Measurement Study of Low-bitrate Internet Video Streaming

Dmitri Loguinov and Hayder Radha
CS Dept at CUNY NY and EE/ECE at MSU.

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Introduction

- Many studies of Internet performance
  - Paxson, Mogul, Caceres…
  - Across countries, many sites
  - Well-connected (often schools on backbone)
- But few look at it from the point of dialup user
  - About 50% of home users dialup
    - Peak, but will remain majority for 3-5 years
  - ISP cannot always do 56 kb/s

Introduction

- Most studies involve TCP
  - 90-99% of traffic on Internet is TCP
- But MM prefers UDP
  - (*Why?)*
- Also, TCP uses ACK-based scheme
  - MM protocols prefer NACK to scale (*why?)*
- Video studies have done few paths

Introduction

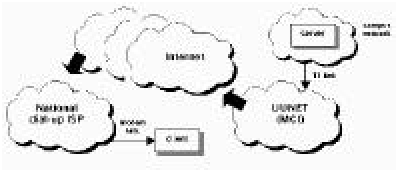
- Video streaming experiment
  - Seven month long
  - MPEG-4 (low-bandwidth) over UDP
  - Over dialup
  - 600 major cities
  - 50 States

Outline

- Introduction (done)
- Methodology ←
  - Setup
  - Streaming
  - Client-Server architecture
- Results
- Analysis
- Summary

Setup

- Clients connected to each long-distance
- Server was in NY



- 3 ISPs in all 50 states
- 1813 different access points
- 1188 major U.S. cities

Setup

- Dialer
  - Connect to ISP with PPP connection
  - *traceroute*from sender→receiver and receiver→sender
    - Parallel paths
- Detect when modem connection was bad
  - If *r* is target bitrate, *p* is packet loss
  - If $B_r < 0.9r$ then bad (toss)
  - If B_p is > 15% then bad (toss)
- Good data was time-stamped
  - Day of week plus 3 eight hour slots each data
  - At least one from each day for each state for each slot

Streaming

- MPEG-4 stream
  - 2 ten-minute QCIF (176x144) streams
  - S_1 14 kbps (Nov-Dec 1999)
  - S_2 25 kbps (Jan-May 2000)
- Server split into 576 byte packets
  - With overhead S_1 16 kbps and S_2 27.4 kbps
  - About 6 packets/sec (for S_2)
- To remove jitter, had delay buffer
  - (*What is this?)*
- Chose 2.7 seconds (1.3 ideal in pilots stud)
  - (*Why might this be a bad idea?)*

Client-Server Architecture

- Server
  - Multithreaded
  - Bursts of packets (340-500 ms)
- Client
  - Recover lost packets through NACK
  - Collect RTT delay
    - Based on NACK
      - *(When might this not work well?)*
    - Probes every 30 seconds if loss < 1%
- Evaluated for 9 months
  - Whew!

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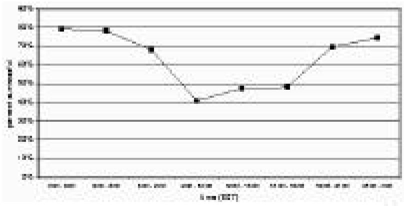
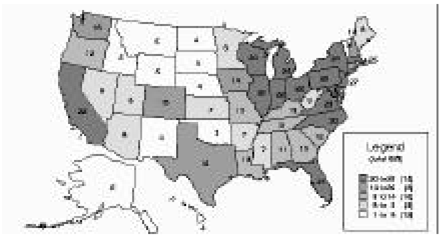
Results

- Two datasets
  - D_1 16.783 connections, 8429 successful
  - D_2 17,465 connections, 8423 successful
  - → To get MPEG-4, need 2 attempts on avg

