DSP , Final Exam (AYHAM HASHESH)

1. The discrete signal $x(n) = e^{j(3\pi n/4)}$

a. Periodic with a Period of 3

b.

c. " " " " " 4

d. " " " " " 8

Non - Periodic

$\frac{3\pi}{4}$ Period

$WN = 2\pi K$

$N = \frac{2\pi K}{W}$

$\Rightarrow N = \frac{4}{3\pi} \cdot 2\pi K = \frac{8}{3} K$

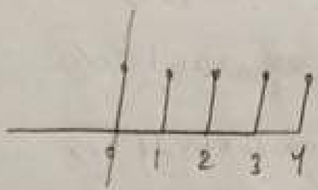
take $K = 3$

$\Rightarrow \boxed{N = 8}$

2. If the Nyquist rate for $x_a(t)$ is Ω_s What is the Nyquist rate for $x_a(2t)$

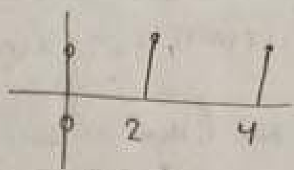
a. $2\Omega_s$ b. $\frac{\Omega_s}{2}$ c. Ω_s d. $\frac{\Omega_s}{4}$

assume $x(t)$



$x(2t)$

$2t$: compression



3. IIR Filter

a. use feed back

b. Are sometimes called recursive filters

c. Can oscillate (i.e not stable) if not properly design

d. All of the above

4. Find two different Continuous-time signals that will produce the
$X(n) = \cos(0.15\pi n)$ when sampled with a sampling frequency of 8K

a. $\sin(1200\pi t)$ and $\cos(17200\pi t)$ ×

b. $\cos(1200\pi t)$ and $\sin(17200\pi t)$

c. $\cos(1200\pi t)$ and $\cos(17200\pi t)$

d. $\sin(1200\pi t)$ and $\sin(17200\pi t)$ ×

$\cos(0.15\pi n)$

$0.15\pi * f_s = 1200\pi \Rightarrow$ Both a & d wrong

$\sin\left(\frac{17200\pi}{8000}\right) = \sin(2.15\pi) = 0.0375$

$\cos\left(\frac{17200}{8000}\pi\right) = \cos(2.15\pi) = 0.999$

$\cos(0.15\pi) = 0.999$

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 Reject filter

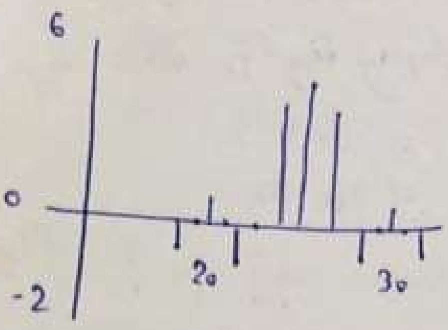
البسط : Corresponding with input
المقام : " " output

$$H(z) = \frac{0.25 + 0.25z^{-1} + 0.25z^{-2} + 0.25z^{-3}}{1}$$

$h(n) = \{\frac{1}{4}, \frac{1}{4}, \frac{1}{4}, \frac{1}{4}\} \Rightarrow$ Moving average
$\Rightarrow$ LPF

z^{-n} ← index

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

Symmetry ⇒ FIR

7. Coefficients symmetry is important in FIR Filters because it Provides

a. A smaller transition bandwidth

b. Less Passband ripple

c. " Stopband "

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 " "

c. Convolving " "

d. Averaging their " and then use hamming window

9. If a linear phase filter has a phase response of 80 degrees at 400 HZ. what will its Phase response freq. of 200 HZ (assuming that both freq. are in the Passband of the filter

a. 35° b. 40° c. 45° d. 80°

* Linear phase ⇒ نسبة وتناسب

freq degree

400 ⟶ 80

200 ⟶ X

$400X = 1600$

$X = 40$

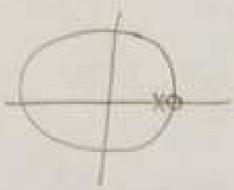
3

10. The Continuous time signal $x(t) = \sin(20\pi t) + \cos(40\pi t)$ is sampled with a sampling freq 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 freq F_s which is consistent with the information is

