Chapter 3
Transport Layer

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Chapter 3: Transport Layer

our goals:

- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control

- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control
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
Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP
Transport vs. network layer

- **network layer**: logical communication between hosts
- **transport layer**: logical communication between processes
  - relies on, enhances, network layer services

**household analogy:**

12 kids in Ann’s house sending letters to 12 kids in Bill’s house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service
Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of “best-effort” IP
- services not available:
  - delay guarantees
  - bandwidth guarantees
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Multiplexing/demultiplexing

**Multiplexing at sender:**
handle data from multiple sockets, add transport header (later used for demultiplexing)

**Demultiplexing at receiver:**
use header info to deliver received segments to correct socket

---

Transport Layer 3-8
How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses **IP addresses & port numbers** to direct segment to appropriate socket

TCP/UDP segment format:

- source port #
- dest port #
- other header fields
- application data (payload)
Connectionless demultiplexing

recall: created socket has host-local port #:
DatagramSocket mySocket1 = new DatagramSocket(12534);

recall: when creating datagram to send into UDP socket, must specify
- destination IP address
- destination port #

when host receives UDP segment:
- checks destination port # in segment
- directs UDP segment to socket with that port #

IP datagrams with same dest. port #, but different source IP addresses and/or source port numbers will be directed to same socket at dest
Connectionless demux: example

```java
DatagramSocket serverSocket = new DatagramSocket(6428);

DatagramSocket mySocket1 = new DatagramSocket(5775);

mySocket2 = new DatagramSocket(9157);

source port: 9157en)
dest port: 6428

source port: ?
dest port: ?

source port: 6428
dest port: 9157

source port: ?
dest port: ?
```

Transport Layer 3-11
Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request
Connection-oriented demux: example

three segments, all destined to IP address: B, dest port: 80 are demultiplexed to different sockets
Connection-oriented demux: example

Transport Layer 3-14
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UDP: User Datagram Protocol [RFC 768]

- “no frills,” “bare bones” Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- connectionless:
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP

- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!
**UDP: segment header**

<table>
<thead>
<tr>
<th>source port #</th>
<th>dest port #</th>
</tr>
</thead>
<tbody>
<tr>
<td>length</td>
<td>checksum</td>
</tr>
</tbody>
</table>

- **UDP segment format**
- **32 bits**
- **length, in bytes of UDP segment, including header**

**why is there a UDP?**

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired
**UDP checksum**

**Goal:** detect “errors” (e.g., flipped bits) in transmitted segment

**sender:**
- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one’s complement sum) of segment contents
- sender puts checksum value into UDP checksum field

**receiver:**
- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless? More later* ....
## Internet checksum: example

**example:** add two 16-bit integers

<p>| 1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0 |</p>
<table>
<thead>
<tr>
<th>1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>wraparound</strong></td>
</tr>
<tr>
<td>1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1</td>
</tr>
<tr>
<td><strong>sum</strong></td>
</tr>
<tr>
<td>1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0</td>
</tr>
<tr>
<td><strong>checksum</strong></td>
</tr>
<tr>
<td>0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1</td>
</tr>
</tbody>
</table>

*Note:* when adding numbers, a carryout from the most significant bit needs to be added to the result
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Principles of reliable data transfer

- Important in application, transport, link layers
  - Top-10 list of important networking topics!

- Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

(a) provided service
Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!

  - characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)
Principles of reliable data transfer

- important in application, transport, link layers
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- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

(a) provided service

(b) service implementation
Reliable data transfer: getting started

**rdt_send()**: called from above, (e.g., by app.). Passed data to deliver to receiver upper layer.

**send side**

**udt_send()**: called by rdt, to transfer packet over unreliable channel to receiver.

**receive side**

**data**

**deliver_data()**: called by rdt to deliver data to upper.

**rdt_rcv()**: called when packet arrives on rcv-side of channel.

**unreliable channel**

**reliable data transfer protocol (sending side)**

**reliable data transfer protocol (receiving side)**

Transport Layer 3-24
Reliable data transfer: getting started

We'll:
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver
**rdt 1.0**: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel

```
rdt_send(data)
```

```
extract (packet, data)
```

```
deliver_data(data)
```

sender

receiver
rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- the question: how to recover from errors:

