



**MANIPAL
UNIVERSITY**



Question Paper

19-Apr-2017 10:49:03

GOWTHAM S

[Support](#) | [Change Password](#) | [Logout](#)
[Basic Masters](#)[Question Paper](#)[Exam Config.](#)[Device & Dockets](#)[Evaluation](#)[Evaluation Reports](#)[Reports](#)[Utilities](#)[Imports](#)

Select Exam Event SOIS End Semester April 2017

Refresh

☐ Show answer keys for MCQ.[Back](#)

MANIPAL UNIVERSITY

SCHOOL OF INFORMATION SCIENCES (SOIS)

SECOND SEMESTER MASTER OF ENGINEERING - ME (EMBEDDED SYSTEMS) / FOURTH SEMESTER MSc Tech(VLSI
DESIGN / EMBEDDED SYSTEMS)

DEGREE EXAMINATION - APRIL / MAY 2017

Wednesday, 19, 2017

Time : 10:00 AM - 1:00 PM

Digital Signal Processing [ESD 602]

Marks: 100

Duration: 180 mins.

A

Answer all the questions.

- 1) Find the DFT of $x(n) = [1, 2, 3, 4, 4, 3, 2, 1]$ using DIF-FFT algorithm (10)
- 2) Realize the following system function using Direct form-I, Direct form-II and Cascade / Parallel forms (10)

$$H(z) = \frac{[2(1-z^{-1})(1+1.414z^{-1}+z^{-2})]}{[(1+0.5z^{-1})(1-0.9z^{-1}+0.81z^{-2})]}$$
- 3) Design a linear phase bandpass FIR filter having cut-off frequencies of 1 KHz and 3 KHz using Frequency Sampling Technique. The system employs a sampling frequency of 8 KHz. Assume 10 tap coefficients. (10)
- 4) Design an FIR linear phase lowpass filter using windows to meet the following specifications. (10)

$$0.99 < |H(e^{jw})| \leq 1.01; \quad \text{for } 0 \leq |w| \leq 0.19\pi \text{ rads}$$

$$|H(e^{jw})| \leq 0.01; \quad \text{for } 0.21\pi \leq |w| \leq \pi \text{ rads}$$
- 5) Design using impulse invariance technique, a digital Butterworth lowpass filter for the following specifications. (20)

$$|H(j\Omega)| \geq -1 \text{ dB} \quad 0 \leq \Omega \leq 100 \text{ rad/sec}$$

$$|H(j\Omega)| \leq -40 \text{ dB} \quad \Omega \geq 2000 \text{ rad/sec.}$$

Sampling frequency = 8000 rad/sec. Realize the filter structure.
- 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) (10)

Explain how multirate signal processing can be used in the analysis and synthesis of sub-band coding of speech signals.

- 8) Explain analytically, how optimum filter coefficients are obtained on Mean Square Error sense in Wiener Filter Configuration. (10)
- 9) Draw the architecture of a TMS320C6X DSP processor and give the functionality of each of the block. (10)



Copyright @ 2017 Littlemore Innovation Labs Pvt IP: 52.66.163.73 epCloud 1.5
Ltd. All rights reserved.