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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 transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented 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 sending letters to 12 kids
  - processes = kids
  - app messages = letters in envelopes
  - hosts = houses
  - transport protocol = Ann and Bill
  - 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

Demultiplexing at rcv host:
delivering received segments to correct socket

Multiplexing at send host:
gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

= socket  = process

application | P3
transport
network
link
physical

application | P1
transport
network
link
physical

application | P2
transport
network
link
physical

P4
application
transport
network
link
physical

host 1

host 2

host 3
How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 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
Connectionless demultiplexing

- **Create sockets with port numbers:**
  
  ```java
  DatagramSocket mySocket1 = new DatagramSocket(99111);
  DatagramSocket mySocket2 = new DatagramSocket(99222);
  ```

- **UDP socket identified by two-tuple:**
  
  `(dest IP address, dest port number)`

- **When host receives UDP segment:**
  
  - checks destination port number in segment
  - directs UDP segment to socket with that port number

- **IP datagrams with different source IP addresses and/or source port numbers directed to same socket**

---

TransportLayer 3-10
Connectionless demux (cont)

DatagramSocket serverSocket = new DatagramSocket(6428);

SP provides "return address"
Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host 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 (cont)

Client
IP: A

P1

P2

P3

Server
IP: C

SP: 9157
DP: 80
S-IP: A
D-IP: C

SP: 5775
DP: 80
S-IP: B
D-IP: C

Client
IP: B

SP: 9157
DP: 80
S-IP: B
D-IP: C
Connection-oriented demux: Threaded

Web Server
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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

Why is there a UDP?
- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired
**UDP: more**

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!

![UDP segment format](chart)

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

Length, in bytes of UDP segment, including header

Application data (message)
**UDP checksum**

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

**Sender:**
- treat segment contents as sequence of 16-bit integers
- checksum: addition (1'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**

- **Note**
  - When adding numbers, a carryout from the most significant bit needs to be added to the result

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

```
1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0
1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1
```

**Wraparound**

```
1 1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1
```

**Sum**

```
1 0 1 1 1 0 1 1 1 0 1 1 1 1 1 0 0
```

**Checksum**

```
0 1 0 0 0 1 0 0 0 1 0 0 0 0 0 1 1
```
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Principles of Reliable data transfer

- important in app., 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
(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.
- **udt_send()**: called by rdt, to transfer packet over unreliable channel to receiver.
- **deliver_data()**: called by rdt to deliver data to upper.
- **rdt_rcv()**: called when packet arrives on rcv-side of channel.
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

Diagram:

- State 1:
  - When in this state, the next state is uniquely determined by the next event.

- State 2:
  - Event causing state transition:
  - Actions taken on state transition:

- Event:
  - Actions:
Rdt1.0: **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

sender

```
wait for call from above

packet = make_pkt(data)
udt_send(packet)

```

receiver

```
wait for call from below

rdt_rcv(packet)
extract (packet, data)
deliver_data(data)
```
**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:
  - acknowledgments (ACKs): receiver explicitly tells sender that pkt received OK
  - negative acknowledgments (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
  - receiver feedback: control msgs (ACK, NAK) rcvr→sender
**rdt2.0: FSM specification**

Sender

- `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)`

Receiver

- `rdt_rcv(rcvpkt) && isACK(rcvpkt)`
- `\Lambda`

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

- `rdt_rcv(rcvpkt) &&`
- `udt_send(NAK)`

Wait for call from below
**rdt2.0: operation with no errors**

- `rdt_send(data)`
- `snkpkt = make_pkt(data, checksum)`
- `udt_send(snkpkt)`
- `rdt_rcv(rcvpkt) && isNAK(rcvpkt)`
- `udt_send(snkpkt)`
- `rdt_rcv(rcvpkt) && isACK(rcvpkt)`
- `Lambda

Wait for call from above

Wait for ACK or NAK

Wait for call from below

- `rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)`
- `extract(rcvpkt, data)`
- `deliver_data(data)`
- `udt_send(ACK)`
- `udt_send(NAK)`
**rdt2.0 error scenario**

