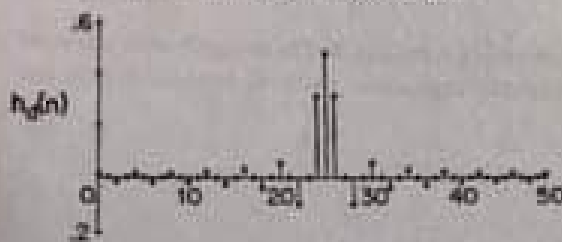


4. Find two different continuous-time signals that will produce the sequence  $x[n] = \cos(0.15n)$  when sampled with a sampling frequency of 8 kHz.
- $\sin(1200\pi t)$  and  $\cos(17200\pi t)$
  - $\cos(1200\pi t)$  and  $\sin(17200\pi t)$
  - $\cos(1200\pi t)$  and  $\cos(17200\pi t)$
  - $\sin(1200\pi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

5. A DSP system with the following difference equation:
- $$y[n] = 0.25x[n] + 0.25x[n-1] + 0.25x[n-2] + 0.25x[n-3]$$
- Low-pass filter
  - High-pass filter
  - Band-pass filter
  - Band-stop filter

6. The following is an impulse response for:



- An IIR high-pass filter
  - An FIR band-pass filter
  - An IIR low-pass filter
  - An FIR low-pass filter
7. Coefficients symmetry is important in FIR filters because it provides
- A smaller transition bandwidth
  - Less passband ripple
  - Less stopband ripple
  - A linear phase response
8. Two digital filters can be operated in cascade. Or, the same effect can be achieved by
- Adding their coefficients
  - Subtracting their coefficients
  - Convolving their coefficients
  - 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 response be at a frequency of 200Hz (assuming that both frequencies are in the passband of the filter)
- 35 degrees
  - 40 degrees
  - 45 degrees
  - 80 degrees

Specifications of a digital filter:

Passband: 0 to 10 KHz

Attenuation: -3dB (at 10 KHz)

Stopband:  $f_s = 100$  KHz

to 20 KHz

Attenuation: 10dB (starting at 20 KHz)

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

The filter is required to be designed using Butterworth model and mapped to

frequency response of the specified digital filter and its equivalent analogue filter. Plot the magnitude and phase response of the filter. [4pts]

**iii) Parallel of first-order sections**

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

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

[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

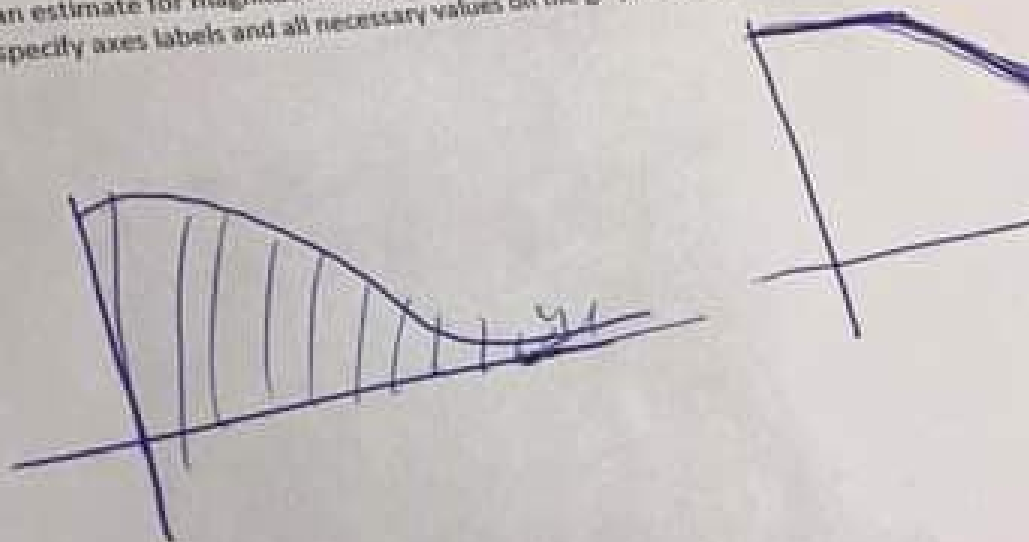
Linear Transform method used to design digital IIR filter for ~~low~~ high sampling frequency

i) Given the following specifications of a digital filter:

