SC250 Computer Networking 1: Quiz

Prof. Matthias Grossglauser

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1. Every host implements the transport layer.

   Answer: True. In fact, every host implements the complete protocol stack.

2. Every router implements the transport layer.

   Answer: False. Routers are part of the network core, and only implement the network layer and below.

3. Every router implements the network layer.

   Answer: True.

4. It is possible that a router implements several types of link layers.

   Answer: True. In fact, this was a major design goal for the IP protocol, i.e., to have a single network layer protocol that can run over a wide variety of link layer technologies, including high-speed optical fiber, dial-up modems, and wireless and satellite connections.

5. It is the network layer's responsibility to forward packets reliably from the source to the destination.

   Answer: False. The network layer only offers a "best-effort" service, i.e., it will attempt to deliver the packet if network conditions permit. However, the network will drop packets if, for example, a buffer overflow occurs somewhere in a router, and the network layer will not try to recover this packet. The transport layer, specifically the TCP protocol, running on top of the network layer will ensure reliable end-to-end delivery if such a service is required.

6. Suppose you access a web page from your computer. When you examine a packet resulting from this access on an Ethernet link, you find first the HTTP header, followed by the TCP header, followed by the IP header, followed by the Ethernet header.

   Answer: False. Every layer encapsulates the PDU it receives from the layer above. Therefore, on the wire one observe first the lowermost header (Ethernet header), then IP, then TCP, and finally HTTP.

7. The advantage of packet-switching over circuit-switching is more pronounced for bursty traffic (the rate fluctuates a lot) than for smooth traffic (the rate fluctuates little or not at all).

   Answer: True. Statistical multiplexing in a packet-switched network exploits a statistical averaging effect, in the following way. The total traffic generated by a number of users exhibits less fluctuation *relative to the mean* than an individual user's traffic. If an individual user's traffic is not bursty, i.e., the rate does not fluctuate, then there is no benefit to packet switching over circuit switching. For more details, see the answer to the next question.

8. Assume that a set of users share a link, i.e., they all send traffic over the same link using packet switching. The traffic of the users is independent and follows the same probability law.

   Let N denote the number of users, R the capacity (bandwidth) of the link. We are interested in the probability p_e that the N users overload the link.

   Now suppose we multiply the number of users by 10, and we also multiply the link capacity by 10, i.e., $N' = 10N, R' = 10R$. Then $p'_e \leq p_e$, i.e., the faster link is less likely to be overloaded.

   Answer: True. We can illustrate this and the former question through the Central Limit Theorem (CLT). Let the random variable X_i denote the traffic rate generated by user i, where we assume that the X_i are independent of each other. The total traffic over the link is then $\sum_i^N X_i$. Suppose we let N grow large, and we let the link capacity scale as NR. We are interested in the overflow probability $p_e = \mathsf{P}\left\{\sum_i^N X_i > NR\right\}$.

The CLT says that

$$\frac{\sum_i^N X_i - N\mu}{\sigma\sqrt{N}} \rightarrow N(0,1), \tag{1}$$

i.e., for large N, the left-hand side has a probability distribution very similar to that of the right-hand side, where $\mu = \mathsf{E}[X_i]$, $\sigma^2 = \mathrm{Var}[X_i]$ and $N(0,1)$ is a standard normal random variable, i.e., a Gaussian with zero mean and unit variance. This means that for large N, a good approximation for p_e is

$$\begin{aligned} p_e &= \mathsf{P}\left\{\sum_i^N X_i > NR\right\} = \mathsf{P}\left\{\sum_i^N X_i - N\mu > N(R-\mu)\right\} \\ &= \mathsf{P}\left\{\frac{\sum_i^N X_i - N\mu}{\sigma\sqrt{N}} > \frac{N(R-\mu)}{\sigma\sqrt{N}}\right\} \approx Q\left(\frac{\sqrt{N}(R-\mu)}{\sigma}\right). \end{aligned} \tag{2}$$

Suppose we fix the capacity per user R and study how p_e changes with increasing N. It is clear from (2) that p_e decreases as N increases. This is the result of the averaging effect mentioned above. Also, a higher variance increases the overflow probability p_e.

Now suppose we fix the probability of overflow p_e and study how R changes with increasing N. Suppose X_i is such that it has some maximum x_{max}. With circuit switching, we have to allocate x_{max} to every user. With packet switching, it follows from (2) that we have to allocate $R = \mu + \frac{\sigma Q^{-1}(p_e)}{\sqrt{N}}$ for a desired overflow probability p_e. This does *not depend on* x_{max}, but only on (μ, σ^2). As N grows larger, the rate per user R is more and more close to the mean rate μ.

Bursty traffic tends to have a high ratio x_{max}/μ, while smooth traffic tends to have a ratio close or equal to one. Therefore, the more bursty the traffic, the larger the gap between the capacity required by circuit-switching (x_{max}), and the capacity required for packet-switching (slightly more than μ).

