Distributed Systems

15-440/640

Fall 2018

3 – Communication: The Internet in a Day (ctnd)

Announcements

- Recitations
 - Tomorrow (9/5) Wean 7500 to go over the basics of Golang at 6pm and again at 7pm
 - Prepare by going over the Tour of Go <u>https://tour.golang.org</u>
 - In general: will be on Wednesday evenings
 - Not every week
 - Will be announced

Thursday's Lecture

- Network links and LANs
- **Inter-network Communication**
- Layering & Protocols
- Internet design
- **Transport protocols**
- Application design

Today's Lecture

- Network links and LANs
- Inter-network Communication
- Layering & Protocols
- Internet design
- **Transport protocols**
- Application design

Network Service Model

- What is the *service model* for inter-network?
 - Defines what promises that the network gives for any transmission
 - Defines what type of failures to expect

- <u>best-effort</u>
 - Ethernet/Internet– packets can get lost, etc.

⇒ Development of "failure models" in DS design

Possible Failure models

- Fail-stop:
 - When something goes wrong, the process stops / crashes / etc.
- Fail-slow or fail-stutter:
 - Performance may vary on failures as well
- Byzantine:
 - Anything that can go wrong, will.
 - Including malicious entities taking over your computers and making them do whatever they want.
- These models are useful for proving things;
- The real world typically has a bit of everything.
- Deciding which model to use is important!

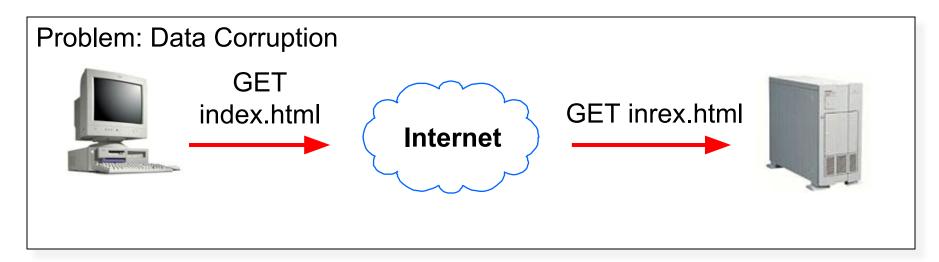
Fancier Network Service Models

- What if you want more?
 - Performance guarantees (QoS)
 - Reliability
 - Corruption
 - Lost packets
 - Flow and congestion control
 - Fragmentation
 - In-order delivery
 - Etc...

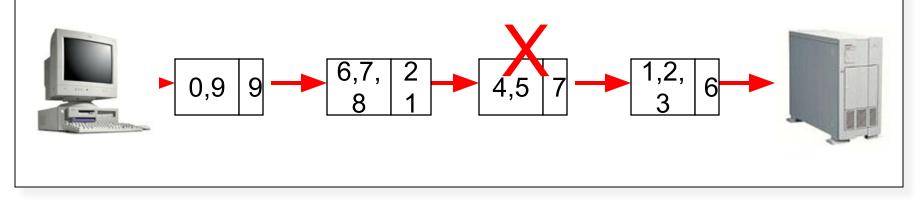
If network provides this \Rightarrow reuse across applications

How would you implement these?

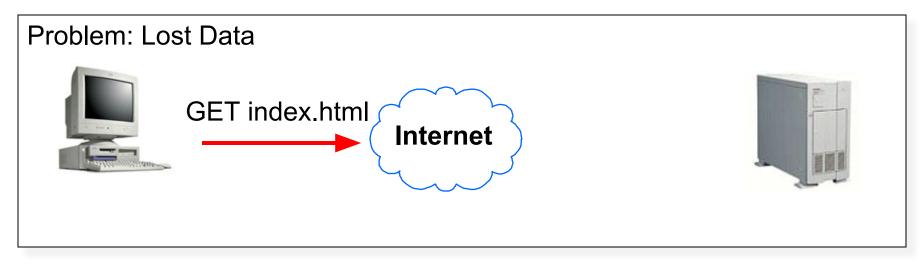
What if the Data gets Corrupted?



