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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH] DECEMBER 2017

Paper Code: IT-307 ***Subject: Digital Signal Processing***

Time: 3 Hours ***Maximum Marks:*** 75

Note: Attempt any five questions including Q.no.1 which is compulsory. Assume missing data if any.

Q1 Attempt **any five:** **(5x5=25)**

(a) Find the Fourier Coefficient of the signal $f(t) = \sin \omega_0 t$.

(b) Verify following system for Linearity and Time Invariance:
(i)$y(t) = x^2(t)$, (ii) $y(t) = \sin t . x(t)$,(iii) $y(t) = x(at)$, and (iv) $y(t) = \log x(t)$.

(c) What is the difference between Causal System or Non-Causal System.

(d) Prove that discrete time harmonics are not always periodic in frequency.

(e) Find the Fourier Coefficient of the signal which is full wave rectifier signal.

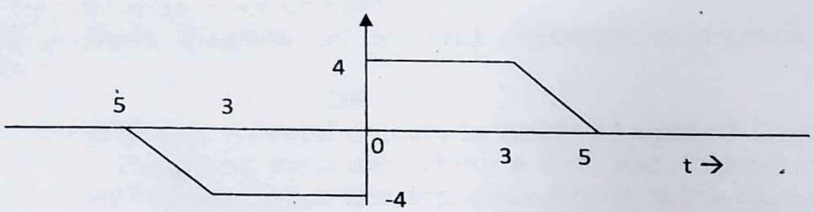
(f) Write a short note filter bank.

(g) Compare IIR and FIR.

(h) Explain the need of low pass filter with a decimator and mathematically prove that $\omega_x = \omega_y D$.

(i) Short note on Frequency Sampling realization of FIR filters.

Q2 (a) Signal f(t) is defined as below: **(6.5)**

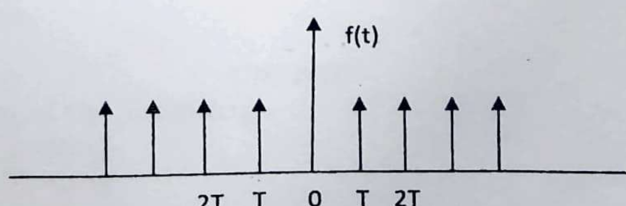


A signal g(t) is realized by multiplying f(t) with $\partial(t+4) + \partial(t-4)$ is the integral of the signal or power signal. Hence find the Energy or Power.

(b) Find the response of discrete time LTI system having the input and impulse responses as given below $f[n] = a^n u[n]$, $h[n] = a^n u[n]$. **(6)**

Q3 (a) Derive the relationship between Trigonometric Fourier Series and Exponential Fourier Series. **(6.5)**

(b) Draw the Complex Spectrum of the given below and also find the Fourier series. **(6)**



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Q4 (a) Find the Fourier Transform of the signal.

(i) $f(t) = \frac{1}{\pi t}$, (ii) $f(t) = t\left(\frac{\sin t}{\pi t}\right)^2$ (6.5)

(b) Find the number of complex additions and complex multiplications required to find DFT for 16 point signal. Compare them with number of computations required, if FFT algorithm is used. (6)

Q5 (a) Compute DFT of a sequence, $x(n) = \{1,2,2,2,1,0,0,0\}$ using DIF-FFT algorithm. Sketch its magnitude spectrum. (6.5)

(b) Find 8-point FFT of, $x(n) = \{1,2,2,2,1\}$ using signal flow graph of Radix-2 Decimation in frequency FFT. (6)

Q6 Derive the Expression for impulse invariance technique for obtaining transfer function of digital filter from analog filter. Derive necessary equation for relationship between frequency of analog and digital filter. (12.5)

Q7 Compare various windows used for designing FIR filters. (12.5)

Q8 Compare various windows used for designing FIR filters. (12.5)

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END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DEC. 2014-JAN. 2015

Paper Code: IT307 | **Subject: Digital Signal Processing**

Time : 3 Hours | **Maximum Marks :60**

Note: Attempt any five questions including Q.no.1 which is compulsory.

Q1 Explain the following briefly:- (2x10=20)

(a) Give the properties of Z-transformation.

(b) What is signal processing?

(c) Give some properties of DFT.

(d) Differentiate between FIR and IIR.

(e) Define Convolution.

(f) Why do we need FFT algorithms?

(g) What are the computational saving in using N point FFT algorithm?

(h) What are the advantages of FIR filters?

(i) Differentiate between DIT and DIF.

(j) Give some applications of DSP.

Q2 (a) What are typical signals? Give some examples of typical signal. (5)
(b) Explain the time-domain LTI system with an example. (5)

Q3 (a) Discuss the design procedure of FIR filter using frequency sampling method. (6)
(b) Give the block diagram representation of digital filter. (4)

Q4 (a) Derive the butterfly diagram of 8 point radix 2 DIF FFT algorithm and fully label it. (6)
(b) How can we classify signals? (4)

Q5 (a) Compute linear convolution of the two sequence x(n)={1,2,2,2} and h(n)={1,2,3,4}. (6)
(b) Derive expressions to relate z-transfer and DFT. (4)

Q6 (a) State and explain the scaling and time delay properties of z transform. (5)
(b) Describe different types of sampling methods. (5)

Q7 (a) Explain the classification of discrete signals. (4)
(b) Determine the response of LTI system when the input sequence is x(n)={-1,1,2,1,-1} using radix 2 DIF FFT. The impulse response is h(n)={-1,1,-1,1}. (6)

Q8 (a) Give some approaches of reducing the computation of an algorithm. (4)
(b) An 8 point sequence is given by x(n)={2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by radix DIT-FFT method. (6)

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END TERM EXAMINATION

SEVENTH SEMESTER [B.TECH./M.TECH.]- DECEMBER 2010

Paper Code: IT401	**Subject: Digital Signal Processing**
Paper ID: 15401	

Time : 3 Hours **Maximum Marks : 60**

Note: Attempt five questions including Q.1 which is compulsory.

Q1 (a) Discuss minimum phase and maximum phase transfer function. **(5)**

(b) A linear time invariant system is characterized by system function $H(z)=\dfrac{3-4z^{-1}}{1-3.5z^{-1}+1.5z^{-2}}$. Determine h(n). **(5)**

(c) Check if the system is LTI or not (i) $y(n)=\sum_{K=-\infty}^{n} x(k)$ (ii) y(n)=x(-n). **(5)**

(d) Derive the relationship between DFT (i) Z-transform (ii) Fourier series. **(5)**

Q2 (a) Compute the convolution y(n)=x(n)*h(n), $x(n)=\begin{cases}1 & n=-2,0,1\\ 2 & n=-1\\ 0 & elsewhere\end{cases}$,

$h(n)=\delta(n)-\delta(n-1)+\delta(n-4)+\delta(n-5)$. **(8)**

(b) Determine whether the signals are energy or power signal and also compute its value (i) $x(n)=e^{2n}u(n)$ (ii) $x(n)=(1/3)^n u(n)$. **(2)**

Q3 (a) Check if It following signal are causal or not:- **(3)**

(i) $y(n)=x(n)+x^2(n-1)$

(ii) $y(n)=x(2n)$

(iii) $y(n)=\sum_{K=-\infty}^{n+1} x(k)$

(b) Discuss sampling theory in frequency domain. **(2)**

(c) Find the cross correlation of two finite length sequence x(n)={1,2,1,1} and y(n)={1,1,2,1}. Also, show that $r_{xy}(l)=x(l)*y(-l)$. **(5)**

Q4 (a) Given the sequence x_1(n)={1,2,3,4}, x_2(n)={1,1,2,2}. Compute- **(8)**

(i) $x_3(n)=x_1(n)\;\; x_2(n)$.

