EECS 489
Computer Networks

Winter 2024

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Material with thanks to Aditya Akella, Sugih Jamin, Philip Levis, Sylvia Ratnasamy, Peter Steenkiste, and many other colleagues.

Logistics

- Open book/text/notes, but OFFLINE
  - Except for taking the exam over the Internet
- You're NOT allowed to write/run any programs
- You're NOT allowed to collaborate with anyone
- You're NOT allowed to use any AI

General guidelines (1)

- Test only assumes material covered in lecture, discussion sections, quizzes, and assignments
  - Text: only to clarify details and context for the above
- The test doesn't require you to do complicated calculations
  - Use this as a hint to determine if you're on right track
- You don't need to memorize anything
- You do need to understand how things work

General guidelines (2)

- Be prepared to:
  - Weigh design options outside of the context we studied them in
  - Contemplate new designs we haven't covered in detail but can be put together
    - e.g., I introduce a new IP address format; how does this affect..."
  - Reason from what you know about the pros/cons of solutions we did study

General guidelines (3)

- Exam format
  - Q1: True-False questions
  - Q2: MCQ questions
  - Q3-QN networking use cases
    - Questions not ordered in terms of complexity
- 60 minutes

This review

- Walk through what you're expected to know at this point: key topics, important aspects of each
- Not covered in review does NOT imply you don't need to know it
  - But if it's covered today, you should know it
- Summarize, not explain
  - Stop me when you want to discuss something further!

Topics

- Basics (lectures 1–2)
- Application layer (lectures 3–5)
  - HTTP, DNS, CDN, Video Streaming, and Cloud
- Transport layer (lectures 6–9)
  - UDP vs. TCP
  - TCP details: reliability and flow control
  - TCP congestion control: general concepts only
- Network layer (lecture 10–11)
  - Overview
  - Data plane

Basic concepts

- You should know:
  - Packet vs. circuit switching
  - Statistical multiplexing
  - Link characteristics
  - Packet delays

Switched networks

- End-systems and networks connected by switches instead of directly connecting them
- Allows us to scale
  - For example, directly connecting N nodes to each other would require N^2 links!

Two approaches to sharing

- Packet switching
  - Network resources consumed on demand per-packet
  - Admission control: per packet
- Circuit switching
  - Network resources reserved a priori at “connection” initiation
  - Admission control: per connection

Statistical multiplexing

- Allowing more demands than the network can handle
  - Hoping that not all demands are required at the same time
  - Good for bursty traffic (average << peak demand)
  - Packet switching exploits statistical multiplexing better than circuit switching

Delay

- Consists of four components
  - Transmission delay
  - Propagation delay
  - Queuing delay
  - Processing delay

due to link properties

due to traffic mix and switch internals

End-to-end delay

transmission

propagation

queueing

processing

transmission

propagation

queueing

processing

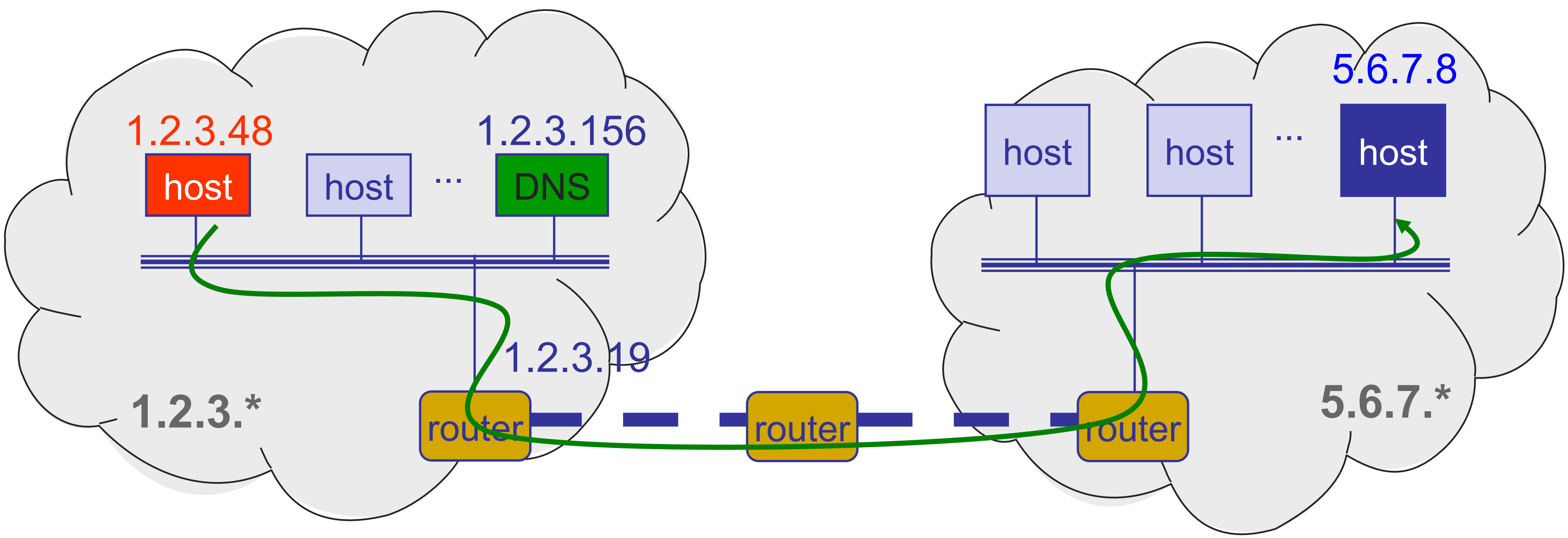
transmission

propagation

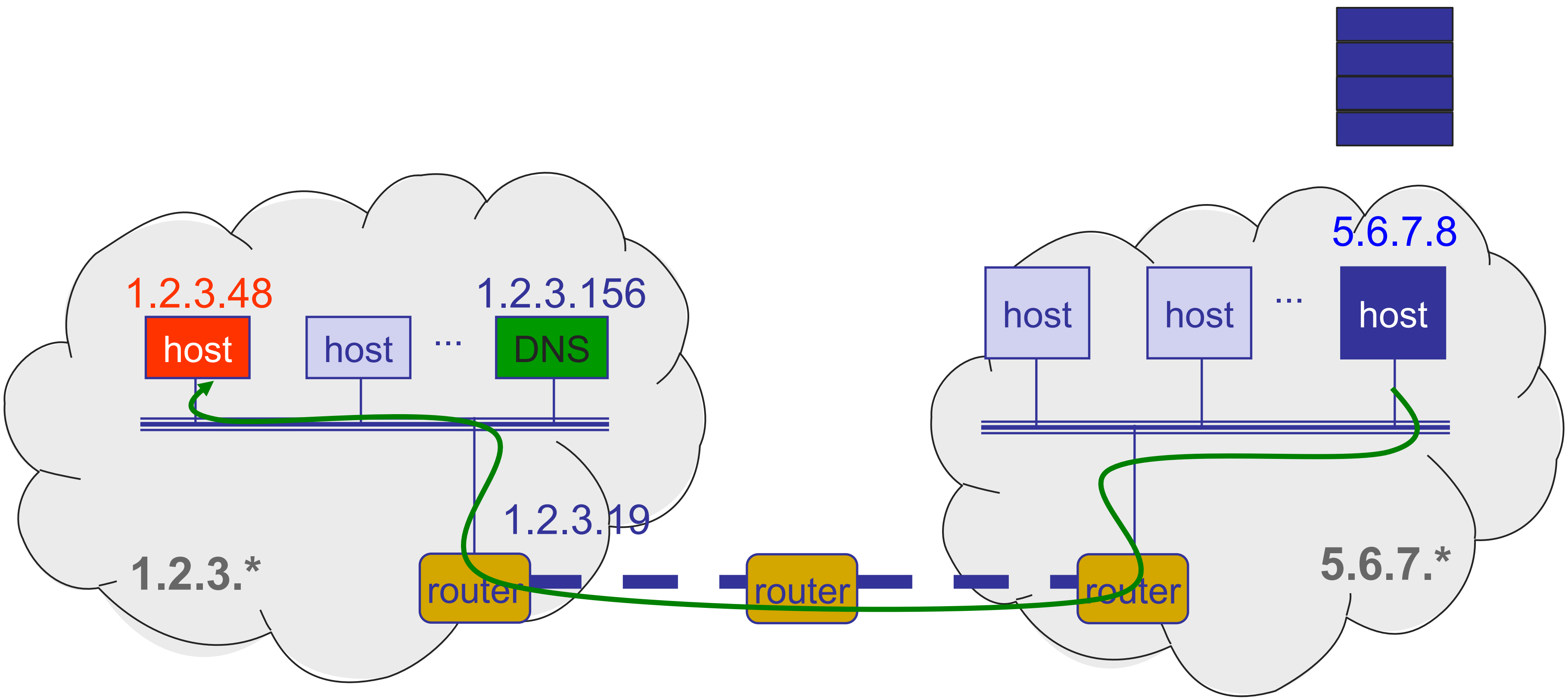
We only consider store-and-forward in this class

What we want

(Some of) What happens...



