Communication Networks

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Materials inspired from Scott Shenker, Jennifer Rexford, and Sharon Goldberg

Last week on

Communication Networks

Border Gateway Protocol

policies and more

1 **BGP Policies**

Follow the Money

2 **Protocol**

How does it work?

3 **Problems**

security, performance, …

Border Gateway Protocol

policies and more

1 **BGP Policies**

Follow the Money

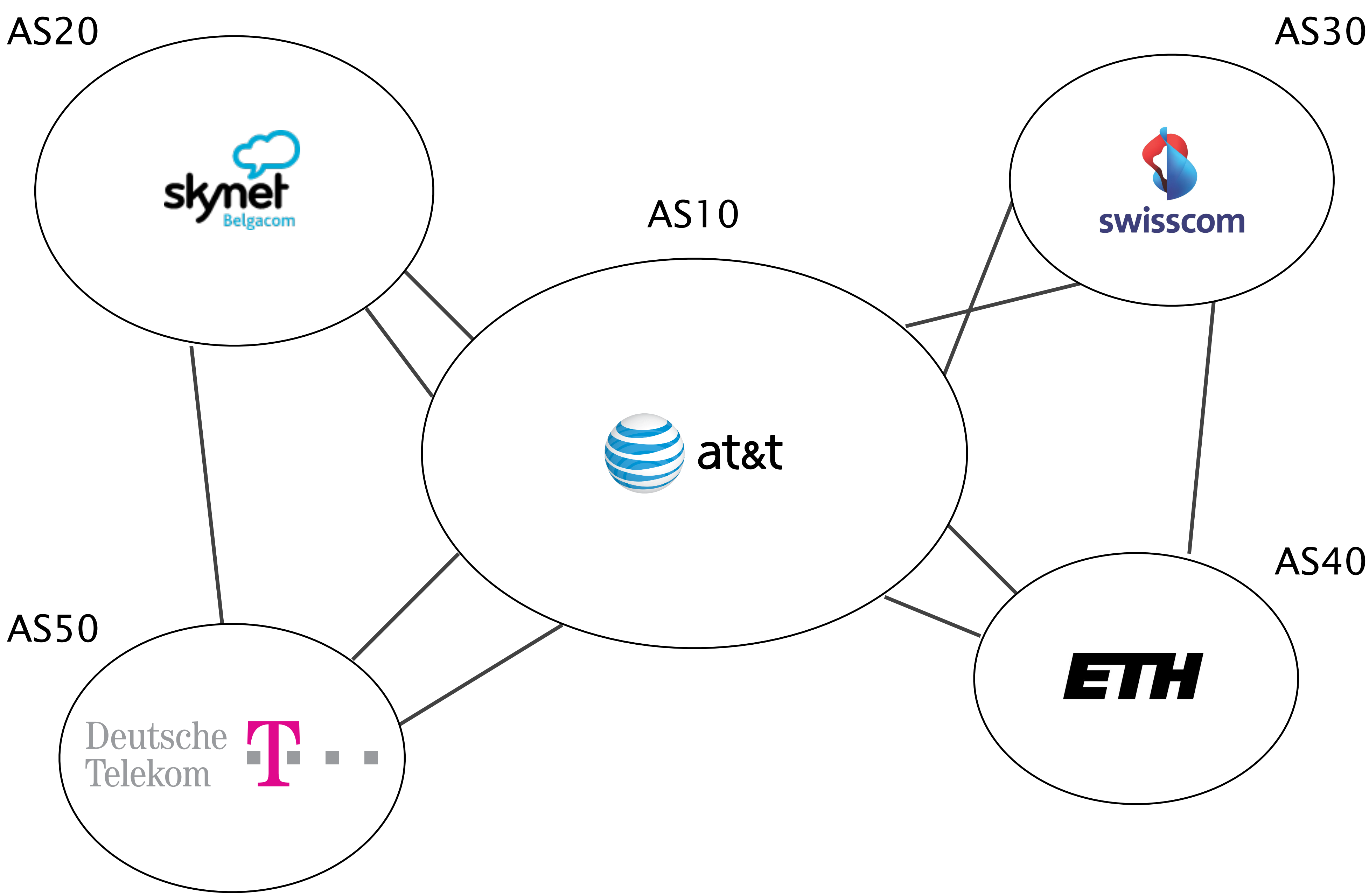
Protocol

How does it work?

Problems

security, performance, …

The Internet topology is shaped according to *business* relationships



There are 2 main business relationships today:

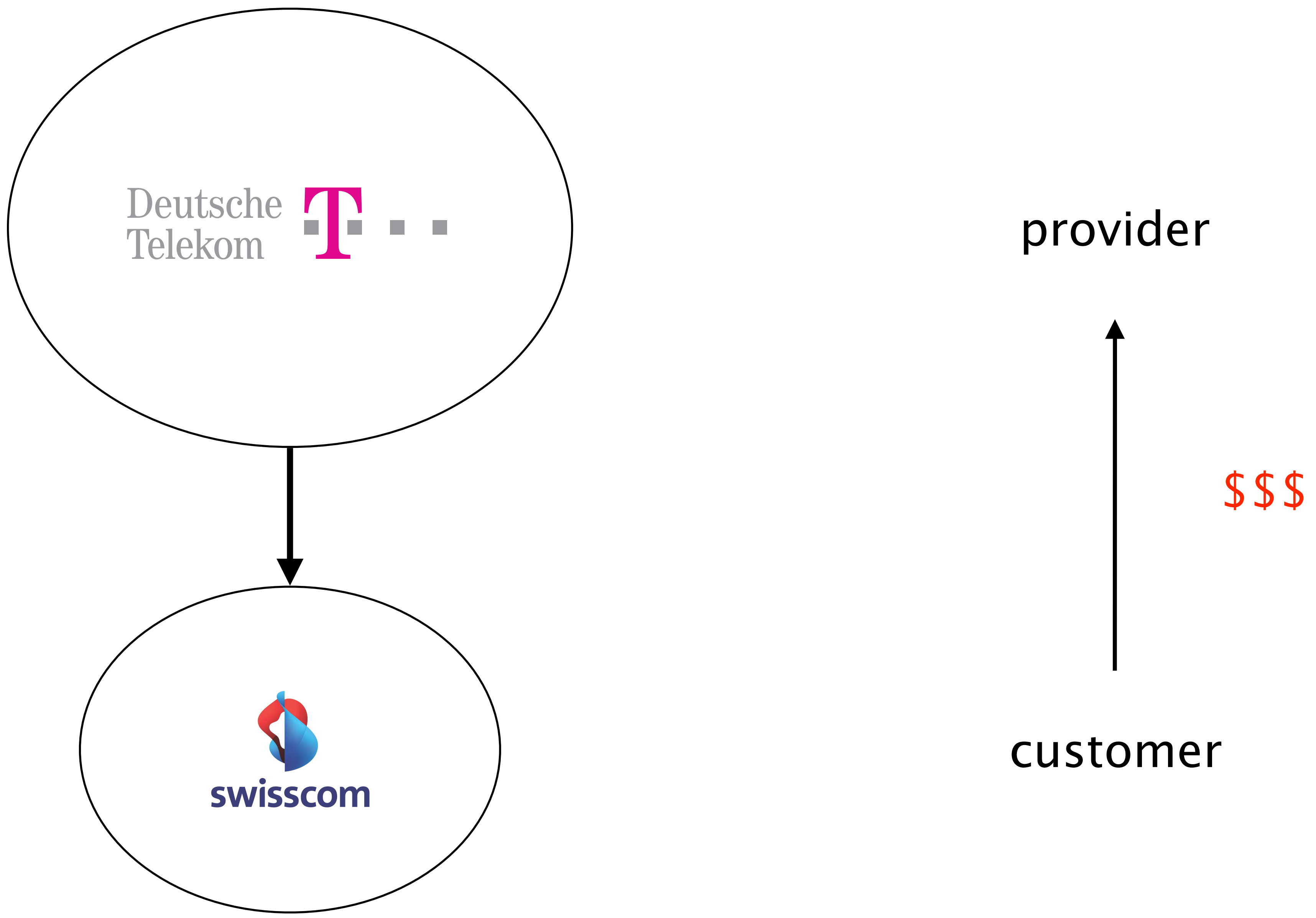
- customer/provider
- peer/peer

many less important ones (siblings, backups,…)

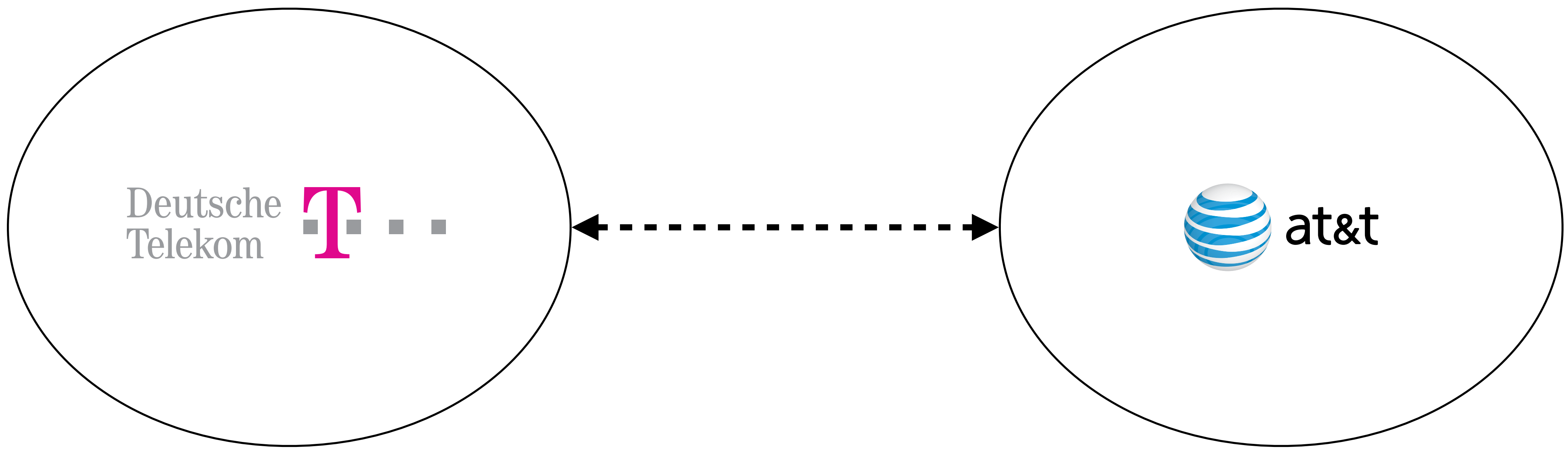
There are 2 main business relationships today:

- customer/provider
- peer/peer

Customers pay providers
to get Internet connectivity



Peers don’t pay each other for connectivity,
they do it *out of common interest*



peer peer

DT and ATT exchange *tons* of traffic.
they save money by directly connecting to each other

Business relationships conditions

route selection

For a destination p, prefer routes coming from

- customers over
- peers over
- providers

route type

Business relationships conditions

route exportation

		send to		
		customer	peer	provider
	customer			
from	peer			
	provider			

Routes coming from customers are propagated to everyone else

		send to		
		customer	peer	provider
	customer	✓	✓	✓
from	peer			
	provider			

Routes coming from peers and providers are only propagated to customers

		send to		
		customer	peer	provider
	customer	✓	✓	✓
from	peer	✓	-	-
	provider	✓	-	-

Border Gateway Protocol

policies and more

BGP Policies

Follow the Money

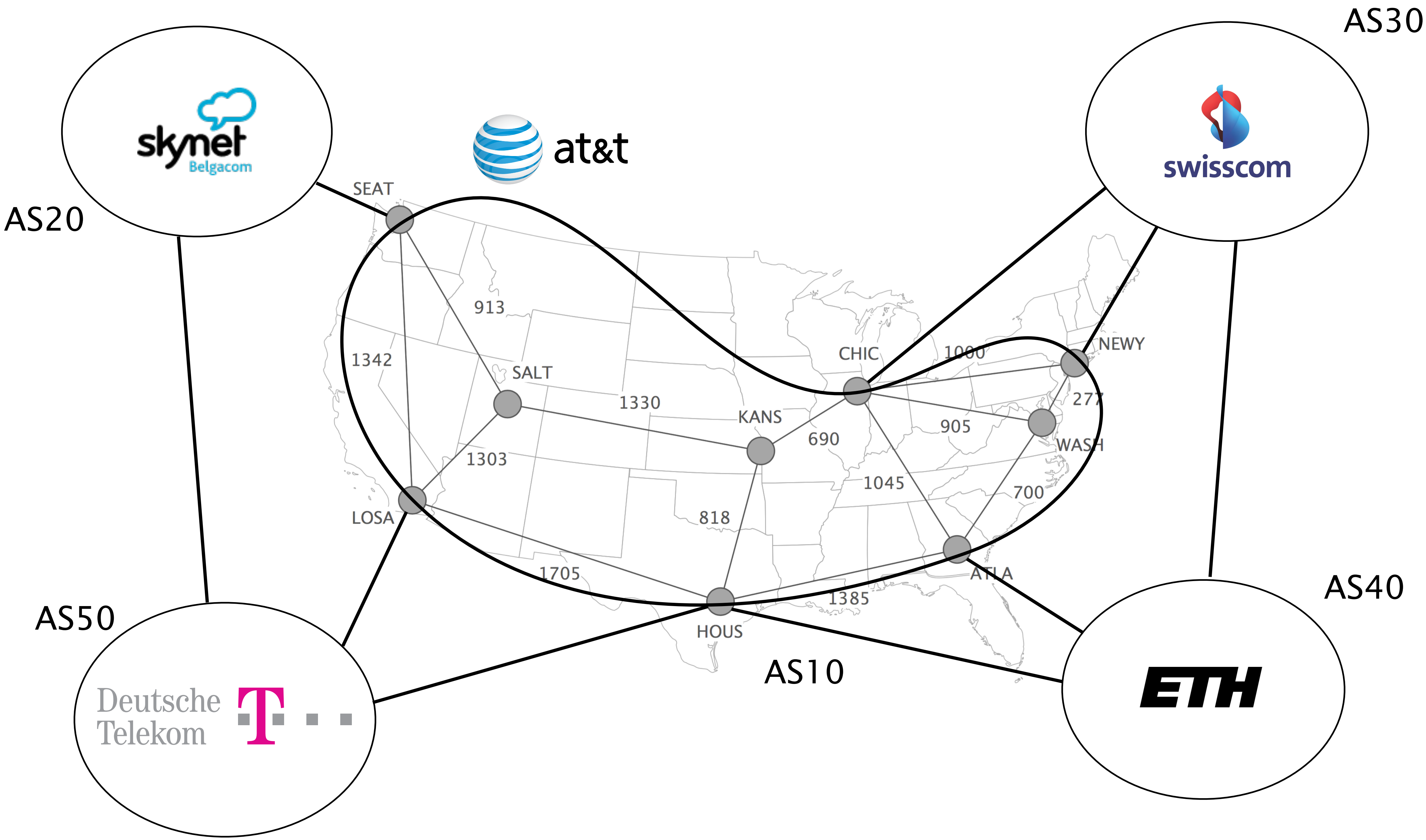
2 Protocol

How does it work?

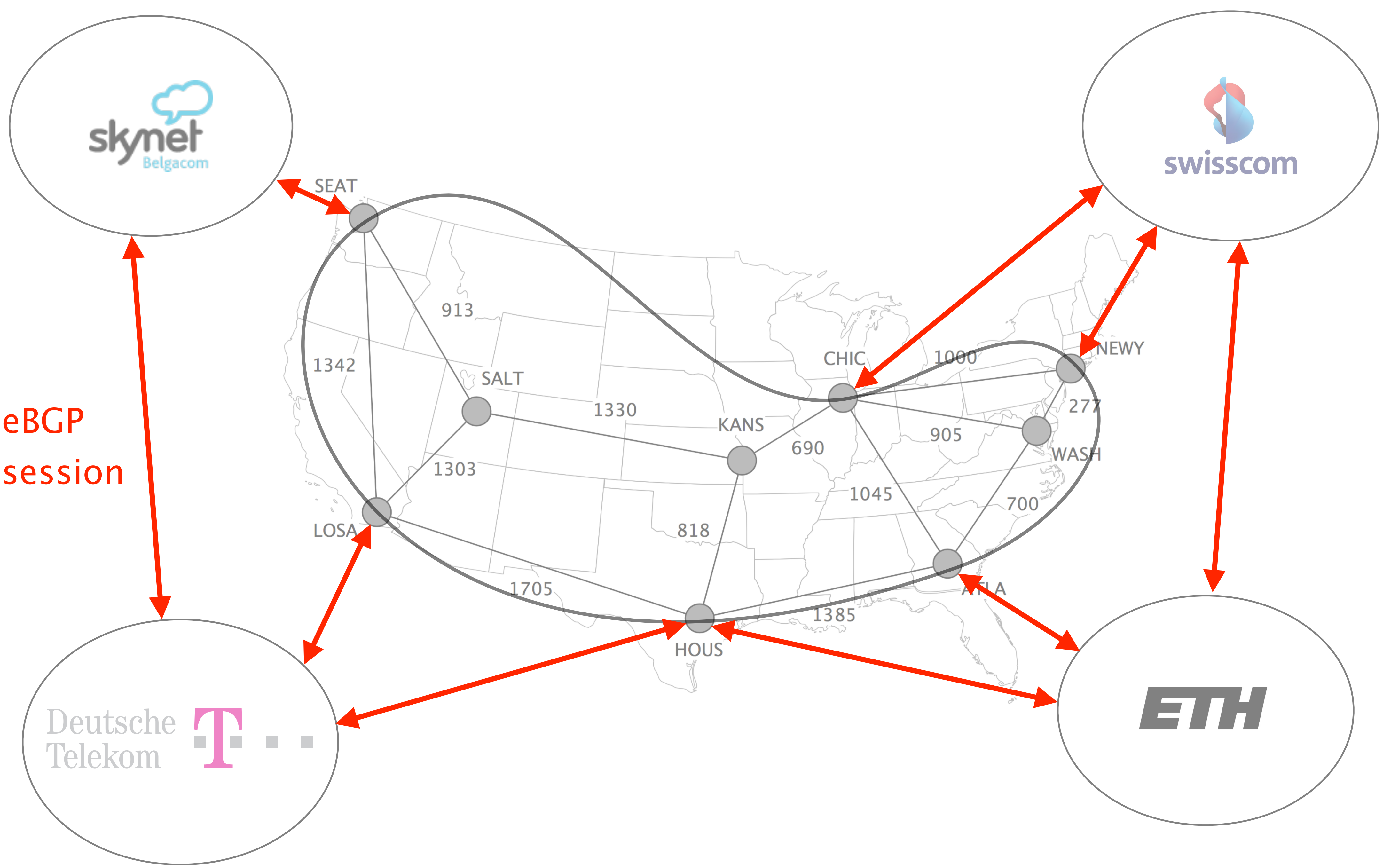
Problems

security, performance, …

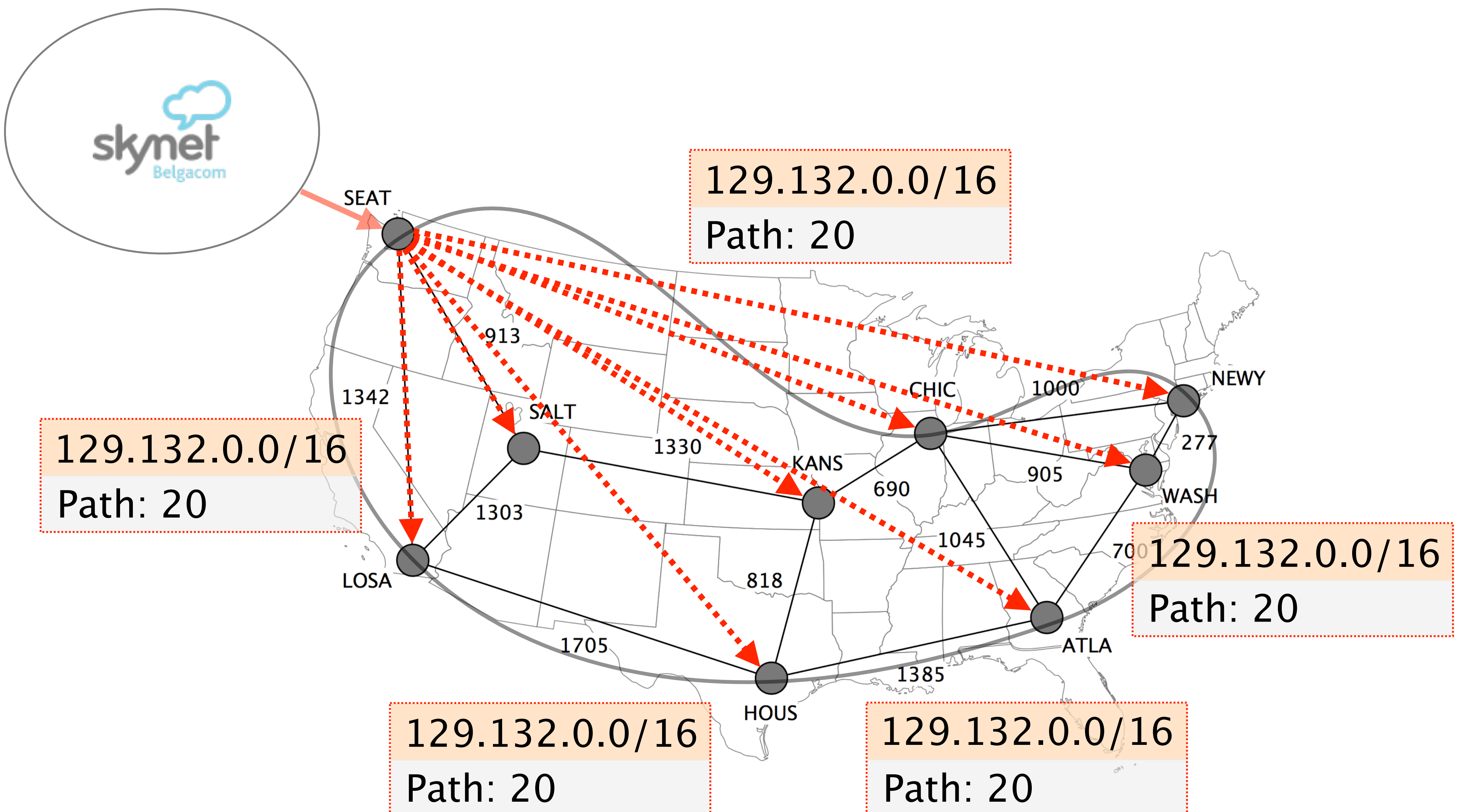
BGP sessions come in two flavors



external BGP (eBGP) sessions
connect border routers in different ASes



iBGP sessions are used to disseminate externally-learned routes internally



BGP UPDATEs carry an IP prefix together with a set of attributes

IP prefix

Attributes

Describe route properties

used in route selection/exportation decisions

are either local (*only* seen on iBGP)

or global (seen on iBGP *and* eBGP)

Attributes	Usage
NEXT-HOP	egress point identification
AS-PATH	loop avoidance outbound traffic control inbound traffic control
LOCAL-PREF	outbound traffic control
MED	inbound traffic control

Prefer routes…

with higher LOCAL-PREF

with shorter AS-PATH length

with lower MED

learned via eBGP instead of iBGP

with lower IGP metric to the next-hop

with smaller egress IP address (tie-break)

Border Gateway Protocol

policies and more

BGP Policies

Follow the Money

Protocol

How does it work?

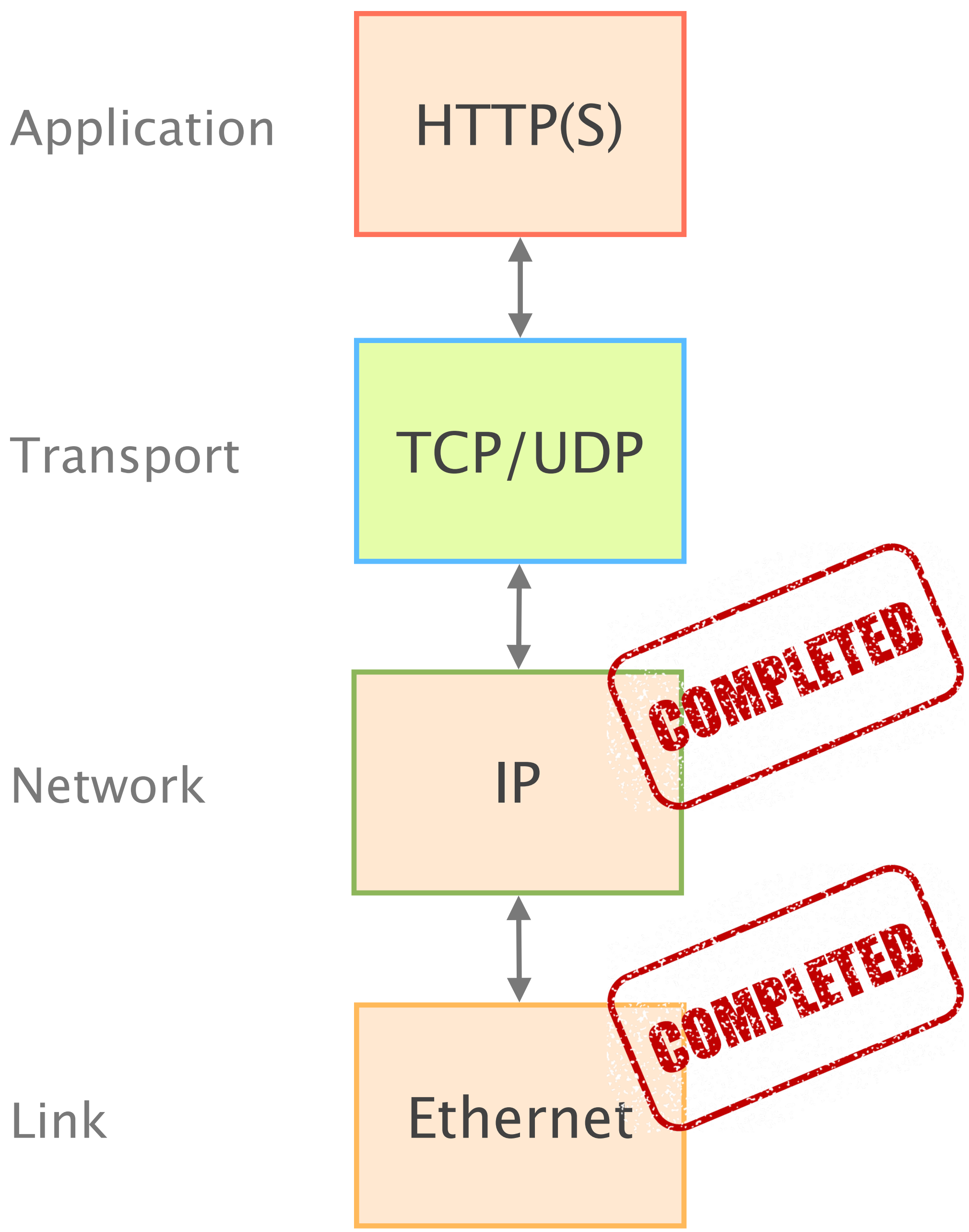
3 **Problems**

security, performance, …

This week on

Communication Networks

We’re continuing our journey up the layers, now looking at the transport layer



But first…

Let's finish BGP

BGP UPDATEs carry an IP prefix together with a set of attributes

IP prefix

Attributes

Describe route properties

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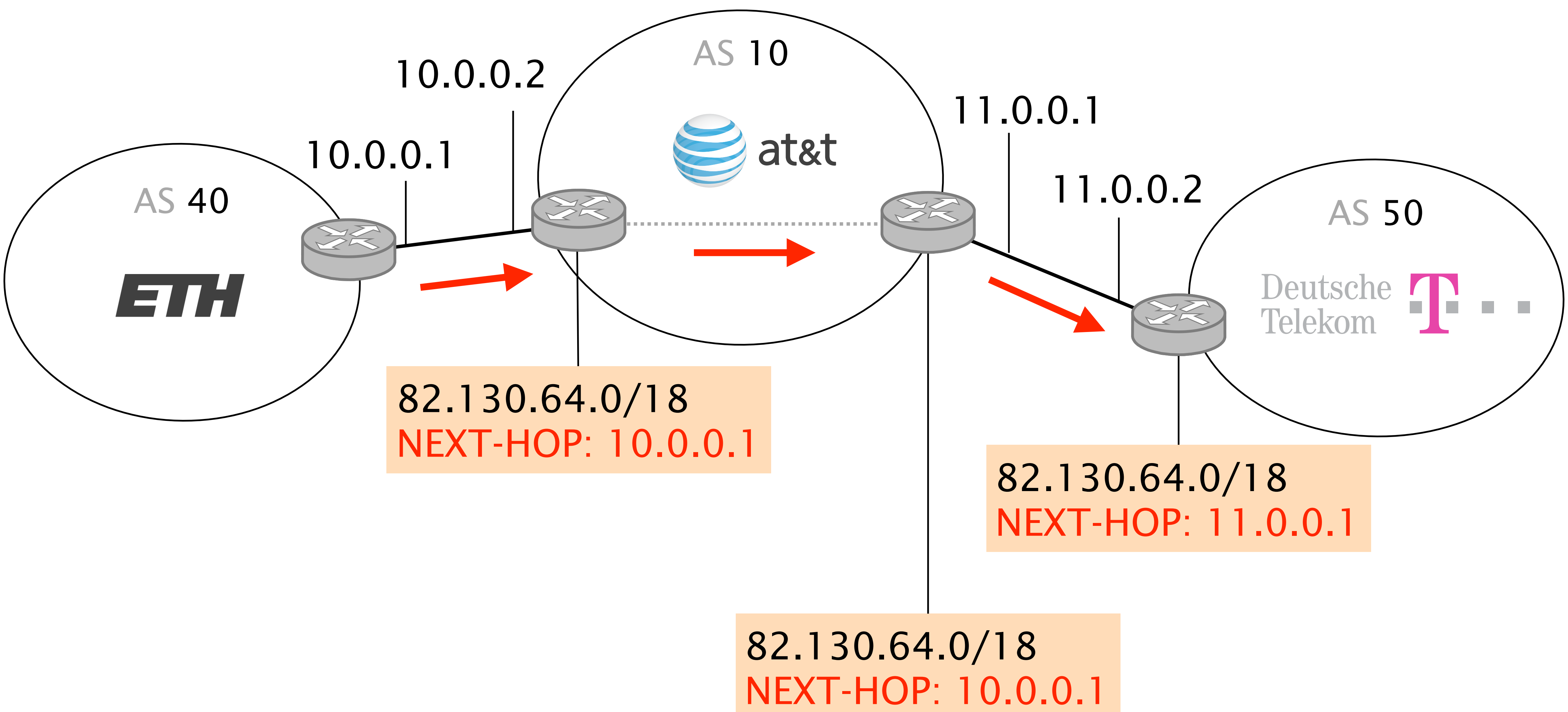
or global (seen on iBGP *and* eBGP)

