Advanced Networking

Renato Lo Cigno
Renato.LoCigno@dit.unitn.it - Tel: 2026
Csaba Kiraly (helps with projects & Seminars)

Dipartimento di Informatica e Telecomunicazioni

Homepage:
www.dit.unitn.it/locigno/didattica/AdNet/

What do you find on the web site

- Exam Rules
- Exam Details … should be on ESSE3, but …
- Generic (useful) information
- Teaching Material: normally posted at least the day before the lesson
- Additional Material and links
- News, Bulletin, How to find and meet me, etc.
- …

The web site is work in progress and updated frequently, so please drop by frequently and don’t blame ME if you did’t read the last news ☺

Program

- Course Perspective
  - what do we learn and what we do not
  - are there other "networks"

- Reharsal of basics
  - Internet and TCP/IP
  - THE network? or YetAnother network
  - IP
  - UDP/TCP

Program

- IP and routing
  - OSPF and link-state protocols
    - Intra AS routing
    - performance driven routing
  - BGP and policy-based protocols
    - External routing
    - Cost (economical!) based routing
  - Global routing and Internet topology
    - How things look and works end-to-end

Program

- Network congestion
  - Network load and stability
  - Call Admission Control
  - Reactive congestion control
    - Closed-loop systems
    - Implicit/Explicit
    - Forward
    - Backward
  - TCP
    - How it really works
  - TCP stabilization methods: mith and reality
    - RED, RIO, ...

Program

- **Multicast**
  - Abstract multicasting
  - Multicast groups and addresses
  - Internet and multicast: IGMP
  - Multicast routing
  - Application level multicast
    - why it's absurd ...
    - ... why it works!!!

Program

- **Internet multimedia communications**
  - **Voice and Video services on packet-based networks**
  - **Transport: TRP/RTCP**
  - **H.323 standard**
  - **SIP standard**
  - **Skype and P2P approaches**

  - **IP TV**
    - VoD/Broadcast/Live
    - Traditional approach
    - P2P systems

- **Recalling known topics:**

  - Internet
  - IP
  - UDP/TCP

Acknowledment:
The following slides are based on the slides developed by J.Kurose and K.Ross to accompany their book “Computer Networks: A Top Down Approach Featuring the Internet” by Wiley edts.

Internet

What we see:

- Services
- Applications we use
- Some "application level" protocols
- Throughput
- Losses
- Delay (sometimes)
- Delay Jitter (if we're really skilled!)

What is it:

- A collection of protocols
- Mainly centered around two centerpieces:
  - **IP** (network layer)
  - **UDP/TCP** (transport layer)
- Does not mandate a physical medium or format
- Does not mandate or limit the services/applications above (integrates services)

IP: The Network Layer

Goals:

- recall principles behind network layer services:
  - routing (path selection)
  - dealing with scale
  - how a router works
- instantiation and implementation in the Internet

Overview:

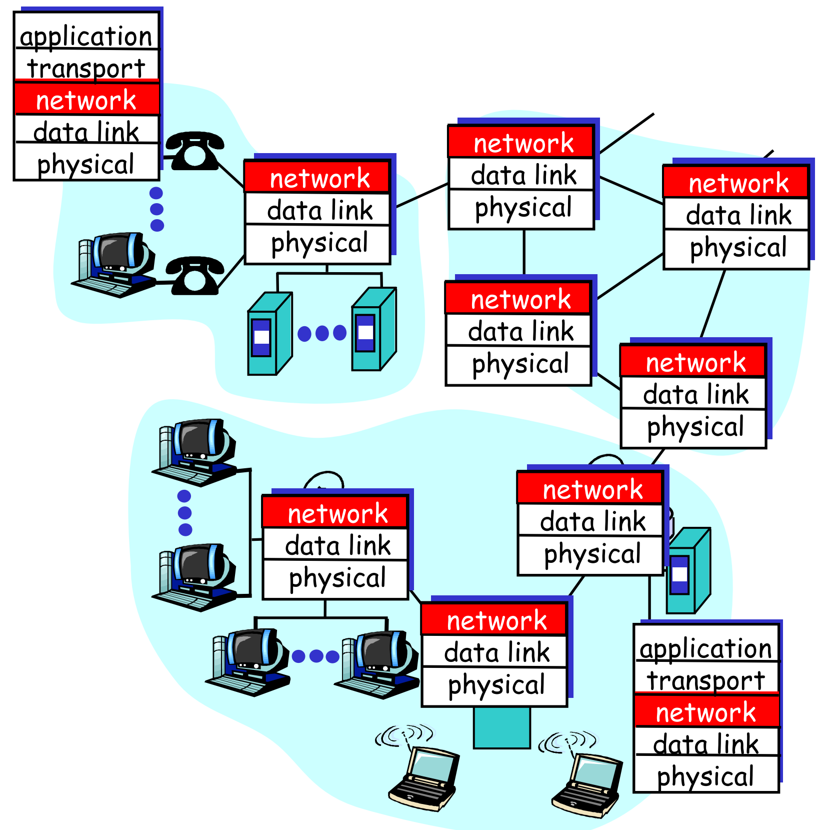
- network layer services
- routing principle: path selection
- IP
- Internet routing protocols reliable transfer
  - intra-domain
  - inter-domain
- what's inside a router?

Network layer functions

- transport packet from sending to receiving hosts
- network layer protocols in *every* host, router

three important functions:

- *path determination:* route taken by packets from source to dest. *Routing algorithms*
- *switching:* move packets from router's input to appropriate router output
- *call setup:* some network architectures require router call setup along path before data flows



Network service model

Q: What *service model* for "channel" transporting packets from sender to receiver?

service abstraction

- guaranteed bandwidth?
- preservation of inter-packet timing (no jitter)?
- loss-free delivery?
- in-order delivery?
- congestion feedback to sender?

The most important abstraction provided by network layer:

virtual circuit
or
datagram?

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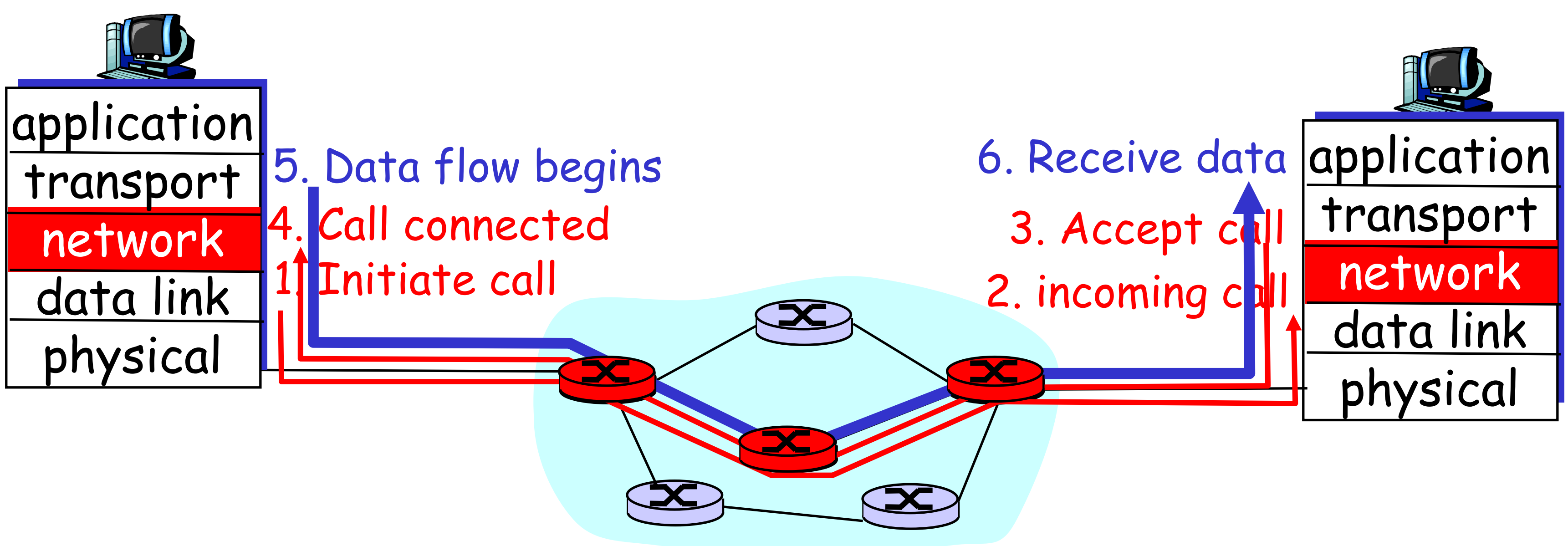
Virtual circuits

"source-to-dest path behaves much like telephone circuit"
- performance-wise
- network actions along source-to-dest path

- call setup, teardown for each call *before* data can flow
- each packet carries VC identifier (not destination host OD)
- *every* router on source-dest path s maintain "state" for each passing connection
  - transport-layer connection only involved two end systems
- link, router resources (bandwidth, buffers) may be *allocated* to VC
  - to get circuit-like perf.

Virtual circuits: signaling protocols

- used to setup, maintain teardown VC
- used in ATM, frame-relay, X.25
- not used in today's Internet



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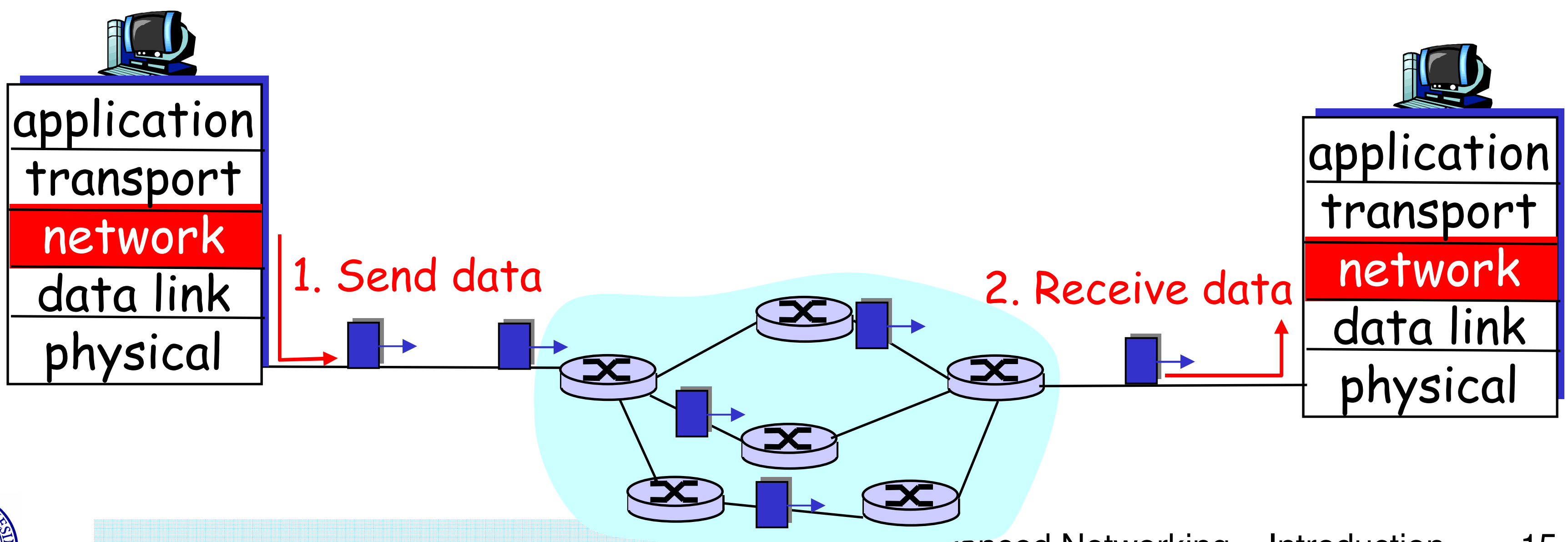
Datagram networks: the Internet model

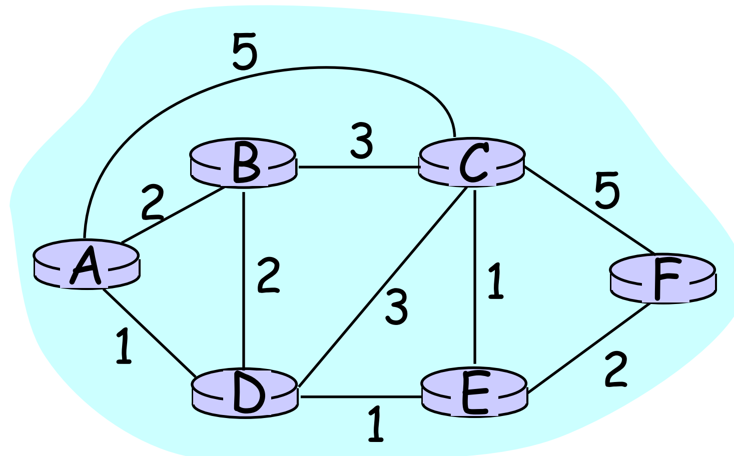
- no call setup at network layer
- routers: no state about end-to-end connections
  - no network-level concept of “connection”
- packets typically routed using destination host ID
  - packets between same source-dest pair may take different paths



Routing

Routing protocol

Goal: determine "good" path (sequence of routers) thru network from source to dest.



Graph abstraction for routing algorithms:

- graph nodes are routers
- graph edges are physical links
  - link cost: delay, $ cost, or congestion level

- "good" path:
  - typically means minimum cost path
  - other def's possible

Routing Algorithm classification

Global or decentralized information?

Global:

- all routers have complete topology, link cost info
- "link state" algorithms

Decentralized:

- router knows physically-connected neighbors, link costs to neighbors
- iterative process of computation, exchange of info with neighbors
- "distance vector" algorithms

Static or dynamic?

Static:

- routes change slowly over time

Dynamic:

- routes change more quickly
  - periodic update
  - in response to link cost changes

A Link-State Routing Algorithm

Dijkstra's algorithm

- net topology, link costs known to all nodes
  - accomplished via "link state broadcast"
  - all nodes have same info
- computes least cost paths from one node ('source") to all other nodes
  - gives routing table for that node
- iterative: after k iterations, know least cost path to k dest.'s

Notation:

- c(i,j): link cost from node i to j. cost infinite if not direct neighbors
- D(v): current value of cost of path from source to dest. V
- p(v): predecessor node along path from source to v, that is next v
- N: set of nodes whose least cost path definitively known

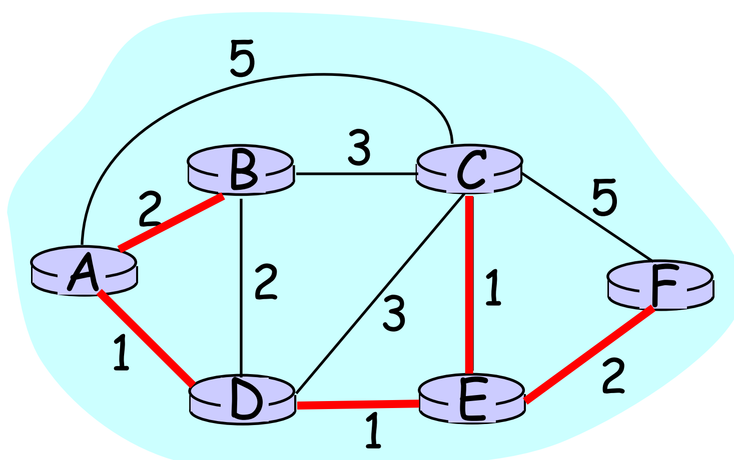
Dijsktra's Algorithm

```
Initialization:
  N = {A}
  for all nodes v
    if v adjacent to A
      then D(v) = c(A,v)
      else D(v) = infty

Loop
   find w not in N such that D(w) is a minimum
   add w to N
   update D(v) for all v adjacent to w and not in N:
      D(v) = min( D(v), D(w) + c(w,v) )
   /* new cost to v is either old cost to v or known
    shortest path cost to w plus cost from w to v */
 until all nodes in N
```

Dijkstra's algorithm: example

Step	start N	D(B),p(B)	D(C),p(C)	D(D),p(D)	D(E),p(E)	D(F),p(F)
0	A	2,A	5,A	1,A	infinity	infinity
1	AD	2,A	4,D		2,D	infinity
2	ADE	2,A	3,E			4,E
3	ADEB		3,E			4,E
4	ADEBC					4,E
5	ADEBCF					



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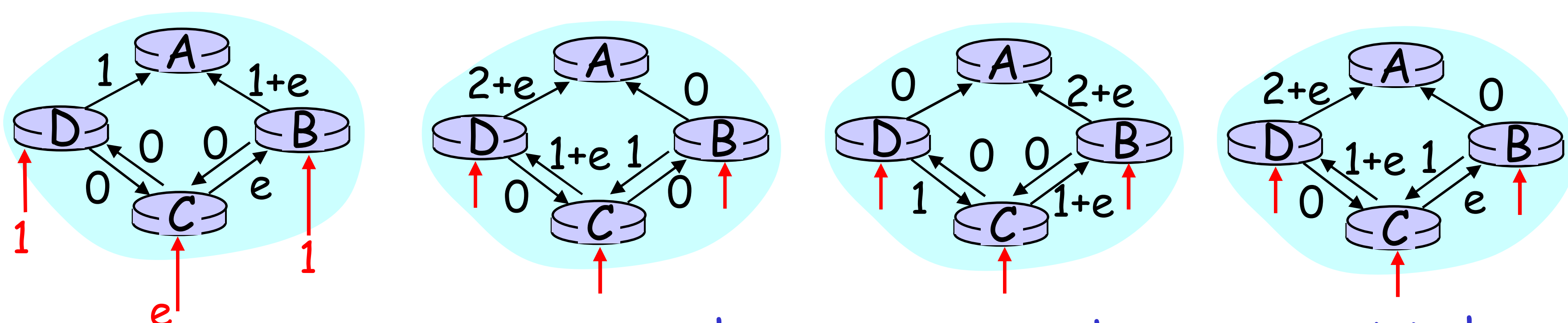
Dijkstra’s algorithm, discussion

Algorithm complexity: n nodes

- each iteration: need to check all nodes, w, not in N
- n*(n+1)/2 comparisons: O(n**2)
- more efficient implementations possible: O(nlogn)

Oscillations possible:

- e.g., link cost = amount of carried traffic



initially … recompute routing … recompute … recompute

Distance Vector Routing Algorithm

iterative:

- continues until no nodes exchange info.
- *self-terminating*: no "signal" to stop

asynchronous:

- nodes need *not* exchange info/iterate in lock step!

distributed:

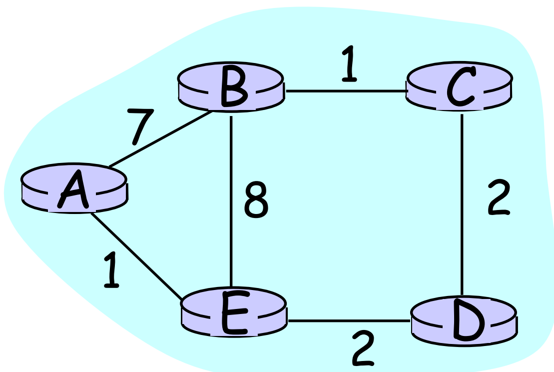
- each node communicates *only* with directly-attached neighbors

Distance Table data structure

- each node has its own
- row for each possible destination
- column for each directly-attached neighbor to node
- example: in node X, for dest. Y via neighbor Z:

$$D^X(Y,Z) = \text{distance } \textit{from } X \textit{ to } Y, \textit{ via } Z \text{ as next hop} = c(X,Z) + \min_w\{D^Z(Y,w)\}$$

Distance Table: example



$$D^E(C,D) = c(E,D) + \min_w\{D^D(C,w)\}$$
$$= 2+2 = 4$$

$$D^E(A,D) = c(E,D) + \min_w\{D^D(A,w)\}$$
$$= 2+3 = 5 \quad \text{loop!}$$

$$D^E(A,B) = c(E,B) + \min_w\{D^B(A,w)\}$$
$$= 8+6 = 14 \quad \text{loop!}$$

cost to destination via

$D^E()$ (destination)	A	B	D
A	1	14	5
B	7	8	5
C	6	9	4
D	4	11	2

Distance table gives routing table

cost to destination via

$D^E()$	A	B	D
A	1	14	5
B	7	8	5
C	6	9	4
D	4	11	2

destination

	Outgoing link to use, cost
A	A,1
B	D,5
C	D,4
D	D,4

destination

Distance table → Routing table

Distance Vector Routing: overview

Iterative, asynchronous:

each local iteration caused by:

- local link cost change
- message from neighbor: its least cost path change from neighbor

Distributed:

- each node notifies neighbors *only* when its least cost path to any destination changes
  - neighbors then notify their neighbors if necessary

Each node:

wait for (change in local link cost of msg from neighbor)

recompute distance table

if least cost path to any dest has changed, *notify* neighbors

Distance Vector Algorithm:

At all nodes, X:

```
Initialization:
 for all adjacent nodes v:
   D^X(*,v) = infty        /* the * operator means "for all rows" */
   D^X(v,v) = c(X,v)
 for all destinations, y
   send min_w D^X(y,w) to each neighbor  /* w over all X's neighbors */
```

Distance Vector Algorithm (cont.):

loop
  wait (until I see a link cost change to neighbor V
        or until I receive update from neighbor V)

  if (c(X,V) changes by d)
    /* change cost to all dest's via neighbor v by d */
    /* note: d could be positive or negative */
    for all destinations y: $D_X(y,V) = D_X(y,V) + d$

