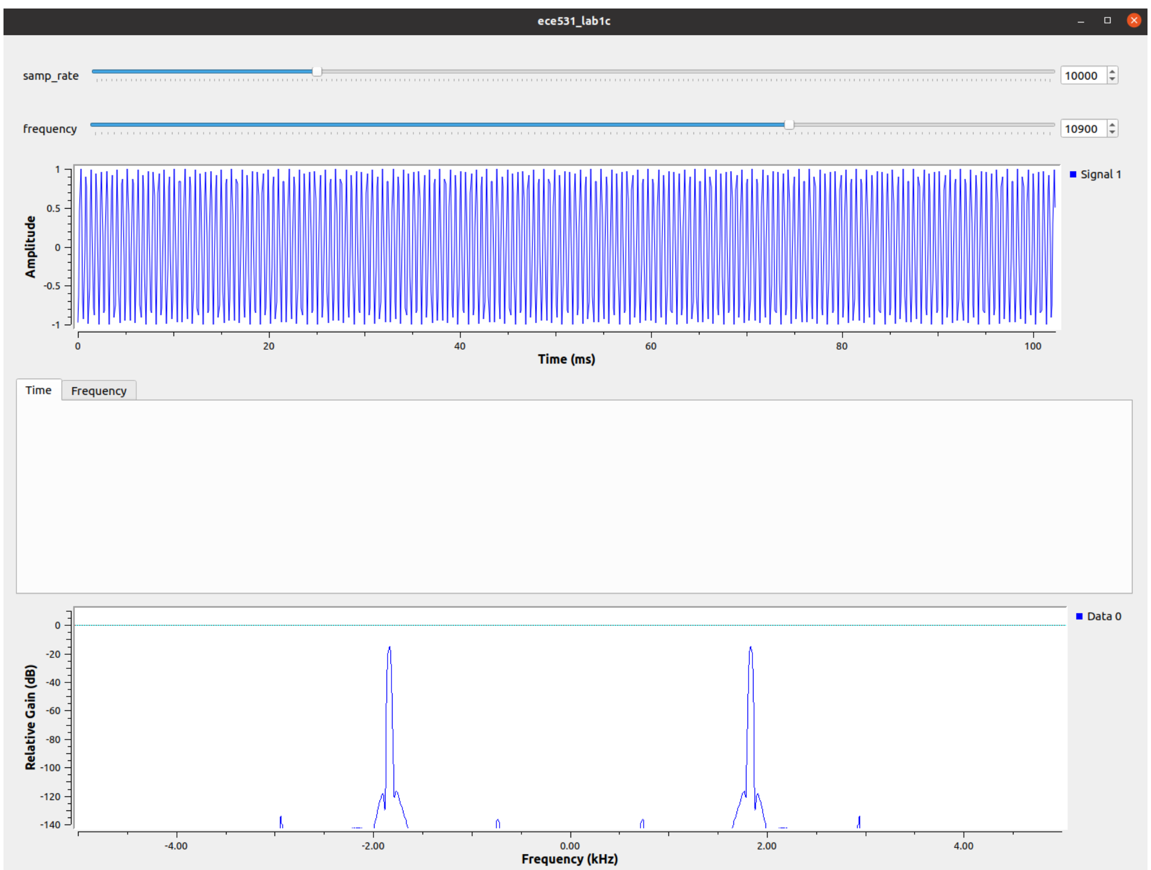

ece531_lab1c
samp_rate
10000
frequency
2000
Signal 1
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)

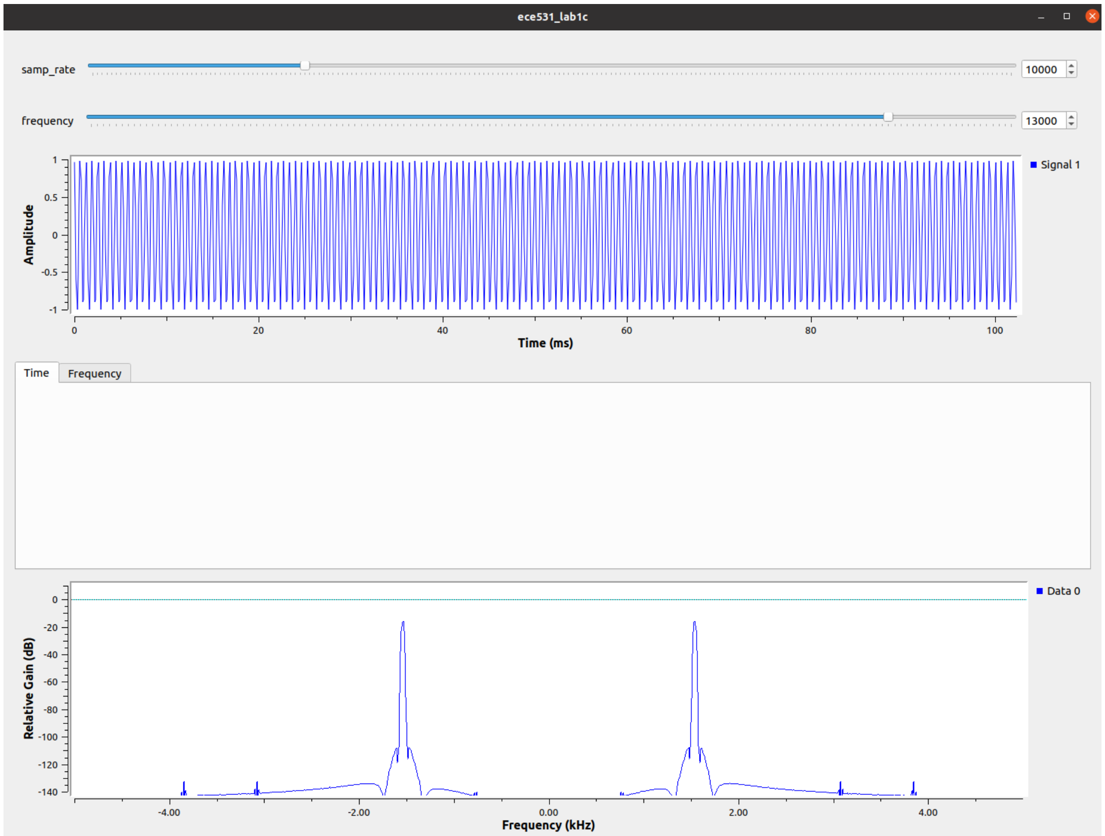

ece531_lab1c
samp_rate
10000
frequency
2800
Signal 1
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)

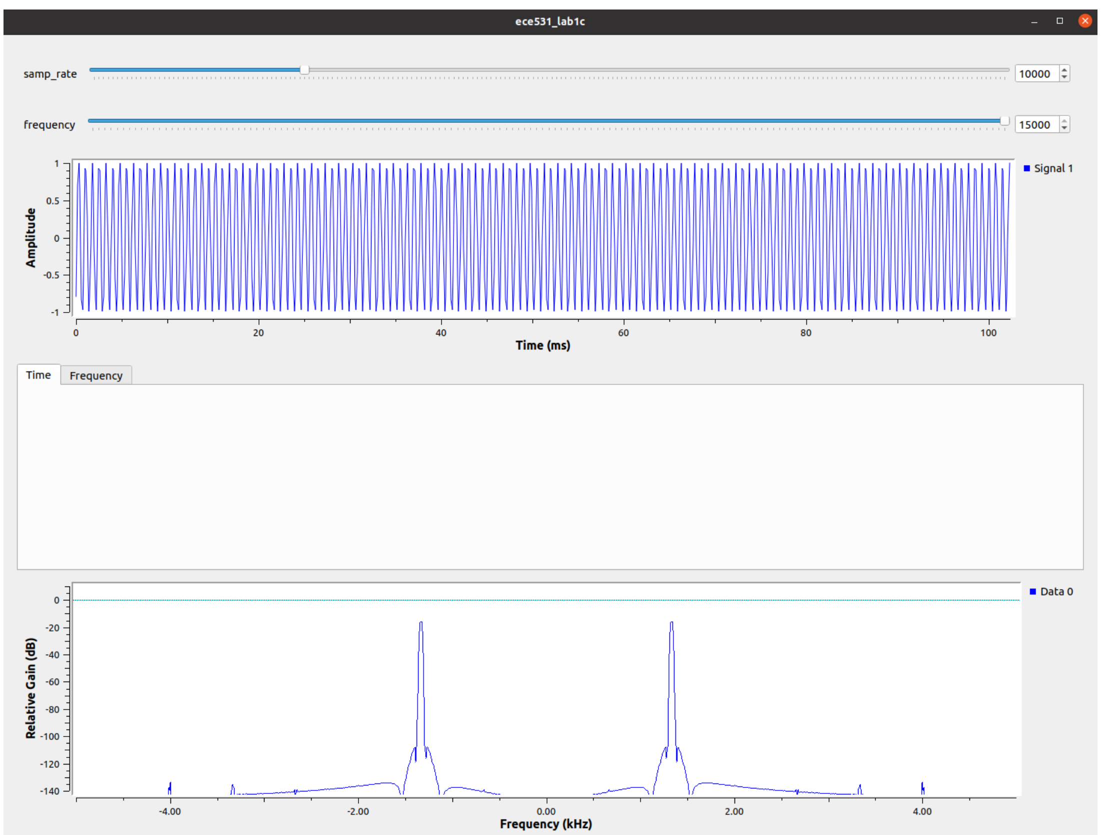

ece531_lab1c
samp_rate
10000
frequency
3700
Signal 1
Amplitude
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)


ece531_lab1c
samp_rate
10000
frequency
6400
Signal 1
Time (ms)
Time
Frequency
Data 0
Relative Gain (dB)
Frequency (kHz)


ece531_lab1c
samp_rate
10000
frequency
8600
Amplitude
Signal 1
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)

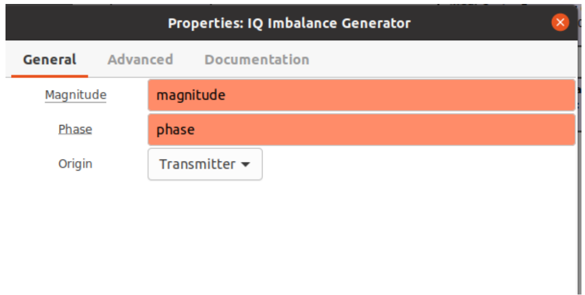

ece531_lab1c
samp_rate
10000
frequency
10900
Amplitude
Signal 1
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)

