CS 438 Final 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!
- We're just planning to cover second-half material, but have slides for the entire course- please let us know what you want to cover
- 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.

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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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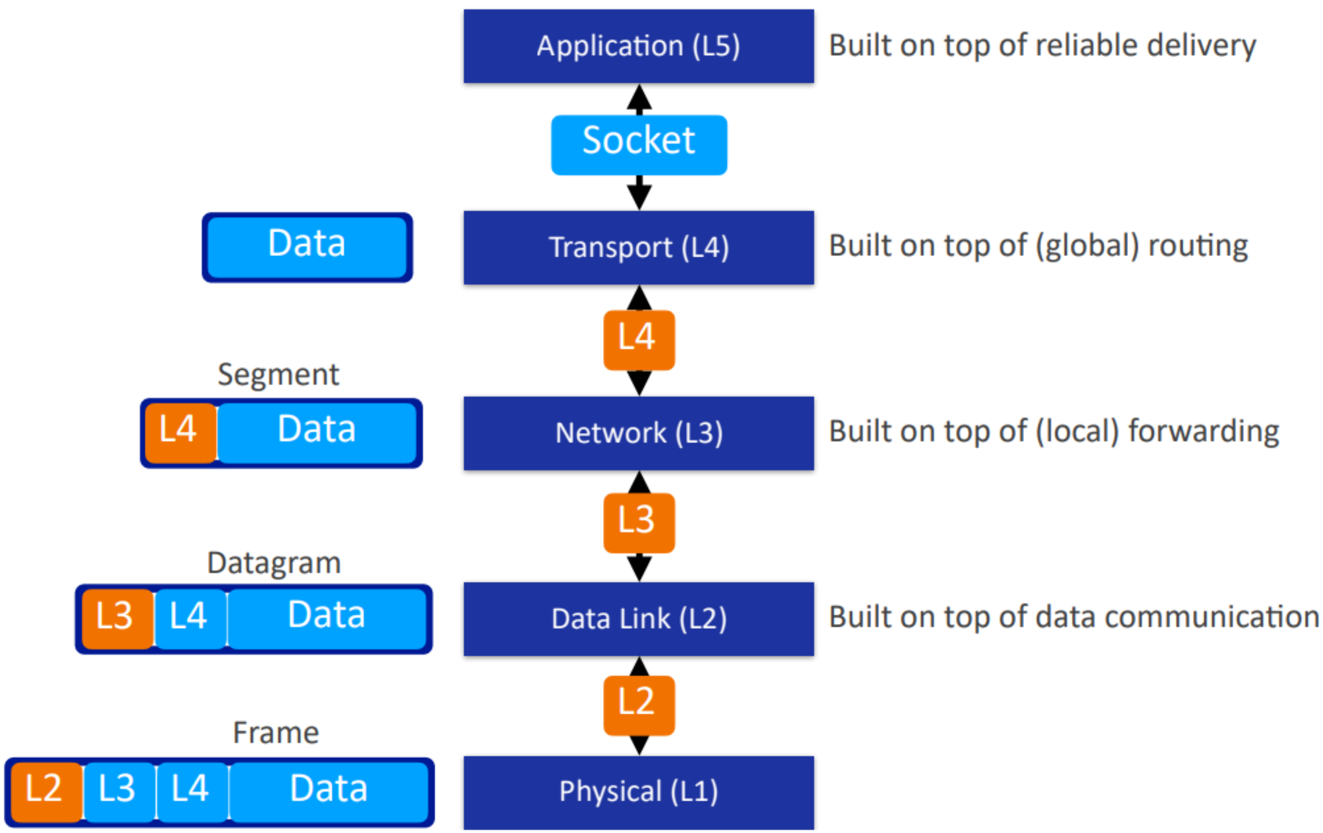
- When packet arrives at destination, forward to correct application

- 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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- 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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- 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

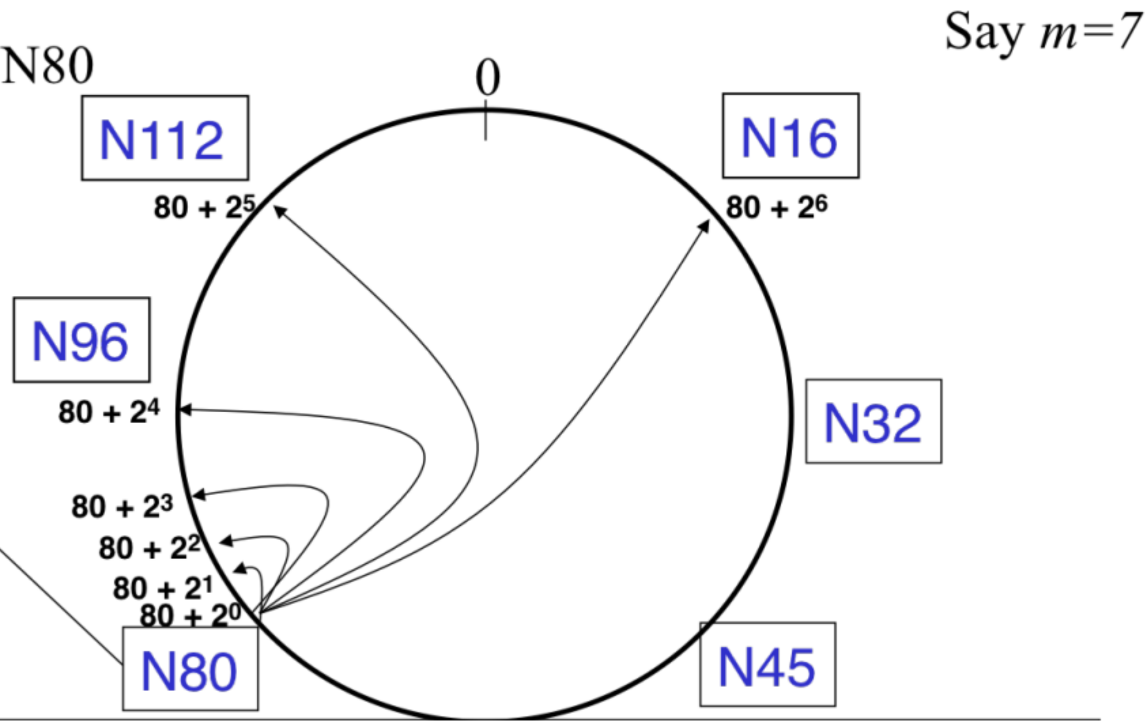
- Uses TCP on port 25 to send mail
- Sending mail server acts as “client”, while recieving server acts as “server”. This makes it a “push” protocol, rather than a “pull” protocol (like HTTP)
- Three phases of transfer: handshake, message transfer, closure. Commands in ASCII, response is status code + 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.

