An Empirical Evaluation of TCP Performance in Online Games

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Talk Outline

- Motivation
- Trace collection
- Game traffic characteristics
- Analysis of TCP performance
- Design Implications
- Conclusion

Motivation

- TCP is generally considered not suitable for real-time transmission
- All first-person shooting games use UDP
- No consensus has been reached for MMORPGs
  - Heated debate occurs at game developers' forums
  - Each protocol has quite a few proponents

Protocol	MMORPGs
TCP	World of Warcraft, Lineage I/II, Guild Wars, Ragnarok Online, Anarchy Online, Mabinogi
UDP	EverQuest, Star Wars Galaxies, City of Heroes, Ultima Online, Asherons Call, Final Fantasy XI
TCP+UDP	Dark Age of Camelot

Motivation (cont.)

Q1: Is TCP appropriate for MMORPGs?

Q2: If not, how to design a suitable protocol?

We try to answer these problems by empirical evidence.

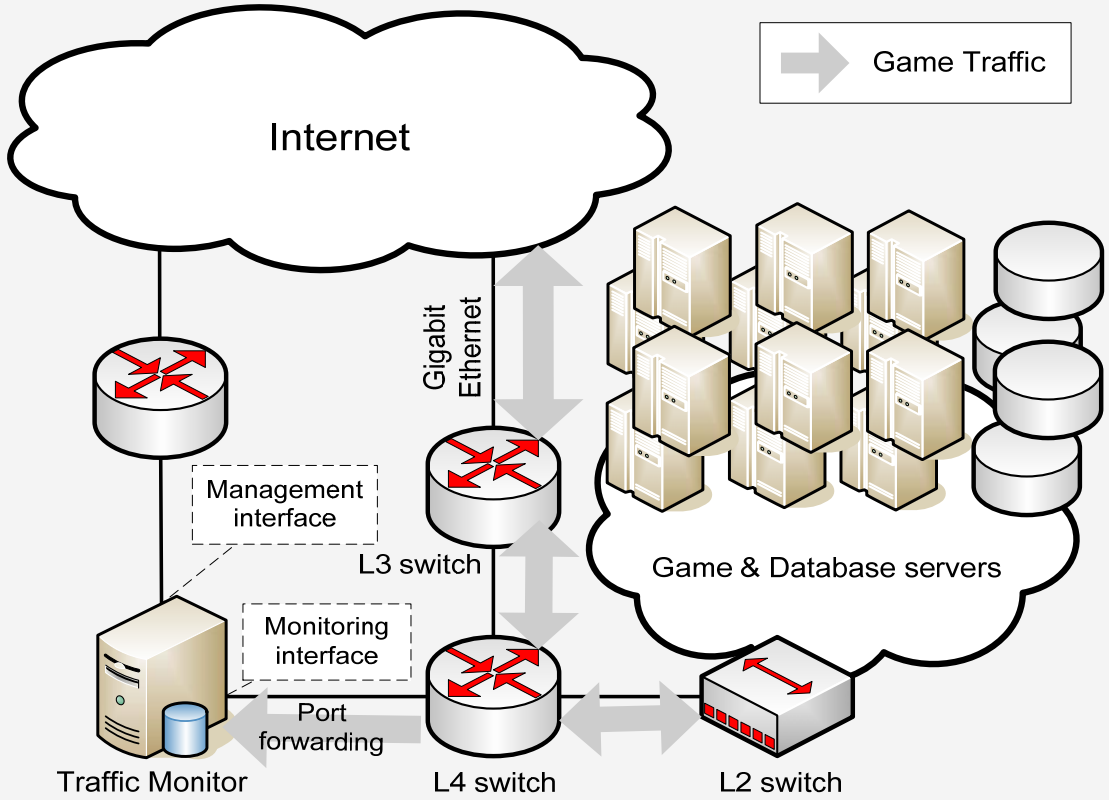
Key Contributions / Findings

- TCP is unwieldy and inappropriate for MMORPGs
- Estimate how long users will stay if a more suitable protocol is used (100 minutes ➔ 135 minutes)
- Proposed a number of design guidelines which exploit the unique characteristics of game traffic

