PRINCE OF SONGKLA UNIVERSITY

FACULTY OF ENGINEERING

Midterm Examination: Semester 1 | **Academic Year:** 2006-2007

Date: August 2, 2006 | **Time:** 09:00 – 12:00

Subject Number: 240-462 | **Room:** A401

Subject Title: Multimedia Networking

Exam Duration: 3 hours

This paper has 8 pages (including this page).

- This exam is divided into two parts.
- The first part should be answered in the **BLUE** answer book,
- The second part in the **YELLOW** answer book.
- There are 200 marks total for this exam, 60 marks are from part 1, the remaining 140 marks from part 2. This will contribute 50% of the course total.

Authorised Materials:

- Anything the student can carry (except mobile/cell phones.)

Instructions to Students:

- Attempt all 5 questions.
- **Clearly Number** the answers. It is **not** required that questions be answered in order.
- Anything illegible is incorrect.
- Show all calculations, not just the final result.
- Answer briefly where possible, essays are **not** required.
- Start the answer to each question on a new page.
- The marks allocated for each question are shown next to that question. There are 200 marks total for this examination.

Instructions to Students for Part 1:

- Answer Questions from Part 1 in the **BLUE** answer book.
- Answer questions in Thai or English.

Instructions to Students for Part 2:

- Answer Questions from Part 2 in the **YELLOW** answer book.
- *Answer questions in English.* Good English is **not** required.

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? Why do we need it? (5 marks)

   1.2 From the figure below, please explain process number 1 to 5 (e.g. how it works, what it affects in terms of performance, and quality) (10 marks).

