

Miscellaneous Topics

EE122 Fall 2012

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Materials with thanks to Jennifer Rexford, Ion Stoica, Vern Paxson and other colleagues at Princeton and UC Berkeley

Q/A on Project 3

Today's Lecture: Dim Sum of Design

- Quality-of-Service
- Multicast

Announcements...

- Wireless addendum
- Advanced CC addendum

Extending the Internet Service Model

Internet Service Model

- Best-Effort: everyone gets the same service
 - No guarantees

Unicast: each packet goes to single destination

Extending the service model

Better than best-effort: Quality of Service (QoS)

More than one receiver: Multicast

Quality of Service (QoS)

Summary of Current QoS Mechanisms

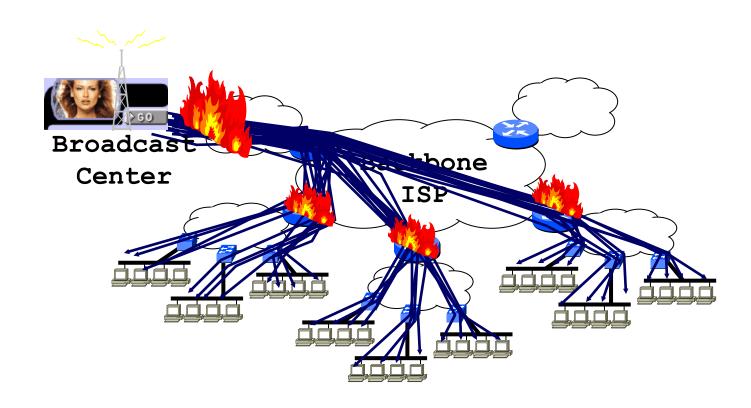
Blah, priority scheduling, blah, blah



Motivating Example: Internet Radio

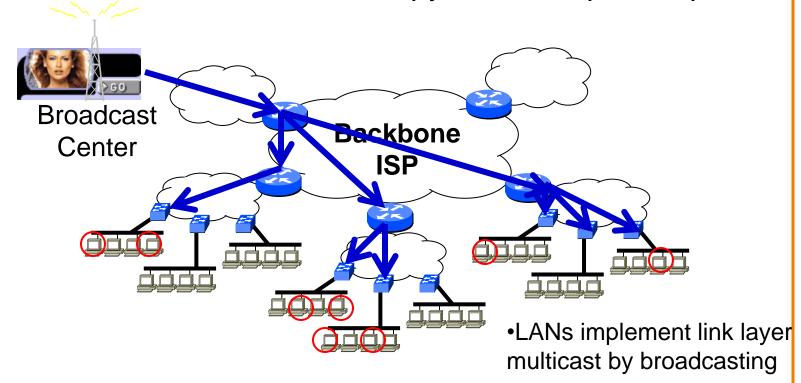
- Internet concert
 - More than 100,000 simultaneous online listeners
 - Could we do this with parallel unicast streams?
- Bandwidth usage
 - If each stream was 1Mbps, concert requires > 100Gbps
- Coordination
 - Hard to keep track of each listener as they come and go
- Multicast addresses both problems....

Unicast approach does not scale...



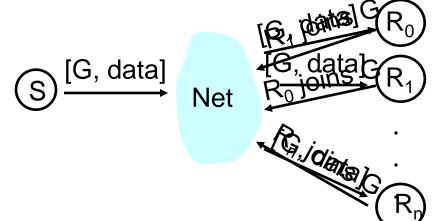
Instead build data replication trees

- Copy data at routers
- At most one copy of a data packet per link



- Routers keep track of groups in real-time
- Routers compute trees and forward packets along them

Multicast Service Model



- Receivers join multicast group with address G
- Sender(s) send data to address G
- Network routes data to each of the receivers
- Multicast both delivery and rendezvous mechanism
 - Senders don't know list of receivers
 - The latter is often more important than the former

Multicast and Layering

- Multicast can be implemented at different layers
 - Link layer
 - e.g. Ethernet multicast
 - Network layer
 - e.g. IP multicast
 - Application layer
 - e.g. End system multicast
- Each layer has advantages and disadvantages
 - Link: easy to implement, limited scope
 - IP: global scope, efficient, but hard to deploy
 - Application: less efficient, easier to deploy [not covered]

Multicast Implementation Issues

How is join implemented?

