La qualité de service

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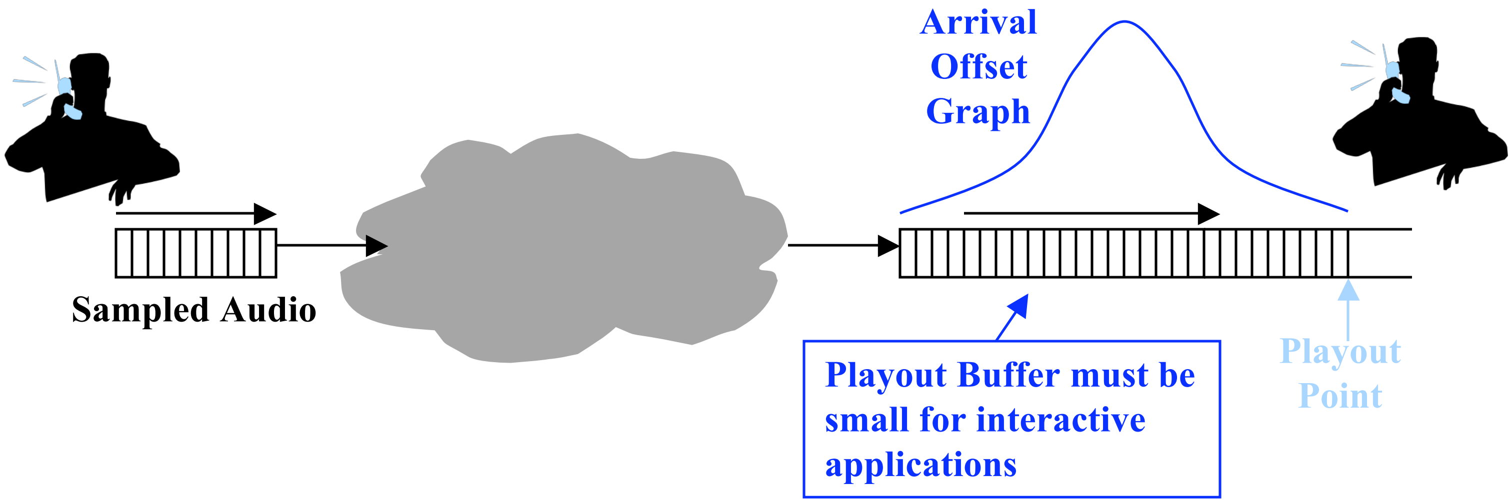
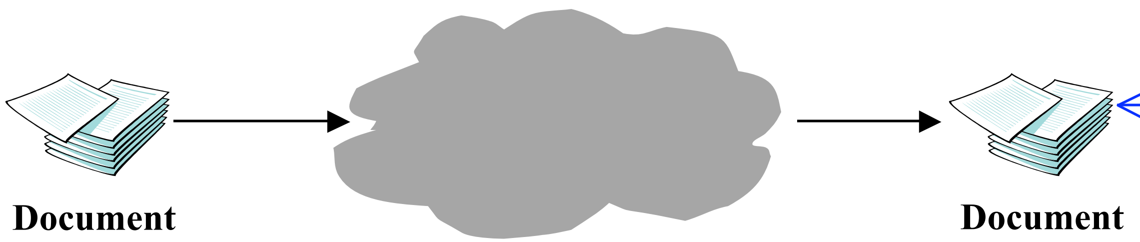
Multimedia, real time on the Internet



- **Real-time applications**
  - Interactive applications are sensitive to packet delays (telephone)
  - Non-interactive applications can adapt to a wider range of packet delays (audio, video broadcasts)
  - Guarantee of maximum delay is useful

Time-constrained applications



Document is only useful when it is completely received. This means *average* packet delay is important, not *maximum* packet delay.

- **Elastic applications**
  - Interactive data transfer (e.g. HTTP, FTP)
    - Sensitive to the average delay, not to the distribution tail
  - Bulk data transfer (e.g. mail and news delivery)
    - Delay insensitive
  - Best effort works well

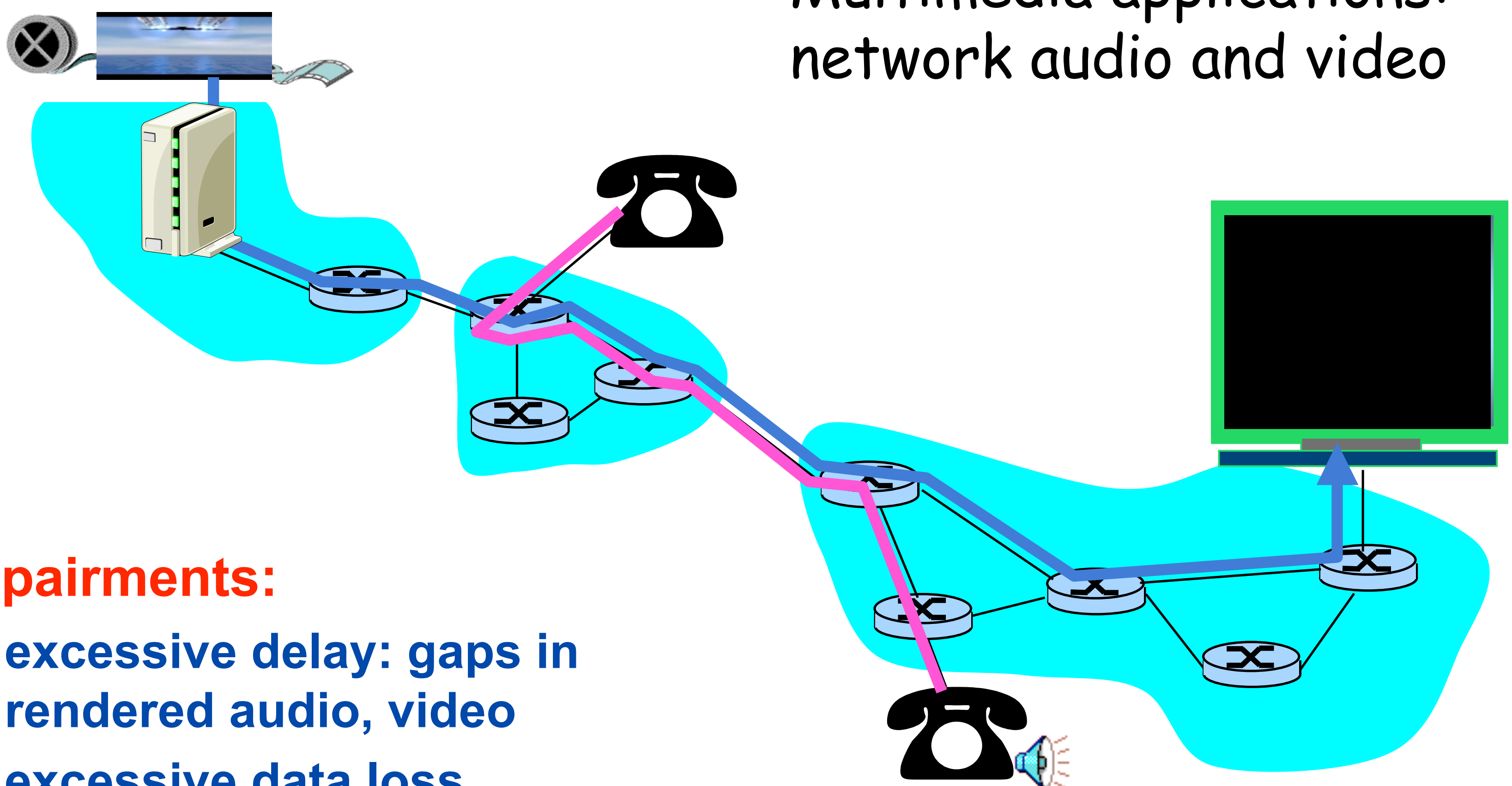
Discussion

- **What is the problem?**
  - Different applications have different delay, bandwidth, and jitter needs
  - Some applications are very sensitive to changing network conditions: the packet arrival time distribution is important
- **Solutions**
  - Make applications adaptive
  - Build more flexibility into network

Why Better-than-Best-Effort (QoS)?

- **To support a wider range of applications**
  - Real-time, Multimedia, etc
- **To develop sustainable economic models and new private networking services**
  - Current flat priced models, and best-effort services do not cut it for businesses

What do we have now?



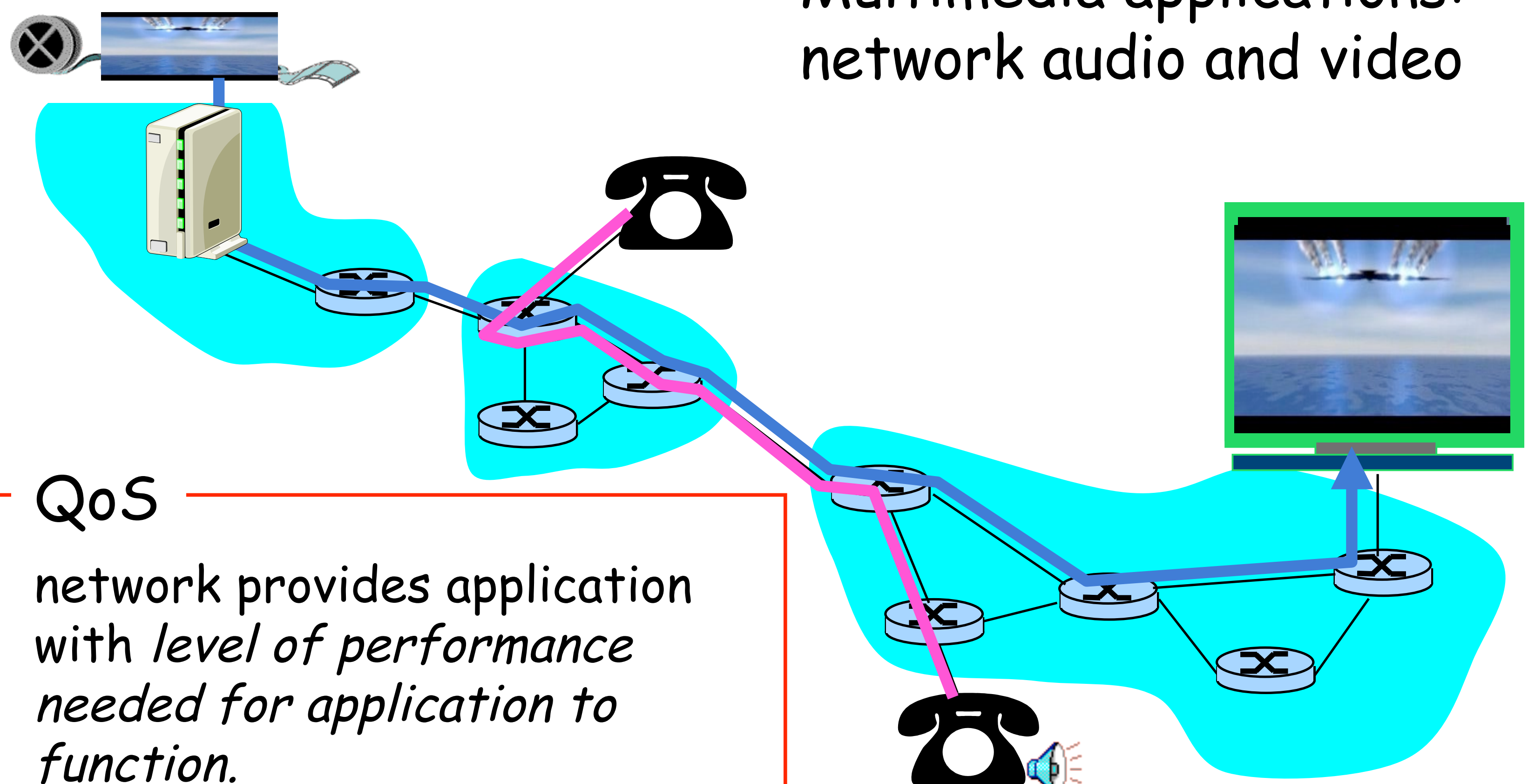
Multimedia applications:
network audio and video

Impairments:

- excessive delay: gaps in rendered audio, video
- excessive data loss

Quality of Service: What is it?

Multimedia applications:
network audio and video



QoS

network provides application with *level of performance needed for application to function.*

What is QoS?

- **“Better performance” as described by a set of parameters or measured by a set of metrics.**

- **Generic parameters:**
  - Bandwidth
  - Delay, Delay-jitter
  - Packet loss rate (or loss probability)

- **Transport/Application-specific parameters:**
  - Timeouts
  - Percentage of “important” packets lost

What is QoS (contd) ?

- These parameters can be measured at several granularities:
  - “micro” flow, aggregate flow, population.
- QoS considered “better” if
  - more parameters can be specified
  - QoS can be specified at a fine-granularity.
- QoS spectrum:

QoS: why don't we have it?

- QoS a concern since early 1980's
- Look at what's happen
  - Internet n
  - applicatio y, Gopher

The Internet is a huge transformative success !

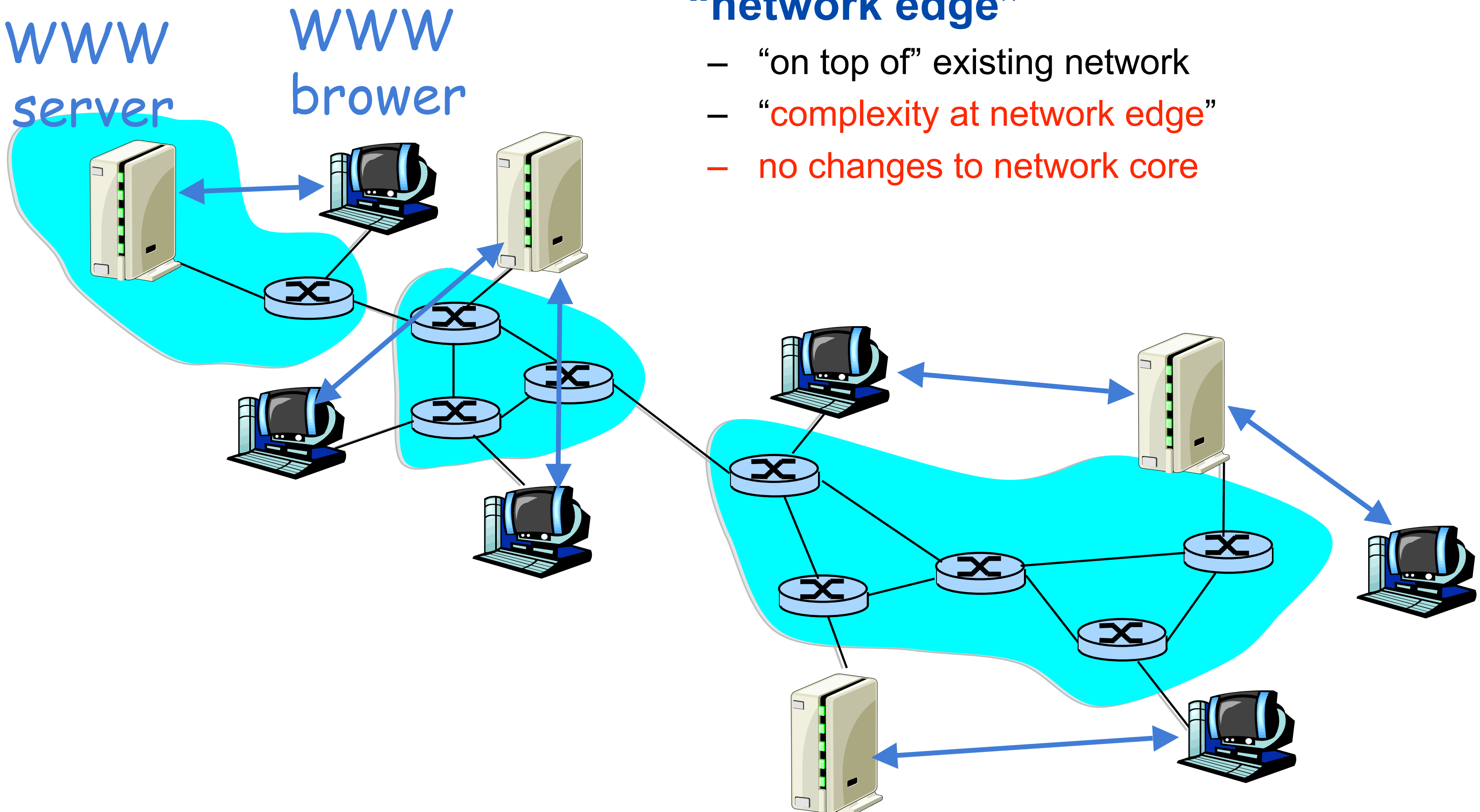
- Why limited progress on QoS?
  - lots of smart people working on it!

Why is the QoS problem so hard?

Why was the WWW so “easy”?

Implemented in hosts, servers at “network edge”

- “on top of” existing network
- “complexity at network edge”
- no changes to network core



Why is QoS more difficult?

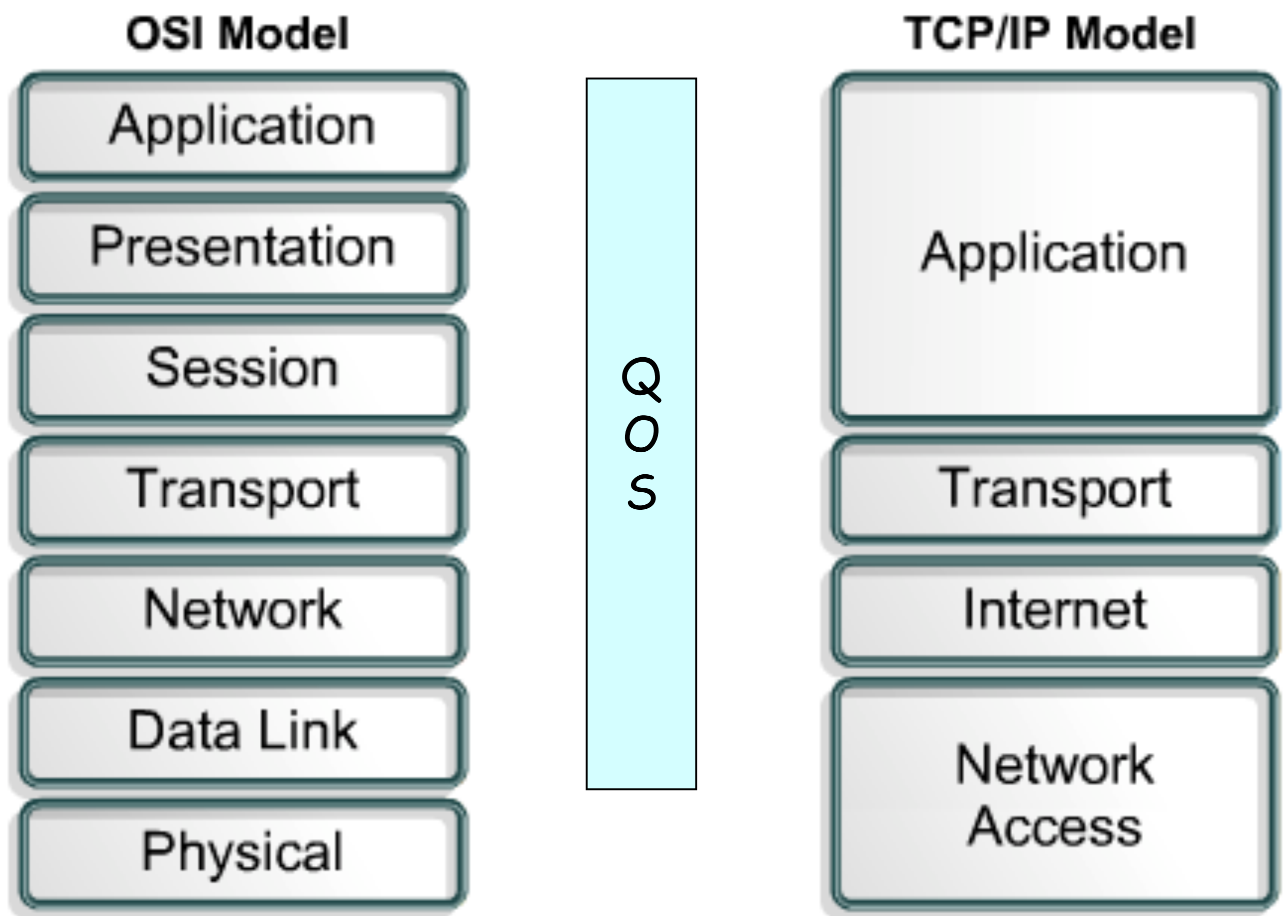
- **today's Internet core provides "best effort" service**
  - network congestion causes delays, loss
  - no timing guarantees
  - no loss guarantees
- **multimedia requires loss, timing constraints met**

"The different timing and reliability constraints of real-time communication require new protocols and architectures to be developed"

wet-behind-the-ears researcher, 1982.

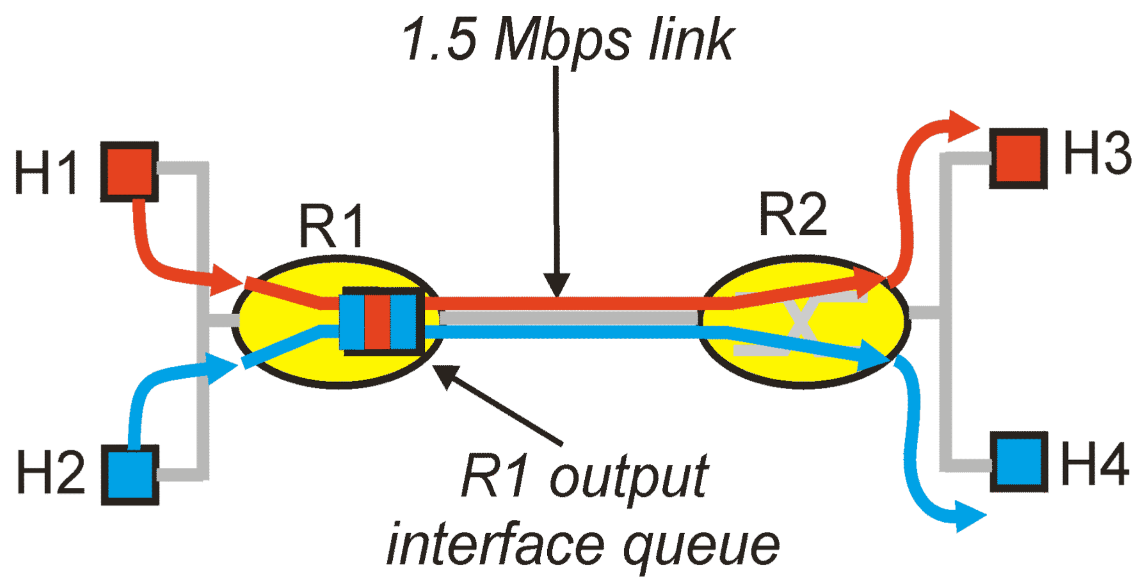
New architecture needed for network core!

Where to put QoS?



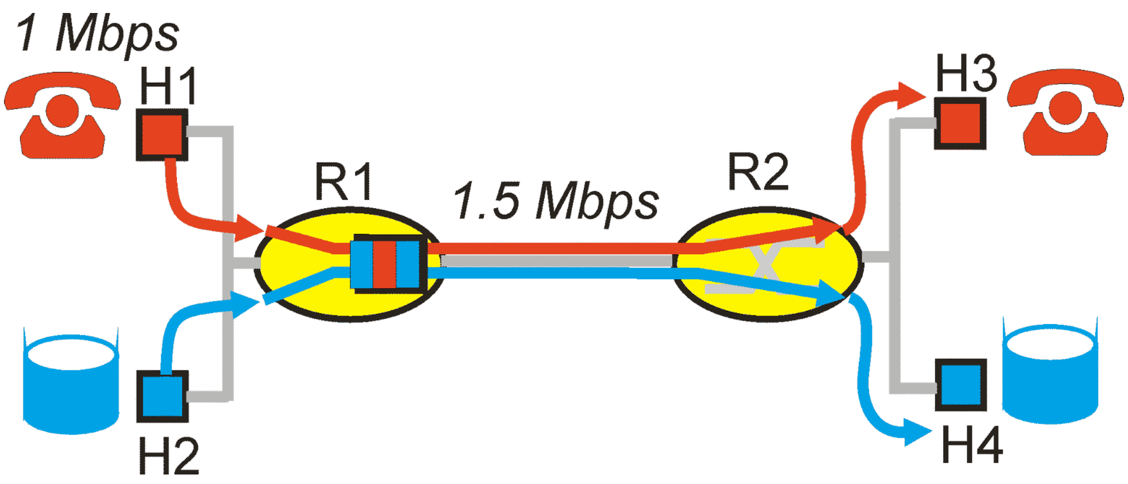
Améliorer la QoS dans les réseaux IP

- IETF travaille sur des propositions afin de fournir une meilleure QoS dans les réseaux IP, càd aller au-delà du service *best-effort*
- Les études en cours sont par exemple RSVP, RED, Differentiated Services
- Modèle simple pour notre étude:



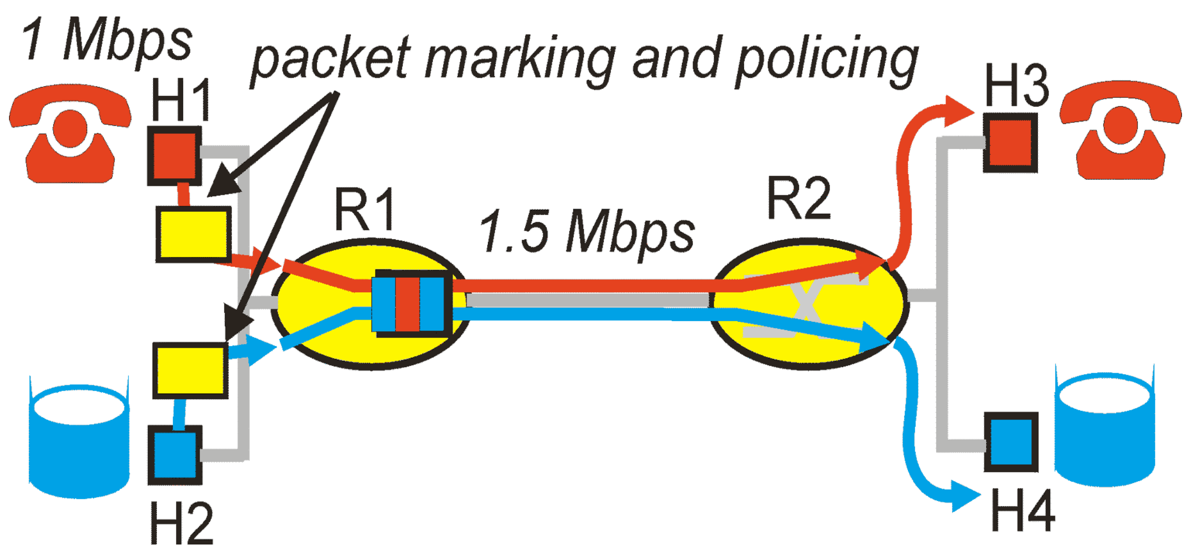
Principes pour garantir la QOS

- **Consider a phone application at 1Mbps and an FTP application sharing a 1.5 Mbps link.**
  - bursts of FTP can congest the router and cause audio packets to be dropped.
  - want to give priority to audio over FTP
- **PRINCIPLE 1: Marking of packets is needed for router to distinguish between different classes; and new router policy to treat packets accordingly**