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

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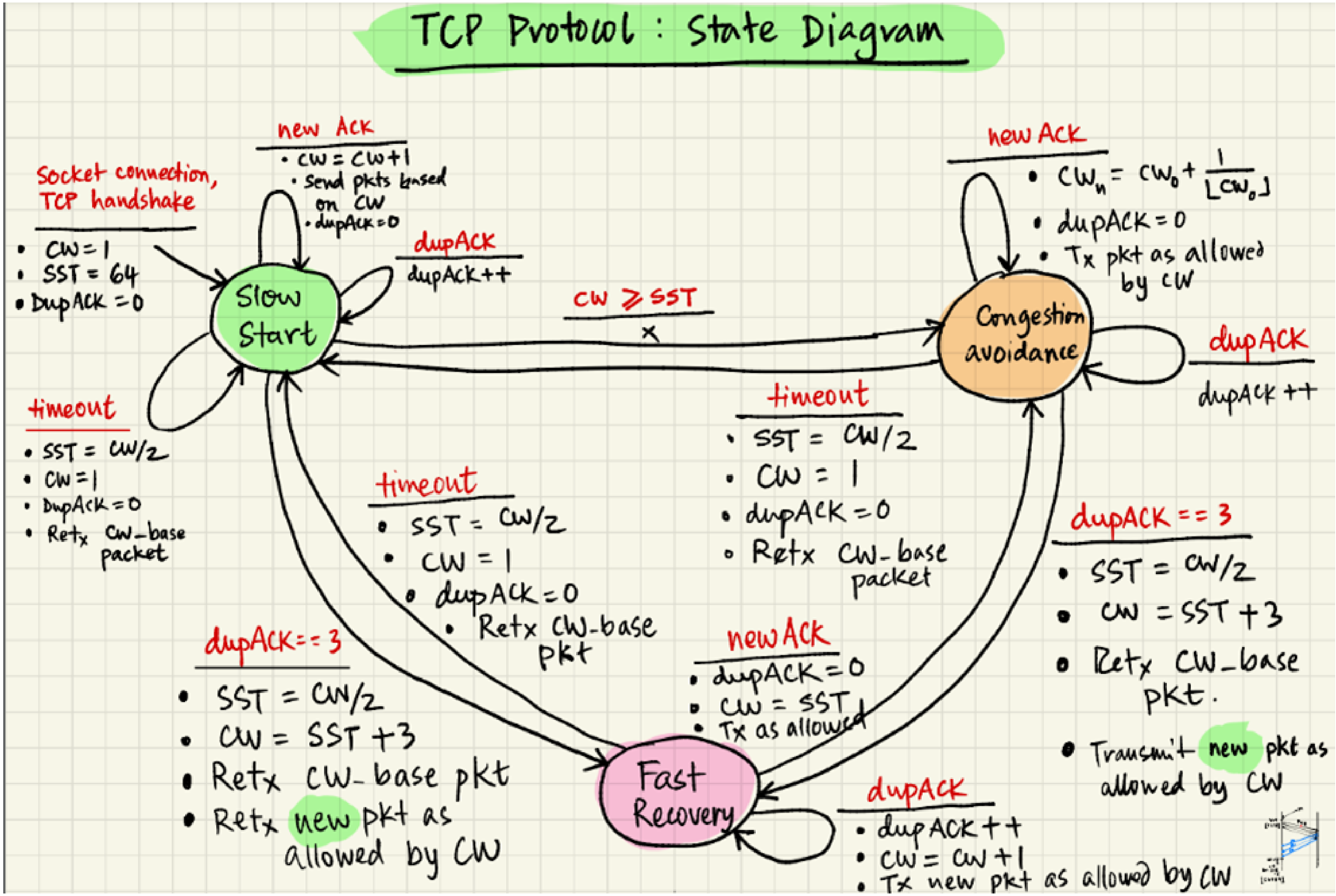
- DupACKs aren't necessarily a bad sign, but might be indicator of missed packets. If 3 dupACKs in a row, retransmit DupACK packet but don't reset SST, slightly cut CW (**fast recovery**).

Transport Layer: TCP State Machine



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- "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.

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- Routers examine headers of all IP packets, figure out where to pass it along

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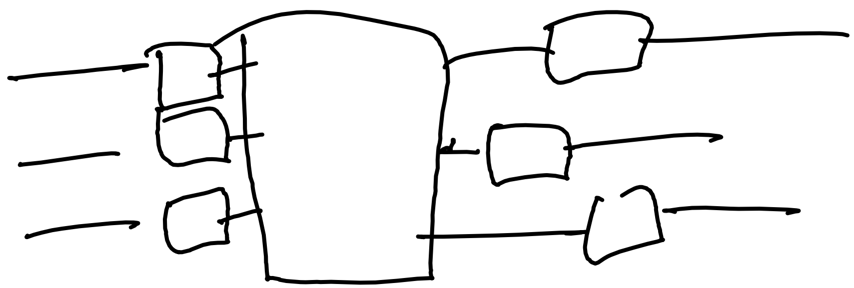

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- Routers run routing algorithms to determine **forwarding table**

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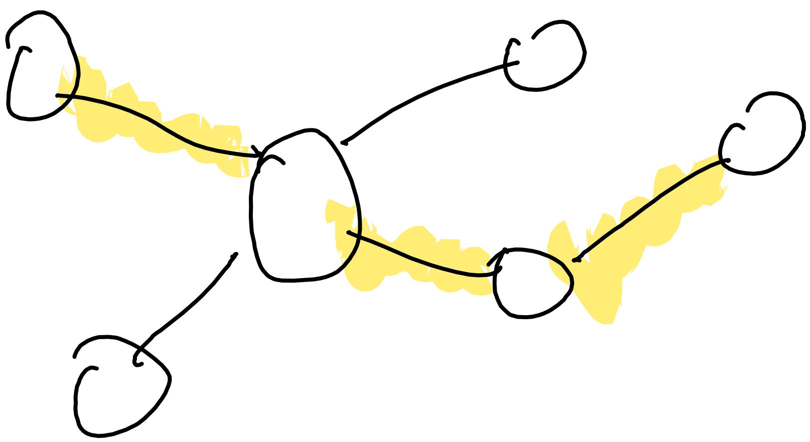

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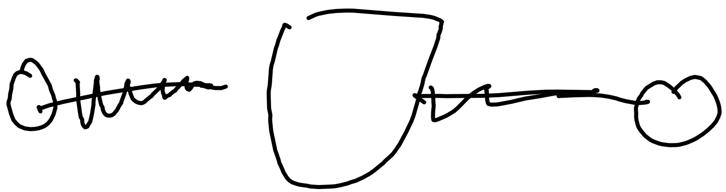

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- This sucks (**lots** of residual capacity along links left open), so it's not really used outside of old ATM networks.

