24 TRIBHUVAN UNIVERSITY

INSTITUTE OF ENGINEERING

Examination Control Division

2075 Bhadra

Exam.		Regular	
Level	BE	**Full Marks**	**80**
Programme	BEX	**Pass Marks**	**32**
Year / Part	IV / II	**Time**	**3 hrs.**

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. Describe Spectral properties of time domain sampling principle with a suitable example. List out the disadvantages of Digital Signal Processing. [4+2]

2. Define the recursive and non-recursive systems with examples. Find the output of LTI system having impulse response $h[n] = 2^n * \{u[n] - u[n-3]\}$ and input signal $x[n] = \delta[n] + \delta[n-1] - \delta[n-3]$. [4+6]

3. List out the properties of Region of convergence and locate the ROC of the following signal
   $x[n] = (0.25)^n u[n] + (0.6)^n u[-n-1]$ [4+6]

4. Why we need FFT? Draw the butterfly structure to compute X (4) and X (7) of 8-point DFT values of given sequence as $x[n] = 0.2*n$ using Radix-2 Decimation in Frequency algorithm. [2+8]

5. What do you mean by zero padding? If $X_1(k)$ and $X_2(k)$ are DFT of sequence $x_1[n] = (3)^n$, $0 \le n \le 3$ and $x_2[n] = (2)^n$, $0 \le n \le 4$ respectively, then find the sequence $x_3[n]$, if DFT of $x_3[n]$, is given by $X_3(k) = X_1(k)X_2(k)$. [1+5]

6. The Poles of a system are located at: 0.45 - 0.77i and -2 ±0.3i and zeroes at: 1.2 ± 3i. Map the poles and zero in the z-plane and plot the magnitude response of the system. [2+6]

7. Draw lattice-ladder structure for given IIR system
   $H(z) = (0.6 - 0.45z^{-1} - 0.25z^{-2})/(1 + 0.27z^{-1} + 0.06z^{-2} - 0.75z^{-3})$. Also check the stability of given system. [7+1]

8. Design an analog filter using Chebyshev approximation which has maximum passband attenuation of 5dB at frequency 20 rad/sec and stopband attenuation of 20dB at frequency 50 rad/sec. Taking sampling frequency of 0.5 Hz convert the obtained analog filter to digital by using Bilinear transformation method. [10]

9. Find out the first three coefficients of impulse response of a low pass FIR filter having Pass band edge frequency $\omega_p = 0.24\pi$ radian, Stop band edge frequency $\omega_s = 0.54\pi$ radian and Stop band attenuation $\alpha_s = 42.2$ dB using any appropriate window function. [6]

10. Write short notes on bit-serial arithmetic and pipelined implementation of DSP processors. [6]

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Exam.	**Regular**		
Level	BE	**Full Marks**	80
Programme	BEX	**Pass Marks**	32
Year / Part	IV / II	**Time**	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. What are the basic elements of Digital Signal Processing (DSP) system? Explain Analog to Digital conversion process in brief. [3+4]
2. Define recursive and non recursive system with the suitable examples. State the condition for the stability and causality of LTI system in terms of ROC and pole zero. [3+3]
3. Impulse response of an LTI system is given as $h(n) = \{2,2,-1,1\}$. Find the output y(n) of the system to an input $x(n) = \{1,2,1,2\}$. [6]
4. Define Z-transform. How is it related to DTFT? Explain scaling property of Z-transform with suitable example. [3+3]
5. Find the inverse Z-transform of $X(z) = (2z^2 + 2z^2 + 3z + 5) / (z^2 - 0.1z - 0.2)$, ROC: $|z| < 0.4$ [5]
6. Discuss the computational efficiency of FFT algorithm. Find DFT of a sequence given a $x[n] = \{1,1,0,0,0\}$ using DIT FFT algorithm. [2+5]
7. What is zero padding? Find the linear convolution through circular convolution with padding of zeros for the following sequence: $x[n] = \{-1,1\}$ and $h[n] = \{2,3,1,-2\}$. [2+6]
8. Write short notes on: [2+2]

   i) IIR filter

   ii) Kaiser window
9. Draw lattice-ladder structure for given IIR system. [5+1]

   $$H(z) = (0.62 + 0.42z^{-1} - 0.25z^{-2})/(1 + 0.27z^{-1} + 0.06z^{-2} - 0.75z^{-3})$$

