Communication Networks

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Communication Networks

Spring 2018

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April 23 2018

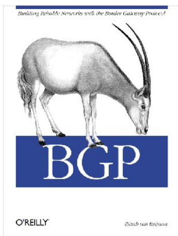
Materials inspired from Scott Shenker & Jennifer Rexford

Last week on

Communication Networks

Border Gateway Protocol

policies and more



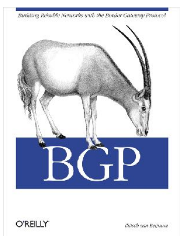
1 BGP Policies
Follow the Money

2 Protocol
How does it work?

3 Problems
security, performance, …

Border Gateway Protocol

policies and more

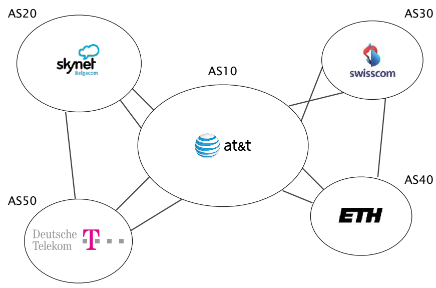


1 BGP Policies
Follow the Money

Protocol
How does it work?

Problems
security, performance, …

The Internet topology is shaped according to *business* relationships



There are 2 main business relationships today:

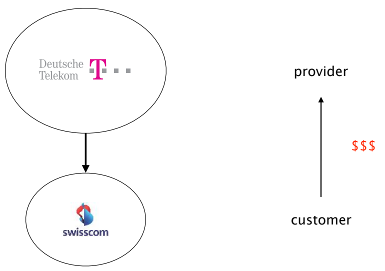
- customer/provider
- peer/peer

many less important ones (siblings, backups,…)

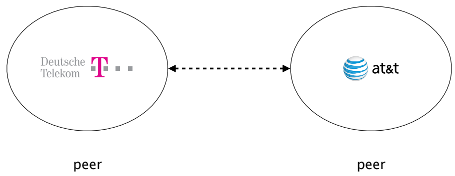
There are 2 main business relationships today:

- 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*



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

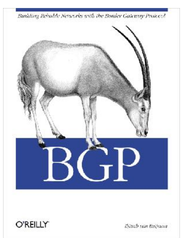
		customer	peer	provider
	customer	✓	✓	✓
from	peer			
	provider			

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

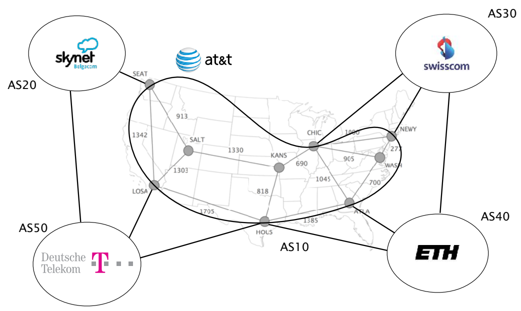


BGP Policies
Follow the Money

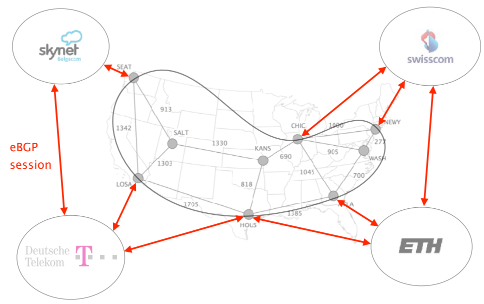
2 Protocol
How does it work?