ShenZhou Online

- A commercial MMORPG in Taiwan
- Thousands of players online at anytime
- TCP-based client-server architecture

Trace Collection



Trace Time	# TCP Conn	# Packets	# Bytes
20 hours	112,369	1,356 million	58.5 TB

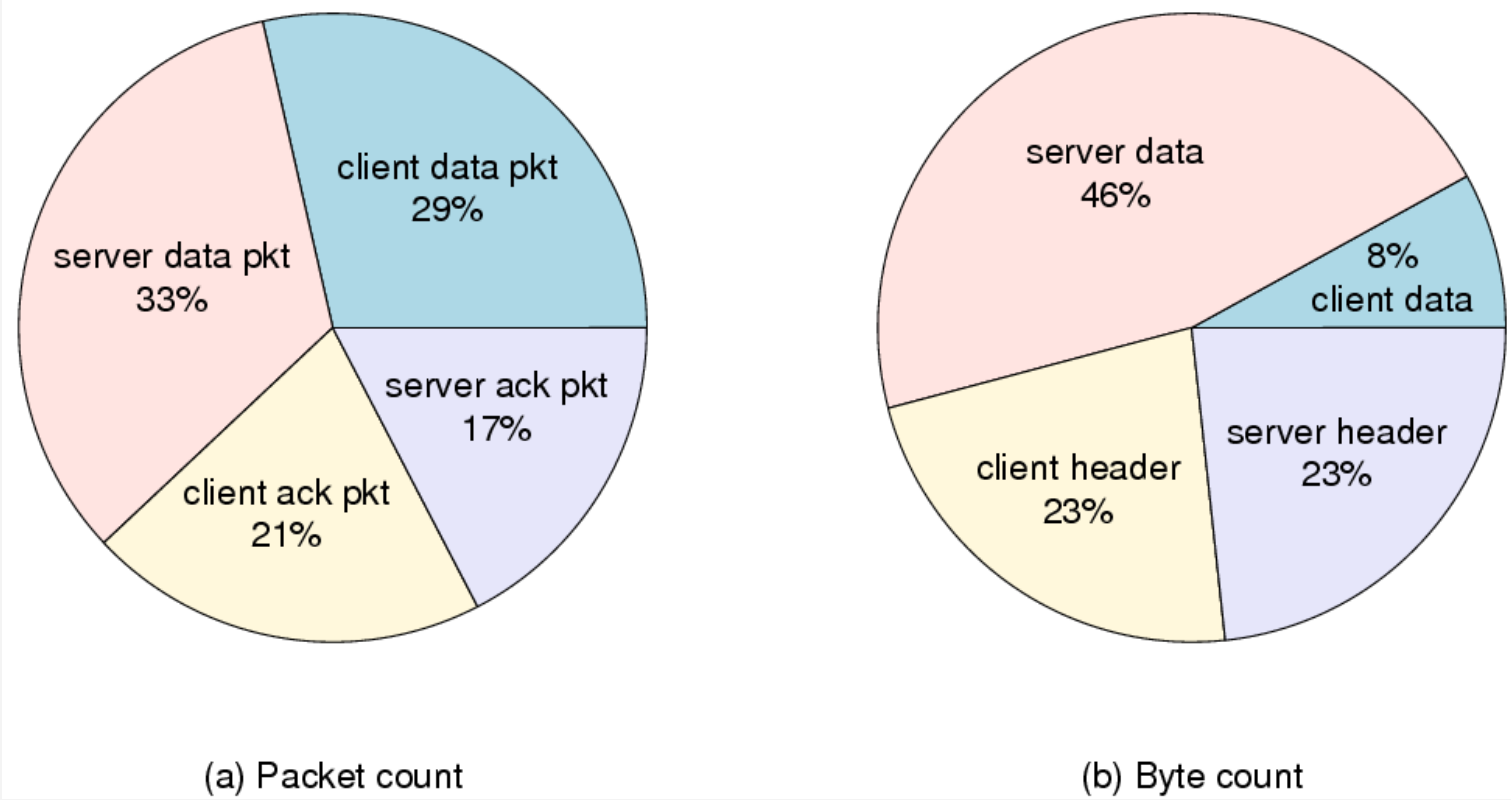
Game Traffic Characteristics

- Small packet size
  - Most of client payload < 32 bytes
  - Average pkt size 84 bytes
- Low packet rate
  - Client packet rate is 8 pkt/sec (avg.)
  - Most of server packet rate < 5 pkt /sec

