Lecture 6: Moving Transport Protocols to the Application Layer

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Revisiting Reliable Data Transport

- Remember the questions we asked a couple weeks ago:
  - Are there any drawbacks to requiring completely reliable, in-order packet delivery?
  - Any types of applications this might cause problems for?

Challenges for a completely reliable, ordered service

- What happens when a packet is lost?
  - **All** following packets are blocked from being delivered until that packet is recovered (head of line blocking)
- Good fit for certain applications (e.g. file transfer, remote login)

- When might this be a problem?
  - When applications have **real-time constraints** and would prefer to drop the packet (e.g. interactive video, audio, gaming)
  - When the application doesn't need **a single total order** on all packets
    - e.g. Web browsing: each webpage typically consists of multiple distinct files (base html file, images, embedded videos other elements)

Supporting new applications (or improving performance of old ones)

- What can we do for these applications?
- One option: just use UDP
- But, that doesn’t give us what we want either…
  – Best effort isn’t good enough!

Supporting new applications (or improving performance of old ones)

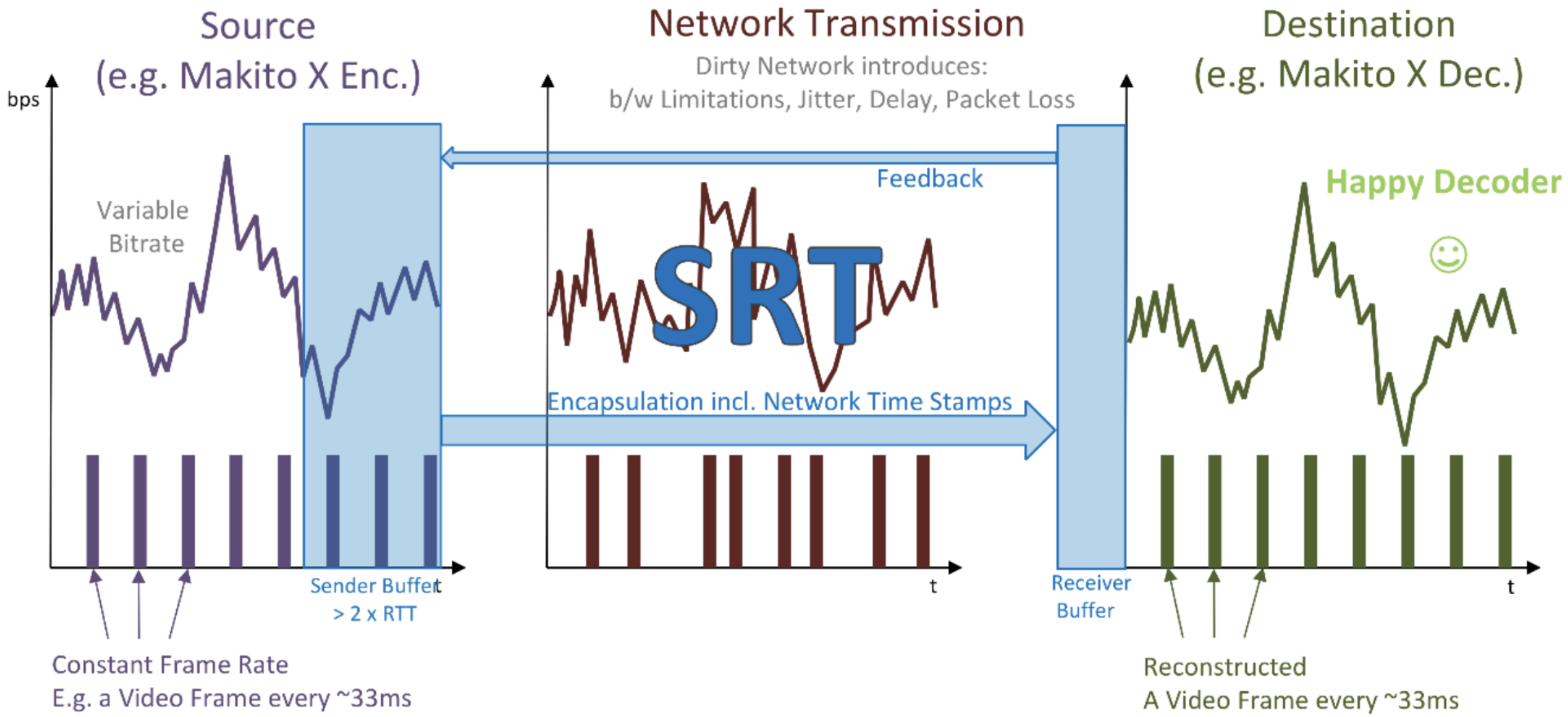
- We want new protocols to match application needs
  - Are there transport layer protocols other than TCP and UDP?
    - Yes! but, often not supported in all OSes, and therefore not widely used
    - Current trend is toward implementing **new transport layer functionality in userspace / at application layer**
  - Why? Making transport layer changes is hard!
    - Breaks others' assumptions…Firewalls block unfamiliar traffic, NAT relies on knowing header structure to rewrite IP address/ports. Timescale for standardization + adoption > 10 years
    - Requiring OS updates (to change TCP implementation in the kernel) -> slow adoption

SRT: Secure Reliable Transport

- Overall goal: provide high quality of experience for real-time video
  - Specifically targets ***video contribution*** - e.g. interview, coverage in the field that needs to be transported back to a TV studio at high quality for further distribution to end users
    - Traditional approach: satellite links – slow and expensive
  - Note that this is different from what you see as an end user watching video from Youtube, Facebook, etc
    - Those are mainly using adaptive bitrate protocols that run over HTTP (over TCP). Works well with existing CDN architectures

SRT: Secure Reliable Transport

- Overall goal: provide high quality of experience for real-time video
  - Low latency, low jitter, high reliability (few dropped packets)
  - Output stream pattern should **match** input stream pattern



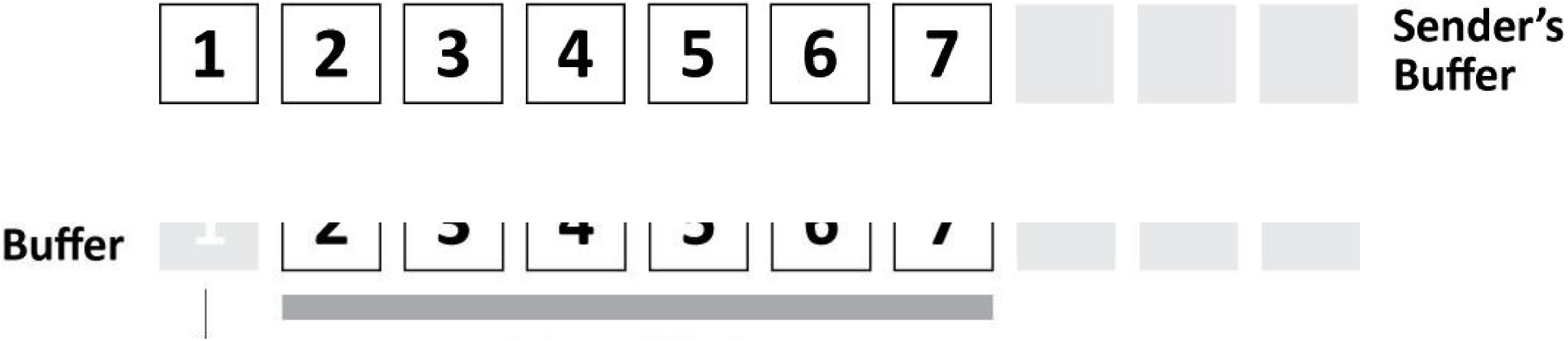
https://github.com/Haivision/srt/blob/master/docs/misc/why-srt-was-created.md

SRT: Secure Reliable Transport

- **Custom transport protocol, implemented in userspace on top of UDP**
- **Why not try to improve TCP?**
  - Reliable semantics are a bad fit for real-time video applications
    - Prefer to drop some packets vs. wait for retransmissions
    - Congestion control is problematic for fixed bit rate video
- **Why isn't UDP already good enough?**
  - *Perfect* reliability isn't a good fit, but we still want to **try** to recover lost packets (if we can do it fast enough)
  - We want to maintain a consistent delivery pattern that matches input

SRT: Key Features

- Packet-based sliding window ARQ protocol that uses cumulative ACKs and explicit NACKs (*should sound familiar!*)



So, what's new?

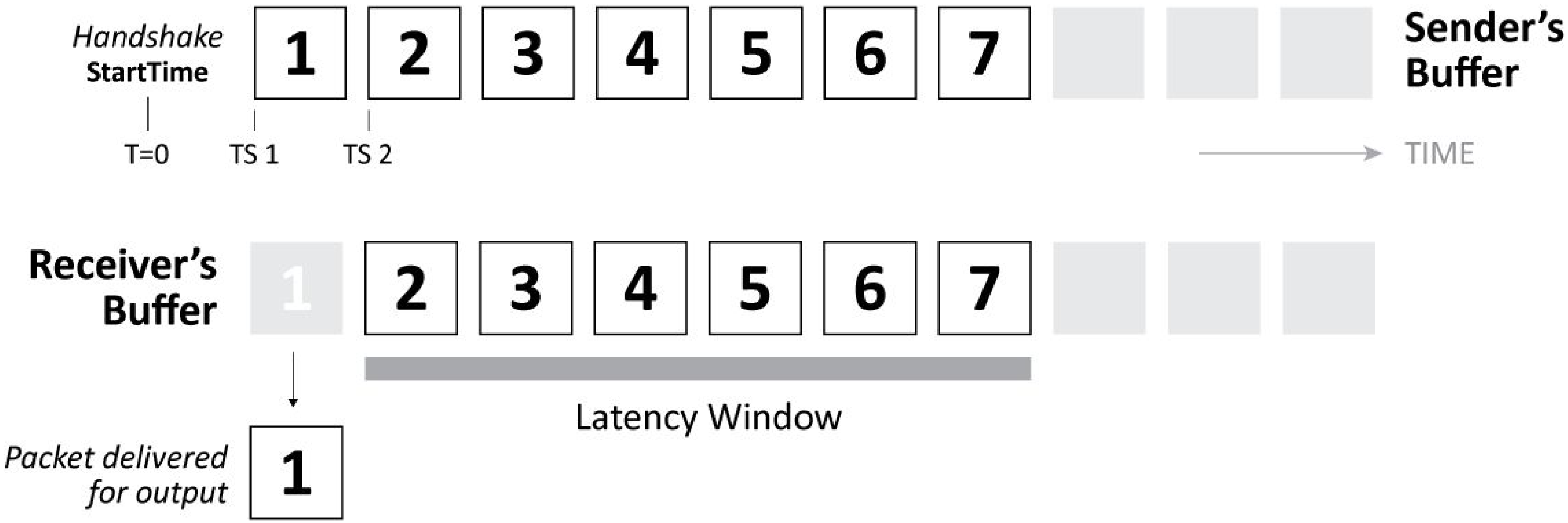
SRT: Sacrificing Reliability & Reproducing Input Timing

- **In the protocols you designed for project 1, when does your receiver decide to “deliver” a packet (write its data to file) from its buffer?**

- Is this a good fit for the application requirements we discussed for real-time video?
- What would you want instead?

