

Communication Networks

Prof. Laurent Vanbever

Communication Networks

Spring 2019



Laurent Vanbever
nsg.ee.ethz.ch

ETH Zürich (D-ITET)
April 8 2019

Materials inspired from Scott Shenker & Jennifer Rexford

Last week on
Communication Networks

Border Gateway Protocol policies and more



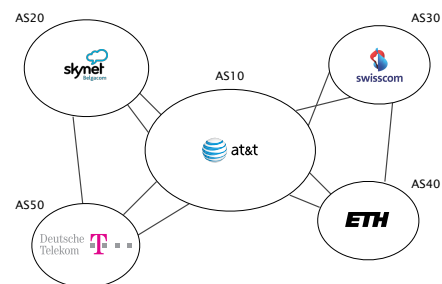
- 1 **BGP Policies**
Follow the Money
- 2 **Protocol**
How does it work?
- 3 **Problems**
security, performance, ...

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:

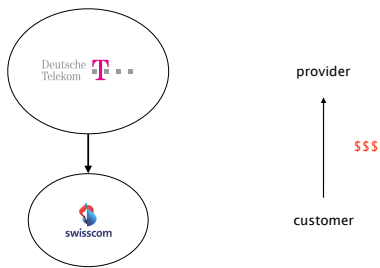
- customer/provider
- peer/peer

many less important ones (siblings, backups,...)

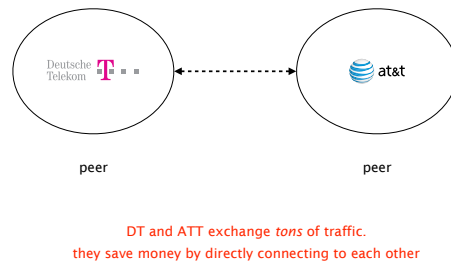
There are 2 main business relationships today:

- **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*

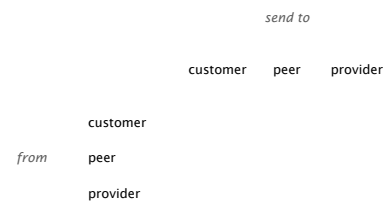


Business relationships conditions
route selection

For a destination p , prefer routes coming from

- customers over
 - peers over
 - providers
- route type

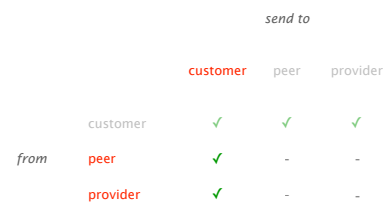
Business relationships conditions
route exportation



Routes coming from customers
are propagated to everyone else



Routes coming from peers and providers
are only propagated to customers



Border Gateway Protocol
policies and more

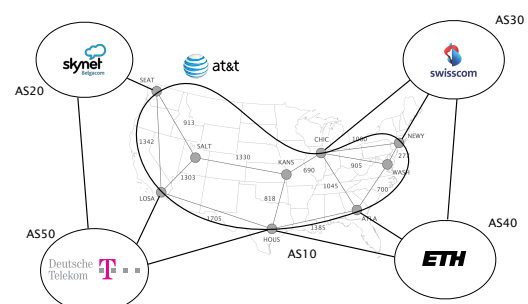


BGP Policies
Follow the Money

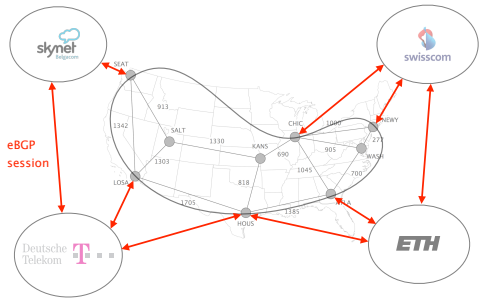
2 Protocol
How does it work?

Problems
security, performance, ...

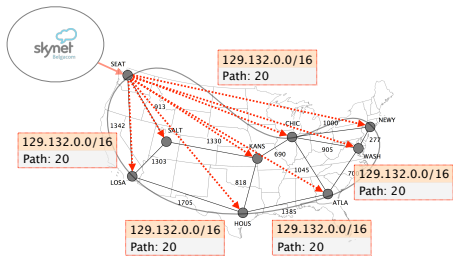
BGP sessions come in two flavors



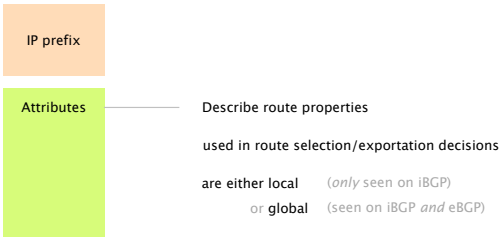
external BGP (eBGP) sessions
connect border routers in different ASes



iBGP sessions are used to disseminate
externally-learned routes internally



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



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)

