CS514: Intermediate Course in Computer Systems

Lecture 6: Sept. 17, 2003

"Performance of the Internet WAN"

What performance are we interested in?

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- Latency
  - Small web access
  - Interactive voice/video or gaming
- Jitter (variation in inter-packet arrival)
  - Interactive voice/video
  - Streaming could benefit (less buffering)
- Throughput
  - Large web access
  - Interactive or streaming video
- Problems
  - Packet loss, outages, out-of-order packets, packet corruption

This lecture

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- Shows why measuring performance is difficult
- Shows how some of the measurement takes place
- Tries to give some sense of how well the internet works
  - Though really nobody knows!!!

Email with Vern Paxson*

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Q: What are the best papers for describing the performance of the internet more generally?

A: There's no broad-perspective available like that. Very hard to do, given the immense diversity and difficulty of attaining sufficient measurement perspectives.

* God of internet measurement

Paxson 1994-1995 Study

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- 35 sites
- Measured TCP bulk transfers
  - Idea is that measurements of TCP can be applied to understanding TCP as well as understanding the Internet
  - Problem though is that TCP backs-off, so don't have complete control over your measuring tool
- Built filter to measure exact times
  - Some packets missed by filter, but this is detectable

TCP-centric measurements

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Simple UDP or ICMP-based measurements don't tell you how well TCP will perform

- You have to look at things that will hurt TCP performance:
  - Out of order delivery
  - Replication (rare, it turns out)
  - Packet corruption

Out of order delivery: Why does TCP care?

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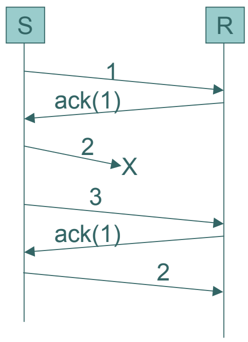
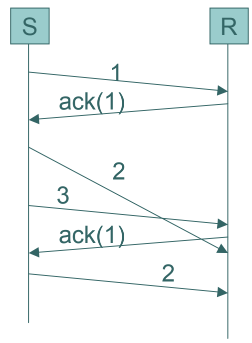


If packet lost, best strategy is to retransmit immediately after duplicate ACK



If packet reordered, best strategy is to wait for another ACK

Out of order delivery

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- It does happen (~0.2% in 1995)
  - Route changes? Path splitting?
- It varies a lot (up to 15% seen)
  - Thus nice if TCP could measure it dynamically
- Different in either direction
  1. Asymmetric paths
  2. Data more frequently reordered than ACKs
  - Therefore can't determine sending reorder rate from measuring received reorder rate

Packet corruption

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- Happens (1 in 5000 data packets)
  - Since 16-bit TCP checksum misses 1/65000 errors, 1/300M corrupted packets go undetected
  - Not a lot, but if your data is important, check at higher layers
  - Encryption will do this for you
- Happens much less for short packets (TCP ACKs)
  - Suggests that corruption happens in routers
  - Because link-layer checksum shouldn't care about packet length

Measuring bottleneck bandwidth: principle

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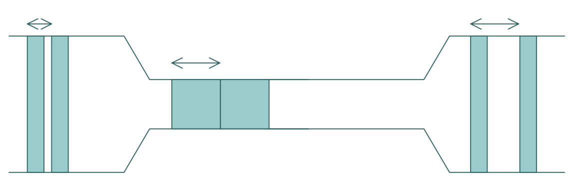

Send two packets within a short time space

Bottleneck link will increase the spacing

Measured receive spacing tells you the size of the bottleneck link

Why do we care?

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- If we want to understand effect of queuing delays, processing delays, etc., we must know bottleneck bandwidth
  - For instance, difference between bottleneck BW and available BW tells what fraction of the bottleneck link we got
- TCP never wants to send faster than the bottleneck bandwidth
  - Though often TCP wants to send much slower because of competing traffic (available bandwidth)

Why is it hard?

