Communication Networks

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Communication Networks

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Materials inspired from Scott Shenker & Jennifer Rexford

Internet Routing Hackathon, Edition 2020

Thursday March 26, 18:00 in ETZ foyer

Register your group (3 students) starting from

Thursday March 5 (see website)

Last week on

Communication Networks

Communication Networks

Part 1: General overview

What is a network made of?

How is it shared?

How is it organized?

#4 How does communication happen?

How do we characterize it?

The Internet should allow

processes on different hosts

to exchange data

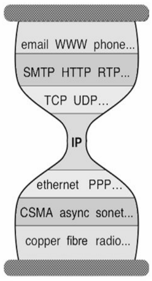
everything else is just commentary…

In practice, there exists **a lot** of network protocols.

How does the Internet organize **this?**

Each layer provides a service to the layer above
by using the services of the layer directly below it

Applications
…built on…
Reliable (or unreliable) transport
…built on…
Best-effort global packet delivery
…built on…
Best-effort local packet delivery
…built on…
Physical transfer of bits



Communication Networks
Part 1: General overview

What is a network made of?

How is it shared?

How is it organized?

How does communication happen?

#5 How do we characterize it?

A network *connection* is characterized by
its delay, loss rate and throughput

delay

loss

throughput

How long does it take for a packet to reach the destination

What fraction of packets sent to a destination are dropped?

At what rate is the destination receiving data from the source?

This week on
Communication Networks

We will dive in the two fundamental
challenges underlying networking

routing

reliable delivery

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routing

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

reliable delivery

How do you ensure reliable transport
on top of best-effort delivery?

---

routing

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

reliable delivery

---

question 1 How do we verify that a forwarding state is valid?

question 2 How do we compute valid forwarding state?

question 1 How do we verify that a forwarding state is valid?

How do we compute valid forwarding state?

Verifying that a routing state is valid is easy

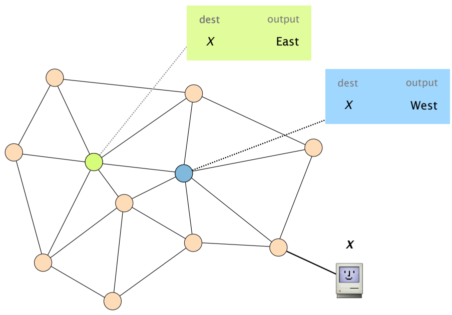
simple algorithm
for one destination

Mark all outgoing ports with an arrow

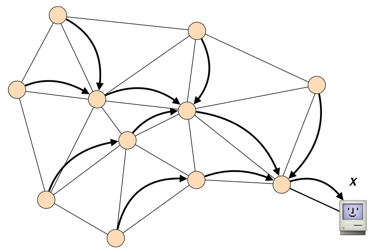
Eliminate all links with no arrow

State is valid *iff* the remaining graph is a spanning-tree

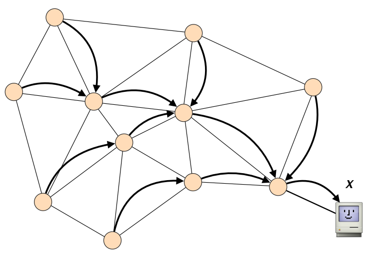
Given a graph with the corresponding forwarding state

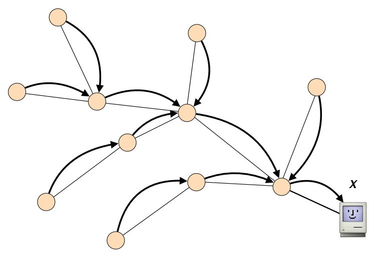


Mark all outgoing ports with an arrow

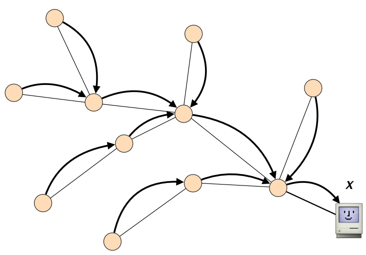


Eliminate all links with no arrow

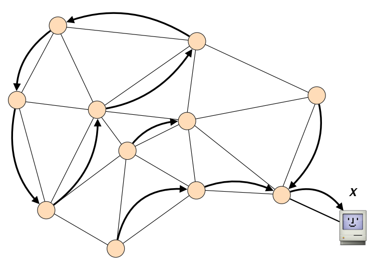




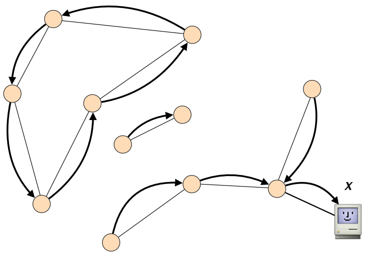
The result is a spanning tree.
This is a valid routing state



