

Seat
No.

--	--	--	--	--	--



मानव - 009

Advanced Digital Signal Processing & Processors

P. Pages : 2

Time : Three Hours

Max. Marks : 100

Instructions to Candidates :

1. Do not write anything on question paper except Seat No.
2. Answersheet should be written with blue ink only. Graph or diagram should be drawn with the same pen being used for writing paper or black HB pencil.
3. Students should note, no supplement will be provided.
4. Attempt **any five** questions.
5. Assume suitable data if necessary.
6. Figures to the right indicate full marks.

1. a) Compare Linear and circular convolution. How Linear convolution is calculated using circular convolution. Obtain the linear convolution using circular convolution using for following sequence. 10
$$x_1(n) = \left\{ \underset{\uparrow}{2}, 2, 2, \right\}$$
$$x_2(n) = \left\{ \underset{\uparrow}{2}, 2, 2, \right\}$$
- b) Explain following properties of DFT. 10
 - i) Circular convolution.
 - ii) Circular symmetry.
2. a) What is principal of interpolation ? Derive the expression for interpolated signal at the output. 10
- b) Design a 4 stage decimator is used to reduce the sampling rate from 96KHz to 1KHz. Assume decimation factors 4,3,4,2 pass band ripple $\delta_p = 0.01$ and passband deviation $\delta_s = .001$. Design and efficient decimeter. Calculate MPS and TSR for the design. 10
3. a) What are FIR filter characteristics. Show that for symmetric or antisymmetric impulse response, it gives linear phase. 10
- b) Design high pass digital filter to meet the following specifications : 10

passband : 2 – 4 KHZ
stopband : 0 – 500 Hz
 δ_p : 3 dB
 δ_s : 20 dB

Assume butterworth approximation.

4. a) Explain Impulse invariance transformation. What is its drawback and how Bilinear transformation overcomes it. Show graphical representation. Explain concept of frequency pre-wrapping. 6
- b) A digital low pass filter is required to meet the following specifications : 8
- Passband ripple : $\leq 1\text{dB}$
 Passband edge : 4KHz
 Stopband Attenuation : $\geq 40\text{dB}$
 Stopband edge : 6KHz
 Sample rate : 24KHz
- c) Obtain the direct form - I and direct form - II structures for the following system.
 $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.255x(n-2)$. 6
5. a) Derive the basic LMS algorithm and explain the steps to implement it. Comment on the robustness and convergence of the same. Discuss statistical LMS theory. 8
- b) Define parametric and non - parametric methods of power spectrum estimation. Explain the non - parametric methods of power spectrum estimation. 12
6. a) For the 8 sample sequence $x[n] = [1, 2, 3, 5, 5, 3, 2, 1]$, the first five DFT coefficient are $[22, -7.5355 - j3.1213, 1+j, -0.4645 - j1.1213, 0]$. Determine the remaining three DFT coefficients. 10
- b) Compute the eight - point of a sequence, $x(n) = \{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\}$ using radix - 2FFT algorithm. 10
7. a) Draw the architecture of TMS320c62XX DSP processor and explain in short. 12
- b) Discuss the application of DSP in Radar signal processing. 8
8. Write short notes on : 20
- Sub band coding of speech signal.
 - Least square filter design.
 - Goertzel algorithm.
 - FIR differentiator.