Name______________ Student ID_____________________ Department/Year_________

Final Examination

Introduction to Computer Networks
Class#: 901 31110
Fall 2003

9:20-11:00 Tuesday
January 13, 2004

Prohibited

1. You are not allowed to write down the answers using pencils. Use only black- or blue-inked pens.
2. You are not allowed to read books or any references not on the question sheets.
3. You are not allowed to use calculators or electronic devices in any form.
4. You are not allowed to use extra sheets of papers.
5. You are not allowed to have any oral, visual, gesture exchange about the exam questions or answers during the exam.

Cautions

1. Check if you get 13 pages (including this title page), 6 questions.
2. Write your name, student ID, and department/year down on top of every page.
3. You have 100 minutes to answer the questions. Skim through all questions and start from the questions you feel more confident with.
4. Use only English to answer the questions. Misspelling and grammar errors will be tolerated, but you want to make sure with those errors your answers will still make sense.
5. If you have any extra-exam emergency or problem regarding the exam questions, raise your hand quietly. The exam administrator will approach you and deal with the problem.

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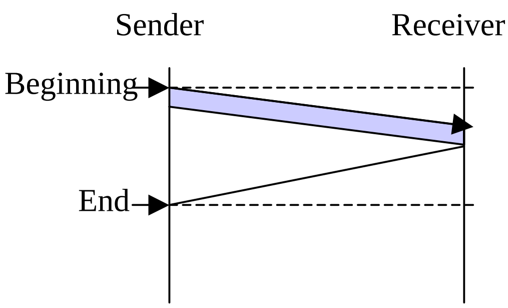
1. (Reliable Data Transfer) We come to derive rdt 3.0 that reliably transfers data for channels with bit errors and losses. It works but the performance sucks. Assume *L* is the packet size in bits, *R* is the transmission rate in bits per second, and *RTT* is the round-trip propagation delay. The transmission diagram below shows the operation of rdt 3.0 when there are no bit errors and losses. Due to the nature that it stops the data packet transmission and waits until receiving the acknowledgement packet to send another, rdt 3.0 is also referred to as the stop-and-wait protocol.



(1) What is the sender utilization (fraction of time the sender is busy) of the stop-and-wait protocol? (5%)
(2) What is the sender utilization if 3 data packets are pipelined within one round trip of transmission? (5%)
(3) Pipelining seems to help the sender utilization but additional extensions to the stop-and-wait rdt are necessary to support pipelining. What might the required extensions be? (5%)
(4) Name the two generic pipelined protocols discussed in the lectures? (5%)

Answer:
(1) (L/R) / {RTT+(L/R)}
(2) (3L/R)/{RTT+(L/R)}
(3) Larger sequence number range and buffering space at the senders and receivers
(4) Go-back-N and selective repeat

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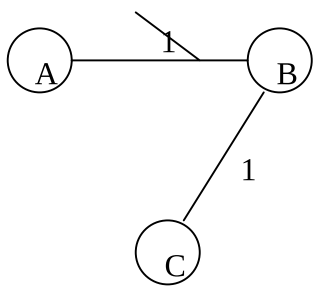
2. (Routing)
   (1) Describe how Distance Vector routing works in principle. Name one example of DV routing protocols. (5%)
   (2) Describe the 'Count To Infinity' problem in DV routing. (Hint: easier by an example) (5%)
   (3) State the main difference between Path Vector and Distance Vector routing. Name one example of PV routing protocols. (5%)
   (4) Would the 'Count To Infinity' problem exist is PV routing? (5%)

Answer:

(1) Each node on the network keeps a vector of best (next hop, distances) to every other node. Whenever a route report is received, the node updates the distance vector if the route report provides a better route to a particular destination via the neighbor from which the report is received. If this results in changes in the route (next hop or distance) to that destination, a route report is sent which might in turn change the distance vector of the node's neighbors. In principle, each node will tell the neighbors the best information it's got. RIP is a DV routing protocol

(2) Consider the scenario below. A goes to B through link A-B, to C through A-B-C. B goes to A through link A-B, to C through link B-C. C goes to A through C-B-A, to B through link B-C. Suddenly, link A-B breaks down.



1. In B, the distance to go to A via A is set to infinity. Therefore, B decides going via C to A is a better route (distance of 3, B-C-B-A). B reports to C that its route to A is now via C with distance 3.
2. C updates the distance to A via B to 4. C reports to B that its route to A is still via B

but with distance 4.

