Student ID_____Name-Surname_____

PRINCE OF SONGKLA UNIVERSITY

FACULTY OF ENGINEERING

Mid-Term Examination: Semester I Academic Year: 2014

Date: 15 October 2014 Time: 09.00 – 11.00 (2 hrs)

Subject: 242-460 Multimedia Networks Room: A400

ทุจริตในการสอบ โทษขั้นต่ำคือ ปรับตกในรายวิชาที่ทุจริต และพักการเรียน 1 ภาคการศึกษา

Instructions

Part I

- There are 100 questions, attempt to do them all
- Marking scheme
  - 2 marks for the right answer of each answer,
  - -1 (minus one) for each wrong answer
  - No penalty if you leave an empty answer.

Part II

- There are 5 questions, attempt to do all of them.

1. Which one is not associated to PSTN?
   a) Circuit Switch
   b) SS7 (Signal System 7)
   c) PABX
   d) Telephone Trunk
   e) No correct answer
2. What signal is used between trunk groups in PSTN?
   a) SS7
   b) SIP
   c) H.323
   d) TCP/IP
   e) VoIP
3. Which one is not used for VoIP?
   a) Voice CODEC
   b) Circuit switch
   c) TCP/IP
   d) Soft phone
   e) No correct answer
4. What is echo canceller?
   a) Sound level booster
   b) Microphone driver
   c) Voice CODEC
   d) Voice signal feedback canceller
   e) No correct answer
5. What is a voice encoder?
   a) voice compression and decompression
   b) voice transmitter
   c) voice security mechanism
   d) Voice signal feedback canceller
   e) No correct answer

6. Which one is NOT true?
   a) Humans can detect impacts beginning at 125 millisecond.
   b) 0 to 150 ms is Acceptable for most user applications.
   c) no more than 50 ms of one-way processing time is recommended for each of the national systems.
   d) No correct answer.
7. Which one is a cause of echo?
   a) Large end-to-end delay
   b) PSTN hybrid refection
   c) The local microphone picks up the acoustic energy from the output of the loudspeaker.
   d) Impedance matching problem
   e) All of above
8. Which one is NOT a cause of voice quality for PSTN?
   a) Loudness
   b) Delay
   c) Echo
   d) Latency
   e) No correct answer
9. Which one is a cause only effecting to VoIP?
   (a) Loudness
   (b) Delay
   (c) echo
   (d) Latency
   (e) No correct answer
10. What is the meaning of voice clarity?
    a) Reflection of the originating signal at the far
    b) the time a signal needs to traverse the network
    c) It can be described as speech intelligibility
    d) ability to handle non-speech signals
    e) None of above
11. The clarity of PSTN and VoIP have some different factors. Which one is applied to ONLY VoIP?
    a) Noise
    b) Fading
    c) crosstalk
    d) jitter
    e) loudness
12. Which one is the factor that impact voice clarity?
    a) Analog/Digital conversion, Quantization Distortion
    b) Voice Compression: Non-linear Distortion
    c) Packet Loss
    d) Delay, Jitter
    e) All of them
13. What is the cause of 'Talker overlap'?
    a) A large round trip time delay
    b) Packet loss
    c) Echo
    d) Packet jitter
    e) None of above
14. What is Accumulation Delay (or algorithmic 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.
15. What is Processing Delay (or packetise 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.

16. 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.

17. 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.

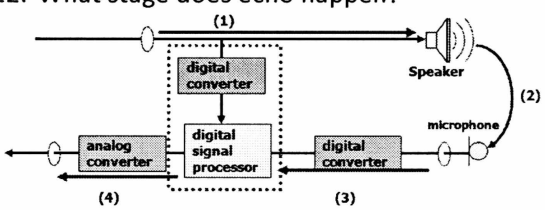
18. Which one is not true for causes of packet loss?
a) Network congestion
b) Time expiry
c) Time-out
d) Buffer over flow
e) No correct answer

19. 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

20. 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.