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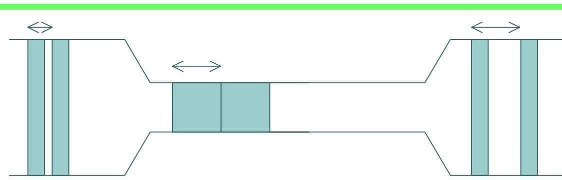
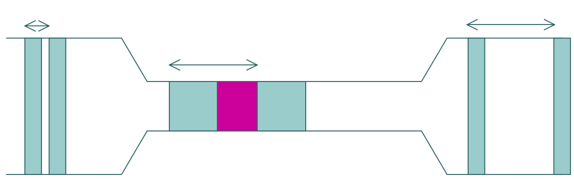


In part, because any competing traffic changes the spacing



Other complications

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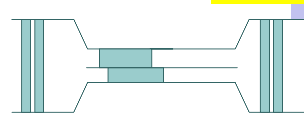
- Multi-channel effects



- Bottleneck may change
  - Route change
  - Dynamic bandwidth allocation (ISDN)
- Poor clock resolution
- Out-of-order delivery

Hard to do at one end

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- If ICMP echo, can’t tell in which direction bottleneck occurs
- If TCP, ACK spacing may be compressed on return path
  - Queue emptying effect

Basic approach to bottleneck bandwidth est.

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- Send stream of evenly spaced packets
  - Look for smallest spacing
- If possible link bandwidths known apriori, then look for peaks near these speeds
  - (modem speeds, T1, E1, multiples of these, Ethernet, etc.)
- This actually works pretty well
  - But impossible to do good bottleneck bandwidth estimation without loading the network

Measuring packet loss

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- Not real hard
  - Though can't use TCP data to measure loss rate, because TCP backs off in order to prevent loss
  - But can use ACKs
    - If loss in both directions is not correlated, which is the case
- Lots of variation
- Lots of correlation
  - Likelihood packet is lost if predecessor was lost
  - 25%-50%
  - But this is pre-RED

One-way Transit Time (Latency)

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- To do with absolute accuracy, required synchronized clocks
  - GPS, which is fairly inexpensive these days (<$500)
- Paxson didn't have this
  - But with analysis could determine relative skew between clocks
  - Basically by sampling only packet round trips with low and similar delay, and tracking timestamps against these
- He really only measured relative OTT

Other Paxson '97 factoids

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- Queuing time scales
  - Period of time over which a queue's delay changes
  - We care because if time is too small, no point in trying to adapt
  - Typically 0.1 – 1 sec, but can be much larger
    - 60 second spike, but due to routing oscillations

Other Paxson '97 factoids

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- Available bandwidth
  - Percentage of bottleneck bandwidth allocated to a given connection
  - Essentially actual_bw/bottleneck_bw
  - Wide range:
    - From 5% to 100%
    - Less as bottleneck BW grows

Large scale measurement now common

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- Many ongoing measurement projects
  - NIMI (Paxson), RIPE TTM, Caida skitter, etc.
- Standard measurement metrics
  - IP Performance Metrics (IPPM)
  - RFC 2330: IPPM Framework
  - Basic concepts and terms
  - Allows results from different measurement infrastructures to be meaningfully compared and combined

IPPM Metrics

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- RFC 2678: IPPM Metrics for Measuring Connectivity
- RFC 2679: A One-way Delay Metric for IPPM
- RFC 2680: A One-way Packet Loss Metric for IPPM
- RFC 2681: A Round-trip Delay Metric for IPPM
- Series ended in 1999, with metrics clearly missing
  - Bandwidth especially, also jitter, packet order, packet corruption, …