(ii) Linear convolution using circular convolution.

(b) Derive the parseval's theorem. **(2)**

Q5 (a) Determine the causal signal x(n) having Z-transform $X(Z)=\dfrac{1}{(1-2z^{-1})(1-z^{-1})^2}$. **(6)**

(b) Prove the following property of DFT where X(K) is the N point DFT of x(n). if x(n) is real and even then X(K) is real and even. **(4)**

Q6 Find the DFT of a sequence x(n)={1,2,3,4,4,3,2,1} using OIT algorithm. Draw the structure and also show bit reversal. **(10)**

Q7 Find the IDFT of the sequence X(K)={4, 1-j2.414, 0, 1-j0.414, 0, 1+j0.414, 0, 1+j2.414} using DIF algorithm. **(10)**

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Q8 Draw the direct form II, cascade and parallel structure for the system described by the difference equation $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$. **(10)**

Q9 (a) Explain the design of IIR filter using (i) impulse invariance method (ii) Bilinear transformation method. **(4)**

(b) Realize the following system function using minimum no. of multipliers:- **(6)**

(i) $H(z) = 1 + \frac{1}{3}z^{-1} + \frac{1}{4}z^{-2} + \frac{1}{4}z^{-3} + \frac{1}{3}z^{-4} + z^{-5}$

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

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2013

Paper Code: IT307 **Subject: Digital Signal Processing**

Time : 3 Hours **Maximum Marks :60**

Note: Attempt any five questions. Usage of calculators is allowed.

Q1. Attempt any four parts. (3x4)

(i) Explain one-dimensional signal with suitable examples.

(ii) Distinguish between continuous time and discrete time signals.

(iii) Explain periodic signals.

(iv) Define the term "stability" for a linear time invariant system.

(v) Determine the *z*-transform as well as the Region of Convergence for $x[n]=(\frac{1}{2})^n u[n]$.

(vi) State and establish the circular – shift property for the Discrete Fourier Transform (DFT).

Q2. Attempt any three parts. (4x3)

(i) Consider the discrete time Linear Time Invariant (Linear Shift Invariant) system with input x[n] and output is y[n] for which y[n-1] – (10/9)y[n]+y[n+1] = x[n]. Determine the unit-response in *z*-domain.

(ii) Find the *z*-transform of the following:

(a) $x[n] = -n\, a^n\, u[-n-1]$

(b) $x[n] = a^n \sin(w\, n)\, u[n]$

(iii) Find the inverse *z*-transform of . $X(z)=\dfrac{1+z^{-1}+2z^{-2}}{(1-\frac{1}{2}z^{-1})(1-\frac{1}{3}z^{-1})(1-\frac{1}{4}z^{-1})};|z|>\frac{1}{2}$

(iv) Given that the *z*-transform of x[n] is X(z), find the *z*-transform of x[n] – x[n-1]. Establish your result.

Q3. Attempt any three parts. (4x3)

(a) Determine the output of the linear filter whose impulse response is h[n] = {1,-2,3} and the input signal is x[n] = {-1,2,-3,4,-5,6,-8} using either overlap-save or overlap-add method. State the method used.

(b) State and establish the Parseval's property/theorem for DFT.

(c) If the DFT of two N point sequences x[n] and y[n] is X[k] and Y[k], respectively. What is the DFT of x[n]y[n].

(d) Find the circular convolution of the given sequences: x[n] = {1,3,5,7} and y[n] = {2,4,6,8}.

Q4. Attempt all parts. (9+3)

(a) Determine the DFT of the given data sequence: x[n] = {2,1,4,6,5,8,3,9} using decimation in time FFT.

(b) What is the computational complexity of the FFT algorithm. Write a brief note.

Q5. For the system described by the difference equation: (6+6)

$y[n] - (13/12)\, y[n-1] - (1/24)\, y[n-3] = x[n]+2x[n-1]$

obtain the following realizations:

i. Direct Form I

ii. Parallel

Q6. Obtain the direct form structure and the cascade structure form for: (6+6)

$H(z) = 1 + 8z^{-1} + 21z^{-2} + 35z^{-3} + 28z^{-4} + 15z^{-5}$.

Q7. Attempt any 3 parts. (4x3)

i. Determine the impulse invariant digital filter transfer function corresponding to the transfer function for an anlaog filter given by

$$H(s)=\frac{s+2}{(s+2)^2+4}$$

ii. Compare and contrast IIR and FIR filters.

iii. Write short note on the sampling theorem. Determine the Nyquist rate / sampling rate for the given signal:

$x(t) = 2\cos(50\pi t) + 3\sin(150\pi t) - 4\cos(300\pi t)$

iv. Write short note on linear phase filter.

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH.] DECEMBER 2016

Paper Code: IT-307 **Subject: Digital Signal Processing**

Time: 3 Hours **Maximum Marks: 60**

Note: Attempt any five questions including Q no.1 which is compulsory. Assume suitable missing data, if any.

Q1 (a) Check the system $y(n) = a^n u(n)$ for stability. **(3)**

(b) Check $y(n) = \sin(n)x(2n-5)$ for Time-Invariance and $y(n) = \sin(n+3)x(n-4) + x(n+2)$ for Causality. **(3)**

(c) Find IR of system $y(n) + 4y(n-1) + 4y(n-2) = r(n-2)$ **(3)**

(d) Perform convolution of two periodic sequences $x_1(n) = \{1,2,3,4\}$ and $x_2(n) = \{5,6,7,8\}$ using Circular convolution. **(3)**

Q2 (a) The IR of a FIR filter, $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) + \frac{1}{16}\delta(n-2)$. Find the response of this filter to $x(n) = \sin\left(\frac{n\pi}{2}\right)u(n)$ **(6)**

(b) Frequency response of a FIR filter is given as $H(e^{Jw}) = e^{-3Jw}[2 + 1.8\cos 3\omega t + 1.2\cos 2\omega t + 0.5\cos \omega t]$ Find IR of filter and identify filter type based on its passband. **(6)**

Q3 (a) Prove Initial Value Theorem of Z Transform. **(6)**

(b) Two systems having IR $h_1(n) = \left(\frac{1}{4}\right)^n u(n)$. And $h_2(n) = \left(\frac{1}{2}\right)^n u(n)$ are connected in cascade find the next IR. (6)