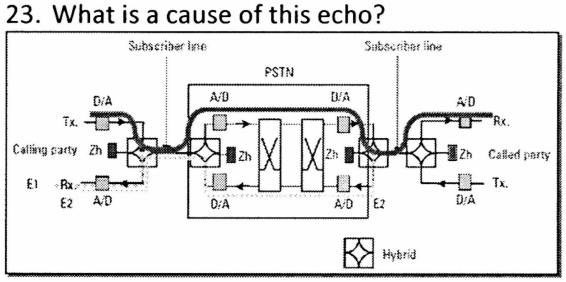
21. What happen when an out of order condition is detected?
a) Out of order packets are played in the order they arrive.
b) Out of order packets are re-ordered and inserted.
c) Out of order packets are dropped.
d) Ask for a re-transmit of these out of order packets.
e) No correct answer

22. What stage does echo happen?



a) (1)
b) (2)
c) (3)
d) (4)

23. What is a cause of this echo?



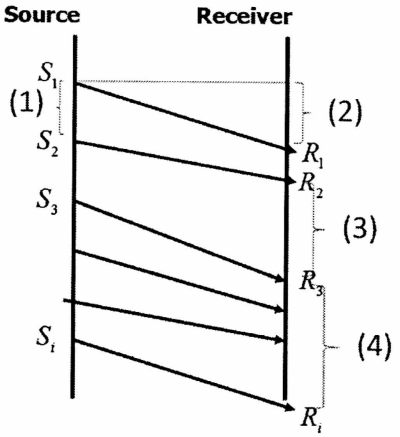
a) Hybrid
b) Long delay
c) A/D and D/A problem
d) Low signal to noise ration

e) No correct answer

24. 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

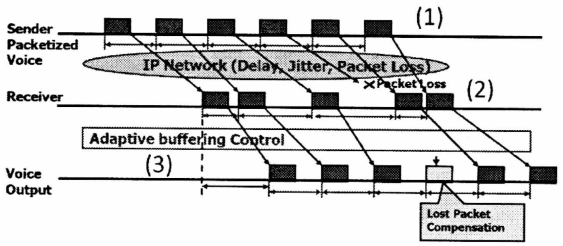
This part is TRUE or FALSE answer (10 questions):

25. local microphone picks up the acoustic energy from the output of the loudspeaker is so called "Acoustic Echo"
    • True
    • False
26. Acoustic echo can be removed by using "echo booster"
    • True
    • false
27. Hybrid is a problem of 2-to-4 wire conversion.
    • True
    • False
28. Hybrid is a problem in IP backbone
    • True
    • False
29. Acoustic echo is not a problem if delay is below 25 msec
    • True
    • False
30. Longer delay creates higher echo level
    • True
    • False
31. Longer delay creates lower clarity
    • True
    • False
32. Higher echo creates lower clarity
    • True
    • False
33. Clarity is the perceptual fidelity, the clearness and the non-distorted nature of voice signals.
    • True
    • False
34. With longer delay, echo must have a lower relative signal level.
    • True
    • False
35. Which one is a value of Jitter?



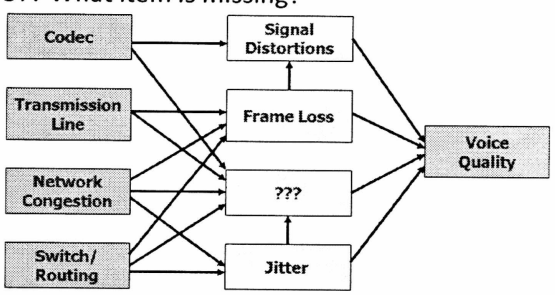
    a) (1)
    b) (2)
    c) (3)
    d) (4)
    e) No correct answer

36. What state is jitter occurring?



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

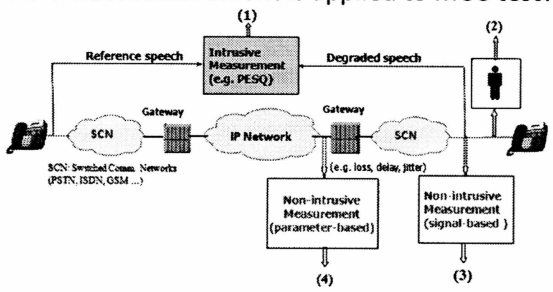
37. What item is missing?



