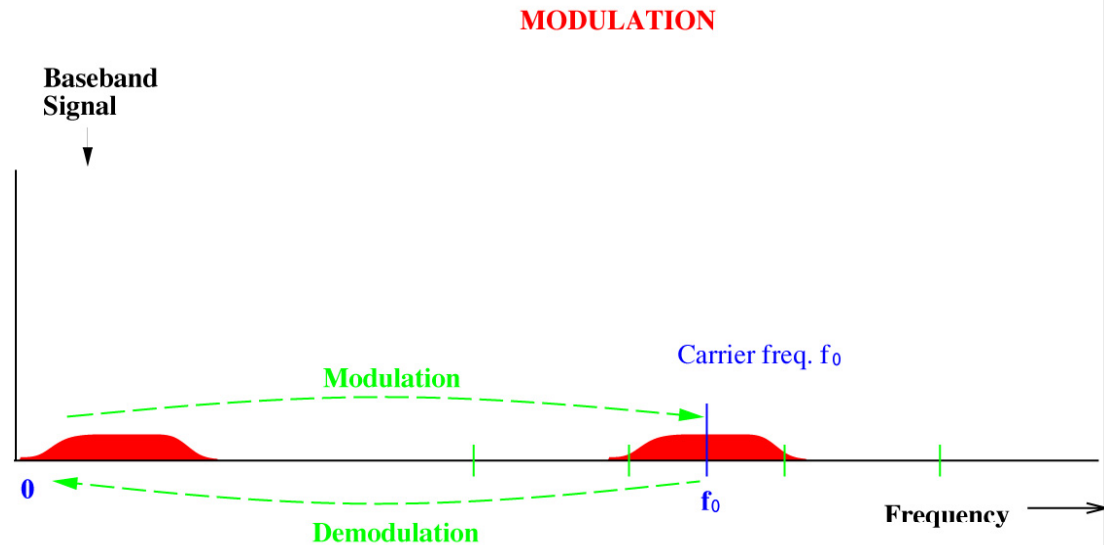


Introduction to Networks

Modulation and Demodulation

- Common examples: radio, television channels for **analog** signals

- Bandwidth in hertz



- Can also be used for **digital** signals (**encoding** binary data)

CARRIER WAVEFORM



$$A \cos(2\pi f_0 t + \theta)$$

Shannon's Theorem

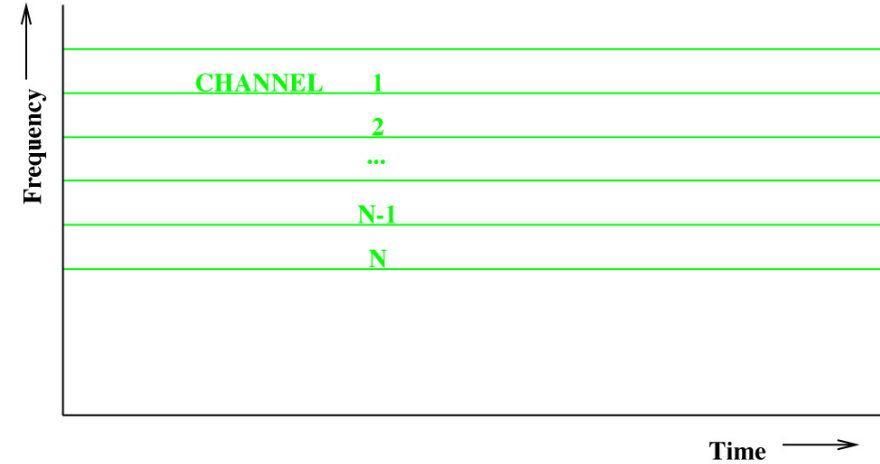
$$C = B \log_2 (1 + S/N)$$

where

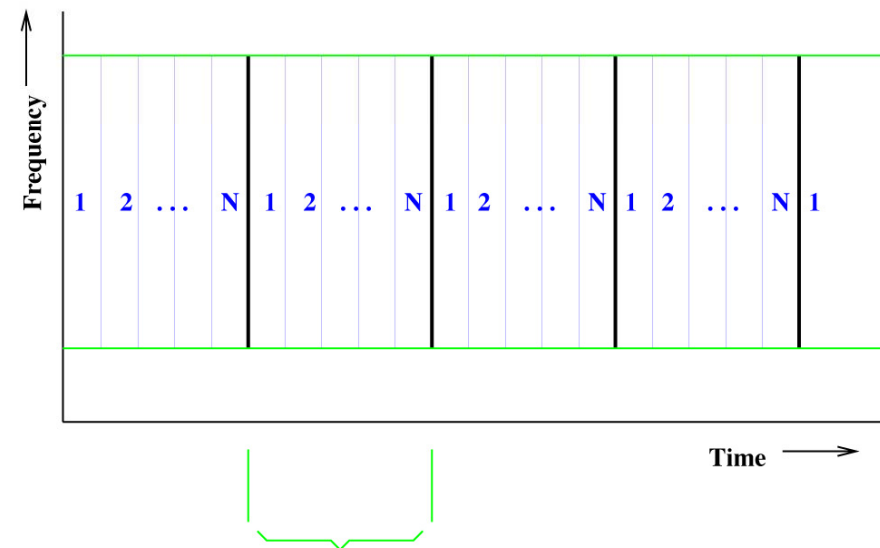
C	max capacity in bits/sec
B	bandwidth in hertz
S/N	signal to noise ratio

FDM vs. TDM

FDM



TDM



Duration of frame (or superframe) is 125 μsec in digital telephone networks

Duration of frame or superframe (125 μsec in digital telephone network)

1/24/2017

Intro to networks (Simon S. Lam)

1-4

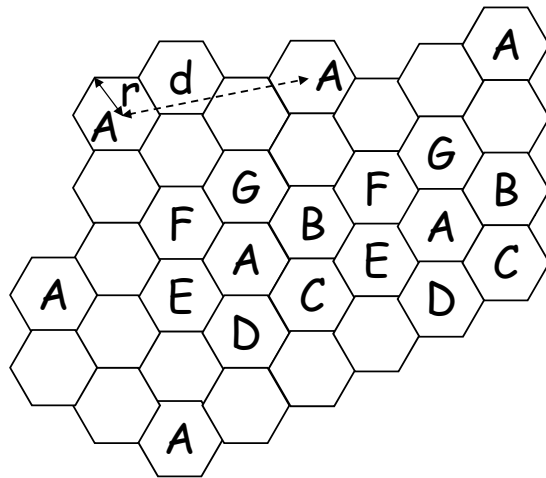
TDM in Telephone Networks

- ❑ Why 125 μ sec for frame duration?
- ❑ **Sampling Theorem:**
An analog signal can be reconstructed from samples taken at a rate equal to twice the signal bandwidth
- ❑ Bandwidth for voice signals is 4 KHz; for hi fidelity music, 22.05 KHz per channel
- ❑ Sampling rate for voice = 8000 samples/sec or one voice sample every 125 μ sec
- ❑ Digital voice channel (uncompressed),
8 bits $\times$ 8000/sec = 64 Kbps

Other Multiplexing Techniques

❑ Space division multiplex

- Same frequency used in different cables
- Same frequency used in different (nonadjacent) cells



❑ Wavelength division multiplex

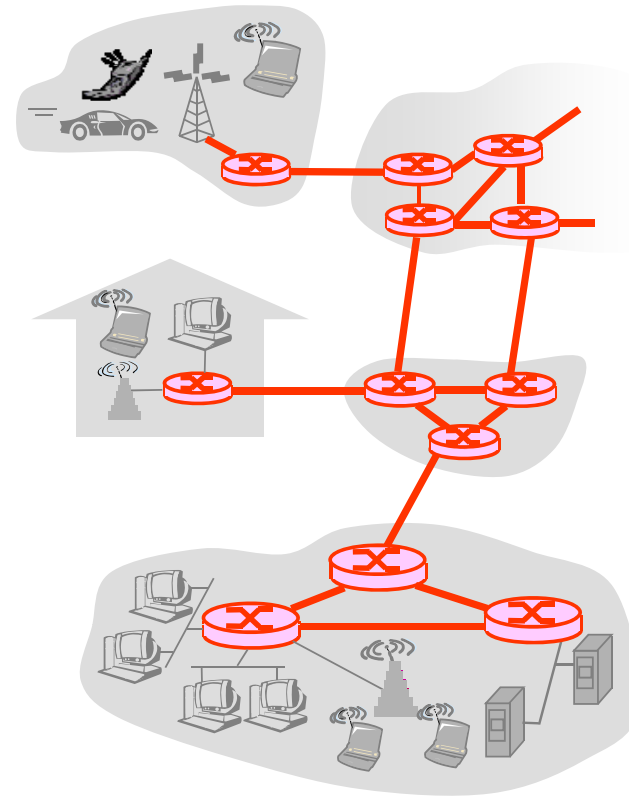
- ❖ Light pulses sent at different wavelengths in optical fiber

