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Sound check

Virtual Analog Modeling

Vesa Välimäki
Aalto University, Dept. Signal Processing and Acoustics
(Espoo, FINLAND)

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

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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

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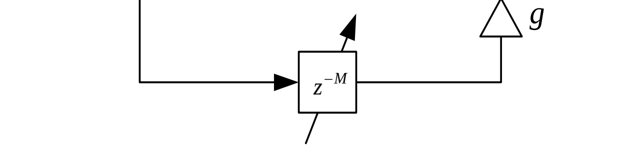
Digital Flanging Effect

- The delay-line length must vary smoothly to avoid clicks
  - ✓ Interpolation (fractional delay filter)
- Otherwise “zipper noise” is produced



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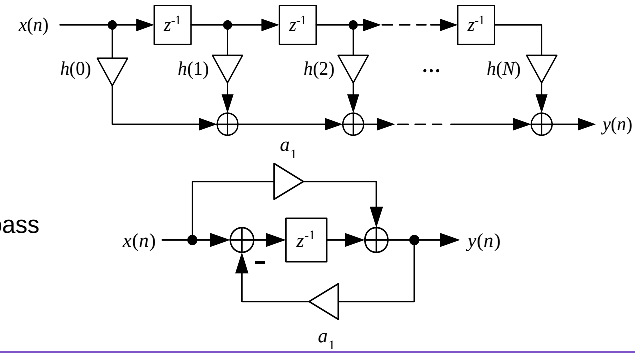
Flanging Effect with Fractional Delay

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

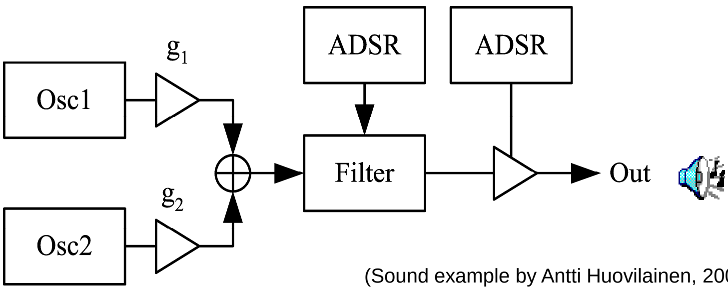
✓ FIR

✓ Allpass



Digital Subtractive Synthesis

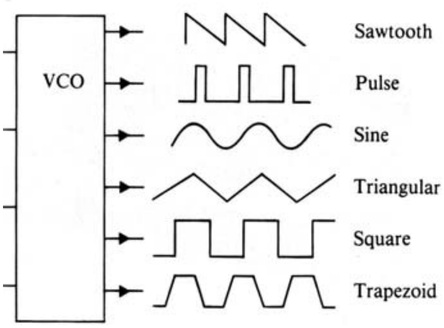
- Emulation of analog synthesizers of the 1970s
- One or more oscillators, e.g., an octave apart or detuned
- Second- or fourth-order resonant lowpass filter
- At least two envelope generators (ADSR)



(Sound example by Antti Huovilainen, 2005)

Oscillators in Subtractive Synthesis

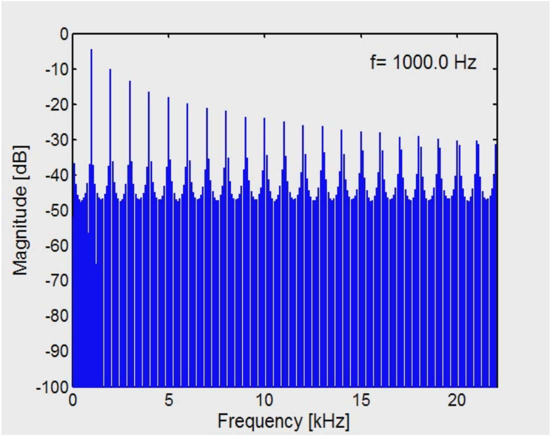
- Usually periodic waveforms
  - All harmonics or only odd harmonics of the fundamental
- Digital implementation must **suppress aliasing**



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

Aliasing – The Movie

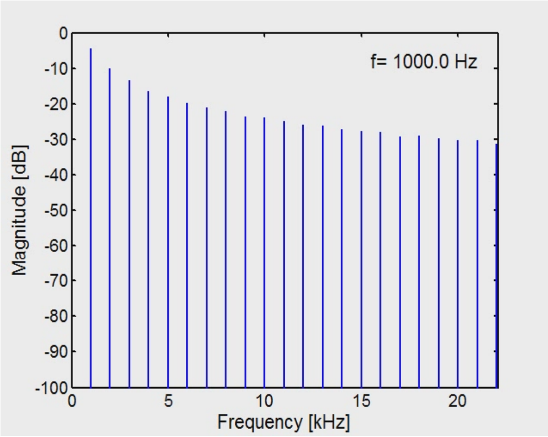
- Trivial sampling of the sawtooth signal
- Harsh aliasing particularly at high fund. frequencies
  - Inharmonicity
  - Beating
  - Heterodyning



