Modified 29/6/2019 3-50pm

CBCS SCHEME

USN

15EE63

Sixth Semester B.E. Degree Examination, June/July 2019
Digital Signal Processing

Time: 3 hrs. Max. Marks: 80

Note: ***Answer any FIVE full questions, choosing ONE full question from each module.***

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

Module-1

1 a. Determine DFT of sequence $x(n) = \frac{1}{3}$ for $0 \le n \le 2$ for N = 4. Plot magnitude and phase spectrum. **(08 Marks)**

b. Two length – 4 sequence are defined below :

$$x(n) = \cos\left(\frac{\pi n}{2}\right) \quad n = 0, 1, 2, 3$$

$$h(n) = 2^n \quad n = 0, 1, 2, 3$$

i) Calculate $x(n) \circledast_4 h(n)$ using circular convolution directly.

ii) Calculate $x(n) \circledast_4 h(n)$ using Linear convolution. **(08 Marks)**

OR

2 a. Compute circular convolution using DFT + IDFT for following sequence :

$$x_1(n) = \left\{\underset{\uparrow}{2}, 3, 1, 1\right\}, \quad x_2(n) = \left\{\underset{\uparrow}{1}, 3, 5, 3\right\}.$$ **(08 Marks)**

b. Find the output of the LTI system whose impulse h(n) = {1, 1, 1} and the input signal is x(n) = {3, –1, 0, 1, 3, 2, 0, 1, 2, 1}. Using the overlap save method. Use 6-pt circular convolution. **(08 Marks)**

Module-2

3 a. What are FFT algorithms? Explain the advantages of FFT algorithms over the direct computations of DFT for a sequence x(n). **(04 Marks)**

b. What are the differences and similarities between DIT and DIF –FFT algorithms? **(04 Marks)**

c. Find the 8-pt DFT of the sequence x(n) = {1, 2, 3, 4, 4, 3, 2, 1}. Using DIT – FFT radix – 2 algorithm. **(08 Marks)**

OR

4 a. Find the 4-pt circular convolution of x(n) and h(n) given. Using radix-2 DIF – FFT algorithm. **(08 Marks)**

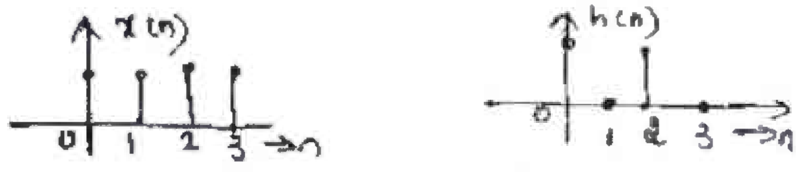

Fig.Q4(a)

b. Given x(n) = (n + 1) and N = 8. Determine X(K). Using DIF – FFT algorithm. **(08 Marks)**

Module-3

5 a. Convert the analog filter with system transfer function :

$$H(s) = \frac{(s+0.1)}{(s+0.1)^2 + 3^2}$$

into a digital IIR filter by mean of the impulse invariant method. **(06 Marks)**

b. Design a butter worth digital IIR lowpass filter using bilinear transformation by taking $T = 0.1$sec, to satisfy the following specification :

$$0.6 \le \left|H(e^{j\omega})\right| \le 1.0; \quad \text{for } 0 \le \omega \le 0.35\pi$$
$$\left|H(e^{j\omega})\right| \le 0.1; \quad \text{for } 0.7\pi \le \omega \le \pi$$

(10 Marks)

OR

6 a. Compare analog and digital filters. **(04 marks)**

b. Determine the poles of lowpass Butterworth filter for $N = 2$. Sketch the location of poles on s-plane and hence determine the normalized transfer function of lowpass filter. **(08 Marks)**

c. Write difference between IIR and FIR filter. **(04 Marks)**

Module-4

7 a. Design a Chebyshev digital IIR lowpass filter using impulse invariant transformation by taking $T = 1$ sec to satisfy the following specifications;

$$0.9 \le \left|H(e^{j\omega})\right| \le 1.0; \quad \text{for } 0 \le \omega \le 0.25\pi$$
$$\left|H(e^{j\omega})\right| \le 0.24; \quad \text{for } 0.5\pi \le \omega \le \pi$$

Draw direct form – I and II structure of the filter. **(12 Marks)**

b. Write the relation between analog and digital frequency in Billnear transformation. **(04 Marks)**

OR

8 a. Obtain the direct form – I, direct form II realization of the LTI system governed by the relation.

$$y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64y}y(n-3) + x(n) + 3x(n-1) + 2x(n-2).$$ **(08 Marks)**

b. Realize the given system in cascade and parallel form :

$$H(z) = \frac{1+0.25z^{-1}}{(1-2z^{-1}+0.25z^{-2})(1-3z^{-1}+0.25z^{-2})}.$$ **(08 Marks)**

Module-5

9 a. The frequency response of a filter is described by : $H(\omega) = j\omega,\ -\pi \le \omega \le \pi$. Design the filter using a rectangular window. Take $N = 7$. **(08 Marks)**

b. Design a lowpass digital filter to be used in A/D – H(z) – D/A structure that will have – 3dB cutoff at 30π rad/sec and attenuation factor of 5dB at 45π rad/sec. The filter is required to have a linear phase and the system will use sampling frequency of 100 samples/sec. **(08 Marks)**