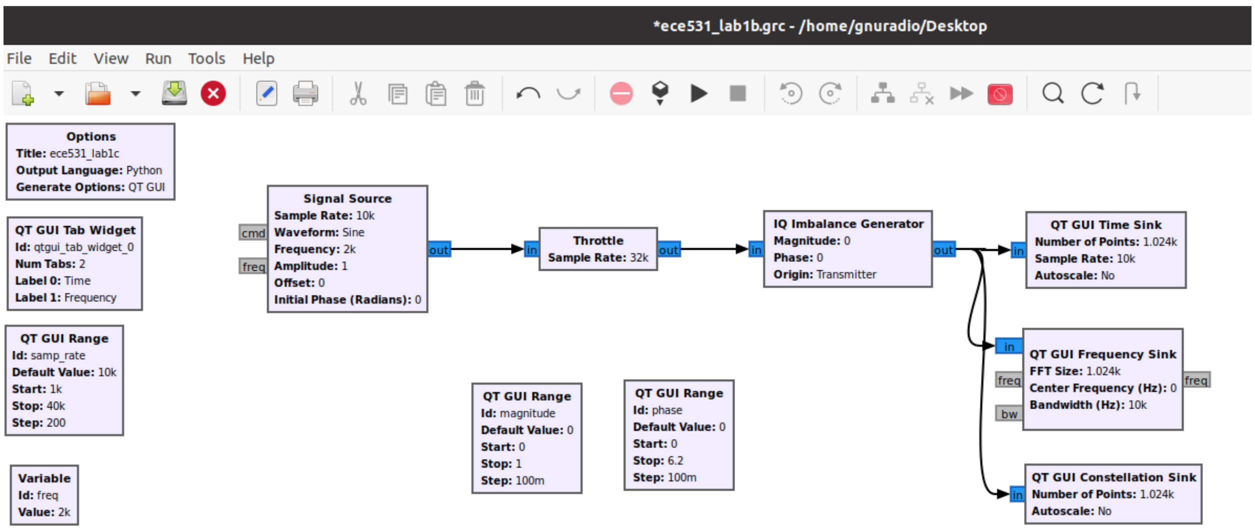




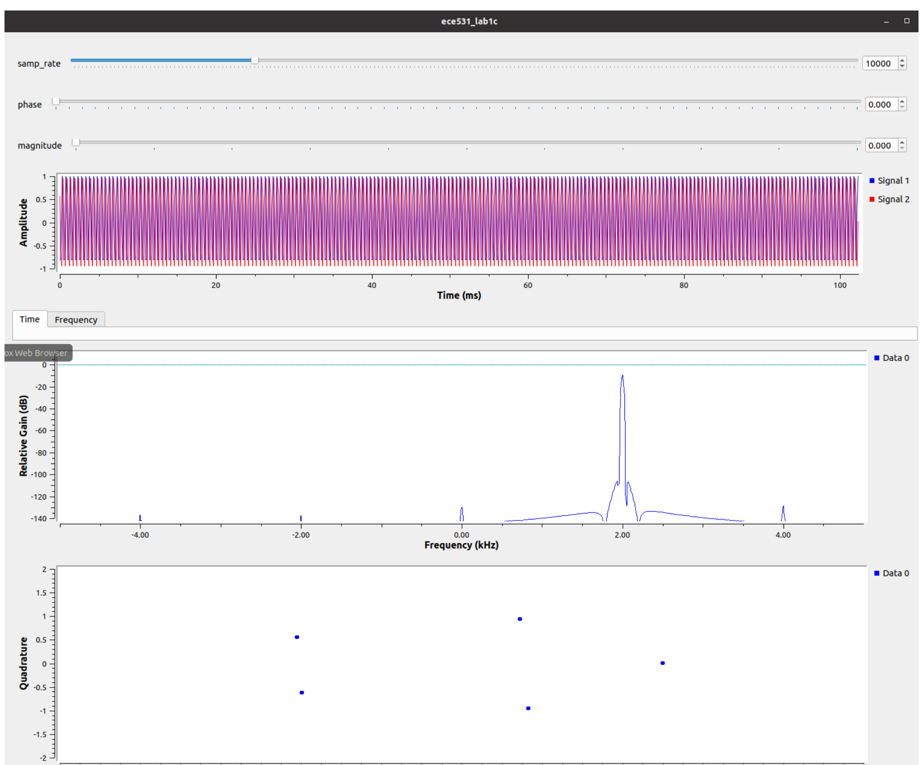
4. It seems like in the frequency domain that it "oscillates" from when the frequency is 100-200 range, it seems that the time domain oscillates from -1 to 1, which changes the gain of the frequency peak. After the 4000 range, the 2 signal peaks gradually get closer to one another. This anomaly is called folding, there is a mirror/symmetry being done on the [0 to the sampling rate] [17].

3.5 I/Q IMBALANCE

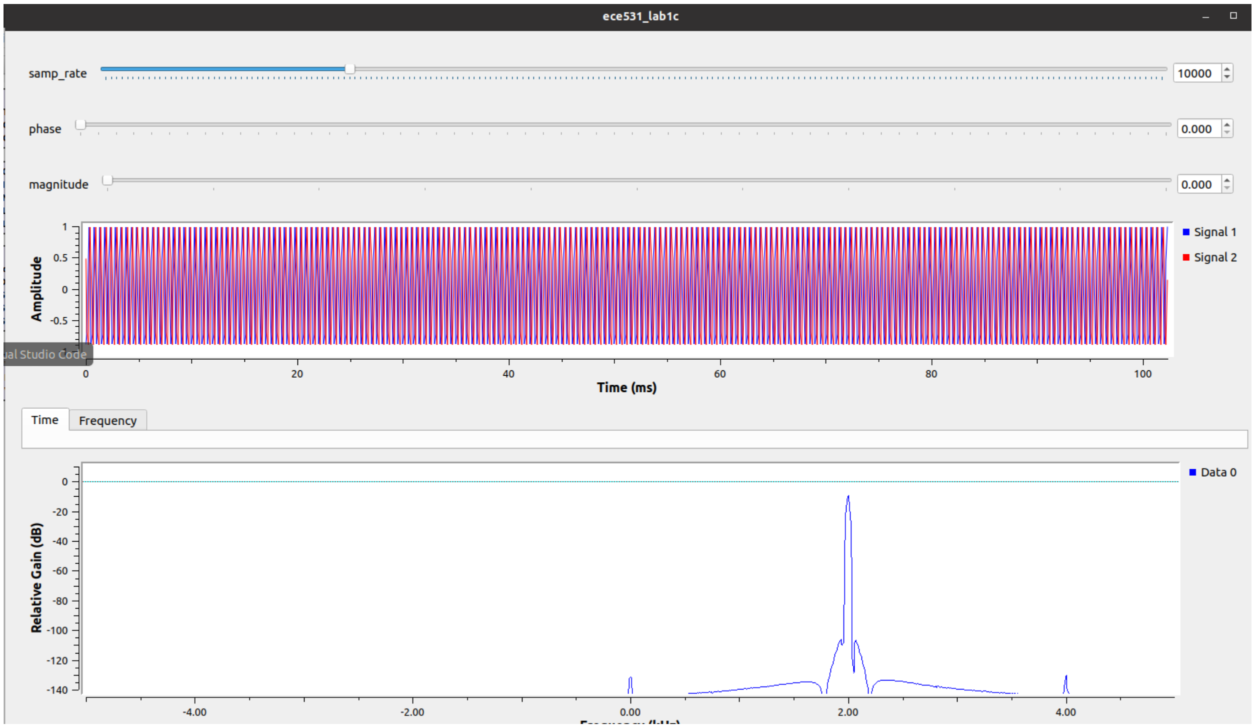


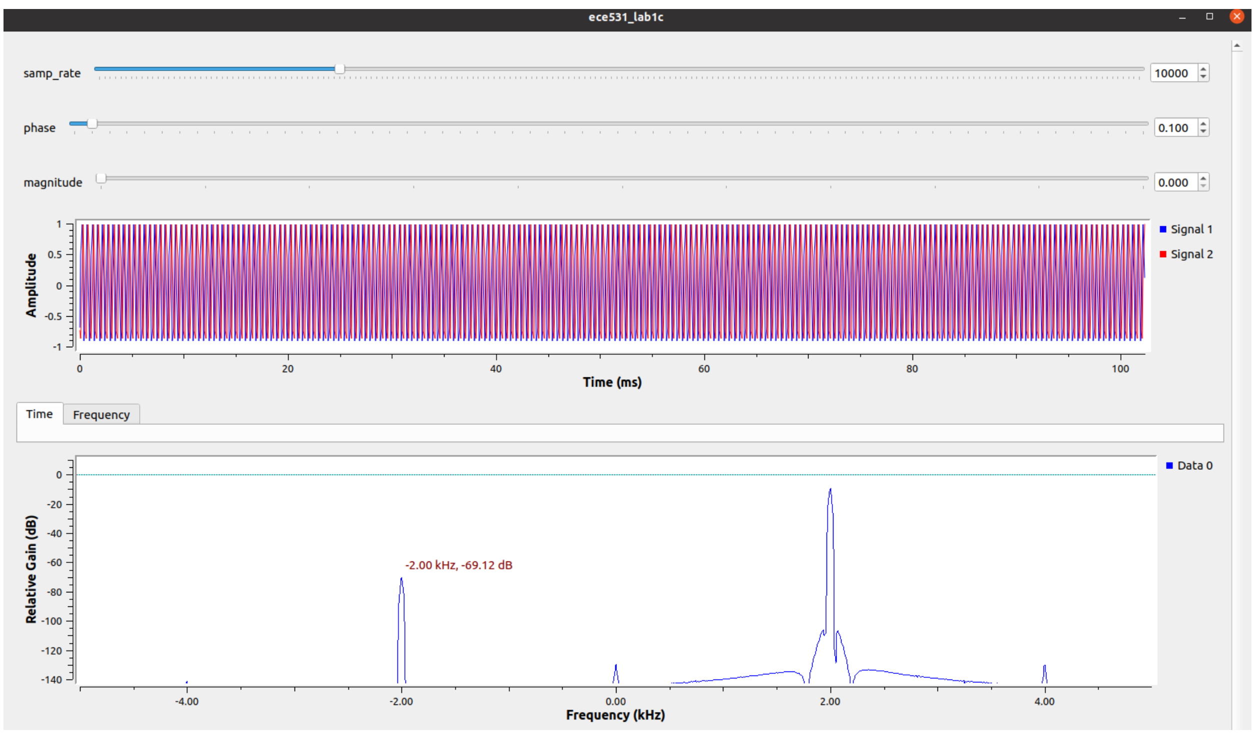


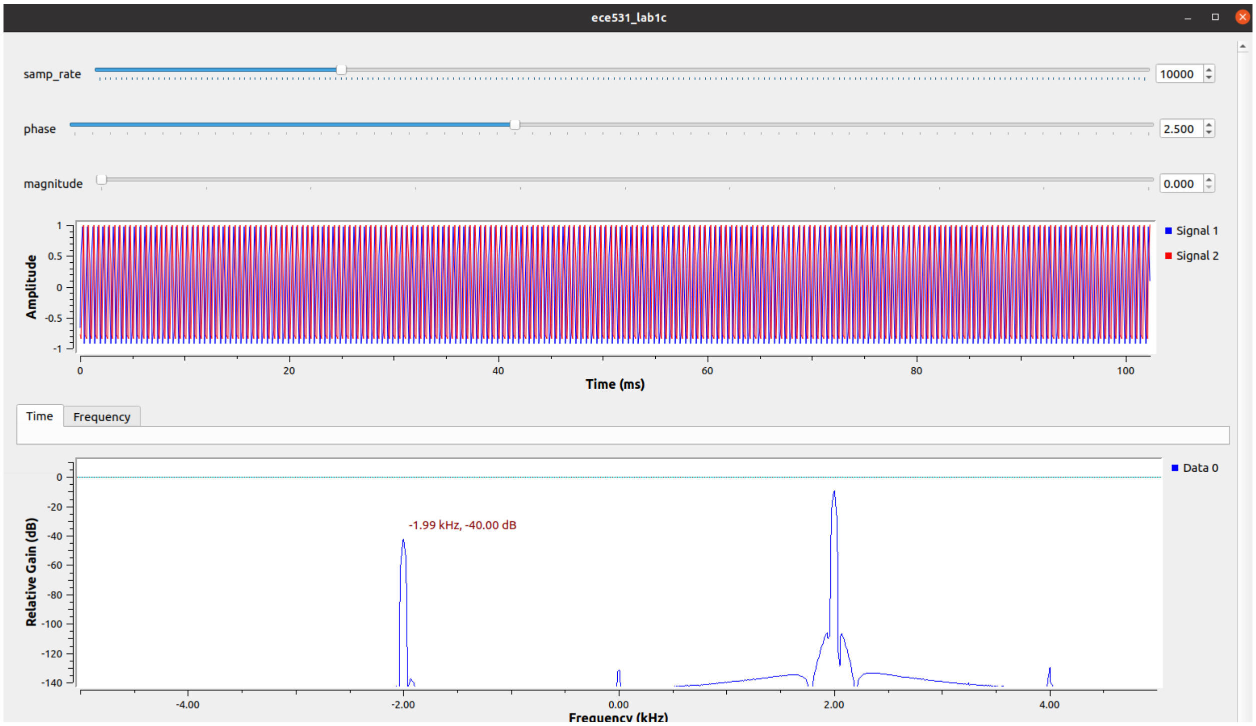
6.