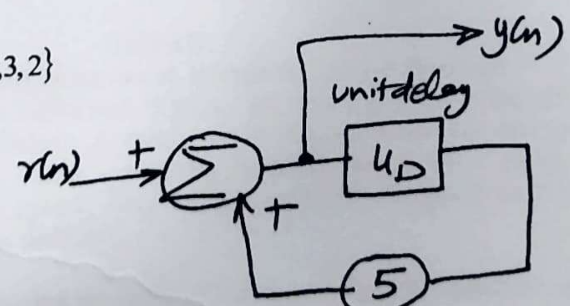
Q4 (a) Show that the magnitude response of an FIR filter at DC can be obtained as $|H(0)| = \left|\sum_{n=0}^{N-1} h(n)\right|$ and at frequency $w = \pi$ as $|H(\pi)| = \left|\sum_{n=0}^{N-1} \cos n\pi h(n)\right|$. **(6)**

(b) For the DTS shown, find **(6)**

(i) LDE

(ii) IR

(iii) Output if $r(n) = \{1,3,2\}$



Q5 Discuss-

(a) Properties of z-transform. **(6)**

(b) Linear convolution using DFT. **(6)**

Q6 The TF of a DT Causal system is $H(z) = \dfrac{1 - \frac{1}{2}z^{-1}}{1 - z^{-1} + \frac{3}{16}z^{-2}}$ obtain **(12)**

(a) Difference Equation.

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(b) Show DF-I, DF-II, Cascade and Parallel realization of this system.
(c) Find IR, step response and response to input

(i) $x(n)=\left(\frac{1}{2}\right)^n u(n)$ [an exponential excitation]

(ii) $x(n)=2\sin\left(\frac{\pi n}{3}-\frac{\pi}{5}\right)$ [a sinusoidal excitation]

Q7 Derive & explain the decimation in Time & Decimation in Frequency techniques for evaluating FFT. **(12)**

Q8 (a) The signal $f(t)=(0.8)^t u(t)$ is discretized to $f(n)=(0.8)^n u(n)$ having infinite length. Find the DFT of this signal, may be evaluated through an 8-point rectangular window. **(6)**
(b) Write short note on IIR filters. **(6)**

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END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2015

Paper Code: IT307	*Subject: Digital Signal Processing*
Time : 3 Hours	*Maximum Marks :60*

Note: Attempt any five questions. Use of calculator is permitted.

Q1. Let x[n], y[n] and w[n] denote three arbitrary sequences. Show that: (6+6)

(a) Discrete convolution is commutative, i.e.,

$x[n] * y[n] = y[n] * x[n]$

(b) Discrete convolution is associative, i.e.,

$x[n] * (y[n] * w[n]) = (x[n] * y[n]) * w[n]$

Q2. For each of the following systems, determine whether or not the system is (1) stable), (2) casual, (3) linear, and (4) shift-invariant: (2x6)

(a) $y[n] = g[n]\, x[n]$

(b) $y[n]=\sum_{k=n_0}^{n} x[k]$

(c) $y[n]=\sum_{k=n-n_0}^{n+n_0} x[k]$

(d) $y[n] = x[n-n_0]$

(e) $y[n]=e^{x[n]}$

(f) $y[n] = ax[n] + b$ where $a,b > 0$

Q3 Find the z-transform of the following: (6+6)

(a) $x[n]=a^n \sin(\omega n) u[n]$ (b) $x[n]=a^n u[n]-b^n u[-n-1]$, (a and b) <1, b>a

Q4 Determine the impulse response of the FIR filter whose impulse response is h[n] = {1,-2,3} and the input signal is x[n] = {1,-2,3,-4,5,-6,7,-8,9}. Use any method for calculation of the concerned DFT and then use the following method for calculation of the linear convolution: (6+6)

(a) Overlap Save (b) Overlap Add

Q5 A system is described by the difference equation (4x3)

$y[n] - (3/4)\, y[n-1] + (1/2)\, y[n-2] = x[n] + (1/2)\, x[n-1]$

Draw a signal flow graph to implement this system in each of the following forms:

(a) Direct form I,

(b) Direct for II,

(c) Cascade and

(d) Parallel

Q6 Design a digital lowpass Butterworth filter worth a passband magnitude characteristic that is constant within 0.75 dB for a frequency below $w = 0.2613\pi$ and stopband attenuation of at least 20dB for frequencies between $w = 0.4018\pi$ and π. Use the Impulse Invariant Design Method. (12)

Q7a Determine the DFT of the signal x[n] = {2,1,4,6,5,8,3,9} by decimation in time FFT. (8)

Q7b What is the time complexity of the (naive) DFT algorithm, and the time complexity of the radix-2 Decimation in time FFT algorithm. (4)

Q8 Write short notes on any two of the following: (6+6)

(a) Sampling Theorem

(b) FIR filter design with windows

(c) Bi-linear transformation for Filter design

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH] NOVEMBER –DECEMBER 2019

Paper Code: IT 307	**Subject: Digital Signal Processing**
Time : 3 Hours	**Maximum Marks :75**

Note: Attempt any five questions including Q. No. 1 which is compulsory. Assume missing data if any.

Q1. a) Compare between DFT and FFT. **(5)**

b) Define linearity and shift invariance properties of the discrete time systems verify there conditions for the following systems: **(5)**

i) $T[x(n)] = \sum_{k=n_0}^{n} x^{(k)}$ ii) $T[x(n)] = e^{x(n)}$

c) Describe methods for finding Inverse Z- bantam. **(5)**

d) Discuss the design for FIR differentiator. **(5)**

e) Compare FIR and IIR system. **(5)**

Q2. a) Discuss the Z-transform theorems and properties. **(6)**

b) Perform linear convolution for the input sequence:- X(n)= {1, 2,3,1,4} and h(n)={ 1,2,3,4}. **(6.5)**

Q3. a) Explain DFT. Prove the following properties of DFT when x(k) is the N-point. **(6)**
i) If x(n) is real and odd.
ii) If x(n) is imaginary and odd.

b) Determine the Z-transform of the following sequences and give their region of convergence: **(6.5)**

i) $\left(\frac{1}{2}\right)^n u(n)$ ii) $\left(\frac{1}{2}\right)^n (u(n)-u(n-10))$

Q4. Explain decimation in-time FFT algorithm for computing DFT. Compute DFT for the sequence {1,4,8,6,3,5,6,2} using FFT algorithm. **(12.5)**

Q5. a) Give the symmetry properties of the DFT of a complex sequence and explain them. **(6)**

b) What are the sample-hold circuits? Explain with the help of an example. **(6.5)**

Q6. a) Discuss the frequency response of the discrete-time system. **(6)**

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b) A casual linear shift invariant filter system has the system function. **(6.5)**

$$H(z) = \frac{1+0.875Z^{-1}}{(1+0.2Z^{-1}+0.9Z^{-2})(1-0.7Z^{-1})}$$

Draw the signal flow graph using

i) Direct form –II
ii) Cascade of the first and second order systems in transposed direct form II.

