Internet in a Day
Day 2 of 1

Carnegie Mellon University
15-440, Distributed Systems

Administrivia

- Recitations start tomorrow!
  - You can attend either one *unless* things get crazy overcrowded
- See newsgroup for andrew linux EDITOR variable fix. It's an andrew bug, but fix is easy; andrew staff are working on it.
- Project 1 stage 0 due Thursday. It's negative points only, and *utterly* trivial, so do it soon!

Last Time

- Modularity, Layering, and Decomposition
  - Example: UDP layered on top of IP to provide application demux ("ports")
- Resource sharing and isolation
  - Statistical multiplexing - packet switching
- Dealing with heterogenity
  - IP "narrow waist" -- allows many apps, many network technologies
  - IP standard -- allows many impls, same proto

- Models to reason about network and system behavior
  - Fail-stop, fail-stutter, byzantine failures
  - Sync/Async communication model
  - etc.

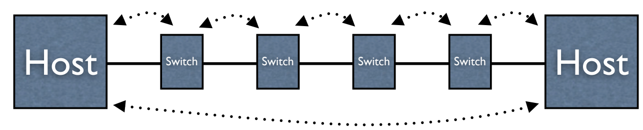
Today: TCP and Apps

- Remember from last time...
- IP service model: “best-effort”
  - Can drop, mangle, re-order, delay packets
- Easy model to *provide* (imposes few requirements on underlying layers--widely applicable)
- Less fun model to *program to* if you happen to need reliability, in-order, correct data

Design Question

- If you want reliability, etc.
- Where should you implement it?

Option 1: Hop-by-hop



Option 2: end-to-end

Options

- Hop-by-hop: Have each switch/router along the path ensure that the packet gets to the next hop
- End-to-end: Have just the end-hosts ensure that the packet made it through
- What do we have to think about to make this decision??

A question

- Is hop-by-hop enough?
  - [hint: What happens if a switch crashes? What if it’s buggy and goofs up a packet?]

The End-to-End Argument

If you have to implement a function end-to-end *anyway* (e.g., because it requires the knowledge and help of the end-point host or application),

don't implement it inside the communication system

unless there's a compelling performance enhancement

Further Reading: "End-to-End Arguments in System Design." Saltzer, Reed, and Clark.

Keep in mind

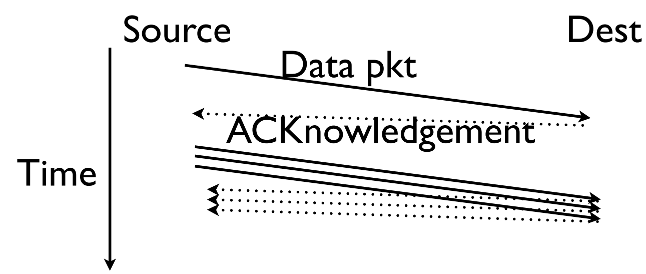
- This is an engineering rule of thumb to be weighed against other design guidelines
- not a law
- But in practice, it's proved to be a nice way to think about things
- You may encounter situations where you can't (for whatever reason - technical, financial, political) implement things in the way the argument suggests. The real world can be an ugly place. :)

Let's apply that to our question...

- Of where to do retransmissions
- What does e2e argument argue for here?
- TCP uses end-to-end retransmissions
- Can you think of times we might want to *also* implement hop-by-hop retransmission?
  - Hop-by-hop retransmission is cheaper and faster (count the "packet-miles" that are traveled)
  - So maybe a very high-loss link -- like wireless!
  - Your wireless card handles a few retransmissions on its own

Rough view of TCP

(This is a *very* incomplete view - take 15-441. :)