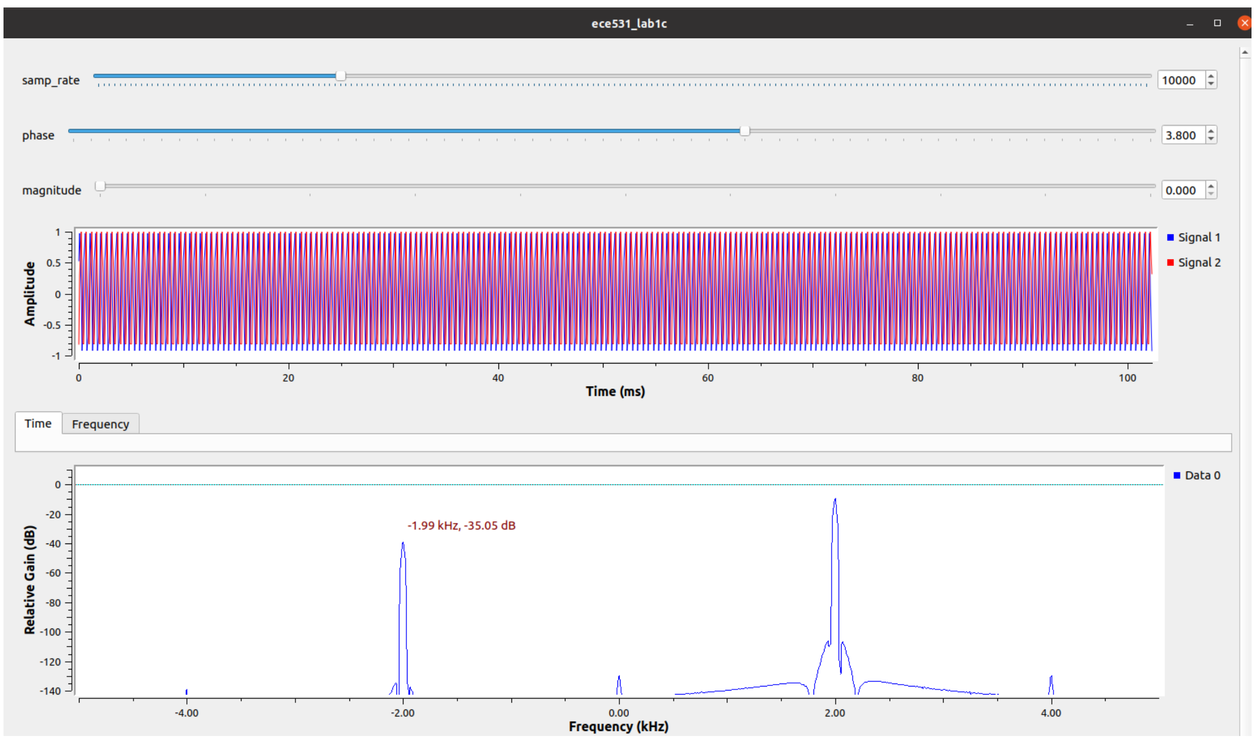


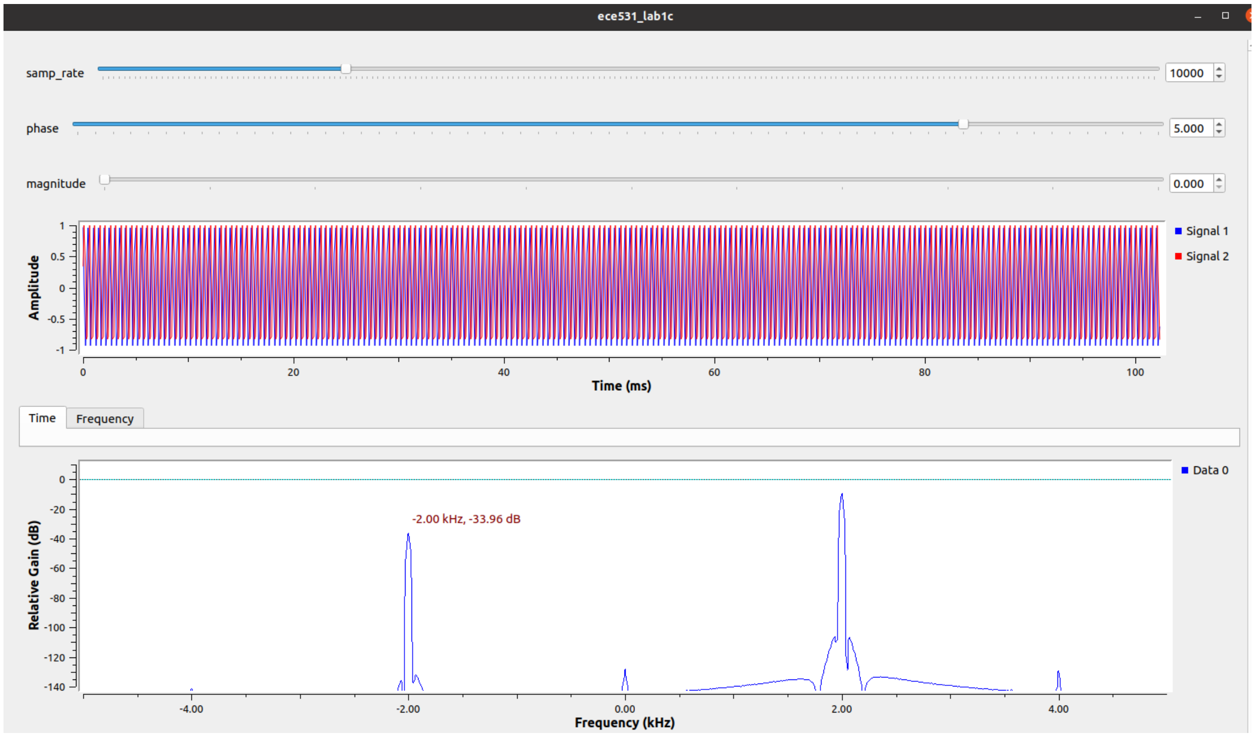
7.

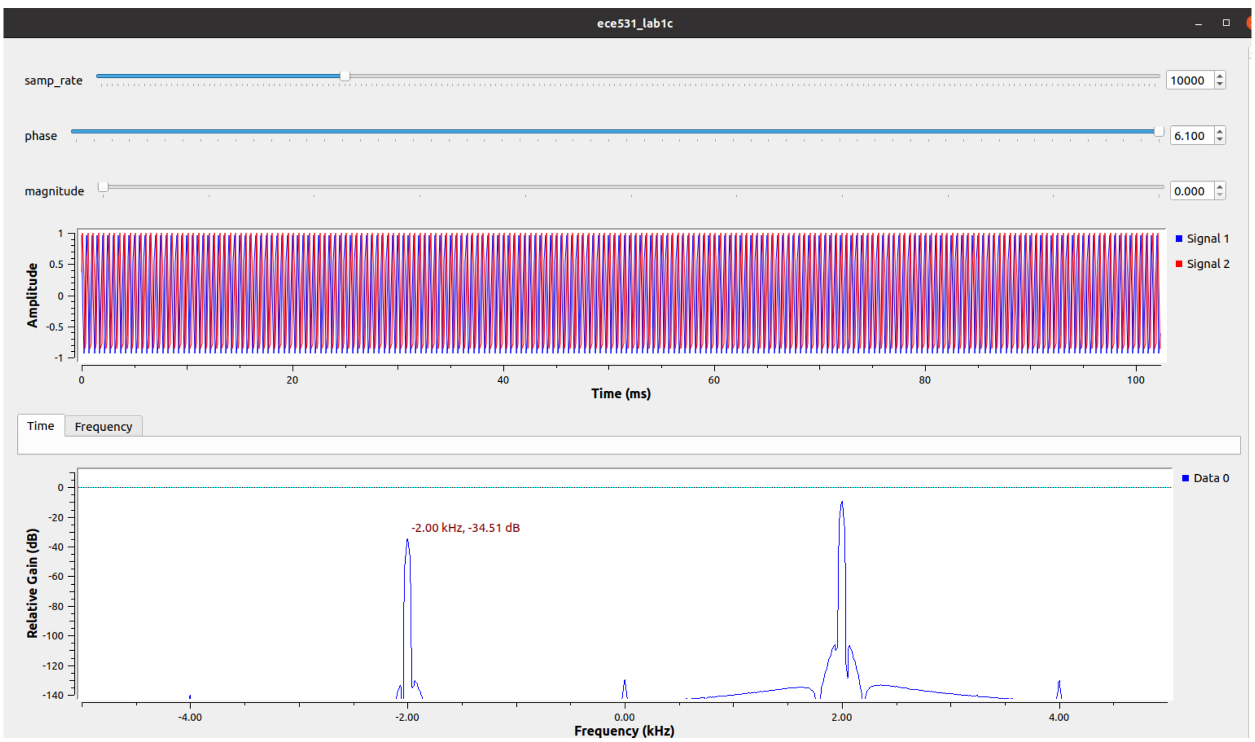

ece531_lab1c
samp_rate
10000
phase
0.000
magnitude
0.000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0


ece531_lab1c
samp_rate
10000
phase
0.100
magnitude
0.000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-2.00 kHz, -69.12 dB
Frequency (kHz)


ece531_lab1c
samp_rate
10000
phase
2.500
magnitude
0.000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-1.99 kHz, -40.00 dB
Frequency (kHz)


ece531_lab1c
samp_rate
10000
phase
3.800
magnitude
0.000
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-1.99 kHz, -35.05 dB
Frequency (kHz)

