Shared, Multi-Hop Networks and End-to-End Arguments

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CS 3035/GZ01
3rd October 2013

Networks Are Shared

- Any host may want to reach any other
- Dedicated links: N hosts → $O(N^2)$ links
  - "Fewer links" implies "shared links"
- CPUs, memory, disks, network cards ever-cheaper, quickly upgradeable
- Adding network links different:
  - Non-decreasing costs of tearing up street, laying cable, launching satellite, radio spectrum…
  - Approval often involves regulatory bodies → glacial pace
  - Users economically motivated to share network resources, despite declining costs of network electronics

Sharing of Multi-Hop Networks

- Link multiplexing among users
- Packet forwarding and delay
- Best-effort delivery: packet buffering, buffer overflow, packet dropping
- Packet duplication
- Packet corruption
- Link breakage
- Packet reordering

Link Multiplexing

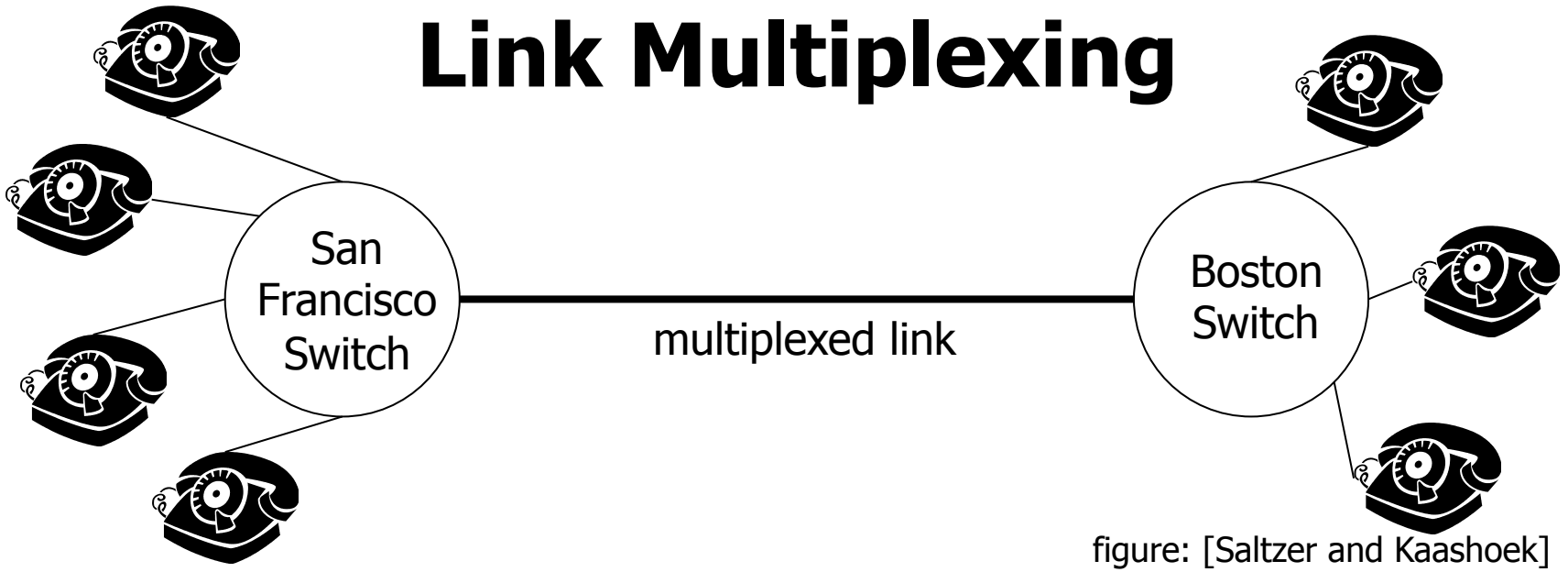

figure: [Saltzer and Kaashoek]

- Link between cities carries many conversations simultaneously; multiplexed
- Earthquake in SF; heavy call load from Boston to SF
- Link capacity limited; some Boston callers will be told "network busy"
- N^{th} caller's call completes, $n+1^{st}$'s fails; why?

Telephony: Isochronous Link Multiplexing

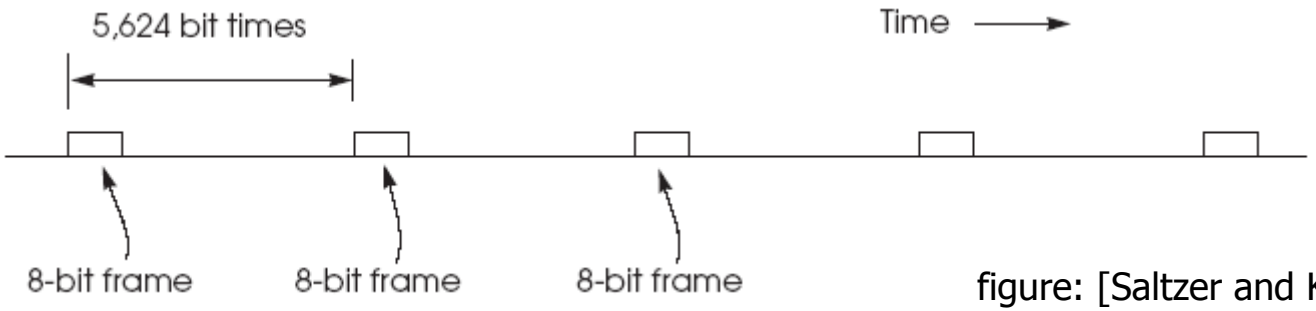

figure: [Saltzer and Kaashoek]

- Assume shared link capacity 45 Mbps
- Voice call: 8-bit samples (frames) spaced perfectly evenly to achieve 64 Kbps
  - one frame per 5624 bit times, or 125 us; 8000 frames per second
- Between one call's successive frames, room for 702 other frames; 703 calls total capacity
- Hard-edged: 1st 703 win, 704th loses

Telephony: Isochronous Link Multiplexing

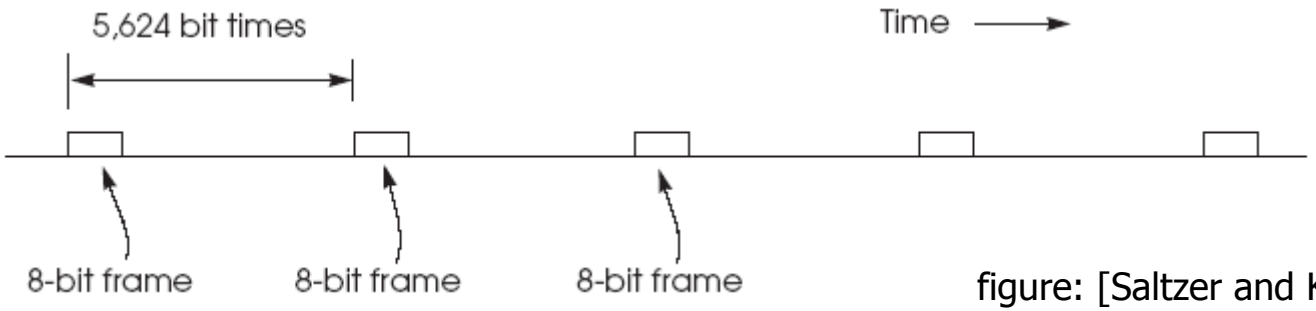

figure: [Saltzer and Kaashoek]

Time-Division Multiplexing (TDM): equal-sized frames at equal intervals, perfectly predictable rate and delay

per second

- Between one call's successive frames, room for 702 other frames; 703 calls total capacity
- Hard-edged: 1st 703 win, 704th loses