(More of) What happens



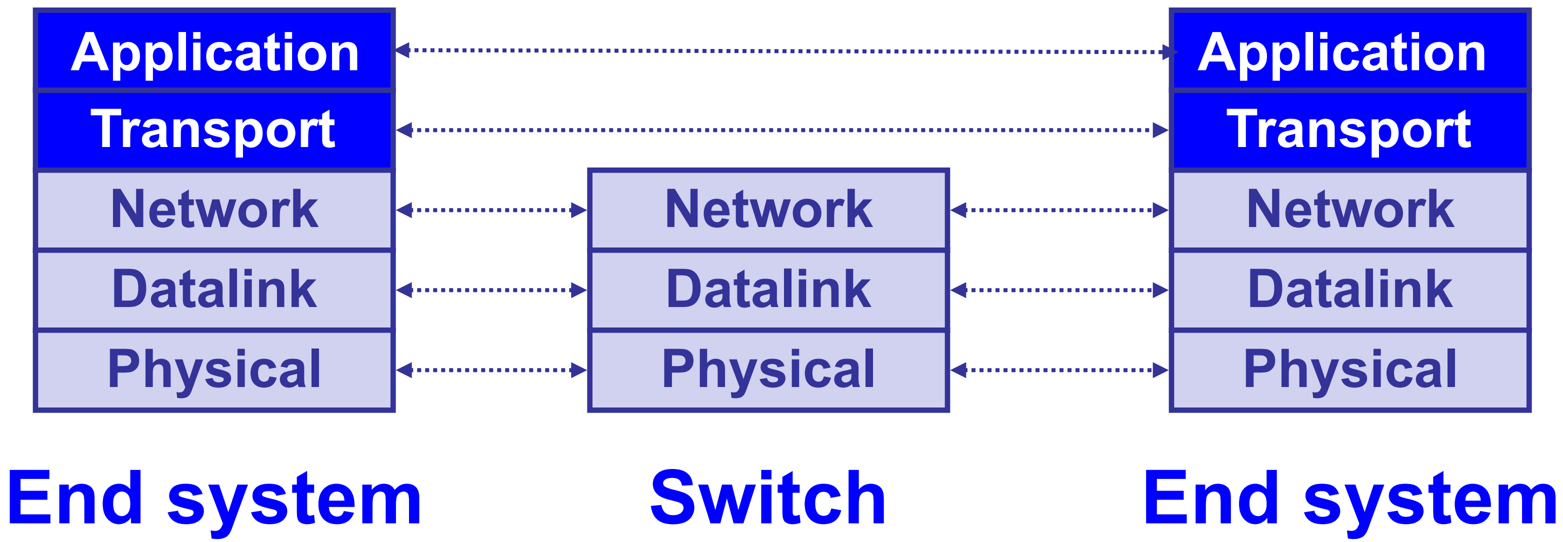
What we get

Layers

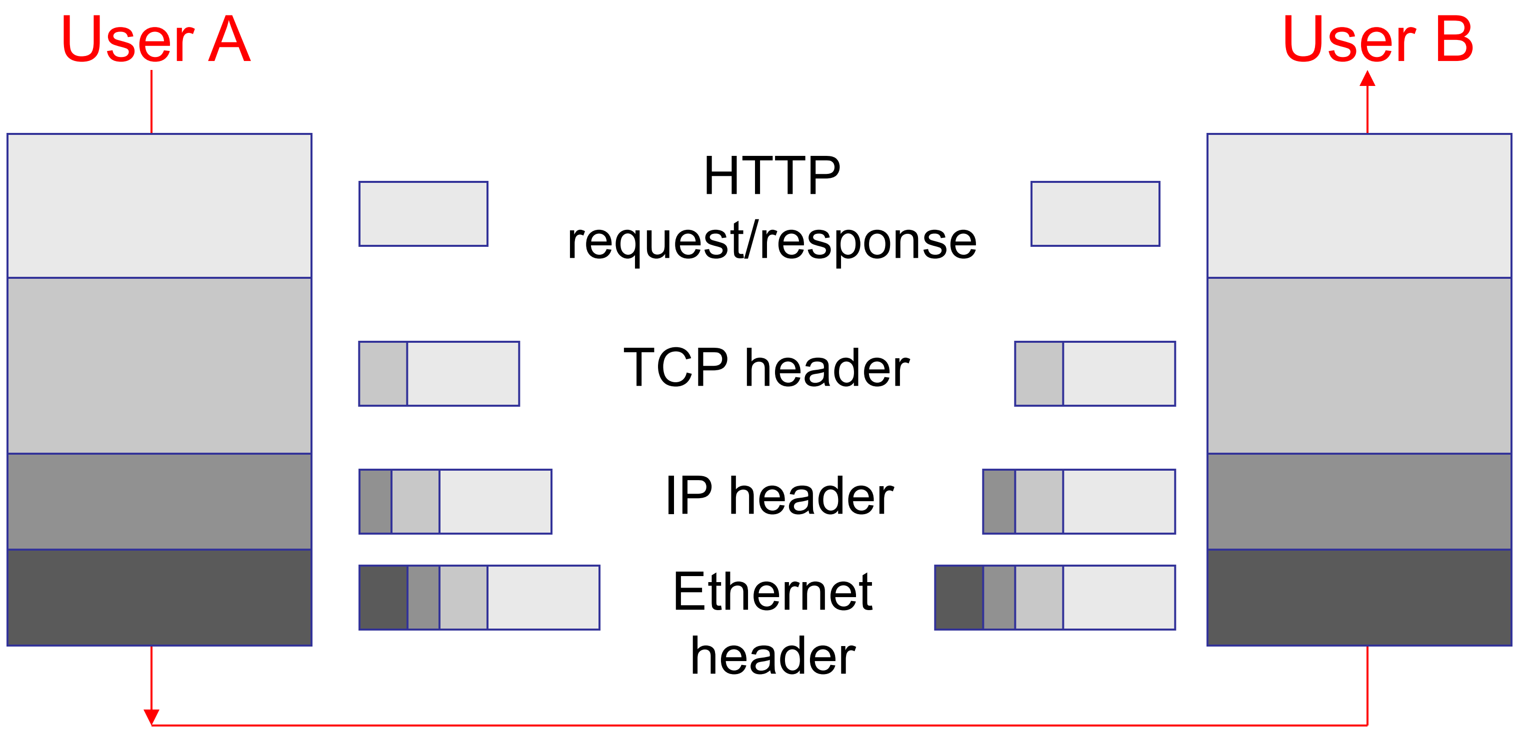
- Layer: a part of a system with well-defined interfaces to other parts
- One layer interacts only with layer above and layer below
- Two layers interact only through the interface between them

Layers in practice

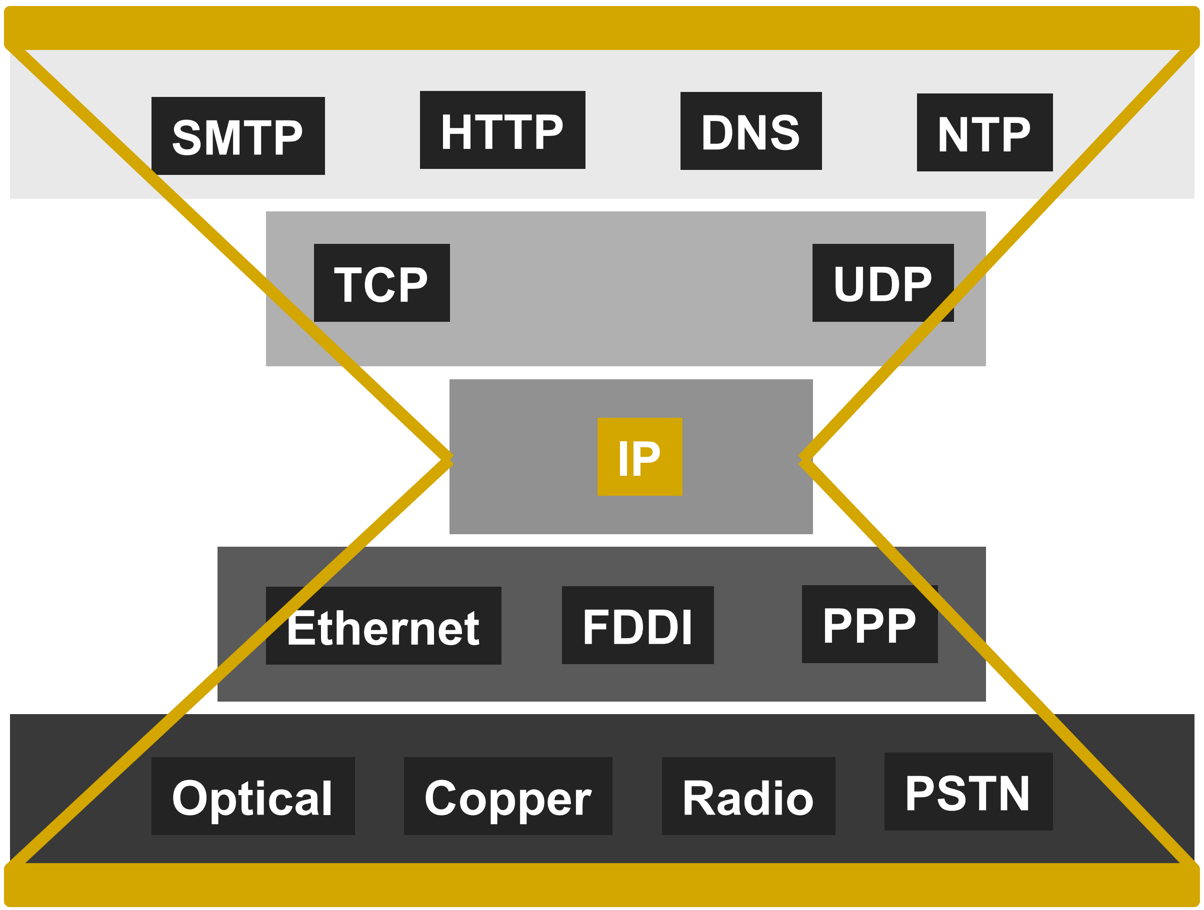
- Lower three layers implemented everywhere
- Top two layers implemented only at hosts



Layer encapsulation: Protocol headers



IP is the narrow waist of the layering hourglass



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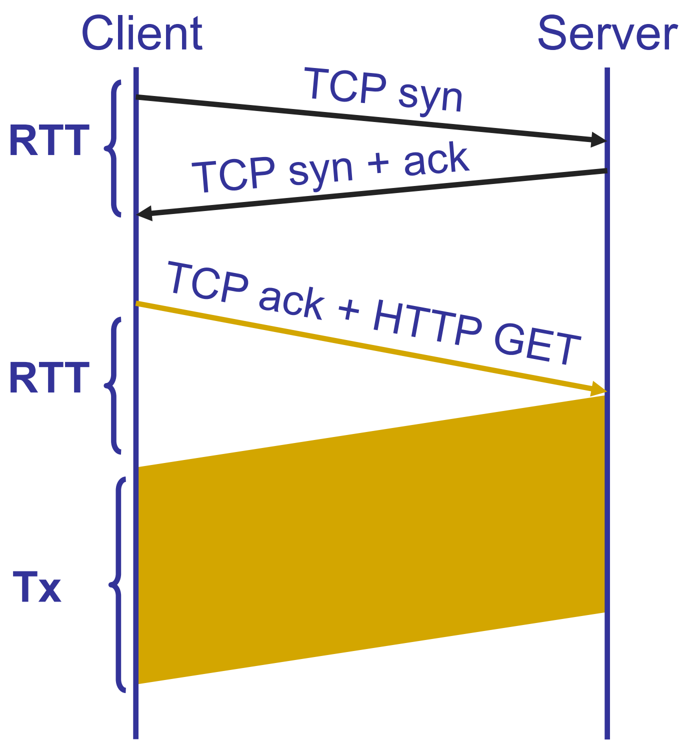
Hyper Text Transfer Protocol (HTTP)

- Client-server architecture
  - Server is “always on” and “well known”
  - Clients initiate contact to server
- Synchronous request/reply protocol
  - Runs over TCP, Port 80
- Stateless
- ASCII format
  - Before HTTP/2

Object request response time

- RTT (round-trip time)
  - Time for a small packet to travel from client to server and back
- Response time
  - 1 RTT for TCP setup
  - 1 RTT for HTTP request and first few bytes
  - Transmission time
  - Total = 2RTT + Transmission Time



Improving HTTP performance

- Optimizing connections using three "P"s
  - Persistent connections
  - Parallel/concurrent connections
  - Pipelined transfers over the same connection
- Caching
  - Forward proxy: close to clients
  - Reverse proxy: close to servers
- Replication

Scorecard: Getting n small objects

- Time dominated by latency

- One-at-a-time: ~2n RTT
- m concurrent: ~2[n/m] RTT
- Persistent: ~ (n+1) RTT
- Pipelined: ~2 RTT
- Pipelined and Persistent: ~2 RTT first time; RTT later for another n from the same site

Scorecard: Getting n large objects each of size F

- Time dominated by TCP throughput B_C (<= B_L), where bottleneck link bandwidth is B_L
  - Assuming all TCP connections go through the same bottleneck link
- One-at-a-time: ~ nF/B_C
- m concurrent: ~ $nF/(mB'_C)$
  - Assuming each TCP connection gets the same throughput (B'_C), where $mB'_C <= B_L$
- Pipelined and/or persistent: ~ nF/B_C
  - The only thing that helps is higher throughput

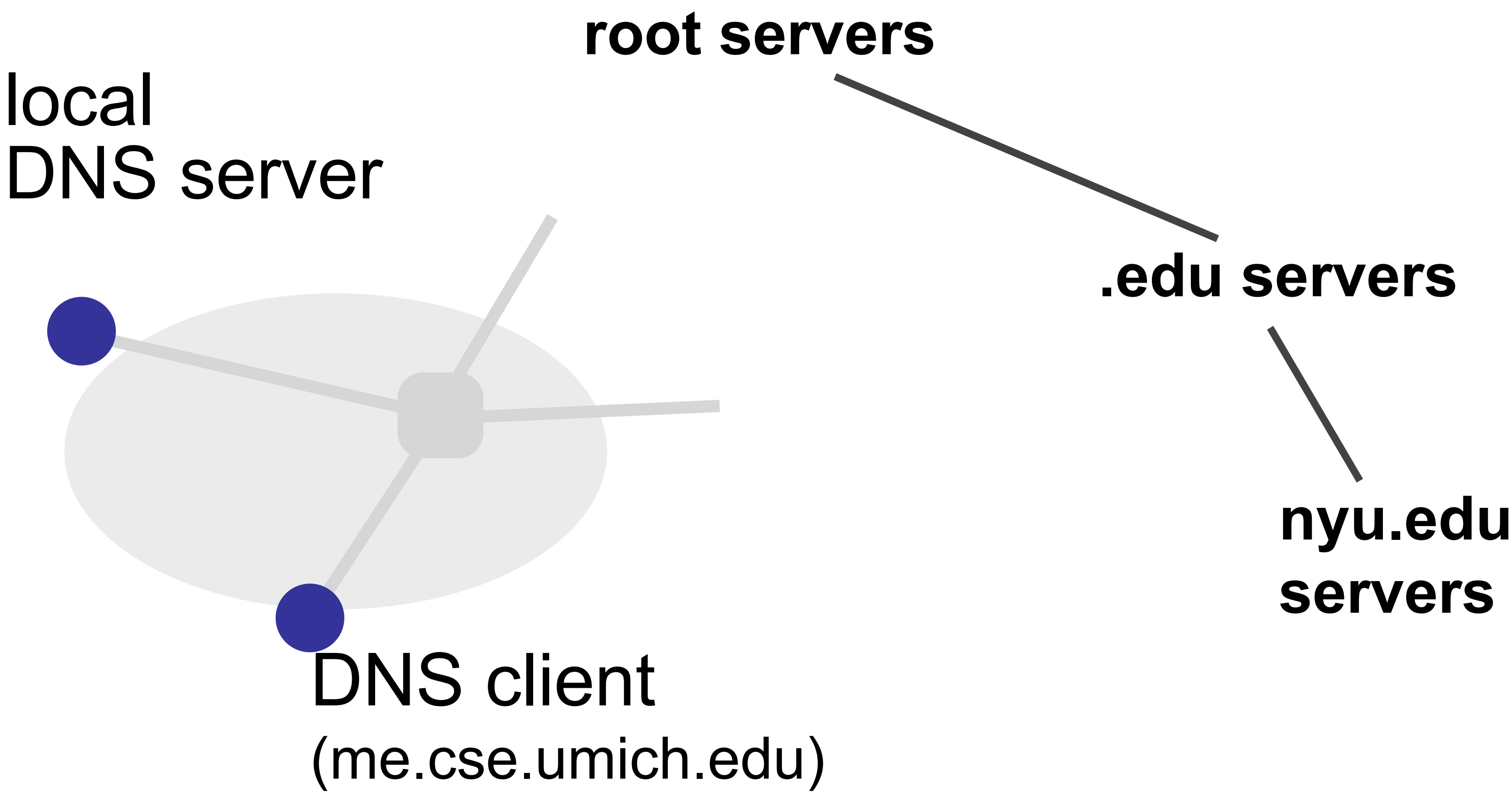
Content Distribution Networks (CDN)