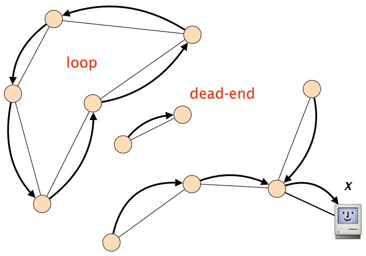
Mark all outgoing ports with an arrow



Eliminate all links with no arrow



The result is not a spanning-tree.
The routing state is not valid



How do we verify that a forwarding state is valid?

question 2 How do we compute valid forwarding state?

Producing valid routing state is harder

prevent dead ends	prevent loops
easy	hard

Producing valid routing state is harder
but doable

prevent dead ends	prevent loops
easy	hard

This is the question
you should focus on

Existing routing protocols differ in
how they avoid loops

prevent loops
hard

Before I give you all the answers
it's your turn

...to figure out a way to route traffic in a network

instructions given in class

Essentially,
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

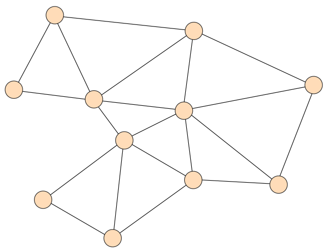
Essentially,
there are three ways to compute valid routing state

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

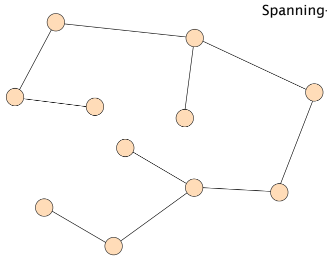
The easiest way to **avoid loops** is to route traffic
on a loop-free topology

simple algorithm	Take an arbitrary topology
	Build a spanning tree and ignore all other links
	Done!
Why does it work?	Spanning-trees have only one path between any two nodes

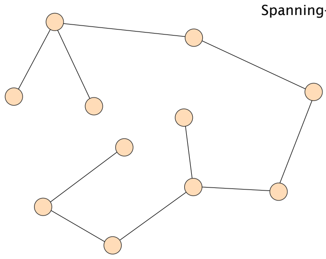
In practice,
there can be ***many*** spanning-trees for a given topology



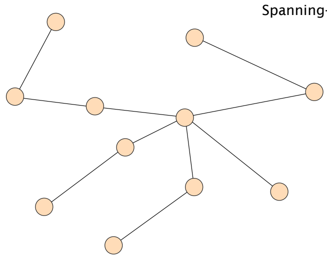
Spanning-Tree #1



Spanning-Tree #2



Spanning-Tree #3

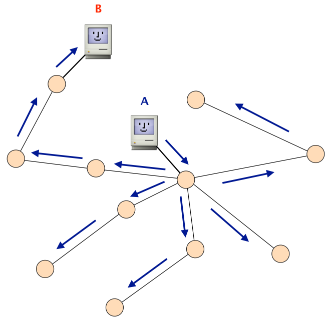


We'll see how to compute spanning-trees in 2 weeks.
For now, **assume it is possible**