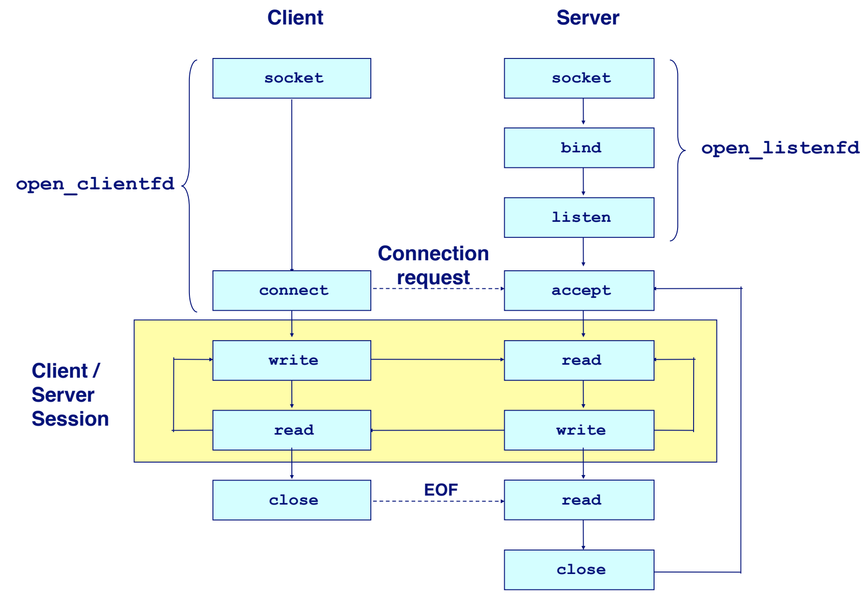


What TCP does:
1) Figures out which packets got through/lost
2) Figures out how fast to send packets to use all of the unused capacity,
- But not more
- And to share the link approx. equally with other senders

Application View of TCP

- Remember socket API basics from 15-213
    - We'll remind you about these a bit in recitations, but for now...

Socket API Operation Overview



Blocking sockets

- What happens if an application write()s to a socket waaaaay faster than the network can send the data?
    - TCP figures out how fast to send the data...
    - And it builds up in the kernel socket buffers at the sender... and builds...
    - until they fill. The next write() call *blocks* (by default).
    - What's blocking? It suspends execution of the blocked thread until enough space frees up...

In contrast to UDP

- UDP doesn't figure out how fast to send data, or make it reliable, etc.
- So if you write() like mad to a UDP socket...
- It often silently disappears. *Maybe* if you're lucky the write() call will return an error. But no promises.

take a breath.

Rehashing all of that...

- TCP is layered on top of IP
  - IP understands only the IP header
  - The IP header has a “protocol” ID that gets set to TCP
  - The TCP at the receiver understands how to parse the TCP information
- IP provides only “best-effort” service
- TCP adds value to IP by adding retransmission, in-order delivery, data checksums, etc., so that programmers don’t have to re-implement the wheel every time. It also helps figure out how fast to send data. This is why TCP sockets can “block” from the app perspective.
- The e2e argument suggests that functionality that *must* be implemented end-to-end anyway (like retransmission in the case of dead routers) should probably be implemented only there -- unless there’s a compelling perf. optimization

Questions to ponder

- What does the end-to-end argument say about where to implement encryption for confidentiality?
- If you have a whole file to transmit,
  how do you send it over the Internet?
  - You break it into packets (packet-switched medium)
  - TCP, roughly speaking, has the sender tell the receiver “got it!” every time it gets a packet. The sender uses this to make sure that the data’s getting through.
  - But by e2e, if you have to acknowledge the correct receipt of the entire file... why bother acknowledging the receipt of the individual packets???

<break>

Answers

- 1) Encrypt end-to-end.
  - A notable exception: The military sometimes uses hop-by-hop so that they can run unencrypted on physically secure links ... so that they can monitor the traffic there.
- 2) This is a bit of a trick question -- it's not asking e2e vs in-network. :)
  The answer: Imagine the waste if you had to retransmit the entire file because one packet was lost. Ow.

Application Requirements

Q: If you're building an application...

How do you choose what transport service to use?

A: That depends what the application's communication requirements are...

What Transport Service Does an Application Need?

