UNIVERSITY OF LONDON

GOLDSMITHS COLLEGE

B. Sc. Examination 2012

Department of Computing

IS51012C/16A/17A Audio-Visual Information Systems, Audio-Visual Computing, Creative Audio-Visual Computing

Duration: 2 hour 15 minutes

There are five questions in this paper. You should answer no more than THREE questions. Full marks will be awarded for complete answers to a total of THREE questions. Each question carries 25 marks. The marks for each part of a question are indicated at the end of the part in [.] brackets.

There are 75 marks available on this paper.

No calculators should be used.

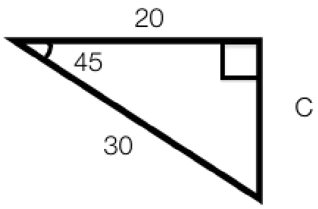
THIS PAPER MUST NOT BE REMOVED FROM THE EXAMINATION ROOM

Question 1

(a) In the following equation, what is the phase of the sine wave, a, b, c, or d: [3]

$$f(t) = a \sin(bt + c) + d$$

(b) Which is the correct formula for C in the following diagram [3]

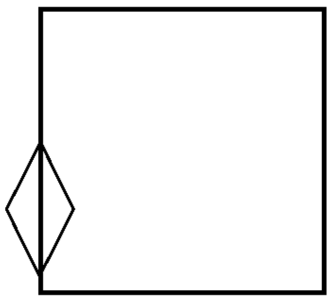


a) $C = \sin(20/30)$
b) $C = \sin(30/20)$
c) $C = 20 \tan^{-1}(45)$
d) $C = 20 \sin(45)$

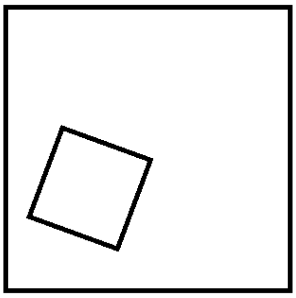
(c) Which image does the following code produce? [3]

```
rect(0, 0, 300, 300);

translate(100, 100);
scale(0.5, 1);
rotate(radians(45));

rect(0, 0, 100, 100);
```

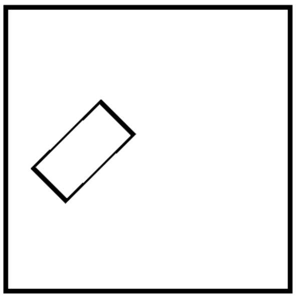
a)



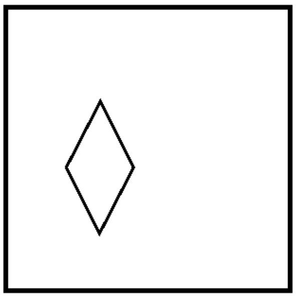
b)



c)



d)



(d) A point has polar coordinates (R, θ). What is it's x coordinate in cartesian coordinates? [3]

a) $x = R\cos(\theta)$
b) $x = \theta\cos(R)$
c) $x = R\sin(\theta)$
d) $x = \theta\sin(R)$

(e) You are provided with the following difference equation for a digital filter:
$y[n] = -\frac{1}{5}y[n-1] + \frac{1}{5}y[n-2] + \frac{1}{3}x[n] + 2x[n-1] + \frac{1}{2}x[n-2]$
Implement a function which applies this filter to an input signal:
`float[] applyFilter(float[] input)`

i. Decide how many coefficients there are, and declare variables for them. [2]
ii. Initialise the ceofficients. [1]
iii. Declare an array of an appropriate length to store the output signal. [2]
iv. Write the loop which applies the filter to the signal. [8]

Question 2

(a) What does the following code do? [3]

```
void draw()
{
        velocity.add(G);
        position.add(velocity);
        pushMatrix();
                translate(position.x, position.y);
                ellipse(0, 0, 20, 20);
        popMatrix();
}
```

a) move an ellipse at constant speed

b) apply gravity to an ellipse

c) move an ellipse at constant acceleration

d) move an ellipse with damping

(b) If v is the velocity of an object and p is it's position, which of the following formulae apply damping? [3]

a) $p = v - 0.01$

b) $p = v * 0.99$

c) $v = v * 0.99$

d) $v = v * 1.01$

(c) If you are generating random numbers with a gaussian distribution, what is the effect of increasing the variance. [3]

a) on average, the numbers increase

b) on average, the numbers decrease

c) the numbers are concentrated closer to the average

d) the numbers are more spread out away from the average

(d) In which of the following situations could you use motion detection? [3]

a) tracking a red object to control music

b) separating the image of a person from the background

c) detecting intruders to your home

d) a face recognition system

(e) For this question, you will be writing a function and associated classes that allow the representation of MIDI and OSC messages and the conversion from MIDI to OSC.

