CSCI-1680
Transport Layer I

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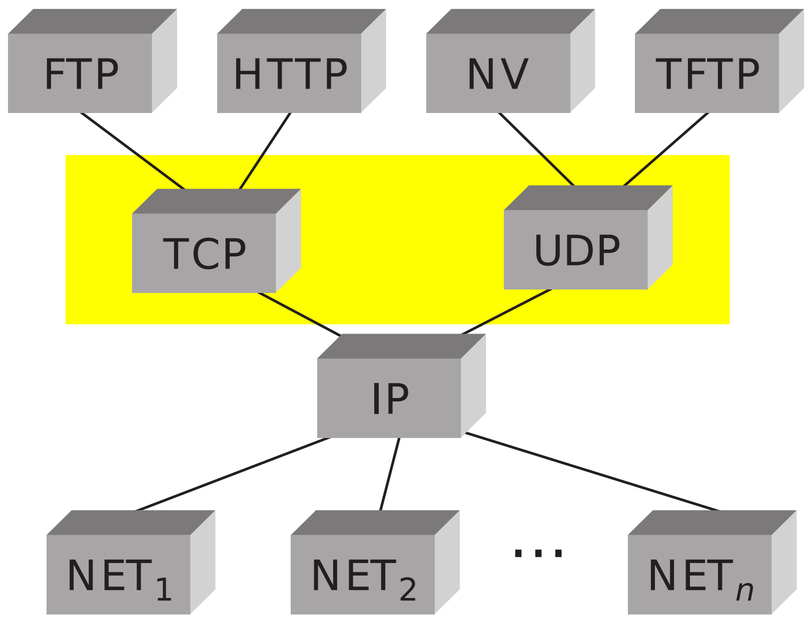
Based partly on lecture notes by David Mazières, Phil Levis, Rodrigo Fonseca