Attributes	Usage
NEXT-HOP	egress point identification
AS-PATH	loop avoidance outbound traffic control inbound traffic control
LOCAL-PREF	outbound traffic control
MED	inbound traffic control

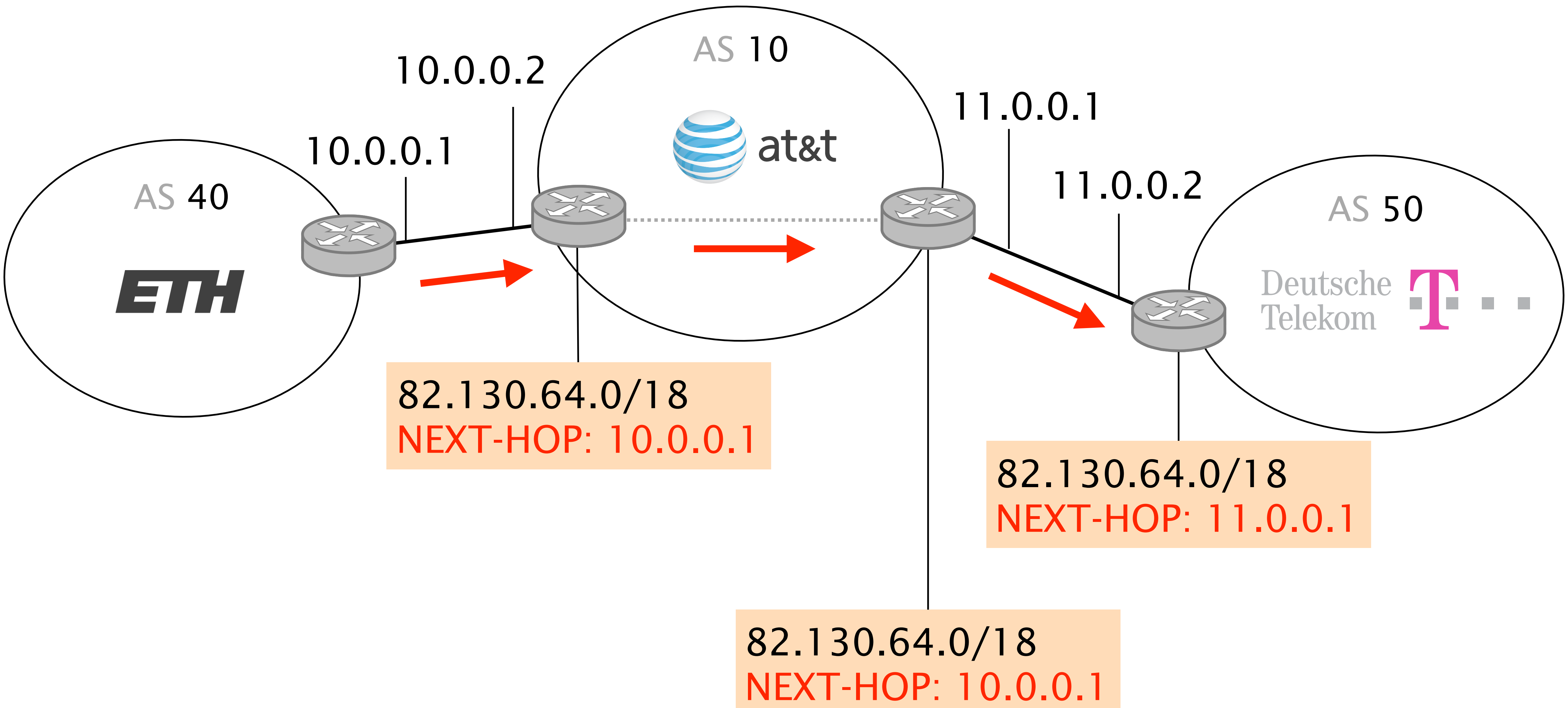
The NEXT-HOP is a global attribute which indicates where to send the traffic next

The NEXT-HOP is set when the route enters/exits an AS,

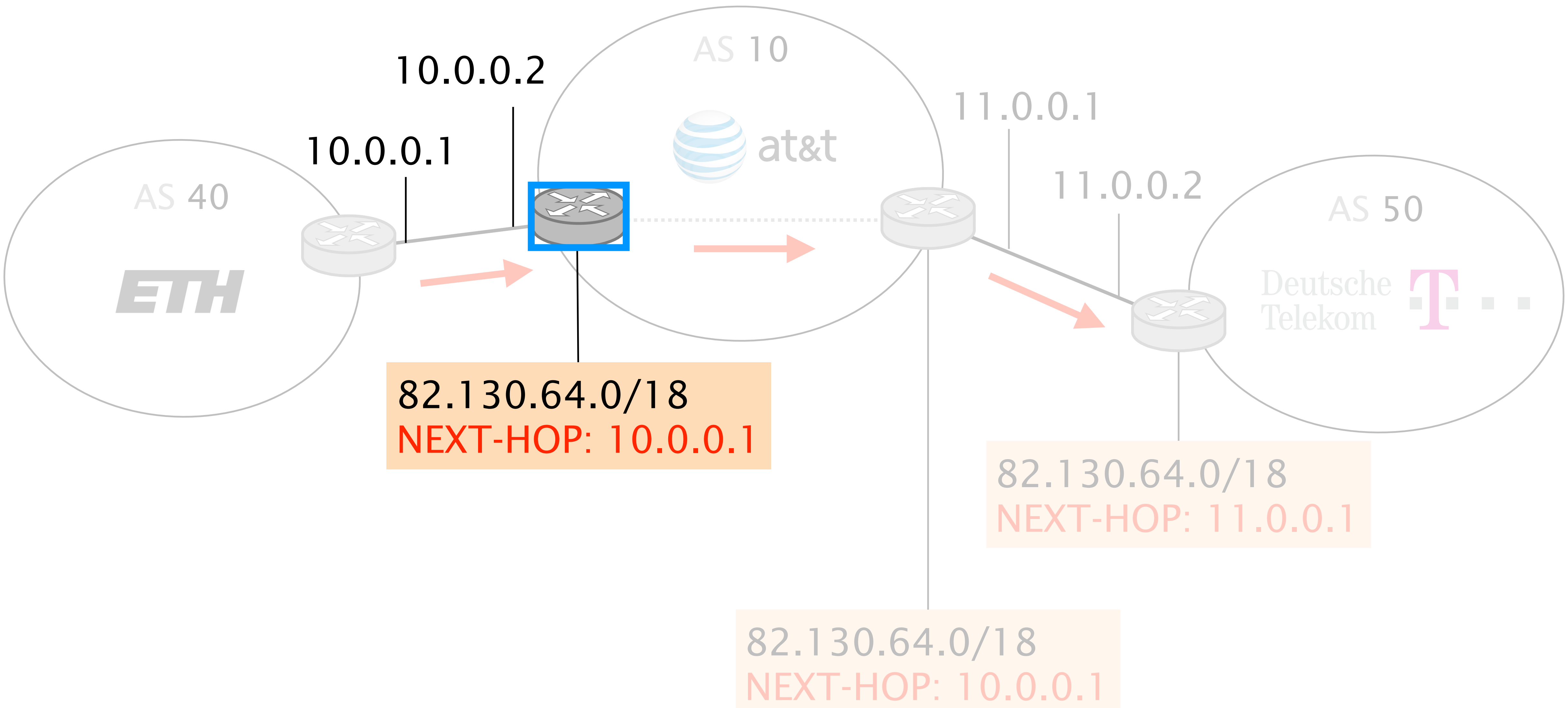
it does not change within the AS



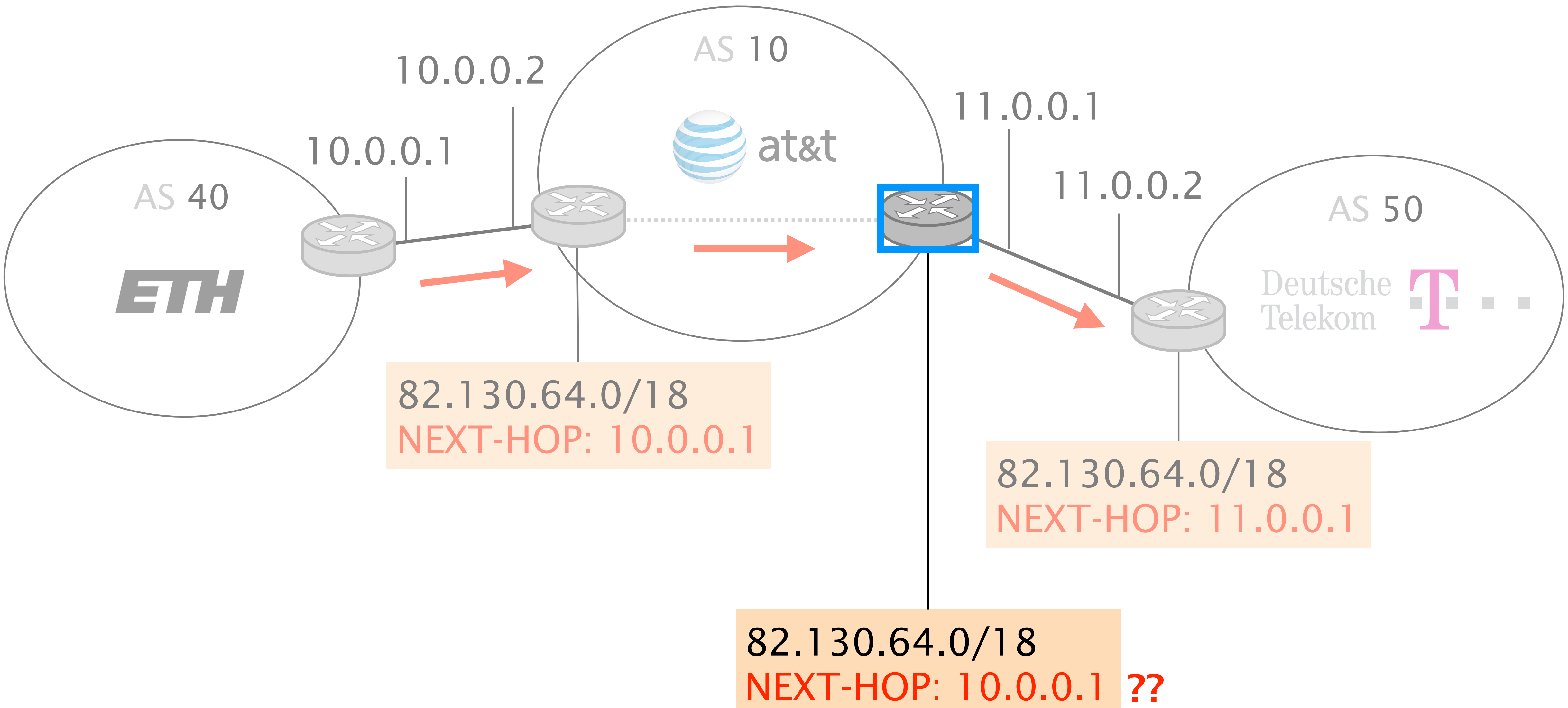
For externally-learned route, this means that the NEXT-HOP is the IP address of the neighbor's eBGP router, here 10.0.0.1



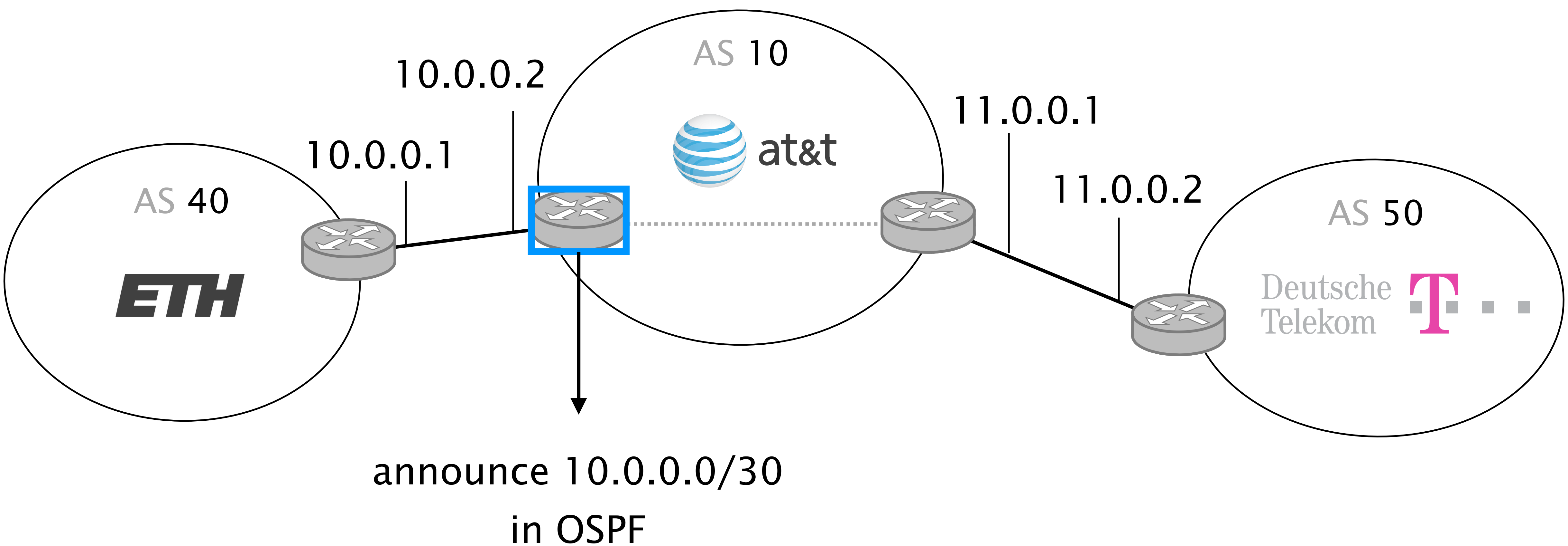
For this router, reaching 10.0.0.1 is not a problem as it is directly connected to the corresponding subnet (10.0.0.0/30)



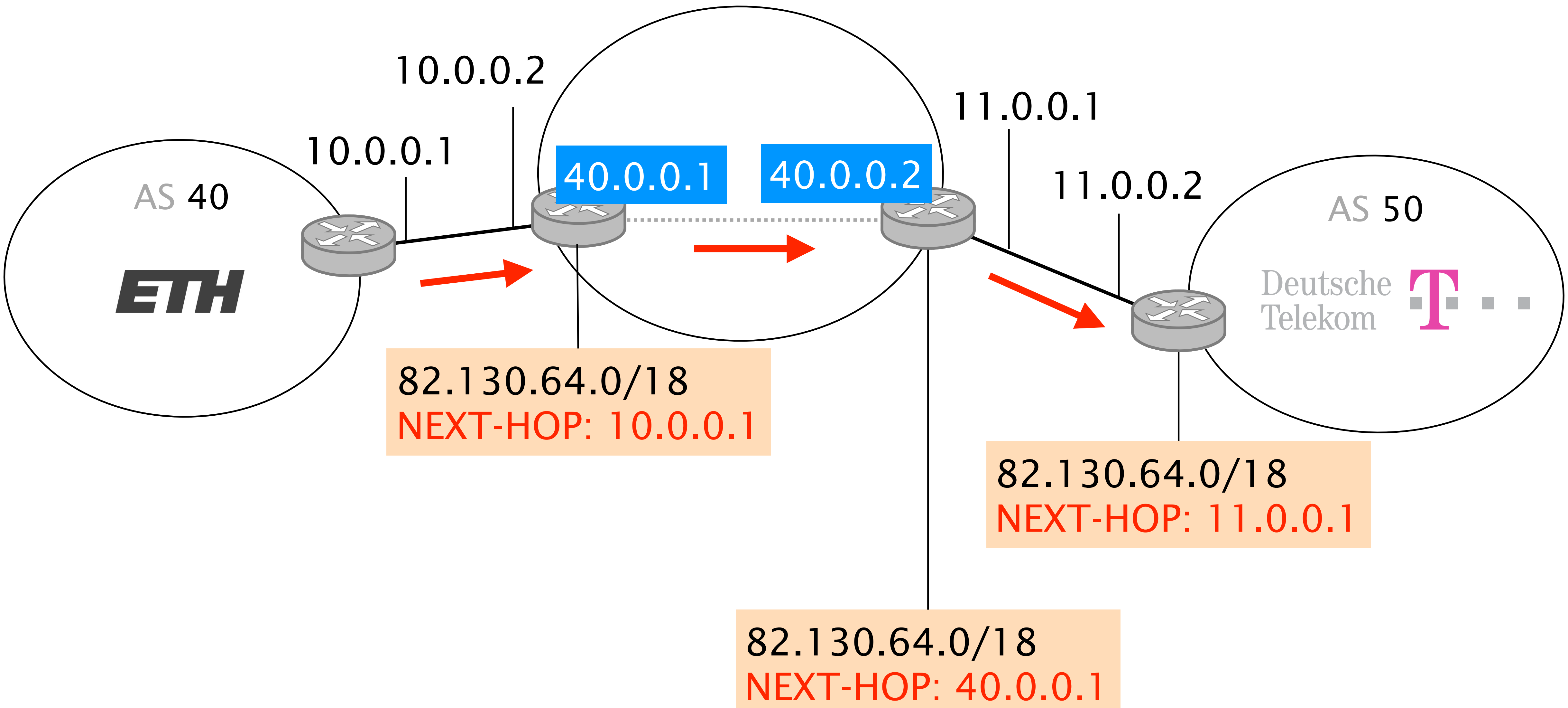
That router is *not* directly to the NEXT-HOP subnet (10.0.0.0/30)
and does not know how to reach it, it will therefore drop the BGP route…



One solution is for the external router to redistribute the prefixes attached to the external interfaces into the IGP

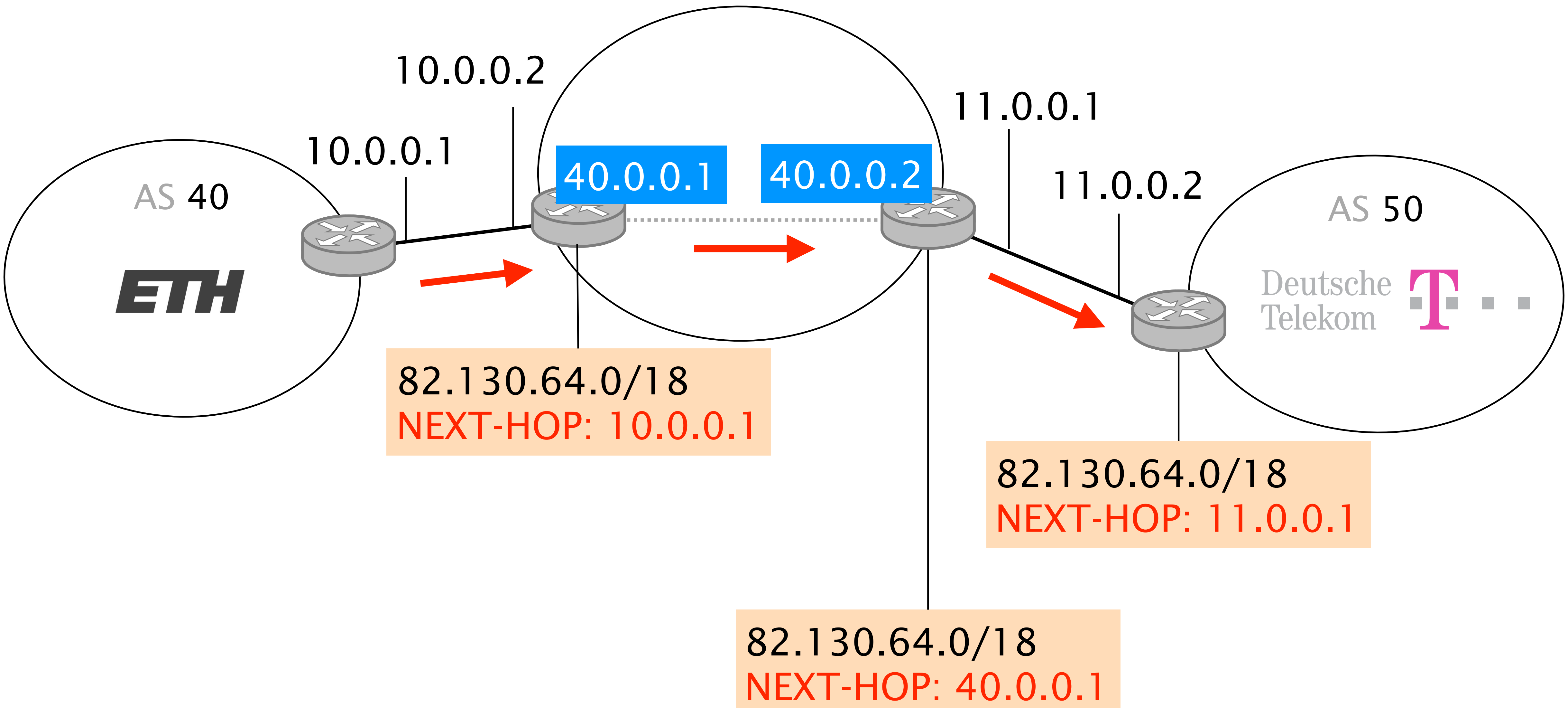


Another solution is for the border router to rewrite the NEXT-HOP before sending it over iBGP, usually to its loopback address

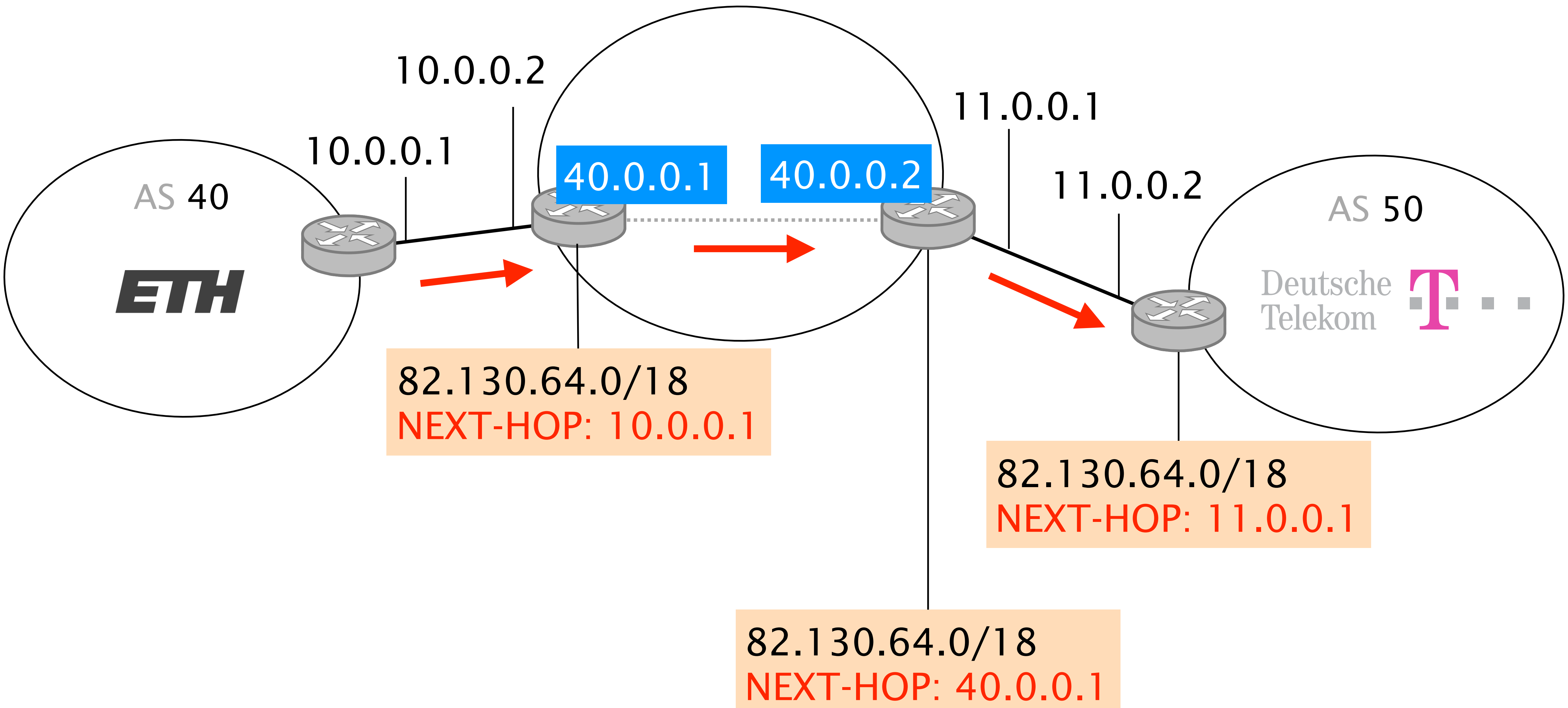


Of course, loopback address need to be reachable network-wide.

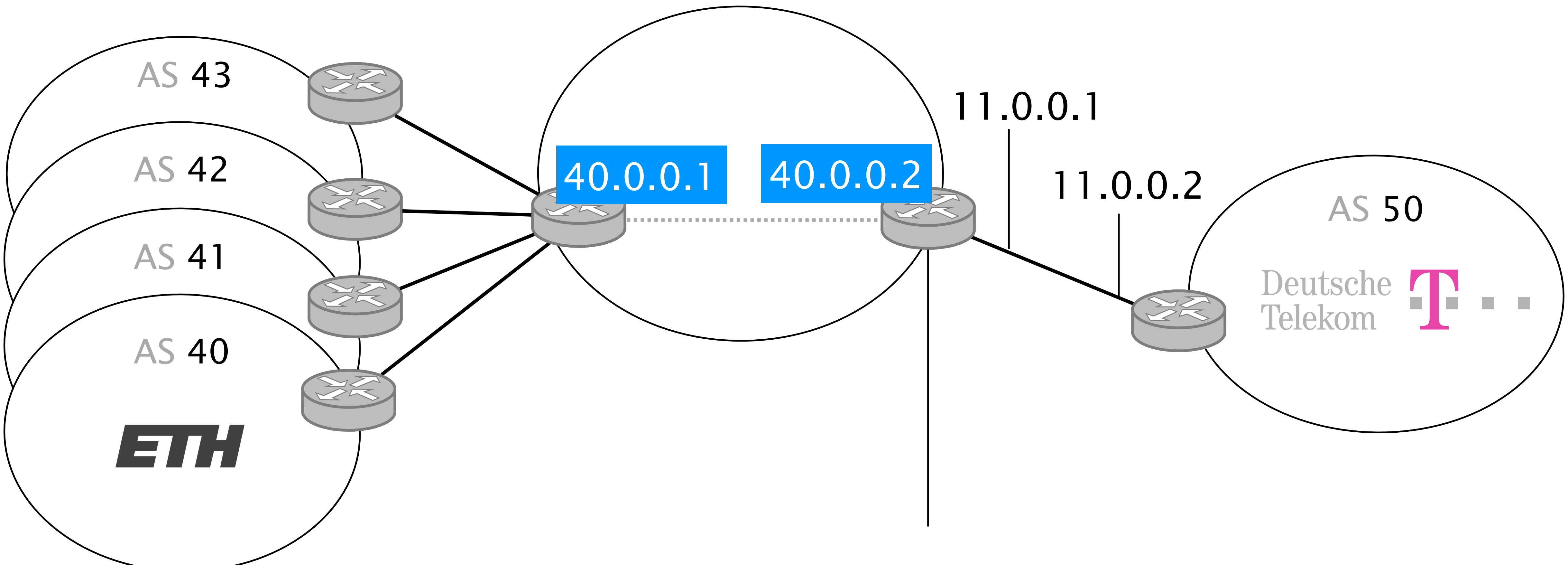
Typically, each router advertise its loopback (as a /32) in the IGP



This is the infamous **next-hop-self policy**



The advantage of next-hop self is to spare the need to advertise *each* prefix attached to an external link in the IGP



one NEXT-HOP, 40.0.0.1, is used
to reach routes announced by AS 40, 41, 42, 43

BGP suffers from many rampant problems

Problems

- Reachability
- Security
- Convergence
- Performance
- Anomalies
- Relevance

Problems	Reachability	covered last week
	Security	
	Convergence	
	Performance	
	Anomalies	
	Relevance	

Many security considerations are simply absent from BGP specifications

ASes can advertise any prefixes

even if they don’t own them!

