Delay, Loss & Throughput

SCC. 203 – Computer Networks

Geoff Coulson

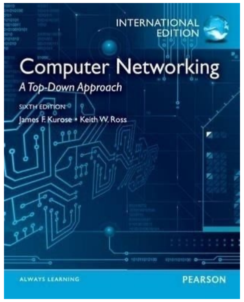
Week 12 Lecture 1

Refresher

- A *packet* is a unit of data carried by a network, and made up of a number of bytes
- The Internet enables end-to-end communication of packets between hosts, across multiple ISPs
  - Hosts can be laptops, PCs, phones or servers; any type of connected end-systems
- The Internet is a *best-effort* network
  - Sometimes packets do not reach their destination
  - Any reliability comes from the protocols (see later) - not come from the devices themselves!
    - Sometime we don’t need reliability - this way, we can choose

Lancaster University

Chapters 1.4.1, 1.4.2 & 1.4.3



Delay

Overview

- Delay is a measure of the time taken for a packet to travel across the network
- Measured in fractions of a second
  - Usually milliseconds – i.e., thousandths of a second

Types of Delay

- There are four main types of delay:
  - Processing Delay
  - Queuing Delay
  - Transmission Delay
  - Propagation Delay

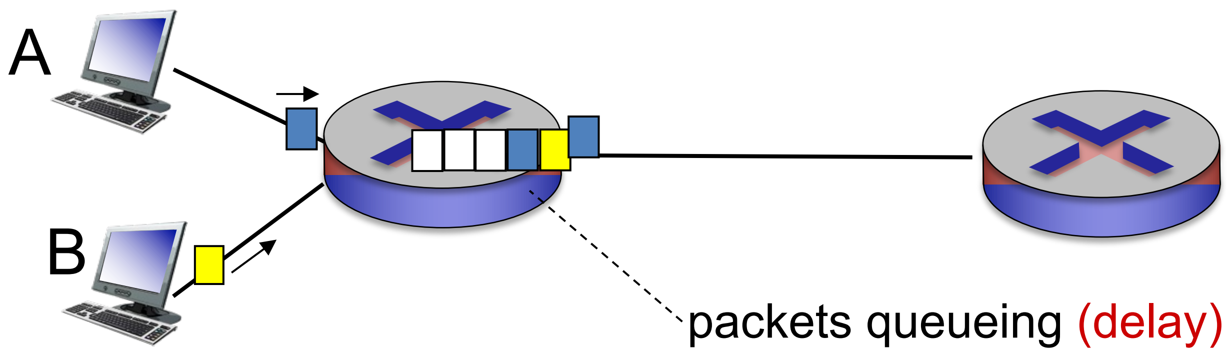
Processing Delay

- Time taken for a device to examine a packet's header and decide where to direct the packet
- May include a check of bit-level errors (as caused during transmission)
- Typically, this delay is very small
  - Usually microseconds – i.e., millionths of a second
- Can vary depending on how busy the device is

Queuing Delay

- Once a packet has been processed, it will join a *queue*
  - It waits here to leave the device
  - It will not be sent until it reaches the head of the queue
- Queuing Delay is the time spent waiting in the queue before a packet is transmitted
- The length of the queue, and thus the delay, is dependent on the congestion level of the node/router
- N.B. A queue only develops if the packet arrival rate to link (temporarily) exceeds output link capacity

