ELT-51106 computer networking 1

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Student name, in block capitals: Student number:

PAY ATTENTION to the exam rules.

Not reading them may harm your final score.

- Exam structure:
 - The exam lasts 3 hours. The questions are split into 3 sections, each with its own kind of questions. Guideline times are 45, 45 and 90 minutes for each section respectively. If you fall behind, proceed to the next section, and then return to questions you skipped.
 - The maximum number of points you can get from this exam is 100. 50 points are necessary to pass. Amount of points earned per question is marked next to the respective question.
- Mark your answers on the sheets given to you do not submit any other papers. You may use the free space of the exam papers as scratch paper. Scratches and drafts are not checked.
- Read questions very carefully before answering.
 - If you find any issues in the exam questions (typos, formulation problems etc), please indicate them for possible bonus points. If you believe you have to make additional assumptions to give an answer, do so, and write them down next to question text. The additional assumptions you make do not affect the score as long as they result in correct answers.
- For multiple choice questions:
 - circle all the correct options. Please, do not tick, underline, strike out or otherwise indicate any of the options. Remember that any number of options may need to be circled;
 - marking all correct answers results in the maximum score. Wrongful marking results in a -1 penalty.
- For open questions:
 - give the answer in the space provided. If it is insufficient, you are probably doing something wrong. In addition, one picture is often worth more than a 1000 words;
 - open questions are rated based on teacher's decisions, unless otherwise indicated in the question text.
- I have read and understood the rules and regulations Tick here

Good luck!

Exam questions

A Basic theory (40 points)

A.1 Symbols and alphabets

How many bits are required to uniquely encode a word written in an alphabet $\{a, A, b, B, c, C, d, D\}$? The word has exactly 7 letters.

1.	4	3.	14
2.	21	4.	56

A.2 Repetition coding

For the purposes of error correction one adds three copies of each symbol in the transmitted message, e.g. if AB was supposed to be sent, AAABBB will be sent instead. This is called repetition coding. The question is, how many symbols must get corrupted during transmission to break error correction mechanism, in this example?

1.	One is enough	3.	Any two symbols
2.	Either first two or last two	4.	Any three symbols

A.3 Fragmentation

What are true statements about fragmentation in digital communications?

- 1. Addressing of network nodes
- 2. Enabling partial retransmission of large messages
- 3. Increase overhead in the link
- 4. Enable multihoming on switches to their connected devices

A.4 NAT

Which of the following statements are true about NAT.

- 1. N to 1 NAT only allows for outgoing connections to be initiated
- 2. NAT stands for Nation Addressing Technique
- 3. Using 1-to-1 NAT techniques, with only 5 connections available, the 6th connection will be overloaded and port translated.
- 4. Dynamic pool NAT and cone NAT block possible incoming connections

A.5 Network addressing

The networks are typically divided into hierarchical and flat addressing networks. Indicate the examples of **flat** addressing schemes from the list below.

- $1.~\mathrm{IP}$ addresses: 10.5.12.0
- 2. Postal addresses: Finland, 33700 Tampere, Korkakoulunkatu 10, office 1, Pekka Postinen
- 3. P2P network hashes: btih:a9878dd831f70edd641d1b8dbcb174fce7b3dee2
- 4. VLAN tag: 1453

A.6 Circuit switching

Select the true statements about circuit switching networks

- 1. A three-way handshake is required before communication can begin
- 2. Two nodes may have multiple circuits established between them
- 3. Circuit switched networks reserve bandwidth for a connection that will be established, thus providing high reliability
- 4. Circuit switched networks cannot handle packets being transferred over their links

[4 points]

[4 points]

[4 points]

[4 points]

[4 points]

[4 points]

A.7 Forwarding in IP networks

Which of the following statements are true about the IP protocol?

- 1. Forwarding in IP is determined by the destination address of the packet
- 2. The routers in IP use routing tables to determine where to forward packets
- 3. The IP addresses are 32 bit integers, where each byte represents a single forwarding decision to be taken
- 4. IP is a flow-based protocol, such that a given destination is always reachable through the same set of intermediate nodes.

A.8 Time to live

It is common for network protocols to include a time to live (TTL) value in the headers of the packets. What is the **purpose** of this field?

- 1. To indicate how much time (miliseconds) has passed since the first transmission
- 2. To indicate when the remote device should power down its interface and stop receiving data
- 3. To prevent requests and packets from traveling through the network indefinitely
- 4. To make it possible to ask a remote device to send you a packet, if it happens to be somewhere on the path between you and the destination

A.9 DHCP

Indicate the services provided by the DHCP.

- 1. Distribution of MAC addresses to the connecting devices
- 2. Informing about available network services (like DNS servers)
- 3. Setting up of security codes
- 4. Distribution of the routing tables to different routers.