ASes can arbitrarily modify route content

e.g., change the content of the AS-PATH

ASes can forward traffic along different paths

than the advertised one

BGP (lack of) security

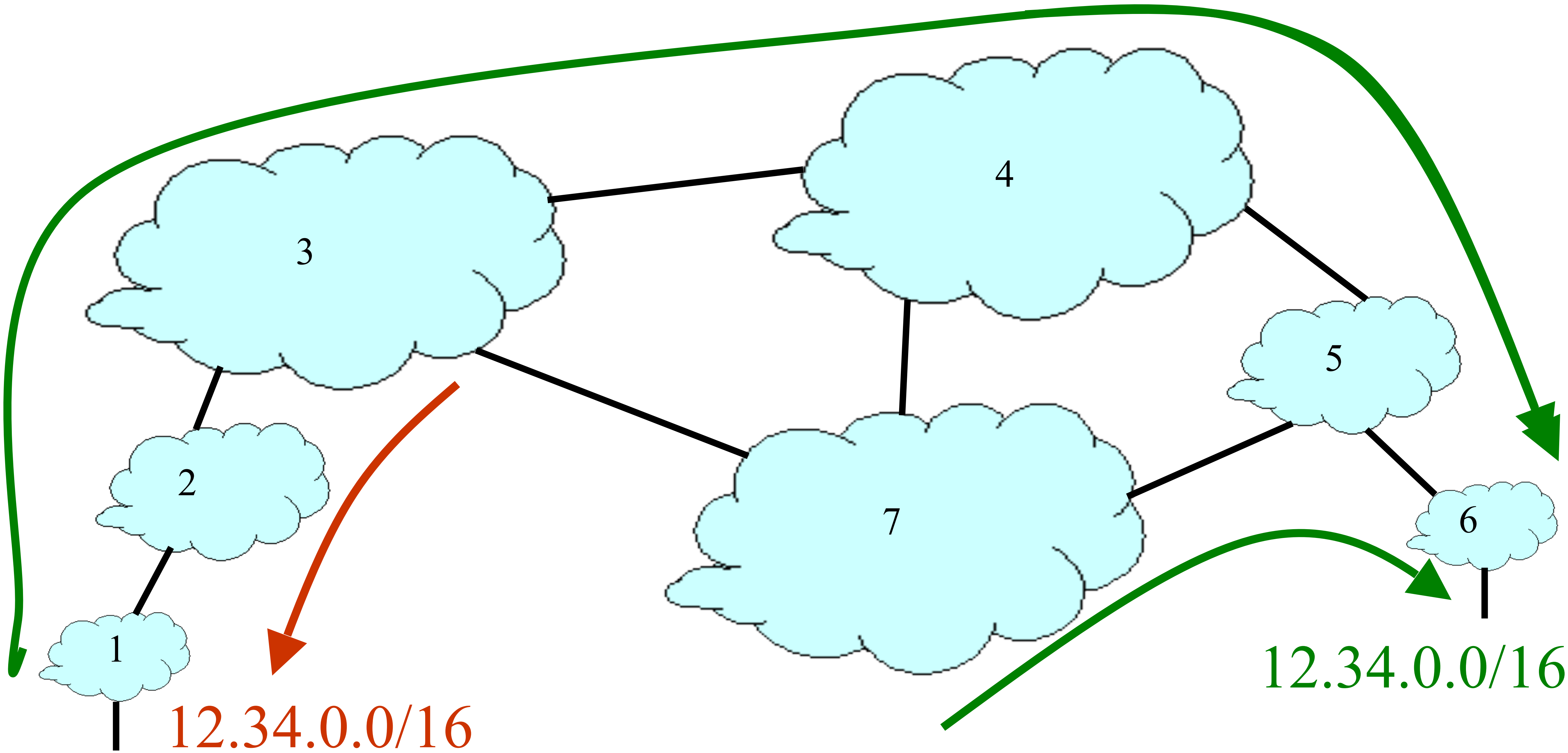
#1	BGP does not validate the origin of advertisements
#2	BGP does not validate the content of advertisements

BGP (lack of) security

#1 BGP does not validate the origin of advertisements

#2 BGP does not validate the content of advertisements

Prefix Hijacking

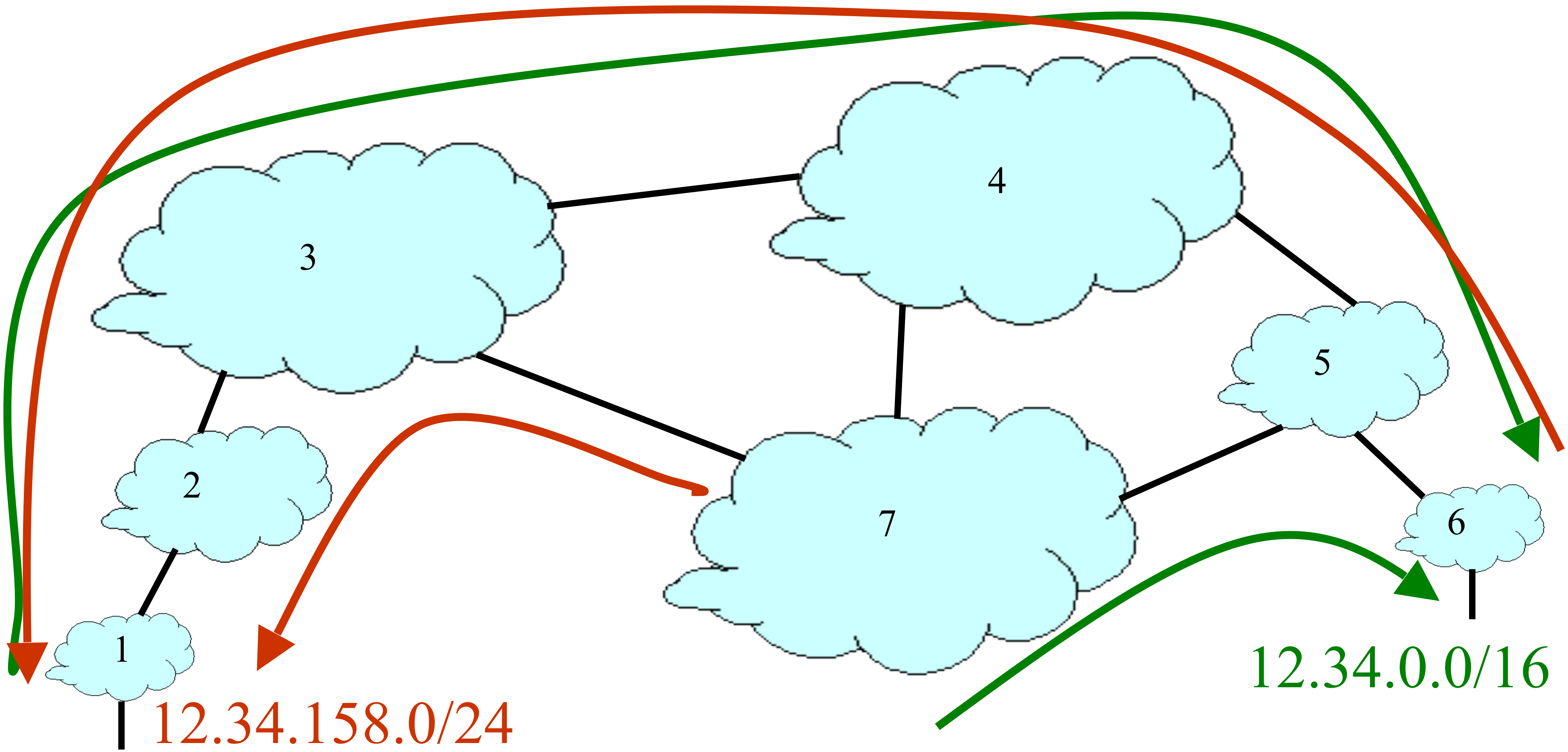


- **Blackhole:** data traffic is discarded
- **Snooping:** data traffic is inspected, then redirected
- **Impersonation:** traffic sent to bogus destinations

Hijacking is Hard to Debug

- The victim AS doesn't see the problem
  - Picks its own route, might not learn the bogus route
- May not cause loss of connectivity
  - Snooping, with minor performance degradation
- Or, loss of connectivity is isolated
  - E.g., only for sources in parts of the Internet
- Diagnosing prefix hijacking
  - Analyzing updates from many vantage points
  - Launching traceroute from many vantage points

Sub-Prefix Hijacking



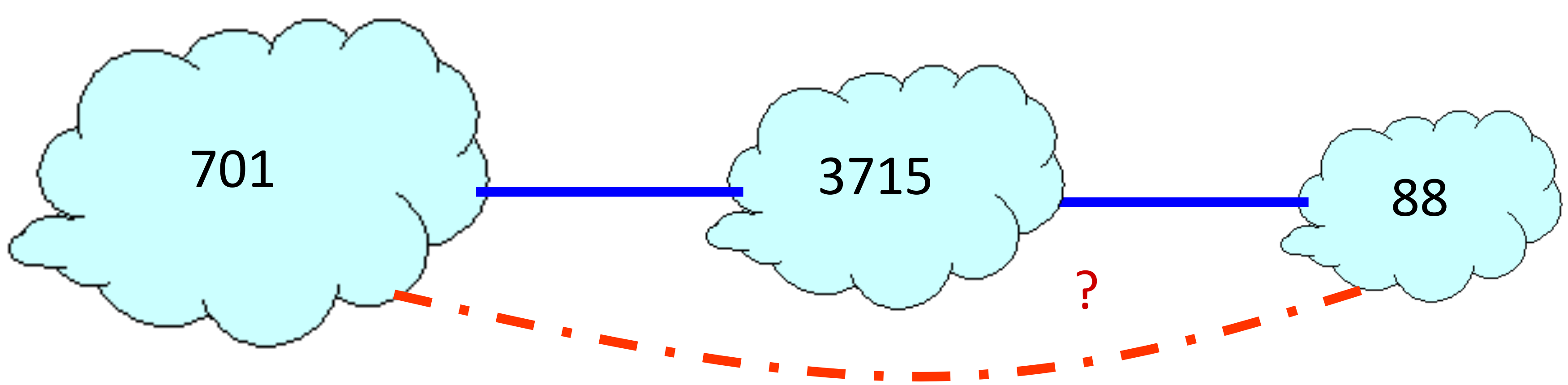
- Originating a more-specific prefix
  - Every AS picks the bogus route for that prefix
  - Traffic follows the longest matching prefix

BGP (lack of) security

#1	BGP does not validate the origin of advertisements
#2	BGP does not validate the content of advertisements

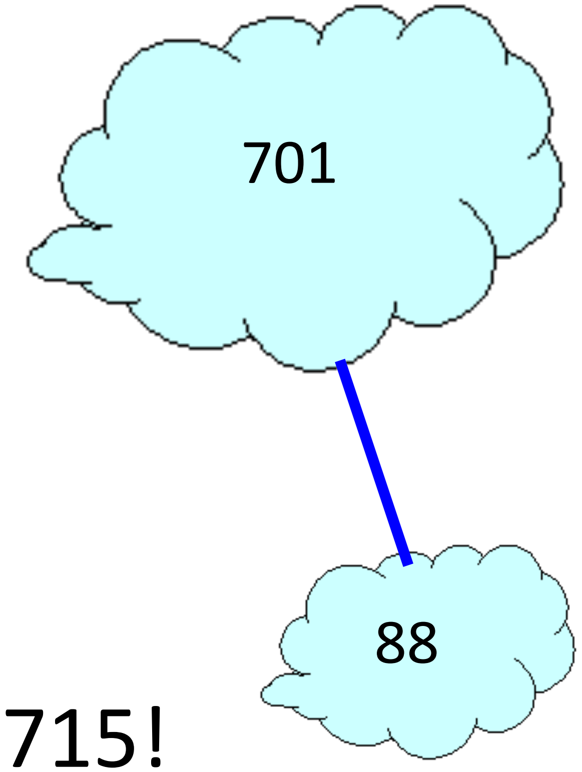
Bogus AS Paths

- Remove ASes from the AS path
  - E.g., turn "701 3715 88" into "701 88"
- Motivations
  - Attract sources that normally try to avoid AS 3715
  - Help AS 88 look like it is closer to the Internet's core
- Who can tell that this AS path is a lie?
  - Maybe AS 88 *does* connect to AS 701 directly



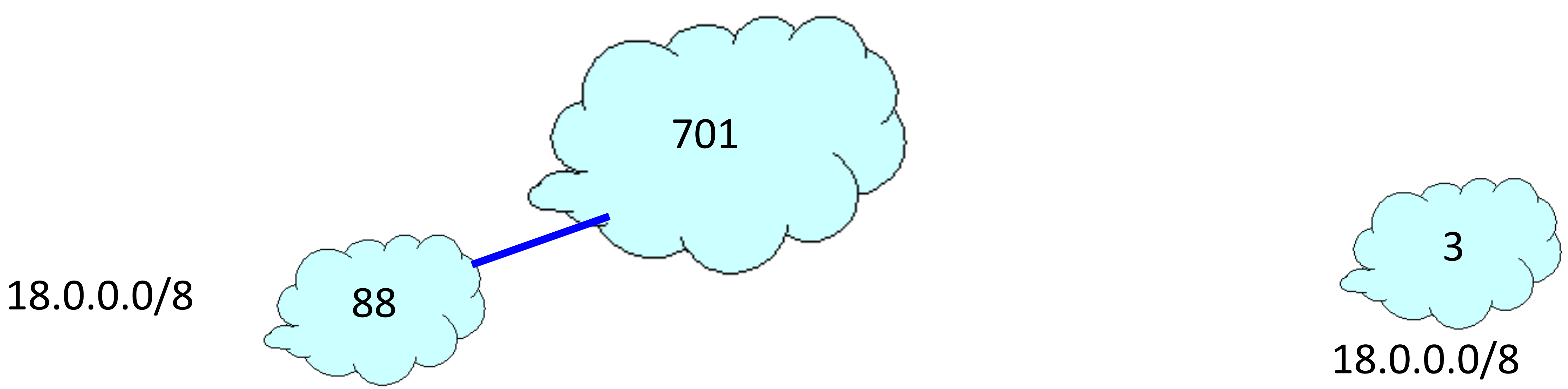
Bogus AS Paths

- Add ASes to the path
  - E.g., turn “701 88” into “701 3715 88”
- Motivations
  - Trigger loop detection in AS 3715
    - Denial-of-service attack on AS 3715
    - Or, blocking unwanted traffic coming from AS 3715!
  - Make your AS look like is has richer connectivity
- Who can tell the AS path is a lie?
  - AS 3715 could, if it could see the route
  - AS 88 could, but would it really care?



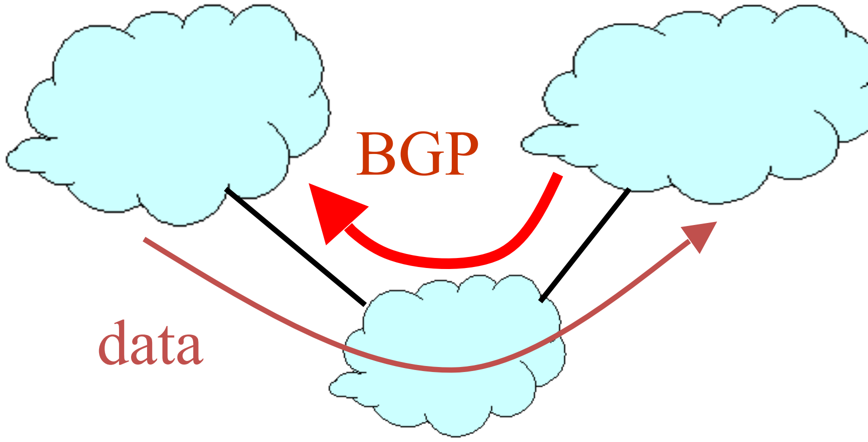
Bogus AS Paths

- Adds AS hop(s) at the end of the path
  – E.g., turns “701 88” into “701 88 3”
- Motivations
  – Evade detection for a bogus route
  – E.g., by adding the legitimate AS to the end
- Hard to tell that the AS path is bogus…
  – Even if other ASes filter based on prefix ownership



Invalid Paths

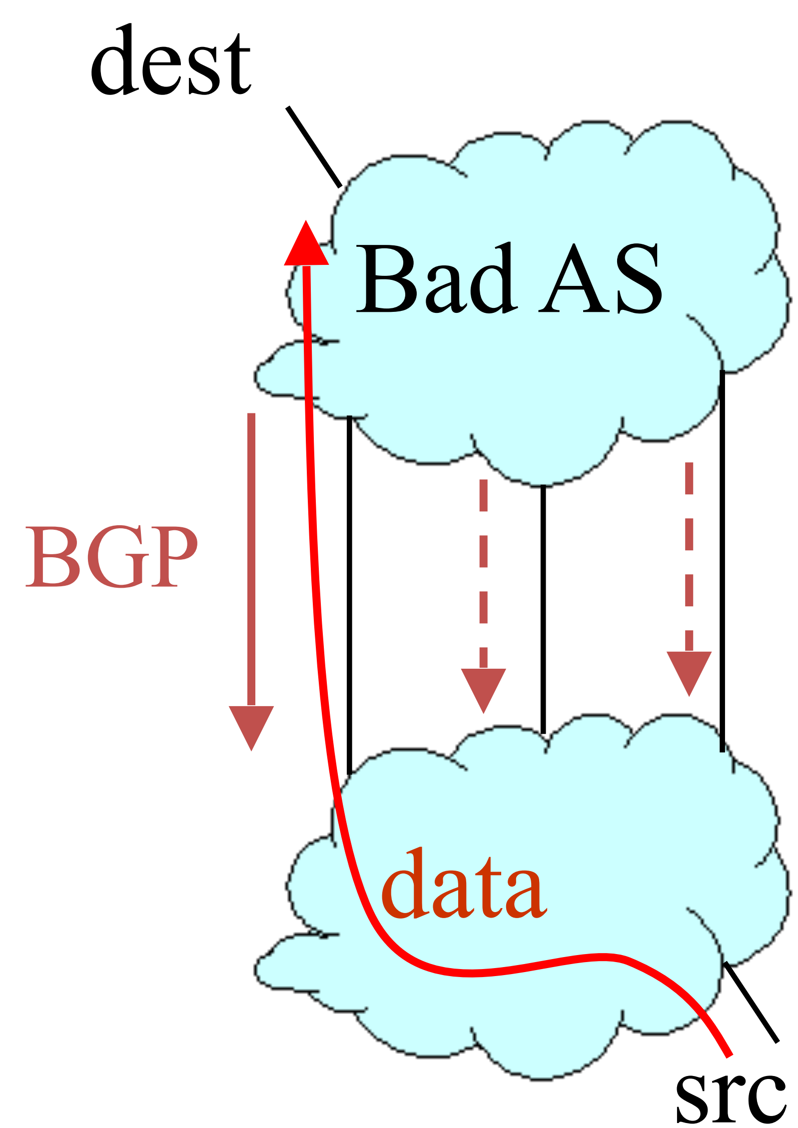
- AS exports a route it shouldn't
  – AS path is a valid sequence, but violated policy
- Example: customer misconfiguration
  – Exports routes from one provider to another
- Interacts with provider policy
  – Provider prefers customer routes
  – Directing all traffic through customer



- Main defense
  – Filtering routes based on prefixes and AS path

Missing/Inconsistent Routes

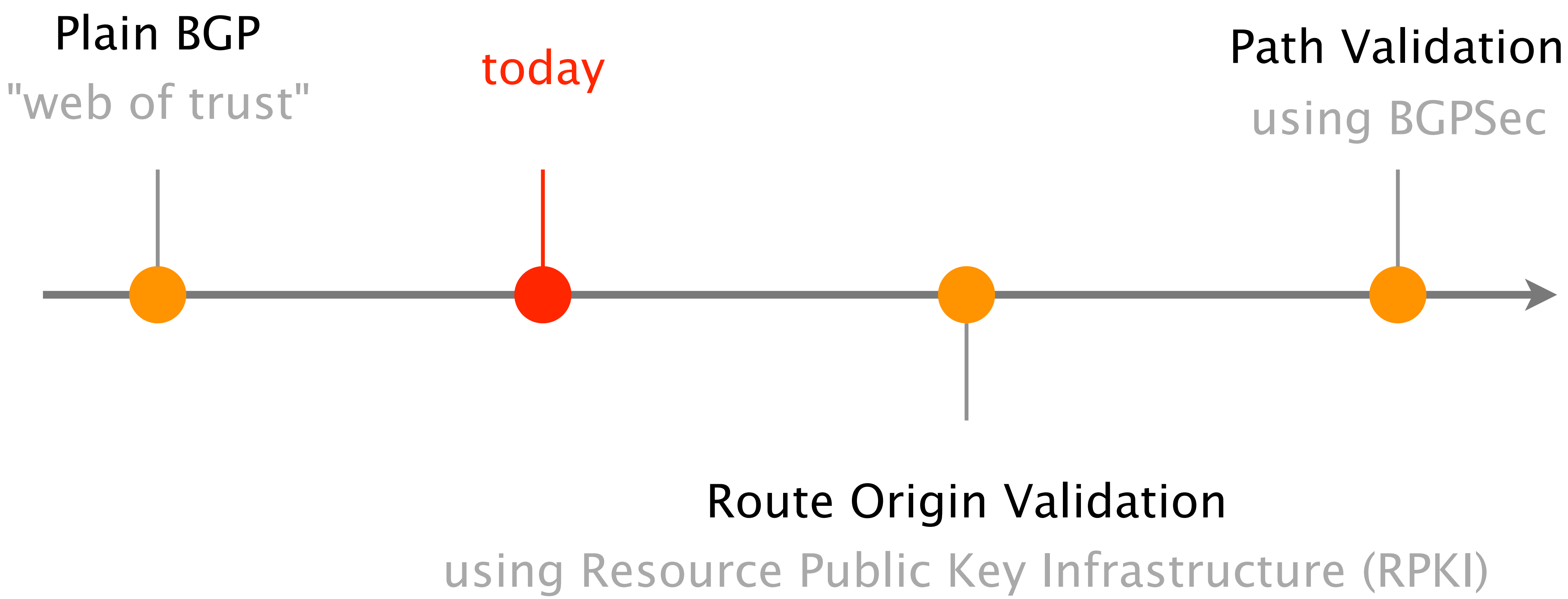
- Peers require consistent export
  - Prefix advertised at all peering points
  - Prefix advertised with same AS path length
- Reasons for violating the policy
  - Trick neighbor into "cold potato"
  - Configuration mistake
- Main defense
  - Analyzing BGP updates, or traffic,
  - … for signs of inconsistency

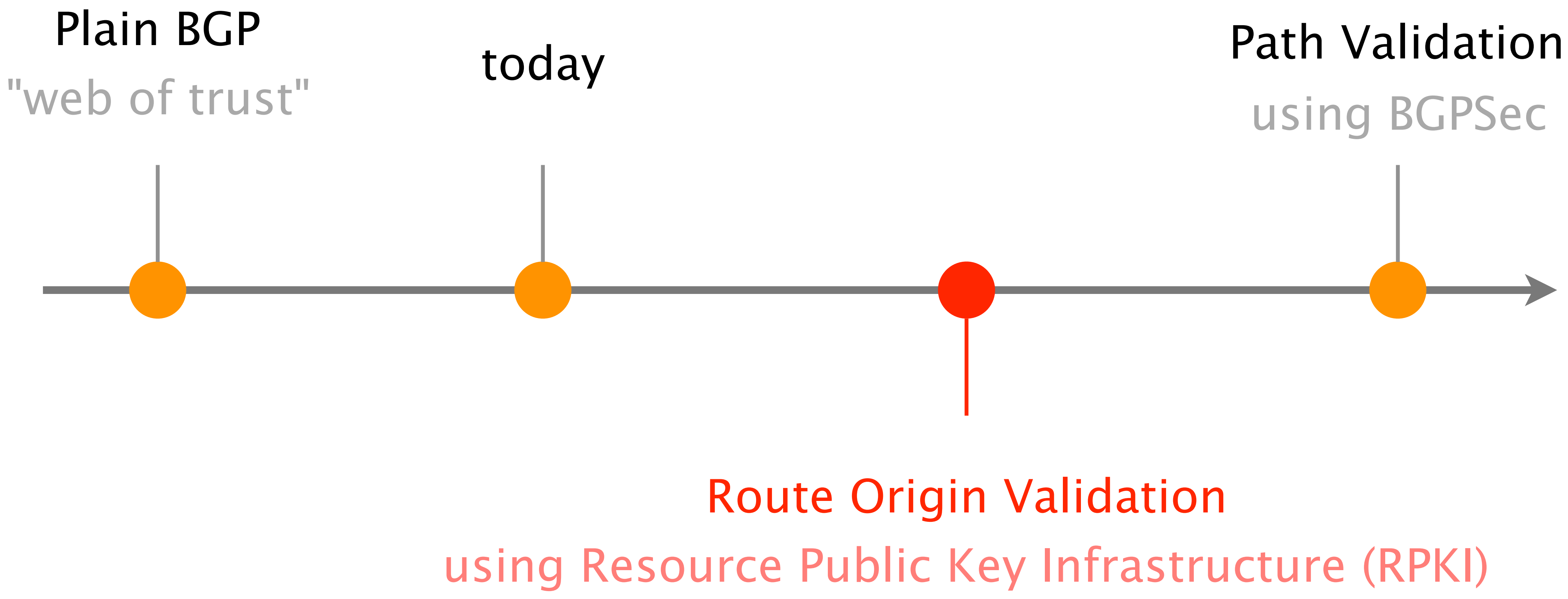


BGP Security ~~Today~~ Yesterday

- Applying best common practices (BCPs)
  - Securing the session (authentication, encryption)
  - Filtering routes by prefix and AS path
  - Packet filters to block unexpected control traffic
- This is not good enough
  - Depends on vigilant application of BCPs
  - Doesn't address fundamental problems
    - Can't tell who owns the IP address block
    - Can't tell if the AS path is bogus or invalid
    - Can't be sure the data packets follow the chosen route

BGP today is *slowly* becoming more secure thanks to cryptography




Plain BGP
"web of trust"
today
Path Validation
using BGPSec
Route Origin Validation
using Resource Public Key Infrastructure (RPKI)

RPKI enables to validate the origin of a BGP route by certifying IP prefixes allocations

RPKI is a database storing Route Origin Authorization

ROAs map prefix space (130.0.0.0/8–32) to an origin AS

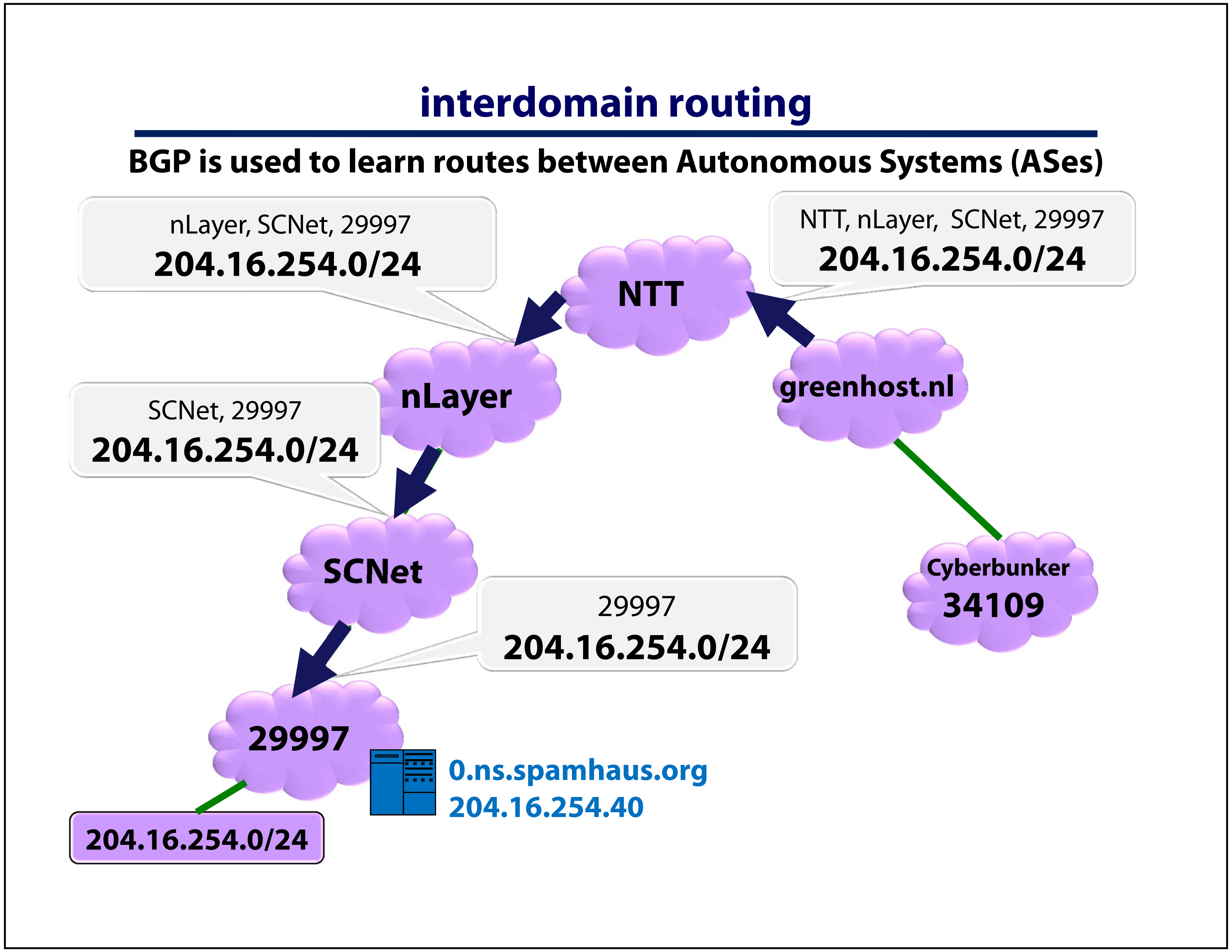
Routers consult this database to verify BGP messages

BGP messages are *not* changed, RPKI works "out-of-band"

RPKI has been standardized in 2012 (RFC 6480)

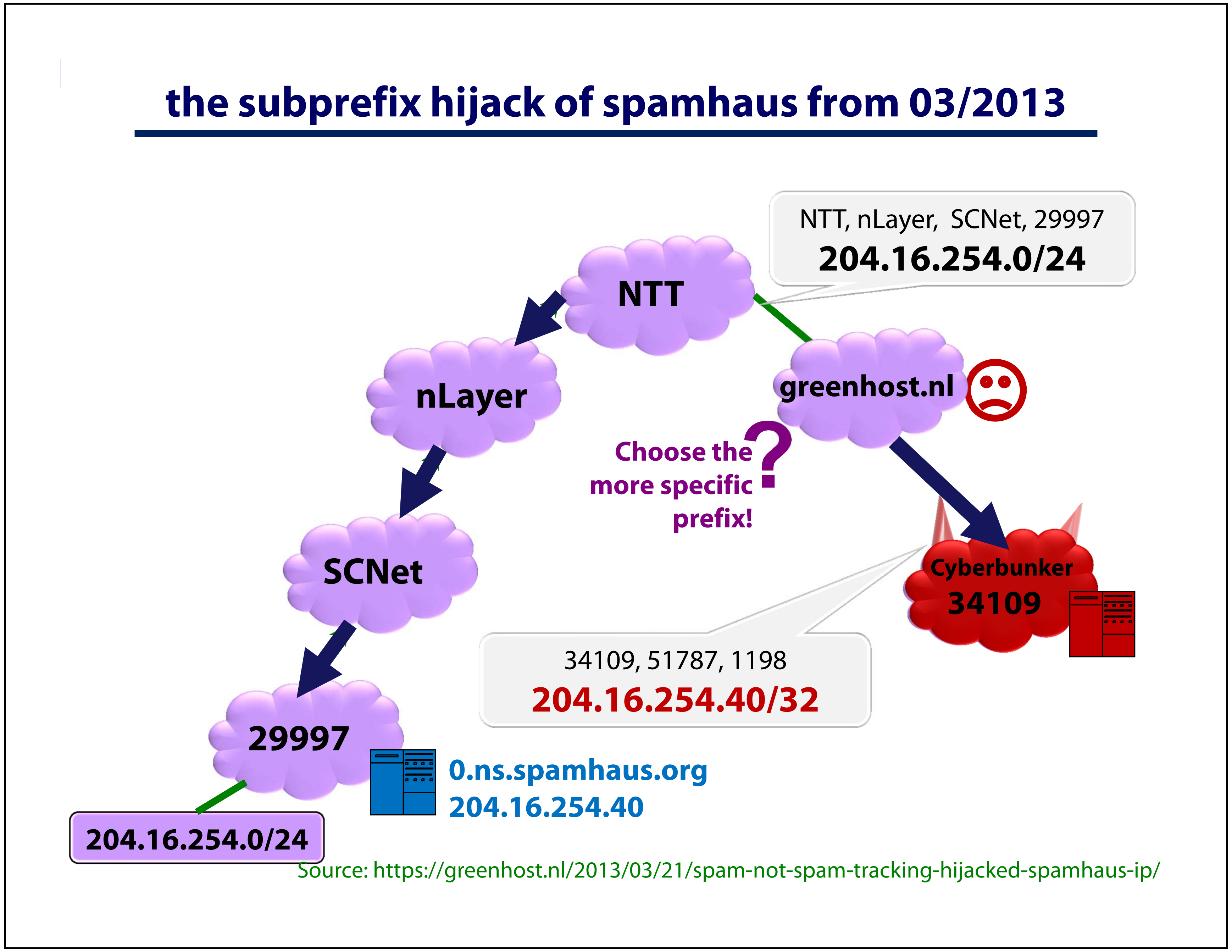
today, RPKI can validate ~19% of the IPv4 prefixes

