TCP Protocol Details, Part 2

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Receiver Window

- The receiver window manages flow control.
  - Receiver adjusts size to reflect buffer space.
  - Sends window size updates via “Window” field in TCP segment header.
- Sender won't send more data than receiver can handle.
  - ... even in the case where receiving application is busy elsewhere.

Sources of Slowness

- Receiver
  - Slow computer
  - Distracted program
    - Dealing with other tasks...
    - Processing received data is complicated...
  - *Receiver buffer fills and receiver window shrinks.*
- Network
  - Slow links
  - High traffic
  - *How is this handled?*

Congestion Window

- TCP maintains a second window.
  - *Estimate of the network's capacity to transmit data*.
  - Sender must compute the size of this window
    - Based on implicit feedback from the receiver
      - Successful ACKs
      - Timeouts
    - *Assumption: Lost segments are due to network congestion* (is this really true?)
- Actual window used for transmission is the smallest of (receiver, congestion).

Slow Start

- Congestion window size (*cwnd*) starts small and grows to “probe” the network capacity.
  - In what follows “one segment” means the MSS used by the sender (typically 1460 bytes on ethernet).
- Initialize with *cwnd* = 1 segment.
- Increment *cwnd* by 1 segment for each segment acknowledged.
  - This increases *cwnd* exponentially!

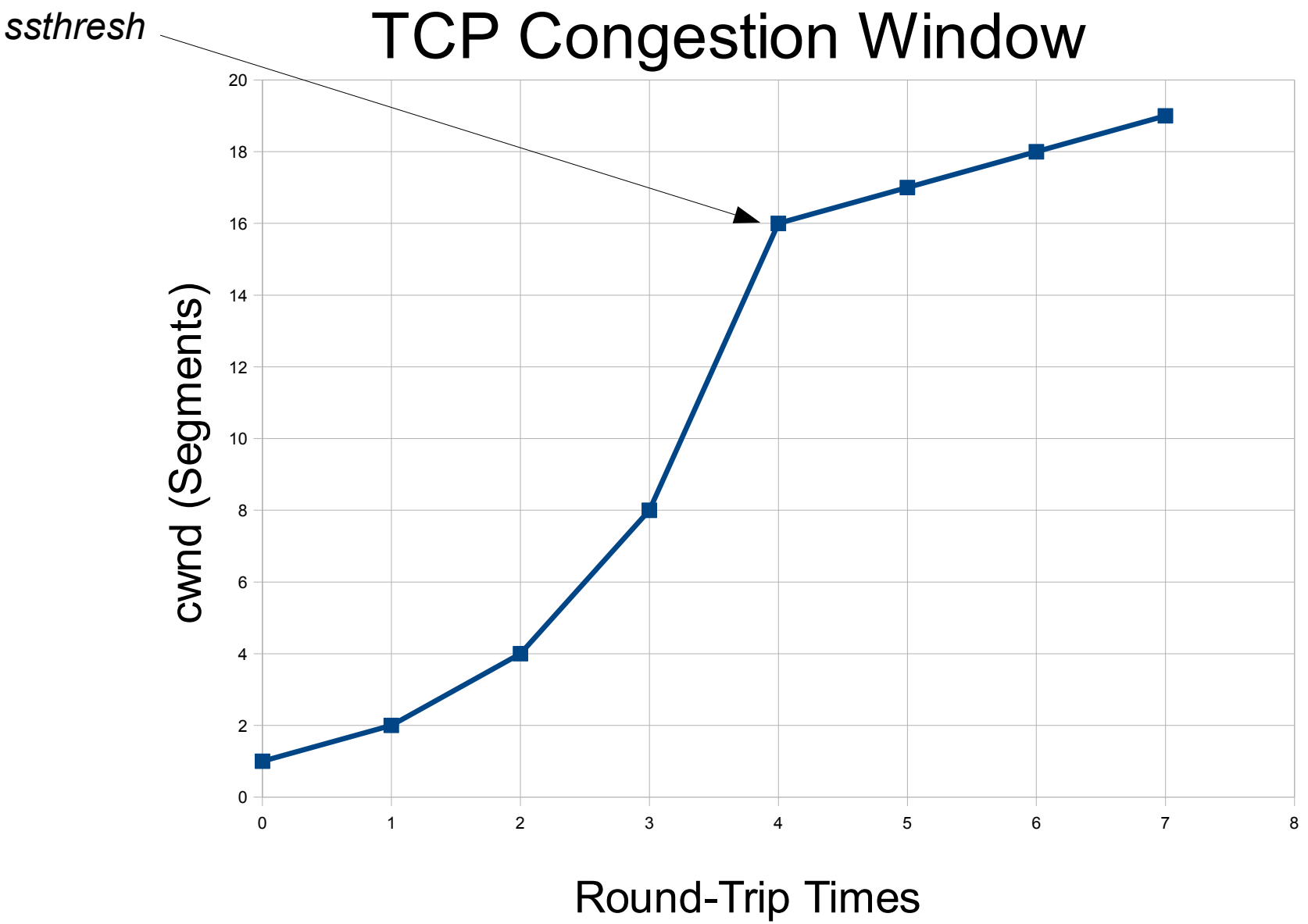
Exponentially?

