

ECE280 - Lab 5: Sampling and Aliasing

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I have adhered to the Duke Community Standard in completing this assignment.

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1 Objectives

1. Use Simulink and a function generator to study aliasing
2. Use Simulink and a function generator to study anti-aliasing
3. Analyze a real world application anti-aliasing systems
4. Create a MATLAB function that produces samples of simple sinusoidal signals

2 Background

Sampling is the process of reducing a continuous time signal into a discrete time signal. Due to sampling, some information about the continuous time signal is lost which can result in aliasing, a state where different signals in discrete time become indistinguishable from each other.

In telephone systems, the input signal (human voice) passes through a low-pass filter with cut-off frequency of 4000 Hz (human voice is 3.5 kHz but approximated to 4 kHz to avoid losing information). The continuous time signal is then sampled at 8 kHz, abiding by the Nyquist criterion and effectively preventing aliasing. This system is known as anti-aliasing system.

Another example for when a continuous time signal is processed digitally is in MP3 audio players. When recording in an MP3, the audio continuous time signal is input and then converted to a digital signal via sampling before being digitally processed to encode and save the file.

3 Results and Discussion

- Explain the purpose of each block in the **Sampling_wAliasing** model in Exercise #1.
The audio device reader block allows for taking the sinusoidal signal from the function generator as input. The down sample block with value of 6 reduces the analog signal sampling rate by a factor 6 from 48 kHz to 8 kHz. The subsystem block samples the input signal at 48 kHz. The audio device writer plays the audio of the input signal. The spectrum analyzer block is used to display the frequency spectra of the input signal.
- Using the plots generated by your Matlab simulation, explain how the system without anti-aliasing works and why it behaves the way it does.
The system without anti-aliasing works by creating a copy of the signal being processed every 4000 Hz. The frequency at 100 Hz is the frequency at the rising side of the triangle, where as 7900 is the same point, but at falling side of the copy of original signal. This is also the case with the signal at 200 Hz and 7800 Hz; 400 Hz and 7600 Hz; and 800 Hz and 7200 Hz as proven by figures 1-4: the plot at the right hand side is the same as the one on the left hand side multiplied by a factor of -1.
- Describe, in your own words, how the **Sampling_AntiAliasing** model from Exercise #2 differs from the **Sampling_wAliasing** model in Exercise #1. Why is the anti-aliasing filter necessary?
The anti-aliasing model in exercise 2 is very similar to the aliasing model in exercise 1. The only difference in design of the anti-aliasing model is that the input signal passes through a low pass filter which removes frequencies beyond the cut off frequency of 4000 Hz.
- Figure 5 shows the spectrum of the signal at all wanted points specified in the lab manual.

- Answer all questions in **bold** asked in the instructions.

Exercise 1 Questions

- What do you expect to hear? Is this actually the case?
A musical tone is expected, since the input signal is sinusoidal and based on experience in previous laboratories. Yes, an audible tone was heard.
- What do you hear/see at the output when the frequency of the sinusoidal signal is 200 Hz? Make similar observations when the frequency is 400 Hz and 800 Hz. Describe what you hear and explain why this is what you would expect.
At frequency 100 Hz, the tone is constant at a low pitch. At frequency 200 Hz, the tone is constant at a higher pitch. At 400 Hz and 800 Hz the tone is constant and a very high pitched. This is expected because the pitch increases with increasing frequency.
- What do you hear/see at the output when the frequency of the sinusoidal signal is 7900 Hz, 7800 Hz, 7600 Hz, and 7200. Describe what you hear and explain why this is what you would expect.
The tone at 7900 Hz sounds the same as the tone at 100 Hz. The tone at 7800 Hz sounds the same as the tone at 200 Hz. The tone at 7600 Hz sounds the same as the tone at 400 Hz. The tone at 7200 Hz sounds the same as the tone at 800 Hz. This is because the maximum frequency that will pass through without aliasing is 4000 Hz; the input signal is repeated every 4000 Hz
- What is the maximum frequency that will pass through the system without aliasing?
4000 Hz is the maximum frequency that will pass through the system without aliasing.
- What do you observe about the quality of the output signal? Why is this the case?
When the wave of the provided sweep file passes through the filter, it generates a tone that rises in pitch/frequency. The tone starts at 0 Hz rises till 4000 Hz, then decays back down 0; rises back to 4000, then decays to 0. This cycle keeps repeating. This is expected because the input signal repeats at frequencies that are multiples of 4000 Hz due to this being an aliasing system.

