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Contact during exam

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**TTM4150 INTERNET NETWORK ARCHITECTURE**

**August 9, 2007**
**0900 - 1300**

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No remedies.

Results will be ready in 4 weeks.

**Glance over all pages before you start answering the exercises.**
**Take care to share your time between the exercises.**
**It is better to answer a little on all the exercises than to answer a lot on a few.**

**If you feel there is a lack of information to solve an exercise, state the assumptions you make.**

## Exercise 1 Internet network architecture

**(a)** Justify why the IP layer is considered not trustworthy. Focus on the properties of unicast IPv4.

*The source address is not authenticated. The content is not protected against modifications, or eavesdropping. The delivery is not guaranteed. For full score the source address property and one or more of the other factors must be included.*

**(b)** Describe how the lack of trust at the IP layer may affect the deployment of multicast. Include a description of the factors that have the largest impact.

*As the source address cannot be authenticated, additional protocols must be used in order to avoid hijacking of a streaming group, falsification of content. Full score must include hijacking*

**(c)** Describe the end-to-end (e2e) argument and how it has been used for placement of functionality in the Internet routers, servers and end systems. Justify whether the multicast routing protocol PIM-SM is in accordance with the e2e principle for functionality placement.

*E2e: The function in question can completely and correctly be implemented only with the knowledge and help of the applications standing at the endpoints of the communication system. Therefore providing that function as a feature of the communication system itself is not possible. The result is that functiona are placed in the end system if feasible. Multicast requires functionality in the routers. Correct to answere both not inaccordance (typical understanding with function placement in the end system) or in accordance ( e2e argument states that place the functionality where it fully can be implemented) Full score depends on the argument of the answer and indefication of multicazst functionality is in routers*

**(d)** What does implementation of QoS (quality of service) in the network add to the original best-effort internet service model?

*"Performance assurance": To make internet more predictable, guaranteed packet delivery, an attempt to define characteristics of specific services. "Service differentiation": To let some users/flow/classes get better service than other*

## Exercise 2 Quality of service

**(a)** What are the four different QoS parameters?

*Bandwidth/throughput [bit/s]: (Max, Average, Min)*

*Delay [s]: (One-way, Round-tri, Variation in delay (jitter))*

*Error rate [%]: Packet loss – arise from congestion, Packet error, Bit error*

*Availability [%]: Internet connection reliability*

**(b)** Comment on the statement "End systems cannot always correct for late arriving packets as they correct for lost packets" with respect to functions performed at the receiving side.

*Buffering at receiver outweights delay variations (jitter). Interactive real-time applications tolerate less buffering/delay than one-way streaming applications, which again tolereate less delay than eg file transfers.*

**(c)** Explain how one of the basic building blocks/mechanisms to provide network quality of service makes use of the Internet Protocol field "Type of service".

*<u>Packet classification</u> associates each packet with a corresponding reservation/traffic class for the packet to be treated correctly*

**(d)** One-way delay of 150 ms end-to-end (from mouth to ear) ensures user satisfaction for telephony applications. What are the various components of delay (network and service) that a VoIP (Voice over IP) system design needs to take into account considering the end-to-end delay?

*Network delay: propagation delay through the backbone, queuing/scheduling delay, and access link serialization delay, and service delay: VoIP gateway codec and dejitter buffer.*

## Exercise 3 Multicast

An Internet service provider uses IP multicast for TV distribution of the three TV channels: NRK1 (market share 20%), NRK 2 (market share 5%) and TV2 (market share 20%). Market share is defined as the percentage of users viewing the channel at any time. We use the simplified assumption that the number of viewers are constant through out the day. A single TV server is connected to an ethernet over a fiber ring with 4 ADSL multiplexing points (ADSL mux). The fiber ring is unidirectional, and the packets inserted by the TV server circulate only to the last ADSL mux requiring the packet. Each ADSL mux serves 2000 customers. The connection between a user and the ADSL multiplexer point is ethernet over twisted pair. Each ADSL mux is a fully functional IP router.

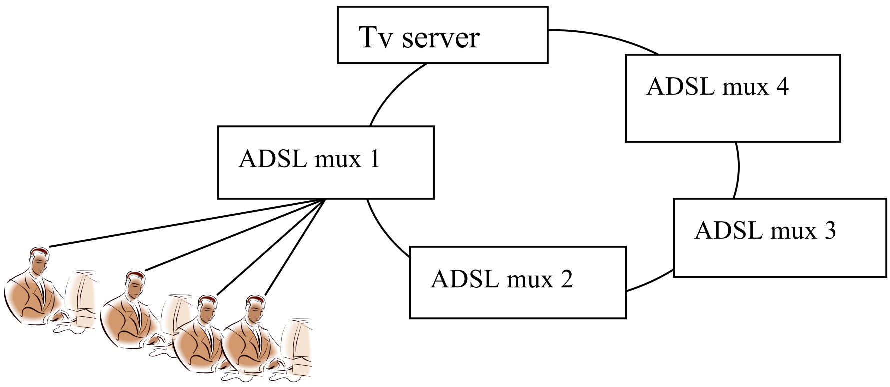


Figure 1 TV distribution architecture

**(a)** Which differences are there between shared and source specific multicast distribution trees? Which tree is the most suited for the system described above? Discuss both bandwidth efficiency and overhead.