Solution: Add a checksum

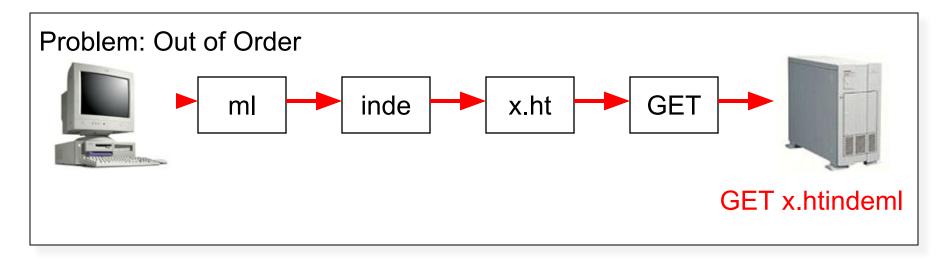


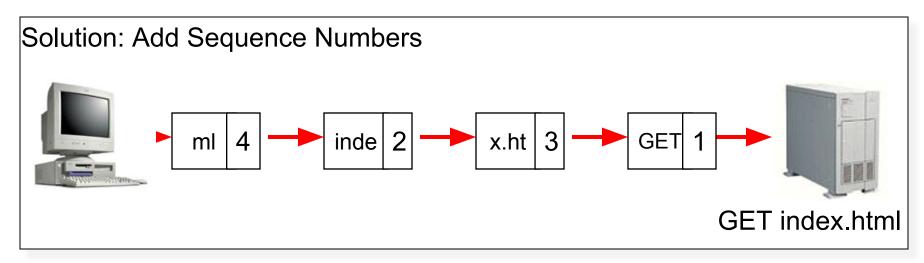
What if the Data gets Lost?



Solution: Timeout and Retransmit GET index.html GET index.html

What if the Data is Out of Order?





Networks [including end points] Implement Many Functions

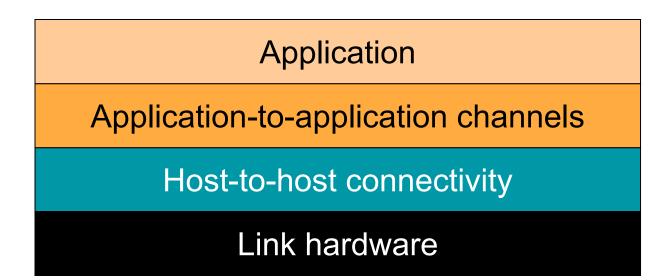
- Link
- Multiplexing
- Routing
- Addressing/naming (locating peers)
- Reliability
- Flow control
- Fragmentation
- Etc....

But note limitations: these can't turn a byzantine failure model into a fail-stop model!

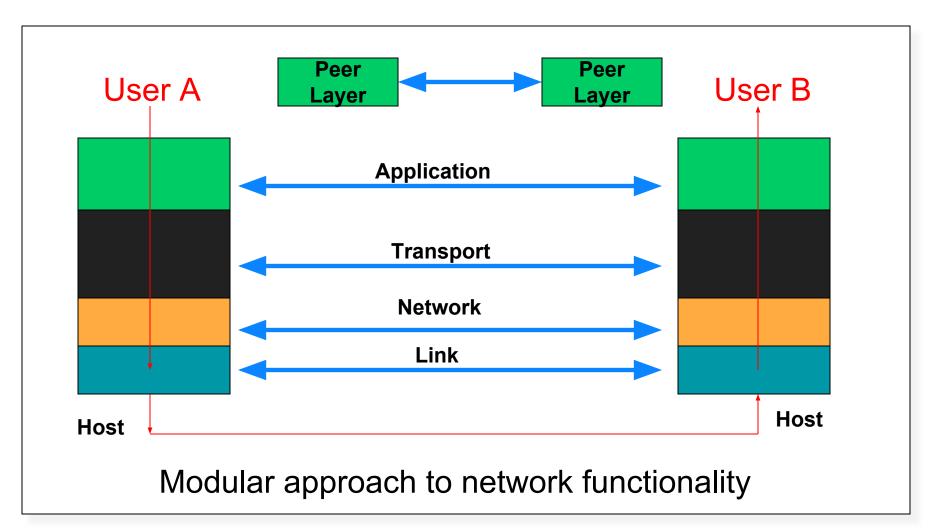
Where would you implement each function?

What is Layering?

- Modular approach to network functionality
- Example:



What is Layering?



