521150A Introduction to Internet

Exercise 3B

Welcome to calculation exercises

In this non-mandatory part of the course, you will learn to

- answer concrete, numerical questions on the subject matter
- model different network scenarios
- use the presented algorithms and methods

Contact info for issues with these exercises

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- room: TS387 (prefer email for contacting)

Before the session

Complete pre-exercises in Moodle

- you will be better prepared for the exercise
- you can earn up to **1 point for final grading** (in Moodle, the points are scaled by 10)

During the session

Follow, participate and solve

- revisiting pre-exercises
- solving example problems together
- describing and going through how to solve the advanced problems

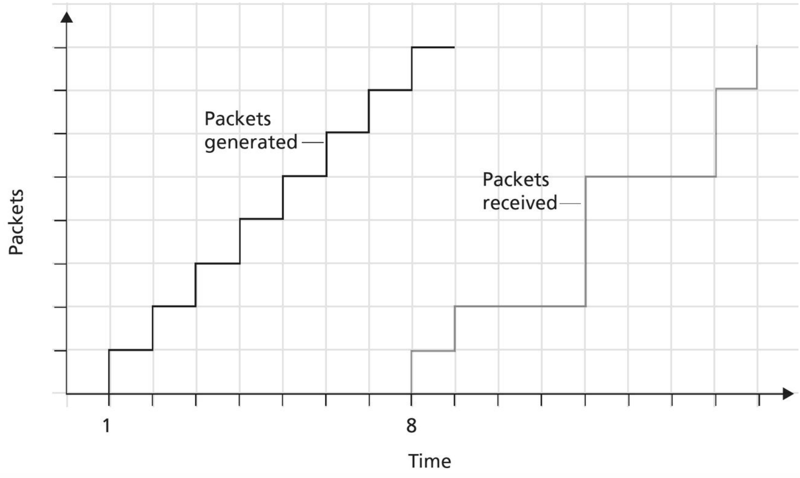
After the session

Solve the rest of the problems presented in this document

- return a scanned version (or good photo) as PDF of your hand-written solutions **with your name on it** to Moodle **before the next exercise**
- you can earn up to **0.5 points for each solved problem**, so a maximum of 1.5 points per exercise (in Moodle, the points are scaled by 10)

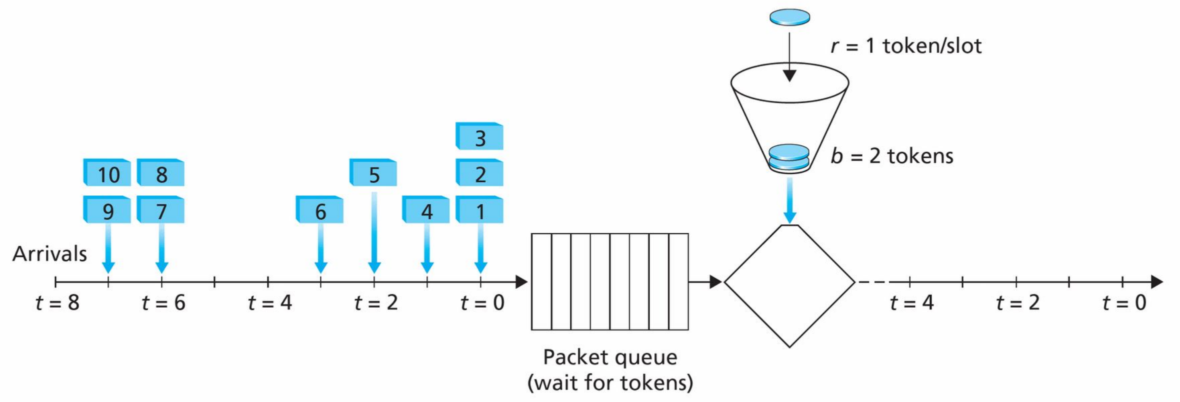
Pre-assignments

1. **In the diagram, one can seen when packets are generated at the sender and when they arrive at the repeater. How many time units does the playout delay have to be in order for all eight packets to arrive in time for playout?**



   9, toistamalla yksiköt tasan yhdeksän aikayksikön jälkeen tallennusaikaleimasta, kaikki paketit ehtivät ajoissa.

