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Department of Computer Science and Engineering

IT6502 Digital Signal Processing Question Bank

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Unit – I Signals and Systems

Two Marks

1.	Define and express the transfer function of Nth order LTI system.	Nov/ Dec 2011
2.	Compare Linear Convolution and Circular convolution	Nov/ Dec 2011
3.	State low pass sampling theorem.	Nov/ Dec 2012
4.	What is meant by energy and power signal?	Nov/ Dec 2012
5.	State the convolution property of Z transforms.	Nov/ Dec 2013
6.	Define sampling theorem.	Nov/ Dec 2013
7.	A discrete time signal $x(n) = \{0, 0, 1, 1, 2, 0, 0, \ldots \ldots\}$. Sketch the $x(n)$ and $x(-n+2)$ signals.	Nov/ Dec 2014
8.	Determine whether the following sinusoids are periodic; if periodic then compute their fundamental period. (a) $\cos(0.01nn)$ (b) $\sin\left(\frac{62nn}{10}\right)$	Nov/ Dec 2014
9.	What do you understand by the term signal processing?	May/ Jun 2014
10.	What is time invariant system?	May/ Jun 2014
11.	State Sampling theorem.	Apr/ May 2015
12.	What is quantization error?	Apr/ May 2015
13.	A discrete time signal $x(n) = \{-2, -1, 0, \underset{\uparrow}{1}, -1, 1\}$ is multiplied by $u(-n+2)$. What is the resultant signal?	Nov/Dec 2015
14.	What is a shift invariant system? Give an example.	Nov/Dec 2015
15.	Define DT system.	Nov/Dec 2015
16.	How do you obtain a digital signal from DT signal?	Nov/Dec 2015
17.	Find the circular convolution of two sequences $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 1\}$.	Nov/Dec 2015

16 Marks

1.	(i)	Find the inverse Z transform of $X(z) = \frac{z^2+z}{(z-1)(z-3)}$, ROC: $\lvert z\rvert > 3$ using (1) Residue method and (2) Convolution method. (8 Marks)	Nov/ Dec 2011
	(ii)	State and prove circular convolution. (8 Marks)	Nov/ Dec 2011
2.		LTI System is described by the difference equation $y(n) = ay(n-1) + bx(n)$. Find the impulse response, magnitude function and phase function. Solve b, if $\lvert H(m)\rvert = 1$. Sketch the magnitude and phase response for $a = 0.6$. (16 Marks)	Nov/ Dec 2011
3.		A causal system is represented by the following difference equation $y(n) + \frac{1}{4}y(n-1) = x(n) + \frac{1}{2}x(n-1)$ Find the system transfer function H(z), unit sample response, magnitude and phase function of the system. (16 Marks)	Nov/ Dec 2012
4.		Determine the causal signal x(n) for the following Z transform:	
	(i)	$X(z) = \frac{z^2+z}{\left(z-\frac{1}{2}\right)^3\left(z-\frac{1}{4}\right)}$ (8 Marks)	Nov/ Dec 2012

	(ii)	$$X(z) = \frac{1 + z^{-1}}{1 - z^{-1} + 0.5z^{-2}}$$ (8 Marks)	Nov/ Dec 2012
5.	(i)	Compute the convolution of the signals $x(n) = \{1, 2, 3, 4, 5, 3, -1, -2\}$ and $h(n) = \{3, 2, 1, 4\}$ using tabulation method. (6 marks)	Nov/ Dec 2013
	(ii)	Check whether the following systems are static or dynamic, linear or non linear, time variant or time invariant, causal or non causal, stable or unstable. (a) $y(n) = \cos[x(n)]$ (b) $y(n) = x(-n + 2)$ (c) $y(n) = x(2n)$ (d) $y(n) = x(n)\cos(m_0 n)$ (10 Marks)	Nov/ Dec 2013
6.	(i)	Describe the different types of digital signal representation. (8 Marks)	Nov/ Dec 2013
	(ii)	What is Nyquist rate? Explain its significant while sampling the analog signal. (8 Marks)	Nov/ Dec 2013
7.	Check whether the systems described by the equations are (i) $y(n) = x(n)\cos m_0 n$ (ii) $y(n) = \lvert x(n)\rvert$ (iii) $y(n) = \mathrm{sgn}\ (x(n))$ Static or Dynamic Linear or Non Linear Shift invariant or variant Causal or non causal Stable or Unstable (16 Marks)		Nov/ Dec 2014
8.	Compute the linear convolution of the following sequence using mathematical equation, multiplication, and tabular methods. (16 Marks) $$x(n) = \{0, 2, 2, 3\} \text{ and } h(n) = \sin\left(\frac{3nn}{8}\right), 0 \le n \le 4$$		Nov/ Dec 2014
9.	Find the Z transform of the following discrete time signals and find ROC.		
	(i)	$$x(n) = \left(-\frac{1}{5}\right)^{n} u(n) + 5\left(\frac{1}{2}\right)^{-n} u(-n-1)$$ (8 Marks)	May/ Jun 2014
	(ii)	$$x(n) = u(n-2)$$ (8 Marks)	May/ Jun 2014
10.	Find whether the following systems are (a) Linear (b) Time Invariant		
	(i)	$$y(n) = e^{-s(n)}$$ (8 Marks)	May/ Jun 2014
	(ii)	$$y(n) = x(n)\cos mn$$ (8 Marks)	May/ Jun 2014
11.	(i)	Find the convolution of given signals. (8 Marks) $$x(n) = 3^{n}u(-n) \text{ and } h(n) = \left(\frac{1}{3}\right)^{n} u(n-2)$$	Apr/ May 2015
	(ii)	Applying concentric circle method, compute circular convolution of the sequences $h(n) = \{1, 2, 3, 4\}$ and $x(n) = \{1, 2, 3\}$. (8 Marks)	Apr/ May 2015