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

- `wait for call from above`

- `rdt_rcv(rcvpkt) && isACK(rcvpkt)`

- `Lambda` (Λ)

- `wait for ACK or NAK`

- `rdt_rcv(rcvpkt) && isNAK(rcvpkt)`
  - `udt_send(snkpkt)`

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

- `wait for call from below`

- `rdt_rcv(rcvpkt) && corrupt(rcvpkt)`
  - `udt_send(NAK)`
**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 its current pkt if ACK/NAK garbled
- 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**

**Algorithm**

1. **rdt_send(data)**
   - `sndpkt = make_pkt(0, data, checksum)`
   - `udt_send(sndpkt)`

2. **Wait for call 0 from above**
3. **Wait for ACK or NAK 0**
   - `rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt)`
   - `udt_send(sndpkt)`
   - `Lambda` (denotes end of cycle)
4. **Wait for ACK or NAK 1**
   - `rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt)`
   - `Lambda`
5. **rdt_send(data)**
   - `sndpkt = make_pkt(1, data, checksum)`
   - `udt_send(sndpkt)`

**Implementations**

- For garbled ACKs:
  - `rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isNAK(rcvpkt))` (check for corrupt or NAK frames)

- For garbled NAKs:
  - `rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isNAK(rcvpkt))`
**rdt2.1: receiver, handles garbled ACK/NAKs**

```
rdt_rvc(rcvpkt) && notcorrupt(rcvpkt)
    && has_seq0(rcvpkt)
    
    extract(rcvpkt,data)
    deliver_data(data)
    sndpkt = make_pkt(ACK, checksum)
    udt_send(sndpkt)
```

```
rdt_rvc(rcvpkt) &&
    (corrupt(rcvpkt)
    sndpkt = make_pkt(NAK, checksum)
    udt_send(sndpkt)
```

```
rdt_rvc(rcvpkt) &&
    not corrupt(rcvpkt) &&
    has_seq1(rcvpkt)
    
    sndpkt = make_pkt(ACK, checksum)
    udt_send(sndpkt)
```

```
rdt_rvc(rcvpkt) &&
    (corrupt(rcvpkt)
    sndpkt = make_pkt(NAK, checksum)
    udt_send(sndpkt)
```

```
rdt_rvc(rcvpkt) &&
    not corrupt(rcvpkt) &&
    has_seq0(rcvpkt)
    
    sndpkt = make_pkt(ACK, checksum)
    udt_send(sndpkt)
```

```
rdt_rvc(rcvpkt) &&
    (corrupt(rcvpkt)
    sndpkt = make_pkt(NAK, checksum)
    udt_send(sndpkt)
```

TransportLayer 3-31
rdt2.1: discussion

**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 “current” pkt has 0 or 1 seq. #

**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**
- `rdt_send(data)`
- `sndpkt = make_pkt(0, data, checksum)`
- `udt_send(sndpkt)`
- Wait for call 0 from above

**receiver FSM**
- `rdt_rcv(rcvpkt) & & (corrupt(rcvpkt) || isACK(rcvpkt,1))`
- `udt_send(sndpkt)`
- `rdt_rcv(rcvpkt) & & notcorrupt(rcvpkt) & & isACK(rcvpkt,0)`
- Wait for 0 from below

- `extract(rcvpkt,data)`
- `deliver_data(data)`
- `sndpkt = make_pkt(ACK1, checksum)`
- `udt_send(sndpkt)`

**Fragment Diagram**
**rdt3 0: channels with errors and loss**

**New assumption:** underlying channel can also lose packets (data or 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 use of seq# 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`

- **rdt_rcv(rcvpkt)**
  - `wait for call 0 from above`
  - `timeout`
  - `udt_send(sndpkt)`
  - `start_timer`

- **rdt_send(data)**
  - `sndpkt = make_pkt(1, data, checksum)`
  - `udt_send(sndpkt)`
  - `start_timer`