   Also check the stability of given system.
10. Design a digital low-pass filter with the following specification: [9]

    i) Pass-band magnitude characteristics constant to 0.7 dB below the frequency of 0.15π

    ii) Stop-band attenuation of at least 14 dB for the frequencies between 0.6π to π.

    Use Butterworth approximation as a prototype and use bilinear transformation method to obtain the digital filter.
11. Why Remez exchange algorithm is required? Describe Remez exchange algorithm with a flow chart. [5]
12. What is Gibb's phenomena? Explain how it occurs while designing FIR filter based on windowing technique. How can it be minimized? [1+4]
13. What are the various DSP processor chips? Explain Bit-Serial implementation in DSP architecture. Why is it used? [2+4]

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2074 Magh

Exam.		Back	
Level	BE	**Full Marks**	80
Programme	BEX	**Pass Marks**	32
Year / Part	IV / II	**Time**	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. Consider the analog signal $x_a(t) = 3\cos 100\pi t$. [7]
   i) Determine the minimum sampling rate required to avoid aliasing.
   ii) Suppose that signal is sampled at Fs = 75Hz. What is the discrete time signal obtained after sampling?
   iii) Assume now that we sample this signal using a sampling rate Fs = 5000 samples/sec. What is discrete time signal after sampling?
2. Differentiate between energy and power, period and a periodic, symmetric and anti-symmetric signals. Check whether the following signal is linear or non-linear: $y[n] = x^2[n]$. [3+3]
3. Find the response of an FIR system with input $x[n] = \{0,2,4,6\}$ and impulse response $h[n] = \{5,3,4,2,0\}$. [5]
4. Write the properties for Region of Convergence (ROC). A discrete-time signal is expresses as $x[n]=\cos\omega n$ for $n\geq 0$. Find its Z-transform. [3+8]
5. Differentiate between DFT and DTFT. Find the circular convolution of $x_1[n] = \{2,1,2,1\}$ and $x_2[n] = \{1,2,3,4\}$ [1+5]
6. Find 8 point DFT of {1,1,0,-1,2} using DFTFFT algorithm. [7]
7. Draw cascaded and parallel form structure of [7]

$$H(z) = \frac{5(1-0.25z^{-1})(1-0.667z^{-1})(1+2z^{-1})}{(1-0.75z^{-1})(1-0.125z^{-1})[1-(0.5+j0.5)z^{-1}][1-(0.5-j0.5)z^{-1}]}$$

8. Given three-stage lattice filter with coefficients K1=1/4, K2=1/4 and K3=1/3. Determine the FIR filter coefficients for direct-form structure. [7]
9. Design a discrete time low pass filter by applying bilinear transformation method to approximate a Chebyshev type I filter if the pass brand frequency is 0.2π radians and maximum deviation of 1dB below 0 dB gain in the pass band. The maximum gain of -15dB and frequency is 0.3π radians in the stop band. Consider sampling frequency of 1Hz. [9]
10. Explain the FIR filter design using the Remez exchange algorithm. [9]
11. Explain Bit Serial Arithmetic with suitable diagram. [6]

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2073 Bhadra

Exam.	Regular		
Level	BE	Full Marks	80
Programme	BEX	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

Subject: - Digital Signal Processing *(EX 753)*

✓ Candidates are required to give their answers in their own words as far as practicable.
✓ *Attempt* ***All*** *questions.*
✓ *The figures in the margin indicate* ***Full Marks.***
✓ *Assume suitable data if necessary.*

