**ECE 3512: SignalS – Continuous and Discrete**

# Recitation No. 11: Sampling Revisited

In this recitation, we will explore the impact of aliasing on sampling. Record a signal at 48 kHz using MATLAB, Audacity or any other audio tool. We will call this the original signal, .

(1) Using the MATLAB filter design tools, generate a low-pass filtered version of this system using a low-pass filter with a cutoff of 7 kHz and a steep attenuation (e.g. 60 dB down at 8 kHz). We will call this . Listen to this signal. Compute its averaged Fourier Transform to verify that the low-pass filter worked properly (use your code developed previously for the averaged FFT). Note that this signal is still sampled at 48 kHz.

(2) Using the interpolation function explored in Recitation 8:



compute new signals from  and  where fs = 4000 Hz, 8000 Hz, 16000 Hz and 32000 Hz. Make sure L is sufficiently long (apply what you learned in recitation 8).

Listen to these signals. Compute the averaged spectrum. Explain your results (hint: some of the signals will sound good while others will sound distorted – why?).

(3) Repeat (2), but this time low-pass filter the signal with an anti-aliasing filter before downsampling it (e.g., for the 4000 Hz case, use a 2000 Hz low-pass filter). Compute the averaged spectrum and compare each signal to the corresponding signal in (2).

(4) Record a signal at 8 kHz. Without altering the data, change the sample frequency to 16 kHz. Listen to the signal. Explain what you hear using concepts introduced in this class (e.g., frequency scaling).

(5) Generate a 900 Hz sinewave sampled at 10 kHz. Listen to it. Now add a second sinewave at 1100 Hz. Listen to the file again. Add another sinewave at 2900 Hz, and at 3100 Hz. Repeat this pattern out to 5 kHz.

It has been argued by some students that a sampled signal does not really have energy at higher frequencies as our analysis suggests. We would all argue the first signal could be sampled at 2 kHz. Sketch the spectrum of that sampled signal. How does that relate to the signal you constructed above? Why do the two sound so different?