Note that these calculations are simply a way to illustrate how this "averaging effect" in statistical multiplexing helps. The three simple lessons to retain are: (1) for bursty traffic, the advantage of statistical multiplexing in packet switching over circuit switching is more pronounced than for less bursty traffic; (2) as the number of users N grows, this advantage is more pronounced; (3) as N grows large, the rate to be allocated per user approaches the mean rate per user, which is the best we can hope for.

9. The phone network uses circuit switching.

   Answer: True. The phone network allocates 64kbps for every call. If the two participants do not speak, then this capacity remains unused.

10. Assume we send packets of some fixed size through a network, and assume routes do not change. Then the only random component in total packet delay is the queuing delay.

    Answer: True. Propagation delay depends on the distance the packet travels, and transmission delay depends on the packet size and the speeds of the links is travels over. Queuing delay depends on other packets that happen to be present in buffers along the way, and is essentially unpredictable and random. Note that processing delay can also be considered random, but tends to be small compared to queuing delay.

11. Which of the four sources of delay (transmission, propagation, processing, queueing) depend on the packet size?

    Answer: Transmission delay.

12. In a datagram network (such as IP), routers keep track of connections between end systems.

    Answer: False. Routers only keep track of destinations, i.e., over what outgoing link each destination in the network is reachable. This is independent of how many connections exist from or to that destination. Connections only exist in the transport layer, and therefore in hosts.

13. In a virtual circuit network, routers allocate resources (link capacity, buffer space) to every circuit.

    Answer: False. Virtual circuits are only predetermined routes for a connection. This means that a packet that belongs to a particular connection always follows this same route, and the switches in the network keep track of these connections (contrary to datagram networks such as IP, see above). However, no resources (link capacity, buffer space) are allocated to the connection, and there is still a possibility of delaying or losing packets because of congestion.

14. In a circuit-switched network, switches allocate resources (link capacity, buffer space) to every circuit.

    Answer: True.

15. A router decides on a route for an IP packet based on its source and destination address.

    Answer: False. It is only based on the destination address. Note that this means that the packet will be delivered to the destination even if the source address is false. This is frequently used in denial-of-service attacks.

16. The TTL (time to live) field is decremented at every hop in the network in order to avoid that a packet can accidentally loop in the network. This TTL field is part of the TCP header.

    Answer: False. The TTL field could not possibly be part of the TCP header, as the TCP header cannot be modified inside the network (routers do not implement the transport layer). The TTL field is part of the IP header.

17. A cable (TV) access network is a shared medium.

    Answer: True.

18. ADSL (Asynchronous Digital Subscriber Line) is a shared medium.

    Answer: False.

19. HTML is a protocol to describe web content.

    Answer: False. HTML is not a protocol, just a standardized data format to describe web content. A protocol consists of two or more communicating entities, along with a definition of what messages they can exchange and what actions they should take in response to these messages.

20. The end-to-end argument that is key to the design of TCP/IP recommends placing functionality whenever possible at the network edge (hosts) rather than the network core (routers).

    Answer: True. An example of this is the fact that the transport layer only exists in end-systems. We could try to design a different network protocol stack where the network layer, for example, ensures reliable delivery (in fact, this was tried in the early days of the Internet). However, it turns out that this results in a very complex network layer and a less flexible architecture.

    The chief example of a network that follows a different design methodology is the phone network: the "end systems" (your phone) has very limited functionality, and it is the network that contains all the "intelligence".

21. When you download a web page, it results always in exactly one TCP connection.

    Answer: False. This is true only if the web page contains no embedded objects such as images, or if the download happens over HTTP/1.1 in persistent mode and all the objects reside on the same server.

22. When you download a web page containing images, the HTML page and the images come from the same server.

    Answer: False. They can very well come from different servers. In fact, there are companies, such as Akamai, that specialize in hosting images and other objects on behalf of other companies to reduce the load on their servers. This is known as "content distribution".

23. You issue the command `ifconfig` and observe the following output:

```
lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:126647 errors:0 dropped:0 overruns:0 frame:0
          TX packets:126647 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:24457818 (23.3 Mb)  TX bytes:24457818 (23.3 Mb)
```

    Your computer is currently connected to the network.

    Answer: False. The only network interface you see if the loopback interface, which is a virtual interface representing your own machine.

24. The network layer modifies the UDP header.

    Answer: False. UDP is a transport layer protocol; therefore, for the network layer, the UDP header is just a bunch of bits.

25. Tier-1 ISPs (Internet Service Provider) are customers of Tier-2 ISPs.

    Answer: False. It is the other way around: Tier-2 providers are customers of Tier-1 providers, Tier-3 of Tier-2, etc.

