EC/TE5T2

Reg. No.

Fifth Semester B.E. Degree Examination, January/February 2003

Electronics & Communication/Telecommunication Engineering

Digital Signal Processing

Time: 3 hrs.] [Max.Marks : 100

Note: ***Answer any FIVE full questions.***

1. (a) Give the magnitude characteristics of four classical analog filters. (4 Marks)

   (b) The square magnitude response of an analog Butterworth L.P.F is

   $|H_a(s)|^2 = \frac{1}{[1+(\frac{s}{2})^8]}$

   Determine the order and the cut off frequency of this filter. (4 Marks)

   (c) Derive the transfer function of a normalised Butterworth filter of order N=6. Show the pole locations in the S-plane. (12 Marks)

2. (a) Show that the DFT and IDFT form a consistent discrete Fourier transform pairs. (4 Marks)

   (b) Let $X(e^{jw})$ denote the Fourier transform of the sequence $x(n) = (\frac{1}{2})^n u(n)$. Let $y(n)$ denote a finite-duration sequence of length 10; i.e., $y(n) = 0$, $n < 0$ and $y(n) = 0$, $n \geq 10$. The 10 point DFT of $y(n)$, denoted by $Y(k)$, corresponds to 10 equally spaced samples of $X(e^{jw})$; i.e., $Y(k) = X(e^{j\frac{2\pi k}{10}})$. Determine $y(n)$. (6 Marks)

   (c) Let $X(k)$ denote the N-point DFT of the N-point sequence $x(n)$.

   i) Show that if $x(n)$ satisfies the relation

   $x(n) = -x(N-1-n)$

   Then $X(0) = 0$

   ii) Show that with N even and if

   $x(n) = x(N-1-n)$

   The $X(\frac{N}{2}) = 0$ (10 Marks)

3. (a) Derive DIF-FFT algorithm for N = 8 and draw the complete signal graph. Using this signal flow graph compute the DFT of the sequence

   $x(n) = \{\underset{\uparrow}{1}, -1, 1, -1, 1\}$ of length $N = 5$. (12 Marks)

   (b) Develop DIT-FFT algorithm for $N = 9 = 3 \times 3$ and draw the complete signal flow graph. (8 Marks)

4. (a) What is Chirp Z-transform (CZT). Derive the CZT algorithm. (8 Marks)

   (b) What is fast convolution? Name the different types of fast convolution. (2 Marks)

Contd.... 2

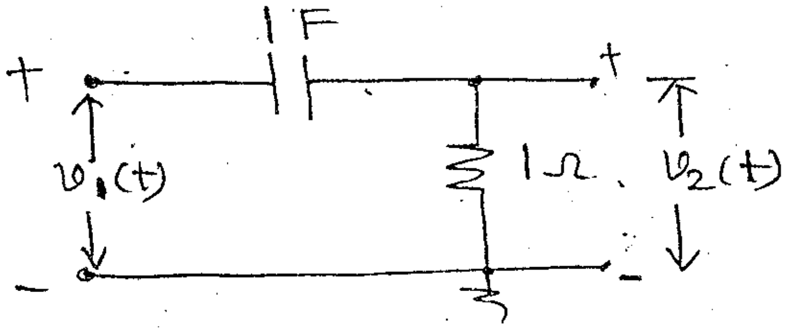
(c) Determine the circular convolution using DFT and IDFT method, for

$$x_1(n) = \{\underset{\uparrow}{1}, 2, -2, -1\}$$

$$x_2(n) = \{\underset{\uparrow}{-3}, -2, 1, 0\}$$

Verify your answer by matrix method. **(10 Marks)**

5. (a) Derive the impulse invariant tranformation. List its properties. Using this transformation obtain the digital filter equivalent of the following analog filter.