How is send implemented?

How much state is kept and who keeps it?

Link Layer Multicast

- Join group at multicast address G
 - NIC normally only listens for packets sent to unicast address A and broadcast address B
 - After being instructed to join group G, NIC also listens for packets sent to multicast address G
- Send to group G
 - Packet is flooded on all LAN segments, like broadcast
- Scalability:
 - State: Only host NICs keep state about who has joined
 - Bandwidth: Requires broadcast on all LAN segments
- Limitation: just over single LAN

Network Layer (IP) Multicast

- Performs inter-network multicast routing
 - Relies on link layer multicast for intra-network routing
- Portion of IP address space reserved for multicast
 - -2²⁸ addresses for entire Internet
- Open group membership
 - Anyone can join (sends IGMP message)
 - Internet Group Management Protocol
 - Privacy preserved at application layer (encryption)
- Anyone can send to group
 - Even nonmembers

How Would YOU Design this?

• 5 Minutes....

IP Multicast Routing

- Intra-domain (know the basics here)
 - Source Specific Tree: Distance Vector Multicast Routing Protocol (DVRMP)
 - -Shared Tree: Core Based Tree (CBT)

Inter-domain [not covered]

Distance Vector Multicast Routing Protocol

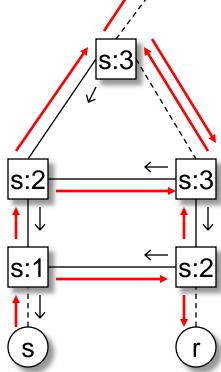
- Elegant extension to DV routing
 - Using reverse paths!
- Use shortest path DV routes to determine if link is on the source-rooted spanning tree
- Three steps in developing DVRMP
 - Reverse Path Flooding
 - Reverse Path Broadcasting
 - Truncated Reverse Path Broadcasting (pruning)

Reverse Path Flooding (RPF)

If incoming link is shortest path to source

Send on all links except incoming

Otherwise, drop

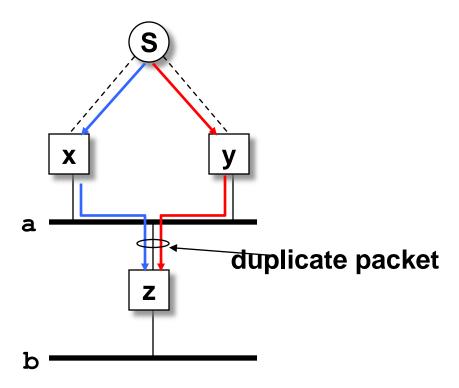


Issues: (fixed with RPB)

- Some links (LANs) may receive multiple copies
- Every link receives each multicast packet

Other Problems

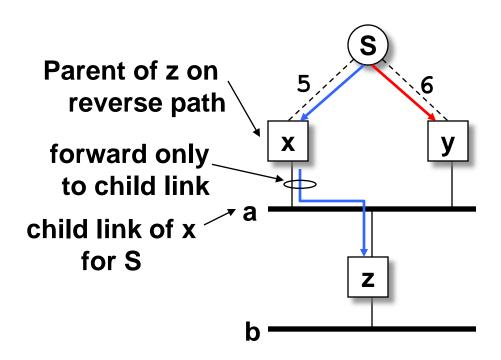
 Flooding can cause a given packet to be sent multiple times over the same link



Solution: Reverse Path Broadcasting

Reverse Path Broadcasting (RPB)

- Choose single parent for each link along reverse shortest path to source
- Only parent forwards to child link
- Identifying parent links
 - Distance
 - Lower address as tiebreaker



Even after fixing this, not done

- This is still a broadcast algorithm the traffic goes everywhere
- Need to "Prune" the tree when there are subtrees with no group members
- Networks know they have members based on IGMP messages
- Add the notion of "leaf" nodes in tree
 - They start the pruning process

