CS2105

An Awesome Introduction to Computer Networks

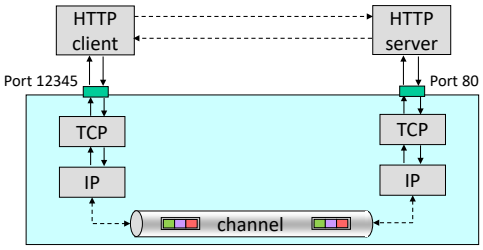
Lectures 4&5: The Transport Layer

NUS National University of Singapore | Department of Computer Science, School of Computing

Web and HTTP

PREVIOUS LECTURE

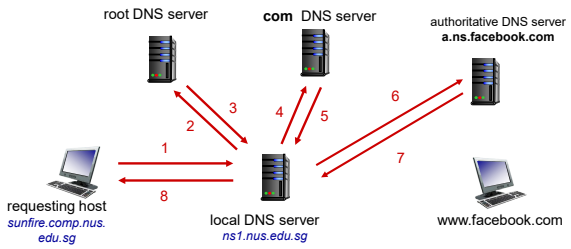
- A Web page consists of a *base HTML file* and *some other objects* referenced by the HTML file.
- HTTP uses TCP as transport service.
  - TCP, in turn, uses service provided by IP!



Domain Name System

PREVIOUS LECTURE

- DNS is the Internet's primary directory service.
  - It translates host names, which can be easily memorized by humans, to numerical IP addresses used by hosts for the purpose of communication.



Socket

PREVIOUS LECTURE

- Applications (processes) send messages over the network through sockets.
  - Conceptually, socket = IP address + port number
  - Programming wise, socket = a set of APIs
- UDP socket
  - Server uses one socket to serve all clients.
  - No connection is established before sending data.
  - Sender explicitly attaches destination IP address + port #.
- TCP socket
  - Server creates a new socket for each client.
  - Client establishes connection to server.
  - Server uses connection to identify client.

Lectures 4&5: The Transport Layer

After this class, you are expected to:

- appreciate the simplicity of UDP and the service it provides.
- know how to calculate the checksum of a packet.
- be able to design your own reliable protocols with *ACK*, *NAK*, *sequence number*, *timeout* and *retransmission*.
- understand the working of *Go-Back-N* and *Selective Repeat* protocols.
- understand the operations of TCP.

Lectures 4&5: Roadmap

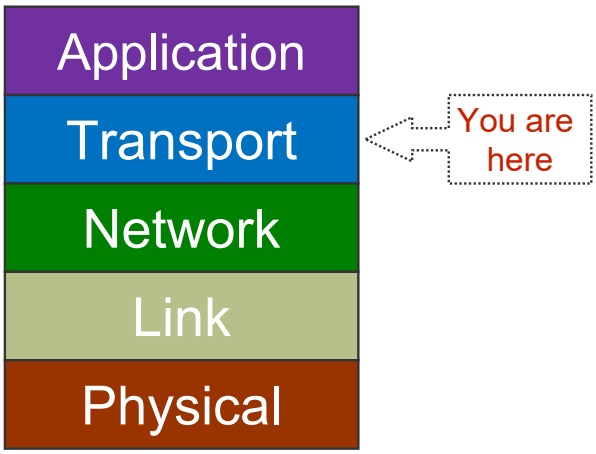
3.1 Transport-layer Services

3.3 Connectionless Transport: UDP

3.4 Principles of Reliable Data Transfer

3.5 Connection-oriented Transport: TCP

Kurose Textbook, Chapter 3
(Some slides are taken from the book)

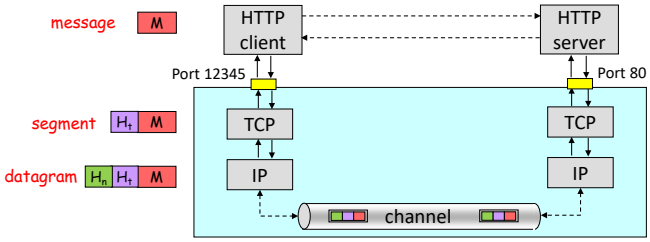


Transport Layer Services

- Deliver messages between application processes running on different hosts
  - Two popular protocols: TCP and UDP
- Transport layer protocols run in hosts.
  - Sender side: breaks app message into *segments* (as needed), passes them to network layer (aka IP layer).
  - Receiver side: reassembles segments into message, passes it to app layer.
  - Packet switches (routers) in between: only check destination IP address to decide routing.

Transport / Network Layers

- Each IP datagram contains source and dest IP addresses.
  - Receiving host is identified by dest IP address.
  - Each IP datagram carries one transport-layer segment.
  - Each segment contains source and dest port numbers.



Lectures 4&5: Roadmap

3.1 Transport-layer Services

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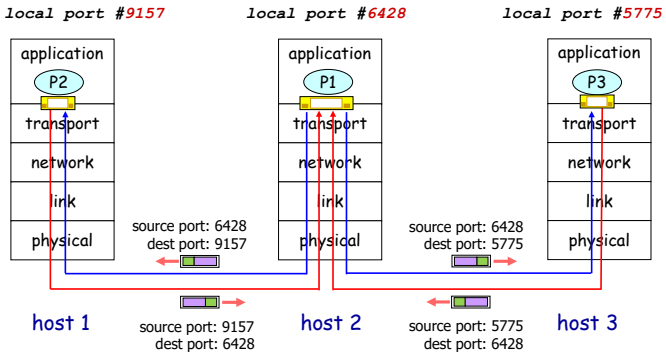
UDP: User Datagram Protocol [RFC 768]

- UDP adds very little service on top of IP:
  - Connectionless multiplexing / de-multiplexing
  - Checksum
- UDP transmission is unreliable
  - Often used by streaming multimedia apps (loss tolerant & rate sensitive)
- To achieve reliable transmission over UDP
  - Application implements error detection and recovery mechanisms!

