

**PRINCE OF SONGKLA UNIVERSITY
FACULTY OF ENGINEERING**

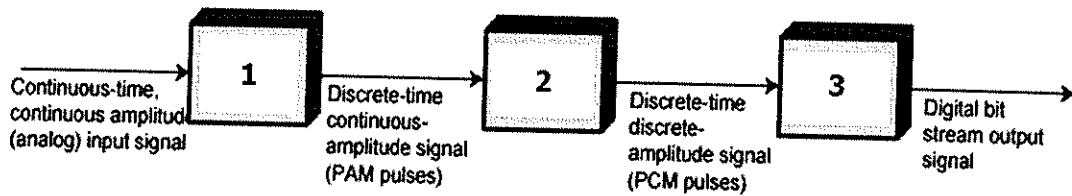
Mid-Term Examination: Semester I
Date: 12 October 2016
Subject: 242-460 Multimedia Networks

Academic Year: 2016
Time: 09.00-11.00 (2 hrs)
Room: S103

ทฤษฎีในการสอบ โทษขั้นต่ำคือ ปรับตกในรายวิชาที่ทฤษฎี และพักการเรียน 1 ภาคการศึกษา

- All types of calculators, dictionaries and electronic devices are not allowed.
- All notes and books are not allowed.

1. Below is the process how to digitise analog voice to digital voice. Each box receives its input and produces the output as described in the picture below. Please state what process is:



Answer

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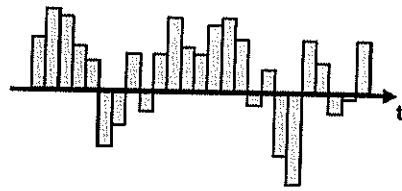
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2. Below is the process steps how to process analog voice input to voice packet. Please describe what the step names are.

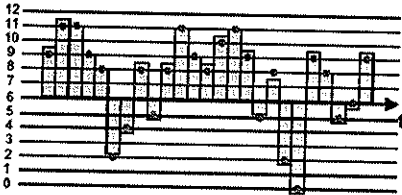
3.



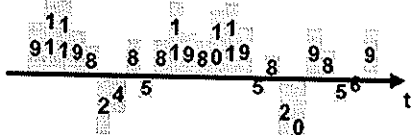
1) Analog input signal



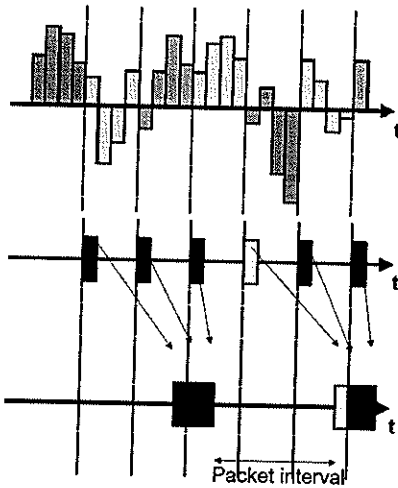
2)



3)



4)



5) Voice Codec encoding

6)

4. VoIP performance

4.1 E-Model is described in ITU G.107 as follows:

$$R = R_o - I_s - I_d - I_e + A$$

- *R* Transmission rating factor
- *R_o* Basic signal-to-noise ratio (SNR)
- *I_s* All simultaneous impairments to voice signal, e.g. loudness, PCM quantization distortion
- *I_d* All delays (impairments after voice signal caused by delays), e.g. echo, delay
- *I_e* Distortion impairment caused by Equipment Impairment factor, low bit rate codec, packet loss
- *A* Expectation factor

The following test conditions are used:

- Codec G.729A (+VAD)
- The packet loss is 2%.

- End-to-end delay is 150 msec.
- Echo lose is 51 dB.

