USN

DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

UG Semester End Examination, December 2017

Course: Digital Signal Processing **Code: EC53**

Maximum marks: 100 Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	Derive the relationship between DFT and Z Transform.	04
	b	State and prove Circular convolution in time domain property of DFT.	08
	c	Find IDFT of 4 point sequence X(K) = { 4, -j2, 0, j2 } using Matrix method.	08
		(OR)	
2	a	For the sequences $x_1(n)= \cos\frac{2\pi}{4}n$ and $x_2(n) = \sin\frac{2\pi}{4}n$, $0\le n \le 3$, Find Circular convolution using DFT and IDFT.	08
	b	Let X(k) be a 14 point DFT of length 14 real sequence x(n). The first 8 samples of X(k) are : X(0)=12, X(1)=-1+3j, X(2)=3+4j, X(3)=1-5j, X(5)=6+3j, X(6)= -2-3j, X(7)=10. Find the remaining samples of X(k). Also evaluate the following : **(i)** X(0) **(ii)** X(7) **(iii)** $\sum_{n=0}^{13} x(n)$ **(iv)** $\sum_{n=0}^{13} \lvert x(n)\rvert$	12
3	a	Derive DIF FFT algorithm for N=8. Draw the complete signal flow graph.	12
	b	In direct computation of N point DFT of x(n), find **(i)** Complex Multiplications **(ii)** Complex Additions **(iii)** Real Multiplications **(iv)** Real Additions are required. Write the necessary steps required	08
		(OR)	
4	a	Find 4 point circular convolution of x(n)= {1,2,3,4} and h(n)= {1,1,1,1} using Radix 2 DIF FFT Algorithm.	10
	b	Find 8 point DFT of x(n)={1,1,1,1,1,1,1,1} using Radix 2 DIT FFT algorithm.	10
5	a	Design a Chebyshev Analog low pass filter that has a **-3dB** cutoff frequency of **100 rad/sec** and a stopband attenuation **25dB** of greater for all radian frequencies past **250 rad/sec.**	10
	b	Let **H(s)** = $\frac{1}{s^2+s+1}$ represent the transfer function of LPF with a passband of 1 rad/sec. Use frequency transformation (Analog to Analog) to find the transfer function of a band pass filter with passband **10 rad/sec** and a centre frequency of **100 rad/sec.**	10
		(OR)	

6 a For Analog Butterworth filter, derive an expression for order, cut off frequency for design of low pass filter. 08

b Design a digital filter H(z) that when used in an **A/D - H(z) – D/A** structures given an **equivalent analog filter** with the following **specifications** : 12

Passband ripple : ≤ 3.01 dB, Passband edge: 500Hz

Stopband attenuation: ≥15dB, Stopband edge: 750 Hz

Sample Rate: 2 KHz. Use Bilinear transformation to design the filter on an analog system equation. Use Butterworth filter prototype. Also, obtain the difference equation.

7 a A LPF is to be designed with the following desired frequency response: 10

$$H_d(e^{j\omega}) = H_d(\omega) = e^{-j3w} \quad ;0\le |\omega| \le \pi/4$$

$$0 \quad ; \pi/4 \le \omega \le \pi$$

Determine the filter coefficients h(n) , given rectangular window w(n) defined

$$w(n) = \{1 \ ; 0\le n \le 4$$

$$\{0 \ ; \text{otherwise}$$

b Design a linear phase FIR bandpass filter to pass frequencies in the range of 0.4π to 0.65π rad/secby using 7 samples of Hanning window sequence 10

(OR)

8 a Design a linear phase highpass filter using the Hamming window for the following desired frequency response. 10

$$Hd(\omega)=\begin{cases} e^{-j3\omega} & \frac{\pi}{6} \le |\omega| \le \pi \\ 0 & |\omega| < \frac{\pi}{6} \end{cases}$$

$\omega(n)=0.54-0.46\cos\left(\frac{2\pi n}{N-1}\right)$, where N is the length of the Hamming window.

b Design a linear phase **lowpass FIR filter with 7taps** and a cut off frequency of **ωc= 0.3π** using the **frequency sampling** method. 10

9 a A FIR filter is given by y(n)=x(n)+2/5x(n-1)+3/4x(n-2)+1/3x(n-3). Draw DFI and lattice structure. 10

b Obtain the direct form I , parallel form structures of the following systems. 10

i) y(n) = 3/4 y(n-1) – 1/8y(n-2) + x(n) + 1/3 x(n - 1)

ii) y(n) = -0.1 y(n-1) + 0.2y(n-2) + 3x(n) +3.6 x(n – 1) + 0.6x(n – 2)

(OR)

10 a Write briefly on the following problems dealing with finite word length effects: 10

i)Parameter quantisation in digital filters **ii)**Round off noise in multiplication **iii)** Overflow in addition **iv)** Limit Cycles

b Obtain direct form I , direct form II and cascade realization for 10

y(n)=0.75y(n-1)-0.125y(n-2)+6x(n)+7x(n-1)+x(n-2)

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

UG Make-up Examination, January/ February 2018

Course: **Digital Signal Processing** Code: **EC53**

Maximum marks: 100 Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	Let X(k) be a 14-point DFT of length - 14 real sequence x(n). The first K-samples of X(k) are given by **X(0)=12, X(1)=-1+3j, X(2)=3+4j, X(3)=1-5j, X(4)=1+j, X(5)=6+3j, X(6)=-2-3j), X(7)= 10.** Find the remaining samples of X(k).Also evaluate the following **i) X(0) ii) X(7) iii)**$\sum_{n=0}^{13} x(n)$ **iv)** $\sum_{n=0}^{13} \lvert x(n)\rvert^{2}$	10
	b	Compute DFT of the sequence **x(n)=cos(** $\pi n/4$**)** for N=6 . Plot magnitude and phase of X(k)	10
		(OR)	
2	a	Determine the 8-point DFT of the sequence **x(n)={8, 8, 8, 8,1, 1, 0, 0},** Sketch the magnitude and phase spectra.	10
	b	A 4-point sequence **x(n)={1,2,3,4}** has DFT X(k) for $0 \le K \le 3$. Find the signal values y(n) which has DFT **Y(K)=X((K-1))**$_N$ without performing DFT and IDFT.	04
	c	Compute 4 point circular convolution for the sequence **x(n)={1,8,1,8}** and **h(n)={2,9,2,9}**using DFT and IDFT method.	06
3	a	Find the IDFT **X(k)={36, -4+i9.7, -4+i4, -4+i1.7, -4, -4-i1.7, -4-i4, -4-j9.7}** using DIF-FFT algorithm show clearly all intermediate steps.	10
	b	Determine the circular convolution of the sequence **x(n)={1,2,3,1} and h(n)={4,3,2,2}** using DIT FFT Method.	10
		(OR)	
4	a	(a) Find the IDFT of X(k) = {4, 1 -j, -2, 1+j}using DIT- FFT algorithm.	08
	b	Determine the circular convolution of the sequence **x(n)={1,2,3,1} and h(n)={0,3,1,2}** using DIT FFT.	10
	c	What is the need of FFT?	02
5	a	Design a digital Butterworth filter H(z) given an equivalent analog filter with the following specifications : **passband ripple ≤ 3db , stopband edge frequency of 750Hz , stopband attenuation of 15db, passband edge frequency = 500Hz and sampling rate is 2KHz.Design using bilinear transformation.**	10
	b	Design a Chebyshev analog filter with ripple of **0.5db in band |Ω|≤ 1 and at Ω = 3, amplitude if down by 3db.**	10
		(OR)	
6	a	Let **H(s)=**$\frac{1}{s^2+s+1}$ represents transfer function of a low pass filter with a passband of 1 rad/sec. Use frequency transformation to find the transfer function of the following analog filters. **i)A LPF with Ω`p=10 rad/sec. ii)** A HPF with **Ω`p=100 rad/sec.**	04

