# 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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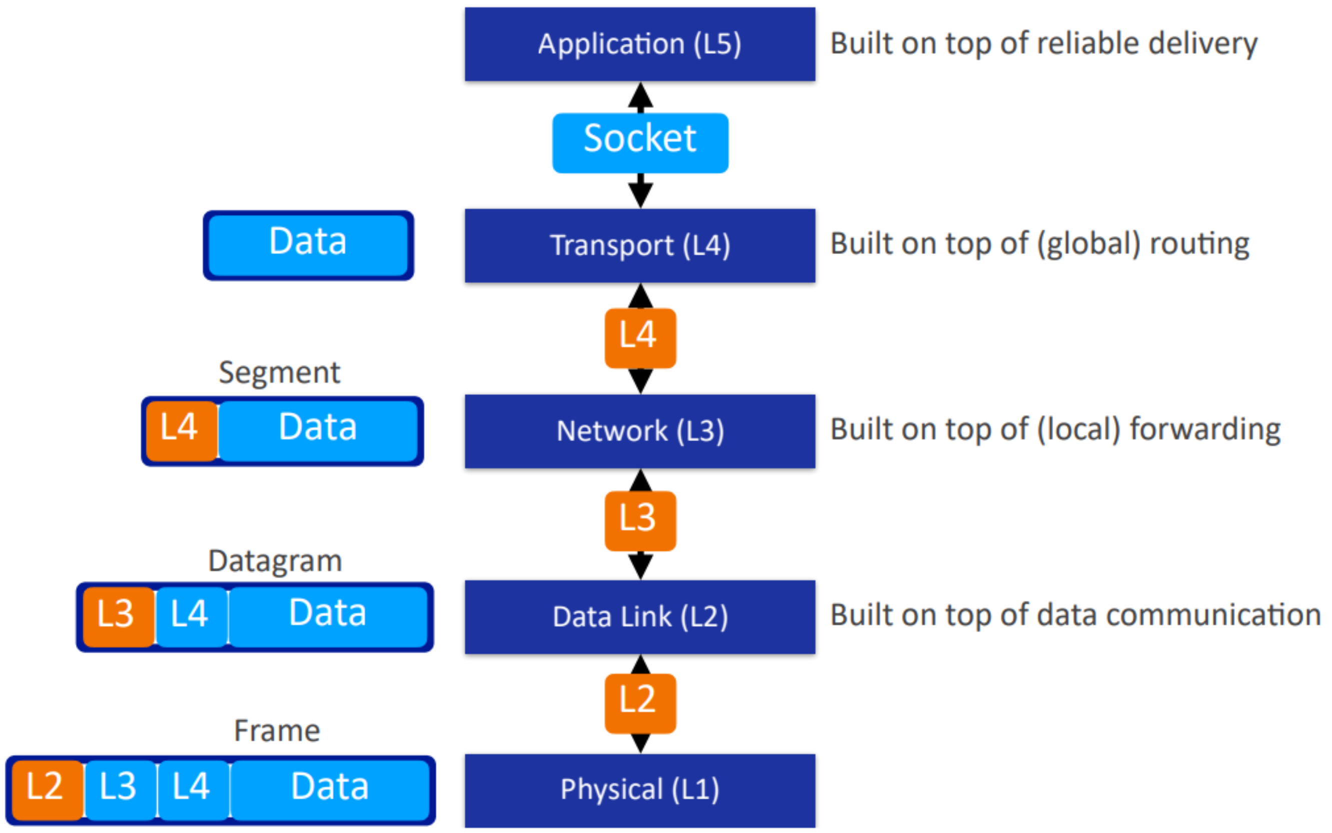
- When packet arrives at destination, forward to correct application

- 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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- **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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```
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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- 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.