Connection-Oriented Forwarding

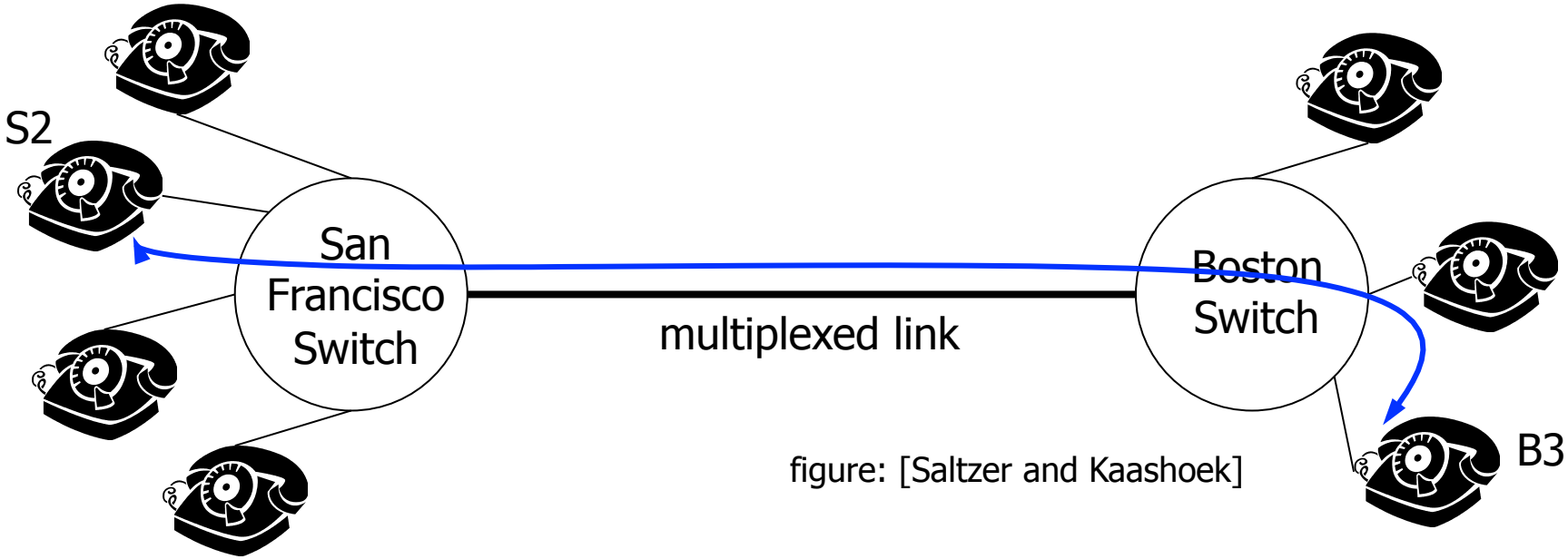

figure: [Saltzer and Kaashoek]

- Connection setup: Boston switch asks SF switch to forward B3's frames to S2
- Switches store state concerning how to forward frames for the call; slot determines forwarding
- Connection tear down: at end of call, two switches delete state concerning call

Data Networks: Asynchronous Link Multiplexing

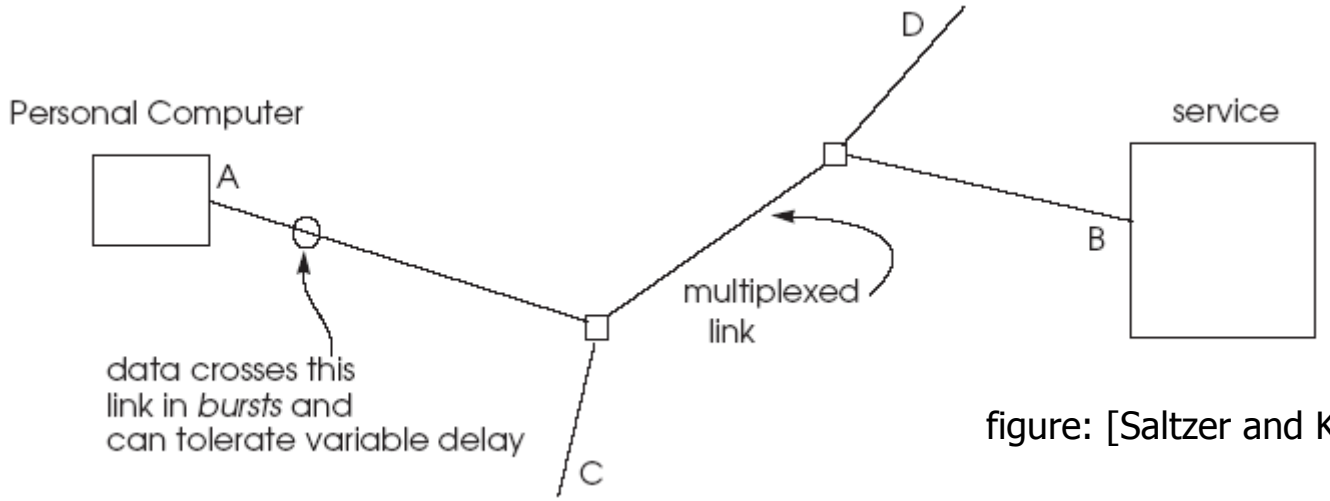

figure: [Saltzer and Kaashoek]

- Computers (and users) send data in bursts; not in a steady stream, like telephony
  - Sender may have nothing to send when its TDM slot comes around
  - Sender may have more than one frame to send at once, but can only use its own TDM slots
- More fitting to send data as they become available, rather than in scheduled fashion

Data Networks: Asynchronous Link Multiplexing

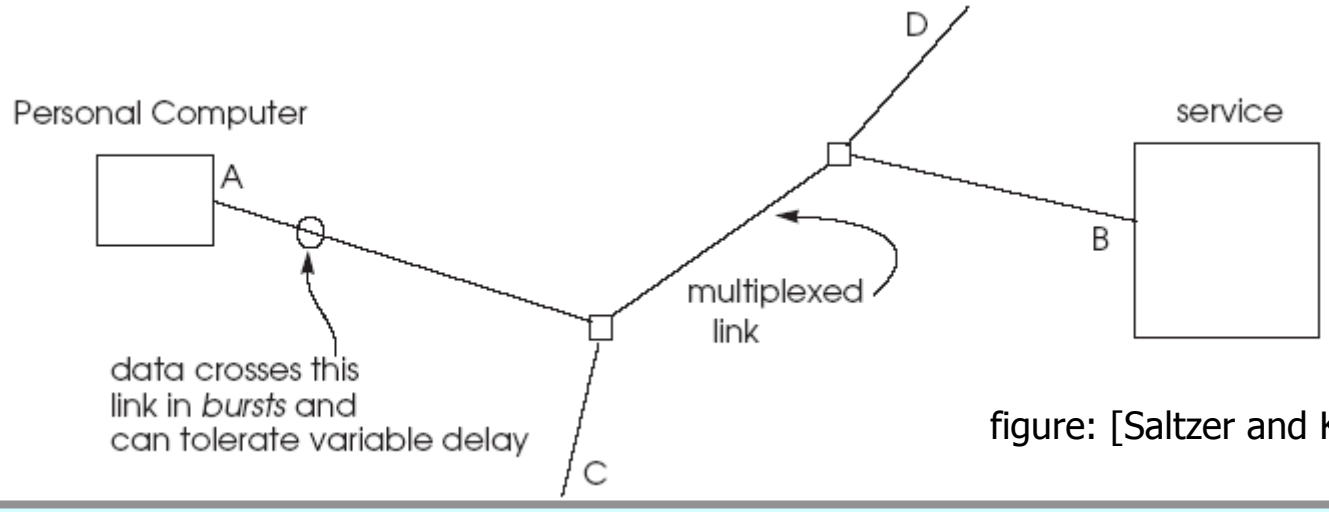

figure: [Saltzer and Kaashoek]

Asynchronous Multiplexing: give up predictable data rate and latency in favor of delivering entire message more quickly

- Sender may have more than one frame to send at once, but can only use its own TDM slots

- More fitting to send data as they become available, rather than in scheduled fashion

Async Link Multiplexing (cont'd)

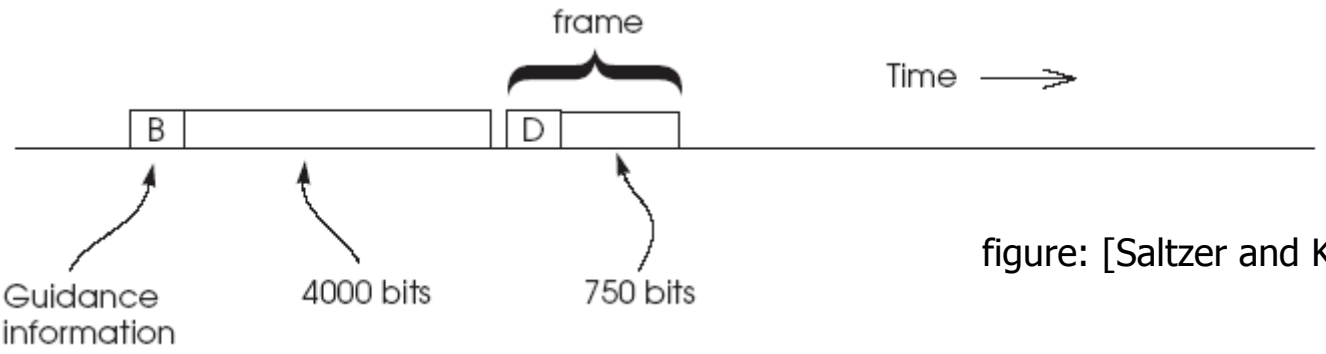

figure: [Saltzer and Kaashoek]