- Lowpass filter: 0 – 10 KHz passband
- Maximum passband ripple: -3dB (at 10 KHz)
- Sampling frequency:  $f_s = 100$  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:

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



**Question 4:** [20 marks]

(a) Compare between the impulse invariance and bilinear transform methods and list their disadvantages. [4pts]

Given the following specifications of a digital filter:

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

## بسم الله الرحمن الرحيم

a) Resonant filter

b) Notch filter

c) low-pass filter

d) High-pass filter

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,  $h_c(t)$ , is sampled with a sampling frequency  $F_s$ , what happens to the cutoff frequency,  $\omega_c$ , of the discrete-time filter as  $F_s$  increased?

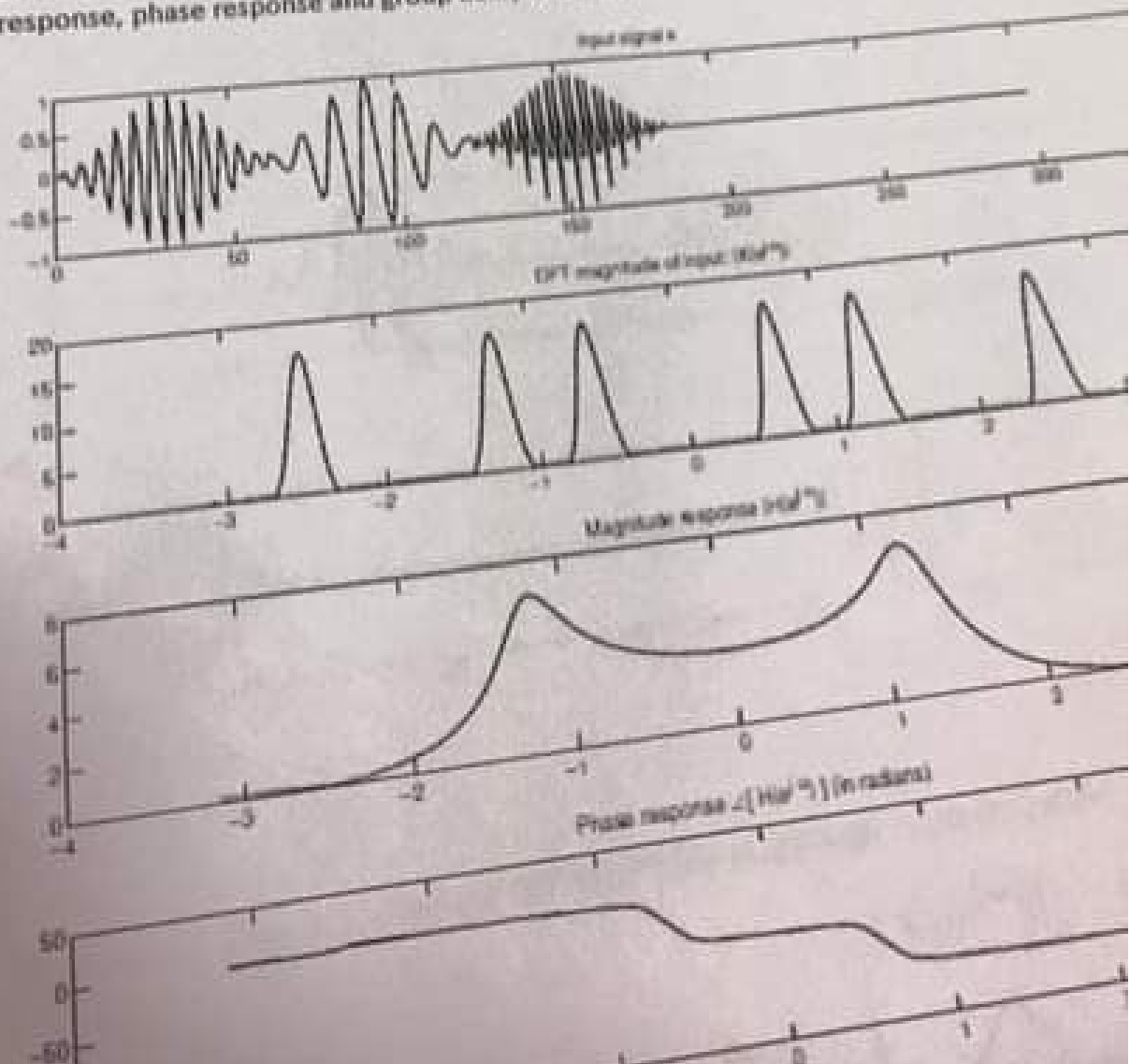
a)  $\omega_c$  increases

b)  $\omega_c$  decreases

c)  $\omega_c$  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  $X(e^{j\omega})$ , its 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:  $F_s = 100$  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

