Communication Links:
- Fiber, Copper, Radio, Satellite
- Transmission Rate: *Bandwidth*

Packet Switches: Forward Packets (chucks of data)
- Routers and Switches

Q: How to connect end systems to edge router?
- Residential Access Nets
- Institutional Access Networks
- Mobile Access Networks

Breaks message into smaller chucks, **packets**, of length **L** bits.

Host transmits packet into access network at **transmission rate R** bits/seconds.

$$\text{packet transmission delay} = \text{time needed to transmit } L\text{-bit packet into link} = \frac{L \text{ (bits)}}{R \text{ (bits/sec)}}$$

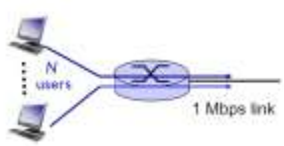
Routing: Determines source-destination route taken by packets. - Routing Algorithms

Forwarding: Move packets from router's input to appropriate router output.

Packet switching versus circuit switching

packet switching allows more users to use network!

example:
- 1 Mb/s link
- each user:
  - 100 kb/s when "active"
  - active 10% of time



- *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?

Q: what happens if > 35 users ?

P8) 3 Mbps link
150 Kbps Transmitting
Each user only transmits 10% of the time

a) $\frac{3 \text{ Mbps}}{150 \text{ Kbps}} = \frac{3000 \text{ Kbps}}{150 \text{ Kbps}} = \boxed{20 \text{ users}}$ when circuit-switching used

b) "Each user transmits only 10 percent of the time"
$\boxed{P(\text{Given user is transmitting}) = 0.1}$

c) $n = 120$ users $p = 0.1$
Binomial Distribution:

$$F(x) = \left(\frac{n!}{x!(n-x)!}\right) p^x \cdot (1-p)^{n-x}$$

$$P(x=n) = \left(\frac{120!}{n!(120-n)!}\right)(0.1)^n \cdot (1-0.1)^{120-n}$$

$$\boxed{P(x=n) = \left(\frac{120!}{n!(120-n)!}\right)(0.1)^n (0.9)^{120-n}}$$

d) $P(x \geq 21) = ? \rightarrow P(x=21) + P(x=22) + \cdots + P(x=120)$

$$P(x=21) = \left(\frac{120!}{21!(120-21)!}\right)(0.1)^{21}(0.9)^{120-21}$$

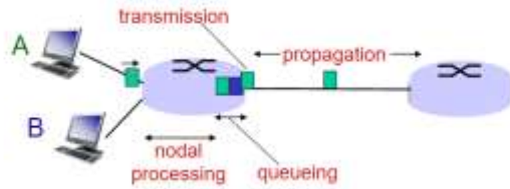
$$P(x=21) = \left(\frac{120!}{21!(99)!}\right)(0.1)^{21}(0.9)^{99}$$

$$P(x=21) = 0.0044$$

So.

$\boxed{P(x \geq 21) \text{ less than or equal to } 0.0044}$

1 bits = 0.125 Byte 1 Byte = 8 bits



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

d_{proc}: nodal processing
- check bit errors
- determine output link
- typically < msec

d_{queue}: queueing delay
- time waiting at output link for transmission
- depends on congestion level of router

d_{trans}: transmission delay:
- *L*: packet length (bits)
- *R*: link bandwidth (bps)
- $d_{trans} = L/R$

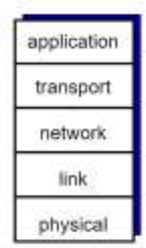
d_{trans} and d_{prop} *very different*

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

a: Average packet arrival rate
- $La/R \sim 0$: avg. d_{queue} small
- $La/R \rightarrow 1$: avg. d_{queue} large
- $La/R > 1$: more "work" arriving than can be serviced, avg. d_{queue} infinite!

Throughput: Rate (bits/time unit) at which bits transferred between sender/receiver
- *Instantaneous*: rate at given point in time
- *Average*: rate over longer period of time

- *application*: supporting network applications
  - FTP, SMTP, HTTP
- *transport*: process-process data transfer
  - TCP, UDP
- *network*: routing of datagrams from source to destination
  - IP, routing protocols
- *link*: data transfer between neighboring network devices
  - Ethernet, 802.111 (WiFi), PPP
- *physical*: bits "on the wire"



Viruses and **Worms**

Denial of Service (DoS): Attackers make resources (server, bandwidth) unavailable to legitimate traffic by overwhelming resource with bogus traffic.

Packet "Sniffing" and **IP Spoofing**

F bits
R = R bps
Host A → Host B
→ Segments of S bits each + 80 bits
Packets → L = 80 + S bits
- 3 links
- 2 Switches
- No Queuing Delays (d_{queue})
- Disregard Propagation Delays (d_{prop})

S = ? bits to minimize the delay of the moving file

Number of Packets = $\frac{F}{S}$ $L = 80 + S$ bits

First calculating the overall delay.