Queuing Delay

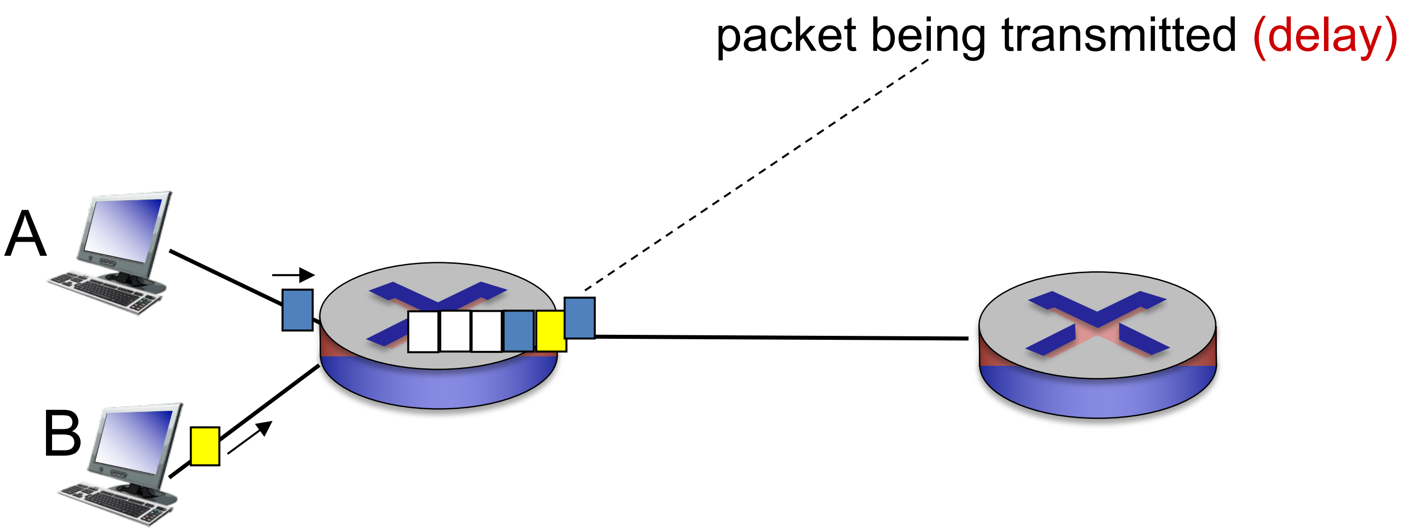


Transmission Delay

- Packets are transmitted once they reach the head of the queue…
- Networks are *Store-and-Forward*
  - An entire packet must be received before it's forwarded
- *Transmission Delay* is the amount of time required to push (transmit) all of the packet's bits onto the link

- Transmission Delay = L/R
- L: packet length (bits)
- R: link bandwidth (bps)

Transmission Delay



Propagation Delay

Lancaster University

- Once a bit has been pushed onto a link, it needs to propagate to the next device
- Each link has an associated *Propagation Delay*
  - Measured as the time needed to get over the link from one end to another
  - Propagation delay is dependent on the physical type of the link*

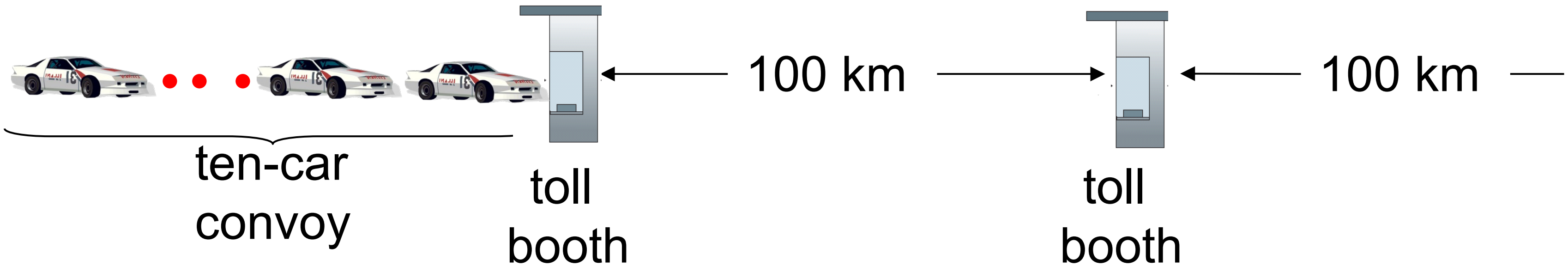
- Propagation Delay = d/s
- d: length of physical link
- s: link's propagation speed (~2×10^8 m/sec)

