Ways to deal with congestion

- Host-centric versus router-centric
- Reservation-based versus feedback-based
- Window-based versus rate-based
- The Internet is: host-centric, feedback-based, and window-based
  - Because that’s what TCP is
  - But this is to some extent an “accident” of TCP’s history
Alternative approaches

- In the mid-90’s, there was a concerted effort to make Internet QoS more router-centric, reservation-based, and rate-based
  - An architecture called Integrated-Services ("intserv")
  - And a resource reservation protocol called RSVP
  - This didn’t take off, but it’s interesting to look at, and to see where things stand now
Queuing disciplines

- We talked a bit about RED
- But in fact, most queuing in the internet is FIFO with tail-drop
  - FIFO means First-In-First-Out, like the queue in a bank
    - This is a scheduling discipline
  - Tail drop means that, if the queue overflows, you drop the last packet received
    - This is a drop policy
Limitations of FIFO . . .

- The problem with FIFO is that aggressive flows can squeeze out conservative flows
  - A TCP that doesn’t follow AIMD rules can grab all the bandwidth
  - Non-TCP connections (voice) can grab all the bandwidth

- It just isn’t fair!
Fair Queuing

- Use multiple FIFO queues instead of just one
- Assign traffic to queue according to some policy
  - Such as type of traffic (voice versus TCP)
  - Or by TCP flow
- Service each queue in turn
Fair Queuing
Fair Queuing issues

- Scheduling must be (conceptually) per bit, not per packet
  - Else large packet flows get more bandwidth
- Often you want to schedule a short packet from Q1 that arrived after a long packet in Q2
  - Unless of course the long packet is already in transit . . .
Fair Queuing issues

- Perfect “per bit” scheduling of fair queues a bit expensive
  - Book defines giving a sending timestamp to each packet, and then sending in order, but now you have an ordering job
- I’ll ask you to build simple “reasonable approximations” in project 4!
Weighted Fair Queuing

- If we want to give higher priority to some queues over others, we can schedule bits from some queues more often than others.
  - “Q1 gets 2 bits for every 1 bit from Q2”
- Why do this rather than a strict priority queuing scheme???
  - Service Q2 only if Q1 empty, service Q3 only if Q2 empty, etc…
Fair Queuing is “work conserving”

- “Work conserving” means:
  - If there is work to be done, it will be done
- With fair queuing, if any queue has something to send, it will be sent
- Note that this is not the case with pure circuit switches, where BW has been reserved whether it is used or not!
What is a “real-time” application?

- One where the time at which a packet is “played out” is important
  - Voice or video . . .
- But real-time applications can have extremely different network requirements
  - Voice conversation is very bad if delay > 200ms or so
  - Streaming media can be delayed for many seconds
    - Telnet has much stricter delay requirements!
Play-out (or playback) buffer

Stream Source -> Network -> Playback Buffer

- even spaced
- jitter
- even spaced (with delay introduced)
Real-time applications

- Some video applications can adapt bandwidth requirements over a large range
  - High-fidelity versus low-fidelity bits
- And can therefore tolerate wide BW variance
- Others won’t or can’t do this…
IETF Intserv (Integrated Services)

- IETF attempt at fine-grained (per-flow) QoS
  - Resource reservation with admission control
- Settled on two types of service:
  - Guaranteed
    - i.e. conversational voice
  - Controlled Load
    - More tolerant/adaptive realtime applications
- (In addition to existing “best effort” service)
Guaranteed versus Controlled Load

- Guaranteed really requires reserved resources, careful packet scheduling.
- Controlled Load is based on the notion that most realtime apps work well as long as the network is lightly loaded.
  - Simply give this class adequate bandwidth (WFQ), but otherwise treat FIFO.
Flowspec

- Recall that the more bursty traffic is, the quicker queues build up.
- To make admission control decisions, the network needs to know how bursty a given flow is going to be.
- And, it needs to guarantee that the flow is no more bursty than it claimed.
- A flowspec is what describes the traffic (TSpec) and the network requirements (RSpec).
Token bucket (aka Leaky bucket)

- A simple and common way to describe traffic is with a token bucket

- Two parameters:
  - Rate $r$ (bits per second)
    - The size of the hole in the bucket
    - (average throughput)
  - and bucket Depth $B$ (bits)
    - The size of the bucket itself
    - (max burst size)
Token bucket policing

