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CMPSC 176B HW2

Introduction

In this assignment, I attempted to simulate the transmission of an audio file through a lossy channel. This simulation contained three significant parameters: packet size, packet loss, and packet replacement policy. Packet size can be varied from very small packets, to large packets up to a size of 1500 or more. Packet loss rate determines the chance that any given packet is lost in transmission. Packet replacement policy chooses how to replace packets that are lost. For this assignment, I worked on three replacement policies: replacing a lost packet with all 0s, replacing a lost packet with the most recent sample taken, and replacing a packet with the most recent packet received.

Generating the simulated files:

When generating the simulated files, I had two settings. First, I generated files individually, allowing inputs to determine the three parameters listed above. After doing some basic testing to verify that the program was working correctly, I wrote a second part to the program to generate all song outputs for a given array of each of the parameters. At the current time, my program tests for the packet sizes [100, 200, 400, 600, 800, 1000, 1200, 1500], for the loss rates [0.005, 0.01, 0.02, 0.04, 0.08, 0.12, 0.16, 0.25, 0.36, 0.49, 0.64, 0.80], and for all three packet replacement policies.

Testing Methodology and Data:

To create a reasonably effective quantitative measurement of the individual pieces of audio at different settings, I created a scale from 1 to 10 to approximate the quality of the output audio files. The scale works as follows:

Packet Descriptions:	
1	Completely Inaudible
2	Words can be heard although not understood
3	Audio can be understood with great difficulty
4	Audio can be understood, but is difficult to listen to
5	Audio can be understood, but is blurry
6	Audio has significan quality drops

7	Audio is scratchy or has noticeable quality drops
8	Minor disruptions noticeable
9	Minor disruptions noticeable, but only if listening for problems
10	Sounds like the original, no noticeable changes

With these standards, I listened to each of the pieces of audio I generated and rated them appropriately on this scale. This allowed met to get the following results.

packet replacement: zeroes

Loss	psize:100	psize:200	psize:400	psize:600	psize:800	psize:1000	psize:1200	psize:1500
0.005	10	10	10	10	10	10	10	9
0.01	10	10	10	10	10	10	9	9
0.02	10	9	8	8	9	9	9	8
0.04	9	9	9	8	8	8	8	8
0.08	8	8	8	8	7	7	6	7
0.12	7	6	5	6	6	5	5	6
0.16	7	6	6	6	5	5	5	5
0.25	5	5	4	4	4	4	4	4
0.36	4	4	4	4	4	3	3	3
0.49	3	3	3	3	3	3	3	3
0.64	3	3	3	2	2	3	2	2
0.8	1	1	1	1	1	1	1	1

packet replacement: most recent sample

Loss	psize:100	psize:200	psize:400	psize:600	psize:800	psize:1000	psize:1200	psize:1500
0.005	10	10	10	10	10	10	10	9
0.01	10	10	10	10	10	10	10	9
0.02	9	9	8	8	8	8	8	9
0.04	9	8	8	8	8	8	8	8
0.08	8	8	8	7	7	7	5	6
0.12	7	7	6	6	6	6	5	5
0.16	7	7	6	6	6	5	5	5
0.25	6	5	5	5	5	4	4	4
0.36	5	5	4	4	4	3	3	3
0.49	3	3	3	3	3	2	2	2
0.64	2	2	2	2	2	2	2	2
0.8	1	1	1	1	1	1	1	1

packet replacement: most recent packet

Loss	psize:100	psize:200	psize:400	psize:600	psize:800	psize:1000	psize:1200	psize:1500
0.005	10	10	10	10	10	10	10	10
0.01	10	10	10	10	10	10	10	10
0.02	10	10	9	9	9	9	8	9
0.04	10	9	9	9	8	8	6	6
0.08	9	8	8	8	8	7	7	6
0.12	7	7	7	7	6	5	4	4
0.16	5	5	4	5	6	6	5	5
0.25	4	4	4	4	4	3	3	3
0.36	4	4	4	3	2	3	2	2
0.49	3	3	3	3	3	2	2	2
0.64	2	2	2	2	2	2	2	2
0.8	1	1	1	1	1	1	1	1

Effects of Packet Size:

As shown above, I tested for 8 different packet sizes, 100, 200, 400, 600, 800, 1000, 1200, and 1500 bytes. For this test, I was primarily writing 0s when packets were dropped. At relatively low loss rates (below 10 percent), smaller packets disguise the loss making it almost unnoticeable while the larger packet sizes, especially 1500 cause large disruptions in the audio when they come up, greatly reducing the quality. At a more medium loss rate (0.12, 0.16, 0.25, 0.36) low packet

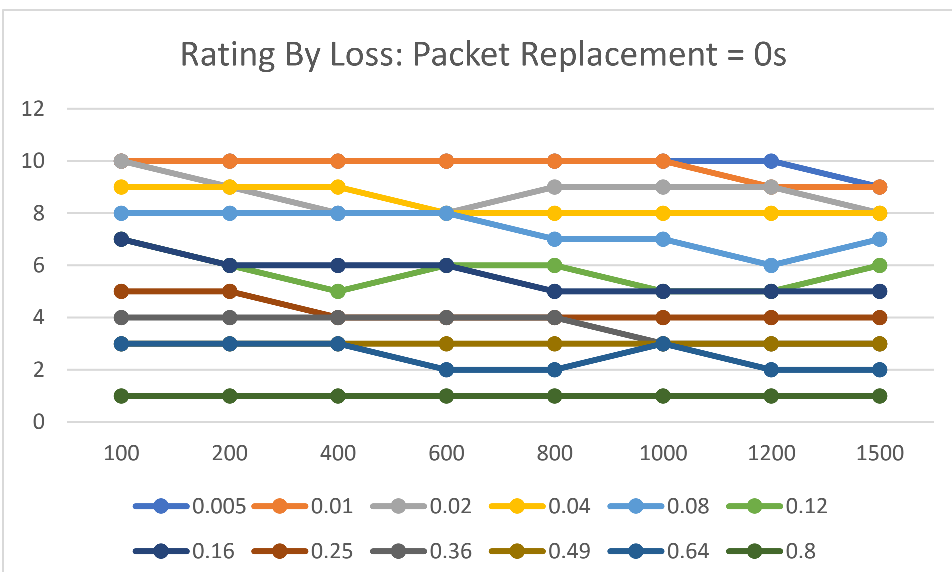

Figure 1

sizes make the music appear scratchy while large packet sizes make the music sound like it's starting and stopping. Interestingly, medium packet sizes (400, 600, 800) at medium in this range of loss rates tend to sound the best, striking a decent balance between a scratchy sounds and missing sound. At the highest loss rates I tested (0.49, 0.64, 0.80), the smaller packets sound incredibly scratchy, nearing the point of being inaudible. The larger packets are less scratchy, but sometimes the sound stops for more than a second at a time, making it almost impossible to listen to. Middling packet sizes at these higher loss rates sound similar to the higher packet sizes at medium loss rates, with the sound constantly stopping and starting. Through all of these tests, I got figure 1 which shows the general trend of different loss rates through each range of packet sizes. Although the data yields mostly horizontal lines, it does generally trend downward as the packet size gets larger. This indicates that in general, packet size decreases audio quality across a lossy channel. This generally makes sense as an increased packet size means that any loss of packets causes a greater disruption to the music.

Effects of Loss rate:

As noted above, I tested for 12 different loss rates: [0.005, 0.01, 0.02, 0.04, 0.08, 0.12, 0.16, 0.25, 0.36, 0.49, 0.64, 0.80] in my main tests. At low packet loss rates (0.005, 0.01, 0.02), the packet losses are almost completely unnoticeable, especially at lower packet sizes. At around 4% loss, the loss is really only noticeable when paying strict attention to the music, but does not pop out. Loss is definitely noticeable at 8% loss rate, at which point there is a distinct difference between