Problems
security, performance, …

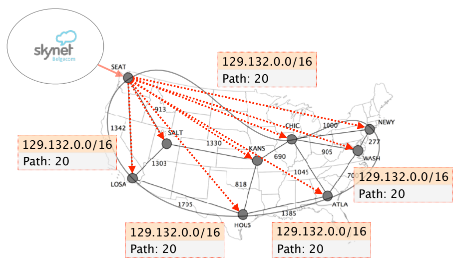
BGP sessions come in two flavors



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

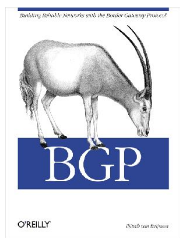
with lower MED

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



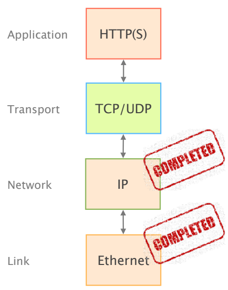
BGP Policies
Follow the Money

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



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):

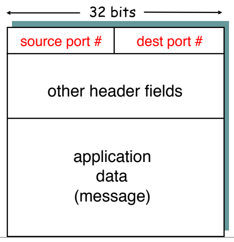
- Given to clients (at random)

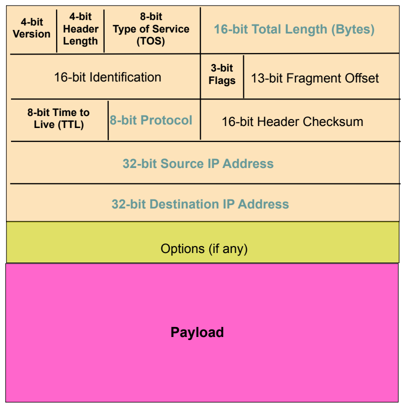
Multiplexing and Demultiplexing

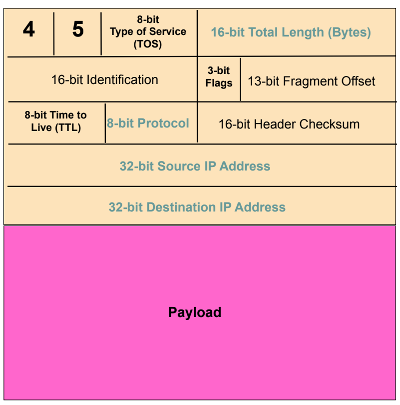
Host receives IP datagrams

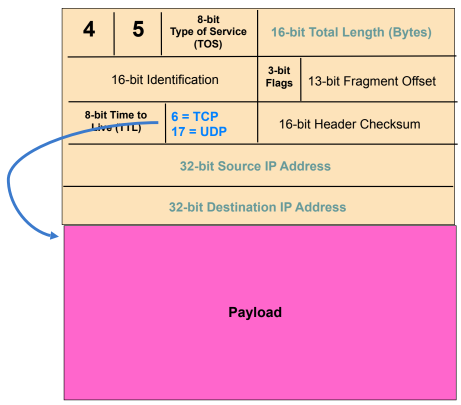
- Each datagram has source and destination IP address,
- Each segment has source and destination port number

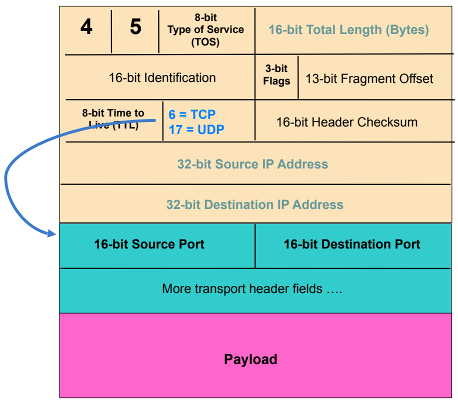
Host uses IP addresses and port numbers to direct the segment to appropriate socket











Connection Mappings

For UDP ports (SOCK_DGRAM)

- OS stores (local port, local IP address) ←→ socket

For TCP ports (SOCK_STREAM)

- OS stores (local port, local IP, remote port, remote IP) ←→ 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

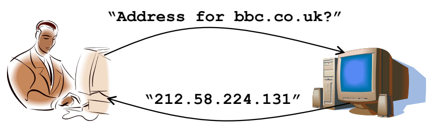
Popular Applications That Use UDP

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

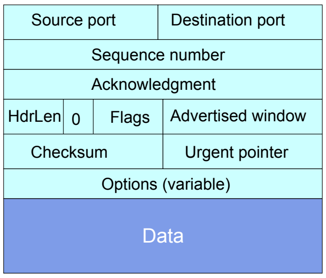
Sliding window flow control
- Allow W contiguous bytes to be in flight

