Lab 5

name of the lab?

Elena Montalvo
Data collected with Darren Ray
ECE3101L - Andrew Pagnon
October 13, 2022

Objective

Analyze converting a continuous analog signal to a discrete sample signal. Find the Nyquist sampling rate for a given signal. Be able to explain signal analog-to-digital (ADC) and digital-to-analog (DAC) conversion

Introduction

The ananlog signal must be analyze and process at continues time. We convert the signal into digital by sampling the continues time signal so the signal could be process by a computer or a device then we reconstruct the signal for digital to analog.

1. A sampling system,

2. A digital signal processor,

3. A reconstruction system.

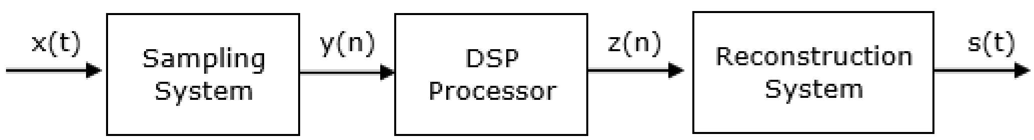


Figure 1: Example of a typical digital signal processing system

Its is essential that when we digitalize the anolog signal we take efficient sampling so when we correctly reconstruct the signal

Sampling

Sampling is the process of measuring the value of a continuous-time signal at certain instants of time. Typically, these measurements are uniformly separated by the sampling period, Ts.

If x(t) is the input signal, then the sampled signal, y(n), is as follows:’

$$y(n) = x(t)|t = n * Ts$$

Fourier transform (DTFT), . However, is a continuous-time signal, requiring the use of the continuous-time Fourier transform (CTFT), denoted as X(f). Fortunately, can be written in terms of X(f):

$$Y\left(e^{jw}\right) = \left(\frac{1}{Ts}\right) * \sum_{k=-\infty}^{\infty} X(f)|f \ = \frac{w-2*\pi*k}{2*\pi*Ts} = \left(\frac{1}{Ts}\right) * \sum_{k=-\infty}^{\infty} X\left(\frac{w-2*\pi*k}{2*\pi*Ts}\right) \ (1)$$

Consistent with the properties of the DTFT, is periodic with a period 2 π. It is formed by rescaling the amplitude and frequency of X(f), and then repeating it in frequency every 2 π .If X(f) has frequency components that are above 1/(2Ts), the repetition in frequency will cause these components to overlap with the components below 1/(2Ts). This causes an unrecoverable distortion, known as aliasing, that will prevent a perfect reconstruction of X(f).

To prevent aliasing, most sampling systems first low pass filter the incoming signal to ensure that its frequency content is below the Nyquist frequency through an anti-aliasing low pass filter.

Here, it is understood that is periodic with period 2*π. Note in this expression that and X(f) are related by a simple scaling of the frequency and magnitude axes. Also note that ω = π in corresponds to the Nyquist frequency, f = 1/(2Ts) in X(f).

After the sampled signal has been digitally processed, it then must be reconstructed. Conceptually, this may be done by converting the discrete-time signal to a sequence of continuous-time impulses that are weighted by the sample values. If this continuous-time “impulse train” is filtered with an ideal low pass filter, with a cutoff frequency equal to the Nyquist frequency, a scaled version of the original low pass filtered signal will result. The spectrum of the reconstructed signal S(f) is given by

No need to add portions of the manual to the report. Howe

Shannon’s Sampling Theorem

A bandlimited signal x(t) is sampled at an interval of Ts = 1/ $f_{sampling}$, where Ts is the sampling interval and $f_{sampling}$ is the sampling rate. Let the bandwidth of x(t) be B Hz. Then, the sampling does not destroy any information content of x(t) if the sampling rate is greater than or equal to two times the highest frequency content of x(t), that is,

fsampling ≥ 2B.

The original signal x(t) can be reconstructed from the sampled signal by passing it through a low pass filter with cutoff frequency $f_{sampling}/2$. This theorem is called Shannon’s sampling theorem, and 2B is called Nyquist’s sampling rate and 1/(2B) is called Nyquist’s sampling interval. If the sampling rate is exactly Nyquist’s sampling rate, an ideal low pass filter is needed to recover the original signal. However, an ideal low pass filter is physically unrealizable. Thus, the actual sampling rate should be higher than the Nyquist’s sampling rate.

Reconstruction using a Sample-and-Hold

In practice, signals are reconstructed using digital-to-analog converters. These devices work by reading the current sample, and generating the corresponding output voltage for a period of Ts seconds. The combined effect of sampling and D/A conversion may be thought of as a single sample-and-hold device. Unfortunately, the sample-and-hold process distorts the frequency spectrum of the reconstructed signal. In this section, we will analyze the effects of using a zeroth − order sample-and-hold in a sampling and reconstruction system. Later in the laboratory, we will see how the distortion introduced by a sample-and-hold process may be reduced with discrete-time interpolation.

Figure 2 illustrates a system with a low-pass input filter, a sample-and-hold device, and a low-pass output filter. If there were no sample & hold, this system would simply be two analog filters in cascade. We know the frequency response for this simpler system. Any differences between this and the frequency response for the entire system is a result of the sample & hold reconstruction. Our goal is to compare the two frequency responses using MATLAB. For this analysis, we will assume that the filters are Nth order Butterworth filters with a cutoff frequency of fc, and that the sample-and-hold runs at a sampling rate of fs = 1/Ts.

We will start the analysis by first examining the ideal case. Consider replacing the sample and-hold with an ideal impulse generator, and assume that instead of the Butterworth filters we use perfect low-pass filters with a cutoff of fc. After analyzing this case, we will modify the results to account for the sample-and-hold and Butterworth filter roll-off.

If an ideal impulse generator is used in place of the sample-and-hold, then the frequency spectrum of the impulse train can be computed by combining the sampling equation (1) with the reconstruction equation (2).

If we assume that F_s>2*f_c, then the infinite sum reduces to one term. In this case, the reconstructed signal is given by

Notice that the reconstructed signal is scaled by the factor 1/ Ts.