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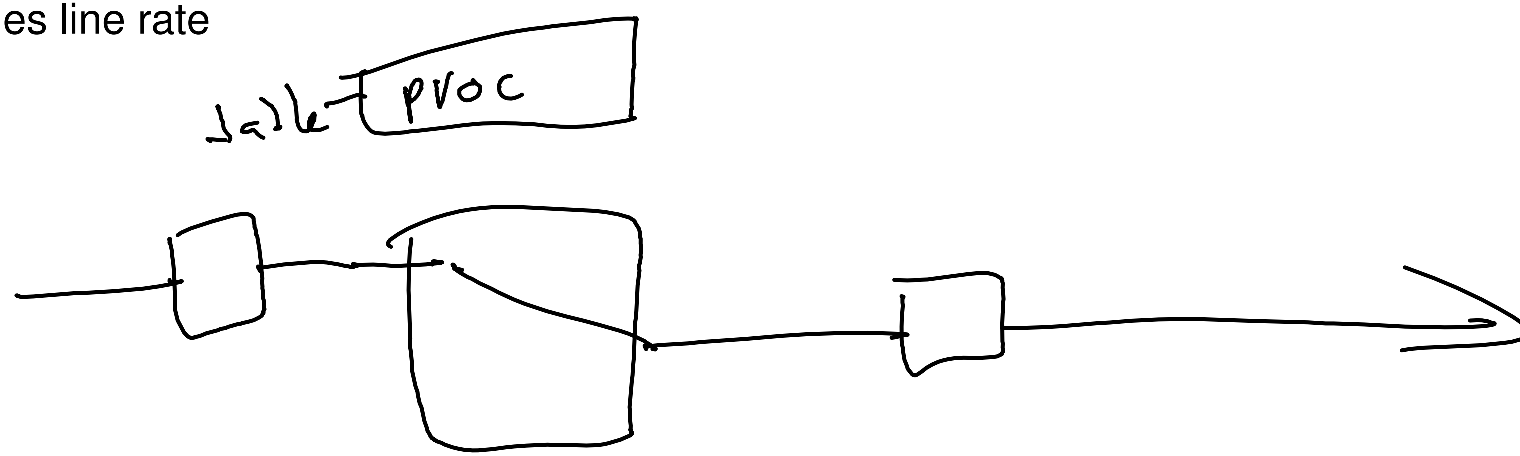
- To allow for more finegrained control, allow different prefixes, map packets to **longest** prefix that fully matches (ex: 240.128.x.x/5 is more specific than 128.x.x.x/1, so if both rules in table, choose first)

Subnet mask
first n bits

Routers: The Low-Level View

- Routers consist of processors (which control routing tables and act in millisecond time frames) and switching fabric (which just simply forwards packets along and acts in nanosecond time frames)
- On link-layer receive: using header values, lookup port to send to and queue for transmit. Traditionally, destination-based forwarding (only use destination IP address) is used, but **generalized forwarding** (based on any set of header values) can help at times.
- Switching fabric transfers packets between input + output links, goal is to have a high **switching rate** (rate at which inputs -> outputs can be transmit, measured as multiple of input/output line rate). Goal is with *N* inputs, switching rate of *N* times line rate



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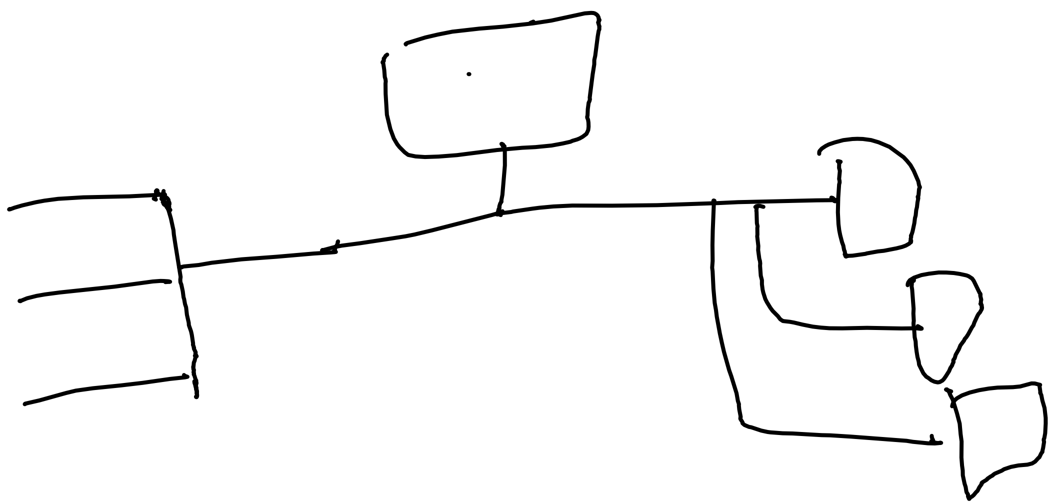

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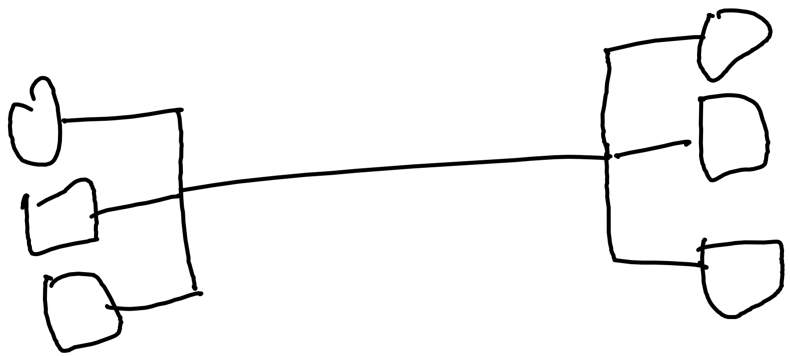

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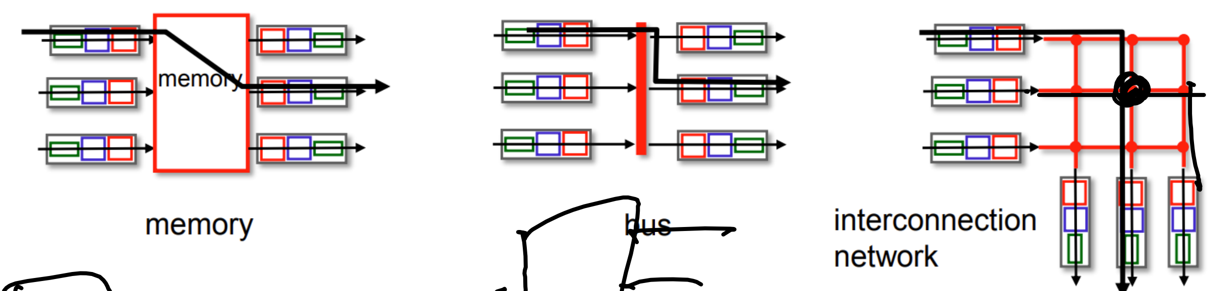
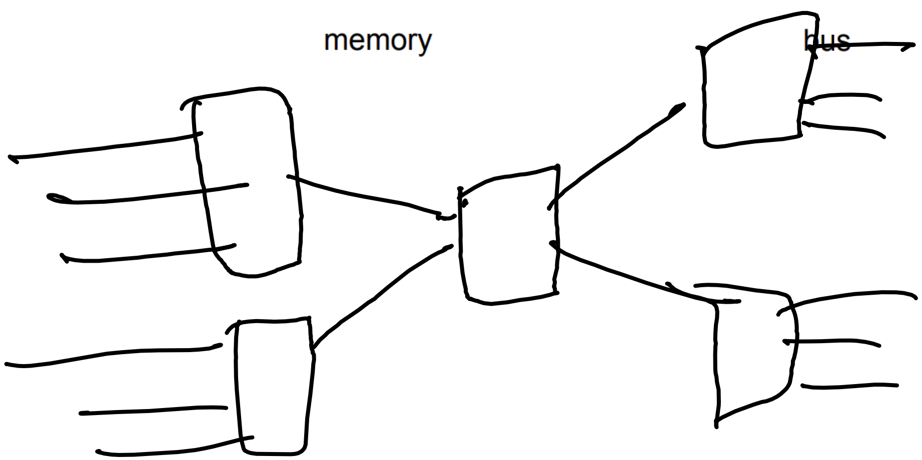
- **Others**: Crossbar (only fails when intersection between paths), multistaged switching (switch formed by switches), etc. We can also exploit parallelism by breaking packets up and reassembling them on transmit.