SRT: Sacrificing Reliability & Reproducing Input Timing

- **Latency window** is used to determine when to deliver (or give up on) a packet in the window

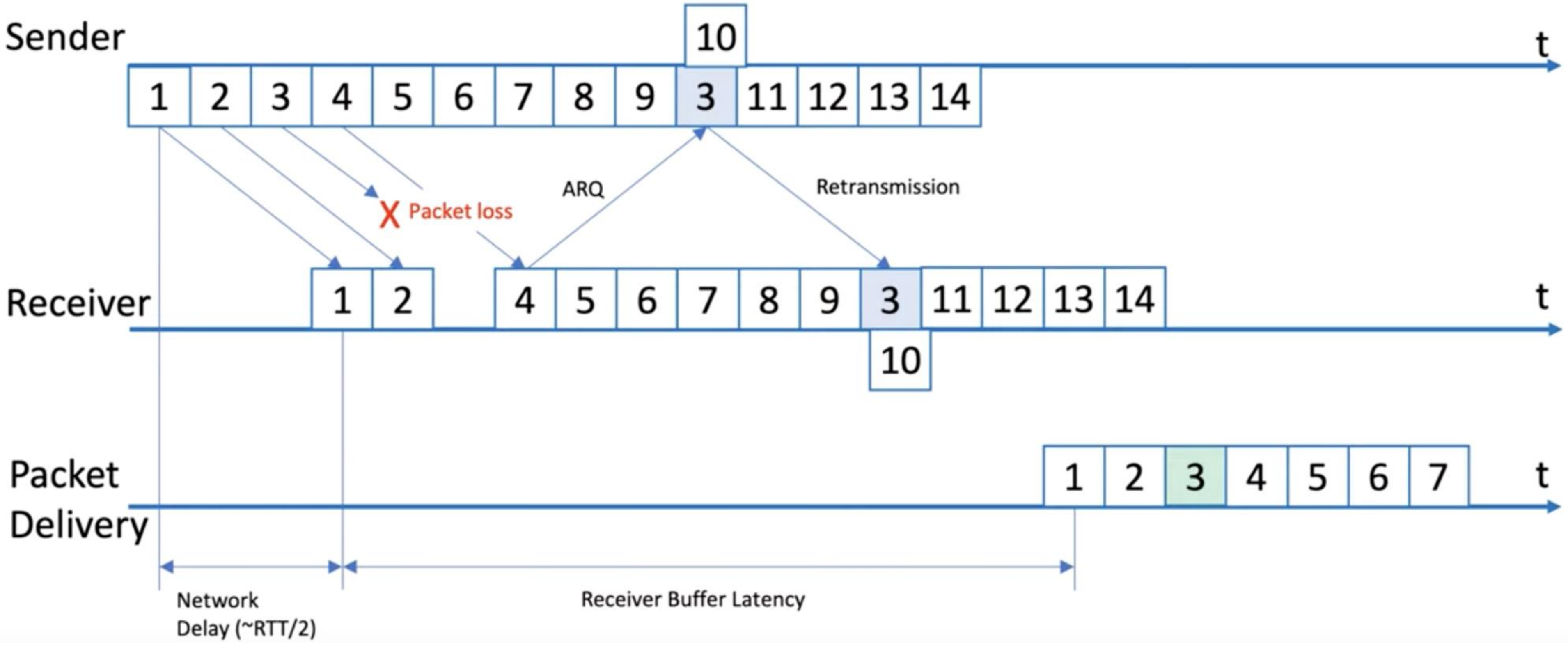


In more detail…

- Each packet gets a **timestamp** from the sender (TS1, TS2, …)
- The **latency window** is a value in milliseconds that says how long to buffer the packet
- At time TS+latency_window:
  - Receiver delivers the packet
  - Sender can remove it from its buffer (even if it hasn't been ACKed)

In more detail…

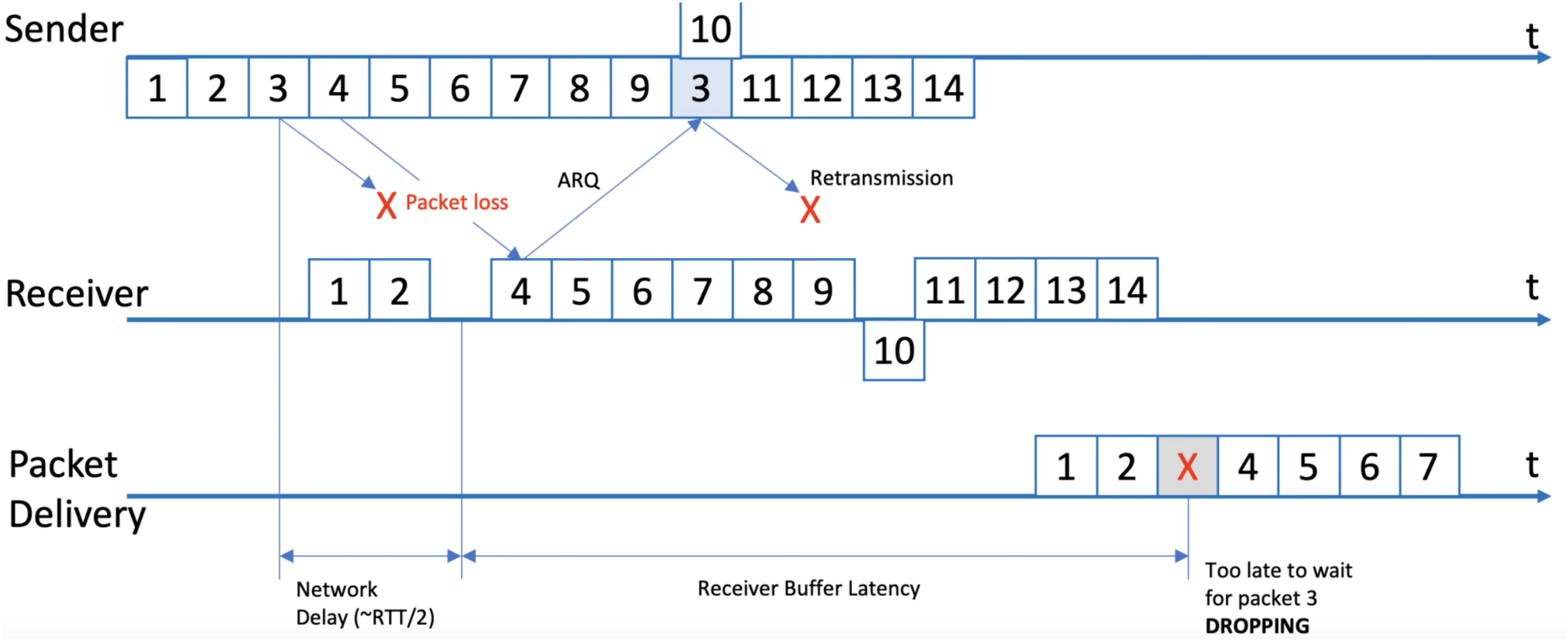
- Latency window gives some time to recover requested packets…



https://www.youtube.com/watch?v=VrE3dJej5IE

In more detail…

- But, if the time to deliver a packet arrives, the receiver delivers it, *even if some prior packets are still missing*



https://www.youtube.com/watch?v=VrE3dJej5IE

A complication

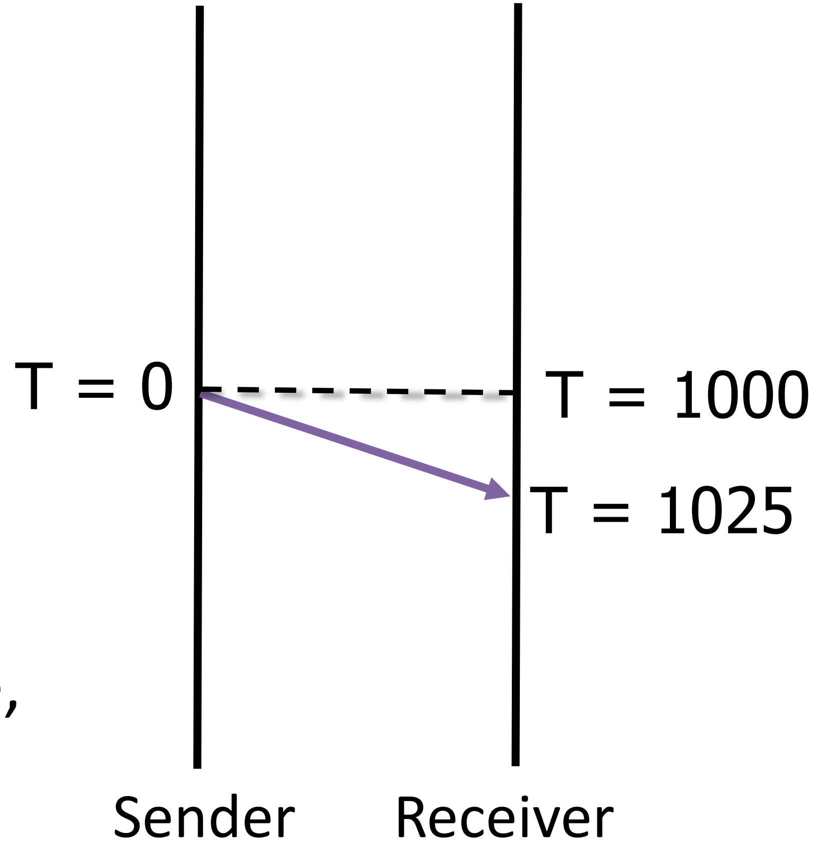
- Do you see any problem in implementing this latency window?
- Any assumptions we're making?

