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a) Butterworth filter

b) Notch filter

c) low-pass filter

d) ...

21. The system having frequency response $H(e^{j\omega}) = \frac{2e^{-j\omega}}{1+3e^{-j2\omega}}$ is:

a) $y(n-1) = \frac{1}{2}x(n) + \frac{1}{2}x(n-2)$

b) $y(n) + 3y(n-2) = 2x(n-1)$

c) $y(n) = x(n) + 3x(n-$ ...

d) $y(n) + 3y(n-1) = 2x(n)$

e) $y(n) + 3y(n-1) = 2x(n-2)$

22. If a continuous-time filter with an impulse response, $h_c(t)$, is sampled with a sampling frequency Fs, what happens to the cutoff frequency, ω_c, of the discrete-time filter as Fs increased?

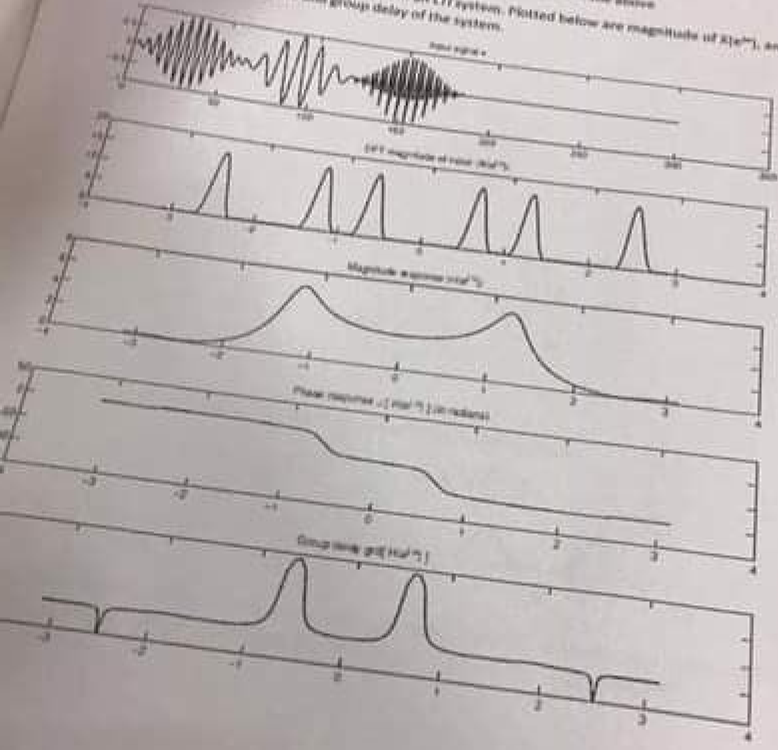
a) ω_c increases

b) ω_c decreases

c) ω_c remains constant

d) None of the above

23. A discrete-time signal x(n) is applied to an LTI system. Plotted below are magnitude of $X(e^{j\omega})$, and magnitude response, phase response and group delay of the system.



10. The continuous-time signal x(t) = sin(20πt) + cos(40πt) is sampled with a sampling frequency Fs to obtain the discrete-time signal $x(n)=\sin(\frac{\pi n}{5})+\cos(\frac{2\pi n}{5})$. The sampling frequency, Fs, which is consistent with this information is

a) 40 Hz
b) 80 Hz
c) 100 Hz
d) 120 Hz

11. A DSP system with the following system function $H(z)=\frac{0.97(z-1)}{z-0.94}$ is a

a) Low-pass filter
b) Band-pass filter
c) High-pass filter
d) band-stop filter

12. The pole-zero diagram of DSP system is shown in the figure below. If this system is implement with a sampling frequency of 10KHz, the magnitude response of this system at frequency 5KHz is:

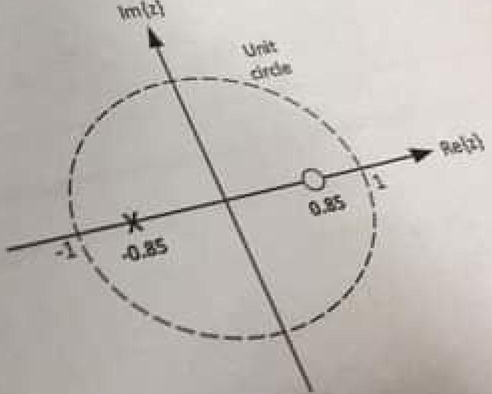
c) 12.33 d) 0.081

$z-0.94$

a) Low-pass filter
b) Band-pass filter
c) High-pass filter
d) band-stop filter

12. The pole-zero diagram of DSP system is shown in the figure below. If this system is implement with a sampling frequency of 10KHz, the magnitude response of this system at frequency 5KHz is:

a) 1 b) 0 c) 12.33 d) 0.081

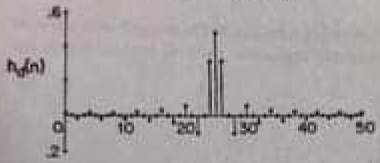


...ke response of an LTI system, then the output y...

4. Find two different continuous-time signals that will produce the sequence $x(n) = \cos(0.15\pi n)$ when sampled with a sampling frequency of 8 kHz.
   a) Sin(1200πt) and cos(17200πt)
   b) Cos(1200πt) and sin(17200πt)
   c) Cos(1200πt) and cos(17200πt)
   d) Sin(1200πt) and sin(17200πt)

5. A DSP system with the following difference equation:
   $y(n) = 0.25x(n) + 0.25x(n-1) + 0.25x(n-2) + 0.25x(n-3)$ is
   a) Low-pass filter
   b) High-pass filter
   c) Band-pass filter
   d) Band-stop filter

6. The following is an impulse response for



   a) An IIR high-pass filter
   b) An FIR band-pass filter
   c) An IIR low-pass filter
   d) An FIR low-pass filter

7. Coefficients symmetry is important in FIR filters because it provides
   a) A smaller transition bandwidth
   b) Less passband ripple
   c) Less stopband ripple
   d) A linear phase response

8. Two digital filters can be operated in cascade. Or, the same effect can be achieved by
   a) Adding their coefficients
   b) Subtracting their coefficients
   c) Convolving their coefficients
   d) Averaging their coefficients and then use Hamming window

9. If a linear phase filter has a phase response of 80 degrees at 400Hz. What will its phase re[...] frequency of 200Hz (assuming that both frequencies are in the passband of the filter)
   a) 35 degrees
   b) 40 degrees
   c) 45 degrees
   d) 80 degrees

ations of a digital filter:

0 KHz passband

ipple: -3dB (at 10 KHz)

Fs = 100 KHz

z to 20 KHz

-10dB (starting at 20 KHz)

onic in the pass and stop bands, (i.e. no ripple).

tions, is required to be designed using Butterworth model and mapped to

ency response of the specified digital filter and its equivalent analogue fil
ry values on the graphs. [4pts]

iii) Parallel of first-order sections

17. H(z) designed using bilinear transform is:

a) $\frac{1}{2}(1+z^{-1})$ b) $\frac{2z}{z-e^{-2}}$ c) $\frac{20z}{z-e^{-2}}$ d) $\frac{1}{2}(1-z^{-1})$ e) $\frac{z}{z-0.1}$

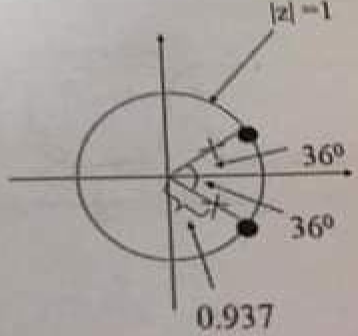
18. If input is a 300 Hz sinusoid, which DSP system outputs only a 450 Hz sinusoid?
LPF is an Ideal Low Pass Filter (not bandpass) with a single cutoff frequency.