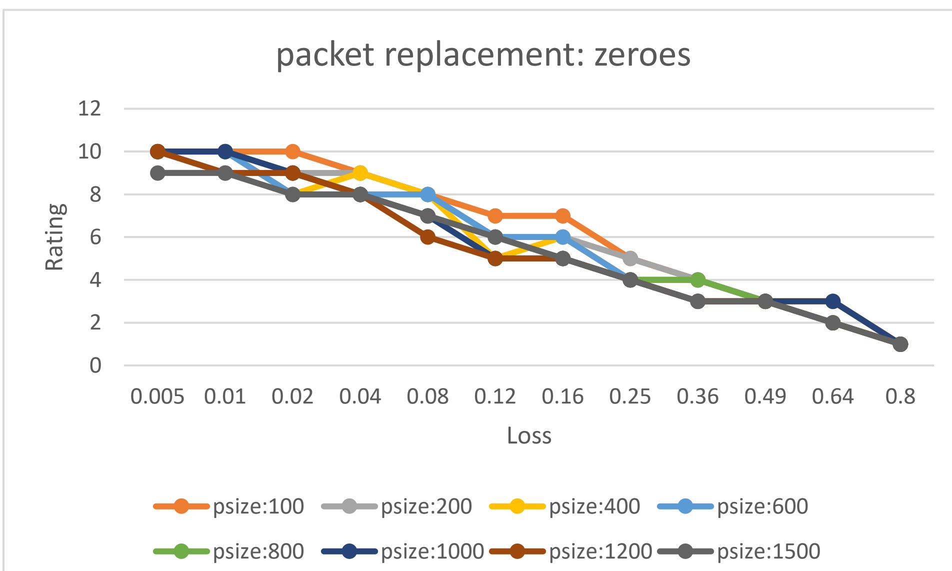

Figure 2

small packets, which only make the music scratchy, and large packets which greatly interrupt the sound. As loss continues increasing, audio quality gets generally worse. Around 0.25 and 0.36 loss rates, the audio becomes very difficult to listen to due to the highly reduced quality. Past this, the audio becomes effectively unintelligible at between 0.64 and 0.8 loss rates. Figure 2 shows the general relationship between loss rate and audio quality over a series of different packet sizes. Although there are some anomalies in the data, likely due to human error in assessing audio quality, the audio quality generally trends downward as loss increases. Additionally, across most of the graph, the higher packet size audio tends to be rated lower than the lower packet size audio. This correlates with our previous assessment above.

Effect of Packet Replay Policy

To test for different options of packet replacement, I repeated the above tests with three different packet replacement policies: replace lost packets with 0s, replace lost packets with the last sample, replace lost packets with the last received packet. Packet replacement with 0s can be seen in figure 2 and packet replacement with the last sample can be seen in figure 3.