OR

10 a. Deduce the equation for the following frequency spectrum for rectangular window sequence defined by :

$$w_f(n) = \begin{cases} 1, & \dfrac{-(N-1)}{2} \le n \le \dfrac{N-1}{2} \\ 0, & \text{otherwise} \end{cases}.$$

(06 Marks)

b. A lowpass filter has the desired frequency response :

$$H_d(\omega) = \begin{cases} e^{-j\omega 3}, & 0 < \omega < \pi/2 \\ 0, & \text{otherwise} \end{cases}.$$

Determine h(n) based on frequency sampling method. Take $K = 7$. **(06 Marks)**

c. Realize the linear phase FIR filter having the following impulse response :

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4).$$

(04 Marks)

* * * * *

Gmail pmanjunath p <pmanjunathvtu@gmail.com>

New Doc 2019-06-29 11.53.45 - Page 1

4 messages

Manja Naik <manjubdt2009@gmail.com> Sat, Jun 29, 2019 at 12:47 PM
To: pmanjunathvtu@gmail.com

Sir
For question 4a wrong solution uploaded pls rectify
Correct solution is attached

New Doc 2019-06-29 11.53.45_1.pdf
221K

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pmanjunath p <pmanjunathvtu@gmail.com> Sat, Jun 29, 2019 at 12:56 PM
To: shivashantala@gmail.com

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pmanjunath p <pmanjunathvtu@gmail.com> Sat, Jun 29, 2019 at 3:12 PM
To: sumathisrinivasan1@gmail.com

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sumathi srinivasan <sumathisrinivasan1@gmail.com> Sat, Jun 29, 2019 at 3:44 PM
To: pmanjunath p <pmanjunathvtu@gmail.com>

Sir , The solution to question 4.a of the subject 15EE63 attached in this mail is correct.
[Quoted text hidden]

" APPROVED ".
Registrar (Evaluation)
Visvesvaraya Technological Univers
BELAGAVI - 590018

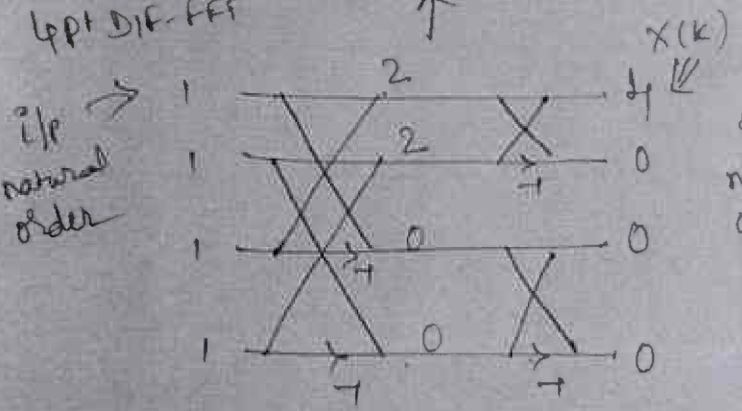
Paper :- 15EE63. Digital Signal Processing

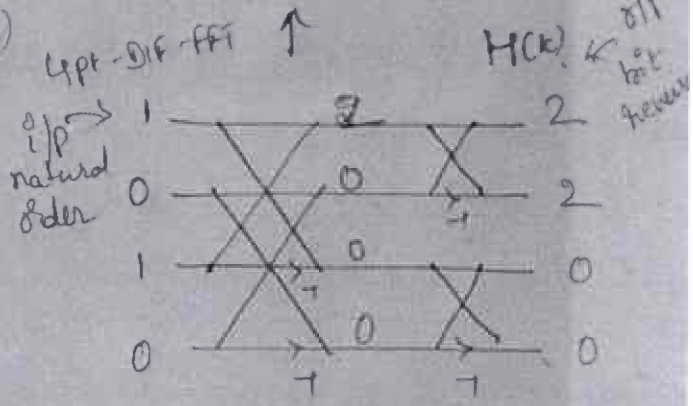
Q. No. 4.a

for above Question no., wrong solution has been uploaded in scheme.

The correct soln is given below

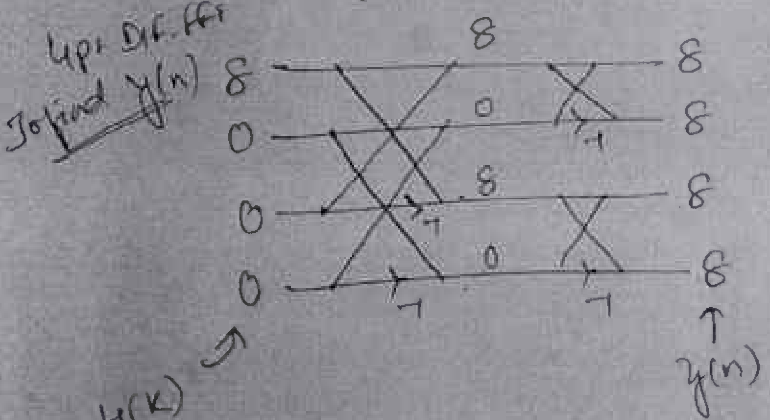
$x(n) = \{\underset{\uparrow}{1}, 1, 1, 1\}$ $h(n) = \{\underset{\uparrow}{1}, 0, 1, 0\}$