Connectionless De-multiplexing

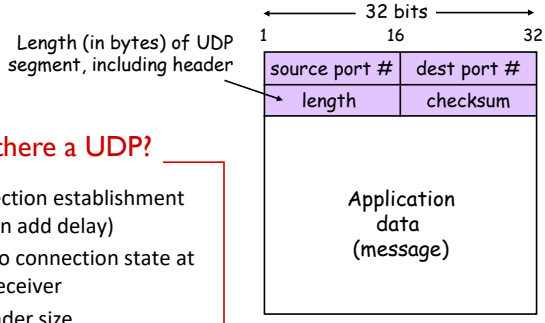
- UDP sender:
  - Creates a socket with local port #.
  - When creating a datagram to send to UDP socket, sender must specify dest. IP address and port #.
- When UDP receiver receives a UDP segment:
  - Checks destination port # in segment.
  - Directs UDP segment to the socket with that port #.
  - IP datagrams (from different sources) with the same destination port # will be directed to the same UDP socket at destination.

Connectionless De-multiplexing



IP datagrams (from different sources) with the same destination port # will be directed to the same UDP socket at destination.

UDP Header



UDP segment format

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: to detect "errors" (i.e., flipped bits) in transmitted segment.

Sender:

- compute checksum value (next page)
- put 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 really no error?)

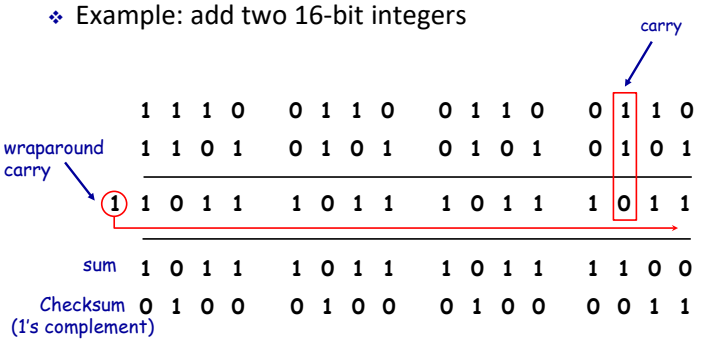
Checksum Computation

- How is UDP checksum computed?
  1. Treat UDP segment as a sequence of 16-bit integers.
  2. Apply binary addition on every 16-bit integer (checksum field is currently 0).
  3. Carry (if any) from the most significant bit will be added to the result.
  4. Compute 1's complement to get UDP checksum.

x	y	x ⊕ y	carry
0	0	0	-
0	1	1	-
1	0	1	-
1	1	0	1

Checksum Example

- Example: add two 16-bit integers



Lectures 4&5: Roadmap

3.1 Transport-layer Services

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3.4 Principles of Reliable Data Transfer

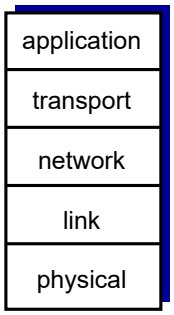
3.5 Connection-oriented transport: TCP

"Sending Data Reliably Over the Internet is Much Harder Than You Think. The Intricacy Involved in Ensuring Reliability Will Make Your Head Explode."

Transport vs. Network Layer

- **Transport layer** resides on end hosts and provides process-to-process communication.
- **Network layer** provides host-to-host, best-effort and unreliable communication.

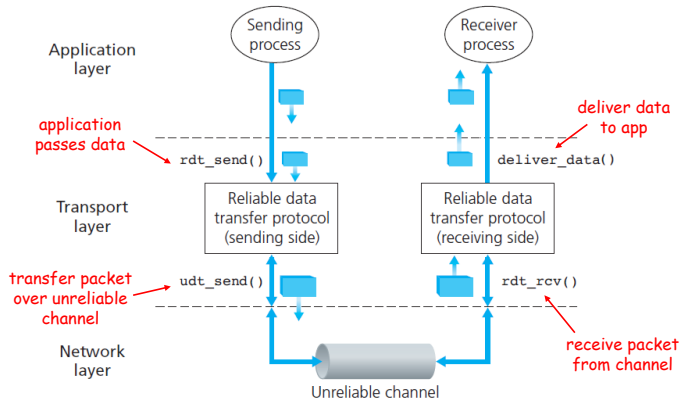
Question: How to build a **reliable transport layer protocol** on top of **unreliable communication**?



Reliable Transfer over Unreliable Channel

- Underlying network may
  - corrupt packets
  - drop packets
  - re-order packets (not considered in this lecture)
  - deliver packets after an arbitrarily long delay
- End-to-end reliable transport service should
  - guarantee packets delivery and correctness
  - deliver packets (to application) in the same order they are sent

Reliable Data Transfer: Service Model

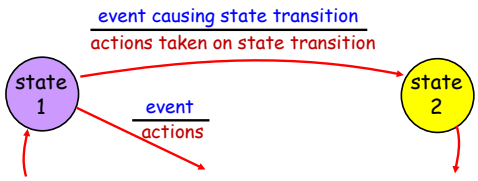


Reliable Data Transfer Protocols

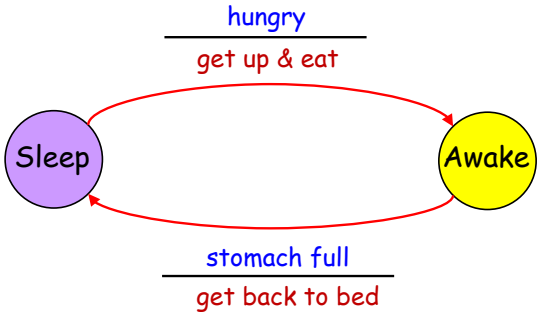
- Characteristics of unreliable channel will determine the complexity of reliable data transfer protocols (rdt).
- We will incrementally develop sender & receiver sides of rdt protocols, considering increasingly complex models of unreliable channel.
- We consider only unidirectional data transfer
  - but control info may flow in reverse direction!