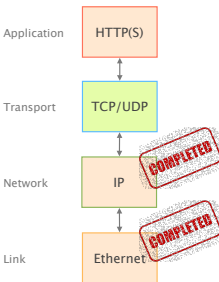
Border Gateway Protocol policies and more



- 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**



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):

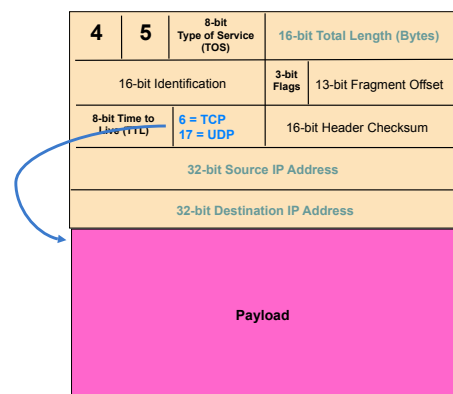
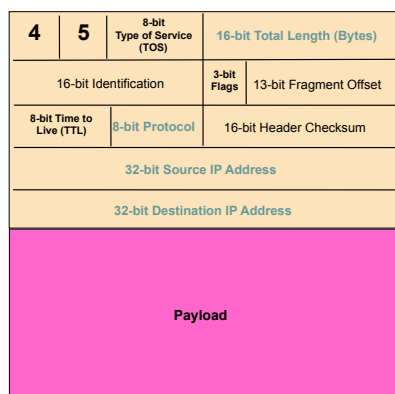
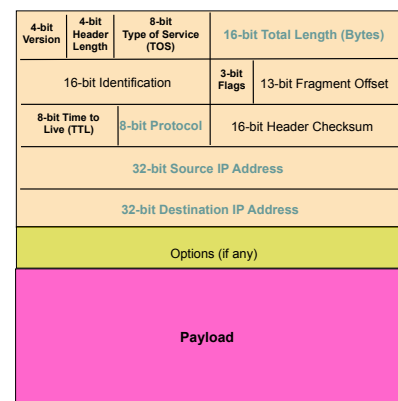
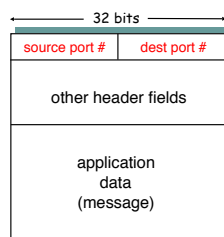
- Given to clients (at random)

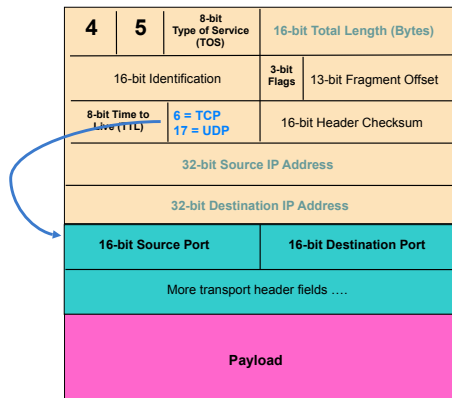
Multiplexing and Demultiplexing

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**





Connection Mappings

For UDP ports (SOCK_DGRAM)

- OS stores (local port, local IP address) \leftrightarrow socket

For TCP ports (SOCK_STREAM)

- OS stores (local port, local IP, remote port, remote IP) \leftrightarrow socket

Why the difference?

Implications for mobility

Why do you need to include local IP?

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

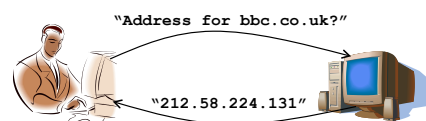
Popular Applications That Use UDP

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

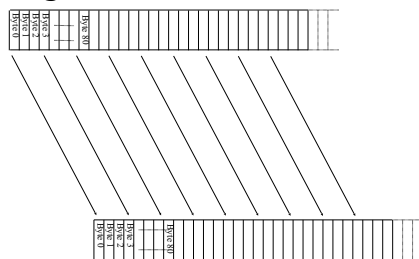
TCP Header

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...

