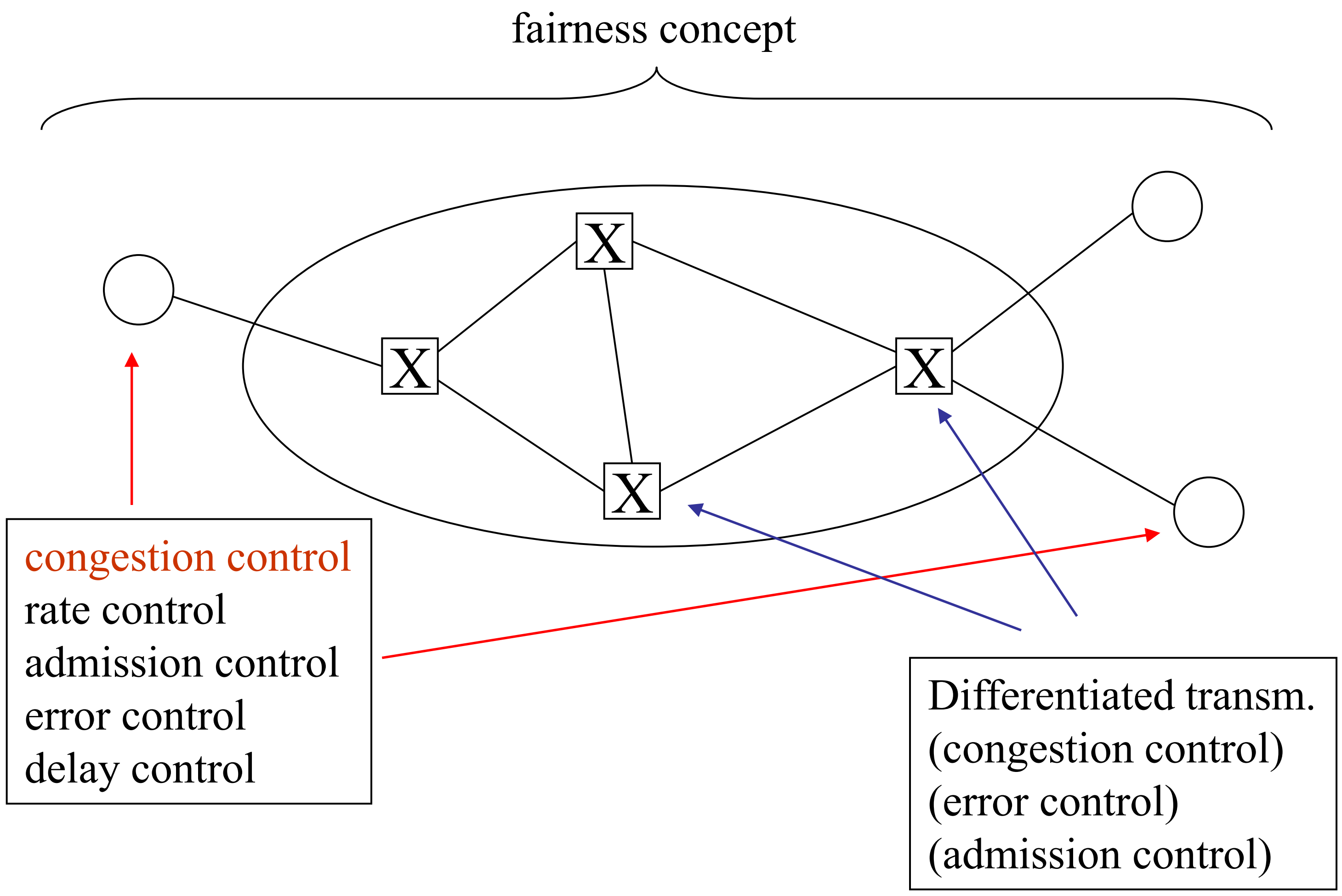
EP2200
Performance analysis of Communication networks

