Introduction to Computer Networks

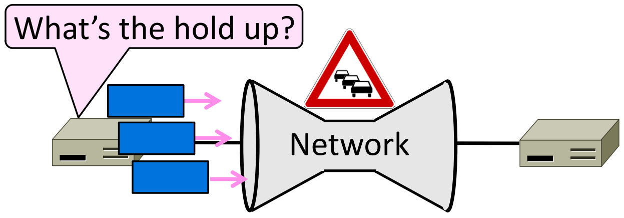
Congestion Overview
(§6.3, §6.5.10)

Computer Science & Engineering

UNIVERSITY *of* WASHINGTON

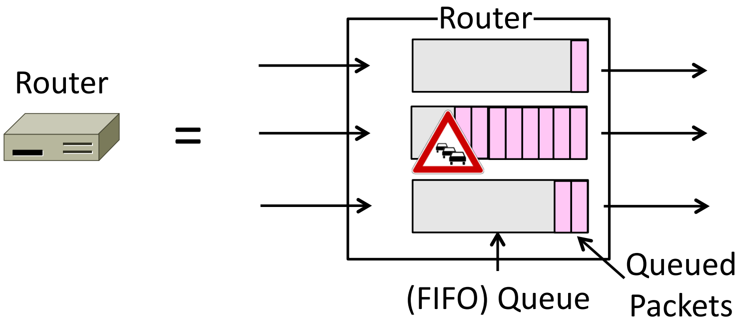
Topic

- Understanding congestion, a "traffic jam" in the network
    - Later we will learn how to control it



Nature of Congestion

- Simplified view of per port output queues
  - Typically FIFO (First In First Out), discard when full

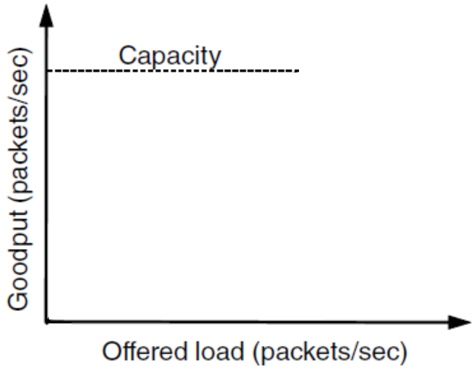


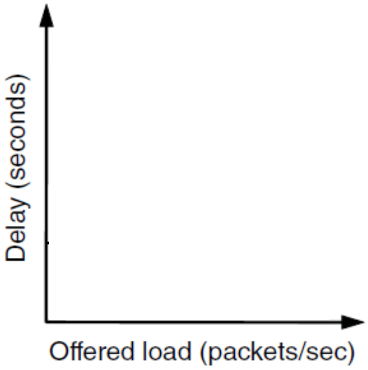
Nature of Congestion (2)

- Queues help by absorbing bursts when input > output rate
- But if input > output rate persistently, queue will overflow
  - This is congestion
- Congestion is a function of the traffic patterns – can occur even if every link have the same capacity

Effects of Congestion

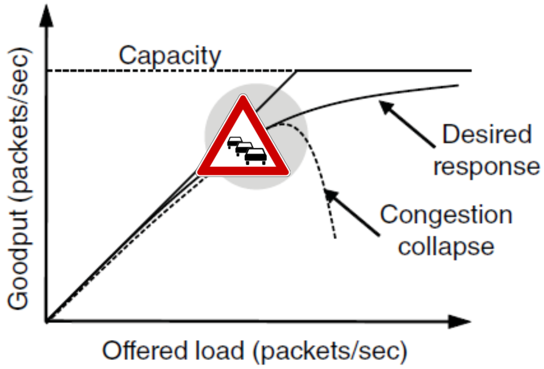
- What happens to performance as we increase the load?

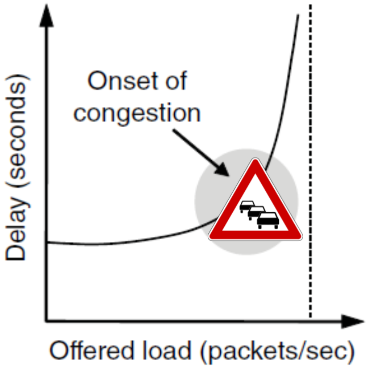




Effects of Congestion (2)

- What happens to performance as we increase the load?





Effects of Congestion (3)

- As offered load rises, congestion occurs as queues begin to fill:
  - Delay and loss rise sharply with more load
  - Throughput falls below load (due to loss)
  - Goodput may fall below throughput (due to spurious retransmissions)
- None of the above is good!
  - Want to operate network just ore the onset of congestion

Bandwidth Allocation

- Important task for network is to allocate its capacity to senders
  - Good allocation is efficient and fair
- <u>Efficient</u> means most capacity is used but there is no congestion
- <u>Fair</u> means every sender gets a reasonable share the network

Bandwidth Allocation (2)

- Why is it hard? (Just split equally!)
  - Number of senders and their offered load is constantly changing
  - Senders may lack capacity in different parts of the network
  - Network is distributed; no single party has an overall picture of its state

Bandwidth Allocation (3)

- Key observation:
  - In an effective solution, Transport and Network layers must work together
- Network layer witnesses congestion
  - Only it can provide direct feedback
- Transport layer causes congestion
  - Only it can reduce offered load

Bandwidth Allocation (4)

- Solution context:
  - Senders adapt concurrently based on their own view of the network
  - Design this adaption so the network usage as a whole is efficient and fair
  - Adaption is continuous since offered loads continue to change over time

---

Introduction to Computer Networks

Fairness of Bandwidth Allocation (§6.3.1)

Computer Science & Engineering

UNIVERSITY *of* WASHINGTON

Topic

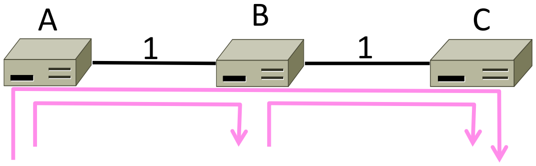
- What's a "fair" bandwidth allocation?
  - The max-min fair allocation

Recall

- We want a good bandwidth allocation to be fair and efficient
  - Now we learn what fair means
- Caveat: in practice, efficiency is more important than fairness

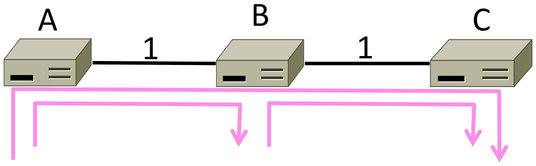
Efficiency vs. Fairness

- Cannot always have both!
  - Example network with traffic A→B, B→C and A→C
  - How much traffic can we carry?



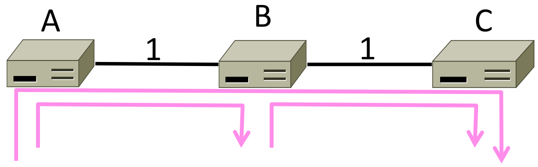
Efficiency vs. Fairness (2)

- If we care about fairness:
  - Give equal bandwidth to each flow
  - A→B: ½ unit, B→C: ½, and A→C, ½
  - Total traffic carried is 1 ½ units



Efficiency vs. Fairness (3)

- If we care about efficiency:
  - Maximize total traffic in network
  - A→B: 1 unit, B→C: 1, and A→C, 0
  - Total traffic rises to 2 units!

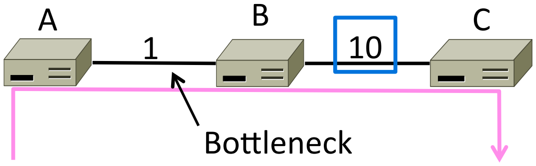


The Slippery Notion of Fairness

- Why is "equal per flow" fair anyway?
  - A→C uses more network resources (two links) than A→B or B→C
  - Host A sends two flows, B sends one
- Not productive to seek exact fairness
  - More important to avoid <u>starvation</u>
  - "Equal per flow" is good enough

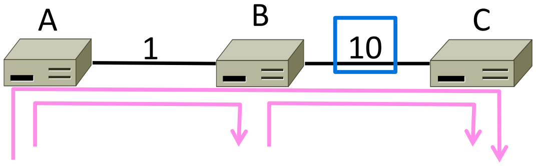
Generalizing "Equal per Flow"

- Bottleneck for a flow of traffic is the link that limits its bandwidth
  - Where congestion occurs for the flow
  - For A→C, link A–B is the bottleneck



Generalizing "Equal per Flow" (2)

- Flows may have different bottlenecks
  - For A→C, link A–B is the bottleneck
  - For B→C, link B–C is the bottleneck
  - Can no longer divide links equally …

