PRINCE OF SONGKLA UNIVERSITY

FACULTY OF ENGINEERING

Final Examination: Semester 2 **Academic Year: 2010**

Date: 22/2/2011 **Time: 1330-1630**

Subject: 241-464 Multimedia Networking **Room: Robot**

ชื่อ-นามสกุล .... รหัสนักศึกษา ....

<u>หมายเหตุ</u>

1. ข้อสอบมีทั้งหมด 2 ตอน ตอนที่ 1 จำนวน 85 ข้อ และตอนที่ 2 จำนวน 3 ข้อในกระดาษคำถาม 15 หน้า (ไม่รวมหน้านี้ และคำแนะนำเพิ่มเติมของตอนที่ 1)
2. ห้ามการหยิบยืมสิ่งใด ๆ ทั้งสิ้น จากผู้อื่น ๆ เว้นแต่ผู้คุมสอบจะหยิบยืมให้
3. ห้ามนำส่วนใดส่วนหนึ่งของข้อสอบออกจากห้องสอบ
4. ผู้ที่ประสงค์จะออกจากห้องสอบก่อนหมดเวลาสอบ **แต่ต้องไม่น้อยกว่า 30 นาที** ให้ยกมือขออนุญาตจากผู้คุมสอบก่อนจะลุกจากที่นั่ง
5. เมื่อหมดเวลาสอบ ผู้เข้าสอบต้องหยุดการเขียนใด ๆ ทั้งสิ้น
6. ผู้ที่ปฏิบัติเข้าข่ายทุจริตในการสอบ ตามประกาศคณะวิศวกรรมศาสตร์ **มีโทษ คือ ปรับตกในรายวิชาที่ทุจริต และพักการเรียน 1 ภาคการศึกษา**
7. ให้นักศึกษาสามารถนำสิ่งต่อไปนี้เข้าห้องสอบได้

   ☐ ตำรา ☐ หนังสือ

   ☐ เครื่องคิดเลข ☐ กระดาษ A4 ...... แผ่น

   ☐ พจนานุกรม

   ☐ อื่น ๆ ....
8. ให้ทำข้อสอบโดยใช้

   ☑ ดินสอ ☑ ปากกา

ผู้ออกข้อสอบ อ.สินชัย กมลภิวงศ์ และ อ. สุธน แซ่ว่อง

นักศึกษารับทราบ ลงชื่อ ....

รหัสนักศึกษา ....

Part I: อ.สินชัย

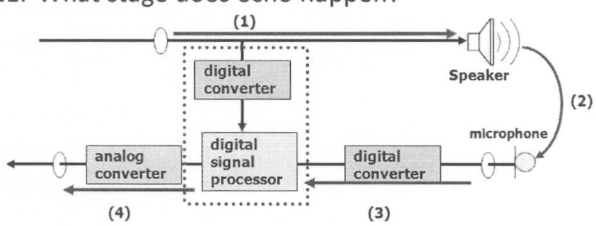
Instructions

- There are 85 questions, attempt to do them all
- Marking scheme
    - 2 marks for the right answer of each question,
    - -1 (minus one) for each wrong answer
    - No penalty if you leave an empty answer.

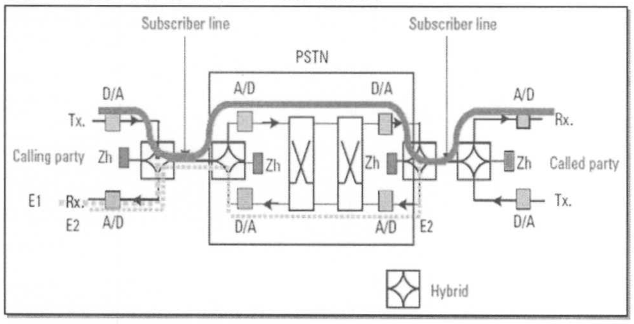
1. What is Packet delay?
    a) This delay is caused by the actual process of collecting the encoded samples into a packet for transmission over the packet network
    b) This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder.
    c) This delay is caused by the physical medium and protocols used to transmit the voice data.
    d) The delay problem is compounded by the need to remove a variable inter-packet timing caused by the network a packet traverses.
    e) None of above.
2. What is Jitter delay?
    a) This delay is caused by the actual process of collecting the encoded samples into a packet for transmission over the packet network
    b) This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder.
    c) This delay is caused by the physical medium and protocols used to transmit the voice data.
    d) The delay problem is compounded by the need to remove a variable inter-packet timing caused by the network a packet traverses.
    e) None of above.
3. Which one is true for jitter?
    a) A jitter buffer temporarily stores arriving packets in order to minimize delay variations.
    b) If a jitter buffer is too small then an excessive number of packets may be discarded.
    c) If a jitter buffer is too large then the additional delay can lead to conversational difficulty.
    d) All of them.
4. What is an effect if jitter buffer is too large?
    a) packets are dropped.
    b) lead to conversational difficulty.
    c) Packet overhead is large.
    d) Echo is removed.
    e) All of above.
5. Which one is not true for causes of packet loss?
    a) Network congestion
    b) Time expiry
    c) Time-out
    d) Buffer over flow
    e) No correct answer
6. If we would like to increase a number of voice channels, what techniques can be used
    a) Using voice codec
    b) Using voice multiplexing
    c) Increasing a play load size
    d) Increasing a packetise time
    e) All of above
7. If we would like to increase a number of voice channels, what techniques can be used
    a) Using voice codec
    b) Using voice multiplexing
    c) Increasing a play load size
    d) Increasing a packetise time
    e) All of above
8. What is a delay boundary (for one-way delay) echo cancellation is required?