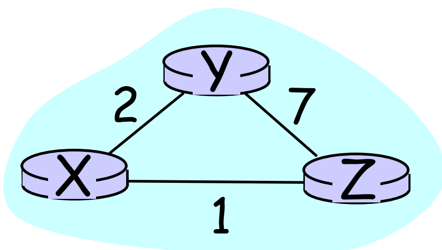
  else if (update received from V wrt destination Y)
    /* shortest path from V to some Y has changed */
    /* V has sent a new value for its min DV(Y,w) */
    /* call this received new value is "newval" */
    for the single destination y: $D_X(Y,V) = c(X,V) + newval$

  if we have a new $\min_w D_X(Y,w)$ for any destination Y
    send new value of $\min_w D_X(Y,w)$ to all neighbors

forever

Renato.LoCigno@dit.unitn.it

Distance Vector Algorithm: example



D^X (dest)	cost via Y	cost via Z
Y	2	∞
Z	∞	7

D^Y (dest)	cost via X	cost via Z
X	2	∞
Z	∞	1

D^Z (dest)	cost via X	cost via Y
X	7	∞
Y	∞	1

D^X (dest)	cost via Y	cost via Z
Y	2	8
Z	3	7

D^Y (dest)	cost via X	cost via Z
X	2	8
Z	9	1

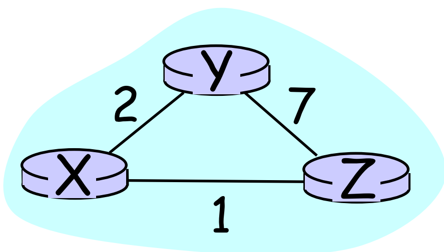
D^Z (dest)	cost via X	cost via Y
X	7	3
Y	9	1

D^X (dest)	cost via Y	cost via Z
Y		
Z		

D^Y (dest)	cost via X	cost via Z
X		
Z		

D^Z (dest)	cost via X	cost via Y
X		
Y		

Distance Vector Algorithm: example



D^X (dest)	cost via Y	cost via Z
Y	2	∞
Z	∞	7

D^Y (dest)	cost via X	cost via Z
X	2	∞
Z	∞	1

D^Z (dest)	cost via X	cost via Y
X	7	∞
Y	∞	1

D^X (dest)	cost via Y	cost via Z
Y	2	8
Z	3	7

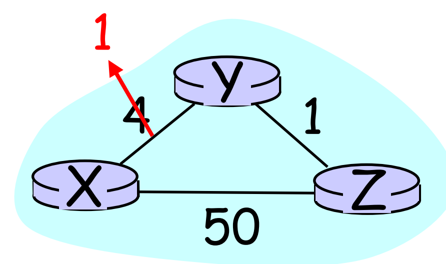
$$D^X(Y,Z) = c(X,Z) + \min_w\{D^Z(Y,w)\}$$
$$= 7+1 = 8$$

$$D^X(Z,Y) = c(X,Y) + \min_w\{D^Y(Z,w)\}$$
$$= 2+1 = 3$$

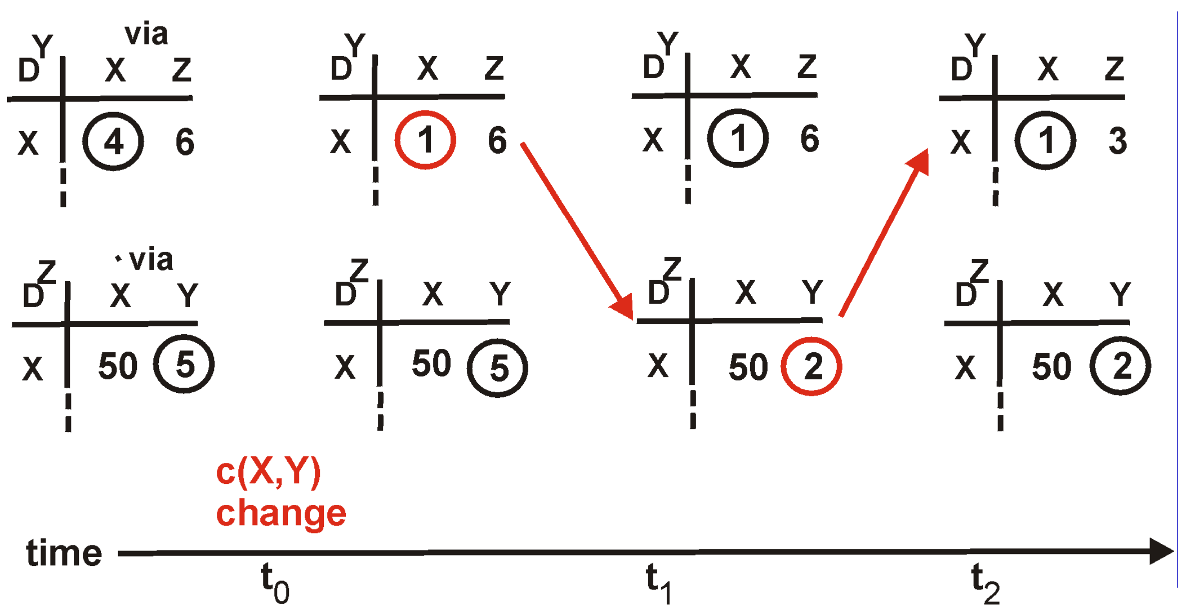
Distance Vector: link cost changes

Link cost changes:

- node detects local link cost change
- updates distance table (line 15)
- if cost change in least cost path, notify neighbors (lines 23,24)



"good news travels fast"

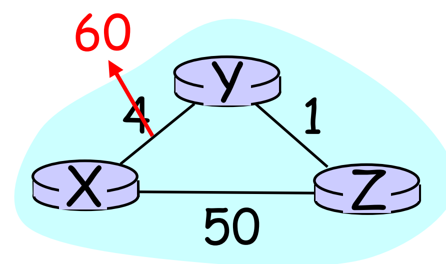


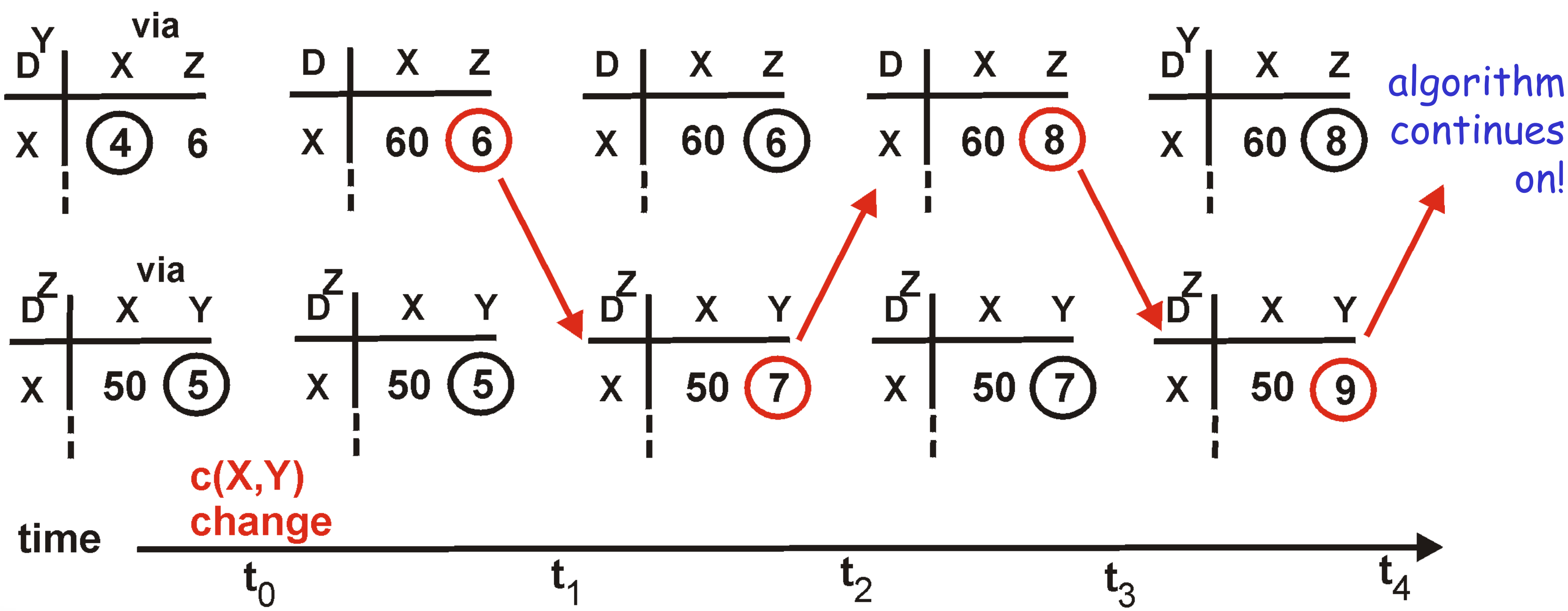
algorithm terminates

Distance Vector: link cost changes

Link cost changes:

- good news travels fast
- bad news travels slow - "count to infinity" problem!

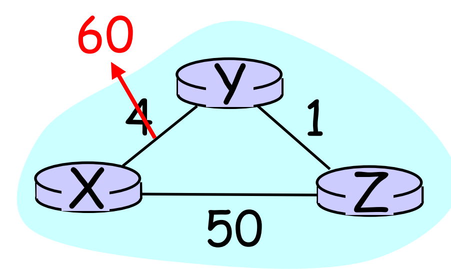


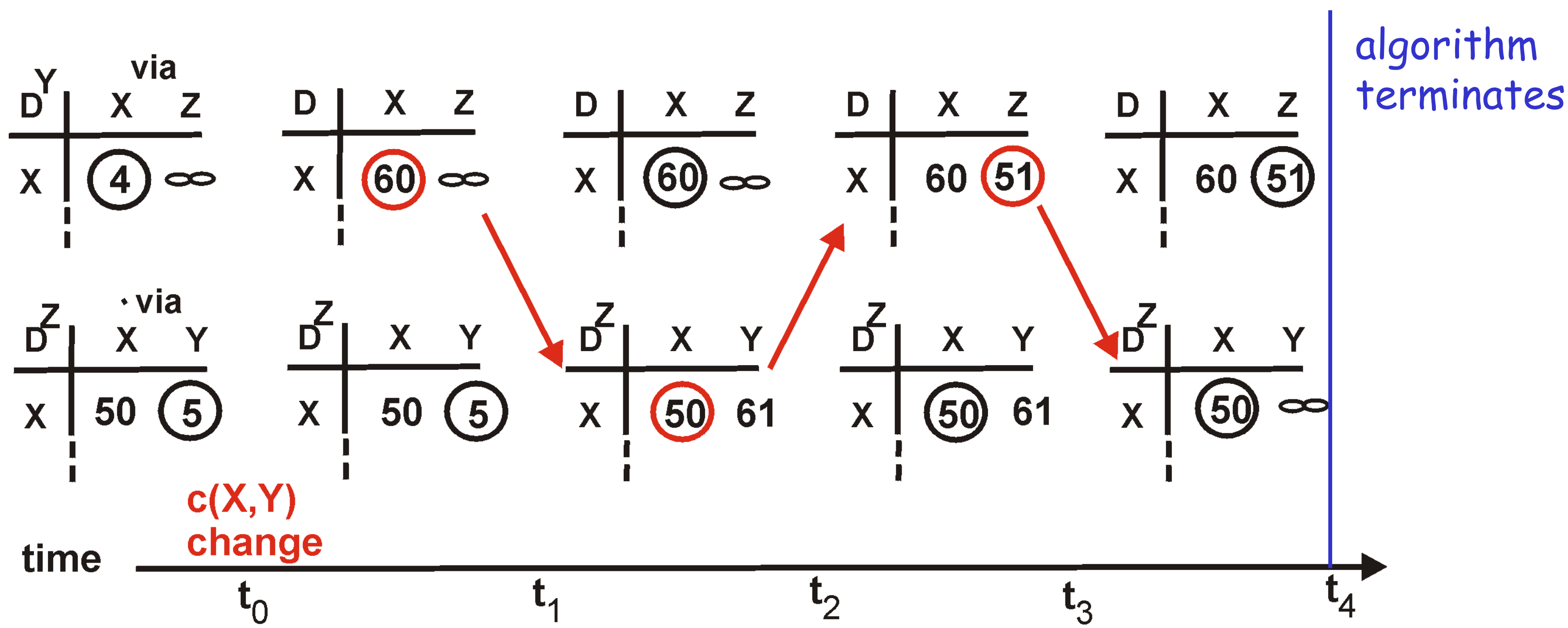


Distance Vector: poisoned reverse

If Z routes through Y to get to X :

- Z tells Y its (Z's) distance to X is infinite (so Y won't route to X via Z)
- will this completely solve count to infinity problem?





Comparison of LS and DV algorithms

Message complexity

- LS: with n nodes, E links, O(nE) msgs sent each
- DV: exchange between neighbors only
  - convergence time varies

Speed of Convergence

- LS: O(n**2) algorithm requires O(nE) msgs
  - may have oscillations
- DV: convergence time varies
  - may be routing loops
  - count-to-infinity problem

Robustness: what happens if router malfunctions?

LS:

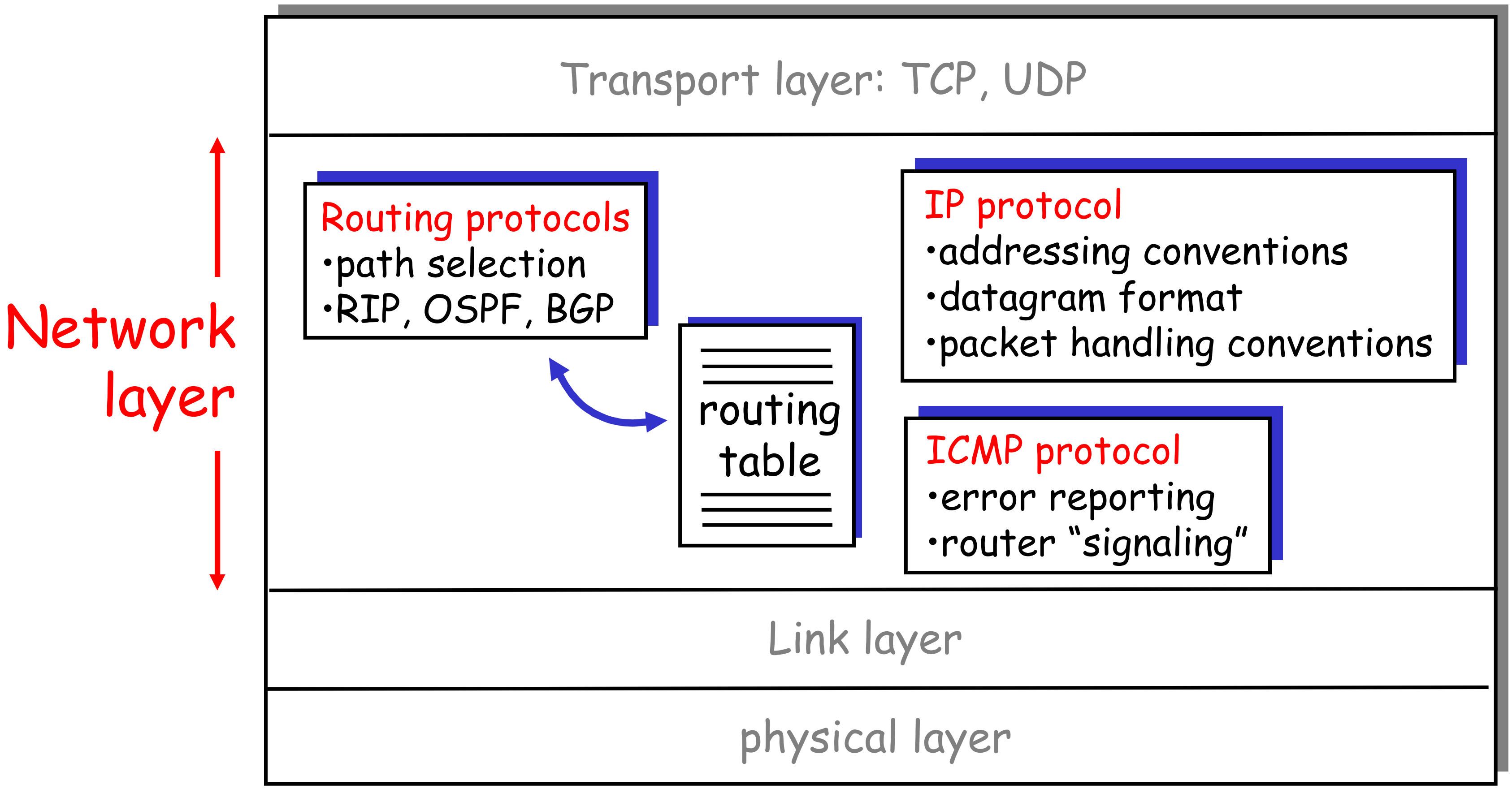
- node can advertise incorrect *link* cost
- each node computes only its *own* table

DV:

- DV node can advertise incorrect *path* cost
- each node's table used by others
  - error propagate thru network

The Internet Network layer

Host, router network layer functions:



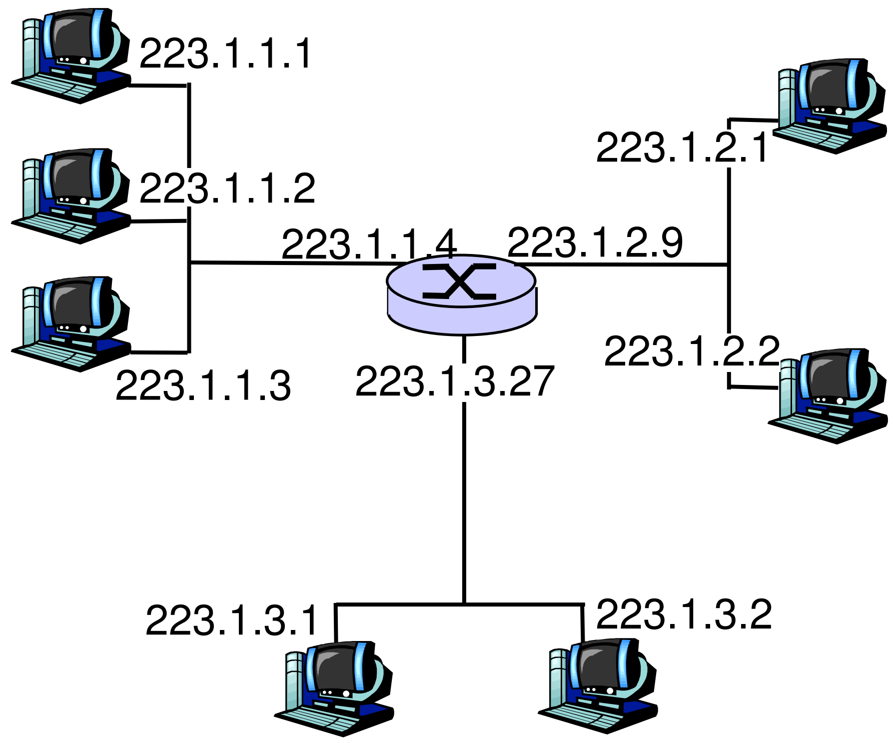
Why different Intra- and Inter-AS routing ?

- **Policy**: Inter is concerned with policies (which provider we must select/avoid, etc). Intra is contained in a single organization, so, no policy decisions necessary
- **Scale**: Inter provides an extra level of routing table size and routing update traffic reduction above the Intra layer
- **Performance**: Intra is focused on performance metrics; needs to keep costs low. In Inter it is difficult to propagate performance metrics efficiently (latency, privacy etc). Besides, policy related information is more meaningful.

We need **BOTH**!