How do humans recover from “errors” during conversation?
rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- the question: how to recover from errors:
  - acknowledgements (ACKs): receiver explicitly tells sender that pkt received OK
  - negative acknowledgements (NAKs): receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in rdt2.0 (beyond rdt1.0):
  - error detection
  - feedback: control msgs (ACK, NAK) from receiver to sender
**rdt2.0: FSM specification**

**sender**

- `rdt_send(data)`
- `sndpkt = make_pkt(data, checksum)`
- `udt_send(sndpkt)`

**receiver**

- `rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)`
  - `extract(rcvpkt, data)`
  - `deliver_data(data)`
  - `udt_send(ACK)`

- `wait for ACK or NAK`
- `wait for call from above`
- `rdt_rcv(rcvpkt) && isNAK(rcvpkt)`
  - `udt_send(sndpkt)`
- `rdt_rcv(rcvpkt) && isACK(rcvpkt)`
  - `udt_send(sndpkt)`
- `corrupt(rcvpkt)`
- `notcorrupt(rcvpkt)`
**rdt2.0: operation with no errors**

- `rdt_send(data)`
- `snkpkt = make_pkt(data, checksum)`
- `udt_send(sndpkt)`
- `wait for call from above`
- `wait for ACK or NAK`
- `rdt_rcv(rcvpkt) && isNAK(rcvpkt)`
- `udt_send(sndpkt)`
- `wait for call from below`
- `rdt_rcv(rcvpkt) && isACK(rcvpkt)`
- `udt_send(ACK)`
- `corrupt(rcvpkt)`
- `udt_send(NAK)`
- `notcorrupt(rcvpkt)`
- `extract(rcvpkt, data)`
- `deliver_data(data)`
- `udt_send(ACK)`

Diagram:
- Flowchart with nodes and edges illustrating the operations and decision points in the rdt2.0 protocol.
### rdt2.0: error scenario

- **rdt_send(data)**
  - `snkpkt = make_pkt(data, checksum)`
  - `udt_send(sndpkt)`

- **Wait for call from above**
  - `wait_for_ACK_or_NAK`

- **Wait for ACK or NAK**
  - `rdt_rcv(rcvpkt) && isNAK(rcvpkt)`
  - `udt_send(sndpkt)`

- **Wait for ACK or NAK**
  - `rdt_rcv(rcvpkt) && isACK(rcvpkt)`
  - `udt_send(ACK)`

- **Wait for ACK or NAK**
  - `rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)`
  - `extract(rcvpkt, data)`
  - `deliver_data(data)`
  - `udt_send(ACK)`

- **Wait for ACK or NAK**
  - `rdt_rcv(rcvpkt) && corrupt(rcvpkt)`
  - `udt_send(NAK)`

- **Wait for ACK or NAK**
  - `wait_for_ACK_or_NAK`

- **Transport Layer 3-31**
rdt2.0 has a fatal flaw!

what happens if 
ACK/NAK corrupted?

- sender doesn’t know what happened at receiver!
- can’t just retransmit: possible duplicate

handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- sender adds sequence number to each pkt
- receiver discards (doesn’t deliver up) duplicate pkt

stop and wait
sender sends one packet, then waits for receiver response
rdt2.1: sender, handles garbled ACK/NAKs

```c
rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

Wait for
ACK or
NAK 0

rdt_rcv(rcvpkt) && (notcorrupt(rcvpkt) && isACK(rcvpkt))
\Lambda

Wait for
call 0 from
above

Wait for
ACK or
NAK 1

rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isNAK(rcvpkt))

udt_send(sndpkt)

rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

Wait for
ACK or
NAK 1

rdt_rcv(rcvpkt) && (notcorrupt(rcvpkt) && isACK(rcvpkt))
\Lambda

Wait for
call 1 from
above

rdt_rcv(rcvpkt) && corrupt(rcvpkt) || isNAK(rcvpkt)
```