Today

- **Transport Layer**
  - UDP
  - TCP Intro
    - Connection Establishment

Transport Layer

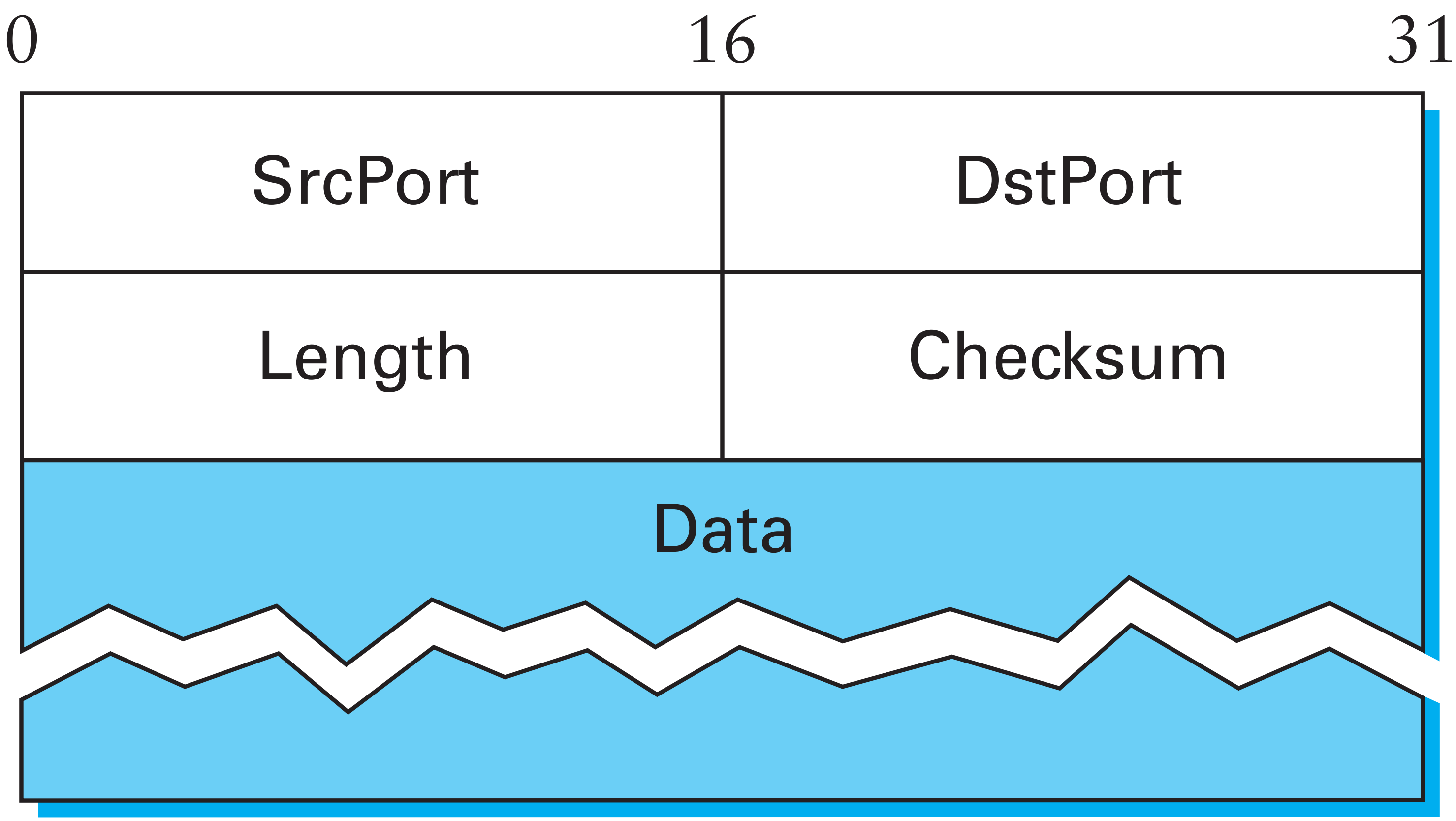


- **Transport protocols sit on top of the network layer**
- **Problem solved: communication among processes**
  - Application-level multiplexing (“ports”)
  - Error detection, reliability, etc.

UDP – User Datagram Protocol

- **Unreliable, unordered datagram service**
- **Adds multiplexing, checksum**
- **End points identified by *ports***
  - Scope is an IP address (interface)
- **Checksum aids in error detection**

UDP Header



UDP Checksum

- **Uses the same algorithm as the IP checksum**
  - Set Checksum field to 0
  - Sum all 16-bit words, adding any carry bits to the LSB
  - Flip bits to get checksum (except 0xffff->0xffff)
  - To check: sum whole packet, including sum, should get 0xffff
- **How many errors?**
  - Catches any 1-bit error
  - Not all 2-bit errors
- **Optional in IPv4: not checked if value is all 0**

Pseudo Header

```
 0      7 8     15 16    23 24    31
+--------+--------+--------+--------+
|          source address           |
+--------+--------+--------+--------+
|        destination address        |
+--------+--------+--------+--------+
|  zero  |protocol|   UDP length    |
+--------+--------+--------+--------+
```

- **UDP Checksum is computer over *pseudo-header* prepended to the UDP header**
  - For IPv4: IP Source, IP Dest, Protocol (=17), plus UDP length
- **What does this give us?**
- **What is a problem with this?**
  - Is UDP a layer on top of IP?

Next Problem: Reliability

- **Review: reliability on the link layer**

Problem	Mechanism
Dropped Packets	Acknowledgments + Timeout
Duplicate Packets	Sequence Numbers
Packets out of order	Receiver Window
Keeping the pipe full	Sliding Window (Pipelining)

