

## Information

- This exam is for course codes IK2215, IK2204, and 2G1701
- The duration of the exam is 4 hours (14.00-18.00).
- Answers should be well structured and readable.
- Write your name and personal-id/date-of-birth on each page.
- No help material is allowed.
- Answers will be posted on the course web within 3 working weeks after the exam.
- Results will be published no later than November 9, 2018. Requests for grading re-evaluations should be made according to the routines specified by KTH School of Information and Communication Technology.
- The exam consists of 2 parts; Part A and Part B. Part A is a set of questions with short answers. **Respect the word limits!** Answers longer than the word limit will be truncated, meaning that we will disregard from the part of your answer that exceeds the word limit during the exam marking. Part B is a smaller set of questions that require more elaborative answers. To pass the exam you need to attain a certain number of points (preliminary 75%) on Part A. Higher grades (A-C or 4-5) will be based on the total score (Part A + Part B). **Part B will not be graded for those who do not pass Part A.**
- Preliminary grading is as follows:

Points	Grade (A-F)
23-30 points on Part A and 45-50 points in total	A
23-30 points on Part A and 40-44 points in total	B
23-30 points on Part A and 35-39 points in total	C
23-30 points on Part A and 23-34 points in total	D
21-22 points on Part A and passed complementary assignment	E
21-22 points on Part A (complementary assignment offered)	Fx
0-20 points on Part A	F (Fail)

Points	Grade (U-5)
23-30 points on Part A and 42-50 points in total	5
23-30 points on Part A and 37-41 points in total	4
23-30 points on Part A and 23-36 points in total	3
21-22 points on Part A (complementary assignment offered)	U
0-20 points on Part A	U (Fail)

**Good Luck!**

**Exam Part A (30p) (Note the word limits)****1) Various true/false statements (10p)**

Mark the following statements as **true** or **false**. Don't write "t" or "f", since indistinct hand-writing makes it hard to differ between the two.

**Note:**

- you will get 1p for each correct answer
  - you will get -1p for each wrong answer
  - you will get 0p for each "no answer"
  - you will **not** get less than 0p in total on this question
- 
- A. OSPF can be configured into several areas to support a large number of routers in the network.
  - B. UDP has an optional sequence number so that UDP datagrams can be re-ordered at the receiving side. (1p)
  - C. BGP uses an enhancement of distance vectors called path vectors. (1p)
  - D. The overall purpose with RPM (Reverse Path Multicasting) is to deal with flooding and packet duplicates. (1p)
  - E. According to GeSI (Global e-Sustainability Initiative), the greenhouse gas (GHG) emissions from ICT is only about 10% of the GHG emissions from the aviation industry. (1p)
  - F. In int-serv (integrated services), traffic is divided into a small set of traffic classes and resources are allocated on a per-class basis. (1p)
  - G. The number of available IPv6 addresses is four times higher than the number of IPv4 addresses. (1p)
  - H. IP multicast only works for link technologies with multicast-capable networking hardware, such as Ethernet. (1p)
  - I. The main purpose of RTP (Real-time Transport Protocol) is to provide time-stamps and sequence numbers when sending real-time data. (1p)
  - J. In a hierarchical peer-to-peer overlay network, flooding is normally limited to an overlay of super peers. (1p)

**Answer:**

- A. True
- B. False
- C. True
- D. True
- E. False
- F. False
- G. False
- H. False
- I. True
- J. True

**2) Various questions with short answers (10p)**

Answer the following questions with short answers.

**Note:**

- You will get 1p for each entirely correct answer
- Word limit per question: 30 words

- A. Place the following four protocols at the correct layer in the TCP/IP protocol stack: SMTP, ICMP, TCP, and PPP. (1p)
- B. Can you aggregate the prefixes 199.1.1.0/25 and 199.1.1.128/25 to one single subnet? If so, specify (in CIDR notation) the resulting aggregated prefix. If not, explain why. (1p)
- C. At a given time, a TCP sender has a congestion window of 2200 bytes and a receiver-advertised window of 1800 bytes. It has just sent 900 bytes without receiving any ACK. How many more bytes can be transmitted before the sender has to wait for an ACK? (1p)
- D. What is the purpose with a playback buffer in multimedia networking? (1p)
- E. There are two main types of delivery trees in multicast routing. Which ones? (1p)
- F. Is the EEE (Energy-Efficient Ethernet) standard based on rate switching (switch to lower transmission rate when possible) or low-power idle (sleep between packets when possible)? (1p)
- G. Router A is running RIP and sends out a distance vector. Which routers are the receivers of this distance vector? (1p)
- H. IP options have been removed in IPv6. What does IPv6 use instead of options? (1p)
- I. What does policing mean in the context of IP QoS (Quality-of-Service)? (1p)
- J. What is the overall purpose with 6LoWPAN? (1p)

