End-to-End Protocols: UDP and TCP

Hui Chen, Ph.D.
Dept. of Engineering & Computer Science
Virginia State University
Petersburg, VA 23806

Acknowledgements

- Some pictures used in this presentation were obtained from the Internet
- The instructor used the following references
  - Larry L. Peterson and Bruce S. Davie, Computer Networks: A Systems Approach, 5th Edition, Elsevier, 2011
  - Andrew S. Tanenbaum, Computer Networks, 5th Edition, Prentice-Hall, 2010
  - James F. Kurose and Keith W. Ross, Computer Networking: A Top-Down Approach, 5th Ed., Addison Wesley, 2009
  - Larry L. Peterson's (http://www.cs.princeton.edu/~llp/) Computer Networks class web site

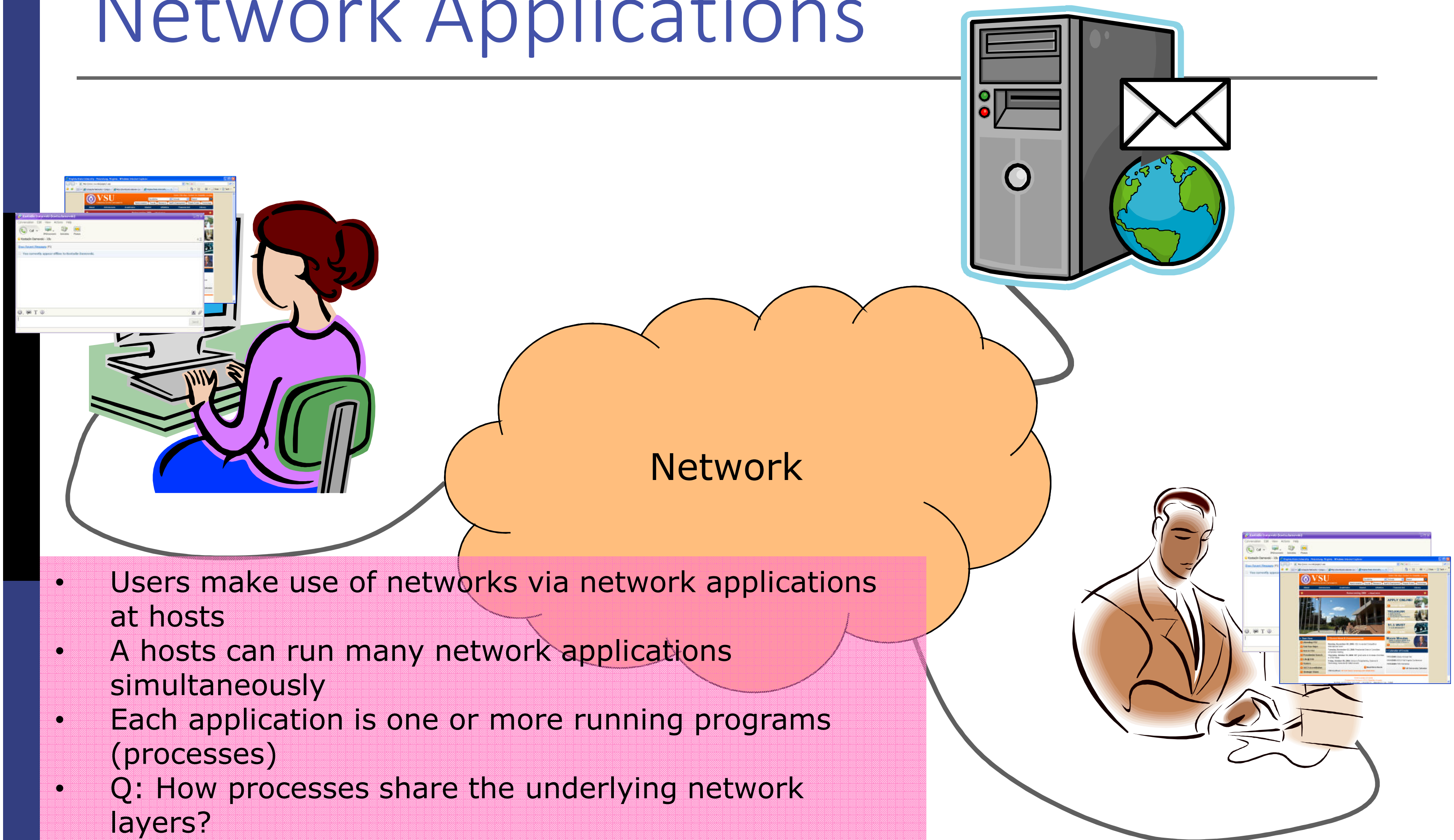
Acknowledgements

- Animations in the PDF version of the slides is produced using
  - PPspliT
  - http://www.dia.uniroma3.it/~rimondin/downloads.php

Outline

- User Datagram Protocol
- Transmission Control Protocol

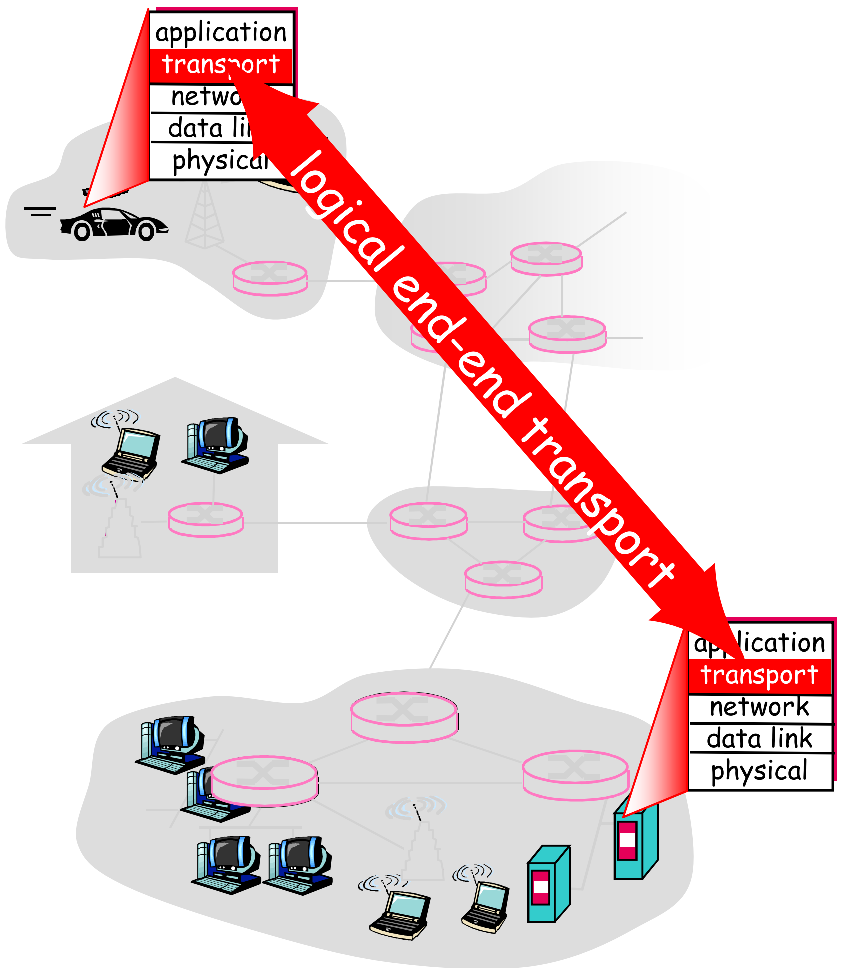
Network Applications



- Users make use of networks via network applications at hosts
- A hosts can run many network applications simultaneously
- Each application is one or more running programs (processes)
- Q: How processes share the underlying network layers?

Transport Layer Services and Protocols

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



Transport vs. Network Layer (1)

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

Household analogy:

12 kids sending letters among themselves via their parents

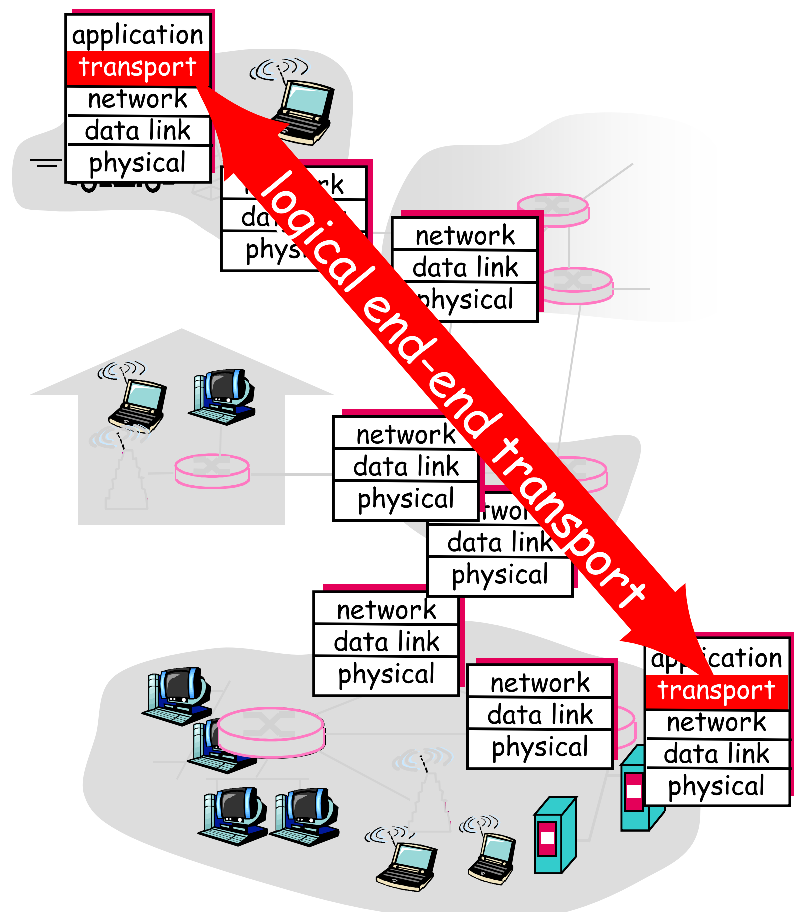
- processes = kids
- application messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill (parents)
- network-layer protocol = postal service

Transport vs. Network Layer (2)

- Network layer: Underlying best-effort network
  - drop messages
  - re-orders messages
  - delivers duplicate copies of a given message
  - limits messages to some finite size
  - delivers messages after an arbitrarily long delay
- Transport Layer: Common end-to-end services
  - guarantee message delivery
  - deliver messages in the same order they are sent
  - deliver at most one copy of each message
  - support arbitrarily large messages
  - support synchronization
  - allow the receiver to flow control the sender
  - support multiple application processes on each host

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



Multiplexing/Demultiplexing

Host-to-host delivery ←→ process-to-process delivery

Multiplexing/Demultiplexing

Host-to-host delivery ←→ process-to-process delivery

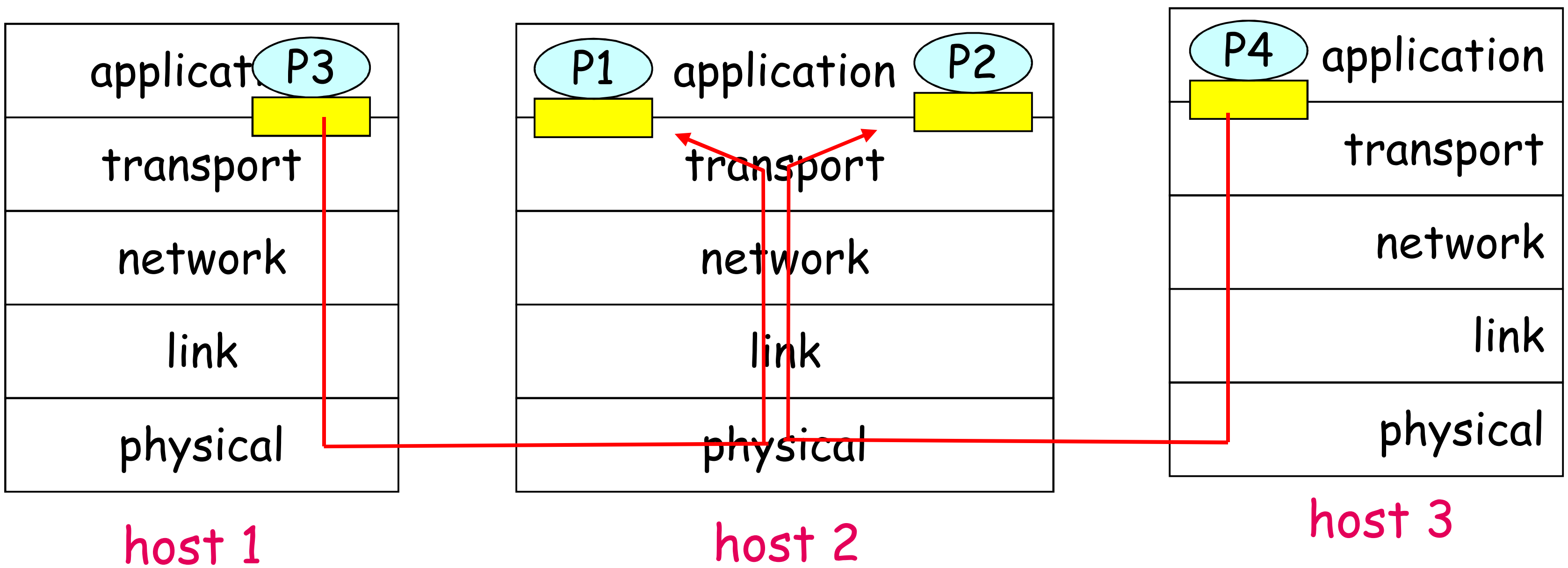
Demultiplexing at rcv host:

delivering received segments to correct socket

Multiplexing at send host:

gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

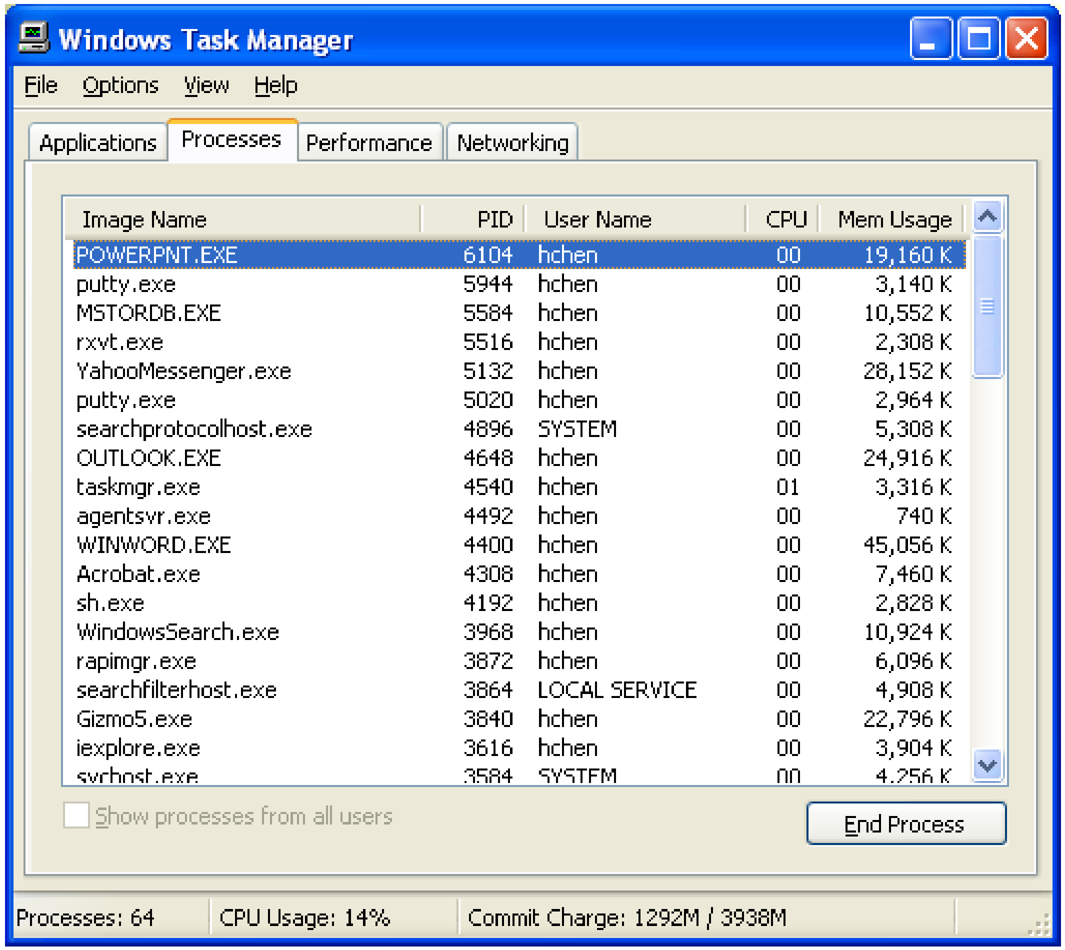
= socket = process



Simple Demultiplexer (1)

- Need to know to or from which process the data is sent or come
  - Identify processes on hosts
- How to identify processes on hosts?
  - Introduce concept of “port”
  - *Q: why not to use process id?*

Processes ID: Windows Example



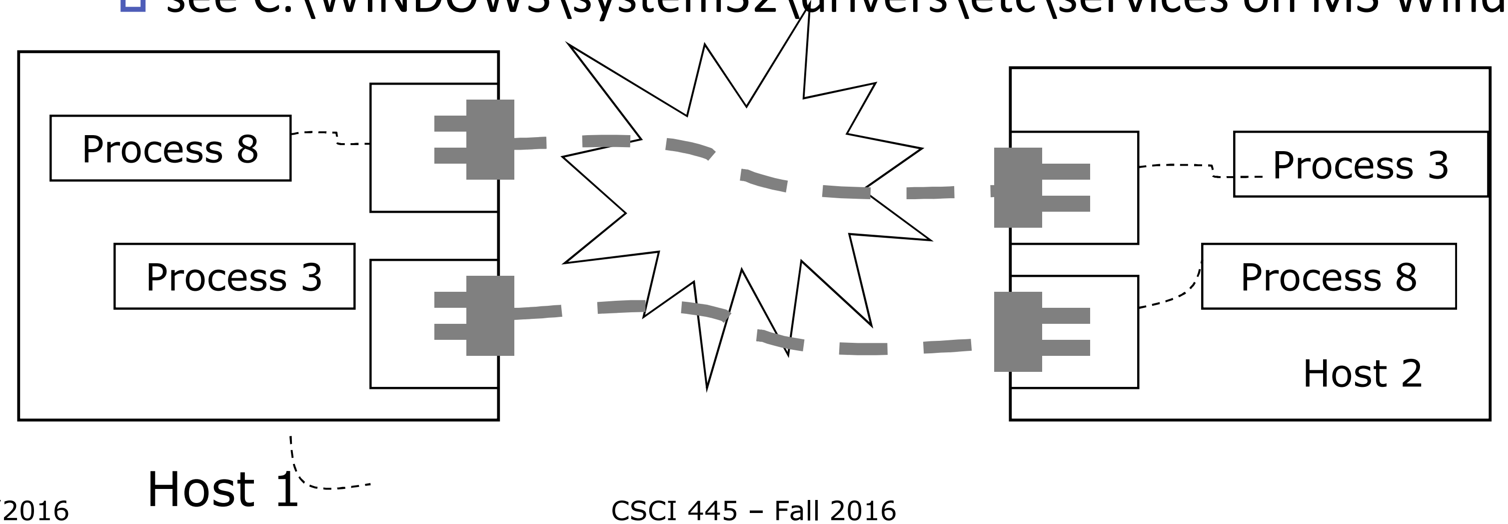
Processes ID: Linux Example

hchen@turing:~

```
[hchen@turing ~]$ ps ax
  PID TTY      STAT   TIME COMMAND
    1 ?        Ss     0:02 init [5]
    2 ?        S<     0:00 [migration/0]
    3 ?        SN     0:00 [ksoftirqd/0]
    4 ?        S<     0:00 [watchdog/0]
    5 ?        S<     0:00 [migration/1]
    6 ?        SN     0:00 [ksoftirqd/1]
    7 ?        S<     0:00 [watchdog/1]
    8 ?        S<     0:00 [migration/2]
    9 ?        SN     0:00 [ksoftirqd/2]
   10 ?        S<     0:00 [watchdog/2]
   11 ?        S<     0:00 [migration/3]
   12 ?        SN     0:00 [ksoftirqd/3]
   13 ?        S<     0:00 [watchdog/3]
   14 ?        S<     0:00 [migration/4]
   15 ?        SN     0:00 [ksoftirqd/4]
   16 ?        S<     0:00 [watchdog/4]
   17 ?        S<     0:00 [migration/5]
   18 ?        SN     0:00 [ksoftirqd/5]
   19 ?        S<     0:00 [watchdog/5]
   20 ?        S<     0:00 [migration/6]
   21 ?        SN     0:00 [ksoftirqd/6]
   22 ?        S<     0:00 [watchdog/6]
```

Simple Demultiplexer (2)

- How to identify processes on hosts?
  - *Q: why not to use process id?*
  - Introduce concept of "port"
    - Endpoints identified by ports
    - servers have well-known ports
    - see /etc/services on Unix/Linux
    - see C:\WINDOWS\system32\drivers\etc\services on MS Windows

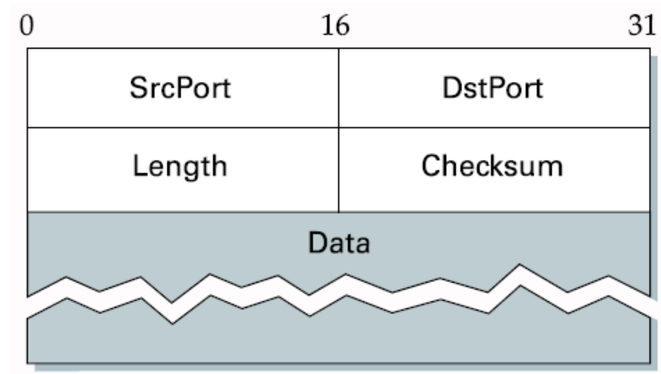


Simple Demultiplexer: UDP

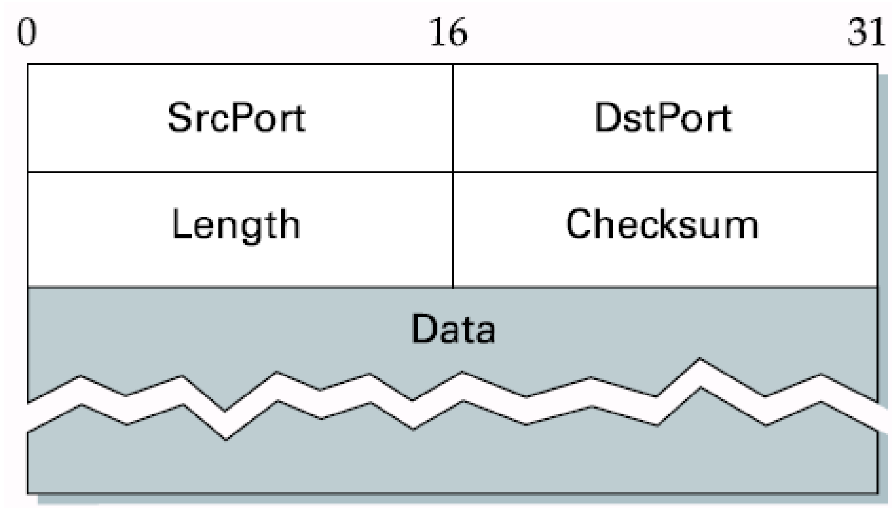
- Adds multiplexing to Internet Protocol
  - Endpoints identified by ports (UDP ports)
  - Demultiplex via ports on hosts
  - Nothing more is added
    - Unreliable and unordered datagram service
    - No flow control
  - User Datagram Protocol (UDP)
    - A process is identified by <host, port>
    - Connectionless model
- Header format
  - Optional checksum
    - psuedo header + UDP header + data
    - pseudo header = protocol number + source IP address and destination IP address + UDP length field

From IP header

From UDP header



In-Class Exercise L15-1



- Q1: How many UDP ports are there?
- Q2: How big are UDP headers?
- Q3: How much data does a UDP datagram can carry?

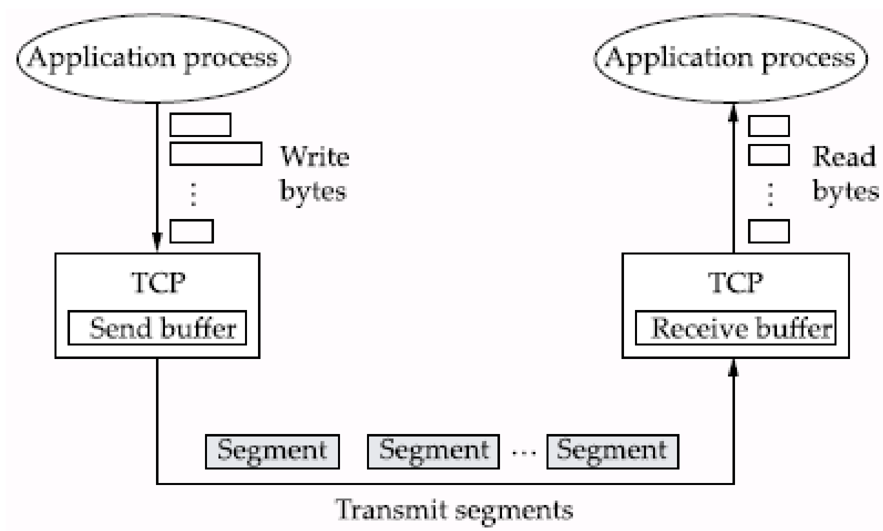
- Turn your work in before you leave!