Pruning Details

- Prune (Source, Group) at leaf if no members
 - Send Non-Membership Report (NMR) up tree
- If all children of router R send NMR, prune (S,G)
 - Propagate prune for (S,G) to parent R
- On timeout:
 - Prune dropped
 - Flow is reinstated
 - Down stream routers re-prune
- Note: a soft-state approach

Distance Vector Multicast Scaling

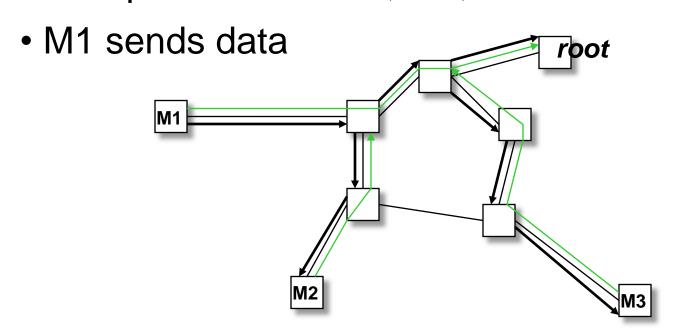
- State requirements:
 - O(Sources × Groups) active state
- How to get better scaling?
 - Hierarchical Multicast
 - Core-based Trees

Core-Based Trees (CBT)

- Pick "rendevouz point" for the group (called core)
- Build tree from all members to that core
 - Shared tree
- More scalable:
 - Reduces routing table state from O(S x G) to O(G)

Use Shared Tree for Delivery

• Group members: M1, M2, M3



→ control (join) messages

→ data

Core-Based Tree Approach

- Build tree from all members to core or root
 - Spanning tree of members
- Packets are broadcast on tree
 - We know how to broadcast on trees

Requires knowing root per group

Barriers to Multicast

- Hard to change IP
 - Multicast means changes to IP
 - Details of multicast were very hard to get right

- Not always consistent with ISP economic model
 - Charging done at edge, but single packet from edge can explode into millions of packets within network

Multicast vs Caching

 If delivery need not be simultaneous, caching (as in CDNs) works well, and needs no change to IP

Announcements

Clarification on Homework 3

- All links in the homework are full duplex.
 - Can send full line rate in both directions simultaneously

Announcements

- HW4 will just be a nongraded worksheet
- Review sessions (i.e., extended office hours) will be scheduled during class time of reading week
 Will ask you to send in questions beforehand....
- In response to several queries, sometime after the SDN lecture I will give a short (and very informal) talk on my experience with Nicira.

Wireless Review

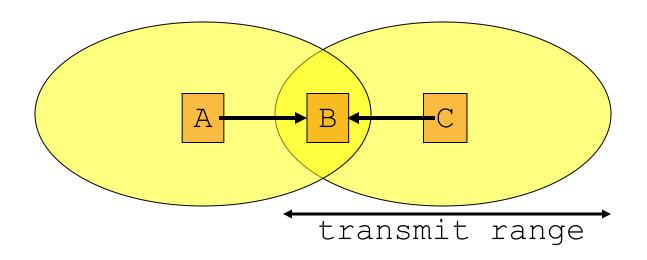
History

- MACA proposal: basis for RTS/CTS in lecture
 - -MACA proposed as replacement for carrier sense
 - Contention is at receiver, but CS detects sender!
- MACAW paper: extended and altered MACA
 - Implications of data ACKing
 - Introducing DS in exchange: RTS-CTS-DS-Data-ACK
 - Shut up when hear DS or CTS (DS sort of like carrier sense)
 - Other clever but unused extensions for fairness, etc.
- 802.11: uses carrier sense and RTS/CTS
 - -RTS/CTS often turned off, just use carrier sense
 - When RTS/CTS turned on, shut up when hear either
 - RTS/CTS augments carrier sense

Why Isn't MACA Enough?

