PGT Exam Performance Feedback 2016/2017 Semester 2

COMP61242 Mobile Communications

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Comments Q1

1.a. Mostly reasonable answers were given. But some did not answer the question just describing protocols or behaviours at each layer and therefore accumulating few marks.

1.b. This got a few irrelevant answers but most mentioned costs/economics of satellite launches, extra timing delays/latency but not distinguishing between geostationary with very long latency compared to low earth orbits at a few hundred miles but needed frequent handovers. Mention of bandwidth usage and wide coverage area leading to resourcing issues was mentioned by some answers.

1.c. This was all about knowing roughly what should happen and listing a sequence of actions and the types of frames and their contents needed. A lot of students, perhaps sensibly stuck just to the routing part as this is where they had the most specific and usable taught knowledge. This was OK as the marking scheme was just looking for relevant items of knowledge and sensible suggestions with no partitioning into the 3 suggested areas necessary. However, there were some answers with next to nothing useful said.

Q2
This question was very similar to previous problem solving questions. Most did very well losing just a few marks down to mistakes not true errors.

2.a. Almost everybody got the marks for this.

2.b. Again almost everybody got the marks for this. I gave the specification in both decimal and octal which most understood. For those who assumed an 8 bit not 4 bit generator function the marking scheme was adapted so marks were not taken away.

2.c.
Most did a good job of this. But almost everybody got 1 or 2 bits wrong which was easily done.

2.d. There was a big mixture of answers that were blank, wrong and some completely correct in roughly even proportions. The word 'frame' should have said 'slot'. This does not effect this part but did effect part e below.

2.e. Due to the word 'slot' being wrong in 2.d most simply did the sum and got an OK answer. Some had problems deciding whether 114 or 148 bits were sent. I'd intended it to be 148 but was not too upset if 114 was assumed. Therefore, this part was almost a marks giveaway for most people!

Q3
3.a. One way delay is hard to measure accurately because it has to rely on two almost never synchronized clocks. Therefore it will be open to the clock having been set wrong, having drifted.... Round trip, one way to resolve this works because time is measured only by one clock and usually only after a very short delay between sending and return meaning any clock drift will be minimal. However, with round trip you have to divide by 2 and make an allowance for delays varying in the network and also delay in between the far end receiving and replying. A better, sometimes used system is to synchronize both clocks to an external source such as GPS's atomic clocks.

3.b. This was the hardest bit of the question. Little or no jitter means there are no packets lost and every packet arrives with well in time for playback. Therefore, it might naively seem that PLC is never going to be needed because all the data is being delivered in time to be played through the sound card. However, as mentioned above, there is always some clock drift! Imagine the receiver’s clock is running very slightly faster than the senders. After a bit, the receiver will start playing packets early. Typical jitter buffers only hold a few packets say 2 or 3. In a longish conversation the drift may cause the playback to empty the jitter buffer even though nothing has gone wrong or failed, then, with nothing to play the playback would crash so the receiver has to insert an extra bit of sound. The extra sound could be a complete sound or as little as 1 extra sample but the extra added is generated by some PLC algorithm. Many systems add a few samples as needed every so often to correct for drift before it becomes a problem.

Alternately, the source may be sending at a slightly faster rate than the receiver can play the sound. In this case the buffer might eventually overflow losing a complete packet. This can be avoided by timing a few samples from packets as needed to ensure the buffer does not overflow or need to be made larger which would imply a larger pre playback delay.

Some answers talked about bit errors in delivered packets, this was OK and earned marks but in practice error detection in a wireless system would probably detect this and drop the packet leading to a big jump in jitter - which you are told doesn't happen. On a wire only network, errors like this are very rare but could find their way through to playback as CRC in UDP is optional and there is no checksum in RTP, so this is a genuine though rare issue. But, the error would probably, in the wired case be played to the listener.

3.c. Here there was no expectation of a highly technical answer with details of Codes and signal processing. Most answers discussed that only a fairly narrow band of frequencies is needed for told quality speech so the war's bandwidth of roughly 20KHz can be cut to 3.5-4KHz. 4KHz signals can be accurately reconstructed with 8K samples per second. But if a sample is 8 bits that is equivalent to ITU G711's 64kbps which is too big. So a way to send fewer bits had to be found. There were a few variants mentioned such as nonlinear frequency to same mapping, Varying the accuracy of frequency representation such as mapping, for example (not a real case!) , all frequencies between 2.5KHz and 4KHz to a single value so playback would perhaps produce 3250Hz as output.

3.d almost everybody realised zero stuffing is horrible for the ear so selected better PLC methods. Dealing with the congestion seemed to be harder with many suggestions that would lead to more traffic not less. However, the 200ms delay left some scope to increase the delay between frames beyond 50ms making frames bigger but having fewer of them. The amount of data could also be cut by using Codes with a lower bit rate thus allowing fewer or smaller packets. The last two ideas reducing the congestion a bit.

Overall very good performance in Q4 and Q5.

Q4a was answered by most students as if the question was cellular vs wifi, which was not the case. The main idea here being that wireless is about the communication technology not using any wires whereas mobility is about allowing devices to move across networks seamlessly. For instance, you can have non-mobile wireless (e.g. a sensor network based on static sensors that communicate over wifi) or mobility over wired (e.g multi-site offices where you'll plug your laptop to different sockets/subnets).

Q4b was overall better with most students being able to differentiate between different levels of addressing and their uses. One of the main points that was missing is that you could do global routing based on MAC, but given the lack of hierarchy, it'd require too large routing tables.

Q4c had a good performance overall as well. Here the most missing point was the idea of leasing and its management.

Q5a was answered correctly by most students.

Q5b had good performance in general as well. Students mostly focused on Reactive algorithms, and only a few mentioned more sophisticated routings such as hierarchical, geographic or opportunistic.

Q5c was generally quite poor.