Video by Andreas Franck, 2012

No Aliasing

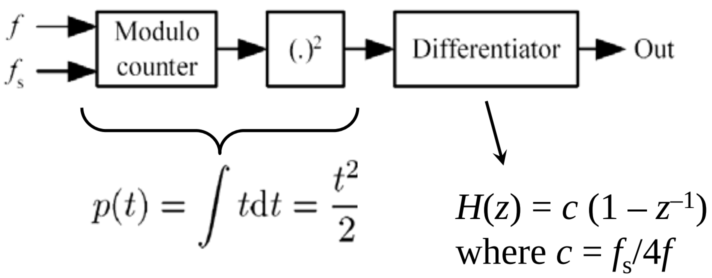
- Additive synthesis of the sawtooth signal
- Contains harmonics below the Nyquist limit only



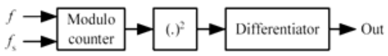
Video by Andreas Franck, 2012

Differentiated Parabolic Wave Algorithm

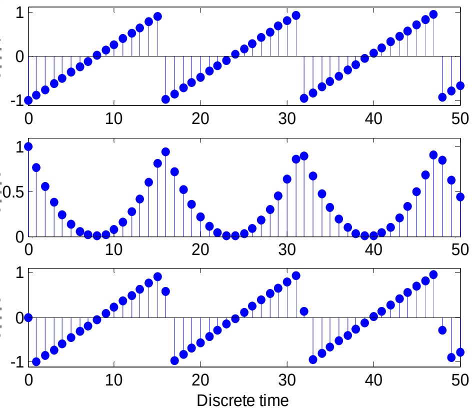
- A method to produce a sawtooth wave with reduced aliasing (Välimäki, 2005)
  - 2 parameters: fundamental frequency f and sampling frequency f_s



Signal Generation in DPW Algorithm

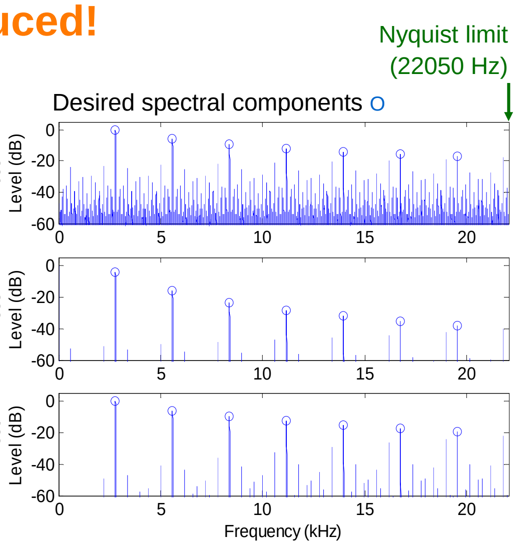


- Output of modulo counter $x(n)$
  - A 'trivial' sawtooth wave
- Squared signal $x^2(n)$
  - Piecewise parabolic wave
- Differentiated signal $c\,[x^2(n) - x^2(n-1)]$
  - Difference of neighbors



Aliasing is Reduced!

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



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 kHz

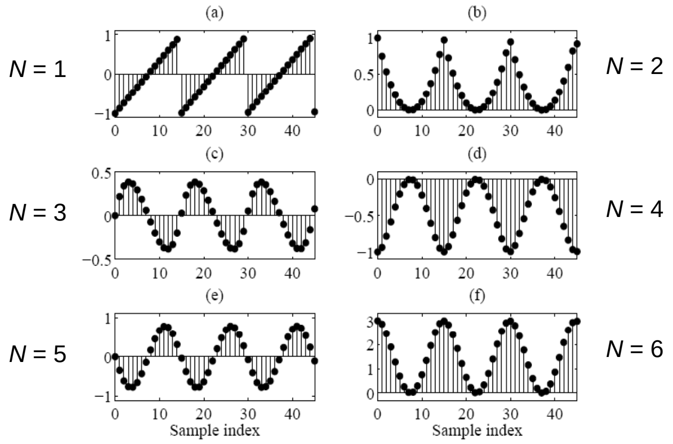
Higher-order DPW Oscillators

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

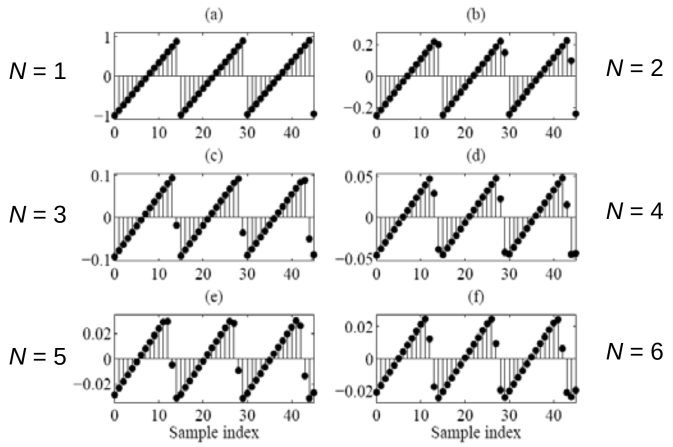
Polynomial order	Polynomial function
$N = 1$	x
$N = 2$	x^2
$N = 3$	$x^3 - x$
$N = 4$	$x^4 - 2x^2$
$N = 5$	$x^5 - 10x^3/3 + 7x/3$
$N = 6$	$x^6 - 5x^4 + 7x^2$