IP Addressing

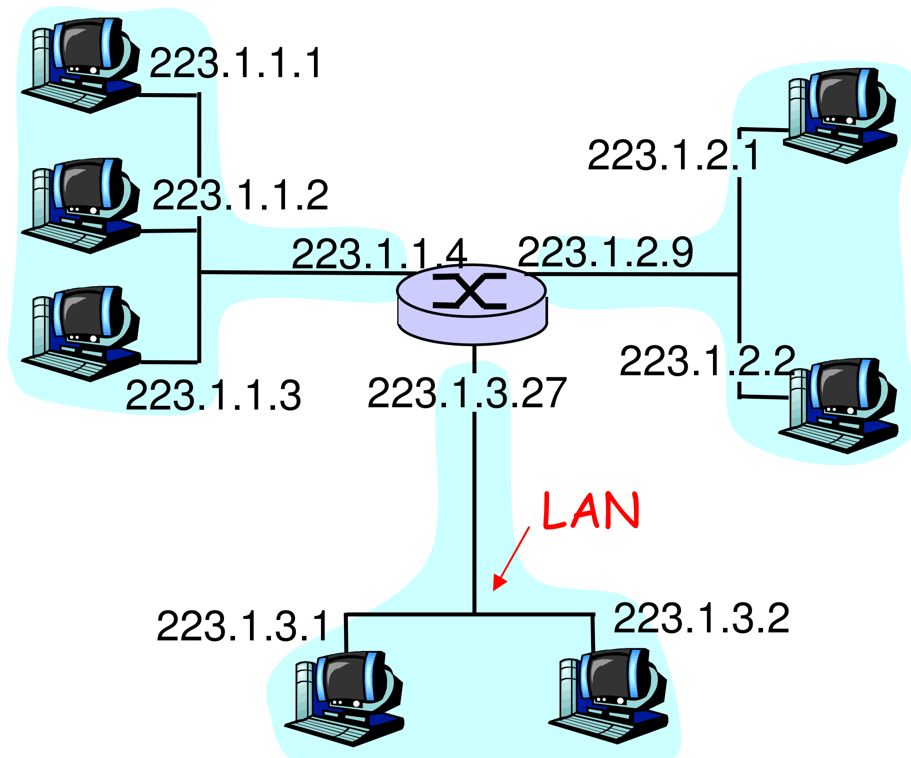
- IP address: 32-bit identifier for host, router *interface*
- *interface:* connection between host, router and physical link
  - router's typically have multiple interfaces
  - host may have multiple interfaces
  - IP addresses associated with interface, not host, router



$$223.1.1.1 = \underbrace{11011111}_{223}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}$$

IP Addressing

- *IP address:*
  - network part (high order bits)
  - host part (low order bits)
- *What's a network ?*
  (from IP address perspective)
  - device interfaces with same network part of IP address
  - can physically reach each other without intervening router



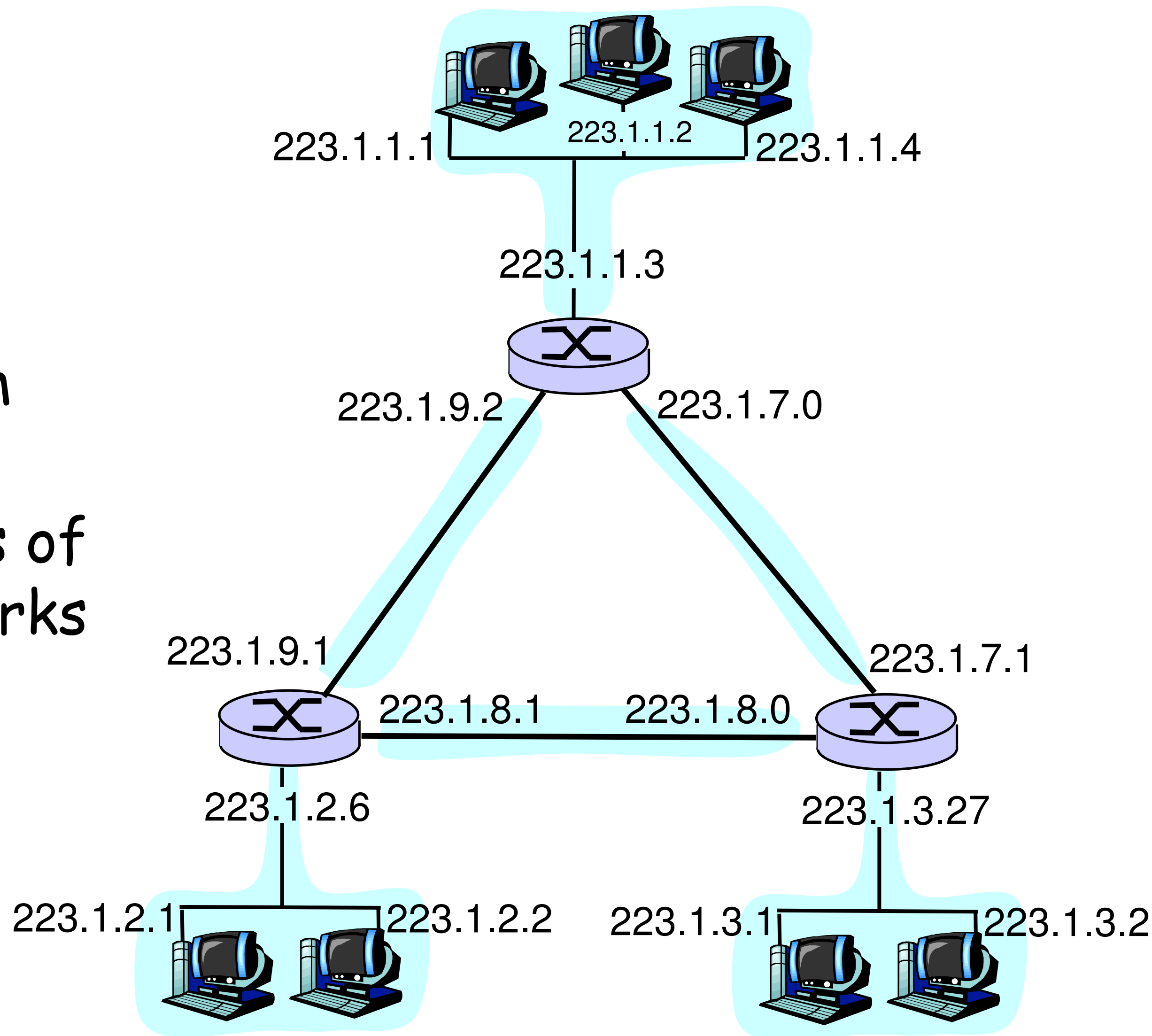
network consisting of 3 IP networks (for IP addresses starting with 223, first 24 bits are network address)

Renato.LoCigno@dit.unitn.it

IP Addressing

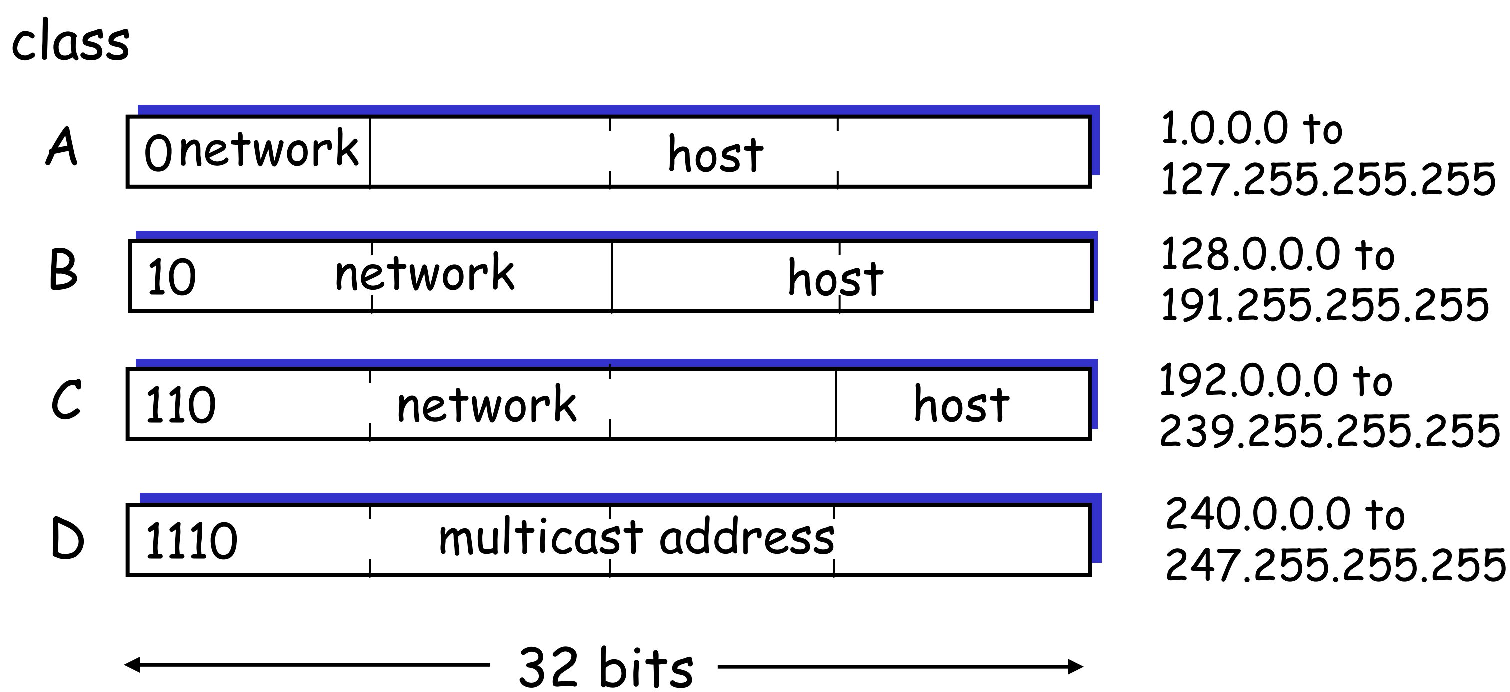
How to find the networks?

- Detach each interface from router, host
- create "islands of isolated networks



Interconnected system consisting of six networks

IP Addresses

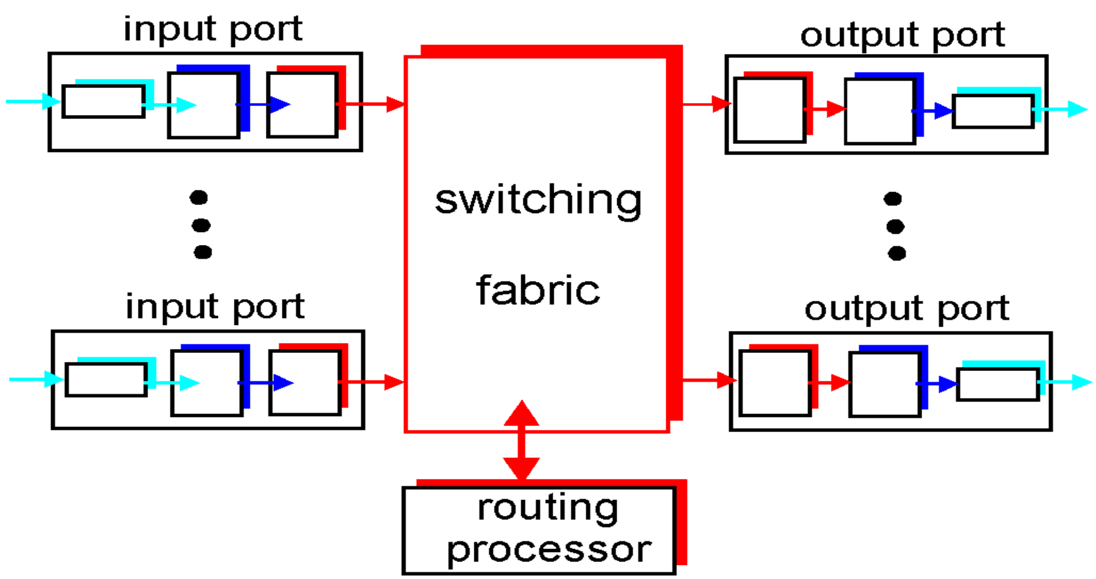


Address Management

- As Internet grows, we run out of addresses
- Solution (a): **subnetting**. Eg, Class B Host field (16bits) is subdivided into <subnet;host> fields
- Solution (b): **CIDR** (Classless Inter Domain Routing): assign block of contiguous Class C addresses to the same organization; these addresses all share a common prefix
- repeated "aggregation" within same provider leads to shorter and shorter prefixes
- CIDR helps also routing table size and processing: Border Gwys keep only prefixes and find "longest prefix" match

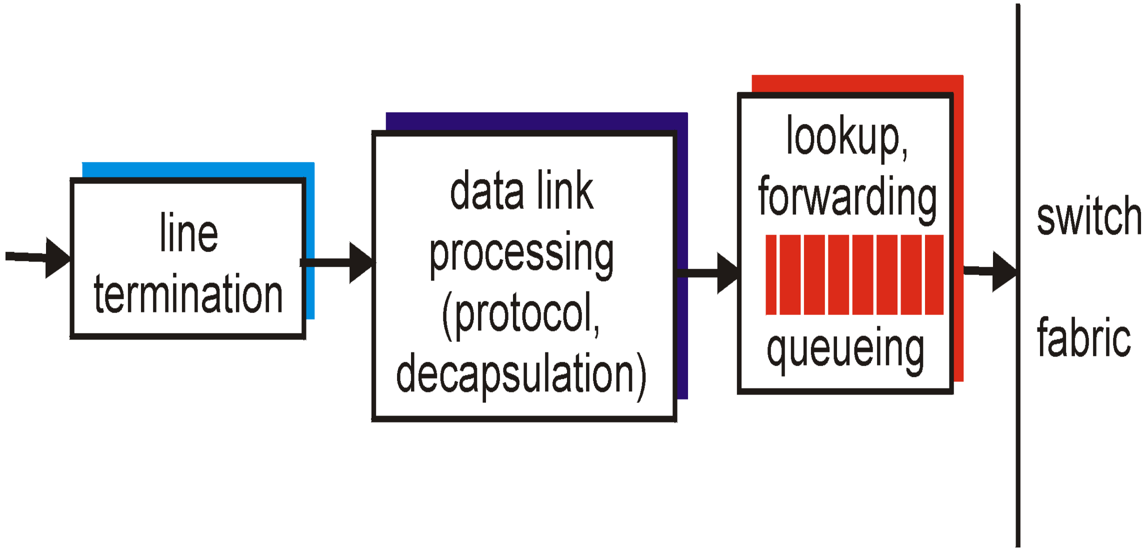
Router Architecture Overview

- Router main functions: *routing* algorithms and protocols processing, *switching* datagrams from an incoming link to an outgoing link



Router Components

Input Ports

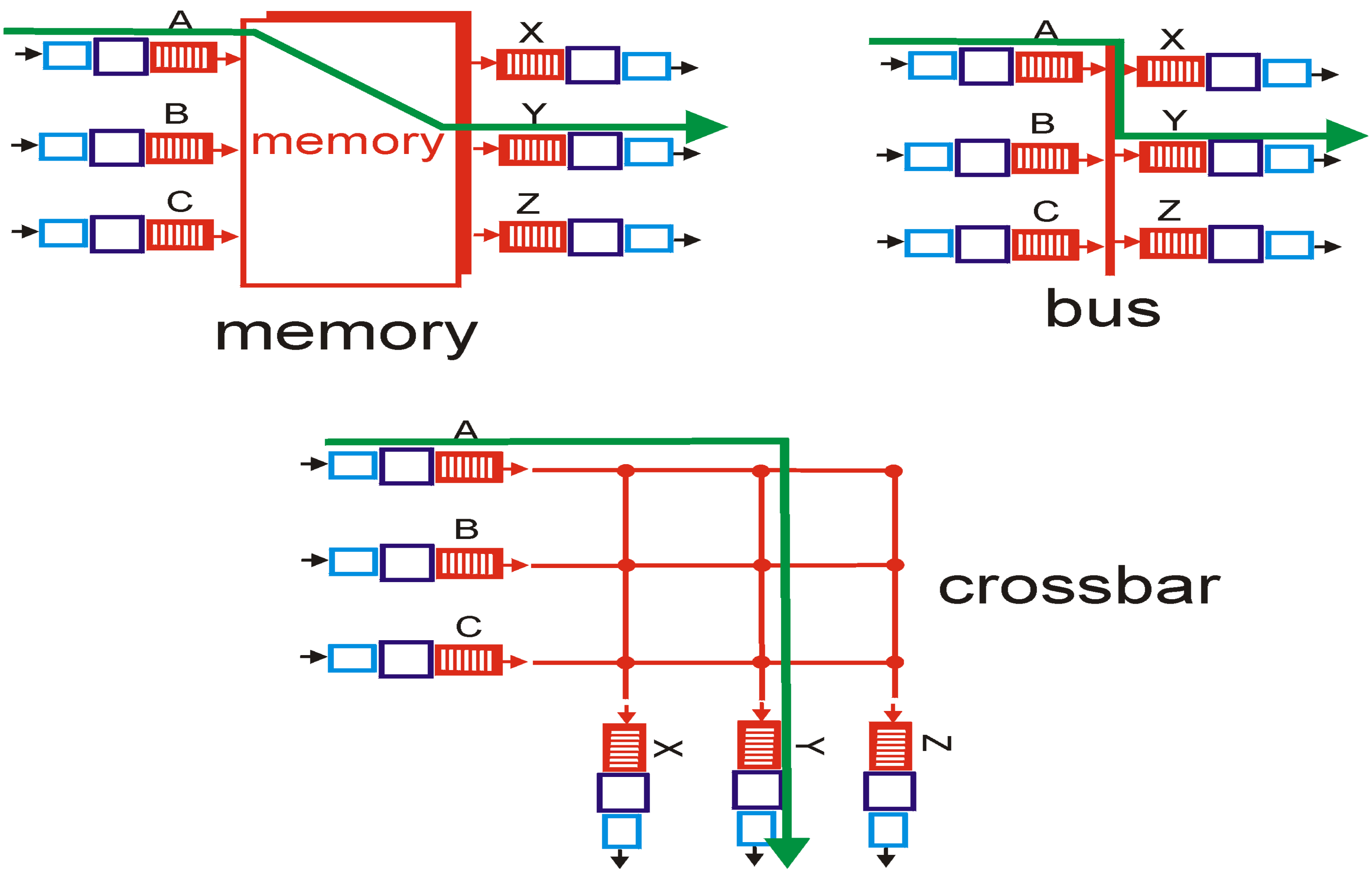


- **Decentralized switching**: perform routing table lookup using a copy of the node routing table stored in the port memory
- Goal is to complete input port processing at 'line speed', ie processing time =< frame reception time (eg, with 2.5 Gbps line, 256 bytes long frame, router must perform about 1 million routing table lookups in a second)
- Queuing occurs if datagrams arrive at rate higher than can be forwarded on switching fabric

Speeding Up Routing Table Lookup

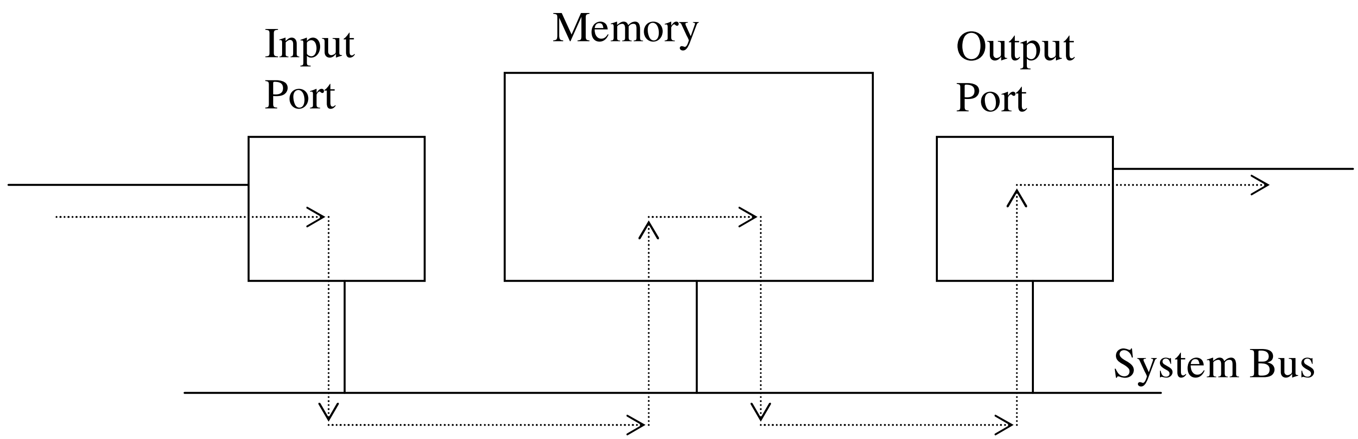
- Table is stored in a tree structure to facilitate binary search
- Content Addressable Memory (associative memory), eg Cisco 8500 series routers
- Caching of recently looked-up addresses
- Compression of routing tables

Switching Fabric



Switching Via Memory

- *First generation routers*: packet is copied under system's (single) CPU control; speed limited by Memory bandwidth. For Memory speed of B packet/sec or pps, throughput is B/2 pps



- *Modern routers*: input ports with CPUs that implement output port lookup, and store packets in appropriate locations (= switch) in a shared Memory; eg Cisco Catalyst 8500 switches

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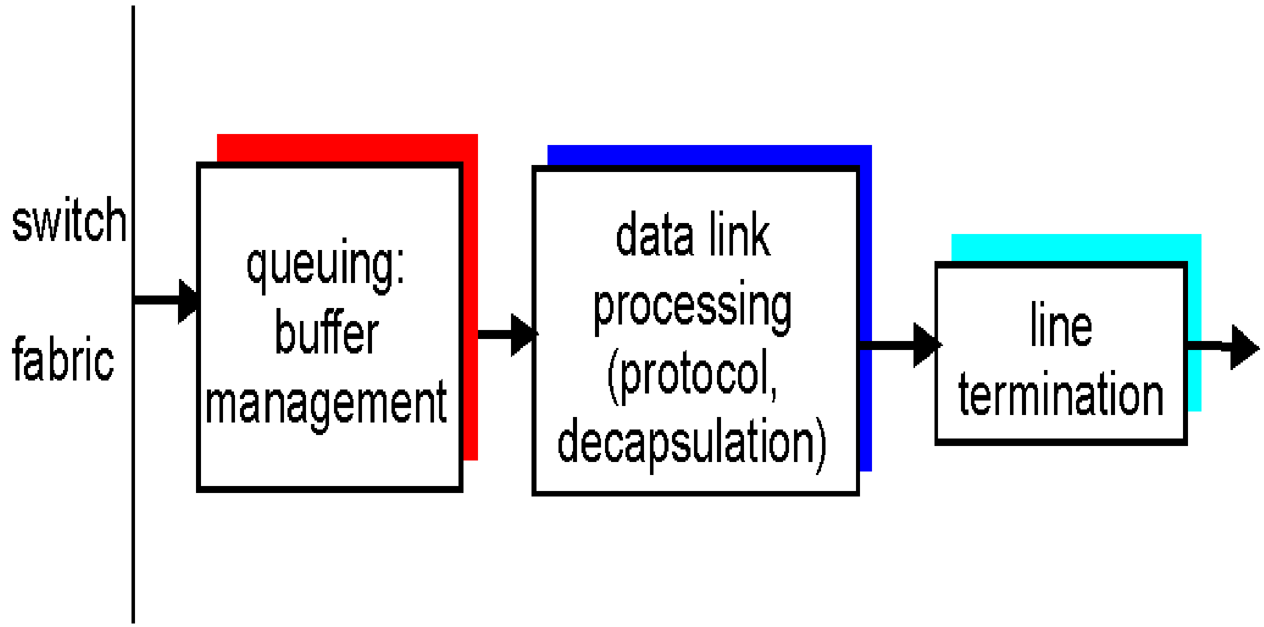
Switching Via Bus