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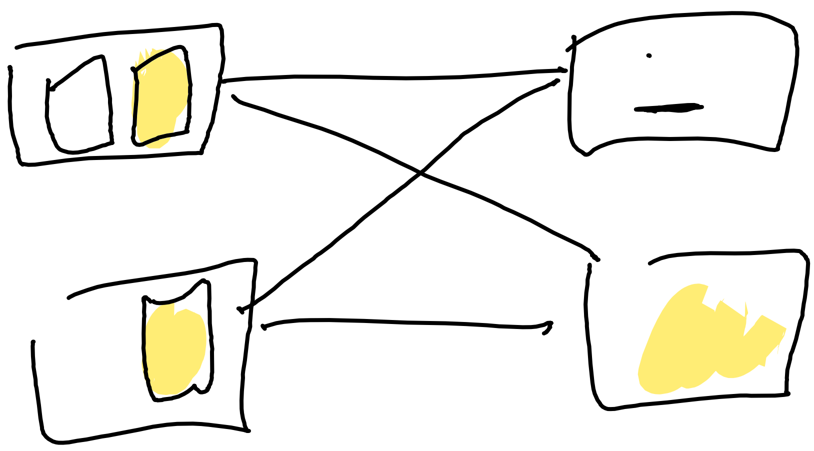

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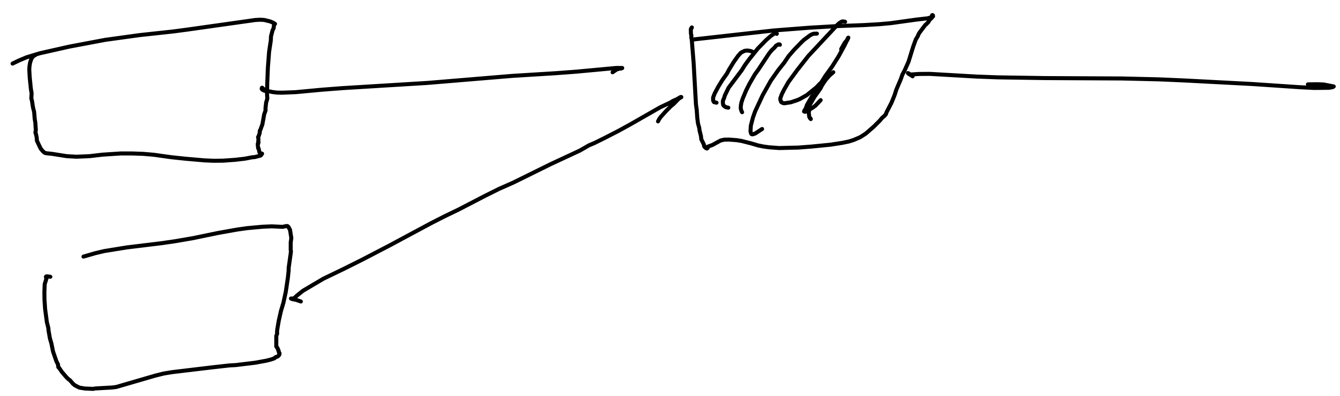

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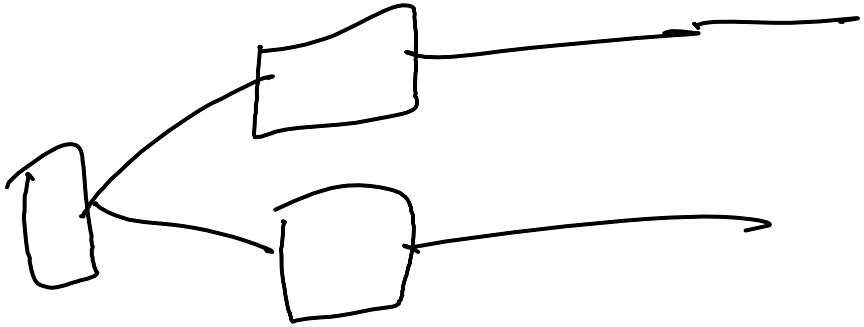

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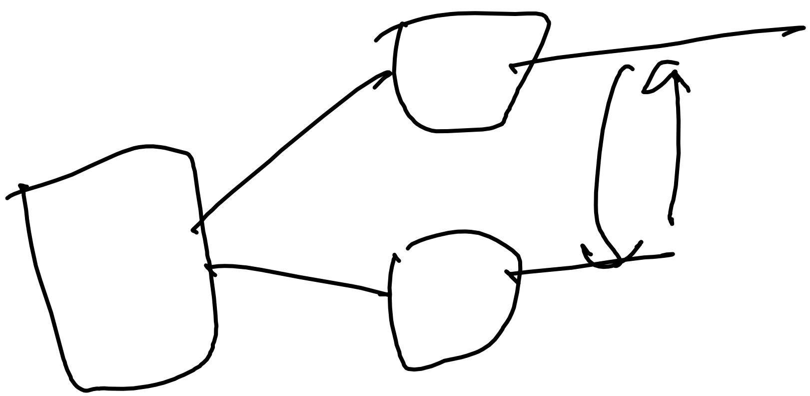

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  - **Weighted Fair Queuing**: give each queue a weight, per cycle, spend $\frac{w_i}{\sum_j w_j}$ of the cycle on sending from this queue.

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  - Client broadcasts request for DCHP server, which issues address. Client confirms use. Can also assign first-hop router, DNS server, network mask.