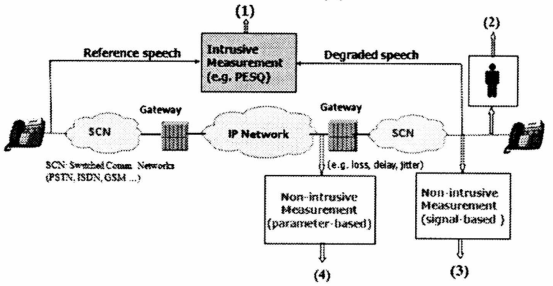
    a) Buffering
    b) Latency
    c) Packet delay
    d) Noise

e) No correct answer

38. 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
39. 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
40. 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
41. 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.
42. 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
    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.
43. 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.
44. What measurement is applied to MOS test?



    a) (1)
    b) (2)
    c) (3)
    d) (4)
    e) No correct answer
45. What measurement is applied to E-Model?

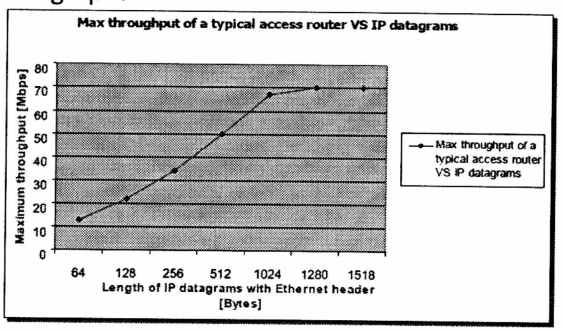


    a) (1)
    b) (2)
    c) (3)
    d) (4)
    e) No correct answer
46. If we would like to increase a number of voice channels, what techniques can be used.
    a) Using voice codec
    b) Using voice multiplexing
    c) Increasing a play load size
    d) Increasing a packetise time
    e) All of above
47. Which one is not a source of fixed delay
    a) Algorithmic Delay
    b) Serialization Delay
    c) Propagation Delay

d) Component Delay
e) Network delay

48. 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

49. 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.

50. 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

51. 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.

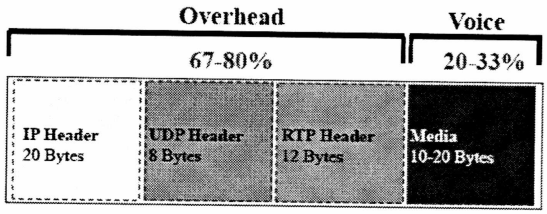
52. From the table below, which one offers the highest voice utilisation?

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

53. 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

54. 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

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

Origin	standard	Codec bit rate (kb/s)	Month-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

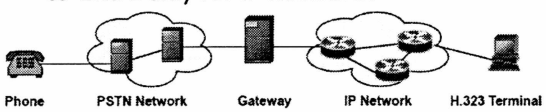
b) G.728
c) G.729
d) GSM-FR
e) GSM-EFR

56. 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

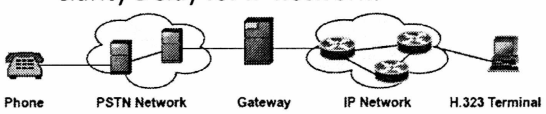
a) G.711 without PLC
b) G.711 with PLC
c) G.729(A) + VAD
d) G.723 +VAD
e) GSM-EFR

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



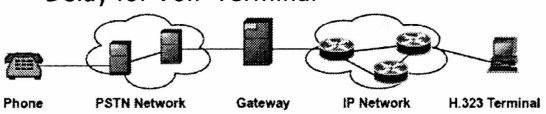
a) Fixed transmission time
b) Voice signal processing, Receive Jitter buffering
c) Buffering, Queuing
d) Transmit - packetization
e) All of above

58. 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

59. 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

60. 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

61. 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)

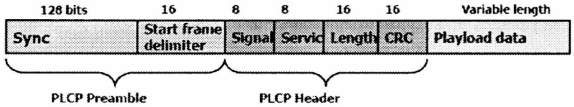
a) 24 kbps
b) 40 kbps
c) 46 kbps
d) 80 kbps
e) 96 kb

62. 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)

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

63. 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

64. 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.

