COMP9121: Design of Networks and Distributed Systems

Week 7: Network Layer 4 and Transport Layer 1

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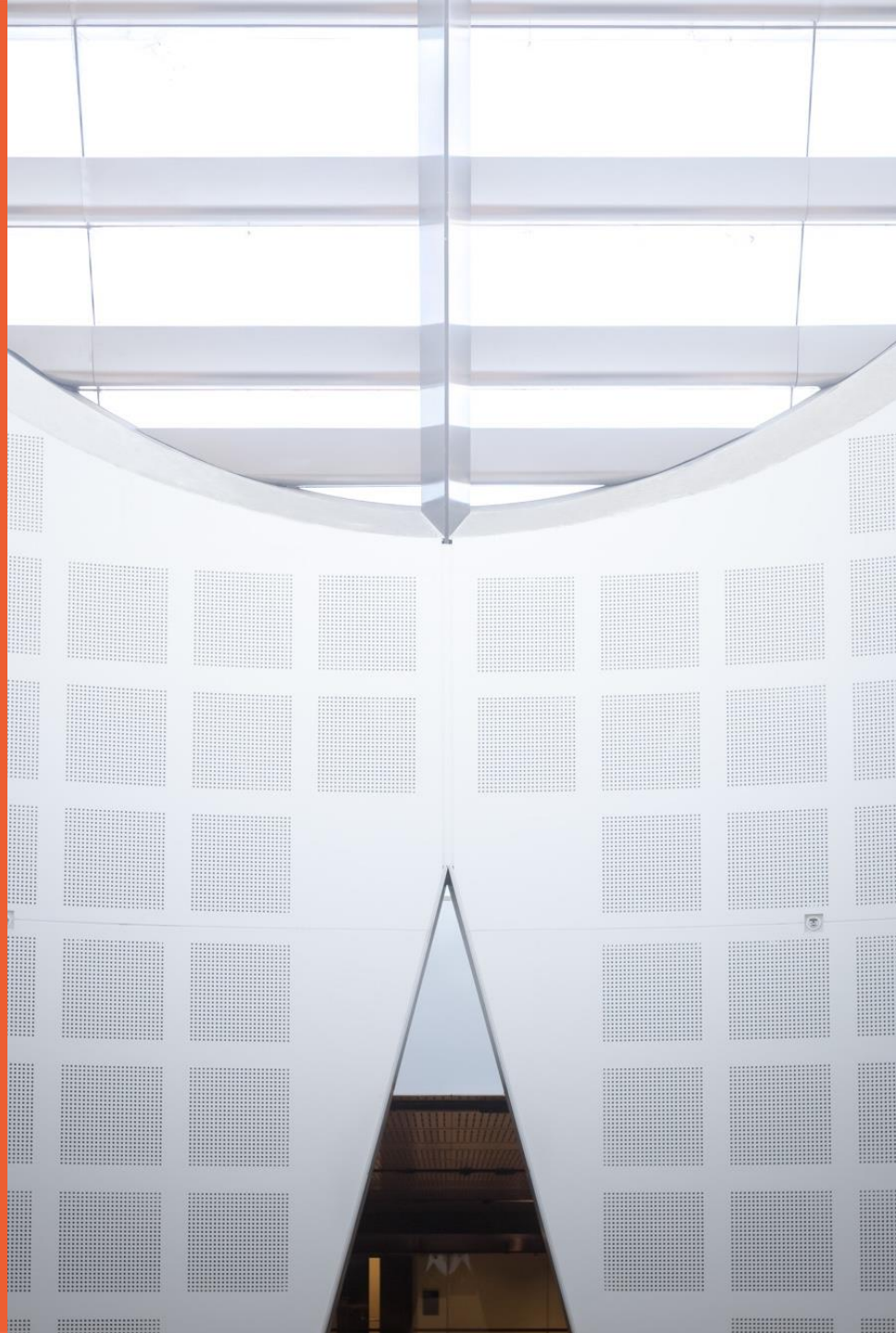
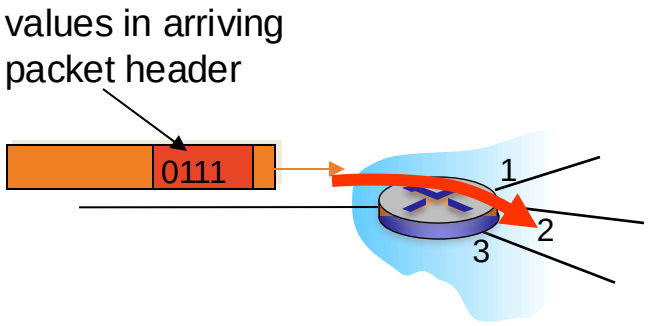


第一部分：结束 Network layer

Network layer: data plane, control plane

Data plane

- local, per-router function
- determines how datagram arriving on router input port is forwarded to router output port
- forwarding function

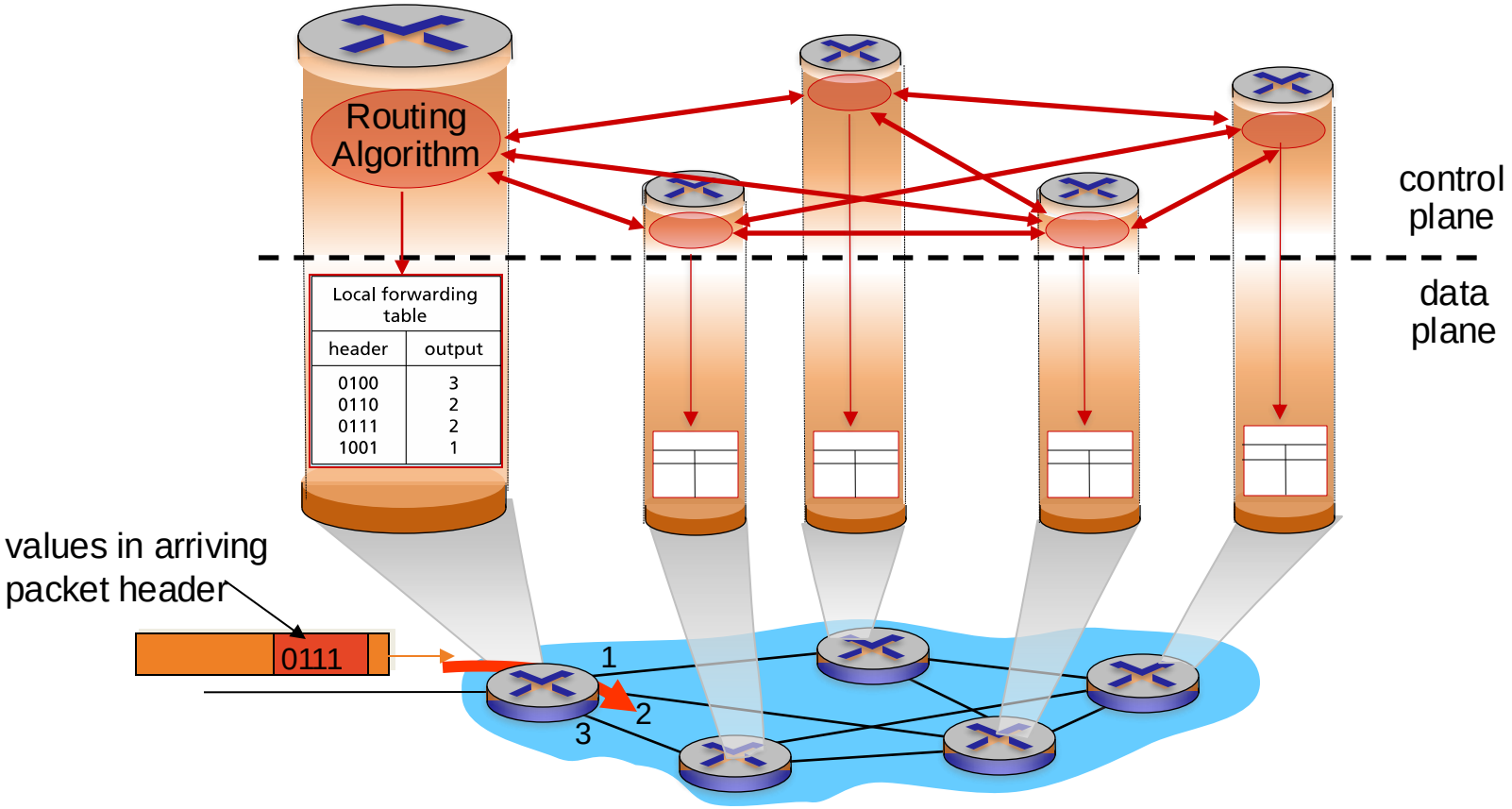


Control plane

- network-wide logic
- determines how datagram is routed among routers along end-end path from source host to destination host
- two control-plane approaches:
  - *traditional routing algorithms:* implemented in routers
  - *software-defined networking (SDN)*: implemented in (remote) servers

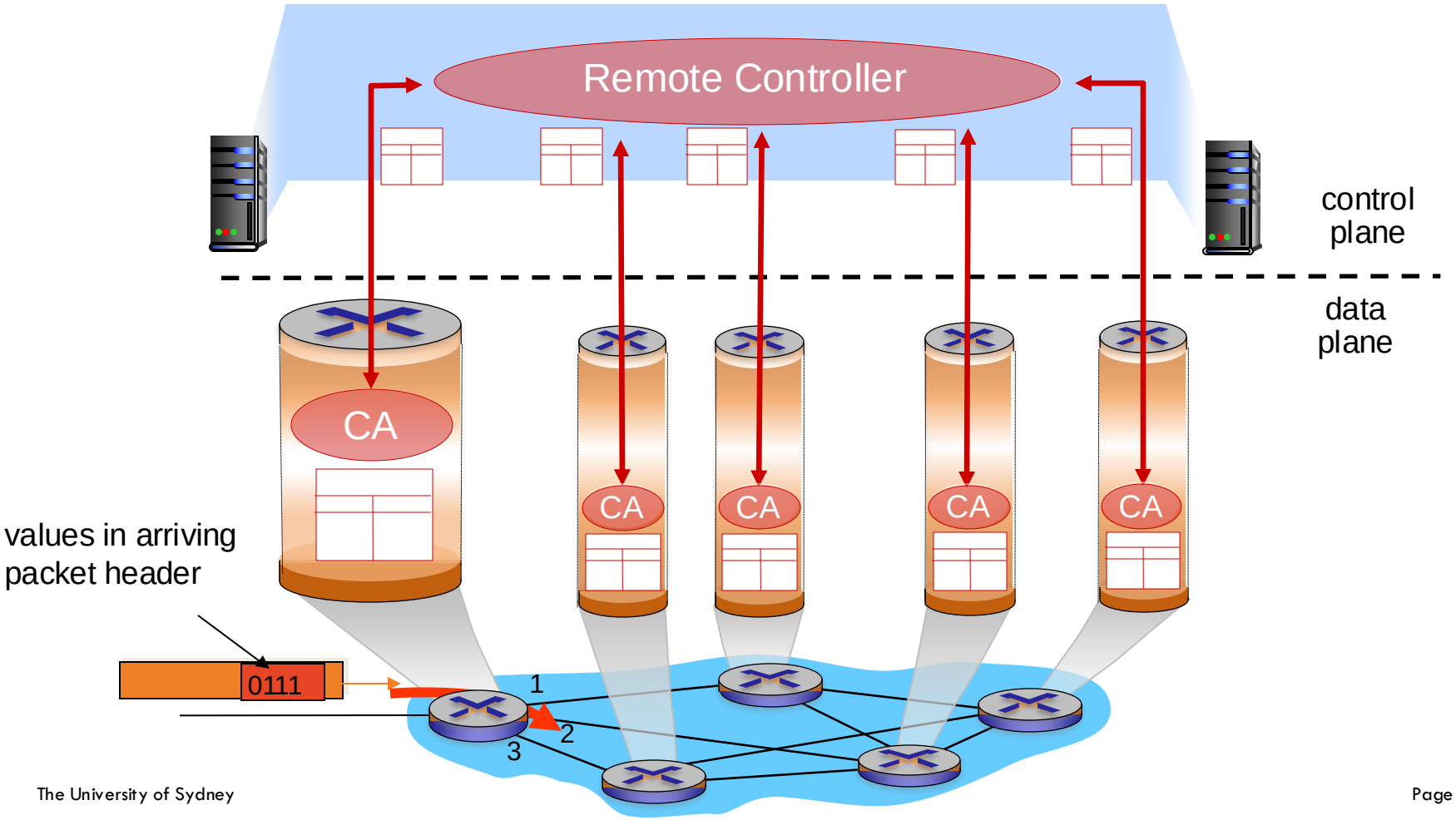
Traditional: Per-router control plane

Individual routing algorithm components *in each and every router* interact in the control plane



SDN: Logically centralized control plane

A distinct (typically remote) controller interacts with local control agents (CAs)



Software defined networking (SDN)

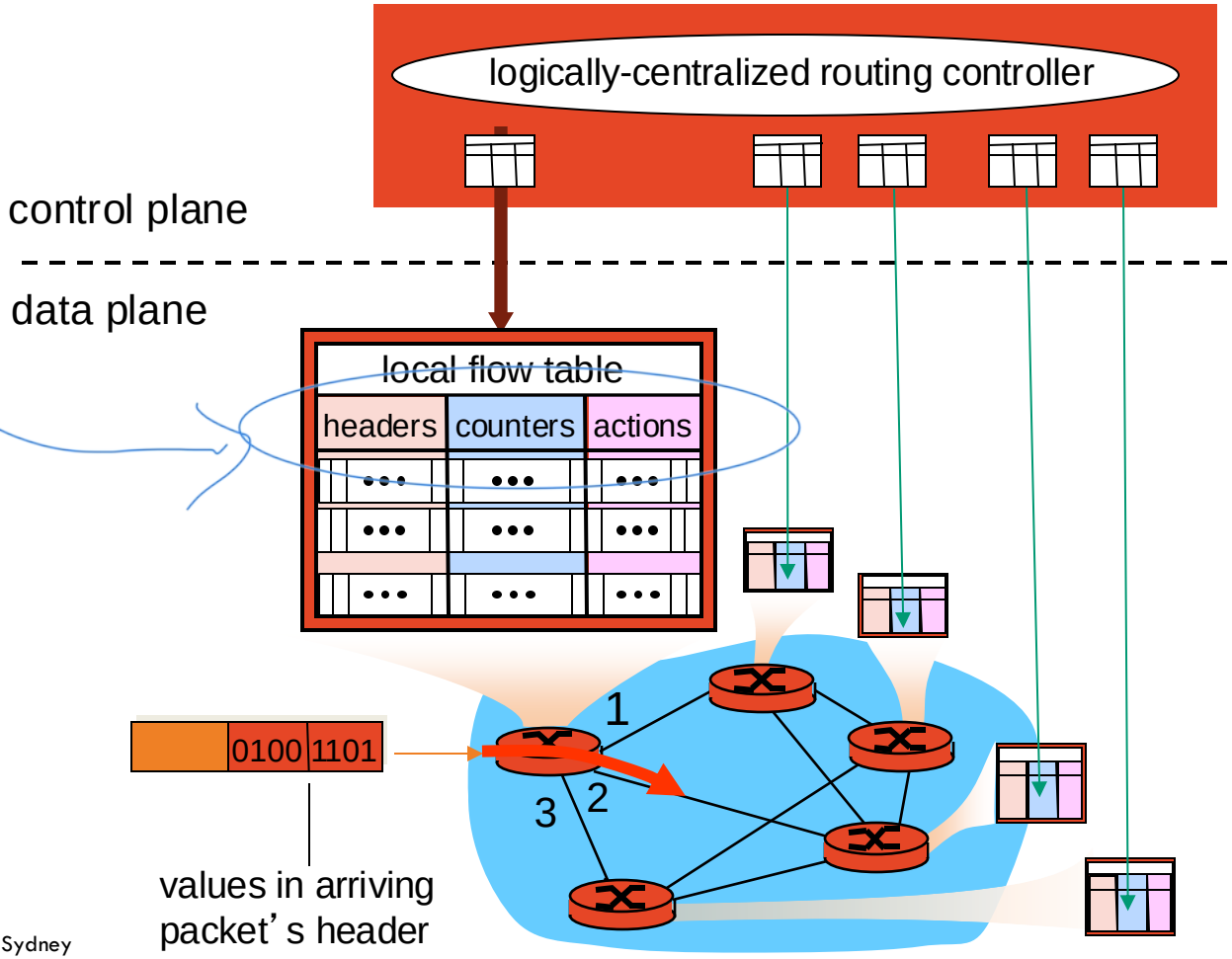
- Internet network layer: historically has been implemented via distributed, per-router approach
  - *monolithic* router contains switching hardware, runs proprietary implementation of Internet standard protocols (IP, RIP, IS-IS, OSPF, BGP) in proprietary router OS (e.g., Cisco IOS)
  - different "middleboxes" for different network layer functions: firewalls, load balancers, NAT boxes, ..

