# CPT250 - Data Communication and Net-Centric Computing
# Study Period 3, 2017
# Assignment 2

**Due: end of Week 11, 11:59 pm on Sunday November 12, 2017 AEDT (Melbourne time)**

**Warning regarding Plagiarism**

- Please remember that plagiarism and other forms of cheating are considered academic offences, which in turn have academic penalties. The School of Science now routinely uses plagiarism detection software on electronically submitted assignments.
- All other material that is not original must be fully credited. That is, any material that is copied or derived from another source must be clearly identified as such and the original author or source must be identified.
- Sometimes students assist each other with an assignment, but end up working together too closely, so that the students' separate answers have significant parts in common; unless answers were developed independently, they are regarded as plagiarized.
- The minimum penalty for plagiarism is loss of marks for assignment. Please be careful!!!

**Introduction and Objectives of Assignment 2**

This assignment has **a total of 75 marks** and is worth **30%** of your final score.

**This assignment is not a hurdle - rather it contributes towards the practical work component (and thus your final result) for this course.**

**NOTE: This assignment is to be undertaken individually – no group work is permitted.**

**Due: end of Week 11, 11:59 pm on Sunday November 12, 2017 AEDT (Melbourne time)**

**Write all your answers in a .pdf file and submit that file in the Assignment section of Canvas.**

You should ensure the file you submit contains your name and student number. The file name should be your student number.

There will be a **late submission period of 5 days** for this assignment, which will expire at **11:59pm on Friday November 17, 2017 AEDT (Melbourne time)**. Late submissions that are received before the end of this late submission period will **attract a late penalty of 10% per day (or part thereof) of the marks awarded on the assignment**, unless a previously submitted application for an extension of time has been approved.

Do NOT leave submitting your assignment until the last day; let alone the last hour or even minute. **You can resubmit many times up until the final cut-off time**. If you leave submissions until it is too late and find you have a problem (e.g. unable to connect) that is entirely your own fault and no special consideration will be given.

You **must** monitor Canvas Discussion section and Announcements for variations and updates to the assignment.

**Submissions that are received after the late submission period expires at 11:59pm on Friday October 13, 2017 AEDT (Melbourne time) will not be assessed** unless a previously submitted application for an extension of time has been approved.

Your primary resource for assistance with the assignment is the Lecture slides followed by tutorial material and the textbook. You can also ask questions in the Discussion. You can also send questions to the Instructor.

**Q1. (Total: 6 marks)**

A multiplexer combines five 400 kbps channels A, B, C, D, E (see Figure 1 below). Each frame has 1 synchronisation bit added to the beginning of the frame and has slot size of 5 bits. Synchronous time division multiplexing will start from Channel A, then B, C, D, E and then back to channel A, then B, C, D, E and so on...

**a)** Draw a diagram to show the content of the first two frames of the output for the inputs as shown in Figure 1. In your diagram, for each frame, show all the slots containing bits of each channel and the synchronisation bit. **(2 marks)** (It is up to you to choose what sync bit for each frame, as long as there is 1 sync bit at the beginning of each frame. The data generated by each channel is shown in Figure 1)

**b)** What is the frame rate (frame per second) **(1 mark)** and the frame duration (in microsecond) **(1 mark)**?

**c)** What is the bit rate (bps) of the MUX output link? **(2 marks)**

Use 100 kbps = 100000 bps.