- Caching and replication as a service
- Combination of caching and replication
  - Pull: Direct result of clients' requests (caching)
  - Push: Expectation of high access rate (replication)

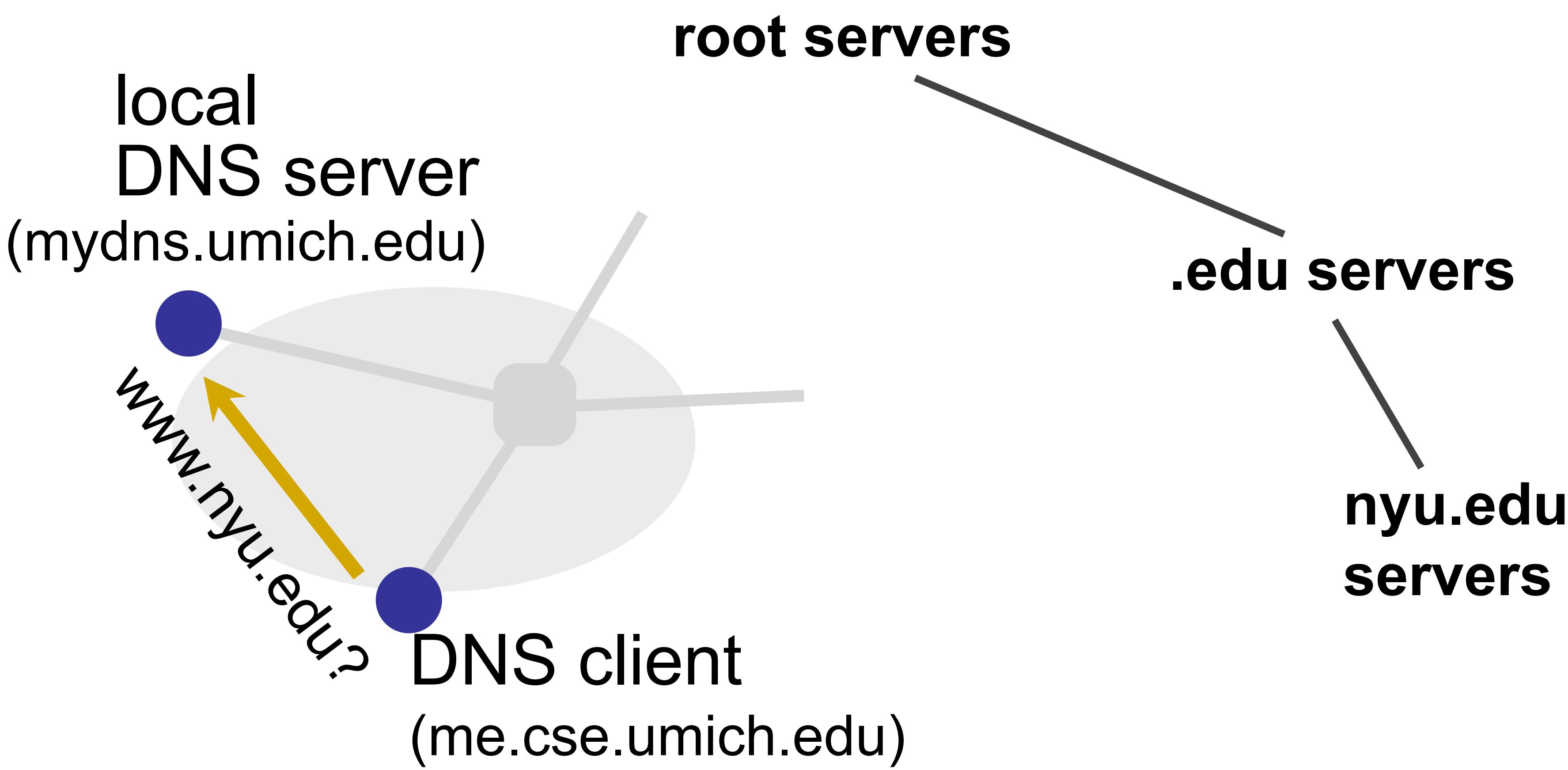
Hierarchies in the DNS

- Three intertwined hierarchies
  - Hierarchical namespace
    - As opposed to flat namespace
  - Hierarchically administered
    - As opposed to centralized
  - (Distributed) hierarchy of servers
    - As opposed to centralized storage

Name resolution



Name resolution



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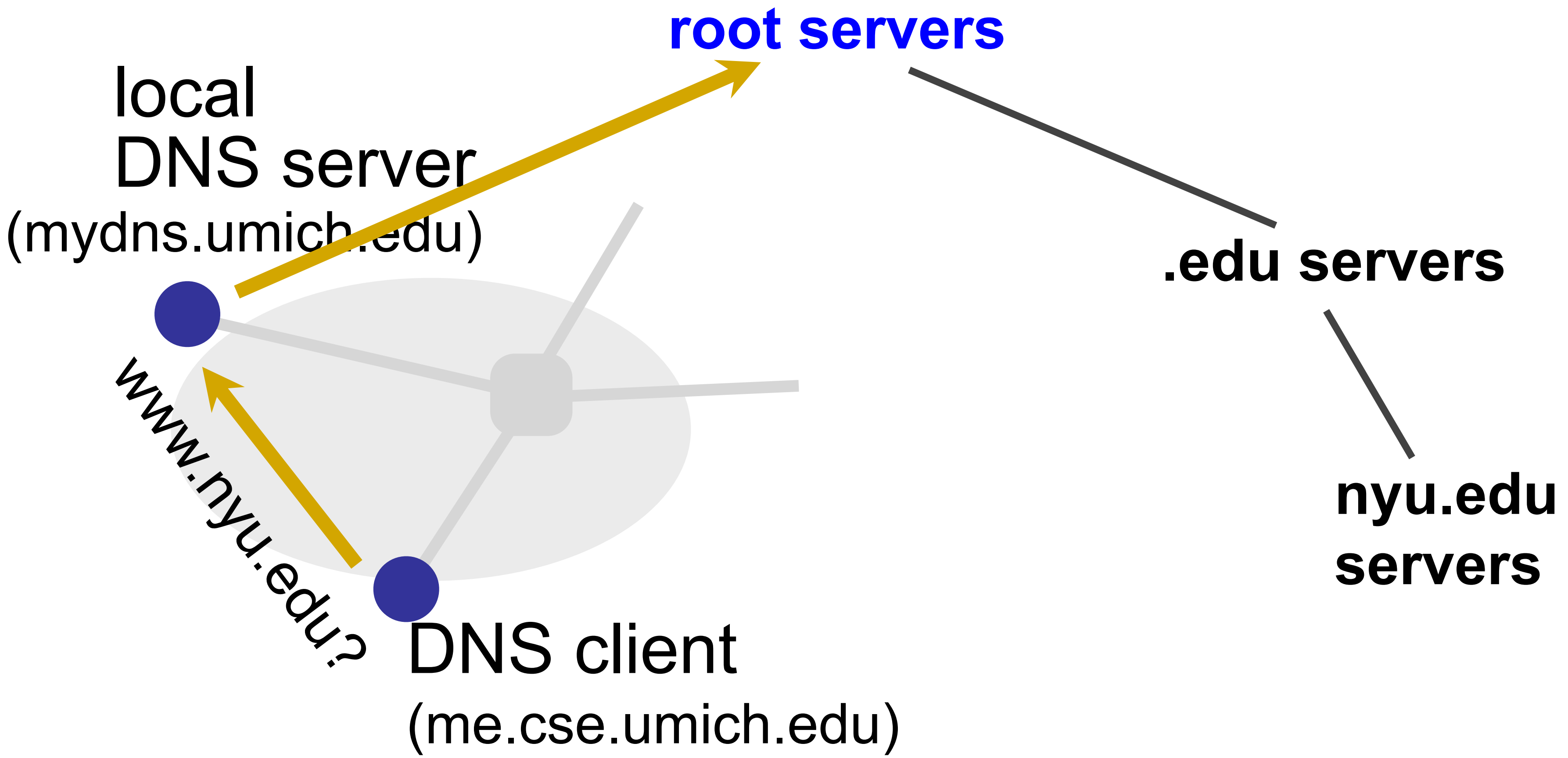
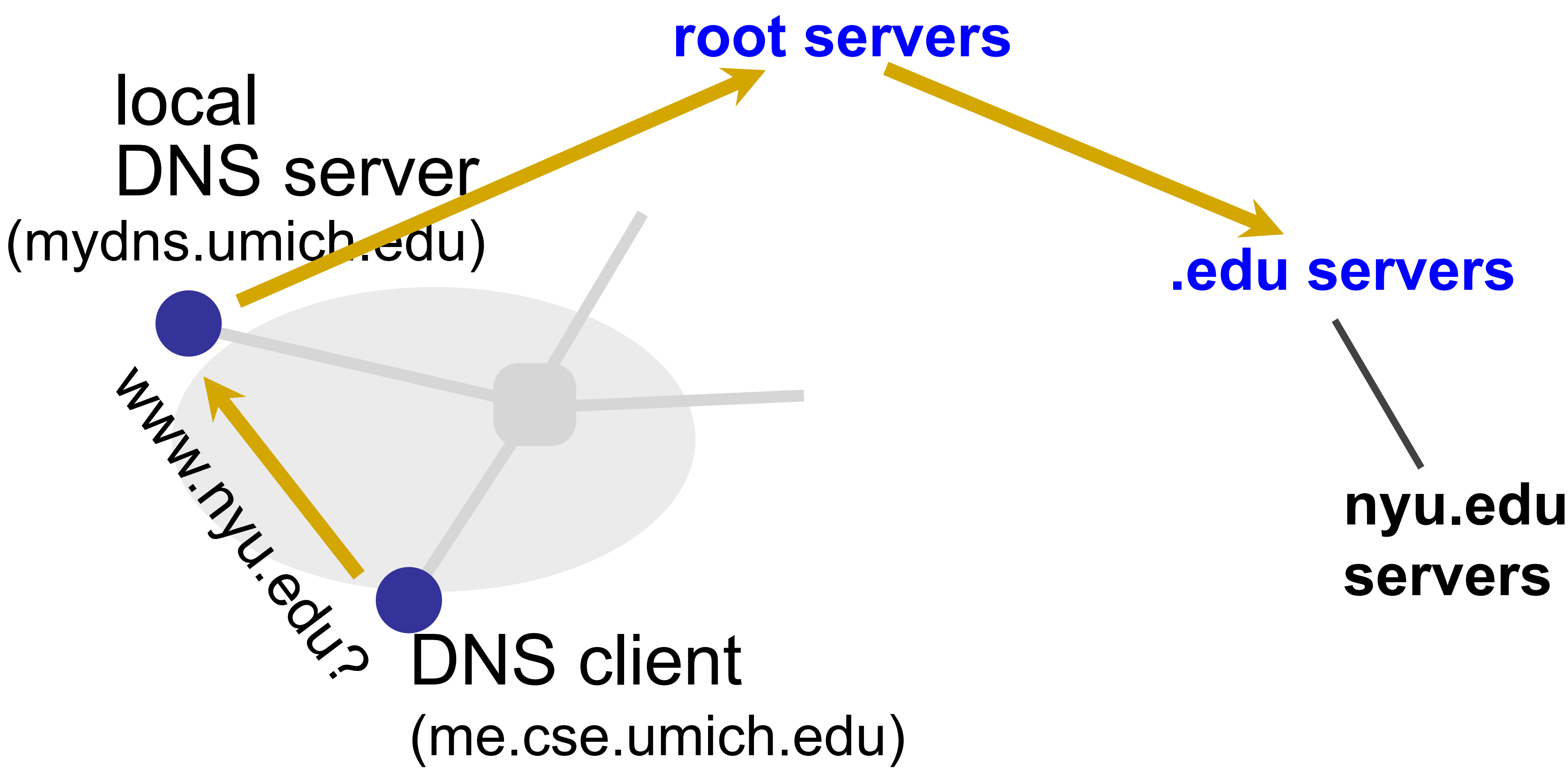
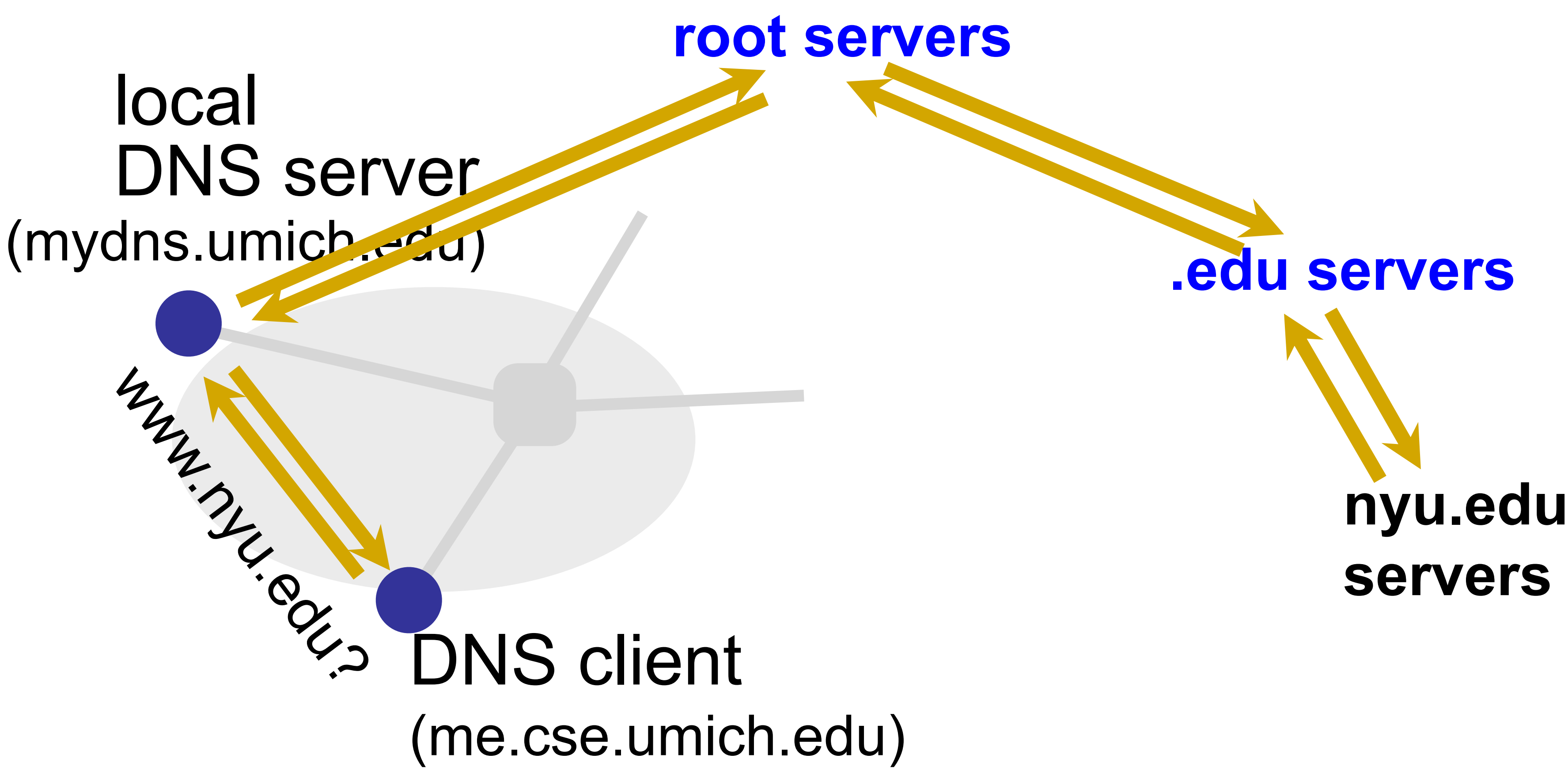


