Congestion Control

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Slides courtesy Ion Stoica with
slight adaptation by Brighten

Today

TCP congestion control

Limitations of TCP CC

RED

Today's Lecture

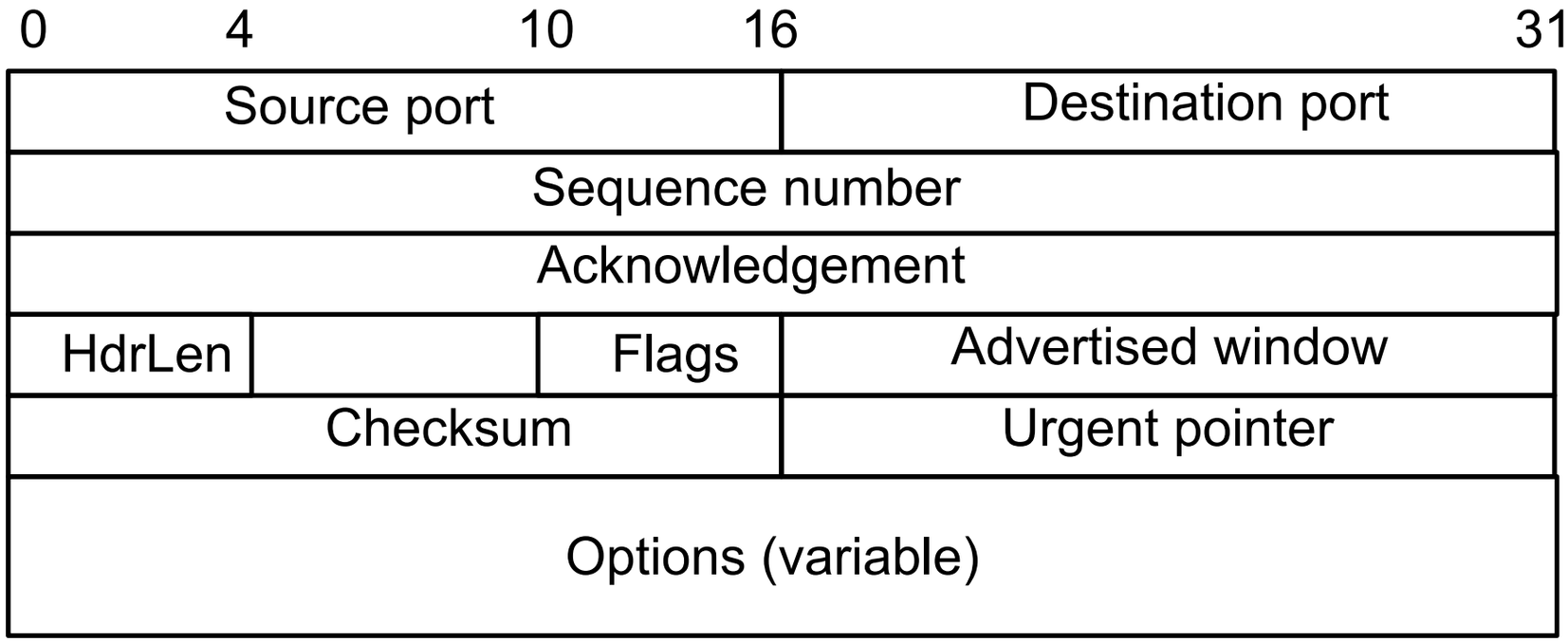
- Basics of Transport
- Basics of Congestion Control
- Comments on Congestion Control

Duties of Transport

- Demultiplexing:
  - IP header points to protocol
  - Transport header needs demultiplex further
    - UDP: port
    - TCP: source and destination address/port
      - Well known ports and ephemeral ports

- Data reliability (if desired):
  - UDP: checksum, but no data recovery
  - TCP: checksum and data recovery

TCP Header



- Sequence number, acknowledgement, and advertised window – used by sliding-window based flow control
- Flags:
  - SYN, FIN – establishing/terminating a TCP connection
  - ACK – set when Acknowledgement field is valid
  - URG – urgent data; Urgent Pointer says where non-urgent data starts
  - PUSH – don't wait to fill segment
  - RESET – abort connection

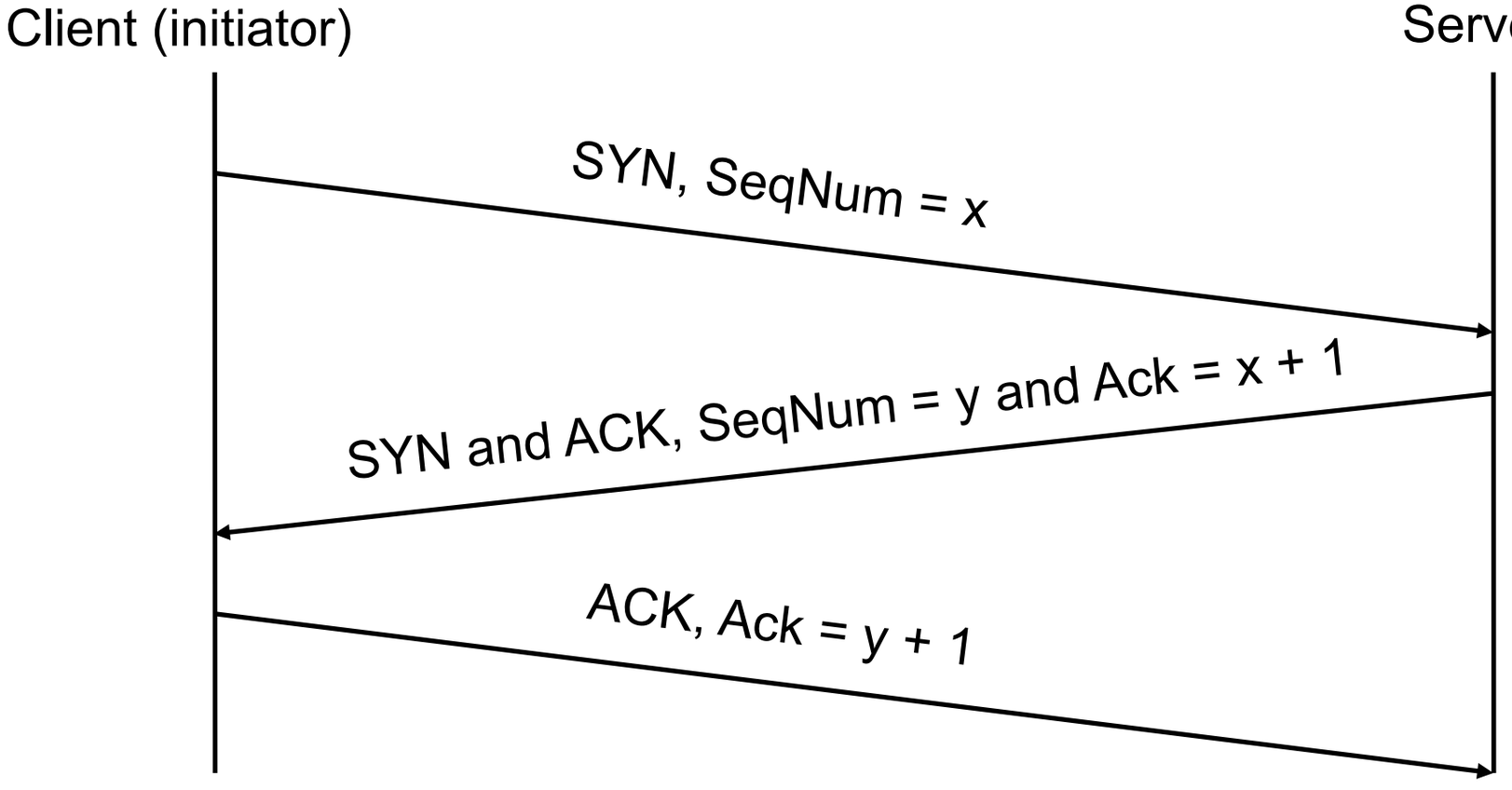
TCP Header (Cont)

- Checksum – 1's complement and is computed over
  - TCP header
  - TCP data
  - Pseudo-header (from IP header)
    - Note: breaks the layering!

Source address		
Destination address		
0	Protocol (TCP)	TCP Segment length

TCP Connection Establishment

- Three-way handshake
  - Goal: agree on a set of parameters: the start sequence number for each side



TCP Issues

- Connection confusion:
  - ISNs can’t always be the same

- Source spoofing:
  - Need to make sure ISNs are random

- SYN floods:
  - SYN cookies

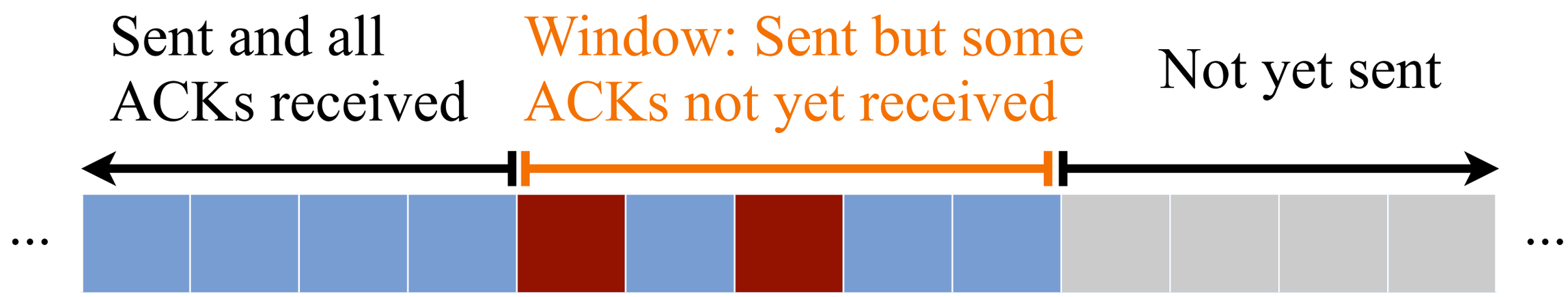
- State management with many connections
  - Server-stateless TCP (NSDI 05)

