Virtual Analog Modeling

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Outline

Signal processing techniques for modeling analog audio systems used in music technology

• Introduction
• 1. Reduction of digital artifacts
• 2. Introducing analog ‘feel’
• 3. Emulation of analog systems
• Case studies:
  Virtual analog oscillators and filters, guitar pickups, spring reverb, ring modulator, carbon mic, antiquing
Introduction

• Virtual analog modeling = Imitate analog systems with digital ones
• Digitization, a current megatrend of turning everything digital
  ✓ CD, MP3, DAFX, digital music studios, laptop music…
• Analog music technology is getting old and expensive
  ✓ Software emulation is cheaper and nicer, if it sounds good…
• Examples: virtual analog filters and synthesizers, electromechanical reverb emulations, guitar amplifier models, and virtual musical instruments.

Three Different Goals

1. Reduce digital artifacts
2. Add analog ‘feel’
3. Emulate

Ref: Julian Parker, PhD thesis, to be published later this year
1. Reduce Digital Artifacts

- Digital signal processing has limits and undesirable side-effects
  - Quantization noise
  - Discrete time (unit delays)
  - Aliasing, imaging (periodic frequency domain)
  - Frequency warping (caused by, e.g., bilinear transformation)
  - Instabilities under coefficient modulation (time-variance)
- Solutions take us closer to analog
  - Use more bits (24 bits) or floating-point numbers
  - Oversampling
  - Interpolated delay lines
  - Antialiasing techniques

Digital Flanging Effect

- The delay-line length must vary smoothly to avoid clicks
  - Interpolation (fractional delay filter)
- Otherwise “zipper noise” is produced
Flanging Effect with Fractional Delay

- Use FIR or allpass fractional delay filter to vary delay smoothly (Laakso et al. 1996)

FIR

Allpass

Digital Subtractive Synthesis

- Emulation of analog synthesizers of the 1970s
- One or more oscillators, e.g., an octave apart or detuned
- Second- or fourth-order resonant lowpass filter
- At least two envelope generators (ADSR)

(Sound example by Antti Huovilainen, 2005)
Oscillators in Subtractive Synthesis

- Usually periodic waveforms
  - All harmonics or only odd harmonics of the fundamental
- Digital implementation must **suppress aliasing**

(Figure from: T. D. Rossing: *The Science of Sound.* Second Edition. Addison-Wesley, 1990.)

Aliasing – The Movie

- Trivial sampling of the sawtooth signal
- Harsh aliasing particularly at high fund. frequencies
  - Inharmonicity
  - Beating
  - Heterodyning

Video by Andreas Franck, 2012
No Aliasing

- Additive synthesis of the sawtooth signal
- Contains harmonics below the Nyquist limit only

Differentiated Parabolic Wave Algorithm

- A method to produce a sawtooth wave with reduced aliasing (Välimäki, 2005)
  - 2 parameters: fundamental frequency $f$ and sampling frequency $f_s$

$$p(t) = \int t dt = \frac{t^2}{2}$$

$$H(z) = c (1 - z^{-1})$$

where $c = f_s / 4f$
Signal Generation in DPW Algorithm

- Output of modulo counter \( x(n) \)
  - A ‘trivial’ sawtooth wave
- Squared signal \( x^2(n) \)
  - Piecewise parabolic wave
- Differentiated signal \( c \left[ x^2(n) - x^2(n-1) \right] \)
  - Difference of neighbors

Aliasing is Reduced!