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8000 80

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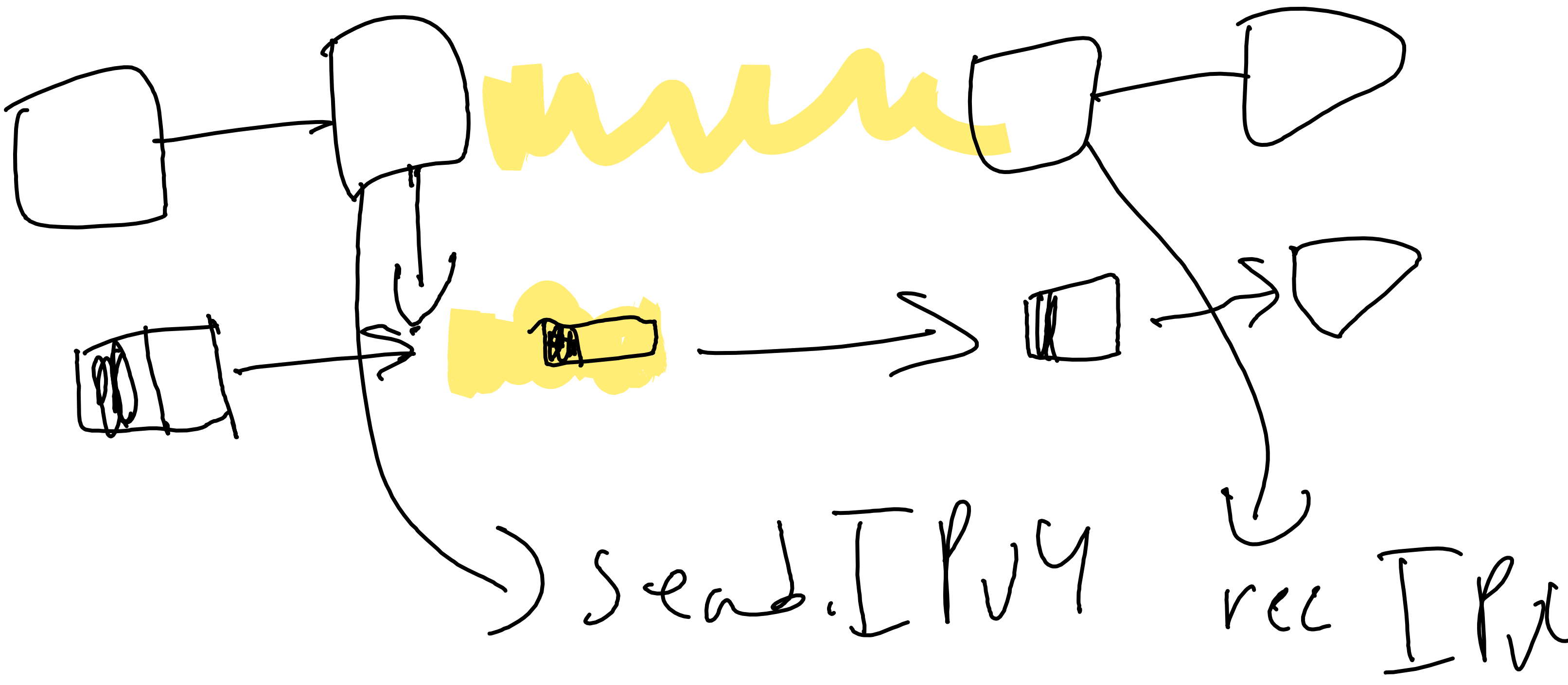
- Routers translate (source, port) to (NAT, NAT port) on outgoing packets, and (NAT, NAT port) to (internal, port) on incoming packets.
- Controversial as this breaks layer separation, but extensively used as IPv4 ~~ports~~ adds, are expensive.

IPv6

- The “real” solution to IPv4 address exhaustion, 128 bit address but fewer additional headers, so 40-byte fixed length
- Not every device supports it, so sometimes we need to use IPv4 tunneling: simply encode your IPv6 packet with headers packet as a payload in an IPv4 packet. Set source to the node converting to IPv4, and destination to the node that is back on IPv6.



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- Algorithms can either rely on **global** information (all routers have knowledge of network topology, even in areas that are far away) or be built to only use **decentralized** information (routers only know about their own physical connections, and gather everything else from each other)

Link state: Dijkstra's

Pseudocode

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procedure LinkState(G, s)
    Initialize a table distances to all be ∞
    Initialize a priority queue pq
    for all s → t do
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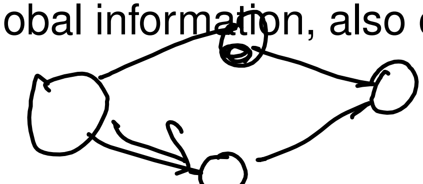
- **Downsides**: needs global information, also oscillations are possible if two paths are nearly equal.



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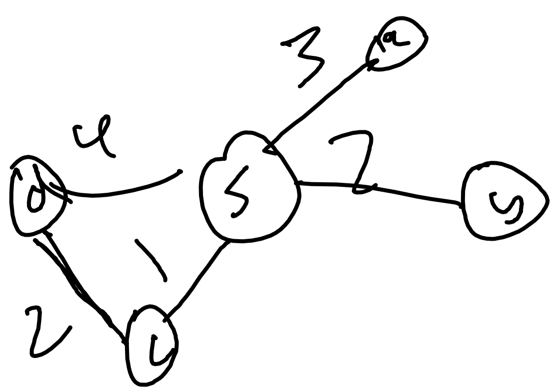

s

a	b	c	d
3	2	1	~~4~~ 3

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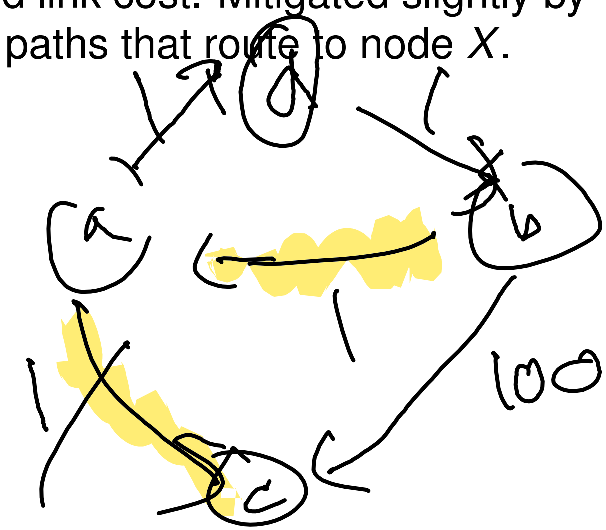

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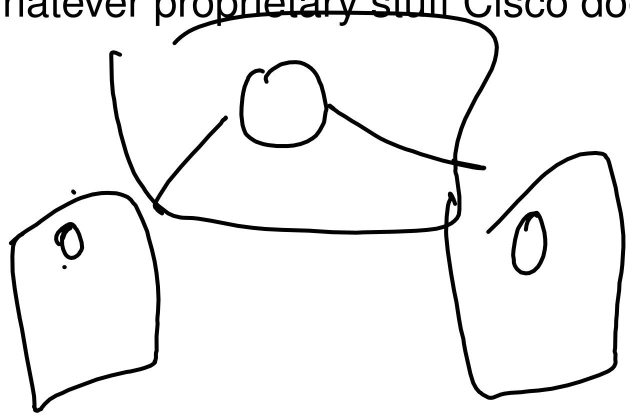
- Intra-domain protocols/**Interior Gateway Protocols** (IGP) include **RIP** (distance vector), **OSPF** (Link state, with routers flooding entire AS with advertisements with message authentication, multicast, and hierarchical OSPF as “advanced” features), and whatever proprietary stuff Cisco does.



