

COSC 4P98 TERM TEST

Instructor: B. Ross

Date: November 17, 2017

NAME (print): _____

STUDENT NUMBER: _____

A 3" by 5" card of hand-written (not photocopied or digitally reproduced) notes may be used.

A calculator may be used. No other aids are permitted.

Please answer all questions on the exam paper. Use the backs of pages if necessary.

Keep written answers brief and to the point. Write neatly and legibly.

Carefully read all questions!

Question	Total	Mark
1	16	
2	24	
3	15	
4	20	
5	22	
6	12	
7	1	
Total:	110	

Question 1 [16] Matching: Write the letter of the most appropriate answer.
[no penalty for wrong answers]

1. _____ Used by musicians to create both MIDI and samples for performances.
2. _____ Remove grain distortion with this.
3. _____ A more “algorithmic” music language than Csound.
4. _____ This can integrate with MIDI to control stage lighting and effects.
5. _____ The synthesis technique used by old analog Moog synths.
6. _____ Contemporary programming framework popular for music applications.
7. _____ A standard plug-in format, defined via an SDK.
8. _____ Suitable for real-time audio processing.
9. _____ Don’t try sampling about this frequency!
10. _____ A serial protocol for musical events.
11. _____ This can give snapshots of the harmonic spectra for real-time audio.
12. _____ Synthesis technique where carrier multiplied by unipolar signal.
13. _____ Term that means you are repeatedly playing a sample over and over.
14. _____ A wave that is an integral multiple of another.
15. _____ A form of aliasing, akin to pixelization in bitmaps.
16. _____ Useful as a generator of music-like note sequences.

- | | |
|---------------------------|--------------------------------------|
| A. Harmonic | L. DMX |
| B. DAW | M. MIDI |
| C. Frequency modulation | N. Wave terrain |
| D. Fast Fourier Transform | O. Supercollider |
| E. 8000 KHz | P. 100 decibels |
| F. Pink noise | Q. Short-time Fourier Transformation |
| G. VST | R. Amplitude modulation |
| H. JUCE | S. visualization |
| I. Subtractive synthesis | T. Nyquist |
| J. Amplitude envelope | U. White noise |
| K. Quantization error | V. Looping |

Question 2 [24] Multiple choice: select the **best** answer to each question

[+2 correct, 0 nothing, -0.5 wrong guess; one free wrong guess permitted without penalty]

1. The decibel scale is:

- a. $O(N^2)$
- b. logarithmic
- c. uses complex numbers
- d. between 20 Hz to 20 kHz

2. Csound might be best described as:

- a. a music programming library callable in C programs
- b. an audio processing dataflow language
- c. a visual programming environment
- d. appropriate when your composition is in the key of C

3. Which of the following is not a standard MIDI port?

- a. IN
- b. THRU
- c. CLOCK
- d. OUT

4. A harmonic frequency associated with human speech is called a:

- a. vocoder
- b. autotune
- c. fundamental
- d. formant

5. CSound's medium data rate (k-rate) is recommended for altering variables that denote:

- a. audio data
- b. envelopes
- c. notes
- d. MIDI

6. VST stands for:

- a. Virtual Synthesizer Toolkit
- b. Voltage-controlled Sequencer Tracks
- c. Virtual Studio Technology
- d. VCO Sample Table

7. You are playing music in your car. You decide to double the volume. You double it again. Then, to impress Europe, you double it again. The overall volume increase is approximately:

- a. 3 decibels
- b. 6 decibels
- c. 8 decibels
- d. 18 decibels

8. This can be used to remove noise and clicks in recorded audio:

- a. low-pass filter
- b. envelope the grains
- c. Nyquist feedback
- d. high-pass filter

9. CSound has different rates for processing variable changes. The reason that the fastest (audio sample rate) is not used by default is:

- a. it would munchkin-ize all the instruments
- b. it would require too much memory
- c. it would be needlessly inefficient for low-rate changes such as notes
- d. none of the above

10. Ring modulation involves:

- a. adding two unipolar signals
- b. enveloping a modulator with a ringer
- c. multiplying two bipolar signals
- d. computing and playing a moving average of sample sequences

11. One of the two methods by which audio can be altered in VST plug-in implementations:

- a. processReplacing
- b. feedback
- c. quantize
- d. audioGain

12. Suffering succotash! The professor is fixated on this project idea:

- a. Raspberry Pi-based granular synthesis with mappable hardware control
- b. autotuning everything in existence
- c. any song in the key of B-flat
- d. piano keyboards with exactly 43 keys

Question 3 [15] Define each term. Include a brief discussion of each as well.

a) stochastic music

b) the 0th harmonic in a DFT

c) oscillator

d) circular buffer

e) grain-size parameter in granular synthesis

Question 4 [20]

a) [12] Draw a diagram showing the complete processing for sampling and playback of digital audio. Include a descriptive discussion of all the major steps in the diagram.

b) [4] Why does the sampling rate have to be at least twice the frequency of the highest frequency to be recorded? Illustrate what happens if it is not. What term is used to describe the undesirable effect?

c) [4] Audio equipment with high sampling rates of 192 kHz and higher are controversial. Briefly discuss 2 points against the usefulness of such high sampling rates.

Question 5 [22] Consider the discrete Fourier transformation formula:

$$f(t) = \sum_{n=1}^{T/2} \left(a_n \cos\left(\frac{2\pi nt}{T}\right) + b_n \sin\left(\frac{2\pi nt}{T}\right) \right) + a_0$$

$$a_0 = \sum_{k=0}^{T-1} \frac{x[k]}{T}$$

$$a_n = \left(\sum_{k=0}^{T-1} x[k] \cos\left(\frac{2\pi nk}{T}\right) \right) \cdot \left(\frac{2}{T}\right)$$

$$b_n = \left(\sum_{k=0}^{T-1} x[k] \sin\left(\frac{2\pi nk}{T}\right) \right) \cdot \left(\frac{2}{T}\right)$$

a) [6] Define and briefly describe the following terms:

t:

T:

x[k]:

n:

b) [4] The Fourier transformation is said to work in the frequency domain. Explain what this means.

c) [4] The Fourier series is characterized as the sum of sine waves. Why does the above use sine and cosine waves instead? How do the a and b coefficients relate to this?

d) [4] What is the technique used for applying Fourier transformations to long, continuous signals. Briefly explain how it works.

f) [2] Why do we not need to compute more than $T/2$ harmonics

Question 6 [12]

a) [8] Define granular synthesis. Describe how it works with a sample table.

b) [4] List 4 parameters that can be defined to control aspects of granular synthesis.

Question 7 [1] Briefly describe what you are thinking of doing for the course project.