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

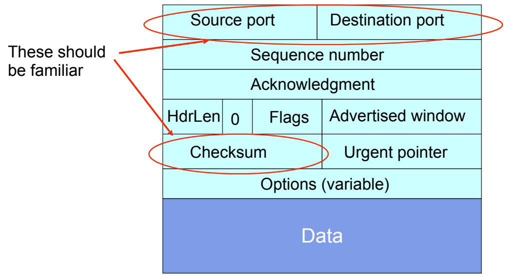
Single timer set after each payload is ACKed
- Timer is effectively for the "next expected payload"
- When timer goes off, resend that payload and wait
  - And double timeout period

Various tricks related to "fast retransmit"
- Using duplicate ACKs to trigger retransmission

TCP Header



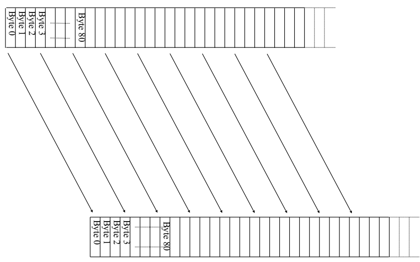
TCP Header



Segments and Sequence Numbers

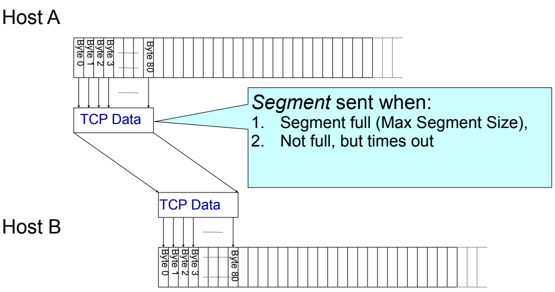
TCP "Stream of Bytes" Service…

Application @ Host A

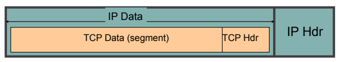


Application @ Host B

… Provided Using TCP "Segments"



TCP Segment

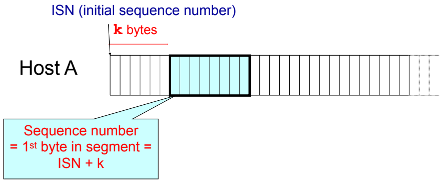


IP **packet**
- No bigger than Maximum Transmission Unit (MTU)
- E.g., up to 1500 bytes with Ethernet

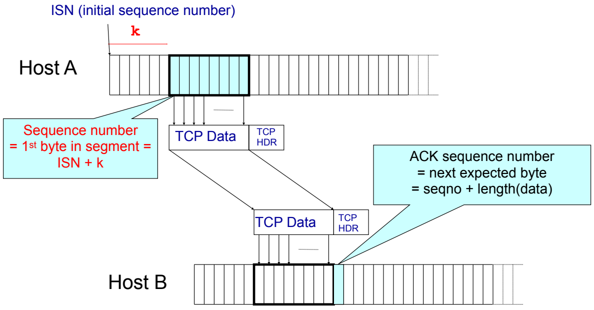
TCP **packet**
- IP packet with a TCP header and data inside
- TCP header ≥ 20 bytes long

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

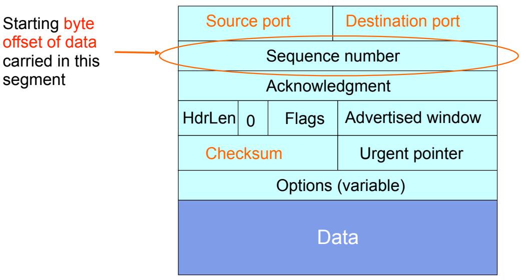
Sender: seqno=X, length=B
Receiver: ACK=X+B

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



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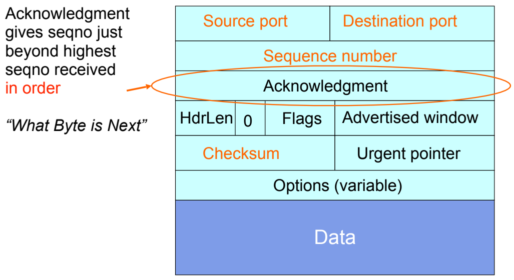

