Re-exam for:
Multimedia and Video Communications (SSY 150) - August 30, 2012

Department of Signals and Systems, Chalmers University of Technology
30 Aug 2012 - 2.00 pm, Sven Hultins gata 6,
Lecturer and Examiner: Prof. Irene Gu

This written exam may yield a maximum of 40 points.

The total exam yields 100 points (where 3 laboratory work yields a maximum 60 points)

To pass the examination, a minimum of 50 points is required.

<u>Grades</u> are defined as follows:

<u>TOTAL</u> points (p)	<u>Grade</u>:
$p < 50$	Fail;
$50 \le p < 70$	Pass with grade 3;
$70 \le p < 85$	Pass with grade 4;
$p \ge 85$	Pass with grade 5.

<u>Aids allowed:</u>
The mathematical handbook 'Beta' and other books,
A calculator (of any type),
One A4 page (can be in 2 sides) of hand written notes (can include notes, equations and drawings etc).

Success and good luck!

Problem 1. (20p)

1.1. (4p) Compression of video signals:

(a) Write down the name of a key "parameter" in video compression (which leads to a fundamental difference to 2D image compression), that enables one to achieve high video compression*;*

(b) Briefly describe the differences between I frames, B frames and P frames used in many video coding standards*;*

(c) To adapt to the dynamic bandwidth in networks, current video coding standards allow some options such as using "*progressive transmission*". Describe 3 most relevant ways to introduce *scalability* that are suitable for progressive transmission of compressed video (write down the names, and use one sentence to explain each way of scalability) (*maximum 50 words);*

(d) There are *two modes* used in the current video compression standards. Do you know them? If so, write down the names of these 2 modes, and briefly describe the difference of these 2 modes.

1.2. (4p) Compression of speech signals:

(a) Mention ONE name of model-based speech compression method that you know. Briefly describe how compression is achieved in the method;

(b) Mention another NAME of non-parametric speech compression method that you know. Briefly describe how compression is achieved in the method;

(c) In speech signal coding and compression, a speech signal sequence is usually divided into blocks (e.g. 20ms) where coding is then applied to each block of speech. Briefly describe the reasons on why this is applied;

(d) For obtaining a high compression rate with good sound quality, psycho-acoustics, or human auditory system models, are often considered in speech coding.
Write down some names of speech coding standards that you know, which have exploited some psycho-acoustic features in the coding, and briefly specify which feature(s) is exploited in each of these standards (briefly);

(e) List several important psychoacoustic-related 'features' that maybe exploited to achieve high quality speech with a high compression rate.

1.3. (1p) Write down the names of main parameters related to Quality of Service (QoS) for transporting compressed multimedia data through an error prune network.

1.4 (2p) Describe the basic principle that the Shannon theory of communications holds for the separate source and channel coding. Which condition(s) is/are violated in transmitting packets of compressed video? *(maximum 50 words).*

1.5. (2p) For transmitting compressed video packets, we assume that the main error is caused by the *bursty packet loss* in the network layer. Two-state Markov model is frequently used to model the packet loss networks.

(a) Sketch the two-state Markov model;

(b) Describe how "probability of packet loss" and "bursty length of packet loss" are computed under this model (describe them by giving mathematic formulae).

1.6. (2p) Given the following IP protocol stack (Fig,1), you are asked to choose the best combination of protocols for transmitting compressed video through the Internet.
(a) Write down the protocol combination you have chosen.
(b) *Briefly* state the reasons of your choice (*maximum 20 words*).

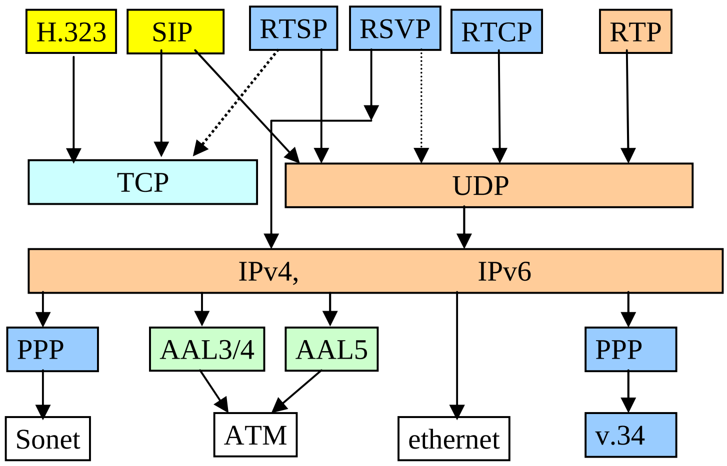

Fig.1. an IP protocol stack

1.7. (5p) The aim here is to generate a video communication system that delivers the best end-to-end performance when transporting compressed video through a network system (as described by a 5-layer model in Fig.2).
Briefly describe the parameters (or methods) in each layer that can be used to adjust the end-to-end performance *(maximum 20 words for each layer).*