- **rdt_rcv(rcvpkt)**
  - `wait for call 1 from above`
  - `timeout`
  - `udt_send(sndpkt)`
  - `start_timer`

- **(corrupt(rcvpkt) || isACK(rcvpkt,1))**
rdt3.0 in action

(a) operation with no loss

(b) lost packet
**rdt3.0 in action**

(c) lost ACK

(d) premature timeout
Performance of rdt3.0

- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e2e prop. delay, 1KB packet:

\[ T_{transmit} = \frac{L}{R} = \frac{8kb/pt}{10^{\times 9} b/sec} = 8 \text{ microsec} \]

\[ U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027 \]

- \( U_{sender} \): utilization — fraction of time sender busy sending
- 1KB pkt every 30 m sec \( \rightarrow \) 33kB/sec throughput over 1 Gbps link
- network protocol limits use of physical resources!
**rdt3 0: stop-and-wait operation**

First packet bit transmitted, $t = 0$

Last packet bit transmitted, $t = L / R$

ACK arrives, send next packet, $t = RTT + L / R$

First packet bit arrives

Last packet bit arrives, send ACK

$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{0.008}{30.008} = 0.00027$
**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
Pipelining: increased utilization

first packet bit transmitted, $t = 0$
last bit transmitted, $t = \frac{L}{R}$

first packet bit arrives
last packet bit arrives, send ACK
last bit of 2nd packet arrives, send ACK
last bit of 3rd packet arrives, send ACK

ACK arrives, send next packet, $t = \text{RTT} + \frac{L}{R}$

Increase utilization by a factor of 3!

$$U_{\text{sender}} = \frac{3 \times \frac{L}{R}}{\text{RTT} + \frac{L}{R}} = \frac{.024}{30.008} = 0.0008$$
Go-Back-N

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

ACK (n): ACKs all pkts up to, including seq # n — “cumulative ACK”
- may deceive duplicate ACKs (see receiver)
- timer foreach in-flight pkt
- timeout(n): retransmit pkt 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, cksum)
    udt_send(sndpkt[nextseqnum])
    if (base == nextseqnum)
        start_timer
    nextseqnum++
}
else
    refuse_data(data)