Q7. Implement the all pass filter $H_a P(Z) = \frac{-0.5120Z^{-1}-0.8Z^{-2}+Z^{-3}}{1-0.8Z^{-1}+0.6402Z^{-2}-0.512Z^{-3}}$ using a lattice filter structure. **(12.5)**

Q8. a) How digital filter specification are given? Explain with the help of magnitude response specifications. **(6)**

b) Explain the process of IIR filter design using a bilinear transformation. **(6.5)**

Q9. Discuss the cascade, parallel and transposed terms of the IIR filter structure. **(12.5)**

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END TERM EXAMINATION

FIFTH SEMESTER [B.TECH.] DECEMBER 2016

Paper Code: IT-307 **Subject: Digital Signal Processing**

Time: 3 Hours **Maximum Marks: 60**

Note: Attempt any five questions including Q no.1 which is compulsory. Assume suitable missing data, if any.

Q1 (a) Check the system $y(n) = a^n u(n)$ for stability. **(3)**

(b) Check $y(n) = \sin(n)x(2n-5)$ for Time-Invariance and $y(n) = \sin(n+3)x(n-4) + x(n+2)$ for Causality. **(3)**

(c) Find IR of system $y(n) + 4y(n-1) + 4y(n-2) = r(n-2)$ **(3)**

(d) Perform convolution of two periodic sequences $x_1(n) = \{1,2,3,4\}$ and $x_2(n) = \{5,6,7,8\}$ using Circular convolution. **(3)**

Q2 (a) The IR of a FIR filter, $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) + \frac{1}{16}\delta(n-2)$. Find the response of this filter to $x(n) = \sin\left(\frac{n\pi}{2}\right)u(n)$ **(6)**

(b) Frequency response of a FIR filter is given as $H(e^{jw}) = e^{-3Jw}[2 + 1.8\cos 3\omega t + 1.2\cos 2\omega t + 0.5\cos\omega t]$ Find IR of filter and identify filter type based on its passband. **(6)**

Q3 (a) Prove Initial Value Theorem of Z Transform. **(6)**

(b) Two systems having IR $h_1(n) = \left(\frac{1}{4}\right)^n u(n)$. And $h_2(n) = \left(\frac{1}{2}\right)^n u(n)$ are connected in cascade find the next IR. (6)

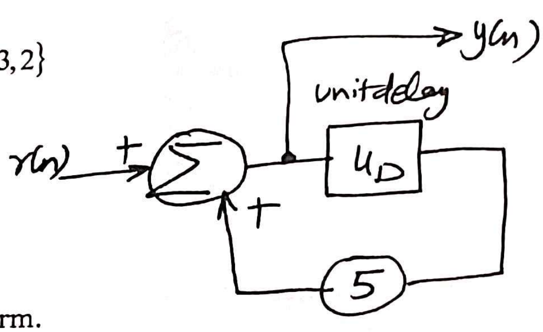
Q4 (a) Show that the magnitude response of an FIR filter at DC can be obtained as $|H(0)| = \left|\sum_{n=0}^{N-1} h(n)\right|$ and at frequency $w = \pi$ as $|H(\pi)| = \left|\sum_{n=0}^{N-1} \cos n\pi h(n)\right|$. **(6)**

(b) For the DTS shown, find **(6)**

(i) LDE

(ii) IR

(iii) Output if $r(n) = \{1,3,2\}$



Q5 Discuss-

(a) Properties of z-transform. **(6)**

(b) Linear convolution using DFT. **(6)**

Q6 The TF of a DT Causal system is $H(z) = \dfrac{1 - \frac{1}{2}z^{-1}}{1 - z^{-1} + \frac{3}{16}z^{-2}}$ obtain **(12)**

(a) Difference Equation.

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(b) Show DF-I, DF-II, Cascade and Parallel realization of this system.
(c) Find IR, step response and response to input

(i) $x(n)=\left(\frac{1}{2}\right)^n u(n)$ [an exponential excitation]

(ii) $x(n)=2\sin\left(\frac{\pi n}{3}-\frac{\pi}{5}\right)$ [a sinusoidal excitation]

Q7 Derive & explain the decimation in Time & Decimation in Frequency techniques for evaluating FFT. **(12)**

Q8 (a) The signal $f(t)=(0.8)^t u(t)$ is discretized to $f(n)=(0.8)^n u(n)$ having infinite length. Find the DFT of this signal, may be evaluated through an 8-point rectangular window. **(6)**
(b) Write short note on IIR filters. **(6)**

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FIFTH SEMESTER [B.TECH.] DECEMBER 2016

Paper Code: IT-307 | **Subject: Digital Signal Processing**

Time: 3 Hours | **Maximum Marks: 60**

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(b) Check $y(n) = \sin(n)x(2n-5)$ for Time-Invariance and $y(n) = \sin(n+3)x(n-4) + x(n+2)$ for Causality. (3)

(c) Find IR of system $y(n) + 4y(n-1) + 4y(n-2) = r(n-2)$ (3)

(d) Perform convolution of two periodic sequences $x_1(n) = \{1,2,3,4\}$ and $x_2(n) = \{5,6,7,8\}$ using Circular convolution. (3)

Q2 (a) The IR of a FIR filter, $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) + \frac{1}{16}\delta(n-2)$. Find the response of this filter to $x(n) = \sin\left(\frac{n\pi}{2}\right)u(n)$ (6)

(b) Frequency response of a FIR filter is given as $H(e^{jw}) = e^{-3Jw}[2 + 1.8\cos 3\omega t + 1.2\cos 2\omega t + 0.5\cos\omega t]$ Find IR of filter and identify filter type based on its passband. (6)

Q3 (a) Prove Initial Value Theorem of Z Transform. (6)

(b) Two systems having IR $h_1(n) = \left(\frac{1}{4}\right)^n u(n)$. And $h_2(n) = \left(\frac{1}{2}\right)^n u(n)$ are connected in cascade find the next IR. (6)

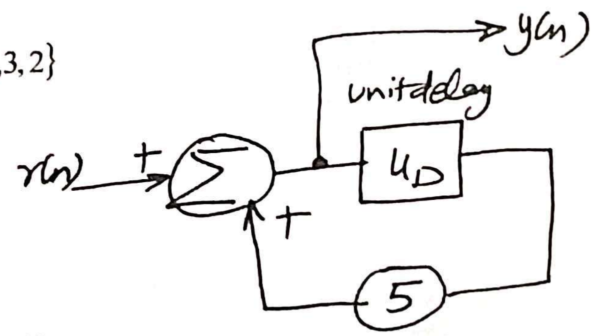
Q4 (a) Show that the magnitude response of an FIR filter at DC can be obtained as $|H(0)| = \left|\sum_{n=0}^{N-1} h(n)\right|$ and at frequency $w = \pi$ as $|H(\pi)| = \left|\sum_{n=0}^{N-1} \cos n\pi h(n)\right|$. (6)

(b) For the DTS shown, find (6)

(i) LDE

(ii) IR

(iii) Output if $r(n) = \{1,3,2\}$



Q5 Discuss-

(a) Properties of z-transform. (6)

(b) Linear convolution using DFT. (6)

Q6 The TF of a DT Causal system is $H(z) = \dfrac{1 - \frac{1}{2}z^{-1}}{1 - z^{-1} + \frac{3}{16}z^{-2}}$ obtain (12)

(a) Difference Equation.