An IIR filter, meets the above specifications, is required to be designed in the s-plane and then transformed to the z-plane using bilinear transform:

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

Fall Semester 23/01/2017

Name: \_\_\_\_\_

ID: \_\_\_\_\_

CR: \_\_\_\_\_

**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 Level
Q1	60		
Q2-(a)	10		
Q2-(b)	10		
Q3	20		
Total	100		

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

1. The discrete signal  $x(n] = e^{j\pi n/4}$  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  $x_s(t)$  is  $\Omega_c$ , what is the Nyquist rate for  $x_s(2t)$ 
  - a)  $2\Omega_c$
  - b)  $\Omega_c/2$
  - c)  $\Omega_c$
  - d)  $\Omega_c/4$
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:

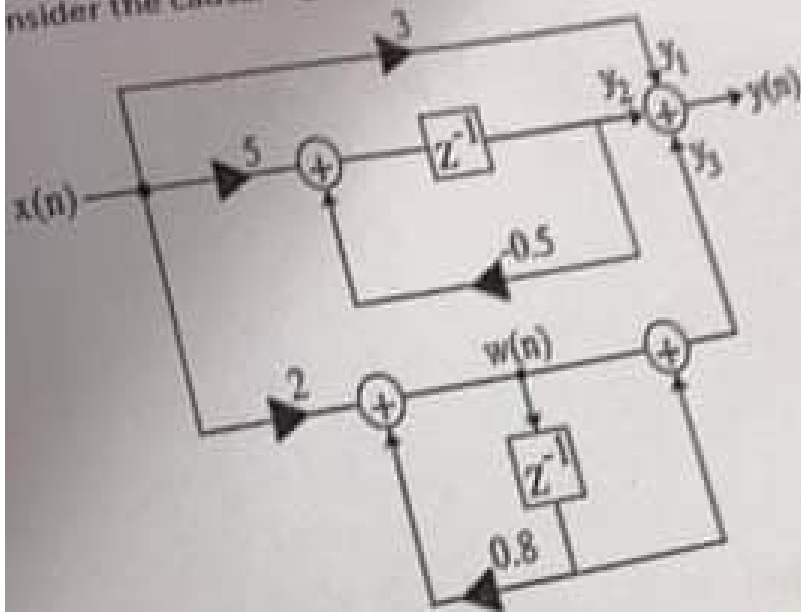
- Lowpass filter: 0 – 10 KHz passband
- Maximum passband ripple: -3dB (at 10 KHz)
- Sampling frequency:  $f_s = 100$  KHz
- Transition band: 10 KHz to 20 KHz
- Stopband attenuation: -10dB (starting at 20 KHz)
- The filter must be monotonic in the pass and stop bands, (i.e. no ripple).

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

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

(20 marks)

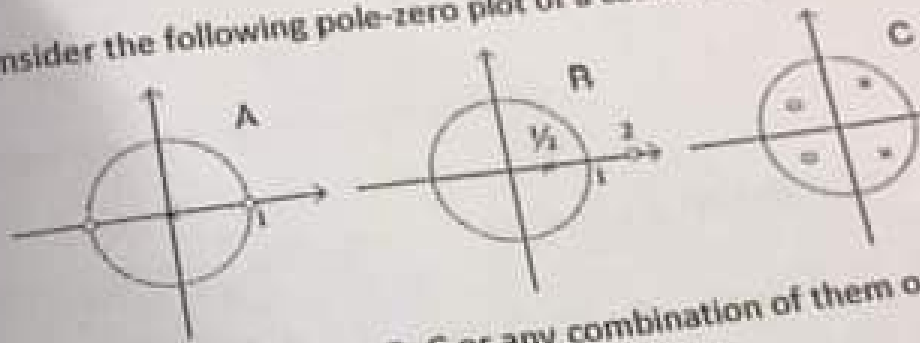
Consider the causal digital filter shown in the following figure.