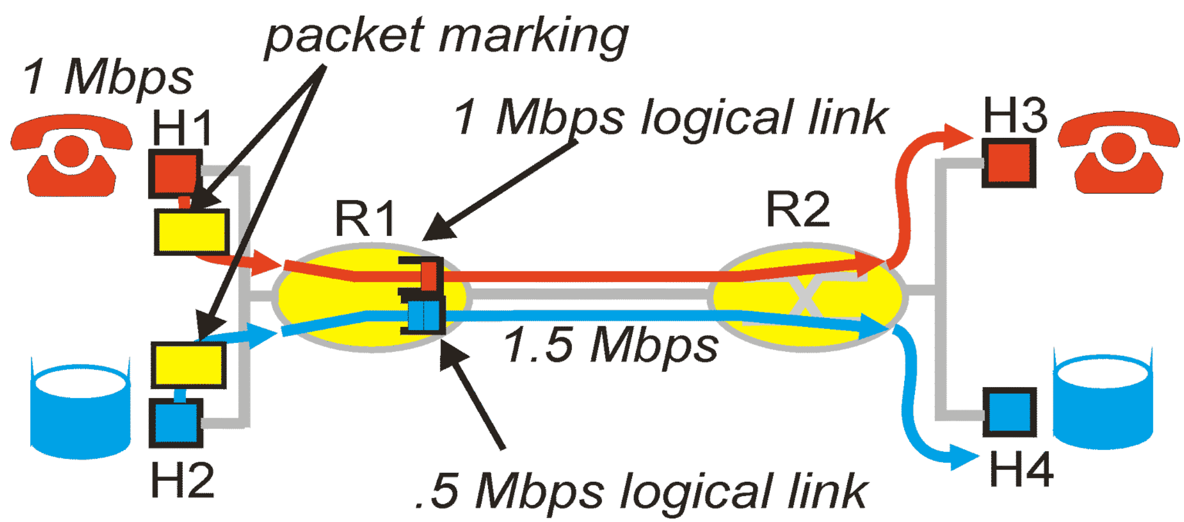
Principles for QOS Guarantees (more)

- Applications misbehave (audio sends packets at a rate higher than 1Mbps assumed above);
- PRINCIPLE 2: provide protection (isolation) for one class from other classes
- Require Policing Mechanisms to ensure sources adhere to bandwidth requirements; Marking and Policing need to be done at the edges:



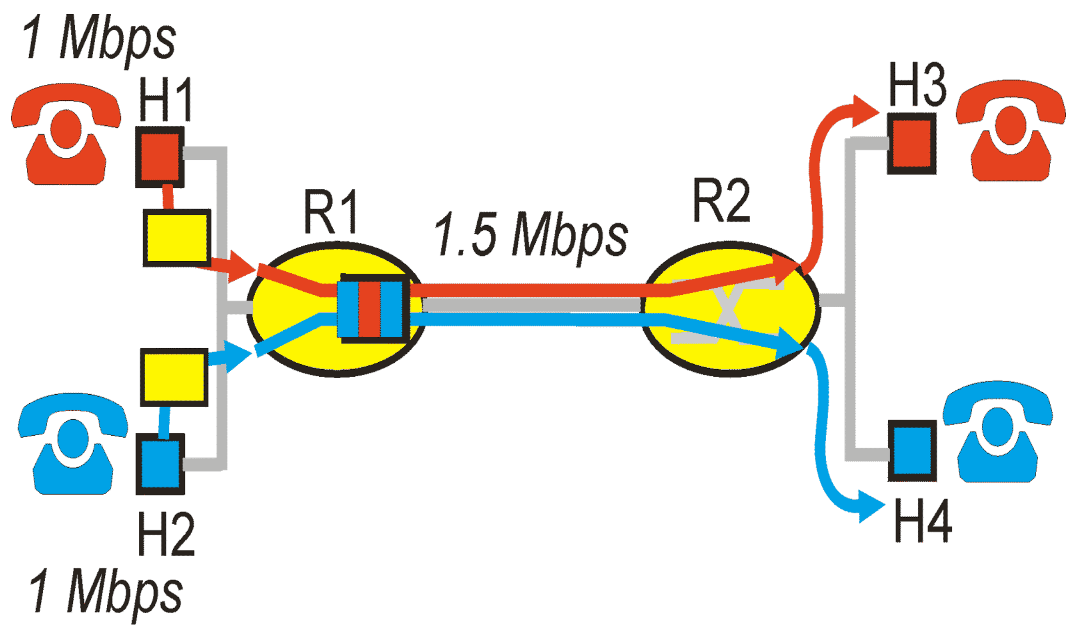
Principles for QOS Guarantees (more)

- Alternative to Marking and Policing: allocate a set portion of bandwidth to each application flow; can lead to inefficient use of bandwidth if one of the flows does not use its allocation
- PRINCIPLE 3: While providing isolation, it is desirable to use resources as efficiently as possible

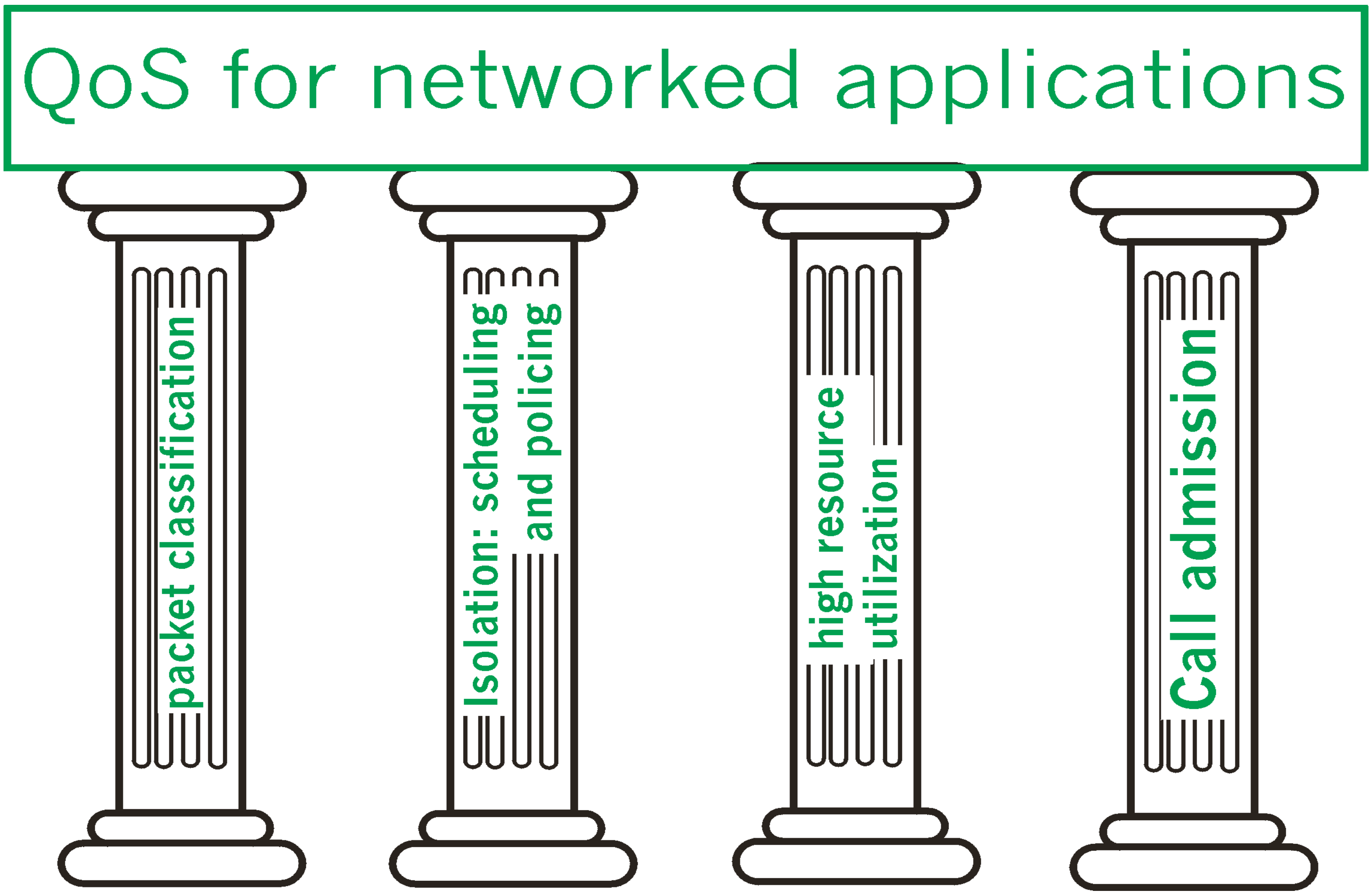


Principles for strong QOS Guarantees

- Cannot support traffic beyond link capacity
- PRINCIPLE 4: Need a Call Admission Process; application flow declares its needs, network may block call if it cannot satisfy the needs

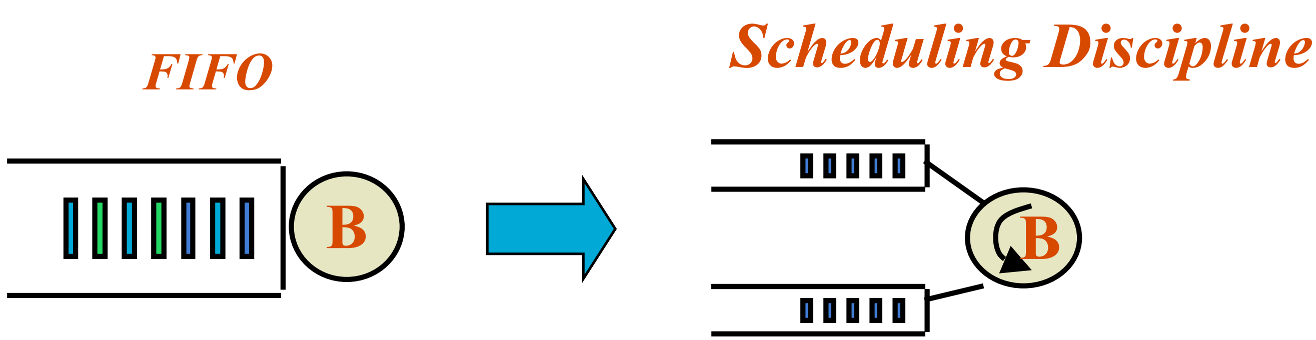


Summary

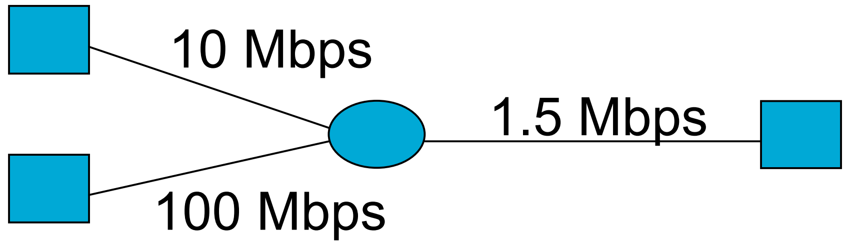

QoS for networked applications
packet classification
Isolation: scheduling and policing
high resource utilization
Call admission

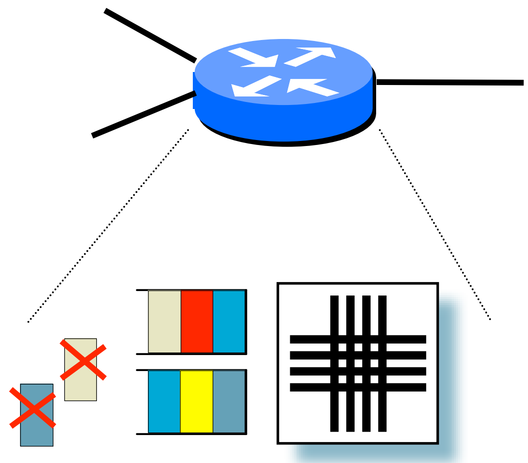
Maintenir une qualité de service

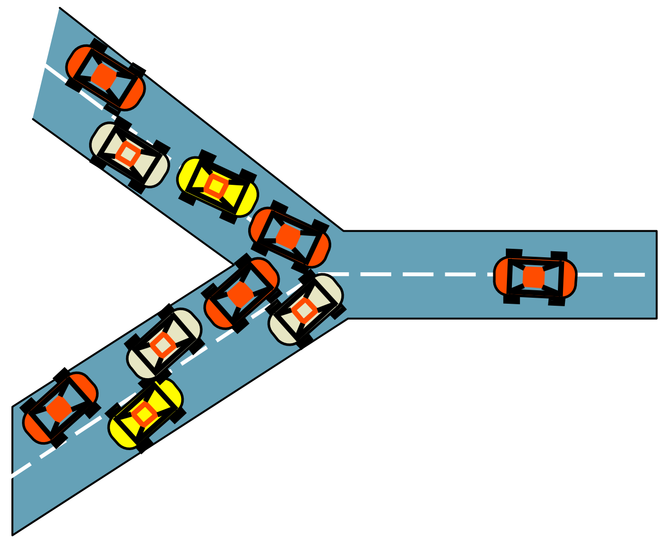
- **Eviter les pertes de paquets**
  - dans le réseau:
    - contrôle de l'admission, contrôle de congestion,contrôle de flux,
  - au récepteur:
    - régulation de l'émetteur (fenêtrage), contrôle de flux.
- **Eviter les trop longues latences**
  - notion de priorité
  - ordonnancement intelligents des paquets



The congestion phenomenon



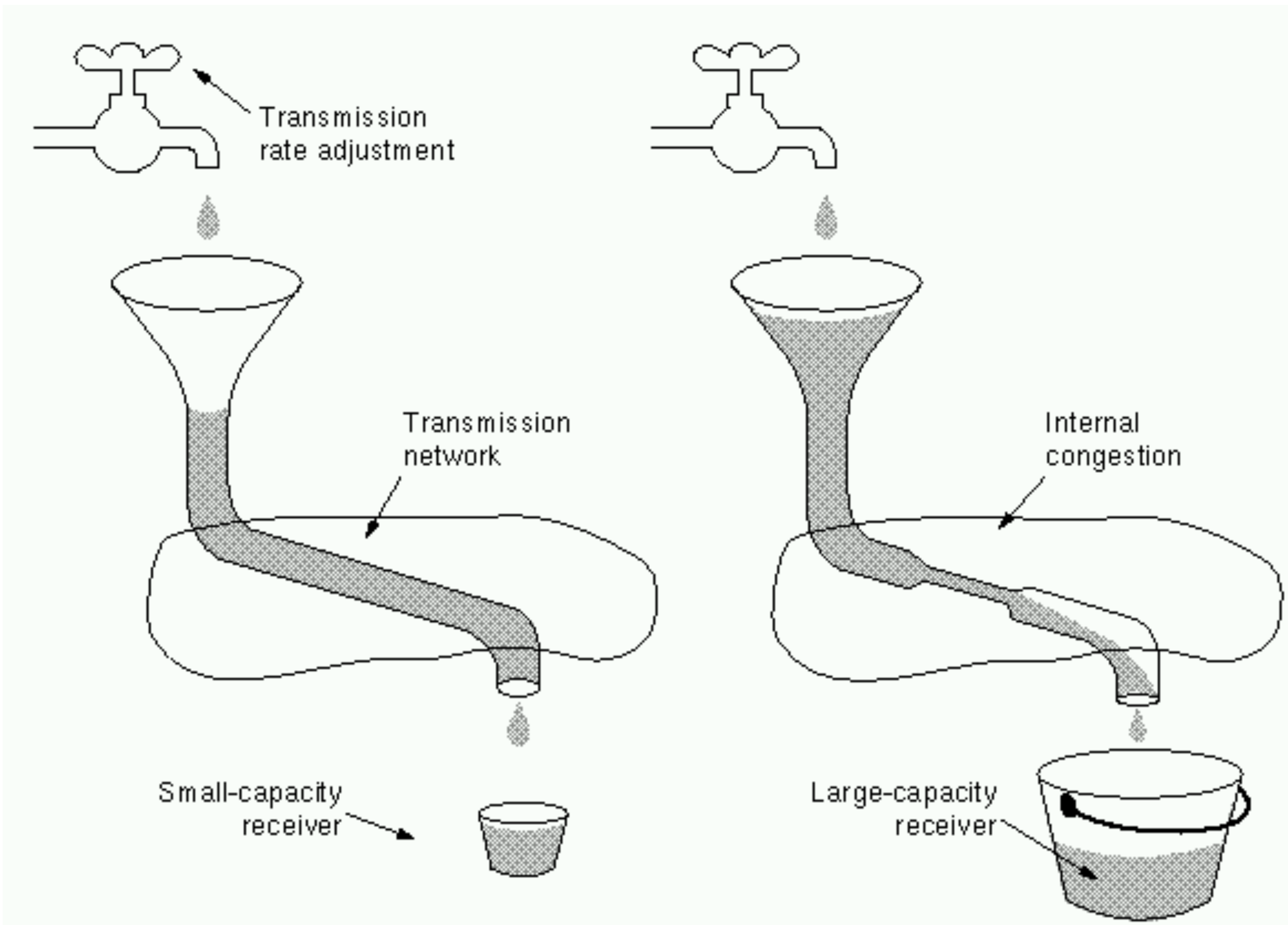




- Too many packets sent to the same interface.
- Difference bandwidth from one network to another

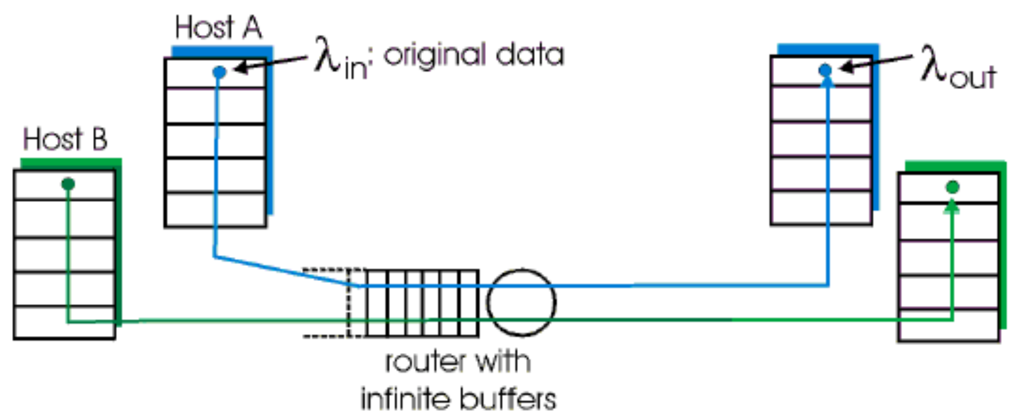
Main consequence: packet losses in routers

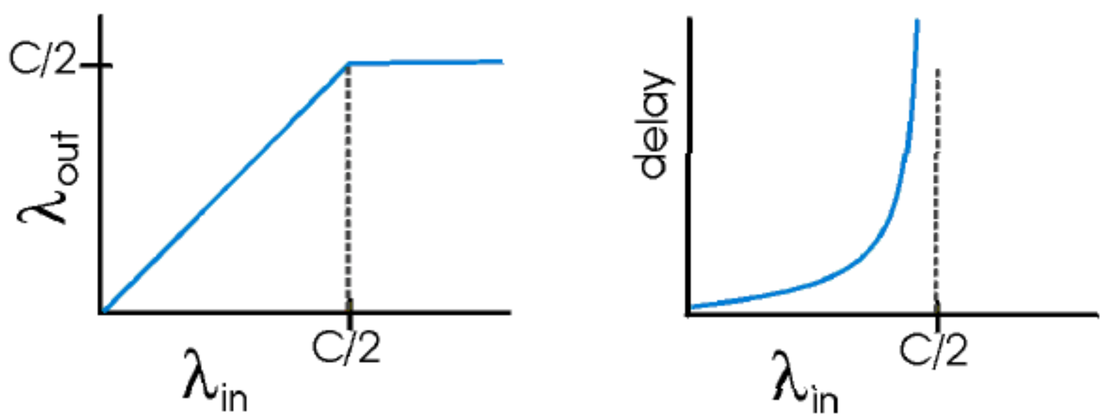
The problem of bottlenecks in networks



Causes/coûts de la congestion: scenario 1

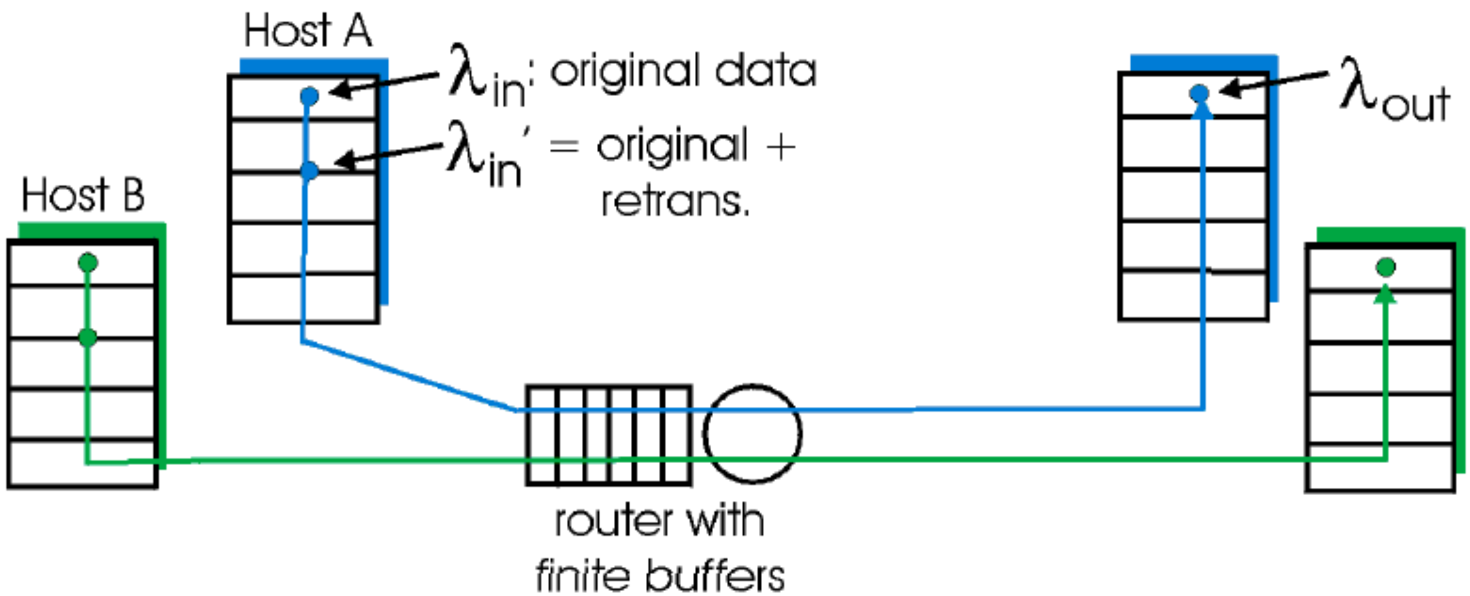
- Deux émetteurs, deux récepteurs
- un routeur, mémoire infinie
- Pas de retransmission





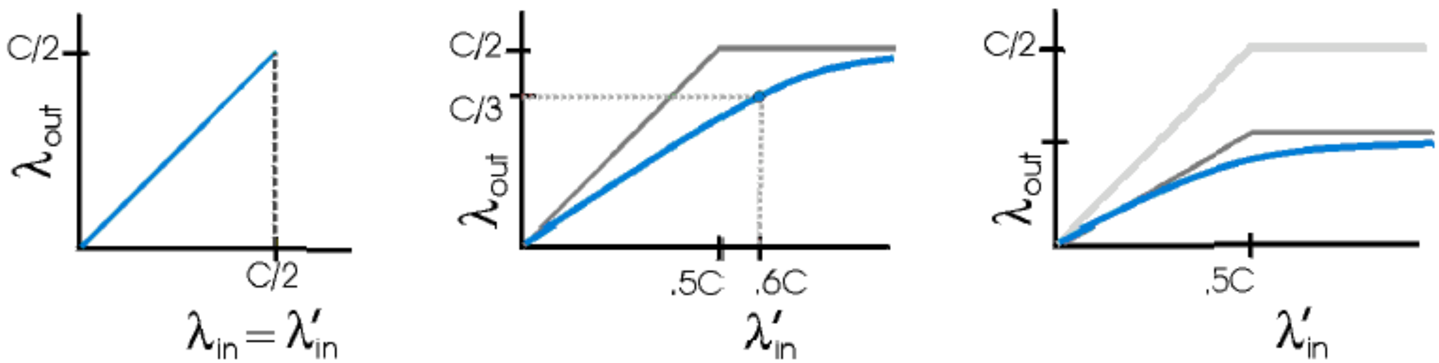
Causes/coûts de la congestion: scenario 2

- Un routeur, *mémoire finie*
- L'émetteur retransmet les paquets perdus



Causes/coûts de la congestion: scenario 2

- $\lambda_{in} = \lambda_{out}$ (goodput)
- Si la retransmission est parfaite : $\lambda'_{in} > \lambda_{out}$
- La retransmission de paquet non perdu rend ? $_{in}$ que dans le cas parfait