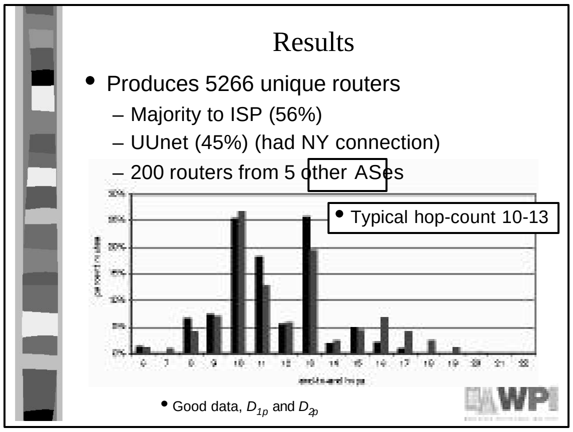


- Time of day matters

Results



- D1 had 962 dialup points, 637 cities
- D2 had 880 dialup points, 575 cities



Results

- Produces 5266 unique routers
  - Majority to ISP (56%)
  - UUnet (45%) (had NY connection)
  - 200 routers from 5 other ASes

- Typical hop-count 10-13

- Good data, D_{1p} and D_{2p}

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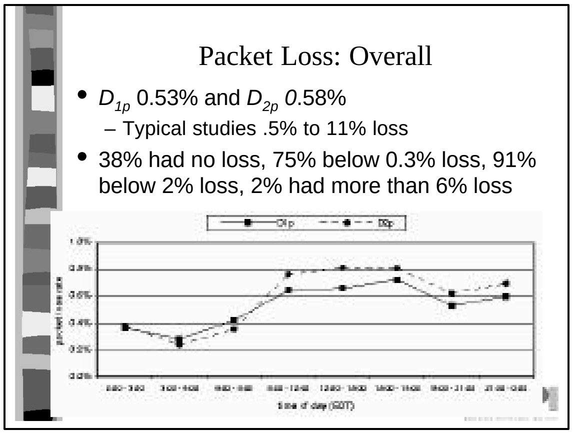
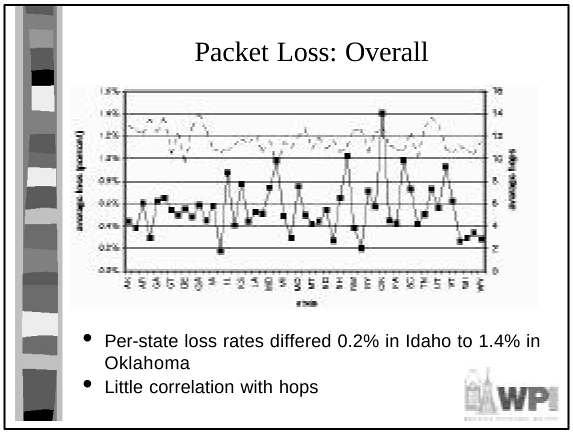
  - Packet Loss
  - Underflow
  - Delay and Jitter
  - Reordering
  - Assymetric
- Summary



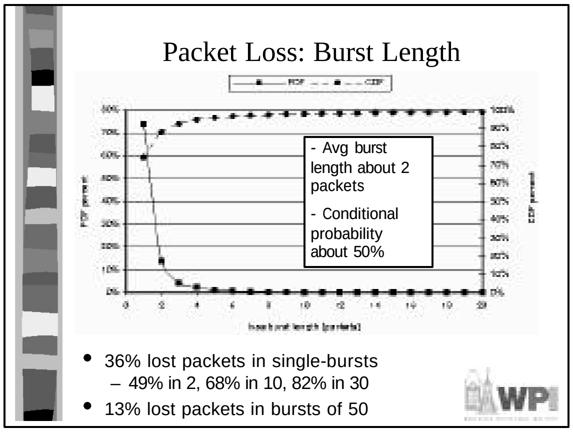
Packet Loss: Overall

- D_{1p} 0.53% and D_{2p} 0.58%
  - Typical studies .5% to 11% loss
- 38% had no loss, 75% below 0.3% loss, 91% below 2% loss, 2% had more than 6% loss