1. Explain the general application areas of digital Signal Processing. Consider a continuous time signal $x(t) = \sin 2000\pi t + 5\cos 12000\pi t + 10\sin 6000\pi t$.

   a) What is the discrete time signal obtained after sampling the signal at a sampling rate of 5000 samples per second for $0 \le n \le 3$? [3+4]

2. Define causal and stable system with examples. [3]

3. Find a convolution between two signals x[n] and h[n] where,

$$x[n] = \begin{cases} 1 & 0 \le n \le 4 \\ 0 & \text{otherwise} \end{cases} \text{ and } h[n] = \begin{cases} a^n & 0 \le n \le 6 \\ 0 & \text{otherwise} \end{cases} \quad (a > 1).$$ [8]

4. Using long division determine the inverse Z-transform of;

$$X(z) = \frac{1}{1 - \left(\frac{3}{2}\right)z^{-1} + \left(\frac{1}{2}\right)z^{-2}} \text{ when, ROC : } |Z| > 1 \text{ and ROC:} |Z| < \frac{1}{2}$$ [6]

5. Define ROC. Explain the properties of ROC with suitable examples. [1+4]

6. Why we need a DFT? Find 8-point DFT of sequence x[n] = {1,0,2,0,-1,1,1} using Decimation in Time Fast Fourier Transform (DITFFT) algorithm. [1+6]

7. State Multiplication of two DFTs property of DFT. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k)X_2(k)$ where $X_1(k)$ and $X_2(k)$ are 5-point DFT of $x_1[n]$ = {1,2,3,-1,5} and $x_2[n]$ = {2,1,-3} respectively. [1+5]

8. For a system with poles at $0.45 \pm j1.06$ and zero at $0.58 \pm j2.06$. Plot the location of poles and zeroes in the z-plane and also plot the magnitude response of the system. [8]

9. Convert the following filter into a lattice ladder structure.
   $H(z) = 1 + 2z^{-1} + 2z^{-2} + z^{-3}$ [6]

10. Design a low pass digital filter by Bilinear Transformation method to an approximate Butterworth filter, if passband edge frequency is 0.25π radians and maximum deviation of 0.99 dB below 0 dB gain in the passband. The maximum gain of -14.85 dB and frequency is 0.59π radians in stopband, Consider sampling frequency 0.5 Hz. [11]

11. List out the key points of windowing and design the symmetric FIR low pass filter for which desired frequency response is expressed as

$$H_d(\omega) = \begin{cases} e^{-j\omega\tau} & \text{for} |\omega| \le \omega_c \\ 0 & \text{elsewhere} \end{cases}$$

The length of the filter should be 7 and ωc = 1 radians/sample. Use Hanning window as a prototype. [2+7]

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Exam.	New Back (2066 & Later Batch)		
Level	BE	Full Marks	80
Programme	BEX	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

Subject: - Digital Signal Processing *(EX 753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. Describe the different application of a digital signal processing. [3]

2. Consider a continuous time signal $x(t) = \cos 1000\pi t + 3\sin 3000\pi t + 5\cos 5000\pi t$. Calculate the discrete time signal for sampling rate of 6000 samples per second for $0 \le n \le 3$. [4]

3. Define a Causal and Stable system in terms of impulse response. Find the output of LTI system having impulse response $h[n] = u[n+1]-u[n-4]$ and input signal $x[n] = (1/2)^n u[n]$. [2+6]

4. What is a convolution sum and derive its equation? [3]

5. Define a Region of Convergence (ROC). Given a discrete time signal $x[n] = (0.5 + j0.2)^n u[n] + (-j)^n u[-n-1]$. Find X(z) at z = 0.4 and z = 0.7; where x(z) is bilateral Z-transform of x[n]. [1+4]

