

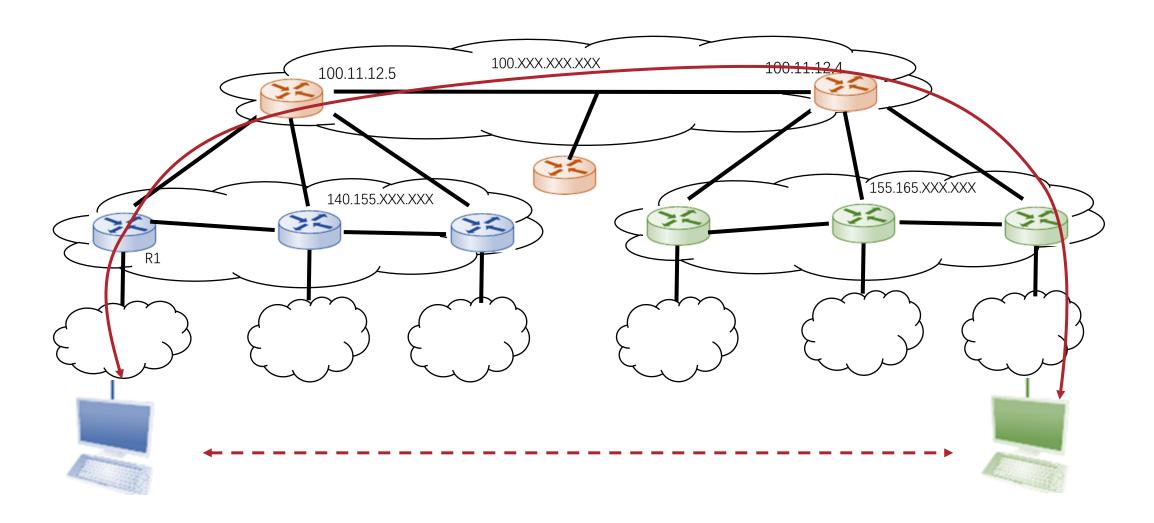
CS120: Computer Networks

Lecture 16. TCP 1

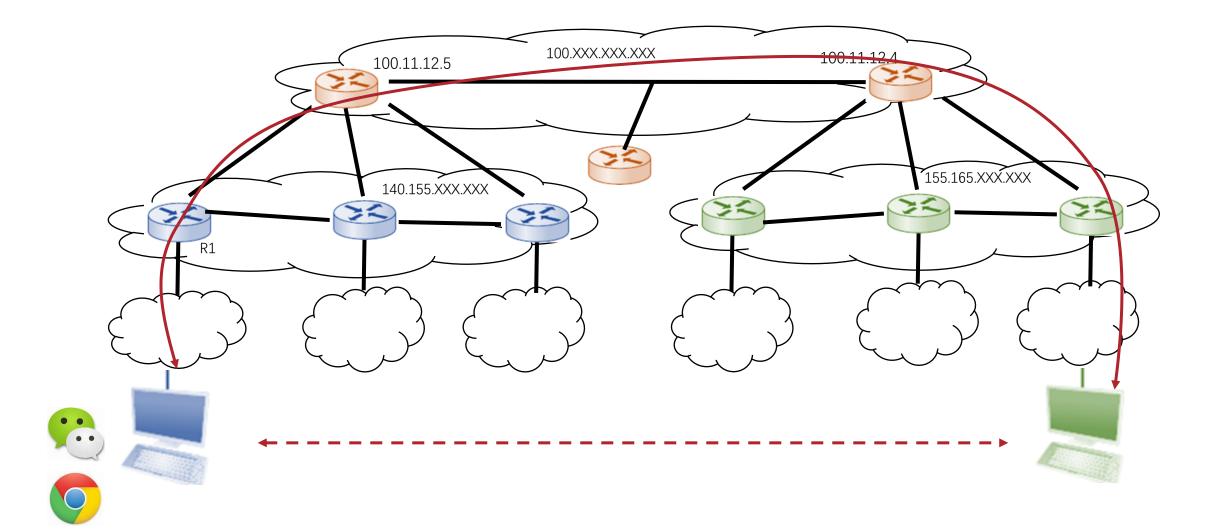
Haoxian Chen

Slides adopted from: Zhice Yang

IP: Host-to-host Protocol



Process-to-process Communication



Process-to-process Communication

 Problem: How to turn host-to-host packet delivery service into a process-to-process communication channel

Possible Application Level Requirements:

- Supports multiple application processes
- Reliable message delivery
- Messages are in order
- At most one copy
- Guaranteed delay
- Support arbitrarily large messages
- etc.