Packet Loss: Overall

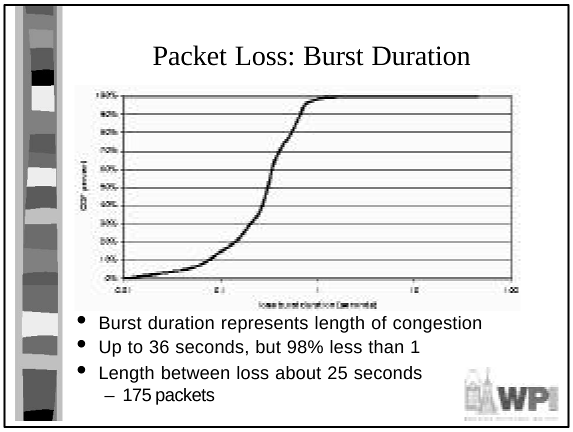
- Per-state loss rates differed 0.2% in Idaho to 1.4% in Oklahoma
- Little correlation with hops



Packet Loss: Burst Length

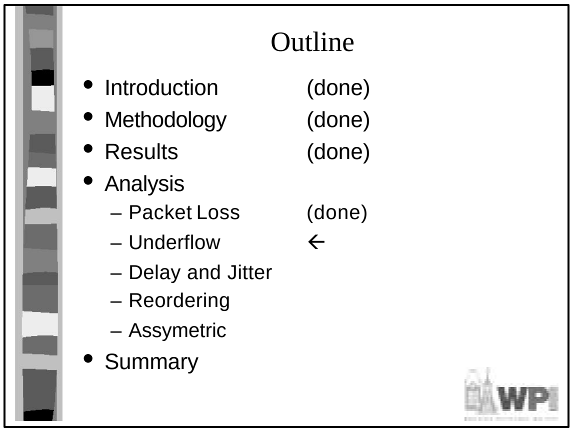
- Avg burst length about 2 packets
- Conditional probability about 50%

- 36% lost packets in single-bursts
  - 49% in 2, 68% in 10, 82% in 30
- 13% lost packets in bursts of 50



Packet Loss: Burst Duration

- Burst duration represents length of congestion
- Up to 36 seconds, but 98% less than 1
- Length between loss about 25 seconds
  - 175 packets



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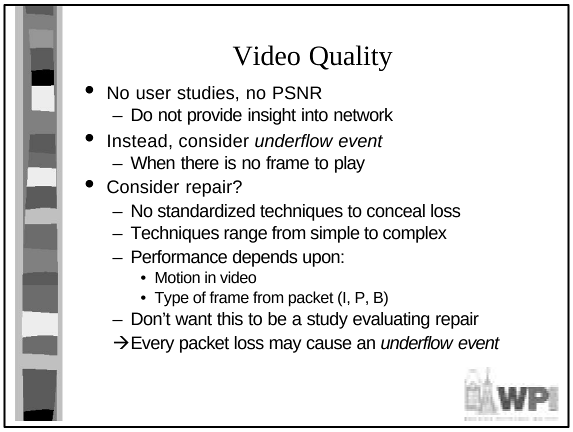
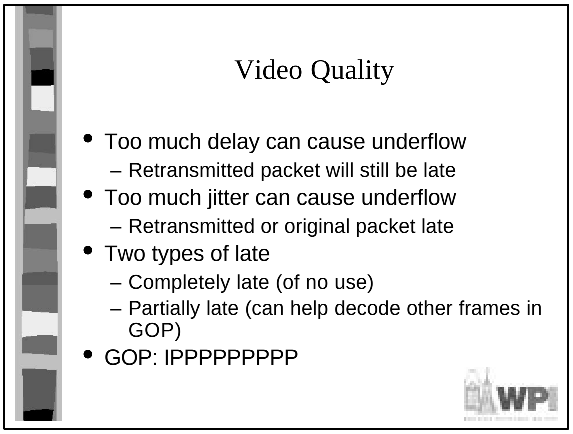