The polynomial signal must be **differenced** $N - 1$ times and **scaled** to get the sawtooth wave

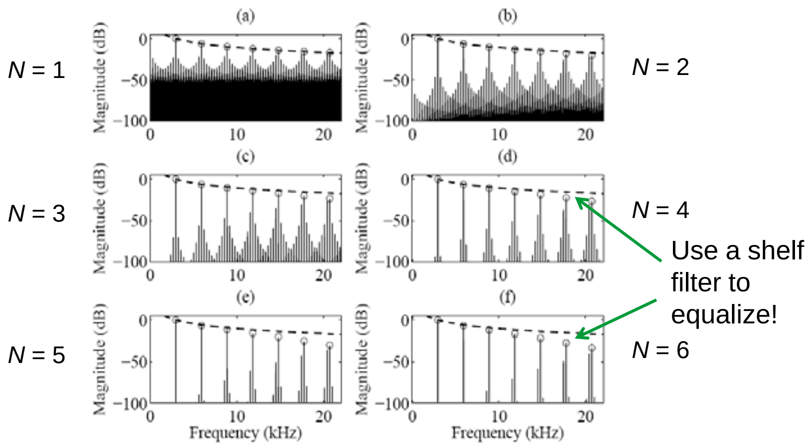
Integrated Polynomial Waveforms



Differenced Polynomial Waveforms

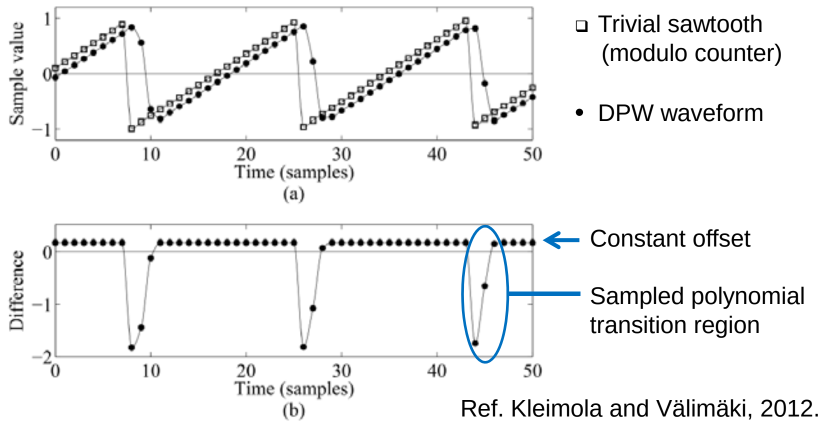


Spectra of Differenced Waveforms



Polynomial Transition Region (PTR)

- The PTR algorithm implements DPW efficiently and extends it



Ref. Kleimola and Välimäki, 2012.

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

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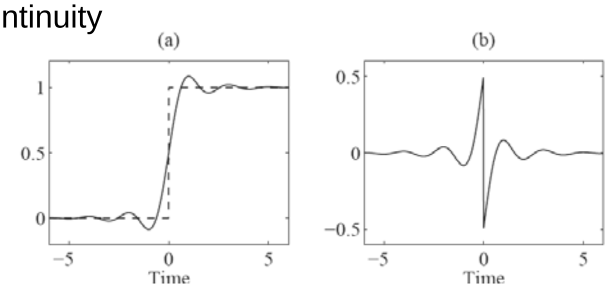
BLEP Method

- 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



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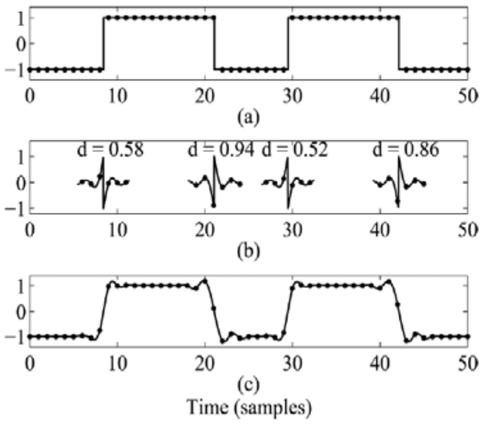
BLEP Method Example

- 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)

