Sound synthesis with Periodically Linear Time Varying Filters

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Seminários CompMus - 2015/03/23
Linux Audio Conference soon!

- LTV theory approach to distortion techniques
- New synth sounds
- Virtual Analog Oscillators
- Usage as audio effect

- LTV theory approach to distortion techniques
- New synth sounds
- Virtual Analog Oscillators
- Usage as audio effect
- The challenge:

"When I first got some - I won't call it music - sounds out of a computer in 1957, they were pretty horrible. (...) Almost all the sequence of samples - the sounds that you produce with a digital process - are either uninteresting, or disagreeable, or downright painful and dangerous. **It's very hard to find beautiful timbres.**"

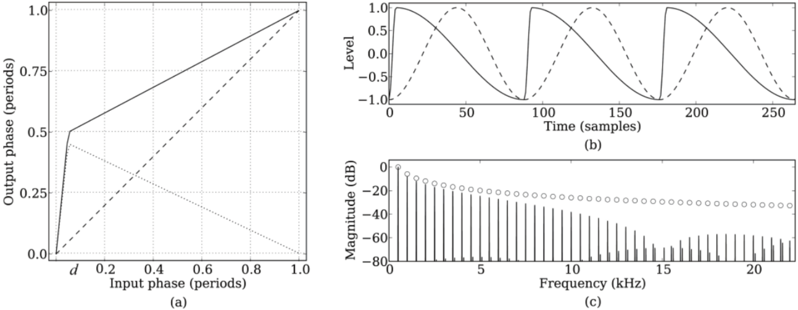
Max Mathews, 2010.

Phaseshaping - US patent 4658691 Casio - CZ

Add a phase distortion function to the regular phase generator
Sawtooth: Inflection point on the regular (dashed) index

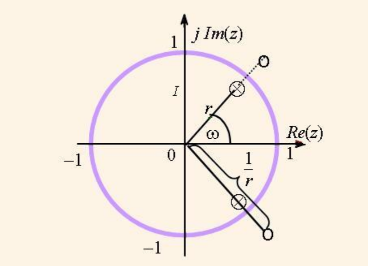
$$g(t) = \begin{cases} 0.5\frac{t}{d}, & 0 \leq t \leq d \\ 0.5\frac{t-d}{1-d} + 0.5, & d < t < 1 \end{cases}$$

For $d = 0.05$



The allpass filter

$$H(z) = \frac{-a + z^{-1}}{1 - az^{-1}}$$



Flat magnitude response

Frequency dependent phase shift
(T.Laakso, V.Valimaki, M.Karjalainen, U.Laine)

$$\phi(\omega) = -\omega + 2\tan^{-1}\left(\frac{-a\sin(\omega)}{1 - a\cos(\omega)}\right)$$

Jussi Pekonen, 2008

Coefficient-modulated first-order allpass filter as distortion effect

- Suggests the method for sound synthesis and audio effects
- Recall that classic PD is restricted to cyclic tables (Adaptive PD requires the delay line)
- Derives stability condition

$$|m(n)| \leq 1 \quad \forall n$$

- Recommends appropriate values for $m(n)$
- Dispersion problem on low frequencies

$$\phi_{DC}(n) = \frac{1 - m(n)}{1 + m(n)}$$

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Spectrally rich phase distortion sound synthesis using allpass filter

Time-varying allpass transfer function

$$H(z, n) = \frac{-m(n) + z^{-1}}{1 - m(n)z^{-1}}$$

Phase distortion

$$\phi(\omega, n) = -\omega + 2\tan^{-1}\left(\frac{-m(n)\sin(\omega)}{1 - m(n)\cos(\omega)}\right)$$

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Spectrally rich phase distortion sound synthesis using allpass filter

Using $tan(x) \approx x$, and knowing $\phi(\omega, n)$

$$m(n) = \frac{-(\phi(\omega, n) + \omega)}{2\sin(\omega) - (\phi(\omega, n) + \omega)\cos(\omega)}$$

Range for the allpass modulation should be $[-\omega, -\pi]$

$$\phi(\omega, t) = \frac{g(t)((1-2d)\pi - \omega)}{(1-2d)\pi} - (1-2d)\pi - \omega$$