b Design an analog bandpass filter to meet the following frequency domain specifications: **(i) -3.0103dB upper and lower cut-off frequency of 50Hz and 20kHz (ii)A stopband attenuation of at least 20dB at 20hz and 45kHz (iii)Monotonic frequency response** 12

c Distinguish between IIR and FIR filters. 04

7 a A filter is designed with the following desired frequency specifications using. use rectangular window. 10

$$Hd(w) = \begin{cases} 0, & -\frac{\pi}{4} \le w \le \frac{\pi}{4} \\ e^{-j2w}, & \frac{\pi}{4} \le |w| \le \pi \end{cases}$$ Find the frequency response of the FIR filter

b For the desired frequency response. Find $H(\omega)$ for N = 7 using hamming window. 10

$H_d(e^{j\omega}) = \{e^{-j3w}$; $-3\pi/4 \le \omega \le 3\pi/4$

$\{0$; $3\pi/4 \le |\omega| \le \pi$.

(OR)

8 a Determine the filter coefficients h(n) obtained by sampling $H_d(\omega)$ given by 12

$H_d(e^{j\omega}) = \{e^{-j3w}$; $0 \le |\omega| \le \pi/2$

$\{0$; $\pi/2 \le \omega \le \pi$. Also obtain the frequency response $H(\omega)$. N = 7.

b Consider the pole zero plot, as shown in below fig. 8b 04

(i) Does it represent an FIR filter? **(ii)**Is it a linear phase system?

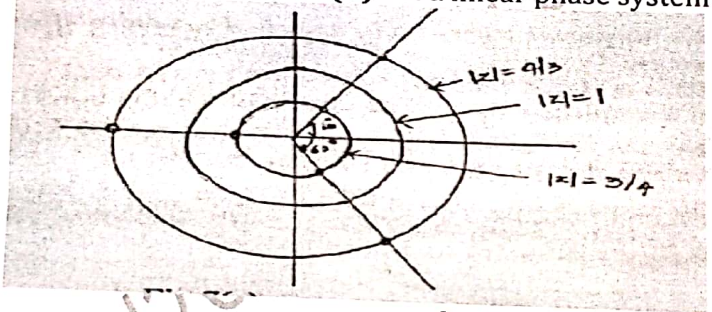

fig. 8b

c Mention few advantages and disadvantages of windowing technique 04

9 a Consider the LTI system, initially at rest , described by the difference equation 14

y(n) = ¼ y(n-2) + x(n)

(i) Determine the impulse response of the system

(ii) Determine the system transfer function H(Z)

(iii) Determine the direct form II

(iv) Cascade form realization for this system.

b The input to the system **y(n) = 0.999 y(n-1) + x(n)** is quantized **to b = 8 bits.** What is the power produced by the quantization noise at the output of the filter? 06

(OR)

10 a Write briefly on the following problems dealing with finite word length effects: **(i)**Parameter quantization in digital filters **(ii)**Round off noise in multiplication **(iii)**Overflow in addition **(iv)**Limit Cycles 10

b Consider a FIR filter with system function

$H(z) = 1 + 2.88z^{-1} + 3.4048z^{-2} + 1.74z^{-3} + 0.4z^{-4}$. Sketch the **(i)** direct form and **(ii)** lattice realizations of the filter. 10

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UG Semester End Examination, December 2017

Course: **Digital signal processing** Code: **TE53**

Maximum marks: 100 Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	For the following sequence, find: **i)** N point DFT of $x(n) = \cos\frac{2\pi}{N}K_0 n$ **ii)** 5 point DFT of $x(n) = \{1,1,1,\}$	10
	b	State and prove: **i)** Linearity property of DFT **ii)** DFT of real and even sequence	10
		(OR)	
2	a	Find six point of DFT of sequence x(n)={1,1,2,2,3,3} and hence draw magnitude and phase spectra.	10
	b	Prove that the sampling of Fourier transform of a sequence x(n) results in N point DFT using which the sequence can be reconstructed.	10
3	a	Determine circular convolution of x(n) = { 1 2 3 5} and { 4 5 2 3} using Time domain and frequency domain approach.	10
	b	Find the output y(n) of a filter whose impulse response is h(n)={1, 2} and the input signal to the filter is x(n)={1,2,-1,2,3,-2,-3,-1,1,1,2,-1} using overlap-save method.	10
		(OR)	
4	a	Let x(n)={1, 2, 0, 3, -2, 4, 7, 5} with a 8-point DFT x(k). Evaluate the following without explicitly computing x(k): **i)** x(0) **ii)** x(4) **iii)** $\sum_{k=0}^{7} x(k)$ **iv)** $\sum_{k=0}^{7} \lvert x(k)\rvert^2$	10
	b	State and prove Circular convolution property of DFT.	10
5	a	Derive the radix 2 decimation in time FFT algorithm to compute the DFT of a N=8 point sequence and draw the final complete signal flow graph.	10
	b	Given x(n)={1,2,3,4,4,3,2,1}, find X(K) using DIT FFT algorithm.	10
		(OR)	
6	a	Derive the radix-2 DIF FFT algorithm to compute DFT of an N =8 point sequence and draw the complete signal flow graph.	10
	b	Find the 8 point DFT of {2, 1, 2, 1} using DIF-FFT. Draw the signal flow graph for N=8 with intermediate values, stuff appropriate zeros.	10

7 a Derive the expression for N order and cut-off frequency Ω_c for a low pass Butterworth filter starting from the frequency domain specification of a low-pass filter. 08

b Obtain a cascade and parallel realization for the system described by 08

$$H(z) = \frac{(1+z^{-1})(1+2z^{-1})}{(1+\frac{1}{2}z^{-1})(1-\frac{1}{4}z^{-1})(1+\frac{1}{8}z^{-1})}$$

c Transform the third order Butterworth normalized low pass filter to high pass filter with pass band edge at 2 rad/sec.(Transfer function can be directly written) 04

(OR)

8 a Design a Chebyshev type1 analog filter to meet the following specifications. Passband attenuation of 2dB at 4 r/s, stopband attenuation of 10dB at 7 r/s. 10

b Design a digital low pass filter with the following specifications 10

i)Monotonic in the pass band and stop band

ii)-3.01dB cutoff frequency of 0.5π radians.

iii) Stop band attenuation of 156dB at 0.75π radians

Use bilinear transformation. Assume T = 1 sec.