*for wireless communications – a little less than the speed of light

Transmission vs. Propagation Delay

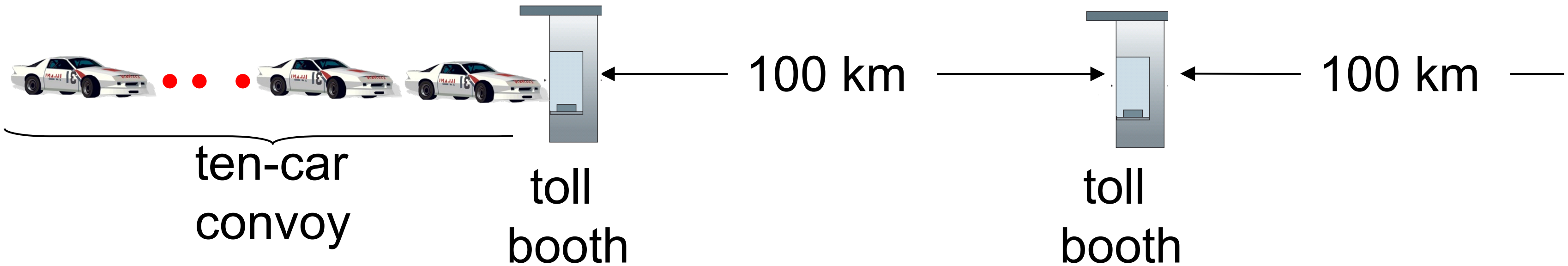
- A subtle difference, but important!
- *Transmission Delay* is the time required to push a packet out
  - It's a function of the packet's length and the transmission rate of the link interface
  - Nothing to do with the distance between the two devices
- *Propagation Delay* is the time taken for a bit to propagate from one device to the next
  - Function of the link technology and the distance between two devices
  - Nothing to do with the packet's length or the transmission rate of the link interface

Convoy Analogy



- Car ~ bit; convoy ~ packet
- Cars "propagate" at 100 km/hr
- Toll booth takes 12 sec to service a car (bit transmission time)
- *Q:* How long until convoy is lined up before 2nd toll booth?
- Time to "push" entire convoy through toll booth onto highway = 12*10 = 120 sec
- Time for last car to propagate from 1st to 2nd toll both: 100km/(100km/hr) = 1 hr
- *A:* 62 minutes