Let's look back at an example, first without RPKI



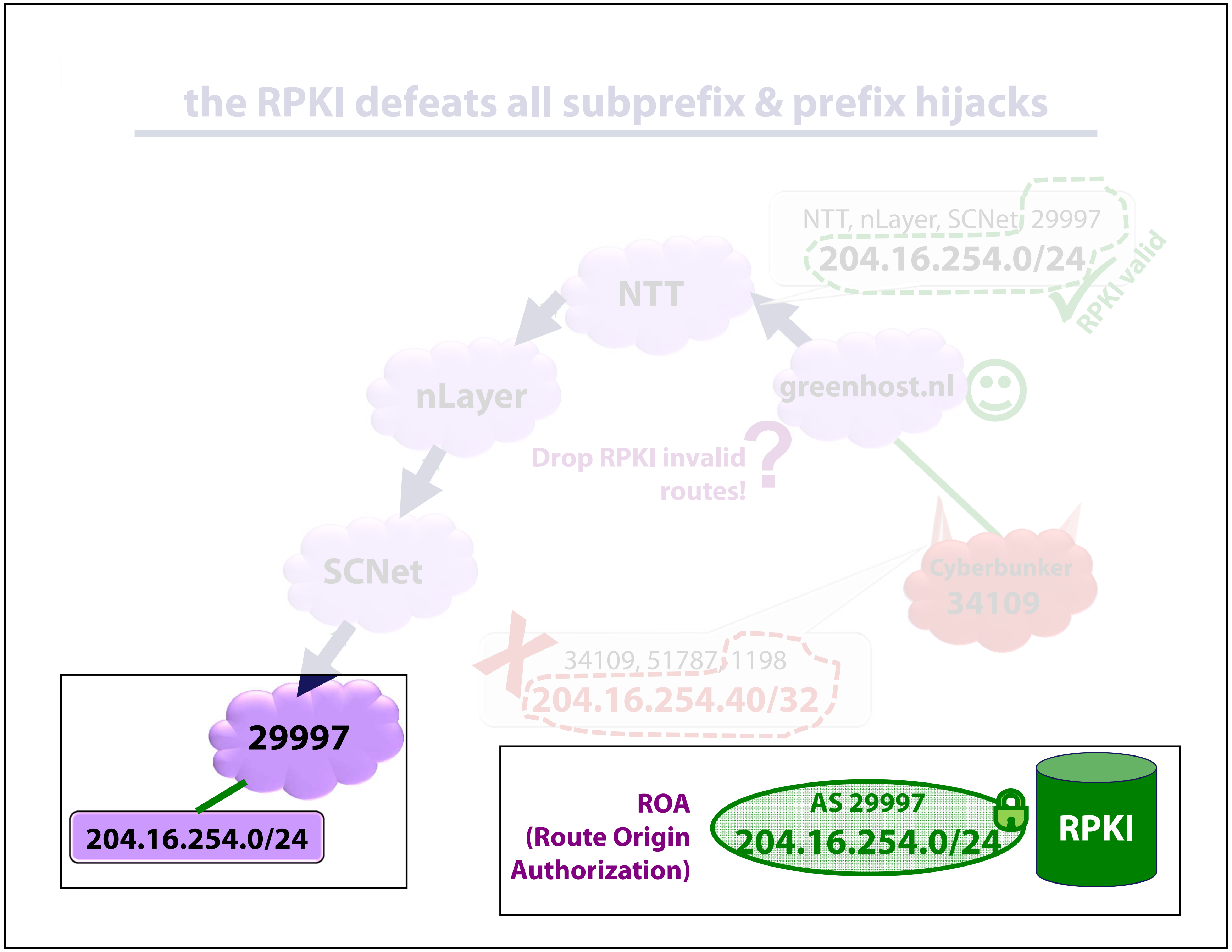
Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

Here, we see that the attack is successful



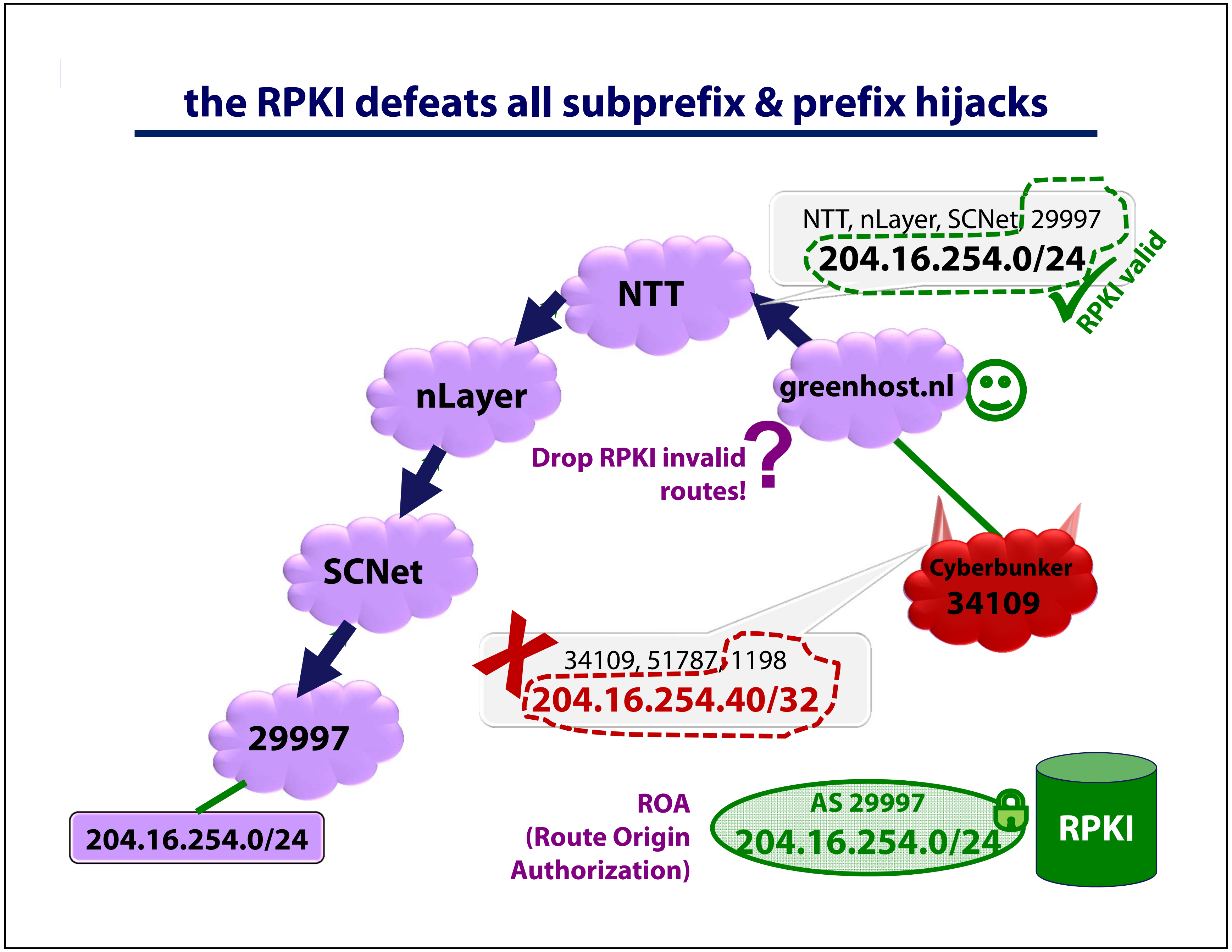
Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

Let's assume now that AS 29997 registers (204.16.254.0/24-32, 29997) as a new ROA



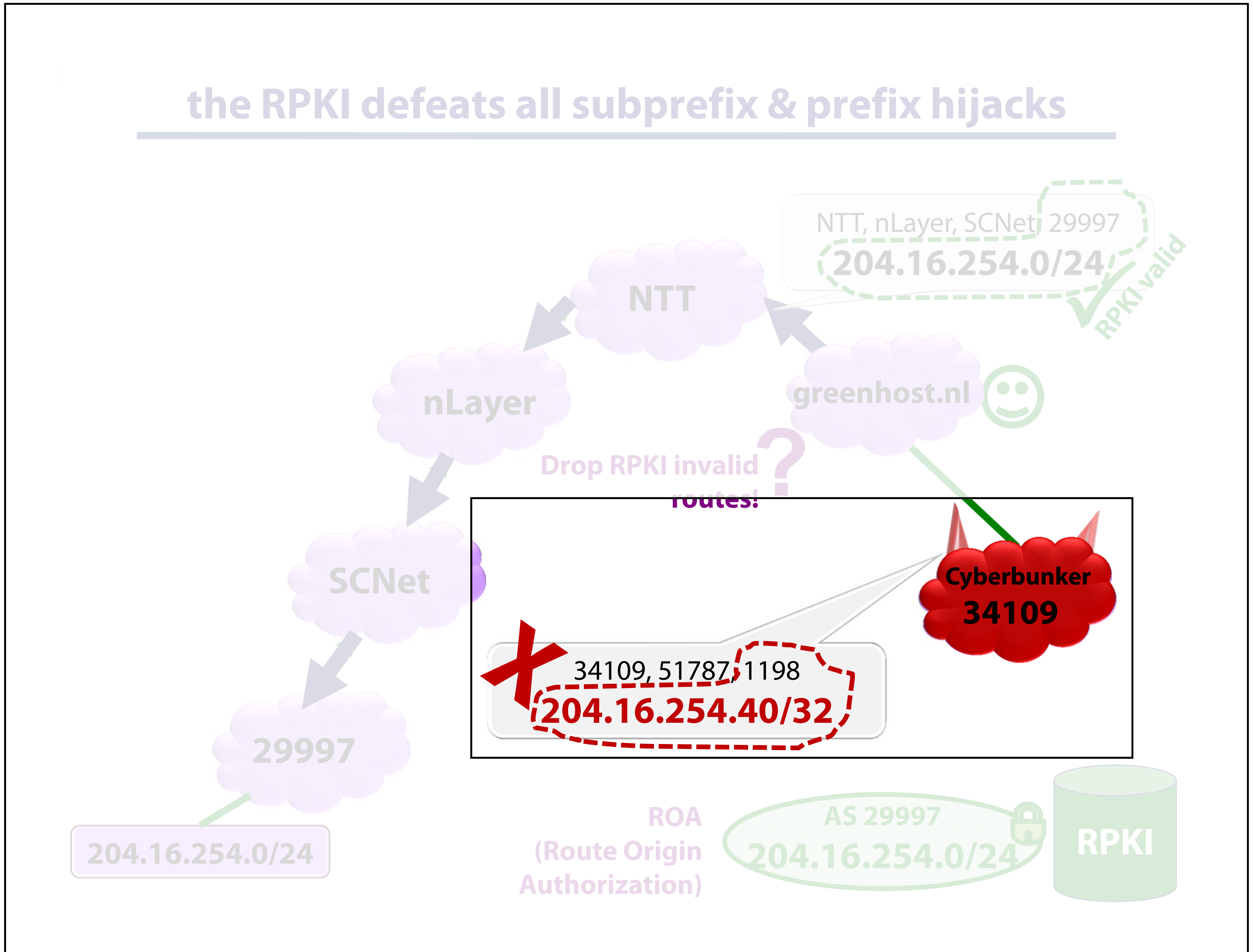
Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

Using the RPKI, greenhost.nl sees that AS34109 is *not* a valid origin for 204.16.254.40/32



Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

This announcement is said to be *INVALID*



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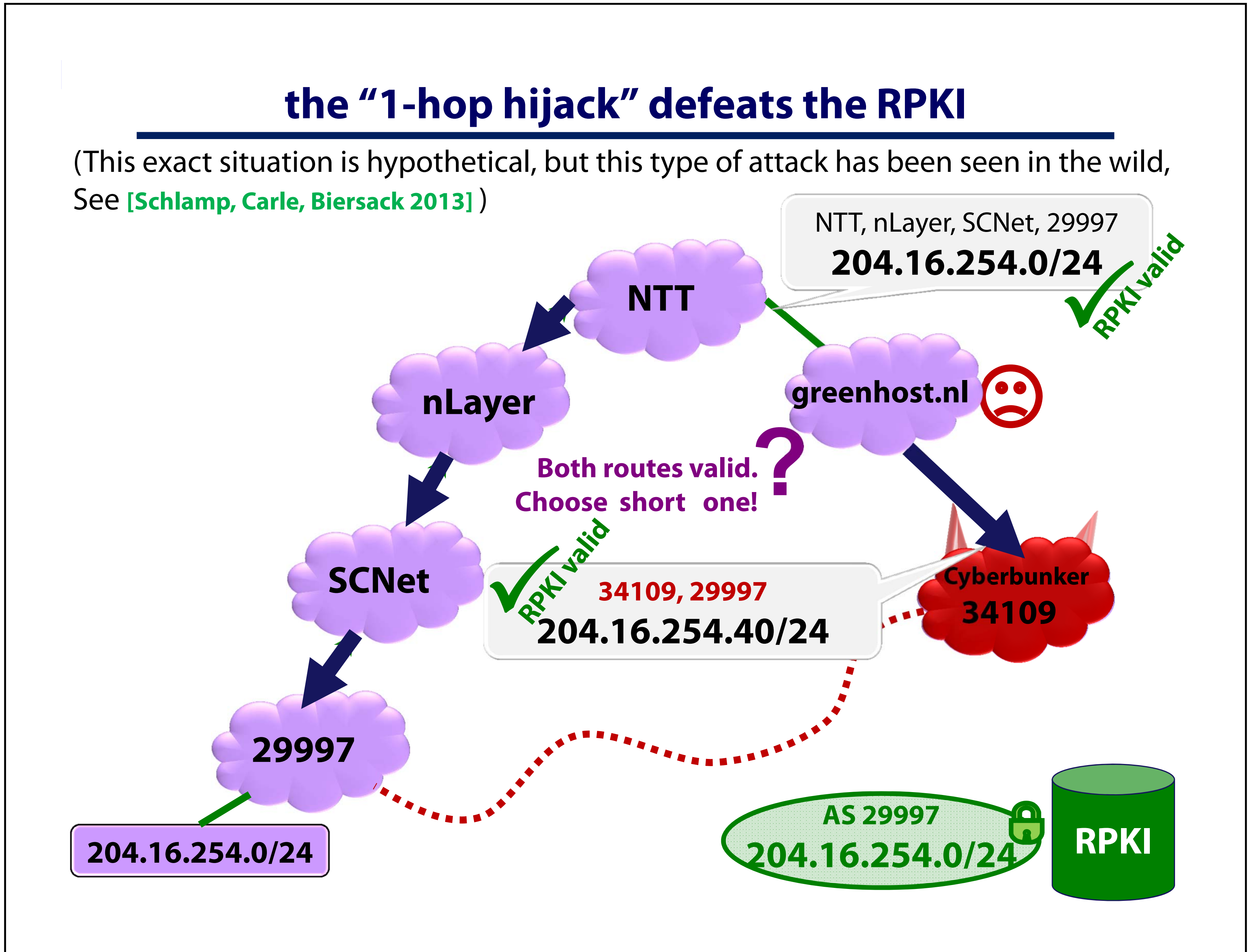
Now what if AS34109 announce AS29997 as the origin?



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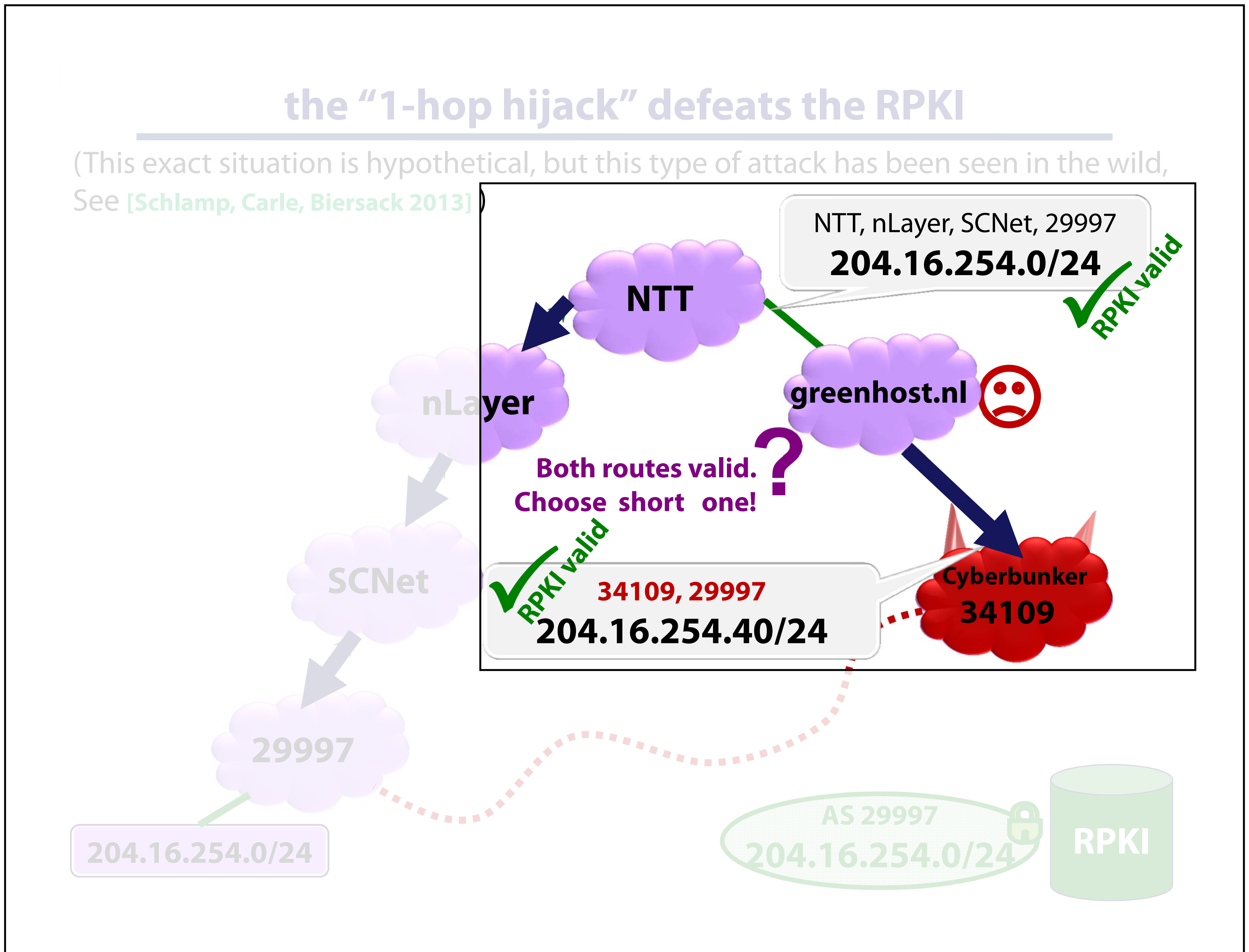
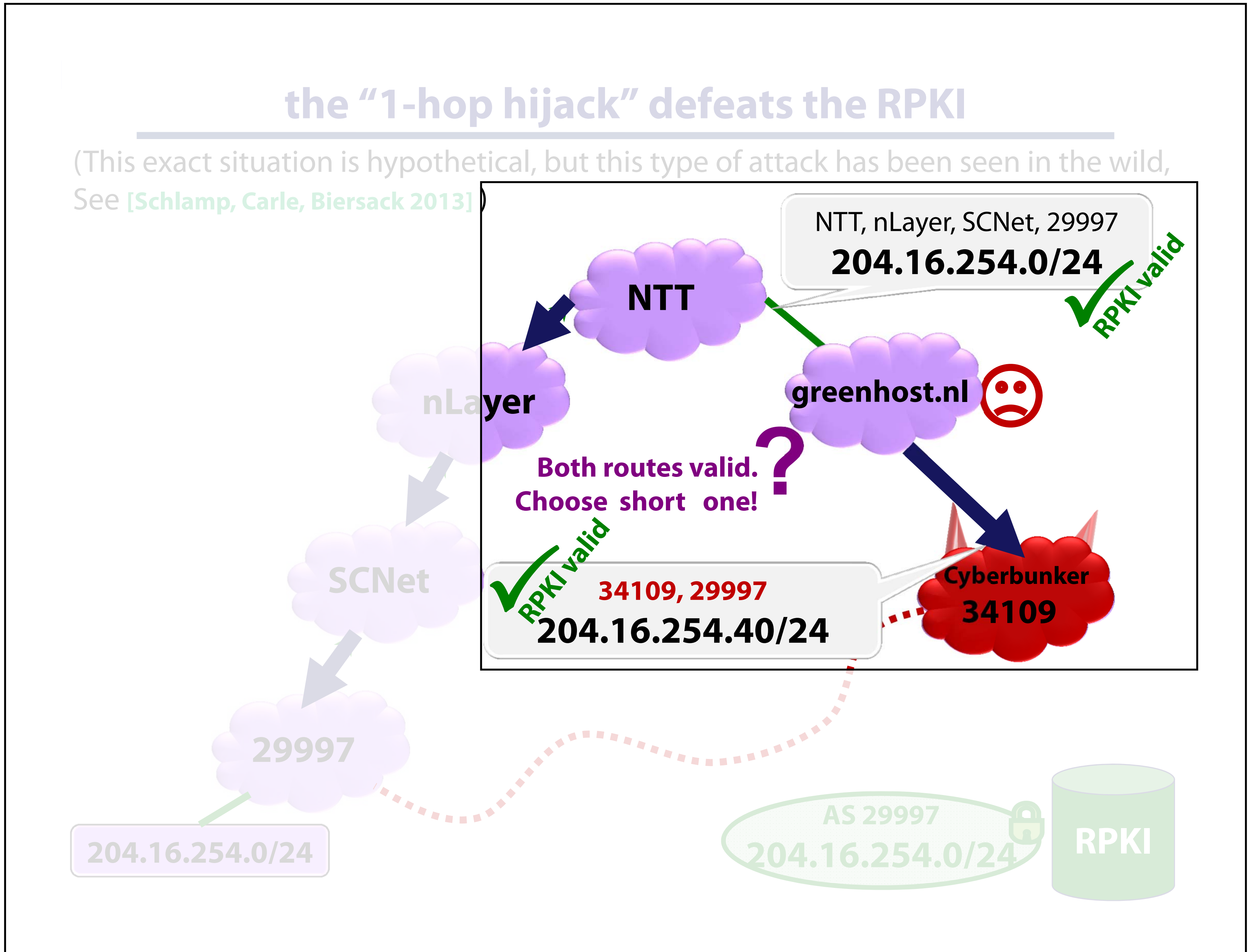
Here greenhost.nl receives 2 valid RPKI routes: one via NTT and another one via 34109



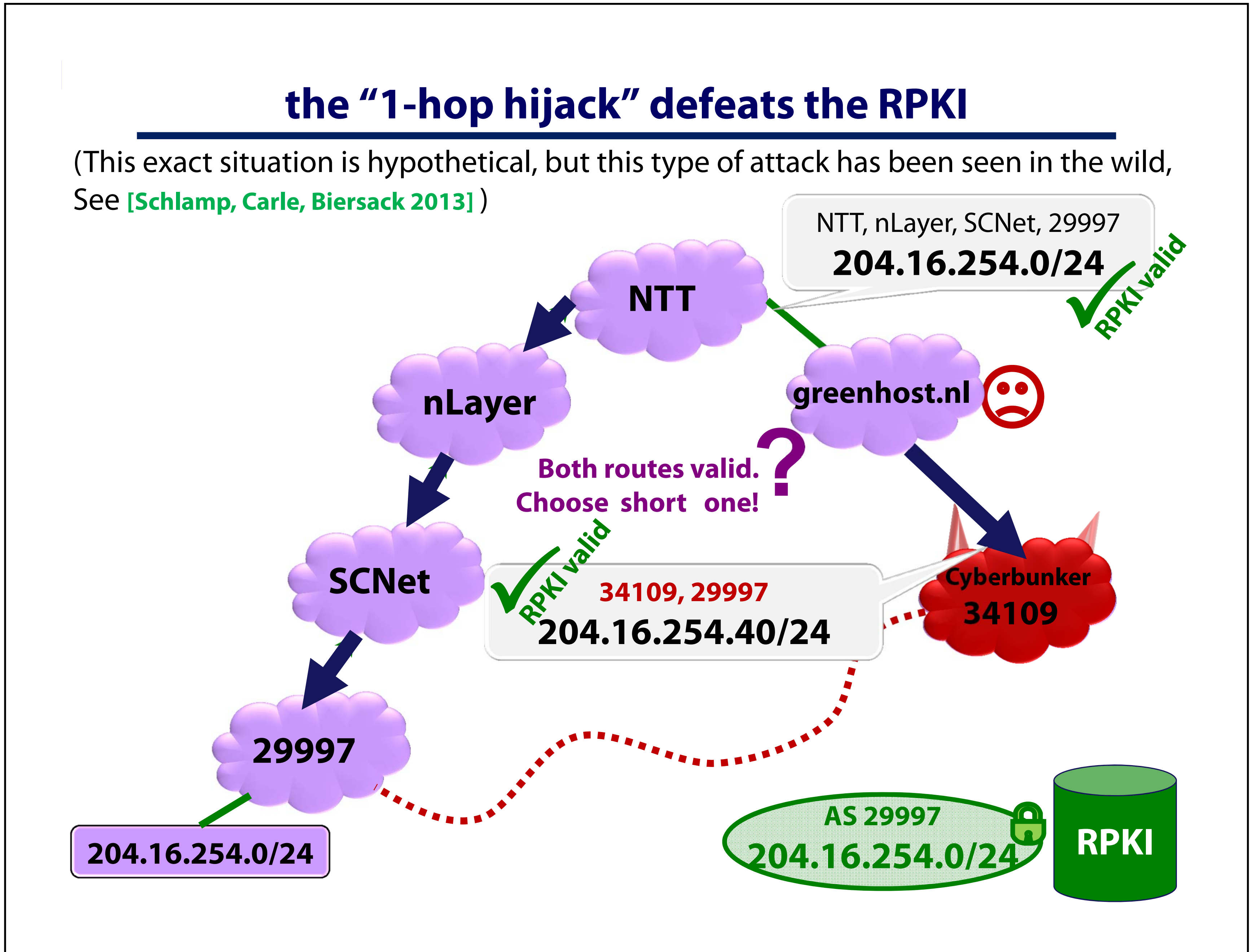
Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

As the route via 34109 has a shorter path, it is preferred… the attack works again!



Source: Sharon Goldberg, "The Transition to BGP Security. Is the Juice Worth the Squeeze?"

We see that RPKI does not protect against *all* attacks



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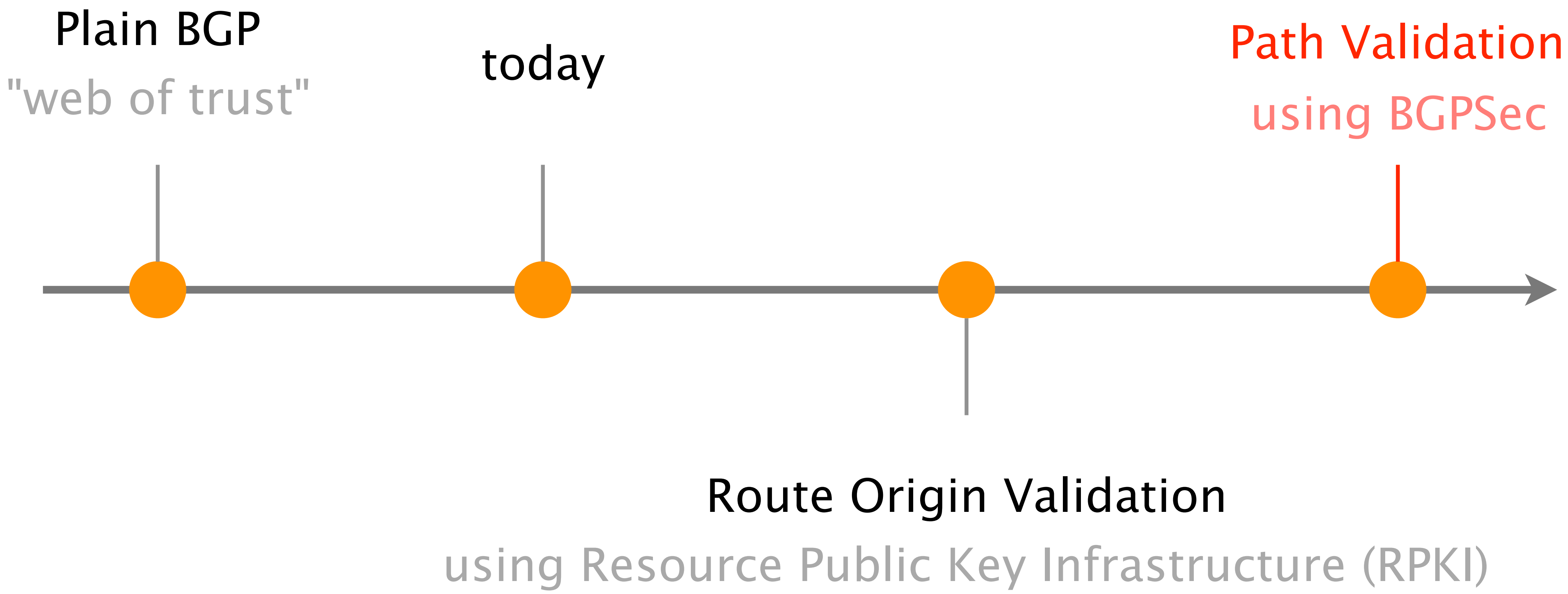
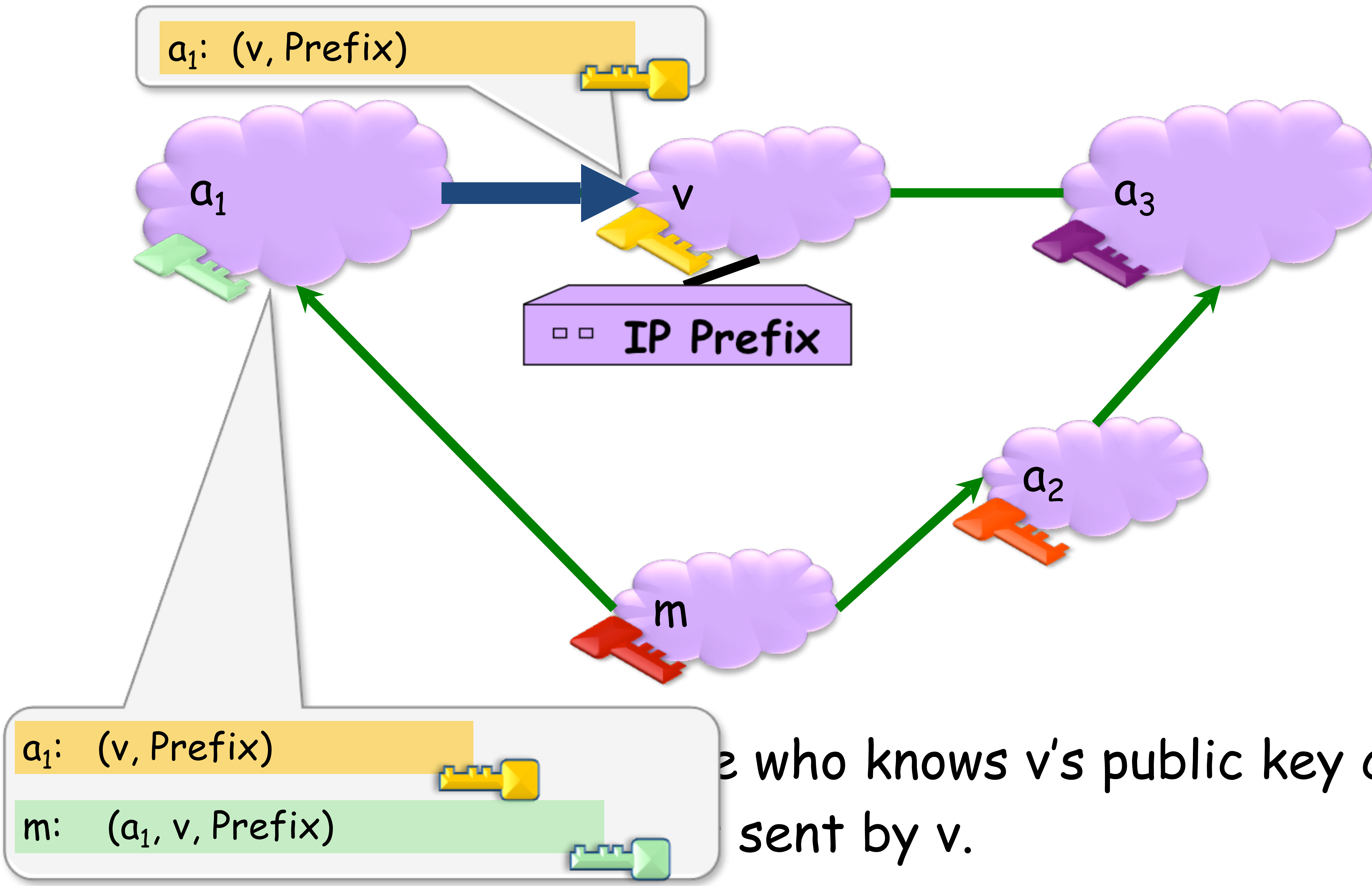

Plain BGP
"web of trust"
today
Path Validation
using BGPSec
Route Origin Validation
using Resource Public Key Infrastructure (RPKI)

Secure BGP

Origin + Path Authentication using cryptographic signatures



e who knows v's public key can

sent by v.