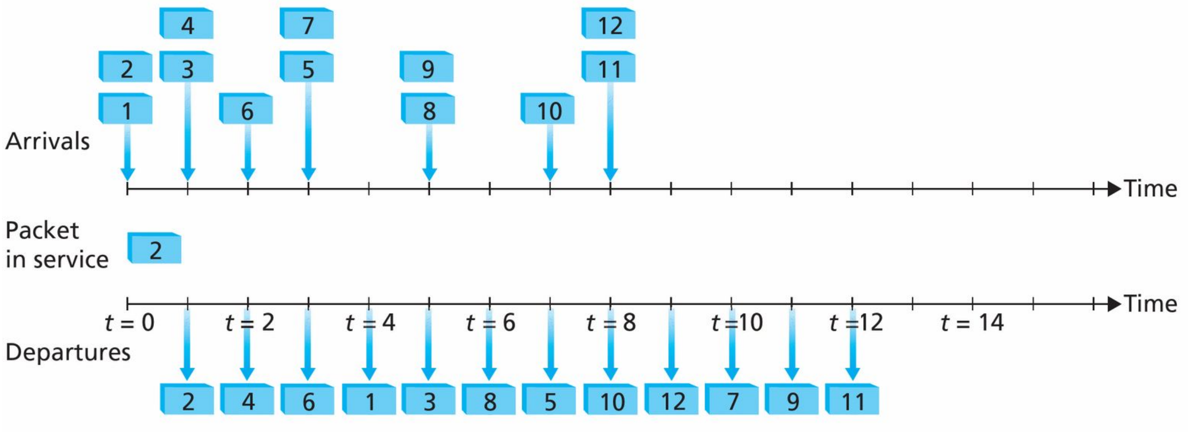
2. **In the situation seen in the diagram, token bucket algorithm is used to limit burstiness. Tokens are put into the bucket at constant rate of one token per time slot. The bucket has capacity of two tokens and it is full at start. Packets arrive at the queue from the left as seen in the diagram (for example, #6 arrives just before time slot 3, and so is processed at time slot 3). Because the bucket has two tokens, packets one and two are sent at time slot 0. When are other packets sent?**



   1,2,3,4,6,6,7,8

TIME	t=0	t=1	t=2	t=3	t=4	t=5	t=6	t=7	t=8
TOKENS	2	0+1	0+1	0+1	0+1	0+1	1+1	0+1	0+1
PACKETS	0+3	1+1	1+1	1+1	1+0	0+0	0+2	0+2	0+0
SENT (#)	2 (1,2)	1 (3)	1 (4)	1 (5)	1 (6)	0	2 (7,8)	1 (9)	1 (10)

Pre-assignments



3. The system described in the diagram can send one packet per time slot. A packet can be sent out at the next time slot after it's arrival. For example, packet #1 has queuing delay of zero because the queue is empty; therefore, the packet can depart immediately (at t=1). Compute the average queuing delay, when priority queue method is used. Packets with even index have higher priority than packets with odd index. Round to three decimal places (e.g., 15.100).

TIME	t=0	t=1	t=2	t=3	t=4	t=5	t=6	t=7	t=8	t=9	t=10	t=11	t=12
ARRIVAL (WAIT)	1,2	3,4	6	5,7		8,9		10	11,12				
IN QUEUE (WAIT)	-	1(1)	1(2),3(1)	1(3),3(2)	3(3),5(1),7(1)	5(2),7(2)	5(3),7(3),9(1)	7(4),9(2)	7(5),9(3)	7(6),9(4),11(1)	9(5),11(2)	11(3)	
SENT (WAIT)		2(0)	4(0)	6(0)	1(3)	3(3)	8(0)	5(3)	10(0)	12(0)	7(6)	9(5)	11(3)

(0+0+0+3+3+0+3+0+0+6+5+3)/12 = 1.917

4. A router is using a token bucket algorithm for congestion control. It has a capacity (C) of 250 KB and the maximum output rate (M) of 25 MB/s. Tokens are inserted into the bucket at rate (r) of 2 MB/s. What is the maximum burst length (T)? Round your answer to milliseconds with no decimals.

T=250KB/(25MB/s-2MB/s)=0.010869 s = 11ms

C = token bucket capacity (B) = 250KB = 250,000 B
r = token arrival rate (B/s) = 2MB/s = 2,000,000 B/s
M = maximum output rate (B/s) = 25MB/s = 25,000,000B/s
T = burst length (s) = C/(M-r)

5. A host needs to send 1MB of burst data, but the router is using the token bucket algorithm described in question #4. Assume that the bucket is already full, so M*T of the data is transmitted during the burst. How long does it take to transmit the rest of the data? Write your answer in ms without decimals.