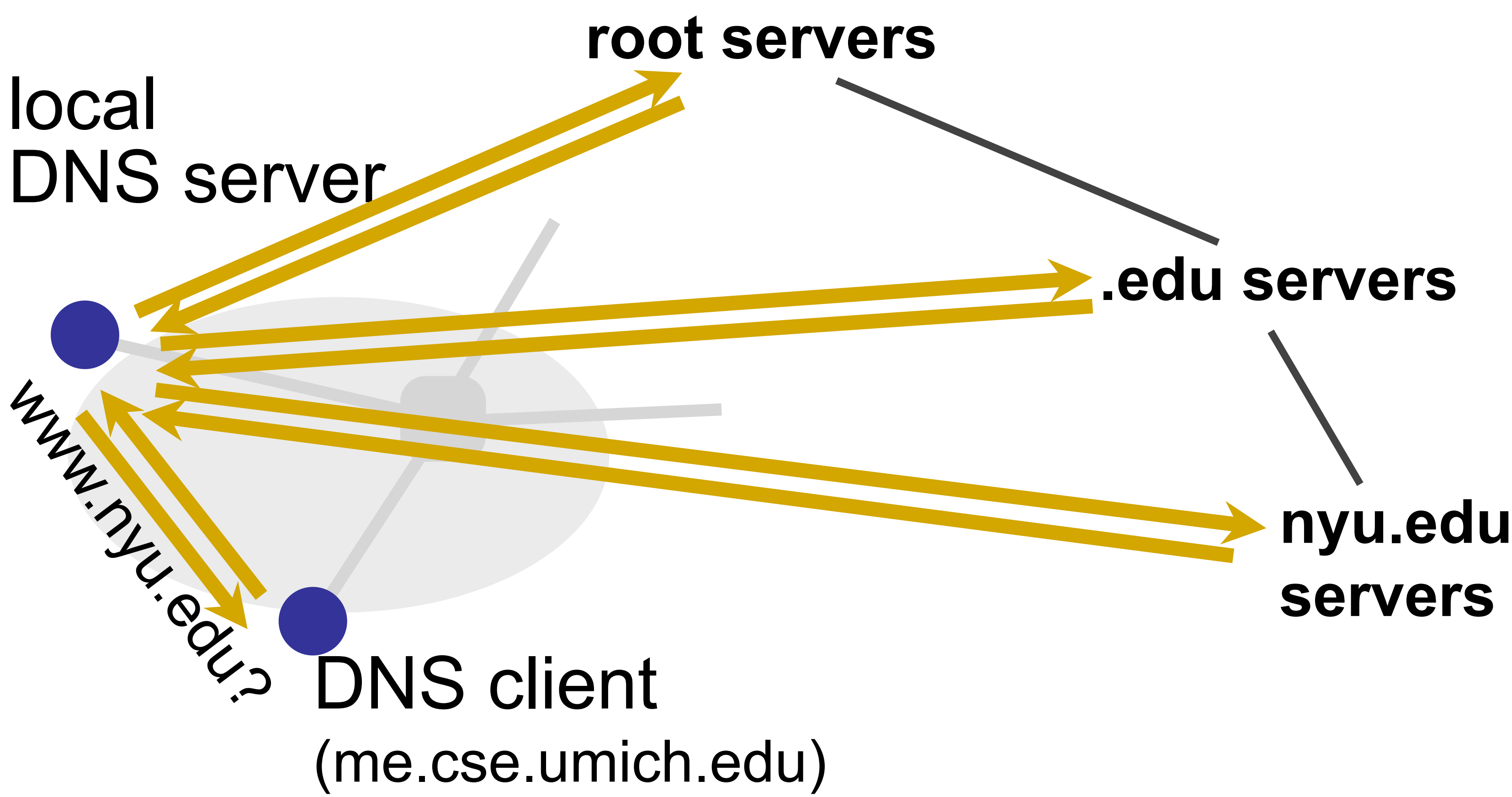
Name resolution



Name resolution: Recursive



Name resolution: Iterative



DNS caching

- Performing all these queries takes time
  - Up to 1-second latency before starting download
- Caching can greatly reduce overhead
  - The top-level servers very rarely change
  - Popular sites (e.g., www.google.com) visited often
  - Local DNS server often has the information cached
- How DNS caching works
  - DNS servers cache responses to queries
  - Responses include a "time to live" (TTL) field
  - Server deletes cached entry after TTL expires

HTTP streaming

- Video is stored at an HTTP server with a URL
- Clients send a GET request for the URL
- Server sends the video file as a stream
- Client first buffers for a while to minimize interruptions later
- Once the buffer reaches a threshold
  - The video plays in the foreground
  - More frames are downloaded in the background

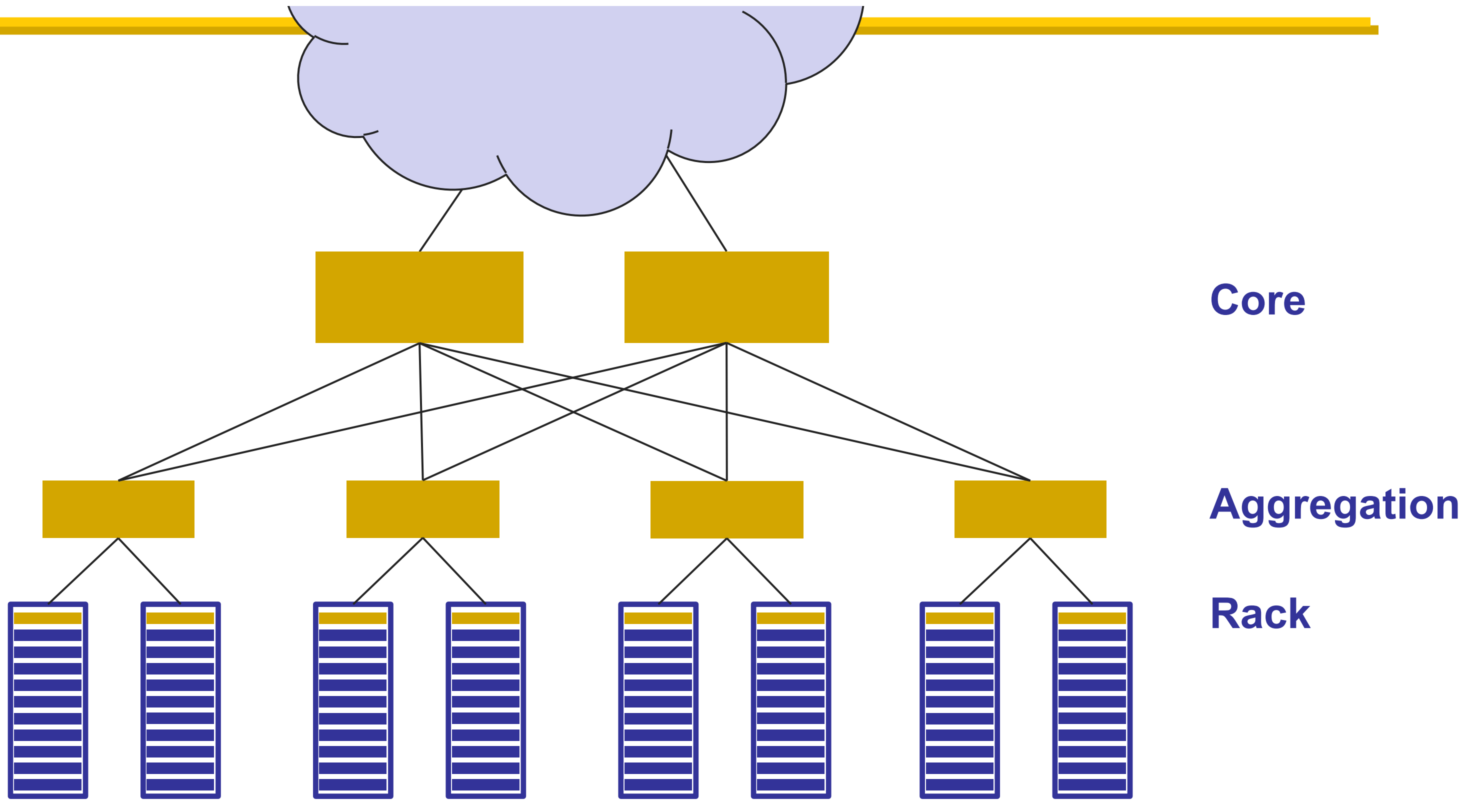
DASH : Dynamic Adaptive Streaming over HTTP