- Frames of any length (up to link limit)
- Frames may be sent anytime link not already in use
- Timing of frame doesn't imply forwarding; must explicitly include guidance information
- Variable-length frames require framing to demarcate inter-frame boundaries
- No need for per-connection state at switches; **connectionless**

Async Link Multiplexing (cont'd)

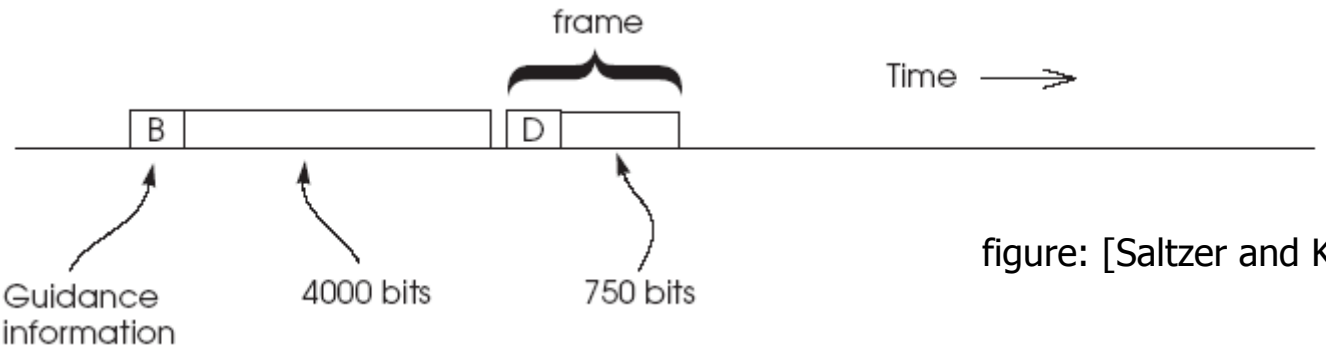

figure: [Saltzer and Kaashoek]

Asynchronous Multiplexing may be usable for telephony; depends on variability of delay and rate offered by network

include guidance information

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Packet Forwarding

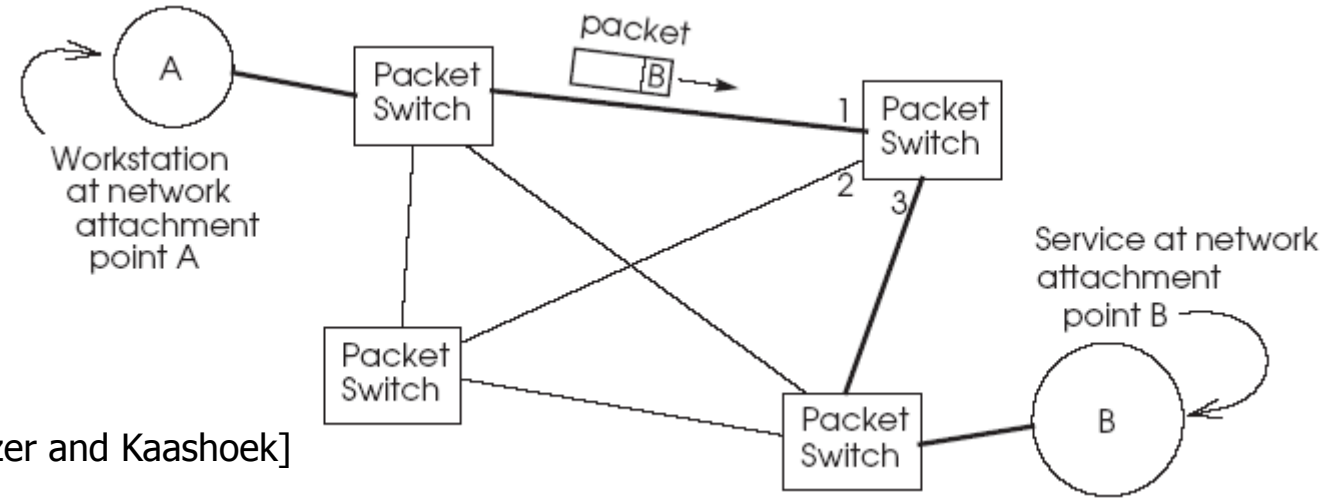

figure: [Saltzer and Kaashoek]

- Packet switches, or routers, connected by async links
- Multiple paths, or routes, possible
- Forwarding: examine destination of arrived packet, look it up in table, forward on appropriate link

Transit Time

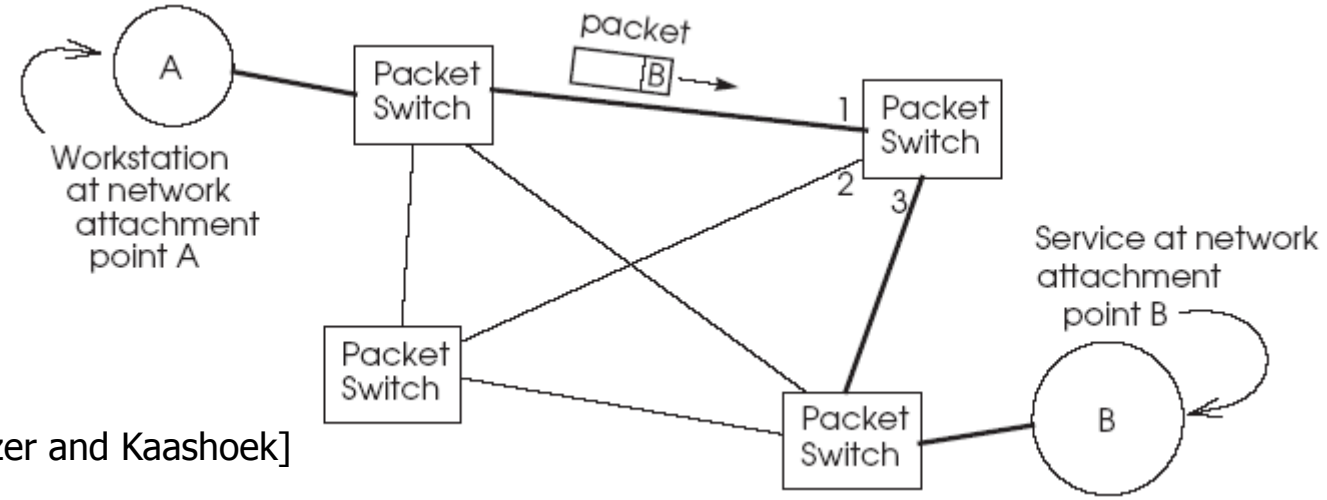

figure: [Saltzer and Kaashoek]

- Propagation delay: speed of light over medium across link length; fixed for given distance and medium
- Transmission delay: serialization of packet's bits at link rate on each transmission; varies depending on packet length and link speed

Transit Time (cont'd)

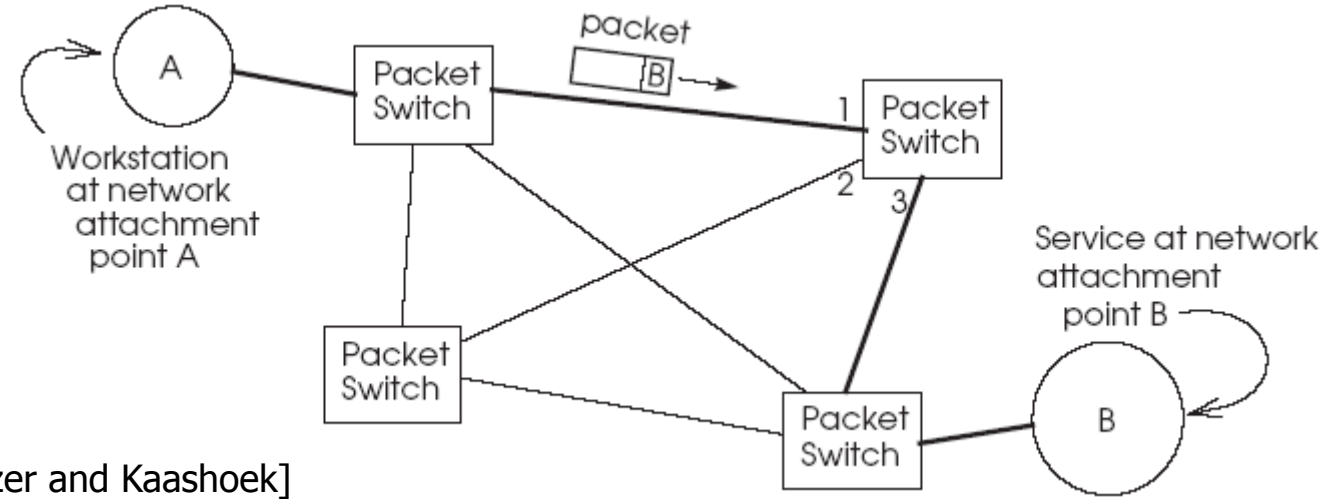

figure: [Saltzer and Kaashoek]