- A token bucket flowspec \((r,B)\) can be enforced with a queue of size \(B\) that is serviced at a rate of \(r\).
- The network can therefore enforce compliance.
- The network will tag a non-compliant packet as “out of spec” rather than drop it.
  - And then drop with higher priority should there actually be congestion.
Resource Reservation Protocol (RSVP)

- IETF’s version of a “call setup” protocol
- Different from a virtual circuit network in several interesting ways
- VCs couple routing and resource reservation (RR), whereas Internet already has routing (decoupled from RR)
- IETF wanted to allow router failure and not lose the “call”
- IETF wanted to accommodate multicast
RSVP

- Recipient makes the actual resource reservation
  - But initiator gives the recipient the path to use
  - Resource reservation is on reverse path

- Reservation is “soft-state”
  - Network will “forget” the reservation if recipient doesn’t refresh it
  - Essentially, the recipient refreshes the reservation every minute or so!
An aside: soft-state

- Internet community was (still is???) big on the notion of soft state
- Idea is to allow control state to “age” (timeout) rather than require explicit deletion of state
- More robust, because if state creator crashes, state goes away naturally
- Simpler, because only need state create commands
An aside: soft-state

- Can be a nice principle if functions degrade gracefully rather than stop working when state disappears
  - RSVP: packet still forwarded, but just without requested QoS
- Or if actual usage (user data packets) is what refreshes the state
  - LRU caching is a form of soft state
- A nice design principle to keep in mind, but don’t be religious about it…
RSVP with multicast

Diagram: Initiator sends a message labeled 'a' to nodes b, c, d, e, and f. Nodes b, c, and d send back messages labeled 'ad', 'aded', and 'adhs' respectively. The final node i receives a 'PATH Message'.
RSVP with multicast

Initiator

Individual Receiver can drop out if reservation fails.

Reserve Messages (combined as go up-tree!)
Intserv failed in the commercial marketplace

- Scaling issues
  - Core routers can’t handle so much flow state
  - Rather spend energy on high speed (rightfully)
- Lack of business model?
- Requires buy-in from too many communities
  - ISPs, OS vendors, application developers
Differentiated Services (DiffServ)

- A more modest (and realistic) proposal from IETF
- No resources reservations
- No per-flow handling
- Simply define a smallish number of service classes, encoded in IP’s ToS bits
- These bits can be set by ISP edge routers, handled by internal routers according to ISP policies
Example Diffserv deployment model

Customer A

Customer B

Customer C

Set Diffserv Tos mapping

Execute Diffserv Queuing

Customer A

Customer B

Customer C
Service classes reflect those of Intserv

- **EF** ( Expedited Forwarding )
  - For highly delay sensitive and intolerant apps

- **AF** ( Assured Forwarding )
  - To give “high priority” traffic the effect of a lightly loaded network
  - 12 classes of this
Several approaches to AF

- Weighted RED (or RIO: Red with In and Out)
  - Different drop thresholds for different classes
- WFQ
- Combinations of these
Status of Diffserv

- I’ve seen it defined for cellular wireless data networks
  - Where there is a clear bottleneck and need for differentiated services
- Certainly people believe that this is the best usage of the IP ToS bits
- Not aware that this has taken off for backbone services
What if you don’t need TCP’s reliability/sequencing, but want to be TCP friendly?
- A BW-flexible realtime video that can tolerate some packet loss
- TCP behavior can be described by an equation
  - Some time called “equation-based congestion control”
Approaches for TCP-friendly congestion control

- Round-trip delay $R$
- Packet size $s$
- Loss event rate $p$ (receiver feedback every RTT)
- Retransmission timeout $t_{RTO} \sim 4R$

$$T = \frac{s}{R \sqrt{\frac{2p}{3}} + t_{RTO} (3 \sqrt{\frac{3p}{8}}) p (1 + 32p^2)}$$
Simple TCP model

- Bandwidth as function of packet loss:
  \[ B(p) = \frac{1}{RTT} \sqrt{\frac{3}{2bp}} + o(1/\sqrt{p}) \]

- Assumes triple-duplicate-ACK triggering retransmission
- Does not take timeout into account
- Model: single saturated TCP pumping data into bottleneck
  - other flows only modeled through packet loss
TFRC

- Defines an algorithm for
  - Measuring loss at receiver
  - Feeding back that info to sender
  - Measuring RTT at sender
  - Adjusting send rate accordingly