a) 15 ms
b) 25 ms
c) 50 ms
d) 150 ms
e) No correct answer

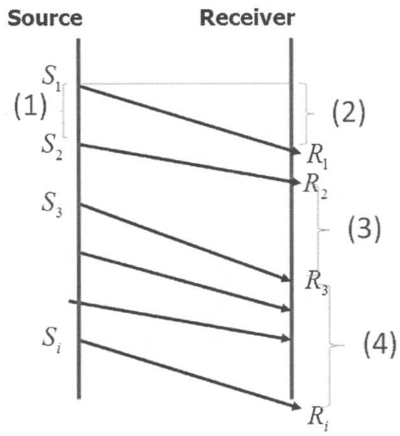
9. What is a cause that the listener hears annoying pops & clicks?
   a) replays the last successfully received packet.
   b) Packet loss during pay-out.
   c) Jitter remove process.
   d) Voice encoding.
   e) Voice buffering and queueing delay.
10. What happen when an out of order condition is detected?
   a) Out of order packets are played in the order they arrive.
   b) Out of order packets are re-ordered and inserted.
   c) Out of order packets are dropped.
   d) Ask for a re-transmit of these out of order packets.
   e) No correct answer
11. What stage does echo happen?



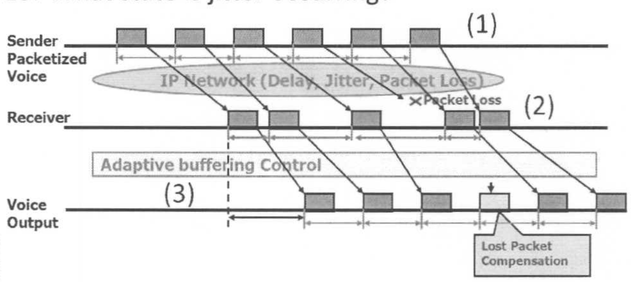
   a) (1)
   b) (2)
   c) (3)
   d) (4)
12. What is a cause of this echo?



   a) Hybrid
   b) Long delay
   c) A/D and D/A problem
   d) Low signal to noise ration
   e) No correct answer
13. Which one is a cause of QoS degradation
   a) CPU overloaded
   b) Network congested
   c) Router overloaded
   d) Gateway too busy
   e) All of them
14. Which one is a value of Jitter?

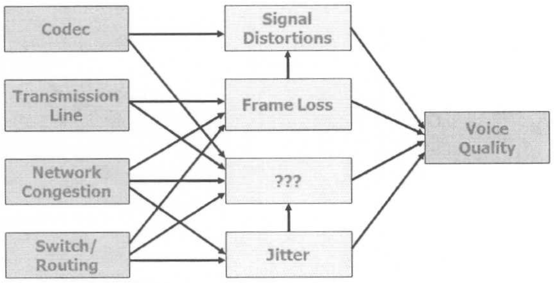


   a) (1)
   b) (2)
   c) (3)
   d) (4)
   e) No correct answer
15. What state is jitter occurring?



   a) (1)
   b) (2)
   c) (3)
   d) All of them
   e) No correct answer
16. What item is missing?

รหัสนักศึกษา ....



a) Buffering
b) Latency
c) Packet delay
d) Noise
e) No correct answer

17. What is “jitter”?
a) Variable of voice packet
b) Packet delay variation
c) Packet interval time delay variation
d) All of them
e) No correct answer

18. What is a cause of ‘jitter’?
a) Router is busy
b) Network is over-load
c) Packet delay is vary
d) Queueing delay
e) All of above

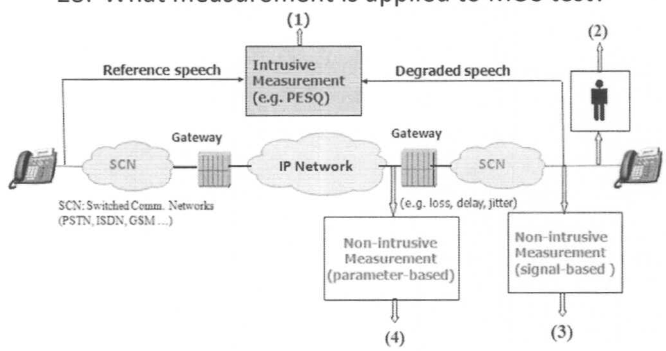
19. Which one is phenomenon of packet loss?
a) The more significant the change in the inter-arrival time.
b) abrupt rises in jitter value.
c) A large value of end-to-end delay.
d) All of above.
e) No correct answer

20. Which one is NOT voice quality measurement?
a) Mean Opinion Score (MOS).
b) Perceptual Speech Quality Measure (PSQM).
c) Measuring Normalizing Blocks (MNB).
d) Talker Echo Loudness Rating (TELR).
e) E-Model.

21. Which one does it describe for MOS (Mean Opinion Score)?
a) Computes the auditory distance based o how humans psycho-acoustically adjust for certain degradations
b) A computation model for use in transmission planning
c) Listening test conducted by real people
d) Using a speech-like test signal which consists of 30 seconds of male and female phonetic sounds.
e) No correct answer.