Assume the sampling rate $f_s = f_0$ where f_0 is the cut off frequency of the analog filter shown. **(15 Marks)**

(b) Obtain the digital filter using Bilinear transformation

i) $H_a(s) = \frac{1}{(s^2+5s+6)}$

ii) $h_a(t) = e^{-2t}u(t) + e^{-3t}u(t)$

where $h_a(t)$ is the unit impulse response and $H_a(s)$ is the transfer function of the analog filter. Assume T=1sec. **(5 Marks)**

6. (a) Design a Butterworth Filter (Digital) using Bilinear Transformation and hence realize it in Direct form - II. The filter specifications are

$$0.8 \le |H(e^{jw})| \le 1, \quad 0 \le w \le 0.2\pi$$

$$|H(e^{jw})| \le 0.2, \quad 0.6\pi \le w \le \pi$$

Assume $T = 1sec$ **(15 Marks)**

(b) Find the cutoff frequency of the digital filter whose system transfer function is

$$H(Z) = \frac{1-Z^{-1}}{1+Z^{-1}}$$ **(5 Marks)**

7. (a) For the constraints

$$0.8 \le |H(e^{jw})| \le 1, \quad 0 \le w \le 0.25\pi$$

$$|H(e^{jw})| \le 0.1, \quad 0.5\pi \le w \le \pi$$

design a digital Chebyshev filter using $T = 1sec$ and the Bilinear transformation. **(14 Marks)**

Contd...

(b) Distinguish between IIR and FIR digital filters. (6 Marks)

8. (a) Design an ideal bandpass filter with a frequence response.

$$H_d(e^{jw}) = 1, \quad \text{for} \quad \frac{\pi}{4} \le |w| \le \frac{3\pi}{4}$$
$$= 0, \quad \text{otherwise}$$

Use rectangular window with N=11 in your design. (10 Marks)

(b) Explain the frequency sampling technique of FIR filter design with necessary mathematical treatment. (6 Marks)

(c) Obtain the cascade realisation of system function

$$H(Z) = 1 + \tfrac{5}{2}Z^{-1} + 2Z^{-2} + 2Z^{-3}$$ (4 Marks)

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EC/TE5T2

USN

Fifth Semester B.E. Degree Examination, January/February 2004

Electronics & Communication/Telecommunication Engineering

Digital Signal Processing

Time: 3 hrs.] [Max.Marks : 100

Note: *1. Answer any FIVE full questions.*
2. Use of Normalized Butterworth filter table not permitted.

1. (a) Compare the main features of analog and digital filters. (6 Marks)

(b) Obtain the digital filter equivalent of the analog filter shown in fig.1(b) using

i) Impulse invariant transformation ii) Bilinear transformation

Assume the sampling frequency $f_s = 8f_o$, *where* f_o is the cutoff frequency of the filter. (14 Marks)

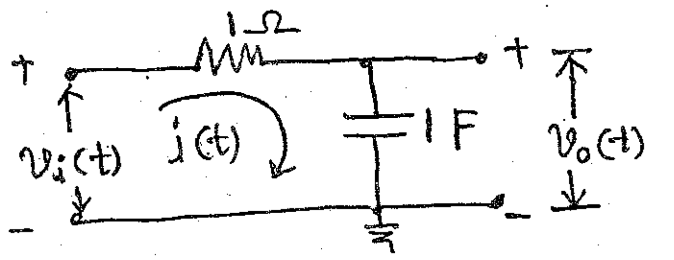

Fig. 1 (b)

2. (a) Design a Chebyshev analog filter with a maximum passband attenuation of 2.5 dB at $\Omega_p = 20\ rad/sec$ and the stop band attenuation of 30 dB at $\Omega_s = 50\ rad/sec$. (10 Marks)

(b) Derive the S to Z plane transformation based on finite backward difference method. Also show that the entire left half S-plane poles are mapped inside the smaller circle of radius $\frac{1}{2}$ centered at $Z = \frac{1}{2}$ inside the unit circle in the Z-plane. (10 Marks)

3. (a) Consider two periodic sequences $\tilde{X}(n)$ and $\tilde{y}(n)$. $\tilde{x}(n)$ has a period N and $\tilde{y}(n)$ has a period M. The sequence $\tilde{\omega}(n)$ is defined as $\tilde{\omega}(n) = \tilde{x}(n) + \tilde{y}(n)$.

i) Show that $\tilde{\omega}(n)$ is periodic with period MN

ii) Also show that $\tilde{\omega}(k)$ represents the M.N point DFT of an MN point sequence $\tilde{\omega}(n)$. (6 Marks)

(b) Compute the circular convolution between the following sequences using DFT and IDFT method.

$$\tilde{x}(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$$

$$\tilde{y}(n) = \{\underset{\uparrow}{-1}, -2, -3, -4\}$$

$\tilde{x}(n)$ & $\tilde{y}(n)$ are periodic sequences with period N=4. (14 Marks)

