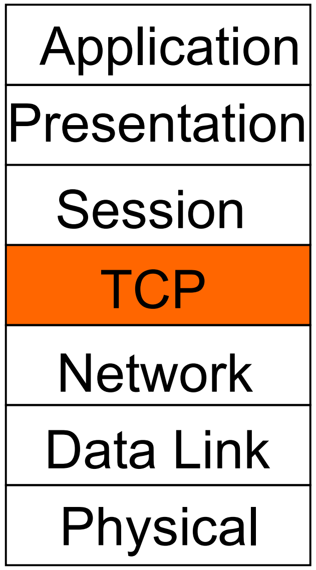
CSE 461 – Module 12

TCP End-to-End

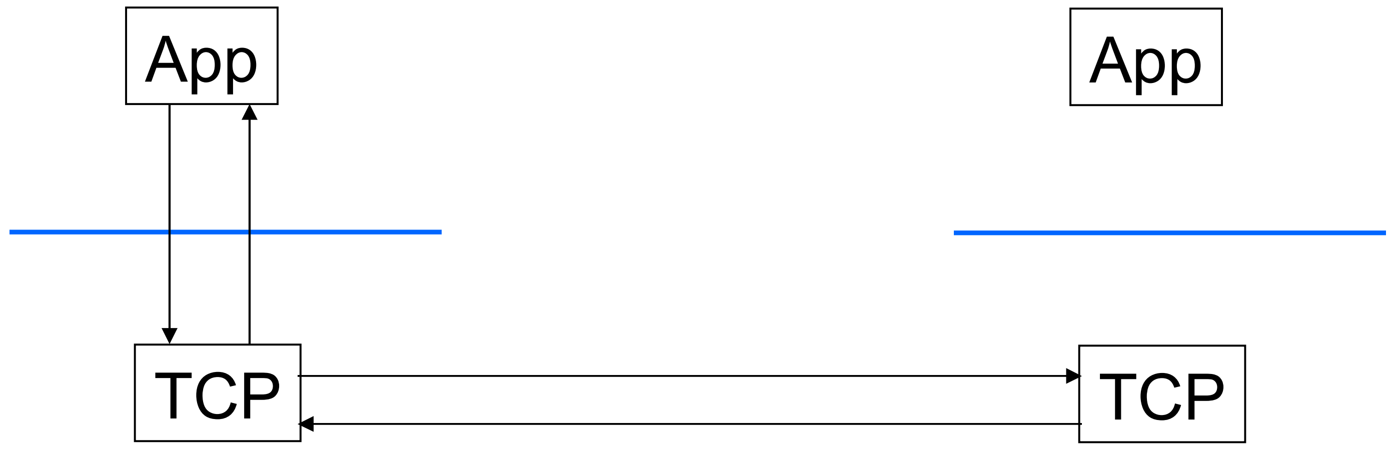
This Time

- End-to-end considerations for TCP
  - How is *connect()* different from *send(SYN)?*
  - Concurrency / blocking issues
  - What does receiver do?
  - What does sender do?
    - When should data be sent?
    - When should it be resent?
    - When should it conclude connectivity has been lost?



1. connect() vs. send(SYN)

- Q: Is *connect()* the same thing as *send(syn)* (if the interface allowed the latter)?



A: No. (How are they different?)