**Answers**

- A. SMTP: Application layer, ICMP: Network layer, TCP: Transport layer, PPP: Link layer.
- B. Yes, the aggregated prefix is 199.1.1.0/24.
- C. 900 bytes (1800-900).
- D. To absorb network-induced delay jitter.
- E. Source based trees and group shared trees.
- F. EEE uses low-power idle.
- G. Router A's neighbors.
- H. Extension headers
- I. Ensuring that a reserved flow conforms to the traffic specification..
- J. Make it possible to run IPv6 on constrained low-power devices.

**3) Internet-of-Things (2p) (Word limit: 50)**

Compare CoAP and MQTT when it comes to different properties. Use the table below and identify which one of the protocols (CoAP or MQTT) that best matches the given property. Redraw the table on your answer sheet and fill in the protocol column.

Property	Protocol
Specifically designed for constrained devices	
Uses UDP as the transport protocol	
Originally designed after a publish-subscribe model	
Relays data via a broker	

**Answer:**

Property	Protocol
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Specifically designed for constrained devices	CoAP
Uses UDP as the transport protocol	CoAP
Originally designed after a publish-subscribe model	MQTT
Relays data via a broker	MQTT

**4) IP multicast (2p) (Word limit: 80)**

In a large subnet there could be many simultaneous receivers of the same multicast group. When a router sends an IGMP query, this could then generate a large amount of IGMP reports. How is this avoided in an IGMP implementation?

**Answer:**

IGMP reports are sent to the group address. The hosts can then snoop for other host's IGMP reports. So, before sending its IGMP report a host will set a random timer and suppress its IGMP report if it detects that another host on the same subnet sends the report for this group.

**5) Interdomain routing (2p) (Word limit: 100)**

Inter-domain routing protocols are responsible for computing routes across Internet networks.

- A. Explain why RIP would not be a good inter-domain routing protocol. Mention two reasons.
- B. Are RIP and BGP guaranteed to converge to a stable routing state when operated in a static topology (i.e., no failures in the network)? Briefly motivate your answer.

**Answer:**

- A. Reason 1: RIP does not support expressive enough ranking of routes, e.g., per-neighbor ranking.  
Reasons 2: RIP does not allow to filter or export routes in a selective manner.
- B. RIP is a shortest-path distance-vector routing protocol. We proved in class that such protocols are guaranteed to converge to a stable routing in the absence of failures. In contrast, BGP is a policy-based path-vector protocol where operators are allowed to override the shortest-path best path selection with local-preferences over routes. When local-preferences among different networks are conflicting, the routing may never converge to a stable state.

**6) IPv6 (2p) (Word limit: 100)**

IP fragmentation is handled differently in IPv6 compared to IPv4. How is it different and what is the motivation for this change? Why is reassembly always done at the receiver?

**Answer:**

IPv6 does not allow for fragmentation by the intermediate routers. This operation is done only by the source. Fragmentation is a time-consuming process, so removing this from the routers releases the burden on the network.

Reassembly has to be done at the receiving host since different fragments may take different paths through the network. A router can thus not be assumed to receive all fragments of a packet.

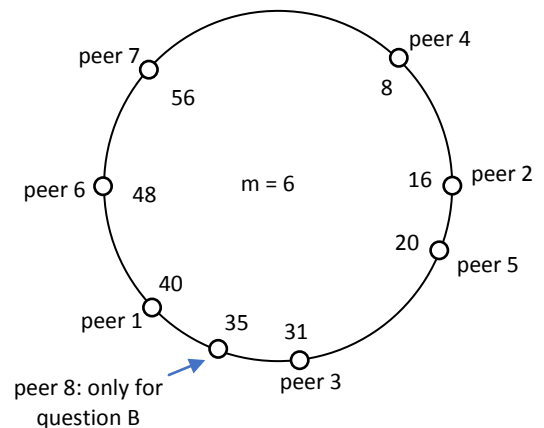
**7) Peer-to-peer networking (2p) (Word limit: 100)**

Chord is a Distributed Hash Table that strikes a good trade-off between communication and memory overheads. Finger tables are the main data structures used in Chord to identify the peer that has information about a resource. Consider a small Chord DHT network with 64 IDs, from 0 to 63, and 7 peers that participate in the network (see figure below).