Transport Layer 3-33
**rdt2.1: receiver, handles garbled ACK/NAKs**

```
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
  sndpkt = make_pkt(NAK, chksum)
  udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
  sndpkt = make_pkt(NAK, chksum)
  udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
  sndpkt = make_pkt(ACK, chksum)
  udt_send(sndpkt)
```

Transport Layer 3-34
sender:
- seq # added to pkt
- two seq. #’s (0, 1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must “remember” whether “expected” pkt should have seq # of 0 or 1

receiver:
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can not know if its last ACK/NAK received OK at sender
rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*
**rdt2.2: sender, receiver fragments**

- **Sender FSM fragment**
  - `rtt_send(data)`
  - `sndpkt = make_pkt(0, data, checksum)`
  - `udt_send(sndpkt)`
  - `rtt_rtcv(rcvpkt) && (corrupt(rcvpkt) || has_seq1(rcvpkt))`
  - `udt_send(sndpkt)`

- **Receiver FSM fragment**
  - `rtt_rtcv(rcvpkt) && notcorrupt(rcvpkt) && has_seq1(rcvpkt)`
  - `extract(rcvpkt, data)`
  - `deliver_data(data)`
  - `sndpkt = make_pkt(ACK1, checksum)`
  - `udt_send(sndpkt)`

- **Flowchart**
  - `Wait for call 0 from above`
  - `Wait for ACK 0`
  - `IsACK(rcvpkt, 1)`
  - `udt_send(sndpkt)`

- **Conditions**
  - `rtt_rtcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt, 0)`

---

*Transport Layer 3-37*
**rdt3.0: channels with errors and loss**

**new assumption:** underlying channel can also lose packets (data, ACKs)
- checksum, seq. #, ACKs, retransmissions will be of help ... but not enough

**approach:** sender waits “reasonable” amount of time for ACK
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #’s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer
**rdt3.0 sender**

```
rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

wait for ACK0

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

Lambda

Wait for call 0 from above

Lambda

rdt_rcv(rcvpkt)
Lambda

 && notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)
stop_timer

Wait for ACK1

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,0))

Lambda

udt_send(sndpkt)
start_timer

Wait for call 1 from above

Lambda

rdt_rcv(rcvpkt)
Lambda

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

timeout

stop_timer

transport Layer 3-39
```
rdt3.0 in action

(a) no loss

(b) packet loss

Transport Layer 3-40
rdt3.0 in action

**sender**
- send pkt0
- rcv pkt0
- rcv ack0
- send pkt1
- rcv pkt1
- rcv ack1
- timeout
- resend pkt1
- rcv pkt1
- rcv ack1
- send ack1
- rcv ack1
- send pkt0
- rcv pkt0
- send ack0

**receiver**
- send pkt0
- rcv pkt0
- send ack0
- rcv ack0
- send pkt1
- rcv pkt1
- send ack1
- rcv pkt1
- send ack1
- timeout
- resend pkt1
- rcv pkt1
- send ack1
- rcv ack1
- send pkt0
- rcv pkt0
- send ack0

(c) ACK loss

(d) premature timeout/ delayed ACK
Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

\[ D_{\text{trans}} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs} \]

- \( U_{\text{sender}} \): utilization – fraction of time sender busy sending

\[ U_{\text{sender}} = \frac{L / R}{\text{RTT} + L / R} = \frac{0.008}{30.008} = 0.00027 \]

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec throughput over 1 Gbps link

- network protocol limits use of physical resources!
rdt3.0: stop-and-wait operation

\[ U_{\text{sender}} = \frac{L/R}{RTT + L/R} = \frac{0.008}{30.008} = 0.00027 \]
Pipelined protocols

**pipelining:** sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

- two generic forms of pipelined protocols: *go-Back-N*, *selective repeat*

(a) a stop-and-wait protocol in operation  
(b) a pipelined protocol in operation

*Transport Layer 3-44*
Pipelining: increased utilization

3-packet pipelining increases utilization by a factor of 3!

