Communication Networks

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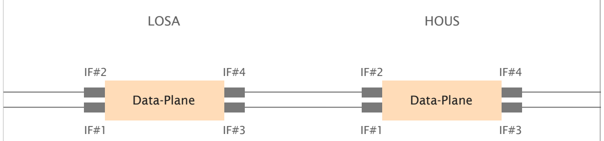
Material inspired from Scott Shenker & Jennifer Rexford

Last week on
Communication Networks

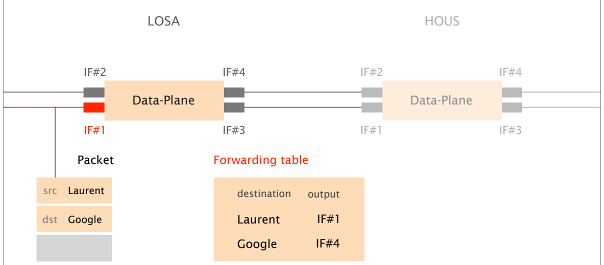
We explored the concepts behind routing

Last week	This week
routing	reliable delivery

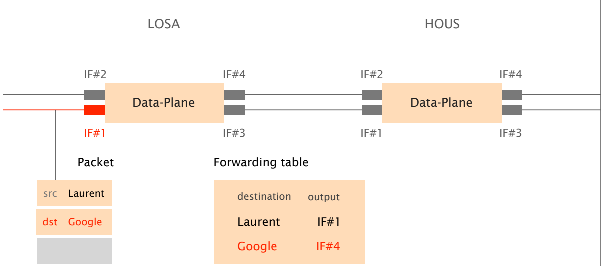
How do you guide IP packets
from a source to destination?

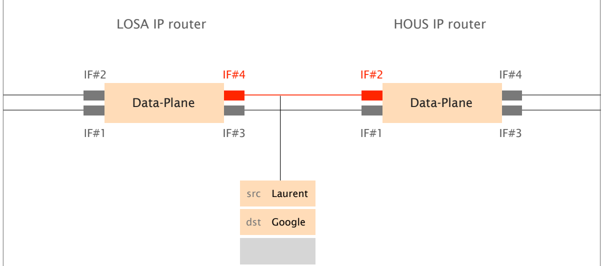


Upon packet reception, routers locally look up their forwarding table to know where to send it next

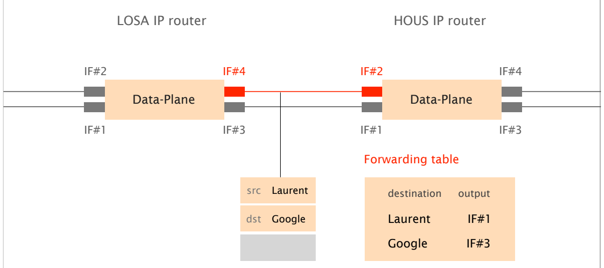


Here, the packet should be directed to IF#4

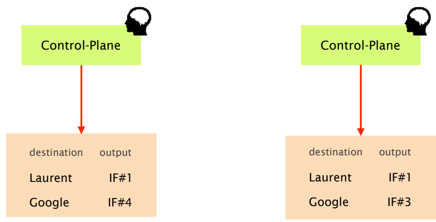




Forwarding is repeated at each router, until the destination is reached



Routing is the control-plane process that computes and populates the forwarding tables



Forwarding *vs* Routing

summary

	forwarding	routing
goal	directing packet to an outgoing link	computing the paths packets will follow
scope	local	network-wide
implem.	hardware usually	software always
timescale	nanoseconds	10s of ms hopefully

The goal of routing is to compute valid global forwarding state

definition

a global forwarding state is valid if

it always delivers packets
to the correct destination

sufficient and necessary condition

Theorem a global forwarding state is valid if and only if

- there are no dead ends
  i.e. no outgoing port defined in the table
- there are no loops
  i.e. packets going around the same set of nodes

observation 1 Verifying that a forwarding state is valid is easy

observation 2 There are 3 ways to compute valid forwarding state

There are three ways to compute valid routing state

	Intuition	Example
#1	Use tree-like topologies	Spanning-tree
#2	Rely on a global network view	Link-State SDN
#3	Rely on distributed computation	Distance-Vector BGP

