CARLETON UNIVERSITY

Department of Systems and Computer Engineering

SYSC4700 Telecommunications Engineering Winter 2008

Term Exam – 14 February 2008

Duration: 80 minutes

Instructions:

1. **Closed-book exam (no aid-sheet). Use of non-programmable, non-communicating calculators is permitted.**
2. **Write answers in the spaces provided on the question sheet.**
3. **If necessary, use both sides of a page.**
4. **Write legibly, and state any assumptions that you make.**

Name:

Student Number:

Question	Mark	Max possible mark
1		25
2		27
3		27
Total		79

Question 1 [25 marks] – Link Budget

In this question we will determine the maximum achievable transmission rate in the downlink of a cellular wireless network. Here are the specifications of interest:

- BS transmit power: P_{TX} = 200 mW
- Transmitter (BS) antenna gain: G_{TX} = 7 dB
- Receiver (terminal) antenna gain: G_{RX} = 3 dB
- Quality requirement: SNR > 5 dB
- Carrier frequency: f = 2 GHz
- Receiver noise figure: F = 3.98 = 6 dB
- Ambient temperature: $T = 290^{o}$K
- Boltzmann constant: $k = 1.38\text{x}10^{-23}$ joule/^{o}K
- Path loss (PL): $(4\pi/\lambda)^2 d^4$, where
    - Distance between BS and a terminal: d
    - Carrier wavelength: λ
    - Speed of light: $c = 3\text{x}10^{8}$ m/sec.
- Cell radius: 200 m
- Maximum spectral efficiency according to Shannon's channel capacity theorem: $\mu_{\max} = \log_2(1+\text{SNR})$ bits/sec/Hz.

Calculate the maximum achievable rate in bits/sec at the cell edge.

[**Help 1**: $P_n = kTBF$ in linear scale.]
[**Help 2**: Note that B in not given in the question; it is to be calculated.]

Solution:

$$SNR_{dB} = 5\,\text{dB} \Rightarrow SNR = 10^{\frac{5}{10}} = 3.16$$

$$\mu_{\max} = \log_2(1 + SNR) = 2.06\,\text{bits/sec/Hz}$$

$$R_{\max} = \mu_{\max} B = 2.06B \quad (1)$$

Calculation of *B*:

$$SNR_{dB} = P_{TX} - P_L + G_{TX} + G_{RX} - P_n \quad (2)$$

$$P_{TX} = 10\log_{10}(0.2) = -7\,\text{dBW}$$

$$P_L = 10\log_{10}(\frac{4\pi}{\lambda} d^4) = 10\log 10(\frac{4\pi f}{c} d^4) = 130.5\,\text{dB}$$

$$P_n = 10\log_{10}(kTBF) = 10\log_{10}(k) - 197.98\,\text{dBW}$$

Substitute P_{TX}, P_L, P_n **in (2)** $\Rightarrow$

$$5 = -7 - 130.5 + 7 + 3 - 197.98 - 10\log_{10} B \Rightarrow$$

$$B = 3.53\,\text{MHz}$$

Substitute *B* in (1) $\Rightarrow$

$$R_{\max} = \mu_{\max} B = 2.06 \times 3.53\,\text{MHz} = 7.28\,\text{Mbits/sec}$$

[Blank Page for Question 1]

Question 2 [27 pts] – A/D Conversion and Time-Division Multiplexing

There is a high-speed line which can carry traffic at a rate 3.024 Mbits/sec. A number of analog voice signals will first be digitized and then will be multiplexed on to this high-speed line through TDM (time-division multiplexing). There is no control bits appended during the multiplexing operation.

There are two types of A/D conversion, low quality and high quality, with the below parameters.

- Low-Quality A/D Conversion: Sampling Rate = 6,000 samples/sec
Quantizer: 64-level
- High-Quality A/D Conversion: Sampling Rate = 18,000 samples/sec
Quantizer: 2048-level

Let us call the users whose voice signals go through the low-quality A/D conversion as type L users, and those whose voice signals go through the high-quality A/D conversion as type H users.

(a) **[5 pts]** How many type L users can be multiplexed on to the high-speed carrier?

$$\text{Rate for one L user } (\mathrm{R}_{\mathrm{L}}) = 6000 \text{ sample/sec} \times \log_2 64 \text{ bit/sample} = 36 \text{ kbits/sec}$$

$$\text{Number of L users} = \frac{3.024\text{ M}}{36\text{ k}} = 84 \text{ users}$$

(b) **[5 pts]** How many type H users can be multiplexed on to the high-speed carrier?

$$\text{Rate for one H user } (\mathrm{R}_{\mathrm{H}}) = 18000 \text{ sample/sec} \times \log_2 2048 \text{ bit/sample} = 198 \text{ kbits/sec}$$

$$\text{Number of H users} = \left\lfloor \frac{3.024\text{ M}}{198\text{ k}} \right\rfloor = 15 \text{ users}$$

(c) **[5 pts]** Next, consider that case where a mix of type L and type H users will be multiplexed. If there are *n* type L users and *n* type H users to be multiplexed, find *n*.

$$nR_L + nR_H = 3.024\, Mbits/\sec \Rightarrow$$

$$n = \left\lfloor \frac{3.024\text{ M}}{R_L + R_H} \right\rfloor = 12 \text{ users}$$

(d) **[12 pts]** Sketch the frame structure for the case described in Part (c).