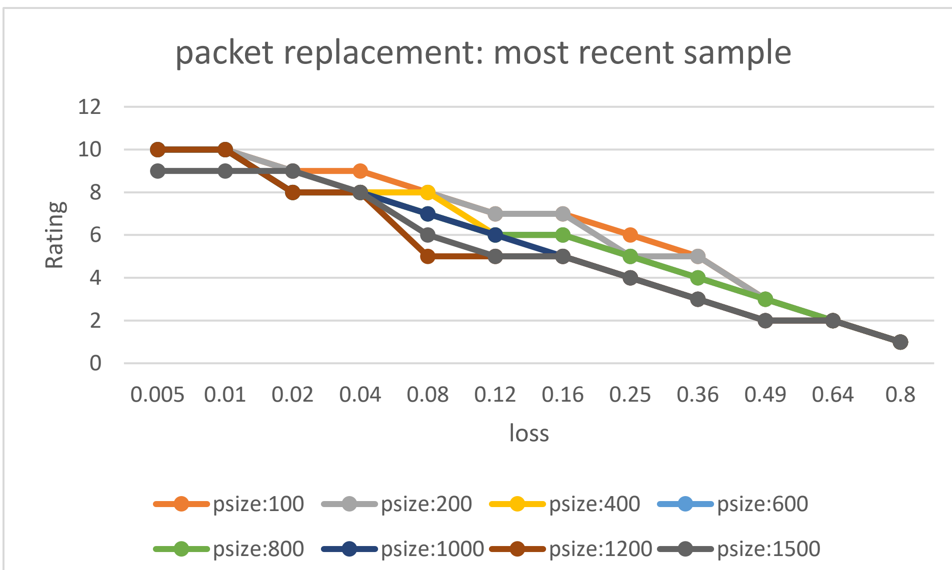

Figure 3

Interestingly, these two graphs have very little difference. Additionally, the subjective sound of replacing a packet with the most recent sample sounds almost exactly the same as replacing it with 0s. Intuitively, this makes sense. Humans hear sounds primarily as frequencies. A series of identical samples, be they 0s or some number would not show any frequency information. This means that on hearing the sound, there should be very little noticeable difference between the different packet replacement policies. Packet replacement with the most recent packet can be seen in figure 4. The most noticeable immediate difference is the greatly widened graph at lower loss rates. At low loss rates and low packet sizes, replacing a packet with the most recent packet caused a noticeable increase in audio quality. Conversely, larger packet sizes tended to have slightly lower ratings at these lower loss rates. This seems to be because the packet replacement disguises the lost packet at lower packet sizes but the disruption is clearly audible at higher packet sizes. As loss rates increased, replacing with the most recent packet causes a significant decrease in audio to the point where it is generally equal to or worse than the other two methods. Although there is no objectively clear reason for this decrease, I noticed that when two packets in a row are lost with this method, it is much more noticeable than with either of the other two methods. Additionally, at high packet loss rates (0.36 to 0.64) and low packet sizes, I noticed that voices begin to sound robotic, as if they are missing some significant degree of tone and inflection. This could explain some degree of the decrease in audio quality.

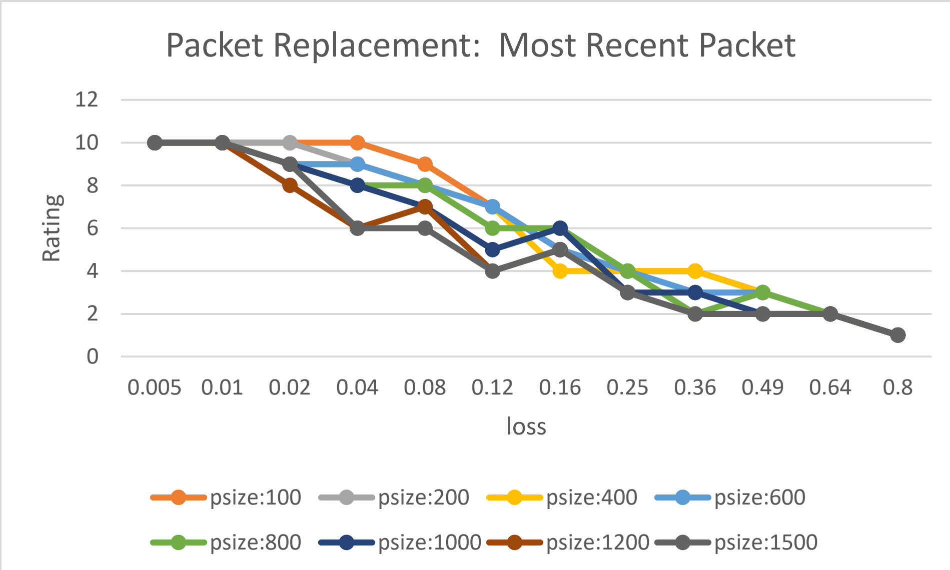

Figure 4

Generally between the three packet replacement policies, it seems that replacing with the most recent packet is the best policy at lower loss rates, especially at lower packet sizes, and that either replacing with 0 or with the most recent sample is the best policy at higher loss rates.

Given that the best packet replacement policy is unclear, it may be good to create a packet replacement policy that stays best across the board. The most obvious replacement policy would be to replace lost packets with a repeated overlay of the last x recent samples where x is some small number (less than 400). This would likely retain the quality increases seen in most recent packet replacement at low loss rates and packet sizes because it retains frequency information, and would remain equal to or better than all the previous packet replacement policies at different loss rates and packet sizes by lessening the disruption of larger packets. Another packet replacement policy, although more computationally complex, could be to replace packets with sound generated from an fft of the most recent samples. This could work very well on more instrumental sounds, although it may have some problems with human voices given the rapid tone changes we experience.

Conclusions

This experiment revealed a few reasonably important conclusions. First, smaller packet size increases audio quality in almost all cases. Second, as long as high quality audio is not necessary any loss rate below 5 percent is generally acceptable and if only understanding is needed, loss rates below 30 percent are generally acceptable. Audio then becomes almost completely obfuscated starting around 60 percent up through around 80 percent loss rate. Finally, replacing lost packets with the most recent packet can greatly increase audio quality at low packet sizes and low loss rates, but may decrease quality away from these conditions.