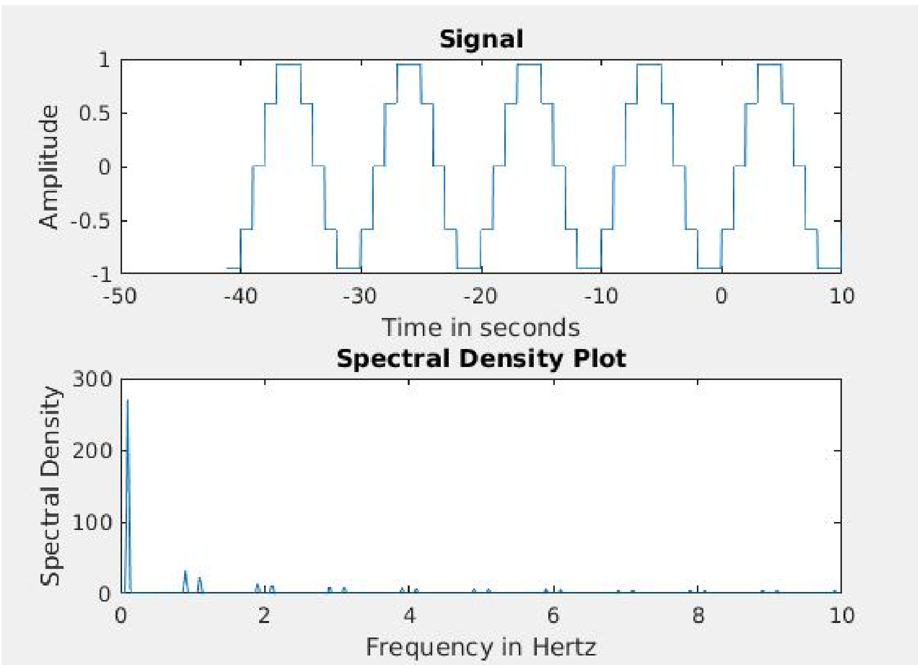
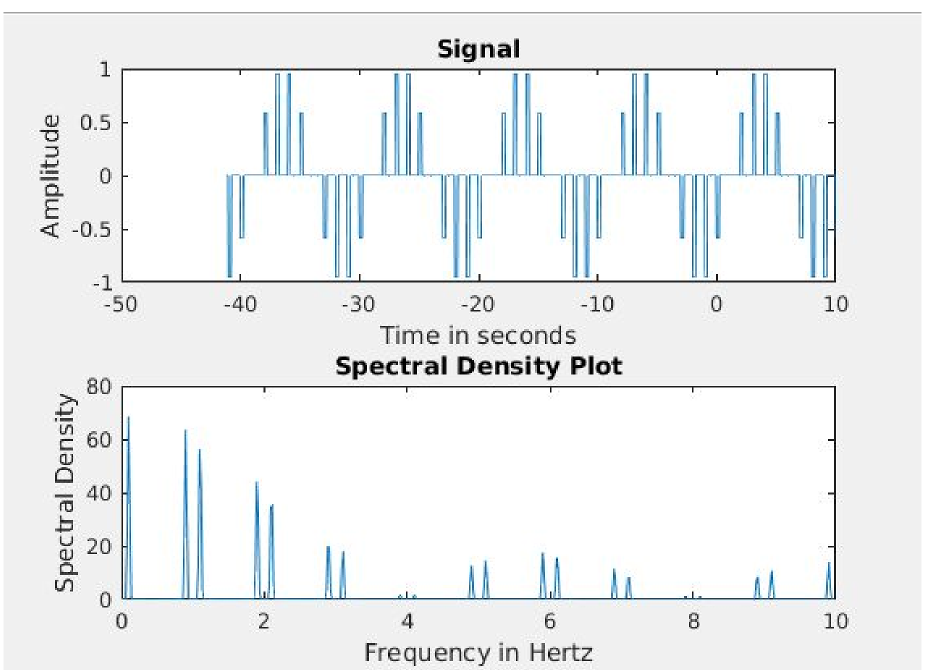
ECE 438 Lab 4b
David Dang & Benedict Lee

1 Discrete-Time Interpolation

INLAB REPORT: Submit your plot of signal c and its frequency spectrum. Circle the aliased components in your plot.

