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PRESIDENCY UNIVERSITY BENGALURU

 SCHOOL OF ENGINEERING

SUMMER TERM END TERM EXAMINATION - AUGUST 2024

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| **Semester : V** | **Date :05-AUG-2024** |
| **Course Code :EEE3014** | **Time :9:30AM-12:30PM** |
| **Course Name :** **Digital Signal Processing Systems** | **Max Marks :100** |
| **Program :B.Tech**  | **Weightage :50%** |

**Instructions:**

1. *Read all questions carefully and answer accordingly.*
2. *Question paper consists of 3 parts.*
3. *Scientific and non-programmable calculator are permitted.*
4. *Do not write any information on the question paper other than Roll Number.*

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| **PART A** |
|  **ANSWER ANY 4 QUESTIONS 4Q X 5M=20M** |
| 1 | Define Convolution. Write the applications of convolution | (CO 1) | [Knowledge] |
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| 2 | What is linear phase FIR filter? List the different ways of analyzing the FIR filter. | (CO 4) | [Knowledge] |
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| 3 | Write the differences between analog and digital filters.  | (CO 3) | [Knowledge] |
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| 4 | The DFT allows one to convert a set of digital time samples to its frequency domain representation. In contrast, the IDFT can be used to invert the DFT samples, allowing one to reconstruct the signal samples x(k) directly from its frequency domain form. Write the equations as per the statement for DFT and IDFT. | CO 2) | [Knowledge] |
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| 5 | Write the differences between IIR and FIR filters. | (CO 4) | [Knowledge] |
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| 6 | What are Infinite Impulse Response (IIR) filters, and how do they differ from Finite Impulse Response (FIR) filters? | (CO 3) | [Knowledge] |
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| **PART B** |
|  **ANSWER ANY 5 QUESTIONS 5Q X 10M=50M** |
| 7 | Mr. Rajesh wants to use a Chebyshev filters with the requirements as pass band attenuation of 3 dB at a pass band frequency of 1kHz, stop band attenuation of 16dB at 2kHz in power electronics for tasks such as harmonic filtering. Their steep roll-off helps in attenuating unwanted harmonic frequencies in power supply systems. Put yourself  in the place of Mr. Rajesh and design Chebyshev filter | (CO 3) | [Comprehension] |
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| 8 | For audio processing the given difference equation is used.y(n)=x(n)+3x(n-1)+2x(n-2)+3x(n-3)+x(n-4)obtain the non-recursive structure of y(n) using minimum multiplier. | (CO 4) | [Comprehension] |
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| 9 | Linear FIR (Finite Impulse Response) filtering is a commonly used technique in digital signal processing for filtering a long sequence of data. The long sequence is divided into overlapping blocks. Each block is then filtered using the FIR filter in the frequency domain. The filtered blocks are then added together, taking into account the overlap between adjacent blocks. Finally, the resulting sequence is obtained by discarding the overlapping portions. For the given signals use the overlap add procedure and estimate the resulting sequence.x(n) = {1,2,-1,2,3,-2,-3,-1,1,1,2,-1}h(n)= {1,1,1} | (CO 1) | [Comprehension] |
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| 10 | Linear FIR (Finite Impulse Response) filtering is a commonly used technique in digital signal processing for filtering a long sequence of data. The long sequence is divided into overlapping blocks. Each block is then filtered using the FIR filter in the frequency domain. The filtered blocks are then added together, taking into account the overlap between adjacent blocks. Finally, the resulting sequence is obtained by discarding the overlapping portions. For the given signals use the above mentioned procedure and estimate the resulting sequence.    | (CO 1) | [Comprehension] |
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| 11 | The Radix-2 FFT is extensively used in signal processing applications, such as audio and image processing, telecommunications, and various scientific and engineering fields. The following signal has been captured from the fault diagnosis. x(n) = (1,2,3,4,5,6,7,8). Using radix2-DIFFFT algorithm, compute Fourier Transform of x(n) sequence.  | (CO 2) | [Comprehension] |
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| 12 | In image and video processing, digital filters are applied for tasks such as blurring, sharpening, edge detection, and noise reduction. They contribute to enhancing image quality and extracting relevant features. For the analog transfer function H(s) = 1 / *(s+1)(s+2).*Evaluate the digital transfer function using Impulse Invariant method by considering T= 1 sec | (CO 3) | [Comprehension] |
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| 13 | In image and video processing, digital filters are applied for tasks such as blurring, sharpening, edge detection, and noise reduction. They contribute to enhancing image quality and extracting relevant features. For the analog transfer function H(s) = 2 / *(s+1)(s+2).*Evaluate the digital transfer function using bilinear transformation method by considering T= 1 sec | (CO 3) | [Comprehension] |
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| **PART C** |
|  **ANSWER ANY 2 QUESTIONS 2Q X 15M=30M** |
| 14 | Butterworth filters are employed in radar systems for tasks such as pulse shaping, target detection, and clutter rejection. Their smooth frequency response is beneficial in these applications and it is required to design an analog Butterworth filter for target detection that has a -2db pass band attenuation at a frequency of 20 rad/sec an at least 10 db stop band attenuation at 30 rad/sec. | (CO 3) | [Application] |
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| 15 | With the developments of data analysis technology, signal analysis methods become gradually mature and have been widely applied in the field of condition monitoring and fault diagnosis of various kinds of rotating machineries. The signal analysis could extract useful information from the original signal, and judge the operational states of the equipment based on the obtained information. Signal analysis is of great significance to the rotating machineries both for condition monitoring and fault diagnosis. The following signals has been captured from the fault diagnosis.x(n)={1,2,3,4,4,3,2,1} & x(k) = { 7, -0.707-j0.707, -j, 0.707-j0.707}List any two methods of analyzing the signals.Apply the listed methods one for each given signal to compute the signal with fastest computation techniques.         | (CO 2) | [Application] |
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| 16 | Design band pass filter whose desired frequency response is given asHd (ω) = **1**for  (− π /4) ≤ ω ≤  (3π/4)            = 0            otherwiseThe length of the filter should be 11 . Use rectangular window for the design of filter. Assume necessary data if required. Also find the frequency response. | (CO 4) | [Application] |
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