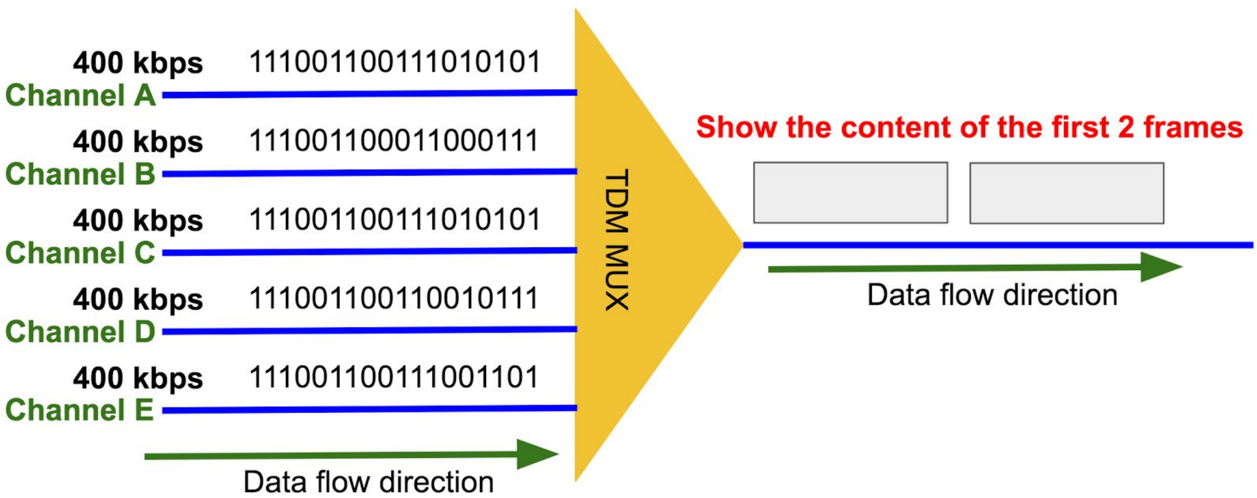


**Figure 1**

**Q2. (Total: 19 marks)**

**(i)** Suppose that frames are 600 bytes long which includes 47 bytes of overhead. Also assume that ACK frames are 78 bytes long. **(Total: 8 marks)**

**a)** The transmission uses Stop-and-Wait ARQ. Let the transmission rate of the system be R where R = 1.5 Mbps. For convenience calculation, the Processing Times at each end is 1.2 ms. Calculate the efficiency of the system if RTT takes the following values: 1.5 ms, 13 ms, 117 ms, and 1.25 seconds. **(4 marks)**

**b)** Repeat if R = 1.5 Gbps. **(4 marks)**

[Note: this question can give some very small numbers. Round your results up to 5 decimal places if possible]
1 Mbps = 1,000,000 bps
1 Gbps = 1,000,000,000 bps

Efficiency of the system is calculated using the following formula:

$$\text{Efficiency} = \frac{(T_{FRAME} - T_{OVERHEAD})}{T_O} \times 100$$

Where $T_{FRAME}$ is the transmission time of a frame, $T_{OVERHEAD}$ is the transmission time of the frame overhead and $T_O$ is the overall time. You need to fully understand how Stop-and-Wait ARQ works to be able to determine $T_{FRAME}$ and $T_{OVERHEAD}$ and calculate the overall time $T_O$.

**(ii)** Three ARQ protocols are covered in this course. Discuss how each ARQ protocol will respond when it detects a frame with errors? Explain how recovery is achieved. **(5 marks)**

**(iii)** In a Stop-and-Wait ARQ system, the bandwidth of the line is 512 kbps, and 1 bit takes 37 ms to make a round trip. **(Total: 6 marks)**

**a)** What is the bandwidth-delay product? **(2 marks)**

**b)** If the system data frames are 128 bytes in length, what is the utilization percentage of the link as estimated from the BDP in (a). **(2 marks)**

**c)** What is the utilization percentage of the link if the link uses Go-Back- N ARQ with window size of 9? **(2 marks)**

**Note:**

1 kbps = 1000 bps
Round your numerical answers to 3 decimal places where possible.

**Q3. (Total: 20 marks)**

**(i)** This is an analysis question.

You are a network engineer who is setting up a highly reliable data communication network for your company. To guarantee packet delivery, you have decided that your network will use Flooding as a routing method. One technical challenge you face when using Flooding is that packets can stay circulating around in circles on your network "forever", causing congestion in your network.

To address this challenge, you include a time to live (TTL) field in each packet. This value is an integer which takes into account the number of nodes that a packet may have to pass through on the way to its destination. **You are thinking about what value you should set the TTL field for each packet**.

Your colleague suggests an approach to set the TTL value based on **Network Diameter**. It means that when calculating a routing table, you will have a list of paths from one node to every other node in the network. From the routing table of every nodes in your network, you can determine the paths from every node to every other node. From this list of paths, the path with the highest hop count will be the network diameter.

If you choose to set the TTL field of each packet to the "diameter" of the network as suggested by your colleague, then as a packet traverses the network, the TTL field is decremented after each hop until the TTL value reaches zero, then the router processing that packet will discard that packet, unless that router is the destination of the packet.