S-BGP Secure Version of BGP

- Address attestations
  - Claim the right to originate a prefix
  - Signed and distributed out-of-band
  - Checked through delegation chain from ICANN
- Route attestations
  - Distributed as an attribute in BGP update message
  - Signed by each AS as route traverses the network
- S-BGP can validate
  - AS path indicates the order ASes were traversed
  - No intermediate ASes were added or removed

S-BGP Deployment Challenges

- Complete, accurate registries of prefix “owner”
- Public Key Infrastructure
  - To know the public key for any given AS
- Cryptographic operations
  - E.g., digital signatures on BGP messages
- Need to perform operations quickly
  - To avoid delaying response to routing changes
- Difficulty of incremental deployment
  - Hard to have a “flag day” to deploy S-BGP

Problems

Reachability

Security

Convergence switch back to last week's slides

Performance

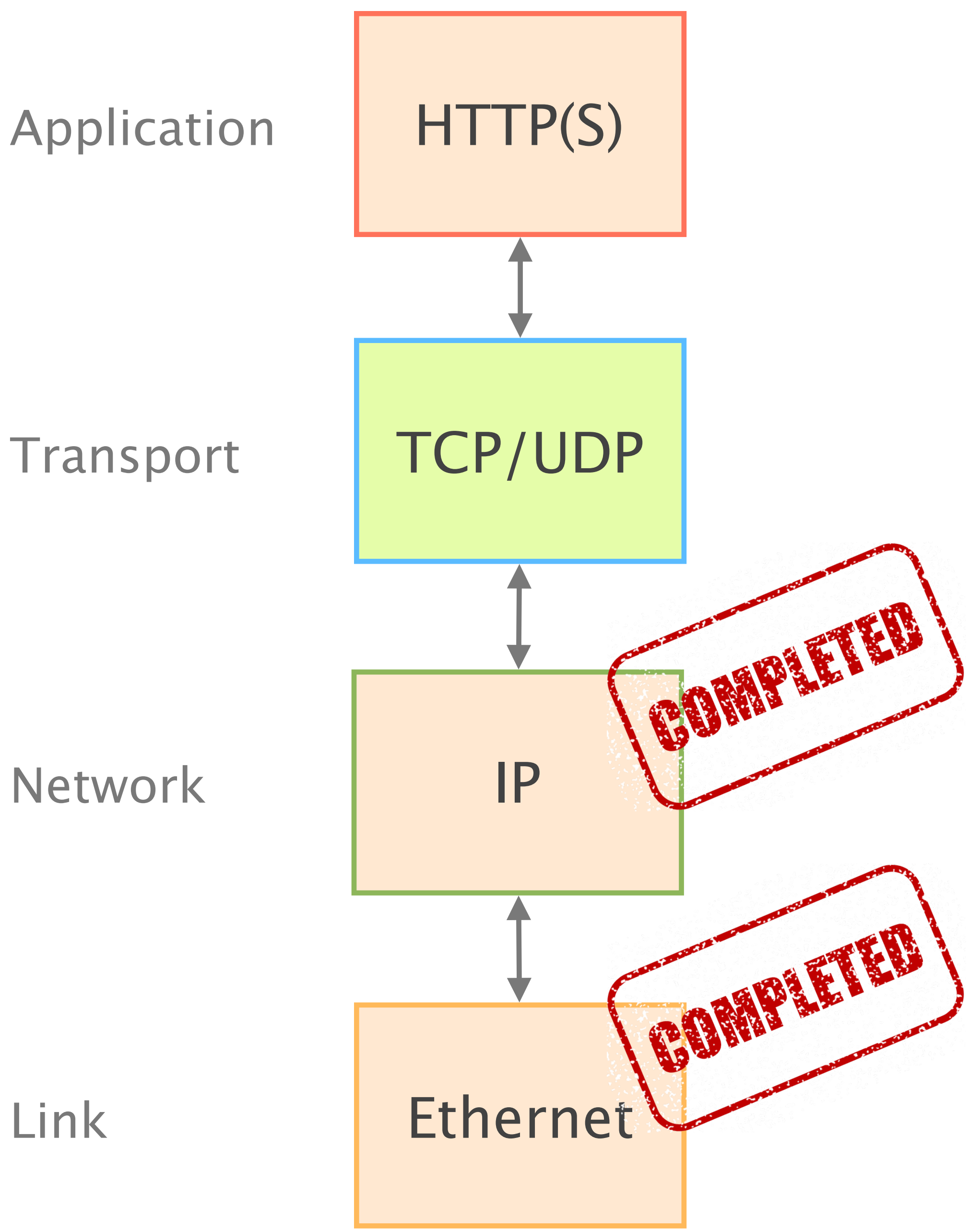
Anomalies

Relevance

That's it!

for the network layer, and for now…

We're continuing our journey up the layers, now looking at the transport layer



What do we need in the Transport layer?

Functionality implemented in network

- Keep minimal (easy to build, broadly applicable)

Functionality implemented in the application

- Keep minimal (easy to write)
- Restricted to application-specific functionality

Functionality implemented in the "network stack"

- The shared networking code on the host
- This relieves burden from both app and network
- The transport layer is a key component here

What do we need in the Transport layer?

Application layer

- Communication for specific applications
- *e.g.*, HyperText Transfer Protocol (HTTP),
  File Transfer Protocol (FTP)

Network layer

- Global communication between hosts
- Hides details of the link technology
- *e.g.*, Internet Protocol (IP)

What Problems Should Be Solved Here?

Data delivering, to the *correct* application

- IP just points towards next protocol
- *Transport needs to demultiplex incoming data (ports)*

Files or bytestreams abstractions for the applications

- Network deals with packets
- *Transport layer needs to translate between them*

Reliable transfer (if needed)

Not overloading the receiver

Not overloading the network

What Is Needed to Address These?

Demultiplexing: identifier for application process

- Going from host-to-host (IP) to process-to-process

Translating between bytestreams and packets:

- Do segmentation and reassembly

Reliability: ACKs and all that stuff

Corruption: Checksum

Not overloading receiver: “Flow Control”

- Limit data in receiver’s buffer

Not overloading network: “Congestion Control”

UDP: Datagram messaging service

UDP provides a connectionless, unreliable transport service

- No-frills extension of “best-effort” IP
- UDP provides only two services to the App layer
  - Multiplexing/Demultiplexing among processes
  - Discarding corrupted packets (optional)

TCP: Reliable, in-order delivery

TCP provides a connection-oriented, reliable, bytestream transport service

What UDP provides, plus:

- Retransmission of lost and corrupted packets
- Flow control (to not overflow receiver)
- Congestion control (to not overload network)
- “Connection” set-up & tear-down

Connections (or sessions)

Reliability requires keeping state

- Sender: packets sent but not ACKed, and related timers
- Receiver: noncontiguous packets

Each bytestream is called a connection or session

- Each with their own connection state
- State is in hosts, not network!

What transport protocols do not provide

Delay and/or bandwidth guarantees

- This cannot be offered by transport
- Requires support at IP level (*and let's not go there*)

Sessions that survive change-of-IP-address

- This is an artifact of current implementations
- As we shall see….

Important Context: Sockets and Ports

Sockets: an operating system abstraction

Ports: a networking abstraction

- This is not a port on a switch (which is an interface)
- Think of it as a *logical interface* on a host

Sockets

A socket is a software abstraction by which an application process exchanges network messages with the (transport layer in the) operating system

- socketID = socket(…, socket.TYPE)
- socketID.sendto(message, …)
- socketID.recvfrom(…)

Two important types of sockets

- UDP socket: TYPE is SOCK_DGRAM
- TCP socket: TYPE is SOCK_STREAM

Ports

Problem: which app (socket) gets which packets

Solution: port as transport layer identifier (16 bits)

- Packet carries source/destination port numbers in transport header

OS stores mapping between sockets and ports

- Port: in packets
- Socket: in OS

More on Ports

Separate 16-bit port address space for UDP, TCP

“Well known” ports (0-1023)

- Agreement on which services run on these ports
- *e.g.*, ssh:22, http:80
- Client (app) knows appropriate port on server
- Services can listen on well-known port

Ephemeral ports (most 1024-65535):

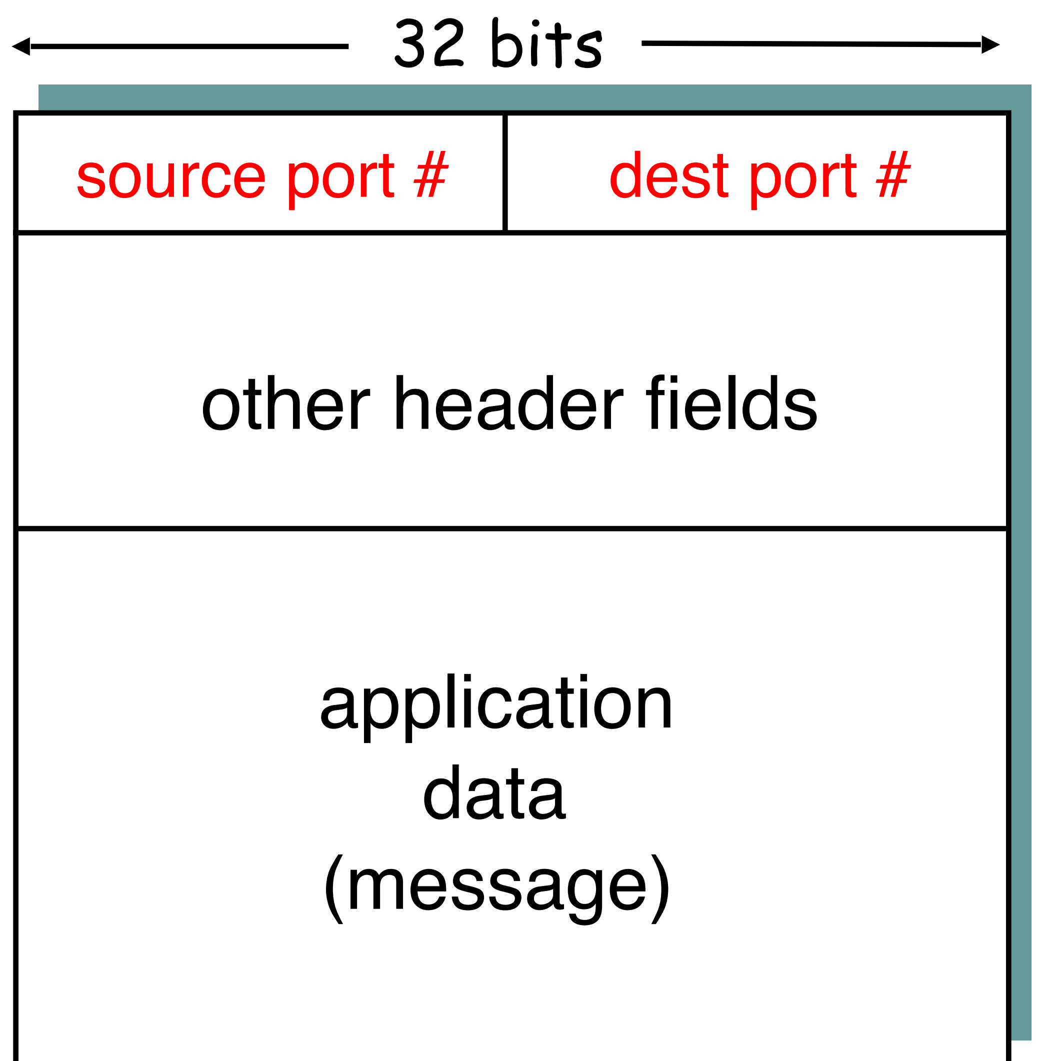
- Given to clients (at random)

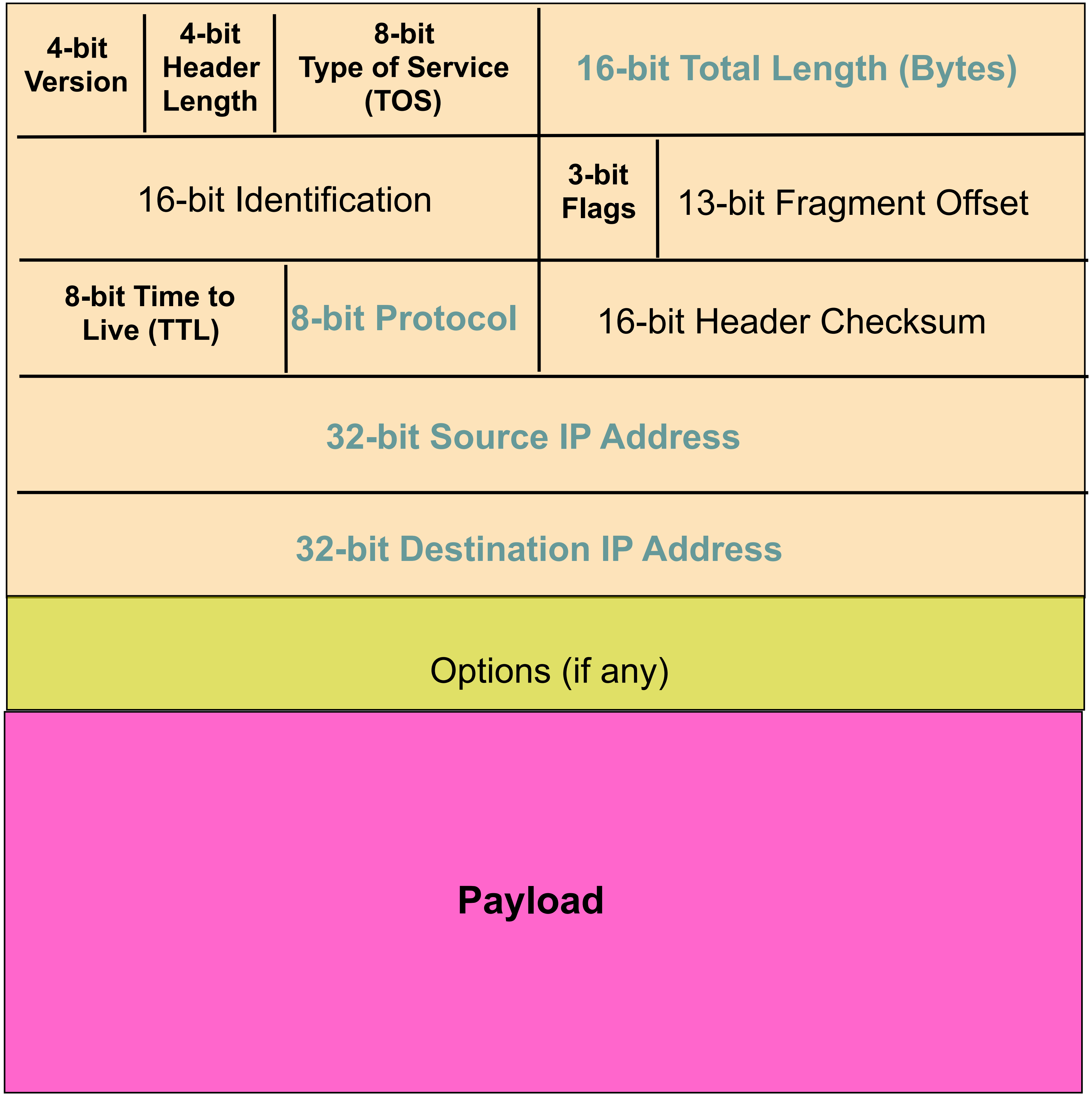
Multiplexing and Demultiplexing

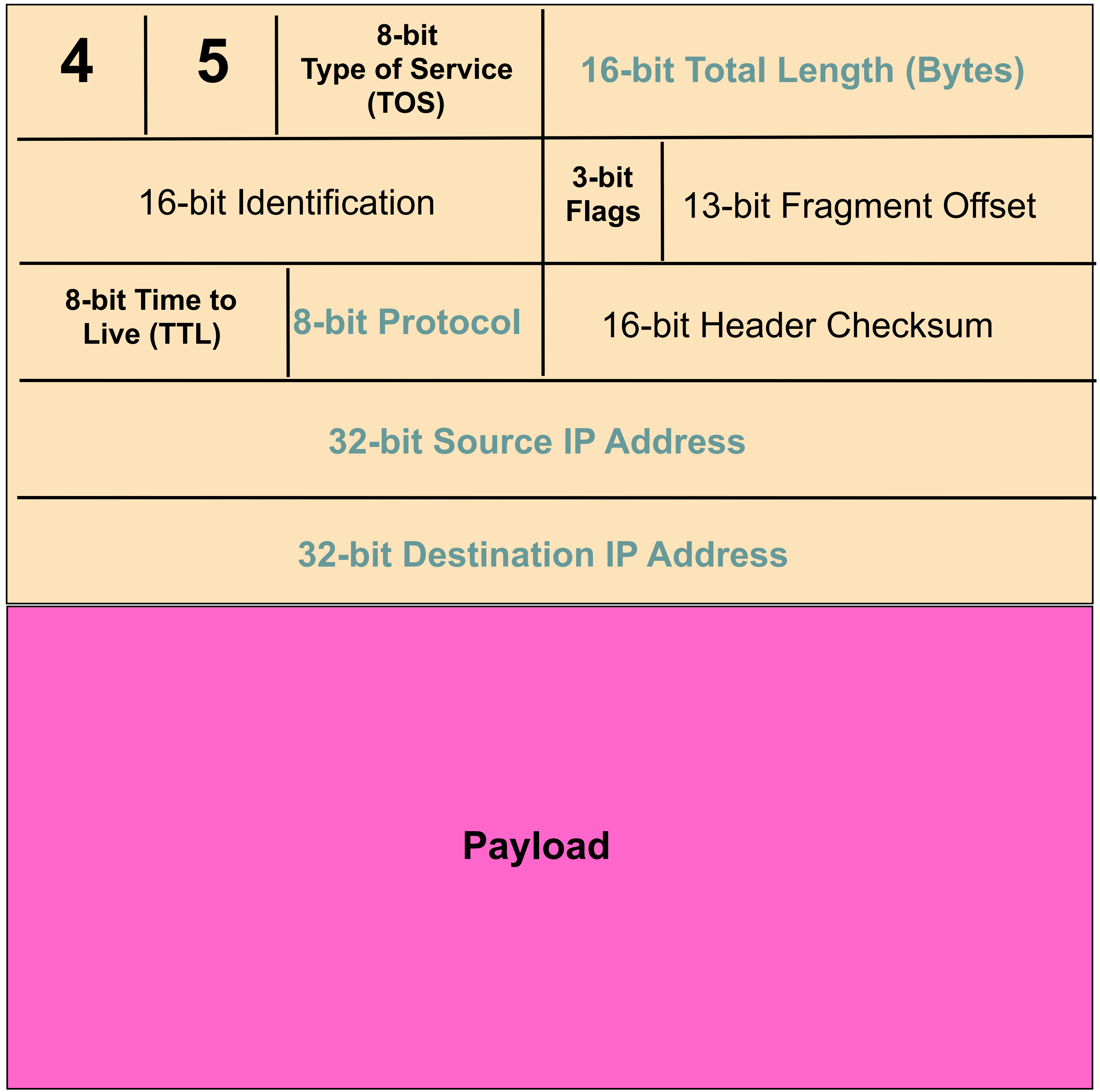
Host receives IP datagrams

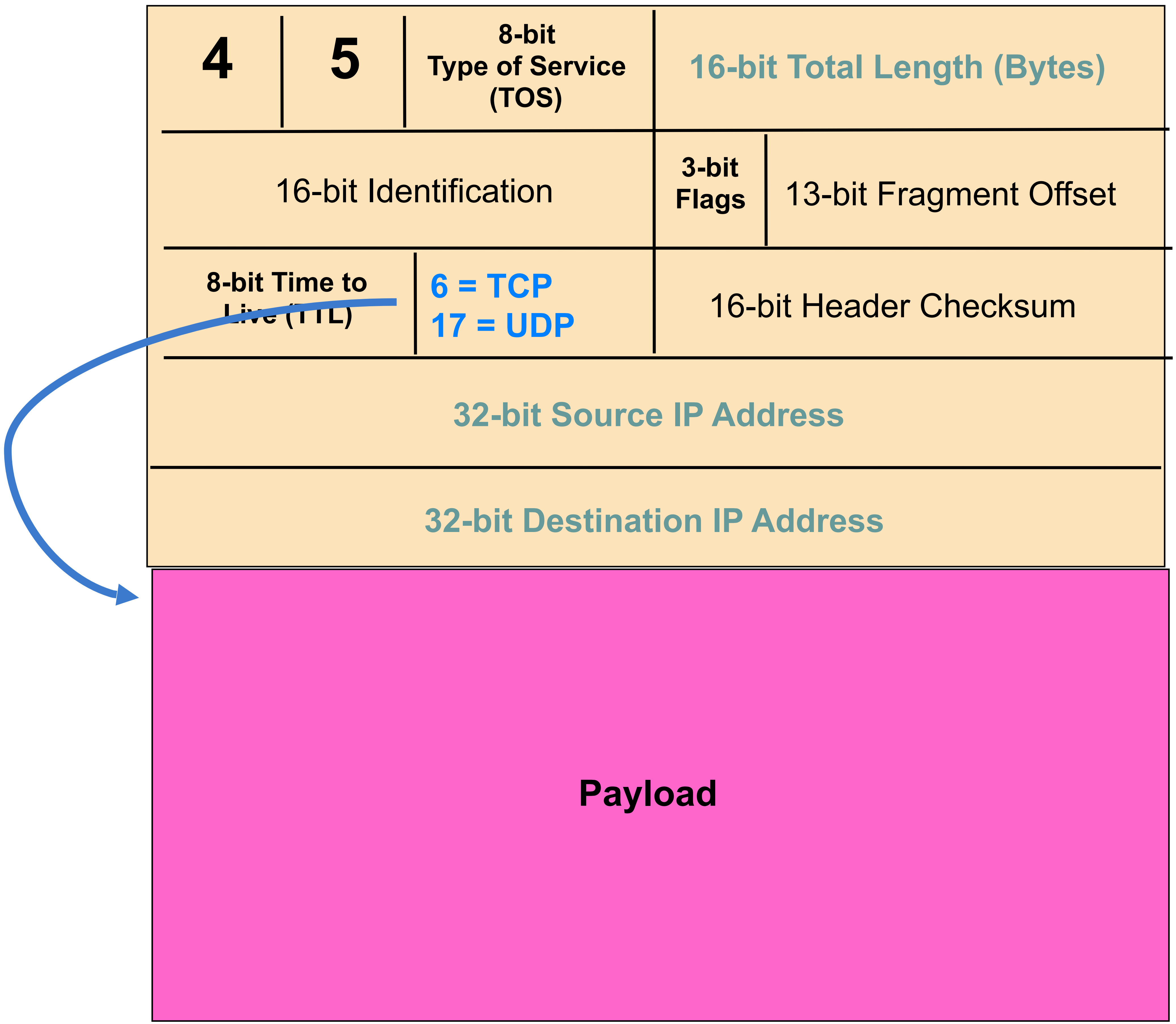
- Each datagram has source and destination IP address,
- Each segment has source and destination port number

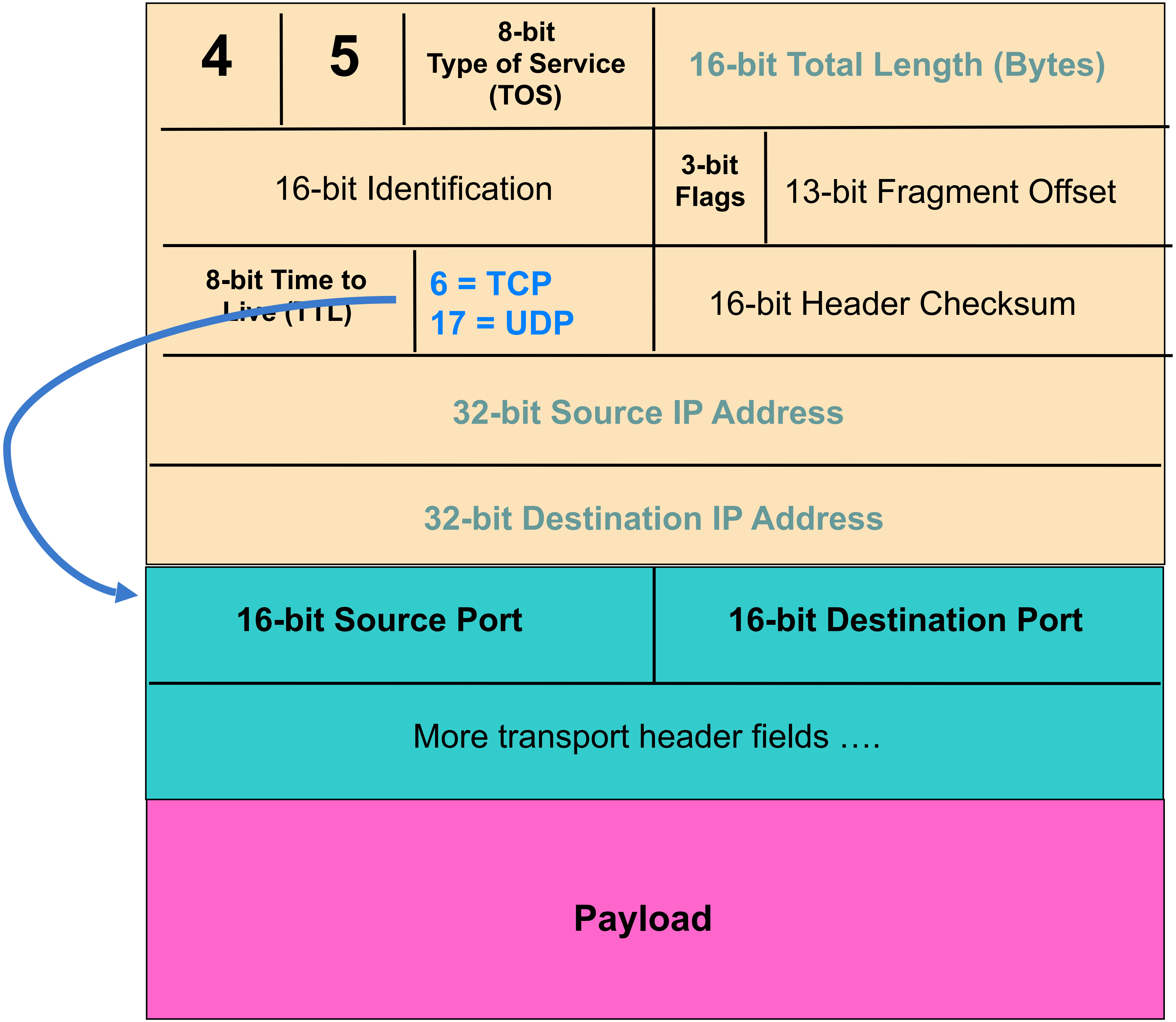
Host uses IP addresses and port numbers to direct the segment to appropriate socket




4-bit Version
4-bit Header Length
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
8-bit Protocol
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Options (if any)
Payload


4
5
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
8-bit Protocol
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Payload


4
5
8-bit
Type of Service
(TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit
Flags
13-bit Fragment Offset
8-bit Time to
Live (TTL)
6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Payload


4
5
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
16-bit Source Port
16-bit Destination Port
More transport header fields ….
Payload

UDP

UDP: User Datagram Protocol

Lightweight communication between processes

- Avoid overhead and delays of ordered, reliable delivery
- Send messages to and receive them from a socket

UDP described in RFC 768 – (1980!)

- IP plus port numbers to support (de)multiplexing
- Optional error checking on the packet contents
  - (checksum field = 0 means “don’t verify checksum”)

SRC port	DST port
checksum	length
DATA	

Why Would Anyone Use UDP?

