Computer Network Problem Set5

Chapter 7 1. P3 2. P7 3. P11 4. P13 5. P18 6. P20 7. P23

P3. Recall the simple model for HTTP streaming shown in Figure 7.3. Suppose the buffer size is infinite but the server sends bits at variable rate $x(t)$. Specifically, suppose $x(t)$ has the following sawtooth shape. The rate is initially zero at time $t = 0$ and linearly climbs to H at time $t = T$. It then repeats this pattern again and again, as shown in the figure below.

a What is the server's average send rate?
Answer: the average send rate is H/2

b Suppose that Q=0,sothattheclientstartsplaybackassoonasitreceives a video frame. What will happen?
Answer: Some time the video would play slow, sometimes the video may play fast, so it will cause uncomfortable for the users.

c Now suppose Q > 0. Determine as a function of Q, H, and T the time at which playback first begins.
Answer: When Q > 0,
If(H*T*1/2) > Q
Then t*(H/T)*t*1/2 = Q
T = (2Q*T/H)(1/2)
If((H*T*1/2)) < Q;
Then determine the N that
N(H*T*1/2) < Q and (N+1) (H*T*1/2) >=Q;
t(H/T*t) * 1/2 = N*T - (Q-N(H*T*1/2))
t = ((N*T - (Q-N(H*T*1/2)))*2 *T/H)(1/2)

d Suppose $H > 2r$ and $Q = HT/2$. Prove there will be no freezing after the initial playout delay.
Answer: proof it

So H > 2r and Q = HT/2 then.The video will be played at Time T, so because H > 2r, the average sned rate is lager than the play rate. So for every time slot, there would be packets left. And because initially, there

is HT/2 packets to consume which is packets for one time slot. So the buffer can never be empty.

e Suppose $H > 2r$. Find the smallest value of Q such that there will be no freezing after the initial playback delay.
Answer: When H > 2r.

Q = t*(H/T)t* 1/2;

And at time T/2, we need to make srue that the buffer>=0; the value we choose is 0;

So (T/2 - t)(2)*(H/T)*(1/2) = t*(H/T)t* 1/2

So (T/2 –t)(2) = t(2)

So t = T/4

f Now suppose that the buffer size B is finite. Suppose $H > 2r$. As a function of Q, B, T, and H, determine the time $t = t_f$ when the client application buffer first becomes full.
Answer: for everytime slot the left packtes is (H/2 –r)*T
The initialized packets is Q.
So we need to determine that N that Q + N*(H/2 - r)*T < B
Whiel Q+(N+1)(H/2 - r)*T > B
Now we now the t, when we do calculation to get the t.

P7. Consider the procedure described in Section 7.3 for estimating average delay d_i. Suppose that $u = 0.1$. Let $r_1 - t_1$ be the most recent sample delay, let $r_2 - t_2$ be the next most recent sample delay, and so on.
s
a For a given audio application suppose four packet shave arrived at the receiver with sample delays $r_4 - t_4$, $r_3 - t_3$, $r_2 - t_2$, and $r_1 - t_1$. Express the estimate of delay d in terms of the four samples.
Answer: the answer should be (((r4-t4))*0.9 + (r3- d3)*0.1)*0.9 + r2-d2)*0.9 +(r1-d1) * 0.1
Generalize your formula for n sample delays.

Answer: the formula is when y
For all I (ri-ti)(i - 1)* 0.1
For u it is u* for all u * (1-u)(j-1)(rj - tj) + (1-u)(n)(rn - tn)

C For the formula in Part b, let *n* approach infinity and give the resulting formula. Comment on why this averaging procedure is called an exponential moving average.
Answer: for i, when it is far from now, then it would increase with more 0.9. so that means with the time passes the weight would decrease with 0.9. So that is the exponential moving average.

P11. Consider the figure below (which is similar to Figure 7.7). 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 play out 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.