This week on
Communication Networks

Reliable Transport

1 Correctness condition
if-and-only if again

2 Design space
timeliness *vs* efficiency *vs* …

3 Examples
Go-Back-N & Selective Repeat

Reliable Transport

1 Correctness condition
if-and-only if again

Design space
timeliness *vs* efficiency *vs* …

Examples
Go-Back-N & Selective Repeat

In the Internet, reliability is ensured by the end hosts, not by the network

Internet layer puts reliability in L4, just above the Network layer

goals	Keep the network simple, dumb make it relatively "easy" to build and operate a network
	Keep applications as network "unaware" as possible a developer should focus on its app, not on the network
design	Implement reliability in-between, in the networking stack relieve the burden from both the app and the network

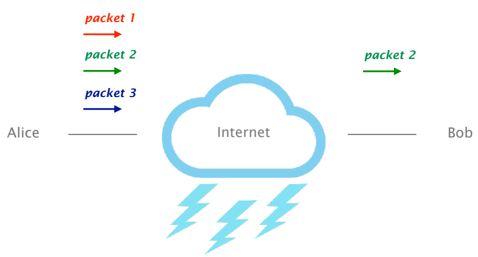
Internet layer puts reliability in L4, just above the Network layer

	layer	
	Application	
L4	Transport	reliable end-to-end delivery
L3	Network	global best-effort delivery
	Link	
	Physical	

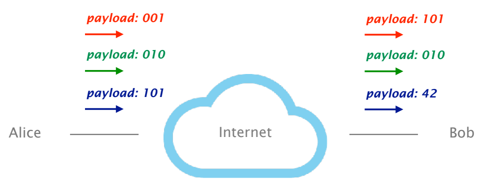
Recall that the Network provides a best-effort service, with quite poor guarantees

	layer	
	Application	
L4	Transport	reliable end-to-end delivery
L3	Network	global best-effort delivery
	Link	
	Physical	

IP packets can get lost or delayed

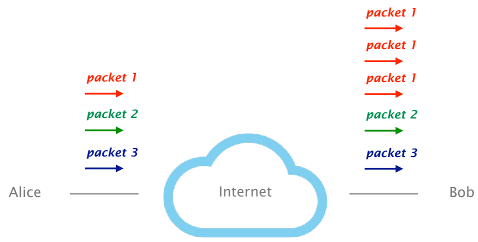


IP packets can get corrupted



IP packets can get reordered

IP packets can get duplicated



The four goals of reliable transfer

goals	
correctness	ensure data is delivered, in order, and untouched
timeliness	minimize time until data is transferred
efficiency	optimal use of bandwidth
fairness	play well with concurrent communications

goals	
correctness	ensure data is delivered, in order, and untouched

Routing had a clean sufficient and necessary correctness condition

sufficient and necessary condition

Theorem a global forwarding state is valid if and only if

- there are no dead ends
  no outgoing port defined in the table
- there are no loops
  packets going around the same set of nodes

We need the same kind of "if and only if" condition for a "correct" reliable transport design

A reliable transport design is correct if…

attempt #1 packets are delivered to the receiver

Wrong Consider that the network is partitioned

We cannot say a transport design is *incorrect*
if it doesn't work in a partitioned network…

A reliable transport design is correct if…

attempt #2 packets are delivered to receiver if and only if it was possible to deliver them

Wrong If the network is only available one instant in time, only an oracle would know when to send

We cannot say a transport design is *incorrect*
if it doesn't know the unknowable

A reliable transport design is correct if…

attempt #3

It resends a packet if and only if
the previous packet was lost or corrupted

Wrong

Consider two cases

- packet made it to the receiver and
  all packets from receiver were dropped
- packet is dropped on the way and
  all packets from receiver were dropped

A reliable transport design is correct if…

attempt #3

It resends a packet if and only if
the previous packet was lost or corrupted

Wrong

In both case, the sender has no feedback at all
Does it resend or not?

A reliable transport design is correct if…

attempt #3

It resends a packet if and only if
the previous packet was lost or corrupted

Wrong

but better as it refers to what the design does (which it can control),
not whether it always succeeds (which it can't)

A reliable transport design is correct if…

attempt #4

A packet is always resent if
the previous packet was lost or corrupted

A packet may be resent at other times

Correct!

A transport mechanism is correct
if and only if it resends all dropped or corrupted packets

Sufficient
"if"

algorithm will always keep trying
to deliver undelivered packets

Necessary
"only if"

if it ever let a packet go undelivered
without resending it, it isn't reliable

Note

it is ok to give up after a while but
must announce it to the application

Reliable Transport

Correctness condition
if-and-only if again

2 Design space
timeliness *vs* efficiency *vs* …

Examples
Go-Back-N & Selective Repeat

Now, that we have a correctness condition
how do we achieve it and with what tradeoffs?

