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[simula . research laboratory]

Congestion Control

INF3190

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Congestion and Flow Control

Opposite objectives

- End-system
  - Optimize its own throughput
  - Possibly at the expense of other end-systems

Opposite objectives

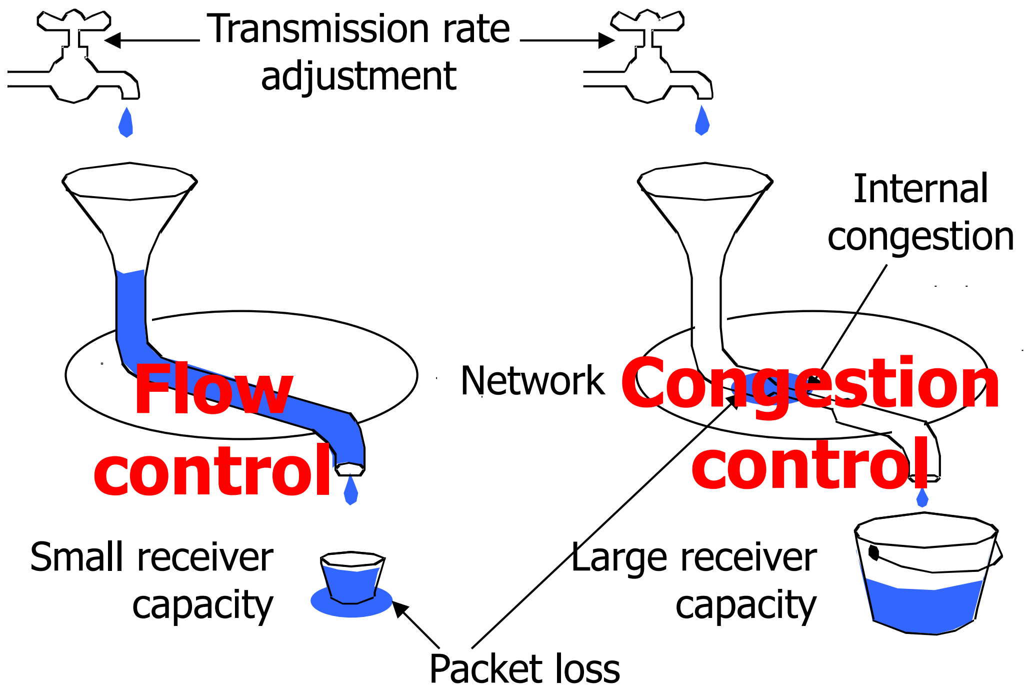
- Network
  - Optimize overall throughput

Two different problems

- Receiver capacity
- Network capacity

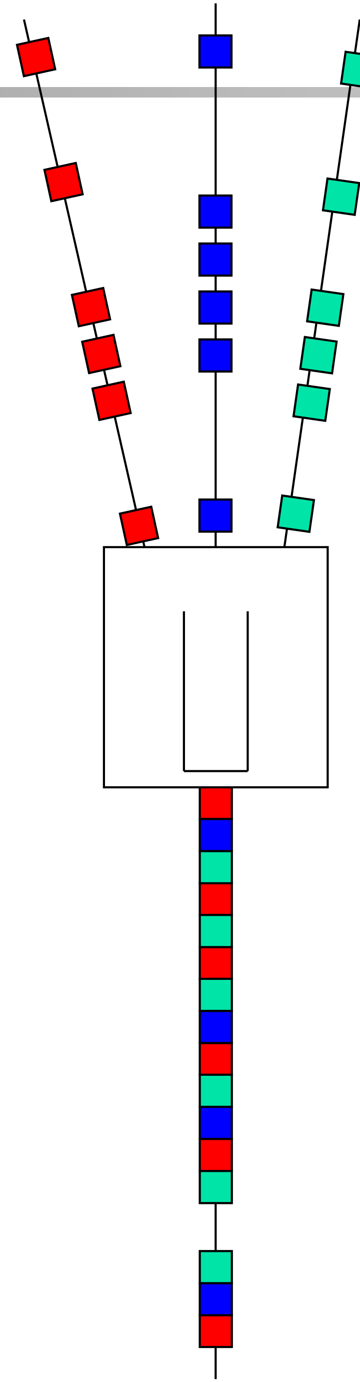
Cannot be distinguished easily at all places

But should be differentiated



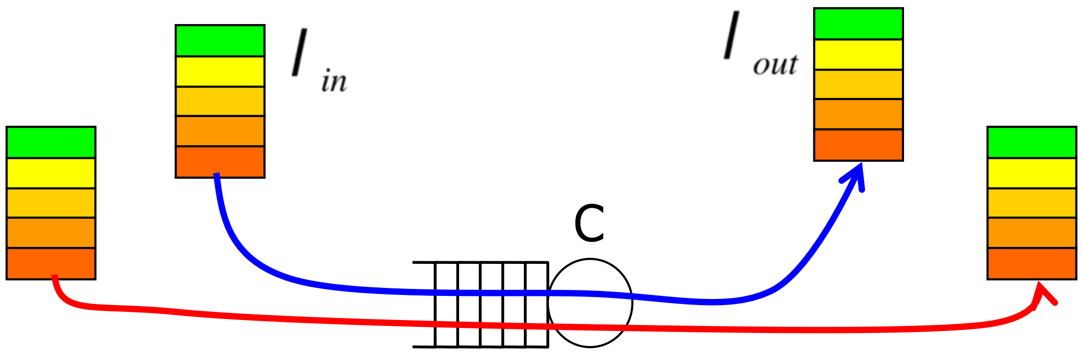
Congestion

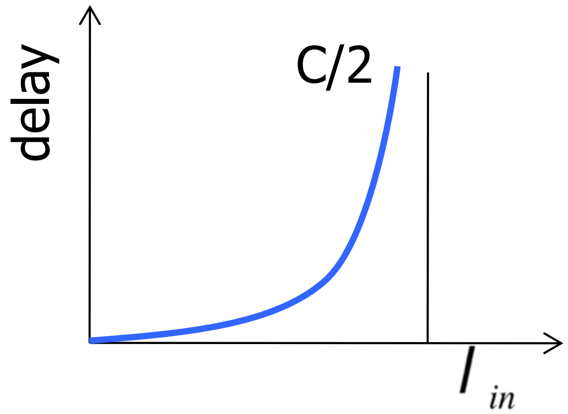
- ***Traffic***
  - All traffic from all sources
- ***Traffic class***
  - All packets from all sources with a common distinguishing property, e.g. priority
- ***Persistent congestion***
  - Router stays congested for a long time
  - Excessive traffic offered
- ***Transient congestion***
  - Congestion occurs for a while
  - Router is temporarily overloaded
  - Often due to ***burstiness***

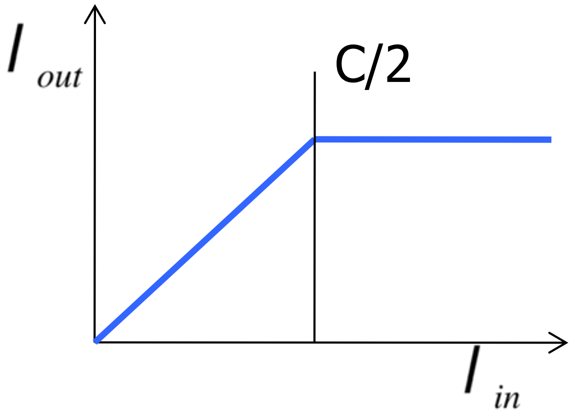


Reasons for congestions

- Two senders, two receivers
- One router, infinite buffers
- No retransmissions



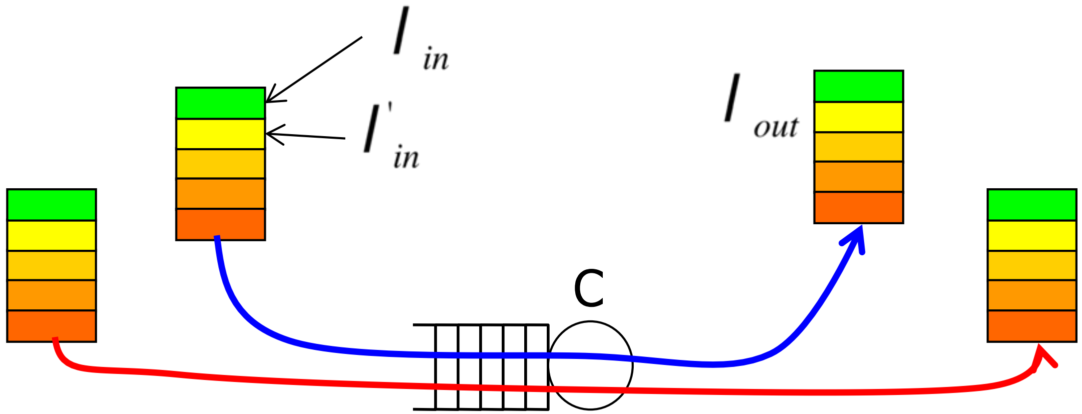




- Very long delays in case of congestion
- Maximum utilization of the networks

Reasons for congestions

- Two senders, two receivers
- One router, finite buffers
- *Retransmission* of lost packets



