BITS Pilani K K Birla Goa Campus, Semester 1, 2019-20

ECE/EEE F434 - Digital Signal Processing

Comprehensive Examination, Date -14/12/19

Max. Marks 30 | Part-A, Regular, Close Book | Time -80 Mins

Q. 1 Impulse responses of certain FIR filters are provided below. Find the expressions of $\breve{H}(\omega)$ (Amplitude response –in terms of cos or sin), and values of c and β. Note that frequency response can be expressed as $\boldsymbol{H(e^{j\omega}) = e^{j(c\omega+\beta)}\breve{H}(\omega)}$. (Expressions should be in standard and reduced form only). Here, Starting index value is n=0, and impulse response is zero at sample points other than specified. **4×3 =12M**

a) h1(n) = {2, 1, 2} b) h2(n) = {1, 1} c) h3(n) = {1, 0, -1} d) h4(n) = {1, 3, 5, 3, 1}

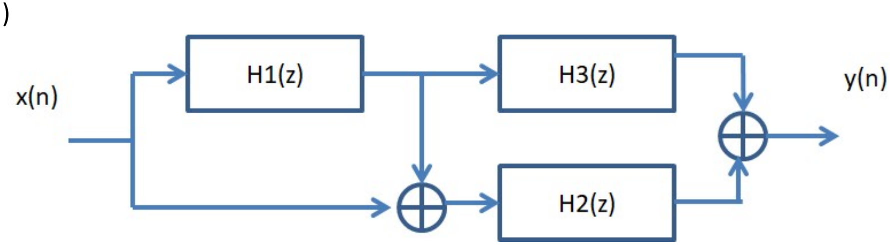
Q.2 A causal LTI discrete-time system has system function $\boldsymbol{H(z) = \frac{(1-0.5z^{-1})(1+4z^{-2})}{(1-0.64z^{-2})}}$

Find expressions for a minimum-phase system $H_1(z)$ & all-pass system $H_{ap}(z)$ such that $H(z) = H_1(z)H_{ap}(z)$. **4M**

Q.3 Consider the digital filter structure as follows, where,

$H1(z) = 2.1 + 3.3\, z^{-1} + 0.7\, z^{-2}$, $H2(z) = 1.4 - 5.2\, z^{-1} + 0.8z^{-2}$, and $H3(z) = 3.2 + 4.5\, z^{-1} + 0.9\, z^{-2}$,

Determine the transfer function H(z) of the composite filter. (Express in the standard and reduced form only.) **3M**



Q.4 Filter with the specifications as given below needs to be designed using Interpolated-FIR method. Given: $\omega_p = 0.08\pi, \omega_s = 0.1\pi, \delta_p = 0.002, \delta_s = 0.001.$ **5M**

For this design, provide the (largest) value of Sparsity factor L and Specifications of the Shaping filter, i.e., $\omega_p^{(F)}$, $\omega_s^{(F)}$ and specifications of the Interpolator, i.e. $\omega_p^{(I)}$, $\omega_s^{(I)}$.

Q.5 A linear convolution $x_3(n)$ of two sequences $x_1(n)$ and $x_2(n)$ is to be found out using DFT method. For that, solve the following as specified. **3M**

a) From a given sequence $x_1(n)$ =[4 11 3], find the 4-point DFT $X_1(k)$.
b) From $X_2(k)$ =[15 7-8j -1 7+8j], find the sequence $x_2(n)$ using 4-point IDFT.
c) Calculate $X_3(k)$ as $X_3(k) = X_1(k). X_2(k)$, find the linear convoluted sequence $x_3(n)$ using 4-point IDFT of $X_3(k)$.

Q.6 Consider that an impulse response of Type 4 (Even length and Anti-Symmetric) is available. Specify which type of filter among Low-pass, high-pass, band-pass and band-stop cannot be designed using this impulse response, and why? **3M**

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Max. Marks 30 Part-A, Regular, Close Book –Answer Sheet Time -80 Mins

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Name -____________________________________ ID no. -______________________________

Q.1 a)	C =	$\beta =$
	$\breve{H}1(\omega)$	
Q.1 b)	C =	$\beta =$
	$\breve{H}2(\omega)$	
Q.1 c)	C =	$\beta =$
	$\breve{H}3(\omega)$	
Q.1 d)	C =	$\beta =$
	$\breve{H}4(\omega)$	

Q.2	$H_1(z)$	
	$H_{ap}(z)$	

Q.4	Sparsity Factor L =		
$\omega_p^{(F)}$		$\omega_s^{(F)}$	
$\omega_p^{(I)}$		$\omega_s^{(I)}$	

Q.3	H(z) =

Q.5	$X_1(k)$	
	$x_2(n)$	
	$x_3(n)$	

Q.6	

Recheck Request if any -

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Max. Marks 80 | Part-B, Regular, Close Book | Time -180 Mins

