1 Packet forwarding efficiency and protocol overhead

This exercise examines the overhead caused by packet headers and you will also see how header compression helps to resolve the situation.

1.1 Prerequisites

Before you start solving the exercises, please study the following subjects:

1. The header lengths of RTP, UDP and TCP -protocols and the effect of compression to the aforementioned header lengths(RFC 2508, http://www.ietf.org/)
2. The encoding speeds of the following voice codecs: G.711 (PCM) ja G.723.1 (ACELP)
   - http://standard.pictel.com/reference/summary_itu_codecs.htm
   - http://keskus.hut.fi/opetus/s38117/k2000/Aiheet/Esitelmat/5-sanna_lahde.pdf
3. OSI -reference model
   - http://www.ictp.trieste.it/~radionet/1998_school/networking_presentation/OSI-layers.html
   - http://www.geocities.com/SiliconValley/Monitor/3131/ne/osimodel.html

In packet networks all of the data is forwarded within packets. A packet usually consists of a header and the actual data. It is possible to send just the headers and no data. If the packet lengths are short the header forms a significant proportion of the packet length. Therefore, the overhead/data -ratio is smaller and, in a way, reduces the goodput (the relay of user data) in the network. To avoid this effect the header can be reduced in size, or compressed.

1.2 Packet overhead

- What is the true use of bandwidth (coded speech rate + headers) on an VoIP-connection if you use G.711- ja G.723.1 -coding (ACELP) and the packet length is 20, 40 and 80 ms with and without header compression. Your answer should contain 18 values per each codec.

- Determine also how many %:s the header is from the total use of the bandwidth.

Return your answers in the answer sheet available from the course web-pages. Remember to include all the calculations you used.

1.3 OSI-reference model and a VoIP-call in the Internet

A corpoation uses a VoIP-system to connect its branches located around Finland. The CEO in Helsinki wants to confer with a sales manager in Oulu via VoIP-conversation. Fill the Figure 1 and explain what layers in the OSI-model are used in different network elements along the VoIP-conversation. Name also all the protocols you know (or suspect) that participate in forming the call and explain also how the protocols effect the call.

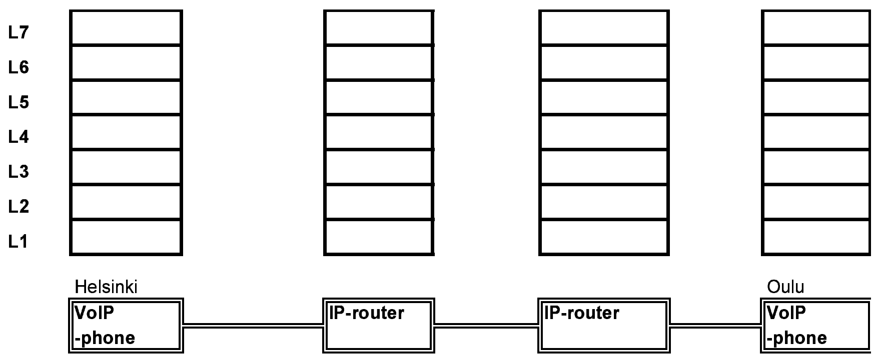

Figure 1: OSI-reference model in VoIP-conversation

2 Statistical multiplexing in packet networks

An ATM device has six subscribers attached to it. The subsribers offer independent of each other variable rate traffic containing a video broadcast.

The subscribers come in two different forms (and there are three of each in the network, $n_a = n_b = 3$).

1. Source A sends at basic rate, $\lambda_{Abasic} = 15$ Mbit/s, and for 10% of the time the send rate is $\lambda_{Amax} = 30$ Mbit/s
2. Source B sends at basic rate, $\lambda_{Bbasic} = 15$ Mbit/s, and for 30% of the time the send rate is $\lambda_{Bmax} = 20$ Mbit/s

The maximum combined send rate available on the link for all of the traffic is $\lambda_{link} = 148.75$ Mbit/s

- What is the average load of the link?
- What is the probability of link overload?
- How large a percentage of the traffic offered to the link is lost?
- How much of the traffic is lost by A and how much by B? We may assume that the dropped packets are picked randomly out of the traffic flow.

Determine the previous values also when there are 24 subscribers($n_a = n_b = 12$) and the basic rate for A is 3.75 Mbit/s and the maximum for 10% of the time is 7,5 Mbit/s; for B the basic rate is 3.75 Mbit/s and for 30 % of the time B sends at its maximum rate of 5 Mbit/s.

Compare your results? What can you observe?