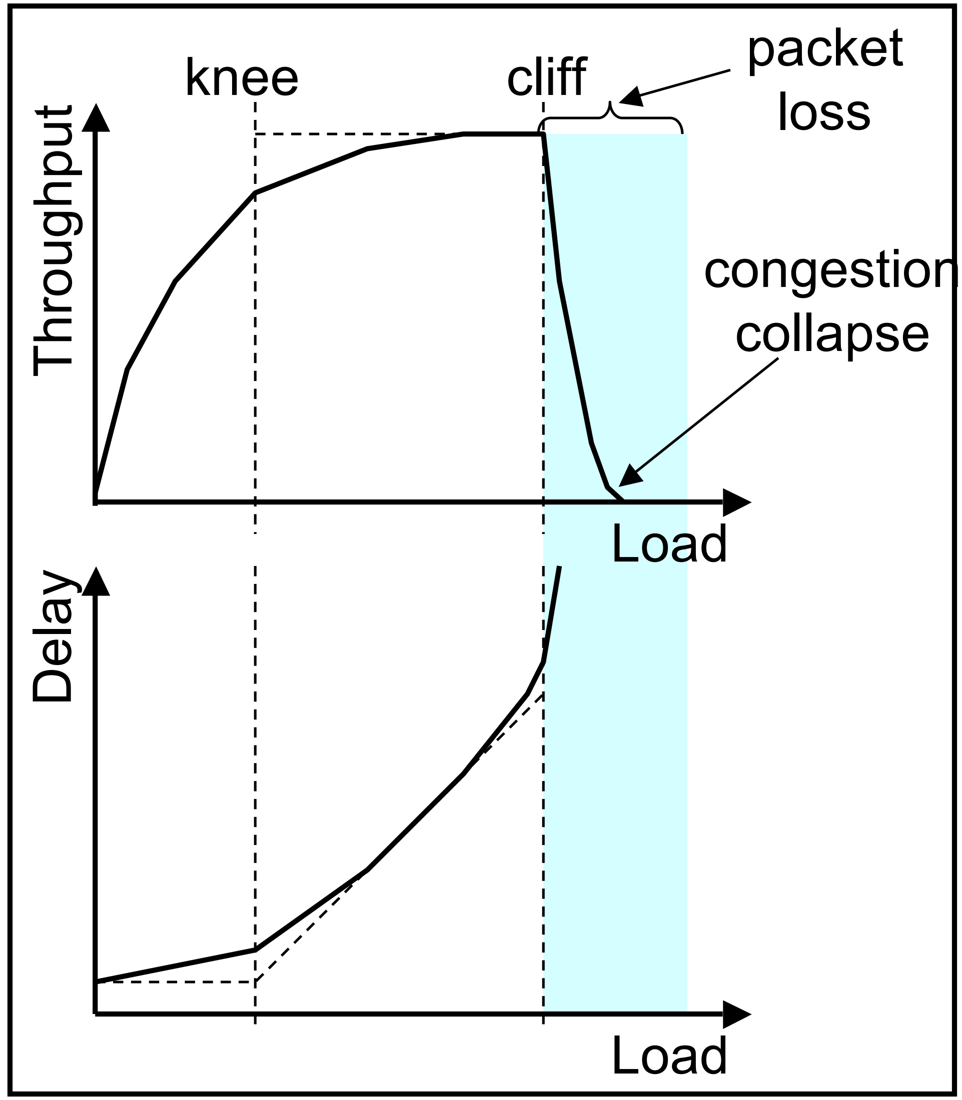


"coûts" de la congestion:

- Plus de travail (retrans) pour un même débit utile ("goodput")
- Retransmissions redondantes

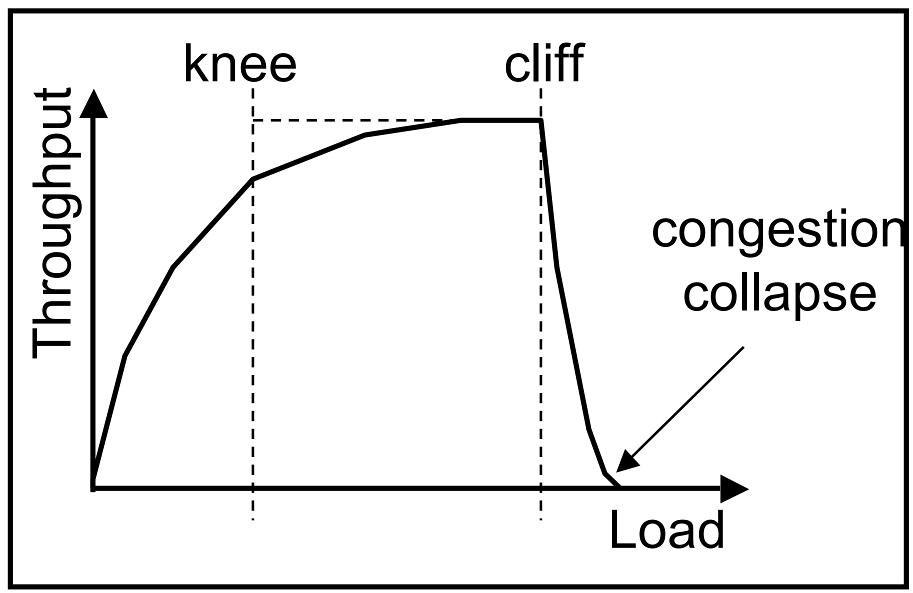
Congestion: A Close-up View

- **knee – point after which**
  - throughput increases very slowly
  - delay increases fast
- **cliff – point after which**
  - throughput starts to decrease very fast to zero (congestion collapse)
  - delay approaches infinity
- **Note (in an M/M/1 queue)**
  - delay = 1/(1 – utilization)

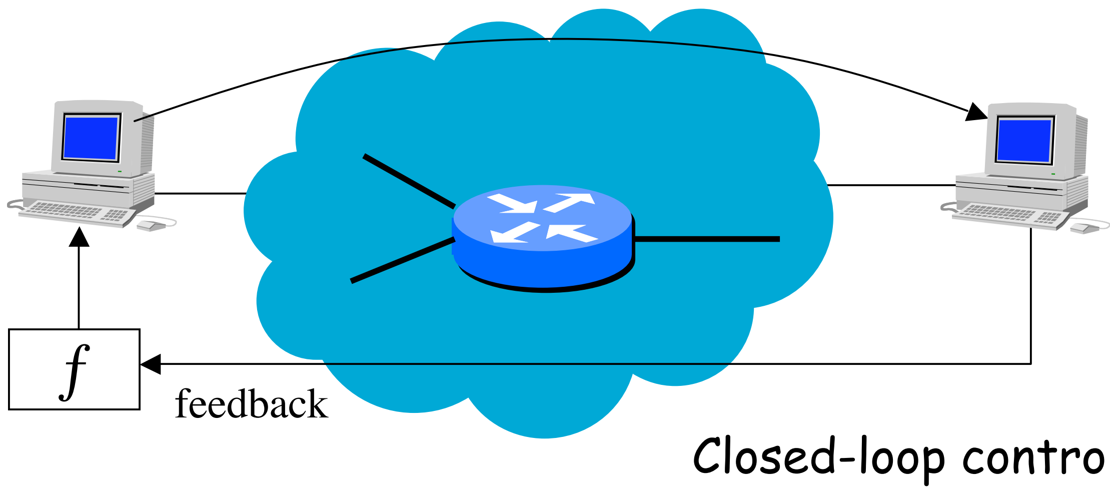


Congestion Control vs. Congestion Avoidance

- **Congestion control goal**
  - stay left of cliff
- **Congestion avoidance goal**
  - stay left of knee
- **Right of cliff:**
  - Congestion collapse



From the control theory point of view



- Feedback should be frequent, but not too much otherwise there will be oscillations
- Can not control the behavior with a time granularity less than the feedback period

Le contrôle de congestion: principes

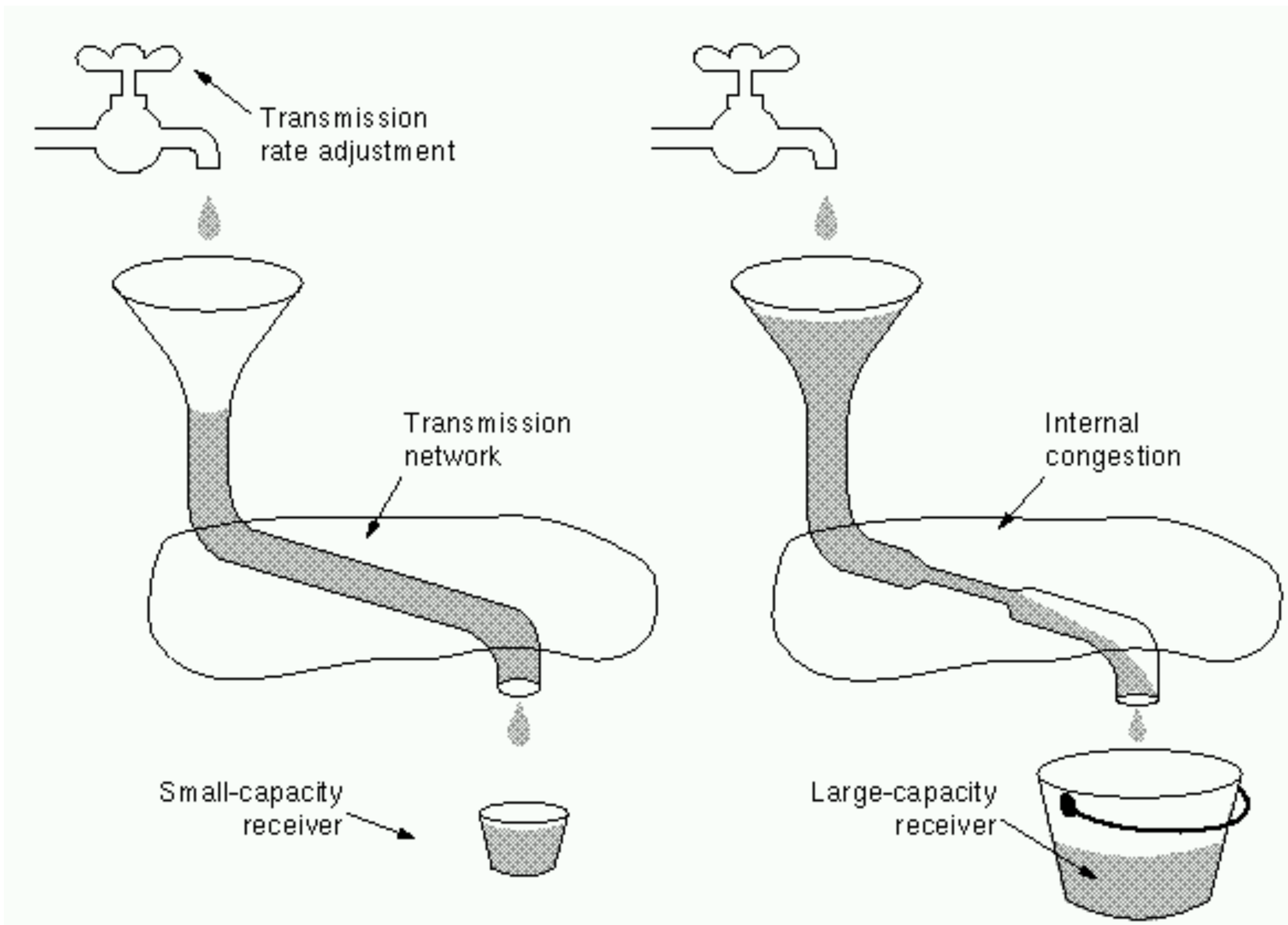
- **Réactif**
  - lorsque la congestion est détectée, informer les noeuds en amont et en aval,
  - puis, marquer des paquets, rejeter des paquets, traiter les paquets prioritaires.
- **Préventif**
  - diffusion périodique d'informations d'états (taille des buffers)
  - contrôle continue de la source (Leacky Bucket, Token Bucket...),
  - contrôle de flux, contrôle d'admission.
- **De bout en bout**
  - pas de retour du réseau
  - la congestion est estimée grâce à l'observation des pertes et des délais de bout-en-bout
- **Assisté par le réseau**
  - bit d'annonce de congestion (SNA, DECbit, TCP/ECN, FR, ATM)

Le contrôle de flux, pour le récepteur

- **Fenêtrage**
  - l'émetteur utilise une fenêtre d'anticipation dans laquelle il va pouvoir envoyer une certaine quantité de données sans acquittements
  - la taille de cette fenêtre peut être choisie par le récepteur à la phase de connexion
  - si l'émetteur respecte les règles, le récepteur ne sera pas surchargé.

Cela ne garantit pas que le contrôle de flux sera efficace pour le réseau (voir figure suivante).

Problème d’un réseau trop faible

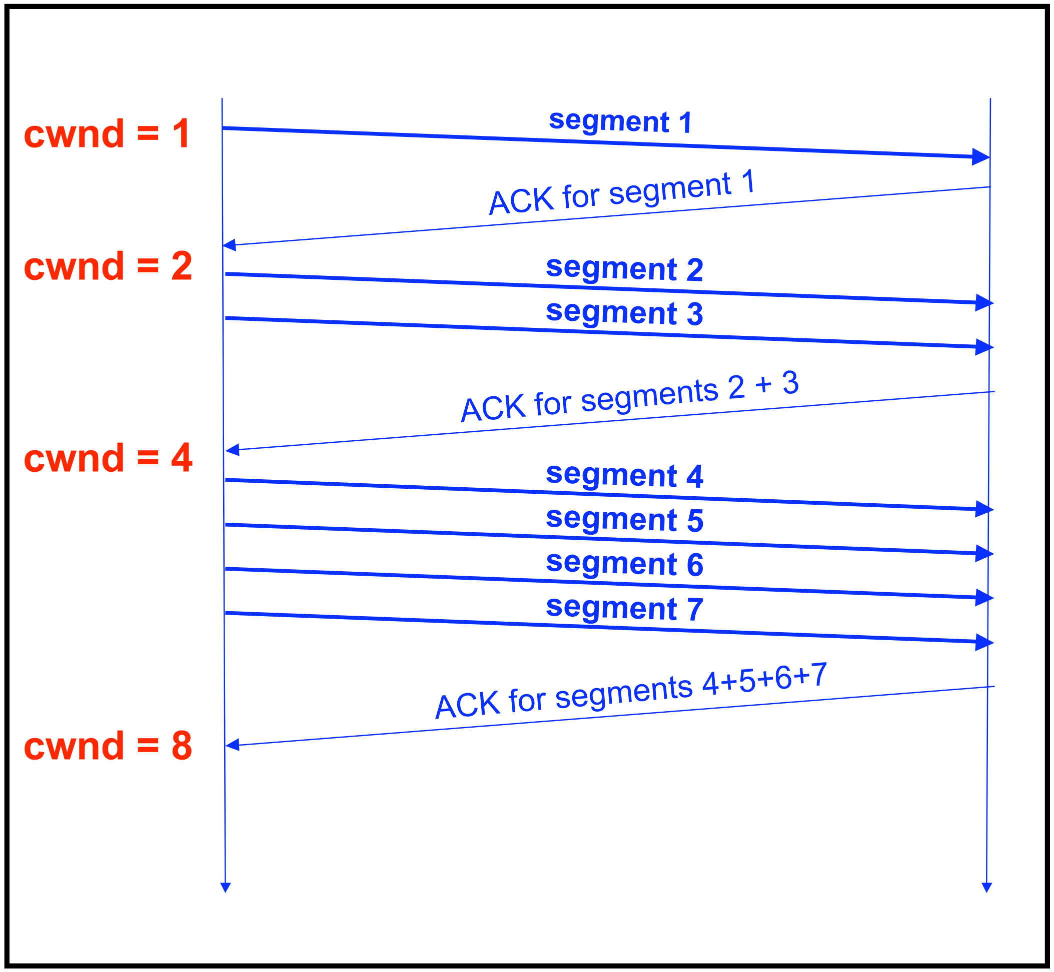


Le contrôle de flux pour le réseau

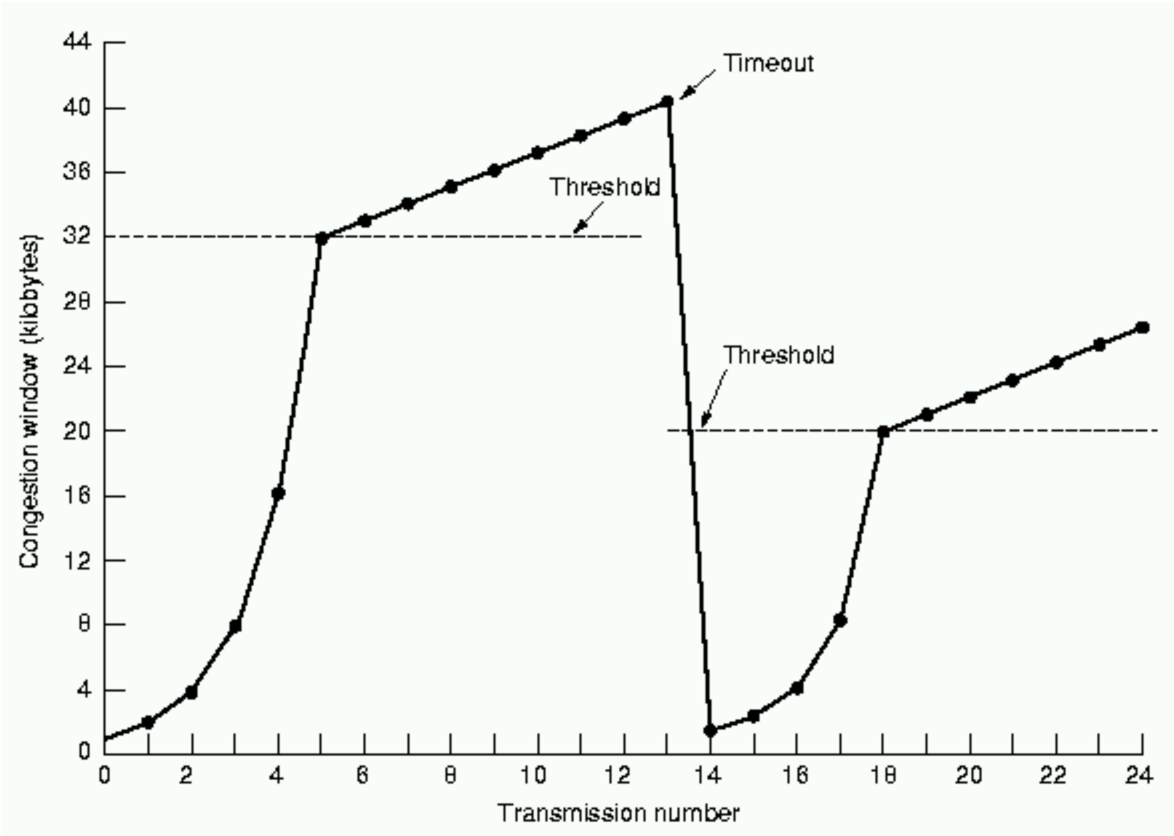
- **Ex: principe du contrôle de congestion dans TCP**
  - chaque émetteur maintient une deuxième fenêtre de congestion pour le réseau,
  - la quantité d'information qu'il est autorisé à transmettre par anticipation est le minimum des 2 fenêtres
  - initialement, la fenêtre de congestion est mise à K octets, l'émetteur envoie les données et arme un temporisateur,
  - si les données sont acquittées avant l'expiration du temporisateur, on augmente K, et ainsi de suite jusqu'à (i) l'expiration d'un temporisateur ou, (ii) la taille de la fenêtre du récepteur a été atteinte.
  - C'est le principe du "*slow start*"

Slow Start

- La fenêtre de congestion augmente en fait très rapidement!

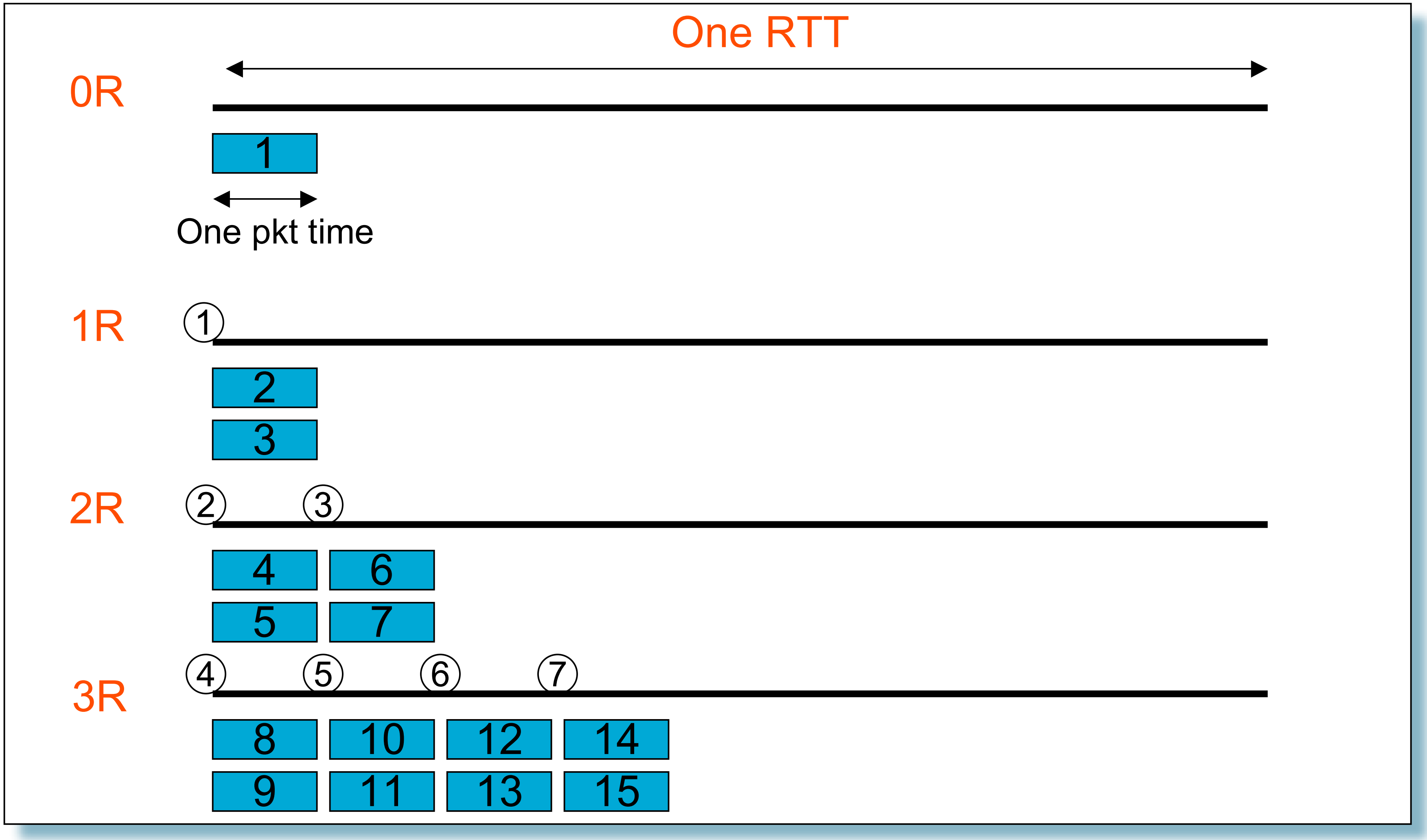


Le contrôle de congestion dans TCP

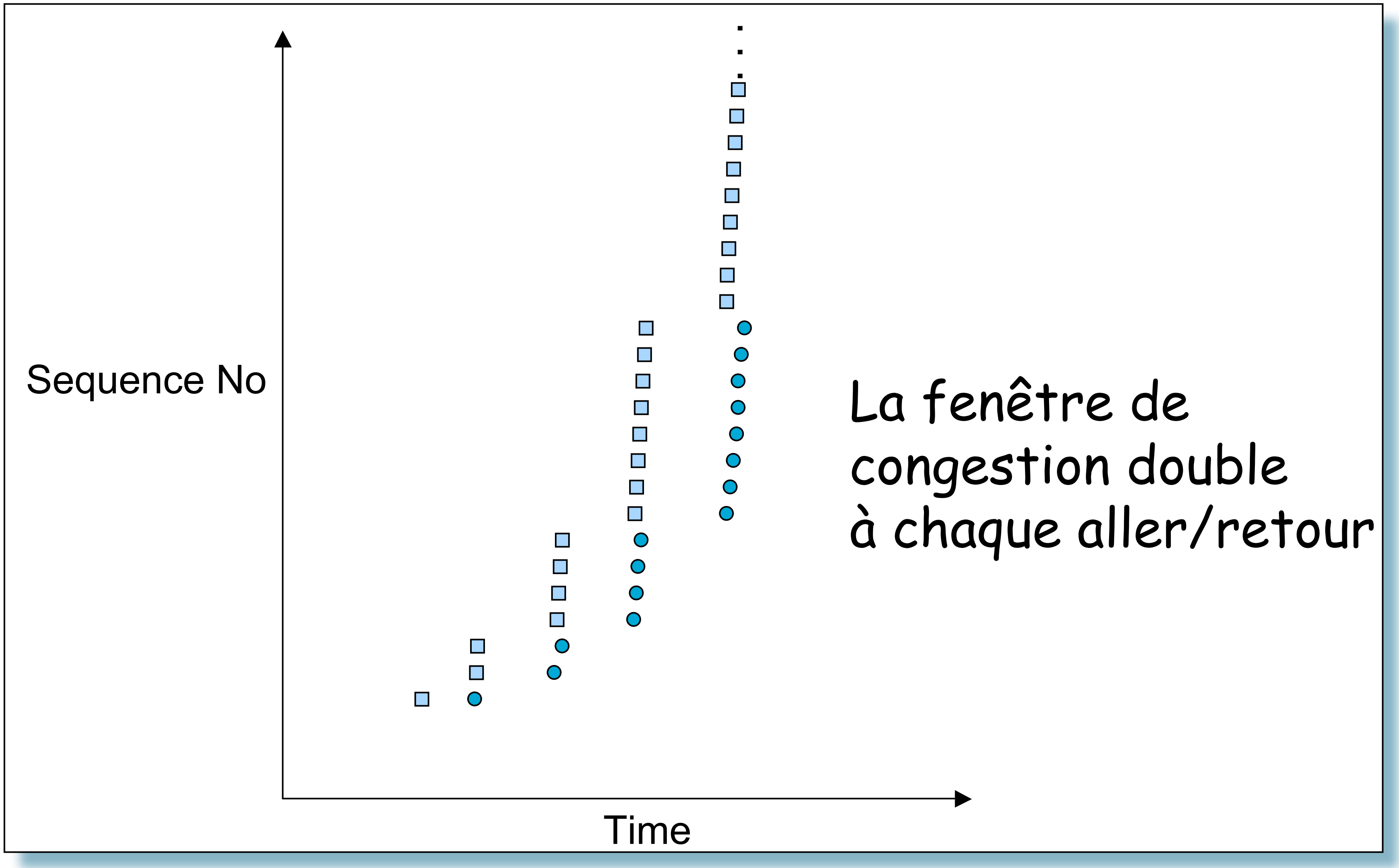


- **seuil initial a 64K, on augmente K exponentiellement avant et linéairement après (*congestion avoidance*),**
- **si perte, divise le seuil par 2, et on recommence avec K=1**

