B. TECH. SEM -VI (E & TC ENGG.) (2014 COURSE) (CBCS) : SUMMER-2018

SUBJECT: DIGITAL SIGNAL PROCESSING

Day: **Friday** Time: **02.30 PM TO 05.30 PM**

Date: **01/06/2018** **S-2018-2457** Max. Marks: 60

N.B.:

1) All questions are **COMPULSORY**.
2) Figures to the right indicate **FULL** marks.
3) Use of non-programmable scientific **CALCULATOR** is allowed.

Q.1 What are criteria for causality and stability in the z domain for linear time invariant the systems? **(10)**

OR

Determine the magnitude and phase of H (ω) for the system described by **(10)**

$y\,(n) = \frac{1}{3}[x(n) + x(n-1) + x(n-2)]$ and plot the two functions for $0 \le \omega \le \pi$

Q.2 Find the discrete Fourier transform of a sequence **(10)**

$$x(n) = \{\underset{\uparrow}{2},\ 0,\ 3,\ 1\}$$

OR

Draw the diagram for the Eight-point decimation-in-time FFT algorithm. **(10)**

Q.3 Obtain an expression for the frequency response function H (ω) for a symmetric FIR of length M = 5. **(10)**

OR

Find impulse response h (n) for an antisymmetric FIR filter of length M=4 with frequency response function specified as **(10)**

$H_r(0) = 0$, $H_r\left(\frac{\pi}{2}\right) = \frac{1}{\sqrt{2}}$ *and* $H_r(\pi) = \frac{3}{2}$.

Q.4 Can we use the impulse invariance method for the design of high pass filter? Justify your answer. **(10)**

OR

Convert the analog filter with system function **(10)**

$$H(s) = \frac{1}{(s+0.1)^2 + 16}$$

into digital IIR filter using the approximation of derivatives method.

Q.5 How to reduce the errors arising out of coefficient quantization in digital filters? **(10)**

OR

What is the purpose of scaling with reference to the finite word length effects? **(10)**

Q.6 What is the purpose of address generators in the DSP processors? **(10)**

OR

What are the types of musical sound processing which can be done using DSP? **(10)**

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