❑ Code division multiplex

e.g., CDMA for cell phones

The Network Core

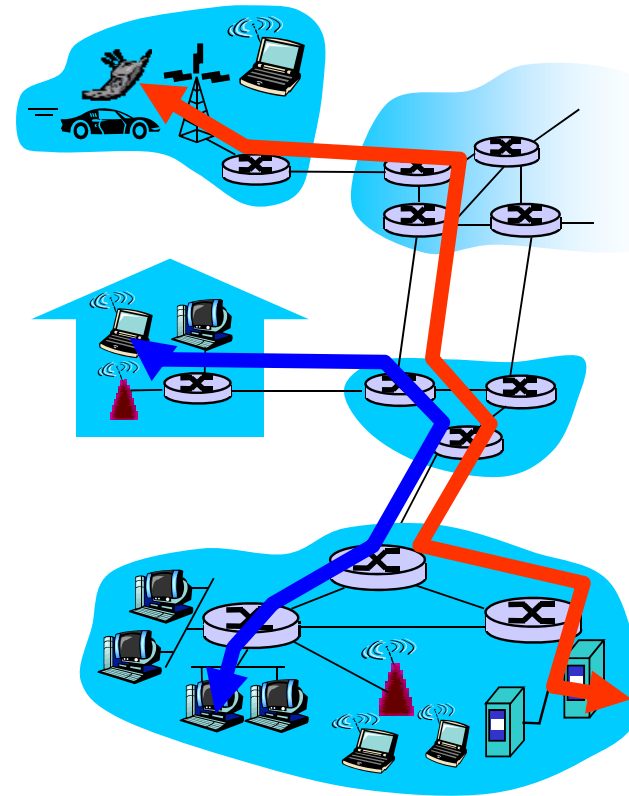
- ❑ mesh of interconnected routers
- ❑ the fundamental question: how is data transferred through net?
 - circuit switching: dedicated circuit per call: telephone net
 - packet-switching: data sent thru net in discrete "chunks"



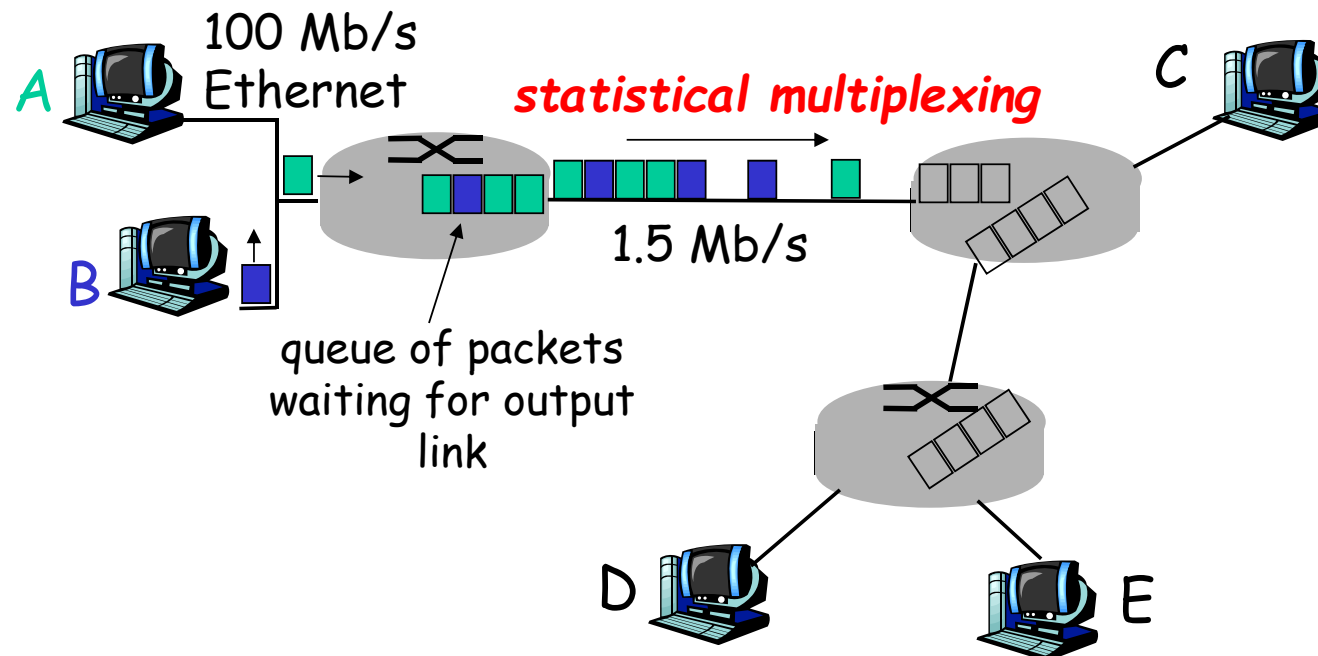
Network Core: Circuit Switching

End-to-end resources reserved for each "call"

- ❑ E.g., link bandwidth
 - FDM, TDM
- ❑ end-to-end circuit-like (guaranteed) performance
- ❑ call setup required
 - resource piece *idle* if not used by the call (no sharing)
 - *state information each step along the way*



Packet Switching: Statistical Multiplexing



- ❑ Sequence of A & B packets does not have fixed pattern
bandwidth shared on demand → *statistical multiplexing*
- ❑ queueing delay, packet loss

Network Core: Packet Switching

*each end-end data stream
divided into packets*

- ❑ packets of different users share network resources
- ❑ each packet uses full link bandwidth

Bandwidth division into "pieces"
Dedicated allocation
Resource reservation



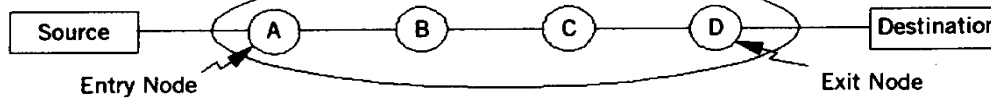
resource contention:

- ❑ aggregate resource demand can exceed amount available

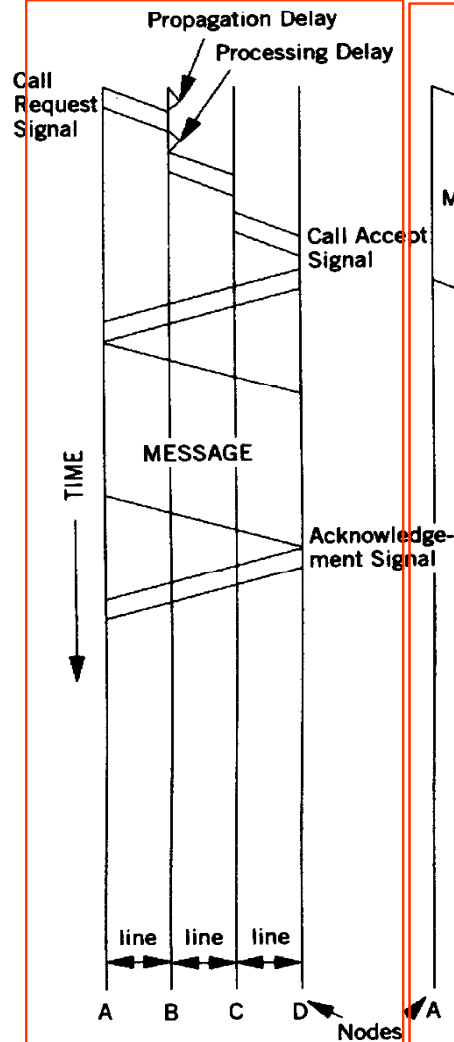
congestion: packets queue, wait for link use

- ❑ **store and forward:** packets move one hop at a time

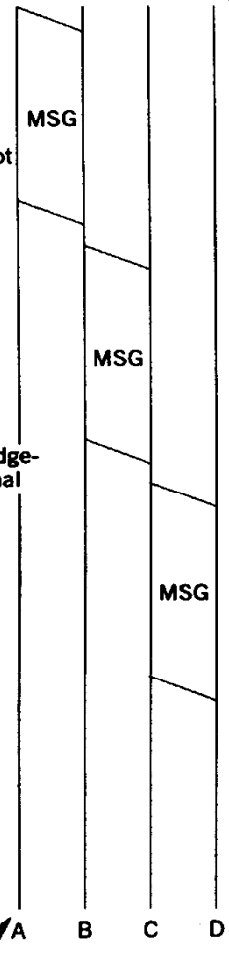
❖ Each node receives the **complete** packet before forwarding it