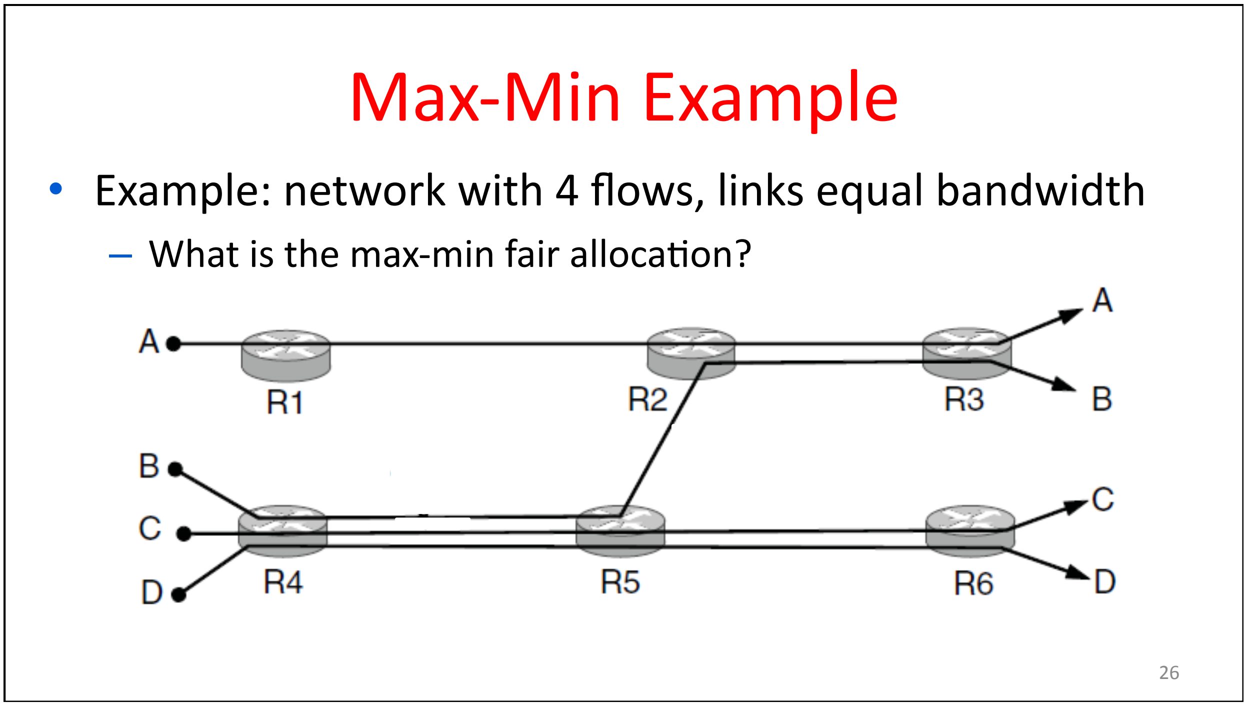


Max-Min Fairness

- Intuitively, flows bottlenecked on a link get an equal share of that link
- <u>Max-min fair allocation</u> is one that:
  - Increasing the rate of one flow will decrease the rate of a smaller flow
  - This “maximizes the minimum” flow

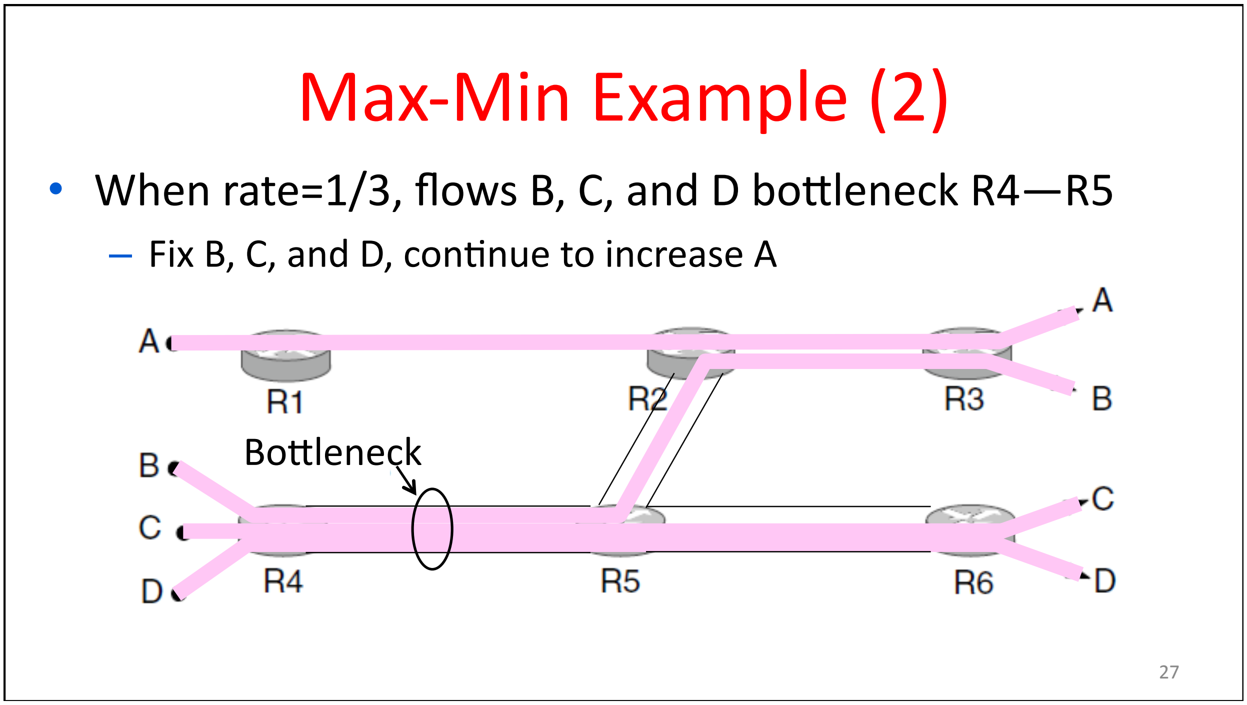
Max-Min Fairness (2)

- To find it given a network, imagine “pouring water into the network”
  1. Start with all flows at rate 0
  2. Increase the flows until there is a new bottleneck in the network
  3. Hold fixed the rate of the flows that are bottlenecked
  4. Go to step 2 for any remaining flows



Max-Min Example

- Example: network with 4 flows, links equal bandwidth
  - What is the max-min fair allocation?

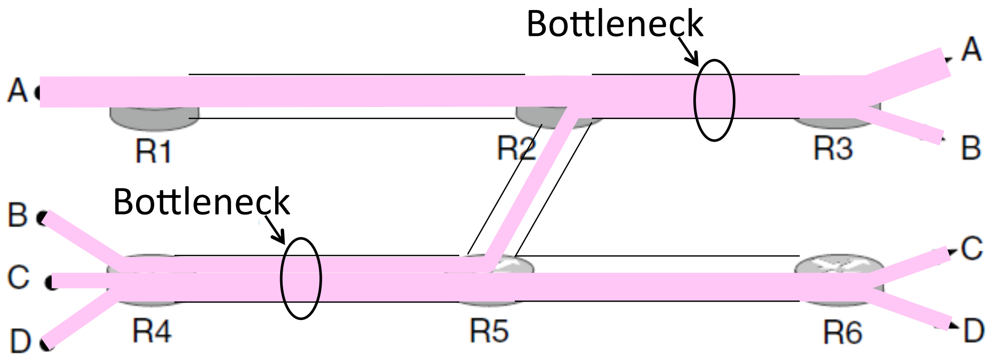


Max-Min Example (2)

- When rate=1/3, flows B, C, and D bottleneck R4—R5
  - Fix B, C, and D, continue to increase A

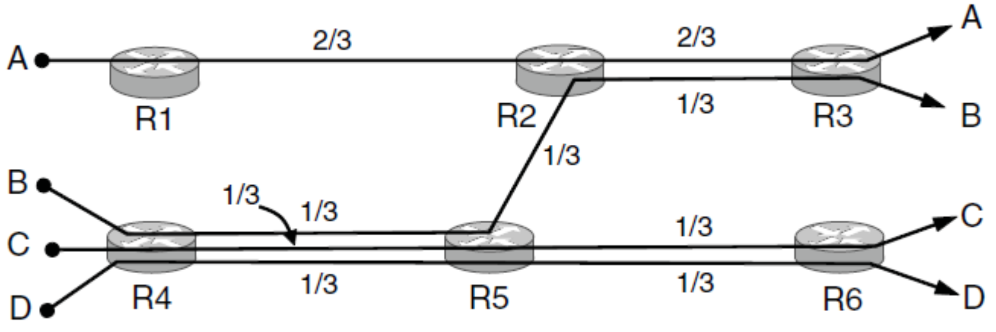
Max-Min Example (3)

- When rate=2/3, flow A bottlenecks R2—R3. Done.