Finite State Machine (FSM)

- We will use finite state machines (FSM) to describe sender and receiver of a protocol.
  - We will learn a protocol by examples, but FSM provides you the complete picture to refer to as necessary.

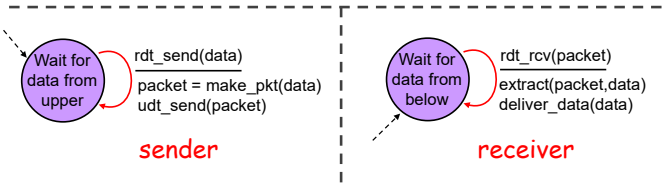


Example FSM



rdt 1.0: Perfectly Reliable Channel

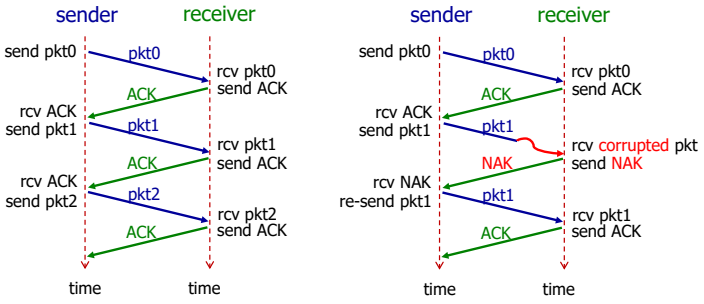
- Assume underlying channel is perfectly reliable.
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying (perfect) channel
  - Receiver reads data from underlying (perfect) channel



rdt 2.0: Channel with *Bit Errors*

- Assumption:
  - underlying channel may flip bits in packets
  - other than that, the channel is perfect
- Receiver may use ***checksum*** to detect bit errors.
- Question: how to recover from bit errors?
  - *Acknowledgements (ACKs):* receiver explicitly tells sender that packet received is OK.
  - *Negative acknowledgements (NAKs):* receiver explicitly tells sender that packet has errors.
  - Sender retransmits packet on receipt of NAK.

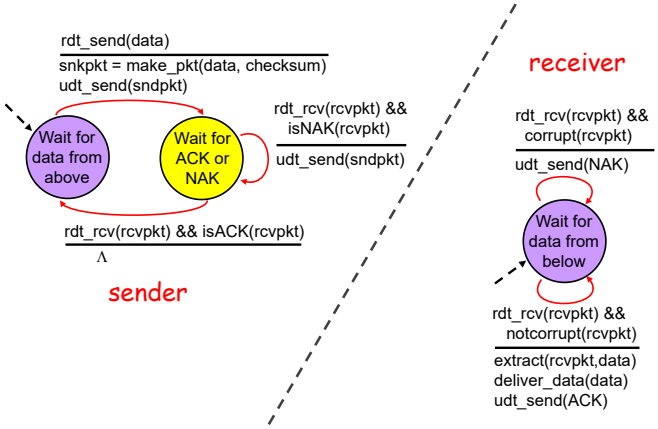
rdt 2.0 In Action



stop and wait protocol

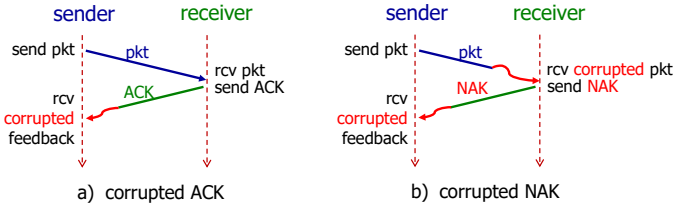
Sender sends one packet at a time, then waits for receiver response

rdt 2.0: FSM



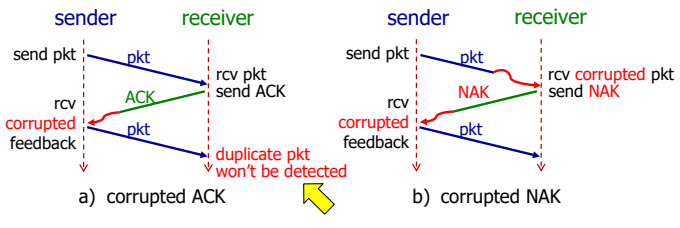
rdt 2.0 has a Fatal Flaw!

- What happens if ACK/NAK is corrupted?
  - Sender doesn't know what happened at receiver!
- So what should the sender do?
  - Sender just retransmits when receives garbled ACK or NAK.
  - Questions: does this work?



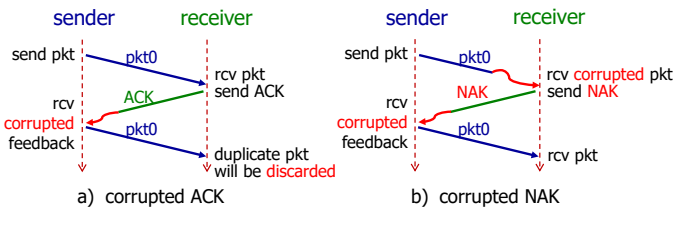
rdt 2.0 has a Fatal Flaw!

- Sender just retransmits when it receives garbled feedback.
  - This may cause retransmission of correctly received packet!
  - Question: how can receiver identify duplicate packet?

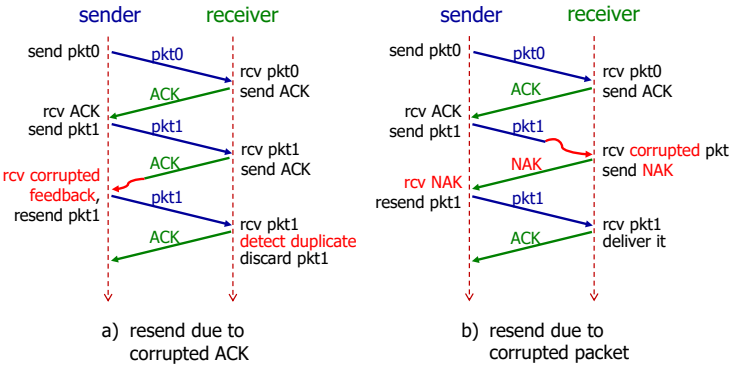


rdt 2.1: rdt 2.0 + Packet Seq.