- Spectrum of modulo counter signal \( x(n) \)
- Spectrum of squared signal \( x^2(n) \)
- Spectrum of differentiated signal \( c \left[ x^2(n) - x^2(n-1) \right] \)

Nyquist limit (22050 Hz)
Compare Sawtooth Wave Algorithms

- A scale at high fundamental frequencies
  - Trivial sawtooth (modulo counter signal)
  - DPW sawtooth
  - Ideal sawtooth (additive synthesis)

\[ f_s = 44.1 \text{ kHz} \]

Higher-order DPW Oscillators

- Trivial sawtooth can be integrated multiple times
  (Välimäki et al., 2010)

<table>
<thead>
<tr>
<th>Polynomial order</th>
<th>Polynomial function</th>
</tr>
</thead>
<tbody>
<tr>
<td>( N = 1 )</td>
<td>( x )</td>
</tr>
<tr>
<td>( N = 2 )</td>
<td>( x^2 )</td>
</tr>
<tr>
<td>( N = 3 )</td>
<td>( x^3 - x )</td>
</tr>
<tr>
<td>( N = 4 )</td>
<td>( x^4 - 2x^2 )</td>
</tr>
<tr>
<td>( N = 5 )</td>
<td>( x^5 - 10x^2/3 + 7x^3/3 )</td>
</tr>
<tr>
<td>( N = 6 )</td>
<td>( x^6 - 5x^4 + 7x^3 )</td>
</tr>
</tbody>
</table>

The polynomial signal must be differentiated \( N - 1 \) times and scaled to get the sawtooth wave.
Integrated Polynomial Waveforms

\[ N = 1 \]

\[ N = 2 \]

\[ N = 3 \]

\[ N = 4 \]

\[ N = 5 \]

\[ N = 6 \]

Differenced Polynomial Waveforms

\[ N = 1 \]

\[ N = 2 \]

\[ N = 3 \]

\[ N = 4 \]

\[ N = 5 \]

\[ N = 6 \]
Spectra of Differenced Waveforms

$N = 1$

$N = 2$

$N = 3$

$N = 4$

$N = 5$

$N = 6$

Use a shelf filter to equalize!

Polynomial Transition Region (PTR)

- The PTR algorithm implements DPW efficiently and extends it
  - Trivial sawtooth (modulo counter)
  - DPW waveform

Efficient Polynomial Transition Region Algorithm (EPTR)

- Ambrits and Bank (Budapest Univ. Tech. & Econ.) proposed an improvement (SMC-2013, Aug. 2013)
  - Eliminates the 0.5-sample delay and the constant offset
  - Reduces the computational load by 30% (first-order polyn. case)
  - Extends the PTR method to asymmetric triangle waveform synthesis

BLEP Method

- BLEP = Bandlimited step function (Brandt, ICMC’01), which is obtained by integrating a sinc function
  - Must be oversampled and stored in a table
- BLEP residual samples are added around every discontinuity
**BLEP Method Example**

- A shifted and sampled BLEP residual is added onto each discontinuity.
- The shift is the same as the fractional delay of the step.
- The BLEP residual is inverted for downward steps.
- The ideal BLEP function is the sine integral (Matlab function `sinint`) (Välimäki et al., 2012)

**Polynomial BLEP Method (PolyBLEP)**

- The BLEP residual table can be replaced with a polynomial approximation (Välimäki et al., 2012)
- Lagrange polynomials can be integrated and used for approximating the sinc function.
- Low-order cases are of interest: $N = 1$ (Välimäki and Huovilainen, 2007)
  - $N = 2$ (Välimäki et al., 2012)
  - $N = 3$ (Välimäki et al., 2012)
Goals

1. Reduce digital artifacts
2. Add analog ‘feel’
3. Emulate

2. Digital Versions of Analog ‘Feel’

- Digital systems are too good
  - Analog systems are noisy and change when they warm up, produce distortion when input amplitude gets larger, …
- Solutions
  - Simulated parameter drift
  - Nonlinearities (Rossum, ICMC-1992, …)
  - Additional noises
  - Imperfect delays (Raffel & Smith, DAFX-2010)
Biquad Filter with a Nonlinearity

- Dave Rossum proposed to insert a saturating nonlinearity inside a 2nd-order IIR filter (Rossum, ICMC 1992)

Audio Antiquing*

- Render a new recording to sound aged
  - For example, imitate the lo-fi sound of LP, gramophone, or phonograph recordings
- Simulate degradations with signal processing techniques (González, thesis 2007; Välimäki et al., JAES 2008)
  - Local degradations: clicks and thumps (low-frequency pulses)
  - Global degradations: hiss, wow, distortion, limited dynamic range, frequency band limitations, resonances

