CS 438 MT1 Review

ACM @ UIUC

September 26, 2024

Disclaimers and Logistics

- **Disclaimer:** We have not seen the exam. We have no idea what the questions are. However, we've taken the course and reviewed material/practice exams, so we have **suspicions** as to what the questions will be like.
- This review session is being recorded. Recordings and slides will be distributed on Piazza and the ACM site at the end.
- **Agenda:** We'll review all topics likely to be covered, working through some examples that may look like exam questions as we go, then review individual topics by request.
  - Questions are designed to be written in the same style as previous exams but to be *slightly* harder, so don't worry if you don't get everything right away!
- Please let us know if we're going too fast/slow, not speaking loud enough/speaking too loud, etc.
- If you have a question anytime during the review session, please ask! Someone else almost surely has a similar question.
- We'll provide a feedback form at the end of the session.

Foundations I: Resource Sharing

- A **network** is just a set of **elements** (servers, routers, etc.) connected together, that implements a set of **protocols** for the purpose of **sharing resources** at the end hosts
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  - **Packets/Datagrams**: Packets contain data (body) + information on how/where to send it and where it came from (headers). No underutilization/blocked connections/setup costs and can route around link failures, but no guarantees on availiability/delay, and overhead from headers. Used basically everywhere.

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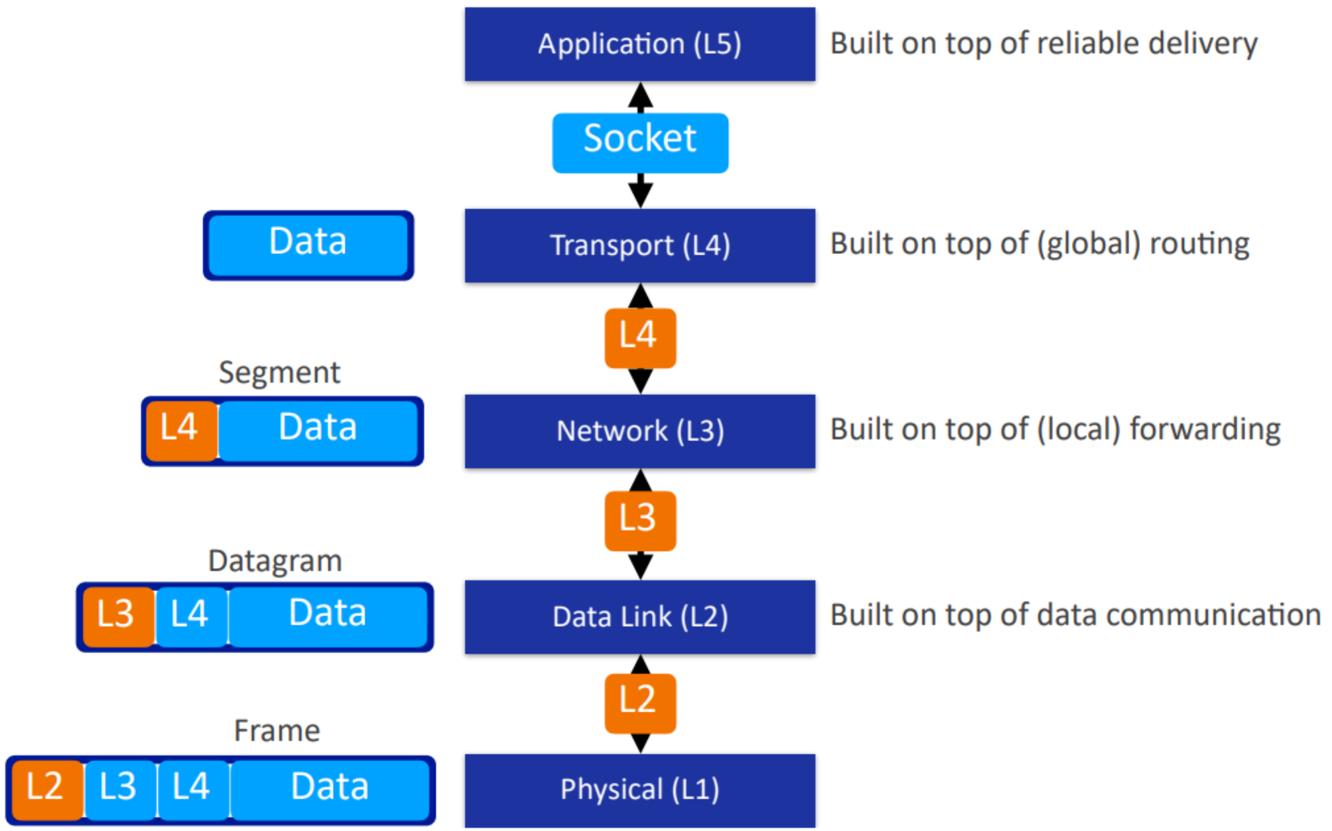
- When packet arrives at destination, forward to correct application

