Scheduling

An Engineering Approach to Computer Networking

Outline

- What is scheduling
- Why we need it
- Requirements of a scheduling discipline
- Fundamental choices
- Scheduling best effort connections
- Scheduling guaranteed-service connections
- Packet drop strategies

Scheduling

- Sharing always results in contention
- A *scheduling discipline* resolves contention:
  - who's next?
- Key to *fairly sharing resources* and *providing performance guarantees*

Components

- A scheduling discipline does two things:
  - decides service order
  - manages queue of service requests
- Example:
  - consider queries awaiting web server
  - scheduling discipline decides service order
  - and also if some query should be ignored

Where?

- Anywhere where contention may occur
- At every layer of protocol stack
- Usually studied at network layer, at output queues of switches

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Why do we need one?

- *Because future applications need it*
- We expect two types of future applications
  - best-effort (adaptive, non-real time)
    - e.g. email, some types of file transfer
  - guaranteed service (non-adaptive, real time)
    - e.g. packet voice, interactive video, stock quotes

What can scheduling disciplines do?

- Give different users different qualities of service
- Example of passengers waiting to board a plane
  - early boarders spend less time waiting
  - bumped off passengers are ‘lost’!
- Scheduling disciplines can allocate
  - bandwidth
  - delay
  - loss
- They also determine how *fair* the network is

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Requirements

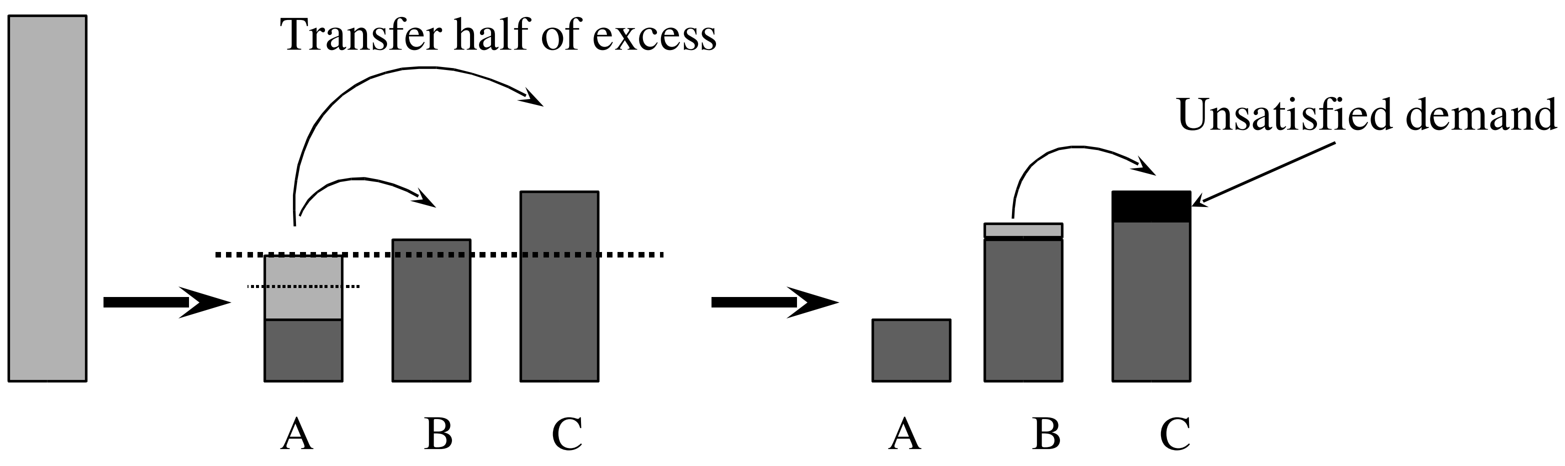
- An ideal scheduling discipline
    - is easy to implement
    - is fair
    - provides performance bounds
    - allows easy *admission control* decisions
        - to decide whether a new flow can be allowed

Requirements: 1. Ease of implementation

- Scheduling discipline has to make a decision once every few microseconds!
- Should be implementable in a few instructions or hardware
  - for hardware: critical constraint is VLSI *space*
- Work per packet should scale less than linearly with number of active connections

Requirements: 2. Fairness

- Scheduling discipline *allocates* a *resource*
- An allocation is fair if it satisfies *min-max fairness*
- Intuitively
  - each connection gets no more than what it wants
  - the excess, if any, is equally shared



Fairness (contd.)

