Code.No: **RR311102** | RR | SET-1

III B.TECH – I SEM EXAMINATIONS, NOVEMBER - 2010
DIGITAL SIGNAL PROCESSING
(COMMON TO BME, ECC)

Time: 3hours **Max.Marks:80**

Answer any FIVE questions
All questions carry equal marks
- - -

1.a) Define: (i) Signal ii) Signal processing
b) Discuss the basic elements in a digital processing system
c) Show that the even and odd parts of a real sequence are, even and odd sequences respectively. [2+8+6]

2.a) State and prove time and frequency shifting properties of Fourier transform.
b) Show that the frequency response of a discrete system is a periodic function of frequency. [8+8]

3.a) State and prove circular time shifting and frequency shifting properties of the DFT.
b) Compute the circular convolution of the following two sequences, using DFT approach

$$x_1(n) = \{1, 2, 0, 1\}$$
$$x_2(n) = \{2, 2, 1, 1\}$$
[8+8]

4.a) Implement the decimation in frequency FFT algorithm of N-point DFT where N=8.
b) Compute the FFT for the sequence, $x(n) = \{1, 1, 0, 0, 1, 1, 1, 0\}$ [8+8]

5.a) Determine the frequency response and magnitude response for the system:

$$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) - x(n-1).$$

b) Determine the signal x(n), where its z-transform is given by,

$$x(z) = \frac{z^2 + z}{\left(z - \frac{1}{2}\right)^2 \left(z - \frac{1}{4}\right)}$$
[6+10]

6.a) Give the relation between analog and digital filter poles in I I M transformation and explain.
b) Discuss the aliasing effect due to impulse invariance transformation.
c) Compare bi-linear and impulse invariance transformation methods. [4+8+4]

7.a) Using a rectangular window technique, design a low pass filter with pass band gain of unity, cut-off frequency of 1kHz and working at a sampling frequency of 5kHz. The length of the impulse response is 7.
b) Compare I I R filters and FIR filters. [10+6]

8.a) Design the following systems with minimum number of multipliers:

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

$$H(z)=\left[1+\frac{1}{2}z^{-1}+z^{-2}\right]\left[1+\frac{1}{4}z^{-1}+z^{-2}\right]$$

b) Write briefly about digital processing of speech. [8+8]

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SET-2

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DIGITAL SIGNAL PROCESSING
(COMMON TO BME, ECC)

Time: 3hours **Max.Marks:80**

Answer any FIVE questions
All questions carry equal marks
- - -

1.a) State and prove circular time shifting and frequency shifting properties of the DFT.

b) Compute the circular convolution of the following two sequences, using DFT approach

$$x_1(n) = \{1, 2, 0, 1\}$$
$$x_2(n) = \{2, 2, 1, 1\}$$

[8+8]

2.a) Implement the decimation in frequency FFT algorithm of N-point DFT where N=8.

b) Compute the FFT for the sequence, $x(n) = \{1,1,0,0,1,1,1,0\}$ [8+8]

3.a) Determine the frequency response and magnitude response for the system:

$$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) - x(n-1).$$

b) Determine the signal x(n), where its z-transform is given by,

$$x(z) = \frac{z^2 + z}{\left(z - \frac{1}{2}\right)^2\left(z - \frac{1}{4}\right)}$$

[6+10]

4.a) Give the relation between analog and digital filter poles in I I M transformation and explain.

b) Discuss the aliasing effect due to impulse invariance transformation.

c) Compare bi-linear and impulse invariance transformation methods. [4+8+4]

5.a) Using a rectangular window technique, design a low pass filter with pass band gain of unity, cut-off frequency of 1kHz and working at a sampling frequency of 5kHz. The length of the impulse response is 7.

b) Compare I I R filters and FIR filters. [10+6]

6.a) Design the following systems with minimum number of multipliers:

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

$$H(z) = \left[1 + \frac{1}{2}z^{-1} + z^{-2}\right]\left[1 + \frac{1}{4}z^{-1} + z^{-2}\right]$$

b) Write briefly about digital processing of speech. [8+8]

7.a) Define: (i) Signal ii) Signal processing
b) Discuss the basic elements in a digital processing system
c) Show that the even and odd parts of a real sequence are, even and odd sequences respectively. [2+8+6]

8.a) State and prove time and frequency shifting properties of Fourier transform.
b) Show that the frequency response of a discrete system is a periodic function of frequency. [8+8]

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SET-3

III B.TECH – I SEM EXAMINATIONS, NOVEMBER - 2010
DIGITAL SIGNAL PROCESSING
(COMMON TO BME, ECC)