Data loss

- Some applications (e.g., audio) can tolerate some loss
- Other applications (e.g., file transfer, telnet) require 100% reliable data transfer

Timing

- Some applications (e.g., Internet telephony, interactive games) require low delay to be "effective"

Bandwidth

- Some applications (e.g., multimedia) require a minimum amount of bandwidth to be "effective"
- Other applications ("elastic apps") will make use of whatever bandwidth they get

User Datagram Protocol(UDP): An Analogy

UDP	Postal Mail
Single socket to receive messages	Single mailbox to receive letters
No guarantee of delivery	Unreliable ☺
Not necessarily in-order delivery	Not necessarily in-order delivery
Datagram – independent packets	Letters sent independently
Must address each packet	Must address each reply

Example UDP applications
Multimedia, voice over IP

Transmission Control Protocol (TCP): An Analogy

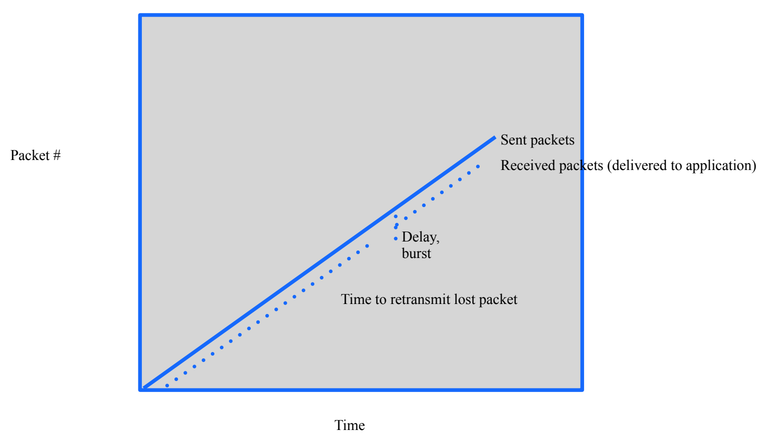
TCP	Telephone Call
• Reliable – guarantee delivery • Byte stream – in-order delivery • Connection-oriented – single socket per connection • Setup connection followed by data transfer	• Guaranteed delivery • In-order delivery • Connection-oriented • Setup connection followed by conversation

Example TCP applications
Web, Email, Telnet

Why not always use TCP?

- TCP provides “more” than UDP
- Why not use it for everything??
- A: Nothing comes for free...
- 1) Connection setup (take on faith) -- TCP requires one round-trip time to setup the connection state before it can chat...
- How long does it take, using TCP, to fix a lost packet?
  - » At minimum, one “round-trip time” (2x the latency of the network)
  - » That could be 100+ milliseconds!
- If I guarantee in-order delivery, what happens if I lose one packet in a stream of packets?

One lost packet



Design trade-off

- If you’re building an app...
- Do you need everything TCP provides?
- If not: Can you deal with its drawbacks to take advantage of the subset of its features you need?
- If not: You’re going to have to implement the ones you need on top of UDP
  - Caveat: There are some libraries, protocols, etc., that can help provide a middle ground.
  - Takes some looking around - they’re not as standard as UDP and TCP.

Transport Service Requirements of Common Applications

Application	Data loss	Bandwidth	Time Sensitive
file transfer	no loss	elastic	no
e-mail	no loss	elastic	no
web documents	no loss	elastic	no
real-time audio/ video	loss-tolerant	audio: 5Kb-1Mb video:10Kb-5Mb	yes, 100's msec
stored audio/video	loss-tolerant	same as above	yes, few secs
interactive games	loss-tolerant	few Kbps	yes, 100's msec
financial apps	no loss	elastic	yes and no

- **Interactions between layers are important.**
  - **persistent HTTP**
  - **encryption and compression**
  - **MPEG frame types. Loss & real-time video.**

Proj 1 and today's material

- You'll use UDP. Why?
  - A1: The course staff is full of sadists who want you to do a lot of work. This is true in part: timeouts and retransmission are a core aspect of using the network.
  - A2: The communication needed is *very* small, and you have to implement a lot of reliability stuff anyway to ensure that the work gets done...
  - Honestly? This one seems to me like a middle ground. You might use TCP for "other" reasons (firewalls that block everything but TCP), or to avoid the need for the "job ack" part of the protocol. Or you might stick with UDP to reduce the overhead at the server.