- Classic distributed systems problem:
  – What happens if our sender and receiver clocks aren't synchronized?!

Clock synchronization issues

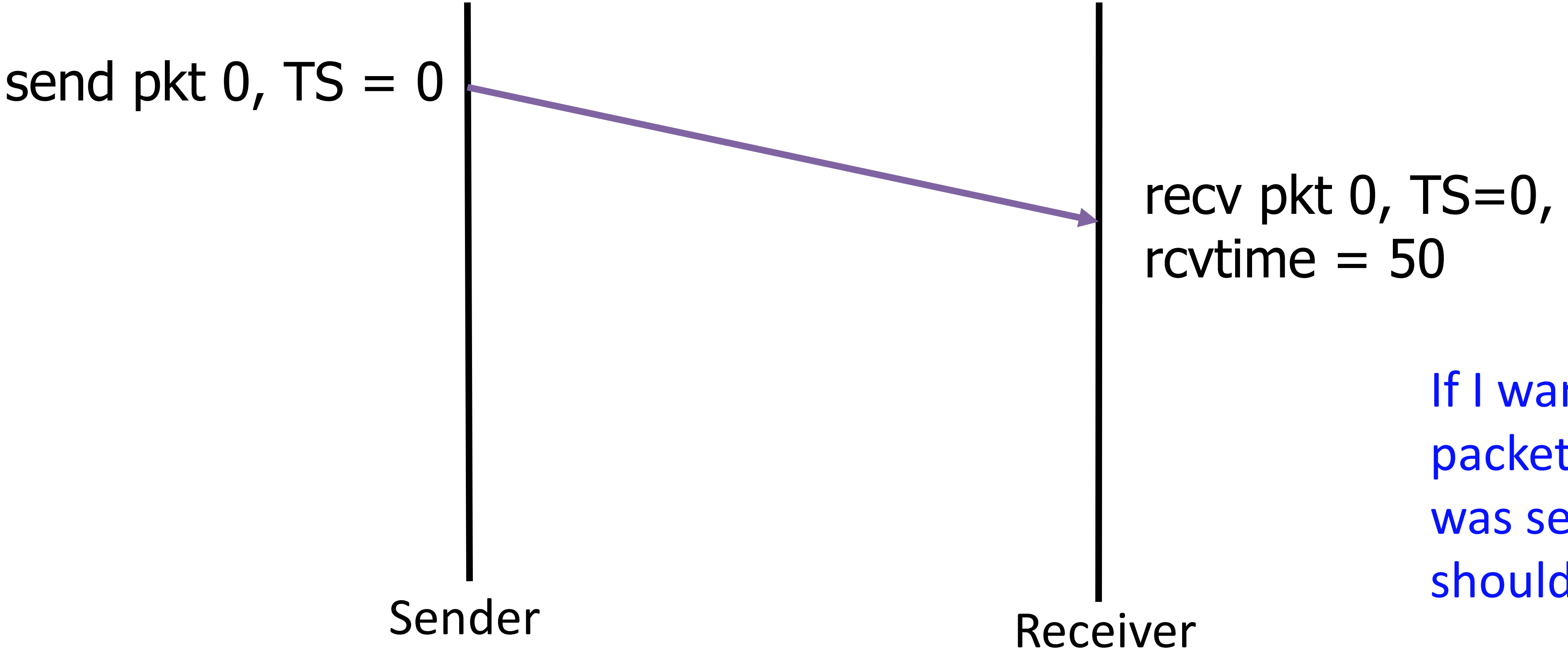
- Say our sender clock is 1 second (1000 ms) *behind* our receiver clock
- RTT between sender and receiver is 50ms
- Latency window is 500 ms

- What happens?
  - Sender sends packet with TS=0
  - Delivery time **should** be TS+latency_window = 500
  - Receiver gets it 25 ms (RTT/2) after it was sent. To the receiver, this looks like time 1025, packet is late, should just drop it.



Clock Skew Compensation

- Goal: map time on sender's clock (pkt timestamp) to time on receiver's clock



If I want to deliver a packet 500ms after it was sent, how long should I buffer it?

Clock Skew Compensation

- rcvtime – sendTS = 50
- I want to deliver the packet 500ms after it was sent (i.e. time 500 on sender’s clock)
- When do I deliver it?

- If clocks are synchronized, this is easy:
  - Target delivery time = 500, so wait 450ms, then deliver
- If I know the sender’s clock is 10ms behind mine, this is also easy:
  - Target delivery time = 500+10, so wait 460ms
  - But I don’t know this!
- **Observation: rcvtime – sendTS = oneway_delay + clock_diff**

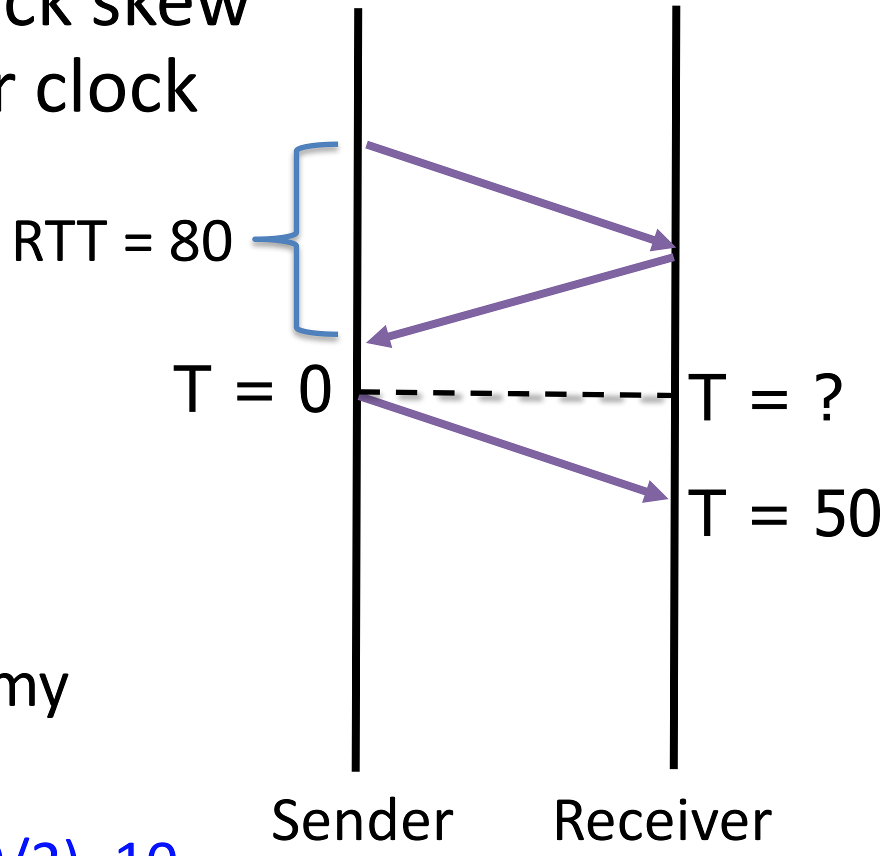
Clock Skew Compensation

- *If I know the normal one-way delay* between sender and receiver, I can estimate the clock skew to translate sender timestamps to receiver clock
- But, I don't know the one-way delay!
- But I can measure the roundtrip…

rcvtime – sendTS = RTT/2 + clock_diff

clock_diff = rcvtime – sendTS - RTT/2

then, I can use clock diff to translate sendTS to my (receiver's) local clock



clock_diff = 50-0-(80/2)=10

Clock Skew Compensation

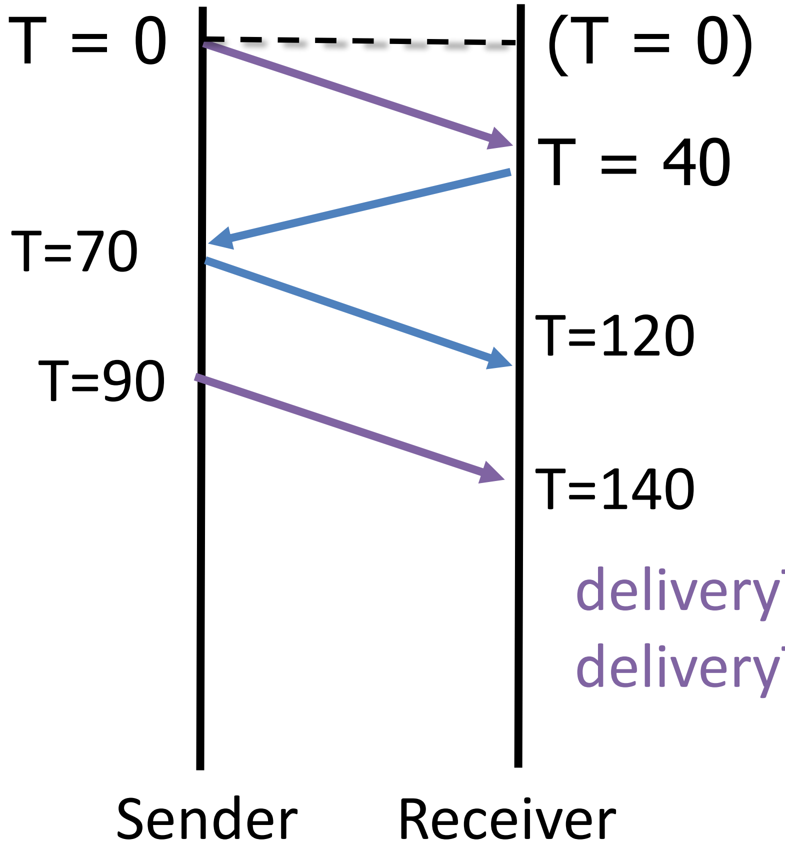
- In practice, it’s a bit more complicated…
- Clocks can drift, so offset between sender clock and receiver clock changes over time
- RTT can change over time
  - Rerouting, congestion
- Need to measure periodically and update our estimates