Internet Flow Rates

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- Zhang, Breslau, Paxson, Shenker
- Study of flow rates and sizes
- Particular interest in causes of different flow characteristics
- Traced flows at ISPs and campus access links
- Developed a tool (T-RAT) to analyze cause of rate limiting

  http://www.research.att.com/projects/T-RAT/

Subsequent slides taken from Sigcomm 2002 presentation

Internet Flow Rates Data Set

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- Packet traces at ISP backbones and campus access links
  - 8 datasets; each lasts 0.5 – 24 hours; over 110 million packets
- Summary flow statistics collected at 19 backbone routers
  - 76 datasets; each lasts 24 hours; over 20 billion packets

Subsequent slides taken from Sigcomm 2002 presentation

Flow Rate Characteristics

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- Rate distribution
  - Most flows are *slow,* but most bytes are in *fast* flows
  - Distribution is skewed
    - Not as skewed as size distribution
    - Consistent with log-normal distribution [BSSK97]
- Correlations
  - Rate and size are strongly correlated
  - Not due to TCP slow-start
    - Removed initial 1 second of each connection; correlations increase
  - What users download is a function of their bandwidth

Rate Limiting Factors (Bytes)

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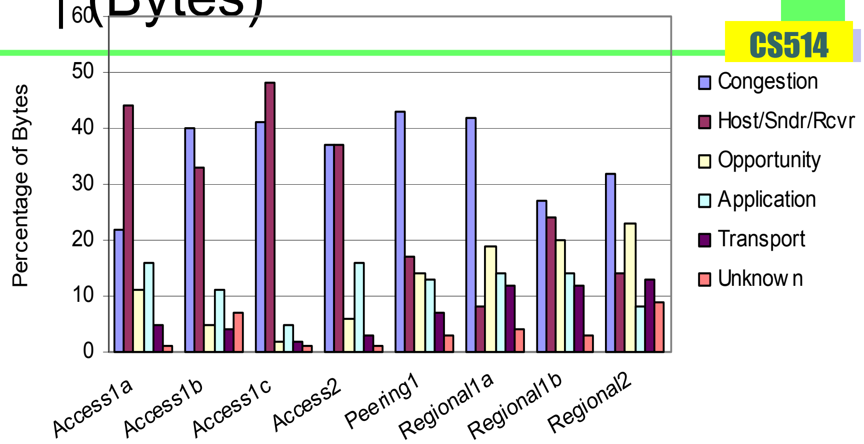

Dominant causes by bytes: Congestion, Receiver

Rate Limiting Factors (Bytes---i.e. long flows)

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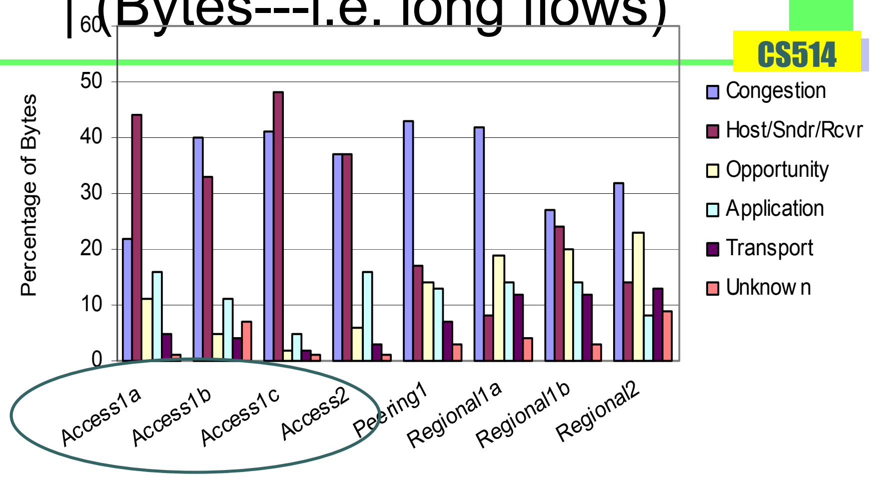

This population tends to have better internet connectivity, and so is more often receiver limited