Topic 3
Congestion and rate control

Congestion, rate and error control

- Lecture material:
  - Bertsekas, Gallager, *Data networks*, 6.1-2
  - I. Kay, *Stochastic modeling*, 3.3.4
  - J-Y Le Boudec, “Rate adaptation, congestion control and fairness: a tutorial,” Nov. 2005

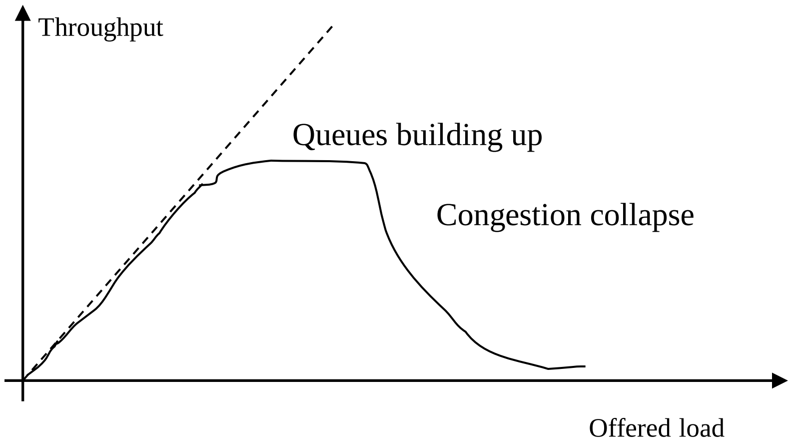
Control functions in communication networks



Congestion control

- To regulate the packet population in the network
- To share resources (link bandwidth, buffer space)
- *Flow control: between two users for speed matching,* sometimes means congestion control in the literature
- The main objectives of congestion control
  - Efficiency
    - high utilization (from the network provider's perspective)
    - high per flow throughput, low delay (from users' perspective)
  - Fairness: fair allocation of resources

Congestion control



- What happens if the incoming traffic is not restricted?
  - Bottleneck links: the offered traffic is higher than the link transmission capacity: temporarily (bursts arriving) or permanently
- What happens at bottleneck links?
  - queue sizes grow, end-to-end delays increase
  - queue space fills up, packets get dropped
  - packets are retransmitted by the applications, further increasing the load
- Congestion collapse: the network throughput decreases and delays become excessive

Congestion control techniques

- Should depend on the service requirements of the application
  - voice, video: minimum bandwidth requirement, delay sensitive
  - data: lossless end-to-end transmission required

- Call blocking at the network edge + rate control
  - calls blocked if resources are not available
  - the rate of accepted calls is controlled
- Packet discarding at a network node
  - at buffer overflow or earlier
  - discarding policy - fairness, service differentiation
- Packet blocking at the network edge
  - packet waits in a queue outside the network

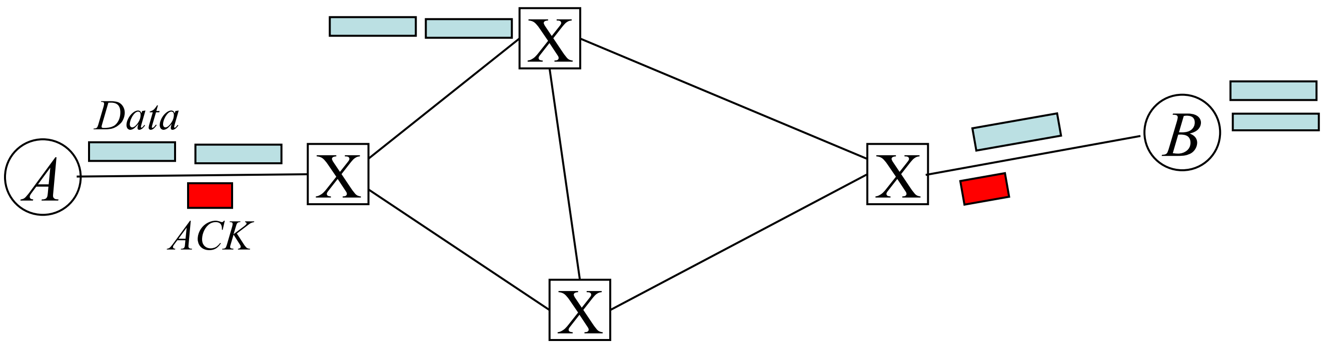
Group work:

- select solutions for streaming and for elastic flows
- find examples for the techniques above.