SRT Approach

- SRT's goal is to maintain the same delivery offset from senderTS despite clock drift and changing RTT
- Slightly different approach than what we described: not trying to "subtract out" network delay
- Basic mechanism:
  - Initialize baseDelta between sender and receiver as rcvtime – sendTS (includes both network delay and clock skew)
    - deliveryTime = sendTS + baseDelta + latencyWindow
  - **RTT Measurement**: sender sends ACKACK in response to ACKs to allow receiver to calculate RTT (time from sending ACK to getting ACKACK) and update its estimate on each ACK
    - use these measurements to adjust baseDelta

Maintaining smooth delivery with clock drift

clock drift! network delay is 40ms, but sender clock slowed down, T=70 (instead of 80)



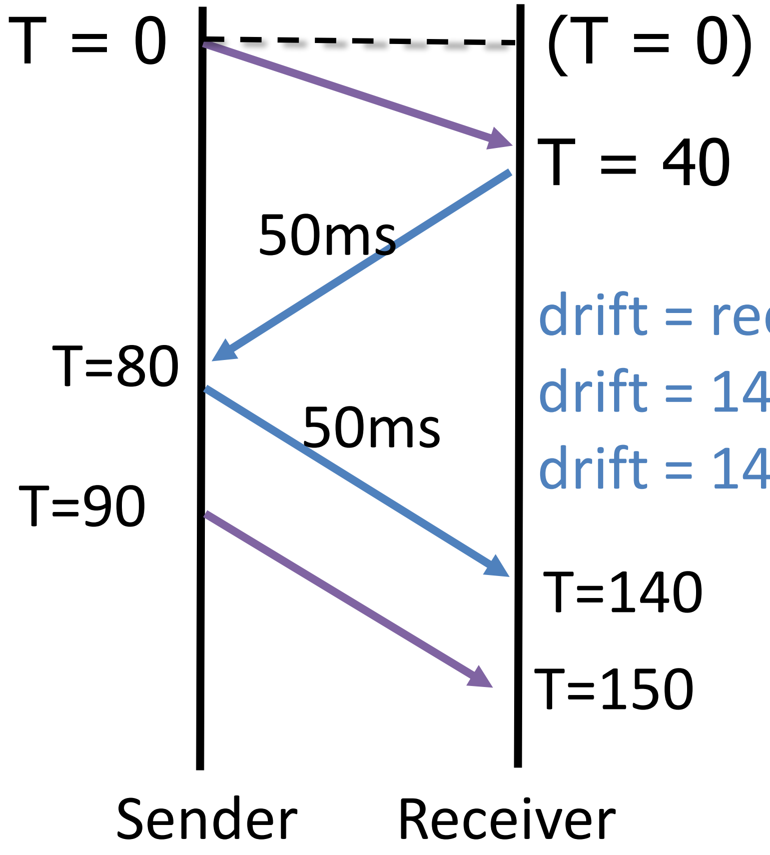
let latencyWindow = 500
baseDelta = 40-0 = 40
deliveryTime = 0+40+500 = 540

drift = recvTime – (ACKACK_TS + baseDelta)
drift = 120 – (70+40) = 10

deliveryTime = sendTS + baseDelta + drift + latencyWin
deliveryTime = 90 + 40 + 10 + 500 = 640

Maintaining smooth delivery with clock drift + changing RTT

clock drifted (sender is now 10ms behind) AND RTT increased from 80ms to 100ms



let latencyWindow = 500
baseDelta = 40-0 = 40
deliveryTime = 0+40+500 = 540

drift = recvTime – (ACKACK_TS + baseDelta) - **deltaRTT/2**
drift = 140 – (80+40) - (100-80)/2
drift = 140 – 120 - 10 = 10

So, I can say T=90 -> T=100 on **rcv clock** if I want to maintain 540ms offset from when the packet was really sent to when it is delivered, deliver at time T=640

Congestion control?

- LiveMode vs FileMode
- LiveMode is what we've been discussing, FileMode is for file transfers (and uses TCP-like congestion control)
- For video, we have a fixed rate we want to send at…just reducing the sending rate would cause problems
- Options:
  - Do congestion control by *dropping* some packets at the sender
  - Signal application to change the bitrate

QUIC Motivation

- Overall goal: Reduce latency for web applications
- New protocol, implemented in userspace on top of UDP

- Why not just try to improve TCP?
  - Making changes is hard…
  - Layering has a cost
    - TCP handshake + TLS handshake -> 3 RTTs before data can be sent
  - Totally ordered bytestream abstraction limits performance (head of line blocking)

QUIC Role in the Web Stack

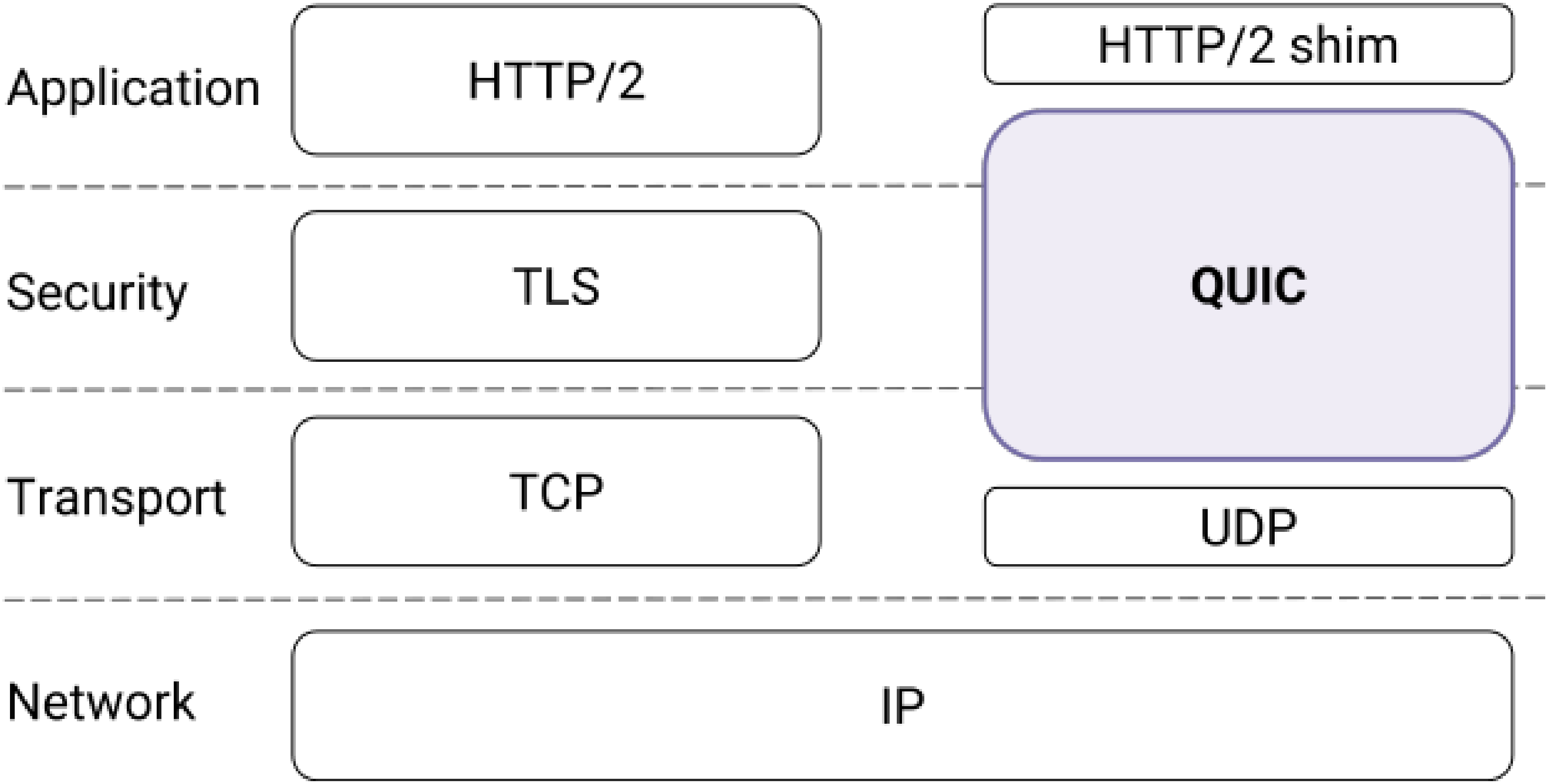
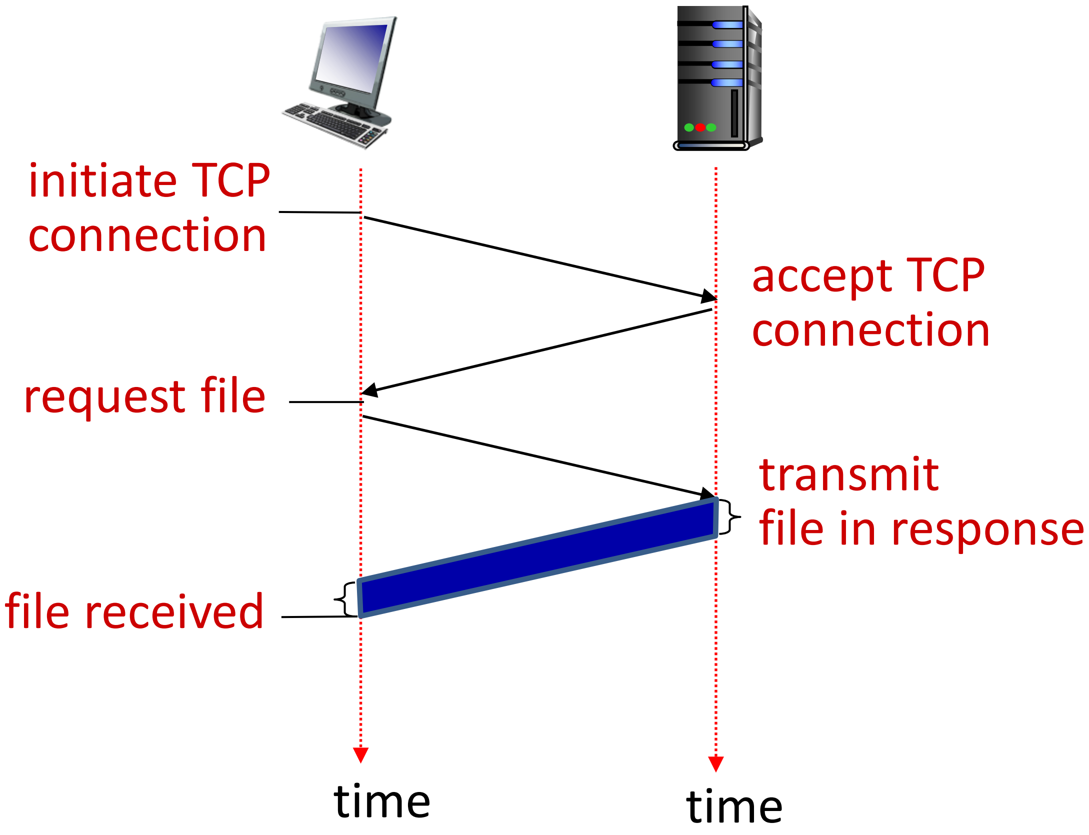