12.	Explain the process of analog to digital conversion of signals in terms of sampling, quantization and coding. (16 Marks)		Apr/ May 2015
13.	(i)	Determine whether each of the following systems below is (1) causal (2) Linear (3) Dynamic (4) Time Invariant (5) Stable. (8 Marks) (a) $y(n) = e^{-s(n)}$ (b) $y(n) = x(n)\,\Sigma_{k=-œ}^{œ}\,ð(n-2k)$	Nov/Dec 2015
	(ii)	Explain sampling theorem and reconstruction of the analog signal from its samples. (8 Marks)	Nov/Dec 2015
14.	(i)	Explain the properties of cross correlation and auto correlation sequences. (8 marks)	Nov/Dec 2015
	(ii)	Find the discrete convolution of the following sequences $u(n) * u(n-3)$	Nov/Dec 2015
15.	(i)	Check whether the following systems are linear: (4+4) (a) $y(n) = \frac{1}{N}\Sigma_{n=0}^{N-1}\, x(n-n)$ (b) $y(n) = [x(n)]^2$	Nov/Dec 2015
	(ii)	Find the impulse response of the causal system $y(n) - y(n-1) = x(n) + x(n-1)$. (8 Marks)	Nov/Dec 2015
Unit – II Frequency Transformations			
Two Marks			
1.	What is the relation between DFT and Z transform?		Nov/ Dec 2011
2.	What is Phase factor or Twiddle factor?		Nov/ Dec 2011
3.	What is twiddle factor?		Nov/ Dec 2012
4.	List the uses of FFT in linear filtering.		Nov/ Dec 2012
5.	Find the DTFT of $x(n) = -b^n u(-n-1)$.		Nov/ Dec 2013
6.	Compute the IDFT of $Y(k) = \{1, 0, 1, 0\}$		Nov/ Dec 2013
7.	Using the definition $W = e^{-j(2n/N)}$, and the Euler identity $e^{\pm i8} = \cos 8 \pm i \sin 8$, the value of $w^{(N/3)}$ is ______________________________		Nov/ Dec 2014
8.	In the direct computation of N point DFT of a sequence, how many multiplications and additions are required?		Nov/ Dec 2014
9.	List any two properties of DFT.		May/ Jun 2014
10.	What is meant by radix 2 FFT algorithm?		May/ Jun 2014
11.	Compute DFT of the signal $x(n) = ð(n)$		Apr/ May 2015
12.	What is meant by radix 2 FFT?		Apr/ May 2015
13.	State the advantages of FFT over DFTs.		Apr/ May 2011
14.	What is meant by bit reversal?		Apr/ May 2011
15.	State the difference between DFT and DTFT.		May/ Jun 2014
16.	What is bit reversal?		May/ Jun 2014
17.	Write the analysis and synthesis equations of DFT,		Apr/ May 2015
18.	Is the DFT of the finite length sequence is periodic? If so state the theorem?		Apr/ May 2015
19.	Obtain the circular convolution of the following sequences $x(n) = \{1, 2, 1\}$ and $h(n) = \{1, -2, 2\}$.		Nov/ Dec 2010
20.	How many multiplications and additions are required to compute N point DFT using radix 2 FFT?		Nov/ Dec 2010
21.	Distinguish between DFT and DTFT.		Nov/ Dec 2011
22.	What is zero padding? What are its uses?		Nov/ Dec 2011
23.	What is twiddle factor?		Nov/ Dec 2012

24.	How many stage of decimations are required in the case of a 64 point radix 2 DIT FFT algorithm?		Nov/ Dec 2012
25.	What is zero padding? What is the purpose of it?		Nov/ Dec 2013
26.	How many multiplications and additions are required to compute N point DFT using radix 2 FFT?		Nov/ Dec 2013
27.	Calculate the number of multiplications required to compute the DFT of a 64 point sequence using direct computation and that using FFT.		Nov/ Dec 2014
28.	What is meant by 'in place' in DIT and DIF algorithm?		Nov/ Dec 2014
29.	Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence.		Nov/Dec 2015
16 Marks			
1.	(i)	Evaluate the 8 point DFT for the following sequences using DIT FFT algorithm. $$x(n) = \begin{cases} 1, & -3 \leq n \leq 3 \\ 0, & otherwise \end{cases}$$ (8 Marks)	Nov/ Dec 2011
	(ii)	Calculate the percentage of saving in calculations in a 1024 point radix 2 FFT when compared to Direct DFT. (8 Marks)	Nov/ Dec 2011
2.	Determine the response of LTI system when the input sequence $x(n) = \{-1, 1, 2, 1, -1\}$ by radix 2 DIT FFT. The impulse response of the system is $h(n) = \{-1, 1, -1, 1\}$. (16 marks)		Nov/ Dec 2011
3.	(i)	The input x(n) and impulse response $h(n)$ of a system are given by $x(n) = \{\underset{\uparrow}{-1}, 1, 2, -2\}$ and $h(n) = \{0.5, \underset{\uparrow}{1}, -1, 2, 0.75\}$. Determine the response of the system using DFT. (10 Marks)	Nov/ Dec 2012
	(ii)	State and prove convolution property of DFT. (6 Marks)	Nov/ Dec 2012
4.	Compute the FFT of the sequence $x(n) = n^2 + 1$ for $0 \leq n \leq N-1$, where $N = 8$ using DIT algorithm.		Nov/ Dec 2012
5.	(i)	Discuss the properties of DFT. (8 Marks)	Nov/ Dec 2013
	(ii)	Discuss the uses of FFT algorithm in linear filtering and correlation. (8 Marks)	Nov/ Dec 2013
6.	Find DFT for $\{1, 1, 2, 0, 1, 2, 0, 1\}$ using FFT DIT butterfly algorithm and plot the spectrum.		Nov/ Dec 2013
7.	(i)	State and prove the periodicity and time reversal properties of DFT. (8 Marks)	Nov/ Dec 2014
	(ii)	Obtain the 4 point DFT of the following sequences (a) $x(n) = 2^n$ (b) $x(n) = \{0, 1, 0, -1\}$	Nov/ Dec 2014
8.	Compute the 8 point DFT of the equation $x(n) = n + 1$ using radix 2 DIF FFT algorithm. (16 Marks)		Nov/ Dec 2014
9.	Find 8 point DFT of the sequence using radix 2 DIT algorithm. (16 marks) $$x(n) = \{1, -1, 1, -1, 0, 0, 0, 0\}$$		May/ Jun 2014
10.	Using radix 2 DIT FFT algorithm, determine DFT of the given sequence for N = 8. (16 Marks) $$x(n) = n, 0 \leq n \leq 7$$		May/ Jun 2014
11.	Find the 8 point DFT of a sequence using radix 2 DIT algorithm. (16 Marks) $$x(n) = 1, 0 \leq n \leq 2$$		Apr/ May 2015
12.	Compute 8 point DFT of the sequence $x(n) = \{1, -1, 1, -1, 0, 0, 0, 0\}$ using radix 2 DIF algorithm. (16 Marks)		Apr/ May 2015
13.	With appropriate diagrams describe		