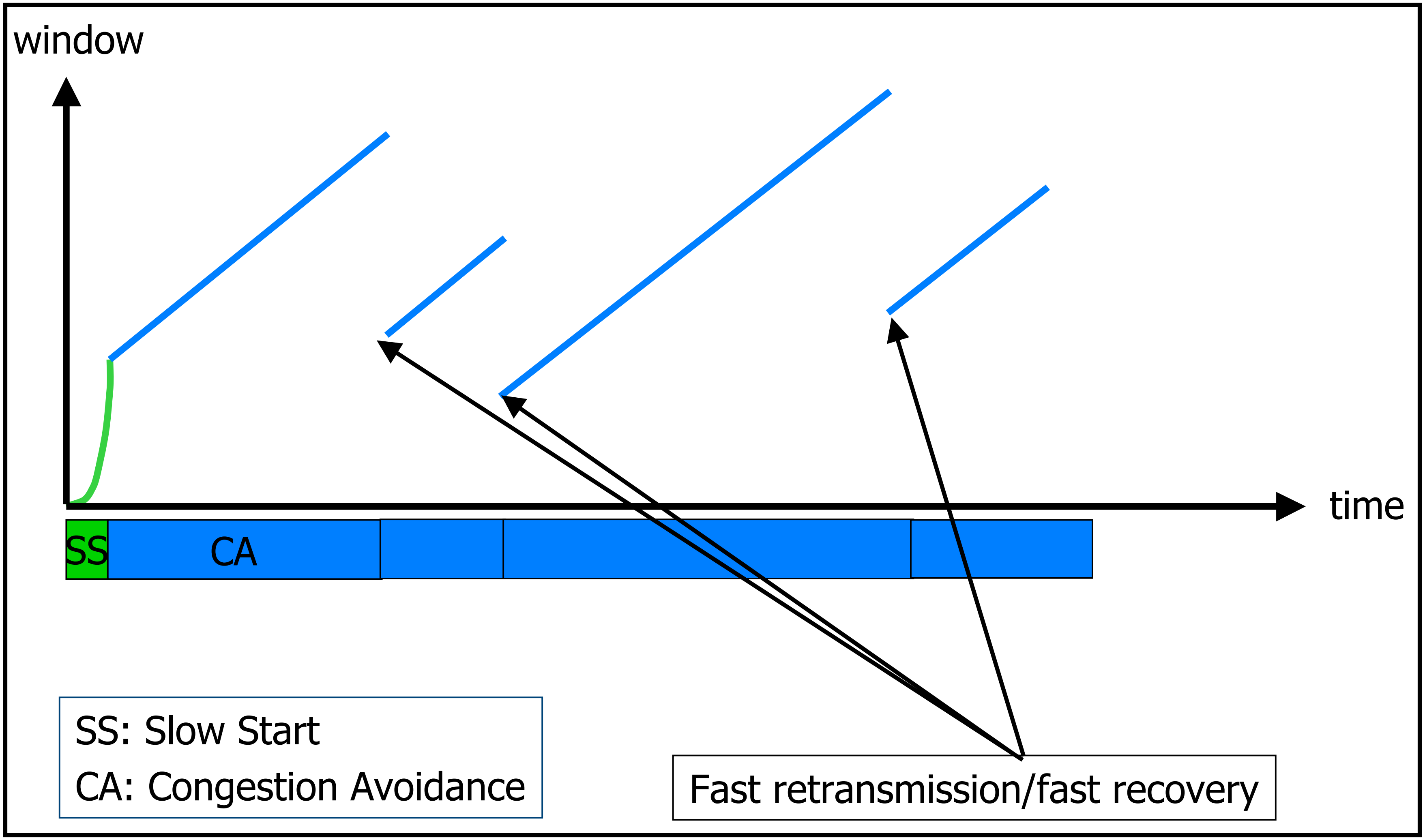
Utilisation du Round Trip Time



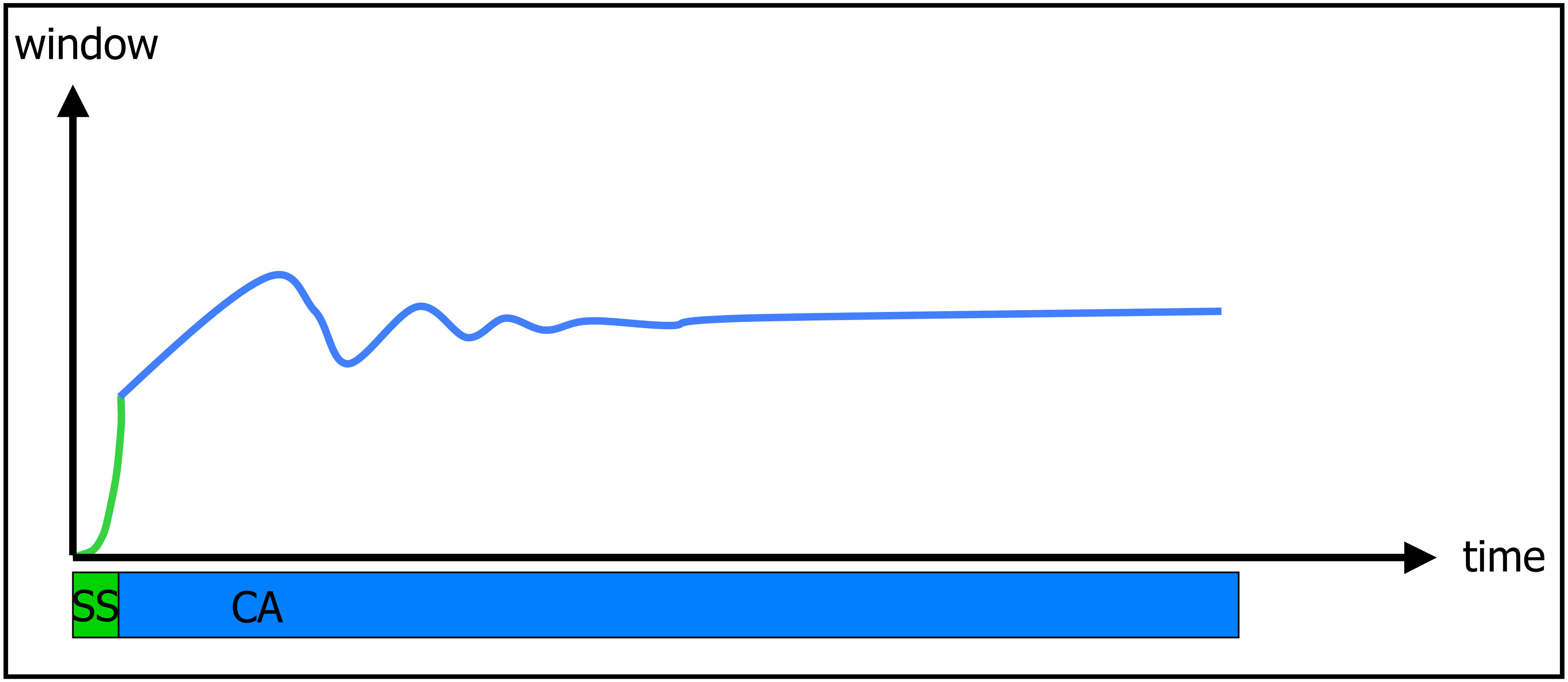
Slow Start Sequence Plot



TCP Reno (Jacobson 1990)



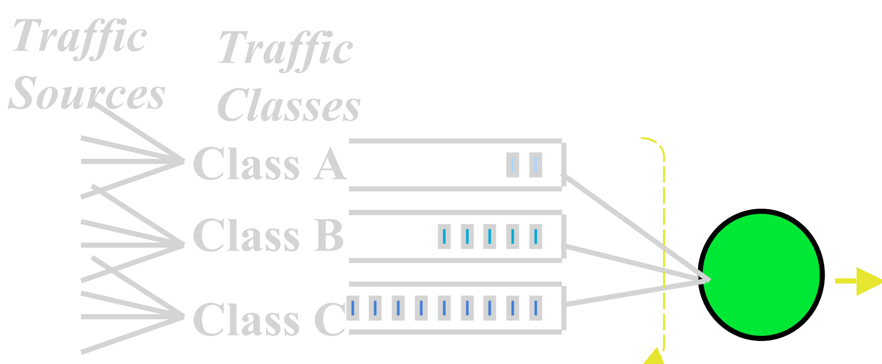
TCP Vegas (Brakmo & Peterson 1994)



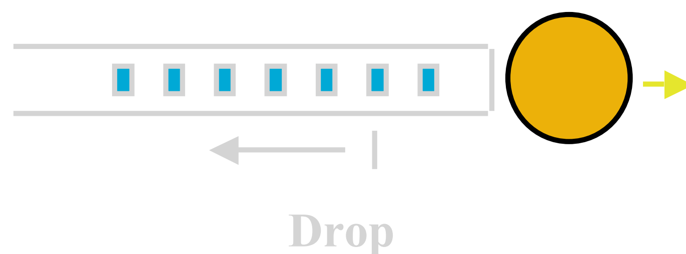
- Converges, no retransmission
- … provided buffer is large enough

Queuing Disciplines

- **Each router must implement some queuing discipline**
- **Queuing allocates bandwidth and buffer space:**
  - Bandwidth: which packet to serve next (scheduling)
  - Buffer space: which packet to drop next (buff mgmt)
- **Queuing also affects latency**



Scheduling



Buffer Management

Typical Internet Queuing

- FIFO + drop-tail
  - Simplest choice
  - Used widely in the Internet
- FIFO (first-in-first-out)
  - Implies single class of traffic
- Drop-tail
  - Arriving packets get dropped when queue is full regardless of flow or importance
- Important distinction:
  - FIFO: scheduling discipline
  - Drop-tail: drop (buffer management) policy

FIFO + Drop-tail Problems

- FIFO Issues: In a FIFO discipline, the service seen by a flow is *convoluted* with the arrivals of packets from all other flows!
  - No isolation between flows: full burden on e2e control
  - No policing: send more packets → get more service
- Drop-tail issues:
  - Routers are forced to have have large queues to maintain high utilizations
  - Larger buffers => larger steady state queues/delays
  - Synchronization: end hosts react to same events because packets tend to be lost in bursts
  - Lock-out: a side effect of burstiness and synchronization is that a few flows can monopolize queue space

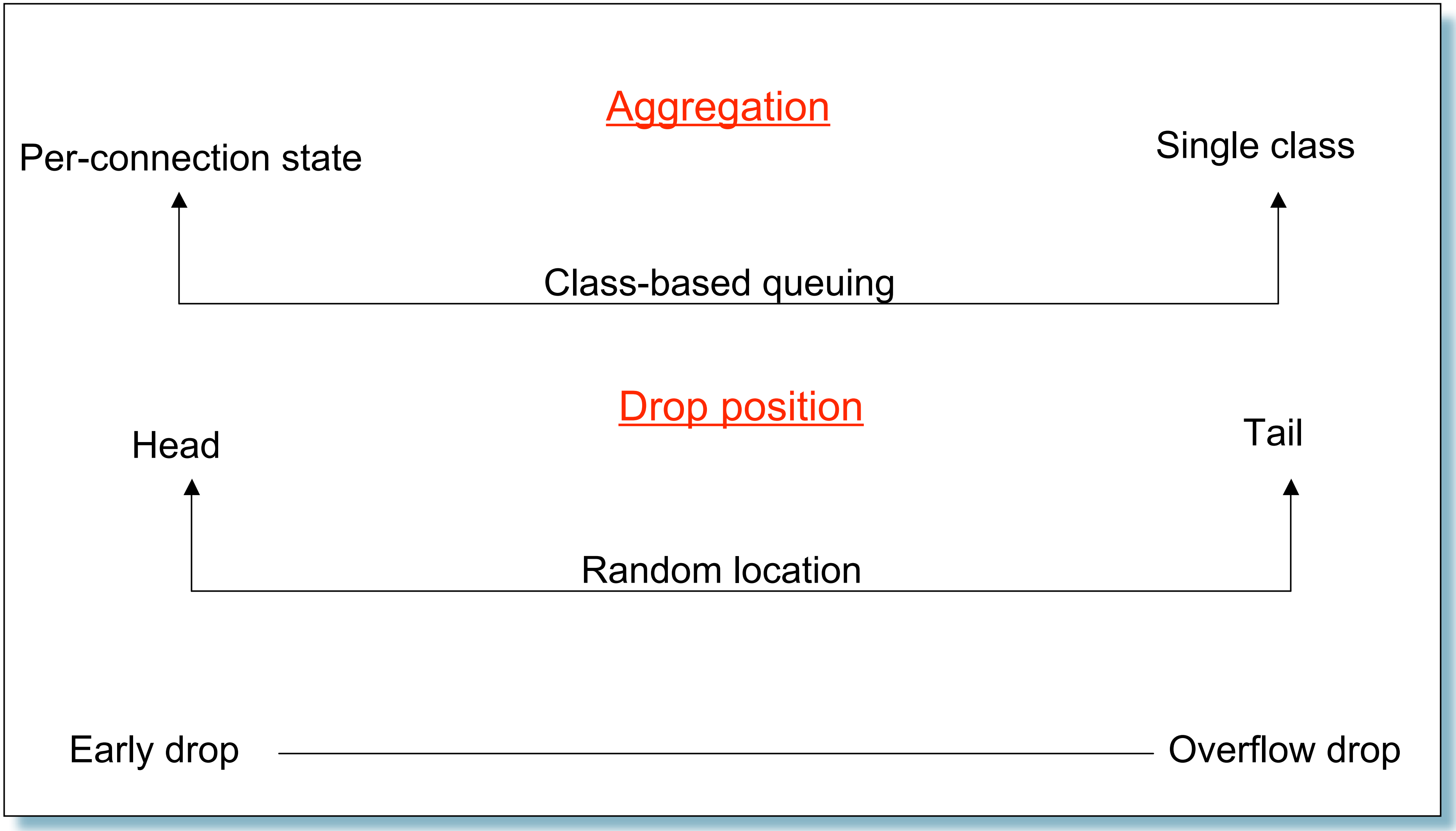
Design Objectives

- Keep throughput high and delay low (i.e. knee)
- Accommodate bursts
- Queue size should reflect ability to accept bursts rather than steady-state queuing
- Improve TCP performance with minimal hardware changes

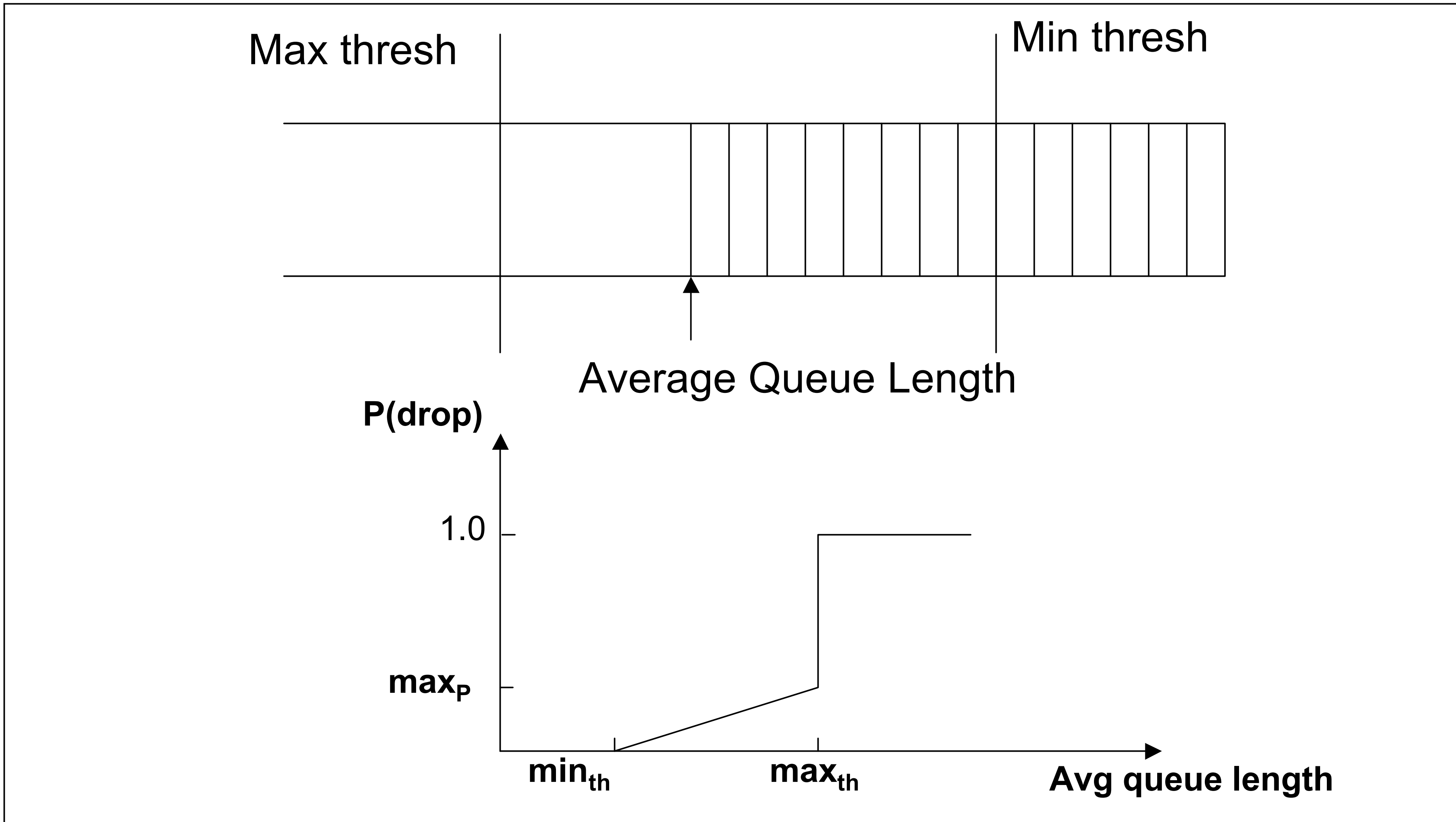
Queue Management Ideas

- **Synchronization, lock-out:**
  - Random drop: drop a randomly chosen packet
  - Drop front: drop packet from head of queue
- **High steady-state queuing vs burstiness:**
  - Early drop: Drop packets before queue full
  - Do not drop packets “too early” because queue may reflect only burstiness and not true overload
- **Misbehaving vs Fragile flows:**
  - Drop packets proportional to queue occupancy of flow
  - Try to protect fragile flows from packet loss (eg: color them or classify them on the fly)
- **Drop packets vs Mark packets:**
  - Dropping packets interacts w/ reliability mechanisms
  - Mark packets: need to trust end-systems to respond!

Packet Drop Dimensions



Random Early Detection (RED)



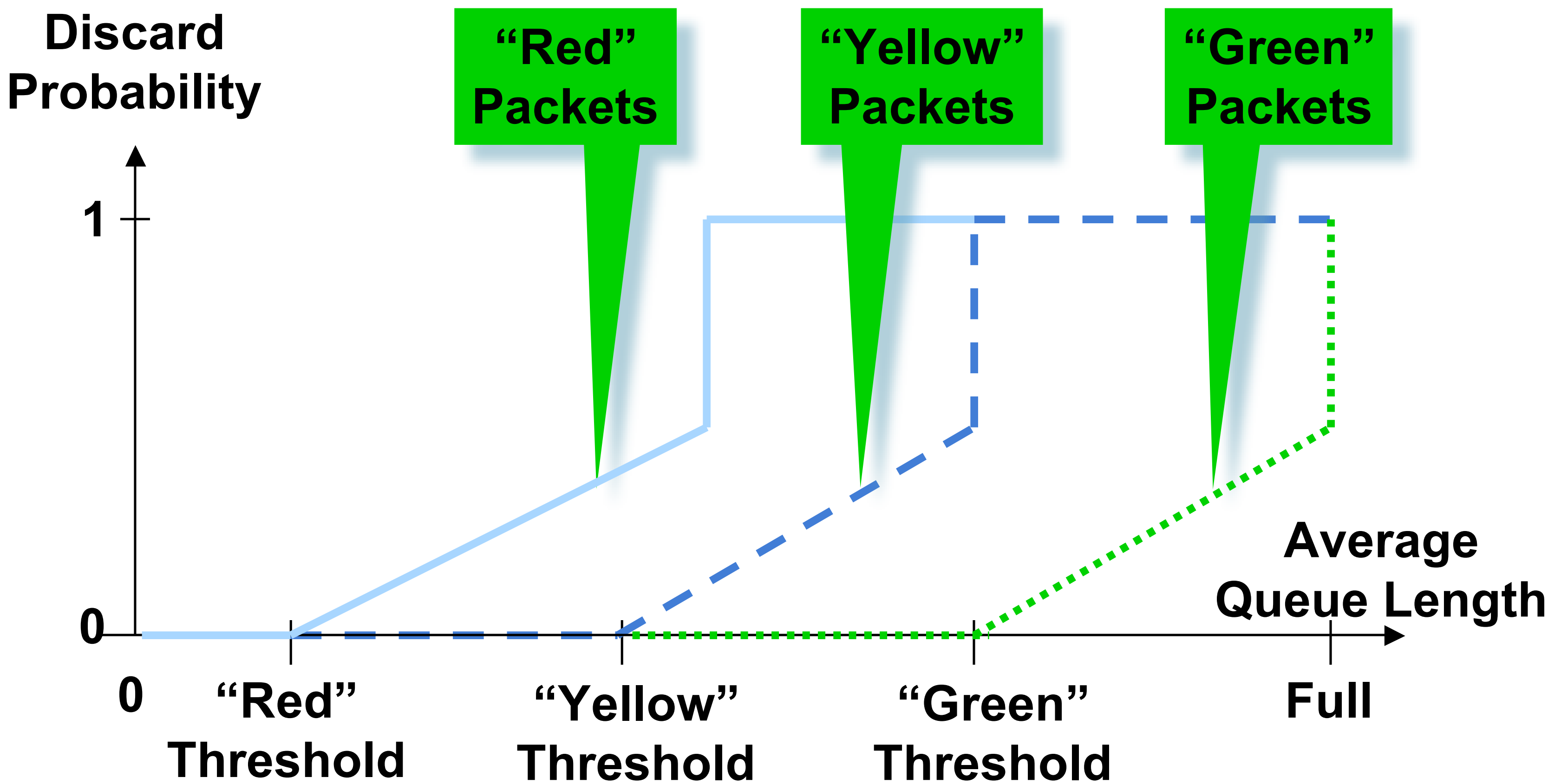
Random Early Detection (RED)

- **Maintain running average of queue length**
  - Low pass filtering
- **If avg Q < min_{th} do nothing**
  - Low queuing, send packets through
- **If avg Q > max_{th}, drop packet**
  - Protection from misbehaving sources
- **Else mark (or drop) packet in a manner proportional to queue length & bias to protect against synchronization**
  - $P_b = max_p(avg - min_{th}) / (max_{th} - min_{th})$
  - Further, bias P_b by history of unmarked packets
  - $P_a = P_b/(1 - count*P_b)$

RED Issues

- **Issues:**
  - Breaks synchronization well
  - Extremely sensitive to parameter settings
  - Wild queue oscillations upon load changes
  - Fail to prevent buffer overflow as #sources increases
  - Does not help fragile flows (eg: small window flows or retransmitted packets)
  - Does not adequately isolate cooperative flows from non-cooperative flows
- **Isolation:**
  - Fair queuing achieves isolation using per-flow state
  - RED penalty box: Monitor history for packet drops, identify flows that use disproportionate bandwidth

RED with Multiple Thresholds

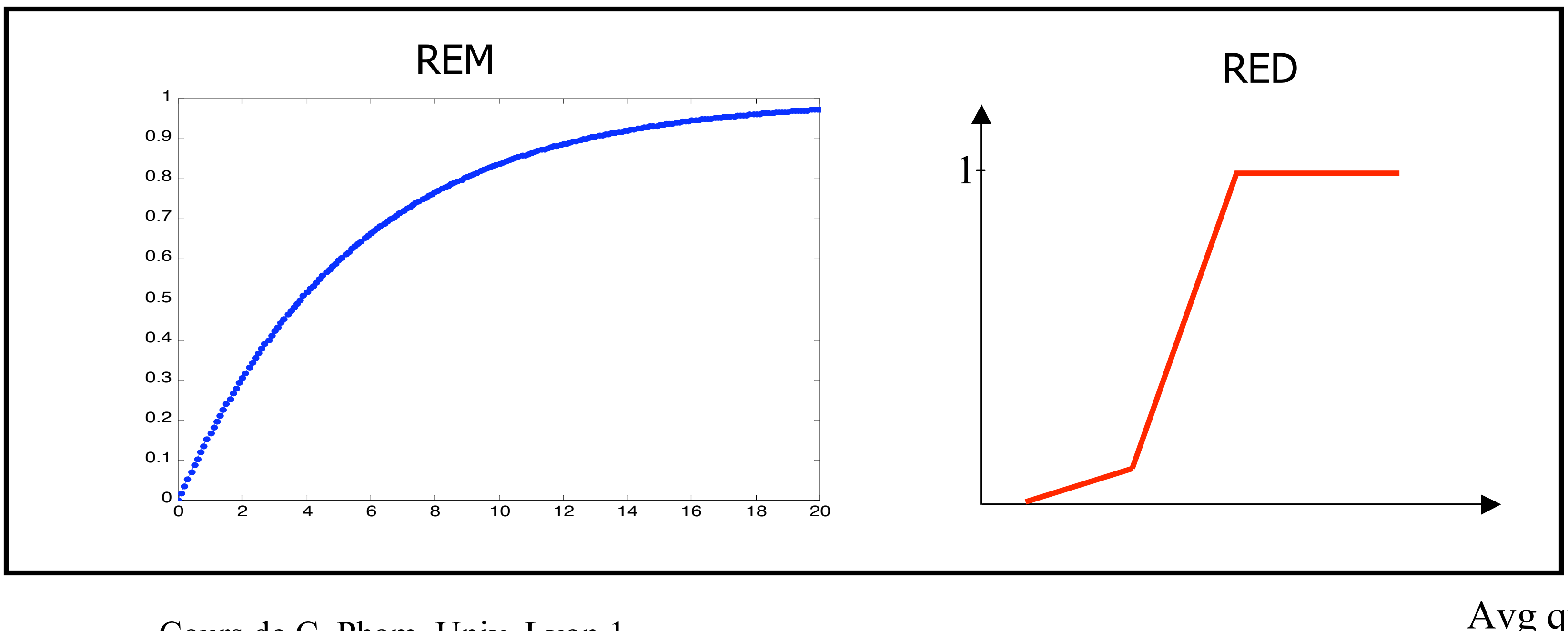


source Juha Heinänen

REM Athuraliya & Low 2000

Main ideas

- Decouple congestion & performance measure
- "Price" adjusted to match rate and clear buffer
- Marking probability exponential in `price'