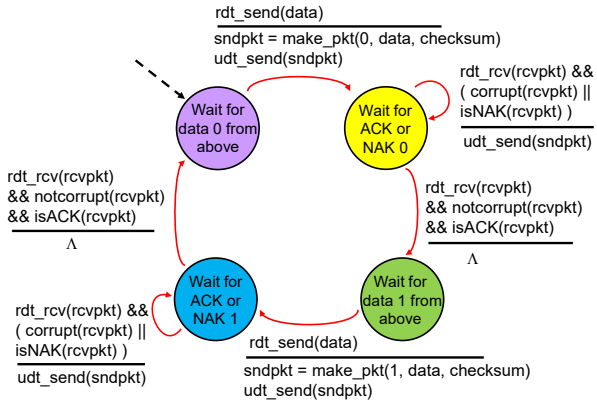
- To handle duplicates:
  - Sender retransmits current packet if ACK/NAK is garbled.
  - Sender adds *sequence number* to each packet.
  - Receiver discards (doesn't deliver up) duplicate packet.
- This gives rise to protocol rdt 2.1.



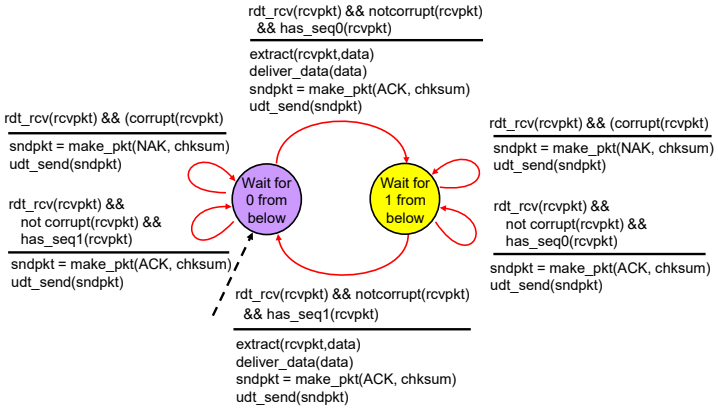
rdt 2.1 In Action



rdt 2.1 Sender FSM



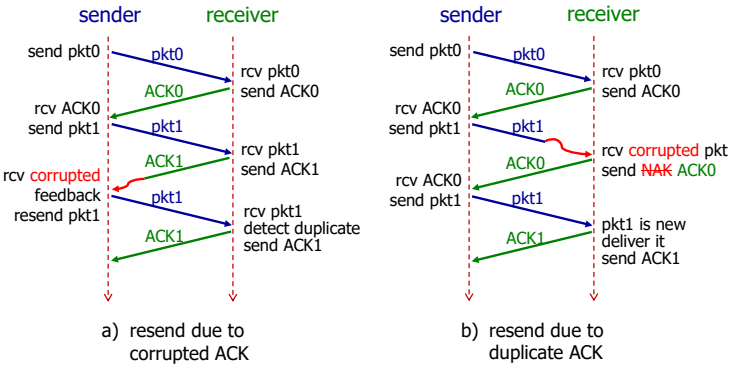
rdt 2.1 Receiver FSM



rdt 2.2: a NAK-free Protocol

- Same assumption and functionality as rdt 2.1, but use ACKs only.
- Instead of sending NAK, receiver sends ACK for the last packet received OK.
  - Now receiver must *explicitly* include seq. # of the packet being ACKed.
- Duplicate ACKs at sender results in same action as NAK: *retransmit current pkt*.

rdt 2.2 In Action



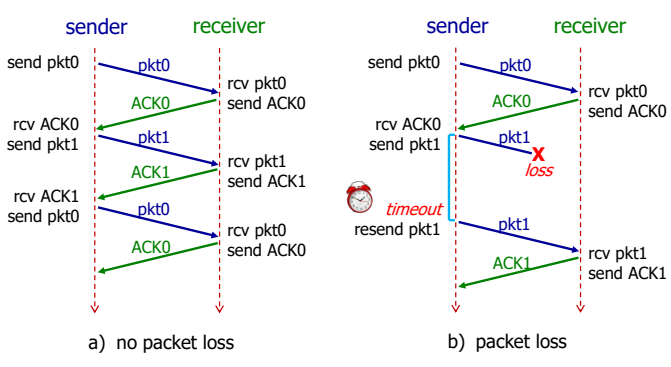
rdt 3.0: Channel with *Errors* and *Loss*

- Assumption: underlying channel
  - may flip bits in packets
  - may lose packets
  - may incur arbitrarily long packet delay
  - but won't re-order packets
- Question: how to detect packet loss?
  - checksum, ACKs, seq. #, retransmissions will be of help… but not enough

rdt 3.0: Channel with *Errors* and *Loss*

- To handle packet loss:
  - Sender waits "reasonable" amount of time for ACK.
  - Sender retransmits if no ACK is received till *timeout*.
- Question: what if packet (or ACK) is just delayed, but not lost?
  - Timeout will trigger retransmission.
  - Retransmission will generate duplicates in this case, but receiver may use seq. # to detect it.
  - Receiver must specify seq. # of the packet being ACKed (check scenario (d) two pages later).