- Fairness is *intuitively* a good idea
- But it also provides *protection*
    - traffic hogs cannot overrun others
    - automatically builds *firewalls* around heavy users
- Fairness is a *global* objective, but scheduling is local
- Each endpoint must restrict its flow to the smallest fair allocation
- Dynamics + delay => global fairness may never be achieved

Requirements: 3. Performance bounds

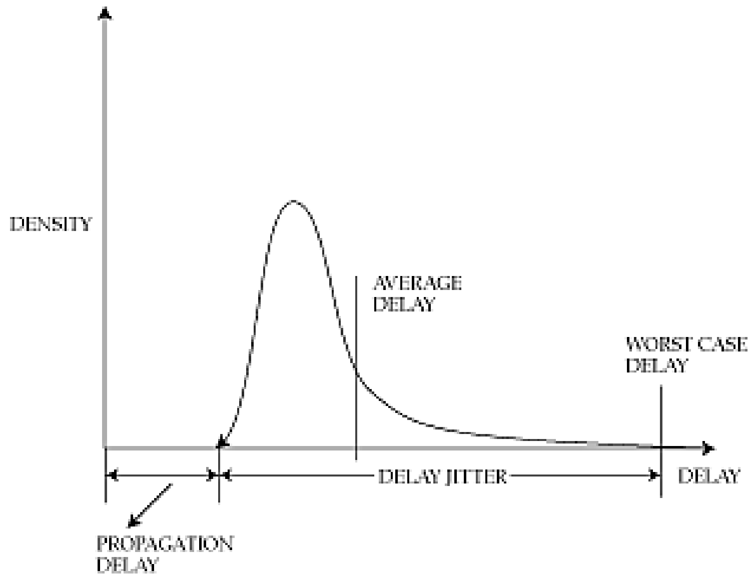
- What is it?
  - A way to obtain a desired level of service
- Can be *deterministic* or *statistical*
- Common parameters are
  - bandwidth
  - delay
  - delay-jitter
  - loss

Bandwidth

- Specified as minimum bandwidth measured over a prespecified interval
- E.g. > 5Mbps over intervals of > 1 sec
- Meaningless without an interval!
- Can be a bound on average (sustained) rate or peak rate
- Peak is measured over a ‘small’ inteval
- Average is asymptote as intervals increase without bound

Delay and delay-jitter

- Bound on some parameter of the delay distribution curve



Req'ments: 4. Ease of admission control

- Admission control needed to provide QoS
- Overloaded resource cannot guarantee performance
- Choice of scheduling discipline affects ease of admission control algorithm

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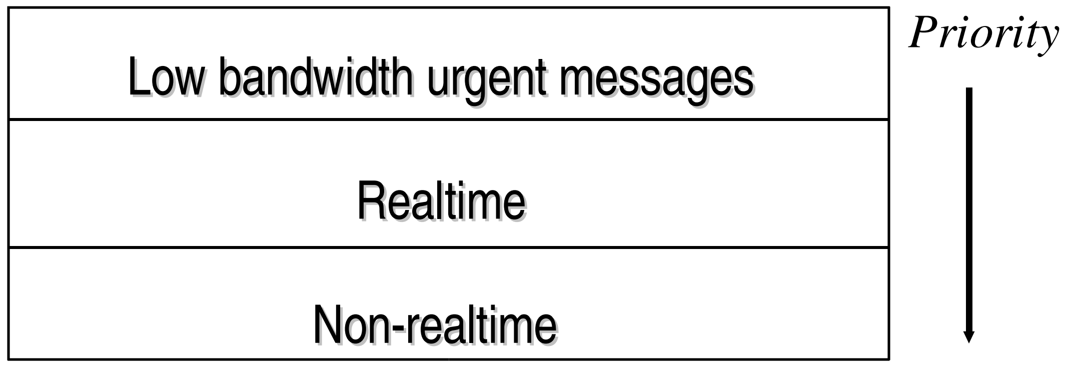
Fundamental choices

