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Experimental Study on Sampling Theorem in Signal Processing

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Abstract

This practicum is to define the study properties of the sampling theorem. Understand the effect of selecting the sample size and its effect on the signal recovery process. The experiment utilizes a computer or portable workstation to run an examination of hypothesis reenactment program. From the test information gotten, it can be concluded that the more noteworthy the frequency of the signal to be inspected, the closer the signal will be to the initial signal. The time and frequency of the examining signal are conversely relative. The higher the frequency, the lower the time will be. The magnitude of the amplitude of the output signal is indeterminate.

Introduction

Research by Tlemcani et.al (2013) stated that a continuous signal time x (t) is sampled at a frequency (Hz) to produce a sample signal xs (t). We model xs (t) as an impulse with the area and the impulse given by x (nTs). An ideal low-pass filter with frequency is used to obtain the reconstructed signal xr (t). The digital signal processing takes a slightly different form (Casey & Christensen, 2015). The main component of this system is a digital processor capable of working when the input is a digital signal. For an input in the form of an analog signal, an initial process called digitization is needed through a device called an analog-to-digital converter (ADC), where the analog signal must go through a process of sampling, quantizing and coding. By estimating the highest-frequency component in x (t) at the frequency fm. Then the Theorema sampling states for fs> 2fm no loss of information on sampling. In this case, selecting fc in the range fm <fc <fs-fm gives xr (t) = x (t). This result can be understood by Fourier's test changing the form X (f), Xs (f) and Xr (f). If fs <2fm or fc is poorly chosen, then xr (t) will not resemble x (t).

Methods

This experiment uses a computer or laptop to run a sampling theorem simulation program which can be downloaded from internet. Practicum Procedure includes (1) Run the sampling program (2) Select the wave type (3) Shifting the sampling frequency to move the sampling line from the main signal. (4) Repeat step two for the second experiment.

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Result and Discussion

Experiment 1

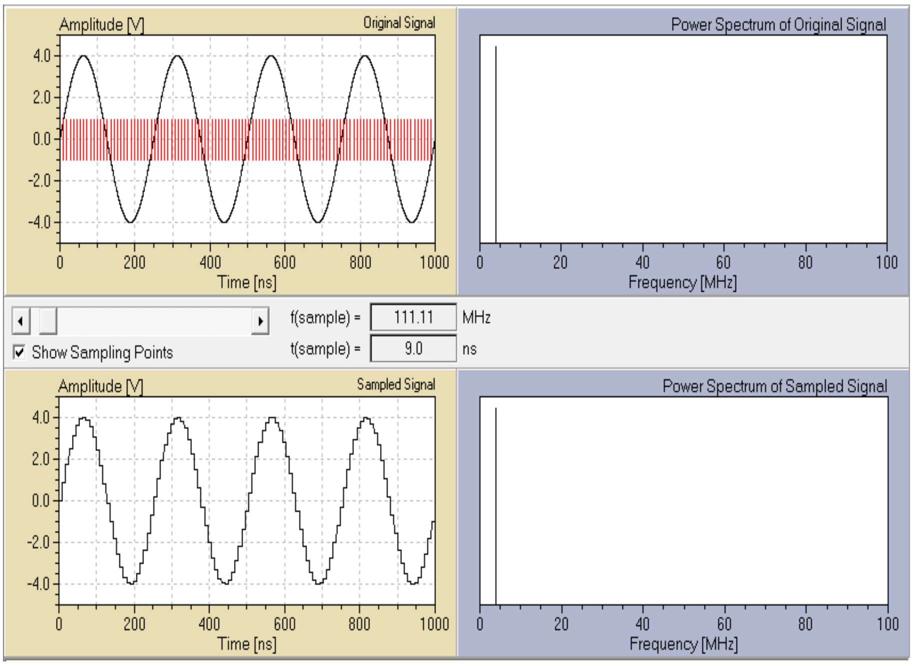

Figure 1.The simulation results of the sampling theorem 1

Where:

f = 111,11 MHz

t = 9,0 ns

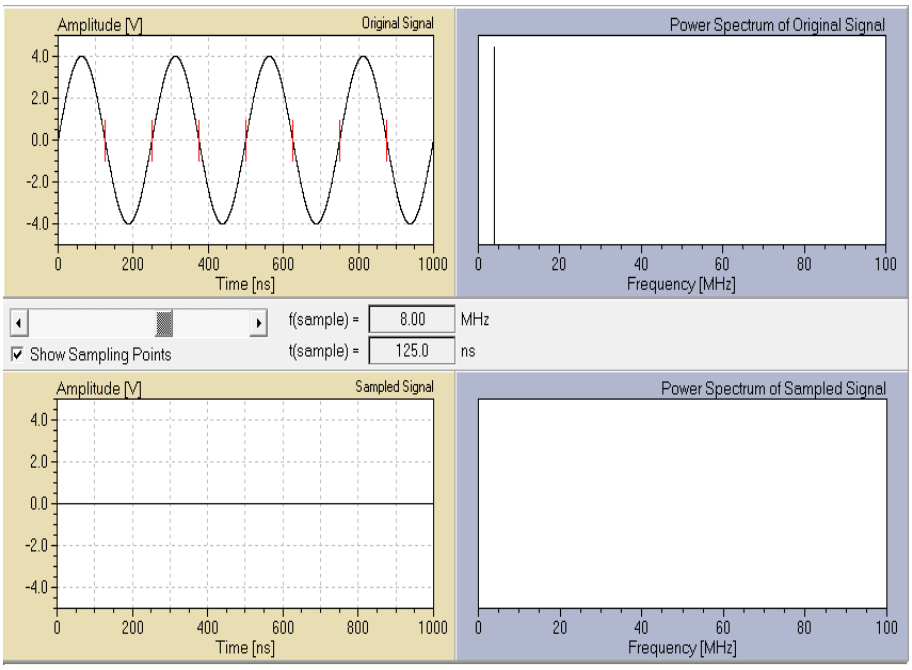

Figure 2. The simulation results of the sampling theorem 2

Where:

f = 8,00 MHz

t = 125,0 n

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Experiment 2

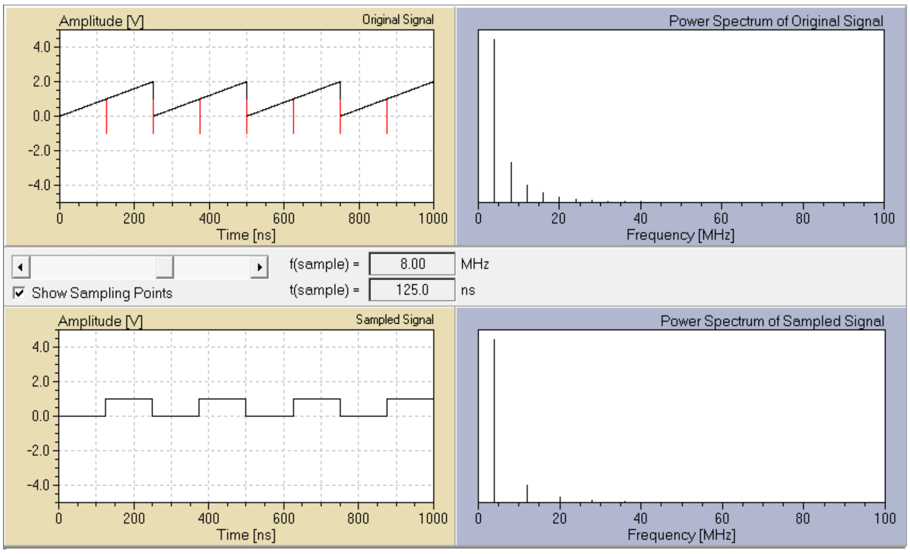

Figure 3. The simulation results of the sampling theorem 1

Where:

f = 8,00 MHz

t = 125,0 ns

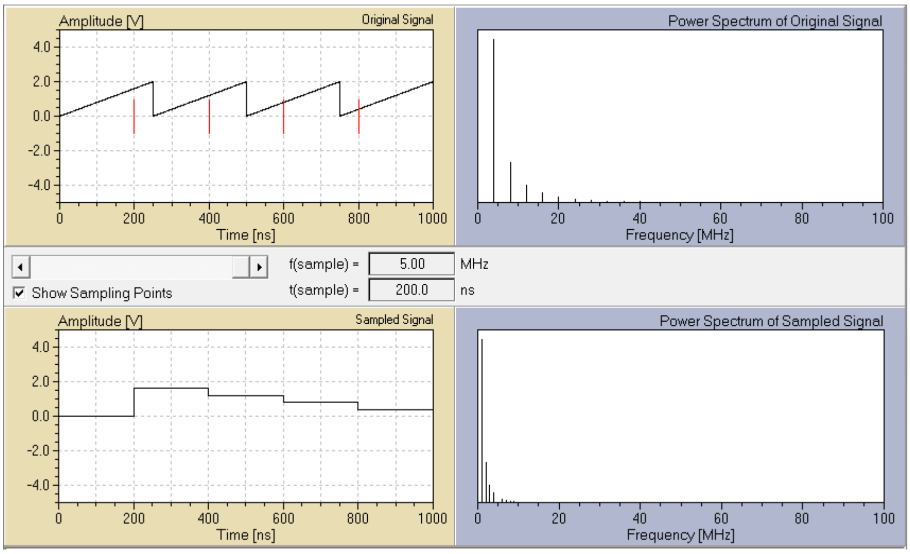

Figure 4. The simulation results of the sampling theorem 2

Where:

f = 5,00 MHz

t = 200,0 ns

According to Grafakos (2008) the sampling theorem is part of the Fourier analysis, where the discrete time system signal is obtained from the continuous time system signal. The main purpose in this process is so that information is not lost at all. If the continuous-time signal is a finite band, which does not contain other signals above a certain frequency, this can be done

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by passing the signal through a low pass filter, such as sound waves transmitted on a wired telephone, where the frequency limit is above 4KHz.