- **Single link: things were easy… ☺**

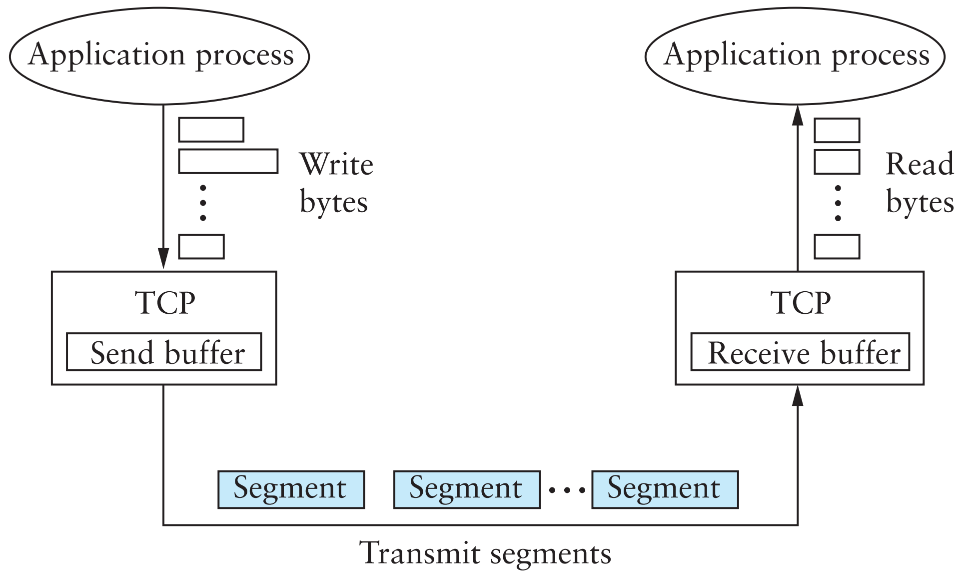
Transport Layer Reliability

- **Extra difficulties**
  - Multiple hosts
  - Multiple hops
  - Multiple potential paths
- **Need for connection establishment, tear down**
  - Analogy: dialing a number versus a direct line
- **Varying RTTs**
  - Both across connections and *during* a connection
  - Why do they vary? What do they influence?

Extra Difficulties (cont.)

- **Out of order packets**
  - Not only because of drops/retransmissions
  - Can get very old packets (up to 120s), must not get confused
- **Unknown resources at other end**
  - Must be able to discover receiver buffer: flow control
- **Unknown resources in the network**
  - Should not overload the network
  - But should use as much as safely possible
  - Congestion Control (next class)

TCP – Transmission Control Protocol



- **Service model: “reliable, connection oriented, full duplex byte stream”**
  – Endpoints: <IP Address, Port>
- **Flow control**
  – If one end stops reading, writes at other eventually stop/fail
- **Congestion control**
  – Keeps sender from overloading the network

TCP

- **Specification**
  - RFC 793 (1981), RFC 1222 (1989, some corrections), RFC 5681 (2009, congestion control), …
- **Was born coupled with IP, later factored out**
  - We talked about this, don't always need everything!
- **End-to-end protocol**
  - Minimal assumptions on the network
  - All mechanisms run on the end points
- **Alternative idea:**
  - Provide reliability, flow control, etc, link-by-link
  - Does it work?

Why not provide (*) on the network layer?

- **Cost**
  - These functionalities are not free: don't burden those who don't need them
- **Conflicting**
  - Timeliness and in-order delivery, for example
- **Insufficient**
  - Example: reliability

* may be security, reliability, ordering guarantees, …

End-to-end argument

- **Functions placed at lower levels of a system may be redundant or of little value**
  - They may **need** to be performed at a higher layer anyway
- **But they may be justified for performance reasons**
  - Or just because they provide *most* of what is needed
  - Example: retransmissions
- **Lesson: weigh the costs and benefits at each layer**
  - Also: the *end* also varies from case to case

TCP Header

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          Source Port          |       Destination Port        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                        Sequence Number                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Acknowledgment Number                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Data |           |U|A|P|R|S|F|                               |
| Offset| Reserved  |R|C|S|S|Y|I|            Window             |
|       |           |G|K|H|T|N|N|                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           Checksum            |         Urgent Pointer        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Options                    |    Padding    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                             data                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

