KTN Lab 7

Adrian Langseth

March 2019

Question 1: What is the rate at which data is generated at the sender (in bytes)?

As the sender uses normal RTP and transmits G.722-encoded voice at 48 Kbps, we have that the sender generates data at $48000/8 = 6000$kB/s

Question 2: What is the size of the IP datagrams sent?

The datagram consists of two elements, the headers and the data. The header of the RTP packet is 12 bytes, the header of the UDP is 8B, the header of IP is 20B and the data of each packet is $6000B/s * 16ms = 96B$. This gives us a total of 136 Bytes per datagram sent. This gives a transmission rate of $136B/0.016s = 8.5kB/s = 68kbps$

Question 3: Explain how an arbitrary RTP packet in the application will look like.

The RTP packet will have an IP header(20B), UDP Header(8B), an RTP header of 12B and a payload, which is the data of 96B.
The two first bits of the RTP Header is the version which is set to 2(or 11) because it is always set to 2 when dealing with RTP. If it is the first speech packet after a silence period, the seventh bit, the marker bit, is set HIGH. The 9th to 15th bits of the RTP header signifies the payload type, and is 13 (or 0000101) as it reflects that the audio is G.722 encoded. Then follows the sequence number, which is a incrementing number starting at a randomly generated point, and is used by the receiver to detect missing packages and restoring the original order sequence. After this we have a time stamp of 32 bits signalizing the time of sending. The 32-bits after the timestamp is the SSRC identifier, which is a identifier for the RTP stream. This is randomly generated onto the header by the sender. The other bits undescribed here are miscellaneous small fields such as padding, extension, etc.

Question 4: Show how much additional bandwidth each of them require of our application.

i: forward error correction (FEC) with redundant encoded chunks

n is 5, Transmission rate increases by $1/n = 20\%$ This gives us a new transmission rate of $68kbps * 1.2 = 81kbps$

ii: FEC with redundant lower-resolution audio stream

GSM has a rate of 13, which gives a new transmission rate of $68kbps + 13kbps = 81kbps$, an increase of 19%.

iii: Interleaving

Interleaving has the same transmission as it does not send anything more, only shuffles around the order of what one sends.

Question 5: What happens if the first packet is lost in every group of five packets? Which scheme will have the better audio quality?

The best would be the FEC with redundant encoded chunks.

i: forward error correction (FEC) with redundant encoded chunks

Will be able to reconstruct the HQ audio.

ii: FEC with redundant lower-resolution audio stream

This will use the LQ audio encoding to cover for the lost packets, resulting in a lower quality overall.

iii: Interleaving

Interleaving will result in losing a smaller part of several packets.

Question 6: Make an illustration of the scenario in question 5 for FEC with redundant encoded chunks?

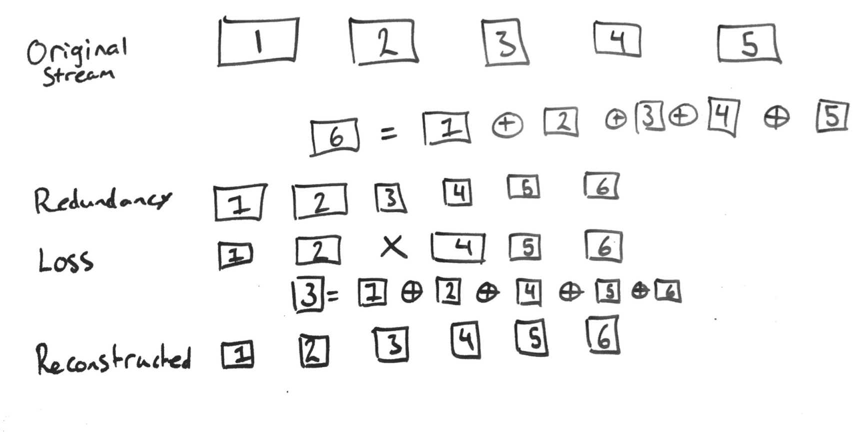

Figure 1: Illustration of FEC with redundant encoded chunks

Question 7: What happens if the first packet is lost in every group of two packets. Which scheme will have the better audio quality?

i: forward error correction (FEC) with redundant encoded chunks

The audio quality will be very poor and many of the original packets will be lost.

ii: FEC with redundant lower-resolution audio stream

The audio will be available for the receiver, although every other chunk will be of low quality. This will result in a decent audio quality.

iii: Interleaving

Interleaving will result in noticeable holes in the audio since so many packets will be lost.

Question 8: Make an illustration of the scenario in question 7 for FEC with redundant lower-resolution audio stream.

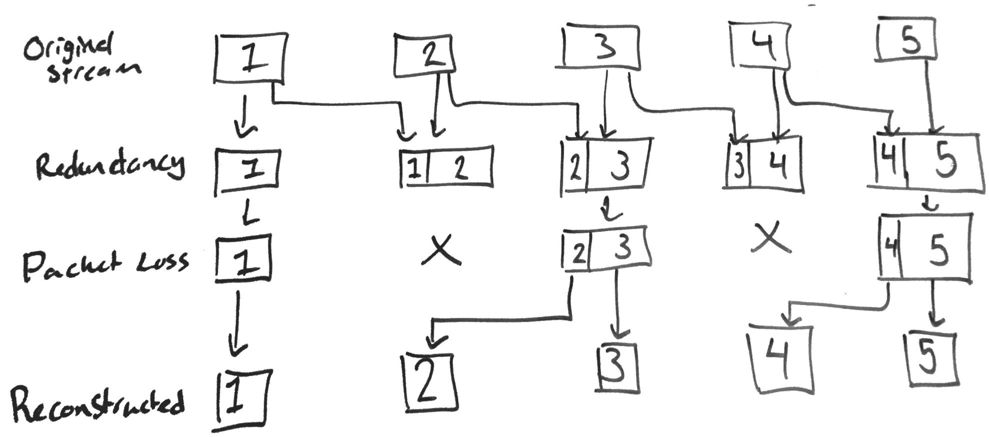

Figure 2: Illustration of FEC with redundant lower-resolution audio stream

Question 9: For each of the FEC schemes, how much playback delay does each scheme add? What can be said about the delay of the interleaving scheme?

i: forward error correction (FEC) with redundant encoded chunks

have to wait for n + 1 = 6 packets. The playback delay will be 6 added.

ii: FEC with redundant lower-resolution audio stream

This scheme only has to wait for two packets to begin playback. The added playback delay will be 2.

iii: Interleaving

Interleaving needs for all packets, then interleave them, stream them and then reconstruct them. This adds significant playback delay.