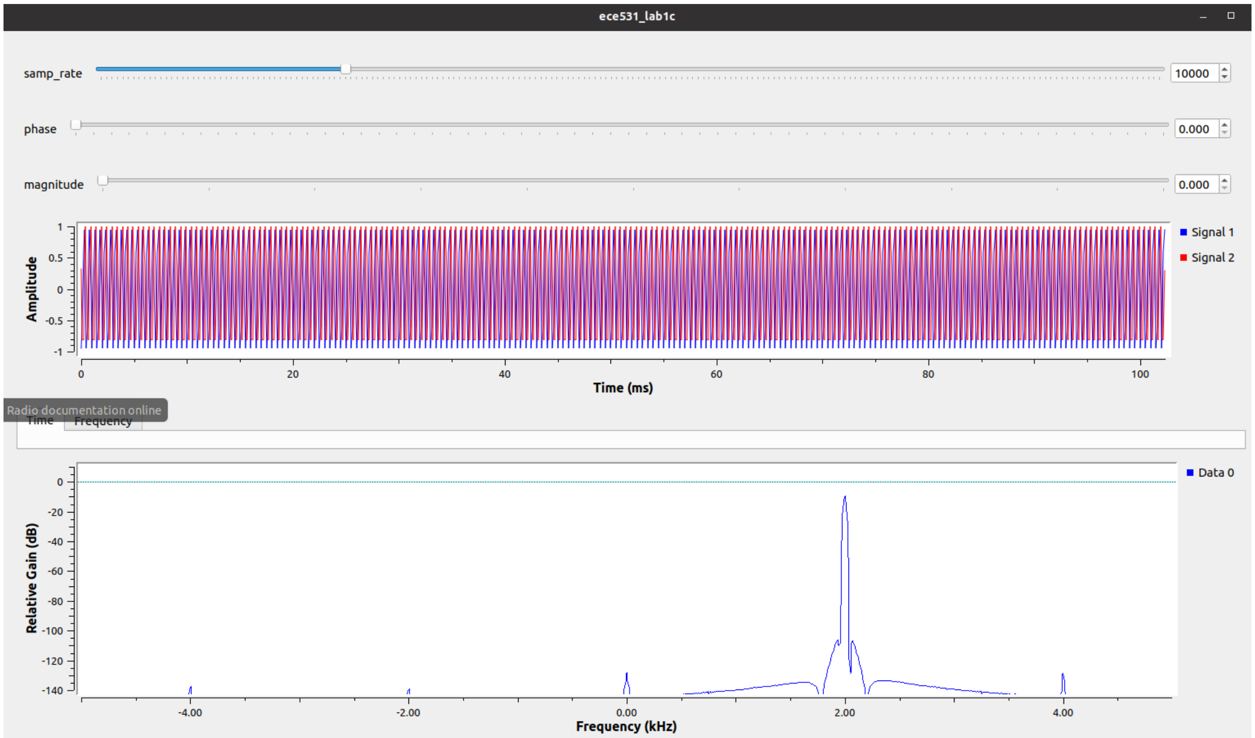


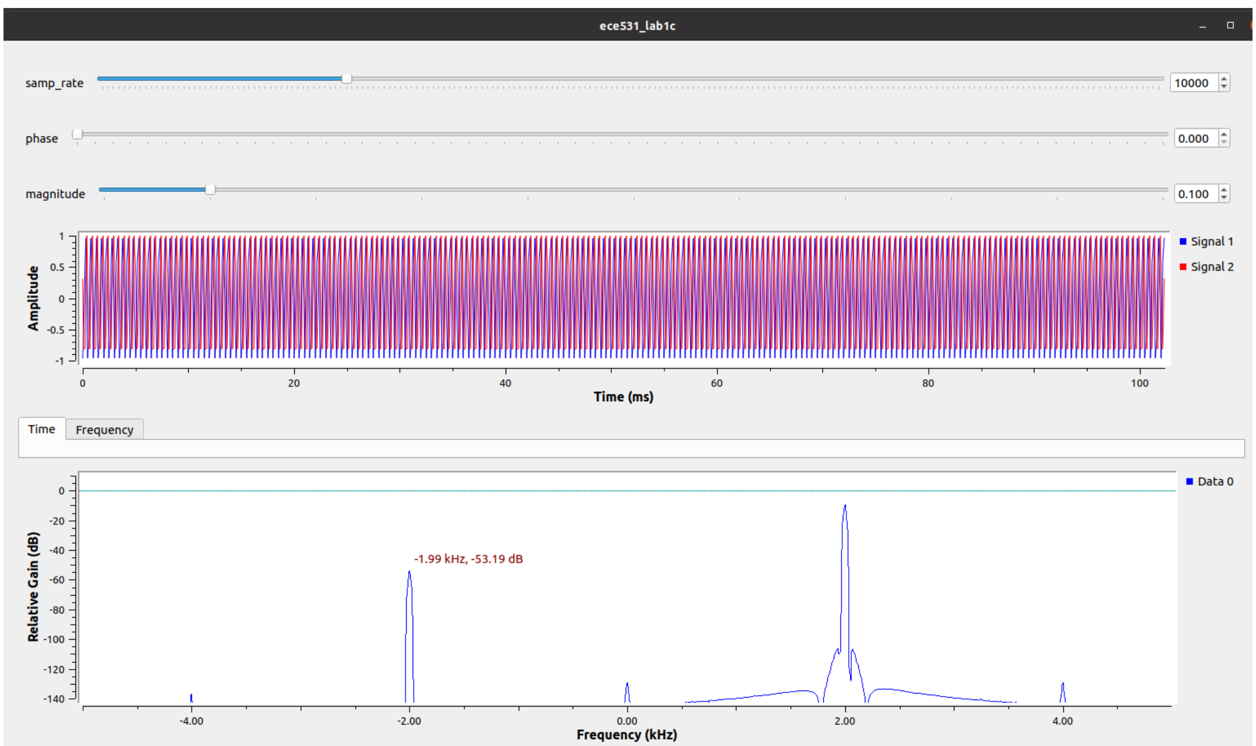


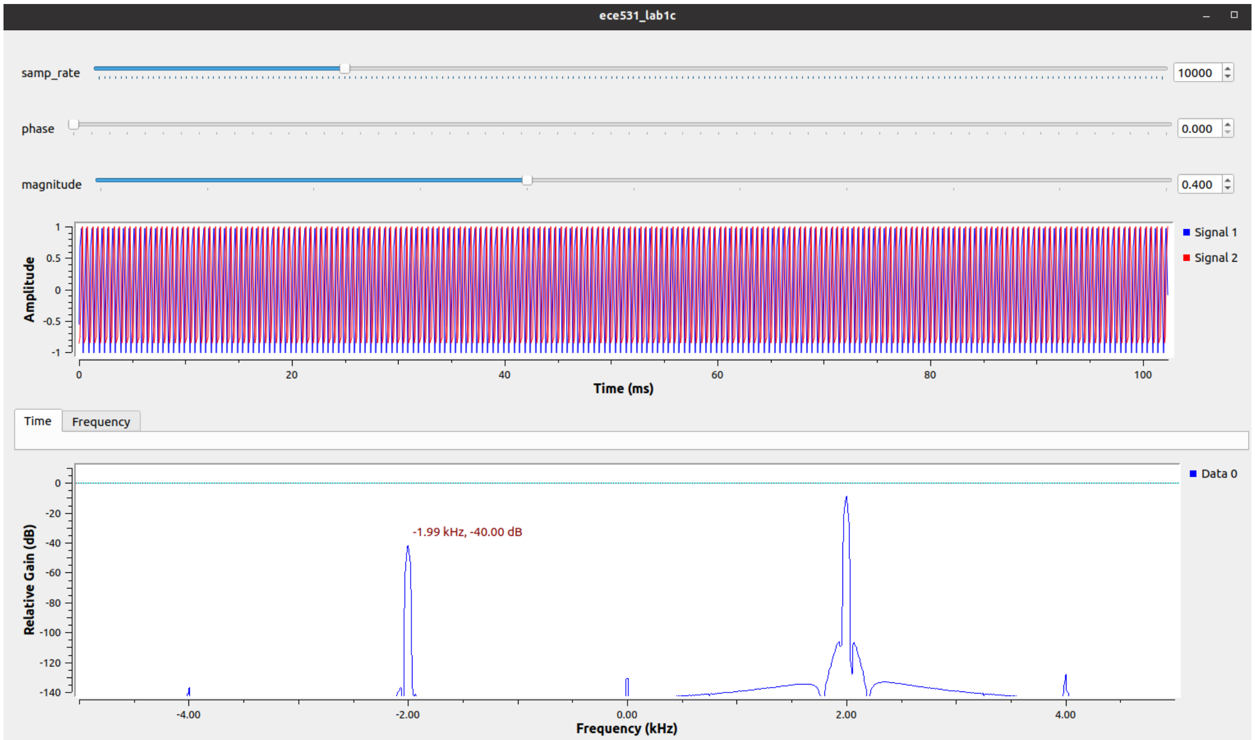
The phase angle is increasing and creating a larger imbalance between the cosine and sine I and Q values. In this case the I does not equal Q. The phase difference is no longer 90 degrees which will have an IQ imbalance which will need to be corrected; else there will be errors if kept.

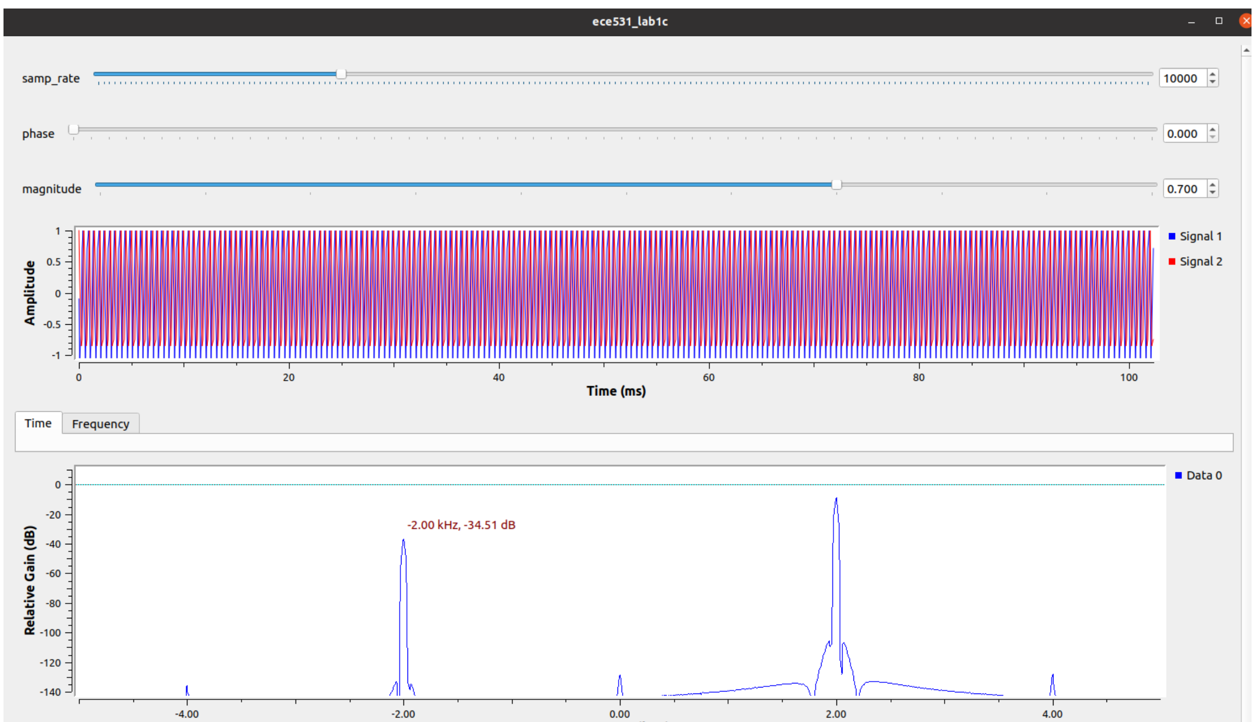
“Mismatches between the two LO signals and/or along the two branches…Cause the quadrature baseband signals to be corrupted, either due to amplitude or phase differences” [16].

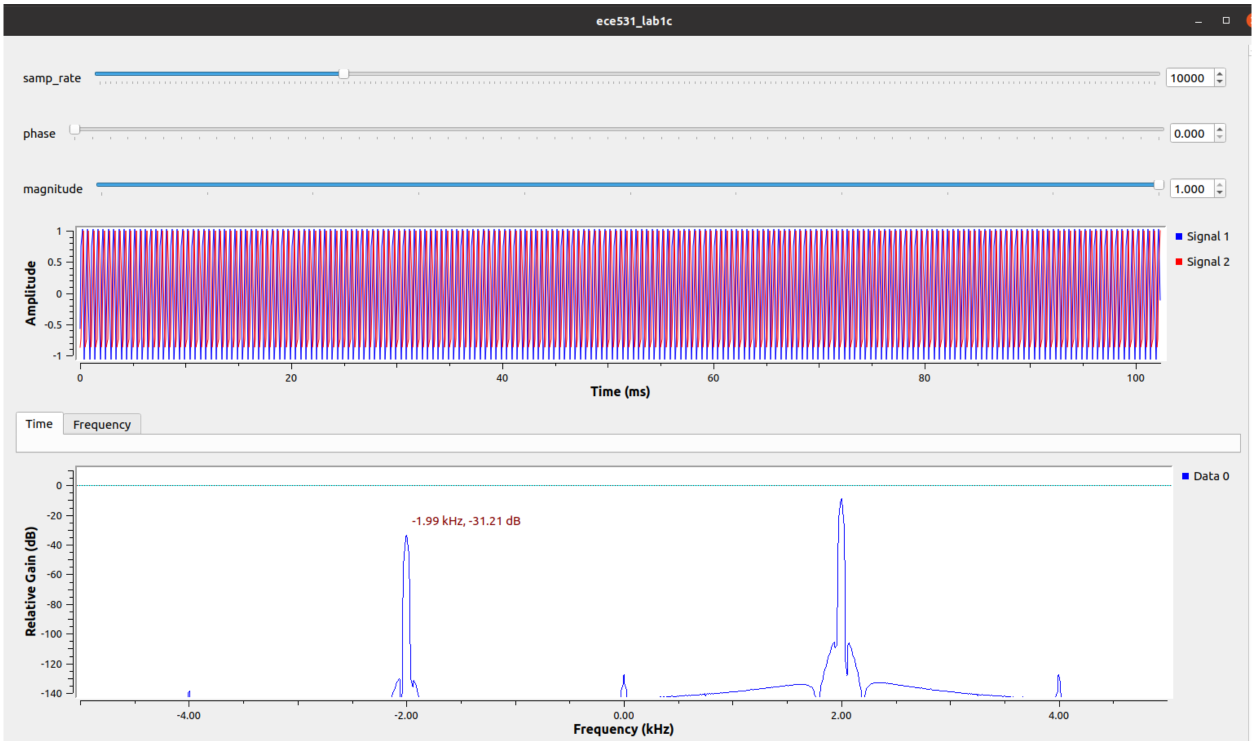
8.


ece531_lab1c
samp_rate
10000
phase
0.000
magnitude
0.000
Amplitude
Signal 1
Signal 2
Time (ms)
Radio documentation online
Time
Frequency
Relative Gain (dB)
Data 0
Frequency (kHz)


ece531_lab1c
samp_rate
10000
phase
0.000
magnitude
0.100
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-1.99 kHz, -53.19 dB
Frequency (kHz)

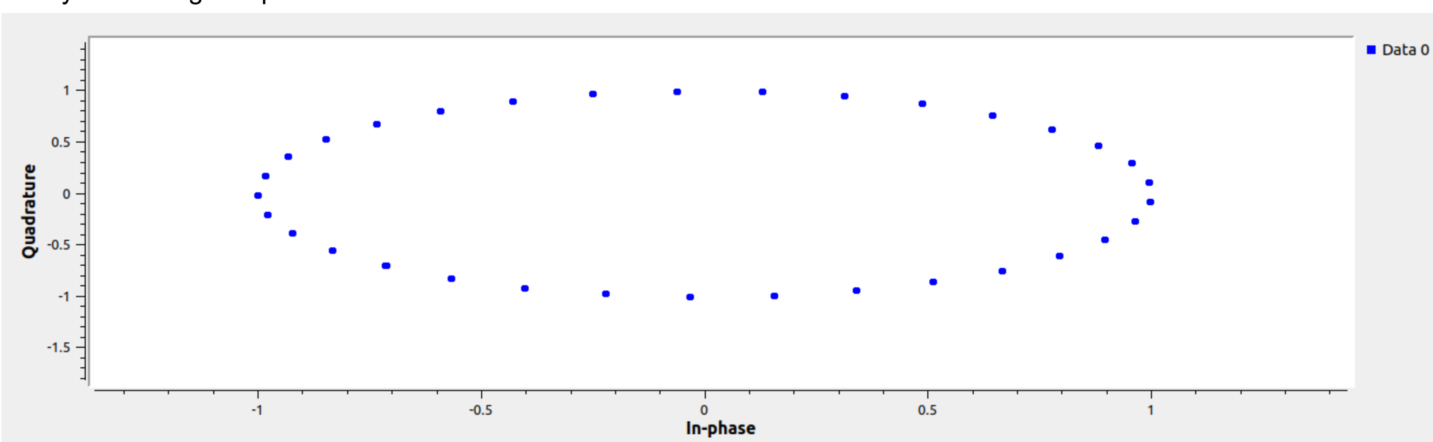

ece531_lab1c
samp_rate
10000
phase
0.000
magnitude
0.400
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-1.99 kHz, -40.00 dB
Frequency (kHz)

