# Advanced Computer Networks

Fair Queuing

## Overview

- TCP and queues
- Queuing disciplines
- RED
- Fair-queuing
- Core-stateless FQ
- XCP



## Example

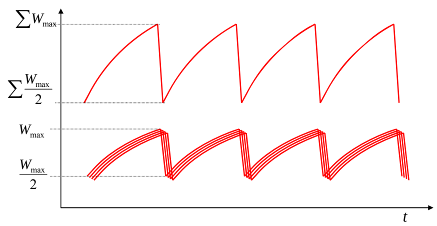
- 10Gb/s linecard
  - Requires 300Mbytes of buffering.
  - Read and write 40 byte packet every 32ns.
- Memory technologies
  - DRAM: require 4 devices, but too slow.
  - SRAM: require 80 devices, 1kW, $2000.
- Problem gets harder at 40Gb/s
  - Hence new memory access technologies: RLDRAM, FCRAM, etc.



## Rule-of-thumb

- Rule-of-thumb makes sense for one flow
- Typical backbone link has > 20,000 flows
- Does the rule-of-thumb still hold?

## If flows are synchronized




- Aggregate window has same dynamics
- Therefore buffer occupancy has same dynamics
- Rule-of-thumb still holds.

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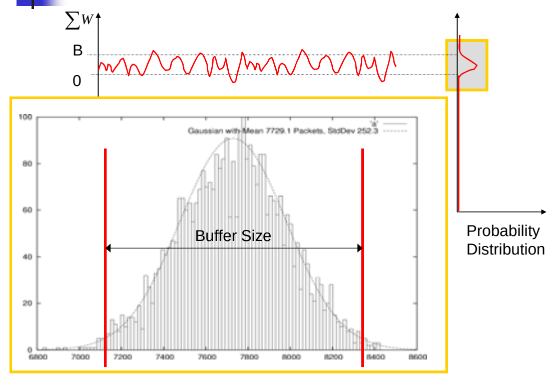
## If flows are not synchronized

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## Central Limit Theorem

- Central Limit Theorem: If samples $X_1, \dots X_n$ independent and from same population with population mean $\mu$ and standard deviation $\sigma$, then

sample mean: $$\bar{X} = \frac{\sum_{j=1}^{n} X_j}{n}$$

is approximately normally distributed with mean $\mu$ and standard deviation $\frac{\sigma}{\sqrt{n}}$

## Central Limit Theorem

- CLT tells us that the more variables (Congestion Windows of Flows) we have, the narrower the Gaussian (Fluctuation of sum of windows)
  - Width of Gaussian decreases with $\frac{1}{\sqrt{n}}$
  - Buffer size should also decreases with $\frac{1}{\sqrt{n}}$

$$B \rightarrow \frac{B_{n=1}}{\sqrt{n}} = \frac{2T \times C}{\sqrt{n}}$$

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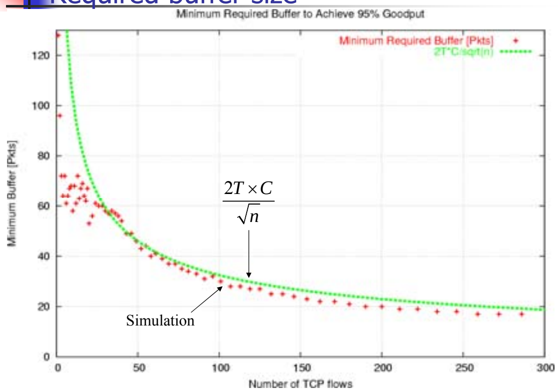
## Required buffer size



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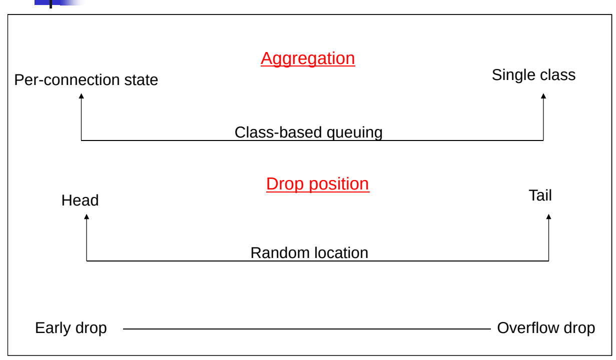
## Queuing Disciplines