Design a *correct, timely, efficient* and *fair* transport mechanism
knowing that

packets can get lost → let's focus on these aspects first
corrupted
reordered
delayed
duplicated

You already solved the problem
for the single packet case

Alice

```
for word in list:
    send_packet(word);
    set_timer();

    upon timer going off:
        if no ACK received:
            send_packet(word);
            reset_timer();
    upon ACK:
        pass;
```

Bob

```
receive_packet(p);
if check(p.payload) == p.checksum:
    send_ack();

    if word not delivered:
        deliver_word(word);
else:
    pass;
```

There is a clear tradeoff between timeliness and efficiency in the selection of the timeout value

```
for word in list:
    send_packet(word);
    set_timer();

    upon timer going off:
        if no ACK received:
            send_packet(word);
            reset_timer();
    upon ACK:
        pass
```

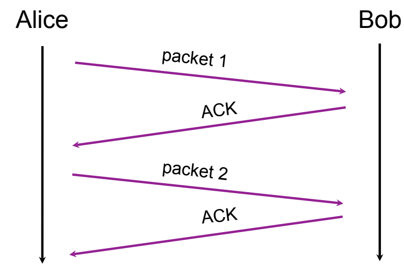
```
receive_packet(p);
if check(p.payload) == p.checksum:
    send_ack();

    if word not delivered:
        deliver_word(word);
else:
    pass;
```

Timeliness argues for small timers,
efficiency for large ones

timeliness	efficiency
small timers	large timers
risk: unnecessary retransmissions	risk: slow transmission

Even with short timers, the timeliness of our protocol is extremely poor: one packet per Round-Trip Time (RTT)



An obvious solution to improve timeliness is
to send multiple packets at the same time

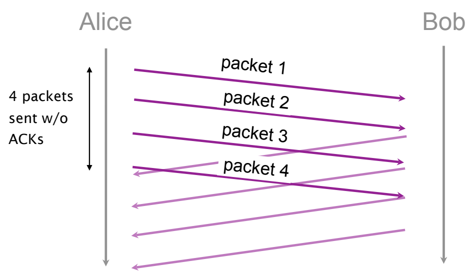
approach

add sequence number inside each packet

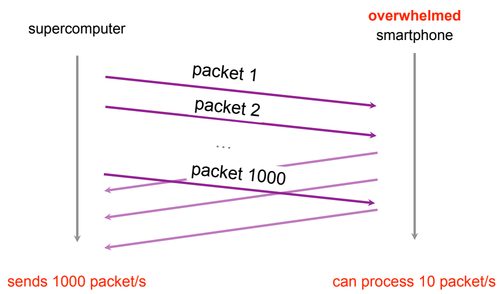
add buffers to the sender and receiver

sender store packets sent & not acknowledged

receiver store out-of-sequence packets received



Sending multiple packets improves timeliness,
but it can also overwhelm the receiver



To solve this issue,
we need a mechanism for flow control

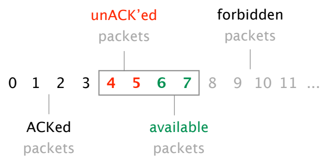
Using **a sliding window** is one way to do that

Sender keeps a list of the sequence # it can send
known as the *sending window*

Receiver also keeps a list of the acceptable sequence #
known as the *receiving window*

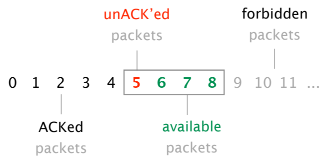
Sender and receiver negotiate the window size
sending window <= *receiving window*

Example with a window composed of 4 packets



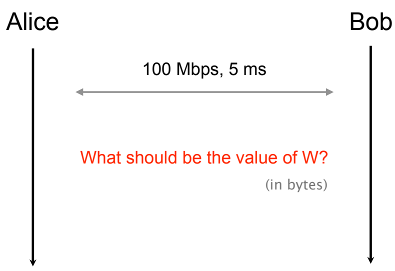
Window after sender receives ACK 4

Window after sender receives ACK 4



Timeliness of the window protocol depends on
the size of the sending window

Assuming infinite buffers,
how **big** should the window be to **maximize timeliness?**



Timeliness matters,
but what about efficiency?

The **efficiency** of our protocol
essentially depends on two factors

receiver feedback	behavior upon losses
How much information does the sender get?	How does the sender detect and react to losses?

The efficiency of our protocol
essentially depends on two factors

receiver feedback	behavior upon losses
How much information does the sender get?	