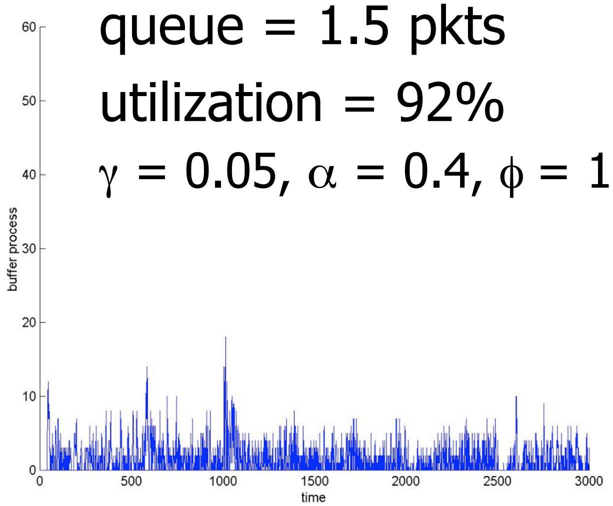
Comparison of AQM Performance

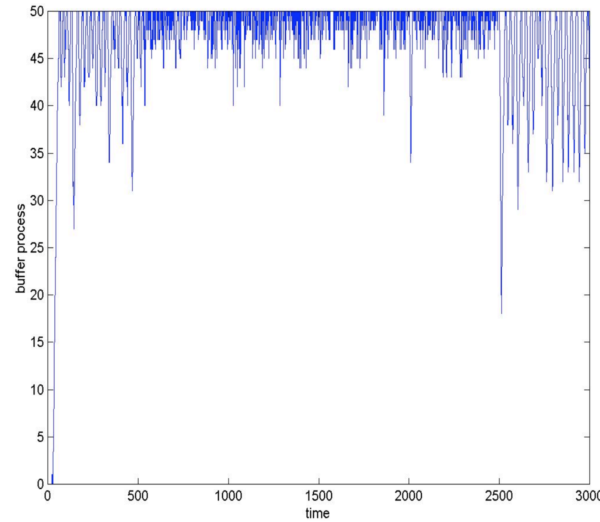
REM

queue = 1.5 pkts

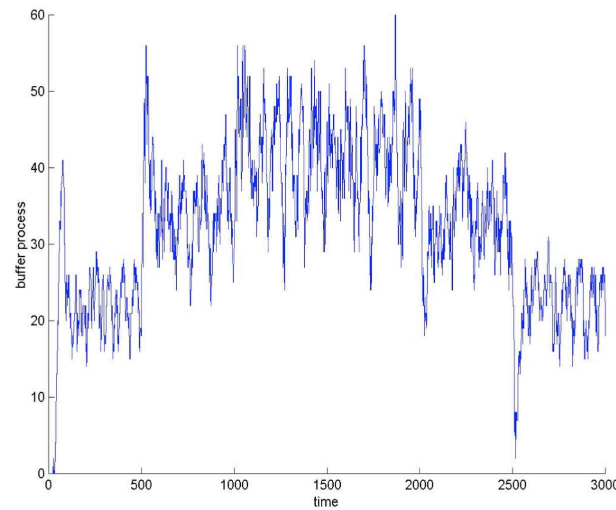
utilization = 92%

$\gamma = 0.05$, $\alpha = 0.4$, $\phi = 1.15$





DropTail



RED

min_th = 10 pkts

max_th = 40 pkts

max_p = 0.1

Service Specification

- **Loss:** probability that a flow’s packet is lost
- **Delay:** time it takes a packet’s flow to get from source to destination
- **Delay jitter:** maximum difference between the delays experienced by two packets of the flow
- **Bandwidth:** maximum rate at which the soource can send traffic
- **QoS spectrum:**

Hard Real Time: Guaranteed Services

- **Service contract**
  - Network to client: guarantee a deterministic upper bound on delay for each packet in a session
  - Client to network: the session does not send more than it specifies
- **Algorithm support**
  - Admission control based on worst-case analysis
  - Per flow classification/scheduling at routers

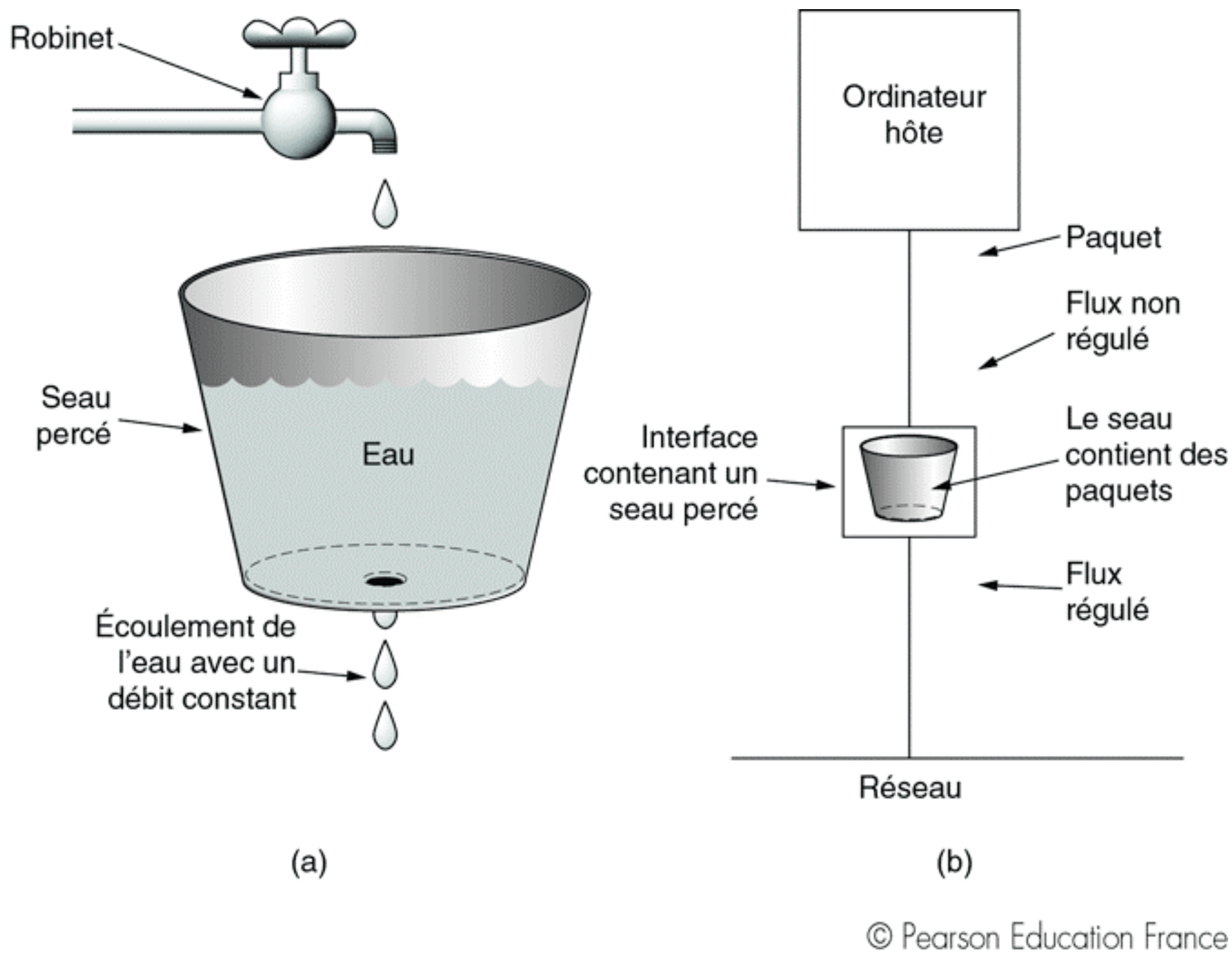
Soft Real Time: Controlled Load Service

- **Service contract:**
  - Network to client: similar performance as an unloaded best-effort network
  - Client to network: the session does not send more than it specifies
- **Algorithm Support**
  - Admission control based on measurement of aggregates
  - Scheduling for aggregate possible

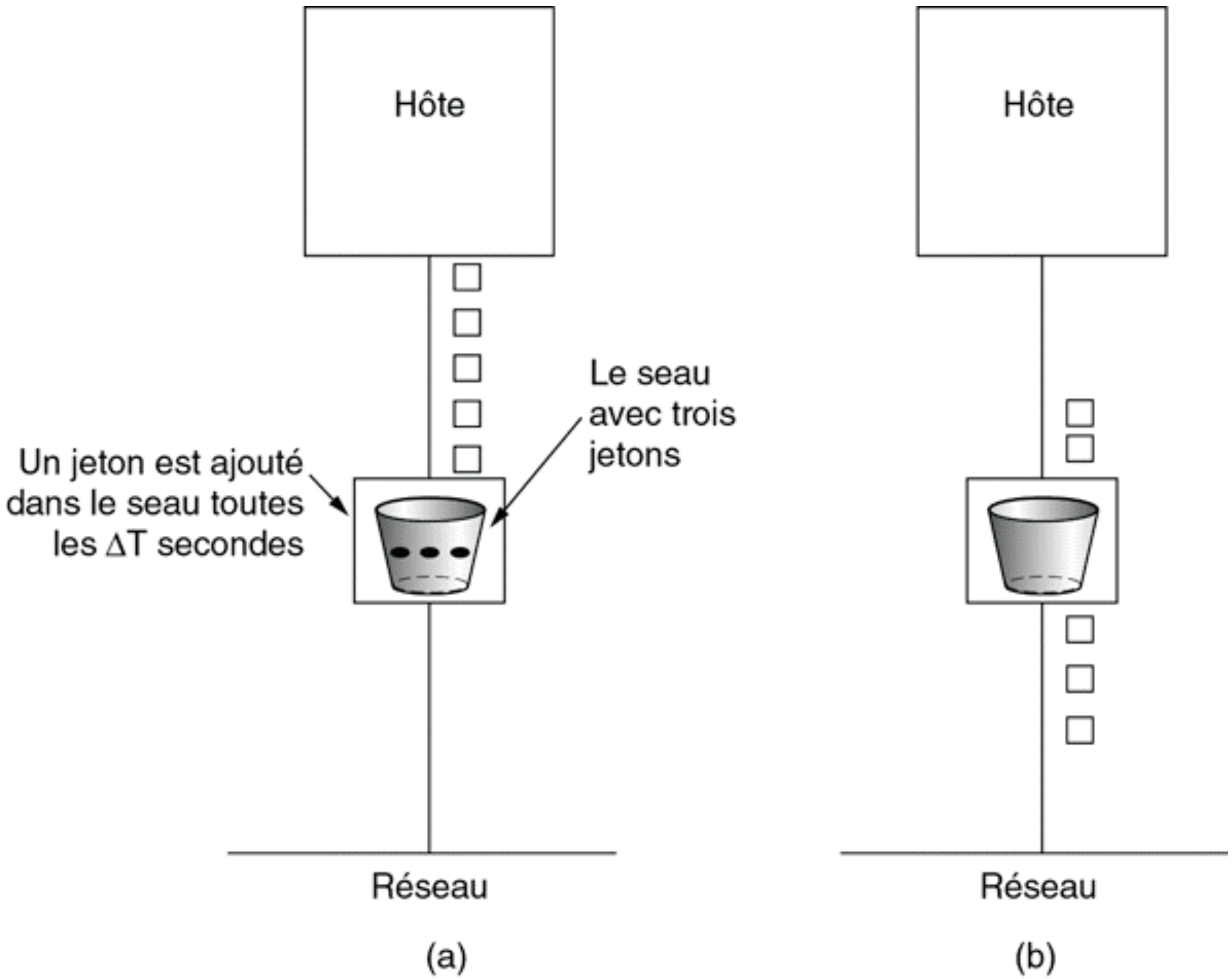
Traffic and Service Characterization

- **To quantify a service one has two know**
  – Flow’s traffic arrival
  – Service provided by the router, i.e., resources reserved at each router
- **Examples:**
  – Traffic characterization: token bucket
  – Service provided by router: fix rate and fix buffer space
    - Characterized by a service model (service curve framework)

Régulation du trafic: Leacky Bucket



Régulation du trafic: Token Bucket



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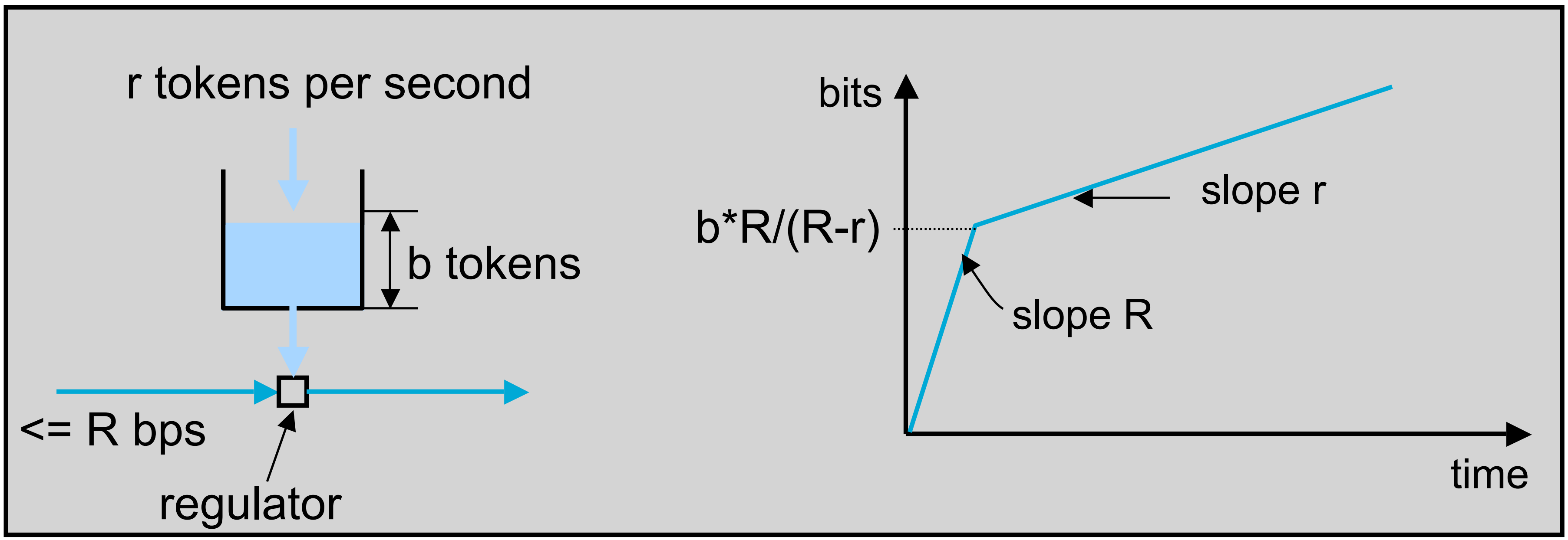
Token Bucket

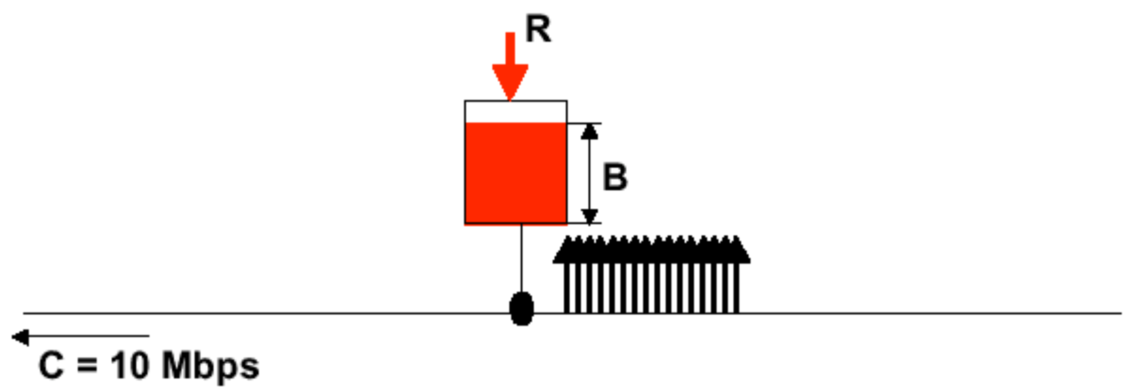
- **Caractérisé par 3 paramètres (b, r, R)**
  - b: capacité en jetons
  - r: taux de génération des tokens
  - R: taux d'émission maximum (e.g., R = capacité du lien)
- **Un bit est transmis s'il y a un jeton**
  - Quand un bit est transmis, un jeton est consommé



Token Bucket

Example

- B = 4000 bits, R = 1 Mbps, C = 10 Mbps
- Packet length = 1000 bits
- Assume the bucket is initially full and a "large" burst of packets arrives



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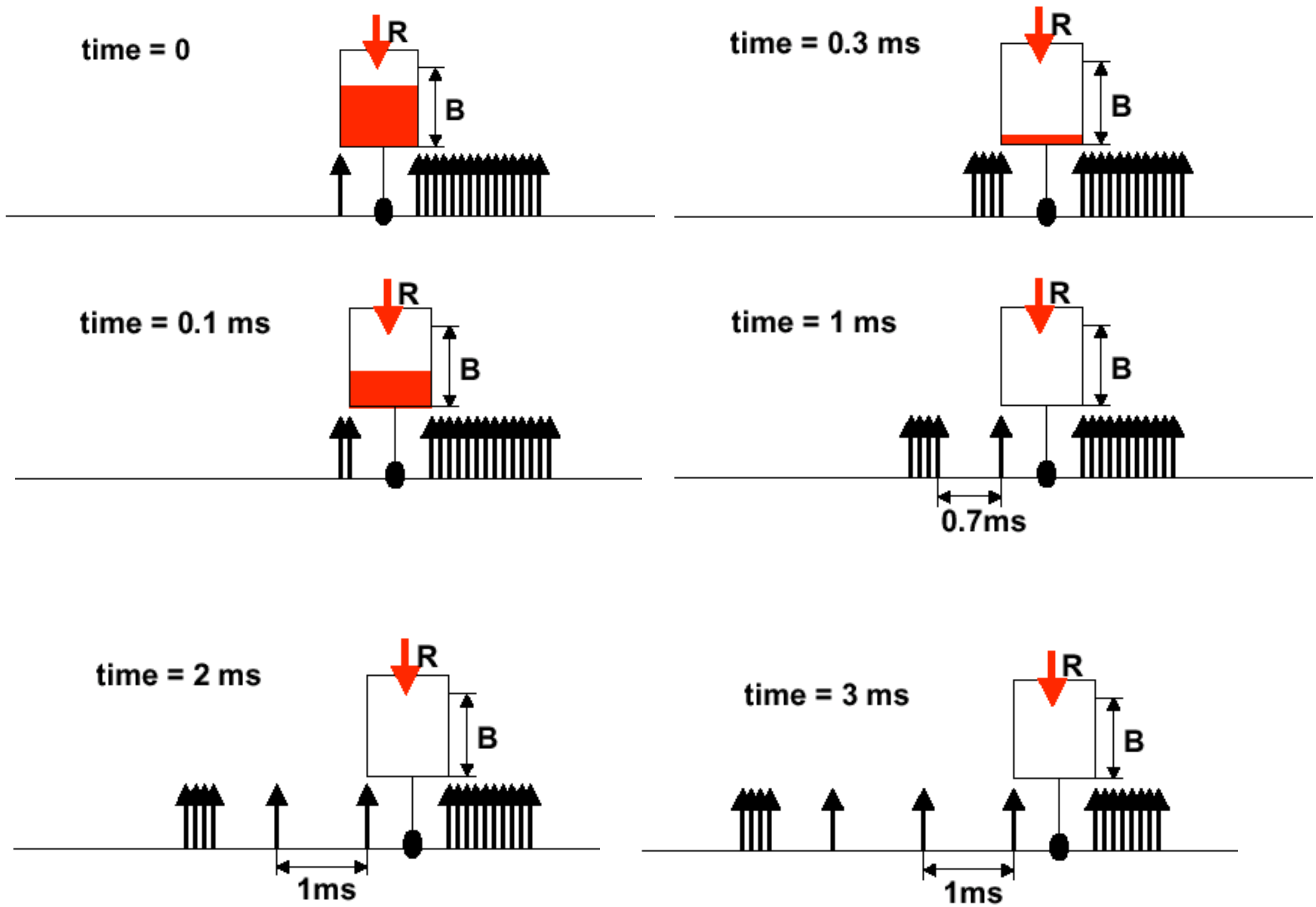
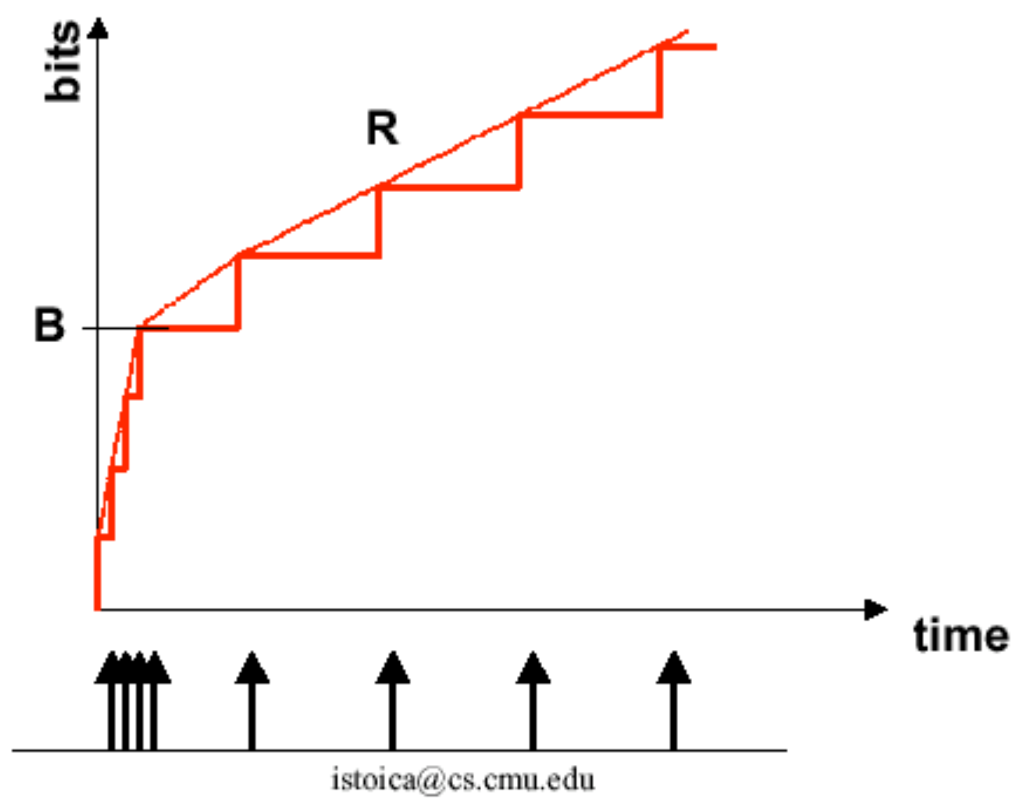
Token Bucket



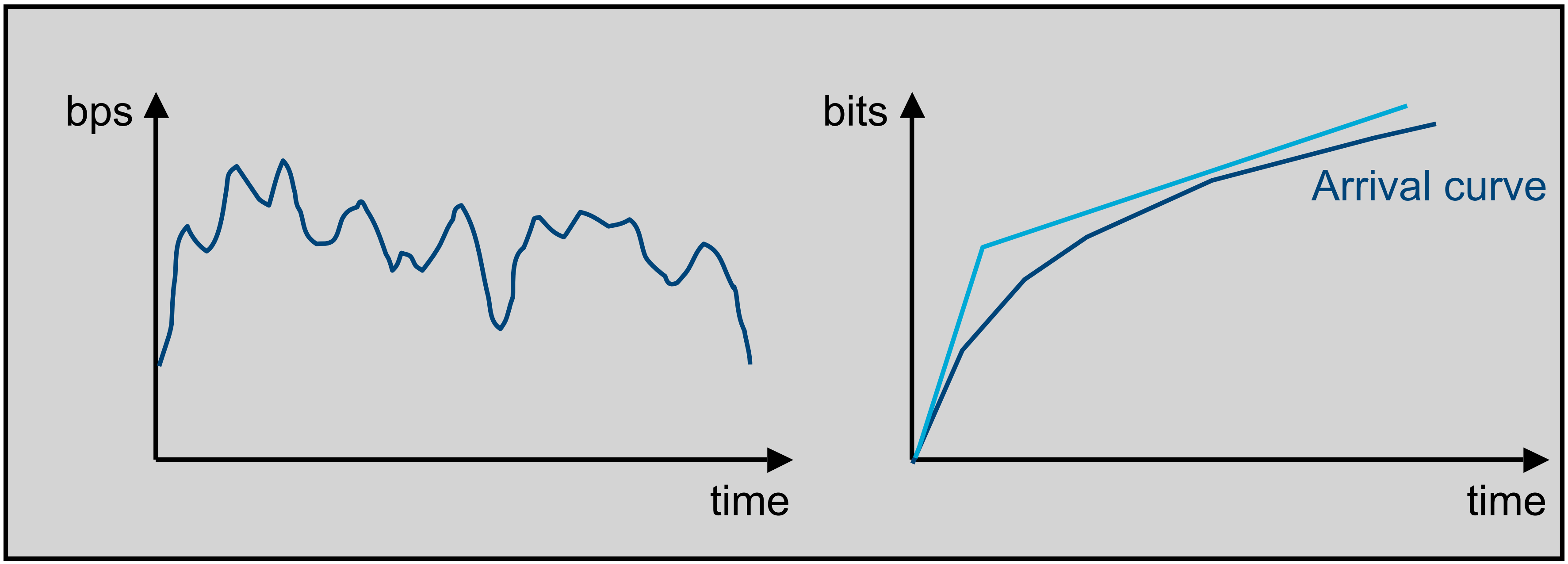
Courbe des arrivées

A(t) – number of bits received up to time t

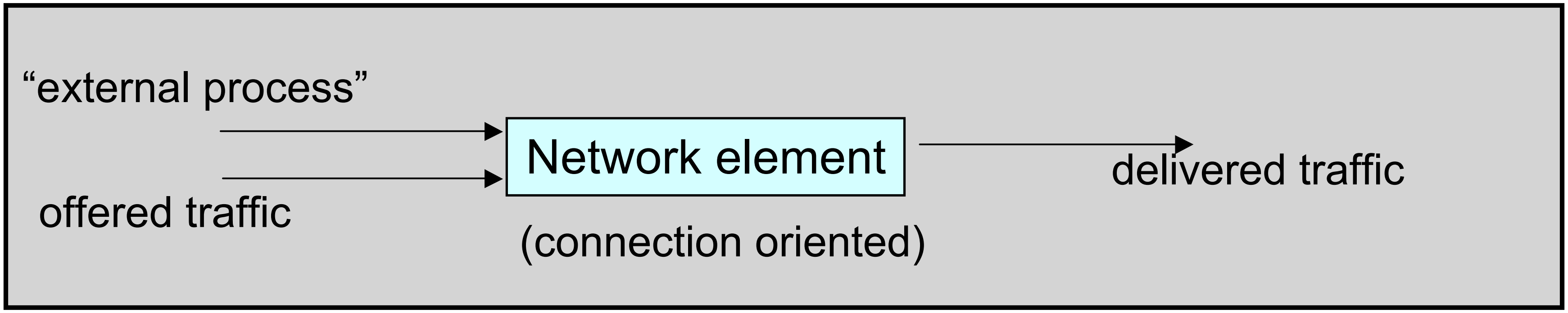


Caractérisation avec un Token Bucket

- Courbes des arrivées – nombre maximum de bits transmis pendant un temps t
- Utilise le Token Bucket pour borner le taux d'arrivée

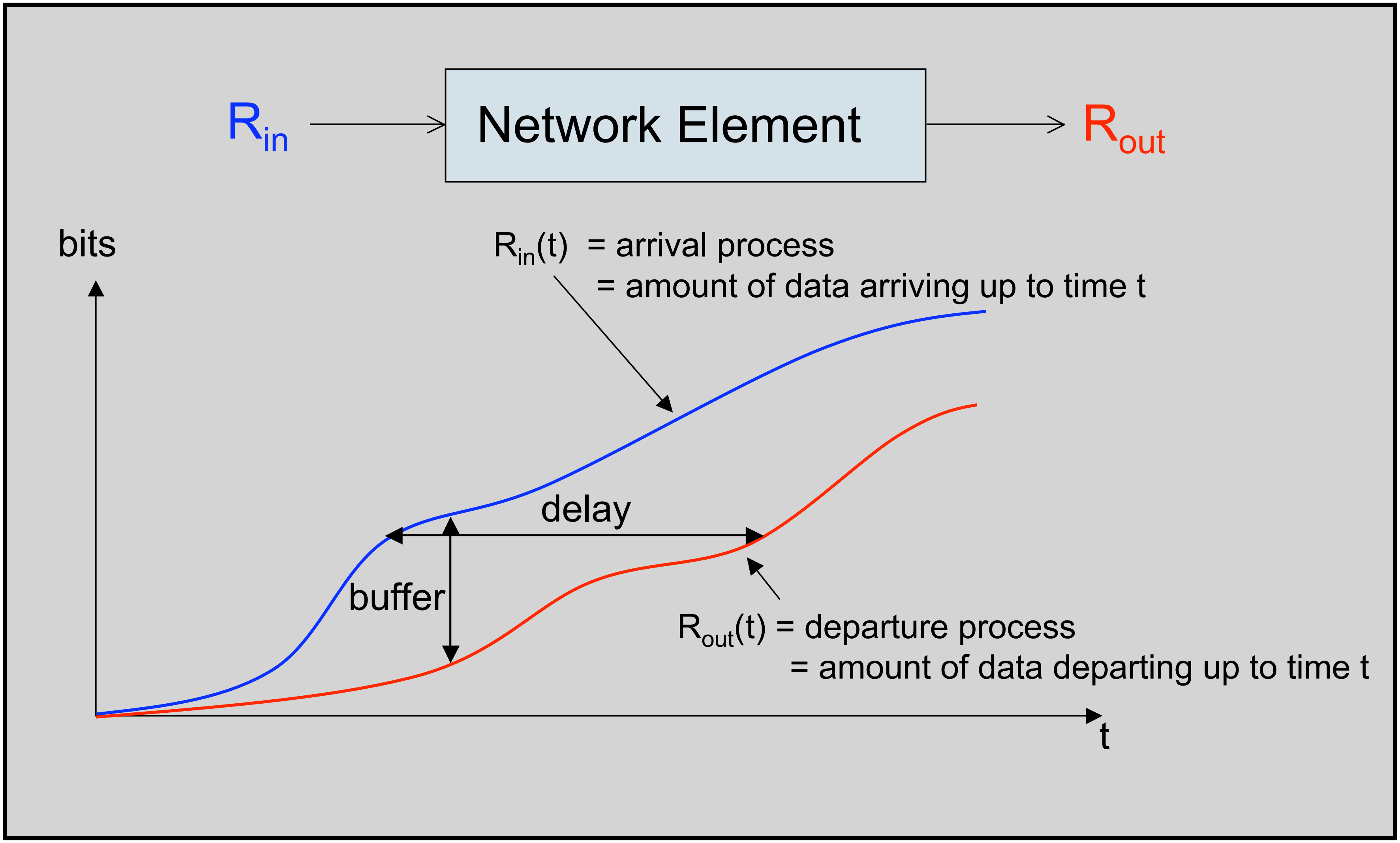


What is a Service Model?