Rate Limiting Factors (Flows---I.e. short flows)

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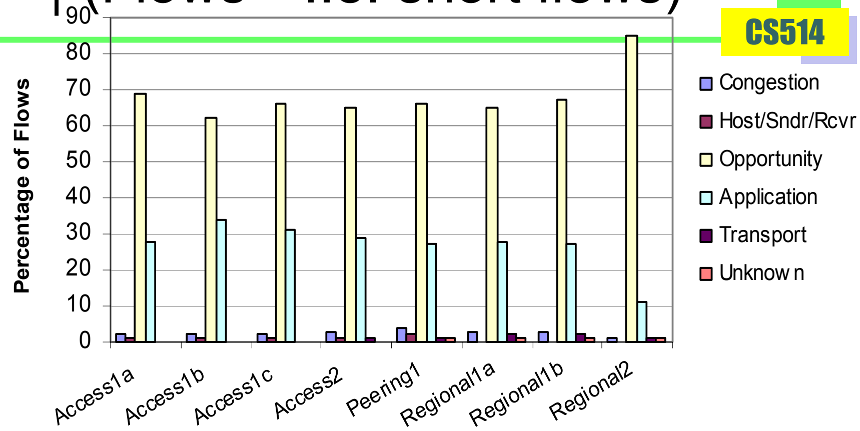

Dominant causes by flows: Opportunity, Application

Rate Limiting Factors (Flows)

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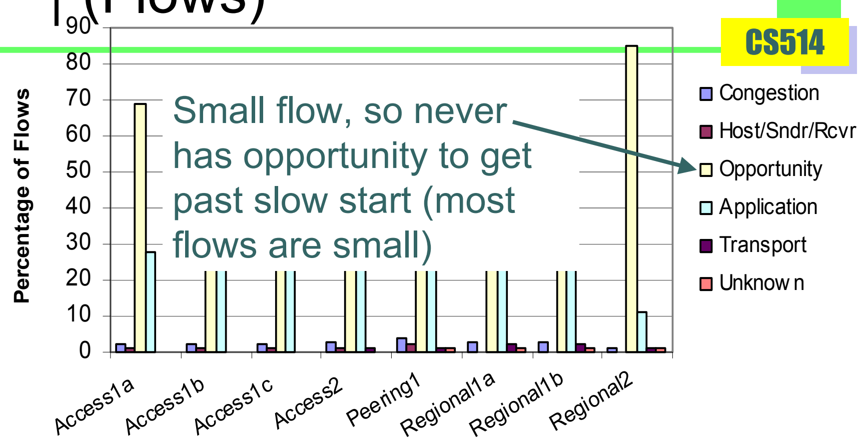

Small flow, so never has opportunity to get past slow start (most flows are small)

Dominant causes by flows: Opportunity, Application

Rate Limiting Factors (Flows)

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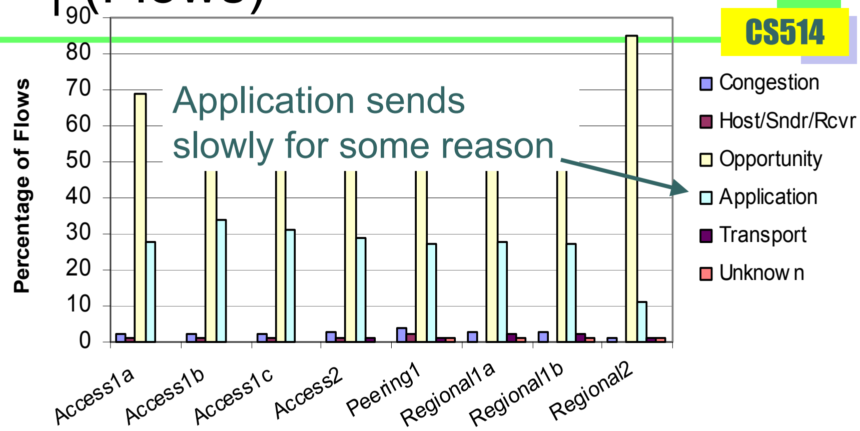