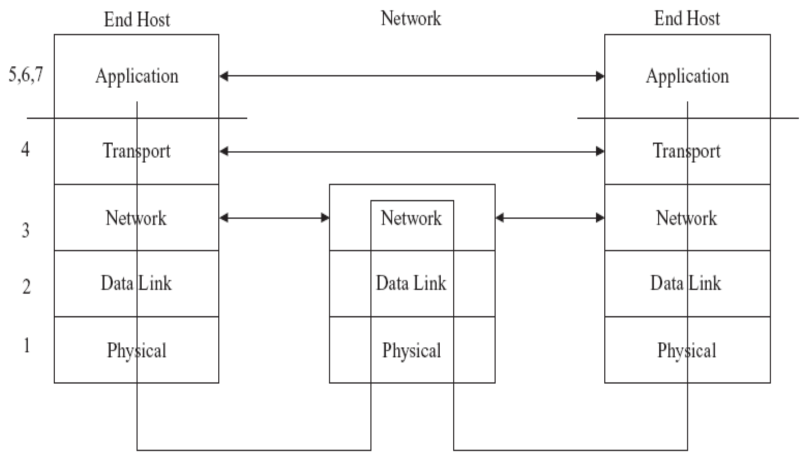

Fig.2 A 5-layer model

Problem 2. (10p)
In order to design an end-to-end performance optimized video communication system, the following conditions in the system are specified and can be employed for your design process:

(a) The selected network is able to acknowledge the packet loss and re-transmit the lost packets. Further, the sender's buffer can store up to 2 frames of video, each frame of video contains M packets. The re-transmission parameter for the k-th packet in the n-th frame is $\sigma_k^{(n)} \in \{0,1\}$, where 0 denotes no packet re-transmission, and 1 with packet re-transmission.

(b) RS codes is used for channel coding. To allow different error protection for the codewords obtained from the inter-mode and the intra-mode in video source codec, the RS codes allows 2 rate-modes $\mathrm{RS}(n_i, k)$, $i = 1, 2$: that lead to two different channel rates $c_i = k / n_i$, where *k* is the number of source symbols, and n_i is the length of RS codewords.

(c) For video source coding: both one directional and bidirectional predictions are employed (i.e. P frames and B frames are used) for inter-coding; for intra coding, the quantization step size can be adjusted that may impact the coding errors and the coding rate.

Further, the following delay constraint is specified: The maximum allowed delay time for receiving each video packet is: $T_0 = R_0 / R_T$ (where R_0 is the rate constraint, and R_T is the throughput).

Your aim is to design an end-to-end performance optimized video system, such that <u>the expected distortion of the reconstructed video in the receiver side is minimized</u> (assume the distortion function is D). *You are asked to formulate a criterion function **L** to design this video system,* where ***L*** is a function of the parameters in source coding, channel coding and packet re-transmission, and is subject to the delay constraint. You should then specify on how these parameters can be adjusted, in order to yield an optimized system.

(hint: specify the parameter sets for the source coding, channel coding and re-transmission; formulate the criterion function and the constraint condition; formulate a criterion function L using the Lagrange multiples, and specify whether to minimize or maximize the function L, and specify how the parameter sets can be estimated.
Note, you do not need to get the final solution in numbers!)

Problem 3 (10p)
In order to obtain the end-to-end video (i.e. global) performance optimization (i.e. minimizing the expected 'distortion' between the transmitted and received video), cross-layer design is desirable instead of achieving local optimization in each individual layer.

Using the knowledge you have gained in this course and your creativity, suggest (briefly) a *novel* approach or some ideas (that has NOT been included in our lecture notes) for cross-layer design (e.g. 2 or 3 layers), which could potentially improve the end-to-end video performance when transporting compressed video over packet error prune network (give your intuitive reasoning).