26. Tier-1 ISPs may exchange traffic (packets) both at private and public peering points.

    Answer: True. In addition to the public peering points, where many ISPs of different size (tiers) interconnect, ISPs at the same level may elect to make private arrangements to exchange traffic over dedicated links. They may do this, for example, to improve performance and to have better control over their connectivity to the wider Internet. Note in general that the way ISPs interconnect is as much a business issue as a technical issue.

27. You send a large message through a network. You have the option of sending the message as a unit, or to break it up into much smaller packets. In the second option, which components of end-to-end delay are reduced?

    Answer: Transmission delay. Obviously the propagation delay does not change. The queuing delay does not change (on average), as it depends on the other traffic in the network. The total transmission delay is reduced thanks to pipelining.

28. You send two pings from your machine A to another machine B on an otherwise empty network. You note that a ping with a 40byte packet results in a RTT of 15.7ms, with a 1500byte packets the RTT is 224.3ms. What is the link speed (in kbps) of the bottleneck link?

    Answer: We have $d(L) = d_p + d_t = d_p + L/R$, where d_p is the propagation delay, d_t the transmission delay, L the packet size, and R the link speed. Given $d(L_1 = 40\text{bytes}) = 15.7\text{ms}$ and $d(L_2 = 1500\text{bytes}) = 15.7\text{ms}$, then

    $$R = \frac{L_2 - L_1}{d(L_2) - d(L_1)} = 56\text{kbps}. \tag{3}$$

    Note: we have assume dhere that the return packet is of negligible size, and/or the return link speed is very high.

29. In the previous question, what is the (round-trip) propagation delay?

    Answer:

    $$d_p = d(L_1) - L_1/R = 10\text{ms}. \tag{4}$$

30. What transport protocol should an elastic application with tight delay requirements use?

    Answer: It should use UDP. TCP incurs more delay than UDP, because TCP has to ensure reliable, in-order delivery to the application. This requires exchanges of control messages (such as ACK/NAK), retransmissions, etc. For an application that can tolerate some packet loss and possibly packet reordering, UDP may be a better option.

31. Suppose you access a very small web page using HTTP from your browser. The lowest delay you can hope for between the moment when your browser issues the request until the page is on your screen is how many times the RTT?

    Answer: The minimum is 2RTT. The first RTT is incurred when the client sends a packet requesting a TCP connection, and the server's response back to the client. The second RTT is incurred when the client sends the HTTP GET request to the server, and the server's response containing the web page.

32. SMTP possesses a command called "SUBJ" to allow the client to send the message subject string to the server.

    Answer: False. The subject line is simply part of the message body, and as such not part of the SMTP protocol header.

33. When you send an email message containing attachments (such as images), a separate TCP connection is opened for every object.

    Answer: False. Such objects will be part of the same email message. It is the responsibility of the MIME encapsulation to "pack" these objects into the email message and to "unpack" them at the recipient.

34. It is possible for a finite state machine (FSM) to possess, for a state X, multiple transitions $X \rightarrow X$.

    Answer: True. This makes perfect sense, if these transitions have different activation conditions.

35. RSA is a public key cryptographic algorithm. In case one loses the private key, it is possible to reconstruct it from the public key.

    Answer: False. This would defeat the very purpose of cryptography, as anyone could then recover the private key (a contradiction!), and decrypt the message.

36. With public key cryptography, Alice uses her private key to ensure the confidentiality of messages sent to Bob.

    Answer: False. Alice (or anyone else sending confidential messages to Bob) will use Bob's public key to encrypt the message. Bob will then be the only one who can decrypt the message, as only he possesses his private key.

37. Alice uses her private key to authenticate herself to Bob.

    Answer: True. Alice will crypt a random number sent to her by Bob using her private key. Bob, who must already possess a copy of Alice's public key, can then verify that the person pretending to be Alice must in fact be Alice. Note that ensuring that Bob possesses Alice's real public key is a nontrivial problem known as the key distribution problem.

38. ssh establishes symmetric per-session keys because symmetric key encryption/decryption is faster than public key encryption/decryption.

    Answer: True. It would conceptually be possible to rely entirely on public key crypto, but symmetric crypto is faster.

39. When you use ssh to connect to a server (sshd), both the server and the client authenticate themselves to each other before the connection is established.

    Answer: True. You have seen this in the TP. On the client side, there is a file `known_hosts` containing the public keys of known servers. On the server side, there is a file `authorized_keys` containing the public keys of known clients. ssh will connect to a server only after authentication has happened in both directions.

40. One advantage of the Napster peer-to-peer network is that it requires less query traffic than the Gnutella network.

    Answer: True. In a decentralized architecture such as Gnutella, there is no directory where one or several nodes record on which node each file available in the network is stored. Therefore, for every request for a file, a query has to be flooded through the entire overlay network (or a reasonable portion of the network). As the network becomes large, this overhead becomes prohibitive.