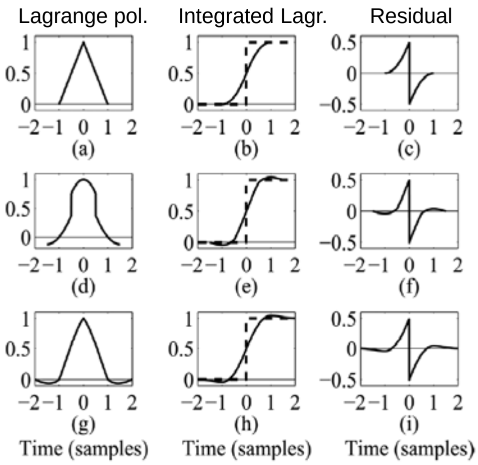
- The BLEP residual table can be replaced with a polynomial approximation

  (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)

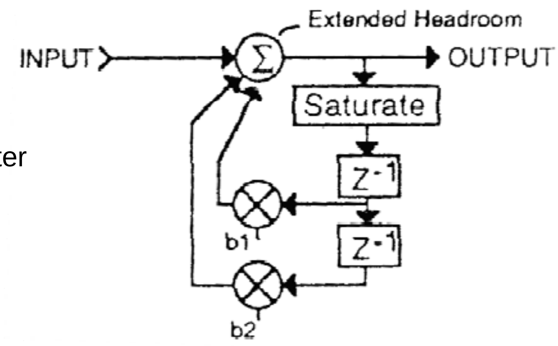


Figure: D. Rossum, Proc. ICMC 1992.

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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!

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Audio Antiquing Example #1: Phonograph

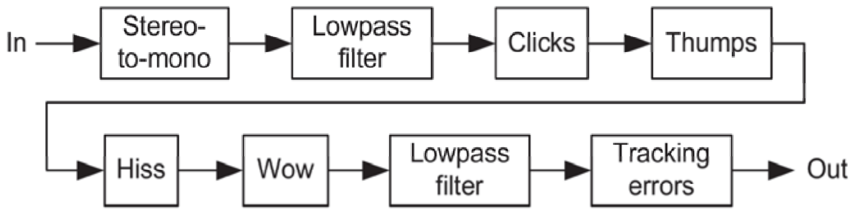
1. CD (original)
2. Phonograph cylinder (new – best quality)
3. Phonograph cylinder (worn)

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

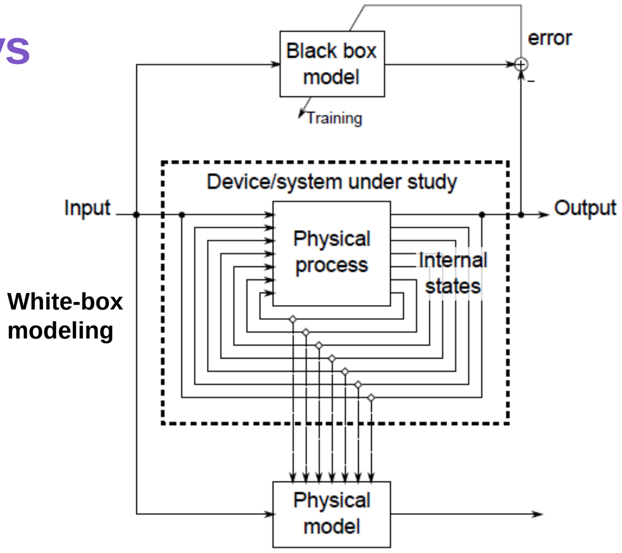
Approaches

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

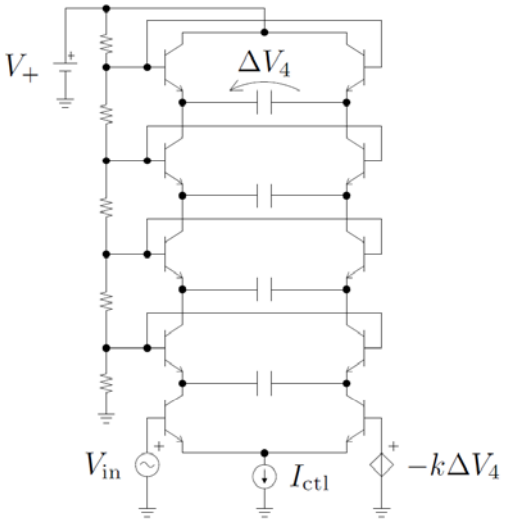
Black-Box vs White-Box Modeling



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



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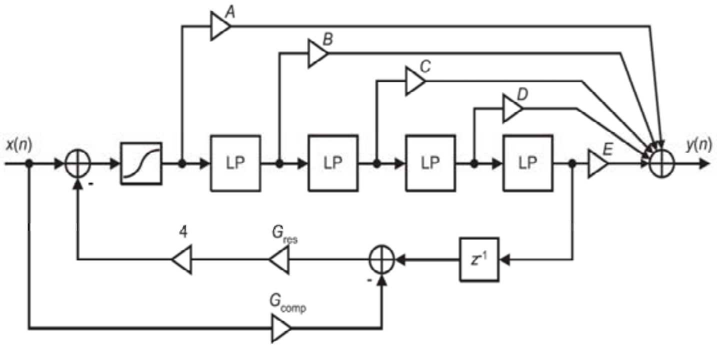
Digital Moog Filter