Figure 1: QUIC in the traditional HTTPS stack.

But first...a review of HTTP communication patterns

User enters URL: **`www.someSchool.edu/someDepartment/home.index`**

1. Client **initiates TCP connection** to server at www.someSchool.edu on port 80
2. Server listening for TCP connections on port 80 **accepts** the connection
3. Client sends **HTTP request** message for object someDepartment/home.index to server on TCP connection
4. Server receives request and sends **HTTP response** message containing requested object to client on TCP connection

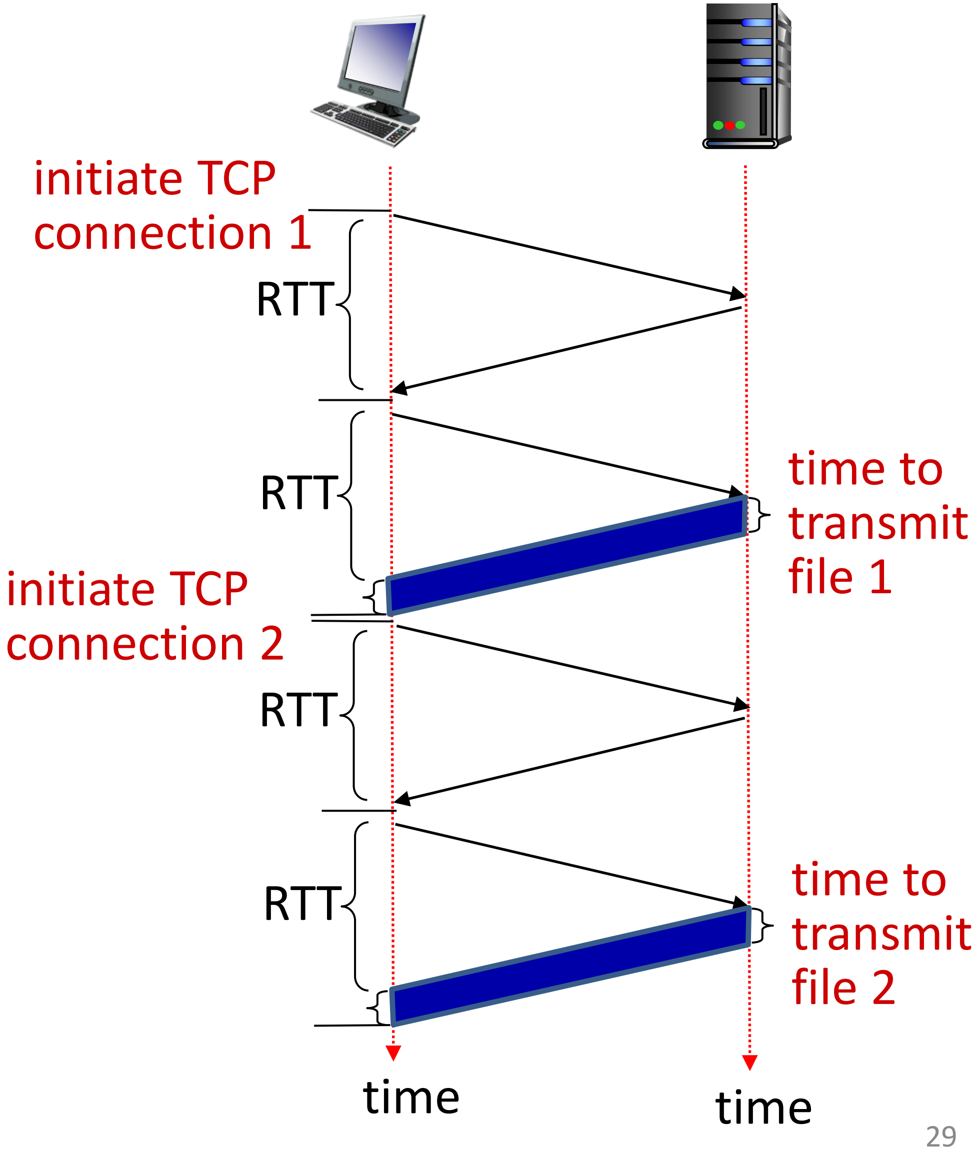


HTTP Performance

- Most webpages include more than one object…how do we retrieve all of them?

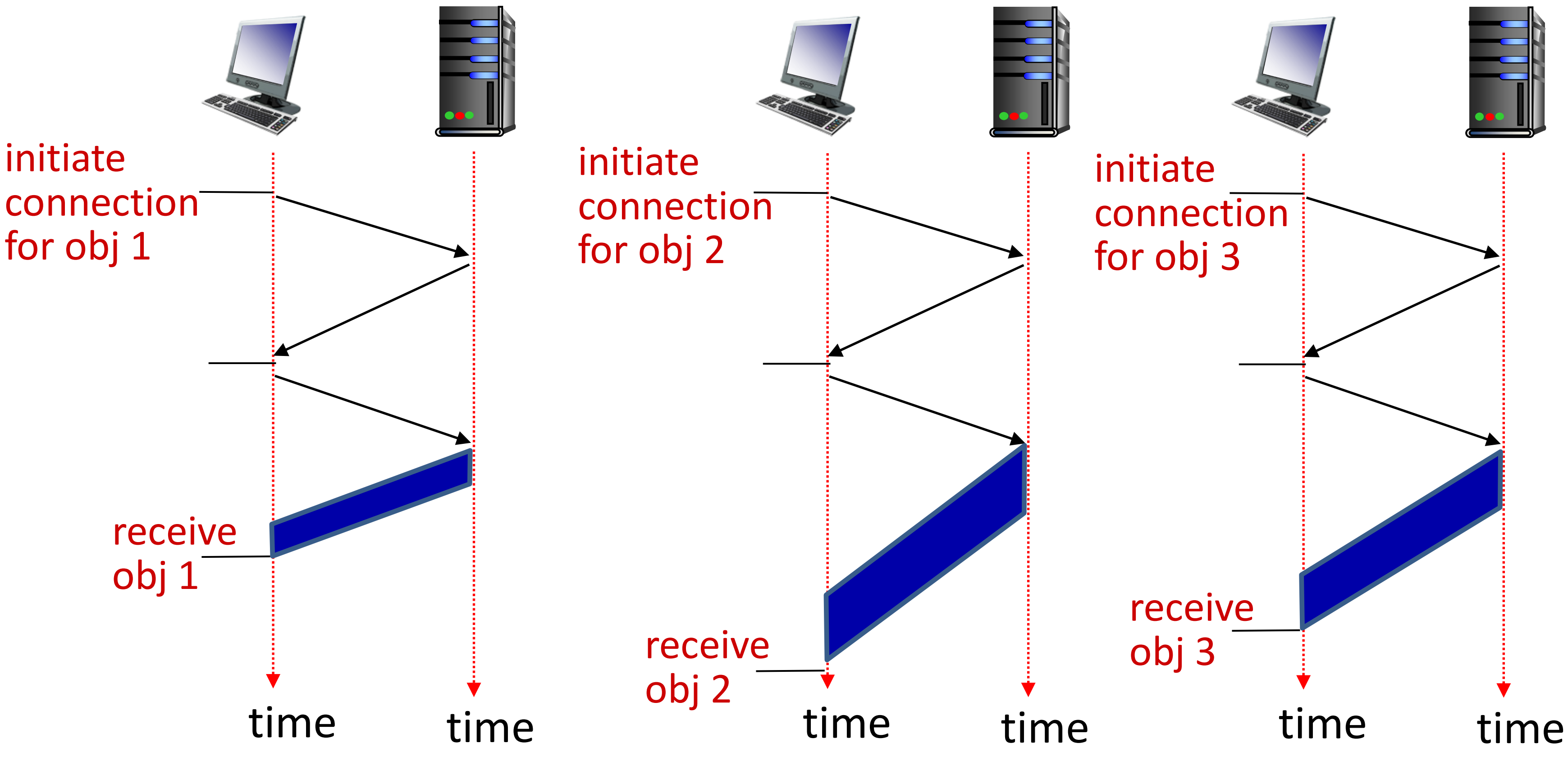
Non-Persistent HTTP (HTTP/1.0)