* Thanks to Perry Cook!
Audio Antiquing Example #1: Phonograph

1. CD (original)
2. Phonograph cylinder (new – best quality)
3. Phonograph cylinder (worn)

Audio Antiquing Example #2: Vinyl LP

1. CD (original)
2. LP (new – best quality)
3. LP (worn)
Vinyl LP Simulation Algorithm

- Adjust parameters or skip processing steps for better quality
- For thumps and tracking errors, time of revolution:
  60/33 sec = 1.8 sec

Ref. Välimäki et al., JAES 2008

Approaches

1. Reduce digital artifacts
2. Add analog ‘feel’
3. Emulate
Black and White-Box Models

- Black-box models attempt to imitate the analog system based on its input-output relationship
- Swept-sine methods (Farina, 2000; Novák et al., 2010; Pakarinen, 2010)
- Volterra filters (for weakly nonlinear systems) (Hélie, DAFX 2006, 2010)
- Grey-box models use some information about the system structure, then use black-box techniques
- White-box methods are physical models of the circuitry
  - Also antiquing can be based on physical modeling

Ref: Rafael de Paiva, PhD thesis, 2013, to be published
Moog Ladder Filter

- Bob Moog introduced an analog resonant lowpass filter design, which became famous
- Four lowpass transistor ladder stages and a differential pair

Digital Moog Filter

- Simplified version of the digital nonlinear 4th-order Moog ladder filter (Huovilainen, DAFx-2004; Välimäki & Huovilainen, CMJ 2006)
Sweeping the Resonance Frequency

- Changing the resonance frequency does not affect the Q value (much)

Video by Oskari Porkka & Jaakko Kestilä, 2007
Self-Oscillation

- When \( \text{Cres} = 1 \), the digital Moog filter oscillates for some time.
- However, \( \text{Cres} \) can be made larger than 1, because the \( \tanh \) limits the amplitude!

Image by Oskari Porkka & Jaakko Kestilä, 2007

Improved Digital Moog Filter (2013)

- Novel version derived using the bilinear transform (D’Angelo & Välimäki, ICASSP 2013; Smith, LAC-2012)
  - Self-oscillates well!
  - More accurate modeling of the nonlinearity
Novel Digital Moog Filter (2013)

Guitar Pickup Modeling

- The pickup is a magnetic device used for capturing string motion
  - Useful in steel-stringed instruments: guitars, bass, the Clavinet

Ref: D’Angelo & Välimäki, ICASSP 2013

Ref. Paiva et al., JAES, 2012.
Magnetic Induction in Guitar Pickup

- String proximity increases the magnetic flux
- The change causes an alternating current in the winding

Ref. Paiva et al., JAES, 2012.

Pickup Nonlinearity

- Sensitivity is different for the vertical and horizontal polarizations
- 2-D FEM simulations using Vizimag

Ref. Paiva et al., JAES, 2012.
**Pickup Nonlinearity**

a) String displacement in the **vertical** direction leads to harmonic asymmetric distortion (*all* harmonics)

b) String displacement in the **horizontal** direction leads to harmonic symmetric distortion (*even* harmonics)


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**Spring Reverberation**

- Spring reverberators are an early form of artificial reverberation

- Reminiscent of room reverberation, but with distinctly different qualities

- Recent research characterizes the special sound of the spring reverberator, and models it digitally (Abel, Bilbao, Parker, Välimäki...)

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Parametric Spring Reverberation Model

- Many (e.g. 100) allpass filters produce a chirp-like response
- A feedback delay loop produces a sequence of chirps
- Random modulation of delay-line length introduces smearing

![Diagram of Parametric Spring Reverberation Model]

Ref. V. Välimäki et al., JAES, 2010.