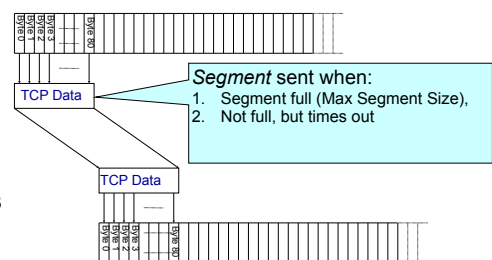
Application @ Host A



Application @ Host B

... Provided Using TCP "Segments"

Host A



Host B

TCP Segment



IP packet

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

TCP packet

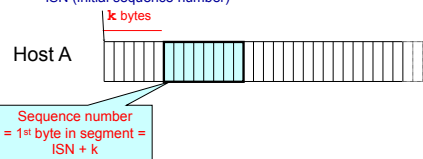
- IP packet with a TCP header and data inside
- TCP header ≥ 20 bytes long

TCP segment

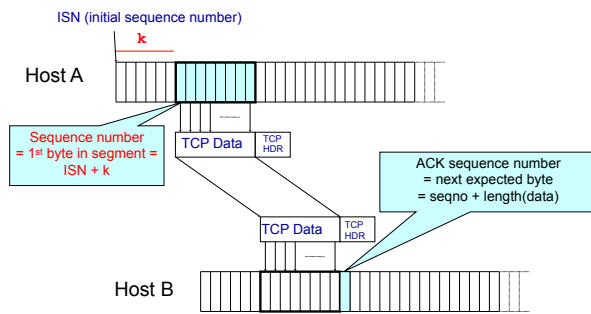
- No more than Maximum Segment Size (MSS) bytes
- E.g., up to 1460 consecutive bytes from the stream
- MSS = MTU - (IP header) - (TCP header)

Sequence Numbers

ISN (initial sequence number)



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 **slow** 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

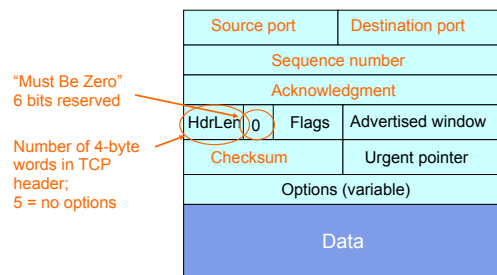
Sender: window **advances** when new data ack'd

Receiver: window advances as receiving process **consumes** data

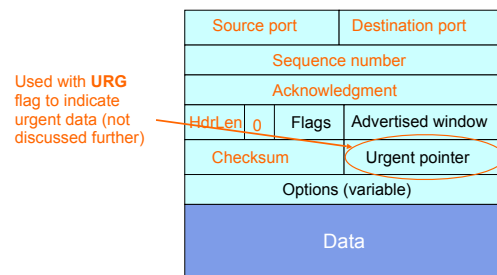
Receiver **advertises** to the sender where the receiver window currently ends ("righthand edge")

- 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

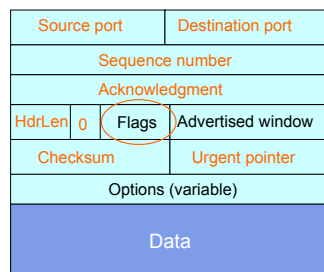
TCP Header: What's left?



TCP Header: What's left?



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**

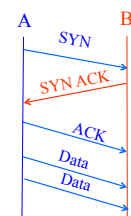
TCP therefore **requires** changing ISN

- 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

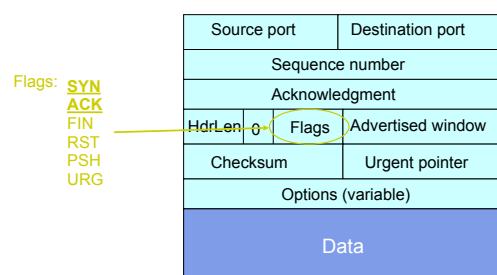


Each host tells its ISN to the other host.

Three-way handshake to establish connection

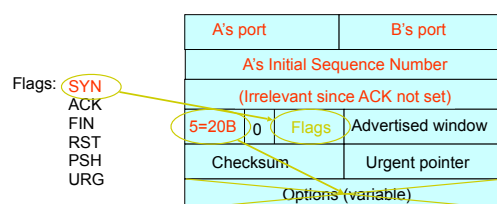
- Host A sends a **SYN** (open; "synchronize sequence numbers")
- Host B returns a SYN acknowledgment (**SYN ACK**)
- Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header



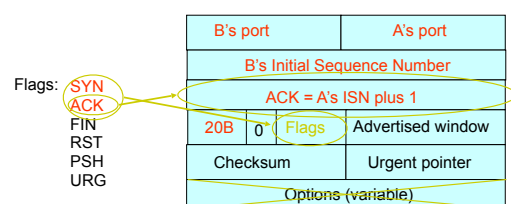
See `/usr/include/netinet/tcp.h` on Unix Systems

Step 1: A's Initial SYN Packet



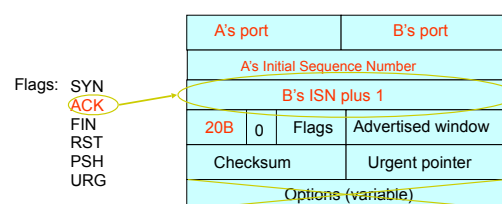
A tells B it wants to open a connection...