- Separate TCP connection for each object
- Naively, request objects one at a time (serially)… 2RTT + Transmission time **for each object**



Non-Persistent HTTP (HTTP/1.0)

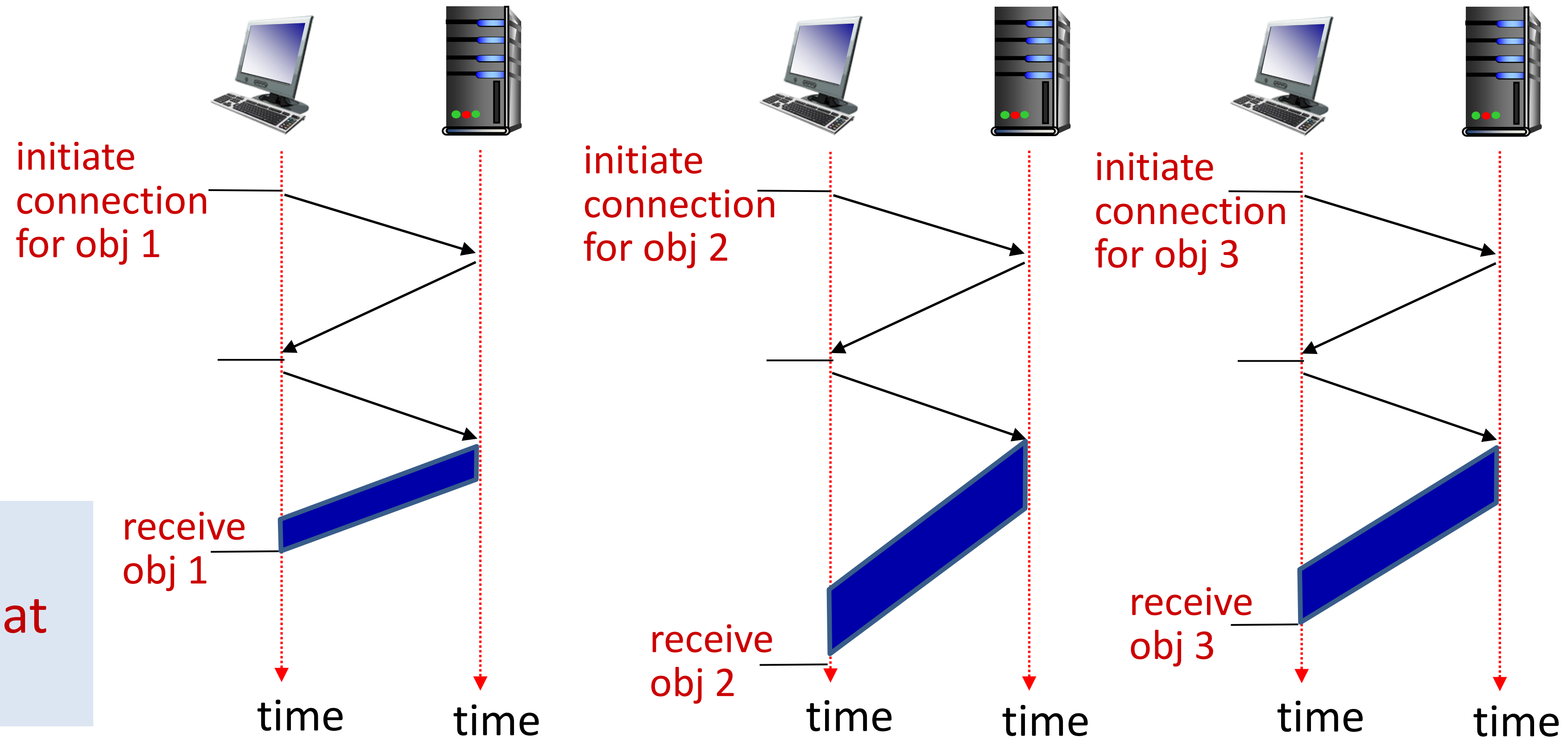
- **How can we reduce response time?**
- Separate TCP connection per object, but run them in **parallel**



Non-Persistent HTTP (HTTP/1.0)

- How can we reduce response time?
- Separate TCP connection per object, but run them in parallel

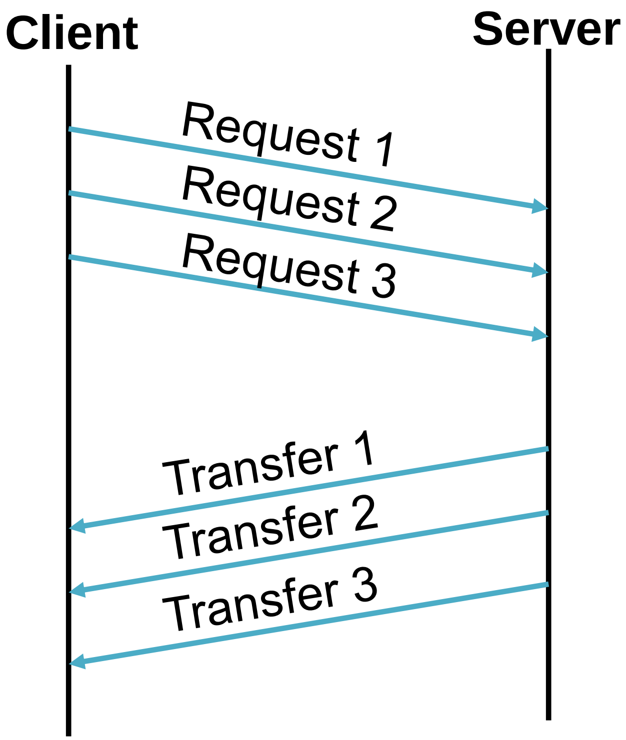
But, not so nice from a network perspective…what about TCP fairness?



Persistent HTTP (HTTP/1.1)

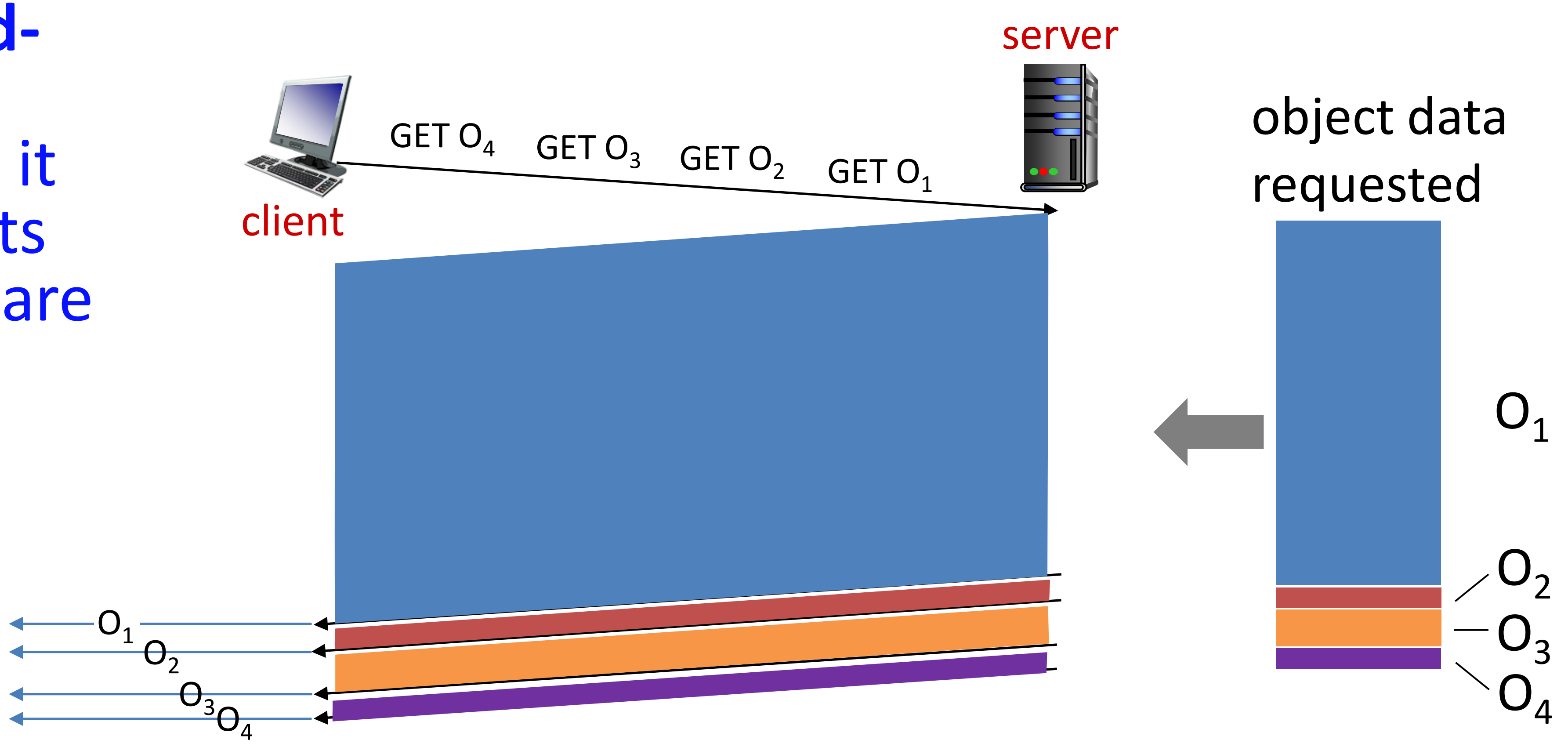
- **Maintain TCP connection across multiple requests**
  - Avoid overhead of setting up and tearing down many connections
  - Better match TCP expectations: allow TCP to learn RTT and bandwidth characteristics, support fair bandwidth sharing
- **Pipelining** to further reduce response time