- Processing delay:
  - forwarding decision: fixed component
  - other packet processing (e.g., checksum, copying): varying, length-proportional component
- Queuing delay:
  - output link may be busy when packet arrives
  - store packet in memory, or queue
  - varies with total traffic volume and pattern

Diversion: Queuing Theory

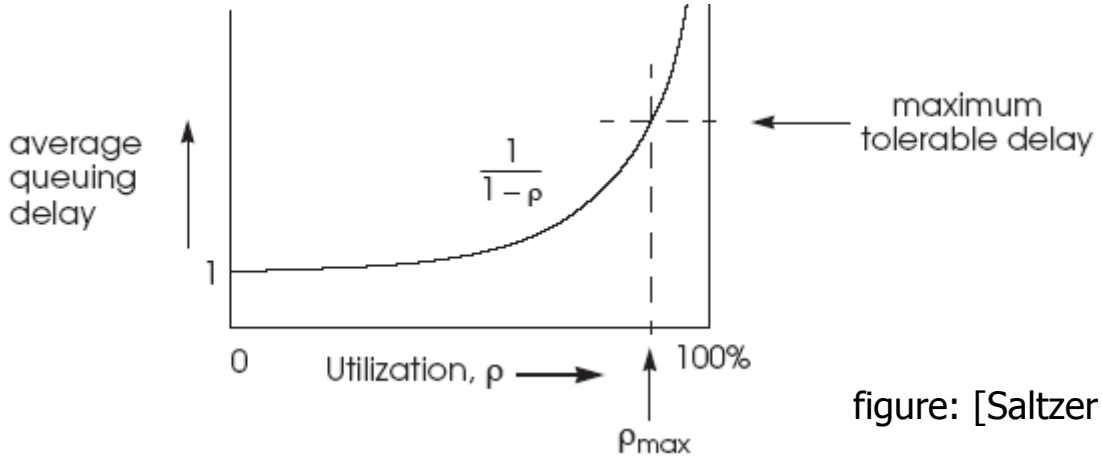

figure: [Saltzer and Kaashoek]

- How can we estimate queuing delay?
- Assume
  - Packets arrive according to random, memoryless process
  - Packets have randomly distributed service times (i.e., transmission delays)
  - Utilization of outgoing link is ρ
- Average queuing delay in packet service times (including this packet's) is 1/(1-ρ)

Utilization and Delay

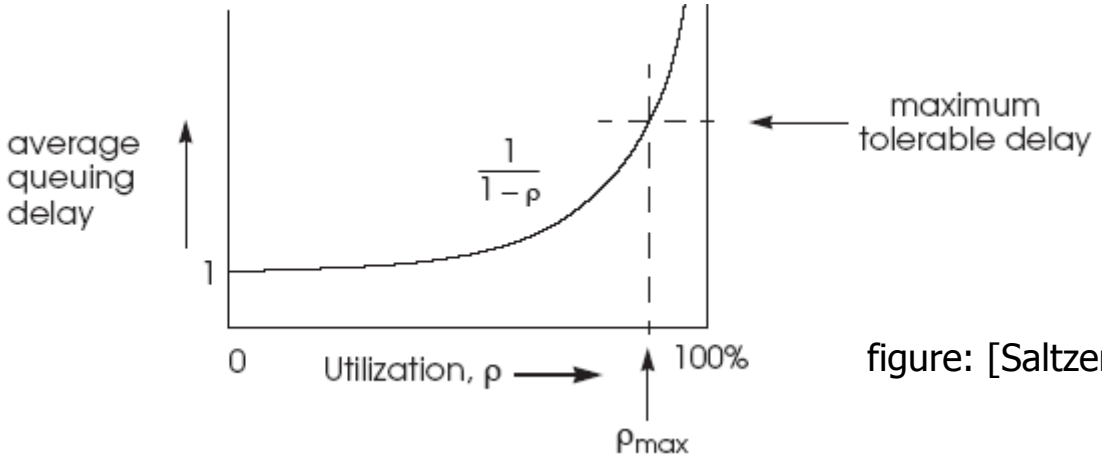

figure: [Saltzer and Kaashoek]

- Trade-off: bounding max delay also bounds max utilization
- Network operators like high utilization; links cost money
- Users like low latency; e.g., interactive voice
- Isochronous: abrupt exclusion of N+1st user
- Asynchronous: delay grows as load grows

Utilization and Delay

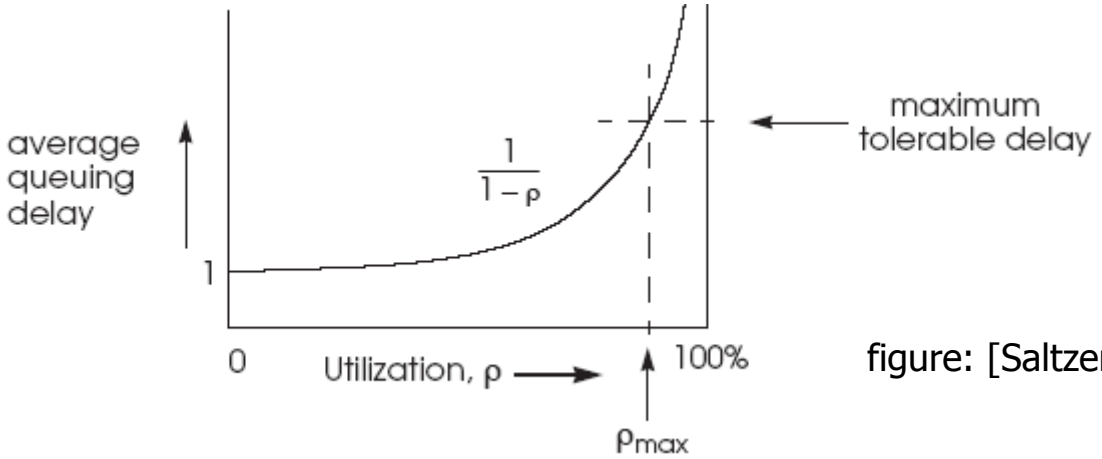

figure: [Saltzer and Kaashoek]

Delay is **average;** to bound worst-case delay, must target utilization below ρ_{max}

money

- Users like low latency; e.g., interactive voice
- Isochronous: abrupt exclusion of N+1st user
- Asynchronous: delay grows as load grows

Queuing Theory: Final Words

- Utilization/delay trade-off true throughout computer systems (CPU, disk scheduling)
- Warning: queuing theory assumptions don't hold on real data networks!
  - Packet arrivals not Poisson process; burstier
  - Router queues of finite length; bounds queuing delay

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Queue Sizes (and Packet Drops)

- A router's packet queue is just memory
- How much memory does a router need?
- Strategies:
  - Plan for worst case: enough memory to buffer longest possible queue length
  - Plan for average case, slow senders: enough memory for common case; when queue full, tell senders to slow down
  - Plan for average case, drop extra packets: enough memory for common case; when queue full, drop extras

Worst-Case Memory Size

- Memory is relatively cheap
- How can we predict worst-case queue size?
- Bursts of load caused by users (and bugs in code!)
  - highly unpredictable
  - orders of magnitude worse than average case
- Very long queues mean very long delays
  - Would you wait 2 minutes to learn if you had new email?