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



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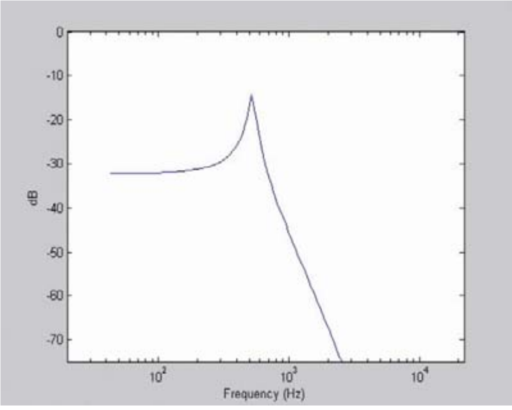
Sweeping the Resonance Frequency

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



Video by Oskari Porkka & Jaakko Kestilä, 2007

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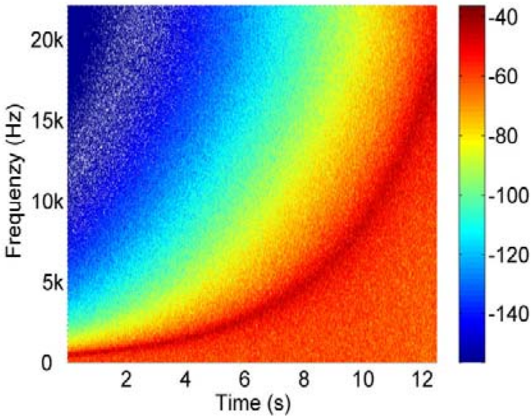

Image by Oskari Porkka & Jaakko Kestilä, 2007

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Self-Oscillation

- When Cres = 1, the digital Moog filter oscillates for some time
- However, Cres can be made larger than 1, because the TANH limits the amplitude!

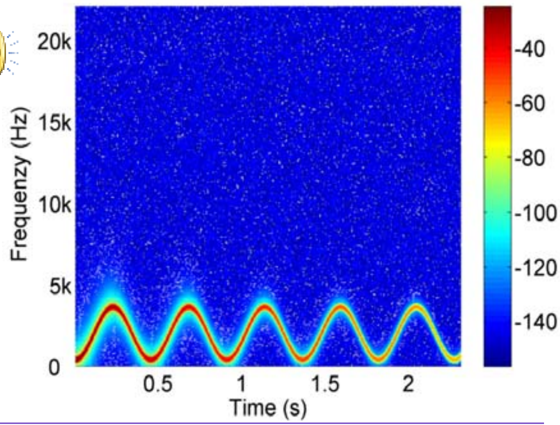
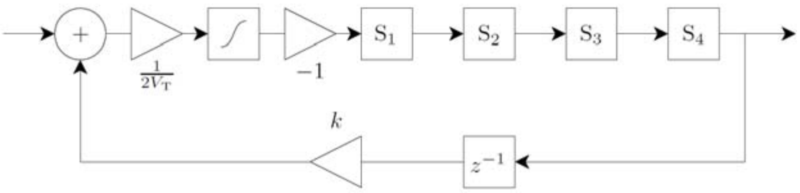


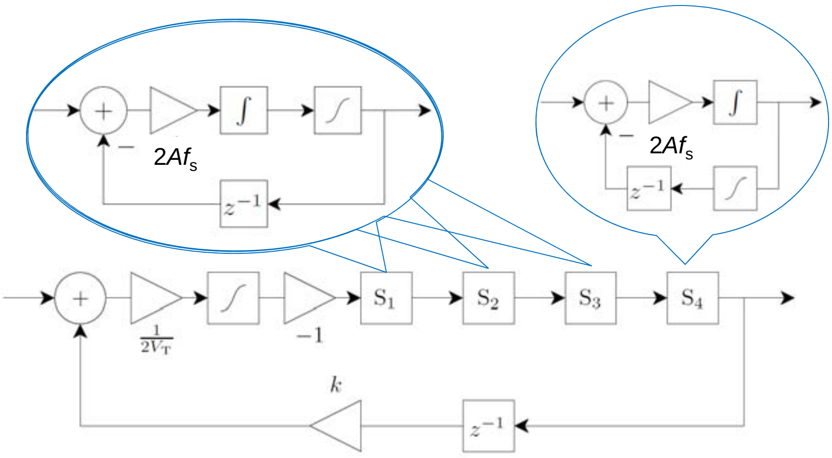
Image by Oskari Porkka & Jaakko Kestilä, 2007

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)