(a) → [3] → [2] → [LPF] → (b) → [3] → [2] → [LPF] → (c) → [3] → [LPF] → [2] →
(d) → [3] → [LPF] → [2] → (e) → [2] → [3] → [LPF] → (f) → [2] → [LPF] → [3] →

19. x(n)= $\cos(\frac{\pi}{2}n)$ is an input into a LTI system with $H(e^{j\omega}) = 1 + 0.5e^{j\omega} + e^{j2\omega}$. The output, y(n) is:

a) $\sqrt{2}\cos(\frac{\pi}{2}n)$ b) $2.5\cos(\frac{\pi}{2}n)$ c) $1.5\cos(\frac{\pi}{2}n)$ d) $\sin(\frac{\pi}{2}n)$ e) $-0.5\sin(\frac{\pi}{2}n)$

20. The pole-zero diagram of a digital filter is shown in the figure below. This filter is:



b) Draw the realization block diagram of the following filter usin

$$H(z)=\frac{1+2z^{-1}-6z^{-2}}{1+0.3z^{-1}-0.1z^{-2}}$$

[3pts]

i) Direct Form I

ii) Transposed Direct form II

[3pts]

13. If input signal $x(n) = h(-n)$, where $h(n)$ is the impulse response of an LTI system, then the output $y(n)$ is
   a) $y(n) = 2h(n)$
   b) $y(n) = [h(n)]^2$
   c) $y(n)$ is the autocorrelation of $x(n)$
   d) None of the above

14. If an input signal is applied to two LTI systems with respective unit-impulse responses $h(n)$ and $3h(n-2)$, then the response of the second system is the response of the first
   a) Amplitude scaled by 3 and delayed by 2
   b) Amplitude scaled by 3 and advanced by 2
   c) Amplitude scaled by 3
   d) None of the above

...nce: an method used to design discrete h(n).

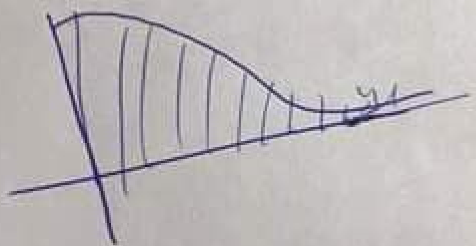
...linear Trans method used to design digital IIR filter for ~~low~~ high sampling frequency

(b) Given the following specifications of a digital filter:

- Lowpass filter: 0 – 10 KHz passband
- Maximum passband ripple: -3dB (at 10 KHz)
- Sampling frequency: Fs = 100 KHz
- Transition band: 10 KHz to 20 KHz
- Stopband attenuation: -10dB (starting at 20 KHz)
- The filter must be monotonic in the pass and stop bands, (i.e. no ripple).

An IIR filter, meets the above specifications, is required to be designed using Butterworth model and mapped ... digital using bilinear transform:

i) Draw an estimate for magnitude frequency response of the specified digital filter and its equivalent analo... Clearly specify axes labels and all necessary values on the graphs. [4pts]



Question 4: [20 marks]

(a) Compare between the impulse invariance and bi
their disadvantages. [4pts]

Given the following specifications of a digital filte

- **Lowpass filter:** 0 – 10 KHz passband
- **Maximum passband ripple:** -3dB (at 10
- **Sampling frequency:** Fs = 100 KHz
- **Transition band:** 10 KHz to 20 KHz

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a) Resonant filter b) Notch filter c) low-pass filter d) high-pass f...

