Transport Protocols

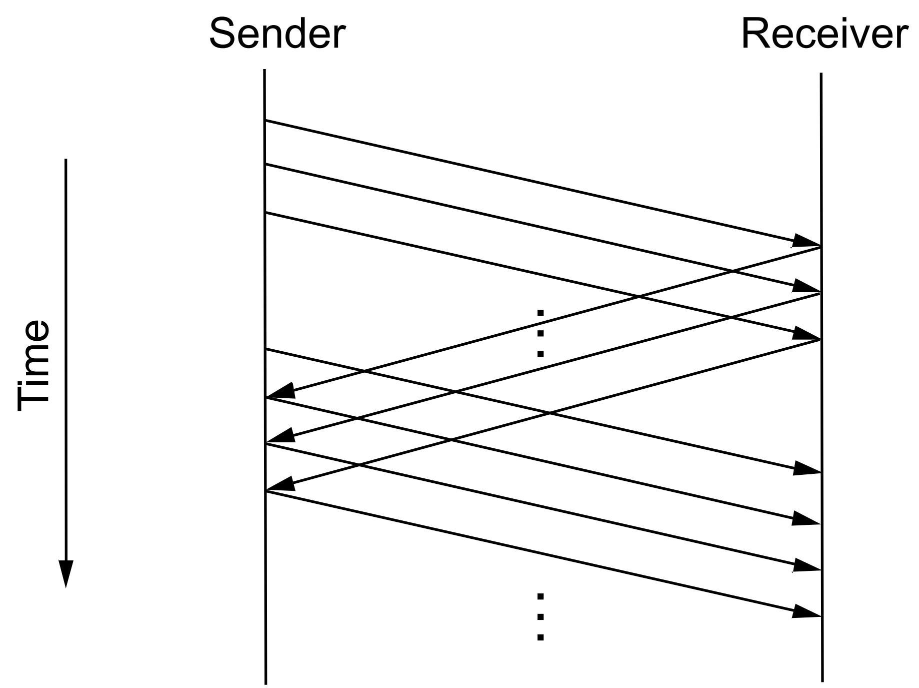
Reliability

Sliding Window Revisited

- TCP's variant of the sliding window algorithm, which serves several purposes:
  - (1) it guarantees the reliable delivery of data,
  - (2) it ensures that data is delivered in order, and
  - (3) it enforces flow control between the sender and the receiver.

Solution: Pipelining via Sliding Window

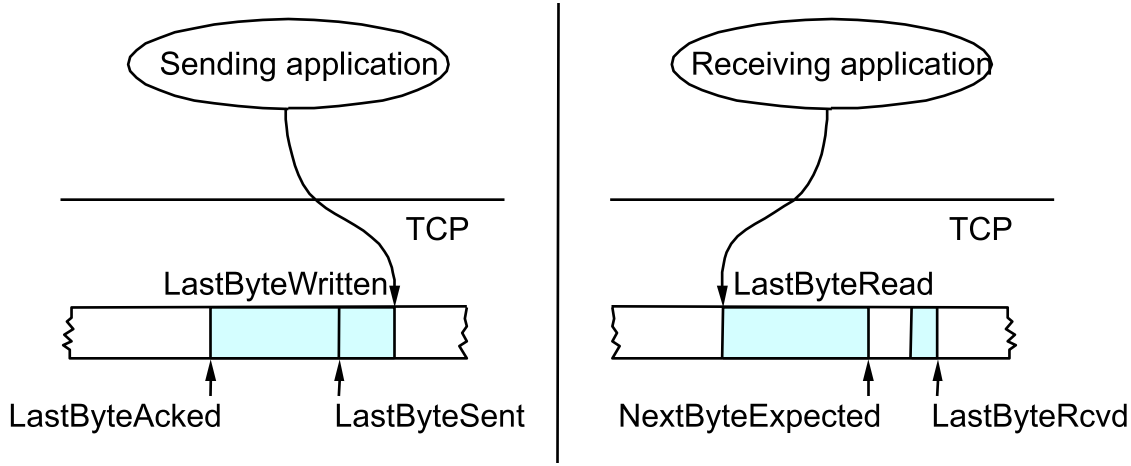
- Allow multiple outstanding (un-ACKed) frames
- Upper bound on un-ACKed frames, called *window*



