Agenda for Today

- Delay
- Reverberation
- Filters in Music Software
- Simple FIR Filters
- Impulse Response
- Simple IIR Filters
- Frequency Response
- Fourier Transform, z Transform
- Building Filters According to Cookbooks

Delay

Demo:

- delay time, amount of feedback, dry/wet ratio
- clicks when changing values: interpolation of control values?
- changing the pitch by modulating the delay time
- using linear interpolation to simulate entries at non-integer addresses in the delay line

Reverberation

- continuous set of reflections, no distinguishable single echos
- cannot be realized with a set of delay lines alone (shatter effect like in a bathroom or metallic character of sound)
- typical real-time solution: use a set of delay lines and smear out the echos using special diffusion filters
- now also a real-time solution: convolution reverb. The impulse response of a room is applied to each sample.

Filters in Music Software

- Demo: EQ in Steinberg Cubase
- Demo: resonating filter
  in virtual synthesizers

Simple FIR Filters

- blurring images in Adobe Photoshop through averaging
- building a low-pass audio filter using averaging with past values
- sharpening images in Adobe Photoshop by subtracting the average
- building a high-pass audio filter by subtracting the average
- all these filters have
  a finite impulse response (FIR),
  because there is no feedback

Impulse Response

- a linear and continuous filter is completely defined by its impulse response
- filter can be built using only its impulse response if that is finite and obeys causality
- FIR filters are:
  + easy to understand
  – expensive to run
  – expensive to tune in real time

Simple IIR Filters

- infinite impluse response (IIR)
  by using feedback
- example: the smoothing we used
  yesterday for the envelope
  env := 0.999*env+0.001*in
- IIR filters are:
  – difficult to understand
  + cheap to run
  + quite inexpensive to tune in real time

Frequency Response

- A linear and continuous filter is completely defined by its frequency response, i.e., information about how it alters the amplitude and phase of a given sine wave, depending on the frequency.
- Such filters will only change the amplitude and the phase of a sine wave, neither the frequency nor the waveform.

Fourier Transform, z Transform

- Computing the frequency response using $\sin(2\pi ft)$ is a tedious task.
- Complex Fourier transform: Rather use $\exp(2\pi ift)$ and employ the laws for exponentials. To delay by n samples means to multiply by $\exp(-2\pi inf/f_s)$.
- z Transform: Use $z := \exp(2\pi if/f_s)$. To delay by n samples means to multiply by z^{-n}.

Building Filters According to Cookbooks

- Examples from the book DAFX, ed. Udo Zölzer