- ~2005: renewed interest in rethinking network control plane

Data Plane

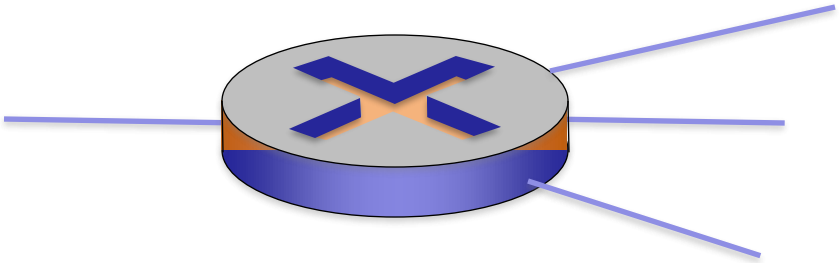
Data Plane: Generalized Forwarding

Each router contains a *flow table* that is computed and distributed by a *logically centralized* routing controller



OpenFlow data plane abstraction

- *flow:* defined by header fields
- generalized forwarding: simple packet-handling rules
  - *Pattern:* match values in packet header fields
  - *Actions: for matched packet:* drop, forward, modify, matched packet or send matched packet to controller
  - *Priority*: disambiguate overlapping patterns
  - *Counters:* #bytes and #packets



Flow table in a router (computed and distributed by controller) define router's match+action rules

OpenFlow data plane abstraction

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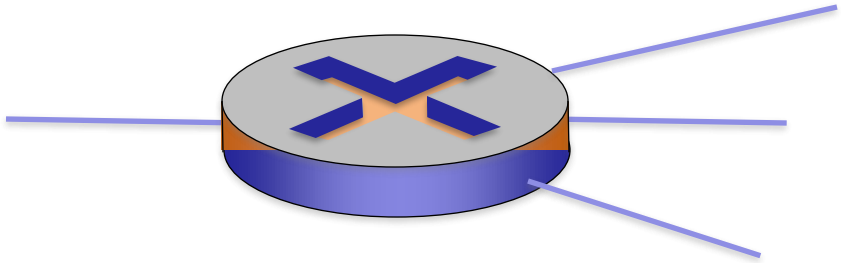
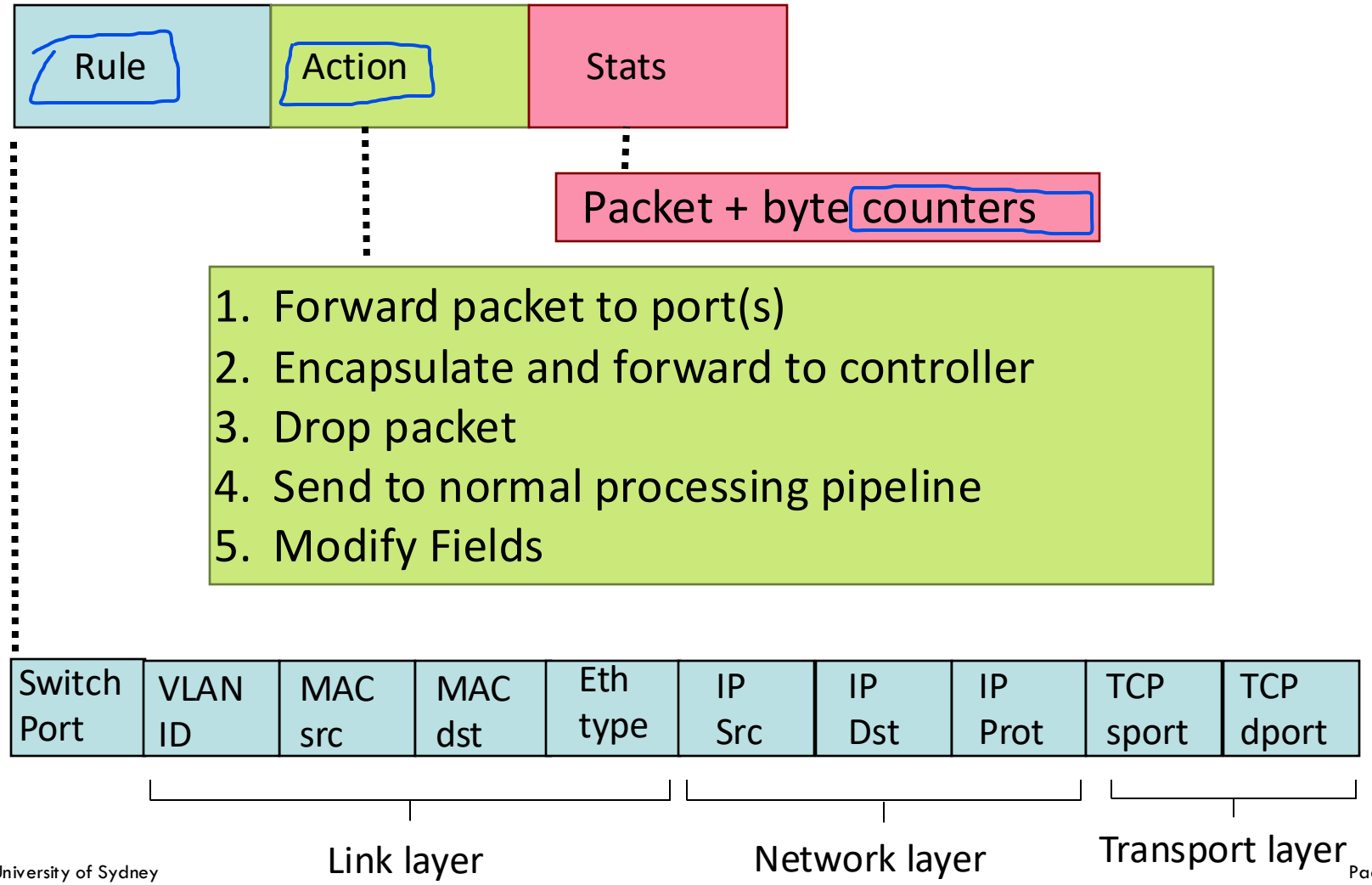


* : wildcard

Ex

1. src=1.2.*.*, dest=3.4.5.* → drop
2. src = *.*.*.*, dest=3.4.*.* → forward(2)
3. src=10.1.2.3, dest=*.*.*.* → send to controller

OpenFlow: Flow Table Entries



Examples

Destination-based forwarding:

Switch Port	MAC src	MAC dst	Eth type	VLAN ID	IP Src	IP Dst	IP Prot	TCP sport	TCP dport	Action
*	*	*	*	*	*	51.6.0.8	*	*	*	port6

IP datagrams destined to IP address 51.6.0.8 should be forwarded to router output port 6

Firewall:

Switch Port	MAC src	MAC dst	Eth type	VLAN ID	IP Src	IP Dst	IP Prot	TCP sport	TCP dport	Action
*	*	*	*	*	*	*	*	*	22	drop

do not forward (block) all datagrams destined to TCP port 22

Switch Port	MAC src	MAC dst	Eth type	VLAN ID	IP Src	IP Dst	IP Prot	TCP sport	TCP dport	Action
*	*	*	*	*	128.119.1.1	*	*	*	*	drop

do not forward (block) all datagrams sent by host 128.119.1.1

Examples

Destination-based layer 2 (switch) forwarding:

Switch Port	MAC src	MAC dst	Eth type	VLAN ID	IP Src	IP Dst	IP Prot	TCP sport	TCP dport	Action
*	22:A7:23: 11:E1:02	*	*	*	*	*	*	*	*	port3

layer 2 frames from MAC address 22:A7:23:11:E1:02 should be forwarded to output port 3

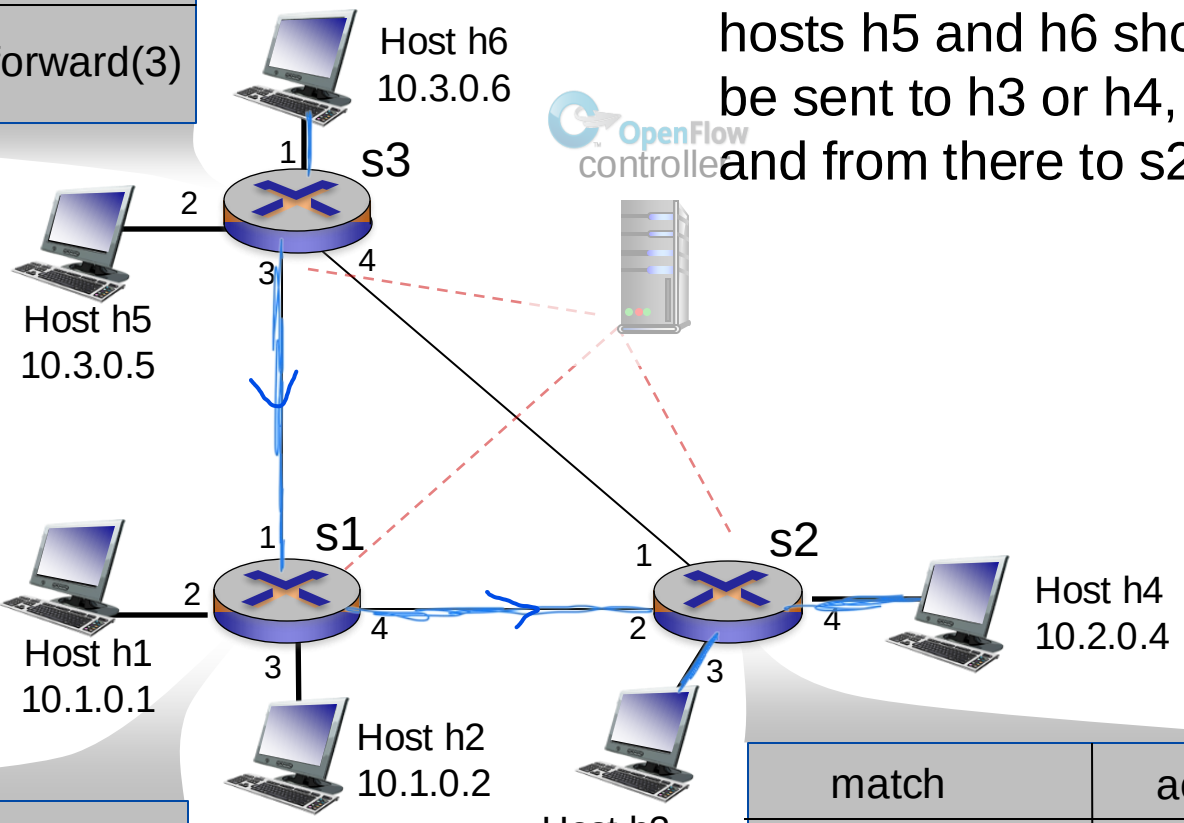
OpenFlow abstraction

- *match+action:* unifies different kinds of devices

- Router
  - *match:* longest destination IP prefix
  - *action:* forward out a link
- Switch
  - *match:* destination MAC address
  - *action:* forward or flood
- Firewall
  - *match*: IP addresses and TCP/UDP port numbers
  - *action:* permit or deny
- NAT
  - *match:* IP address and port
  - *action:* rewrite address and port

OpenFlow example

match	action
IP Src = 10.3.*.* IP Dst = 10.2.*.*	forward(3)



Example: datagrams from hosts h5 and h6 should be sent to h3 or h4, via s1 and from there to s2

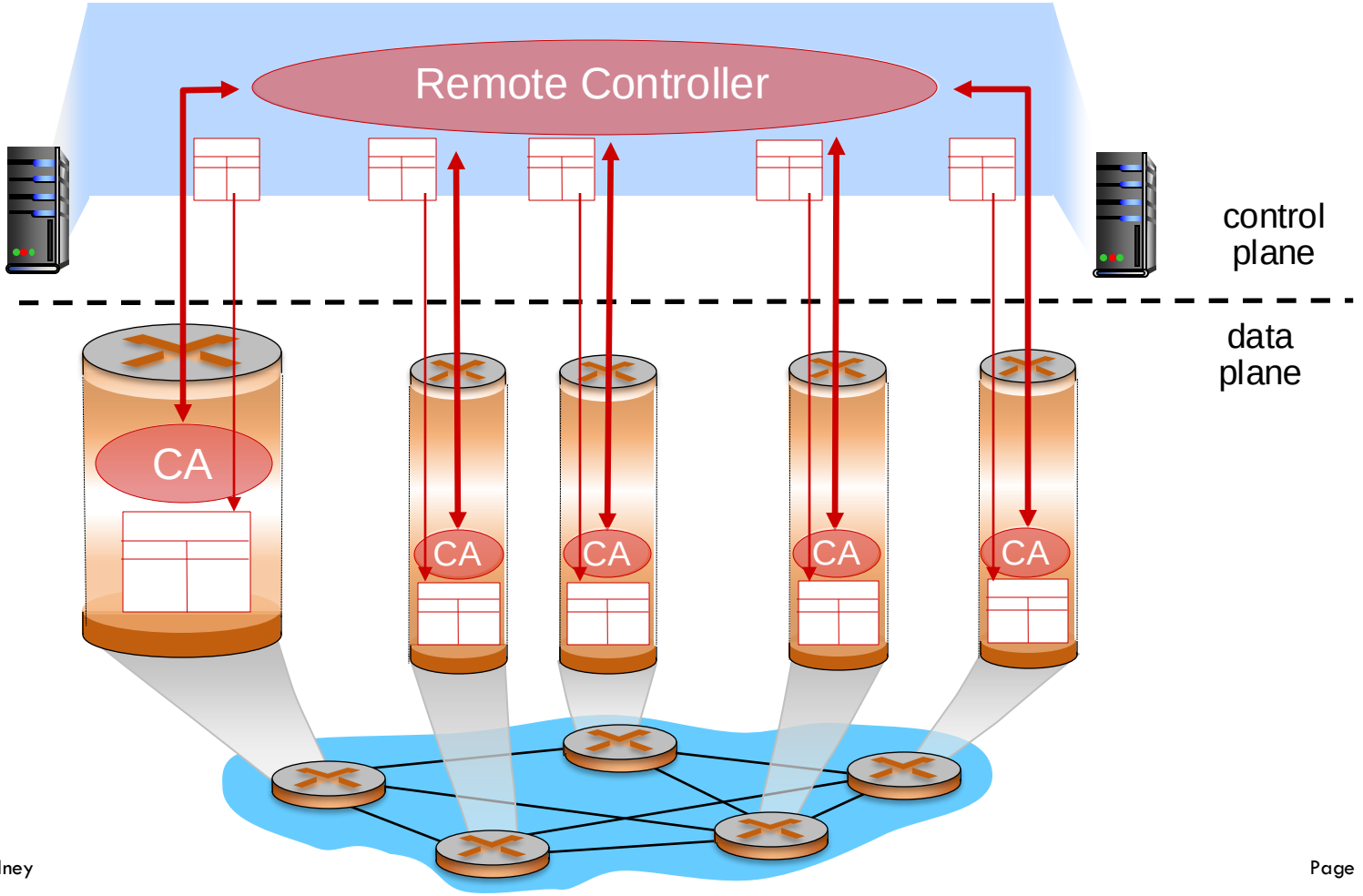
match	action
ingress port = 1 IP Src = 10.3.*.* IP Dst = 10.2.*.*	forward(4)

match	action
ingress port = 2 IP Dst = 10.2.0.3	forward(3)
ingress port = 2 IP Dst = 10.2.0.4	forward(4)

Control Plane

Recall: logically centralized control plane

A distinct (typically remote) controller interacts with local control agents (CAs) in routers to compute forwarding tables

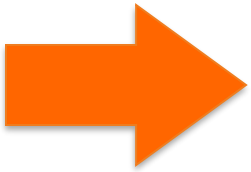


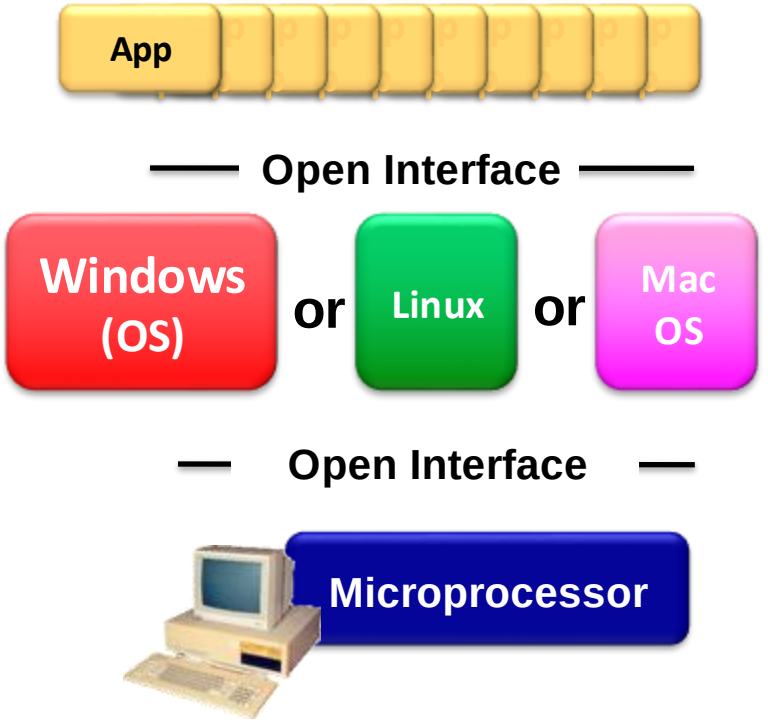
Software defined networking (SDN)

Why a *logically centralized* control plane?

