End-to-End Transport Over Wireless I: Preliminaries, Split Connection

COS 463: Wireless Networks
Lecture 2

Kyle Jamieson

[Various parts adapted from S. Das, B. Karp, N. Vaidya]

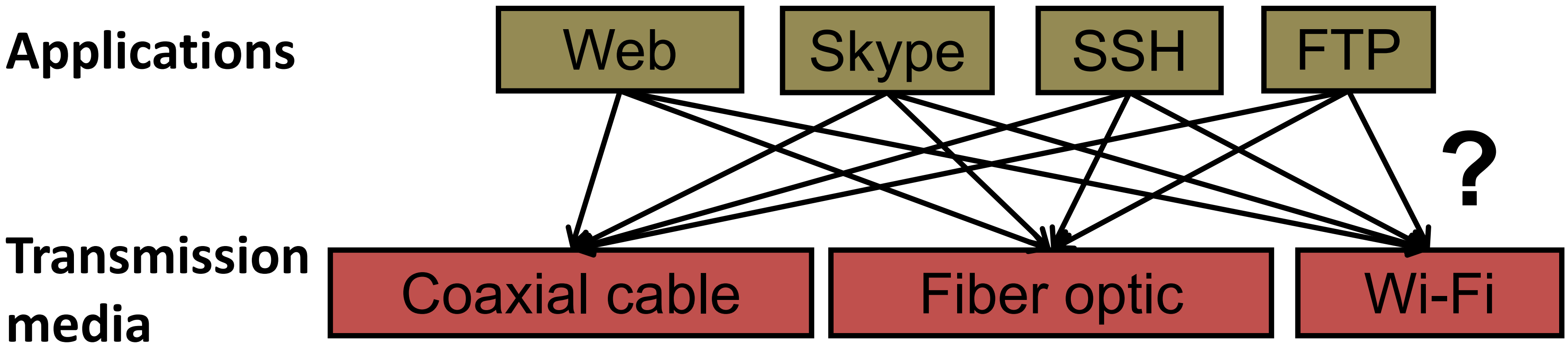
Today

1. **Layering** and the End-to-End Argument

2. Transmission Control Protocol (TCP) primer

3. Split Connection TCP over wireless

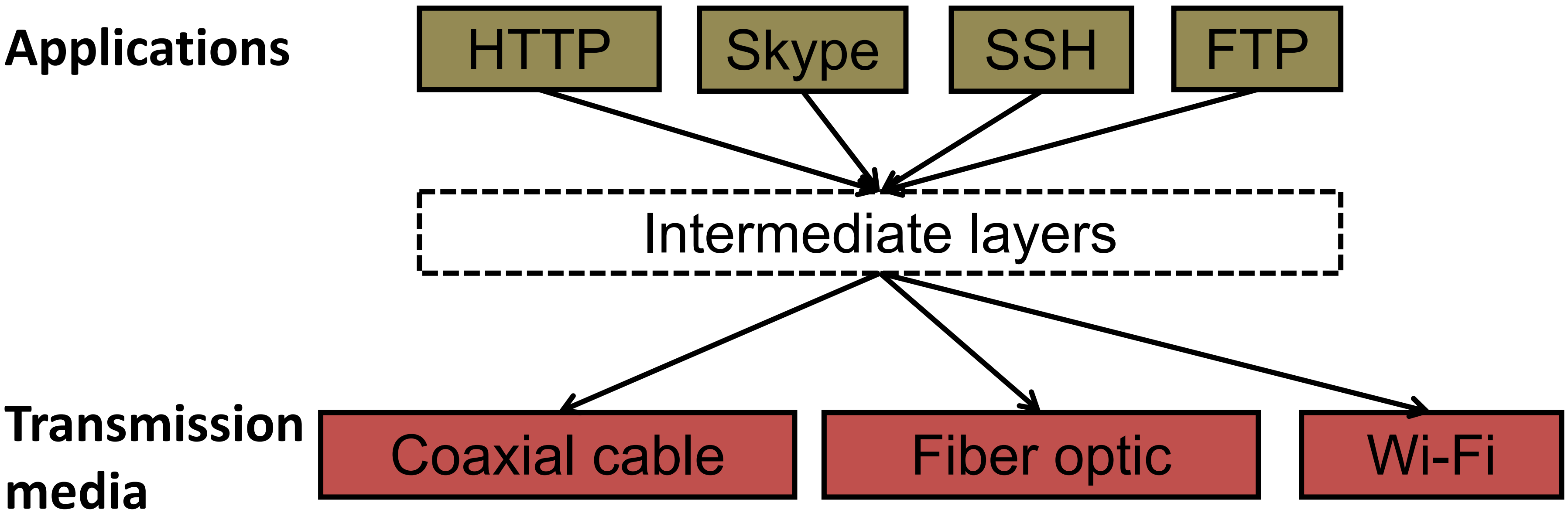
Layering: Motivation



- **Re-implement every application** for every new underlying transmission medium?
  - **Change** every application on any **change** to an underlying transmission medium (and vice-versa)?

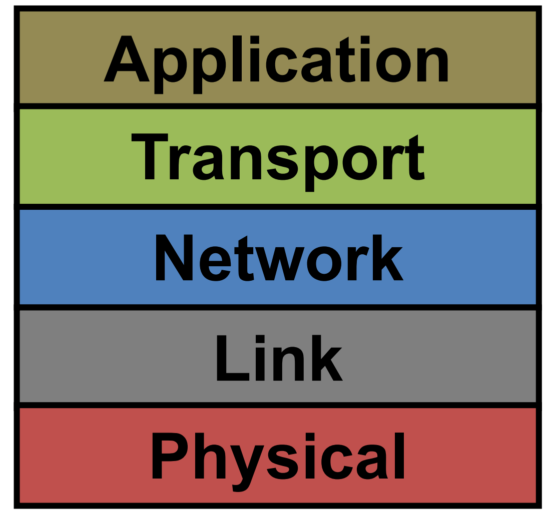
- **No!** But how does the Internet design avoid this?

Internet solution: Intermediate layers



- **Intermediate layers** provide a **set of abstractions** for applications and media
- New applications or media **need only implement for** intermediate layer's **interface**

Properties of layers



- **Service:** **What** a layer does
- **Service interface:** **How to access** the service
  - Interface for the layer **above**
- **Protocol interface:** **How peers communicate** to implement service
  - Set of rules and formats that govern the communication **between two Internet hosts**

Physical layer (L1)

- **Service**: **Move bits** between two systems connected by a **single physical link**
- **Interface**: specifies **how to send, receive bits**
  - *e.g.*, require quantities and timing
- **Protocols**: coding scheme used to represent bits, voltage levels, duration of a bit

Data link layer (L2)

- **Service**: **End hosts exchange atomic messages**
  - Perhaps over multiple physical links
    - But using same ***framing*** (headers/trailers)
  - **Arbitrates access** to common physical media
  - Implements **reliable transmission, flow control**
- **Interface**: send messages (frames) to other end hosts; receive messages addressed to end host
- **Protocols**: Addressing, routing, medium access control

Network layer (L3)

- **Service**: **Deliver *datagrams* to other networks**
  - Cross-technology (*e.g.*, Ethernet, 802.11, optical, …)
  - Possibly includes packet scheduling/priority
  - Possibly includes buffer management
  - **Best effort service:** may **drop, delay, duplicate** datagrams