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

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...

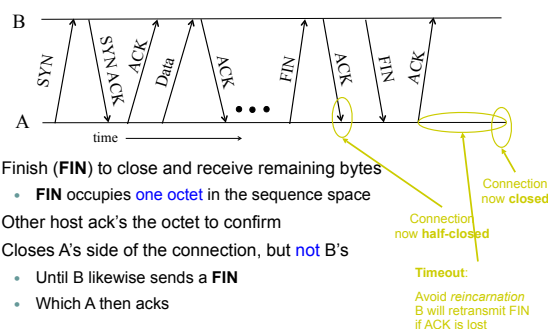
- 3-6 seconds of delay: can be **very long**
- User may become impatient
- ... and click the hyperlink again, or click "reload"

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

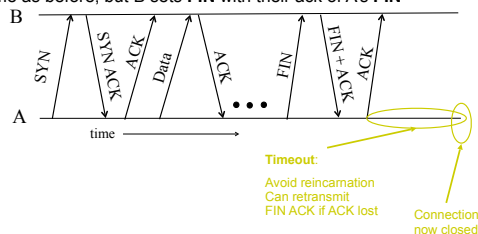
Other host ack's the octet to confirm

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

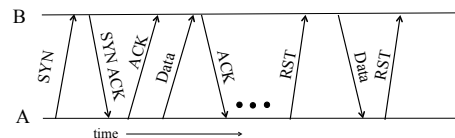
- Until B likewise sends a **FIN**
- Which A then acks

Normal Termination, Both Together

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



Abrupt Termination



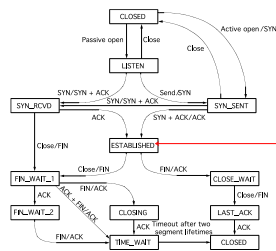
A sends a RESET (**RST**) to B

- 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



Reliability: TCP Retransmission

Timeouts and Retransmissions

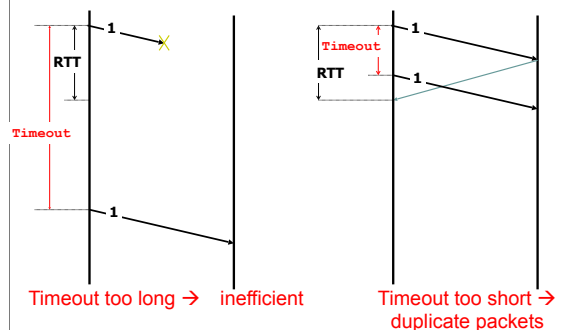
Reliability requires retransmitting lost data

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



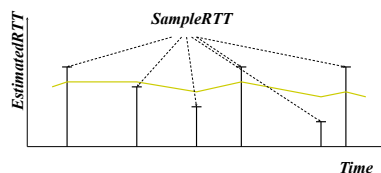
RTT Estimation

Use exponential averaging of RTT samples

$$\text{SampleRTT} = \text{AckRcvdTime} - \text{SendPacketTime}$$

$$\text{EstimatedRTT} = \alpha \times \text{EstimatedRTT} + (1 - \alpha) \times \text{SampleRTT}$$

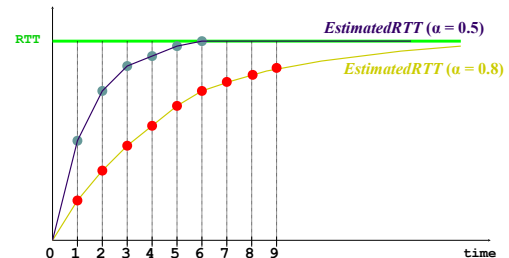
$$0 < \alpha \leq 1$$



Exponential Averaging Example

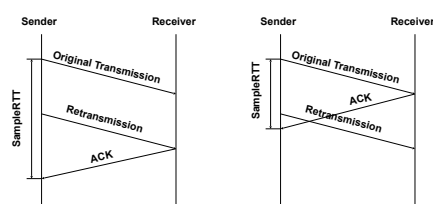
$$\text{EstimatedRTT} = \alpha \times \text{EstimatedRTT} + (1 - \alpha) \times \text{SampleRTT}$$

Assume RTT is constant \rightarrow $\text{SampleRTT} = \text{RTT}$



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) = $2 \times \text{EstimatedRTT}$

Use exponential backoff for repeated retransmissions

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

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, ...

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 2019



Laurent Vanbever
nsg.ee.ethz.ch

ETH Zürich (D-ITET)
April 8 2019