- Goal: Nodes shouldn't have to worry about the implementation details of other nodes, just the correct decision locally (modularity)

Foundations III: Layering



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  - **Queuing Delay**: Time that a packet waits in queue because link is busy. In expectation, proportional to $\frac{La}{r}$ with a packets in queue.

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- Frequency bands may be divided up into smaller **channels** for simultaneous communication
- **Carrier Frequency**: Fixed (higher) frequency used to carry signal. Options include **Amplitude Shift Keying**, **Frequency Shift Keying**
- **Signal to Interference and Noise Ratio**: $\frac{P_{\text{signal}}}{P_{\text{noise}}+P_{\text{interference}}}$. Bit error rate is a function of this.

Theorem (Shannon Capacity)

$C = B \log_2(1 + SINR)$

- *Capacity (C) in bits per second*
- *Bandwidth (B) in Hz*

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- Most internet protocols (HTTP/FTP/SMTP/etc.) are built on TCP, but a lot of video streaming/VoIP/trading systems use UDP

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- A **distributed database** implemented as a hierarchy of **name servers** to **resolve** domain names as IP addresses at the application layer
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- Inserting Records: Provide registrar with name and IP of authorative name server, registrar inserts NS record for auth server name and A record for auth server IP

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- Many services use hybrid (ex: video conferencing/instant messaging: users directly connect with each other but use central server to register/look up *where* users are)

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- **Persistent HTTP *with* pipelining** (HTTP/1.1 default): Connection stays open, *and* client requests a file as soon as it's referenced. What's the min possible total response time? **Setup + data transmit + 1 RTT for *all* objects**.

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```
HTTP/1.1 200 OK
Date: Mon, 21 Oct 2024 23:15:43 GMT
Server: Apache/2.4.57 (Red Hat Enterprise Linux) OpenSSL/3.0.7
Last-Modified: Mon, 23 Sep 2024 21:24:01 GMT
ETag: "eac6-622d001ecb792"
Accept-Ranges: bytes
Content-Length: 60102
Content-Type: text/html; charset=UTF-8
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- Browser sends all requests to cache, which acts as both client and server. If cache has file, returns immediately, else requests from server and returns. **Which requests does this help?**

Application Layer: Caching

- **Goal**: Satisfy client request without involving origin server
- Browser sends all requests to cache, which acts as both client and server. If cache has file, returns immediately, else requests from server and returns. **Which requests does this help?**
- Can use **conditional GET requests**. Add `If-modified-since` field to headers; if not modified, return status 304, else return file. Ensures that requests are up-to-date while still saving bandwidth. **Why?**

Application Layer: SMTP

- Uses TCP on port 25 to send mail
- Sending mail server acts as “client”, while recieving server acts as “server”. This makes it a “push” protocol, rather than a “pull” protocol (like HTTP)
- Three phases of transfer: handshake, message transfer, closure. Commands in ASCII, response is status code + message.
- Users access email boxes via **user agents** (POP3/IMAP/webmail).
- Lots more details, but they’re highly unlikely to come up on an exam.

