Digital Signal Processing VTU Question Paper Set 2017

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10EE64

Sixth Semester B.E. Degree Examination, Dec.2016/Jan.2017
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions, selecting atleast TWO questions from each part.***

PART – A

1 a. If X(k) is N – point DFT of N-length sequence x(n), and if $x_1(n)$ is DFT of X(k), then determine $x_1(n)$ interms of x(n). **(05 Marks)**

b. Compute 8 – point DFT of the sequence x(n) = {1, 2, 2, 1, 2, 2} and verify conjugate symmetry about k = N/2. **(10 Marks)**

c. If X(k) represent 6-point DFT of sequence. X(n) = {2, –1, 3, 4, 0, 5}, then find y(n) of same length as x(n) such that its DFT $Y(k) = W_3^{2k} X(k)$. **(05 Marks)**

2 a. Using Stockham's method find circular convolution of the sequences : $g(n) = \delta(n) + 2\delta(n-1) + 3\delta(n-2) + 4\delta(n-3)$ and h(n) = n for $0 \le n \le 3$. **(07 Marks)**

b. Obtain output of the system having impulse response $h(n) = \cos\left(\frac{2\pi n}{N}\right)$ and input $x(n) = \sin\left(\frac{2\pi n}{N}\right)$, through N – point circular convolution. **(06 Marks)**

c. Use sectional convolution approach to find the response of filter having impulse response h(n) = {1, 2} and input x(n) = {1, 2, –1, 2, 3, –2, –3, –1, 1, 1, 2, –1}. Use 5-point circular convolution use overlap and add method. **(07 Marks)**

3 a. Develop DIF FFT algorithm for N = 8 from basic principles of decomposition of radix–2. **(10 Marks)**

b. Using time decomposition approach find the DFT of sequence for N point such that $N = 2^M$ and M = 3, the given sequence is y(n) = {1, 1, 1, 1 1}. **(10 Marks)**

4 a. The first five points of DFT of a sequence are given as {7, –0.707–j0.707, –j, 0.707–j0.707, 1}. Obtain the corresponding time domain sequence of length–8 using DIF FFT algorithm. **(10 Marks)**

b. Develop a N–composite DIT FFT algorithm for evaluating 9 point DFT. **(10 Marks)**

PART – B

5 a. A lowpass Butterworth filter has to meet the following specifications :
Passband gain, $K_p = -1$dB at $\Omega_p = 4$ rad/sec
Stopband attenuation greater than or equal to 20 dB at $\Omega_S = 8$ rad/sec.
Determine the transfer function $H_a(s)$ of the lowest order Butterworth filter to meet the above specifications. **(10 Marks)**

b. Design a Chebyshev – I filter to meet the following specifications :

Passband ripple	: $\le$ 2dB
Passband edge	: 1 rad/sec
Stopband attenuation	: $\ge$ 20 dB
Stopband edge	: 1.3 rad/sec.

(10 Marks)

6 a. Using impulse invariant transformation, design a digital Chebyshev I filter that satisfies the following constraints. $0.8 \le |H(\omega)| \le 1, \quad 0 \le \omega \le 0.2\pi$
$|H(\omega)| \le 0.2, \quad 0.6\pi \le \omega \le \pi.$ **(12 Marks)**

b. Define the following windows along with their impulse response :
i) Rectangular window
ii) Hamming window
iii) Hanning window. **(08 Marks)**

7 a. The desired frequency response of a lowpass FIR filter is given by :

$$H_d(\omega) = \begin{cases} e^{-j3\omega}, & |\omega| < \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} < |\omega| < \pi \end{cases}$$

Determine the frequency response of the filter using Hamming window for N =7. **(10 Marks)**

b. Determine the filter coefficients h(n) obtained by sampling $H_d(\omega)$ given by :

$$H_d(\omega) = \begin{cases} e^{-j3\omega}, & 0 < \omega \le \frac{\pi}{2} \\ 0, & \frac{\pi}{2} < \omega < \pi \end{cases}$$

