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PURWANCHAL UNIVERSITY
VIII SEMESTER FINAL EXAMINATION- 2004

LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital signal Processing.

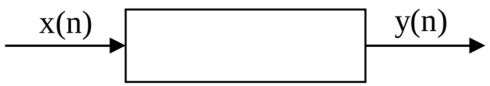
Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.

Attempt ALL questions.

Q. [1] [a] What are the advantage of digital signal processor over an analog signal processor? [6]

[b] Consider a system. [2×3]



Where y(n) = x^2(n-1)+2. Determine whether the system is [i] Linear [ii] Stable [iii] Causal

Q. [2] [a] Determine the sequence x_3(n) corresponding to the circular convolution of x_1(n) and x_2 (n) given as x_1(n)= {**2**,1,2,1} x_2 (n) = { **1**,2,3,4} using DFT and IDFT. [6]

[b] Define linear time invariant system ? Derive the input output relationship for an LTI system. [6]

Q. [3] [a] For a LTI system determine the signal x(n) whose z transform is as X(z) = log (1+az^{-1}) $|z| > |a|$. Find the z transform of x(n) = 2^n 4(n-2) [6]

[b] Derive the formula for the inverse Z transform by contour Integration. [6]

Q. [4] [a] The system function of a causal filter is

$$H(z) = \frac{1}{1 + 0.9z^{-1} - 0.8z^{-2} + 0.5z^{3}}$$

Draw the lattice ladder implementation of the above filter.
Is the filter stable?

[b] Explain the limit cycle oscillations for the recursive filter. [6]

Q. [5] [a] Describe the Remez exchange algorithm for the design of the liner phase FIR filter. [6]

[b] Differenciate between FIR and IIR filter? [6]

Q. [6] Convert a signal pole low pass Butter worth filter with the system function $H(z)\frac{0.245(1+z^{-1})}{1-0.509z^{-1}}$ into a band pass filter with the upper and the lower cut off frequencies ω_u=3π/5 ω_l = 2π/5 respectively. The low pass filter has the 3-dB band width of ω_p =0.2 π [6]

[b] Convert the analog filter into a digital filter whose system function is expressed as $H(s)\frac{1}{(s+1)(s+2)}$ using impulse invariant technique. [6]

Q. [7] Write a short note on (any Two): [4×2=8]

[a] Bit serial arithmetic implementation of DSP process.
[b] Round of errors.
[c] Sampling theorem
[d] Chebyshev Filters.

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PURWANCHAL UNIVERSITY
VIII SEMESTER FINAL EXAMINATION- 2005
LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital Signal Processing.
Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.

Attempt ALL questions.

Q. [1] [a] Define causal system. Explain why causality is important in the realization of Digital filters. Is the system described y(n) = x(-n) is causal. Justify. [2+4+4]

[b] The digital communication link carries binary coded words representing samples of an input signal $x_a(t) = 3\cos 600\ \pi t + 2\cos 1800\ \pi t$. [3+3]

Calculate:
[i] Nyquist rate for the signal $X_a(t)$.
[ii] What will be the discrete time signal if it is sampled at the sampling rate of F_s= 300 sampling per sec.

Q. [2] [a] Differentiate between Discrete time fourier transform and Discrete fourier transform. Explain the periodicity property of DFT. [5+5]

[b] Find the DFT of the signal x[n] = [3,4,2,1] using DIT FFT algorithm. [6]

Q. [3] [a] Show that the z-transform of nx(n) = $X(a^{-1} z)$. Determine also the ROC. (Region of Convergence) [6]
[b] For the second order system $H(z) = 1/(1-2r\ \cos\theta z^{-1}+r^2z^{-2})$. Plot the magnitude phase and group delay (not in scale) for r= 0.8 and O = $\pi/4$. [10]

Q. [4] [a] Determine the Lattice coefficients corresponding to the FIR filter given by $H(z) -1 +1/2z^{-1}+5/8z^{-2}/5z^{3}$ [6]
[b] Design a IIR Digital filter using bilinear transformations method. The designed filter must be a butter worth low pass with the following specifications. ω – o, 4 π , ω_s – 0.55 π , α_p -1 db, α_s = 15 db. [6]

Q. [5] [a] Consider a transfer function of a third order low pass Butterworth filter ω_p = 0.25 π is $H_3(z) = 0.0662\ (1+z^{-1})^3/\ (1-0.2593z^{-1})\ (1-\ 0.6762z^{-1}+0.3917z^{-2})$. [8]
[b] Discuss Remez exchange algorithm in FIR filter design. Explain each steps in brief. [8]

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PURWANCHAL UNIVERSITY
VIII SEMESTER BACK-PAPER EXAMINATION- 2005

LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital Signal Processing.

Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.
Attempt ALL questions.

Q. [1] [a] Explain the basic elements of digital processing system. [8]

[b] Consider a analog signal $x(t) = 3\cos 100\pi t$.
[i] Determine the minimum sampling rate required to avoid aliasing. [7]
[ii] Suppose that the signal is to be sampled at the rate of $F_s = 200$ Hz. What is the discrete time signal obtained after sampling?