Layering Characteristics

- Each layer relies on services from layer below and exports services to layer above
- Interface defines interaction with peer on other hosts
- Protocols define:
 - Interface to higher layers (API)
 - Interface to peer (syntax & semantics)
 - Actions taken on receipt of a messages
 - Format and order of messages
 - Error handling, termination, ordering of requests, etc.
- Hides implementation layers can change without disturbing other layers (black box)

Today's Lecture

Network links and LANs

Inter-network Communication

Layering & Protocols

Internet design

Transport protocols

Application design

Goals [Clark88]

0 Connect existing networks

initially ARPANET and ARPA packet radio network

1. Survivability

ensure communication service even in the presence of network and router failures

- 2. Support multiple types of services
- 3. Must accommodate a variety of networks
- 4. Allow distributed management
- 5. Allow host attachment with a low level of effort
- 6. Be cost effective
- 7. Allow resource accountability

Gateway Alternatives

- Translation
 - Difficulty in dealing with different features supported by networks
 - Scales poorly with number of network types (N^2 conversions)
- Standardization
 - "IP over everything" (<u>Design Principle 1</u>)
 - Minimal assumptions about network
 - Hourglass design

Goal 1: Survivability

- If network is disrupted and reconfigured...
 - Communicating entities should not care!
 - No higher-level state reconfiguration
- How to achieve such reliability?
 - Where can communication state be stored?

	Network	Host
Failure handing	Replication	"Fate sharing"
Net Engineering	Tough	Simple
Switches	Maintain state	Stateless
Host trust	Less	More

Fate Sharing



- Lose state information for an entity if and only if the entity itself is lost.
- Examples:
 - OK to lose TCP state if one endpoint crashes
 - NOT okay to lose if an intermediate router reboots
- Tradeoffs
 - Survivability: Heterogeneous network → less information available to end hosts and Internet level recovery mechanisms
 - Trust: must trust endpoints more

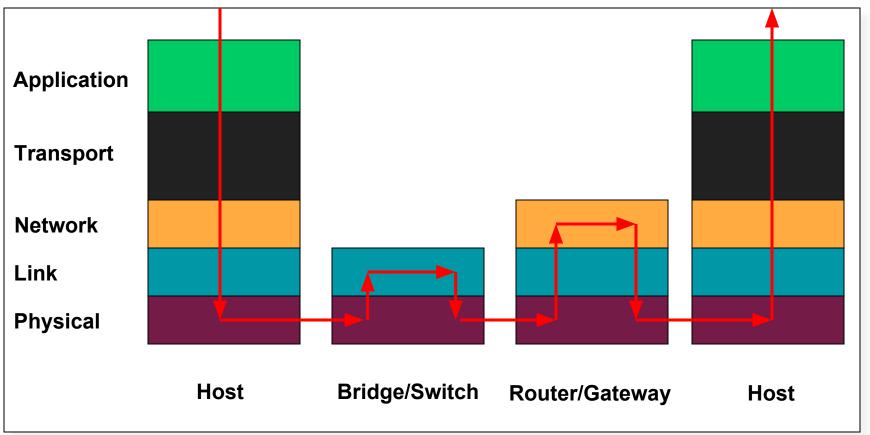
End-to-End Argument

- Deals with where to place functionality
 - Inside the network (in switching elements)
 - At the edges
- 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
- Key motivation for split of functionality between TCP,UPD and IP