Also obtain frequency response taking N = 7. **(10 Marks)**

8 a. For a LTI system described by following input-output relation :
$2y(n) - y(n-2) - 4y(n-3) = 3x(n-2)$
Realize the system in following forms :
i) Direct form – I
ii) Direct form – II transposed realization. **(10 Marks)**

b. Obtain cascade realization for the system function given below :

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

(06 Marks)

c. Compare direct from – I and II realizations. **(04 Marks)**

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Sixth Semester B.E. Degree Examination, June/July 2016
Digital Signal Processing

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Note: ***Answer FIVE full questions, selecting at least TWO questions from each part.***

PART – A

1 a. Compute the N – point DFT of $x[n] = a^n$ for $0 \le n \le N-1$. Also find the DFT of the sequence $x[n] = 0.5^n u[n]$; $0 \le n \le 3$. **(07 Marks)**

b. Find the DFT of a sequence $x[n] = \begin{cases} 1 \text{ for } 0 \le n \le 3 \\ 0 \text{ otherwise} \end{cases}$

For N = 8. Plot magnitude of the DFT x(k). **(10 Marks)**

c. If $x[n] \xleftrightarrow[N]{DFT} x(k)$ then prove that DFT {x(k)} = N x(-ℓ) **(03 Marks)**

2 a. The first values of an 8 point DFT of a real value sequence is {28, -4.966j, 4+4j, -4+1.66j, -4}. Find the remaining values of the DFT. **(04 Marks)**

b. Obtain the circular convolution of $x_1[n]$ = [1, 2, 3, 4] with [1, 1, 2, 2]. **(06 Marks)**

c. A long sequence x[n] is filtered though a filter with impulse response h(n) to yield the output y[n]. if x[n] = {1, 4, 3, 0, 7, 4, -7, -7, -1, 3, 4, 3}, h(n) = {1, 2} compute y[n] using overlap add technique. Use only a 5 point circular convolution. **(10 Marks)**

3 a. Prove the symmetry and periodicity property of a twiddle factor. **(04 Marks)**

b. Develop an 8 point DIT – FFT algorithm. Draw the signal flow Graph. Determine the DFT of the sequence x[n] = {1, 1, 1, 1, 0, 0, 0, 0,} using signal flow graph. Show all the intermediate results on the signal flow graph. **(12 Marks)**

c. What is FFT algorithm? State their advantages over the direct computation of DFT. **(04 Marks)**

4 a. Find 4 point circular convolution of x[n] and h[n] using radix 2 DIF FFT algorithm x[n] = [1, 1, 1, 1] and h[n] = [1, 0, 1, 0]. **(08 Marks)**

b. Calculate the IDFT of x(k) = {0, 2.828 – j2.828, 0, 0, 0, 0, 0, 2.82 + j 2.82} using iniverse radix 2 DIT FFT algorithm. **(12 Marks)**

PART – B

5 a. The transfer function of an analog filter is given as $H_a(s) = \frac{1}{(s+1)(s+2)}$: obtain H(z) using impulse invariant method. Take sampling frequency of 5 samples/sec. **(05 Marks)**

b. Obtain H(z) using impulse invariance method for following analog filter $H_a(s) = \frac{1}{(s+0.5)(s^2+0.5s+2)}$. Assume T = 1 sec. **(10 Marks)**

c. Convert the analog filter into a digital filter whose system function is $H(s) = \frac{2}{(s+1)(s+3)}$ using bilinear transformation, with T = 0.1 sec. **(05 Marks)**