- Input port processors transfer a datagram from input port memory to output port memory via a shared bus
- Main resource contention is over the bus; switching is limited by bus speed
- Sufficient speed for access and enterprise routers (not regional or backbone routers) is provided by a Gbps bus; eg Cisco 1900 which has a 1 Gbps bus

Switching Via An Interconnection Network

- Used to overcome bus bandwidth limitations
- Banyan networks and other interconnection networks were initially developed to connect processors in a multiprocessor computer system; used in Cisco 12000 switches provide up to 60 Gbps through the interconnection network
- Advanced design incorporates fragmenting a datagram into fixed length cells and switch the cells through the fabric; + better sharing of the switching fabric resulting in higher switching speed

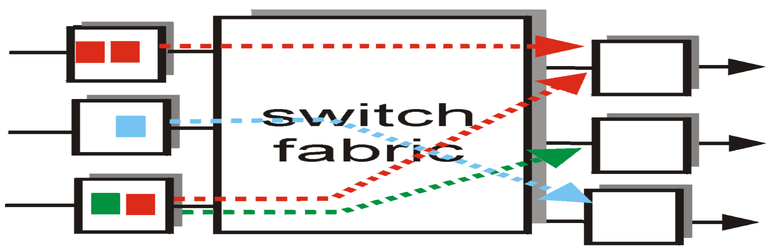
Output Ports



Buffering is required to hold datagrams whenever they arrive from the switching fabric at a rate faster than the transmission rate

Queuing At Input and Output Ports

- Queues build up whenever there is a rate mismatch or blocking. Consider the following scenarios:
  - Fabric speed is faster than all input ports combined; more datagrams are destined to an output port than other output ports; queuing occurs at output port
  - Fabric bandwidth is not as fast as all input ports combined; queuing may occur at input queues;
  - HOL blocking: fabric can deliver datagrams from input ports in parallel, except if datagrams are destined to same output port; in this case datagrams are queued at input queues; there may be queued datagrams that are held behind HOL conflict, even when their output port is available



output port contention at time t - only one red packet can be transferred

green packet experiences HOL blocking

Transport Layer: UDP & TCP

Goals:

- Recall principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet

Overview:

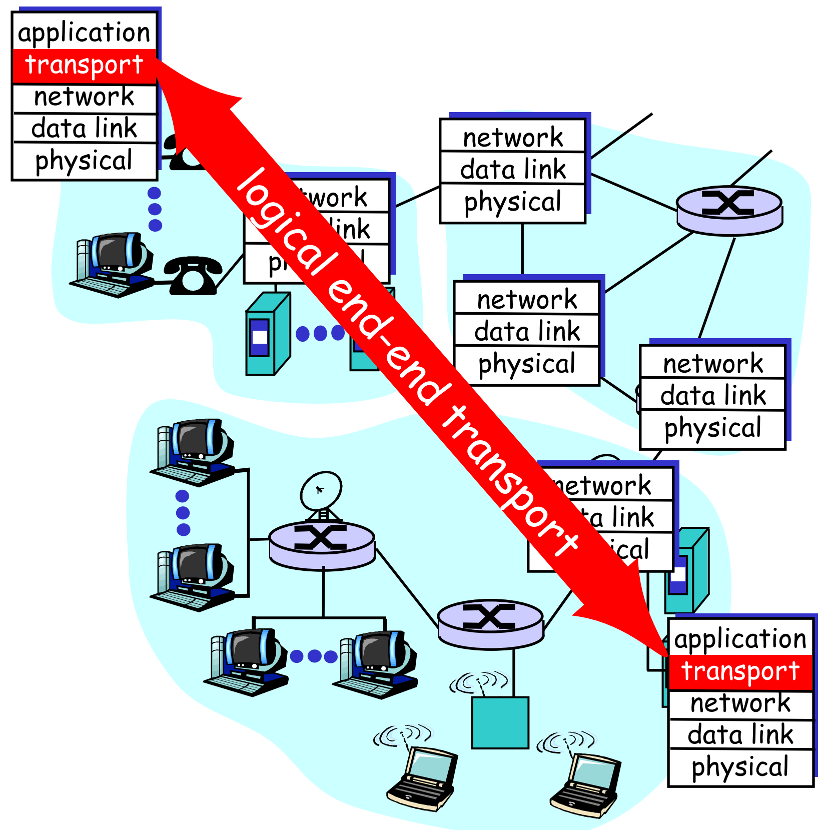
- transport layer services
- multiplexing/demultiplexing
- connectionless transport: UDP
- principles of reliable data transfer
- connection-oriented transport: TCP
  - reliable transfer
  - flow control
  - connection management

Transport services and protocols

- provide *logical communication* between app' processes running on different hosts
- transport protocols run in end systems (primarily)

transport vs network layer services:

- *network layer:* data transfer between end systems
- *transport layer:* data transfer between processes
  - relies on, enhances, network layer services

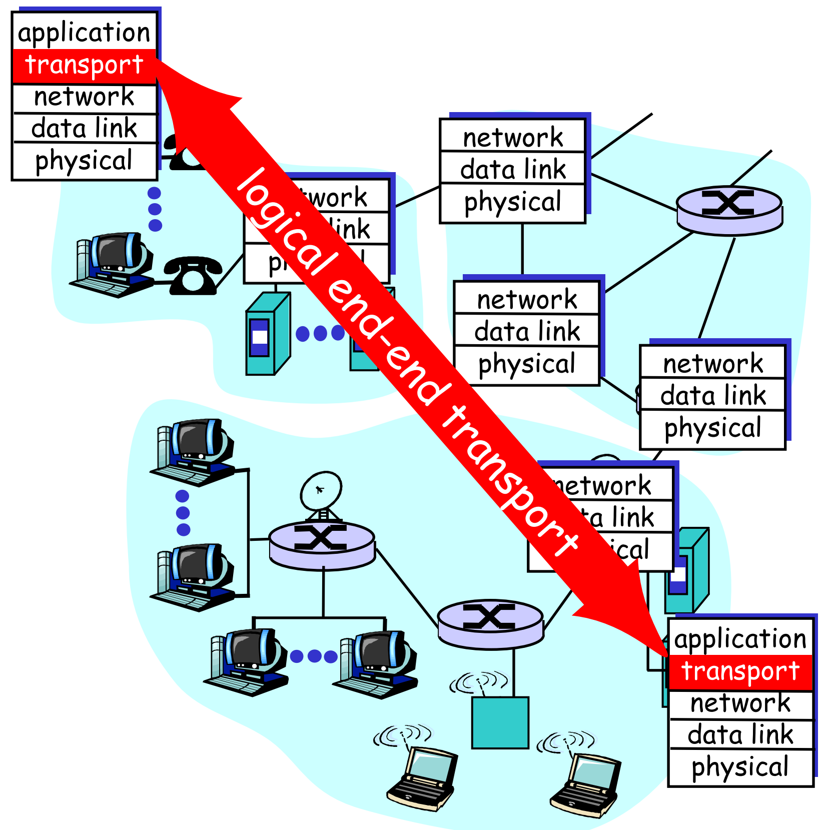


Renato.LoCigno@dit.unitn.it

Transport-layer protocols

Internet transport services:

- reliable, in-order unicast delivery (TCP)
  - congestion
  - flow control
  - connection setup
- unreliable (“best-effort”), unordered unicast or multicast delivery: UDP
- services not available:
  - real-time
  - bandwidth guarantees
  - reliable multicast

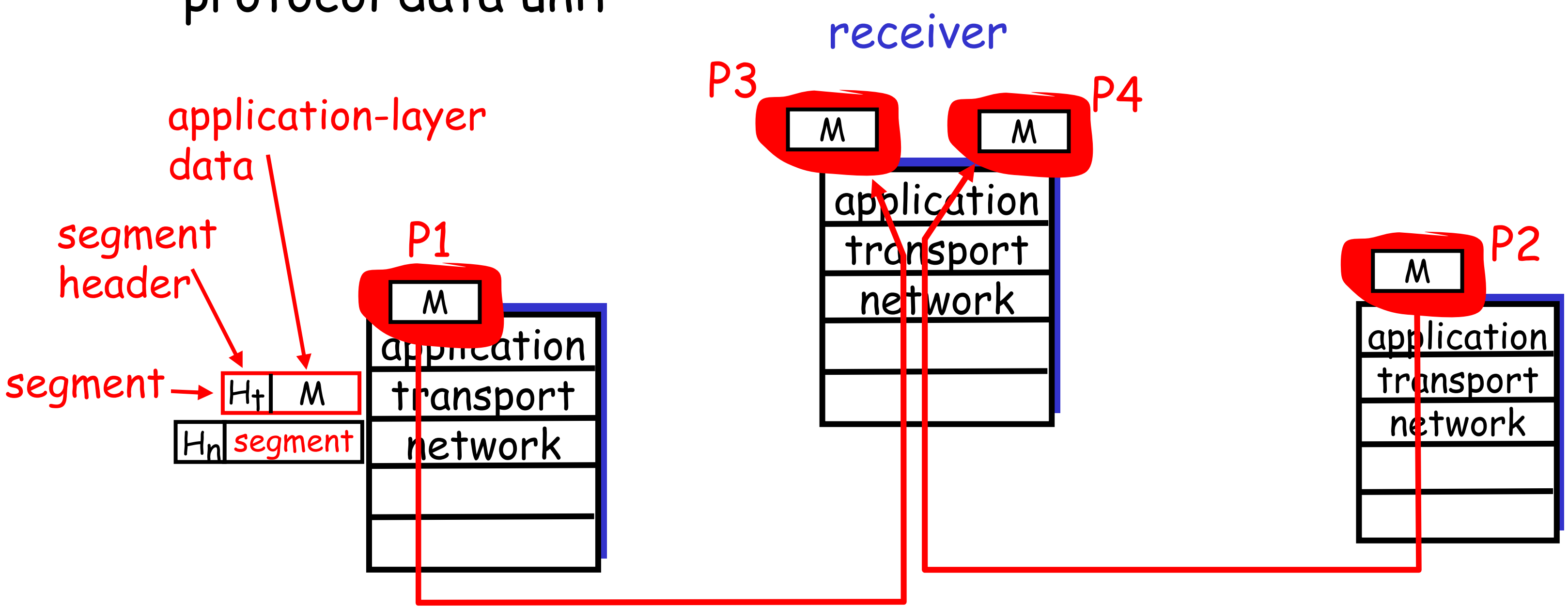


Renato.LoCigno@dit.unitn.it

Multiplexing/demultiplexing

Recall: *segment* - unit of data exchanged between transport layer entities
- aka TPDU: transport protocol data unit

Demultiplexing: delivering received segments (TPDUs)to correct app layer processes

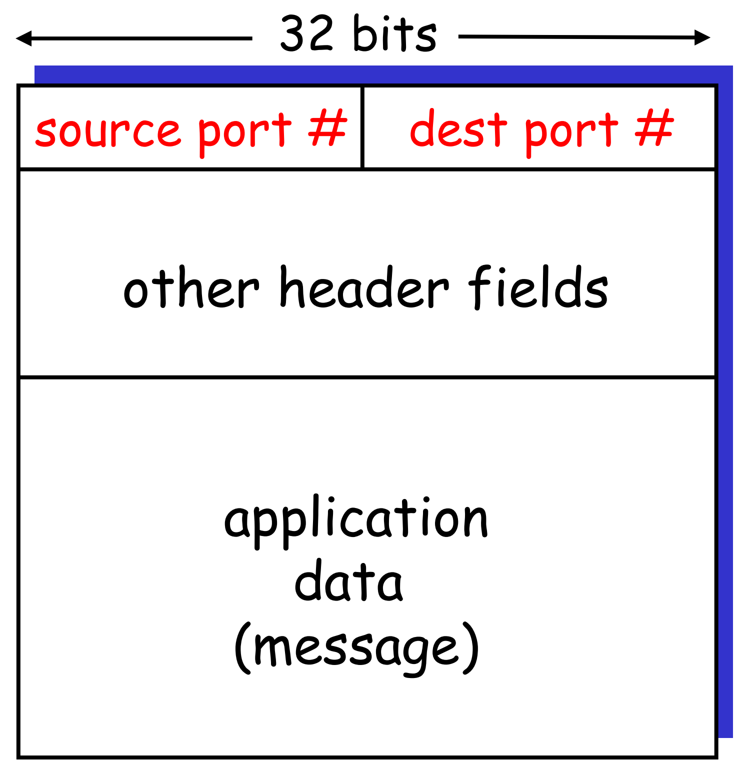


Multiplexing/demultiplexing

Multiplexing:
gathering data from multiple app processes, enveloping data with header (later used for demultiplexing)

multiplexing/demultiplexing:

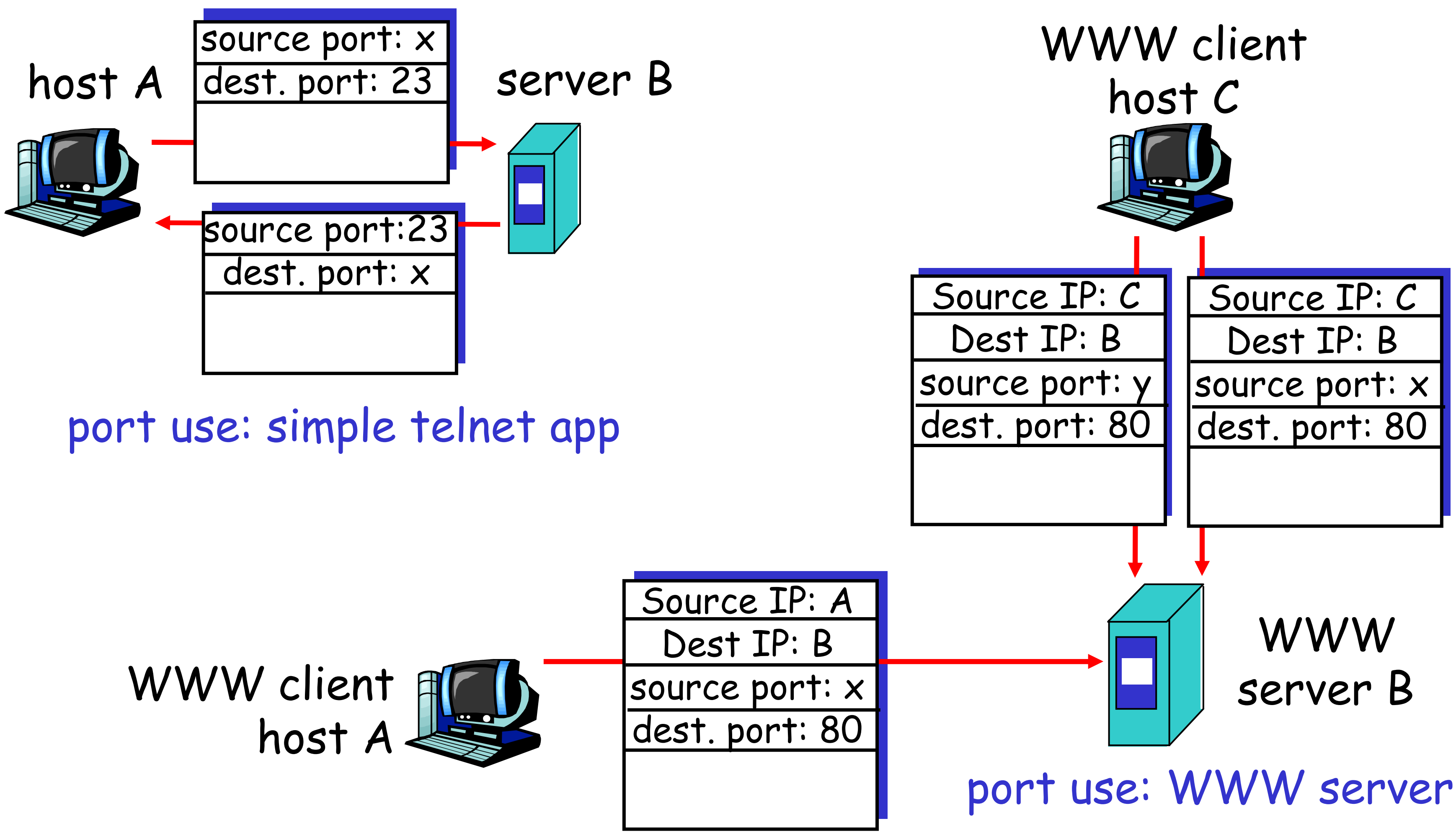
- based on sender, receiver port numbers, IP addresses
  - source, dest port #s in each segment
  - recall: well-known port numbers for specific applications



TCP/UDP segment format

Renato.LoCigno@dit.unitn.it

Multiplexing/demultiplexing: examples



UDP: User Datagram Protocol [RFC 768]

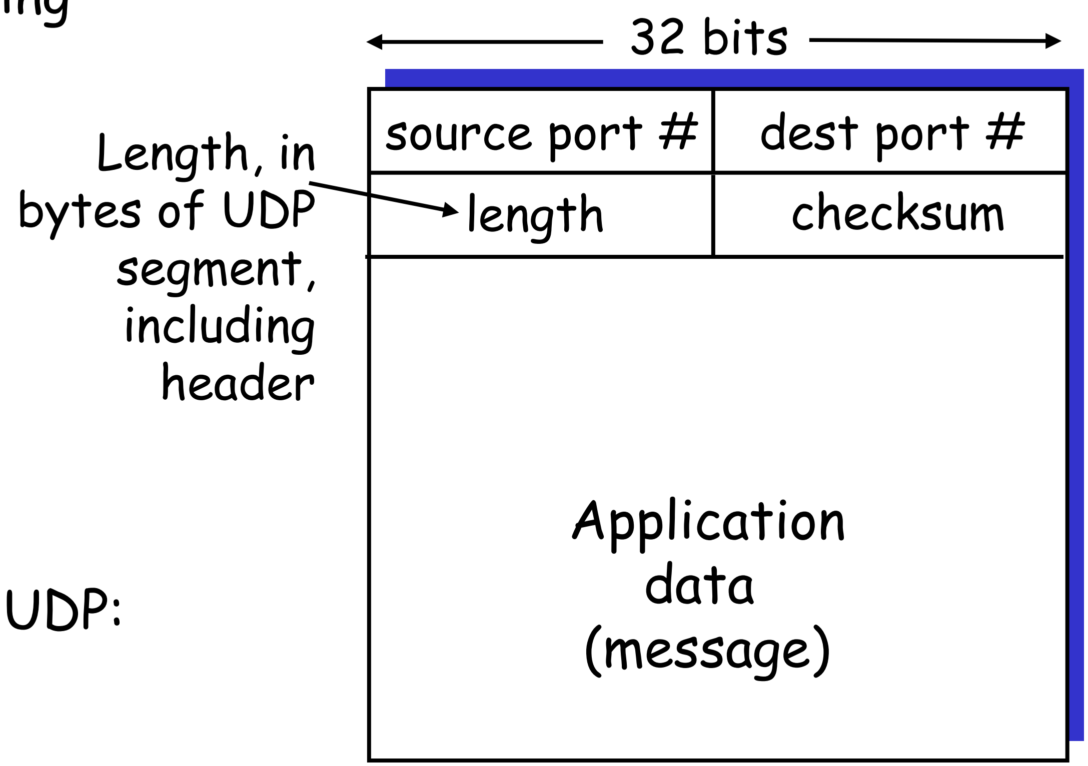
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