\[ U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{0.0024}{30.008} = 0.00081 \]
## 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
Go-Back-N: sender

- k-bit seq # in pkt header
- “window” of up to N, consecutive unack’ed pkts allowed

Diagram:
- send_base
- nextseqnum
- Window size N
- already ack’ed
- sent, not yet ack’ed
- usable, not yet sent
- not usable

- ACK(n): ACKs all pkts up to, including seq # n - “cumulative ACK”
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- timeout(n): retransmit packet n and all higher seq # pkts in window
GBN: sender extended FSM

rdt_send(data)

if (nextseqnum < base+N) {
    sndpkt[nextseqnum] = make_pkt(nextseqnum, data, chksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
        start_timer
    nextseqnum++
} else
    refuse_data(data)

base = 1
nextseqnum = 1

rdt_rcv(rcvpkt)
    && corrupt(rcvpkt)

timeout
    start_timer
    udt_send(sndpkt[base])
    udt_send(sndpkt[base+1])
    ...
    udt_send(sndpkt[nextseqnum-1])

rdt_rcv(rcvpkt)
    && notcorrupt(rcvpkt)

base = getacknum(rcvpkt)+1
if (base == nextseqnum)
    stop_timer
else
    start_timer
GBN: receiver extended FSM

ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #
- may generate duplicate ACKs
- need only remember `expectedseqnum`

- out-of-order pkt:
  - discard (don’t buffer): *no receiver buffering!*
  - re-ACK pkt with highest in-order seq #
GBN in action

**sender window (N=4)**

<table>
<thead>
<tr>
<th>012345678</th>
<th>012345678</th>
<th>012345678</th>
<th>012345678</th>
</tr>
</thead>
<tbody>
<tr>
<td>sender</td>
<td>sender</td>
<td>receiver</td>
<td>receiver</td>
</tr>
<tr>
<td>send pkt0</td>
<td>send pkt1</td>
<td>receive pkt0, send ack0</td>
<td>receive pkt0, send ack0</td>
</tr>
<tr>
<td>send pkt1</td>
<td>send pkt2</td>
<td>receive pkt1, send ack1</td>
<td>receive pkt1, send ack1</td>
</tr>
<tr>
<td>send pkt2</td>
<td>send pkt3</td>
<td>receive pkt3, discard, (re)send ack1</td>
<td>receive pkt3, discard, (re)send ack1</td>
</tr>
<tr>
<td>(wait)</td>
<td></td>
<td>receive pkt4, discard, (re)send ack1</td>
<td>receive pkt4, discard, (re)send ack1</td>
</tr>
<tr>
<td>rcv ack0, send pkt4</td>
<td>rcv ack1, send pkt5</td>
<td>receive pkt5, discard, (re)send ack1</td>
<td>receive pkt5, discard, (re)send ack1</td>
</tr>
<tr>
<td>ignore duplicate ACK</td>
<td>pkt 2 timeout</td>
<td></td>
<td></td>
</tr>
<tr>
<td>send pkt2</td>
<td>send pkt3</td>
<td>rcv pkt2, deliver, send ack2</td>
<td>rcv pkt2, deliver, send ack2</td>
</tr>
<tr>
<td>send pkt3</td>
<td>send pkt4</td>
<td>rcv pkt3, deliver, send ack3</td>
<td>rcv pkt3, deliver, send ack3</td>
</tr>
<tr>
<td>send pkt4</td>
<td>send pkt5</td>
<td>rcv pkt4, deliver, send ack4</td>
<td>rcv pkt4, deliver, send ack4</td>
</tr>
<tr>
<td>send pkt5</td>
<td></td>
<td>rcv pkt5, deliver, send ack5</td>
<td>rcv pkt5, deliver, send ack5</td>
</tr>
</tbody>
</table>

Transport Layer 3-50
Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - $N$ consecutive seq #’s
  - limits seq #s of sent, unACKed pkts
Selective repeat: sender, receiver windows