Of course, the sample-and-hold does not generate perfect impulses. Instead, it generates a pulse of width Ts, and magnitude equal to the input sample. Therefore, the new signal out of the sample-and-hold is equivalent to the old signal (an impulse train) convolved with the pulse

Convolution in the time domain is equivalent to multiplication in the frequency domain, so this convolution with p(t) is equivalent to filtering with the Fourier transform P(f) where

Finally, the magnitude of the frequency response of the N-th order Butterworth filter is given by

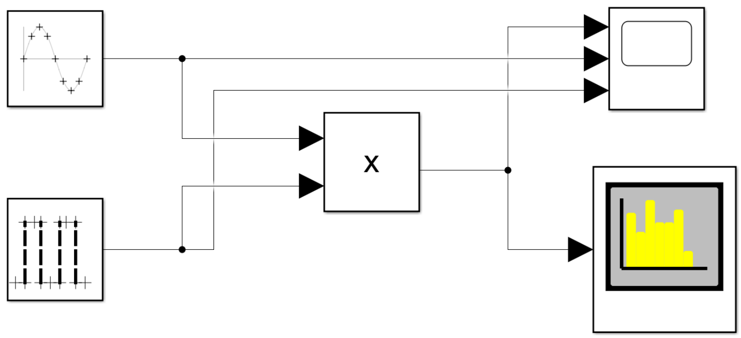
We may calculate the complete magnitude response of the sample-and-hold system by combining the effects of the Butterworth filters in equation (5), the ideal sampling system in equation (3), and the sample-and-hold pulse width in equation (4). This yields the final expression

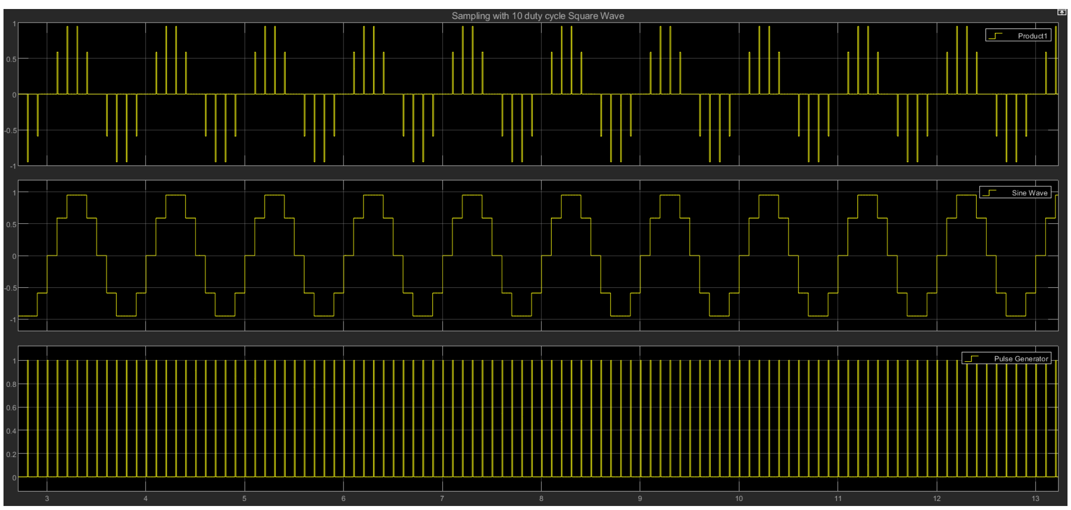
Notice that the expression produces a roll-off in frequency, which will attenuate frequencies close to the Nyquist rate. Generally, this roll-off is not desirable.

Results

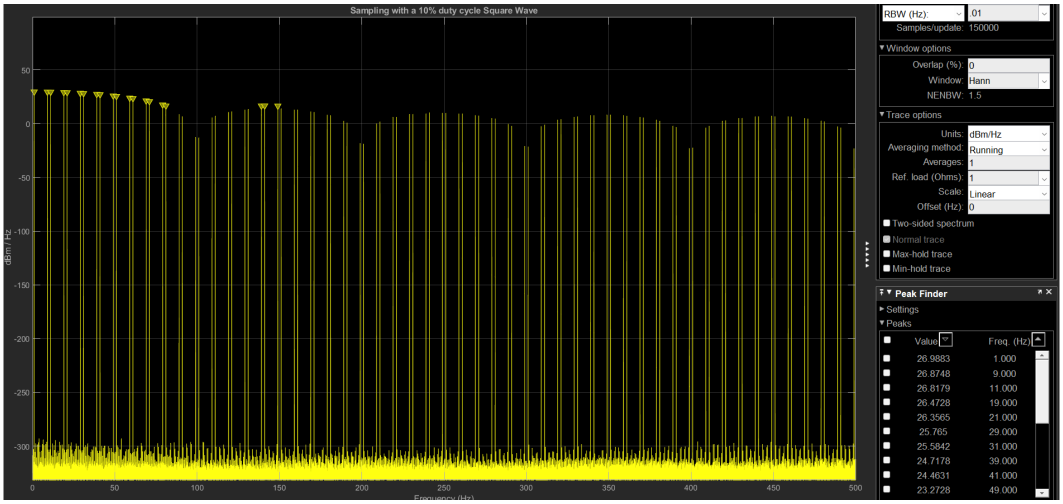
Sampling 10%

Sampling with 10% duty Cycle Square wave

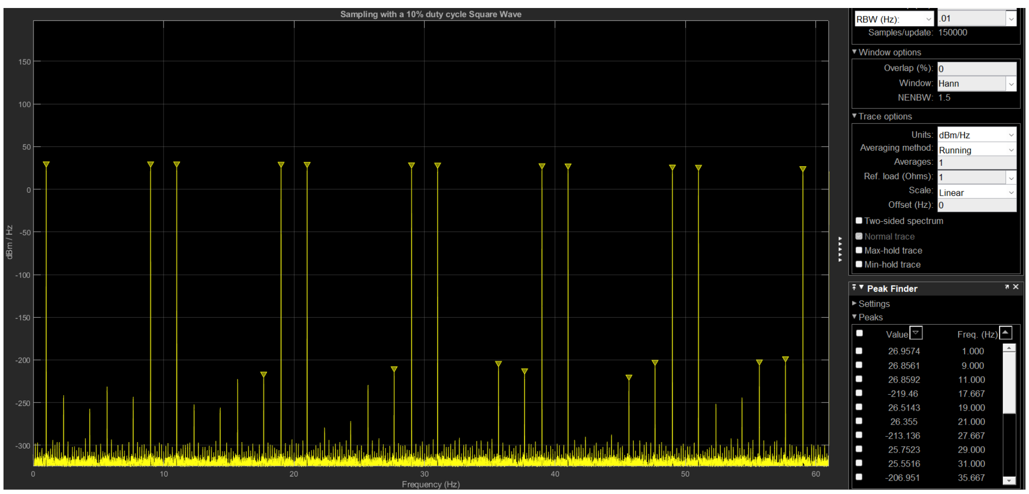




The 1st wave represents the product between the sine wave and the 10% pulse wave. The 2nd wave represents the sine wave and the 3rd wave is the pulse wave at 10%