Router with finite buffers

l_{in} Data rate sent by the application

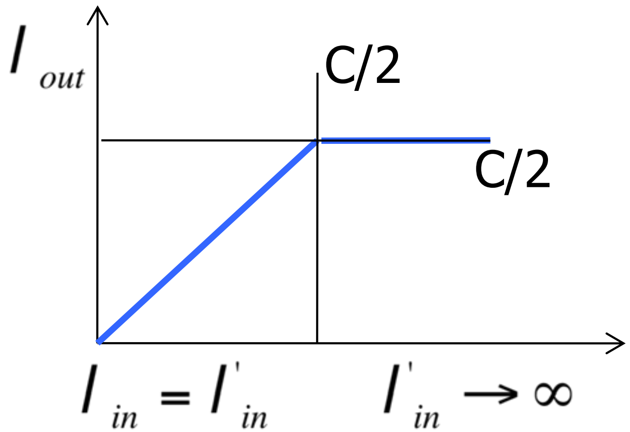
l'_{in} Higher data rate sent by the transport layer including retransmissions

Reasons for congestions

- Always $\lambda_{in} = \lambda_{out}$
- Perfect retransmission only in case of loss $\lambda'_{in} > \lambda_{out}$
- Retransmission of delayed (but not lost) packets increases λ'_{in} above the perfect value, without increasing λ_{out}

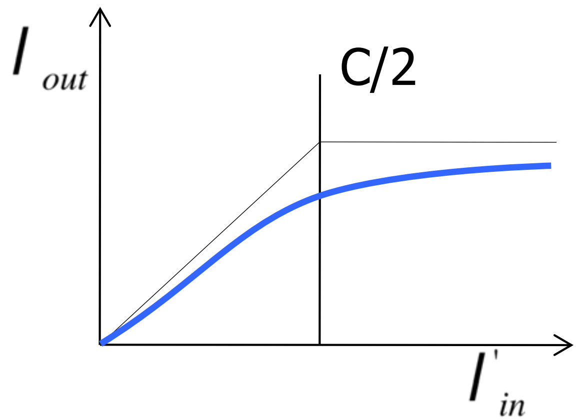
- "Cost of congestion":
  - More work (retransmissions) for a desired throughput
  - Useless retransmissions, some links transmit several copies of the same packet

Constant bitrate (non-bursty) traffic



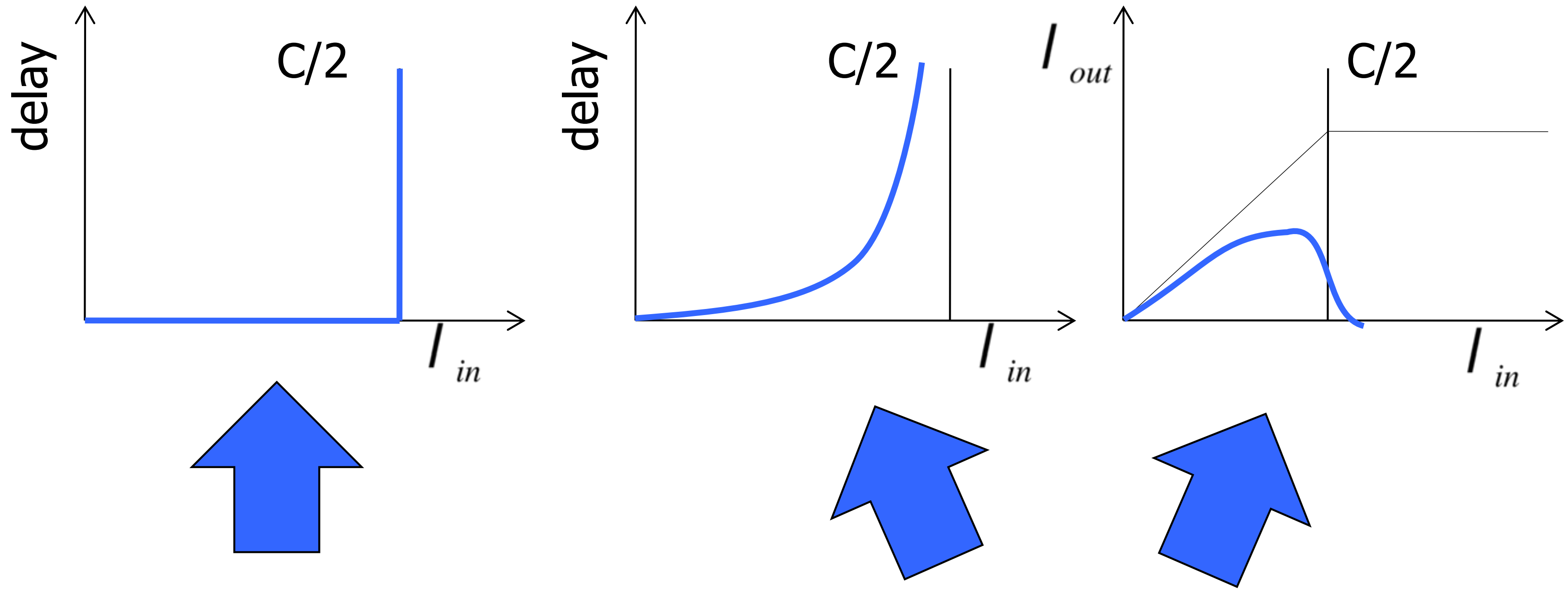
No congestion / persistent congestion

Bursty traffic



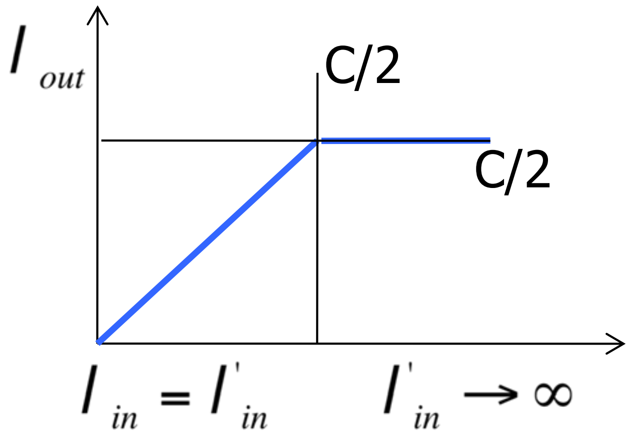
Transient congestion

Reasons for congestions

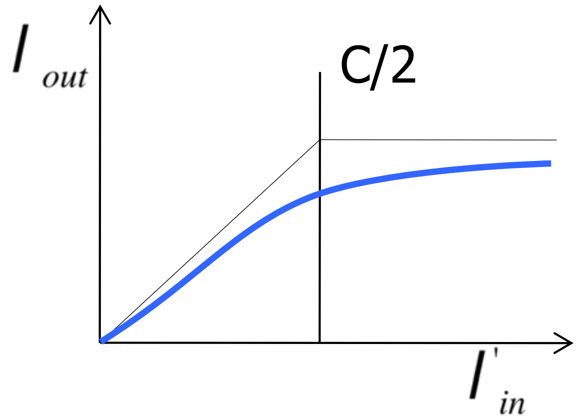


Constant bitrate (non-bursty) traffic

Bursty traffic



No congestion / persistent congestion



Transient congestion

Approaches to congestion control

Congestion Control

- Strategies
  - Increase capacity
  - Decrease traffic

Congestion Control

- Strategies
  - End-to-end congestion control
    - no explicit feedback from the network
    - congestion is detected by observed packet loss and delay
    - this is the approach of basic (regular) TCP
  - Network-assisted congestion control
    - router give feedback to end systems
    - choke packets (SNA, DECbit, ICMP, TCP/IP ECN, ATM)
    - explicit send rate (ATM, XCP)

Congestion Control

- Strategies
  - Repair
    - when congestion is noticed
    - explicit feedback (packets are sent from the point of congestion)
    - implicit feedback (source assumes that congestion occurred due to other effects)
    - Methods: drop packets, choke packets, hop-by-hop choke packets, fair queuing, …
  - Avoid
    - before congestion happens
    - initiate countermeasures at the sender
    - initiate countermeasures at the receiver
    - Methods: leaky bucket, token bucket, isarithmic congestion control, reservation, …

Repair

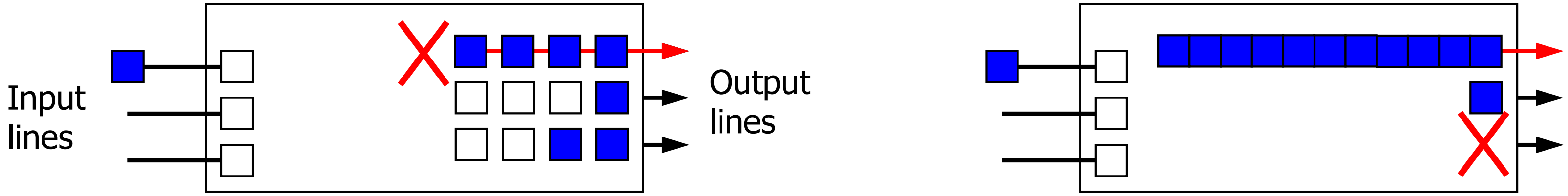
- Principle
  - No resource reservation
  - Necessary steps
    - Congestion detected
    - Introduce appropriate procedures for reduction

Repair by Packet dropping

- Principle
  - At each intermediate system
  - Queue length is tested
  - Incoming packet is dropped if it cannot be buffered
    - We may not wait until the queue is entirely full

- To provide
  - Unreliable service
    - No preparations necessary
  - Reliable service
    - Buffer packet until reception has been acknowledged

Repair by Packet dropping



- Assigning buffers to queues at output lines
  1. Fair distribution of buffers per output line
     - Packet may be dropped although there are free buffers
  2. Minimal number (usually 1) of buffers per output line
     - Sequences to same output line (“bursts”) lead to drops
  3. Dynamic buffer assignment
     - React badly to load shifting loads