rdt 3.0 In Action



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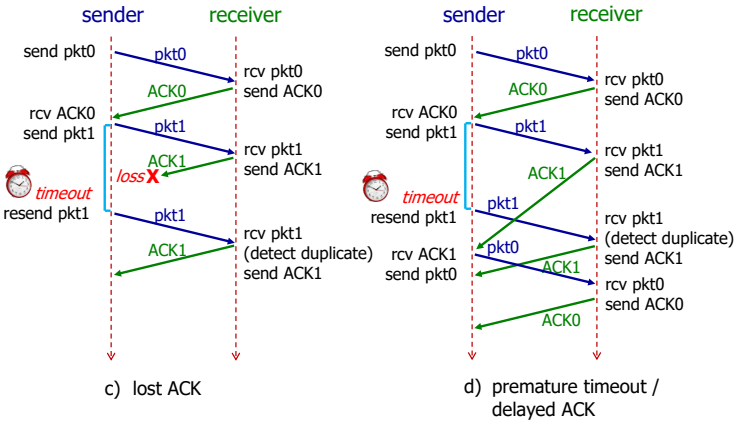
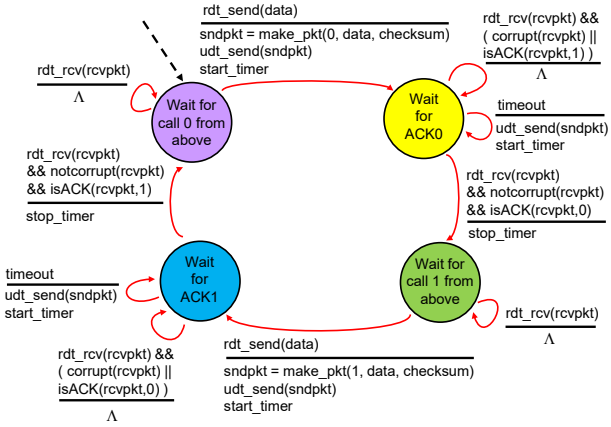


rdt 3.0 Sender FSM



Performance of rdt 3.0

- rdt 3.0 works, but performance stinks.
- Example: packet size = 8000 bits, link rate = 1 Gbps:

$$d_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 0.008 \text{ msec}$$

  - If RTT = 30 msec, sender sends 8000 bits every 30.008 msec.

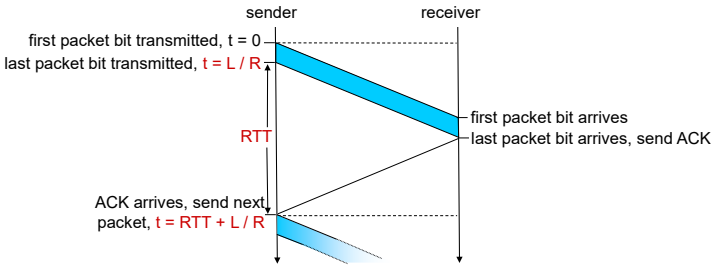
$$\text{throughput} = \frac{L}{RTT + d_{trans}} = \frac{8000}{30.008} = 267 \text{ kbps}$$

  - U_{sender}: *utilization* – fraction of time sender is busy sending

$$U_{sender} = \frac{d_{trans}}{RTT + d_{trans}} = \frac{0.008}{30 + 0.008} = 0.00027$$

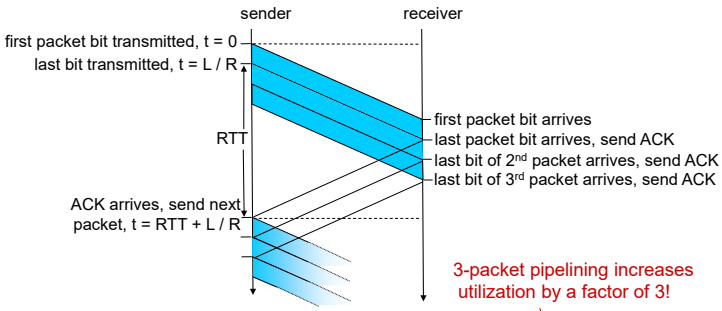
rdt 3.0: Stop-and-wait Operation

- Network protocol limits use of physical resources!



$$U_{sender} = \frac{L / R}{RTT + L/R} = \frac{0.008}{30 + 0.008} = 0.00027$$

Pipelining: Increased Utilization



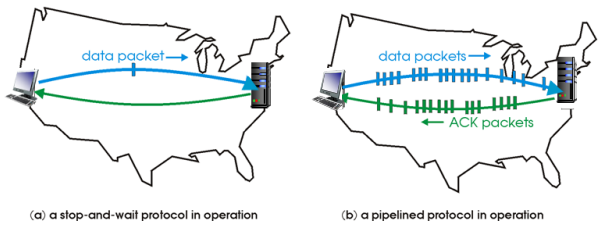
3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L/R} = \frac{0.024}{30 + 0.008} = 0.00081$$

Pipelined Protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged packets.

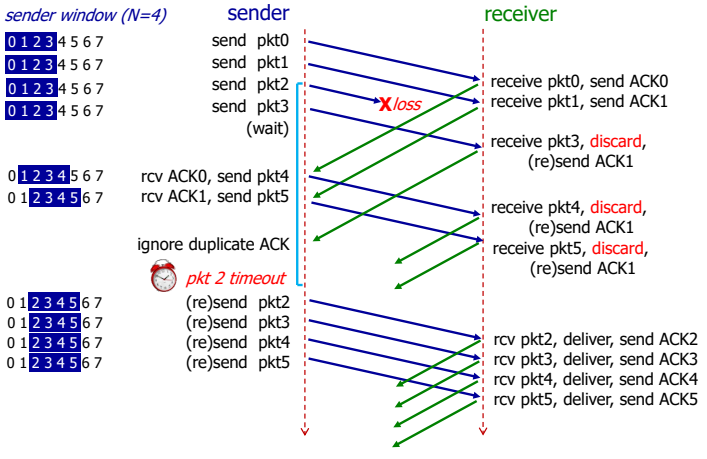
- range of sequence numbers must be increased
- buffering at sender and/or receiver