65. 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
66. 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
67. 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
68. What is a delay boundary (for one-way delay) echo cancellation is required?
    a) 15 ms
    b) 25 ms
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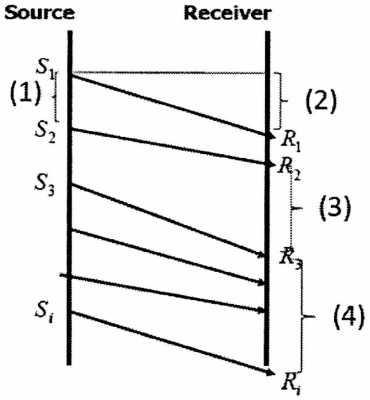
    d) Component Delay
    e) Network delay
70. What is "jitter"?
    a) Variable of voice packet
    b) Packet delay variation
    c) Packet interval time delay variation
    d) All of them
71. Which one is a value of Jitter?



    a) (1)
    b) (2)
    c) (3)
    d) (4)
    e) No correct answer
72. What is a cause of 'jitter'?
    a) Router is busy
    b) Network is over-load
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73. Which one is phenomenon of packet loss?
    a) The more significant the change in the interarrival time.
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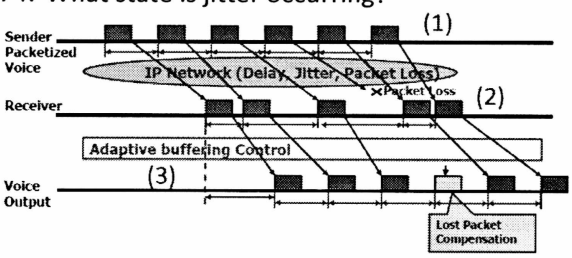
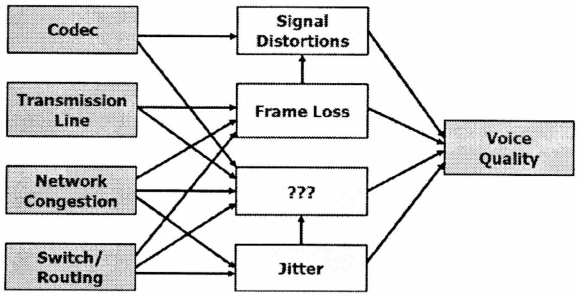
74. What state is jitter occurring?



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



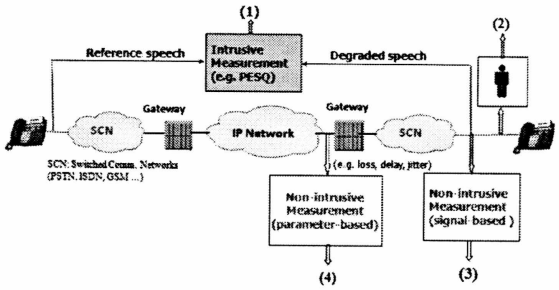
a) Buffering
b) Latency
c) Packet delay
d) Noise
e) No correct answer

76. 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.

77. 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.

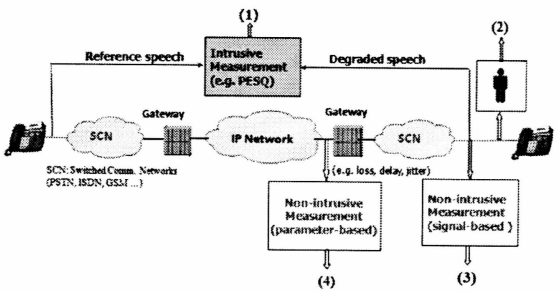
78. Which one is true for MOS and PSQM?
a) MOS and PSQM can be used to accurately measure impairments as a result of voice coding.
b) They can also reflect impairments as a result of frame loss.
c) However, it is more difficult to measure the effect of latency and latency variations using MOS and PSQM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

79. What measurement is applied to MOS test?



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

80. What measurement is applied to E-Model?



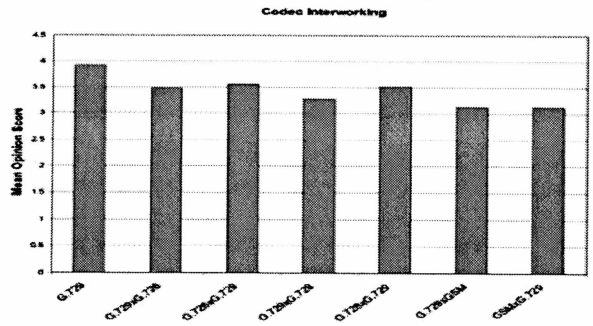
a) (1)
b) (2)
c) (3)
d) (4)
e) No correct answer

81. 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

82. 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

83. 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

84. 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.