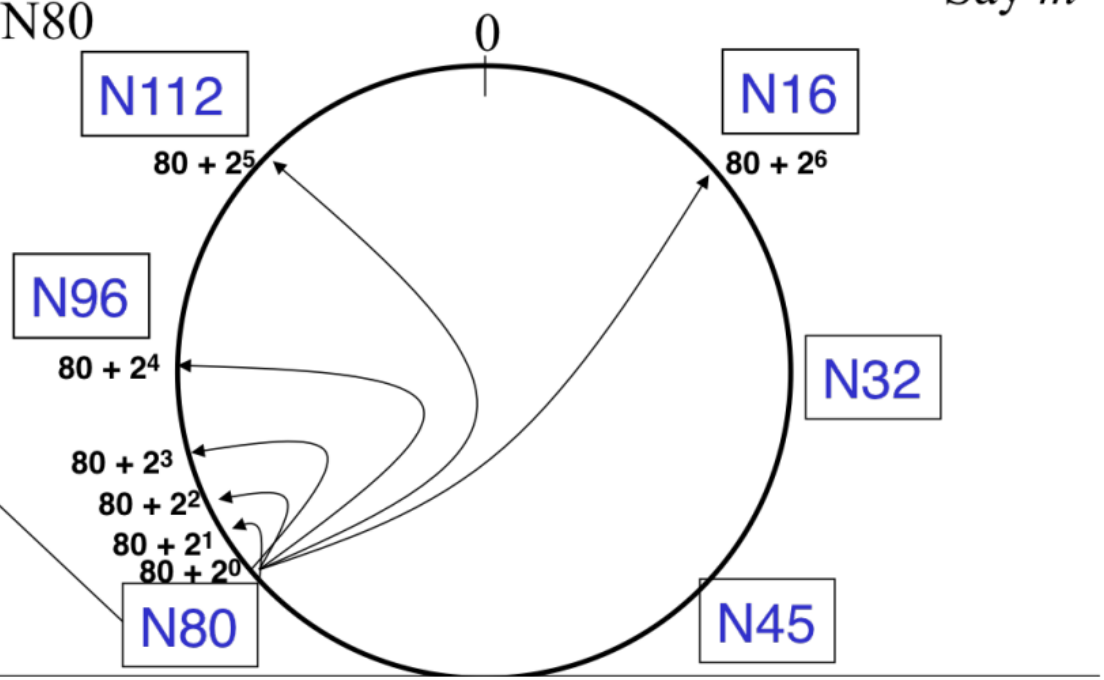
Application Layer: CHORD

- Each file assigned a hash and assigned to the next highest node, each server knows a "finger table" of nodes exponentially far away from current id, recursive lookup structure.

Say m=7

Finger Table at N80

i	$ft[i]$
0	96
1	96
2	96
3	96
4	96
5	112
6	16



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- We don't want to use any information about lower/higher layers
- Also, distributed consensus is *hard*. Some of what we want to do is the **Two Generals' Problem**: since message acknowledgments are as likely to be lost as messages, we'd potentially need infinite messages to come to consensus safely.

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- Idea: Instead of verifying message/ACK reception, have sender simply resend the packet if no ACK has been received after some time. If receiver receives duplicate packet (by sequence number), acknowledge but throw out. **How does this avoid two generals?** Receiver doesn't know (or care) *which* ACKs have been received, so no distributed consensus.

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procedure RECIEVE(k)
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  - Sender considers multiple ACK(i)s as **dupACK**s, fresh i in ACK(i) **newACK**. Useful for estimating congestion.