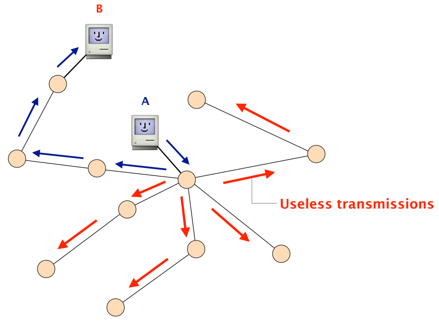
Once we have a spanning tree,
forwarding on it is **easy**

literally just flood
the packets everywhere

When a packet arrives,
simply send it on all ports



While flooding works,
it is quite wasteful



The issue is that nodes do not know their
respective locations

Nodes can learn how to reach nodes
by remembering where packets came from

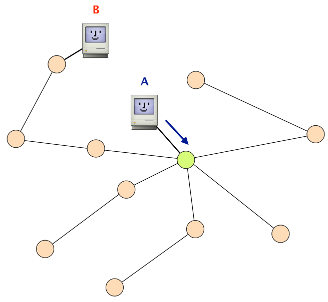
intuition

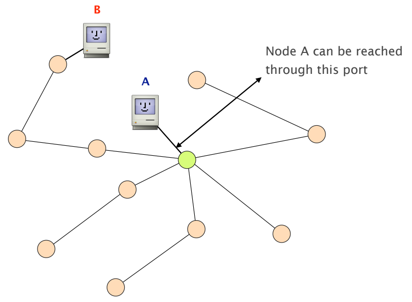
if

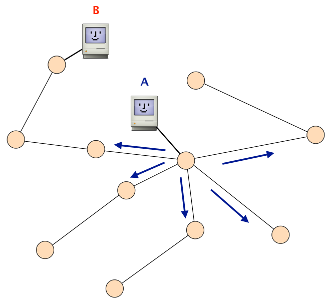
flood packet from node *A*
entered switch *X* on port *4*

then

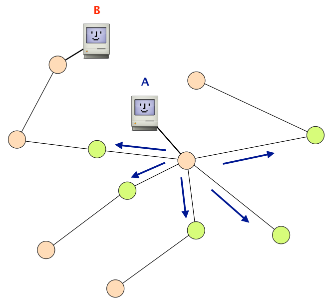
switch *X* can use port *4*
to reach node *A*







All the green nodes learn how to reach A



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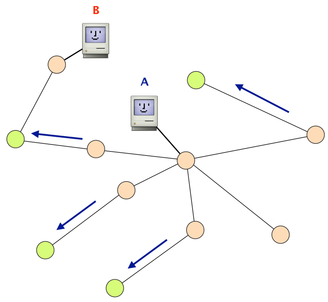
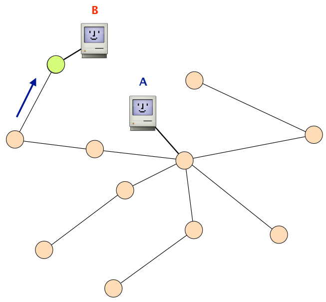
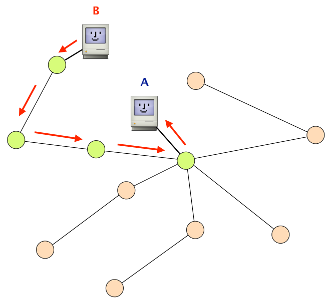


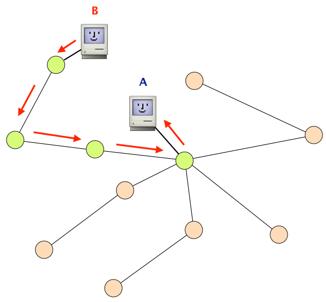
All the nodes know on which port
A can be reached



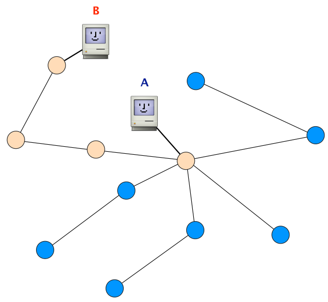
B answers back to A
enabling the green nodes to also learn where B is



There is no need for flooding here
as the position of A is already known by everybody



Learning is topology-dependent
The blue nodes only know how to reach A (not B)



Routing by flooding on a spanning-tree
in a nutshell

Flood first packet to node you're trying to reach
all switches learn where you are

When destination answers, some switches learn where it is
some because packet to you is not flooded anymore

The decision to flood or not is done on each switch
depending on who has communicated before

Spanning-Tree in practice
used in Ethernet

advantages	disadvantages
plug-and-play configuration-free	mandate a spanning-tree eliminate many links from the topology
automatically adapts to moving host	slow to react to failures host movement

Essentially,
there are three ways to compute valid routing state

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

If each router knows the entire graph,
it can locally compute paths to all other nodes