Average-Case Memory Size, with Quench

- Central worry is congestion: sometimes, memory not big enough, queue will fill
- When queue fills, send a quench message on incoming link, asking sender (router or host) to slow down
- Problems:
  - Respond to congestion by generating traffic?
  - Whom should be quenched?
  - Quenched source may no longer be sending
  - Quench itself delayed by queuing; worse congestion is, longer the delay
- Essentially not used in practice

Average-Case Memory Size, with Dropping

- When queue full, drop packets!
- Some entity must resend dropped packet
- **Lack of end-to-end acknowledgement now indicates congestion!**
  - Implicit signal to sender—without adding to traffic load
  - Introduces possibility of sender slowing automatically in response to congestion
- What the Internet does

Isochronous vs. Asynchronous

- Isochronous:
  - when capacity remains, get fixed transmission rate, independent of other users' activity
  - when capacity fully subscribed, get nothing
- Asynchronous:
  - transmission rate depends on instantaneous load from those with whom you share links
- Preferable regime depends on application
  - After earthquake, better to be able to send "I'm alive" slowly than not to be able to send at all
  - Remote surgery requires guaranteed bit rate

Best Effort

- Networks that never discard packets termed guaranteed-delivery
  - more mechanism required to guarantee (and track) delivery than to drop packets
- Networks willing to discard packets under congestion termed best-effort
- Fuzzy meaning of "best effort": far greater chance of undetected loss than with guaranteed delivery
- Internet delivers packets best-effort
- Internet email guarantees delivery (atop best-effort packet delivery service)

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Packet Duplication

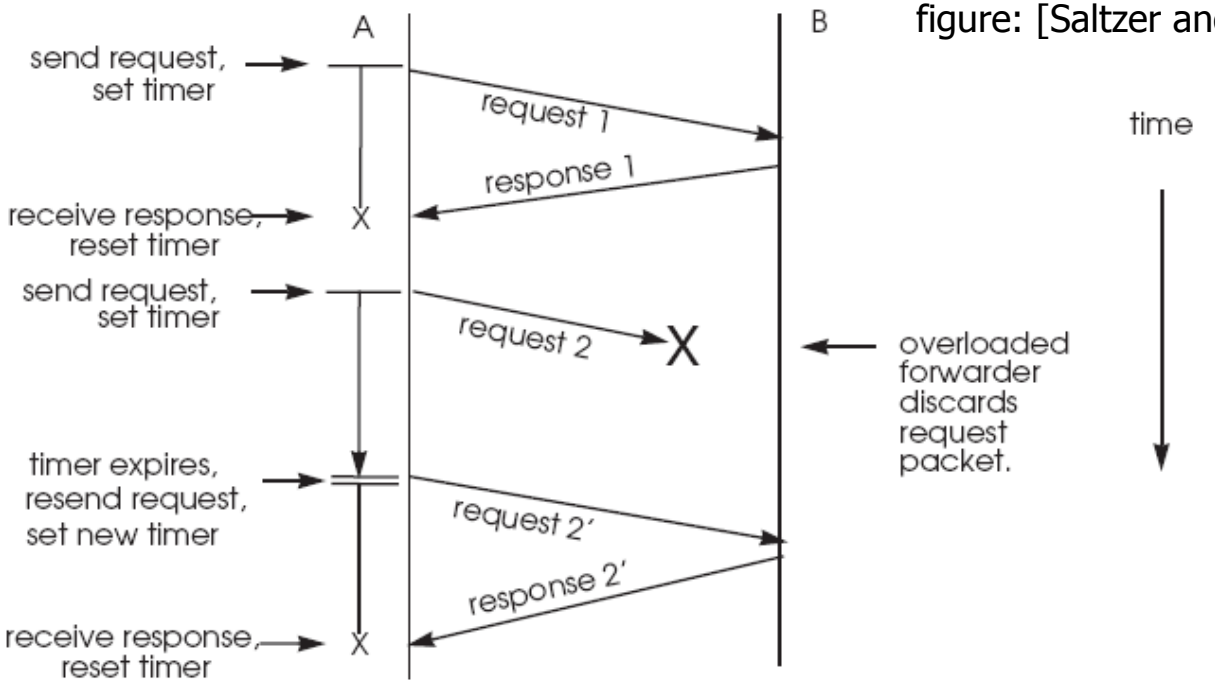


figure: [Saltzer and Kaashoek]

- Best-effort delivery drops packets when queues overflow
- End-to-end principle suggests original sender should retry
  - destination could also be down

Packet Duplication (cont'd)

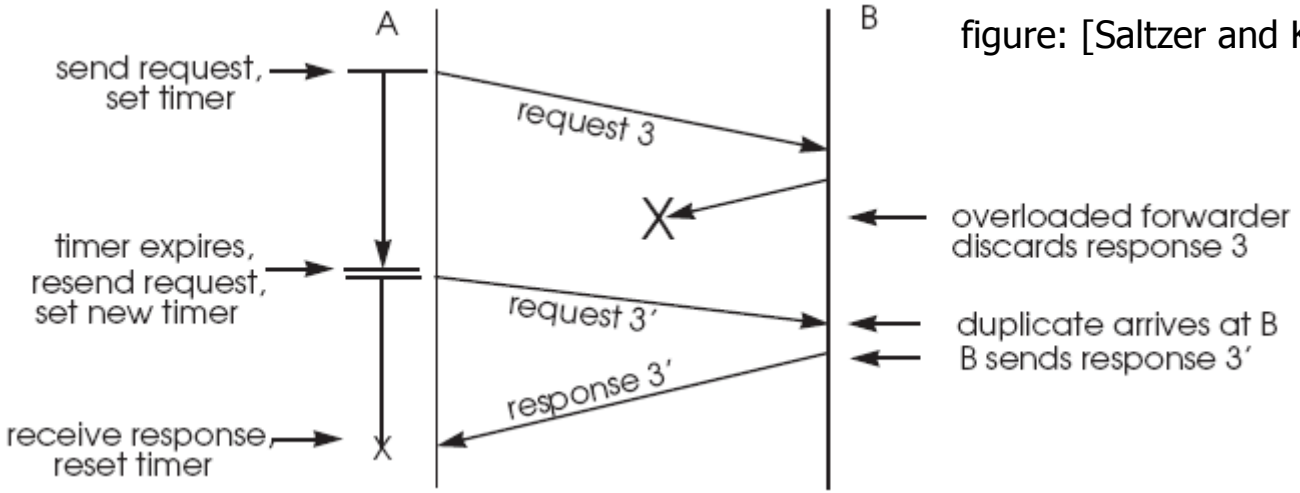


figure: [Saltzer and Kaashoek]

- Responses can be dropped
- Consequence: client's resent request appears as duplicate to server
- Problem? Depends on application and request type:
  - Bank withdrawal
  - File write

Packet Duplication (cont'd)

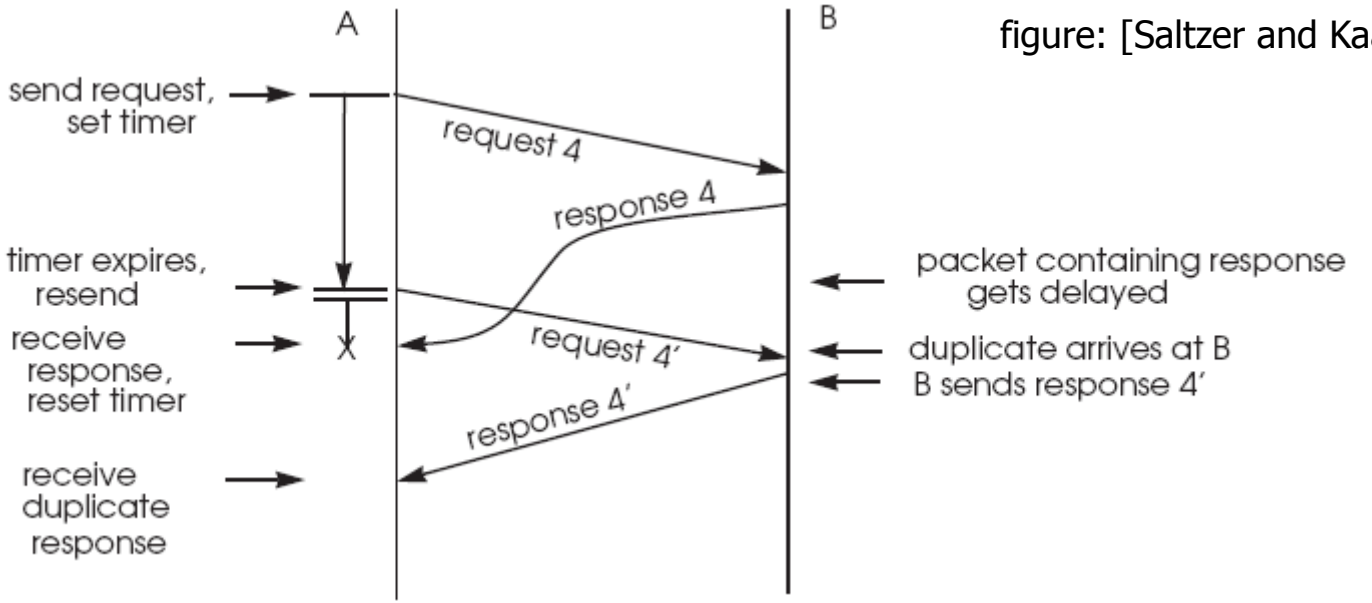


figure: [Saltzer and Kaashoek]