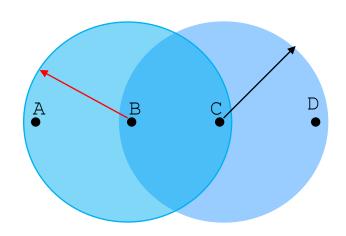
- That's what we now discuss, by repeating a few slides from previous lecture
- In what follows, we assume that basic reception is symmetric (if no interference):
 - If A is in range of B, then B is in range of A
- Any asymmetries in reception reflect interference from other senders

Hidden Terminals: Why MACA is Needed



- A, C can both send to B but can't hear each other
 - A is a hidden terminal for C and vice versa
- Carrier Sense by itself will be ineffective
 - If A is already sending to B, C won't know to defer.

Exposed Terminals: Alleged Flaw in CS



- Exposed node: using carrier sense
 - B sends a packet to A
 - C hears this and decides not to send a packet to D
 - Despite the fact that this will not cause interference!
- Carrier sense prevents successful transmission!

Key Points in MACA Rationale

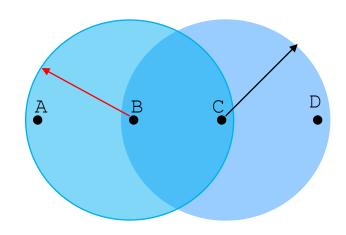
- No concept of a global collision
 - Different receivers hear different signals
 - Different senders reach different receivers

- Collisions are at receiver, not sender
 - -Only care if receiver can hear the sender clearly
 - It does not matter if sender can hear someone else
 - As long as that signal does not interfere with receiver
- Goal of protocol:
 - Detect if receiver can hear sender
 - Tell senders who might interfere with receiver to shut up

What Does This Analysis Ignore?

- Data should be ACKed!
 - In wireless settings, data can be easily lost
 - Need to give sender signal that data was received
- How does that change story?
- Connection is now two-way!
 - Congestion is at both sender and receiver
 - Carrier-sense is good for detecting the former
 - MACA is good for detecting the latter

Exposed Terminals Revisited



- Exposed node: with only MACA, both B and C send
- But when A or D send ACKs, they won't be heard!
- Carrier-sense prevents B, C sending at same time

802.11 Overview (oversimplified)

- Uses carrier sense and RTS/CTS
 - -RTS/CTS often turned off, just use carrier sense
- If RTS/CTS turned on, shut up when hear either
 - -CTS
 - Or current transmission (carrier sense)

- What if hear only RTS, no CTS or transmission?
 - You can send (after waiting small period)
 - CTS probably wasn't sent

What Will Be on the Final?

- General awareness of wireless (lecture, section)
- Reasoning about a given protocol
 - If we used the following algorithm, what would happen?
- You are **not** expected to know which algorithm to use; we will tell you explicitly:
 - -RTS/CTS
 - -Carrier Sense
 - -Both

Back to Congestion Control

Quick Review of Advanced CC

• Full queues: RED

Non-congestion losses: ECN

High-speeds: Alter constants: HSTCP

Fairness: Need isolation

• Min. flow completion time: Need better adjustment

Router-Assisted CC: Isolation: WFQ, AFD

Adjustment. RCP

Why is Scott a Moron?

Or why does Bob Briscoe think so?

Giving equal shares to "flows" is silly

- What if you have 8 flows (to different destinations), and I have 4...
 - Why should you get twice the bandwidth?
- What if your flow goes over 4 congested hops, and mine only goes over 1?
 - Why not penalize you for using more scarce bandwidth?
- And what is a flow anyway?
 - -TCP connection
 - Source-Destination pair?
 - -Source?

flow rate fairness dismantling a religion

<draft-briscoe-tsvarea-fair-01.pdf>

status: individual draft

final intent: informational

intent next: tsvwg WG item after (or at) next draft

Bob Briscoe Chief Researcher, BT Group IETF-68 tsvwg Mar 2007





Charge people for congestion!

- Use ECN as congestion markers
- Whenever I get ECN bit set, I have to pay \$\$\$
- No debate over what a flow is, or what fair is...
 - -Just send your packets and pay your bill

- Can avoid charges by sending when no congestion
- Idea started by Frank Kelly, backed by much math
 - Great idea: simple, elegant, effective

Why isn't this the obvious thing to do?