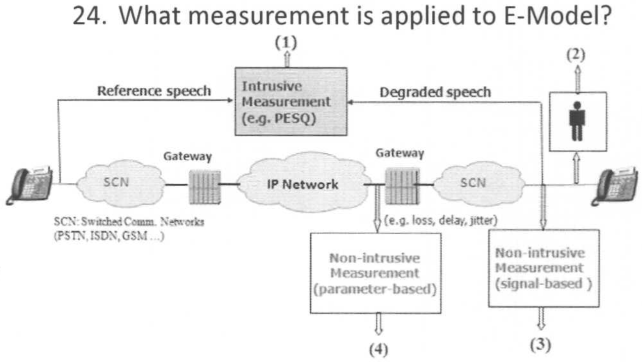
22. Which one is true for MOS and PSQM?
a) MOS and PSQM can be used to accurately measure impairments as a result of voice coding.
b) They can also reflect impairments as a result of frame loss.
c) However, it is more difficult to measure the effect of latency and latency variations using MOS and PSQM alone.
d) MOS and PSQM also do not provide information about the source of the impairment.
e) All of them.

23. What measurement is applied to MOS test?



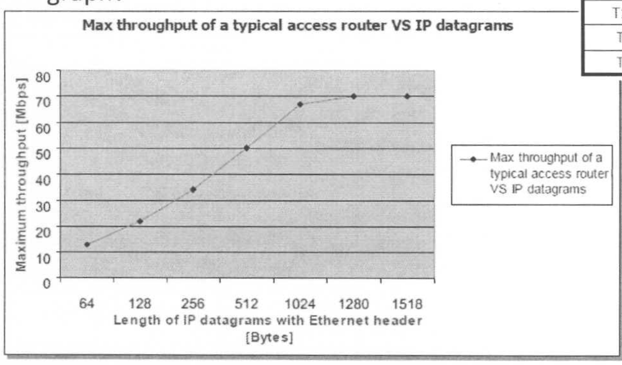
a) (1)
b) (2)
c) (3)
d) (4)
e) No correct answer

24. What measurement is applied to E-Model?



a) (1)
b) (2)
c) (3)
d) (4)
e) No correct answer

25. If we would like to increase a number of voice channels, what techniques can be used.
    a) Using voice codec
    b) Using voice multiplexing
    c) Increasing a play load size
    d) Increasing a packetise time
    e) All of above
26. Which one is not a source of fixed delay
    a) Algorithmic Delay
    b) Serialization Delay
    c) Propagation Delay
    d) Component Delay
    e) Network delay
27. Regarding to encoding standard, which one has the fastest voice encoding
    a) G.711
    b) G.723.1
    c) G.726
    d) G.728
    e) G.729
28. Which one is true regarding to the below graph?



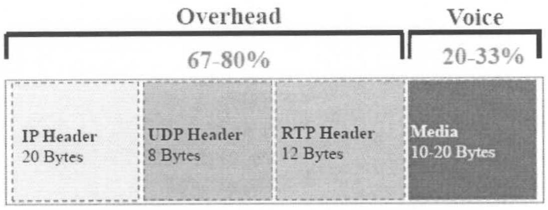
    a) A shorter packet gives higher throughput.
    b) A longer packet give a higher throughput.
    c) A longer packet gives a lower throughput.
    d) A shorter packet gives a moderate throughput.
    e) No correct answer.
29. What is a serialised delay?
    a) A delay time consumes during a data collection.
    b) A delay time consumes during a packet is shifted via a transmitter.
    c) A delay time consumes during voice encoding.
    d) A delay time consumes during voice compression.
    e) No correct answer
30. Which one is true?

Encoding Format	Bit Rate (kbits/s)	Packetization Interval (msec)	RTP Payload Size (Bytes)	Required Bandwidth (kbits/s)
G.711	64	20	160	80
		10	80	96
G.729	8	20	20	24
		10	10	40*1

    a) A longer packetization interval reduces a required bandwidth.
    b) A higher bit rate gives a larger RTP payload.
    c) Bandwidth requirement is based on packetization interval
    d) RTP payload is based on packetization interval.
    e) All of them.
31. From the table below, which one offers the highest voice utilisation?

Transmission facility (Mb/s)	Maximum delay variation (ms)	Number of voice calls supported			
		AAL-2	Frame relay	TDM	AAL-1/AAL-5
T1 (1.536)	20	123	125	24	72
T1(1.536)	5	104	108	24	72
T3(44.7)	20	4,090	3,500	672	2,108
T3(44.7)	5	3,964	3,024	672	2,108

    a) AAL-2 with 20 ms delay using T1
    b) AAL-2 with 5 ms delay using T1
    c) AAL-2 with 20 ms delay using T3
    d) Frame delay with 20 ms delay using T3
    e) Frame delay with 20 ms delay using T3
32. If we would like to increase a number of voice channels, what techniques can be used
    a) Using voice codec
    b) Using voice multiplexing
    c) Increasing a play load size
    d) Increasing a packetise time
    e) All of above
33. Calculate the bandwidth required for G.729 when packetization time is 10 msec



a) 24 kbps
b) 40 kbps
c) 45 kbps
d) 80 kbps
e) 96 kbps

34. From the given table, which codec can be the most tolerable mouth-to-ear delay?

Origin	standard	Codec bit rate (kb/s)	Mouth-to-ear delay bound (ms)
ITU-T	G.711	64	400
	G.728	12.8	212
		16	324
	G.729(A)	8	296
	G.723.1	5.3	221
		6.3	253
ETSI	GSM-FR	13	212
	GSM-HR	5.6	180
	GSM-EFR	12.2	345

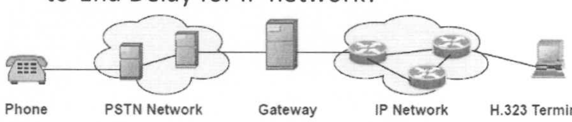
a) G 711
b) G.728
c) G.729
d) GSM-FR
e) GSM-EFR