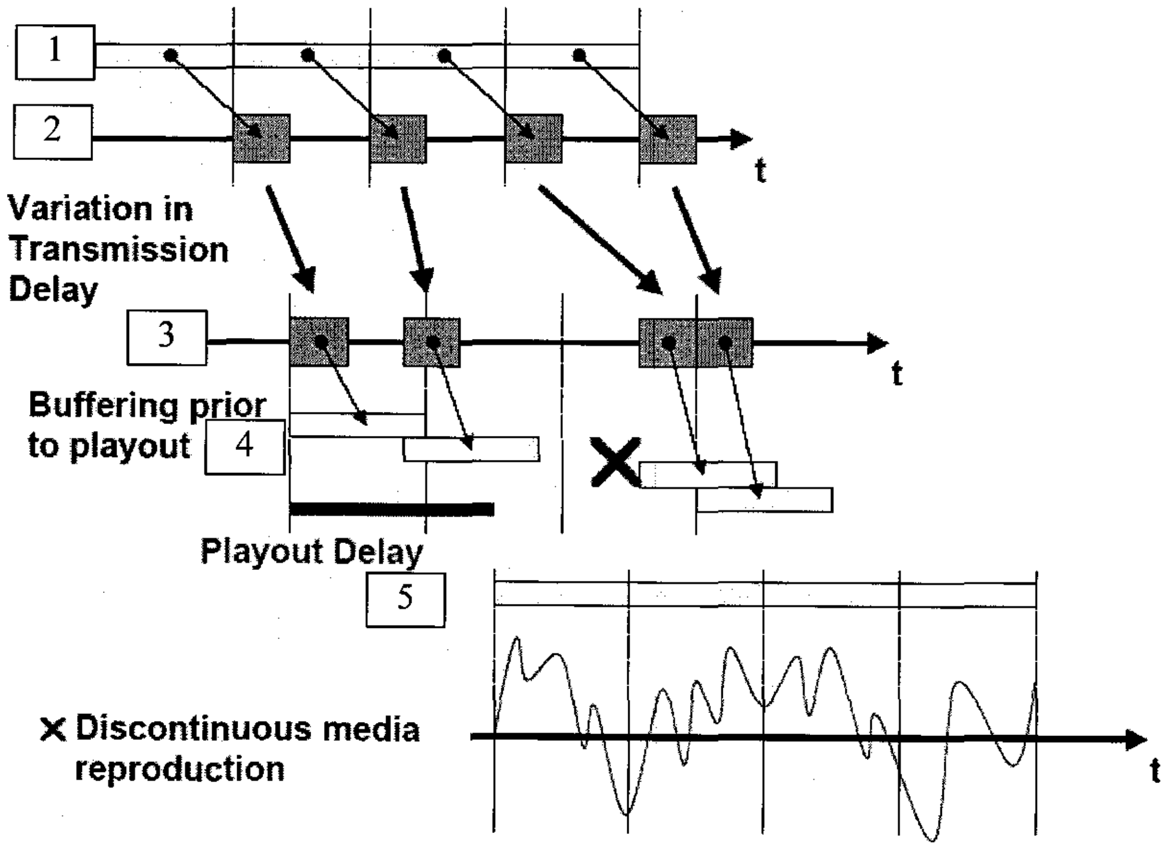

Figure 1 for question 1.2

   1.3 The following sub-questions are about jitter (5 marks)
      1.3.1 What is jitter?
      1.3.2 How does it happen?
      1.3.3 What are its effects in terms of QoS?
      1.3.4 How can we remove the jitter?

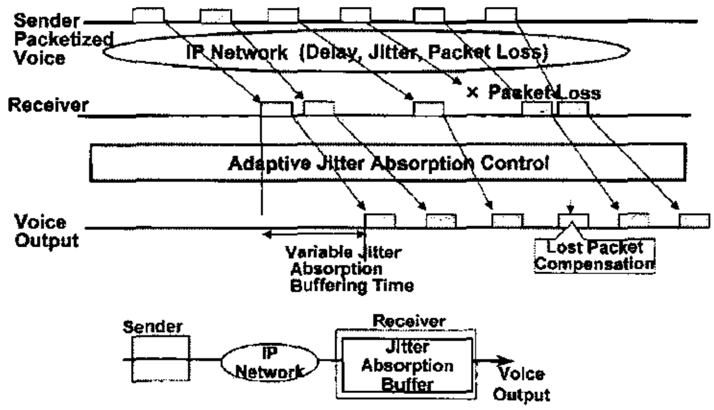

Figure 2 for question 1.3

1.4 Please explain the figure below (5 marks)

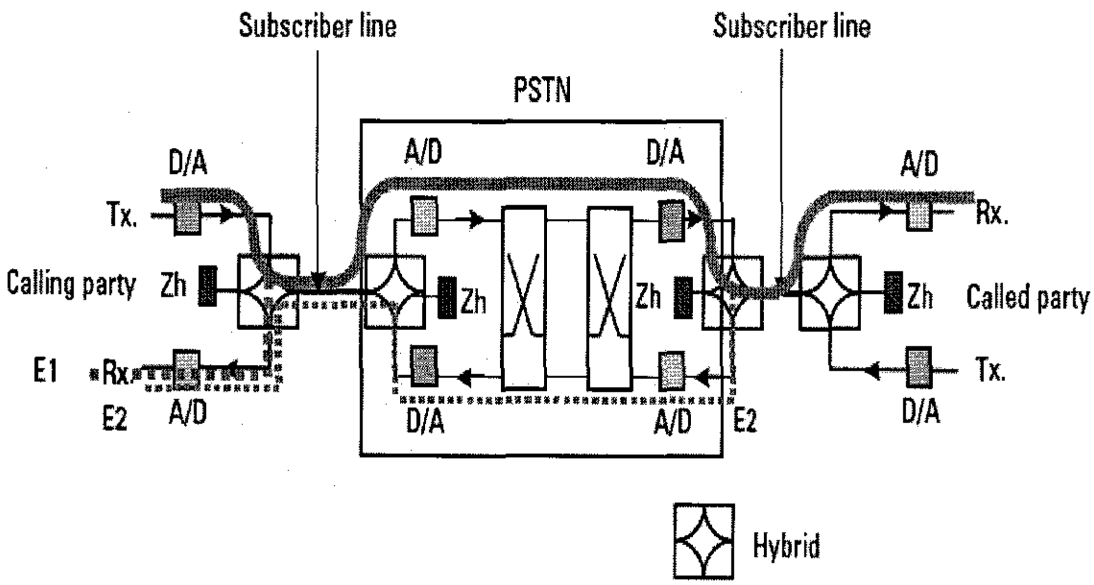

Figure 3 for question 1.4

1.5 Below are **clarity** factors in PSTN: (5 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 over head and performance

   2.1 Please complete the table below (**Table 1**), (a), (b), (c), (d), (e), and (f), for the required bandwidth in network layer (10 marks).