IP Layer Provides:

- Host to host communication service But:
- Messages may be dropped
- Messages may be reordered
- Messages may be duplicated
- Delivering delay is not guaranteed
- Message size is limited

Outline

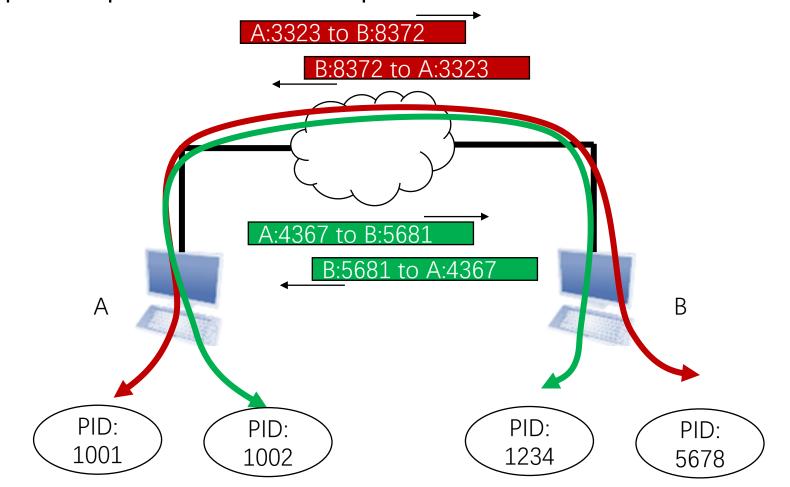
- Simple Demultiplexer (UDP)
- Reliable Byte Stream (TCP)

User Datagram Protocol (UDP)

• RFC 768

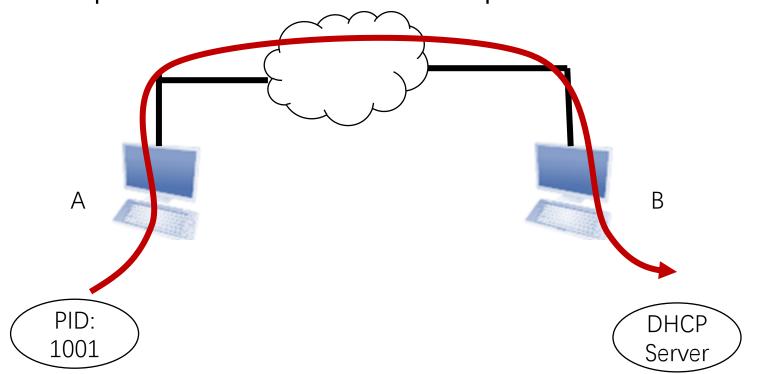
- Adds multiplexing support
 - Multiple process on the same host can share the same IP.
 - Each process is identified by port number.
- Direct Extension of IP
 - Best effort
 - Connection Less
 - No Guarantees

• The <IP, port> pair identifies a process in the network:



How does a process learn the port of the destination process?

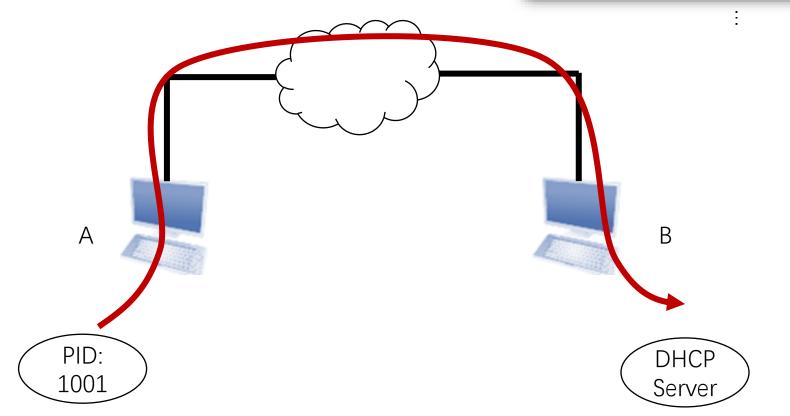
- 1. Server can accept messages at a well-known port.
- 2. Server then replies to client via the 'srcport' field.