- A. Fill the finger table of peer node 7. Write the calculations of the key id field and the successor peer number.

Peer 7 finger table

i	key id	successor
0	56 +	peer
1	56 +	peer
2	56 +	peer
3	56 +	peer
4	56 +	peer
5	56 +	peer



- B. Which nodes will update their finger tables after a new peer with ID 35 joins the network? Write "YES" or "NO" and explain the YES answers. The word limit apply for this explanation.

peer	Update finger table?
1	
2	
3	
4	
5	
6	
7	

**Answer:**

A.

i	key id	successor
0	$56 + 1 \bmod 64 = 57$	peer 4
1	$56 + 2 \bmod 64 = 58$	peer 4
2	$56 + 4 \bmod 64 = 60$	peer 4
3	$56 + 8 \bmod 64 = 0$	peer 4
4	$56 + 16 \bmod 64 = 8$	peer 4
5	$56 + 32 \bmod 64 = 24$	peer 3

B.

peer	Update finger table?
1	NO
2	YES
3	YES

4	NO
5	NO
6	NO
7	NO

Peer 2 updates its finger table because  $16 + 2^4 \bmod 64 = 32$ , which falls between 35 and 32.

Peer 3 updates its finger table because  $31 + 2^0 \bmod 64 = 32$ , which falls between 35 and 32.

**Exam Part B (20p)****8) Quality of service in IP networks (5p)**

Suppose that the following flows, specified with token bucket traffic specifications, have been accepted by an int-serv (Integrated Services) capable router:

R (rate in packets/second)	B (bucket depth in no of packets)
2	9
5	6
8	5

All flows are in the same direction and the router forwards 20 packets per second. Note that the example is unrealistic in its use of packets, instead of bytes.

- What is the maximum delay a packet may face? (2p)
- What is the maximum number of packets from the third flow ( $r = 8$ ,  $B = 5$ ) that the router would send over 4.0 seconds, assuming the router sends packets at its maximum rate uniformly? (1p)
- What are the main drawbacks with the integrated service model for IP quality of service? (2p)

**Answer:**

- Max delay is given by max queue length, which is the sum of all buckets.  $B_{tot} = 9+6+5 = 20$ . Max delay =  $B_{tot}/\text{link capacity} = 20/20 = 1$  second.
- Max no of packets over 4 seconds for the third flow =  $rT + B = 8 \times 4 + 5 = 37$  pkts.
- The end-to-end connection set-up and the resource reservations on a per flow basis make int-serv very unpractical (impossible) to scale. There could literally be millions of flows to keep track of, each requiring its own buffer. The required state per flow in each router along the path is very costly for routers.

**9) TCP (5 p)**

Transport protocols are used to exchange data among processes in a network.