1. Number of priority levels
2. Work-conserving vs. non-work-conserving
3. Degree of aggregation
4. Service order within a level

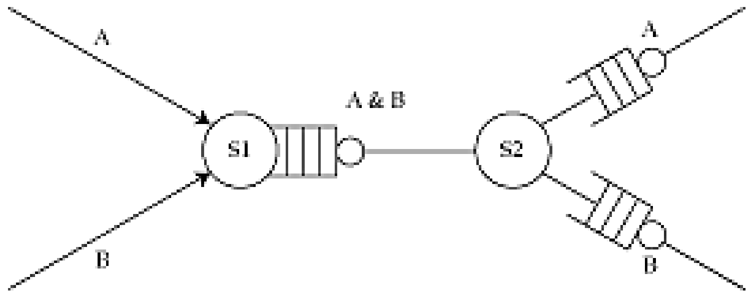
Choices: 1. Priority

- Packet is served from a given priority level only if no packets exist at higher levels (*multilevel priority with exhaustive service*)
- Highest level gets lowest delay
- Watch out for starvation!
- Usually map priority levels to delay classes



Choices: 2. Work conserving vs. non-work-conserving

- Work conserving discipline is never idle when packets await service
- Why bother with non-work conserving?



Non-work-conserving disciplines

- Key conceptual idea: delay packet till *eligible*
- Reduces delay-jitter => fewer buffers in network
- How to choose eligibility time?
  - rate-jitter regulator
    - bounds maximum outgoing rate
  - delay-jitter regulator
    - compensates for variable delay at previous hop

Do we need non-work-conservation?

- Can remove delay-jitter at an endpoint instead
  - but also reduces size of switch buffers…
- Increases mean delay
  - not a problem for *playback* applications
- Wastes bandwidth
  - can serve best-effort packets instead
- Always punishes a misbehaving source
  - can't have it both ways
- Bottom line: not too bad, implementation cost may be the biggest problem

Choices: 3. Degree of aggregation

- More aggregation
  - less state
  - cheaper
    - smaller VLSI
    - less to advertise
  - BUT: less individualization
- Solution
  - aggregate to a *class*, members of class have same performance requirement
  - no protection within class

Choices: 4. Service within a priority level

- In order of arrival (FCFS) or in order of a service tag
- Service tags => can arbitrarily reorder queue
  - Need to sort queue, which can be expensive
- FCFS
  - bandwidth hogs win (no protection)
  - no guarantee on delays
- Service tags
  - with appropriate choice, both protection and delay bounds possible

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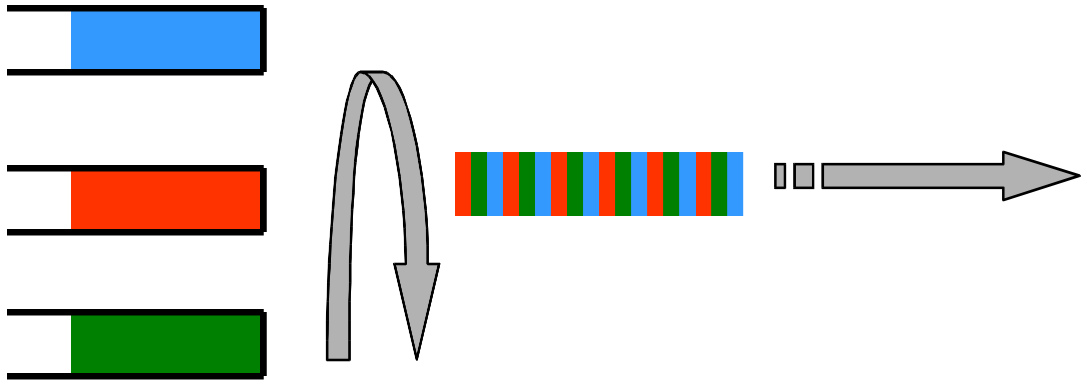
Scheduling best-effort connections

- Main requirement is *fairness*
- Achievable using *Generalized processor sharing (GPS)*
  - Visit each non-empty queue in turn
  - Serve infinitesimal from each
  - Why is this fair?
  - How can we give weights to connections?