Ref: D'Angelo & Välimäki, ICASSP 2013

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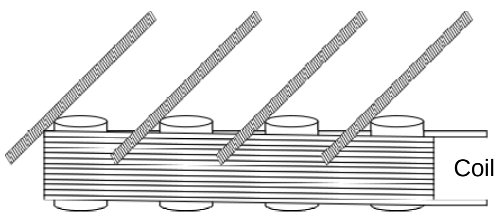
Guitar Pickup Modeling

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



Stratocaster *Les Paul*

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

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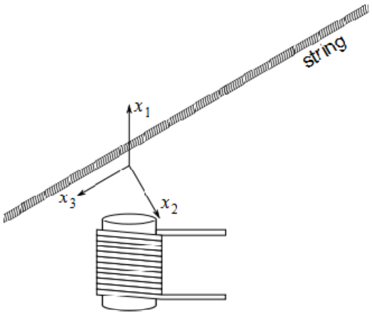
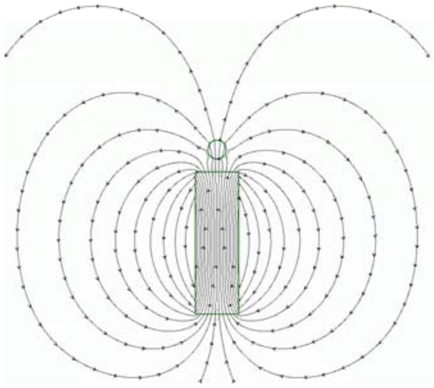
Magnetic Induction in Guitar Pickup

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

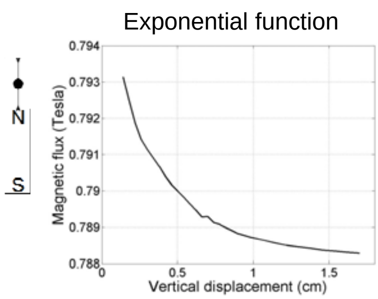


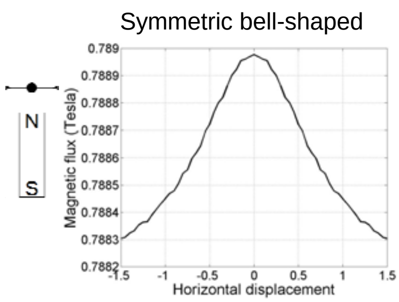


Ref. Paiva *et al*., JAES, 2012.

Pickup Nonlinearity

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



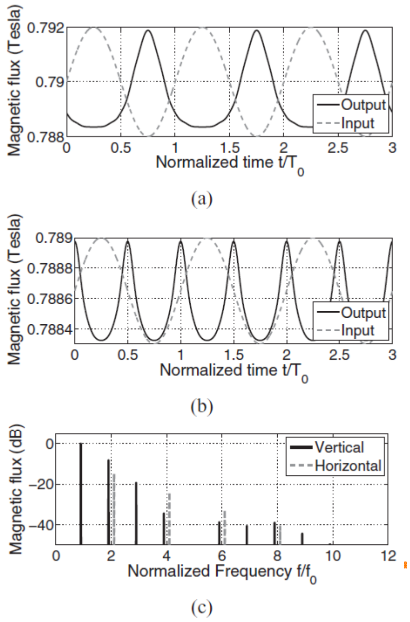


Ref. Paiva *et al*., JAES, 2012.

Pickup Nonlinearity

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

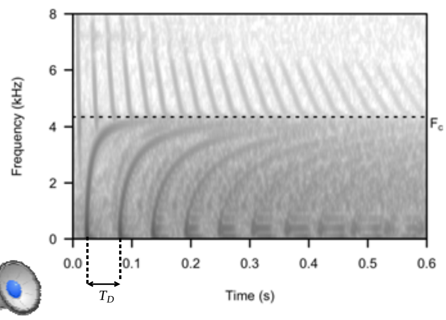
b) String displacement in the **horizontal** direction leads to harmonic symmetric distortion (*even* harmonics)



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

Spring Reverberation

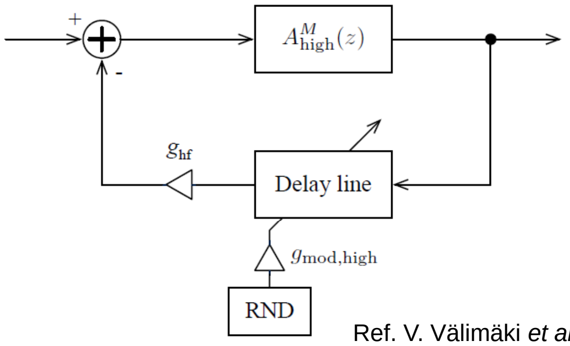
- Spring reverberators are an early form of artificial reverberation



- 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…)