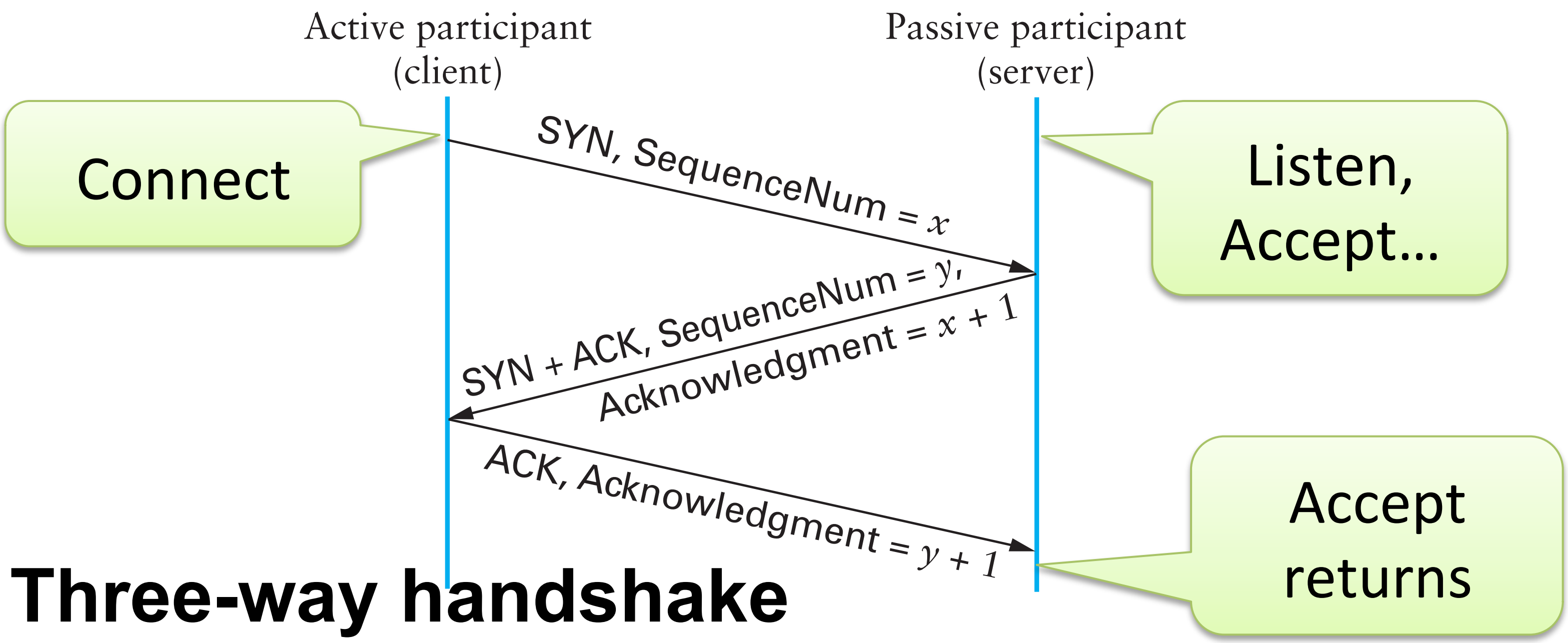
Header Fields

- **Ports: multiplexing**
- **Sequence number**
  - Correspond to *bytes*, not packets!
- **Acknowledgment Number**
  - Next expected sequence number
- **Window: willing to receive**
  - Lets receiver limit SWS (even to 0) for flow control
- **Data Offset: # of 4 byte (header + option bytes)**
- **Flags, Checksum, Urgent Pointer**

Header Flags

- URG: whether there is urgent data
- ACK: ack no. valid (all but first segment)
- PSH: push data to the application immediately
- RST: reset connection
- SYN: synchronize, establishes connection
- FIN: close connection

