**SIDDHARTH INSTITUTE OF ENGINEERING AND TECHNOLOGY :: PUTTUR**
**(AUTONOMOUS)**
Siddharth Nagar, Narayanavanam Road – 517583

**QUESTION BANK**

**Subject with Code : Advanced DSP& Applications**(16EC5504)

**Course & Branch**: M.Tech – (ES) **Year & Sem:** I-M.Tech & I-Sem

## UNIT-I

## LIT DISCRETE-TIMESYSTEMS IN THE TRANSFORM DOMAIN

1.(a) Discuss the principle of extracting minimum phase function from the magnitude spectrum for digital filters. [5M]

(b)Determine the minimum phase transfer function following magnitude squared function for the given $H(j\Omega)$ $H(j\Omega) = \frac{5+4.\cos(\Omega)}{17+8..\cos(\Omega)}$ [5M]

2.(a) Explain about the poly phase filters structures and how they are used in interpolation. [5M]

(b) Write down differences between FIR and IIR filter [5M]

3. (a) Design of linear phase FIR filters using windows. [5M]

(b)Write the different types of linear – phase transfer functions [5M]

4. (a) Explain about IIR trapped cascaded lattice filters. [5M]

(b).Write a short note digital sine-cosine generator. [5M]

5. (a) Consider two LTI causal digital filters with impulse responses given by: [5M]

$$h_1(n) = 0.5\delta(n) - \delta(n-1) + 0.5\delta(n-2)$$
$$h_2(n) = 0.5\delta(n) + \delta(n-1) + 0.5\delta(n-2)$$

Sketch the magnitude response of the two filters and compare their characteristics

(b) State and explain the process of deconvolution in inverse systems with suitable example [5M]

6. (a) Draw the cascaded lattice structure that can be used to realize an arbitrary FIR transfer function and develop the realization algorithm for the same. [5M]

(b) Derive a single multiplier structure for generating sine-cosine sequences from a general second order digital filter structure. [5M]

7. (a) Explain algebraic stability test [5M]

(b) With an example explain the all pass realization of IIR transfer functions [5M]

8. (a) Physically realizable and stable IIR filters cannot have linear phase. Prove [5M]

(b) Describe the characteristics of IIR and FIR systems. [5M]

9 .(a) Explain the poly-phase digital filter structure [5M]

(b)Explain state space structure and poly phase structure. [5M]

10.(a) What is an all pass filter and write down in properties? [5M]

(b). (i) Define a casual system. [5M]

(ii)Differentiate stable from a unstable system.

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**QUESTION BANK (DESCRIPTIVE)**

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**Year & Sem:** M.TECH-I SEM

## UNIT-II

1.(a) Explain the design of an IIR filter using Pades's approximation method. [5M]

(b)Explain the advantages of Pades's approximation method. [5M]

2. (a).Explain the method of designing IIR filters using Pade approximation [5M]

(b) Explain the complexity of digital filter structure [5M]

3. (a) What are the advantages of FIR filter? [5M]

(b). Write a short note on narrow frequency band of DFT. [5M]

4.(a). Discuss about fast DFT algorithm based on index mapping [5M]

(b). Explain about least square design method [5M]

5.(a) Compare the filter design for FIR and IIR filter. [5M]

(b). If $H(S) = \dfrac{2}{(S+1)(S+4)}$ [5M]

determine $H(Z)$ using impulse invariance method for T = 0.1 sec & 1 sec

6. What do you understand by pade approximation method to design an IIR filter? [10M]

In what respect, this method is different from the method of frequency transformation

7. Explain the design of computationally Efficient FIR Filters. [10M]

8. (a) Explain the principle of spectral transformations of IIR filters. [5M]

(b).What are the issues in the design of FIR filters? [5M]

9. List out the least square design methods ?Explain each with one example. [10M]

10. A band limited analog signal is sampled (with no aliasing) at 500 Hz and [10M]

980 samples are collected. DFT of these 980 samples is computed. It is desired to computeThevalueofthespectrumoftheesampledsignalat120Hz.

(i)Which DFT index is nearest to 120 Hz and what is its physical frequency in hertz?

(ii)What is the minimum number of zeros that need to be padded onto the 980 samples to obtain a DFT at 120 Hz exactly? What is the DFT index k then corresponding to 120 Hz.