1MB-275KB=725KB, use token arrival rate to calculate 725KB / 2MB/s=362.5ms

Pre-assignments

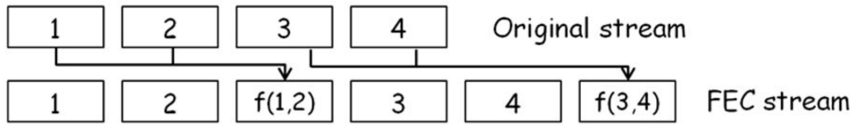
6. **RTP is used to transmit stereo audio packets. A system samples both channels at rate of 44100 samples per second, each of them containing 16 bits of data. How many packets have to be transmitted using RTP, if audio data is segmented into 1024 byte packets? Give your answer as a decimal number, round to one decimal place.**

   44100*2*2B/1024B=172,3

7. **In a multimedia system, errors can be corrected using Forward Error Correction (FEC) methods. The system in question implements error detection and correction by introducing redundancy into the system by encoding 4 bit data words into 7 bit data words (e.g., using Hamming(7,4)). Choose the correct statements - how the chosen FEC method affect the network traffic characteristics:**

   - Dataliikenteen vaatima kaista 1,75-kertaistuu - Requires 1,75x bandwidth for data
   - Toistoviive pysyy kutakuinkin samana - Does not affect playout delay noticeably (additional delay for decoding)

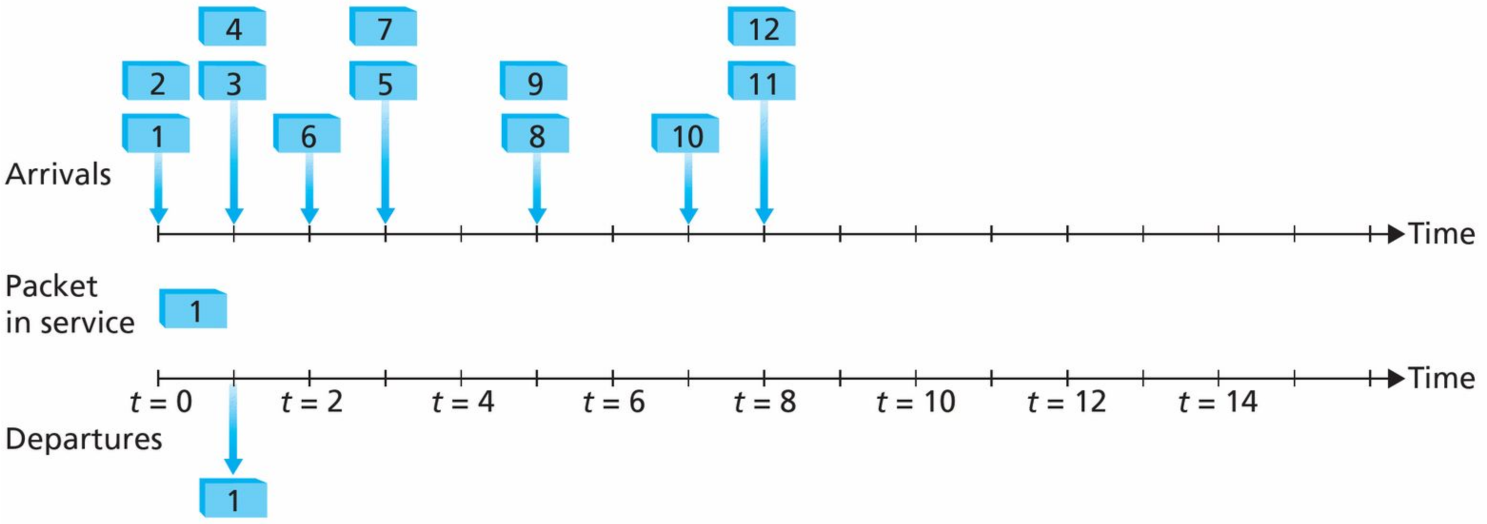
8. **In a multimedia system, errors can be detected and corrected using FEC methods. The system in question transmits 8 bit data words to the receiver. The error correction is implemented by computing a third packet: a parity packet from the two data packets, which can be used to recover one packet of the two packets using the other packet and the parity packet (see the picture). Choose the correct statements - how does the FEC method affect the network traffic characteristics?**



   - Laatu pysyy häiriöiden kasvaessa pitkään korkeana, mutta tippuu jyrkästi kun paketteja alkaa hukkumaan enemmän - Quality does not degrade until the communication channel experiences major packet loss, which causes significant degradation in quality (original n chunks can be reconstructed if there is at most one lost chunk from the n+1 chunks)
   - Toistoviiveen on varmistettava, että pariteettipaketti ehtii myös saapua perille - In order to playout media chunk, all three packets have to arrive; therefore, playout delay increases (playout delay needs to be fixed to the time to receive all n+1 packets!)

Problem #12

a) Given the following queuing system, determine the total average queuing delay experienced by all packets, the delay experienced by class A=[1,4,7,9], and the delay experienced by class B=[2,3,5,6,8,10,11,12] when using i) FIFO, ii) Priority A (group A has higher priority than B), iii) Round Robin as your queuing method.



b) Now, divide the incoming packets as follows: odd numbered packets are from class A and even-numbered packets are from class B. Class A has a WFQ weight of 2, while class B has WFQ weight of 1. What are the average delays experienced by classes A and B compared to the total average?

