Code No: 09A60405 **R09** **Set No. 1**

III B.Tech II Semester Examinations,April/May 2012
DIGITAL SIGNAL PROCESSING
Electronics And Communication Engineering

Time: 3 hours Max Marks: 75

Answer any FIVE Questions
All Questions carry equal marks
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1. (a) Give the differences between analog and digital system.
   (b) Give the block diagram of Analog Signal Processing and compare with Digital signal Processing system and list out the applications of each. [7+8]
2. (a) Define DFT and IDFT. Prove Circular convolution, Circular correlation and Time reversal properties of DFT.
   (b) Find the IDFT of the sequence X(K) = {2, 2-3i. 4. 2+3j}. [7+8]
3. (a) Compare Butterworth and Chebyshev approximation techniques of filter designing.
   (b) Design a Digital Butterworth LPF using Bilinear transformation technique for the following specifications
   $0.707 \leq |H(w)| \leq 1; 0 \leq w \leq 0.2\pi$
   $|H(w)| \leq 0.08; 0.4\pi \leq w \leq \pi$. [7+8]
4. (a) Compare DIT and DIF FFT algorithms.
   (b) Develop the signal flow graph in computing 16-point FFT using DIT-FFT algorithm. [7+8]
5. (a) Define Phase delay and group delay.
   (b) The following transfer function characterises an FIR filter (M=11). Determine the magnitude response and show that the phase and group delays are constant. $H(z) = \sum_{n=0}^{M-1} h(n) z^{-n}$. [7+8]
6. (a) Define Frequency response, Magnitude spectrum, phase spectrum and time delay.
   (b) Determine the frequency response, magnitude response and phase response of the second order system
   y(n)+1/2y(n-1) = x(n)-x(n-1). [7+8]
7. (a) Discuss the effect of ADC Quantization noise on Signal Quality.
   (b) Discuss finite word length effects of implementation of FFT algorithm. [7+8]
8. (a) Design a poly phase filter structure for a sequence x(n) = { x(0), x(1), x(2), x(3) } Interpolated by a factor 3 and consider the filter length N=9.
   (b) Explain the process of performing subband coding for speech signals. [15]

Code No: 09A60405 **R09** **Set No. 2**

III B.Tech II Semester Examinations,April/May 2012
DIGITAL SIGNAL PROCESSING
Electronics And Communication Engineering

Time: 3 hours Max Marks: 75

Answer any FIVE Questions
All Questions carry equal marks
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1. (a) What is Signal Processing and list the advantages, Limitations of Digital Signal Processing. List out some applications of it.
   (b) Discuss in brief about the classification of signals. [7+8]
2. (a) What are the desirable characteristics of windowing function to be satisfied in filter design.
   (b) Design an FIR Digital Low pass filter using Blackman-Tukey window whose cutoff freq is 1.2 rad/s and length of window N=5. [7+8]
3. (a) What are Multirate Systems? Discuss their importance in real time processing of signals.
   (b) Explain the process of Interpolation by a factor I and also discuss how the images are eliminated with a neat block diagram. [7+8]
4. (a) Discuss the effects due to finite word length in Direct form - I and II structures.
   (b) Discuss the effect of quantization of coefficients in FIR filters. [7+8]
5. (a) Determine the relationship between DFT and Fourier transform of an aperiodic sequence.
   (b) Perform Linear convolution of the two sequences x(n) = {12, 3, -1, -2, -3, 4, 5, 3} and h(n) = {2, 1, -1} using over-lap save method and verify the result using Over-lap add method. [7+8]
6. (a) In a speech recording system with a sampling frequency of 10,000 Hz, the speech is corrupted by random noise. To remove the random noise while preserving speech information, the following specifications are given.
   Speech frequency range : 0 - 3000 KHz.
   Stop band range : 4,000 - 5,000 KHz.
   Passband ripple : 3 dB
   Stopband attenuation : 25 dB.
   Determine the filter order and transfer function using butterworth IIR filter.
   (b) How Chebyshev filter approximation is superior than butterworth filter approximation. [7+8]
7. (a) What is DIF algorithm? Give the mathematical analysis of DIF algorithm using Radix-2 In-place algorithm.
   (b) Compute 8-point DFT of the given sequence x(n) = {1, 2, 1, 2, 1, 2, 1, 2} using DIF FFT algorithm. [7+8]