else
    refuse_data(data)
```

Diagram:
- Wait
- rdt_send(data)
- rdt_recv(rcvpkt) & & corrupt(rcvpkt)
- base = getacknum(rcvpkt)+1
- If (base == nextseqnum)
  - stop_timer
- else
  - start_timer
- timeout
- start_timer
- udt_send(sndpkt[base])
- udt_send(sndpkt[base+1])
- ...
- udt_send(sndpkt[nextseqnum-1])
- udt_send(sndpkt[nextseqnum-1])
- notcorrupt(rcvpkt)

TransportLayer 3-44
**GBN: receiver extended FSM**

- Default:
  - `udt_send(sndpkt)`

- `udt_send(sndpkt)`

- `expectedseqnum = 1`

- `sndpkt = make_pkt(expectedseqnum, ACK, checksum)`

- `rdt_rcv(rcvpkt)`
  - `&& notcurrept(rcvpkt)`
  - `&& hasseqnum(rcvpkt, expectedseqnum)`
  - `extract(rcvpkt, data)`
  - `deliver_data(data)`

- `sndpkt = make_pkt(expectedseqnum, ACK, checksum)`

- `udt_send(sndpkt)`

- `expectedseqnum++`

---

**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

send pkt0
send pkt1
send pkt2
send pkt3 (wait)

receiver

rcv pkt0
send ACK0

rcv pkt1
send ACK1

rcv pkt3, discard
send ACK1

rcv pkt4, discard
send ACK1

rcv pkt5, discard
send ACK1

pkt2 timeout

send pkt2
send pkt3
send pkt4
send pkt5

rcv pkt2, deliver
send ACK2
rcv pkt3, deliver
send ACK3
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 unACK ed pkt
- sender window
  - $N$ consecutive seq #’s
  - again limits seq #’s of sent, unACK ed 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

pkt0 sent
0 1 2 3 4 5 6 7 8 9

pkt1 sent
0 1 2 3 4 5 6 7 8 9

pkt2 sent
0 1 2 3 4 5 6 7 8 9

pkt3 sent, window full
0 1 2 3 4 5 6 7 8 9

ACK0 rcvd, pkt4 sent
0 1 2 3 4 5 6 7 8 9

ACK1 rcvd, pkt5 sent
0 1 2 3 4 5 6 7 8 9

pkt2 TIMEOUT, pkt2 resent
0 1 2 3 4 5 6 7 8 9

ACK3 rcvd, nothing sent
0 1 2 3 4 5 6 7 8 9

pkt0 rcvd, delivered, ACK0 sent
0 1 2 3 4 5 6 7 8 9

pkt1 rcvd, delivered, ACK1 sent
0 1 2 3 4 5 6 7 8 9

pkt3 rcvd, buffered, ACK3 sent
0 1 2 3 4 5 6 7 8 9

pkt4 rcvd, buffered, ACK4 sent
0 1 2 3 4 5 6 7 8 9

pkt5 rcvd, buffered, ACK5 sent
0 1 2 3 4 5 6 7 8 9

pkt2 rcvd, pkt2,pkt3,pkt4,pkt5 delivered, ACK2 sent
0 1 2 3 4 5 6 7 8 9
Selective repeat: dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size = 3

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?
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
TCP: Overview
RFCs: 793, 1122, 1323, 2018, 2581

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

- **reliable, in-order byte stream:**
  - no "message boundaries"

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

- **send & receive buffers**

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

- **connection-oriented:**
  - handshaking (exchange of control messages) in its sender, receiver state before data exchange

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

- **URG**: urgent data (generally not used)
- **ACK**: ACK # valid
- **PSH**: push data now (generally not used)
- **RST, SYN, FIN**: connection establish (setup, teardown commands)
- **Internet checksum**: as in UDP

**TCP Segment Structure Diagram**

- **Source port #**
- **Destination port #**
- **Sequence number**
- **Acknowledgement number**
- **Receive window**
- **Checksum**
- **Urgent data pointer**
- **Options (variable length)**
- **Application data (variable length)**
**TCP seq. #’s and ACKs**

**Seq. #’s:**
- byte stream “number” of first byte in segment’s data

**ACKs:**
- 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

---

**User**

```
<table>
<thead>
<tr>
<th>Seq</th>
<th>ACK</th>
<th>Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>42</td>
<td>79</td>
<td>‘C’</td>
</tr>
<tr>
<td>79</td>
<td>43</td>
<td>‘C’</td>
</tr>
<tr>
<td>43</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```

**HostA**

```
Seq=42, ACK=79, data = ‘C’
```

**HostB**

```
Seq=79, ACK=43, data = ‘C’
```

```
Seq=43, ACK=80
```

**Time**

**simple telnet scenario**
TCP Round Trip Time and 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 and Timeout

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

- Exponential weighted moving average
- Influence of past sample decreases exponentially fast
- Typical value: \(\alpha = 0.125\)
Example RTT estimation:

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr

SampleRTT vs Estimated RTT over time in milliseconds.
TCP Round Trip Time and Timeout

Setting the timeout

- EstimatedRTT plus “safety margin”
  - large variation in EstimatedRTT \(\rightarrow\) larger safety margin
- First estimate of how much SampleRTT deviates from EstimatedRTT:

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

(typically, \(\beta = 0.25\))

Then set timeout interval:

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

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer

- Retransmissions are triggered by:
  - timeout events
  - duplicate acks

- 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:**
ExpirationInterval

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

**Ack rcvd:**
- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
    switch(event)

    event: data received from application above
        create TCP segment with sequence number NextSeqNum
        if (timer currently not running)
            start timer
        pass segment to IP
        NextSeqNum = NextSeqNum + length(data)

    event: timer timeout
        retransmit not-yet-acknowledged segment with
            smallest sequence number
        start timer

    event: ACK received, with ACK field value of y
        if (y > SendBase) {
            SendBase = y
            if (there are currently not-yet-acknowledged segments)
                start timer
        }
} /* end of loop forever */
TCP: retransmission scenarios

Host A

Seq = 100, 20 bytes data

ACK = 100

timeout

Seq = 92, 8 bytes data

ACK = 100

loss

SendBase = 100

Host B

Seq = 92, 8 bytes data

ACK = 120

timeout

Seq = 92, 8 bytes data

ACK = 120

SendBase = 120

SendBase = 100

Premature timeout
TCP retransmission scenarios (more)

Host A
Seq = 92, 8 bytes data
ACK = 100
Timeout
Cumulative ACK scenario

Host B
Seq = 100, 20 bytes data
ACK = 120
SendBase = 120

CumulativeACK scenario
## TCP ACK generation [RFC 1122, RFC 2581]

<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>
Fast Retransmit

- Timeout 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 the same data, it supposes that segment after ACKed data was lost:
  - Fast retransmit: resend segment before timer expires
**Fast retransmit algorithm:**

Event: ACK received, with ACK field value of $y$

- If ($y > \text{SendBase}$) {
  - $\text{SendBase} = y$
  - If (there are currently not-yet-acknowledged segments)
    - Start timer
}
- Else {
  - Increment count of dup ACKs received for $y$
  - If (count of dup ACKs received for $y = 3$) {
    - Resend segment with sequence number $y$
  }
}

A duplicate ACK for already ACK'ed segment
Chapter 3 outline

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- 3.7 TCP congestion control
TCP Flow Control

- receive side of TCP connection has a receive buffer.

- flow control
  sender won't overflow receiver's buffer by transmitting too much, too fast

- speed-matching service: matching the send rate to the receiving app's drain rate

- app process may be slow at reading from buffer
TCP Flow control: how it works

- Rcvr advertises spare room by including value of RcvWindow in segments
- Sender limits unACKed data to RcvWindow
  - guarantees receive buffer doesn’t overflow

Suppose TCP receiver discards out-of-order segments

- spare room in buffer
  - \( RcvWindow \)
  - \( RcvBuffer - [ \text{LastByteRcvd} - \text{LastByteRead} ] \)
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**TCP Connection Management**

*Recall:* TCP sender, receiver establish “connection” before exchanging data segments

- **initialize TCP variables:**
  - seq.#s
  - buffers, flow control info (e.g. RcvWindow)

- **client: connection initiator**
  
  Socket clientSocket = new Socket("hostname","port number");

- **server: contacted by client**
  
  Socket connectionSocket = welcomeSocket.accept();

---

**Three way handshake:**

**Step 1:** client host sends TCP SYN segment to server
  - specifies initial seq #
  - no data

**Step 2:** server host receives SYN, replies with SYNACK segment
  - server allocates buffers
  - specifies server initial seq. #

**Step 3:** client receives SYNACK, replies with ACK segment, which may contain data
**TCP Connection Management (cont.)**

**Closing a connection:**

Client closes socket:
```java
clientSocket.close();
```

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

**Step 2:** Server receives FIN, replies with ACK. Closes connection, sends FIN.

![Diagram of TCP connection closing](image)
**TCP Connection Management (cont.)**