	(i)	Overlap save method (8 Marks)	Apr/ May 2011
	(ii)	Overlap add method (8 Marks)	Apr/ May 2011
14.	Explain radix 2 DIF FFT algorithm. Compare it with DIT FFT algorithms. (16 Marks)		Apr/ May 2011
15.	Explain in detail about overlap add method and overlap save method for filtering long data sequences using DFT. (16 Marks)		May/ Jun 2014
16.	Develop a 8 point DIT FFT algorithm. Draw the signal flow graph. Determine the DFT of the following sequence $x(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$ using the signal flow graph. Show all the intermediate results on the signal flow graph. (16 Marks)		May/ Jun 2014
17.	(i)	Determine the 8 point DFT of the sequence $x(n) = \{0, 0, 1, 1, 1, 0, 0, 0\}$. (8 Marks)	Apr/ May 2015
	(ii)	What is decimation in frequency algorithm? Write the similarities and differences between DIT and DIF algorithms. (8 Marks)	Apr/ May 2015
18.	With appropriate diagrams, discuss how overlap save method and overlap add methods are used. (16 Marks)		Apr/ May 2015
19.	(i)	Compute the 8 point DFT of the sequence $$x(n) = \{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\}$$ using the radix 2 decimation in time algorithm. (10 Marks)	Nov/ Dec 2010
	(ii)	Explain the overlap add method for linear FIR filtering of a long data sequence. (6 marks)	Nov/ Dec 2010
20.	(i)	Compute the 8 point DFT of the sequence $$x(n) = \begin{cases} 1, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$$ Bu using the decimation in frequency FFT algorithm. (10 Marks)	Nov/ Dec 2010
	(ii)	Summarize the properties of DFT. (6 Marks)	Nov/ Dec 2010
21.	(i)	Determine the N point DFT of the following sequences. (6 Marks) (a) $x(n) = ð(n)$ (b) $x(n) = ð(n - n_0)$	Nov/ Dec 2011
	(ii)	Compute 8 point DFT of the sequence $x(n) = \{0, 1, 2, 3, 4, 5, 6, 7\}$ using radix 2 DIF algorithm. (10 Marks)	Nov/ Dec 2011
22.	Compute the linear convolution of finite duration sequences $h(n) = \{1, 2\}$ and $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$ by overlap add method. (16 Marks)		Nov/ Dec 2011
23.	(i)	Differentiate DFT from DTFT. (4 Marks)	Nov/ Dec 2012
	(ii)	Compute an 8 point DFT of the sequence $x(n) = \{1, 0, \underset{\uparrow}{1}, -1, 1, -1, 0, 1\}$. (12 Marks)	Nov/ Dec 2012
24.	(i)	Prove that FFT algorithms help in reducing the number of computations involved in DFT computation. (6 Marks)	Nov/ Dec 2012
	(ii)	Compute a 8 point DFT of the sequence using DIT FFT algorithm. (10 Marks) $$x(n) = \{1, 2, \underset{\uparrow}{3}, 2, 1, 0\}$$	Nov/ Dec 2012
25.	(i)	Compute the DFT of the sequence whose values for one period is given by $x(n) = \{1, 1, 1 - 2, -2\}$. (8 Marks)	Nov/ Dec 2013
	(ii)	Compute the 8 point DFT of the sequence $x(n) = \begin{cases} 1, & 0 \le n \le 7 \\ 0, & otherwise \end{cases}$ by using DIT and DIF algorithm. (8 Marks)	Nov/ Dec 2013
26.	(i)	Summarize the difference between overlap save and overlap add method. (8 Marks)	Nov/ Dec 2013