$X(k) = \{\underset{\uparrow}{4}, 0, 0, 0\}$ $H(k) = \{\underset{\uparrow}{2}\ 0\ 2\ 0\}$

$Y(k) = X(k) \cdot H(k) = \{8, 0, 0, 0\}$



$y(n) = \frac{1}{N}\{\underset{\uparrow}{8}, 8, 8, 8\}$

$= \frac{1}{4}\{8, 8, 8, 8\}$

$y(n) = \{2, 2, 2, 2\}$

"APPROVED"

Registrar (Evaluation)
Visvesvaraya Technological University
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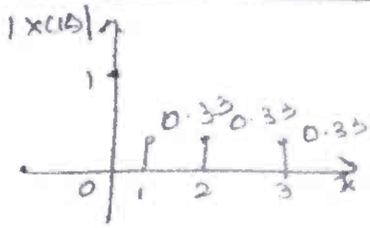
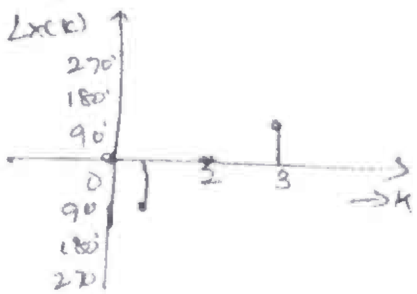
Visvesvaraya Technological University

Belagavi, Karnataka - 590 018

Signature of Scrutinizer

Scheme & Solution

Subject Title: Digital signal processing **Subject Code:** 15EE63

Question Number	Solution	Marks Allocated
1 a)	$x(n) = \{\underset{\uparrow}{1/3}, 1/3, 1/3\}$	
	$N = 4$	
	$\therefore x(n) = \{\underset{\uparrow}{1/3}, 1/3, 1/3, 0\}$	1
	$X(k) = \sum_{n=0}^{N-1} x(n)\, e^{\frac{-j2\pi nk}{N}}$	1
	$X(k) = \{1, -j/3, 1/3, 1/3\,j\}$	4
	magnitude response phase plot	
	 	1+1 = 2
		8
b)	Given	
	$x(n) = \{1, 0, -1, 0\}$	
	$h(n) = \{1, 2, 4, 8\}$	
	using circular convolution directly	
	$y(n) = x(n) \circledast_4 h(n)$	
	$y(n) = \begin{bmatrix} 1 & 0 & -1 & 0 \\ 0 & 1 & 0 & -1 \\ -1 & 0 & 1 & 0 \\ 0 & -1 & 0 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 2 \\ 4 \\ 8 \end{bmatrix}$	3
	$y(n) = \{-3, -6, 3, 6\}$	

Subject Title: Digital signal processing   Subject Code: 15EE63

Question Number	Solution	Marks Allocated

using Linear convolution

$$y(n) = x(n) * h(n)$$

$$\begin{array}{c|cccc} & 1 & 0 & -1 & 0 \\ \hline 1 & 1 & 0 & -1 & 0 \\ 2 & 2 & 0 & -2 & 0 \\ 4 & 4 & 0 & -4 & 0 \\ 8 & 8 & 0 & -8 & 0 \end{array}$$

3

$$y(n) = \{1, 2, 3, 6, -4, -8\}$$

* circular convolution is nothing but linear convolution plus aliasing

$$y(n) = y_l(0) + y_l(4)$$
$$= 1 - 4 = -3$$

2

$$y_c(1) = y_l(1) + y_l(5)$$
$$= 2 - 8 = -6$$

8

$$y_c(2) = y_l(2) = 3$$
$$y_c(3) = y_l(3) = 6$$

2a) $x_1(n) = \{\underset{\uparrow}{2}, 3, 1, 1\}$, $x_2(n) = \{\underset{\uparrow}{1}, 3, 5, 3\}$

$N = 4$

$$DFT\{x_1(n)\} = X_1(k)$$

$$X_1(k) = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ -1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} 2 \\ 3 \\ 1 \\ 1 \end{bmatrix}$$

3

$$= \{7, 1-2j, -1, 1+2j\}$$

$$DFT\{x_2(n)\} = X_2(k)$$

$$X_2(k) = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -j & -1 & j \\ -1 & -1 & 1 & -1 \\ 1 & j & -1 & -j \end{bmatrix} \begin{bmatrix} 1 \\ 3 \\ 5 \\ 3 \end{bmatrix}$$