- **Interface**:
  - Send packets to specified internetwork destination
  - Receive packets destined for end host

- **Protocols**:
  - Define inter-network addresses (globally unique)
  - Construct routing tables and forward datagrams

Transport layer (L4)

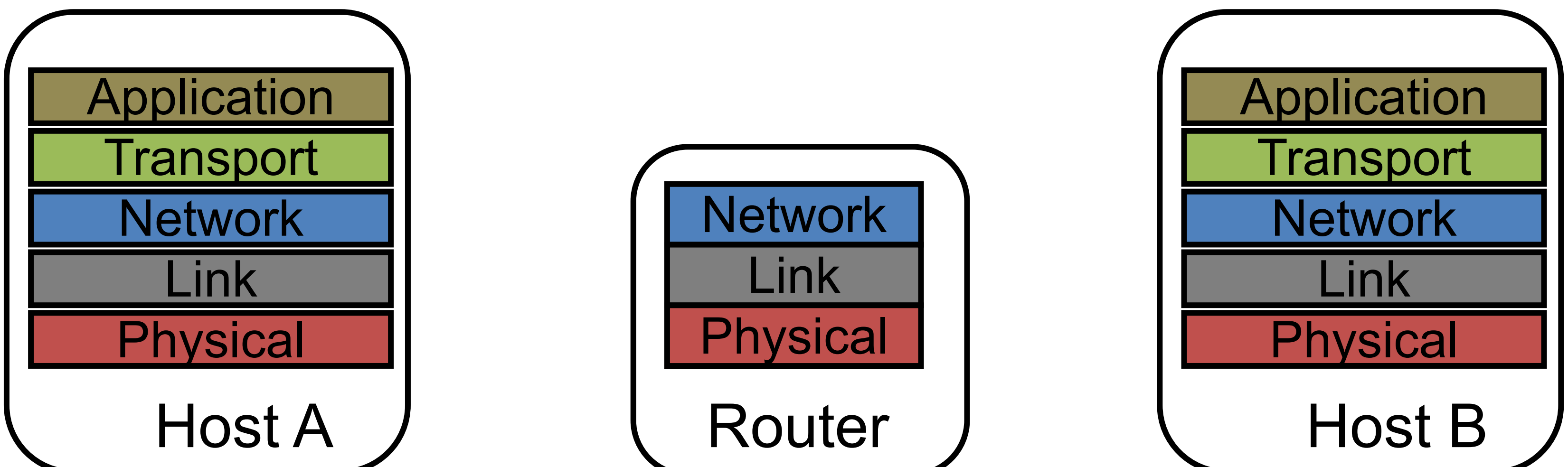
- **Service**: Provide **end-to-end** communication between **processes on different hosts**
  - Demultiplex communication between hosts
  - Possibly reliability in the presence of errors
  - Rate adaptation (**flow control, congestion control**)

- **Interface**: send message to specific process at given destination; local process receives messages sent to it

- **Protocol**: perhaps implement reliability, flow control, packetization of large messages, framing

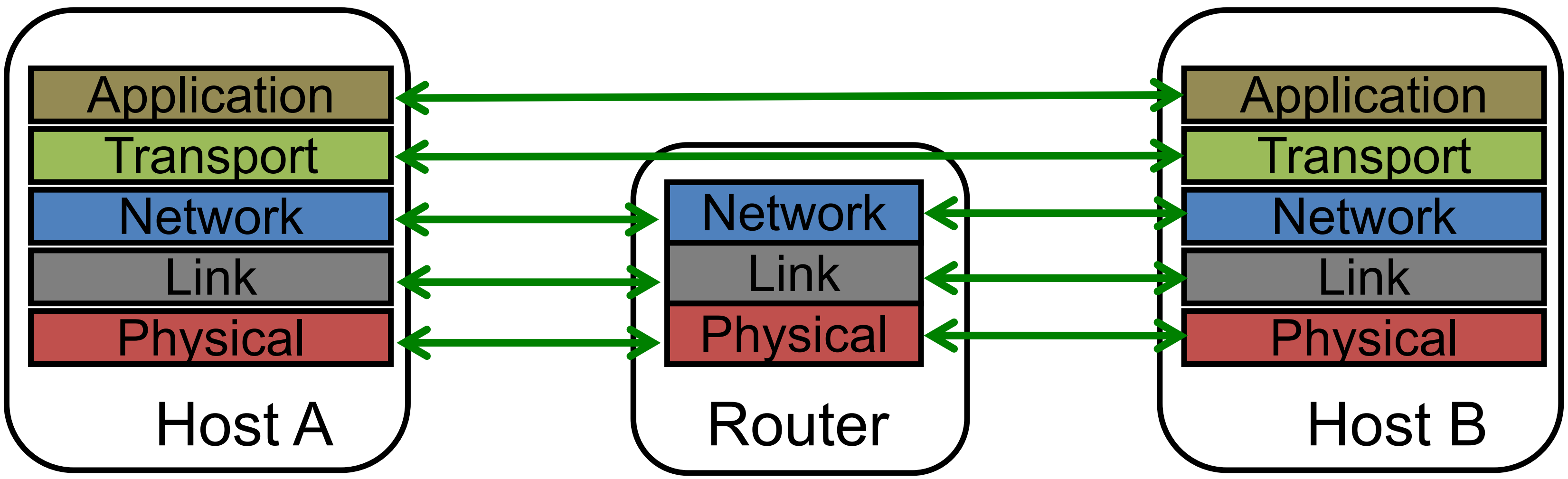
Who does what?

- Five layers
  - **Lower three layers** are implemented **everywhere**
  - **Top two layers** are implemented **only at end hosts**
    - **Their protocols are *end-to-end***



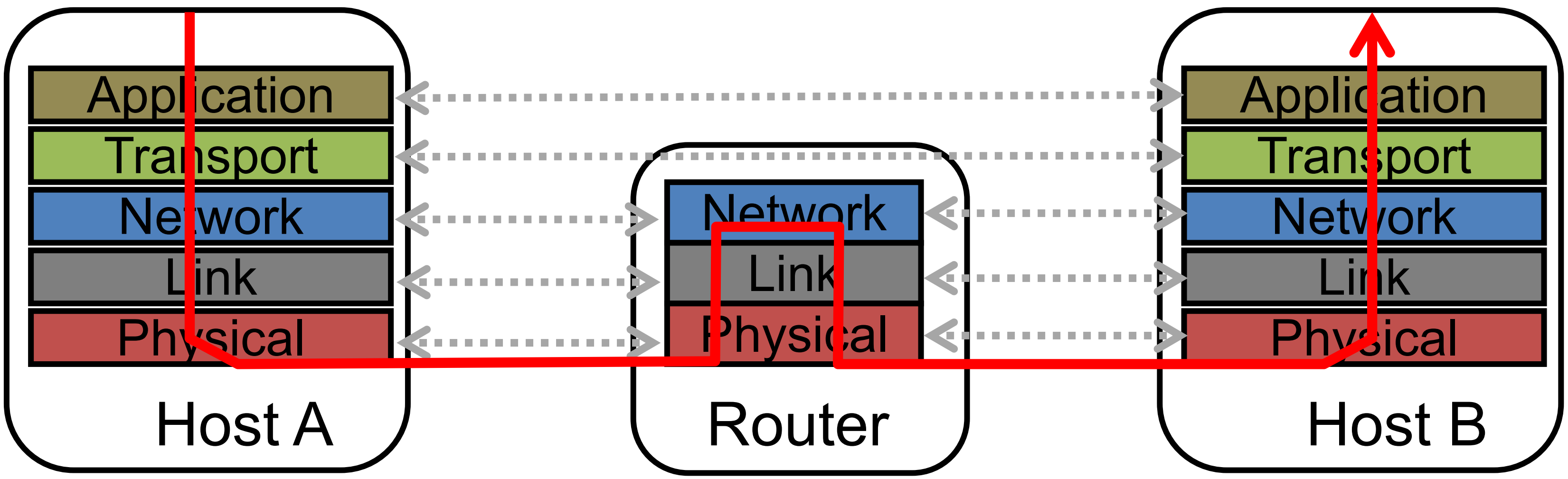
Logical communication

- Each layer on a host interacts with its **peer** host's corresponding layer via the **protocol interface**



