16.

Exam.Code:1005
Sub. Code: 7694

1128
M. E. (Information Technology)
First Semester
MEIT-103/MEIT-113: Advanced Digital Signal Processing

Time allowed: 3 Hours **Max. Marks: 50**

NOTE: Attempt five questions in all, including Question No. I (Part-I) which is compulsory and selecting two questions each from Part II - III.

x-x-x

Part-A

I (a) What is differentiation property of Z transform? 10
(b) Why FIR filters are called linear filters?
(c) Why and how time domain signal is converted into frequency domain?
(d) What is LMS algorithm?
(e) What is active noise cancellation ?

Part-B

II (a) Perform the Z transform of the signal using various properties 5
$x(n) = n^2 u(n-5)$ Also find ROC for $x(n)$
(b) Find inverse Z-transform of the signal, 5
$X(z) = \frac{1}{(z-2)(z-1)}$ ROC: $|z| > 2$.

III. Find IFFT of the signal. Signal is 10
$X(K) = \{6, (-2+j2), -2, (-2-2j)\}$

IV A low pass filter is to be designed with the following desired frequency response. Determine filter coefficients for rectangular window for the window length 4. 10
$H_d(e^{j\omega}) = \{e^{-3j\omega}, \ -\pi/4 \le \omega \le \frac{\pi}{4} \quad and\ 0\ for\ \pi/4 \le \omega \le \pi$

Part-C

V. Perform Direct form I, direct form II, cascade and parallel realization of the following 10
$H(z) = \frac{1-25z^{-1}}{(1+z^{-1}-72z^{-2})}$

VI. (a) What are adaptive filters? Explain using block diagram. 5
(b) What is the use of decimation and interpolation? Explain decimation and interpolation by using an example. 5

VII Write note on the following 10

(a) Subband coding of speech signal
(b) Multirate signal processing.

x-x-x