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

UDP: more

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



UDP segment format

UDP checksum

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

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

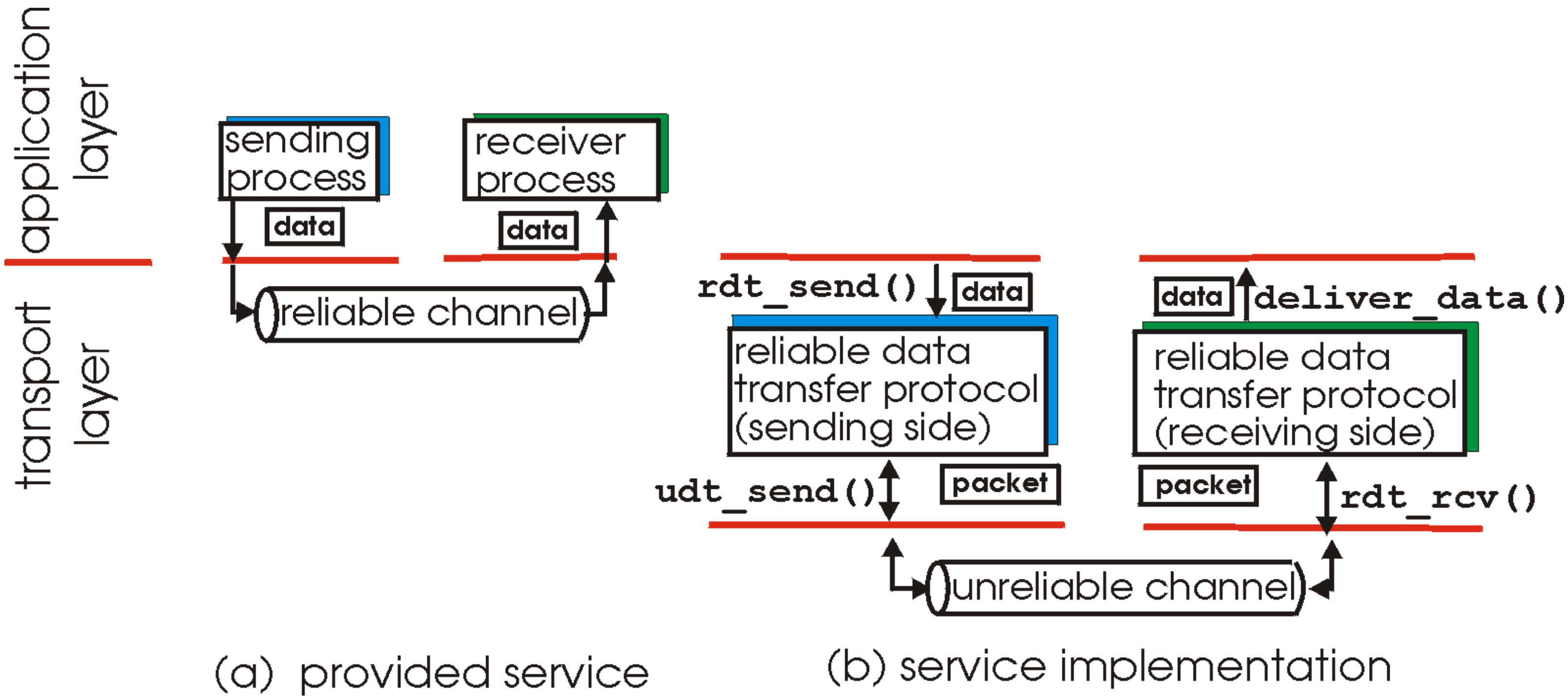
Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonethless?*

Renato.LoCigno@dit.unitn.it

Principles of Reliable data transfer

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



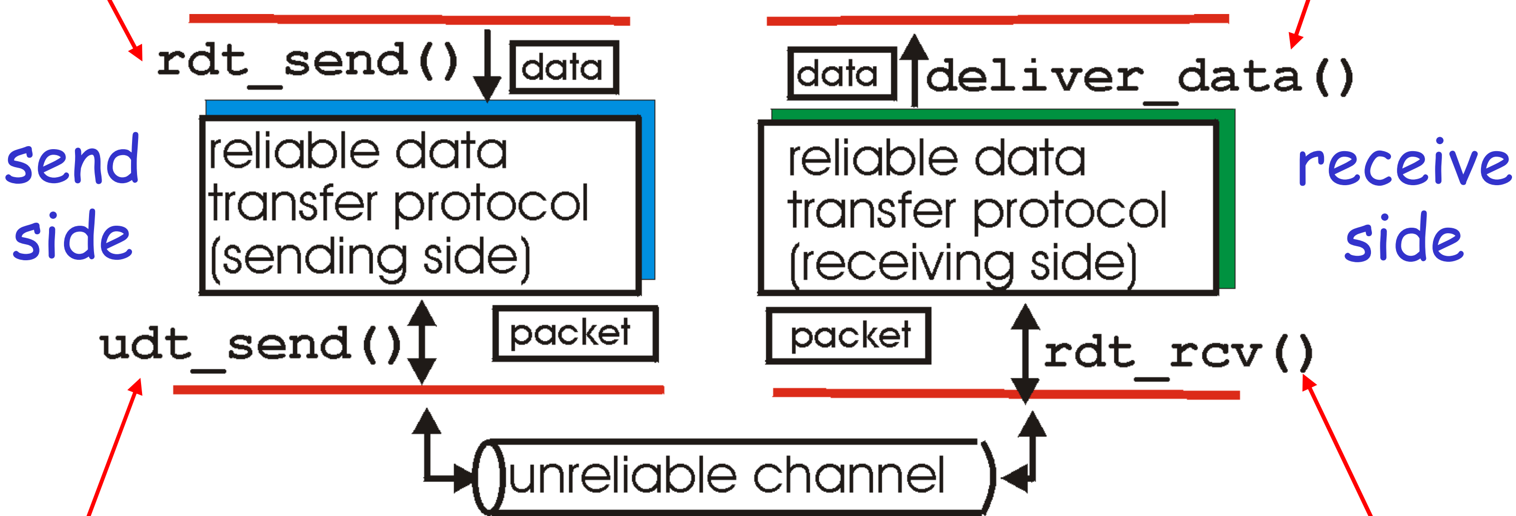
- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Renato.LoCigno@dit.unitn.it

Reliable data transfer: getting started

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by **rdt** to deliver data to upper



udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

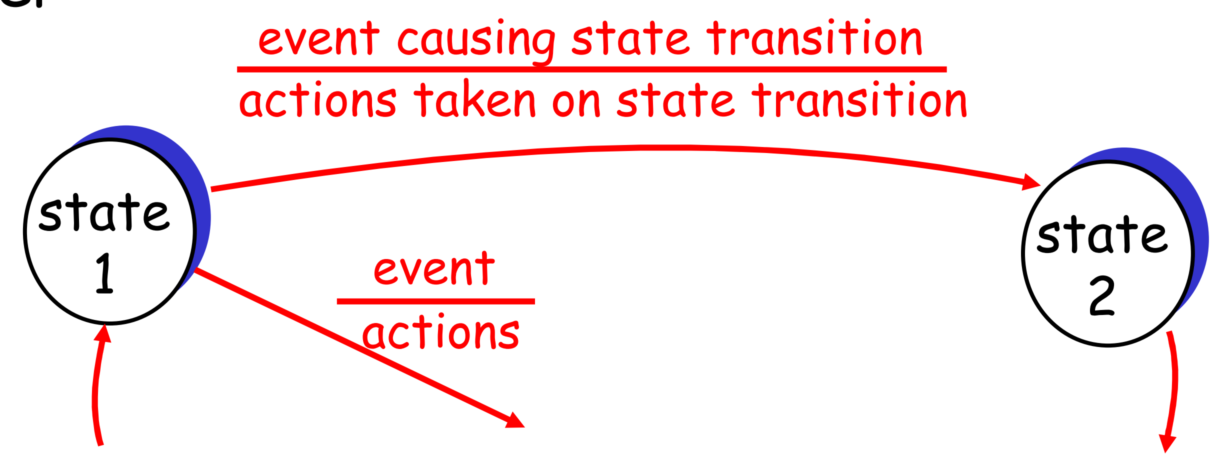
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

We'll:

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

state: when in this "state" next state uniquely determined by next event



rdt: channels with errors *and* loss

Sssumption: underlying channel can lose packets (data or ACKs)

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

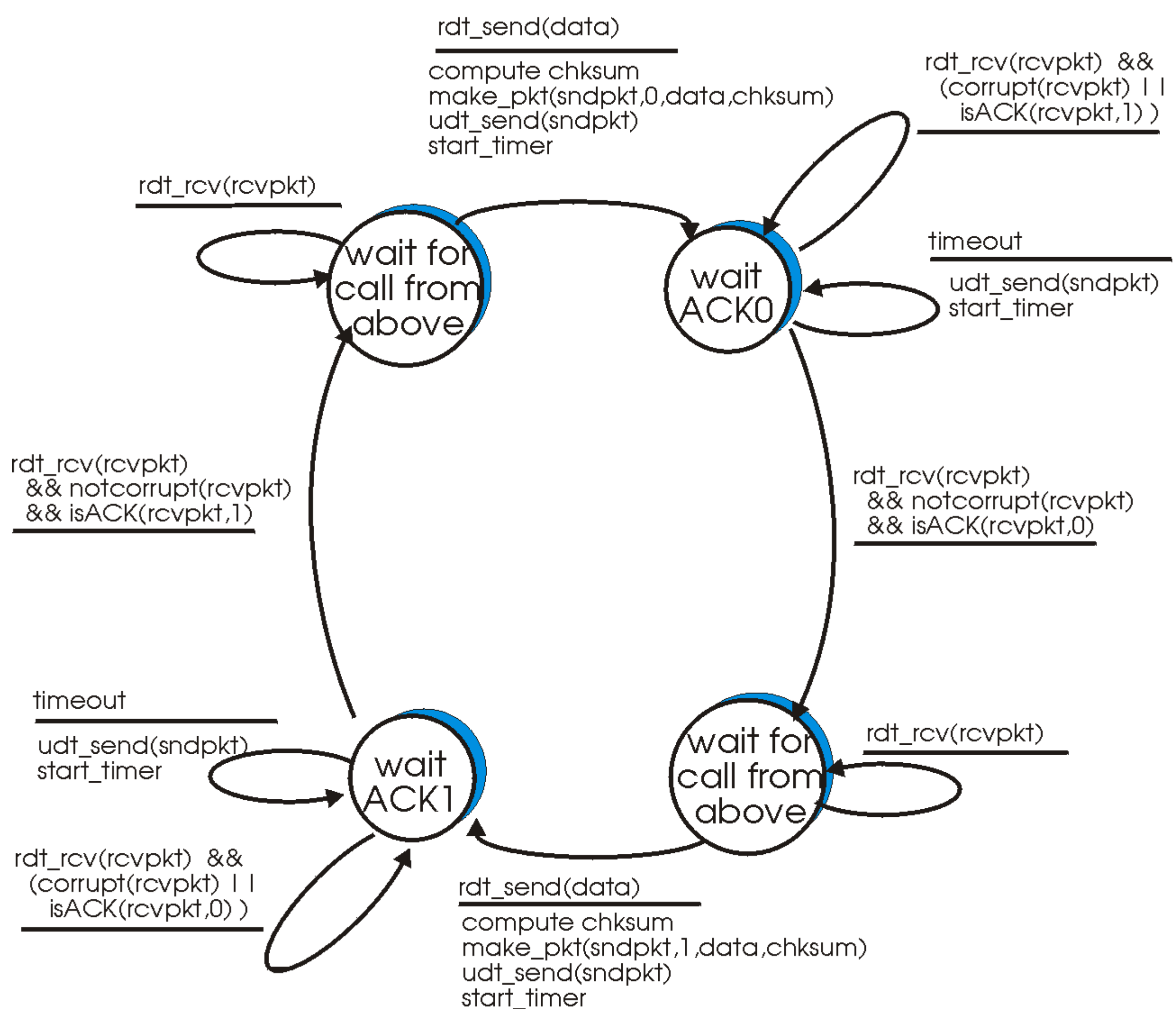
Q: how to deal with loss?

- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

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

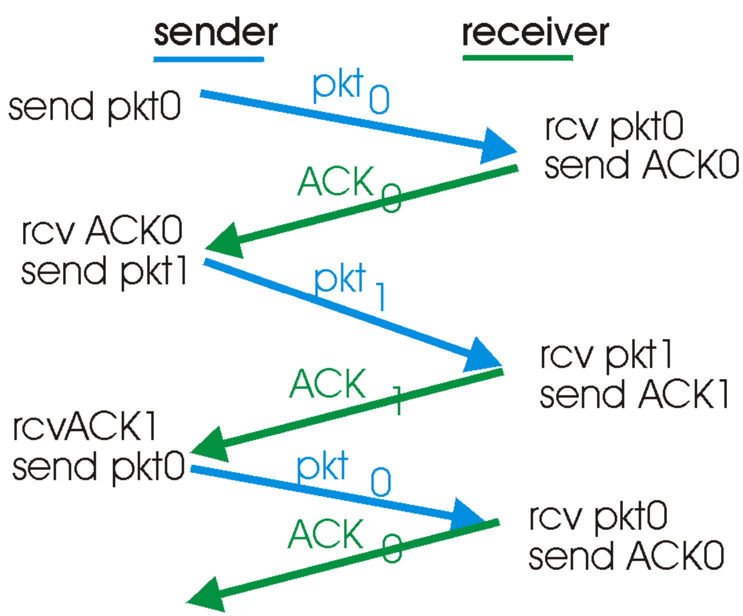
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

rdt: sender

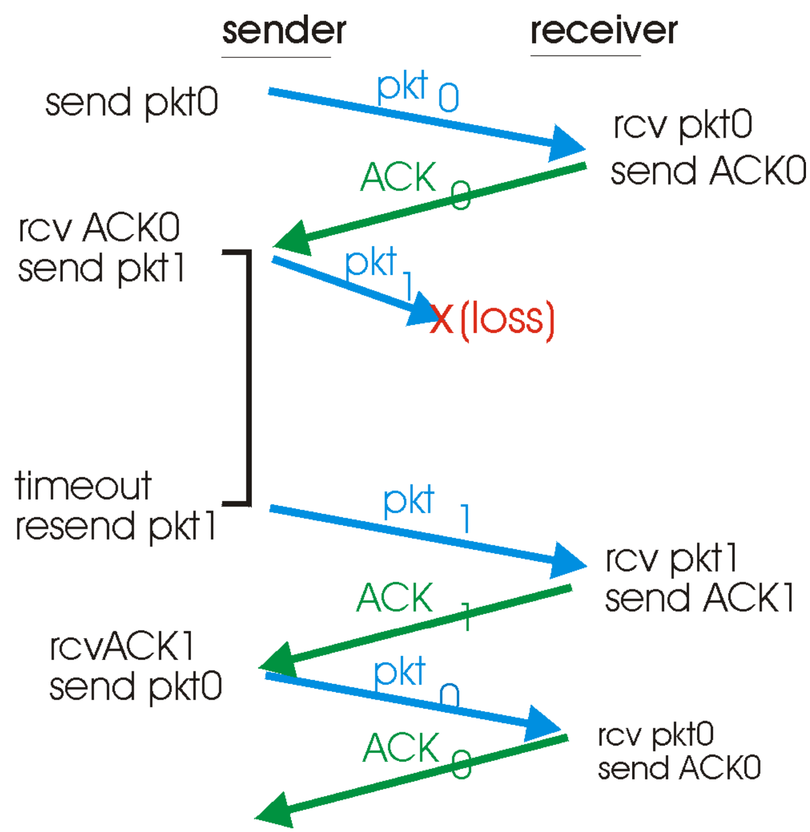


rdt_send(data)
compute chksum
make_pkt(sndpkt,0,data,chksum)
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) | | isACK(rcvpkt,1))
rdt_rcv(rcvpkt)
wait for call from above
wait ACK0
timeout
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,1)
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,0)
timeout
udt_send(sndpkt)
start_timer
wait ACK1
wait for call from above
rdt_rcv(rcvpkt)
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) | | isACK(rcvpkt,0))
rdt_send(data)
compute chksum
make_pkt(sndpkt,1,data,chksum)
udt_send(sndpkt)
start_timer

rdt in action

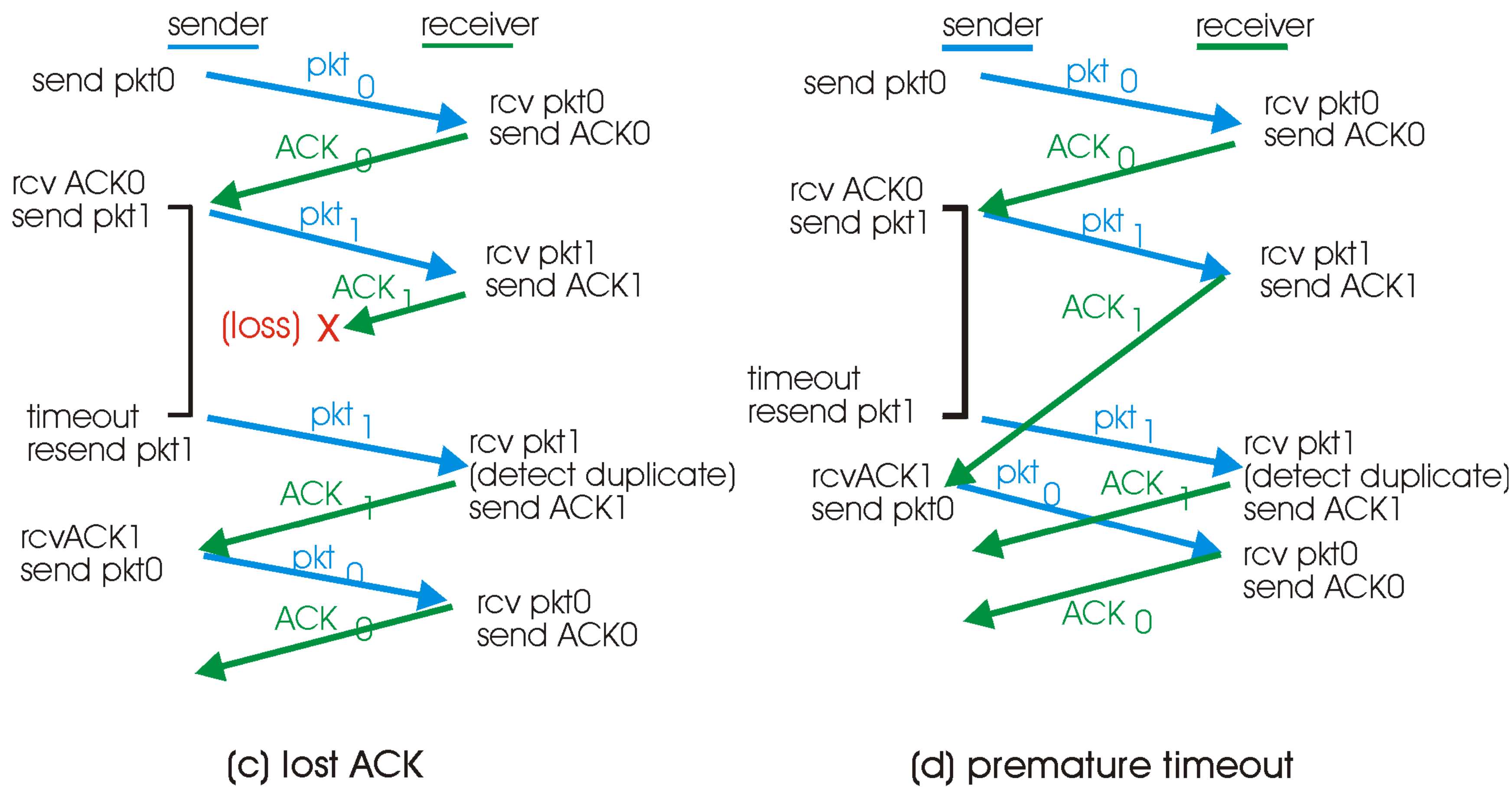


(a) operation with no loss



(b) lost packet

rdt in action



Performance of rdt

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

$$T_{transmit} = \frac{8\text{kb/pkt}}{10^{**}9\text{ b/sec}} = 8\text{ microsec}$$

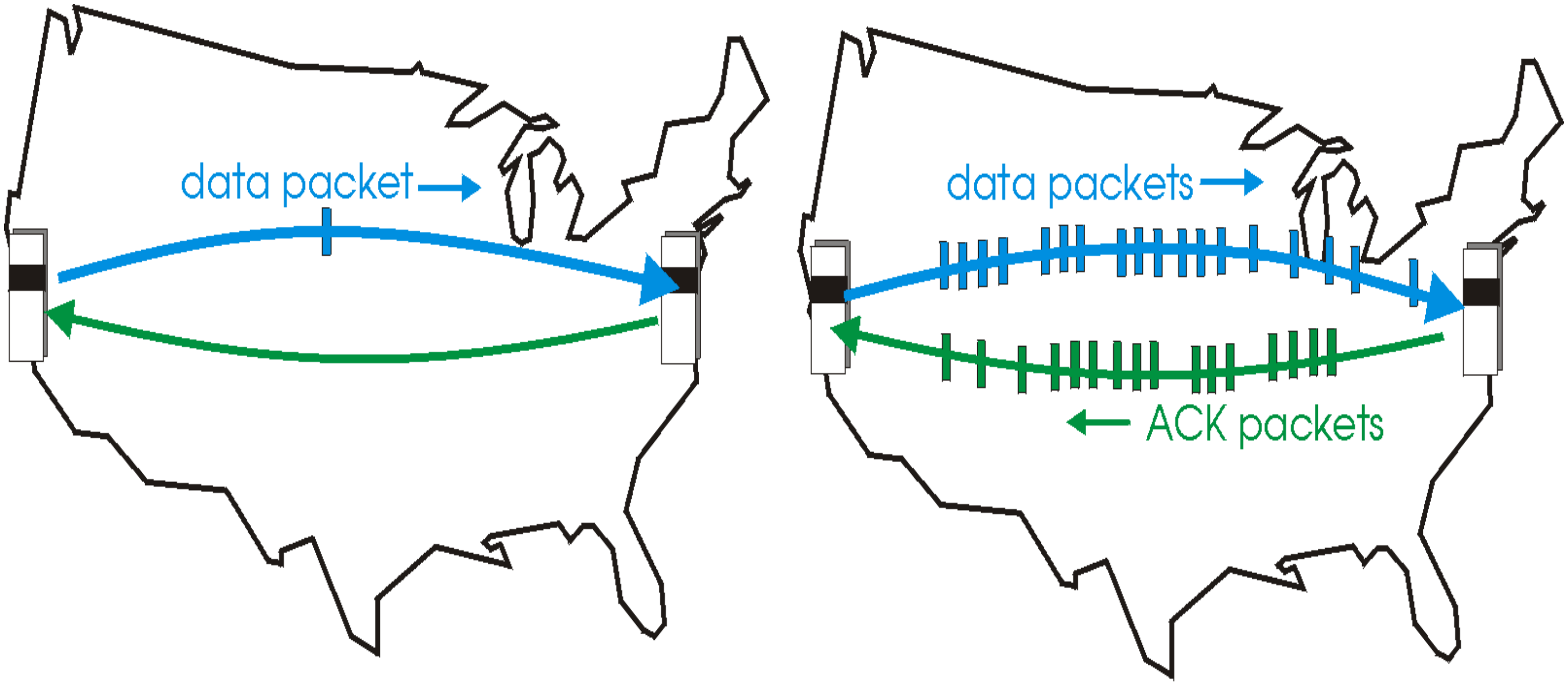
$$\text{Utilization} = U = \frac{\text{fraction of time}}{\text{sender busy sending}} = \frac{8\text{ microsec}}{30.016\text{ msec}} = 0.00015$$