Repair by Packet dropping

4. Content-related dropping: relevance

- Relevance of data connection as a whole
  or every packet from one end system to another end system
  - Examples
    - Favor IPv6 packets with flow id 0x4b5 over all others
    - Favor packets of TCP connection (65.246.255.51,80,129.240.69.49,53051) over all others

- Relevance of a traffic class
  - Examples
    - Favor ICMP packets over IP packets
    - Favor HTTP traffic (all TCP packets with source port 80) over FTP traffic
    - Favor packets from 65.246.0.0/16 over all others

Repair by Packet dropping

- Properties of packet dropping
  - Very simple

- But
  - Retransmitted packets waste bandwidth
  - Packet has to be sent 1 / (1 - p) times before it is accepted
    - (p ... probability that packet will be dropped)

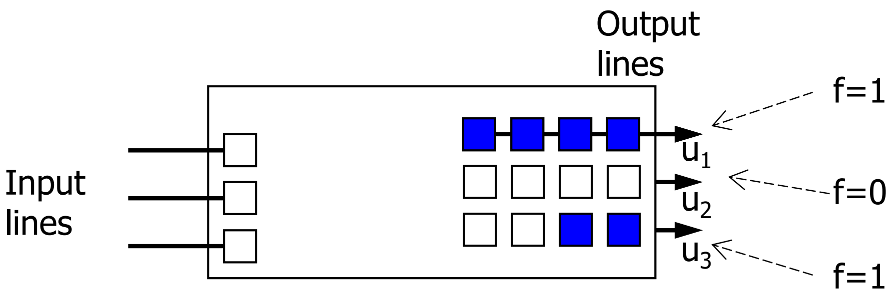
- Optimization necessary to reduce the waste of bandwidth
  - Dropping packets that have not gotten that far yet

  ⇒ e.g. Choke packets

Repair by Choke Packets

- Principle
  - Reduce traffic during congestion by telling source to slow down

- Procedure for router
  - Each outgoing line has one variable
    - Utilization u ($0 \le u \le 1$)



  - Calculating u: Router checks the line usage f periodically (f is 0 or 1)
    - $u = a * u + (1 - a) * f$
    - $0 \le a \le 1$ determines to what extent "history" is taken into account

  - u > threshold: line changes to condition "warning"
    - Send choke packet to source (indicating destination)
    - Tag packet (to avoid further choke packets from down stream router) & forward it

Repair by Choke Packets

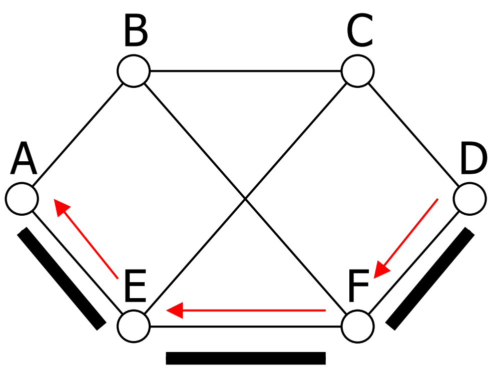
- Principle
  - Reduce traffic during congestion by telling source to slow down

- Procedure for source
  - Source receives the choke packet
    - Reduces the data traffic to the destination in question by x_1%
  - Source recognizes 2 phases
    (gate time so that the algorithm can take effect)
    - Ignore: source ignores further Choke packets until timeout
    - Listen: source listens if more Choke packets are arriving
      - yes: further reduction by x_2%; go to Ignore phase
      - no: increase the data traffic

Repair by Choke Packets

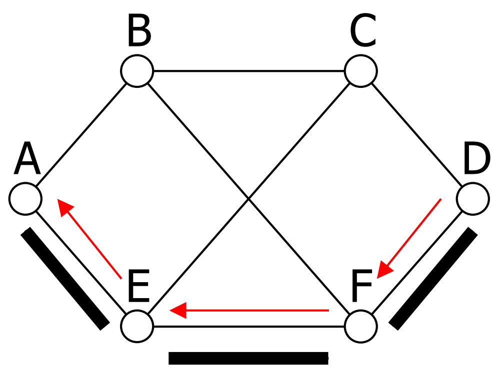
- Hop-by-Hop Choke Packets
- Principle
  - Reaction to Choke packets already at router (not only at end system)

Plain Choke packets



A heavy flow is established
Congestion is noticed at D
A Choke packet is sent to A
The flow is reduced at A
The flow is reduced at D

Hop-by-hop Choke packets



A heavy flow is established
Congestion is noticed at D
A Choke packet is sent to A
The flow is reduced at F
The flow is reduced at D

Repair by Choke Packets

- End-to-end variation
  - u > threshold: line changes to condition "warning"
    - Procedure for router
      - do not send choke packet to source (indicating destination)
      - tag packet (to avoid further choke packets from down stream router) & forward it
    - Procedure at receiver
      - send choke packet to sender
- Other variations
  - Varying choke packets depending on state of congestion
    - Warning
    - acute warning
  - For u instead of utilization
    - queue length
    - ....

Repair by Choke Packets

- Properties
  - Effective procedure
  - But:
    - possibly many choke packets in the network
      - even if Choke bits may be included in the data at the senders to minimize reflux
    - end systems can (but do not have to) adjust the traffic
    - choke packets take time to reach source
      - transient congestion may have passed when the source reacts
    - oscillations
      - several end systems reduce speed because of choke packets
      - seeing no more choke packets, all increase speed again

Transport Layer

Congestion Avoidance

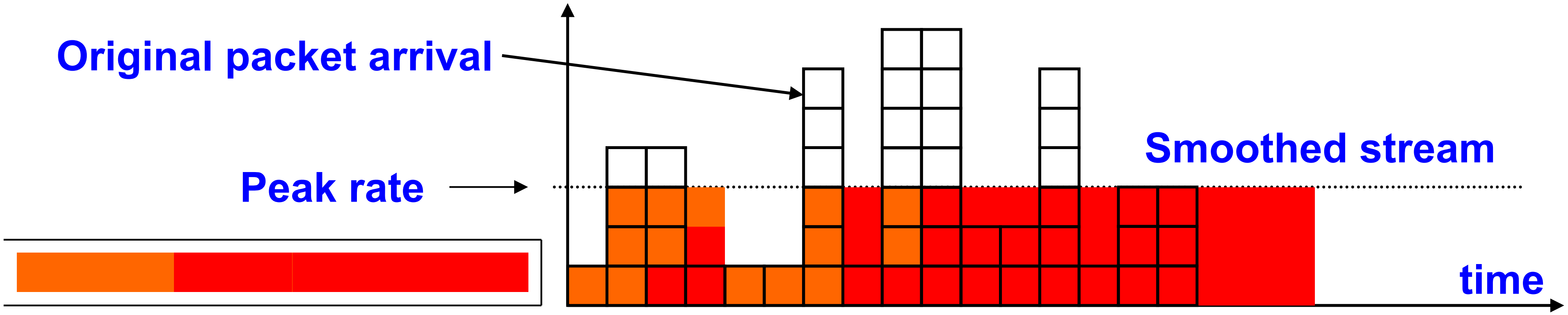
Avoidance

- Principle
  - Appropriate communication system behavior and design

- Policies at various layers can affect congestion
  - Data link layer
    - Flow control
    - Acknowledgements
    - Error treatment / retransmission / FEC
  - Network layer
    - Datagram (more complex) vs. virtual circuit (more procedures available)
    - Packet queueing and scheduling in router
    - Packet dropping in router (including packet lifetime)
    - Selected route
  - Transport layer
    - Basically the same as for the data link layer
    - But some issues are harder (determining timeout interval)

Avoidance by Traffic Shaping

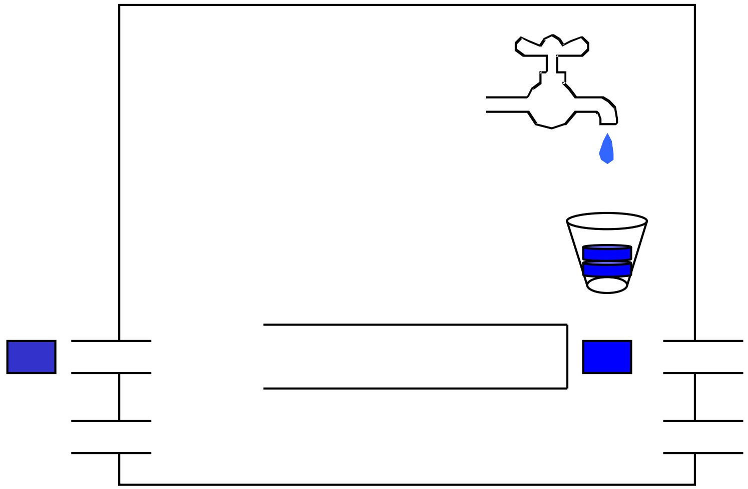
- Motivation
  - Congestion is often caused by bursts
  - Bursts are relieved by smoothing the traffic (at the price of a delay)



- Procedure
  - Negotiate the traffic contract beforehand (e.g., flow specification)
  - The traffic is shaped by sender
    - Average rate
    - Burstiness
  - Applied
    - In ATM
    - In the Internet ("DiffServ" – Differentiated Services)

Token Bucket

- Principle
  - Permit a certain amount of data to flow off for a certain amount of time
  - Controlled by "tokens‟
  - Number of tokens limited
  - Number of queued packets limited