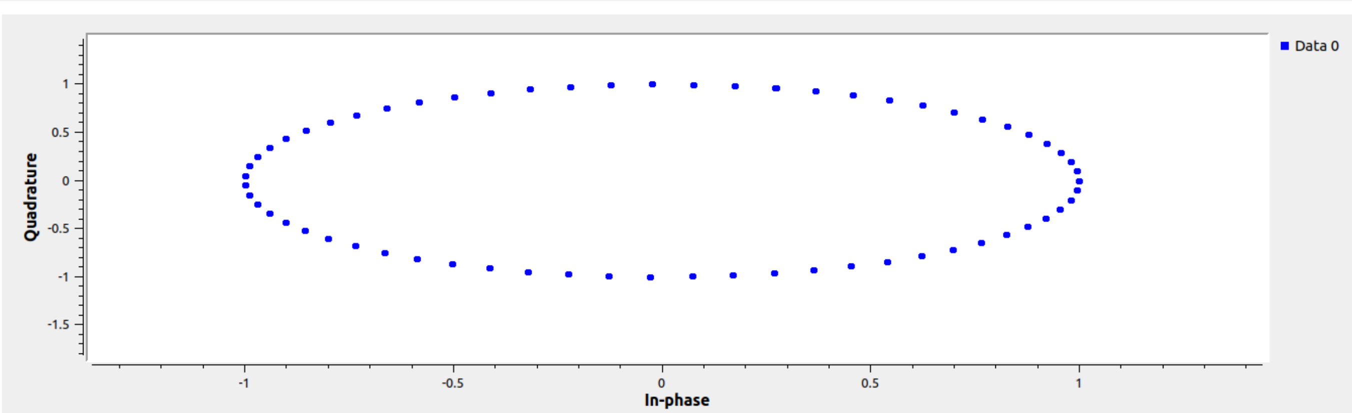

ece531_lab1c
samp_rate
10000
phase
0.000
magnitude
0.700
Amplitude
Signal 1
Signal 2
Time (ms)
Time
Frequency
Relative Gain (dB)
Data 0
-2.00 kHz, -34.51 dB

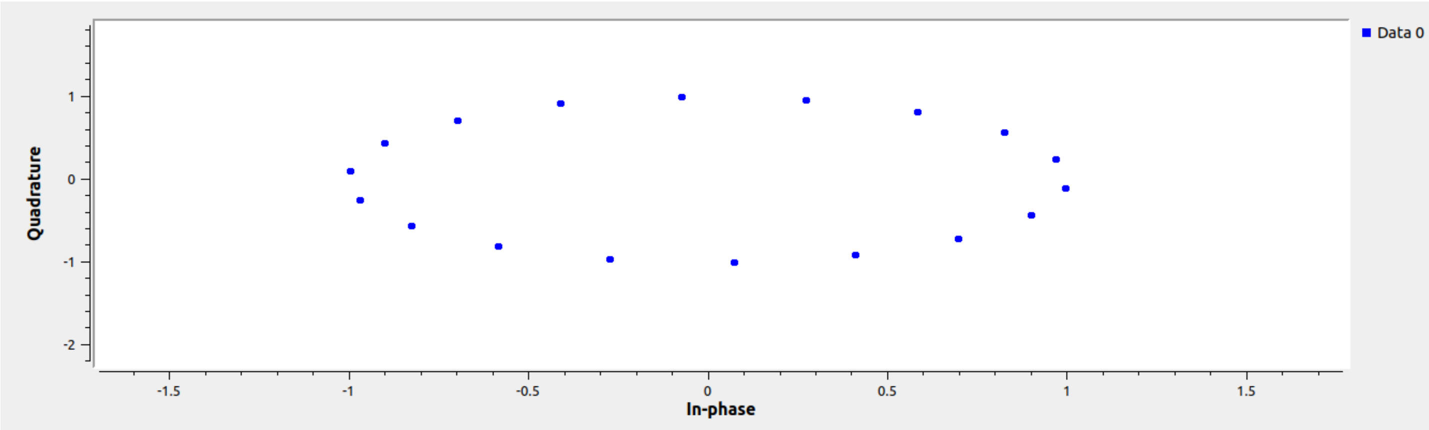


Increasing the amplitude of the complex signal creates an imbalance between the amplitudes of I and Q. They are no longer equal and the frequency domain is similar to when the phase was imbalanced. This will also affect the phase difference to no longer be 90 degrees.

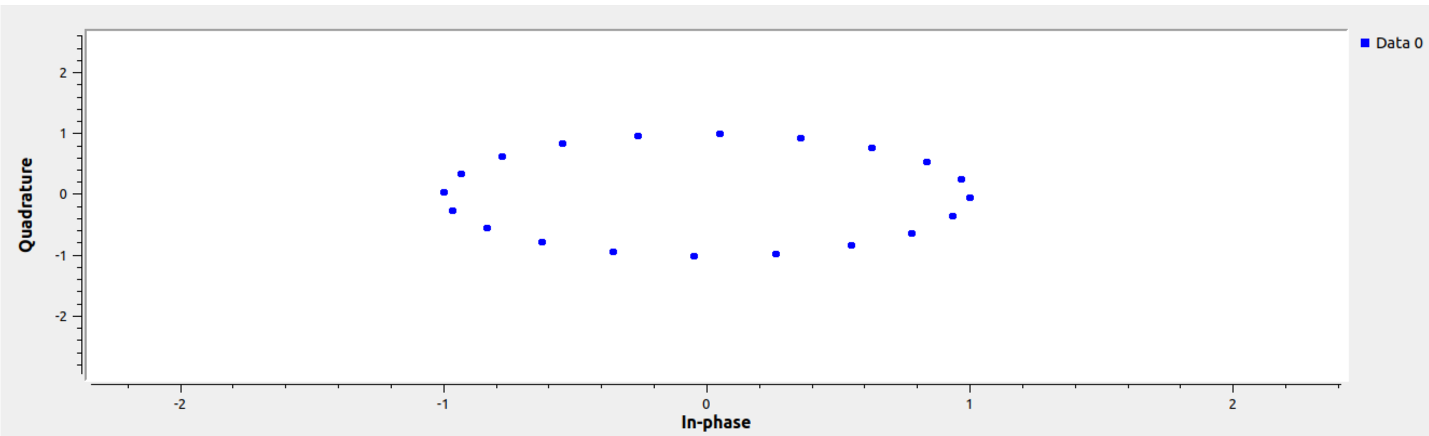
9.
Slowly increasing sample rate



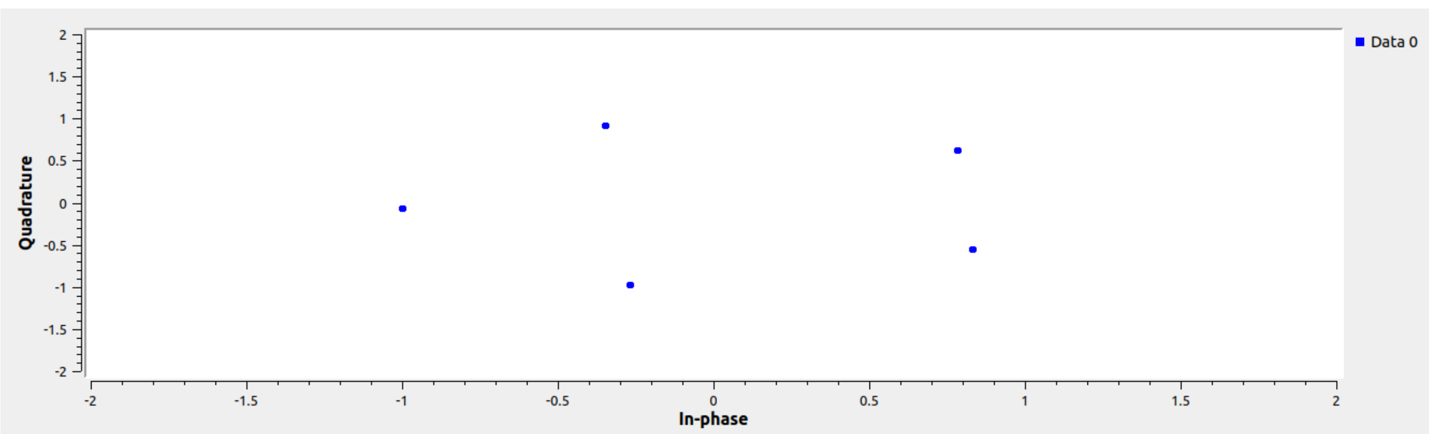


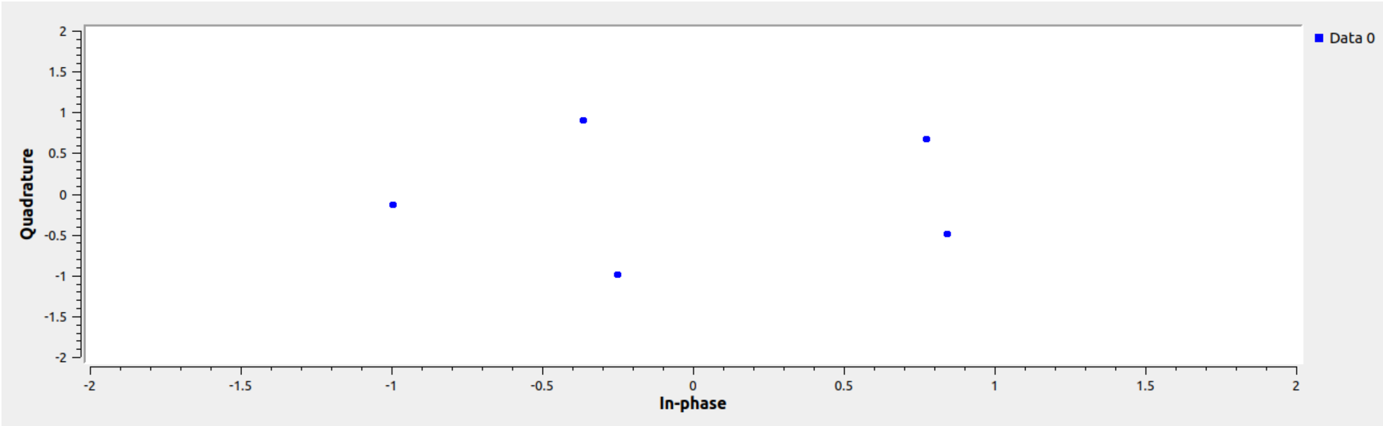


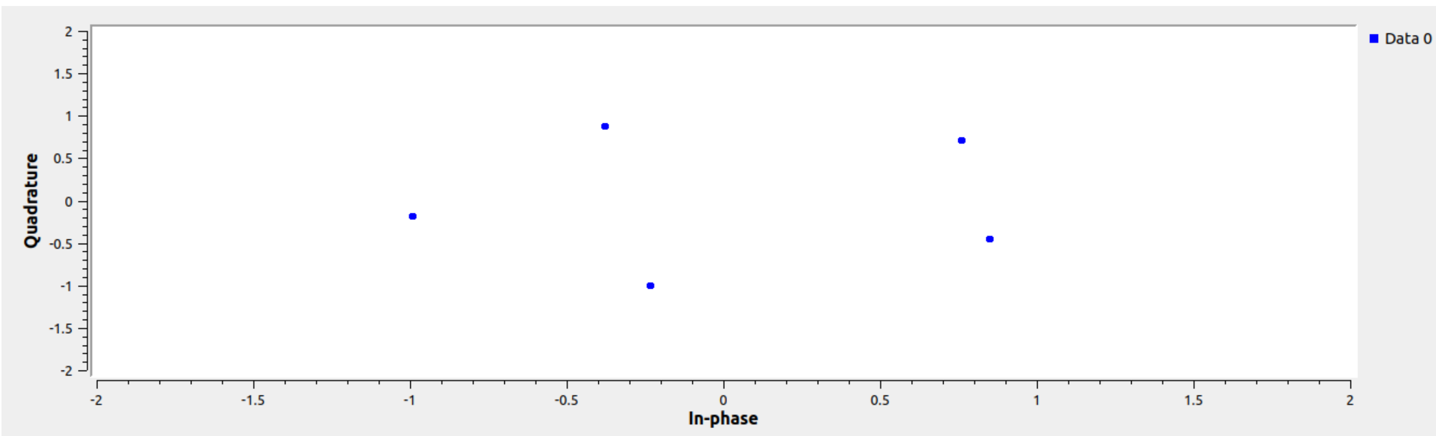
Max value of sampling rate:



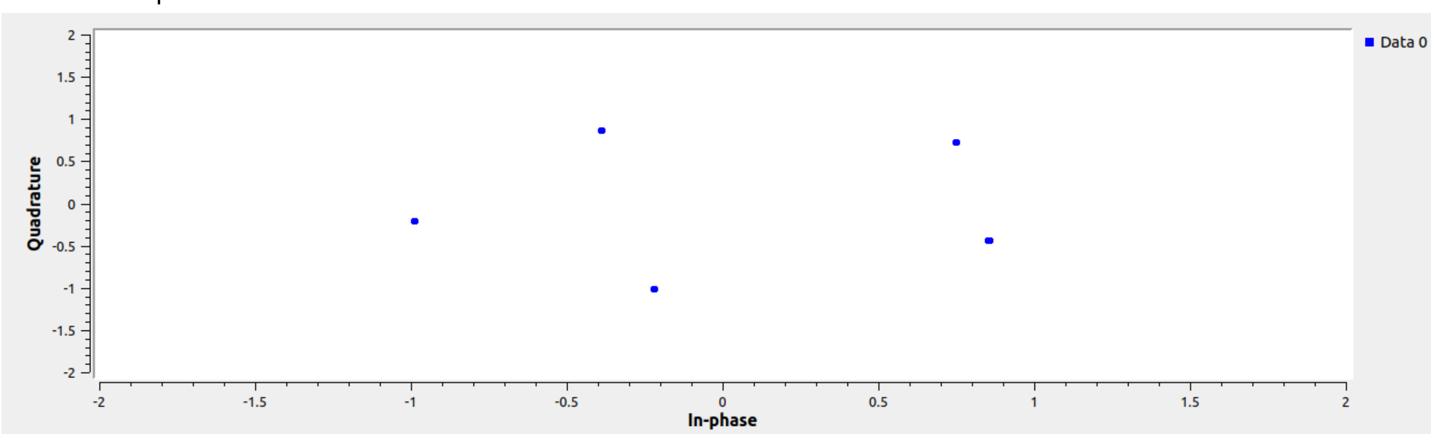
Increasing the sample rate, increases the number of samples being produced in the constellation domain.
Slowly increasing phase

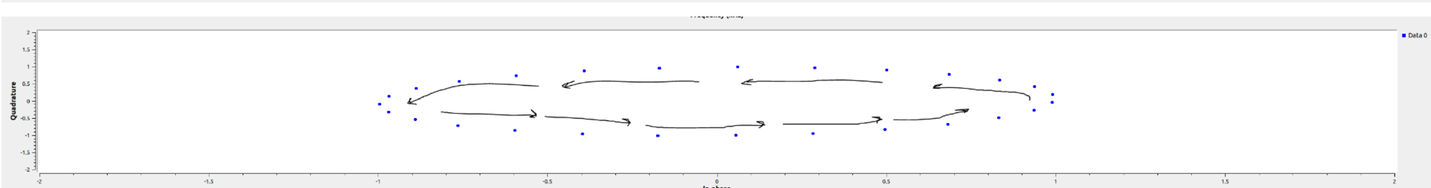




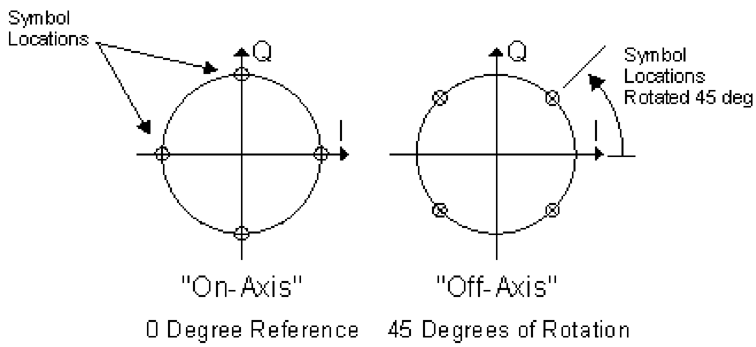


Max value of phase imbalance

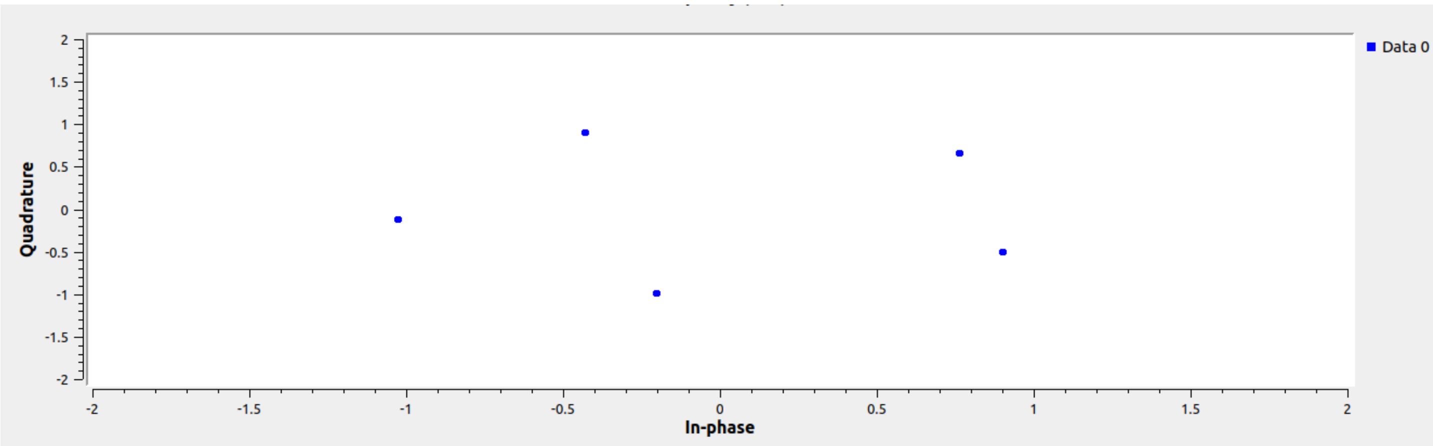


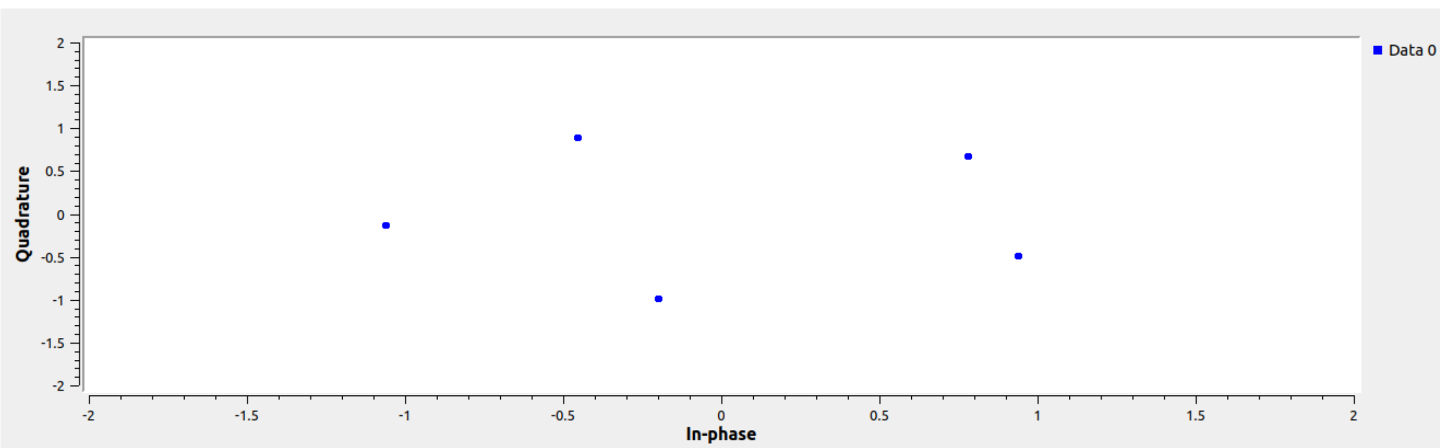


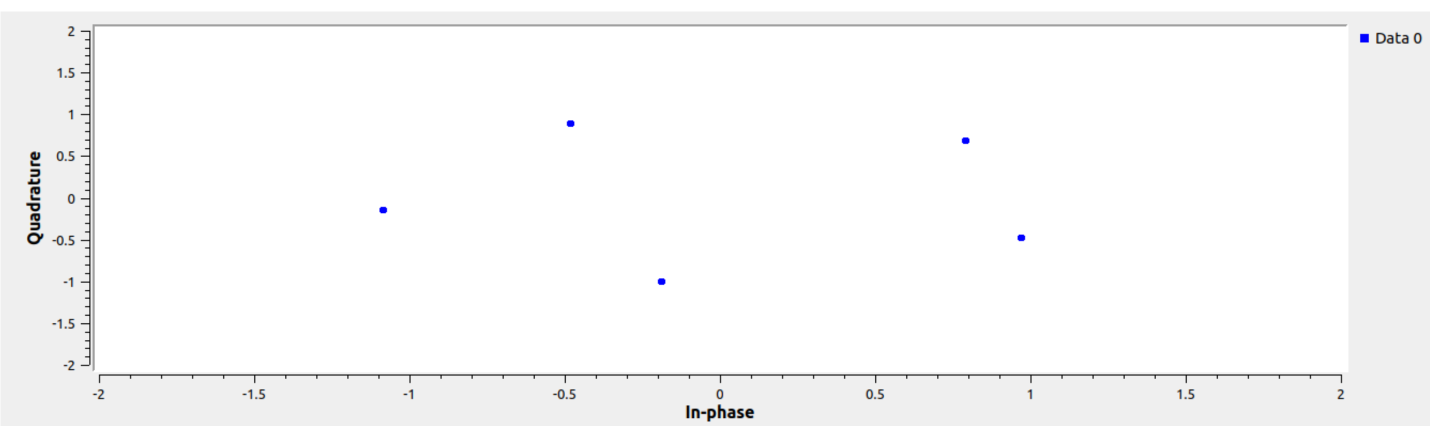
Maxing out the phase imbalance, shifts the constellation in a counter-clockwise direction based off the imbalance created by the phase.