TCP Performance Analysis

- Protocol overhead
- In-order delivery
- Congestion control
- Loss recovery

Protocol Overhead

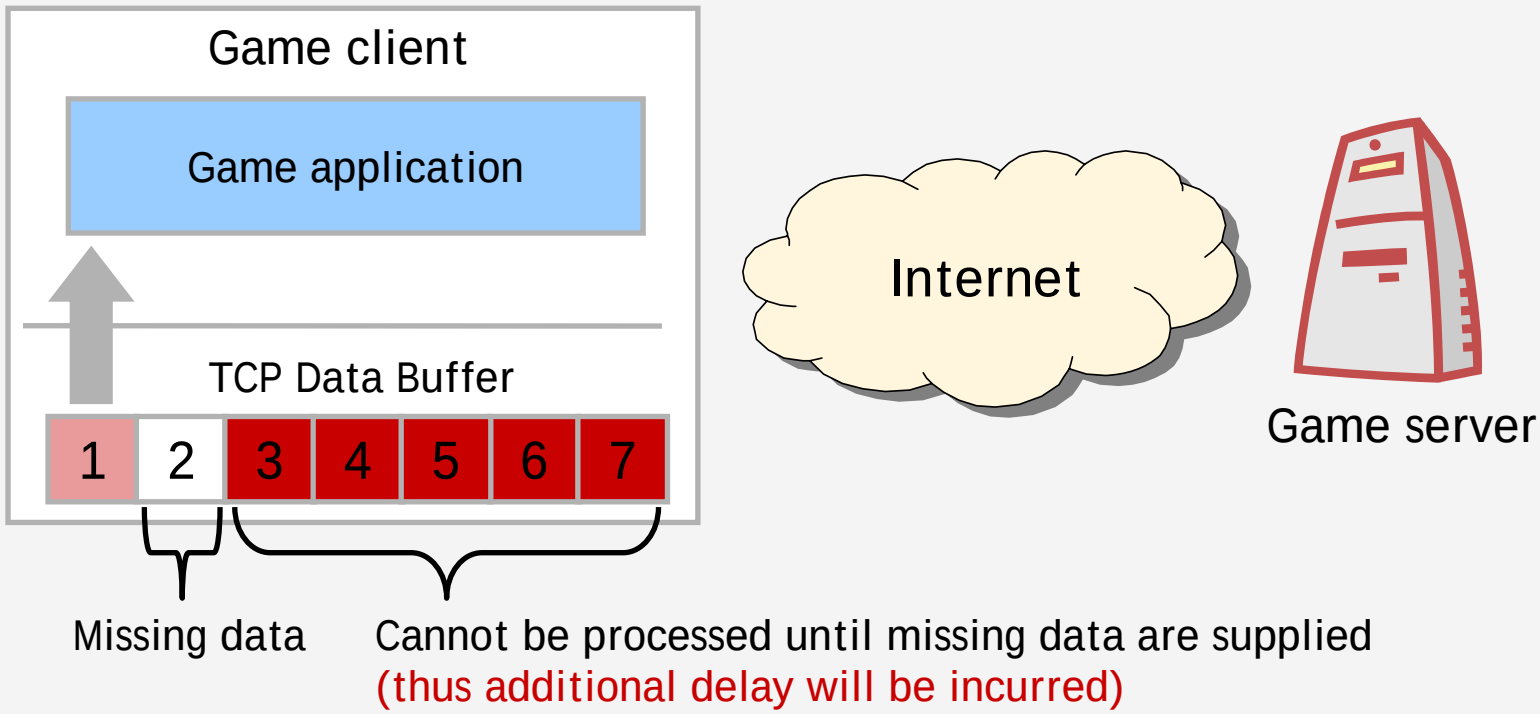


38% packets are TCP ACK packets

46% bandwidth are used by TCP/IP headers

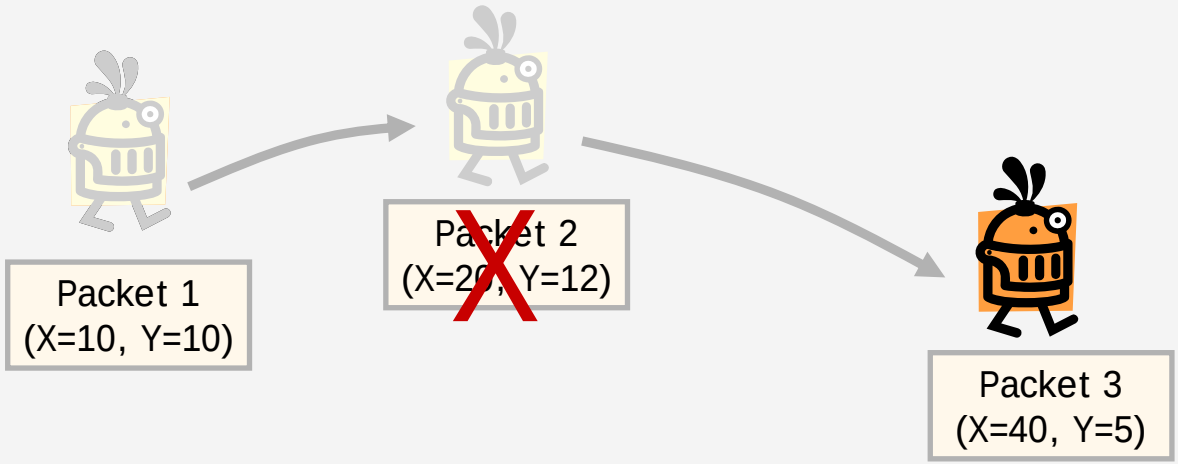
In-Order Delivery

- TCP enforces byte-level in-order delivery
