PRINCE OF SONGKLA UNIVERSITY
FACULTY OF ENGINEERING

Mid-Term Examination: Semester I | Academic Year: 2005
Date: August 1, 2005 | Time: 09.00 – 12.00
Subject: 240-462 Multimedia Networking | Room: R300

ทุจริตในการสอบ โทษขั้นต่ำคือ ปรับตกในรายวิชาที่ทุจริต และพักการเรียน 1 ภาคการศึกษา

- There are TWO parts of this exam paper,
- **In part I**, there are TWO questions. Answers can be either in Thai or English. Write answers in the **BLUE** answer book.
- **In part II**, there are FOUR questions. Answers must be in English. Write answers in the **YELLOW** answer book
- All notes, books and calculator are allowed,

PART I

1. Explain the following technical terms clearly, more marks will be given if you demonstrate some examples:
   1.1 What is the voice packetisation? How does it work? (4 marks)
   1.2 There are 5 sources of delay, what they are (you should make their distinction clearly) (4 marks).
   1.3 What is jitter? (adding pictures to your explanation). (4 marks)
   1.4 What are the differences between E-Model and MOS (Mean Opinion Score)? (4 marks)
   1.5 Below are **clarity** factors in PSTN: (4 marks)
   - Intelligibility (capability of being understood
   - Noise
   - Fading (to lose strength)
   - Crosstalk

   The above factors appear in PSTN, however, there are other factors that only appear in VoIP (don't appear in PSTN). Please give such factors and their details.

2. Voice overhead and E-Model question (20 marks):

   2.1 Please complete the table below **(Table 1)**, (a) to (f), for the payload size and required bandwidth in network layer. Marks will not be given if you don't show your result calculation. (10 marks)

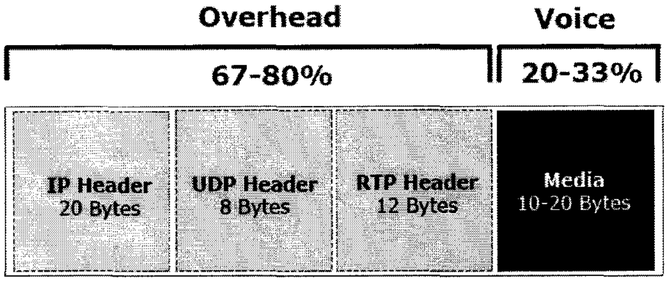

Figure 1 Over head in network layer

Table 1 Required bandwidth for G.711 and G.729 CODECes in Network layer

Encoding Format	Bit rate (kbits/s)	Packetisation interval (msec)	RTP payload size (bytes)	Required bandwidth (kbits/s)
G.711	64	30	(a)	(e)
		15	(b)	(f)
G.729	8	30	(c)	(g)
		15	(d)	(h)

2.2 **Figure 2** shows distortion impairment Ie as a function of the packet loss. **Figure 3** shows the influence of packet loss on distortion to R-factor. **Table 2** shows distortion impairment Ie for standardised low bit rate codes in E-Model. Please answer the below question based on information in **Figure 2**, **Figure 3** and **Table 2**. (10 marks)

2.2.1 Which codec gives the best performance in terms of R rating when % of packet loss is high.

2.2.2 Which codec gives good performance but when facing packet loss its performance drops significantly (getting worst).

2.2.3 At 2 percent of packet lose, which codec gives 2nd better choice.

2.2.4 From **Figure 3**, please indicate the maximum of packet loss of each codec (e.g. G.711, G.723, G.729) if the acceptable R-factor is 70.