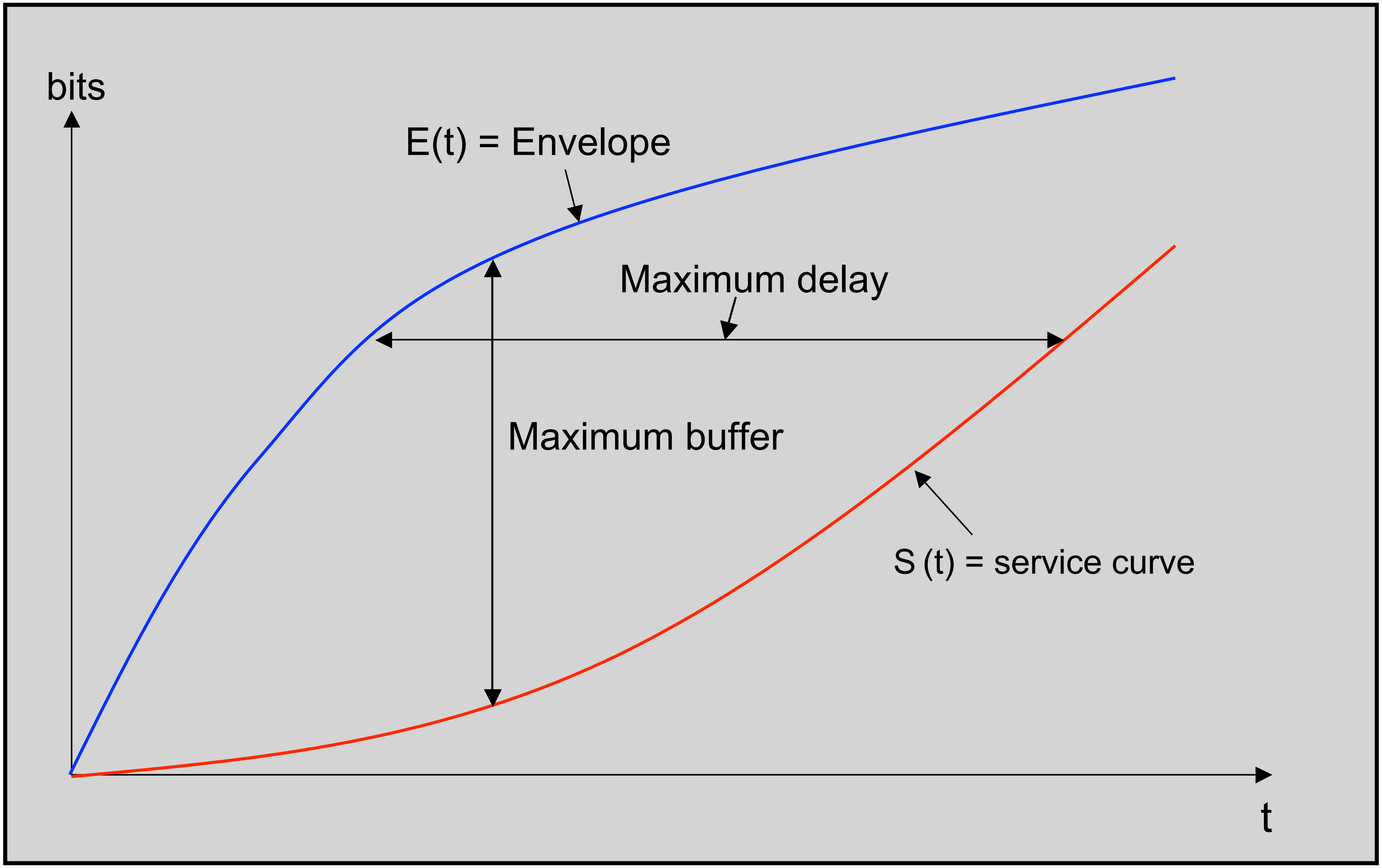


- **The QoS measures (delay,throughput, loss, cost) depend on offered traffic, and possibly other external processes.**
- **A service model attempts to characterize the relationship between offered traffic, delivered traffic, and possibly other external processes.**

Arrival and Departure Process



Delay and Buffer Bounds



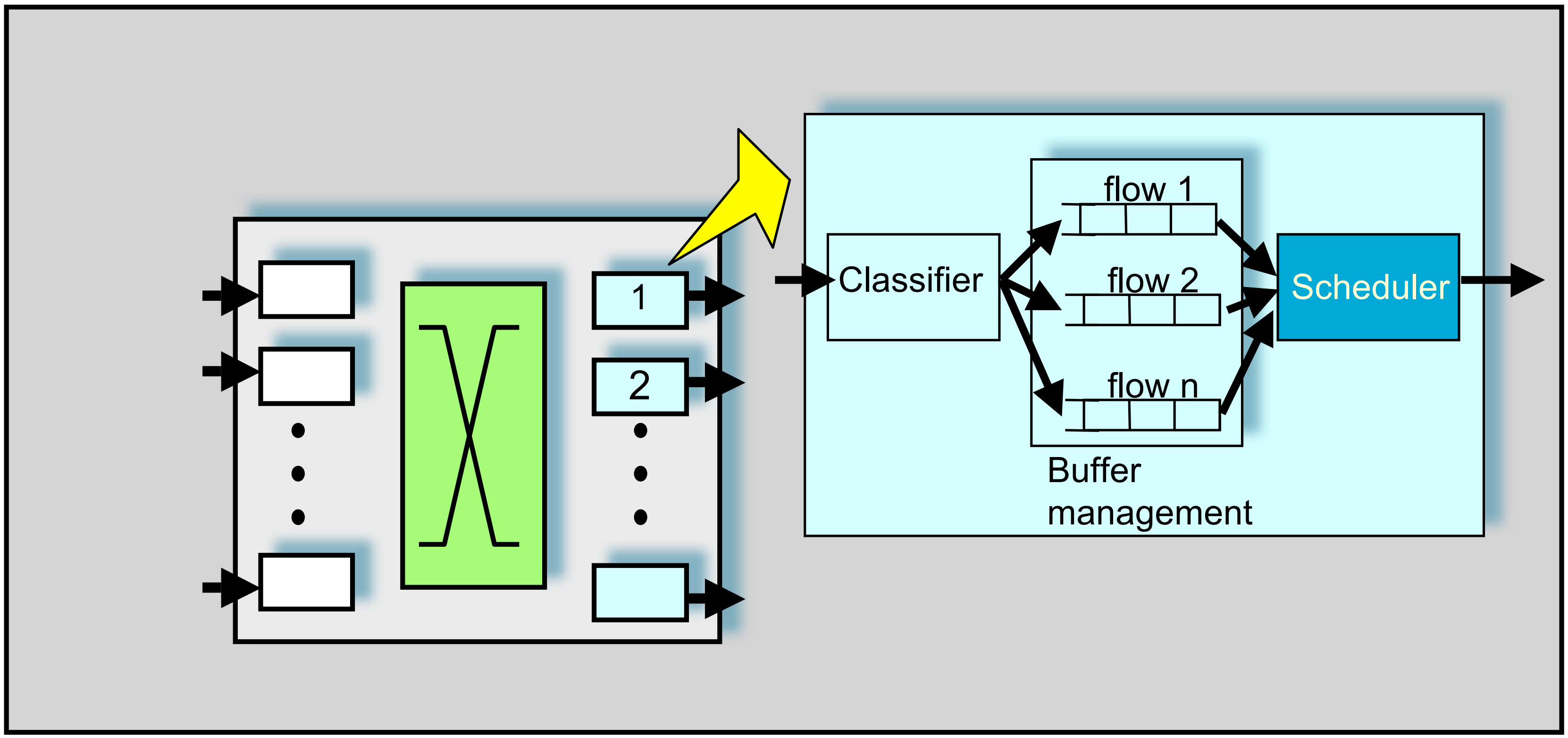
Le contrôle d'admission: principes

- **Pessimiste**
  - admission sur le débit crête D_{max},
  - pour des paquets de taille fixe, on utilise le temps entre 2 paquets,
  - pour des paquets de taille variable, on doit en plus définir un intervalle de temps pour le calcul de D_{max},
  - simple, sûr, mais gaspille de la bande passante.

- **Plus optimiste**
  - plusieurs critères en plus comme de débit moyen, le débit burst,...
  - plus complexe à mettre en oeuvre, mais exploite au mieux l'aspect statistique du trafic.
  - Ex: utilisation d'un Token Bucket (RSVP)

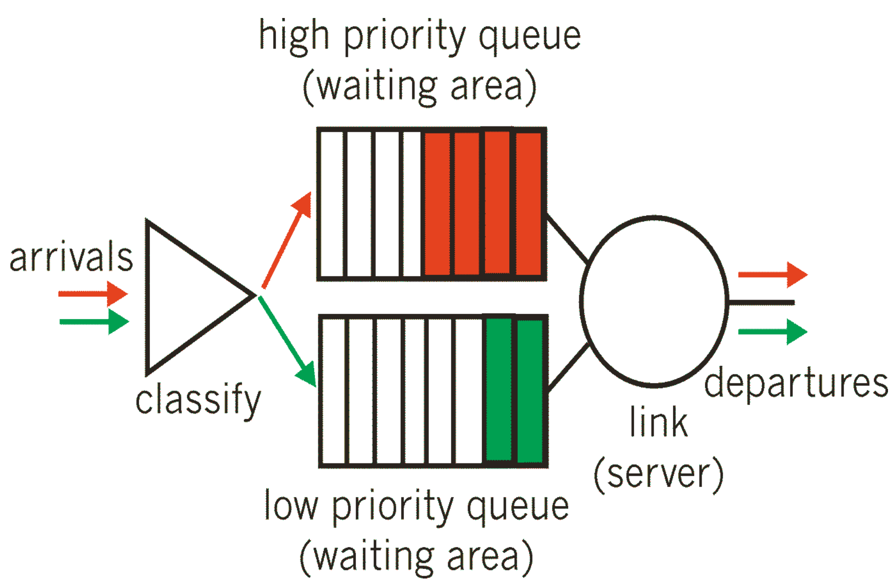
L'ordonnancement des paquets: principes

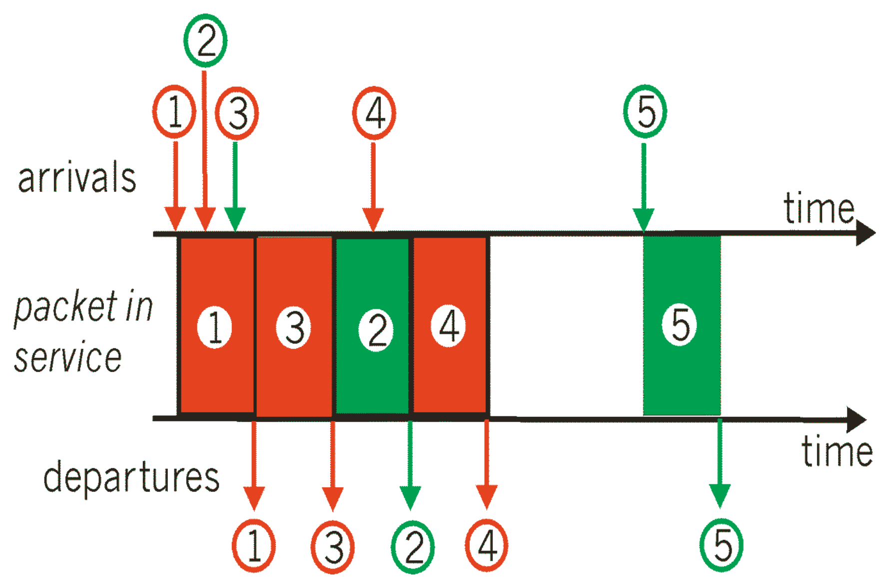
- **Décider quand et quel paquet envoyer sur la ligne**
  - Généralement réalisé sur l'interface de sortie



Politique d'ordonnancement (1)

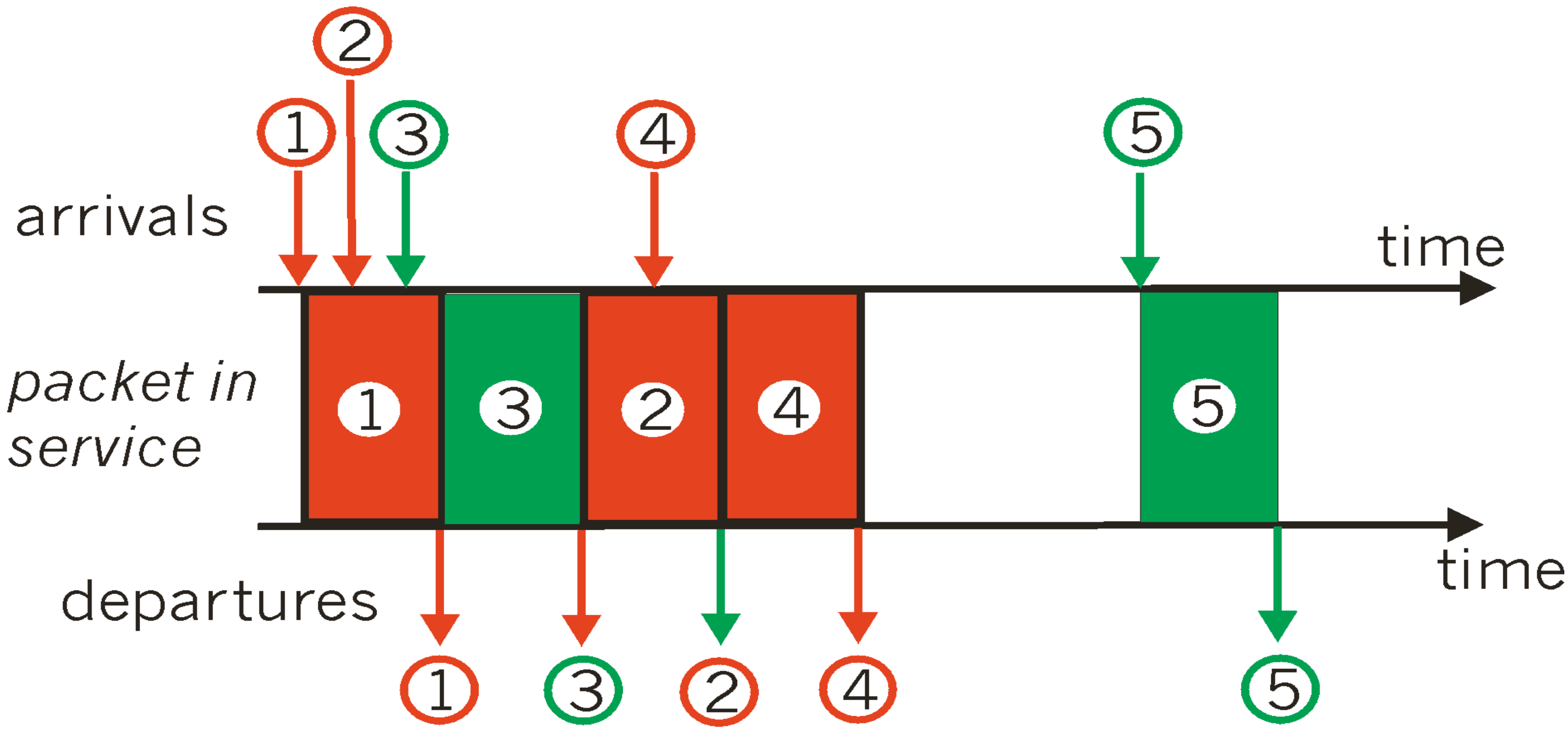
- **Priority Queuing: classes have different priorities; class may depend on explicit marking or other header info, eg IP source or destination, TCP Port numbers…**
- **Transmit a packet from the highest priority class with a non-empty queue**
- **Preemptive and non-preemptive versions**





Politique d'ordonnancement (2)

- **Round Robin: scan class queues serving one from each class that has a non-empty queue**



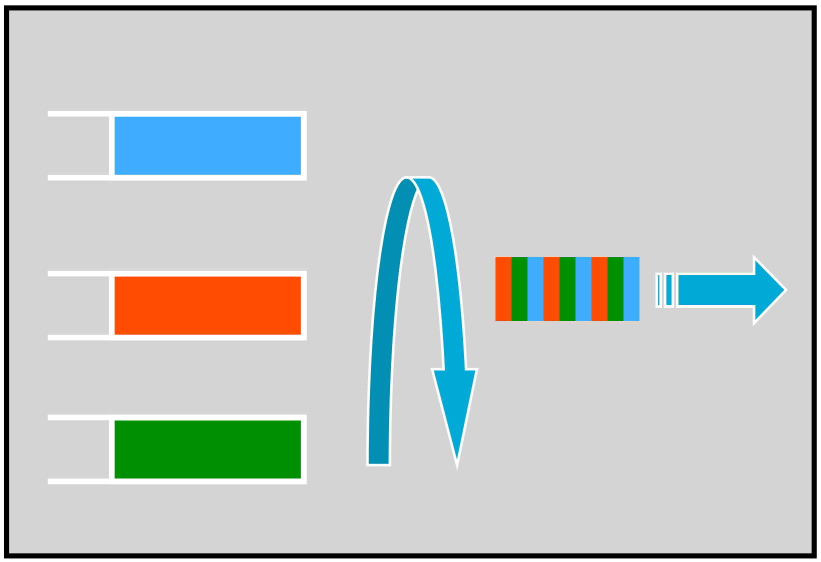
Discussion sur le Round-Robin

- **Advantages: protection among flows**
  - Misbehaving flows will not affect the performance of well-behaving flows
    - Misbehaving flow – a flow that does not implement any congestion control
  - FIFO does not have such a property

- **Disadvantages:**
  - More complex than FIFO: per flow queue/state
  - Biased toward large packets – a flow receives service proportional to the number of packets

Generalized Processor Sharing(GPS)

- **Assume a fluid model of traffic**
  - Visit each non-empty queue in turn (RR)
  - Serve infinitesimal from each
  - Leads to “max-min” fairness
- **GPS is un-implementable!**
  - We cannot serve infinitesimals, only packets



max-min fairness

Soit un ensemble de sources 1,..,n demandant des ressources $x_1,..,x_n$ avec $x_1<x_2..<x_n$ par exemple. Le serveur a une capacité C.

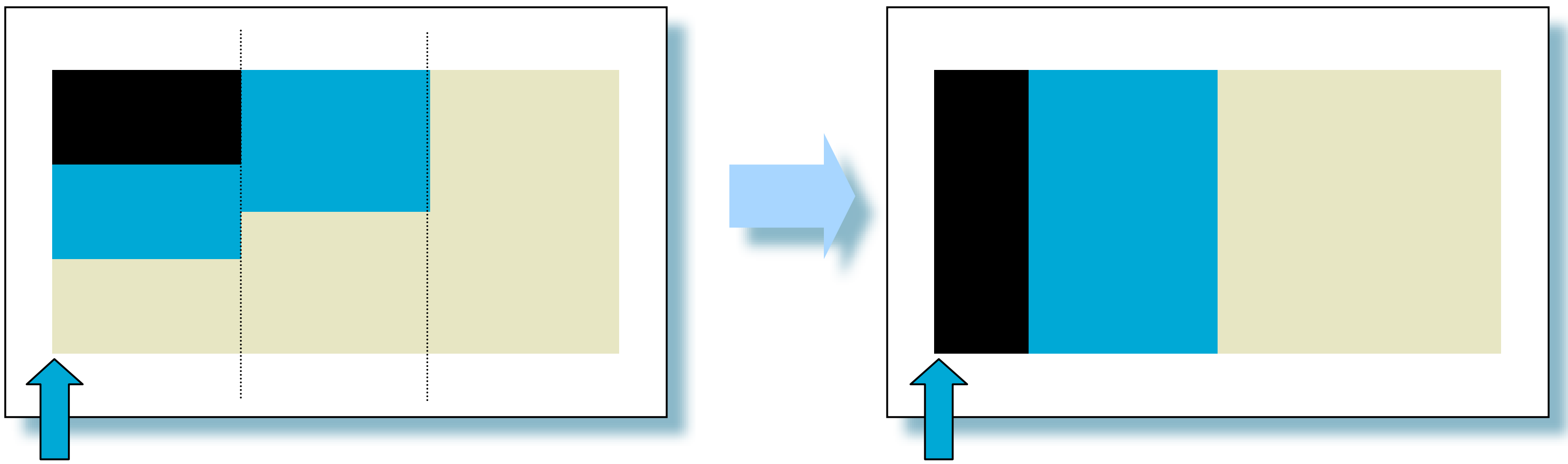
On donne alors C/n à la source 1. Si $C/n>x_1$, on donne $C/n+(C/n-x_1)/(n-1)$ aux (n-1) sources restantes. Si cela est supérieur à x_2, on recommence.