Benchmark Pipelined Protocols

- Two generic forms of pipelined protocols:
  - *Go-Back-N*
  - *Selective repeat*
- Assumption (same as rdt 3.0): underlying channel
  - may flip bits in packets
  - may lose packets
  - may incur arbitrarily long packet delay
  - but won't re-order packets

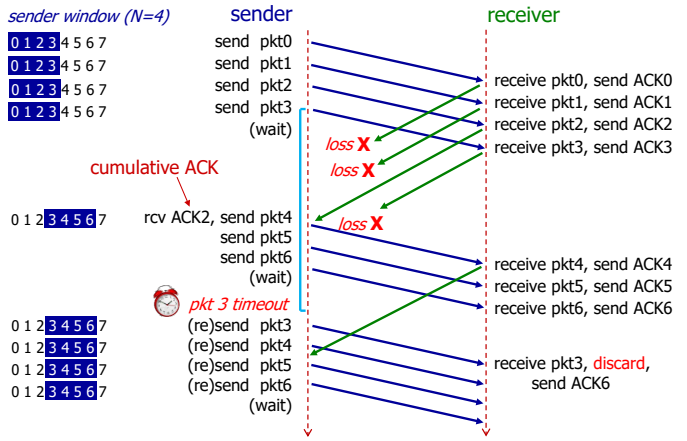
Go-back-N In Action



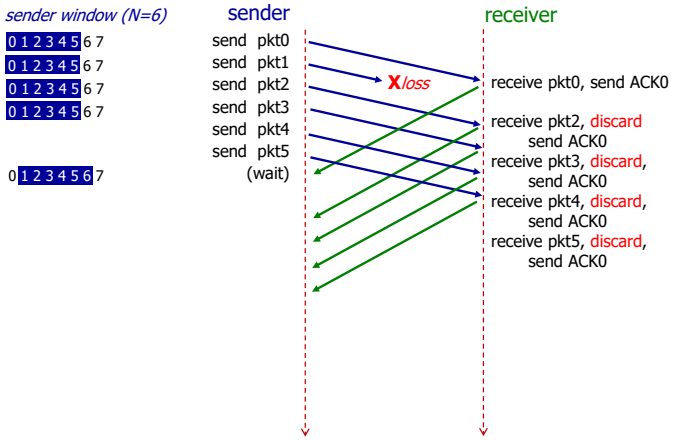
Go-back-N: Key Features

- GBN Sender
  - can have up to *N* unACKed packets in pipeline.
  - insert k-bits sequence number in packet header.
  - use a "sliding window" to keep track of unACKed packets.
  - keep a timer for the oldest unACKed packet.
  - *timeout(n)*: retransmit packet *n* and all subsequent packets in the window.
- GBN Receiver
  - only ACK packets that arrive in order.
    - simple receiver: need only remember `expectedSeqNum`
  - discard out-of-order packets and ACK the last in-order seq. #.
    - *Cumulative ACK*: "*ACK m*" means all packets up to *m* are received.

Go-back-N In Action



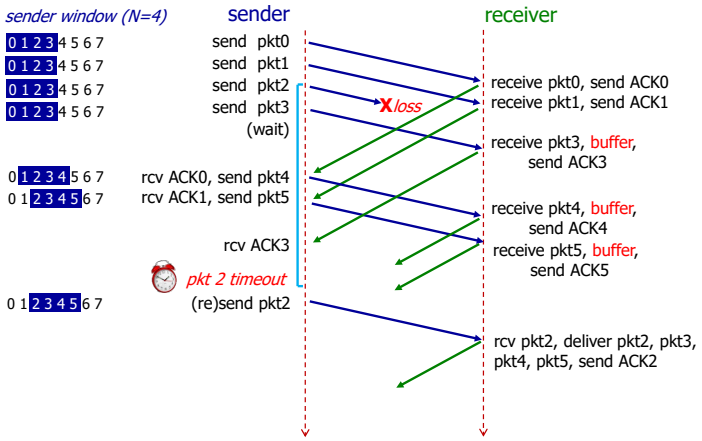
Go-back-N In Action



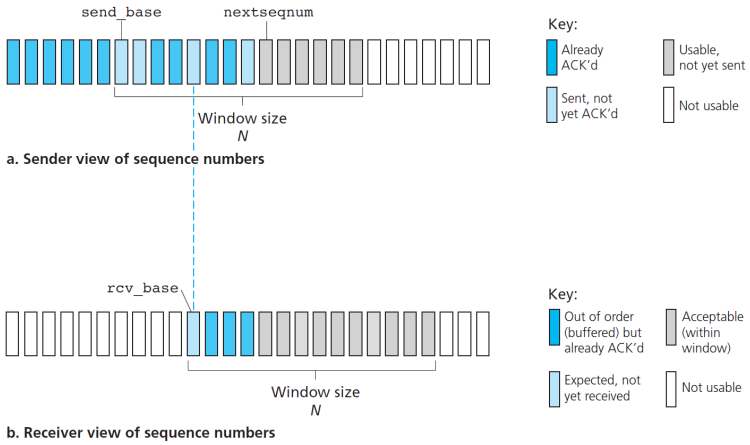
Selective Repeat: Key Features

- Receiver *individually* acknowledges all correctly received packets.
  - Buffers out-of-order packets, as needed, for eventual in-order delivery to upper layer.
- Sender maintains timer for each unACKed packet.
  - When timer expires, retransmit only that unACKed packet.

