COMS W4119: Computer Networks
Homework 1

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Bits on Wire

(a) The largest IP datagram that can be transmitted over Ethernet is 1500 bytes. [1]

OC-1 has a transmission rate of 51.84 Mb/s [2]

Thus, the transmission delay is:

$$\frac{0.0015\text{Mb}}{51.84\text{Mb/s}} = 0.00002894\text{s}$$

Since the speed of light in vacuum is $2.998 * 10^8$ m/s, the distance of the wire is:

$$0.00002894\text{s} * 2.998 * 10^8\text{m/s} * \frac{2}{3} = \boxed{5.78\text{km}}$$

(b) The width of a packet on a wire becomes shorter because its transmission delay is shorter, allowing it to be sent out and received by the other end in a shorter period of time.

Circuit Switching and Packet Switching

(a) Circuit switching provides bandwidth guarantees for resource intensive applications, such as audio/video apps, since it offers dedicated resources.

(b) Packet switching allows for better variation in use and better resource utilization, both in terms of the types of applications supported, frequency of use amongst users, and number of users.

(c) Two users

(d) For a specific user, that probability is 20%.

However, if the question asks what the probability that some user is using the network (say within a n person network), then that would be $1 - 0.8^n$

(e) The probability of k users using the network simultaneously, and the others not using the network is:

$$0.2^k * 0.8^{N-k}$$

[1] https://en.wikipedia.org/wiki/Maximum_transmission_unit#cite_note-11

[2] https://en.wikipedia.org/wiki/Optical_Carrier_transmission_rates

The number of k user groups that can be selected from N total users is:

$$\binom{N}{k}$$

Thus, the overall probability is:

$$\boxed{0.2^k * 0.8^{N-k} * \binom{N}{k}}$$

(f) 3 or more users must be using the network simultaneously in order to overwhelm the network. We will solve this problem by solving its inverse.

Fraction of the time that 0 users are using the network simultaneously:

$$0.8^{10} = 0.107$$

Fraction of the time that 1 user is using the network:

$$0.2 * 0.8^9 * 10 = 0.268$$

Fraction of the time that 2 users are using the network simultaneously:

$$0.2^2 * 0.8^8 * 45 = 0.302$$

Fraction of time that 3+ users are using the network (i.e network overwhelmed):

$$1 - 0.107 - 0.268 - 0.302 = \boxed{0.323}$$

Message Switching & Segmentation

(a) I would propose a **packet-switching** design because it would support more users than a circuit switching alternative. Packet switching also supports a variety of applications, which is critical for this company, unlike circuit switching. Furthermore, audio and video streaming allows for some tolerance in packet loss, negating much of the downsides of packet switcing.

(b) Because there are N routers between Alice and Bob, this means that there are $N + 1$ hops that the message must travel through.

The time it takes for the message to travel to one node before continuing on is:

$$\frac{M}{R} \quad \text{seconds}$$

Thus the total time for delivery is:

$$\frac{M(N+1)}{R} \quad \text{seconds}$$

(c) Across the trip, there is a transmission delay and a queueing delay that needs to be accounted for.

Transmission delay: Since each packet is $\frac{M}{k}$ bits, the delay for each node is:

$$\frac{M}{k * R}$$

Queueing delay: Initially, the first router is congested because all k packets arrive. Due to transmission delay, queueing delay does not impact any of the other routers. The last packet transmitted has a queueing delay of:

$$\frac{M(k-1)}{k * R}$$

Tracking the final packet, we can figure out what the time of delivery is. It undergoes the queueing delay of the first router, and then undergoes standard transmission delay for all of the routers in the path:

$$\frac{M(k-1)}{k * R} + \frac{N * M}{k * R} = \boxed{\frac{M(N+k-1)}{k * R} \quad \text{seconds}}$$

(d) With the addition of the header, the total number of bits transferred will change.

For the message switching scenario, the total time of delivery is:

$$\frac{(M+h)(N+1)}{R} \quad \text{seconds}$$

For the packet switching scenario, the total time of delivery is:

$$\frac{(M+kh)(N+k-1)}{k * R} \quad \text{seconds}$$

The same end-to-end delay occurs between both mechanisms when

$$\begin{aligned}
\frac{(M+h)(N+1)}{R} &= \frac{(M+kh)(N+k-1)}{k * R} \\
(M+h)(N+1) &= \frac{M+kh}{k}(N+k-1) \\
(M+h)(N+1) &= (\frac{M}{k}+h)(N+k-1) \\
MN + M + h(N+1) &= \frac{MN}{k} + M - \frac{M}{k} + h(N+k-1) \\
MN - \frac{MN}{k} + \frac{M}{k} &= h(N+k-1-N-1) \\
\frac{MN(k-1)}{k} + \frac{M}{k} &= h(k-2) \\
h = \boxed{\frac{M(N(k-1)+1)}{k(k-2)}}
\end{aligned}$$

With the above condition for h satisfied, both mechanisms have the same end-to-end delay.