Finer control over what data is sent and when

- As soon as an application process writes into the socket
- … UDP will package the data and send the packet

No delay for connection establishment

- UDP just blasts away without any formal preliminaries
- … which avoids introducing any unnecessary delays

No connection state

- No allocation of buffers, sequence #s, timers …
- … making it easier to handle many active clients at once

Small packet header overhead

- UDP header is only 8 bytes

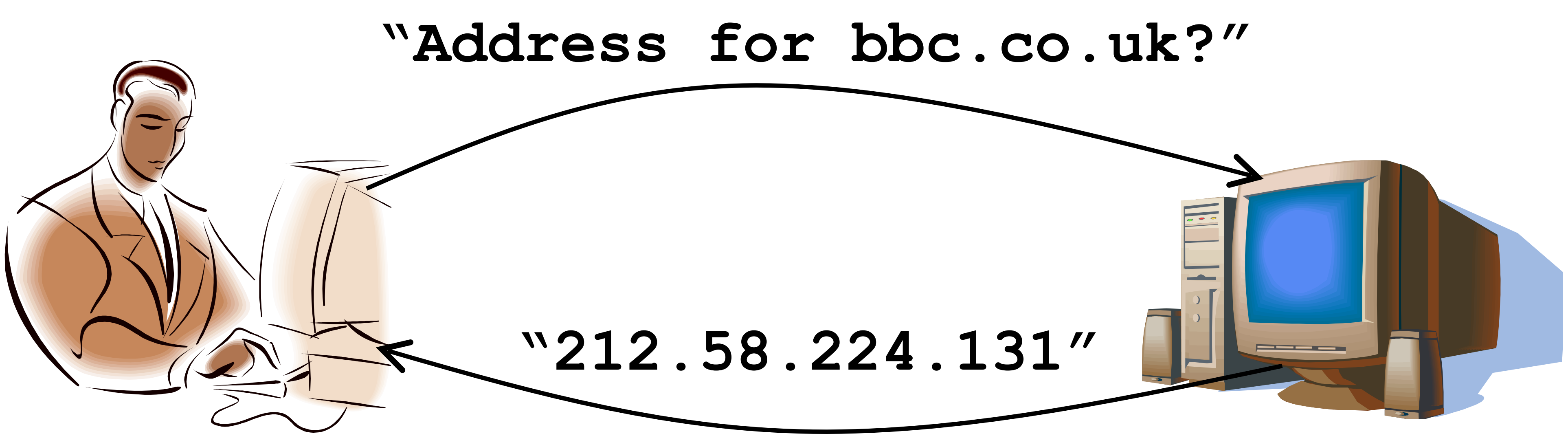
Popular Applications That Use UDP

Some interactive streaming apps

- Retransmitting lost/corrupted packets often pointless: by the time the packet is retransmitted, it's too late
  - telephone calls, video conferencing, gaming…
  - ***Modern streaming protocols using TCP (and HTTP)***

Simple query protocols like Domain Name System (DNS)

- Connection establishment overhead would double cost
- Easier to have application retransmit if needed



TCP

Transmission Control Protocol (TCP)

Reliable, in-order delivery *(previously, but quick review)*

- Ensures byte stream (eventually) arrives intact
  - In the presence of corruption and loss

Connection oriented *(today)*

- Explicit set-up and tear-down of TCP session

Full duplex stream-of-bytes service *(today)*

- Sends and receives a stream of bytes, not messages

Flow control *(previously, but quick review)*

- Ensures that sender doesn't overwhelm receiver

Congestion control *(next week)*

- Dynamic adaptation to network path's capacity

Basic Components of Reliability

ACKs

- Can’t be reliable without knowing whether data has arrived
- ***TCP uses byte sequence numbers to identify payloads***

Checksums

- Can’t be reliable without knowing whether data is corrupted
- ***TCP does checksum over TCP and pseudoheader***

Timeouts and retransmissions

- Can’t be reliable without retransmitting lost/corrupted data
- ***TCP retransmits based on timeouts and duplicate ACKs***
- *Timeout based on estimate of RTT*

Other TCP Design Decisions

Sliding window flow control

- Allow W contiguous bytes to be in flight

Cumulative acknowledgements

- Selective ACKs (full information) also supported (ignore)

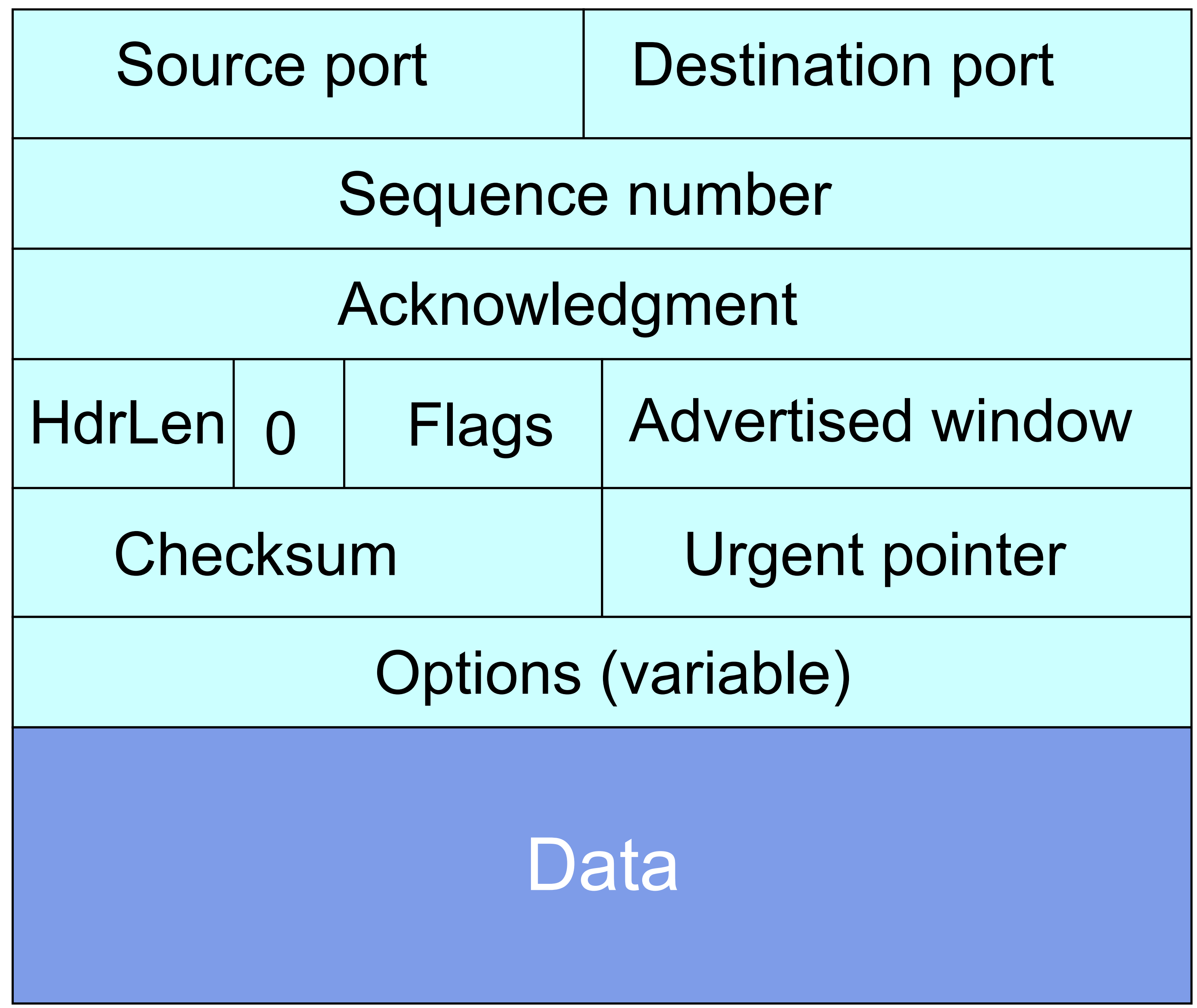
Single timer set after each payload is ACKed

- Timer is effectively for the “next expected payload”
- When timer goes off, resend that payload and wait
  - And double timeout period

Various tricks related to “fast retransmit”

- Using duplicate ACKs to trigger retransmission

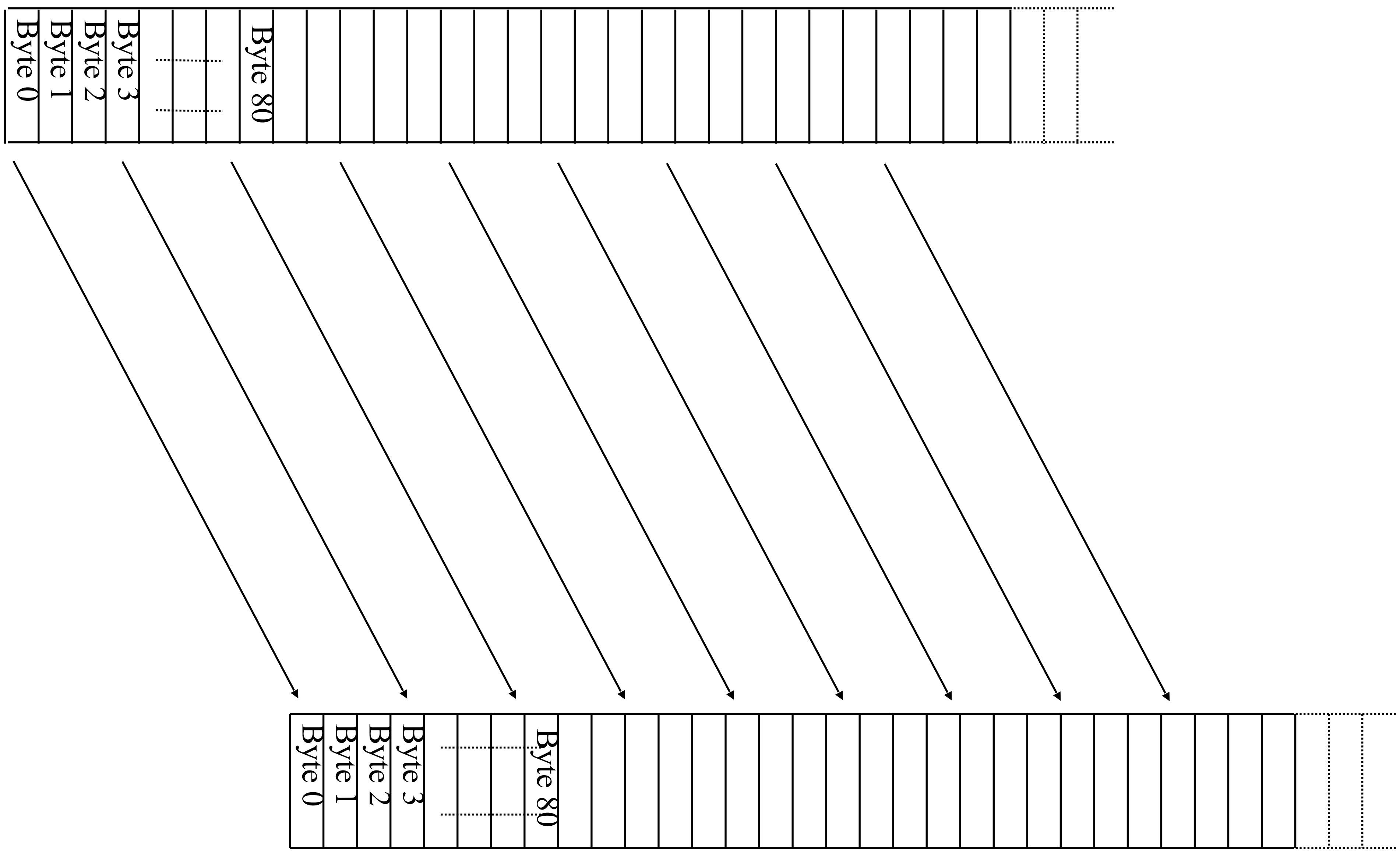
TCP Header


Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

Segments and Sequence Numbers

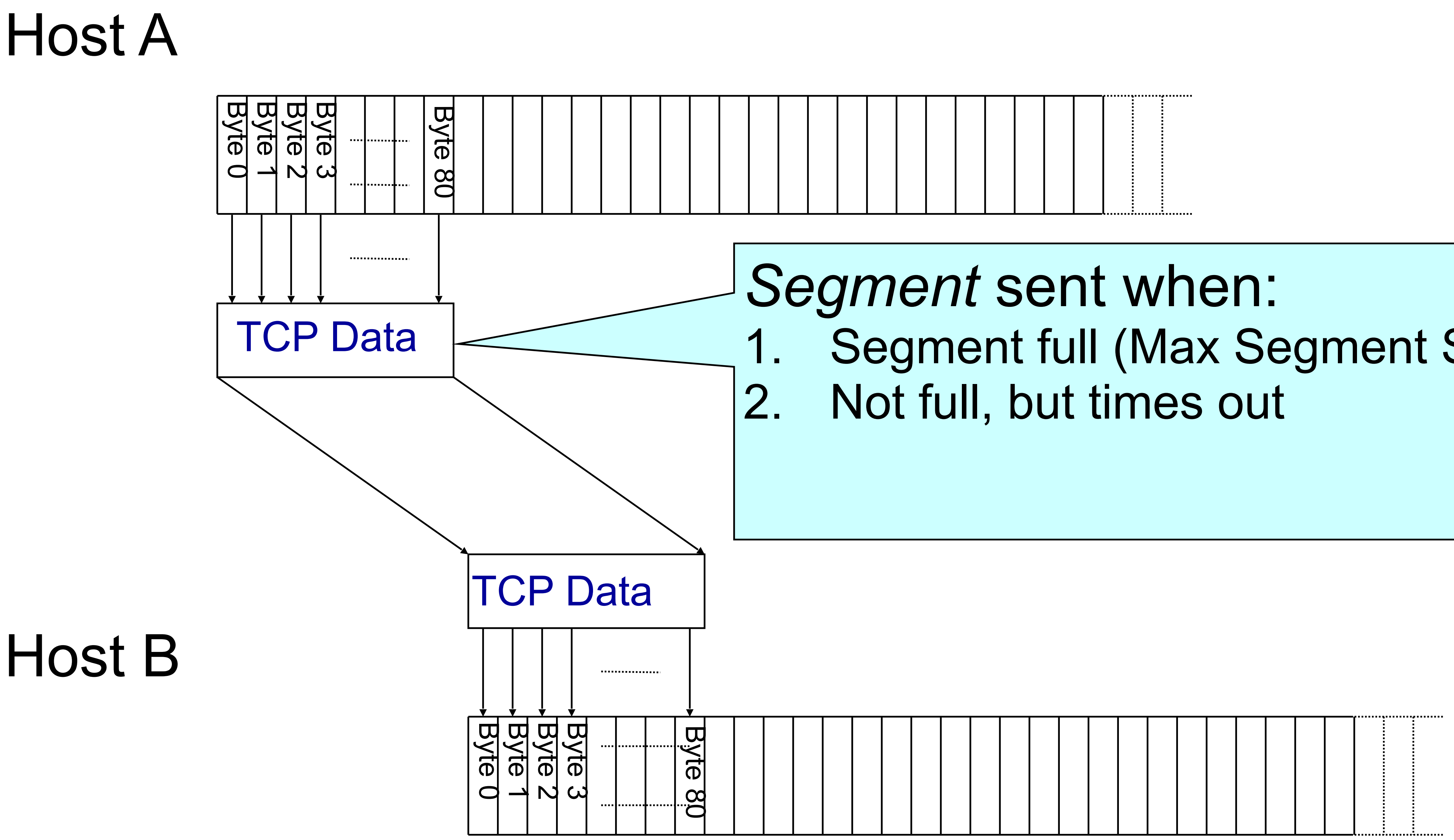
TCP "Stream of Bytes" Service…

Application @ Host A



Application @ Host B

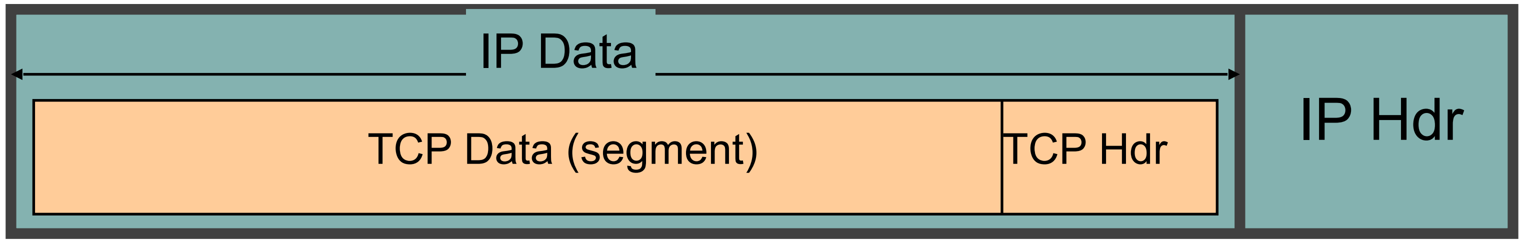
… Provided Using TCP "Segments"



Segment sent when:

1. Segment full (Max Segment Size),
2. Not full, but times out

TCP Segment



IP **packet**

- No bigger than Maximum Transmission Unit (MTU)
- E.g., up to 1500 bytes with Ethernet

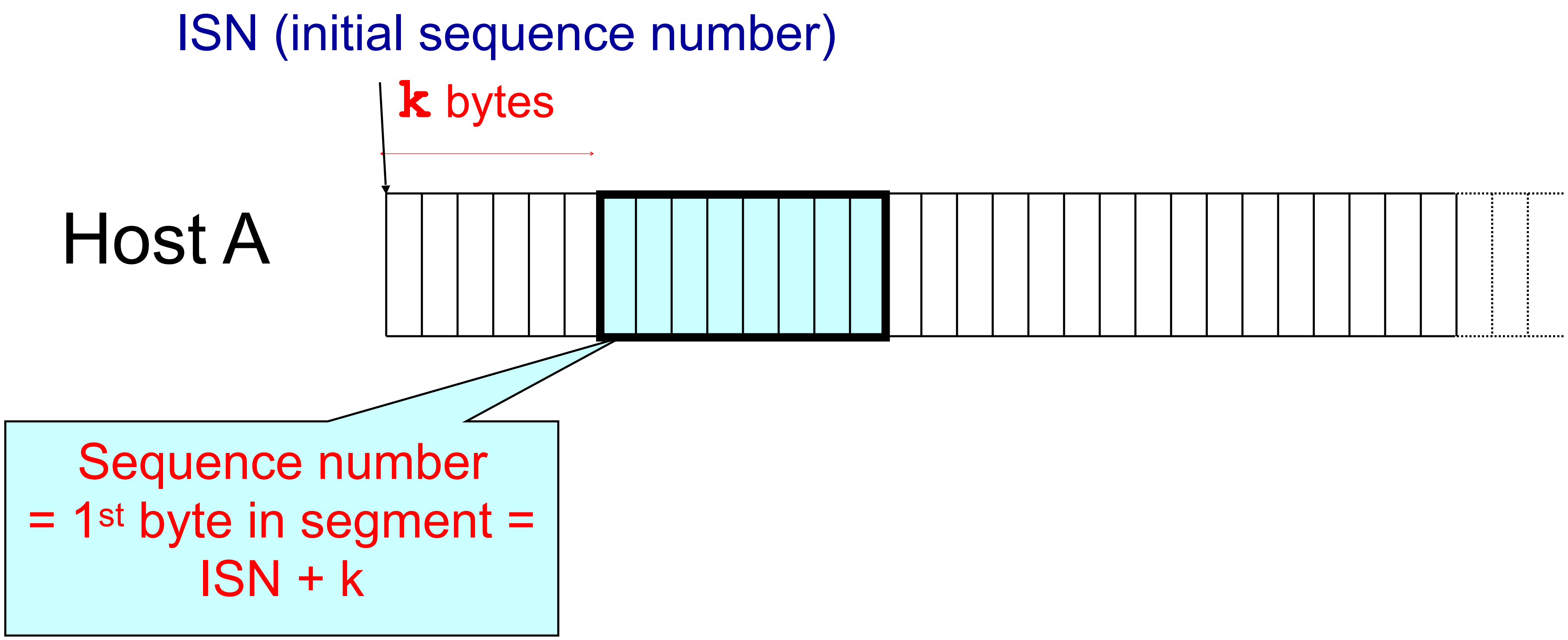
TCP **packet**

- IP packet with a TCP header and data inside
- TCP header ≥ 20 bytes long

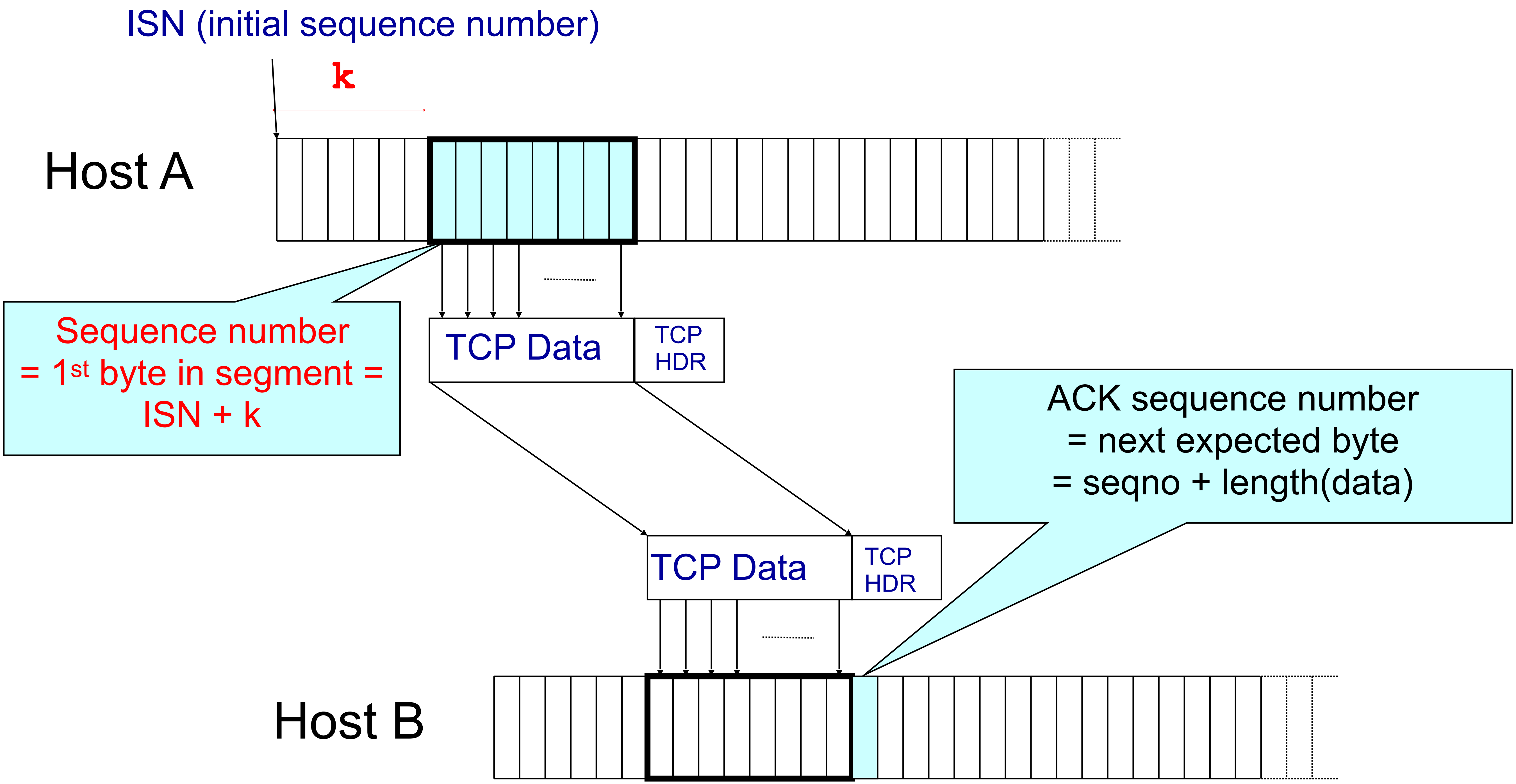
TCP **segment**

- No more than Maximum Segment Size (MSS) bytes
- E.g., up to 1460 consecutive bytes from the stream
- MSS = MTU – (IP header) – (TCP header)

Sequence Numbers



Sequence Numbers



ACKing and Sequence Numbers

Sender sends packet

- Data starts with sequence number X
- Packet contains B bytes
  - X, X+1, X+2, ….X+B-1

Upon receipt of packet, receiver sends an ACK

- If all data prior to X already received:
  - ACK acknowledges X+B (because that is next expected byte)
- If highest contiguous byte received is smaller value Y
  - ACK acknowledges Y+1
  - Even if this has been ACKed before

Normal Pattern

Sender: seqno=X, length=B

Receiver: ACK=X+B

Sender: seqno=X+B, length=B

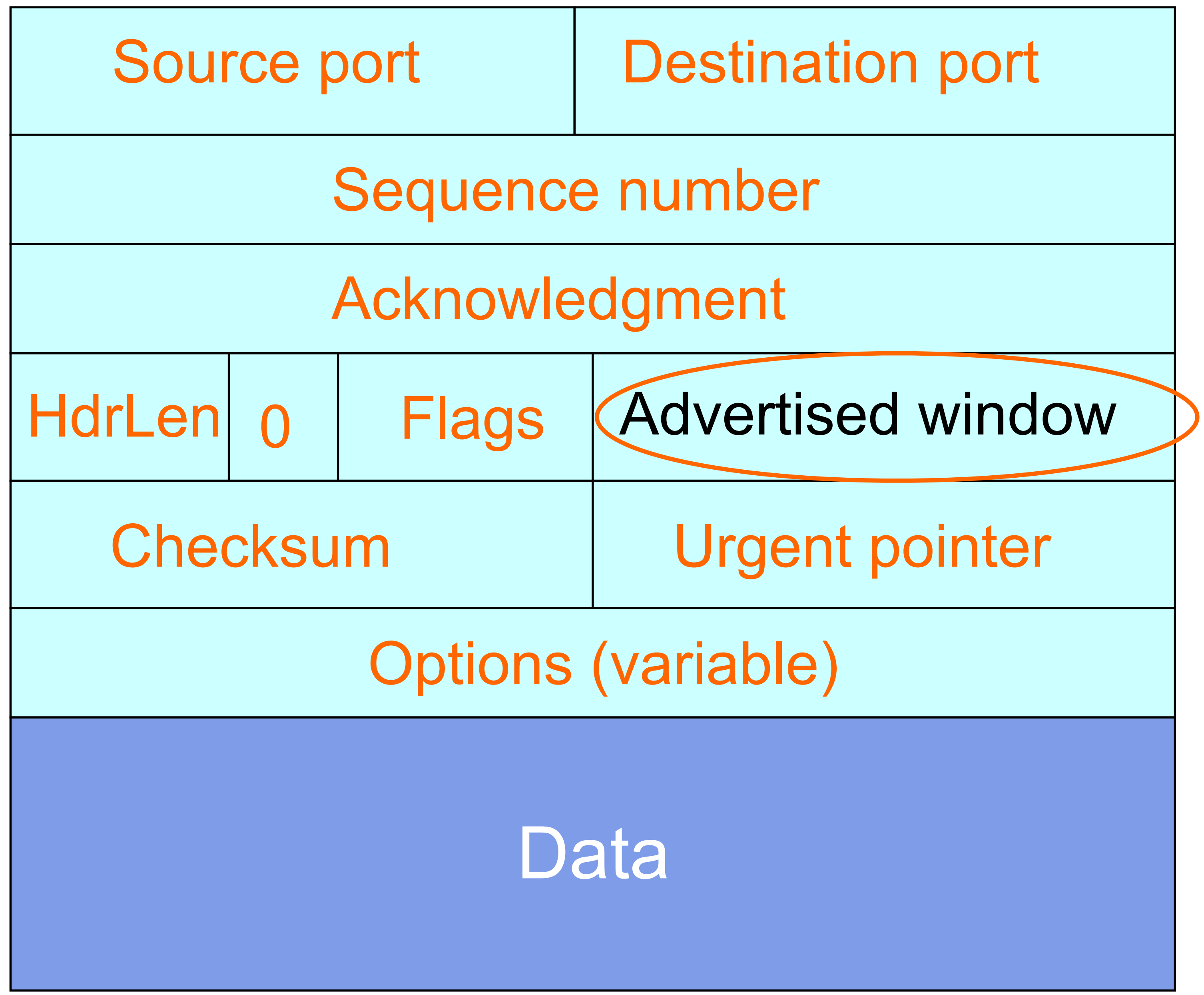
Receiver: ACK=X+2B

Sender: seqno=X+2B, length=B

…

Seqno of next packet is same as last ACK field

TCP Header


Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

Sliding Window Flow Control

Advertised Window: W

- Can send W bytes beyond the next expected byte

Receiver uses W to prevent sender from overflowing buffer

Limits number of bytes sender can have in flight

Advertised Window Limits Rate

Sender can send no faster than W/RTT bytes/sec

Receiver only advertises more space when it has consumed old arriving data

In original TCP design, that was the **sole** protocol mechanism controlling sender's rate

What's missing?

