INTENSIVE GRADUATE COURSE "INTERNET MULTIMEDIA" 20.12.2002

1. Brief questions (please answer with max. 100 words per item, 2 pt per item, total 8 pt).

a. Compare the properties of the international telephone numbering system on one hand and IP addresses and DNS host names on the other.
b. Give an example of an RSVP killer reservation using a drawing. What feature of RSVP causes this problem?
c. What components are necessary and sufficient to create a packet network with delay bounds? What does the delay bound depend on? Explain the components briefly.
d. Sketch the SIP INVITE request, with SDP, when you call up *presidentti@tpk.fi* from your office.

2. At 10 pm on some evening in the not too distant future, 1000 users on a CATV network (bandwidth 10 Mb/s) have set their timers to start their video program to watch the evening news. The news are carried as a 20 kb/s bitstream using RTP and RTCP over multicast IP. How long, on average, does it take until everybody knows the number of other receivers? RTCP packets are 100 bytes long and the standard RTCP bandwidth (5% of data stream) is used. (6 pt)

3. You can run SIP in proxy or redirect mode. Compute the call-setup delay for both modes and compare the server processing requirements for a server handling the *oulu.fi* domain, assuming that the caller is in New York and the callee is at *ees1.oulu.fi*, with the one-way delay between New York and Finland being 50 ms. You can ignore the delay within the University of Oulu's local area network. (6 pt)

4. *Design task*. Design a video conferencing service that integrates IP-based PCs as well as traditional telephones. Each conference can have anywhere from two to one hundred participants. The system should support many simultaneous conferences. On occasion, there will be large lectures where a lecture, for example, is being distributed to a large audience watching on PCs distributed across a modest number of local area networks. Describe the architecture and protocol components. Also, size the conference mixer and its network connection, assuming the following:

- 10 simultaneous conferences with 20 participants each;
- each participant listens to audio, but only at most two speak at any given time;
- there is one video channel per conference, for the active speaker;
- video uses H.261 at 128 kb/s and 15 frames per second;
- audio is G.729 with 10 ms packetization.

To estimate the necessary server size for the server(s), describe the number of encodings and decodings that the server needs to perform. Describe the protocols needed for this service. How would you assure quality-of-service for the participants? How do participants dial into the conference? How do they find out who else is in the conference? Can you improve the efficiency of audio for low-bandwidth users (that may only get audio, rather than video and audio)? Do end systems need any special capabilities? (10 pt)