The function signature should look like this:

```
OSCMessage mIDIToOSC(MIDIMessage mIDIMessage)
```

i. Define the fields for the OSCMessage class, which should be able to represent a simple OSC message containing numerical data. [4]

ii. Define the fields for the MIDIMessage class, which should be able to represent a MIDI channel message. [4]

iii. Define the function itself, which converts the properties from one format to the other. [5]

Question 3

(a) If a single channel, digitised audio signal has a sampling rate of 44100Hz and a bit depth of 16 bits and a length of 25 seconds, how much memory is required to store it? [3]

a) 2205000 bytes

b) 17640000 bytes

c) 50 bytes

d) 1102500 bytes

(b) How can a digital audio signal be played at half the speed? [3]

a) Move the playhead at a rate of twice the sampling rate.

b) Move the playhead at a rate of half the sampling rate.

c) Ignore the second half of the sample data during playback.

d) Reduce the bit depth during playback to half the original.

(c) What is an advantage of a higher sampling rate for digital audio signals? [3]

a) The resulting digital signal is more compact that it would be with a lower sampling rate.

b) The digital signal measures amplitude more precisely.

c) The digital signal can be processed more rapidly.

d) The digital signal can encode a wider frequency range.

(d) If you want to record a signal with lots of low frequency components very precisely, why would you opt for a high bit depth? [3]

a) The high bit depth will capture the deeper frequencies more accurately.

b) The high bit depth would provide high precision on the frequency scale.

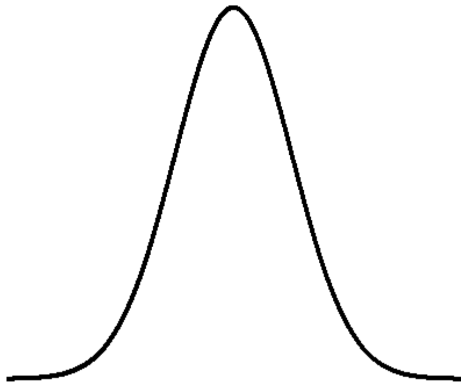
c) The high bit depth would provide high precision on the amplitude scale.

d) The high bit depth would increase the frequency bandwidth.

(e) The gaussian function, often called the bell curve, is defined by the following equation: [13]

$$G(x) = exp(\frac{-(x-m)^2}{2s^2})$$

Where *exp* is the exponential function. The graph of the function looks like this:



m and s are parameters, m defines where the middle of the "bell" is and s defines how wide it is.

Assuming that the *exp* function is built into processing, write a program that plots the gaussian function with a vertex curve.

Question 4

(a) In digital signal processing, a system is: [3]

a) A digital representation of audio data.
b) Anything which produces an output signal in response to an input signal.
c) A digital audio workstation.
d) A piece of hardware such as a computer.

(b) A filter is defined as follows: [3]
$y[n] = 0.33x[n] + 0.33x[n-1] + 0.33x[n-2]$ This filter is a:

a) Low pass filter.
b) High pass filter.
c) Chorus effect.
d) Phaser effect.

(c) Which of the following is a delay with feedback? [3]

a) $y[n] = y[n] * y[n-1]$
b) $y[n] = x[n] + 0.6y[n-100]$
c) $y[n] = x[n] + 0.6x[n-100]$
d) $y[n-1] = x[n-1] + x[n-10]$

(d) Higher order filters ... [3]

a) are more precise than lower order filters.
b) can handle more inputs.
c) are more efficient than lower order filters.
d) are more computationally expensive than lower order filters.

(e) Green screening is a common method in film and television production. An actor is filmed in front of a green screen. All green parts of the image are then removed so that the actor can then be inserted into a new background. Explain, with code examples, how you would write a programme to perform green screening. [13]

Question 5

(a) MIDI channel messages contain: [3]

a) A status byte and two data bytes.
b) A note on byte and a velocity byte.
c) A control message and a note message.
d) A note on message and a note off message.

(b) MIDI is still used today, even on modern platforms such as Apple's iOS. This is because: [3]

a) It is an effective way to describe the notes and rhythms of Western music.
b) It is very efficient.
c) It works well over networks.
d) It is popular amongst developers.

(c) OSC can be described as: [3]

a) A peer to peer protocol.
b) The HTTP protocol applied to music.
c) A client/ server protocol.
d) A closed protocol.

(d) An OSC message consists of: [3]

a) An address, a target node and the data.
b) An address, a type tag string and the data.
c) An address, a set of tags and the data in the tags.
d) A status byte and many data bytes.

(e) Audio filters are operations on time varying signals, so in principle, they could be applied to any time varying data. The filter below defines a filter that performs feedback with delay. Explain, with code examples, how you would apply that filter to a video. Try to describe the effect. [13]

$$y[t] = 0.2x[t] + 0.8y[t-1]$$