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(b) Show DF-I, DF-II, Cascade and Parallel realization of this system.
(c) Find IR, step response and response to input

(i) $x(n) = \left(\frac{1}{2}\right)^n u(n)$ [an exponential excitation]

(ii) $x(n) = 2\sin\left(\frac{\pi n}{3} - \frac{\pi}{5}\right)$ [a sinusoidal excitation]

Q7 Derive & explain the decimation in Time & Decimation in Frequency techniques for evaluating FFT. (12)

Q8 (a) The signal $f(t) = (0.8)^t u(t)$ is discretized to $f(n) = (0.8)^n u(n)$ having infinite length. Find the DFT of this signal, may be evaluated through an 8-point rectangular window. (6)

(b) Write short note on IIR filters. (6)

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2015

Paper Code: IT307 Subject: Digital Signal Processing

Time : 3 Hours Maximum Marks :60

Note: Attempt any five questions. Use of calculator is permitted.

Q1. Let x[n], y[n] and w[n] denote three arbitrary sequences. Show that: (6+6)

(a) Discrete convolution is commutative, i.e.,

x[n] * y[n] = y[n] * x[n]

(b) Discrete convolution is associative, i.e.,

x[n] * (y[n] * w[n]) = (x[n] * y[n]) * w[n]

Q2. For each of the following systems, determine whether or not the system is (1) stable), (2) casual, (3) linear, and (4) shift-invariant: (2x6)

(a) y[n] = g[n] x[n]

(b) $y[n]=\sum_{k=n_0}^{n} x[k]$

(c) $y[n]=\sum_{k=n-n_0}^{n+n_0} x[k]$

(d) $y[n] = x[n-n_0]$

(e) $y[n]=e^{x[n]}$

(f) y[n] = ax[n] +b where a,b >0

Q3 Find the z-transform of the following: (6+6)

(a) $x[n]=a^n \sin(\omega n) u[n]$ (b) $x[n]=a^n u[n]-b^n u[-n-1]$, (a and b) <1, b>a

Q4 Determine the impulse response of the FIR filter whose impulse response is h[n] = {1,-2,3} and the input signal is x[n] = {1,-2,3,-4,5,-6,7,-8,9}. Use any method for calculation of the concerned DFT and then use the following method for calculation of the linear convolution: (6+6)

(a) Overlap Save (b) Overlap Add

Q5 A system is described by the difference equation (4x3)

y[n] - (3/4) y[n-1] + (1/2) y[n-2] = x[n] + (1/2) x[n-1]

Draw a signal flow graph to implement this system in each of the following forms:

(a) Direct form I,

(b) Direct for II,

(c) Cascade and

(d) Parallel

Q6 Design a digital lowpass Butterworth filter worth a passband magnitude characteristic that is constant within 0.75 dB for a frequency below w = 0.2613π and stopband attenuation of at least 20dB for frequencies between w = 0.4018π and π. Use the Impulse Invariant Design Method. (12)

Q7a Determine the DFT of the signal x[n] = {2,1,4,6,5,8,3,9} by decimation in time FFT. (8)

Q7b What is the time complexity of the (naive) DFT algorithm, and the time complexity of the radix-2 Decimation in time FFT algorithm. (4)

Q8 Write short notes on any two of the following: (6+6)

(a) Sampling Theorem

(b) FIR filter design with windows

(c) Bi-linear transformation for Filter design

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH] NOVEMBER -DECEMBER 2019

Paper Code: IT 307	**Subject: Digital Signal Processing**
Time : 3 Hours	**Maximum Marks :75**

Note: Attempt any five questions including Q. No. 1 which is compulsory. Assume missing data if any.

Q1. a) Compare between DFT and FFT. **(5)**

b) Define linearity and shift invariance properties of the discrete time systems verify there conditions for the following systems: **(5)**

i) $T[x(n)] = \sum_{k=n_0}^{n} x^{(k)}$ ii) $T[x(n)] = ex^{(n)}$

c) Describe methods for finding Inverse Z- bantam. **(5)**

d) Discuss the design for FIR differentiator. **(5)**

e) Compare FIR and IIR system. **(5)**

Q2. a) Discuss the Z-transform theorems and properties. **(6)**

b) Perform linear convolution for the input sequence:-
X(n)= {1, 2,3,1,4} and h(n)={ 1,2,3,4}. **(6.5)**

Q3. a) Explain DFT. Prove the following properties of DFT when x(k) is the N-point. **(6)**
i) If x(n) is real and odd.
ii) If x(n) is imaginary and odd.

b) Determine the Z-transform of the following sequences and give their region of convergence: **(6.5)**

i) $\left(\frac{1}{2}\right)^n u(n)$ ii) $\left(\frac{1}{2}\right)^n (u(n)-u(n-10))$

Q4. Explain decimation in-time FFT algorithm for computing DFT. Compute DFT for the sequence {1,4,8,6,3,5,6,2} using FFT algorithm. **(12.5)**

Q5. a) Give the symmetry properties of the DFT of a complex sequence and explain them. **(6)**

b) What are the sample-hold circuits? Explain with the help of an example. **(6.5)**

Q6. a) Discuss the frequency response of the discrete-time system. **(6)**

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b) A casual linear shift invariant filter system has the system function. **(6.5)**

$$H(z) = \frac{1+0.875Z^{-1}}{(1+0.2Z^{-1}+0.9Z^{-2})(1-0.7Z^{-1})}$$

Draw the signal flow graph using

i) Direct form –II
ii) Cascade of the first and second order systems in transposed direct form II.

Q7. Implement the all pass filter $H_a P(Z) = \frac{-0.5120Z^{-1}-0.8Z^{-2}+Z^{-3}}{1-0.8Z^{-1}+0.6402Z^{-2}-0.512Z^{-3}}$ using a lattice filter structure. **(12.5)**

Q8. a) How digital filter specification are given? Explain with the help of magnitude response specifications. **(6)**

b) Explain the process of IIR filter design using a bilinear transformation. **(6.5)**

Q9. Discuss the cascade, parallel and transposed terms of the IIR filter structure. **(12.5)**

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2015

Paper Code: IT307	Subject: Digital Signal Processing
Time : 3 Hours	Maximum Marks :60

Note: Attempt any five questions. Use of calculator is permitted.