	(ii)	Evaluate the 8 point DFT of the following sequence using DIT FFT algorithm. $x(n) = \begin{cases} 1, & -3 \le n \le 3 \\ 0, & otherwise \end{cases}$ (8 Marks)	Nov/ Dec 2013
27.	(i)	Find the DFT of a sequence $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT algorithm. (10 Marks)	Nov/ Dec 2014
	(ii)	State any 6 properties of DFT. (6 Marks)	Nov/ Dec 2014
28.	(i)	Using linear convolution find $y(n) = x(n) * h(n)$ for the sequences $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 1, 2, -1\}$ and $h(n) = \{1, 2\}$. Compare the result by solving the problem using overlap add method and overlap save method. (12 Marks)	Nov/ Dec 2014
	(ii)	Find the IDFT of the sequence $X(k) = \{6, -2 + j2, -2, -2 - j2\}$ using DIF algorithm. (4 Marks)	Nov/ Dec 2014
29.	(i)	Explain any four properties of DFT. (8 Marks)	Nov/Dec 2015
	(ii)	Calculate the 8 point DFT of the sequence $x(n) = \{0.5, 0.5, 0.5, 0.5, 1, 2, -1, 0\}$ using the in place radix 2 DIT algorithm. (8 Marks)	Nov/Dec 2015
30.	(i)	Draw the flow graph of the two point radix 2 DIF FFT algorithm. What is the basic operation of DIF algorithm? (8 Marks)	Nov/Dec 2015
	(ii)	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. (8 Marks)	Nov/Dec 2015
31.	(i)	Illustrate the construction of an 8 point DFT from two 4 point DFTs. (8 Marks)	Nov/Dec 2015
	(ii)	Illustrate the reduction of an 8 point DFT to two 4 point DFTs by decimation in frequency. (8 Marks)	Nov/Dec 2015
Unit – III IIR Filter Design			
Two Marks			
1.	Sketch the various tolerance limits to approximate an ideal low pass and high pass filter.		Nov/ Dec 2011
2.	What is the importance of poles in filter design?		Nov/ Dec 2011
3.	What is frequency warping?		Nov/ Dec 2011
4.	What is Butterworth approximations?		Nov/ Dec 2011
5.	Compare bilinear and impulse invariant transformation.		Nov/ Dec 2012
6.	What is aliasing?		Nov/ Dec 2012
7.	Define bilinear transformation with expressions.		Nov/ Dec 2013
8.	Mention the properties of Butterworth filter.		Nov/ Dec 2013
9.	Compare analog and digital filters.		Nov/ Dec 2014
10.	Sketch the mapping of S plane and Z plane in approximation of derivatives.		Nov/ Dec 2014
11.	What are the properties of impulse invariant transformation?		May/ Jun 2014
12.	Draw the direct form structure of IIR filter.		May/ Jun 2014
13.	What is meant by bilinear transformation method of designing IIR filter?		Apr/ May 2015
14.	Draw the direct form realization of IIR system.		Apr/ May 2015
15.	Why do we go for analog approximation to design a digital filter?		Apr/ May 2011
16.	Give any two properties of Chebyshev filters.		Apr/ May 2011
17.	Sketch the frequency response of an odd and an even order chebyshev low pass filters.		May/ Jun 2014
18.	What is bilinear transformation? What is the main advantages and disadvantages of this technique?		May/ Jun 2014
19.	Find H(z) for the IIR filter whose $H(s) = \frac{1}{c+6}$ with $T = 0.1$ sec.		Apr/ May 2015

20.	Draw the response curve for Butterworth, Chebyshev and Elliptic Filter.	Apr/ May 2015
21.	What is prewarping?	Nov/ Dec 2010
22.	What is the advantage of direct form II realization when compared to direct form I realization?	Nov/ Dec 2010
23.	List the properties of Chebyshev filter.	Nov/ Dec 2011
24.	Draw the direct form structure of IIR filter.	Nov/ Dec 2011
25.	What are the properties of IIR filter?	Nov/ Dec 2011
26.	Why is the Butterworth response called a maximally flat response?	Nov/ Dec 2012
27.	What is frequency warping?	Nov/ Dec 2012
28.	Give the steps in the design of a digital filter from analog filters.	Nov/ Dec 2013
29.	What are the disadvantages of direct form realization?	Nov/ Dec 2013
30.	Distinguish between Butterworth and Chebyshev Filter.	Nov/ Dec 2014
31.	What is prewarping?	Nov/ Dec 2014
32.	Compare digital and analog filters.	Nov/Dec 2015
33.	What is meant by impulse invariant method of designing IIR filters?	Nov/Dec 2015
34.	Define pass band.	Nov/Dec 2015
35.	Use the backward difference for the derivative to convert analog LPF with system function $H(s) = \frac{1}{c+2}$	Nov/Dec 2015
16 marks		
1.	The specification of the desired Low pass filter is $$\frac{1}{\sqrt{2}} \le \lvert H(m)\rvert \le 1, 0 \le m \le 0.2n$$ $$\lvert H(m)\rvert \le 0.08, 0.4n \le m \le n$$ Design a Butterworth digital filter using bilinear transformation. (16 marks)	Nov/ Dec 2011
2.	The specification of the desired Low pass filter is $$0.9 \le \lvert H(m)\rvert \le 1, 0 \le m \le 0.25n$$ $$\lvert H(m)\rvert \le 0.24, 0.5n \le m \le n$$ Design a Chebyshev digital filter using impulse invariant transformation. (16 Marks)	Nov/ Dec 2011
3.	Design a Butterworth digital filter using bilinear transformation that satisfy the following specifications $$0.89 \le \lvert H(m)\rvert \le 1, 0 \le m \le 0.2n$$ $$\lvert H(m)\rvert \le 0.18, 0.3n \le m \le n$$	Nov/ Dec 2012
4.	The specification of the desired Low pass filter is $$0.9 \le \lvert H(m)\rvert \le 1, 0 \le m \le 0.25n$$ $$\lvert H(m)\rvert \le 0.24, 0.5n \le m \le n$$ Design a Chebyshev digital filter using impulse invariant transformation. (16 Marks)	Nov/ Dec 2012
5.	The specification of the desired Low pass filter is $$0.8 \le \lvert H(m)\rvert \le 1, 0 \le m \le 0.2n$$ $$\lvert H(m)\rvert \le 0.2, 0.32n \le m \le n$$ Design Butterworth digital filter using impulse invariant transformation. (16 marks)	Nov/ Dec 2013
6.	(i) Discuss the limitations in designing IIR filter using impulse invariant method. (6 Marks)	Nov/ Dec 2013
	(ii) Convert the analog filter with the system transfer function $$H_a(s) = \frac{s + 0.3}{(s + 0.3)^2 + 16}$$ using bilinear transformation.	Nov/ Dec 2013