Transmission Control Protocol (TCP)

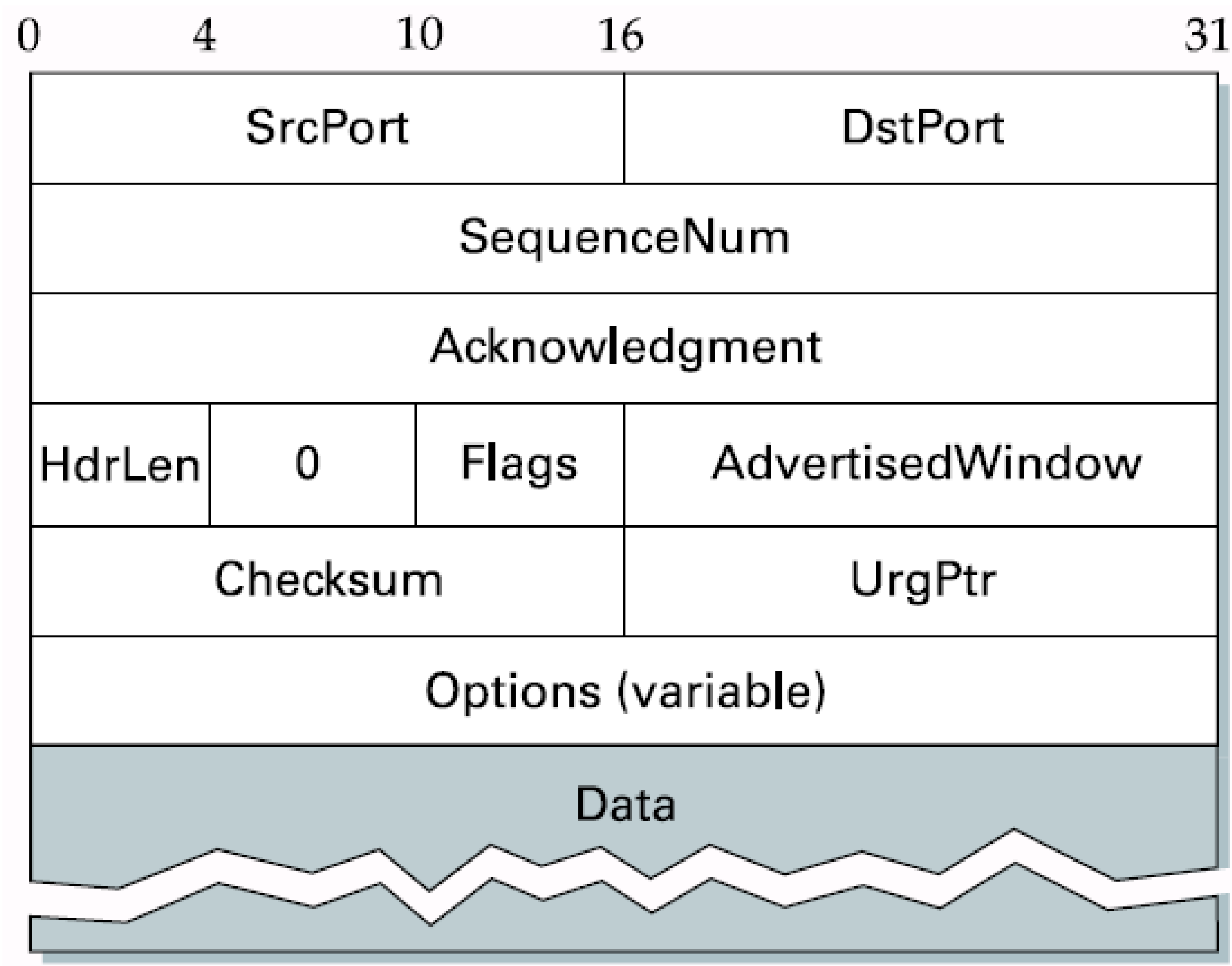
- Connection-oriented
- Byte-stream
  - applications writes bytes
  - TCP sends segments
  - applications reads bytes
- Full duplex
- Flow control: keep sender from overrunning receiver
- Congestion control: keep sender from overrunning network

Data Link Versus Transport

- Potentially connects many different hosts
  - need explicit connection establishment and termination
- Potentially different RTT
  - need adaptive timeout mechanism
- Potentially long delay in network
  - need to be prepared for arrival of very old packets
- Potentially different capacity at destination
- need to accommodate different node capacity
- Potentially different network capacity
- need to be prepared for network congestion

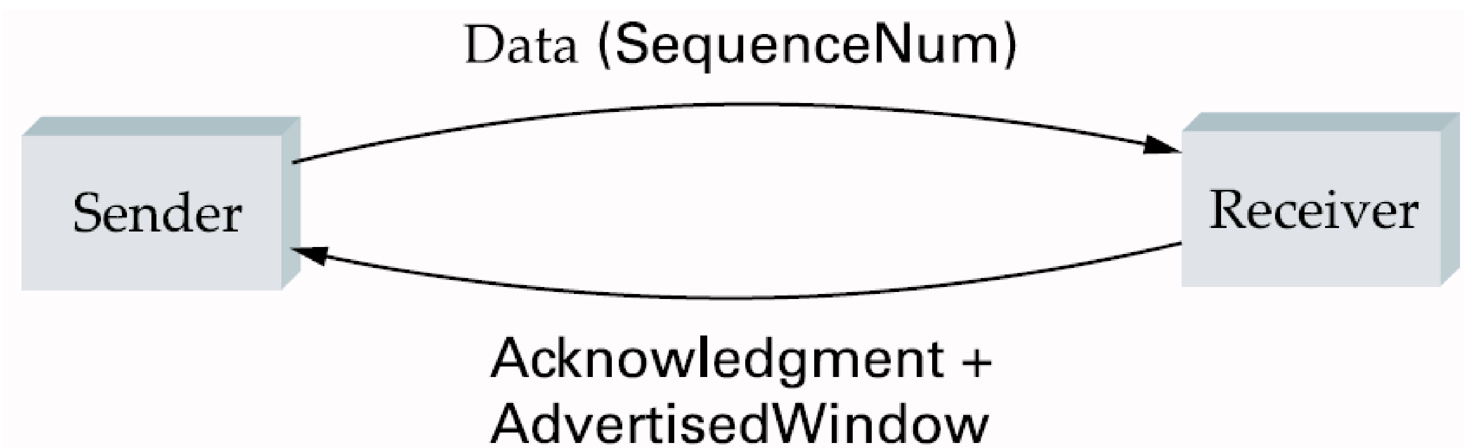


Segment Format (1)



Segment Format (2)

- Each connection identified with 4-tuple:
  - (SrcPort, SrcIPAddr, DsrPort, DstIPAddr)
- Sliding window + flow control
  - acknowledgment, SequenceNum, AdvertisedWinow
- Flags
  - SYN, FIN, RESET, PUSH, URG, ACK
- Checksum
  - pseudo header + TCP header + data



Sequence and Acknowledgement Numbers (1)

- Host A sends a file of 500,000 bytes over a TCP connection with Maximum Segment Size (MSS) as 1,000 bytes to host B
  - How many segments? 500,000/1,000 = 500
  - Sequence number assignments
    - Sequence number of 1st segment? 0
    - Sequence number of 2nd segment? 1,000
    - Sequence number of 3rd segment? 2,000
    - ……

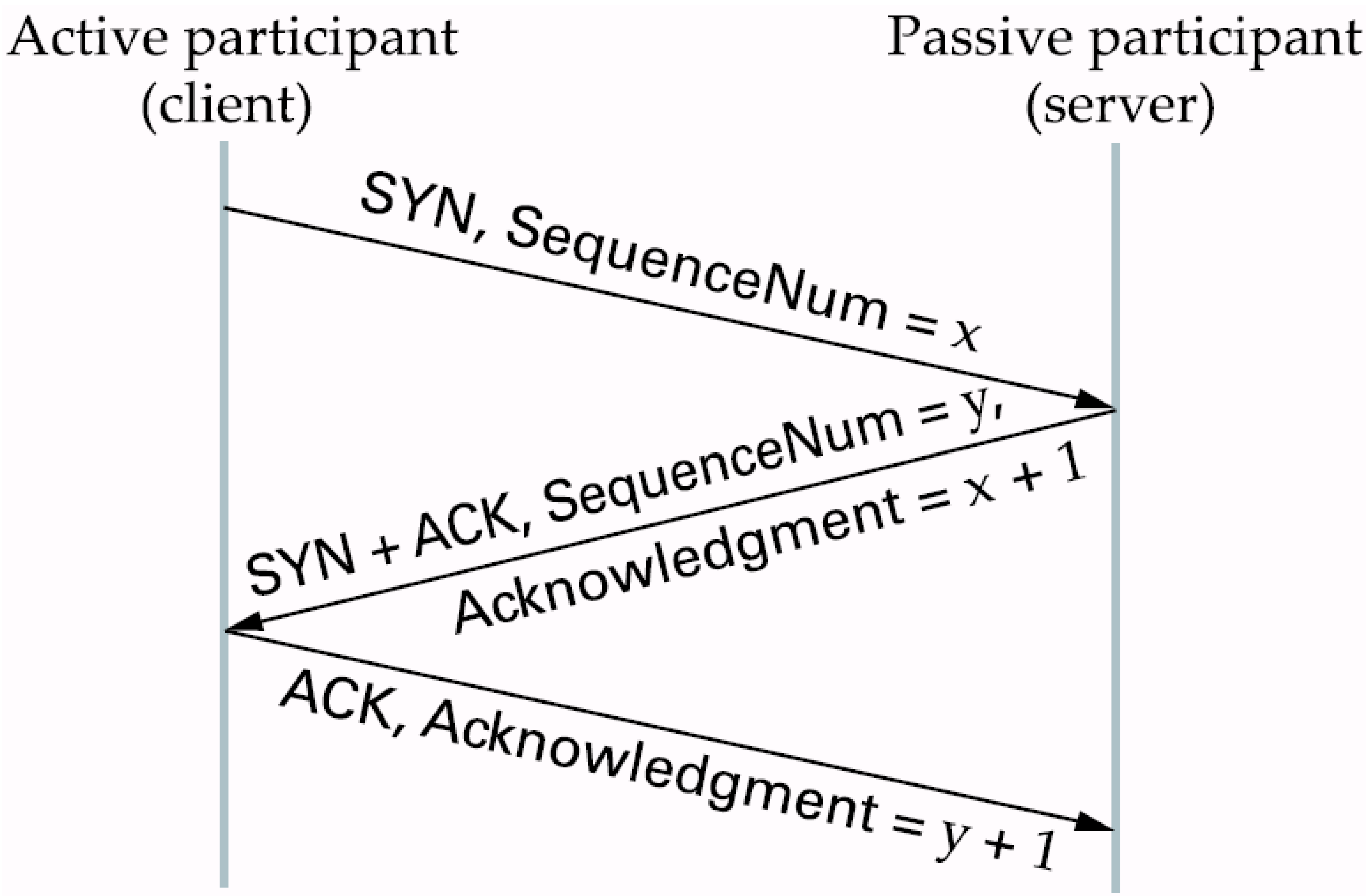
Sequence and Acknowledgement Numbers (2)

- Scenario 1
  - Host B received all bytes numbered 0 to 1,999 from host A
  - What would host B put in the acknowledgement number field of the segment it sends to A?
    - 2,000: the sequence number of the next byte host B is expecting
- Scenario 2
  - Host B received two segments containing bytes from 0-999, and 2,000-2,999, respectively?
  - What would host B put in the acknowledgement number field of the segment it sends to A?
    - 1000: TCP only acknowledges bytes up to the first missing byte in the stream, and it is the next byte host B is expecting
- Scenario 3
  - Host B received 1st segment containing bytes from 0-999. Somehow, next it received 3rd segment containing bytes from 2,000-2,999.
  - What does host B in this case that the segments arrive out of order?
    - TCP does not specify how to deal with this situation. Hence, it is up to the implementation.
      - Option 1: Host B immediately discards out-of-order segment → simple receiver design
      - Option 2: Host B keeps the out-of-order segment and waits for missing bytes to fill in the gaps → more efficient on bandwidth utilization → taken in practice

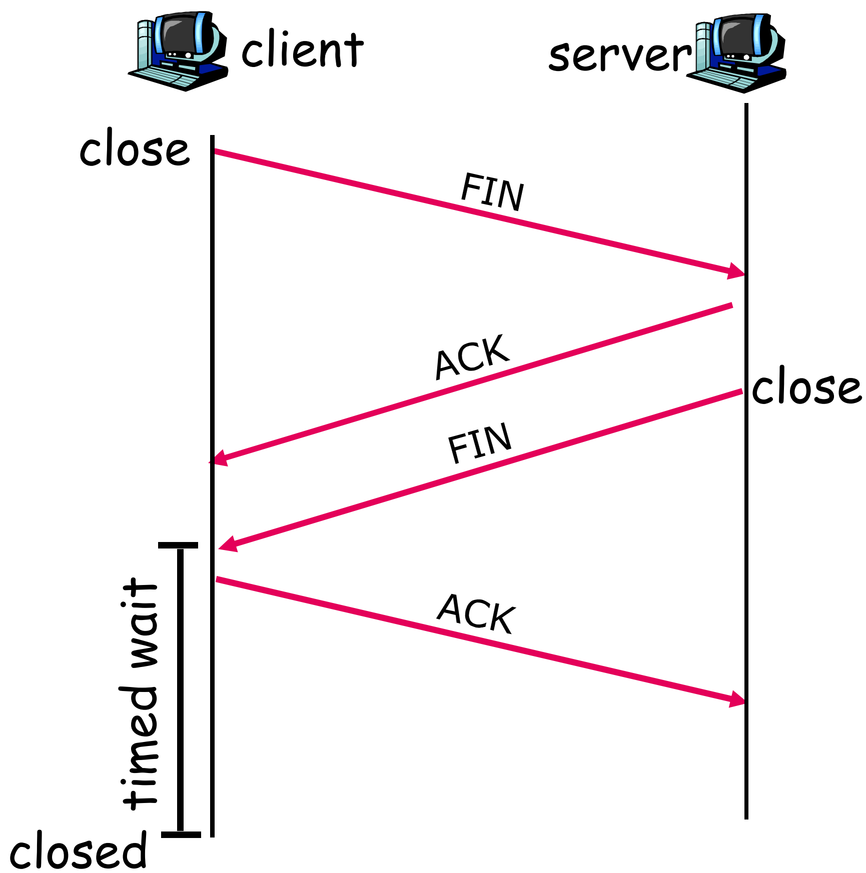
TCP is Connection-Oriented

- Keep track of states of receiver and sender
  - Connection Establishment
  - Connection Termination
  - TCP finite state machine and state transition

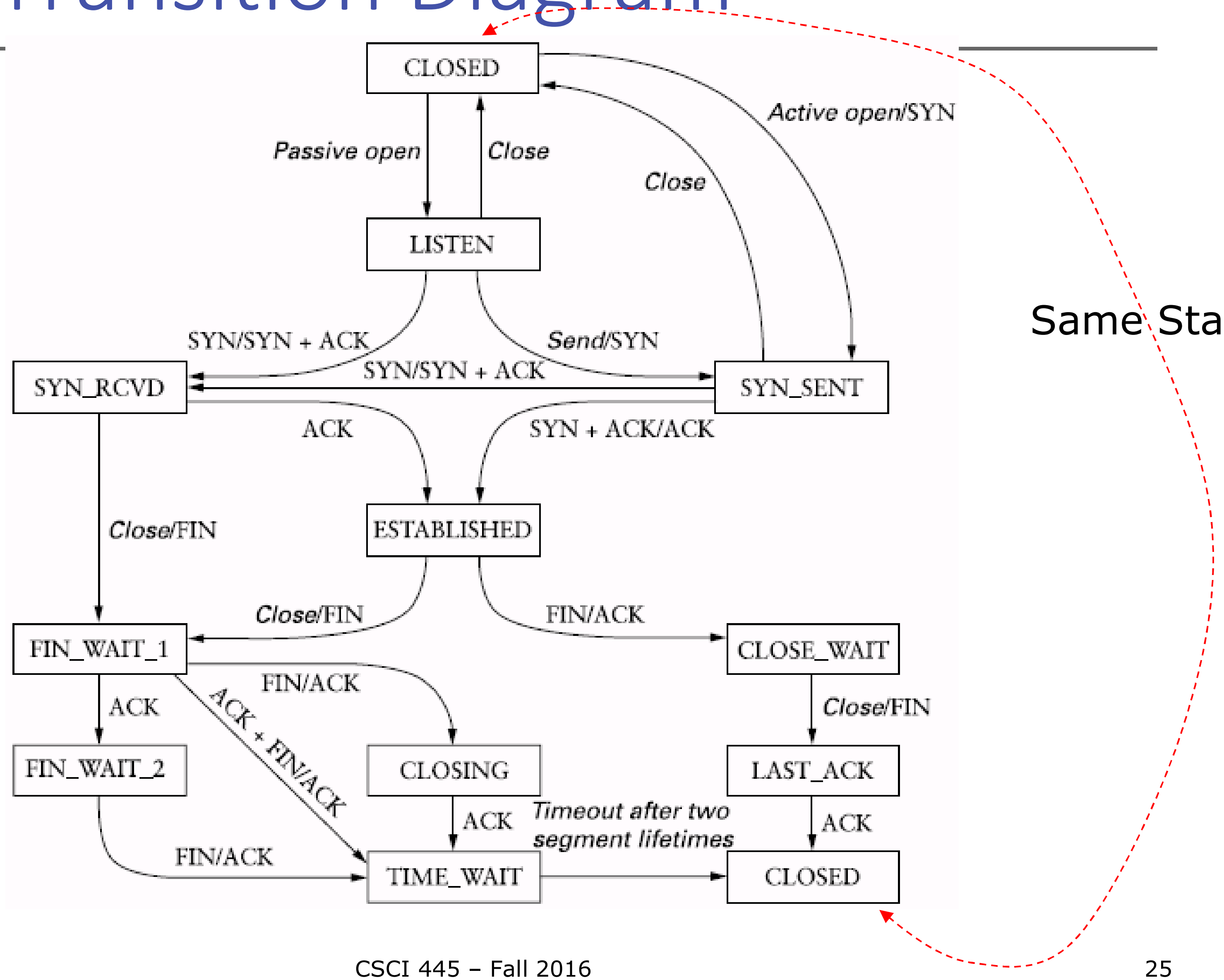
Connection Establishment



Connection Termination



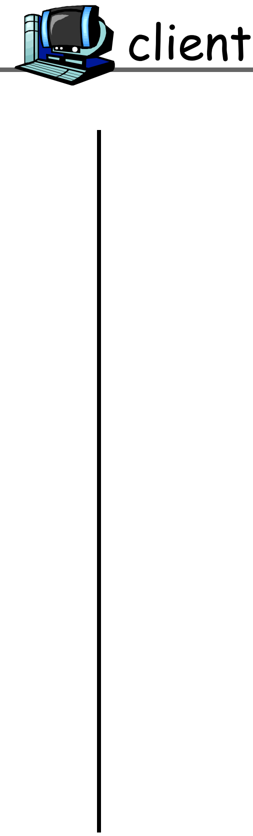
State Transition Diagram



Same State

Connection Establishment and State Transition

Connection Establishment and State Transition



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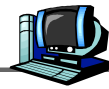
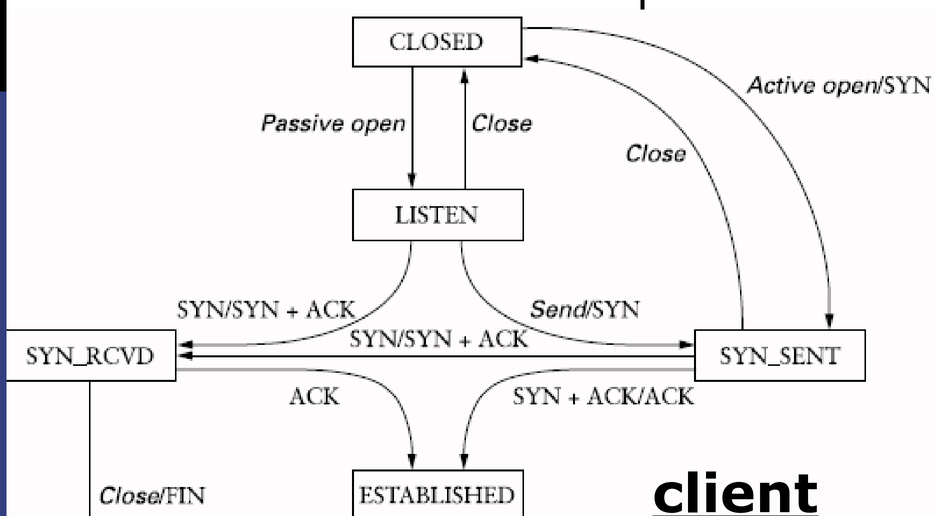


client



client

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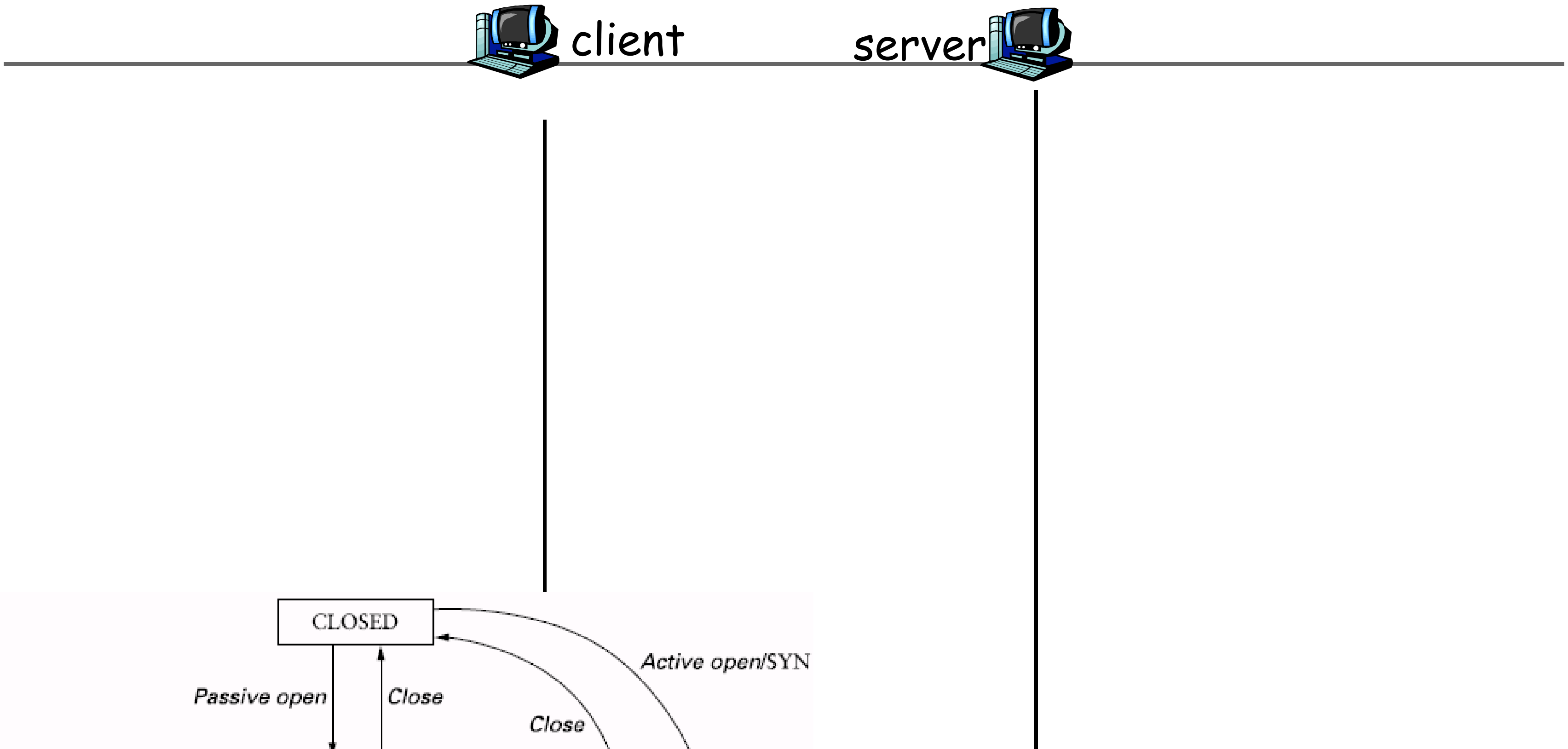

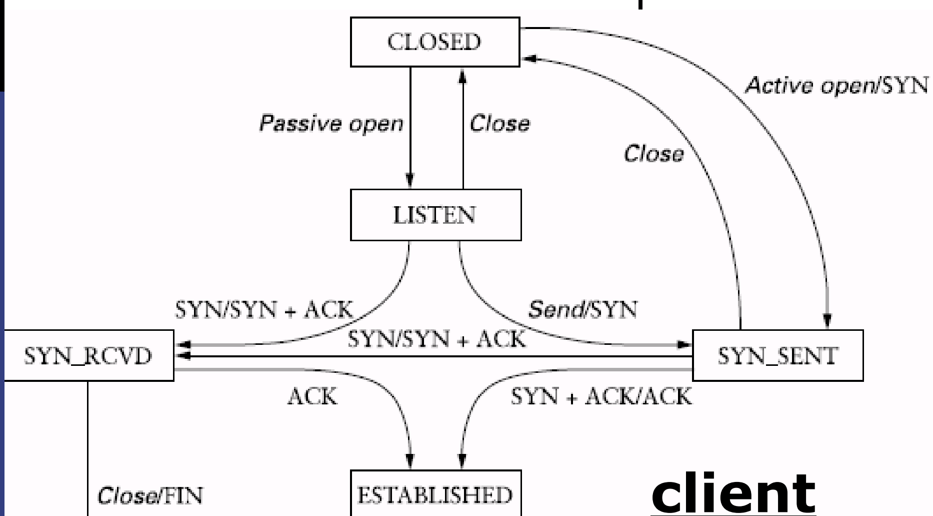

client

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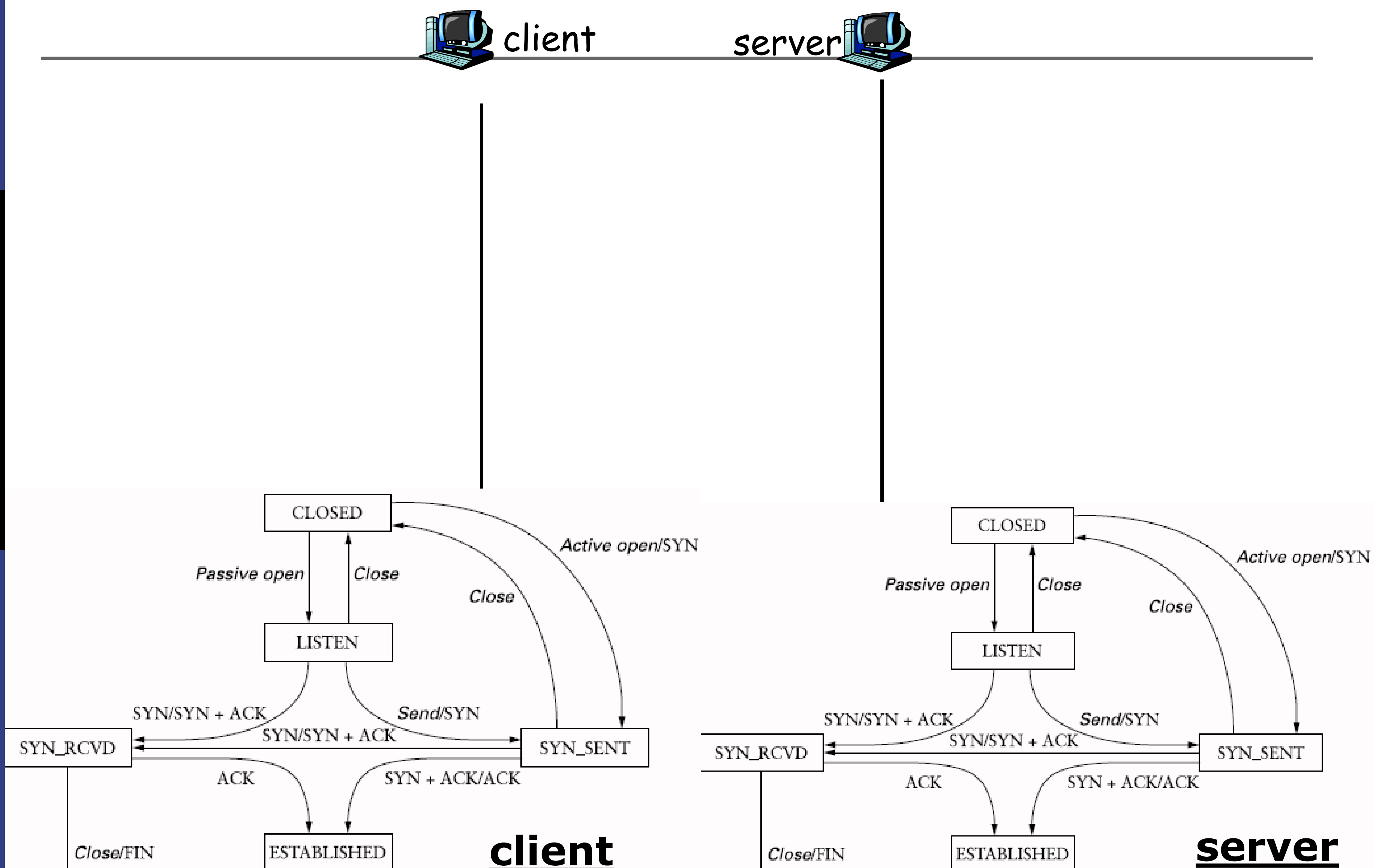