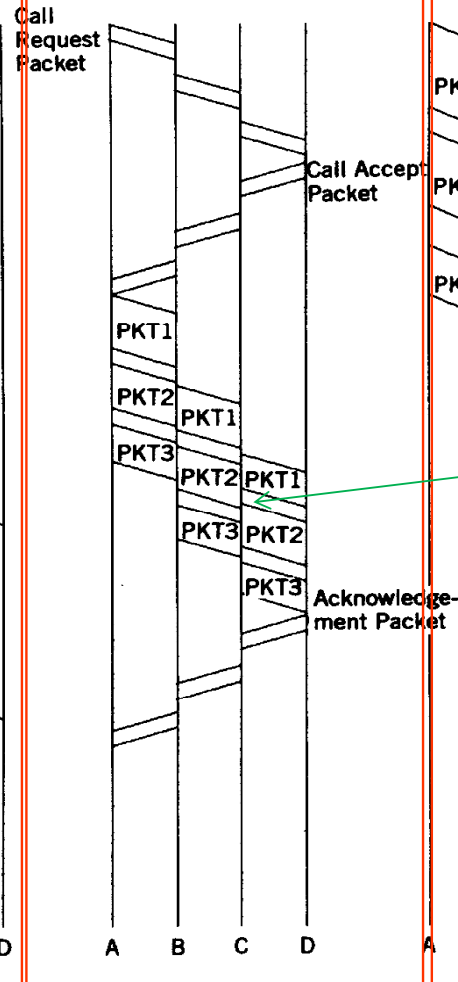
(a) Communication Network



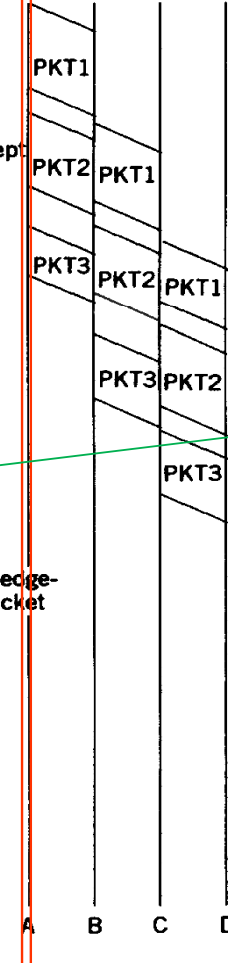
(b) Circuit Switching



(c) Message Switching



(d) Virtual Circuit
(Packet) Switching



(e) Datagram Packet
Switching

Circuit vs. Message vs. Packet Switching

violates store-and-forward?

Packet switching versus circuit switching

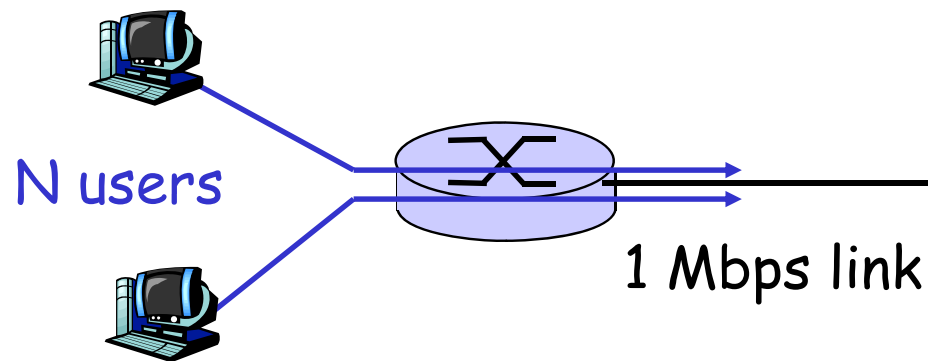
- ❑ 1 Mb/s link
- ❑ each user:
 - 100 kb/s when "active"
 - active 10% of time (a "bursty" user)

- ❑ *circuit-switching:*

- 10 users

- ❑ *packet switching:*

- with 35 users, probability > 10 active at same time is less than .0004



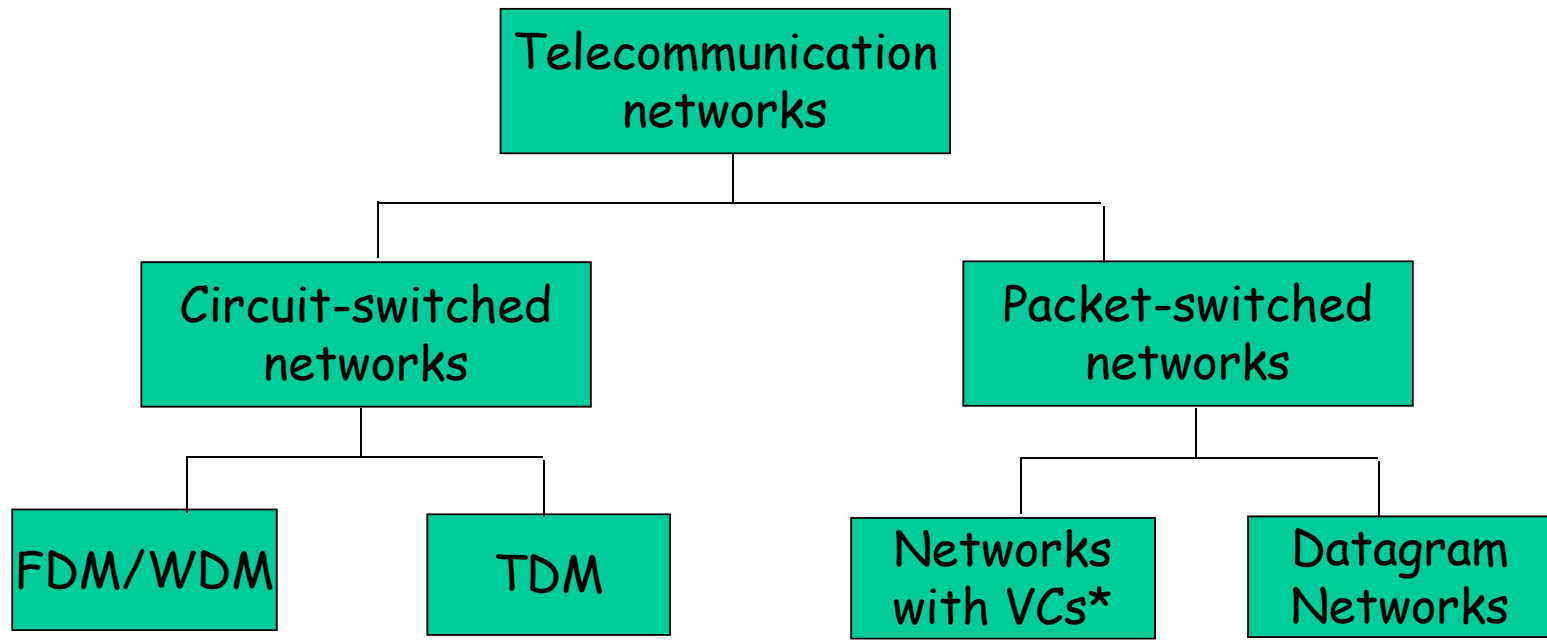
Q: how did we get value 0.0004?

Packet switching versus circuit switching

Is packet switching a "slam dunk winner?"

- ❑ great for bursty data
 - resource sharing
 - simpler, no call setup
- ❑ excessive congestion -> packet delay and loss
 - protocols needed for reliable data transfer, congestion control
- ❑ Q: How to provide circuit-like behavior?
 - bandwidth guarantees needed for
 - interactive audio/video apps
 - providing virtual links to enterprise network customers (under service contracts)

Network Taxonomy



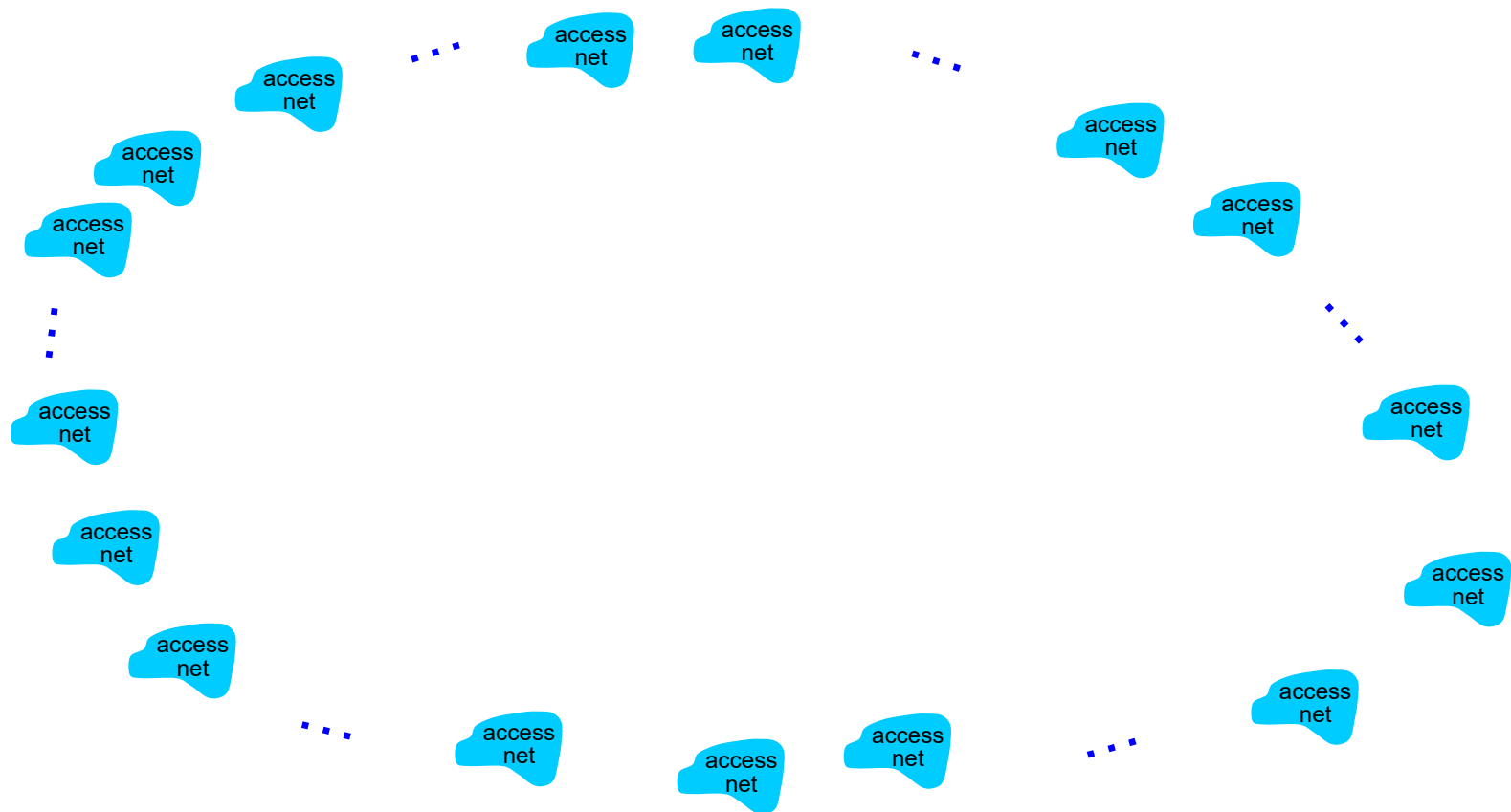
Any technology can be used in
link layer of Internet under IP