9 a A low pass filter is to be designed with the following desired frequency response: 10

i. $H_d(e^{jw}) = H_d(w) = e^{-jw}$ ....$|w| < \frac{\pi}{4}$

ii. 0........... $\frac{\pi}{4} < |w| < \pi$

b Derive the frequency response of a symmetric FIR low pass filter for both N even and N odd. 10

(OR)

10 a Consider a FIR filter with system function: Sketch the direct form and lattice realizations of the filter. $H(z) = 1 + 2.82z^{-1} + 3.4048z^{-2} + 1.74z^{-3}$ 10

b Realize the linear phase FIR filter having the following impulse response 05

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

c Compare IIR and FIR filters. 05

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UG Semester End Examination, December 2017

Course: **Digital Signal Processing** Code: **ML53**

Maximum marks: 100 Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	State and prove the following properties of DFT **i)** Circular Convolution **ii)** Periodicity	08
	b	Compute DFT of a sequence $(-1)^n$ for N=4	05
	c	Using Linear transformation method, obtain the DFT of the four point (N=4) sequence $x(n) = 2^n$	07
		(OR)	
2	a	Compute Circular Convolution between following sequences using DFT and IDFT method x(n) = {1, 2, 3 , 4} and y(n) = {-1,-2,-3,-4} x(n) and y(n) are periodic sequences with N=4	10
	b	Let X(k) denote the N-point DFT of the N-point sequence x(n) **i)** Show that with N even, and if x(n) = x(N-1-n), then X(N/2) = 0 **ii)** Show that if x(n) = -x(N-1-n) then X(0) = 0	10
3	a	If x(n) = {3,1,5,4,2,1,0,1}, find X(K) by using decimation in time FFT algorithm.	12
	b	Define different properties of twiddle factor.	08
		(OR)	
4	a	If x(n) = {1,2,2,3,1,1,4,2} find X(K) by using decimation in frequency FFT algorithm.	12
	b	Derive the Radix 2 DIT FFT algorithm for N=8	08
5	a	Illustrate how to determine the order and Cut off frequency of a filter with monotonous pass band and stop band.	10
	b	Design a butter worth analog filter for the following specifications 0.8<= IH(ejw)I <= 1 for 0 <= w <= 0.2π IH(ejw)I <= 0.2 for 0.6 π >= w <= π	10
		(OR)	
6	a	Derive the relation between an analog and Digital Filter using impulse invariance transformation method.	08

	b	Design a digital Butterworth filter that satisfies the following constraints using bilinear transformation. Assume T = 1 Sec. $0.9 <= \|H(e^{jw})\| <= 1 \qquad 0<=w<= \pi/2$ $\|H(e^{jw})\| <= 0.2 \qquad 3\pi/2<= w<= \pi$	12
7	a	Name any four types of windows used in the design of FIR filters. Write the analytical equations and magnitude response of each window.	10
	b	Realize the given system function using **i)** Direct Form – I **ii)** Direct Form –II $H(Z) = 0.3(1-0.25Z^{-2}) / 1+0.1Z^{-1} - 0.72Z^{-2}$	10

(OR)

8	a	Distinguish between IIR and FIR filters.	08
	b	A low pass filter is to be designed with the following desired frequency response $H_d(w) = e^{-2/w} \; ; \|w\| < \pi/4$ $\; ; \pi/4 < \|w\| < \pi$ Determine the filter coefficients hd(n) and h(n) if $w_R(n)$ is a rectangular window defined as below. Also, find the resulting FIR filter. $w_R(n) = 1 \; ; 0 < n < 4$ $= 0 \; ;$ otherwise	12
9	a	Discuss the implementation of multirate DSP in interfacing digital systems with different sampling rates.	10
	b	Explain the implementation of digital filter banks.	10

(OR)

10	a	Construct a two channel quadrature mirror filter bank with appropriate equations and block diagram.	10
	b	Discuss Decimation by a factor D and Interpolation by a factor I along with necessary equations and Block diagram.	10

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

UG Make-up Examination, January/ February 2018

Course: Digital Signal processing | **Code: ML53**

Maximum marks: 100 | Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	State and Prove the Complex Conjugate property of Discrete Fourier Transforms.	04
	b	Find the IDFT of 4-point sequence, x(k) = (4, -j2, 0, j2)	06
	c	Consider a sequence : $x(n) = \delta(n) + 2\delta(n-2) + \delta(n-3);\ 0 \le n \le 3$ **i)** Find the 4 pt. DFT of the sequence $x(n)$ **ii)** If $y(n)$ is the 4 pt. circular convolution of $x(n)$ with $x(n)$ itself, then find $y(n)$ & 4 pt. DFT $Y(K)$	10
		(OR)	
2	a	State and Prove the Linearity property of Discrete Fourier Transforms.	04
	b	Compute the 4-point DFT of the sequence $x(n) = (1,0,1,0)$. Also, find $y(n)$ If $Y(k) = X((k-2))_4$.	06
	c	Consider the sequences $x_1(n)$ and $x_2(n)$ given below: $x_1(n) = \{1,2,3,4,5\}$, $x_2(n) = \{1,3,6,7,2\}$ **i)** Find the 5-point circular convolution of the sequences $x_1(n)$ and $x_2(n)$ **ii)** Obtain the linear convolution of the sequences $x_1(n)$ and $x_2(n)$ and hence obtain the result of 5-point circular convolution.	10
3	a	Give the relationship between DFT and other transforms.	04
	b	Compute the 4-pt circular convolution of the 2 sequences given below using DIT radix 2 FFT algorithm. $x_1(n) = \{1,0,1,0\}$, $x_2(n) = \{1,1,1,1\}$	08
	c	Develop the FFT algorithm by using DIF approach for computing N-point DFT of a length –N sequence x(n). Illustrate for N=8.	08
		(OR)	
4	a	Give the need for efficient computation of DFT (FFT algorithm).	04
	b	Compute 8-pt DFT of the sequence $x(n) = 2^n$; ; $0 \le n \le 7$ using decimation in frequency, radix2, FFT algorithm.	08
	c	Develop the FFT algorithm by using DIT approach for computing N-point DFT of a length –N sequence x(n). Illustrate for N=8.	08
5	a	Explain mapping S-plane to Z-plane in Bilinear transformation.	05
	b	Derive the transfer function of a normalized LP Butterworth filter for N=2, 3	07
	c	Design an analog lowpass (maximally flat) filter that will have a -2 dB cutoff frequency at 75 Hz and have greater than 20 dB of attenuation for all frequencies greater than 150 Hz.	08