4. (a) Analog data to be spectrum-analyzed are sampled at 10 kHz and the DFT of

1024 samples computed. Determine the frequency spacing between spectral samples. Justify your answer. (4 Marks)

(b) Derive the DIT-FFT algorithm for N=8 and using the resulting signal flow graph compute the 8-point DFT of an 8-point sequence
$x(n) = \{\underset{\uparrow}{1}, 1, 0, 0, -1, -1, 0, 0\}$

(16 Marks)

5. (a) Design a digital low pass filter using Chebyshev filter design procedure that meets the following specifications :

passband magnitude characteristics that is constant to within 1 dB for frequencies below $w = 0.2\pi$ and stop band attenuation of atleast 15 dB for frequencies between $w = 0.3\pi$ *and* π. Use bilinear transformation. (12 Marks)

(b) Distinguish between IIR and FIR digital filters. (8 Marks)

6. (a) Let $h_a(t)$ denote the impulse response of a linear time-invariant analog filter and $h_d(n)$ the unit-sample response of a linear shift invariant digital filter.

i) If $h_a(t) = e^{-at}u(t)$, determine the analog filter frequency response and sketch its magnitude.

ii) If $h_a(n) = h_a(nT)$ *with* $h_a(t)$ as given above, determine the digital filter frequency response and sketch its magnitude. (10 Marks)

(b) Explain various types of windows used in the design of FIR filters. Write their analytical equations and draw the frequency response characteristics of each window. (10 Marks)

7. (a) Using a rectangular window technique, design a low pass filter with passband gain of unity, cutoff frequency of 1000Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. (10 Marks)

(b) With necessary mathematical analysis explain the frequency sampling technique of FIR filter design. (10 Marks)

8. (a) The unit impulse response of a linear phase filter is $h(n) = \{\underset{\uparrow}{1}, -2, 3, 4, -5\}$

Determine its step response. (4 Marks)

(b) Obtain the direct form II (canonic) and cascade realization of

$$H(Z) = \frac{(Z-1)(Z^2+5Z+6)(Z-3)}{(Z^2+6Z+5)(Z^2-6Z+8)}$$

The cascade section should consist of two biquadratic sections. (12 Marks)

(c) Determine the cutoff frequency of the digital filter whose difference equation is $y(n) - \frac{1}{2}y(n-1) = x(n)$ (4 Marks)

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EC/TE5T2

USN

Fifth Semester B.E. Degree Examination, July/August 2004

Electronics & Communication/Telecommunication Engineering

Digital Signal Processing

Time: 3 hrs.] [Max.Marks : 100

Note: *1. Answer any FIVE full questions.*
2. Use of Normalized Chebyshev and Butterworth prototype table is NOT allowed.

1. (a) How is the design of filters related to approximation problem in network theory in both time and frequency domain. What are the different types of error criteria used in the approximation. **(6+4 Marks)**

(b) Design a lowest order analog filter with maximally flat response in the pass-band and an acceptable attenuation of –2.5 dB at 15 rad/sec. The attenuation in the stopband should be more than –12dB beyond 25 rad/sec. Sketch the pole-zero's of the filter. **(10 Marks)**

2. (a) $x(n)$ is a N-point sequence with N-point DFT X(k). If DFT of X(k) is computed resulting in $x_1(n)$ establish the relation ship between $x(n)$ and $x_1(n)$. **(6 Marks)**

(b) A long sequence $x(n)$ is filtered through a filter of impulse response $h(n)$ to give output $y(n)$. Given $x(n)$ and $h(n)$ as follows, compute $y(n)$ using over lap add technique.

$x(n) = [1111131142113111]$

$h(n) = [1\ \ -1]$

Use only 5-point circular convolution in your approach. **(14 Marks)**

3. (a) A complex sequence $z(n)$ with $DFT\ Z(k)$ is formed as $z(n) = x(n) + j\,y(n)$, where $x(n)$ and $y(n)$ are real sequences with corresponding DFT's $X(k)$ *and* $Y(k)$ respectively. Express $X(k)$ & $Y(k)$ in term of DFT $Z(k)$. Given

$$Z(k) = [12 + j12, 1.414 + j3.414,\ 0,\ -0.5858 + j1.414, 0,$$
$$-1.414 + j0.5858,\ 0,\ -3.414 - j1.414]$$

