Lab 4

31 March 2014

During this lab session you should gain some insight into how the concepts explained during the lectures can affect the quality of a video stream.

1. **Packet Analysis**

For this exercise, you will have to use *Wireshark* packet analyzer. This program is used to capture and display network packets.

Wireshark has been used to capture network traffic during the window of time in which a streaming unicast communication occurred. This data has been stored in the file *StreamingVideo.pcapng* (or *StreamingVideo.pcap* in case you have an old *Wireshark* version). Open the file with Wireshark to check its content. Before answering the questions bellow there may be some steps that you would need to follow.

In case that the protocol shown of the packets is UDP, you will need to decode them as RTP. To do so, right click over a UDP packet, select "Decode As" option, and then select RTP. Repeat this process, for each UDP packet type.

(a) What is the server and client IP addresses?

(b) What protocol is used for the data transmission? Why?

(c) Can you differentiate the audio and video data? Explain why. After identifying video packets, you may need to indicate the payload type as H.264 (if it is not available in the protocol column). To do so, go to Edit > Preferences > Protocols > H.264 and indicate the payload type number.

(d) **RTP stream analysis.** To analyse the RTP stream use the menu Statistics or Telephony > RTP > Show All Streams and Statistics or Telephony > RTP > Stream Analysis. You can get the delay, jitter, bandwidth, etc. of the RTP stream. Also get general statistics like packet loss, maximum delay and sequence errors. Use the "Graph" button to see the jitter and difference between packets over time.

Notes: A text file *StreamingVideo.txt* is also available, in case you are not able to see the content of the pcapng or pcap files.

Extra information on RTP payload format can be found in `http://tools.ietf.org/html/rfc3984`

2. **Matlab Assignment: Intermediate Buffer Size Influence on Media Streaming over Variable Networks**

Setting : A network scenario with 2 disjoint paths between the Server and the Client, each path contains 2 network segments linked by an intermediate node, modelled by a FIFO buffer. Each segment is characterized by rate, which varies in time.

Files : You will only need to modify the *lba3_sliding.m* file.

What to do : Just change in the *lba3_sliding.m* file the values for B1(2) with the values given in the same line in the code. The graphs will show the difference in packet scheduling with the different values of the buffer on the first path, compared to the case of infinite buffer size. If time permits, repeat the operation for all 3 rate sets commented out in the code.

To see : **Figure 1** shows the total cumulative rate of the video sequence (red line), the total cumulative scheduled rate with infinite intermediate buffers (blue), and the total cumulative scheduled rate with constrained buffer size on one network path (green).
Figure 2 shows the cumulative scheduled rate on each of the two paths in the case of unlimited and constrained buffer size.
Figure 3 shows the instantaneous buffer content during streaming, for the two network paths.

(a) Interpret the changes in the streaming process with the varying sizes of the buffer on one network path.

(b) Interpret the influence of the intermediate buffer size on the streaming process, depending on the average available network rate, and on the network rate variations in time.

3. **Study of a RTSP communication** The next table gives you a simplified view of some packets that have been captured in the given order between a Client `C` and a server S:
Afterwards, RTP Packets with the sequence numbers 9810092, 9810093, 9810094 and 9810096 are captured.

(a) What is happening between the server and the client?

(b) What can you say about the used channel?

(c) What packet(s) do you expect to see if the client decides to stop playback?

(d) Suppose the channel has an average rate of 512kbps, and that the client starts playing the video as soon as it receives data. Will the playback of the stream be continuous? Why?

```
C->S    DESCRIBE rtsp://foo/twister RTSP/1.0
        CSeq: 1

S->C    RTSP/1.0 OK
        CSeq: 1
        Content-Type: application/sdp
        Content-Length: 164
        o=- 2890844256 2890842807 IN IP4 128.178.121.68
        s=RTSP Session
        i=An Example of RTSP Session Usage
        a=control:rtsp://foo/twister
        m=video 0 RTP/AVP 26
        avg_bitrate=600000

C->S    SETUP rtsp://foo/twister/video RTSP/1.0
        CSeq: 2
        Transport: RTP/AVP;unicast;client_port=8002-8003
        Session: 12345678

S->C    RTSP/1.0 OK
        CSeq: 2
        Transport: RTP/AVP;unicast;client_port=8002-8003;
        server_port=9004-9005
        Session: 12345678

C->S    PLAY rtsp://foo/twister RTSP/1.0
        CSeq: 3
        Range: npt=0-inf
        Session: 12345678

S->C    RTSP/1.0 OK
        CSeq: 3
        Session: 12345678
        RTP-Info: url=rtsp://foo/twister/video;
        seq=9810092;rtptime=3450012
```