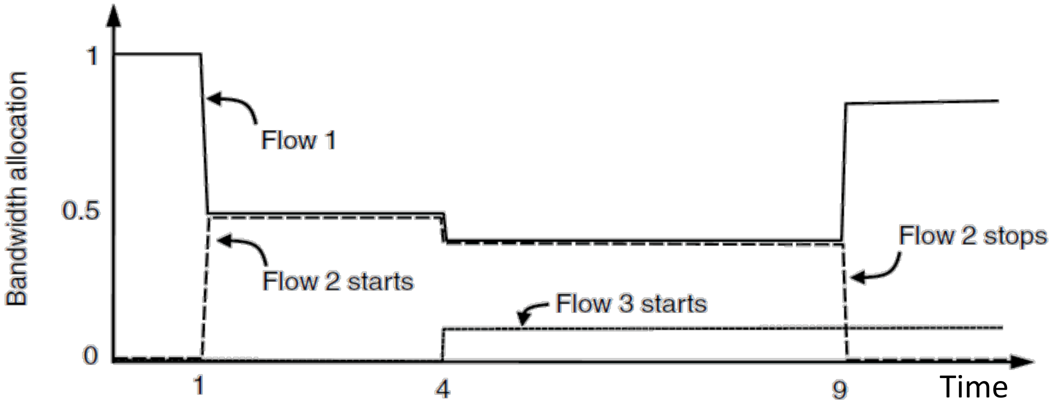
Max-Min Example (4)

- End with A=2/3, B, C, D=1/3, and R2—R3, R4—R5 full
  - Other links have extra capacity that can't be used

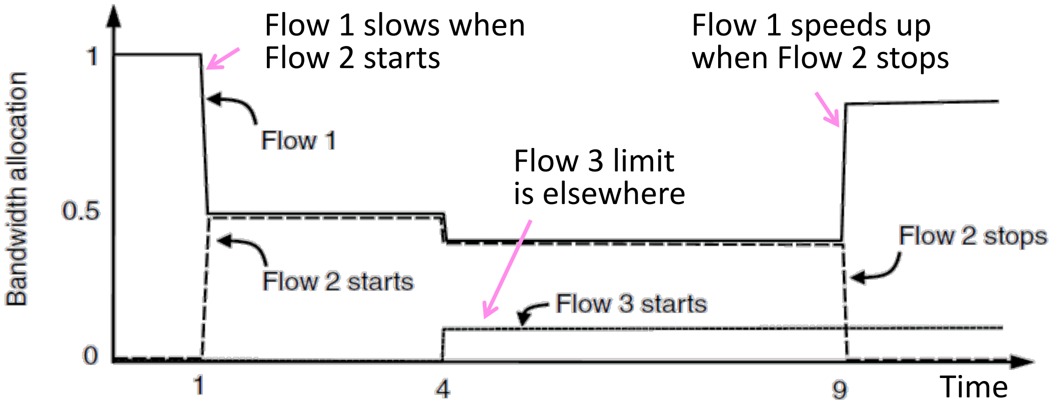


Adapting over Time

- Allocation changes as flows start and stop



Adapting over Time (2)



Flow 1 slows when
Flow 2 starts

Flow 1 speeds up
when Flow 2 stops

Flow 3 limit
is elsewhere

Introduction to Computer Networks

Additive Increase Multiplicative Decrease (AIMD) (§6.3.2)

Computer Science & Engineering

UNIVERSITY *of* WASHINGTON

Recall

- Want to allocate capacity to senders
  - Network layer provides feedback
  - Transport layer adjusts offered load
  - A good allocation is efficient and fair
- How should we perform the allocation?
  - Several different possibilities …

Bandwidth Allocation Models

- Open loop versus closed loop
  - Open: reserve bandwidth before use
  - Closed: use feedback to adjust rates
- Host versus Network support
  - Who sets/enforces allocations?
- Window versus Rate based
  - How is allocation expressed?

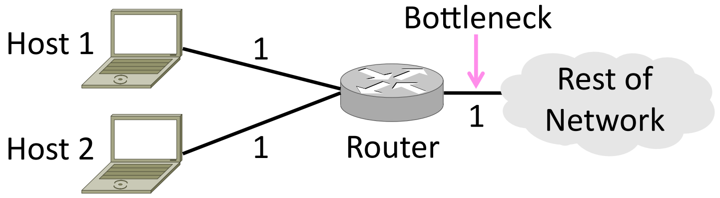
TCP is a closed loop, host-driven, and window-based

Additive Increase Multiplicative Decrease

- AIMD is a control law hosts can use to reach a good allocation
  - Hosts additively increase rate while network is not congested
  - Hosts multiplicatively decrease rate when congestion occurs
  - Used by TCP
- Let's explore the AIMD game …

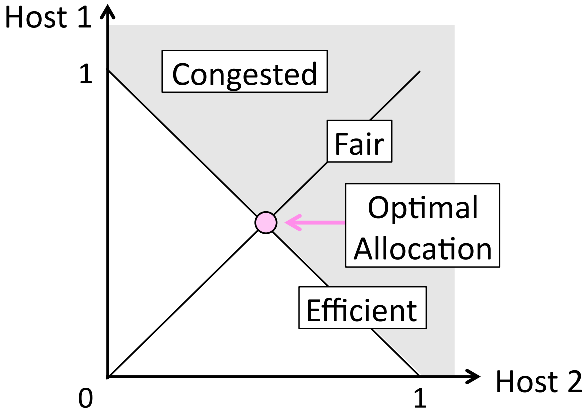
AIMD Game

- Hosts 1 and 2 share a bottleneck
  - But do not talk to each other directly
- Router provides binary feedback
  - Tells hosts if network is congested



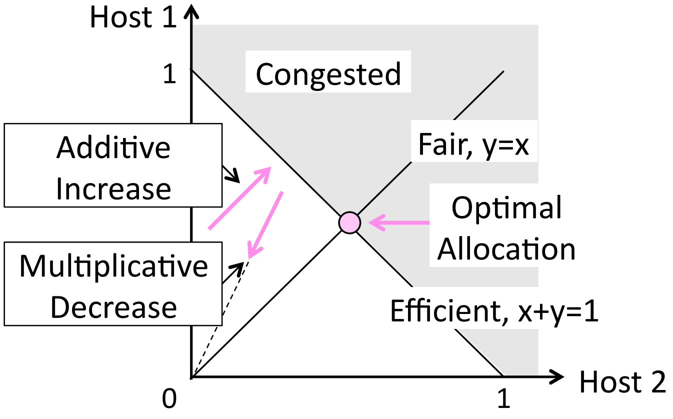
AIMD Game (2)

- Each point is a possible allocation



AIMD Game (3)

- AI and MD move the allocation



Host 1

1

Congested

Fair, y=x

Additive Increase

Optimal Allocation

Multiplicative Decrease

Efficient, x+y=1

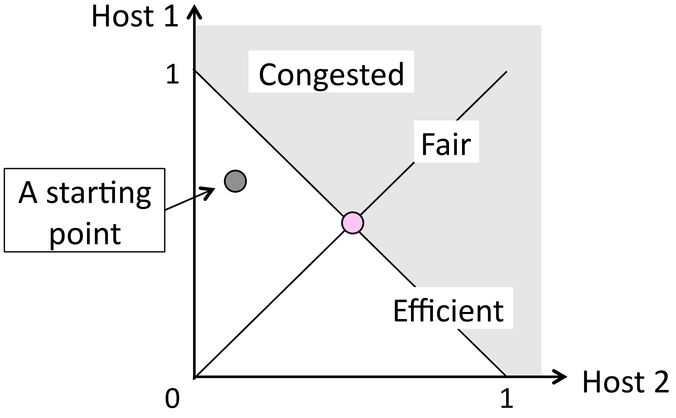
0

1

Host 2

AIMD Game (4)

- Play the game!



Host 1

1

Congested

Fair

A starting point

Efficient

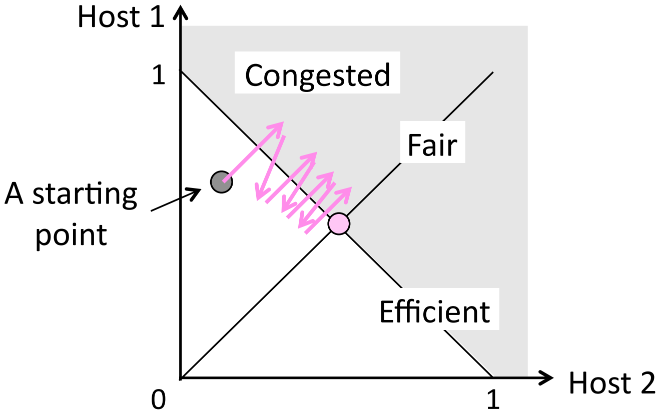
0

1

Host 2

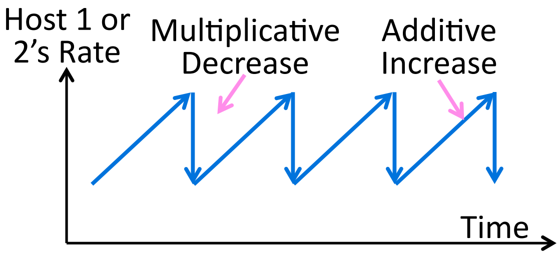
AIMD Game (5)