Packet Approximation of Fluid System

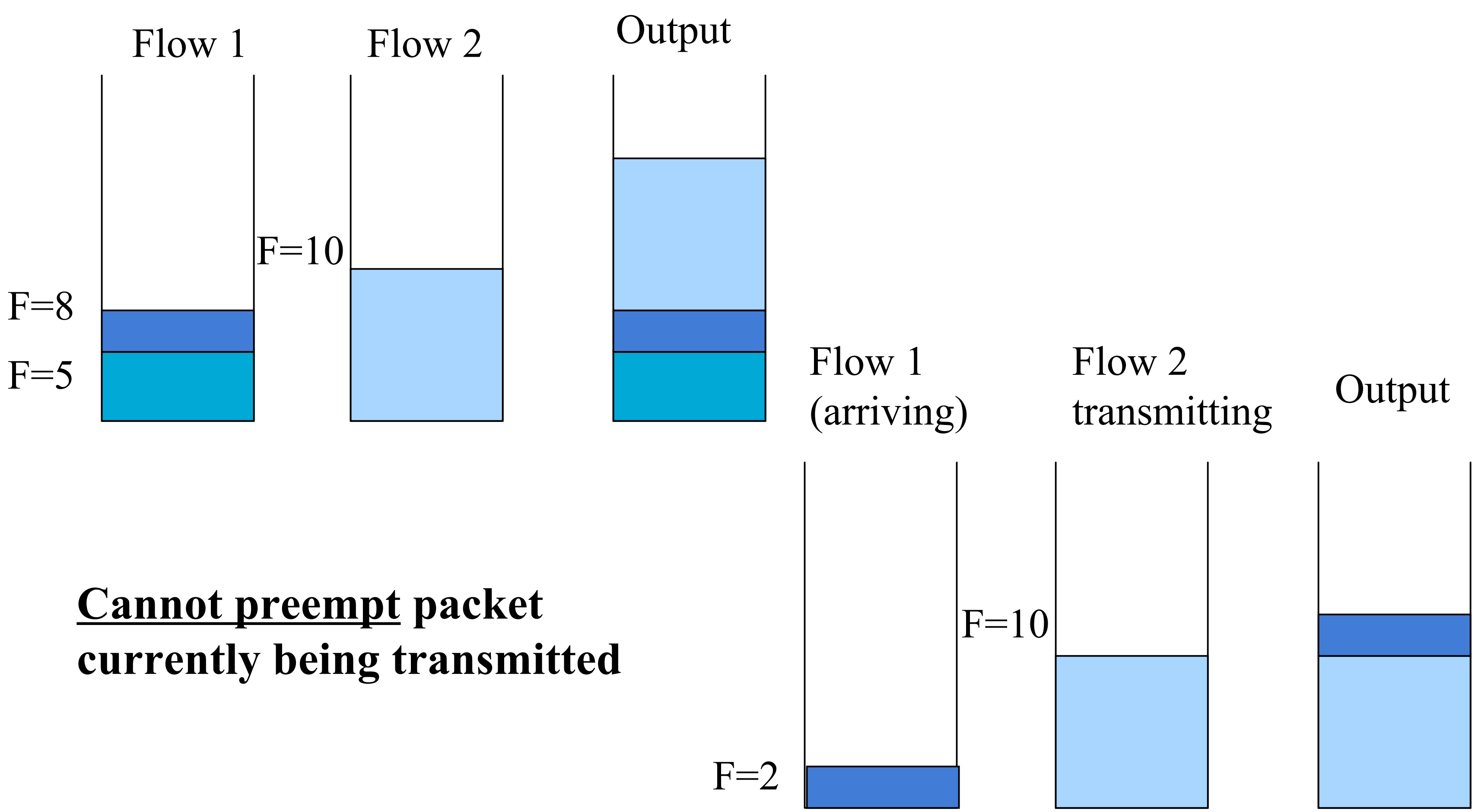
- **GPS un-implementable**
- **Standard techniques of approximating fluid GPS**
  - Select packet that finishes first in GPS assuming that there are no future arrivals (emulate GPS on the side)
- **Important properties of GPS**
  - Finishing order of packets currently in system independent of future arrivals
- **Implementation based on virtual time**
  - Assign virtual finish time to each packet upon arrival
  - Packets served in increasing order of virtual times

Fair Queuing (FQ)

- Idea: serve packets in the order in which they would have finished transmission in the fluid flow system
- Mapping bit-by-bit schedule onto packet transmission schedule
- Transmit packet with the lowest finish time at any given time



FQ Simple Example



<u>Cannot preempt</u> packet currently being transmitted

Round Number and Finish Number

- **Single flow: *clock ticks when a bit is transmitted.* For packet k:**
  - P_k = length, A_k = arrival time, S_k = begin transmit time, F_k = finish transmit time
  - $F_k = S_k + P_k = \max(F_{k-1}, A_k) + P_k$
- **Multiple flows: *clock ticks when a bit from all active flows is transmitted* → round number**
  - Can calculate F_k for each packet if number of flows is known at all times
    - F_k = current round number + size of packet k, inactive case
    - F_k = largest F_k in the queue + size of packet k, active case
  - $F_{i,k,t} = \max(F_{i,k-1,t}, R_t) + P_{i,k,t}$
  - In packet approximation, finish number indicate a relative order (service tag) in which a packet is to be served. finish time≠finish number

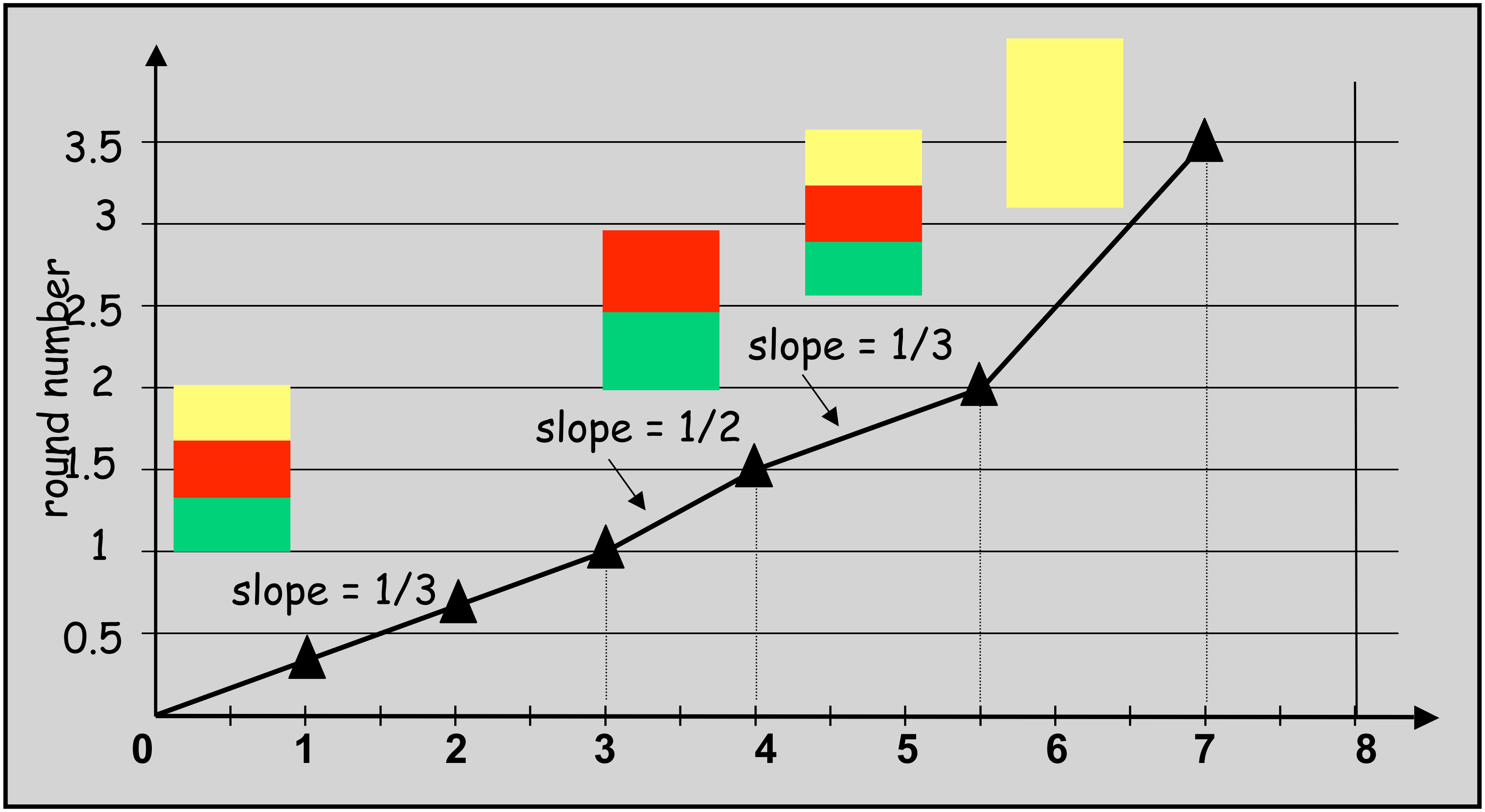
Example

- **The round number increases at a rate inversely proportional to the number of active connections**
  - Thus is only used for computing finish numbers

- **Largest finish number in a connection's queue is the connection's finish number**

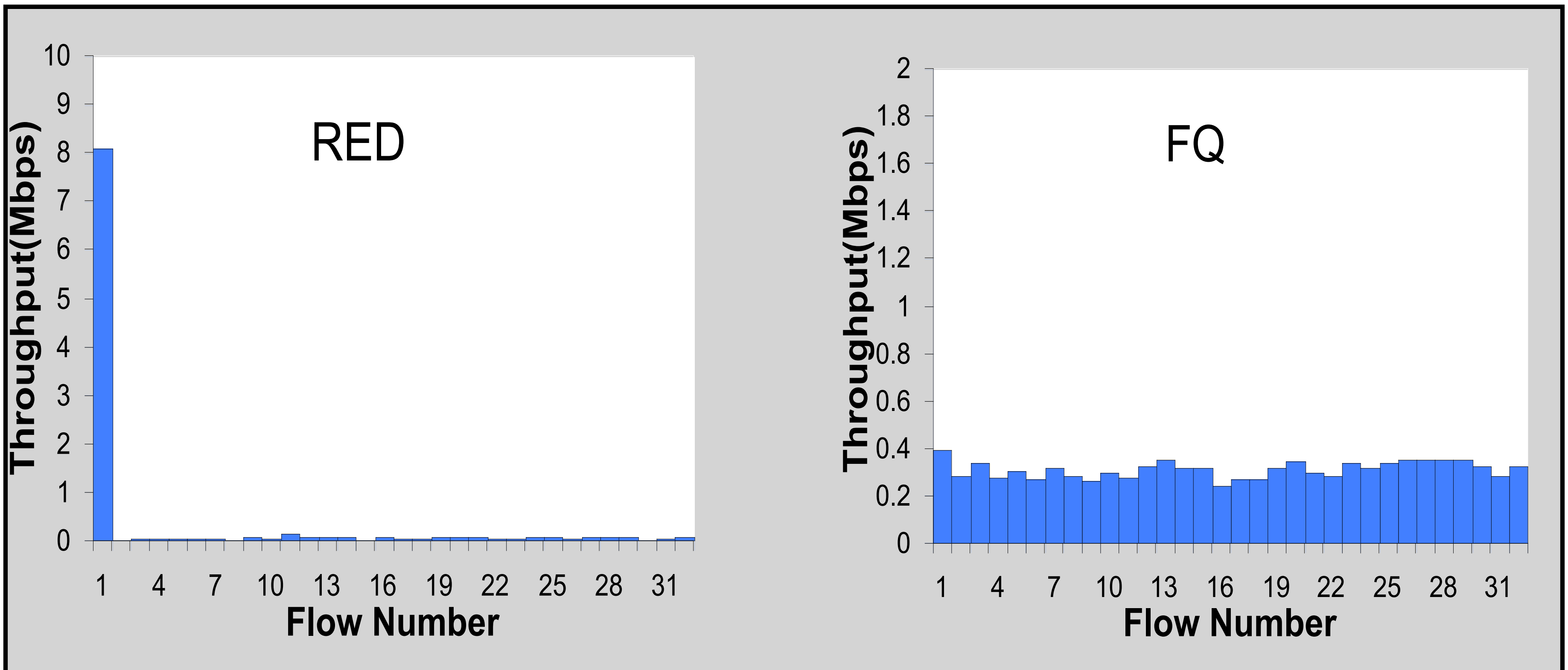
- **Example**
  - Suppose packets of size 1, 2 and 2 units arrive at a FQ scheduler at time for connection A, B and C. Also, assume that a packet of size 2 arrive for connection A at time 4. The link service rate is 1 unit/s. Compute the finish number of all packets.

Illustration



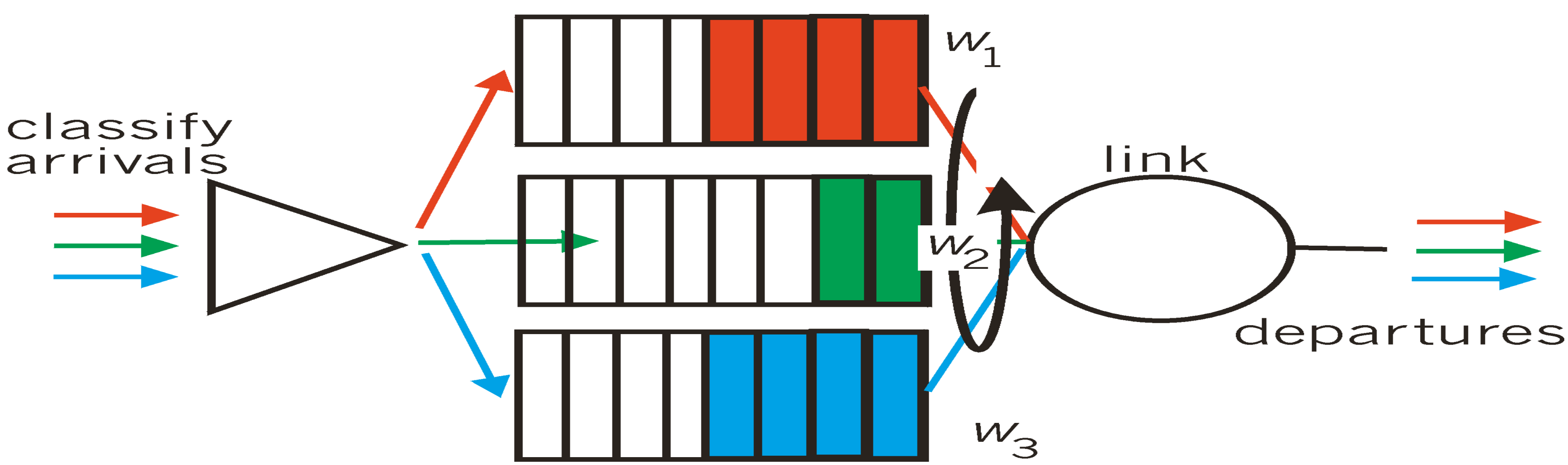
FQ Advantages

- **FQ protect well-behaved flows from ill-behaved flows**
- **Example: 1 UDP (10 Mbps) and 31 TCP's sharing a 10 Mbps link**



Weighted Fair Queueing

- Variation of FQ: Weighted Fair Queuing (WFQ)
- Weighted Fair Queuing: is a generalized Round Robin in which an attempt is made to provide a class with a differentiated amount of service over a given period of time



Implementing WFQ

- **WFQ needs per-connection (or per-aggregate) scheduler state→implementation complexity.**
  - complex iterated deletion algorithm
  - complex sorting at the output queue on the service tag
- **WFQ needs to know the weight assigned for each queue →manual configuration, signalling.**
- **WFQ is not perfect…**
- **Router manufacturers have implemented as early as 1996 WFQ in their products**
  - from CISCO 1600 series
  - Fore System ATM switches

Big Picture

- **FQ does not eliminate congestion → it just manages the congestion**
- **You need both end-host congestion control and router support for congestion control**
  - end-host congestion control to adapt
  - router congestion control to protect/isolate
- **Don’t forget buffer management: you still need to drop in case of congestion. Which packet’s would you drop in FQ?**
  - one possibility: packet from the longest queue

Further readings

- See http://www.cnaf.infn.it/~ferrari/ispn.html for Quality of Service list of papers
- See http://www.cnaf.infn.it/~ferrari/sched.html for scheduling list of papers

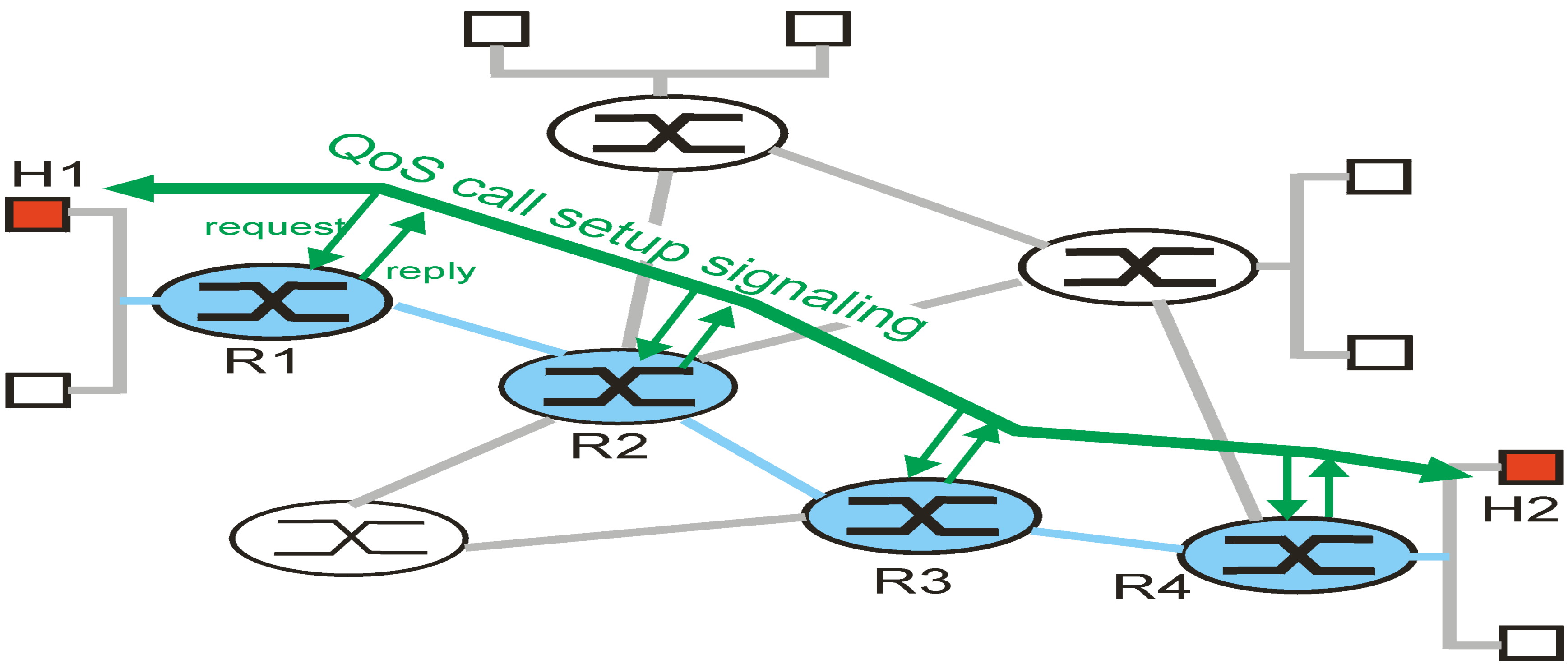
QoS ARCHITECTURES

Stateless vs. Stateful QoS Solutions

- **Stateless solutions – routers maintain no fine grained state about traffic**
  - ↑ scalable, robust
  - ↓ weak services
- **Stateful solutions – routers maintain per-flow state**
  - ↑ powerful services
    - guaranteed services + high resource utilization
    - fine grained differentiation
    - protection
  - ↓ much less scalable and robust

Integrated Services (IntServ)

- An architecture for providing QOS guarantees in IP networks for individual application sessions
- Relies on resource reservation, and routers need to maintain state information of allocated resources (eg: g) and respond to new Call setup requests



Integrated Services Model

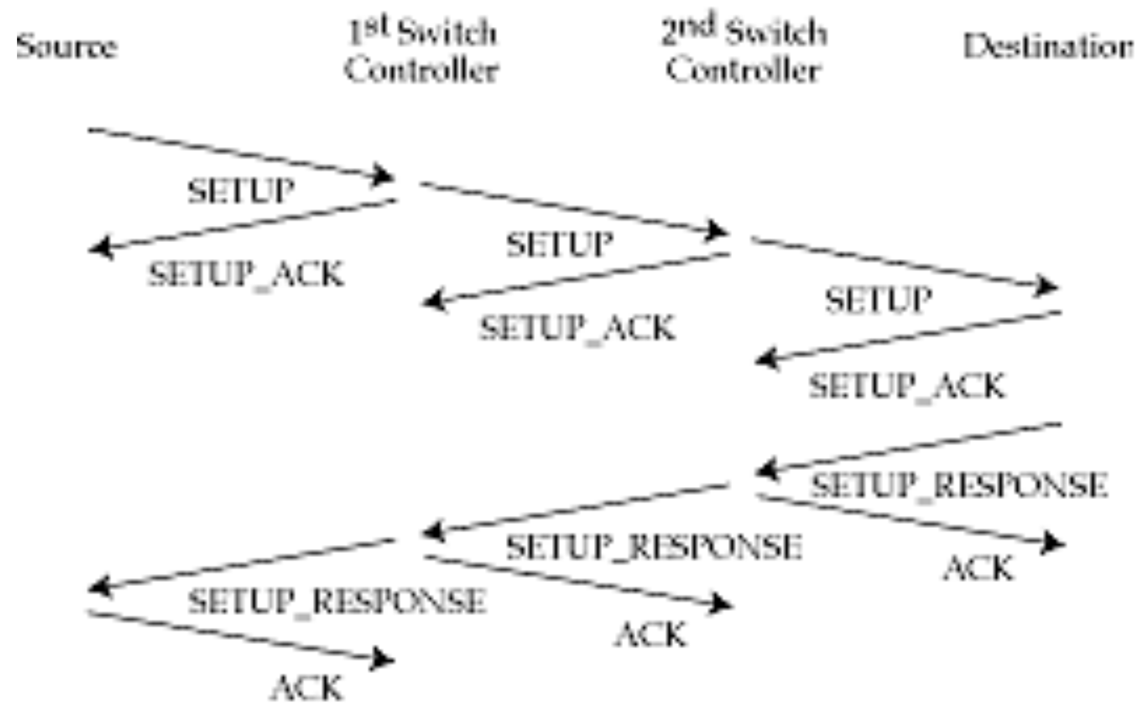
- **Flow specification**
  – Leacky Bucket, Token Bucket
- **Routing**
- **Admission control**
- **Policy control**
- **Resource reservation**
  – RSVP
- **Packet scheduling**
  – WFQ, CBQ, RED

Integrated Services: Classes

- Guaranteed QOS: **this class is provided with <u>firm bounds</u> on queuing delay at a router; envisioned for hard real-time applications that are highly sensitive to end-to-end delay expectation and variance**
- Controlled Load: **this class is provided a QOS <u>closely approximating</u> that provided by an unloaded router; envisioned for today's IP network real-time applications which perform well in an unloaded network**

Signaling semantics

- Classic scheme: sender initiated
- SETUP, SETUP_ACK, SETUP_RESPONSE
- Admission control
- Tentative resource reservation and confirmation
- Simplex and duplex setup; no multicast support



RSVP for the IntServ approach

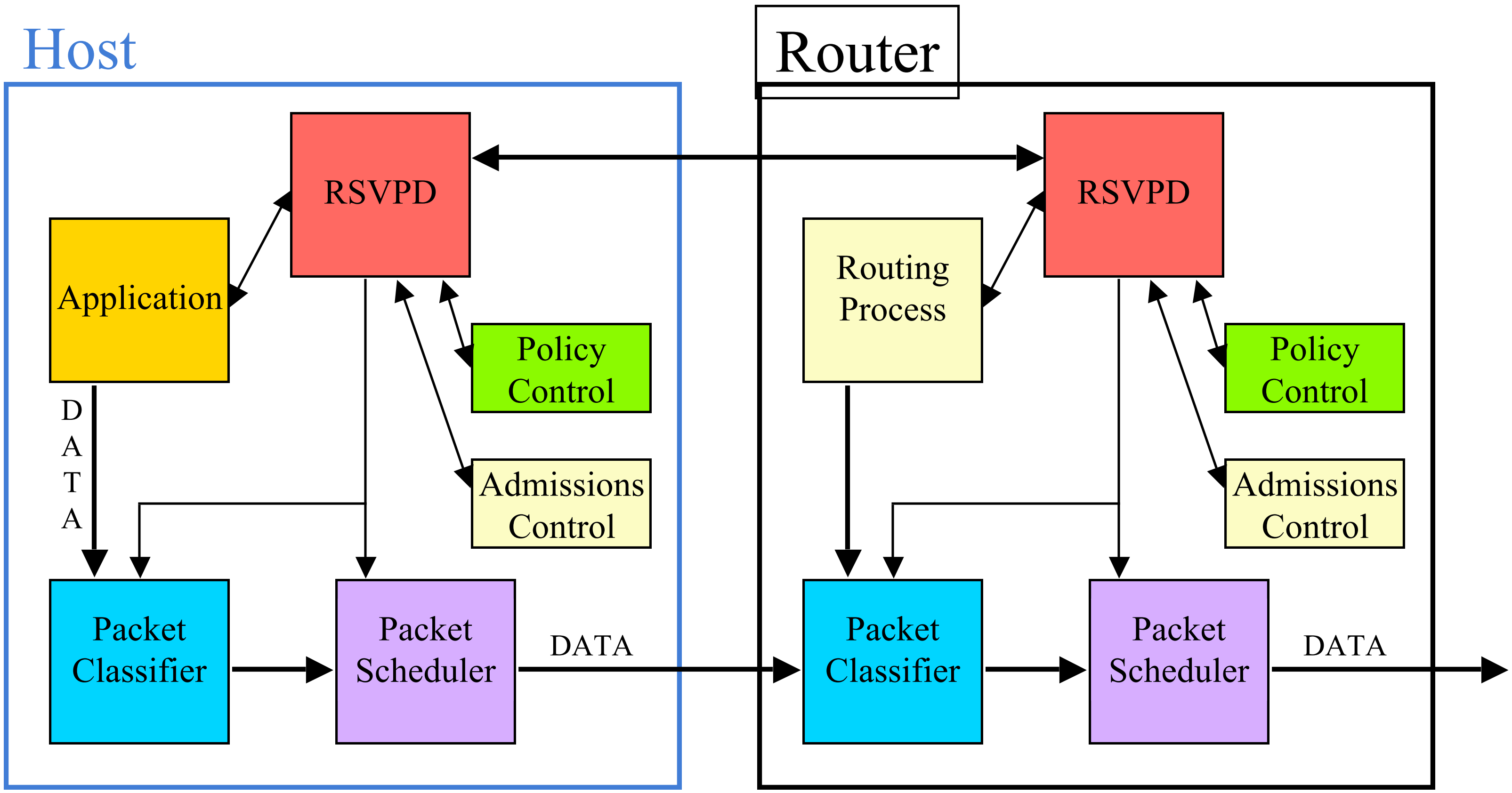
- **Resource reSerVation Protocol**
- **What is RSVP?**
  - Method for application to specify desired QoS to net
  - Switch state establishment protocol (signaling)
  - Multicast friendly, receiver-oriented
  - Simplex reservations (single direction)
- **Why run RSVP?**
  - Allows precise allocation of network resources
  - Guarantees on quality of service
  - Heterogeneous bandwidth support for multicast
  - Scalable (?)

source Gordon Schaffee

Resource Reservation

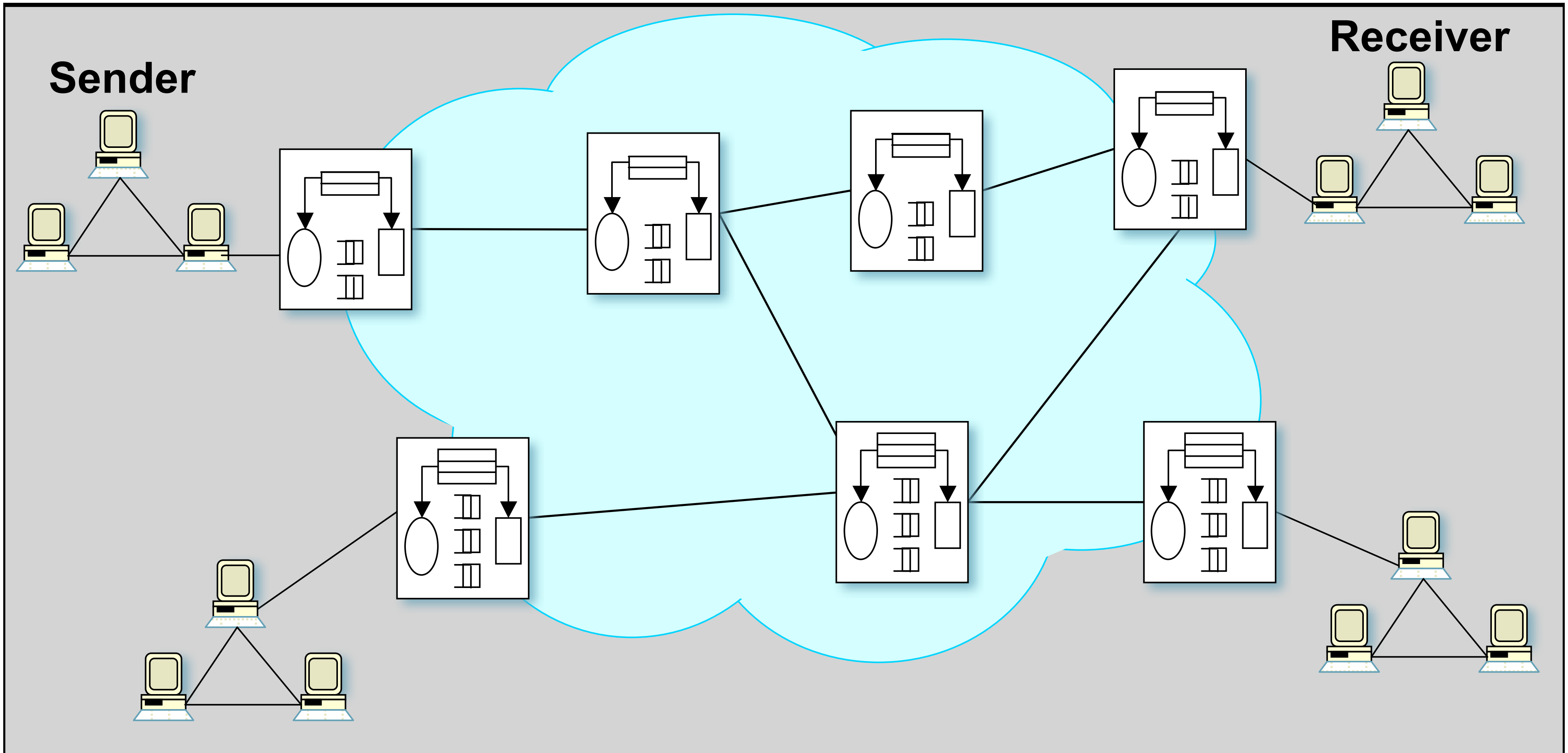
- **Senders advertise using PATH message**
- **Receivers reserve using RESV message**
  - Flowspec + filterspec + policy data
  - Travels upstream in reverse direction of Path message
- **Merging of reservations**
- **Sender/receiver notified of changes**