Pipelined communication pattern



But, persistent pipelined option didn't really stop browsers from opening parallel connections…

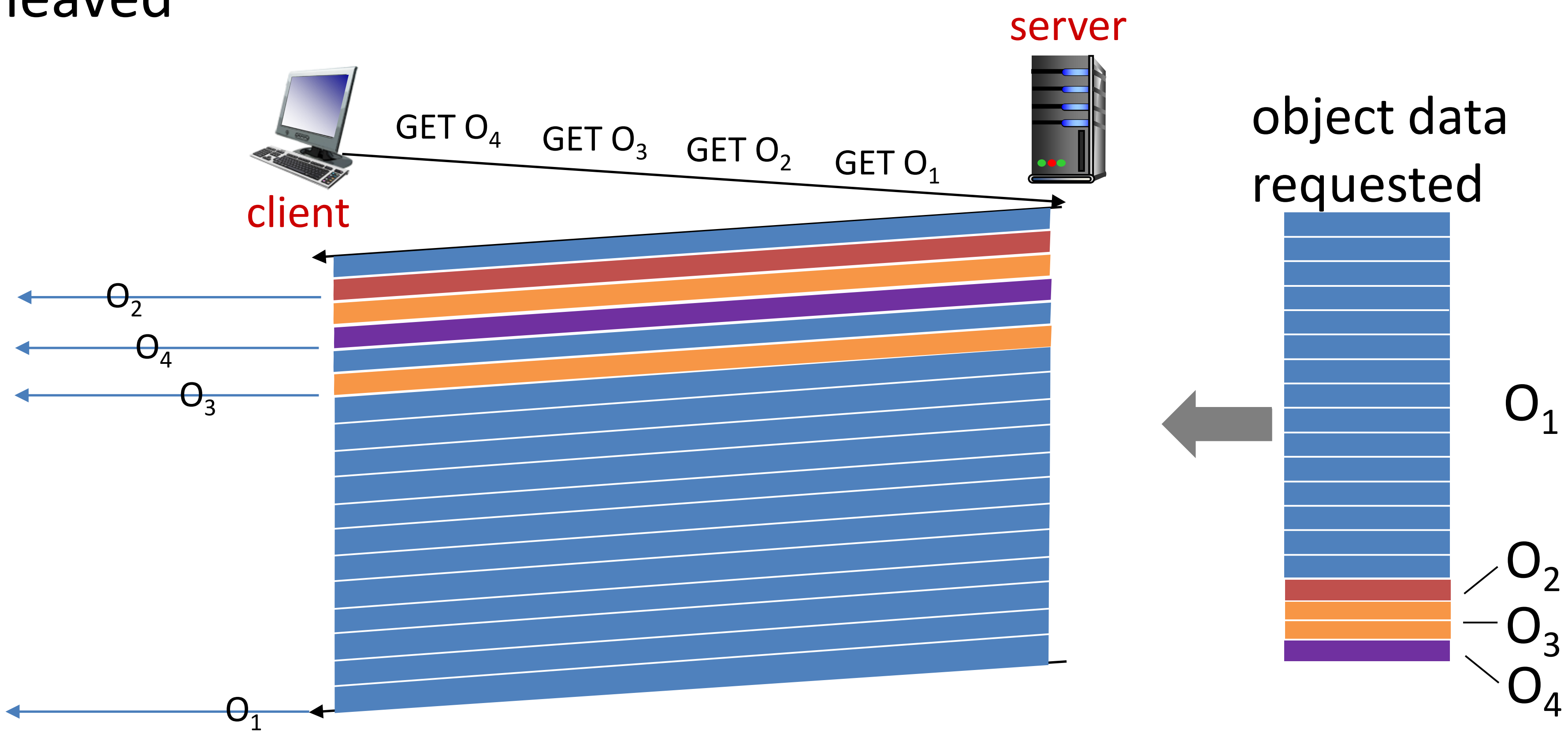
HTTP/1.1 pipelining suffers from **Head-of-Line (HOL) Blocking** because it processes requests in the order they are received



objects delivered in order requested: O_2, O_3, O_4 wait behind O_1

HTTP/2 Multiplexing

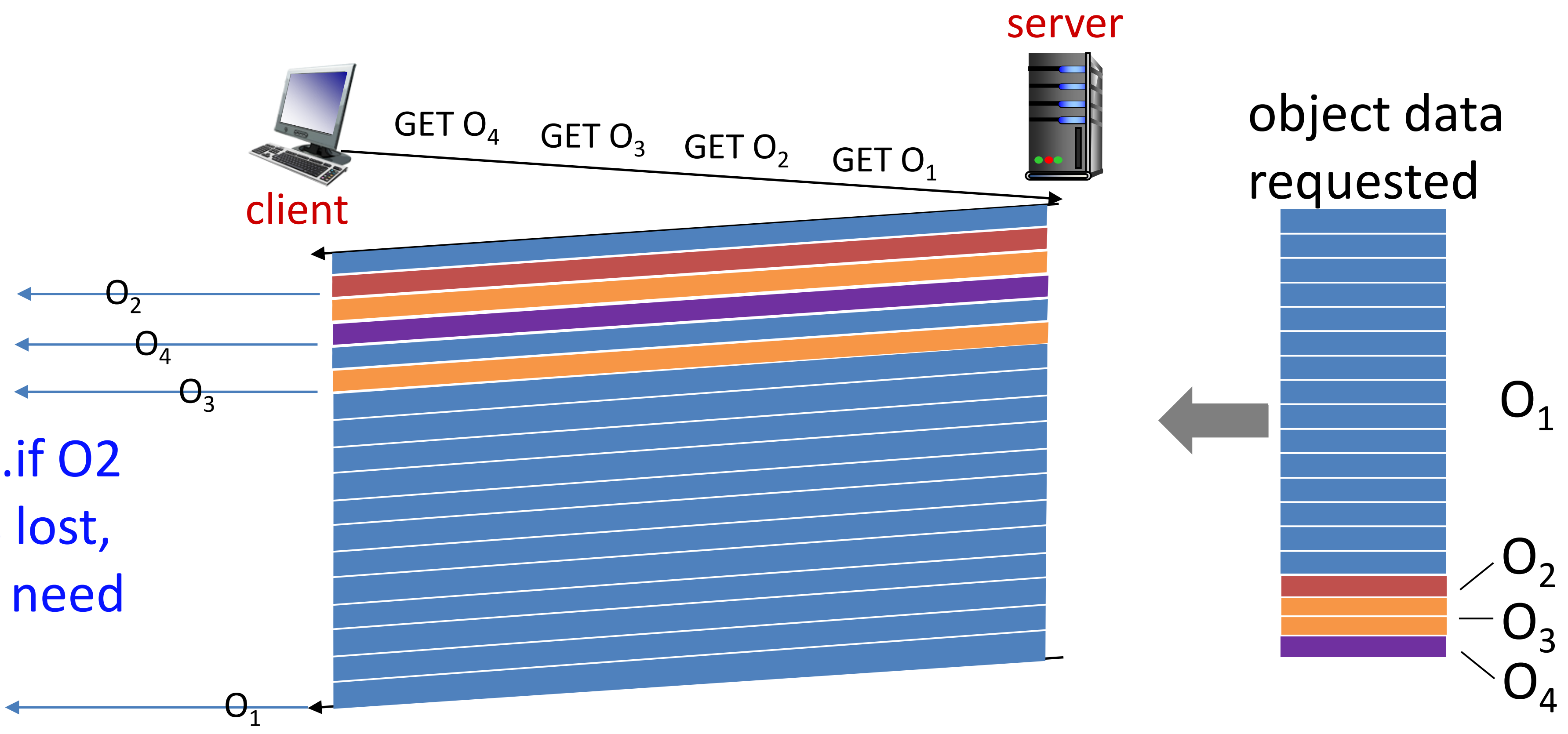
- Objects are divided into *frames*, and frames for different objects can be interleaved



O_2, O_3, O_4 delivered quickly, O_1 slightly delayed

HTTP/2 Multiplexing

- Does this solve head of line blocking problem??



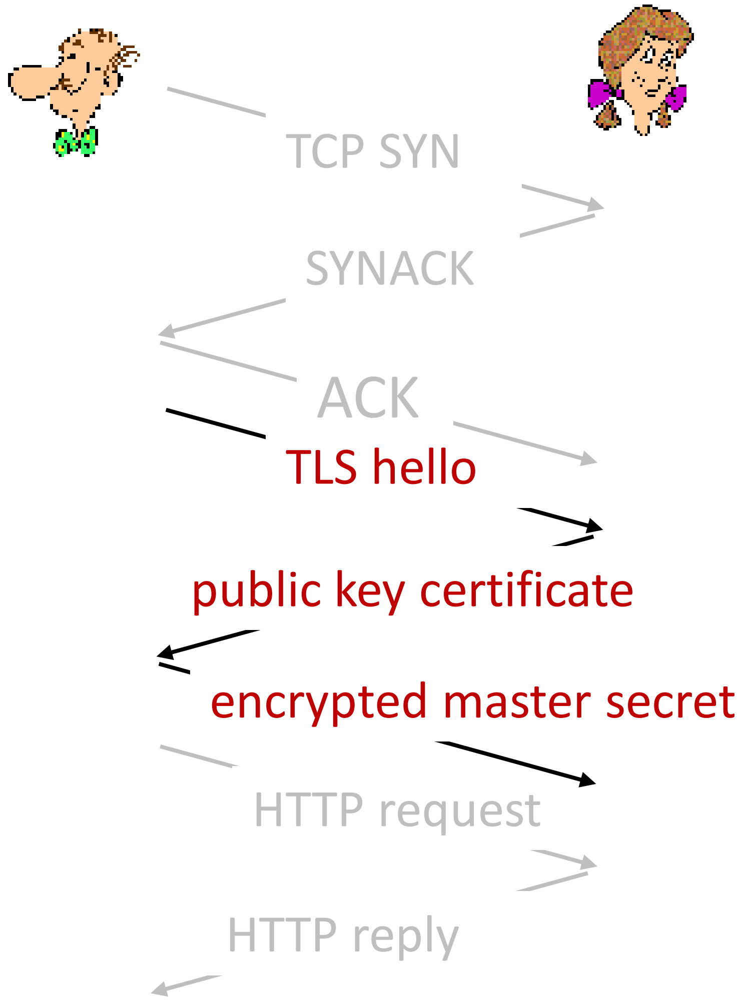
Not quite…if O2 frame gets lost, others still need to wait

O_2, O_3, O_4 delivered quickly, O_1 slightly delayed

Another performance challenge

- We care a lot more about web security than we used to!
- That's mostly a good thing, but what does HTTPS adoption mean for performance?

