Name: **Enrolment No:**	UPES

UNIVERSITY OF PETROLEUM AND ENERGY STUDIES
End Semester Examination, December 2020

Course: Digital Signal Processing (ECEG2013) **Semester: V**
Program: B Tech ECE
Time: 03 hrs. **Max. Marks: 100**
No. of page/s: 3

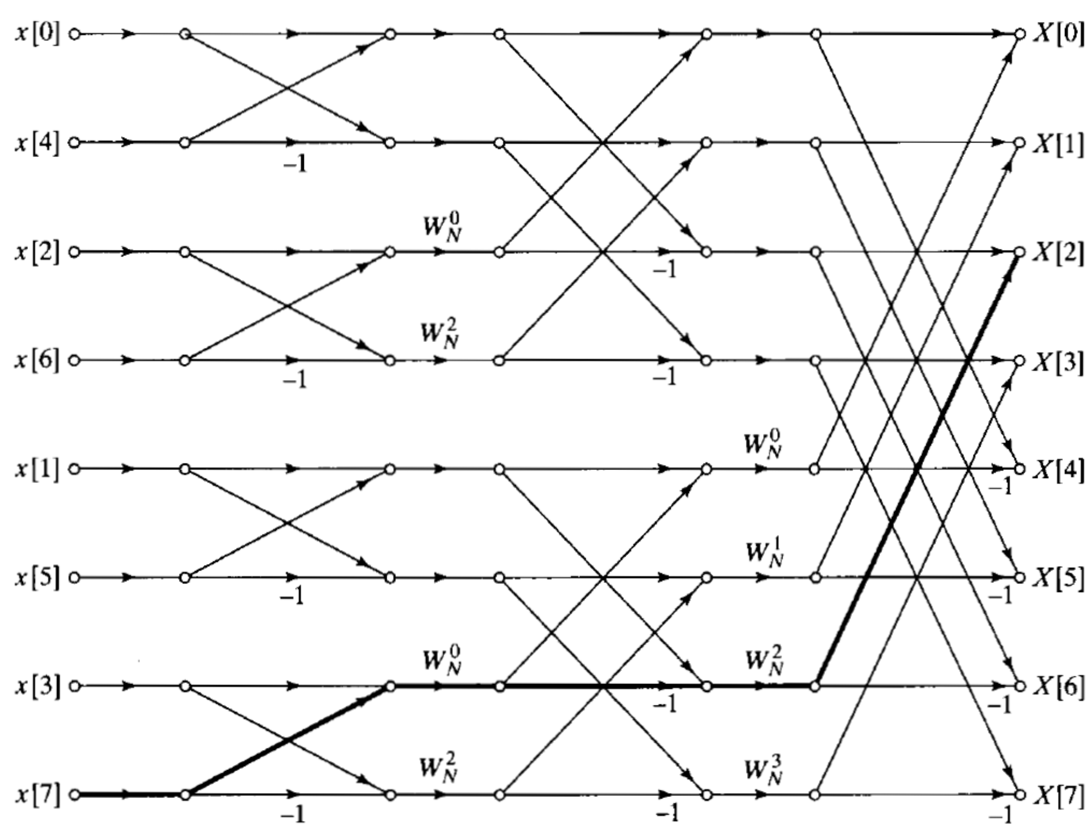
Instructions:
- The question paper contains three sections namely Section-A, Section-B and Section-C.
- Attempt all questions. The number of marks for each question is mentioned on the right side of it.
- Assume any data if required and indicate the same clearly. Unless otherwise indicated symbols and notations have their usual meanings.

SECTION A (30 Marks)

S. No.		**Marks**	**CO**
Q 1	Define the following properties of the discrete time system: Linearity and non-linearity; causality and non-causality; time invariant and time invariant; stability and un-stability	**5**	**CO1**
Q 2	What is FFT? Calculate the percentage saving in calculations in a computation of 256-point DFT using direct computation and that using FFT?	**5**	**CO2**
Q 3	Define the following terms: Phase delay, Group delay, linear phase response. Also mention how phase distortion and delay distortion are introduced?	**5**	**CO3**
Q 4	Write the steps involved in the following digital filter design: **(a)** IIR filter design; **(b)** FIR filter design.	**5**	**CO4**
Q 5	Define cross-correlation and auto-correlation sequence. Also write relation between linear convolution and correlation.	**5**	**CO1**
Q 6	What are the possible types of impulse response for linear phase FIR filters? Briefly describes the characteristics of each type.	**5**	**CO3**