Internet
won!

VC examples: ATM networks, MPLS tunnels

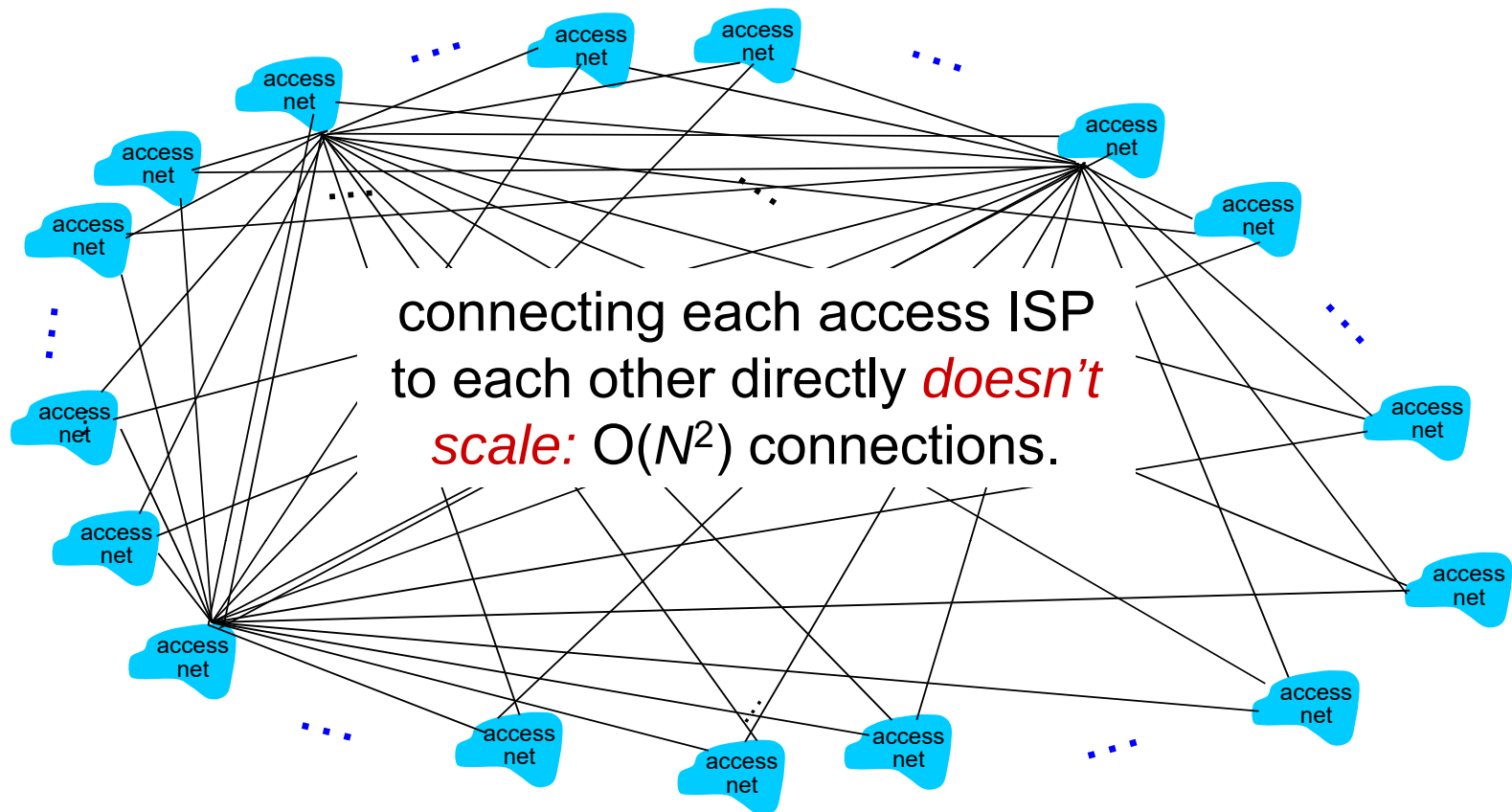
Internet structure: network of networks

Question: given millions of access ISPs, how to connect them together?



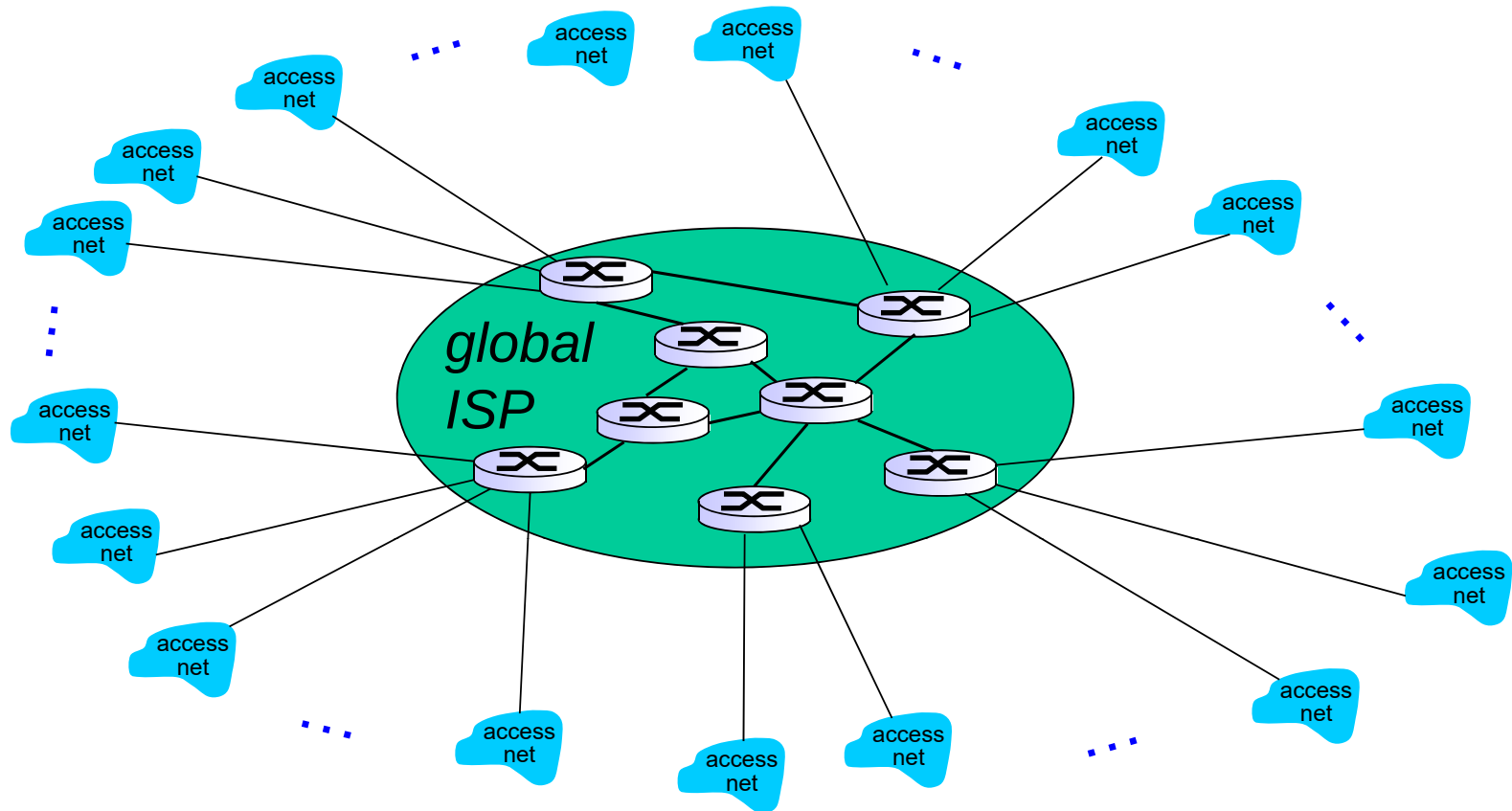
Internet structure: network of networks

Option: connect each access ISP to every other access ISP?