- Cons: a dropped packet causes a stall in subsequent network data until that packet is delivered



In-Order Delivery is Not Always Required

- Server to client: state updates, dialogue messages, responses to users' queries
- Client to server: many commands are accumulative, e.g., position updates

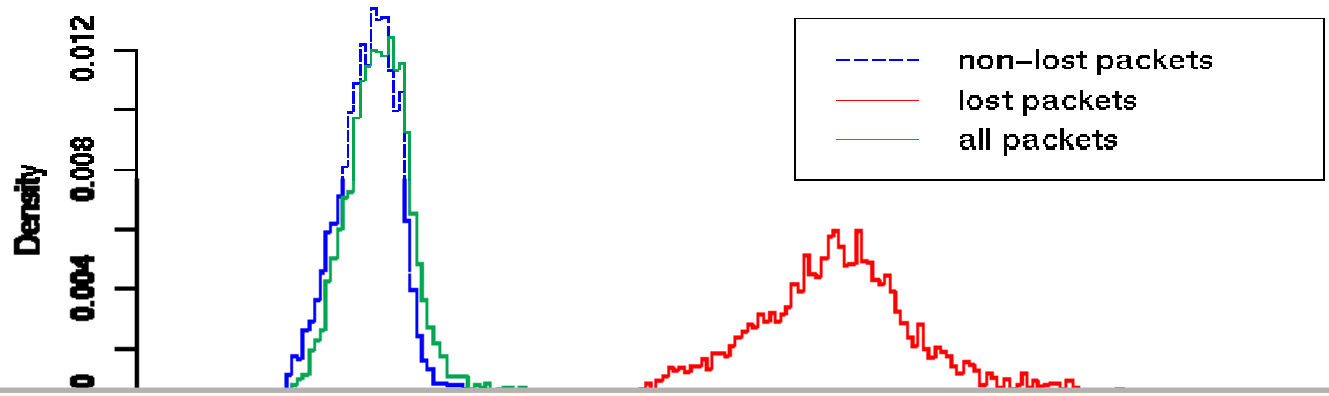


Processing packet 3 without waiting for packet 2 is OK and leads to more smooth game play