2. Concurrency and blocking

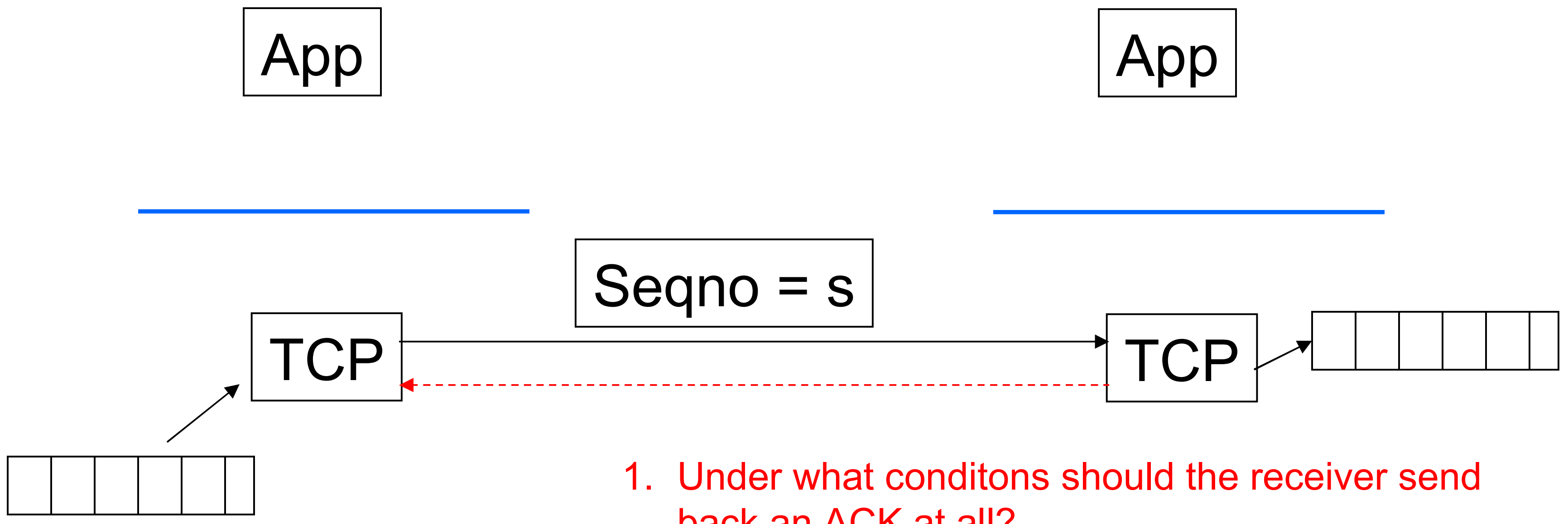
- Protocol implementation involves a lot of concurrency
  - E.g., (S1) sending app thread adds to send buffer; (S2) sending TCP thread removes from buffer and sends; (R1) receving TCP thread puts in buffer; (R2) receiving app reads from buffer

- Whether or not the app thread is blocked is an important part of the semantics
  - Why should app thread block on *connect()*?
  - Why shouldn't it block on *send()?*
    - Why should it block on *send()?*
  - Must *receive()* be blocking?
  - Must *close()* be blocking?

Socket Semantics vs. Application Architecture

- The application knows best what semantics it needs
  - Suppose your application establishes a data connection and a control connection to some peer
  - Can't do a *read()* from either one without ignoring the other
- One way to get around blocking semantics at lower level: spawn more threads, and synchronize as necessary at the user level
- Problem: performance
- "Solution": Most interfaces provide some form of non-blocking mechanism
  - Usually you can:
    - Ask if some operation would block or not (*poll*)
    - Wait for any of a number of distinct events to happen (*select*)
- A half-way measure: often you can specify a timeout for how long the thread should block (e.g., *receive(250)*)