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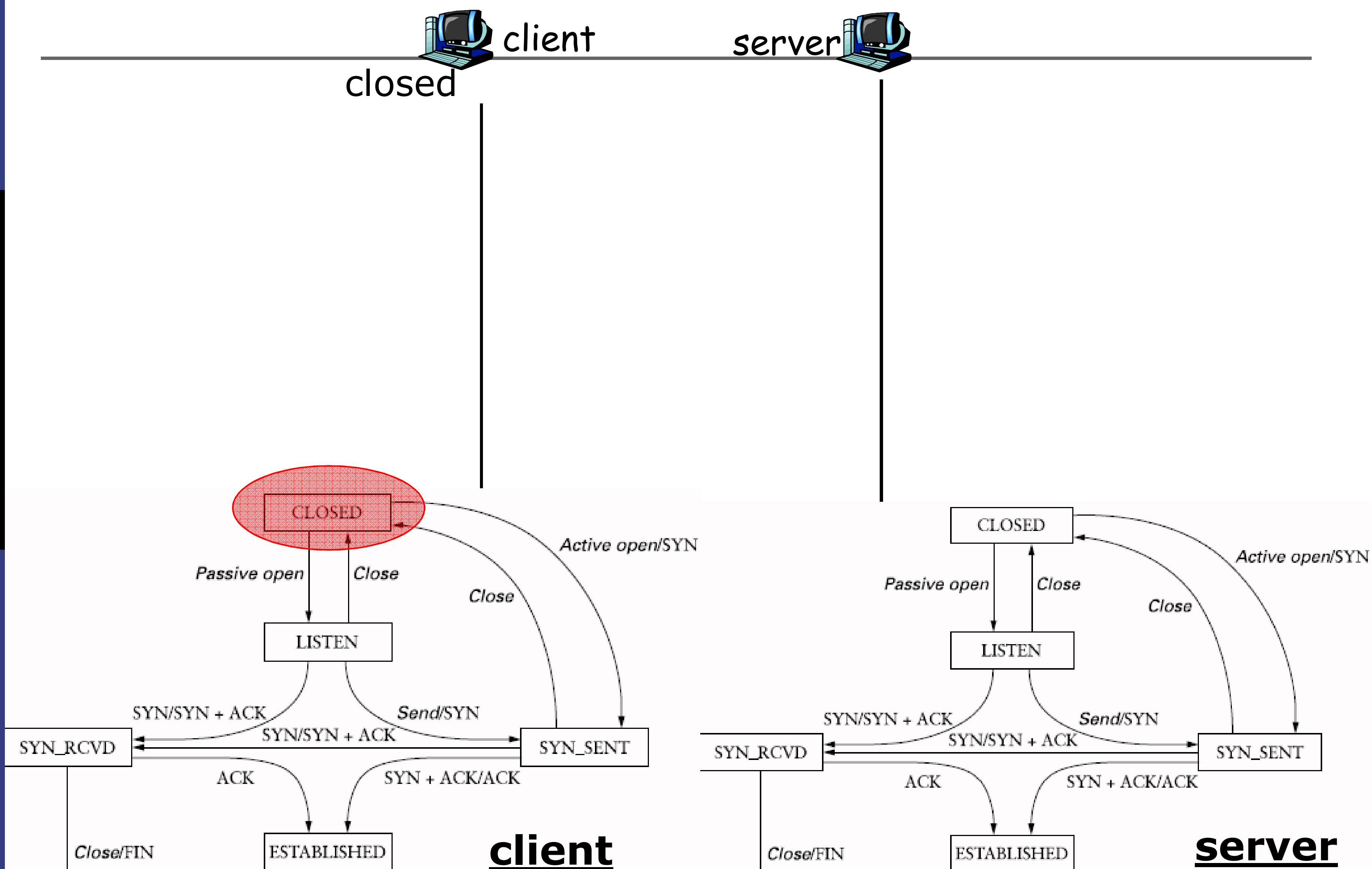

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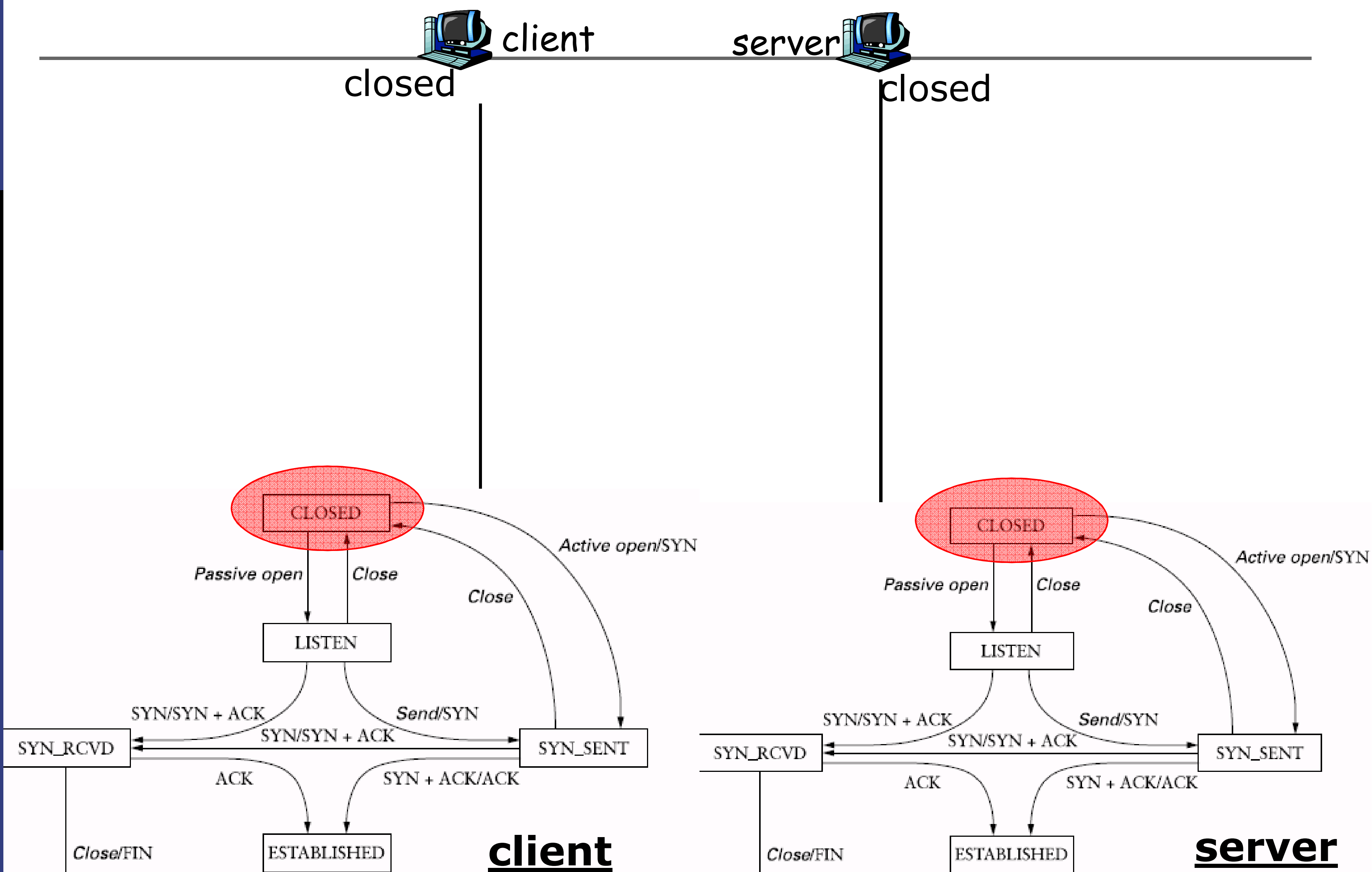

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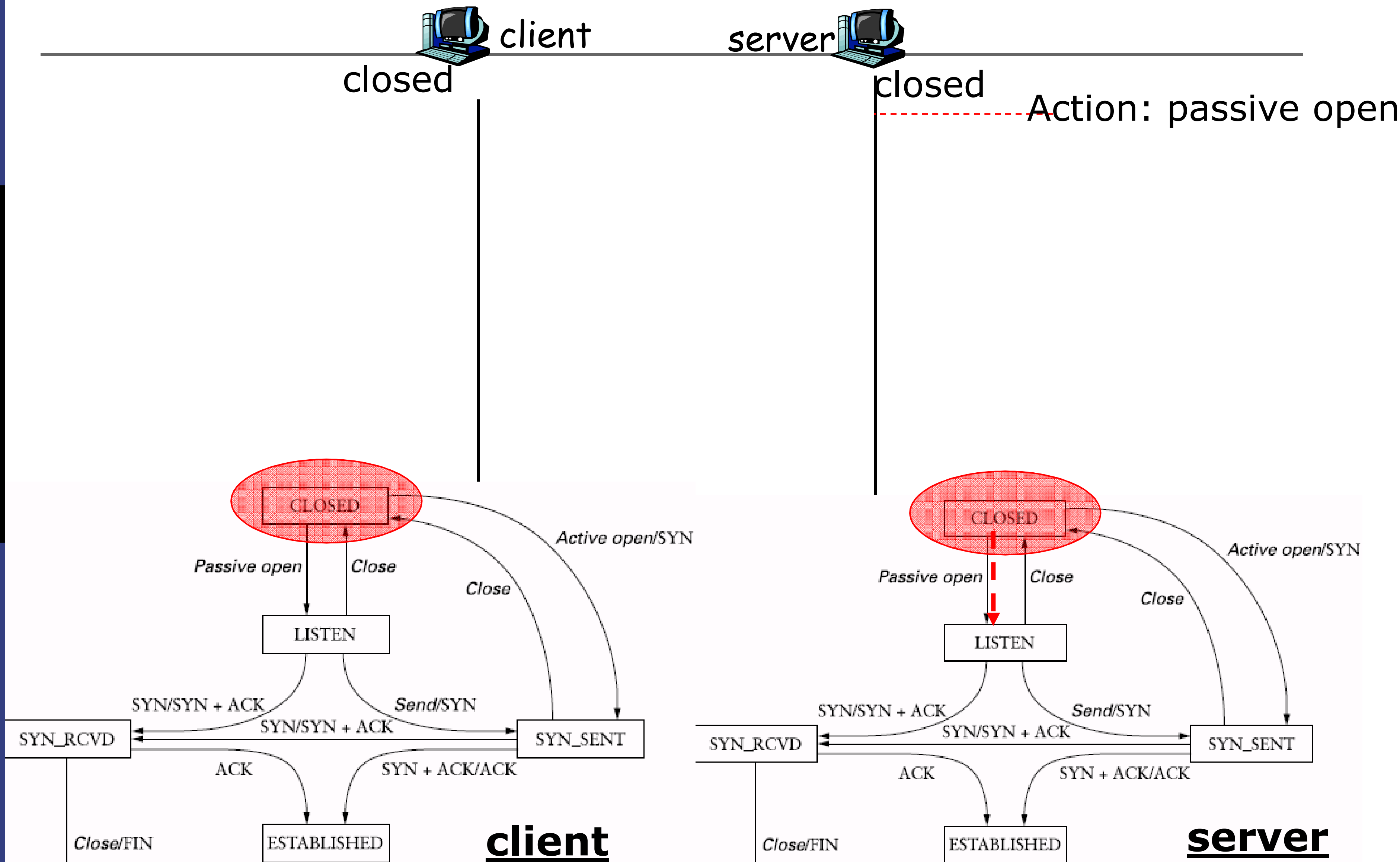

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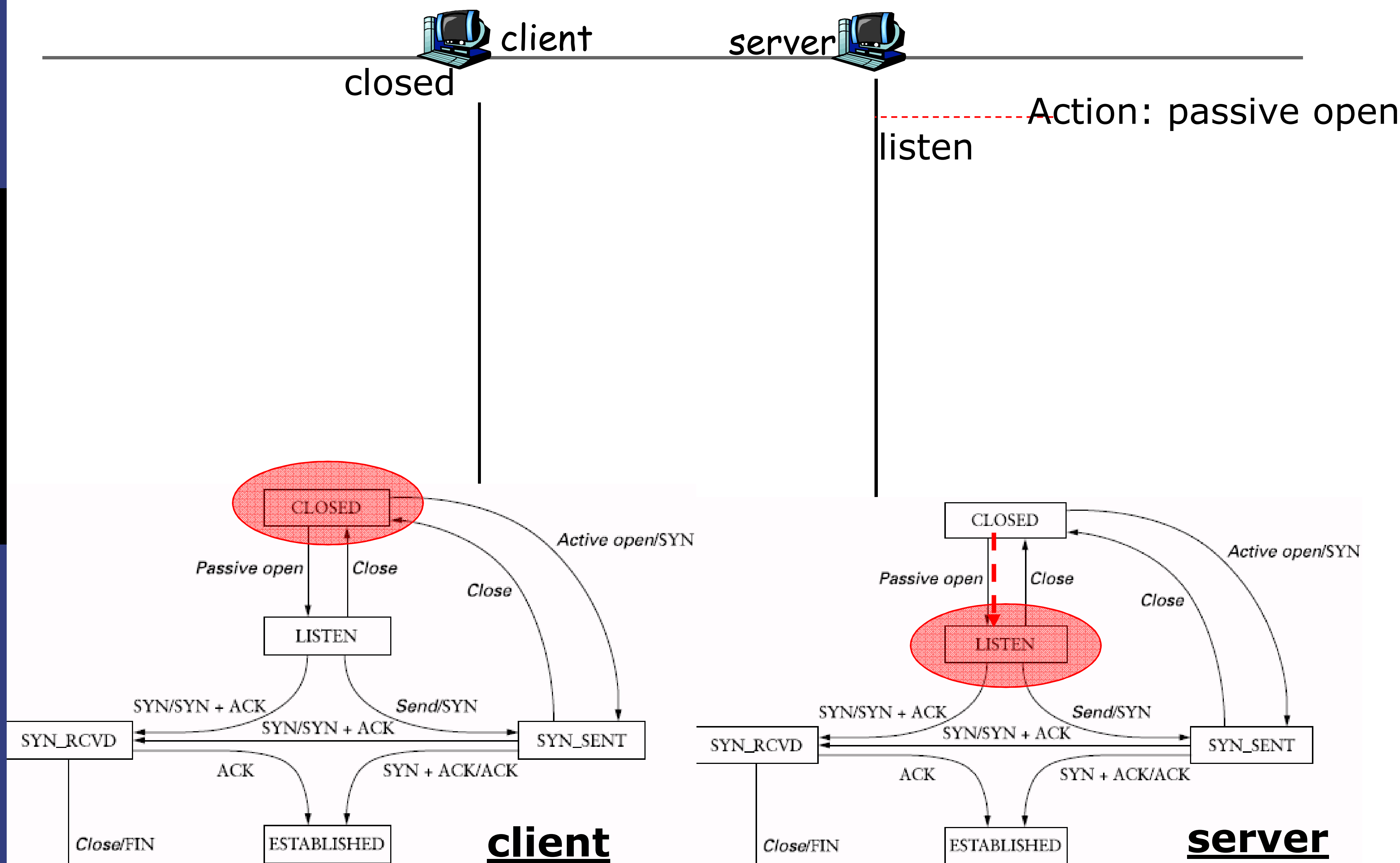

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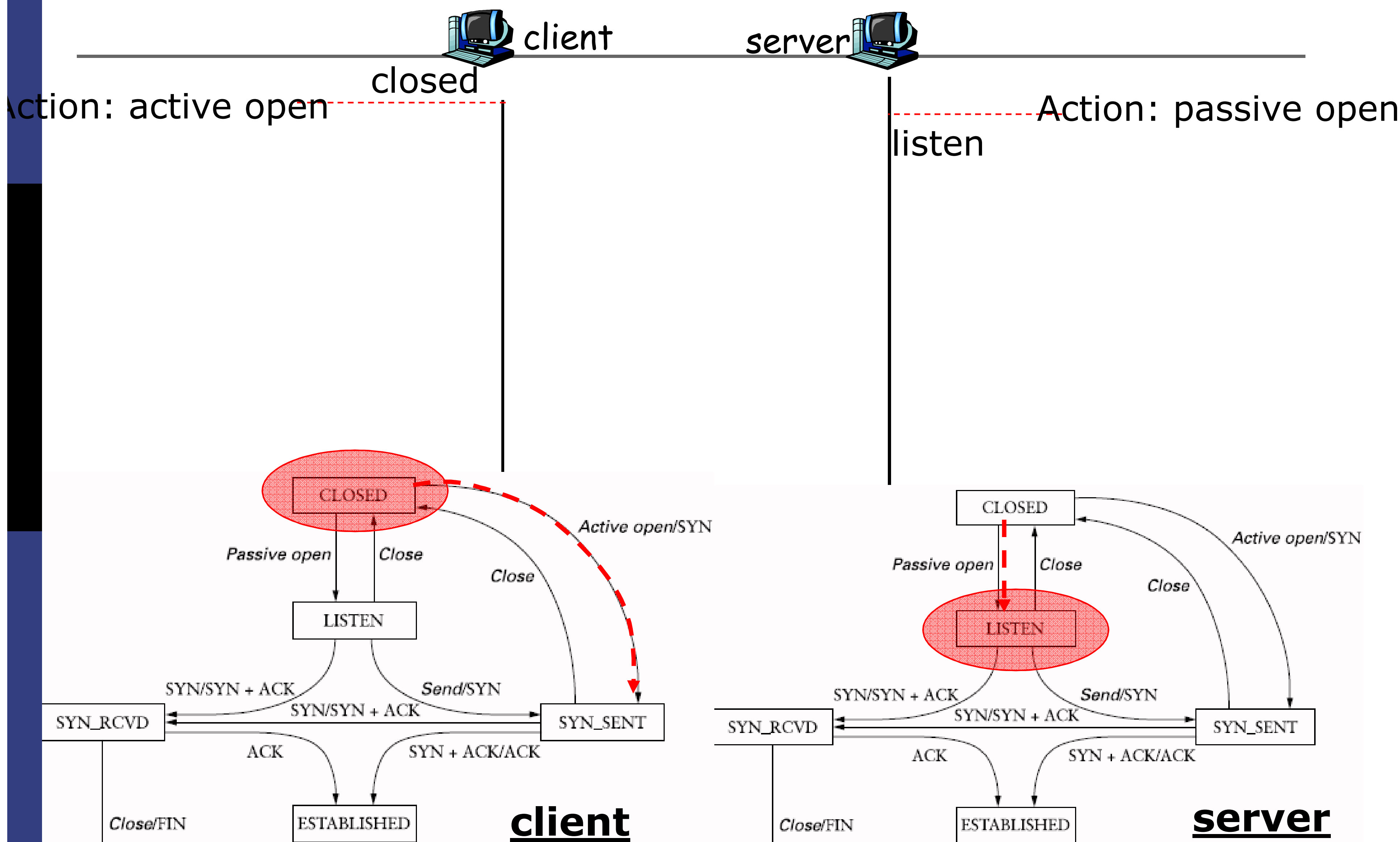

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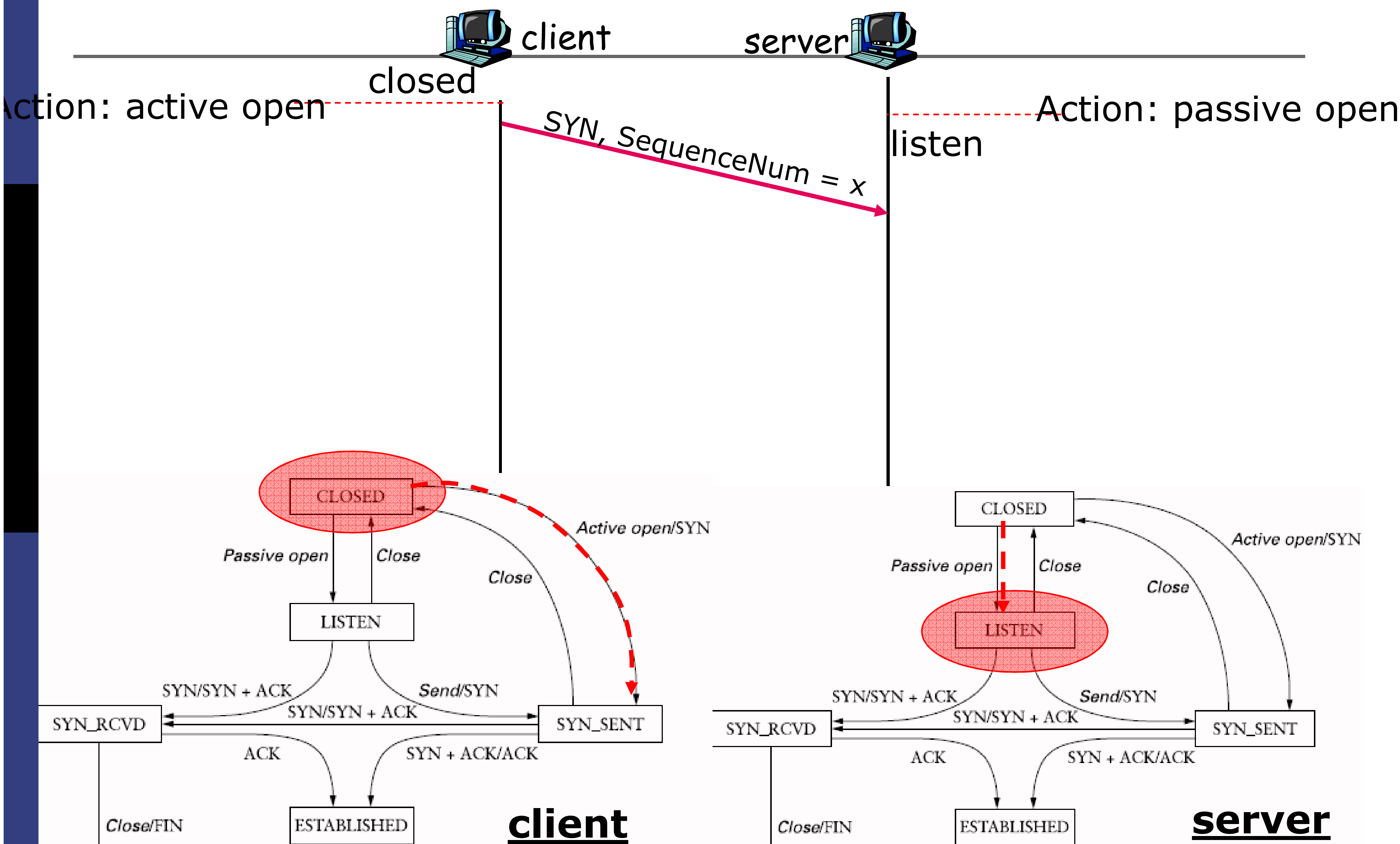

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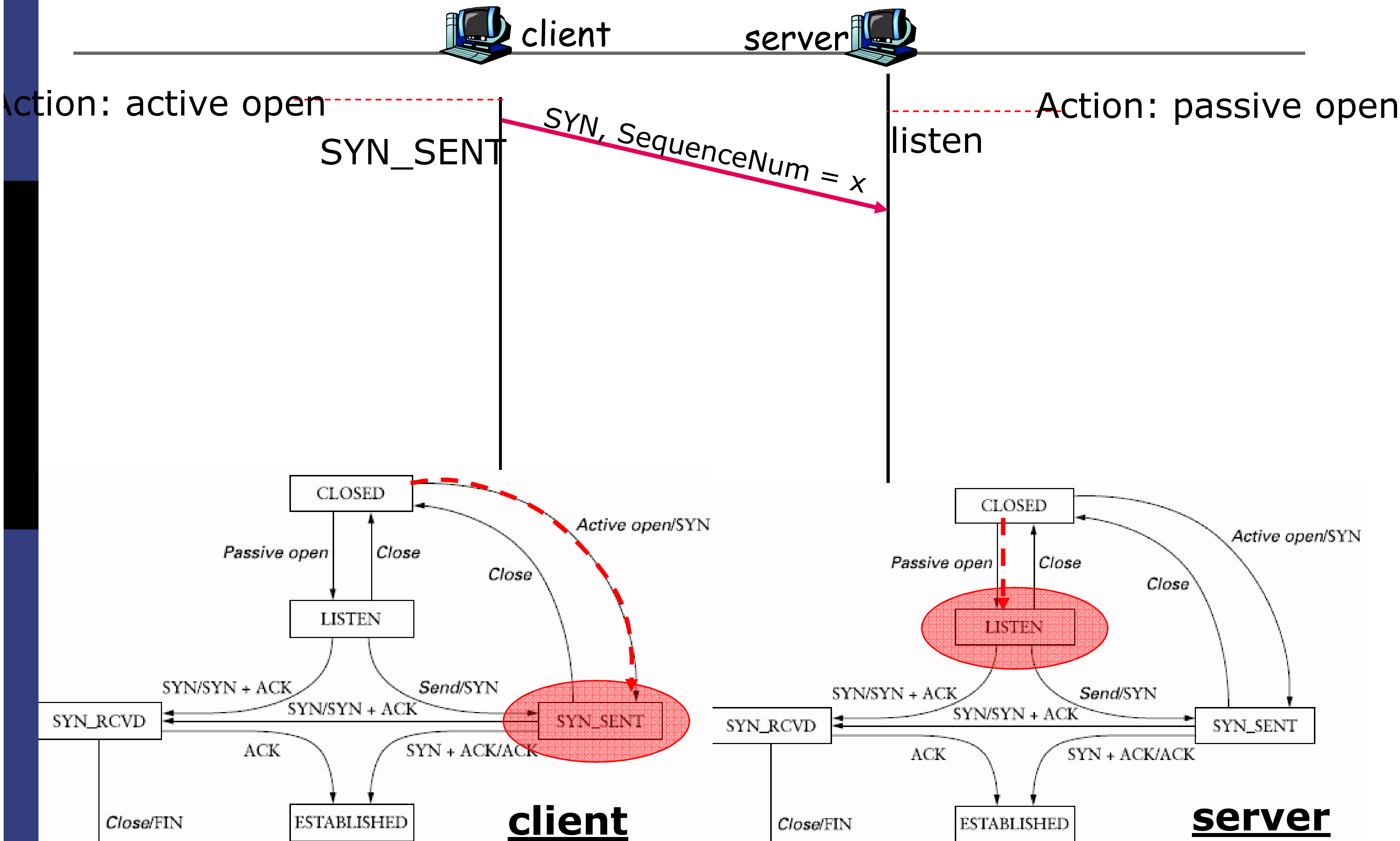