- Initiate Connection: Default Port
 - stored in /etc/services

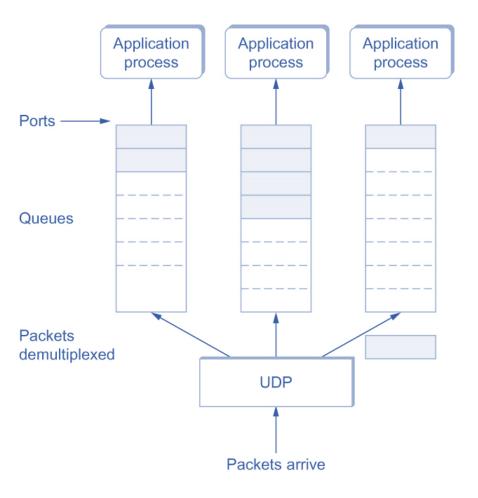
A:3323 to B:67

Port Number	Protocol	Function
21	TCP	FTP (File Transfer Protocol)
22	TCP/UDP	SSH (ssh,scp copy or sftp)
23	TCP/UDP	Telnet
25	TCP/UDP	SMTP (for sending outgoing emails)
43	TCP	WHOIS function
53	TCP/UDP	DNS Server (Domain name service for DNS requests)
67 68	UDP TCP	DHCP Server DHCP Client
70	TCP	Gopher Protocol
79	TCP	Finger protocol
110	TCP	POP3 (for receiving email)
119	TCP	NNTP (Network News Transfer Protocol)
143	TCP/UDP	IMAP4 Protocol (for email service)

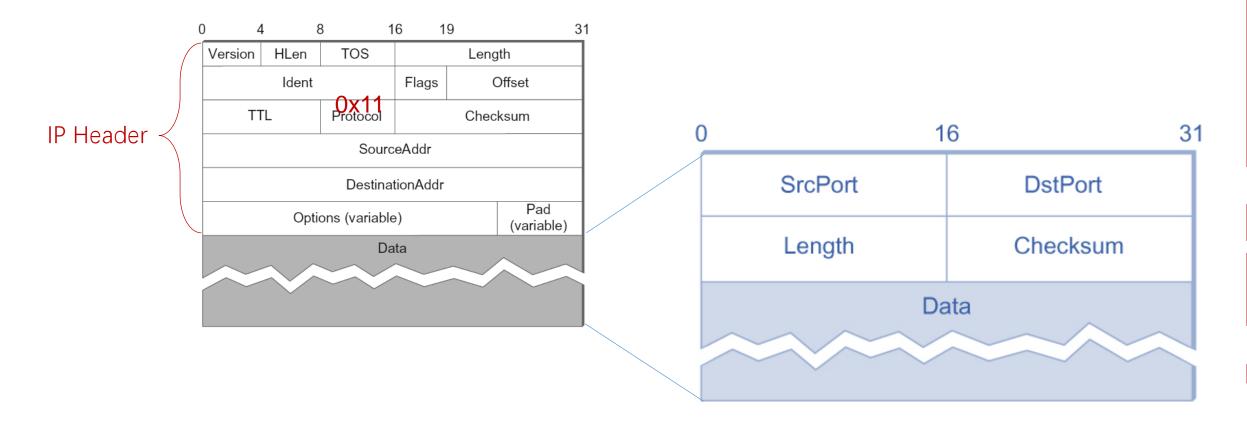


 Ports are implemented as message queues

• No flow control: messages are discarded when queue is full.

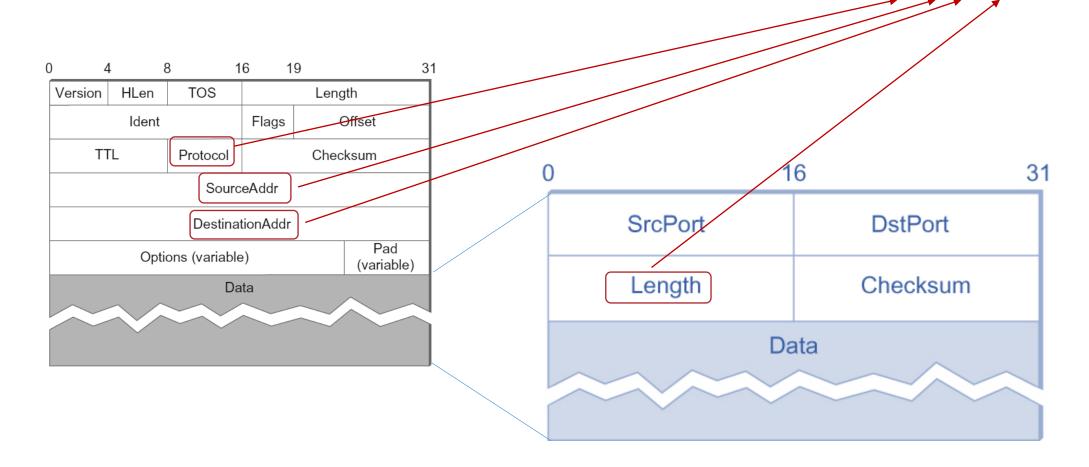


UDP Header



UDP Checksum

- UDP Checksum Range: UDP Header + UDP Data + Pseudoheader
 - Simple end-to-end integrity