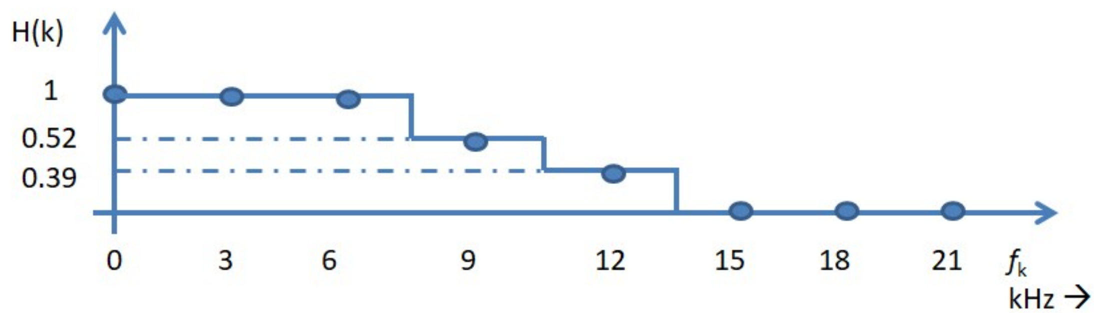
Instructions –

1) Filter should be represented in standard format only. i.e. for IIR filter T.F. should be of the form $H(z) = \frac{\sum_{k=0}^{M} b_k z^{-k}}{1+\sum_{k=1}^{N} a_k z^{-k}}$
2) Output of all the important stages should be highlighted/ enclosed in rectangular boxes.
3) All parts of the question should be solved together. New question should start on new page.
4) Assumptions if any, should be clearly specified.

Q.1 Design a Digital **Bandpass filter** with following specifications: **24M**

Lower Passband edge frequency = 2940 Hz, Upper Passband edge frequency = 4060 Hz
Lower Stopband edge frequency = 1200 Hz, Upper Stopband edge frequency = 5600 Hz
Passband ripples is 3 dB and Stopband attenuation is 14 dB. Sampling Frequency = 14000 Hz.
Design a **Butterworth Bandpass IIR filter** to satisfy stopband specifications correctly. Use Bilinear transformation with T=2. For geometric symmetry, if necessary, change the lower stopband frequency of the filter. Indicate new value of this corner frequency in Hz. Implement the filter by using D.F.II realization structure. From the transfer function, provide the values of poles, zeros and constant factor. Draw the pole-zero plot of this fiter.

Q.2 Design the following **FIR filter using frequency sampling method** **12M**



For this filter, Sampling frequency = 45 kHz and Filter length =15. Sketch the magnitude response H(k) vs. k values. Provide the impulse response of the filter in causal form, i.e. h(0), h(1), h(2) and so on. Make a table with one column of 'index n' and other column of 'filter coefficients'. State the correct formula being used for the complete credit.

Q.3 Design a lowpass filter for the following specifications using Hanning window. **14M**

$$H(e^{j\omega}) = \begin{cases} e^{-j2\omega} & |\omega| \le 0.25\pi \\ 0 & 0.25\pi \le |\omega| \le \pi \end{cases}$$

Hann window $w[n] = \frac{1}{2}\left[1 + \cos\left(\frac{n\pi}{M}\right)\right] \quad -M \le n \le M$

Provide the value of

a) Length of filter
b) h(n) (i.e. coefficients of impulse response of this filter)
c) For h(n) calculated in part b, determine whether the magnitude in passband is normalized to 1 (or 0 dB in dB scale)? if not, specify the scaling factor to be used for the same. Denote this new set of impulse response coefficients as h1(n).
d) Provide the expression for transfer function H1(z), and frequency response in the form of $H1(e^{j\omega}) = e^{j(c\omega+\beta)}\widetilde{H1}(\omega)$.
e) Tabulate the magnitude response of the filter as per the given table.

Omega (in radians)	0	0.2	0.4	0.6	0.8	1
Mag (Normalized to 1)	1					
Mag (in dB)	0 dB					

f) Draw the optimum direct form structure for the designed filter using H1(z). (Optimum – to save no. of coefficients).
g) What difference in shape of the filter that you observe between the filter asked to design in the question and the one which you have designed. Specify how this difference can be minimized?

Table -1 : 3 dB Butterworth lowpass prototype transfer functions ($\varepsilon = 1$)

N (order)	H(s) (Transfer Function)
1	$\frac{1}{s+1}$
2	$\frac{1}{s^2 + 1.4142\,s + 1}$
3	$\frac{1}{s^3 + 2s^2 + 2\,s + 1}$
4	$\frac{1}{s^4 + 2.6131\,s^3 + 3.4142\,s^2 + 2.6131\,s + 1}$