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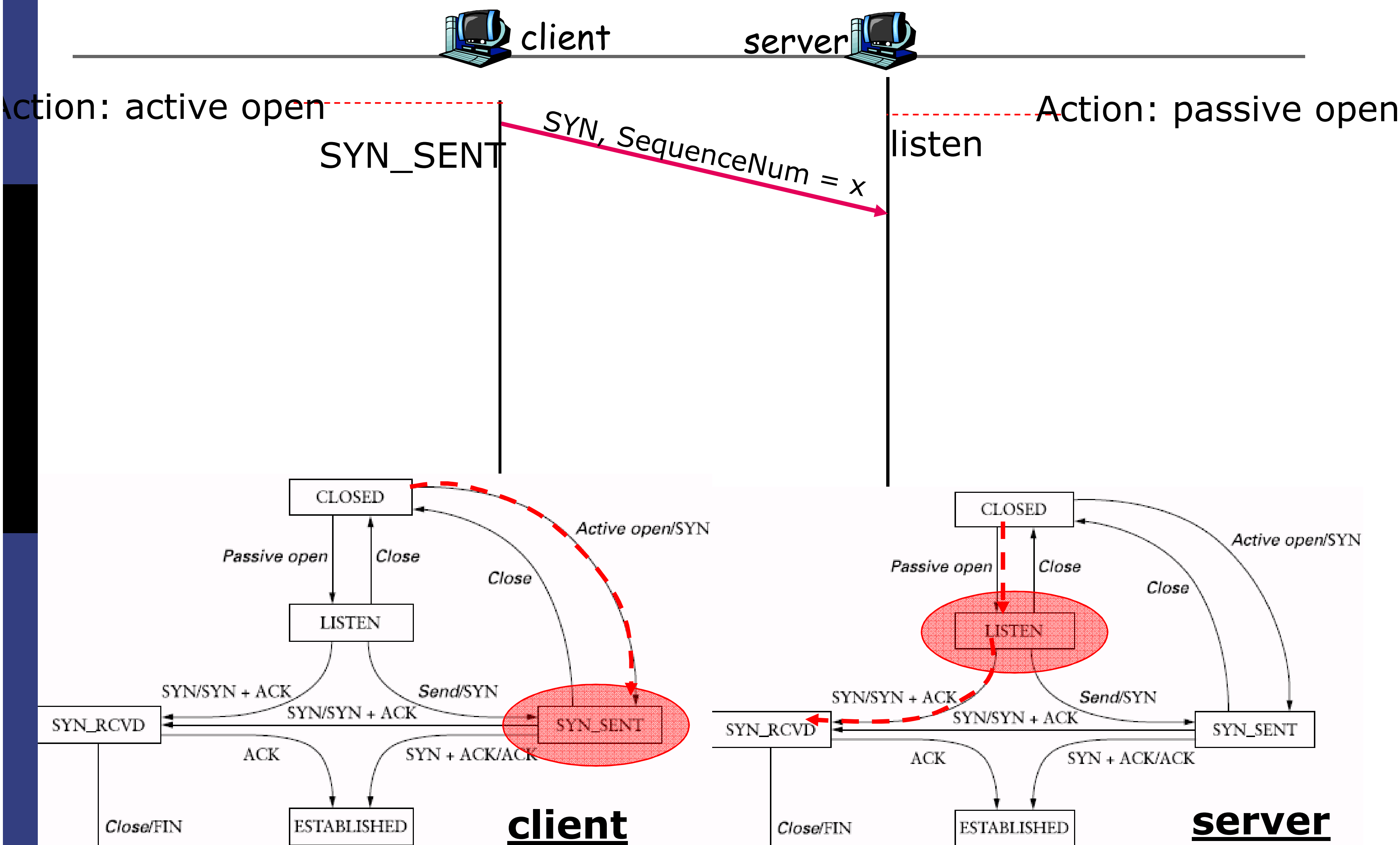

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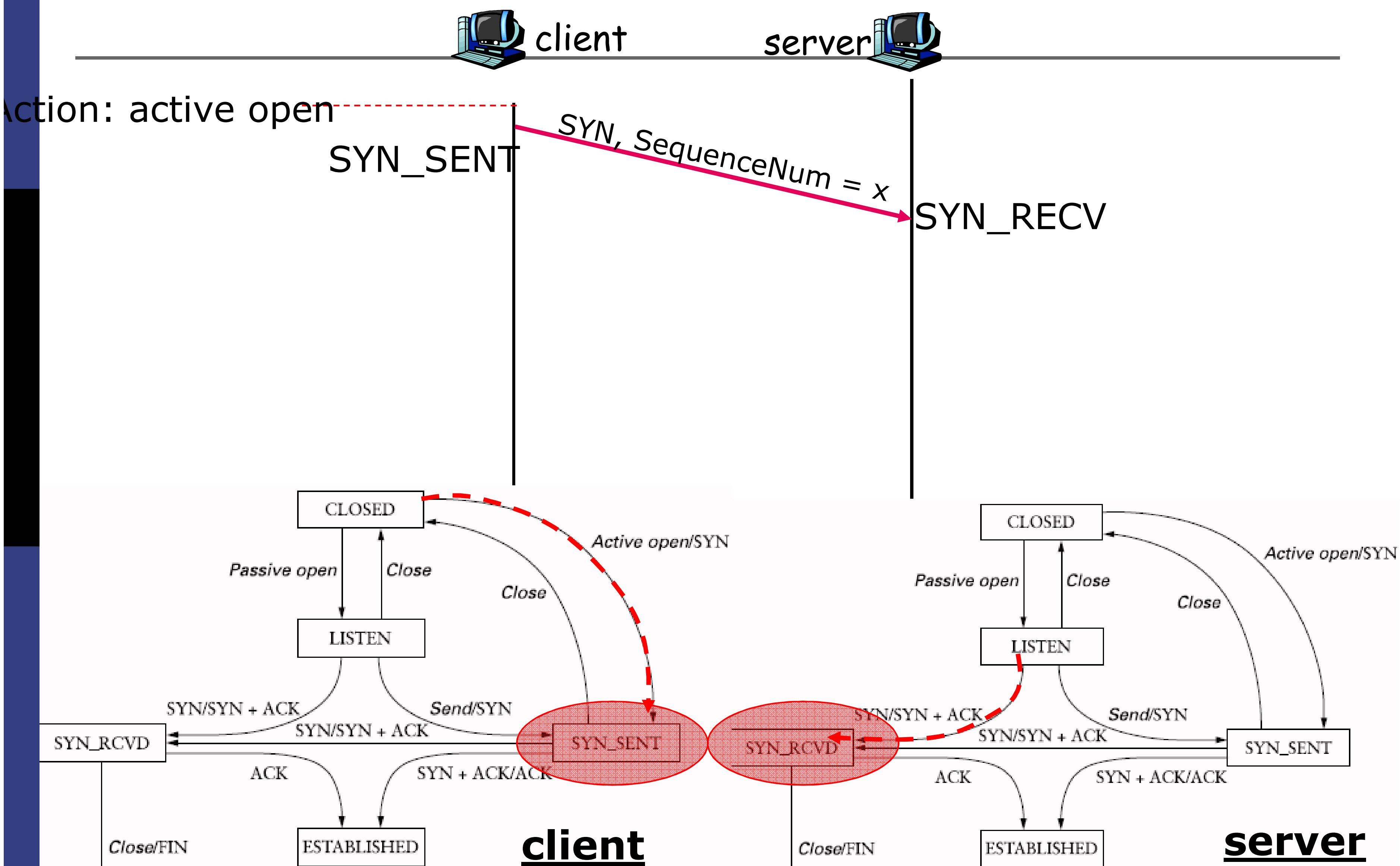

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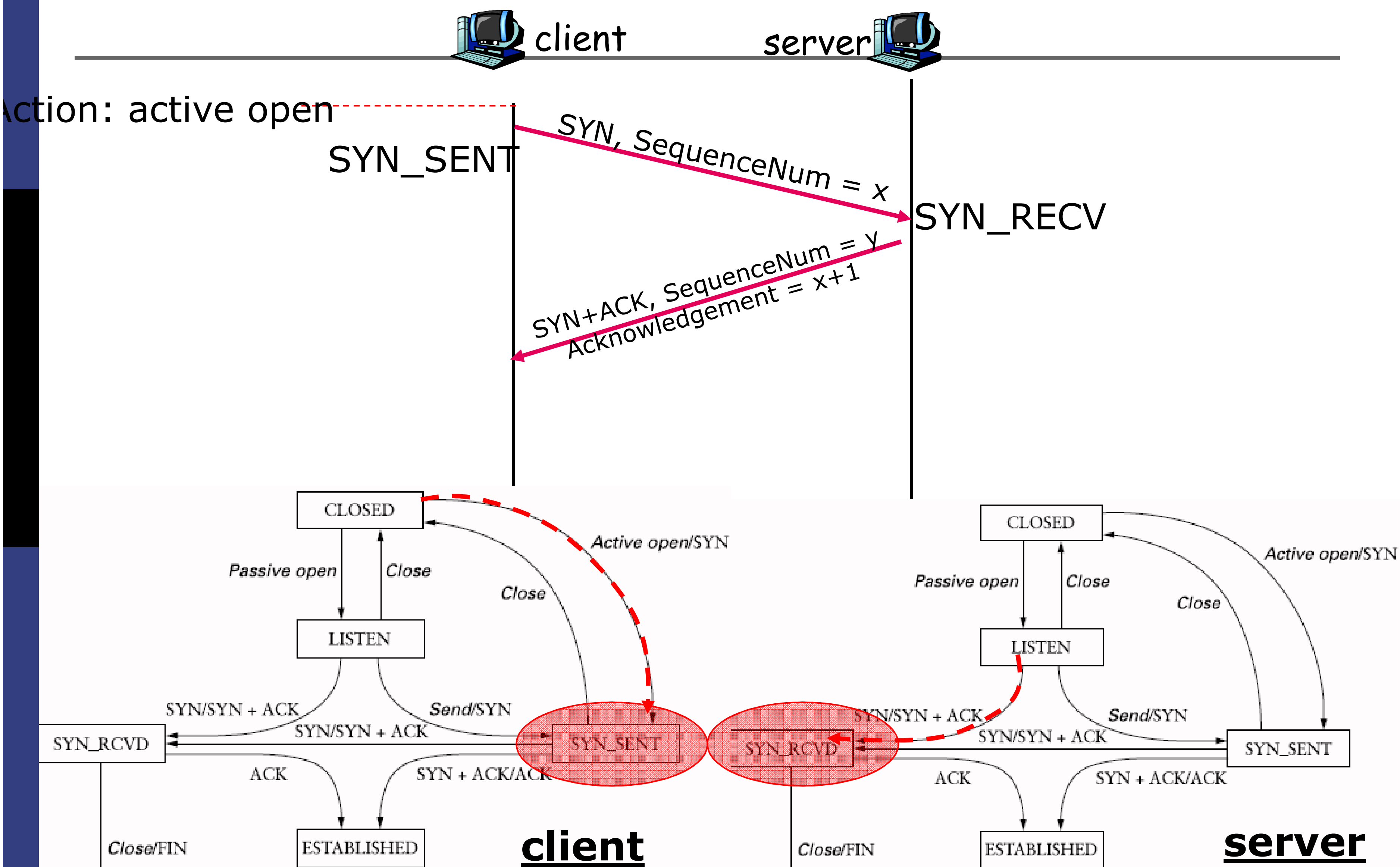

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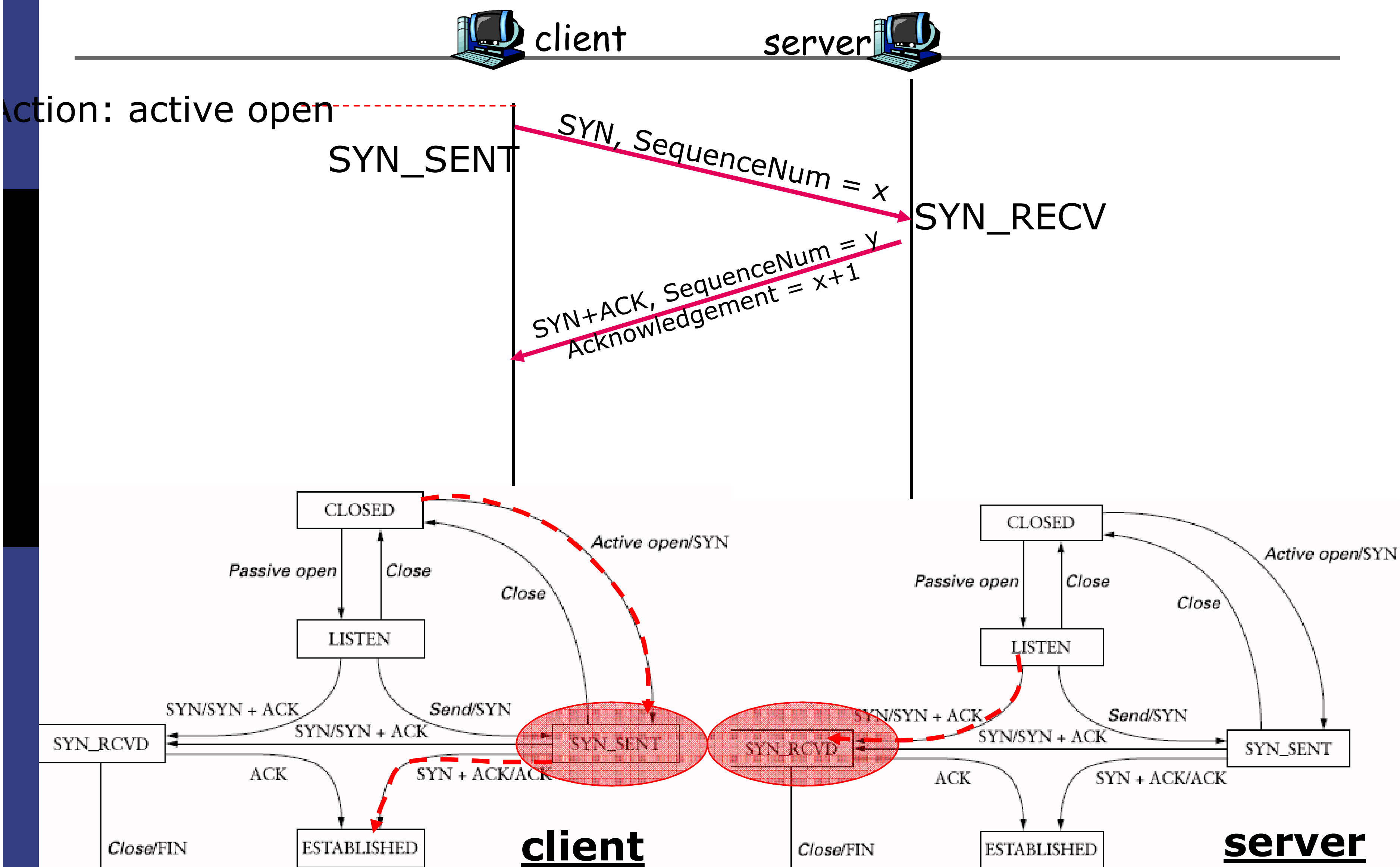

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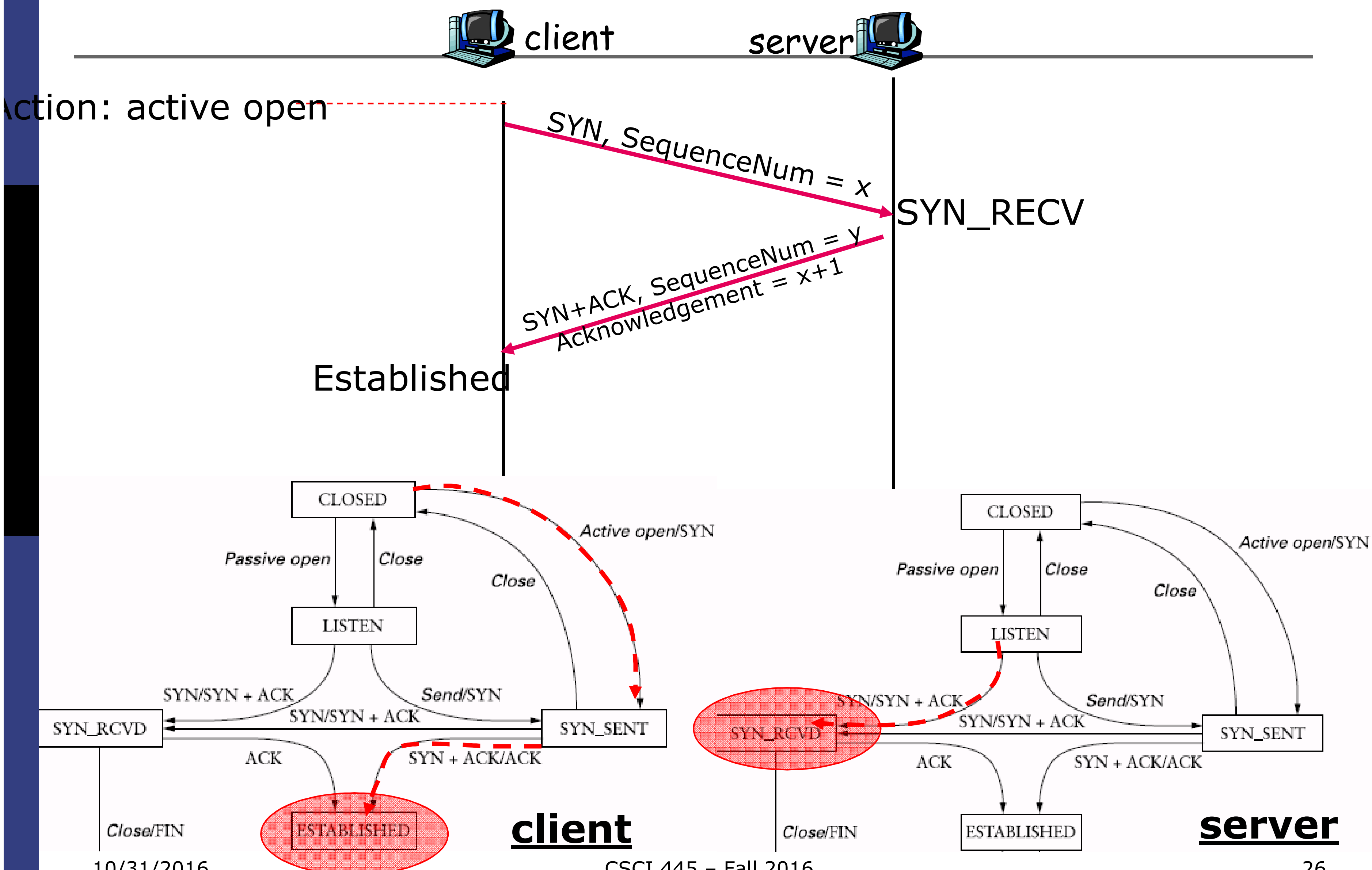

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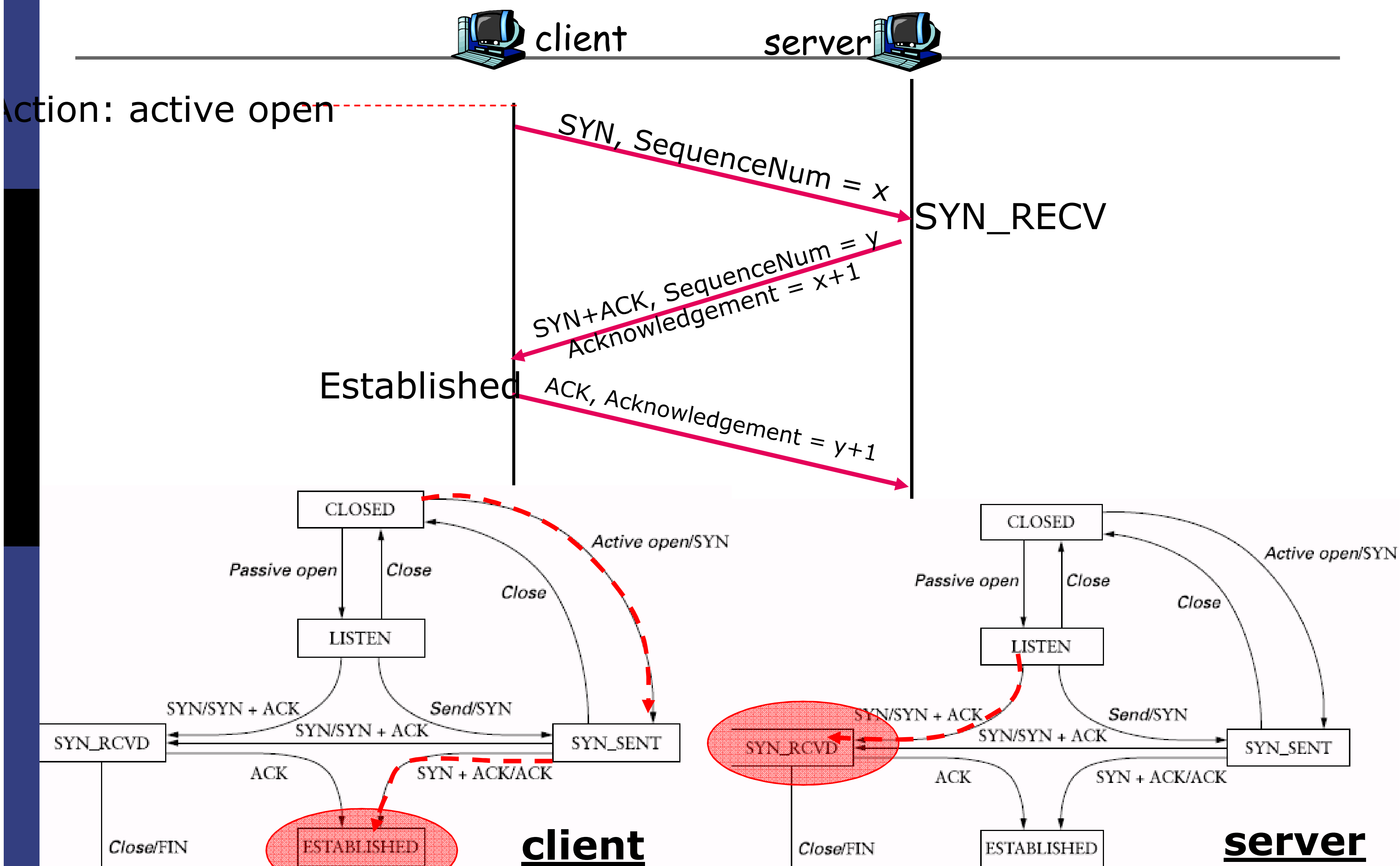

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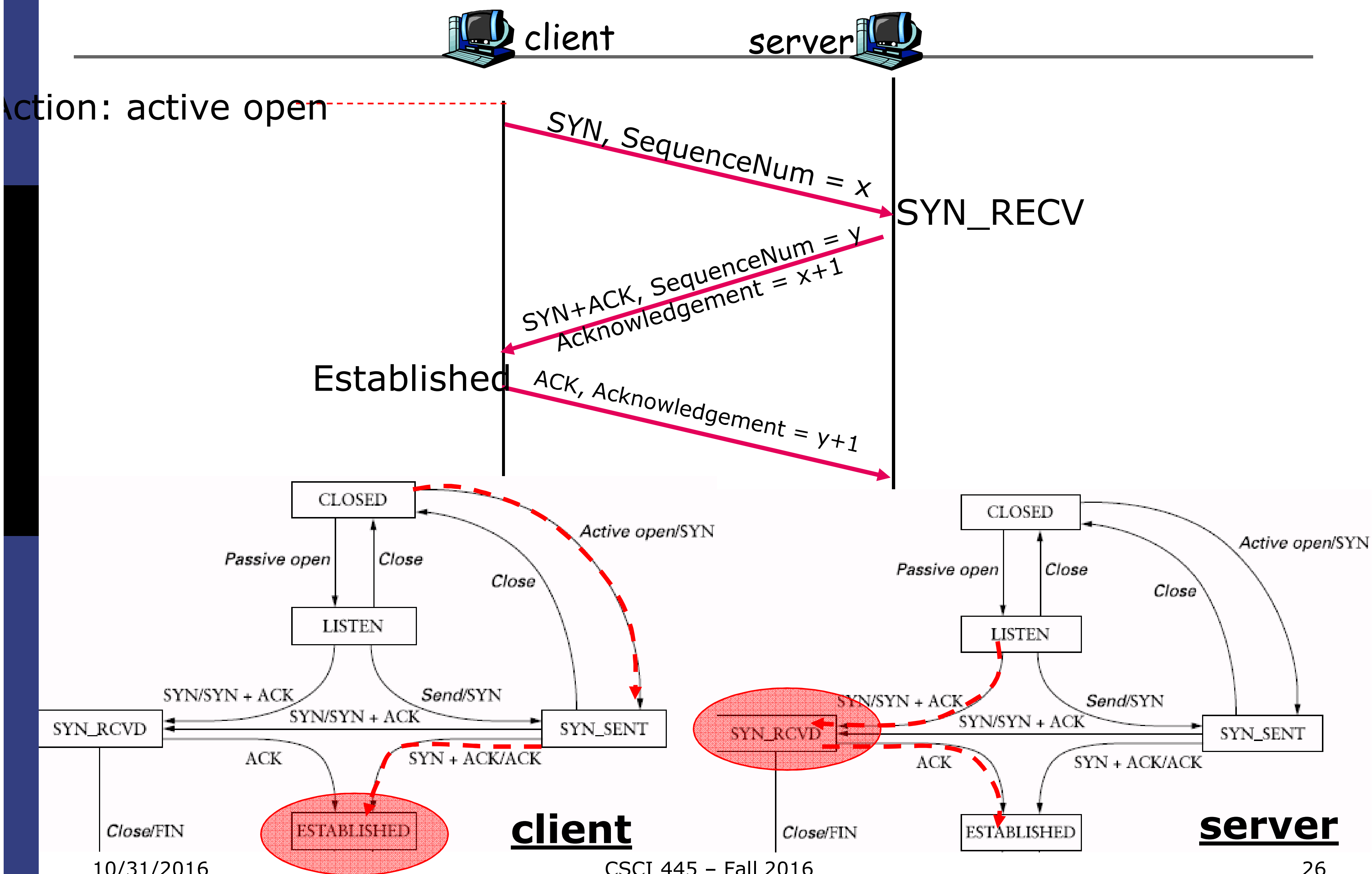

