COMP 790-088
Networked and Distributed Systems

Reliability and Sliding Window

Jasleen Kaur

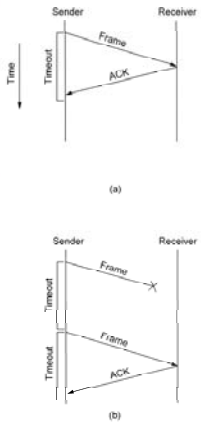
September 30, 2009

Recovering Lost Data
ARQ

- Segment loss may occur due to:
  - » non-recoverable bit errors
  - » buffer overflow
  - » multiple attempts at shared medium access
- Need mechanisms to recover lost frames
- Basic idea: use Automatic Repeat Request (ARQ)
  - » Acknowledgements (sent by receiver)
    - ❖ Indicate successful delivery
  - » Timeouts (maintained by sender)
    - ❖ Indicate when packet is likely to be lost

The Stop-and-Wait Protocol
ARQ Example

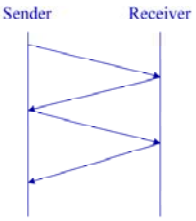
- Sender sends next packet *only* after it receives ACK for transmitted packet
  - » At most one unacknowledged packet in flight
  - » Uses sequence numbers to distinguish retransmissions from new packets
    - ❖ How many distinct sequence numbers needed?



COMP 790-088 3

Stop-and-Wait
Inefficiency

- Expected throughput in a stop-and-wait transfer?
  - » May prevent transfer from utilizing available capacity
  - » e.g., 1 KB frame size, 1.5 Mbps link, 45 ms RTT
    - ❖ ⟹ 1/8th link utilization



- Keeping the pipe full:
  - » How much data should be in transit to completely utilize the bottleneck bandwidth?
  - » 1.5Mbps link x 45ms RTT = 67.5Kb (8KB)

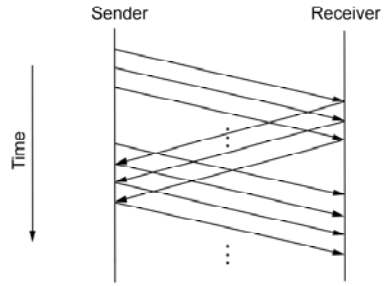
Should allow multiple un-acknowledged frames

COMP 790-088 4

Sliding Window

Efficient ARQ Example

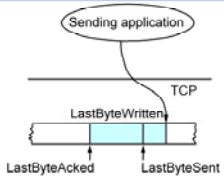
- Allows multiple unacknowledged packets
  - » Number of unacknowledged packets upper bounded by *window*

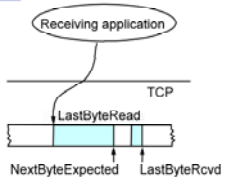


COMP 790-088 5

Sliding Window

Reliable, In-order, Byte-stream Delivery





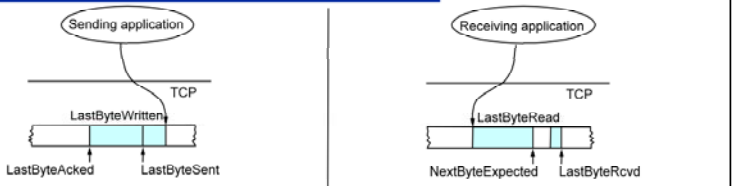
- Sender assigns seq no. to every byte
- Sender maintains:
  - » Last Byte Written (LBWritten)
  - » Last Byte Sent (LBSent)
  - » Last Byte Acked (LBAcked)
- Invariant:
  LBAcked ≤ LBSent ≤ LBWritten
- Sender buffers only bytes between: (LBAcked, LBWritten]
  - » Retransmits packets that time-out waiting for ACKs

- Receiver maintains:
  - » Last Byte Read (LBRead)
  - » Next Byte Expected (NBExp)
  - » Last Byte Received (LBRcvd)
- Receiver (cumulative) ACKs send NBExp
- Invariant:
  LBRead ≤ NBExp ≤ LBRcvd + 1
- Receiver buffers only bytes between: [LBRead, LBRcvd]

COMP 790-088 6

Sliding Window

Flow Control



- New invariants:
  - LBSent – LBAcked ≤ AdvWin
  - EffectiveWin = AdvWin – (LBSent – LBAcked)
- Sender buffer limits (MaxSendBuf) add invariant:
  - LBWritten – LBAcked ≤ MaxSendBuf
- Receiver buffer limits (MaxRcvBuf) add new invariant
  - LBRead – LBRcvd ≤ MaxRcvBuf
- Receiver advertises to sender:
  - AdvWin = MaxRcvBuf – [(NBExp – 1) – LBRead]

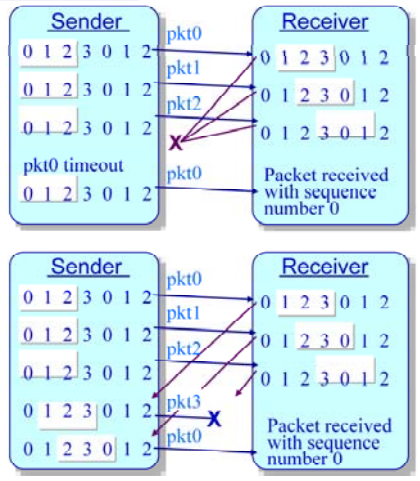
Is this guaranteed to prevent buffer overflow at either ends?