*In a shared tree all sources use the same distribution tree. In a source specific tree each source has its own tree. The result is fewr hops between a source and a receiver for SST, a better spread over links, and larger overhead, since the signaling must be to each source. With the topology given the two tree types will have the same topology and there will be no difference. Each tv channel will be a separate multicast address. The argument is independent of whether a multicast ehternet is used between the tv server and the muxes or IP transmissions. Any assumption is valid. The number of packets will be the same since a fiber ring and unidirectional. The important assumption is that all muxes will always be interested in all multicast addresses 1-P(0, bin ( 2000,0,05)) If asked, each mux is an IP multicast router, but it is given with the statement fully functional IP router.*

**(b)** The internet group management protocol (IGMP) is used between the user and the TV server. If we consider only the subsystem consisting of the fiber ring, and the TV distribution server and the 4 ADSL mux, is there a difference in efficiency between using IGMP v1 and IGMP v2? Justify your answer.

*IGMP v1 times out on leave, while v2 has explicit signalling A tv channel will therefore continue to send after there is no interest. Over the adsl wire it is important* *to avoid, since there is only one user downstream. On the fiber ring it is of no importance since the probability that the mux that other users conentcted to the mux is interested is for all practical purposes = 1*

**(c)** If we consider the whole system, i.e. also include the communication between the users and the ADSL mux, is there a difference in efficiency between IGMPv1 and IGMPv2. Justify your answer.

*See above*

**(d)** In PIM-SM why must a source initially tunnel packets to the RP (Rendezvous point)?

*In PIM-SM the distribution is downwards in the shared tree. For a packet to be distributed, it must be inserted at the top, the RP. Since the source will not initially have a distribution tree, the packet must be routed to the RP. This can only be done within a tunnel, since the multicast address is not routable by a router not part of the distribution tree. A source need not be part of the distribution tree. Eventually, the RP will build a source specific distribution tree towards the source.*

**(e)** Are unicast and multicast packets treated the same in the forwarding plane of a router? Describe common functionality and differences based on multicast forwarding under the PIM-SM.

*In pim-sm there is sahred tree, source specific trees in addition source can send without previous signaling. Fowarding of a multicast packets must therefore check whether any of these condations apply and if there is a corresponding routing entry. Different routing tables there is also a reversed path forwarding check to only forward packets arriving on the interface pointing back to the source. For unicast the decision is simple, if route forward else drop. No RPF check. Commonality would be the QoS processing with AQM and scheduling. The latter is not needed for full score.*

## Exercise 4 Mobility and Ad-hoc networks

**(a)** Mobile IP uses two tiered addressing. Describe what is meant by two tiered addressing, and describe the two alternative mechanisms that could be used to implement it in mobile IP.

- *Ip address has two functions: 1) Mobile node point of attachment, Used as a routing directive 2) Component Mobile node end-point identifier, Remains static for the lifetime of a mobile node*
- *In Mobile IP implemented either through IP tunneling or Loose source routing*

**(b)** Describe the purpose of the Host Identifier protocol (HIP). Will two tiered addressing be required at the IP layer to handle mobility if HIP is used?

*Add a host identifier defining an endpoint Fixed size, low prob of collision Authentication, Dynamic binding of Host identifier and location, Protection against DOS attacks. With an end point identifier, the IP address can be used for routing and two tiered addressing not required. However, there must be a dynamic binding between Host identity and IP address. This is yet to be defiend*

**(c)** In mobile IP, describe the mechanism in route optimization to handle the lack of trust at the IP layer.

*In route optimization, the corresponding node is informed of the care address, so the corresponding node can tunnel traffic directly to the mobile node or the FA. The HA normally does the updating of the binding to the corresponding node. The mechanism includes security mechanisms to ensure that hostile nodes cannot hijack an address. Every aspect of the design is influenced by the need to allow the correspondent nodes to be sure of the authenticity of the updates. Binding updates are therefore accompanied by a binding update acknowledgment.*

**(d)** In the proactive ad-hoc routing protocol, OLSR, the concept of Multipoint relay node (MPR) is used. What is the function of an MPR node, and what are the benefits?

*The MPR is used for flooding the protocol control traffic except Hello, i.e limit the bradcasting. All packets on a path with more than 2 hops are routed through a MRP. The connections between the MPR are treated as the backbone of the network. Only links between a node and the MRPS are announced. Thereby limiting the number of links reported. The benefit, limit number of broadcasts of the same packet, limit the number of links announced, saving size of overhead packets and therefore resources*

**(e)** Compare the mechanisms used to avoid stale routes (old routes that no longer are valid) in the proactive routing protocol OLSR and the reactive protocol AODV.