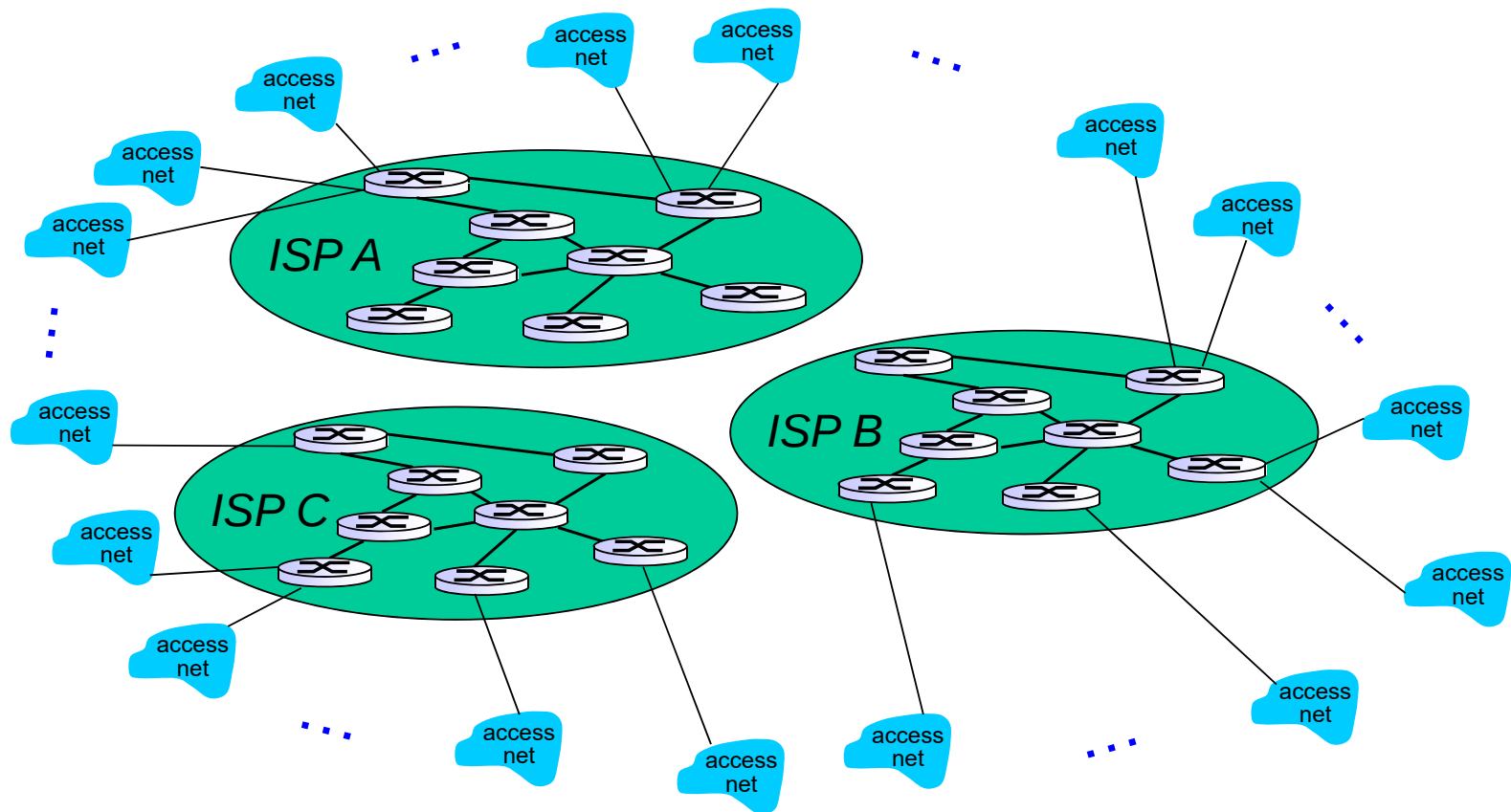
Internet structure: network of networks

1970-1991: connect each access ISP to a global transit ISP:
1. Financed by US government: **ARPAnet, NSFnet**



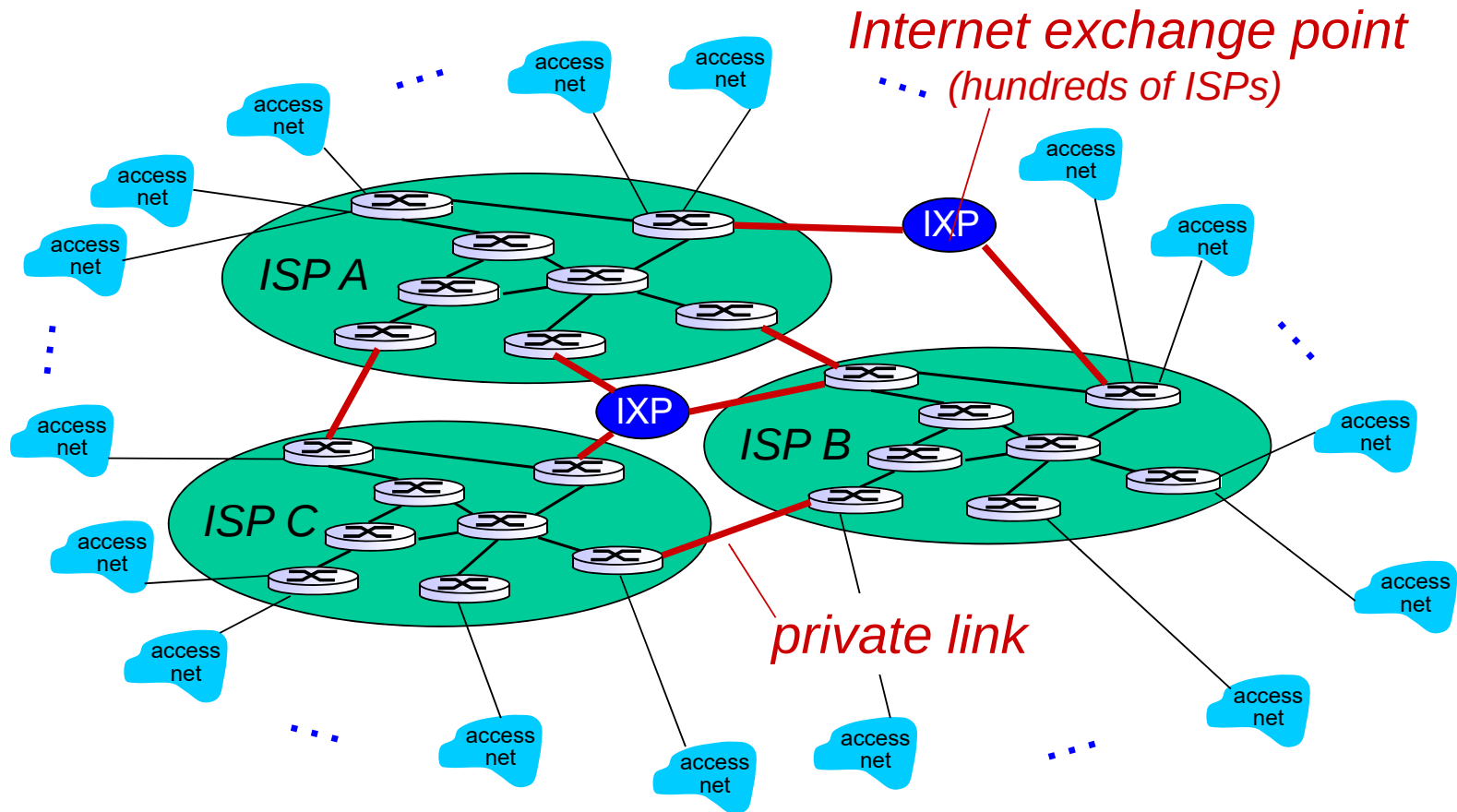
Internet structure: network of networks

Post 1991 - Transition to commercial ISPs - If one global ISP is viable business, there will be competitors