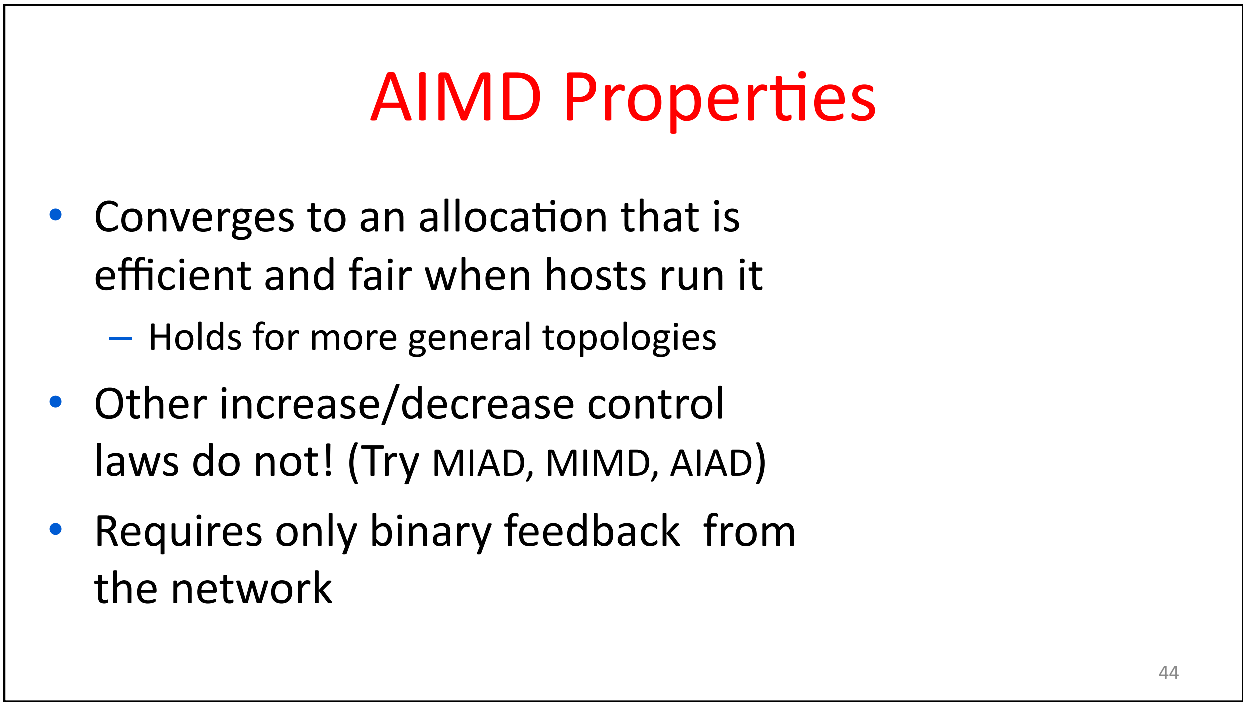
- Always converge to good allocation!



AIMD Sawtooth

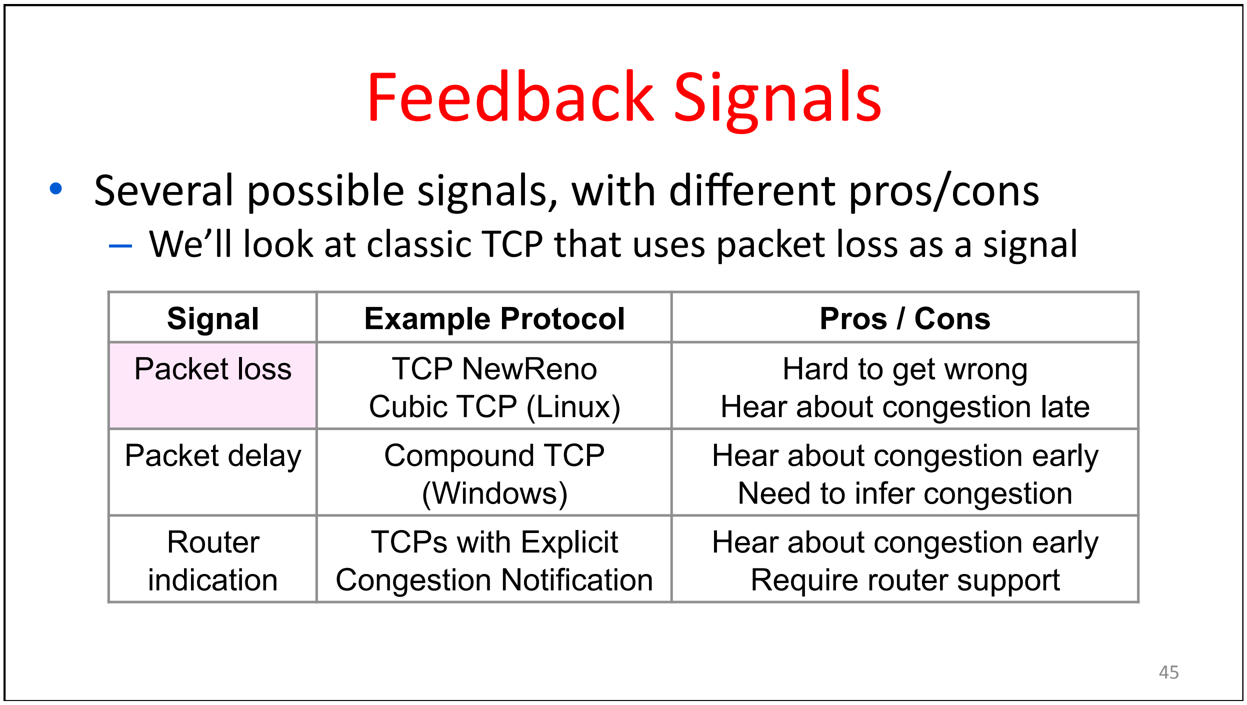
- Produces a “sawtooth” pattern over time for rate of each host
  - This is the TCP sawtooth (later)





AIMD Properties

- Converges to an allocation that is efficient and fair when hosts run it
  - Holds for more general topologies
- Other increase/decrease control laws do not! (Try MIAD, MIMD, AIAD)
- Requires only binary feedback from the network



Feedback Signals

- Several possible signals, with different pros/cons
  - We'll look at classic TCP that uses packet loss as a signal

Signal	Example Protocol	Pros / Cons
Packet loss	TCP NewReno Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	Compound TCP (Windows)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification	Hear about congestion early Require router support

TCP Tahoe/Reno

- Avoid congestion collapse without changing routers (or even receivers)
- Idea is to fix timeouts and introduce a congestion window (cwnd) over the sliding window to limit queues/loss
- TCP Tahoe/Reno implements AIMD by adapting cwnd using packet loss as the network feedback signal

TCP Tahoe/Reno (2)

- TCP behaviors we will study:
  - ACK clocking
  - Adaptive timeout (mean and variance)
  - Slow-start
  - Fast Retransmission
  - Fast Recovery
- Together, they implement AIMD

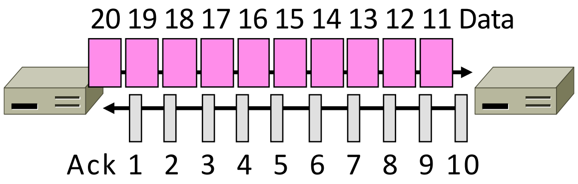
Introduction to Computer Networks

TCP Ack Clocking (§6.5.10)

Computer Science & Engineering
UNIVERSITY of WASHINGTON

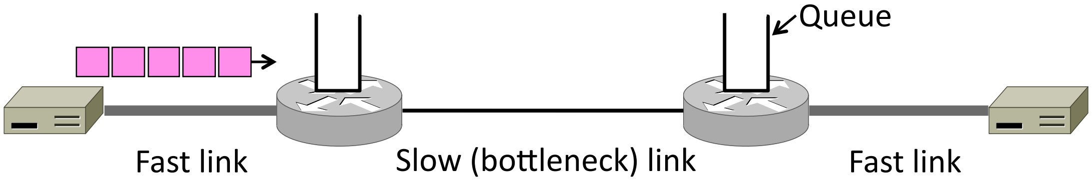
Sliding Window ACK Clock

- Each in-order ACK advances the sliding window and lets a new segment enter the network
  - ACKs “clock” data segments