21. The system having frequency response $H(e^{j\omega}) = \frac{2e^{-j\omega}}{1+3e^{-j2\omega}}$ is:

a) $y(n-1) = \frac{1}{2}x(n) + \frac{3}{2}x(n-2)$ b) $y(n)+3y(n-2) = 2x(n-1)$ c) $y(n) = x(n) + 3x(n-2)$

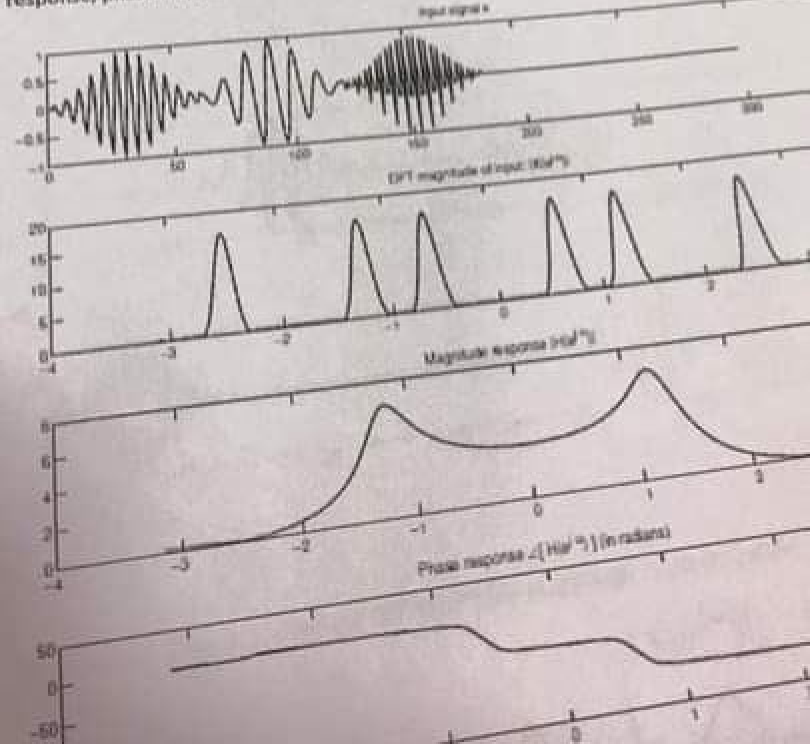
d) $y(n)+3y(n-1) = 2x(n)$ e) $y(n)+3y(n-1) = 2x(n-2)$

22. If a continuous-time filter with an impulse response, $h_a(t)$, is sampled with a sampling frequency Fs, w... happens to the cutoff frequency, ω_c, of the discrete-time filter as Fs increased?

a) ω_c increases b) ω_c decreases

c) ω_c remains constant d) None of the above

23. A discrete-time signal x(n) is applied to an LTI system. Plotted below are magnitude of $X(e^{j\omega})$, an... response, phase response and group delay of the system.



(b) Given the following specifications of a digital filter:

- **Lowpass filter:** 0 – 10 KHz passband
- **Maximum passband ripple:** -3dB (at 10 KHz)
- **Sampling frequency:** Fs = 100 KHz
- **Transition band:** 10 KHz to 20 KHz
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- The filter must be **monotonic** in the pass and stop ban

An IIR filter, meets the above specifications, is required to be desi
digital using **bilinear transform:**

i) Draw an estimate for magnitude frequency response of the sp
Clearly specify axes labels and all necessary values on the graph

e) FIR ______________

For questions #16-17: T=0.2. The analogue filter is: $H_a(s)=\dfrac{10}{s+10}$ and $h_a(t)=10e^{-10t}u(t)$:

16. h[n] designed using impulse invariance technique is:

a) $2e^{-2n}u(n)$ b) $10e^{-10n}u(n)$ c) $20e^{-20n}u(n)$ d) $0.1e^{-0.1n}u(n)$ e) $0.2e^{-0.2n}u(n)$

17. H(z) designed using bilinear transform is:

a) $\frac{1}{2}(1+z^{-1})$ b) $\dfrac{2z}{z-e^{-2}}$ c) $\dfrac{20z}{z-e^{-2}}$ d) $\frac{1}{2}(1-z^{-1})$ e) $\dfrac{z}{z-0.1}$