Once a node u knows the entire topology,
it can compute shortest-paths using **Dijkstra's algorithm**

Initialization

S = {*u*}
for all nodes *v*:
    if (*v* is adjacent to *u*):
        D(*v*) = c(*u*,*v*)
    else:
        D(*v*) = ∞

Loop

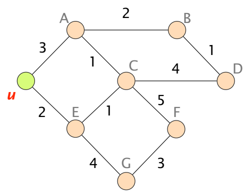
while *not* all nodes in S:
    add *w* with the smallest D(*w*) to S
    update D(*v*) for all adjacent *v* not in S:
        D(*v*) = min{D(*v*), D(*w*) + c(*w*,*v*)}

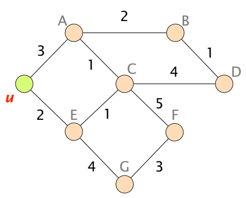
u is the node running the algorithm

S = {*u*}
for all nodes *v*:
    if (*v* is adjacent to *u*):
        D(*v*) = c(*u*,*v*) *c(u,v)* is the weight of the link connecting *u* and *v*
    else:
        D(*v*) = ∞

D(*v*) is the smallest distance currently known by *u* to reach *v*

Let's compute the shortest-paths
from *u*





Initialization

S = {*u*}
for all nodes *v*:
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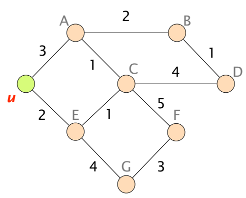
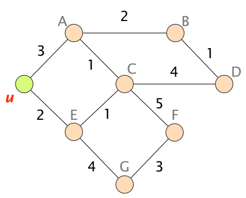
D is initialized based on u's weight,
and S only contains u itself



D(.) =   S = {u}

A	3
B	∞
C	∞
D	∞
E	2
F	∞
G	∞



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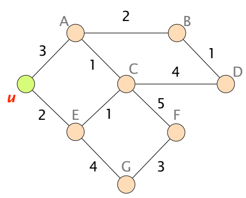
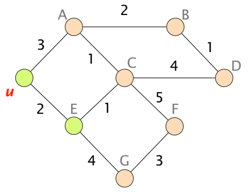


D(.) =   S = {u}

A	3	
B	∞	
C	∞	
D	∞	
E	2	smallest D(*w*)
F	∞	
G	∞	

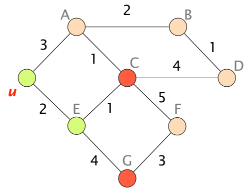
add E to S

D(.) = S = {u, E}



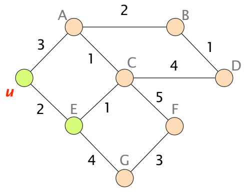
D(.) =	
A	3
B	∞
C	∞
D	∞
E	2
F	∞
G	∞

D(.) = S = {u, E}



D(.) =		
A	3	
B	∞	
C	3	D(*v*) = min{∞, 2 + 1}
D	∞	
E	2	
F	∞	
G	6	D(*v*) = min{∞, 2 + 4}

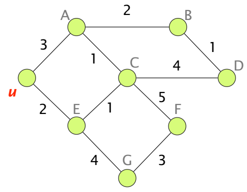
Now, do it by yourself



D(.) = S = {u, E}

D(.) =	
A	3
B	∞
C	3
D	∞
E	2
F	∞
G	6

Here is the final state



D(.) = S = {u, A, B, C, D, E, F,G}

D(.) =	
A	3
B	5
C	3
D	6
E	2
F	8
G	6

This algorithm has a $O(n^2)$ complexity
where n is the number of nodes in the graph

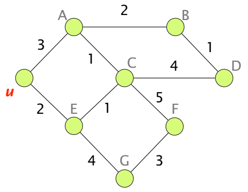
iteration #1	search for minimum through n nodes
iteration #2	search for minimum through n-1 nodes
iteration n	search for minimum through 1 node

$\frac{n(n+1)}{2}$ operations => $O(n^2)$

This algorithm has a $O(n^2)$ complexity
where n is the number of nodes in the graph

Better implementations rely on a heap
to find the next node to expand,
bringing down the complexity to $O(n \log n)$