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

$$\frac{20\pi}{F_s} = \frac{\pi}{5} \Rightarrow F_s = 100 \text{ Hz}$$

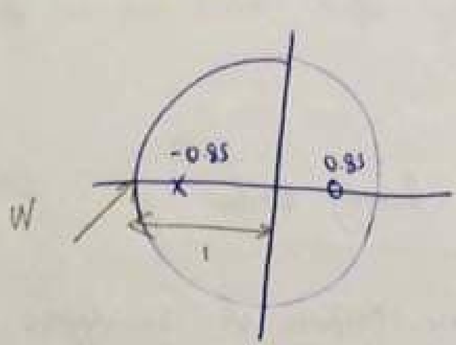
11. A DSP system with the following system function $H(z) = \frac{0.97(z-1)}{z-0.94}$ is a :- a. LPF b. ~~HPF~~ Band Pass Filter (Reso..) c. HPF d. = Stop = (Notch)



Zeros at $z = 1$

Poles at $z = 0.94$

12. The Pole/Zero diagram of DSP system is shown in the following figure. If this system Implemented with $f_s = 10$ KHz the Mag. response of this system at freq = 5K Hz is



a. 1 b. 0

c. 12.33 d. 0.81

$$W = \frac{2\pi f}{f_s} = \frac{2\pi\, 5K}{10K} = \pi$$

$$\pi \times \frac{180}{\pi} = 180^\circ$$

$$\text{Mag} = \frac{\text{Distance with zero}}{\text{Distance with Poles}} = \frac{1.85}{0.15} = 12.33$$

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

14. if an input signal is applied to two LTI system with respective unit impulse resonance $h(n)$ and $3h(n-2)$, then the response of the second system is the response of the first system

a. Amplitude scaled by 3 and delayed by 2

b. " " " 3 " advanced " 2

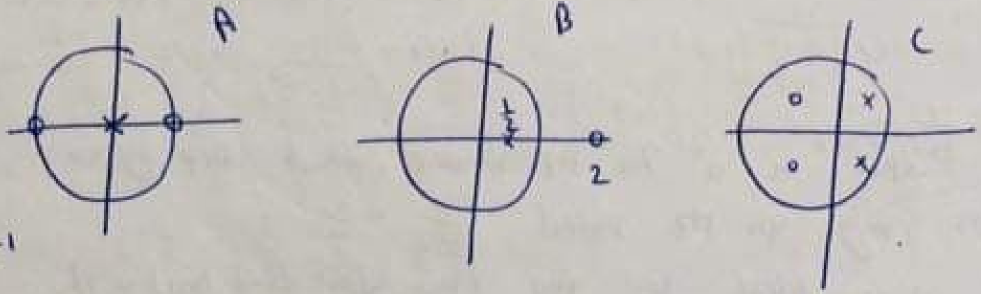
c. " " " 3

d. None

Note Min phase ⇒ causal & stable [& its inverse] [zero & Poles in U.C]
All Pass ⇒ zero & Poles in conjugate, if Pole on $\frac{1}{2}$ ⇒ zero on 2 [illegible]

15. Consider the following Pole-Zero plot of a causal LTI system [illegible]

decide if All-pass, Minimum phase, stable, IIR, and FIR



$\frac{1}{1-0z^{-1}}$

A	B	C
Stable	Stable	Stable
FIR	IIR	IIR ⇒ Poles exist
~~All-pass~~	All-Pass	Minimum phase ⇒ U.C جوا Z/P [illegible]

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* For question 16 & 17, $T = 0.2$

analogue filter is $H_a(s) = \frac{10}{s+10}$

$h_a(t) = 10e^{-10t}u(t)$

16. $h(n)$ designed using impulse invariance technique is

a. $2e^{-2n}u(n)$ sampling $h_a(t)$
b. $10e^{-10n}u(n)$ sampling $H_a(s)$
c. $20e^{-20n}u(n)$
d. $0.1e^{-0.1}u(n)$
e. $0.2e^{-0.2n}u(n)$