*AODV use a seqno for each route A req for a route contains a seq no higher than the previous to know seq no ensuring that intermediate nodes with old knowledge (i.e lowers seq no) do not report the route.*

*In the proactive OSLR, routes are reported periodically, ensuring a limited lifetime of stale routes.*

## Exercise 5 Transport protocols

**(a)** Explain why TCP sliding window is more complex than a link layer sliding window.

*Potentially (4 gives full score)*

- *Interconnection among several hosts across several network nodes*
- *Requires explicit establishment and termination of connection*
- *varying RTT: need adaptive timeout mechanism*
- *longer delay in network: old datagrams may arrive*
- *different receiver capacity: varying buffer requirements*
- *different network capacities: may be prepared for network congestion*

**(b)** Describe two options/extensions to TCP that have an affect on measured end-to-end performance.

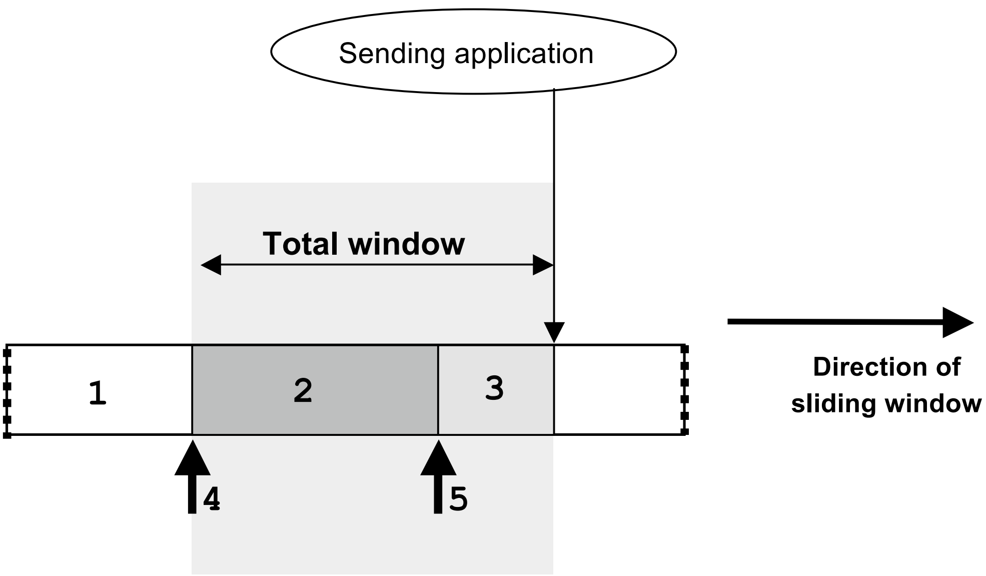


Figure 2 TCP sliding window

- *Larger buffers with Window scale" option (shift-operation on advertised window)*
- *Selective acknowledgment" option: Avoid retransmission of data that actually reached the receiver, sender can learn about several lost packets per RTT*
- *Consider "delayed ACK": Increase slow start congestion window with 2 MSS*
- *Larger initial window size: Small transfers can finish within 1 RTT*
- *Limited transmit: Extension of fast "retransmit/fast recovery" when do not receive duplicate acks*

- *Explicit congestion notification (ECN): Faster respons to congestion by explicitly notification without having to invoke packet retransmission, eg don't do congestion control if packet loss due to bit error.*

Figure 2 illustrates part of a sending TCP byte stream.

**(c)** Describe the byte areas at the sender marked with 1, 2, and 3 and describe what makes the state variables indicated by 4 and 5 change value.

*1: Sent and ack received*
*2: Sent but no ack received*
*3: Written from application to sender buffer, but not sent*
*4: Receiving ack ( **LastByteAcked**)*
*5: send() – ie. When a new segment is sent (**LastByteSent**)*

The transport protocol DCCP (datagram congestion control protocol) has an underlying motivation of avoiding congestion collapse.

**(d)** Explain what is meant by "congestion collapse".

*Congestion collapse; everything works fine, but the <u>fraction of data that makes it to its final destination becomes small</u>, so the effective throughput of the network is small: <u>Internet meltdown</u>: Congestion collaps due to unnecessary restransmissions and due to undelivered packets.*

**(e)** Briefly describe the service model of the transport-layer protocol DCCP.

*DCCP is a minimal general purpose transport-layer protocol providing a <u>congestion-controlled unreliable service</u>:*

- *the establishment, maintenance and teardown of an unreliable packet flow*
- *congestion control of that packet flow*

*DCCP is intended for applications that currently use UDP: streaming media and Internet telephony, where the costs of long delays can exceed the costs of loss and out-of-order delivery.*