Congestion control techniques

- Should depend on the service requirements of the application
  - voice, video: minimum bandwidth requirement, delay sensitive
  - data: requires strict error control
- Call blocking at the network edge + rate control
  - calls blocked if resources are not available
  - the rate of accepted calls is controlled
- Packet discarding at a network node
  - buffer overflow or active queue management
  - discarding policy – fairness, service differentiation
- Packet blocking at the network edge
  - packet waits in a queue outside the network

End-to-end window congestion control



- Basic idea:
- Window: Upper bound on the number of packets transmitted by A and not acknowledged by B.
- Input rate reduced if acknowledgements arrive slowly – achieves congestion control

Fixed window based congestion control

- Congestion is indicated by increased RTT
- Simple case:
  - ACK after each packet reception
  - Window size W is
    * decremented after each transmission
    * incremented if ACK arrives
  - Transmission, if W > 0
  - No lost Data packet or ACK
  - Parameters
    - constant maximum window size (W_m)
    - constant packet transmission time (X: packet size/link rate [sec])
    - round-trip delay (rtt: propagation, **transmission** and queuing delays)
  - To calculate:
    maximum packet transmission rate (r: [packet/sec])

$$r = \min\left(\frac{1}{X}, \frac{W_m}{rtt}\right)$$

- Rate inversely proportional to round-trip delay
- Congestion control: large queuing delays – large rtt – lower rate
  - Reacts to congestion in W packets transmission time – large window means slow reaction to congestion
  - What rtt value means congestion? When should the congestion control “be activated” ($rtt=W_mX$)?

Window control - packet delay

- Packet delay T ?
  - Delay: propagation, transmission and waiting at the queues in the multihop path
  - Queuing analysis in "steady state", for given network load
  - Model the network as a "black box", apply Little theorem for the entire network $(N = \lambda T)$
  - Assumptions:
    - arrival intensity – constant (steady state) – arrival process is arbitrary
    - Infinite buffers – no loss

$$N = \sum_{i=1}^{n} const_i W_i$$

$$T = \frac{N}{\lambda} = \frac{\sum_{i=1}^{n} const_i W_i}{\lambda}$$

n: active flows

W_i : max window size for flow *i*

N: number of packets in the network

λ: aggregate throughput of all flows (no loss – infinite buffers!)

- For constant aggregate throughput (λ)
- Delay is proportional to the number of active sessions and to the window sizes

Window control

- How to select the window size?
  - Small window: OK packet delay, but low throughput (congestion control starts with low rtt)
  - Large window: OK throughput, but high packet delay and long reaction time

$$r = \min\left\{\frac{1}{X}, \frac{W}{rtt}\right\}$$

$$T = \frac{\sum_{i=1}^{n} const_i W_i}{\lambda}$$

- Dynamic window sizes are necessary to follow the network load
  - small window if the network is congested
  - large window if the network is low loaded
  - congestion is controlled by the window size – not by the *rtt*
  - *But what indicates the congestion: packet drop, increased rtt, …*

Congestion control-dynamic window size

- How to select the window size?
  - Small window: OK packet delay, but low throughput
  - Large window: OK throughput, but high packet delay
- Dynamic window sizes are necessary to follow the network load
  - Congestion is controlled by the window size – not by the *rtt*
- TCP
  - Performance with dynamic window sizes - comes now
  - More realistic models for TCP performance – reading assignment