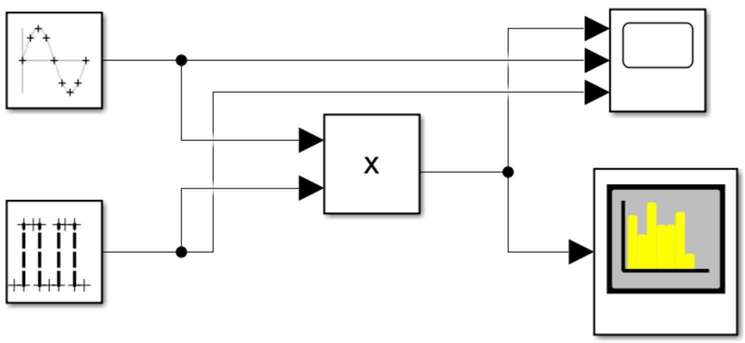
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 500Hz

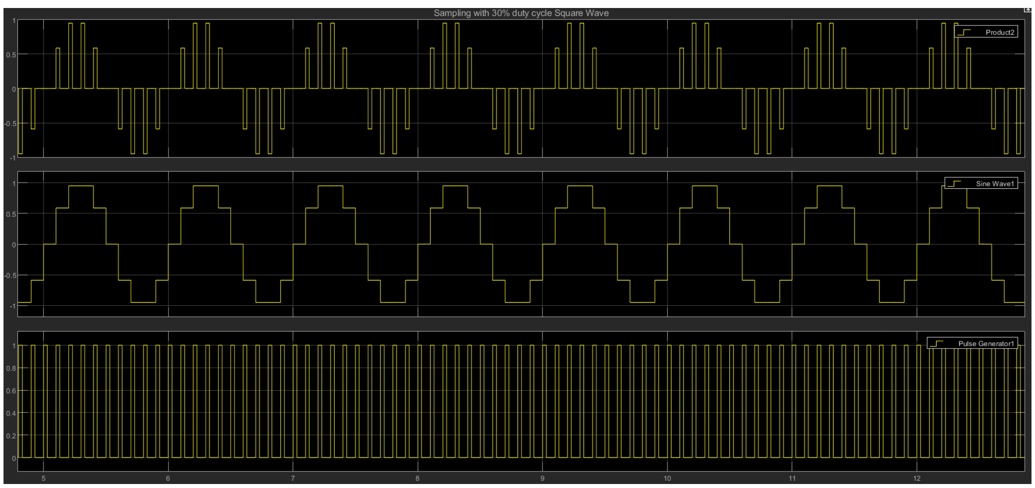


The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 61Hz

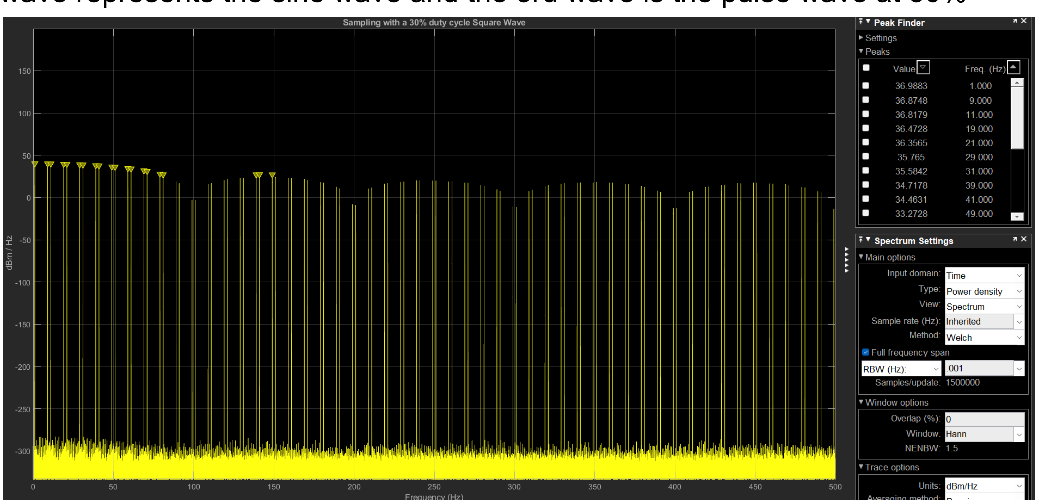
Sampling 30%

Sampling with 30% duty Cycle Square wave

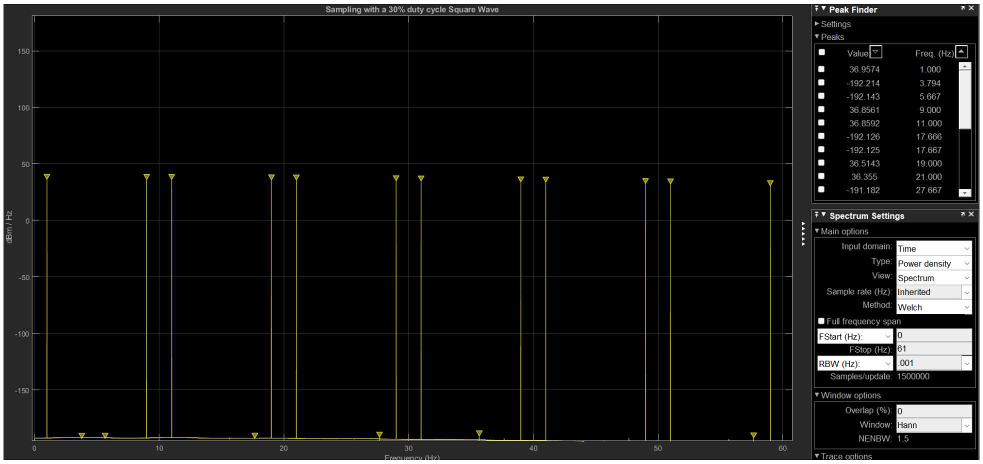




The 1st wave represents the product between the sine wave and the 30% pulse wave. The 2nd wave represents the sine wave and the 3rd wave is the pulse wave at 30%



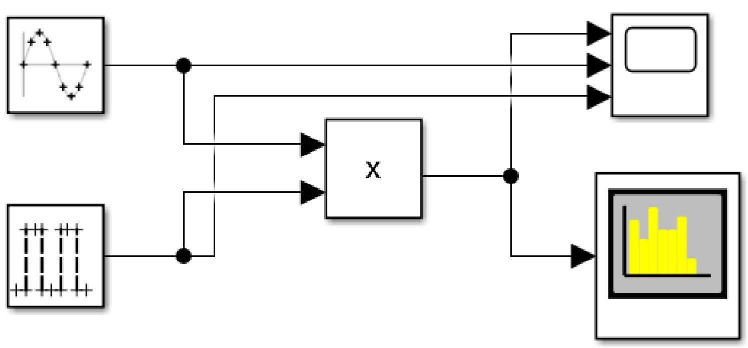
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 500Hz