3

$$X_2(k) = \{12, -4, 0, -4\}$$

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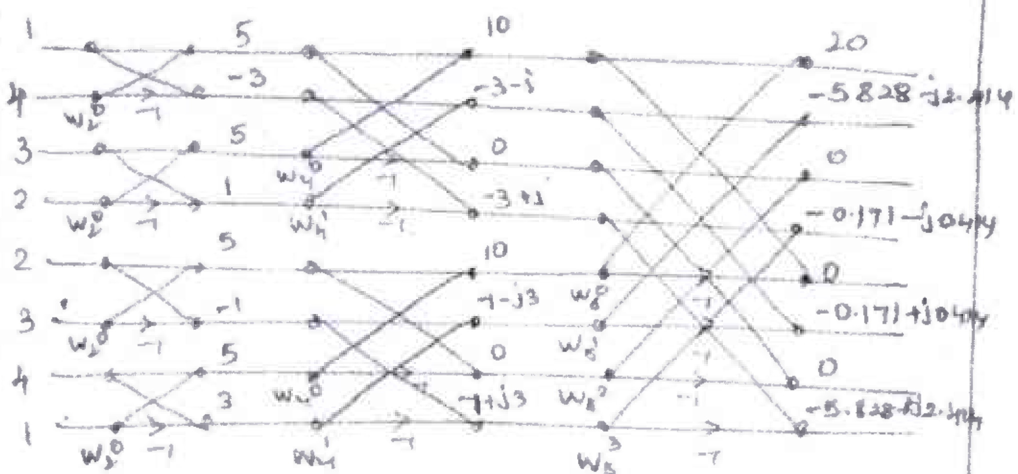
Question Number	Solution	Marks Allocated
	$y(k) = x_1(k) \cdot x_2(k)$ $= [84, -4+8j, 0, -4-8j]$ $IDFT\{y(k)\} = \frac{1}{N}\left[\sum_{k=0}^{N-1} y(k) W_N^{-nk}\right]$	2
	$= \{19, 17, 23, 25\}$	8
b)	$h(n) = \{1, 1, 1\}$ $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ $N = 2^M = 2^3 = 8$ $L = N - M + 1$ $L = 6 - 3 + 1$ $\boxed{L = 6}$	1
	$h(n) = \{1, 1, 1, 0, 0, 0, 0, 0\}$ $x_1(n) = \{0, 0, 3, -1, 0, 1, 3, 2\}$ $x_2(n) = \{3, 2, 0, 1, 2, 1, 0, 0\}$ $y_1(n) = x_1(n) * h(n)$ $y_1(n) = \{5, 2, 3, 2, 2, 0, 4, 6\}$	3
	$y_2(n) = x_2(n) * h(n)$ $y_2(n) = \{3, 5, 5, 3, 3, 4, 3, 1\}$	3
	O/P $y(n) = \{3, 2, 2, 0, 4, 6, 5, 3, 3, 4, 3, 1\}$	1
		8
	Module - 2	
3a)	Fast fourier Transform is an algorithm used to compute the DFT. It makes use of the symmetry and periodicity properties of Twiddle factor W_N^k to reduce DFT computations time	2

Subject Title: Digital signal processing Subject Code: 15EE63

Question Number	Solution	Marks Allocated
	Advantages:	
	1. The computations complexity of FFT algorithm greatly reduced compared to direct computation	
	2. FFT algorithms can be used to calculate DFT as well as IDFT very efficiently	2M
		4M
b)	Differences:	
	1. In DIT, the i/p is bit reversed, while the o/p is in normal order	
	2. In DIF, the i/p is normal order, and o/p is bit reversed	2
	similarities:	
	1. Both the algorithm require same number of operations to compute DFT.	
	2. Both algorithm require bit reversal at some place during computations	2
		4M
c)	$x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$.	



6

$X(K) = \{20, -5.828 - j2.414, 0, -0.171 - j0.414, 0, -0.171 + j0.414, 0, -5.828 + j2.414\}$ 2

8

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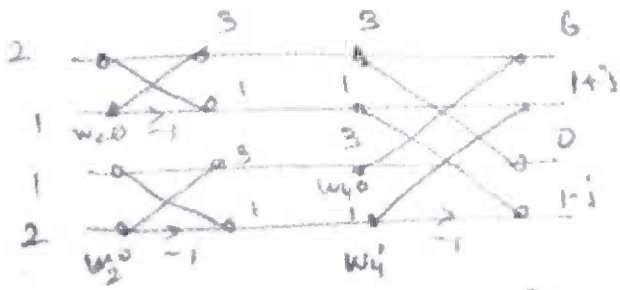
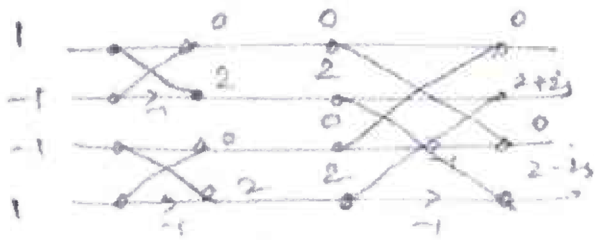
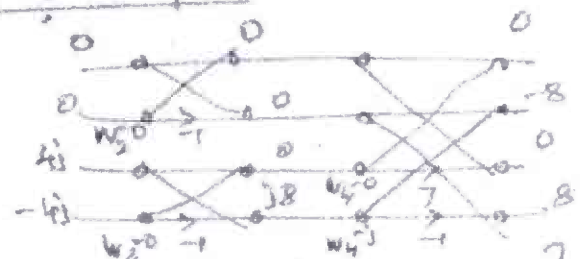
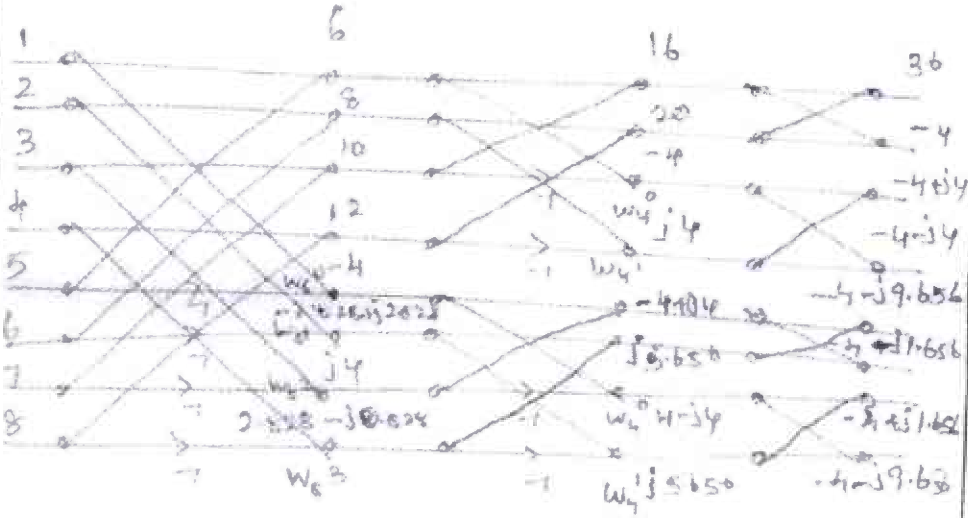
Question Number	Solution	Marks Allocated
4a	sol:	
	$x_1(n) = \{2, 1, 1, 2\}$, $x_2(n) = \{1, -1, -1, 1\}$	
	To find $X_1(K)$ using DIT FFT	
		2
	$X_1(K) = \{6, 1+j, 0, 1-j\}$	
	To find $X_2(K)$ using DIT FFT.	
		2
	$X_2(K) = \{0, 2+2j, 0, 2-2j\}$	
	$Y(K) = X_1(K) \cdot X_2(K) = \{0, 4j, 0, -4j\}$	1
	To find $y(n)$	
		3
		8
	$y(n) = \{0, -2, 0, 2\}$	

