The Piano

Acoustics of the Piano

• Five Lectures on the Acoustics of the Piano
• The Piano Hammer as a Nonlinear Spring

Piano Synthesis

We know how to

• simulate strings efficiently
• couple strings at the bridge

We will now discuss how to

• commute resonators and strings (for speed)
• convolve resonator impulse response with excitation (hammer)
• simulate a piano hammer in more detail
• identify the string-loop filter for stiff strings

http://www.speech.kth.se/music/5_lectures/
http://www.acs.psu.edu/drussell/Piano/NonlinearHammer.html
Commuted Synthesis

Features of Commuted Synthesis

- Enormous resonators can be implemented inexpensively (three orders of magnitude less computation for typical stringed instruments)
- Good qualitative excitation signals are easy to measure (just tap on the bridge)
- Apparent “resonator size” can be modulated by changing the playback rate of the excitation table

Drawbacks:

- Requires linearity and time invariance

Features:

- Trigger
- Excitation
- String
- Resonator
- Output

Schematic diagram of a stringed musical instrument.

Equivalent diagram in the linear, time-invariant case.

Use of an aggregate excitation given by the convolution of original excitation with the resonator impulse response.

Possible components of a guitar resonator.

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Committed Piano Synthesis

Hammer-string interaction pulses (force):

- Vertical lines = locations and amplitudes of three impulses
- Hammer-string interaction signal synthesized using one to three digital filters
- Filters depend on striking velocity

Synthesis of One Hammer-String Interaction Pulse

- Filter input = impulse
- Filter output = desired hammer-string force pulse
- As input amplitude increases, output pulse narrows => nonlinear filter
- For each specific impulse, filter is linear time-invariant
- Piano is “linearized” separately for each hammer velocity
Synthesis of Multiple Hammer-String Interaction Pulse

Multiple hammer-string interaction pulses = superposition of several individual pulses:

- Input to each filter is a single impulse
- Sum of outputs = superposition of hammer-string force pulses
- As impulse amplitude grows, output pulses become taller and thinner, showing less overlap.

Complete Piano Model

Natural Ordering:

- Soundboard and enclosure are commuted
- Only need a stored recording of their impulse response
- An enormous digital filter is otherwise required
Simplified Variations of the Committed Piano

One interaction pulse:

Up to three identical interaction pulses:

Three General Interaction Pulses

Differentially Filtered Interaction Pulses

Each new delay is equal to the travel from the hammer, to the agraffe, and back to the hammer.
Striking a Digital Waveguide String

Waveguide string in “physical canonical form”:

- Delay lines contain traveling force waves
- Hammer-string force pulses are summed into both directions (by physical symmetry)

Equivalent Computational Models

- Hammer position set by feedforward comb filter
- String model independent of hammer position (more valuable for plucked strings)
- Same factoring applies to pickup locations
Use of a Tapped Delay Line

- Delay-line tap simulates hammer-strike echoes returning from the far end of the string
- Total delay line length equals twice the travel time along the entire length of the string
- Placement is not critical, so an interpolating tap is not necessary
- Tap motion gives a free built-in “flanger” effect
- Since force waves are most convenient, no sign inversion at the string endpoints

Excitation Factoring

Commuted Piano Synthesizer:

- Excitation & Nonlinear Filtering
- String(s)
- Resonator(s) and/or Equalization and/or Reverb and/or Comb Filter(s)

A few small filters are retained in parametric form for

- Soundboard and enclosure
- Equalization for color variations
- Reverberation
- Flange
- Chorus
- Simulated hammer-strike echoes
- Multiple outputs

Filters are easily modified, even while a note is playing
Piano and Harpsichord Sound Examples

- [http://ccrma.stanford.edu/~jos/wav/Harpsichord.wav](http://ccrma.stanford.edu/~jos/wav/Harpsichord.wav)
- Julien Bensa’s synthetic piano

The Ideal (Lossless) Stiff String

Simulation of a rigidly terminated, stiff string.

