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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  - **Reservations/Circuit Switching**: Source sends call request, path between source and destination reserved + blocked off, communication happens, then circuit teardown. Used in some telephone + ATM protocols. Can share a channel using FDM (split frequencies) or TDM (round-robin whole resource)

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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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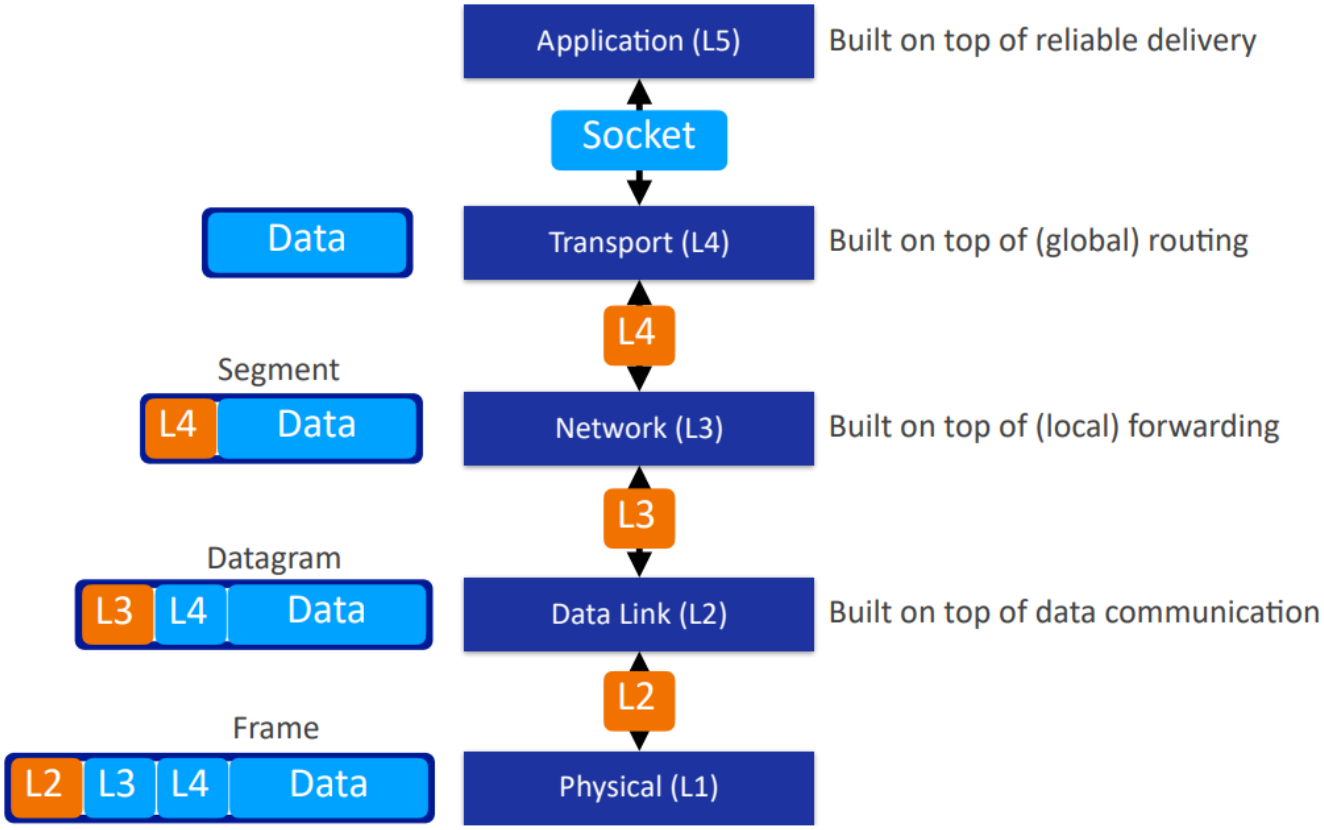
- When packet arrives at destination, forward to correct application

- 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.