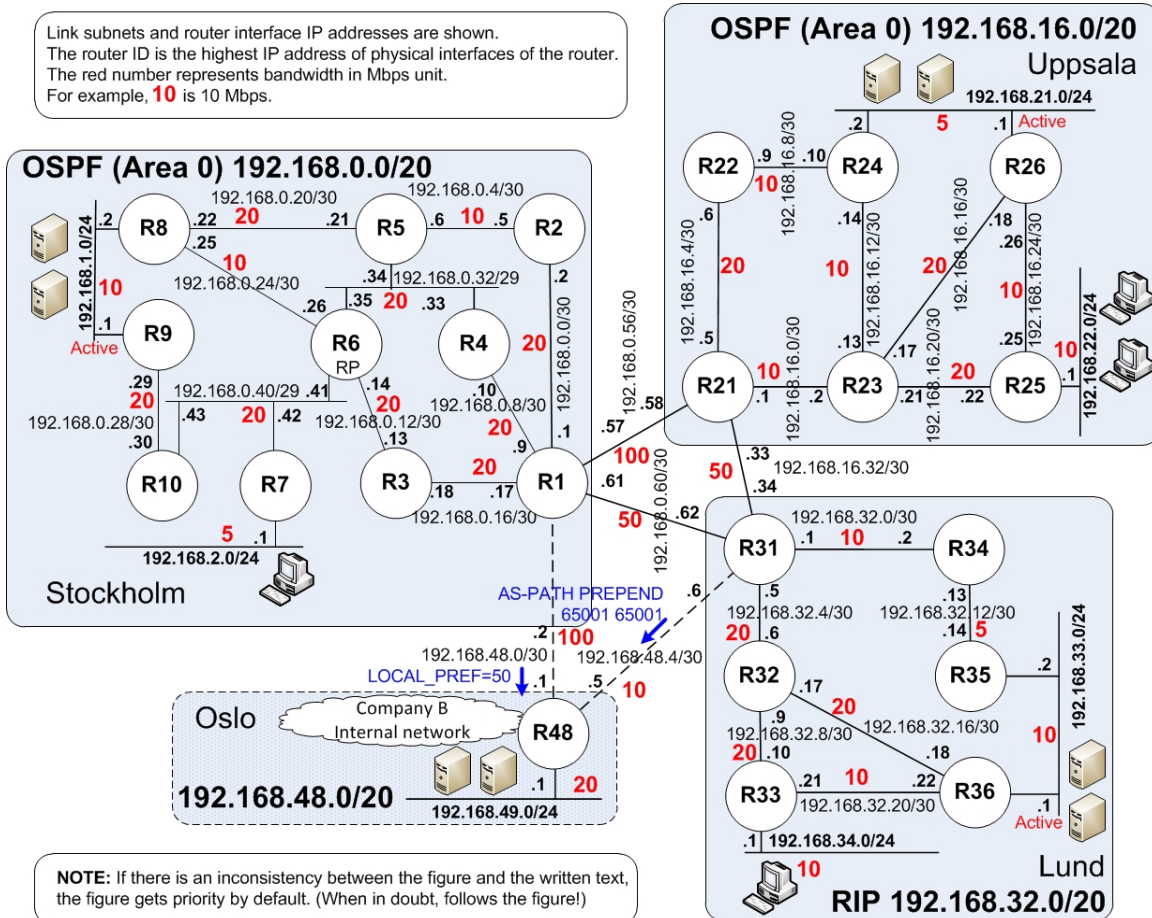
- Can one build a reliable communication on top of UDP? (Think carefully). Justify your answer. (1p)
- Consider a 10-second long TCP Reno connection between two hosts A and B connected by a 5ms link. Host A sends a large file to B. Let 100B be the average packet size and 100 packets the average sender congestion window of the connection. Assume the receiving window is not a bottleneck. Estimate the amount of data transmitted from the sender to the receiver. (1p)
- Explain how the throughput of a TCP Reno connection is affected by the RTT between the sender and the receiver and the drop rate. How can a sender improve its own throughput when competing against a TCP connection with a smaller RTT? (2p)
- What problem of HTTP/2 does QUIC solve and how? (1p)

**Answer:**

- A. Yes, one can implement the acknowledgement of packets on top of UDP within the process that relies on UDP. See QUIC.
- B. The average throughput is given by ("average window" X "average packet size" ) / RTT. Since the RTT is 10ms, we obtain an average throughput of  $10^6$  bytes per second. Since the connection is active for 10 seconds, the amount of data transmitted is  $10^7$  bytes.
- C. For a single flow, the higher the RTT the lower the throughput. The higher the packet drop, the lower the throughput. A user could open multiple TCP connections to increase the throughput.
- D. In HTTP/2 all streams are multiplexed within a single TCP connection. If there is a packet drop on one stream, all the other streams are also affected because TCP delivers packets in order. In QUIC, streams are multiplexed using UDP, which does not enforce packet ordering.



## 10) Large-scale network scenario (10p)



The above figure illustrates Company A's network topology. Company A has three branch offices in different cities, each with different networks. The first office is located in Stockholm using the 192.168.0.0/20 subnet. The second office is located in Uppsala using the 192.168.16.0/20 subnet. The last office is located in Lund using the 192.168.32.0/20 subnet. All routers are interconnected with different link bandwidths as shown in the figure. All routers are Cisco routers with default parameters set.

Each office has designed its own internal network and runs its own routing protocol internally. The Stockholm office uses OSPF as its sole routing protocol within its network. All routers (R1-R10) are running in the backbone area (OSPF area 0). Similar to the Stockholm office, the Uppsala office uses OSPF as its sole routing protocol within its network (on routers R21-R26). All routers are running in the backbone area (OSPF area 0). The Lund office uses RIPv2 on all its routers (R31-36). These routing protocols are not running on the links between the offices (no OSPF or RIP on the R1-R21 link, the R21-R31 link, or the R1-R31 link). However, different routing schemes are used for communication between branch offices.

