Chapter 3 outline

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
   - connection management
3.6 principles of congestion control
3.7 TCP congestion control
Pipelined protocols: overview

**Go-back-N:**
- sender can have up to $N$ unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn’t ack packet if there’s a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

**Selective Repeat:**
- sender can have up to $N$ unack’ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet
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TCP: Overview  RFCs: 793, 1122, 1323, 2018, 2581

- **point-to-point:**
  - one sender, one receiver

- **reliable, in-order byte steam:**
  - no “message boundaries”

- **pipelined:**
  - TCP congestion and flow control set window size

- **full duplex data:**
  - bi-directional data flow in same connection
  - MSS: maximum segment size

- **connection-oriented:**
  - handshaking (exchange of control msgs) initiates sender, receiver state before data exchange

- **flow controlled:**
  - sender will not overwhelm receiver
TCP segment structure

- **Source port #**
- **Destination port #**
- **Sequence number**
- **Acknowledgement number**
- **Receive window**
- **Checksum**
- **Urg data pointer**
- **Options (variable length)**
- **Application data (variable length)**

**Fields and Their Meanings**

- **URG**: Urgent data (generally not used)
- **ACK**: ACK # valid
- **PSH**: Push data now (generally not used)
- **RST, SYN, FIN**: Connection establishment (setup, teardown commands)
- **Internet checksum**: As in UDP

**Additional Notes**

- Counting by bytes of data (not segments!)
- # bytes rcvr willing to accept
TCP seq. numbers, ACKs

**Sequence numbers:**
- Byte stream “number” of first byte in segment’s data

**Acknowledgements:**
- Seq # of next byte expected from other side
- Cumulative ACK

---

**Outgoing segment from sender**

<table>
<thead>
<tr>
<th>Source port #</th>
<th>Dest port #</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequence number</td>
<td></td>
</tr>
<tr>
<td>Acknowledgement number</td>
<td></td>
</tr>
<tr>
<td>Rwnd</td>
<td></td>
</tr>
<tr>
<td>Checksum</td>
<td></td>
</tr>
<tr>
<td>Urg pointer</td>
<td></td>
</tr>
</tbody>
</table>

**Sender sequence number space**

- Window size $N$

**Incoming segment to sender**

<table>
<thead>
<tr>
<th>Source port #</th>
<th>Dest port #</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequence number</td>
<td></td>
</tr>
<tr>
<td>Acknowledgement number</td>
<td></td>
</tr>
<tr>
<td>A</td>
<td></td>
</tr>
<tr>
<td>Rwnd</td>
<td></td>
</tr>
<tr>
<td>Checksum</td>
<td></td>
</tr>
<tr>
<td>Urg pointer</td>
<td></td>
</tr>
</tbody>
</table>
TCP seq. numbers, ACKs

User types ‘C’

Seq=42, ACK=79, data = ‘C’
Seq=79, ACK=43, data = ‘C’
Seq=43, ACK=80

Host A

host ACKs receipt of echoed ‘C’

Host B

host ACKs receipt of ‘C’, echoes back ‘C’

simple telnet scenario
TCP round trip time, timeout

Q: how to set TCP timeout value?
- longer than RTT
  - but RTT varies
- too short: premature timeout, unnecessary retransmissions
- too long: slow reaction to segment loss

Q: how to estimate RTT?
- SampleRTT: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- SampleRTT will vary, want estimated RTT “smoother”
  - average several recent measurements, not just current SampleRTT
TCP round trip time, timeout

Estimated\text{RTT} = (1- \alpha)\text{EstimatedRTT} + \alpha\text{SampleRTT}

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$
TCP round trip time, timeout

- **timeout interval**: EstimatedRTT plus “safety margin”
  - large variation in EstimatedRTT -> larger safety margin

- estimate SampleRTT deviation from EstimatedRTT:
  \[
  \text{DevRTT} = (1-\beta) \times \text{DevRTT} + \beta \times |\text{SampleRTT} - \text{EstimatedRTT}|
  \]
  
  (typically, \( \beta = 0.25 \))

\[
\text{TimeoutInterval} = \text{EstimatedRTT} + 4 \times \text{DevRTT}
\]
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TCP reliable data transfer

TCP creates rdt service on top of IP’s unreliable service:
- pipelined segments
- cumulative acks
- single retransmission timer

Retransmissions triggered by:
- timeout events
- duplicate acks

Let’s initially consider simplified TCP sender:
- ignore duplicate acks
- ignore flow control, congestion control
TCP sender events:

**data rcvd from app:**
- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unacked segment
  - expiration interval: TimeOutInterval

**timeout:**
- retransmit segment that caused timeout
- restart timer

**ack rcvd:**
- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments
TCP sender (simplified)

wait for event

\[ \text{NextSeqNum} = \text{InitialSeqNum} \]
\[ \text{SendBase} = \text{InitialSeqNum} \]