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

**Step 4:** server receives ACK.
Connection closed.

**Note:** with small modification, can handle simultaneous FINs.
TCP Connection Management (cont)

TCP client lifecycle:
- CLOSED
  - wait 30 seconds
  - receive FIN, send ACK
- TIME_WAIT
  - receive ACK, send nothing
- FIN_WAIT_2
  - receive FIN, send ACK
- FIN_WAIT_1
  - send FIN
- ESTABLISHED
  - client application initiates a TCP connection
- SYN_SENT
  - receive SYN & ACK, send ACK
- SYN_RCVD
  - receive FIN, send ACK
  - ESTABLISHED
  - receive ACK, send nothing
  - CLOSE_WAIT
  - send FIN
  - LAST_ACK
  - receive ACK, send nothing
  - CLOSED
  - server application creates a listen socket
  - LISTEN
  - receive SYN, send SYN & ACK
  - CLOSE_WAIT
  - send FIN
  - SYN_RCVD
  - receive ACK, send nothing
  - ESTABLISHED
  - receive FIN, send ACK
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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
- no retransmission

- large delays when congested
- maximum achievable throughput
Causes/costs of congestion: scenario 2

- one router, finite buffers
- sender retransmission of lost packet

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

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

finite shared output link buffers
**Causes/costs of congestion: scenario 2**

- **always:** \( \lambda_{in} = \frac{\lambda}{\lambda_{out}} \) (goodput)
- "perfect" retransmission only when loss:
- retransmission of delayed (not lost) packet makes same \( \lambda_{in} > \lambda_{out} \)

### “costs” of congestion:

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

- four senders
- multihop paths
- timeout/retransmit

Q: what happens as \( \lambda \) and \( \lambda' \) increase in finite shared output link buffers?
Causes/costs of congestion: scenario 3

Another “cost” of congestion:

- when packet dropped, any “upstream transmission capacity used for that packet was wasted!”
Approaches to end's congestion control

Two broad approaches to end's 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 sender should 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
  - Sender's send rate thus minimum supportable rate on path

- **EFCI bit in data cells: set to 1 in congested switch**
  - If data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell
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**TCP Congestion Control**

- end-end control (no network assistance)
- sender limits transmission:  
  \[
  \text{LastByteSent} - \text{LastByteAcked} \leq \text{CongWin}
  \]
- Roughly,

\[
\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}
\]

- CongWin is dynamic, function of perceived network congestion

**How does sender perceive congestion?**

- loss event = timeout or 3 duplicate acks
- TCP sender reduces rate (CongWin) after loss event

**Three mechanisms:**

- AIMD
- slow start
- conservative after timeout events
**TCP AIMD**

**Multiplicative Decrease:**
- Cut CongWin in half after loss event

**Additive Increase:**
- Increase CongWin by 1 MSS every RTT in the absence of loss events: probing

*Long-lived TCP connection*
**TCP Slow Start**

- *When connection begins, CongWin = 1 MSS*
  - Example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps

- *available bandwidth may be >> MSS/RTT*
  - desirable to quickly ramp up to respectable rate

- *When connection begins, increase rate exponentially fast until first loss event*
TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
  - double \textcolor{blue}{\text{CongWin}} every RTT
  - done by incrementing \textcolor{blue}{\text{CongWin}} for every ACK received

- \textbf{Summary: initial rate is slow but ramps up exponentially fast}
Refinement

- After 3 dup ACKs:
  - CongWin is cut in half
  - window then grows linearly

- But after timeout event:
  - CongWin instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

Philosophy:
- 3 dup ACKs indicates network capable of delivering some segments
- timeout before 3 dup ACKs is “more alarming”
**Refinement (more)**

**Q:** When should the exponential increase switch to linear?

**A:** When $CongWin$ gets to $1/2$ of its value before timeout.

**Implementation:**
- Variable Threshold
- At loss event, Threshold is set to $1/2$ of $CongWin$ just before loss event
Summary: TCP Congestion Control

- When CongWin is below Threshold, sender is in slow-start phase, window grows exponentially.

- When CongWin is above Threshold, sender is in congestion-avoidance phase, window grows linearly.

- When a triple duplicate ACK occurs, Threshold set to CongWin/2 and CongWin set to Threshold.