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

بسم الله الرحمن الرحيم

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



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

- a) All-pass \_\_\_\_\_
- b) Minimum phase \_\_\_\_\_
- c) Stable \_\_\_\_\_
- d) IIR \_\_\_\_\_
- e) FIR \_\_\_\_\_

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

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 system

is an ideal Low Pass Filter (not bandpass) with

Find minimum order of the designed IIR filter,  $N$ , and the corresponding cut-off frequency,  $\Omega_c$ . Show all steps. (4pts)

iii) Find system function  $H_a(s)$  of the designed analogue filter and the corresponding digital filter.

End of exam

مسألة للتمرين

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

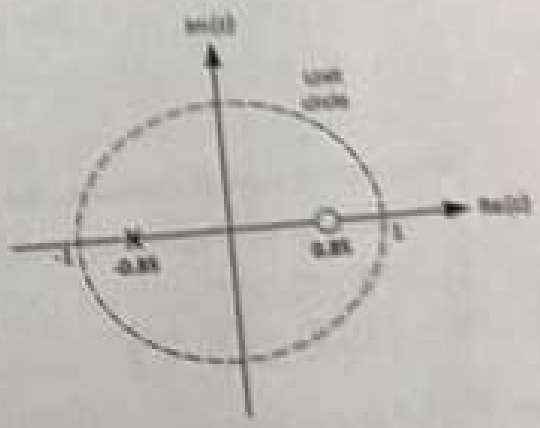
- a) 40 Hz
- b) 80 Hz
- c) 100 Hz
- d) 120 Hz

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

- a) 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 100Hz, the magnitude response of this system at frequency 30Hz is:

- a) 1
- b) 0
- c) 12.31
- d) 0.081



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  $h_1[n]$  and  $h_2[n]$ , the response of the second system is the response of the first

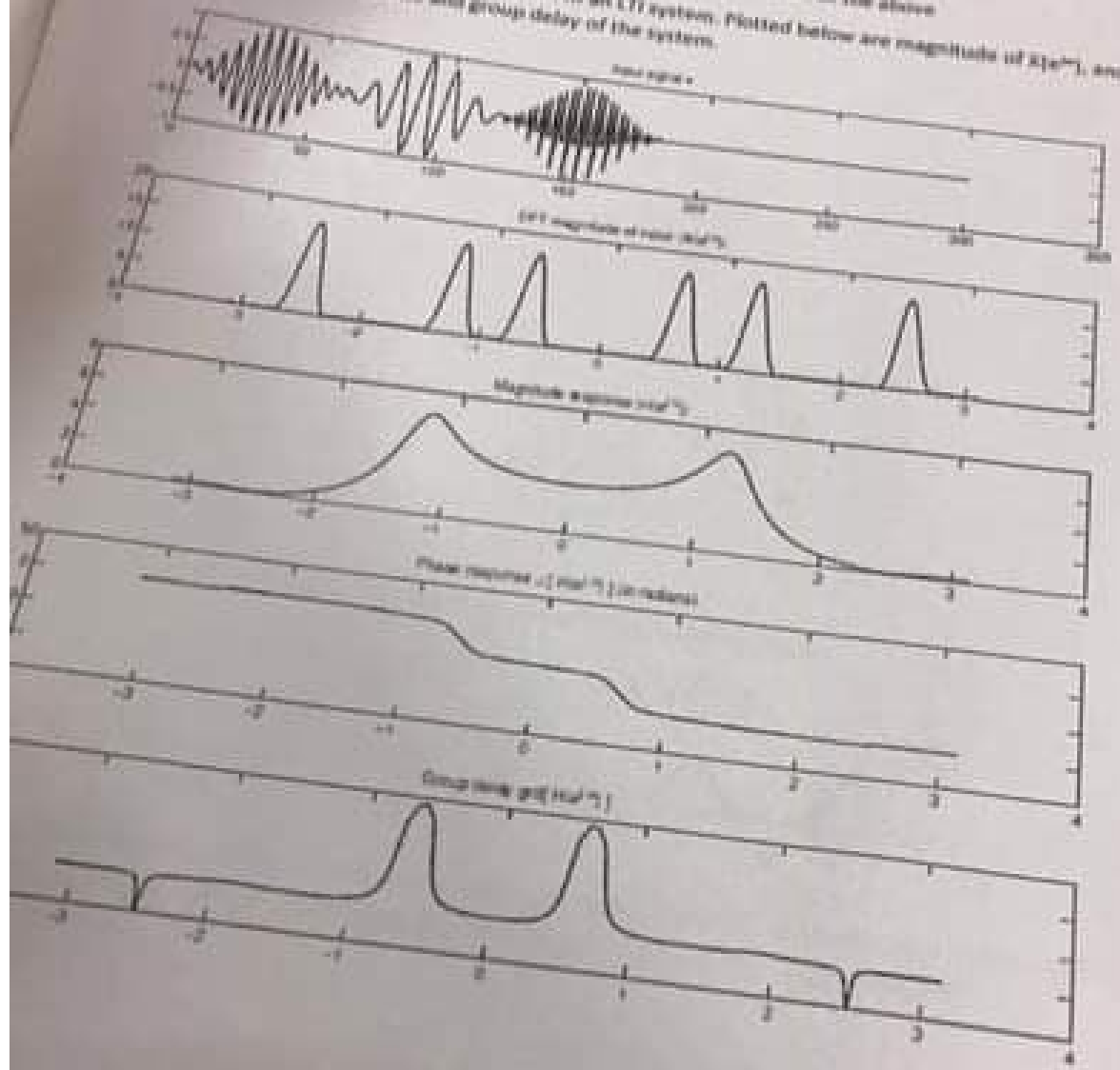
- a) Amplitude scaled by 3 and delayed by 2
- b) Amplitude scaled by 3 and advanced by 2
- c) Amplitude scaled by 3
- d) None of the above