35. Which codec is the most tolerable of the packet loss?

Origin	standard	Codec bit rate (kb/s)	Packet loss bound (%)
ITU-T	G.711 without PLC	64	1
	G.711 with PLC	64	10
	G.729(A) + VAD	8	3.4
	G.723.1@6.3 kb/s + VAD	6.3	2.1
ETSI	GSM-EFR	12.2	2.7

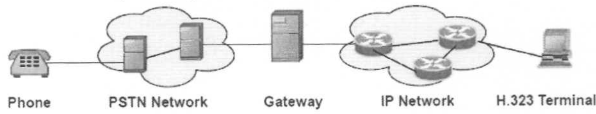
a) G.711 without PLC
b) G.711 with PLC
c) G.729(A) + VAD
d) G.723 +VAD
e) GSM-EFR

36. Which one is the Influencing Factors of End-to-End Delay for IP network?



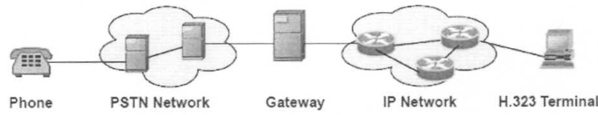
a) Fixed transmission time
b) Voice signal processing, Receive Jitter buffering
c) Buffering, Queuing
d) Transmit - packetization
e) All of above

37. Which one is the Influencing Factors of clarity Delay for IP network?



a) Microphone, loudspeaker quality
b) Hybrid (echo source)
c) Silence suppression
d) Packet loss
e) Speech codec

38. Which one is the Influencing Factors of clarity Delay for VoIP Terminal



a) Microphone, loudspeaker quality
b) Hybrid (echo source)
c) Silence suppression
d) Packet loss
e) Speech codec

39. Which component does affect the quality of CODEC?
a) Analog-to-digital conversion
b) Digital-to-analog conversion
c) Signal distortion
d) Linearity
e) All of them

40. What is a bandwidth required for (A)?

Encoding Format	Bit Rate (kbits/s)	Packetization Interval (msec)	RTP Payload Size (Bytes)	Required Bandwidth (kbits/s)
G.711	64	20	160	(A)
		10	80	96
G.729	8	20	20	24
		10	10	(B)

a) 24 kbps
b) 40 kbps
c) 46 kbps
d) 80 kbps
e) 96 kb

41. What is a bandwidth required for (B)?

Encoding Format	Bit Rate (kbits/s)	Packetization Interval (msec)	RTP Payload Size (Bytes)	Required Bandwidth (kbits/s)
G.711	64	20	160	(A)
		10	80	96
G.729	8	20	20	24
		10	10	(B)

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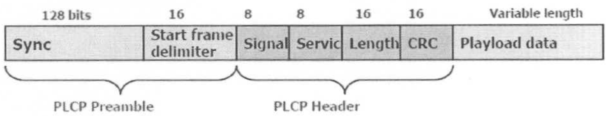
a) 24 kbps
b) 40 kbps
c) 46 kbps
d) 80 kbps
e) 96 kbps

42. Regarding to the picture below, what overhead is.



a) 18 bytes
b) 22 bytes
c) 24 bytes
d) Can not determine
e). No correct answer

43. Which one is true?
a) the mouth to-ear delay is smaller than 25 ms does not need echo canceller.
b) a mouth-to-ear delay of up to 150 ms is acceptable for most user applications,
c) a mouth-to-ear delay between 150 ms and 400 ms is acceptable.
d) a mouth-to-ear delay above 400 ms is unacceptable
e) All of them.

44. Which one is not a function of voice codec?
a) packetisation
b) Analog-to-digital conversion
c) Digital-to-analog conversion
d) Signal distortion
e). No correct answer

45. Which codec is tolerable mouth-to-ear delay bounds when there is no packet loss?
a) G.729A with VAD
b) G.723.1 (6.3kbps) with VAD
c) G.711 with PLC
d) G.711 wlo Packet Loss Concealment
e) GSM-EFR

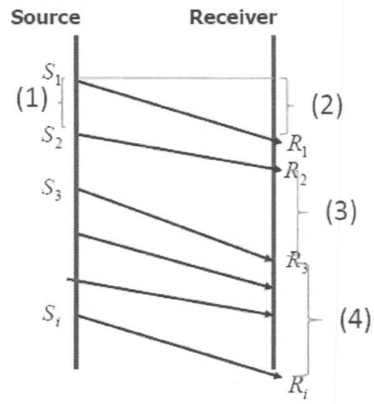
46. Which codec can tolerate a highest packet loss?
a) G.729A with VAD
b) G.723.1 (6.3kbps) with VAD
c) G.711 with PLC
d) G.711 wlo Packet Loss Concealment
e) GSM-EFR

47. What is a delay boundary (for one-way delay) echo cancellation is required?
a) 15 ms
b) 25 ms
c) 50 ms
d) 150 ms
e) No correct answer

48. Which one is not a source of fixed delay
a) Algorithmic Delay
b) Serialization Delay
c) Propagation Delay
d) Component Delay
e) Network delay

49. What is "jitter"?
a) Variable of voice packet
b) Packet delay variation
c) Packet interval time delay variation
d) All of them

50. Which one is a value of Jitter?



a) (1)
b) (2)
c) (3)
d) (4)
e) No correct answer

51. What is a cause of 'jitter'?
a) Router is busy
b) Network is over-load
c) Packet delay is vary
d) Queueing delay
e) All of above

52. Which one is phenomenon of packet loss?
a) The more significant the change in the interarrival time.
b) abrupt rises in jitter value.
c) A large value of end-to-end delay.