Demo

netstat

User Datagram Protocol (UDP)

- RFC 768
- Direct Extension of IP
 - Best effort
 - Connection Less
 - No Guarantees
- Support Process Multiplexing

- UDP Use:
 - Loss tolerant, Rate sensitive
 - Video Stream
 - No Connection Setup delay, "One Time" Transfer
 - DNS
 - DHCP
 - Reliable Transfer over UDP
 - Add reliability at application layer, e.g., QUIC

Process-to-process Communication

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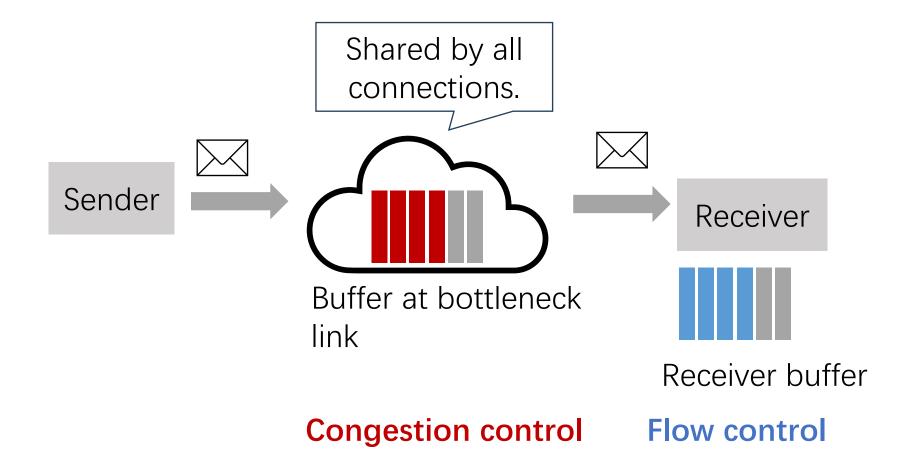
IP Layer Provides:

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Transmission Control Protocol (TCP)

- RFC: 793,1122,1323, 2018, 2581
- Goal: Reliable, In-order Delivery
 - Connection oriented
 - Reliable message delivery
 - Messages are in order
 - At most one copy
 - Flow control
 - Congestion control

Flow control vs. congestion control



What's the difference with sliding window in link layer?

1. Connection establishment

Link: always connects the **same** two computers.

TCP: Can connect any two computers on the internet.

→ Needs explicit connection establishment: like dial up a phone.

What's the difference with sliding window in link layer?

2. Adaptive timeout for retransmission

Link: has a **fixed** round-trip time (RTT)

TCP: RTT varies.

e.g., Shanghai - New York: 100 ms,

Two computers in the same room: 1ms

What's the difference with sliding window in link layer?

- 3. Packets arrive out of order, due to:
- Packet loss: congestion, time out
- Packets travel along different paths: network dynamic, traffic engineering

What's the difference with sliding window in link layer?

4. Receiver capacity varies

Link: the same host has the same capacity.

TCP: connects to any computers.

sender needs to learn the receiver's available buffer size.

What's the difference with sliding window in link layer?

5. Congestion control

Directly connected link: has **fixed** bandwidth and delay. only one sender.

TCP: network conditions are **unknown**.

share the network with many connections.

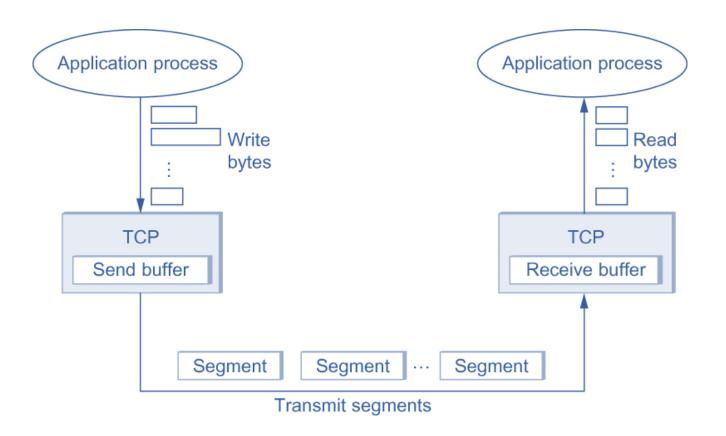
Needs congestion control mechanism.