From the shortest-paths,
u can directly compute its forwarding table

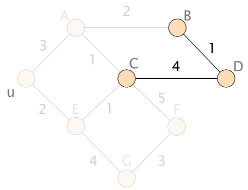


Forwarding table

destination	*next-hop*
A	A
B	A
C	E
D	A
E	E
F	E
G	E

To build this global view
routers essentially solve a jigsaw puzzle

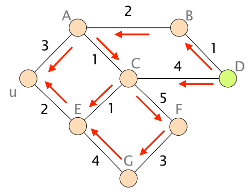
Initially,
routers only know **their ID and their neighbors**



D only knows,
it is connected to B and C

along with the weights to reach them
(by configuration)

Each routers builds a message (known as Link-State)
and floods it (reliably) in the entire network

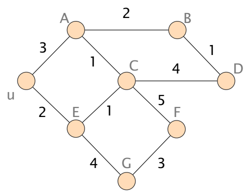


D's Advertisement

edge (D,B); cost: 1
edge (D,C); cost: 4

At the end of the flooding process,
everybody share the exact same view of the network

required for correctness
see exercise



Dijkstra will always converge to a unique stable state
when run on *static* weights

cf. exercice session
for the dynamic case

Essentially,
there are three ways to compute valid routing state

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

Instead of **locally** compute paths based on the graph,
paths can be computed in a distributed fashion

Let $d_x(y)$ be the cost of the least-cost path
known by x to reach y

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Each node bundles these distances
into one message (called a vector)
until convergence
that it repeatedly sends to all its neighbors

Let $d_x(y)$ be the cost of the least-cost path known by x to reach y

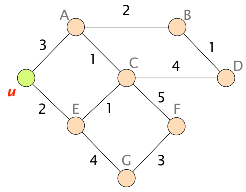
until convergence

Each node bundles these distances into one message (called a vector) that it repeatedly sends to all its neighbors

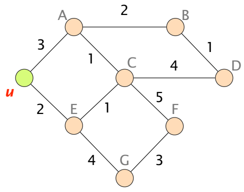
Each node updates its distances based on neighbors' vectors:

$d_x(y) = \min\{ c(x,v) + d_v(y) \}$ over all neighbors v

Let's compute the shortest-path from u to D



The values computed by a node u depends on what it learns from its neighbors (A and E)

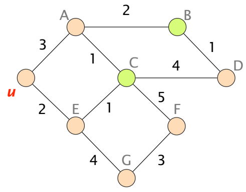


$d_x(y) = \min\{ c(x,v) + d_v(y) \}$

over all neighbors v

$d_u(D) = \min\{ c(u,A) + d_A(D),$
$c(u,E) + d_E(D) \}$

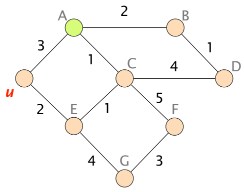
To unfold the recursion, let's start with the direct neighbor of D



$d_B(D) = 1$

$d_C(D) = 4$

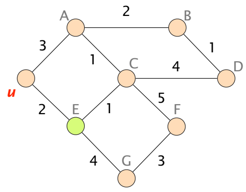
B and C announce their vector to their neighbors, enabling A to compute its shortest-path



$d_A(D) = \min\{ 2 + \overbrace{d_B(D)}^{1},$
$1 + \underbrace{d_C(D)}_{4} \}$

$= 3$

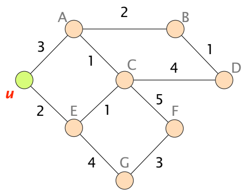
As soon as a distance vector changes, each node propagates it to its neighbor



$d_E(D) = \min\{ 1 + d_C(D),$
$4 + d_G(D),$
$2 + d_u(D) \}$

$= 5$

Eventually, the process converges to the shortest-path distance to each destination



$d_u(D) = \min\{ 3 + d_A(D),$
$2 + d_E(D) \}$

$= 6$

As before, u can directly infer its forwarding table by directing the traffic to the best neighbor

the one which advertised the smallest cost

Evaluating the complexity of DV is harder,
we'll get back to that in a couple of weeks

Communication Networks

Part 2: Concepts

routing

reliable delivery

How do you ensure reliable transport
on top of best-effort delivery?