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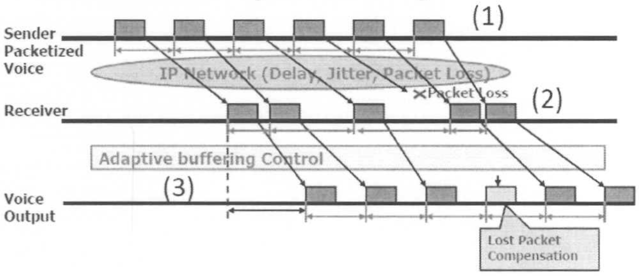
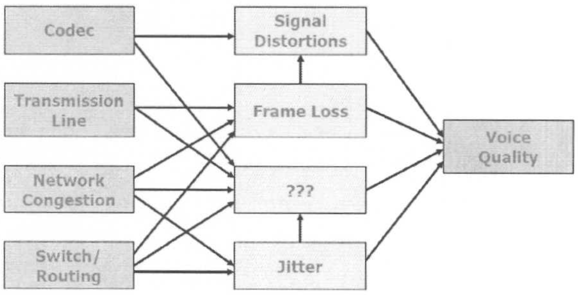
d) All of above.
e) No correct answer

53. What state is jitter occurring?



a) (1)
b) (2)
c) (3)
d) All of them
e) No correct answer

54. What item is missing?



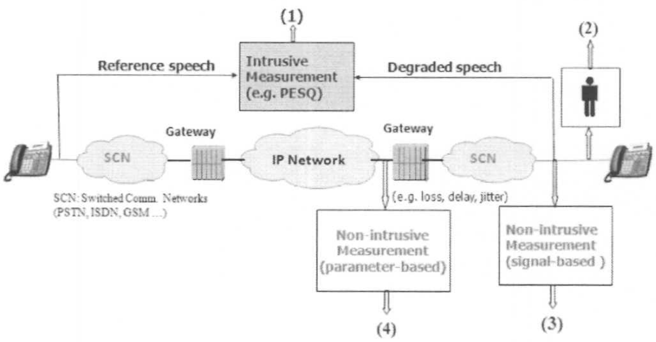
a) Buffering
b) Latency
c) Packet delay
d) Noise
e) No correct answer

55. Which one is NOT voice quality measurement?
a) Mean Opinion Score (MOS).
b) Perceptual Speech Quality Measure (PSQM).
c) Measuring Normalizing Blocks (MNB).
d) Talker Echo Loudness Rating (TELR).
e) E-Model.

56. Which one does it describe for MOS (Mean Opinion Score)?
a) Computes the auditory distance based on how humans psycho-acoustically adjust for certain degradations
a) A computation model for use in transmission planning
b) Listening test conducted by real people
c) Using a speech-like test signal which consists of 30 seconds of male and female phonetic sounds.
d) All of above
e) No correct answer.

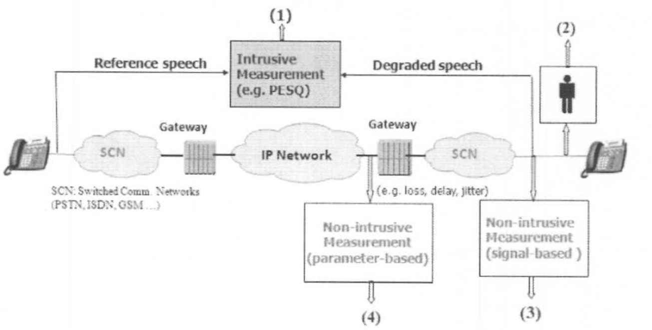
57. Which one is true for MOS and PSQM?
a) MOS and PSQM can be used to accurately measure impairments as a result of voice coding.
b) They can also reflect impairments as a result of frame loss.
c) However, it is more difficult to measure the effect of latency and latency variations using MOS and PSQM alone.
d) MOS and PSQM also do not provide information about the source of the impairment.
e) All of them.

58. What measurement is applied to MOS test?



a) (1)
b) (2)
c) (3)
d) (4)
e) No correct answer

59. What measurement is applied to E-Model?



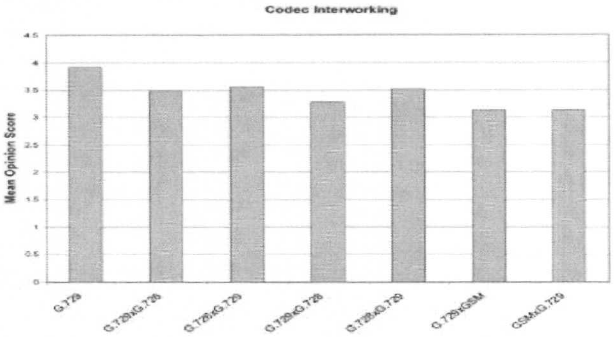
a) (1)
b) (2)

c) (3)
d) (4)
e) No correct answer

60. Which one is a best describe to MOS?
a) Listening test conducted by real people
b) Subjective measure of voice quality
c) Score ranges from 5 to 1
d) Difficult to repeat and time consuming
e) All of them

61. Which one is a main drawback of MOS?
a) it is difficult to measure the effect of latency and latency variations.
b) Difficult to repeat and time consuming.
c) It does not provide information about the source of the impairment.
d) All of above.
e) No correct answer