Token Bucket

- Principle
  - Permit a certain amount of data to flow off for a certain amount of time
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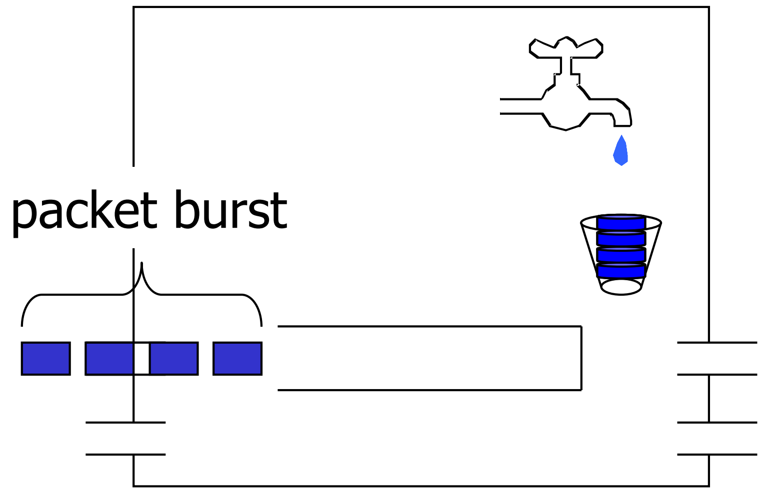
- Implementation
  - Add tokens periodically
    - Until maximum has been reached
  - Remove token
    - Depending on the length of the packet (byte counter)



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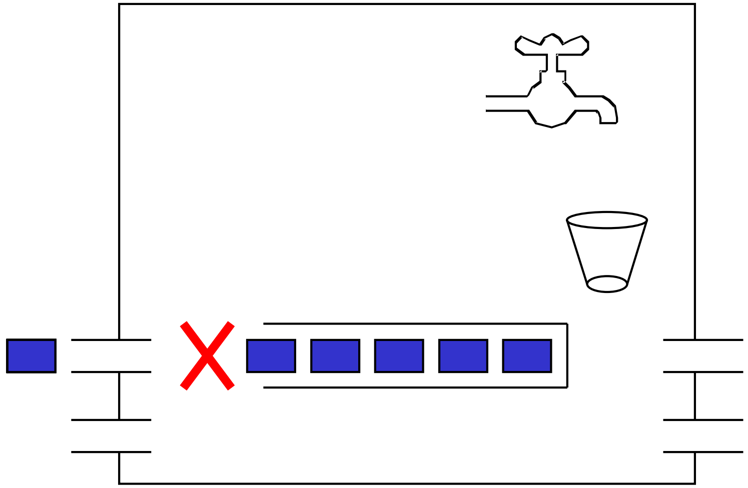
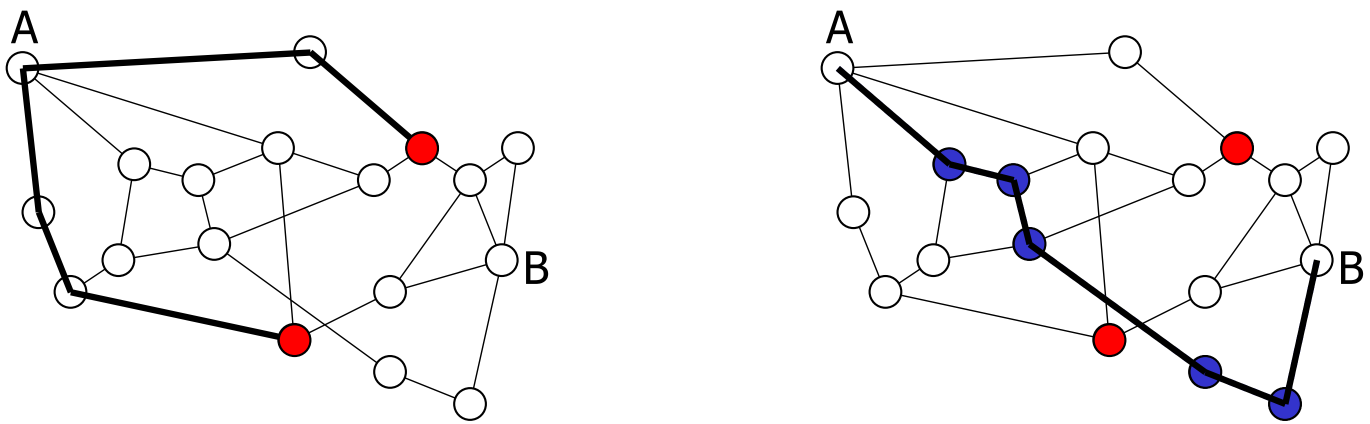


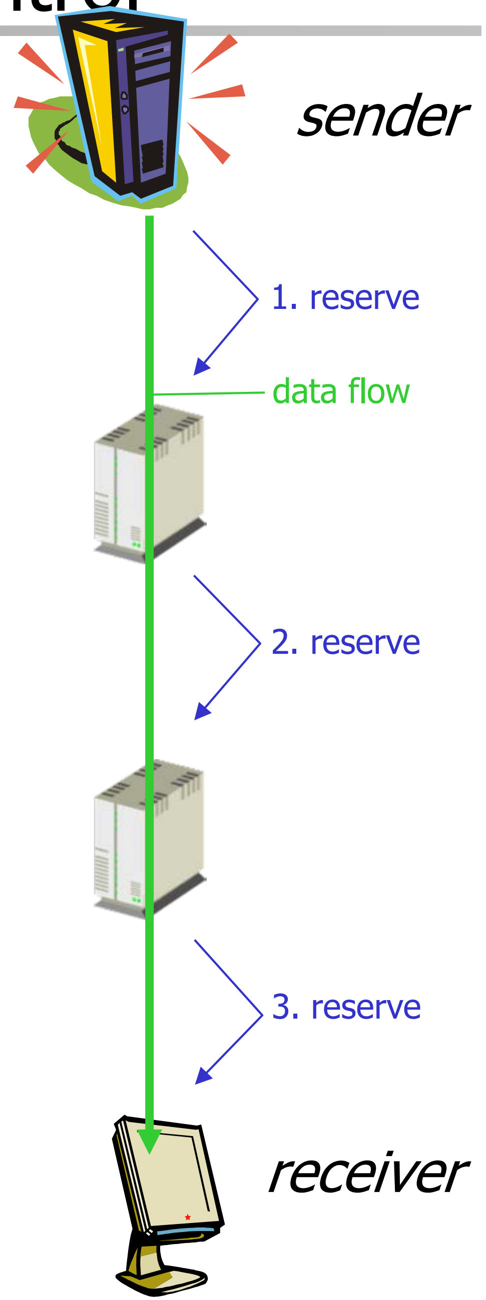
Avoidance by Reservation: Admission Control



- Principle
  - Prerequisite: virtual circuits
  - Reserving the necessary resources (incl. buffers) during connect
  - If buffer or other resources not available
    - Alternative path
    - Desired connection refused
- Example
  - Network layer may adjust routing based on congestion
  - When the actual connect occurs

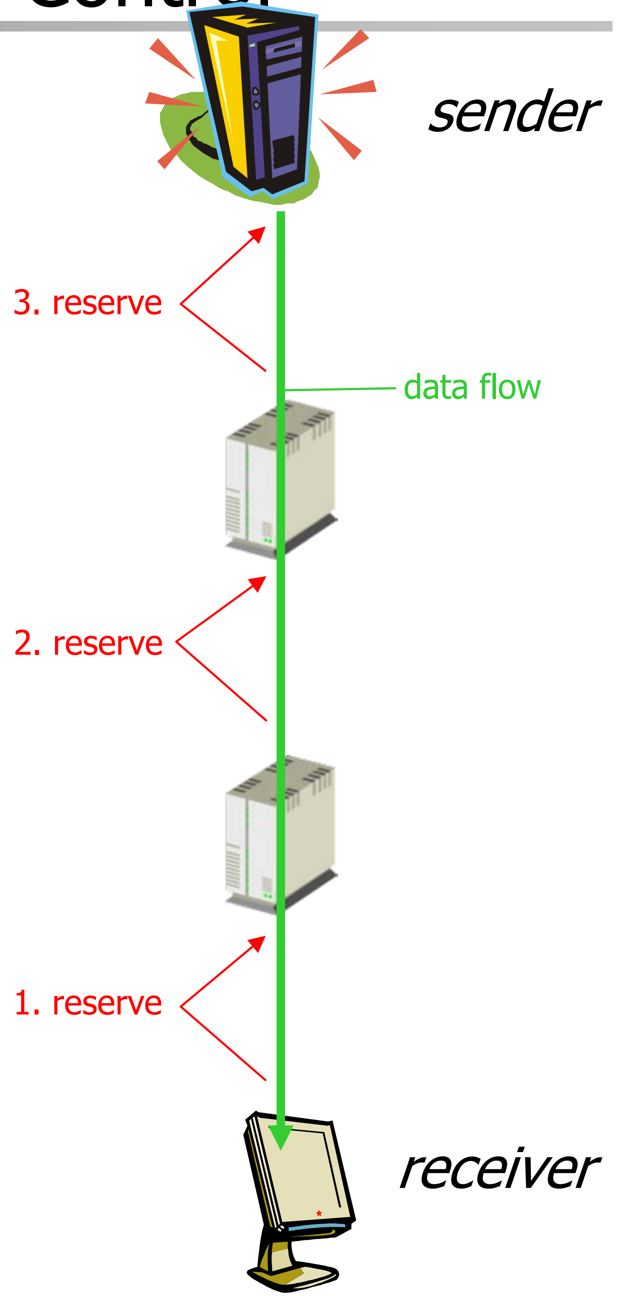
Avoidance by Reservation: Admission Control

- Sender oriented
  - Sender (initiates reservation)
    - Must know target addresses (participants)
    - Not scalable
    - Good security