**Please answer the following question**: Does this approach always ensure that a packet will reach its destination if there is at least one functioning path (to the destination) exists? Why or why not?
**(4 marks)**

**(ii)** Consider the network shown in Figure 2 **(Total: 16 marks)**

> **a)** Using Dijkstra's algorithm, compute the shortest paths from node H to all other network nodes showing your work in a table. **(8 marks)**
>
> **b)** Using Bellman-Ford algorithm, compute the shortest paths from H to all network nodes showing your work in a table. **(8 marks)**

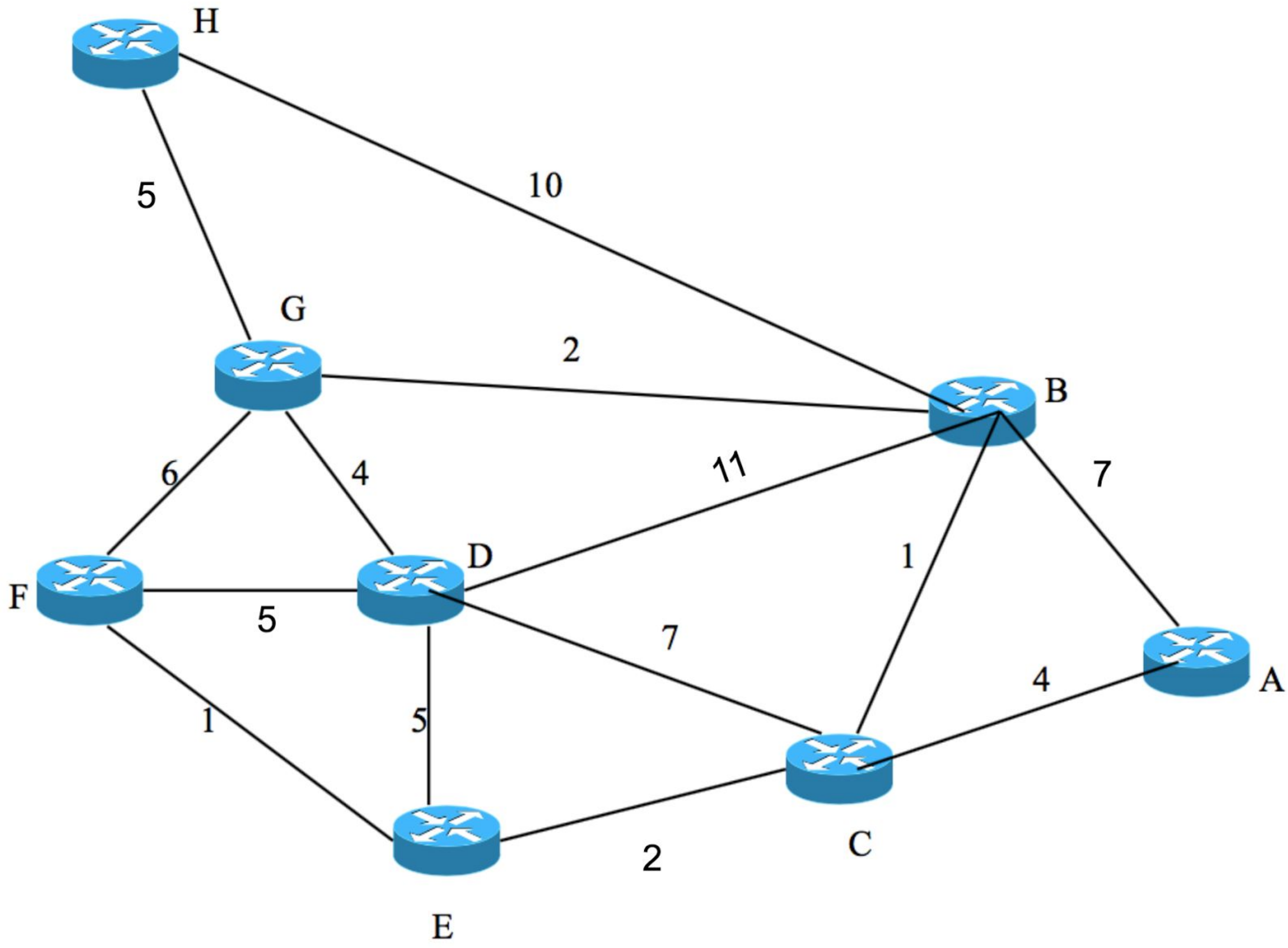


**Figure 2**

**Q4.** **(Total: 6 marks)**

Suppose N stations are connected to a single Ethernet LAN, operating at the rate of 270 Mbps. Assume that the efficiency of each Ethernet is 33 percent.