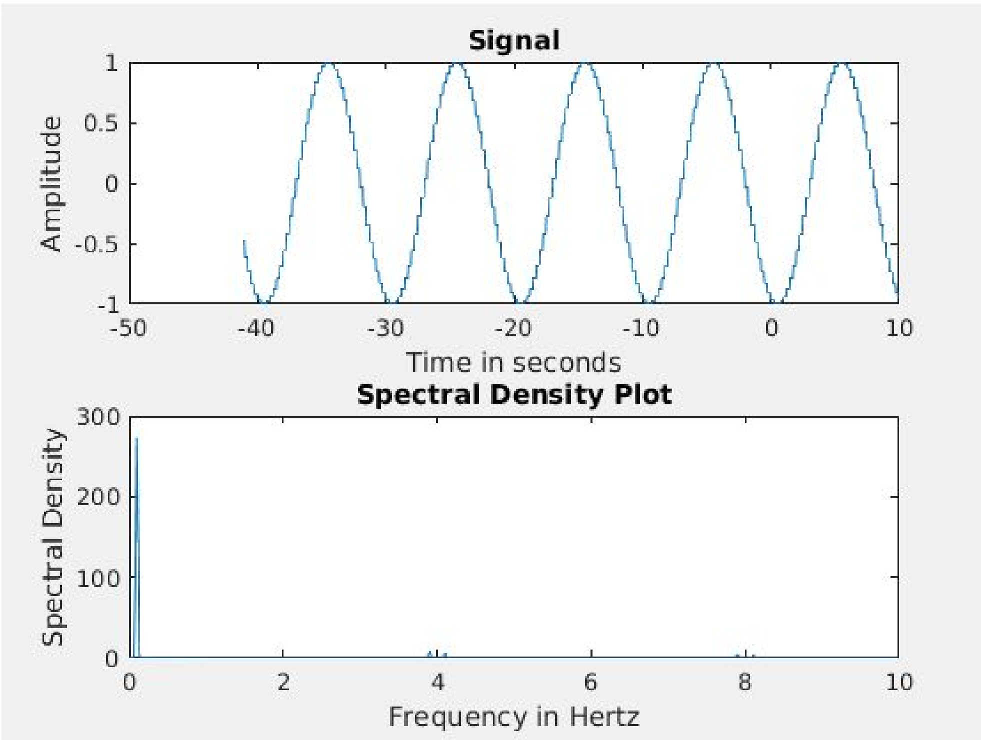


INLAB REPORT: Submit your plot of signal c and its frequency spectrum. On your frequency plot, circle the first aliased component and label the value of its center frequency. Comment on the shape of the envelope of the spectrum.



- Center frequency is 1 Hz. The envelope of the spectrum is an absolute sinc signal with zero-crossings every 4k Hz, where k is the range of natural numbers. Also, its amplitude is scaled by a factor of ¼ with respect to the original spectrum envelope.

INLAB REPORT: Submit your plot of signal c and its frequency spectrum. Give the values of the cutoff frequency and gain that were used. On your frequency plot, circle the location of the first aliased component. Explain why discrete-time interpolation is desirable before reconstructing a sampled signal.



- A cutoff frequency of pi/4 and gain of 4 was used to get the interpolated signal. Discrete-time interpolation is desirable before reconstructing a sampled signal because it negates the roll-off of imperfect low-pass filtering by spacing out the gap from the aliased component.

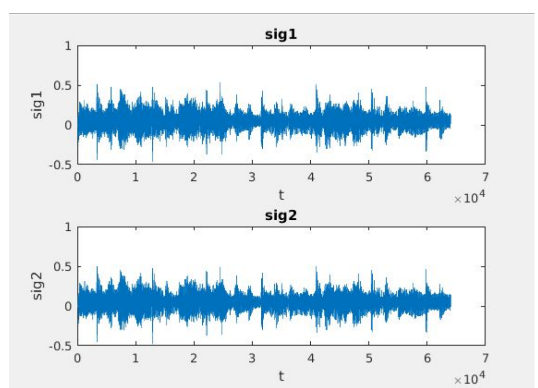
2 Discrete-Time Decimation

INLAB REPORT: Hand in the Matlab code for this exercise. Also, comment on the quality of the audio signal generated by using the two decimation methods. Was there any noticeable distortion in sig1? If so, describe the distortion.

```
audio_vector = audioread('music.au');
%sound(audio_vector)

sig1 = audio_vector(1:2:end);
%sound(sig1)

h = fir1(20, [1/2]);
filterOut = conv(audio_vector, h);
sig2 = filterOut(1:2:end);
sound(sig2)
```



- There could be distortion in sig1 due to the absence of the low pass filter, but qualitatively, the difference was not tangible to the ears.