- Each router must implement some queuing discipline
- Queuing allocates both bandwidth and buffer space:
  - Bandwidth: which packet to serve (transmit) next
  - Buffer space: which packet to drop next (when required)
- Queuing also affects latency



## Packet Drop Dimensions



## 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 policy



## FIFO + Drop-tail Problems

- Leaves responsibility of congestion control to edges (e.g., TCP)
- Does not separate between different flows
- No policing: send more packets → get more service
- Synchronization: end hosts react to same events



## Active Queue Management

- Design active router queue management to aid congestion control
- Why?
  - Routers can distinguish between propagation and persistent queuing delays
  - Routers can decide on transient congestion, based on workload



## Active Queue Designs

- Modify both router and hosts
  - DECbit – congestion bit in packet header
- Modify router, hosts use TCP
  - Fair queuing
    - Per-connection buffer allocation
  - RED (Random Early Detection)
    - Drop packet or set bit in packet header as soon as congestion is starting

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## Internet Problems

- Full queues
  - Routers are forced to have have large queues to maintain high utilizations
  - TCP detects congestion from loss
    - Forces network to have long standing queues in steady-state
- Lock-out problem
  - Drop-tail routers treat bursty traffic poorly
  - Traffic gets synchronized easily → allows a few flows to monopolize the queue space

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## Design Objectives

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

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## Lock-out Problem

- Random drop
  - Packet arriving when queue is full causes some random packet to be dropped
- Drop front
  - On full queue, drop packet at head of queue
- Random drop and drop front solve the lock-out problem but not the full-queues problem

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## Full Queues Problem

- Drop packets before queue becomes full (early drop)
- Intuition: notify senders of incipient congestion
  - Example: random early drop (RED):
    - If qlen > drop level, drop each new packet with fixed probability *p*
    - Does not control misbehaving users

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## Random Early Detection (RED)

- Detect incipient congestion, allow bursts
- Keep power (throughput/delay) high
  - Keep average queue size low
  - Assume hosts respond to lost packets
- Avoid window synchronization
  - Randomly mark packets
- Avoid bias against bursty traffic
- Some protection against ill-behaved users

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## RED Algorithm

- Maintain running average of queue length
- If avgq < $min_{th}$ do nothing
  - Low queuing, send packets through
- If avgq > $max_{th}$, drop packet
  - Protection from misbehaving sources
- Else mark packet in a manner proportional to queue length
  - Notify sources of incipient congestion

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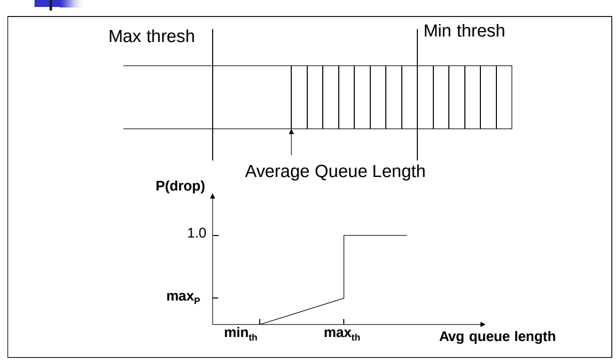
## RED Operation

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## RED Algorithm

- Maintain running average of queue length
  - Byte mode vs. packet mode
- For each packet arrival
  - Calculate average queue size (avg)
  - If $min_{th} \leq avgq < max_{th}$
    - Calculate probability $P_a$
    - With probability $P_a$
      - Mark the arriving packet
    - Else if $max_{th} \leq avg$
      - Mark the arriving packet



## Queue Estimation

- Standard Exponentially weighted moving average (EWMA) :
  - $avgq = (1-w_q)\ avgq + w_q qlen$
  - Special fix for idle periods – why?
- Upper bound on $w_q$ depends on $min_{th}$
  - Want to ignore transient congestion
  - Can calculate the queue average if a burst arrives
    - Set $w_q$ such that certain burst size does not exceed $min_{th}$
- Lower bound on $w_q$ to detect congestion relatively quickly
- Typical $w_q = 0.002$