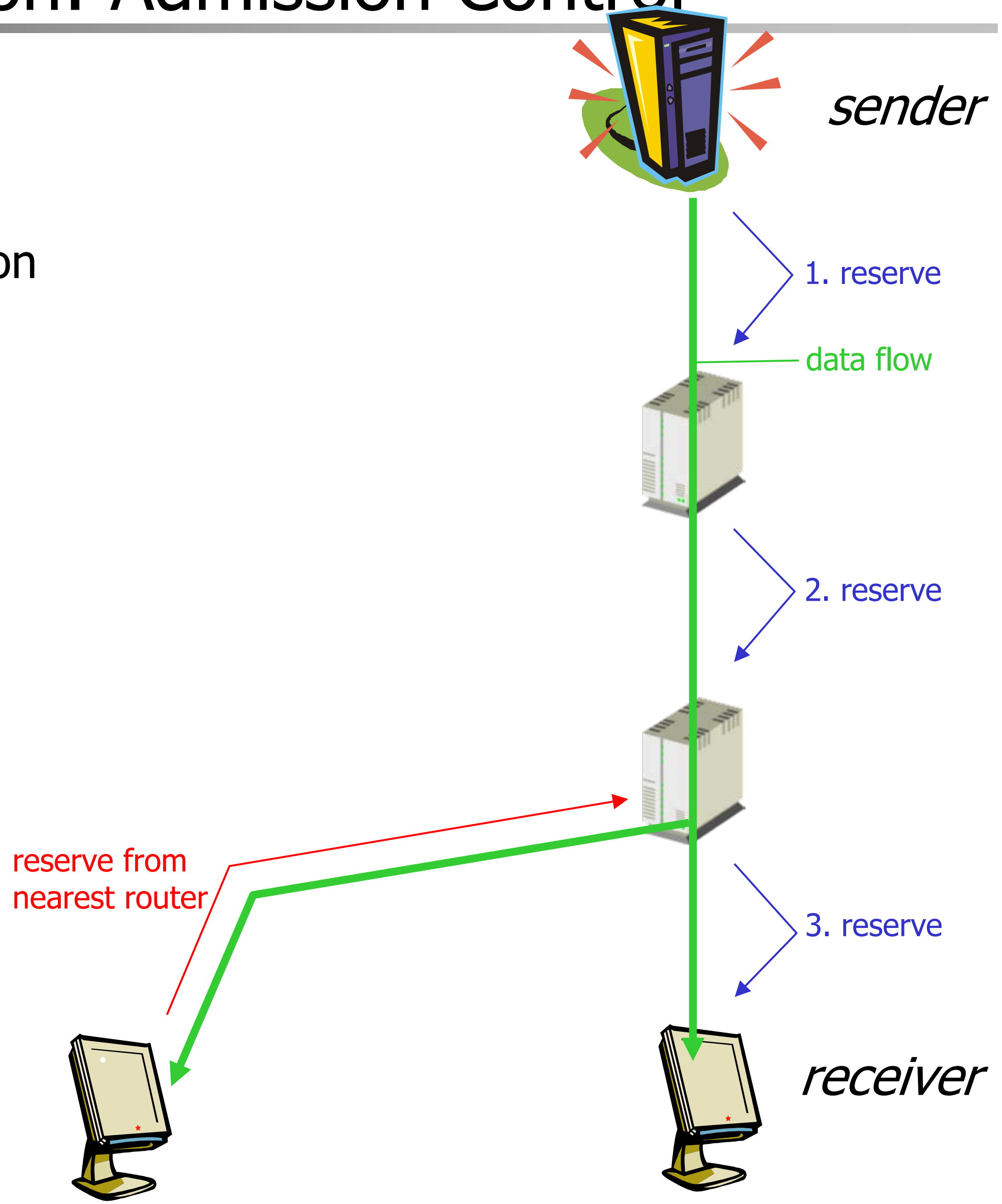
Avoidance by Reservation: Admission Control

- Receiver oriented
  - Receive (initiates reservation)
    - Needs advertisement before reservation
    - Must know "flow" addresses
  - Sender
    - Need not to know receivers
    - More scalable
    - Insecure



Avoidance by Reservation: Admission Control

- Combination?
  - Start sender oriented reservation



Avoidance: combined approaches

- **Controlled load**
  - Traffic in the controlled load class experiences the network as empty
  - Traffic class for the IntServ/RSVP reservation mechanism

- Approach
  - Allocate ***few*** buffers for this class on each router
  - Use admission control for these few buffers
    - Reservation is in packets/second (or Token Bucket specification)
    - Router knows its transmission speed
    - Router knows the number of packets it can store
  - Strictly prioritize traffic in a controlled load class

- Effect
  - Controlled load traffic is hardly ever dropped
  - Overtakes other traffic

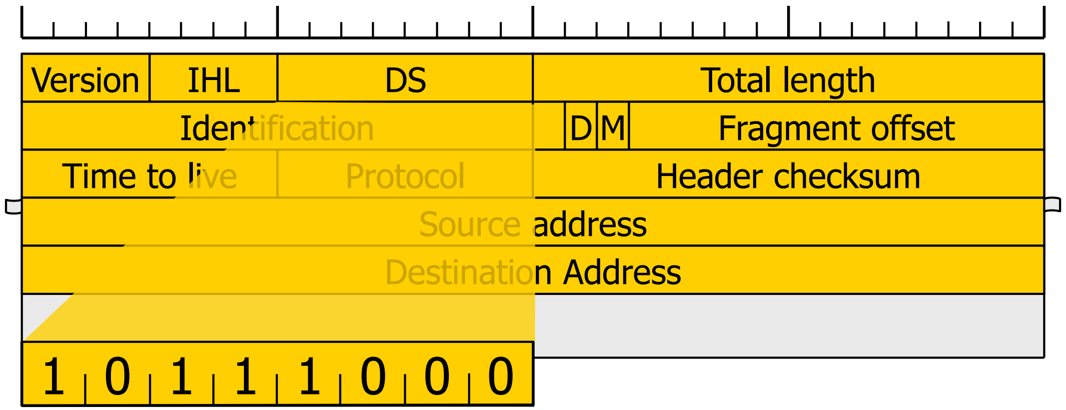
Avoidance: combined approaches

- **Expedited forwarding**
  - Very similar to controlled load
  - A differentiated services PHB (per-hop-behavior) for the DiffServ mechanisms

- Approach
  - Set aside ***few*** buffers for this class on each router
  - Police the traffic
    - Shape or mark the traffic
    - Only at senders, or at some routers
  - Strictly prioritize traffic in a controlled load class

Effect

Shapers drop excessive traffic
EF traffic is hardly ever dropped
Overtakes other traffic



Transport Layer

Congestion Avoidance: TCP

TCP Congestion Control

- TCP limit sending rate as a function of perceived network congestion
  - little traffic – increase sending rate
  - much traffic – reduce sending rate

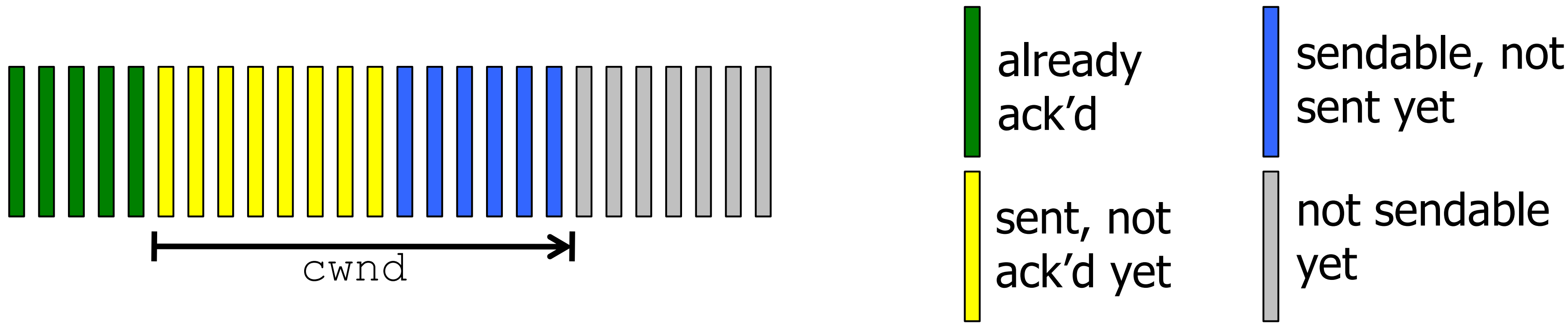
- Congestion algorithm has three major “components”:
  - additive-increase, multiplicative-decrease (AIMD)
  - slow-start
  - reaction to timeout events

TCP Congestion Control

- Testing for available bandwidth
  - Ideally: send as fast as possible (`cwnd` as large as possible) without loss
  - Increase congestion window until you have loss
  - If loss, reduce congestion window, try increasing again
- Two phases
  - Slow start
  - Congestion avoidance
- Important variables
  - `cwnd` (congestion window)
  - `ssthresh` – defines the threshold where the transition from slow start to congestion avoidance is made