6 a. Design a Digital Butterworth filter using the bilinear transformation for the following specifications:

$$0.8 \le \left|H(e^{jw})\right| \le 1 \quad \text{for } 0 \le w \le 0.2\pi$$

$$\left|H(e^{jw})\right| \le 0.2 \quad \text{for } 0.6\pi \le w \le \pi$$

(12 Marks)

b. Determine the order of a Chebyshev digital low pass filter to meet the following specifications: In the passband extending from 0 to 0.25π a ripple of not more than 2dB is allowed. In the stop band extending form 0.4π to π, attenuation can be more than 40dB. Use bilinear transformation method. **(08 Marks)**

7 a. The frequency response of a filter is given by $H(e^{jw}) = jw\,;\, -\pi \le w \le \pi$. Design the FIR filter, using a rectangular window function. Take N = 7. **(12 Marks)**

b. The desired frequency response of the low pass FIR filter is given by

$$H_d(e^{jw}) = H_d(w) = \begin{cases} e^{-j3w}; & |w| < 3\pi/4 \\ 0 \quad ; & 3\pi/4 < |w| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if the hamming window is used with N = 7. **(08 Marks)**

8 a. A FIR filter is given by $y[n] = x[n] + \frac{2}{5}x(n-1) + \frac{3}{4}x(n-2) + \frac{1}{3}x(n-2)$. Draw the direct and linear form realization. **(10 Marks)**

b. Obtain the direct form II and cascade realization of the following function.

$$H(z) = \frac{8z^3 - 4z^2 + 11z - 2}{(z-0.25)(z^2 - z + 0.5)}$$

(10 Marks)

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Sixth Semester B.E. Degree Examination, Dec.2015/Jan.2016
Digital Signal Processing

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PART – A

1 a. List and state any four properties of DFT. **(06 Marks)**

b. Find the DFT of a sequence $x(n) = \{1, 1, 0, 0\}$ and find the IDFT of $Y(K) = (2, 1+j, 0, 1-j)$ **(08 Marks)**

c. Consider the finite length sequence x(n) shown in Fig. Q1 (c). The five point DFT of x(n) is denoted by X(K). Plot the sequence whose DFT is $Y(K) = e^{\frac{-4\pi K}{5}} X(K)$. **(06 Marks)**

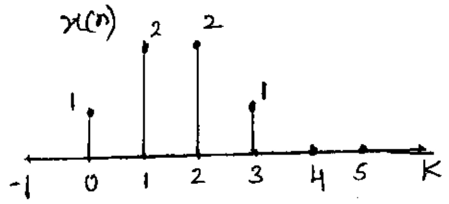

Fig. Q1 (c)

2 a. Perform the circular convolution of the following sequence $x(n) = \{1, 1, 2, 1\}$, $h(n) = \{1, 2, 3, 4\}$ using DFT and IDFT method. **(08 Marks)**

b. Find the output y(n) of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and input signal $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using overlap–add method. Use 5-point circular convolution in your approach. **(12 Marks)**

3 a. What is FFT? Explain Decimation-in-Time algorithm. **(08 Marks)**

b. Given the sequences $x_1(n)$ and $x_2(n)$ below. Compute the circular convolution $x_1(n) \circledast x_2(n)$ for N = 4. Use DIT – FFT algorithm. **(12 Marks)**

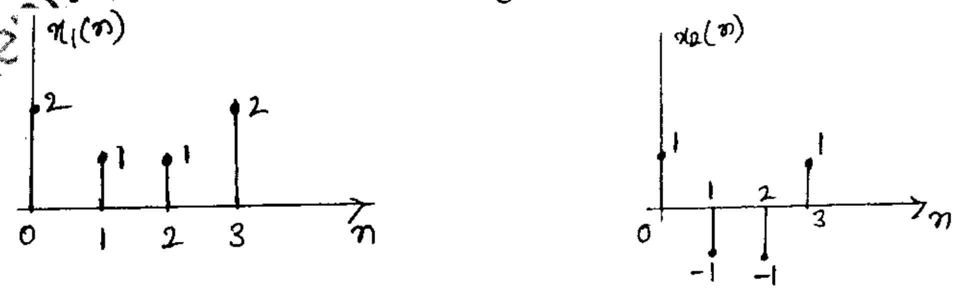

Fig. Q3 (b)

