CNT5106C Homework 3 Solutions

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1. (a) Since you're not waiting for the frames to go all the way around the token-ring there's only 1 propagation time incurred for the token, i.e.

$$
\begin{aligned}
u &= \frac{T_1 + T_2 + \cdots + T_n}{T_1 + T_2 + \cdots + T_n + Prop} \\
&= \frac{\sum_{i=1}^{n} T_i}{\sum_{i=1}^{n} T_i + Prop} \\
&\approx \frac{n \cdot E(T)}{n \cdot E(T) + Prop} \\
&\approx \frac{n \cdot E(T)}{n \cdot E(T)} \frac{1}{1 + \frac{Prop}{n \cdot E(T)}} \\
&\approx \frac{1}{1 + \frac{a}{n}}
\end{aligned}
$$

where $a = Prop/E(T)$.

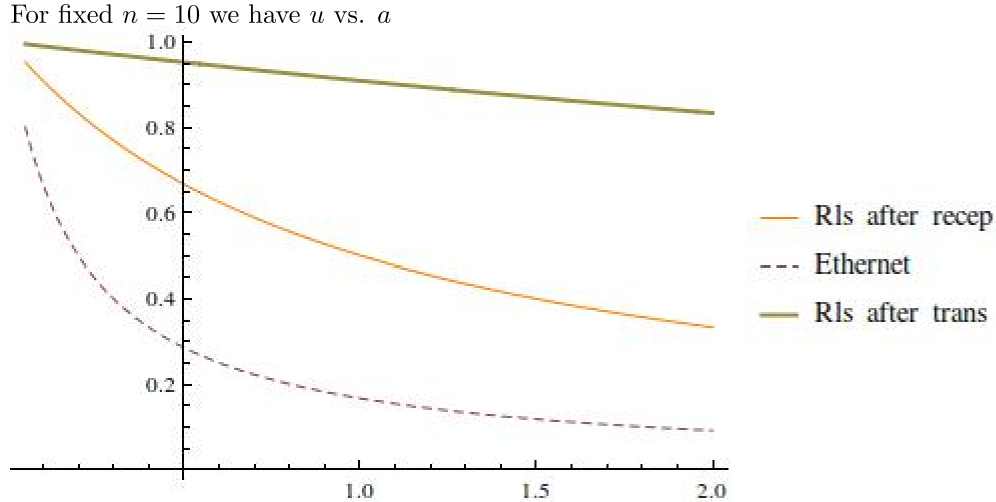
(b) For release after reception

$$u = \frac{1}{1+a}$$

and for CSMA/CD Ethernet

$$u = \frac{1}{1+5a}$$

For fixed $n = 10$ we have u vs. a



(c) For release after transmission the utilization obviously goes up as the number of nodes n increases, since the $a/n \to 0$ implies that u increases. For the release after reception the utilization does not change as a function of number of nodes. For Ethernet the constant coefficient 5 increases because propagation delay increased as the number of collisions increase, which is a function of the number of nodes.

2. (a) For CSMA/CD Ethernet

$$
\begin{aligned}
u &= \frac{1}{1+5a} \\
&= \frac{1}{1+5\frac{t_{prop}}{t_{trans}}}
\end{aligned}
$$

Supposing that propagation delay, t_{prop}, is fixed, and that $t_{trans} \propto \frac{1}{R}$, where R is the bus data rate then for $R_1 = 10Mbps$ and $R_2 = 100Mbps$ we have that

$$
\begin{aligned}
\frac{u_1}{u_2} &= \frac{\frac{1}{1+5\frac{t_{prop}}{t_{trans}}}}{\frac{1}{1+5\frac{t_{prop}}{t_{trans}}}} \\
&= \frac{\frac{1}{1+5\frac{t_{prop}}{k(1/R_1)}}}{\frac{1}{1+5\frac{t_{prop}}{k(1/R_2)}}} \\
&= \frac{\frac{1}{1+5R_1\frac{t_{prop}}{k}}}{\frac{1}{1+5R_2\frac{t_{prop}}{k}}} \\
&= \frac{k+5R_2t_{prop}}{k+5R_1t_{prop}} \\
&= \frac{k+5\cdot 100\cdot t_{prop}}{k+5\cdot 10\cdot t_{prop}}
\end{aligned}
$$

And supposing that $k \ll 5\cdot 10\cdot t_{prop}$ $u_1 \approx 10u_2$.

