UNIVERSITI SAINS MALAYSIA

First Semester Examination

Academic Session 2004/2005

October 2004

EEE512 / EEE502 – ADVANCED DIGITAL SIGNAL AND IMAGE PROCESSING

Time : 3 Hours

INSTRUCTION TO CANDIDATE:-

Please ensure that this examination paper contains **SEVEN (7)** printed pages and **SIX (6)** questions before answering.

Answer **FIVE (5)** questions.

Distribution of marks for each question is given accordingly.

All questions must be answered in English.

Note : Use SI system of units

Assume suitable data with justification where necessary.

1. (a) A discrete signal is corrupted by additive random noise d(n) resulting in the noisy signal x(n) = s(n) + d(n). A two point causal moving average filter is to be designed that will operate on x(n) to give an output y(n) that is a reasonable approximation to s(n). Determine the impulse response and the transfer function of such a filter.

(50%)

(b) Obtain the expression for the output y(n) in terms of the input x(n) for the multirate system given in Figure 1.

(50%)

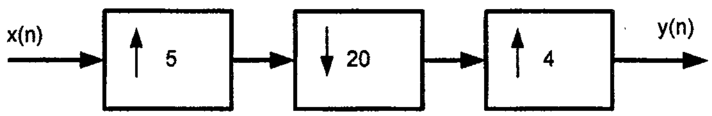

Figure 1

2. (a) The following difference equation represents a linear time invariant system.

$$y(n) = x(n) + \frac{1}{4}x(n-1) - y(n-1) + y(n-2).$$

Obtain the frequency response of the system. (50%)

(b) A DSP chip used in real-time signal processing applications has an instruction cycle time of 100 ns. One of the instructions in the instruction set MACD will fetch a value from data memory (input signal), fetch another data value from program memory (filter coefficient), multiply the two numbers together, add the product to the accumulator, and then move a number in data memory into the next memory location (this corresponds to a shift or delay of the data sequence). Thus, for an FIR filter of order N to find the value of the output at times n, one instruction to read the new input value, x(n), into the processor, we need (N+1) MACD instructions to evaluate the sum

$$y(n) = \sum_{k=0}^{N} h(k)x(n-k)$$

and we need one instruction to output the value of y(n). In addition there are eight other cycles required for each n in order to perform such functions as setting up memory pointers, zeroing the accumulator and so on.

(50 %)

[i] With the above requirements in mind determine the maximum bandwidth signal that may be filtered with an FIR filter of order N=255, in real time, using a single DSP chip.

[ii] A speech waveform $x_a(t)$ is sampled at 8kHz.Determine the maximum length FIR filter that may be used to filter the sampled speech signal in real time.

3. (a) Consider a LTI system with a system function

$$H(z) = \frac{1 - 0.4z^{-1}}{(1 - 0.6z^{-1})(1 - 0.8z^{-1})}$$

Suppose that the system is implemented on a 16-bit fixed-point processor and that the sums of the products are accumulated prior to quantization. Let σ^2 be the variance of the round-off noise. If the system is implemented in direct form II, find the variance of the round-off noise at the output of the filter.

(50%)

(b) In order to cut-off the very low frequency components of speech signal a high pass digital filter is required. The filter should have the following specifications

Cut-off frequency 30 Hz
Sampling frequency 150 Hz.

Starting from a simple low pass analog filter with transfer function 1/(1+s), determine the transfer function of the digital high pass filter. Also comment on the stability of the system.

Take the analog low pass to high pass frequency transformation as $s \rightarrow \Omega_c \, \Omega^*_c / s$ where the symbols have their usual meaning.

(50%)

4. (a) Consider the following 8×8 image

$$A = \begin{bmatrix} 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 10 & 10 & 10 & 10 & 0 & 0 \\ 0 & 0 & 10 & 10 & 10 & 10 & 0 & 0 \\ 0 & 0 & 10 & 10 & 10 & 10 & 0 & 0 \\ 0 & 0 & 10 & 10 & 10 & 10 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{bmatrix}$$

Perform filtering in spatial domain using the filter function defined as follows:

$$g(x,y)=\frac{1}{4}\left[f(x,y+1)+f(x+1,y)+f(x-1,y)+f(x,y-1)\right]$$

(40 %)

(b) A filtered function in spatial domain is given by,

$$g(x,y)=f(x,y)-f(x+1,y)+f(x,y)-f(x,y+1)$$

Prove that the filter function $H(u,v)$ for performing the equivalent process in the frequency domain is given by,

$$H(u,v)=-2j\left[\sin\left(\frac{\pi u}{M}\right)e^{j\frac{\pi u}{M}}+\sin\left(\frac{\pi v}{N}\right)e^{j\frac{\pi v}{N}}\right]$$

(40 %)

(c) Without detailed mathematical calculation, show that $H(u,v)$ is a high pass filter.

(20 %)

Given

$$f(x-x_0,y-y_0)=F(u,v)e^{-j2\pi\left(\frac{ux_0}{M}+\frac{vy_0}{N}\right)}$$

$$2j\sin(x)=e^{jx}-e^{-jx}$$

6. (a) Briefly explain some unique properties of Haar and Walsh-Hadamard transforms for image compression.

(40 %)

(b) Consider the following 8×8 image

$$f = \begin{bmatrix} 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{bmatrix}$$

(i) Construct the Haar matrix needed to perform the transformation of f.

(15 %)

(ii) Perform Haar transformation on f.

(15 %)

(iii) Reconstruct f using the first two Haar basis images.

(20 %)

(iv) From (iii), calculate the mean square of reconstruction.

(10 %)

<u>Given</u>

Haar functions are defined as:

$$H_0(t) = 1 \;;\; 0 \le t < 1$$

$$H_1(t) = \begin{cases} 1 \;;\; 0 \le t < \frac{1}{2} \\ -1 \;;\; \frac{1}{2} \le t < 1 \end{cases}$$

$$H_{2\;p\;+1}(t) = \begin{cases} \sqrt{2^p} \;;\; \frac{n}{2^p} \le t < \frac{n+0.5}{2^p} \\ -\sqrt{2^p} \;;\; \frac{n+0.5}{2^p} \le t < \frac{n+1}{2^p} \\ 0 \;;\; elsewhere \end{cases}$$