- 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

Pipelined Protocols

- Channel utilization under a Stop&Wait protocol is not high when the propagation time is long relative to the transmission time
- Solution: pipelined protocols, where more than one packet can be sent without waiting for feedback, thus filling the 'pipeline'
- Two major versions (and lots of variations on the theme):
  - Go-Back-N
  - Selective Repeat
- New requirements:
  - Buffering more than one packet at sender, and possibly at receiver too
  - Larger sequence numbers for identifying packets in transit

Filling the Pipeline



(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

Stop&Wait Efficiency

$$U = \frac{T_{datatrans}}{T_{datatrans} + 2*T_{prop} + T_{proc} + T_{acktrans}}$$

For relatively small T_{proc} and $T_{acktrans}$

$$U \approx \frac{T_{datatrans}}{T_{datatrans} + 2*T_{prop}}, \text{ or}$$

$$U \approx \frac{1}{1 + 2*a}, \text{ where } a = \frac{T_{prop}}{T_{datatrans}}$$

$$T_{datatrans} = \frac{L}{C}, \text{ where L is the Packet}$$

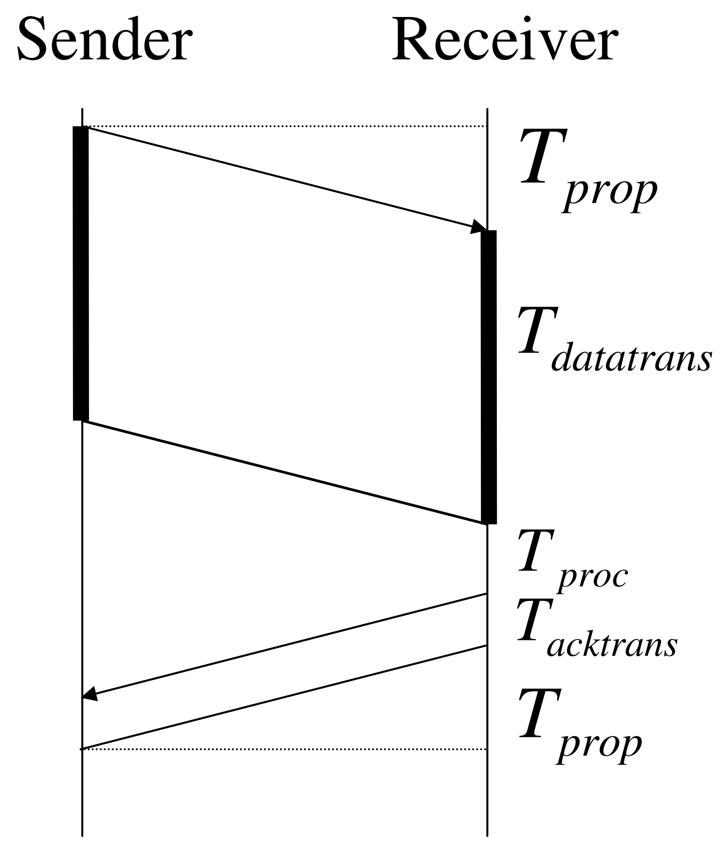
length and C is the Transmission Speed.

For one bit of data, $T_{datatrans} = 1/C$;

in this case $a = CT_{prop}$, which is

called the "Bandwidth - Delay"

product

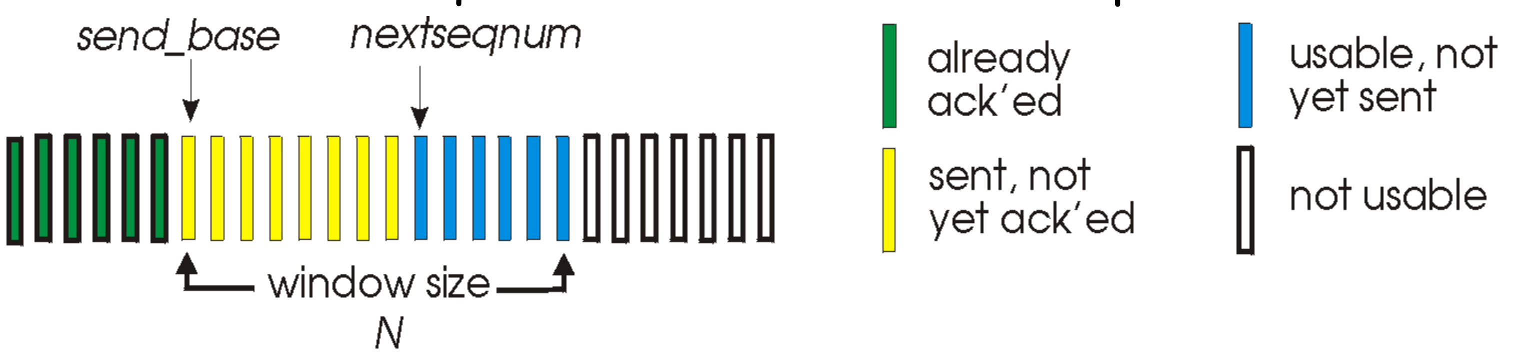


Go-Back-N

- Sender can go ahead and transmit packets without waiting for feedback up to some number of packets (for flow control reasons, details later)
- Definitions:

  N: maximum allowable number of transmission without feedback

  Base: lowest sequence number of unacked packets



Renato.LoCigno@dit.unitn.it

Go-Back-N Window

- From definitions and figure above:

  [0, base-1] transmitted and acked

  [base, nextseqnum-1] transmitted and waiting for feedback, or 'outstanding'

  [nextseqnum, base+N-1] numbers that can be used when packets are provided by higher layer for transmission

  [base+N, maxseqnum] numbers that cannot be used until more packets are acked

Go-Back-N Window (Cont.)

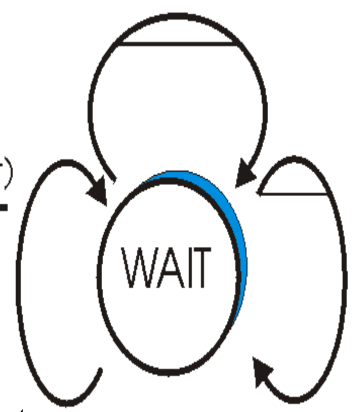
- Because of the window metaphor, these protocols are also referred to as *sliding window protocols*
- Stop&Wait can be viewed as a sliding window protocol, with window size N = 1, and sequence space = [0,1]
- Sequence number is carried in a fixed length field in the packet header; with k bits in the Sequence number field, the sequence space is
- Since sequence numbers must wrap around, all sequence number arithmetic is modulo

Go-Back-N Sender

```
rdt_send(data)
______________
if (nextseqnum < base+N) {
  compute chksum
  make_pkt(sndpkt(nextseqnum)),nextseqnum,data,chksum)
  udt_send(sndpkt(nextseqnum))
  if (base == nextseqnum)
    start_timer
  nextseqnum = nextseqnum + 1
  }
else
  refuse_data(data)
```

Window NOT full

No other packets outstanding



Acks are cumulative

```
rdt_rcv(rcv_pkt) && notcorrupt(rcvpkt)
______________________________________
base = getacknum(rvcpkt)+1
if (base == nextseqnum)
   stop_timer
 else
   start_timer
```

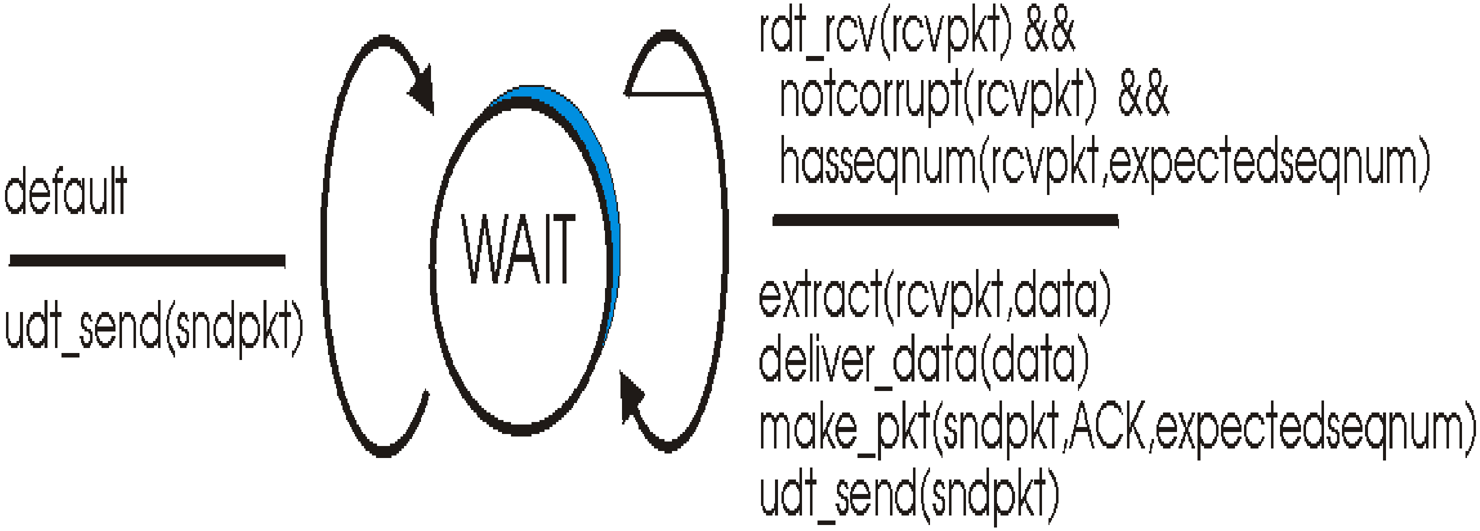
?

No packets outstanding

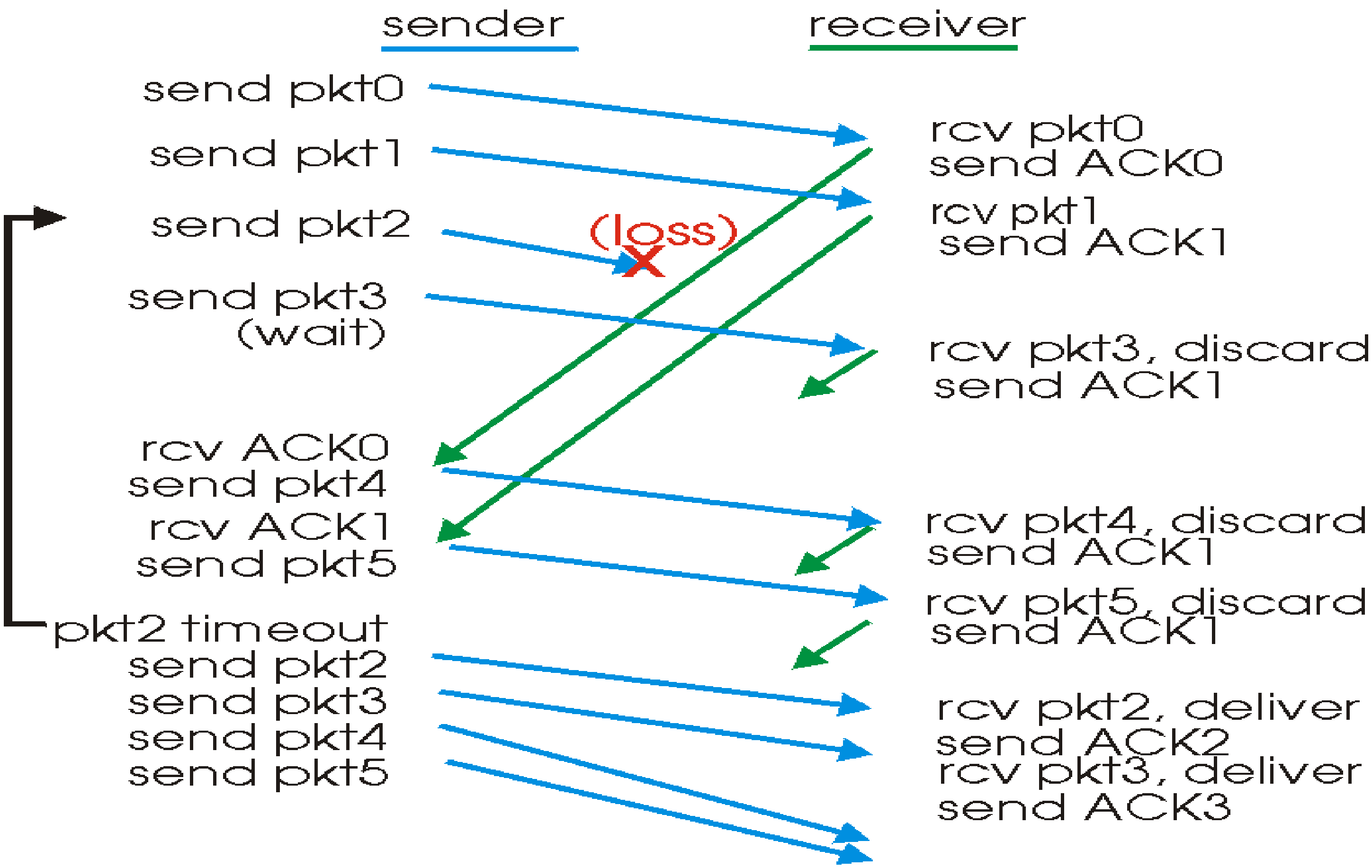
```
timeout
_______
start_timer
udt_send(sndpkt(base))
udt_send(sndpkt(base+1))
......
udt_send(sndpkt(nextseqnum-1))
```

Go-Back-N Receiver

• Receiver accepts packets in order only! out-of-order packets are simply dropped



Go-Back-N Example (N=4)



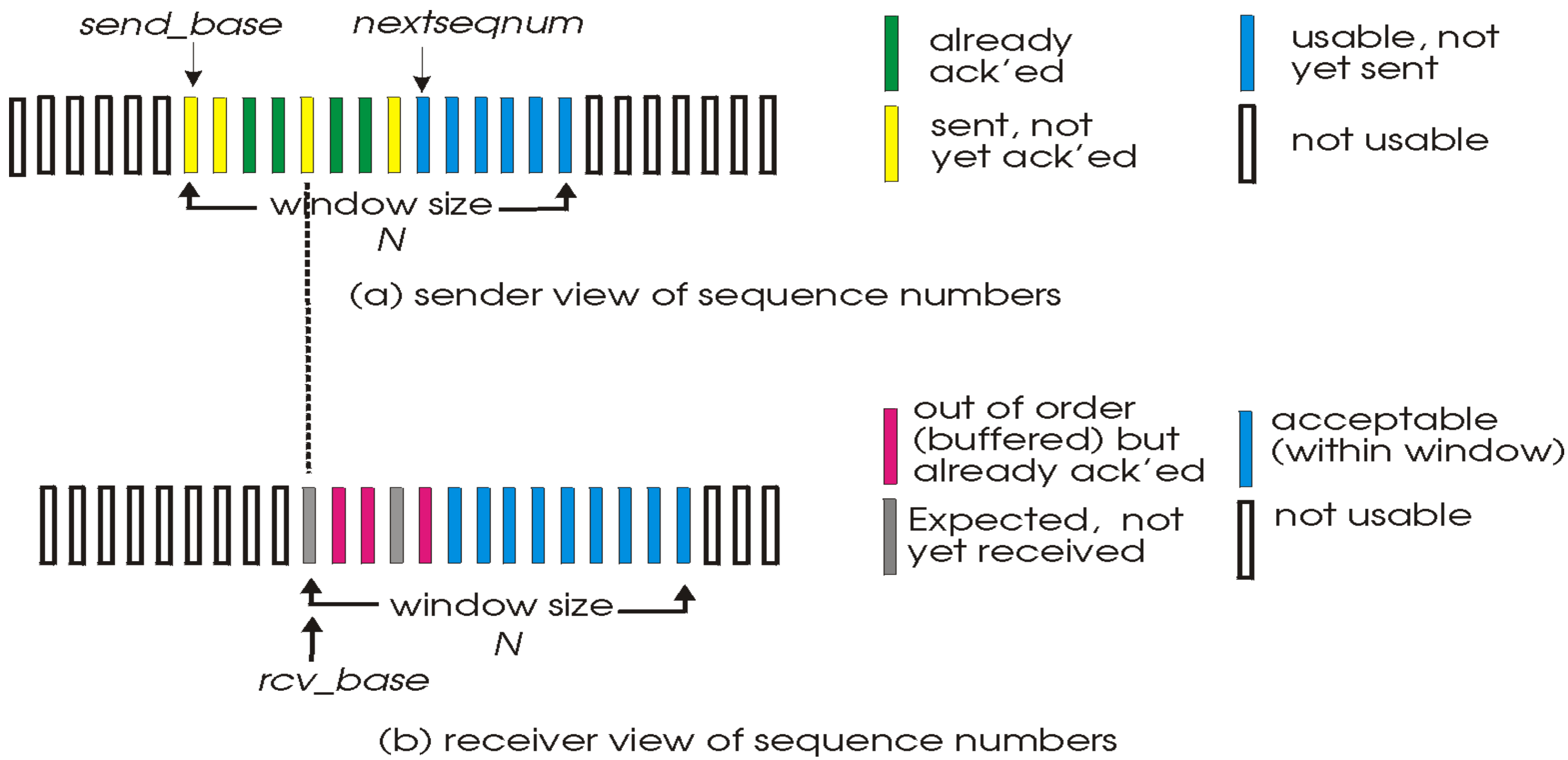
Go-Back-N Performance

- *Bandwidth-Delay Product* (ie "pipeline size") is defined as the product of the channel transmission speed and the propagation delay
- As transmission speed or propagation delay increases, more packets can be transmitted to "fill the pipeline"
- For channels with high Bandwidth-Delay product, Go-Back-N performance may deteriorate: the number of outstanding packets may be large and all these packets will be unnecessarily retransmitted when an error occurs

Selective Repeat

- Selective Repeat addresses the performance limitation of Go-back-N mentioned above
- Receiver indicates to sender which packet needs to be retransmitted; sender retransmits only that packet
- Receiver accepts and buffers packets received out of order within a limit imposed by a *receiver window*
- Groups of packets with <u>consecutive sequence numbers</u> (or completed sequences) are delivered to the higher layer at the sender
- A timer must be associated with each packet (but we can use one hardware timer to implement multiple logical timers)

Selective Repeat Windows



Selective Repeat Sender Event-Driven Algorithms

- Higher layer calls to transmit data:

  if there are unused sequence numbers

  then packetize and transmit;

  else reject the data;

- Timeout occurs:

  transmit the (single) packet which timed out;

- Ack is received:

  mark packet acked;

  if base can be moved

  then move it to the unacked packet with the lowest sequence number;

Selective Repeat Receiver Event-Driven Algorithms

- <u>Packet received, not corrupted, within current receive window:</u>

  Ack the received packet;

  if not previously received

      then buffer the packet;

  deliver consectively sequenced received packets to higher layer;

  move window forward;

- <u>Packet received, not corrupt, sequence number below window base</u>:

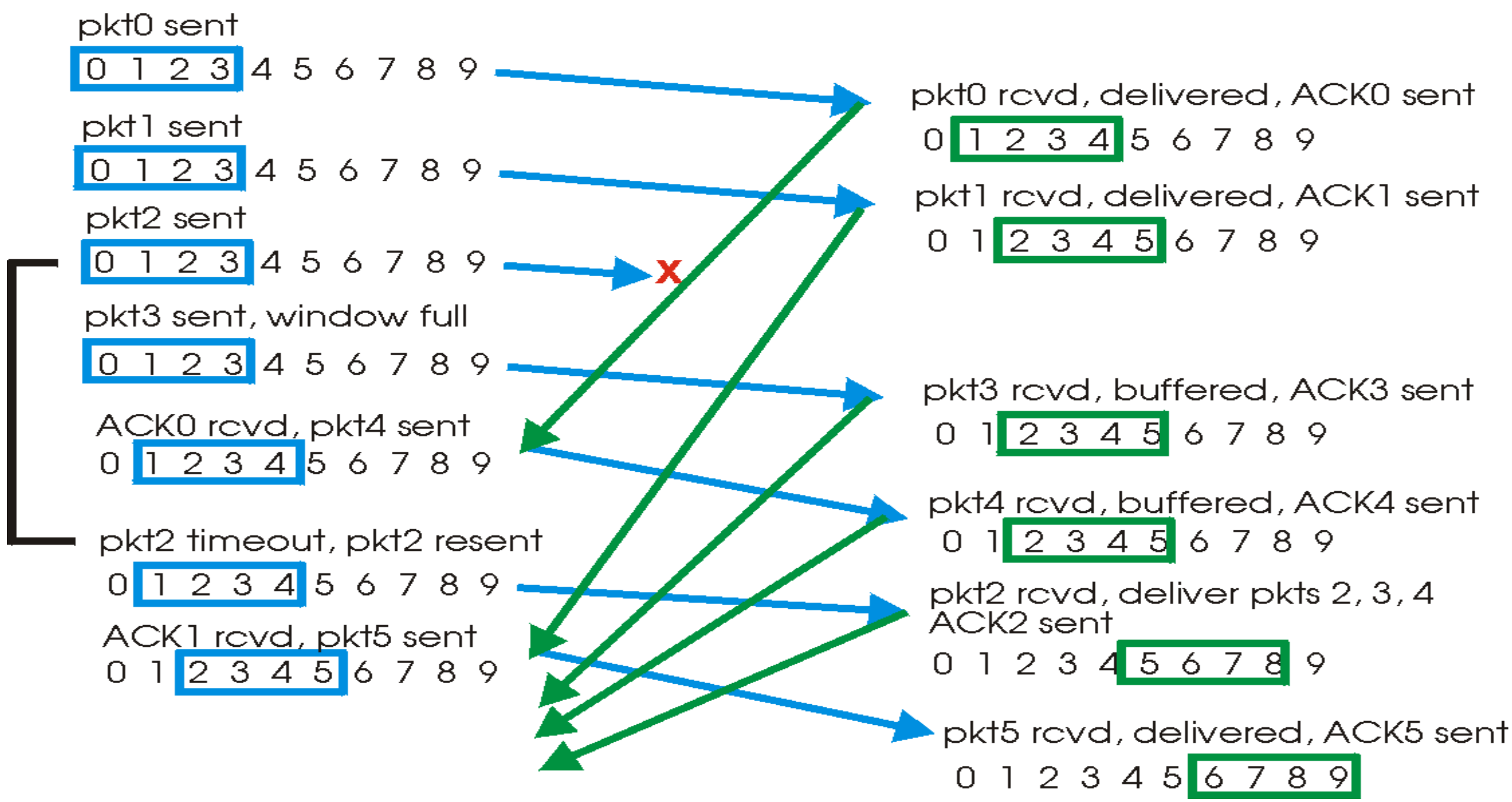
  Ack the received packet; /* packet previously acked and already

      delivered to higher layer*/

- <u>Packet received, corrupt, or sequence number beyond window</u>:

  Ignore the packet

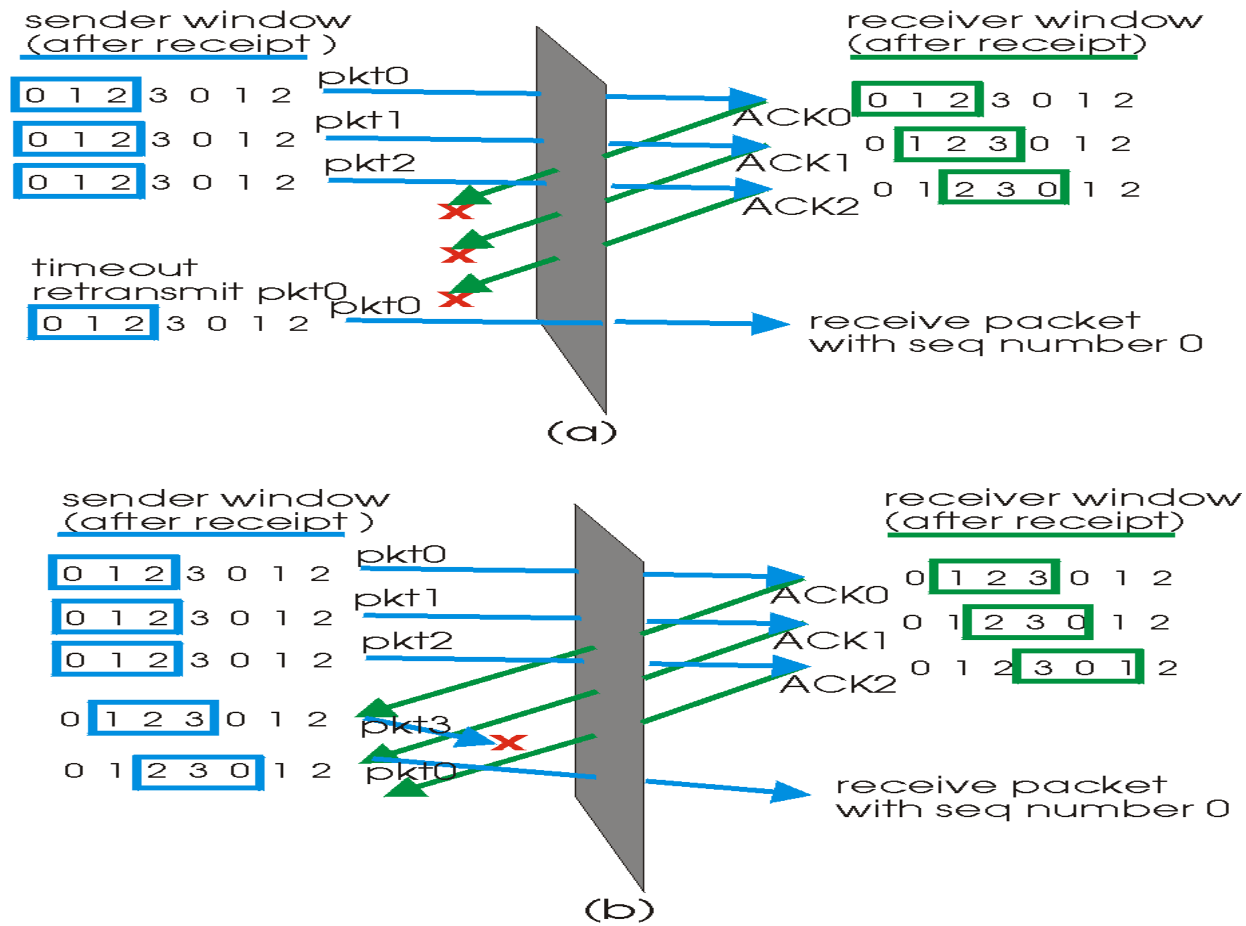
Selective Repeat Example



Setting The Window Size

- The window size N is an important parameter
- N should be large enough to allow filling the pipeline, thus making better utilization of the channel
- On the other hand, N is limited by the protocols (ensure receiver correctly identifies packets)
- It was found that N cannot be larger than half the sequence space length

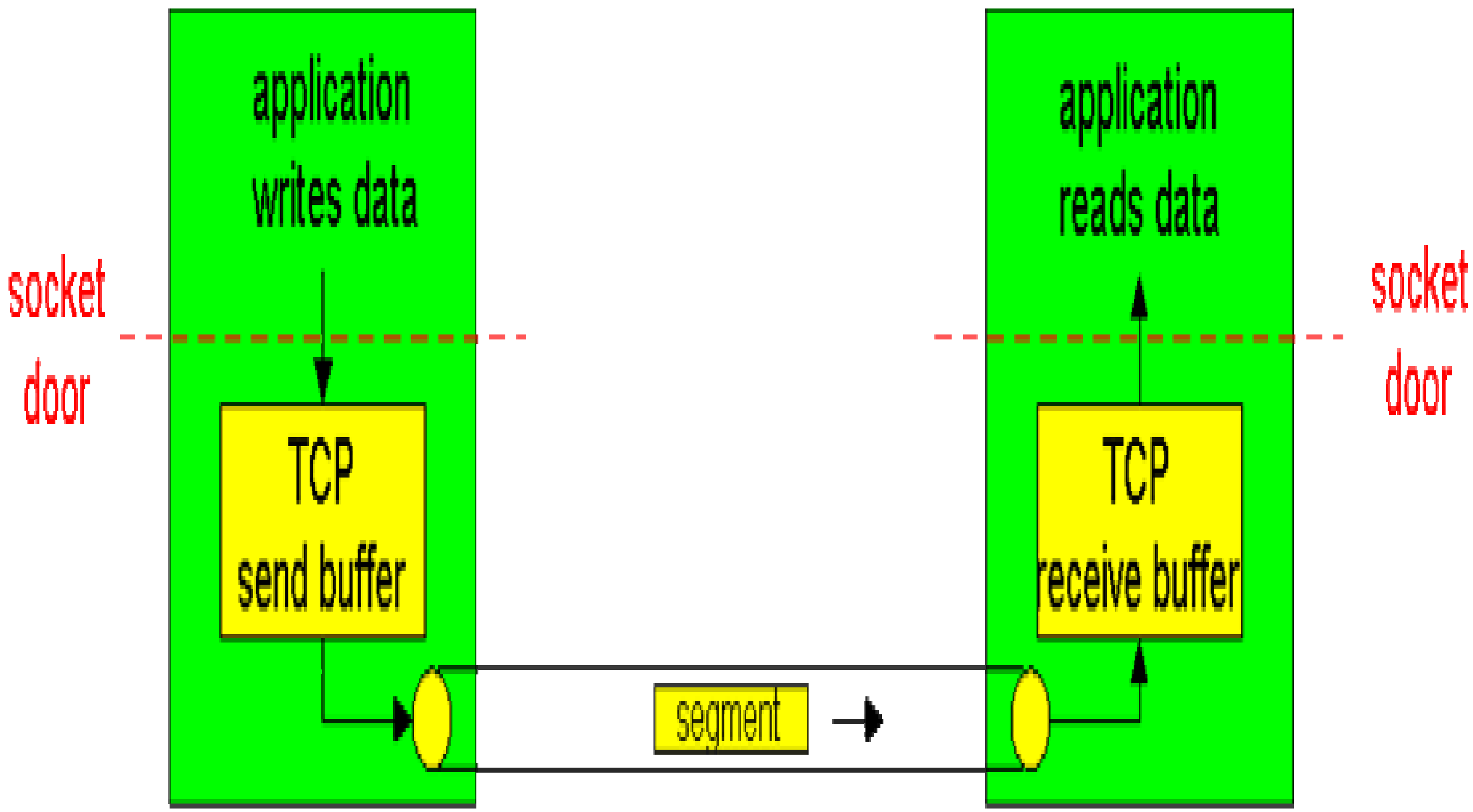
Misidentification Example



Reliable Transport Layer: TCP

- Full-duplex
- End-to-end protocol, transparent to network and lower layers in routers
- Connection-oriented, connection established through "three way handshake" protocol
- Byte Stream transfer, stream is divided into segments with a *maximum segment size* (MSS)
- Reliability through an ARQ type protocol
- Flow Control: receiver controls the amount of bytes a sender is allowed to send
- Point-to-point connection, no multicasting with TCP

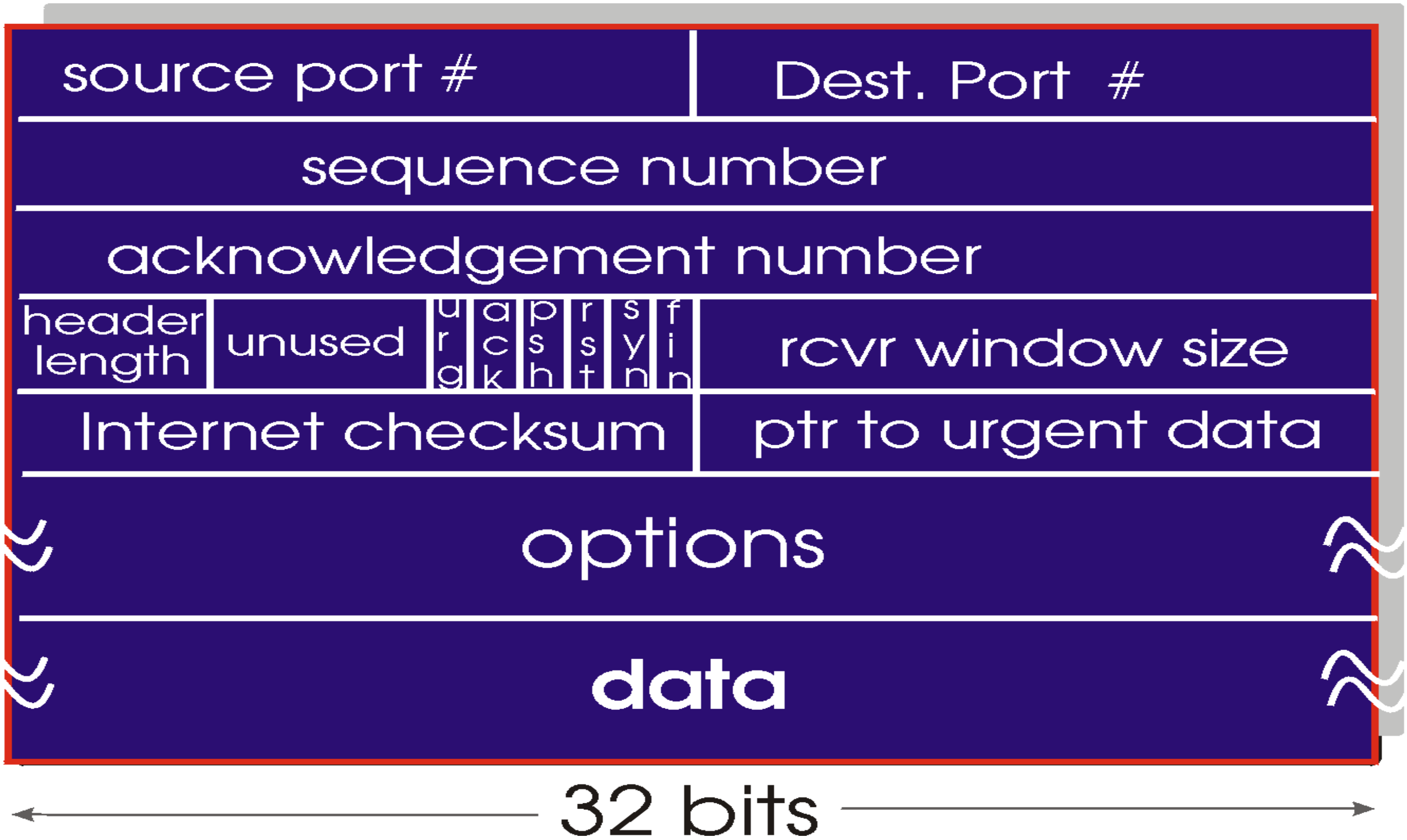
TCP Connection Model



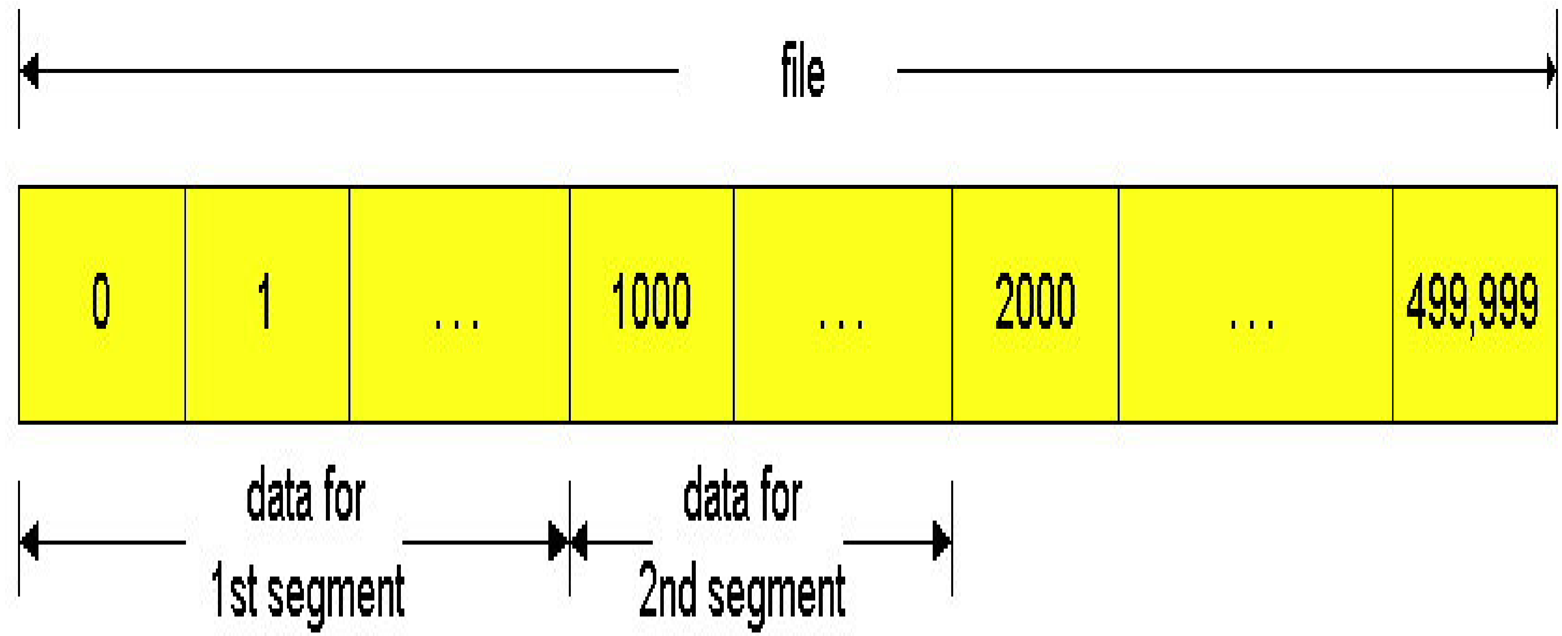
Segment Format

- Header contains:
  - Source and Destination Ports
  - Segment Sequence Number: that of the first byte in the segment (Byte Stream model)
  - Acknowledgment Number: sequence number of byte expected from the other side next
  - Header length: header as a fixed part of 20 bytes + optional fields
  - Receiver Window Size: the maximum number of bytes that the other side is allowed to send next
  - Header checksum: to ensure correctness of header field
  - Flags
  - 4 unused bits!

Segment Format (Cont.)



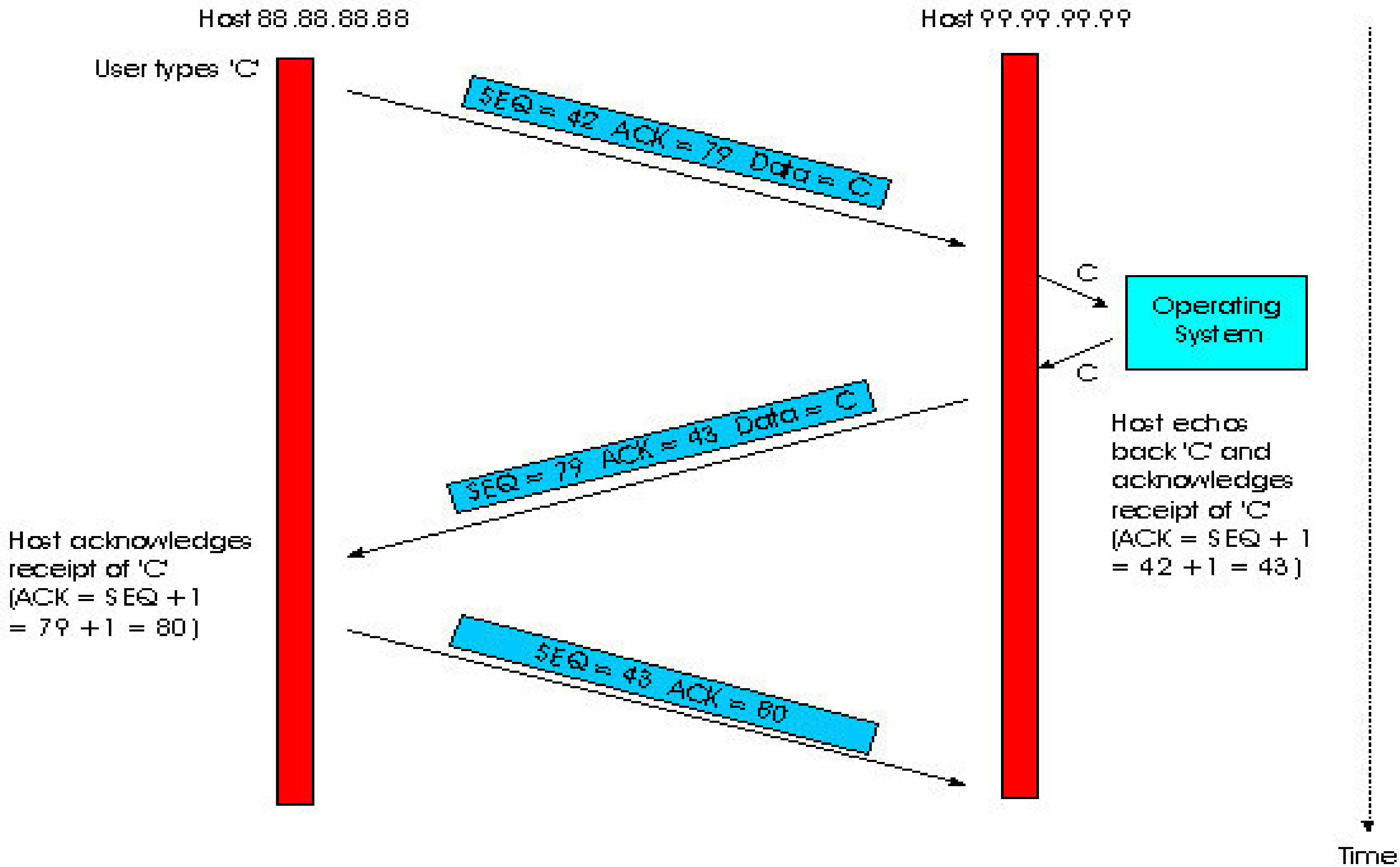
Segmented Byte Stream



Telnet: A TCP ACK example

- Telnet: appl. level protocol for remote login
- Interactive mode; typed characters are "echoed back" by remote Host (each character traverses the network twice)
- Full duplex stream of characters provides opportunity for ACK piggybacking
- In simplex (one way) data transfer, explicit ACKs are required

Telnet: ACK Example



TCP Reliable Data Transfer

- IP layer is often unreliable: packet drop (due to buffer overflow); data corruption (eg, noise, collisions).
- TCP approach: data is retransmitted following error detection (bad checksum) or packet loss detection (timeout or out of sequence reception)
- TCP uses pipelining to improve efficiency over paths with many hops and large end to end delays
- TCP error recovery mechanism similar to Go-Back-N
- TCP RFCs do not require receivers to drop out-of-order packets; some implementation keep such packets to save channel bandwidth

Three Key Events In Reliable TCP

- Event 1: TCP releases data segment to IP layer; segment retx timer started
- Event 2: segment timeout expires: segment is retransmitted
- Event 3: sender receives an ACK:
  - (a) First Time ACK, ie the ACK is for data not acked before (nextseqnum > ACK # > sendbase); the sender updates TCP state variables (sendbase, timer etc)
  - (b) Duplicate ACK (ACK # < or = sendbase); it re-ACKs old segments.