- Do you really want to sell links to highest bidder?
 - Will you ever bandwidth when Bill Gates is sending?
 - He just sends as fast as he can, and pays the bill

- Can we embed economics at such a low level?
 - Charging mechanisms usually at higher levels
- Is this the main problem with congestion control?
 - Is this a problem worth solving? Bandwidth caps work…
- This approach is fundamentally right...
 - ...but perhaps not practically relevant

Datacenter Networks

What makes them special?

- Huge scale:
 - 100,000s of servers in one location
- Limited geographic scope:
 - High bandwidth (10Gbps)
 - Very low RTT
- Extreme latency requirements (especially the tail)
 - With real money on the line
- Single administrative domain
 - No need to follow standards, or play nice with others
- Often "green field" deployment
 - So can "start from scratch"...

This means....

- Can design "from scratch"
- Can assume very low latencies
- Need to ensure very low queuing delays
 - -Both the average, and the tail....
 - -TCP terrible at this, even with RED
- Can assume plentiful bandwidth internally
- As usual, flows can be elephants or mice
 - Most flows small, most bytes in large flows...

Deconstructing Datacenter Packet Transport

Mohammad Alizadeh, Shuang Yang, Sachin Katti, Nick McKeown, Balaji Prabhakar, Scott Shenker

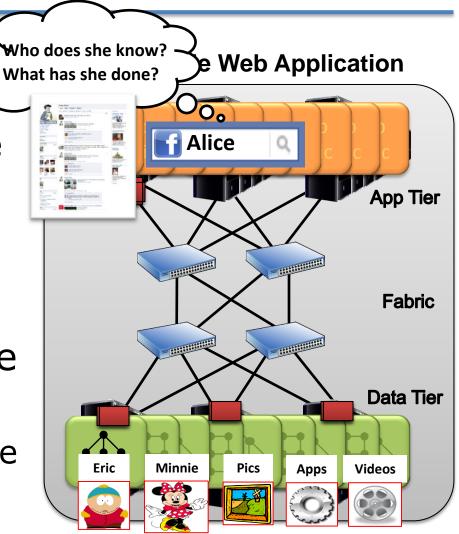
Stanford University

U.C. Berkeley/ICSI

HotNets 2012

Transport in Datacenters

- Latency is King
 - Web app response time depends on completion of 100s of small RPCs
 - Tail of distribution is particularly important
- But, traffic also diverse
 - Mice AND Elephants
 - Often, elephants are the root cause of latency



Transport in Datacenters

- Two fundamental requirements
 - High fabric utilization
 - Good for all traffic, esp. the large flows
 - Low fabric latency (propagation + switching)
 - Critical for latency-sensitive traffic
- Active area of research
 - DCTCP[SIGCOMM'10], D3[SIGCOMM'11]
 HULL[NSDI'11], D2TCP[SIGCOMM'12]
 PDQ[SIGCOMM'12], DeTail[SIGCOMM'12]

vastly improve performance, but very complex

Goal of pFabric

- Subtract complexity, not add to it
- Start from scratch and ask:
 - What components do we need for DC CC?
 - How little mechanism can we get away with?