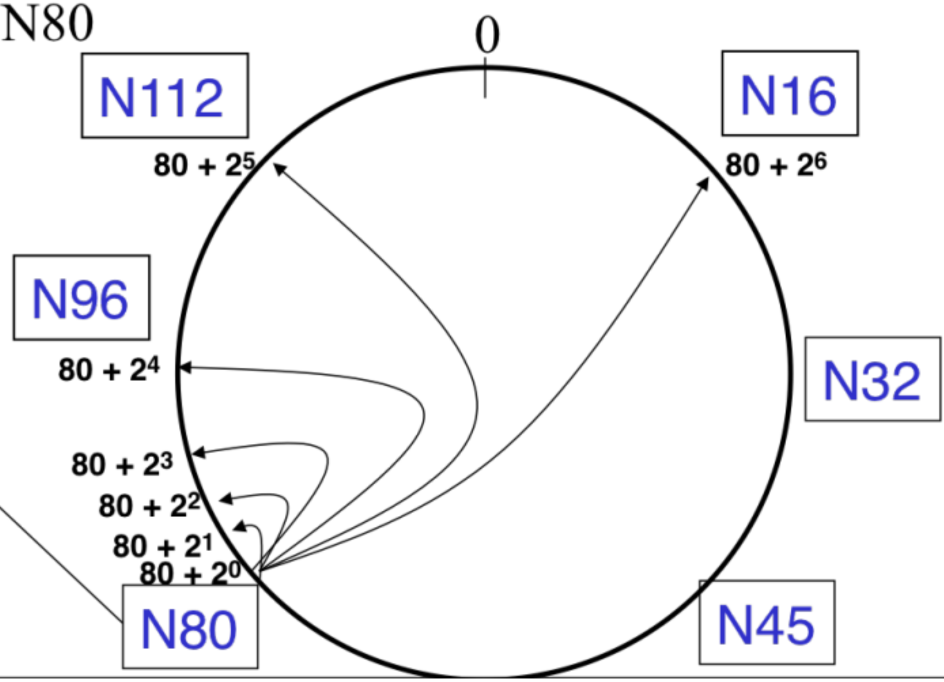
# Application Layer: CHORD

- 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 |



# Transport Layer: Goals

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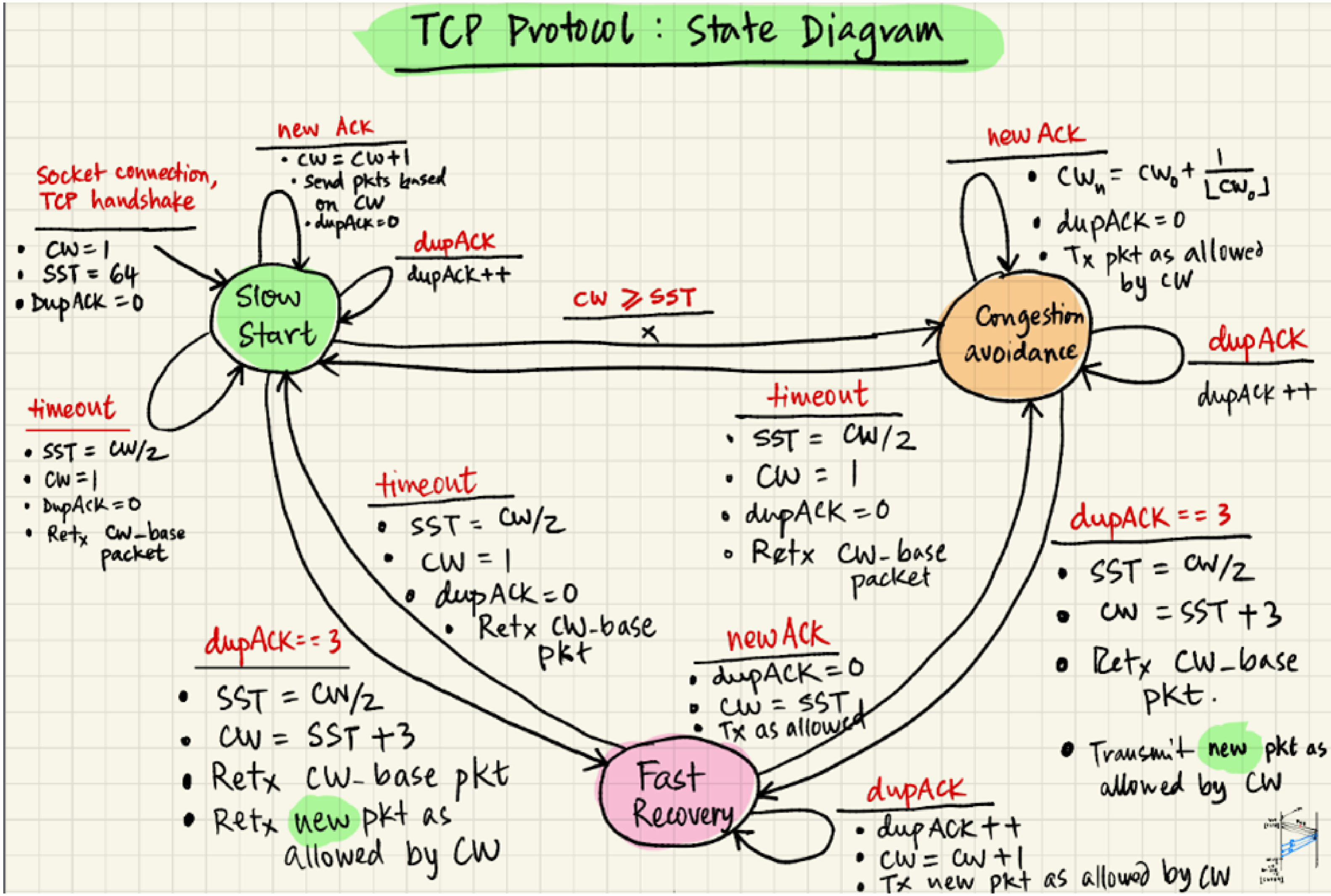
- DupACKs aren't necessarily a bad sign, but might be indicator of missed packets. If 3 dupACKs in a row, retransmit DupACK packet but don't reset SST, slightly cut CW (**fast recovery**).

# 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