Homework 1

Gabriel Emerson Due 9/2

R16: Processing, transmission, propagation and queueing delays.

Constant are: Processing, transmission, propagation.
Variable: queueing.

R18: It takes 10msec

No it doesn't depend on packet length

No it doesn't depend on transmission rate

R19: a.) 500 Kbps

b.) 64 seconds

c.) 100Kbps & 320 seconds

R26: Virus requires human interaction to spread

Worms don't need user replication. These scan IP's and Ports looking for vulnerabilities.

P2: At time $N(L/R)$ the first packet has reached destination. The second has been stored in last router, and third is stored in next-to-last router. This can be incremented by $N(L/R)+(L/R)$. This means using P to show packet, this equals $N(L/R)+(P-1)(L/R)$.

P3: a.) Circuit switching would be better because it takes a long time and is a steady transmission.

b.) Worst case all application will simultaneously transmit over one or more networks. However each link has enough bandwidth to handle the sum of all data rates. This means we do not need congestion control mechanisms.

P6: a.) $d_{prop} = m/s$

b.) $d_{trans} = L/R$

c.) $d_{end\text{-}to\text{-}end} = (m/s + L/R)$

d.) the bit is leaving Host A.

e.) the first bit is in link, has not reached Host B

P6: f.) the first bit has reached Host B.

g.) $(L/R)s = \frac{120}{56\times10^3}\cdot(2.5\times10^8) = 536\text{ Km}$

P10: This will be equal to the 5 delays sum.

$d_{end\text{-}to\text{-}end} = L/R_1 + d_1/s_1 + L/R_2 + d_2/s_2 + L/R_3 + d_3/s_3 + d_{proc} + d_{proc}$

$\Rightarrow 6+6+6+20+16+4+3+3 = 64\text{ms}$

P31: a.) $\frac{8\times10^6}{2\times10^6} = 4\text{sec.}$ = Host to first packet sw

Source to destination = $4\text{sec}\cdot3 = 12\text{s}$.

b.) First packet Host to packet switch

$\Rightarrow \frac{1\times10^4}{2\times10^6} = 5\text{ms}$

Time second is recieved at first switch

$\Rightarrow 2\cdot5\text{ms} = 10\text{ms}$

c.) First packet to destination Host

$\Rightarrow 5\text{ms}\cdot3 = 15\text{ms}.$

After First packet, new one will be recieved every 5ms. This means the last will be recieved at $15\text{ms}+(799\cdot5\text{ms}) = 4.01\text{ s}.$

This shows without segmentation delay is significantly less.

P31: d.) Without segmentation if there is any bit errors, the whole message must be retransmitted, as opposed to just one packet.

Also without segmentation, huge packets being sent to network cause big delays even if it is followed by small packets

e.) 1.) Packets have to be put in sequence.
2.) They cause smaller packets. This leads to the amount of header bytes is more

P33: Each pack is $S = 80$ bits.
Last packet is at first router $= \frac{S+80}{R} \cdot \frac{F}{S}$
This means $\frac{F}{S} - 2$ is at destination.
The delay then $= \left(\frac{S+80}{R}\right) + \left(\frac{F}{S} + 2\right)$

$S = $ min delay $= 0 \Rightarrow S = \sqrt{40F}$

P34: Telephone are internet are connected through gateways. Skype voice is sent to gateway, is reconstructed and sent through. This can be done since both are circuit-switched, we send packets from internet to gateway. In other way voice goes to gateway and is packetized, then sent to internet circuit (skype)