Sender Reaction To Duplicate ACKs

- Duplicate ACK (last ACK #) returned by receiver if:
  - (a) segment received out of order (seq num larger than expected)
  - (b) old segment received
- Sender ignores first two duplicate ACKs (timers still in force)
- Upon receiving THIRD duplicate ACK, the sender infers that the segment was indeed lost (as opposed to delayed); sender retransmits segment without waiting for timeout.

TCP Receiver ACK Generation

EVENT:

Arrival of in-order segment with expected sequence number. All data up to expected sequence number already acknowledged. No gaps in the received data.

ACTION:

Delayed ACK. Wait up to 500 ms for arrival of another in-order segment. If next in-order segment does not arrives in this interval, send an ACK

EVENT:

Arrival of in-order segment with expected sequence number. One other in-order segment waiting for ACK transmission. No gaps in the received data.

ACTION:

Immediately send single cumulative ACK, ACKing both in-order segments

TCP Receiver ACK Generation (Cont.)

EVENT:

Arrival of out-of-order segment with higher-than expected sequence number. Gap detected.

ACTION:

Immediately send duplicate ACK, indicating sequence number of next expected byte

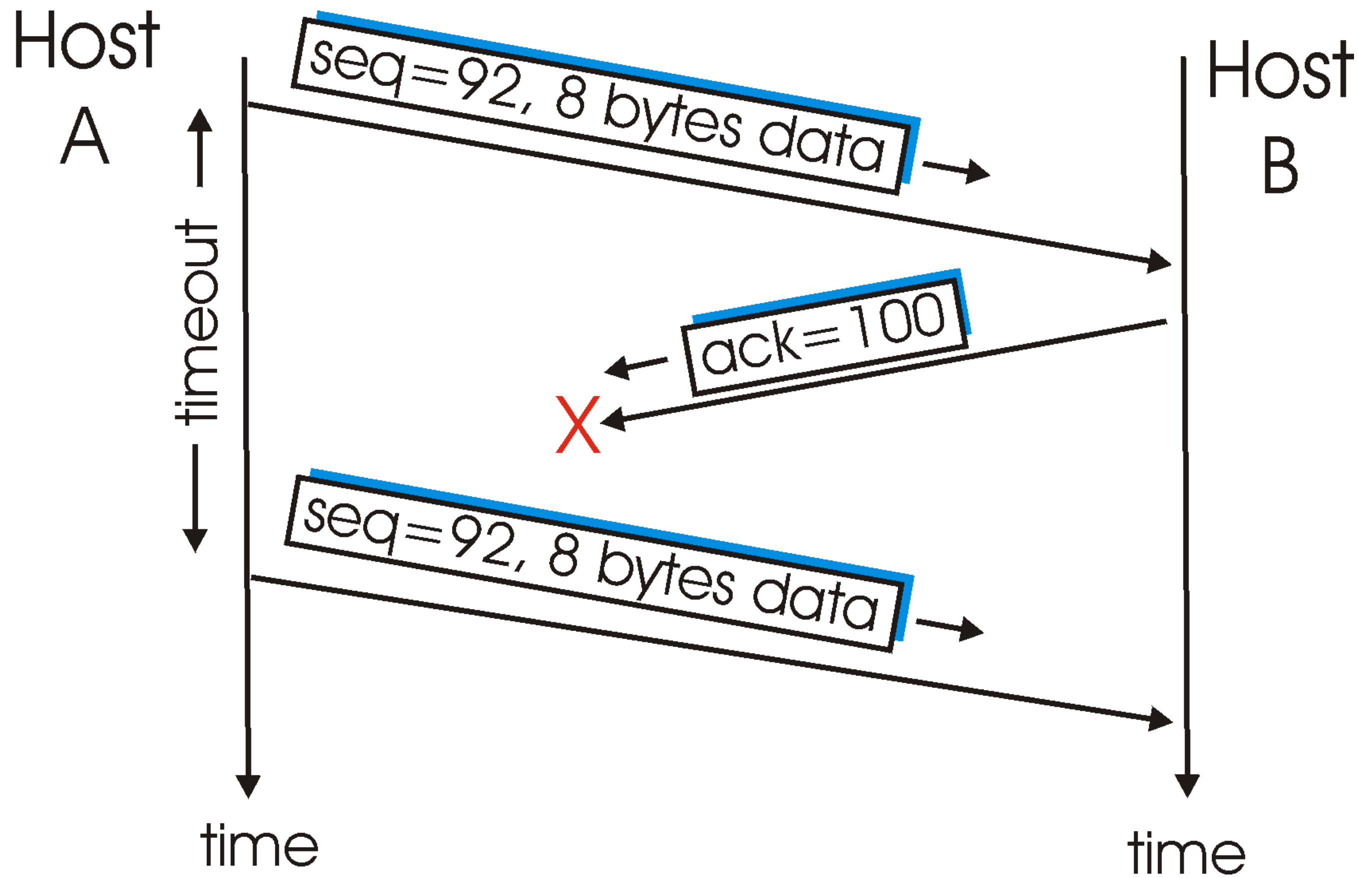
EVENT:

Arrival of segment that partially or completely fills in gap in received data

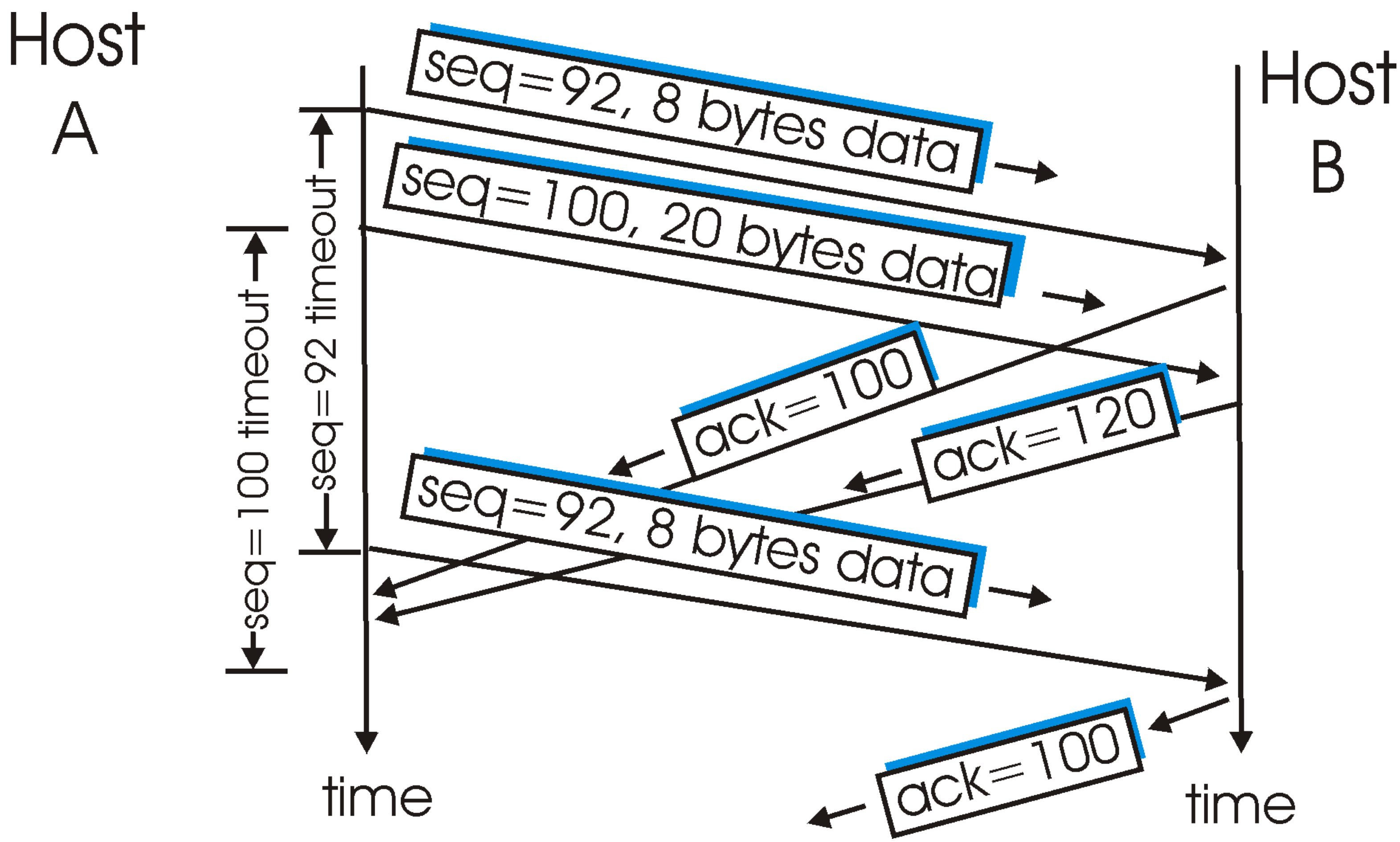
ACTION:

Immediately send ACK, provided that segment starts at the lower end of gap.

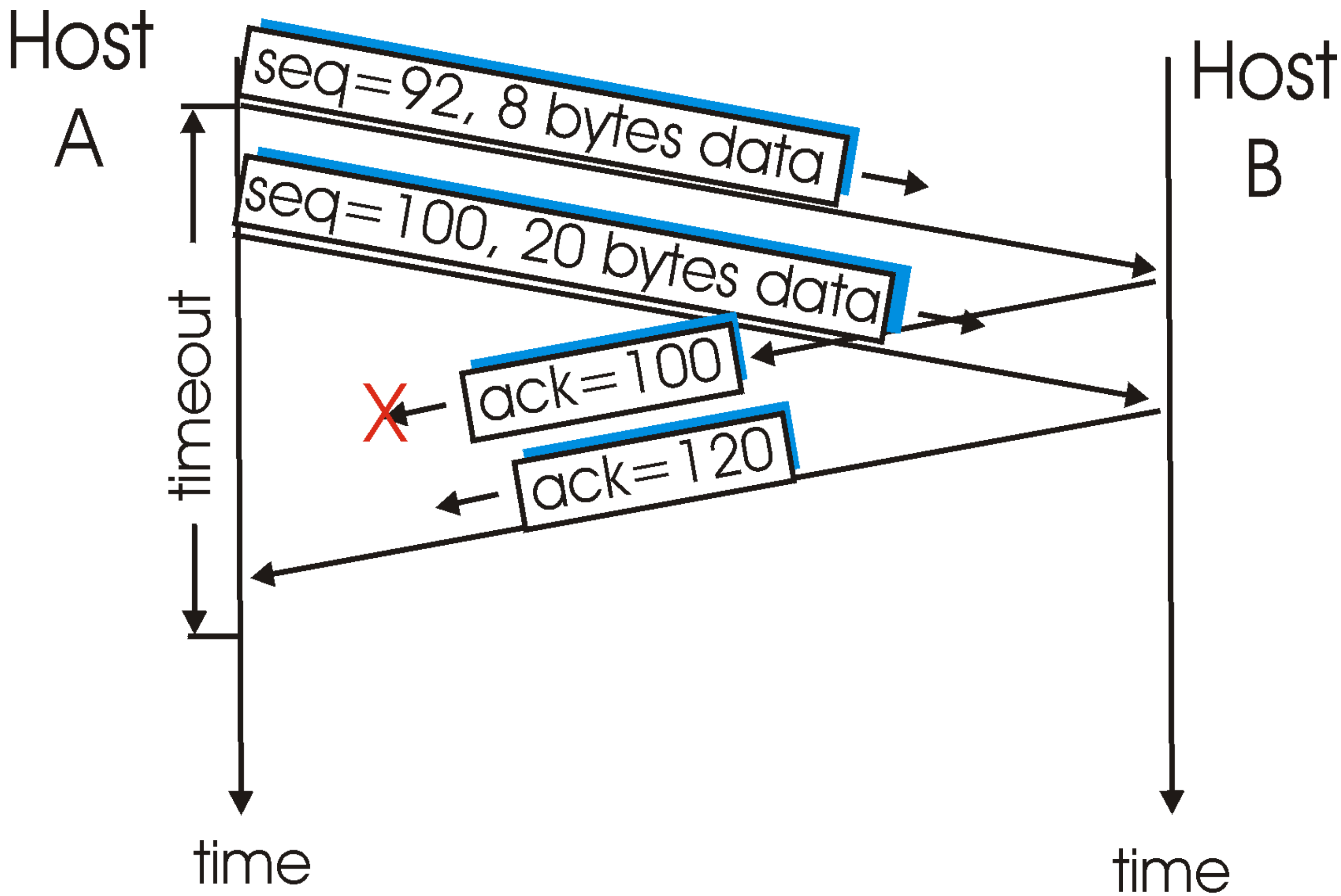
Example: TCP ACK loss



Example: TCP Timeout



Example: TCP ACK Loss



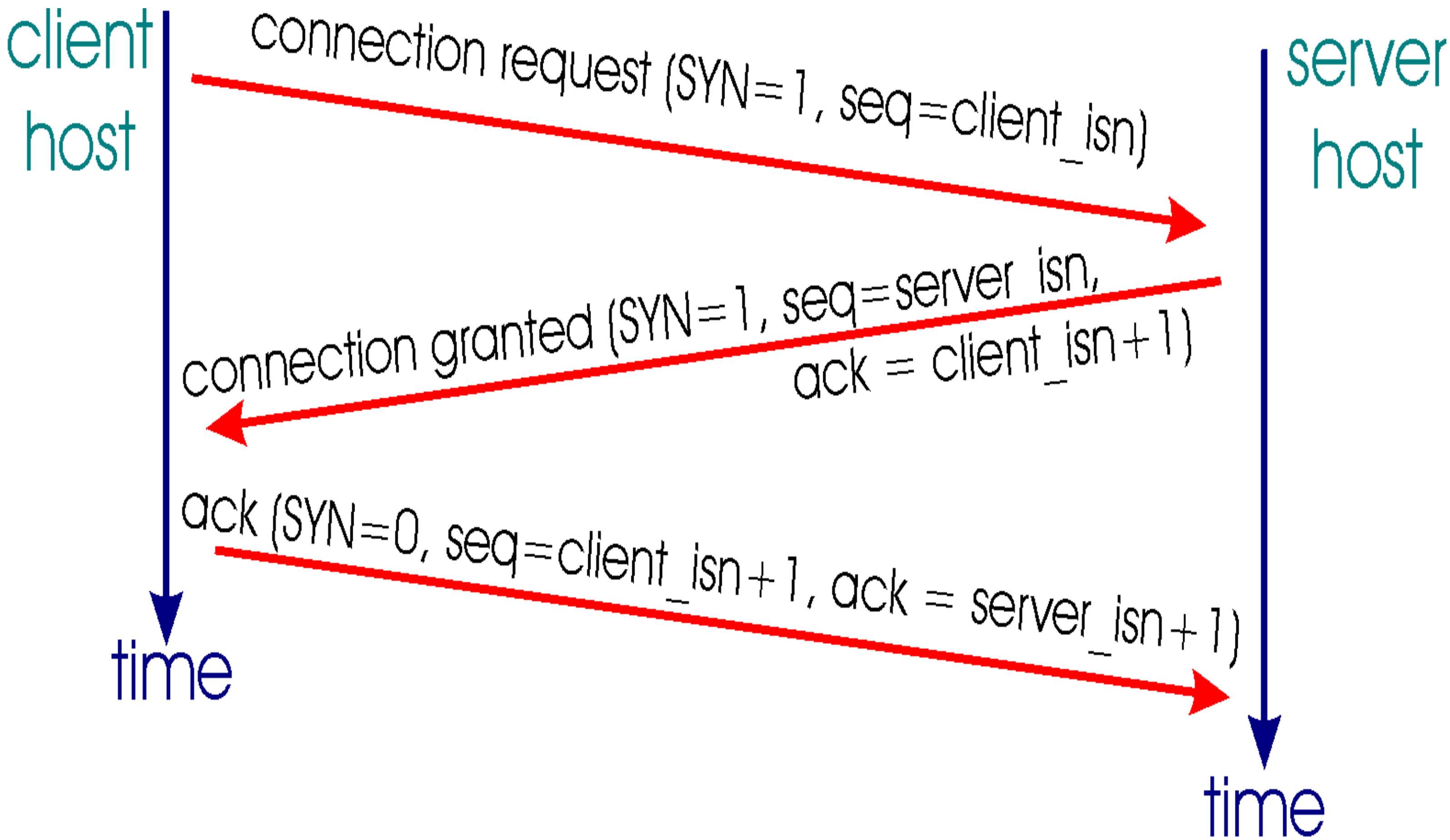
Flow/Congestion Control

- Flow Control (strict definition): regulate TCP flow so as to prevent receive buffer overflow at destination
- Flow Control (more general definition): regulate TCP flow so as to prevent buffer overflow anywhere along the path
- Congestion Control: regulate TCP flow(s) so as to avoid congestion in the entire network and to achieve efficient, fair sharing of resources.
- Key TCP flow/congestion mechanism: adjustable sender window

TCP Connection Management

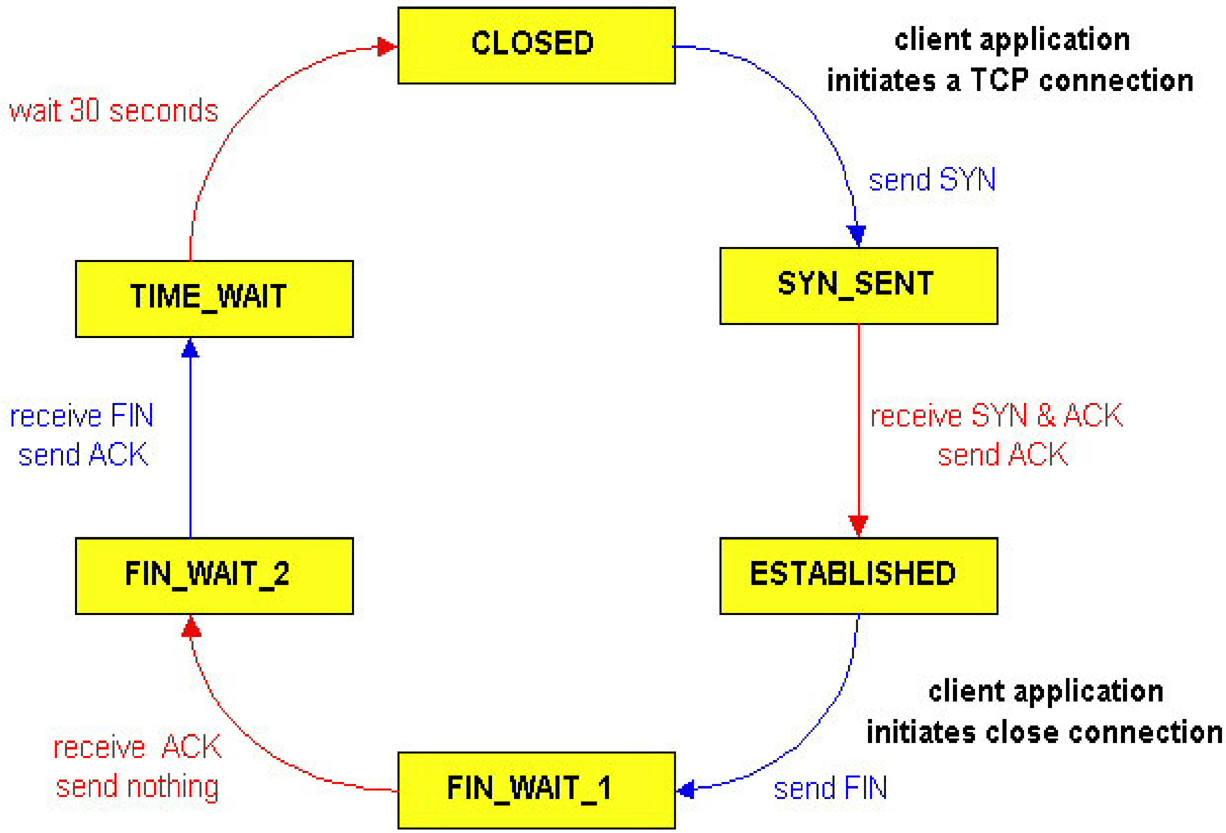
- TCP connection is set up using the *three way handshake* protocol
- Special segments (SYN segment, SYNACK segment) exchange initial client and server sequence numbers and allocate buffers
- Three Way Handshake protocol allows to detect and eliminate "old" connection requests (more robust than two separate handshakes)
- Another Three Way Handshake (with FIN flag turned on) is used to close the connection, releasing all resources

Three Way Handshake



Renato.LoCigno@dit.unitn.it

TCP Connection States (Client)



Renato.LoCigno@dit.unitn.it

TCP Connection States (Server)