4 a. What is DIF algorithm? Draw the 4-point radix-2 DIF-FFT Butterfly structure for DFT. **(06 Marks)**

b. Find the 4-point real sequence x(n), if its 4-point DFT samples are $X(0) = 6$, $X(1) = -2 + j2$, $X(2) = -2$. Use DIF-FFT algorithm. **(08 Marks)**

c. Find the 4-point DFT of the sequence, $x(n) = \cos\left(\frac{\pi}{4} n\right)$ using DIF-FFT algorithm. **(06 Marks)**

PART – B

5 a. Distinguish between analog and digital filters. **(04 Marks)**

b. Design an analog Bandpass filter to meet the following frequency-domain specifications:
   i) a -3.0103 dB upper and lower cutoff frequency of 50 Hz and 20 kHz.
   ii) a stopband attenuation of atleast 20 dB at 20 Hz and 45 kHz and
   iii) a monotonic frequency response. **(10 Marks)**

c. The system function of the analog filter is given by $H_a(s) = \dfrac{s+0.1}{(s+0.1)^2+9}$.
   Obtain the system function of the IIR digital filter by using Impulse invariance method. **(06 Marks)**

6 a. A Chebyshev – I filter of order N = 3 and unit bandwidth is known to have a pole at $s = -1$.
   i) Find the two other poles of the filter and parameter ε.
   ii) The analog filter is mapped to the z-domain using the bilinear transformation with T = 2. Find the transfer function H(z) of the digital filter. **(12 Marks)**

b. Distinguish between Butterworth and Chebyshev filter. **(04 Marks)**

c. What is Bilinear transformation? Explain warping and prewarping effect. **(04 Marks)**

7 a. What is Gibb's phenomenon? **(04 Marks)**

b. Distinguish between FIR and IIR filters. **(04 Marks)**

c. A filter is to be designed with the following desired frequency response:

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

Find the frequency response of the FIR filter designed using a rectangular window defined below:

$$W_R(n) = \begin{cases} 1 & 0 < n < 4 \\ 0 & \text{Otherwise} \end{cases}$$

(12 Marks)

8 a. Sketch the direct form-I, direct form-II realizations for the system function given below:
$H(z) = \dfrac{2z^2 + z - 2}{z - 2}$. **(10 Marks)**

b. Obtain a Cascade realization for a system having the following system function:

$$H(z) = \frac{(z-1)(z-2)(z+1)z}{\left(z-\frac{1}{2}-j\frac{1}{2}\right)\left(z-\frac{1}{2}+j\frac{1}{2}\right)\left(z-j\frac{1}{4}\right)\left(z+j\frac{1}{4}\right)}.$$

(10 Marks)

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Sixth Semester B.E. Degree Examination, June/July 2015
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions, selecting atleast TWO questions from each part.***

PART – A

1 a. Determine DFT of sequence $x(n) = \frac{1}{3}$ for $0 \le n \le 2$ for $N = 4$. Plot magnitude and phase spectrum. **(06 Marks)**

b. Compute the 4 – point DFT of the sequence $x(n) = (1, 0, 1, 0)$. Also find $y(n)$, if $y(k) = x((k-2))_4$. **(06 Marks)**

c. Compute circular convolution using DFT + IDFT for the following sequences.
$x_1(n) = \{2, 3, 1, 1\}$ $x_2(n) = \{1, 3, 5, 3\}$. **(08 Marks)**

2 a. Two length - 4 sequences are defined below :
$x(n) = \cos(\pi n/2)$ $n = 0, 1, 2, 3$
$h(n) = 2^n$ $n = 0, 1, 2, 3$

i) calculate $x(n) \circledast_4 h(n)$ using circular convolution directly

ii) calculate $x(n) \circledast_4 h(n)$using linear convolution. **(10 Marks)**

b. Find the output $y(n)$ of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and input signal $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using :
i) overlap – save method
ii) overlap – add method.
Use circular convolution. **(10 Marks)**