To get the first packet through along with the idea that every packet after the first reaches to Host E every $\frac{S+80}{R}$ seconds:

$$3\left(\frac{L}{R}\right)+\left(\frac{F}{S}-1\right)\left(\frac{L}{R}\right) = \text{Total Delay}$$

$$3\left(\frac{80+S}{R}\right)+\left(\frac{F}{S}-1\right)\left(\frac{80+S}{R}\right)$$

$$\left(\frac{80+S}{R}\right)\left(3+\left(\frac{F}{S}-1\right)\right)$$

$$✓\left(\frac{80+S}{R}\right)\left(\frac{F}{S}+2\right)$$ → Now use this delay formula in order to find where on its graph the value of the equation is the smallest

- So one way is where the slope = 0, by making the derivate of the equation equal to zero in terms of S

$$\frac{d(\text{total delay})}{dS} = \frac{80F}{SR} + \frac{SF}{SR} + \frac{160}{R} + \frac{2S}{R}$$

$$= \left(\frac{80F}{R}\right)\left(\frac{1}{S}\right) + \frac{F}{R} + \frac{160}{R} + \left(\frac{2}{R}\right)(S)$$

$$= -\left(\frac{80F}{R}\right)(S)^{-2} + 0 + 0 + \left(\frac{2}{R}\right)(S)$$

$$\frac{-\left(\frac{2}{R}\right)}{-\left(\frac{80F}{R}\right)} = \frac{-\cancel{\left(\frac{80F}{R}\right)}(S)^{-2}}{-\cancel{\left(\frac{80F}{R}\right)}} + \frac{\cancel{\left(\frac{2}{R}\right)}}{-\cancel{\left(\frac{2}{R}\right)}}$$

$$\frac{2\cancel{R}}{80F\cancel{R}} = (S)^{-2} \rightarrow \frac{1}{S^2} = \frac{2}{80F} \rightarrow \frac{1}{S^2} = \frac{1}{40F}$$

$$\sqrt{S^2} = \sqrt{40F} \qquad \boxed{S = \sqrt{40F}} \text{ (bits) } ✓$$

Total Non-Persistent HTTP Response Time PER OBJECT:

2RTT + File Transmission Time

Total Delay = Internet Delay + Access Delay + LAN Delay

time to distribute F to N clients using client-server approach $D_{cs} \geq max\{NF/u_s, F/d_{min}\}$

increases linearly in N

time to distribute F to N clients using P2P approach $D_{P2P} \geq max\{F/u_s, F/d_{min}, NF/(u_s + \Sigma u_i)\}$

increases linearly in N …

… but so does this, as each peer brings service capacity

Cross-Country Example, Figure 3.17
Channel Utilization > 98%
L = 1500 bytes = 12000 bits
RTT ≈ 30 ms ≈ 0.030 s
R = 1 Gbps = 10^9 bps

Window Size = ?

$$U_{sender} = \frac{L/R}{RTT + L/R}$$

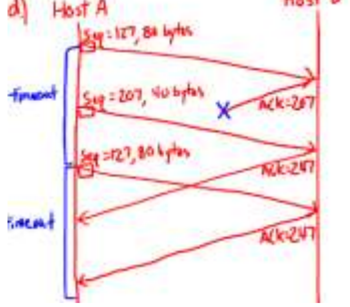
$$d_{trans} = \frac{L}{R}$$

$$d_{trans} = \frac{12000}{10^9} = 1.2 \times 10^{-5} \text{ s}$$

$$0.98 = \frac{(1.2\times10^{-5})N}{0.03 + (1.2\times10^{-5})} \rightarrow N = 2450.98$$

The window size would have to be about $\boxed{\text{2451 packets}}$/big.

a) Seq = 207, Source port = 302, Dest. port = 80
b) ACK = 207, Source port = 80, Dest. port = 302
c) ACK = 127 (since cumulative ACKs)
d) Host A Host B



h)

Round	Packets Sent
1	1
2	2-3
3	4-7
4	8-15
5	16-31
6	32-63
7	64-96

Packet #70 is sent during 7th round

i) ssthresh = 8/2 = 4
CW = 4+3 = 7

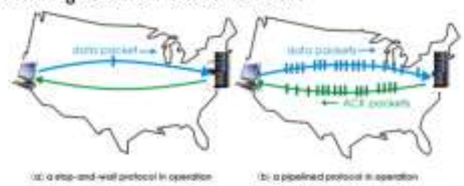
j) ssthresh = 21
CW = 1

k)

Round	# Sent
17	1
18	2
19	4
20	8
21	16
22	+ 21
	52 packets

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

- two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

At most half of the window size of sequence numbers.

- timeout interval: `EstimatedRTT` plus "safety margin"
  - large variation in `EstimatedRTT` -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

```
DevRTT = (1-β)*DevRTT +
             β*|SampleRTT-EstimatedRTT|
         (typically, β = 0.25)
```

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT "safety margin"

two broad approaches towards congestion control:

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at

- loss indicated by timeout:
  - `cwnd` set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - `cwnd` is cut in half window then grows linearly
- TCP Tahoe always sets `cwnd` to 1 (timeout or 3 duplicate acks)