UNIVERSITY OF SWAZILAND

MAIN EXAMINATION, SECOND SEMESTER MAY 2011

FACULTY OF SCIENCE

DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING

TITLE OF PAPER:	DIGITAL SIGNAL PROCESSING
COURSE CODE:	E420
TIME ALLOWED:	THREE HOURS

INSTRUCTIONS:

1. **Answer all five questions.**
2. **Each question carries 20 marks.**
3. **Tables of selected Z-transform pairs are attached.**

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THIS PAPER CONTAINS SEVEN (7) PAGES INCLUDING THIS PAGE

QUESTION ONE ***(20 marks)***

(a) Name 4 advantages and 4 applications of DSP. *(4 marks)*

(b) Draw a block diagram of a typical DSP system and briefly describe the function of each block. *(7 marks)*

(c) The transfer function of an IIR filter is given as

$$H(z) = \frac{0.5 - 0.5z^{-2}}{1 + 1.3z^{-1} + 0.36z^{-2}}$$

(i) Derive and sketch the Direct Form I realization of the filter. *(5 marks)*

(ii) Derive and sketch the Direct Form II realization of the filter. *(3 marks)*

(iii) What advantage has the Direct Form II realization over the Direct Form I realization? *(1 mark)*

QUESTION TWO **(20 *marks)***

A 3-tap FIR lowpass digital filter has cutoff frequency of 1800 Hz and a sampling arte of 8 kHz.

(a) Using the Fourier Transform method, calculate its filter coefficients.

You are given that the ideal impulse response for a low pass FIR filter is given by

$$h(n) = \begin{cases} \dfrac{\Omega_c}{\pi}, & n = 0 \\ \dfrac{\sin(n\Omega_c)}{n\pi}, & -M \le n \le M \end{cases}$$

(8 marks)

(b) Determine the transfer function and difference equation of the designed filter.

(2 marks)

(c) Calculate and plot its magnitude and phase frequency response for $\Omega = 0, \pi/4, \pi/2, 3\pi/4$ and π rad. *(10 marks)*

QUESTION THREE **(20 *marks)***

a) Explain recovery of analog signal from its sampled signal version. Discuss the different cases for recovery of the original signal spectrum. *(10 marks)*

b) Analog signal is given by

$$x(t) = 5\cos(2\pi.2000t) + 3\cos(2\pi.3000t)$$

and it is sampled at the rate of 8000 Hz.

(i) Sketch the spectrum of the sampled signal up to 20 kHz.

(ii) Sketch the recovered analog signal spectrum if an ideal low pass filter with a cutoff frequency of 4 kHz is used to filter the sampled signal to recover the original signal.

(4 marks)

c) A 3-bit ADC channel accepts analog input ranging from 0 to 5 volts determine:

(i) Number of quantization levels

(ii) Step size of the quantizer

(iii) Quantization level when the analog voltage is 3.2 volts.

(6 marks)

QUESTION FOUR **(20 *marks)***

a) Explain the basic mechanism of circular buffering for a buffer having eight data samples. *(8 marks)*

b) What are the differences between general microprocessor to that of digital signal processor? Draw architecture of both processors. *(8 marks)*

c) Find the signal Q-15 representation for the decimal number 0.560123. *(4 marks)*

QUESTION FIVE **(20 *marks*)**

a) What is meant by Quantization Error? *(2 marks)*

b) Sketch a neat circuit diagram of a 2-bit Digital-to-Analogue Converter which uses R – 2R ladder, and explain its operation. *(8 marks)*

c) What is meant by fixed-point and floating-point format in DSP? Give one example of each format. Discuss and compare the operation of fixed-point and floating–point digital signal processors. *(10 marks)*

TABLE OF Z-TRANSFORMS OF SOME COMMON SEQUENCES

Discrete-time sequence $x(n),\ n \geq 0$	**Z-transform** $H(z)$
$k\delta(n)$	k
k	$\frac{kz}{z-1}$
$ke^{-\alpha n}$	$\frac{kz}{z-e^{-\alpha}}$
$k\alpha^n$	$\frac{kz}{z-\alpha}$
kn	$\frac{kz}{(z-1)^2}$
kn^2	$\frac{kz(z+1)}{(z-1)^3}$
$kn\alpha^n$	$\frac{k\alpha z}{(z-\alpha)^2}$