62. Which one is true for a transcoding?



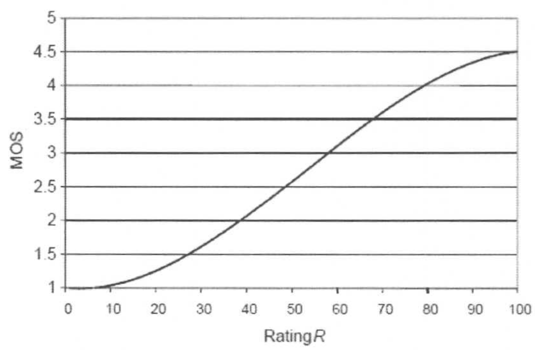
a) Doing a transcoding will improve voice quality.
b) Doing a transcoding will decrease voice quality.
c) Doing a transcoding will increase a packet loss.
d) Doing a transcoding will reduce a bandwidth.
e) No correct answer

63. Which one is NOT true for E-Model
a) A computation model
b) it does not involve any tests.
c) The model predicts the voice quality based on the network configuration and performance metrics.
d) Subjective measure of voice quality.
e) No correct answer.

64. Which one does it describes for E-Model >
a) Computes the auditory distance based on how humans psycho-acoustically adjust for certain degradations
a) A computation model for use in transmission planning
b) Listening test conducted by real peopleฆ
c) Uses a speech-like test signal which consists of 30 seconds of male and female phonetic sounds.
d) All of above
e) No correct answer.

65. Describe the right meaning of
R = Ro - Is - Ie + A
a) Basic signal-to-noise ratio, Impairments which occur simultaneously with voice signal, Impairments caused by delay, Distortion Impairment, Expectation Factor.
b) Basic signal-to-noise ratio, Impairments caused by delay, Impairments which occur simultaneously with voice signal, Distortion Impairment, Expectation Factor.
c) Basic signal-to-noise ratio, Impairments which occur simultaneously with voice signal, Distortion Impairment, Impairments caused by delay, Expectation Factor.
d) Basic signal-to-noise ratio, Distortion Impairment, Impairments which occur simultaneously with voice signal, Impairments caused by delay, Expectation Factor.
e) No correct answer

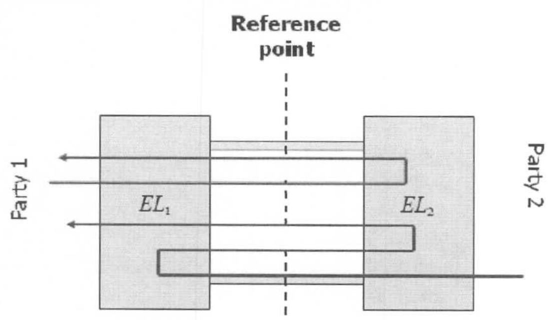
66. What is a minimum score of E-Model that satisfy PSTN voice quality?



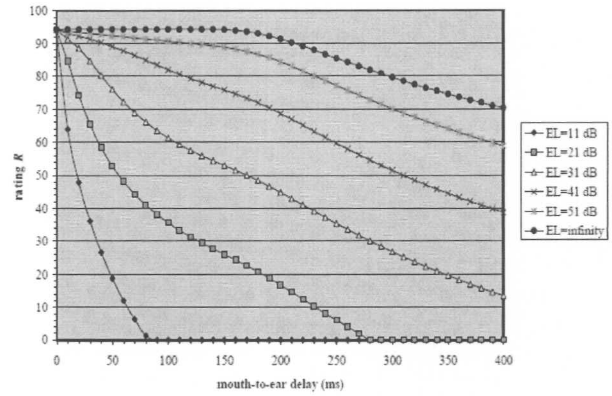
a) 60

b) 70
c) 80
d) 90
e) No correct answer

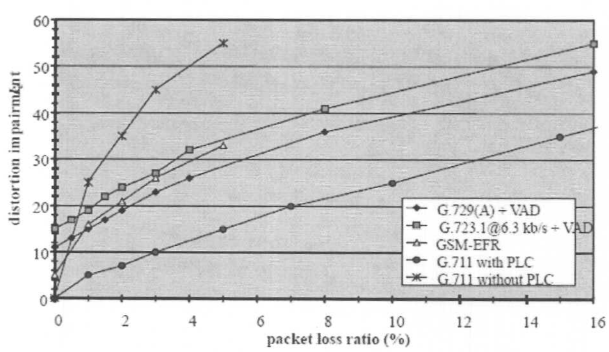
67. Regarding to E-Model calculation, which one is a factor of Is
   a) signal-to-noise ratio
   b) loudness.
   c) Echo
   d) Packet loss
   e) User mobility
68. Regarding to E-Model calculation, which one is a factor of Id
   a) signal-to-noise ratio
   b) loudness.
   c) Echo
   d) Packet loss
   e) User mobility
69. Which one does it give the higher score for the Advantage factor, A?
   a) Wireline telephone.
   b) GSM phone.
   c) 3G phone.
   d) Satellite phone.
   e) No correct answer
70. What is the cause of the below picture?



   a) Talker and listener echo.
   b) Acoustic echo
   c) Voice reflection
   d) End-to-end delay
   e) No correct answer.
71. Which one is associated to Id?
   a) loss of interactivity
   b) talker echo
   c) listener echo
   d) All of them.
   e) No correct answer
72. Which one is the impairment associated with distortion?
   a) VAD (Voice Activity Detection)
   b) Transcoding
   c) Packet loss
   d) All of them
   e) No correct answer
73. If we want R rating = 70, which EL is possible.



