Written Exam in
Image and Audio Coding
TSBK02

30th October 2018 8:00 - 12:00

Location:	G34
Examiner:	Harald Nautsch
Teacher:	Harald Nautsch, 1361
Department:	ISY
Exam code:	TEN1
Number of problems:	6
Number of pages:	4 + formula collection
Permitted equipment:	Calculator, "Tables and Formulas for Image Coding and Data Compression"
Grades:	0-13 U 14-19 3 20-25 4 26-30 5
Other:	Answers can be given in English or Swedish. The teacher will visit around 9:15 and 10.45

1 Explain how the coding works in each of these still image coding standards

a) GIF

(1 p)

b) PNG

(1 p)

c) JPEG

(1 p)

d) JPEG-2000

(1 p)

2 a) When coding general audio signals, a *psychoacoustic model* is often used to get a more efficient coding. Explain how the model works.

(2 p)

b) Draw a block schedule over a typical hybrid coder for video signals and explain how the parts work.

(2 p)

c) Explain what the LBG algorithm is and how it works.

(2 p)

3 A memoryless source has the alphabet $\mathcal{A} = \{a, b, c\}$ with the symbol probabilities $P(a) = 0.65,\ P(b) = 0.25$ and $P(c) = 0.1$.

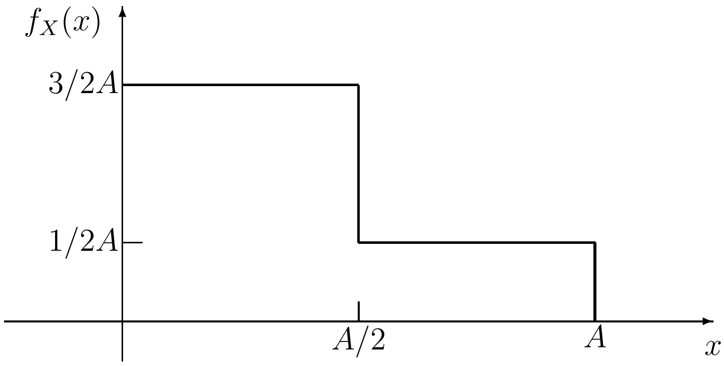
a) Construct a Huffman code for the source that gives an average rate of at most 1.3 bits/symbol.

(2 p)

b) Code the sequence a, b, a, a from the source using arithmetic coding. Give both the resulting interval and the binary codeword.

(2 p)

4 The random variable X is distributed according to the figure below. X is to be quantized to two levels. Calculate the reconstruction levels and decision regions such that the expected mean square error is minimized. Calculate the resulting mean square error.



(4 p)

5 Linnea wants to construct a coder for transmitting mono audio files over the Internet.

In order to get a coder that is robust to packet losses, she splits the audio stream into its odd and even samples, respectively, and codes the two streams independently of each other, ie given the original signal X_n she creates two new signals Y_m and Z_m according to

$$Y_m = X_{2m} \ , \quad Z_m = X_{2m+1}$$

Assume that X_n is modelled as a stationary gaussian AR(1) process with mean 0 and auto correlation function $R_{XX}(k)$

$$R_{XX}(k) = E\{X_n \cdot X_{n+k}\} = \sigma_X^2 \cdot \rho^{|k|} \ , \quad |\rho| < 1$$

Given that we code Y_m and Z_m using optimal linear predictors, how much higher rate must we use to achieve the same signal-to-noise ratio as an optimal linear predictive coder working directly on the original signal X_n? The quantization can be assumed to be fine.

(6 p)

6 Linus wants to use transform coding to code the same type of audio data that Linnea coded in problem 5. Assume that the audio signal is modelled as a stationary gaussian process X_n with mean 0 and auto correlation function $R_{XX}(k)$

$$R_{XX}(k) = E\{X_n \cdot X_{n+k}\} = \sigma_X^2 \cdot 0.95^{|k|}$$

Given that the transform used is a 4 point Hadamard transform and that the transform components are Lloyd-Max quantized.

What is the lowest average data rate that can be used if the resulting signal-to-noise ratio (SNR) has to be at least 13 dB?

What is the lowest average data rate that can be used if we Lloyd-Max quantize the signal directly, without using a transform?

(6 p)