Implementing Sliding Window

Both sender & receiver maintain a **window**

- Sender: not yet ACK'ed
- Receiver: not yet delivered to application

Left edge of window:

- Sender: beginning of unacknowledged data
- Receiver: beginning of undelivered data

For the sender:

- Window size = maximum amount of data in flight

For the receiver:

- Window size = maximum amount of undelivered data

Sliding Window Summary

Sender: window advances when new data ack'd

Receiver: window advances as receiving process consumes data

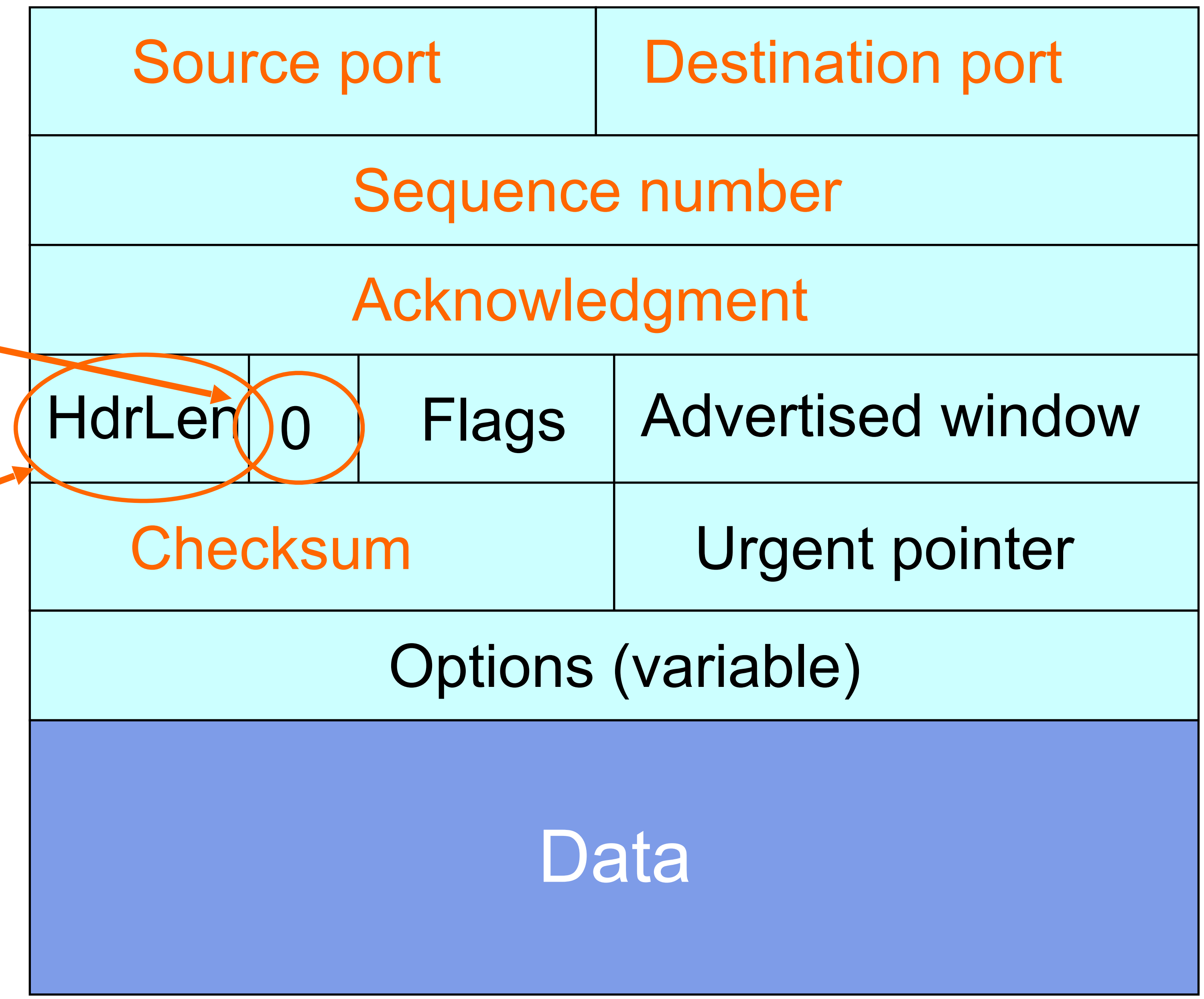
Receiver advertises to the sender where the receiver window currently ends ("righthand edge")

- Sender agrees not to exceed this amount
- It makes sure by setting its own window size to a value that can't send beyond the receiver's righthand edge

TCP Header: What's left?

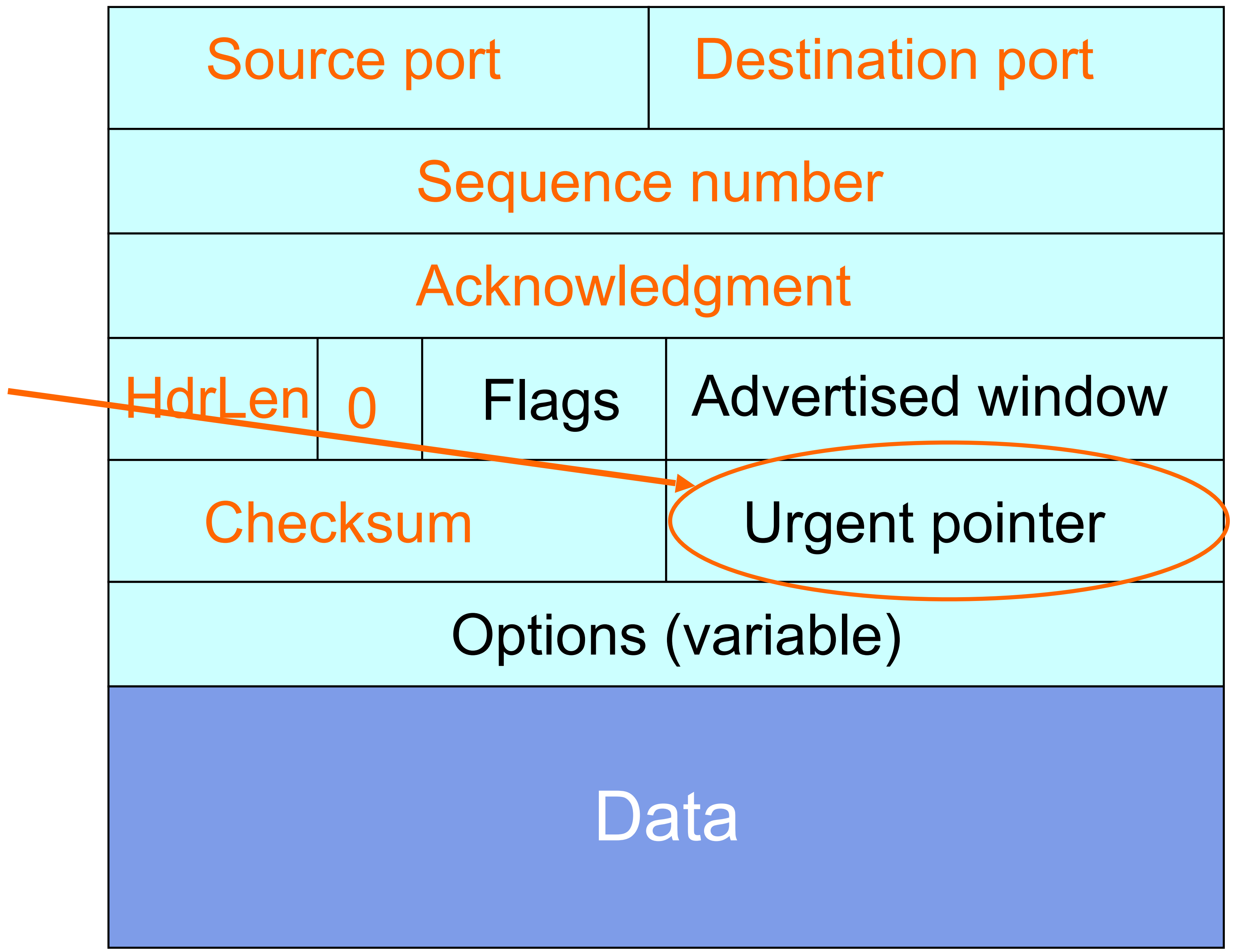
"Must Be Zero"
6 bits reserved

Number of 4-byte words in TCP header;
5 = no options

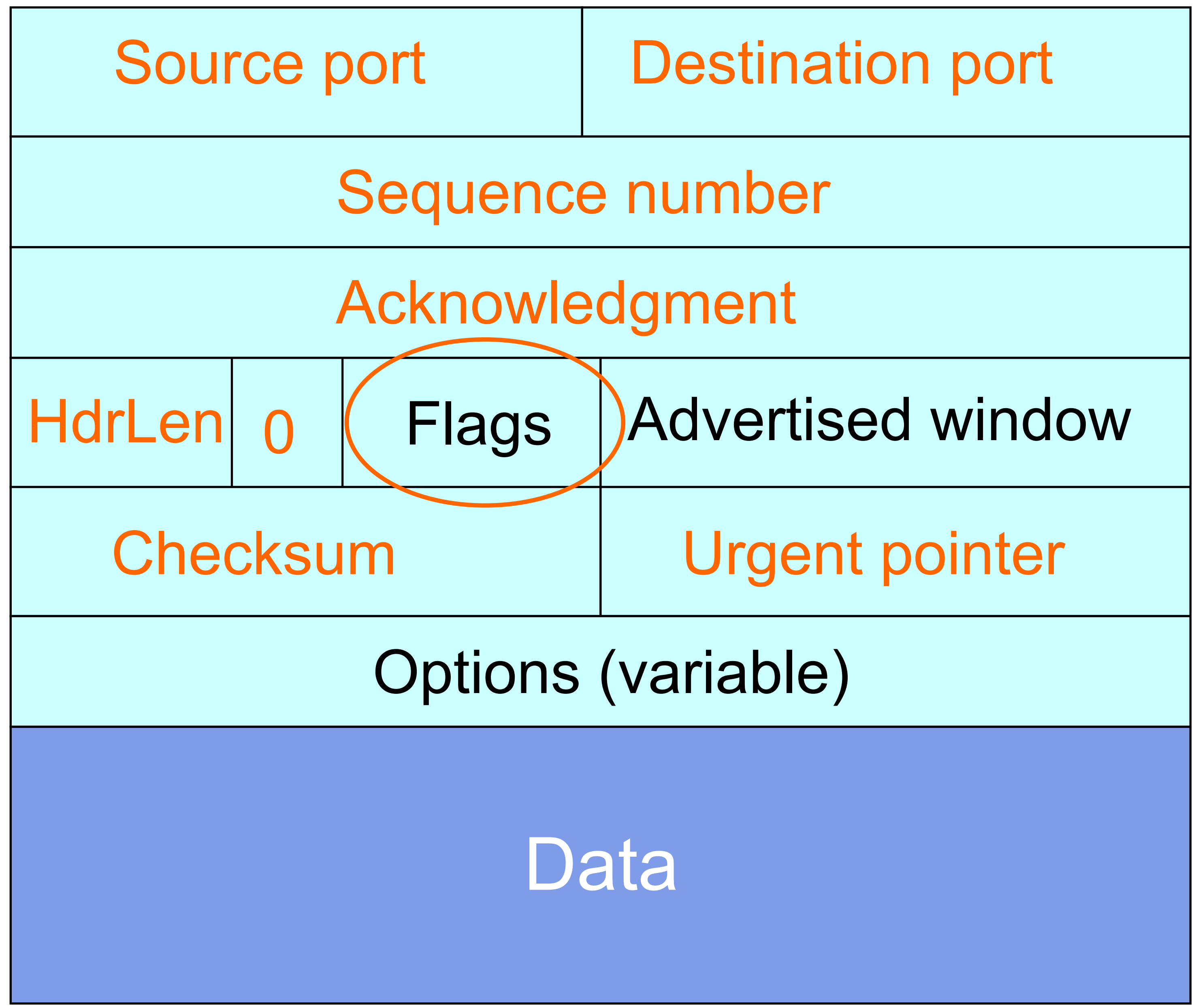


TCP Header: What's left?

Used with **URG** flag to indicate urgent data (not discussed further)



TCP Header: What's left?



TCP Connection Establishment and Initial Sequence Numbers

Initial Sequence Number (ISN)

Sequence number for the very first byte

- E.g., Why not just use ISN = 0?

Practical issue

- IP addresses and port #s uniquely identify a connection
- Eventually, though, these port #s do get used again
- … small chance an old packet is still in flight

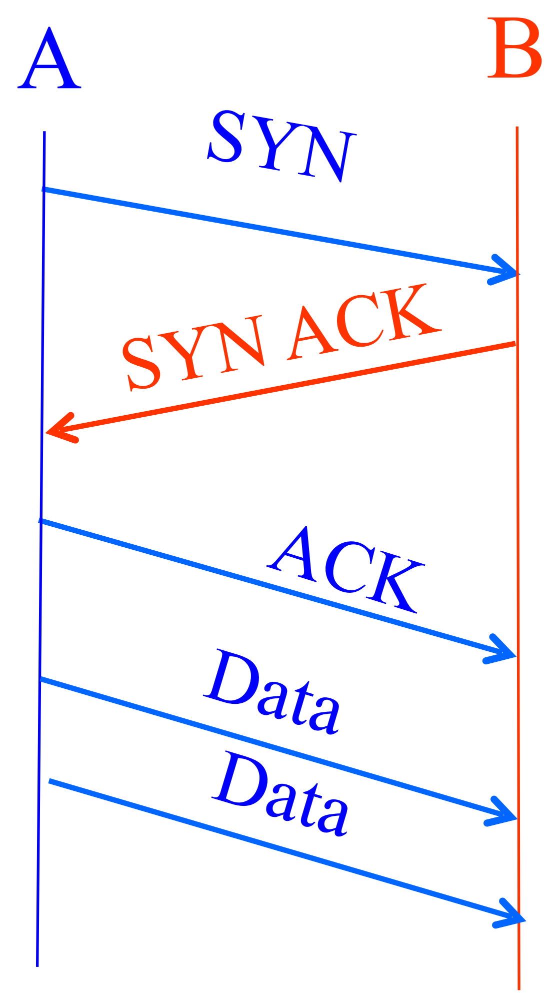
TCP therefore requires changing ISN

- initially set from 32-bit clock that ticks every 4 microseconds
- now drawn from a pseudo random number generator (security)

To establish a connection, hosts exchange ISNs

- **How does this help?**

Establishing a TCP Connection

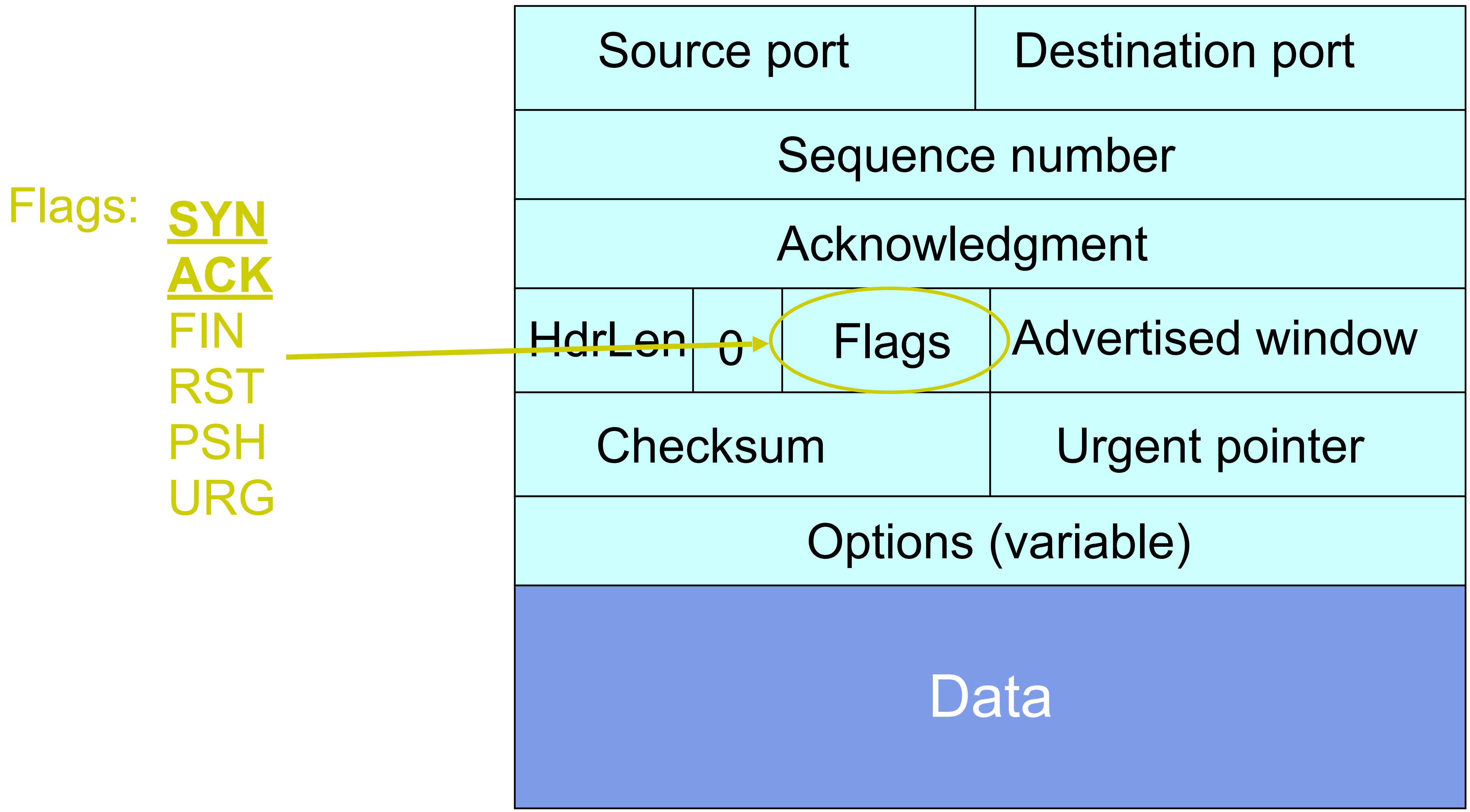


Each host tells its ISN to the other host.

Three-way handshake to establish connection

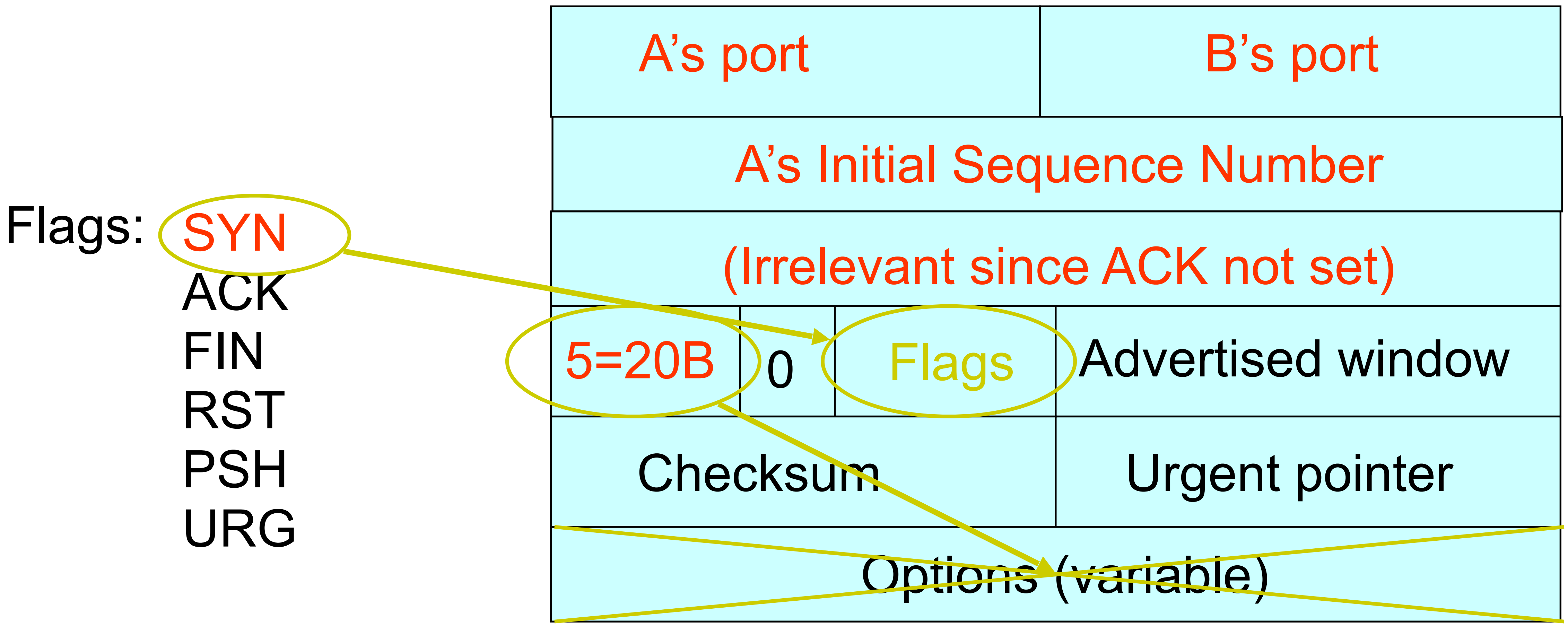
- Host A sends a **SYN** (open; "synchronize sequence numbers")
- Host B returns a SYN acknowledgment (**SYN ACK**)
- Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header



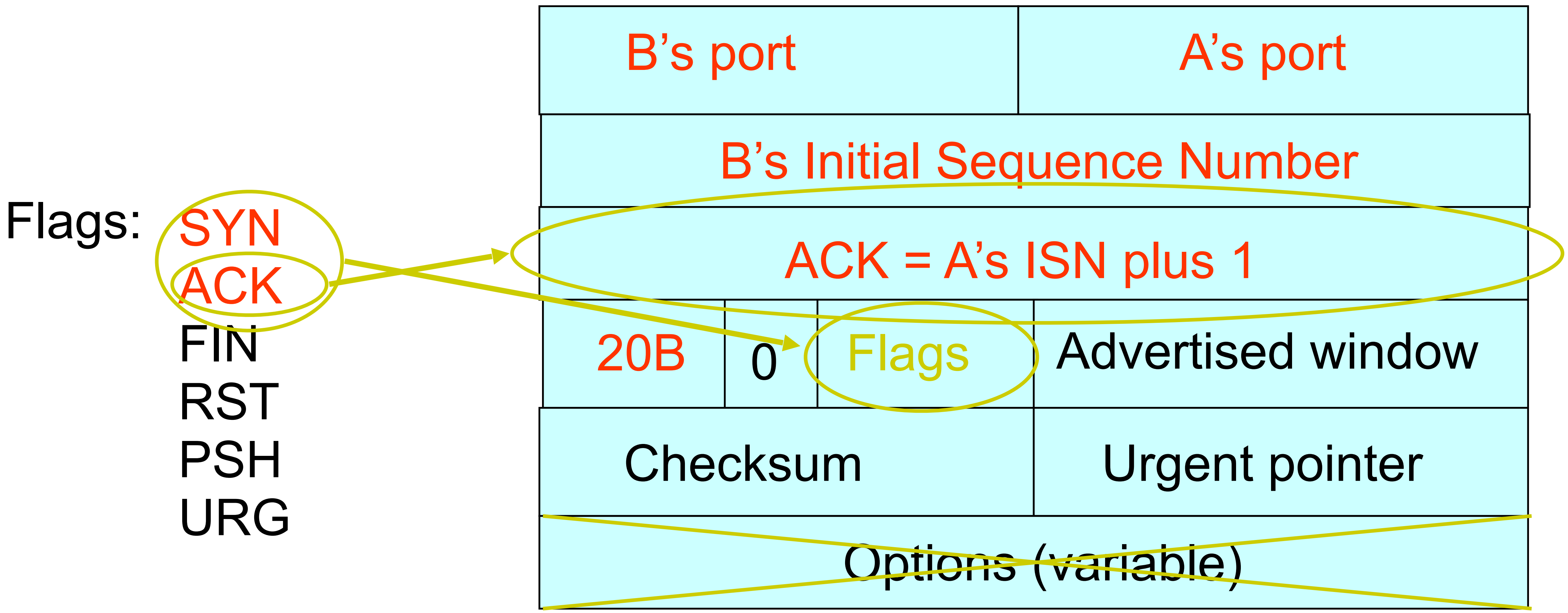
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

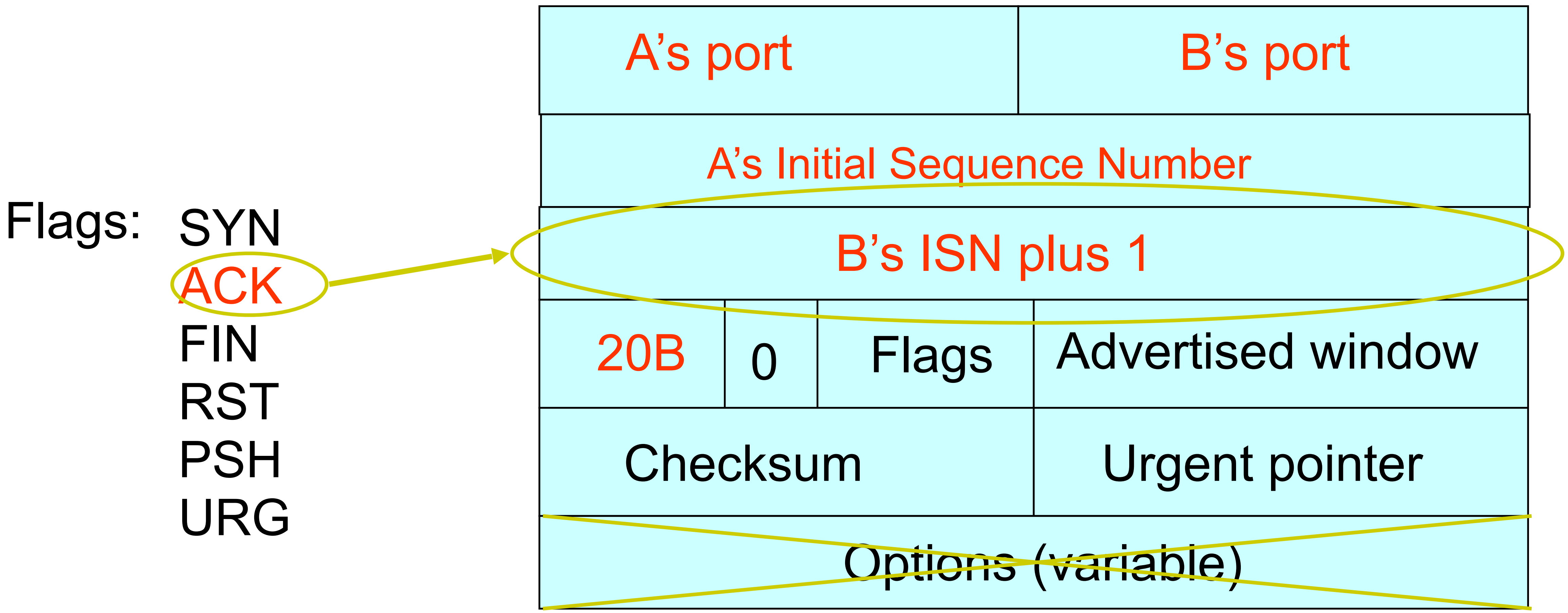
Step 2: B’s SYN-ACK Packet



B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

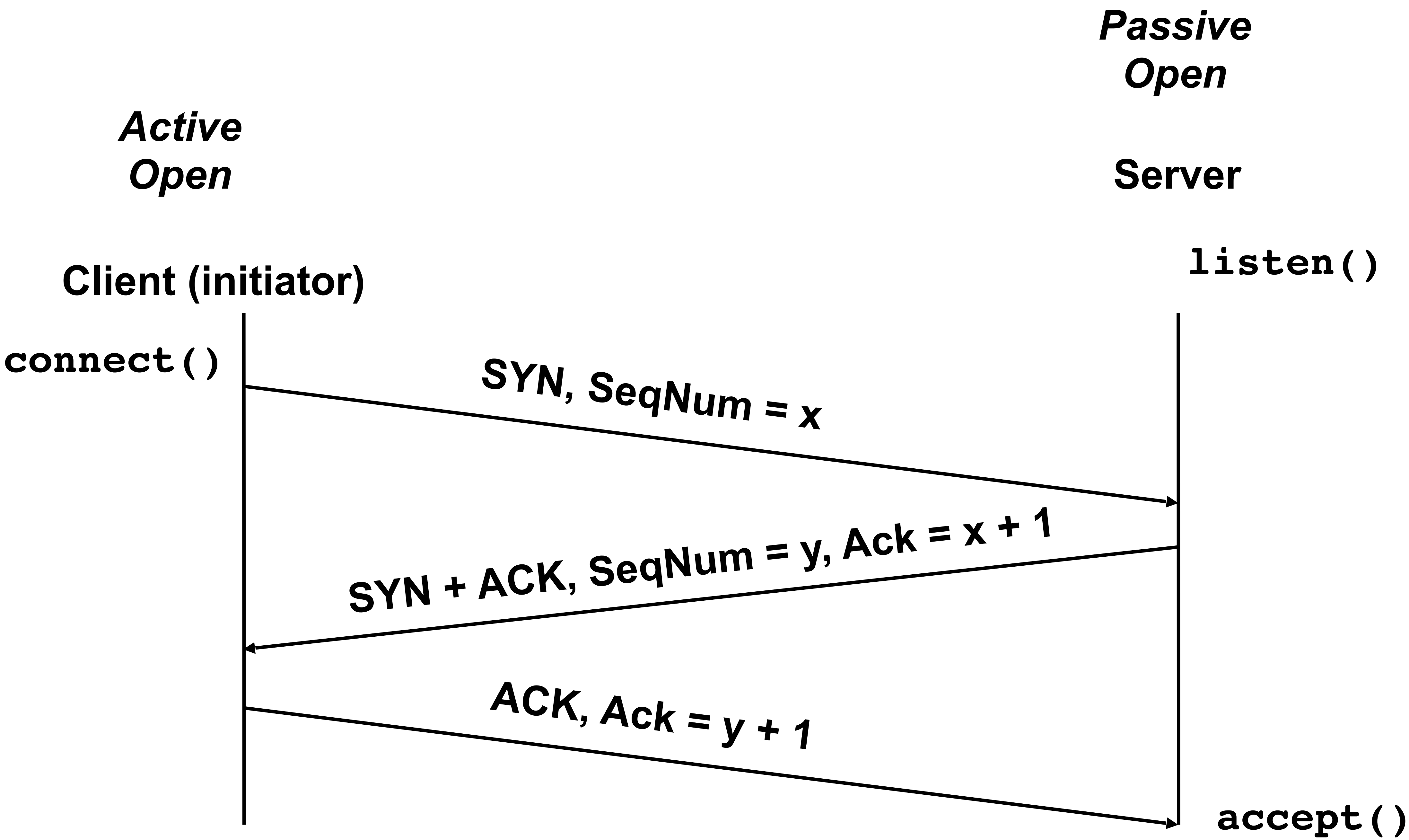
Step 3: A's ACK of the SYN-ACK



A tells B it's likewise okay to start sending

… upon receiving this packet, B can start sending data

Timing Diagram: 3-Way Handshaking



What if the SYN Packet Gets Lost?

Suppose the SYN packet gets lost

- Packet is lost inside the network, or:
- Server discards the packet (e.g., listen queue is full)

Eventually, no SYN-ACK arrives

- Sender sets a timer and waits for the SYN-ACK
- … and retransmits the SYN if needed

How should the TCP sender set the timer?