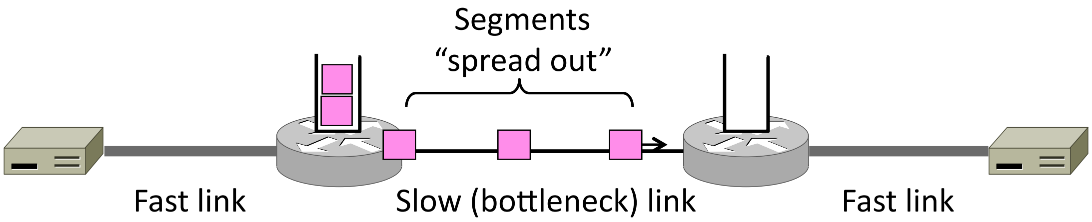
Benefit of ACK Clocking

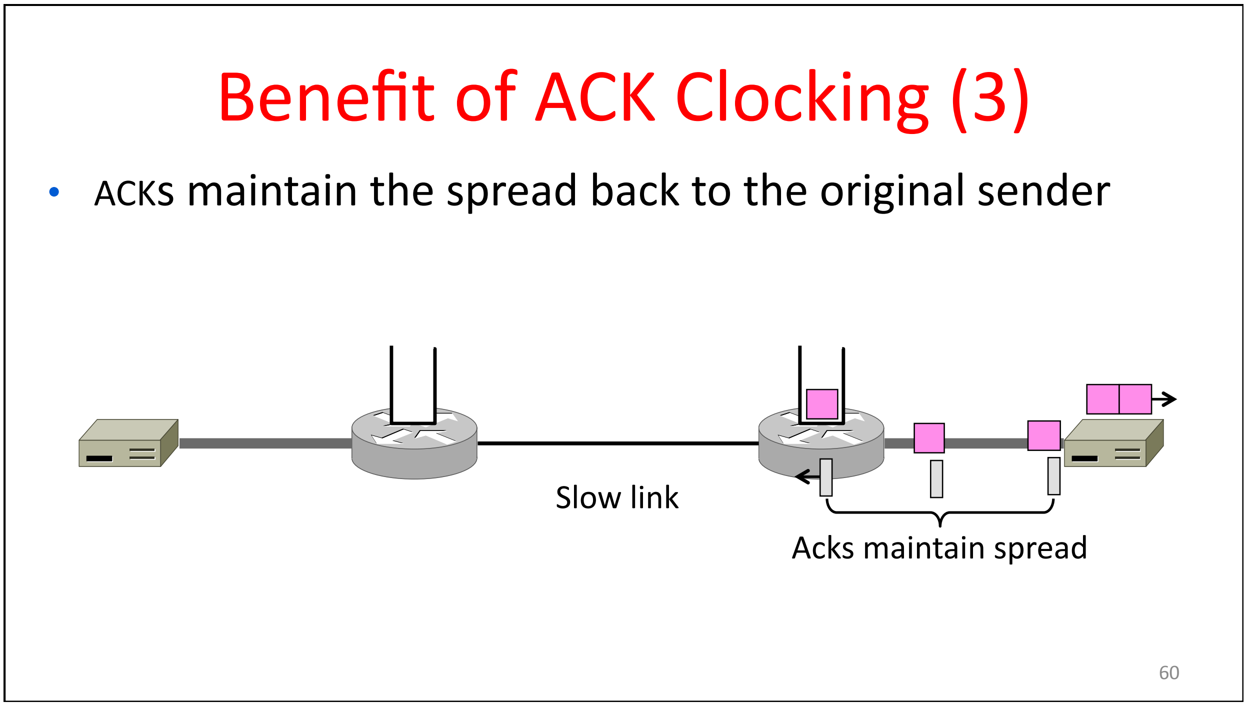
- Consider what happens when sender injects a burst of segments into the network



Benefit of ACK Clocking (2)

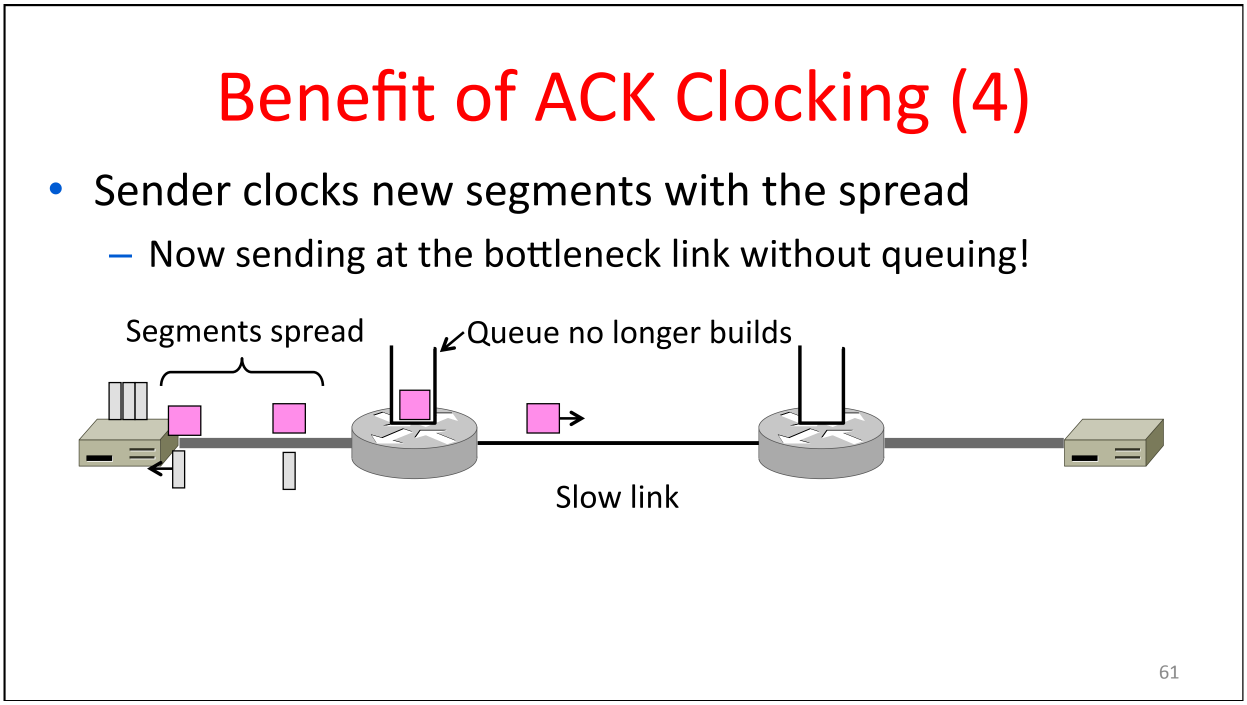
- Segments are buffered and spread out on slow link





Benefit of ACK Clocking (3)

- ACKs maintain the spread back to the original sender



Benefit of ACK Clocking (4)

- Sender clocks new segments with the spread
  - Now sending at the bottleneck link without queuing!

Benefit of ACK Clocking (4)

- Helps the network run with low levels of loss and delay!
- The network has smoothed out the burst of data segments
- ACK clock transfers this smooth timing back to the sender
- Subsequent data segments are not sent in bursts so do not queue up in the network

---

Introduction to Computer Networks

TCP Slow Start (§6.5.10)

Computer Science & Engineering

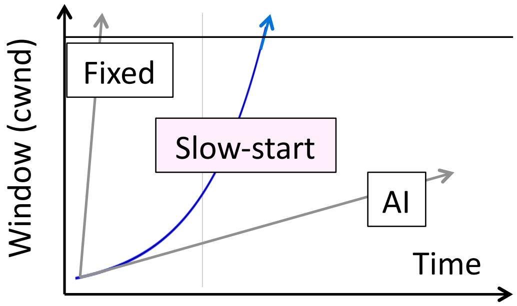
UNIVERSITY *of* WASHINGTON

TCP Startup Problem

- We want to quickly near the right rate, cwnd$_{IDEAL}$, but it varies greatly
  - Fixed sliding window doesn't adapt and is rough on the network (loss!)
  - AI with small bursts adapts cwnd gently to the network, but might take a long time to become efficient

Slow-Start Solution

- Start by doubling cwnd every RTT
  - Exponential growth (1, 2, 4, 8, 16, …)
  - Start slow, quickly reach large values

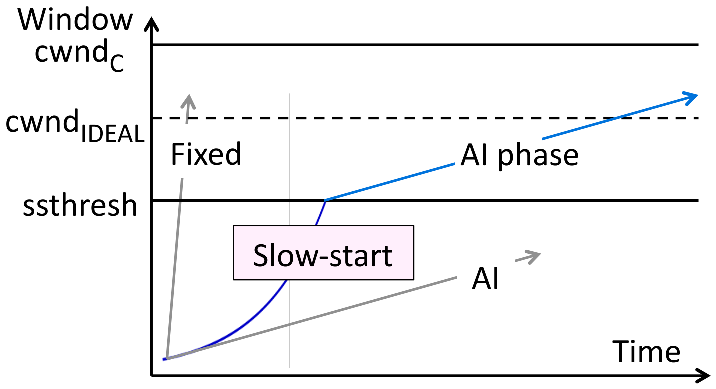


Slow-Start Solution (2)

