



**MANIPAL  
UNIVERSITY**



# Question Paper

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## MANIPAL UNIVERSITY

SCHOOL OF INFORMATION SCIENCES

SECOND SEMESTER MASTER OF ENGINEERING - ME(EMBEDDED SYSTEMS)

DEGREE MAKE-UP EXAMINATION - JULY 2017

Monday, 10 July, 2017

Time: 10:00 to 13:00

### Digital Signal Processing [ESD 602]

Marks: 100

Duration: 180 mins.

**Answer all the questions.**

- 1) Compute the DFT of the sequence  $x(n) = [1, -1, -1, -1, 1, 1, 1, -1]$  using DIT-FFT algorithm. Draw the flow graph indicating the intermediate values (10)
- 2) Realize the following system function using Direct form-I, Direct form-II and Cascade / Parallel forms (10)  

$$H(z) = [(z^2 + 0.5z + 1)(z + 0.6)] / [(z^2 + 0.6z + 0.2)(z - 0.8)]$$
- 3) Design an ideal linear phase FIR low pass filter with a cutoff frequency of  $\pi/2$  radians, using frequency sampling technique. Assume 11 tap coefficients (10)
- 4) It is desired to remove low frequencies of an analog signal with a digital linear phase FIR filter. The 3 dB frequency is 2 KHz, transition width is 500 Hz and the stop band attenuation is 50 dB. Use suitable window function to design the filter to meet the above specification. The filter employs a sampling frequency of 10 KHz. (10)
- 5) A third ordered Chebychev lowpass filter with 3 dB frequency of 5 KHz is to be realized using digital system. The sampling period is 10  $\mu$ sec. Realize the filter using Impulse Invariance technique (20)

- 6) Provide polyphase filter structures of interpolator and decimator. Explain how these structures are in a position to provide the required sampling rate conversion (10)
- 7) What is a digital filter bank? Explain how an uniform DFT filter bank can be implemented using multirate signal processing (10)
- 8) Explain LMS adaptive algorithm. Explain how LMS adaptive algorithm is made use to make the Weiner Filter Configuration adaptive based on the steepest descent technique (10)
- 9) Explain the internal and external memory organization in TMS320C6X DSP Processor (10)



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