## Thresholds

- $min_{th}$ determined by the utilization requirement
  - Tradeoff between queuing delay and utilization
- Relationship between $max_{th}$ and $min_{th}$
  - Want to ensure that feedback has enough time to make difference in load
  - Depends on average queue increase in one RTT
  - Paper suggest ratio of 2
    - Current rule of thumb is factor of 3



## Packet Marking

- $max_p$ is reflective of typical loss rates
- Paper uses 0.02
  - 0.1 is more realistic value
- If network needs marking of 20-30% then need to buy a better link!
- Gentle variant of RED (recommended)
  - Vary drop rate from $max_p$ to 1 as the avgq varies from $max_{th}$ to 2* $max_{th}$
  - More robust to setting of $max_{th}$ and $max_p$

## Extending RED for Flow Isolation

- Problem: what to do with non-cooperative flows?
- Fair queuing achieves isolation using per-flow state – expensive at backbone routers
  - How can we isolate unresponsive flows without per-flow state?
- RED penalty box
  - Monitor history for packet drops, identify flows that use disproportionate bandwidth
  - Isolate and punish those flows

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## Fairness Goals

- Allocate resources fairly
- Isolate ill-behaved users
  - Router does not send explicit feedback to source
  - Still needs e2e congestion control
- Still achieve statistical muxing
  - One flow can fill entire pipe if no contenders
  - Work conserving → scheduler never idles link if it has a packet

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## What is Fairness?

- At what granularity?
  - Flows, connections, domains?
- What if users have different RTTs/links/etc.
  - Should it share a link fairly or be TCP fair?
- Maximize fairness index?
  - Fairness = $(\Sigma x_i)^2/n(\Sigma x_i^2)$ $0<\text{fairness}<1$
- Basically a tough question to answer – typically design mechanisms instead of policy
  - User = arbitrary granularity

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## Max-min Fairness

- Allocate user with "small" demand what it wants, evenly divide unused resources to "big" users
- Formally:
    - Resources allocated in terms of increasing demand
    - No source gets resource share larger than its demand
    - Sources with unsatisfied demands get equal share of resource



## Max-min Fairness Example

- Assume sources 1..n, with resource demands X1..Xn in ascending order
- Assume channel capacity C.
    - Give C/n to X1; if this is more than X1 wants, divide excess (C/n - X1) to other sources: each gets C/n + (C/n - X1)/(n-1)
    - If this is larger than what X2 wants, repeat process



## Implementing max-min Fairness

- Generalized processor sharing
    - Fluid fairness
    - Bitwise round robin among all queues
- Why not simple round robin?
    - Variable packet length → can get more service by sending bigger packets
    - Unfair instantaneous service rate
        - What if arrive just before/after packet departs?

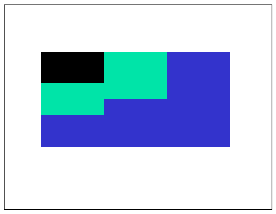


## Bit-by-bit RR

- Single flow: clock ticks when a bit is transmitted. For packet i:
    - $P_i$ = length, $A_i$ = arrival time, $S_i$ = begin transmit time, $F_i$ = finish transmit time
    - $F_i = S_i+P_i = \max(F_{i-1}, A_i) + P_i$
- Multiple flows: clock ticks when a bit from all active flows is transmitted → round number
    - Can calculate $F_i$ for each packet if number of flows is know at all times
        - This can be complicated

## Bit-by-bit RR Illustration

- Not feasible to interleave bits on real networks
  - FQ simulates bit-by-bit RR



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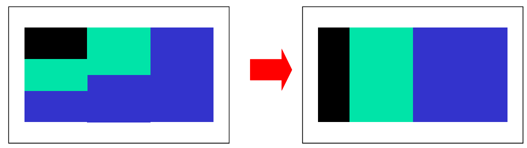
## Fair Queuing

- Mapping bit-by-bit schedule onto packet transmission schedule
- Transmit packet with the lowest $F_i$ at any given time
  - How do you compute $F_i$?