6. Explain one sided Z-transform with its importance. Explain the time delay property of one sided Z-transform. [3+3]

7. Discuss the computional efficiency of FFT algorithm. Find DFT of a sequence given as $x[n] = \{1, -1, 2\}$ using DIT FFT algorithm. [2+5]

8. Find $x_3[n]$ if DFT is given by $x_3(k) = x_1(k)x_2(k)$. where $x_1[n] = \{3, 2, 1, -1, -2\}$ and $x_2[n] = \{1, 2, 3\}$ for 5 points DFT. [6]

9. Plot Magnitude Response (not to the scale) of the system described by difference equation [7]

   $$y[n] - 0.3\, y[n-1] + 0.2y[n-2] = x[n] + 0.7x[n-1]$$

10. Draw lattice-ladder structure for given IIR system

    $H(z) = (0.5 + 0.65z^{-1} + 0.2z^{-2}) / (1 - 0.45z^{-1} + 0.3z^{-2} + 0.5z^{-3})$. Also check the stability of given system. [6+1]

11. Design a low pass Butterworth filter using bilinear method to meet the following specifications: [11]

    Maximum deviation of 0.98dB below 0dB gain in the pass band.
    Stop band attenuation = 20dB
    Pass band edge frequency = 0.2π radian
    Stop band edge frequency = 0.5π radian
    Sampling frequency = 1Hz

12. Explain a Remez exchange algorithm for FIR filter design. Design the FIR filter using suitable window for the following specifications: [4+5]

    $$0.899 \le \left|H(e^{jw})\right| \le 1; \quad \text{for } |w| \le 0.2\pi$$

    $$\left|H(e^{jw})\right| \le 0.01; \quad \text{for } 0.4\pi \le |w| \le \pi$$

13. Where are Digital Signal Processor Chips used? What are the General and special

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Exam.	**Regular**		
Level	BE	**Full Marks**	80
Programme	BEX	**Pass Marks**	32
Year / Part	IV / II	**Time**	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
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1. What are the applications of digital signal processing? Consider the analog signal, $x(t) = 2\cos 3000\pi t + 3\sin 4000\pi t + 7\cos 6000\pi t$. If sampling rate is 8000 samples per second and quantized at 8 bits, find: [2+5]
   i) Discrete values at any two points
   ii) Quantization errors at those points
2. Explain the properties of LTI systems with suitable examples. [8]
3. Find inverse Z-transform of: [5]

   $X(z) = (z^4 + 2z^3 - z + 4)/(z^2 - 1.5z - 1)$, ROC: $|z| < 0.5$
4. State and prove convolution property of z-transform. Describe causality and stability of system in terms of ROC with suitable examples. [1+3+3]
5. Find the linear convolution of $x_1[n] = \{1,1,1\}$ and $x_2[n] = [2,2,2]$ using circular convolution method. [5]
6. How fast is FFT? Use the FFT Algorithm to compute IDFT of a sequence given by $X(k) = \{6, -2+2j, -2-2-2j\}$ [2+6]
7. Plot Magnitude Response (not to the scale) of the system described by difference equation. $y[n] - 0.4y[n-1] + 0.25y[n-2] = x[n] + 0.5x[n-1]$ [7]
8. Compute Lattice coefficients and draw Lattice structure for given IIR system

   $H(z) = 1/(1 - 0.525z^{-1} + 0.6125z^{-2} + 0.3z^{-3})$. Also check the stability of given system. [4+2+1]
9. Design a low pass digital filter by Bilinear Transformation method to an approximate Butterworth filter, if passband edge frequency is 0.24 π radians and maximum deviation of 1 dB below 0 dB gain in the passband. The maximum gain of -14.9 dB and frequency is 0.57 π radians in stopband, consider sampling frequency 0.5 Hz. [12]
10. Define Gibbs phenomena in FIR filter design. Explain the steps to design the FIR filter using Kaiser Window. [2+7]
11. Explain Bit Serial Arithmetic in Digital Signal Processor. [5]

