

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 redundancy:** Redundancy within the same image, e.g. consecutive pixels with the same color. We could compress the video frame by sending just the color value and its count (instead of the color value  $N$  times).

**Temporal redundancy:** Redundancy between multiple images, e.g. consecutive frames that are very similar and only have a small change between them. We could compress the second video frame by sending only its difference from the original frame (instead of the full uncompressed frame).

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?

**Each sample would require  $\log_2 1024 = 10$  bits encode its data. Hence, sampling at 16,000 times per second would generate a signal of 160,000 bits per second = 160 kbps.**

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?

**No, they are not the same. TCP and application are at different layers of the OSI model. TCP receive buffer (at transport layer) receives the TCP packets from the network layer, which are then processed and forwarded as HTTP packets to the client's application buffer (at the application layer). The client (e.g. Chrome browser), upon receiving these HTTP packets, then processes and renders the video playback to the user.**

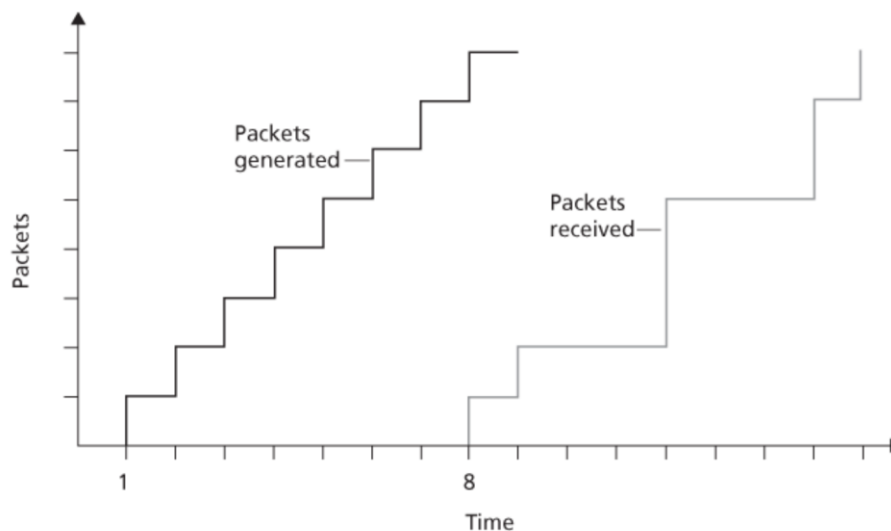
4. [KR, Chapter 9, R10] Why is a packet that is received after its scheduled playout time considered lost?

**When an application receives a packet that is *late*, i.e. received after its scheduled playout time, the application would not be able to play that packet anymore. Hence, from the perspective of the application, the packet has been lost.**

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

RTP sends media flow (e.g. multiple video packets) while RTSP sends playback commands (e.g. "Play" and "Pause"). Hence, it may be preferred that RTP prioritizes speed over reliability (UDP), so that the client could playback the video with minimal delay (losing a few video packets in-between is a relatively small cost since its impact on playback experience is generally tolerable). On the other hand, it may be preferred that RTSP prioritizes reliability (TCP), so that we do not lose any of these commands requested by the client.

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  $t = 8$ .



- 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: 7 units, Packet 3: 9 units, Packet 4: 8 units,**

**Packet 5: 7 units, Packet 6: 9 units, Packet 7: 8 units, Packet 8: 8 units**

- 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?

**Packet 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?

**Packets 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?

**Minimum playout delay is 2, so playout begins at  $t = 10$**

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)

- a) How much additional bandwidth does each scheme require?

**Scheme 1: Every 4 original chunks will have 1 redundant chunk = 25% additional bandwidth; Scheme 2: Every chunk will have its redundant low-quality chunk “piggyback” on the next transmission = 25% additional bandwidth. Hence, both schemes will require additional 25% bandwidth.**

- 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 will be able to reconstruct the original high-quality audio, while Scheme 2 will get a low-quality audio packet in every 5 packets. Hence, Scheme 1 will have better overall audio quality.**

- 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 will have better audio quality as Scheme 1 is unable to recover from the packet losses while Scheme 2 can (albeit with low-quality). Note, redundancy-based FEC (e.g., XOR in Scheme 1) has a fixed loss limit up to which we can recover the data. If the losses exceed that limit, then all data lost in that group cannot be recovered.**