SECTION B (50 Marks)

Q 1	Consider a system with input x[n] and output y[n] that satisfy the difference equation $y[n] = ny[n-1] + x[n]$. The system is causal and satisfies zero initial conditions. (a) If $x[n] = \delta[n]$, determine y[n] for all n (b) Is the system linear? Justify your answer. (c) Is the system time invariant? Justify your answer.	**10**	**CO1**
Q 2	Consider two finite-length sequences x[n] and h[n] for which $x[n] = 0;\ for\ n \leq 0\ and\ n \geq 24$ and h[n] = 0; for $n \leq 0\ and\ n \geq 9$	**10**	**CO2**

	(a) what is the maximum number of non-zero values in the linear convolution of x[n] and h[n] (b) The 25-point circular convolution of x[n] and h[n] is $x[n] \circledast h[n] = 5;\ for\ 0 \le n \le 24$ and the first five points of linear convolution of x[n] and h[n] are $x[n] * h[n] = 2;\ for\ 0 \le n \le 4$ Determine as many points as possible of the linear convolution of x[n] and h[n]		
Q 3	**Figure.1**, shows the graphical representation of decimation in time FFT algorithm for N = 8. The heavy line shows a path from input sample x[7] to X[2].  **Figure.1** **(a)** What is the gain along the path that darken in **Figure.1**. **(b)** How many other paths in the flow graph begin from x[7] to X[2]? Also find how many paths are there between each input sample x[n] and each output sample X[k] **(c)** Now consider the DFT sample X[2]. By tracing paths in the flow graph of **Figure.1**, show that each input sample contributes the proper amount to the output DFT sample; i.e., verify that $X[2] = \sum_{n=0}^{N-1} x[n]\, e^{-j\frac{2\pi}{N}2n}$	**10**	**CO2**
Q 4	An IIR filter has the following transfer function: $$H(z) = \frac{0.1436 + 0.2872z^{-1} + 0.1436z^{-2}}{1 - 1.8353z^{-1} + 0.9748z^{-2}}$$ (a) Determine the position of the poles and zeros and sketch pole-zero plot (b) Realize the filter using the direct form-I and direct form-II. Also, determine the difference equations for implementation	**2+4+4**	**CO3**

	(c) Compute the first 4-samples (n = 0,1,2,3) of the filter output signal y[n] if $x[n] = \delta[n] - \delta[n-1]$. Assume zero initial conditions.		
Q 5	(a) Given a fourth-order filter transfer function $$H(z) = \frac{(1.0215z + 0.5108)\,(0.4371z^2 + 0.8742z + 0.4371)}{(z^2 + 0.5654z + 0.4776)(z^2 - 0.1316z + 0.1733)}$$ Realize the digital filter using the cascade (series) form via second order sections using the direct form II (b) Digitize the analog filter with transfer function $H(s) = \frac{s+1}{s(s+2)}$ using the impulse invariant method, Assume the sampling frequency of 10 Hz	**5+5**	**CO3**
SECTION-C (20 Marks)			
Q 1	A digitally recorded speech in the noisy environment can be enhanced using a lowpass filter if the recorded speech with a sampling rate of 8000Hz contains information within 1600Hz. Design a lowpass filter using window method to remove the high-frequency noise above 1600Hz with following filter specifications: passband frequency range: 0–1600Hz; passband ripple: 0.02 dB; stopband frequency range: 1800–4000Hz; and stopband attenuation: 50dB. Also realize with suitable structure. **Or** A bandpass digital IIR filter with Butterworth response is required to remove baseline wander and artefacts due to body movement in a certain biomedical application. The filter requires to meet the following requirements: Passband: 20 - 50 Hz Stopband: 0 - 5 Hz and 70 -128 Hz Passband ripple: 3 dB Stopband attenuation: 20 dB Sampling frequency: 256 Hz Obtain the suitable transfer function for the filter using the bilinear transformation method. Also realize with suitable structure.	**20**	**CO4**