---

18. 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) $\rightarrow\boxed{\uparrow 3}\rightarrow\boxed{\downarrow 2}\rightarrow\boxed{\text{LPF}}\rightarrow$ (b) $\rightarrow\boxed{\downarrow 3}\rightarrow\boxed{\uparrow 2}\rightarrow\boxed{\text{LPF}}\rightarrow$ (c) $\rightarrow\boxed{\uparrow 3}\rightarrow\boxed{\text{LPF}}\rightarrow\boxed{\downarrow 2}\rightarrow$

(d) $\rightarrow\boxed{\downarrow 3}\rightarrow\boxed{\text{LPF}}\rightarrow\boxed{\uparrow 2}\rightarrow$ (e) $\rightarrow\boxed{\uparrow 2}\rightarrow\boxed{\downarrow 3}\rightarrow\boxed{\text{LPF}}\rightarrow$ (f) $\rightarrow\boxed{\uparrow 2}\rightarrow\boxed{\text{LPF}}\rightarrow\boxed{\downarrow 3}\rightarrow$

19. $x(n)=\cos(\frac{\pi}{\;}n)$ is an input into a LTI system with $H(e^{j\omega})=1+0.5e^{j\omega}+e^{j2\omega}$. The output, y(n) is:

Fall Semester 29/01/2017

Name: ID: Gr...

Instructions:

- You have 145 minutes, budget your time carefully!
- Turn OFF your mobile!
- This exam consists of **Three** questions, you have to answer all

Question	Maximum	Mark	ABET Lea...
Q1	60		
Q2-(a)	10		
Q2-(b)	10		
Q3	20		
Total	100		

Question 1: multiple choice (60 marks; 2pts each)

1. The discrete signal $x(n) = e^{[illegible]}$ is
   a) Periodic with a period of 3.
   b) Periodic with period of 4.
   c) Periodic with period 8.
   d) Non-periodic
2. If the Nyquist rate for $x_a(t)$ is Ω_s, what is the Nyquist rate for $x_a(2t)$
   a) $2\,\Omega_s$
   b) $\Omega_s/2$
   c) Ω_s
   d) $\Omega_s/4$
3. IIR filters
   a) Use feedback
   b) Are sometimes called recursive filters
   c) Can oscillate (i.e. not stable) if not properly designed
   d) All of the above

Question 4: [20 marks]

(a) Compare between the impulse invariance and bilinear transformation methods in terms of their advantages and their disadvantages. [4pts]

b) Given the following specifications of a digital filter:

- **Lowpass filter:** 0 – 10 KHz passband
- **Maximum passband ripple:** -3dB (at 10 KHz)
- **Sampling frequency:** Fs = 100 KHz
- **Transition band:** 10 KHz to 20 KHz
- **Stopband attenuation:** -10dB (starting at 20 KHz)
- The filter must be **monotonic** in the pass and stop bands, (i.e. **no ripple**).

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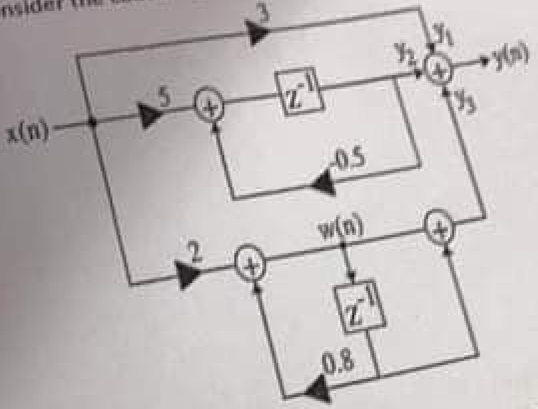
ital using **bilinear transform**:

raw an estimate for magnitude frequency response of the specified digital filter and its equivalent a

arly specify axes labels and all necessary values on the graphs. [4pts]

(20 marks)

onsider the causal digital filter shown in the following figure:



a) Find the unit sample response, $h(n) = Z^{-1}\{H(z)\}$, of this filter. [10pts]