- All $N$ allpass filters $H_a(z)$ have had a minimum sample of delay extracted, corresponding to the delay at $f_s/2$
- All have been combined at a single point
- The delay-line length $N$ is the number of samples in $K$ periods of $f_K$, where $K$ is the number of the highest partial (usually the last one before $f_s/2$)
- In stiff strings, high frequencies travel faster $\Rightarrow$ allpass delay decreases with frequency
- An order 10 allpass gives high quality piano synthesis
- FFT convolution most efficient for maximum accuracy (FIR filter length $\approx N/2$ for note F1)
Nonparametric Model of Measured Piano-String Frequency Response (from Julien Bensa’s thesis)

High-quality frequency-response model based on measurements from a grand-piano string, note F1.
A basic problem in waveguide synthesis of string is identifying the loop filter $H_l(z)$ which controls

- Partial overtone decay time (due to $|H_l(e^{j\omega})|$)
- Partial overtone tuning perturbations (due to $\angle H_l(e^{j\omega})$)

### Loop-Filter Design Methods

One of the earliest methods was “periodic linear prediction” — a linear combination of a small group of samples predicts a sample one period away from the midpoint of the group.

More recently, methods have been developed based on measurements of the time-constant of decay for each partial overtone, since this is what matters the most.

- Record a plucked (or struck) string.
- Measure frequency $f_k$ and decay time $\tau_k$ for each overtone.
  - Energy Decay Relief (EDR) useful for this purpose
  - Sinusoidal modeling has been used as well
  - In either case:
    * Fit a straight line to the log of the amplitude envelope (e.g., using polyfit in Matlab or Octave).
    * Decay time $\tau_k$ is simply related to the slope of this line.

- Translate each decay-time $\tau_k$ into a desired loop-gain at each frequency $f_k$, thus determining $|H_l(e^{j\omega_k})|$ for each partial.

- If desired, use the inharmonic spacing of the $f_k$ to compute samples of the desired phase of the loop filter, $\angle H_l(e^{j\omega_k})$.

- Use a general purpose digital filter design routine to design the loop filter which best approximates the desired frequency-response samples $H_l(e^{j\omega_k})$.

- Phase-matching filter design utilities:
  - Steiglitz-McBride algorithm (IIR or FIR)
    See `stmcb` in Matlab Signal Processing Tool Box
  - Equation-Error Minimization (IIR or FIR)
    See `invfreqz` in matlab
  - Hankel-Norm Minimization (IIR)
Also called a “balanced truncation model” (state space)
– Linear-phase FIR (fir1, firls, remez)
• Phase-insensitive filter design:
  – Linear Prediction
  See lpc or yulewalk in the Matlab SPTB
  – Create a desired linear or minimum phase and use a phase-sensitive design method.
• The fit at low frequencies should be weighted higher.
• A simple $1/f$ weighting usually works well.
• Bark spectral warping is commonly used. It implicitly weights low frequencies more by “stretching out” the low-frequency frequency axis and compressing the high-frequency frequency axis.
• Piano strings are highly dispersive:
  – Recall that every LTI filter can be factored into a linear phase filter in series with an allpass filter
  – The allpass part for a low-pitched string requires on the order of 100 poles (and 100 zeros) to provide a closely matched impulse response
  – Ten poles is a typical number used in practice (can be very good perceptually)
– Most methods for allpass filter design fail numerically at this high order
– The linear-phase part, in contrast, is well approximated using only an 8-tap FIR filter, and can be designed by any good method for FIR filter design (e.g., invfreqz in Matlab/Octave)

Order Estimation

Given a desired frequency response, a rough digital filter order estimate can be obtained as follows:
• FIR: Look at the IFFT duration (e.g., $20 \log(|h(n)|)$)
• IIR: Start with half of the FIR order (equal degrees of freedom)
• In either case, convert desired frequency response to minimum phase whenever possible: Matlab code: [link]
• For piano strings, a low-order minimum-phase damping filter is fine because we’re going to need a high-order allpass dispersion filter anyway