More on GPS

- GPS is unimplementable!
  - we cannot serve infinitesimals, only packets
- No packet discipline can be as fair as GPS
  - while a packet is being served, we are unfair to others
- Degree of unfairness can be bounded
- **Define**: *work(I,a,b)* = # bits transmitted for connection I in time [a,b]
- *Absolute* fairness bound for discipline S
  - Max (work_GPS(I,a,b) - work_S(I, a,b))
- *Relative* fairness bound for discipline S
  - Max (work_S(I,a,b) - work_S(J,a,b))

What next?

- We can’t implement GPS
- So, lets see how to emulate it
- We want to be as fair as possible
- But also have an efficient implementation

Weighted round robin

- Serve a packet from each non-empty queue in turn
- Unfair if packets are of different length or weights are not equal
- Different weights, fixed packet size
  - serve more than one packet per visit, after normalizing to obtain integer weights
- Different weights, variable size packets
  - normalize weights by mean packet size
    - e.g. weights {0.5, 0.75, 1.0}, mean packet sizes {50, 500, 1500}
    - normalize weights: {0.5/50, 0.75/500, 1.0/1500} = { 0.01, 0.0015, 0.000666}, normalize again {60, 9, 4}

Problems with Weighted Round Robin

- With variable size packets and different weights, need to know mean packet size in advance
- Can be unfair for long periods of time
- E.g.
  - T3 trunk with 500 connections, each connection has mean packet length 500 bytes, 250 with weight 1, 250 with weight 10
  - Each packet takes 500 * 8/45 Mbps = 88.8 microseconds
  - Round time =2750 * 88.8 = 244.2 ms

Weighted Fair Queueing (WFQ)

- Deals better with variable size packets and weights
- GPS is fairest discipline
- Find the *finish time* of a packet, *had we been doing GPS*
- Then serve packets in order of their finish times

WFQ: first cut

- Suppose, in each *round,* the server served one bit from each active connection
- *Round number* is the number of rounds already completed
  - can be fractional
- If a packet of length *p* arrives to an empty queue when the round number is *R*, it will complete service when the round number is *R + p => finish number* is *R + p*
  - independent of the number of other connections!
- If a packet arrives to a non-empty queue, and the previous packet has a finish number of *f*, then the packet's finish number is *f+p*
- Serve packets in order of finish numbers

A catch

- A queue may need to be considered non-empty even if it has no packets in it
  - e.g. packets of length 1 from connections A and B, on a link of speed 1 bit/sec
    - at time 1, packet from A served, round number = 0.5
    - A has no packets in its queue, yet should be considered non-empty, because a packet arriving to it at time 1 should have finish number 1+ *p*
- A connection is *active* if the last packet served from it, or in its queue, has a finish number greater than the current round number

WFQ continued

- To sum up, assuming we know the current round number *R*
- Finish number of packet of length *p*
    - if arriving to active connection = previous finish number + *p*
    - if arriving to an inactive connection = *R* + *p*
- (How should we deal with weights?)
- To implement, we need to know two things:
    - is connection active?
    - if not, what is the current round number?
- Answer to both questions depends on computing the current round number (why?)

