姓名 学号 级 密 封 专业 线 学院

山东大学 2019-2010 学年 一 学期 数字信号处理 课程试卷（A）

题号	一	二	三	四	五	六	七	八	九	十	总分	阅卷人
得分												

备注：闭卷，可以带计算器

得分	阅卷人

一、Fill the blanks(40%)

1. Let there is a continuous time sequences x(t)= x1(t)+ x2(t)+ with two continuous components, x1(t)= cos(3πt), x2(t)= cos(7πt), x3(t)= cos(10πt), if the combined digital signal is x[n]= x1[n]+ x2[n]+x3[n], then

(1) if sampled by 10Hz, then the signal will get aliasing or not ①and then sampled signal in time domain x[n]=②;t

(2) based on (1) , to filter the x1[n] by passing x[n] through a digital filter, choose one kind of digital filter ③and determine its cut-off frequency④

(3) based on (2), determine the period of x1[n], N=⑤.

2. If $x[n]=\{1,\ 4,\ 5,-4,\ -1\},-2\le n\le 2$, then

(1) $x[n]$ can be expressed in terms of the unit impulse signal $\delta[n]$ as ⑥;

(2) If the impulse response of a LTI system is h[n]=$\{1,\ 4,\ 5\},\ 0\le n\le 3$, then given by x[n], the output sequence y[1]=⑦and the range of y[n] is between ⑧and⑨;

(3) if calculate $y_c[n]$ by 6-points circulation convolution, the value of $y_c[1]$= ⑩;

(4) without computing the DTFT, $X(e^{j0})=$ ⑪; $\int_{-\pi}^{\pi}|X(e^{j\omega})|^2\,d\omega$=⑫;

(5) to make h[n] to be a Type1 linear phrase FIR, then write out one of the possible new linear phase h'[n]= ⑬.

(5) Without computing the DTFT, determine what kind of $X(e^{j\omega})$ is ⑭; (real-valued or imaginary-valued or others types of DTFT).

3. If x[n] is a length-120 sequence and h[n] is a FIR filter with $0\le n\le 4$. If using 10-points overlap method to computer y[n] , then the length of each small segment of x[n] is ⑮and their whole length output y[n] is ⑯.

4. Some samples of the 5-point DFT of a length-5 real sequence are given by X[0]=-4.7, X[2]=1.2-j2, X[4]=-3.5+j3. The X[1] should be ⑰.

5. An IIR digital filter has the unit pulse response $h[n]=(0.5)^n\mu[-n]+(0.2)^n\mu[n]$, then

(1) z-transform of H(z) in closed form is ⑱ and its R.O.C is ⑲.

(2) whether h[n] is BIBO stable or not ⑳.

得分	阅卷人

二、Comprehensive problems(60%)

1. (40%) A causal LTI system is described by the recursive difference equation

$$y[n]=2x[n]-x[n-1]+\frac{7}{12}y[n-1]-\frac{1}{12}y[n-2]$$

(1) Find the transfer function of H(z) and its R.O.C.(6%)

(2) Draw the diagram of the system in parallel form. (8%)

(3) If using bilinear form to transfer the digital filter to analog filter, write out the transform equation. (4%)

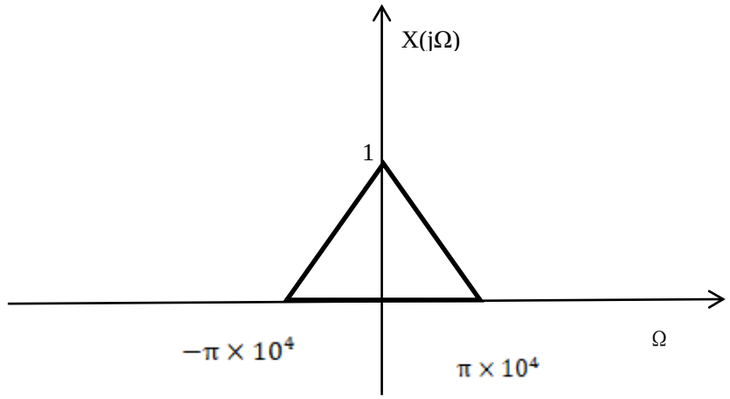
(4) Find the impulse response h[n] by solving differential equations. (8%)

(5) Write out the magnitude function of the frequency response $H(e^{j\omega})$.(6%)

(6) If using FIR filter to approximate h[n] and N=10, then what its magnitude function of the frequency response $H(e^{j\omega})$ will be. (6%)

(7) How to make (6) to be linear phase.(2%)

2. (20%) For a continuous time signal x(t) with frequency spectrum X(j Ω), which $-\pi\times10^4 r/s\le\Omega\le\pi\times10^4 r/s$ as following.



(1) Plot corresponding frequency spectrum of $X(e^{j\omega})$ and Xs(jΩ) with a proper sampling period T=0.5×10^{-4}s (5%).

(2) If there is a LPF H($e^{j\omega}$) with cut-off frequency $-\pi/4\le\omega_c\le\pi/4$, Plot the frequency spectrum of Y($e^{j\omega}$)and its 10-points DFT Y[k] .(5%)

(3) Determine the range of sampling rate Ω_s if Ys(jΩ) is reconstructed by filter in (2)without aliasing and Draw the Ys(jΩ) . (5%)

(4) Design an antialiasing filter if T=0.2×10^{-4}s and plot the frequency spectrum of Xs(jΩ)after using anti-aliasing filter . (5%)