(a) sender view of sequence numbers

(b) receiver view of sequence numbers
Selective repeat

sender

data from above:
- if next available seq # in window, send pkt

timeout(n):
- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:
- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]:
- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

pkt n in [rcvbase-N,rcvbase-1]
- ACK(n)

otherwise:
- ignore
Selective repeat in action

**sender window (N=4)**

- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8
- 0 1 2 3 4 5 6 7 8

**sender**

- send pkt0
- send pkt1
- send pkt2
- send pkt3 (wait)

**receiver**

- receive pkt0, send ack0
- receive pkt1, send ack1
- receive pkt3, buffer, send ack3
- receive pkt4, buffer, send ack4
- receive pkt5, buffer, send ack5
- receive pkt2
- deliver pkt2, pkt3, pkt4, pkt5; send ack2

**Q: what happens when ack2 arrives?**

- record ack3 arrived
- record ack4 arrived
Selective repeat: dilemma

example:
- seq #’s: 0, 1, 2, 3
- window size=3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)

Q: what relationship between seq # size and window size to avoid problem in (b)?
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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) inits sender, receiver state before data exchange

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

- **source port #**
- **dest port #**
- **sequence number**
- **acknowledgement number**
- **receive window**
- **checksum**
- **Urg data pointer**
- **options (variable length)**
- **application data** (variable length)

**URG:** urgent data (generally not used)

**ACK:** ACK # (valid)

**PSH:** push data now (generally not used)

**RST, SYN, FIN:** connection estab (setup, teardown commands)

**Internet checksum** (as in UDP)

**counting by bytes of data (not segments!)**

**# bytes rcvr willing to accept**

Transport Layer 3-58
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

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

**Simple telnet scenario**

Host A

User types 'C'

Seq=42, ACK=79, data = ‘C’

Seq=79, ACK=43, data = ‘C’

host ACKs receipt of echoed ‘C’

Seq=43, ACK=80

Host B

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

EstimatedRTT = (1 - α)*EstimatedRTT + α*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, \( \beta = 0.25 \))

\[
\text{TimeoutInterval} = \text{EstimatedRTT} + 4 \times \text{DevRTT}
\]

- estimated RTT
- "safety margin"
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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)

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

ACK received, with ACK field value y
if (y > SendBase) {
    SendBase = y
    /* 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

timeout

Host B

Seq=92, 8 bytes of data

X

premature timeout

Host A

SendBase=92
Seq=92, 8 bytes of data

SendBase=100
Seq=100, 20 bytes of data

ACK=100

SendBase=120

ACK=120

Host B

Seq=92, 8 bytes of data

ACK=120

SendBase=120

ACK=120
TCP: retransmission scenarios

Host A
Seq=92, 8 bytes of data
Seq=100, 20 bytes of data
timeout
Seq=120, 15 bytes of data
cumulative ACK

Host B
ACK=100
X
ACK=120
<table>
<thead>
<tr>
<th>event at receiver</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 duplicate ACK, 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.

  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

Host B

Seq=100, 20 bytes of data

ACK=100

ACK=100

ACK=100

ACK=100

ACK=100

Seq=100, 20 bytes of data

fast retransmit after sender receipt of triple duplicate ACK
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TCP flow control