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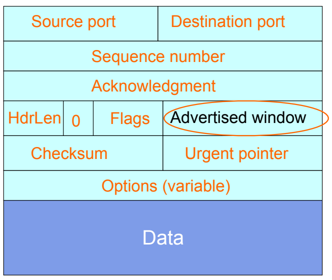


Sliding Window Flow Control

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

Filling the Pipe

Simple example:

- W (in bytes), which we assume is constant
- RTT (in sec), which we assume is constant
- B (in **bytes**/sec)

How fast will data be transferred?

If W/RTT < B, the transfer has speed W/RTT

If W/RTT > B, the transfer has speed B

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

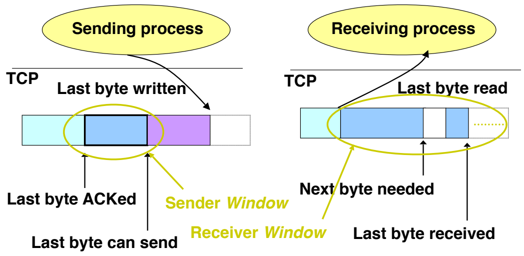
For the receiver:

- Window size = maximum amount of undelivered data

Sliding Window

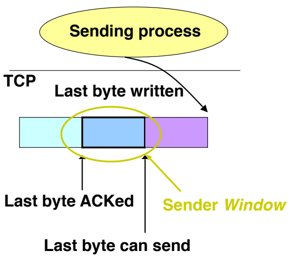
Allow a larger amount of data "in flight"

- Allow sender to get ahead of the receiver
- … though not *too far* ahead



Sliding Window

For the sender, when receives an acknowledgment for new data, window advances (*slides* forward)



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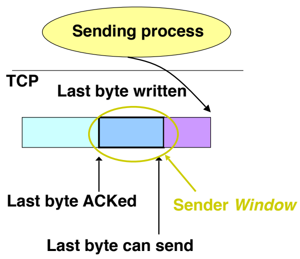
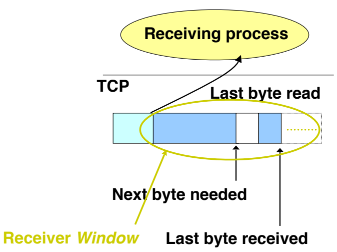


Sliding Window

For the receiver, as the receiving process consumes data, the window slides forward



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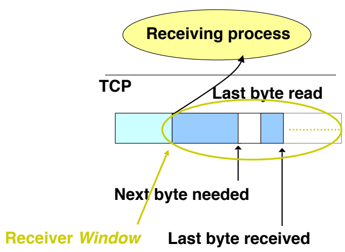


Sliding Window Summary

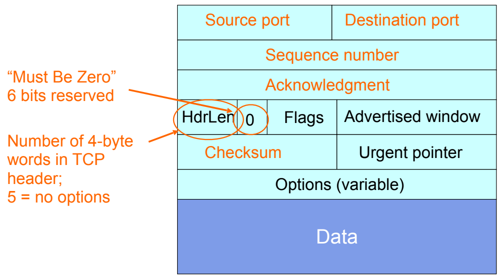
Sender: window advances when new data ack'd

Receiver: window advances as receiving process consumes data

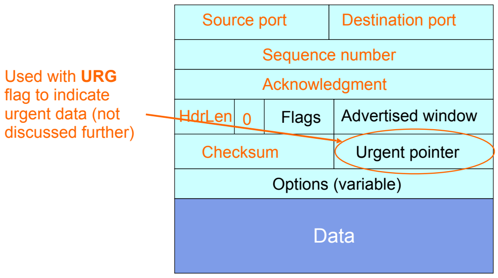
Receiver advertises to the sender where the receiver window currently ends ("righthand edge")

- 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?