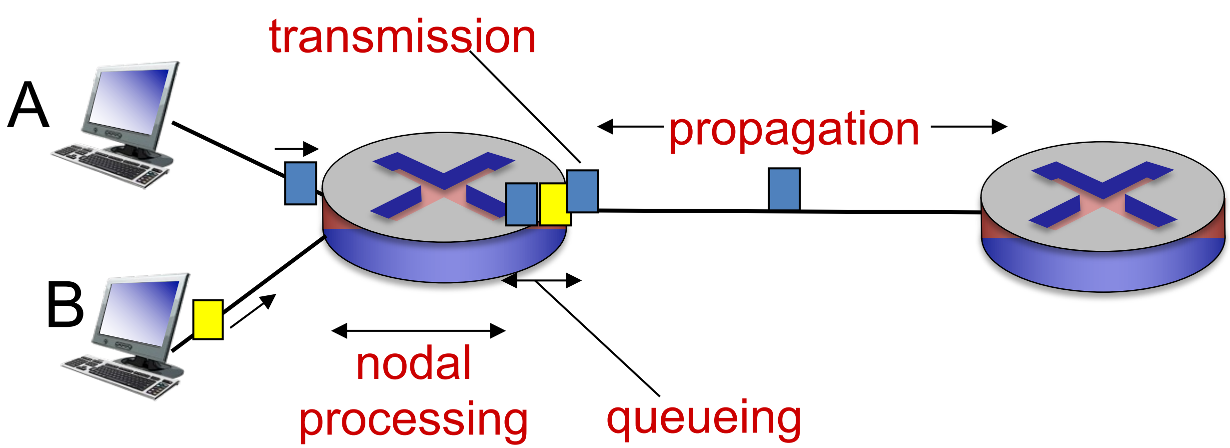
Convoy Analogy



- Suppose cars now "propagate" faster - at 1000 km/hr
- And that toll booths now take one min to service a car
- *Q:* Will cars arrive at the 2nd booth before all cars have been serviced at the1st?
- *A: Yes!* after 7 min, first car arrives at the 2nd booth; three cars are still at the 1st booth
  - First car spends 1 minute at toll booth, and then 1/10 of an hour = 6 mins propagating through the link until arriving at 2nd booth
  - At this point, car 8 is still at 1st booth (with 2 cars waiting behind)

Nodal Delay

- Total of all previously-mentioned types of delay
- Measured per node (i.e., per each device in a network)



$$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$$

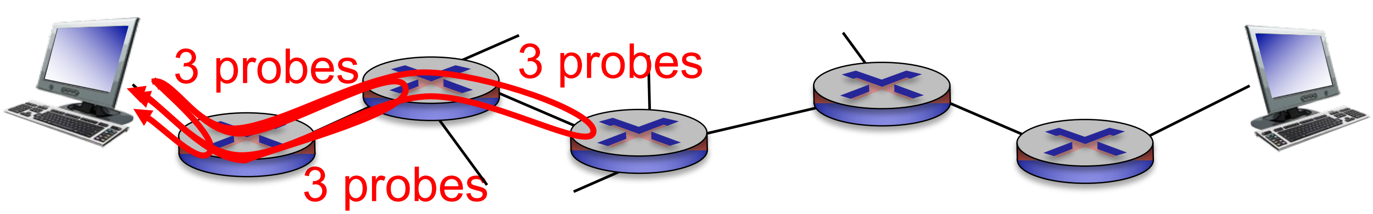
End-to-End Delay and RTT

- *End-to-End Delay* is an often-used measurement that is the *total of all nodal delays, from one host to another*
  - May include numerous nodes
- Varies over time as the various component sources of delay increase and decrease

- *Round Trip Time (RTT)*: End-to-End Delay measured in both directions
  - From one to host to another, and then back again
  - Doesn't necessarily have to use the same path/route!

Measuring Delay
Traceroute

- What does "real" Internet delay look like?
- The traceroute program measures delay from the source to each router on the path to a destination
- Records RTT (there and back)
- For each router, the source:
  - Sends three packets (probes) to a router that lies on the path to the destination
  - Routers return packets to the sender
  - Sender measures lag between transmission and reply



Demo

Traceroute

Measuring Delay
Traceroute

3 delay measurements

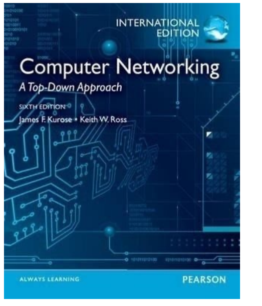
```
traceroute to gaia.cs.umass.edu (128.119.245.12), 64 hops max, 52
    byte packets
1.  192.168.1.1 (192.168.1.1)  2.911 ms  1.106 ms  1.061 ms
2.  212.69.63.54 (212.69.63.54)  25.012 ms  23.756 ms  23.824 ms
3.  212.69.63.126 (212.69.63.126)  24.398 ms  23.826 ms  23.822 ms
4.  ge-5-2-7.edge6.london1.level3.net (212.113.9.65)  23.798 ms
    24.030 ms  23.827 ms
5.  * * *
6.  university.ear3.newyork1.level3.net (4.71.230.234)  97.623 ms
    97.235 ms  97.021 ms
7.  core2-rt-et-8-3-0.gw.umass.edu (192.80.83.113)  97.005 ms  97.210
    ms  96.794 ms
8.  n5-rt-1-1-et-7-0-0.gw.umass.edu (128.119.0.10)  96.930 ms  97.168
    ms  98.084 ms
9.  cics-rt-xe-0-0-0.gw.umass.edu (128.119.3.32)  96.987 ms  97.519 ms
    96.960 ms
10. nscs1bbs1.cs.umass.edu (128.119.240.253)  101.099 ms  99.015 ms
    98.820 ms
11. gaia.cs.umass.edu (128.119.245.12)  97.101 ms  97.026 ms  96.965
    ms !Z
```