TCP Congestion Control

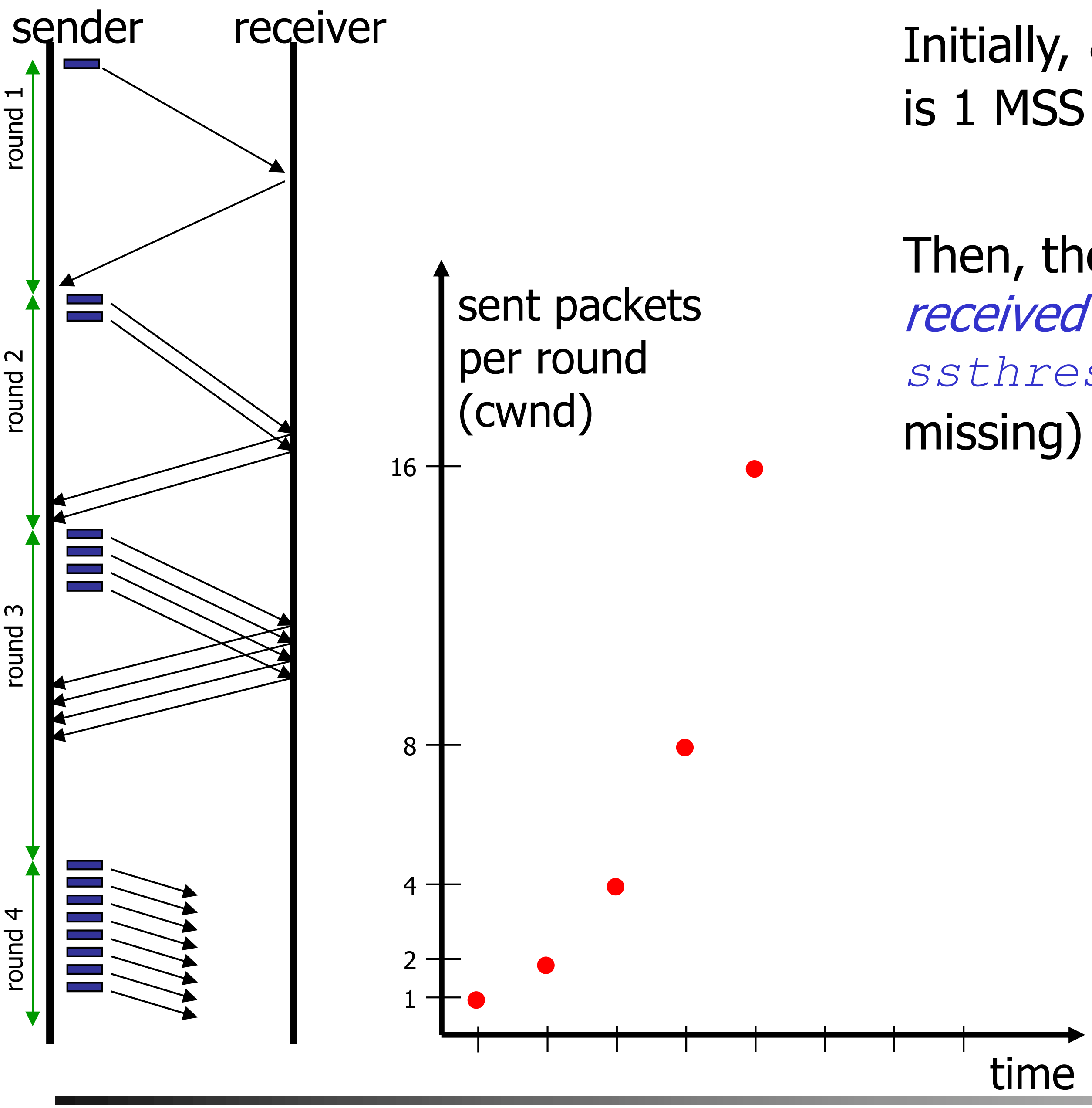
- End-to-end control (no support from the network layer)
- Send rate is limited by the size of a congestion window, `cwnd`, that is measured in bytes



- w segments, each of size MSS, can be sent in each RTT:

$$throughput = \frac{w * MSS}{RTT} bytes/\sec$$

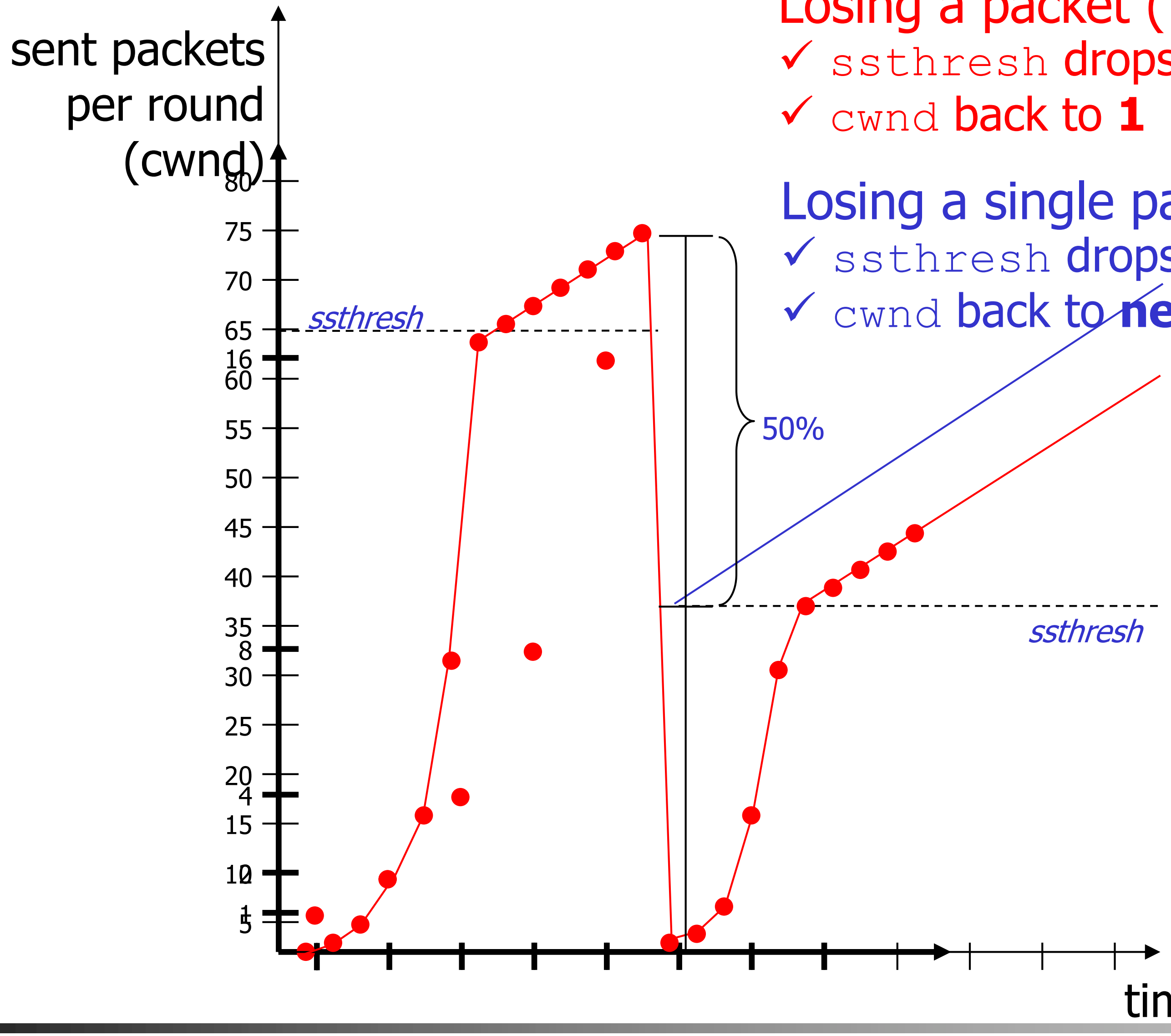
TCP Congestion Control



Initially, `cwnd` is 1 MSS (message segment size)

Then, the size *increases by 1 for each received ACK* (until threshold *`ssthresh`* is reached or an ACK is missing)

TCP Congestion Control



Normally, the threshold is 65 K

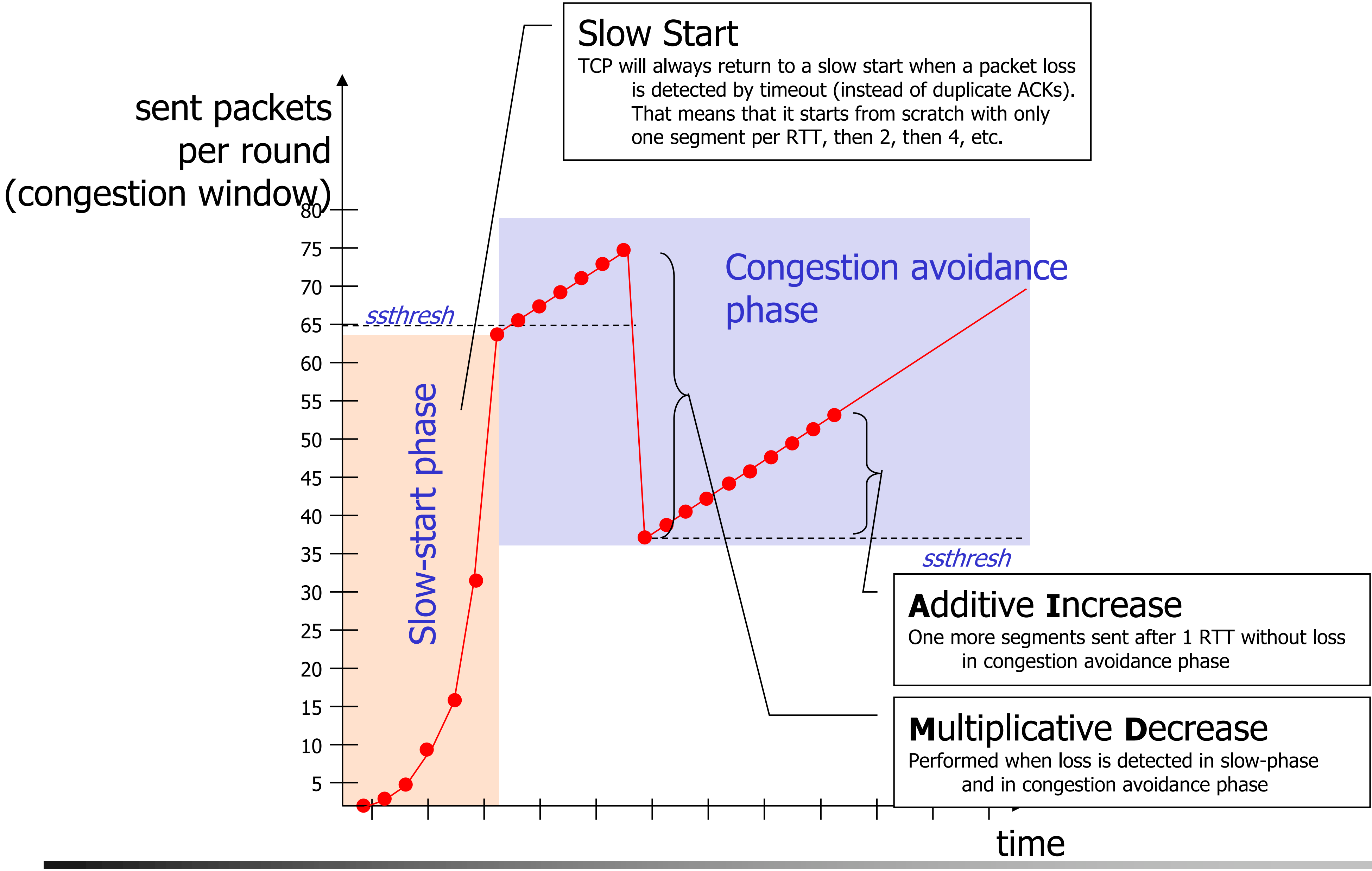
Losing a packet (TCP Tahoe):