TCP Flow Control

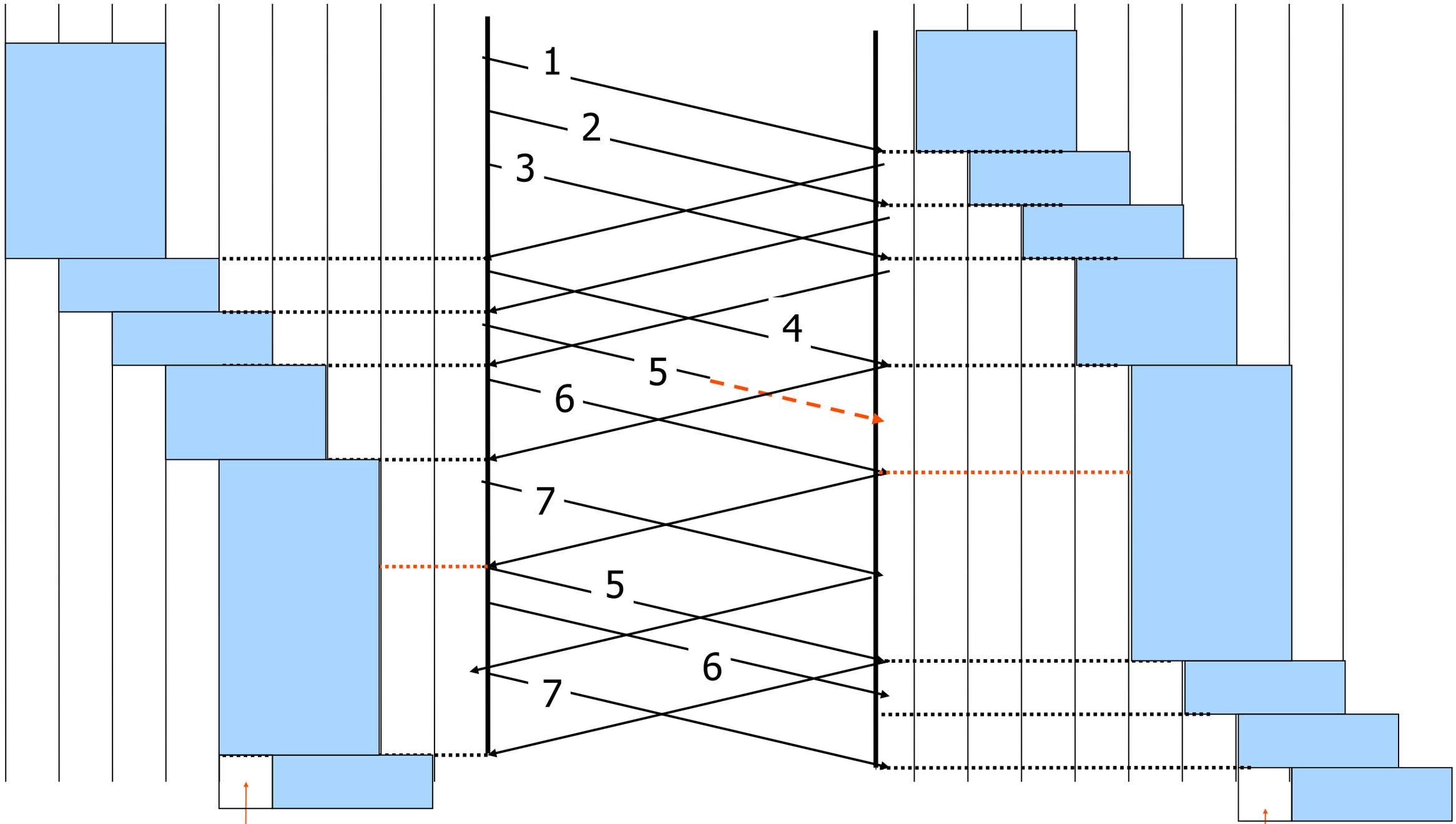
- Make sure receiving end can handle data
- Negotiated end-to-end, with no regard to network
- Ends must ensure that no more than W packets are in flight if buffer has size W
  - Receiver ACKs packets
  - When sender gets an ACK, it knows packet has arrived

Sliding window-based flow control

At the sender...



Sliding Window



Observations

- What is the throughput in terms of RTT?
  - Throughput is ~ (w/RTT)

- Sender has to buffer all unacknowledged packets, because they may require retransmission

- Receiver may be able to accept out-of-order packets, but only up to its buffer limits

What Should the Receiver ACK?

1. ACK every packet, giving its sequence number

2. Use *negative ACKs* (NACKs), indicating which packet did not arrive

3. Use *cumulative ACK*, where an ACK for number *n* implies ACKS for all $k < n$

4. Use *selective ACKs* (SACKs), indicating those that did arrive, even if not in order

Error Recovery

- Must retransmit packets that were dropped
- To do this efficiently
  - Keep transmitting whenever possible
  - Detect dropped packets and retransmit quickly
- Requires:
  - Timeouts (with good timers)
  - Other hints that packet were dropped

Timer Algorithm

- Use exponential averaging:

T(n) = measured RTT of this packet

```
A(n) = b*A(n- 1) + (1 – b)T(n)
D(n) = b*D(n-1) + (1 – b)*(T(n) – A(n))
Timeout(n) = A(n) +4D(n)
```

Question: Why not set timeout to average delay?

Notes

1. Measure T(n) only for original transmissions
2. Double `Timeout` after timeout (why?)

Hints

- When should I suspect a packet was dropped?

- When I receive several duplicate ACKs
  - Receiver sends an ACK whenever a packet arrives
  - ACK indicates seq. no. of last received *consecutively* received packet
  - Duplicate ACKs indicates missing packet

TCP Congestion Control

- Can the network handle the rate of data?

- Determined end-to-end, but TCP is making guesses about the state of the network

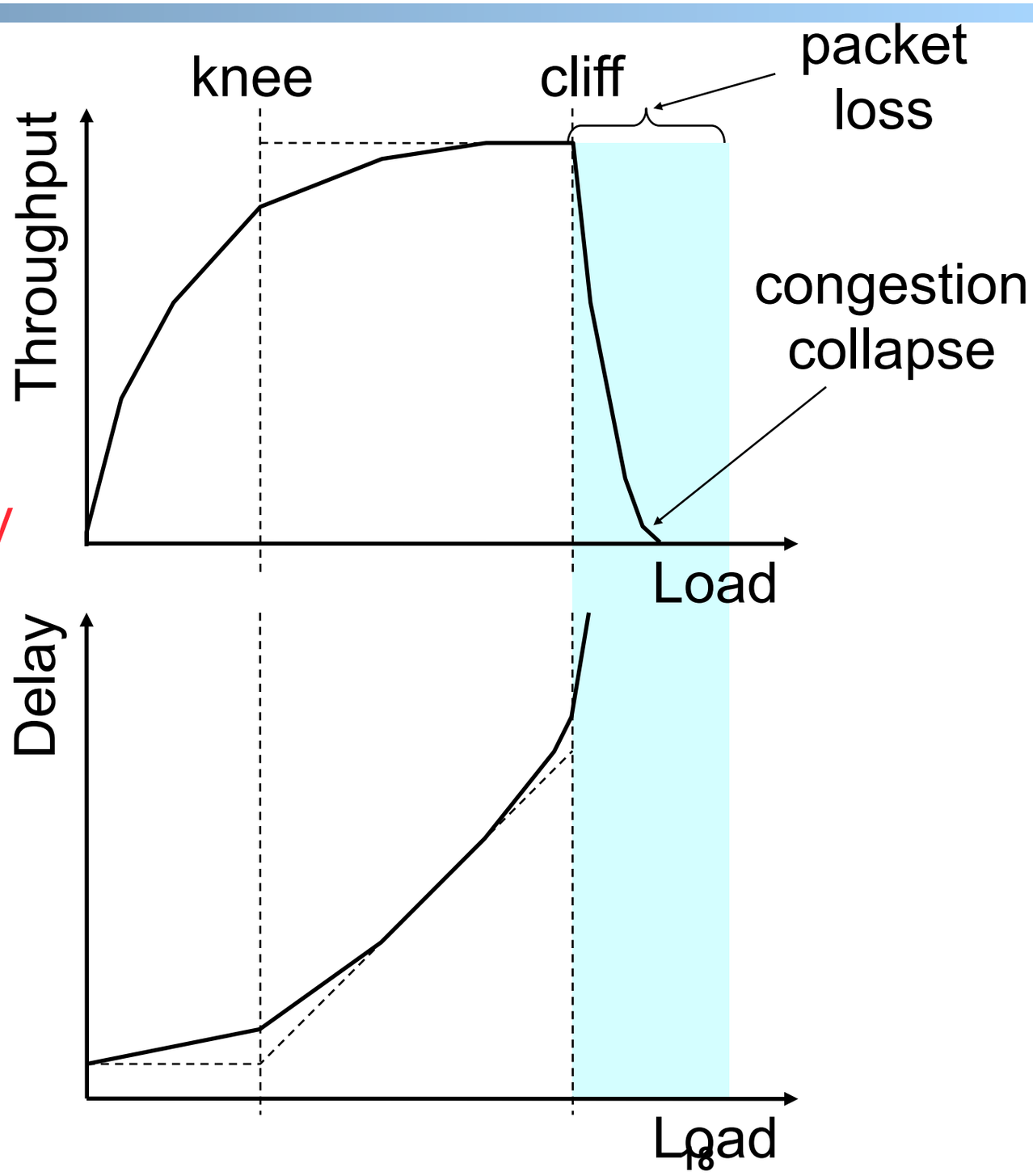
- Two papers:
  - Good science vs great engineering

Dangers of Increasing Load

- Knee – point after which
  - Throughput increases very slow
  - Delay increases fast

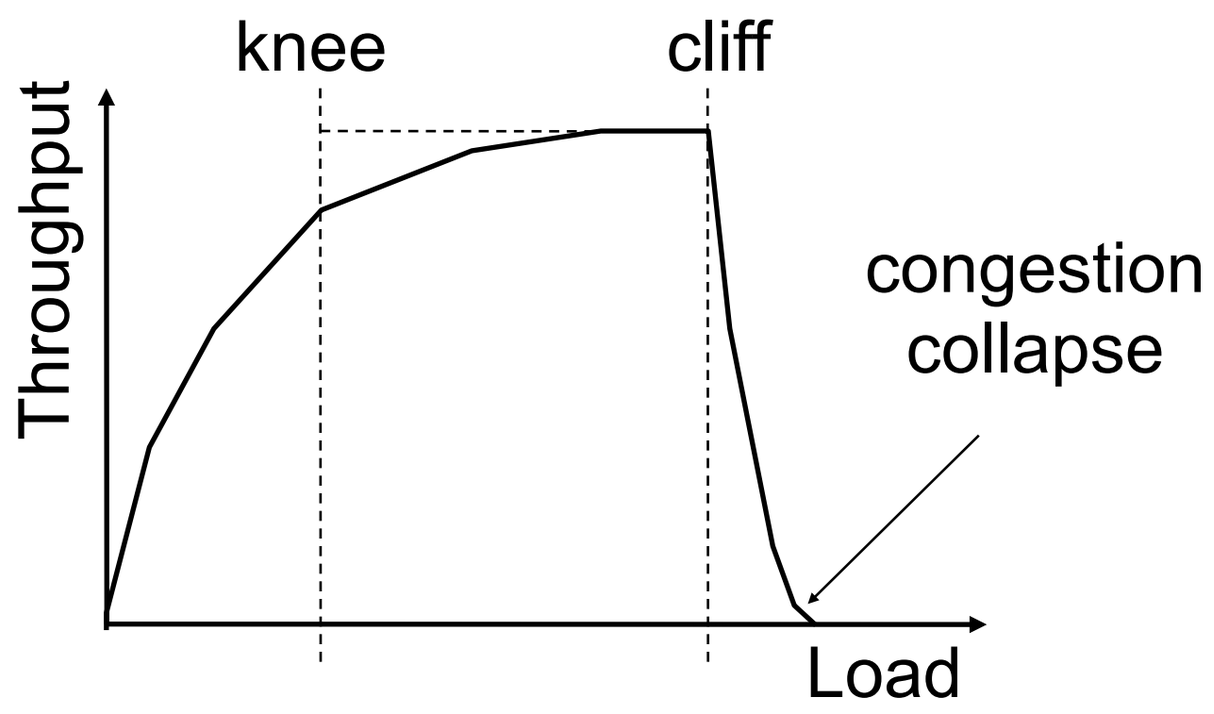
- Cliff – point after which
  - Throughput starts to decrease very fast to zero (congestion collapse)
  - Delay approaches infinity

- In an M/M/1 queue
  - Delay = 1/(1 – utilization)