Answer: the overall rate for the first 8 bits are 8 time unit.
The delay for the packet 1 is 8 slots.
The delay for the packet 2 is 7 slots
The delay for the packet 3 is 9 slots.
The delay for the packet 4 is 8 slots.
The delay for the packet 5 is 7 slots.
The delay for the packet 6 is 9 slots.
The delay for the packet 7 is 8 slots.
The delay for the packet 8 is 8 slots.

b If audio play out 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 play out?
Answer: the number would be 3,4, 6,7

c If audio play out begins at t=9,which of the first eight packets sent will not arrive in time for play out?
Answer: 3 and 6

d What is the minimum play out delay at the receiver that results in all of the first eight packets arriving in time for their play out?
Answer: 3. We should begin at time slot 10.

P13. Recall the two FEC schemes for VoIP described in Section 7.3. Suppose the first scheme generates a redundant chunk for every four original chunks. Suppose the second scheme uses a low-bit rate encoding whose transmission rate is 25 percent of the transmission rate of the nominal stream.
How much additional band width does each scheme require? How much playback delay does each scheme add?

The additional bandwidth it needs it 1/4 the original bandwidth. Because it now cause 1/4 more redundancy

The first one will cause 4 delay of 5 packets.

The second one will cause delay of 1 packets.

Because 1 packet is the 25% of the original packets

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?

Answer: the first schema will use the redundant packet to reconstruct the original high-quality packet, and the second one will use the low-quality packet to replace the origin one.
Answer: Use the first schema, because we can reconstruct the missed schema if there is one lost in 5 packets.

C How do the two schemas perform if the first packet is lost in every two packets? Which scheme will have better audio quality?
Answer

For the first schema, we can not reconstruct the high quality packet using the redundant packet.But for the second schema, we could use the low quality packet the replace the high quality packet. The second one would be good, because we can use the low quality packet to replace the high-quality packet, so that we can get the acceptable version of video.

(read the book for the problem 18)

P18. Consider the figure below. Answer the following questions:

a.Assuming FIFO service, indicate the time at which packets 2 through 12 each leave the queue. For each packet, what is the delay between its arrival and the beginning of the slot in which it is transmitted? What is the average of this delay over all 12 packets?

For the timeline, for the input schema.

FIFO

Number	**Arrive time**	**Transmitted time**	**delay**
1	**0**	**0**	**0**
2	**0**	**1**	**1**
3	**1**	**2**	**1**
4	**1**	**3**	**2**
5	**3**	**5**	**2**
6	**2**	**4**	**2**
7	**3**	**6**	**3**
8	**5**	**7**	**2**
9	**5**	**8**	**3**
10	**7**	**9**	**2**
11	**8**	**10**	**2**
12	**8**	**11**	**3**

So the average dalay time should be 23/12 = 1.91.

b. Now assume a priority service, and assume that odd-numbered packets are high priority, and even-numbered packets are low priority. Indicate the time at which packets 2 through 12 each leave the queue. For each packet, what is the delay between its

arrival and the beginning of the slot in which it is transmitted? What is the average of this delay over all 12 packets?

Number	Arrive time	Transmitted time	delay
1	0	0	0
2	0	2	2
3	1	1	0
4	1	6	5
5	3	3	0
6	2	7	5
7	3	4	1
8	5	9	4
9	5	5	0
10	7	10	3
11	8	8	0
12	8	11	3

Answer: So the average delay time is get the average number.

So the average delay is 23/12 = 1.91

c.Now assume round robin service. Assumethatpackets1,2,3,6,11,and 12 are from class 1, and packets 4, 5, 7, 8, 9, and 10 are from class 2. Indi- cate the time at which packets 2 through 12 each leave the queue. For each packet, what is the delay between its arrival and its departure? What is the average delay over all 12 packets?

Number	Arrive time	Transmitted time	delay
1	0	0	0
2	0	2	2
3	1	4	3
4	1	1	0
5	3	3	0
6	2	6	4
7	3	5	2
8	5	7	2

9	**5**	**9**	**4**
10	**7**	**11**	**4**
11	**8**	**8**	**0**
12	**8**	**10**	**2**

So the average delay is 23/12 = 1.91

d. Now assume weighted air queueing(WFQ) service. Assume that odd- numbered packets are from class 1, and even-numbered packets are from class 2. Class 1 has a WFQ weight of 2, while class 2 has a WFQ weight of 1. Note that it may not be possible to achieve an idealized WFQ schedule as described in the text, so indicate why you have chosen the particular packet to go into service at each time slot. For each packet what is the delay between its arrival and its departure? What is the average delay over all 12 packets?