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 rwnd value in TCP header of receiver-to-sender segments
  - RcvBuffer size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust RcvBuffer
- sender limits amount of unacked (“in-flight”) data to receiver’s 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

```java
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. req_conn(x)) due to message loss
- message reordering
- can’t “see” other side
Agreeing to establish a connection

2-way handshake failure scenarios:

1. Client chooses to initiate a connection (req_conn(x)).
2. Server accepts the request (acc_conn(x)).
3. Client completes the connection.
4. Server forgets the connection.
5. Client sends data (data(x+1)).
6. Server accepts the data.
7. Client terminates the connection.
8. Server forgets the connection.

**Half open connection!** (no client!)

Transport Layer 3-79
TCP 3-way handshake

**client state**
- **LISTEN**
- choose init seq num, x
- send TCP SYN msg

**server state**
- **LISTEN**

SYNSENT
- SYNbit=1, Seq=x
- SYNbit=1, Seq=y
- ACKbit=1; ACKnum=x+1
- ACKbit=1, ACKnum=y+1

ESTAB
- received SYNACK(x)
- ACKbit=1, ACKnum=x+1
- received ACK(y)
- indicates server is live;
  send ACK for SYNACK;
  this segment may contain client-to-server data
- received ACK(y)
- indicates client is live
TCP 3-way handshake: FSM

Socket connectionSocket =
welcomeSocket.accept();

SYN(x)

SYNACK(seq=y,ACKnum=x+1)
create new socket for
communication back to client

ACK(ACKnum=y+1)

Socket clientSocket =
newSocket(“hostname”, “port
number”);

SYN(seq=x)

SYN rcvd

ESTAB

SYN sent

Transport Layer 3-81
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
    - FINbit=1, seq=x
      - can no longer send but can receive data
      - clientSocket.close()
  - FIN_WAIT_2
    - wait for server close
  - TIMED_WAIT
    - timed wait for 2*max segment lifetime
  - CLOSED

**server state**

- ESTAB
  - FIN bit=1, seq=x
  - ACKbit=1; ACKnum=x+1
  - can still send data
- CLOSE_WAIT
  - FINbit=1, seq=y
  - ACKbit=1; ACKnum=y+1
  - can no longer send data
- LAST_ACK
  - can no longer send data
- CLOSED
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Principles of congestion control

congestion:

- informally: “too many sources sending too much data too fast for network to handle”
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!
Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission

Original data:\n\[ \lambda_{in} \]

Throughput:\n\[ \lambda_{out} \]

Unlimited shared output link buffers

Host A

Host B

- maximum per-connection throughput: \( \frac{R}{2} \)
- large delays as arrival rate, \( \lambda_{in} \), approaches capacity
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{\text{in}} = \lambda_{\text{out}}$
  - transport-layer input includes *retransmissions*: $\lambda_{\text{in}}' \geq \lambda_{\text{in}}$
Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

\[ \text{in: original data, plus retransmitted data} \]

\[ \lambda^{\prime}_{\text{in}} : \text{original data, plus retransmitted data} \]

\[ \lambda_{\text{in}} : \text{original data} \]

\[ \text{free buffer space!} \]

finite shared output link buffers

Host B

Transport Layer 3-88
Causes/costs of congestion: scenario 2

**Idealization: known loss**

- packets can be lost, dropped at router due to full buffers
- sender only resends if packet known to be lost

If $\lambda_{\text{in}}$ : original data, then $\lambda'_{\text{in}}$ : original data, plus retransmitted data
**Causes/costs of congestion: scenario 2**

*Idealization: known loss*

packets can be lost, dropped at router due to full buffers

- sender only resends if packet known to be lost

![Diagram](image)

\[ \lambda_{\text{in}} : \text{original data} \]

\[ \lambda'_{\text{in}} : \text{original data, plus retransmitted data} \]

\[ \lambda_{\text{out}} \]

when sending at R/2, some packets are retransmissions but asymptotic goodput is still R/2 (why?)

Transport Layer 3-90
Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending two copies, both of which are delivered

![Diagram showing network traffic and buffer space](Transport_Layer_3-91)
Causes/costs of congestion: scenario 2

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending two copies, both of which are delivered

“costs” of congestion:

- more work (retrans) for given “goodput”
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput
Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

Q: what happens as $\lambda_{in}$ and $\lambda_{in}'$ increase?

A: as red $\lambda_{in}'$ increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow 0$
Causes/costs of congestion: scenario 3

another “cost” of congestion:

- when packet dropped, any “upstream transmission capacity used for that packet was wasted!
Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:
- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:
- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at
Case study: ATM ABR congestion control

**ABR: available bit rate:**
- “elastic service”
- if sender’s path “underloaded”:
  - sender should use available bandwidth
- if sender’s path congested:
  - sender throttled to minimum guaranteed rate

**RM (resource management) cells:**
- sent by sender, interspersed with data cells
- bits in RM cell set by switches (“network-assisted”)
  - *NI bit:* no increase in rate (mild congestion)
  - *CI bit:* congestion indication
- RM cells returned to sender by receiver, with bits intact
Case study: ATM ABR congestion control

- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - senders’ send rate thus max supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, receiver sets CI bit in returned RM cell
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TCP congestion control: additive increase
multiplicative decrease

- **Approach**: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - **Additive increase**: increase cwnd by 1 MSS every RTT until loss detected
  - **Multiplicative decrease**: cut cwnd in half after loss

AIMD saw tooth behavior: probing for bandwidth
TCP Congestion Control: details

TCP sending rate:
- roughly: send cwnd bytes, wait RTT for ACKS, then send more bytes

sender limits transmission:
- LastByteSent - LastByteAcked ≤ cwnd

- cwnd is dynamic, function of perceived network congestion

\[
\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}
\]
TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially \( cwnd = 1 \) MSS
  - double \( cwnd \) every RTT
  - done by incrementing \( cwnd \) for every ACK received

- summary: initial rate is slow but ramps up exponentially fast
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - $cwnd$ set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly

- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - $cwnd$ is cut in half window then grows linearly

- TCP Tahoe always sets $cwnd$ to 1 (timeout or 3 duplicate acks)
TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?
A: when cwnd gets to 1/2 of its value before timeout.

Implementation:
- variable sssthresh
- on loss event, sssthresh is set to 1/2 of cwnd just before loss event
Summary: TCP Congestion Control

- **Slow Start**
  - $cwnd = 1$ MSS
  - $ssthresh = 64$ KB
  - $dupACKcount = 0$
  - If timeout, $ssthresh = cwnd/2$
  - $cwnd = 1$ MSS
  - $dupACKcount = 0$

- **Fast Recovery**
  - $cwnd = ssthresh$
  - $dupACKcount = 0$
  - $ssthresh = cwnd/2$
  - $cwnd = ssthresh + 3$
  - $dupACKcount = 3$

- **Congestion Avoidance**
  - $cwnd \geq ssthresh$
  - If timeout, $ssthresh = cwnd/2$
  - $cwnd = 1$ MSS
  - $dupACKcount = 0$
  - $dupACKcount = 3$

- **New ACK**
  - $cwnd = cwnd + MSS$
  - $cwnd = cwnd + MSS$ (MSS/cwnd)
  - $dupACKcount = 0$
  - Transmit new segment(s), as allowed

- **Slow Start**
  - $dupACKcount++$
  - Transmit new segment(s), as allowed

- **Fast Recovery**
  - $dupACKcount = 3$
  - Transmit new segment(s), as allowed

- **Congestion Avoidance**
  - $dupACKcount = 0$
  - Transmit new segment(s), as allowed

Transport Layer 3-104
TCP throughput

- avg. TCP throughput as function of window size, RTT?
  - ignore slow start, assume always data to send

- $W$: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is $\frac{3}{4} W$
  - avg. throughput is $\frac{3}{4}W$ per RTT

$$\text{avg TCP throughput} = \frac{3}{4} \frac{W}{\text{RTT}} \text{ bytes/sec}$$
TCP Futures: TCP over “long, fat pipes”

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L
  \[\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT} \sqrt{L}}\]

  \(\rightarrow\) to achieve 10 Gbps throughput, need a loss rate of \(L = 2 \cdot 10^{-10}\) – a very small loss rate!

- new versions of TCP for high-speed
**TCP Fairness**

*fairness goal:* if K TCP sessions share same bottleneck link of bandwidth $R$, each should have average rate of $R/K$
Why is TCP fair?

two competing sessions:
- additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally

connection diagram with line and arrows indicating additive increase and multiplicative decrease upon loss.
Fairness (more)

**Fairness and UDP**
- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss

**Fairness, parallel TCP connections**
- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate $R$ with 9 existing connections:
  - new app asks for 1 TCP, gets rate $R/10$
  - new app asks for 11 TCPs, gets $R/2$
Chapter 3: summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control

- instantiation, implementation in the Internet
  - UDP
  - TCP

next:
- leaving the network “edge” (application, transport layers)
- into the network “core”