ECE 3210 Midterm 2

Week of: November 1, 2022

Student's name: ______________________________

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You have 2 hours for 5 problems.

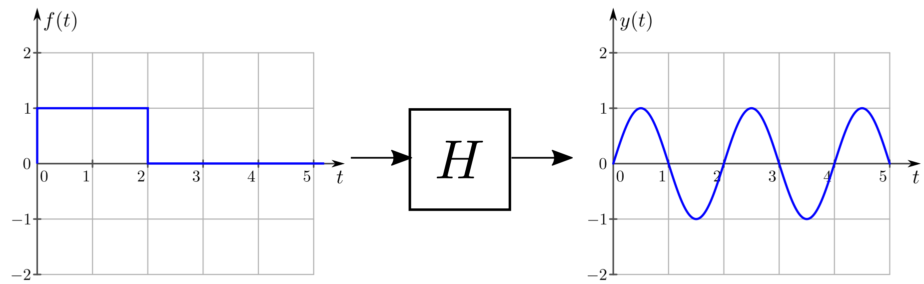
- Show enough (neat) work in the clear spaces on this exam to convince us that you derived, not guessed, your answers.
- Put your final answers in the boxes at the bottom of the page.

You may use two US letter-style size page of notes, front and back.

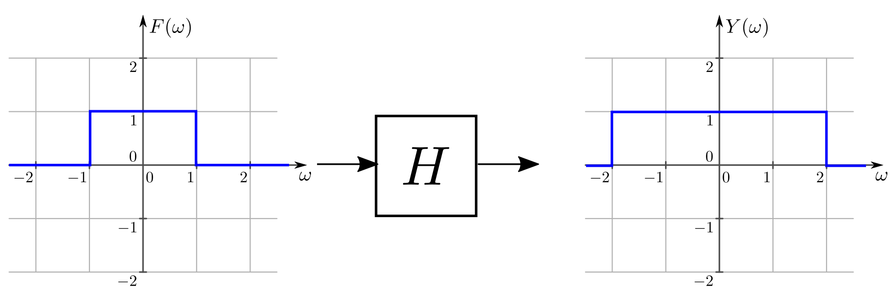
Problem	Score	Possible Points
1		20
2		20
3		20
4		20
5		20
Total score		100

1 Systems

(a) Given the shown inputs and outputs, is the following system LTI?



(b) Given the shown inputs and outputs, is the following system LTI?



(c) Consider the system described by the differential equation

$$2\frac{d^3y}{dt^3} - \frac{dy}{dt} + 4y(t) = 5\frac{df}{dt} - f(t).$$

Sketch the block diagram representation below.

2 Differential equations

Consider the following differential equation

$$(D^2 + 4D + 4)y(t) = (2D^2 + 11D + 8)f(t)$$

with initial conditions $y(0^-) = 4$ and $y'(0^-) = -6$ as well as an input $f(t) = u(t)$.

(a) Find the zero-input response using the Laplace transform.

$y_{zi}(t) =$

(b) Find the zero-state response using the Laplace transform.

$y_{zs}(t) =$

(c) Find the total response.

$y(t) =$

3 Sampling

(a) Consider the following discrete-time signals

$$f_k = \{1, 2, 3, 4, 1, 2, 3, 4\}$$
$$g_k = \{1, 1, 1, 1, 0, 0, 0, 0\}.$$

please compute the *circular* convolution of g_k and f_k.

$y_{k,\mathrm{cir}} =$

(b) Consider bandlimited signals f_1 with a bandwidth of 150 Hz and f_2 with a bandwidth of 75 Hz.

(i) What is the Nyquist sampling rate for $f_1 - f_2$?

$f_s =$

(ii) What is the Nyquist sampling rate for f_1^2?

$f_s =$

(c) Suppose we have a sound wave defined as

$$x(t) = A\sin(2\pi f_0 t)$$

where $f_0 = 7700$ Hz. If we were to sample it at $f_s = 8000$ Hz and were able to reconstruct the signal with an ideal low-pass filter with a frequency cutoff at $f_c = {}^{f_s}/_2$, what tone (i.e., f_0) should you hear? (*Hint:* think of what you observed in Lab 06.)