Compute $X(k)$ & $Y(k)$ using above relations without computing any DFT. **(12 Marks)**

(b) Explain chirp z-transform and bringout its differences from usual z-transform. **(8 Marks)**

4. (a) Derive the necessary conditions for FIR filters to have linear phase characteristics. **(6 Marks)**

(b) Enumerate the differences between IIR and FIR filters. **(4 Marks)**

(c) A low-pass FIR causal filter is to be designed with the following desired frequency response.

$$H_d(e^{jw}) = \begin{cases} e^{-j2w} & \frac{-\pi}{4} \le w \le \frac{\pi}{4} \\ 0 & \frac{\pi}{4} \le |w| \le \pi \end{cases}$$

Determine the filter coefficients $h(n)$ if a rectangular window of width 5-samples is used. **(10 Marks)**

Contd.... 2

5. (a) What is FFT? Explain radix-2 DIT-FFT algorithm. (6 Marks)

(b) Develop Decimation-in-frequency (DIF) FFT algorithm with all necessary steps and neat signal flow diagram used in computing N-point DFT X(k) of a N-point sequence $x(n)$. Using the same, compute the DFT of a sequence.
$x(n) = [44\ 22\ 33\ 22]$
What is the number of computations required in this computation? (14 Marks)

6. (a) Explain impulse invariant method of designing a digital filter. (6 Marks)

(b) Determine $H(z)$ for a lowest order Butterworth filter satisfying following constraints.

$$\sqrt{0.5} \leq |H(e^{jw})| \leq 1 \quad 0 \leq |w| \leq \frac{\pi}{2}$$

$$|H(e^{jw})| \leq 0.2 \quad \frac{3\pi}{4} \leq w \leq \pi$$

with $T = 1s$. Apply impulse invariant transformation. Realise $H(z)$ in parallel form. (14 Marks)

7. (a) Show that mapping function used with bilinear transformation satisfies all requirements in transforming analog filter to digital filter effectively. (6 Marks)

(b) Design a digital Chebyshev filter to satisfy the constraints

$$0.707 \leq |H(e^{jw})| \leq 1 \quad 0 \leq w \leq 0.2\pi$$

$$|H(e^{jw})| \leq 0.1 \quad 0.5\pi \leq w \leq \pi$$

Using bilinear transformation with $T = 1sec$. (14 Marks)

8. A discrete-time system $H(z)$ is expressed as,

$$H(z) = \frac{10(1-\frac{1}{2}z^{-1})\,(1-\frac{2}{3}z^{-1})\,(1+2z^{-1})}{(1-\frac{3}{4}z^{-1})\,(1-\frac{1}{8}z^{-1})\,(1-(\frac{1}{2}+j\frac{1}{2})z^{-1})\,(1-(\frac{1}{2}-j\frac{1}{2})z^{-1})}$$

i) Find the difference equation for the system. (2 Marks)

ii) Realize the system in direct form I and II. (4+4 Marks)

iii) Realize parallel and cascades forms using second order sections. (6+4 Marks)

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EC/TE5T2

USN

Fifth Semester B.E. Degree Examination, July/August 2005

Electronics & Communication/Telecommunication Engineering

(Old Scheme)

Digital Signal Processing

Time: 3 hrs.] [Max.Marks : 100

Note: ***1. Answer any FIVE full questions.***
2. Any missing data may suitably be assumed.
3. All assumptions must be stated. Symbols have usual meanings.

1. (a) The Four-Point DFT of a 4 - point sequence $x(n)$ is given by $X(k) = \{10, -2+j2, -2, -2-j2\}$

Without computing DFT or IDFT determine DFT of the following :

i) $x_1(n) = x(n)(-1)^n$

ii) $x_2(n) = x(4-n)$

iii) $x_3(n) = x((n-1))_4 + x((n-2))_4$ (12 Marks)

(b) Given two sequences $x_1(n)$ & $x_2(n)$ of length N, obtain an expression to compute circular convolution of these sequences. What changes are required if the circular convolution output to be same as linear convolution output. (8 Marks)

2. (a)

$$\text{Given} \quad x_1(n) = e^{j\pi n} \quad 0 \le n \le 7$$
$$x_2(n) = u(n) - u(n-5)$$

Determine $x_3(n) = x_1(n)$ *and* $x_2(n)$. Sketch all the sequences. (10 Marks)