WFQ: computing the round number

- Naively: round number = number of rounds of service completed so far
  - what if a server has not served all connections in a round?
  - what if new conversations join in halfway through a round?
- *Redefine* round number as a real-valued variable that increases at a rate inversely proportional to the number of currently active connections
  - this takes care of both problems (why?)
- With this change, WFQ emulates GPS instead of bit-by-bit RR

Problem: iterated deletion

$$\begin{tikzcd}
\textit{Round number} \arrow[r, bend left] & \textit{\# active conversations} \arrow[l, bend left]
\end{tikzcd}$$

- A sever recomputes round number on each packet arrival
- At any recomputation, the number of conversations can go up at most by one, but can go down to zero
- => overestimation
- Trick
  - use previous count to compute round number
  - if this makes some conversation inactive, recompute
  - repeat until no conversations become inactive

WFQ implementation

- On packet arrival:
    - use source + destination address (or VCI) to classify it and look up finish number of last packet served (or waiting to be served)
    - recompute round number
    - compute finish number
    - insert in priority queue sorted by finish numbers
    - if no space, drop the packet with largest finish number
- On service completion
    - select the packet with the lowest finish number

Analysis

- Unweighted case:
  - if GPS has served *x* bits from connection A by time t
  - WFQ would have served at least *x* - *P* bits, where *P* is the largest possible packet in the network
- WFQ could send *more* than GPS would => absolute fairness bound > *P*
- To reduce bound, choose smallest finish number only among packets that have started service in the corresponding GPS system (WF^{2}Q)
  - requires a regulator to determine eligible packets

Evaluation

- Pros
  - like GPS, it provides protection
  - can obtain worst-case end-to-end delay bound
  - gives users incentive to use intelligent flow control (and also provides rate information implicitly)
- Cons
  - needs per-connection state
  - iterated deletion is complicated
  - requires a priority queue

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Scheduling guaranteed-service connections

- With best-effort connections, goal is fairness
- With guaranteed-service connections
  - what performance guarantees are achievable?
  - how easy is admission control?
- We now study some scheduling disciplines that provide performance guarantees

WFQ

- Turns out that WFQ also provides performance guarantees
- Bandwidth bound
  - ratio of weights * link capacity
  - e.g. connections with weights 1, 2, 7; link capacity 10
  - connections get at least 1, 2, 7 units of b/w each
- End-to-end delay bound
  - assumes that the connection doesn’t send ‘too much’ (otherwise its packets will be stuck in queues)
  - more precisely, connection should be *leaky-bucket* regulated
  - # bits sent in time $[t_1, t_2] <= \square\ (t_2 - t_1) + \square$

Parekh-Gallager theorem

- Let a connection be allocated weights at each WFQ scheduler along its path, so that the least bandwidth it is allocated is *g*
- Let it be leaky-bucket regulated such that # bits sent in time $[t_1, t_2]$ <= □ $(t_2 - t_1)$ + □
- Let the connection pass through *K* schedulers, where the *k*th scheduler has a rate *r(k)*
- Let the largest packet allowed in the network be *P*

$$\boxed{end\_to\_end\_delay \qquad / g \qquad \overset{K\quad 1}{\underset{k\quad 1}{}} \; P / g \qquad \overset{K}{\underset{k\quad 1}{}} \; P / r(k)}$$

Significance

- Theorem shows that WFQ can provide end-to-end delay bounds
- So WFQ provides both fairness and performance guarantees
- Boud holds regardless of cross traffic behavior
- Can be generalized for networks where schedulers are variants of WFQ, and the link service rate changes over time

Problems

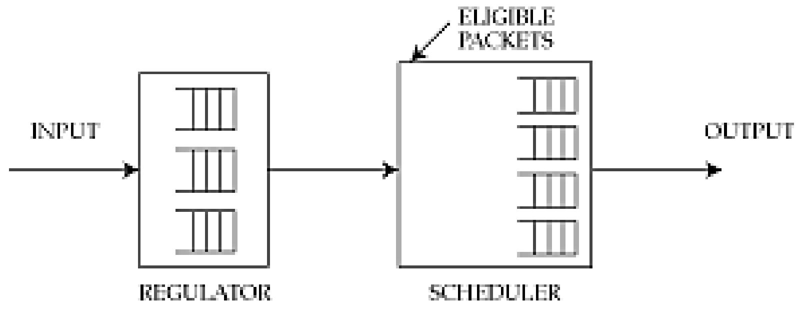
- To get a delay bound, need to pick *g*
  - the lower the delay bounds, the larger *g* needs to be
  - large *g* => exclusion of more competitors from link
  - *g* can be very large, in some cases 80 times the peak rate!
- Sources must be leaky-bucket regulated
  - but choosing leaky-bucket parameters is problematic
- WFQ couples delay and bandwidth allocations
  - low delay requires allocating more bandwidth
  - wastes bandwidth for low-bandwidth low-delay sources