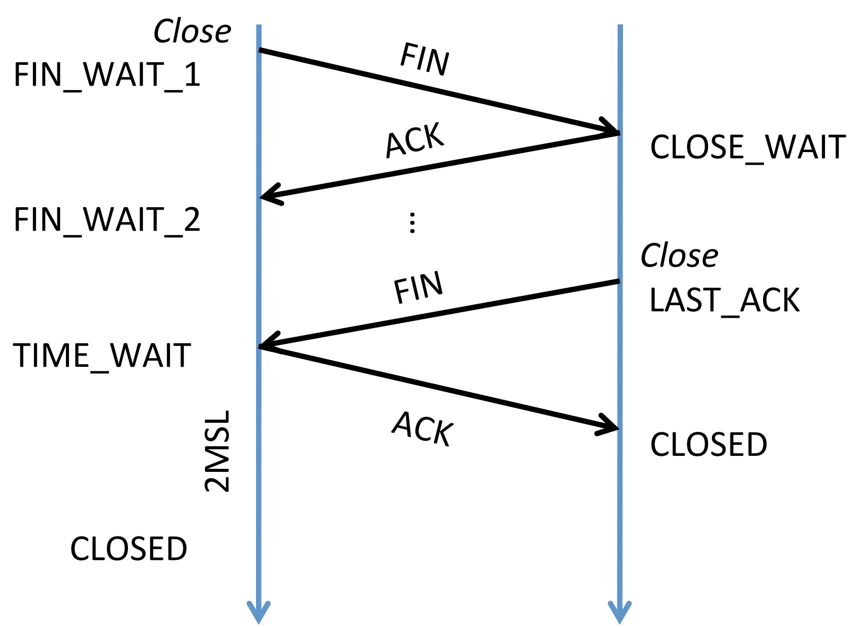
Establishing a Connection



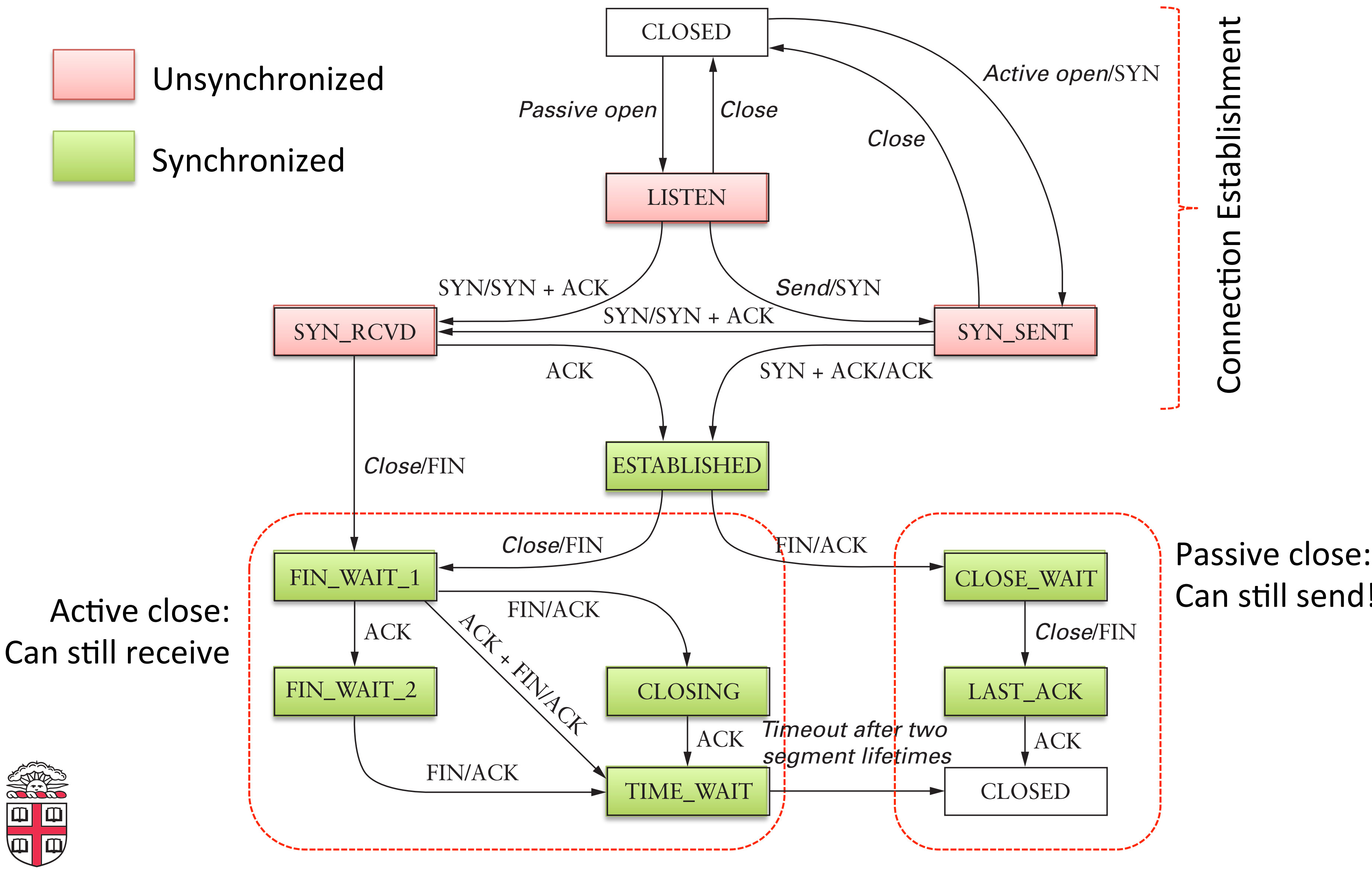
- **Three-way handshake**
  - Two sides agree on respective initial sequence nums
- **If no one is listening on port: server sends RST**
- **If server is overloaded: ignore SYN**
- **If no SYN-ACK: retry, timeout**

