MUS424:
Signal Processing Techniques for Digital Audio Effects Handout #11
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Homework #7: Reverberation Analysis & Synthesis
Due Date: June 7, 2012

Submission Instructions

Submit via coursework's DropBox. Create **a single compressed file** (`.zip`, `.tar` or `.tar.gz`) containing all your submitted files. Place the compressed file at the top level of your Drop Box. Name the file using the following convention:

`<suid>_hw<number>.zip`

where `<suid>` is your Stanford username and `<number>` is the homework number. For example, for Homework #1 my own submission would be named `jorgeh_hw1.zip`.

Each problem should be in its own subfolder, ideally named `problemX`. Therefore, if different problems ask to to modify the same code, you'll have to turn in different projects, one for each problem.

What to submit?

· Matlab problems: submit all the files necessary to run your code.

· VST implementations: only submit the corresponding `.h` and `.cpp` files.

· Theory problems: submit the solutions in **PDF format only**. LaTeX or other equation editors are preferred, but scans are also accepted. In case of scanned handwriting, make sure the scan is legible.

1 Laboratory

Problem 1. [120 Points]

In this lab, the plugin Reverb will be modified to incorporate filters which control the decay time and output equalization.

1(a) [70 Points] The feedback delay network reverberator Reverb as implemented has first-order filters in its feedback loop. Based on user settings specifying the low-frequency and high-frequency decay times and a transition frequency, design first-order shelf filter coefficients for each delay lines filter. The shelf filters should have DC and band-edge gains designed to give the selected decay times, taking into account the associated delay line lengths. For each filterset the pole frequency equal to the transition frequency.

Remember that an analog shelf filter has the following prototype:

$$h(s) = \frac{\ell_\pi s/\rho + \ell_0}{s/\rho + 1}$$

where ℓ_0 and ℓ_π define the DC and high-frequency gains and ρ controls the transition frequency (see page 250 of the course notes for more details).

Verify that you are approximately getting the desired decay time as a function of frequency for the low-frequency decay time set to 2.0 seconds and the high-frequency decay time set to 0.5 seconds, with the transition frequency set to 1.5 kHz. Hand in a spectrogram of the sample impulse response.

1(b) [20 points] Implement a parametric section to control the equalization of the wet signal. You can either re-use your own code from previous labs or you can borrow code from the posted solutions. Explain the difference between putting the equalization on the output vs. on the input. Finally, connect the parametric section to allow user control of the Reverb's equalization.

1(c) [30 points] Consider Table 1 showing estimates of the $T_{60}(\omega)$ and initial late field $EQ(\omega)$ for a reverb impulse response `memchu_bpeq.wav`, synthesized from a Memorial Church balloon pop response. Find settings of the T_{60}, EQ parameters and wet/dry mix controls such that the impulse response of the FDN reverberator sounds like the impulse response provided. This can be done by ear or using Matlab to estimate the correct parameters.

frequency [Hz]	125	250	500	1000	2000	4000	8000	16000
T_{60} [ms]	2,729	3,694	4,308	4,407	3,625	2,336	1,351	760
EQ [dB]	-11.48	-3.79	0	-5.07	-12.03	-13.97	-20.44	-47.08

Table 1: Synthetic Impulse Response T_{60} and initial late field EQ as a function of frequency