TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2081 Chaitra

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. Consider the analog signal $x_a(t) = 3\cos 100\pi t$ [1+3+3]

   a) Determine the minimum sampling rate required to avoid aliasing.

   b) Suppose that signal is sampled at $F_s = 75$Hz. What is the discrete time signal obtained after sampling?

   c) Assume now that we sample this signal using a sampling rate $F_s = 5{,}000$ samples/sec. What is discrete time signal after sampling?

2. Define a Causal and Stable system in terms of impulse response. Find the output of LTI system having impulse response $h[n] = 2u[n]-2u[n-5]$ and input signal $x[n]=(1/3)^n u[n]$. [2+2+6]

3. Determine the inverse Z-transform of [6]

$$X(z) = \frac{(z^4 + z^3 - 3z + 5)}{z^2 - 1.5z - 1}$$

   When ROC: $|z| < 0.5$

4. Why we need FFT? Compute the 8-point DFT of sequence x(n) = {-0.5,- 0.5, 0.5, 0.5, 0,0, 0, 0) using Decimation in Frequency Fast Fourier Transform (DIFFFT) algorithm. [2+6]

5. What do you mean by zero padding? 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 $x_1[n] = (3)^n$, $0 \le n \le 3$ and $x_2[n] = (2)^n$, $0 \le n \le 4$ respectively. [1+6]

6. 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]

7. 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]

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 I Hz. [12]

9. Define Gibbs phenomena in FIR filter design. Design the FIR filter using suitable window for the following specifications: [2+6]

$$0.899 \le |H(e^{jw})| \le 1 \qquad for\ [w] \le 0.2\pi$$
$$|H(e^{jw})| \le 0.01 \qquad for\ 0.4\pi \le |w| \le \pi$$

10. What are the Round-off effects in Digital Filters? Explain Limit Cycle Oscillation with an example. [3+4]

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TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2080 Chaitra

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. Define causal and stable system with examples. Consider a continuous time signal $x(t) = \sin 2000\pi t - 5\cos 12000\pi t + 10\sin 6000\pi t$.
   What is the discrete time signal obtained after sampling the signal at a sampling rate of 5000 samples per second for $0 <= n <= 3$? [3+4]
2. Define a Causal and Stable system in terms of impulse response. Find the output of LTI system having impulse response $h[n] = 2u[n]- 2u[n-5]$ and input signal $x[n]=(1/3)^n u[n]$. [2+2+6]
3. Determine the causal signal x[n] if its z-transform X(z) is given by [6]
$$X(z) = \frac{1 + 2z^{-1} + z^{-2}}{1 + 4z^{-1} + 4z^{-2}}$$
4. 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+7]
5. 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. [7]
6. Plot Magnitude Response (not to the scale) of the system described by difference equation. [8]
   $y[n] - 0.4 y[n-1] + 0.25y[n-2] = x[n] + 0.5x[n-1]$
7. Given three-stage lattice filter with coefficients $K_1=1/4$, $K_2=1/4$ and $K_3=1/3$. Determine the FIR filter coefficients for direct-form structure. [7]
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 I Hz. [12]
9. Explain a Remez exchange algorithm for FIR filter design. Explain the steps to design the FIR filter using Kaiser Window. [5+4]
10. What are the Round-off effects in Digital Filters? Explain Limit Cycle Oscillation with an example. [2+4]

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TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2079 Chaitra

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. What are the applications of digital signal processing? Consider a continuous time signal $x(t) = 2\cos 3000\pi t + 3\sin 4000\pi t + 7\cos 6000\pi t$. If the sampling rate is 8000 samples per second and quantized at 8 bits, find:

   a) Discrete values at any two points.

   b) Quantization error at those points. [1+3+3]

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

3. Find the inverse Z- transform of:

   $X(z) = (z^4-2z^3-z+4) / (z^2-1.5z-1)$ ROC: $|z|< 0.5$ [6]

4. Why do we need Discrete Fourier Transform (DFT) although we have Discrete-time Fourier Transform (DTFT)? Find the circular convolution between $x[n] = \{1, 2\}$ and $y[n] = u[n] - u[n-4]$ [3+5]

5. Explain the Fast Fourier transform algorithm. Discuss the advantages of the Fast Fourier transform. [4+2]

6. Compute Lattice coefficient and draw Lattice structure for given IIR system

   $H(z) = 1/(1-0.9z^{-1}+0.64z^{-2}-0.576z^{-3})$. Also check the stability of given system. [4+2+1]

7. Design a low pass discrete IIR filter by Bilinear Transformation method to an approximate Butterworth filter, if pass band edge frequency $(\omega_p) = 0.24\pi$ radians and maximum deviation of 1 dB below 0 dB gain in the pass band. The maximum gain of -14.9 dB and Stop band edge frequency $(\omega_s) = 0.57\pi$ radians. Consider the sampling frequency of 0.5 Hz. [15]