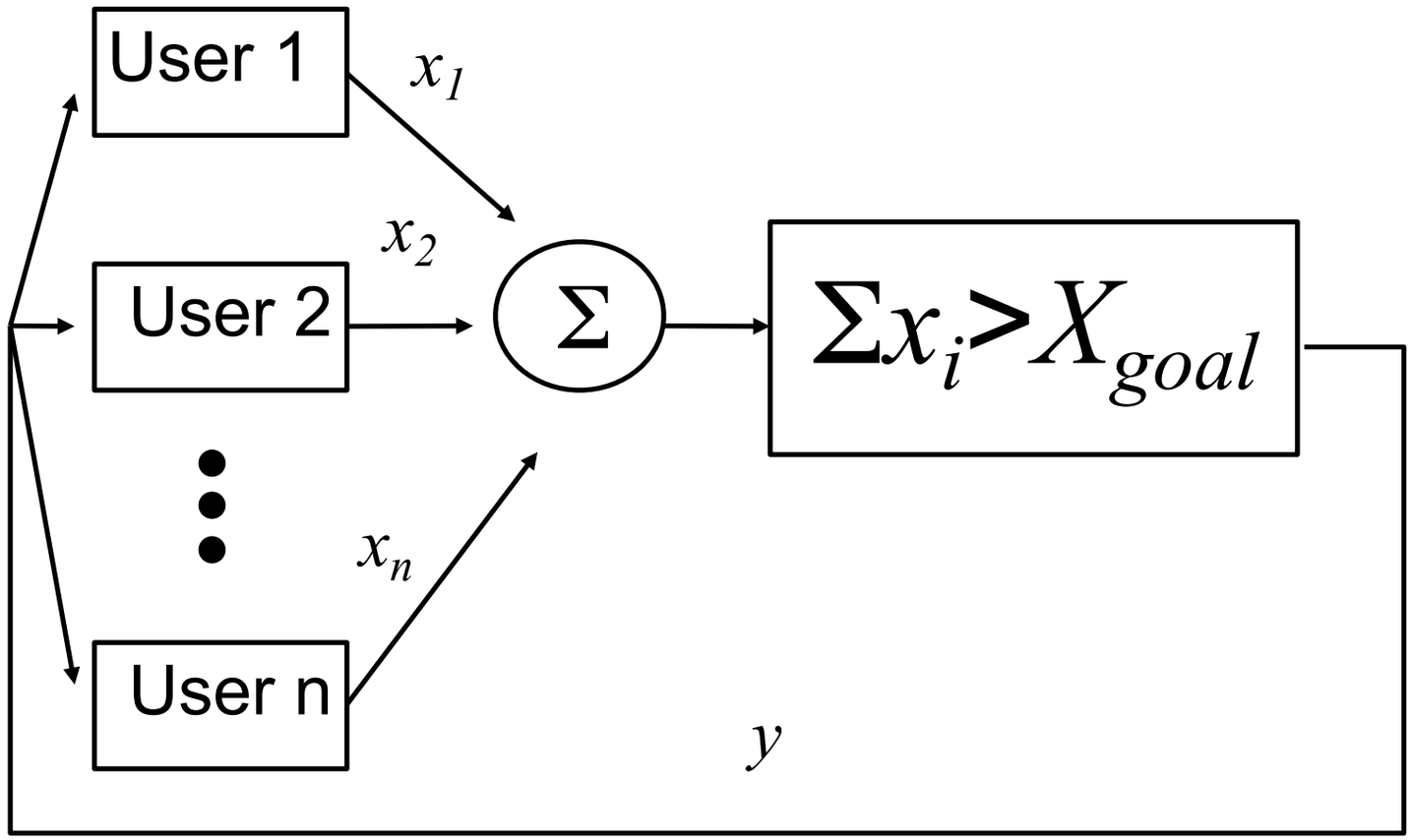


Cong. Control vs. Cong. Avoidance

- Congestion control goal
  - Stay left of cliff
- Congestion avoidance goal
  - Stay left of knee



Control System Model [CJ89]



- Simple, yet powerful model
- Explicit binary signal of congestion

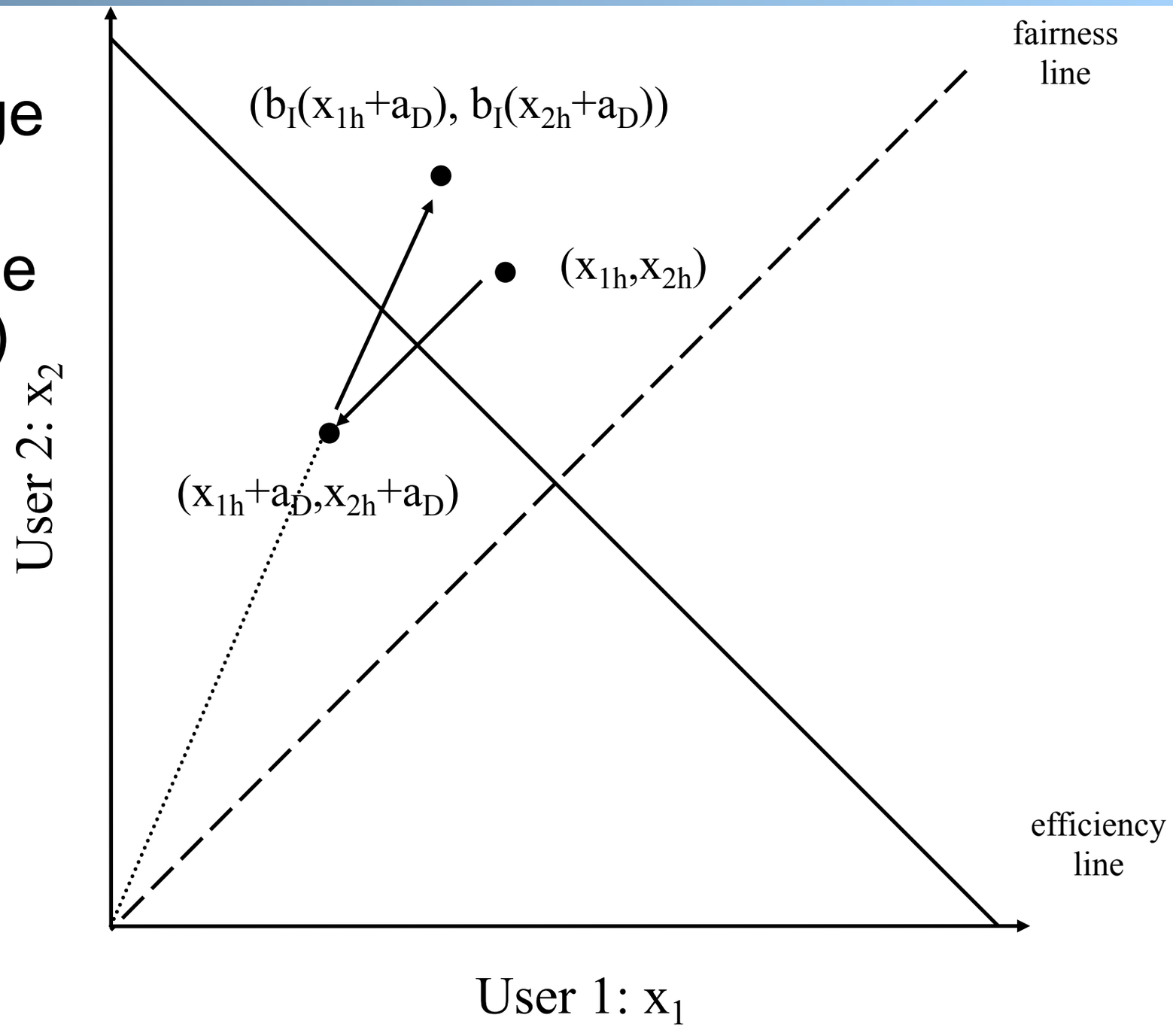
Possible Choices

$$x_i(t+1) = \begin{cases} a_I + b_I x_i(t) & increase \\ a_D + b_D x_i(t) & decrease \end{cases}$$

- Multiplicative increase, additive decrease
  - $a_I=0$, $b_I>1$, $a_D<0$, $b_D=1$
- Additive increase, additive decrease
  - $a_I>0$, $b_I=1$, $a_D<0$, $b_D=1$
- Multiplicative increase, multiplicative decrease
  - $a_I=0$, $b_I>1$, $a_D=0$, $0<b_D<1$
- Additive increase, multiplicative decrease
  - $a_I>0$, $b_I=1$, $a_D=0$, $0<b_D<1$
- Which one should we pick?

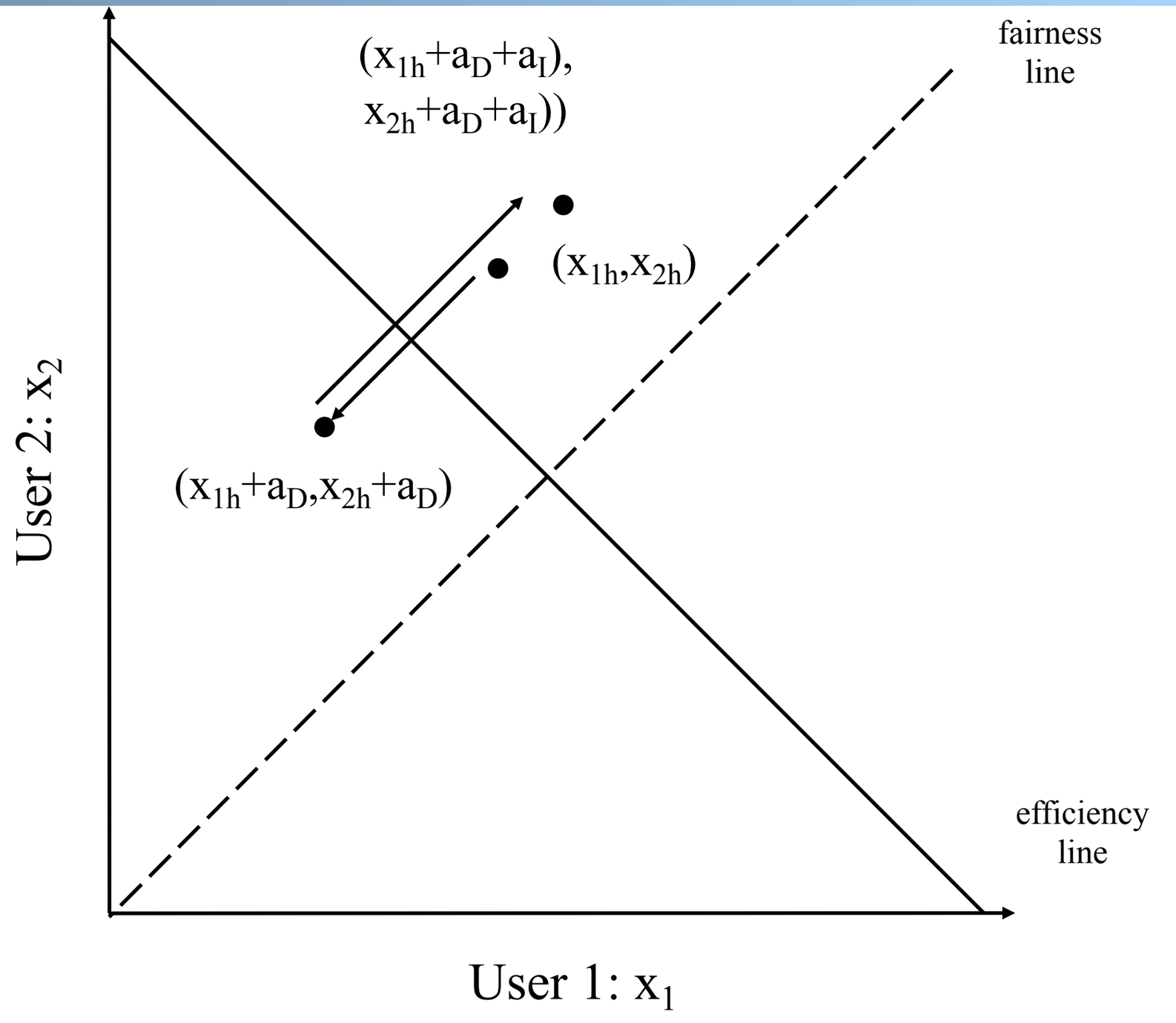
Multiplicative Increase, Additive Decrease

- Does not converge to fairness
- (Additive decrease worsens fairness)