Difference from Simple Sliding Window

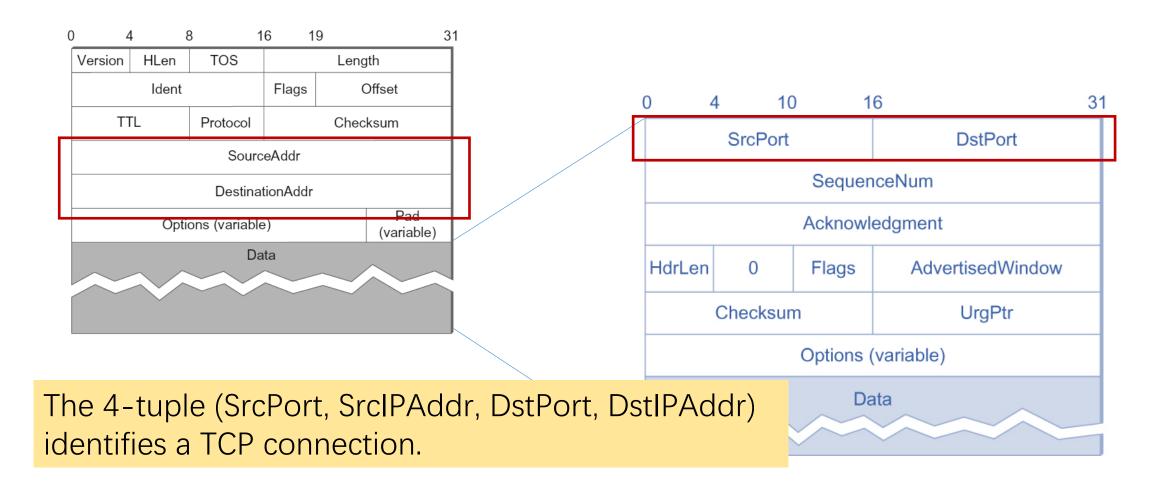
- Connection Establishment
 - Need to share connection parameters
- Adaptive Timeout
 - Need to handle dynamic RTT in IP network
- Timeout Packet
 - Need to distinguish old packets
- Flow Control
 - Need to know the receiver's capability
- Congestion Control
 - Need to estimate the network capacity