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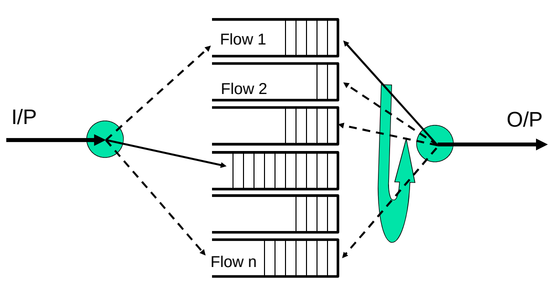
## FQ Illustration




Variation: Weighted Fair Queuing (WFQ)

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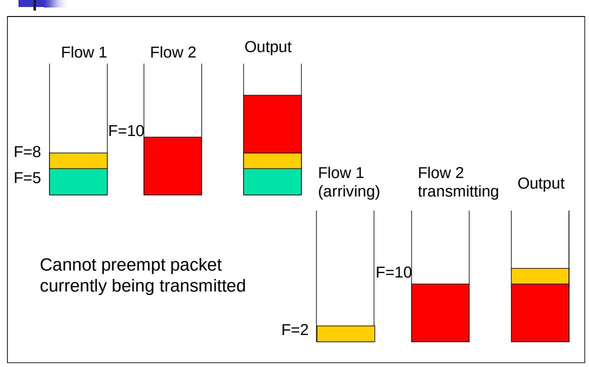
## Bit-by-bit RR Example

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## Fair Queuing Tradeoffs

- FQ can control congestion by monitoring flows
  - Non-adaptive flows can still be a problem – why?
- Complex state
  - Must keep queue per flow
    - Hard in routers with many flows (e.g., backbone routers)
    - Flow aggregation is a possibility (e.g. do fairness per domain)
- Complex computation
  - Classification into flows may be hard
  - Must keep queues sorted by finish times
  - Finish times change whenever the flow count changes

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## Core-Stateless Fair Queuing

- Key problem with FQ is core routers
  - Must maintain state for 1000's of flows
  - Must update state at Gbps line speeds
- CSFQ (Core-Stateless FQ) objectives
  - Edge routers should do complex tasks since they have fewer flows
  - Core routers can do simple tasks
    - No per-flow state/processing → this means that core routers can only decide on dropping packets not on order of processing
    - Can only provide max-min bandwidth fairness not delay allocation

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## Core-Stateless Fair Queuing

- Edge routers keep state about flows and do computation when packet arrives
- DPS (Dynamic Packet State)
  - Edge routers label packets with the result of state lookup and computation
- Core routers use DPS and local measurements to control processing of packets

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## Edge Router Behavior

- Monitor each flow i to measure its arrival rate ($r_i$)
  - EWMA of rate
  - Non-constant EWMA constant
    - $e^{-T/K}$ where T = current interarrival, K = constant
    - Helps adapt to different packet sizes and arrival patterns
- Rate is attached to each packet

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## Core Router Behavior

- Keep track of fair share rate α
  - Increasing α does not increase load (F) by N * α
  - $F(\alpha) = \Sigma_i \min(r_i, \alpha)$
  - Periodically update α
  - Keep track of current arrival rate
    - Only update α if entire period was congested or uncongested
- Drop probability for packet = max(1- α/r, 0)

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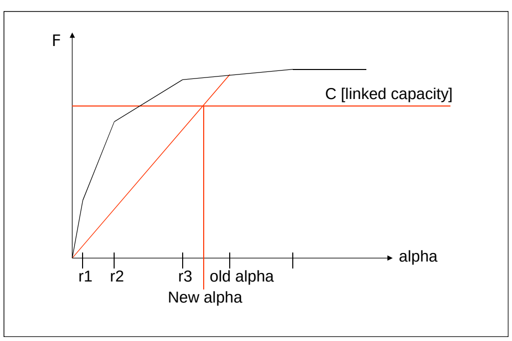
## F vs. Alpha

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## Estimating Fair Share

- Need F(α) = capacity = C
  - Can't keep map of F(α) values → would require per flow state
  - Since F(α) is concave, piecewise-linear
    - $F(0) = 0$ and $F(\alpha_{old})$ = current accepted rate = $F_c$
    - Draw a line: from (0,0) to ($\alpha_{old}$, $F_c$)
    - Hopefully, when $\alpha_{new}$, the $F(\alpha_{new})$ = the capacity of the link C
    - $F(\alpha_{new}) = C \rightarrow \alpha_{new} / \alpha_{old} = C/F_c \rightarrow \alpha_{new} = \alpha_{old} * C/F_c$
- What if a mistake was made?
  - Forced into dropping packets due to buffer capacity
  - When queue overflows α is decreased slightly