ACKing individual packets provides detailed feedback, but triggers unnecessary retransmission upon losses

advantages	disadvantages
know fate of each packet	loss of an ACK packet requires a retransmission
simple window algorithm W single-packet algorithms	causes unnecessary retransmission
not sensitive to reordering	

Cumulative ACKs enables to recover from lost ACKs, but provides coarse-grained information to the sender

approach	ACK the highest sequence number for which all the previous packets have been received
advantages	recover from lost ACKs
disadvantages	confused by reordering incomplete information about which packets have arrived causes unnecessary retransmission

Full Information Feedback prevents unnecessary retransmission, but can induce a sizable overhead

approach	List all packets that have been received highest cumulative ACK, plus any additional packets
advantages	complete information resilient form of individual ACKs
disadvantages	overhead (hence lowering efficiency) *e.g.*, when large gaps between received packets

We see that Internet design is
all about balancing tradeoffs (again)

The efficiency of our protocol
essentially depends on two factors

receiver feedback	behavior upon losses
	How does the sender detect and react to losses?

As of now, we detect loss by using timers.
That's only one way though

Losses can also be detected by relying on ACKs

With **individual ACKs**,
missing packets (gaps) are **implicit**

Assume packet 5 is lost
but no other

ACK stream

1
2
3
4
6 — sender can infer that 5 is missing
and resend 5 after k subsequent packets
7
…

With **full information**,
missing packets (gaps) are **explicit**

Assume packet 5 is lost
but no other

ACK stream

up to **1**
up to **2**
up to **3**
up to **4**
up to **4**, plus **6** — sender learns that 5 is missing
retransmits after k packets
up to **4**, plus **6—7**
…

With **cumulative ACKs**,
missing packets are harder to know

Assume packet 5 is lost
but no other

ACK stream

1
2
3
4
4 sent when 6 arrives
4 sent when 7 arrives
…

Duplicated ACKs are a sign of isolated losses.
Dealing with them is trickier though.

situation

Lack of ACK progress means that 5 hasn't made it

Stream of ACKs means that (some) packets are delivered

Sender could trigger resend
upon receiving k duplicates ACKs

but *what* do you resend?
only 5 or 5 and everything after?

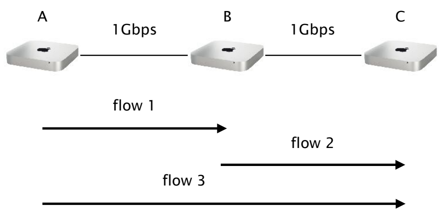
What about fairness?

Design a *correct*, *timely*, *efficient* and *fair* transport mechanism
knowing that

packets can get lost
corrupted
reordered
delayed
duplicated

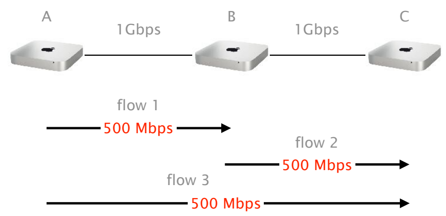
When n **entities** are using our transport mechanism,
we want a **fair allocation of the available bandwidth**

Consider this simple network
in which **three hosts are sharing two links**



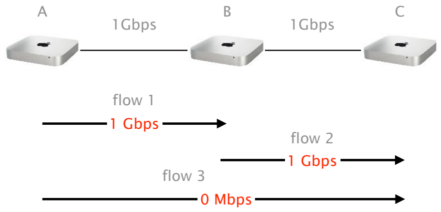
What is a fair allocation for the 3 flows?

An equal allocation is certainly "**fair**",
but what about the **efficiency of the network**?



Total traffic is 1.5 Gbps

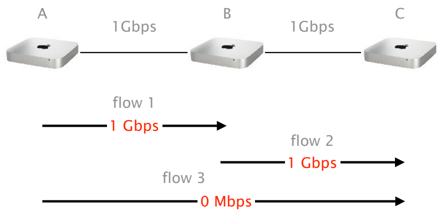
Fairness and efficiency don't always play along, here an unfair allocation ends up *more efficient*



Total traffic is 2 Gbps!

What is fair anyway?

equal per flow isn't really fair as (A,C) crosses two links: it uses *more* resources



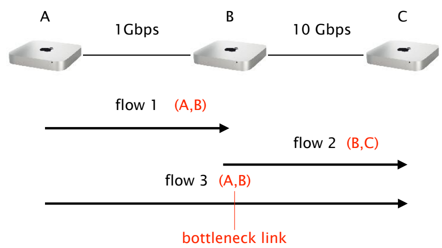
With equal-per-flow, A ends up with 1 Gbps because it sends 2 flows, while B ends up with 500 Mbps

Is it fair?