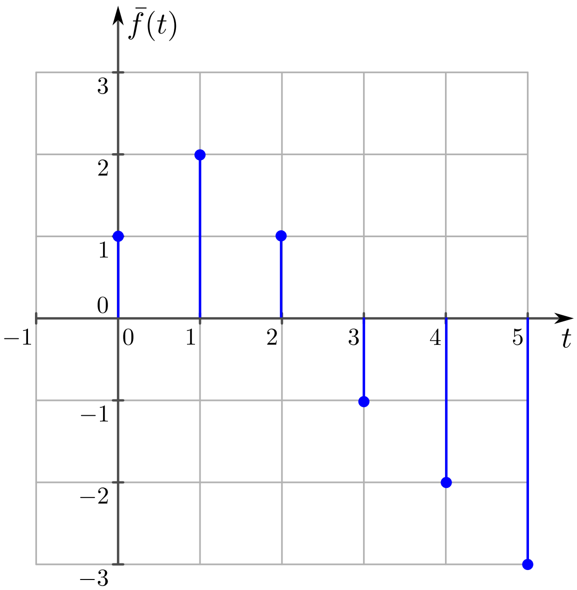
$f_0 =$

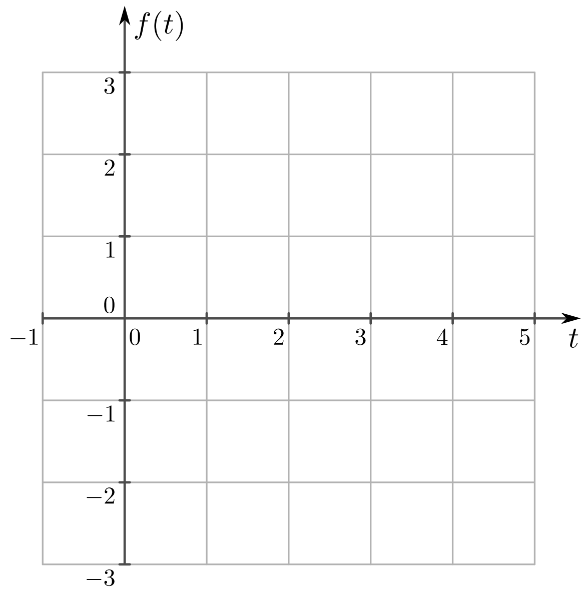
4 More sampling

(a) A first-order hold system can be used to reconstruct a signal $f(t)$ from its samples. The impulse response of this system is

$$h(t) = \Delta\left(\frac{t}{2}\right)$$

where T is the sampling interval. Consider a sampled signal $\bar{f}(t)$ below. Sketch the system output $f(t)$ on the plot marked $f(t)$. (*Hint:* the system is LTI therefore the system output is $f(t) = h(t) * \bar{f}(t)$.)





(b) What type of interpolation is this?

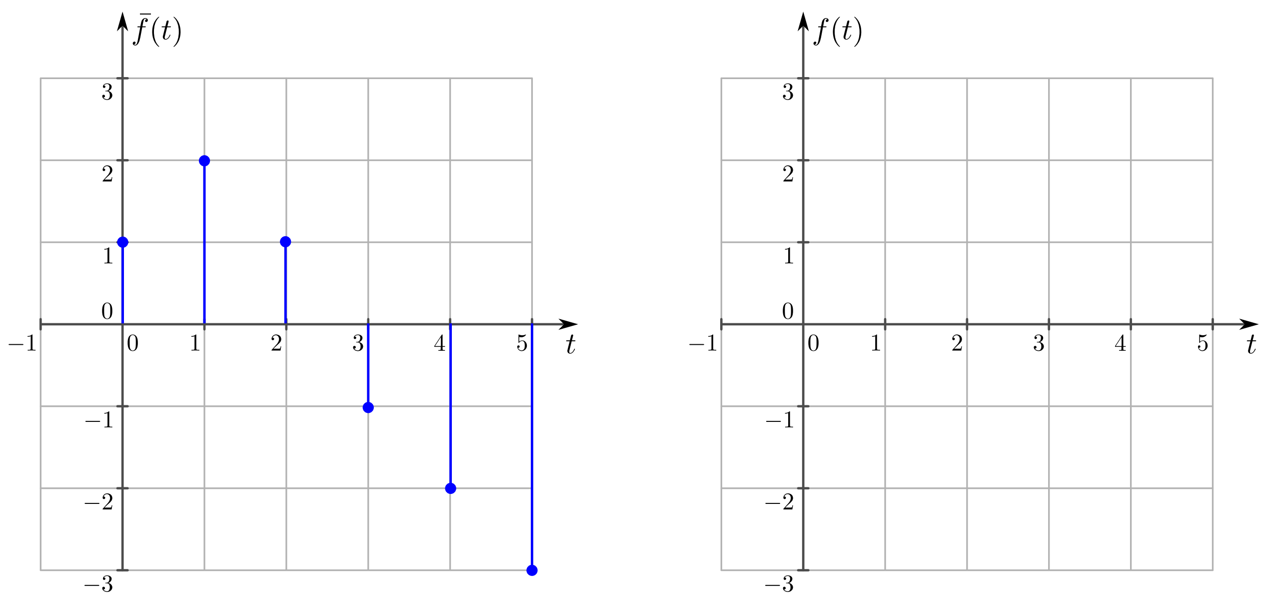
(c) Determine the frequency response $H(\omega)$ of this filter.

$H(\omega) =$

(d) This filter, being non-causal, is unrealizable. By delaying its impulse response, the filter can be made realizable. What is the minimum delay required to make the filter realizable?

$T =$

(e) How would this delay affect the reconstructed signal and the filter frequency response? Sketch the reconstruction on the plot below with the *delayed* $h(t)$.

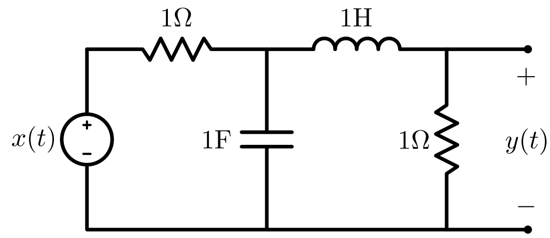


(f) How would this delay affect the system's frequency response? What is the new $H(\omega)$?

$H(\omega) =$

5 Transfer functions

Consider the circuit below.



(a) Derive the transfer function $H(s)$ for this circuit.

$H(s) =$

(b) Given an input $x(t) = e^{-t}u(t)$, find the zero-state response for $y(t)$.

$y_{zsr}(t) =$