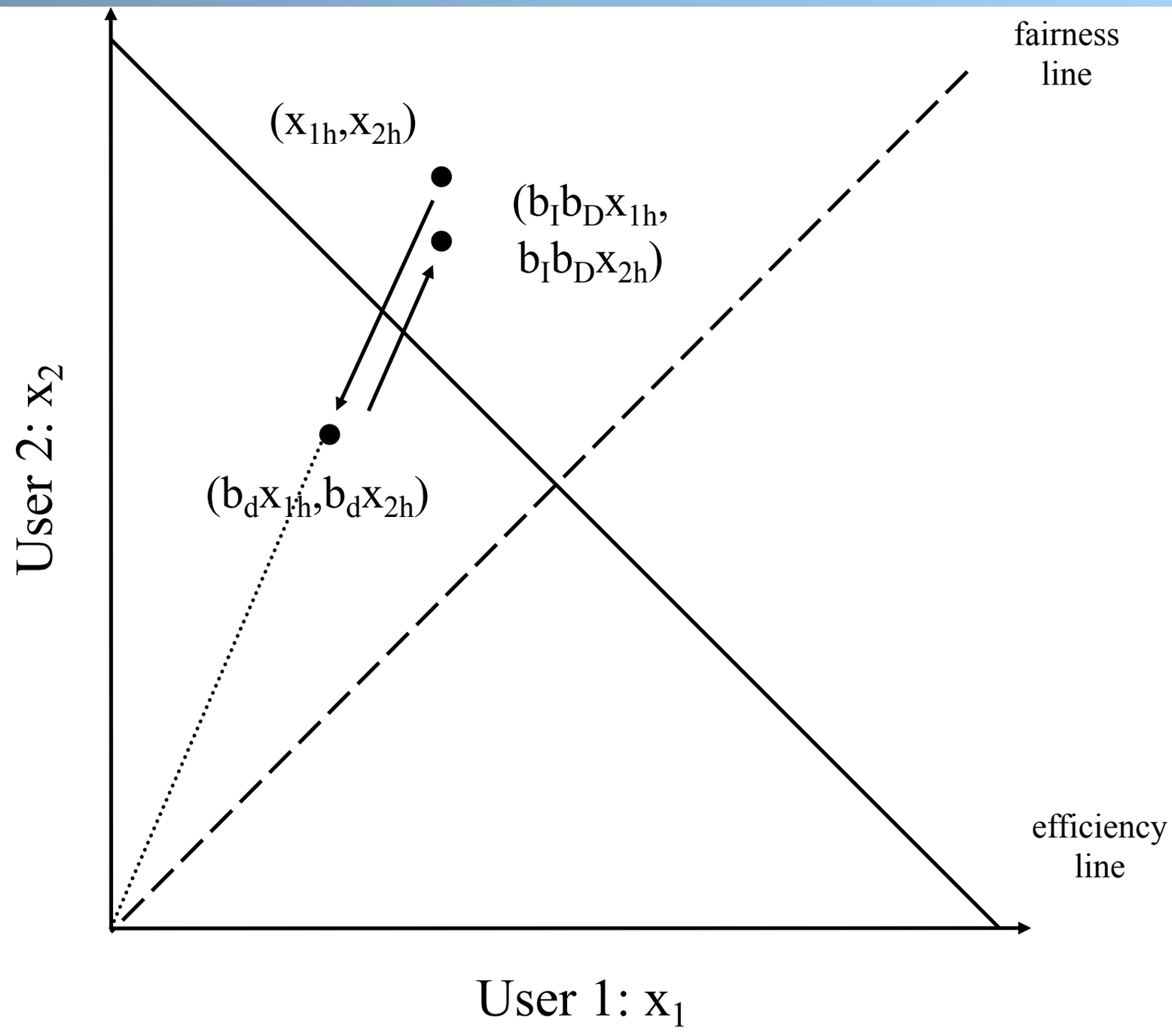
Additive Increase, Additive Decrease

- Reaches stable cycle, but does not converge to fairness



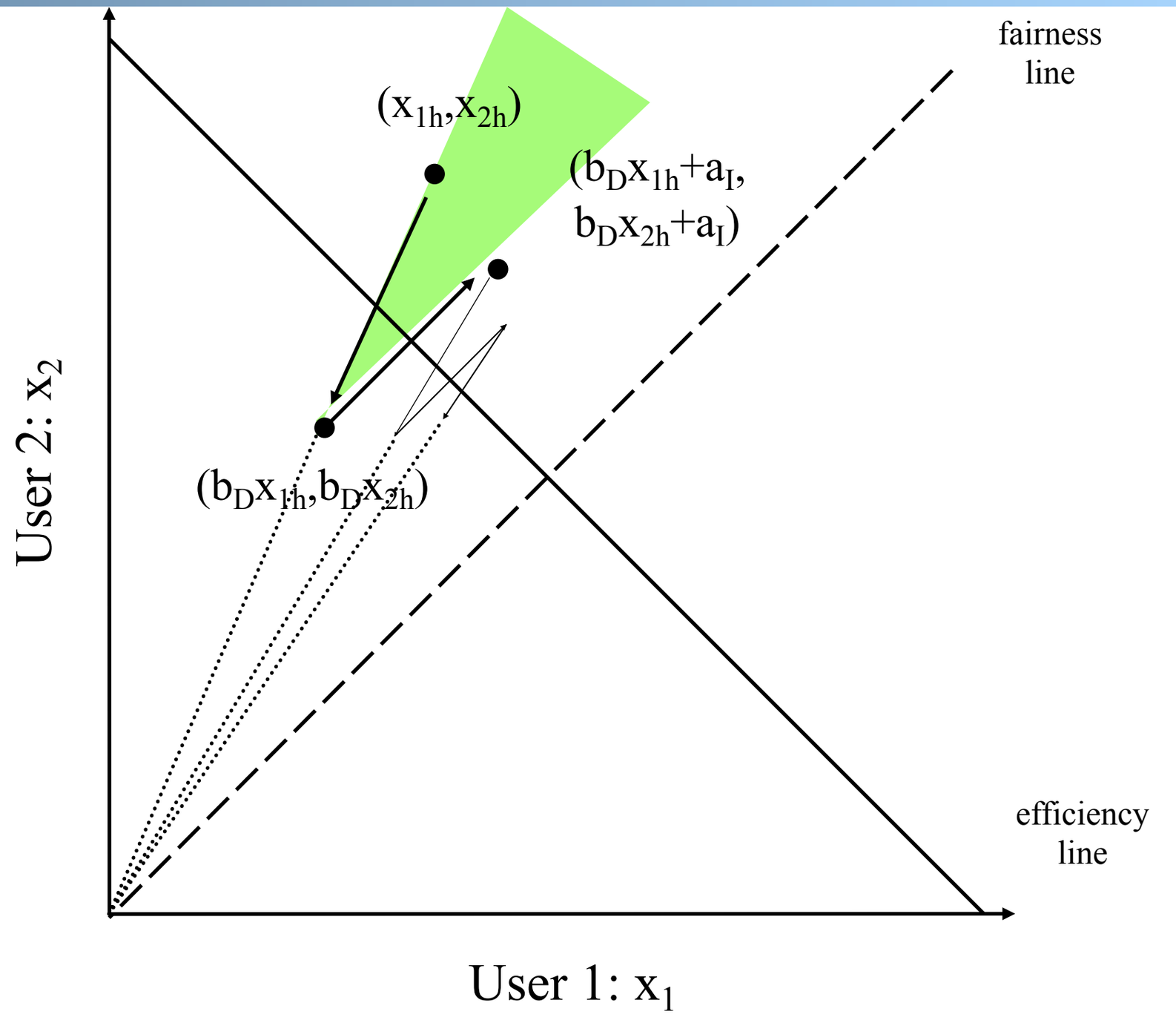
Multiplicative Increase, Multiplicative Decrease

- Converges to stable cycle, but is not fair



Additive Increase, Multiplicative Decrease

- Converges to stable and fair cycle



Modeling

- Critical to understanding complex systems
  - [CJ89] model relevant after 15 years, 10^6 increase of bandwidth, 1000x increase in number of users

- Criteria for good models
  - Two conflicting goals: reality and simplicity
  - Realistic, complex model → too hard to understand, too limited in applicability
  - Unrealistic, simple model → can be misleading

TCP Congestion Control

- [CJ89] provides theoretical basis for basic congestion avoidance mechanism

- Must turn this into real protocol

TCP Congestion Control

- Maintains three variables:
  - cwnd: congestion window
  - flow_win: flow window; receiver advertised window
  - Ssthresh: threshold size (used to update cwnd)
  -
- For sending, use: win = **min**(flow_win, cwnd)

TCP: Slow Start

- Goal: reach knee quickly

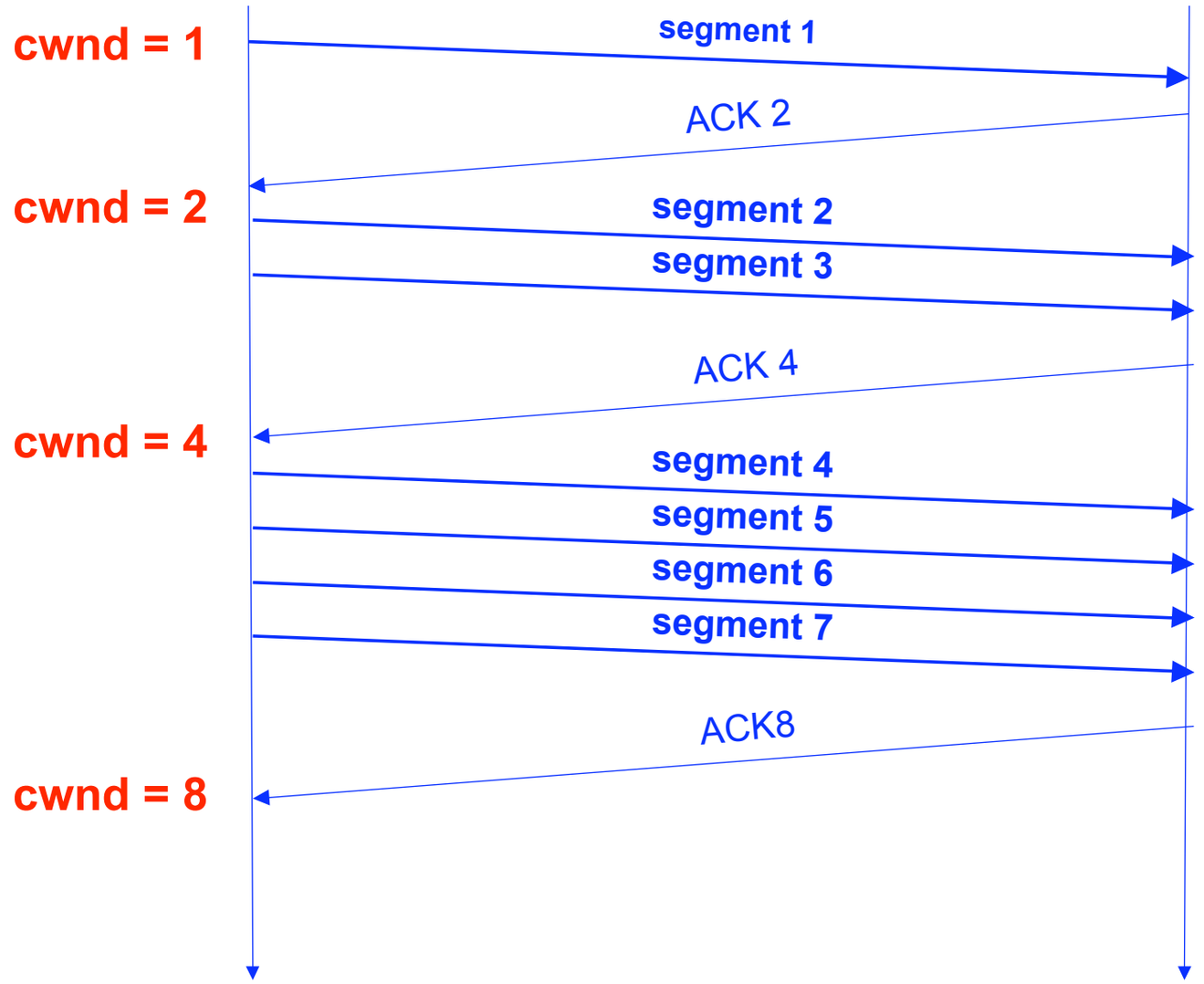
- Upon starting (or restarting):
  - Set *cwnd* =1
  - Each time a segment is acknowledged increment *cwnd* by one (*cwnd*++).

