Transport Layer IV

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Chapter 3: Roadmap

3.1 Transport-layer services
3.2 Multiplexing and demultiplexing
3.3 Connectionless transport: UDP
3.4 Principles of reliable data transfer
3.5 Connection-oriented transport: TCP
   - Segment structure
   - Reliable data transfer
   - Flow control
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TCP: Overview [RFCs: 793, 1122, 1323, 2001, 2018, 2581, 3390, 5681]

- Point-to-point (unicast):
  - One sender, one receiver
- Reliable, in-order *byte stream*:
  - Packet boundaries are not visible to the application
- Pipelined:
  - TCP congestion and flow control set window size
- Send & receive buffers

- Full duplex data:
  - Bi-directional data flow in same connection
- MSS: maximum segment size (excluding headers)
- Connection-oriented:
  - Handshaking (exchange of control msgs) initializes sender/receiver state before sending data
- Flow controlled:
  - Sender will not overwhelm receiver
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TCP Segment Structure

- Sequence/ACK numbers
  - Count bytes, not segments
  - ACKs piggybacked on data packets
- Flags (U-A-P-R-S-F)
  - Urgent data (not used)
  - ACK field is valid
  - PUSH (reduce latency)
  - RST (reset connection)
  - SYN (connection request)
  - FIN (connection close)
- Hdr length in DWORDs (4-bit field)
  - Normally 20 bytes, but longer if options are present
**TCP Seq. #'S and ACKs**

**Seq. #’s:**
- Sequence number of the first byte in segment’s data

**ACKs:**
- Seq # of next byte expected from sender
- Cumulative ACK

Q: how receiver handles out-of-order segments?
A: TCP spec doesn’t say, up to implementor

**Simple telnet scenario**

- **Host A**
  - User types ‘C’
  - Seq=42, ACK=79, data = ‘C’
  - Seq=79, ACK=43, data = ‘C’
  - Seq=43, ACK=80

- **Host B**
  - host ACKs receipt of ‘C’, echoes back ‘C’

- host ACKs receipt of echoed ‘C’
**TCP Round Trip Time and Timeout**

**Q:** how to set TCP timeout value (RTO)?

- Want it slightly larger than the next RTT
  - But the RTT varies
- **Too short:** premature timeout
  - Unnecessary retransmissions
- **Too long:** slow reaction to segment loss
  - Protocol may stall, exhibit low performance

**Idea:** dynamically measure RTT, average these samples, then add safety margin

**SampleRTT:** measured time from segment transmission until ACK receipt

- Ignore retransmissions, why?

**SampleRTT will vary,** want estimated RTT “smoother”

- Average several recent measurements, not just current SampleRTT
TCP Round Trip Time and Timeout

EstimatedRTT(n) = (1-\(\alpha\))EstimatedRTT(n-1) + \(\alpha\)SampleRTT(n)

- **Exponentially weighted moving average (EWMA)**
  - Influence of past sample decreases exponentially fast
  - Typical value: \(\alpha = 1/8\)

- **Task: derive a non-recursive formula for EstimatedRTT(n)**
  - **Assume** EstimatedRTT(0) = SampleRTT(0)
  - **Let** \(Y(n) = EstimatedRTT(n)\) and \(y(n) = SampleRTT(n)\)

\[
Y(n) = (1 - \alpha)^n y(0) + \alpha \sum_{i=0}^{n-1} (1 - \alpha)^i y(n - i)
\]
Example RTT Estimation:

![Graph showing RTT estimation examples](image)

- Sampled RTT (black diamonds)
- Estimated RTT (pink squares)

- X-axis: Sample number
- Y-axis: RTT (ms)
TCP Round Trip Time and Timeout

- **Setting the timeout:**
- EstimatedRTT plus a “safety margin”
  - Larger variation in EstimatedRTT → larger safety margin
- First estimate how much SampleRTT deviates from EstimatedRTT (typically, $\beta = 1/4$):

\[
\text{DevRTT}(n) = (1-\beta) \times \text{DevRTT}(n-1) + \beta \times |\text{SampleRTT}(n) - \text{EstimatedRTT}(n)|
\]

Then set retransmission timeout (RTO):

\[
\text{RTO}(n) = \text{EstimatedRTT}(n) + 4 \times \text{DevRTT}(n)
\]
Example Timeout Estimation:

![Graph showing example timeout estimation]

- **sampled RTT**: Blue diamonds
- **estimated RTT**: Pink squares
- **timeout**: Green triangles

The graph illustrates the sample number on the x-axis and RTT (ms) on the y-axis, with trends indicating the sampled RTT, estimated RTT, and timeout values over time.
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TCP Reliable Data Transfer

- TCP creates rdt service on top of IP’s unreliable service
  - Hybrid of Go-back-N and Selective Repeat
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
  - For the oldest unACK’ed packet
  - Retx only the base

