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Question Paper Code : 31877

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2013.

Fifth Semester

Medical Electronics Engineering

BM 3302/080290029 — DIGITAL SIGNAL PROCESSING

(Common to 080290029 Digital Signal Processing for Electronics and Communication Engineering)

(Regulation 2008)

Time : Three hours | Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. What do you mean by fast convolution?

2. Write Radix 2 FFT Algorithm.

3. Why we use Hamming window instead of rectangular window for designing FIR filters? Write the window function of Hamming window.

4. What is advantage of Kaiser window over Hamming window for spectral analysis?

5. How are digital IIR filters designed?

6. What is the difference between Chebyshev type 1 and type 2 filters? Write the equation for magnitude square response of the two filters.

7. What do you mean by limit cycle oscillations?

8. Define dead band.

9. List out the various addressing modes used in TMS320C50 Processor.

10. Draw the DSP Building blocks.

PART B — ($5 \times 16 = 80$ marks)

11. (a) Compute the output using 8 point DIT-FFT algorithm for the sequence $\{1\ 2\ 3\ 4\ 5\ 5\ 3\ 1\}$.

Or

(b) Using Overlap Add Block convolution method, find the convolution of two given sequences.

$x(n) = [1\ 2\ 34\ 5\ 6\ -1\ -2\ -3\ -4\ -5\ -6]$ and $h(n) = [1\ 2\ -1\ -2]$.

12. (a) Using the Kaiser window, design a bandpass digital filter with the following specifications:

$f_s = 20\ kHz$; $f_{sa} = 3kHz$; $f_{pa} = 4\,kHz$ $f_{pb} = 6\,kHz$ $f_{sb} = 8kHz$

$A_{pass} = 0.1\,dB$, $A_{stop} = 80\,dB$.

Or

(b) A signal consisting of four sinusoids of frequencies of 3, 3.5, 4 and 4.35kHz is sampled at a rate of 20kHz. What is a minimum number of samples that should be collected for the frequency spectrum to exhibit four distinct peaks at these frequencies? How many samples should be collected if they are going to be preprocessed by a Hamming Window and then fourier transformed? How many samples should be collected if they are going to be preprocessed by a Kaisor Window with a side lobe level is 60dB and then fourier transformed?

13. (a) Using bilinear transformation and a lowpass analog Butterworth prototype, design a lowpass digital filter operating at a rate of 20kHz and having passband extending to 5kHz with maximum passband attenuation of 0.5dB and stopband starting at 6kHz with minimum stopband attenuation of 10dB.

Or

(b) Using bilinear transformation and a lowpass analog Chebyshev Type 1 prototype, design a lowpass digital filter operating at a rate of 20kHz and having passband extending to 3kHz with maximum passband attenuation of 0.5dB and stopband starting at 4kHz with minimum stopband attenuation of 8dB.

14. (a) Derive the quantization noise power for the communication system.

Or

(b) Derive the expression for quantization error in the communication system.

15. (a) Draw the TMS320C50 Architecture in detail.

Or

(b) Explain in detail about pipelining.
