$\qquad$

| Question: | 1 | 2 | 3 | 4 | 5 | 6 | Total |
| :--- | :---: | :---: | :---: | :---: | :---: | :---: | :---: |
| Points: | 8 | 11 | 12 | 10 | 5 | 4 | 50 |
| Score: |  |  |  |  |  |  |  |



Figure 1: A sequence $x[n]$ and its amplitude spectrum. The sequence $x[n]$ is not aliased and consists of 101 samples of a continuous signal $x_{c}(t)$.

Question 1 (8 points)
(a) [2 points] What is the time sampling interval $T$, in seconds?
(b) [2 points] What is the Nyquist frequency, in Hz ?
(c) [2 points] What is the frequency (in Hz ) of the periodic fluctuations in $x[n]$.
(d) [2 points] Label the frequency axis below the amplitude spectrum.
 Consider the moving average filter with system function $H(z)=\frac{1}{5}\left(z^{2}+z+1+\right.$ $\left.z^{-1}+z^{-2}\right)$.
(a) [2 points] Sketch the impulse response $h[n]$ of this system.
(b) [2 points] Use a well-known formula to rewrite $H(z)$ as the ratio of two polynomials in $z$.
(c) [2 points] For what two frequencies $\omega$ between 0 and $\pi$ is the amplitude response $A(\omega)$ of this filter zero?
(d) [3 points] Assume that the sequence displayed in Figure 1 is $x[n]=c n+$ $\sin \left(\omega_{0} n\right)$, for some constant $c$ and frequency $\omega_{0}$. (You should already have determined the frequency $\omega_{0}$.) For this input $x[n]$, and ignoring samples near the ends, what is the output $y[n]$ of this moving average filter? Hint: the input $x[n]$ to this LTI system is the sum of two sequences.
(e) [2 points] Write the main loop of a computer program that implements this filter.

Design a causal time-domain notch filter to attenuate the periodic fluctations in $x[n]$ of Figure 1, as follows:
(a) [2 points] What is the frequency to be attenuated, in radians/sample?
(b) [2 points] Sketch the locations of filter poles and zeros in the complex $z$ plane.
(c) [2 points] What is the system function $H(z)$ for your filter? (Include the region of convergence.)
(d) [2 points] Modify your system function $H(z)$ so that your filter does nothing at frequency zero (DC).
(e) [2 points] Write a constant-coefficient difference equation relating filter output $y[n]$ to input $x[n]$.
(f) [2 points] Write the main loop of a computer program that implements your filter.
 Assume that you are given a discrete Fourier transform $X[k]$ of the sequence $x[n]$ of Figure 1. Also assume that a fast Fourier transform (FFT) was used, and that the number of samples after padding $x[n]$ with zeros was $N=128$.
(a) [2 points] Why typically are only positive frequencies $\omega$ sampled in the array $X[k]$ ?
(b) [2 points] Assuming that only positive frequencies $\omega$ are sampled, how many values are provided in the array $X[k]$ ?
(c) [2 points] What is the frequency sampling interval $\Delta \omega$, in radians/sample?
(d) [2 points] What is the index $k_{0}$ of the sample corresponding to the frequency nearest that of the periodic fluctuations in the sequence $x[n]$ ?
(e) [2 points] Describe in words how you would use this index $k_{0}$ to implement a linear time-invariant filter that attenuates the periodic fluctuations in the sequence $x[n]$ ?

Question 5
Given the sequence $x[n]=x_{c}(n T)$ from Figure 1, $\ldots$
(a) [3 points] Implement the transformation $y_{c}(t)=x_{c}(\sqrt{t})$. That is, write an expression that exactly defines a new output sequence $y[n]=y_{c}(n T)$ in terms of the samples of the input sequence $x[n]$.
(b) [2 points] In practice, we often sacrifice precision for efficiency in such transformations. How would you modify your computation of $y[n]$ to make it faster, while still approximately correct?

Question 6
(4 points)
Give an example of $H(z)$ (system function with ROC) for a linear time-invariant system that is
(a) [2 points] Stable but not causal.
(b) [2 points] Causal but not stable.