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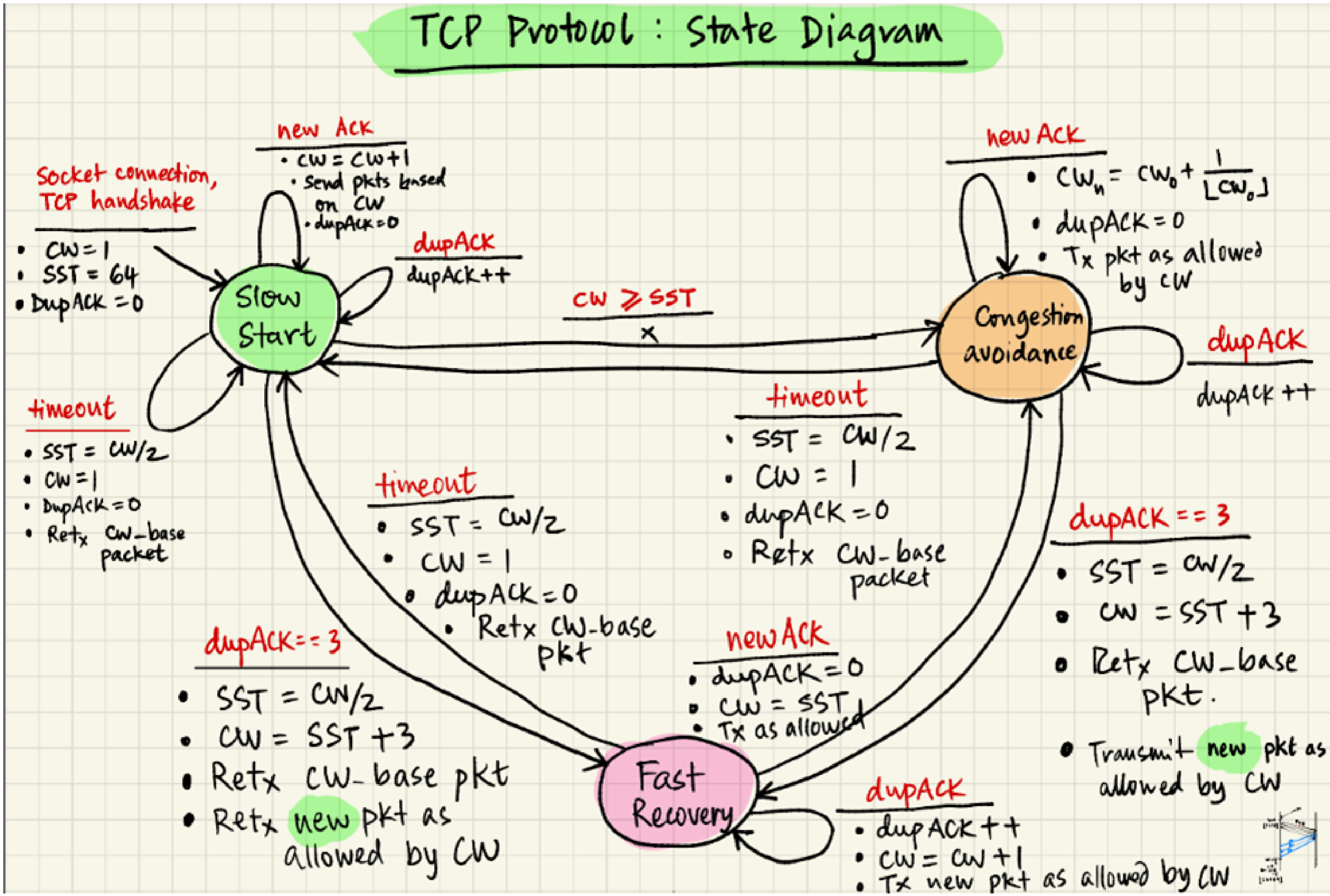
- DupACKs aren't necessarily a bad sign, but might be indicator of missed packets. If 3 dupACKs in a row, retransmit DupACK packet but don't reset SST, slightly cut CW (**fast recovery**).

Transport Layer: TCP State Machine



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- AvgRTT estimated by rolling average: $RTT_{avg} \leftarrow (1-\alpha)RTT_{avg} + \alpha\, RTT_{packet}$
- "guard factor" can be a deviation estimate:

$$
\begin{aligned}
devRTT_{avg} &\leftarrow (1-\beta)\, devRTT_{avg} + \beta(|RTT_{packet} - RTT_{avg}|) \\
RTO &\leftarrow RTT_{avg} + 4\, devRTT_{avg}
\end{aligned}
$$

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- TCP guarantees **max-min fairness** (in stable state): All flows requesting less than fair share get their request. Remaining flows divide equally.

Feedback

http://go.acm.illinois.edu/cs438_mt1_feedback