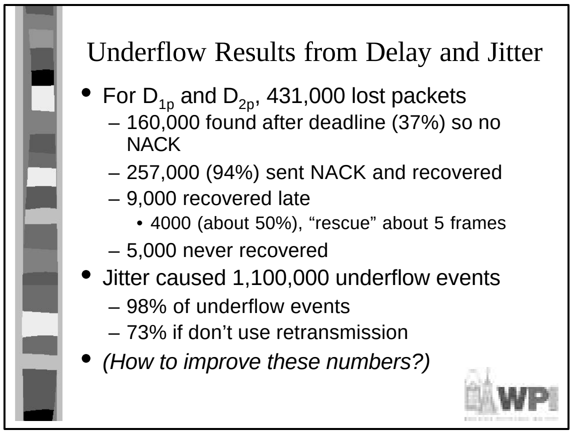
Video Quality

- No user studies, no PSNR
  - Do not provide insight into network
- Instead, consider *underflow event*
  - When there is no frame to play
- Consider repair?
  - No standardized techniques to conceal loss
  - Techniques range from simple to complex
  - Performance depends upon:
    - Motion in video
    - Type of frame from packet (I, P, B)
  - Don't want this to be a study evaluating repair
  - →Every packet loss may cause an *underflow event*



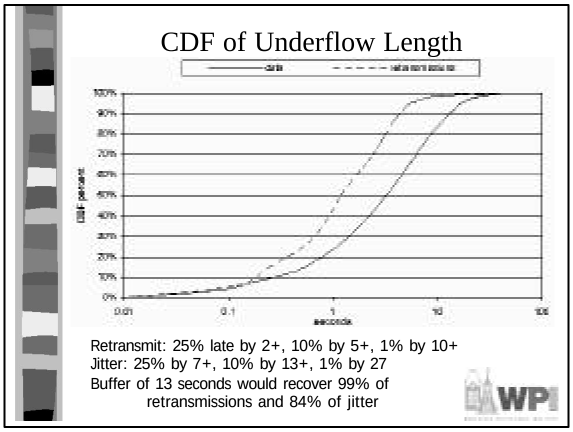
Video Quality

- Too much delay can cause underflow
  - Retransmitted packet will still be late
- Too much jitter can cause underflow
  - Retransmitted or original packet late
- Two types of late
  - Completely late (of no use)
  - Partially late (can help decode other frames in GOP)
- GOP: IPPPPPPPPP



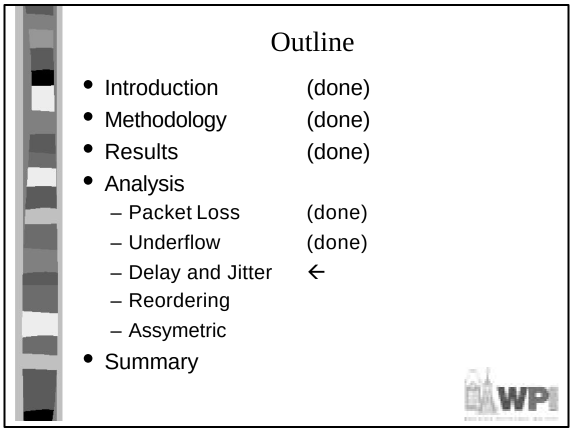
Underflow Results from Delay and Jitter

- For D_{1p} and D_{2p}, 431,000 lost packets
  - 160,000 found after deadline (37%) so no NACK
  - 257,000 (94%) sent NACK and recovered
  - 9,000 recovered late
    - 4000 (about 50%), "rescue" about 5 frames
  - 5,000 never recovered
- Jitter caused 1,100,000 underflow events
  - 98% of underflow events
  - 73% if don't use retransmission
- *(How to improve these numbers?)*



CDF of Underflow Length

Retransmit: 25% late by 2+, 10% by 5+, 1% by 10+
Jitter: 25% by 7+, 10% by 13+, 1% by 27
Buffer of 13 seconds would recover 99% of retransmissions and 84% of jitter



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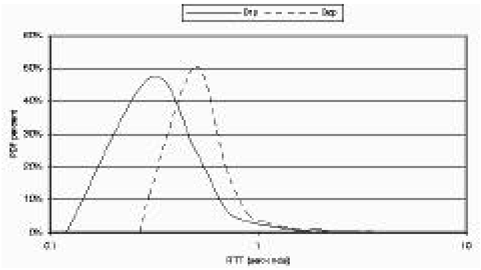
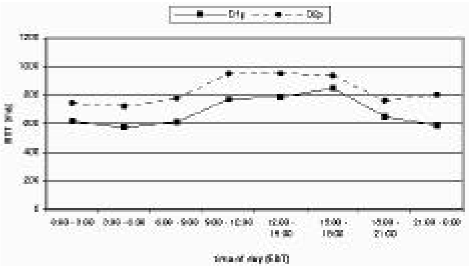
Round-Trip Time