(OR)

6	a	What should be the Buttterworth LPF order so as to have minimum rulloff response of 70dB.	05
	b	Derive an expression for Impulse Invariant Technique(IIT) for the design of Digital Filters.	07
	c	Design a lowpassChebyshev filter to meet the following frequency-domain specifications: **i)** Acceptable passband ripple: 2 dB **ii)** Passband edge frequency: 40 rad/sec **iii)** Stopband attenuation: ≥ 20 dB **iv)** Stopband edge frequency: 52 rad/sec	08
7	a	Explain the design procedure for the linear-phase FIR filters using frequency sampling method.	06
	b	Realize the following FIR system with minimum number of multipliers $H(Z)=1/2+1/4Z^{-1}+1/8Z^{-2}+1/5\,Z^{-3}+1/8\,Z^{-4}+1/4\,Z^{-5}+1/2\,Z^{-6}$	06
	c	A FIR filter is to be designed with the following frequency response: $H_d(w) = \begin{cases} e^{-jw\alpha} & ; \quad w_1 < \lvert w \rvert < w_2 < \pi \\ 0 & ; \text{ otherwise} \end{cases}$ Take $\alpha = 3, w_1 = 1$ rad $\& w_2 = 2$ rad. The FIR filter is to be designed using Hamming window. Find the co-efficients of FIRF and also its frequency response	08

(OR)

8	a	Give the advantages and disadvantages of window technique. Explain the FIR filter using frequency sampling technique.	06
	b	Obtain the parallel form of structure, for the following Digital filter $H(z)= (1+1/2\,Z^{-1}) \;/\; ((1+1/3\,Z^{-1})\,(1-1/4\;Z^{-1}))$	06
	c	Design an FIR digital filter to approximate an ideal LPF with a gain of unity, cut off frequency 850 Hz and working at a sampling frequency = 5000 Hz. The length of the impulse response should be 5. Use rectangular window.	08
9	a	Construct a 2 channel quadrature mirror filter bank with appropriate equations and block diagram.	10
	b	Explain the implementation of digital filter banks.	10

(OR)

10	a	Demonstrate an application of multirate DSP in interfacing digital systems with different sampling rates.	10
	b	How would you increase or decrease the sampling rates? Explain in details.	10

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)
Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Make-Up Examination, February 2019

Course: **Digital Signal Processing**
Course Code: **ML53**
Semester: **V**

Maximum marks: **100**
Duration: **3 hours**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			**Marks**
1	a)	The number of real additions for direct evaluation of 8-point DFT is ---------.	01
	b)	The computation efficiency of 64 point FFT over 64 point DFT is ------------.	01
	c)	DFT of $X((-n))_N$ =	01
	d)	If $x(n) = \delta(n) + \delta(n-5)$, find X(K).	01
	e)	If x(n) = { 1,3,2,5,,-2,7} find $x((-n)_6$	01
	f)	The IDFT of i) X*(N-K) is ---------------. ii) $X(K-2)\ )_N$ is ----------------.	01
	g)	Given x(n) = { 1 2 0 1} X(N-2) is	01
	h)	$N\log_2 N$ is the number of complex multiplications needed to find N values of DFT in DIT FFT; State true or false.	01
	i)	Find the values of W_N^K When N=8, K=2.	01
	j)	In IIR systems, the ------------ structure relates the time domain and z-domain.	01
	k)	The condition for symmetry of impulse response of FIR system is -----------.	01
	l)	Write the relation between s plane and z plane in bilinear transformation.	01
	m)	Given Y(K) = { -6,1+j,0,1-j} using radix-2 DIT FFT algorithm, the four point IDFT of Y(K) is --------------.	01
	n)	Name the techniques for designing FIR filters.	01
	o)	Given x(n) = {1,2,1} using DIT –FFT algorithm X(K) is -------------.	01
	p)	The periodicity property of twiddle factor is ------------------.	01
	q)	The basic butterfly of decimation in time FFT is --------------.	01
	r)	For the FIR system h(n) = {-0.5, 0.8,-.5}, the realization with minimum number of multipliers is -------------.	01
	s)	List the applications of DSP.	01
	t)	What is the effect of undersampling a signal?	01
2	a)	State and prove the circular shift and circular symmetry of a sequence.	08
	b)	Let x(n) be the sequence where $x(n) = \delta(n) + 2\delta(n-2) + \delta(n-3)$, i) Find the four point DFT of x(n).	08

ii) If y(n) is the four point circular convolution of x(n) with itself, find y(n) and the four point DFT Y(K).

3 a) If x(n) = { 3,1,5,4,2,1,0,1} find X(K) by using decimation in Time FFT algorithm. 12

b) Compute X(K) by using decimation in frequency FFT algorithm, if x(n) = {2 1 3 1}. 04

4 a) Design a low pass Butterworth filter using bilinear transformation method for satisfying the following constraints: 11

Pass band wp: 0.162 rad

Stop band ws:1.63 rad

Pass band ripple: 3dB

Stop band attenuation: 30 dB

Sampling frequency: 8 KHZ.

b) Use impulse invariance method to design a digital IIR filter from an analog prototype that has a system function: 05

$H_a(S) = s+a / (s+a)+b^2$

5 a) Compare the difference between FIR and IIR Filters. 04

b) Realize the following system function using: 12

i) Direct form I

ii) Direct form II

iii) Cascade form

iv) Parallel form

$H(Z) = [0.3(1-0.25Z^{-2})] / [1+0.1Z^{-1} - 0.72Z^{-2}]$

OR

6 a) Design an ideal differentiator with frequency response 10

$H(e^{jw}) = jw \quad -\pi \leq w \leq \pi$

Using hamming window with N=7.Plot frequency response.

b) Name any four types of windows used in the design of FIR filters. Write the analytical equations and the magnitude response of each window. 06

