Department of Electrical Engineering and Computer Sciences
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EECS 122 : Introduction to Communication Networks
Spring 2005 - Assignment 1 Solutions – Available: February 4, 2005

Problem 1 [20 Pts]

Consider the problem of transmitting one file of K bits. We are comparing packet switching and circuit switching.

For the circuit-switched approach, it takes 1 second to set up the circuit. Once the circuit is set up, the transmission occurs at 1Mbps and the signals take 50ms from the source to the destination.

For the packet-switched approach, the network transports the information as packets of 1kbits along lines with a transmission rate of 1Mbps; however, each packet must contain additional information that amounts to 100 bits. We neglect the packet switching delays. The signals again take 50ms from the source to the destination.

(a) Calculate the delays to deliver the file using circuit-switching.

(b) Assume that in packet switching one sends one packet, waits until we get an acknowledgement (assume that this takes 55ms after the packet has been completely received by the destination), then sends the next packet, and so on. Calculate how long it takes to deliver the file.

(c) Assume that we use packet switching but that we send all the packets back to back, without waiting for acknowledgments. Calculate how long it takes to deliver the file.

(d) For what values of K is approach (a) faster than approach (c)?

Solution 1

(a) The time to deliver the file consists of the time it takes to establish the circuit (1 sec), the propagation delay (50 ms), and the time it takes to transmit the file over the link ($\frac{K}{1\times 10^6}$)

$$1.00 + 0.05 + \left(\frac{K}{1 \times 10^6}\right) seconds$$

(b) Each packet is 1100 bits in length (1kbit data plus 100 bit header). After every packet is received by the destination, we wait 55 ms for the acknowledgement. Thus, the time it takes to deliver the file is

$$\left(\frac{K}{1000}\right)\left(\frac{1100}{1 \times 10^6} + 0.055 + 0.05\right) seconds$$

One could also interpret that the packet has a total size of 1kbit, of which 900 bits are data and 100 bits are the header. In this case, the 1000 in the above solution is replaced by 900, and the 1100 by 1000.

(c) As before, we must send the data in 1100-bit packets, but this time we do not need to wait for acknowledgements. The file will be delivered after its transmission time plus the propagation delay.

$$0.05 + \left(\frac{K}{1000}\right)\left(\frac{1100}{1 \times 10^6}\right) seconds$$

(d) We set (a) less than (c) and solve for K.

$$\begin{aligned}
1.00 + 0.05 + \left(\frac{K}{1 \times 10^6}\right) &< 0.05 + \left(\frac{K}{1000}\right)\left(\frac{1100}{1 \times 10^6}\right) \\
1.00 + \left(\frac{K}{1 \times 10^6}\right) &< \left(\frac{11 \times K}{10 \times 10^6}\right) \\
1.00 &< \left(\frac{K}{10 \times 10^6}\right) \\
10 \times 10^6 &< K
\end{aligned}$$

Thus, approach (a) is faster for $K > 10$ Mbits (or 9 Mbits for packets of 1000 bits of total size).

Problem 2 [30 Pts]

Network functions can either be implemented in the end hosts or in the network devices (such as routers and switches). For each of the following network functions, state whether it should be implemented in the end hosts or in the network devices, and briefly justify your answer.

(a) packet retransmission

(b) security

(c) address lookup and routing decisions

(d) multicasting (sending the same data to multiple hosts)

(e) error detection/correction

(f) congestion control

Solution 2

(a) **End hosts**. Making the routers responsible for connection reliability would increase complexity and require too much memory. Furthermore, retransmission is not always necessary or desired (live video streaming, for instance).

(b) **End hosts**. You shouldn't rely on others for keeping your data secure. Furthermore, if a new encryption technology comes out, it would be a great burden to upgrade all of the routers on the Internet.

(c) **Routers**. The routers are responsible for address lookup and determining where to send packets, based on their destinations. If we were to require the end hosts to make all routing decisions, they would each need to have knowledge of the entire Internet's topology, the location of every other end host and how to get to it, and constantly keep this information up to date.

(d) **Both**. We would optimally want the routers to take care of multicasting, avoiding the same data being transmitted multiple times. Unfortunately, this increases router complexity and raises some other issues (such as how to acknowledge the packets and retransmit lost ones). Although IP supports multicasting (and there is a special backbone for it called MBONE), it is not widely deployed. Thus, current efforts to multicast over the Internet are usually handled by the end hosts.

(e) **End hosts**. It is not the routers' responsibility to understand the content of the data they are transmitting, thus they cannot check it for errors. Assuming they were given that information, they would still need to retain a copy of the packet until the other end acknowledges it, and possibly retransmit it. As discussed in (a), this is undesirable. Furthermore, any changes in the higher levels that would require a different error detection system would necessitate that all the Internet's routers be upgraded. Note that there could also be some error detection and correction capabilities in the network devices (e.g. in the link layer or in the IP header).

(f) **End hosts**. If the router throttled down its sending rate while its receiving rate remained the same, its queue would eventually fill up, leading to congestion in the link before the router. If there is congestion somewhere in the connection, the end hosts must slow down, not the routers. However, network devices can play a role in congestion control, mainly by notifying the end hosts before congestion occurs (using, for instance, explicit congestion notification (ECN) or random early detection (RED)).

There are many possible correct answers, so other reasonably justified explanations were also accepted for the above problems.

Problem 3 [25 Pts]

Perform a Traceroute between a source and destination on the same continent, and another Traceroute between a source and destination on different continents, at three different hours of the day (a total of 6 Traceroutes).

(a) Find the average and standard deviation of the round-trip delays at each of the three hours.

(b) Find the number of routers in the path at each of the three hours. Did the paths change during any of the hours?

(c) Try to identify the number of ISP networks that the Traceroute packets pass through from source to destination. Routers with similar names and/or similar IP addresses should be considered as a part of the same ISP. In your experiments, do the largest delays occur at the peering interfaces between adjacent ISPs?

Solution 3
(a) 6 Traceroutes with their appropriate averages and standard deviations.
(b) The paths may have changed over the hours.
(c) The largest delays are likely to occur between adjacent ISPs.

Problem 4 [25 Pts]
Consider sending voice from Host A to Host B over a packet-switched network (for example, Internet phone). Host A converts analog voice to a digital 64 kbps bit stream on the fly. Host A then groups the bits into 48-byte packets. There is one link between Host A and B; its transmission rate is 1 Mbps and its propagation delay is 2 ms. As soon as Host A gathers a packet, it sends it to Host B. As soon as Host B receives an entire packet, it converts the packet's bits to an analog signal. What is the maximum time that elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Solution 4
The maximum time elapses if the bit falls at the beginning of a new packet. We must first wait for the entire packet to be generated, for the packet to be transmitted over the 1 Mbps link, and for one propagation delay:

$$\left(\frac{48 \times 8}{64000}\right) + \left(\frac{48 \times 8}{1 \times 10^6}\right) + 0.002 = .008384 sec = 8.384 ms$$

One may have interpreted that Host B reproduces the first bit as soon as the first bit of the first packet arrives (without waiting for the entire packet to arrive). In this case, the second term in the above sum (the transmission time) goes away. In practice, it is customary to wait for the full packet to arrive (to be able to check it for errors, among other reasons).