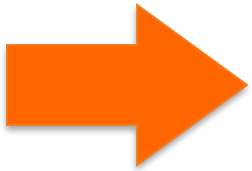
- easier network management: avoid router misconfigurations, greater flexibility of traffic flows
- table-based forwarding allows "programming" routers
  - centralized programming easier: compute tables centrally and distribute
  - distributed programming: more difficult: compute tables as result of distributed algorithm (protocol) implemented in each and every router
- open (non-proprietary) implementation of control plane

Analogy: mainframe to PC evolution





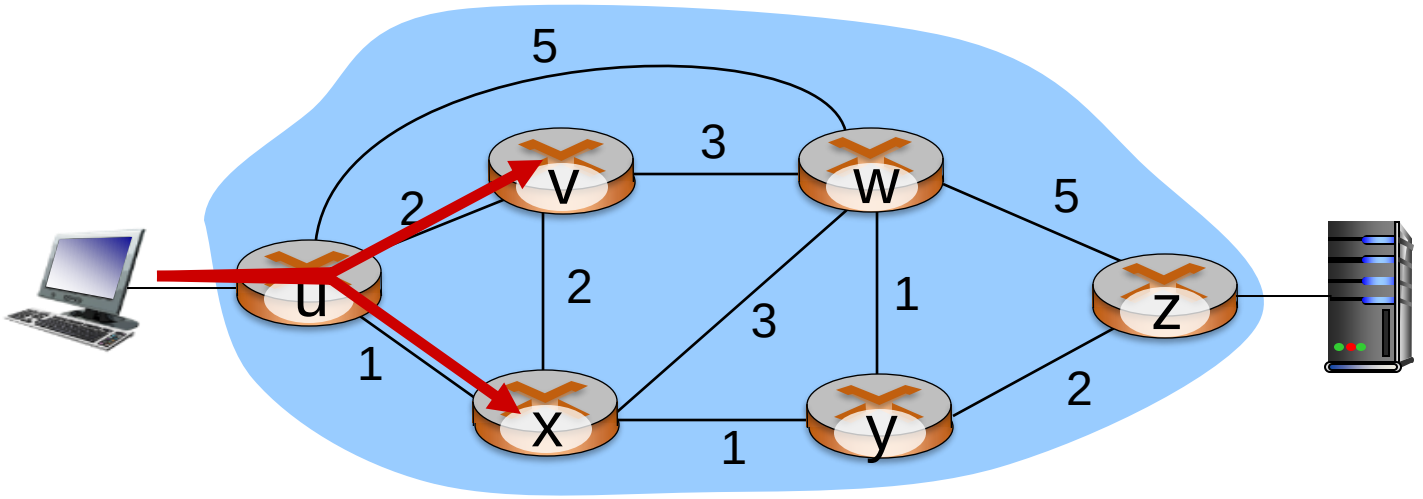
Vertically integrated
Closed, proprietary
Slow innovation
Small industry



Horizontal
Open interfaces
Rapid innovation
Huge industry

* Slide courtesy: N. McKeown

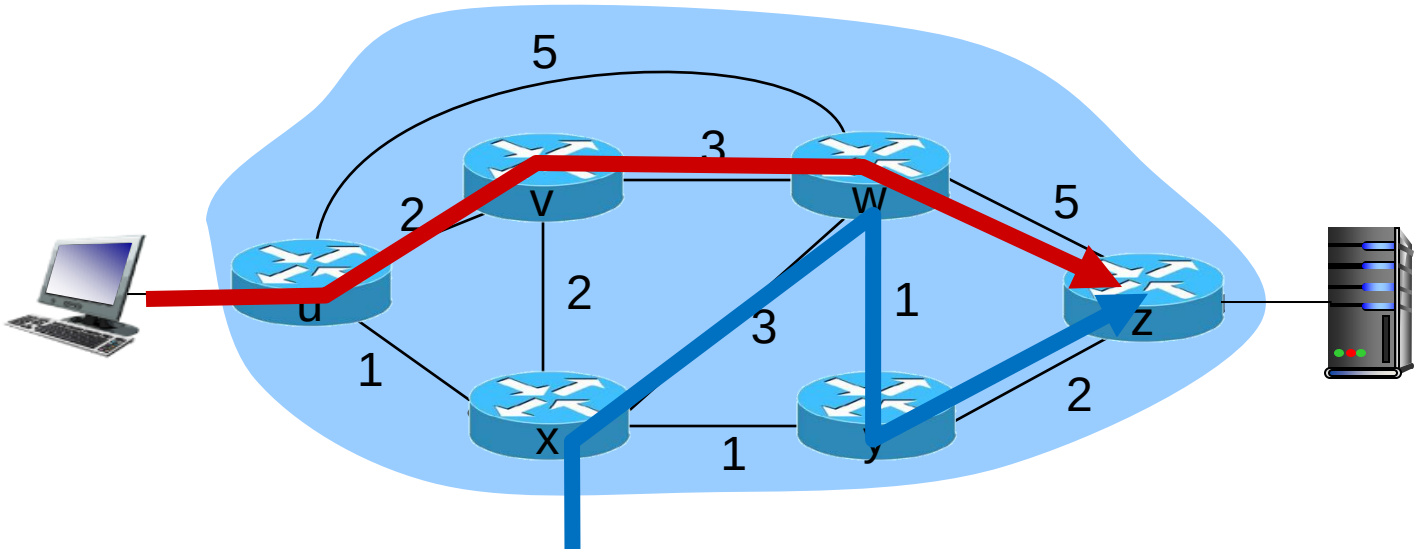
Traffic engineering: difficult traditional routing



Q: what if network operator wants to split u-to-z traffic along uvwz *and* uxyz (load balancing)?

A: can't do it (or need a new routing algorithm)

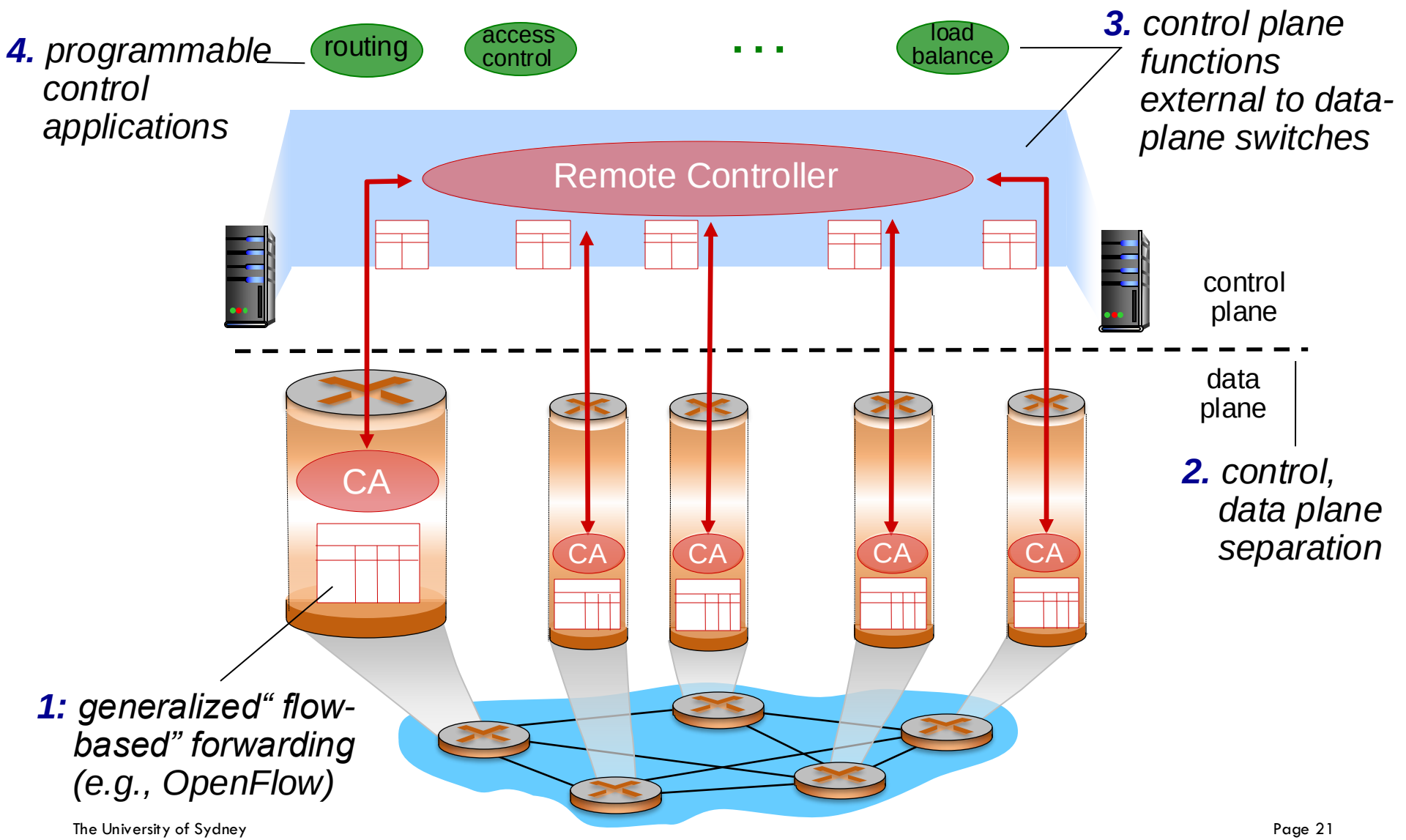
Traffic engineering: difficult traditional routing



Q: what if w wants to route blue and red traffic differently?

A: can't do it (with destination based forwarding, and LS, DV routing)

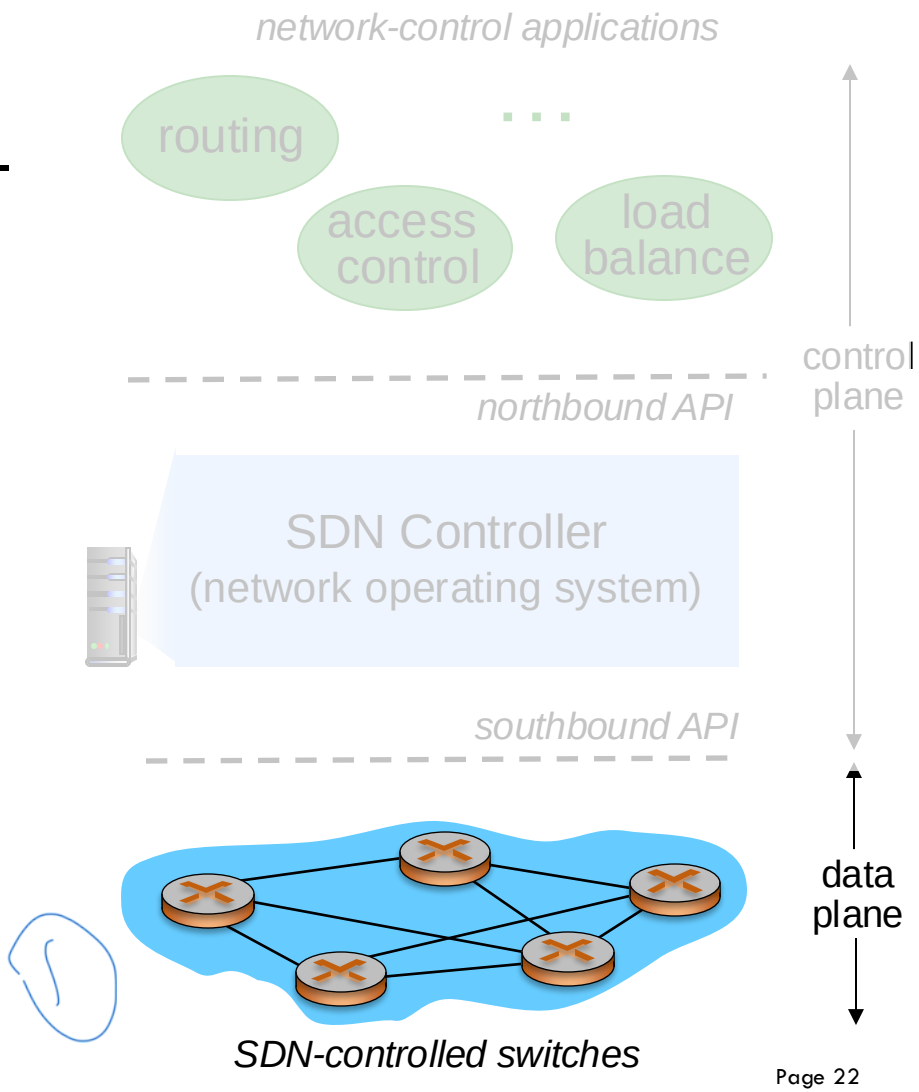
Software defined networking (SDN)



SDN perspective: data plane switches

Data plane switches

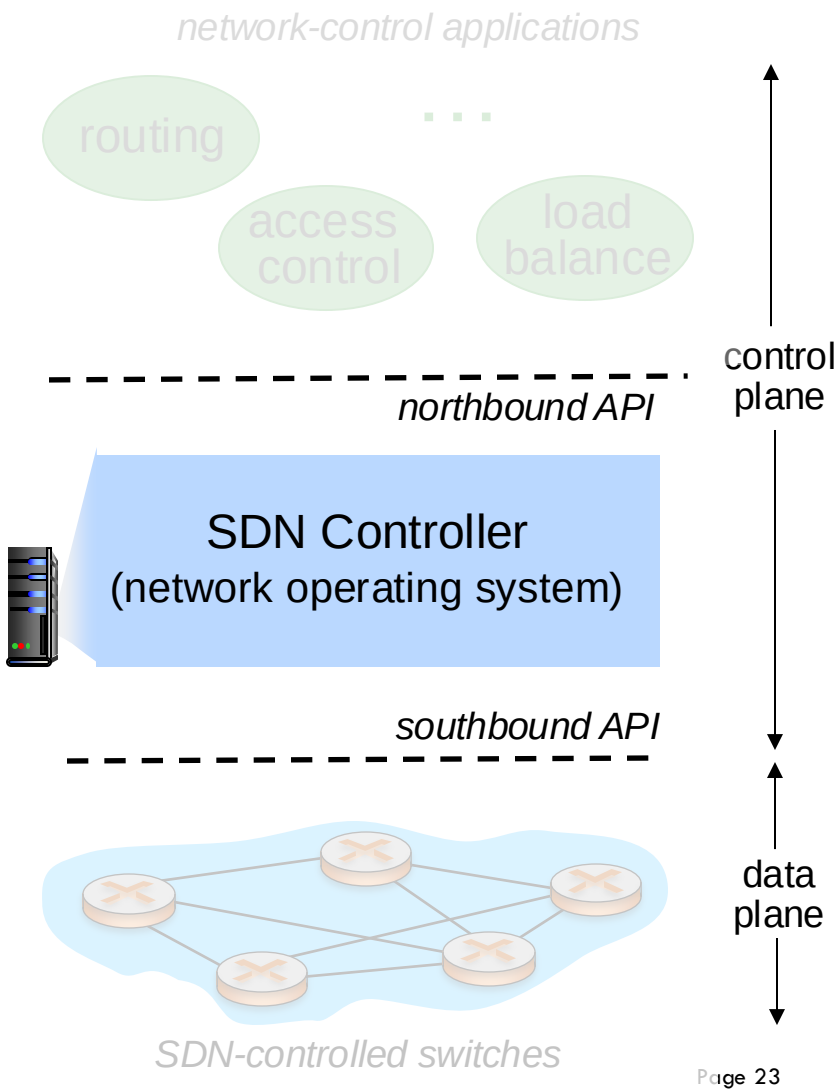
- fast, simple, commodity switches implementing generalized data-plane forwarding in hardware
- switch flow table computed, installed by controller
- API for table-based switch control (e.g., OpenFlow)
  - defines what is controllable and what is not
- protocol for communicating with controller (e.g., OpenFlow)



SDN perspective: SDN controller

SDN controller (network OS):

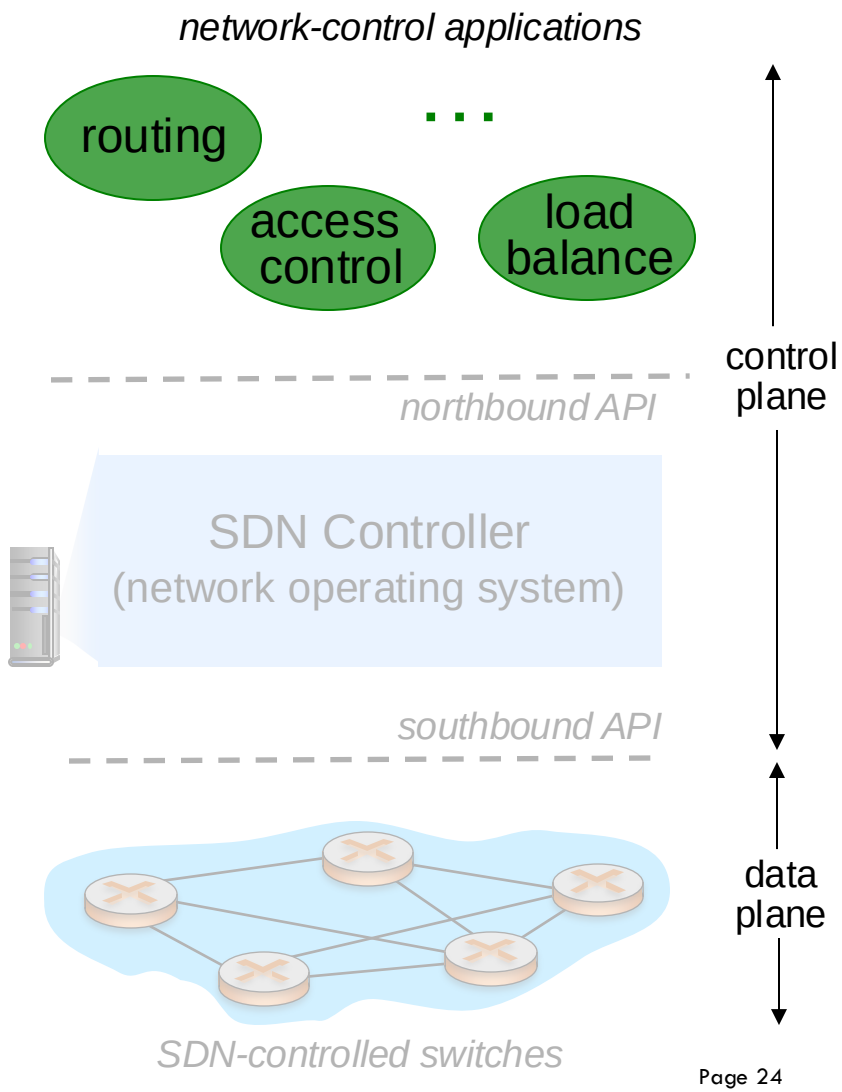
- maintain network state information
- interacts with network control applications "above" via northbound API
- interacts with network switches "below" via southbound API
- implemented as distributed system for performance, scalability, fault-tolerance, robustness



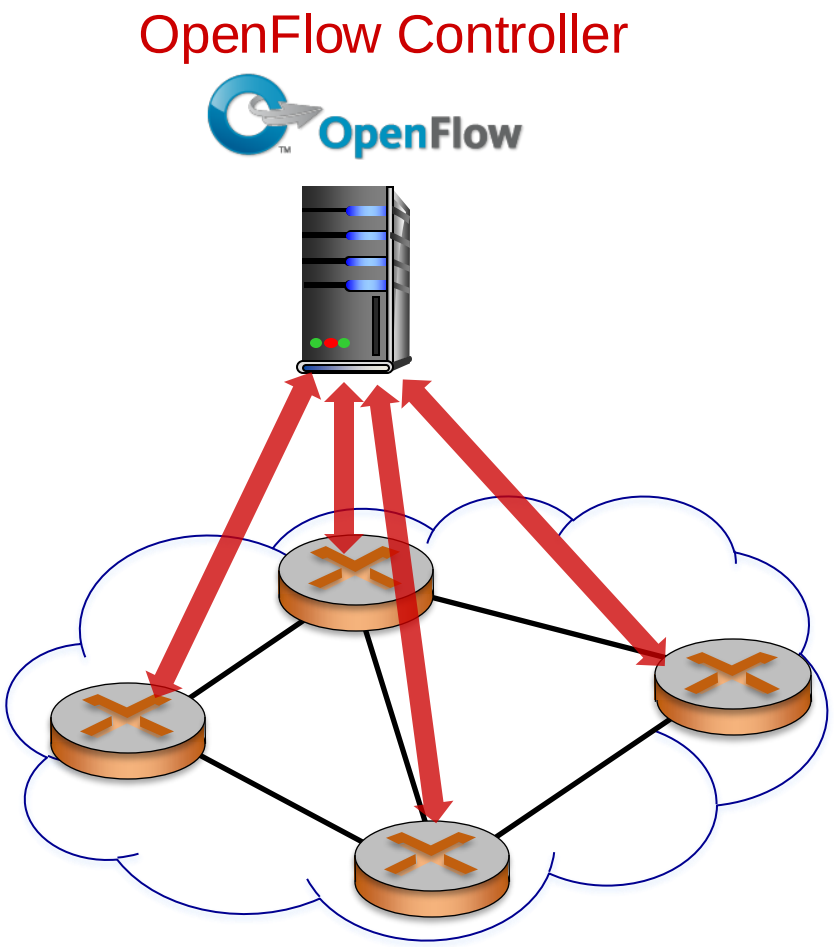
SDN perspective: control applications

network-control apps:

- "brains" of control: implement control functions using lower-level services, API provided by SND controller
- *unbundled*: can be provided by 3rd party: distinct from routing vendor, or SDN controller



OpenFlow protocol

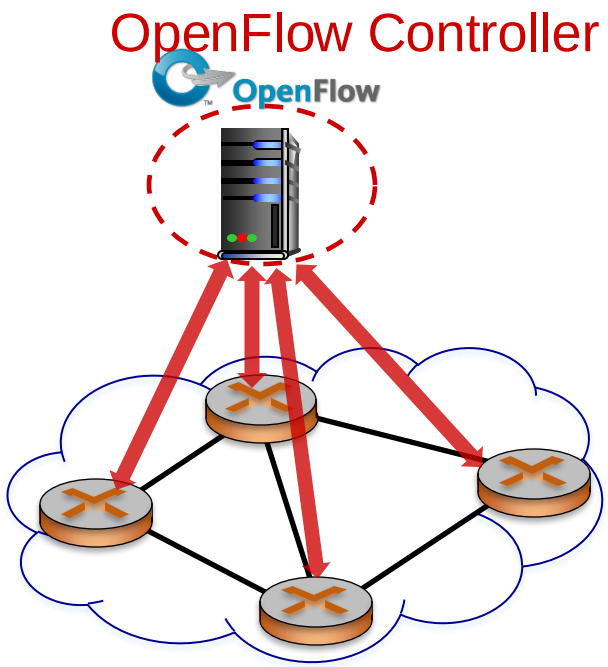


- operates between controller, switch
- TCP used to exchange messages
  - optional encryption
- OpenFlow messages:
  - controller-to-switch
  - switch-to-controller

OpenFlow: controller-to-switch messages

Key controller-to-switch messages

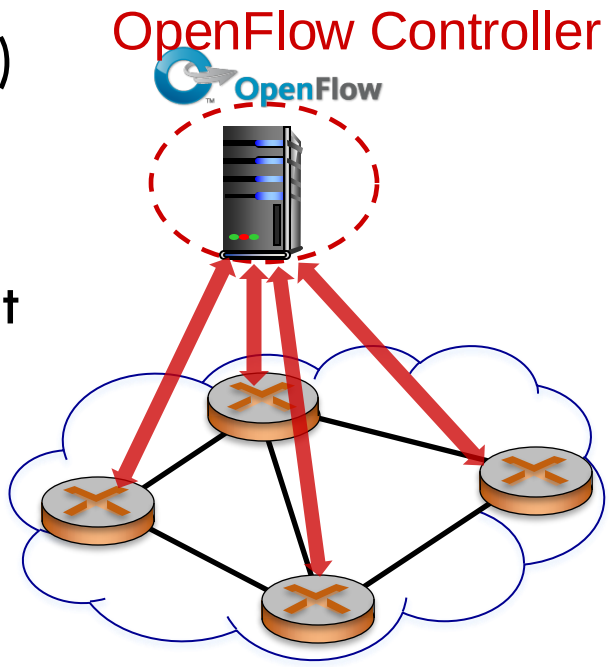
- *features:* controller queries switch features, switch replies
- *configure:* controller queries/sets switch configuration parameters
- *modify-state:* add, delete, modify flow entries in the OpenFlow tables
- *packet-out:* controller can send this packet out of specific switch port



OpenFlow: switch-to-controller messages

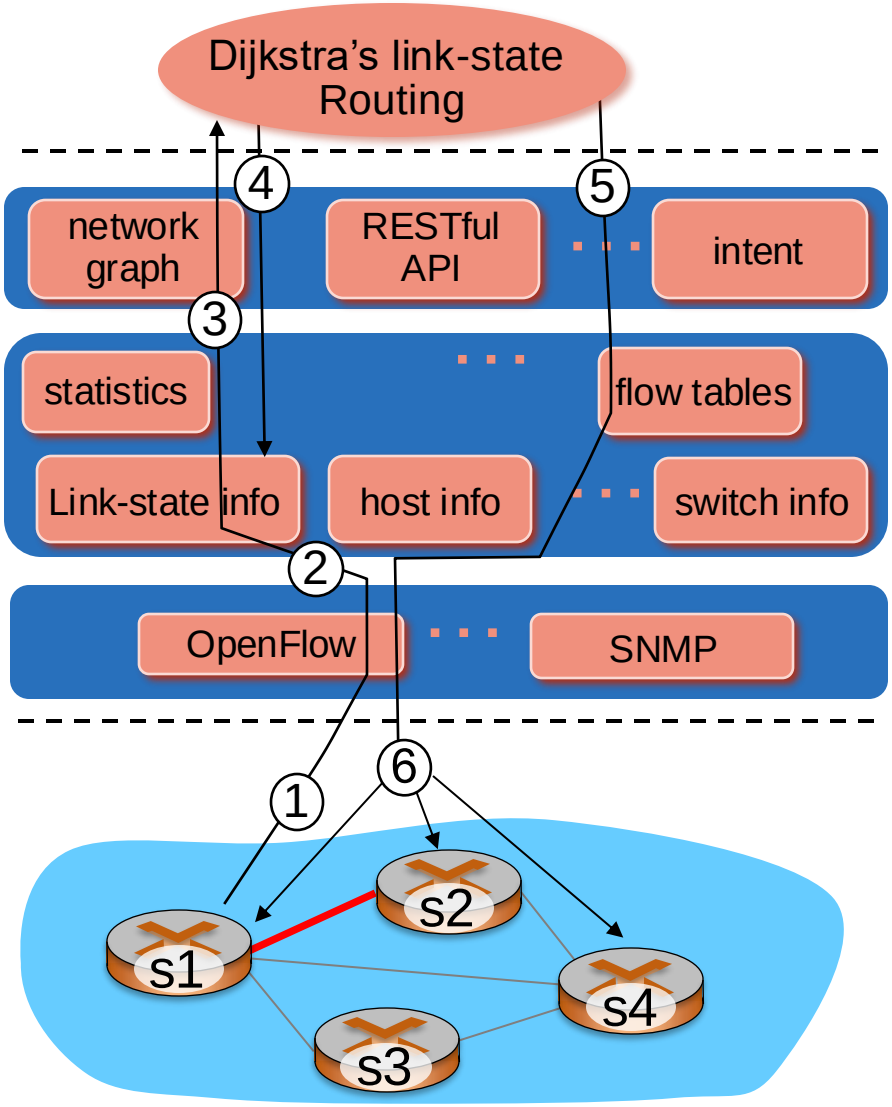
Key switch-to-controller messages

- *packet-in:* transfer packet (and its control) to controller. See packet-out message from controller
- *flow-removed:* flow table entry deleted at switch
- *port status:* inform controller of a change on a port.



Fortunately, network operators don't "program" switches by creating/sending OpenFlow messages directly. Instead use higher-level abstraction at controller

SDN: control/data plane interaction example



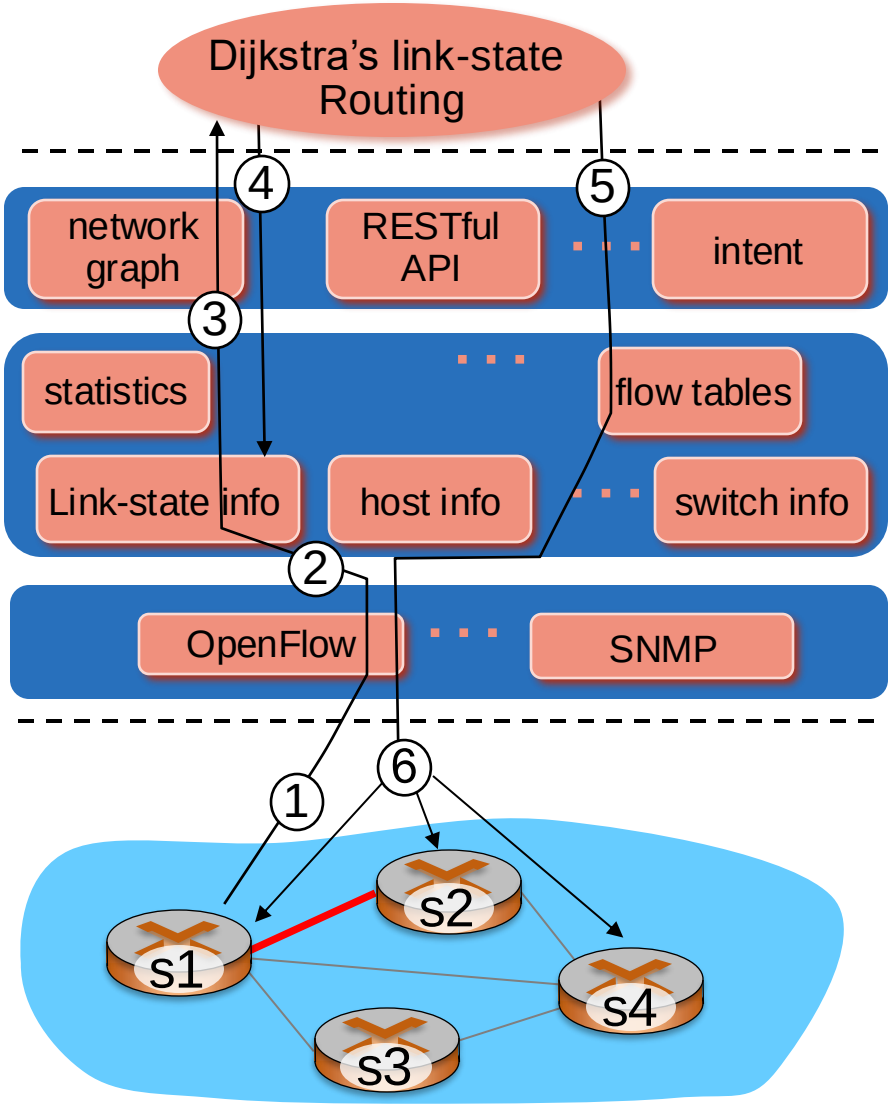
① S1, experiencing link failure using OpenFlow port status message to notify controller

② SDN controller receives OpenFlow message, updates link status info

③ Dijkstra's routing algorithm application has previously registered to be called when ever link status changes. It is called.

④ Dijkstra's routing algorithm access network graph info, link state info in controller, computes new routes

SDN: control/data plane interaction example



⑤ link state routing app interacts with flow-table-computation component in SDN controller, which computes new flow tables needed

⑥ Controller uses OpenFlow to install new tables in switches that need updating

Transport Layer

Transport Layer

our goals:

- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

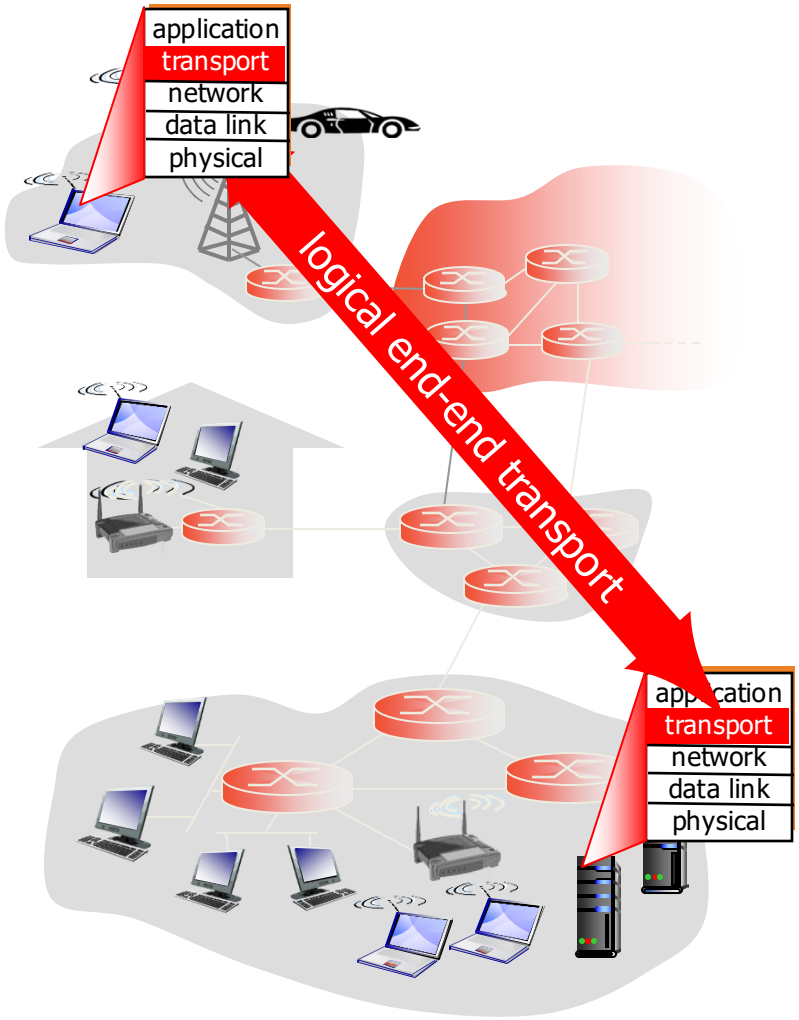
Outline

- Transport-layer services
- Multiplexing/demultiplexing
- Connectionless transport (UDP)
- Principles of reliable data transfer
- TCP

Transport Services

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



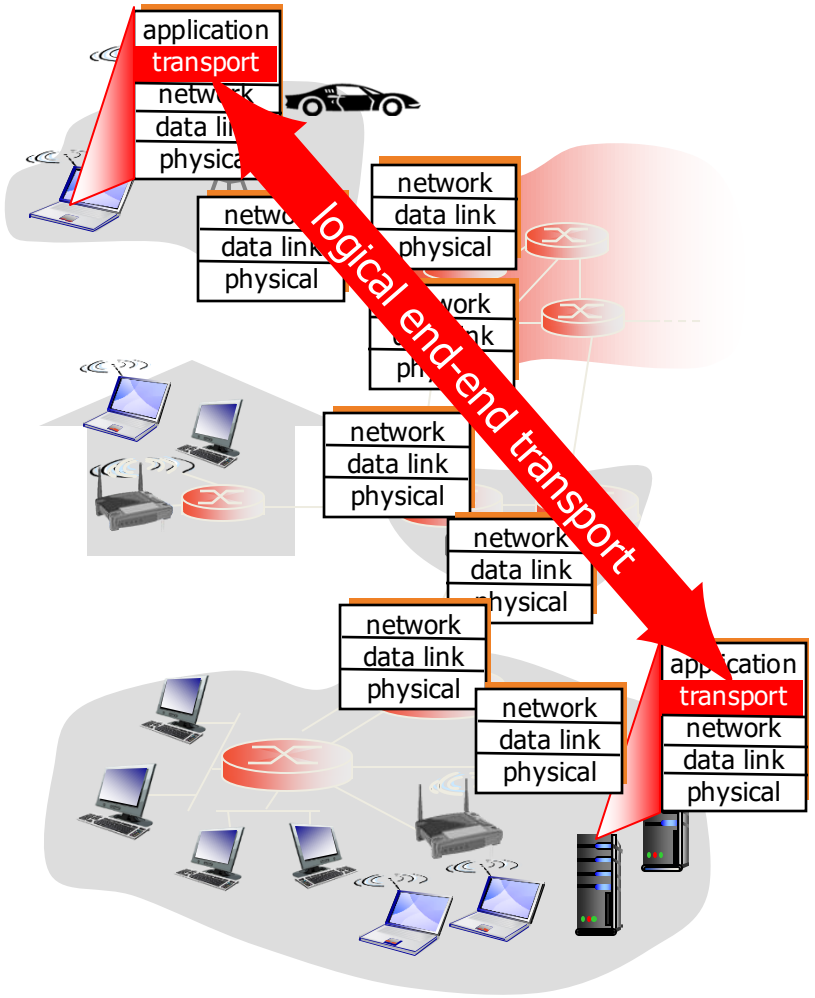
Transport vs. network layer

- *network layer:* host-to-host communication
  - best-effort, unreliable

- *transport layer:* process-to-process communication
  - relies on, enhances, network layer services

Internet transport-layer protocols

- IP: best effort service
- 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