TCP: Communicate Model

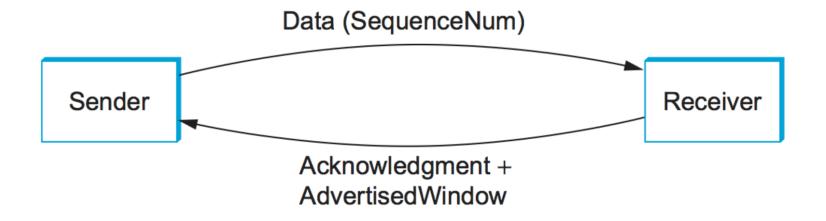
TCP Peers Communicate through Segments



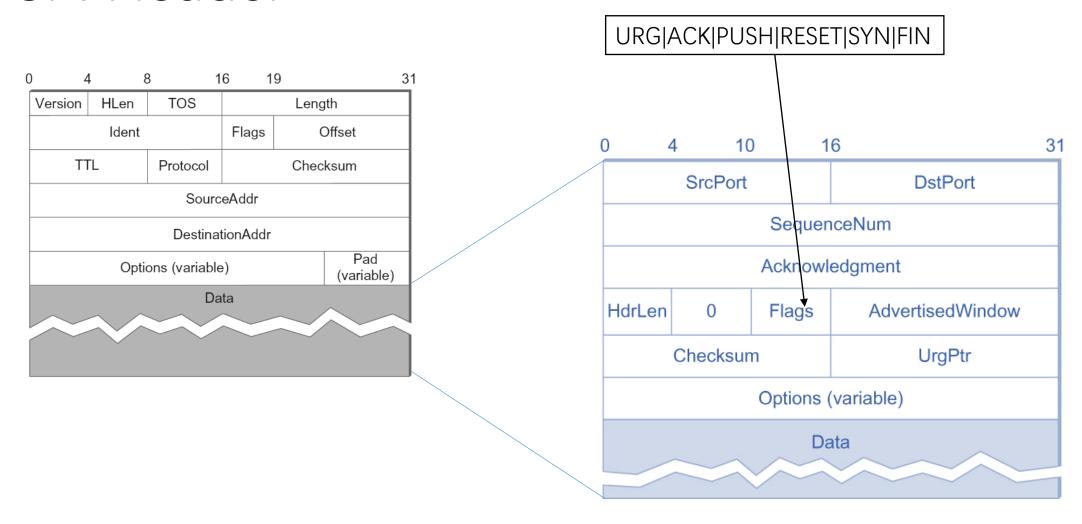
TCP: Header



TCP: SequenceNum and Acknowledgment

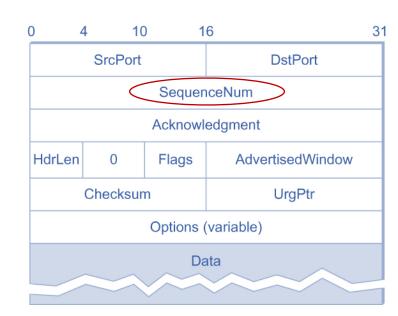


TCP: Header

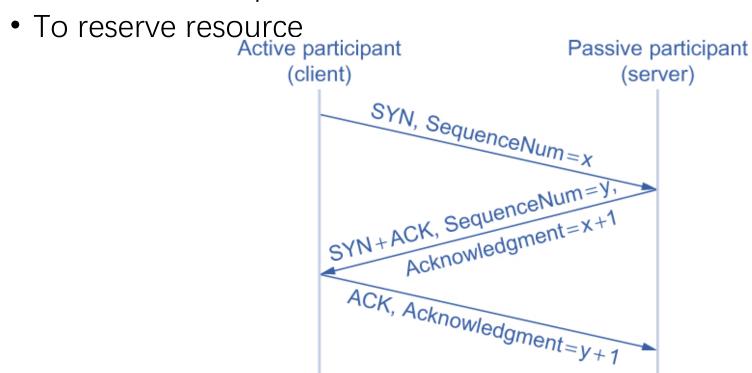


- Why?
 - Reserve Connection Resource (buffer, etc.)
 - Negotiate Sequence Number
 - Reject Out-of-time Connection Request

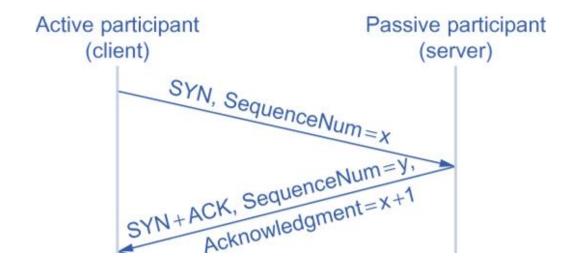
- Sequence Number
 - The pointer of the data byte in the segment
 - Initial sequence number is exchanged in connection establishment
 - Initial sequence number is a random number (32bits):
 - To avoid segments with same sequence number from dead connections
 - maximum segment lifetime: 120 seconds
 - Security concern
 - Sequence number prediction attack
- Acknowledgement
 - Next sequence number expected



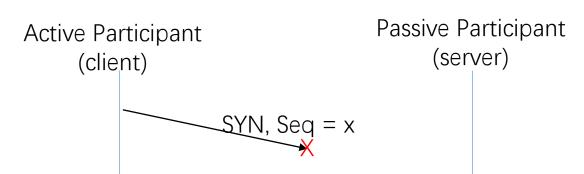
- Three-way Handshake
 - To share the sequence number



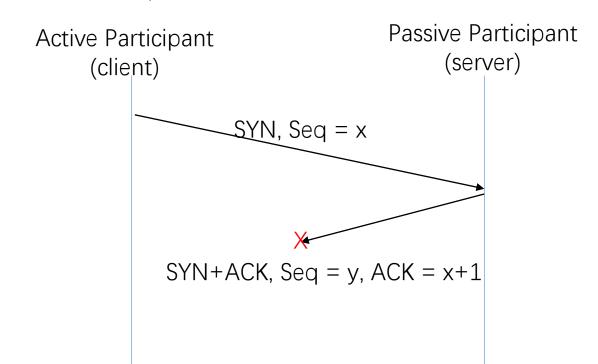
- Why two-way handshake is not enough?
 - To eliminate out-of-order connection request
 - Three-way: client will not response to the SYN+ACK if the connection request is old
 - To confirm the client knows the server is ready
 - In case SYN+ACK loss



- Three-way Handshake
 - SYN loss
 - Client retransmits, until receives SYN+ACK from server

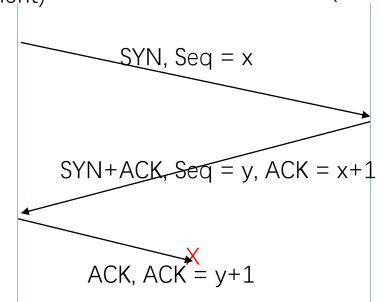


- Three-way Handshake
 - SYN+ACK loss
 - Server retransmits, until receive ACK from client