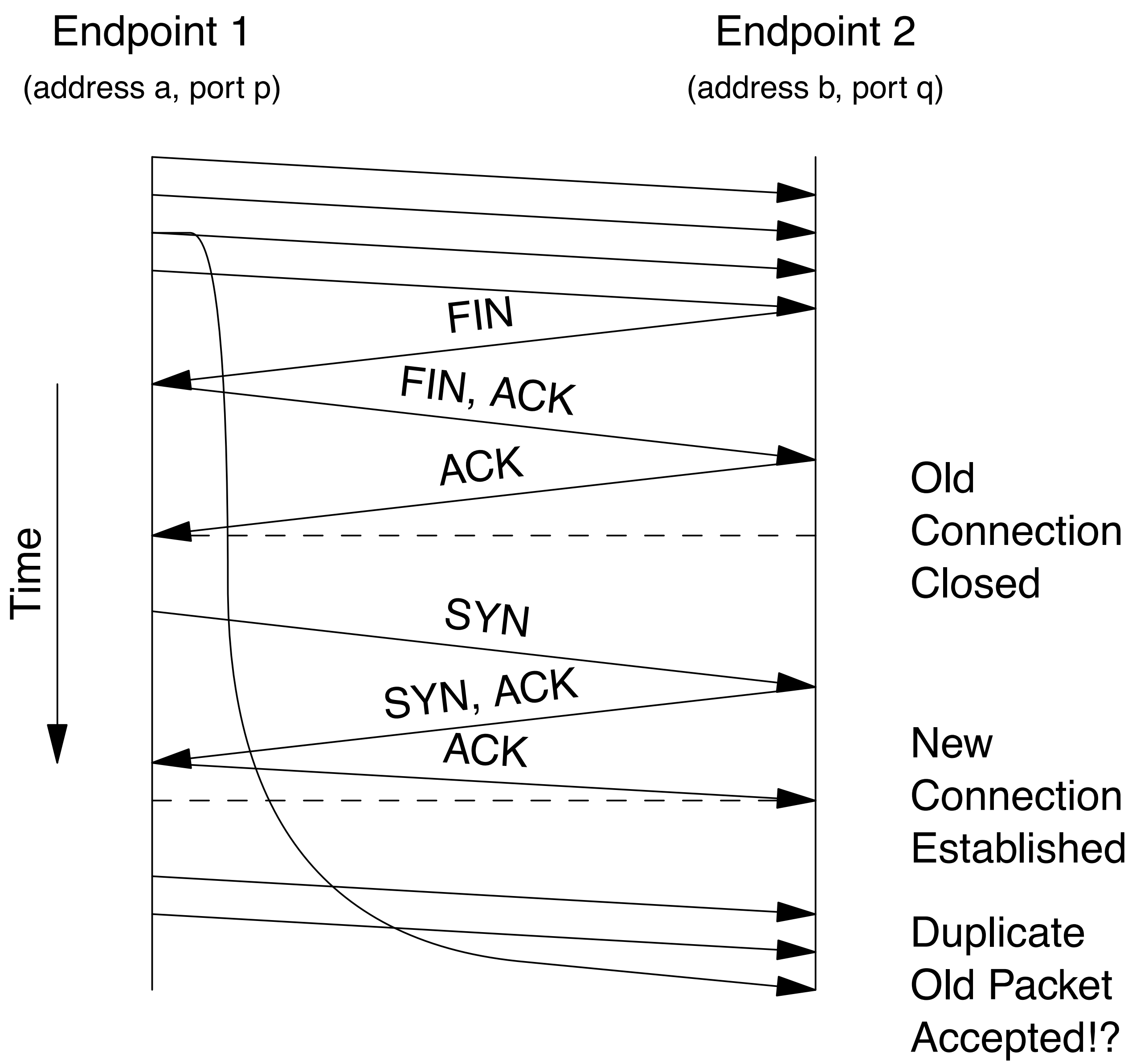
Connection Termination

- **FIN bit says no more data to send**
  – Caused by close or shutdown
  – Both sides must send FIN to close a connection
- **Typical close**



Summary of TCP States





From: The TIME−WAIT state in TCP and Its Effect on Busy Servers, Faber and Touch Infocom 1999

TIME_WAIT

- **Why do you have to wait for 2MSL in TIME_WAIT?**
  - What if last ack is severely delayed, AND
  - Same port pair is immediately reused for a new connection?
- **Solution: active closer goes into TIME_WAIT**
  - Waits for 2MSL (Maximum Segment Lifetime)
- **Can be problematic for active servers**
  - OS has too many sockets in TIME_WAIT, can accept fewer connections
    - Hack: send RST and delete socket, SO_LINGER = 0
  - OS won't let you re-start server because port in use
    - SO_REUSEADDR lets you rebind

Next class

- Sending data over TCP