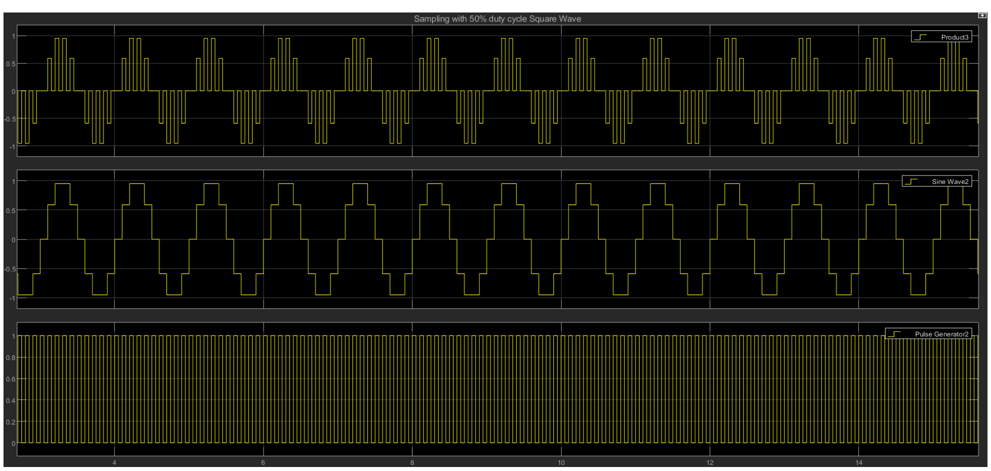


The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 61Hz

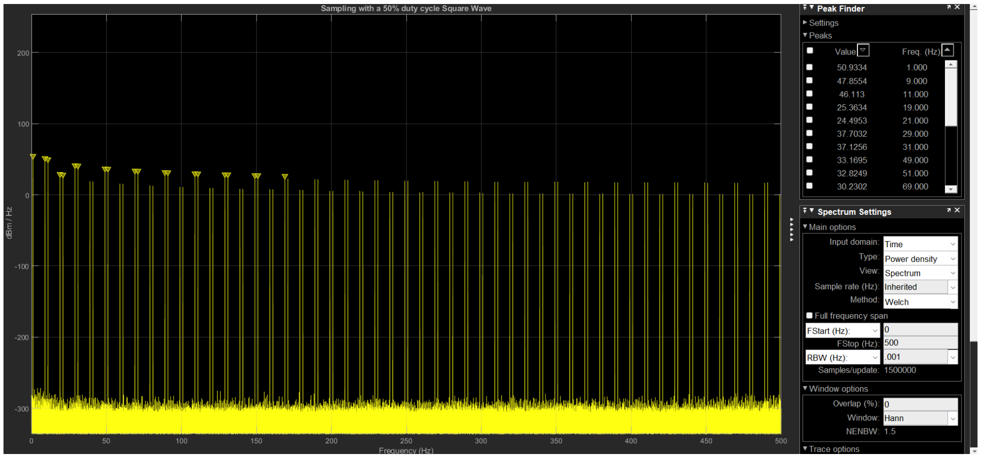
Sampling 50%

Sampling with 50% duty Cycle Square wave

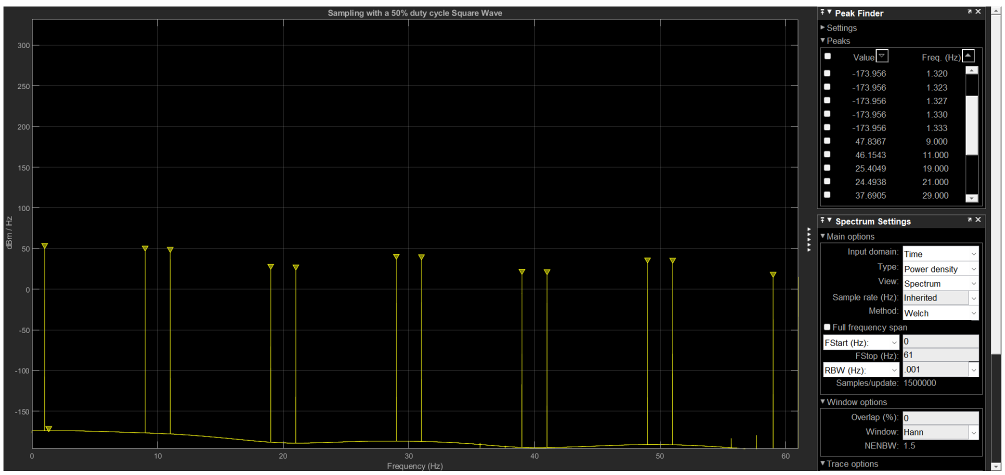




The 1st wave represents the product between the sine wave and the 50% pulse wave. The 2nd wave represents the sine wave and the 3rd wave is the pulse wave at 50%



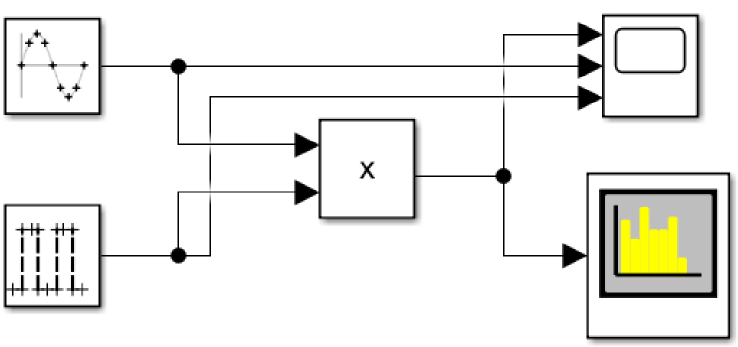
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 500Hz

