BENC 4173: MULTIMEDIA TECHNOLOGY & APPLICATION
TUTORIAL CHAPTER 4: DIGITAL AUDIO

1. If a 22000 Hz sound wave is sampled at a rate of 16000 samples/second, what will be the frequency of the resulting digitized sound wave?
   a. 6000 Hz
   b. 8000 Hz
   c. 10000 Hz
   d. 16000 Hz

2. What is the signal-to-quantization noise ratio of an audio file stored with 16 bits per sample?
   a. 8 decibels
   b. 96 decibels
   c. 32 decibels
   d. 48 decibels

3. Convert 20 N/m2 (approximately the air pressure level of very loud music) to dB_SPL. Show your work.

4. Audio dithering is applied so that
   a. low amplitude samples are quantized more coarsely and high amplitude samples are quantized more finely
   b. there are fewer breaks and "jumps" in the audio caused by quantization error
   c. the quantization error can be pulled up to a higher frequency and filtered out
   d. the sampling rate for lower amplitude samples can be lowered

5. Non-linear quantization
   a. allocates fewer bits per sample to low amplitude samples
   b. allocates more bits per sample to low amplitude samples
   c. quantizes high amplitude samples more finely (i.e, with more detail) than low amplitude ones.
   d quantizes low amplitude samples more finely (i.e, with more detail) than high amplitude ones.

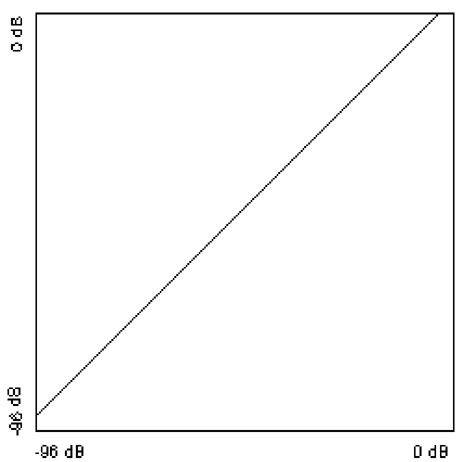
6. Say that you want to normalize a 16-bit audio file that has a maximum amplitude of -40 dBFS and a minimum amplitude of -80 dBFS. You want to use the maximum amplitude possible. After normalization, what will be the value of samples that were originally -50 dBFS?

a. about -10 dBFS
b. about -25 dBFS
c. about -40 dBFS
d. about -66 dBFS

7. The compression ratio for MPEG-I audio is reported as 4:1 with a corresponding data rate of 384 kb/sec. The compression rate for MPEG-II audio is reported as about 8:1 with a corresponding data rate of 192 kb/sec. Why does a higher compression ratio imply a lower data rate?

a. Because a higher compression rate implies that the data moves at a higher data rate through its transmission medium.
b. Because a higher compression rate yields less data, and therefore less audio data needs to be transmitted per second when the audio file plays.
c. These numbers don't demonstrate any relationship between compression rate and data rate. There is no such relationship. MPEG-II just made improvements to both.
d. Because a higher compression rate yields more data after compression, and therefore you have bigger files.

8. What does the following transfer function imply in terms of dynamics processing?



a. The audio signal is expanded.
b. The audio signal is attentuated by a constant amount.
c. The audio signal is boosted by a constant amount.
d. The function is linear so the amplitudes of the audio signal remain the same.

9. How can masking tones be used for compression?

a. The amplitude of the signal is boosted so that it is louder than the quantization error.
b. For each band of frequencies, the frequencies that are masked by the masking tones in that band can be filtered out.
c. Both a and b are true.
d. Neither a nor b is true.

10. How does mu-law encoding work?

a. A 16-bit audio file is compressed into 8 bits per sample, but when it is decompressed to be played, you get back the equivalent of 12 bits of dynamic range.
b. Mu-law encoding uses non-linear quantization to reduce the percentage quantization error for low amplitude samples.
c. Both a and b are true.
d. Neither a nor b is true.

11. Assume we want to transmit the audio signal with frequency range from 200 Hz to 3 kHz through telephone line.

(i) If we want to transmit the signal at 12 bit per sample, estimate the minimum bandwidth required for the transmission.

(ii) By using non-linear quantization, we can reduce the data size, say at rate 8 bps and at the same time still maintain the perceived level of audio quality. Explain how it is possible.

(iii) If we transmit the data at rate 4000 sample/s, it will cause the aliasing at the receiver. Explain the meaning of aliasing and calculate the aliasing frequency.

12. Describe in detail noise shaping. How it help to compensate quantization error?

13.

(a) Analog-to-digital conversion requires two steps: sampling and quantization.

(i) Explain what is quantization and quantization noise (using illustrations if necessary).

(ii) Describe how audio dithering is able to compensate for quantization error. *Hint: Think about the advantages of the method.*

(b) Assume you have 12-bit audio samples. You are going to transmit the signal at a bit depth of 6, and expand it back to 12 bits at the receiving end. Given an initial 12-bit sample of -600, apply nonlinear companding method to determine the sample value at the receiving end. Compute the error in percentage.

14. Describe the psychoacoustics phenomena of frequency masking.

15. In PCM, what is the delay, assuming 8kHz sampling? Generally, delay is the time penalty associated with any algorithm due to sampling, processing and analysis.