3 a. Explain Decimation–in–time algorithm. Draw the basic butterfly diagram for DIT algorithm. **(08 Marks)**

b. Find the 8–point DFT of the sequence, $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$. Using DIT–FFT radix–2 algorithm. The basic computational block known as the butterfly should be as shown in Fig. Q3(b). **(12 Marks)**

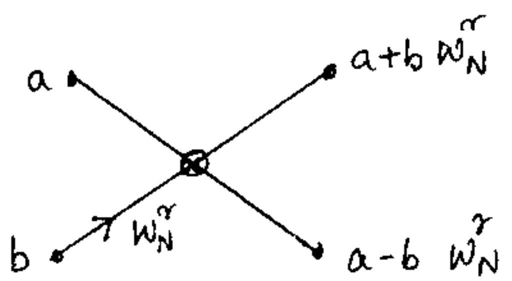

Fig.Q3(b)

4 a. Find the 4 – point DFT of the sequence, $x(n) = \cos\left(\frac{\pi}{4}n\right)$ using DIF–FFT algorithm. **(08 Marks)**

b. Using linear convolution find $y(n) = x(n) * h(n)$ for the sequences :
$x(n) = (1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1)$ and $h(n) = (1, 2)$.
Compare the result by solving the problem using :
i) Overlap – save method
ii) Overlap – add method. **(12 Marks)**

PART – B

5 a. Compare analog and digital filters. **(04 Marks)**

b. For the given specifications $k_p = 3\text{dB}$; $k_s = 15$ dB; $\Omega_p = 1000$ rad/sec; $\Omega_s = 500$ rad/sec. Design analog Butterworth high-pass filter. **(08 Marks)**

c. Design a Chebyshev analog low-pass filter that has a –3 dB cut off frequency of 100 rad/sec and a stop-band attenuation of 25 dB or greater for all radian frequencies past 250 rad/sec. **(08 Marks)**

6 a. Design a high-pass filter H(z) to meet the specifications shown in Fig. Q6(a). The sampling rate is fixed at 1000 samples/sec. Use Bilinear transformation. **(12 Marks)**

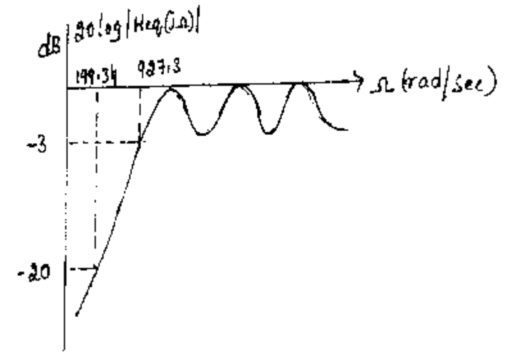

Fig.Q6(a)

b. Transform the analog filter :

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

into H(z) using impulse invariant transformation. Take T = 0.1 sec. **(08 Marks)**

7 a. Explain why windows are necessary in FIR filter design. What are the different windows in practice? Explain in brief. **(08 Marks)**

b. A filter is to be designed with the following desired frequency response :

$$H_d(\omega) = \begin{cases} 0, & -\frac{\pi}{4} < \omega < \frac{\pi}{\omega} \\ e^{-j2\omega}, & \frac{\pi}{4} < |\omega| < \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangular window defined below : $\omega_R(n) = \begin{cases} 1 & 0 \le n \le 4 \\ 0 & \text{otheriwse} \end{cases}$. **(12 Marks)**

8 Realize the following transfer function using :

$$H(z) = \left\{ \frac{0.7 - 0.25z^{-1} - z^{-2}}{1 + 0.1z^{-1} - 0.72z^{-2}} \right.$$

i) Direct form – I
ii) Direct form – II
iii) Cascade form
iv) Parallel form. **(20 Marks)**