21. The system having frequency response  $H(e^{j\omega}) = \frac{2e^{-j\omega}}{1 + 2e^{-j2\omega}}$  is
- (a) low-pass filter
  - (b) high-pass filter
  - (c) band-pass filter
  - (d) all-pass filter

- (a)  $y(n-1] = \frac{1}{2}x[n] + \frac{1}{2}x[n-2]$       (b)  $y[n] = 2x[n-2] - 2x[n-1]$       (c)  $y[n] = x[n] + 2x[n-1]$
- (d)  $y[n] = 2x[n-1] - 2x[n]$       (e)  $y[n] = x[n-1] - 2x[n-2]$

22. If a continuous-time filter with an impulse response,  $h_c(t)$ , is sampled with a sampling frequency  $F_s$ , happens to the cutoff frequency,  $\omega_c$ , of the discrete-time filter as  $F_s$  increased?
- (a)  $\omega_c$  increases
  - (b)  $\omega_c$  decreases
  - (c)  $\omega_c$  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 its response, phase response and group delay of the system.





## تمرين الاختبار الرابع

a) Resonant filter

b) Notch filter

c) Low-pass filter

d) High-pass filter

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

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,  $h_c(t)$ , is sampled with a sampling frequency  $F_s$ , what happens to the cutoff frequency,  $\omega_c$ , of the discrete-time filter as  $F_s$  increased?

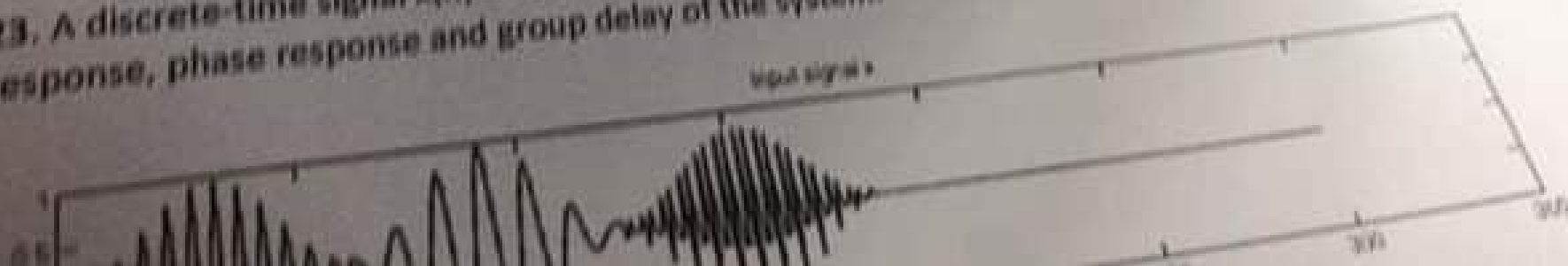
a)  $\omega_c$  increases

b)  $\omega_c$  decreases

c)  $\omega_c$  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.