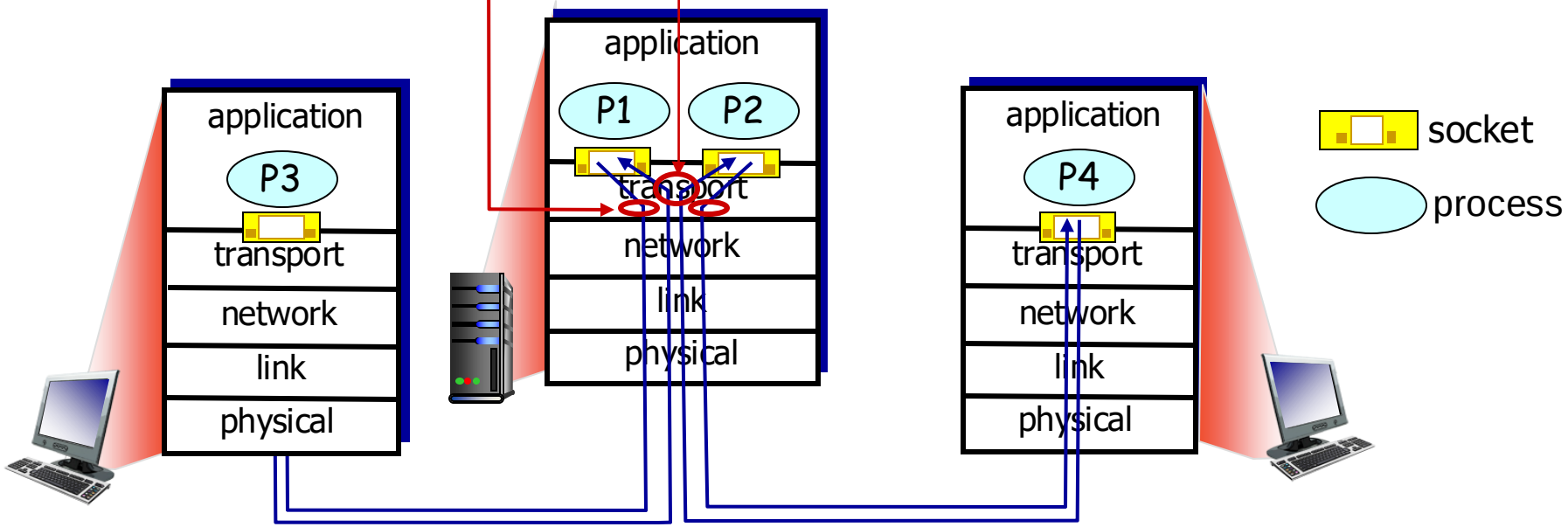


Transport Services

Multiplexing/demultiplexing

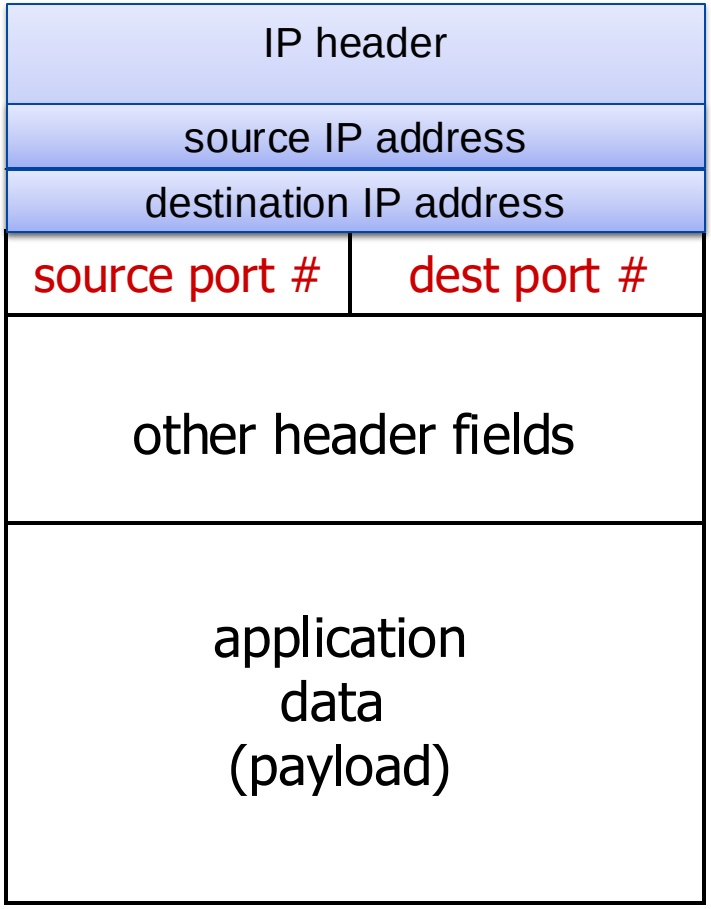
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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



How demultiplexing works

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



TCP/UDP segment format

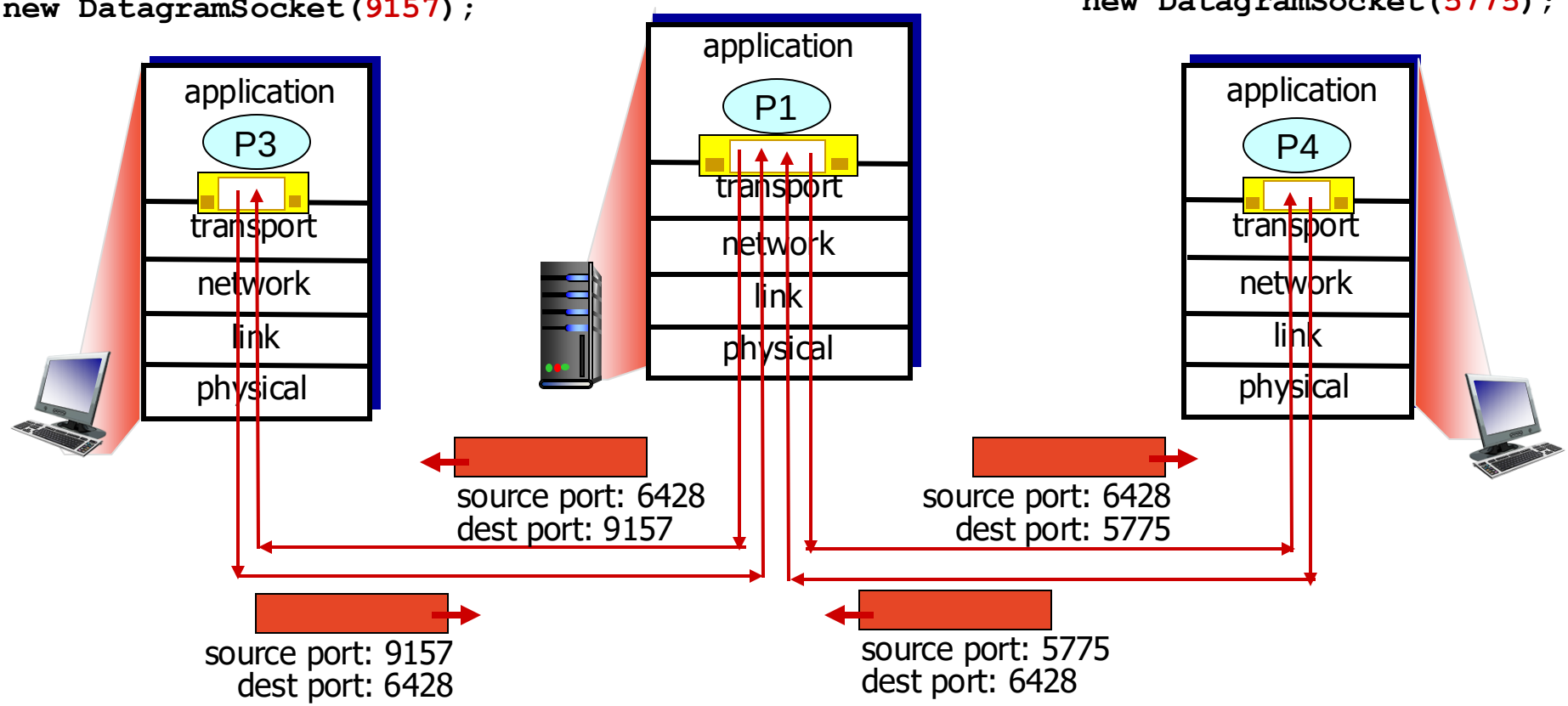
Connectionless demux: example

UDP

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

```
DatagramSocket mySocket2 =
new DatagramSocket(9157);
```

```
DatagramSocket mySocket1 =
 new DatagramSocket(5775);
```

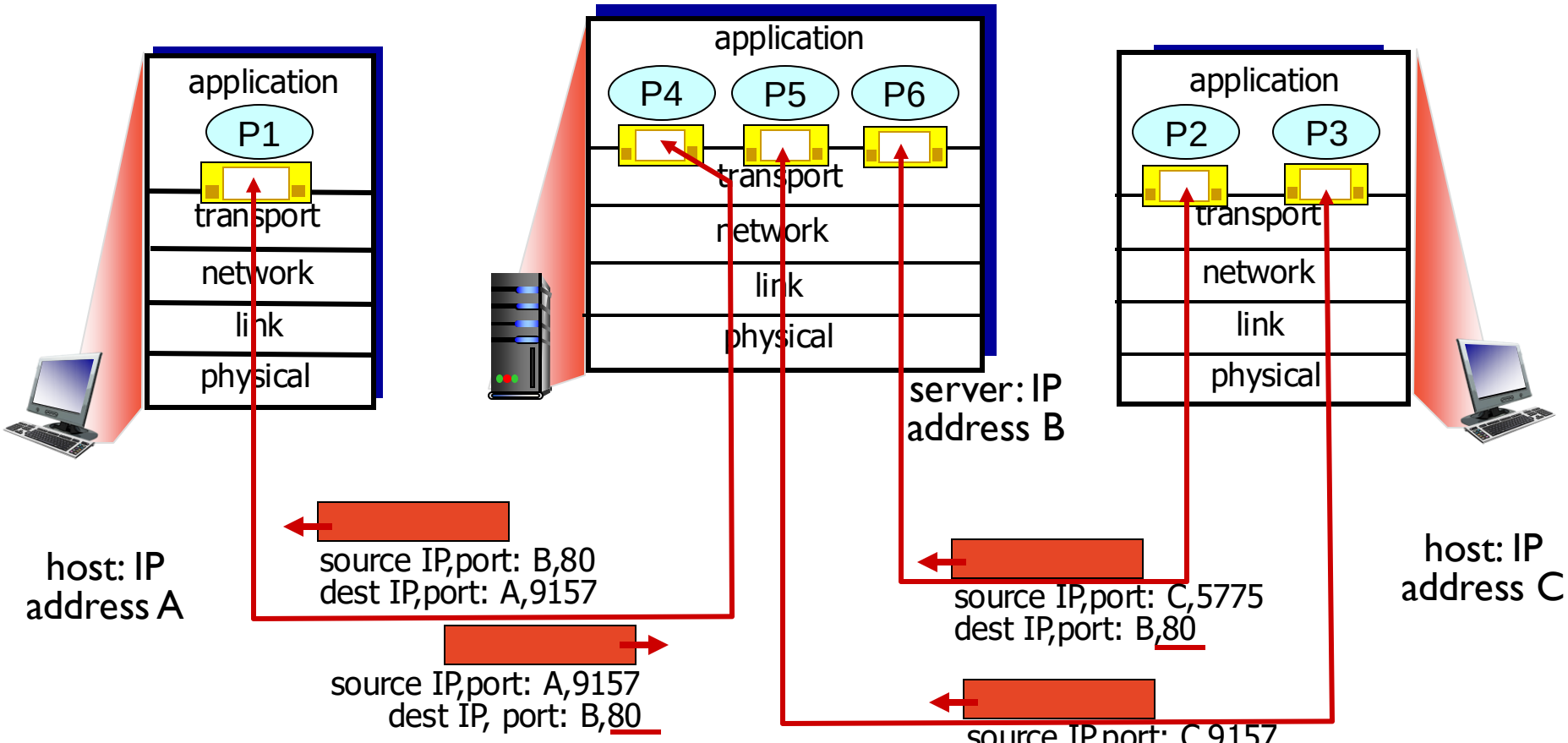


Connection-oriented demux

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

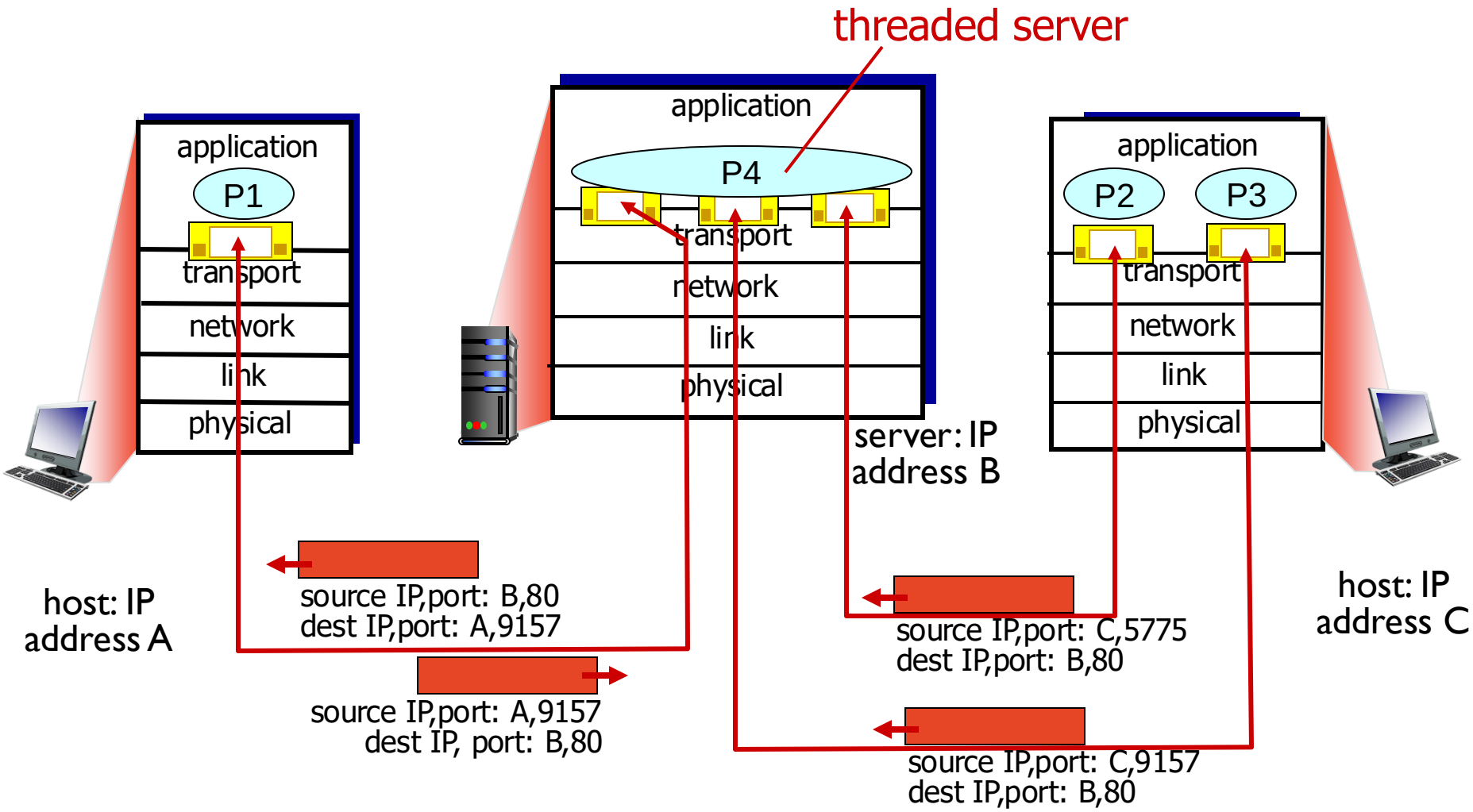
Connection-oriented demux: example

TCP



three segments, all destined to IP address: B, dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



Connectionless Transport UDP

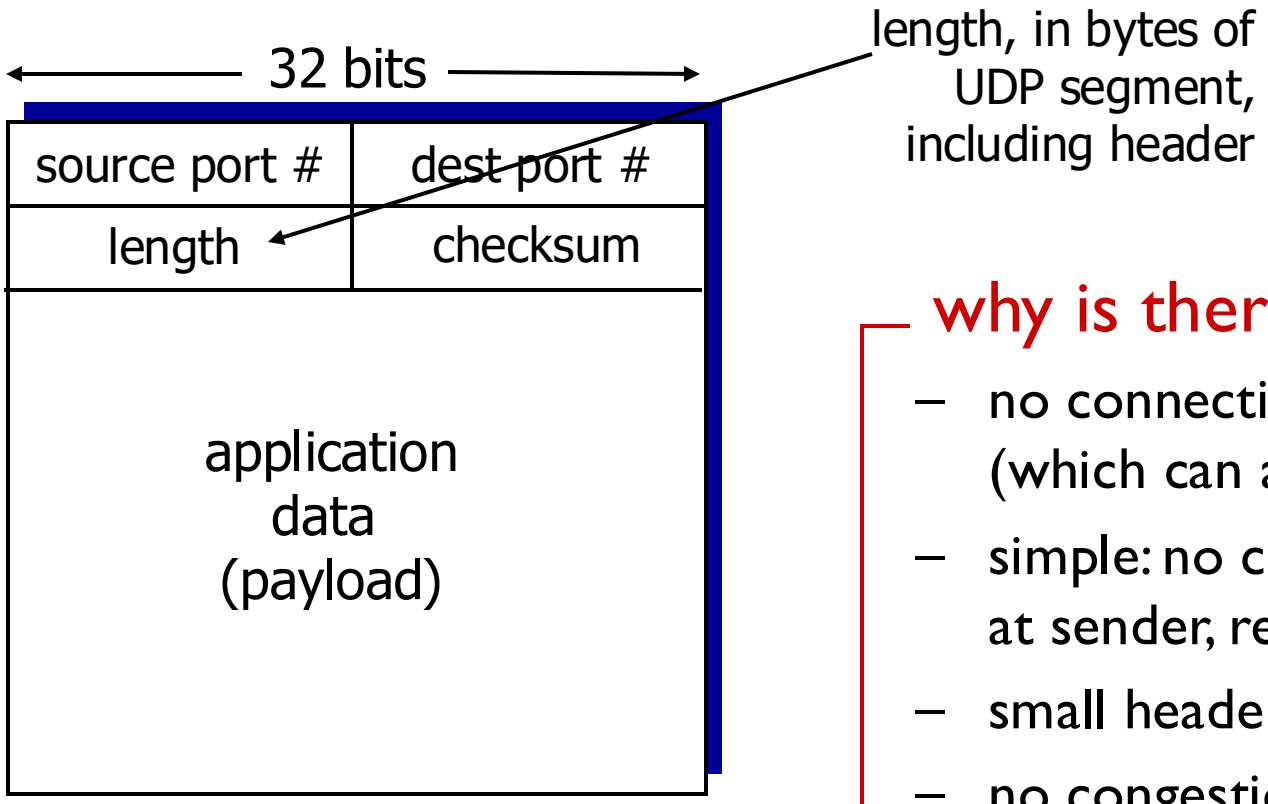
UDP: User Datagram Protocol [RFC 768]

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

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

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

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

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- sum: addition (one's complement sum) of segment contents
- checksum: complement of sum
- 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.