All workstations on the LAN transmit data at a rate of R= 4 Mbps. Note that all traffic generated goes to the internet and the amount of data from the internet is negligible.

The current design of the LAN is shown in Figure 3.

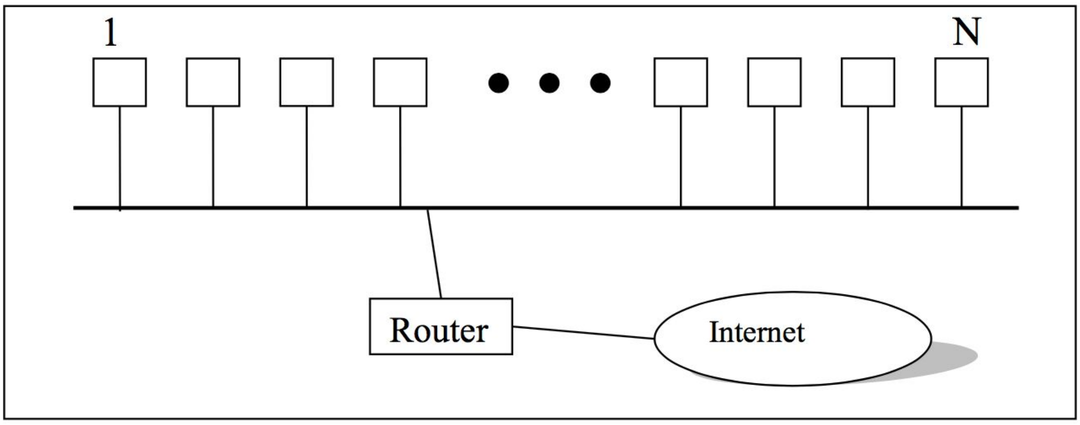


**Figure 3**

a) What is the maximum number of stations, N, that can be supported by the current design in Figure 3? **(2 marks)**

b) To increase the number of nodes on the network, it is proposed to reduce R by half and re-design the network. The new proposed design is shown in Figure 4. Calculate the number of nodes N that can be supported. **(2 marks)**

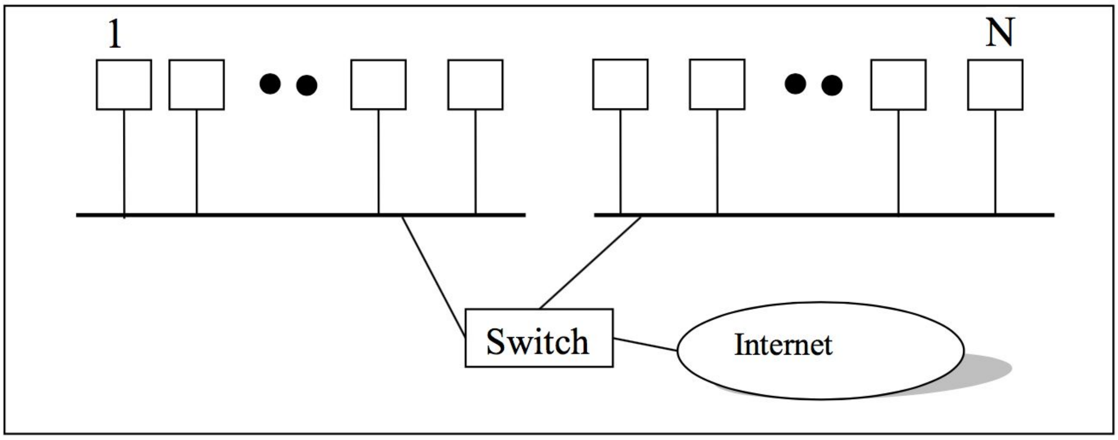


**Figure 4**

c) The Accountant thinks he knows a bit about data networking and can see a way to redesign the network re-use existing equipment, saving money. He claims that in his solution, R remains the same thereby saving even more money. Analyse his design shown in Figure 5 and calculate how many nodes can be supported. **(2 marks)**