(b) Define chirp Z - transform. Describe the computational scheme to revaluate CZT. Enumerate differences between CZT and DFT. (10 Marks)

3. (a) Derive the complete decimation - in- frequency FFT algorithm for a 8 - point sequence. Draw the neat signal flow graph mentioning all intermediate outputs. (12 Marks)

(b) Compute the IDFT using the decimation - in frequency of Q 3.(a) given

$X(k) = \{4, 1-j2.414, 0,\ 1-j0.410, 0, 1+j0.414, 0, 1+j2.414\}$ (8 Marks)

4. (a) How do you formulate analog filter design as a approximation problem in time-domain. What error criteria can be used ? (5 Marks)

(b) Drive the necessary frequency transformation expressions used in converting low pass analog filter to high pass and band pass analog filters. (6 Marks)

(c) Determine the order and cutoff frequency of the analog butterworth filter such that it has a -2dB pass band attenuation at a frequency of 20 rad/sec and at least 15 dB stop band attenuation at 30 rad/sec. (9 Marks)

Contd.... 2

5. Design a analog filter which has equiripple characteristics in passband and monotonic fall-off characteristics in stopband given maximum passband attenuation of $2.5dB$ at $\Omega_p = 20rad/sec$ and the stop band attenuation of $30dB$ at $\Omega_s = 30rad/sec$. Transform the analog filter to digital filter using impulse invariance method. **(20 Marks)**

6. (a) Show that if the impulse response has the even symmetry then FIR filter possess linear phase characteristics. Comment on position of zeros on the Z-plane. **(8 Marks)**

(b) Design a ideal high pass filter with a frequency response.

$$H_d(3^{jw}) = \begin{cases} 1 & \frac{\pi}{4} \leq |\ w\ | \leq \pi \\ 0 & |\ w\ | < \pi/4 \end{cases}$$

Find the values of $h(n)$ for N = 9 and plot the sequence. **(12 Marks)**

7. (a) Draw the direct form II, cascade and parallel form (using first order section) structure for the following system

$$H(z) = \frac{1-3/4z^{-1}+\frac{1}{8}z^{-2}}{(1+z^{-1}+2/9z^{-2})\ (1+\frac{1}{4}z^{-1})}$$ **(16 Marks)**

(b) Compare Butterworth and Chebyshev filter characteristics **(4 Marks)**

8. Write short notes on :

a) Window based FIR filter design **(6 Marks)**

b) Bilinear transformation **(8 Marks)**

c) Butterfly operation and inplace computation **(6 Marks)**

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NEW SCHEME

EC52

Reg. No.

Fifth Semester B.E. Degree Examination, January/February 2006

Electronics & Communication/Telecommunication Engineering

Digital Signal Processing

Time: 3 hrs.) (Max.Marks : 100

Note: ***Answer any FIVE full questions.***

1. (a) What is the difference between the discrete Fourier series and the discrete Fourier transform? **(3 Marks)**

   (b) Consider the finite-length sequence $x(n)$; $x(n) = \left(1, \frac{3}{4}, \frac{1}{2}, \frac{1}{4}\right)$. The 4-point DFT of $x(n)$ is $X(K)$. Plot the sequence $y(n)$ whose DFT is $Y(K) = W_4^{3K}\ X(K)$. **(5 Marks)**

   (c) Let $X(K)$ denote the N-point DFT of an N-point sequence $x(n)$. $X(K)$ itself is an N-point sequence. If the DFT of $X(K)$ is computed to obtain a sequence $x_1(n)$, determine $x_1(n)$ in terms of $x(n)$. **(6 Marks)**

   (d) $x(n)$ denotes a finite-length sequence of length N. Show that $x\left[(-n)_N\right] = x\left[(N-n)_N\right]$. **(6 Marks)**

2. (a) Prove Parseval's relation as applied to DFT. **(5 Marks)**

   (b) Let $x_1(n) = \left(1, \frac{1}{2}, \frac{1}{4}, \frac{1}{8}\right)$ and $x_2(n) = (1,\ 1,\ 1,\ 1)$. Compute the DFT of $x_1(n)$ by the decimation in time FFT algorithm and that of $x_2(n)$ by the decimation in frequency FFT algorithm. Using the above results, evaluate the circular convolution of $x_1(n)$ and $x_2(n)$. **(3+3+4 Marks)**