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## Other Issues

- Punishing fire-hoses – why?
  - Easy to keep track of in a FQ scheme
- What are the real edges in such a scheme?
  - Must trust edges to mark traffic accurately
  - Could do some statistical sampling to see if edge was marking accurately

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## Overview

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- XCP (eXplicit Control Protocol)

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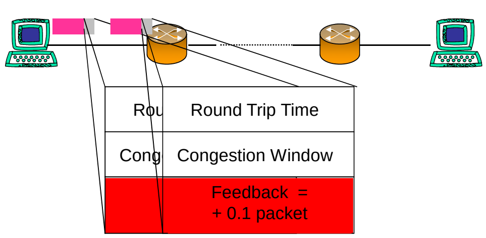
## How does XCP Work?




Congestion Header

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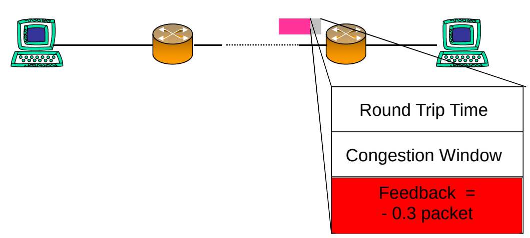
## How does XCP Work?

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## How does XCP Work?

Congestion Window = Congestion Window + Feedback

XCP extends ECN and CSFQ

**Routers compute feedback without any per-flow state**

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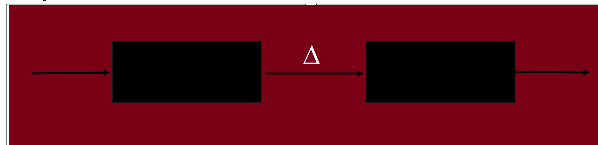
## How Does an XCP Router Compute the Feedback?




queue

MIMD

Algorithm:

Aggregate traffic changes by Δ

Δ ~ Spare Bandwidth

Δ ~ - Queue Size

So, $\Delta = \alpha\, d_{avg}\, Spare - \beta\, Queue$

Congestion He

AIMD

Algorithm:

If Δ > 0 ⇒ Divide Δ equally between flows

If Δ < 0 ⇒ Divide Δ between flows proportionally to their current rates

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## Getting the devil out of the details …

**Congestion Controller**

$\Delta = \alpha\, d_{avg}\, Spare - \beta\, Queue$

Theorem: System converges to optimal utilization (i.e., stable) for any link bandwidth, delay, number of sources if:

$$0 < \alpha < \frac{\pi}{4\sqrt{2}} \quad and \quad \beta = \alpha^2\sqrt{2}$$

No Parameter Tuning

**Fairness Controller**

Algorithm:

If Δ > 0 ⇒ Divide Δ equally between flows

If Δ < 0 ⇒ Divide Δ between flows proportionally to their current rates

Need to estimate number of flows $N$

$$N = \sum_{pkts\ in\ T} \frac{1}{T \times (Cwnd_{pkt} / RTT_{pkt})}$$

$RTT_{pkt}$ : Round Trip Time in header

No Per-Flow State

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## Discussion

- RED
  - Parameter settings
- RED vs. FQ
  - How much do we need per flow tracking? At what cost?
- FQ vs. XCP/CSFQ
  - Is coarse-grained fairness sufficient?
  - Misbehaving routers/trusting the edge
  - Deployment (and incentives)
  - How painful is FQ
- XCP vs CSFQ
  - What are the key differences
- Granularity of fairness
  - Mechanism vs. policy → will see this in QoS

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## Important Lessons

- How does TCP implement AIMD?
  - Sliding window, slow start & ack clocking
  - How to maintain ack clocking during loss recovery → fast recovery
- How does TCP fully utilize a link?
  - Role of router buffers
- TCP alternatives
  - TCP being used in new/unexpected ways
  - Key changes needed