- ✓ `ssthresh` drops to **half** `cwnd`
- ✓ `cwnd` back to **1**

Losing a single packet (TCP Reno):

- ✓ `ssthresh` drops to **halve** `cwnd`
- ✓ `cwnd` back to **new threshold**

TCP Congestion Control



TCP Fairness

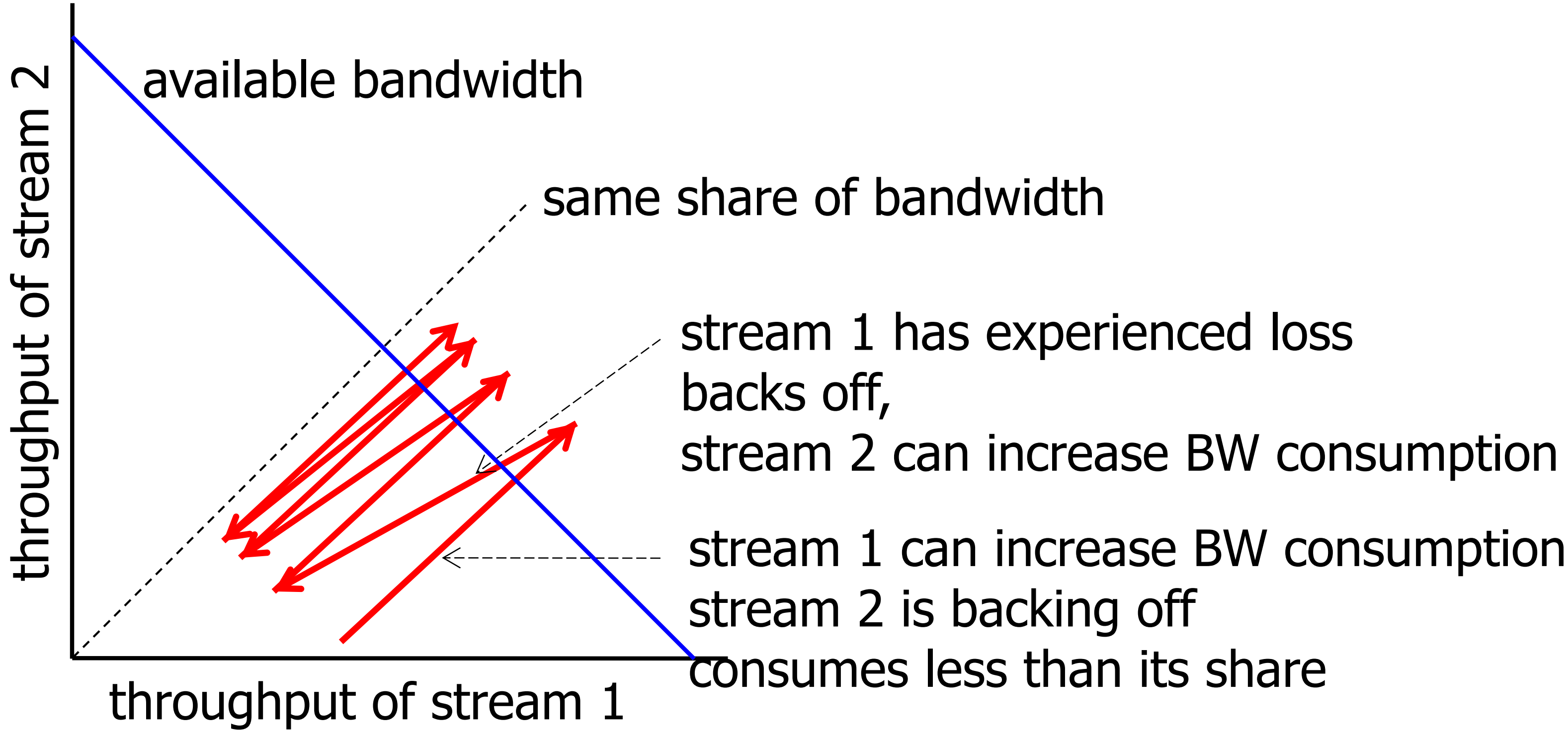
- Goal of fairness
  - When N TCP streams share a bottleneck, each TCP stream should receive an n^{th} of the bottleneck bandwidth

- more realistic demand
  - When N TCP streams with the same RTT and loss rate share a bottleneck, and they are infinitely long, each TCP stream receives an n^{th} of the bottleneck bandwidth

- but the approximation is in many cases good

TCP Fairness

- In which way is TCP fair?
  - Two competing streams
    - Additive increase adds 1 MSS per RTT until loss occurs
    - Multiplicative decrease reduces throughput proportionally



Transport Layer

Congestion Avoidance: RED and ECN

Random Early Detection (RED)

- Random Early Detection (Discard/Drop) (RED) uses active queue management

- Drops packet in an intermediate node based on average queue length exceeding a threshold
  - TCP receiver reports loss in ACK
  - sender applies MD

- Why?
  - if not, many TCPs loose packets at the same time
  - many TCP streams probe again at the same time
  - oscillating problems

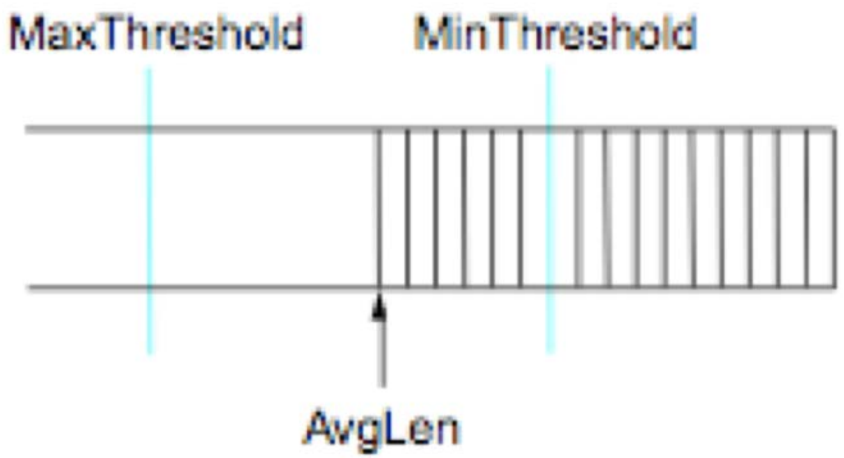
Random Early Detection (RED)

- Information flow is implicit
  - just drop one packet (TCP will retransmit it and back off)
  - can be made explicit by marking packets
- Random early discard of packets
  - instead of waiting until the queue is full, drop some packets considering the probability (drop probability) when the queue length exceeds a certain level (drop level)
- RED: details
  - compute the average queue length
  - AvgLen = (1-Weight) * AvgLen + Weight * SampleLen
  - 0 < Weight < 1 (usually 0.002)
  - SampleLen is the queue length every time a packet arrives

Random Early Detection (RED)

- Two tresholds for queue length

```
if AvgLen <= MinTreshold then
   enqueue the packet
if MinTreshold < AvgLen < MaxThreshold
   compute probability P
   drop packet with probability P
if MaxThreshold <= AvgLen
   drop packet
```



Random Early Detection (RED)

- Probability P
  - Floating
  - Function of `AvgLen` and how long since the previous drop
  - variable count keeps the number of new packets that have been added to the queue (not dropped) while `AvgLen` was between the two thresholds

$$TempP = MaxP * \frac{AvgLen - MinThreshold}{MaxThreshold - MinThreshold}$$

$$P = 1 - \frac{TempP}{count * TempP}$$

Random Early Detection (RED)