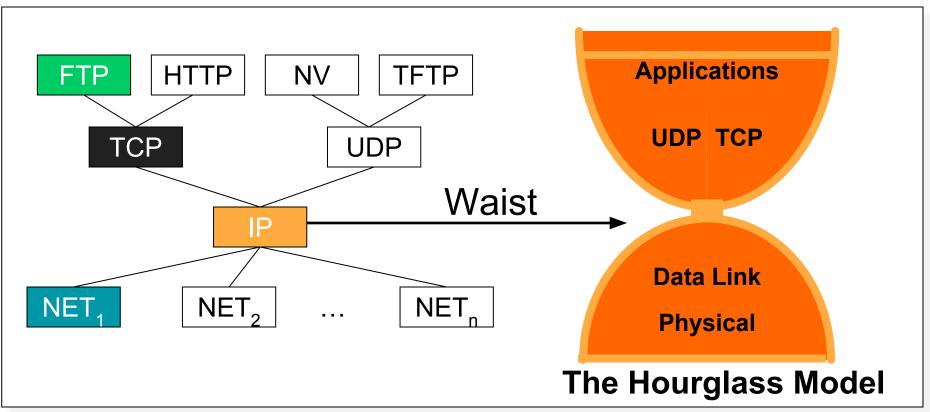
20

IP Layering

• Relatively simple



The Internet Protocol Suite

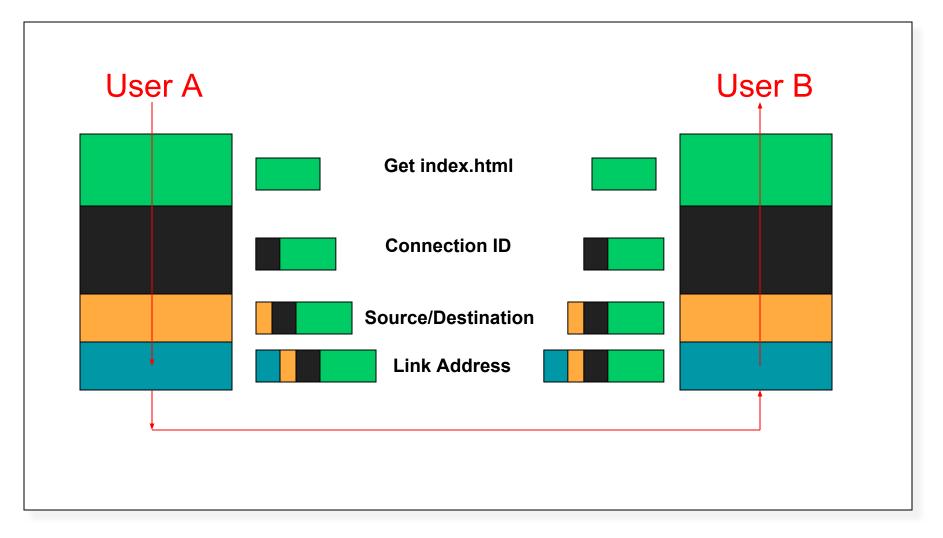


The waist facilitates interoperability

What's the disadvantage of the "IP waist"?

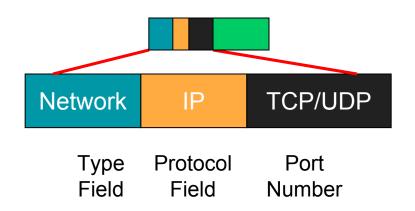
What creates the "edges" in the tree above?

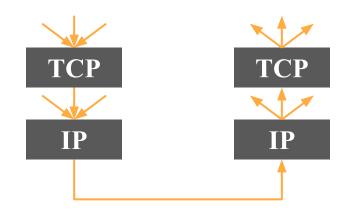
Layer Encapsulation



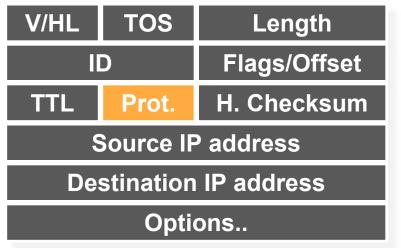
Multiplexing and Demultiplexing

- There may be multiple implementations of each layer.
 - How does the receiver know what version of a layer to use?
- Each header includes a demultiplexing field that is used to identify the next layer.
 - Filled in by the sender
 - Used by the receiver







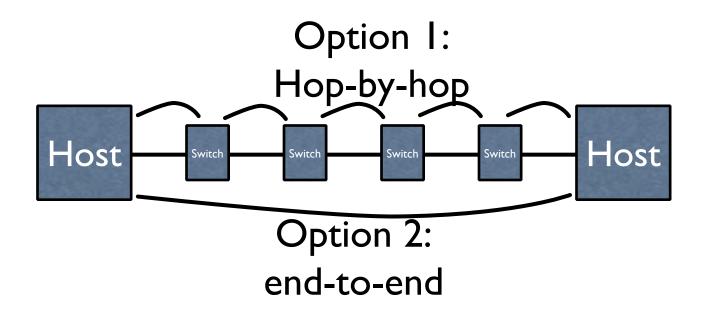


Today's Lecture

- Network links and LANs
- Inter-network Communication
- Layering & Protocols
- Internet design
- **Transport protocols**
- Application design