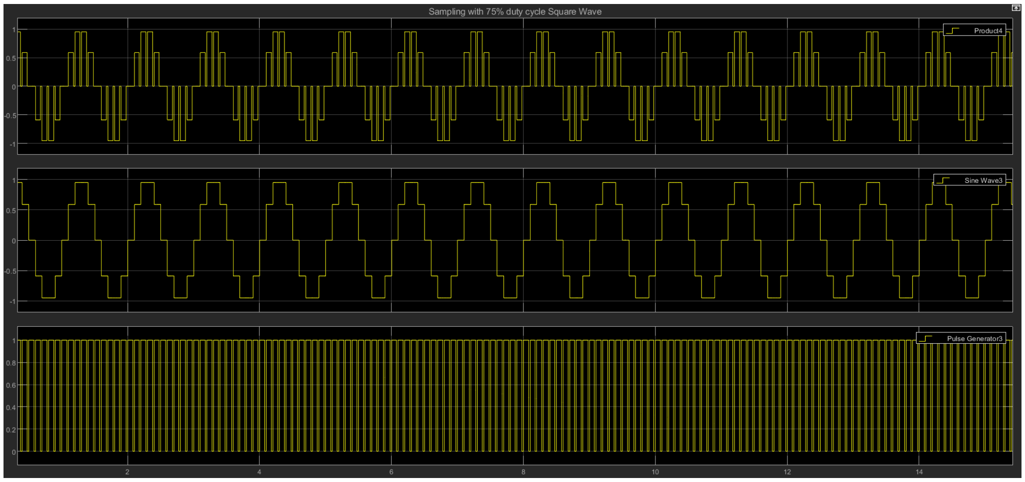


The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 61Hz

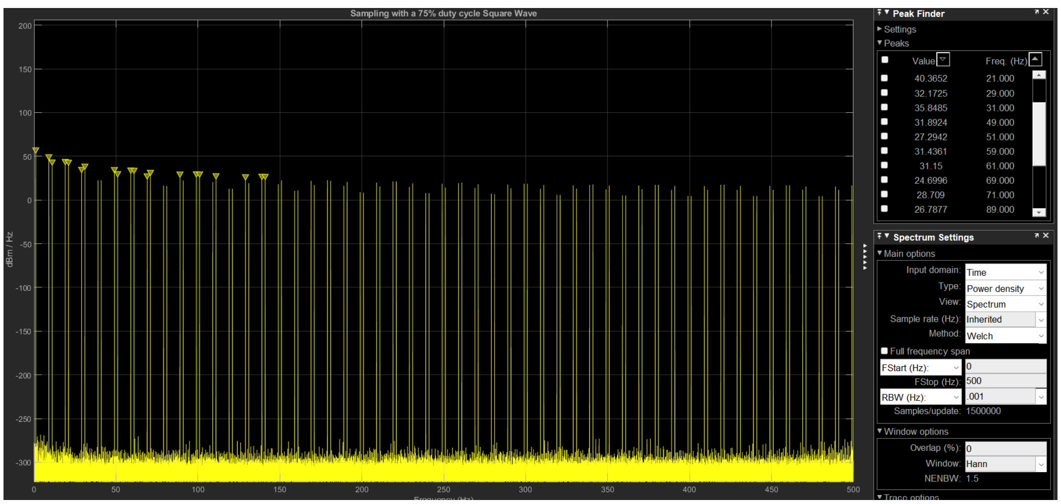
Sampling 75%

Sampling with 75% duty Cycle Square wave

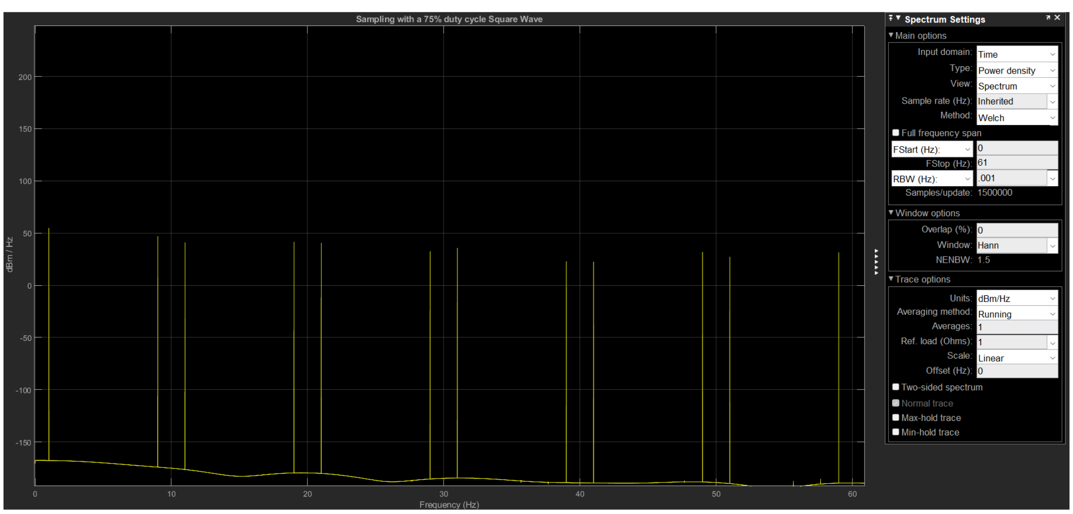




The 1st wave represents the product between the sine wave and the 75% pulse wave. The 2nd wave represents the sine wave and the 3rd wave is the pulse wave at 75%

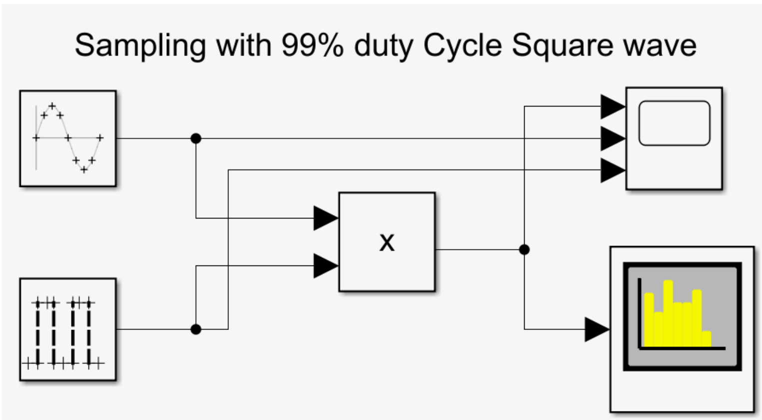


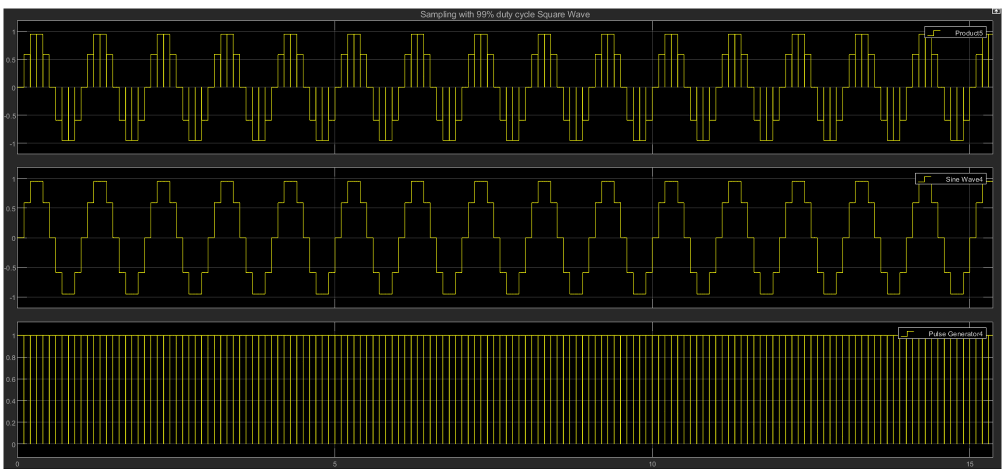
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 500Hz