 None of what we say here applies to general CC, it is limited to DC setting

pFabric in 1 Slide

Packets carry a single priority

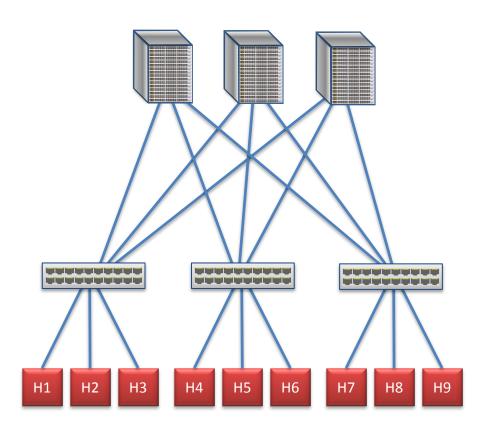
e.g., prio = remaining flow size

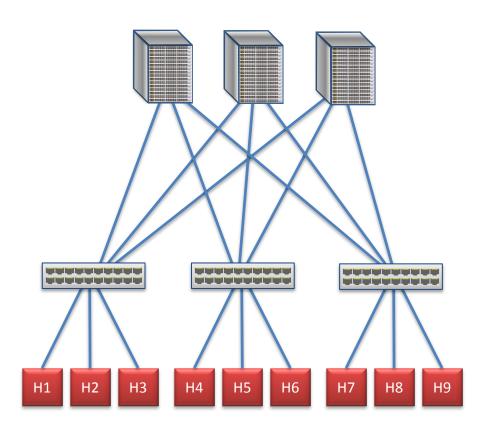
pFabric Switches

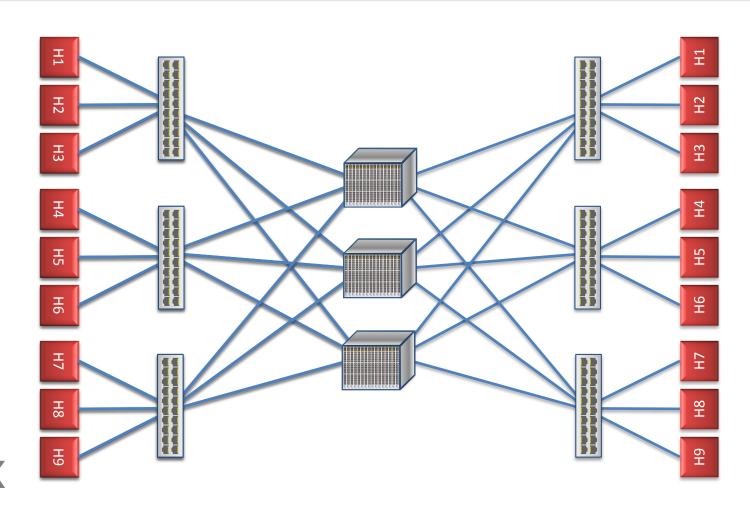
- Very small buffers (e.g., 10-20KB)
- Send highest priority / drop lowest priority pkts
 - Give priority to packets with least remaining in flow

pFabric Hosts

- Send/retransmit aggressively
- Minimal rate control: just prevent congestion collapse