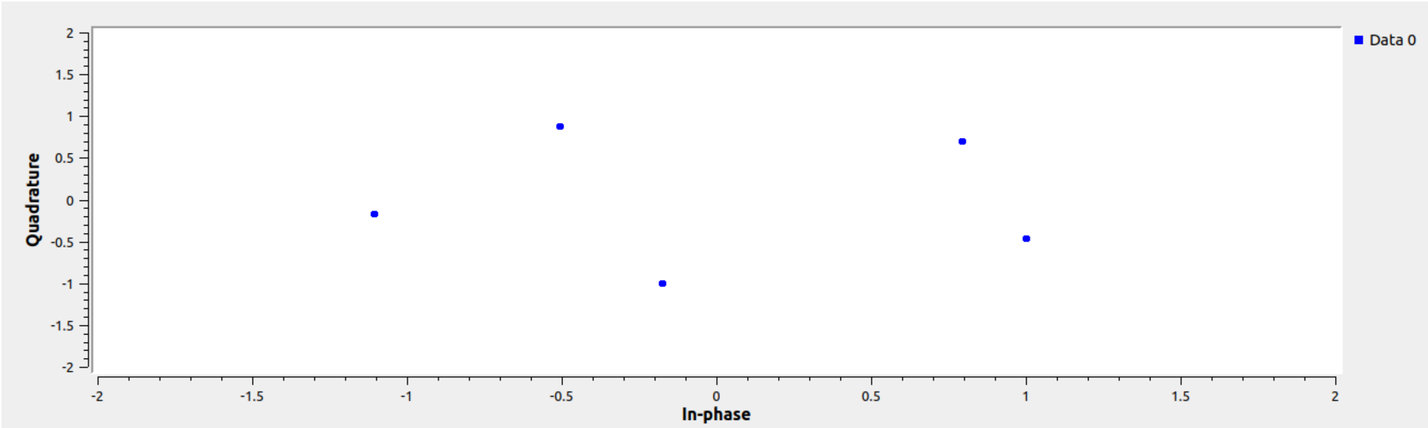
Slowly increasing magnitude

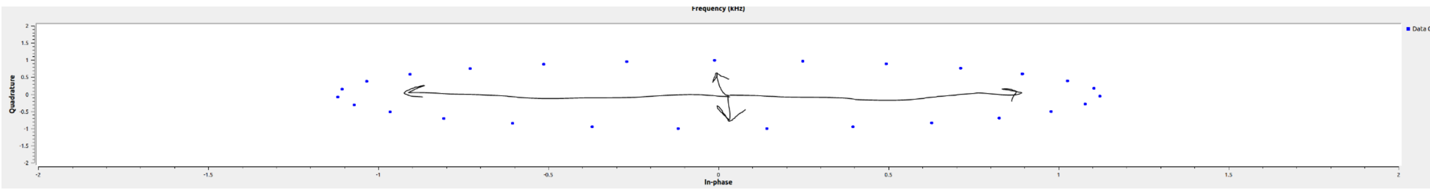






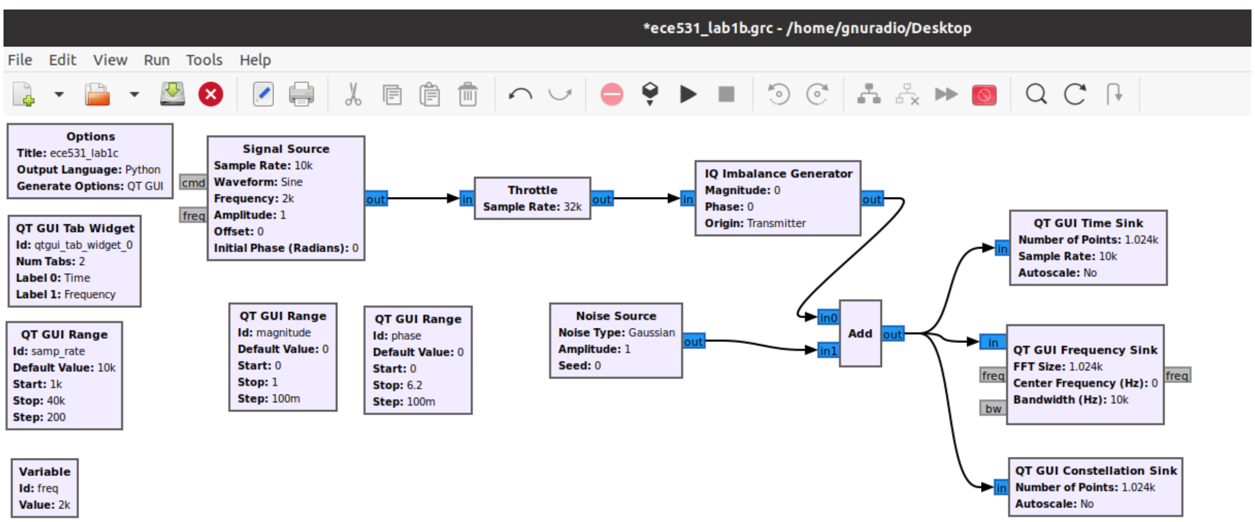
Max value of amplitude imbalance



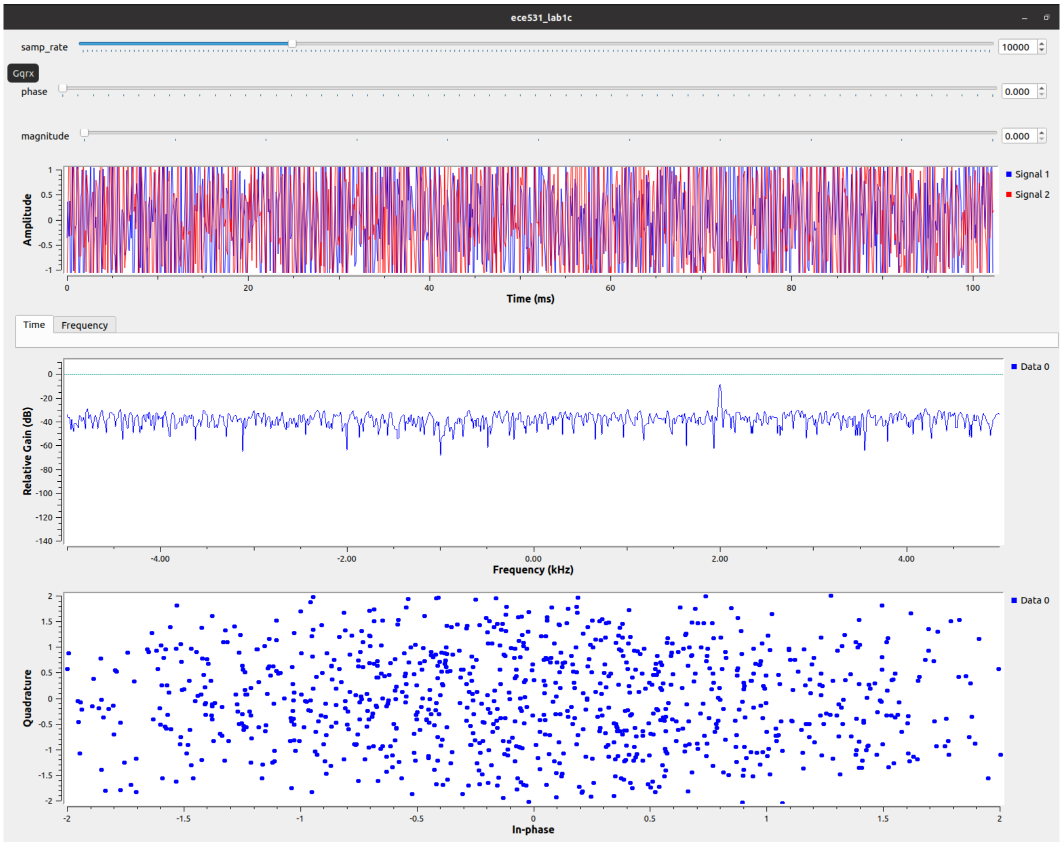


When maxing out the amplitude imbalance, the constellation is expanding outward evenly in all directions.

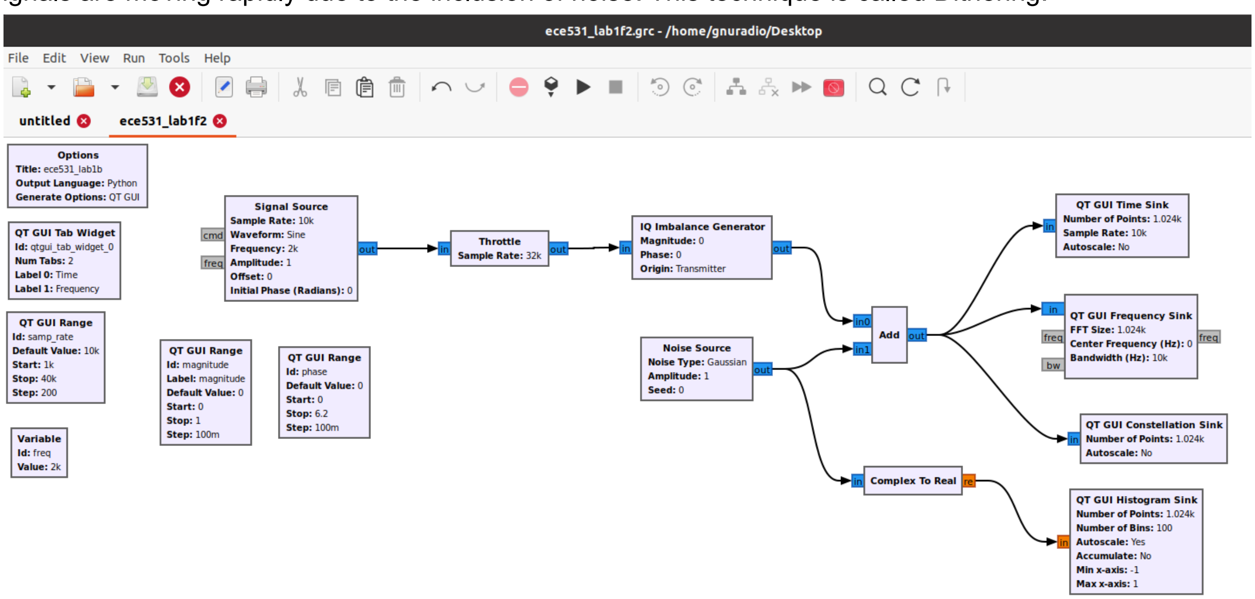
3.6 ADDING NOISE

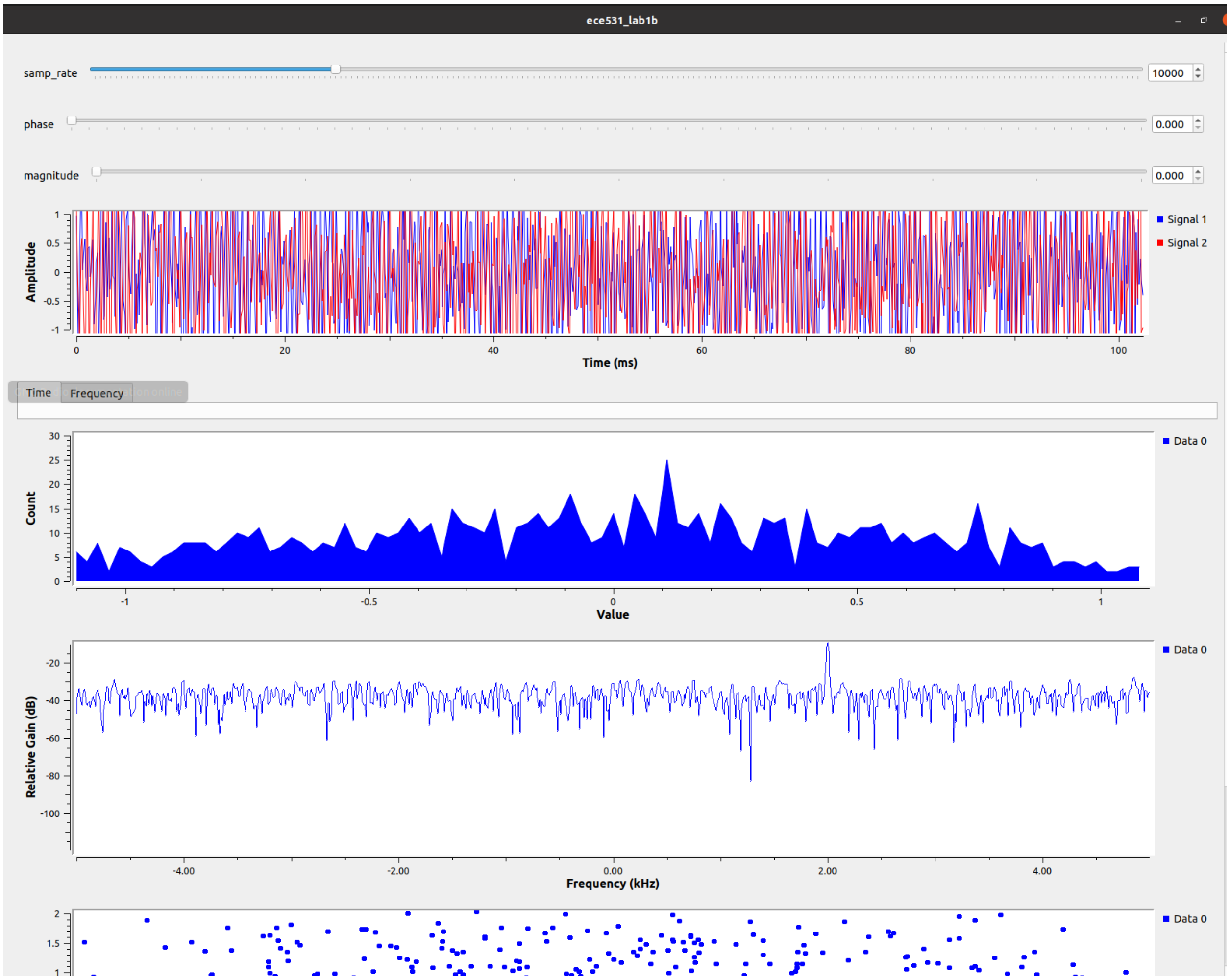


2

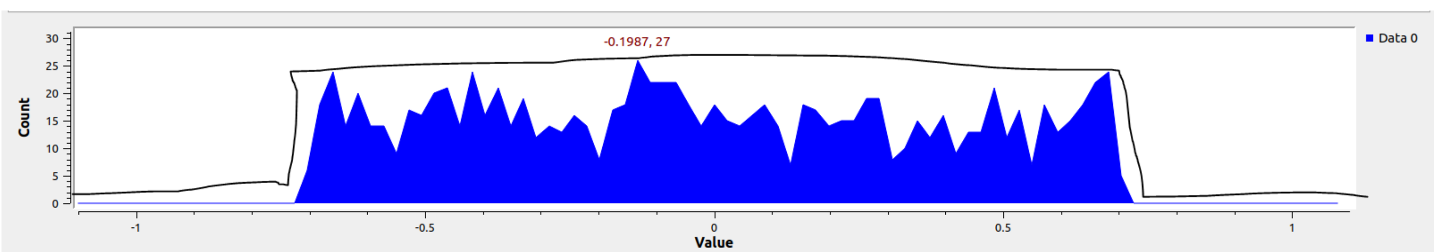


All the signals are moving rapidly due to the inclusion of noise. This technique is called Dithering.