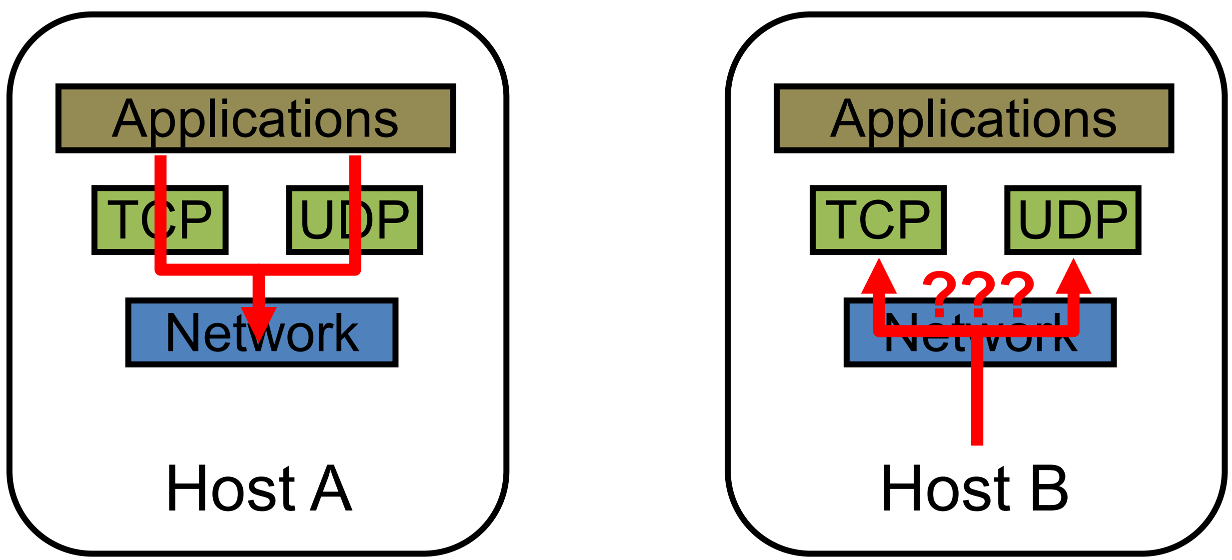
Physical path across the Internet

- Communication **goes down to physical network**
- Then from **network peer to peer**
- Then **up to the relevant layer**



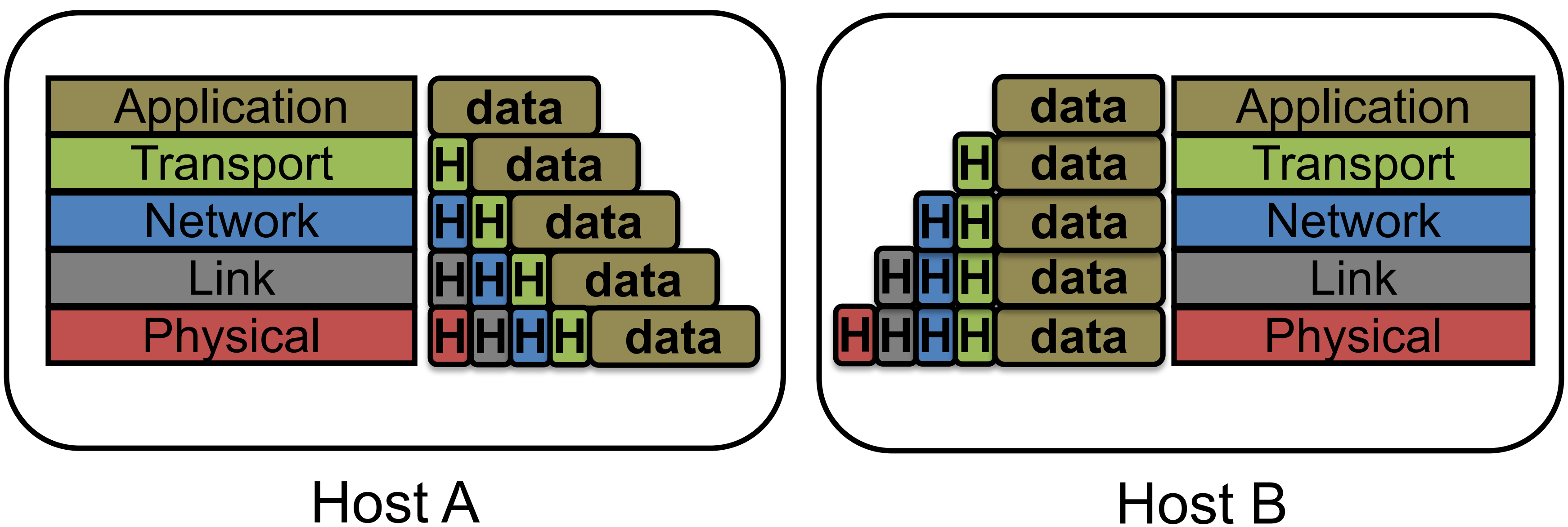
Protocol multiplexing

- **Multiplexing:** Multiple **overlying** protocols share use of a single **underlying** protocol
- **Problem:** How does the underlying protocol decide **which overlying protocol** messages go to?