8. Find out the first three coefficients of impulse response of a low pass FIR filter having Pass band edge frequency $\omega_p = 0.24\pi$, Stop band edge frequency $\omega_s = 0.54\pi$ and Stop band attenuation $\alpha_s = 42.2$ dB using any suitable window function. [8]

9. What is Gibb's phenomena? Explain how it occurs while designing FIR filter based on windowing technique. How can it be minimized? [2+5]

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

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TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
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2078 Chaitra

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. List out the advantages of digital signal processing over analog signal processing. Consider the analog signal $x(t) = 3\cos(100\pi t)$:
   a) Determine the minimum sampling rate required to avoid aliasing.
   b) If the signal is sampled at the rate $F_s = 200$ Hz, what is the discrete time signal obtained after sampling? [2+2+3]
2. Determine whether the following system: $y[n] = y[n-4] + x[n-4]$ is time invariant or not. [3]
3. Find the output of LTI system having impulse response $h[n] = u[n+1] - u[n-3]$ and input signal $x[n] = (0.5)^n u[n]$. [6]
4. State the properties of region of convergence (ROC). Obtain the z-transform of the following signal: $x[n] = na^n u[n]$. [2+4]
5. Find the signal x[n] using inverse of z-transform of $X(z) = z^2(1 - 1.5z^{-1})(1 - z^{-1})(1 + z^{-2})$. [4]
6. Find the circular convolution for $x_1[n] = \{2, 1, 2, 1\}$ and $x_2[n] = \{1, -2, -1, 3\}$. [5]
7. Find the 8-point DFT of the sequence given by $x[n] = 0.5\sin\left(\frac{n\pi}{6}\right)$ using DIFFFT algorithm. [7]
8. A system has an impulse response $h[n] = (0.5)^n u[n] + n(0.2)^n u[n]$. Determine the parallel from realization. [6]
9. Synthesize the IIR filter with transfer function $H(Z) = \dfrac{1 - z^{-1} + 0.5z^{-2}}{1 + 0.2z^{-1} - 0.15z^{-2}}$ using a lattice-ladder network. Also comment on the stability of the filter. [6+2]
10. Using Bilinear transformation, design a Butterworth filter which satisfies the following conditions: [10]

    $0.82 \le |H(e^{jw})| \le 1;\ 0 \le \omega \le 0.22\pi$

    $|H(e^{jw})| \le 0.18;\quad 0.58\pi \le \omega \le \pi$
11. Design a FIR linear phase filter using Kaiser window to meet the following specifications: [8]

    $0.98 \le |H(e^{jw})| \le 1.02;\ 0 \le \omega \le 0.19\pi$

    $|H(e^{jw})| \le 0.02;\quad 0.21\pi \le \omega \le \pi$
12. Explain Remez exchange algorithm for FIR filter design. [4]
13. Explain bit-serial arithmetic and distributed arithmetic of DSP processors. [3+3]

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TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2077 Chaitra

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 basic elements of Digital Signal Processing with necessary block diagram. Also list out the general application areas of Digital Signal Processing. [4+2]
2. Define causality and stability of LTI system in terms of impulse response with a suitable example. Find the output of LTI system having impulse response $h[n] = 2\ u[n+3] - 2\ u[n-4]$ and input signal $x[n] = (1/2)^n\ u[n+1]$. [3+7]
3. List out the properties of Region of Convergence. [4]
4. Find the inverse z-transform for $(z) = \frac{1}{1-0.5z^{-1}+1.5z^{-2}}$, ROC: $|z| > 1$ using long division method. [6]
5. Find the 8-point DFT values of given sequence as $x[n] = \text{Sin}\ (3\pi n/8)$ using Radix-2 Decimation in Frequency algorithm. [8]
6. Find the linear convolution through circular convolution of the following sequences $x[n] = u[n] - u[n] -[n-3]$ and $h[n] = \delta\ [n] - 3\delta\ [n-2]$. Define Parseval's theorem. [6+2]
7. Draw cascaded structure of the system given by
$$(z) = \frac{2(1-0.4z^{-1})(1-0.5z^{-1})}{(1+0.2z^{-1})(1-0.25z^{-1})[1-(-0.5+j0.15)z^{-1}][1-(-0.5-j0.15)z^{-1}]}.$$ [7]
8. Draw the lattice ladder structure for the IIR system given by $H(z) = \frac{0.5-0.4z^{-1}-0.2z^{-2}}{1+0.2z^{-1}+0.7z^{-2}-0.75z^{-3}}$. Also comment on the stability of given system. [7+2]
9. Design a digital low-pass Butterworth filter using Bilinear Transformation method technique whose Passband frequency is 0.2π rad and maximum deviation of 1 dB below 0 dB in Passband. There is a maximum gain of -15 dB in Stopband at frequency 0.3π rad. Assume sampling frequency = 1 Hz. [10]
10. Why Remez exchange algorithm is required? Describe Remez exchange algorithm with mathematical derivation. [1+5]
11. Explain Bit-serial arithmetic used in DSP implementations. [6]