- Delay can also cause duplication

Duplicate Suppression

- A marks requests with monotonically increasing sequence numbers (any non-repeating sequence suffices)
- B remembers request sequence numbers for requests previously processed
- B ignores requests with sequence numbers already completed; repeats response to A

Packet Corruption and Link Breakage

- Noise on links, errors in router memory, software errors, *&c.,* may corrupt packets en route
  - Given best-effort service, error detection is sufficient; drop packets that contain errors
- Links may break (excavation cuts fiber; power failure)
  - Networks typically offer more than one path; routing must find a currently working path

Packet Reordering

- Consider two paths, R1 and R2, with delays D1 and D2, where D1 < D2
  - Sender may send packet P0 along R2
  - Sender may send packet P1 along R1
  - Packets may arrive at receiver in order P1, P0
- For messages divided into packets, reassembly at receiver must be done with care

Summary: Multi-Hop Networks

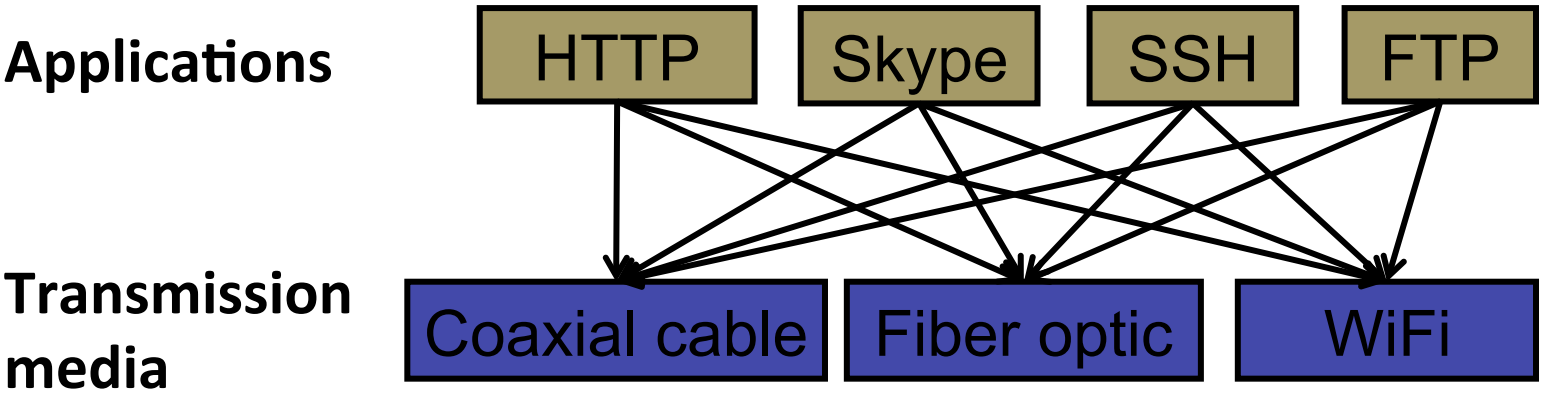
Network Type	Application characteristics: Continuous stream (e.g., interactive voice)	Application characteristics: Bursts of data (most computer-to-computer data)	Response to load variations
isochronous (e.g., telephone network)	good match	wastes capacity	(hard-edged) either accepts or blocks call
asynchronous (e.g., Internet)	variable latency upsets application	good match	(gradual) 1. variable delay 2. discards data 3. rate adaptation

figure: [Saltzer and Kaashoek]

Outline

- Shared, Multi-hop Networks
- Background: Protocol Layering
- End-to-End Arguments

Dealing with Heterogeneity of Applications and Links



- Re-implement every application for every new underlying transmission medium?
- Change every application on any change to an underlying transmission medium (and vice-versa)?

- **No!** But how does the Internet design avoid this?

Computer system modularity

- **Key idea: Partition system into modules and abstractions**

- Well-defined interfaces give flexibility and isolation
  - Hide implementation, thus, it can be freely changed
  - Extend functionality of system by adding new modules

- *e.g.*, libraries encapsulating set of functionality

- *e.g.*, a programming language and compiler abstracts away how a particular CPU and operating system work

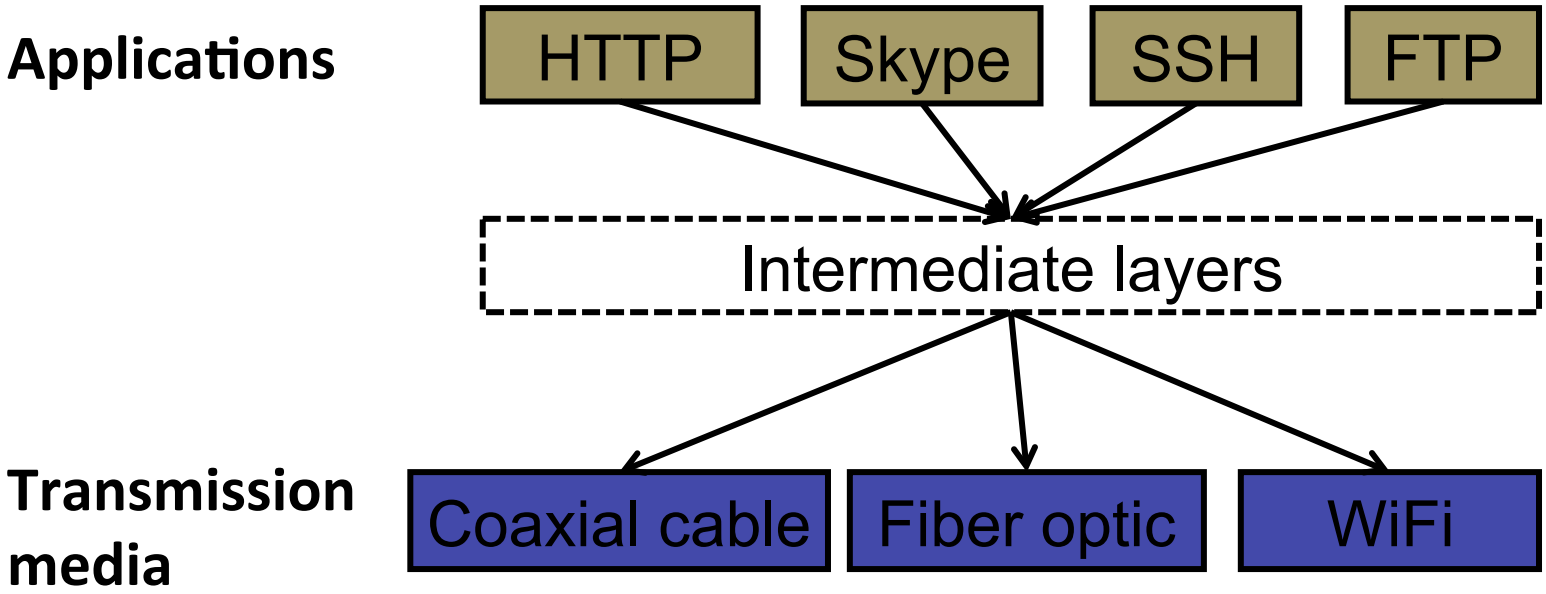
Layering: a modular approach

- Partition protocols on the Internet into **layers**
  - Each layer solely relies on services from layer below
  - Each layer solely exports services to layer above

- Advantages of layering:

1. Decomposes problem of building a network into manageable pieces
2. Results in a more modular design. Additions or changes are usually isolated to one layer
3. Layer *n* hides complexity of layer *n*–1 to higher layers

Internet solution: Intermediate layers



- Intermediate layers provide a set of abstractions for applications and media
- New applications or media need only implement for intermediate layer's interface