Code No: 09A60405 **R09** **Set No. 3**

III B.Tech II Semester Examinations,April/May 2012
DIGITAL SIGNAL PROCESSING
Electronics And Communication Engineering

Time: 3 hours **Max Marks: 75**

Answer any FIVE Questions
All Questions carry equal marks

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1. (a) Compare FIR and IIR filters.
   (b) Justify the statement that FIR filters can have linear phase characteristics. [7+8]
2. (a) Give the Basic butterfly structures of computing DFT using DIT and DIF algorithm.
   (b) Compute the IFFT for the sequence X(K) ={0, 1, 2, 3, 0, 0, 0, 0}, using DIT algorithm. [7+8]
3. (a) Define Z- Transform. State any four properties of Z- transform.
   (b) A system has an impulse response h(n) = {1,2,3} and output response y(n) = {1, 1, 2,-1, 3}. Determine the input sequence. [7+8]
4. (a) Define the operations of a signal - Time scaling, Amplitude Scaling and folding.
   (b) What is an LTI system? Show that an LTI system combined with time scaling property may result in an Time-variant system. [7+8]
5. (a) Discuss the process of Decimation by a factor D with a neat block diagram.
   (b) Plot the signals and their corresponding spectra for rational sampling rate conversion by a) I/D = 5/3 and b) I/D = 3/5. Assume that the spectra of input signal x(n) occupies the entire range $-\pi \leq \omega_x \leq \pi$. [7+8]
6. (a) Discuss in detail the procedure of designing an analog filter using Butterworth approximation technique.
   (b) Explain how to convert an analog filter transfer function into digital filter transfer function using Bilinear transformation. [7+8]
7. (a) State the properties of DFT of a delayed sequence, Time reversed sequence, circular convolution, Circular frequency shift and circular time shift.
   (b) Determine the response of the system whose input x(n) and impulse response h(n) are given by x(n) = {1, 2} and h(n) = {1,2} using DFT and IDFT. [7+8]
8. (a) Discuss the effect of ADC Quantization noise on Signal Quality.
   (b) What are Limit Cycles?Discuss various types of Limit Cycles in brief. [7+8]

Code No: 09A60405 **R09** **Set No. 4**

III B.Tech II Semester Examinations,April/May 2012
DIGITAL SIGNAL PROCESSING
Electronics And Communication Engineering

Time: 3 hours **Max Marks: 75**

Answer any FIVE Questions
All Questions carry equal marks

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1. (a) Discuss the process of Decimation by a factor D and hence explain how the aliasing effect can be avoided.
   (b) Explain the process of performing subband coding for speech signals. [7+8]
2. (a) Compare the computational complexity of DFT and FFT.
   (b) An 8 point sequence is given by x(n) = {1, 2, 1, 2, 1, 2, 1, 2}. Compute 8 point DFT of x(n) using Radix-2 DIT FFT. [7+8]
3. (a) Compute DFT for the given sequence x(n) = {1, 2, 3, 4}.
   (b) Compute linear convolution of two given sequences x(n) ={1, 2, 3} and h(n) = {2, 3}. [7+8]
4. (a) Discuss various Discrete time sequences and mention the importance of each.
   (b) Define an LTI system and derive the expression for the output response of an LTI system whose input sequence is x(n) and impulse function of the system is h(n). [7+8]
5. (a) What is Bilinear transformation and sketch the mapping of S-plane into Z-plane in bilinear transformation.
   (b) Discuss the problems encountered in design of digital filter using Impulse invariant and bilinear transformation techniques. [7+8]
6. (a) FIR filters are always stable and have linear phase characteristics. Justify.
   (b) Design an FIR Digital Band stop filter using rectangular window whose upper and lower cut off frequencies are 4 & 5 rad/s and length of window N=9. Realize the filter using Linear phase Realization structure. [7+8]
7. (a) Determine the impulse response h(n) for he system described by the second order difference equation y(n)-4y(n-1)+4y(n-2) = x(n-1).
   (b) An LTI system is described by the equation y(n) = x(n)+0.8x(n-1)+0.7x(n-2)- 0.45y(n-2). Determine the transfer function of the system. Sketch its poles and zeros on the Z-plane. [15]
8. (a) Discuss finite word length effects of implementation of FFT algorithm.
   (b) What is scaling? Discuss how to reduce finite word length effects using scaling. [7+8]