7.	Determine the system function of the IIR digital filter for the analog transfer function $$H_a(s) = \frac{10}{s^2 + 7s + 10} \text{ with } T = 0.2 \text{ seconds}$$ using impulse invariance method. (16 Marks)		Nov/ Dec 2014
8.	A digital filter with a 3 dB bandwidth of $0.25n$ is to be designed from the analog filter whose system response is $$H_a(s) = \frac{\Omega_c}{s + \Omega_c}$$ using bilinear transformation and obtain H(z).		Nov/ Dec 2014
9.	(i)	Realize the following FIR system with difference equation $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$ in direct form I. (6 Marks)	May/ Jun 2014
	(ii)	Analyze briefly the different structures of IIR filters. (10 marks)	May/ Jun 2014
10.	Design a digital chebyshev filter using bilinear transformation satisfying the following constraints. Assume $T = 1$ seconds. $$0.75 \le \lvert H(m) \rvert \le 1, 0 \le m \le n/2$$ $$\lvert H(m) \rvert \le 0.2, 3n/4 \le m \le n$$		May/ Jun 2014
11.	Design a digital Butterworth filter satisfying the constraints $$0.707 \le \lvert H(m) \rvert \le 1, 0 \le m \le n/2$$ $$\lvert H(m) \rvert \le 0.2, 3n/4 \le m \le n$$ with $T = 1$ second using bilinear transformation method. (16 Marks)		Apr/ May 2015
12.	Obtain the direct form I, direct form II and cascade form realization of the following system functions. (16 Marks) $$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$		Apr/ May 2015
13.	Explain in detail Butterworth filter approximation. (16 Marks)		Apr/ May 2011
14.	Explain the bilinear transform method of IIR filter design. What is warping effect? Explain the poles and zeros mapping procedure clearly. (16 Marks)		Apr/ May 2011
15.	Design a low pass Butterworth digital filter with the following specifications: $m_c = 4000$; $m_p = 3000$; $A_p = 3dB$; $A_c = 20dB$; $T = 0.0001$ sec. (16 Marks)		May/ Jun 2014
16.	A system is represented by a transfer function H(z) is given by $$H(z) = 3 + \frac{4z}{z - \frac{1}{2}} - \frac{z}{z - \frac{1}{4}}$$		
	(i)	Does this H(z) represent a FIR or IIR filter why? (4 Marks)	May/ Jun 2014
	(ii)	Give the difference equation realization of this system using direct form I. (6 Marks)	May/ Jun 2014
	(iii)	Draw the block diagram for the direct form II canonic realization, and give the governing equations for implementations. (6 Marks)	May/ Jun 2014
17.	Explain in detail the steps involved in the design of IIR filter using bilinear transformation. (16 marks)		Apr/ May 2015
18.	(i)	Convert the analog filter with system function $$H_a(s) = \frac{s + 0.1}{(s + 0.1)^2 + 16}$$ into a digital IIR filter by means of the bilinear transformation. The digital filter is to have a resonant frequency of $m_r = \frac{n}{2}$. (10 marks)	Apr/ May 2015
	(ii)	Draw the structure for the IIR filter in direct form II for the following transfer	Apr/ May 2015

		function. (6 marks) $$H(z) = \frac{(2+3z^{-1})(4+2z^{-1}+3z^{-2})}{(1+0.6z^{-1})(1+z^{-1}+0.5z^{-2})}$$	
19.		Determine the system function $H(z)$ of a chebyshev's low pass digital filter with the specifications $$\alpha_p = 1 \text{ dB ripple in the passband } 0 \le m \le 0.2n$$ $$\alpha_c = 15 \text{ dB ripple in the stopband } 0.3n \le m \le n$$ Using bilinear transformation (assume T = 1 sec). (16 marks)	Nov/ Dec 2010
20.		Obtain the direct form I, direct form II, cascade and parallel form realization for the system. (16 Marks) $$y(n) = -0.1\, y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$	Nov/ Dec 2010
21.		Design a digital Butterworth filter using impulse invariance method satisfying the constraints. Assume T = 1 sec. (16 Mark) $$0.8 \le \lvert H(m)\rvert \le 1,\ 0 \le m \le 0.2n$$ $$\lvert H(m)\rvert \le 0.2,\ 0.6n \le m \le n$$	Nov/ Dec 2011
22.		Obtain the direct form I, direct form II and cascade form realization of the following system functions. (16 Marks) $$y(n) = 0.1\, y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$	Nov/ Dec 2011
23.	(i)	Explain the procedure for designing analog filters using the chebyshev approximations. (6 Marks)	Nov/ Dec 2012
	(ii)	Convert the following analog transfer function into digital using impulse invariant mapping with T = 1 sec. (10 Marks)	Nov/ Dec 2012
24.	(i)	Design a digital second order low pass Butterworth filter with cut off frequency 2200 Hz using Bilinear transformation. Sampling rate is 8000 Hz. (8 Marks)	Nov/ Dec 2012
	(ii)	Determine the cascade form and parallel form implementation of the system governed by the transfer function. (8 Marks) $$H(z) = \frac{(1+z^{-1})(1-5z^{-1}-z^{-2})}{(1+2z^{-1}+z^{-2})(1+z^{-1}+z^{-2})}$$	Nov/ Dec 2012
25.		Discuss the steps in the design of IIR filter using Bilinear Transformation for any one type of filter. (16 Marks)	Nov/ Dec 2013
26.		Convert the following pole zero IIR filter into a lattice ladder structure. (16 Marks) $$H(z) = \frac{1+2z^{-1}+2z^{-2}+z^{-3}}{1+\frac{13}{24}z^{-1}+\frac{5}{8}z^{-2}+\frac{1}{3}z^{-3}}$$	Nov/ Dec 2013
27.		Design a digital chebyshev filter to satisfy the constraints $$0.707 \le \lvert H(m)\rvert \le 1,\ 0 \le m \le 0.2n$$ $$\lvert H(m)\rvert \le 0.1,\ 0.5n \le m \le n$$ using bilinear transformation and assuming T = 1 sec. (16 Marks)	Nov/ Dec 2014
28.	(i)	For the analog transfer function $$H(s) = \frac{2}{(s+1)(s+2)}$$ Determine $H(z)$ using impulse invariant method. Assume T = 1 sec. (10 Marks)	Nov/ Dec 2014
	(ii)	Obtain the cascade and parallel realizations for the system function given by	Nov/ Dec 2014