Q1. Let x[n], y[n] and w[n] denote three arbitrary sequences. Show that: (6+6)
(a) Discrete convolution is commutative, i.e.,
$x[n] * y[n] = y[n] * x[n]$
(b) Discrete convolution is associative, i.e.,
$x[n] * (y[n] * w[n]) = (x[n] * y[n]) * w[n]$

Q2. For each of the following systems, determine whether or not the system is (1) stable), (2) casual, (3) linear, and (4) shift-invariant: (2x6)

(a) $y[n] = g[n]\,x[n]$

(b) $y[n]=\sum_{k=n_0}^{n} x[k]$

(c) $y[n]=\sum_{k=n-n_0}^{n+n_0} x[k]$

(d) $y[n] = x[n-n_0]$

(e) $y[n]=e^{x[n]}$

(f) $y[n] = ax[n] + b$ where $a,b > 0$

Q3 Find the z-transform of the following: (6+6)
(a) $x[n]=a^n \sin(\omega n) u[n]$ (b) $x[n]=a^n u[n]-b^n u[-n-1]$, (a and b) <1, b>a

Q4 Determine the impulse response of the FIR filter whose impulse response is h[n] = {1,-2,3} and the input signal is x[n] = {1,-2,3,-4,5,-6,7,-8,9}. Use any method for calculation of the concerned DFT and then use the following method for calculation of the linear convolution: (6+6)
(a) Overlap Save (b) Overlap Add

Q5 A system is described by the difference equation (4x3)
$y[n] - (3/4)\, y[n-1] + (1/2)\, y[n-2] = x[n] + (1/2)\, x[n-1]$
Draw a signal flow graph to implement this system in each of the following forms:
(a) Direct form I,
(b) Direct for II,
(c) Cascade and
(d) Parallel

Q6 Design a digital lowpass Butterworth filter worth a passband magnitude characteristic that is constant within 0.75 dB for a frequency below $w = 0.2613\pi$ and stopband attenuation of at least 20dB for frequencies between $w = 0.4018\pi$ and π. Use the Impulse Invariant Design Method. (12)

Q7a Determine the DFT of the signal x[n] = {2,1,4,6,5,8,3,9} by decimation in time FFT. (8)

Q7b What is the time complexity of the (naive) DFT algorithm, and the time complexity of the radix-2 Decimation in time FFT algorithm. (4)

Q8 Write short notes on any two of the following: (6+6)
(a) Sampling Theorem
(b) FIR filter design with windows
(c) Bi-linear transformation for Filter design

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(Please write your Exam Roll No.)　　　　　　　　**Exam Roll No. ....**

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH.] DECEMBER 2016

Paper Code: IT-307　　　　　**Subject: Digital Signal Processing**

Time: 3 Hours　　　　　**Maximum Marks: 60**

Note: Attempt any five questions including Q no.1 which is compulsory. Assume suitable missing data, if any.

Q1 (a) Check the system $y(n) = a^n u(n)$ for stability. **(3)**

(b) Check $y(n) = \sin(n)x(2n-5)$ for Time-Invariance and $y(n) = \sin(n+3)x(n-4) + x(n+2)$ for Causality. **(3)**

(c) Find IR of system $y(n) + 4y(n-1) + 4y(n-2) = r(n-2)$ **(3)**

(d) Perform convolution of two periodic sequences $x_1(n) = \{1,2,3,4\}$ and $x_2(n) = \{5,6,7,8\}$ using Circular convolution. **(3)**

Q2 (a) The IR of a FIR filter, $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) + \frac{1}{16}\delta(n-2)$. Find the response of this filter to $x(n) = \sin\left(\frac{n\pi}{2}\right)u(n)$ **(6)**

(b) Frequency response of a FIR filter is given as $H(e^{jw}) = e^{-3Jw}[2 + 1.8\cos 3\omega t + 1.2\cos 2\omega t + 0.5\cos\omega t]$ Find IR of filter and identify filter type based on its passband. **(6)**

Q3 (a) Prove Initial Value Theorem of Z Transform. **(6)**

(b) Two systems having IR $h_1(n) = \left(\frac{1}{4}\right)^n u(n)$. And $h_2(n) = \left(\frac{1}{2}\right)^n u(n)$ are connected in cascade find the next IR. (6)

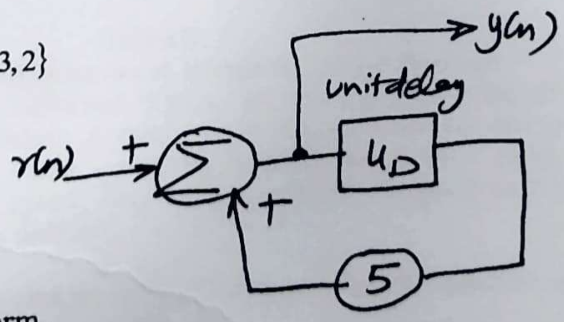
Q4 (a) Show that the magnitude response of an FIR filter at DC can be obtained as $|H(0)| = \left|\sum_{n=0}^{N-1} h(n)\right|$ and at frequency $w = \pi$ as $|H(\pi)| = \left|\sum_{n=0}^{N-1} \cos n\pi h(n)\right|$. **(6)**

(b) For the DTS shown, find **(6)**

(i) LDE

(ii) IR

(iii) Output if $r(n) = \{1,3,2\}$



Q5 Discuss-

(a) Properties of z-transform. **(6)**

(b) Linear convolution using DFT. **(6)**

Q6 The TF of a DT Causal system is $H(z) = \dfrac{1 - \frac{1}{2}z^{-1}}{1 - z^{-1} + \frac{3}{16}z^{-2}}$ obtain **(12)**

(a) Difference Equation.

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(b) Show DF-I, DF-II, Cascade and Parallel realization of this system.
(c) Find IR, step response and response to input

(i) $x(n)=\left(\frac{1}{2}\right)^n u(n)$ [an exponential excitation]

(ii) $x(n)=2\sin\left(\frac{\pi n}{3}-\frac{\pi}{5}\right)$ [a sinusoidal excitation]

Q7 Derive & explain the decimation in Time & Decimation in Frequency techniques for evaluating FFT. **(12)**

Q8 (a) The signal $f(t)=(0.8)^t u(t)$ is discretized to $f(n)=(0.8)^n u(n)$ having infinite length. Find the DFT of this signal, may be evaluated through an 8-point rectangular window. **(6)**
(b) Write short note on IIR filters. **(6)**

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2013

Paper Code: IT307 **Subject: Digital Signal Processing**

Time : 3 Hours **Maximum Marks :60**

Note: Attempt any five questions. Usage of calculators is allowed.

Q1. Attempt any four parts. (3x4)

(i) Explain one-dimensional signal with suitable examples.

(ii) Distinguish between continuous time and discrete time signals.

(iii) Explain periodic signals.

(iv) Define the term "stability" for a linear time invariant system.

(v) Determine the *z*-transform as well as the Region of Convergence for $x[n]=(\frac{1}{2})^n u[n]$.

(vi) State and establish the circular – shift property for the Discrete Fourier Transform (DFT).

Q2. Attempt any three parts. (4x3)

(i) Consider the discrete time Linear Time Invariant (Linear Shift Invariant) system with input x[n] and output is y[n] for which y[n-1] – (10/9)y[n]+y[n+1] = x[n]. Determine the unit-response in *z*-domain.

