



Institute: (0701) SYMBIOSIS INSTITUTE OF TECHNOLOGY, PUNE

Programme: (070121) B.TECH
Electronics & Telecommunication

Batch: 2012-16

Semester: V

Course: Digital Signal Processing

Course Code: 0701210502EC

Date: 10/12/2014

Maximum Marks: 75

Day: Wednesday

Time: 10:00 am - 12:30 pm

Instructions:

- 1) Use of non-programmable scientific calculator is permitted.
- 2) Make suitable assumptions wherever required.
- 3) Draw neat diagrams wherever required.
- 4) Part A is compulsory.
- 5) Answer any 5 questions out of 6 questions in Part B.

Part A

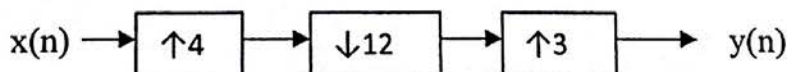
Q.1 a) Compute 4 point DFT of the sequence $x(n) = \sin(n\pi/4)$ 3

b) Compute 5 point circular convolution of two DT signals defined as, 4

$$x(n) = (2)^n \quad 0 \leq n \leq 2$$
$$h(n) = 2n-1 \quad 0 \leq n \leq 3$$

c) How can the transition bandwidth of the FIR filter be reduced in design using windows? Explain it for different window functions. 3

d) Obtain the expression for output $y(n)$ for given multirate system. 3



e) Convert the analog filter with system function 3

$H(s) = \frac{s+0.1}{(s+0.1)^2+9}$ into a digital IIR filter using impulse invariant technique. Assume $T=0.5s$.

f) Justify the necessity of MAC in DSP processor. 2

g) What is meant by fixed point and floating point DSP processor? 2

h) Obtain the direct form II and cascade structures for the following system. 5
 $y(n) = -0.1y(n-1) - 0.72y(n-2) + 0.7x(n) - 0.2x(n-2)$

Part B

- Q.2 a) Draw the flow graph for the implementation of 8 point DIF FFT of the following sequence. 8
 $x(n) = \cos(n\pi/2)$
- b) Give the computational efficiency of FFT over DFT. 2
- Q.3 Using impulse invariant transformation convert the following analog filter transfer function to digital filter function by taking sampling time, $T=0.5s$ 10
$$H(s) = \frac{2.8s^2 + 4.8s + 2.9}{(s+3)(s^2 + s + 0.85)}$$
- Q.4 a) Design a normalized linear phase FIR filter having the phase delay $\tau=3$ and at least 44 dB attenuation in the stopband. Assume cut off frequency of $\pi/4$ rad/sec. 8
- b) Compare FIR and IIR filter. 2
- Q.5 a) Design a two stage decimator with down sample factor of 96 with following specifications. (Assume $M_1=16$ and $M_2=4$) 6
 $F_s = 3.2\text{MHz}$
Highest Frequency of interest in data = 20KHz
Passband ripple = 0.01dB
Stopband ripple = 20dB
- b) Explain the application of multirate DSP in speech processing for sub band coding. 4
- Q.6 a) FIR digital filter has the unit impulse response sequence $h(n) = \{1, 2, 2\}$. Determine the output sequence in response to the input sequence $x(n) = \{2, 1, 0, -2, 0, 2, 1, -2, 2, 1\}$ using overlap-save method. 6
- b) Determine the parallel form realization of the given IIR digital filter transfer function 4
$$H(z) = \frac{3(2z^2 + 5z + 4)}{(2z+1)(z+2)}$$

- Q.7
- a) Explain the various ways of fast computation in DSP processor. 4
 - b) Draw the architectural block diagram of TMS320C54x processor. 4
 - c) State the features of TMS320C54x processor. 2