		$$H(z) = \frac{1+\frac{1}{4}z^{-1}}{(1+\frac{1}{2}z^{-1})\,(1+\frac{1}{2}z^{-1}+\frac{1}{4}z^{-2})}$$ (6 Marks)	
29.	(i)	Design an analog Butterworth filter that has a -2 dB pass band attenuation at a frequency of 20 rad/sec and at least -10 dB stop band attenuation at 30 rad/sec. (10 Marks)	Nov/Dec 2015
	(ii)	Explain the steps of design of digital filters from analog filters. (6 Marks)	Nov/Dec 2015
30.	(i)	Using the bilinear transform, design a high pass filter, monotonic in pass band with cut off frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. (10 Marks)	Nov/Dec 2015
	(ii)	Explain the methods of realization of digital filters. (6 marks)	Nov/Dec 2015
31.	(i)	An analog filter has the following system function. Convert this filter into a digital filter using backward difference for the derivative. (8 Marks) $$H(s) = \frac{1}{(s+0.1)^2+9}$$	Nov/Dec 2015
	(ii)	Convert the analog filter into a digital filter whose system function is $$H(s) = \frac{s+0.2}{(s+0.2)^2+9}$$ Use Impulse Invariance technique. Assume T = 1 sec. (8 Marks)	Nov/Dec 2015
32.	(i)	Convert the analog filter with system function $$H(s) = \frac{s+0.1}{(s+0.1)^2+9}$$ into a digital IIR filter using bilinear transformation. The digital filter should have a resonant frequency of $m_r = n/4$.	Nov/Dec 2015
	(ii)	A digital filter with a 3 dB bandwidth of $0.25n$ is to be designed from the analog filter whose system response is $H(s) = \frac{\Omega_c}{c+\Omega_c}$. Use bilinear transformation and obtain $H(z)$.	Nov/Dec 2015
Unit – IV FIR Filter Design			
Two Marks			
1.	What is Gibb's phenomenon?		Nov/ Dec 2012
2.	What are limit cycles?		Nov/ Dec 2012
3.	What are Gibb's oscillations?		Nov/ Dec 2013
4.	Distinguish between FIR and IIR filter.		Nov/ Dec 2013
5.	What are the characteristic features of FIR filter?		Nov/ Dec 2014
6.	What do you understand by linear phase response in filters?		May/ Jun 2014
7.	What is the reason that FIR filter is always stable?		May/ Jun 2014
8.	What is linear phase response of a filter		Apr/ May 2015
9.	State any two properties of FIR filter.		Apr/ May 2015
10.	State the properties of FIR filters.		Apr/ May 2011
11.	What is meant by Gibb's Phenomenon?		Apr/ May 2011
12.	State the effect of having abrupt discontinuity in frequency response of FIR filters.		May/ Jun 2014
13.	State Gibb's phenomenon.		May/ Jun 2014
14.	Why is window function used in FIR filter design?		Apr/ May 2015
15.	Draw a causal FIR filter structure for length $n = 5$.		Apr/ May 2015

16.	Give the equations for Hamming window and Blackman window.		Nov/ Dec 2010
17.	Determine the transversal structure of the system function $$H(z) = 1 + 2z^{-1} - 3z^{-2} - 4z^{-3}$$		Nov/ Dec 2010
18.	What are the desirable characteristics of window?		Nov/ Dec 2011
19.	What are the features of FIR filter design using Kaiser's approach?		Nov/ Dec 2012
20.	Draw the direct form implementations of the FIR system having difference equation $$y(n) = x(n) - 2x(n-1) + 3x(n-2) - 10x(n-6)$$		Nov/ Dec 2012
21.	State the properties of FIR filter.		Nov/ Dec 2013
22.	Give the desirable characteristics of window.		Nov/ Dec 2013
23.	Give the equation specifying Hamming and Blackman window.		Nov/ Dec 2014
24.	Realize the following causal linear phase FIR system function $$H(z) = \frac{2}{3} + z^{-1} + \frac{2}{3} z^{-2}$$		Nov/ Dec 2014
25.	What conditions on the FIR sequence $h(n)$ are to be imposed in order that this filter can be called a linear phase filter?		Nov/Dec 2015
26.	List the disadvantages of FIR filter.		Nov/Dec 2015
27.	List the desirable window characteristics.		Nov/Dec 2015
16 Marks			
1.	(i)	Design a single tier notch filter to reject frequencies in the range of 1 to 2 rad/sec using rectangular window with $N = 7$. (8 Marks)	Nov/ Dec 2011
	(ii)	Compare Hamming window and Kaiser window. (8 Marks)	Nov/ Dec 2011
2.	Design a FIR Band Stop Filter to reject frequencies in the range 1.2 to 1.8 rad/sec using hamming window with length $N = 6$. Also realize the linear phase structure of the band stop FIR filter. (16 Marks)		Nov/ Dec 2012
3.	Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N-1-n)$. Also discuss the symmetric and anti symmetric cases of FIR filter when N is even. (16 Marks)		Nov/ Dec 2013
4.	Design the symmetric FIR low pass filter whose desired frequency response is given as $$H_a(m) = \begin{cases} e^{-jmt}, & \text{for } \lvert m\rvert \le m_c \\ 0, & otherwise \end{cases}$$ The length of the filter should be 5 and $m_c = 1$ radians/sanple using rectangular window. (16 Marks)		Nov/ Dec 2014
5.	Realize the direct form and linear phase FIR filter structures with the following impulse response. Which is the best realization? Why? (16 Marks) $$h(n) = ð(n) + \frac{1}{3}ð(n-1) - \frac{1}{4}ð(n-2) + \frac{1}{3}ð(n-3) + ð(n-4)$$		Nov/ Dec 2014
6.	Design an ideal Band Reject filter using Hamming window for the given frequency response. Assume $N = 11$. (16 Marks) $$H_d(e^{jm}) = \begin{cases} 1, & \lvert m\rvert \le n/3;\ \lvert m\rvert \ge 2n/3 \\ 0, & otherwise \end{cases}$$		May/ Jun 2014
7.	Design an FIR filter for the ideal frequency response using Hamming window with $N = 7$. (16 Marks) $$H_d(e^{jm}) = \begin{cases} e^{-j3m}, & -n/8 \le m \le n/8 \\ 0, & n/8 \le \lvert m\rvert \le n \end{cases}$$		May/ Jun 2014
8.	Discuss the design procedure of FIR filter using frequency sampling method. (16 Marks)		Apr/ May 2015