Is it possible that an ACK arrival will not allow further transmissions?

Sliding Window

Sequence Number Space Required

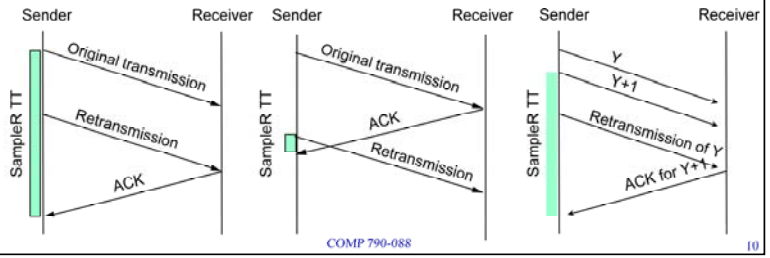
- How large should the sequence number space be?
  - » As large as the maximum window size?
  - » MaxSeqNum > 2*Window



Adaptive Retransmissions

Setting the Retransmission Timer (RTO)

- Central question: How to set retransmission timeout?
  - » Internet paths can vary significantly in their propagation length
  - » End-to-end RTTs can vary significantly over time
    - ❖ RTO should be an adaptive function of current RTT conditions
- Associated question: How to sample path RTTs?
  - » Given retransmitted packets? And given delayed ACKs?
    - ❖ Workable solution: stop sampling during retransmission phases



Adaptive RTO

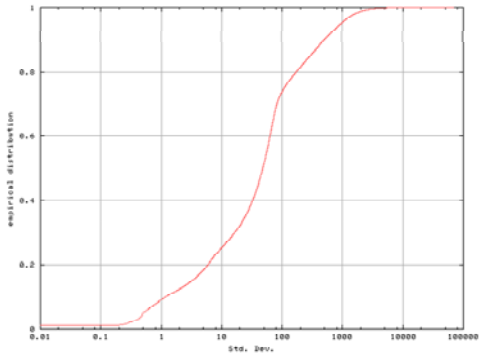
RTO Computation as a Function of RTT

- Original algo:
  - » Maintain running average of RTT samples
    - ❖ Exponentially weighted average
      - ◆ EstimatedRTT = a*EstimatedRTT + (1-a)*SampledRTT
    - ❖ How does a impact computation?
  - » Use RTT estimate to compute timeout:
    - ❖ RTO = 2*EstimatedRTT
    - ❖ Just a conservative value
- Problem: does not take RTT variance into account

Do TCP RTTs vary much?

RTT Variability In Practice

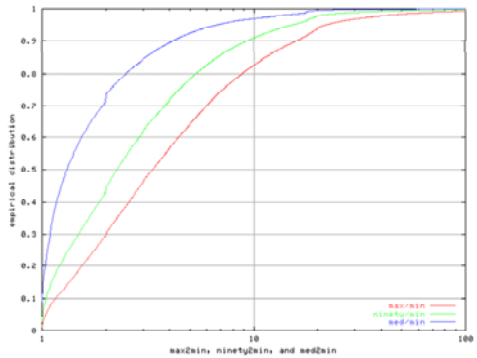
UNC Study of TCP Transfers



- UNC study measured per-segment RTTs for 1 million TCP transfers
  - » 70% transfers see a standard deviation of 10 ms or more

RTT Variability In Practice

UNC Study of TCP Transfers



- Median RTTs can be more than twice of min RTTs for 30% of transfers

Jacobson/Karel's Algo
Incorporating RTT Variability

- Use variance in RTTs to estimate timeout
  - » If variance is small, why set RTO to twice the value of EstimatedRTT?
- Algo:
  - » Calculate running averages of both RTT and its variation
    - Diff = SampledRTT – EstimatedRTT
    - EstimatedRTT = EstimatedRTT + (1-*a*)*Diff
    - Deviation = Deviation + (1-*b*)*(|Diff| - Deviation)
  - » RTO = *u**EstimatedRTT + *p**Deviation
    - Typically, $u = 1$, $p = 4$, $a = b = 0.875$
    - Large variance causes Deviation to dominate calculation of RTO

RTO Timers In Practice
Impact of Implementations

- Most Unix TCP implementations check to see if a timeout should occur, only once every clock tick
  - Ultimately, algorithm only as good as granularity of system clocks
    - 10-100 ms for current OSes
- Most implementations take only one RTT sample per RTT
- Most implementations use a minRTO value of 200 ms – 1000 ms

Timeouts may happen no earlier than 1 second after segment was transmitted !

- TCP extensions:
  - » Instead of relying on coarse-grained timers for measuring RTTs, use per-packet timestamps to measure RTT