A.10 Flow & Congestion Control

Which of the following statements are true about flow & congestion control?

- 1. UDP uses flow control
- 2. Closed loop congestion control uses feedback from the receiver to inform the sender
- 3. Flow control adjusts the flow of traffic based on the network load
- 4. The TCP window size is equal to the minimum value of the receiving window and the congestion window

B Short open questions (30 points max)

B.1 IP subnetting

You are given an address range (subnet) 130.230.0.0/25. Allocate 4 reasonably-sized subnetworks, with following loads: [10,10,20,40] hosts. Fill your answers into table below. Write the IP addresses in dotted decimal format. There may be multiple ways you can distribute the addresses. Choose your router as the first available address (lowest number).

Network address	Mask length	Router address	Broadcast address

[4 points]

[4 points]

[4 points]

[8 points]

B.2 ARP

Consider an Ethernet-based network described in the picture below (it is very similar to the setup we used in the lab).



Now fill in the table with messages that would be sent if **D** was to send a UDP packet to **B**, and **B** would reply with a UDP packet. Consider that the network was just powered up, and all the ARP caches are empty. Assume that the routing and forwarding information is filled in correctly on all hosts.

Ethernet header		IP header (if exists)		Body /payload
MAC source	MAC	IP source	IP	
	destination		destination	
				What is MAC for "IP R2"?
				MAC for "IP R2" is "MAC R2"
				UDP request
				What is MAC for "IP B"?
				MAC for "IP B" is "MAC B"
				UDP request
				UDP response
				UDP response

B.3 Client-server vs P2P

[6 points]

Using the picture below, illustrate the differences between peer to peer and client-server models in case of file distribution. Consider node S to be the one with the original copy of the file, and draw the connections that would be established to distribute the file. Assume that the node S in client-server case can serve up to 3 downloaders at a time, while in P2P mode all the nodes can serve at most 2 downloaders at a time, and can not download and upload at the same time. Assume also that the message can not be fragmented.



How long (assuming 1 transfer takes 1 time unit) would the download take for both systems to distribute the file?

Client server:	P2P:		

Prove the correctness of your statement. (max 40 words)

Why:

B.4 Protocol stack

[8 points]

What is the set of protocols that is used when a typical browser (unsecure) downloads a web page from a typical web server? Assume that the browser runs on a Windows laptop that has just entered a WiFi network, has not yet sent any IP packets anywhere and the configuration has a valid IP address. Also assume that the IP address of the web server is not known in advance, only its hostname.

Layer	Protocol (name all that are used)
Application	
Transport	
Network	
Data-link	

Almost ready, final section ahead! :)

C Short exercises (30 points)

C.1 VoIP

Consider an aggregated voice over IP telecommunications link, for which the following is known: When a user is talking 40 kbit/s of traffic is generated When a user is not talking, 8 kbit/s of traffic is generated The ratio of talking to idle periods during conversation is 1:1. There are 100 of such VoIP sessions in the link of interest. Answer the following questions:

• What is the required average throughput of a link to serve the connections? Answer:	[1 point]
• What is the maximal rate that can be generated by the connections? Answer:] [1 point]
• Which rate will you use when buying equipment for the link? Answer:	[2 points]
• What is the maixmal rate, considering 4:1 ratio of talking to idle? Answer:	[2 points]
Calculations:	

C.2 IoT performance

[12 points]

[6 points]

Consider a tiny sensor which needs to send 20 bytes of data to a remote monitoring host (e.g. temperature reading). The MAC header is 18 bytes, TCP header is 20 bytes, UDP header is 8 bytes, IP header is 20 bytes. What is the **total number of bytes** that will be transferred between the sensor and the remote monitoring host (including the transport layer's connection setup, connection closure and acknowledgments). Assume no piggyback riding is involved in this particular case.

• Case 1: TCP is used, ACK sent on transport layer by TCP	Answer:	[8 points]
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• Case 2: UDP is used, and 1 byte ACK sent on application layer [4 points]

Calculations:	

C.3 Fragmentation, HTTP MTU, TCP

	SENDER	RECEIVER
Consider an HTML webpage, 6900 bytes in length, that has to be transmitted from point A to point B using HTTP. The message is coupled with a 200 byte HTTP header, and transported through TCP socket. The MTU of the link layer is 1500 bytes. Your task is to calculate, what is the minimum number of TCP packets that is needed to get the message across. Keep in mind that in this particular implementation TCP sends an ACK packet at least for every two data packets. Assume that the connection was established in advance, and will not be terminated when the message is delivered. Give the final answer as an integer. Hints: TCP header size is 20 bytes, IP header size is 20 bytes. Draw the transferred packets as arrows on the diagram (no need to add names, sizes or numbers).		
Calculations:		

Answer:

Additional comments on the course and the exam (optional):

Thank you!