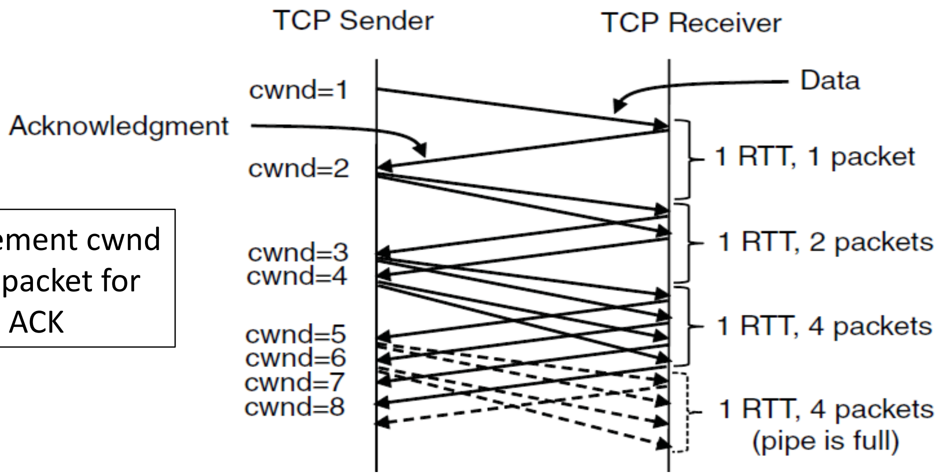
- Eventually packet loss will occur when the network is congested
  - Loss timeout tells us cwnd is too large
  - Next time, switch to AI beforehand
  - Slowly adapt cwnd near right value
- In terms of cwnd:
  - Expect loss for $cwnd_C \approx 2BD+queue$
  - Use ssthresh = $cwnd_C/2$ to switch to AI

Slow-Start Solution (3)

- Combined behavior, after first time
  - Most time spend near right value

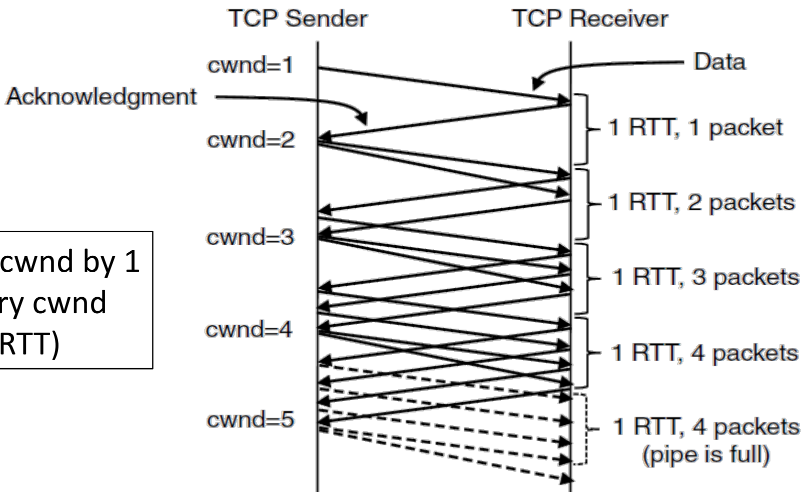


Slow-Start (Doubling) Timeline



Increment cwnd by 1 packet for each ACK

Additive Increase Timeline



Increment cwnd by 1 packet every cwnd ACKs (or 1 RTT)

TCP Tahoe (Implementation)

- Initial slow-start (doubling) phase
  - Start with cwnd = 1 (or small value)
  - cwnd += 1 packet per ACK

- Later Additive Increase phase
  - cwnd += 1/cwnd packets per ACK
  - Roughly adds 1 packet per RTT

- Switching threshold (initially infinity)
  - Switch to AI when cwnd > ssthresh
  - Set ssthresh = cwnd/2 after loss
  - Begin with slow-start after timeout

Timeout Misfortunes

- Why do a slow-start after timeout?
  - Instead of MD cwnd (for AIMD)

Timeout Misfortunes

- Why do a slow-start after timeout?
  - Instead of MD cwnd (for AIMD)
- Timeouts are sufficiently long that the ACK clock will have run down
  - Slow-start ramps up the ACK clock
- We need to detect loss before a timeout to get to full AIMD
  - Done in TCP Reno

Introduction to Computer Networks

TCP Fast Retransmit / Fast Recovery (§6.5.10)

Computer Science & Engineering

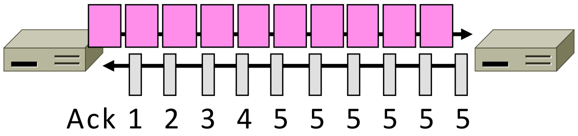
UNIVERSITY *of* WASHINGTON

Inferring Loss from ACKs

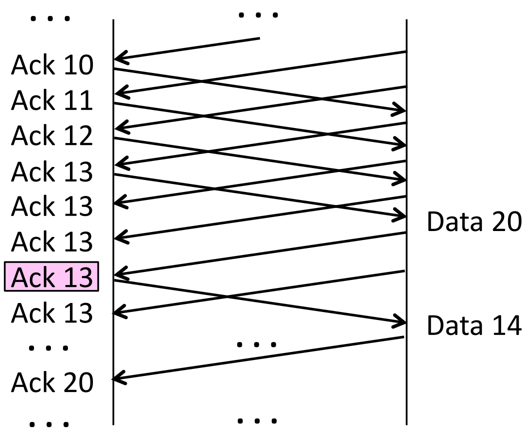
- TCP uses a cumulative ACK
  - Carries highest in-order seq. number
  - Normally a steady advance
- Duplicate ACKs give us hints about what data hasn't arrived
  - Tell us some new data did arrive, but it was not next segment
  - Thus the next segment may be lost

Fast Retransmit

- Treat three duplicate ACKs as a loss
  - Retransmit next expected segment
  - Some repetition allows for reordering, but still detects loss quickly



Fast Retransmit (2)



Data 14 was lost earlier, but got 15 to 20

Third duplicate ACK, so send 14

Retransmission fills in the hole at 14

ACK jumps after loss is repaired

Fast Retransmit (3)

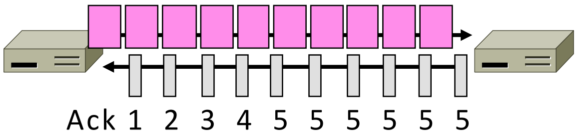
- It can repair single segment loss quickly, typically before a timeout
- However, we have quiet time at the sender/receiver while waiting for the ACK to jump
- And we still need to MD cwnd …

Inferring Non-Loss from ACKs

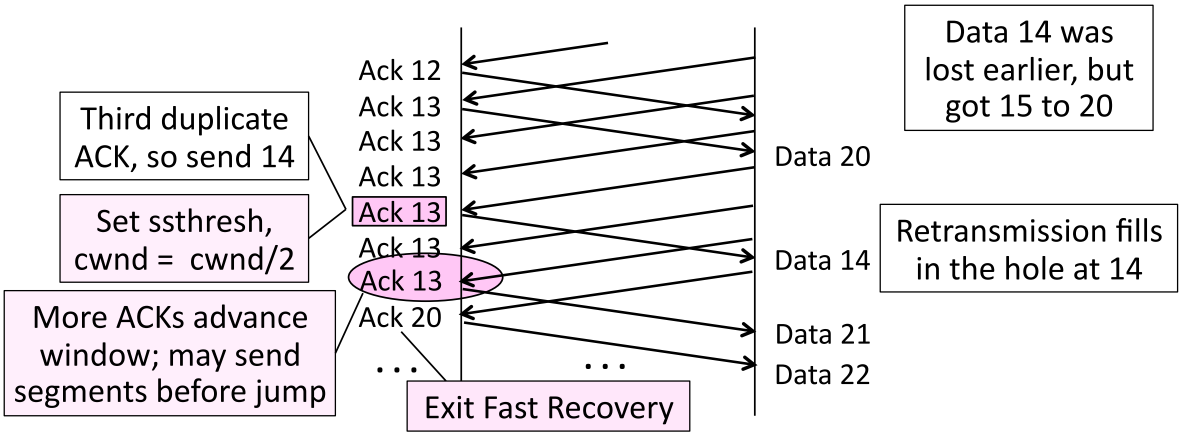
- Duplicate ACKs also give us hints about what data has arrived
  - Each new duplicate ACK means that some new segment has arrived
  - It will be the segments after the loss
  - Thus advancing the sliding window will not increase the number of segments stored in the network

Fast Recovery

- First fast retransmit, and MD cwnd
- Then pretend further duplicate ACKs are the expected ACKs
  - Lets new segments be sent for ACKs
  - Reconcile views when the ACK jumps



Fast Recovery (2)



Third duplicate ACK, so send 14

Set ssthresh, cwnd = cwnd/2

More ACKs advance window; may send segments before jump

Ack 12
Ack 13
Ack 13
Ack 13
Ack 13
Ack 13
Ack 13
Ack 20
. . .

Exit Fast Recovery

. . .

