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=0.2. The analog filter is: $H_a(s)=\frac{10}{s+10}$. $h_a(t)=10e^{-10t}u(t)$.

1. h[n] designed using **impulse invariance** is:
(a) $2e^{-2n}u[n]$ **(b)** $10e^{-10n}u[n]$ **(c)** $20e^{-20n}u[n]$ **(d)** $0.1e^{-0.1n}u[n]$ **(e)** $0.2e^{-0.2n}u[n]$

2. H(z) designed using **bilinear transform** is:
(a) $\frac{1}{2}(1+z^{-1})$ **(b)** $\frac{2z}{z-e^{-2}}$ **(c)** $\frac{20z}{z-e^{-20}}$ **(d)** $\frac{1}{2}(1-z^{-1})$ **(e)** $\frac{z}{z-0.1}$

3. Using a **bilinear transform**, the continuous-time frequency $\Omega = 10$ maps to the discrete-time frequency $\omega =$: **(a)** 10 **(b)** $\frac{\pi}{4}$ **(c)** $\frac{\pi}{3}$ **(d)** $\frac{\pi}{2}$ **(e)** π

4. The IIR filter for an ideal differentiator 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]+y[n-1]=x[n]–x[n-1] **(d)** y[n]–y[n-1]=x[n]–[n-1] **(e)** y[n]=x[n]+x[n-1] **(f)** y[n]=x[n]–x[n-1]

5. 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

For problems #6-8: Design a linear-phase length=3 noncausal FIR low-pass filter having cutoff frequency=$\frac{\pi}{2}$ using each of the following three FIR design techniques:

6. **Frequency sampling** with $H(e^{j0})$=1. $H(e^{j\pi/2})$=1/2. $H(e^{j\pi})$=0:
(a) $\{-\frac{1}{4},\frac{1}{2},-\frac{1}{4}\}$ **(b)** $\{\frac{1}{4},\frac{1}{2},\frac{1}{4}\}$ **(c)** $\{-\frac{\pi}{4},\frac{\pi}{2},-\frac{\pi}{4}\}$ **(d)** $\{\frac{\pi}{4},\frac{\pi}{2},\frac{\pi}{4}\}$ **(e)** $\{\frac{1}{\pi},\frac{1}{2},\frac{1}{\pi}\}$

7. A **rectangular window** on the ideal low-pass filter with cutoff frequency ω_c=$\frac{\pi}{2}$:
(a) $\{-\frac{1}{4},\frac{1}{2},-\frac{1}{4}\}$ **(b)** $\{\frac{1}{4},\frac{1}{2},\frac{1}{4}\}$ **(c)** $\{-\frac{\pi}{4},\frac{\pi}{2},-\frac{\pi}{4}\}$ **(d)** $\{\frac{\pi}{4},\frac{\pi}{2},\frac{\pi}{4}\}$ **(e)** $\{\frac{1}{\pi},\frac{1}{2},\frac{1}{\pi}\}$

8. An **equiripple** filter. Matlab command: **(a)** `fir1(2,[0,.49,.51,1],[0,0,1,1])`
(b) `fir2(2,[0,.49,.51,1],[0,0,1,1])` **(c)** `firpm(2,[0,.49,.51,1],[0,0,1,1])`
(d) `firpm(2,[0,.49,.51,1],[1,1,0,0])` **(e)** `firpm(2,[0,.5,1],[0,.5,1])`

9. An FIR filter of the form h[n]=$\{a,b,0,-b,-a\}$ has the following restrictions:
(a) $H(e^{j0})$=0 **(b)** $H(e^{j\pi/2})$=0 **(c)** $H(e^{j\pi})$=0 **(d)** (a) and (c) **(c)** None

Problems #10-13 involve selection of image processing techniques for various tasks:

10. To reduce isolated point noise, use:
(a) 2-D lowpass filter **(b)** Median filter **(c)** Edge detector **(d)** (a)&(b) **(e)** (b)&(c)

11. To reduce white noise, use:
(a) 2-D lowpass filter **(b)** Median filter **(c)** Edge detector **(d)** (a)&(b) **(e)** (b)&(c)

12. To enhance image features, use:
(a) 2-D lowpass filter **(b)** Median filter **(c)** Edge detector **(d)** (a)&(b) **(e)** (b)&(c)

13. Which technique has a very different frequency response from the other four?
(a) Lowpass filter **(b)** Highpass filter **(c)** Differentiator **(d)** Edge detector **(e)** None

For #14-16: x[n]=Acos($\omega_0 n$)+Bcos($\omega_1 n$) for 0≤n≤L-1; use an N-point DFT of x[n].

14. Increasing N helps: **(a)** Resolve 2 peaks **(b)** Reduce sidelobes **(c)** Smoothes spectrum **(d)** (a)&(b) **(e)** (b)&(c)

15. Increasing L helps: **(a)** Resolve 2 peaks **(b)** Reduce sidelobes **(c)** Smoothes spectrum **(d)** (a)&(b) **(e)** (b)&(c)

16. A Hamming window helps: **(a)** Resolve 2 peaks **(b)** Reduce sidelobes **(c)** Smoothes spectrum **(d)** (a)&(b) **(e)** (b)&(c)

For #17-19: A 300 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.

17. 300 Hz→[↓ **2**]→? **(a)** 150 Hz **(b)** 150&350 Hz **(c)** 400 Hz **(d)** 400&600 Hz **(e)** 600 Hz

18. 300 Hz→[↑ **2**]→? **(a)** 150 Hz **(b)** 150&350 Hz **(c)** 400 Hz **(d)** 400&600 Hz **(e)** 600 Hz

19. If input is a 300 Hz sinusoid, which DSP system outputs **only** a 450 Hz sinusoid?
LPF is an ideal Low Pass Filter (not bandpass) with a single cutoff frequency.
(a) →[↑ **3**]→[↓ **2**]→[**LPF**]→ **(b)** →[↓ **3**]→[↑ **2**]→[**LPF**]→ **(c)** →[↑ **3**]→[**LPF**]→[↓ **2**]→
(d) →[↓ **3**]→[**LPF**]→[↑ **2**]→ **(e)** →[↑ **2**]→[↓ **3**]→[**LPF**]→ **(f)** →[↑ **2**]→[**LPF**]→[↓ **3**]→

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