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

The Internet 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

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

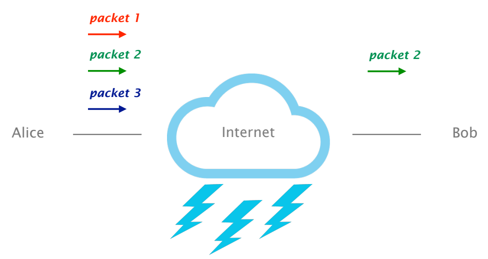
	layer	
	Application	
L4	Transport	reliable end-to-end delivery
L3	Network	global best-effort delivery
	Link	
	Physical	

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

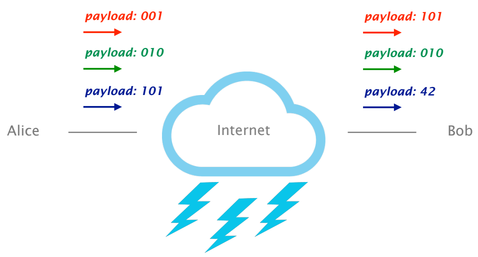
	layer	
	Application	
L4	Transport	reliable end-to-end delivery
L3	Network	global best-effort delivery
	Link	
	Physical	

Let's consider a simple communication
between two end-points, **Alice and Bob**

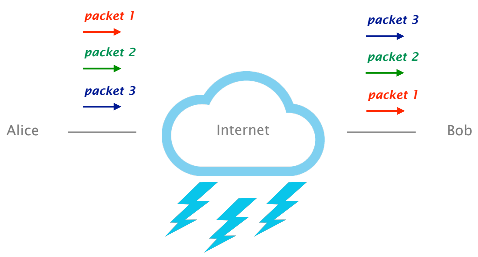
IP packets **can get lost or delayed**



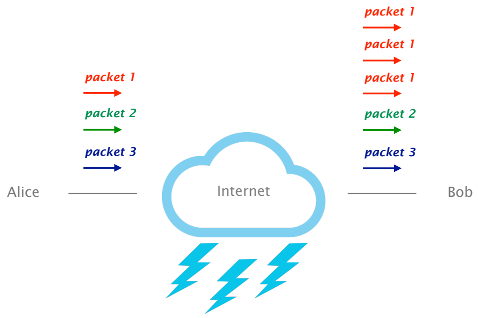
IP packets can get corrupted



IP packets can get reordered



IP packets can get duplicated



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

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

Alice | Bob

```
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;
```

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
```

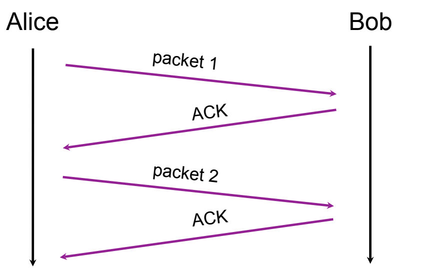
```
receive_packet(p);
if check(p.payload) == p.checksum:
    send_ack();

    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)

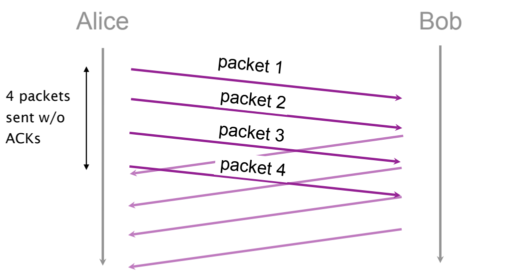


An obvious solution to improve timeliness is to send multiple packets at the same time

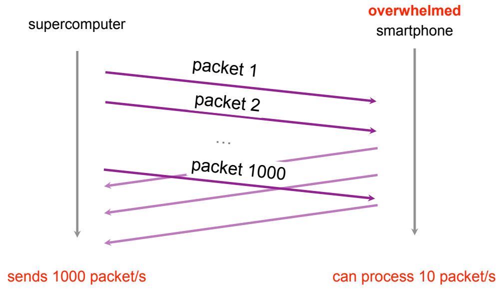
approach: add sequence number inside each packet

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

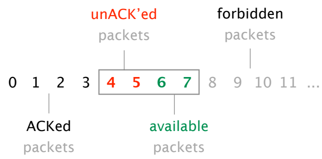
Using **a sliding window** is one way to do that

