Study Material for Midterm
Introduction to Computer Music 15-322 Fall 2017

Some general things to study:

- You should know how a table lookup oscillator works

- You should expect to write some Nyquist expressions or explain some Nyquist code, but you will not have to memorize functions and their parameters (they will be provided if you need them).

- You should know basic sampling theory and understand the effects of limited sample rates and limited precision (quantization). You should know the purpose of a pre-filter (anti-aliasing filter) and a post-filter (reconstruction filter).

- You should know how FM synthesis works, e.g. how to generate a certain set of frequencies, or alternatively, what frequencies are generated, given carrier and modulator frequencies and the modulation depth.

- You should review your notes on music listening examples from class meetings

(The following material is cut and pasted from previous semester homework and quizzes. You should be able to answer these questions. If not, you should brush up on the related concepts to prepare for the midterm exam.)

1. What causes "clipping" in a computer-generated sound?
A. The sound has a frequency that is outside the range of human hearing.
B. The sound has an amplitude that exceeds the maximum value the computer can store.
C. The sound has too many overtones.
D. The sound takes up too much memory.

Answer:___________________

2. From the reading: What hardware device converts the binary representation of samples used in a computer to the analog electrical signal that is sent to the loudspeaker?
A. A “Binary To Voltage Converter” (BVC)
B. A “Digital To Analog Converter” (DAC)
C. A “Computer To Speaker Converter” (CSC)
D. A “Circuit To Speaker Converter” (CTSC)

Answer:______________

3. From the reading: The human range of hearing in the frequency domain extends up to
A. 22,000 Hz
B. 48,000 Hz
C. 76,000 Hz
D. 90,000 Hz

Answer:_______________

4. In an oscillator table lookup, if the phase offset has exceeded the table length, it is modified and used for a valid table lookup by

A: Dividing by two.
B: Dividing by ten.
C: Setting to zero, to go back to the beginning of the table.
D. Subtracting the table length.

Answer:________________

5. Which of the following will most affect timbre while leaving pitch and loudness relatively unchanged?

A. Phase shift by pi/2 (a quarter-cycle).
B. Multiply amplitude by 2.
C. Change wave from a sine wave to a sawtooth wave.
D. Double the frequency of the wave.

Answer:_________________

6. Sometimes the phase offset in an oscillator table lookup is not an integer value, and therefore there is no defined lookup value that should be returned. Which of the following methods approximates lookup values that will produce the best sound?

A. Round the phase offset up to the next highest integer, and use the lookup value associated with that.
B. Round the phase offset up or down to the nearest integer, and use the lookup value associated with that.
C. Interpolate a return value by computing a weighted average of the lookup values immediately associated with the integers immediately above and below the phase offset.
D. Return zero, because non-integer phase offset values are so rare that a zero sample won't affect overall sound quality.

Answer:__________________

7. How is the Fast Fourier Transform (FFT) different from the Discrete Fourier Transform (DFT)?

a. The FFT works on continuous data, the DFT on discrete data
b. The FFT transforms frequency to time, while the DFT converts time to frequency
c. The FFT sacrifices accuracy for speed of computation, while the DFT is slow but always accurate
d. The FFT is an O(n log n) algorithm for computing the DFT, which is a mathematical function

Answer: ____________________

8. Which of these is the best example of a sound with frequency modulation?

a. A trombone glissando
b. Repeated knocking on a door
c. A major scale played on a piano
d. The tone you hear during a hearing test

Answer: ____________________

9. Imagine an A5 (880 Hz) is played on a violin with 3Hz vibrato. Which of the following is true, according to the definitions of carrier and modulation frequency?

a. The carrier frequency is 3Hz, the modulation frequency is 880Hz
b. The carrier frequency is 880Hz, the modulation frequency is 3Hz
c. The carrier frequency is 3Hz, the modulation frequency cannot be determined from the information given
d. The carrier frequency is 880Hz, the modulation frequency cannot be determined from the information given

Answer: ______________________

10. Of the four basic waveform parameters, in which one will a change be least audible?

a. Frequency
b. Amplitude
c. Shape (sine, square, sawtooth, etc)
d. Phase

Answer: ______________________________

11. Vocabulary from class and the reading: Frequencies in a spectrum known as “partials” differ from frequency components known as “harmonics” in that:

a. Harmonics are of equal amplitude to the fundamental frequency, partials are of some fractional amplitude of the fundamental frequency.
b. Harmonics are frequencies of some integer multiple of the fundamental frequency, partials are frequency components of a sound with no specific mathematical relation to any fundamental.
c. Harmonics are in phase with the fundamental, partials are not.
d. There is no significant difference, the terms are interchangeable.