Dominant causes by flows: Opportunity, Application

Flow Characteristics by Cause

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- Different causes are associated with different performance for users
  - Rate distribution
    - Highest rates: Receiver, Transport
    - (Don't experience congestion)
  - Size distribution
    - Largest sizes: Receiver
    - (Size and rate a correlated)
  - Duration distribution
    - Longest duration: Congestion
    - (Congested flows take longer)

Some flow rate study conclusions (mine)

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- Latency matters (most flows are small)
  - Slow start performance dominated by round trip time
- Congestion matters, but most congestion is at the edge
  - Recall that more congestion seen for users with thinner access pipes
  - Web service provider can't do anything about edge congestion

End-to-end effects of Internet Path Selection

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- Savage et. al., Univ of Washington Seattle
- Compared path found by internet routing with alternates
  - Alternates composed by gluing together two internet-routed paths
- Roundtrip time, loss rate, bandwidth
- Data sets: Paxson, plus new ones from UW

Results

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- Round-trip time
  - 30% - 55% of paths had better alternate paths
    - Mostly within 30ms
    - 10% had 50% or better latency
- Loss rate
  - 75% - 80% of paths had better alternate paths
  - 5% - 50% of paths had 5% or better drop rates

Results

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- Bandwidth
  - 70% - 80% of paths have better alternates
  - 10% - 20% by a factor of three
- Results held for different times of day
- Results were not due to only a small number of hosts
- Shorter propagation delay and avoidance of congestion both contributed to better alternate paths

Some conclusions (mine)

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- These results probably apply more to p2p than to popular web services
  - Web services well connected, so almost always a good path
- In cases where congestion was avoided, benefit derives from fact that few flows were going through alternate path
  - If many users took advantage of alternate paths, the alternate paths would no longer be better!
  - MIT RON, Sockeye (global routing service)

E2E WAN Service Availability

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- Chandra et. al., U Texas at Austin
- Understand how network failures effect service availability
- Help web service designers make best use of available tools
- Evaluate likely value of techniques like:
  - replication of active objects
  - overlay objects

Data sets and limitations

- Traceroute data sets
  - Paxson NIMI data sets ('94-'95)
    - Measures middle of network but not the typical edge (NIMI probes well connected)
  - Savage UW data sets ('99)
    - All from Univ. Washington
- HTTP data sets from squid proxies
  - Not reflective of either typical connectivity or typical user
- Authors think they underestimate failure
  - Mainly because of flakeyness of many clients

Model derived from data sets

Parameter	Default Value	Comment
Rate	1.5% (all) 1.25% (>30s)	Varies from 0.4% to 7.4% in different data sets
Location	Src Mid Dst 25% 50% 25%	All locations significant (vaguely defined though)
Duration	avg. = 609 sec	Heavy tailed
Interarrival	avg. = 13 hr.	

Key Findings (but I don't trust them)

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- Failure distributions are heavy-tailed
  - Long failures account for significant fraction of failure duration
- Data caching techniques will have little positive effect
  - Because still uncached content at failure time
- Prefetching and shipping mobile extension code to clients may have order-of-magnitude benefit
  - Prefetch everything before failure
  - Mobile extension: ship code and data to client---not yet practical

What sounds right to me

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- Engineer web services for reliability
- Connect web services to many ISPs to minimize effects of routing failures in the middle
  - On the theory that fewer AS hops means fewer chances for failure
- There isn't much web service can do about failures near the client

Conclusions

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- Network performance varies tremendously
  - WAN, MAN, LAN . . .
  - Even in different WAN settings, performance varies
- You'll need to either make your own measurements, or understand how to interpret those claimed by the ISP
  - Either way, the issues are subtle