Data 20
Data 14
Data 21
Data 22

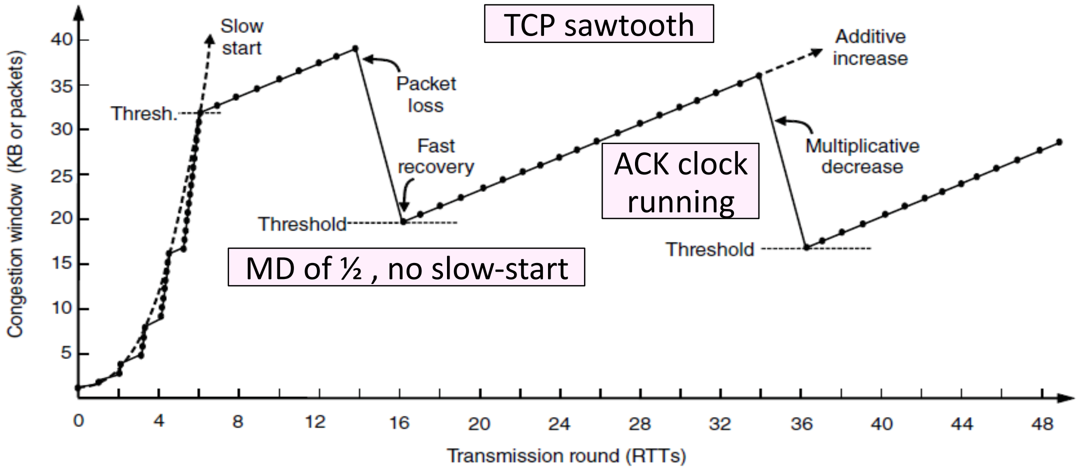
Data 14 was lost earlier, but got 15 to 20

Retransmission fills in the hole at 14

Fast Recovery (3)

- With fast retransmit, it repairs a single segment loss quickly and keeps the ACK clock running
- This allows us to realize AIMD
  - No timeouts or slow-start after loss, just continue with a smaller cwnd
- TCP Reno combines slow-start, fast retransmit and fast recovery
  - Multiplicative Decrease is ½

TCP Reno



TCP sawtooth

ACK clock running

MD of ½ , no slow-start

TCP Reno, NewReno, and SACK

- Reno can repair one loss per RTT
  - Multiple losses cause a timeout
- NewReno further refines ACK heuristics
  - Repairs multiple losses without timeout
- SACK is a better idea
  - Receiver sends ACK ranges so sender can retransmit without guesswork

Introduction to Computer Networks

Explicit Congestion Notification
(§5.3.4, §6.5.10)

Computer Science & Engineering
UNIVERSITY of WASHINGTON

Congestion Avoidance vs. Control

- Classic TCP drives the network into congestion and then recovers
  - Needs to see loss to slow down
- Would be better to use the network but avoid congestion altogether!
  - Reduces loss and delay
- But how can we do this?

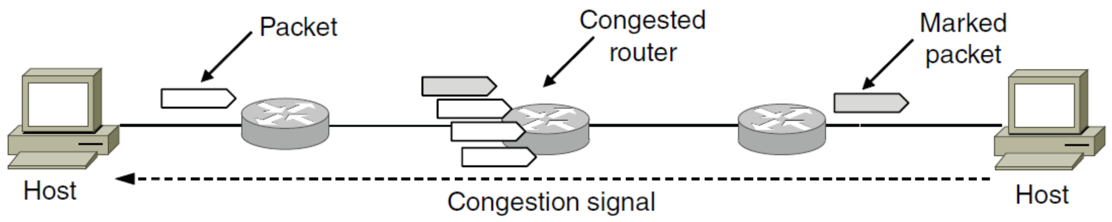
Feedback Signals

- Delay and router signals can let us avoid congestion

Signal	Example Protocol	Pros / Cons
Packet loss	Classic TCP Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	Compound TCP (Windows)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification	Hear about congestion early Require router support

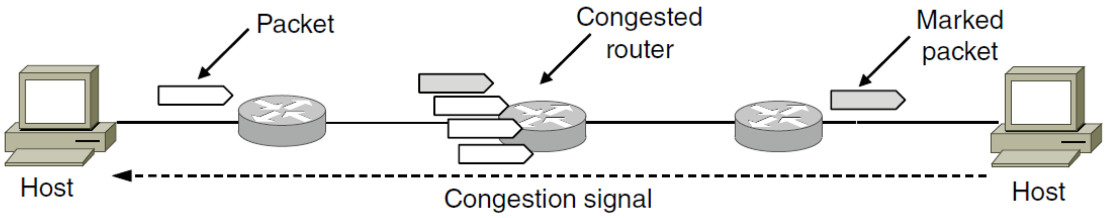
ECN (Explicit Congestion Notification)

- Router detects the onset of congestion via its queue
  - When congested, it marks affected packets (IP header)



ECN (2)

- Marked packets arrive at receiver; treated as loss
  - TCP receiver reliably informs TCP sender of the congestion



ECN (3)

- Advantages:
  - Routers deliver clear signal to hosts
  - Congestion is detected early, no loss
  - No extra packets need to be sent
- Disadvantages:
  - Routers and hosts must be upgraded

Introduction to Computer Networks

TCP Variants

Computer Science & Engineering

UNIVERSITY *of* WASHINGTON

TCP Variants

- There are many different strains of TCP including:
  - Loss-based congestion control: Reno, BIC, Cubic
  - Delay-based congestion control: Vegas, Veno, Westwood
  - High-speed congestion control: Scalable, HighSpeed, HTCP

Delay Based Congestion Control

- Basic idea:
  - Before packet loss occurs, detect the early stage of congestion in the routers between source and destination
  - Additively decrease the sending rate when incipient congestion is detected

TCP Vegas

- *Expected = cwnd/BaseRTT*
- *Actual = cwnd/RTT*
- *DIFF = (Expected-Actual)*

```
if ( $DIFF*BaseRTT < \alpha$ )
        $cwnd = cwnd + 1$
else if ( $DIFF*BaseRTT > \beta$ )
        $cwnd = cwnd - 1$
else $cwnd = cwnd$
```

BaseRTT: the minimum of all measured *RTT*

RTT: the actual round-trip time of a tagged packet

α and β are constant values
that are set by experimentation

TCP Vegas

- Modified Slow Start
  - Try to find the correct window size without incurring a loss
  - exponentially increasing its window every *other* RTT and using the other RTT to calculate *DIFF*
  - As soon as Vegas detects queue buildup during slow start, it transitions to congestion avoidance

TCP Veno

- TCP Vegas has some limitations:
  - Not robust to RTT changes
  - Does not compete well with loss-based congestion techniques
- TCP Veno is designed to address these limitations:
  - Combines Vegas with Reno
  - Exponential start as in Reno
  - Modifies additive increase/multiplicative decrease phases

TCP Veno Algorithm

- Multiplicative decrease algorithm

```
if ($DIFF*BaseRTT < \beta$)          //random loss due to bit errors is most
                                     //likely to have occurred
    $ssthresh = cwnd_{loss} * (4/5)$;
else
    $ssthresh = cwnd_{loss} /2$;     //congestive loss is most
                                     //likely to have occurred
```

TCP Veno

- Additive increase algorithm
  - Reduce increments when buffers are getting filled up; more aggressive than Vegas, but less aggressive than Reno

```
if ( $DIFF*BaseRTT < \beta$ ) // available bandwidth under-utilized
    $cwnd=cwnd+1/cwnd$ when every new ack received
else if ($DIFF*BaseRTT \geq \beta$ ) // available bandwidth fully utilized
    $cwnd=cwnd+1/cwnd$ when every other new ack received
```

TCP Westwood

Packet pair:

effective under random loss, overestimates under congestion

Packet train:

fair estimate under congestion, underestimates under random loss

- To obtain rate estimate: adapt the sample interval T_k according to congestion level
- Need to be careful about dupacks, delayed acks, etc.

High BDP Variants

- Represents a class of algorithms that are much more aggressive than traditional TCP

Traditional TCP

cwnd ← cwnd + 1/cwnd;
- if no loss was detected

cwnd ← cwnd/2;
- if a loss was detected

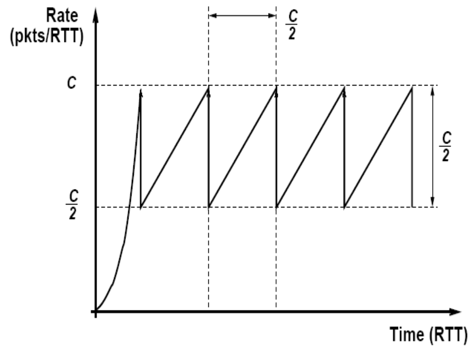
Scalable TCP

cwnd ← cwnd + 0.01;
- if no loss was detected

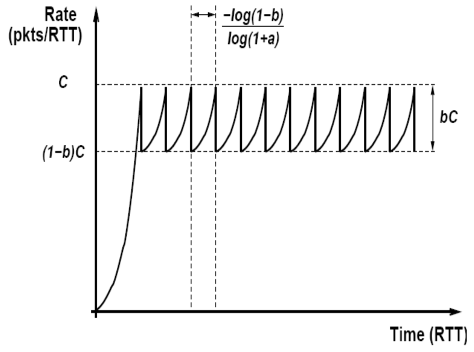
cwnd ← cwnd*0.875;
- if a loss was detected

Comparison

Traditional TCP



Scalable TCP



Cubic

- Two key modifications:
  - Cubic window growth with inflection point at congestion window at previous loss

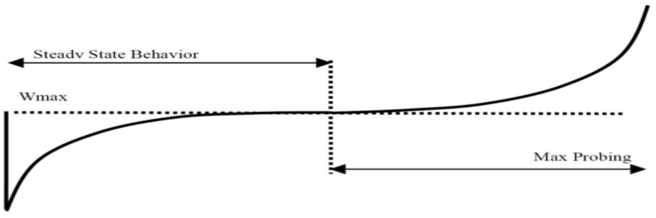


Fig. 2: The Window Growth Function of CUBIC

  - Safe exit for slow start (i.e., transition from exponential growth to linear growth)

Introduction to Computer Networks

PCP – Probe Control Protocol

Computer Science & Engineering

UNIVERSITY *of* WASHINGTON

Resource Allocation Problem

- How to allocate network bandwidth resources when multiple flows share common links?