Subject Title: Digital signal processing Subject Code: 15EE63

Question Number	Solution	Marks Allocated
b)	$x(n) = (n+1)$, $N = 8$ $n = 0, 1, \cdots N-1$	



6

$X(K) = \{36, -4-j9.656, -4+4j, -4+j1.656, -4,$
$-4-j1.656, -4-4j, -4-j9.656\}$ 2

8M

MODULE-3

5a) $H(S) = \frac{S+0.1}{(S+0.1)^2+3^2}$

using transformation equation / impulse invariant.

$$\frac{(S+a)}{(S+a)^2+b^2} = \frac{1-e^{-aT}(\cos bT)z^{-1}}{1-2e^{-aT}(\cos bT)z^{-1}+e^{-2aT}z^{-2}}$$ 2

$$H(z) = \frac{1+0.8938z^{-1}}{1+1.7916z^{-1}+0.8187z^{-2}}$$ $\boxed{T = 1\,sec}$ 4

6M

b) $0.6 \le |H(e^{j\omega})| \le 1.0$, $0 \le \omega \le 0.35\pi$

$|H(e^{j\omega})| \le 0.1$; $0.7\pi \le \omega \le \pi$

specifications

pass band edge digital frequency $\omega_p = 0.35\pi$ rad/sample

Subject Title: Digital Signal processing Subject Code: 15EE63

Question Number	Solution	Marks Allocated
	Stopband edge digital frequency $\omega_s = 0.7\pi$ rad/sample Gain in Normal value at passband $A_p = 0.6$ Gain in Normal value at stop band $A_s = 0.1$ Specification of analog IIR LPF: $A_p = 0.6, \; A_s = 0.1$ (Gain is same in analog & digital) For Bilinear Transformation pass band edge analog frequency $\Omega_p = \frac{2}{T} \tan \frac{\omega_p}{2}$ $= \frac{2}{0.1} \tan \frac{0.35\pi}{2} = 12.256$ rad/sec Stopband edge analog frequency $\Omega_s = \frac{2}{T} \tan \frac{\omega_s}{2}$ $= \frac{2}{0.1} \tan \frac{0.7\pi}{2}$ $= 39.2522$ rad/sec	2
	Order (N): $N = \frac{1}{2} \frac{\log\left[\frac{(1/A_s^2)-1}{(1/A_p^2)-1}\right]}{\log \frac{\Omega_s}{\Omega_p}} = 1.7267$ $\boxed{N = 2}$	3
	Normalized Transfer function $H(S_n)$. $H(S_n) = \prod_{K=1}^{N/2} \frac{1}{S_n^2 + b_k S_n + 1}$, where $b_k = 2\sin\left[\frac{(2k-1)\pi}{2N}\right]$ $H(S_n) = \frac{1}{S_n^2 + 1.414 S_n + 1}$ unNormalized Transfer Function $H(S)$ $H(S) = H(S_n)\big\|_{S_n = \frac{S}{\Omega_c}}$ $H(S) = \frac{154.8506}{S^2 + 17.5982 S + 154.8506}$	3

