National University of Singapore
School of Computing

CS2105 **Tutorial 10** Question paper

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1. **[KR, Chapter 9, R2]** There are two types of redundancy in video. Describe them, and discuss how they can be exploited for efficient compression.

   Spatial – send only 2 values, pixel colour and no. of repeated value, ~~same image~~ Temporal – send differences between frames.

2. **[KR, Chapter 9, R3]** Suppose an analog audio signal is sampled 16,000 times per second, and each sample is quantized into one of 1,024 levels. What would be the resulting bit rate of the PCM digital audio signal?

   2^{10} ∴ bit rate = 16000 × 10
   = 160 000 bps.

3. **[KR, Chapter 9, R7]** With HTTP streaming, are the TCP receive buffer and the client's application buffer the same thing? If not, how do they interact?

   Transport. application.

   at different layers.

   No, TCP receive buffer sends packets to client application buffer. However, the fill rate may fluctuates.

4. **[KR, Chapter 9, R10]** Why is a packet that is received after its scheduled playout time considered lost?

   It is considered lost to maintain a constant bit rate playout at the client.

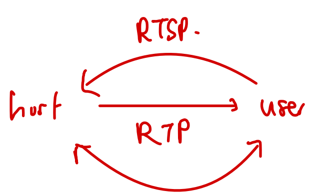
5. In practice, RTP tends to be used over UDP while RTSP tends to be used over TCP. Why might this be so?

   reliablity vs speed

   faster * reliable

   RTP encapsulates video data which is large, more loss tolerant, while RTSP is to control the media which is small, less loss tolerant.

6. ** **[KR, Chapter 9, P11]** Consider the figure below (which is similar to Lecture 10 notes page 26). A sender begins sending packetized audio periodically at $t = 1$. The first packet arrives at the receiver at $t = 8$.

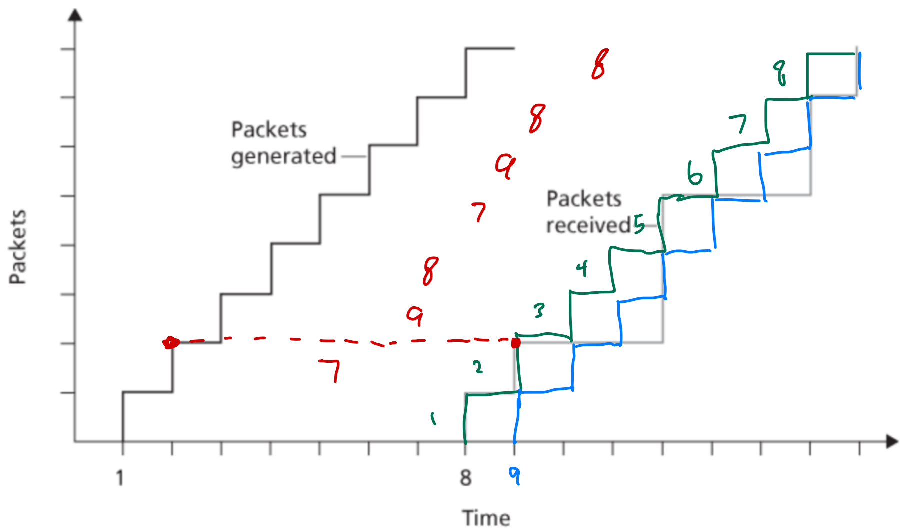


RTCP – Control: streaming quality, user bandwidth

– Send back to host

– TCP.

RTP



a) What are the delays (from sender to receiver, ignoring any playout delays) of packets 2 through 8? Note that each vertical and horizontal line segment in the figure has a length of 1, 2, or 3 time units.

packet	2	3	4	5	6	7	8
delay	7	9	8	7	9	8	8.

b) If audio playout begins as soon as the first packet arrives at the receiver at $t = 8$, which of the first eight packets sent will *not* arrive in time for playout? 3, 4, 6, 7, 8

c) If audio playout begins at $t = 9$, which of the first eight packets sent will not arrive in time for playout? 3, 6.

d) What is the minimum playout delay at the receiver that results in all of the first eight packets arriving in time for their playout? t = 10.

or 9s delay.

① XOR all lines → send the line together.
→ can reverse the XOR.

7. **[Modified from KR, Chapter 9, P13]** Recall the two FEC schemes for VoIP described in lecture. Suppose the first scheme (Scheme 1) generates a redundant chunk for every four original chunks. Suppose the second scheme (Scheme 2) uses a low-bit rate encoding whose transmission rate is 25 percent of the transmission rate of the nominal stream. (Note: we ignore the effects of playout delay in this question as we assume that all packets, including FEC packets, will be received prior to reconstruction and playback)

② low-quality piggyback

a) How much additional bandwidth does each scheme require? Both 25%

b) How do the two schemes perform if the first packet is lost in every group of five packets? Which scheme will have better audio quality? Scheme 1.

low quality vs high quality
5th packet.

c) How do the two schemes perform if the first packet is lost in every group of two packets? Which scheme will have better audio quality? Scheme 2.

Both schemes are bad. → cannot be recovered by scheme 2 if

Scheme 1: cannot recover,

⇒ cannot recover.