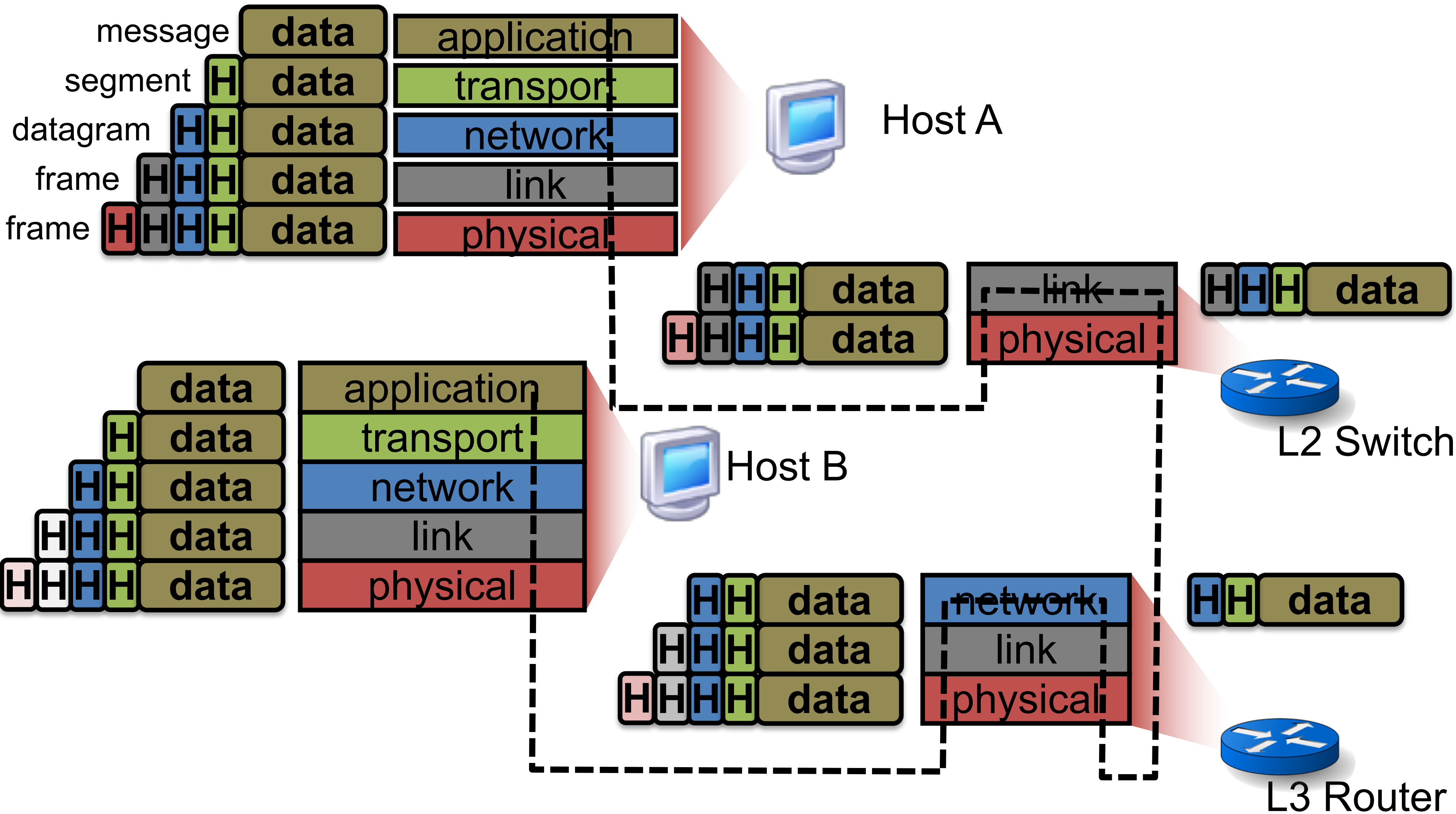


Protocol headers

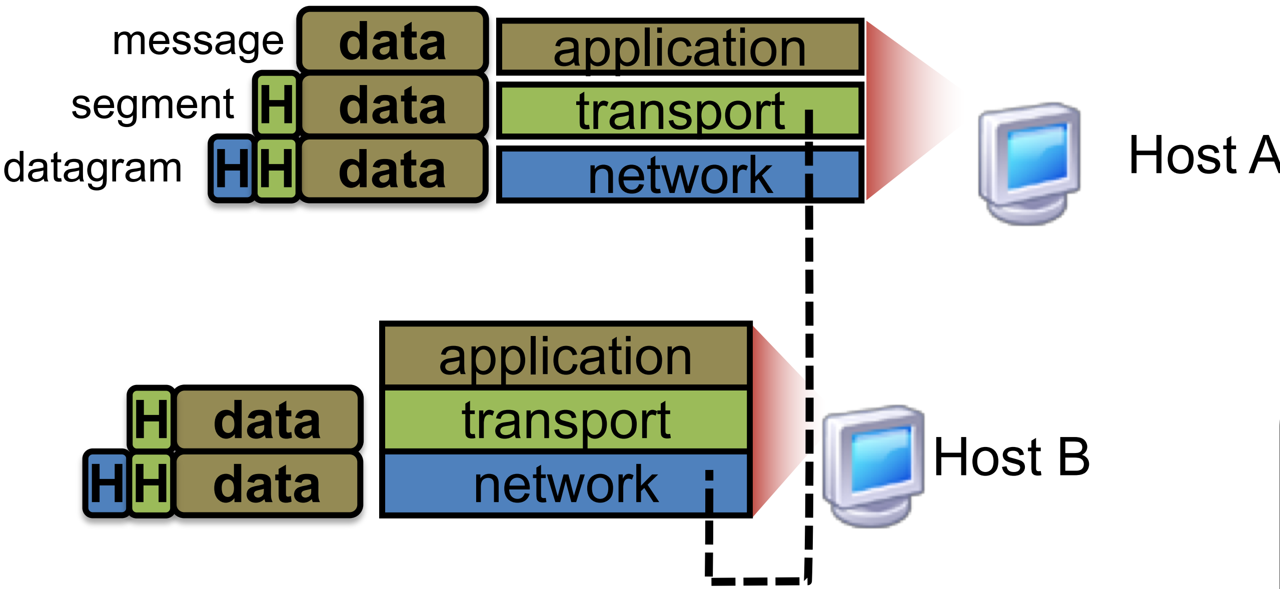
- Each layer attaches its own header (H) to facilitate communication between peer protocols
- On reception, layer **inspects and removes** its own header
  - Higher layers **don't see** lower layers' headers

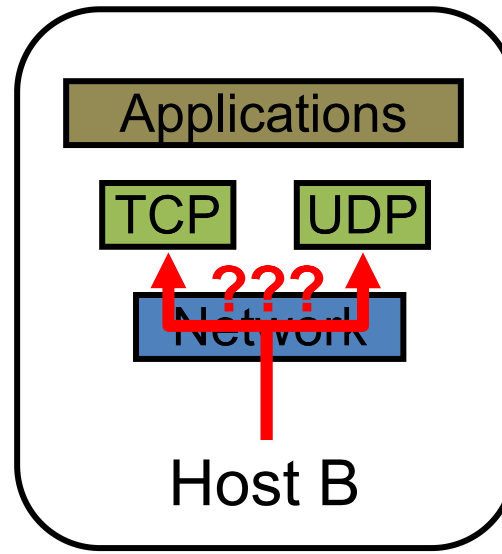


Encapsulation in the Internet



Protocol demultiplexing





- Lower-layer header contains demultiplexing information

- **Network header** contains **Protocol** field specifying overlying protocol

Drawbacks of layering

- Layer *n* may **duplicate** lower level functionality
  - *e.g.*, error recovery to retransmit lost data

- Layers may need **same information in headers**
  - *e.g.*, timestamps, maximum transmission unit size

- Layering can **hurt performance**
  - *e.g.*, previous lecture

Layer violations

- Two types:

1. **Overlying** layer examines **underlying** layer's state
   – *e.g.*, transport monitors wireless link-layer to see whether packet loss from congestion or corruption

2. **Underlying** layer inspecting **overlying** layer's state
   – *e.g.*, firewalls, NATs (network address translators), "transparent proxies"

Today

1. Layering and **the End-to-End Argument**
   – **Reading:** "End-to-End Arguments in System Design" by Saltzer, Reed, Clark