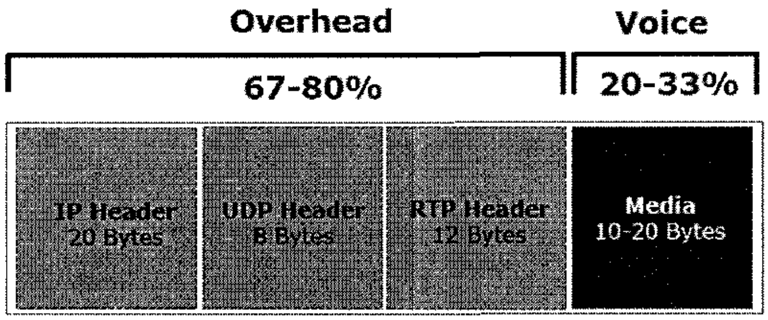

Figure 4 Over head in network layer

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

Encoding Format	Bit rate (kbites/s)	Packetisation interval (msec)	RTP payload size (bytes)	Required bandwidth (kbits/s)
G.711	64	20	160	(a)
		30	(e)	(b)
G.729	8	20	20	(c)
		30	(f)	(d)

2.2 **Figure 5** shows distortion impairment Ie as a function of the packet loss. **Figure 6** shows the influence of packet loss on distortion to R-factore. **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 5**, **Figure 6** and **Table 2**.

2.2.1 What does codec give the best performance in terms of R rating when % of packet loss is high (5 marks).

2.2.2 What is codec giving good performance but when facing packet loss its performance drops significantly (getting worst) (5 marks).