Interpolated Stretched Allpass Filter

- A low-frequency chirp is produced by a cascade of ~100 ISAFs

![Diagram of Interpolated Stretched Allpass Filter]

Ref. V. Välimäki et al., JAES, 2010.
Carbon Microphone Modeling

- The sandwich structure is used (Välimäki et al., DAFX book 2e, 2011)

Pre-filter consists of 2 or 3 EQ filters
- Nonlinearity is a polynomial waveshaper (order 2…5)
  (Oksanen & Välimäki, 2011)
### Modeling of the Carbon Microphone Nonlinearity for a Vintage Telephone Sound Effect

<table>
<thead>
<tr>
<th>Sample type</th>
<th>Input</th>
<th>Linear &amp; BP filtered</th>
<th>Nonlinear processing</th>
<th>Nonlinear processing + noise</th>
</tr>
</thead>
<tbody>
<tr>
<td>Male Speech</td>
<td><img src="image1" alt="Male Speech" /></td>
<td><img src="image2" alt="Male Speech" /></td>
<td><img src="image3" alt="Male Speech" /></td>
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<td><img src="image11" alt="Music" /></td>
<td><img src="image12" alt="Music" /></td>
</tr>
</tbody>
</table>

### Ring Modulator

- Julian Parker proposed a model for the ring modulator (DAFx-11)

![Ring Modulator Circuit](image13)
Ring Modulator

- Julian Parker proposed a model for the ring modulator (DAFx-11)

Digital

\[ V_o \]

\[ V_m \]

\[ 0.5 \]

\[ -1 \]

500 Hz

Ring Modulator

- BBC Research implemented Parker’s ring modulator: http://webaudio.prototyping.bbc.co.uk/ring-modulator/

Simulation of Analog Synth Waveforms

- For example, the Moog Voyager analog music synthesizer
- Waveform can be imitated using *phase distortion synthesis* or by filtering a *sawtooth oscillator signal*
- Alternatively, use a wave digital filter model of the osc. circuit (De Sanctis & Sarti, IEEE ASL 2010)

Ref. Pekonen, Lazzarini, Timoney, Kleimola, Välimäki, JASP 2011
Novel Audio DSP Algorithms Inspired by Virtual Analog Research

- Same tools, different uses
- The integration-differentiation idea (DPW) for wavetable and sampling synthesis (Geiger, DAFX-2006; Franck & Välimäki, DAFX-2012; JAES, TBP in 2013)
- Linear dynamic range reduction with dispersive allpass filters (Parker & Välimäki, IEEE SPL, 2013)

Integrated Wavetable and Sampling Synthesis

- The integration-differentiation idea helps pitch-shifting in wavetable and sampling synthesis (Geiger, DAFX, 2006; Franck & Välimäki, DAFX 2012; JAES, TBP in 2013)
- Transient problems in the time-varying case
- How about real-time implementation?
Dynamic Range Reduction using an Allpass Filter Chain

- Dispersive allpass filters, like in spring reverb models (Parker & Välimäki, IEEE SPL, 2013)
- Use golden-ratio coefficients for the allpass filters ($g = \pm 0.618$)
- Delay-line lengths of 3 AP filters are adjusted by trial and error

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Dynamic Range Reduction using an Allpass Filter Chain

- About 2.5 dB (even 5 dB) reduction in amplitude
Future Work

- Automatic modeling of nonlinear analog audio systems
- Alias reduction in nonlinear audio processing systems
- Subjective evaluation of virtual analog models – how to compare?
- Modeling of all electronic musical instruments and devices

Conclusion

- Virtual analog modeling provides software versions of analog hardware
  - Sound quality is improving
- Many successful examples from the past 15 years, e.g. virtual analog synths, virtual effects processing, guitar amp models
- Create also something new: novel signal processing methods, new effects?
Thanks to All My Collaborators in Virtual Analog Research

- Julian Parker
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- Andreas Franck (Fraunhofer IDMT)

Recommended Reading

References

References (Page 3)

- C. Raffel and J. Smith, "Practical modeling of bucket-brigade device circuits," in Proc. DAFX-10, Graz, Austria, Sept. 2010.