Prof. Noronha
Spring 2001

EE384B: Multimedia Networking and Communications

Quiz #1
Closed Book
Time: 30 minutes

April 19, 2001

Score: ___ / 36

Name:____

E-mail:_______________________________________

☐ SITN student ☐ On-campus student

I agree to abide by the Stanford Honor Code:____
(signature)

Question 1: Audio Compression Techniques (5 points)

There are several methods for coding and compressing voice (PCM, ADPCM, CELP). The MPEG standards also address audio compression, using completely different techniques. Why are the techniques used in MPEG audio compression so different from the voice compression techniques?

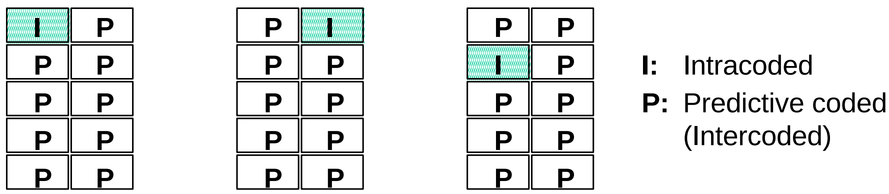
Question 2: Latency in Video Encoding (5 points)

In general, when encoding video, there is a tradeoff between latency and quality. Why? Explain why latency suffers if you want the best possible quality, and why you need to give up quality to get low latency.

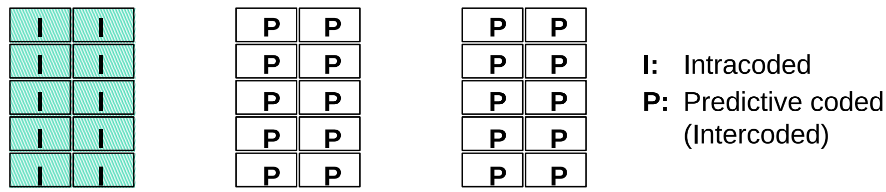
Question 3: Latency in H.261 (8 points)

The following two options are available in H.261:

Technique 1: Rotational intracoding of GOBs:



Technique 2: One completely intracoded picture followed by a number of predictive pictures:



a) Technique 1 above is used to reduce latency, as compared with technique 2. Why does it reduce latency? (4 points)

b) Assume that the video encoded using techniques 1 and 2 is transmitted over a channel with bit errors. Qualitatively compare the two techniques in regards to glitch duration. For technique 2, consider that the number of P frames between I frames is a design value, and discuss the effect it has on the glitch duration. (4 points)

Question 4: Delay Jitter in MPEG (5 points)

Explain what is the effect of delay jitter in a live MPEG transmission system (i.e., a system where the output of a real-time encoder is being transmitted through a network to a real-time decoder).

Question 5: Encoder/Decoder Synchronization (5 points)

In a live MPEG transmission system (a real-time encoder transmitting to a real-time decoder), why is it necessary to synchronize the encoder and the decoder? Briefly explain two synchronization techniques.

Question 6: Scalable Video (8 points)

Using MPEG-4, it is possible to generate *scalable video*. A scalable video stream has a *base layer*, which provides a certain minimum quality, and a number of *enhancement layers*. Combining the base layer with the enhancement layers provides increasing video quality. Consider a scenario where MPEG-4 is being used in a live point-to-multipoint video distribution system, where there are heterogeneous receivers. Receivers vary in their connection speed to the network; some have low speed links, some have very fast connections, and others have available something in between. The backbone links between the receivers and the video source also vary. Explain how you would use the following network features to get the best possible video quality for each receiver:

- IP Multicast.
- IP Precedence.

Justify any statement you make.

BONUS QUESTION (Extra Credit): (4 extra points)

When coding P or B macroblocks, the encoder finds the best match block in the reference frame, computes the difference between that block and the block being coded, and takes the DCT of this difference. Alternatively, it could take the DCT of the reference block and the block being coded, and compute the difference in the frequency domain by subtracting the coefficients. What is the effect of doing this? Is there any advantage of doing it one way or another? Justify.

Since this is a closed-book test, here is the DCT equation in case you need it:

$$S_{vu} = \frac{1}{\sqrt{N1N2}} c_u v_u \sum_{x=0}^{N1-1} \sum_{y=0}^{N2-1} s_{yx} \cos\frac{(2x+1)u\pi}{2N1} \cos\frac{(2y+1)v\pi}{2N2}$$

$$c_u, v_u = \frac{1}{\sqrt{2}}, u = 0, v = 0$$

$$c_u, v_u = 1, else$$