1. (2 points) Define “discrete-time.” Discrete-time means a signal is discrete, or sampled, along its time dimension. Thus, the signal only has defined values at specific, countable points in time. Normally the sampling is with a constant period, $T$, such that $t = nT$, where $t$ is the continuous time variable and $n$ is any integer.

2. (1 point) Besides being discrete-time, which is the other key property of a digital signal relative to an analog signal? Quantized.

3. (4 points) Draw the basic DSP system block diagram including anti-alias and reconstruction filters, an ADC, and a DAC.

   \[ x(t) \rightarrow \text{[anti-alias filter]} \rightarrow \text{[analog/digital converter]} \rightarrow x(n) \rightarrow \text{[digital signal processing]} \rightarrow y(n) \rightarrow \text{[digital/analog converter]} \rightarrow \text{[reconstruction filter]} \rightarrow y_a(t) \]

4. (2 points) What is the purpose of an anti-alias filter? The AA filter is a lowpass filter that blocks high frequencies from entering the digital system. These high frequencies would be mistaken for lower frequencies due to not being sampled at enough points during a cycle; it is better to eliminate them than to mistake them for another frequency. For example if we sample at 2 Hz it turns out that (if we start at DC, which we do in this class) we can only represent frequencies up to 1 Hz. If a higher frequency (1.1 Hz, 1.5 Hz, 3.9 Hz, etc.) enters the digital system and is sampled, its phase increases by more than 180° between samples and its samples are equally explained by a lower frequency “alias” of the signal.

5. (1 point) What is an advantage of DSP over analog SP?
   a. It is portable (implement on various types of hardware, or in software)
   b. It is stable (no component drift or uncertainty)
   c. It enables complex designs (due to being precisely defined mathematically and having no analog component uncertainty)