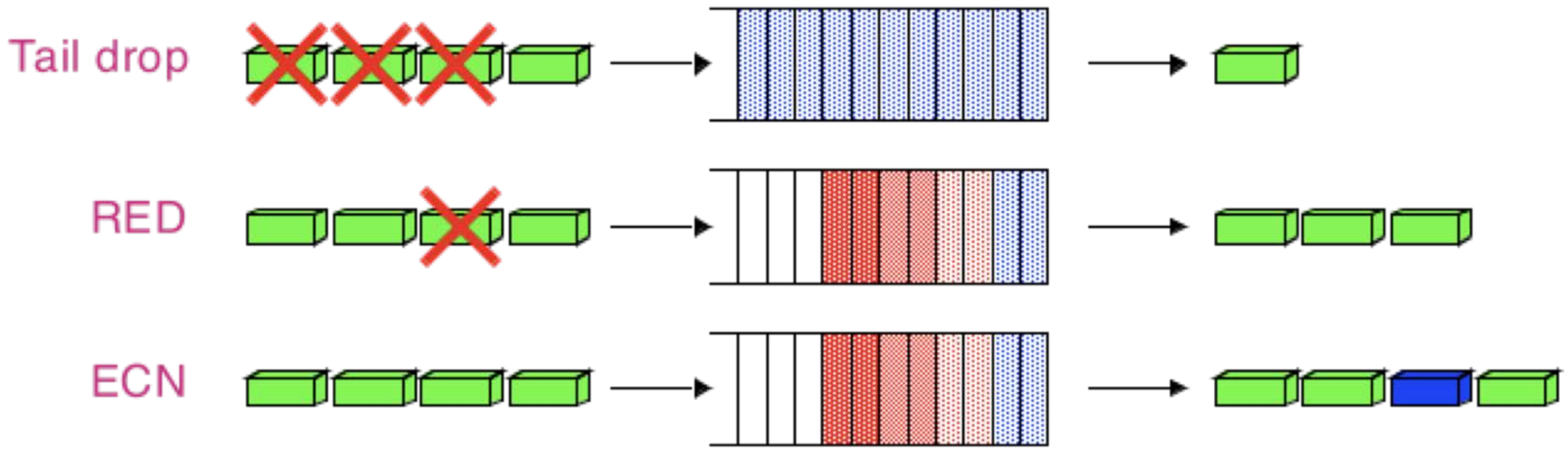
- Notes
  - probability for dropping a packet of a given flow approx. proportional to the size of the flow
  - `MaxP` will usually by 0.02. That means that when the average queue length is in the middle of the thresholds, roughly 1 in 50 packets will be dropped
  - when the traffic is "bursty", `MinThreshold` should be high enough to allow an acceptable utilization of the links
  - the difference between the thresholds should be bigger than the typical increase in the computed queue length in one RTT
    for today's Internet traffic it is reasonable to set
    `MaxThreshold = 2 * MinThreshold`

Early Congestion Notification (ECN)

- Early Congestion Notification (ECN) - RFC 2481
  - an end-to-end congestion avoidance mechanism
  - implemented in routers and supported by end-systems
  - not multimedia-specific, but very TCP-specific
  - two IP header bits used
    - ECT - ECN Capable Transport, set by sender
    - CE - Congestion Experienced, may be set by router

- Extends RED
  - if packet has ECT bit set
    - ECN node sets CE bit
    - TCP receiver sets ECN bit in ACK
    - sender applies multiple decrease (AIMD)
  - else
    - Act like RED

Early Congestion Notification (ECN)



- Effects
  - Congestion is not oscillating - RED & ECN
  - ECN-packets are never lost on uncongested links
  - Receiving an ECN mark means
    - TCP window decrease
    - No packet loss
    - No retransmission

Transport Layer

Congestion Avoidance: TCP Vegas

TCP Vegas

- Idea
  - Source is looking for signs that a router is close to congestion
  - By checking whether
    - The RTT is growing
      implying that other traffic leads to a growth of the router queue
    - The distance between the received ACKs grows
      implying that the flow itself is too fast

TCP Vegas

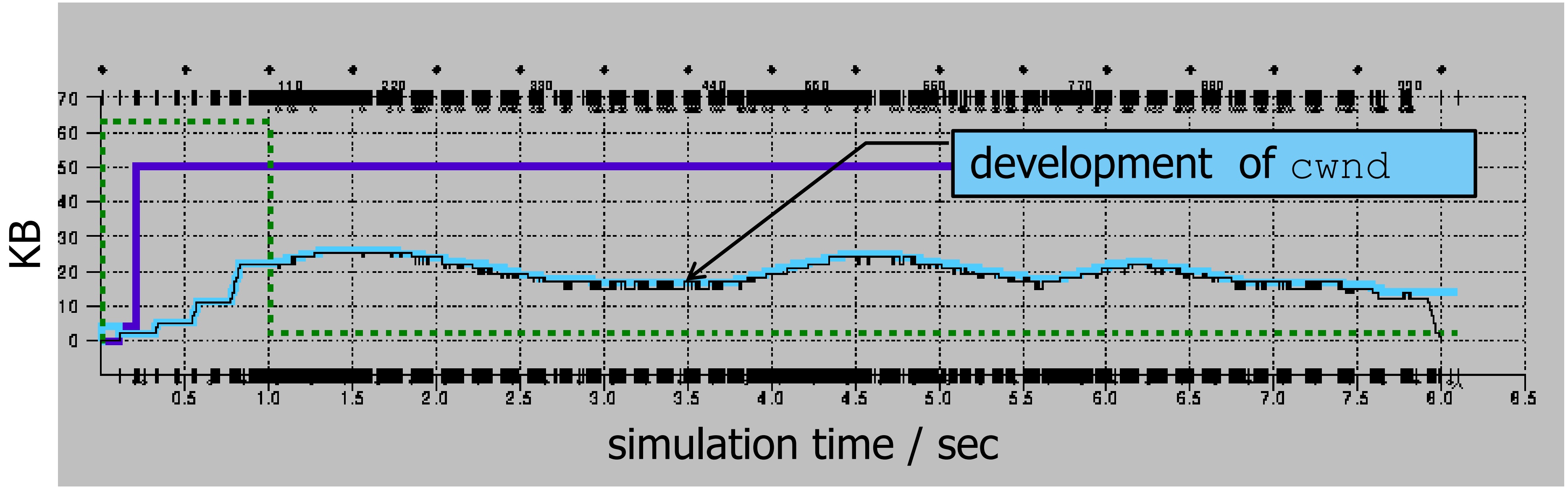
- Algorithm
  - Let `BaseRTT` be the minimum of all measured RTTs (usually the RTT of the first packet)
  - If we don't congest the connection, we can assume

$$ExpectedRate = \frac{cwnd}{BaseRTT}$$

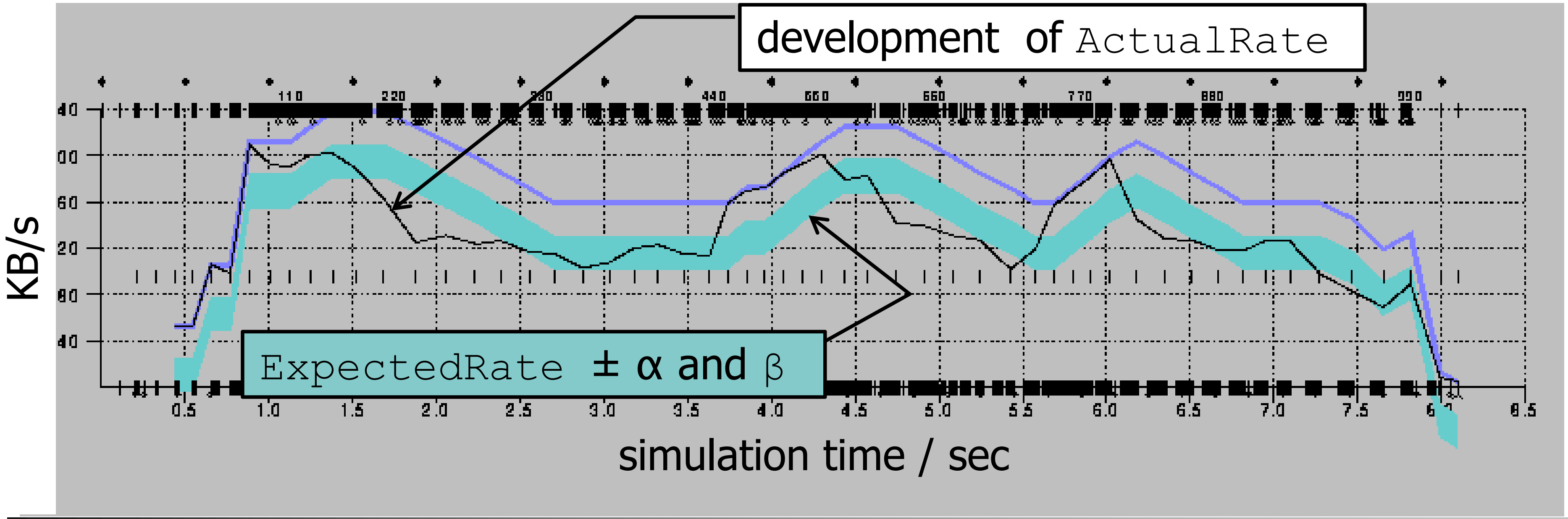
  - The source computes the actual sending rate (`ActualRate`) once per RTT
  - The source compares `ActualRate` to `ExpectedRate`

```
Diff = ExpectedRate - ActualRate
if Diff < α
    increase cwnd linearly
if diff > β
    decrease cwnd linearly
Else
    leave cwnd unchanged
```

TCP Vegas



TCP Vegas simulation results: from http://www.cs.arizona.edu/projects/protocols/



Summary

Congestion control

- reason for congestion
- congestion repair
- congestion avoidance
- TCP congestion control
- Variants of TCP congestion control