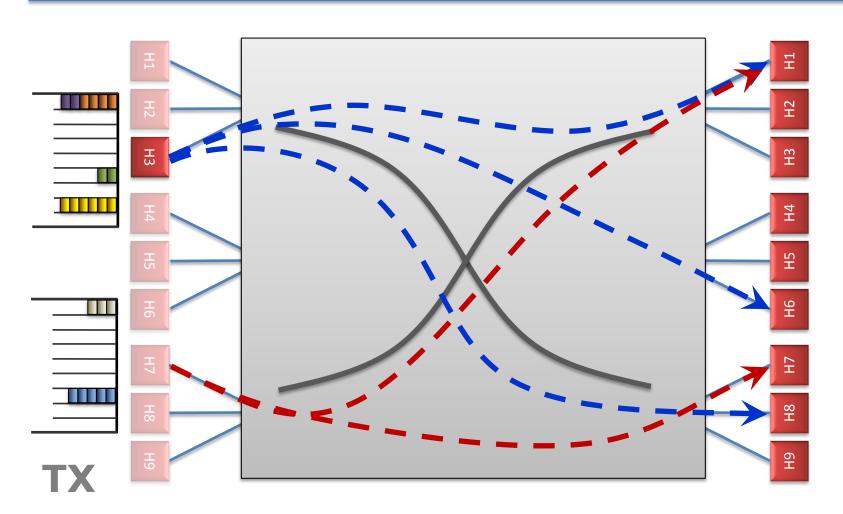






RX

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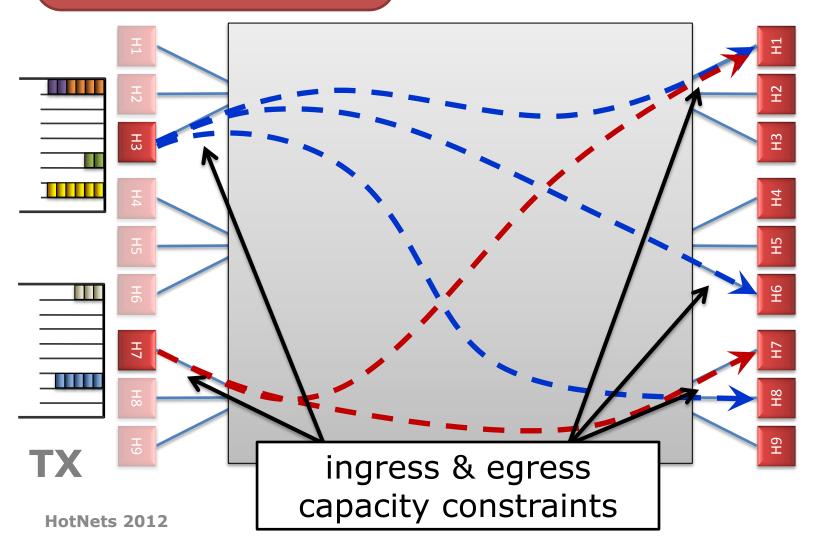


RX

DC transport = Flow scheduling on giant switch

Objective?

➤ Minimize avg FCT



RX

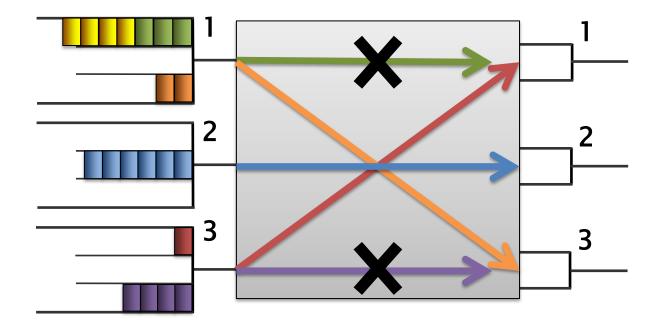
Min. FCT with Simple Queue

- Just use shortest-job first.....
- But here we have many inputs and many outputs with capacity constraints at both

"Ideal" Flow Scheduling

Problem is NP-hard ⊗ [Bar-Noy et al.]

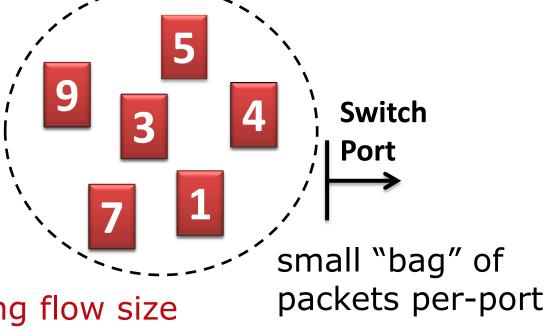
- Simple greedy algorithm: 2-approximation



pFabric Design

pFabric Switch

Priority Scheduling send higher priority packets first Priority Dropping drop low priority packets first



prio = remaining flow size

Near-Zero Buffers

- Easiest way to keep delays small?
 - Have small buffers!

- Buffers are very small (~1 BDP)
 - e.g., C=10Gbps, RTT=15 μ s → BDP = 18.75KB
 - Today's switch buffers are 10-30x larger

pFabric Rate Control

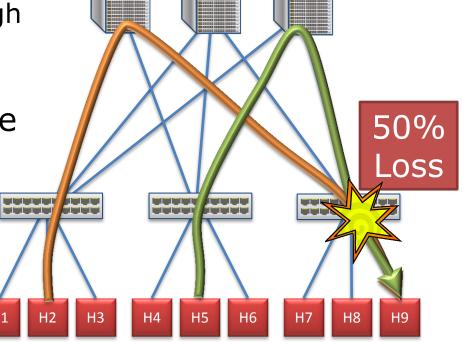
 Priority scheduling & dropping in fabric also simplifies rate control

Queue backlog doesn't matter

Priority packets get through

One task:

Prevent congestion collapse when elephants collide



pFabric Rate Control

- Minimal version of TCP
 - 1. Start at line-rate
 - Initial window larger than BDP
 - 2. No retransmission timeout estimation
 - Fix RTO near round-trip time
 - 3. No fast retransmission on 3-dupacks
 - Allow packet reordering

Why does this work?