TCP congestion control

- Window based congestion control
- Dynamic window size:
  - decrease window size if congestion is detected (e.g., by packet loss)
  - increase window size if current rate does not cause congestion (e.g., no loss)
- How to increase and decrease the window sizes?
  - Additive-increase, multiplicative decrease (AIMD)
  - Efficient and fair (if all users get the same immediate indication of congestion): Chiu and Jain, 1989
  - Probing - increase phase in each rtt: $w_{i+1} = w_i + b, 0 < b << w_{max}$
  - Congestion - decrease phase: $w_{i+1} = aw_i, 0 < a < 1$

- TCP: a=0.5, b=1

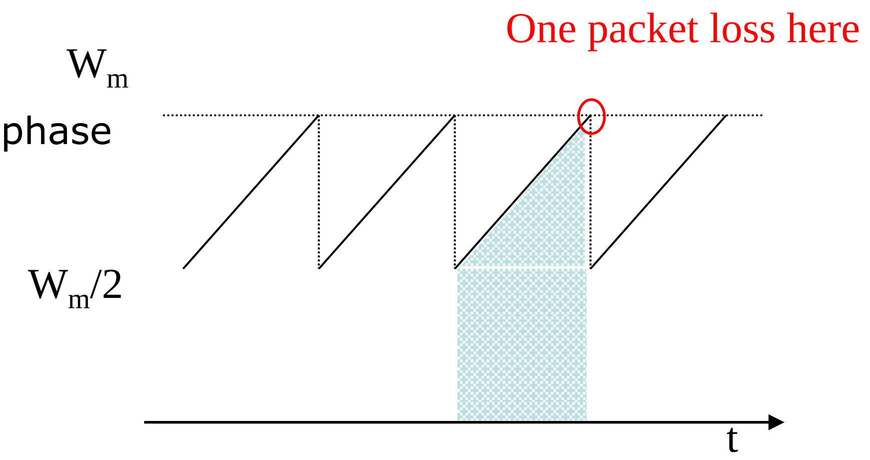
- TCP additional phases: slow start, fast recovery – not considered now

Analysis – AIDM model

- Question: How does the throughput depend on the loss rate?

- Assumptions – for a very simplified case
  - saturated source (always has packet to send)
  - constant background traffic (all other traffic), loss at the same window size
  - loss due to congestion only
  - loss is the only congestion indicator
  - transients negligible (long flows)

  ⇒ we model a static congestion avoidance phase
  - constant round trip time (rtt)
  - rtt >> transmission time
  - constant packet size L
  - low loss probability
    (to simplify calculations)



- W increased to W+1 after rtt if there was no loss
- W decreased to W/2 after rtt if there was a loss

TCP throughput vs. packet loss

- *Th* as a function of *p* and *rtt*

$$Th = \frac{(N-1)L}{T_0}$$

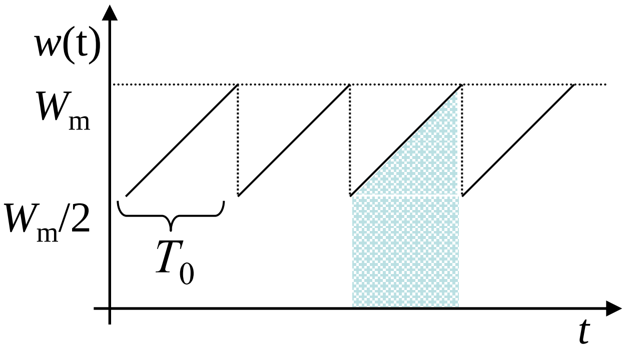
$$p = \frac{1}{N}, \quad N = \frac{1}{p}$$

$$T_0 = \frac{W_m}{2} rtt$$

$$N = \left(\frac{W_m}{2}\right)^2 + \frac{1}{2}\left(\frac{W_m}{2}\right)^2 = \frac{3}{8} W_m^2$$

$$W_m = \sqrt{\frac{8}{3}}\sqrt{N} = 2\sqrt{\frac{2}{3}}\frac{1}{\sqrt{p}}$$

$$Th = \frac{(N-1)L}{T_0} = \frac{\left(\frac{1}{p}-1\right)L}{\sqrt{\frac{2}{3}}\frac{1}{\sqrt{p}} rtt} = \sqrt{\frac{3}{2}}\frac{L}{rtt}\left(\frac{1-p}{p}\right)\sqrt{p} \approx \sqrt{\frac{3}{2}}\frac{L}{rtt}\frac{1}{\sqrt{p}}$$



rtt: round trip time (s)
p: packet loss probability
L: packet length (bit)