Parametric Spring Reverberation Model

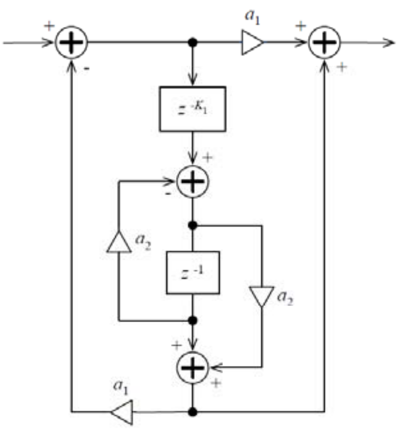
- Many (e.g. 100) allpass filters produce a chirp-like response
- A feedback delay loop produces a sequence of chirps
- Random modulation of delay-line length introduces smearing

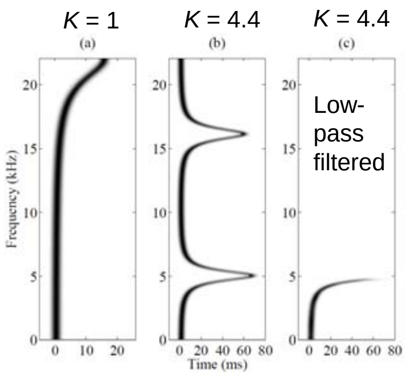


Ref. V. Välimäki *et al*., JAES, 2010.

Interpolated Stretched Allpass Filter

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

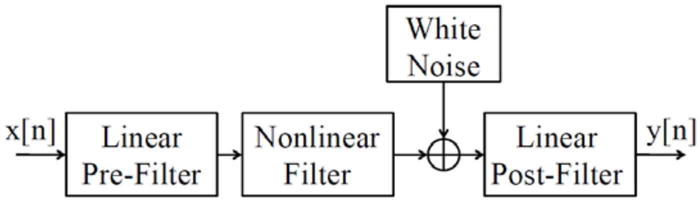




Ref. V. Välimäki *et al*., JAES, 2010.

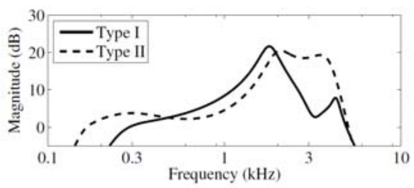
Carbon Microphone Modeling

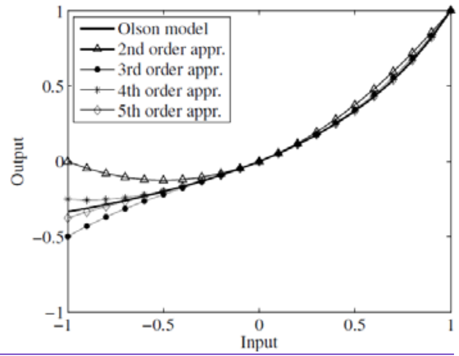
- The sandwich structure is used (Välimäki *et al.*, DAFX book 2e, 2011)



Carbon Microphone Modeling

- 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

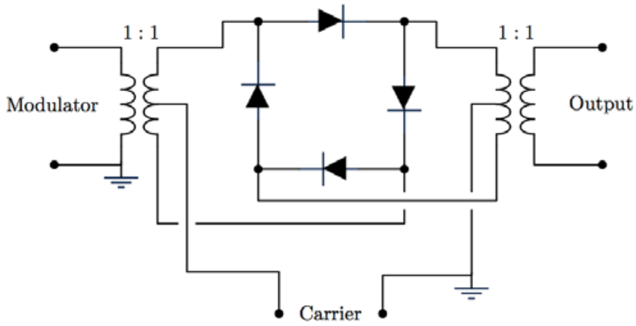
Sample type	Input	Linear & BP filtered	Nonlinear processing	Nonlinear processing + noise
Male Speech				
Female Speech				
Music				

DEMO

Ring Modulator

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

Analog



Ring Modulator

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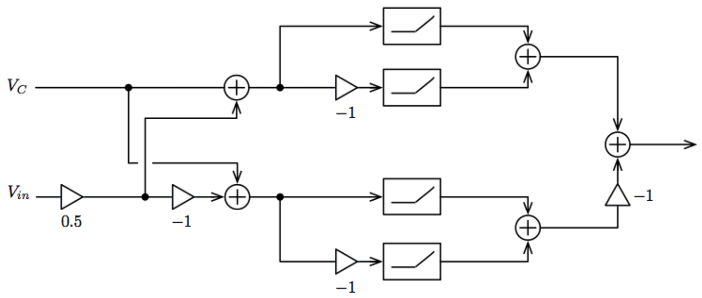
Digital