(ii) Find the *z*-transform of the following:

(a) $x[n] = -n\, a^n\, u[-n-1]$

(b) $x[n] = a^n \sin(w\, n)\, u[n]$

(iii) Find the inverse *z*-transform of . $X(z)=\dfrac{1+z^{-1}+2z^{-2}}{(1-\frac{1}{2}z^{-1})(1-\frac{1}{3}z^{-1})(1-\frac{1}{4}z^{-1})};|z|>\frac{1}{2}$

(iv) Given that the *z*-transform of x[n] is X(z), find the *z*-transform of x[n] – x[n-1]. Establish your result.

Q3. Attempt any three parts. (4x3)

(a) Determine the output of the linear filter whose impulse response is h[n] = {1,-2,3} and the input signal is x[n] = {-1,2,-3,4,-5,6,-8} using either overlap-save or overlap-add method. State the method used.

(b) State and establish the Parseval's property/theorem for DFT.

(c) If the DFT of two N point sequences x[n] and y[n] is X[k] and Y[k], respectively. What is the DFT of x[n]y[n].

(d) Find the circular convolution of the given sequences: x[n] = {1,3,5,7} and y[n] = {2,4,6,8}.

Q4. Attempt all parts. (9+3)

(a) Determine the DFT of the given data sequence: x[n] = {2,1,4,6,5,8,3,9} using decimation in time FFT.

(b) What is the computational complexity of the FFT algorithm. Write a brief note.

Q5. For the system described by the difference equation: (6+6)

$$y[n] - (13/12)\, y[n-1] - (1/24)\, y[n-3] = x[n]+2x[n-1]$$

obtain the following realizations:

i. Direct Form I

ii. Parallel

Q6. Obtain the direct form structure and the cascade structure form for: (6+6)

$$H(z) = 1 + 8z^{-1} + 21z^{-2} + 35z^{-3} + 28z^{-4} + 15z^{-5}.$$

Q7. Attempt any 3 parts. (4x3)

i. Determine the impulse invariant digital filter transfer function corresponding to the transfer function for an anlaog filter given by

$$H(s)=\frac{s+2}{(s+2)^2+4}$$

ii. Compare and contrast IIR and FIR filters.

iii. Write short note on the sampling theorem. Determine the Nyquist rate / sampling rate for the given signal:

$x(t) = 2\cos(50\pi t) + 3\sin(150\pi t) - 4\cos(300\pi t)$

iv. Write short note on linear phase filter.

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DECEMBER 2010

Paper Code: IT307 Subject: Digital Signal Processing
Paper ID: 15307

Time : 3 Hours Maximum Marks : 60

Note: Attempt any five questions.

Q1 Define (a) linearity (b) shift-invariance (c) causality and (d) stability of a discrete time system. Verify these conditions for the following systems:- **(12)**

(i) $T[x(n)] = \sum_{k=n_0}^{n} x(k)$

(ii) $T[x(n)] = \sum_{k=n-n_0}^{n+n_0} x(k)$

(iii) $T[x(n)] = x(n-n_0)$

(iv) $T[x(n)] = e^{x(n)}$

Q2 (a) Define z-transform. Determine the z-transform of the following sequences and give their region of convergence:- **(6)**

(i) $\left(\frac{1}{2}\right)^n u(n)$ (ii) $\left(\frac{1}{2}\right)^n (u(n) - u(n-10))$

(b) Discuss the z-transform theorems and properties. **(6)**

Q3 (a) Explain DFT. Give matrix relations for computing DFT and IDFT. **(6)**
(b) Define circular convolution. Evaluate circular convolution of the following sequences- x(n)={1 3 4 2 1} and h(n)={2 0 1 0 1}. **(6)**

Q4 (a) Explain the Overlap-Add method for evaluating convolution of infinite length sequence with finite length sequence. **(6)**
(b) Give the symmetry property of the DFT of a complex sequence and explain them. **(6)**

Q5 (a) Give and explain the network structures for IIR filters. **(6)**
(b) Discuss the polyphase realization of FIR filters. **(6)**

Q6 (a) How digital filter specifications are given? Explain with the help of magnitude response specifications. **(4)**
(b) Explain the process of IIR filter design using bilinear transformation. **(8)**

Q7 What are the approaches for decreasing the computational complexity of the DFT? Explain decimation-in-time FFT algorithm with the help of an example. **(12)**

Q8 (a) Compare IIR and FIR filters. **(4)**
(b) Describe procedure for designing FIR filters using windows. **(8)**

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(Please write your Exam Roll No.) **Exam Roll No. ………………**

END TERM EXAMINATION

SEVENTH SEMESTER [B.TECH./M.TECH.]- DECEMBER 2010

Paper Code: IT401 **Subject: Digital Signal Processing**
Paper ID: 15401

Time : 3 Hours **Maximum Marks : 60**

Note: Attempt five questions including Q.1 which is compulsory.

Q1 (a) Discuss minimum phase and maximum phase transfer function. **(5)**

(b) A linear time invariant system is characterized by system function $H(z)=\frac{3-4z^{-1}}{1-3.5z^{-1}+1.5z^{-2}}$. Determine h(n). **(5)**

(c) Check if the system is LTI or not (i) $y(n)=\sum_{K=-\infty}^{n}x(k)$ (ii) y(n)=x(-n). **(5)**

(d) Derive the relationship between DFT (i) Z-transform (ii) Fourier series. **(5)**

Q2 (a) Compute the convolution y(n)=x(n)*h(n), $x(n)=\begin{cases}1 & n=-2,0,1\\ 2 & n=-1\\ 0 & elsewhere\end{cases}$,

$h(n)=\delta(n)-\delta(n-1)+\delta(n-4)+\delta(n-5)$. **(8)**

(b) Determine whether the signals are energy or power signal and also compute its value (i) $x(n)=e^{2n}u(n)$ (ii) $x(n)=(1/3)^n u(n)$. **(2)**

Q3 (a) Check if It following signal are causal or not:- **(3)**

(i) $y(n)=x(n)+x^2(n-1)$

(ii) $y(n)=x(2n)$

(iii) $y(n)=\sum_{K=-\infty}^{n+1}x(k)$

(b) Discuss sampling theory in frequency domain. **(2)**

(c) Find the cross correlation of two finite length sequence x(n)={1,2,1,1} and y(n)={1,1,2,1}. Also, show that $r_{xy}(l)=x(l)*y(-l)$. **(5)**

Q4 (a) Given the sequence x_1(n)={1,2,3,4}, x_2(n)={1,1,2,2}. Compute- **(8)**

(i) $x_3(n)=x_1(n)\ \ x_2(n)$.

(ii) Linear convolution using circular convolution.