- Consider...
  - Set *cwnd* = 1 segment. Send it.
  - Wait for ACK. Set *cwnd* = 2 segments. Send them.
  - After both ACKs...
    - Set *cwnd* = 2 + 1 + 1 = 4 segments. Send them.
  - After all four ACKS...
    - Set *cwnd* = 4 + 1 + 1 + 1 + 1 = 8 segments. Send them.
- In real life it is more complicated.
  - ACKs don't really arrive all together (in general).
  - TCP follows the same basic rule, however.

Slow Start Threshold

- A second value, *ssthresh*, defines when slow start ends and “congestion avoidance” begins.
  - After *cwnd* reaches *ssthresh*...
    - Increment *cwnd* by *1/cwnd* (as measured in segments) for each ACK.
      - Example: If *cwnd* = 4 segments, then add ¼ segment to *cwnd* in response to the next ACK.
      - Thus 4 ACKs needed to increase *cwnd* by 1 segment.
    - Thus *cwnd* increases by 1 for each round trip time regardless of segment count.
  - Causes a linear increase of *cwnd*.

Summary


ssthresh
TCP Congestion Window
cwnd (Segments)
Round-Trip Times
0
2
4
6
8
10
12
14
16
18
20
0
1
2
3
4
5
6
7
8

Timeout!

- When a timeout occurs...
  - *ssthresh* is set to ½ the current *cwnd* value.
  - *cwnd* is set to 2.
  - Slow start begins again.
- TCP assumes timeout means data loss.
  - Backs off by reducing the congestion window size.
  - Begins probing the network again in case source of congestion is gone.

Remember...

- TCP uses the smallest of (receiver, congestion) windows.
  - Once *cwnd* exceeds the receiver window, flow is limited by receiver window size.
  - This is the normal case on a clear network.
- On a WAN, however, *cwnd* is often limiting.
- Many details left out of this dicussion.
  - See references slide at end of this slide group.

How Long to Timeout?

- Too long...
  - If TCP waits too long to retransmit a lost segment time is wasted.
  - Slows down transmission.
- Too short...
  - If TCP doesn't wait long enough, it may retransmit unnecessarily.
  - Clogs the network.
  - Wastes bandwidth.

Round Trip Time?

- How long is a normal round trip?
  - LAN...
    - Transit time is sub-millisecond.
    - Usually steady.
  - WAN...
    - Transit time is multiple millisecond.
    - Often tens, hundreds, even thousands of milliseconds.
    - Often highly variable.
  - Computation time is usually short.
    - TCP acknowledges, application not involved.

RTT Estimation (Old)

- RFC-793 contains an algorithm for estimating round trip time (`RTT`).
  - Associate a timer with each outgoing segment.
  - When an ACK comes in, note the measured RTT for that segment (`M`).
  - Compute: $R_{new} = \alpha R_{old} + (1 - \alpha) M$
    - Where *α* is a scale factor (typically 0.9). *R* is an estimate of the RTT.
  - Compute timeout: $T = R_{new} \beta$
    - Where *β* is another scale factor (typically 2).

Problems

- The previous algorithm is not that great.
  - Can't keep up with changes.
  - Doesn't deal with highly variable RTT values.
  - Tends to cause many unnecessary retransmissions.
- What is needed is a way to account for the degree of variability in the RTT.