Implementation with difference equations

$$y(n) = x(n-1) - m(n)(x(n) - y(n-1))$$

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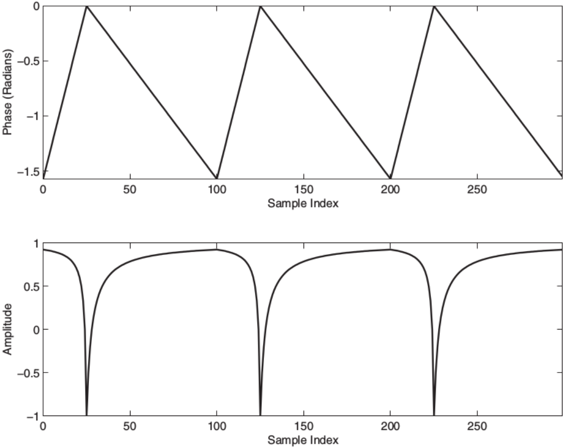
Spectrally rich phase distortion sound synthesis using allpass filter
Phase distortion and coefficient modulation functions



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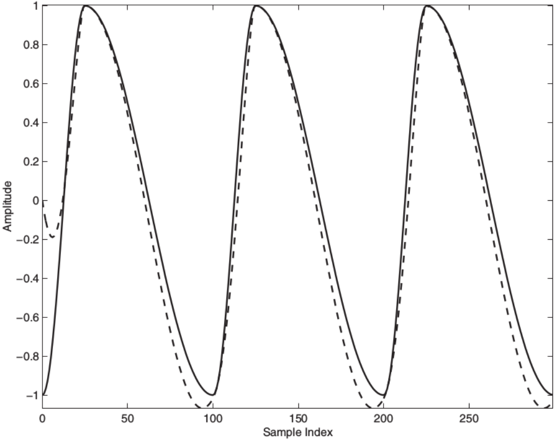
Spectrally rich phase distortion sound synthesis using allpass filter
Outputs with classic PD (solid) and modulated allpass (dashed)



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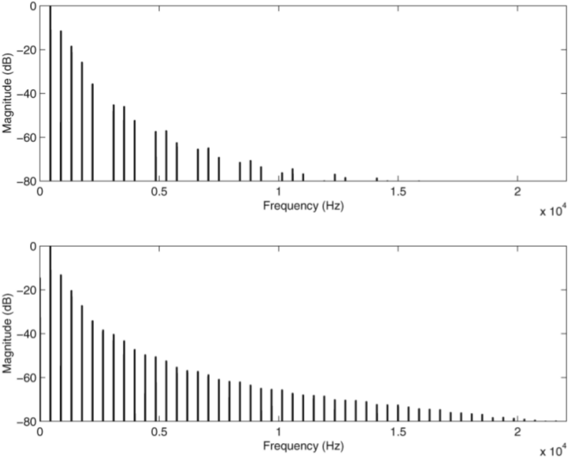
Spectrally rich phase distortion sound synthesis using allpass filter
Classic PD and Modulated allpass spectra



Arbitrary distortion function

$$y(n) = 0.4\cos(f_0) + 0.4\cos\left(2f_0 - \frac{\pi}{3}\right) + 0.35\cos\left(3f_0 + \frac{\pi}{7}\right) + 0.3\cos\left(4f_0 + \frac{4\pi}{3}\right)$$

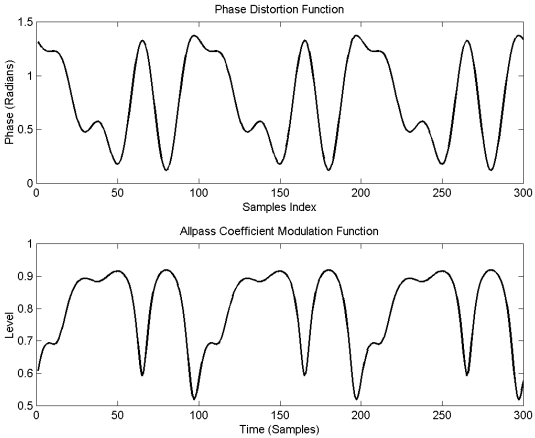
Shift it to the appropriate range

$$y_s(n) = -\frac{\pi}{2}\frac{(y(n)+1)}{2}$$

Technique opens the possibility for coming up with new phase distortion functions and apply them to arbitrary inputs