- When timeout occurs, Threshold set to CongWin/2 and CongWin is set to 1MSS.
### TCP Sender Congestion Control

<table>
<thead>
<tr>
<th>State</th>
<th>Event</th>
<th>TCP Sender Action</th>
<th>Commentary</th>
</tr>
</thead>
<tbody>
<tr>
<td>Slow Start (SS)</td>
<td>ACK receipt for previously unacked data</td>
<td>CongWin = CongWin + MSS, If (CongWin &gt; Threshold) set state to “Congestion Avoidance”</td>
<td>Resulting in a doubling of CongWin every RTT</td>
</tr>
<tr>
<td>Congestion Avoidance (CA)</td>
<td>ACK receipt for previously unacked data</td>
<td>CongWin = CongWin+MSS * (MSS/CongWin)</td>
<td>Additive increase, resulting in increase of CongWin by 1 MSS every RTT</td>
</tr>
<tr>
<td>SS or CA</td>
<td>Loss event detected by triple duplicate ACK</td>
<td>Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance”</td>
<td>Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.</td>
</tr>
<tr>
<td>SS or CA</td>
<td>Timeout</td>
<td>Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start”</td>
<td>Enter slow start</td>
</tr>
<tr>
<td>SS or CA</td>
<td>Duplicate ACK</td>
<td>Increment duplicate ACK count for segment being acked</td>
<td>CongWin and Threshold not changed</td>
</tr>
</tbody>
</table>
**TCP throughput**

- What's the average throughput of TCP as a function of window size and RTT?
  - Ignore slow start
- Let \( W \) be the window size when loss occurs.
- When window is \( W \), throughput is \( W / \text{RTT} \)
- Just after loss, window drops to \( W/2 \), throughput to \( W /2 \text{RTT} \).
- Average throughput: \( .75 W / \text{RTT} \)
TCP Futures

- **Example:** 1500 byte segments, 100 ms RTT, want 10 Gbps throughput
- Requires window size $W = 83,333$ in-flight segments
- Throughput in terms of loss rate:

\[
\frac{1.22 \cdot MSS}{RTT\sqrt{L}}
\]

- $L = 2 \times 10^{10} W \omega$
- New versions of TCP for high-speed needed!
**TCP Fairness**

**Fairness goal:** if $K$ TCP sessions share the same bottleneck link of bandwidth $R$, each should have an average rate of $R/K$. 

![Diagram](attachment:image.png)
Why is TCP fair?

Two competing sessions:
- Additive increase gives slope of 1, as throughput increases
- Multiplicative decrease decreases throughput proportionally

![Diagram showing equal bandwidth share and additive increase and multiplicative decrease in throughput for two competing sessions.](image_url)
**Fairness (more)**

**Fairness and UDP**
- Multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- Instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

**Fairness and parallel TCP connections**
- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: link of rate $R$ supporting 9 connections;
  - new app asks for 1 TCP, gets rate $R/10$
  - new app asks for 11 TCPs, gets $R/2$!
**Delay modeling**

Q: How long does it take to receive an object from a Web server after sending a request?

**Ignoring congestion, delay is influenced by:**
- TCP connection establishment
- data transmission delay
- slow start

**Notation, assumptions:**
- Assume one link between client and server of rate $R$
- $S$: MSS (bits)
- $O$: object size (bits)
- no retransmissions (no loss, no corruption)

**Window size:**
- First assume: fixed congestion window, $W$ segments
- Then dynamic window, modeling slow start
**Fixed congestion window (1)**

*First case:*

\[ W \frac{S}{R} > RTT + \frac{S}{R} : \text{ACK for first segment in window returns before window's worth of data sent} \]

\[ \text{delay} = 2RTT + O/R \]
Second case:


\[
\text{delay} = 2RTT + O/R + (K-1)[S/R + RTT - W S/R]
\]
TCP Delay Modeling: Slow Start (1)

Now suppose window grows according to slow start

We will show that the delay for one object is:

$$\text{Latency} = 2\text{RTT} + \frac{O}{R} + P \left[ \text{RTT} + \frac{S}{R} \right] - (2^P - 1) \frac{S}{R}$$

where $P$ is the number of times TCP idles at server:

$$P = \min\{Q, K - 1\}$$

- where $Q$ is the number of times the server idles if the object were of infinite size.