Q. [2] [a] Compute and plot the convolution of x(n)*h(n) for the pairs of signal x(n) = {n/3 for 0<=n<=6 and 0 otherwise.
h(n) = {1 for -2<= n <=2 and 0 otherwise. [8]

[b] Explain the importance of the transform techniques for the analysis of the LTI system. [7]

Q. [3] [a] For an LTI system determine the Z-transform of the signal $x(n) = \alpha^{-n} u(n-1) = \{0$ for $n > 0$ and $-\alpha^{n}$ for $n <= 1$.

Find the inverse Z transform of $1/(1-0.5z^{-1} + 0.5z^{-2})$ when ROC is $|z| > 1$ and $z_1 < 0.5$ [8]

[b] Derive the formula for the inverse Z transform by Cauchy Integration. [7]

Q. [4] [a] Determine the FIR filter coefficients for the direct form structure having the lattice filter coefficients $k_1 = ¼$, $k_2 = 1/2$ $k_3 = 1/3$. Draw the lattice ladder implementation of the above filter. [7]

[b] Sensitivity of the filter frequency response characteristics to quantization of the filter coefficients is minimized by realizing the filter having a large number of poles and zeros as an interconnection of second order filter sections. Justify the above statement with suitable example. [8]

Q. [5] [a] Convert a analog filter with a system function H(z) = b/(s+a) into a digital filter by means of bilinear transformation method. [8]

[b] Differentiate between analog and digital filter. [7]

Q. [6] **Write a short note on. (Any TWO):** [2×2.5 = 5]
[a] Bit serial arithmetic implementation of DSP process.
[b] Kaisers window.
[c] Matched Z-transform
[d] Importance of chebyshev filters.

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PURWANCHAL UNIVERSITY
VIII SEMESTER FINAL EXAMINATION- 2006
LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital Signal Processing.
Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.
Attempt ALL questions.

Q. [1] [a] Define linearity, shift invariance, causality & stability of discrete time system. [8]

[b] Determine the output y[n] of a relaxed linear time invariant system with impulse response. [8]

$$h[n] = a^n . u[n], \quad |a| < 1$$

When the input is a unit step sequence.

Q. [2] [a] Prove that the multiplication of the DFTs of two sequence is equivalent to the circular convolution of the two sequences in the time domain. [8]

[b] Determine 8-point DFT of the sequence. [8]
X[n] = {1,2,3,4,5,6}

Q. [3] [a] Determine the z-transform of the signal. [8]
$X[n] = a^n u[n] + b^n u[-n-1]$

[b] A filter is characterized by following poles & zeroes on z-plane. [8]

Zeroes at		**Poles at**	
Radius	**Angle**	**Radius**	**Angle**
0.4	π rad	0.5	0 rad
0.9	1.0376 rad	0.892	2.5158 rad

Show z-plane plot & plot the magnitude response (not to scale)

Q. [4] [a] Consider a causal IIR system with system function $H(z) = (1 + 2z^{-1} + 3z^{-2} + 2z^{-3})/(1 + 0.9z^{-1} - 0.8z^{-2} + 0.5z^{-3})$ Determine the equivalent lattice-ladder structure.

[b] Define Butterworth filter. Derive a formula to obtain the order of Buterworth filter with specification given as pass band frequency, stop band frequency, pass band attenuation & stop band attenuation. [8]

Q. [5] [a] Design a low pass IIR filter with the specifications $\omega_p = 0.2\pi$, $\omega_s = 0.65\pi$, $\alpha_p = 0.4$ dB, $\alpha_s = 15$ dB. Use bilinear transformation method. [8]

[b] A low pass filter is required to be designed with the desired frequency response which is expressed as follows: [8]

$$H_d(e^{-j2\omega}) = \begin{cases} e^{-j2\omega} & for \quad -\pi/4 \le |\omega| \le \pi/4 \\ = 0 & for \quad \pi/4 \le |\omega| \le \pi \end{cases}$$

Obtain filter coefficients $h_d(n)$ if the window function is defined as

$$\omega(n) = 1 \quad \text{for } 0 \le n \le 4$$
$$= 0 \quad \text{Otherwise.}$$

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PURWANCHAL UNIVERSITY
VIII SEMESTER BACK-PAPER EXAMINATION- 2006
LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital Signal Processing.
Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.
Attempt ALL questions.

Q. [1] [a] A system is given as: [8]
Figure:

$h_1(n)$ = { 1,2,1,-1} and $h_2(n)$ = {1,2,3,1}. Find the over all response of the system.