7 a) Demonstrate an application of Multirate DSP in interfacing digital systems with different sampling rates. 08

b) How would you increase or decrease the sampling rates? Explain in detail. 08

OR

8 a) Construct a two channel quadrature mirror filter bank with appropriate equations and block diagram. 08

b) Explain the implementation of Digital Filter banks. 08

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

UG Semester End Examination, May 2018

Course: **Digital Signal Processing** Code: **EI63**

Maximum marks: 100 Duration: 3 hours

Answer five full Questions. All questions carry equal marks

			Marks
1	a	Find the 4 point DFT of the sequence $x(n) = 6 + \sin(\frac{2\pi n}{N})$	10
	b	Determine N point DFT of $x(n) = \cos\left(\frac{2\pi}{N}\right) k_0 n$	05
	c	Derive the N point DFT of a real and even sequence.	05
		(OR)	
2	a	The length of the impulse response is h(n)= {2,1,1,2} and input signal x(n)= {1,2,3,4}. Compute the Circular Convolution using circle method and matrix method.	10
	b	Find the output y(n) of a filter whose impulse response is h(n)={1,1,1} and input signal to the filter is x(n)={3,-1, 0, 1, 3, 2, 0, 1 , 2, 1} use overlap save method.	10
3	a	Develop DIT-FFT algorithm for the given sequence $x(n) = \sin\left(\frac{\pi}{2}\right) n$; $0 \leq n \leq 3$	10
	b	Determine the 4 point IDFT of X(K)= {2.5 , -0.25 + j 0.75, 0, -0.25 – j 0.75} using DIF FFT algorithm.	10
		(OR)	
4	a	Apply circular convolution method to determine y(n) using radix 2 DIF FFT algorithm for the given sequence x(n)={ 1,1,1,1} and h(n) = {1, 0, 1, 0}	10
	b	Explain Goertzel algorithm with relavent Equations.	10
5	a	Design a digital Low pass filter with a passband characteristic that is constant to within 0.75dB for frequencies below w=0.2613π and stop band attenuation of at least 20dB for frequencies between w=0.4018 π and π . Use bilinear transformation method.	10
	b	Transform the analog filter $H(s) = \frac{s+0.1}{(s+0.1)^2+9}$ into a digital filter using bilinear transformation. The digital filter should have resonant frequency wr = $\frac{\pi}{4}$	06
	c	Give any four comparison between IIR and FIR filter.	04
		(OR)	
6	a	Design a butterworth LPF to meet the following specifications using bilinear transformation. $0.8 \leq \lvert H(e^{jw}) \rvert \leq 1$; $0 \leq w \leq 0.2\pi$; $\lvert H(e^{jw}) \rvert \leq 0.1$; $0.5 \leq w \leq \pi$	10
	b	Discuss the properties of Chebyshev filter with characteristics.	06
	c	Give any four comparison between analog and digital filter.	04

7 a Design a linear phase FIR filter approximating the ideal frequency response 10

$Hd(w) = 1$ for $|w| \leq \pi/6$

$Hd(w) = 0$ for $\pi/6 < w \leq \pi$

b List out the different type of windows used in the design of FIR filters. Write the analytical equations and draw the magnitude response characteristics of any two windows. 10

(OR)

8 a Analyze the FIR filter design using frequency sampling technique. 10

b Discuss the frequency sampling method of FIR filter design. Applying the above principle, design a FIR filter with h(n) = { 1, 2, 1}. 10

9 a Analyze the Decimation process With the help of block diagram and input output relationships. 10

b The input sequence x(n)=[0 3 6 9 12] is interpolated using interpolation factor L=3 and its low pass filter coefficient bk = [1/3 2/3 1 2/3 1/3]. Determine the output of the linear interpolation process. 10

(OR)

10 a Analyze two types of digital filter banks and realize its structure. 06

b Discuss on the two channel Quadrature mirror filter bank with its characteristics. 10

c Explain Multirate signal Processing with examples. 04

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)
Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Semester End Examination, December 2018/January 2019

Course: **Digital Signal Processing**
Course Code: **ML53**
Semester: **V**

Maximum marks: **100**
Duration: **3 hours**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			Marks
1	a	Multiplication of 2 DFTs in frequency domain leads to circular convolution in time domain: Say true or false.	01
	b	State time reversal property of DFT.	01
	c	Write the number of complex multiplications to determine N values of DFT.	01
	d	Find the value of $(W_N^N)^N$ where W_N is the twiddle factor.	01
	e	Find the DFT of x(n)={1,2} using DIT FFT algorithm.	01
	f	Draw a 4 point DIF FFT algorithm.	01
	g	$W_8^7 = -W_8^3$, State true or false.	01
	h	$N\log_2 N$ is the number of complex multiplications needed to find N values of DFT in DIT FFT; State true or false.	01
	i	Where are the poles of the stable butterworth filter located?	01
	j	Distinguish between butterworth and chebyshev filters.	01
	k	Classify filters based on their impulse responses.	01
	l	Write the relation between s plane and z plane in bilinear transformation.	01
	m	What is linear phase?	01
	n	Name the techniques for designing FIR filters.	01
	o	What are the limitations of FIR filters?	01
	p	Name any 4 types of windows used in FIR filter design.	01
	q	State sampling theorem.	01
	r	Write the analytical equation of a hanning window.	01
	s	What is the effect of undersampling a signal?	01
	t	Name 4 applications of multirate DSP.	01
2	a	Define DFT and IDFT. Describe the importance of DFT in DSP.	08
	b	Determine the DFT of x[n] = {1, 2, 0, 2, 3}	08

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3	a	Define the different properties of twiddle factor	06
	b	Compute IDFT of the sequence X(k) = {7, -0.707-j0.707, -j , 0.707-j0.707, 1, 0.707+j0.707, j , -0.707+ j0.707} using DIT algorithm	10
4	a	Derive expressions for the order and cutoff frequency of an analog type I chebyshev filter.	08
	b	Derive the relation between an analog and digital filter using impulse invariance transformation method.	08
5	a	Distinguish between IIR and FIR filters.	08
	b	Represent the filter in parallel form. $y[n] = -0.1y[n-1] + 0.72y[n-2] + 0.7x[n] - 0.25x[n-2]$	08
		OR	
6	a	Design the direct form I and II form realizations of the following filter. $y[n] - \frac{1}{8}y[n-2] - \frac{3}{8}y[n-4] + x[n] + \frac{1}{5}x[n-2] = 0$	08
	b	Using a rectangular window technique design a low pass filter with passband gain of unity, cutoff frequency of 1000 Hz and working at a sampling frequency of 5kHz. The length of the impulse response should be 7.	08
7	a	Demonstrate interfacing multiple systems with different sampling rates.	08
	b	Briefly explain about DFT filter banks.	08
		OR	
8	a	Find an expression for interpolation by a factor I.	06
	b	Construct 2 channel quadrature mirror filter bank.	10

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Semester End Examination, May-June 2019