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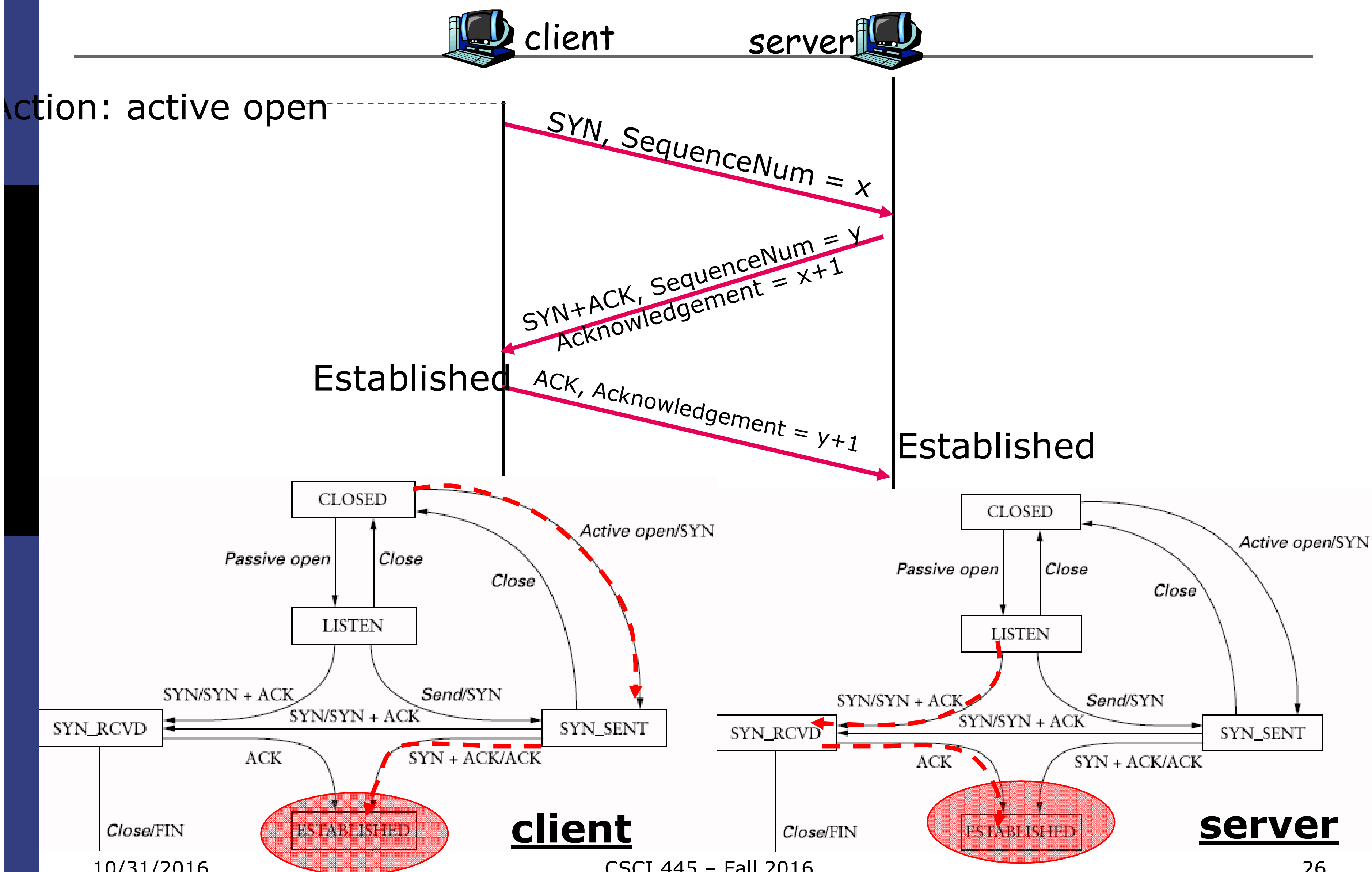
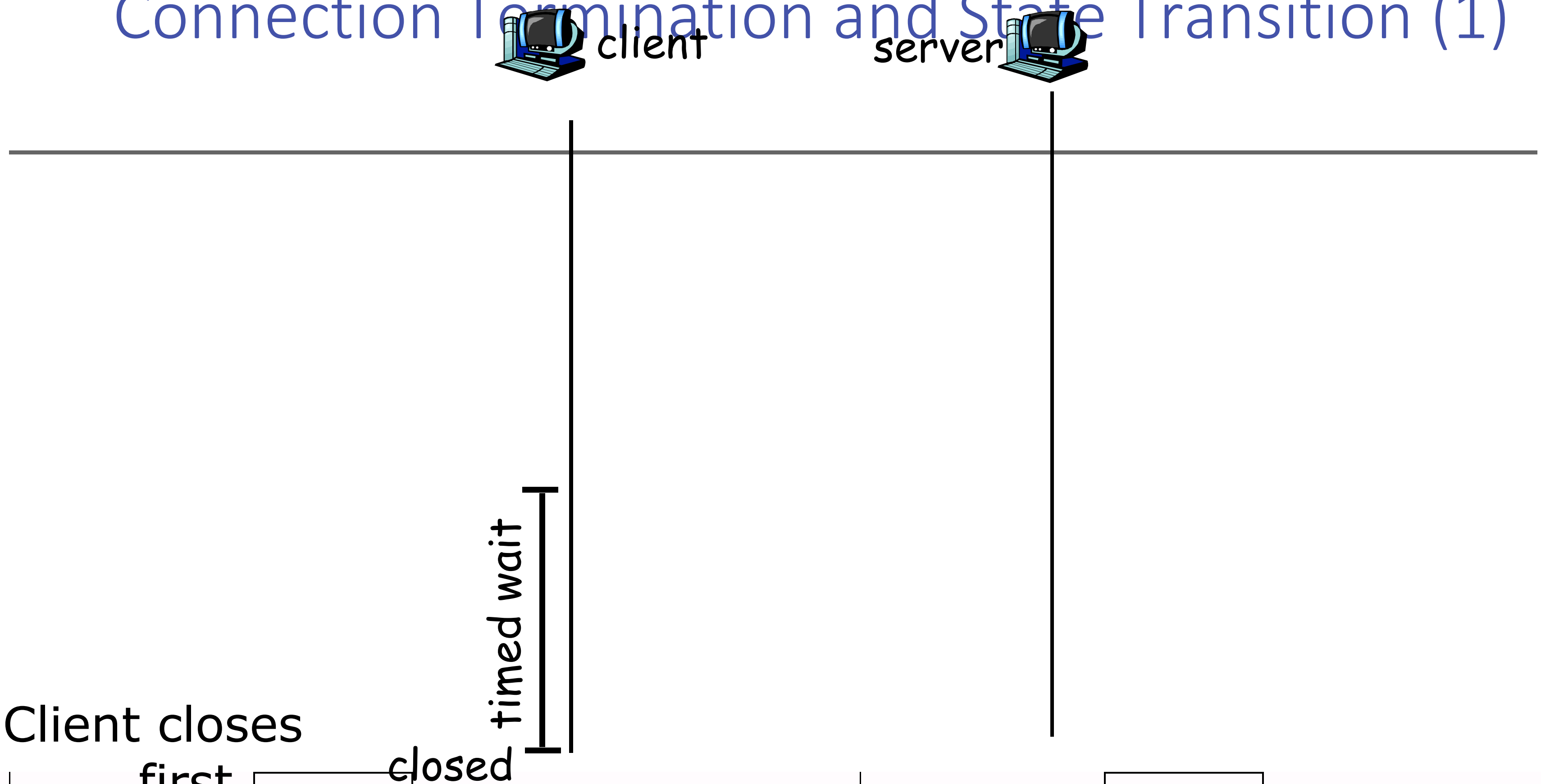
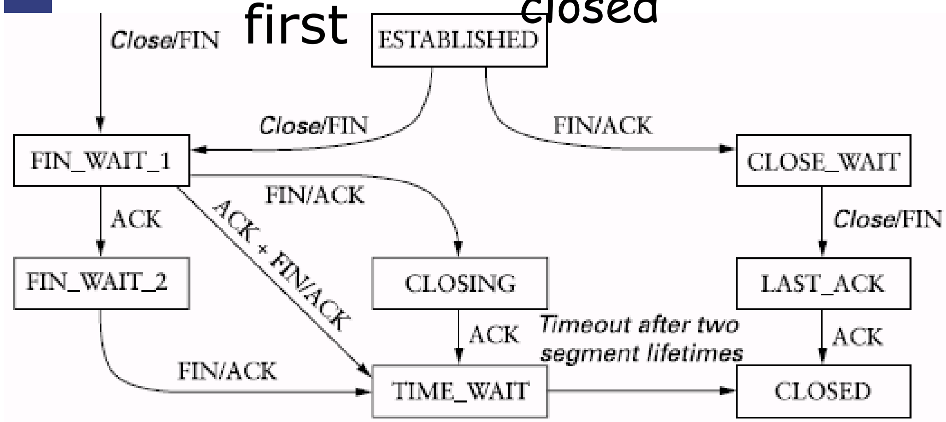
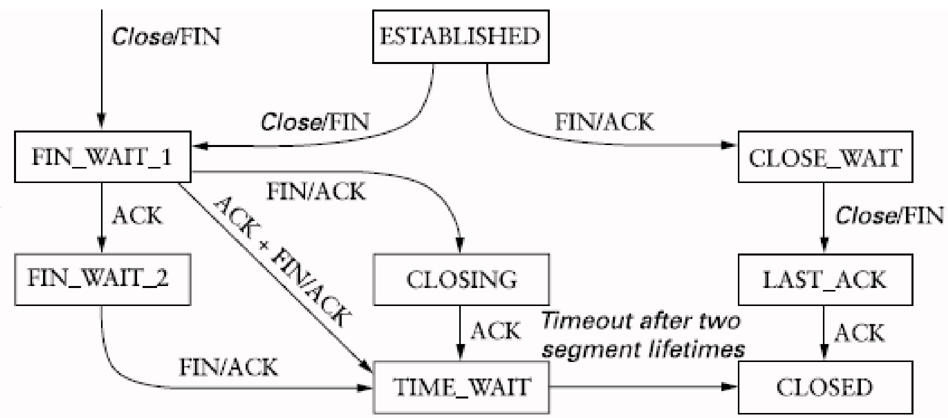


Connection Termination and State Transition (1)



Client closes first





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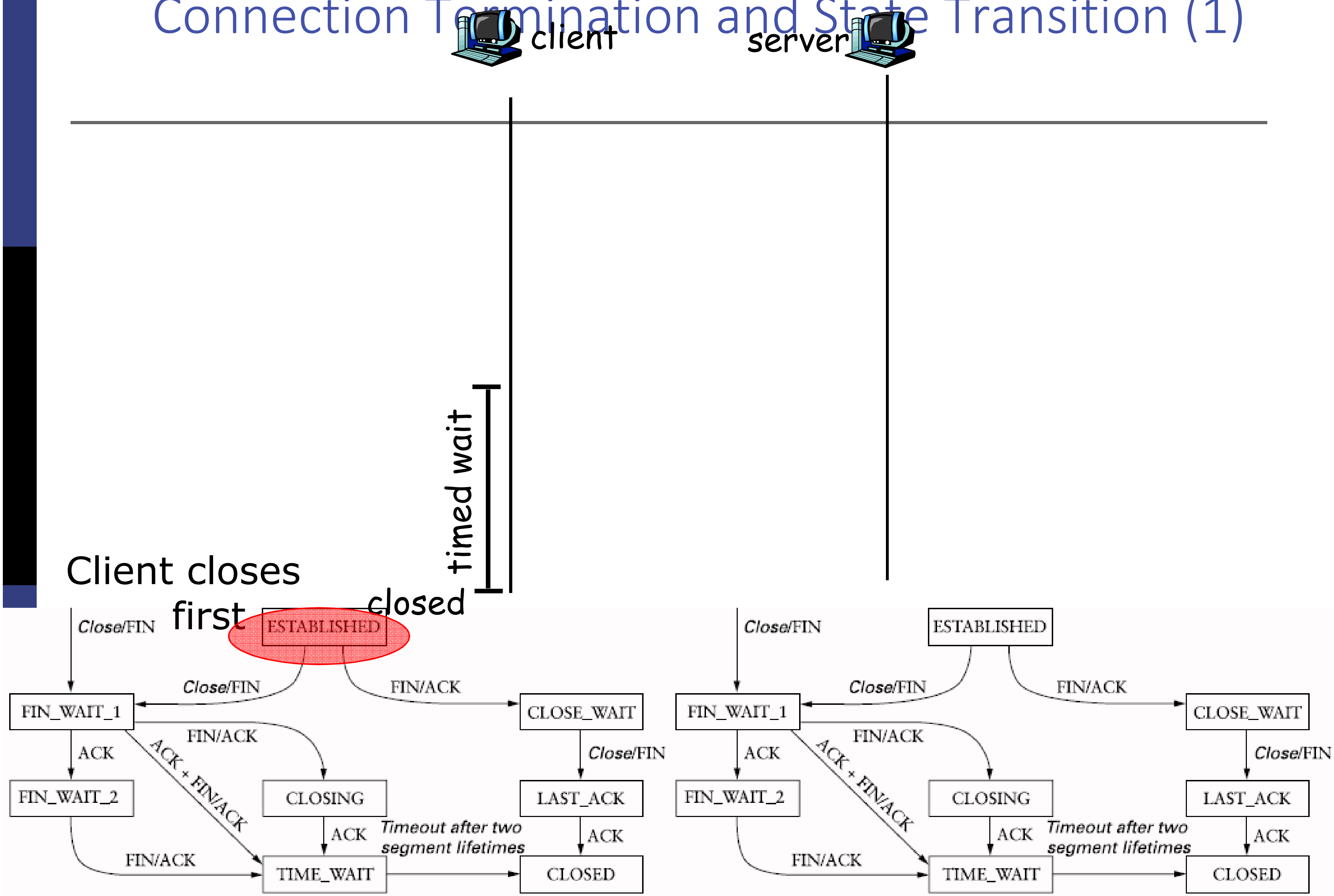

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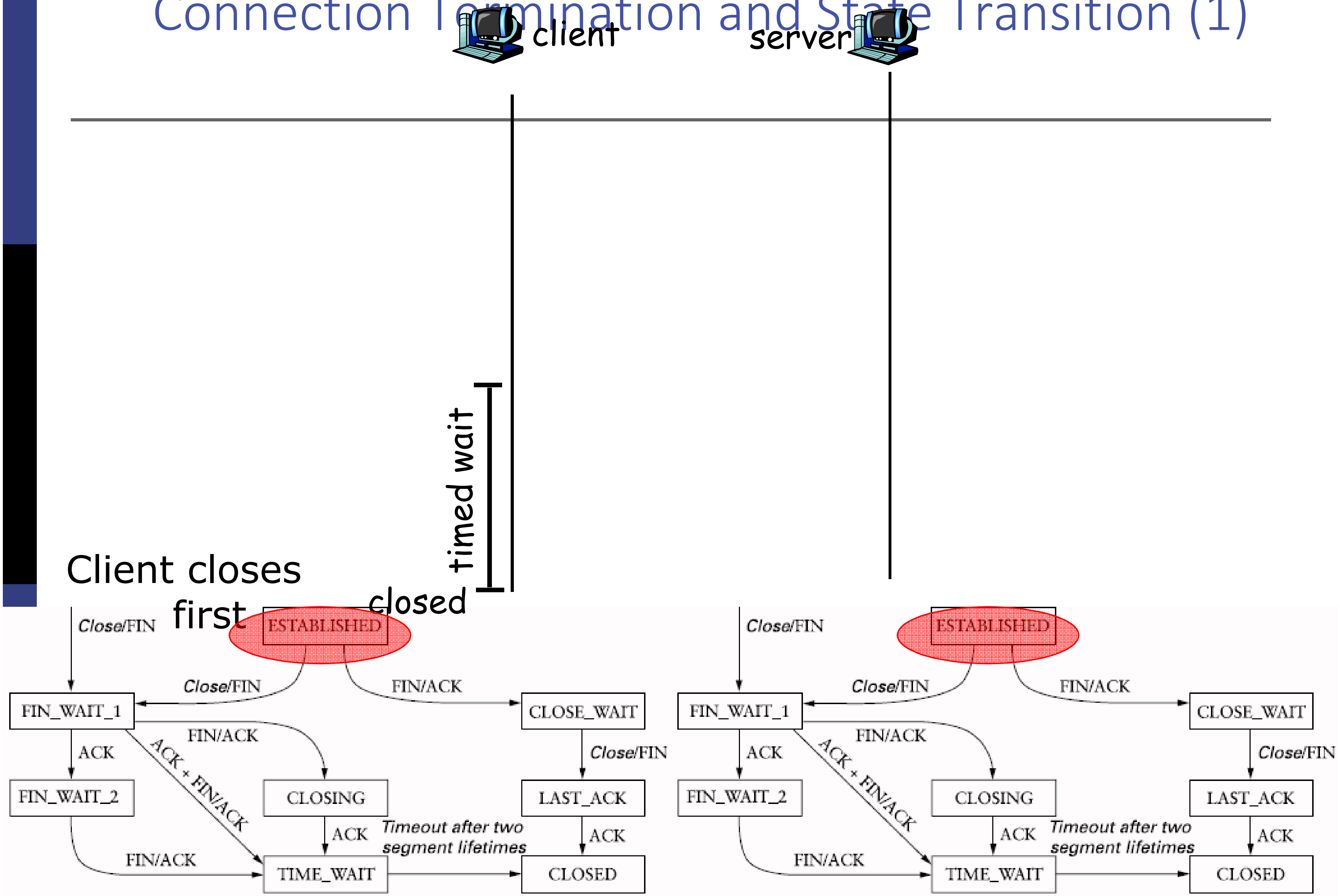

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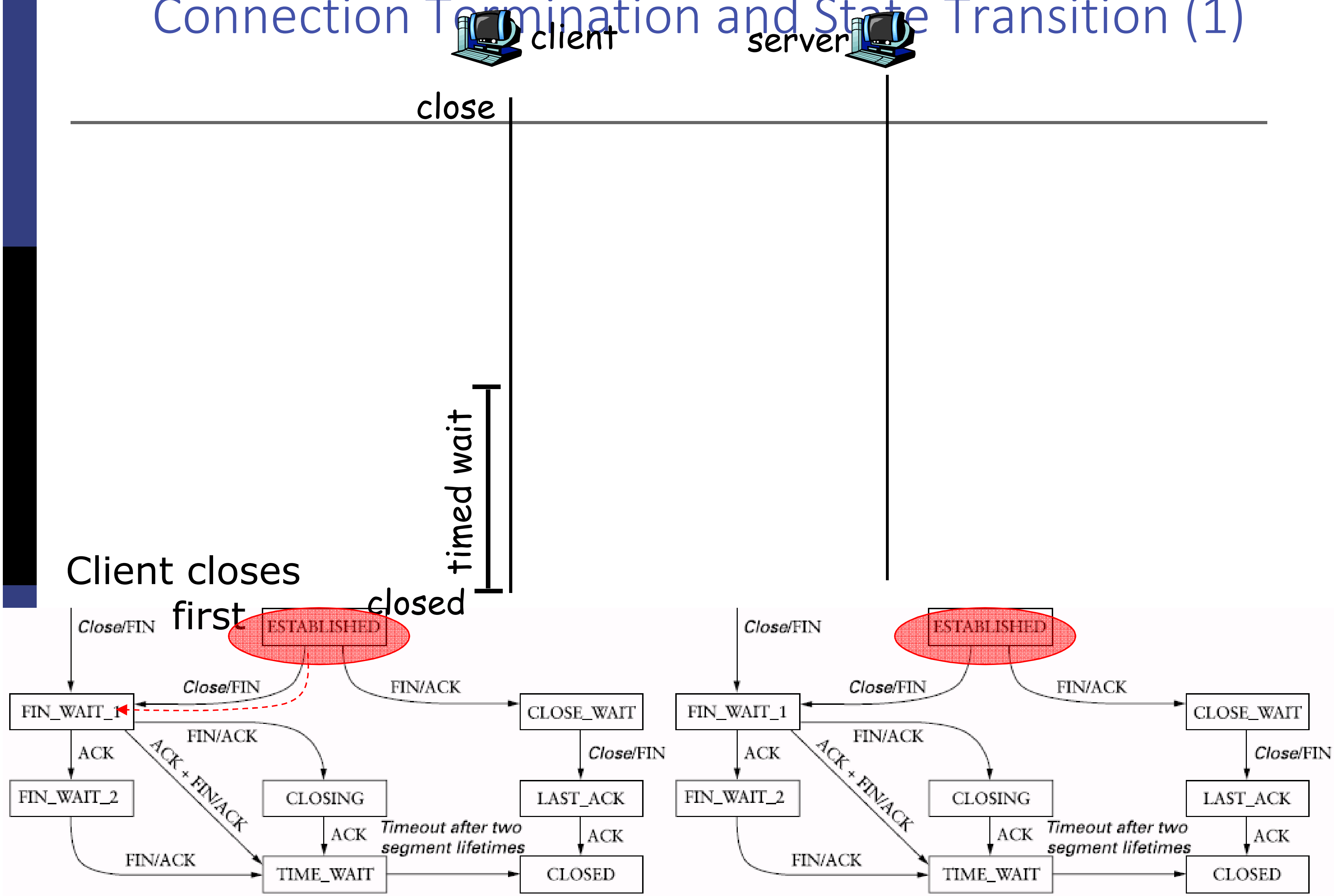

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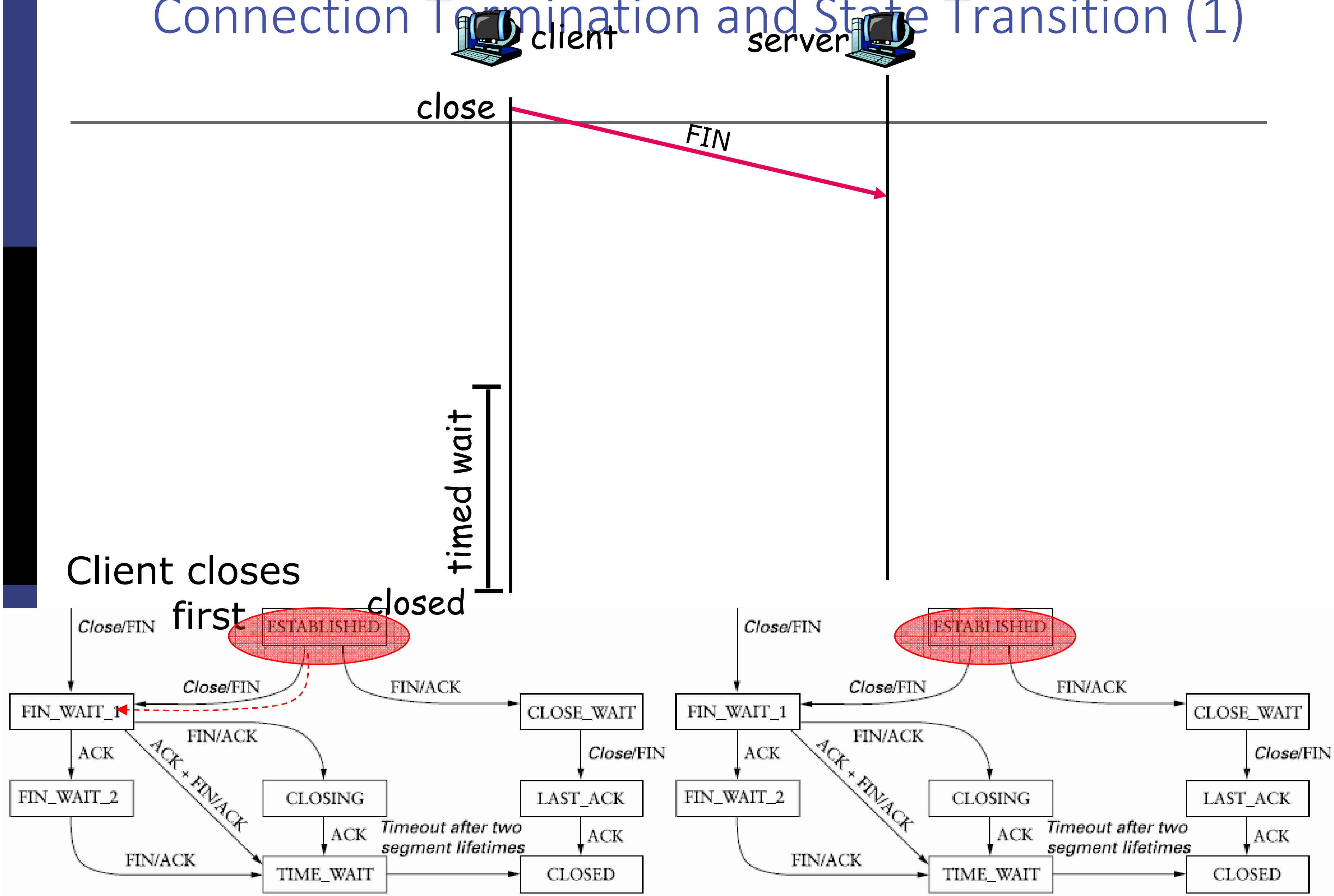