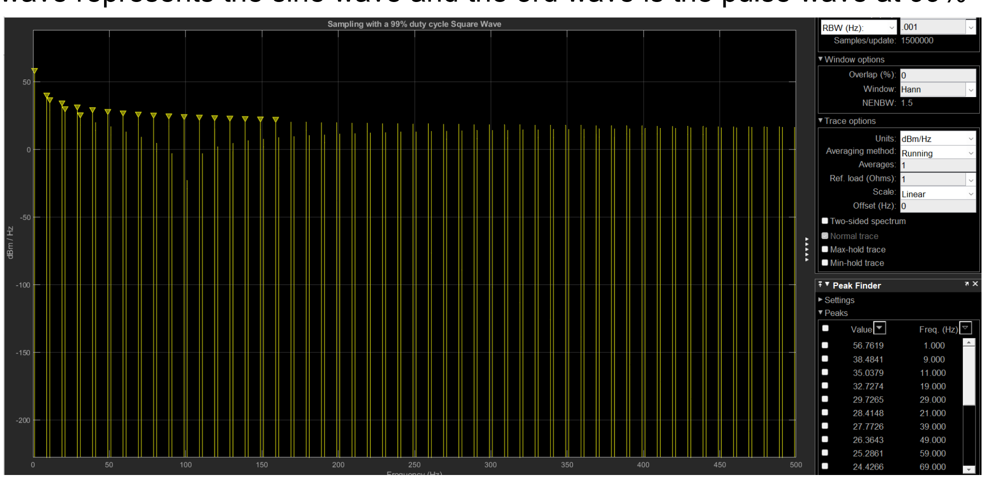
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 61Hz

Sampling 99%

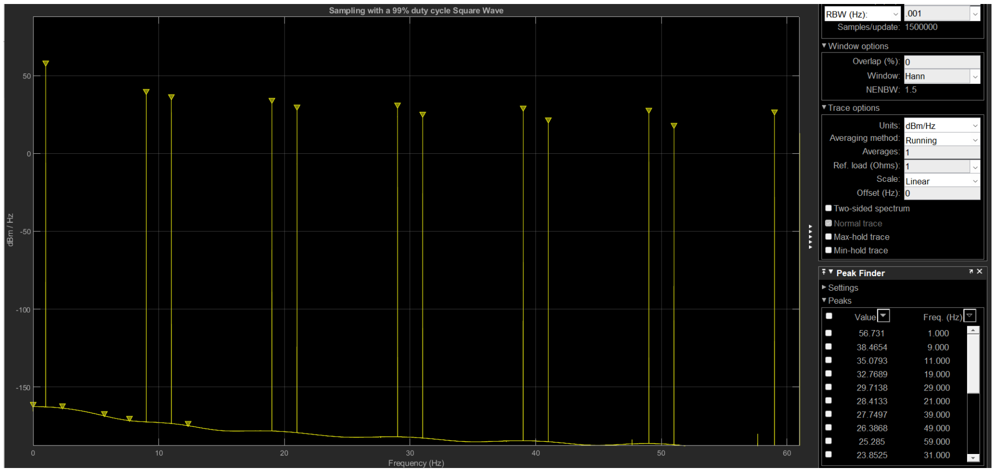




The 1st wave represents the product between the sine wave and the 99% pulse wave. The 2nd wave represents the sine wave and the 3rd wave is the pulse wave at 99%



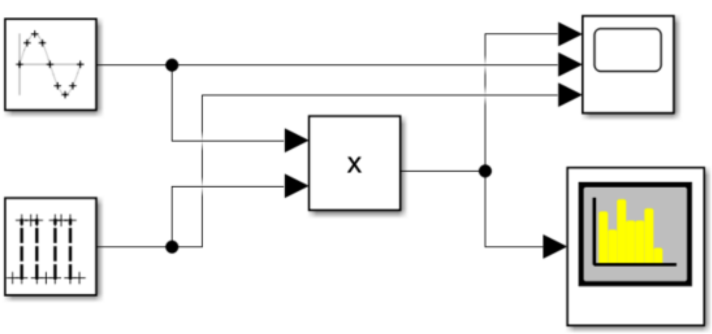
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 500Hz

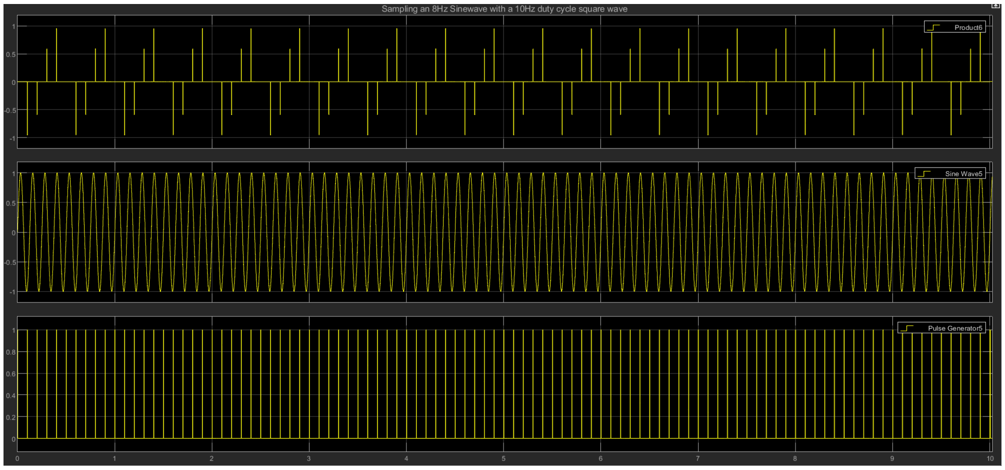


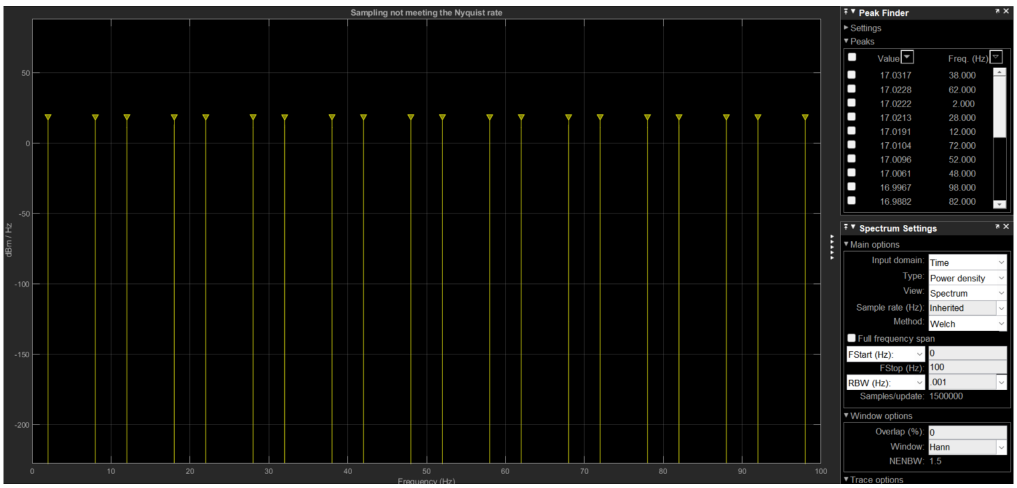
The following is the result of the frequency and the dBm in the spectrum anaulyzer when from DC to 61Hz