Course: **Digital Signal Processing**
Course Code: **EI63**
Semester: **VI**

Maximum marks: **100**
Duration: **03 hours**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			**Marks**
1	a	Find the mathematical relation between DFT and Z transform.	01
	b	Given statement is TRUE or FALSE."The DFT supports only circular convolution"	01
	c	What is the speed improvement factor in calculating 128 point DFT of a sequence using direct computation and FFT algorithm?	01
	d	The complex valued twiddle factor W_N can be represented as ________	01
	e	Draw the basic butterfly diagram of for DIF algorithm.	01
	f	Write the any two LIMITATIONS of Digital Signal Processing.	01
	g	Write the mathematical relation between continuous time frequency and discrete time frequency.	01
	h	Write the similarities between DIT and DIF radix 2 FFT algorithms.	01
	i	Find the value of W_8^{10}	01
	j	What are Gibbs oscillations in FIR filters?	01
	k	Write the application of Goertzel algorithm.	01
	l	Write the two properties of chebyshev filter.	01
	m	What is the basis for fourier series method of design? Why truncation is necessary.	01
	n	The process of signal at different sampling rates is called ________	01
	o	The ________ is the process of increasing sampling rate.	01
	p	If x[n] and y[n] are input and out put of a decimator with sampling rate conversion of D, then relation between x[n] and y[n]	01
	q	IIR filters are implemented by using ________ equations.	01
	r	How the analog frequency mapped to digital frequency in Bilinear transformation.	01
	s	How the poles of Chebyshev transfer function are located in S plane.	01
	t	What is the importance of poles in filter design?	01

2	a	Find the 8 point DFT of the following sequence by using direct DFT formula. Given x[n]= 〚(0.25)〛 ^n;where 0≤ n≤ 7	10
	b	Compute the circular convolution of two periodic sequences using graphical method. given that x1(n)={1,2,3,1} and x2(n)={4,3,2,2}	06
3	a	Apply DIT FFT radix 2 algorithm find the X(k) for the following discrete time sequence: X[n]= $(2)^n$ and N=8.	08
	b	Derive the CHIRP Z transform with necessity mathematical equations and model contour graphs.	08
4	a	Design a butterworth Low pass filter using bilinear transformation for the following specifications: $0.8 \leq \lvert H(e^{j\omega})\rvert \leq 1$; $0 \leq \omega \leq 0.2\pi$ and $\lvert H(e^{j\omega})\rvert \leq 0.2$; $0.6\pi \leq \omega \leq \pi$.	10
	b	Distinguish IIR and FIR digital filters with necessity mathematical equations.	06
5	a	Design an FIR linear phase filter approximating the ideal frequency response $H_d(w)=1$; $\lvert w\rvert \leq \pi/6$; $H_d(w)=0$; $\lvert w\rvert \leq \pi$; using the hamming window technique with N=7 samples.	08
	b	Discuss the FIR filter design using frequency sampling technique.	08
		OR	
6	a	Determine the transfer function of H(z) of an FIR filter to implement impulse sequence h(n)= δ[n]+2 δ(n-1)+2 δ(n-2) using frequency sampling technique.	06
	b	Find an expression for impulse response h[n] of a linear phase low pass FIR filter using KAISER window to satisfy the following magnitude response specifications for the equivalent analog filter, stop band attenuation=40dB, pass band ripple=0.01dB, transition width =1000πrad/sec, Ideal cut off frequency=2400 rad/sec, sampling frequency=10Khz.	10
7	a	Apply multirate signal processing concepts with block diagram, analyze M-channel Quadrature mirror filter (QMF) bank.	08
	b	The sequence x[n]={0,2,4,6,8} is interpolated using interpolator with b_k={0.5,1,0.5}, the interpolation factor is 2.Find interpolated sequence y[m] using input-output relation for the interpolator.	08
		OR	
8	a	Apply interpolation and decimation concepts write design steps of the interfacing of digital systems with different sampling rates.	08
	b	Apply multirate signal processing concepts write design steps of the analysis and synthesis in a filter bank.	08

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Makeup Examination, July 2019

Course: **Digital Signal Processing** Maximum marks: **100**

Course Code: **EI63** Duration: **03 hours**

Semester: **VI**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			Marks
1	a)	Give any two applications of DFT.	01
	b)	Why DFT is performed over DTFT.	01
	c)	What are the two methods of sectioned convolution.	01
	d)	The relation between Z- transform X(Z) & DFT X(K) is _______	01
	e)	State the computational requirements of FFT.	01
	f)	The number of complex multiplications involved in the direct computation of 8-point DFT is________	01
	g)	The DFT of a two sample sequence x(n)= {A,B} is x(k)=________	01
	h)	The number of butterflies in each stage of computation of 64- point radix-2 FFT is ________	01
	i)	What is the relationship between analog & digital frequencies in Impulse Invariant Transformation.	01
	j)	The impulse response is obtained by taking the inverse forward transform of the ______	01
	k)	Aliasing occurs only in _______ transformation.	01
	l)	The type-2 chebyshev response is also called ________ response.	01
	m)	What is the main advantage of winding?	01
	n)	In FIR filter with constant phase delay the impulse response is ________	01
	o)	Delay distortion is _______ with phase distortion.	01
	p)	For a linear phase filter the delay is ______	01
	q)	How is narrow band low pass filter characterised?	01
	r)	How many types of filter banks are there?	01
	s)	The reciprocal of the Nyquist rate is called ______	01
	t)	Define sampling theorem.	01
2	a)	Derive and explain any two properties of DFT.	06
	b)	Perform the linear convolution of the following sequences using overlap- save method x(n)={1,-2, 2,-1,3,-4,4,-3} & h(n)= {1,-1}	10

3 a) Find the 8- point DFT by radix – 2 DTF FFT algorithm $x(n)=\{1,1,1,1,1,1,1,1\}$ 08

b) Determine the DFT of the following sequences using DIF-FFT algorithm 08

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

4 a) The system function of an analog filter is expressed as $H_a(s)= 2/S(S+2)$ find the corresponding H(z) using Impulse Invariant Transformation for a sampling frequency of 4 samples/sec . 06

b) Design low pass butter worth filter using the bilinear transformation for satisfying the following constraints: 10

Pass band: 0-400Hz stop band: 2.1 – 4KHZ

Pass band ripple: 2dB Stop band attenuation: 20db

Sampling freq: 10KHZ

5 a) Derive and explain FIR filter design using frequency sampling technique. 06

b) Design a filter with 10

$$H_d(e^{jw}) = \begin{cases} e^{-j3w}, & -\pi/4 \le \omega \le \pi/4 \\ 0, & \pi/4 \le |\omega| \le \pi \end{cases}$$

Using Hamming window with N=7

OR

6 a) Design a high pass filter using Hamming window, with a cut off frequency of 1.2 rad/sec & N = 9 06

b) Design an FIR low pass filter using Kaiser window satisfying the following specifications. 10