2.2.3 At 2 percent of packet lose, what codec gives 2^{nd} better choice (5 marks).

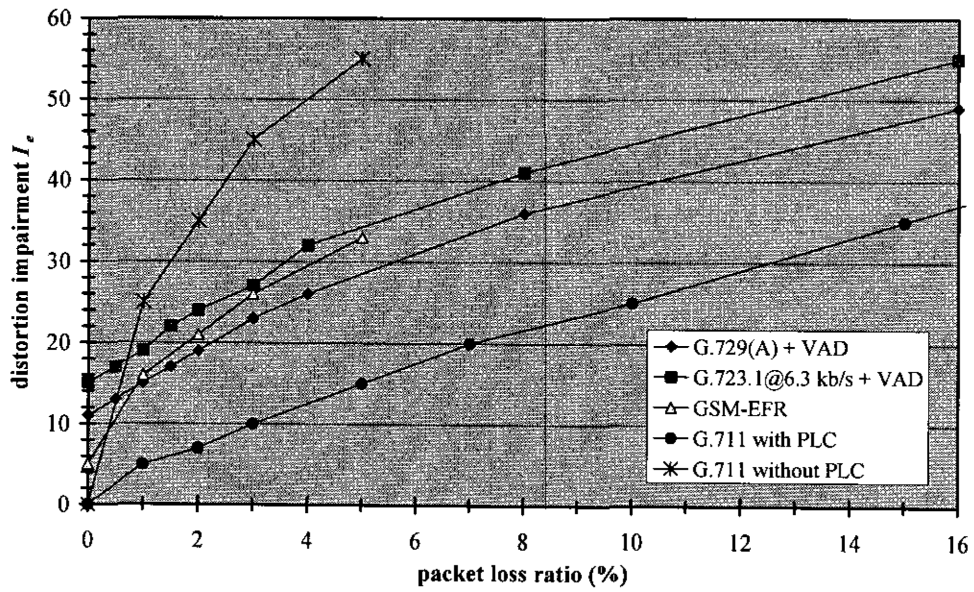

Figure 5 Distortion impairment *Ie* as a function of the packet loss

2.2.4 From **Figure 6**, 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 (5 marks).

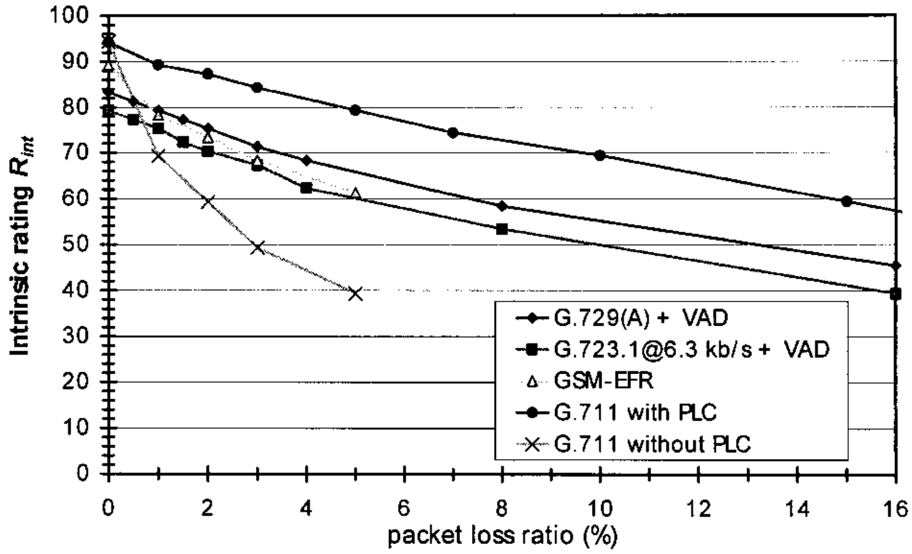

Figure 6 Influence of packet loss on distortion

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	[redacted]
	G.726, G.727	ADPCM	16	50	[redacted]
			24	25	69.3
			32	7	87.3
			40	2	[redacted]
	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 Two

Answer in **English** *in the* **YELLOW** *answer book.*

Question 3. *(80 marks)*

A token bucket is established in a router to control a packet flow. The parameters for the token bucket are:

r: 100 tokens every 20 milliseconds (rate)
b: 500 tokens (token bucket maximum)

The flow the token bucket is controlling is described using the token bucket, and the following additional parameters:

M: 500 bytes (max packet size)
m: 100 bytes (min policed unit size)
p: 10,000,000 bytes/second (peak data rate)

One token is required for this flow for every data byte in a packet that is part of the flow. No tokens are used for header bytes.

The token bucket is initialised (and empty: contains no tokens) at time zero, with the first tokens added 20 ms later. All relevant queues are also empty at time zero, and no packets related to this network flow exist.

For the purposes of this question, assume that the transport and network headers each total 20 bytes (40 bytes for both together), and the link layer header for the outgoing link from this router for the flow also adds 20 bytes of header (including other overhead).

3. A) Calculate the application to application bandwidth (in terms of data bytes per second) described by this flow.

[10 marks]

3. B) Determine the bandwidth (in bytes/second or bits/second) that would need to be reserved on the outgoing link to guarantee that this flow can be handled.

[15 marks]

For the remaining parts of this question, consider the following sequence of arriving packets, with the packet arrival time given in units of milliseconds after time zero, and the packet size measuring the data bytes in the packet. The packet labels are simply names to be used to refer to the individual packets in this question and answer.

Label	Arrival Time	Size
A	30	**100**
B	40	**50**
C	60	**100**
D	70	**100**
E	80	**100**
F	120	**200**
G	130	**50**
H	140	**50**
J	150	**50**

3\. C) Assume that packets arriving when there are insufficient tokens in the token bucket are simply discarded, and that there are no other packets competing for the outgoing link (it is idle, so no packets are ever delayed by queueing), show which packets are transmitted on the outgoing link, and at what times (relative to time zero). You should assume that the processing delay in the router is negligible – that is, it is possible for a packet that arrives at time zero can also be transmitted at time zero.

[30 marks]

3\. D) Assume instead that packets arriving when there are insufficient tokens in the token bucket are queued until sufficient tokens accumulate, with a maximum queue length of 2 packets (with packets arriving when the queue is full being discarded), and that there are no other packets competing for the outgoing link (it is idle, so no packets are ever delayed by queueing), show which packets are transmitted on the outgoing link, and at what times (relative to time zero). You should assume that the processing delay in the router is negligible – that is, it is possible for a packet that arrives at time zero can also be transmitted at time zero.

[25 marks]

Question 4. *(20 marks)*

Explain, **briefly** why Internet Service Providers (ISPs) prefer to implement **Differentiated Services** rather than **Integrated Services** when they implement quality of service.

You should refer to the major distinguishing features of each of those QoS protocols in your answer.

Question 5. *(40 marks)*

Explain the characteristics of a network attempting to be modelled by:

A) The fluid model (water in a pipe)

B) The idle network model

Include in your answer the qualify of service characteristics available in each of those models, and the requirements placed upon applications seeking to use each model.

Give examples of applications (or application protocols) which might prefer a network which offered each of those quality of service models.