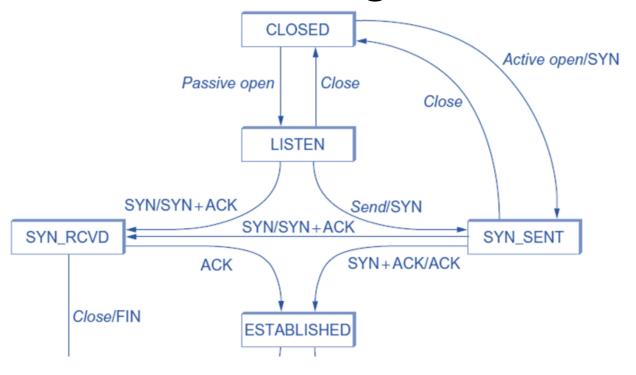


- Three-way Handshake
 - Client ACK loss
 - Server retransmits SYN+ACK, until receive ACK from client
 - or Client transmits DATA+ACK, server treats it as ACK
 Active Participant (client)

 Active Participant (server)



TCP State-transition Diagram



- Four-way Handshake
 - To release resource
 - Can be asymmetric
 - e.g.: Server are transmitting to client; Clients has nothing to transmit, it closes the connection, releases transmission queue

Server

Connection Termination

Four-way Handshake

Use to prevent

new connections

retransmitted FIN (due to

ACK loss) from terminating

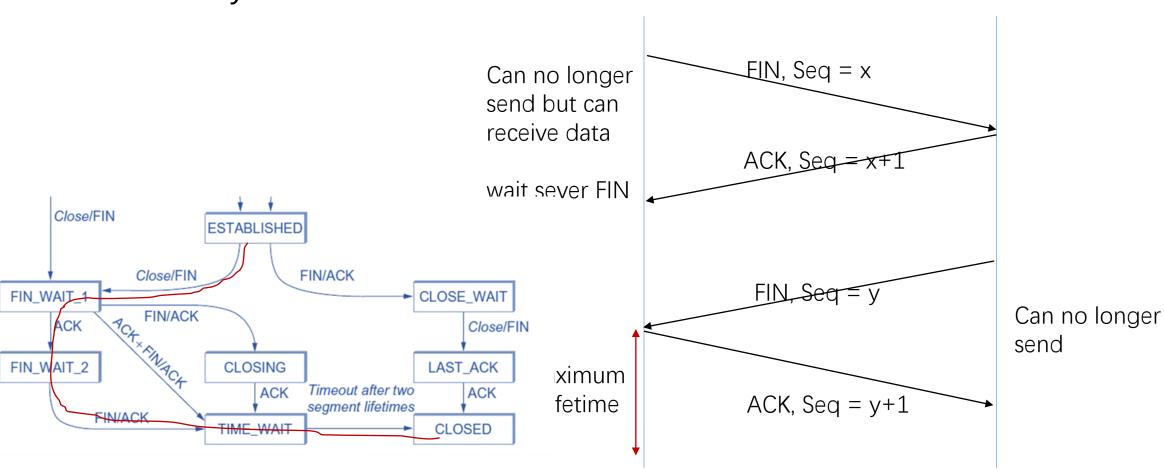
EIN, Seq = x Can no longer send but can receive data ACK, Seq wait sever FIN Can no longer send Wait 2*maximum segment lifetime ACK, Seq = y+1