* means no response (router not replying, probe lost?)

trans-oceanic link

Lancaster University

Chapter 1.4.4

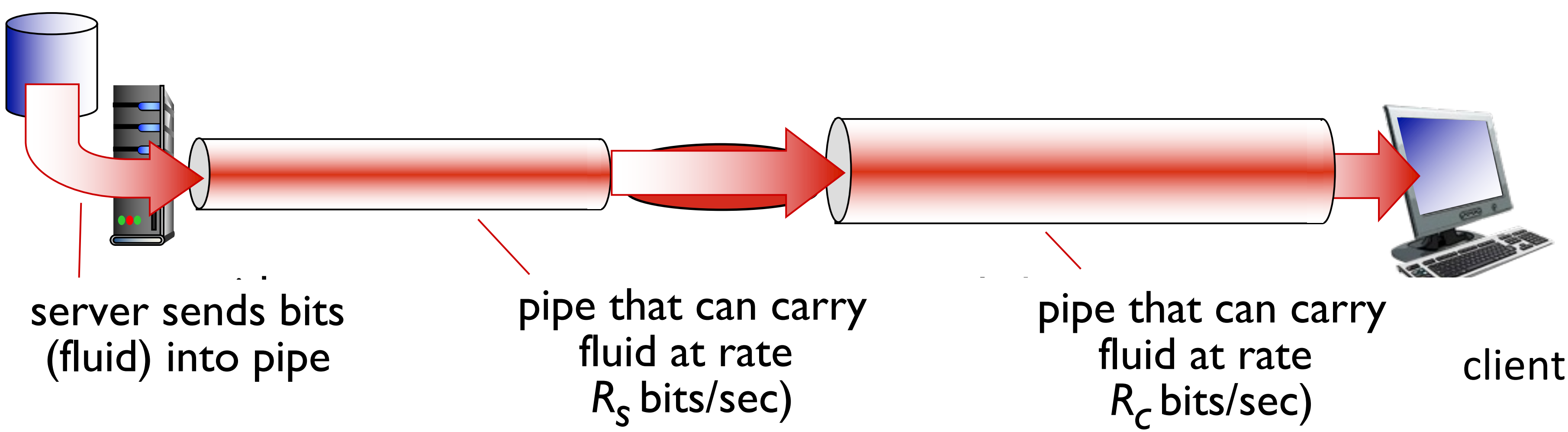


Throughput

Overview

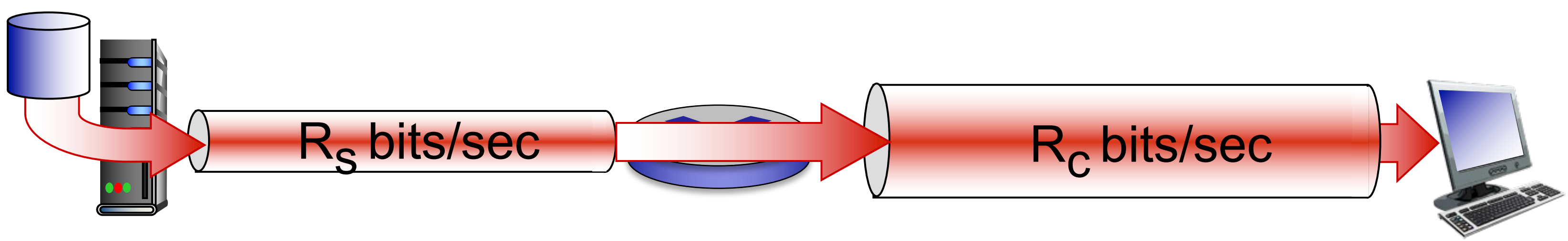
- *Throughput* is the rate (bits/timeunit) at which bits are transferred from a sender to a receiver
  - Instantaneous throughput: rate at given point in time
  - Average throughput: rate over a period of time
  - Peak throughput: highest instantaneous throughput rate seen so far
- Throughput is often restricted by a single-point *bottleneck*
- Some protocols can "throttle" themselves, and reduce their own rate
  - This avoids stressing bottleneck, but at the cost of lower rates