Sampling not meeting the Nyquist rate

Sampling an 8Hz Sinewave with a 10Hz duty cycle Square wave

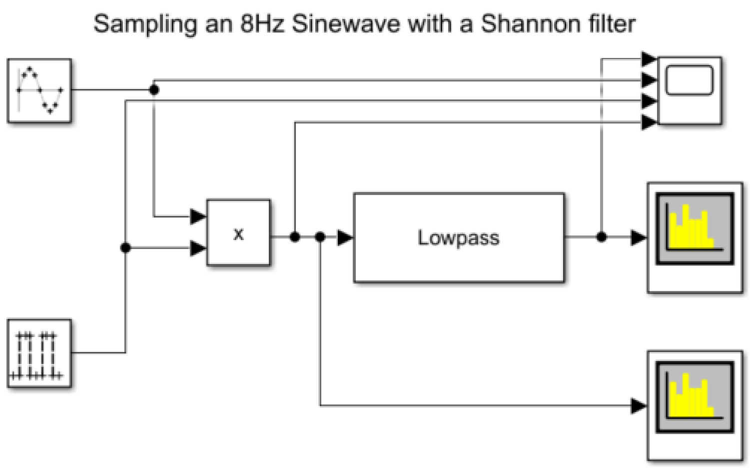


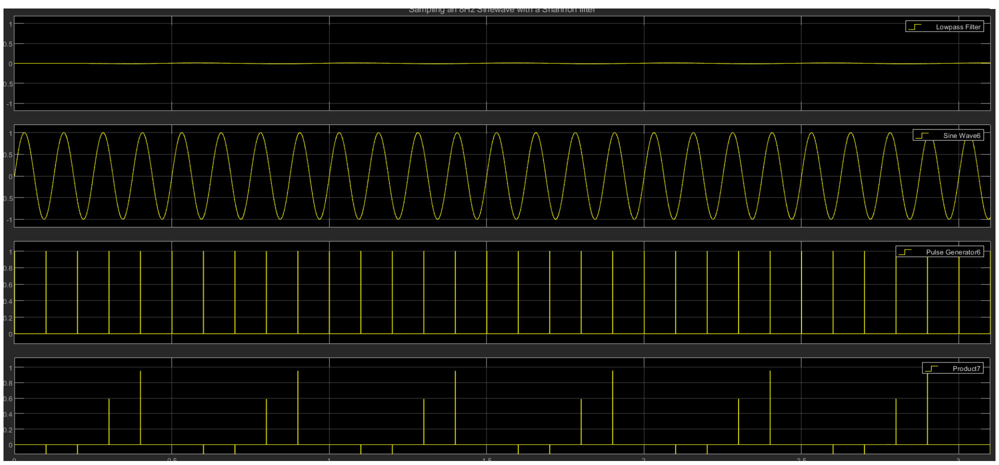


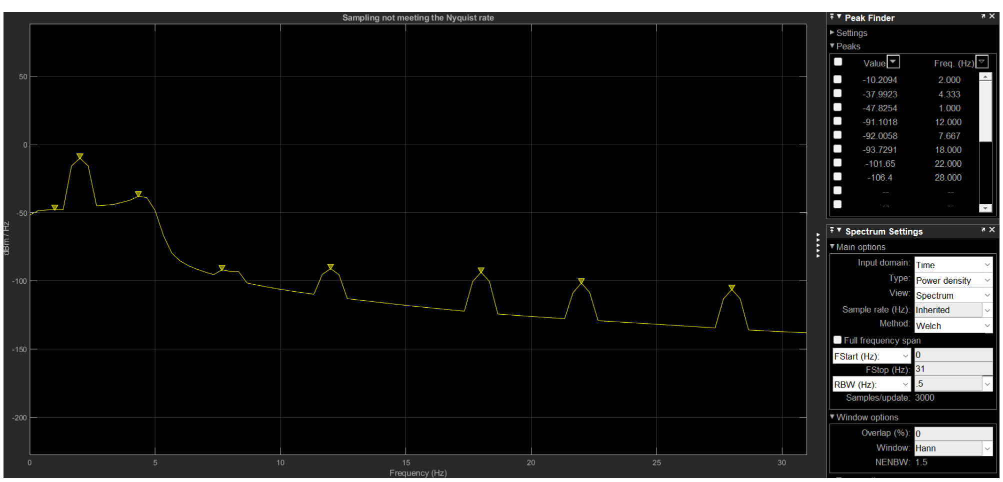


Recovering the 8Hz sampled Sinewave with a Shannon filter

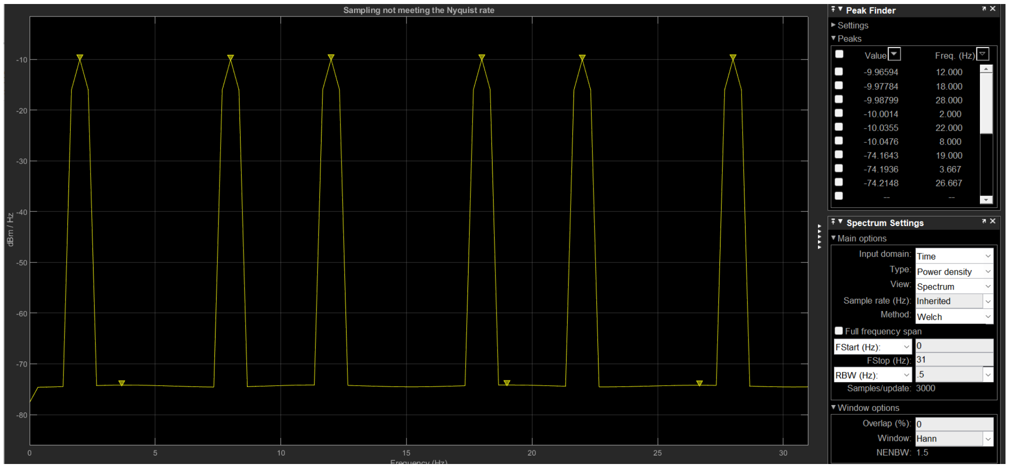
We will set the filter corner frequency to 5Hz, half of the 10Hz sampling rate.