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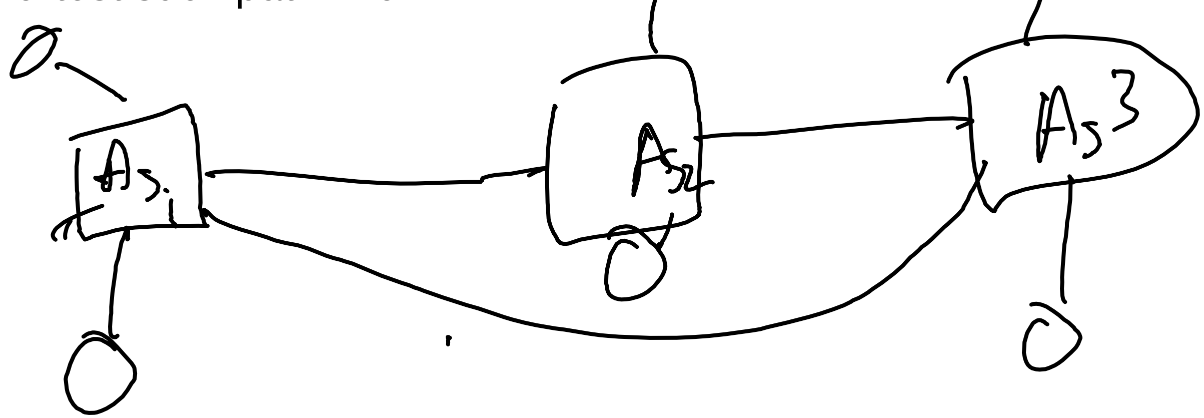

As1, As3, R,

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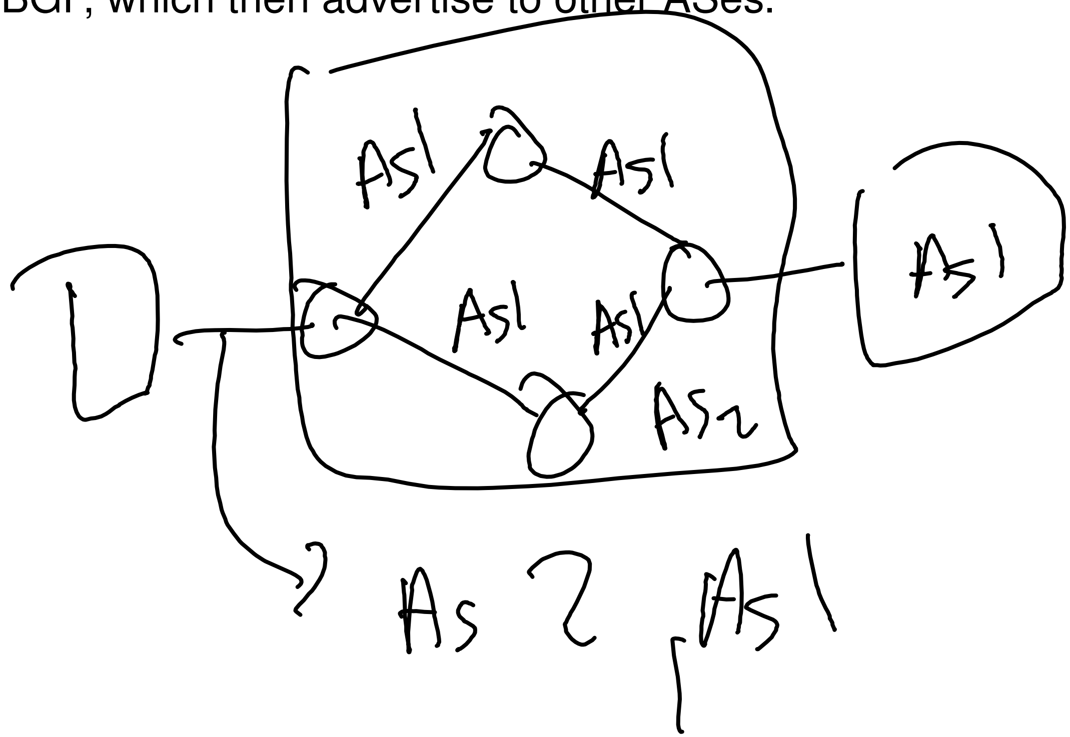

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- Routers can choose how to send to destination routes based on different criteria including shortest AS-PATH and “hot potato” routing (get it out of my AS as quickly as possible)

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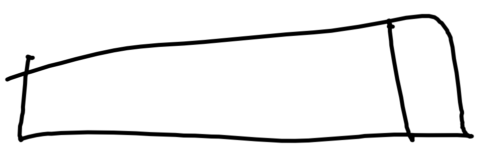
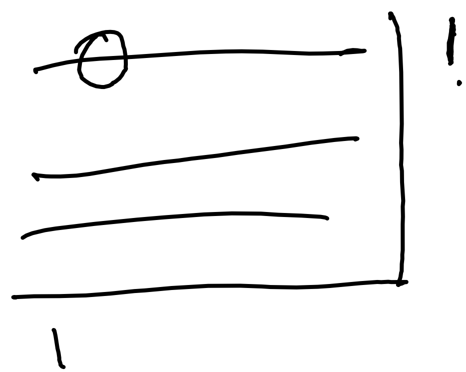
- Bit error correction usually done via **pairity checking**: options include single bit, two dimensional pairity checking, Hamming coding, CRC. More reliable transmission mediums will use less complete correction means or will leave it out entirely.





