MUS424:
Signal Processing Techniques for Digital Audio Effects Handout #24
Jonathan Abel, David Berners May 18, 2006

Homework #3: Laboratory Exercise 3
Due Date: June 1, 2006

Reverberator Equalization and Feedback Filter Design

Problem 1. [80 Points]

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

Source code can be downloaded from the class website. All necessary materials are contained in the file `lab4.tar.gz`.

Solutions for each excercise should include the source files `Reverb.cpp` and `Reverb.hpp`, suitably modified. The files must be able to be compiled. Additional write-up is required for some sections of this problem.

1(a). [30 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 line's filter. The shelf filters should have DC and band-edge gains designed to give the selected decay times, taking into account the associted delay line lengths. The filters should also have gains equal to the geometric mean of their DC and band-edge values at the transition frequency. 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.

1(b). [30 Points] At the reverberator output, place a second-order peak filter specified by center frequency (on a log scale from about 100 Hz to about 10 kHz), Q (also on a log scale from about 0.25 to about 32) and peak gain (from -24 dB to 24 dB). Make the low-frequency and high-frequency decay times equal, and Fourier transform a segment of the reverb impulse response to verify that the equalization filter is performing as expected.

Using a drum track as input, make the decay time long (say, 5.0 seconds or so), and set the filter gain to about 20 dB, with a narrow (that is high) Q, and sweep the equalization filter center frequency. Would this sound different if the equalization were placed on the input?

1(c). [**20 points**] Consider the table below showing estimates of the $T_{60}(\omega)$ and initial equalization $q(\omega)$ for the balloon pop `balloon_c2.wav` recorded during our session at Memorial Church. Find settings of the equalization and T_{60} controls such that the impulse response of the FDN reverberator sounds like the balloon pop. Turn in Matlab plots showing the measured and implemented decay times and equalizations.

frequency, Hz	125	250	500	1000	2000	4000	8000	16000
T_{60}, ms	2806	4045	3922	4551	3915	2380	1179	515
$\lvert q(\omega)\rvert$, dB	-11.1	-6.2	0.0	-9.8	-22.4	-27.2	-30.8	-32.4