- Retransmissions are triggered by:
  - Timeout events
  - Duplicate acks
- Initially consider simplified TCP sender:
  - Ignore duplicate acks
  - Ignore flow control, congestion control
NextSeqNum = InitialSeqNum // random for each transfer
SendBase = InitialSeqNum
loop (forever) {
    switch(event) {
        (a) data received from application above (assuming it fits into window):
            create TCP segment with sequence number NextSeqNum
            if (timer currently not running)
                start timer
            pass segment to IP
            NextSeqNum = NextSeqNum + length(data)
        (b) timeout:
            retransmit pending segment with smallest sequence number (i.e., SendBase); restart timer
        (c) ACK received, with ACK field value of y
            if (y > SendBase) {
                SendBase = y
                if (there are currently not-yet-acknowledged segments)
                    restart timer with latest RTO
                else cancel timer }
    }
} /* end of loop forever */
TCP Seq. #'s and ACKs

FTP Example:
• Suppose MSS = 1,000 bytes and the sender has a large file to transmit (we ignore seq field in ACKs and ACK field in data pkts)

What is the sender window size?

Host A
seq = 0
seq = 1000
seq = 2000
seq = 3000
seq = 4000
seq = 2000
RTO

Host B
ACK = 1000
ACK = 2000
ACK = 2000
ACK = 2000
ACK = 5000
TCP ACK Generation [RFC 1122, RFC 2581]

- Receiver immediately ACKs the base of its window in all cases except Nagle’s algorithm:
  - For *in-order* arrival of packets, send ACKs for every *pair* of segments; if second segment of a pair not received in 500ms, ACK the first one alone

```
seq = 0
seq = 1000
seq = 2000
seq = 3000
seq = 4000
seq = 2000
seq = 1000
seq = 2000

ACK = 2000
ACK = 2000
ACK = 2000
ACK = 2000
ACK = 5000
ACK = 1000
```
Fast Retransmit

- Time-out period often relatively long
  - Especially in the beginning of transfer (3 seconds in RFC 1122)
- **Idea**: infer loss via duplicate ACKs
  - Sender often sends many segments back-to-back
  - If a segment is lost, there will be many duplicate ACKs
- If sender receives 3 duplicate ACKs for its base, it assumes this packet was lost
  - **Fast Retransmit**: resend the base segment immediately (i.e., without waiting for RTO)
- Note that reordering may trigger unnecessary retransmission
  - To combat this problem, modern routers avoid load-balancing packets of same flow along multiple paths
Fast Retransmit Algorithm:

(c) event: ACK received, with ACK field value of y
   if (y > SendBase) {
      SendBase = y; dupACK = 0;
      if (SendBase != NextSeqNum)
         restart timer with latest RTO;
      else
         cancel timer; // last pkt in window
   }
   else if (y == SendBase) {
      dupACK++;
      if (dupACK == 3)
         { resend segment with sequence y; restart timer}
   }

a duplicate ACK for already ACKed segment

fast retransmit
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TCP Flow Control

- Assume packets received without loss, but the application does not call `recv()`
  - How to prevent sender from overflowing TCP buffer?
- Speed-matching service: sender rate to suit the receiving app’s ability to process incoming data

Flow control
Sender won’t overflow receiver buffer by transmitting too much, too fast
**TCP Flow Control: How It Works**

- **Spare room in buffer**
  
  \[ RcvWin = RcvBuffer - \left( LastByteReceivedInOrder - LastByteDelivered \right) \]

- **Receiver advertises spare room by including value of RcvWin in segments**

- **Sender enforces seq < ACK + RcvWin**
  - Guarantees receiver buffer doesn’t overflow

- **Combining both constraints (sender, receiver):**
  
  \[ seq < \min(sndBase+sndWin, ACK+RcvWin) \]
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TCP Connection Management

- Purpose of connection establishment:
  - Exchange initial seq #s
  - Exchange flow control info (i.e., $RcvWin$)
  - Negotiate options (SACK, large windows, etc.)

Three way handshake:

- **Step 1:** client sends TCP SYN to server
  - Specifies initial seq # $X$ and buffer size $RcvWin$
  - No data, ACK bit = 0

- **Step 2:** server gets SYN, replies with SYN+ACK
  - Sends server initial seq # $Y$ and buffer size $RcvWin$
  - No data, ACK val = $X+1$

- **Step 3:** client receives SYN+ACK, replies with ACK segment
  - Seq = $X+1$, ACK val = $Y+1$
  - May contain regular data, but many servers will break

- **Step 4:** regular packets
  - Seq = $X+1$, ACK = $Y+1$
Closing a connection:

- Closing a socket:
  \[ \text{closesocket} \text{(sock)}; \]

**Step 1:** originator end
system sends TCP FIN control segment to server

**Step 2:** other side
receives FIN, replies with ACK. Connection in “closing” state, sends FIN

TCP initiates a close when it has all ACKs for the transmitted data
**TCP Connection Management (Cont.)**

**Step 3:** originator receives FIN, replies with ACK
- Enters “timed wait” - will respond with ACK to received FINs

**Step 4:** other side receives ACK; its connection considered closed

**Step 5:** after a timeout (TIME_WAIT state lasts 240 seconds), originator’s connection is closed as well

Birectional transfer means both sides must agree to close