Subject Title: Digital signal processing Subject Code: 15EE63

Question Number	Solution	Marks Allocated
	Digital Filter $H(z)$ $H(z) = H(s)\Big\|_{s=\frac{2}{T}\frac{1-z^{-1}}{1+z^{-1}}}$ $H(z) = \dfrac{0.1708 + 0.3415z^{-1} + 0.1708z^{-2}}{1 - 0.5407z^{-1} + 0.2237z^{-2}}$	2 10M
6a)	**Digital Filter** | **Analog Filter** 1. operates on digital samples of the signal | 1. operates on analog signal 2. It is governed by linear difference equation | 2. It is defined by linear differential equation 3. It consist of adders, multipliers and delays | 3. It consist of electrical components resistors, capacitors & inductor. 4. In digital filter the filter coefficients are designed to satisfy the desired frequency response | 4. In analog filters the approximation problem is solved to satisfy the desired frequency response	4M.
b)	$N = 2$ $S_n = e^{j\frac{(2k-1)\pi}{4}}$; for $k = 1, 2, 3, 4$ when $k=1$, $S_n = e^{\frac{j(2-1)\pi}{4}} = 0.707 + j0.707 = P_1$ $k=2$, $-0.707 + j0.707 = P_2$ $k=3$, $-0.707 - j0.707 = P_2^*$ $k=4$, $0.707 - j0.707 = P_1^*$ Normalised function $H(S_n) = \dfrac{1}{(S_n - P_2)(S_n - P_2^*)}$ $H(S_n) = \dfrac{1}{S_n^2 + 1.414S_n + 1}$	2 4 2 8M

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Question Number	Solution		Marks Allocated
c)	IIR	FIR	
	1) All the infinite samples of impulse response are considered	1. only N samples of impulse response are considered	4M
	2) The impulse response cannot be directly converted to digital filter transfer function	2. The impulse response can be directly converted to digital filter transfer function	
	3. Linear phase characteristics cannot be achieved	3. Linear phase filters can be easily designed	

MODULE-4

7a) sol: specification of digital IIR filter:

$\omega_p = 0.25\pi$ rad/sample

$\omega_s = 0.5\pi$ rad/sample

Gain $A_p = 0.9$

Gain $A_s = 0.24$.

specification of analog IIR filter:

Gain $A_p = 0.9$

Gain $A_s = 0.24$

(Gain is same in analog & digital filter).

$\Omega_p = \frac{\omega_p}{T} = \frac{0.25\pi}{1} = 0.7854$ rad/sec — 1

$\Omega_s = \frac{\omega_s}{T} = \frac{0.5\pi}{1} = 1.5708$ rad/sec

order:

$$N = \frac{\cosh^{-1}\left[\frac{(1/A_s^2) - 1}{(1/A_p^2) - 1}\right]^{1/2}}{\cosh^{-1}(\Omega_s/\Omega_p)} = 2.1077 \quad \boxed{N = 3}$$ — 2

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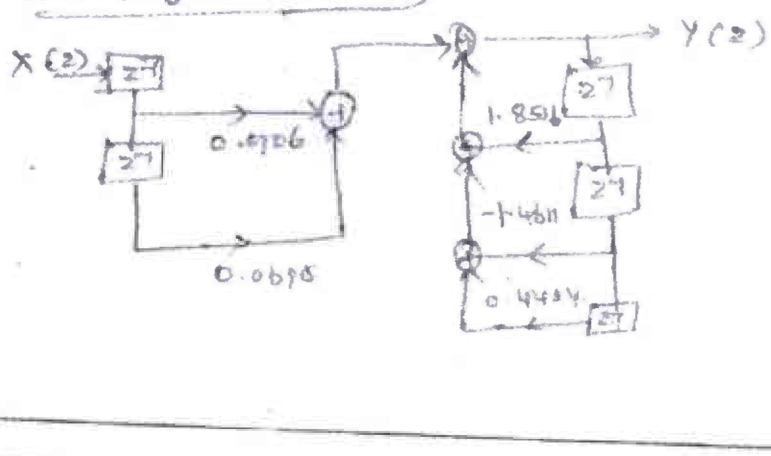
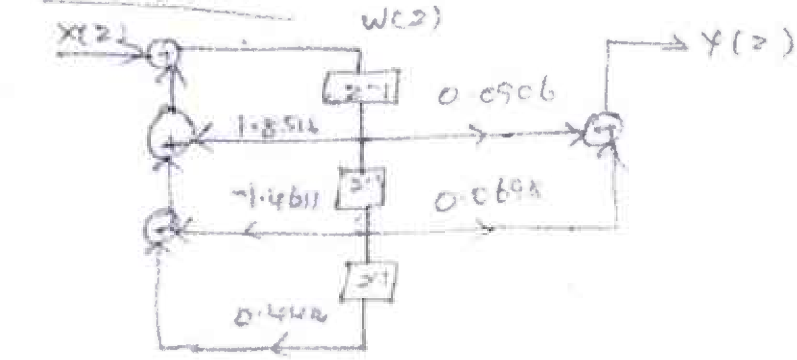
Question Number	Solution	Marks Allocated
	Normalised Transfer Function $H(S_n)$:- $H(S_n) = \frac{B_0}{S_n + C_0} \prod_{K=1}^{\frac{N-1}{2}} \frac{B_K}{S_n^2 + b_K S_n + C_K}$ $H(S_n) = \frac{0.5162}{S_n^3 + 1.0214 S_n^2 + 1.27165 S_n + 0.5162}$	2
	unnormalised transfer function $H(s)$:- $H(s) = H(S_n)\big\rvert_{S_n = \frac{s}{\Omega_c}}$ $\Omega_c = \Omega_p = 0.7854$ rad/sec $H(s) = \frac{0.2501}{(S+0.4011)(S^2 + 0.4011S + 0.6285)}$	2
	$H(s) = \frac{0.4011}{(S+0.4011)} - 0.4011 \frac{S+0.2006}{(S+0.2006)^2 + (0.7637)^2}$ $+ 0.1054 \frac{0.7637}{(S+0.2006)^2 + 0.7637^2}$	
	Digital IIR LPF filter $H(z)$ using impulse invariant transformation. $H(z) = \frac{0.0906 z^2 + 0.0698 z}{z^3 - 1.8516 z^2 + 1.4611 z - 0.4484}$	2
	Direct form-I structure 	$1\frac{1}{2}$