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## UNIT –III

## DSP ALGORITHMS

1. (a).Discuss about fast DFT algorithm based on index mapping. [5M]
   (b).Write a short note on narrow frequency band of DFT [5M]
2. (a) Explain Cooley-Tukey FFT algorithm for composite length N = 15. [5M]
   (b).What is Chirp z-transform? Develop DFT computation using the Chirp z-transform [5M]
3. (a). Discuss the merits of computation of DFT using chirp Z-transform. [5M]
   (b).What is the DFT used in a very long sequence? Give its features. [5M]
4. (a).Explain using an example, the computation of DFT using chirp Z-transform [5M]
   (b).What if FFT? Describe an algorithm for decimation on time FFT. [5M]
5. (a) Explain why sliding discrete Fourier transform is efficient. Also prove that an IIR filter implementation with a comb filter and resonator cascade filter using the concept of sliding DFT is efficient [5M]
   (b).Explain "Divide and Conquer" approach for computation of DFT and write any two algorithms to compute DFT using this approach. [5M]
6. (a ) Discuss about fast DFT algorithm based on index mapping. [5M]
   (b)Write a short note on narrow frequency band of DFT. [5M]
7. (a) Explain the complexity of digital filter structure. [5M]
   (b).What are the advantages of FIR filter? [5M]
8. Explain the computation of DFT approach for linear filtering using chip Z-transform [10M]
9. What if FFT? Describe an algorithm for decimation on time FFT. [10M]
10. Write short notes on following [10M]
    (a) Split radix FFT.
    (b). An pass transfer function
    (c) Difference between analog and digital filter

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## UNIT –IV

## ANANLYSIS OF FINITE WORLD LENGTH EFFECTS

1.(a ) Explain the quantization by truncation and rounding method. [5M]

(b)Explain zero input and overflow limit cycle oscillations with respect to finite word length effects. [5M]

2. (a) Discuss about fast DFT algorithm based on index mapping. [5M]

(b)Write a short note on narrow frequency band of DFT. [5M]

3. (a) What are the errors that effect using feedback? [5M]

(b)Write the limits of IIR digital filters. [5M]

4. (a) Illustrate the process of quantization of fixed point and floating point numbers in the analysis of finite word length effects. [5M]

(b)Explain the effect of input scaling on signal to noise ratio (SNR). [5M]

5. (a)Consider a second order digital filter structure and find its model for product round-off error analysis with an example. [5M]

(b)Discuss about round-off errors in FFT algorithm. [5M]

6. (a) Using an example, explain how the quantization of fixed point numbers is carried out. [5M]

(b).Write short notes on analysis of coefficient quantization effects. [5M]

7. (a) Discuss what do you mean by dynamic range scaling. [5M]

(b).Explain how round of errors in FFT algorithms is done. [5M]

8.(a ) Write short notes on analysis of coefficient quantization effects in FIR filters. [5M]

(b).Using a model diagram, explain A/D conversion noise analysis. [5M]

9. (a)Discuss about quantization process and explain quantization of fixed-point numbers. [5M]

(b).Write a short note on dynamic range scaling [5M]

10. (a) What do you mean by dynamic range scaling? Explain in detail. [5M]

(b).How is the product round off errors reduced? [5M]

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## UNIT –V

## APPLICATIONS OF DSP &MULTIRATE SIHNAL PROCESSING

1.(a) Compare Von Neumann architecture, Harvard and modified Harvard architectures along with the technical details. [5M]

(b).Explain any two applications of DSP processors. [5M]

2. (a) What are the current trends in digital signal processors? [5M]

(b).Write the applications of DSP processor. [5M]

3. Write the following:

(a)Spectral analysis of non-stationary signals. [5M]

(b)Over sampling D/A converter. [5M]

4. (a) Give the functional diagram of DSP processor TMS320C50 and explain the importance of each block briefly. [5M]

(b).List out the addressing modes in a DSP processor. [5M]

5. (a) With DFT sample figures explain dual tone multi frequency signal detection. [5M]

(b)Discuss the effect of over sampling A/D converter [5M]

6. (a) Explain the process of spectral analysis of non-stationary signals. [5M]

(b)Discuss the effect of over sampling D/A convertor. [5M]

7. Write short notes on:

(a).DSP controller. [5M]

(b).Musical sound processing [5M]

8. (a) Describe musical sound processing. [5M]

(b)Explain the following:

(i) Oversampling A/D converter. (ii) Oversampling D/A converter. [5M]

9. (a) Explain the architecture of TMS 320C54 DSP. [5M]

(b).Define multirate systems and sampling rate conversion [5M]

10.(a). Explain the effects of over sampling for ADC and DAC. [5M]

(b). With the help of block diagram explain the architecture of digital signal processors [5M]

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