Internet checksum: example

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
carryout	1															
	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
wraparound	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1
sum	1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum	0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

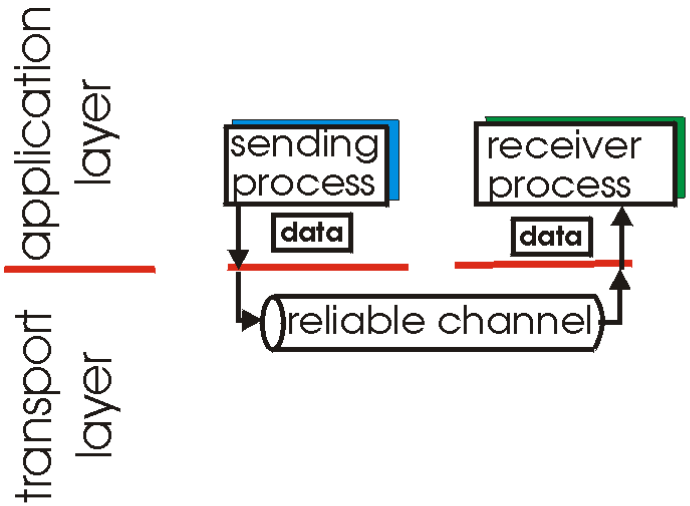
add again

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

Principles of Reliable Data Transfer

Principles of reliable data transfer

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

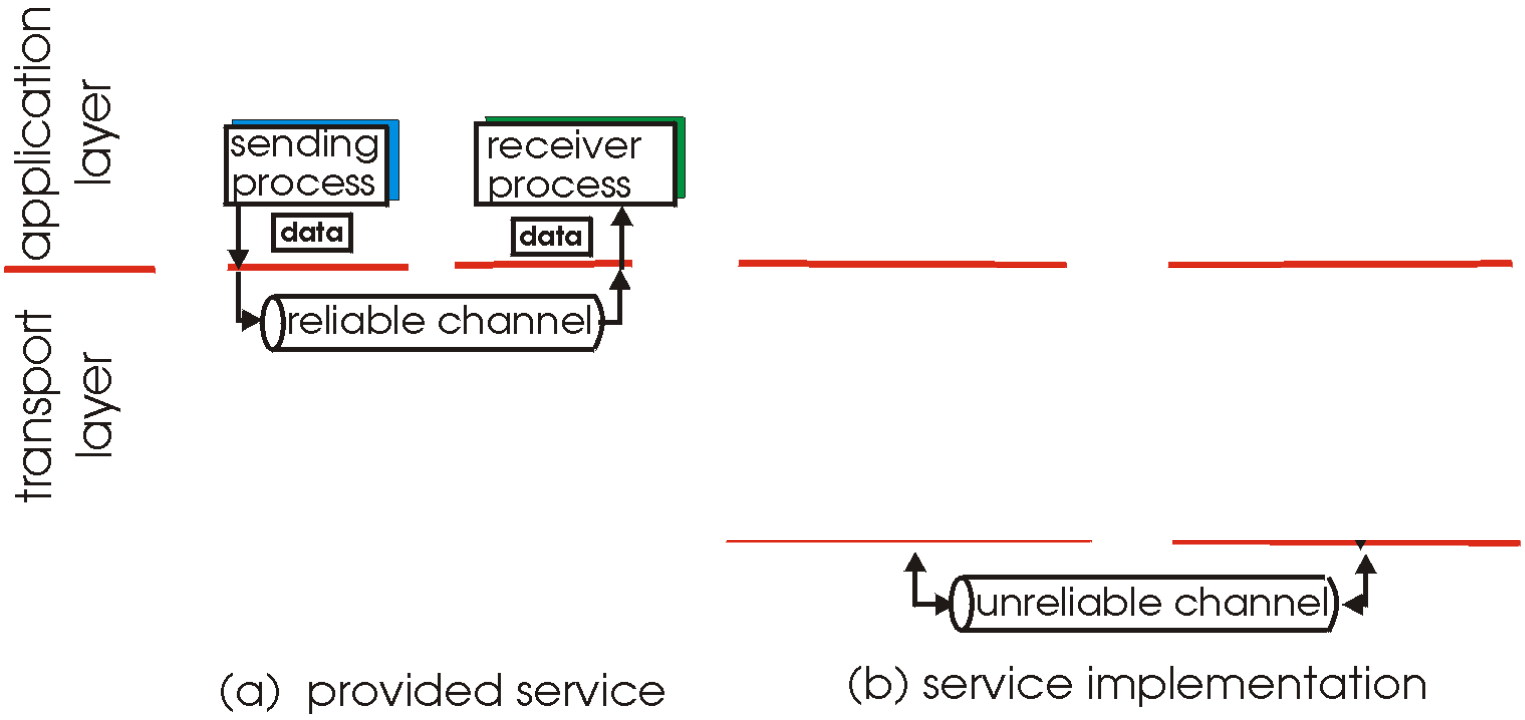


(a) provided service

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

Principles of reliable data transfer

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



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

Principles of reliable data transfer

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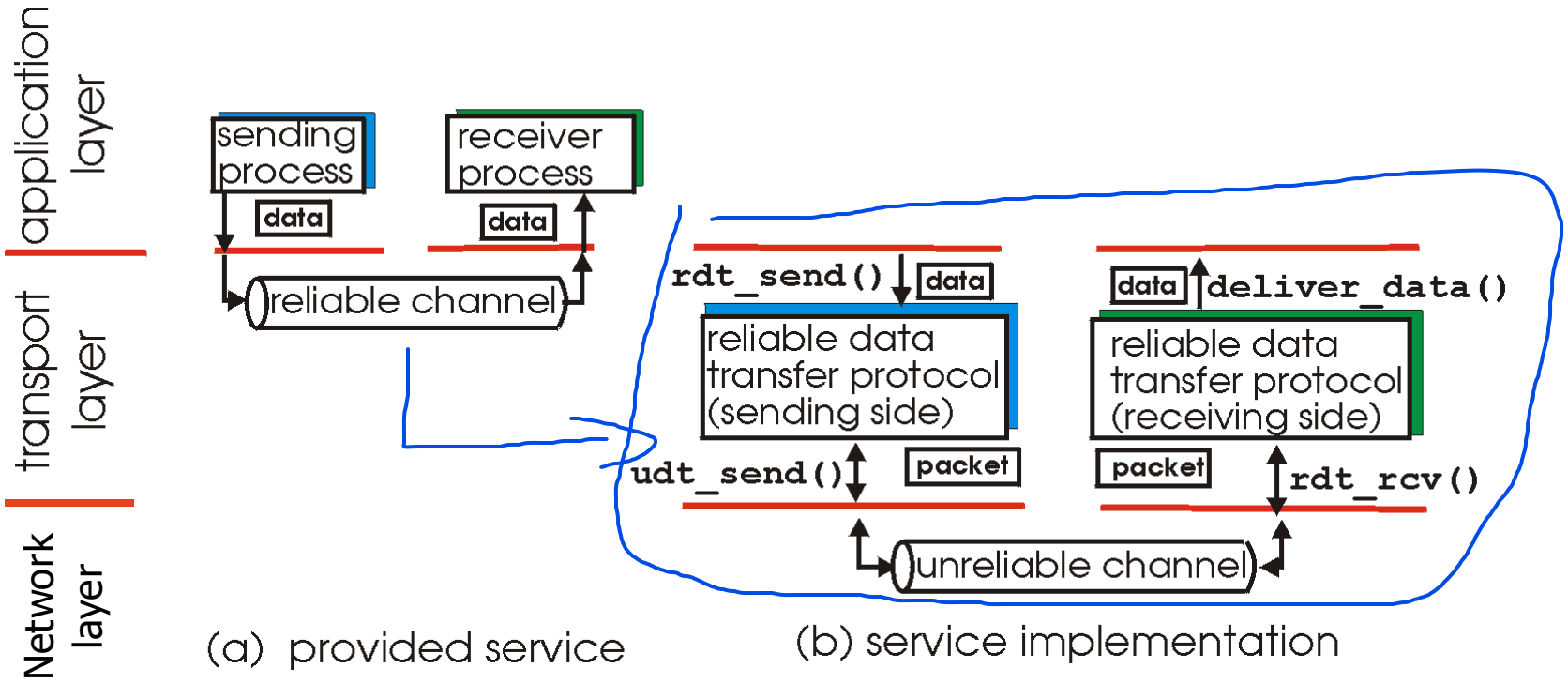
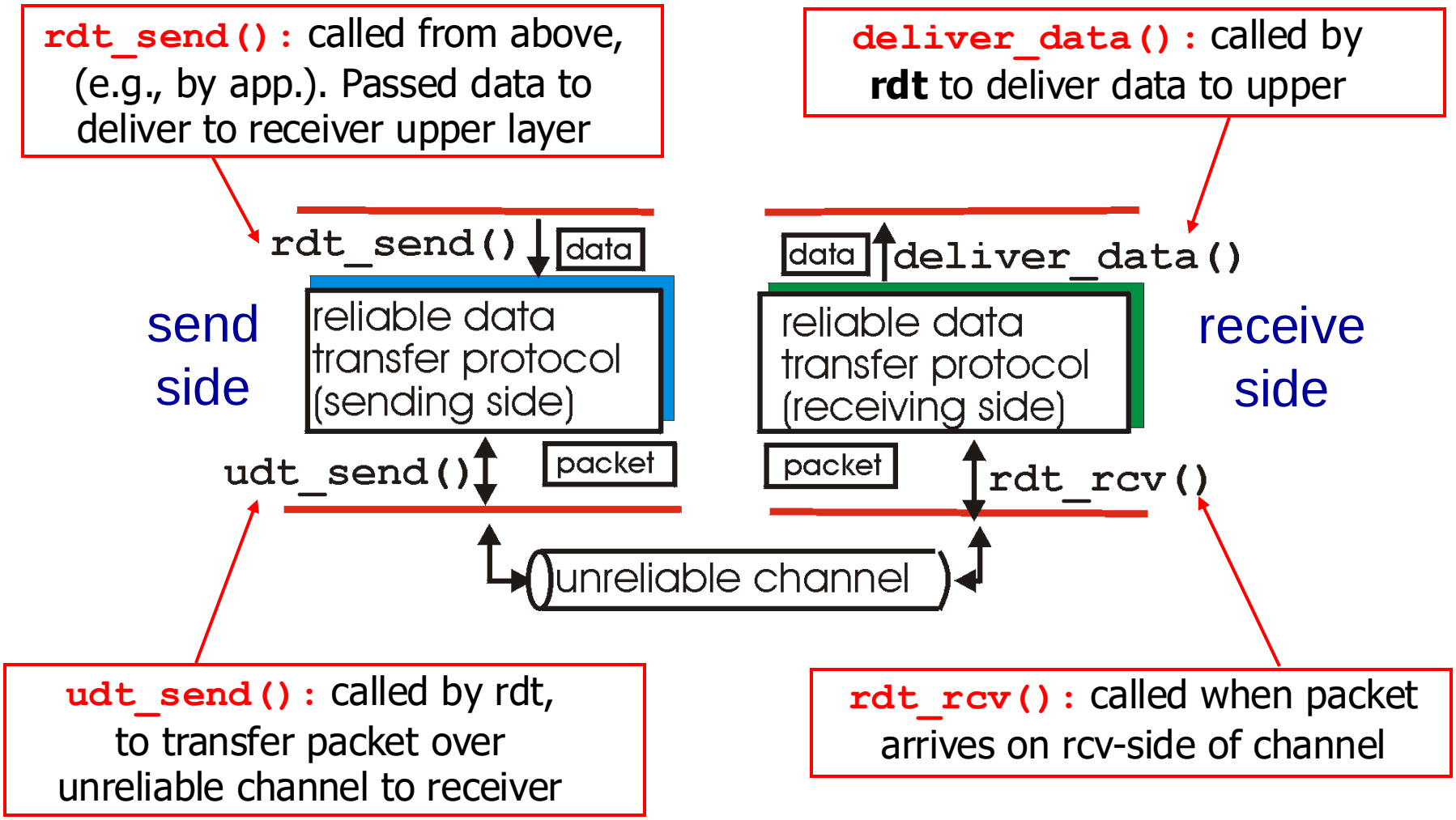


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

RDT

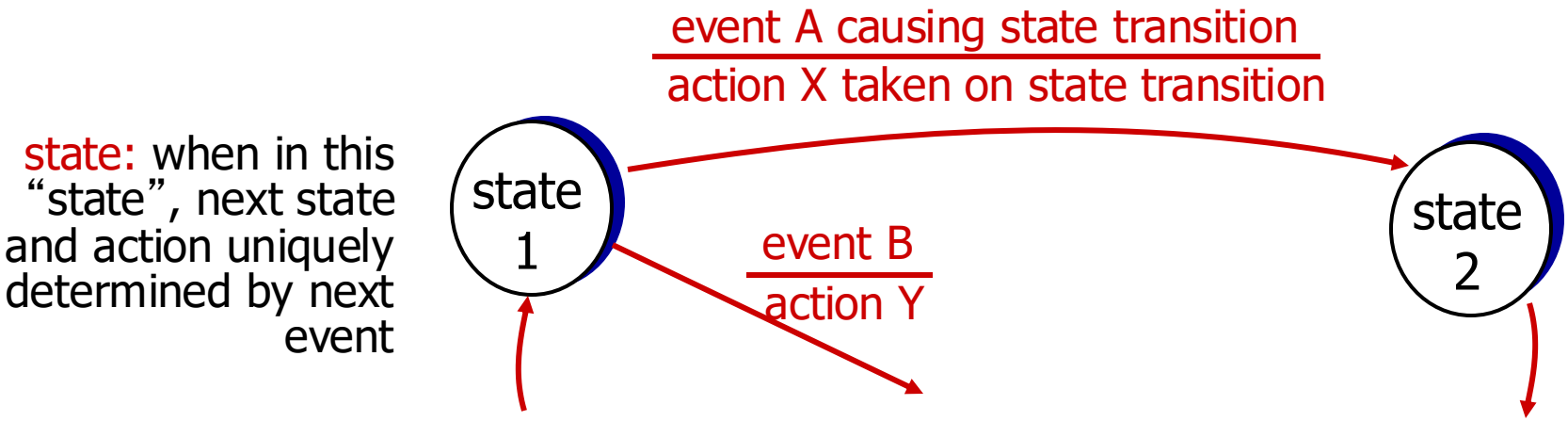
Principles of reliable data transfer



Principles of reliable data transfer

We will:

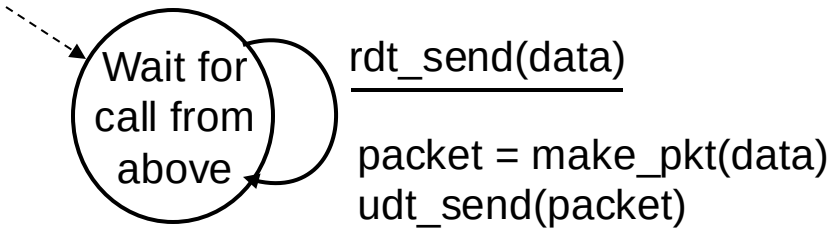
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver



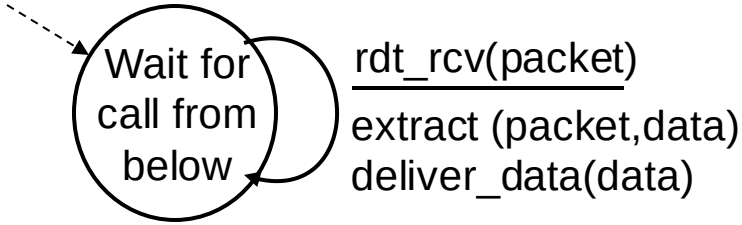
Principles of reliable data transfer

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

RDT 1.0



sender



receiver

rdt2.0: channel with bit errors

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

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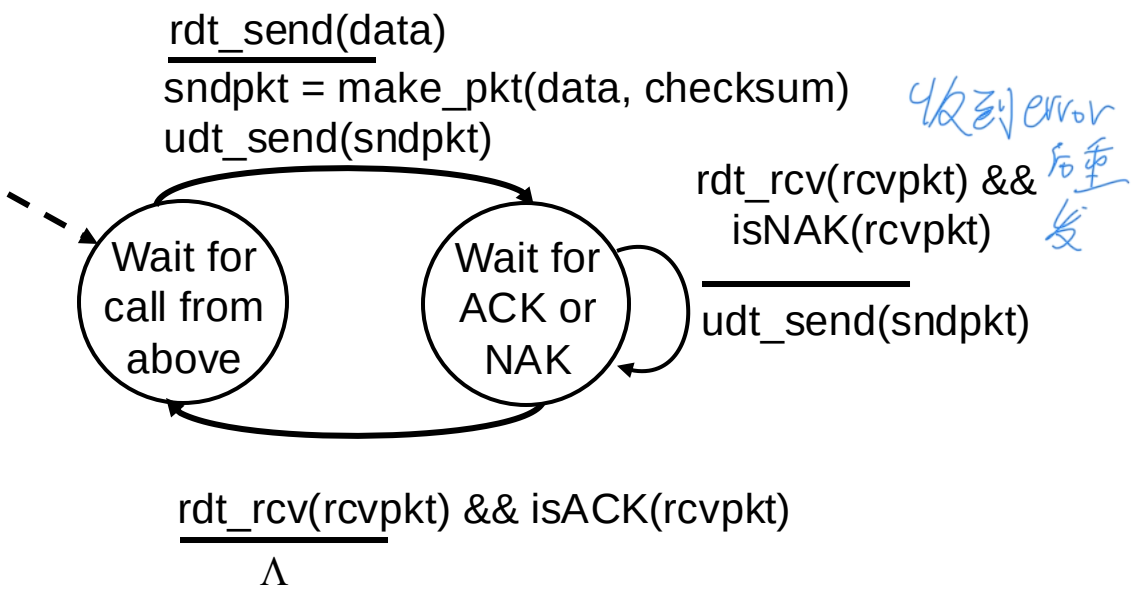
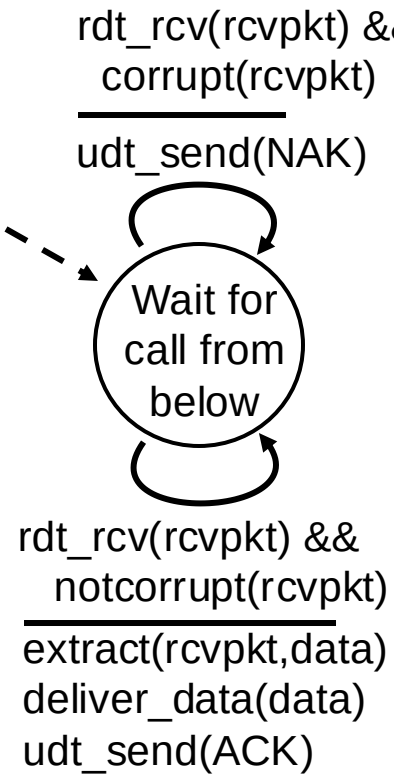
  - feedback: control msgs (ACK,NAK) from receiver to sender