- Slow Start is not actually slow
  - *cwnd* increases exponentially

Slow Start Example

- The congestion window size grows very rapidly

- TCP slows down the increase of *cwnd* when ***cwnd >= ssthresh***

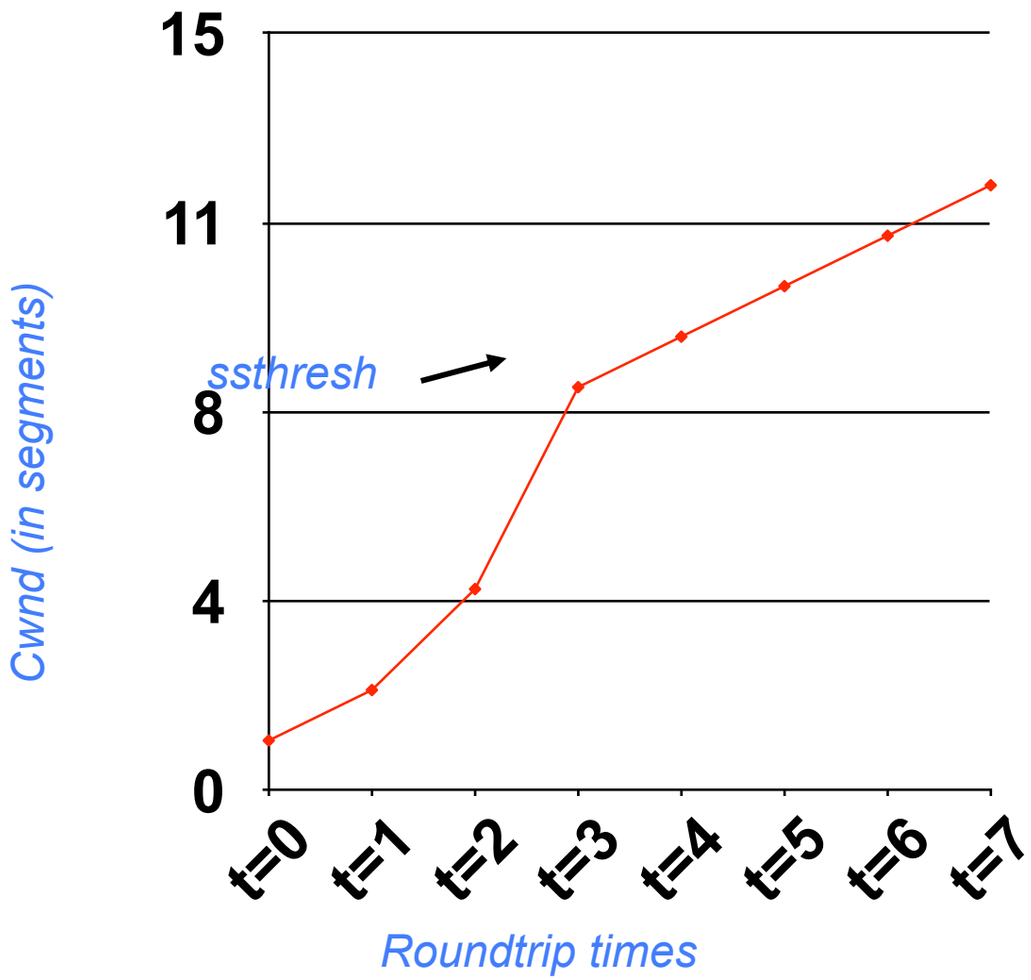


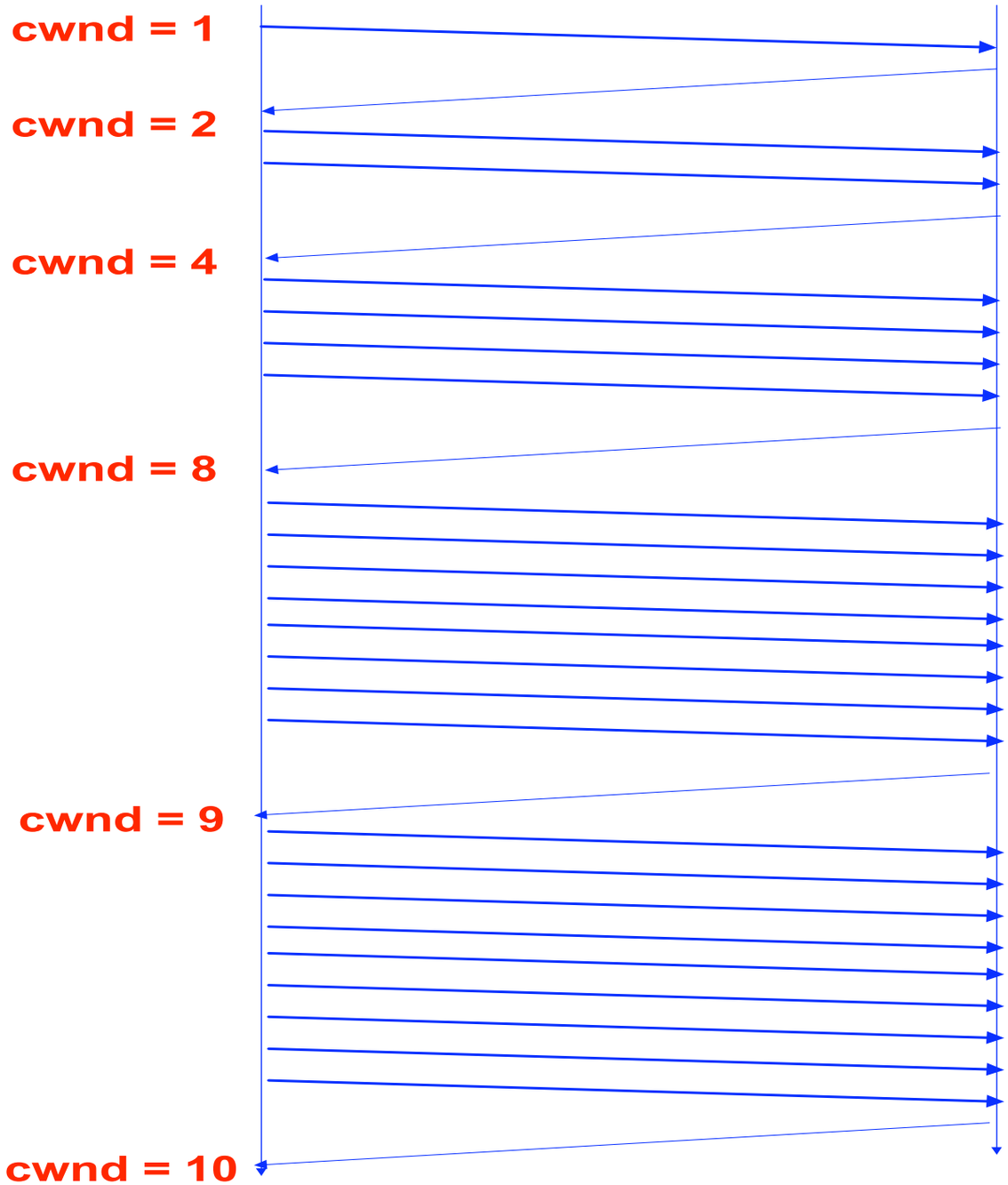
Congestion Avoidance

- Slow down “Slow Start”
- *ssthresh* is lower-bound guess about location of knee
- **If** *cwnd* > *ssthresh* **then**
  each time a segment is acknowledged
  increment *cwnd* by *1/cwnd* (*cwnd* += *1/cwnd*).
- So *cwnd* is increased by one only if all segments have been acknowledged.

Slow Start/Congestion Avoidance Example

- Assume that *ssthresh = 8*





Putting Everything Together: TCP Pseudocode

Initially:

```
    cwnd = 1;
    ssthresh = infinite;
```

New ack received:

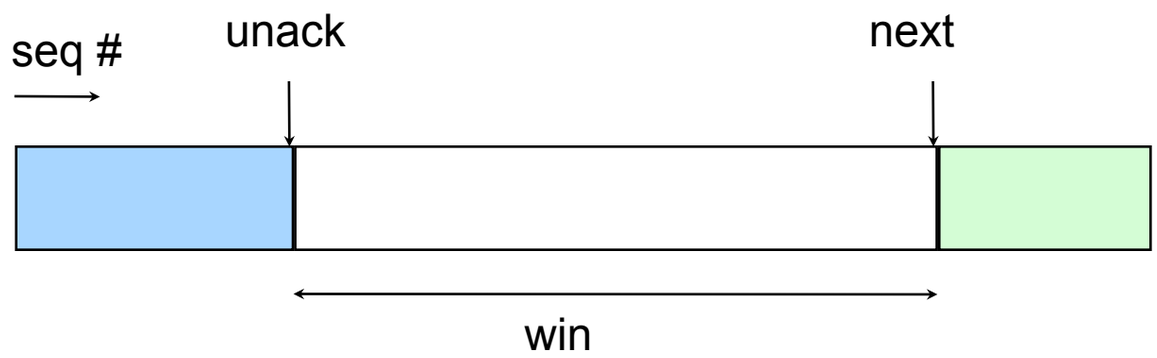
```
    if (cwnd < ssthresh)
        /* Slow Start*/
        cwnd = cwnd + 1;
    else
        /* Additive increase */
        cwnd = cwnd + 1/cwnd;
```

Timeout:

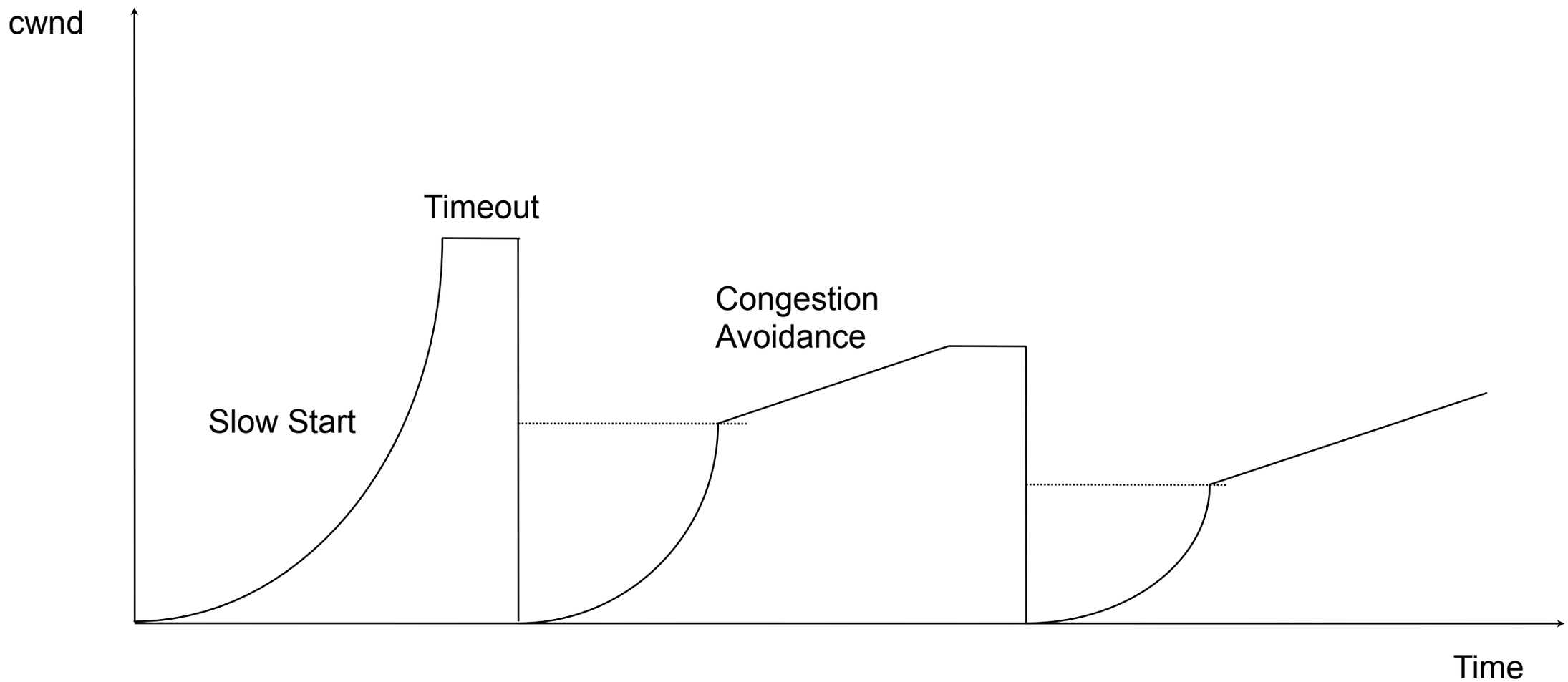
```
    /* Multiplicative decrease */
    ssthresh = cwnd/2;
    cwnd = 1;
```

```
while (next < unack + win)
    transmit next packet;

where   win = min(cwnd,
                  flow_win);
```