(e) Message switching

(f) Both mechanisms get less efficient as the number of hops increase, but packet switching takes $\frac{M}{k*R}$ seconds longer per additional hop while message switching takes $\frac{M}{R}$ seconds longer, making packet switching the far more efficiently scalable mechanism.

(g) The transmission delay per node during packet switching is much lower, by a factor of $\frac{1}{k}$ (see part f). This is because in packet switching, the message is broken up into much smaller packets to be sent out, making the router wait less time to receive the complete datagram.

(h) The transmission delay for each 0.1MB packet is $\frac{1}{20}$ seconds.

The amount of time it takes for a packet to transfer succesfully looks like the following:

$$
\begin{aligned}
&\left(\frac{1}{10}\right)^0 * \frac{9}{10} * \frac{1}{20} && \text{Success on first try} \\
+&\left(\frac{1}{10}\right)^1 * \frac{9}{10} * \frac{2}{20} && \text{Success on second try} \\
+&\left(\frac{1}{10}\right)^2 * \frac{9}{10} * \frac{3}{20} && \text{Success on third try} \\
+&\ldots
\end{aligned}
$$

This can be modeled as

$$\sum_{i=0}^{\infty}\left(\frac{1}{10}\right)^i * \frac{9}{10} * \frac{i+1}{20} = \frac{9}{10}\sum_{i=0}^{\infty}\left(\frac{1}{10}\right)^i * \frac{i+1}{20}$$

Since there are a total of 10 packets that need to be transferred, the expected time for the file to be transferred is:

$$10 * \frac{9}{10}\sum_{i=0}^{\infty}\left(\frac{1}{10}\right)^i * \frac{i+1}{20} = \boxed{9\sum_{i=0}^{\infty}\left(\frac{1}{10}\right)^i * \frac{i+1}{20} \quad \text{seconds}}$$

(i) No

(j) The ideal delay is 1/2 seconds. This number will be used as the baseline for the following parts.

(i) The threshold time would be: $0.5 * 1.2 = 0.6$ seconds.
To meet that threshold, the file must succeed on its first try, hence $\boxed{0.9}$

(ii) The threshold time would be: $0.5 * 2 = 1$ second.
To exceed (or equal) the threshold, the file must fail at least once, hence $\boxed{0.1}$

(iii) The threshold time would be: $0.5 * 1.2 = 0.6$ seconds.
To meet that threshold, there must be at most two packet failures (note that the combination is added because all of the packets are indistinguishable, and thus failures could occur with any set of them)

$$
\begin{aligned}
&0.9^{10} + 0.9^9 * 0.1 * \binom{10}{1} + 0.9^8 * (0.1)^2 * \binom{10}{2} \\
=&0.9^{10} + 0.9^9 * 0.1 * 10 + 0.9^8 * (0.1)^2 * 45 \\
=&\boxed{0.93}
\end{aligned}
$$

(iv) The threshold time would be: $0.5 * 2 = 1$ second.

To exceed (or equal) the threshold, there must be at least 10 packet failures. Since the first retransmission is always successful, then this means all of the packets must fail: $\boxed{0.1^{10}}$

(k) Segmentation significantly improves the performance of file transmission. Part (j) showed that segmentation not only increased the likelihood of file transfers meeting the 120% threshold, but made it nearly impossible for it to exceed the 200% threshold.

The specific mechanism that achieves this is the lower transmission delay achieved when segmenting the file into packets. This makes any failure in the segmented case much less costly (for nonsegmented case, it could not tolerate any failures to reach the 120% threshold).

(l) It would make the influence of segmentation much more significant in a positive way. By segmentating a large file into many small packets, each packet individually has a very low corruption rate, making it much more likely that the file would be transferred in its entirety without any changes. Meanwhile, sending the entire file at once would likely have a high corruption rate. Thus segmentation would have a higher influence with corruption rates taken into consideration.

Protocol Layers and Service Models

(a) A layered internet architecture is important for two reasons:

(1) Modularization makes maintenance, updating, and reuse of a system possible, especially given how distributed it is and vital its continuing uptime is.

(2) It allows for identification and for relationship between the complex system's components.

(b) Layers may duplicate functionality with each other, increasing complexity and redundancy.

(c) The Internet protocol provides that all these services, if needed, must be implemented in the application layer.

(d) The host machine uses the application, transport, network, link, and physical layers.

(e) The router uses the network, link, and physical layers.

(f) The transport layer protocol (TCP vs UDP) does not affect the behavior of a switch, since a switch only operates at a link and physical level.

(g) A data segment is at the transport level, which means it contains information about the transport and application levels. With this in mind, here are the answers for the different parts:

(i) Determinable, since HTTP is in the application layer, which is encapsulated within a data segment.

(ii) Determinable, since TCP is in transport layer, which is encapsulated within a data segment.

(iii) Unclear, since IP is in the network layer, which is below the transport layer and thus out of scope for a data segment.

(iv) Unclear, since Ethernet is in the physical layer, which is below the transport layer and thus out of scope for a data segment.