Multiple Access Protocols

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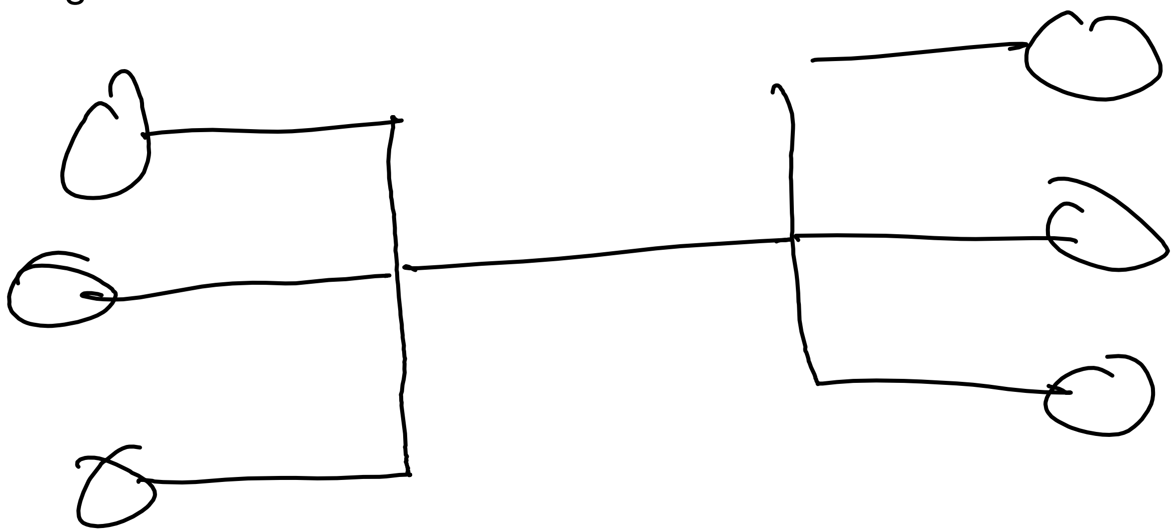

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    - "Taking Turns": Nodes take turns but nodes that have more to send can take longer turns. Includes token-passing (control token giving the right to send passed between nodes)

(Slotted) ALOHA

- If all frames same size and time divided into equal sized slots, when node receives frame, transmit immediately. If collision, retransmit with probability p^k (with k consecutive failures) for each subsequent frame until success.

N_1	T	pT		
N_2			T	pT
N_3	T	p	pT	p^2

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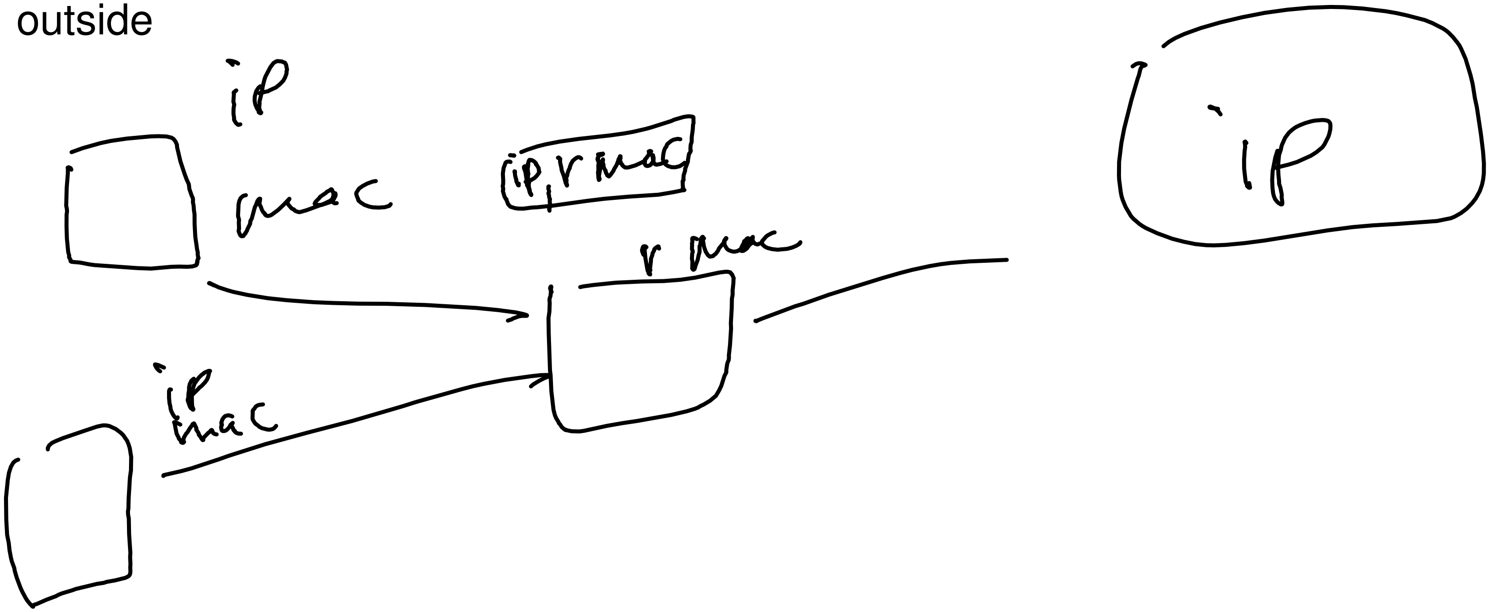
- If N nodes active, the successful transmit probability is $Np(1-p)$, so for large N, successful transmit only happens 37% of the time!
- Unslotted ALOHA doesn't even use slots, synchronization. Performance even worse: 18% expected success rate.

CSMA/CD

- Listen on channel before transmit. If channel busy, wait until free, then transmit full frame. On collision, abort and choose wait time at random between 0 and $2^m - 1$ for m consecutive collisions
- Efficiency trends to 1, so much better than ALOHA

MAC Addressing

- 48 bit address that's hardcoded, can move between LANs using same address. Each node in a LAN has an (IP address, MAC address, TTL) table. If MAC address not known, node broadcasts ARP query, requested device responds in unicast with its address
- Requests outside of LAN are targeted to router, which then will transmit to the outside



Ethernet

- Topologies include ***bus** (all nodes in same collision domain) and **star** (switch in center, no collisions)
- Connectionless (no handshaking), unreliable (no acknowledgement), CSMA/CD with binary backoff for MAC
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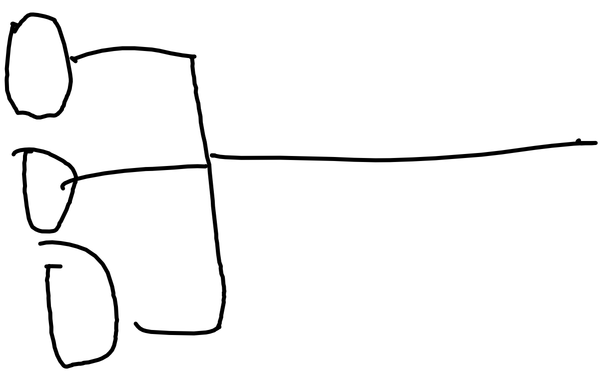