(b) We can either change t_{prop}, making it small would raise the utilization, or we can change the constant coefficient 5 which is also related to the estimated propagation delay. If we decrease t_{prop_2} by a factor of 10 of the original t_{prop_1} then

$$
\begin{aligned}
\frac{u_1}{u_2} &= \frac{k+5\cdot 100\cdot t_{prop_2}}{k+5\cdot 10\cdot t_{prop_1}} \\
&= \frac{k+5\cdot 100\cdot (1/10)t_{prop_1}}{k+5\cdot 10\cdot t_{prop_1}} \\
&= \frac{k+5\cdot 10\cdot t_{prop_1}}{k+5\cdot 10\cdot t_{prop_1}} \\
&= 1
\end{aligned}
$$

(c) No I don't agree. Since token propagation time will increase only negligibly then in either mechanism, "release after transmission" or "release after reception", the times between receipt and release of the the token will be the same. Hence, assuming that all nodes are attempting to transmit at all times and only kept from transmitting by a lack of token, they will all be able to transmit at the same rate, regardless of how many are on the network.

3. (a) Switches allow multiple simultaneous transmissions and the switch does selective forwarding, instead of simply broadcast and filtration. Each host has a dedicated direct connection to the switch, the switch buffers packets sent by hosts, and then retransmits based on a correspondence between MAC address of recipient and interface that that recipient is connected to. Initially this correspondence is not established and so the first time a switch receives a packet for a host on unknown link it floods all the links and waits for a response from the host. From then on it only sends across that link and so there are no collisions - each link is its own collision domain links are full-duplex so no collisions occur. Therefore simultaneous transmission from all hosts is possible.

(b) CSMA/CD is not need in current implementations of Switched Ethernet because every host is on its own collision domain. It is still supported for backwards compatibility and for half-duplex connections, where the switch needs to know when the host is sending so that its responses don't collide with the host's transmissions.

4. No it is difficult to sense collisions when transmitting due to weak/fading received signals. Furthermore you can't in general sense all collisions of hidden nodes, discussed below.

5. The hidden terminal problem is when, for example, 3 nodes A,B,C are communicating and node B can sense both A and C but neither A can sense C nor C sense A. And so collision detection becomes impossible since C can be transmitting to B and yet to A it appears as if no one is transmitting.

6. The way that 802.11 works is that it first senses for an amount of time equal to DIFS, distributed inter-frame spacing. If the channel is clear then it transmits the entire frame. But if the channel is not clear, if the protocol senses that the channel is busy, then a random timer is started,which sets the amount of time the protocol will wait until trying to transmit. Since the timer is completely random it does not give preferential treatment to those who have gotten some data sent already. Alternatively you could have a situation where a node that is sending a large number of frames hogs the link because each time the timers of the users expire the channel is still busy but when the exploitative node finishes sending a frame all of other nodes still have timers running. This is an instance where priorities would be necessary, or where the DIFS helps the fairness.

7. (a) Rate adaption is the selection of the appropriate encoding scheme in order to match the prevailing signal to noise ratio. That is to say that as the signal to noise ratio decreases it behooves the protocol to select a lower and lower encoding/transmission rate in order to space out the data, in order to reduce the bit error rate, so that the receiver has more time to detect it.

   (b) Since TCP depends on round trip time estimates in order to gauge the congestion in the network it will not respond appropriately at time when the encoding rates on wifi are switched (because this will appear to be congestion but in fact will simply be lower signal to noise ratio).

8. Code division multiple access

   (a) A unique "code" is assigned toe each user, i.e. there's partitioning in code set instead of frequency or time, and so all users can share the same frequency. Multiple users can transmit simultaneously with minimal interference, if the codes are "orthogonal".