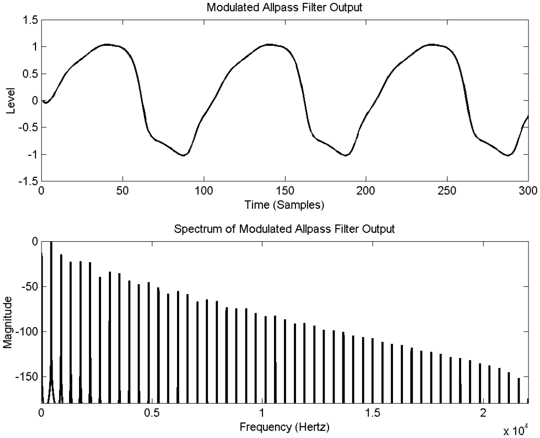
Arbitrary distortion function

Phase distortion and derived modulation functions

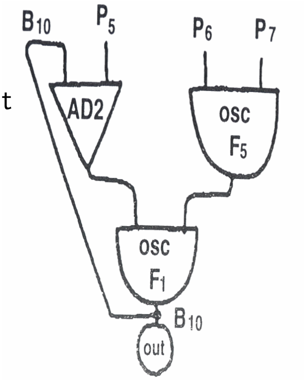


Arbitrary distortion function

Waveform and spectrum



FeedBack Amplitude Modulation



Modulate oscillator amplitude using its output

$y(n) = \cos(\omega_0 n)[1 + \beta y(n-1)]$

with $\omega_0 = 2\pi f_0$ and $y[0] = 0$

LPTV interpretation

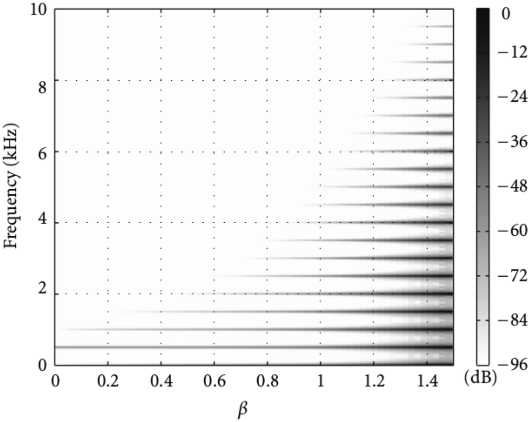
$y(n) = x(n) + \beta a(n) y(n-1)$

$x(n) = a(n) = \cos(\omega_0 n)$
in this case (but could be $\neq$)

1 pole coefficient modulated IIR $\rightarrow$ Dynamic PD

Feedback Amplitude Modulation

β similar to FM's modulation index

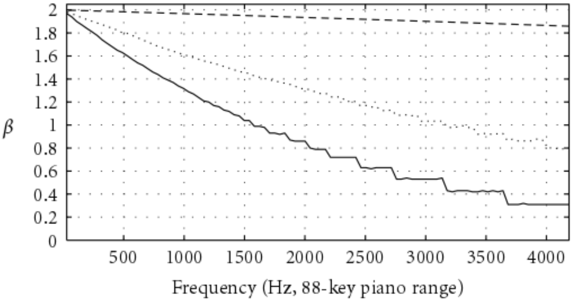


Feedback Amplitude Modulation

Stability condition

$$\left|\beta \prod_{m=1}^{N} \cos\left(\omega_0 m\right)\right| < 1$$

Aliasing before instability

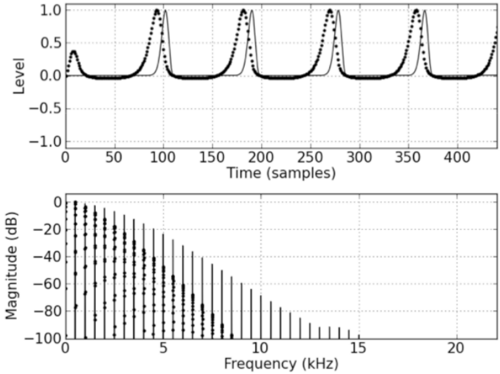


2nd order FBAM

Two previous outputs with individual βs

$$y(n) = \cos(\omega_0 n)[1 + \beta_1 y(n-1) + \beta_2 y(n-2)]$$

Narrower pulse and wider band



Conclusions

- Reissue of a classic technique
- Different kind of implementation
- Enable processing of arbitrary signals
- Studying 2nd and higher order systems stability

Thanks a lot!

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