Exercise 2 Questions

- What is the cut off frequency of the filter? Why is this frequency appropriate for the telephone system application? Generally, how should the cut off frequency of an anti-aliasing filter be chosen?
The cutoff frequency is 4000 Hz. 4000 Hz is an appropriate cut off frequency for telephone system applications to allow long distance transmission and because it meets the Nyquist criterion as it's half the sampling frequency.
- Describe what you hear/see when the frequency is 100 Hz, 200 Hz, 400 Hz, and 800 Hz; explain why this is what you would expect.
The tones at 100 Hz, 200 Hz, 400 Hz, and 800 Hz in the anti-aliasing system sound the same as the tones at 100 Hz, 200 Hz, 400 Hz, and 800 Hz (respectively) in the aliasing system.
- What do you hear/see at the output when the frequency of the sinusoidal signal is 7900 Hz, 7800 Hz, 7600 Hz, and 7200. Describe what you hear and explain why this is what you would expect.
Almost nothing is heard at 7900 Hz, 7800 Hz, 7600 Hz, and 7200 Hz. This is expected because nothing should be heard beyond the cut off frequency (4000 Hz) in an anti-aliasing system.
- What is the maximum frequency that will pass through the system undistorted? How does this compare to your answer in Exercise 1?
4000 Hz is the maximum frequency that will pass undistorted. No frequency is audible beyond 4000 Hz, which is the contrary to what was observed in exercise 1, where the tones repeat beyond 4000 Hz.
- What do you observe about the quality of the output signal? Why is this the case? How does this differ from exercise 1.
When the wave of the provided sweep file passes through the filter, it generates a tone increases in frequency till 4000 Hz beyond which the tone cuts off. After a few seconds, the tone comes back, but cuts off again after reaching 4000 Hz before coming back again. This cycle keeps repeating.

This is expected because nothing should be heard. beyond 4000 Hz in an anti-aliasing system. This is different from exercise 1 where the tone would repeat beyond 4000 Hz. In addition, the quality of the sound in the anti-aliasing system is better.

- Based on what you observed, discuss why the performance of the telephone system would degrade significantly if anti-aliasing pre-filtering were not used. Explain how this filtering prevents these negative effects.

The performance of the telephone system would degrade significantly if anti-aliasing pre-filtering were not used, because the telephone systems would pick up undesired signals of waves with high frequencies, such as radio waves. This would lower the quality of the sound heard. An anti-aliasing filter cuts off the high undesired frequencies that are beyond the cut off frequency.

4 Conclusions

In this laboratory, the difference between aliasing and anti-aliasing system models was examined using Simulink models. Furthermore, an understanding of where/how anti-aliasing filters are used in the real world was gained. Lastly, MATLAB code that samples sinusoidal signals was written.

5 Extension

Anti-aliasing systems use low pass filters (LPFs). What are LPFs; what components make them up; and how is the cut off frequency set? The purpose of this extension is to study the preceding questions.

Low pass filters are circuits that allow input signals with frequencies below the cut off to pass while rejecting higher frequencies, hence called low pass filter. This type of circuit creates the described effect by utilizing properties of resistors, inductors, and/or capacitors together. Figure 6 and 7 show a simple RC and RL LPFs. The cut off frequency of an LPF is determined by the frequency at which the output signal is $.707$ ($1/\sqrt{2}$) times the input signal. Although LPFs theoretically sharply reject any frequency beyond the cut off, practical LPFs attenuate higher frequency signals (the further the frequency of the input from the cut off the greater the attenuation).

References

- <https://www.watelectronics.com/what-is-a-low-pass-filter-circuit-its-working/>
- <https://www.allaboutcircuits.com>