Buffering on Sender and Receiver

- Sender needs to buffer data so that if data is lost, it can be resent
- Receiver needs to buffer data so that if data is received out of order, it can be held until all packets are received
  - Flow control
- How can we prevent sender overflowing receiver's buffer?
  - Receiver tells sender its buffer size during connection setup
- How can we insure reliability in pipelined transmissions?
  - Go-Back-N
    - Send all N unACKed packets when a loss is signaled
    - Inefficient
  - Selective repeat
    - Only send specifically unACKed packets
    - A bit trickier to implement

Sliding Window Revisited



Sending side

LastByteAcked <= LastByteSent

LastByteSent <= LastByteWritten

buffer bytes between **LastByteAcked** and **LastByteWritten**

- Receiving side
  - **LastByteRead < NextByteExpected**
  - **NextByteExpected <= LastByteRcvd +1**
  - buffer bytes between **NextByteRead** and **LastByteRcvd**

Flow Control in TCP

- Send buffer size: **`MaxSendBuffer`**
- Receive buffer size: **`MaxRcvBuffer`**
- Receiving side
  - **`LastByteRcvd`** - **`LastByteRead`** < = **`MaxRcvBuffer`**
  - **`AdvertisedWindow`** = **`MaxRcvBuffer`** - (**`(NextByteExpected - 1) - LastByteRead`**)
- Sending side
  - **`LastByteSent`** - **`LastByteAcked`** < = **`AdvertisedWindow`**
  - **`EffectiveWindow`** = **`AdvertisedWindow`** - (**`LastByteSent`** - **`LastByteAcked`**)
  - **`LastByteWritten`** - **`LastByteAcked`** < = **`MaxSendBuffer`**
  - block sender if (**`LastByteWritten`** - **`LastByteAcked`**) + y > **`MaxSenderBuffer`**
- Always send ACK in response to arriving data segment
- Persist sending one byte seg. when **`AdvertisedWindow = 0`**
  - Keep soliciting ACKs, eventually window opens up

Triggering Transmission

- How does TCP decide to transmit a segment?
  - TCP supports a byte stream abstraction
  - Application programs write bytes into streams
  - It is up to TCP to decide that it has enough bytes to send a segment
- TCP uses “self clocking”
  - Use ACKs as an implicit timer
- ACK info tells if there is enough space

Nagle's Algorithm

- We could use a clock-based timer, for example one that fires every 100 ms
- Nagle introduced an elegant self-clocking solution
- Key Idea
  - As long as TCP has any data in flight, the sender will eventually receive an ACK
  - This ACK can be treated like a timer firing, triggering the transmission of more data

Nagle's Algorithm

When the application produces data to send

if both the available data and the window ≥ MSS `// either at startup or when an ACK arrives`