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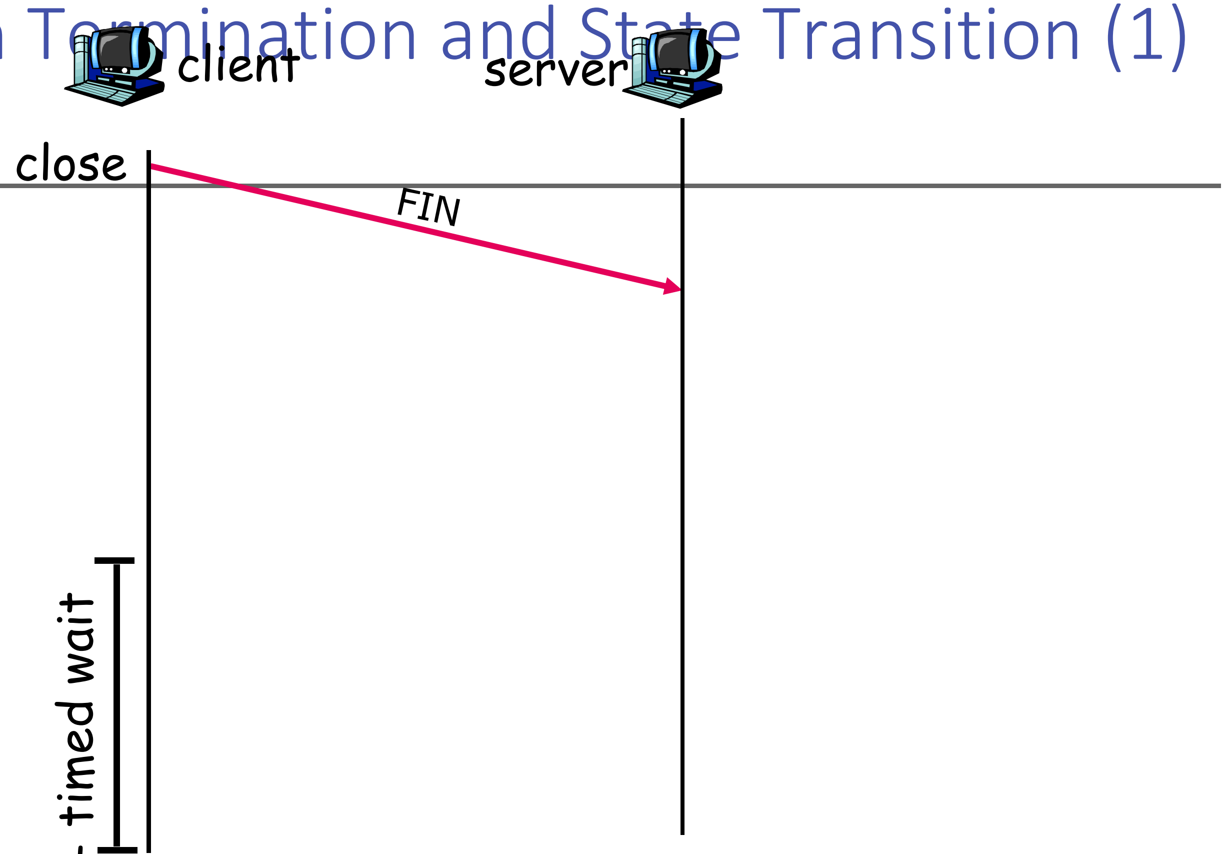

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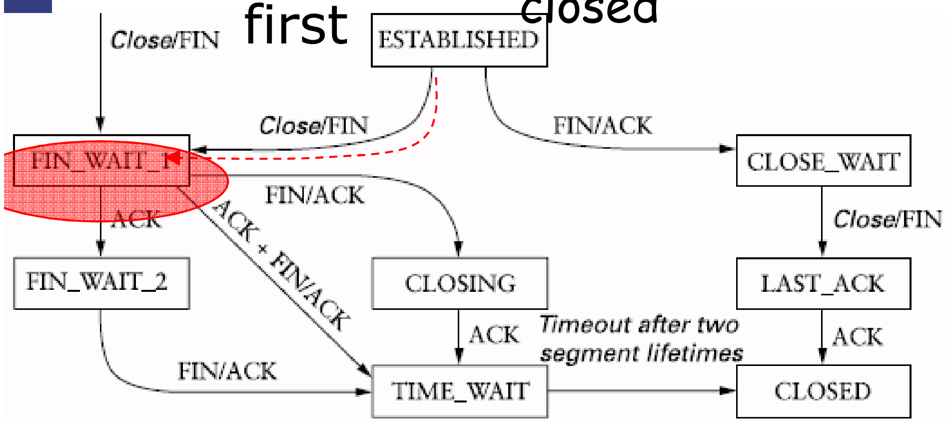

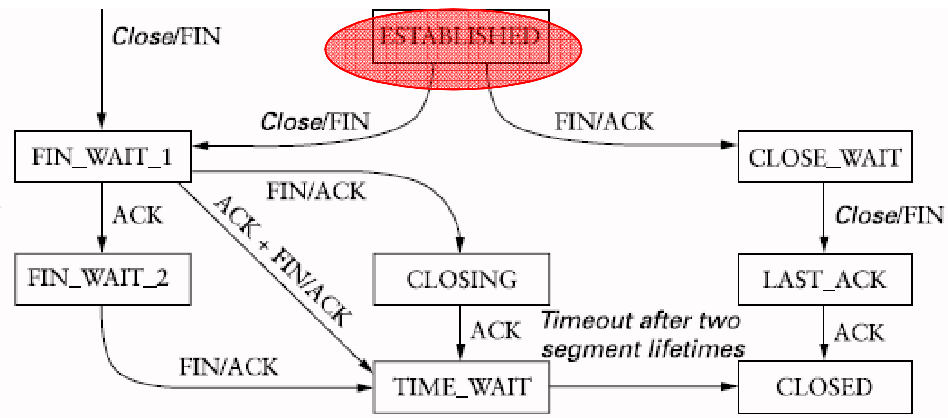

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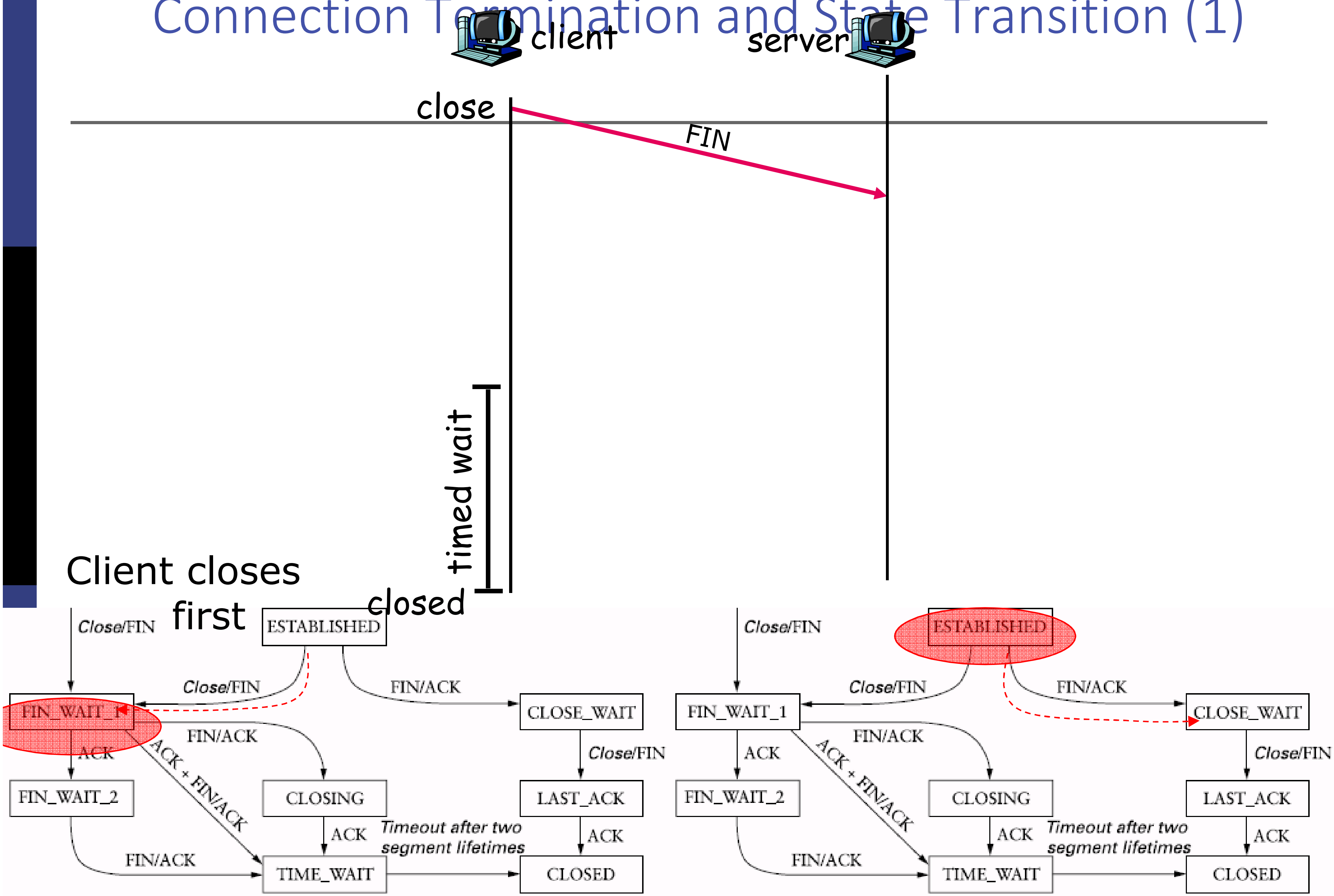

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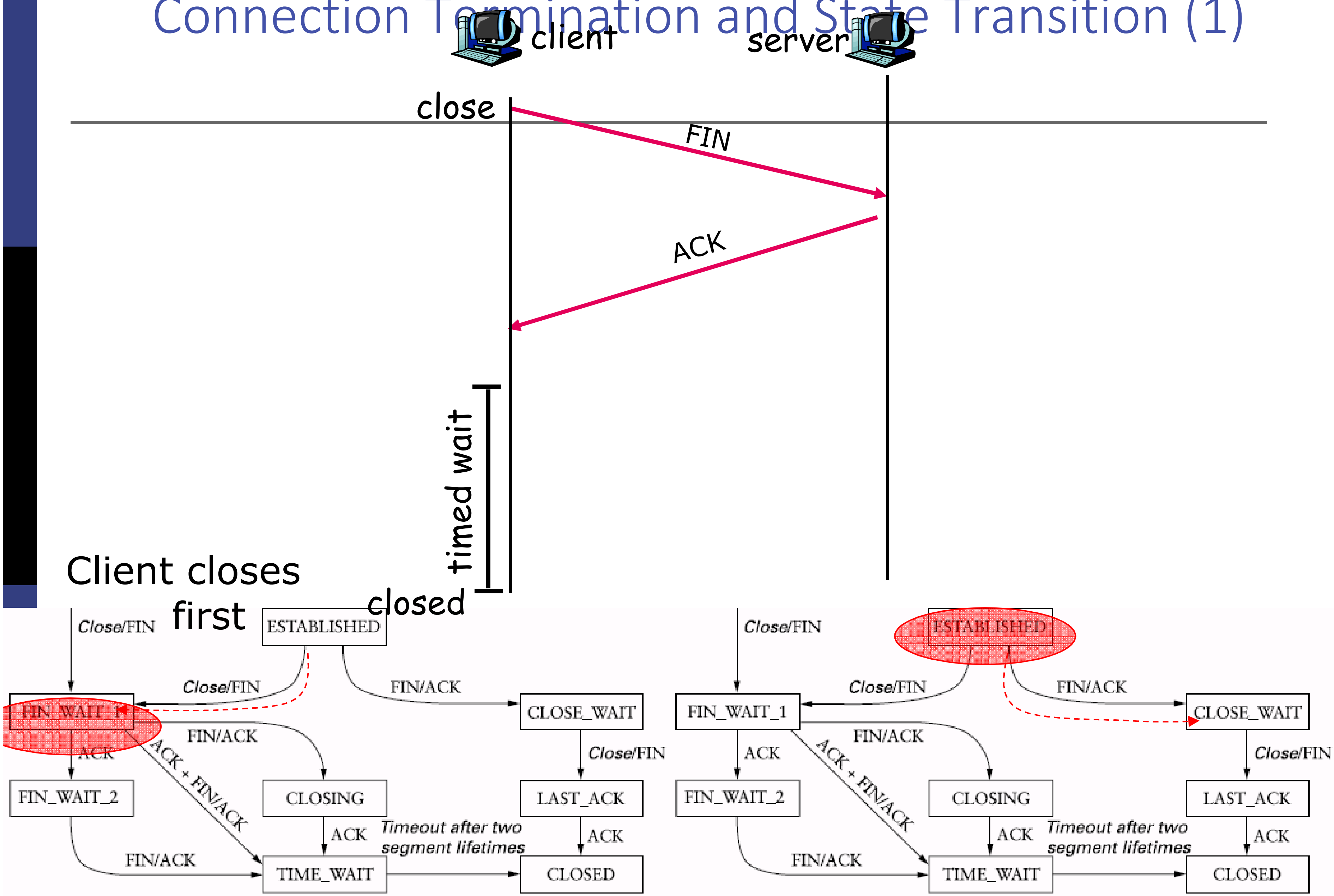

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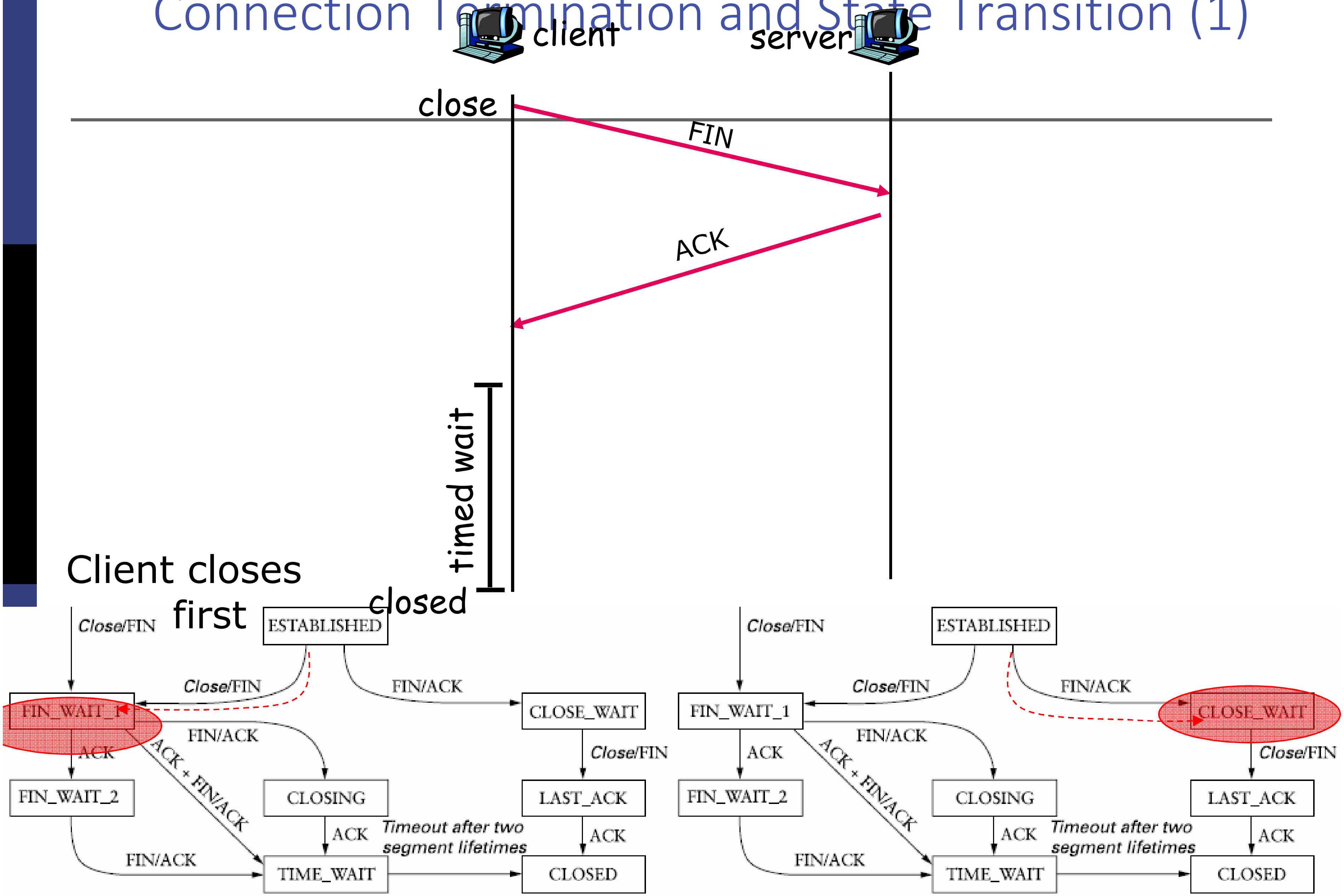

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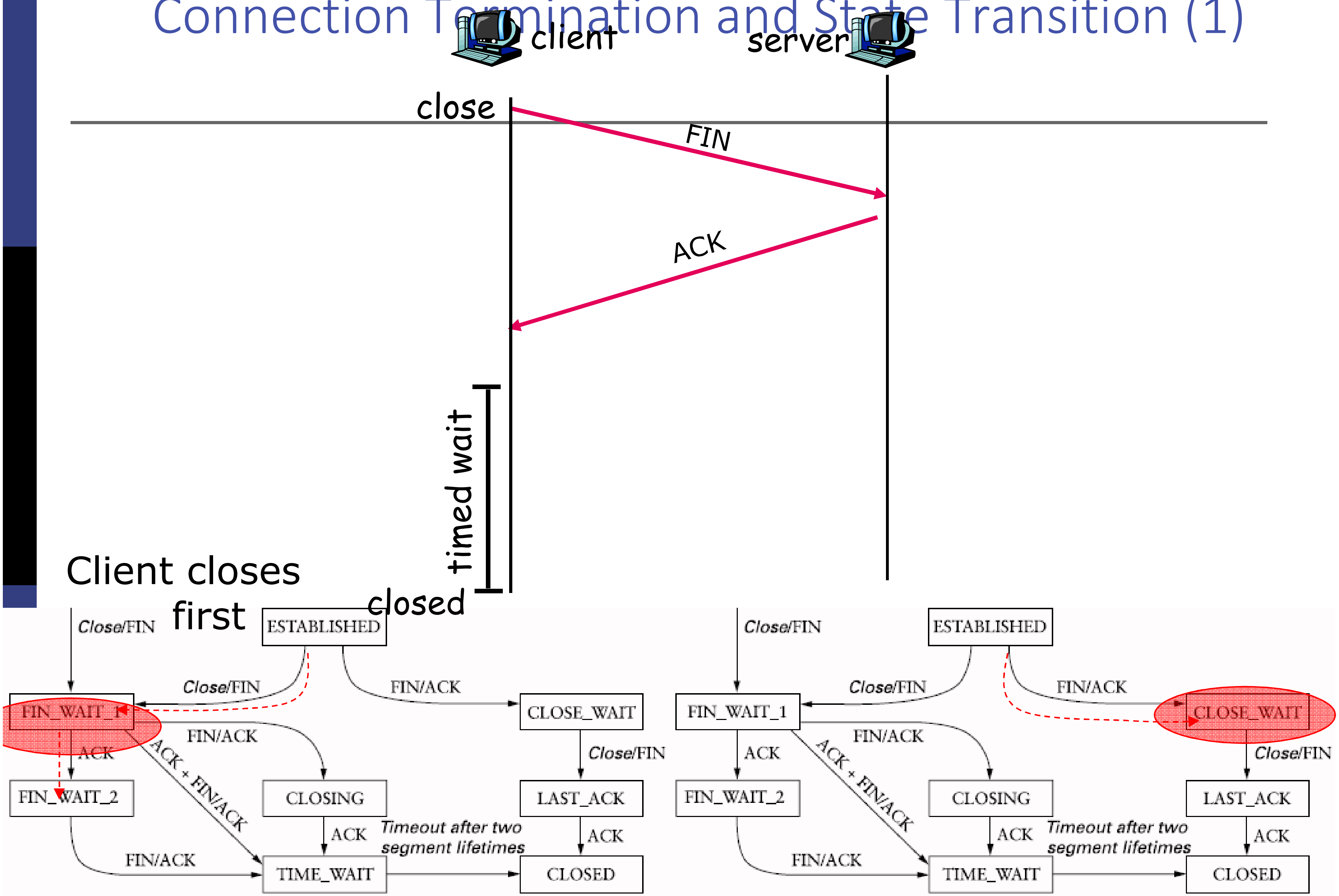

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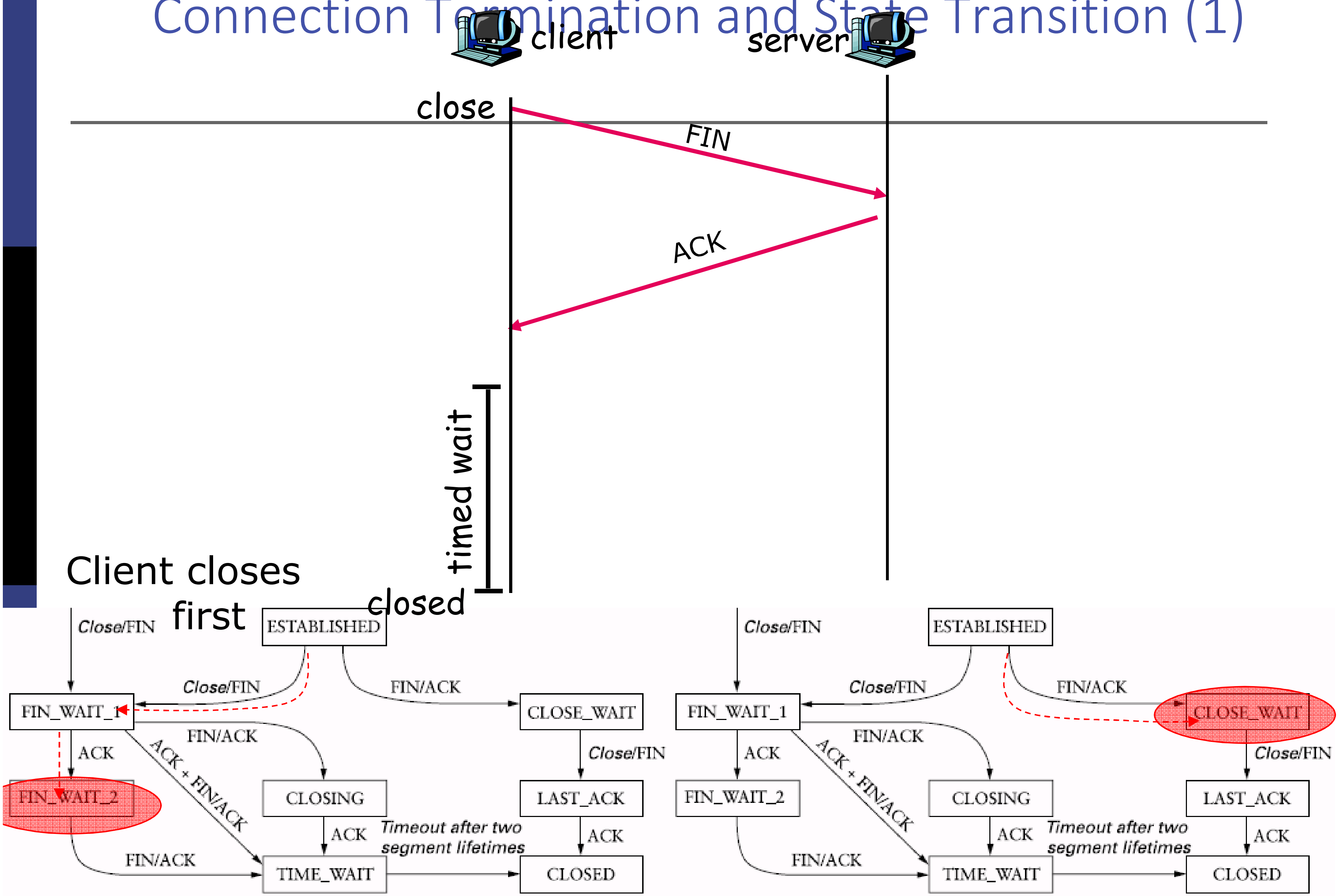