Selective Repeat In Action



SR Sender and Receiver Windows



Selective Repeat: Behaviors

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 is 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

Lectures 4&5: Roadmap

TCP: Transport Control Protocol

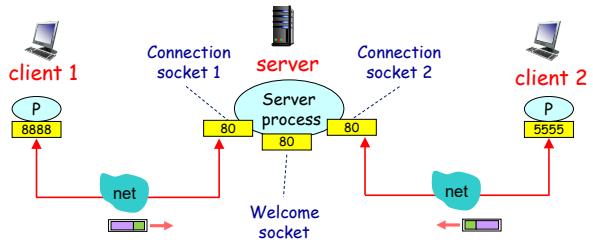
- In contrast to UDP, TCP is complex and is described in tens of RFCs, with new mechanisms or tweaks introduced throughout the years, resulting in many variants of TCP.
- We will only scratch the surface of TCP in CS2105.
  - More will be covered in CS3103.

TCP Overview

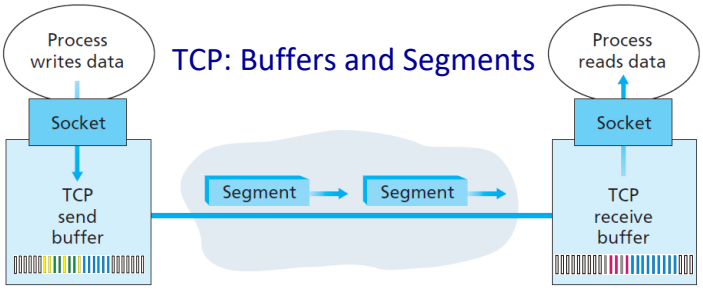
- Connection-oriented:
  - handshaking (exchange of control messages) before sending app data.
- Full duplex service:
  - bi-directional data flow in the same connection
- Reliable, in-order *byte steam:*
  - use sequence numbers to label bytes
- Flow control and congestion control
  - not in syllabus

Connection-oriented De-mux

- A TCP connection (socket) is identified by 4-tuple:
  - (srcIPAddr, srcPort, destIPAddr, destPort)
  - Receiver uses all four values to direct a segment to the appropriate socket.

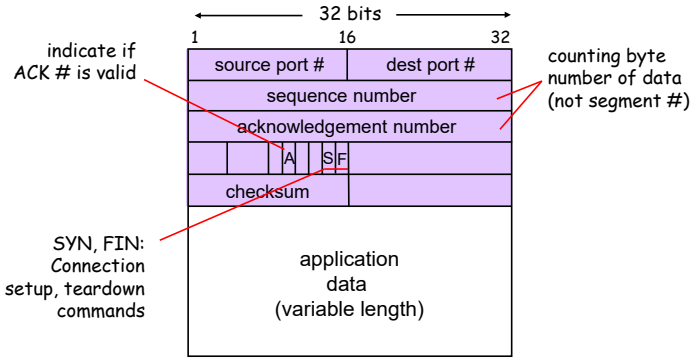


TCP: Buffers and Segments



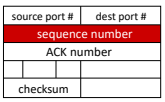
- TCP send and receive buffers
  - two buffers created after handshaking at any side.
- How much app-layer data a TCP segment can carry?
  - maximum segment size (MSS), typically 1,460 bytes
  - app passes data to TCP and TCP forms packets in view of MSS.

TCP Header

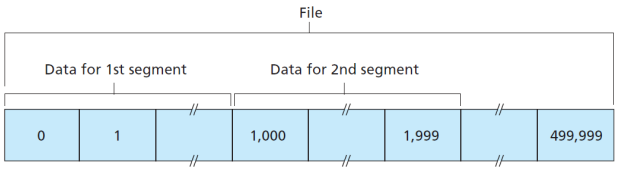


(some fields are not shown)

TCP Sequence Number



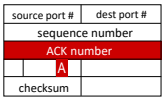
- "Byte number" of the first byte of data in a segment.
- Example: send a file of 500,000 bytes; MSS is 1,000 bytes.



Dividing file data into TCP segments

- Seq. # of 1st TCP segment: 0, 2nd TCP segment: 1,000, 3rd TCP segment: 2,000, 4th TCP segment: 3,000, etc.

TCP ACK Number



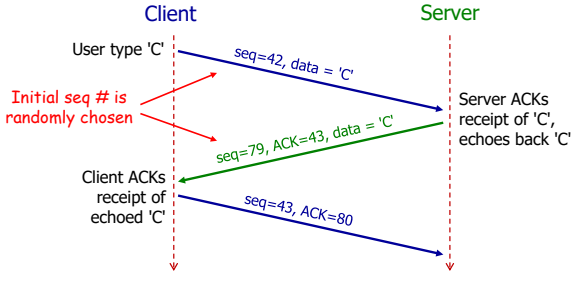
- Seq # of the next byte of data expected by receiver.

Sequence number of a segment	Amount of data carried	Corresponding ACK number
0	1,000	1,000
1,000	1,000	2,000
2,000	1,000	3,000
3,000	1,000	4,000
...	...	...

- TCP ACKs up to the first missing byte in the stream (cumulative ACK).
  - Note: TCP spec doesn't say how receiver should handle out-of-order segments - it's up to implementer.

Example: TCP Echo Server

- TCP (and also UDP) is a full duplex protocol
  - bi-directional data flow in the same TCP connection.
- Example:



TCP Echo Server

TCP Sender Events (simplified)

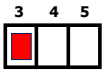
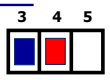
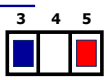
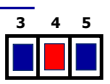
```
NextSeqNum=InitialSeqNumber
SendBase=InitialSeqNumber

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)
        break;

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

    event: ACK received, with ACK field value of y
        if (y > SendBase) {
            SendBase=y
            if (there are currently any not-yet-acknowledged segments)
                start timer
            }
        break;

} /* end of loop forever */
```