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Exam.	**New Back (2066 & Later Batch)**		
Level	BE	**Full Marks**	80
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Year / Part	IV / II	**Time**	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
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1. List the applications of Digital Signal Processing consider the analog signal $x(t) = 3 \cos 2000 \pi t + 5 \sin 4000 \pi t + 8 \cos 7000 \pi t$. If sampling rate is 10,000 samples per second and quantized at 10 bits: [2+5]
   i) Plot any two discrete values
   ii) Quantization errors at two points
2. Find the output of LTI system having impulse response $h[n] = 2u[n]-2u[n-5]$ and input signal $x[n]= (1/3)^n u[n]$. [8]
3. Determine the inverse Z-transform of, $X(z) = \dfrac{1}{1-1.5z^{-1}+0.5z^{-2}}$ [6]

   When,

   (a) ROC : $|Z| > 1.5$ (b) ROC : $|Z| < 0.5$
4. Define ROC in Z transform. State and prove the Convolution property of Z-transform. [2+4]
5. Why we need FFT? Find 8-point DFT of sequence $x[n] = \{1,0,0,0,-1,1,1)$ using Decimation in Frequency Fast Fourier Transform (DIFFFT) algorithm. [1+6]
6. Find $x_3[n]$ if DFT of $x_3[n]$ is given by $X_3(k) = X_1(k)\ X_2(k)$ where $X_1(k)$ and $X_2(k)$ are 4-point DFT of $x_1[n]=\{1,2,3,-1\}$ and $x_2[n]=\{2,1,-3\}$ respectively. [2+4]
7. Compute and draw the lattice structure for the following system: [8]

   $$H(z) = 2.3 + \frac{3.6}{24}Z^{-1} + \frac{101}{8}Z^{2} + \frac{2}{3}Z^{-3}$$
8. Define limit cycle oscillation. Explain it taking an example of ideal single pole recursive system. [5]
9. Design the Chebysev filter using bilinear transformation to meet the following specification. [11]

   $$0.707 \le |H(e^{jw})| \le 1; \qquad \text{for} |w| \le 0.2\pi$$

   $$|H(e^{jw})| \le 0.1; \qquad \text{for} 0.5\pi \le |w| \le \pi$$
10. How FIR filter can be designed using windowing method? Describe with mathematical expression. [4]
11. Describe Remez exchange algorithm for FIR filter design with a flow chart. [6]
12. Explain about bit serial arithmetic. [6]

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2071 Bhadra

Exam.	Regular / Back		
Level	BE	Full Marks	80
Programme	BEX	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
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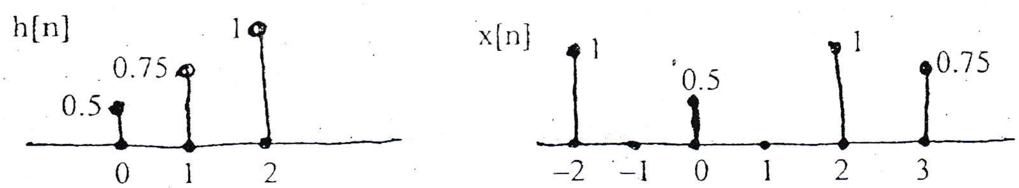
1. What are the advantages of DSP? Explain the significance of A/D and D/A conversion in DSP. [3+4]

