PRINT YOUR NAME HERE:

HONOR CODE PLEDGE: "I have neither given nor received aid on this exam, nor have I concealed any violations of the honor code." Closed book; 4 sides of 8.5×11 "cheat sheet."

SIGN YOUR NAME HERE:

CIRCLE ONE: Undergraduate Graduate

Write your answer to each question in the answer space to the right of that question. Problems #1-20 are multiple choice (here same as fill-in-the-blank) worth 5 points each.

For Problems #1-3: T=0.1. The analog filter is: $H_a(s)=\frac{40s}{s^2+400}$. $h_a(t)=40\cos(20t)u(t)$.

1. h[n] designed using **impulse invariance** is: **(a)** $4\cos(2n)u[n]$ **(b)** $40\cos(20n)u[n]$ **(c)** $400\cos(200n)u[n]$ **(d)** $0.4\cos(0.2n)u[n]$ **(e)** $2\cos(2n)u[n]$
2. H(z) using **bilinear transform** is: **(a)** $\frac{z-1}{z+1}$ **(b)** $\frac{2z}{z^2+2}$ **(c)** $\frac{4z}{z^2+2}$ **(d)** $\frac{2z}{z^2+1}$ **(e)** $\frac{z^2-1}{z^2+1}$
3. Using a **bilinear transform**, the continuous-time frequency $\Omega = 20$ maps to discrete-time frequency $\omega =$: **(a)** 20 **(b)** $\frac{\pi}{4}$ **(c)** $\frac{\pi}{3}$ **(d)** $\frac{\pi}{2}$ **(e)** π
   Can do #3 independently of #2; no partial credit if wrong #2 leads to wrong #3.
4. IIR filters are guaranteed to be stable if designed from a stable analog filter using: **(a)** Impulse invariance **(b)** Bilinear transform **(c)** Both **(d)** Neither **(e)** Can't tell

A differentiator has: $H_a(s)$=s and $H(e^{j\omega})=j\omega$ for $|\omega| < \pi$ and h[n]=$\frac{(-1)^n}{n}, n \neq 0$.
For problems #5-7: Design a linear-phase length=3 noncausal FIR differentiator using:

5. **Frequency sampling** with: $H(e^{j0})$=0. $H(e^{j\pi/2})$=jπ/2. $H(e^{j\pi})$=0 (not π):
   (a) $\{-\frac{1}{4}, \frac{1}{2}, -\frac{1}{4}\}$ **(b)** $\{\frac{\pi}{4}, 0, -\frac{\pi}{4}\}$ **(c)** $\{-\frac{\pi}{4}, \frac{\pi}{2}, -\frac{\pi}{4}\}$ **(d)** $\{\frac{\pi}{4}, \frac{\pi}{2}, \frac{\pi}{4}\}$ **(e)** $\{1, 0, -1\}$
6. A **rectangular window** applied to the ideal digital differentiator:
   (a) $\{-\frac{1}{4}, \frac{1}{2}, -\frac{1}{4}\}$ **(b)** $\{\frac{\pi}{4}, 0, -\frac{\pi}{4}\}$ **(c)** $\{-\frac{\pi}{4}, \frac{\pi}{2}, -\frac{\pi}{4}\}$ **(d)** $\{\frac{\pi}{4}, \frac{\pi}{2}, \frac{\pi}{4}\}$ **(e)** $\{1, 0, -1\}$
7. An **equiripple** filter, now with $H(e^{j\pi})$=jπ. The Matlab command designing it:
   (a) `fir1(2,[0,1],[0,1])` **(b)** `fir2(2,[0,1],[0,1])` **(c)** `firpm(2,[0,1],[0,1])` **(d)** `firpm(2,[0,1],[0,pi/2],'hilbert')` **(e)** `firpm(2,[0,1],[0,pi],'hilbert')`
8. The IIR filter for an ideal **integrator** designed using bilinear transform with T=2:
   (a) y[n]+y[n-1]=x[n]+x[n-1] **(b)** y[n]–y[n-1]=x[n]+x[n-1] **(c)** y[n]=x[n]+x[n-1] **(d)** y[n]–y[n-1] = x[n]–x[n-1] **(e)** y[n]+y[n-1]=x[n]–x[n-1] **(f)** y[n]=x[n]–x[n-1]

For #9-11: A 200 Hz sinusoid is input to a DSP system with sampling rate 1000$\frac{\text{SAMPLE}}{\text{SECOND}}$.
Sampler (A/D) and reconstructor (D/A) are not shown. No antialias filter is used.