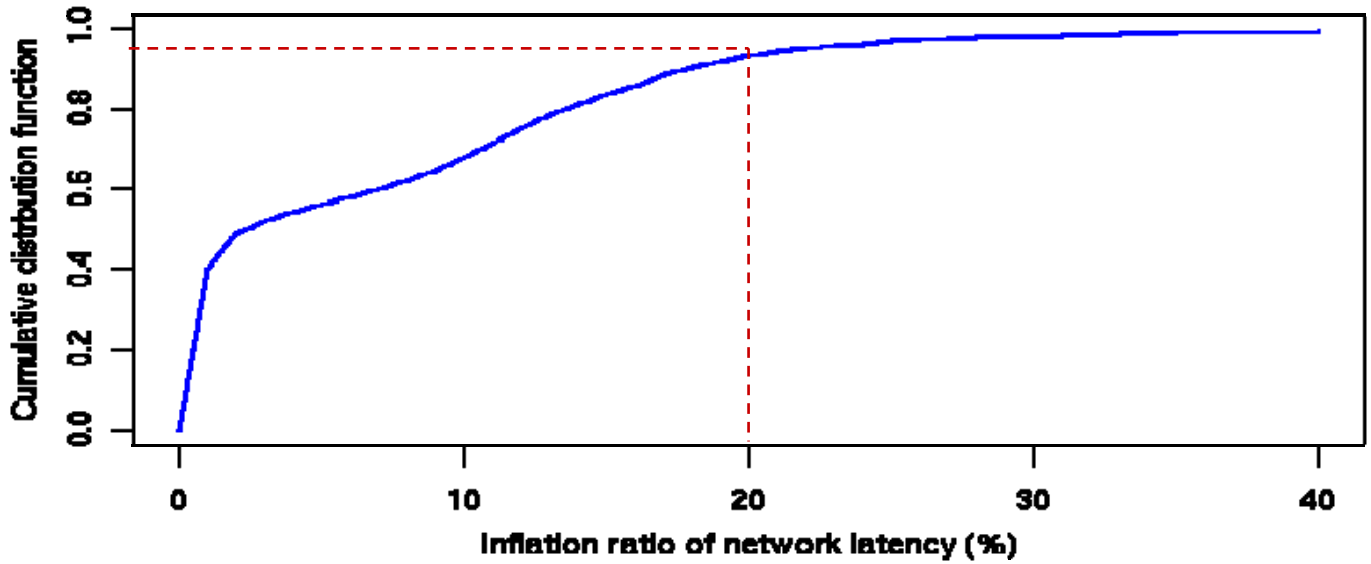
Overhead of In-Order Delivery

- Assuming an ideal case that all packets can be processed in any order
- The overhead of in-order delivery is measured by
  - The increase of round-trip times (RTT)
  - The increase of RTT jitters (std. dev.)

