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COMP 445 M Theory Assignment 1

For Dr. Sandra Cespedes

1. **Circuit Switching aims at providing a better service through the reservation of the circuit (i.e., circuit is dedicated). Now, considering only the perspective of the communicating users over a Circuit Switching network (i.e., you should not be concerned with the entire utilization of the network or the advantages to other users), is it possible that Circuit Switching may end up harming its users instead of providing a better service to them? If yes, provide a scenario/case that shows that. If no, explain why this service will indeed always provide the best service to its users.**

In circuit-switched networks, resources required along a path to provide for communication between the end systems are reserved for the communication’s duration, while packet-switched networks don’t, meaning they have to wait in a queue for access to a communication link.

Circuit-switched networks is considered wasteful due to silent periods that make the dedicated circuits idle, making resources underutilized. Let’s consider the example of the radiologist who uses a circuit-switched network to remotely access a series of x-rays. The radiologist sets up a connection, requests an image, contemplates the image and them requests a new image. Network resources are allocated to the connection but are not used and are instead wasted during the radiologist’s contemplation periods. This shows that circuit-switched networks are not suitable for real-time services, like phone calls, video calls, screen sharing. Packet-switched networks are better alternatives.

1. **With DSL ISPs, dedicated lines usually connect to the local offices to the location where the service is provided (i.e., residential homes). In contrast, with Cable ISPs, the connection is shared between multiple homes in a neighborhood. Despite this configuration that is clearly to the benefit of DSL users, service provided by Cable ISPs may still be superior to the one provided by DSL providers. Explain the reasons behind this. Further, if you are hired by a DSL provider, and considering that changes to the company’s infrastructure (i.e., wiring) are quite costly, what would you propose to speed up the provided service while balancing the cost.**

A Digital Subscriber Line (DSL) provider is an internet service that is provided by the telephone company which is typically distributed to residences for internet access. The same telephone company that provides DSL internet access also provides its wired local phone access, therefore, the telephone company becomes the residence’s ISP as well. The internet connection and the telephone call share the DSL link at the same time.

Cable Internet Access utilizes the television company’s existing cable television infrastructure. A hybrid fiber coax (HFC) system is used to transmit the connection that can support 500 to 5,000 homes. Instead of a DSL modem, Cable Internet Access requires special modems, called cable modems, an external device that connects to the home PC through an Ethernet port.

Cable Internet Access may seem superior to DSL ISPs due to the fact that DSL networks cannot reach the DOCSIS 2.0 standards of 42.8 Mbps downstream and 30.7 Mbps upstream transmission rates. Although the cable network packets are shared across every home, this allows for full cable downstream rate when requests are not made simultaneously.

A more modern, affordable and reliable approach would be Ethernet and Wifi access networks. Ethernets uses twisted-pair copper wires to connect to an Ethernet switch, which is connected to the larger internet. Ethernet provides 100 Mbps or 1 Gbps, whereas servers may have 1 Gbps or 10 Gbps access. Wifi technology has also been an increasingly popular internet access for wireless LAN settings, which now can be found everywhere.

1. **Consider two hosts A and B separated by 2 nodes (switches or routers), A wants to send a file of size M = 15 Mbytes over to B. Each link has the same data rate C = 1.5 Mbits/s.**
   1. **Assume message switching, how long would it take for the whole file to be received by B? Explain your assumptions. Comment. Write first the formula giving the time in terms of C, M, and possibly other parameters.**
   2. **Assume packet switching and that all packets have the same size L=1200 bits, how long would it take for the whole file to be received by B? Explain your assumptions. Write first the formula giving the time in terms of C, M, and possibly other parameters. Comment and compare.**

In message switching, the whole file is sent as a single unit, therefore, the whole message must be loaded after transmitting from node to node:

The number 3 consists of the 3 links made by the 2 nodes. M is the file size in Mbytes, C is the capacity rate and τ is the propagation delay. First, we convert the bytes to bits, therefore:

Plugging everything in properly:

Message is broken in packets of size L, meaning if P is the number of packets, then:

In packet switching, packets are transmitted independently from one another, therefore if the first packet is 3(L/C + τ), then the other packets are (M/L – 1) \* (L/C), giving a final formula:

1. **Suppose there are two links between source and destination, with one router connecting the two links. Each link is 5,000 km long with a transmission rate (R) of 10 Mbps. Assume the propagation speed is 2 x 108 meters/sec. There is a 30 Mbit MP3 file sent as one message. Suppose there is no congestion, so that the message is transmitted onto the second link as soon as the router receives the entire message. The end-to-end delay is**
   1. **6.05 seconds**
   2. **6.1 seconds**
   3. **3.05 seconds**
   4. **none of the above**

Clearly, the answer is B: 6.05 seconds.

1. **Now suppose the same network of question 4, but now the MP3 of file is broken into 3 packets, each of 10 Mbits. Ignore headers that may be added to these packets. Also ignore router processing delays. Assuming store and forward packet switching at the router, the total delay is**
   1. **3.05 seconds**
   2. **4.05 seconds**
   3. **6.05 seconds**
   4. **none of the above**

Clearly, the answer is B: 4.05 seconds.

1. **Now suppose there is only one link between source and destination, and there are 10 FDM channels in the link. The MP3 file is sent over one of the channels. The end-to-end delay is**
   1. **30.05 seconds**
   2. **30 seconds**
   3. **300 microseconds**
   4. **none of the above**

Clearly, the answer is A: 30.05 seconds seconds.

1. **What are the disadvantages of using a layered architecture in computer networks?**

A layered architecture allows us to define and organize a large and complex system. It is a set of rules that allow for a network to operate in a structured manner that provides service and functionality using modularity, making it much easier to change the implementation of the service provided by the layer. However, some researchers and networking engineers are opposed to it. One potential drawback is functionality duplication between layers. Some layers may require information from another layer, which violates the goal of separation of layers. Another drawback is the additional overhead created by the control information appended at each layer. This enlarges the message’s size, meaning greater delays.

1. **What do encapsulation and de-encapsulation mean? Why are they needed in a layered protocol stack?**

Encapsulation is when a message is “packaged” to be sent from one end-system to another. It is divided into multiple transport-layer segments (which might themselves each be divided into multiple network-layer datagrams) which is broken down into parts that are needed to transport the right information to the appropriate destination. The most basic form of encapsulated data includes a header and a message. The header is used to identify the sending and receiving applications. De-encapsulation is when the encapsulated information are removed and these smaller packets are then reconstructed to form the needed data to be used by the target application.

Encapsulation and de-encapsulation allow for modularization and easy maintenance within a layered architecture. It allows for identification of different information and how they are transported from one source to another. This type of system provides an optimal and organized structure within the structure to send information to the right place, using the right identification system for minimal errors.

1. **A message of size 4096 bytes is sent using TCP, IP, and Ethernet as network protocols. The application sending the message uses a 16-byte header. TCP employs a 20-byte header, IP employs a 20-byte header, and Ethernet uses a 14-byte header and a 4-byte trailer. Considering that Ethernet defines a maximum data field size of 1500 bytes (also known as the MTU – Maximum Transmission Unit), what is the percentage of overhead incurred in the transmission of the message.**