What is R value? (10 marks)

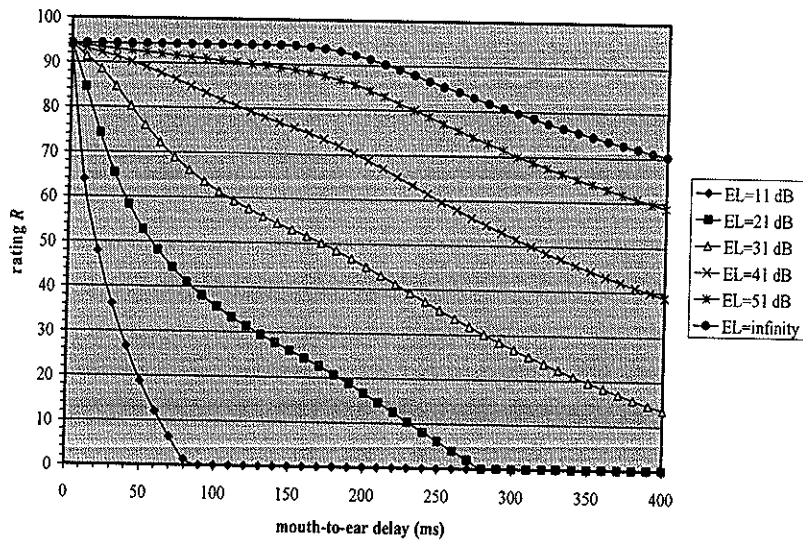


Figure 1 R rating vs mouth-to-ear delay

Table 1 Distortion impairment I_e 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	
	G.726, G.727	ADPCM	16	50	
			24	25	69.0
			32	7	87.5
			40	2	
	G.728	LD-CELP	12.8	20	74.3
			16	7	87.5
			G.729(A)	CS-ACELP	8
	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	89.0

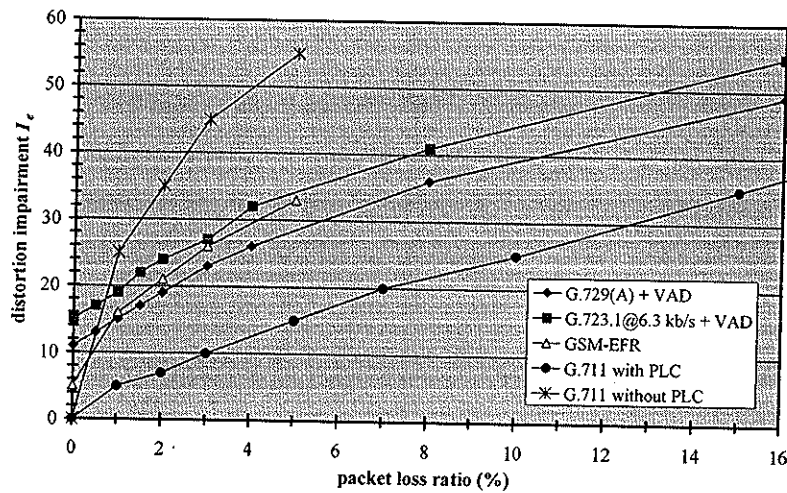


Figure 3 Distortion impairment I_e as a function of the packet loss

5.1.4 From Figure 4, 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).

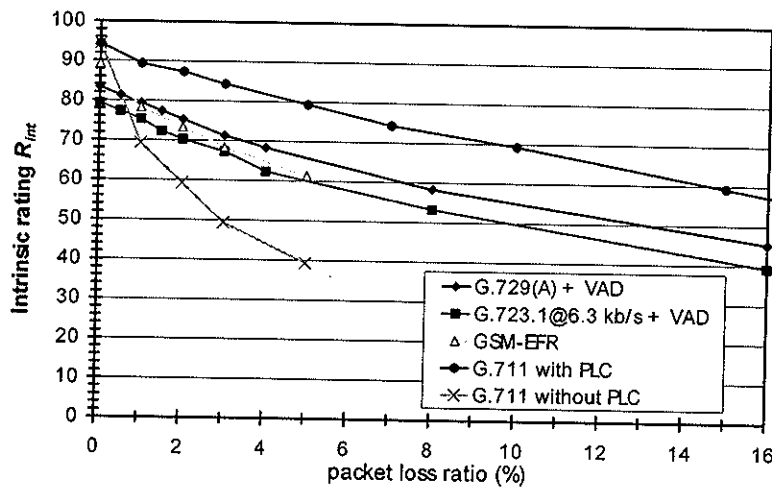


Figure 4 Influence of packet loss on distortion

Table 2 Distortion impairment I_e 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	
	G.726, G.727	ADPCM	16	50	
			24	25	69.3
			32	7	87.3
			40	2	
	G.728	LD-CELP	12.8	20	74.3
			16	7	87.3
			G.729(A)	CS-ACELP	8
	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