- Keep multiple resolutions of the same video
  - Stored in a manifest file in the HTTP server
- Client asks for the manifest file first to learn about the options
- Asks for chunks at a time and measures available bandwidth while they are downloaded
  - Low bandwidth $\Rightarrow$ switch to lower bitrate
  - High bandwidth $\Rightarrow$ switch to higher bitrate

Applications

- Common theme: parallelism
  - Applications decomposed into tasks
  - Running in parallel on different machines
- Two common paradigms
  - Partition-Aggregate
  - Map-Reduce

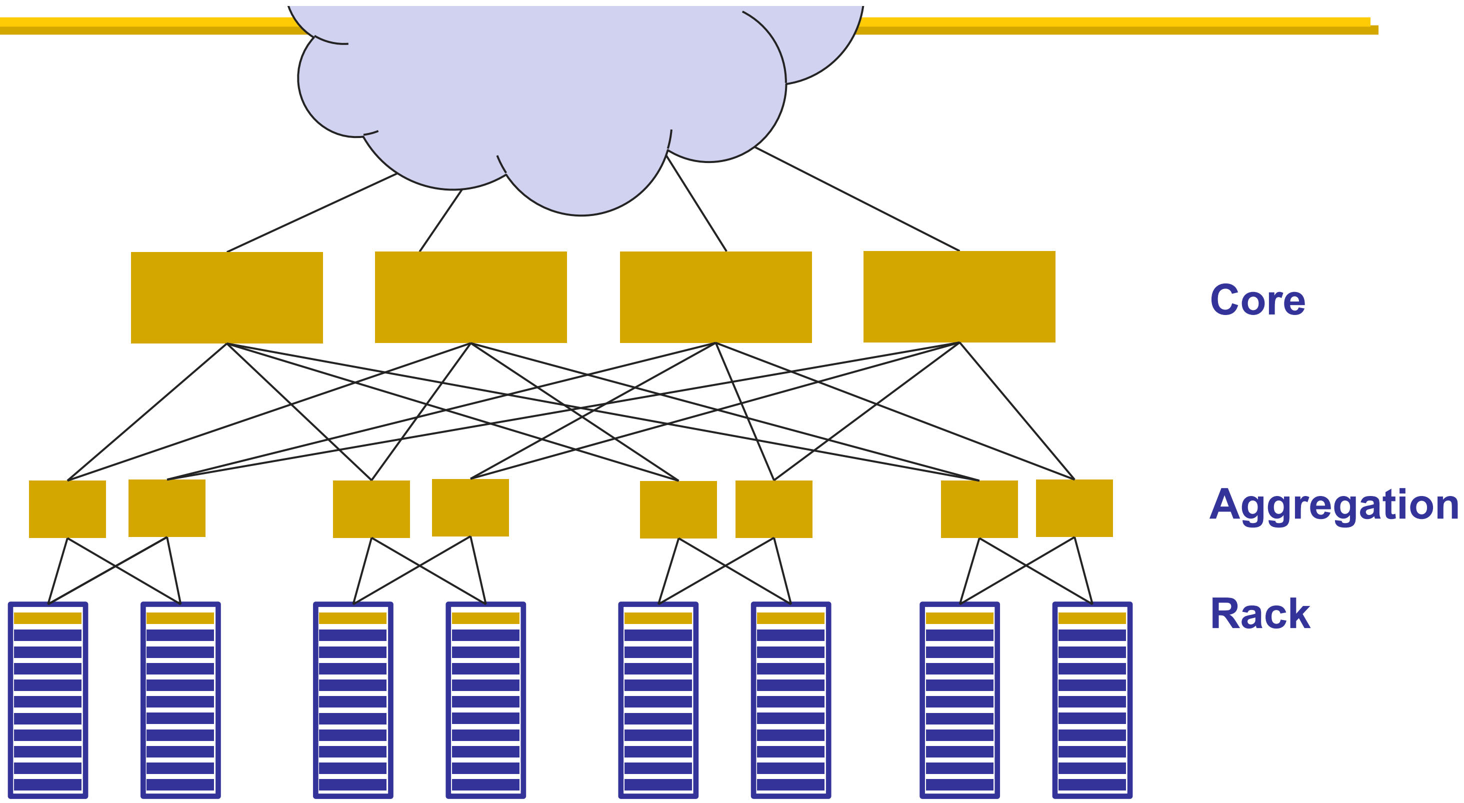
Traditional datacenter networks



Challenges

- Not enough bandwidth
  - Oversubscription: Less bandwidth in the ToR-Agg links than all the servers' bandwidth in the rack
  - Oversubscription ratio: Ratio between bandwidth underneath and bandwidth above
- Not enough paths between server pairs
  - Load balancing issues
  - Failure recovery issues

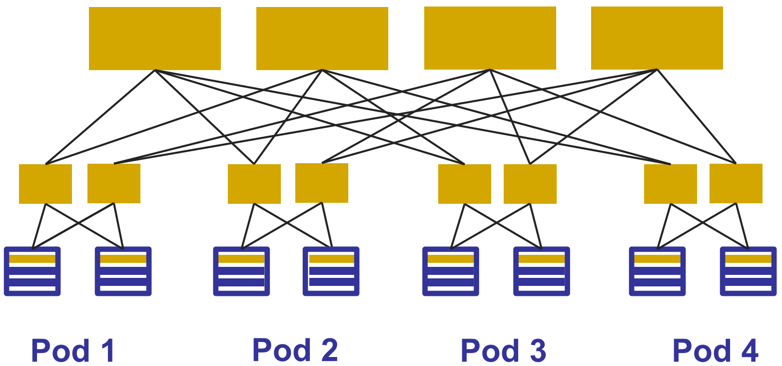
Modern datacenter networks: More bandwidth, more paths



Clos topology

- Multi-stage network
- k pods, where each pod has two layers of k/2 switches
  - k/2 ports up and k/2 down
- All links have the same b/w
- At most $k^3/4$ machines

- Example
  - k = 4
  - 16 machines
- For k=48, 27648 machines



5-MINUTE BREAK!

Announcements

- 60-minute midterm exam on canvas
  - **Mar 6**
  - **Available from 9:30AM**
  - **You MUST start before 11AM**
- Please fill up midterm teaching evaluation

Topics

- Basics (lectures 1–2)
- Application layer (lectures 3–5)
  - HTTP, DNS, CDN, Video Streaming, and Cloud
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  - Overview
  - Data plane

Role of the transport layer

- (1) Communication between application processes
  - Mux and demux from/to application processes
  - Implemented using ports
- (2) Provide common end-to-end services for app layer
  - Reliable, in-order data delivery
  - Well-paced data delivery

UDP vs. TCP

- Both UDP and TCP perform mux/demux via ports

	UDP	TCP
Data abstraction	Packets (datagrams)	Stream of bytes of arbitrary length
Service	Best-effort (same as IP)	•Reliability •In-order delivery •Congestion control •Flow control

Reliable transport: General concepts

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments (feedback from receiver)
  - Cumulative: “received everything up to X”
  - Selective: “received X”
- Sequence no (detect duplicates, accounting)
- Sliding windows (for efficiency)

You should know:

- what these concepts are
- why they exist
- how TCP uses them

Designing a reliable transport protocol

- Stop and wait is correct but inefficient
  - Works packet by packet (of size DATA)
  - Throughput is (DATA/ RTT)
- Sliding window: use pipelining to increase throughput
  - n packets at a time results in higher throughput
  - MIN(n*DATA/RTT, Link Bandwidth)

The TCP abstraction

- TCP delivers a reliable, in-order, byte stream
- Reliable: TCP resends lost packets (recursively)
  - Until it gives up and shuts down connection
- In-order: TCP only hands consecutive chunks of data to application
- Byte stream: TCP assumes there is an incoming stream of data, and attempts to deliver it to app

Things to know about TCP

- How TCP achieves reliability
- RTT estimation
- Connection establishment/teardown
- Flow Control
- Congestion Control (concepts only)

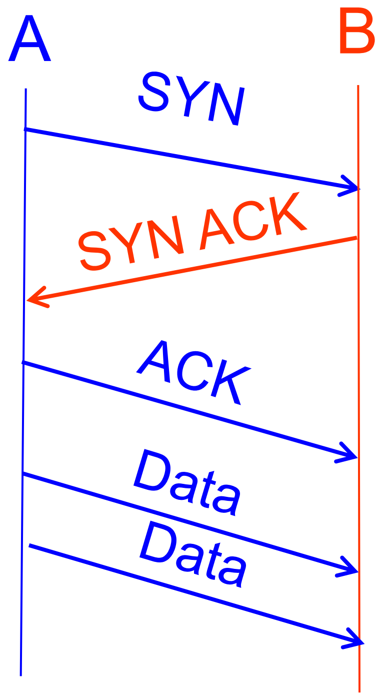
- For each, know how the functionality is implemented and why it is needed

Reliability

- Having TCP take care of it simplifies application development
- How
  - Checksums and timers (for error and loss detection)
  - Fast retransmit (to detect faster-than-timeout loss)
  - Cumulative ACKs (receiver feedback: what’s lost?)
  - Sliding windows (for efficiency)
  - Buffers at sender (hold packets until ACKs arrive)
  - Buffers at receiver (to reorder packets before delivery to application)

Establishing/terminating a TCP connection

- Three-way handshake to establish connection
  - Host A sends a SYN (open; "synchronize sequence numbers") to host B
  - Host B returns a SYN acknowledgment (SYN ACK)
  - Host A sends an ACK to acknowledge the SYN ACK
- Three-way handshake to terminate (normal operation)



Flow control

- Why?
  - TCP at the receiver must buffer a packet until all packets before it (in byte-order) have arrived and the receiving application has consumed available bytes
  - Hence, receiver advances its window when the receiving application consumes data
  - Sender advances its window when new data ACK'd
  - Risk of sender overrunning the receiver's buffers
- How?
  - "Advertised Window" field in TCP header

Congestion control

- Why?
  - Because the network itself can be the bottleneck
  - Should make efficient use of available network capacity
    - While sharing available capacity fairly with other flows
    - And adapting to changes in available capacity
- How?
  - Dynamically adapts the size of the sending window

Put together

- Flow Control
  - Restrict window to RWND to make sure that the receiver isn't overwhelmed
- Congestion Control
  - Restrict window to CWND to make sure that the network isn't overwhelmed
- Together
  - Restrict window to min{RWND, CWND} to make sure that neither the receiver nor the network are overwhelmed

CC implementation

- States at sender
  - CWND (initialized to a small constant)
  - ssthresh (initialized to a large constant)
  - dupACKcount and timer
- Events
  - ACK (new data)
  - dupACK (duplicate ACK for old data)
  - Timeout

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

- *CWND packets per RTT*
- *Hence, after one RTT with no drops:*
  CWND = 2xCWND

Event: ACK (new data)