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

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

17 $S = \frac{2}{T}\left[\frac{1-z^{-1}}{1+z^{-1}}\right]$

substitute in $H(s)$

$\Rightarrow H_s(s) = \frac{10}{\frac{2}{0.2}\left[\frac{1-z^{-1}}{1+z^{-1}}\right]+10}$

$\Rightarrow \frac{\cancel{10}}{\cancel{10}\left[\frac{1-z^{-1}}{1+z^{-1}}\right]+\cancel{10}\,1}$

$\Rightarrow \frac{1+z^{-1}}{1-z^{-1}+1+z^{-1}} = \frac{1}{2}(1+z^{-1})$

16 Impulse Invariance
Map pole to pole in Z-plan
at $S = S_k \Rightarrow Z = e^{S_k T}$

$H_s(s) = \frac{10}{S+10} = 10\frac{1}{S+10} = \underbrace{10e^{-10t}u(t)}_{h(t)}$

$h(n) = 10e^{-10nT}u(t) = 10e^{-2}u(t)$ ($T = 0.2$)

$H(z) = \frac{10}{1-e^{-2}z^{-1}} * 0.2 = 2e^{-2n}u(n)$

18. if input is a 300 Hz sinusoid, which DSP system outputs only 450 Hz sinusoid $\frac{450}{300} = \frac{3}{2}$

* LPF is an ideal low pass filter (not band pass) with a single cut off freq.

a. $\rightarrow \boxed{3\uparrow} \rightarrow \boxed{\downarrow 2} - \boxed{LPF}$
b. $\rightarrow \boxed{\downarrow 3} \rightarrow \boxed{\uparrow 2} \rightarrow \boxed{LPF}$
c. $\boxed{\uparrow 3} \rightarrow \boxed{LPF} \rightarrow \boxed{\downarrow 2}$
d. $\boxed{\downarrow 3} \rightarrow \boxed{LPF} \rightarrow \boxed{\uparrow 2}$
e. $\boxed{\uparrow 2} \rightarrow \boxed{\downarrow 3} \rightarrow \boxed{LPF}$
f. $\boxed{2\uparrow} \rightarrow \boxed{LPF} \rightarrow \boxed{\downarrow 3}$

$x(n) = \cos(\frac{\pi}{2}n)$ is an input into LTI system with

19. $H(e^{jw}) = 1 + \frac{1}{2}e^{jw} + e^{j2w}$. 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. $-\frac{1}{2}\sin(\frac{\pi}{2}n)$

$$H(e^{jw}) = 1 + \frac{1}{2}[\cos w + j\sin w] + \cos 2w + j\sin 2w$$

$$|H(e^{jw})| = \sqrt{(1 + \frac{1}{2}\cos w + \cos 2w)^2 + (\frac{1}{2}\sin w + \sin 2w)^2}$$

$$= \sqrt{\left[1 + \frac{1}{2}\cos(\frac{\pi}{2})^{0} + \cos(2\frac{\pi}{2})^{-1}\right]^2 + \left[(\frac{1}{2}\sin\frac{\pi}{2})^{\frac{1}{2}} + \sin(2\cdot\frac{\pi}{2})^{0}\right]^2}$$

$$= \sqrt{(1+0-1)^2 + (\frac{1}{2})^2}$$

$$= 1/2$$

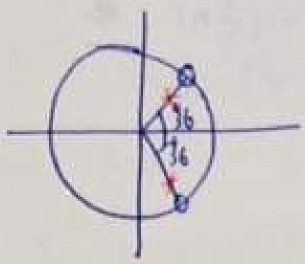
$$\angle H(e^{jw}) = \tan^{-1}\frac{\sin w + \sin 2w}{1 + \frac{1}{2}\cos w + \cos 2w}$$

$$= \tan^{-1}\frac{1+0}{1+0-1} \Rightarrow \theta = \frac{\pi}{2}$$

$$y(n) = \frac{1}{2}\cos(\frac{\pi}{2}n + \frac{\pi}{2})$$

$$= \frac{1}{2} * -\sin(\frac{\pi}{2}n)$$

20. the Pole zero diagram of a digital filter is shown in the figure below. This filter is