## Lessons

- Fairness and isolation in routers
  - Why is this hard?
  - What does it achieve – e.g. do we still need congestion control?
- Routers
  - FIFO, drop-tail interacts poorly with TCP
  - Various schemes to desynchronize flows and control loss rate (e.g. RED)
- Fair-queuing
  - Clean resource allocation to flows
  - Complex packet classification and scheduling
- Core-stateless FQ & XCP
  - Coarse-grain fairness
  - Carrying packet state can reduce complexity



## EXTRA SLIDES

The rest of the slides are FYI

## Overview

- Fairness
- Fair-queuing
- Core-stateless FQ
- Other FQ variants

## Stochastic Fair Queuing

- Compute a hash on each packet
- Instead of per-flow queue have a queue per hash bin
- An aggressive flow steals traffic from other flows in the same hash
- Queues serviced in round-robin fashion
  - Has problems with packet size unfairness
- Memory allocation across all queues
  - When no free buffers, drop packet from longest queue

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## Deficit Round Robin

- Each queue is allowed to send Q bytes per round
- If Q bytes are not sent (because packet is too large) deficit counter of queue keeps track of unused portion
- If queue is empty, deficit counter is reset to 0
- Uses hash bins like Stochastic FQ
- Similar behavior as FQ but computationally simpler

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## Self-clocked Fair Queuing

- Virtual time to make computation of finish time easier
- Problem with basic FQ
  - Need be able to know which flows are really backlogged
    - They may not have packet queued because they were serviced earlier in mapping of bit-by-bit to packet
    - This is necessary to know how bits sent map onto rounds
    - Mapping of real time to round is piecewise linear → however slope can change often

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## Self-clocked FQ

- Use the finish time of the packet being serviced as the virtual time
  - The difference in this virtual time and the real round number can be unbounded
- Amount of service to backlogged flows is bounded by factor of 2

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## Start-time Fair Queuing

- Packets are scheduled in order of their start not finish times
- Self-clocked → virtual time = start time of packet in service
- Main advantage → can handle variable rate service better than other schemes

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## TCP Modeling

- Given the congestion behavior of TCP can we predict what type of performance we should get?
- What are the important factors
  - Loss rate
    - Affects how often window is reduced
  - RTT
    - Affects increase rate and relates BW to window
  - RTO
    - Affects performance during loss recovery
  - MSS
    - Affects increase rate

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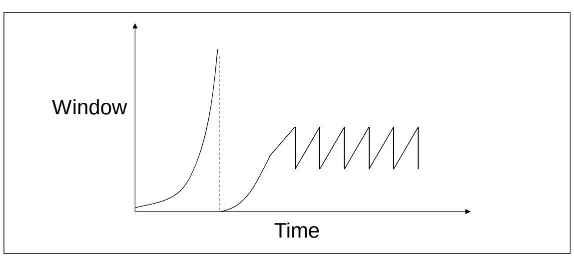
## Overall TCP Behavior

- Let's concentrate on steady state behavior with no timeouts and perfect loss recovery

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## Simple TCP Model

- Some additional assumptions
  - Fixed RTT
  - No delayed ACKs
- In steady state, TCP losses packet each time window reaches W packets
  - Window drops to W/2 packets
  - Each RTT window increases by 1 packet→W/2 * RTT before next loss
  - BW = MSS * avg window/RTT =
    - MSS * (W + W/2)/(2 * RTT)
    - .75 * MSS * W / RTT

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## Simple Loss Model

- What was the loss rate?
  - Packets transferred between losses =
    - Avg BW * time =
    - (.75 W/RTT) * (W/2 * RTT) = $3W^2/8$
  - 1 packet lost → loss rate = p = $8/3W^2$
  - W = sqrt( 8 / (3 * loss rate))
- BW = .75 * MSS * W / RTT
  - BW = MSS / (RTT * sqrt (2/3p))

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## TCP Friendliness

- What does it mean to be TCP friendly?
  - TCP is not going away
  - Any new congestion control must compete with TCP flows
    - Should not clobber TCP flows and grab bulk of link
    - Should also be able to hold its own, i.e. grab its fair share, or it will never become popular
- How is this quantified/shown?
  - Has evolved into evaluating loss/throughput behavior
  - If it shows 1/sqrt(p) behavior it is ok
  - But is this really true?

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