The Effect of In-Order Delivery on RTT

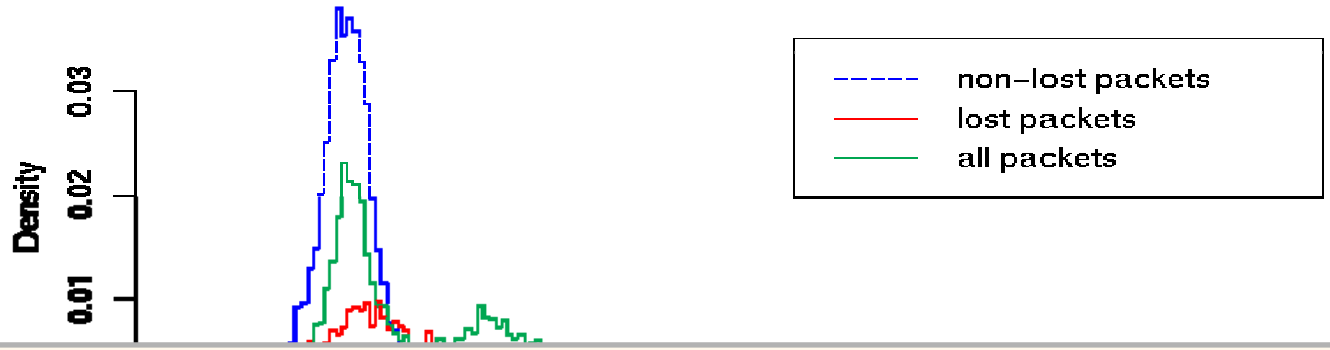


7% connections incurred more than 20% additional average RTTs.



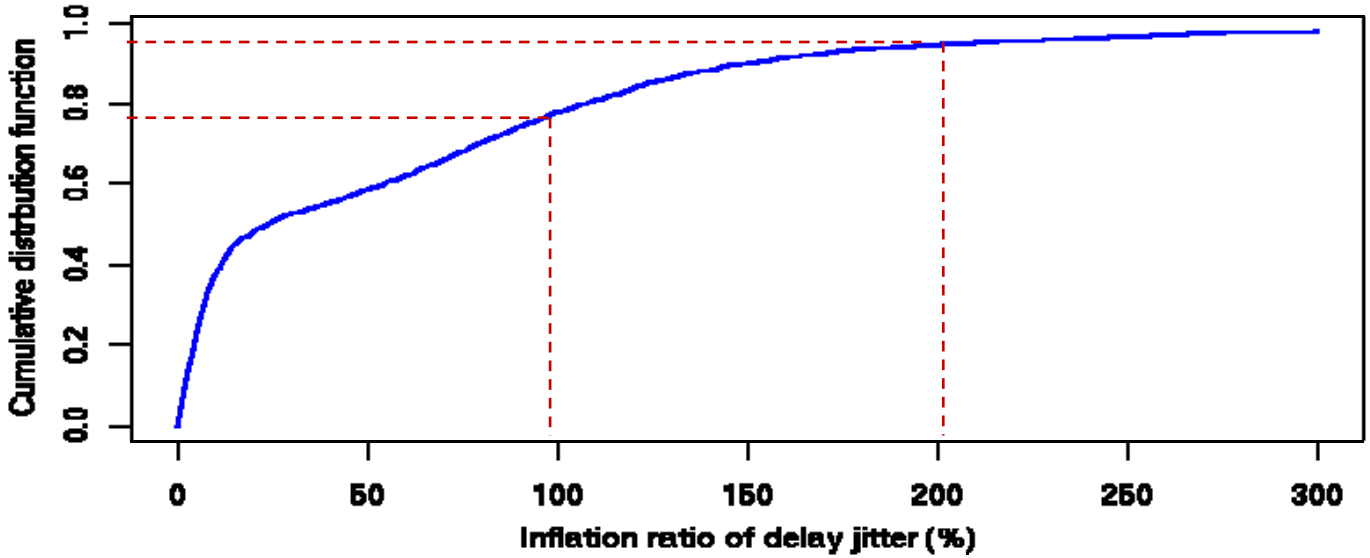
The Effect of In-Order Delivery on RTT Jitter

(std. dev. of RTTs)



22% connections incurred more than 100% additional RTT jitters.

6% connections incurred more than 200% additional RTT jitters.