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

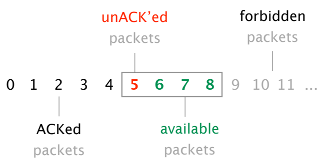
Receiver also keeps a list of the acceptable sequence #
known as the *receiving window*

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

Example with a window composed of 4 packets

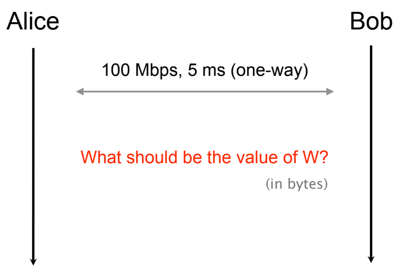


Window after sender receives ACK 4



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	sender can infer that 5 is missing
	6	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
	up to 4, plus 6—7	retransmits after *k* packets
	…	

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?

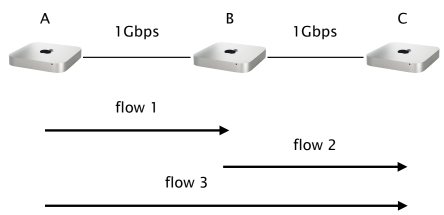
What about fairness?

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

packets can get lost
corrupted
reordered
delayed
duplicated

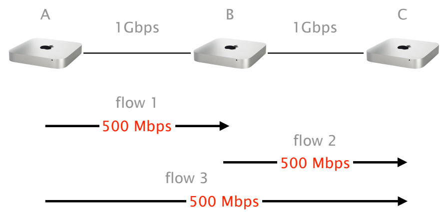
When *n* **entities** are using our transport mechanism,
we want a **fair allocation of the available bandwidth**

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



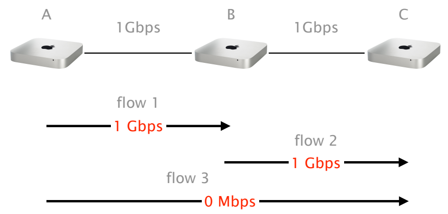
What is a fair allocation for the 3 flows?

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



Total traffic is 1.5 Gbps

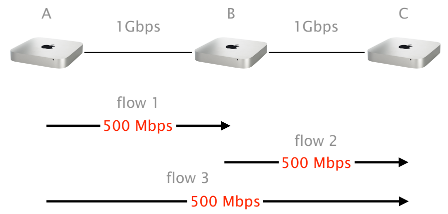
Fairness and efficiency don't always play along,
here an unfair allocation ends up *more efficient*



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



Total traffic is 1.5 Gbps

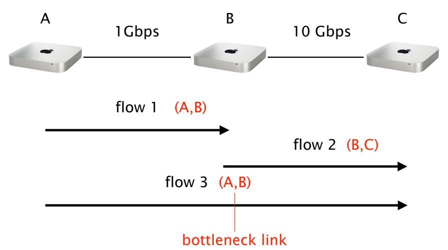
With equal-per-flow, A ends up with 1 Gbps because it sends 2 flows, while B ends up with 500 Mbps

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!

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 new ACK upon timeout, resend all W packets starting with the lost one

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

Let's see how it works in practice
visually

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

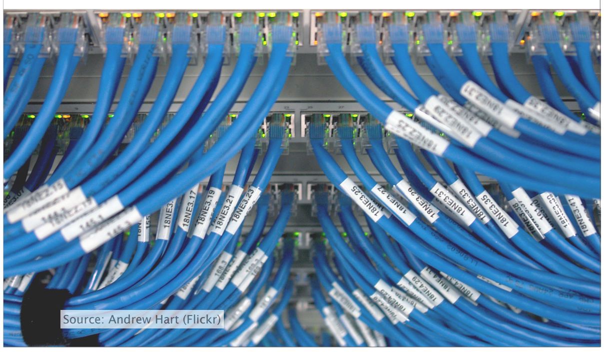
Reliable Transport

Correctness condition
if-and-only if again

Design space
timeliness *vs* efficiency *vs* …

Examples
Go-Back-N & Selective Repeat

Next week on Communication Networks
Ethernet and Switching


Source: Andrew Hart (Flickr)

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