Time: 3hours **Max.Marks:80**

Answer any FIVE questions
All questions carry equal marks
- - -

1.a) Determine the frequency response and magnitude response for the system:
$$y(n)-\frac{3}{4}y(n-1)+\frac{1}{8}y(n-2)=x(n)-x(n-1).$$

b) Determine the signal x(n), where its z-transform is given by,
$$x(z)=\frac{z^2+z}{\left(z-\frac{1}{2}\right)^2\left(z-\frac{1}{4}\right)}$$
[6+10]

2.a) Give the relation between analog and digital filter poles in I I M transformation and explain.

b) Discuss the aliasing effect due to impulse invariance transformation.

c) Compare bi-linear and impulse invariance transformation methods. [4+8+4]

3.a) Using a rectangular window technique, design a low pass filter with pass band gain of unity, cut-off frequency of 1kHz and working at a sampling frequency of 5kHz. The length of the impulse response is 7.

b) Compare I I R filters and FIR filters. [10+6]

4.a) Design the following systems with minimum number of multipliers:
$$H(z)=\frac{1}{4}+\frac{1}{2}z^{-1}+\frac{3}{4}z^{-2}+\frac{1}{2}z^{-3}+\frac{1}{4}z^{-4}$$
$$H(z)=\left[1+\frac{1}{2}z^{-1}+z^{-2}\right]\left[1+\frac{1}{4}z^{-1}+z^{-2}\right]$$

b) Write briefly about digital processing of speech. [8+8]

5.a) Define: (i) Signal ii) Signal processing

b) Discuss the basic elements in a digital processing system

c) Show that the even and odd parts of a real sequence are, even and odd sequences respectively. [2+8+6]

6.a) State and prove time and frequency shifting properties of Fourier transform.

b) Show that the frequency response of a discrete system is a periodic function of frequency. [8+8]

7.a) State and prove circular time shifting and frequency shifting properties of the DFT.

b) Compute the circular convolution of the following two sequences, using DFT approach

$$x_1(n) = \{1, 2, 0, 1\}$$
$$x_2(n) = \{2, 2, 1, 1\}$$

[8+8]

8.a) Implement the decimation in frequency FFT algorithm of N-point DFT where N=8.

b) Compute the FFT for the sequence, $x(n) = \{1,1,0,0,1,1,1,0\}$ [8+8]

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SET-4

III B.TECH – I SEM EXAMINATIONS, NOVEMBER - 2010
DIGITAL SIGNAL PROCESSING
(COMMON TO BME, ECC)

Time: 3hours **Max.Marks:80**

Answer any FIVE questions
All questions carry equal marks
- - -

1.a) Using a rectangular window technique, design a low pass filter with pass band gain of unity, cut-off frequency of 1kHz and working at a sampling frequency of 5kHz. The length of the impulse response is 7.

b) Compare I I R filters and FIR filters. [10+6]

2.a) Design the following systems with minimum number of multipliers:

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

$$H(z) = \left[1 + \frac{1}{2}z^{-1} + z^{-2}\right]\left[1 + \frac{1}{4}z^{-1} + z^{-2}\right]$$

b) Write briefly about digital processing of speech. [8+8]

3.a) Define: (i) Signal ii) Signal processing

b) Discuss the basic elements in a digital processing system

c) Show that the even and odd parts of a real sequence are, even and odd sequences respectively. [2+8+6]

4.a) State and prove time and frequency shifting properties of Fourier transform.

b) Show that the frequency response of a discrete system is a periodic function of frequency. [8+8]

5.a) State and prove circular time shifting and frequency shifting properties of the DFT.

b) Compute the circular convolution of the following two sequences, using DFT approach

$$x_1(n) = \{1, 2, 0, 1\}$$
$$x_2(n) = \{2, 2, 1, 1\}$$

[8+8]

6.a) Implement the decimation in frequency FFT algorithm of N-point DFT where N=8.

b) Compute the FFT for the sequence, $x(n) = \{1, 1, 0, 0, 1, 1, 1, 0\}$ [8+8]

7.a) Determine the frequency response and magnitude response for the system:

$$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) - x(n-1).$$

b) Determine the signal x(n), where its z-transform is given by,

$$x(z) = \frac{z^2 + z}{\left(z - \frac{1}{2}\right)^2\left(z - \frac{1}{4}\right)}$$

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8.a) Give the relation between analog and digital filter poles in I I M transformation and explain.

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