Unbounded Congestion Window

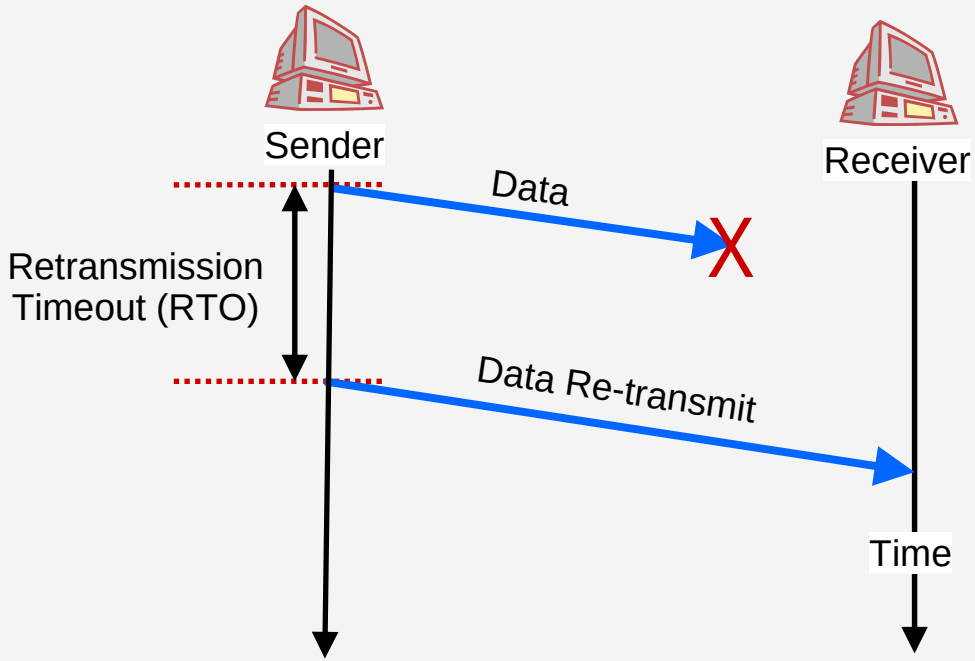
- TCP's congestion control is designed for network-limited application
  - Relies on packet loss to adjust its congestion window
- Data generation in online games is often application-limited
  - 36% connections never experienced a packet loss
- Congestion window may never be reduced ➔ unreasonably large window
- If the game may occasionally generate a large burst of packets ➔ unnecessary network congestion

Unnecessary Congestion Window Reset

- The "restart after idle periods" policy:
  TCP resets its window to 2 if a connection stops sending data for a short interval (usually < 1 second)
- The policy prevents inappropriate bursts of packets being sent due to an out-of-date window
- But, game packet rate is so low ➔ window is occasionally reset (18% of packets faced a window reset)
- In this case, a series of three commands following a thinking time will be penalized (the 3rd packet will be delayed until the first two have been delivered)