e) FIR

For questions #16-17:  $T=0.2$ . The analogue filter is:  $H_a(s) = \frac{10}{s+10}$  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)  $\frac{2z}{z-e^{-1}}$       c)  $\frac{20z}{z-e^{-1}}$       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)  $\rightarrow$   $\boxed{| 3}$   $\rightarrow$   $\boxed{| 2}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$       (b)  $\rightarrow$   $\boxed{| 3}$   $\rightarrow$   $\boxed{| 2}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$       (c)  $\rightarrow$   $\boxed{| 3}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$   $\boxed{| 2}$   $\rightarrow$   
(d)  $\rightarrow$   $\boxed{| 3}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$   $\boxed{| 2}$   $\rightarrow$       (e)  $\rightarrow$   $\boxed{| 2}$   $\rightarrow$   $\boxed{| 3}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$       (f)  $\rightarrow$   $\boxed{| 2}$   $\rightarrow$   $\boxed{\text{LPF}}$   $\rightarrow$   $\boxed{| 3}$   $\rightarrow$

19.  $x[n] = \cos\left(\frac{\pi}{2}n\right)$  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:

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

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

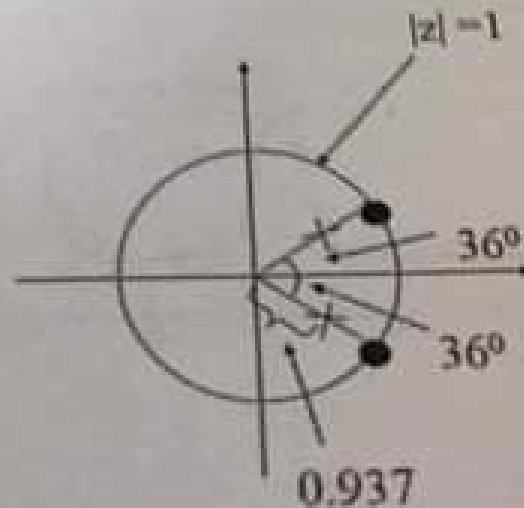
LPF is an ideal Low Pass Filter (not bandpass) with a single cutoff frequency.



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)  $-0.5 \sin(\frac{\pi}{2}n)$

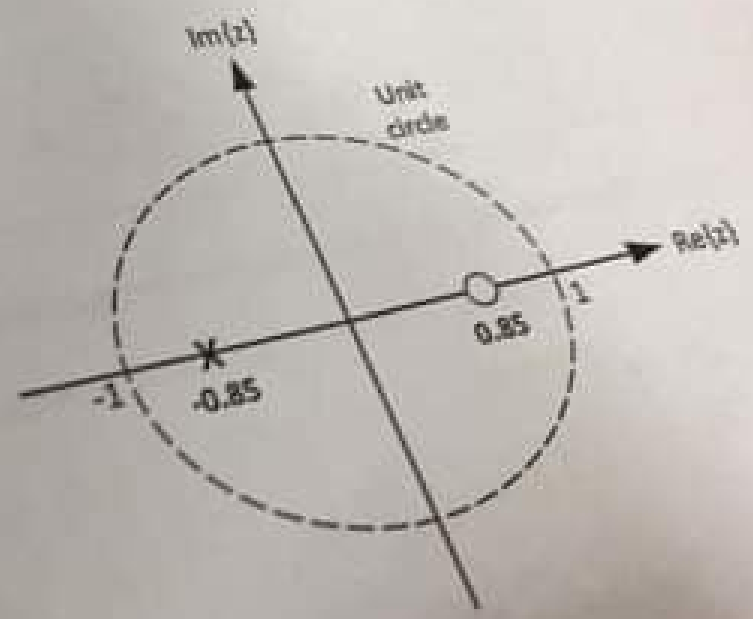
20. The pole-zero diagram of a digital filter is shown in the figure below. This filter is:



- 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 50Hz is:

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



...ike response of an LTI system, then the output y[n]

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