Throughput

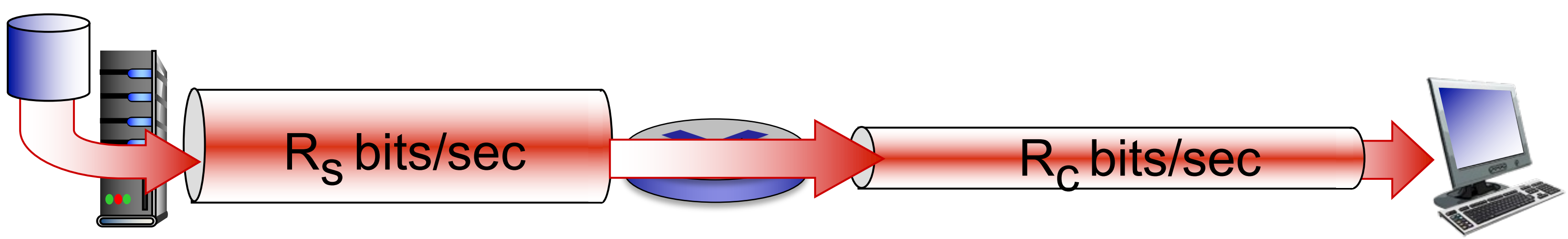

server sends bits
(fluid) into pipe
pipe that can carry
fluid at rate
R_s bits/sec)
pipe that can carry
fluid at rate
R_c bits/sec)
client

Throughput

- $R_s < R_c$ What is average end-end throughput?



- $R_s > R_c$ What is average end-end throughput?