85. 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 people
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.

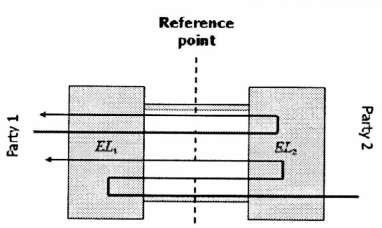
86. Describe the right meaning of
R = Ro - Is - Ie + A
a) Basic signal-to-noise ratio, Impairments which occur simultaneously with voice signal, Impairments caused by delay, Distortion Impairment, Expectation Factor.
b) Basic signal-to-noise ratio, Impairments caused by delay, Impairments which occur 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

87. 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

88. 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

89. 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

90. What is the cause of the below picture?

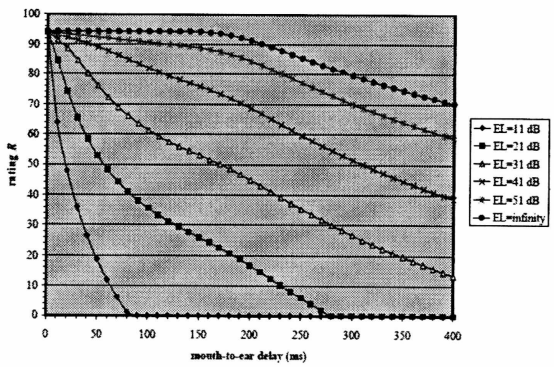


a) Talker and listener echo.
b) Acoustic echo
c) Voice reflection
d) End-to-end delay
e) No correct answer.

91. Which one is associated to Id?
a) loss of interactivity
b) talker echo
c) listener echo
d) All of them.
e) No correct answer

92. 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

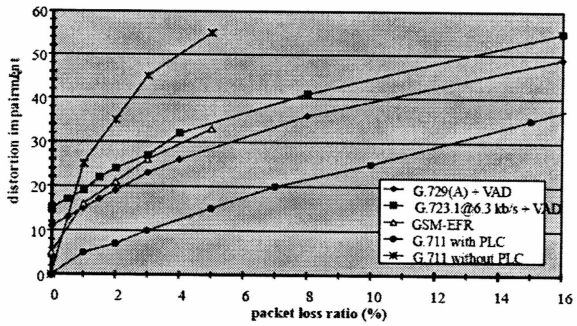
93. If we want R rating = 70, which EL is possible.



a) EL=51
b) EL=41
c) EL=31
d) EL=21 10
e) No correct answer

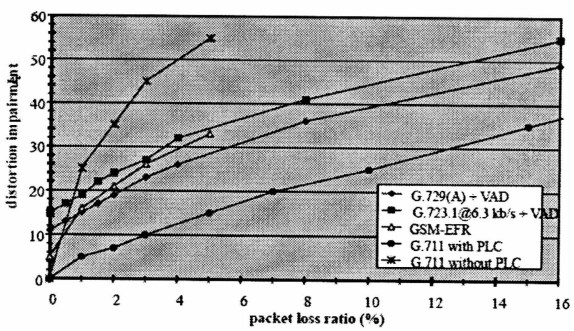
94. Why do we need a transcoding?
a) Bandwidth mis-match
b) CODEC change.
c) To reduce packet loss.
d) To reduce jitter.
e) To reduce echo.