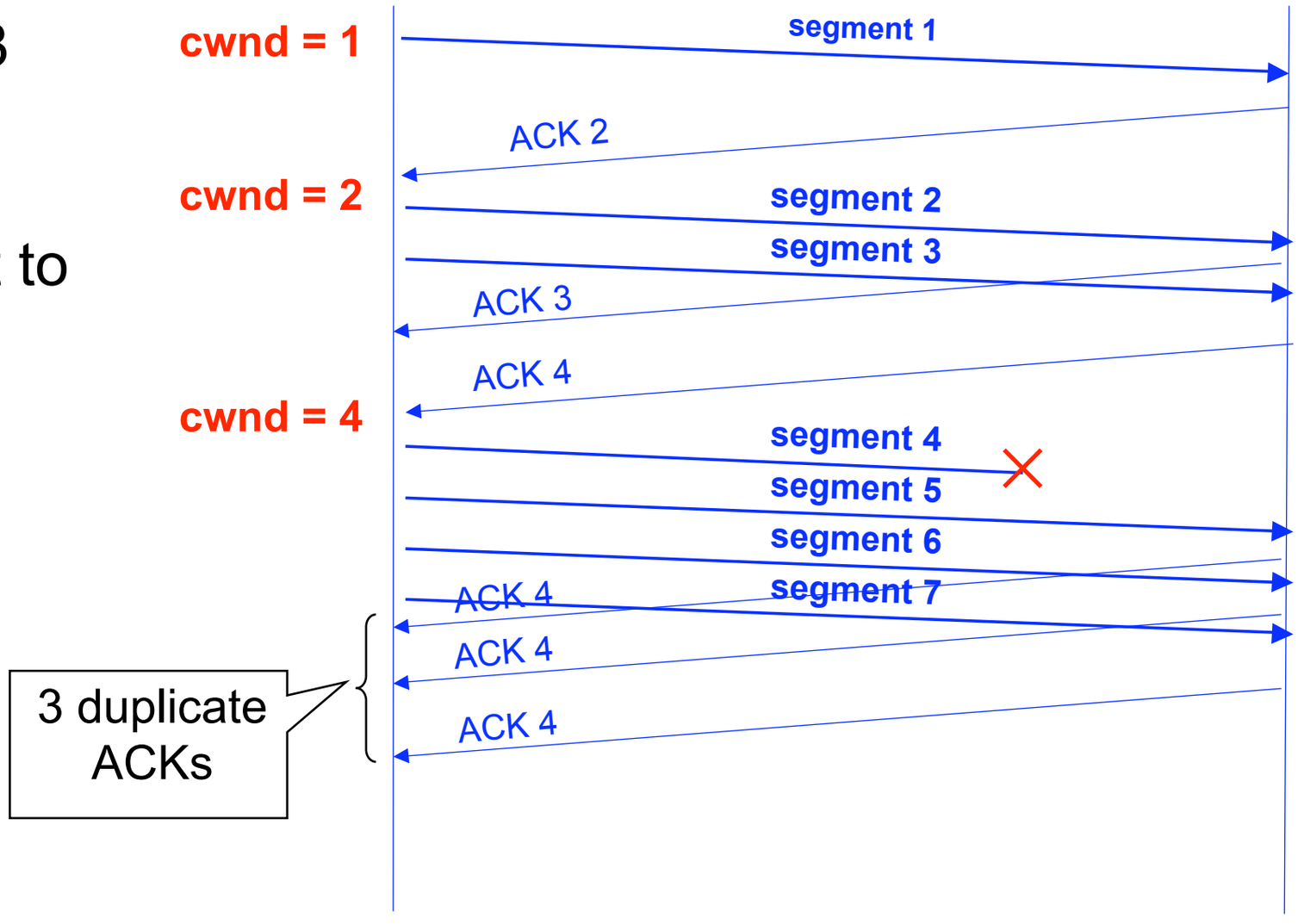


The big picture (so far)



Fast Retransmit

- Resend a segment after 3 duplicate ACKs
- Avoids waiting for timeout to discover loss



Fast Recovery

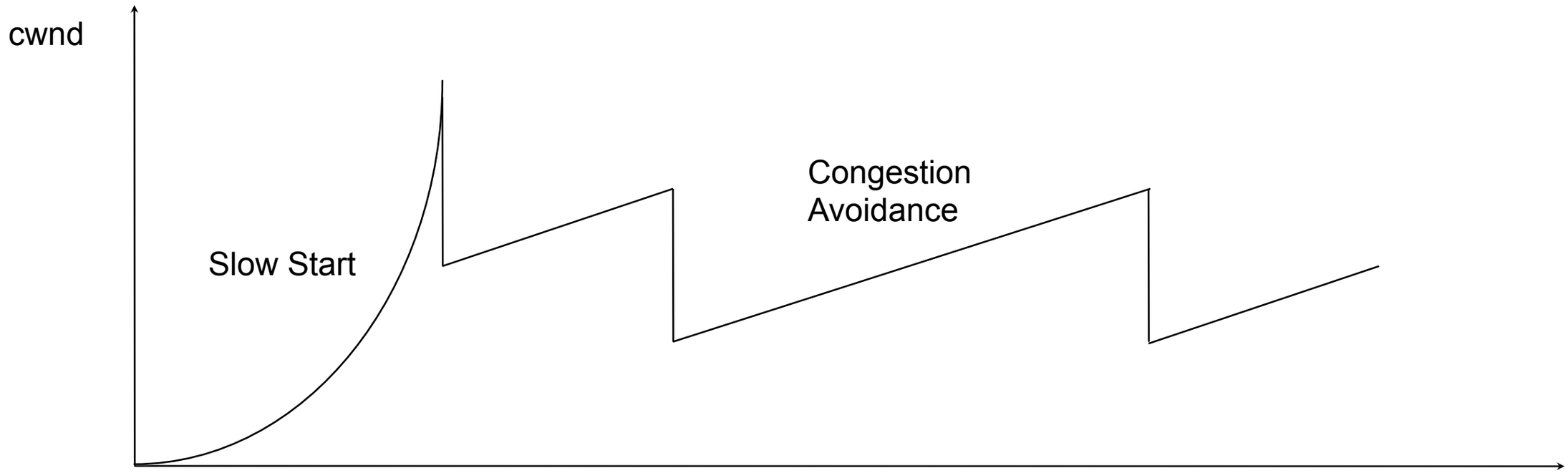
- After a fast-retransmit set *cwnd* to *ssthresh/2*
  - i.e., don't reset *cwnd* to 1

- But when RTO expires still do *cwnd* = 1

- Fast Retransmit and Fast Recovery
  - Implemented by TCP Reno
  - Most widely used version of TCP today

- Lesson: avoid RTOs at all costs!

Big picture with fast retransmit and Fast Recovery



- Retransmit after 3 duplicated acks
  - prevent expensive timeouts
- No need to slow start again
- At steady state, *cwnd* oscillates around the optimal window size.

Engineering vs Science in CC

- Great engineering built useful protocol:
  - TCP Reno, etc.

- Good science by Chiu, Jain and others
  - Basis for understanding why it works so well

Extensions to TCP

- Selective acknowledgements: TCP SACK
- Explicit congestion notification: ECN
- Delay-based congestion avoidance: TCP Vegas
- Discriminating between congestion losses and other losses: cross-layer signaling and guesses
- Randomized drops (RED) and other router mechanisms

Today

TCP congestion control

Limitations of TCP CC

RED

Limitations of TCP CC

- In what ways is TCP congestion control broken or suboptimal?

A partial list...

Efficiency

- Tends to fill queues (adding latency)
- Slow to converge (for short flows or links with high bandwidth•delay product)
- Loss ≠ congestion
- May not fully utilize bandwidth

A partial list...

Fairness

- Unfair to large-RTT flows (less throughput)
- Unfair to short flows if ssthresh starts small
- Equal rates isn't necessarily "fair" or best
- Vulnerable to selfish & malicious behavior
  - TCP assumes everyone is running TCP!