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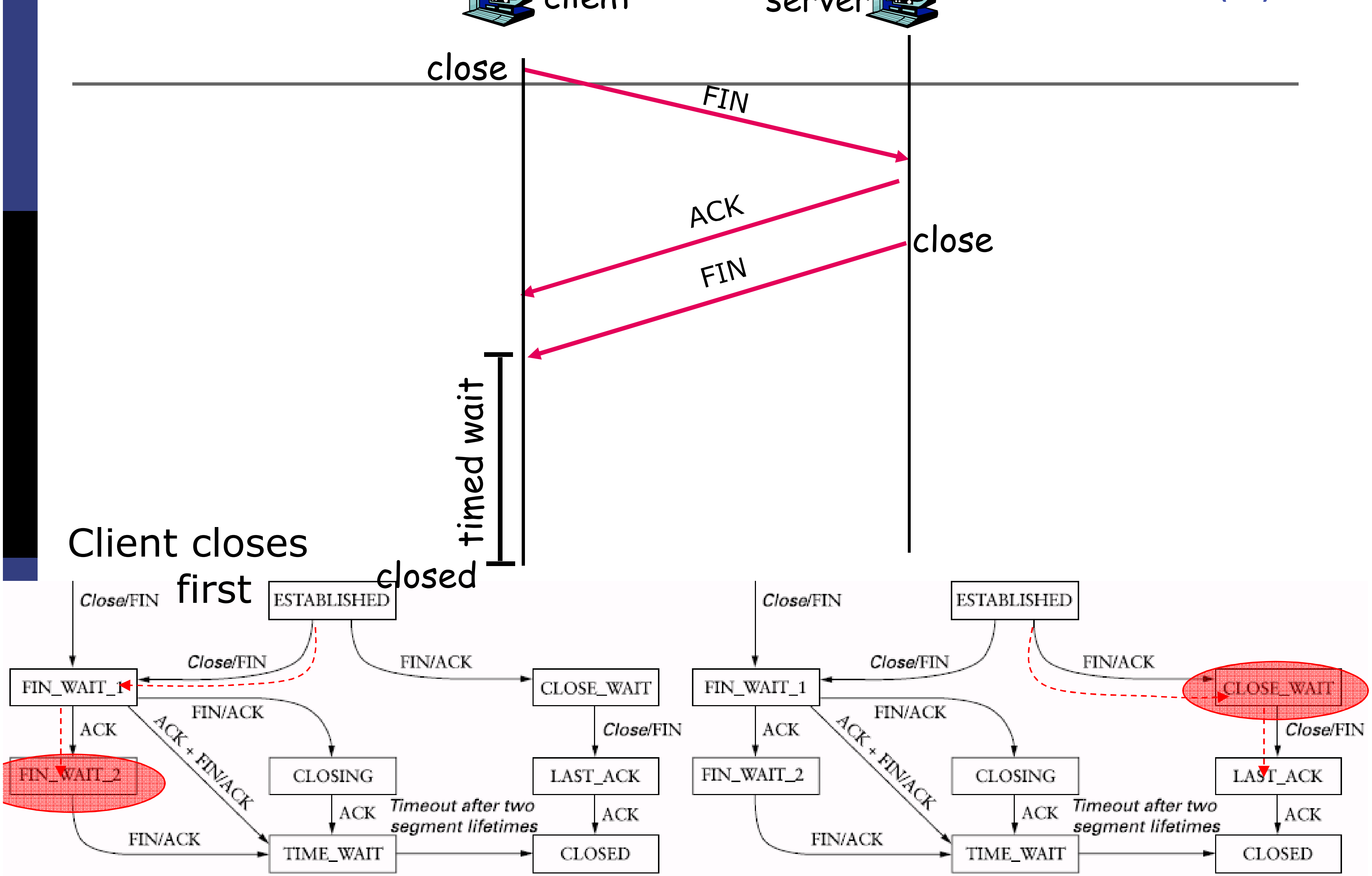
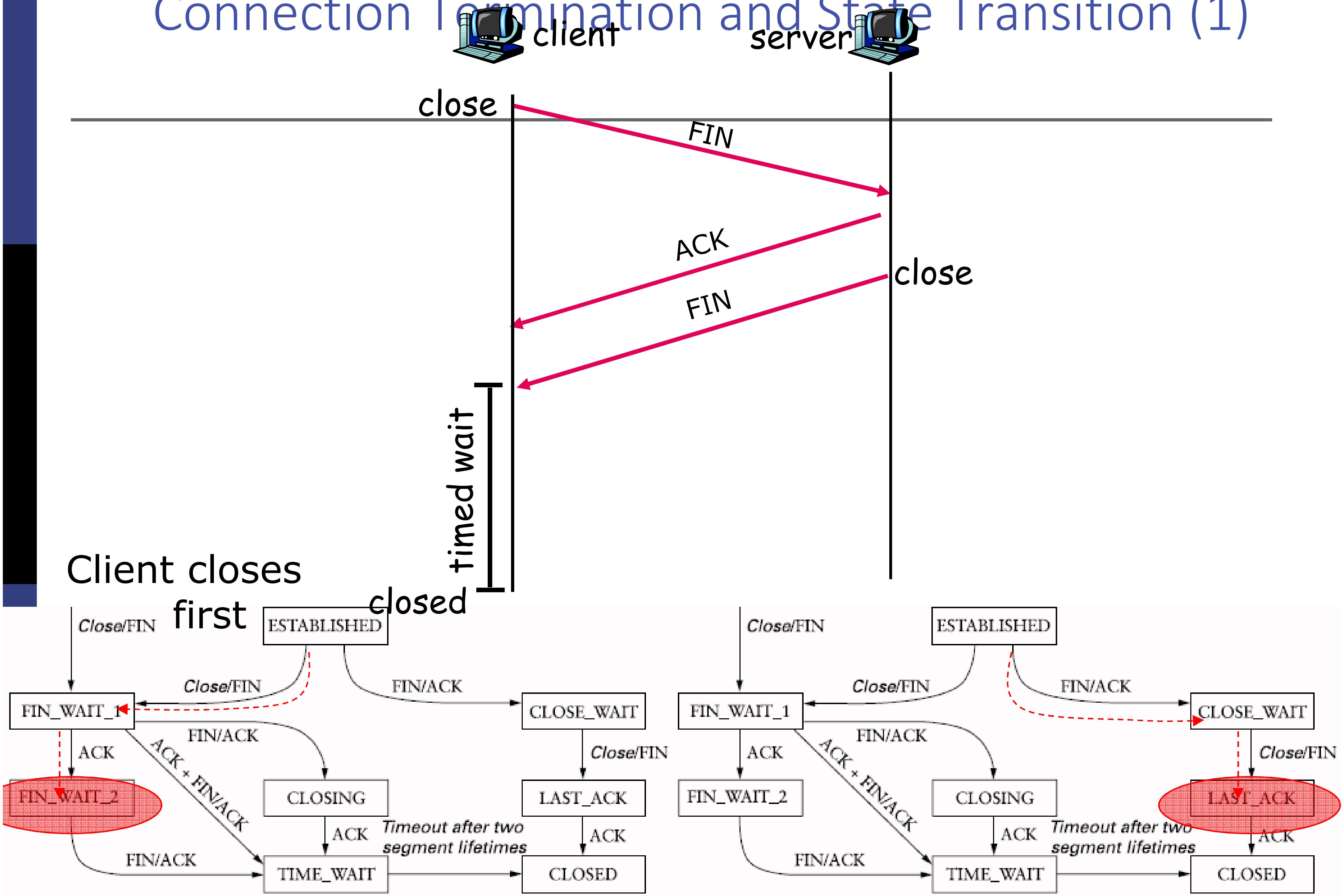


Connection Termination and State Transition (1)



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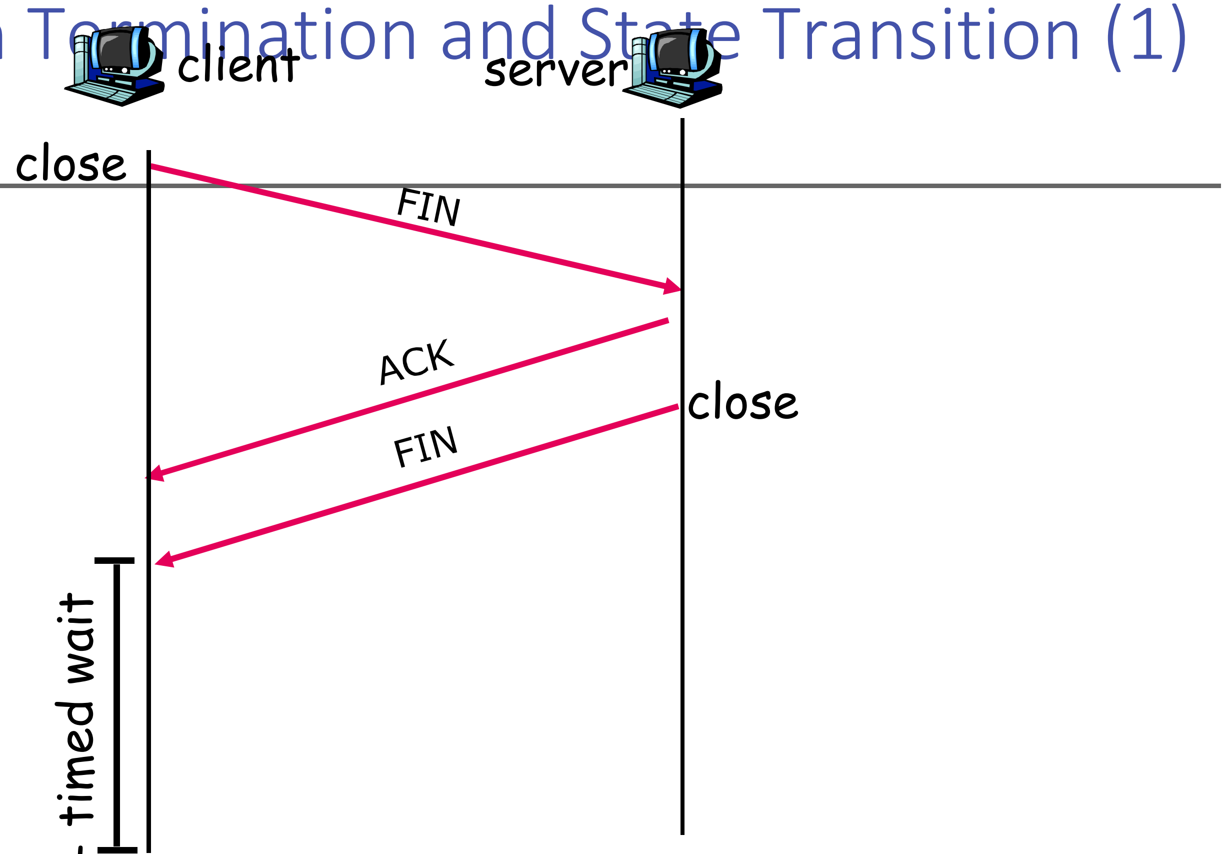
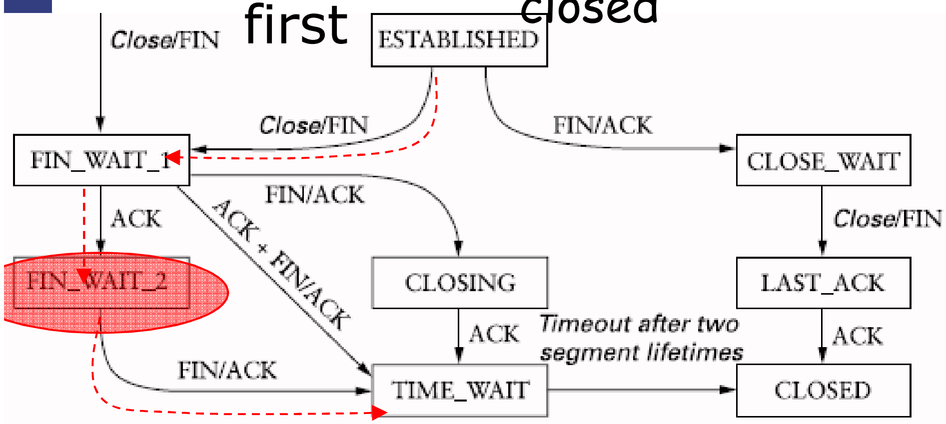
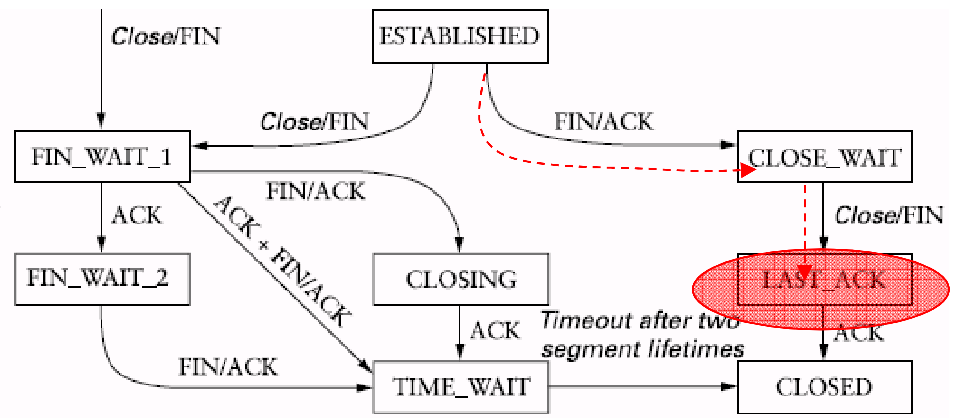
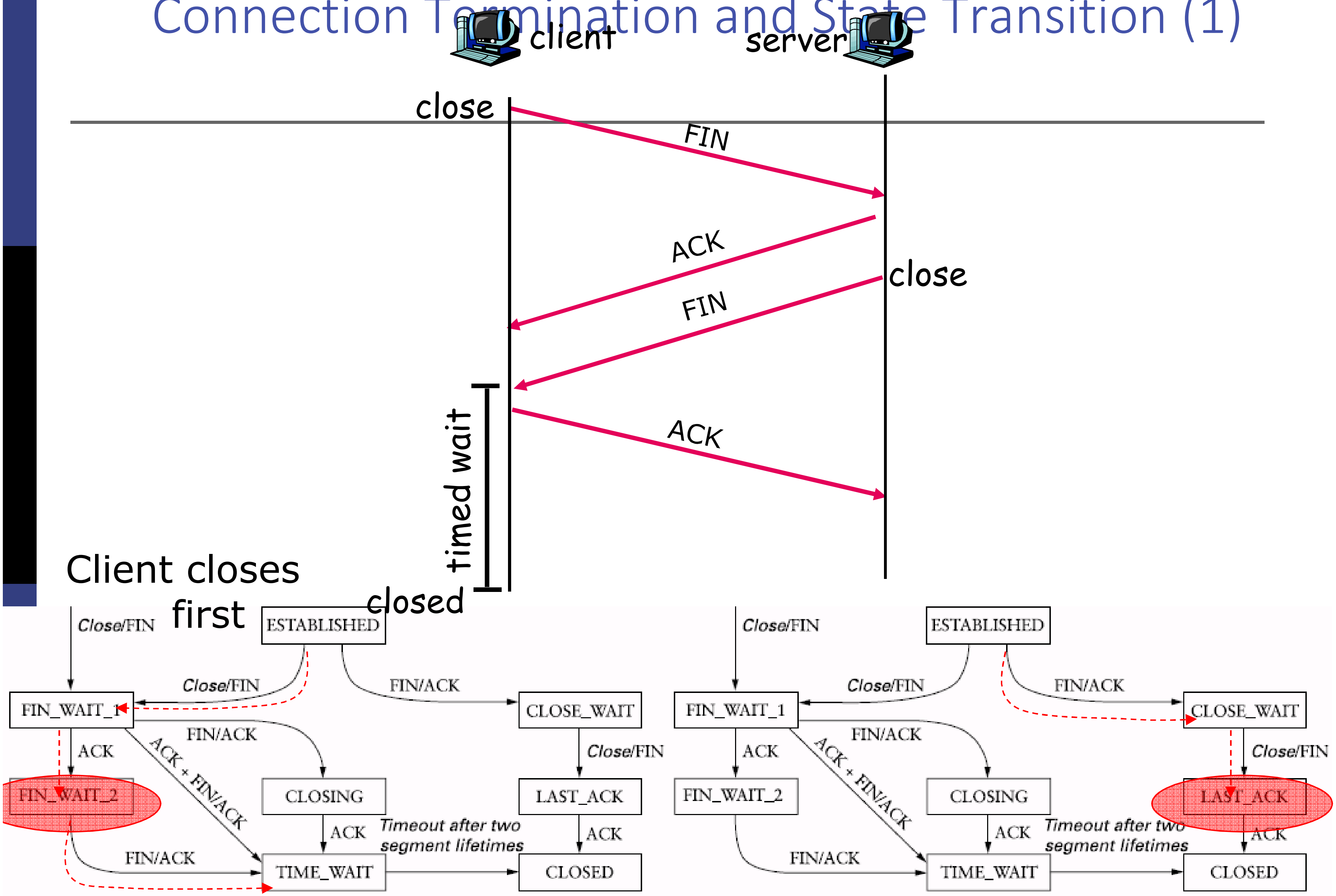


Client closes first





Connection Termination and State Transition (1)



Client closes first

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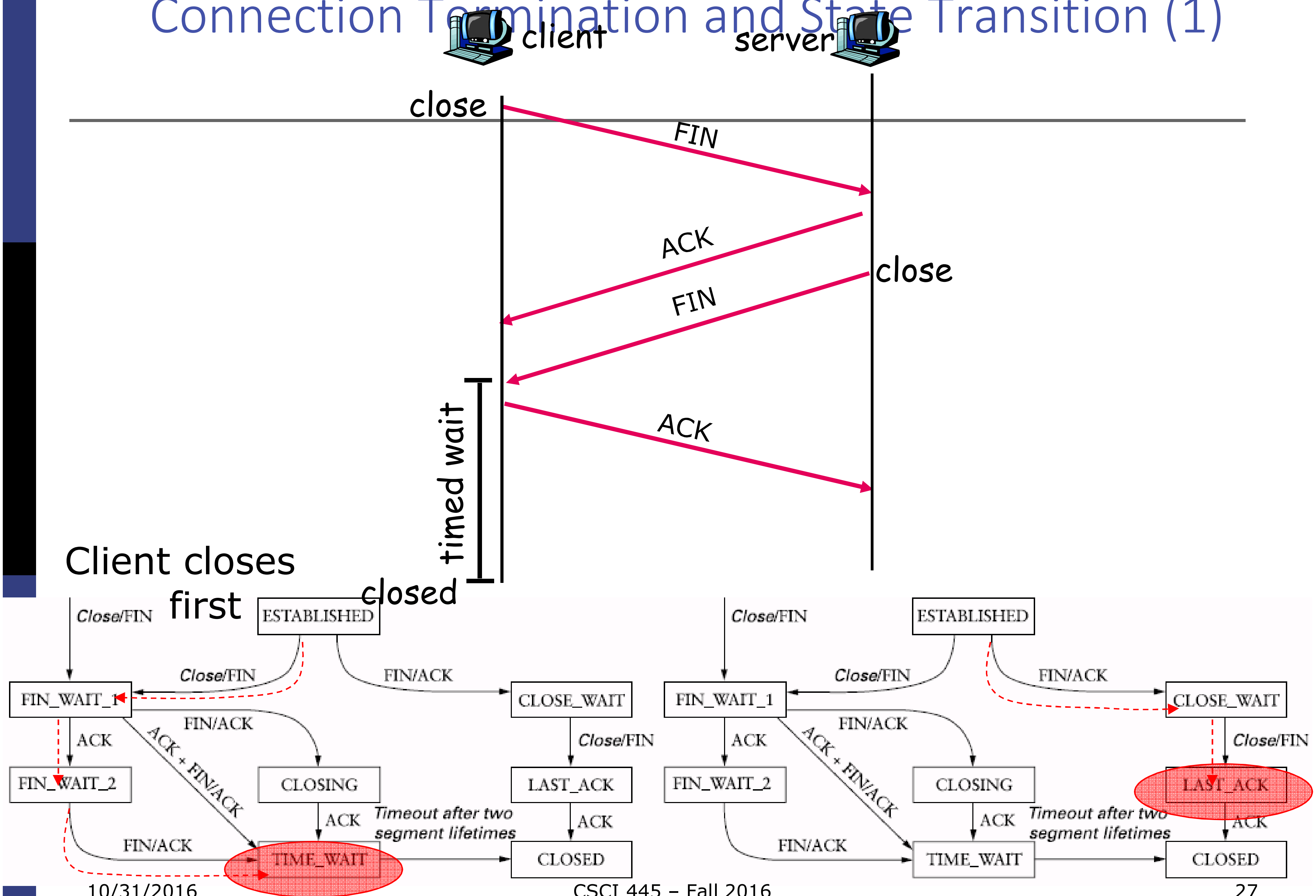

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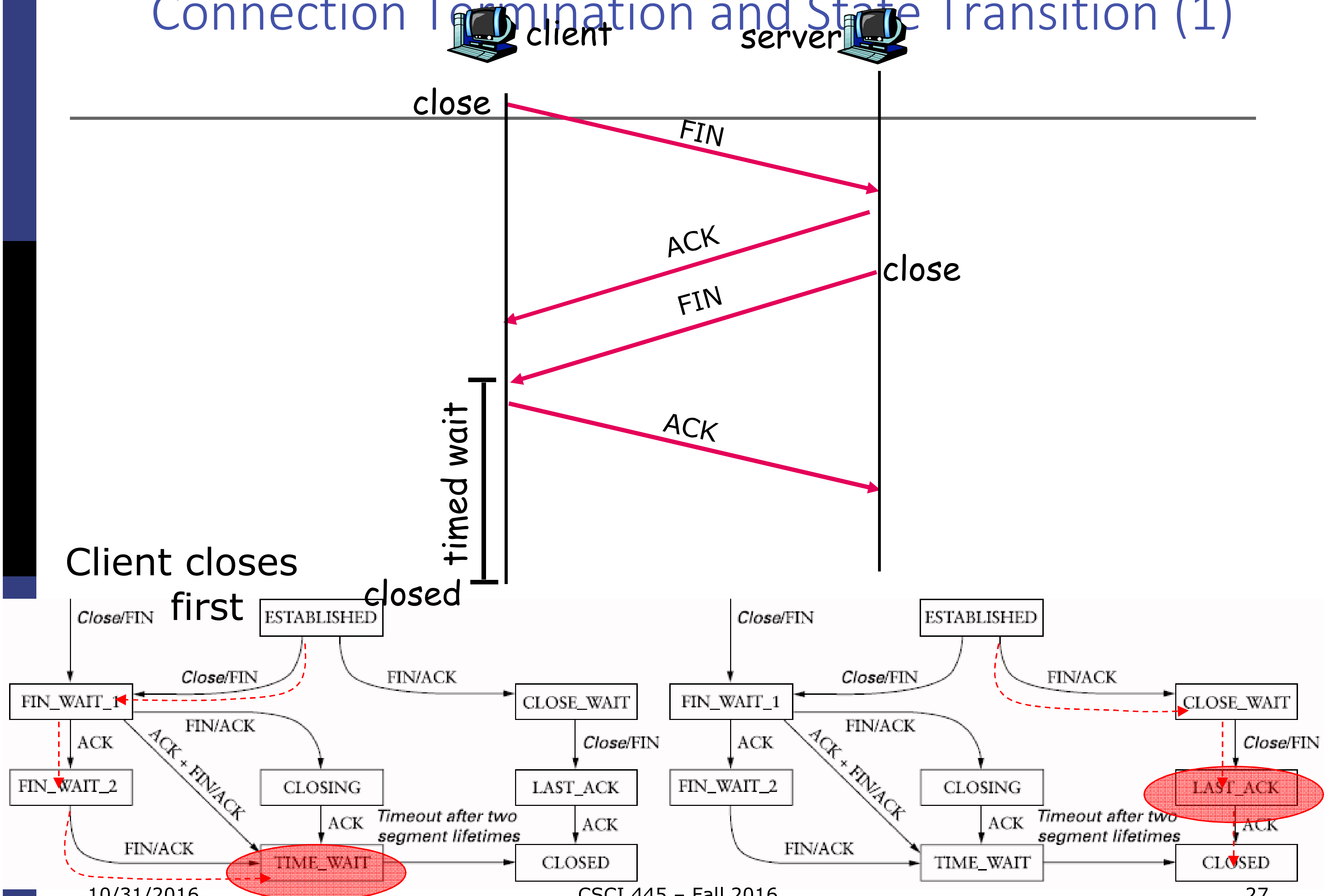

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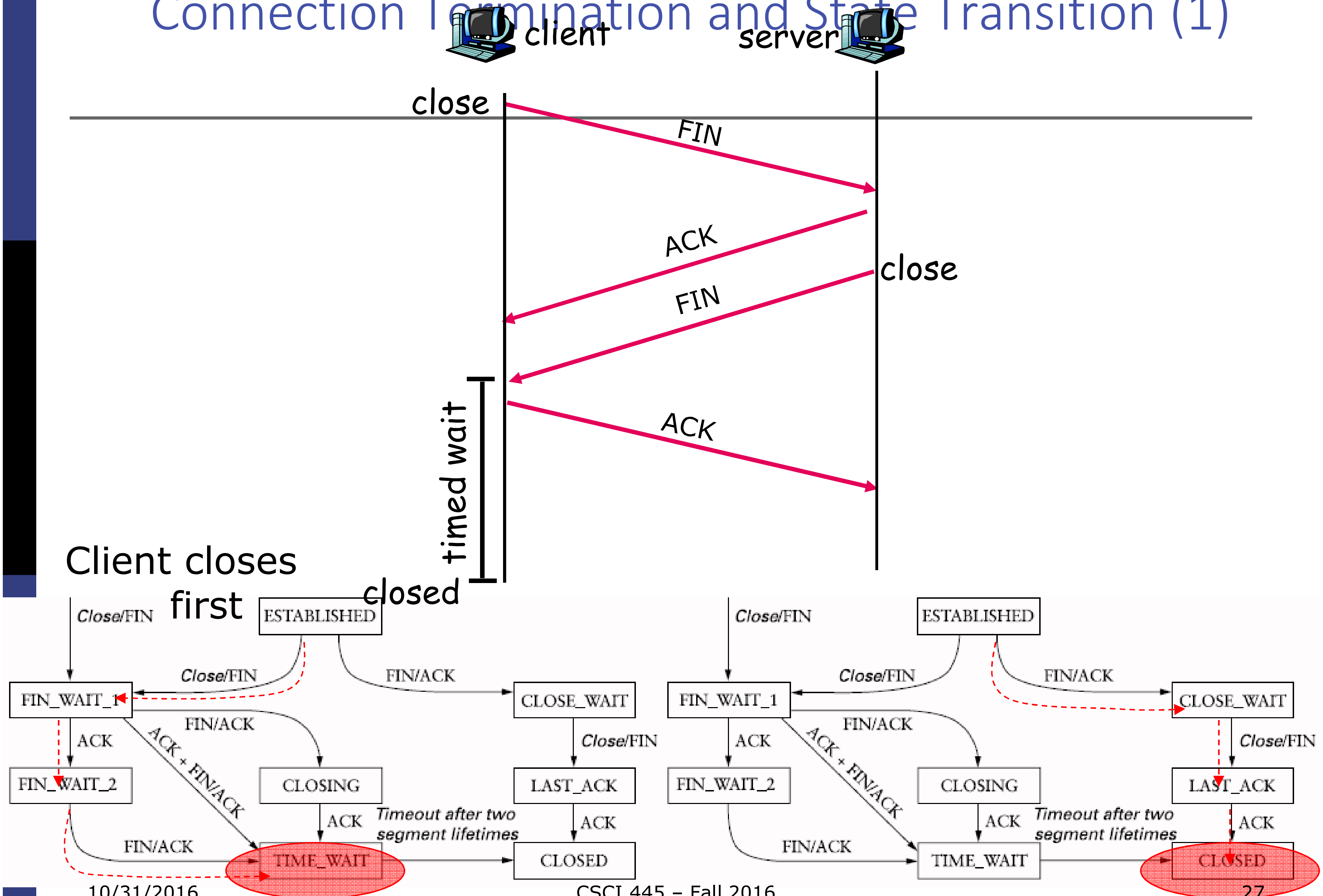