Problem #13

Consider the packet arrival sequence of RTP packets (PCM μlaw encoded audio) below, consisting of two talk spurts. RTP audio sampling clock rate is 8000Hz.

RTP sequence number	RTP timestamp	network delay (seconds)
1	160	0.40
2	320	0.30
3	480	0.35
4	800	0.32
5	960	0.27
6	1120	0.45

Assume that the first packet is transmitted at time zero. Create the following table minimising the playout delay without losing any packets. You can work non-causally, i.e., may look ahead in time. Meaning, that you know in advance the network delay of future packets.

RTP #	RTP timestamp	Network delay	Sending time	Arrival time	Playout time
1	160	0.4 s	…	…	…
…	…	…	…	…	…

Problem #14

Consider the following FEC schemes:

- Scheme A generates a redundant chunk for every four original chunks, by XORing the four original chunks.
- Scheme B piggybacks into the original stream a lower bit-rate stream whose bit rate is 25% of the bit-rate of the original stream.
- Scheme C increases the size of each chunk by 25% by encoding the chunk into a codeword with additional bits
- Scheme D shuffles the data of four chunks evenly between the chunks that it sends

Fill in the table on the right

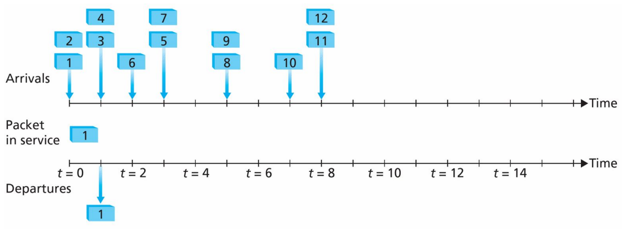
i. How much additional bandwidth does each scheme require?
ii. How much playback delay each scheme adds?
iii. What happens to audio quality if the first packet is lost for every group of five packets in transmission? And which method is the best?
iv. What happens to audio quality if the first packet is lost for every group of two packets in transmission? And which method is the best?

Scheme	Bandwidth (i)	Playback delay (ii)	Audio quality (iii)	Audio quality (iv)
A	+25%	4+1 chunks (+25%) before playback		
B				
C				
D				

Problems 3B

Problem #12:

a) **Given the following queuing system, determine the total average queuing delay experienced by all packets, the delay experienced by class A=[1,4,7,9], and the delay experienced by class B=[2,3,5,6,8,10,11,12] when using i) FIFO, ii) Priority A, iii) Round Robin as your queuing method.**



b) **Now, divide the incoming packets as follows: odd numbered packets are from class A and even-numbered packets are from class B. Class A has a WFQ weight of 2, while class B has WFQ weight of 1. What are the average delays experienced by classes A and B compared to the total average?**

Notes for #12

FIFO: First-In-First-Out
Priority A: class A should be transmitted first
Round-Robin: A-B-A-B-...
Weighted Fair Queuing: A has weight 2 and B weight 1, so in every cycle, 2A and 1B should be put to transmission -> AA-B-AA-B-...

Problem #13: Consider the packet arrival sequence of RTP packets (PCM µlaw encoded audio) below, consisting of two talk spurts. TRP audio sampling clock rate is 8000Hz.

RTP sequence number	RTP timestamp	network delay (seconds)
1	160	0.40
2	320	0.30
3	480	0.35
4	800	0.32
5	960	0.27
6	1120	0.45

Assume that the first packet is transmitted at time zero. Create the following table minimising the playout delay without losing any packets. You can work non-causally, i.e., may look ahead in time. Meaning, that you know in advance the network delay of future packets.

RTP #	RTP timestamp	Network delay	Sending time	Arrival time	Playout time
1	160	0,40s	…	…	…
…	…	…	…	…	…

Notes for #13

- 8KHz sampling rate and RTP timestamps differ 160, so each packet contains 20ms of audio
- two speech bursts separated with silence (which is not sent, so one RTP timestamp is missing)
- playout time between packets should differ 20ms within a speech burst, same with sending time
- arrival time = sending time + network delay

Notes for #14

- see lecture 13 slides
- prefer lower bitrate over silence or gaps in playback

Problem #14: Consider the following FEC schemes:

- **Scheme A** generates a redundant chunk for every four original chunks, by XORing the four original chunks.
- **Scheme B** piggybacks into the original stream a lower bit-rate stream whose bit rate is 25% of the bit-rate of the original stream.
- **Scheme C** increases the size of each chunk by 25% by encoding the chunk into a codeword with additional bits
- **Scheme D** shuffles the data of four chunks evenly between the chunks that it sends

Fill in the table below

i) How much additional bandwidth does each scheme require?
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Scheme	Bandwidth (i)	Playback delay (ii)	Audio quality (iii)	Audio quality (iv)
A				
B				
C				
D				