Seeking an exact notion of fairness is not productive. What matters is to avoid starvation.

equal per flow is good enough for this

Simply dividing the available bandwidth doesn't work in practice since flows can see different bottleneck



Intuitively, we want to give users with "small" demands what they want, and evenly distribute the rest

Max-min fair allocation is such that

the lowest demand is maximized

after the lowest demand has been satisfied,
the second lowest demand is maximized

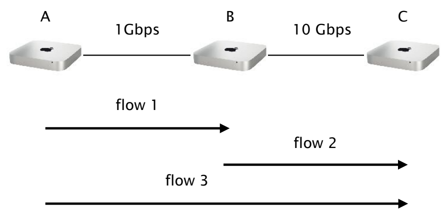
after the second lowest demand has been satisfied,
the third lowest demand is maximized

and so on...

Max-min fair allocation can easily be computed

step 1	Start with all flows at rate 0
step 2	Increase the flows until there is a new bottleneck in the network
step 3	Hold the fixed rate of the flows that are bottlenecked
step 4	Go to step 2 for the remaining flows
	Done!

Let's try on this network



What's the max-min fair allocation?

Max-min fair allocation can be approximated
by slowly increasing *W* until a loss is detected

Intuition	Progressively increase the sending window size	max=receiving window
	Whenever a loss is detected, decrease the window size	signal of congestion
	Repeat	

Design a *correct*, *timely*, *efficient* and *fair* transport mechanism
knowing that

packets can get lost
corrupted
reordered
delayed
duplicated

Dealing with corruption is easy:
Rely on a checksum, treat corrupted packets as lost

The effect of reordering depends on
the type of ACKing mechanism used

individual ACKs	no problem
full feedback	no problem
cumm. ACKs	create duplicate ACKs why is it a problem?

Long delays can create useless timeouts,
for all designs

Packets duplicates can lead to duplicate ACKs whose
effects will depend on the ACKing mechanism used

individual ACKs	no problem
full feedback	no problem
cumm. ACKs	problematic

Design a *correct*, *timely*, *efficient* and *fair* transport mechanism
knowing that

packets can get lost
corrupted
reordered
delayed
duplicated

Here is one correct, timely, efficient and fair transport mechanism

ACKing	full information ACK
retransmission	after timeout
	after *k* subsequent ACKs
window management	additive increase upon successful delivery
	multiple decrease when timeouts

We'll come back to this when we see TCP

Reliable Transport

Correctness condition
if-and-only if again

Design space
timeliness *vs* efficiency *vs* ...

3 Examples
Go-Back-N & Selective Repeat

Go-Back-N (GBN) is a simple sliding window protocol using cumulative ACKs

principle	receiver should be as simple as possible
receiver	delivers packets in-order to the upper layer
	for each received segment, ACK the last in-order packet delivered (cumulative)
sender	use a single timer to detect loss, reset at each ACK
	upon timeout, resend all W packets starting with the lost one

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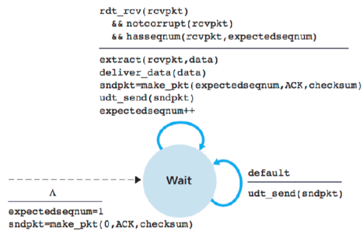
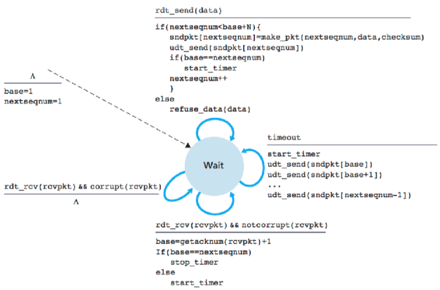
Finite State Machine for the receiver
see Book 3.4.3



Finite State Machine for the sender
see Book 3.4.3



Let's see how it works in practice
visually

http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

Selective Repeat (SR) avoid unnecessary retransmissions by using per-packet ACKs

see Book 3.4.3

principle	avoids unnecessary retransmissions
receiver	acknowledge each packet, in-order or not
	buffer out-of-order packets
sender	use per-packet timer to detect loss
	upon loss, only resend the lost packet

Reliable Transport

Correctness condition
if-and-only if again

Design space
timeliness *vs* efficiency *vs* ...

Examples
Go-Back-N & Selective Repeat