Sampling an 8Hz Sinewave with a Shannon filter
x
Lowpass

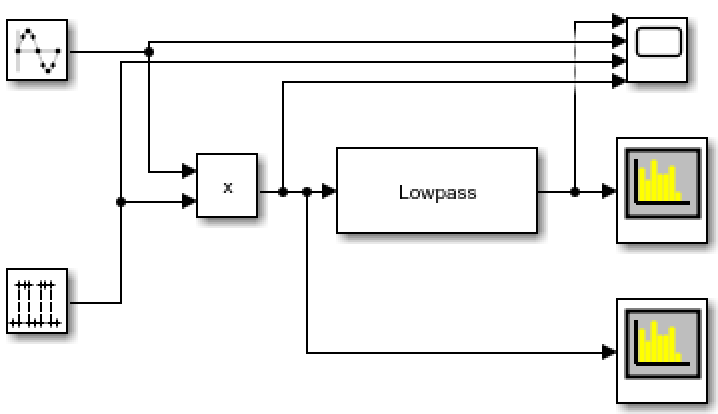

Lowpass Filter
Sine Wave6
Pulse Generator6
Product7


Sampling not meeting the Nyquist rate
Frequency (Hz)
dBm / Hz
Peak Finder
Settings
Peaks
Value
Freq. (Hz)
-10.2094 2.000
-37.9923 4.333
-47.8254 1.000
-91.1018 12.000
-92.0058 7.667
-93.7291 18.000
-101.65 22.000
-106.4 28.000
Spectrum Settings
Main options
Input domain: Time
Type: Power density
View: Spectrum
Sample rate (Hz): Inherited
Method: Welch
Full frequency span
FStart (Hz): 0
FStop (Hz): 31
RBW (Hz): .5
Samples/update: 3000
Window options
Overlap (%): 0
Window: Hann
NENBW: 1.5

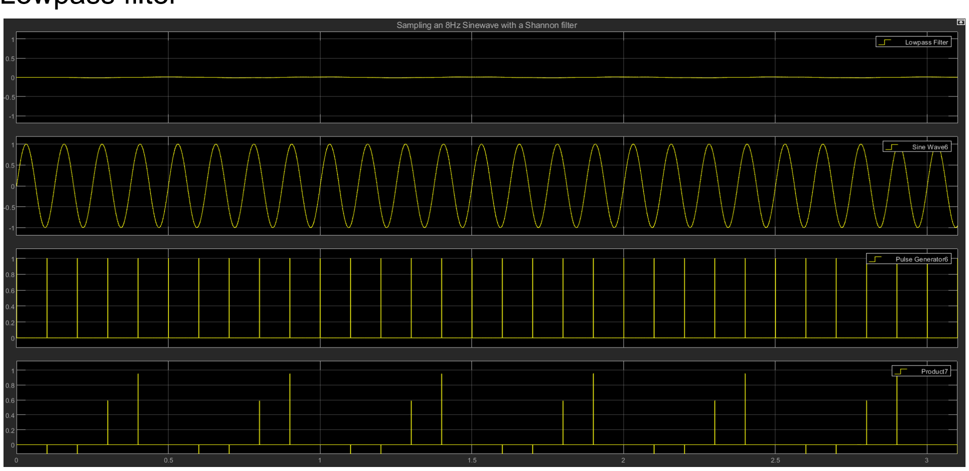
The product

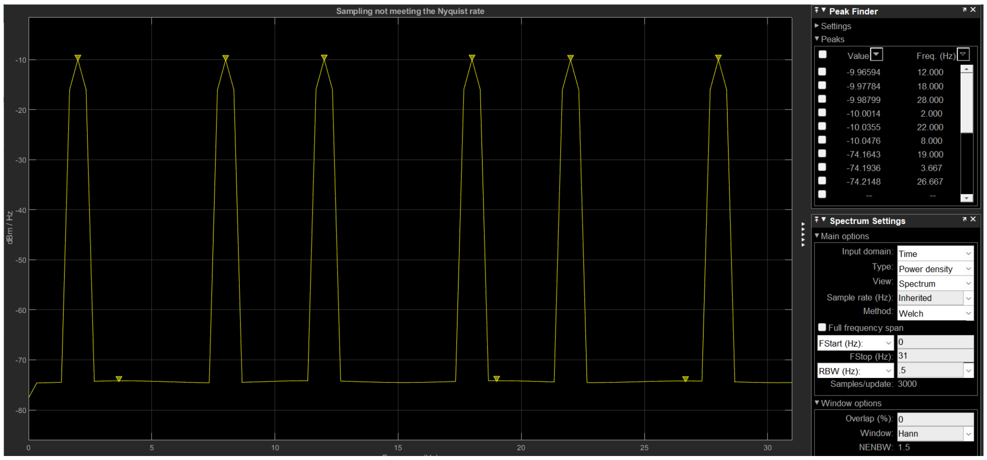


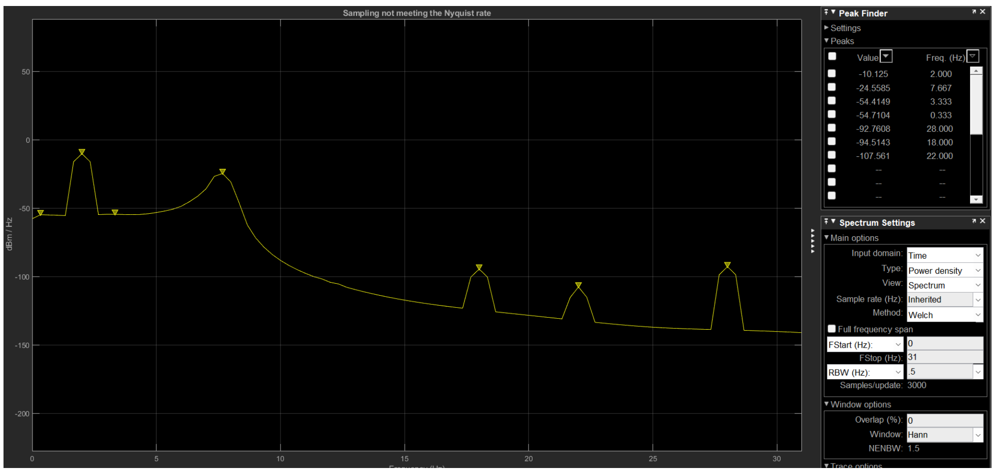
We will set the filter corner frequency to 8Hz, half of twice the 8Hz sampled signal.



Lowpass filter







conclusion

In the following lab we understood the importance of sampling the the anolog frequncy correctly in the first sampling we haw how to sample we provide the same signe sinal and we used different square waves to digitize the signal, we were able to see that when the pulse signal was more thinner close so the theoretincal pulse wave when sampling it was much cleaner. We al recorginize that when the we use Recovering the 8Hz sampled on a 10Hz dity cycle the sampling didnt comply to Nyquist rate and we recontize that alot of the signal got lost when reconstructing the signal.