Static route commands are configured as follows:

On R21: ip route 192.168.32.0 255.255.240.0 192.168.16.34

On R31: ip route 192.168.0.0 255.255.224.0 192.168.16.33

Company A runs two iBGP peerings as follows:

    between Stockholm's R1 and Uppsala's R21 (over R1-R21 link)

    between Stockholm's R1 and Lund's R31 (over R1-R31 link)

All routers run iBGP peering with private AS 65001 and have "no auto-summary" and "no synchronization" commands configured. They also have the following route advertisements:

R1 is configured to generate default route (0.0.0.0/0) in BGP and advertises it to all BGP peers (The generated default route is not installed on R1 routing table). R21 advertises an aggregate address 192.168.16.0/20 with "summary-only" to all BGP peers. Similarly, R31 advertises an aggregate address 192.168.32.0/20 with "summary-only" to all BGP peers.

In addition, the Stockholm office has configured R1 to always originate default route in OSPF. The Uppsala office has configured to redistribute BGP routes into OSPF on R21. The Lund office has configured to redistribute BGP routes into RIP on R31.

The Stockholm office has one server network (192.168.1.0/24) and one user network (192.168.2.0/24). HSRP (Hot Standby Router Protocol) is used to provide fault-tolerant default gateway for the server network. R9 is active router and R8 is passive router. The virtual IP address is 192.168.1.3, which is used as the default gateway by all servers.

The Uppsala office has one server network (192.168.21.0/24) and one user network (192.168.22.0/24). HSRP is used to provide fault-tolerant default gateway for the server network. R26 is active router and R24 is passive router. The virtual IP address is 192.168.21.3, which is used as the default gateway by all servers.

The Lund office has one server network (192.168.33.0/24) and one user network (192.168.34.0/24). HSRP is used to provide fault-tolerant default gateway for the server network. R36 is active router and R35 is passive router. The virtual IP address is 192.168.33.3, which is used as the default gateway by all servers.

Assume that default cost models are used for OSPF, RIP, as well as for static routes (a static route has a fixed cost of 1, OSPF cost formula is  $cost = \frac{100,000,000 \text{ bps}}{\text{bandwidth in bps}}$ ). When a route is redistributed from one protocol to the other, the original cost of the route will be accumulated into the new protocol before the route is forwarded to other routers. For BGP originated routes, the cost will be set to 10. In addition, if a router learns the same route from different routing protocols, it will prefer the route from the routing protocol in the following order: static, eBGP, OSPF, RIP, and iBGP. If a router learns a route with equal cost from the same routing protocol through multiple routers, it will prefer to use the route from the router with the lowest router ID. For example, if R51 (router ID 1.1.1.1) learns an OSPF route 10.0.0.0/24 from R55 (router ID 2.2.2.2) and R60 (router ID 3.3.3.3), it will prefer to use the OSPF route learned from R55 since its router ID 2.2.2.2 is lower than R60 (router ID 3.3.3.3). The router ID is the highest IP address on physical

interfaces. The virtual IP address is not considered when deciding the router ID.

Assume that the topology has converged. Answer the following questions:

- A. What path does a packet traverse when a host in Stockholm with IP address 192.168.2.11 sends an ICMP echo request to a server in Uppsala with IP address 192.168.21.5? (1p)
- B. What path does a packet traverse when a host in Uppsala with IP address 192.168.22.11 sends an ICMP echo request to a server in Lund with IP address 192.168.33.5? (1p)
- C. What path does a packet traverse when a host in Lund with IP address 192.168.34.11 sends an ICMP echo request to a server in Stockholm with IP address 192.168.1.5? (1p)

**IMPORTANT:** You must specify the next-hop IP address of every hop the packet traverses. If the ICMP echo request cannot reach the destination then identify what happens to the request.

**Example answer**

10.0.0.11 -> 10.0.0.1 -> 10.0.1.1 -> 10.0.2.2 -> 10.0.3.3  
1.1.1.1 -> 1.1.1.2 -> 2.2.2.2 -> 3.3.3.3 -> R77 drops the packet  
4.4.4.4 -> 4.4.4.5 <-> 5.5.5.5 (Loop between R88 and R89!)