   a) EL=51
   b) EL=41
   c) EL=31
   d) EL=21 10
   e) No correct answer
74. Why do we need a transcoding?
   a) Bandwidth mis-match
   b) CODEC change.
   c) To reduce packet loss.
   d) To reduce jitter.
   e) To reduce echo.
75. Which on is TRUE?



   a) G.711 with PLC is the best
   b) G.711 without PLC is better than G.729(A)+VAD
   c) G.729(A)+VAD is worst than GSM-EFR
   d) G.723.1 is better than G.729(A)+VAD

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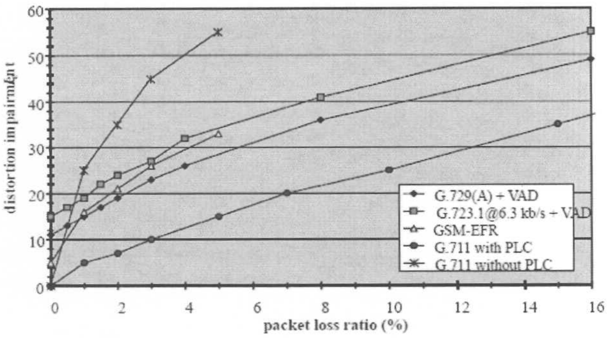
e) No correct answer

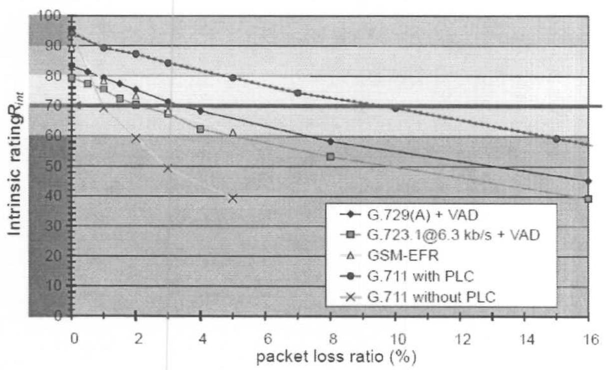
76. Which CODEC is worst when packet loss is 4%?



a) G.711 with PLC.
b) G.729(A)+VAD
c) G.711 wo PLC
d) GSM-EFR
e) No correct answer

77. Make an order from best to worst in MOS score when packet loss is 1%
a) G.711 with PLC, G.729 (VAD), G.711 wo PLC, G.723 (VAD).
b) G.711 with PLC, G.729 (VAD), G.723 (VAD), G.711 wo PLC
c) G.729 (VAD), G.711 with PLC, G.711 wo PLC, G.723 (VAD).
d) G.711 with PLC, G.711 wo PLC, G.729 (VAD), G.723 (VAD).
e) No correct answer

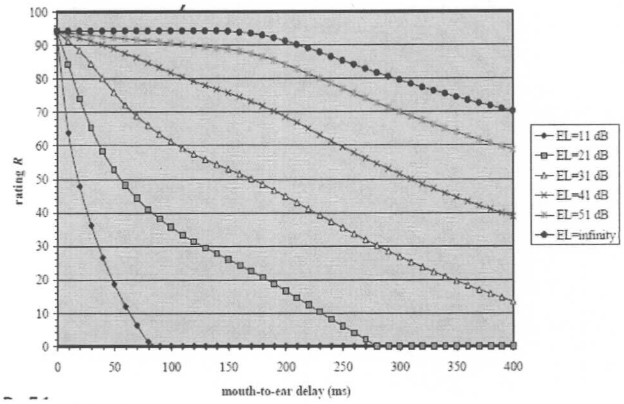
78. What CODEC is accepted when packet loss is 4%?



a) G.729+VAD
b) G.723+VAD
c) GSM-EFR
d) G.711 with PLC
e) G.711 without PLC

79. What is an impairment budget if we use R-factor traditional quality?
a) 30
b) 24
c) 20
d) 14
e) 12

80. If EL=51 db,
Distortion impairment Ie=15, what is R value?



a) R=51
b) R=69
c) R = 76
d) R=81
e) No correct answer

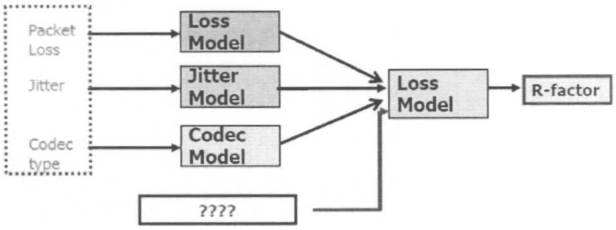
81. What CODEC can tolerate a maximum of month-to-ear delay when packet loss is 0%
a) G.711
b) G.729 (VAD)
c) G.723@6.3 (VAD)
d) GSM-FR
e) GSM-EFR

82. Which parameter does affect distortion of CODEC?
a) level of echo
b) packet loss
c) codec performance
d) All of them
e) No correct answer

83. In R Model for VoIP, Basic signal-to-noise ratio (Ro) is set to 94 (R=94-Id-Ie), why?
a) Because of MOS score comparison.
b) Because of a one-way delay bound between source and destination.
c) Due to a maximum obtainable for G.711.

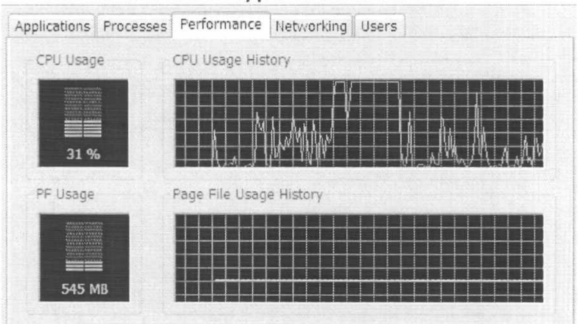
d) Because of loss model maximum value.
e) Because of signal-to-noise impairment factor.