Answer: _____________________________

12. The highest frequency humans can hear is about 20kHz. Which of the following sampling frequencies would give near perfect sound reconstruction for human ears, no matter what the original sound?

a. 10kHz
b. 20kHz
c. 30kHz
d. 40kHz

Answer: ____________________

13. Which of the following constitute minimal conditions for discrete samples to represent a continuous signal without loss?

a. The signal is band-limited
b. The sampling is perfect
c. The signal is band-limited and the sampling is perfect
d. None - it is not possible for discrete samples to represent a continuous signal without loss

Answer: ____________________

14. Dithering is performed so that
a. A sample sounds sharper and crisper
b. The SNR (Signal-To-Noise Ratio) of a sample is reduced (as in a Dolby Noise Reduction system)
c. A sample doesn’t become overloaded with inaudible frequency components.
d. A sample doesn’t become corrupted by artifacts brought about by signal components very near the quantization level.

Answer: ____________________

15. What is the dynamic range (in dB) of a digital system?

a. (number of bits per sample) * 6.11
b. (sample frequency) * 12.6 / (threshold of hearing)
c. (number of bits per sample) *12.6 * (sample frequency) / (threshold of hearing)
d. (number of bits per sample)2 * 6.11

Answer: ___________________

16. The composition “Krib” by the group *Autechre* demonstrates which of the following techniques?

A) Using a vocoder to impress the formants of a human voice onto a synthesized sound

B) Intentionally reducing the sampling rate of audio files to create aliasing
C) Using recordings of a physical object to create complex rhythm patterns
D) Reversing human speech to create ghostly effects

1. ADC stands for: ____

2. Why is ADC usually preceded by low-pass filtering?

3. If we sample a 15 kHz sinusoid using a sample rate of 22050 Hz, the result will be a digitized sinusoid at a new frequency. What is this phenomenon called?

4. If I want to encode speech containing frequencies up to 8kHz, what is the theoretical minimum sampling rate I need to use?

5. If I sample the speech using 8 bits per sample (with PCM = linear encoding), approximately what is the best signal-to-noise ratio I can achieve?

6. What's the advantage of an interpolating oscillator over a non-interpolating oscillator?

7. What is a *Unit Generator*? (In one or two sentences.)

8. Roads says that we can hear a difference in amplitude of somewhat less than 1 dB. What linear scale factor corresponds to a decrease of 1 dB. (Hint: the scale factor corresponds to a change in amplitude. Roads gives the formula
   `dB = 10 × log10(level/reference level)`
   but keep in mind that "level" here refers to power. Power is proportional to the *square* of the amplitude.)

9. What is the purpose of *dither*?

10. What is the memory requirement for CD audio (44.1kHz, 16-bit, stereo) in megabytes per minute?

11. Sketch the magnitude (amplitude) spectrum for a single 800Hz sinusoid. Label the axes.

12. Sketch the magnitude spectrum for the sum of an 800Hz sinusoid and a 200Hz sinusoid.

13. Sketch the magnitude spectrum for the product of an 800Hz sinusoid and a 200Hz sinusoid.

14. Sketch the magnitude spectrum for an 800Hz sinusoid frequency modulated by a 200Hz sinusoid. Assume that there are only 4 “sidebands” generated, 2 above, and 2 below the

carrier frequency.

15. Sketch the magnitude spectrum for an 800Hz sinusoid frequency modulated by a 200Hz sinusoid (same as above), assuming that there are 10 “sidebands” generated, 5 above, and 5 below the carrier frequency. Show phase-inverted reflected components as downward lines as in Figure 6.12 in Roads.

16. Here’s the equation for problems 14 and 15:

    $f(t) = A \times \sin(2\pi tC + B \times \sin(2\pi tM))$,

    where *A, B, C,* and *M* are parameters. For the previous questions which one of these parameters is 800? ______ Which one is 200? _________ To increase the number of sidebands from 4 to 10, parameter ________ should increase from about _________ to about _________.

17. Clarinets are known to have strong odd harmonics, i.e. their spectrum consists of partials with frequencies f, $3f$, $5f$, ... A crude clarinet-like tone can be obtained with a simple FM configuration. In terms of A, B, C, and M from problem 6, what should you do to generate odd harmonics?

18. What are the two functions that invoke sequential behavior and simultaneous behavior respectively?

19. What are tremolo and vibrato?

20. Assume that the frequency of a carrier signal is 1200 Hz and the frequency of a modulation signal is 700 Hz. What are the frequencies of two sidebands in Ring Modulation?

21. When will FM generate inharmonic spectra?

22. As a rule of thumb, what is the number of significant sideband pairs in FM?

23. What is the relationship between modulation index and amplitude? Will amplitude vary a lot if we change modulation index?

24. What is modulation?