2. Determine whether the following systems are (a) casual (b) linear. [4]

   a) $y[n] = x[n] - 3x[n-1]$
   b) $y[n] = x[n+1] + 4x[n]$

3. Find the output sequence y[n] and verify it for [7]



4. Explain the properties of ROC of z-transform. [5]

5. Find inverse Z-transform of [6]

   $X(z) = (z^4 + z^3 - 3z + 5) / (z^2 - 1.5z - 1)$, ROC: $|z| < 0.5$

6. Perform circular convolution of: [5]

   $x_1[n] = [1\ 2\ 0\ 3\ 4\ ]$, $x_2[n] = [2\ -1\ 2\ -1\ 2]$

7. Find DFT for {1, 1, 2, 0, 1, 2, 0, 1} using FFT and plot the spectrum. $x(0)=8$ [8]

8. Plot Magnitude Response (not to the scale) of the system described by difference equation. [6]

   $y[n] - 0.3y[n-1] + 0.225y[n-2] = x[n] - 0.4x[n-1]$

9. Compute Lattice coefficients and draw lattice structure for given IIR system $H(z) = 1/(1 - 0.3z^{-1} + 0.5z^{-2} + 0.25z^{-3})$. [6+2]

10. Design a low pass digital filter by Bilinear Transformation method to an approximate Butterworth filter, if passband edge frequency is 0.25 π radians and maximum deviation of 1 dB below 0 dB gain in the passband. The maximum gain of −15 dB and frequency is 0.55π radians in stopband, consider sampling frequency 0.5 Hz. [9]

11. Explain about the Gibb's phenomenon. Explain Remez Exchange algorithm with flowchart. [1+8]

12. Explain the Bit-serial arithmetic with suitable diagram. [6]

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2070 Bhadra

Exam.	Regular		
Level	BE	Full Marks	80
Programme	BEX	Pass Marks	32
Year / Part	IV / II	Time	3 hrs.

Subject: - Digital Signal Processing *(EX 753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. Explain Digital Signal Processing with its advantages and applications. [7]

2. Determine whether the following systems are (a) casual (b) linear and (c) time invariant. [5]

   a) $y(n) = \log_{10}[\{x(n)\}]$

   b) $y(n) = x(-n-2)$

3. Illustrate the significance of convolution summation in digital signal analysis. Find the output sequence y(n) if: [2+4]

   $h(n) = \{1,1,1\}$ and $x(n) = \{1-2,2,3,4\}$

4. Define region of convergence with its properties. [5]

5. Determine the causal signal x[n] if its z-transform X(z) is given by $X(z) = \dfrac{1+2z^{-1}+z^{-2}}{1+4z^{-1}+4z^{-2}}$. [6]

6. Compute the 8 point DFT of the sequence x(n) = {0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0} using radix-2 DIF algorithm. [8]

7. How the computational complexity of FFT is reduced compared to DFT? Explain with suitable derivations. [5]

8. Show how to use a lattice structure to implement the following all pass filter $H(z) = \dfrac{1+0.4z^{-1}-1.2z^{-2}+2z^{-3}}{2-1.2z^{-1}+0.4z^{-2}+z^{-3}}$. [8]

9. What are the Round-off effects in Digital Filters? Explain Limit Cycle Oscillation with an example. [2+4]

10. Design a digital low pass butterworth filter by applying bilinear transformation technique for the given specifications. [9]

    Pass band edge = 120 Hz
    Pass band attenuation = 1 dB
    Stop band edge = 170 Hz
    Stop band attenuation = 16 dB

    Assume sampling frequency of 256 Hz.

11. Draw the flow chart of Remez exchange algorithm for FIR filter design. [4]

12. ~~Explain window method of FIR filter design.~~ [5]

13. Explain Bit-serial arithmetic implementation. [6]

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Year / Part	IV / II	Time	3 hrs.

Subject: - Digital Signal Processing *(EX753)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