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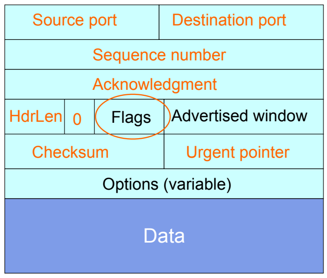


TCP Connection Establishment and Initial Sequence Numbers

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

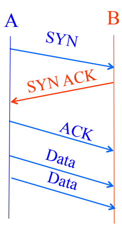
TCP therefore requires changing ISN

- 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

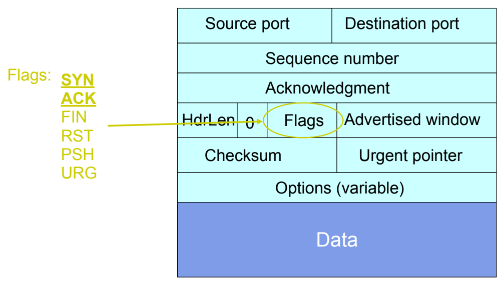


Each host tells its ISN to the other host.

Three-way handshake to establish connection

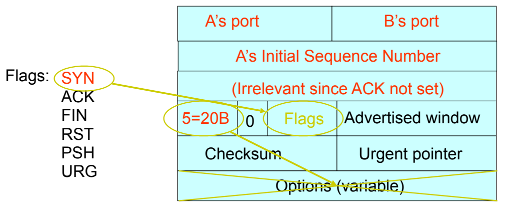
- Host A sends a **SYN** (open; "synchronize sequence numbers")
- Host B returns a SYN acknowledgment (**SYN ACK**)
- Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header



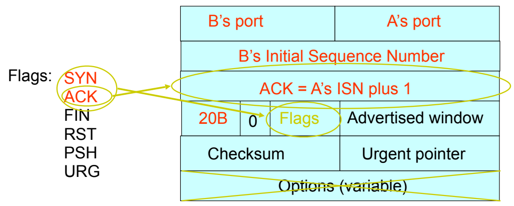
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

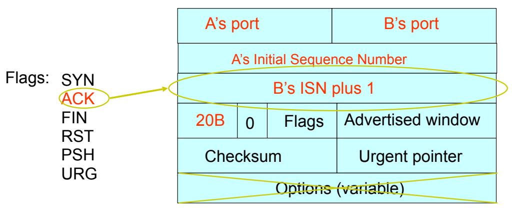
Step 2: B's SYN-ACK Packet



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

… upon receiving this packet, A can start sending data

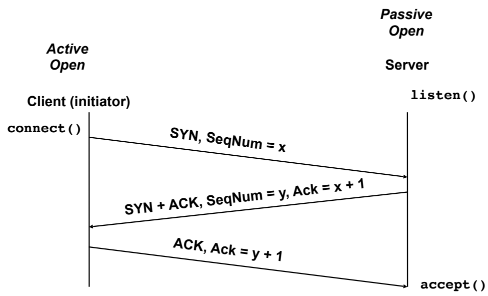
Step 3: A's ACK of the SYN-ACK



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

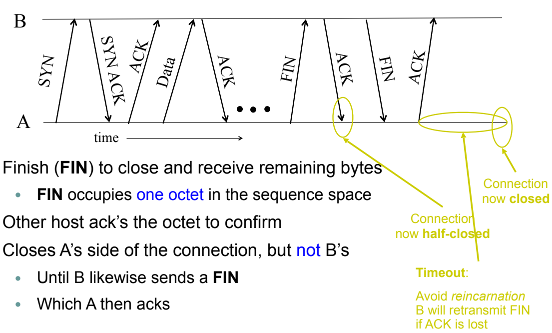
User clicks on a hypertext link
- Browser creates a socket and does a "connect"
- The "connect" triggers the OS to transmit a SYN

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



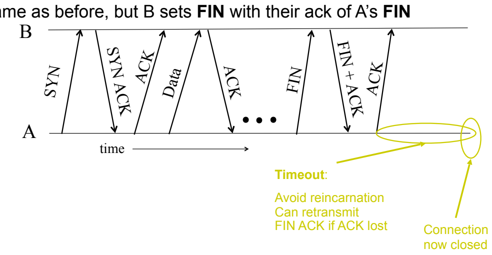
Finish (**FIN**) to close and receive remaining bytes
- **FIN** occupies one octet in the sequence space