Assume that the physical link between Uppsala and Lund (on R21-R31 link) was taken down and the border routers lose all routing information on this link. All routes on this link (including static routes) are removed from the routing tables. Assume that the topology has converged. Answer the following questions:

- D. What path does a packet traverse when a server in Uppsala with IP address 192.168.21.5 sends an ICMP echo request to a host in Lund with IP address 192.168.34.5? (1p)
- E. What path does a packet traverse when a server in Lund with IP address 192.168.33.5 sends an ICMP echo request to a host in Uppsala with IP address 192.168.22.5? (1p)

**IMPORTANT:** if the ICMP echo request cannot reach the destination then identify what happens to the request. See example answer above.

Assume that the R21-R31 link is back to normal and the original topology in the figure has converged. A network administrator would like to run a multicast routing protocol in order to distribute a video from a streaming server (IP 192.168.1.9/24) in the Stockholm office to all users in all branch offices. PIM sparse mode is used for this purpose and R6 is selected as a rendezvous point (RP).

To avoid confusion caused by having two routers on the network (HSRP routers), assume that the links between the passive routers and the server and client networks in each office are removed from the topology (R8-192.168.1.0/24 link, R24-192.168.21.0/24 link, and R35-192.168.33.0/24 link). The routers have HSRP configuration in tact (all virtual IP addresses are still in used)

Assuming that the SPT-threshold is set to infinity (the threshold is never exceeded). Answer the following questions:

- F. Which path is used for streaming from the streaming server to a host with IP 192.168.22.10 in Uppsala office? (1p)
- G. Which path is used for streaming from the streaming server to a host with IP 192.168.34.10 in Lund office? (1p)

Assume the topology is the same as for question F and G above. (All passive routers are still disconnected from the networks). The routers have the same router ID as when the passive links were still intact.

Company A has bought company B in Oslo and decided to connect company B office to Company A's network with two physical links; one from Oslo to Stockholm (R48-R1 link) and another from Oslo to Lund (R48-R31 link). To keep the internal routing policy of Company B intact, the network administrator decided to use BGP as the routing protocol on these links. Company B is assigned a private AS 65002 and runs two eBGP peering sessions with AS 65001; one with R1 in Stockholm and another with R31 in Lund. R48 is configured with "no auto-summary", "no synchronization", and using 192.168.49.1 as its router-id. It is also configured to advertise only an aggregate address 192.168.48.0/20 to all BGP peers with "summary-only" (longer-prefix routes are suppressed and NOT advertised). R48 also redistribute all BGP learned routes into its internal network. In addition, R48 has a BGP policy to set LOCAL\_PREF to 50 on all routes learned via R48-R1 link.

Assuming that the existing iBGP configurations on R1, R21, and R31 as described earlier are still in place.

Then, R1 and R31 each adds the following configurations:

- eBGP peering with R48
- "next-hop-self" command for all iBGP peering

In addition, R31 add a route-map configuration (**set as-path prepend 65001 65001**) to prepend AS path two times on all routes it advertises to R48.

Assume that the topology has converged. Answer the following questions:

- H. What path does a packet traverse when a host in Uppsala with IP address 192.168.22.5 sends an ICMP echo request to a server in Oslo with IP address 192.168.49.5? (1p)
- I. What path does a packet traverse when a server in Oslo with IP address 192.168.49.5 sends an ICMP echo request to a server in Uppsala with IP address 192.168.21.5? (1p)
- J. What path does a packet traverse when a server in Oslo with IP address 192.168.49.5 sends an ICMP echo request to a server in Lund with IP address 192.168.33.5? (1p]

**IMPORTANT:** if the ICMP echo request cannot reach the destination then identify what happens to the request. See example answer above.