- data received from application above
- create segment, seq. #: NextSeqNum
- pass segment to IP (i.e., “send”)
- NextSeqNum = NextSeqNum + length(data)
- if (timer currently not running)
  - start timer
- timeout
  - retransmit not-yet-acked segment with smallest seq. #
  - start timer

ACK received, with ACK field value \( y \)

- if \( y > \text{SendBase} \) {
  - \( \text{SendBase} = y \)
  - /* \text{SendBase}–1: last cumulatively ACKed byte */
  - if (there are currently not-yet-acked segments)
    - start timer
  - else stop timer
- }
TCP: retransmission scenarios

lost ACK scenario

Host A
Seq=92, 8 bytes of data
ACK=100

Host B
timeout

premature timeout

Host A
SendBase=92
Seq=92, 8 bytes of data
ACK=100
SendBase=100
SendBase=120

Host B
Seq=100, 20 bytes of data
ACK=100
ACK=120
Seq=92, 8 bytes of data
ACK=120
SendBase=120
TCP: retransmission scenarios

Cumulative ACK

Host A

Seq=92, 8 bytes of data

Seq=100, 20 bytes of data

Timeout

ACK=100

X

ACK=120

Host B

Seq=120, 15 bytes of data

Seq=100, 20 bytes of data

ACK=120
## TCP ACK generation

**event at receiver**

<table>
<thead>
<tr>
<th>Event Description</th>
<th>TCP receiver action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed</td>
<td>Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK</td>
</tr>
<tr>
<td>Arrival of in-order segment with expected seq #. One other segment has ACK pending</td>
<td>Immediately send single cumulative ACK, ACKing both in-order segments</td>
</tr>
<tr>
<td>Arrival of out-of-order segment higher-than-expect seq. #. Gap detected</td>
<td>Immediately send <em>duplicate ACK</em>, indicating seq. # of next expected byte</td>
</tr>
<tr>
<td>Arrival of segment that partially or completely fills gap</td>
<td>Immediate send ACK, provided that segment starts at lower end of gap</td>
</tr>
</tbody>
</table>
TCP fast retransmit

- time-out period often relatively long:
  - long delay before resending lost packet

- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

*TCP fast retransmit*

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq #
  - likely that unacked segment lost, so don’t wait for timeout
TCP fast retransmit

Host A
Seq=92, 8 bytes of data
Seq=100, 20 bytes of data
ACK=100
ACK=100
ACK=100
ACK=100
timeout
fast retransmit after sender receipt of triple duplicate ACK

Host B
Seq=100, 20 bytes of data
X

Transport Layer 3-20
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TCP flow control

receiver controls sender, so sender won’t overflow receiver’s buffer by transmitting too much, too fast
TCP flow control

- receiver “advertises” free buffer space by including \textit{rwnd} value in TCP header of receiver-to-sender segments
  - \texttt{RcvBuffer} size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust \texttt{RcvBuffer}
- sender limits amount of unacked (“in-flight”) data to receiver’s \textit{rwnd} value
- guarantees receive buffer will not overflow
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Connection Management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters

Socket clientSocket = newSocket("hostname","port number");
Socket connectionSocket = welcomeSocket.accept();
Agreeing to establish a connection

2-way handshake:

Q: will 2-way handshake always work in network?
- variable delays
- retransmitted messages (e.g. \texttt{req\_conn(x)}) due to message loss
- message reordering
- can’t “see” other side
Agreeing to establish a connection

2-way handshake failure scenarios:

- choose x
- retransmit req_conn(x)
- ESTAB
- client terminates
- half open connection! (no client!)

- choose x
- req_conn(x)
- ESTAB
- server forgets x

- choose x
- req_conn(x)
- ESTAB
- client terminates
- connection x completes

- choose x
- req_conn(x)
- ESTAB
- server forgets x

- choose x
- req_conn(x)
- ESTAB
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- choose x
- req_conn(x)
- ESTAB
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- choose x
- req_conn(x)
- ESTAB
- client terminates
- connection x completes

- choose x
- req_conn(x)
- ESTAB
- server forgets x

Transport Layer 3-27
TCP 3-way handshake

**client state**
- **LISTEN**: choose init seq num, x
- **SYNSENT**: send TCP SYN msg
- **ESTAB**: received SYNACK(x) indicates server is live; send ACK for SYNACK; this segment may contain client-to-server data

**server state**
- **LISTEN**: choose init seq num, y
- **SYN RCVD**: send TCP SYNACK msg, acking SYN
- **ESTAB**: received ACK(y) indicates client is live
TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled
TCP: closing a connection

client state

- ESTAB
- FIN_WAIT_1
  - clientSocket.close()
  - can no longer send but can receive data
- FIN_WAIT_2
  - wait for server close
- TIMED_WAIT
  - timed wait for 2*max segment lifetime
- CLOSED

server state

- ESTAB
- CLOSE_WAIT
  - can still send data
- LAST_ACK
  - can no longer send data
- CLOSED
Next class

- Please read Chapter 3.7-3.8 of your textbook BEFORE Class