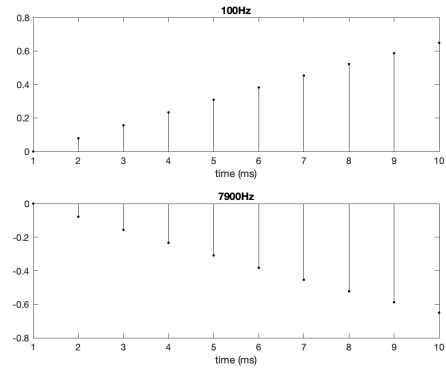


Figure 1: Sampled signal at 100 Hz and 7900 Hz

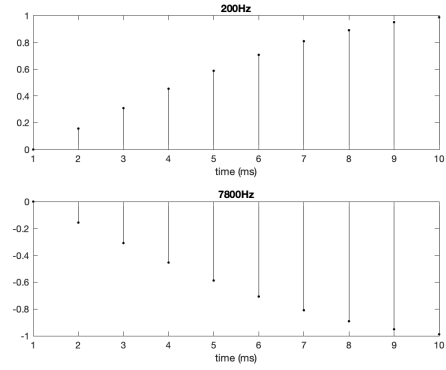


Figure 2: Sampled signal at 200 Hz and 7800 Hz

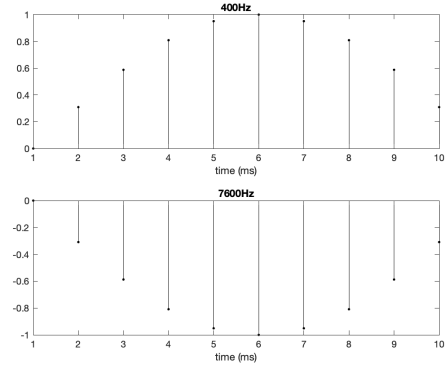


Figure 3: Sampled signal at 400 Hz and 7600 Hz

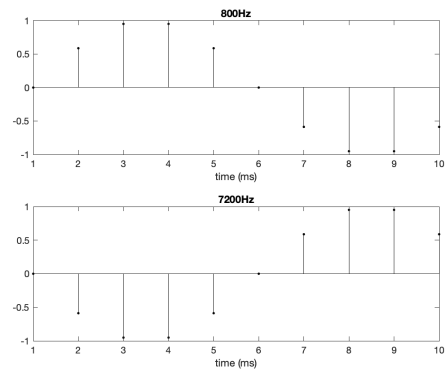
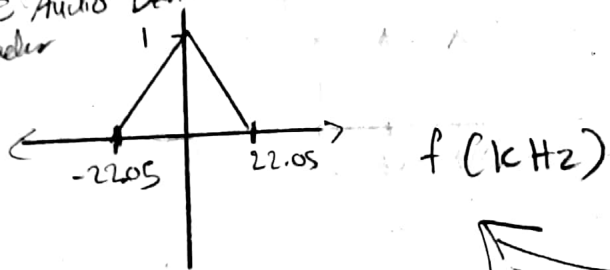


Figure 4: Sampled signal at 800 Hz and 7200 Hz

Sampling w/ aliasing model:

Before Audio Device Reader

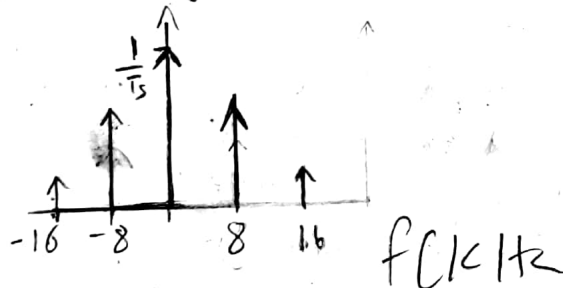


Sampling w/ anti-aliasing

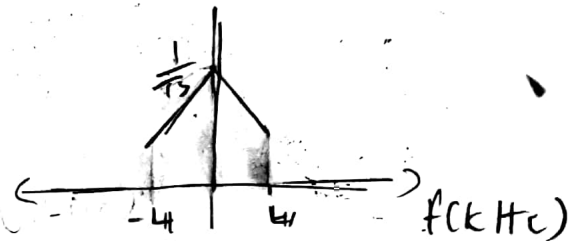
* Before Audio Device

Same

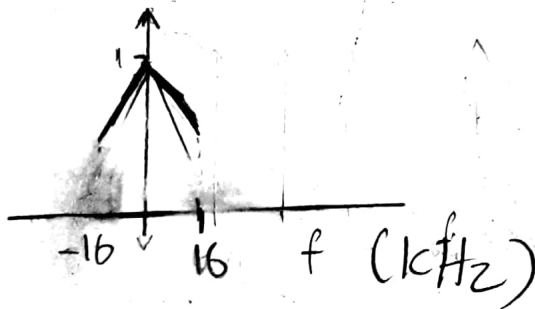
* After Downsampling



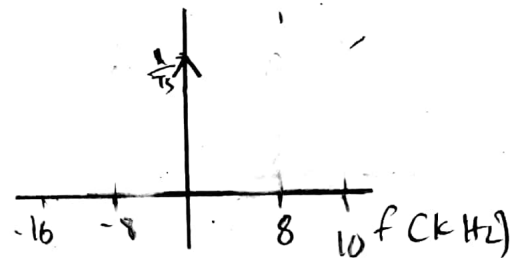
* After LPF



* After Audio Device Writer



* After Down samples



* After Audio Device Writer

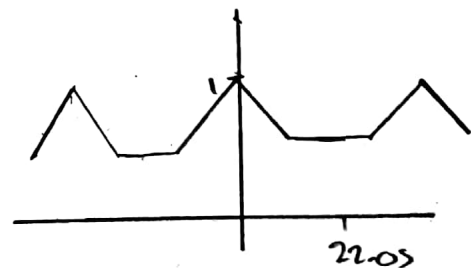


Figure 5: Frequency spectrums

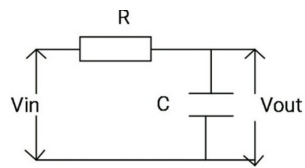


Figure 6: Simple RC, LPF filter

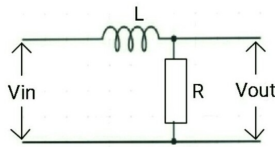


Figure 7: Simple RL, LPF filter