Subject Title: Digital signal processing

Subject Code: 15EE63

Question Number	Solution	Marks Allocated
	Direct form-II	1½
		12 M.



(b) Relation b/w analog & Digital frequency in Bilinear Transformation.

For Bilinear transformation

$$S = \frac{2}{T}\frac{1-z^{-1}}{1+z^{-1}} \quad , \text{ put } s = j\Omega,\ z = e^{j\omega}$$

$$j\Omega = \frac{2}{T}\frac{1-e^{-j\omega}}{1+e^{-j\omega}}$$

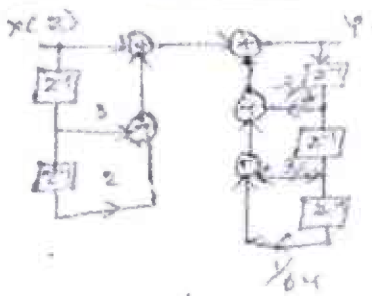
$$\Omega = \frac{2}{T}\tan\frac{\omega}{2}$$ — 2

Analog frequency, $\Omega = \frac{2}{T}\tan\frac{\omega}{2}$ — 1

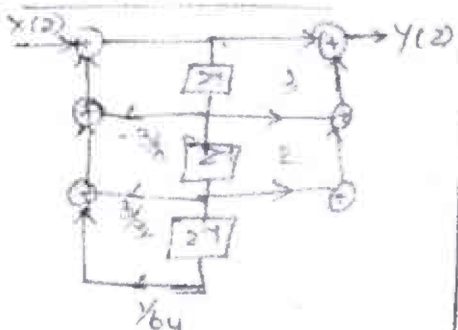
Digital frequency $\omega = 2\tan^{-1}\frac{\Omega T}{2}$ — 1

4M

8a) Direct form-I — 4+4



Direct form-II



8M

$$H_1(z) = \frac{1}{1-2z^{-1}+0.25z^{-2}}$$

$$H_2(z) = \frac{1+0.25z^{-1}}{1-3z^{-1}+0.2z^{-2}}$$

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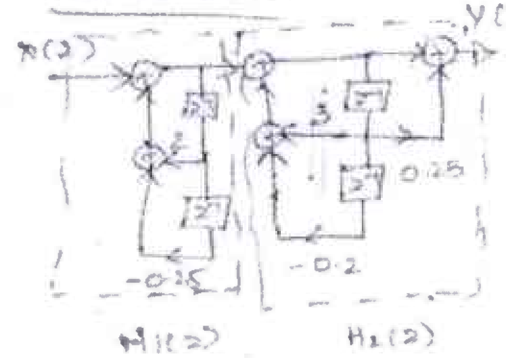
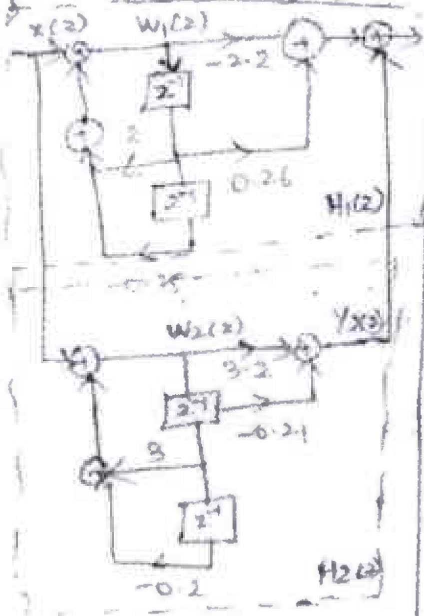
Question Number	Solution	Marks Allocated
b)	Cascade form  Parallel form 	4+4
	$H_1(z) = \dfrac{-2.2 + 0.26z^{-1}}{1 - 2z^{-1} + 0.25z^{-2}}$ $H_2(z) = \dfrac{3.2 - 0.21z^{-1}}{1 - 3z^{-1} + 0.2z^{-2}}$	8M
9a)	$H(\omega) = j\omega, \; -\pi \le \omega \le \pi$ $h_d(n) = \dfrac{1}{2\pi}\displaystyle\int_{-\pi}^{\pi} j\omega e^{j\omega n}\, d\omega$ $= \dfrac{\cos \pi n}{n}$	4
	$n = 0, 1, \ldots N-1$ $h(n) = \{ \frac{1}{3}, -\frac{1}{2}, 1, 0, -1, \frac{1}{2}, -\frac{1}{3} \}$	4 8M
b)	Sol: $\Omega_c = 30\pi$ rad/sec , $\Omega_s = 45\pi$ rad/sec . $f_s = 100$ sam/sec . $\omega_1 = \dfrac{\Omega_c}{f_s} = \dfrac{30\pi}{100} = 0.3\pi$ rad/sec $\omega_2 = \dfrac{\Omega_s}{f_s} = \dfrac{45\pi}{100} = 0.45\pi$ rad/sec	2
	To find N: $K\left(\dfrac{2\pi}{N}\right) = \dfrac{8\pi}{N}$ $\boxed{K = 4}$	2