Physical layer (L1)

- Service: move bits between two systems connected by a single physical link
- Interface: specifies how to send, receive bits
  - *e.g.*, require quantities and timing
- Protocols: coding scheme used to represent bits, voltage levels, duration of a bit

Data link layer (L2)

- Service: enables end hosts to exchange atomic messages with one another
  - Using abstract addresses (*i.e.*, not just direct physical connections)
  - Perhaps over multiple physical links, but using the same framing (headers/trailers)
  - Possibly arbitrates access to common physical media
  - Possibly implements reliable transmission, flow control
- Interface: send messages (frames) to other end hosts; receive messages addressed to end host
- Protocols: addressing, routing, Media Access Control (MAC) (e.g., CSMA/CD - Carrier Sense Multiple Access / Collision Detection)

Network layer (L3)

- Service: Deliver packets to destinations on other networks (inter-network, across multiple L2 networks)
  - Works across networking technologies (*e.g.*, Ethernet, 802.11, frame relay, ATM …)
    - No longer the same framing all the way
  - Possibly includes packet scheduling/priority
  - Possibly includes buffer management
- Interface: send packets to specified internetwork destinations; receive packets destined for end host
- Protocols: define inter-network addresses (globally unique); construct routing tables

Transport layer (L4)

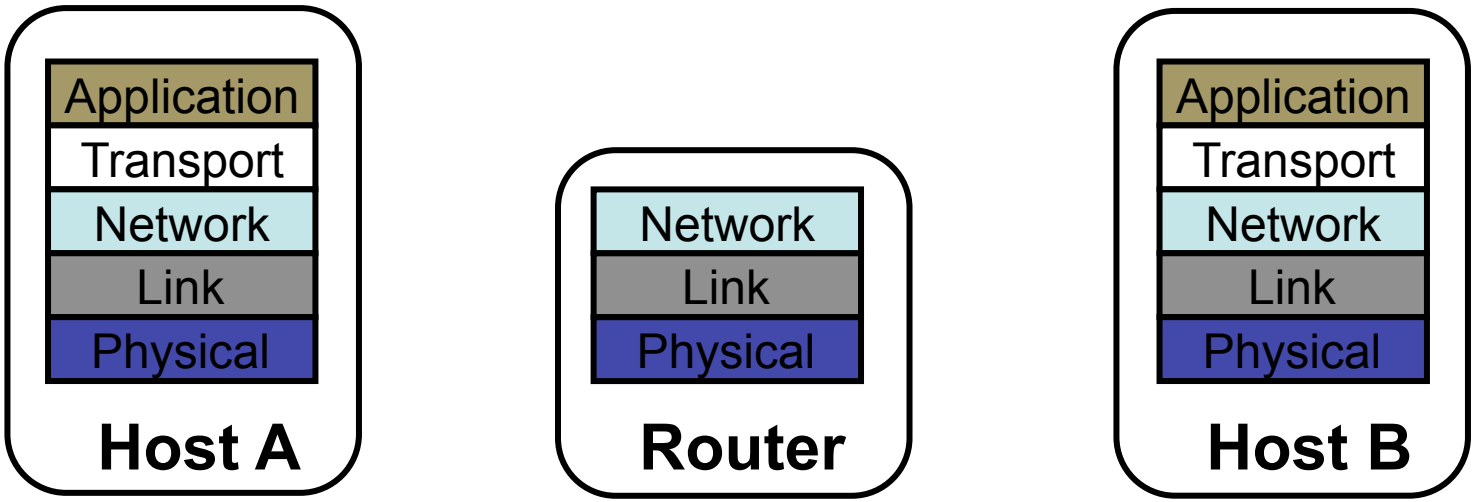
- Service: Provides end-to-end communication between processes on different hosts
  - Demultiplexing of communication between hosts
  - Possibly reliability in the presence of errors
  - Timing properties
  - Rate adaption (flow-control, congestion control)
- Interface: send message to specific process at given destination; local process receives messages sent to it
- Protocol: perhaps implement reliability, flow control, packetization of large messages, framing
- Examples: Transport Control Protocol (TCP), Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP)

Application layer (L7)

- Service: any service provided to the end user
- Interface: depends on the application
- Protocol: depends on the application
- Examples: File Transfer Protocol (FTP), Skype, Simple Mail Transfer Protocol (SMTP), Hypertext Transport Protocol (HTTP), BitTorrent, many others…
- What happened to layers 5 & 6?
  - "Session" and "Presentation" layers
  - Part of OSI architecture, but not Internet architecture

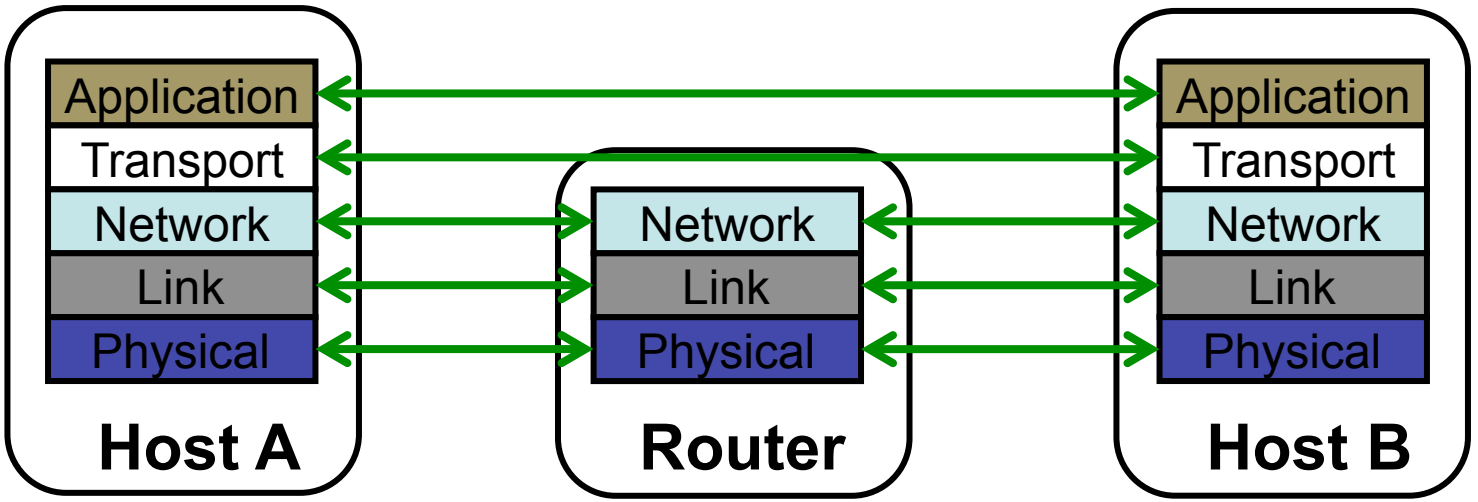
Who does what?

- Five layers
  - Lower three layers are implemented everywhere
  - Top two layers are implemented only at end hosts



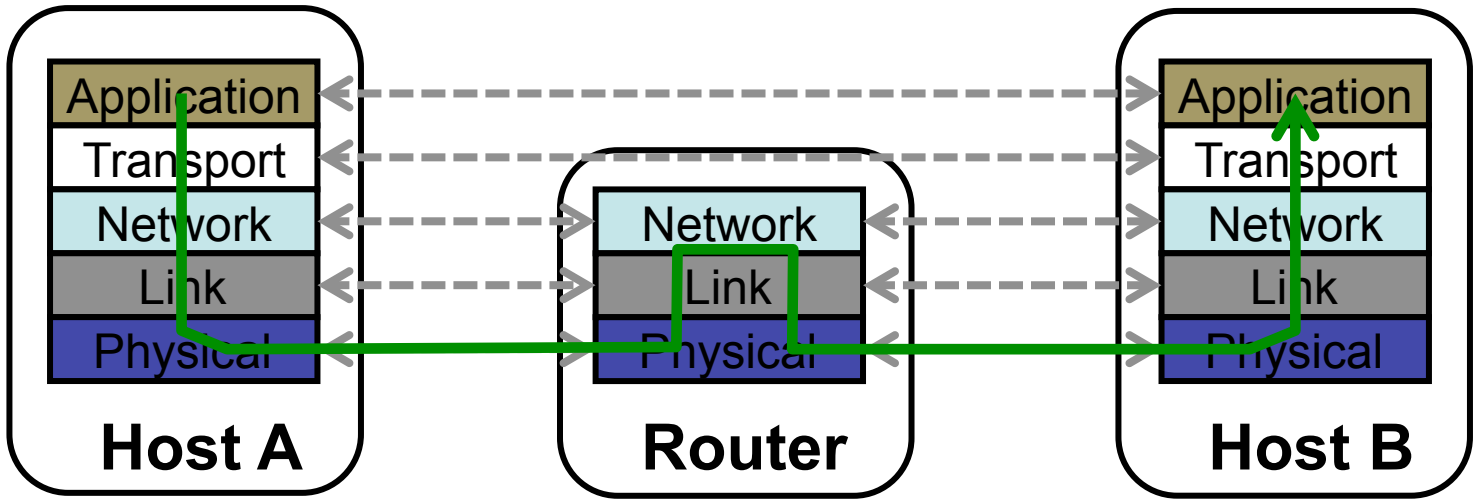
Logical communication

- Each layer on a host interacts with its **peer** host's corresponding layer via the protocol interface