- Sender has no idea how far away the receiver is
- Hard to guess a reasonable length of time to wait
- **SHOULD** (RFCs 1122 & 2988) use default of 3 seconds
  - Other implementations instead use 6 seconds

SYN Loss and Web Downloads

User clicks on a hypertext link

- Browser creates a socket and does a "connect"
- The "connect" triggers the OS to transmit a SYN

If the SYN is lost…

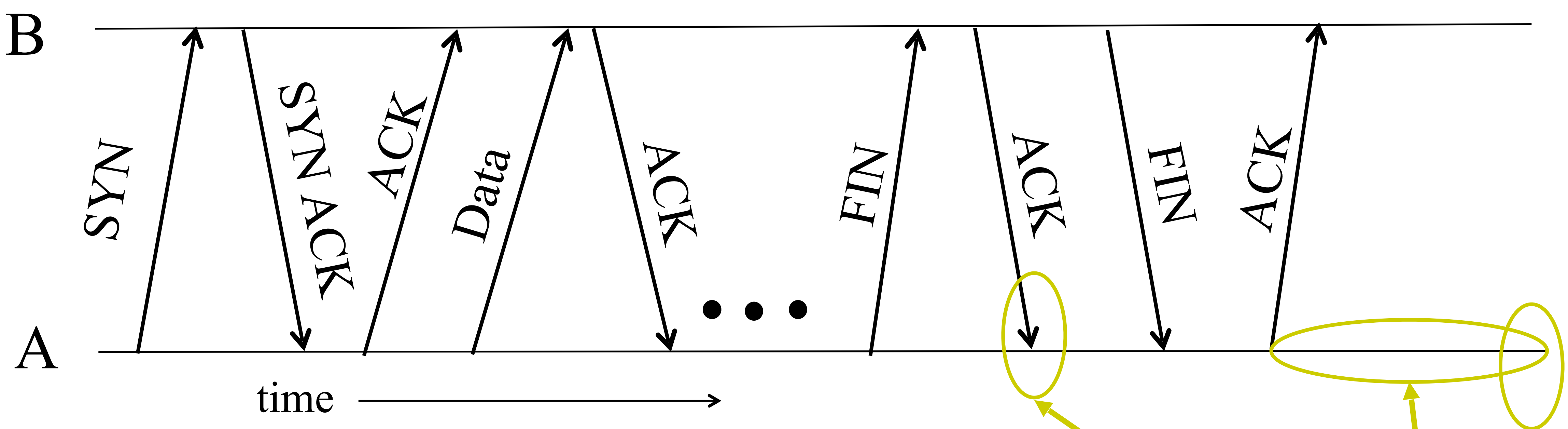
- 3-6 seconds of delay: can be very long
- User may become impatient
- … and click the hyperlink again, or click "reload"

User triggers an "abort" of the "connect"

- Browser creates a new socket and another "connect"
- Essentially, forces a faster send of a new SYN packet!
- Sometimes very effective, and the page comes quickly

Tearing Down the Connection

Normal Termination, One Side At A Time



Finish (**FIN**) to close and receive remaining bytes

- **FIN** occupies one octet in the sequence space

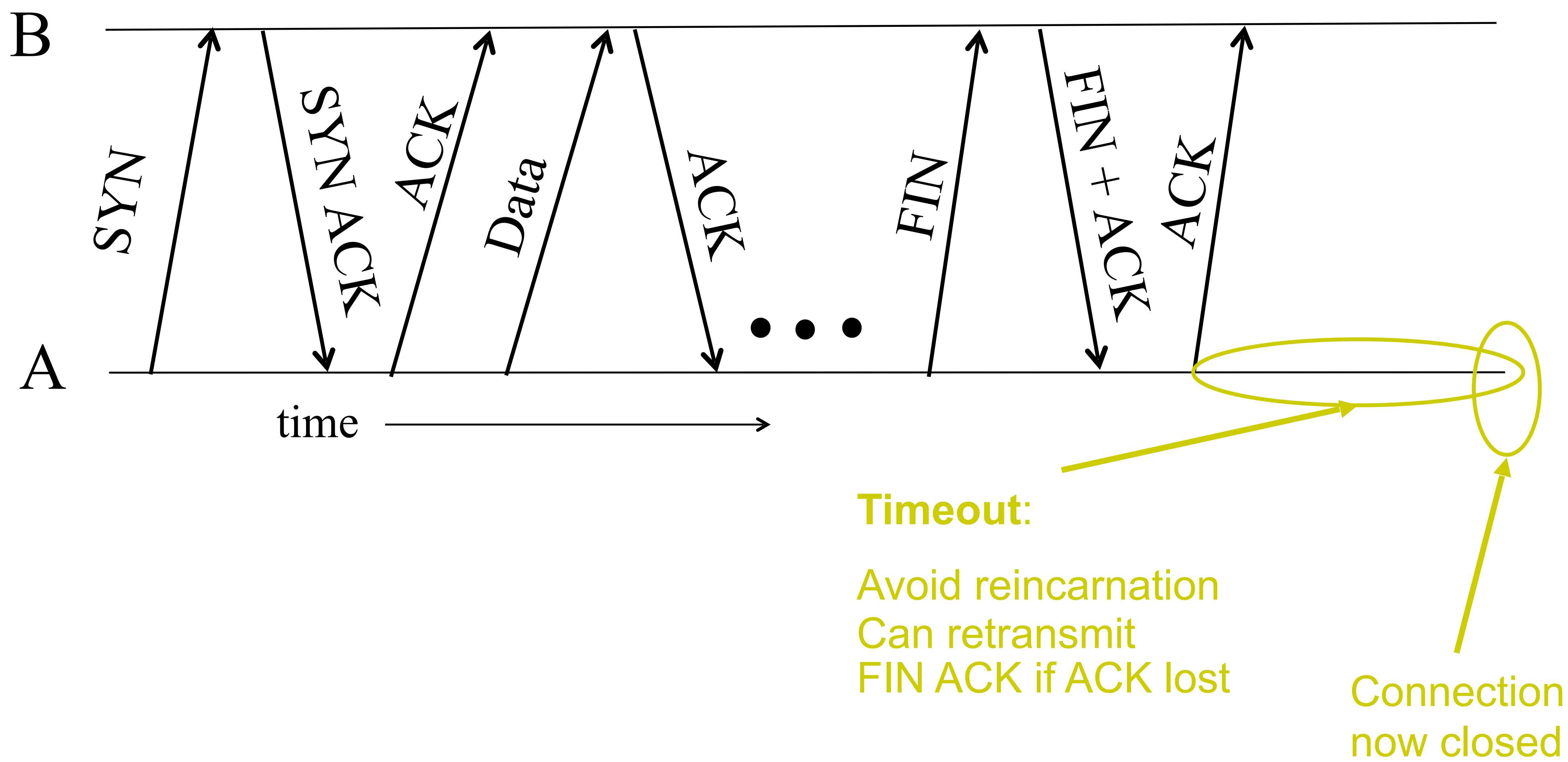
Other host ack's the octet to confirm

Closes A's side of the connection, but not B's

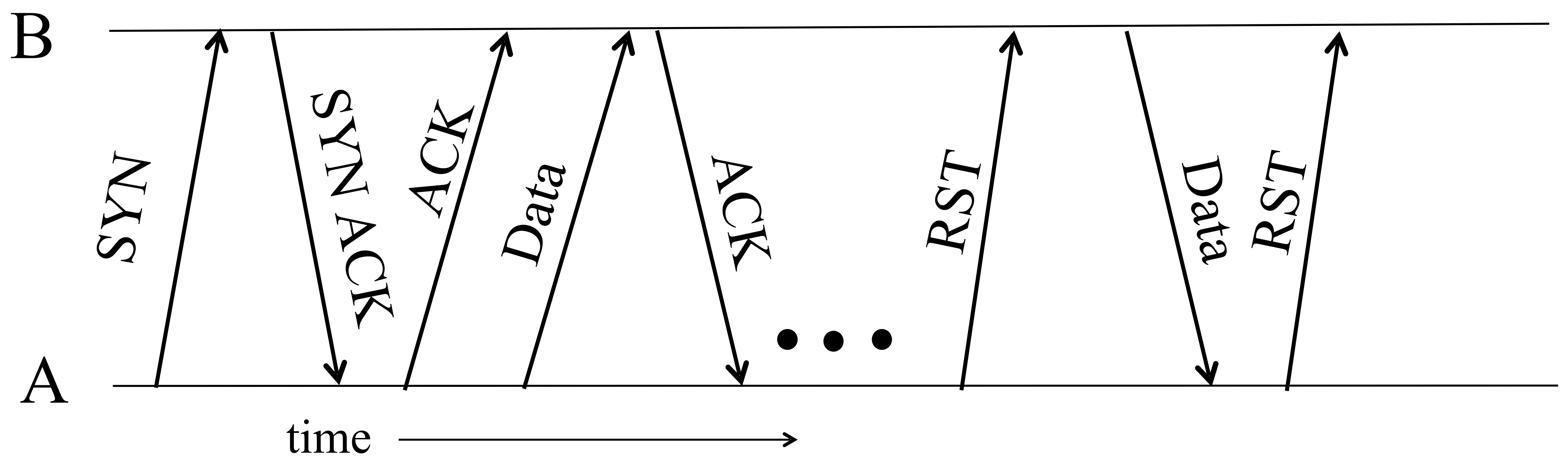
- Until B likewise sends a **FIN**
- Which A then acks

Normal Termination, Both Together

Same as before, but B sets **FIN** with their ack of A's **FIN**



Abrupt Termination



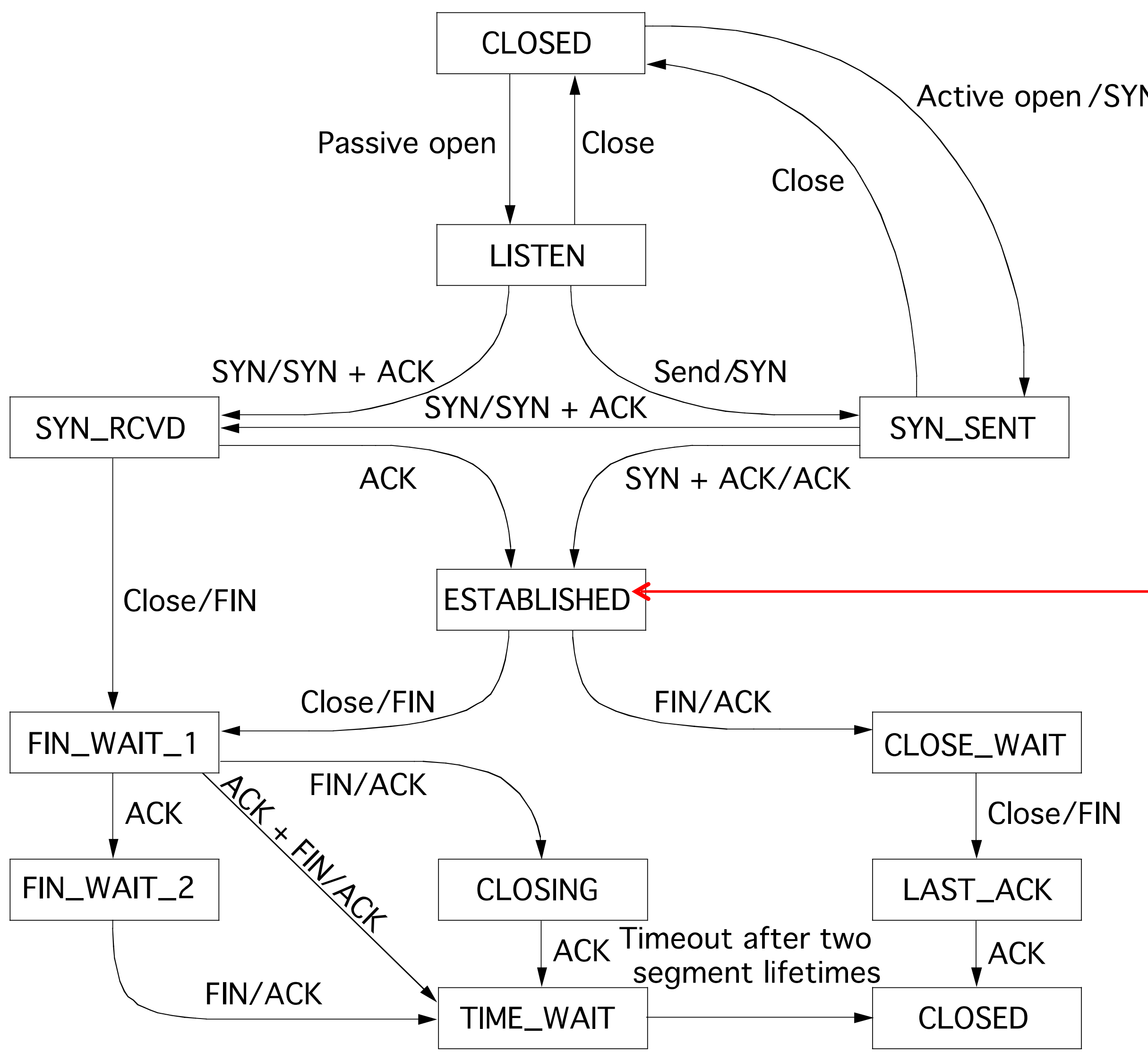
A sends a RESET (**RST**) to B

- E.g., because app. process on A crashed

That's it

- B does not ack the **RST**
- Thus, **RST** is not delivered reliably
- And: any data in flight is lost
- But: if B sends anything more, will elicit another **RST**

TCP State Transitions



Data, ACK exchanges are in here

Reliability: TCP Retransmission

Timeouts and Retransmissions

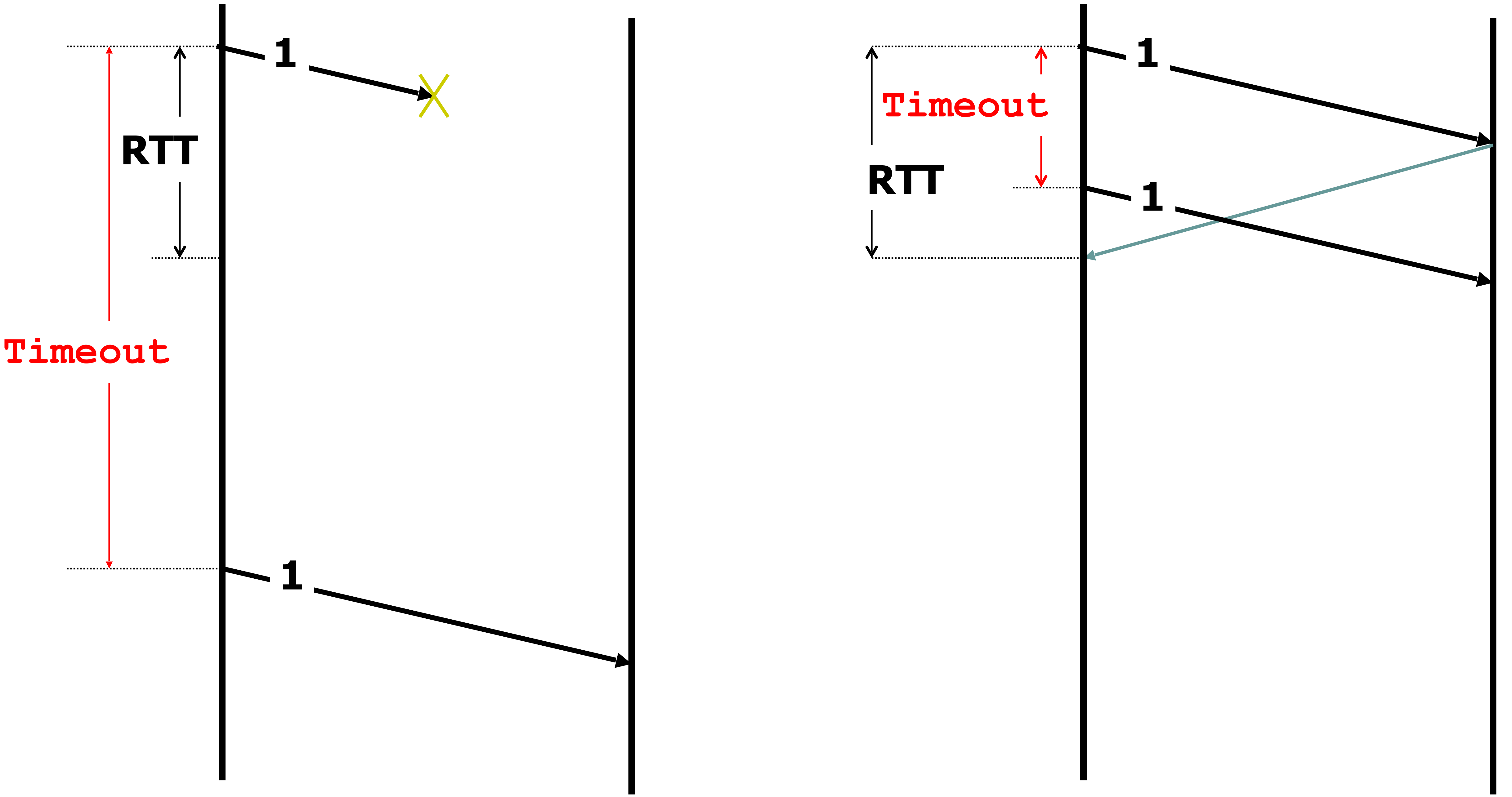
Reliability requires retransmitting lost data

Involves setting timer and retransmitting on timeout

TCP resets timer whenever new data is ACKed

- Retx of packet containing “next byte” when timer goes off

Setting the Timeout Value



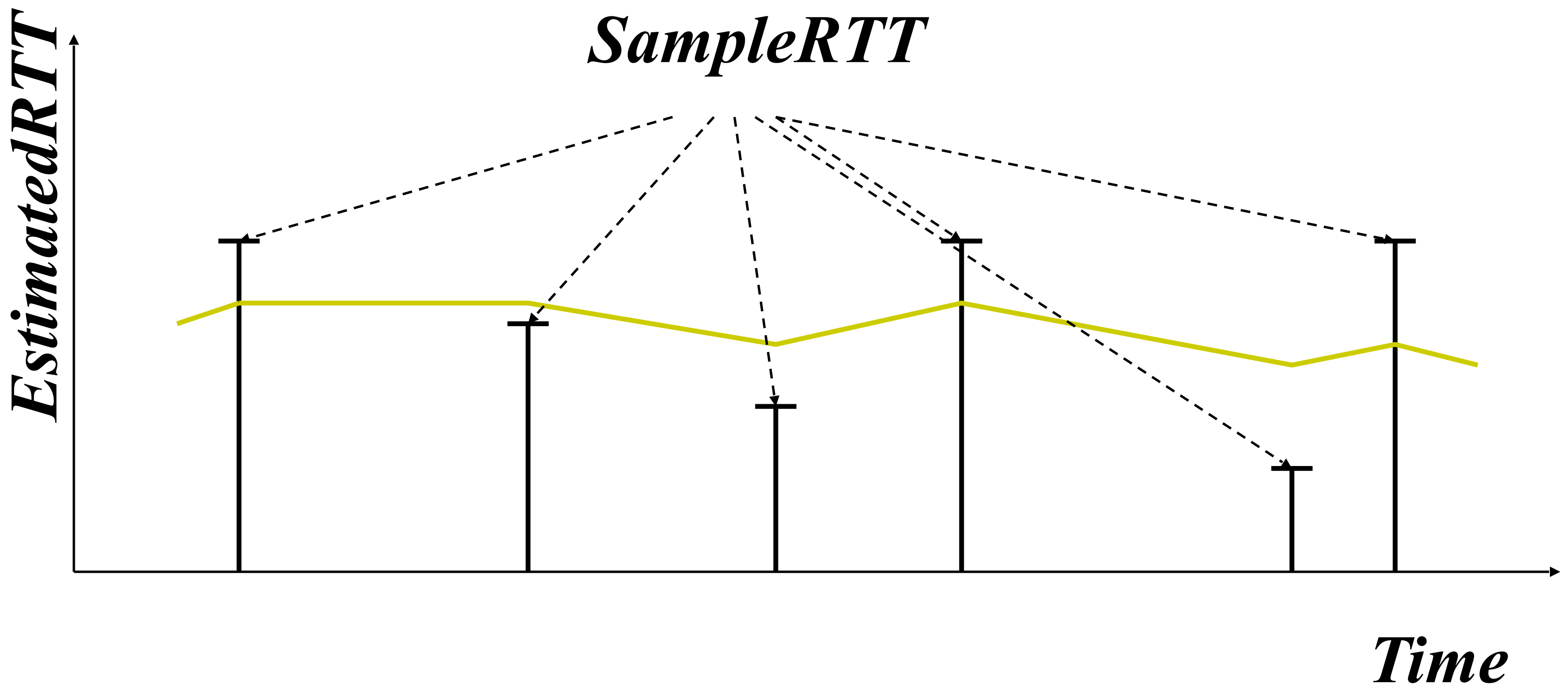
Timeout too long → inefficient

Timeout too short → duplicate packets

RTT Estimation

Use exponential averaging of RTT samples

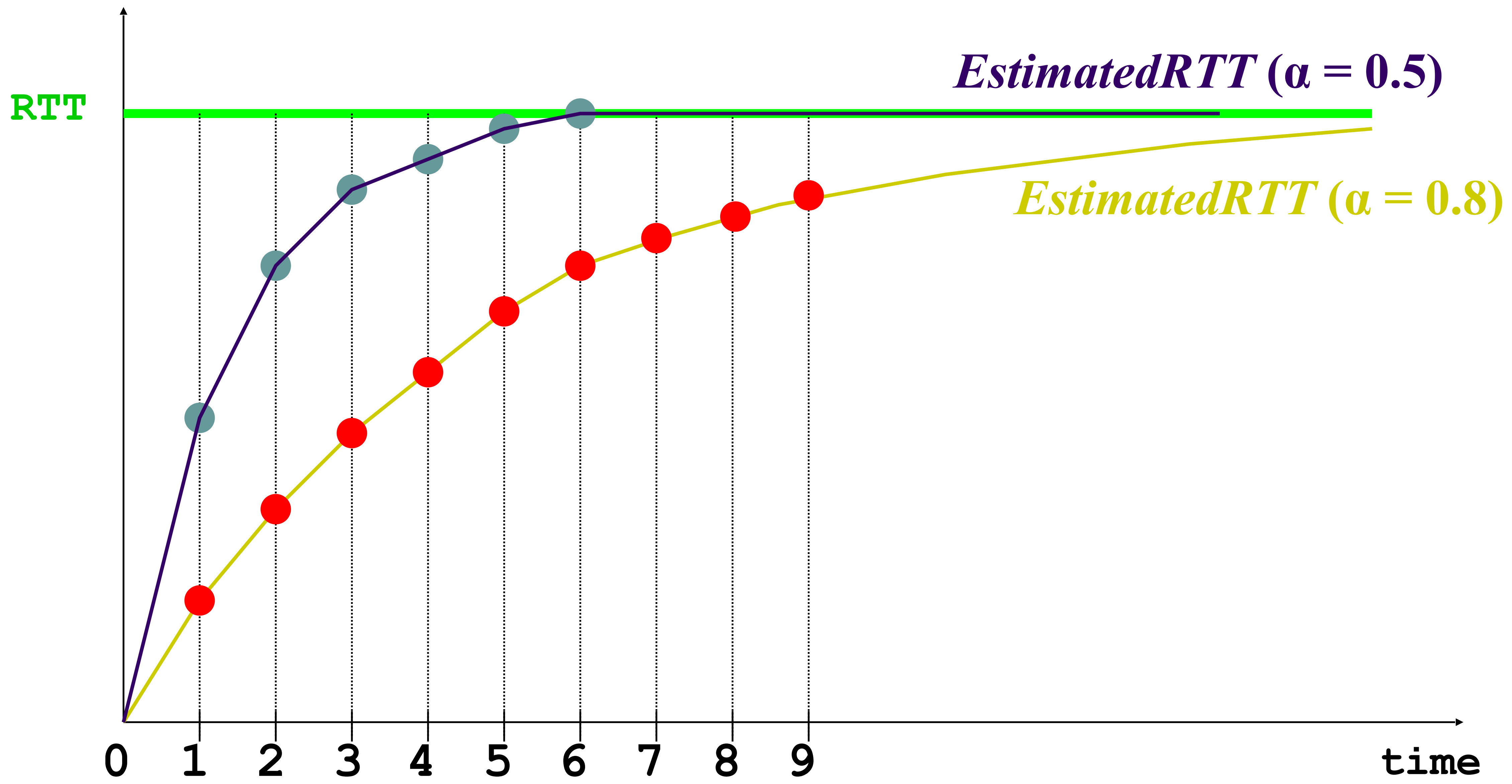
$$SampleRTT = AckRcvdTime - SendPacketTime$$
$$EstimatedRTT = \alpha \times EstimatedRTT + (1-\alpha) \times SampleRTT$$
$$0 < \alpha \leq 1$$



Exponential Averaging Example

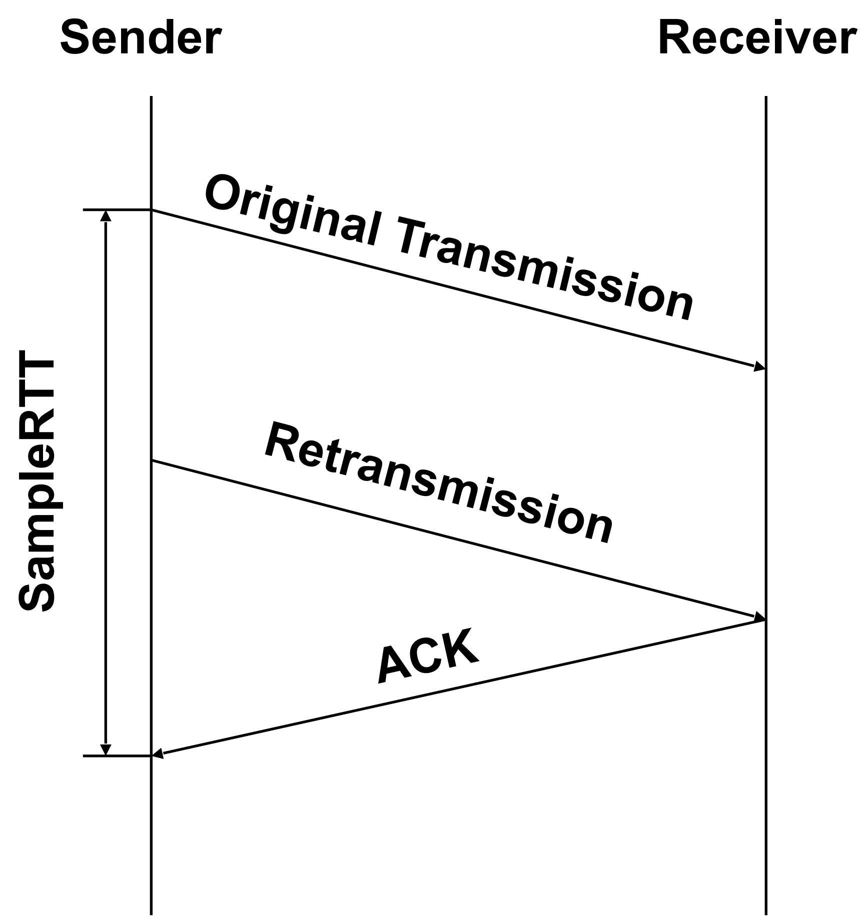
EstimatedRTT = α**EstimatedRTT* + (1 – α)**SampleRTT*

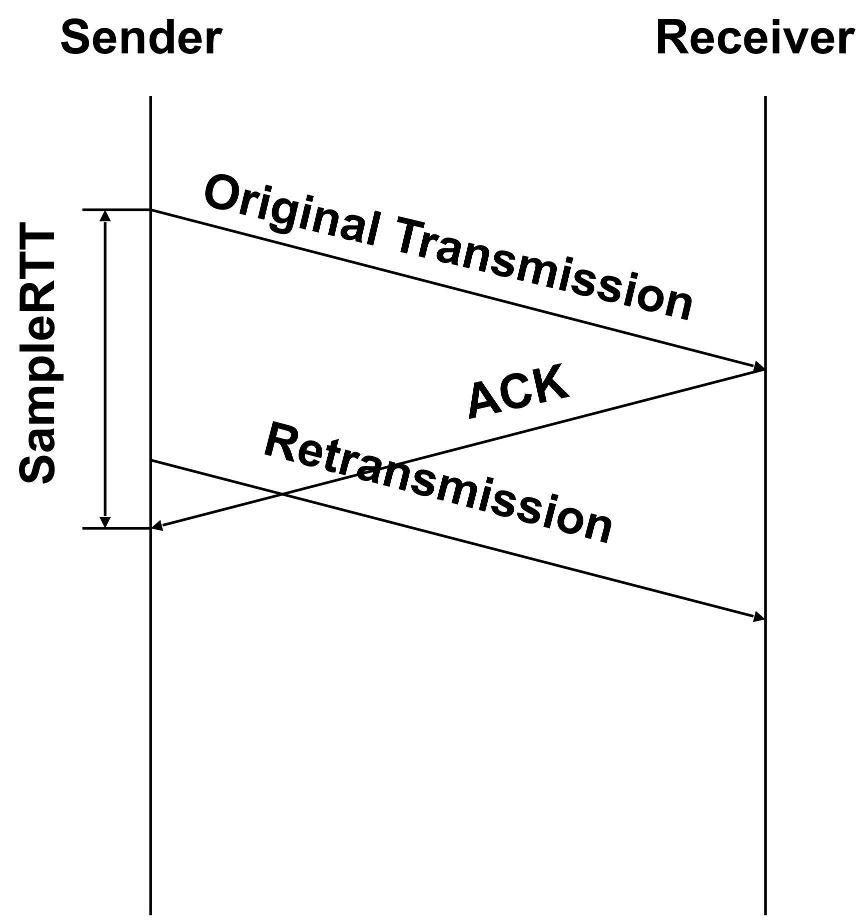
Assume RTT is constant → *SampleRTT* = RTT



Problem: Ambiguous Measurements

How do we differentiate between the real ACK, and ACK of the retransmitted packet?





Karn/Partridge Algorithm

Measure *SampleRTT* only for original transmissions

- Once a segment has been retransmitted, do not use it for any further measurements
- Computes *EstimatedRTT* using $\alpha = 0.875$

Timeout value (RTO) $= 2 \times$ *EstimatedRTT*

Use exponential backoff for repeated retransmissions

- Every time RTO timer expires, set RTO $\leftarrow 2 \cdot$RTO
    - (Up to maximum $\geq$ 60 sec)
- Every time new measurement comes in (= successful original transmission), collapse RTO back to $2 \times$ EstimatedRTT

This is all very interesting, but…..

Implementations often use a coarse-grained timer

- 500 msec is typical

So what?

- Above algorithms are largely irrelevant
- **Incurring a timeout is expensive**

So we rely on duplicate ACKs

Loss with cumulative ACKs

Sender sends packets with 100B and seqnos.:

- 100, 200, 300, 400, 500, 600, 700, 800, 900, …

Assume the fifth packet (seqno 500) is lost, but no others

Stream of ACKs will be:

- 200, 300, 400, 500, 500, 500, 500,…

Loss with cumulative ACKs

"Duplicate ACKs" are a sign of an *isolated* loss

- The lack of ACK progress means 500 hasn't been delivered
- Stream of ACKs means some packets are being delivered

Therefore, could trigger resend upon receiving k duplicate ACKs

- TCP uses k=3

We will revisit this in congestion control

Communication Networks

Spring 2020

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April 6 2020