Jacobson's Algorithm

- Compute both RTT and “deviation” estimates.
  - Compute $E_r = M - R_{old}$
    - Note that the error value is signed.
  - Compute $R_{new} = R_{old} + g\,E_r$
    - Here *g* is typically 1/8.
  - Compute $D_{new} = D_{old} + h\left(\left|E_r\right| - D_{old}\right)$
    - Here *h* is typically ¼. *D* is an estimate of the deviation in observed RTT values.
  - Compute $T_{new} = R_{new} + 4\,D_{new}$
    - Time is RTT with extra to account for variability of RTT.
- Note that computations above are easy.

TCP Performance

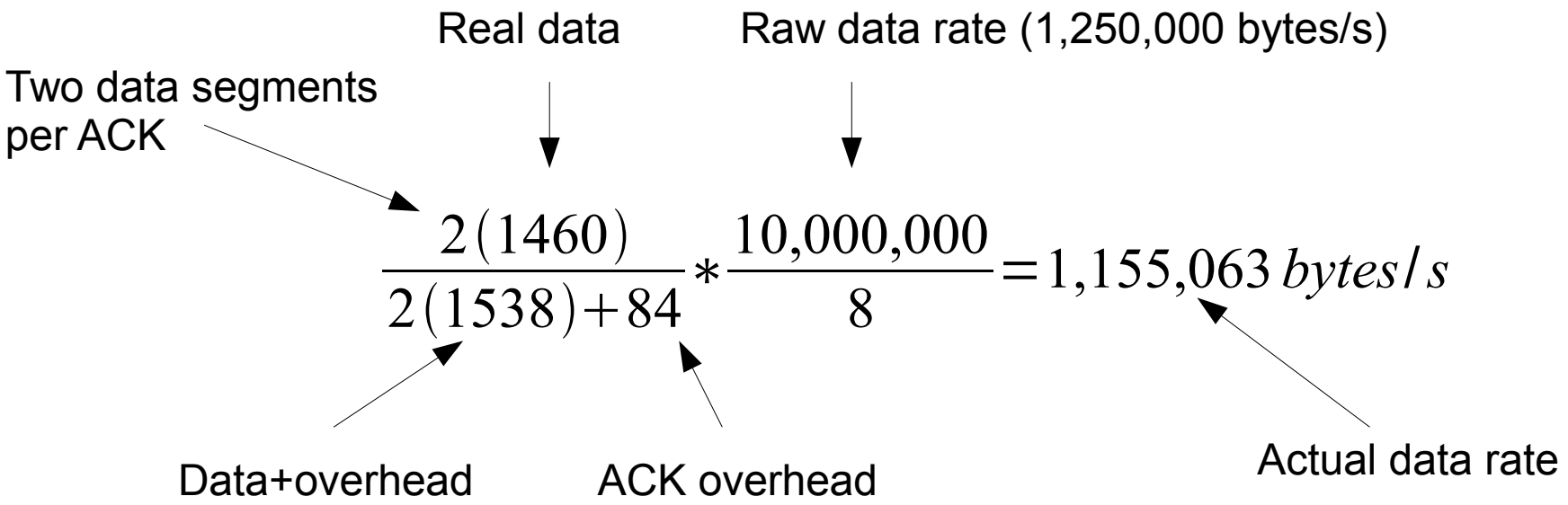
	DATA Bytes	ACK Bytes
Preamble	8	8
Ethernet Header	14	14
IP Header	20	20
TCP Header	20	20
Data	1460	0
Pad	0	6
CRC	4	4
Interpacket Gap	12	12
TOTAL	**1538**	**84**

Needed to meet ethernet minimum of 64 bytes per frame

9.6 microseconds on 10 Mbps ethernet.

Performance Computation

- Assume one ACK for every two data segments
  - In real life there are many possibilities.
- Assume 10 Mbps ethernet.



References

- RFC-793: Transmission Control Protocol.
- RFC-896: Congestion Control in IP/TCP Internetworks. (Describes Nagle's Algorithm for interactive connections).
- RFC-2581: TCP Congestion Control.
- http://en.wikipedia.org/wiki/Transmission_Control_Protocol
- http://www.winlab.rutgers.edu/~hongbol/tcpWeb/tcpTutorialNotes.html