Physical communication

- Communication goes down to physical network
- Then from network peer to peer
- Then up to the relevant layer



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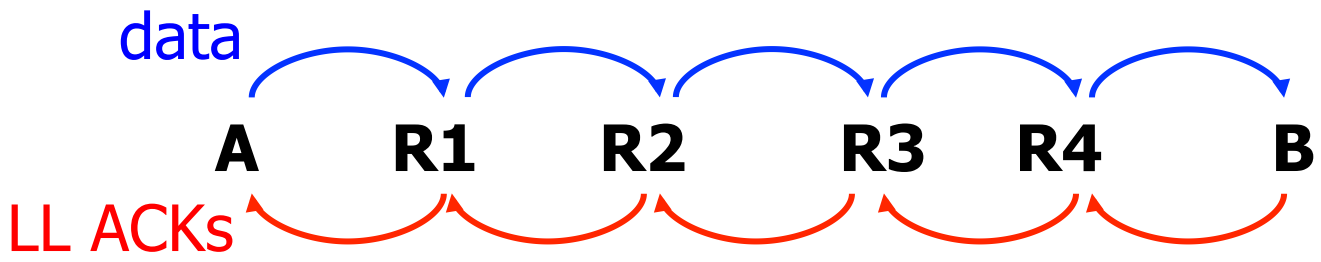
Motivation: End-to-End Argument

- 7 layers in OSI model
- 7 places to solve many of same problems:
  - In-order delivery
  - Duplicate-free delivery
  - Reliable delivery (retransmission) after corruption or loss
  - Encryption
  - Authentication
- In which layer(s) should a particular function be implemented?

Example: Careful File Transfer



- Goal: accurately copy file on A's disk to B's disk
- Straw man:
  - Read file from A's disk
  - A sends stream of packets containing file data to B
  - Link-layer retransmission of lost/corrupted packets at each hop
  - B writes file data to disk
- Does this system meet design goal?
  - Bit errors on links not a problem

Where Can Errors Happen?

- On A's or B's disk
- In A's or B's RAM or CPU
- In A's or B's software
- In the RAM, CPU, or software of any router that forwards packet **(MIT example!)**
- Why might errors be likely?
  - Drive for CPU speed and storage density: pushes hardware to EE limits, engineered to tight tolerances
  - e.g., today's disks return data that are the output of an MLE!

Solution: End-to-End Verification

- A stores checksum with data on disk
  - Why not compute freshly on read?
- B computes checksum over received data, sends to A (or vice-versa)
- Compare two checksums; A resends if not identical
- Can we eliminate hop-by-hop error detection?
  - Suppose there's a router with bad RAM; how will you find it?
- Is a whole-file checksum enough?
  - Poor performance: must resend whole file each time one packet (bit) corrupted!

End-to-End Principle

- Only application at communication endpoints can completely and correctly implement a function
- Processing in middle alone cannot provide function
- Processing in middle may, however, be important performance optimization
- Engineering middle hops to provide guaranteed functionality often wasteful of effort and inefficient

Perils of Low-Layer Implementation

- Entangles application behavior with network internals
- Suppose each IP router reliably transmits to next hop
  - lossless delivery, variable delay
  - ftp: OK, move huge file reliably (just end-to-end TCP works fine, too, though)
  - Skype: terrible, jitter packets when a few corruptions or drops not a problem anyway
- Complicates deployment of innovative applications
  - phone network vs. Internet

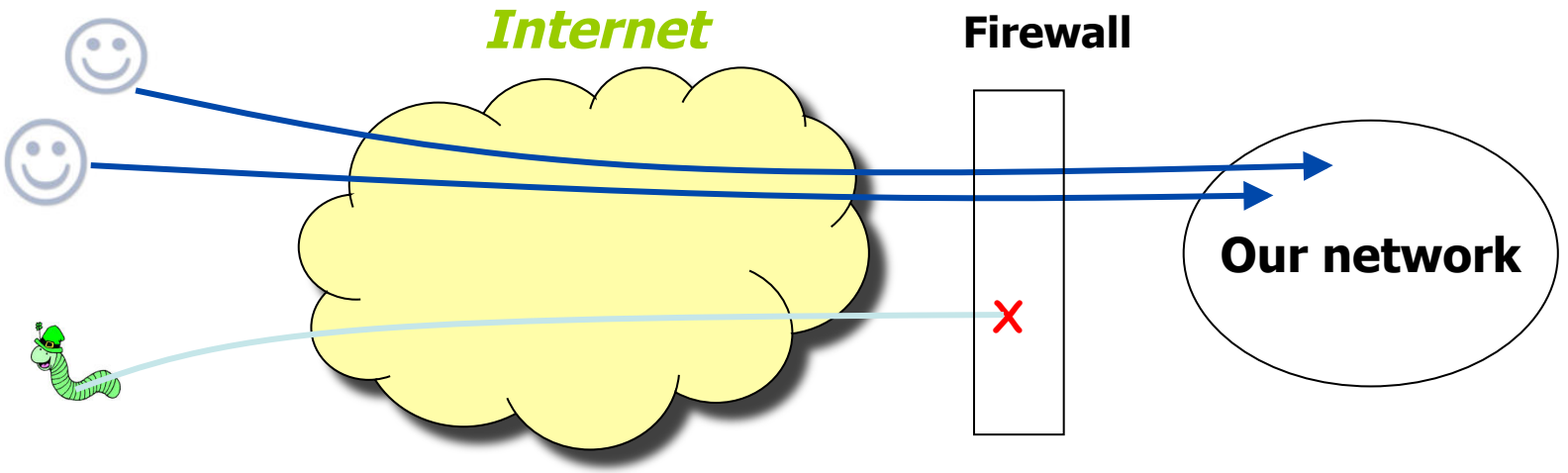
Advantages of Low-Layer Implementation

- Each application author needn't recode a shared function
- Overlapping error checks (e.g., checksums) at all layers invaluable in debugging and fault diagnosis
- If end systems not cooperative (increasingly the case), only way to enforce resource allocation!

Challenge: End-to-End Authentication and Encryption

- Use a public PC to check your email using IMAP & SSL
  - Authenticates server to you and you to server robustly
  - Encrypts between you robustly
- Key security consideration: threat model
  - which attacks are you explicitly defending against?
  - which are you ignoring?
  - what does it cost your adversary to mount an attack?
- What are you trusting?
  - mail reading application (could be trojaned)
  - OS (could also be trojaned)
  - hardware (e.g., "fake ATM" cases)
- End-to-end notion of security must consider integrity of software and hardware at endpoints, possibly even of users!

End-to-End Violation: Firewalls



- Box in middle of network that blocks "malicious" traffic
  - End-host software often vulnerable to worms
  - Users naive, may not keep desktop patched up-to-date
- Clearly violates e2e principle
  - Endpoints capable of deciding what traffic to ignore
  - Firewall entangled with design of network and higher protocol layers and apps, and vice-versa
  - Example: new ECN bit to improve TCP congestion control; many firewalls filtered all such packets!
- Probably need firewalls
- But beware entangling network edge and network interior

Summary: End-to-End Principle

- Many functions **must** be implemented at application endpoints to provide desired behavior, even if implemented in "middle" of network
- End-to-end approach **decouples design** of components in network interior from design of applications at edges
- Some functions still benefit from implementation in network interior; at cost of entangling interior and core
  - Performance (e.g., link-layer retransmission)
  - Security (e.g., firewalls)
  - Scalability (e.g., routing)
- End-to-end principle is **not sacred;** it's a way to think critically about design choices in communication systems