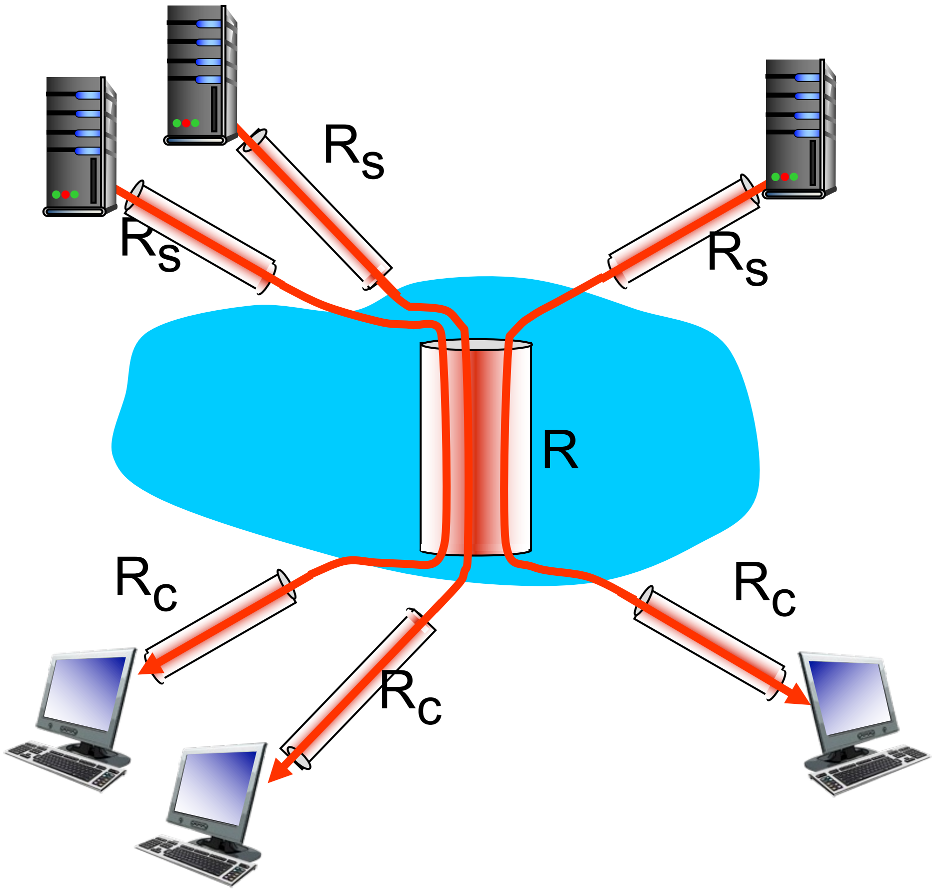


bottleneck link

= link on the end-end path that constrains end-end throughput

Throughput
Internet Scenario

- Per-connection end-end throughput: *min(R_c,R_s,R/10)*
- In practice: R_c or R_s is often the bottleneck



10 connections (fairly) share backbone bottleneck link *R* bits/sec

Goodput

- A term sometimes used when discussing throughput…
- *Goodput* measures throughput at the highest level and aims to be an 'honest' application-meaningful measurement
- It excludes protocol header and retransmission overheads

Measuring throughput
iperf

- Produces standardised throughput measurements
- Normally runs as a “client-server” model
  - Client host requests sample data, server host serves it, generating a stream of sample data across the network
  - Because we know exactly how large the sample data is, and how long it takes to retrieve it, we can calculate throughput
- Very configurable
  - Can use different protocols, data sample sizes, etc.
  - Can also do bi-directional transfer, to test both directions
    - Check for irregularities

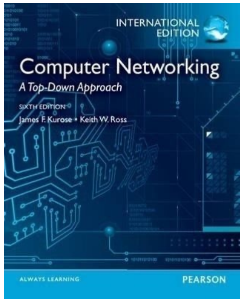
Demo

iperf3

Lancaster University

Chapter 1.4.2

Packet Loss

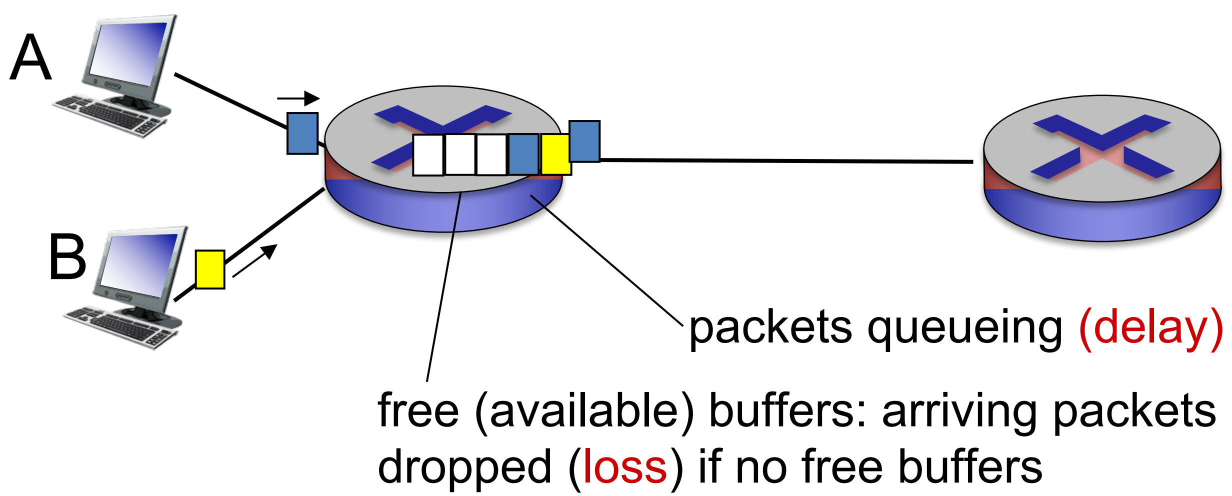


Overview

- *Packet Loss* occurs when a router node drops or discards a packet
- This means that the packet does not reach its destination
- *Packet Loss* reduces the useable throughput (goodput) of a device, as the number of *useful* bits sent is fewer than would otherwise be the case
- For reliable protocols, packet loss causes *retransmission* to occur
  - The destination host sets a timer for the expected arrival time of each next-expected packet
  - If the timer expires before the packet arrives, a retransmission request is sent to the source host
  - This introduces additional delay, as the packet has to make the full journey again, regardless of where it was lost