Delay-Earliest Due Date

- Earliest-due-date: packet with earliest deadline selected
- Delay-EDD prescribes how to assign deadlines to packets
- A source is required to send slower than its *peak rate*
- Bandwidth at scheduler reserved at peak rate
- Deadline = expected arrival time + delay bound
  - If a source sends faster than contract, delay bound will not apply
- Each packet gets a hard delay bound
- Delay bound is *independent* of bandwidth requirement
  - but reservation is at a connection’s peak rate
- Implementation requires per-connection state and a priority queue

Rate-controlled scheduling

- A *class* of disciplines
  - two components: regulator and scheduler
  - incoming packets are placed in regulator where they wait to become eligible
  - then they are put in the scheduler
- Regulator *shapes* the traffic, scheduler provides performance guarantees



Examples

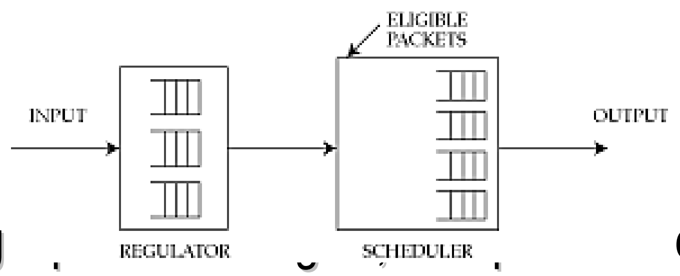
- Recall
  - rate-jitter regulator
    - bounds maximum outgoing rate
  - delay-jitter regulator
    - compensates for variable delay at previous hop
- Rate-jitter regulator + FIFO
  - similar to Delay-EDD (what is the difference?)
- Rate-jitter regulator + multi-priority FIFO
  - gives both bandwidth and delay guarantees (RCSP)
- Delay-jitter regulator + EDD
  - gives bandwidth, delay,and delay-jitter bounds (Jitter-EDD)

Analysis

- First regulator on path monitors and regulates traffic => bandwidth bound
- End-to-end delay bound
  - delay-jitter regulator
    - reconstructs traffic => end-to-end delay is fixed (= worst-case delay at each hop)
  - rate-jitter regulator
    - partially reconstructs traffic
    - can show that end-to-end delay bound is smaller than (sum of delay bound at each hop + delay at first hop)

Decoupling

- Can give a low-bandwidth connection a low delay without overbooking
- E.g consider connection A with rate 64 Kbps sent to a router with rate-jitter regulation and multipriority FCFS scheduling



- After sending eligible at time (now + *l*/64 Kbps)
- If placed at highest-priority queue, all packets from A get low delay
- Can decouple delay and bandwidth bounds, unlike WFQ

Evaluation

- Pros
  - flexibility: ability to emulate other disciplines
  - can decouple bandwidth and delay assignments
  - end-to-end delay bounds are easily computed
  - do not require complicated schedulers to guarantee protection
  - can provide delay-jitter bounds
- Cons
  - require an additional regulator at each output port
  - delay-jitter bounds at the expense of increasing mean delay
  - delay-jitter regulation is expensive (clock synch, timestamps)

Summary

- Two sorts of applications: best effort and guaranteed service
- Best effort connections require fair service
  - provided by GPS, which is unimplementable
  - emulated by WFQ and its variants
- Guaranteed service connections require performance guarantees
  - provided by WFQ, but this is expensive
  - may be better to use rate-controlled schedulers

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Packet dropping

- Packets that cannot be served immediately are buffered
- Full buffers => *packet drop strategy*
- Packet losses happen almost always from best-effort connections (why?)
- Shouldn’t drop packets unless imperative
    - packet drop wastes resources (why?)