Design Question

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



Transport Protocols & Types of Service

- TCP vs. UDP
 - Elastic apps that need reliability: remote login or email
 - Inelastic, loss-tolerant apps: real-time voice or video
 - Others in between, or with stronger requirements
 - Biggest cause of delay variation: reliable delivery
 - Today's net: ~100ms RTT
 - Reliable delivery can add seconds.
- Original Internet model: "TCP/IP" one layer
 - First app was remote login...
 - But then came debugging, voice, etc.
 - These differences caused the layer split, added UDP

Transport Protocols

- UDP provides just integrity and demux
- TCP adds...
 - Connection-oriented
 - Reliable
 - Ordered
 - Point-to-point
 - Byte-stream
 - Full duplex
 - Flow and congestion controlled

User Datagram Protocol (UDP): An Analogy

UDP

- Single socket to receive messages
- No guarantee of delivery
- Not necessarily in-order delivery
- Datagram independent packets
- Must address each packet

Postal Mail

- Single mailbox to receive letters
- Unreliable 😳
- Not necessarily in-order delivery
- Letters sent independently
- Must address each letter

Example UDP applications Multimedia, voice over IP

Transmission Control Protocol (TCP): An Analogy

TCP

- Reliable guarantee delivery
- Byte stream in-order delivery
- Connection-oriented single socket per connection
- Setup connection followed by data transfer

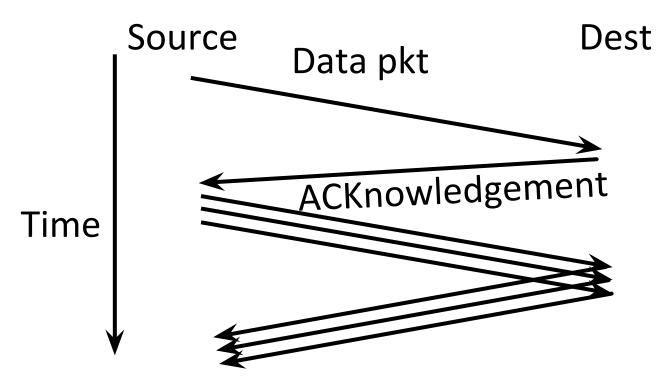
Telephone Call

- Guaranteed delivery
- In-order delivery
- Connection-oriented
- Setup connection followed by conversation

Example TCP applications Web, Email, Telnet

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

Questions to ponder

- 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.
 - If you acknowledge the correct receipt of the entire file (e.g. due to e2e argument)... why bother acknowledging the receipt of the individual packets???

• The answer: Imagine the waste if you had to retransmit the entire file because one packet was lost. Ow.

Today's Lecture

- Network links and LANs
- Inter-network Communication
- Layering & Protocols
- Internet design
- **Transport protocols**
- Application design

Today's Lecture

- Network links and LANs
- Inter-network Communication
- Layering & Protocols
- Internet design
- Transport protocols
- Application design

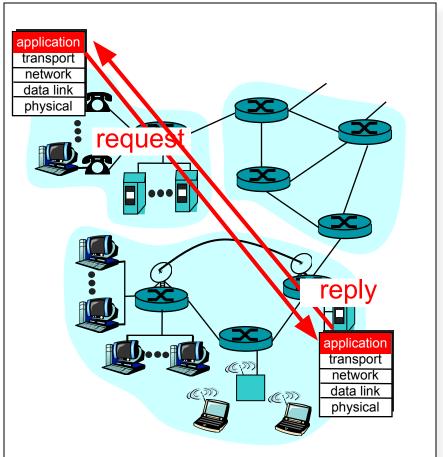
Client-Server Paradigm

Typical network app has two pieces: *client* and *server* Client:

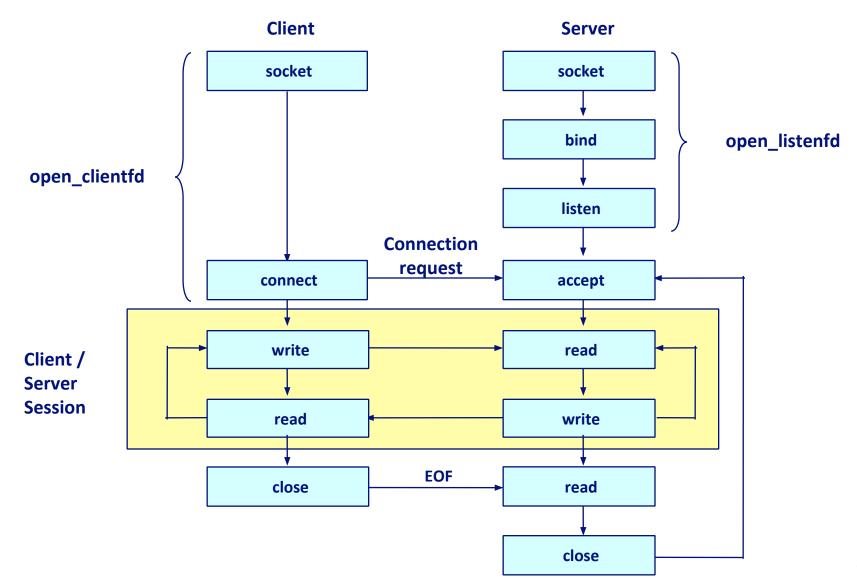
- Initiates contact with server ("speaks first")
- Typically requests service from server,
- For Web, client is implemented in browser; for e-mail, in mail reader

Server:

- Provides requested service to client
- e.g., Web server sends requested Web page, mail server delivers e-mail



Socket API Operation Overview



What Service Does an Application Need?

Data loss

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

Timing

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

Bandwidth

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

Transport Service Requirements of Common Apps

	Application	Data loss	Bandwidth	Time Sensitive
_	file transfer e-mail	no loss no loss	elastic elastic	no
-	web documents	no loss	elastic	no
-	interactive audio/video	loss-tolerant (often)	audio: 5Kb-1Mb video:10Kb-5Mb	yes, 100's msec
-	non-interactive audio/video	loss-tolerant (sometimes)	same as above	yes, few secs
11 —	nteractive games financial apps	loss-tolerant no loss	few Kbps elastic	yes, 100's msec yes and no: µs ?

Why not always use TCP?

- TCP provides "more" than UDP
- Why not use it for everything??
- A: Nothing comes for free...
 - 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



Delayed burst Time to retransmit lost packet Sent packets Received packets (delivered to application)

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?
 OR
 - 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.

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.

Web Page Retrieval

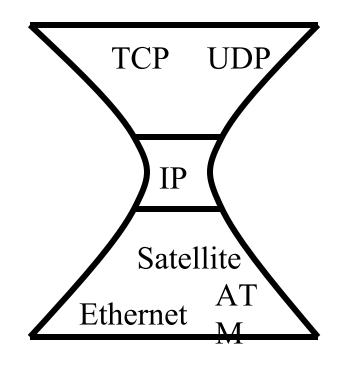
- 1. Static configuration
 - IP address, DNS server IP address, IP address of routers,
- 2. ARP for router
- 3. DNS lookup for web server
 - Several packet exchanges for lookup
- 4. TCP SYN exchange
- 5. HTTP Get request
- 6. HTTP response
 - Slow start, retransmissions, etc.

Caching Helps

- 1. Static configuration
 - IP address, DNS server IP address, IP address of routers,
- 2. ARP for router
- 3. DNS lookup for web server
 - Several packet exchanges for lookup
- 4. TCP SYN exchange
- 5. HTTP Get request
- 6. HTTP response
 - Slow start, retransmissions, etc.

Summary: Internet Architecture

- Packet-switched datagram network
- IP is the "compatibility layer"
 - Hourglass architecture
 - All hosts and routers run IP
- Stateless architecture
 - no per flow state inside network



Summary: Minimalist Approach

- Dumb network
 - IP provide minimal functionalities to support connectivity
 - Addressing, forwarding, routing
- Smart end system
 - Transport layer or application performs more sophisticated functionalities
 - Flow control, error control, congestion control
- Advantages
 - Accommodate heterogeneous technologies (Ethernet, modem, satellite, wireless)
 - Support diverse applications (telnet, ftp, Web, X windows)
 - Decentralized network administration

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