3. B updates the distance to A via C to 5 and reports to C that its route to A is still via C but with distance 5.
4. C updates the distance to A via B to 6 and reports to B that its route to A is via B with new distance 6.
5. The process continues until B updates the distance to A via C to infinity+1 and reports to C that its route to A is now via A with distance infinity.
6. C updates the distance to A via B to infinity+1 and reports to B that its route to A is with distance infinity+1
7. B updates the distance to A via C to infinity+2 and the routing tables finally converge.

This phenomenon that the network needs to wait until the routes are counted to infinity before the routing tables stablize is referred to as the 'Count to Infinity' problem. In the process of the routes counting to infinity, there could be a substantial amount of data looping in between without realizing that the destination is no longer reachable.

(3) Path Vector routing protocols propagate not only the distance, but also the entire path. BGP is a PV routing protocol.

(4) No

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3. (MAC)
   (1) How does CSMA/CD work in principle? (5%)
   (2) Can frames collide in CSMA and how? What is the problem in CSMA that CSMA/CD is trying to resolve? (5%)
   (3) How does CSMA/CA work in principle? (5%)
   (4) How can collisions be detected? What is the problem in CSMA/CD that CSMA/CA is trying to resolve? (5%)

Answer:

(1) Listen before transmit. Send when the channel is sensed idle. Hold when the channel is sensed busy. Abort when collisions are detected. Re-send after a random exponential backoff.

(2) Yes. Multiple CSMA transmissions might start about the same time when the channel is sensed idle. They could collide during the propagation delay. In CSMA, the entire frame transmission time will be wasted as the collision occurs. CSMA/CD tries to stop the frame transmission as soon as the collision is detected so to reduce the channel wastage.

(3) The sender sends a Ready-To-Send (RTS) message to indicate duration of transmission. In reply, the receiver sends a Clear-To-Send (CTS) message to notify reachable (possibly hidden) nodes. Those nodes receiving RTS and CTS but not involved in the transmission will not transmit for the specific interval of Network Allocation Vector (NAV) time. In the meantime, data and acknowledgement are exchanged between the sending and receiving nodes.

(4) Collision can be detected in the wired LAN by measuring signal strength or comparing the sending and receiving signals. It is however not the case for wireless LAN due to the hidden terminal problem, in which transmissions from certain nodes might not be visible by other nodes on the same wireless LAN. CSMA/CA avoids the potential collisions due to the hidden terminal problem by the sending of CTS and RTS frames which in a sense alerts all the visible nodes from the data sender and receiver of the data-ack exchange coming up next.

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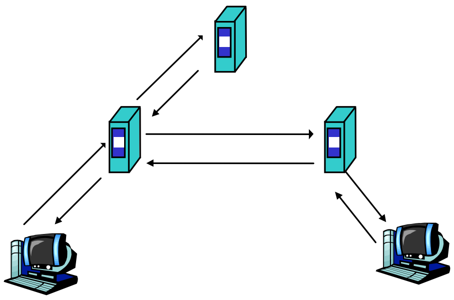
4. (Streaming Media)
   (1) What is delay jitter? Name 2 applications that are sensitive to delay jitter. (5%)
   (2) What can the streaming media client do to compensate for the delay jitter? What is playout delay? (5%)
   (3) What is delay loss? What is the impact in terms of delay loss when the playout delay is set too short? (5%)
   (4) Describe an adaptive playout delay mechanism that minimizes both the playout delay and delay loss rate. (5%)

Answer:

(1) Delay jitter is the variability of packet delays within the same packet stream. Internet phone, TV broadcast, online movie, and online shooting games are all sensitive to delay jitter.
(2) The client usually buffers the initial streaming data and starts the constant-rate replay after a certain period of time. The time period between receiving the first data bit to the playout of the bit is referred to as the playout delay.
(3) Certain data packets may arrive too late for playout at the receiver. If the playout delay is set too early, there tend to be more packets missing the deadline and resulting in more delay losses.
(4) Let the playout delay be the estimated average network delay with deviation. Each data packet is time stamped as the packet is sent. The network delay is computed at the receiver as the difference between the receiving time and the timestamp. A new estimation of average network delay is obtained by taking the weighted average of the previous average network delay estimation and the network delay of the arriving packet. The network delay deviation is computed as the difference between the estimated average network delay and the network delay of the arriving packet. A new estimation of average network delay deviation is obtained by taking the weighted average of the previous average network delay deviation estimate and the network delay deviation of the arriving packet.