2. Transmission Control Protocol (TCP) primer

3. Split Connection TCP over wireless

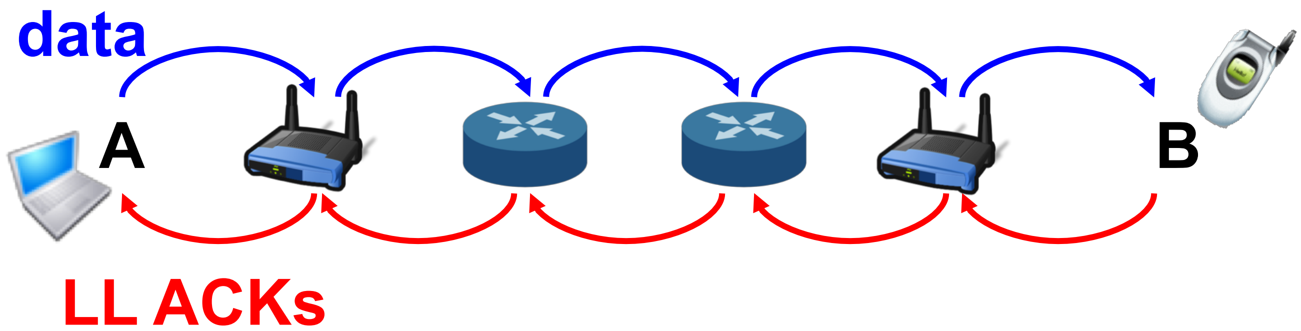
Motivation: End-to-End Argument

- **Five layers** in the Internet architecture model

- **Five places** to solve many of same problems:
  - In-order delivery
  - Duplicate-free delivery
  - Reliable delivery after corruption, loss
  - Encryption
  - Authentication

- ***In which layer(s) should a particular function be implemented?***

Example: Careful file transfer from A to B



- **Goal: Accurately copy file** on **A's** storage to **B's** storage

- Straw man design:
  - Read file from **A**'s storage
  - **A** sends packetized file to **B**
    - Link layer **resends** lost or corrupted packets at each hop
  - **B** writes file data to storage

- ***Does this system meet the design goal?*** Bit errors on links no issue

Where might errors happen?

- On **A**'s or **B**'s disk
- In **A**'s or **B**'s RAM or CPU
- In **A**'s or **B**'s software
- In the RAM, CPU, or **software** of **any router** that forwards packet

- Why might errors be **likely?**
  - Drive for CPU speed and storage density: pushes hardware to EE limits, engineered to tight tolerances
    - *e.g.*, today's disks return data that are the output of an maximum-likelihood estimation!
  - Bugs abound!

Solution: End-to-End verification

1. **A** keeps a **checksum** with the on-disk data
   – *Why not compute checksum at start of transfer?*
2. **B** computes checksum over received data, sends to **A**
3. **A** compares the two checksums and resends if not equal

- Can we eliminate hop-by-hop error detection?

- Is a whole-file checksum, **alone,** enough?

End-to-End Principle

- Only the **application at communication endpoints can completely and correctly** implement a function

- Processing in **middle alone cannot** provide function
  - Processing in **middle may,** however, be an **important performance optimization**

- Engineering middle hops to provide guaranteed functionality is *often* **wasteful of effort, inefficient**

Perils of lower-layer implementation

- **Entangles** application behavior with network internals

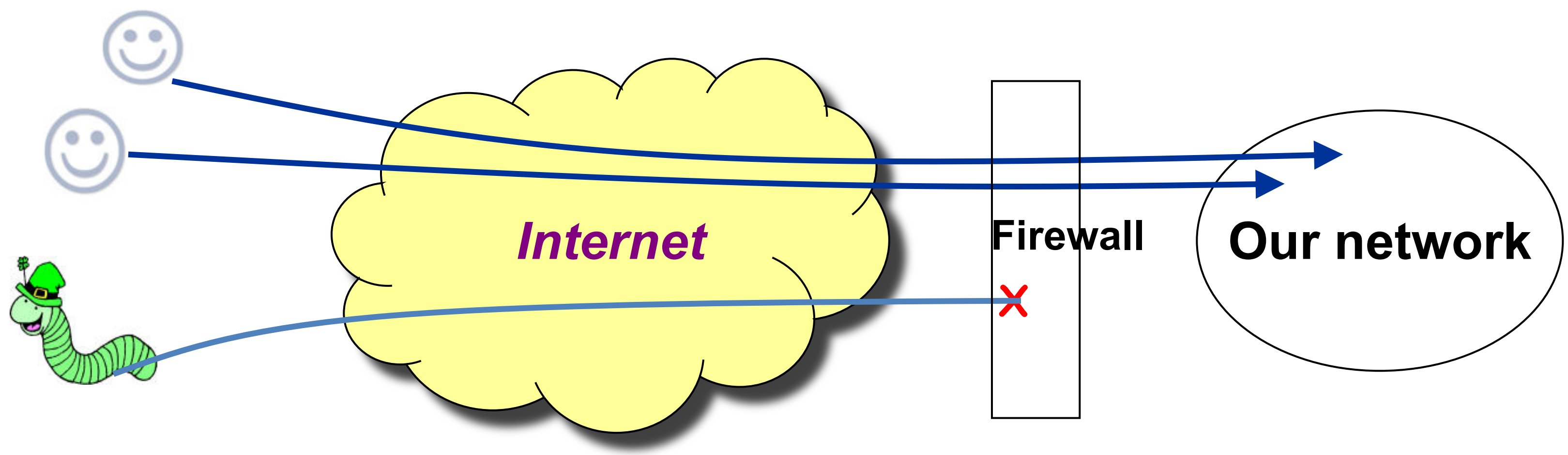
- **Suppose** each IP router **reliably transmitted** to next hop
  - **Result:** Lossless delivery, but **variable delay**
    - ftp: **Okay,** move huge file reliably (just end-to-end TCP works fine, too, though)
    - Skype: **Terrible,** jitter packets when a few corruptions or drops not a problem anyway

- **Complicates deployment** of innovative applications
  - Example: Phone network *v.* the Internet

Advantages of lower-layer implementation

- Can improve **end-to-end system performance**
- Each application author **needn't recode a shared function**
- Overlapping error checks (*e.g.*, checksums) at all layers invaluable in **debugging and fault diagnosis**
- If end systems not cooperative (increasingly the case), **only way to enforce resource allocation!**

End-to-end violation: Firewalls



- Firewalls clearly **violate the e2e principle**
  - **Endpoints** are capable of deciding what traffic to ignore
  - Firewall is **entangled** with network, transport, apps, & vice-versa
    - ***e.g.:*** New header bit to improve congestion control? Many firewalls **filter all such packets!**

- **Yet, we probably do need firewalls**

Summary: End-to-End principle

- Many functions **must** be implemented **at application endpoints** to provide desired behavior
  - Even if implemented in "middle" of network

- End-to-end approach **decouples design** of components in network interior from design of applications at edges
  - Some functions still **benefit** from implementation in **network interior** at cost of entangling interior, edges

- End-to-end principle is **not sacred;** it's just a way to think critically about design choices in communication systems

Five-minute break and Partner Exercises

1. The *end-to-end argument is:*
   A. A guideline for placing functions in computer systems
   B. A rule for placing functions in a computer system
   C. A debate on where to place functions in a computer system
   D. A debate about anonymity in computer networks