3. What does the receiver do?

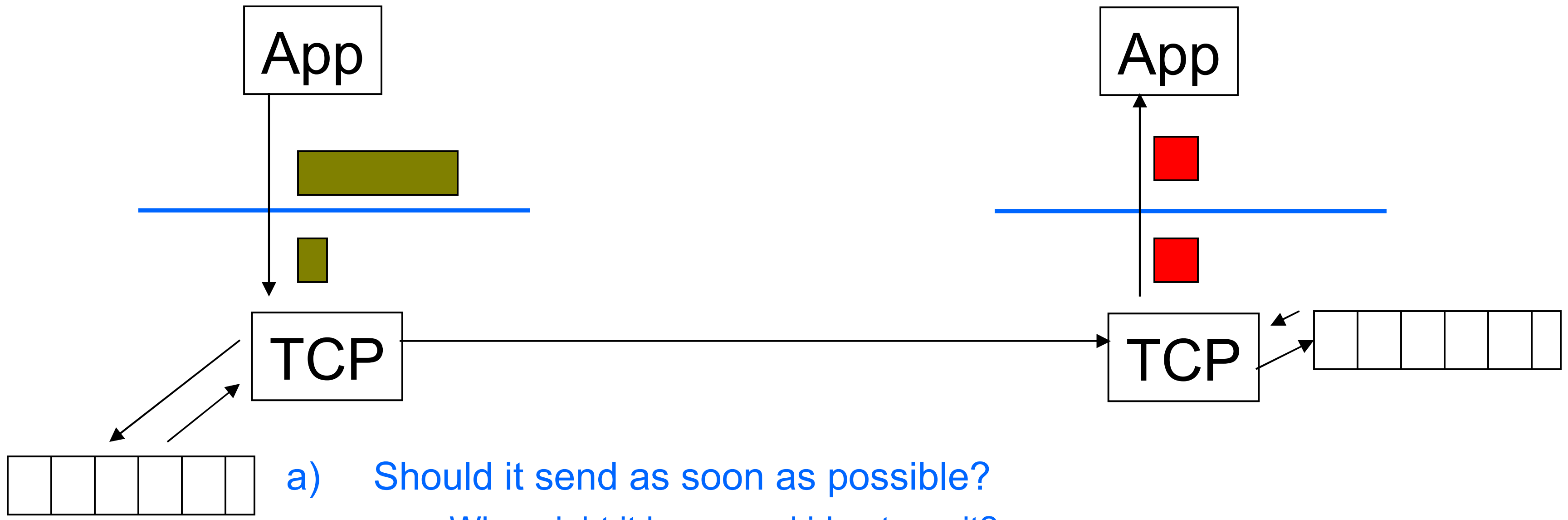


1. Under what conditons should the receiver send back an ACK at all?
2. When it does, what should the ACK seqno be?
3. (What does an ACK tell the sender?)

What should the receiver do?

- General philosophy:
  - keep receiver as simple as possible
  - ACKs are the primary feedback the sender has to work with
- With that in mind:
  - Don't ACK ACK's. (What happens if you do?)
  - Do ACK everything else.
    - Receiver **must** ACK already seen data…
- Many possible choices for what ACK should send back
  - TCP: seqno of first byte not yet received
- Can TCP send a segment with no data bytes?
  - What should happen?

4. What does the sender do?



a) Should it send as soon as possible?

- Why might it be a good idea to wait?

- When it sends, how long should the retry timeout be?
  - Problem with too short? too long?

a) When should it give up?

a) Send as soon as possible?

- "Silly window" problem
  - Reminder:
```
Effective Window =
          Receiver advertised window -
            (LastByteSent - LastByteAcked)
```

- Suppose the sender transmits a small frame for some reason.
  - The ACK for that frame opens the effective window by its size
  - The sender sends an equally small segment
  - Etc…

- Want to avoid this!
  - Either don't send small segments, or
  - Don't open window by a small amount

a) Send as soon as possible?

- Possible receiver side approaches:
  - Could use a timeout at receiver
    - Send an ACK at most once per timeout?
  - Simpler: if window goes to zero, don’t advertise an open window until you have an MSS (maximum segment size) available

- Possible sender side approaches:
  - Could use a sender timeout
  - Could use a Nagle’s Algorithm (self-clocking)

Nagle’s Algorithm

```
send() {
    if both available data and eff window ≥ MSS {
        send MSS bytes
    } else if lastByteSent - lastByteAcked > 0 {
        // expecting another ACK soon -- don’t send
    } else {
        send min(available data, eff window) now
    }
}
```

b) Deciding When to Retransmit

- How do you know when a packet has been lost?
  (Note: It's a little more complicated than this code…)

```
do {
   send(p);
   wait(t);
} while (!p.acked)
```

- How long should the timer t be?
  - Too big: inefficient (large delays ⇒ poor use of bandwidth)
  - Too small: may retransmit unnecessarily (causing extra traffic)
  - A good retransmission timer is important for good performance

- Right timer is based on the round trip time (RTT)
  - Which varies greatly in the wide area (path length and queuing)

b) Setting the Retransmission Timeout

- Boils down to estimating RTT
- Why not `EstimatedRTT = (Sum of SampleRTT’s) / N?`
- The straightforward approach:
  - for each packet, note time sent and time ACK received (RTT sample)
  - compute RTT samples and average recent samples for timeout

```
EstimatedRTT = (1-g)(EstimatedRTT) + g(SampleRTT)
0 ≤ g ≤  1
```

  - this is an **exponentially-weighted moving average** (low pass filter) that smoothes the samples with a gain of *g*
    - big *g* can be jerky, but adapts quickly to change
    - small g can be smooth, but slow to respond
    - typically, g = .1 or .2 $\Rightarrow$ stability is more important than precision
      - (lousy estimate right now does more damage than so-so estimate right now, followed by better one a little later)

Original TCP (RFC793) retransmission timeout algorithm

- Use EWMA to estimate RTT:

```
EstimatedRTT = (1-g)(EstimatedRTT) + g(SampleRTT)
0 ≤ g ≤  1,  usually g = .1 or .2
```

- Conservatively set timeout to small multiple (2x) of the estimate

```
Retransmission Timeout = EstimatedRTT + EstimatedRTT
```

- Why the '`+ EstimatedRTT`'?
  - Better to wait "too long" than not long enough.

