ELEN 4017

Network Fundamentals Lecture 18



Purpose of lecture



Chapter 3: Transport Layer

- Transmission Control Protocol Basics
- Timeout in TCP

Basics of TCP

- Reliable data transfer
- Connection-oriented (handshake)
- Not an end-to-end TDM/FDM connection.
- Not a virtual circuit (resource and path not reserved)
- Full duplex (bi-directional)
- Point-to-point i.e. communications between a single sender and receiver.

Uni and bi-directional communication









Examples ?



Point-to-point vs multipoint



Examples ?



Virtual circuit vs Datagram





Buffers

- TCP employs buffers to store data during transmission.
- Buffers are assigned following a successful handshake.



Figure 3.28

TCP send and receive buffers

Maximum segment size (MSS)



- MSS defines the maximum amount of data that can be grabbed and placed in a segment.
- This is usually dimensioned using the length of largest link-layer frame that can be sent by the local host - called Maximum Transmission Unit (MTU).
- There have been proposals to set the MSS based on the path MTU value. Why ?

MSS



- Determine MTU (i.e. link layer frame)
- Fit a TCP segment into this MTU.
- What is the size of the application data → this determines MSS



TCP Segment

- Source and destination ports
- 32 bit sequence and ACK no.
- Receive window (flow control)
- Header length
- Flag field:
 - RST, SYN, FIN are used for connection setup/teardown.





TCP Reliable transfer



- TCP uses a hybrid of GBN and SR.
- It also piggy-backs messages i.e. the acknowledgment for client-to-server data, is put into the same segment as server-toclient data.
- The sequence numbers are assigned on **byte position**, not packet number (more later).

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Timeout handing



- How do we estimate an optimised timeout value ?
- We don't want a timeout that it is too long since it will delay the pipelining once the window is full.
- We don't want a timeout that is too short, since it means unnecessary retransmissions.

Estimating the RTT



- RTT is the time between sending a segment to IP and receiving the ACK.
- TCP does not measure this for every segment, usually just 1 segment at a time, thus we usually have 1 sample every RTT.
- TCP also does not compute the RTT for a retransmitted segment.
- These values are averaged to smooth out fluctuations.

Exponential Weighted Moving Average (EWMA)



EstimatedRTT = $(1 - \alpha)$.EstimatedRTT + α . SampleRTT

- $\alpha \rightarrow$ recommended = 0.125
- Weighted average, more weight assigned to recent samples.
- Weight of a given sample decays exponentially fast as updates proceed.

RTT sample and RTT estimates



Figure 3.32 • RTT samples and RTT estimates

Variability of RTT (deviation)



$DevRTT = (1 - \beta)$. $DevRTT + \beta$.| SampleRTT – EstimatedRTT|

- β recommended = 0.25
- Weighted moving average of deviation from mean.
- In case of low fluctuation, DevRTT is small.
- TCP thus uses the following timeout interval

TimeoutInterval =EstimatedRTT + 4.DevRTT

Special features of TCP – Doubling timeout interval



- In case of retransmits, TCP sets the timeout interval = twice the currently used interval.
- Thus the interval will grow exponentially for each retransmit.
- This provides a limited form of congestion control.

Fast retransmit



- A problem with **timeout triggered retransmissions** is that the timeout period can be relatively long.
- The sender can often detect packet loss well before the time-out event occurs, by noting so-called duplicate ACKS.
- A **duplicate ACK** is an ACK that re-acknowledges a segment for which the sender already received an earlier ACK.
- If sender receives 3 duplicate ACKS back-to-back, it assumes that the next packet in sequence was lost, and re-transmits.