5. (DNS and SIP)
   (1) Illustrate how the client, local name server, root name server, intermediate name server and authoritative name server interact in DNS iterated queries. Hint: use the reference figure below, name the computers, and number the message exchange accordingly. (6%)
   (2) Similarly, illustrate how the caller client, SIP proxy, SIP home registrar, SIP foreign registrar, and callee client interact in SIP name translation and user location service. (6%)
   (3) Compare and contrast the DNS look up and the SIP name translation and user location services. (8%)



Answer:
   (1) Please refer to the DNS iterated query description in Chapter 2.
   (2) Please refer to the SIP user location description in Chapter 6,
   (3) They are similar at the higher level that 1) they are both name translation services and 2) the query re-direction operation are very alike. SIP proxy's role is similar to the local name server in that it directs the query to the server that is capable of answering the query. The home registrar's role is similar to root name server in that it re-directs the query to the server that knows the more reliable and updated answer for the query. They are different in detail that 1) DNS translates hostname to IP address, as opposed to SIP translates username to the IP of the user's physical location and 2) there is no need of the SIP root proxies like the root name servers. The SIP proxy takes full advantages of the DNS service and directs the query to the SIP registrar at the home domain of the callee. Therefore, SIP doesn't have the server load problem as in the root name servers.

6. (Bonus) You are starting a company aiming to provide teleconferencing service over the Internet. With the service, multiple users can talk to and see each other interactively. Try if you could follow the steps below and come up with a suitable and feasible protocol stack:
   (1) List your application requirements and argue for them. (5%)
   (2) With the application requirement in mind, what transport layer service would you use and why (Hint: think the protocols supporting real-time data transfer)? (5%)
   (3) With the transport layer service of your pick, what functionalities you think are necessary to add at the application layer and why (Hint: think the Internet phone case)? (5%)
   (4) In addition to addressing, unicast routing, and forwarding, what additional network layer services do you see necessary and why (Hint: think for providing for group communication and QoS guarantee)? (5%)

Answer:
No expected solution for this one. You argue for your own claim. As long as the argument makes sense, you get the points. Here's just an example of what one might think towards a successful teleconferencing application start-up.

(1) The application requirement should at least include delay jitter, delay, and bandwidth. Teleconferencing is essentially a service of delivering interactive, synchronized audio/video streams. All audio/video streaming applications are sensitive to delay jitter. Changing in audio/video playout speed hinders user's perception of the visual gesture and speech. Interactive applications are sensitive to end-to-end delay. Long end-to-end delay makes it hard to detect the end of a user's paragraph and therefore hard to react and keep a fluent back-and-forth interaction going. Bandwidth is also required in that Mbps bandwidth per video stream is about the minimum to give perceptible quality with the current state of the art compression technique. In short, the network protocol stack should be gearing towards these delay jitter, delay, and bandwidth needs. Reliability isn't required because this is interactive A/V. People would do the request for retransmission if any speech or gesture being not clear.

(2) Due to the delay requirement, TCP is quickly out of the picture. Since there will be audio and video streams transmitted at the same time and synchronization of the streams is essential to provide a coherent teleconference session. Therefore, the sequence number and clock time are necessary information to be included in the data. RTP stands out comparing to UDP

(3) RTP, however, doesn't provide specific jitter compensation functionality. The options are to do client buffering and to play out the A/V streams adaptively to the network condition. This achieves in the meantime short playout delay and low delay loss rate. Error recovery at the application layer isn't such an urgent issue for the same reason stated for reliability isn't critical for interactive real-time application.

(4) For group video delivery, multicast routing could avoid sending duplicate data repeatedly over the shared path from the source to multiple receivers. From the point of view of efficient use of the network resource and in turn higher transmission quality overall, multicast is an important building block at the network layer. Mechanisms such as IntServ and DiffServ are however too big a deployment challenge even though they promise to provide guaranteed or controlled QoS. Looking at solving the problem from the practical point of view, proper provisioning and planning of the network bandwidth stands out given that labor cost is much higher than the high-speed switches and optical fibers. Finally, the application can only be commercially interesting when the pricing scheme is simple and fair enough for every ISP to understand and agree. And the charging can only work on a robust and well-established and well-connected accounting infrastructure.