9.	Design an ideal differentiator with frequency response $H(e^{jm}) = jm; -n \le m \le n$ using Hamming window with N = 7. (16 Marks)		Apr/ May 2015
10.	Explain the system function $$H(z) = \left(\frac{2}{3}\right) z + 1 + \left(\frac{2}{3}\right) z^{-1}$$ by linear phase FIR structure. (16 Marks)		Apr/ May 2011
11.	Explain the designing of FIR filter using windows. (16 Marks)		Apr/ May 2011
12.	Explain the designing of FIR filter using frequency sampling method. (16 Marks)		May/ Jun 2014
13.	(i)	State and explain the properties of FIR filters. State their importance. (8 Marks)	May/ Jun 2014
	(ii)	Explain linear phase FIR structures. What are the advantages of such structures? (8 Marks)	May/ Jun 2014
14.	Explain the principle and procedure for designing FIR filter using rectangular window. (16 marks)		Apr/ May 2015
15.	Design a FIR linear phase digital filter for the following response. (16 marks) $$H(m) = \begin{cases} 1, & \lvert m\rvert \le n/6 \\ 0, & n/6 < \lvert m\rvert \le n \end{cases}$$		Apr/ May 2015
16.	Design an ideal high pass filter with a frequency response $$H_d(e^{jm}) = \begin{cases} 1, & \frac{n}{4} \le \lvert m\rvert \le n \\ 0, & \lvert m\rvert \le \frac{n}{4} \end{cases}$$ Find the value of $h(n)$ for N = 11 using Hamming Window. Find H(z) and determine the magnitude response. (16 Marks)		Nov/ Dec 2010
17.	(i)	Determine the coefficient $h(n)$ of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the condition (10 Marks) $$H_r\left(\frac{2nk}{15}\right) = \begin{cases} 1, & k = 0,1,2,3 \\ 0, & k = 4,5,6,7 \end{cases}$$	Nov/ Dec 2010
	(ii)	Obtain the linear phase realization of the system function (6 Marks) $$H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$$	Nov/ Dec 2010
18.	Design an ideal differentiator with frequency response $H(e^{jm}) = jm, -n \le m \le n$ using Hamming window with N = 8. (16 Marks)		Nov/ Dec 2011
19.	Design an ideal high pass filter using Hanning window with a frequency response $$H_d(e^{jm}) = \begin{cases} 1, & \frac{n}{4} \le \lvert m\rvert \le n \\ 0, & \lvert m\rvert \le \frac{n}{4} \end{cases}$$ Assume N = 11. (16 Marks)		Nov/ Dec 2011
20.	Design an FIR digital low pass filter by using the frequency sampling method for the following specifications. Cut off frequency = 1500 Hz Sampling frequency = 15000 Hz Order of the filter = N = 10 Filter length required = L = N+1 = 11. (16 Marks)		Nov/ Dec 2012
21.	(i)	Explain with neat sketches the implementation of FIR filters in the (a) Direct form	Nov/ Dec 2012

		(b) Lattice form (6 Marks)	
	(ii)	Design a digital FIR band pass filter with lower cut off frequency 2000 Hz and upper cut off frequency 3200 Hz using Hamming window of length N = 7. Sampling rate is 10000 Hz. (10 Marks)	Nov/ Dec 2012
22.	(i)	Explain briefly how the zeros in FIR filter is located. (7 Marks)	Nov/ Dec 2013
	(ii)	Using the rectangular window technique, design a low pass filter with pass band gain of unity, cut off frequency of 1000 Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. (9 Marks)	Nov/ Dec 2013
23.	Consider an FIR lattice filter with coefficients $k_1 = 1/2; k_2 = 1/3; k_3 = 1/4$. Determine the FIR filter coefficient for the direct form structures. (16 Marks)		Nov/ Dec 2013
24.	(i)	A low pass filter has a desired response as given below $$H_d(e^{jm}) = \begin{cases} e^{-j3m}, & 0 \leq m \leq \frac{n}{2} \\ 0, & \frac{n}{2} \leq m \leq n \end{cases}$$ Determine the filter coefficient $h(n)$ for M = 7 using type 1 frequency sampling technique. (10 Marks)	Nov/ Dec 2014
	(ii)	What is a linear phase filter? What are the condition to be satisfied by the impulse response of an FIR system in order to have a linear phase? (6 marks)	Nov/ Dec 2014
25.	Design a band pass filter which approximates the ideal filter with cut off frequencies at 0.2 rad/sec and 0.3 rad/sec. The filter order is M = 7. Use the Hanning window function. (16 Marks)		Nov/ Dec 2014
26.	(i)	Using a rectangular window technique design a low pass filter with pass band gain of unity, cut off frequency of 1000 Hz, and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. (10 Marks)	Nov/Dec 2015
	(ii)	Compare FIR and IIR filters. (6 Marks)	Nov/Dec 2015
27.	(i)	Obtain the cascade realization of the system function $H(z) = (1 + 2z^{-1} - z^{-2})(1 + z^{-1} - z^{-2})$. (10 Marks)	Nov/Dec 2015
	(ii)	Explain the quantization errors due to finite word length registers in digital filters. (6 marks)	Nov/Dec 2015
28.	(i)	List the steps involved by the general process of designing a digital filter. (8 Marks)	Nov/Dec 2015
	(ii)	List the advantages of FIR filters. (8 Marks)	Nov/Dec 2015
29.	(i)	The transfer function $H(z) = \sum_{n=0}^{M-1} h(n)z^{-n}$ characterizes a FIR filter (M=11). Find the magnitude response. (8 Marks)	Nov/Dec 2015
	(ii)	Use Fourier Series method to design a digital low pass filter to approximate the ideal specification given by $$H(e^{jm}) = \begin{cases} 1, & \lvert f \rvert \leq f_p \\ 0, & f_p < \lvert f \rvert \leq F/2 \end{cases}$$ Where f_p = pass band frequency F = sampling frequency (8 Marks)	Nov/Dec 2015
Unit – V Finite Word Length Effects in Digital Filters			
Two Marks			
1.	Define finite word length effects.		Nov/ Dec 2014
2.	What is meant by fixed point arithmetic? Give example.		Apr/ May 2011
3.	Explain the meaning of limit cycle oscillator?		Apr/ May 2011

