Seat No.: _____ Enrolment No.______

GUJARAT TECHNOLOGICAL UNIVERSITY

M.E Sem-I Remedial Examination January/ February 2011

Subject code:710422

Subject Name: Digital Signal Processing and Applications

Date: 05/02 /2011 **Time: 02.30 pm – 05.00 pm**

Total Marks: 60

Instructions:

1. **Attempt all questions.**
2. **Make suitable assumptions wherever necessary.**
3. **Figures to the right indicate full marks.**

Q.1 **(a)** Give answer of following questions. **06**

i) For the system T(x[n]) = x[n] +3u[n+1], determine whether system is causal, linear and time-variant or not.

ii) Prove the condition for LTI system stability. Test the stability of the system whose impulse response $h(n)=(1/2)^n u[n]$

iii) Prove that for a causal LTI system, impulse response h(n)=0 for n<0.

(b) Give answer of following questions. **06**

i) Define unit sample response of a system and state its significance.

ii) Distinguish between DFT and DTFT.

iii) Determine the initial and final values of the sequence x[n] given $X[Z] = 2+3z^{-1}+4z^{-2}$

Q.2 **(a)** Give answer of following questions. **06**

i) Find the step response of the system described by the differential equation y[n]+y[n-1]=x[n]-2x[n-1]

ii) When the input to an LTI system is

$$x[n]=\left(\frac{1}{3}\right)^n u[n]+(2)^n u[-n-1],$$

the corresponding output is $y[n]=5\left(\frac{1}{3}\right)^n u[n]-5\left(\frac{2}{3}\right)^n u[n]$

1. Find the system function H(z) of the system.
2. Indicate region of convergence in pole-zero plot.

(b) Give answer of following questions. **06**

i) Find inverse z-transform of

$X(z)=\dfrac{1}{1+0.5z^{-1}}$ using long division method if

1. ROC $|z| > 0.5$
2. ROC $|z| < 0.5$

ii) Find inverse z-transform of

$X(z)=\dfrac{z(z^2-4z+5)}{(z-3)(z-1)(z-2)}$ ROC: $2 < |z| < 3$

OR

(b) How to get linear convolution from circular convolution? Find linear convolution of a given sequences x[n]={1,2} & h[n]={2,1} using DFT. **06**

Q.3 **(a)** Give answer of following questions. **06**

i) Write briefly about overlap-save method. State the differences between overlap-save method and overlap-add method also.

ii) How does FFT help in computation savings? What is the saving for 1024 point DFT if FFT is used?

(b) Differentiate Decimation in time and Decimation in frequency FFT algorithm. Explain any one with example. **06**

OR

Q.3 (a) Give answer of following questions. **06**

i) State the advantage of Kaiser window. Find the parameters β and M of the Kaiser window for the filter that meets following specifications:

$|H(\omega)| \leq 0.01 \quad ,0 \leq |\omega| \leq 0.25\pi$

$0.95 \leq |H(\omega)| \leq 1.05 \quad ,0.35\pi \leq |\omega| \leq 0.6\pi$

$|H(\omega)| \leq 0.01 \quad , 0.65\pi \leq |\omega| \leq \pi$

ii) The analog signal has a bandwidth of 4 KHz. If we use N-point DFT with $N=2^m$ (m is an integer) to compute the spectrum of the signal with resolution less than or equal to 25 Hz. Determine the minimum number of required samples and the minimum length of the analog signal record.

(b) Realize the IIR system described by the differential equation $y[n]+0.25y[n-1]-0.125y[n-2] = x[n]-2x[n-1]+x[n-2]$ in each of the following forms: **06**

i) Direct form I

ii) Cascade form

iii) Parallel form

Q.4 (a) Compare FIR and IIR filters. What is the principle of designing FIR filter using windows? **06**

(b) Design a low pass discrete time filter by applying impulse invariance method to an appropriate Butterworth continuous time filter for the following specifications. **06**

i) Passband magnitude is constant within 1 db for frequencies below 0.2 π.

ii) Stopband attenuation is greater than 15 db for frequencies between 0.3π and π. Consider T is equal to unity.

OR

Q.4 (a) Discuss effects of finite word length in FIR digital filters. Also brief on effect of coefficient quantization error. **06**

(b) What are the non-parametric methods of power spectrum estimation? How DFT is useful in power spectral estimation? **06**

Q.5 (a) Give answer of following questions. **06**

i) What is the need for Multirate digital signal processing? What is the need for anti-aliasing filter prior to downsampling?

ii) Discuss how sampling rate can be increased by an integer factor. What is the corresponding effect on the frequency domain? If the z-transform of a sequence x[n] is X[z] then what is the z-transform of a sequence upsampled by a factor M?

(b) Give answer of following questions. **06**

i) Describe application of DSP to radar system in brief.

ii) What is Gibbs phenomenon?

OR

Q.5 (a) Give answer of following questions. **06**

i) Describe Harvard Architecture.

ii) What is pipelining in DSP processor?

(b) Give answer of following questions.

i) Compare the fixed point and floating point arithmetic for DSP Processors. **06**

ii) Compare the architecture of general purpose microprocessors and DSP processors. Also discuss the features of DSP processors.

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