SYSC 4405: Midterm Exam (#:3) October 24, 2008
Carleton University, Systems and Computer Engineering

Background: You are working for a recording company that wants to make its music sound like it was recorded in a concert hall, rather than in a small recording studio. You are given the job of creating a DSP model of a concert hall, which can be applied to music during digital mastering.

To do this, you rent a conference hall, and conduct the following test. A musician plays her instrument on the centre of the stage, which is recorded at a microphone (as $x(t)$). In the centre of the audience, a technician records the sound at another microphone ($y(t)$).

You determine that the recorded sound is the sum of weighted contributions from four paths, each with a different delay. Thus

$$y(t) = \sum_{i \in \text{paths}} (\text{Amplitude}_i) x(t - \text{Delay}_i)$$

Path	Length	Delay	Amplitude
1	6.8 m	20 ms	0.22
2	10.2 m	30 ms	0.08
3	13.6 m	40 ms	0.05

1. (1 point) Your exam is exam number **3**. Write down this number.

2. (5 points) **Write a difference equation for $y(t)$ as a function of $x(t)$.**

3. (5 points) Signals $x(t)$ and $y(t)$ are sampled using an A/D converter with a sampling time $T_s = 0.04$ ms, producing $x[n]$ and $y[n]$. **Sketch the block diagram for the system with input $x[n]$ and output $y[n]$.** To show many delay elements in series, you may write: e.g. *Delay* $\times 15$.

4. (5 points) **Characterize the system in terms of the following properties: a) linear, b) memoryless, c) shift-invariant, d) LSI, e) stable, f) causal.** You only need to list *yes* or *no* for each property.

5. (5 points) The musician plays a note of E_8 (ie. E in the 8^{th} octave: $f = 5.274$ kHz). This note is composed of harmonics as shown at right.

$$x(t) = \sum_{h=1}^{4} A_i cos(2\pi h f t + \phi_i)$$

h	1	2	3	4
A_i [mV]	1.2	0.4	0.3	0.2
ϕ_i [rad]	0	0	$\frac{\pi}{4}$	$\frac{\pi}{2}$

Show a phasor plot of $x(t)$ as well as the frequency f_{max} above which aliasing occurs.

6. (5 points) The signal, $x(t)$, is sampled at T_s to get $x[n]$. **Show an equation for $x[n]$ after accounting for aliasing. Indicate which harmonics, if any, are aliased. For aliased harmonics, indicate whether the are folding or non-folding.**

7. (5 points) (For this question, assume that $x[n]$ is uniformly distributed between -1.60 V and 1.60 V). The signal, $x[n]$, is sampled with a 12-bit A/D differential converter with $X_{max} = -X_{min} = 2$ V. **What is the SNR due to the quantization error?**

8. (5 points) If you choose to implement the concert chamber filter with a DFT using block processing, **is a DFT of length $N = 2048$ sufficient? Calculate values of L and M for this filter and sketch the block filtering process** (indicate the length of any zero-padding and overlapping-signals)