2. Of the following, the best example of an end-to-end argument is:
   A. If you laid all the web programmers in the world end to end, they would reach from Princeton to CERN
   B. Every byte going into the write end of a UNIX pipe eventually emerges from the pipe's read end
   C. Even if a chain manufacturer tests each link before assembly, they had better test the completed chain
   D. All important network communication functions should be moved to the Application layer

Today

1. Layering and the End-to-End Argument

2. **Transmission Control Protocol (TCP) primer**
   - **Over wired networks**
   - **Reading:** "Congestion Avoidance and Control" by Jacobson and Karels

3. TCP over wireless

TCP: Connection-Oriented, Reliable Byte Stream Transport

- Layer-four protocol for **reliable transport**
  - Sending app offers a sequence of bytes: d0, d1, d2, …
  - Receiving app sees all bytes arrive in same sequence: d0, d1, d2…
    - **Result: Reliable byte stream** transport **between endpoints** on the internet
- Each such byte stream is called a ***connection***, or ***flow***

TCP’s Many End-to-End Goals

- Recover from **data loss**
- Avoid receipt of **duplicated** data
- Preserve data **ordering**
- Provide **integrity** against corruption
- Avoid sending faster than **receiver** can **accept** data
- **Avoid congesting** network

Fundamental Problem: Ensuring At-Least-Once Delivery

- Network **drops** packets, so to **ensure delivery:**
  - Sender attaches ***sequence number (seqno)*** to each data packet sent; keeps copy of sent packet
  - Receiver returns ***acknowledgement (ACK)*** to sender for each data packet received, containing seqno

- **Sender** sets a ***retransmit timer*** on each transmission
  - If timer expires < ACK returns: **retransmit** that packet
  - If ACK returns, **cancel timer, forget** that packet

- How long should the retransmit timer be?

Fundamental Problem: Estimating RTT

- Expected time for ACK to return is *round-trip time* (*RTT*)
  - End-to-end delay for data to reach receiver, then its ACK to reach sender
- **Strawman:** use fixed timer (*e.g.,* 250 milliseconds)
  - What if the route/wireless conditions change?
  - What if congestion occurs at one or more routers?

Fixed timer violates E2E argument; details of link behavior should be left to link layer!

Hard-coded timers lead to brittle behavior as technology evolves

- **Too small** a value: needless **retransmissions**
- **Too large** a value: needless **delay detecting loss**

Estimating RTT: Exponentially Weighted Moving Average (EWMA)

- Measurements of RTT readily available
  - Note time ***t*** when packet sent, corresponding ACK returns at time ***t'***
  - **RTT *measurement sample:* m** = t'− t
- Single sa

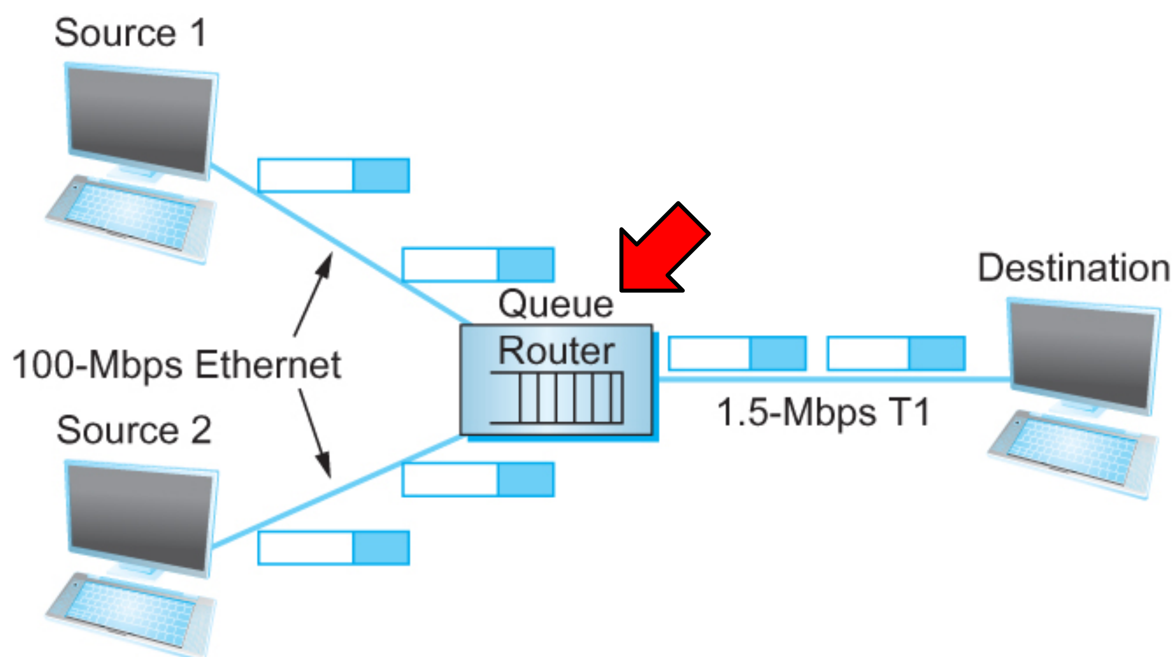
EWMA weights newest samples most

How to choose α? (TCP uses 1/8)

Is mean sufficient to capture RTT behavior over time? (more later)

- Adapt over time, using EWMA:
  - **Measurement samples:** m0, m1, m2, …
  - fractional weight for new measurement, **α**
  - $RTT_{i} = ((1 - \alpha) \times RTT_{i-1} + \alpha \times m_{i})$

What is Congestion?



- Sources may have **sufficient proximal link capacity** to send
- But in the middle of the network may **share capacity**
- **Too many packets in the network** → queue overflows: ***congestion***

How does TCP know congestion occurred?

- *How can packets get lost in wired networks?*
  - Almost exclusively **queue buffer overflows**

- Packet loss is a binary signal

- *How does a TCP **sender** know that a packet loss has occurred?*
  - Lack of receipt of an Acknowledgement → Timeouts

Retransmission and Duplicate Delivery

- When sender's retransmit timer expires, two **indistinguishable cases:**
  - **Data packet dropped** *en route* to receiver, or
  - **ACK dropped** *en route* to sender

- In both cases, **sender retransmits**

- In latter case, **duplicate data packet reaches receiver!**

Eliminating Duplicates: Exactly-Once Delivery

- Sender marks each packet with a **monotonically increasing sequence number seqno**
- Sender includes greatest ACKed seqno in its packets
- Receiver remembers only greatest received sequence number, drops received packets with smaller ones

Doesn't guarantee delivery!
Properties: If delivered, then only once.
If undelivered, sender will not think delivered.
If ACK not seen, data may have been delivered, but sender will not know.