Key observation:

Need the highest priority packet destined for a port available at the port at any given time.

Priority scheduling

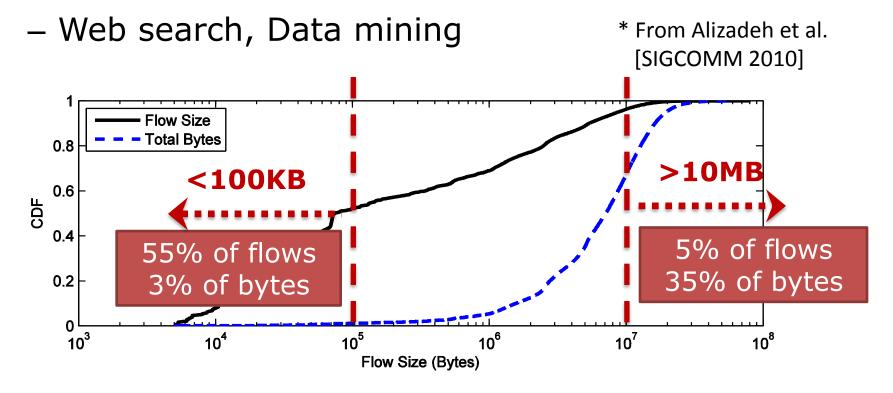
> High priority packets traverse fabric as quickly as possible

What about dropped packets?

- ➤ Lowest priority → not needed till all other packets depart
- ➤ Buffer larger than BDP → more than RTT to retransmit

Evaluation

- 54 port fat-tree: 10Gbps links, RTT = \sim 12 μ s
- Realistic traffic workloads

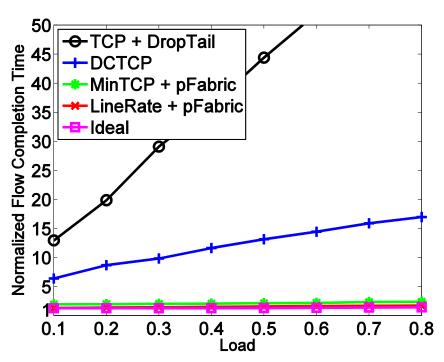


Evaluation: Mice FCT (<100KB)

Average

10 O-TCP + DropTail Normalized Flow Completion Time **DCTCP** MinTCP + pFabric -LineRate + pFabric Ideal 3 0.7 0.1 0.2 0.3 0.4 0.5 0.6 0.8 Load

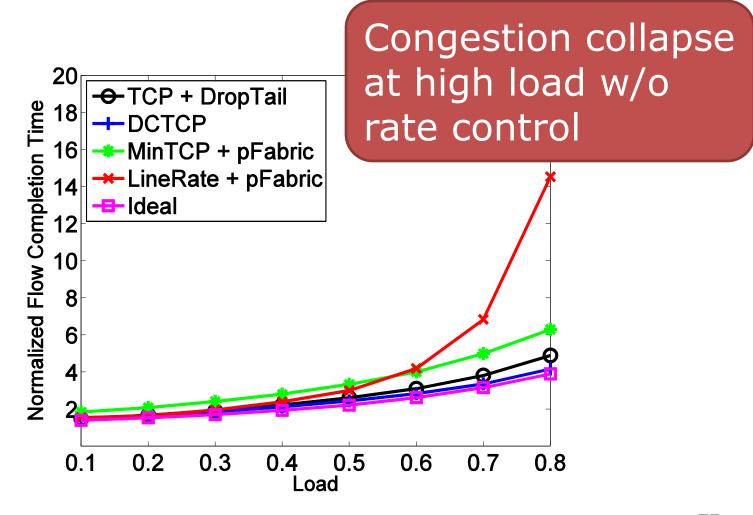
99th Percentile



Near-ideal: almost no jitter

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Evaluation: Elephant FCT (>10MB)



Summary

pFabric's entire design:

Near-ideal flow scheduling across DC fabric

Switches

Locally schedule & drop based on priority

Hosts

- Aggressively send & retransmit
- Minimal rate control to avoid congestion collapse