Other host ack's the octet to confirm

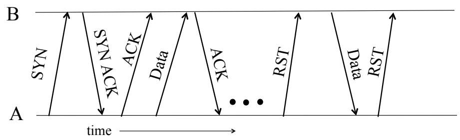
Closes A's side of the connection, but not B's
- Until B likewise sends a **FIN**
- Which A then acks

Normal Termination, Both Together

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



Abrupt Termination



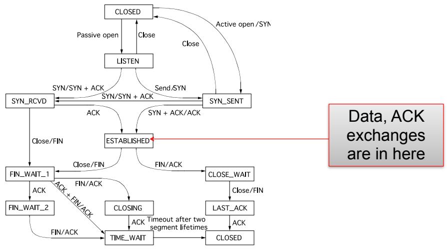
A sends a RESET (**RST**) to B

- 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



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

Example

Arriving ACK expects 100

Sender sends packets 100, 200, 300, 400, 500

- Timer set for 100

Arriving ACK expects 300

- Timer set for 300

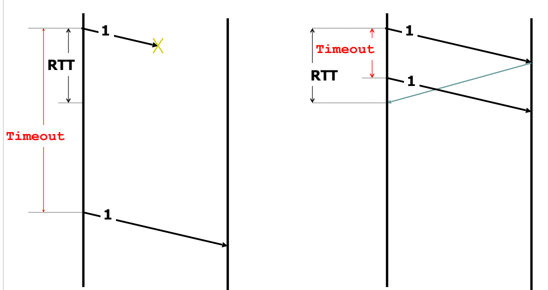
Timer goes off

- Packet 300 is resent

Arriving ACK expects 600

- Packet 600 sent
- Timer set for 600

Setting the Timeout Value



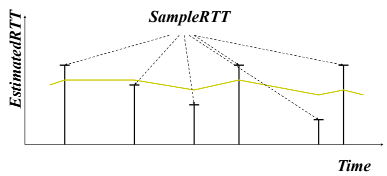
Timeout too long → inefficient

Timeout too short → duplicate packets

RTT Estimation

Use exponential averaging of RTT samples

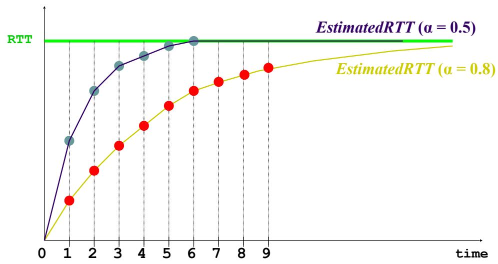
$$SampleRTT = AckRcvdTime - SendPacketTime$$
$$EstimatedRTT = \alpha \times EstimatedRTT + (1-\alpha) \times SampleRTT$$
$$0 < \alpha \le 1$$



Exponential Averaging Example

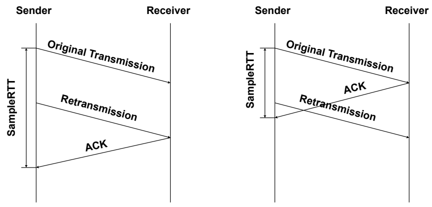
EstimatedRTT = α**EstimatedRTT* + (1 – α)**SampleRTT*

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 α = 0.875

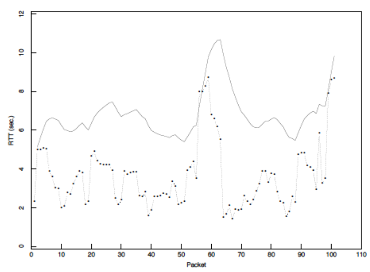
Timeout value (RTO) = 2 × *EstimatedRTT*

Use exponential backoff for repeated retransmissions

- Every time RTO timer expires, set RTO ← 2·RTO
  - (Up to maximum ≥ 60 sec)
- Every time new measurement comes in (= successful original transmission), collapse RTO back to 2 × EstimatedRTT

Karn/Partridge in action

Figure 5: Performance of an RFC793 retransmit timer



from Jacobson and Karels, SIGCOMM 1988

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 2018

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April 23 2018