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Question Number	Solution	Marks Allocated
	$N = K\left(\frac{2\pi}{\omega_2-\omega_1}\right) = 4\left(\frac{2\pi}{0.45\pi - 0.3\pi}\right)$	
	$\boxed{N = 53}$	2
	$h_d(n) = \begin{cases} \frac{1}{\pi(n-\alpha)} \sin \omega_c(n-\alpha) \times 0.54 - 0.46 \cos\left(\frac{2\pi n}{N-1}\right) \\ \frac{\omega_c}{\pi} \times 0.54 - 0.46 \cos\left(\frac{2\pi n}{N-1}\right) \end{cases}$	2
		8M
10a)	Sol: $W(\omega) = \sum_{n=-\left(\frac{N-1}{2}\right)}^{\left(\frac{N-1}{2}\right)} W_R(n)\, e^{-j\omega n}$	
	$= \sum_{n=\left(\frac{N-1}{2}\right)}^{\left(\frac{N-1}{2}\right)} 1 \times e^{-j\omega n} = \sum_{n=-\left(\frac{N-1}{2}\right)}^{\left(\frac{N-1}{2}\right)} (e^{-j\omega})^n$	3
	$\boxed{W(\omega) = \frac{\sin \omega N/2}{\sin \omega/2}}$	3
		6M
b)	Sol: Given $\boxed{N = 7}$, $\alpha = 3$	
	$\omega = \frac{2\pi K}{N} \Rightarrow \alpha = \frac{N-1}{2} \Rightarrow \boxed{N = 7}$	1
	$H(k) = \begin{cases} e^{\frac{-j3 2\pi K}{7}}, & 0 < \frac{2\pi K}{7} < \pi/2 \\ 0, & \text{otherwise} \end{cases}$	1
	$H(k) = \begin{cases} e^{\frac{-6\pi K}{7}}, & 0 < K < 2 \\ 0, & \text{otherwise} \end{cases}$	2

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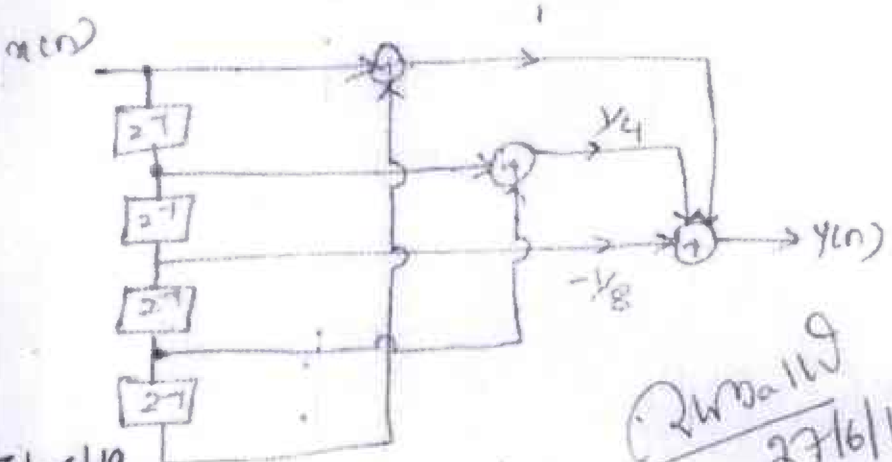
Question Number	Solution	Marks Allocated
	$K=0 \Rightarrow H(0)=1$ $K=1 \Rightarrow H(1) = e^{-j6\pi/7}$ $h(n) = \frac{1}{7}\left[H(0) + 2\sum_{K=1}^{3} Re\, H(k)\, e^{\frac{j2\pi nK}{7}}\right]$ $h(n) = \frac{1}{7}\left[1 + 2\cos\frac{2\pi}{7}(n-3)\right]$.	2 6M
c)	$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4)$ $H(z) = \frac{Y(z)}{X(z)} = 1 + \frac{1}{4}z^{-1} - \frac{1}{8}z^{-2} + \frac{1}{4}z^{-3} + z^{-4}$ Take I.Z.T, we get. $y(n) = x(n) + \frac{1}{4}x(n-1) - \frac{1}{8}x(n-2) + \frac{1}{4}x(n-3) + x(n-4)$. $= (x(n) + x(n-4)) + \frac{1}{4}[x(n-1) + x(n-3)] - \frac{1}{8}x(n-2)$ 	1 1 2 4M

Dr. S. B. Shivakumar 27/06/19 Chairman. B.O.E (E&E)

27/6/19