- If CWND < ssthresh
  - CWND += 1

Slow start **phase**

- Else
  - CWND = CWND + 1/CWND

Congestion avoidance **phase**

- *CWND packets per RTT*
- *Hence, after one RTT with no drops:*

  CWND = CWND + 1

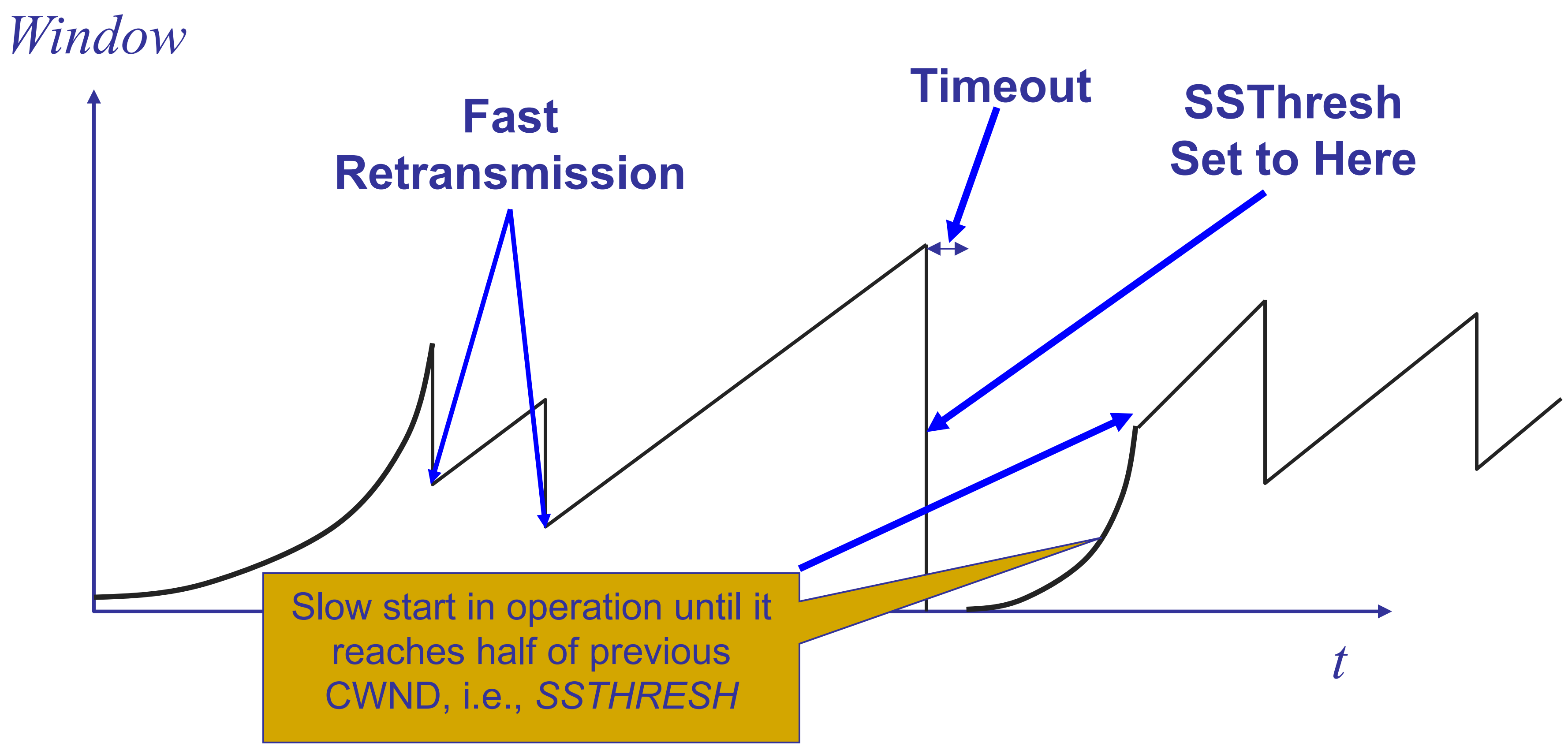
Event: TimeOut

- On Timeout
  - ssthresh ← CWND/2
  - CWND ← 1

Event: dupACK

- dupACKcount ++
- If dupACKcount = 3 /* fast retransmit */
  - ssthresh = CWND/2
  - CWND = CWND/2

Example

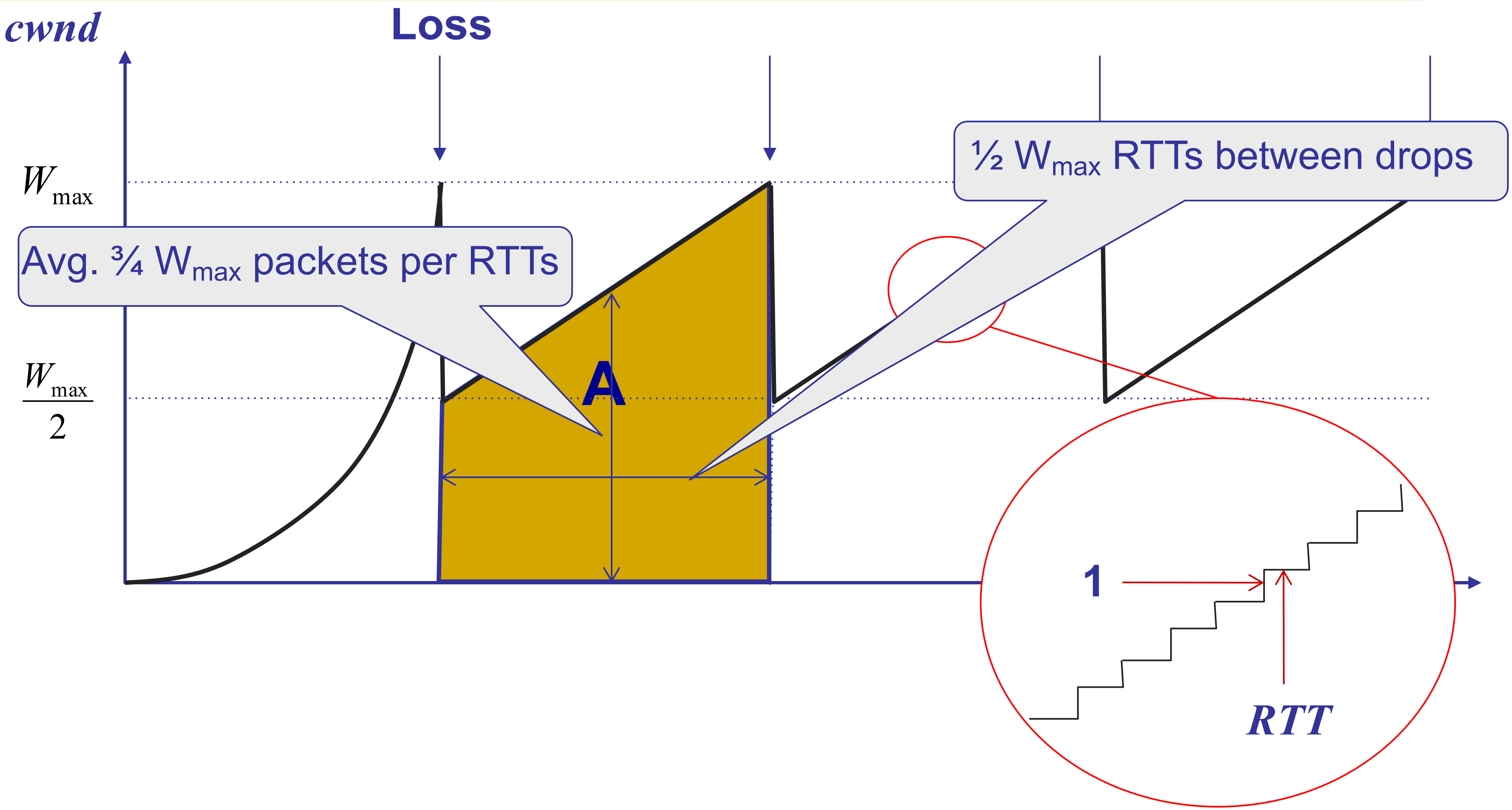


Slow-start restart: Go back to CWND = 1 MSS, but take advantage of knowing the previous value of CWND

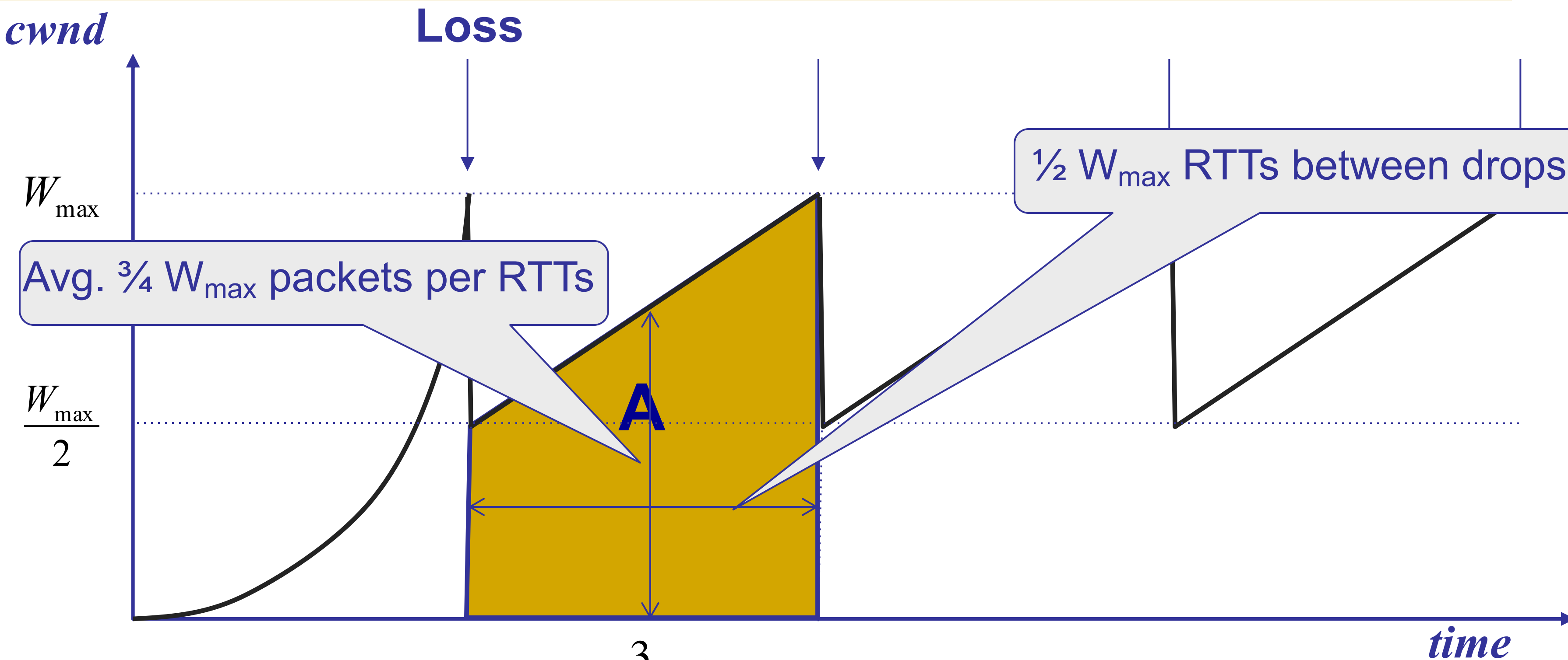
TCP flavors

- TCP-Tahoe
  - CWND =1 on 3 dupACKs
- TCP-Reno
  - CWND =1 on timeout
  - CWND = CWND/2 on 3 dupACKs
- TCP-newReno
  - TCP-Reno + improved fast recovery
- TCP-SACK
  - Incorporates selective acknowledgements