9. 200 Hz$\rightarrow\boxed{\downarrow \mathbf{2}}\rightarrow$? **(a)** 100 Hz **(b)** 100&400 Hz **(c)** 400 Hz **(d)** 400&600 Hz **(e)** 600 Hz
10. 200 Hz$\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow$? **(a)** 100 Hz **(b)** 100&400 Hz **(c)** 400 Hz **(d)** 400&600 Hz **(e)** 600 Hz
11. If input is a 200 Hz sinusoid, which DSP system outputs **only** a 300 Hz sinusoid?
    LPF is an ideal Low Pass Filter (not bandpass) with a single cutoff frequency.
    (a) $\rightarrow\boxed{\uparrow \mathbf{3}}\rightarrow\boxed{\downarrow \mathbf{2}}\rightarrow\boxed{\textbf{LPF}}\rightarrow$ **(b)** $\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\textbf{LPF}}\rightarrow$ **(c)** $\rightarrow\boxed{\uparrow \mathbf{3}}\rightarrow\boxed{\textbf{LPF}}\rightarrow\boxed{\downarrow \mathbf{2}}\rightarrow$
    (d) $\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow\boxed{\textbf{LPF}}\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow$ **(e)** $\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow\boxed{\textbf{LPF}}\rightarrow$ **(f)** $\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\textbf{LPF}}\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow$

12. For which frequency ω_o are the outputs $x[n]$ of these two systems identical?
In both systems, LPF is an ideal lowpass filter with cutoff frequency $\pi/3$.
#1: $\cos(\omega_o n) \rightarrow \boxed{\uparrow \mathbf{3}} \rightarrow \boxed{\mathbf{LPF}} \rightarrow x[n]$ **#2:** $\cos(\omega_o n) \rightarrow \boxed{\downarrow \mathbf{3}} \rightarrow \boxed{\mathbf{LPF}} \rightarrow x[n]$

(a) 0.3π **(b)** 0.5π **(c)** 0.6π **(d)** 0.75π **(e)** (c) & (d)

For #13-16: x[n]=Acos($\omega_0 n$)+Bcos($\omega_1 n$) for 0≤n≤L-1; use an N-point DFT of x[n].

13. To help resolve the two peaks, we should do which of the following:
(a) Increase L **(b)** Increase N **(c)** Use Hamming window **(d)** (a)&(c) **(e)** (b)&(c)

14. To make the spectrum smoother, we should do which of the following:
(a) Increase L **(b)** Increase N **(c)** Use Hamming window **(d)** (a)&(c) **(e)** (b)&(c)

15. To reduce sidelobes around the peaks, we should do which of the following:
(a) Increase L **(b)** Increase N **(c)** Use Hamming window **(d)** (a)&(c) **(e)** (b)&(c)

16. Which of these makes it **harder** to resolve the two peaks:
(a) Decrease L **(b)** Decrease N **(c)** Use Hamming window **(d)** (a)&(c) **(e)** (b)&(c)

17. The filter type designed by `firpm(10,[0,0.2,0.3,0.7,0.8,1],[0,0,1,1,0,0])` is:
(a) Lowpass **(b)** Bandpass **(c)** Highpass **(d)** Band-reject **(e)** Notch **(f)** Comb

18. The filter with h[n]=$\{a, b, c, d, 0, -d, -c, -b, -a\}$ is **guaranteed** to be:
(a) Lowpass **(b)** Bandpass **(c)** Highpass **(d)** Band-reject **(e)** Notch **(f)** Comb

19. The 2-D filter having 2-D impulse response h[i,j]= $\begin{bmatrix} 1 & -2 & 1 \\ -2 & 4 & -2 \\ 1 & -2 & 1 \end{bmatrix}$ is:
(a) Lowpass **(b)** Bandpass **(c)** Highpass **(d)** Band-reject **(e)** Notch **(f)** Comb
HINT: h[i,j] is separable: h[i,j]=h[i]h[j]. What are h[i] and h[j] and what do they do?

20. "DSP" stands for: **(a)** Digital Signal Processing **(b)** Drivel Spewed by Professor
(c) Dumbest Subject Partaken **(d)** Drive Slowly Please **(e)** Dummies Should Pass