B.E. / B.Tech (Full time) ARREAR EXAMINATIONS, NOV / DEC 2011

ELECTRONICS AND COMMUNICATION ENGINEERING BRANCH

FIFTH SEMESTER

(REGULATION 2008)

EC 9304 DIGITAL SIGNAL PROCESSING

Time: 3 Hours Max.marks: 100

PART-A (10X2=20 Marks)

1. If LTI system with input x[n] and output y[n] characterized by its unit sample response $h[n] = a^n u[n]$ for 0 < a < 1. Find y[n] of such a system to the input signal x[n] =u[n].
2. State and prove the circular shift property of DFT.
3. List out the advantages and disadvantages of FIR filter.
4. What is meant by Gibbs phenomenon?
5. An analog filter has a system function $H_a(s) = \frac{1}{s+1}$. A digital filter is obtained using bilinear transform technique. At what digital frequency w_0 will $20 \log\left|H(e^{jw_0})\right|$ equal -3dB? Assume T=2sec.
6. Mention the characteristics of Chebyshev filters.
7. In a pipeline machine, the instruction fetch, decode and execute take 35ns, 25ns, and 40ns respectively. Determine the increase in throughput if the instruction steps were pipelined. Assume a 5ns pipeline overhead at each stage.
8. Distinguish between Von Neumann architecture and Harvard architecture.
9. Show that the up-sampler is a time-varying system.
10. Draw the transversal and efficient transversal structure of a decimator with factor M=4.

PART-B (5X16=80 Marks)

11. (i) Design a 9 tap FIR linear phase digital filter (use Hamming window) approximating the ideal frequency response $H_d(e^{jw}) = \begin{cases} 1 & 0 \le |w| < 0.3\pi \\ 0 & 0.3\pi < |w| < 0.6\pi \\ 1 & 0.6\pi < |w| \le \pi \end{cases}$. (8)

    (ii) Write the equations of various window functions for FIR filter design and compare their important frequency domain characteristics. (8)

12. (a) If the DFT of real x[n] is X(k), compute the 8 point IDFT for the given X[k] = {0, 2-j4.828, 0, 2-j0.8284, 0, _, _, _} using the in place radix 2 decimation in time algorithm. **(16)**

(Or)

12. (b) (i) Compute the circular convolution of the following two sequences using DFT and IDFT. x[n] =[1, 1, -1, -1] and h[n]=[-1, -1, 1, 1]. **(8)**

(ii) Determine the convolution output for the following two sequences using overlap save method. x[n]=[1, 2, 3, 3, 2, 1,1, 3, 2] and h[n]=[1, -1, 1] . **(8)**

13. (a) Design a low pass digital IIR filter using bilinear transform technique meeting the following specifications: pass band frequency: 0-500Hz, stop band frequency: 2-4 kHz, pass band ripple: 3dB, stop band attenuation: 20dB and sampling frequency 8 kHz. Assume Butter worth characteristics for the filter. **(16)**

(Or)

13. (b) (i) Obtain parallel realization for the following H(z) using a partial fraction expansion $H(z) = \dfrac{8z^3 - 4z^2 + 11z - 2}{\left(z - \frac{1}{4}\right)\left(z^2 - z + \frac{1}{2}\right)}$. **(8)**

(ii) Convert the analog filter with system function $H_a(s) = \dfrac{s + 0.9}{(s + 0.9)^2 + 16}$ into a digital IIR filter by means of impulse invariance method. **(8)**

14. (a) (i) Represent the decimal number 0.35 in binary format with 4 bit (including sign bit) using following representations: sign magnitude with truncation, sign magnitude with round off, 2's complement with truncation and 2's complement with round off. Also determine the associated quantization error ranges. **(8)**

(ii) The first order filter described by the linear difference equation is y(n)=ay(n-1)+x(n). If the products are rounded to four bit representation (including sign bit) using the input x(n)=0.1δ(n), Determine: (I) The first five outputs if a=0.5. Does the filter go into limit cycle oscillations? (II) The first five outputs if a= 0.75. Does the filter go into limit cycle oscillations? **(8)**

(Or)

14. (b) (i) Determine the round off noise power at the output of the two cascade realization of the filter with transfer function $H(z) = H_1(z).H_2(z)$, where $H_1(z) = \dfrac{1}{\left(1 - 0.5z^{-1}\right)}$ and $H_2(z) = \dfrac{1}{\left(1 - 0.25z^{-1}\right)}$. **(8)**

(ii) With neat diagram, explain the working of a multiplier- accumulator (MAC) with and without pipelining. **(8)**

15. (a) Derive the relationship between input and output signals of decimator and interpolator. With neat diagram, explain the frequency domain characteristics of decimator and interpolator. **(16)**

(Or)

15. (b) (i) With neat diagram, explain the two channel sub band coding filter bank. **(8)**

(ii) The FIR low pass filter is to be realized, with cutoff frequencies fp=30Hz, fs=40Hz and a sampling rate f_0=8 kHz. The pass- band ripple is to be δp=0.01 and the stop-band ripple δs=0.001. The filter is to be realized using a single stage decimator and interpolator. If the interpolation and decimation filters are chosen to be equal with factor M=8, calculate the total complexity. **(8)**