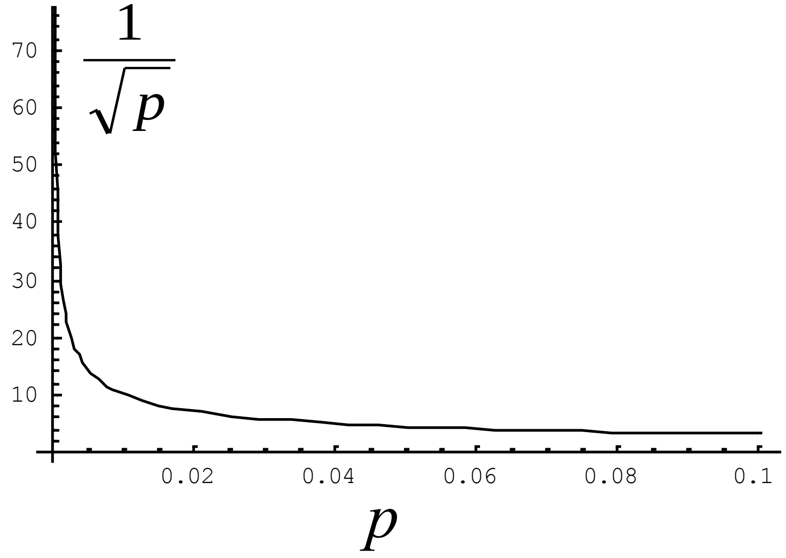
Th: throughput (bit/s)

W_m: max window (packets)
T_0: cycle (s)
N: packets transmitted in one cycle

TCP throughput vs. packet loss

- Statement: throughput in steady state
  - varies as the inverse-square-root of the loss rate, and
  - is inversely proportional to the round trip time.

$$Th \approx \sqrt{\frac{3}{2}}\frac{L}{rtt}\frac{1}{\sqrt{p}}$$



Summary - Reading for next lecture

- Today:
  - Congestion control – definition and possible solutions
  - End-to-end congestion control based on packet blocking at the network edge
  - Fixed window size schemes – throughput, delay conflict
  - Dynamic window congestion control
  - Simple AIMD scheme and throughput analysis – first steps towards evaluating TCP

- Home reading: TCP throughput – random loss due to congestion
- J. Padhye, F. Firoiu, D. Towsley, J. Kurose, "Modeling TCP throughput: a simple model and its empirical validation," Sigcomm, 1998
  - Read sections 1, 2-2.1, 3, 4, 5
  - After reading the text you should be able to answer questions like:
    - What is the difference between the modeling assumptions presented on the lecture and in this paper?
    - What is captured in the paper that is not captured in the simple model from the lecture?
    - What are the factors that affect the throughput of a TCP flow?
    - What is missed in the main paper that is corrected in the “Comments”? Are the main results affected?

Project topics related to TCP

- Research areas where some of the assumptions are released or discussed:
  - TCP performance in high bandwidth-delay product networks
  - loss does not mean congestion: TCP over wireless links
  - loss distribution among sessions: active queue management for TCP