Client

Server

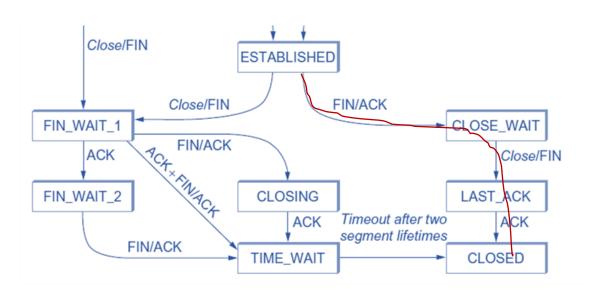
Connection Termination

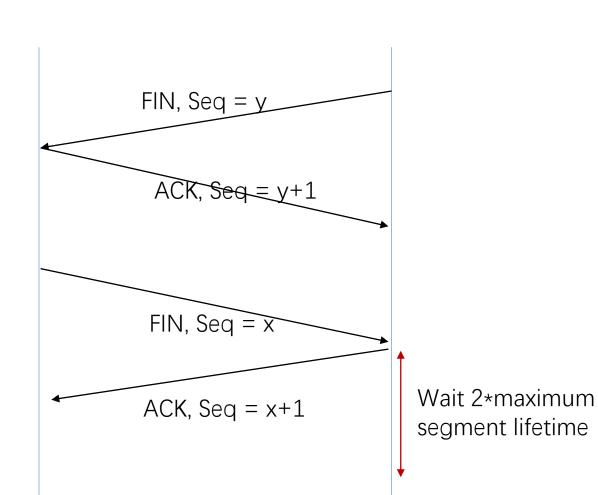
Four-way Handshake



Client

• Case 2:

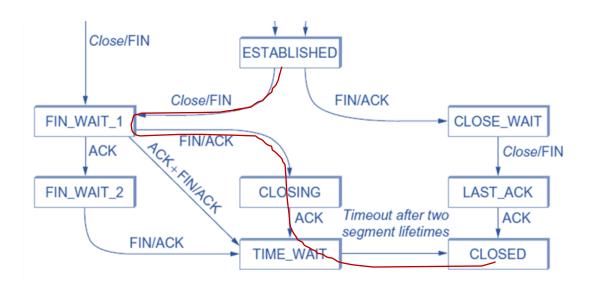


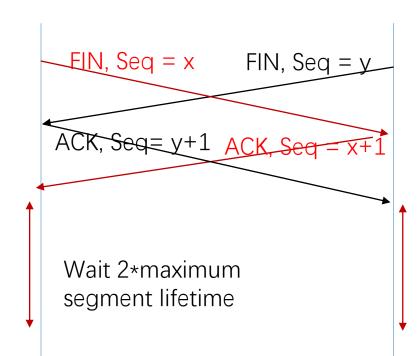


Client

Server

• Case 3:



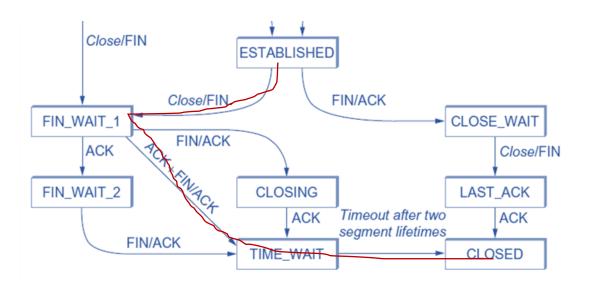


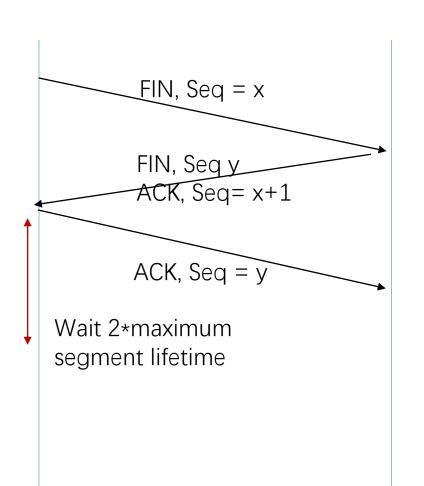
Client

Wait 2*maximum segment lifetime

Server

• Case 4:

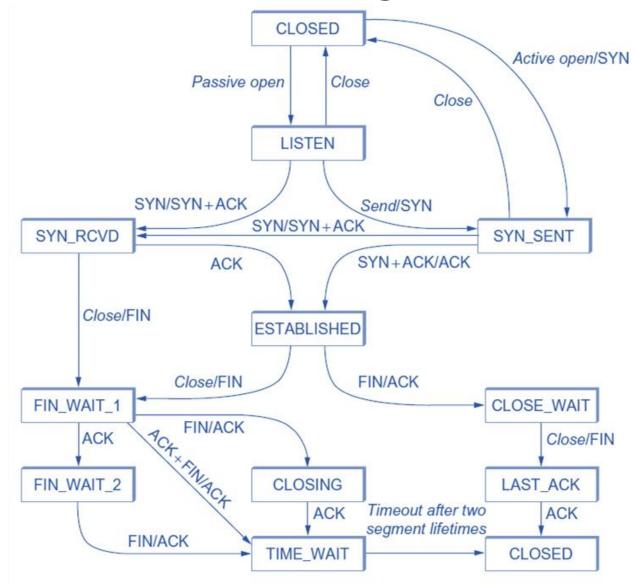




Client

Server

TCP State-transition Diagram



Reference

• Textbook 5.1 5.2