Solutions to Final Exam

Question 1

(a) When the phone is on-hook the ringer is connected across the line through a capacitor and little or no current flows. Current flows when the phone goes off-hook to signal the start of the call.

(b) The differential voltage across the subscriber's equipment will be approximately the same as the battery voltage when the phone is on-hook. It will drop (be lower) when off-hook.

(c) The hybrid on a CO line card separates the two directions of speech. The incoming speech signal is the input to the A/D converter and output from the D/A converter is the outgoing speech signal.

On a telephone set the hybrid also separates the speech signals in the two directions. On an analog phone set the main purpose of the hybrid is set the sidetone level which in turn affects how loudly the user speaks.

Question 2

(a) Signals at frequencies between 100–300 MHz would originate at the head end and be going downstream towards subscribers. Thus the downstream amplifier gains need to be adjusted.

(b) The spectrum analyzer measured the signal level in dBm (dB relative to a milliwatt) while the specifications for CATV signal levels are in dBmV (dB relative to a millivolt). 0 dBmV corresponds to -49 dBm at an impedance level of 75 Ω. The calculation of the power level corresponding to 0 dBmV was done in class and is as follows:

$$P = \frac{V^2}{R} = \frac{(1 \times 10^{-3})^2}{75} = 13 \times 10^{-9} \text{ W}$$

which is -78.8 dBW ≈ -49 dBm.

We need to increase the received level from -60 dBm to -49 dBm by increasing the gain by $-49 - (-60) = 11$ dB.

Question 3

(a) The symbol duration, not including the guard time, is the number of samples per symbol ($N = 256$) times the sample period: $\frac{256}{100 \times 10^6} = 2.56\ \mu$s.

(b) If we add on the 0.44μs guard time, the symbol period is $2.56 + 0.44 = 3\mu$s and the symbol rate is ≈ 333 kHz.

(c) If subcarriers 0–9 and 208–255 are not used then only subcarriers 10–207, a total of $207 - 10 + 1 = 198$, subcarriers are transmitted. 16-QAM-modulation transmits $\log_2(16) = 4$ bits per subcarrier and $198 \times 4 = 792$ bits per OFDM symbol. The bit rate is thus $792 \times 333 \times 10^3 = 264$ Mb/s.

Question 4

For a splitter loss of x, the received power will be 0 dBm (transmit power) minus $20 \text{ km} \times 0.4\text{dB/km} = 8$ dB (fibre loss) minus x (splitter loss). This should be equal to the receiver sensitivity of -25 dBm plus the 5 dB design margin or $-25 + 5 = -20$ dBm. Thus $0 - 8 - x = -20$ dBm and so $x = 12$ dB. An ideal N-way splitter has a loss of $10 \log N$ dB so $N = 10^{(12/10)} \approx 16$. Thus a 16-way splitter ratio could be supported by this system.

Question 5

(a) The shortest shortest possible IP+UDP packet with a 6-byte payload would include a 20 byte IP header, an 8 byte UDP header and 6 bytes for the UDP payload for a total of $20 + 8 + 6 = 34$ bytes.
(b) Each ATM cell except the last can carry 48 bytes while the last can carry 40. Since this IP frame will fit into the last cell, only one ATM cell is required to carry this packet.
(c) Since the last ATM cell can carry a 40 byte payload, $40 - 34 = 6$ bytes of padding are required.

Question 6

The PPP frame, with all values shown in hex, is:

`7E FF 03 11 7D 20 11 7D 5E 20 30 7E`

and starts with a flag character (`7E`), an address byte (`FF`), a control field (`03`) and a protocol type (`11`, only one byte long since the LS bit is 1). By default the frame ends with a 16-bit CRC and a flag character (`20 30 7E`). The remaining bytes (`7D 20 11 7D 5E`) are the PPP payload. However, each `7D` escape character has to be replaced with the next byte XOR'ed with `20`. The byte values are thus `20`⊕`20` = `00`, `11`, and `5E`⊕`20` = `7E`. The payload is thus:

`00 11 7E`

Question 7

(a) The `traceroute` utility sends ICMP echo request packets with small but growing TTL values in the IP header.

(b) `traceroute` discovers the route followed by IP packets by triggering "TTL exceeded" ICMP responses from routers that are successively larger number of hops away.

(c) `traceroute` expects to see ICMP "TTL exceeded" responses from intermediate routers and an ICMP echo response from the final destination.

(d) If no responses are received for TTL values larger than a certain value then the packets are being routed to a router that is not passing the ICMP echo request packets or a host that is not generating echo response packets. This is often the case for security reasons.

Question 8

The response to a TCP packet that has only the `SYN` flag bit set, a source port of 5000, a destination port of 80 and a sequence number of 1000, will contain the following:

(a) TCP flags `SYN` and `ACK` set

(b) source port 80 and destination port 5000

(c) Acknowledgment field set to 1 to acknowledge a "virtual" `SYN` byte before the data

Question 9

(a) If the signal to be digitized contains frequency components up to 500 kHz the sampling rate must be at least twice that: 1 MHz.

To keep the A/D quantization error less than 1 mV for a signal spanning the range ± 2V the number of quantization steps should be $> \frac{2-(-2)}{0.002} = 2000$ so the A/D converter should have a resolution of at least $\log_2(2000) \approx 11$ bits.

(b) The bit rate of an STS-n signal is n times the rate of an STS-1 or $n \times 51.84$ Mb/s. For STS-24 signal the rate is $24 \times 51.84 = 1244.16$ Mb/s.

(c) A T1 receiver that derives its clock from an incoming T1 signal would not see frame slips because the receiver and transmitter clocks are synchronous — running at exactly the same frequency.

(d) Each DHCP TLV-encoded option includes one option type byte, one length byte and as many value bytes as indicated by the length byte. The byte stream `12 03 20 36 3F 15 00 14 02 14 02` thus contains option `12` with 3 bytes of values (`20 36 3F`), option `15` with no (zero) values and option `14` with two bytes of values (`14 02`).

(e) A BIND-format `NS` RR entry for the domain `bcit.ca` with a value of `dns.bcit.ca` would be formatted as:

```
bcit.ca. IN NS dns.bcit.ca.
```

We would also have to include an `A` (address) record for the server with a value of 142.232.19.65:

```
dns.bcit.ca. IN A 142.232.19.65
```

(f) The SIP signaling protocol is only used to set up and terminate a VoIP call while the RTP transport protocol is used continuously to send and receive speech data. We would thus expect to see many more RTP packets than SIP packets.