Classification of drop strategies

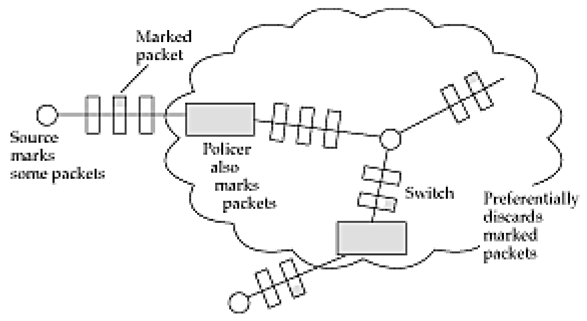
1. Degree of aggregation
2. Drop priorities
3. Early or late
4. Drop position

1. Degree of aggregation

- Degree of discrimination in selecting a packet to drop
- E.g. in vanilla FIFO, all packets are in the same class
- Instead, can classify packets and drop packets selectively
- The finer the classification the better the protection
- Max-min fair allocation of buffers to classes
  - drop packet from class with the longest queue (why?)

2. Drop priorities

- Drop lower-priority packets first
- How to choose?
  - endpoint marks packets
  - regulator marks packets
  - congestion loss priority (CLP) bit in packet header


Marked packet
Source marks some packets
Policer also marks packets
Switch
Preferentially discards marked packets

CLP bit: pros and cons

- Pros
  - if network has spare capacity, all traffic is carried
  - during congestion, load is automatically shed
- Cons
  - separating priorities within a single connection is hard
  - what prevents all packets being marked as high priority?

2. Drop priority (contd.)

- Special case of AAL5
  - want to drop an entire frame, not individual cells
  - cells belonging to the selected frame are preferentially dropped
- Drop packets from ‘nearby’ hosts first
  - because they have used the least network resources
  - can’t do it on Internet because hop count (TTL) decreases

3. Early vs. late drop

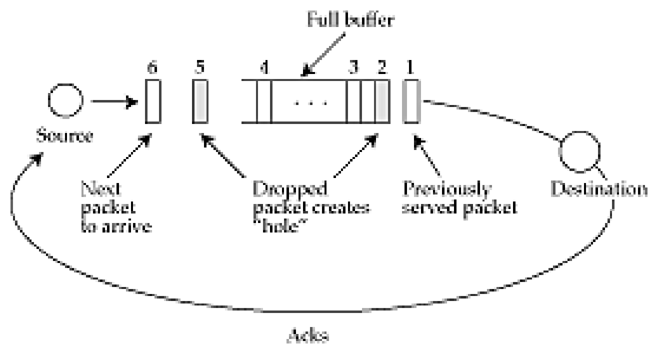
- Early drop => drop even if space is available
    - signals endpoints to reduce rate
    - cooperative sources get lower overall delays, uncooperative sources get severe packet loss
- Early random drop
    - drop arriving packet with fixed drop probability if queue length exceeds threshold
    - intuition: misbehaving sources more likely to send packets and see packet losses
    - doesn’t work!

3. Early vs. late drop: RED

- Random early detection (RED) makes three improvements
- Metric is moving average of queue lengths
  - small bursts pass through unharmed
  - only affects sustained overloads
- Packet drop probability is a function of mean queue length
  - prevents severe reaction to mild overload
- Can mark packets instead of dropping them
  - allows sources to detect network state without losses
- RED improves performance of a network of cooperating TCP sources
- No bias against bursty sources
- Controls queue length regardless of endpoint cooperation

4. Drop position

- Can drop a packet from head, tail, or random position in the queue
- Tail
  - easy
  - default approach
- Head
  - harder
  - lets source detect loss earlier



4. Drop position (contd.)

- Random
  - hardest
  - if no aggregation, hurts hogs most
  - unlikely to make it to real routers
- Drop entire longest queue
  - easy
  - almost as effective as drop tail from longest queue