Recall and outline for today

- Last lecture:
  - Congestion control – definition and possible solutions
  - End-to-end congestion control based on packet blocking at the network edge
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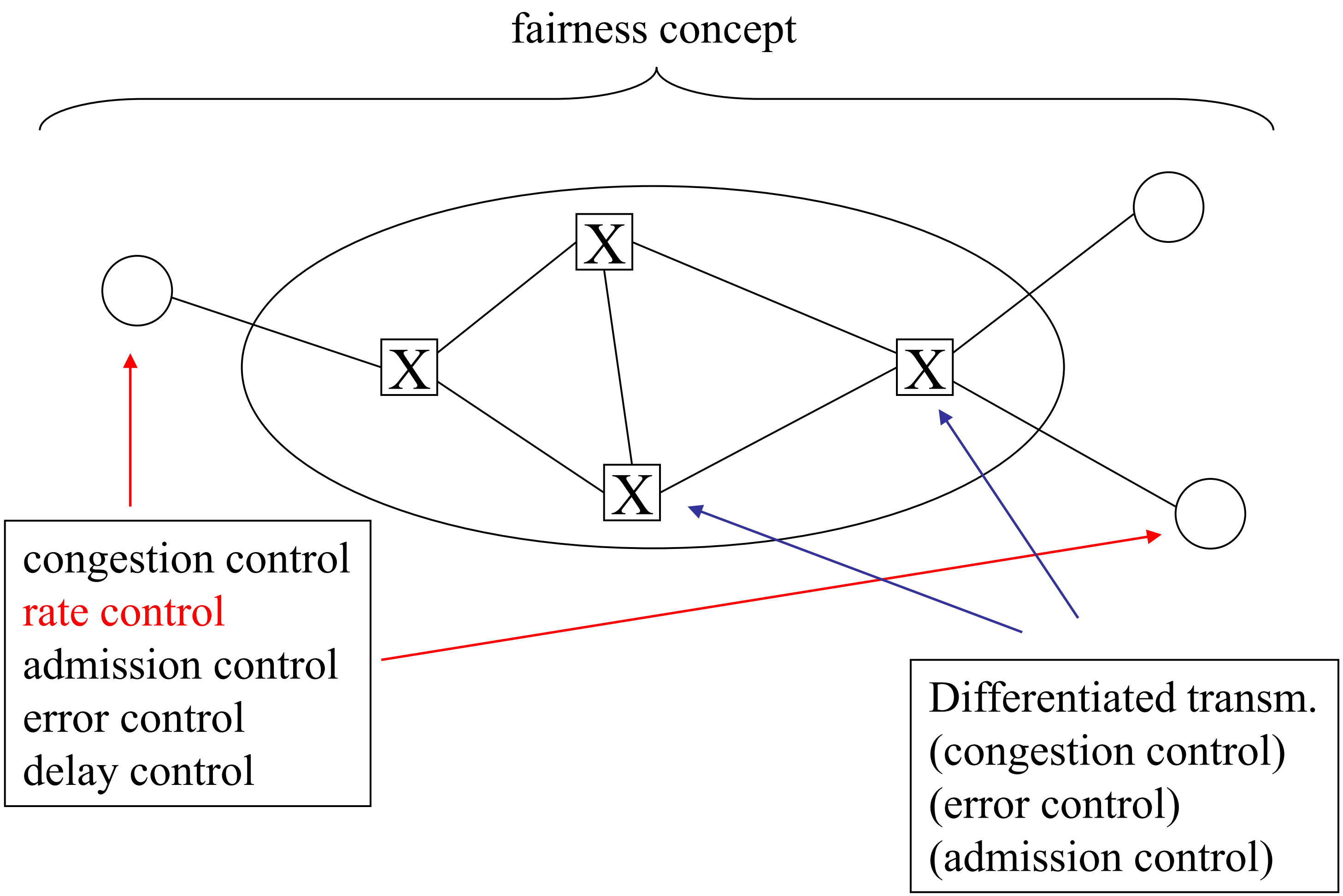


Rate control and guaranteed service

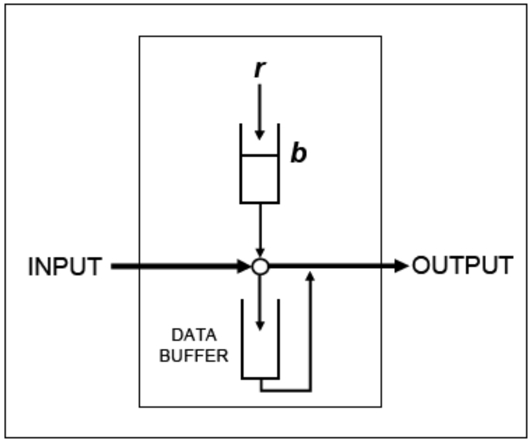
- Contract between the user and the network
  - If the user satisfies a traffic constraint,
  - Then the network provides delay, delay jitter (and loss) limits
- Rate control during the connection time
  - peak rate
  - average rate
  - burstiness (e.g., max. number of packets transmitted without time gap)