1. What are the advantages of DSP over analog signal processing? Explain the significance of A/D and D/A conversion in DSP. [3+4]
2. Explain the conditions for an LTI/LSI system to be causal and BIBO (Bounded Input Bounded Output) Stable. [6]
3. Draw the recursive form of a moving average system. [5]
4. Define one side Z-transform. Find Z-transform of the following signal using properties: $x[n] = na^n u[n]$. Also mention ROC of this signal. [5]
5. Determine the inverse z-transform of $H(z) = \dfrac{1+2z^{-1}+z^{-2}}{1+4z^{-1}+4z^{-2}}$ [6]
   Use long division method for possible ROCs.
6. Define DFT. Compute 4-point DFT of the following sequence: $x[n] = \{2,0,1,2\}$ [7]
7. How do you use DIT FFT Algorithm to compute 8 point DFT of a sequence? Explain with flow graph (structure) of the algorithm. [6]
8. Compute the lattice coefficients and draw the lattice structure of following FIR system $H(z) = 1/(1+2z^{-1}-3z^{-2}+4z^{-3})$. [6]
9. Given H(z) for a system with the following difference equation: [8]
   $$y(n) = x(n) + x(n-2)$$
   Determine its magnitude and phase response. Also, determine whether system is causal and stable.
10. Why do we use transformation of a continuous time filter design methods to design a discrete time IIR filters? Explain IIR filter design using bilinear transformation with its advantages and disadvantages. [3+6]
11. Design a low pass discrete time Chebyshev filter applying bilinear transformation having specifications as follows: [9]
    - Pass band frequency $(W_p) = 0.25\pi$ radians
    - Stop band frequency $(W_s) = 0.55\pi$ radians
    - Minimum attenuation $(\alpha_{min}) = 13.55$ dB
    - Maximum attenuation $(\alpha_{max}) = 1.01$ dB

    Consider sampling frequency 0.5 Hz.
12. Write down the features of special purpose DSP processor. [6]

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Examination Control Division.
2069 Bhadra

Exam.		**Regular / Back.**	
Level	BE	**Full Marks**	80
Programme	BEX	**Pass Marks**	32
Year / Part	IV / II	**Time**	3 hrs.

Subject: - Digital Signal Processing *(EG773EX)*

- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt* ***All*** *questions.*
- ✓ *The figures in the margin indicate* ***Full Marks***.
- ✓ *Assume suitable data if necessary.*

1. What is a convolution summation? Derive its equation and explain it. [1+3]

2. Why do you need different equation? Consider an LTI system with impulse response $h[n] = (1/2)^n u[n]$. Determine y[n], the output of this system, if the input is $x[n] = Ae^{jn\pi/2}$ [2+4]

3. An LTI system is characterized by the system function: [6]

$$H(z) = \frac{(1-1/2\,Z^{-2})}{(1-1/2\,Z^{-1})(1-1/4\,Z^{-1})} \quad |Z| > 1/2$$

Determine the impulse response of the system.

4. Explain how the poles and zeros of H(z) affect the stability and the gain response of a system. Given H(z) for a digital signal processing system with the following difference equation: $y[n]=x[n]+1.21x[n-2]-0.8y[n-1]$ [3+3+2+4]
Plot its poles and zeros on the Z-plane, determine whether it is causal and stable and sketch its gain response.

5. For the given system, $H[z] = \dfrac{1}{1-0.9Z^{-1}+0.64Z^{-2}-0.576Z^{-3}}$ [8]
Compute and draw the lattice structure.

6. Why do you need anti-aliasing filter? Describe the effect of sample and hold circuit at the input of the A/D conversion of Discrete Time Processing of continuous time signal. [2+5]

7. Why is Remez exchange algorithm is generally considered superior to the windowing method as a design technique for digital FIR filters? Explain about Remez exchange algorithm with suitable derivation and flowchart. [3+10]

8. Design a low pass discrete time filter by applying impulse invariance to an approximate Butterworth continuous time filter, if passband frequency is 0.25π radians and maximum deviation of 0.5 dB below 0 dB gain in the passband. The maximum gain of -15 dB and frequency is 0.55π radians in the stopband. Consider sampling frequency 1 Hz. [12]

9. Define one's complement, 2's complement and sign magnitude representation of numbers. Represent -192/220 in 8 bit 1's complement and 2's complement form. [3+4]

10. Find the FFT of the signal x[n]=(3.6, 5.5, 3.3, 6.3) [5]

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