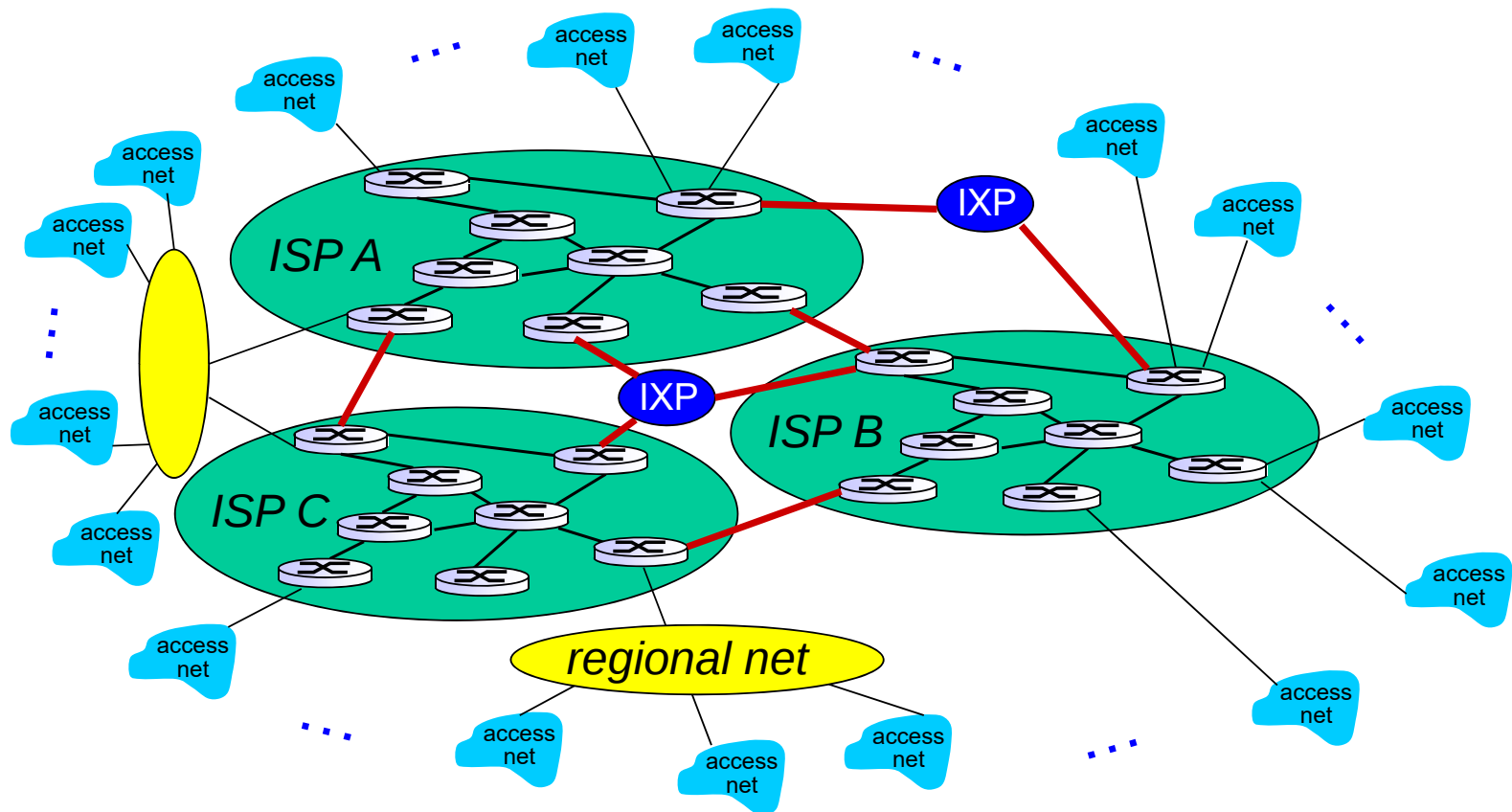
Internet structure: network of networks

Two ISPs are connected in a “provider-customer” or “peer-peer” relationships according to peering agreements



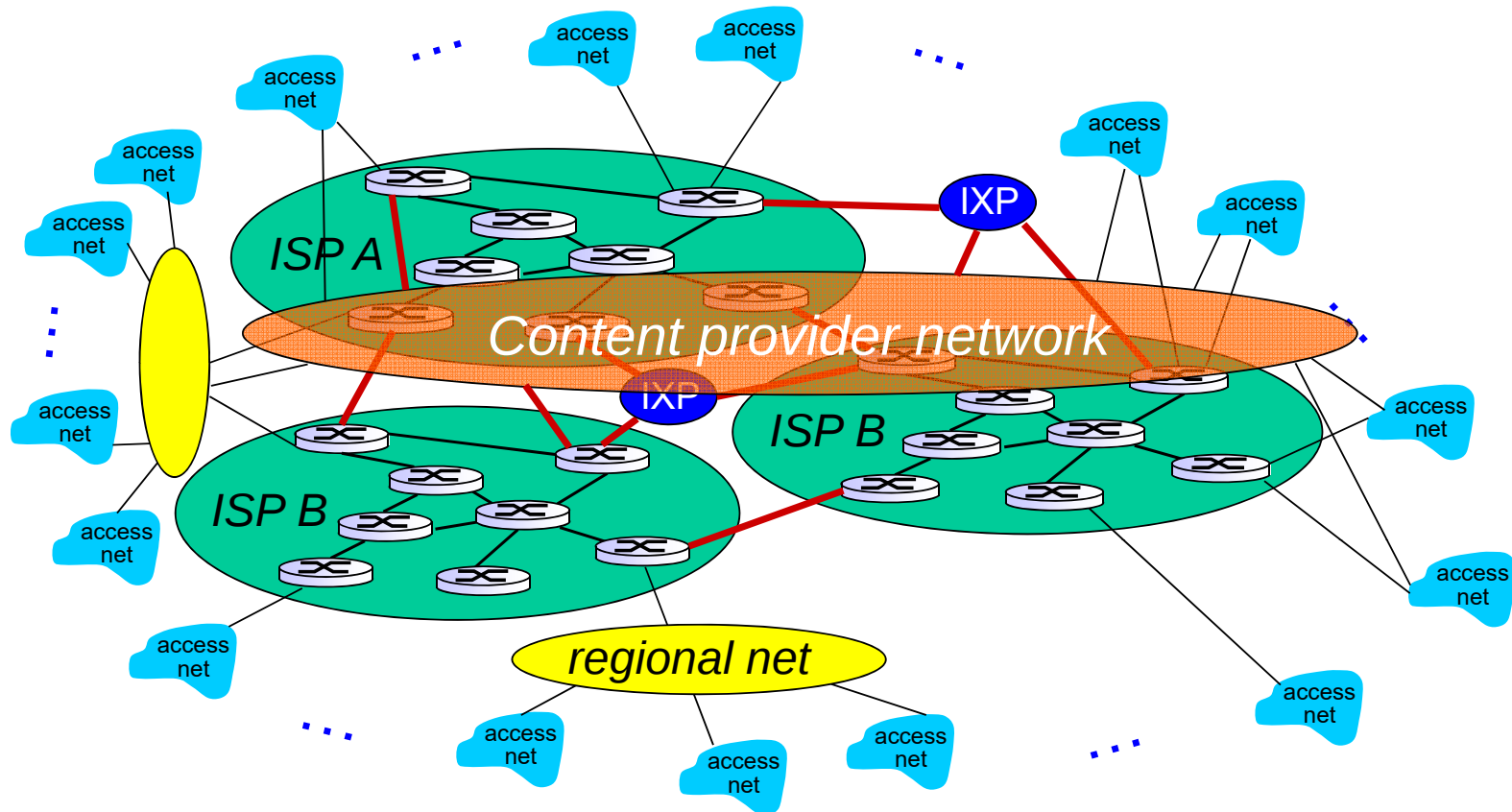
Internet structure: network of networks

... and regional networks may arise to connect access nets to ISPs

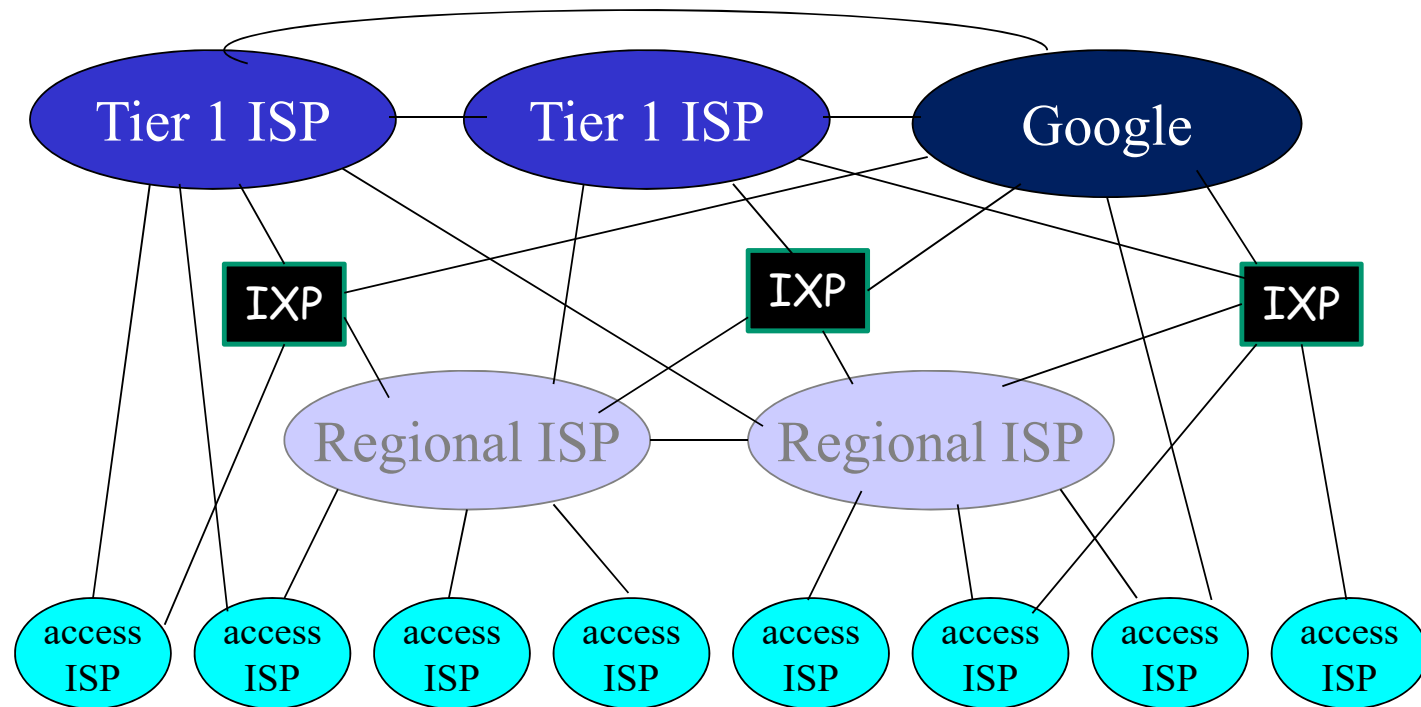


Internet structure: network of networks

... and a content provider network (e.g., Akamai, Google, Microsoft) may run its own network to bring services, content close to end users