$\alpha_p \le 0.1$ db, $\alpha_s \ge 38$ db

$\omega_p = 15$ rad/sec , $\omega_s = 25$ rad/sec & $\omega_{sf} = 80$ rad/sec

7 a) With neat diagram explain the working principle of Quadrature mirror filter banks. 08

b) Derive and explain the interfacing of digital systems with different sampling rates. 08

OR

8 a) With required mathematical equation, explain the process of interpolation & decimation. 08

b) Explain the implementation of the digital filter banks with neat block diagram. 08

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)
Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Make Up Examination, February 2019

Course: **Digital Signal Processing** Maximum marks: **100**
Course Code: **EC53** Duration: **03 hours**
Semester: **V**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			Marks
1	a)	Number of multiplications required for direct computation of 32-Point DFT are____ i) 32 ii) 64 iii)1024 iv) 16	01
	b)	Number of complex additions needed to compute N-point DFT using direct method are ____	01
	c)	Lengths of Linear and circular convolution of two sequences of length M and N are ____ and ______ respectively	01
	d)	Give relationship between DFT and DTFT?	01
	e)	Outputs of DIT-FFT algorithm are in ______ order	01
	f)	Number of memory elements needed for FFT algorithm are______ i) N^2 ii) $N \log_2 N$ iii) $N/2$ iv) $2N$	01
	g)	Give relation between number of stages and number of points in DFT sequence in the implementation of FFT algorithm.	01
	h)	Write two advantages of FFT algorithm over direct computation of DFT.	01
	i)	Write basic butterfly structure of DIT-FFT algorithm.	01
	j)	Define IIR Filters.	01
	k)	The maximum magnitude of $H(\Omega)$ in dB in normalized frequency response of the filter is ______. i)1 dB ii) 0 dB iii) 10 dB iv) None	01
	l)	The order of Chebyshev approximation is greater than that of Butterworth approximation for the given specifications.(TRUE/FALSE)	01
	m)	The relation between analog frequency (Ω) and digital frequency (ω) in impulse invariant transformation is ________.	01
	n)	Analog transformation used to transform a normalized LPF into BPF is ______.	01
	o)	How is the effect of frequency warping overcome in digital filter design?	01
	p)	FIR Filters are _______. i)All zero filters ii) Linear phase iii) both i & ii iv) None	01
	q)	Frequency response of a digital filter is $H(e^{j\omega})=(1/3)(1+\cos(\omega))e^{-j2.5\omega}$. Determine the Phase Delay and Group Delay.	01
	r)	Write general linear constant coefficient difference equation used for representing discrete time LTI system.	01
	s)	Give the mathematical equations for step size in the process of Quantization.	01
	t)	The accuracy of specifying filter coefficients in computer is limited by ______ i) Size of RAM ii) Word length of the computer iii) both i & ii iv) None	01

2 a) Compute 4-point DFT of x(n) = [1,-2, 3,-4] . Also find magnitude and phase. 08

b) Let X(k) be a 14-point DFT of length - 14 real sequence x(n). The first K-samples of X(k) are given by **X(0)= 12, X(1)= -1+3j, X(2)=3+4j, X(3)=1-5j, X(4)=1+2j, X(5)=6+3j, X(6)=-2-3j), X(7)= 10.** Compute the remaining samples of X(k). Also evaluate the following using obtained X(k). 08

i) x(0) **ii)** x(7) **iii)** $\sum_{n=0}^{13} x(n)$ **iv)** $\sum_{n=0}^{13} |x(n)|^2$

3 a) Find 4-point DFT of x(n)= δ(n)+2 δ(n-2)-3 δ(n-3) using DIT-FFT algorithm. 06

b) Determine the response of the LTI system with impulse response h(n)={0.5, 0.5, 0.5 } , x(n)={1,1} and using DIF FFT. 10

4 a) Determine the normalized transfer function of 2^{nd} order Chebyshev filter $H_N(s)$ The filter has pass band attenuation 2db. 06

b) Design a digital filter transfer function H(z) that when used in an A/D - H(z) – D/A structure, given an equivalent analog filter with the following specifications : 10

i) Monotonic in both pass band and stop band.

ii) Passband attenuatin : 1dB, Passband edge : 400π rad/sec

iii) Stopband attenuation: 20 dB, Stopband edge : 800π rad/sec

Sample Rate: 1000 samples/sec. **Use Bilinear transformation** to design the filter on an analog system equation. Use Butterworth filter prototype.

5 a) For the desired frequency response $H_d(e^{j\omega})$ design FIR filter $H(e^{j\omega})$ using hamming window. 12

$H_d(e^{j\omega})$ = {0 ; $-3\pi/4 \le \omega \le 3\pi/4$

{ e^{-j5w} ; $3\pi/4 \le |\omega| \le \pi$

b) Explain phase delay and group delay related to FIR filters. 04

OR

6 a) Determine the filter coefficients h(n) obtained by sampling $H_d(\omega)$ given by 10

$H_d(\omega)$ = { e^{-j3w} ; $0 \le \omega \le \pi/2$

{ 0 ; $\pi/2 \le |\omega| \le \pi$

Also obtain the frequency response H(ω).

b) List the advantages and disadvantages of FIR filter. 06

7 a) Starting from the basic equation for filters show how to realize IIR filter in Direct form-I. 08

b) Explain briefly effects of quantization of filter coefficients while realizing filters on hardware or software. 08

OR

8 a) Explain Limit cycle oscillations in recursive systems and Round off effects in digital filters. 08

b) Consider a IIR filter with system function 08

$$H(z) = \frac{Z^2 + \frac{1}{4}Z - \frac{1}{8}}{Z^2 - 1}$$

Sketch the direct form-I, direct form-II and cascade form realizations of the filter.

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)

Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Semester End Examination, December 2018/January 2019

Course: **Digital signal processing**
Course Code: **TE53**
Semester: **V**

Maximum marks: **100**
Duration: **3 hours**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			Marks
1	a	*Condition for sampling is* ____________	01
	b	$e^{+\frac{j2\pi}{4}}=$______	01
	c	Discrete version of DTFT is DFT . Is it True/False ?	01
	d	DTFT spectrum is ________ and __________	01
	e	DFT of δ(n) is ________	01
	f	The algorithm used to do the discrete convolution between a very long signal and a finite impulse response (FIR) filter is __________	01
	g	DIT algorithm divides the input sequence into__________	01
	h	Given that $W=e^{-j(2\pi/N)}$, where $N=3$. Then $F=W^N$ can be computed as F=	01
	i	The main two advantages of DSP over ASP are________, ________	01
	j	Number of complex multiplications in Radix-2 DIT-FFT are ________	01
	k	Number of complex additions in Radix-2 DIF-FFT are ________	01
	l	The number of stages in Radix-2 DIF-FFT is calculated as ________	01
	m	According to twiddle factor property $W^{(N+k)}$ = __________	01
	n	The order of the ideal band pass filter is ______	01
	o	The order of normalized Butterworth Low Pass filter is given by______	01
	p	The transformation technique in which there is one to one mapping from s-domain to z-domain is ________	01
	q	The 1st order normalized Butterworth low pass filter transfer functions is__	01
	r	Chebyshev Type-1 filter has ripples in _______band.	01
	s	The main advantage of FIR filter compare to IIR filter is ________	01
	t	For the minimum stop band attenuation of -21db , ______window is used.	01
2	a	The given signal is x(n) = $\frac{e^{jan}+e^{-jan}}{2}$, Using Linearity property Find its DFT	08