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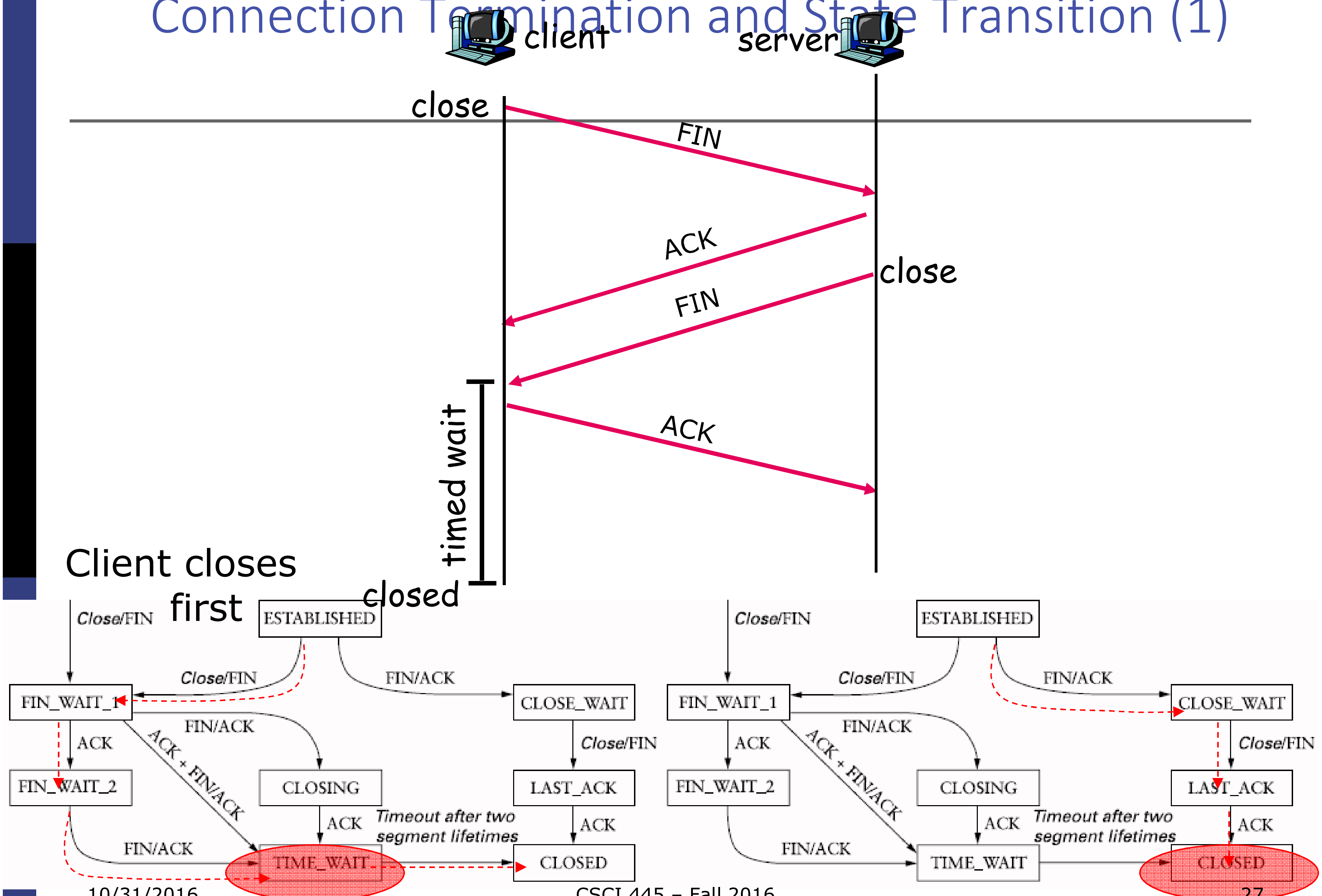

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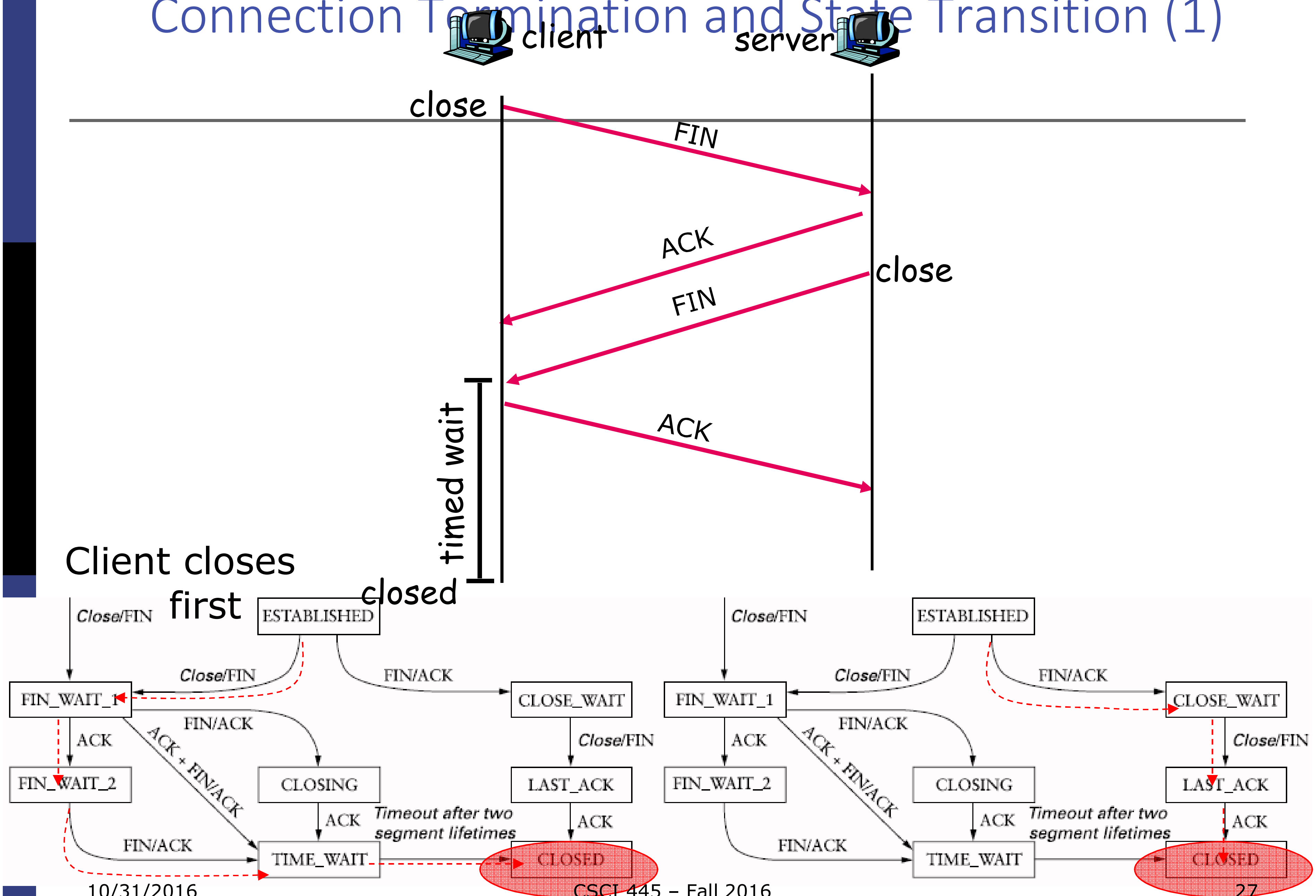
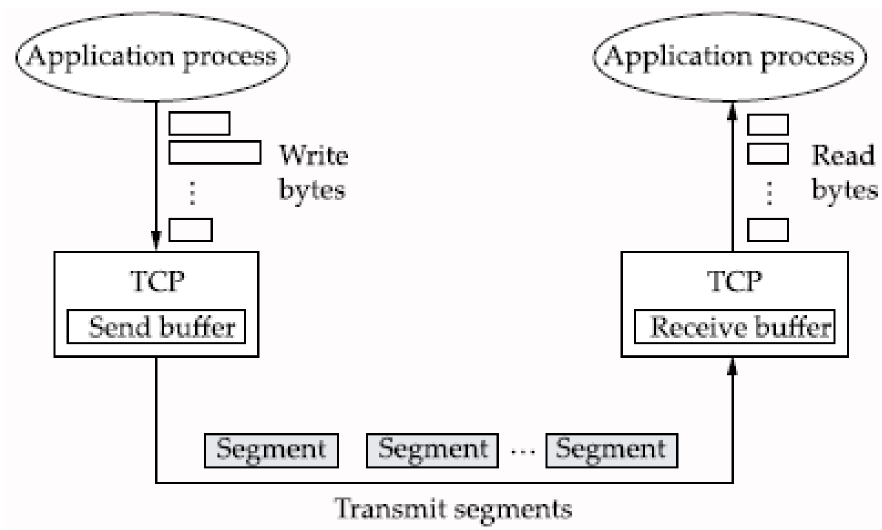


Connection Termination and State Transition (2)

- This side closes first
  - ESTABLISHED → FIN_WAIT_1 → FIN_WAIT_2 → TIME_WAIT
- Other side closes first
  - ESTABLISHED → CLOSE_WAIT → LAST_ACK → CLOSED
- Both sides close at the same time
  - ESTABLISHED → FIN_WAIT_1 → CLOSING → TIME_WAIT → CLOSED

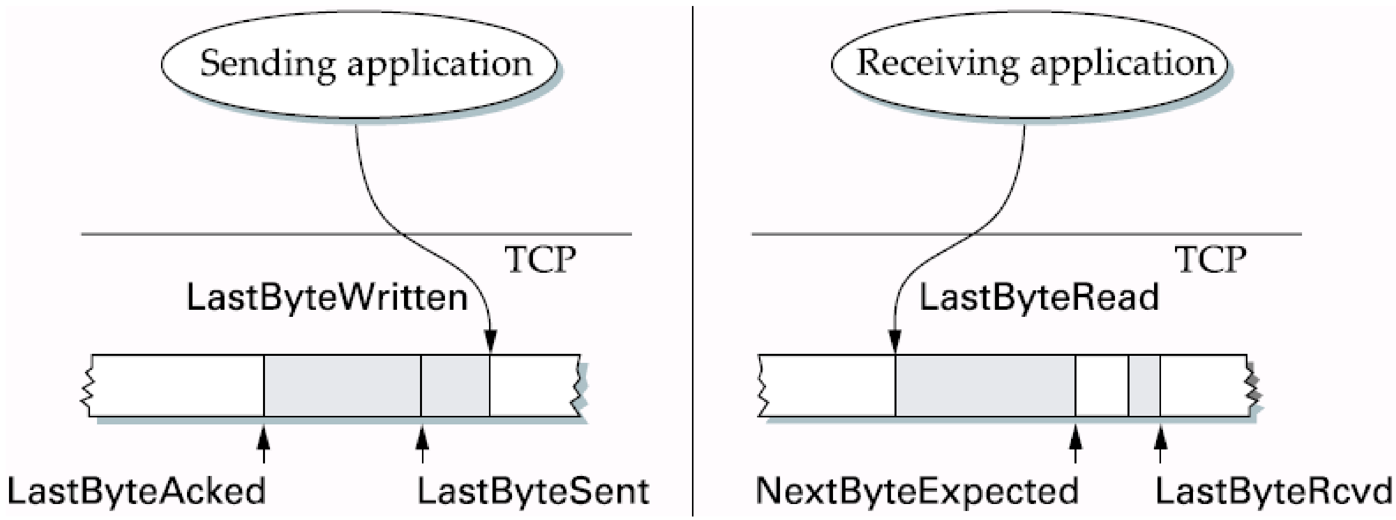
TCP Sliding Window: Why Different?

- Potentially connects many different hosts
  - need explicit connection establishment and termination
- Potentially different RTT
  - need adaptive timeout mechanism
- Potentially long delay in network
  - need to be prepared for arrival of very old packets
- Potentially different capacity at destination
  - need to accommodate different node capacity
- Potentially different network capacity
  - need to be prepared for network congestion



TCP Sliding Window: Reliable and Ordered Delivery

TCP uses cumulative acknowledgements to acknowledge receiving of all the bytes up to the first missing byte



- Sending side
  - LastByteAcked ≤ LastByteSent
  - LastByteSent ≤ LastByteWritten
  - buffer bytes between LastByteAcked and LastByteWritten

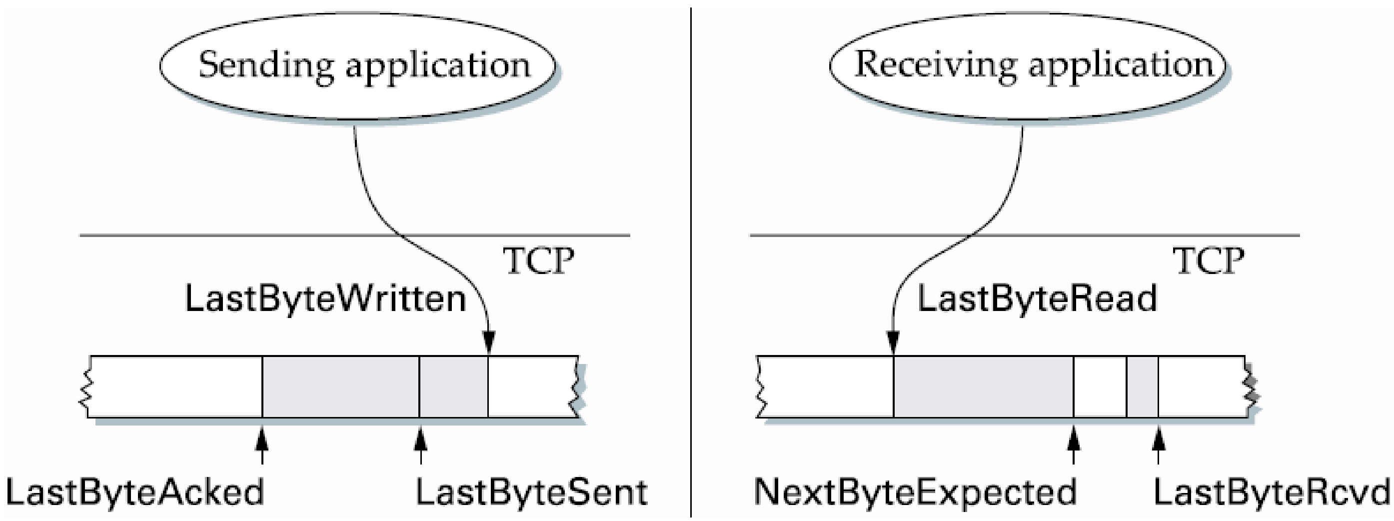
Receiving side
LastByteRead < NextByteExpected
NextByteExpected ≤ LastByteRcvd +1
buffer bytes betweenNextByteRead and LastByteRcvd

TCP Flow Control (1)

- receive side of TCP connection has a receive buffer
- app process may be slow at reading from buffer
- speed-matching service: matching the send rate to the receiving app's drain rate

flow control

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



TCP Flow Control (2)

- Send buffer size: MaxSendBuffer
- Receive buffer size: MaxRcvBuffer
- Receiving side
  - LastByteRcvd - LastByteRead ≤ MaxRcvBuffer
  - AdvertisedWindow = MaxRcvBuffer – ((NextByteExpected -1) - LastByteRead)) → maximum possible free space remaining in the buffer
- Sending side
  - LastByteSent - LastByteAcked ≤ AdvertisedWindow
    - LastByteSent – LastByteAcked: unacknowledged bytes sender has put in TCP
    - Otherwise, the sender may overrun the receiver
  - EffectiveWindow = AdvertisedWindow - (LastByteSent -LastByteAcked) → how much data it can sent
  - LastByteWritten - LastByteAcked ≤ MaxSendBuffer
  - If the sender tries to write y bytes to TCP
    - block sender if (LastByteWritten - LastByteAcked) + y > MaxSenderBuffer
- Always send ACK in response to arriving data segment
- Persist when AdvertisedWindow = 0

Flow Control and Buffering (3)

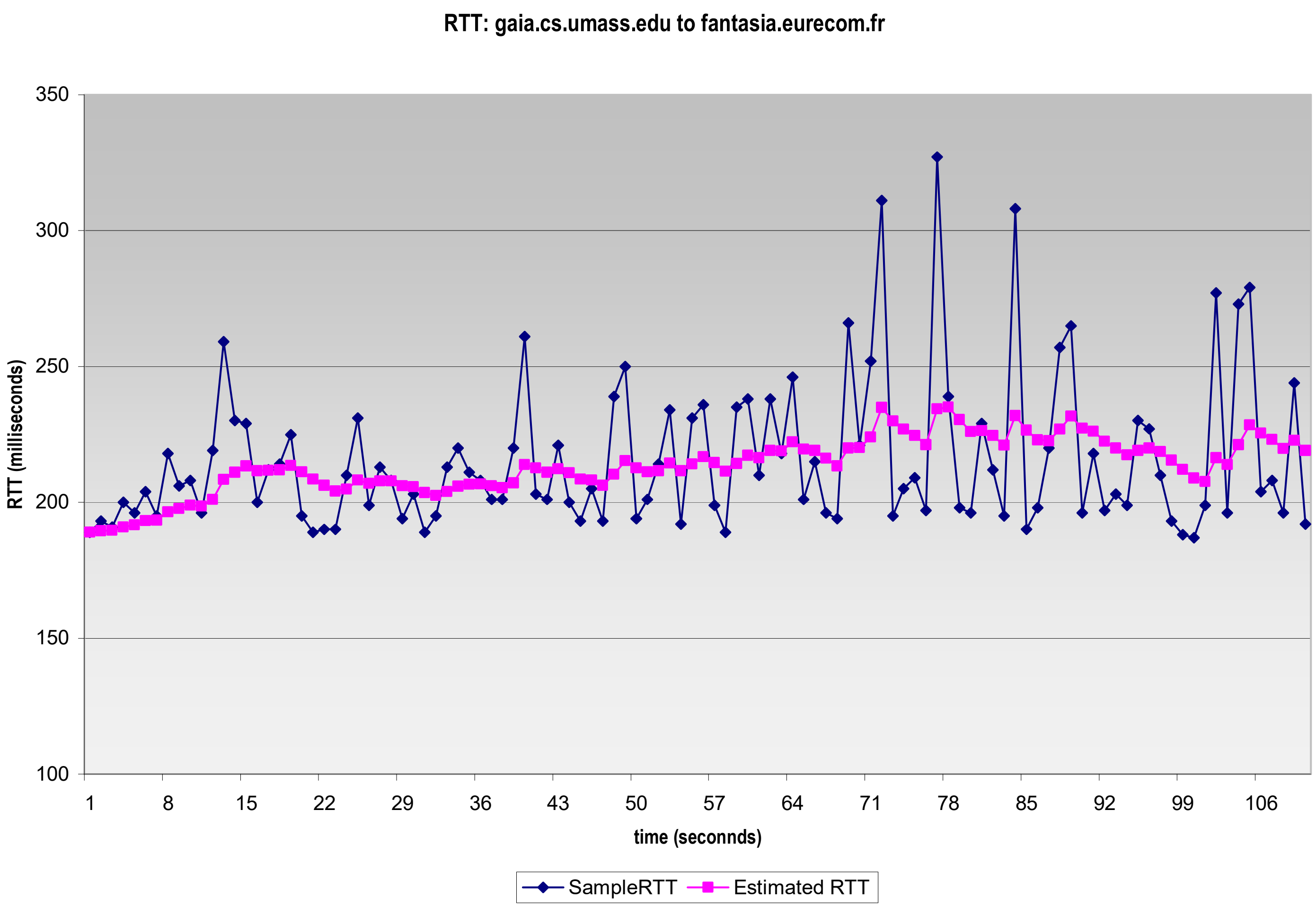
	A	Message	B	Comments
1	→	< request 8 buffers>	→	A wants 8 buffers
2	←	<ack = 15, buf = 4>	←	B grants messages 0-3 only
3	→	<seq = 0, data = m0>	→	A has 3 buffers left now
4	→	<seq = 1, data = m1>	→	A has 2 buffers left now
5	→	<seq = 2, data = m2>	• • •	Message lost but A thinks it has 1 left
6	←	<ack = 1, buf = 3>	←	B acknowledges 0 and 1, permits 2-4
7	→	<seq = 3, data = m3>	→	A has 1 buffer left
8	→	<seq = 4, data = m4>	→	A has 0 buffers left, and must stop
9	→	<seq = 2, data = m2>	→	A times out and retransmits
10	←	<ack = 4, buf = 0>	←	Everything acknowledged, but A still blocked
11	←	<ack = 4, buf = 1>	←	A may now send 5
12	←	<ack = 4, buf = 2>	←	B found a new buffer somewhere
13	→	<seq = 5, data = m5>	→	A has 1 buffer left
14	→	<seq = 6, data = m6>	→	A is now blocked again
15	←	<ack = 6, buf = 0>	←	A is still blocked
16	• • •	<ack = 6, buf = 4>	←	Potential deadlock

Dynamic buffer allocation. The arrows show the direction of transmission. An ellipsis (…) indicates a lost TCP segment

Adaptive Retransmission: Original Algorithm

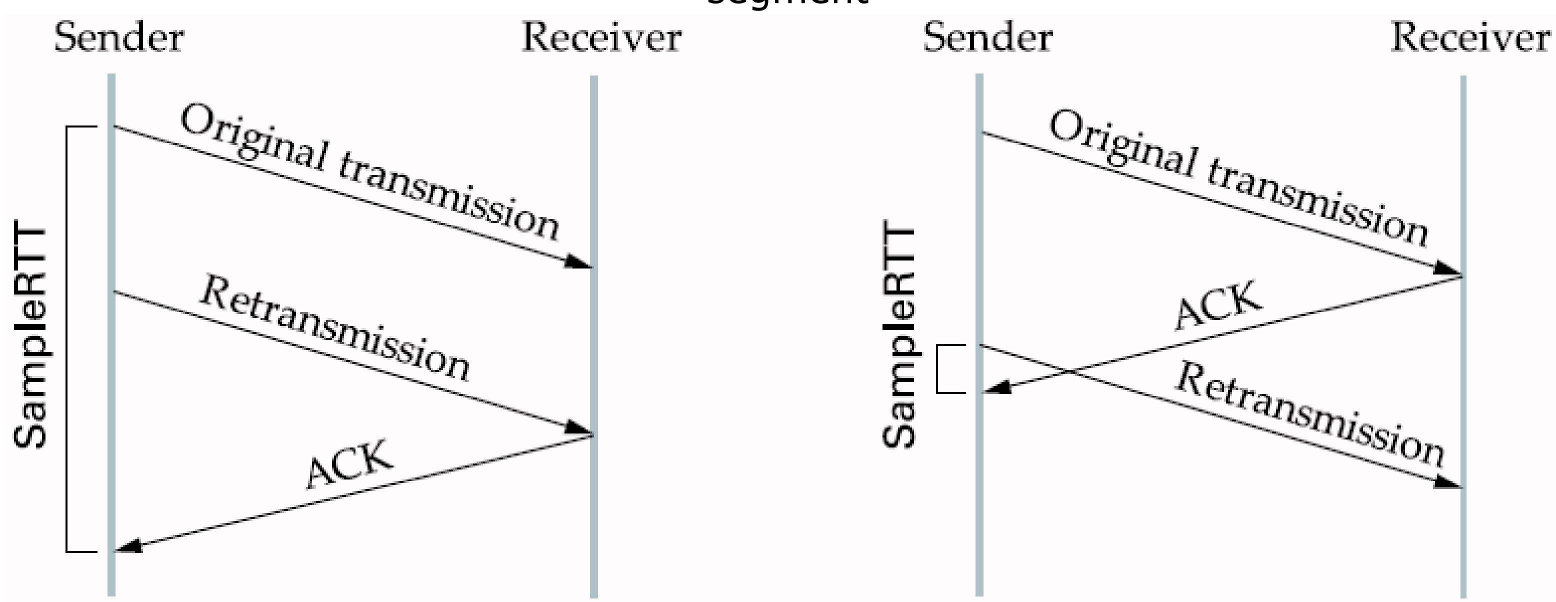
- Measure SampleRTT for each segment/ACK pair
- Compute weighted average of RTT
  - EstimatedRTT = α x EstimatedRTT + β x SampleRTT
  - where α **+** β **= 1**
    - α between 0.8 and 0.9
    - β between 0.1 and 0.2
  - Set timeout based on EstimatedRTT
    - TimeOut = 2 x EstimatedRTT

Example RTT estimation:



Adaptive Retransmission: Karn/Partridge Algorithm

Problem with original algorithm
ACK does not really acknowledge a transmission, it acknowledges the receipt of data → can not distinguish an ACK is for which transmission/retransmission of a segment



- Do not sample RTT when retransmitting
- <u>**Double timeout after each retransmission**</u>
  - Congestion is the most likely cause of lost segments → TCP should not react too aggressively to a timeout

Jacobson/ Karels Algorithm

- Previous approaches did not take the variance of the sample RTT into account
  - If no variance, Estimated RTT is good enough, 2 × Estimated RTT is too pessimistic
  - If variance large, timeout value should not be too dependent on Estimated RTT
- New Calculations for average RTT
  - Difference = SampleRTT – EstimtaedRTT
  - EstimatedRTT = EstimatedRTT + (δ x Difference)
  - Deviation = Deviation + δ(|Difference| - Deviation)
    - where δ is a factor between 0 and 1
  - Consider variance when setting timeout value
    - TimeOut = μ x EstimatedRTT + ϕ x Deviation
    - where μ = 1 and ϕ = 4
- Notes
  - algorithm only as good as granularity of clock (500ms on Unix)
  - accurate timeout mechanism important to congestion control

TCP: Sequence Number Wrap Around

Bandwidth	Time until Wraparound
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
Fast Ethernet (100 Mbps)	6 minutes
OC-3 (155 Mbps)	4 minutes
OC-12 (622 Mbps)	55 seconds
OC-48 (2.5 Gbps)	14 seconds

Time until 32-bit sequence number space wraps around

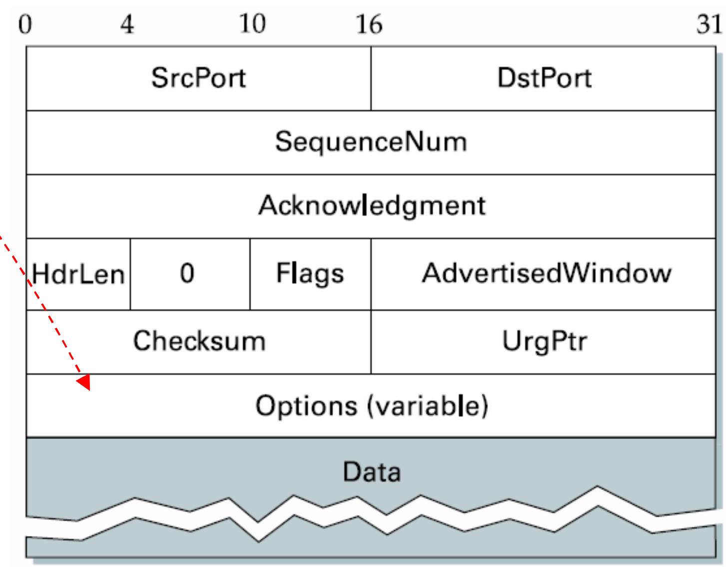
TCP: Can Keep Pipe Full?

Bandwidth	Delay × Bandwidth Product
T1 (1.5 Mbps)	18 KB
Ethernet (10 Mbps)	122 KB
T3 (45 Mbps)	549 KB
Fast Ethernet (100 Mbps)	1.2 MB
OC-3 (155 Mbps)	1.8 MB
OC-12 (622 Mbps)	7.4 MB
OC-48 (2.5 Gbps)	29.6 MB

Required window size for 100-ms RTT.

Solution: TCP Extensions

- Implemented as header options
- Store timestamp in outgoing segments → measure RTT
- Extend sequence space with 32-bit timestamp → protected against sequence number wrap-around
- Shift (scale) advertised window → keep the pipe full
- Selective acknowledgement (SAC) → acknowledge any additional (out-of-order) blocks of received data



TCP Extensions for High Performance
http://tools.ietf.org/html/rfc1323

Summary

- User Datagram Protocol
  - Multiplexer/Demultiplexer for IP
- Transmission Control Protocol
  - Reliable Byte Stream
    - Connection-oriented
      - Connection establishment
      - Connection termination
    - Automatics Repeated-Request: ACKs and NACKs
    - Flow-control
    - Timeout value estimation
    - Extensions
- *Congestion control (future discussions)*