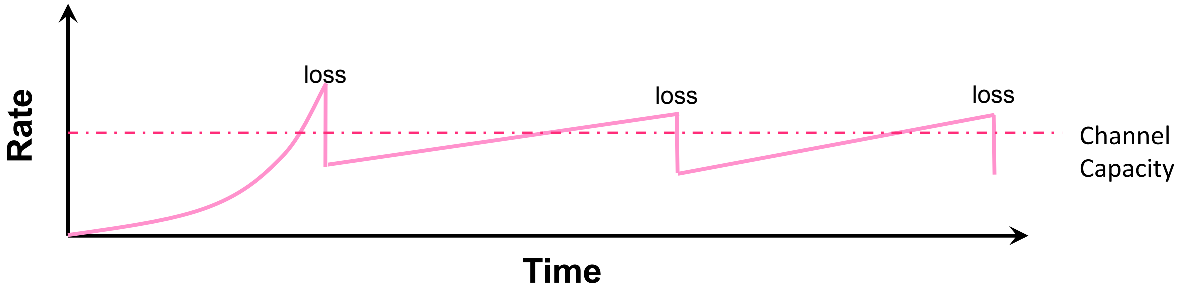
- Goals:
  - Minimize transfer time
  - Negligible packet loss & low queue variability
  - Resources are fully allocated if there is sufficient demand
  - Stable system even under high loads
  - Fairness

TCP: Endpoint Congestion Control

- Allocate resources without requiring network support
- “Try and Backoff” strategy:
  - Start with low transfer rate, ramp up rate
  - FIFO routers drop packets when queues fill up
  - Congestion inferred from packet loss
  - Endpoint responds to packet loss by throttling rate

TCP Endpoint Congestion Control

- Try and Backoff strategy:



Limits of Try-and-Backoff Strategy

- In theory, the link capacity is fully utilized for long flows, but
  - Initial ramp-up takes up most of the response time
  - Channel capacity is left unused
    - If "n" is capacity, takes log(n) steps for the initial ramp-up
    - Wasted capacity during that period: O(n log(n))
  - At the tail of the ramp-up, the rate overshoots the channel capacity
    - Causes multiple packet losses; worse with multiple flows
- Could start with higher transfer rates, but could result in higher packet loss/congestion

Network-assisted Congestion Control

1) Routers provide feedback to end-systems
   - Add TCP-specific support to routers
   - Signal end-hosts to reduce their sending rates
2) Routers explicitly allocate bandwidth to flows
   - Endpoints use a "request and set" strategy
   - Routers enforce resource limits
   - Attains flow isolation

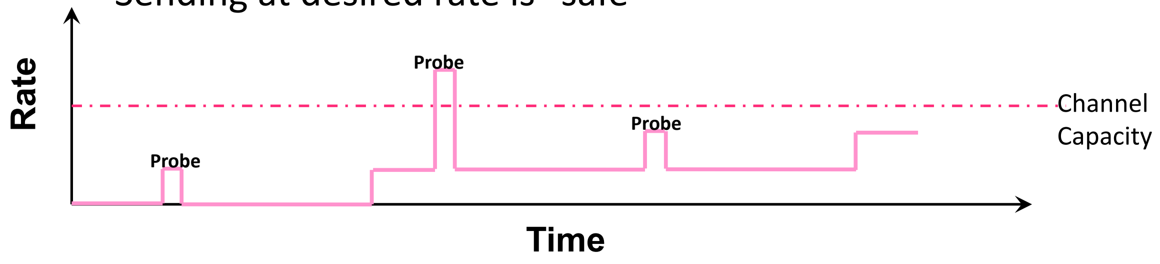
Problem: makes routers complicated and hinders adoption

Previous Work

	Endpoint	Router Support
Try and Backoff	TCP, Vegas, RAP, FastTCP, Scalable TCP, HighSpeed TCP	DecBit, ECN, RED, AQM
Request and Set	?	ATM, XCP, WFQ, RCP

Probe Control Protocol (PCP)

- Probe for required bandwidth using short, non-intrusive probes
- If bandwidth is available, send at the desired uniform rate
  - Sending at desired rate is “safe”



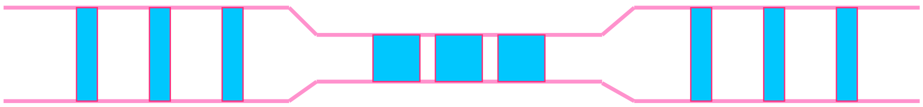
- Probe is a request, successful probe sets the sending rate, other flows cannot acquire the allocated bandwidth

PCP Mechanisms

- Probes: how to check for available bandwidth
- Probe control: how to vary the requests?
- Rate compensation: deal with queue build-ups

Probes

- Send packet train spaced at an interval to achieve desired rate
  - Currently, five packets whose size could be varied
- Check for queuing delays based on reception times



Probe Control

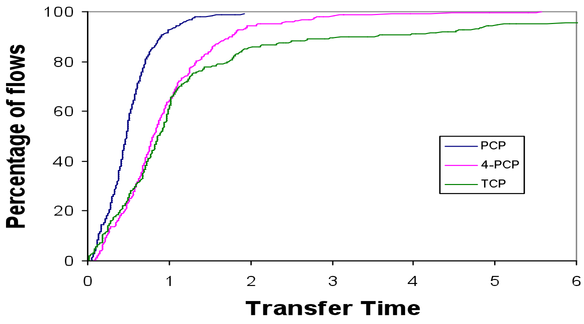
- Base protocol:
  - Start with a baseline rate (one maximum sized packet per round-trip)
  - If probe succeeds, double the requested bandwidth
  - If probe fails, halve the requested bandwidth
  - If probed rate falls below baseline rate:
    - Keep probed rate constant
    - Issue probes less frequently (exponential back-off)
- Augmented with history:
  - Endpoint keeps track of previously used rates for different paths
  - Directly jumps to probe for a rate based on history

Rate Compensations

- Queue build-ups could occur:
  - Probes, even though they are short, could result in additional queueing
  - Almost simultaneous probes could allocate the same bandwidth to two flows
  - Errors in determining success of a probe could result in too much load
- Solution: rate compensation
  - Monitor packet delays
  - Notice queue-buildups
  - Slow down the transmission rate to drain queue

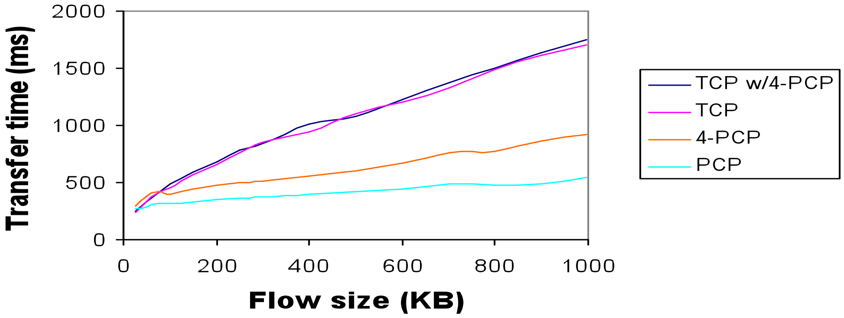
Performance

- User-level implementation tested on WAN infrastructure
  - EMULAB system, twenty nodes
  - 250KB transfers between every pair of nodes
  - PCP vs. TCP vs. four concurrent PCP transmissions



Performance

- Is PCP getting its performance benefits by being aggressive to TCP traffic?
- How does the transfer time vary with flow size?



Summary

- Smart endpoint solution that mimics centralized control
  - More suited for the current day Internet
  - Also leverages a number of hardware developments: better timers, more capable end-hosts, …
  - Combines innovations in networking software: available bandwidth measurement, delay-based rate adjustments, …

- In-kernel implementation of PCP for Linux