Jacobson/Karels Algorithm

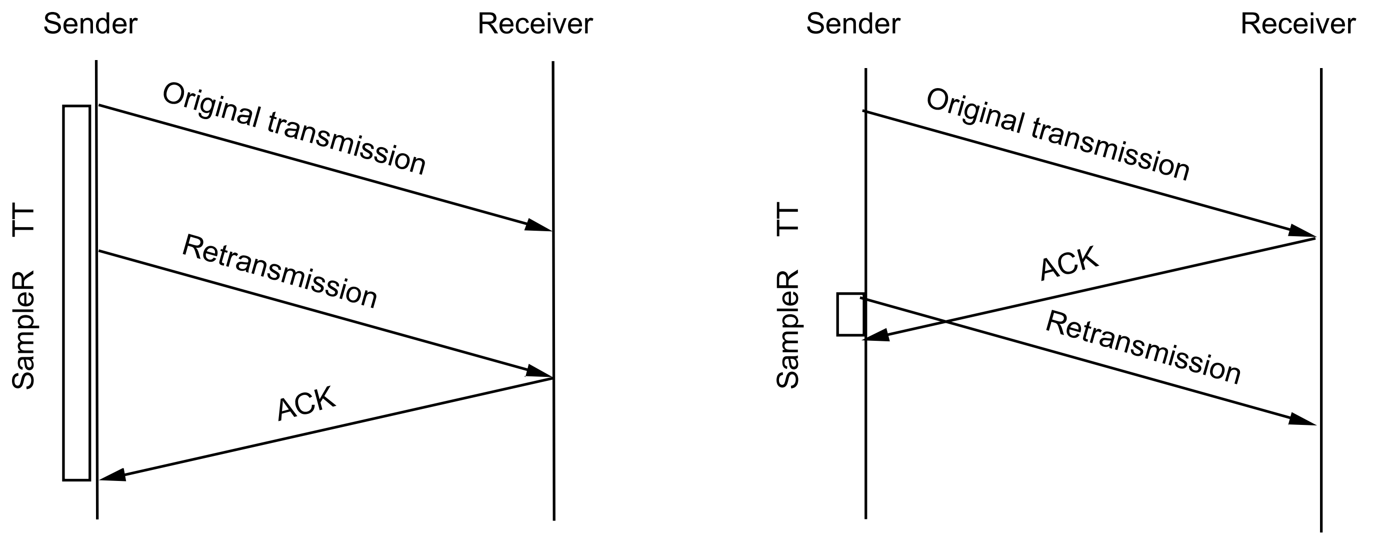
- Replace "`+ EstimatedRTT`" with measured variation in RTT

1. Compute a sample deviation statistic
   - `DevRTT = (1-b)*DevRTT + b*|SampledRTT - EstimatedRTT|`
     - typically, b = .25
2. Set timeout interval as:
   - `retransmission timeout = EstimatedRTT + k * DevRTT`
     - k is generally set to 4

- timeout =~ EstimatedRTT when variance is low (estimate is good)
  - timeout quickly moves away from EstimatedRTT (4x!) when the variance is high (estimate is bad)

Karn/Partridge Algorithm

- Problem: RTT for retransmitted packets ambiguous



- Solution: Don't measure RTT for retransmitted packets
  - Problem: RTT not updated when timeouts occurring
  - Approach: use backoff on timeout until an xmit succeeds with retransmission

c) When do we give up?

RFC 1122 (Requirements for Internet Hosts)

> The following procedure MUST be used to handle excessive retransmissions of data segments:

- There are two thresholds R1 and R2 measuring the amount of retransmission that has occurred for the same segment.
- When the number of transmissions of the same segment reaches or exceeds threshold R1, pass negative advice to the IP layer, to trigger dead-gateway diagnosis.
- When the number of transmissions of the same segment reaches a threshold R2 greater than R1, close the connection.
- An application MUST be able to set the value for R2 for a particular connection. TCP SHOULD inform the application of the delivery problem (unless such information has been disabled by the application; see Section 4.2.4.1), when R1 is reached and before R2.
- The value of R1 SHOULD correspond to at least 3 retransmissions, at the current RTO. The value of R2 SHOULD correspond to at least 100 seconds.