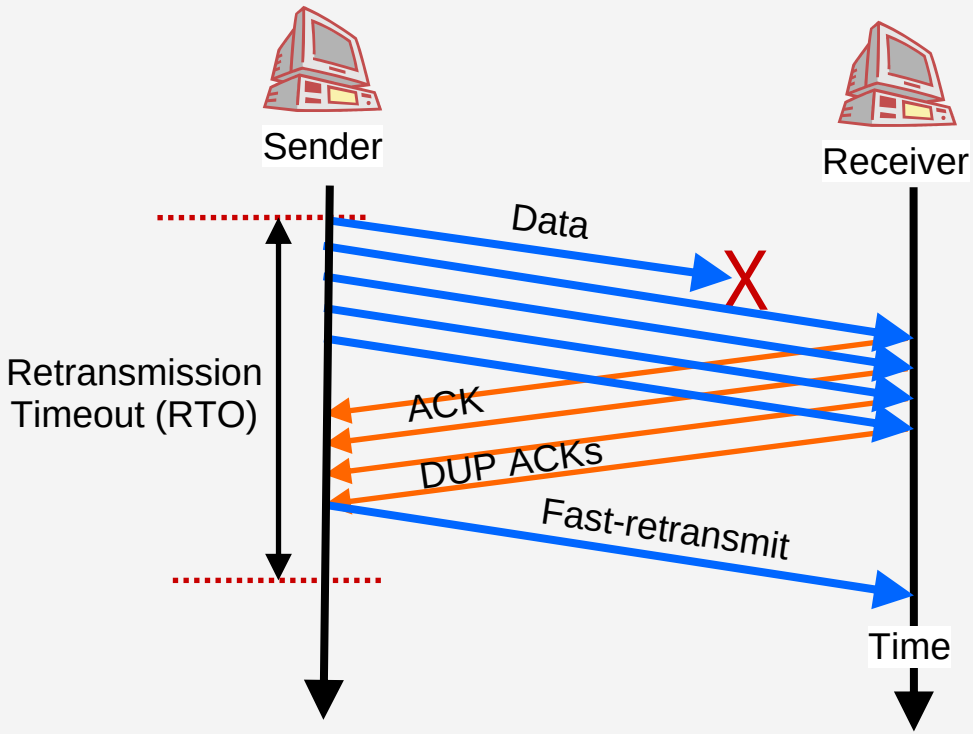
Loss Recovery -- Retransmission TimeOut

- Originally, TCP only relies on a timer to detect packet loss and resend packets



Loss Recovery -- Fast Retransmit

- TCP Tahoe/Reno improves loss detection with duplicate acknowledgement packets



The Failure of Fast Retransmit

- Insufficient duplicate acks
  - The game packet rate is too low
  - More than 99% dropped packets were not detected by fast retransmit
  - Average latency for non-dropped packets was 180 ms; Average latency for dropped packets was 700 ms
  - This is why in-order delivery incurred so much overhead in transmission latency and delay jitters
  - packet not contain data
  - The fast retransmit may not be triggered even sufficient dup acks are generated

Performance Impact on User Experience

- Based on the model describing the relationship between player departure rate and RTT, RTT jitter, and packet loss rate [INFOCOM 2006]

$$\log(\text{departure rate}) \propto 19.2 \times rtt + 4.54 \times rtt.jitter + 0.7 \times \log(closs) + 0.45 \times \log(sloss)$$

- RTT jitter is raised from 77ms to 124ms due to in-order delivery

$$\exp((0.124 - 0.077) \times 4.54) \approx 1.24$$

- Assuming the additional RTT jitters due to TCP in-order delivery can be avoided, the player departure rate is expected to decrease by 20% ($1/1.24 \approx 0.8$)
- This corresponds to an increase of average game playing time from 100 minutes to 135 minutes

Protocol Design Guidelines (1)

- Supporting both reliable and unreliable delivery
  - Some packets can be safely discarded without affecting gaming experience
  - E.g., A gesture of a character faraway from the notified character may need not to be reliably transmitted
- Supporting both in-order and out-of-order delivery
  - Only ordering packets whenever absolutely necessary
  - E.g., Ordering is irrelevant for repeated attack actions (fight the same enemy with the same weapon)

Protocol Design Guidelines (2)

- Accumulative delivery
  - A new messages can override all the previous ones
  - Missing some packets in a series of accumulative commands does not matter (unless the last one in series is also dropped)
  - E.g., position updates
- Multiple streams
  - Make messages as independent as possible
  - Messages need to be ordered only when they are in the same stream
  - E.g., chat messages are independent of game play commands

Protocol Design Guidelines (3)

- Coordinated congestion control
  - Tens of thousands of flows are not unusual
  - Difficult for all the flows to achieve an efficient bandwidth sharing by competition
  - Make congestion control in a coordinated, rather than competitive, manner
  - E.g., avoid dispatching game message synchronously for all game clients ➔ alleviate traffic burstiness ➔ reduce overall packet loss rate

Summary

- TCP is designed for uni-directional bulk transfer, but game traffic has many unique features:
  - Tiny packets
  - Low packet rate
  - Application-limited traffic generation
  - Bi-directional traffic
- TCP is unwieldy and inappropriate for MMORPGs
- The degraded performance did impact users' willingness to continue playing a game
- A number of design guidelines have been proposed

Thank You!

Kuan-Ta Chen