Sender keeps one timer only

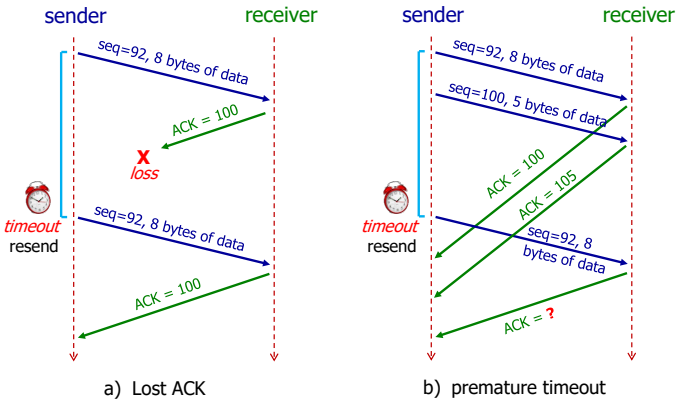
Retransmit only oldest unACKed packet

Cumulative ACK

TCP ACK Generation [RFC 5681]

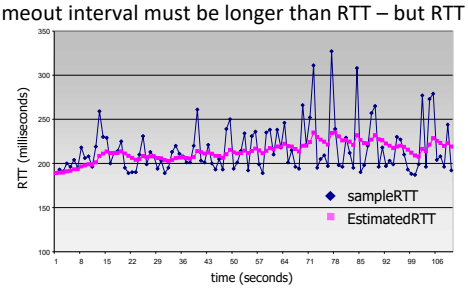
Event at TCP receiver	*TCP receiver action*	
Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	Delayed ACK: wait up to 500ms for next segment. If no next segment, send ACK	
Arrival of in-order segment with expected seq #. One other segment has ACK pending	Immediately send single cumulative ACK, ACKing both in-order segments	
Arrival of out-of-order segment higher-than-expect seq. # (gap detected)	Immediately send *duplicate ACK*, indicating seq. # of next expected byte	
Arrival of segment that partially or completely fills gap	Immediately send ACK, provided that segment starts at lower end of gap	

TCP Timeout / Retransmission



TCP Timeout Value

- How does TCP set appropriate timeout value?
  - too short timeout: premature timeout and unnecessary retransmissions.
  - too long timeout: slow reaction to segment loss.
  - Timeout interval must be longer than RTT – but RTT varies!



TCP Timeout Value

- TCP computes (and keeps updating) timeout interval based on estimated RTT.

EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

(typical value of α : 0.125)

DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|

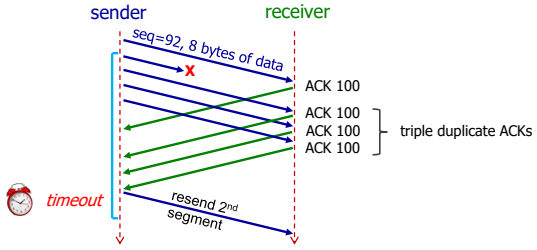
(typical value of β : 0.25)

TimeoutInterval = EstimatedRTT + 4*DevRTT

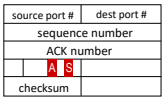
"safety margin"

TCP Fast Retransmission [RFC 2001]

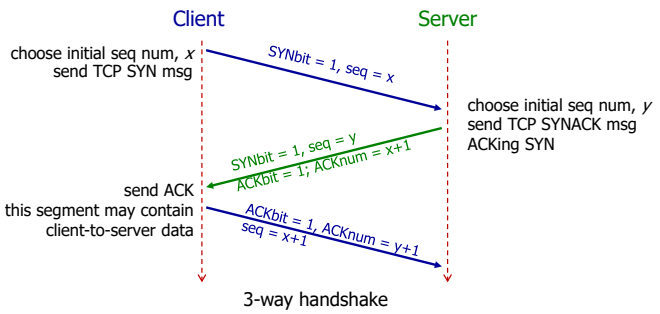
- Timeout period is often relatively long.
  - long delay before resending lost packet
- Fast retransmission:
  - **Event:** If sender receives 4 ACKs for the same segment, it supposes that segment is lost.
  - **Action:** resend segment (even before timer expires).



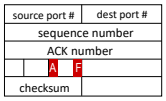
Establishing Connection



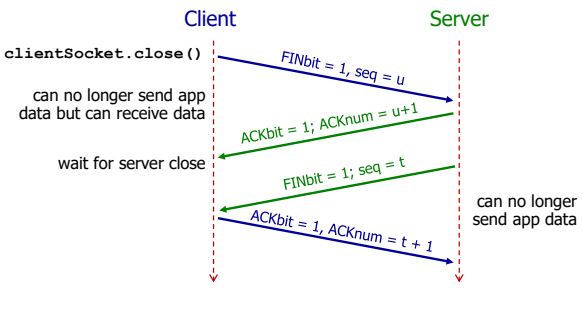
- Before exchanging data, TCP sender and receiver "shake hands".
  - Agree on connection and exchange connection parameters.



Closing Connection



- Client, server each close their side of connection.
  - send TCP segment with FIN bit = 1



What we did not cover….

- TCP flow control (Chapter 3.5.5)
  - Sender won't overflow receiver's buffer by sending too much or too fast.
  - Receiver feeds back to sender how many more bytes it is willing to accept.
- TCP congestion control (Chapter 3.6 & 3.7)
  - Be polite and send less if network is congested.
- They will be covered in the next course (CS3103)

Lectures 4&5: Summary

Go-back-N

- Sender can have up to *N* unACKed packets in pipeline
- Receiver only sends *cumulative ACKs*
  - Out-of-order packets discarded
- Sender sets timer for the oldest unACKed packet
  - when timer expires, retransmit ***all*** unACKed packets

Selective Repeat

- Sender can have up to *N* unACKed packets in pipeline
- Receiver sends *individual ACK* for each packet
  - Out-of-order packets buffered
- Sender maintains timer for ***each*** unACKed packet
  - when timer expires, retransmit only that unACKed packet

Lectures 4&5: Summary

- Connection-oriented transport: TCP
  - Segment structure
  - Reliable data transfer
  - Setting and updating retransmission time interval
  - 3-way handshake
  - Fast retransmission