Foundations V: Signal Processing

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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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- 4 types of records: A (hostname `name` is IP address `value`), NS (authoritative name server for `name` can be found at `value`), CNAME (the "real" name for alias `name` is the canonical name `value`), MX (the mailserver for `name` has name `value`)

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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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  - Example Response:

```
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

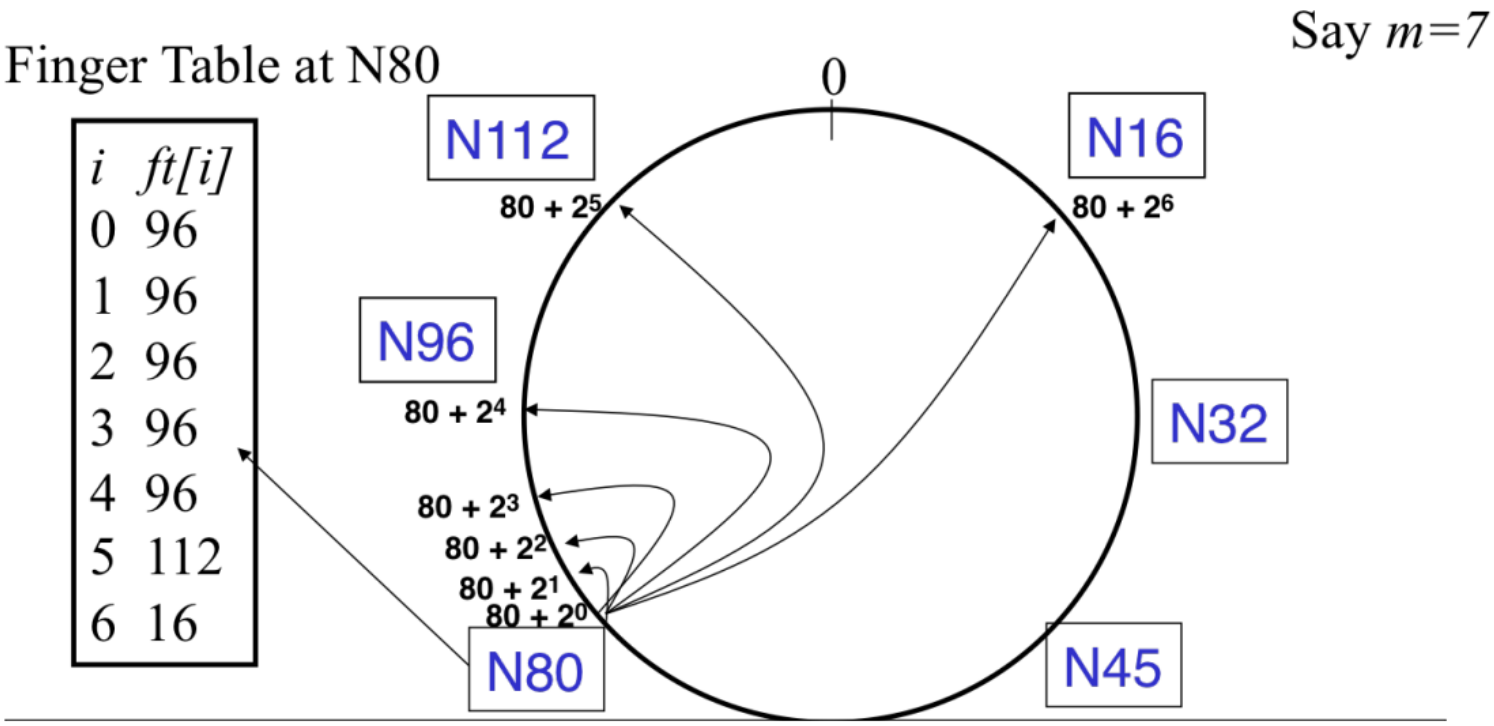
- **Goal**: Satisfy client request without involving origin server
- Browser sends all requests to cache, which acts as both client and server. If cache has file, returns immediately, else requests from server and returns. **Which requests does this help?**
- Can use **conditional GET requests**. Add `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.