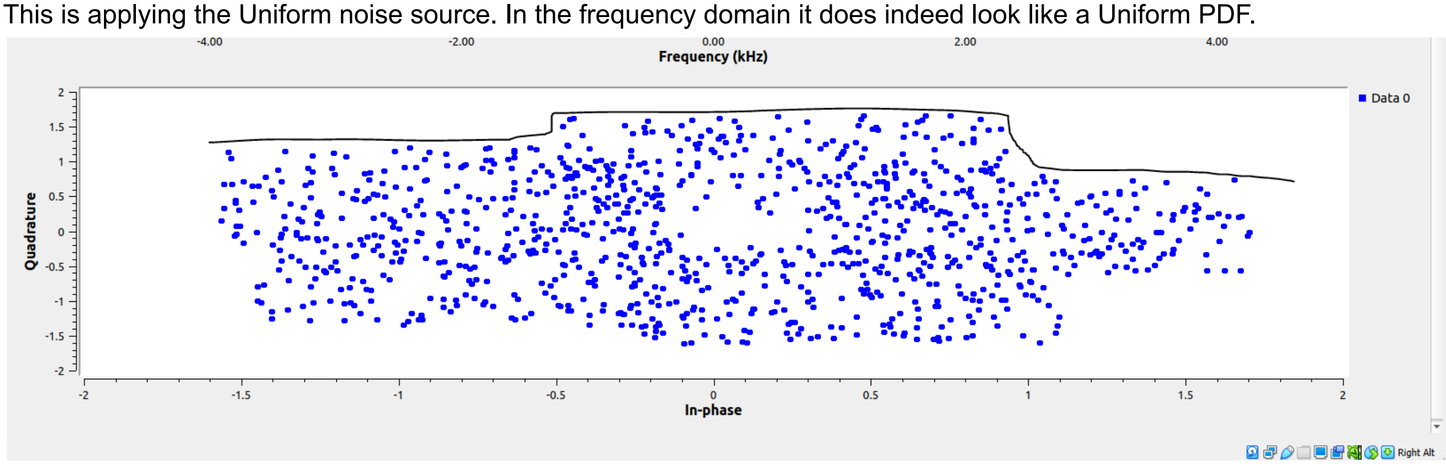




This is applying the Gaussian noise source. Looking in the frequency domain, it looks normally distributed. Gaussian PDF has a normal distribution.



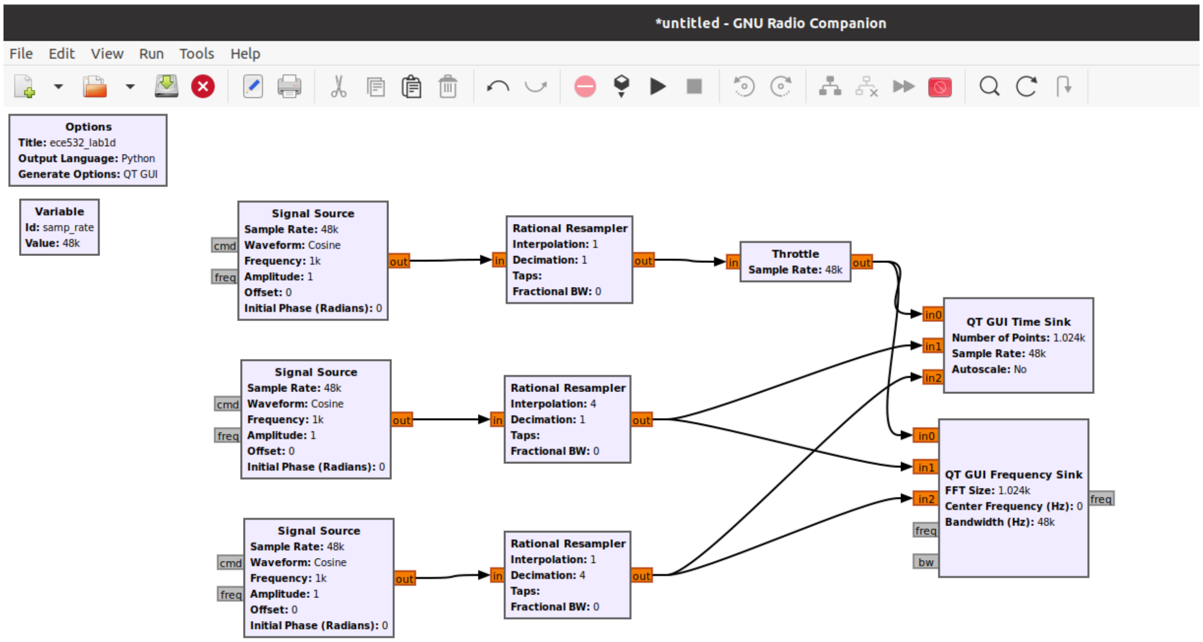
This is applying the Uniform noise source. In the frequency domain it does indeed look like a Uniform PDF.



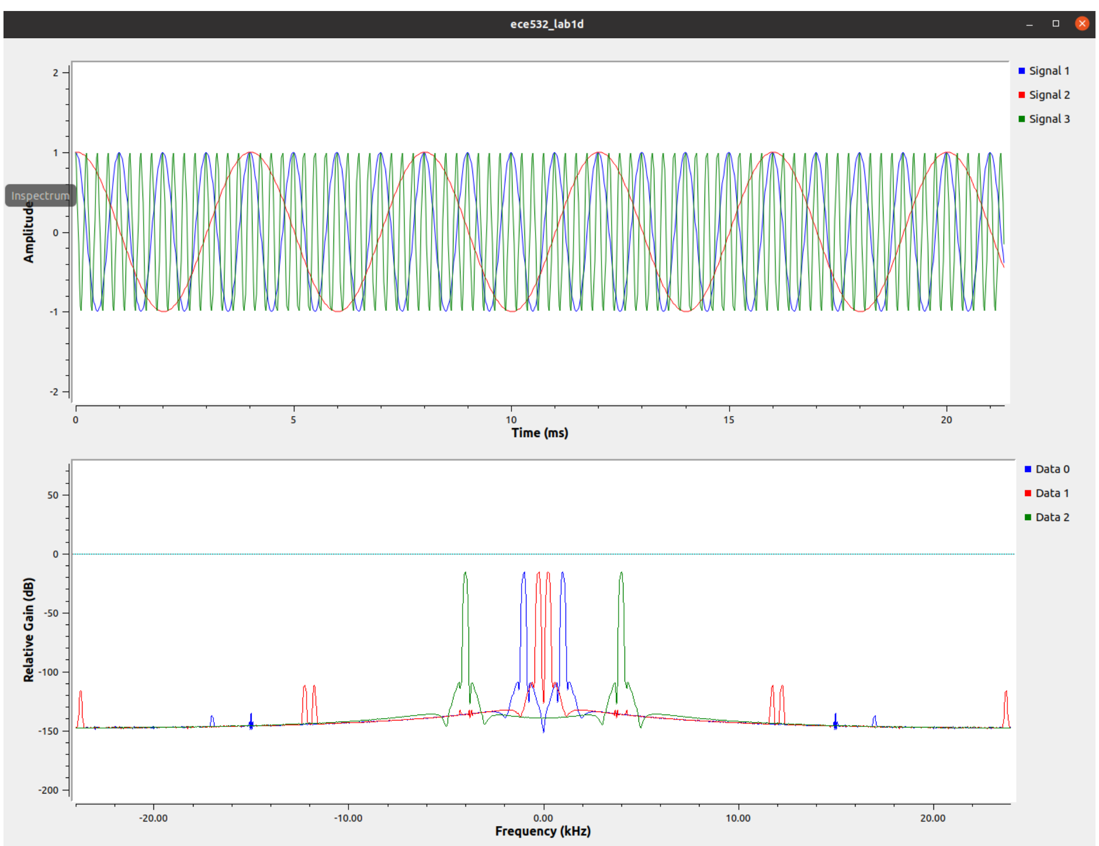
Looking in the constellation sink, it also seems to be applying a uniform distribution. When applying the Laplacian and Impulse noise source type, it seems to not output.

3.7 INTERPOLATION AND DECIMATION

1.



2.



3.

Each signal has a different frequency. In order of highest frequency to lowest: Signal 3, Signal 1, Signal 2.

4.

Each signal has a different resampling factor (interpolation(L) over decimation(M) = L/M). In the case for Signal 1, the resampler factor is 1; Signal 2 is 4, Signal 3 is ¼. This changes the sampling rates of each signal as so: Signal 1 (48kHz*1 = 48kHz); Signal 2 (48kHz*4 = 192kHz) and Signal 3 (48kHz*(¼) = 12kHz). Hence why they differ in the frequency spectrum, since they have a different sampling rate, their bandwidths are also different.

5.

Signal 1 is neither interpolated nor decimated, since the resampling rate is 1, it did not change the sampling rate. Signal 2 has been interpolated or upsampled since the sampling rate has increased. Signal 3 has been decimated since the sampling rate has decreased.

4. QUESTIONS

1. IQ sampling can save the use of a tunable filter. They also allow RF modulation and demodulation. Since IQ signals are represented as discrete time sampled data, through digital signal processing their transmitter and receiver characteristics can be defined. [11]
2. It ensures the average sample rate does not exceed its specified rate. It should be included when there is no limiting block (hardware has limiting blocks, simulation does not) and it needs to be added. Multiple throttle blocks create a latency in the sample rate, and could slow down the simulation. A throttle block is unnecessary when working with hardware, since the hardware itself has a throttle block.
3. A Nyquist zone divides the spectrum into sampling frequency over 2. It helps to reduce the chances of digital signal processing to fail, so it helps in having a smaller SDR bandwidth and having Nyquist zone(s) assist in doing that.
4. It is used to reduce the error output of DAC. DACs themselves produce noise and using a dither will mitigate this.

5. CONCLUSION

My background is not Electrical nor do I have any experience with signals; this lab allowed me to gain a certain fundamental understanding of signal processing and the rules/theorems that apply to them. GNURadio simulation is a useful tool when doing so; allowing the user to essentially “play” with the signals that are being generated and having visualization of this.

References

[1] https://www.youtube.com/watch?v=r46wK-2Ohvw

[2]https://community.sw.siemens.com/s/article/digital-signal-processing-sampling-rates-bandwidth-spectral-lines-and-more#:~:text=Sampling%20rate%20(sometimes%20called%20sampling,points%20are%20acquired%20every%20second.&text=In%20practice%2C%20sampling%20even%20higher,correctly%20in%20the%20time%20domain

[3] https://www.youtube.com/watch?v=h_7d-m1ehoY

[4] http://whiteboard.ping.se/SDR/IQ

[5]https://rfmw.em.keysight.com/wireless/helpfiles/89600b/webhelp/subsystems/digdemod/Content/digdemod_para_interact_iqgainimb_quadskewerr.htm

[6] https://wiki.analog.com/resources/eval/user-guides/ad-fmcomms1-ebz/iq_correction

[7] https://en.wikipedia.org/wiki/Dither

[8] https://dspguru.com/dsp/faqs/multirate/resampling/

[9] https://www.markimicrowave.com/blog/the-why-and-when-of-iq-mixers-for-beginners/

[10] https://wiki.gnuradio.org/index.php/Throttle

[11]https://www.tek.com/en/blog/quadrature-iq-signals-explained

[12] https://lists.gnu.org/archive/html/discuss-gnuradio/2012-02/msg00011.html

[13] http://www.panoradio-sdr.de/analog-digital-conversion/

[14] http://www.precisionmechatronicslab.com/wp-content/uploads/2016/06/hf_dither_v12-1.pdf

[15] Software-Defined Radio for Engineers

[16] https://en.wikipedia.org/wiki/IQ_imbalance

[17]https://en.wikipedia.org/wiki/Aliasing