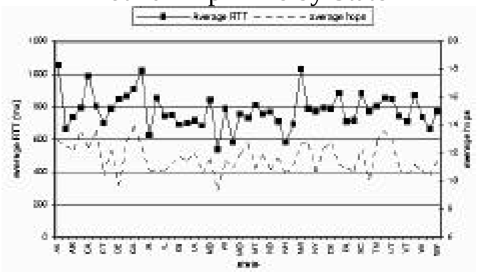
Average: D_{1p} 698 ms, D_{2p} 839 ms
Min: D_{1p} 119 ms, D_{2p} 172 ms
Max: 400+ values over 30 seconds!

Round-Trip Time by Time of Day



Delay correlates with time of day
Increase in min to peak about 30-40%

Round-Trip Time by State



Alaska, Hawaii, New Mexico → high
Maine, New Hamp., Minn → low
→Suggests some correlation with geography, but really very little (Number of hops .5 corr.)

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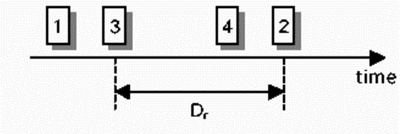
Packet Reordering: Overview

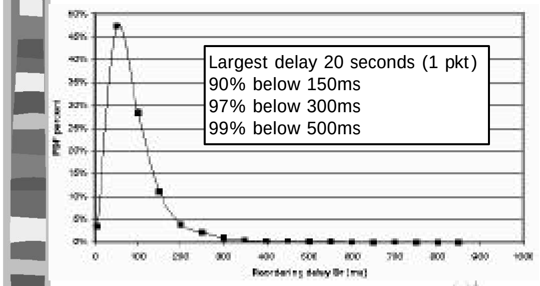
- Gap in sequence numbers indicates loss
  - *(When might this fail?)*
- For D^a_{1p}, 1 in 3 missing packets arrived out of order
  - Simple streaming protocol with NACK could waste bandwidth
- Average
  - was 6.5% of missing
  - 0.04% of sent packets
- Of 16,952 sessions, 9.5% have at least 1
  - ½ of sessions from ISP *a*
- No correlation with time of day

Packet Reordering: Delay and Distance



- Distance is d_r = 2,
- Delay is time from 3 to 2

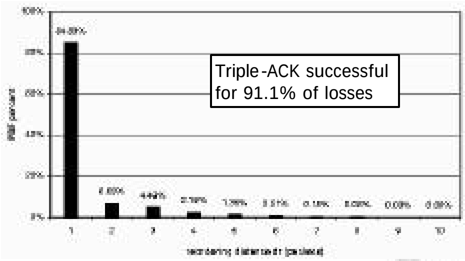
Packet Reordering: Delay



Largest delay 20 seconds (1 pkt)
90% below 150ms
97% below 300ms
99% below 500ms

Packet Reordering: Distance

Largest distance was 10 packets
Triple-ACK in TCP causes duplicate *(why?)*



Triple-ACK successful for 91.1% of losses

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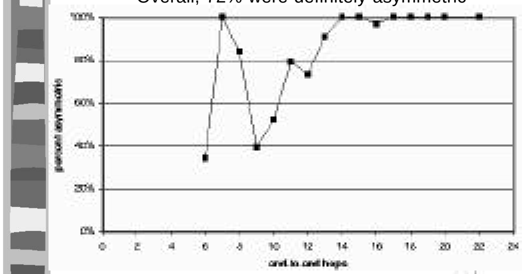
Asymmetric Paths

- If number of hops from sender→receiver different than receiver→sender
  - then *asymmetric*
- If number of hops from sender→receiver same as receiver→sender
  - then *probably symmetric*

Asymmetric Paths

- Overall, 72% were definitely asymmetric



Conclusion

- Internet packet loss is bursty
- Jitter worse than packet loss or RTT
- RTTs on the order of seconds are possible
- RTT correlated with number of hops
- PacktlLoss not correlated with number of hops or RTT
- Most paths asymmetric