A simple model for TCP throughput



A simple model for TCP throughput



$$A = \frac{3}{8} W_{max}^2$$

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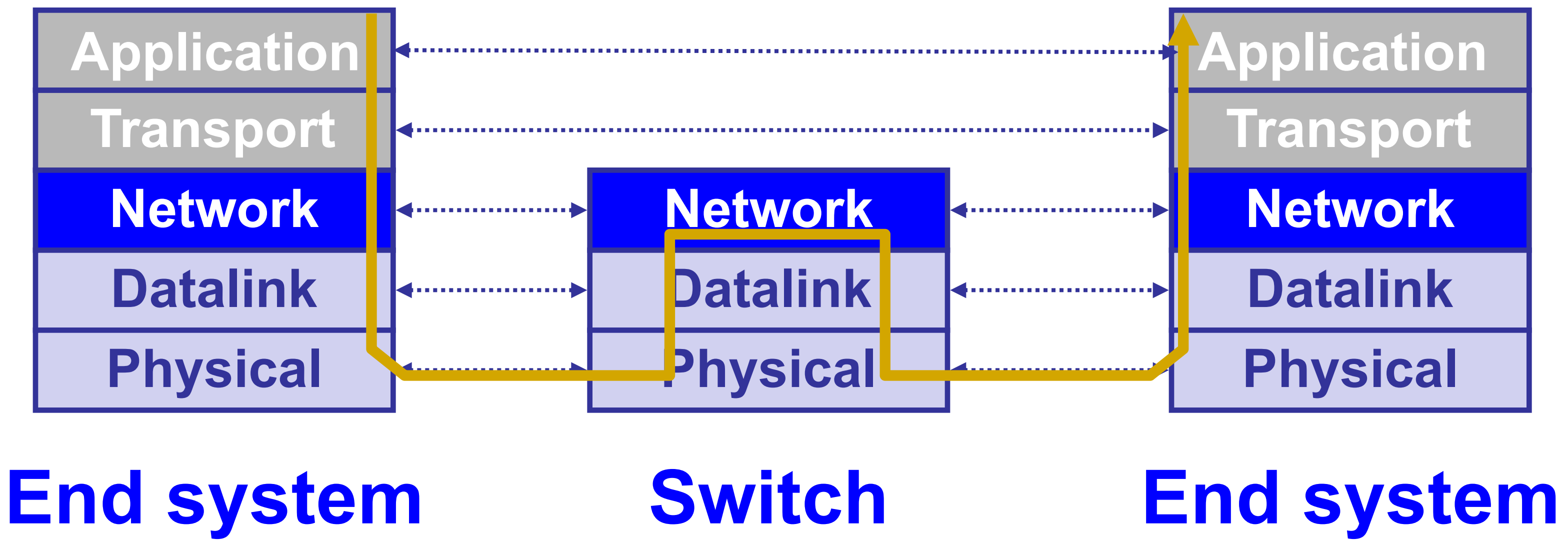
Network layer

- Present everywhere
- Performs addressing, forwarding, and routing, among other tasks



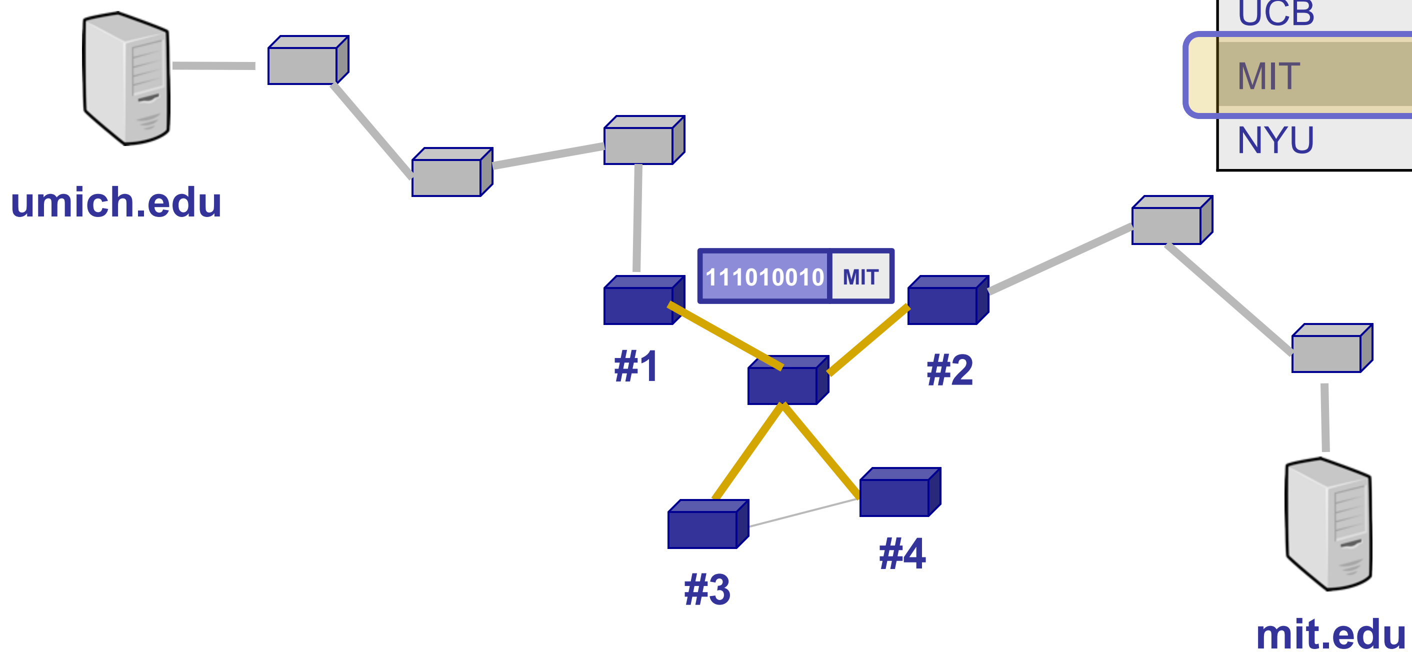
Forwarding vs. routing

- Forwarding: "data plane"
  - Directing one data packet
  - Each router using local routing state
- Routing: "control plane"
  - Computing the forwarding tables that guide packets
  - Jointly computed by routers using a distributed algorithm

Forwarding

Forwarding Table

Destination	Next Hop
UMICH	1
UCB	3
MIT	2
NYU	4



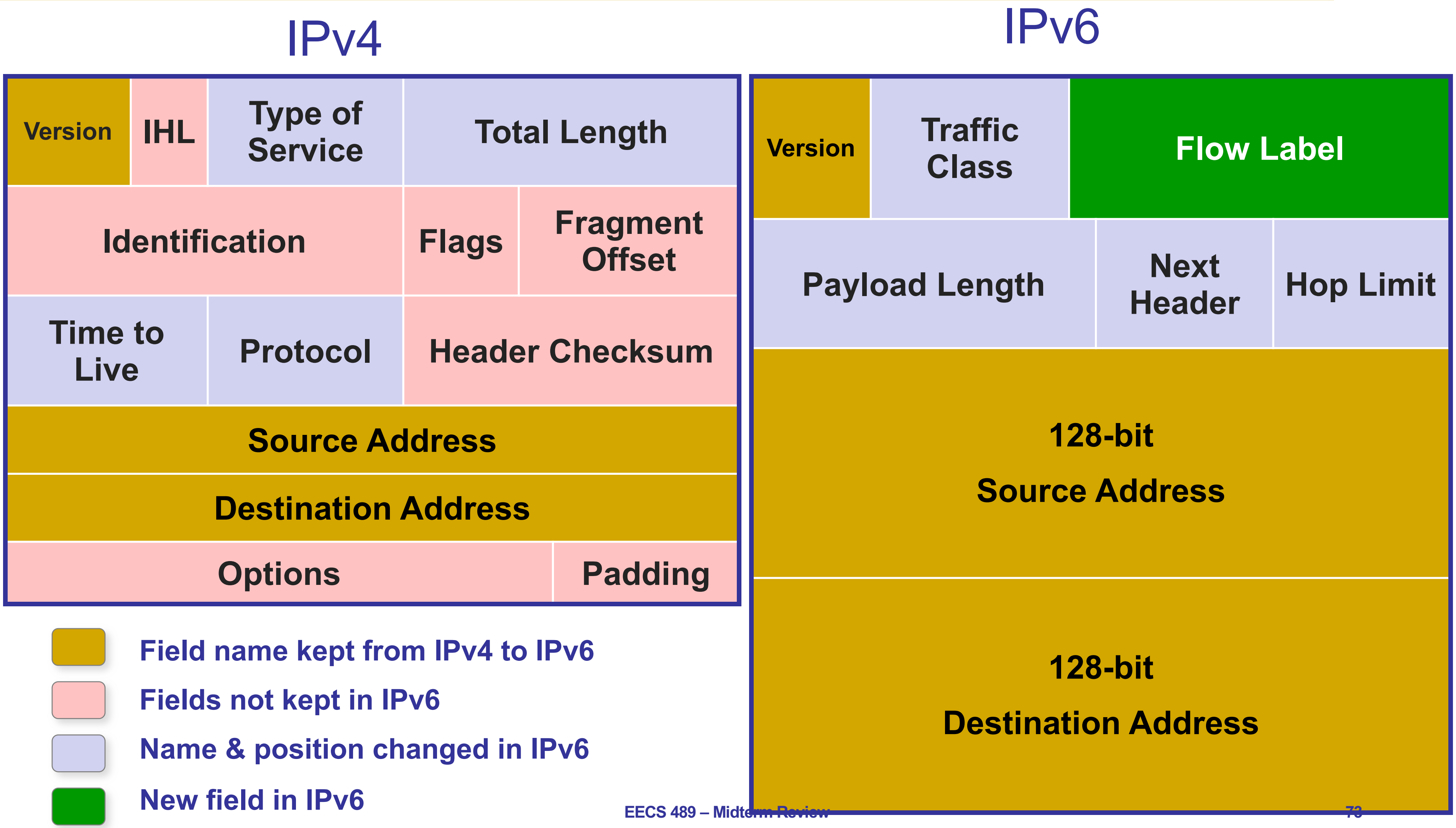
Designing the IP header

- Think of the IP header as an interface
  - Between the source and destination end-systems
  - Between the source and network (routers)
- Designing an interface
  - What task(s) are we trying to accomplish?
  - What information is needed to do it?
- Header reflects information needed for basic tasks

What information do we need?

- Parse packet
  - IP version number (4 bits), packet length (16 bits)
- Carry packet to the destination
  - Destination's IP address (32 bits)
- Deal with problems along the way
  - Loops: TTL (8 bits)
  - Corruption: checksum (16 bits)
  - Packet too large: fragmentation fields (32 bits)

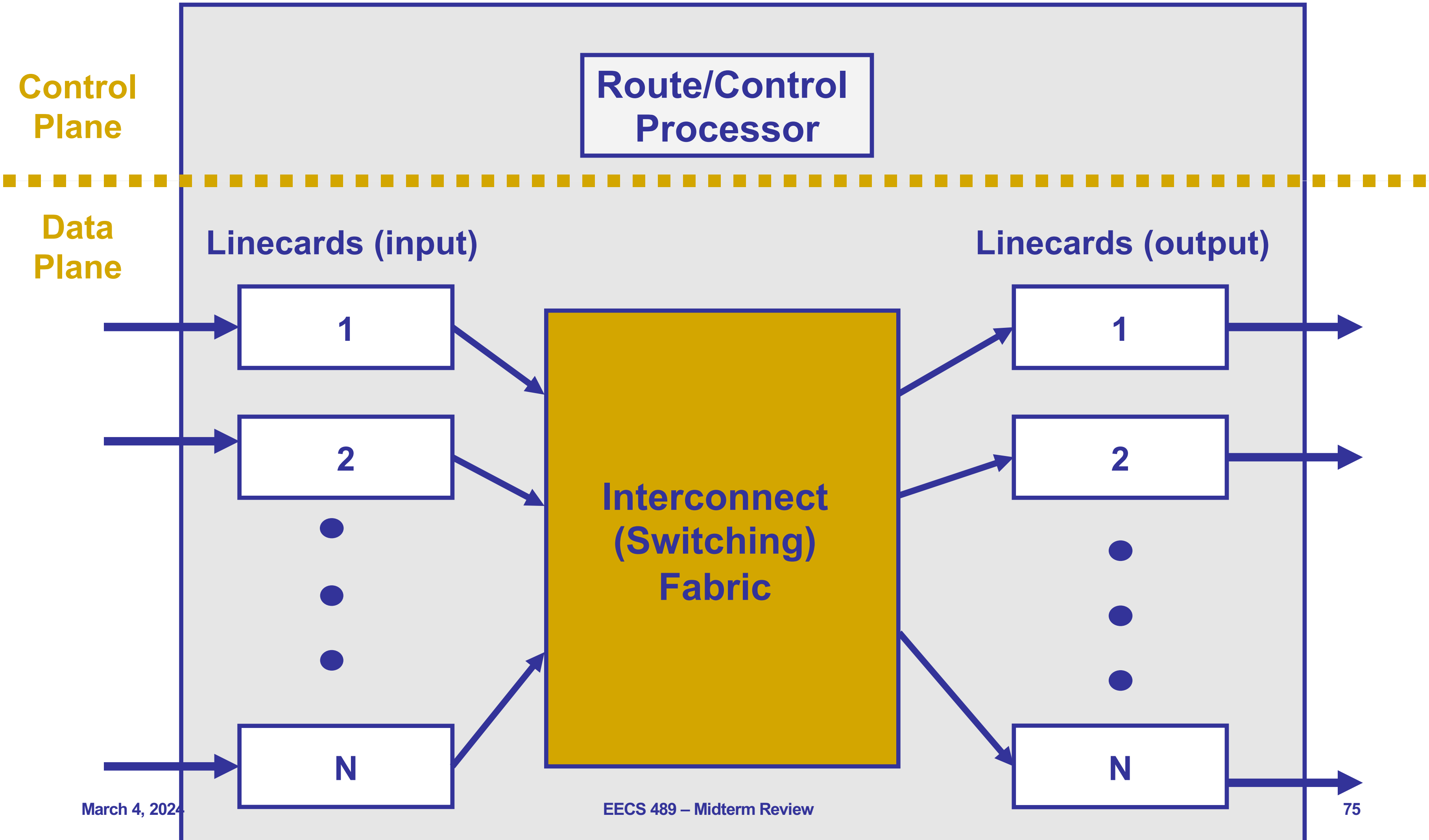
IPv4 and IPv6 header comparison



Philosophy of changes

- Don't deal with problems: leave to ends
  - Eliminated fragmentation and checksum
  - Why retain TTL?
- Simplify handling:
  - New options mechanism (uses next header)
  - Eliminated header length
    - Why couldn't IPv4 do this?
- Provide general flow label for packet
  - Not tied to semantics
  - Provides great flexibility

What's inside a router?