RSVP Functional Diagram



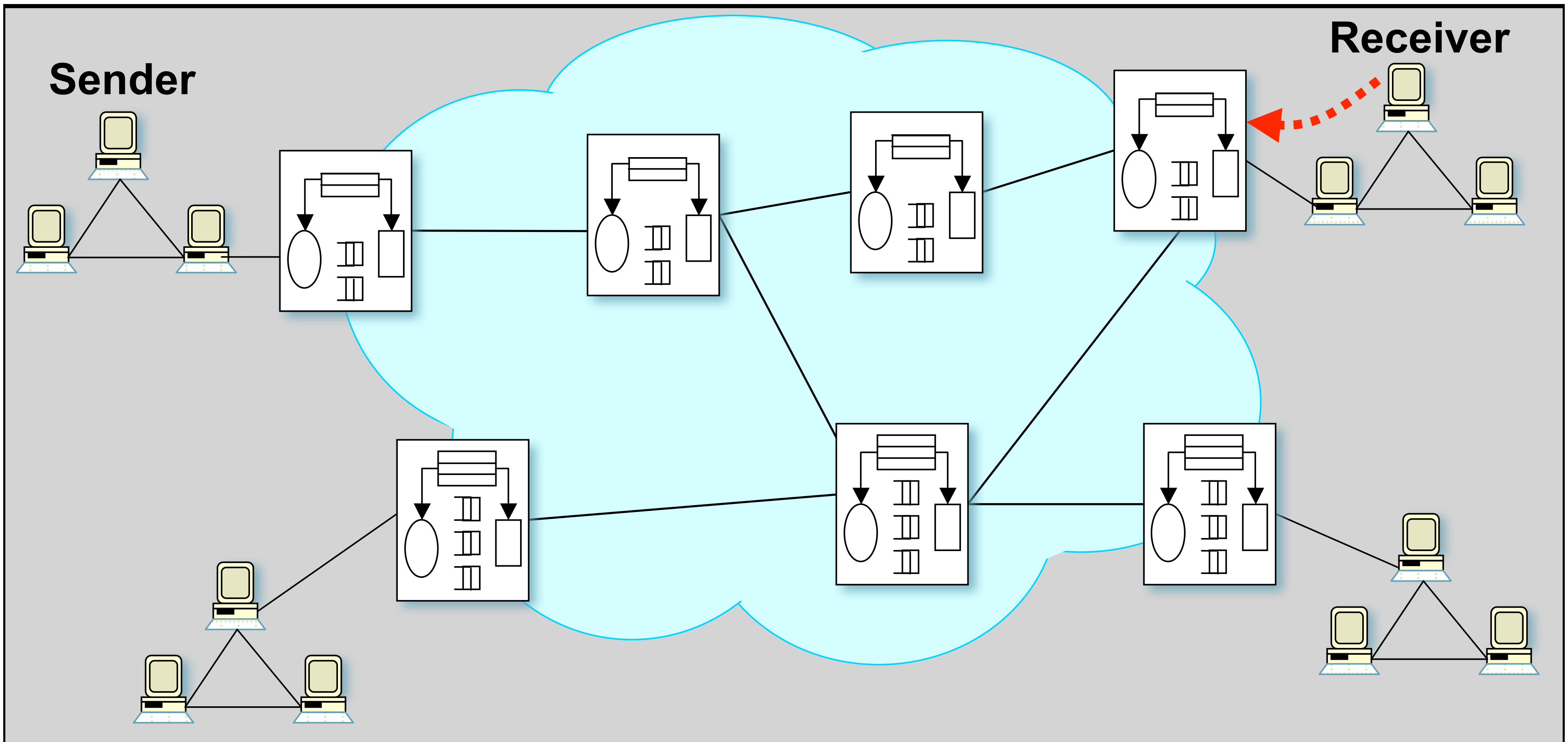
Stateful Solution: Guaranteed Services

- **Achieve per-flow bandwidth and delay guarantees**
  - Example: guarantee 1MBps and < 100 ms delay to a flow



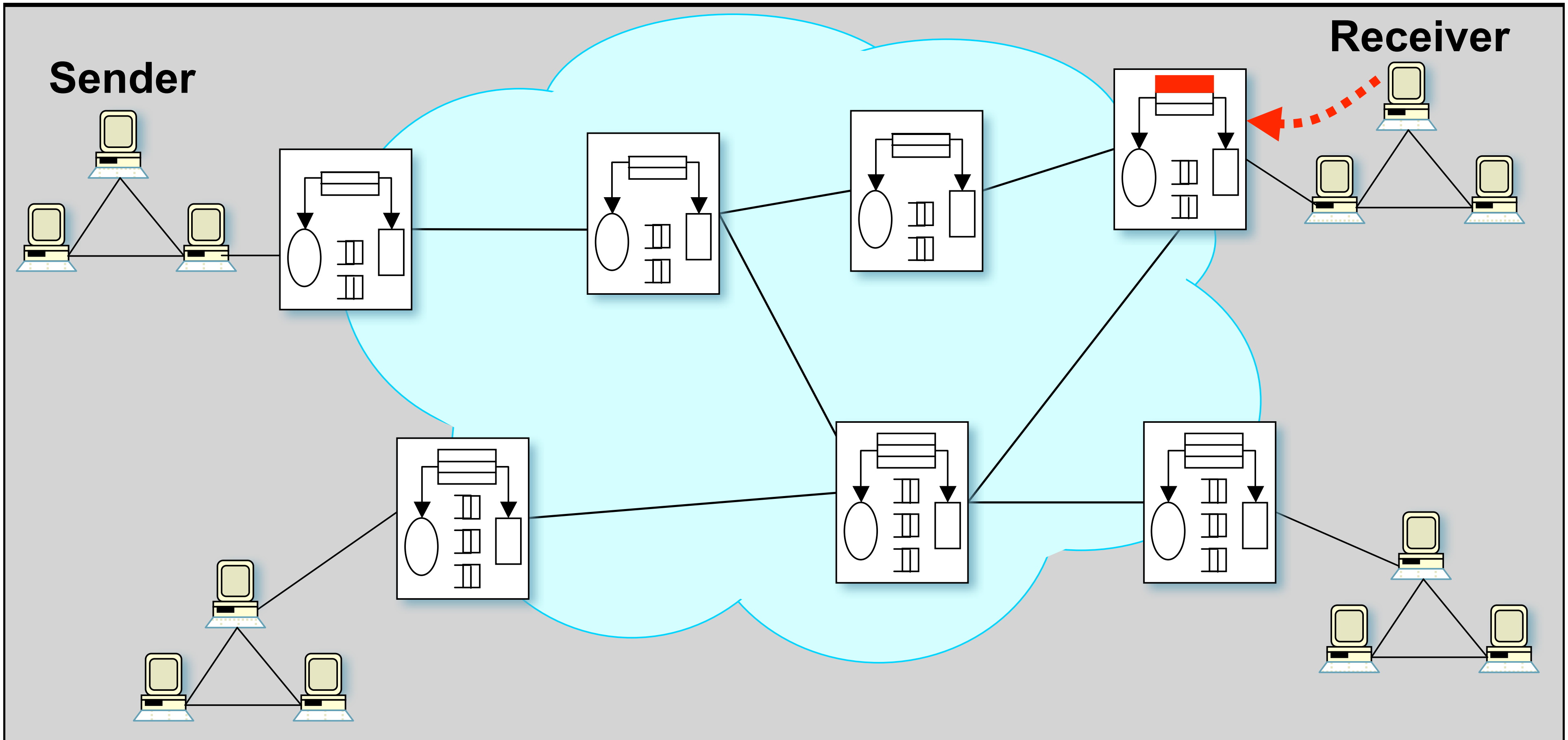
Stateful Solution: Guaranteed Services

- Allocate resources - perform per-flow admission control



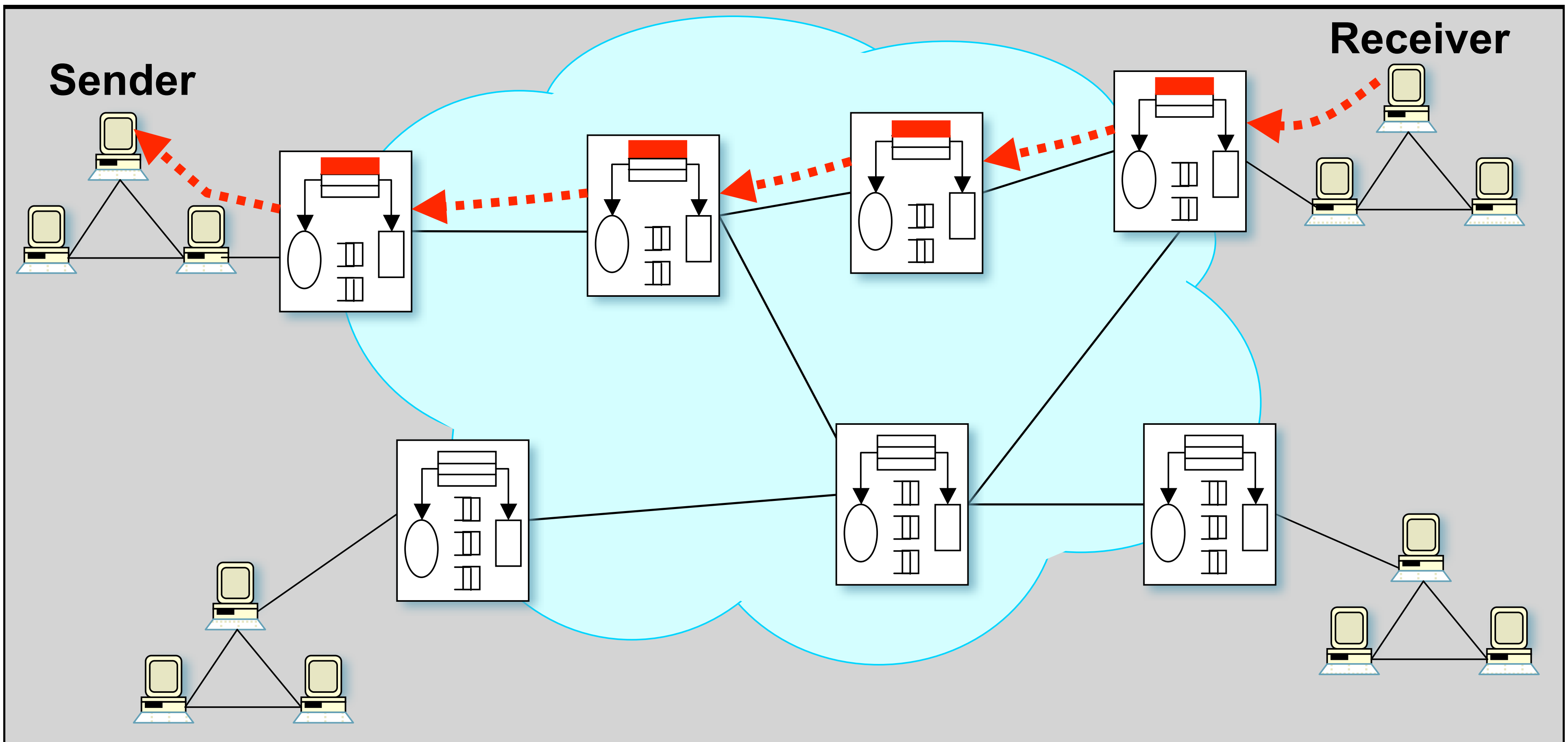
Stateful Solution: Guaranteed Services

- Install per-flow state



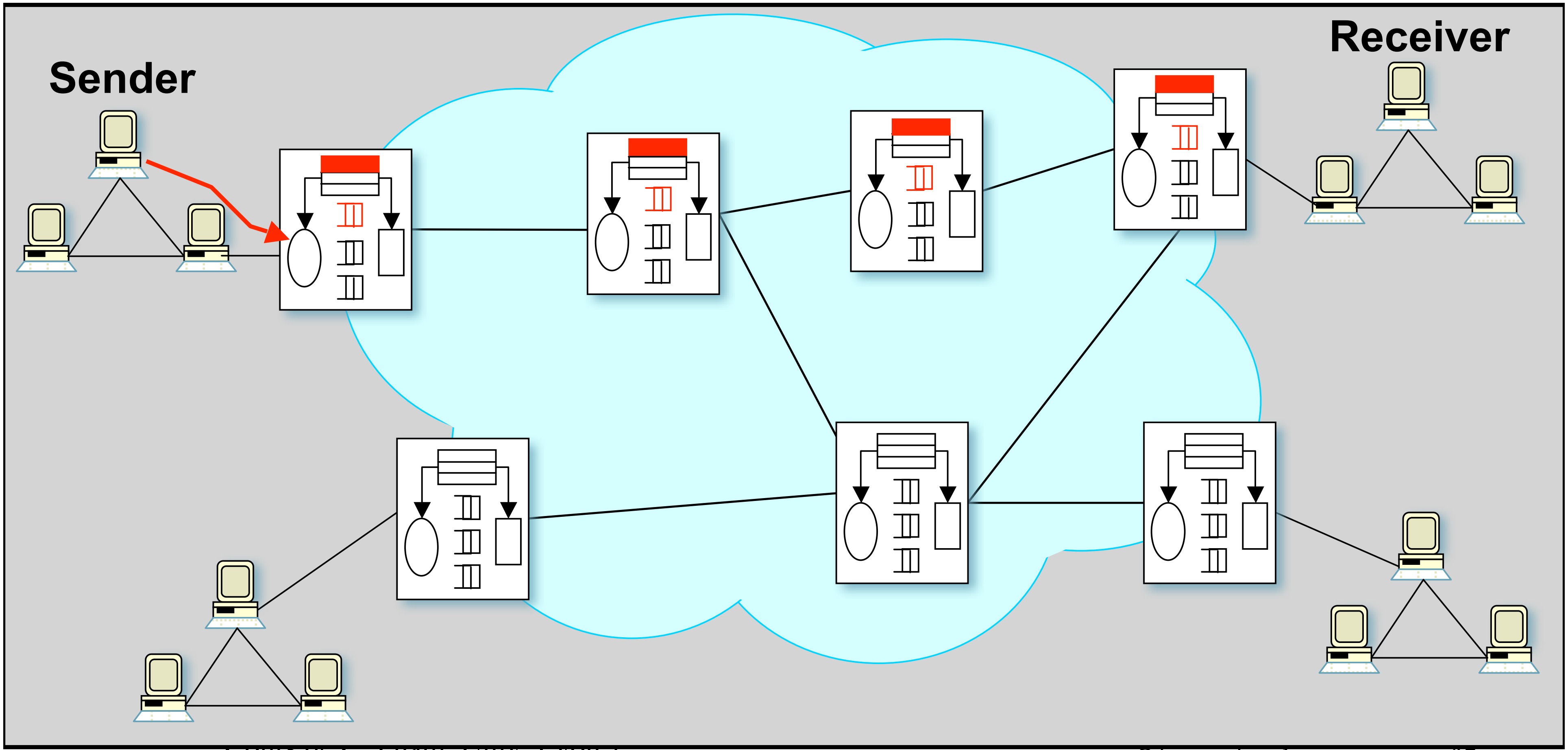
Stateful Solution: Guaranteed Services

- Challenge: maintain per-flow state consistent



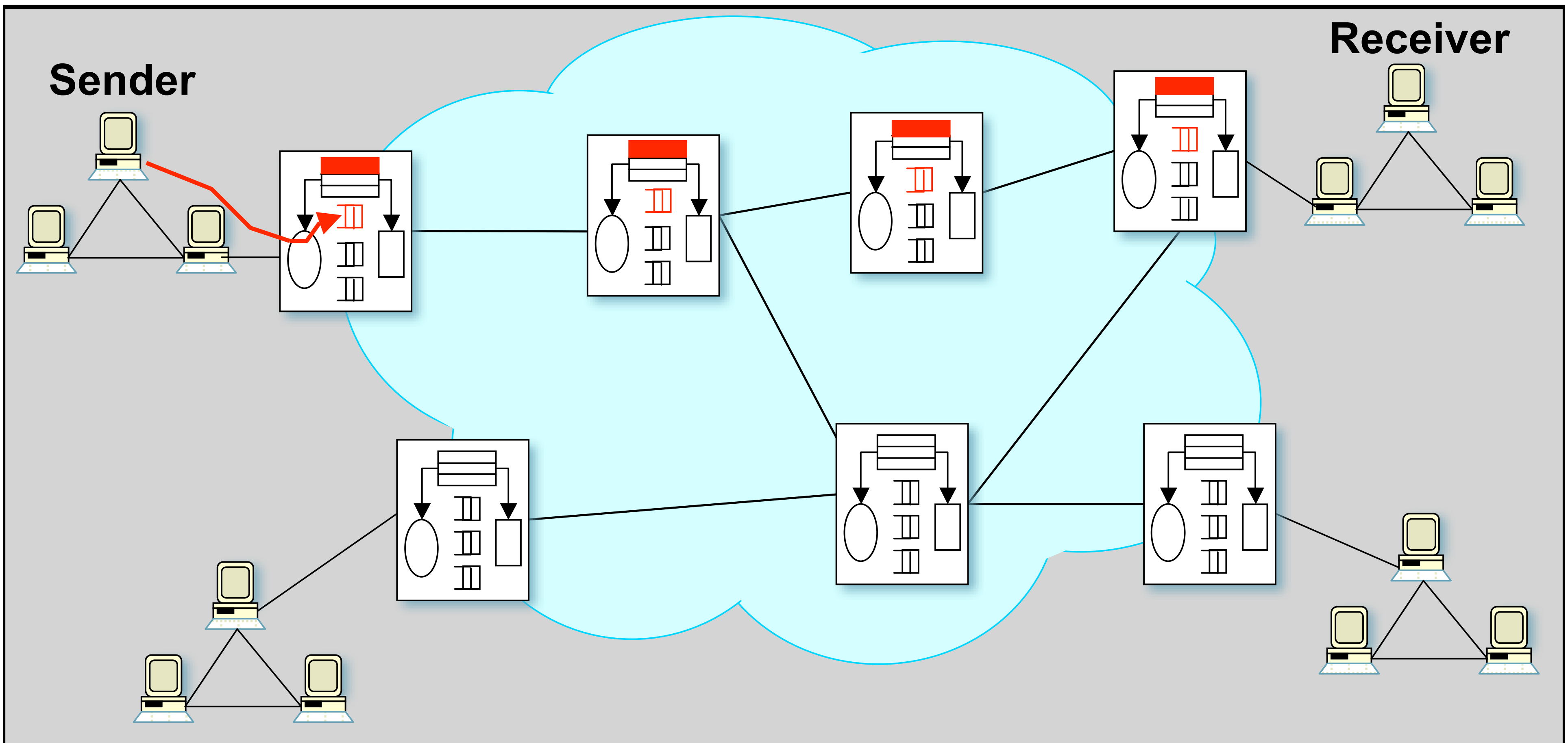
Stateful Solution: Guaranteed Services

- **Per-flow classification**



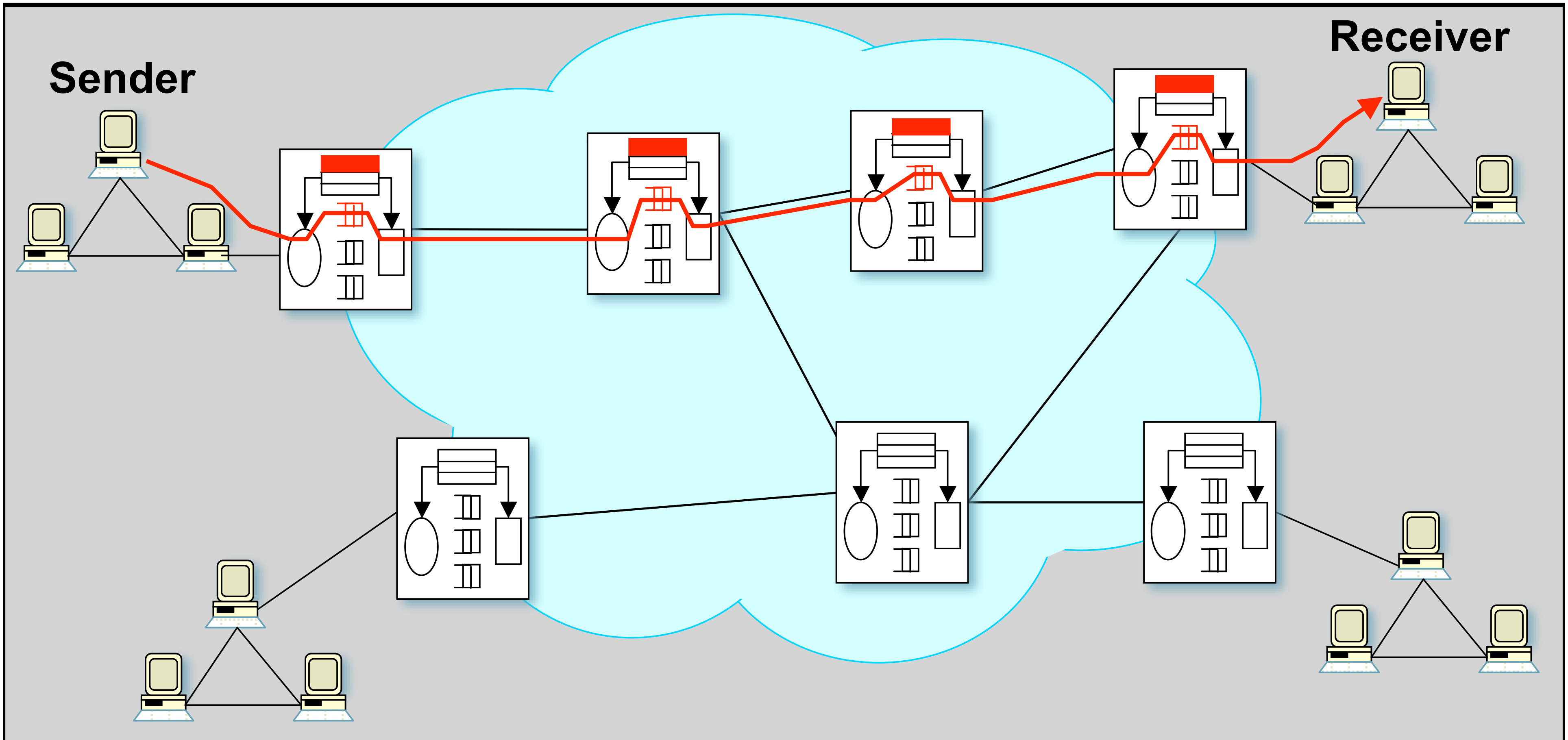
Stateful Solution: Guaranteed Services

- Per-flow buffer management



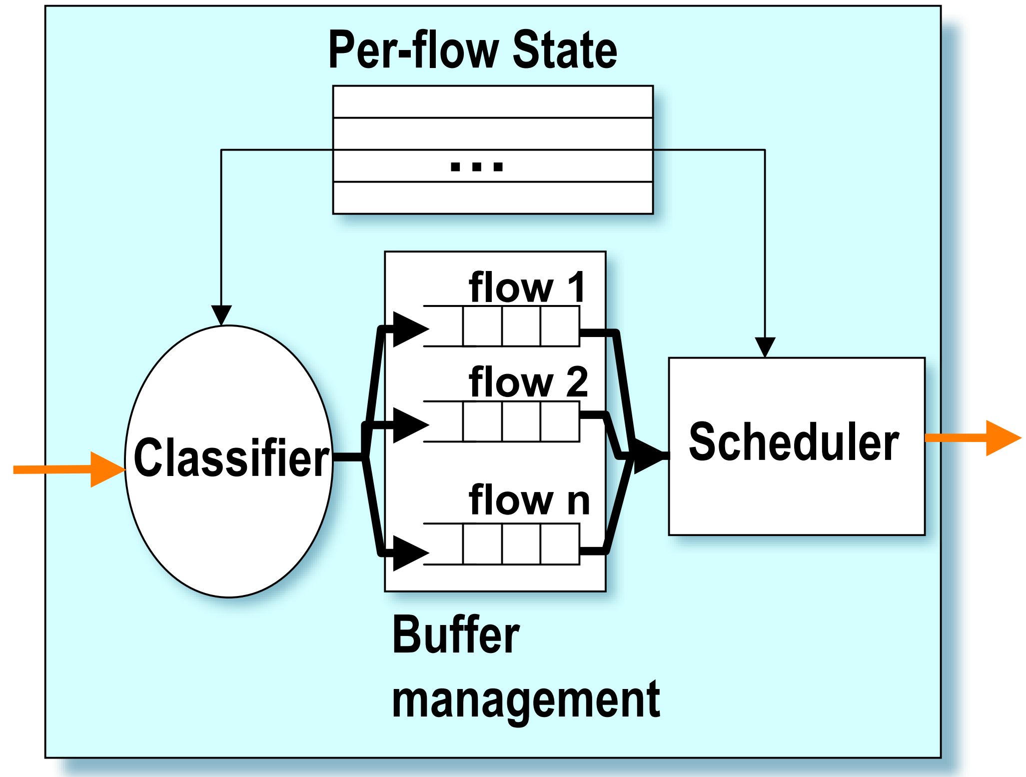
Stateful Solution: Guaranteed Services

- Per-flow scheduling



Stateful Solution Complexity

- **Data path**
  – Per-flow classification
  – Per-flow buffer management
  – Per-flow scheduling
- **Control path**
  – install and maintain per-flow state for data and control paths



Stateless vs. Stateful

- **Stateless solutions are more**
  - scalable
  - robust
- **Stateful solutions provide more powerful and flexible services**
  - guaranteed services + high resource utilization
  - fine grained differentiation
  - protection

Question

- **Can we achieve the best of two worlds, i.e., provide services implemented by stateful networks while maintaining advantages of stateless architectures?**
  - Yes, in some interesting cases. DPS, CSFQ.

- **Can we provide reduced state services, I.e., maintain state only for larger granular flows rather than end-to-end flows?**
  - Yes: Diff-serv