End-to-End Integrity

- Achieved by using **transport checksum**
- Protects against things link-layer reliability cannot:
  - Router memory corruption, software bugs, &c.
- Covers **data in packet, transport protocol header**
- Also should cover **layer-3 source and destination!**
  - Misdelivered packet should not be inserted into data stream at receiver, nor should be acknowledged
  - Receiver drops packets w/failed transport checksum

Today

1. Networking primer/review
2. Transmission Control Protocol (TCP) primer
3. **TCP over wireless**

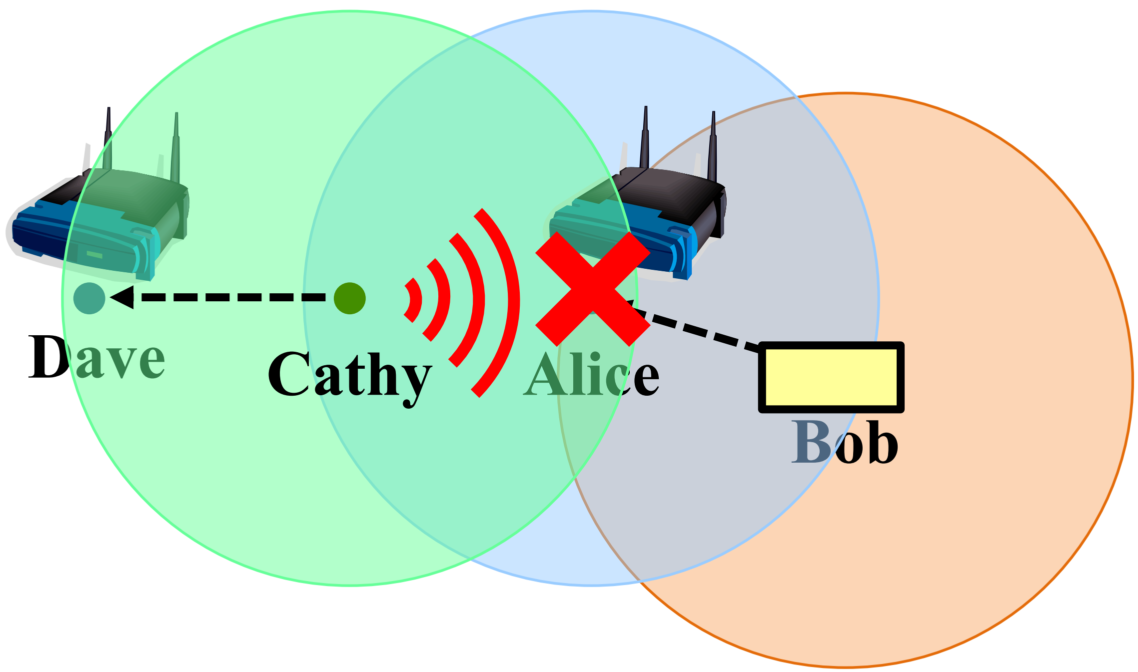
Running TCP on Wireless Links

- Generally, TCP interprets **packet loss** as **queue congestion**
  - TCP sender **reduces congestion window**

- Wireless links have **higher** bit error rates, frame loss rates

- On **wireless links**, packet loss can also occur due to random channel errors, or cellular or WLAN handoffs
  - Temporary loss **not due to congestion**
  - Reducing window **may be too conservative**
  - Leads to **poor throughput**

Wireless can also Cause Packet Loss



Shared wireless medium leads to a *collision* of Bob and Cathy's packets *at* Alice

Wired & Wireless Mix: Best TCP sender strategy becomes unclear

	Congestion loss	Link loss
Wireless link:	**Frequent** (collision)	**Frequent** (multipath, interference)
Wired links:	**Frequent** (queue drop)	**Rare**

Slow down! **Maintain rate**

Fundamental question:

How to differentiate between

1. **Loss due to congestion**
2. **Loss due to wireless link itself**

Hard to do:

TCP is fundamentally an "end-to-end" protocol: only sees a loss

Two Broad Approaches

1. **Mask wireless losses from TCP sender**
   - Then TCP sender will not slow down

   - **Split Connection Approach**
   - TCP Snoop

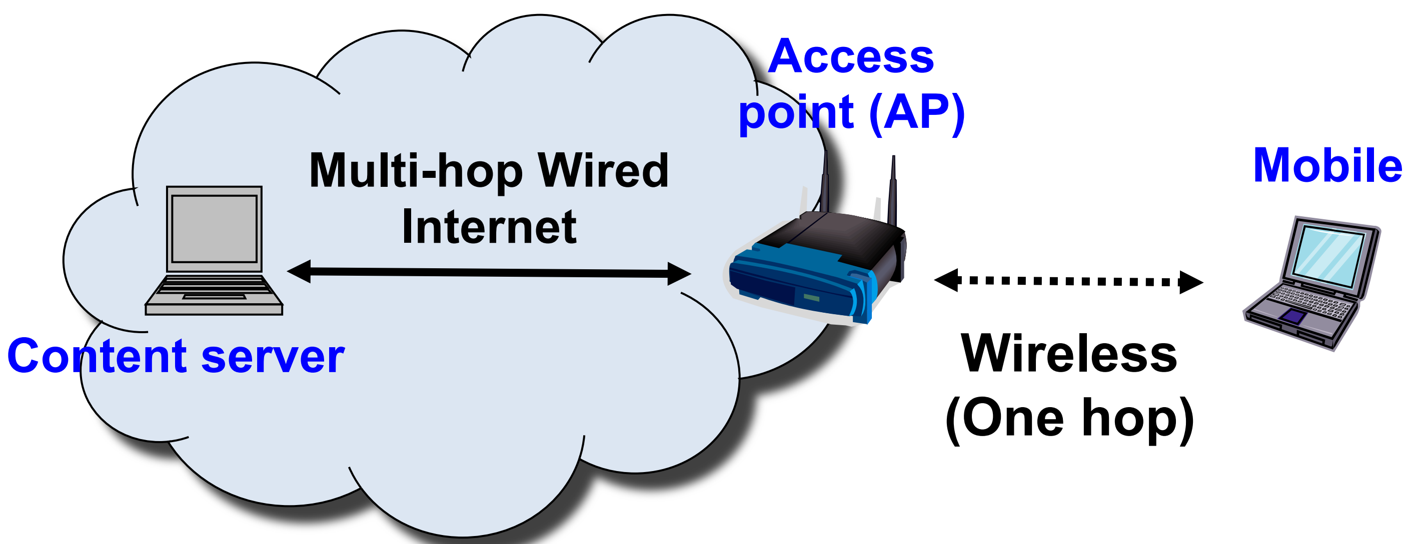
2. Explicitly notify TCP sender about cause of packet loss

