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मुख - 017

Advanced Digital Signal Processing & Processors (1010)

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**. Each question carries equal marks.
5. Draw well label diagram and assume suitable data whenever necessary.

1. a) Explain the following properties of DFT.
 - i) Circular Convolution
 - ii) Duality Property
- b) Obtain the 4 point DFT of following windows function, $w(n) = u(n) - u(n - N)$.
2. a) Explain different properties of FIR Filters.
 - i) Inherently stable property.
 - ii) Symmetric and anti-symmetric property.
- b) Differentiate between following IIR filter design technique.
 - i) Impulse invariant technique.
 - ii) Bilinear Transformation.
3. a) Explain discrete Hilbert Transform and prove the two real and imaginary part.
- b) Explain the Frequency response of Linear Phase FIR Filter.
4. a) Design a normalized linear phase FIR filter having phase delay of $\tau = 4$ and at least 40 dB attenuation in side band. Also obtain magnitude frequency response of the filter.

- b) What is Gibb's phenomenon ? Explain the need of window function in design of FIR filter.
5. a) Perform the circular convolution of the following two sequences
 $x(n) = \{0, 1, 2, 3\}$, $h(n) = \{2, 1, 1, 2\}$.
- b) Design a digital Butterworth filter that satisfies the following constraint using Bilinear Transformation. Assume $T = 1$ Sec.
- $$0.80 \leq |H(e^{j\omega})| \leq 1 \quad 0 \leq \omega \leq 0.2\pi$$
- $$|H(e^{j\omega})| \leq 0.2 \quad 0.6\pi \leq \omega \leq \pi$$
6. a) Explain LMS adaptive algorithm in detail.
- b) Design an FIR Filter to approximate an ideal LPF with passband gain of unity, cut off frequency of 850Hz and working at sampling frequency of 5000Hz. The length of impulse response should be 5. Use rectangular and Hamming window methods.
7. a) Explain sampling rate conversion by rational factor I/D in detail.
- b) Explain how Multirate sampling can be used efficiently in
- Sub-Band coding.
 - Acquisitions of high quality data.
8. a) Explain the desirable features of digital Signal Processor. Draw the Architectural block diagram of DSP Processor and explain the fn of each block.
- b) Draw and Explain the Architecture of TMS 320c62XX DSP processor. Enlist the features.
