2008 Paper 9 Question 11

Digital Signal Processing

(*a*) A radio system outputs signals with frequency components only in the range 2.5 MHz to 3.5 MHz. The analog-to-digital converter that you want to use to digitise such signals can be operated at sampling frequencies that are an integer multiple of 1 MHz. What is the *lowest* sampling frequency that you can use without destroying information through aliasing? [5 marks]

(*b*) Consider a digital filter with an impulse response for which the z-transform is

$$H(z) = \frac{(z+1)^2}{(z-0.7-0.7\mathrm{j})(z-0.7+0.7\mathrm{j})}$$

(*i*) Draw the location of zeros and poles of this function in relation to the complex unit circle. [2 marks]

(*ii*) If this filter is operated at a sampling frequency of 48 kHz, which (approximate) input frequency will experience the lowest attenuation? [2 marks]

(*iii*) Draw a direct form I block-diagram representation of this digital filter. [5 marks]

(*c*) Make the following statements correct by changing one word or number in each case. (Negating the sentence is not sufficient.)

(*i*) Statistical independence implies negative covariance.

(*ii*) Group 3 MH fax code uses a form of arithmetic coding.

(*iii*) Steven's law states that rational scales follow a logarithmic law.

(*iv*) The Karhunen–Loève transform is commonly approximated by the z-transform.

(*v*) 40 dB corresponds to an 80× increase in voltage.

(*vi*) The human ear has about 480 critical bands.

[6 marks]