[b] Find the DFT of the signal x(n) = { 1,2,3} suing DIT FFT algorithm. Why FFT is efficient over DFT. [4+4]

Q. [2] [a] Derive the relation of obtaining X(n) from x(z) using Cauchy integral theorem. [7]

[b] Design a two pole band pass filter that has the center of its pass band at $\omega = \frac{\pi}{2}$, zero in its frequency response characteristics $\omega = 0$ and π and its magnitude response is $-\frac{1}{\sqrt{2}}$ at $\omega = 4\frac{\pi}{9}$. [9]

Q. [3] [a] Obtain the parallel realization of the IIR filter having the transfer function $H(z) = \frac{3(2z^2 + 5z + 4)}{(2z+1)(z+2)}$. [6]

[b] Design a IIR digital filter using impulse invariant method. The designed filter must be butter worth low pass with the following specifications. $\omega_p = 0.4\pi$, $\omega_s = 0.55\pi$ $\alpha_p = 1dB$, $\alpha_s = 15db$. [10]

[Note the student must be provided with the pole and zero table of butter worth filter.]

Q. [4] [a] Explain the effect of the limit cycle effect in the recursive filter. What do you mean dead band of the filter. [8]

[b] Calculate the length and the adjustable parameter required for the design of the low pass filter $\omega_p = 0.3\pi$, $\omega_s = 0.5\pi$, $\alpha_s = 40$ using Kaiser window. [8]

Q. [5] [a] A second order low pass IIR filter with a 3-db cut off frequency at $\omega_s = 0.42\pi$ has transfer function $G_{LP} = \frac{0.223(1+z^{-1})^2}{1-0.2952z^{-1} + 0.187z^{-2}}$. Design a second order low pass filter H_{LP} (z) with a 3-dB cut off frequency at $\omega_p = 0.57\pi$ [10]

[b] Explain the basic operation of TMS320C6713DSK processor. [6]

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PURWANCHAL UNIVERSITY
VIII SEMESTER BACK-PAPER EXAMINATION- 2007

LEVEL : B. E. (Electronics & communication)
SUBJECT : BEG433EC, Digital Signal Processing.

Full Marks: 80
TIME: 03:00 hrs **Pass marks:** 32

Candidates are required to give their answer in their own words as far as practicable.
The Figures in the margin indicate the full marks.

Answer ALL questions.

Q. [1] [a] A digital communication link carries binary-coded words representing samples of an input signal. [2+2+2+2]
$x_n(t) = 3\cos 600\pi t + 2\cos 1800\pi t$
The link is operated at 10,000 bits/s and each input sample is quantized into 1024 different voltage levels.
[i] What is the sampling frequency and folding frequency ?
[ii] What is the Nyquist rate for the signal $x_n(t)$?
[iii] What are the frequencies in the resulting discrete-time signal x(n) ?
[iv] What is the resolution Δ?

[b] The impulse response of a linear time invariant system is [8]
h(n) = { 1,2, 1,-1}
Determine the response of the system to the input signal.
x[n] = {1, 2, 3, 1}

Q. [2] [a] Prove graphically that a circular shift on N-point sequence is equivalent to a liner shift of its periodic extension and vice versa. [8]

[b] Compute 8-point DFT of the sequence x[n] = $\cos n\pi/2$ for $0 \le n \ge 7$ using DIF FFT algorithm. [8]

Q. [3] [a] The well known Fibonacci sequence of integer numbers is obtained by computing each term as the sum of the two previous ones. The first few terms of the sequence are 1, 1, 2,3, 5, 8 ……. [8]

[b] Determine the transient and steady-state response of the system characterized by the different equation. [8]
y[n] = 0.5y[n-1]+x[n]
When the input signal is x[n] = $10\cos(n\pi/4)u[n]$. The system is initially at rest.

Q. [4] [a] Describe limit cycle oscillations in recursive system. [4]

[b] Determine the cascade realization for the system described by the system function.
$H(z) = [\ 10(1-1/2z^{-1})\ (1-2/3z^{-1})\ (1+2z^{-1})]/\ [(1-3/4z^{-1})(1-1/8z^{-1})\ [4+(1/2+j1/2)z^{-1}]\ [1-(1/2-j1/2)z^{-1}]]$ [4]

[c] Use the matched z-transformation to convert the analog filter with system function.
$H(z) = [s+0.1]/[(s+0.1)^2+9]$ into a digital IIR filter. Select T = 0.1 and compare the location of the zeroes in H(z) with the locations of zeroes obtained by applying the impulse invariance method in the conversion of H(s).

Q. [5] [a] Why spectral transformation is needed? Convert the single-pole lowpass Butterworth filter with system function.
$H(z) = [0.245(1+z^{-1})]/[1-0.509z^{-1}]$.
Into a bandpass filter with upper and lower cutoff frequency ω_p and ω_L respectively. The lowpass filter 3-dB bandwidth $\omega_p = 0.2\pi(\omega_u = 3\pi/5 \quad and \quad \omega_L = 2\pi/5)$. [2+8]

[b] A digital filter has impulse response given by h[n] = { 1, 0, 0, 0, 0, 0, -1}. What is its system function? Which class if linear phase filter does this system belongs to? Justify. [4]

[c] A linear phase filter has a phase function e^{-j2w} . What is the order of the filter? [2]

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