- and $K$ is the number of windows that cover the object.
TCP Delay Modeling: Slow Start (2)

Delay components:
- 2 RTT for connection estab and request
- O/R to transmit object
- time server idles due to slow start

Server idles:
\[ P = \min\{K - 1, Q\} \]

Example:
- O/S = 15 segments
- \( K = 4 \) windows
- \( Q = 2 \)
- \( P = \min\{K - 1, Q\} = 2 \)

Server idles \( P = 2 \) times
**TCP Delay Modeling (3)**

\[
\frac{S}{R} + RTT = \text{time from when server starts to send segment until server receives acknowledgment}
\]

\[
2^{k-1} \frac{S}{R} = \text{time to transmit the } k\text{th window}
\]

\[
\left[ \frac{S}{R} + RTT - 2^{k-1} \frac{S}{R} \right]^+ = \text{idle time after the } k\text{th window}
\]

\[
\text{delay} = \frac{O}{R} + 2RTT + \sum_{p=1}^{P} \text{idleTime}
\]

\[
= \frac{O}{R} + 2RTT + \sum_{k=1}^{P} \left[ \frac{S}{R} + RTT - 2^{k-1} \frac{S}{R} \right]
\]

\[
= \frac{O}{R} + 2RTT + P[RTT + \frac{S}{R}] - (2^P - 1) \frac{S}{R}
\]
TCP Delay Modeling (4)

Recall \( K \) = number of windows that cover object

How do we calculate \( K \)?

\[
K = \min\{k : 2^0S + 2^1S + \cdots + 2^{k-1}S \geq O\} \\
= \min\{k : 2^0 + 2^1 + \cdots + 2^{k-1} \geq O / S\} \\
= \min\{k : 2^k - 1 \geq \frac{O}{S}\} \\
= \min\{k : k \geq \log_2\left(\frac{O}{S} + 1\right)\} \\
= \left\lceil \log_2\left(\frac{O}{S} + 1\right) \right\rceil
\]

Calculation of \( Q \), number of idles for infinite-size object, is similar (see HW).
HTTP Modeling

- Assume Web page consists of:
  - 1 base HTML page (of size 0 bits)
  - M images (each of size 0 bits)

- Non-persistent HTTP:
  - \( M + 1 \) TCP connections in series
  - Response time = \((M + 1)\frac{P}{R} + (M + 1)2RTT + \text{sum of idle times}\)

- Persistent HTTP:
  - 2 RTT to request and receive base HTML file
  - 1 RTT to request and receive M images
  - Response time = \((M + 1)\frac{P}{R} + 3RTT + \text{sum of idle times}\)

- Non-persistent HTTP with X parallel connections
  - Suppose \( \frac{M}{X} \) integer.
  - 1 TCP connection for base file
  - \( \frac{M}{X} \) sets of parallel connections for images.
  - Response time = \((M + 1)\frac{P}{R} + (M/X + 1)2RTT + \text{sum of idle times}\)
HTTP Response time (in seconds)

\[ RTT = 100 \text{ m sec}, O = 5 \text{ K bytes}, M = 10 \text{ and } X = 5 \]

For low bandwidth, connection & response time dominated by transmission time.

Persistent connections only give minor improvement over parallel connections.
HTTP Response time (in seconds)

RTT = 1 sec, O = 5 Kbytes, M = 10 and X = 5

For larger RTT, response time dominated by TCP establishment & slow start delays. Persistent connections now give important improvement, particularly in high delay bandwidth networks.
Chapter 3: Summary

- Principles behind transport layer services:
  - Multiplexing, demultiplexing
  - Reliable data transfer
  - Flow control
  - Congestion control

- Instantiation and implementation in the Internet
  - UDP
  - TCP

Next:
- Leaving the network “edge” (application, transport layers)
- Into the network “core”