Today

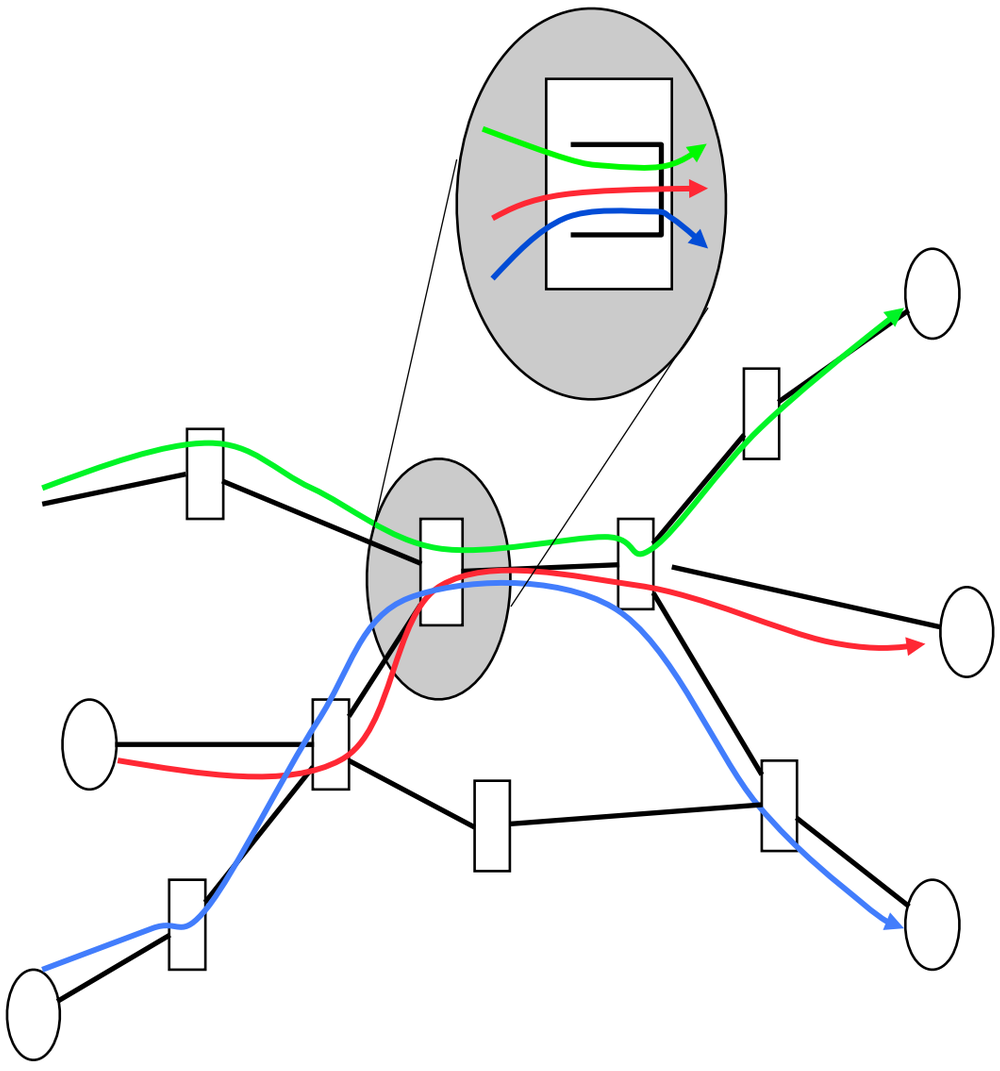
TCP congestion control

Limitations of TCP CC

RED

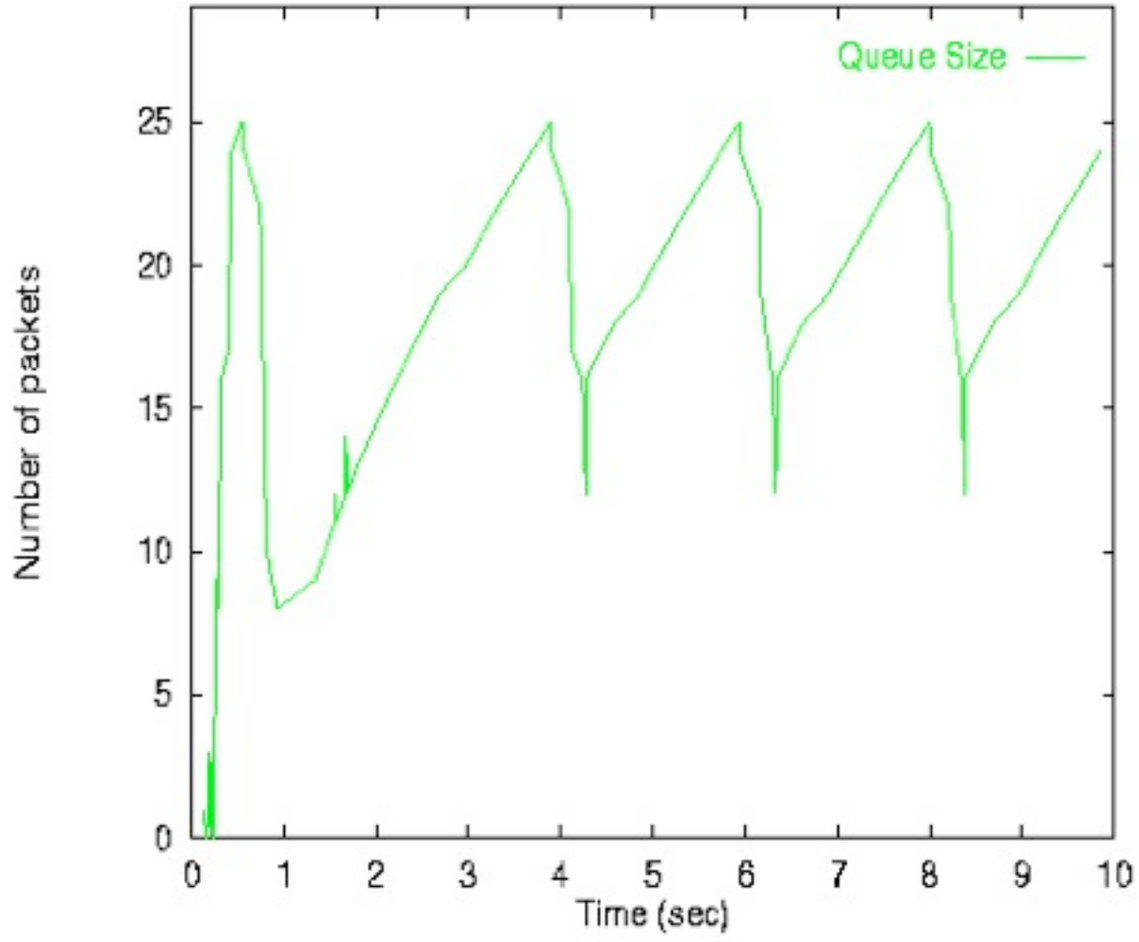
Traditional queueing

- Traditional Internet
  - Congestion control mechanisms at end-systems, mainly implemented in TCP
  - Routers play little role
- Router mechanisms affecting congestion management
  - Scheduling
  - Buffer management
- Traditional routers
  - FIFO
  - Tail drop

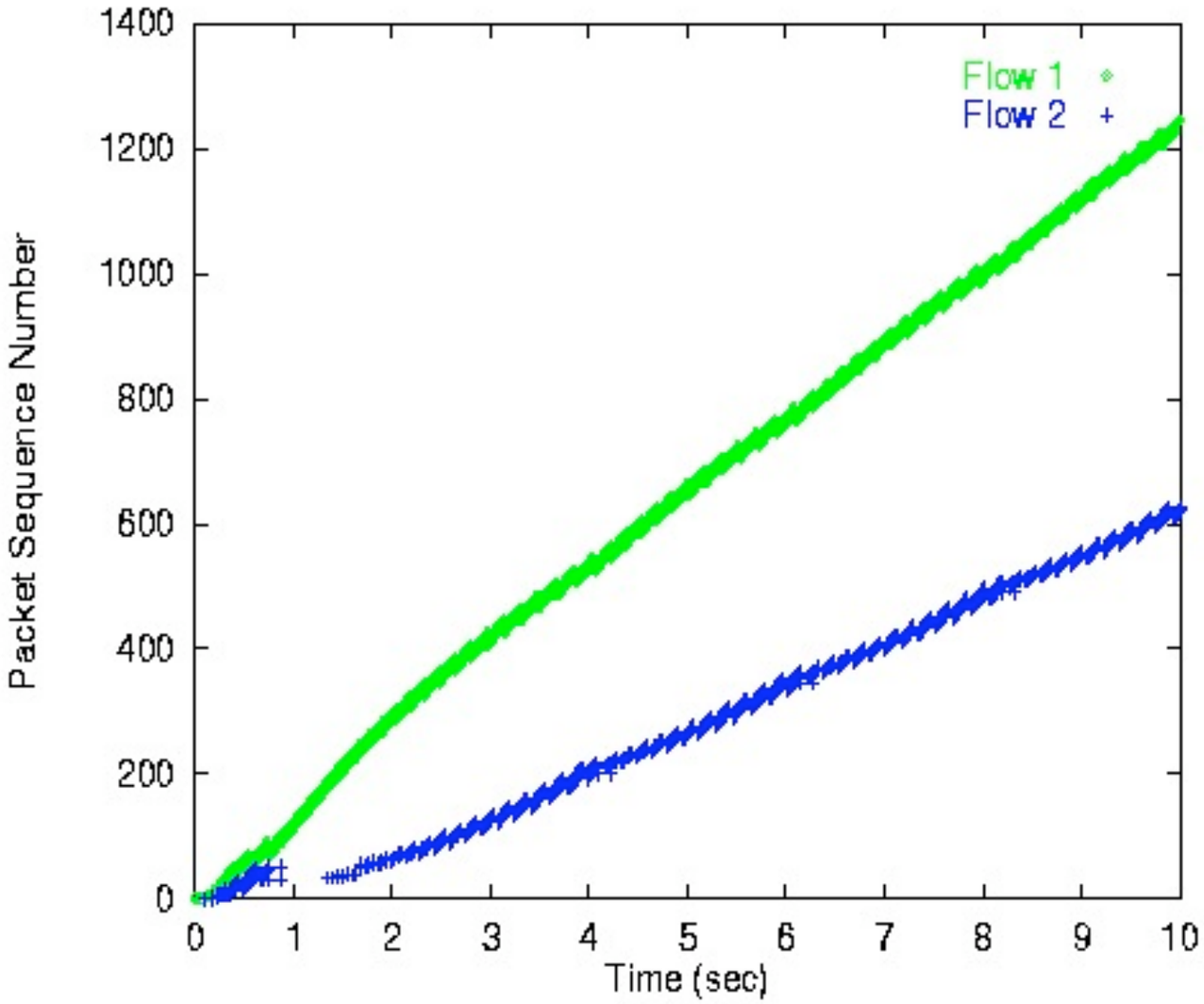


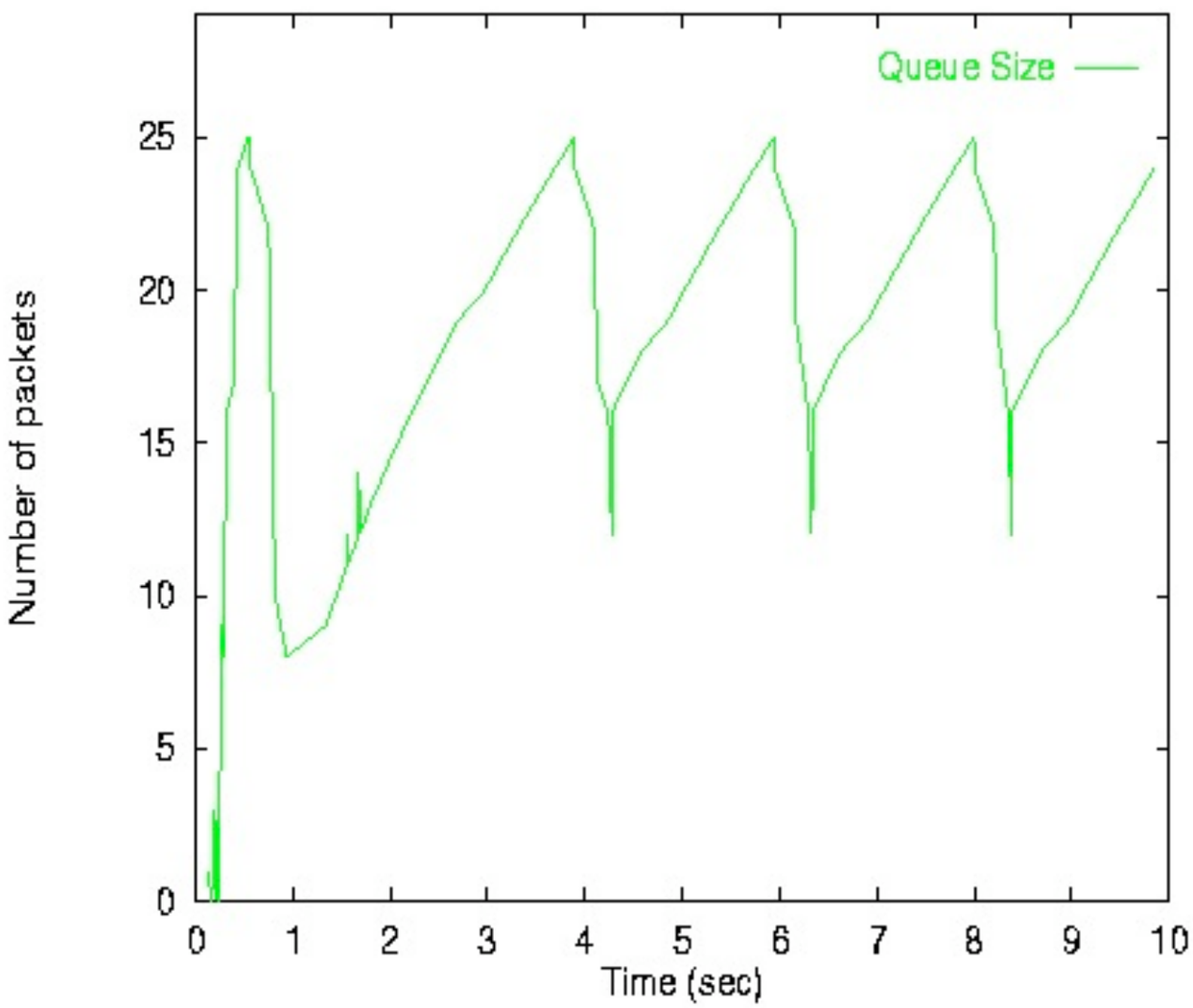
Drawbacks of FIFO with Tail-drop

- Buffer lock out by misbehaving flows
- Synchronizing effect for multiple TCP flows
- Burst or multiple consecutive packet drops
  - Bad for TCP fast recovery