**Answer:**

- A. 192.168.2.11 -> 192.168.2.1 -> 192.168.0.41 -> 192.168.0.13 -> 192.168.0.17 -> 192.168.0.58 -> 192.168.16.6 -> 192.168.16.10 -> 192.168.21.5
- B. 192.168.22.11 -> 192.168.22.1 -> 192.168.16.21 -> 192.168.16.1 -> 192.168.16.34 -> 192.168.32.2 -> 192.168.32.14 -> 192.168.33.5

C. 192.168.34.11 -> 192.168.34.1 -> 192.168.32.9 -> 192.168.32.5 ->  
192.168.16.33 -> 192.168.0.57 -> 192.168.0.10 -> 192.168.0.34 ->  
192.168.0.22 -> 192.168.1.5

D. 192.168.21.5 -> 192.168.21.3 -> 192.168.16.17 -> 192.168.16.1 ->  
192.168.0.57 -> 192.168.0.62 -> 192.168.32.6 -> 192.168.32.10 ->  
192.168.34.5

E. 192.168.33.5 -> 192.168.33.3 -> 192.168.32.17 -> 192.168.32.5 ->  
192.168.0.61 -> 192.168.0.58 -> 192.168.16.2 -> 192.168.16.22 ->  
192.168.22.5

F. 192.168.1.9 -> 192.168.1.3 -> 192.168.0.30 -> 192.168.0.41 ->  
192.168.0.13 -> 192.168.0.17 -> 192.168.0.58 -> 192.168.16.2 ->  
192.168.16.22 -> 192.168.22.10

G. 192.168.1.9 -> 192.168.1.3 -> 192.168.0.30 -> 192.168.0.41 ->  
192.168.0.13 -> 192.168.0.17 -> 192.168.0.62 -> 192.168.32.6 ->  
192.168.32.10 -> 192.168.34.10

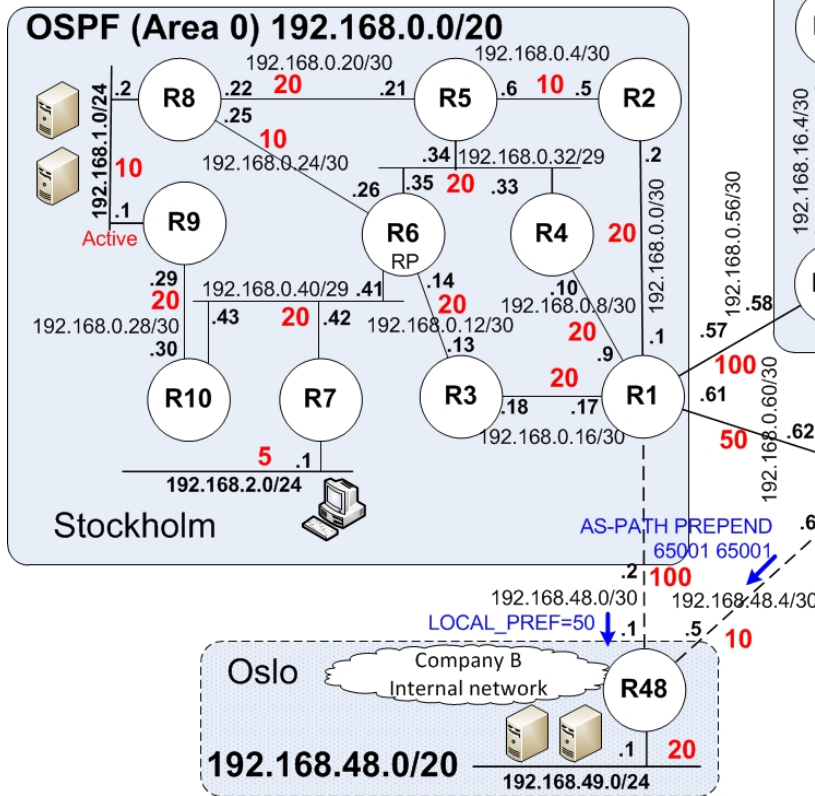
H. 192.168.22.5 -> 192.168.22.1 -> 192.168.16.21 -> 192.168.16.1 ->  
192.168.0.57 -> 192.168.48.1 -> 192.168.49.5

I. 192.168.49.5 -> 192.168.49.1 -> 192.168.48.2 -> 192.168.0.58 ->  
192.168.16.2 -> 192.168.16.18 -> 192.168.21.5

J. 192.168.49.5 -> 192.168.49.1 -> 192.168.48.6 -> 192.168.32.6 ->  
192.168.32.18 -> 192.168.33.5

You may remove this page from the exam and use it as your own note page

Link subnets and router interface IP addresses are shown.  
The router ID is the highest IP address of physical interfaces of the router.  
The red number represents bandwidth in Mbps unit.  
For example, **10** is 10 Mbps.



**NOTE:** If there is an inconsistency between the figure and the written text, the figure gets priority by default. (When in doubt, follows the figure!)