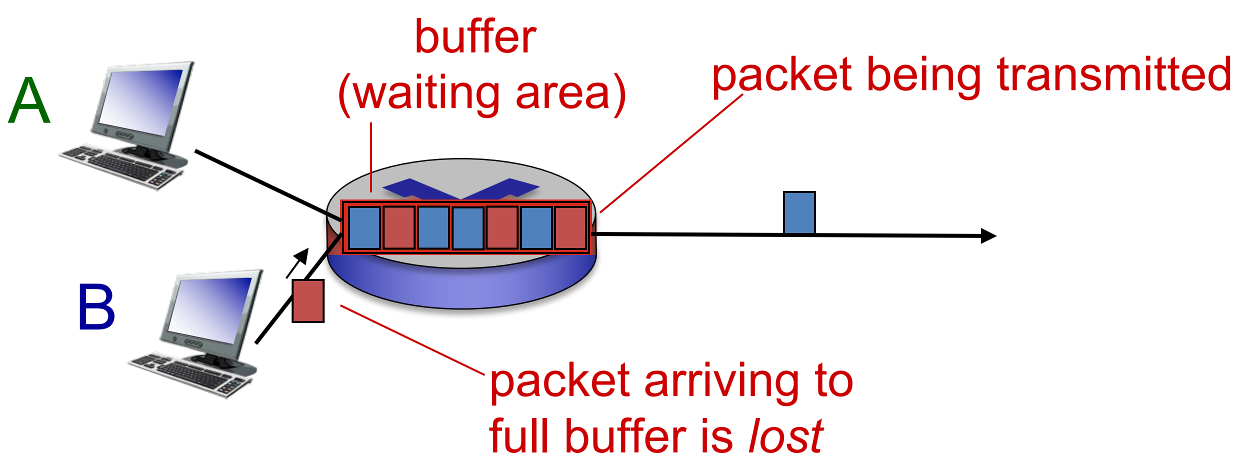
Queuing Delay and Packet Loss

- *Packet Loss* has a close relationship with *Queuing Delay*
- When packets arrive at a rate greater than the maximum supported throughput, the router's queue starts to fill up



Queuing Delay and Packet Loss

- When a queue becomes full, additional packets *cannot* be received
  - As a result, they are discarded and lost
- Lost packets may be retransmitted by the previous router, or by the source host, or not at all



Other Sources of Loss

- Packets can also be lost in the physical medium, particularly with wireless link technologies
- Hardware and software in devices may also malfunction
  - This includes errors and corruption
  - packets may still be sent, but fail a checksum verification
- Devices can also be attacked: “Denial of Service” attack
  - A simple way would be to bombard a device with packets, and so fill up its buffers
  - This prevents other hosts from sending packets, whether completely or at a reduced rate

Measuring Loss
ping

- Ping works in a similar fashion to traceroute but in an end-to-end manner
- It creates messages to be sent out to specific hosts (rather than nodes along the path)
- Measures RTT (as with traceroute)
- Also includes a measurement of packet loss, by keeping track of how many messages were sent, and how many responses were received

Demo

ping

Queuing Disciplines

- When packets must be dropped because of a full queue, we still have control over which packet(s) to drop
- In the examples we have discussed, we have assumed a "tail drop" approach
- However, other techniques can be used:
  - *Random Drop:* Drop any packet within the queue
  - *Quality-of-Service (QoS) Aware*: Packets will be dropped given their priority; provide fairness and guarantee throughput for sensitive services, such as voice calls or live video

Thanks for listening!
Any questions?

g.coulson@lancaster.ac.uk