- Window based control (r,T)
  - n maximum number of packets per T: average rate (r=n/T)
  - jumping or sliding window versions
  - Problems
    - small average rate requires large T – slow reaction
    - no burstiness control within T – max burst size (packets transmitted back to back) is coupled to T and n: (jumping: b=2n=2rT, sliding: b=n=rT)

- Traffic envelope
  - max transmitted data within any time interval is given by a function b(t)
  - how to implement it?

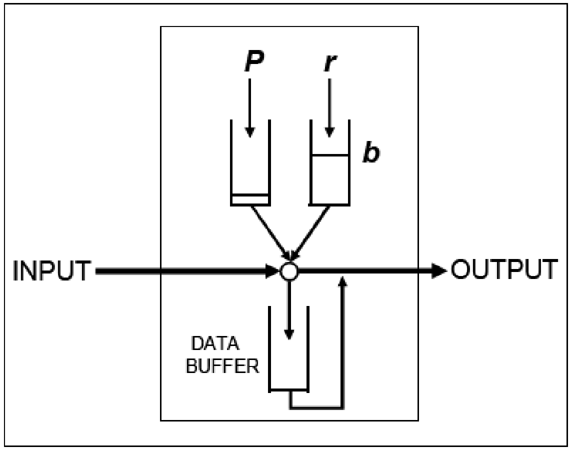
Rate control – Leaky-bucket

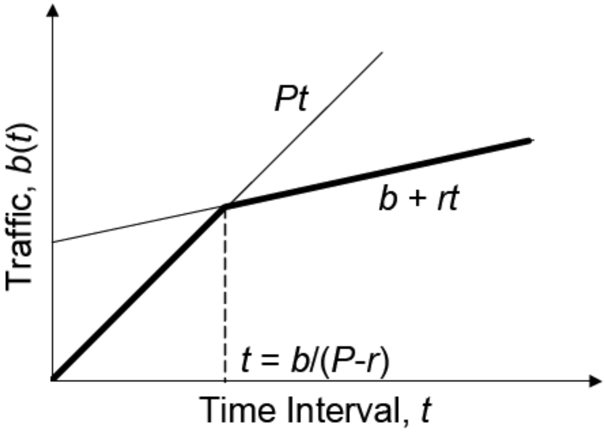
- Leaky-bucket scheme
  - Version 1: average rate and max burst size
  - token buffer (b)
  - token generation rate (r)
  - packets from "input" enter data buffer, waiting for service
  - packet transmitted to "output" if there is a token in the token buffer
  - token removed at packet transmission
  - Result:
    - traffic envelope: $b(t)=b+rt$
    - average rate: $(b+rt)/t$
  - Note: from "output" the packets enter a transmission buffer (MAC)
  - Note: t is not the time from the system start, but *any* interval t during the transmission



Rate control – Leaky-bucket

- Leaky-bucket scheme
  - Version 2: peak rate, average rate and max burst size
  - average rate buffer (b)
  - average token generation rate (r)
  - peak rate buffer (1)
  - peak token generation rate (P>r)
  - packets from “input” enter data buffer, waiting for service
  - packet transmitted to “output” if there is at least one token in both token buffers
  - tokens removed at packet transmission
  - Result:
    - traffic envelope b(t)





Summary

- Congestion control
  - window based congestion control
  - AIMD schemes and TCP

- Rate control
  - to control the average rate, burstiness or peak rate of traffic injected to the network
  - with the goal of providing service guarantees
  - Solutions
    - window based – problem with rate and burtiness coupling
    - traffic envelope based – leaky bucket