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TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2076 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 *(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. List out the major disadvantages of Digital Signal Processing. When does oversampling occur? [2+1+1+2+2]

   Consider the analog signal:

   $x(t) = 3\cos 50\pi t + 10\sin 300\pi t - 0.7\cos 100\pi t$

   a) What is the Nyquist rate for this signal?

   b) Suppose that the signal is sampled at the rate of Fs=5000 samples per second, what is the discrete-time signal obtained after sampling?

   c) What is the analog signal we can reconstruct from the samples?

2. Differentiate between FIR and IIR system. Find the output of LTI system having impulse response $h[n]=2\,u[n+2] - 2u[n-3]$ and input signal $x[n] = (1/3)^n\, u[n-1]$. [2+6]

3. List out the properties of z-transform. Using long division, determine the inverse Z-transform of, $X(Z)=\dfrac{1}{1-1.58z^{-1}+0.5z^{-2}}$ when ROC : $|Z|>1$ and ROC : $|Z|<\dfrac{1}{2}$. [2+6]

4. Why do we need FFT? Determine the 4-point DFT of the following sequence of $x[n]=\{0.5, 0.5, 0.5, 0.5\}$ using radix-2 decimation in time FFT algorithm. [2+5]

5. Determine the circular convolution of the following sequences $x_1[n] = \{1, 0.5, 1, 0.5\}$ and $x_2[n]=\{0, 1, 2, 3\}$. [6]

6. Show the pole-zero in z-plane and plot magnitude response (not to the scale) of the system described by difference equation. [2+5]

   $$y[n] - 0.6\,y[n-1] + 0.35y[n-2] = x[n] - 0.1x[n-1] - 0.2x[n-2]$$

7. Draw direct form and lattice structure for system having system response $H(z) = 1 + 2z^{-1} + 0.62z^{-2} + 0.8z^{-3}$ [2+5]

8. Compare Butterworth approximation and Chebyshev approximation. Design a digital low-pass filter with the following specification:

   i) Passband magnitude characteristics that is constant to 1 dB below the frequency of 0.2π

   ii) Stopband attenuation of a least 15dB for the frequencies between 0.3π to π

   Use Butterworth approximation as a prototype and use bilinear transformation method to obtain the digital filter. The sampling frequency $(f_s)=1$ Hz. [1+10]

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

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

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

10. Explain Bit Serial Arithmetic in digital signal processor. What are the applications of DSP? [6+2]

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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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32 TRIBHUVAN UNIVERSITY
INSTITUTE OF ENGINEERING
Examination Control Division
2074 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. 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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Examination Control Division
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]=cosωn for n≥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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TRIBHUVAN UNIVERSITY

INSTITUTE OF ENGINEERING

Examination Control Division

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 $\omega c = 1$ radians/sample. Use Hanning window as a prototype. [2+7]

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2073 Magh

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]

    $$0.899 \le |H(e^{jw})| \le 1; \quad \text{for } |w| \le 0.2\pi$$
    $$|H(e^{jw})| \le 0.01; \qquad \text{for } 0.4\pi \le |w| \le \pi$$
13. Where are Digital Signal Processor Chips used? What are the General and special

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2072 Ashwin

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 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
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. 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)*

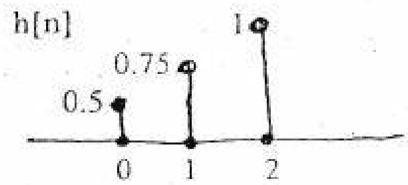
- ✓ Candidates are required to give their answers in their own words as far as practicable.
- ✓ *Attempt **All** questions.*
- ✓ *The figures in the margin indicate **Full Marks**.*
- ✓ *Assume suitable data if necessary.*

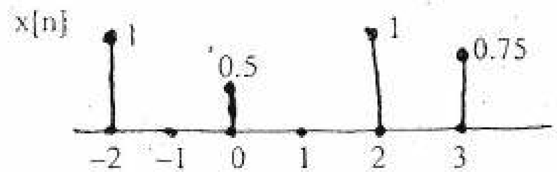
1. What are the advantages of DSP? Explain the significance of A/D and D/A conversion in DSP. [3+4]

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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2070 Magh

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 *(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) = 0.25π radians
    Stop band frequency (W_s) = 0.55π radians
    Minimum attenuation (α_{min}) = 13.55 dB
    Maximum attenuation (α_{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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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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