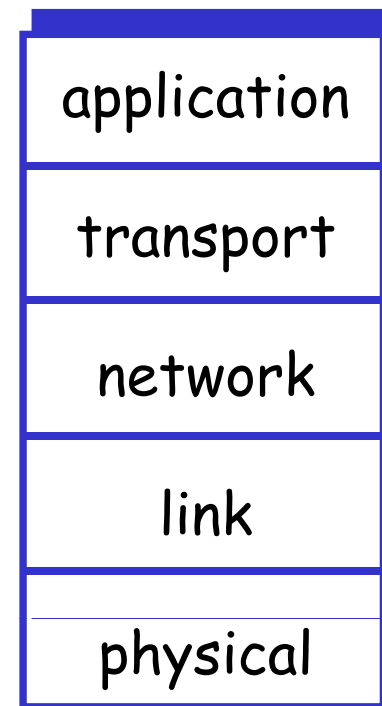
Internet structure: network of networks



- at center: small # of well-connected large networks
 - “**tier-1**” **commercial ISPs** (e.g., Level 3, Sprint, AT&T, NTT), national & international coverage
 - **content provider networks** (e.g., Google): private network that connects its **data centers** to Internet, often bypassing tier-1 and regional ISPs

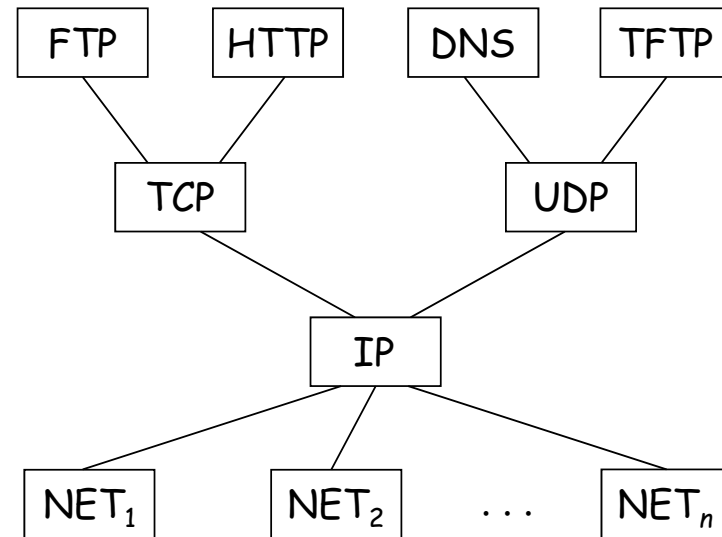
Internet protocol stack

- ❑ **application:** protocols that support network applications
 - SMTP, HTTP, DNS
- ❑ **transport:** process-process data transfer
 - TCP, UDP
- ❑ **network:** routing of datagrams from source to destination
 - IP, routing protocols
- ❑ **link:** data transfer between neighboring network elements
 - PPP, Ethernet, 802.11 (WiFi)
- ❑ **physical:** how to send and receive bits



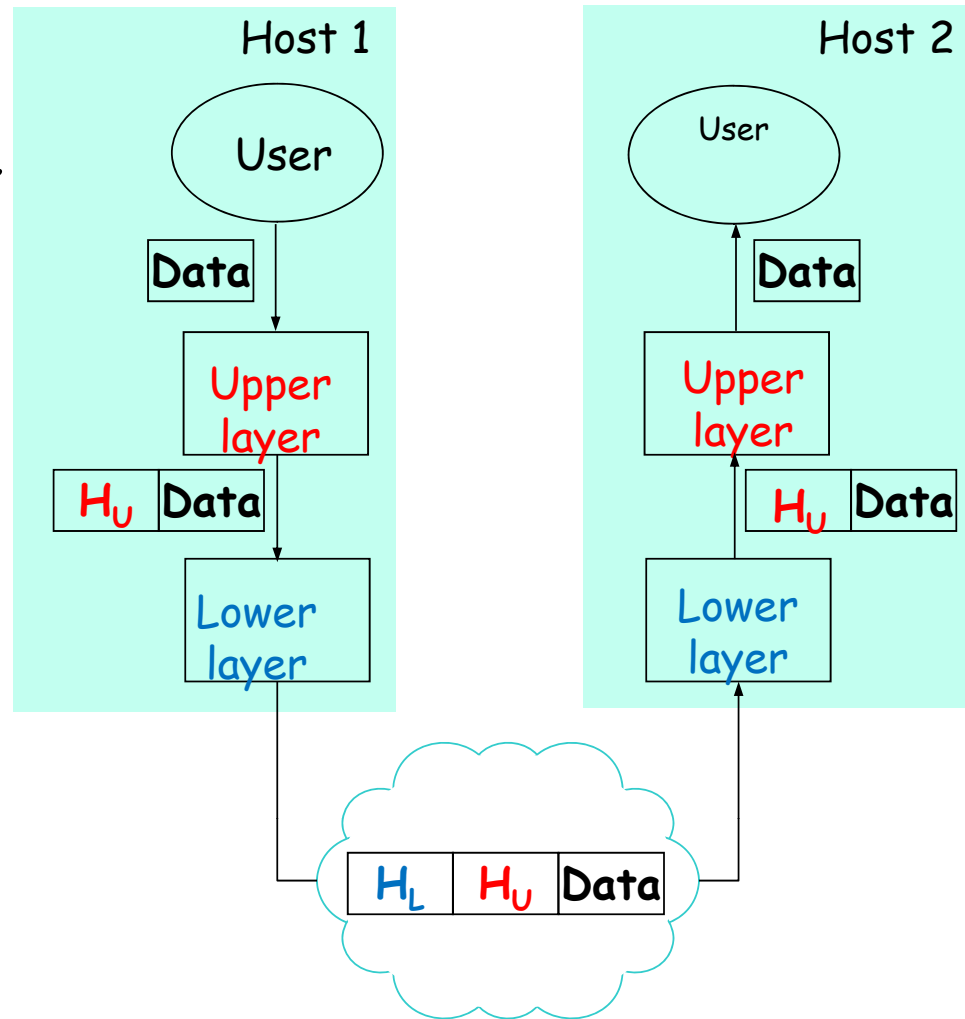
Internet Architecture

- ❑ Internet Engineering Task Force (IETF)
- ❑ application protocols support applications
- ❑ *hourglass* shape (only IP in network layer)
 - best effort service => any delivery service can be used by IP
- ❑ limitation of hourglass



Encapsulation

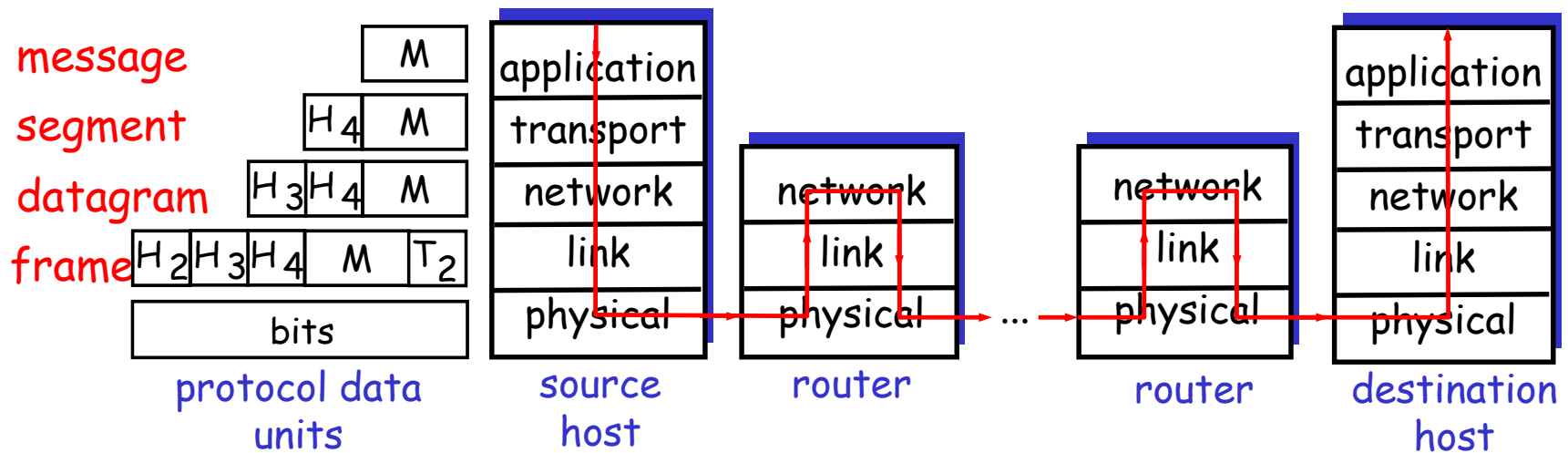
- ❑ Protocol peers provide a data delivery service
- ❑ How do protocol peers in different machines exchange protocol messages between themselves?
 - In **protocol header** encapsulated and de-encapsulated