Ring Modulator

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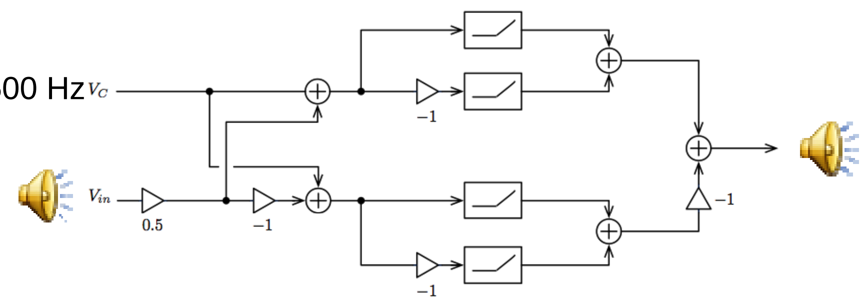
Digital

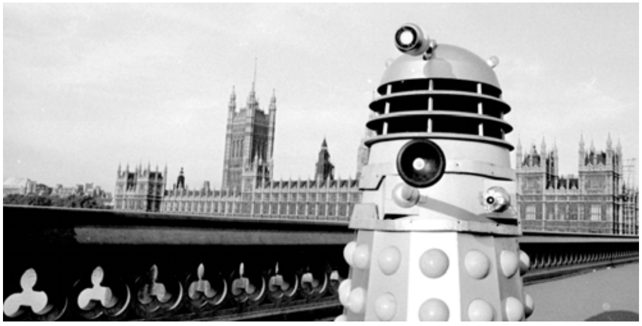
500 Hz



http://www.acoustics.hut.fi/publications/papers/dafx11-ringmod/

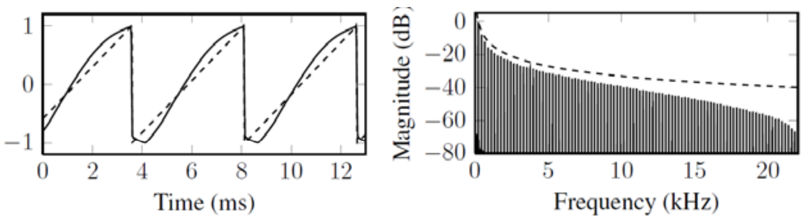
Ring Modulator

- 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)

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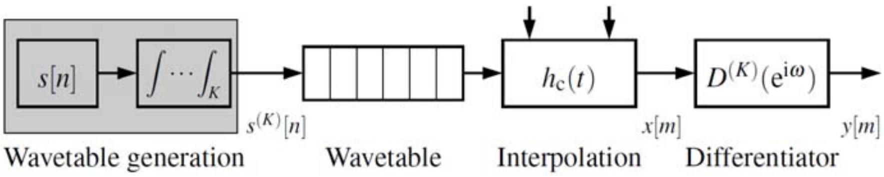
Integrated Wavetable and Sampling Synthesis



- 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?

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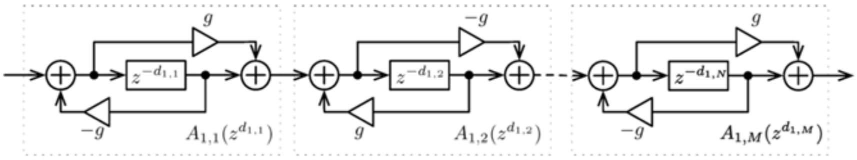
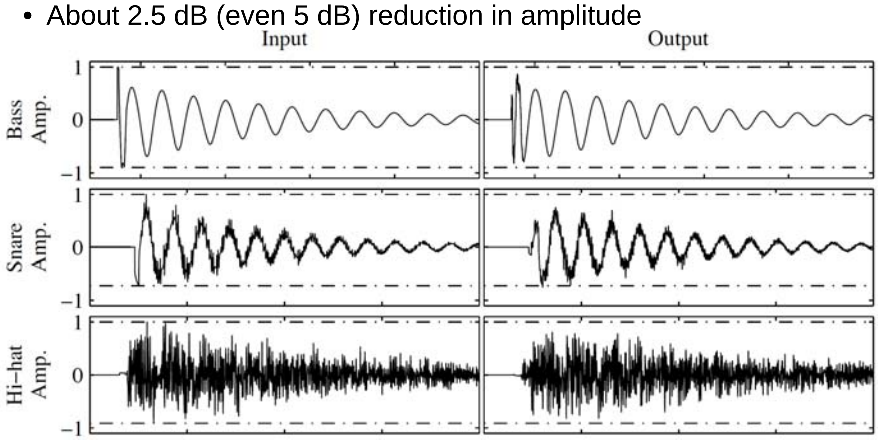
Dynamic Range Reduction using an Allpass Filter Chain

- 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



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

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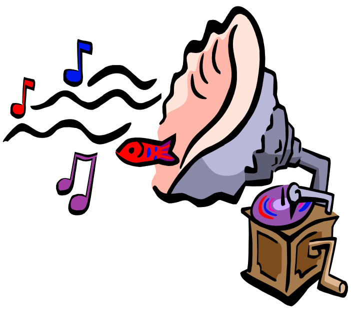
Conclusion

- 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?



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Virtual Analog Modeling

Vesa Välimäki
Aalto University, Dept. Signal Processing and Acoustics
(Espoo, FINLAND)