Poles ⇒ $0.937 \angle 36$

a. Resonant filter

b. Notch filter

c. LPF

d. HPF

21. The system Having freq Response $H(e^{jw}) = \frac{2e^{-jw}}{1+3e^{-j2w}}$ is

corresponding for Input

corresponding for output ⇐

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

c. $y(n) - x(n) + 3x(n-2)$

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

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 $h_c(t)$ has sampled freq $= f_s$, what happen to cutoff freq. w_c of the discrete time filter as f_s increased.

a. w_c increase

b. w_c decrease

c. w_c remain constant

d. None

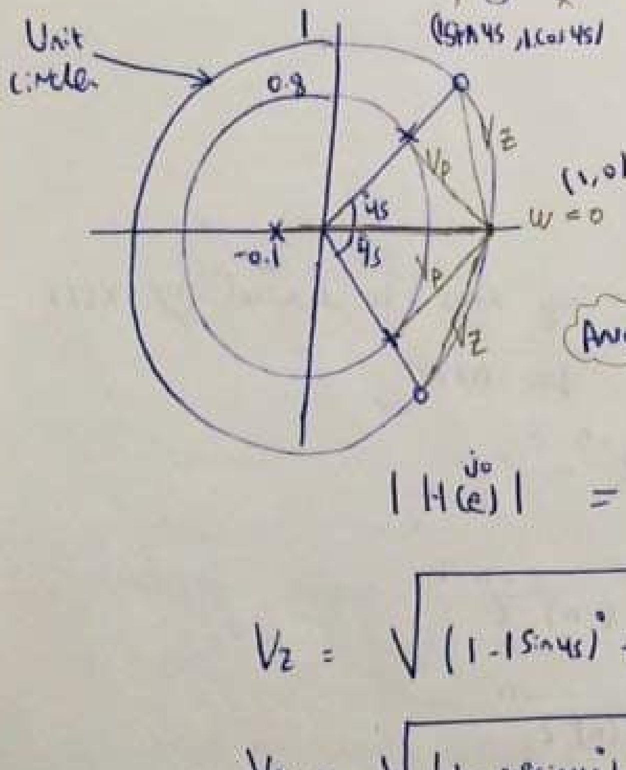
FR : Geometric Interpolation

حاصل ضرب

Magnitude : $H(e^{jw})$ = is Product of Lengths of Vectors from Zeros divide by Product of Lengths of Vectors from Poles

Phase : $\theta(w)$ = is sum of angles of Vectors from Zeros mins sum of angles of vector from Poles

Ex Consider the following figure



Find Mag & Phase at $w=0$ without finding $H(e^{jw})$

Answer $\angle H(e^{j0}) = 45^\circ + 45^\circ - 45^\circ - 45^\circ$
$= 0^\circ$

$$|H(e^{j0})| = \frac{V_z^2}{1.1 * V_p^2}$$

$$V_z = \sqrt{(1-1\sin 45)^2 + (0-1\cos 45)} = 0.7653$$

$$V_p = \sqrt{(1-0.8\sin 45) + (0-0.8\cos 45)} = 0.7132$$

$\Rightarrow$ Answer = 1.3678

Ex the Z-transform of a discrete seq. $x(n)$ is denoted by $X(z)$ derive the z transform of seq. $n x(n)$

Answer

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n}$$

$$\frac{dX(z)}{dz} = \sum x(n) (-n) z^{-n-1}$$

$$= -z^{-1} \sum n x(n) z^{-n}$$

$$= -z^{-1} \underbrace{Z\{n x(n)\}}_{\text{Z-Transform}}$$

$$Z\{n x(n)\} = -z \frac{dX(z)}{dz}$$

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