(b) Derive the parseval's theorem. **(2)**

Q5 (a) Determine the causal signal x(n) having Z-transform $X(Z)=\frac{1}{(1-2z^{-1})(1-z^{-1})^2}$. **(6)**

(b) Prove the following property of DFT where X(K) is the N point DFT of x(n). if x(n) is real and even then X(K) is real and even. **(4)**

Q6 Find the DFT of a sequence x(n)={1,2,3,4,4,3,2,1} using OIT algorithm. Draw the structure and also show bit reversal. **(10)**

Q7 Find the IDFT of the sequence X(K)={4, 1-j2.414, 0, 1-j0.414, 0, 1+j0.414, 0, 1+j2.414} using DIF algorithm. **(10)**

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Q8 Draw the direct form II, cascade and parallel structure for the system described by the difference equation $y(n)=\frac{3}{4}y(n-1)-\frac{1}{8}y(n-2)+x(n)+\frac{1}{3}x(n-1)$. **(10)**

Q9 (a) Explain the design of IIR filter using (i) impulse invariance method (ii) Bilinear transformation method. **(4)**

(b) Realize the following system function using minimum no. of multipliers:- **(6)**

(i) $H(z)=1+\frac{1}{3}z^{-1}+\frac{1}{4}z^{-2}+\frac{1}{4}z^{-3}+\frac{1}{3}z^{-4}+z^{-5}$

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

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END TERM EXAMINATION

FIFTH SEMESTER [B.TECH./M.TECH.] DEC. 2014-JAN. 2015

Paper Code: IT307 | **Subject: Digital Signal Processing**

Time : 3 Hours | **Maximum Marks :60**

Note: Attempt any five questions including Q.no.1 which is compulsory.

Q1 Explain the following briefly:- (2x10=20)

(a) Give the properties of Z-transformation.

(b) What is signal processing?

(c) Give some properties of DFT.

(d) Differentiate between FIR and IIR.

(e) Define Convolution.

(f) Why do we need FFT algorithms?

(g) What are the computational saving in using N point FFT algorithm?

(h) What are the advantages of FIR filters?

(i) Differentiate between DIT and DIF.

(j) Give some applications of DSP.

Q2 (a) What are typical signals? Give some examples of typical signal. (5)
(b) Explain the time-domain LTI system with an example. (5)

Q3 (a) Discuss the design procedure of FIR filter using frequency sampling method. (6)
(b) Give the block diagram representation of digital filter. (4)

Q4 (a) Derive the butterfly diagram of 8 point radix 2 DIF FFT algorithm and fully label it. (6)
(b) How can we classify signals? (4)

Q5 (a) Compute linear convolution of the two sequence x(n)={1,2,2,2} and h(n)={1,2,3,4}. (6)
(b) Derive expressions to relate z-transfer and DFT. (4)

Q6 (a) State and explain the scaling and time delay properties of z transform. (5)
(b) Describe different types of sampling methods. (5)

Q7 (a) Explain the classification of discrete signals. (4)
(b) Determine the response of LTI system when the input sequence is x(n)={-1,1,2,1,-1} using radix 2 DIF FFT. The impulse response is h(n)={-1,1,-1,1}. (6)

Q8 (a) Give some approaches of reducing the computation of an algorithm. (4)
(b) An 8 point sequence is given by x(n)={2,2,2,2,1,1,1,1}. Compute 8 point DFT of x(n) by radix DIT-FFT method. (6)

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(Please write your Exam Roll No.) Exam Roll No. ....

END TERM EXAMINATION

FIFTH SEMESTER [B.TECH] DECEMBER 2017

Paper Code: IT-307	***Subject: Digital Signal Processing***
Time: 3 Hours	***Maximum Marks: 75***

Note: Attempt any five questions including Q.no.1 which is compulsory. Assume missing data if any.

Q1 Attempt **any five**: **(5x5=25)**

(a) Find the Fourier Coefficient of the signal $f(t) = \sin\omega_0 t$.

(b) Verify following system for Linearity and Time Invariance:
(i)$y(t) = x^2(t)$, (ii) $y(t) = \sin t.x(t)$,(iii) $y(t) = x(at)$, and (iv) $y(t) = \log x(t)$.

(c) What is the difference between Causal System or Non-Causal System.

(d) Prove that discrete time harmonics are not always periodic in frequency.

(e) Find the Fourier Coefficient of the signal which is full wave rectifier signal.

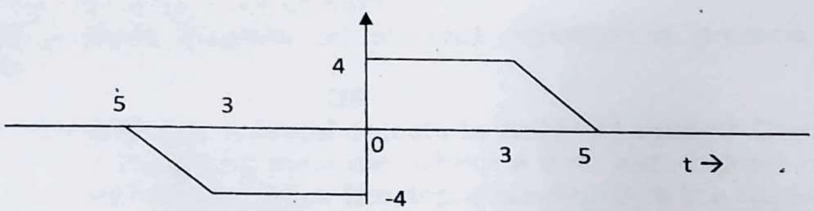
(f) Write a short note filter bank.

(g) Compare IIR and FIR.

(h) Explain the need of low pass filter with a decimator and mathematically prove that $\omega_x = \omega_y D$.

(i) Short note on Frequency Sampling realization of FIR filters.

Q2 (a) Signal f(t) is defined as below: **(6.5)**

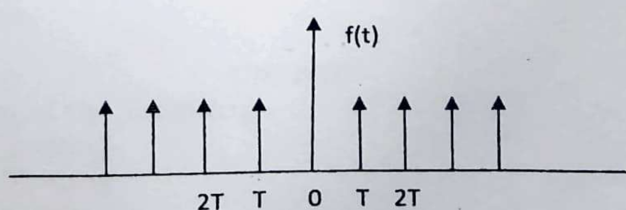


A signal g(t) is realized by multiplying f(t) with $\partial(t+4) + \partial(t-4)$ is the integral of the signal or power signal. Hence find the Energy or Power.

(b) Find the response of discrete time LTI system having the input and impulse responses as given below $f[n] = a^n u[n]$, $h[n] = a^n u[n]$. **(6)**

Q3 (a) Derive the relationship between Trigonometric Fourier Series and Exponential Fourier Series. **(6.5)**

(b) Draw the Complex Spectrum of the given below and also find the Fourier series. **(6)**



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Q4 (a) Find the Fourier Transform of the signal. (6.5)

(i) $f(t) = \frac{1}{\pi t}$, (ii) $f(t) = t\left(\frac{\sin t}{\pi t}\right)^2$

(b) Find the number of complex additions and complex multiplications required to find DFT for 16 point signal. Compare them with number of computations required, if FFT algorithm is used. (6)

Q5 (a) Compute DFT of a sequence, $x(n) = \{1,2,2,2,1,0,0,0\}$ using DIF-FFT algorithm. Sketch its magnitude spectrum. (6.5)

(b) Find 8-point FFT of, $x(n) = \{1,2,2,2,1\}$ using signal flow graph of Radix-2 Decimation in frequency FFT. (6)

Q6 Derive the Expression for impulse invariance technique for obtaining transfer function of digital filter from analog filter. Derive necessary equation for relationship between frequency of analog and digital filter. (12.5)

Q7 Compare various windows used for designing FIR filters. (12.5)

Q8 Compare various windows used for designing FIR filters. (12.5)

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