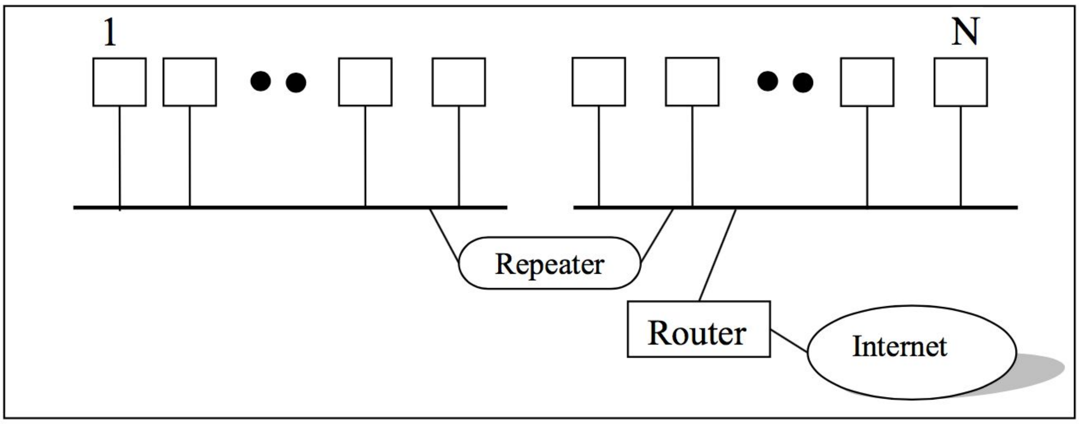


**Figure 5**

**Q5. (Total: 24 marks)**

**(i) (Total: 8 marks)** Consider an application that transmits data at a steady rate (for example, the sender generates an N-bit unit of data every k time units, where k is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions briefly justifying your answer:

**a)** Would a packet-switched or circuit-switched network be more appropriate for the application? Why? **(4 marks)**

**b)** Suppose that a packet-switched network is used and the only traffic in this network comes from such applications as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why? **(4 marks)**

**(ii) (Total: 16 marks)** Recall that ATM uses 53-byte packets consisting of five header bytes and 48 payload bytes. 53 bytes is unusually small for fixed-length packets; most networking protocols (IP, Ethernet, Frame Relay etc.) use packets that are, on average, significantly larger. One of the drawbacks of a small packet size is that a large fraction of link bandwidth is consumed by overhead bytes; in case of ATM, almost 10% of the bandwidth is “wasted” by the ATM header. In this question, we investigate why such a small packet size was chosen.

As part of the discussion, we want to compare the delays of large packets to small packets. There are two delays we will consider: Packetisation Delay and Transmission Delay.

Transmission delay is the delay incurred when moving a cell or packet onto a transmission link. Packetisation delay is the time needed to fill the payload section of a cell or packet from a constant-data-rate source. (It’s a bit like asking “how long does it take to fill a container of a given size from a tap with the water flowing at a given rate?”)

Note that **we will only be considering the payload**, not any of the overhead.

To do such comparisons, we first need a generalised formula. This allows us to vary one parameter and see what happens to the other. We already have a generalized formula for Transmission Delay.

Time = number of bits/data rate = L/R

The first question asks you to construct a formula for Packetisation delay.

**a)** Let the payload of a cell/packet be P bytes. Consider a digitally encoded voice source encoded at a constant rate of 32 kbps. Assume each cell/packet is entirely filled before the cell/packet is sent into the network. Derive a formula for calculating the packetisation delay in millisecond, in terms of P.
**(4 marks)**

When streaming data that is time-sensitive like voice, it is known that gaps in the transmission of 20ms or more cause a noticeable and unpleasant echo. With this in mind, let's compare the packetization delay of small and large cells/packets.

**b)** Use the formula derived in part (a) to determine the packetization delay in millisecond for

i. P = 1,500 bytes (corresponding to a large Ethernet packet payload) **(2 marks)**

ii. P = 48 bytes (corresponding to an ATM cell payload) **(2 marks)**

Next, we use the formula for Transmission delay to compare that delays for small and large cells/packets.

c) A transmission link has a data rate of R=189 Mbps. Calculate the transmission delay in microsecond for:

i. L = 1,500 bytes **(2 marks)**
ii. L = 48 bytes **(2 marks)**

d) Comment on the advantages of using a small ATM cell. **(4 marks)**

Round all your results to 3 decimal places.

# End of Assignment