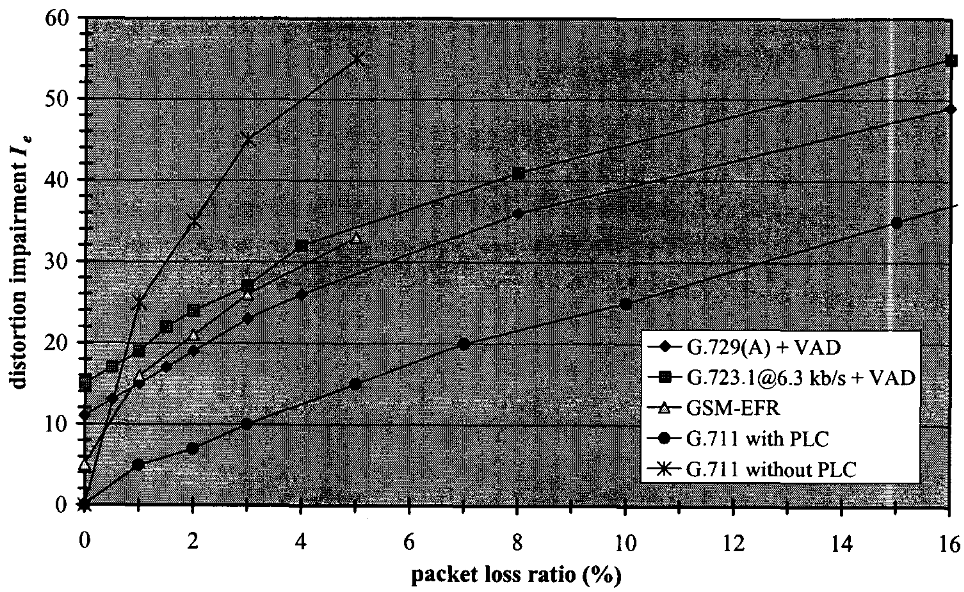

Figure 2 Distortion impairment *Ie* as a function of the packet loss

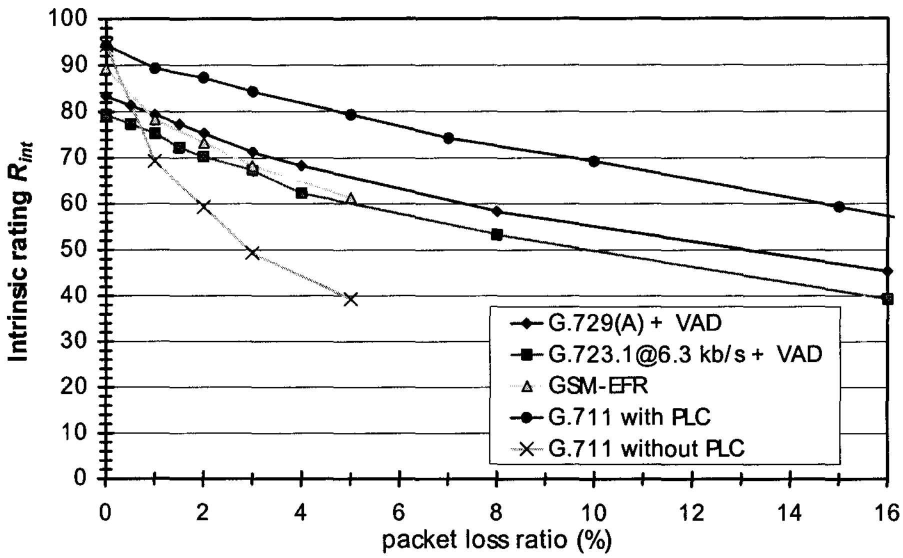

Figure 3 Influence of packet loss on distortion

Table 2 Distortion impairment *Ie* for standardised low bit rate codecs in E-Model

origin	standard	type	codec bit rate (kb/s)	I_e	intrinsic quality R
ITU-T	G.711	PCM	64	0	[illegible]
	G.726, G.727	ADPCM	16	50	[illegible]
			24	25	69.3
			32	7	87.3
			40	2	[illegible]
	G.728	LD-CELP	12.8	20	74.3
			16	7	87.3
	G.729(A)	CS-ACELP	8	10	84.3
	G.723.1	ACELP	5.3	19	75.3
		MP-MLQ	6.3	15	79.3
ETSI	GSM-FR	RPE-LTP	13	20	74.3
	GSM-HR	VSELP	5.6	23	71.3
	GSM-EFR	ACELP	12.2	5	89.3

PART II
(**YELLOW** answer book)

Instructions to Students for Part II:

- *Answer questions in English.* Good English is **not** required.
- Attempt all 4 questions.
- Write answers in the **YELLOW** answer book.
- Start the answer to each question on a new page.
- **Clearly Number** the answers. It is **not** required that questions be answered in order.
- Anything illegible is incorrect.
- Answer briefly where possible, essays are **not** required.
- The marks allocated for each question are shown next to that question. There are 60 marks total for this part of the examination.

Question 1. *(20 marks)*
For each of the following parts of this question, select the most appropriate answer from those given, and **explain** why that answer is the appropriate answer for the question. Simply selecting the answer that happens to be correct, without a good explanation why it is correct, will **not** result in full marks.

1. i) If you wanted to decrease the amount of jitter added to packet delivery times in a network under your control, your best strategy is likely to be:
A) increase network bandwidth
B) increase queue size in routers
C) decrease network bandwidth
D) decrease queue size in routers

1. ii) If your customers complain that too many packets are being dropped inside your network due to network congestion, and you want to reduce the loss rate, your best strategy is likely to be:
A) increase network bandwidth
B) increase queue size in routers
C) decrease network bandwidth
D) decrease queue size in routers

1. iii) Your network carries some data (packets) that must not be delayed any more than cannot be avoided. Which of the following will **NOT** assist in achieving the minimum delay?
A) use a priority queueing scheme
B) classify packets and set Traffic Class
C) implement Congestion Notification (the DEC bit)
D) provide the maximum possible bandwidth (link speed)

1. iv) We sometimes care about the minimum expected packet size when measuring or planning Quality of Service, because:
A) link layer requires at least N bytes/packet
B) small packets mean many packets, which increases the amount of time that is spent routing and queueing and then transmitting the data
C) need to calculate link layer overheads
D) more packets require more queue space

1. v) Integrated Services is not generally implemented on the Internet because:
   A) it is too new and code is not yet written
   B) the RSVP protocol consumes too much network bandwidth
   C) service providers do not want to provide quality of service
   D) there are too many packet flows in the Internet

Question 2. *(10 marks)*
In what ways, if any, does the use of IP multicast for the transmission of the audio (sound) of a meeting, rather than point to pont data transfer, to the participants affect the Quality of Service that the network can provide, or how it is provided?

Question 3. *(15 marks)*
An application needs to send 80,000 bits/second (10,000 bytes/second). It could achieve that by sending 10 packets per second, each containing 1000 bytes of data. Or it could achieve that by sending 100 packets per second, each containing 100 bytes of data. Explain the advantages and disadvantages of each choice.

Question 4. *(15 marks)*
4. i) Explain the purpose of the Realtime Transport Protocol. What functionality does it add that is not provided by other transport protocols? Why does RTP need UDP (the User Datagram Protocol) which is itself a transport protocol? [10 marks]

4. ii) What is the purpose of the Realtime Transport Control Protocol (RTCP)? Is it possible to use RTP without RTCP? If it is, what differences are there from RTP with RTCP? If it is not, what from RTCP is essential to RTP? [5 marks]