I will choose the first one from the class 1, second packet from class 2 and third from class 1, and so one. To make sure I the weight of 2:1

Number	**Arrive time**	**Transmitted time**	**delay**
1	**0**	**0**	**0**
2	**0**	**1**	**1**
3	**1**	**2**	**1**
4	**1**	**4**	**3**
5	**3**	**3**	**0**
6	**2**	**7**	**5**
7	**3**	**5**	**2**
8	**5**	**9**	**4**
9	**5**	**6**	**1**
10	**7**	**10**	**3**
11	**8**	**8**	**0**
12	**8**	**11**	**3**

1 2 3 5 4 7 9 6 11 8 10 12

odd even odd odd even odd odd even odd even even even

I try to implement that, odd even and odd pattern, so that the odd even can be 2:1.

The average delay is 23/12 = 1.91

e.What do you notice about the average delay in all four cases(FIFO,RR, priority, and WFQ)?

Answer: although the priority is different, all methods achieve the same average delay which is 1.91s. And different methods have different implementation, some of teme make more priority one fairness like round robin

P20. Consider the figure below, which shows a leaky bucket policer being fed by a stream of packets. The token buffer can hold at most two tokens, and is ini- tially full at $t = 0$. New tokens arrive at a rate of one token per slot. The out- put link speed is such that if two packets obtain tokens at the beginning of a time slot, they can both go to the output link in the same slot. The timing details of the system are as follows:

1. Packets(if any) arrive at the beginning of the slot. Thus in the figure, packets 1, 2, and 3 arrive in slot 0. If there are already packets in the queue, then the arriving packets join the end of the queue. Packets proceed towards the front of the queue in a FIFO manner.

2. After the arrival shave been added to the queue, if there are any queued packets, one or two of those packets (depending on the number of avail- able tokens) will each remove a token from the token buffer

and go to the output link during that slot. Thus, packets 1 and 2 each remove a token from the buffer (since there are initially two tokens) and go to the output link during slot 0.

3. A new token is added to the token buffer if it is not full, since the token generation rate is $r = 1$ token/slot.

4. Time then advances to the next time slot, and these steps repeat. Answer the following questions:

a. For each time slot, identify the packets that are in the queue and the number of tokens in the bucket, immediately after the arrivals have been processed (step 1 above) but before any of the packets have passed through the queue and removed a token. Thus, for the $t = 0$ time slot in the example above, packets 1, 2 and 3 are in the queue, and there are two tokens in the buffer.

Time slot	Packets in queue	Token Number
0	1,2,3	2
1	3,4	1
2	4,5	1
3	5,6	1
4	6	1
5		1
6	7.8	2
7	9,10	1
8	10	1

Answer: Packet number arrived time and transmitted time.

Packet Number	Arrive Time	Transmit time	Token number	Delay time
1	0	0	1	0
2	0	0	0	0
3	0	1	0	1
4	1	2	0	1
5	2	3	0	1
6	3	4	0	1
7	6	6	1	0
8	6	6	0	0
9	7	7	0	0
10	7	8	0	1

b. For each time slot indicate which packets appear on the output after the token(s) have been removed from the queue. Thus, for the $t = 0$ time slot in the example above, packets 1 and 2 appear on the output link from the leaky buffer during slot 0.

Time slot	Outlink packets
0	1,2
1	3
2	4
3	5
4	6
5	
7	7,8
8	9
9	10

P23. Consider the leaky-bucket policer that polices the average rate and burst size of a packet flow. We now want to police the peak rate, p, as well. Show how the output of this leaky-bucket policer can be fed into a second leaky bucket policer so that the two leaky buckets in series police the average rate, peak rate, and burst size. Be sure to give the bucket size and token generation rate for the second policer.

Answer:

The second bucket should generate one token at one time slot, so that the there is one output per time slot.

We can set the second bucket the token rate is P tokens/second. And the maximum number of tokens is 1. So that we can control the average rate is P packets/second in the long run. And the output could be quit smooth.

For the first bucket, the average token rate is r tokens/second, and the maximum

number is b tokens, so it can control the peak rate of the flow