84. What is a parameter p added of E-model for VoIP?



a) Packet loss
b) Delay, measured using RTCP
c) MOS Score
d) VoIP distortion
e) R rating value

85. What is the type of this cause?



a) Type A – constant jitter.
b) Type B – transient jitter.
c) Type C – short term delay variation

รหัสนักศึกษา ....

Part II: อ.สุธน

ข้อที่ 1 SIP FUNDAMENTAL *(30 คะแนน: 30 นาที)*

1.1 จงอธิบายว่าการกระทำที่กำหนดให้ทั้ง 5 ข้อด้านล่าง ทำให้ SIP มีความสามารถพิเศษอย่างไร

(10 คะแนน)

- มีการ response กับ request method ทุกชนิด (โจทย์ตัวอย่าง นศ. ไม่ต้องทำข้อนี้)
    - ตัวอย่างคำตอบ เนื่องจาก response ทำหน้าที่คล้ายการ acknowledge ว่า UAS ได้รับ request method ที่ UAC ส่งมาให้ จึงทำให้ SIP สามารถทำงานร่วมกับ unreliable transport เช่น UDP ได้
- การใช้ AOR ในการอ้างอิงถึงผู้ใช้ แทน FQDN

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- อนุญาตให้ UA หลายตัว ทำการลงทะเบียนด้วย AOR เดียวกัน

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- สามารถกำหนด record-route header ไปกับสัญญาณ INVITE ได้

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- ผู้ส่ง SIP message จะต้องมีการระบุ contact เป็น FQDN

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รหัสนักศึกษา ....

1.2 SIP Message ข้างล่างเป็น SIP Message ซึ่งเป็นส่วนหนึ่งของสัญญาณ INVITE จงตอบคำถามต่อไปนี้ **พร้อมทั้งอธิบายแนวคิด** (15 คะแนน)

```
Via: SIP/2.0/UDP proxy.munich.de:5060;branch=z9hG4bK8542.1
Via: SIP/2.0/UDP 100.101.102.103:5060;branch=z9hG4bK45a35h76
To: Heisenberg <sip:w.heisenberg@munich.de>;tag=24019385
From: E. Schroedinger <sip:schroed5244@aol.com>;tag=312345
Call-ID: 105637921@100.101.102.103
CSeq: 1 INVITE
Contact: sip:wh@200.201.202.203
Content-Type: application/sdp
Content-Length: 173
```

- SIP Message เป็น Request หรือ Response

- การโทรฯครั้งนี้ เป็นการโทรจากใครหาใคร

- สัญญาณนี้ถูกส่งจาก UA หรือ Proxy

- สัญญาณนี้เป็นสัญญาณที่ถูก Retransmit หรือไม่

1.3 หากภาควิชาฯ ต้องการให้บริการ SIP Server เราควรใช้วิธีใดในการแจ้งแอดเดรสของ SIP Server ให้กับ UA (ให้ข้อดีของวิธีที่เลือก และระบุข้อเสียของวิธีที่ไม่ได้เลือก) (5 คะแนน)

Restricted NAT		
Symmetric NAT		

2.3 หากต้องการเพิ่มระดับความปลอดภัยของสัญญาณ SIP ในลักษณะ End-to-End จะกระทำได้ด้วยวิธีใด

รหัสนักศึกษา ....

ข้อที่ 3 SIP Applications *(30 คะแนน: 30 นาที)*

จงออกแบบบริการเสียงเพลงระหว่างรอสาย กล่าวคือ Caller จะได้ยินเสียงเพลงที่ Callee เลือกไว้ในระหว่างรอสาย โดยให้นักศึกษาระบุ Server ต่าง ๆ ที่เกี่ยวข้องอย่างชัดเจน พร้อมทั้งแสดงลำดับการส่งสัญญาณ SIP ให้ครบถ้วน ทั้งนี้นักศึกษาไม่จำเป็นต้องอธิบายวิธีการที่ Callee เลือกเพลงไว้ เพียงแต่ให้ระบุว่าข้อมูลการเลือกเพลงนั้นควรจะเก็บอยู่หน่วยใด และจะนำข้อมูลนั้นไปใช้อย่างไร

ทั้งนี้แผนภาพจะต้องแสดงให้เห็นตั้งแต่ Caller เริ่มส่งสัญญาณ INVITE ไปจน Caller ได้ยินเสียงจาก Callee (หลัง Callee รับสาย)

รหัสนักศึกษา ....

ข้อที่ 2 SIP ARCHITECTURE *(30 คะแนน: 30 นาที)*

2.1 จงอธิบายว่าเหตุใดเราควรใช้ SIP Stateless Proxy ใน Network Core แทน SIP Stateful Proxy

(10 คะแนน)

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2.2 จงระบุว่าวิธีการที่จะช่วยให้ UA สามารถทำงานผ่าน NAT และ Firewall ในกรณีต่าง ๆได้

(10 คะแนน)

Type	**Description**	**Solution**
Full Cone NAT		
Restricted NAT		
Symmetric NAT		

2.3 หากต้องการเพิ่มระดับความปลอดภัยของสัญญาณ SIP ในลักษณะ End-to-End จะกระทำได้ด้วยวิธีใด

(10 คะแนน)

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