Physical path of data

Each layer takes data (**service data unit**) from above

- adds header to create its own **protocol data unit**
- passes protocol data unit to layer below



Note: In the past, a **switch** implements only two layers (physical and link). Nowadays many switches function as routers (3 layers)

Note: layer $2\frac{1}{2}$ may exist (i.e. virtual circuits)

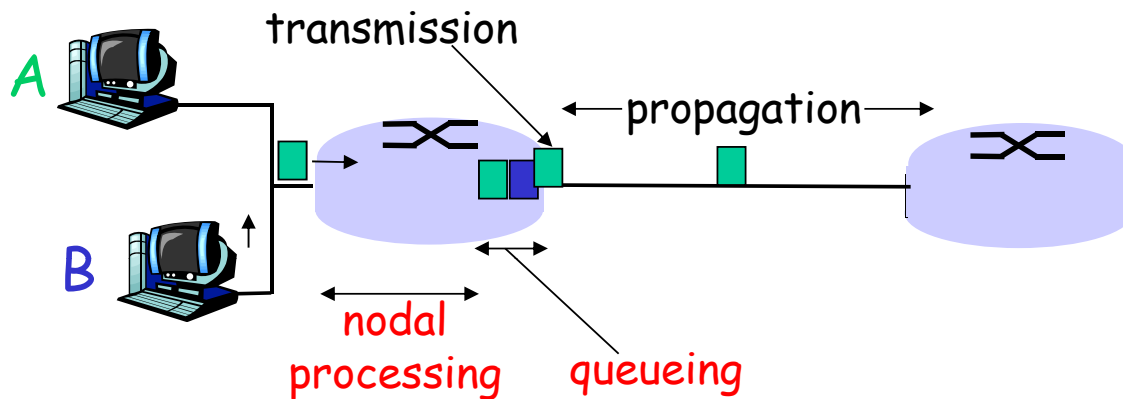
Four sources of packet delay

❑ 1. nodal processing:

- check bit errors
- determine output link

❑ 2. queueing

- ❖ time waiting at output link for transmission
- ❖ depends on congestion level of router



Delay in packet-switched networks

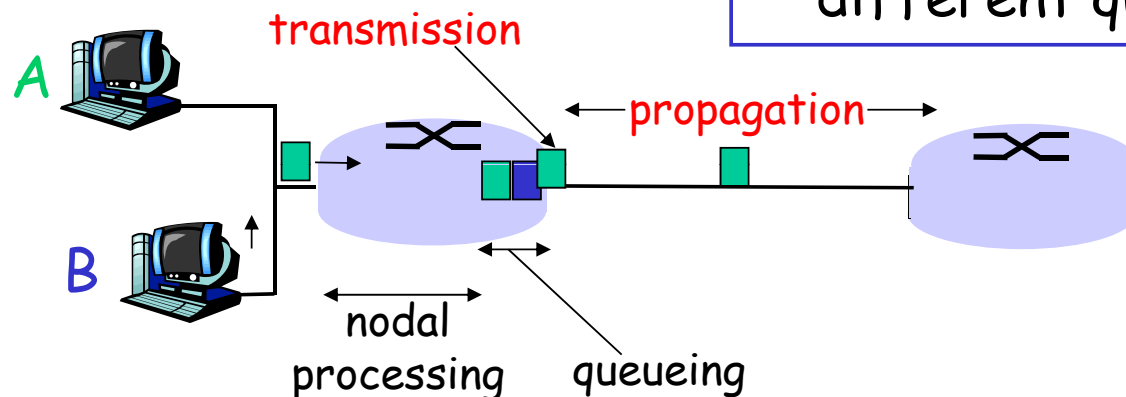
3. Transmission delay:

- R: link bandwidth (bps)
- L: packet length (bits)
- time to send bits into link = L/R

4. Propagation delay:

- d: length of physical link
- s: propagation speed in medium ($\sim 2 \times 10^8$ m/sec)
- propagation delay = d/s

Note: s and R are very different quantities!



End-to-End Delay

- Nodal delay (from when last bit of packet arrives at this node to when last bit arrives at next node)

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

- End-to-end delay over N identical nodes/links from client c to server s (from when last bit of packet leaves client to when last bit arrives at server)

$$d_{c-s} = d_{prop} + Nd_{nodal}$$

- Round trip time (RTT)

$$RTT = d_{c-s} + d_{s-c} + t_{server}$$

where t_{server} is server processing time

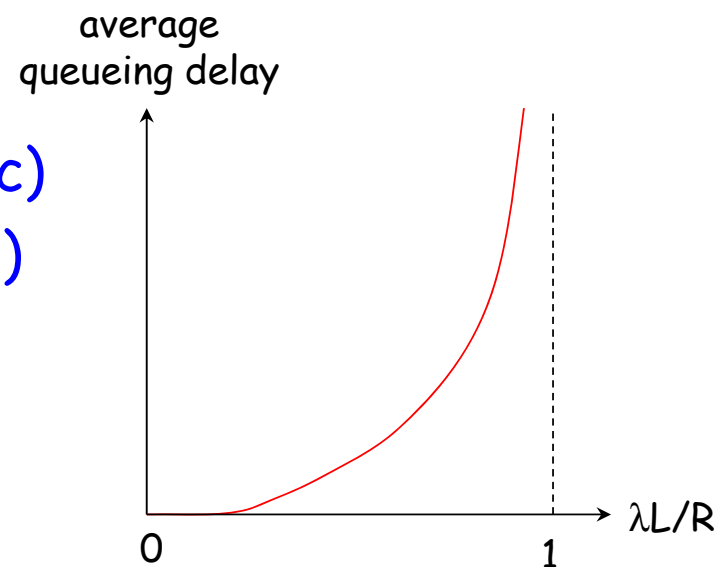
Large delay usually from queueing or loss at "bottleneck"

Queueing delay (waiting time)

- R: link bandwidth (bps)
- L: packet length (bits)
 - service rate = R/L (pkts/sec)
 - ave. service time = L/R (sec)
- λ : packet arrival rate

traffic intensity =

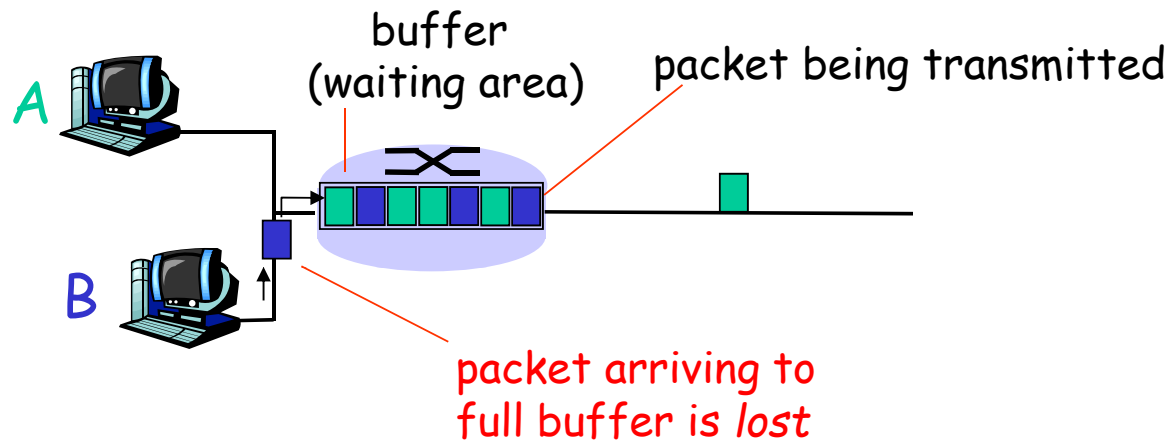
arrival rate/service rate = $\lambda L/R$



- $\lambda L/R \sim 0$: average queueing delay small
- $\lambda L/R \rightarrow 1$: delays become large
- $\lambda L/R > 1$: more "work" arriving than can be served, average delay infinite!
 - ❖ In reality, buffer overflow when $\lambda L/R \rightarrow 1$

Packet loss

- ❑ buffer in router for each link has finite capacity
- ❑ lost packet may be retransmitted by previous node, by source end system, or not at all



End of Introduction