send a full segment

else

if there is unACKed data in flight

buffer the new data until an ACK arrives

else

send all the new data now

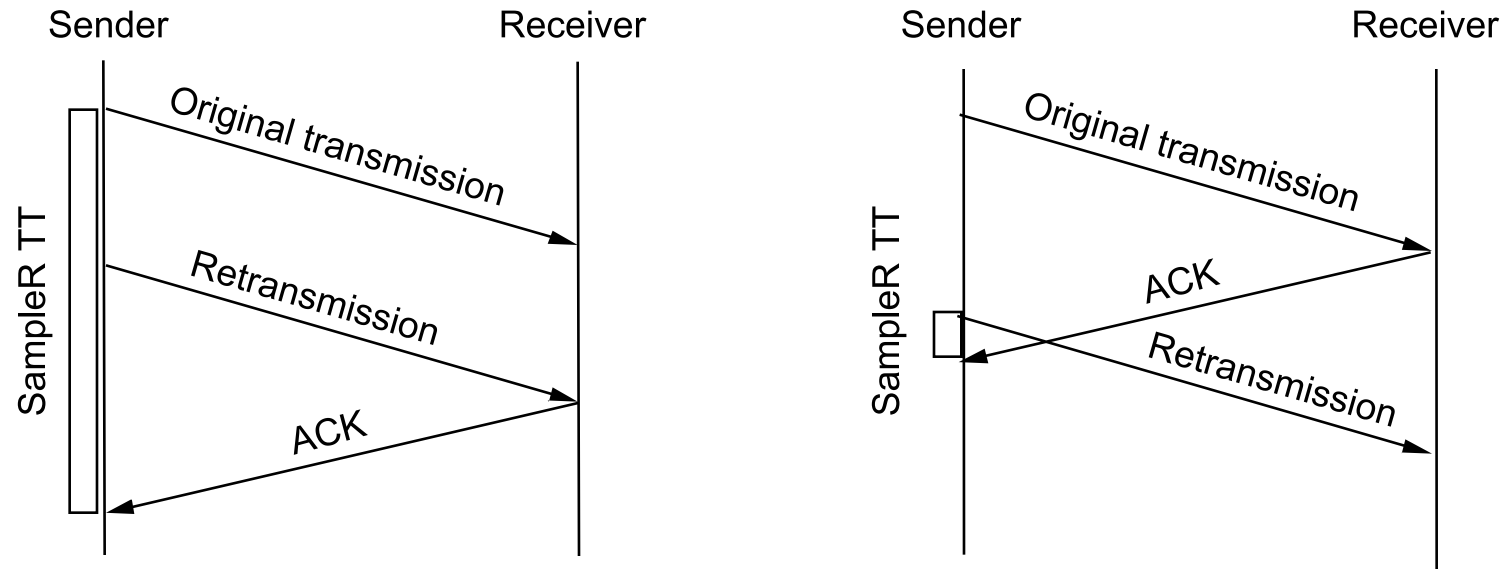
Adaptive Retransmission

- Original Algorithm
  - Measure **`SampleRTT`** for each segment/ ACK pair
  - Compute weighted average of RTT
    - **`EstRTT`** = α x **`EstRTT`** + (1 - α)x **`SampleRTT`**
    - α between 0.8 and 0.9
  - Set timeout based on **`EstRTT`**
    - **`TimeOut`** = **2** x **`EstRTT`**

Original Algorithm

- Problem
  - ACK does not really acknowledge a transmission
    - It actually acknowledges the receipt of data
  - When a segment is retransmitted and then an ACK arrives at the sender
    - It is impossible to decide if this ACK should be associated with the first or the second transmission for calculating RTTs

Karn/Partridge Algorithm for RTO



- Two degenerate cases with timeouts and RTT measurements
  - Solution: Do not sample RTT when retransmitting
- After each retransmission, set next RTO to be double the value of the last
  - Exponential backoff is well known control theory method
  - Loss is most likely caused by congestion so be careful

Karn/Partridge Algorithm

- Karn-Partridge algorithm was an improvement over the original approach, but it does not eliminate congestion

- We need to understand how timeout is related to congestion
  - If you timeout too soon, you may unnecessarily retransmit a segment which adds load to the network

Karn/Partridge Algorithm

- Main problem with the original computation is that it does not take variance of Sample RTTs into consideration.
- If the variance among Sample RTTs is small
  - Then the Estimated RTT can be better trusted
  - There is no need to multiply this by 2 to compute the timeout

Karn/Partridge Algorithm

- On the other hand, a large variance in the samples suggest that timeout value should not be tightly coupled to the Estimated RTT
- Jacobson/Karels proposed a new scheme for TCP retransmission

Jacobson/ Karels Algorithm

- In late ’80s, Internet was suffering from *congestion collapse*
- New Calculations for average RTT – Jacobson ’88
- Variance is not considered when setting timeout value
  - If variance is small, we could set RTO = EstRTT
  - If variance is large, we may need to set RTO > 2 x EstRTT
- New algorithm calculates both variance and mean for RTT
- **`Diff`** = **`sampleRTT`** - **`EstRTT`**
- **`EstRTT = EstRTT + ( d x Diff)`**
- **`Dev = Dev + d ( |Diff| - Dev)`**
  - Initially settings for **`EstRTT`** and **`Dev`** will be given to you
  - where d is a factor between 0 and 1
  - typical value is 0.125

Jacobson/ Karels contd.

- **`TimeOut`** = μ x **`EstRTT`** + ϕ x **`Dev`**
  - where $\mu = 1$ and $\phi = 4$
- When variance is small, TimeOut is close to EstRTT
- When variance is large Dev dominates the calculation
- Another benefit of this mechanism is that it is very efficient to implement in code (does not require floating point)
- Notes
  - algorithm only as good as granularity of clock (500ms on Unix)
  - accurate timeout mechanism important to congestion control (later)
- These issues have been studied and dealt with in new RFC's for RTO calculation.
- TCP RENO uses Jacobson/Karels