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10EE64

Sixth Semester B.E. Degree Examination, June/July 2014
Digital Signal Processing

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PART – A

1 a. Compute N-point DFT of x(n) for N = 4, where,

$$x(n) = \begin{cases} 1/3; & \text{for } 0 \le n \le 2 \\ 0; & \text{otherwise} \end{cases}$$

Draw magnitude and phase spectra. **(10 Marks)**

b. Determine 4-point DFT of x(n) = {0, 1, 2, 3}. Hence verify the result by taking IDFT using linear transformation. **(10 Marks)**

2 a. State and prove the following properties of DFT: i) Periodicity; ii) Linearity. **(08 Marks)**

b. Find the output of LTI system whose impulse response, h(n) = {1, 1, 1} and input signal, x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, ...} using overlap add method. Use block length, N = 5. **(12 Marks)**

3 a. Why FFT is needed? What is the speed improvement factor in calculating 64-pt. DFT of a sequence using direct computation and FFT algorithm? **(08 Marks)**

b. What are the differences and similarities between DIT and DIF-FFT algorithms? **(04 Marks)**

c. Compute the 8-pt. DFT of the sequence, x(n) = {0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0}. Using the in place radix-2 DIT algorithm. **(08 Marks)**

4 a. Develop the DIF-FFT algorithm for N = 8. Using the resulting signal flow graph compute the 8-point DFT of the sequence, $x(n) = \sin\left(\frac{\pi}{2}n\right)$, $0 \le n \le 7$. **(11 Marks)**

b. First five points of eight point DFT of a real valued sequence is given by, x(k) = {0, 2 + j2, -j4, 2-j2, 0}. Determine the remaining points. Hence find the sequence x(n) using DIF-FFT algorithm. **(09 Marks)**

PART – B

5 a. Explain impulse invariance method of designing IIR filter. Hence show that mapping results in many-to-one-mapping on unit circle. **(08 Marks)**

b. Determine H(z) of lowest order Butterworth filter that will meet the following specifications:

i) 1 dB ripple in passband; $0 \le w \le 0.15\pi$ rad.

ii) At least 20dB attenuation in stopband; $0.45\pi \le w \le \pi$ rad.

Use bilinear transformation for T = 1sec. **(12 Marks)**

6 a. Design an analog Chebyshev filter that will meet the following specifications:

i) Maximum pass band attenuation = 3dB at 2 rad/sec.

ii) Minimum stop band attenuation = 20dB at 4 rad/sec. **(10 Marks)**

b. Explain transforming an analog normalized LPF into analog LPF, HPF, BPF and BSF filters using frequency transformation methods. **(06 Marks)**

c. Obtain transfer function of IIR digital filter from given $H_a(s)$, using impulse invariance method, $H_a(s) = \frac{0.5(s+4)}{(s+1)(s+2)}$. **(04 Marks)**

7 a. What are the advantages and disadvantages with the design of FIR filters using window function? **(06 Marks)**

b. Deduce the equation for the frequency spectrum for the rectangular window sequence defined by,

$$W_R(n) = \begin{cases} 1; & \text{for } \frac{-(N-1)}{2} \leq n \leq \frac{(N-1)}{2} \\ 0; & \text{otherwise} \end{cases}.$$

What is the width of main lobe of the spectrum? **(06 Marks)**

c. The frequency response of a filter is given by, $H(e^{jw}) = jw; -\pi \leq w \leq \pi$. Design the filter, using a rectangular window function. Take N = 7. **(08 Marks)**

8 a. A FIR filter is given by, $y(n) = x(n) + \frac{2}{5}x(n-1) + \frac{3}{4}x(n-2) + \frac{1}{3}x(n-3)$. Draw the Lattice structure. **(06 Marks)**

b. A discrete time system H(z) is expressed as,

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

Realize parallel and cascade forms using second order sections. **(14 Marks)**

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