(b) Discuss in details about the decimation and interpolation concept by frequency conversion.

(c) Explain with block diagram, M-channel quadrature mirror filter (QMF) bank.

4 + 5 + 3 = 12

9. Write short notes on any three of the following. **(3 × 4) = 12**
   (i) Energy Spectral Density.
   (ii) Application of Adaptive filters.
   (iii) Frequency convolution.
   (iv) Wiener- Khictchin theorem.

ADVANCED DSP & APPLICATIONS
(ECEN 5202)

Time Allotted : 3 hrs **Full Marks : 70**

Figures out of the right margin indicate full marks.

Candidates are required to answer Group A and any 5 (five) from Group B to E, taking at least one from each group.

Candidates are required to give answer in their own words as far as practicable.

Group – A
(Multiple Choice Type Questions)

1. Choose the correct alternative for the following: **10 × 1 = 10**

(i) What is the DTFT of the following sequence given as $x(n) = \delta(n)$?
(a) -1 (b) 1
(c) 0 (d) ∞.

(ii) What is the process of reducing the sampling rate by a factor D?
(a) Sampling rate conversion (b) Interpolation
(c) Decimation (d) None of the mentioned.

(iii) Which of the following is done to convert a continuous time signal into discrete time signal?
(a) Modulating (b) Sampling
(c) Differentiating (d) Integrating.

(iv) DTFT is a special case of
(a) Z-transform (b) laplace transform
(c) continuous time fourier transform (d) none.

(v) What is the nyquist rate of the signal x(t)=3cos(50*pi*t) + 10sin(300*pi*t) – cos(100*pi*t)?
(a) 50 Hz (b) 100 Hz
(c) 200 Hz (d) 300 Hz.

(vi) ROC is defined as the range of values of z for which X(z)
(a) converges (b) diverges
(c) is zero (d) is finite.

(vii) What are the main characteristics of anti-aliasing filter?
(a) Ensures that bandwidth of signal to be sampled is limited to frequency range
(b) To limit the additive noise spectrum and other interference, which corrupts the signal
(c) Both a and b
(d) None of the mentioned.

(viii) What is the process of converting a signal from a given rate to a different rate?
(a) Sampling
(b) Normalizing
(c) Sampling rate conversion
(d) None of the mentioned.

(ix) The frequency response of LTI system is given by Fourier transform of _______ of the system.
(a) transfer function (b) output
(c) impulse response (d) input

(x) $\delta(n-k) * x(n-k)$ is equal to
(a) x(k) (b) x(n-k)
(c) x(n-2k) (d) $\delta(n-k)$.

Group - B

2. (a) Consider a discrete-time LTI system with impulse response $h(n)=(0.5)^n u(n)$. Use Fourier transform to determine the response to the signal $x(n) = (3/4)^n u(n)$.

(b) The impulse response of a LTI system is given by $h(n) = 0.6^n u(n)$. Find the frequency response, magnitude and phase response.

6 + 6 = 12

3. (a) State and explain Parseval's theorem pertaining to discrete time Fourier transform.

(b) Find the impulse response and frequency response of a discrete time LTI system that has foregoing property, if the input and output are given by x(n) and y(n) respectively.
$x(n) = (0.5)^n u(n) - (0.25)(0.5)^{n-1} u(n-1)$ and $y(n) = (0.5)^n u(n)$.
Also, find the difference equation relating x(n) and y(n) that characterizes the system.

5 + 7 = 12

Group - C

4. (a) Determine the circular convolution of the following sequences and compare the results with linear convolution.
$x(n) = \{1,1,1,1,-1,-1,-1,-1\}$, and
$h(n) = \{0,1,2,3,4,3,2,1\}$.

(b) Compute DFT of the following finite length sequences considered to be of length N.
$x(n) = a^n$ for $0 \le n \le N-1$.

(c) What do you mean by zero padding?

6 + 4 + 2 = 12

5. (a) Explain the use of the FFT algorithm in linear filtering and correlation.

(b) Compute 4 point DFT of the sequence; $x(n) = \sin(n\pi/2)$, where, $0 \le n \le 3$ using decimation in time FFT algorithm.

5 + 7 = 12

Group - D

6. (a) Explain the bilinear transformation method for converting the system function of analog filter into system function of digital filter.

(b) Design a band pass filter (linear phase FIR filter) using windows method.

6 + 6 = 12

7. Design low pass Chebyshev filter using impulse invariant method for satisfying the following constraints.
(i) Pass band ω_p : 0.162 rad
(ii) Stop band ω_s : 1.63 rad
(iii) Pass band ripple : 3 dB
(iv) Stop band attenuation : 30 dB
(v) Sampling frequency : 8 kHz.

12

Group - E

8. (a) What are fixed and adaptive filters?