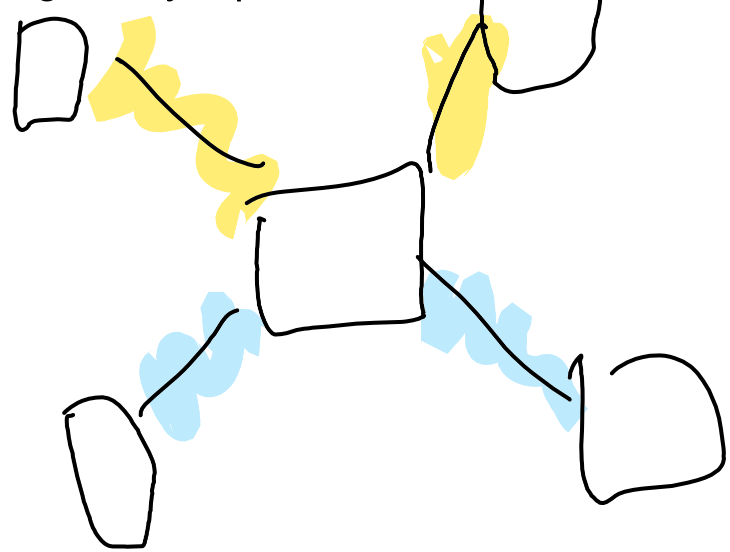

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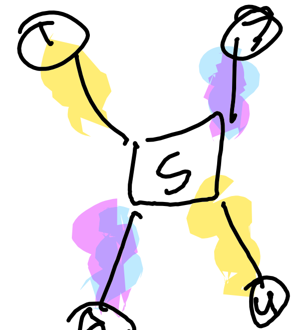

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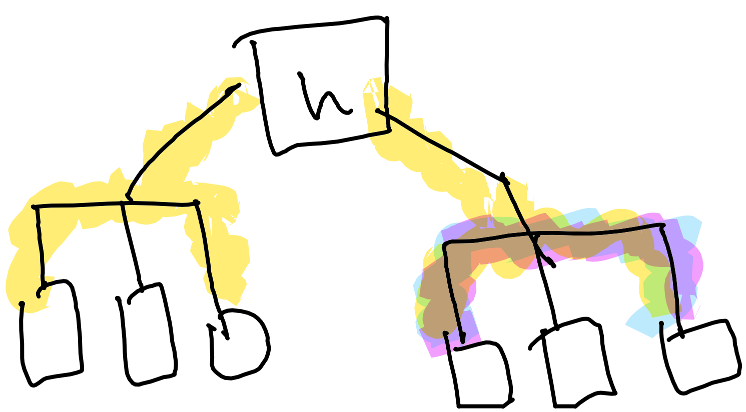
- Destination and source MAC addresses, higher layer protocol in header, CRC at end for bit error correction
- **Switches** take an active role by selectively forwarding frame to outgoing links, using CSMA/CD to access segment. Hosts are unaware of the presence of switches
- Switches are **self learning**: they have a switching table as to what MAC addresses are accessible from which interface. On request coming from node, add its address/interface to table If request designated for known node, send on corresponding interface, else flood.



Wireless

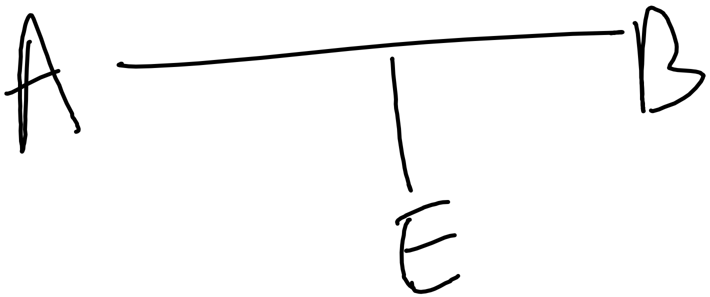
- A bunch of new considerations that don't exist on wired media: can share frequencies of channel, may want to swap between channnels
- Larger issue: signal decay. On wired, you can usually detect all activity over the medium, but on wireless there might be a "hidden terminal" that you cannot detect but can interfere with reciever's ability to recieve your message. Also "exposed terminal" that you can detect but wouldn't interfere with your transmission because the reciever is out of their range.
- Solution: CSMA/CA: Send "Requests to Send" (RTS) messages when wanting to send to someone else, wait for "Clear to Send" (CTS) from reciever. Nodes only need to not send when they hear a CTS. Still possibility of collisions with data and RTS/CTS packets, but those are much shorter so probability lower.

hidden terminal

exposed term.

Security

- **Threat Model**: Attacker has access somewhere along the channel that sender and reciever both have. We want to make sure that attacker can't eavesdrop on messages, modify message content, impersonate sender, etc.



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- More formally, we want to maintain **confidentiality** (only sender and reciever should know what's in the message), **integrity** (no one can modify the message without being detected), and **accessibility** (transmitter should always be able to send to reciever) of data. Also, we want to ensure authentication (reciever should know that transmitter actually sent the data and not someone else).

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- We'll achieve it through processing data through an **encryption algorithm**. Given some message m, we want there to be a K_e, K_d s.t. $K_d(K_e(m)) = m$, but it's hard to recover m without some information internal to K_d

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- Unfortunately, this isn't enough on its own. We don't have a way to transmit decryption keys over an insecure channel ("mailman"/"padlock" problem), and the keys are shared, so this won't work for authentication.

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$$\left(m^{K_+}\right)^{K_-} \quad m^{K_+ \cdot K_-} = m^1 = m$$

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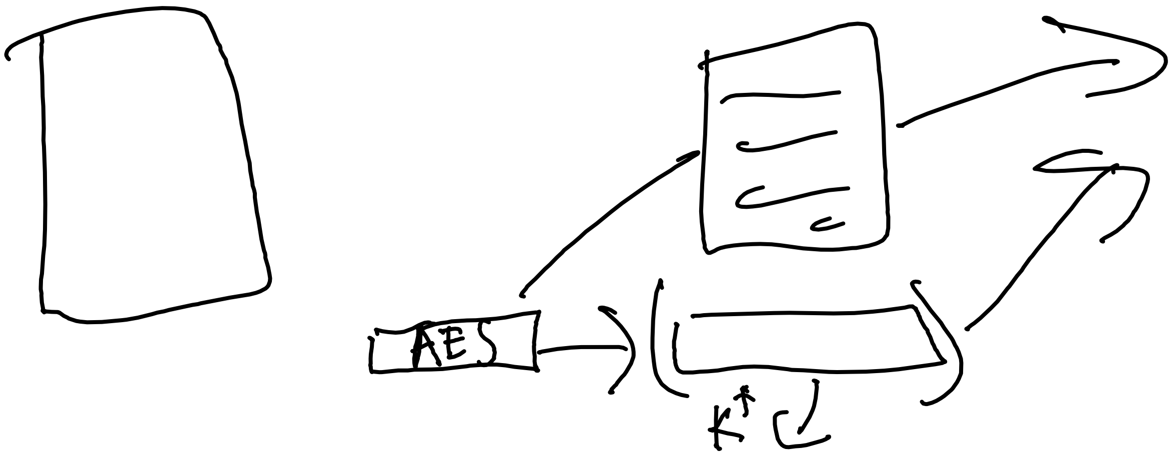

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- Still can't tell who someone is unless you know for a fact that their public key is something specific.Have a trusted source (**Certificate Authority**, CA) store this information.

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5. I want to add a link to my network graph that has a negative weight. My AS runs a distance vector protocol. Will this cause problems for the routers? If not, how quickly will distances using my negative link update on other servers?

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12. Why do we need IP addresses? Why can’t we just route to MAC addresses?

Feedback

http://go.acm.illinois.edu/cs438_final_feedback