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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.

Transport Layer: Pipelined Protocols

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- TCP takes a **hybrid approach**, reports cumulative ACKs (lowest seq # not recieved - 1), but will accept out-of-order packets and reorder them.
  - Sender considers multiple ACK(i)s as **dupACK**s, fresh i in ACK(i) **newACK**. Useful for estimating congestion.

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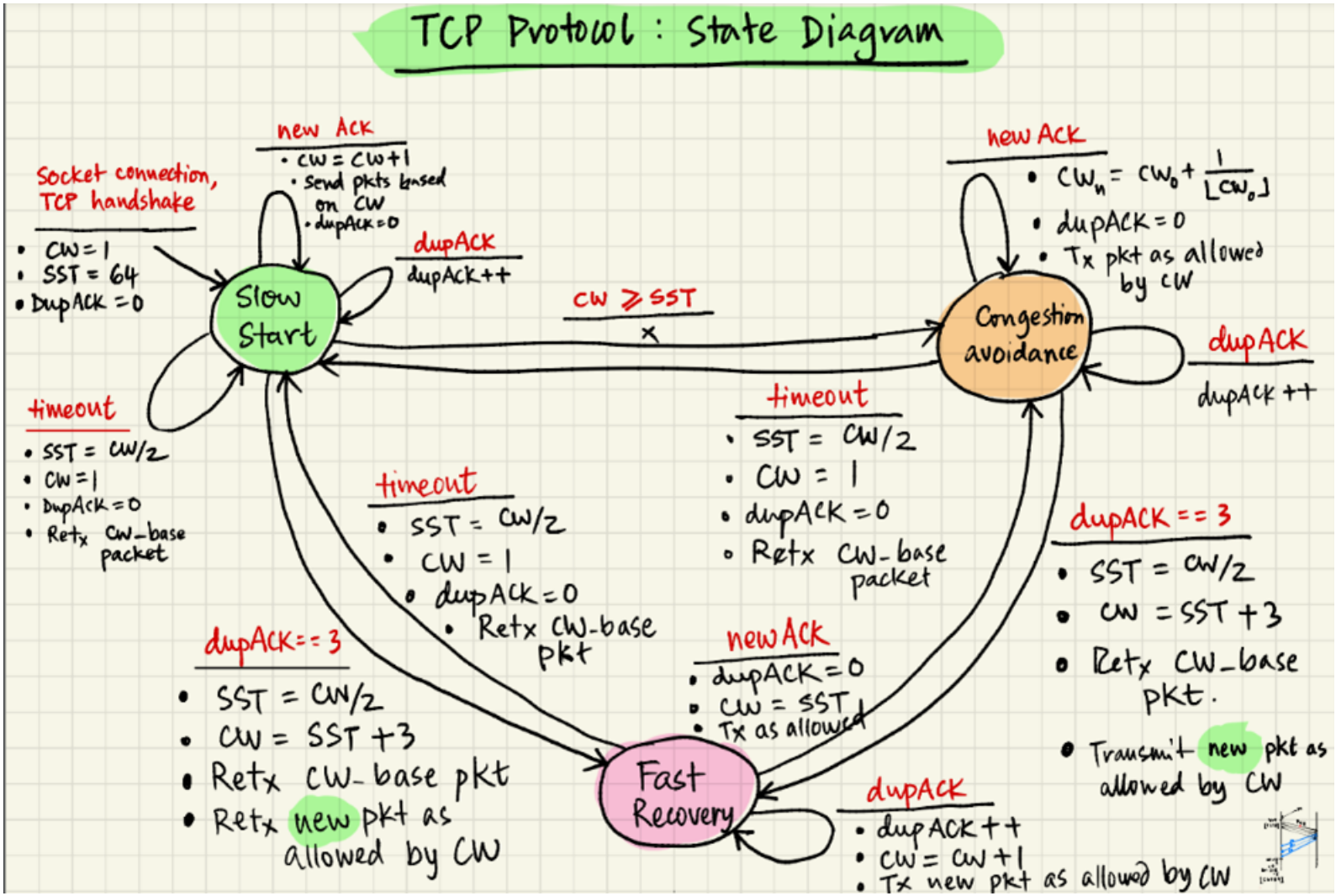
- Sender keeps a timer to interrupt for timeout. When CW shifted, increase the timer by the gap between packets. On timeout, drastically decrease CW, SST, resend packets.
- DupACKs aren't necessarily a bad sign, but might be indicator of missed packets. If 3 dupACKs in a row, retransmit DupACK packet 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:

$$devRTT_{avg} \leftarrow (1-\beta)\ devRTT_{avg} + \beta(|RTT_{packet} - RTT_{avg}|)$$
$$RTO \leftarrow RTT_{avg} + 4\ devRTT_{avg}$$

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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