Split Connection Approach

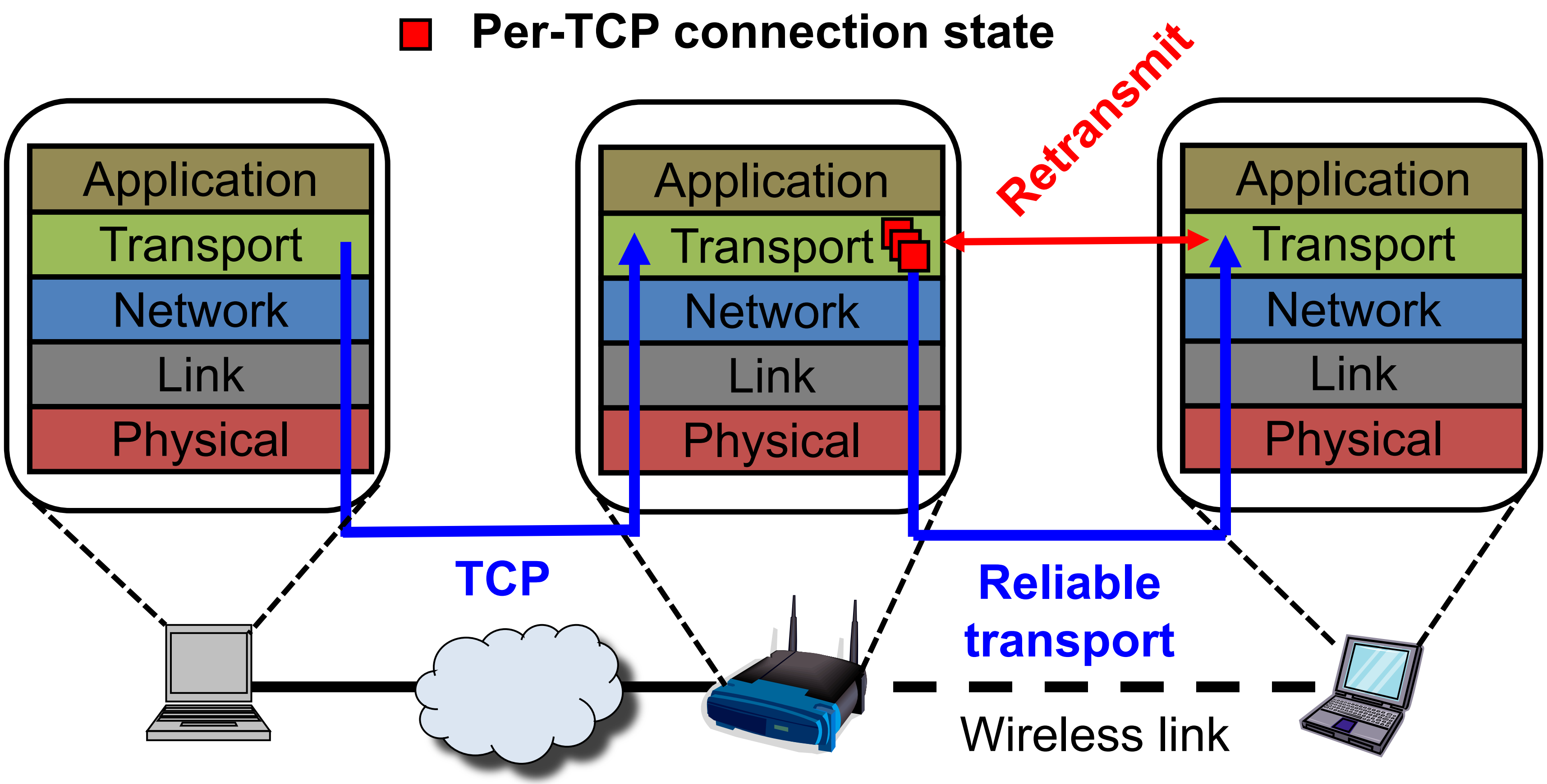
- Also called Indirect TCP (I-TCP)

- Divide the TCP connection into two parts:
  1. TCP connection between content server and AP
  2. Another connection between AP and mobile host
  - **No real end-to-end connection**

- **No changes** to the TCP endpoint at the content server



Split Connection: TCP Implementation



- Maintain **state** for both “halves” of each end-to-end “connection” at the AP

Split Connection: Considerations

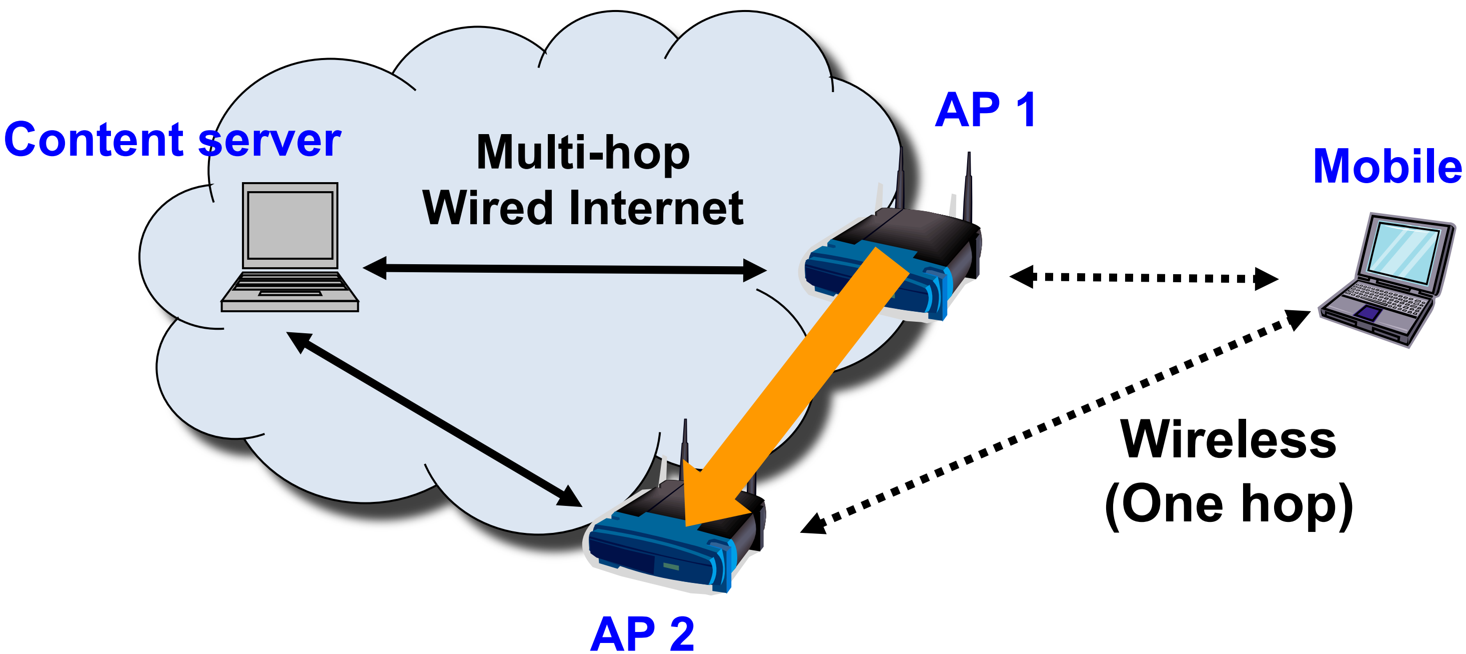
- Connection between AP and mobile need not be TCP
  - Could be *e.g.*, Selective Repeat over UDP

- Assume that the **wireless part is just one hop** (traditional cellular or wireless LAN)

- Assumes wireless losses **not caused by too many packets in the network**
  - But that's **not always true** (*e.g.* collisions)
    - Sender **should slow down, but doesn't**

Split Connection: State Migration

- Consequence of breaking end-to-end connection:
  - On handoff from AP 1 to AP 2, **connection state** must **move from AP 1 to AP 2**



Split Connection: Advantages

- **No changes needed** in wired network or content servers
- Transmission errors on the wireless link do not propagate into the fixed network
  - Local recovery from errors
- Possibility of using custom (optimized) transport protocol for the hop between AP and mobile

Split Connection: Critique

- **Loss of end-to-end semantics:**
  - ACK at TCP sender no longer means that receiver must have received that packet
    - TCP **no longer reliable** if **crash/bug at AP**
- **Large buffer space** may be needed at AP
- AP must maintain **per-TCP connection state**
- **State must be forwarded** to new AP on handoff
  - May cause higher handoff latency

Precepts

Python Intro & Programming Exercises

Location: Friend Center, Room 003

Tuesday Lecture

Transport over Wireless II: Snoop and Explicit Loss Notification