b Identify the following relations and prove the same 08

$DFT(x((-n))_N = X((-k))_N$

3 a Given $x(n) = \{1,2\}$, $w(n) = x(n) * y(n)$ and $w(n) = \{6,-1,10,4\}$. Compute $y(n)$. 06

b Using Overlap add technique find the system output with $h(n) = \{1, 2, 1\}$ and $x(n) = \{1, 2, 0, 2,-2,3,2,3,-1,3,2,1\}$. 10

4 a Explain the Radix-2 DIF FFT algorithm to find DFT of a given sequence. Also draw the 8 Point Butterfly diagram for the same. 08

b Find the 8-Point DFT of $x(n) = \{2,1,2,1\}$ using DIF FFT. Draw the signal flow graph with intermediate values. 08

5 a List the design steps to design Chebyshev type-1 low pass filter with proper design equations. 06

b For the analog transfer function $H(s) = \frac{b}{(s+a)^2+b^2}$ find H(z) using IIT. 08

c What is the normalized Butterworth low pass filter transfer function for order N = 3. 02

OR

6 a With suitable design equations show 10

i. Stability is retained in bilinear transformation technique, which is used to map H(s) to H(z).

ii. Relation between analog angular frequency and digital angular frequency..

b Sketch the direct form-1, direct form-II for the system function given below. 06

$$H(s) = \frac{2Z^2+Z-2}{Z^2-2}$$

7 a List the advantages and disadvantages of a FIR filters. 04

b Compare IIR filters and FIR filters. 04

c Determine the impulse response and frequency response of a low pass FIR filter with $\alpha = 3$ and $w_c = 1$ rad. 08

OR

8 a Consider a three-stage FIR lattice structure having the coefficients: k1 = 0.32, k2=-0.17and k3 = 0.4. draw the lattice structure and direct form -1 structure. 10

b Realize a FIR filter with its impulse response 06

$h(n) = \delta(n) - \frac{1}{35}\delta(n-1) + \frac{2}{15}\delta(n-2) + \frac{2}{15}\delta(n-3) - \frac{1}{35}\delta(n-4) + \delta(n-5)$ using

i. Direct form-1 ii. linear phase structure

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DAYANANDA SAGAR COLLEGE OF ENGINEERING

(An Autonomous Institute Affiliated to VTU, Belagavi)
Shavige Malleshwara Hills, Kumaraswamy Layout, Bengaluru-560078

UG Make-Up Examination, February 2019

Course: **Digital signal processing** Maximum marks: **100**
Course Code: **TE53** Duration: **3 hours**
Semester: **V**

Note: **i). Question ONE (a to t) has to be answered from pages 5 to 7 only.**
ii). Question 1 to 4 is compulsory.
iii). Any missing data should be suitably assumed.

Q. No.			Marks
1	a)	What is system?	01
	b)	The spectrum of DTFT is ________ and ________	01
	c)	What will be the sampling frequency for sampling of a signal with a band of 100Hz to 50KHz?	01
	d)	Define correlation.	01
	e)	The length of the auto correlation is given by ________	01
	f)	In The A/D converter, the meaning of quantization is________	01
	g)	The total trigonometric functions in twiddle factor is__________	01
	h)	The number of real multiplications in Direct computation of DFT is________	01
	i)	The Goertzel algorithm is efficient if the frequency samples are less than _____	01
	j)	What is the meaning of DIF in FFT algorithm?	01
	k)	For N=256 point DFT the total complex additions required in R-2 DIT FFT is____	01
	l)	The FFT is ______________ approach.	01
	m)	According to twiddle factor property $W_N^{(N+k)}$= __________	01
	n)	What are the basic elements to realize Digital filter system ?	01
	o)	The transfer function of 2nd order Normalized low pass Butterworth filter is __	01
	p)	The main advantage of FIR filter compare to IIR filter is ______	01
	q)	The IIR filter difference equation is recursive or Non recursive ?	01
	r)	y(n) = x(n)-2x(n-2)+3y(n-1). Is this difference equation belongs to FIR or IIR system ?	01
	s)	The minimum stopband attenuation of Hammining window is _______	01
	t)	Chebyshev Type-1 filter has ripples in ________band.	01
2	a)	State and prove: Circular Time shifting Property and Circular Frequency shifting Property.	08
	b)	Compute 8-Point DFT of a sequence $x(n) = (-1)^{n+1}$, $0 \le n \le 7$. Also plot the	08

magnitude of DFT.

3 a) Apply the Overlap add algorithm to find the filter output whose impulse response h(n) = {1,2,1} and the input signal to the filter is x(n) = {2,2,-1,1,3,-2,-2,-1,1,2,2,-2}. 10

b) State and prove the circular convolution property with respect to DFT. 06

4 a) Find the DFT of the following sequence using Radix-2 DIT FFT algorithm for N = 8. x(n) = { 4,2,1,4,6,3,5,2 } 08

b) With 8 point flow graph, explain the Radix-2 DIF-FFT algorithm. 08

5 a) Derive an expression for order of a low pass Butterworth filter. 08

b) Explain Bilinear transformation with stability criteria for transforming analog filter into digital filter. 08

OR

6 a) Explain the impulse invariant Transformation Technique. 08

b) Realize the Direct form-2 and parallel form for the given system function. 08

$$H(z) = \frac{(1 - Z^{-1})}{(1 - 3Z^{-1} + 4Z^{-2})}$$

7 a) Compare IIR filters and FIR filters. 05

b) Write equations of any four different windows used in design of FIR filters. 06

c) List the advantages and disadvantages of a FIR filters. 05

OR

8 a) Given the FIR filter with the following difference equation $y(n) = x(n) + 2.1\,x(n-1) + 3.5\,x(n-2) + 3.2x(n-3) + 5.2x(n-4)$. Sketch the lattice realization of the filter. 10

b) Realize a FIR filter with its impulse response 06

h(n) = $\delta(n) - \frac{1}{4}\delta(n-1) + \frac{1}{5}\delta(n-2) + \frac{1}{5}\delta(n-3) - \frac{1}{4}\delta(n-4) + \delta(n-5)$ using

i. linear phase structure, **ii.** Direct form-1