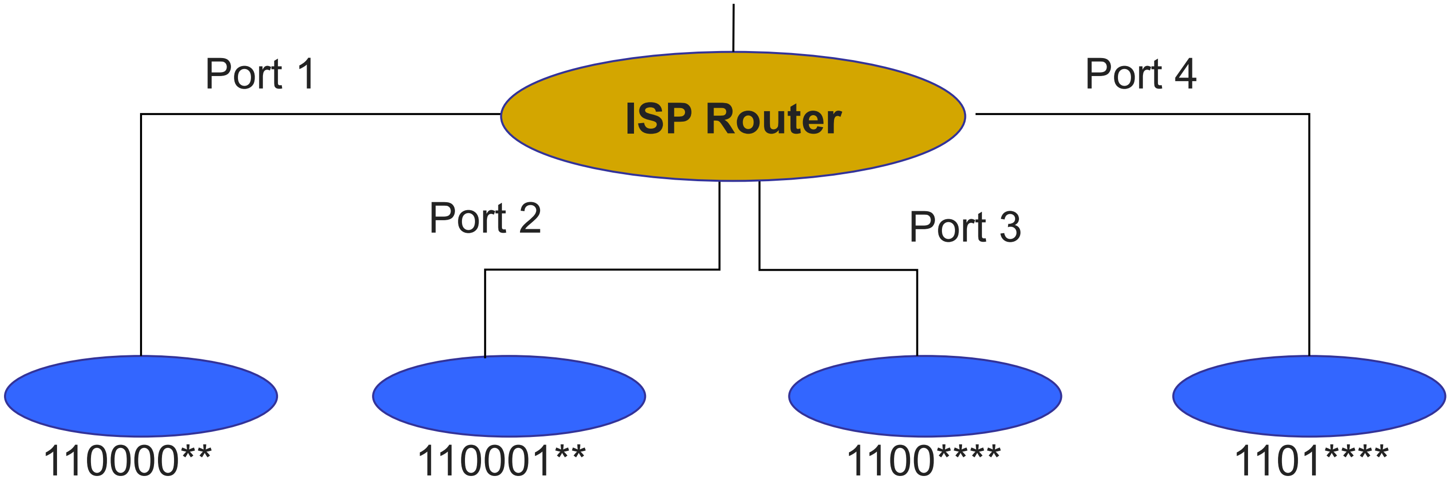
Input linecards

- Main challenge is processing speeds
- Tasks involved:
  - Update packet header (easy)
  - LPM lookup on destination address (harder)
- Mostly implemented with specialized hardware

Looking up the output port

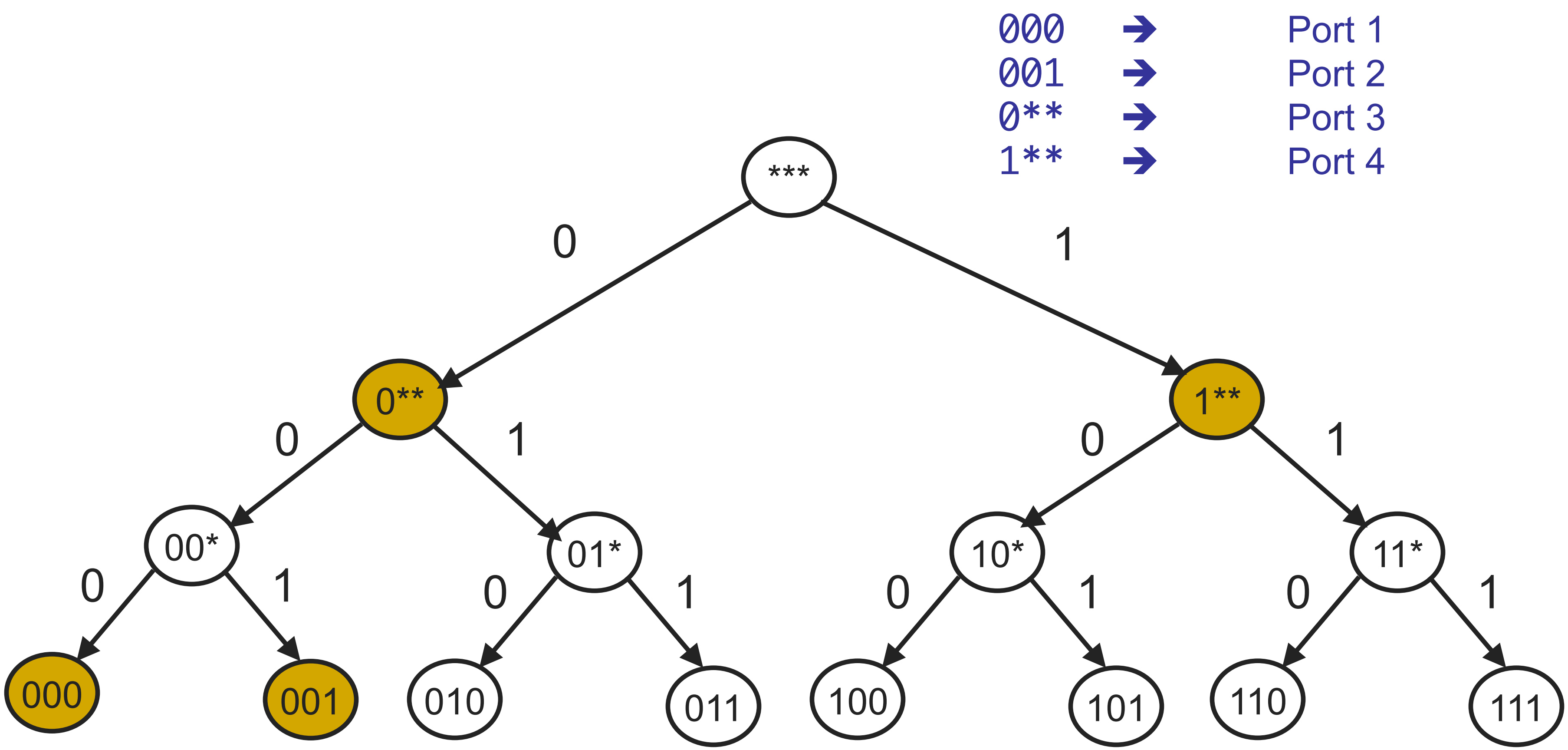
- One entry for each address → 4 billion entries!
- For scalability, addresses are aggregated

Longest prefix matching



Send to the port with the longest prefix match

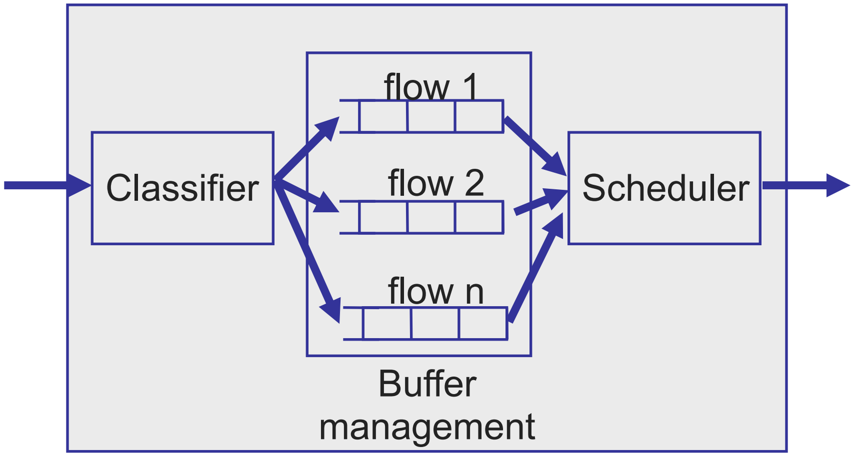
Tree structure



Record port associated with latest match, and only override when it matches another prefix during walk down tree

Output linecards

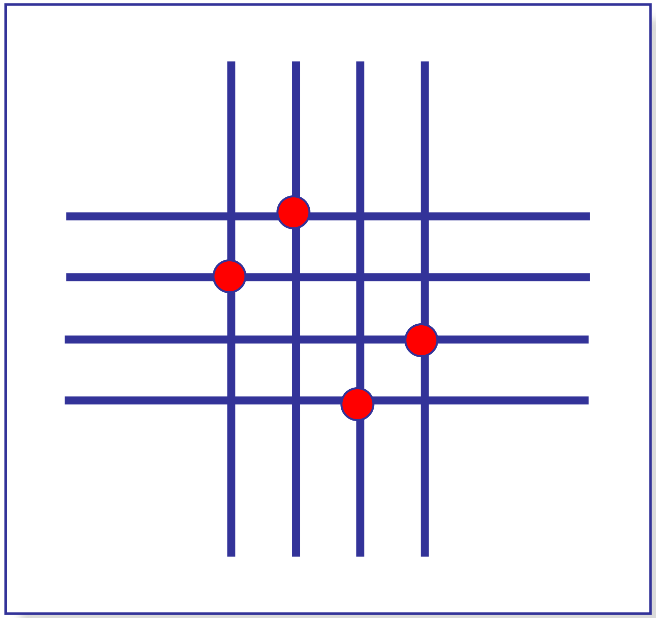
- Packet classification: map packets to flows
- Buffer management: decide when and which packet to drop
- Scheduler: decide when and which packet to transmit



Crossbar interconnect

- 2N buses intersecting with each other:
  - N input
  - N output
- Non-blocking

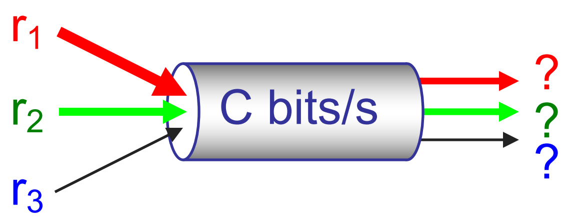
Input ports



Output ports

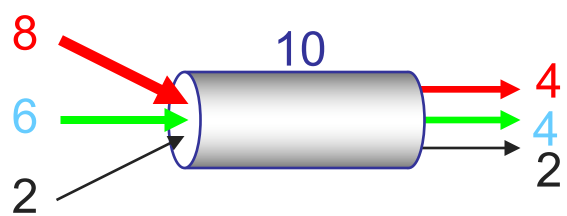
Max-Min fairness

- Given set of bandwidth demands r_i and total bandwidth C, max-min bandwidth allocations are:
  - $a_i = \min(f, r_i)$
  - where f is the unique value such that $\mathrm{Sum}(a_i) = C$



Example

- $C = 10$; $r_1 = 8$, $r_2 = 6$, $r_3 = 2$; $N = 3$
- $C/3 = 3.33 \rightarrow$
  - r_3 needs only 2
    - Can service all of r_3
  - Remove r_3 from the accounting: $C = C - r_3 = 8$; $N = 2$
- $C/2 = 4 \rightarrow$
  - Can't service all of r_1 or r_2
  - So hold them to the remaining fair share: $f = 4$



$f = 4$:
min(8, 4) = 4
min(6, 4) = 4
min(2, 4) = 2

Max-Min fairness

- Given set of bandwidth demands r_i and total bandwidth C, max-min bandwidth allocations are:
  - $a_i = \min(f, r_i)$
  - where f is the unique value such that $\mathrm{Sum}(a_i) = C$
- If you don't get full demand, no one gets more than you
- This is what round-robin service gives if all packets are the same size

Summary

- Demo Exam on Canvas