rdt2.0: FSM specification

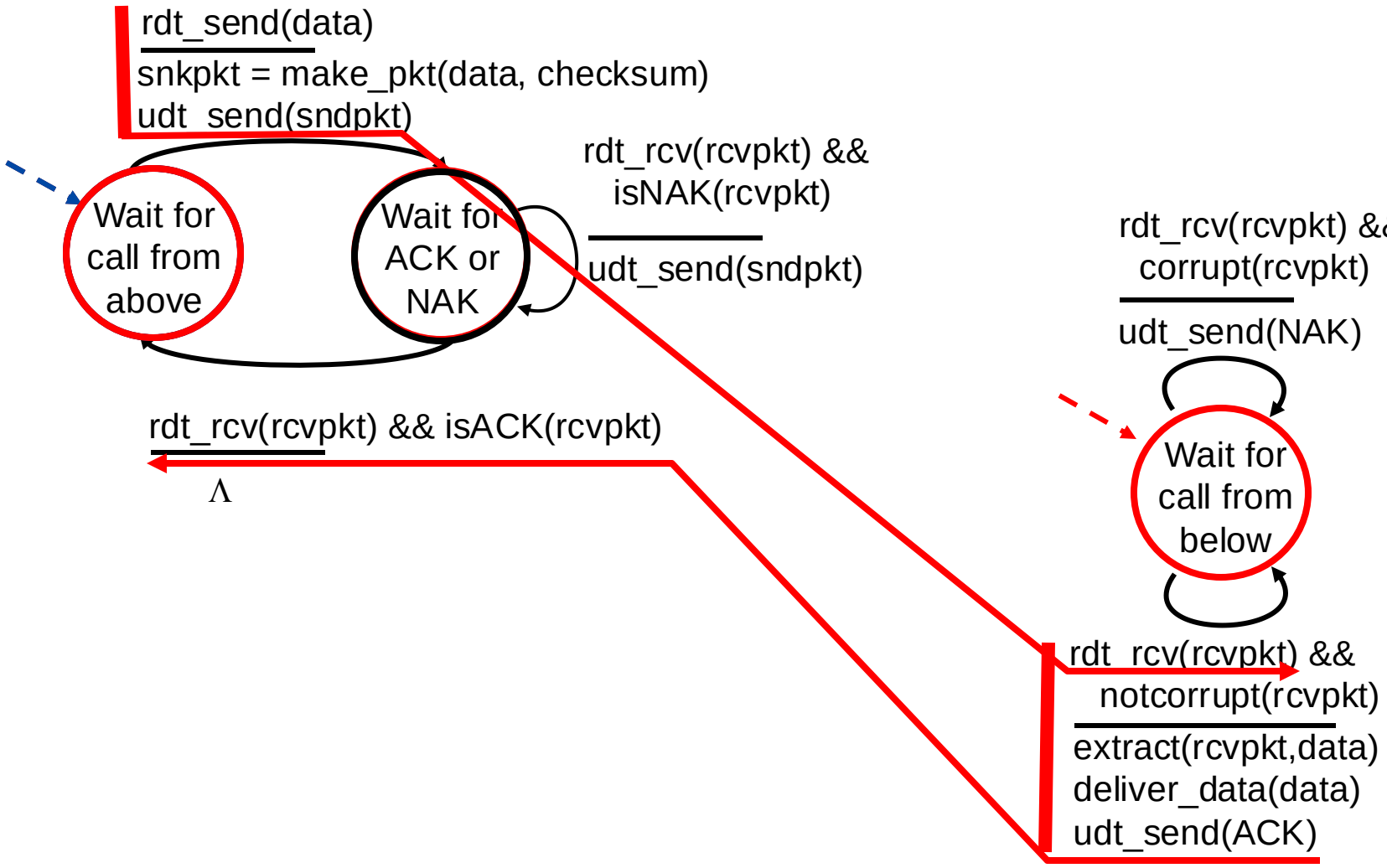


sender

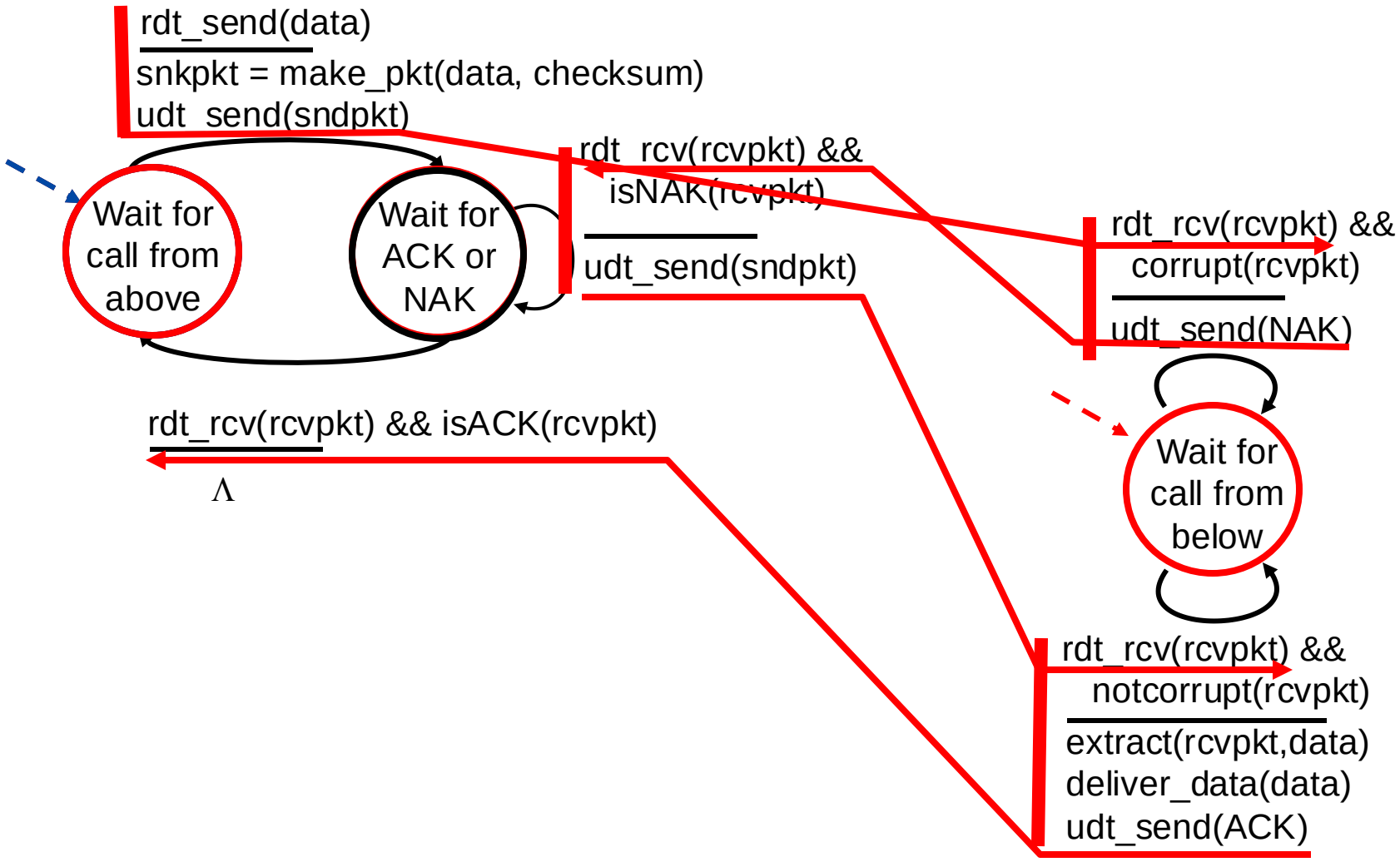
receiver



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

- sender does not know what happened at receiver!
- cannot just retransmit: possible duplicate

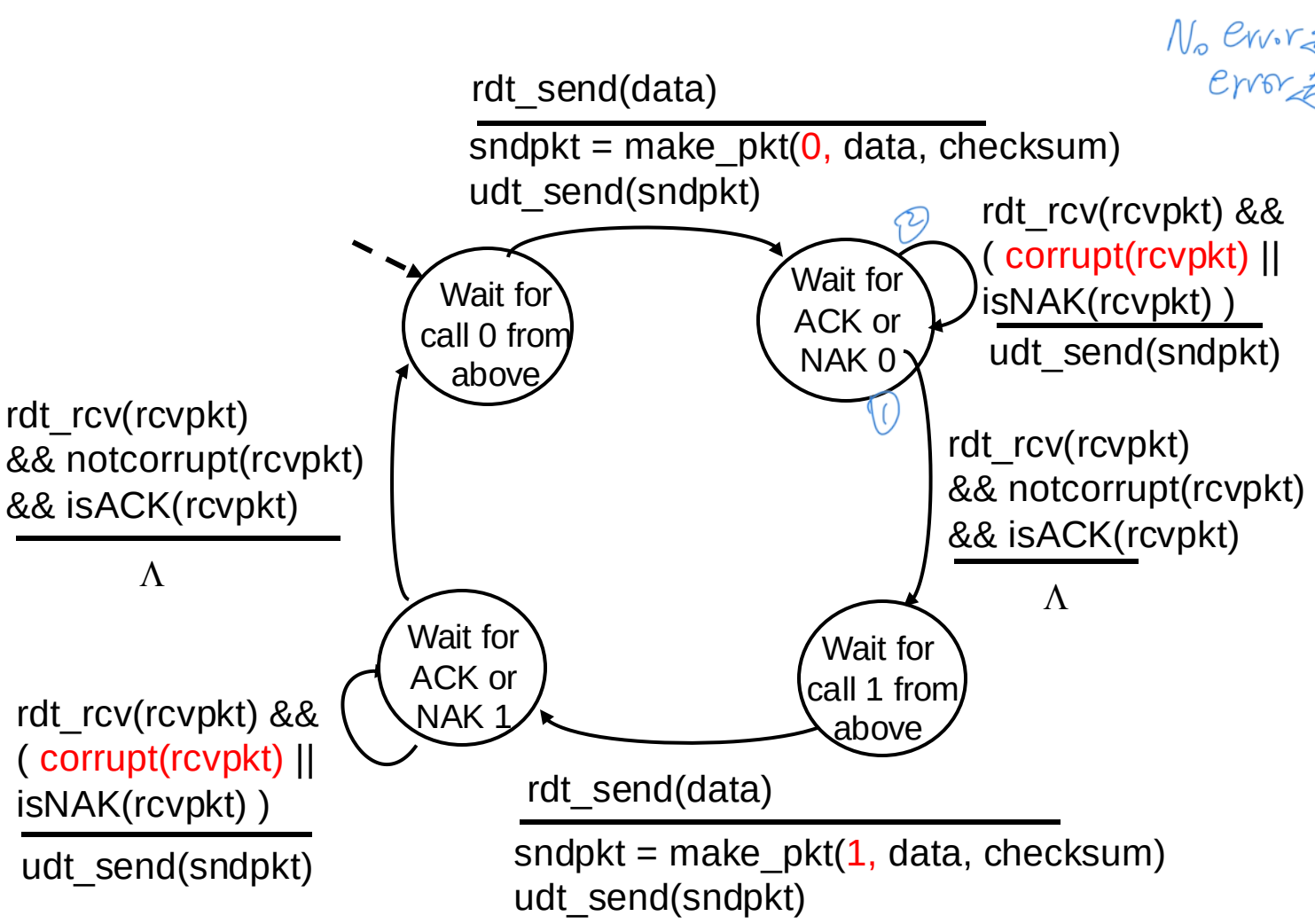
handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- sender adds *sequence number* to each pkt
- receiver discards (does not deliver up) duplicate pkt

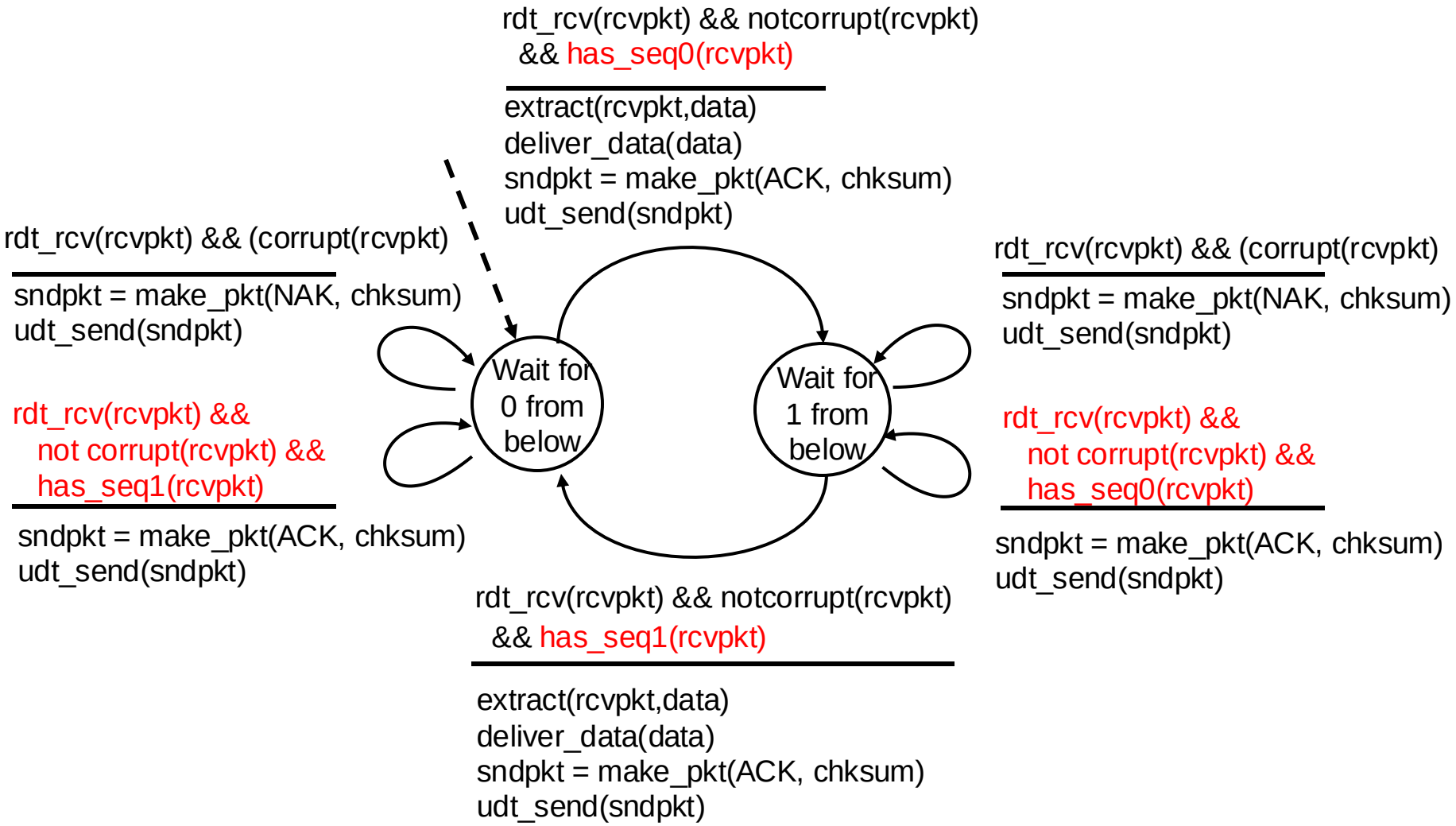
stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

sender:

- seq # added to pkt
- two seq. #' s (0,1) will suffice.
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

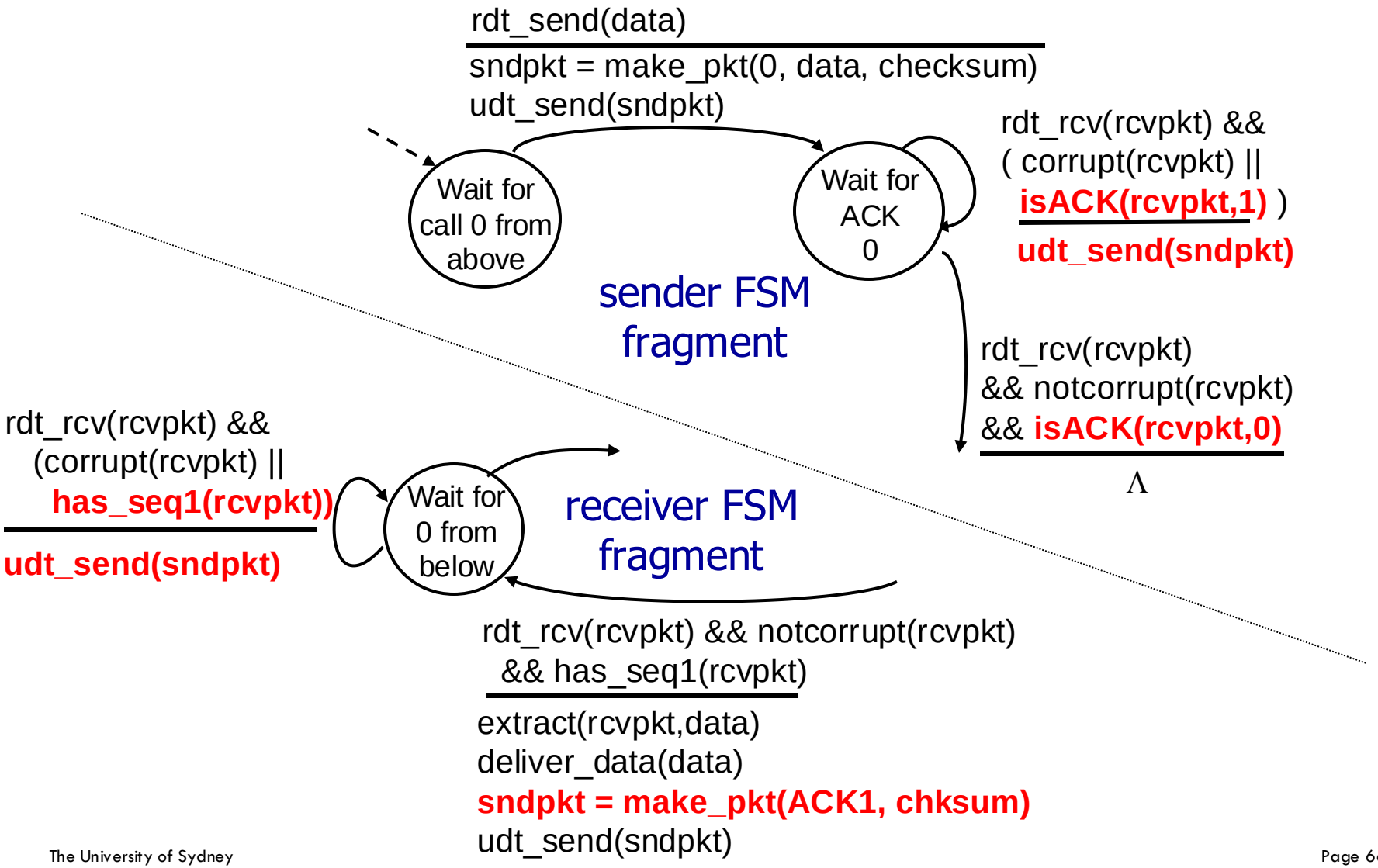
receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #

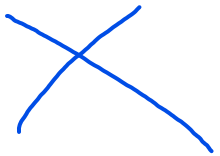
rdt2.2: a NAK-free protocol 另外一种方法

- 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
- "unexpected" ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors and loss



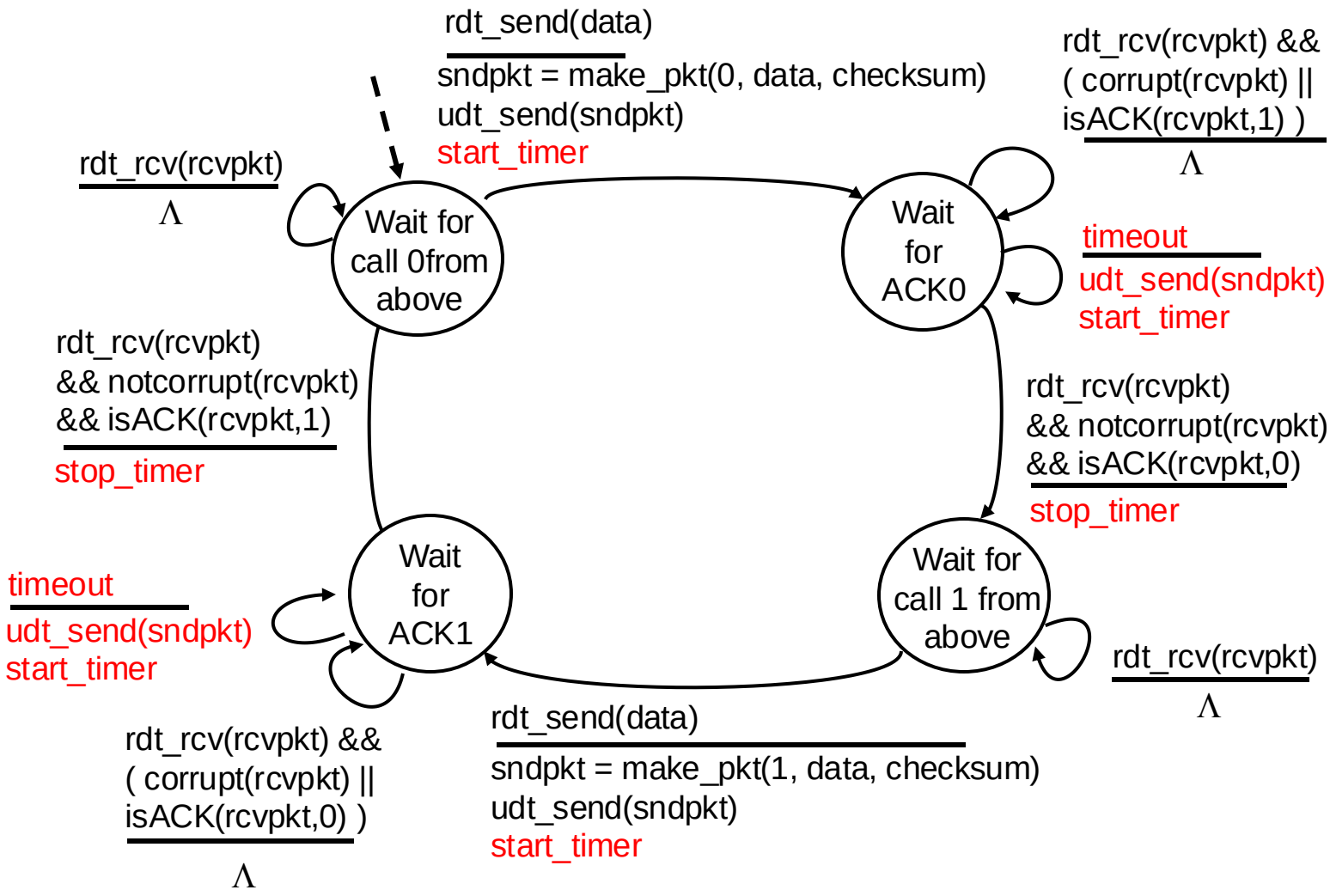
new assumption: underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

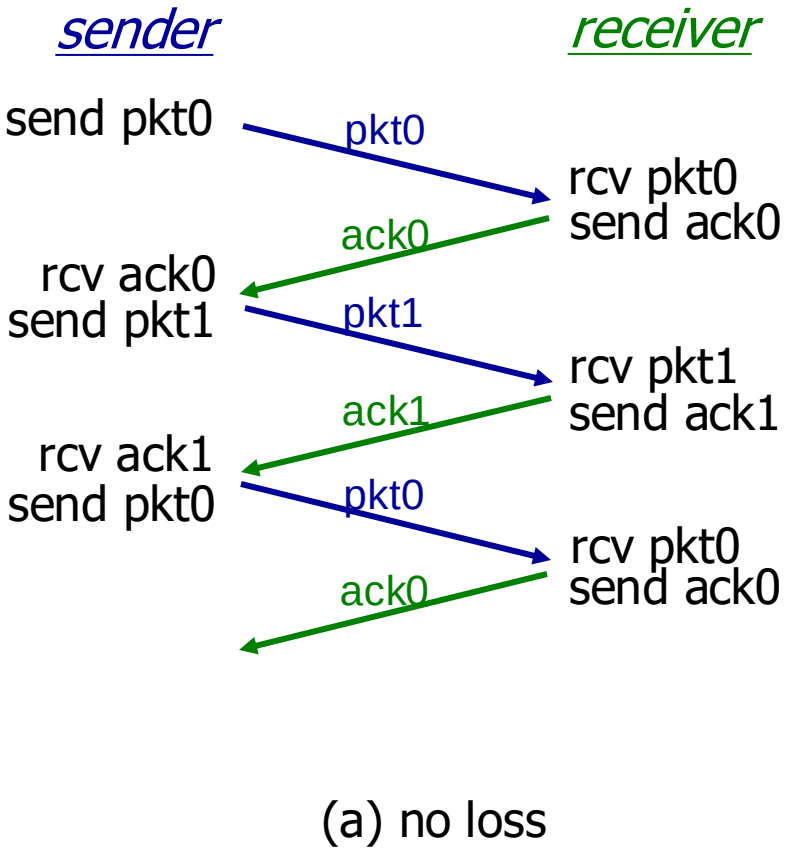
approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

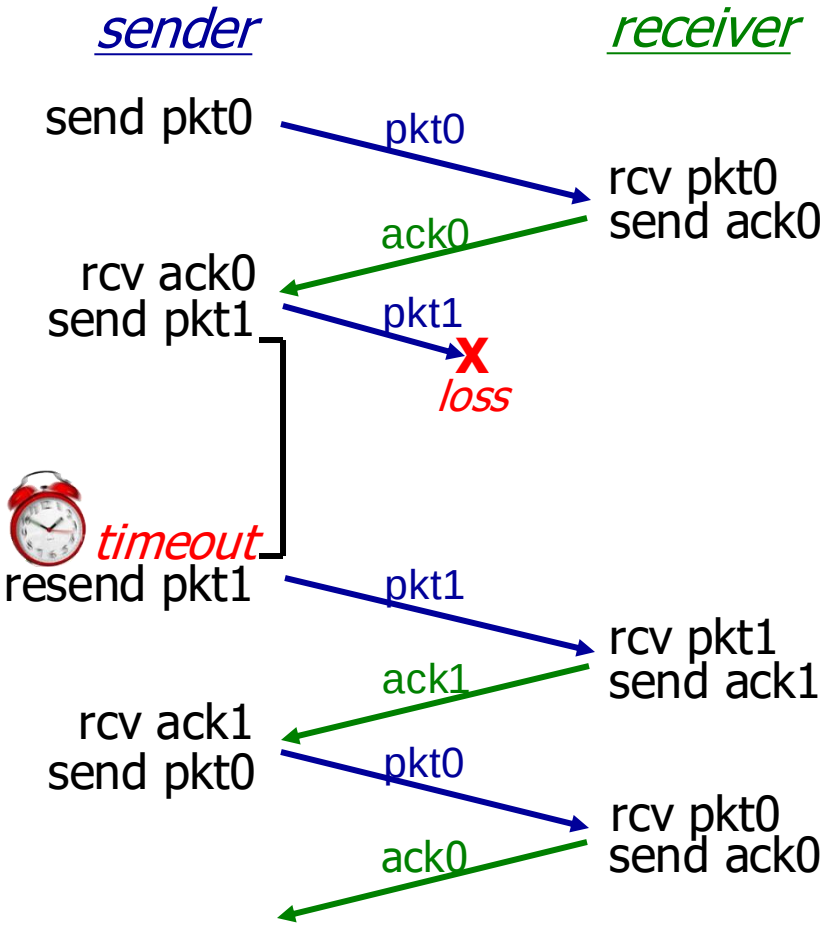
rdt3.0 sender



rdt3.0 in action

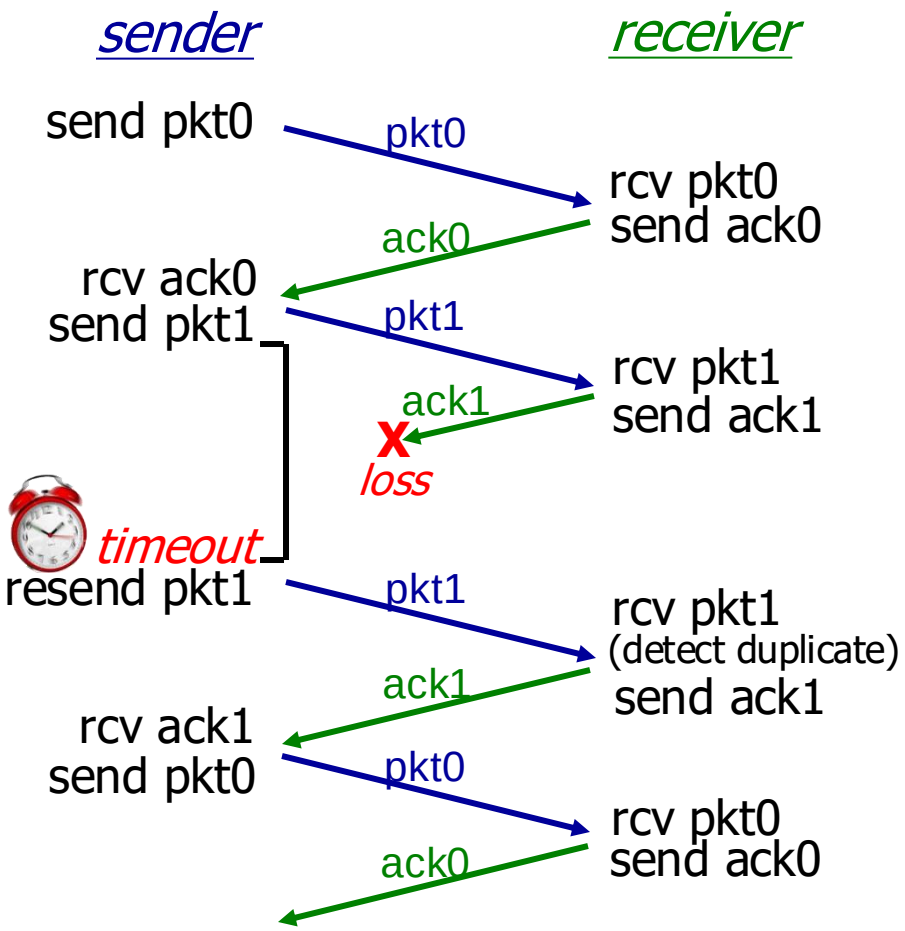


(a) no loss

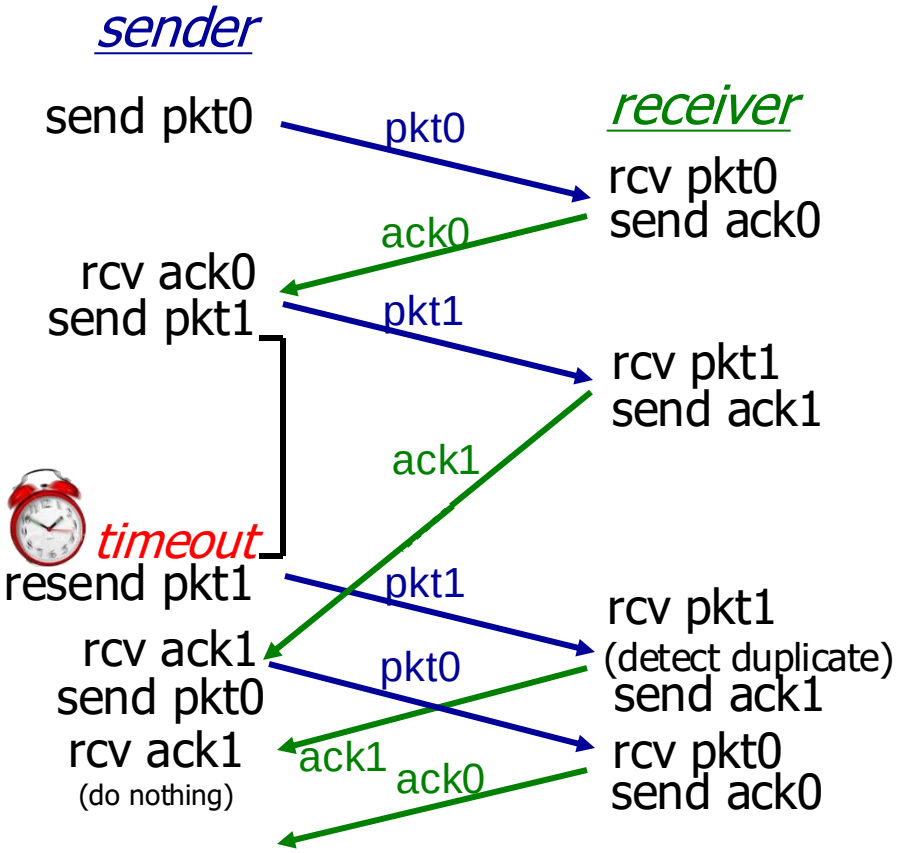


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

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

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

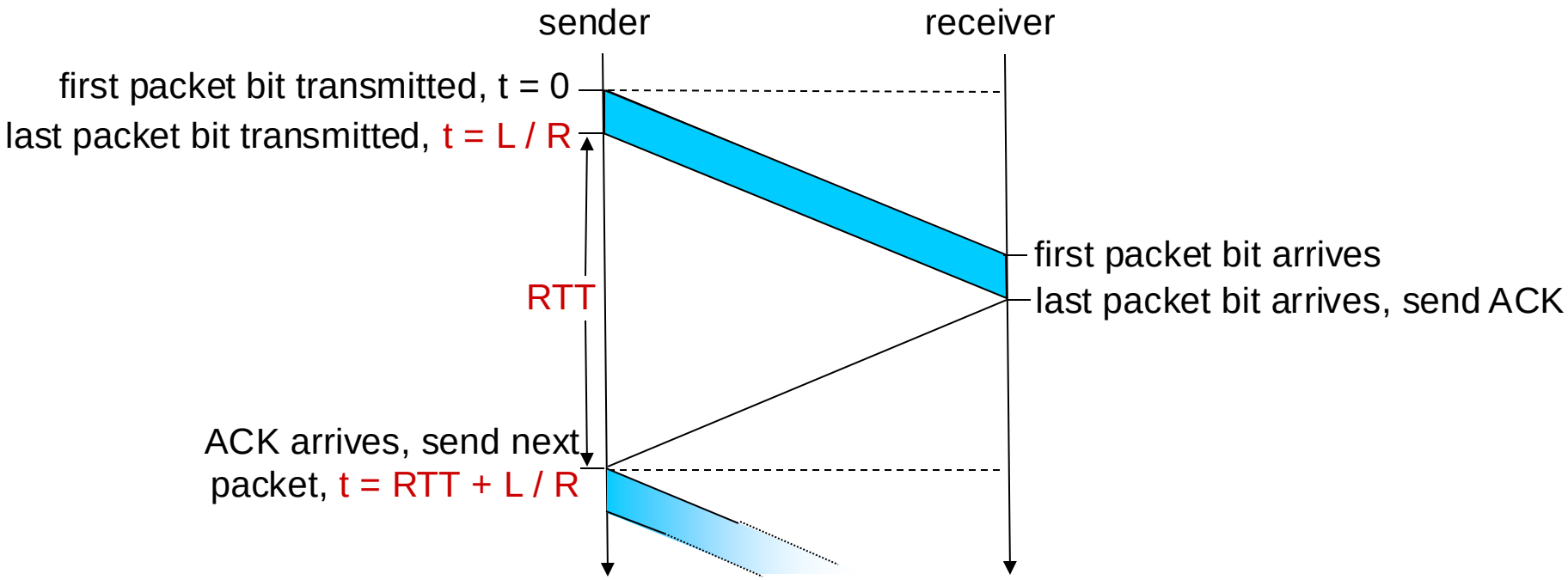
- U_{sender}: *utilization* – fraction of time sender busy sending

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

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

❖ network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation



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