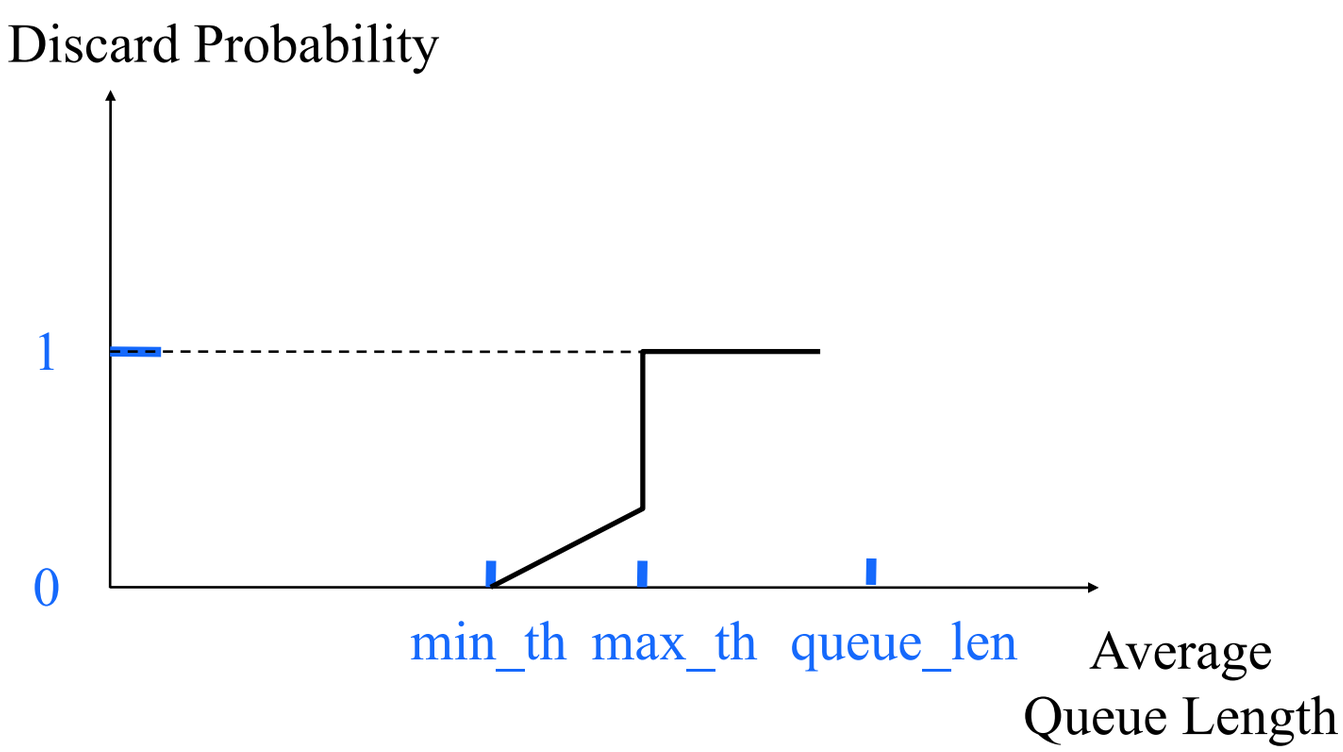
FIFO Router with Two TCP Sessions





RED (Random Early Detection)

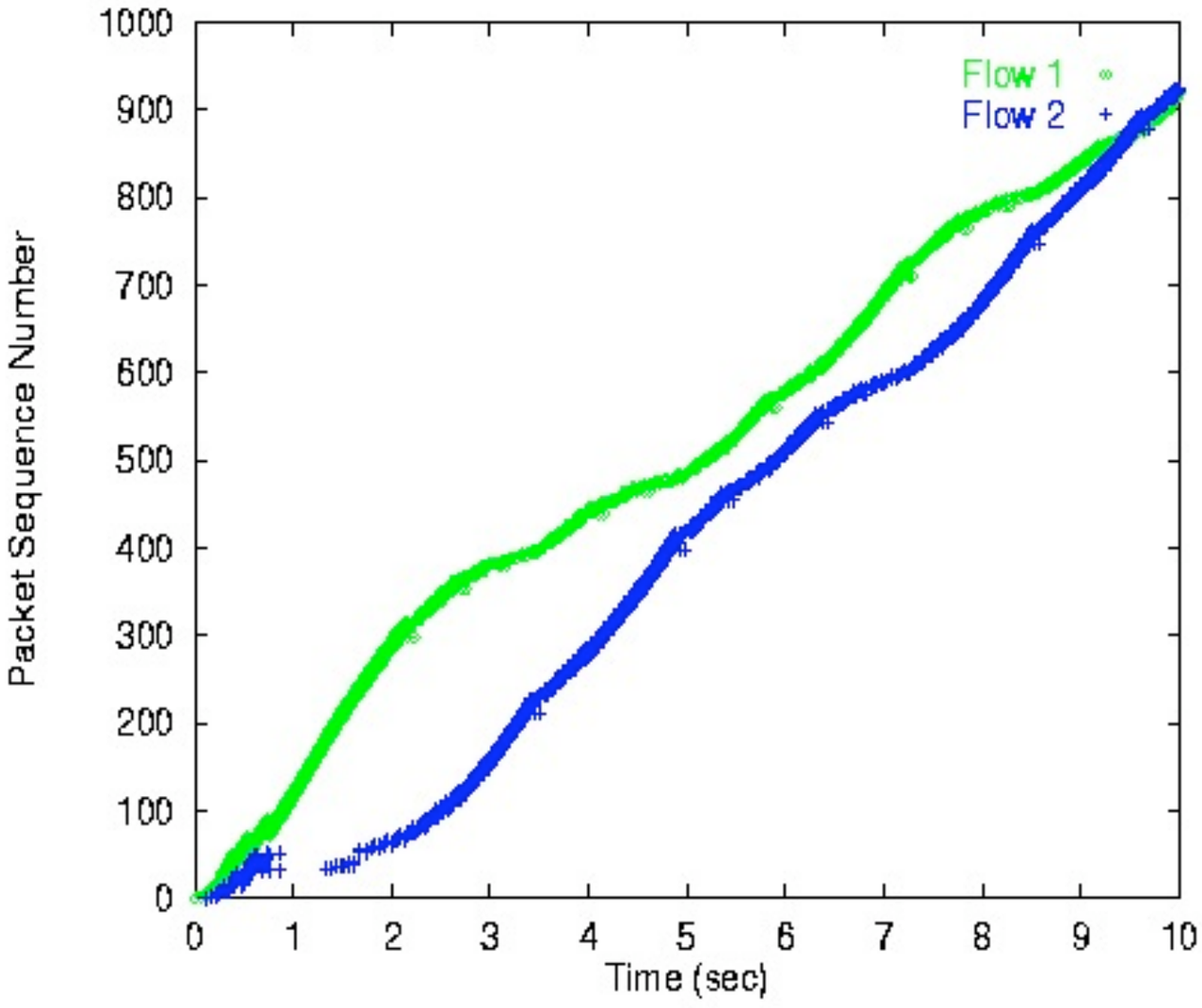
- FIFO scheduling
- Buffer management:
  - Probabilistically discard packets
  - Probability is computed as a function of average queue length (why average?)

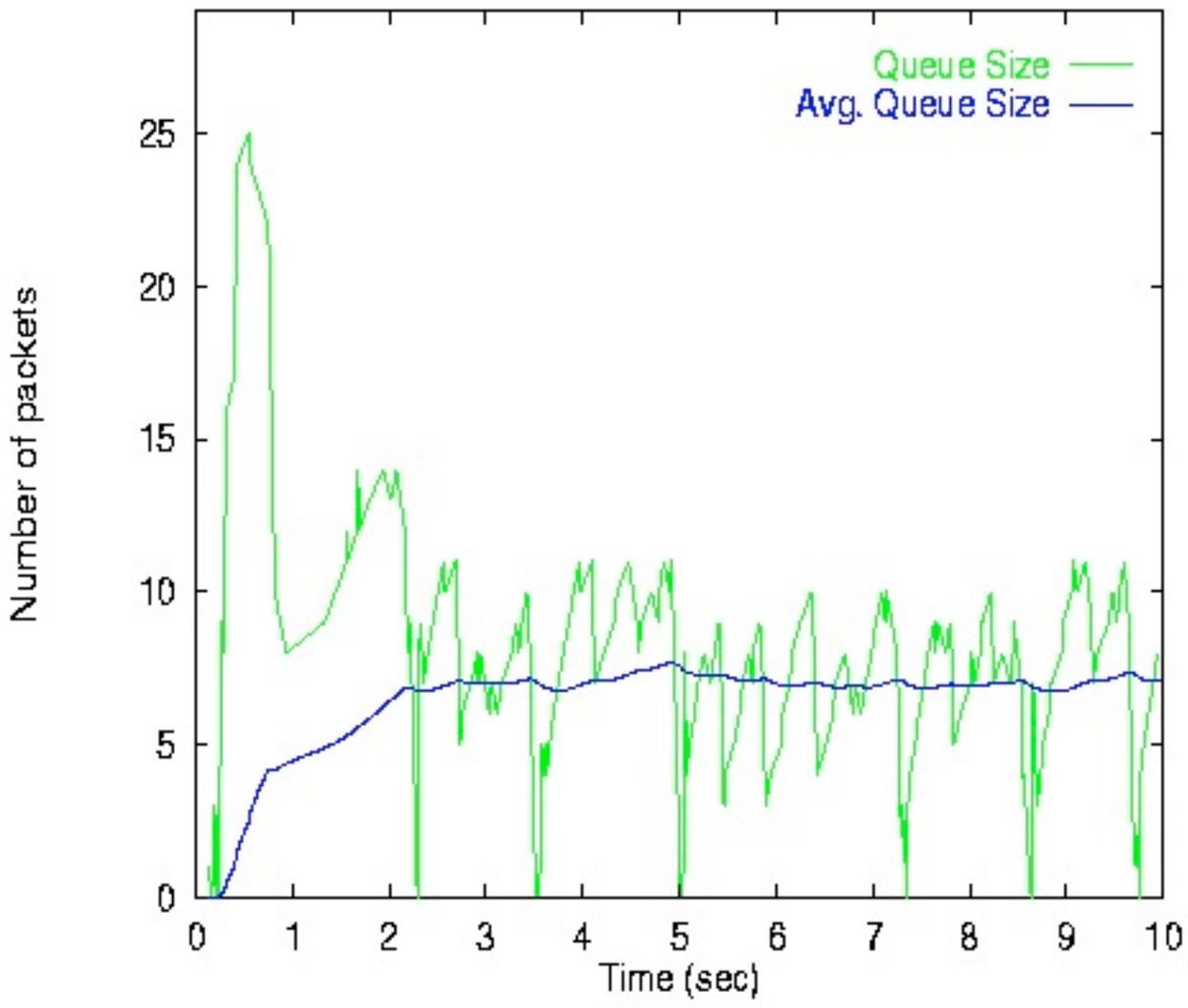


RED Advantages

- Absorb burst better
- Avoids synchronization
- Signal end systems earlier
- And XCP would be even better than RED in these regards

RED Router with Two TCP Sessions





But still no isolation between flows

- No protection: if a flow misbehaves it will hurt the other flows
- Example: 1 UDP (10 Mbps) and 31 TCP's sharing a 10 Mbps link