Sample Questions
CS4551 Multimedia Software Systems

- Solutions are provided for some questions requiring arithmetic computations. The intermediate computation steps may not be provided. However, you SHOULD provide equations and core intermediate computation steps when you take any exams. Otherwise, you will not receive full credit.
- In addition to these questions, all examples (if any) covered during lectures will be considered for the exam.

- Name at least four different media types introduced in the class.

- Explain the difference between multimedia and hypermedia

- What is the core difference between image type and 2D graphics type?

- What is the difference between 8-bit color and 24-bit color image?

- Explain qualitatively

  - What kind of images will give good visual results when quantized with the uniform color quantization scheme.
  - What kind of images won't give a good performance?
  - Explain why the median cut algorithm for color quantization performs very well in such cases.

- What kind of compression algorithm does GIF use?

- Explain why the following matrix: $D = \begin{pmatrix} 0 & 6 & 3 \\ 1 & 7 & 4 \\ 2 & 8 & 5 \end{pmatrix}$ is not a good dithering matrix. Show what happens when the input image has constant brightness $I=6$.

- What does the Error diffusion do?

- What are two steps of digitization process?

- What is Nyquist-Shannon's Sampling Theorem?

- What is the role of pre-filtering for a signal that has band B?

- Suppose we are recording music using a microphone (mono) with cut-off frequency of 44 Khz. We use an ADC (Analog-to-Digital Converter) to digitize the output of the microphone, and then we write the digital signal onto a disk with size 120 MB (1MB=106 bytes). What is the minimum sampling period we should can use if we want to be able to record at least 30 minutes of music (assuming the signal is quantized with 16 bits) with the least aliasing?

  Ans: 0.00003 s (second) or 0.03 ms (millisecond) or 30µs (microsecond)

- Suppose we are recording music using a microphone with cut-off frequency of 30 Khz. We use an ADC (Analog-to-Digital Converter) to digitize the output of the microphone, and then we write the digital signal onto a disk with size 100 MB (1MB=10^6 bytes).

  - What is the ***minimum sampling period*** we should use if we want to be able to record at least 15 minutes of music using 16 quantization bits?

Ans:
Disk size = $8 \cdot 10^8$ bits. 15 minutes = 900 seconds.
Data size = $900 \cdot 16 \cdot R$ must be ≤ disk size
Maximum sampling frequency R = 55,555.56 Hz

Minimum sampling period T = 1/R = 0.018 ms

- What is the bandwidth of the pre-filter we should use in order to avoid aliasing?

Ans: Anti-aliasing filter bandwidth = R/2 = 27,777.78 Hz

- With reference to question above, suppose that we have recorded 15 minutes of music on the disk. Now we want to copy the music content onto a disk which contains only 60 MB, by subsampling the original signal.

  - Which subsampling ratio should we use to make sure that all 15 minutes of music will fit into the new disk using 16 quantization bits?

  Ans: 60 MB = 3/5 · 100 MB
  Subsampling ratio N must be such $1/N \le 3/5$ (or N > 5/3) therefore N=2

  - What is the bandwidth of the pre-filter to avoid aliasing?

  Ans: Antialiasing filter bandwidth = B/2 = **27,777.78 /2** = 13,888.89 Hz

- Suppose that we have 15 minutes of music on the 100 MB ($1MB=10^6$ bytes) disk, which is digitized with the sampling rate *f* and 16 quantization bits. Now we want to copy the music onto a 60MB disk. Instead of subsampling, we want to reduce the size of the digital signal by performing a coarser uniform quantization on the already digitized signal.

  o How many quantization levels should we use to make sure that all 15 minutes of music will fit into the new disk?

  Ans: 16bits should reduce to 9 bits since the disk size reduced from 100MB to 60MB. Therefore, 2^9 quantization levels.

  o What is the difference in quantization SNR between the signal before and after the coarser quantization? (Recall how to compute SNR and its measurement unit. For every bit we add/deduct when quantizing an input sample, by how much does the quantization SNR increase/decrease approximately.)

  Ans: 16-9 = 7 bit reduction. Therefore 7 * 6 = 42 dB decrease.

- Enumerate at least four names of the lossless compression techniques. Pick one technique, describe the advantage(s) and disadvantage(s) of the technique, and name one standard compression method that uses the technique.

- What is the entropy of {0.25, 0.25, 0.25, 0.25}?

  Ans: 2

- What is a prefix coder?

- Suppose we are working with text files with only 5 symbols: {a,b,c,d,e}. After examining many files, we find that, on the average, 5% of the files have symbol a, 5% of the files have symbol b, 10% of the files have symbol c, 10% of the files have symbol d, and the remaining texts of the files have symbol e. We want to design a lossless (entropy) codec that operates on a text file.

  1) What are possible prefix codes for each symbol using the Huffman algorithm?

Ans:

```
000 –a
```

```
001 - b
010 - c
011 - d
1 - e
```

2) What is the lowest average size (in terms of bits per symbol) that an entropy code can achieve?

Ans: 1.45

3) What is the coding efficiency?

Ans: 0.90625

- Given the initial code below, encode "AAABBCCBBCCA" string using the adaptive Huffman coding.

  A 001
  B 010
  C 100

  One additional rule: if any symbol is to be encoded (sent) the first time, it must be preceded by a special symbol, NEW. The initial code for NEW is 0.

  1) What is the final binary stream that encodes the input string?

  2) What is the compression ratio? Compute the original size of the input string using the fixed length encoding scheme.

- Consider the following indices that are generated by LZW algorithm. Based on the indices and the initial dictionary symbols, decode the message and generate the original text message.

  ```
  0 0 1 2 5 2 3 2 6
  ```

Dictionary

Index	Symbol
0	a
1	b
2	c

Ans: aabcbccaaccb

Index	Symbol
0	a
1	b
2	c
3	aa
4	ab
5	bc
6	cb
7	bcc
8	ca
9	aac
10	cc

- Suppose you have a vocabulary consisting of two symbols X and Y and one ending symbol $ having probabilities P(X)=3/5, P(Y)=1/5, and P($) = 1/5. Using arithmetic coding, encode the message "YY$" and generate a binary codeword that encodes the message uniquely.

- Also, review the arithmetic coding (encoding and decoding) example in the lecture note.

- What is differential coding? How does it contribute to the compression?

- Suppose that we have the following 4x4 image block. Encode this block using the lossless mode of JPEG. For prediction, choose A or B or C or A+B-C predictors. The selection criterion is the maximum number of available neighboring pixels. Write the encoded values a 4x4 block in the right-hand side.

C	B		
A	X		

Values before encoding

10	11	12	13
11	12	13	14
12	13	14	14
13	14	14	15

Values after encoding

Ans:

Values before encoding

10	11	12	13
11	12	13	14
12	13	14	14
13	14	14	15

Values after encoding

10	1	1	1
1	0	0	0
1	0	0	-1
1	0	-1	1