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# Fourth Semester B.Tech. Degree Examination, June 2016 (2013 Scheme)

13.404 : DIGITAL SIGNAL PROCESSING (AT)

Time: 3 Hours Max. Marks: 100

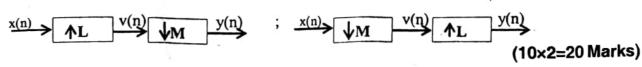
## PART-A

Answer all questions. 2 Marks each.

1. What is meant by zero padding? Why it is used?

+0. Compute IDFT of a sequence :  $\{12, -4 + 4j, -4, -4 - 4j\}$ .

- 3. For what kind of applications, the symmetrical impulse response of FIR filter is used?
  - 4. What are Gibbs oscillations?
- Determine the order of Butterworth filter for a given specifications :  $\alpha_p = 1$  dB;  $\alpha_s = 30$  dB;  $\Omega_p = 200$  rad/sec.,  $\Omega_s = 600$  rad/sec.
- How many number of additions, multiplications and memory locations are required to realize a system H(z) having M zeros and N poles in direct form II realizations?
- Compare the fixed point and floating point arithmetic.
  - 8. What is the steady state variance of the noise in the output due to the quantization of the input for the first order filter?
  - 9. State the identities of interpolator.
  - 10. Under what condition the following systems will become identical?



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## PART-B

Answer one question from each Module.

## Module - 1

necessary butterfly line diagram. Take N = 8.	15
(n) State and prove the complex conjugate property of DFT.	5
12. a) An FIR digital filter has the unit impulse response sequence, $h(n) = \{2, 2, 1\}$ .  Determine the output sequence in the response to the input sequence $x(n) = \{3, 0, -2, 0, 2, 1, 0, -2, -1, 0\}$ using the overlap save convolution method.	12
$ω$ 1 $\checkmark$ 6) Using efficient computation, perform DFT of sequence $x(n) = \{1, 4, 9, 16\}$ .	4
Explain the use of FFT algorithm in correlation.	4
Module – 2	•
13, a) Explain the windows used in FIR filter design with necessary equations.	6
b) A low pass filter has the desired response as given below.	
$H_d(e^{j\omega}) = e^{-j3\omega}; 0 \le  \omega  \le \frac{\pi}{2}$	
$= 0 ; \frac{\pi}{2} <   \omega   \leq \pi$	12
Determine the filter coefficients $h(n)$ for $M = 7$ using frequency sampling technique.	
What is the necessary and sufficient condition for linear phase characteristics in FIR filter?	2
14. a) Determine the system function H(z) of the lowest-order Chebychev digital filter that meets the following specifications:	12
i) 1 – dB ripple in the pass band 0 $\leq$   $\omega$   $\leq$ 0.3 $\pi$	
ii) Atleast 60 dB attenuation in the stop band $0.35\pi \le  \omega  \le \pi$ . Use bilinear transformation.	
Derive mapping formula for impulse invariance technique in IIR filters.	8
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## Module - 3

15 a) A system is represented by a transfer function H(z) is given by

 $H(z) = 3 + \frac{4z}{z - 1/2} - \frac{z}{z - 1/4}$ 

- i) Does this H(z) represent IIR or FIR filter?
- ii) Give a difference equation realization of this system using direct form I. 4
- iii) Draw the block diagram for direct form II realization and give the governing equations for its implementation.
- b) Sketch the ladder structure for the system,

$$H(z) = (1 - 0.5z^{-1} + 1.2 z^{-2})/(1 + 0.15z^{-1} - 0.64z^{-2})$$

Also check the stability of the system.

- 16. a) The input to the system,  $y(n) = 0.999 \ y(n-1) + x(n)$  is applied to an ADC. What is the power produced by the quantization noise at the output of the filter if the input is quantized to 8 bits and 16 bits.
- b) Explain the characteristics of limit cycle oscillations with respect to the system described by the difference equation y(n) = 0.95y(n-1) + x(n) and also determine the dead band of the filter.
- Explain the coefficient quantization error with an example.

#### Module - 4

- 17. a) What is the significance of multirate conversion in sub-band coding? Derive the performance of analysis and synthesis filter in sub-band coding applications.
- Explain the sampling rate conversion by a rational factor of I/D with suitable examples.
- 18. (a) Explain the architecture of TMS320C6713 processor with a neat diagram.
  - b) Draw the block diagram of FDM to TDM trans-multiplexer and explain.