   (b) How many (real) storage registers are required to evaluate the DFT by an in-place FFT algorithm? **(3 Marks)**

   (c) How many twiddles will make a butterfly fast? **(2 Marks)**

3. (a) It is required to take DFT of a data stream of length 8192. However, the analyzer has only a fixed hardware implementation for a 2048 point DFT. Assuming other storage is available along with ways of adding and multiplying, how could the desired 8192-pt transform be obtained? **(8 Marks)**

   (b) Derive the Goertzel algorithm for the computation of DFT. Compare this algorithm with that of the direct method of computing DFT, with respect to number of multiplications and additions. Draw the signal flow graph of the complex recursive computation. Can the said algorithm, at any point of time, be more efficient than the FFT approach? **(6+2+2+2 Marks)**

Contd.... 2

4. (a) Develop a transformation for the solution of a first order linear constant coefficient difference equation by using trapezoidal approximation for the integral approximation. Highlight the features of transformation. **(10 Marks)**

(b) Using the bilinear transformation $S = \frac{1-Z^{-1}}{1+Z^{-1}}$, What is the image of $S = e^{j\pi/2}$ in the Z-plane. **(5 Marks)**

(c) What is the contour in the Z-plane that is the image of $J\Omega$ axis in the S-plane for the mapping $S = \frac{Z-1}{T}$? Are stable system in the S-plane, mapped into stable system in the Z-plane? **(5 Marks)**

5. (a) Discuss the frequency sampling method of FIR filter design. Using the above principle, design a FIR filter with $h(n) = (1,\ 2,\ 1)$. Also indicate the signal flow graph. **(4+4+4 Marks)**

(b) Design a FIR low pass filter with the frequency response, using rectangular window

$$h_d(\omega) = e^{\frac{-j\omega c(N-1)}{2}} \qquad \frac{-\pi}{2} \le \omega \le \frac{\pi}{2}$$
$$= 0 \qquad elsewhere$$

(8 Marks)

6. (a) Let $h(n)$ be the unit sample response of a FIR filter so that $h(n) = 0$ for $n < 0,\ n \ge N$. Assume $h(n)$ is real. The frequency response of this filter can be represented in the form

$$H(e^{j\omega}) = \hat{H}\,(e^{j\omega})\, e^{j\Theta(\omega)}$$

i) Find $\Theta(\omega)$ for $0 \le \omega \le \pi$ When $h(n)$ satisfies the condition

$$h(n) = h(N-1-n)$$

(10 Marks)

ii) If N is even, show that

$$h(n) = h(N-1-n)$$

Implies that $H(\frac{N}{2}) = 0$, where $H(K)$ is the N-point DFT of $h(n)$. **(4 Marks)**

(c) Bring out the comparison between IIR and FIR filters. **(6 Marks)**

7. (a) Design a digital LPF with a passband magnitude characteristic that is constant to within 0.75 dB for frequencies below $\omega = 0.2613\pi$ and stop band attenuation of atleast 20 dB for frequencies between $\omega = 0.4018\pi$ and π. Determine the transfer function $H(Z)$ for the lowest order Butterworth design which meets the specifications. Use bilinear transformation. **(12 Marks)**

(b) Design an analog Chebyshev filter for which the squared magnitude response $|H_a(J\Omega)|^2$ satisfies the condition.

$$20\ log_{10}\ |H_a(J\Omega)|_{\Omega=0.2\pi} \ \ge -1$$
$$20\ log_{10}\ |H_a(J\Omega)|_{\Omega=0.3\pi} \ \le -15$$

(8 Marks)

8. **(a)** Consider an analog system function

$$H_a(S) = \frac{S+a}{(s+a)^2+b^2}$$

Determine the digital filter from an analog filter by means of impulse invariance. When would it produce good results?

(6 Marks)

(b) Obtain a parallel realization for the following $H(Z)$:

$$H(Z) = \frac{8Z^3-4Z^2+11Z-2}{(Z-\frac{1}{4})\,(Z^2-Z+\frac{1}{2})}$$

Also indicate the governing equations. **(8 Marks)**

(c) Implement a digital network whose unit-sample response is $e^{j\omega_o n}\,u(n)$. While implementing this system with a complex unit-sample response, the real and imaginary parts are to be distinguished as separate outputs.

(6 Marks)

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