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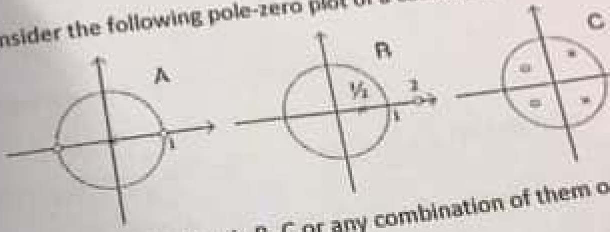
15. Consider the following pole-zero plot of a causal LTI system



Which system is: (put A, B, C or any combination of them o

a) All-pass ________

b) Minimum phase ________

c) Stable ________

d) IIR ________

e) FIR ________

<u>For questions #16-17: T=0.2. The analogue filter is:</u> $H_a(s)$

16. h[n] designed using impulse invariance technique is:

a) $2e^{-2n}u(n)$ b) $10e^{-10n}u(n)$ c) $20e^{-20n}u(n)$

17. H(z) designed using bilinear transform is:

a) $\frac{1}{2}(1+z^{-1})$ b) $\frac{2z}{z-e^{-2}}$ c) $\frac{20}{z-}$

---

18. If input is a 300 Hz sinusoid, which DSP syster

... ideal Low Pass Filter (not bandpass) wi

بسم الله الرحمن الرح

a) Resonant filter   b) Notch filter   c) low-pass filter   d) high-pass filter

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a) $y(n-1) = \frac{1}{2}x(n) + \frac{3}{2}x(n-2)$ b) $y(n)+3y(n-2) = 2x(n-1)$ c) $y(n) = x(n)+3x(n-2)$

d) $y(n)+3y(n-1) = 2x(n)$ e) $y(n)+3\,y(n-1) = 2x(n-2)$

22. If a continuous-time filter with an impulse response, $h_c(t)$, is sampled with a sampling frequency Fs, what happens to the cutoff frequency, ω_c, of the discrete-time filter as Fs increased?

a) ω_c increases   b) ω_c decreases

c) ω_c remains constant   d) None of the above

23. A discrete-time signal x(n) is applied to an LTI system. Plotted below are magnitude of $X(e^{j\omega})$, and magnitude response, phase response and group delay of the system.

ii) Find minimum order of the designed IIR filter, N, and the corresponding cut-off frequency, Ω_c? Show all [illegible]
[illegible]]

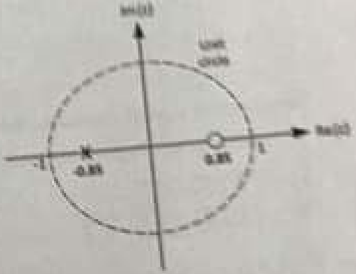
iii) Find system function Ha(s) of the designed analogue filter and the corresponding digital filte

End of exam

10. The continuous-time signal $x(t)$ = [illegible] is sampled with a sampling frequency fs to obtain the discrete-time signal $x(n)$ = [illegible]. The sampling frequency fs which is consistent with this information is
a) 40 Hz
b) 80 Hz
c) 100 Hz
d) 120 Hz

11. A DSP system with the following system function $H(z) = \frac{[illegible](z-1)}{z-[illegible]}$ is a
a) Low-pass filter
b) Band-pass filter
c) High-pass filter
d) band-stop filter

12. The pole-zero diagram of DSP system is shown in the figure below. If this system is implement with a sampling frequency of 10kHz, the magnitude response of this system at frequency 5KHz is:
a) 1 b) 0 c) 12.33 d) 0.082



13. If input signal x(n) = h(-n), where h(n) is the impulse response of an LTI system, then the output
a) y(n) = 2h(n)
b) y(n) = [h(n)]²
c) y(n) is the autocorrelation of x(n)
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14. If an input signal is applied to two LTI systems with respective unit-impulse responses h[n]
the response of the second system is the response of the first
a) Amplitude scaled by 3 and delayed by 2
b) Amplitude scaled by 3 and advanced by 2
c) Amplitude scaled by 3
d) None of the above