95. 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
e) No correct answer

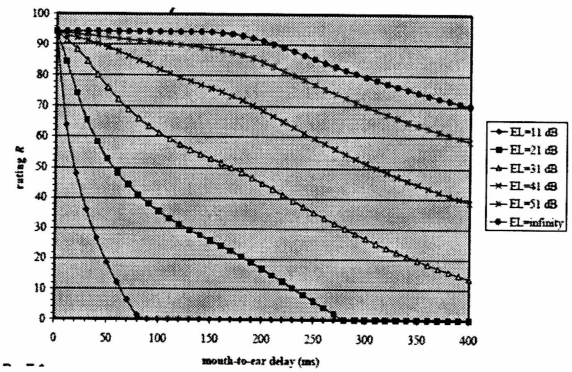
96. Which CODEC is worst when packet loss is 4%?



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

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

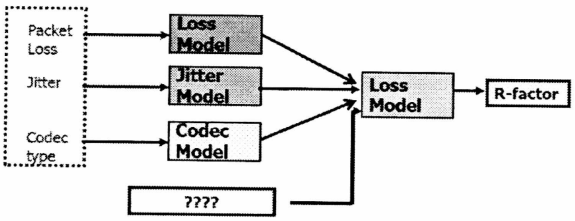
98. If EL=51 db,
Distortion impairment Ie=15, what is R value?



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

99. 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

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



a) Packet loss
b) Delay, measured using RTCP
c) End-to-end delay
d) VoIP distortion
e) R rating value

PART II: Video Communication

1. What is 1) spatial samples, 2) Temporal samples? In video capture (10 marks)

Answer

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2. For MPEG 4 video, please explain the following questions:
   2.1 the encoded frames: I-Frame, P-Frame, and B-Frame, (10 marks)
   2.2 Draw the frame sequence and its intra and/or Inter-relationship. (10 marks)

   Answer
   2.1

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3. Please explain some main features of High Efficiency Video Coding (HEVC), at least 4 of them (10 marks)

Answer

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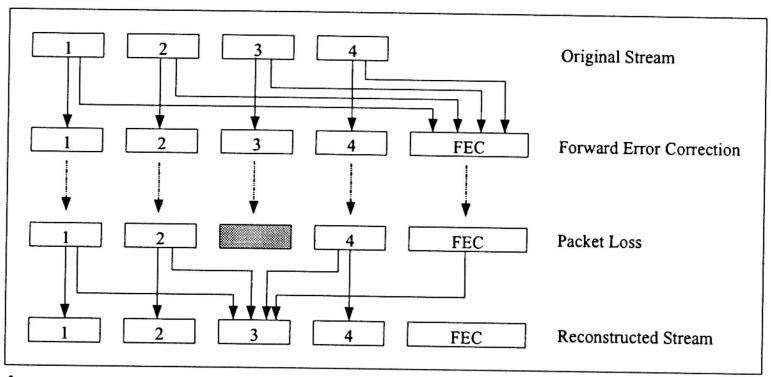
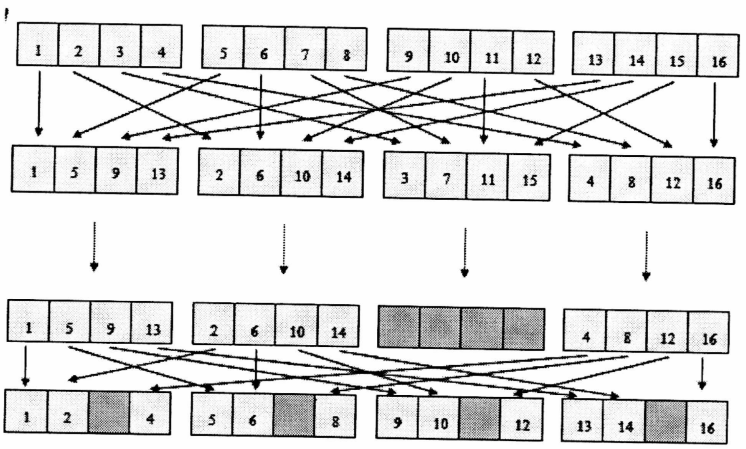
4. Please explain how FEC (Forward Error Collection) in video communications works (give an example of the scheme below), what are the benefits/advantages, and dis-advantages. (15 marks)



Answer

5. The scheme below is one of mechanisms of sender based repairing of video communication. Please explain of this scheme works. What are the benefits and drawbacks of this scheme? (15 marks)



Answer