4.	State the need for scaling in filter implementations.	May/ Jun 2014
5.	What is product round off noise?	May/ Jun 2014
6.	Explain briefly quantization noise.	Apr/ May 2015
7.	List the types of limit cycle oscillation.	Apr/ May 2015
8.	What is truncation?	Nov/ Dec 2010
9.	What is product quantization error?	Nov/ Dec 2010
10.	What is overflow oscillations?	Nov/ Dec 2011
11.	What are advantages of floating point arithmetic?	Nov/ Dec 2011
12.	What are limit cycle oscillations?	Nov/ Dec 2012
13.	What is dead band of the filter?	Nov/ Dec 2012
14.	What do you understand by input quantization error?	Nov/ Dec 2013
15.	State the methods used to prevent overflow.	Nov/ Dec 2013
16.	What is scaling?	Nov/ Dec 2014
17.	What is dead band of a filter?	Nov/ Dec 2014
18.	What is the effect of quantization on pole locations?	Nov/Dec 2015
19.	What does the truncation of data result in?	Nov/Dec 2015
20.	List the representations for which the truncation error is analyzed.	Nov/Dec 2015
16 Marks		
1.	(i) Explain the characteristics of limit cycle oscillation with respect to the system described by the equation $y(n) = 0.95\,y(n-1) + x(n)$. Determine the dead band of the system. (8 Marks)	Nov/ Dec 2011
	(ii) Explain Gibb's Phenomenon or Gibb's Oscillation. (8 marks)	Nov/ Dec 2011
2.	Explain the characteristics of limit cycle oscillation with respect to the system described by the equation $y(n) = 0.85y(n-2) + 0.72y(n-1) + x(n)$. Determine the dead band of the filter. $x(n) = (\frac{3}{4})$ ð(n).	Nov/ Dec 2012
3.	Explain in detail about the finite word length effects in digital filters. (16 Marks)	Nov/ Dec 2013
4.	Explain the quantization process and errors introduced due to quantization. (16 Marks)	Apr/ May 2011
5.	(i) Explain how reduction of product round-off error is achieved in digital filters. (8 Marks)	Apr/ May 2011
	(ii) Explain the effects of coefficient quantization in FIR filters. (8 Marks)	Apr/ May 2011
6.	(i) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation: $y(n) = 0.95y(n-1) + x(n)$; $x(n) = 0$; and $y(-1) = 13$. Determine the dead band range of the system. (10 Marks)	May/ Jun 2014
	(ii) Explain the effects of coefficient quantization in FIR filters. (6 Marks)	May/ Jun 2014
7.	(i) Derive the signal to quantization noise ratio of A/D converter. (6 Marks)	May/ Jun 2014
	(ii) Compare the truncation and rounding errors using fixed and floating point representation. (10 Marks)	May/ Jun 2014
8.	(i) Explain the various formats of the fixed point representation of binary numbers. (8 Marks)	Apr/ May 2015
	(ii) What is meant by finite word length effects on digital filters? List them. (8 Marks)	Apr/ May 2015
9.	Find the output round off noise power for the system having transfer function $H(z) = \frac{1}{(1-0.5z^{-1})(1+0.4z^{-1})}$	Apr/ May 2015

	which is realized in cascade form. Assume word length is 4 bits.		
10.	Discuss in detail the errors resulting from rounding and truncation. (16 marks)		Nov/ Dec 2010
11.	Explain the limit cycle oscillations due to product round off and overflow errors. (16 marks)		Nov/ Dec 2010
12.	With respect to finite word length effects in digital filters, with examples discuss about		
	(i)	Overflow limit cycle oscillation (10 Marks)	Nov/ Dec 2011
	(ii)	Signal Scaling (6 Marks)	Nov/ Dec 2011
13.	(i)	Distinguish between fixed point and floating point arithmetic. (6 Marks)	Nov/ Dec 2011
	(ii)	Consider the second order IIR filter with $$H(z) = \frac{1}{(1-0.5z^{-1})(1-0.45z^{-1})}$$ Find the effect on quantization on pole locations of the given system function in direct form and in cascade form. Assume b = 3 bits. (10 Marks)	Nov/ Dec 2011
14.	(i)	What is quantization of analog signals? Derive the expression for the quantization error. (10 Marks)	Nov/ Dec 2012
	(ii)	Explain coefficient quantization in IIR filter. (6 Marks)	Nov/ Dec 2012
15.	(i)	How to prevent limit cycle oscillation? Explain. (8 Marks)	Nov/ Dec 2012
	(ii)	What is meant by signal scaling? Explain. (8 Marks)	Nov/ Dec 2012
16.	(i)	Discuss the various common methods of quantization. (8 Marks)	Nov/ Dec 2013
	(ii)	Explain the finite word length effect in FIR digital filter. (8 Marks)	Nov/ Dec 2013
17.	Describe the quantization in floating point realization of IIR digital filters. (16 Marks)		Nov/ Dec 2013
18.	Discuss the following:		
	(i)	Product Quantization Error (8 Marks)	Nov/ Dec 2014
	(ii)	Limit Cycle Oscillations (8 Marks)	Nov/ Dec 2014
19.	(i)	Derive the equation for rounding and truncation errors. (8 Marks)	Nov/ Dec 2014
	(ii)	Derive the equation for quantization noise power. (8 Marks)	Nov/ Dec 2014
20.	(i)	The output of an ADC is applied to a digital filter with system function $$H(z) = \frac{0.5z}{z-0.5}$$ Find the output noise power from digital filter when input signal is quantized to have 8 bits. (8 Marks)	Nov/Dec 2015
	(ii)	Prove that $$\sum_{n=0}^{\infty} \lvert x(n) \rvert^2 = \frac{1}{2\pi j} \oint X(z)X(z^{-1})z^{-1}dz$$ (8 Marks)	Nov/Dec 2015
21.	A digital system is characterized by the difference equation $y(n) = 0.9y(n-1) + x(n)$ with $x(0) = 0$ and initial conditions $y(-1) = 12$. Find the dead band of the system. Verify with formula for largest integer.		Nov/Dec 2015