A continuous signal time x (t) is sampled at a frequency (Hz) to produce a sample signal xs (t). We model xs (t) as an impulse with the area and the impulse given by x (nTs) (Brémaud, 2013). An ideal low-pass filter with frequency is used to obtain the reconstructed signal xr (t). The digital signal processing takes a slightly different form. The main component of this system is a digital processor capable of working when the input is a digital signal. For an input in the form of an analog signal, an initial process called digitization is needed through a device called an analog-to-digital converter (ADC), where the analog signal must go through a process of sampling, quantizing and coding.

In this experiment, the aim is to determine the properties of the sampling theorem, the effect of selecting the number of samples and its effect on the signal recovery process and to understand the exact sample value of a signal. In this experiment, the tools and materials used are a laptop to run the sampling theorem simulation program. The work step in this lab is to run a sampling program, then select the type of wave to be used. Next, shift the sampling frequency shift to move the sampling line from the main signal. Then repeat this step for the second experiment with a different signal (Abreu, 2005).

For the first experiment using a sine signal (Kuo & Tian, 2001). In the first signal sample using the sample signal frequency and time. The frequency used is 8.00 MHz and the time used is 125.0 ns. It can be seen in the picture that the sample points are more and more tighter. And the resulting output signal almost resembles a sinusode signal, only jagged or can be said to almost resemble the original signal.

For the second signal, in the first experiment, still using the sine signal but with a different frequency and time. The frequency used is 5.00 MHz with the time used is 200.0 ns. It can be seen in the experimental image that the sampling point of the signal remains in the middle of the wave. This results in no output signal. The amplitude of the output signal is zero.

Then for the second experiment, just like the first experiment, only the signal used is different. In the second experiment using a sawtooth signal. The first sample signal uses a frequency of 8.00 MHz and the time used is 125.0 ns. It can be seen in the experimental image, when the sampling point is right in the half wave, the output signal will be constant.

For the second signal, in the second experiment, the sawtooth signal was still used but with a different frequency and time. The frequency used is 5.00 MHz with the time used is 200.0 ns. It can be seen in the experimental image that the sampling point of the signal is looser than the first signal in the second experiment, so that it produces an output signal like the experimental data image.

From the experimental data obtained, it can be concluded that the greater the frequency of the signal to be sampled, the closer the signal will be to the original signal. The time and frequency of the sampling signal are inversely proportional. The higher the frequency, the lower the time. The magnitude of the amplitude of the output signal is indeterminate (varies). The power spectrum of the original signal does not change, which changes only in the sampled signal. And the greater the sample time, the more irregular the resulting output signal will be. From the journal "Sampling Reduction Methods and the Use of Multiple Sampling Frequencies for Fourier Digital Transformation Analysis in Microcontroller-Based Applications", Fourier analysis is one method that is widely used for various applications (Cabal-Yepez et.al, 2012) . DFT and FFT are models and algorithms commonly used in digital processing systems (Jacobsen & Lyons, 2003). The most common problems associated with DFT or FFT are computation time and memory capacity. If the processing device used is a PC, this is not a problem. However, this is not the case when using a microprocessor or microcontroller. The

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technique used here will greatly reduce computation time and memory requirements which will be suitable for microcontroller based applications. The technique used is to break the wide frequency spectrum of the input signal into several smaller bands by using more than one sampling frequency. And reduce the sampling length from hundreds or thousands of samples to only a few tens of samples.

Conclusion

From the data from practicum results about sampling theory, it can be concluded that: (1) When sampling exactly half a wave, the amplitude of the sampling signal is zero. Can be seen in the experimental results image. (2) The greater the sample time, the more irregular the resulting output signal will be. (3) If the large frequency signal on the signal to be sampled is getting bigger, the results of the sampling signal will be boxes with a smaller size. (4) Period is inversely proportional to frequency. The greater the value of the sampling signal period, the more tenuous the sampling signal is. (5) It can be seen in the experiment even though the sampling signal is different, namely the sine signal and the sawthooth signal, the results of the sampling signal are still square. (6) In the journal on the sampling theorem, it can be proven that the technique used is to split a wide frequency spectrum of the input signal into several smaller bands by using more than one sampling frequency. And reduce the sampling length from hundreds or thousands of samples to only a few tens of samples

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