   (b) Spreading the signal power over a wide range of frequencies reduces fading effects because only part of the spectrum, hence only part of the signal, is affected by fading.

   (c) No frequency planning is required since users can use the same frequency.

   (d) A soft hand-off can be provided since all cells use the same frequency

   (e) If you have the right signal to correlate with, the pseudo-noise signal, then you can decode the signal and if you don't have such a signal you will interpret just noise. This is useful for security since it effectively encrypts the signal.

9. (a) ATM chops up TCP/IP datagrams into many much smaller cells. When one ATM cell is dropped, due to congestion, all other ATM cells that belong to the same IP datagram are uselessly being sent across the network. This is because the datagram isn't reassembled until it gets to the end of the route and only then does TCP check for corruption. Also one IP flow could branch into many ATM virtual circuits per single IP datagram, since IP packets are statistically switched and don't stay together.

   (b) Solution: a discard strategy such that when a cell is discarded you minimize the transmission of other useless cells (potentially those from the same datagram).

i. Partial Packet Discard: The ATM header identifies the end of a datagram by setting the type bit to 1. So a sequence of cells with type bits 000001 indicate that all those cells belong to the same datagram. When a cell is dropped at a switch, all cells belonging to the same datagram are dropped as well. On average this leads to only 1/2 of the cells of a datagram being transmitted uselessly because on average you'll be half way between the end of and beginning of the datagram when you decide to drop the rest of its cells. The only other problem is when ATM flows are mixed and then ATM header type bits might be 0101010 and it's unclear which cells belong to which datagram. So it's important to stratify distinct flows.

ii. Early Packet Discard: Monitor queues and preemptively drop all cells associated with a datagram when queue length exceeds some threshold. The problem with this is that fairness. The shorter the datagram the more likely it is to be dropped because the algorithm simply scans through the queue and looks for the next ATM header with type bit set to 1, indicating the end of a datagram, and drops that one.

iii. Fair buffer allocation: Use Early Packet Discard but drop packets from those flows using more than their fair sure of the queues, according to

$$w\left(i\right) = \frac{n\left(i\right)}{n/v}$$

where n/v is the number of queue slots occupied per flows and $n\left(i\right)$ is the number occupied by the ith flow. Hence drop datagrams from the ith connection if $n > R$, some congestion threshold, and $w\left(i\right) > 1$. The downside is that the switch has to keep track of number of flows, total occupancy, and occupancy per connection, which doesn't scale. The solution is to aggregate flows.

(c) AAL5 divides datagrams into cells and then reassembles these cells into original datagram at the receiving host and does a CRC check. The setting of the "PAYLOAD TYPE" allows the indication of the end of a cell which then allows for algorithms mentioned in part (b) to work. This is called "convergence". AAL5 also does virtual circuit multiplexing which, by reducing number of cells needed to protocol data units, can increase efficiency.

(d) If convergence wasn't available as a service in AAL5 then neither PPD nor EPD could be implemented since it would be impossible to distinguish one datagram from another. Then cells would be dropped randomly and "useless cells being transmitted" would be insoluble. If CRC were not offered it would bit error rates would be much higher and re-transmission would happen much more often.

10. (a) Hand-off between access points.

11. (a) A cellular system services a large number of users over an extended geographical region, and yet has a very limited frequency spectrum. As number of users grow the demand for channels will exceed the available spectrum.

(b) This spectral congestion problem is resolved, high capacity is achieved, by replacing a single high power base stations with many smaller, lower power base stations, and limiting the coverage of the base station to a localized region, a cell. Each smaller base stations is allocated a number of channels and then the same channels are reused in other cells.

(c) i. A problem may occur when moving from one base station to another while keeping the call uninterrupted, i.e. keeping the signal strength above some threshold. This is called the hand-off problem.

ii. Frequency-planning/channel-allocation is another problem. It is the problem of how to allocate in the most efficient way the spectra to each cluster, considering that sometimes there are shifting demands on the clusters. That is to say that sometimes one cluster will have very little demand placed on it while others have much demand and spectra should be allocated proportionally.

iii. You also get co-channel interference and adjacent channel interference.