A simplified view of TLS



TLS handshake phase:

- Bob establishes TCP connection with Alice
- Bob **verifies** that Alice is really Alice
  - public key certificate demonstrates that some trusted authority confirms that Alice really owns this key
- Bob sends Alice a master secret key (MS)
  - MS is **encrypted with Alice's public key** (she can *only read it if she really knows the private part*)
  - MS used to generate session keys used to **encrypt** and **integrity check + authenticate** session data
    - Attacker can't read data or generate valid data without knowing the session keys

QUIC Innovation 1: Squashing the Layers

- **Combines transport handshake with crypto handshake**
- Also improved the handshake itself:
  - Reduces crypto handshake that took 2 RTT in TLS 1.2 to single RTT (now part of TLS 1.3)
  - Caches information about a a given origin to enable 0-RTT handshake for connections to known origin (now part of TLS 1.3)

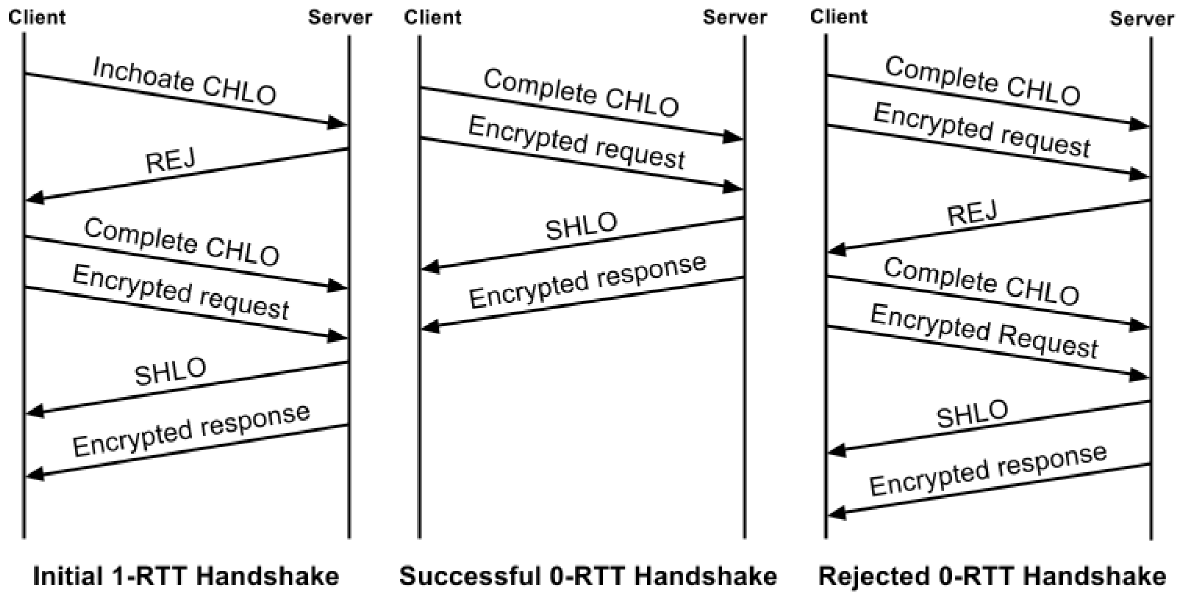
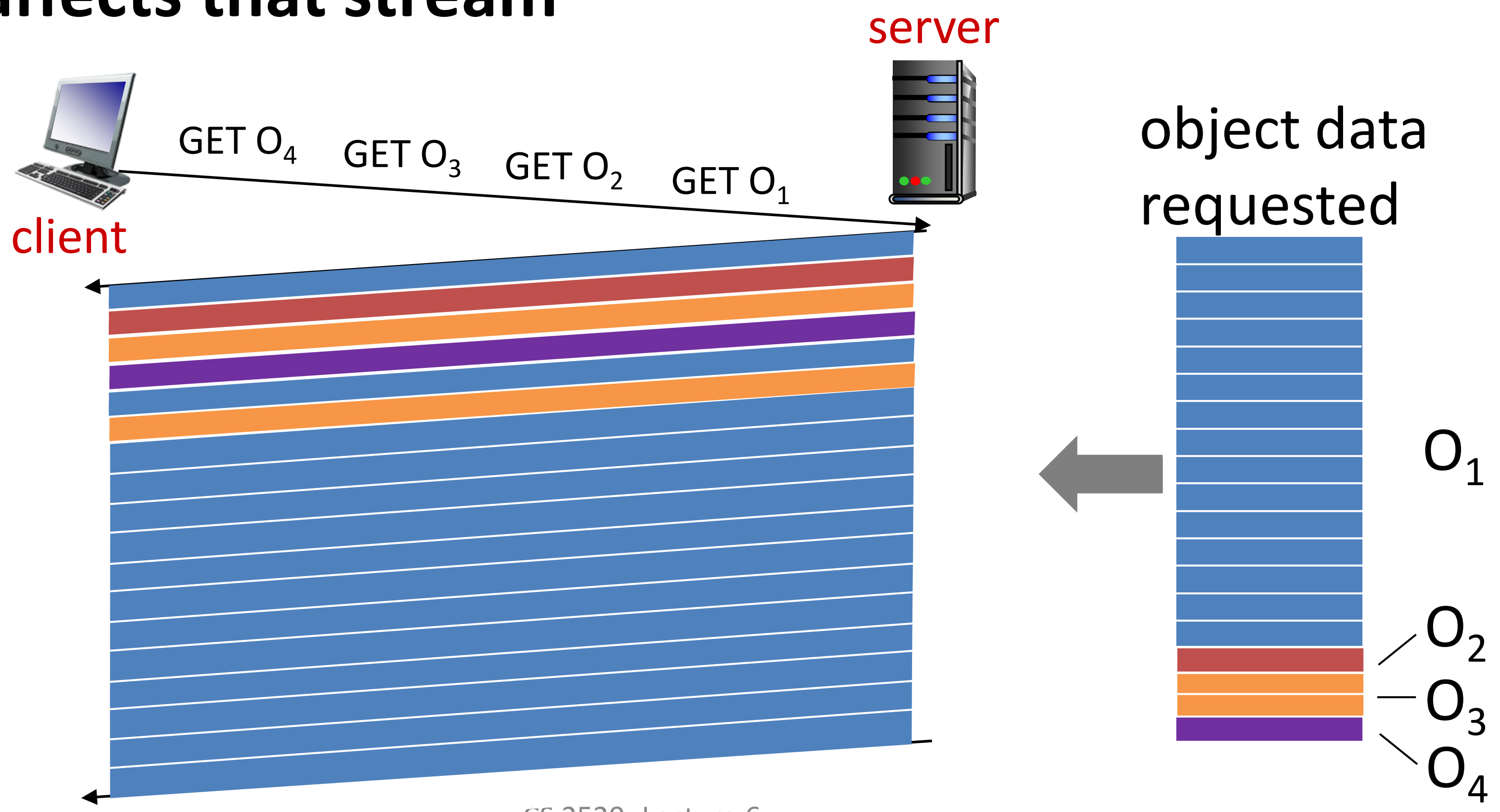


Figure 4: Timeline of QUIC's initial 1-RTT handshake, a subsequent successful 0-RTT handshake, and a failed 0-RTT handshake.

QUIC Innovation 2: Stream Multiplexing

- Changes the single *totally ordered* byte stream abstraction
- Explicitly provides a “stream” abstraction, where a lost packet in one stream **only affects that stream**



“Fixing” TCP issues: a couple other interesting points

- QUIC separates *data* sequence number from *packet* sequence number
  - Retransmission of same data gets a **new packet sequence number**
  - Lets lost retransmissions be identified without waiting for a timeout (can see gap in ACKed packet sequence numbers)
- From its 2017 SIGCOMM paper: “QUIC acknowledgments explicitly encode the delay between the receipt of a packet and its acknowledgment being sent.”
- Why is this useful? (think about BBR)

Summary

- TCP and UDP have worked really well as our foundational transport protocols on the internet, but aren't a perfect fit for all applications
- We can implement custom protocols to get better performance and better match application requirements
- Implementing new protocols at the transport layer (and getting them adopted) is tough due to long timescales, and complex web of assumptions and dependencies
- Innovation at the application layer is much easier! And has been quite successful in practice

Extra resources

- Academic papers:
  - "The QUIC Transport Protocol: Design and Internet-Scale Deployment": https://dl.acm.org/doi/abs/10.1145/3098822.3098842
- Standards docs:
  - https://datatracker.ietf.org/doc/html/draft-sharabayko-srt-01
  - https://datatracker.ietf.org/doc/html/rfc9000/
- Code is available to investigate:
  - https://github.com/Haivision/srt
  - https://github.com/google/quiche