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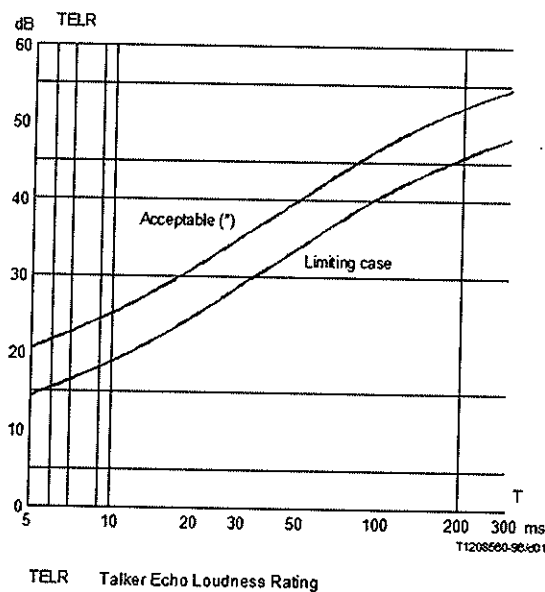
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7. Please explain and interpret the effect of TELR (Talker Echo Loudness Rating) as shown in the graph below (e.g. what the relationship between delay time, echo, and loudness) (5 marks)



Answer

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8. Please use the below information to calculate the voice utilisation as follows:
 The VoIP uses G.729A CODEC with 20 msec packetized interval time. VoIP Clients are using 100 Mbps Ethernet (IEEE 802.3). Please find the following answers:
 a) What is the bandwidth needed for each VoIP client? (5 marks)
 b) If some clients are connected to WiFi, IEEE 802.11g, what is the minimum bandwidth needed for such VoIP clients? (5 marks)

- a) What is R value when G.711 is transcoded to G.729? (5 marks)
- b) What is R value when G.729 is transcoded to G.723.1 (5.3 kbps)? (5 marks)

Answer

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10. Below is the E-model at the Transport Layer. Please use all below information as well as other related information (from previous question) to answer (10 marks).

Table Id value

One-way delay (msec)	Id
0	0
25	0.9
50	1.5
75	2.1
100	2.6
125	3.1
150	3.7
175	5
200	7.4
225	10.6
250	14.1
275	17.4
300	20.6
325	23.5
350	26.2
375	28.7
400	31

Answer

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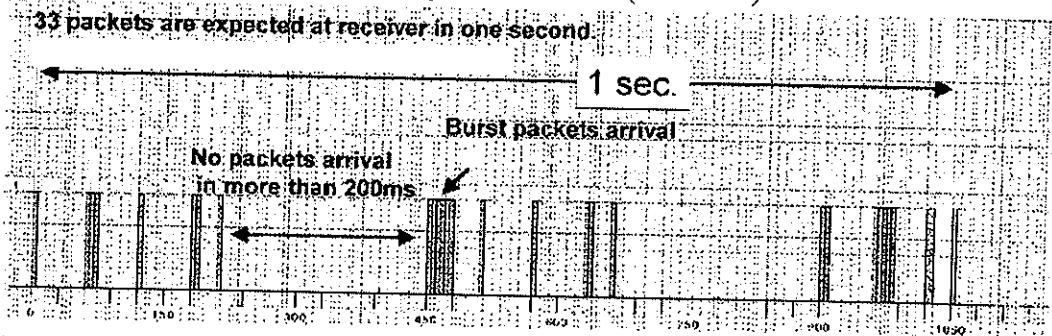
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12. Below is the measuring of VoIP packet arrival at the destination node. The VoIP Codec used is G.723.1 encoded RTP voice packet stream. On the sending side, the packets were sent at rate 33 pk/sec. What is the percentage of packet lose? What is the minimum buffer size required for jitter removal? (10 marks)



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