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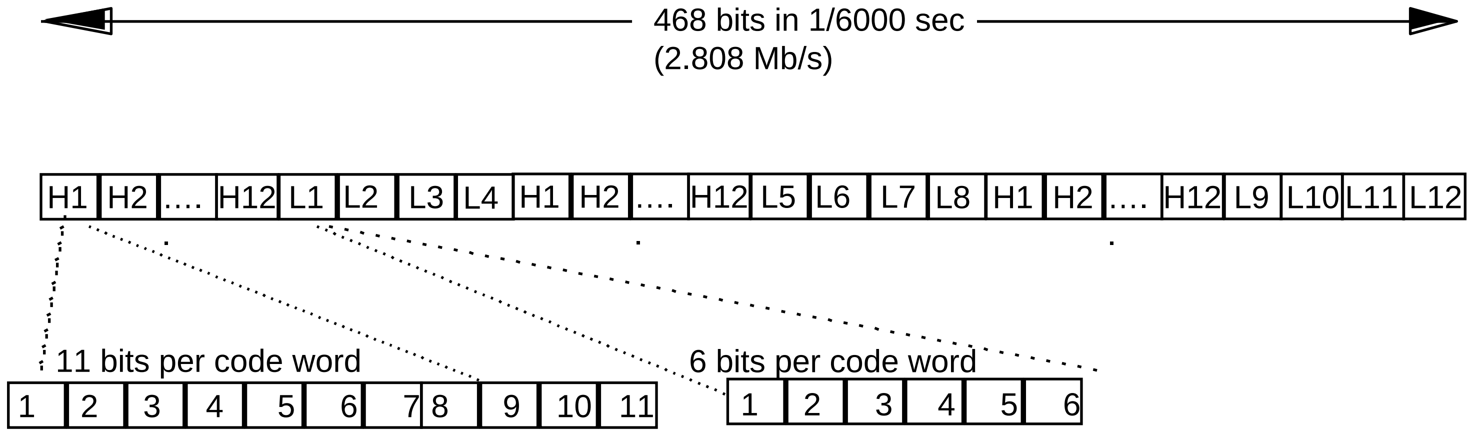

Question 3 [27 marks] – Short Questions on VoIP

(a) [5 pts] Give three reasons for deploying IP rather than TDM.

1. **Easier and cheaper maintenance: Integration of data and voice onto one network**
2. **Lower operating costs: Integration of remote offices over a common corporate data network, rather than through PSTN. Single Dial Plan.**
3. **Access from anywhere: Power users such as Teleworker and sales 'Road Warrior'. Global Access**
4. **Lower product costs: Integration of a voice application onto a central server, e.g. voice mail, means reduced number of devices. The remote sites no longer need their own local VM.**
5. **Security and resiliency: In NY (September 11th) the IP infrastructure kept running; the PSTN didn't**
6. **Future applications will be data centric, e.g. "Presence"**
7. **Displacement of current TDM systems and businesses**

(b) [4 pts] What is "triple play"?

Broadcast TV, Telephony and Internet

(c) [5 pts] What is the most commonly used VoIP signaling protocol? Why this protocol is very popular (give two reasons)?

SIP (Session Initiation Protocol) is the most commonly used VoIP signaling protocol, for the following reasons:

1. **More Client Server based and allowing Peer to Peer interaction.**
2. **Call control can be distributed**
3. **End devices need to be more intelligent than simple phones**
4. **Has the ability to evolve quickly, and scale to large numbers**
5. **Simple protocol, but lacks certain PBX capabilities**
6. **Vendor specific options provide features**
7. **Inter-vendor working is usually determined through "bake-off" but improving as more vendors implement agreed solutions**
8. **Networking features low, but improving**
9. **Open Standards through IETF, agreed by many established industry leaders**
10. **Continual proposal of new features and extensions**

(d) [4 pts] Which two technologies have the potential to become disruptive technologies in displacing the current (TDM) telephone network systems?

SIP and IPv6

(e) [5 pts] Identify three challenges for VoIP.

1. **Voice Quality**
2. **Delay, lost data, jitter, echo**
3. **Network issues, non deterministic, connectionless**
4. **Bandwidth, packet overhead, queue delays**
5. **Clock synchronisation**
6. **Security**
7. **Emergency Location E911**
8. **IP address space**

(f) [4 pts] What is the most popular jitter removal technique?

Leaky Bucket