Complex Nonlinearities for Audio Signal Processing

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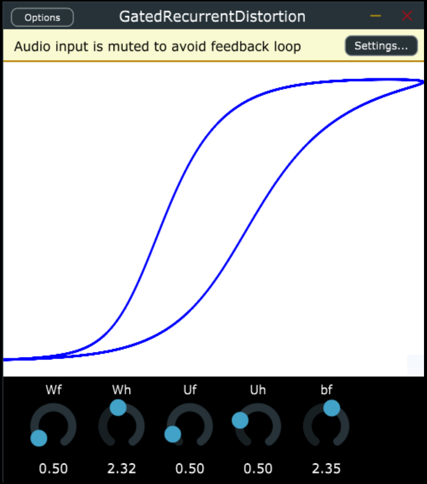
Conclusion

Goals

- Tools for musicians/mixing engineeers
- Inspiration/explanations for audio effect makers
- A academic paper (or two)

Presentation

Audio plugins (VST/AU)



Presentation

Medium articles

M

Upgrade

Complex Nonlinearities Episode 4: Nonlinear Biquad Filters

Jatin Chowdhury
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For today's article, we'll be talking about filters. So far in this series I haven't spoken too much about filters, which might seem odd considering how much of signal processing in general is all about filters. The reason I've avoided filters is that most filters in audio signal processing are implemented as linear processors, and I've been focusing on nonlinear processing concepts.

Presentation

Papers

COMPLEX NONLINEARITIES FOR AUDIO SIGNAL PROCESSING

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ABSTRACT

We present an ongoing study of new and interesting nonlinear structures for audio signal processing, intended to be used for audio effects and synthesis. We give a brief discussion of each structure, and present a series of open-source audio plugins that implement the structures.

1. INTRODUCTION

In digital audio signal processing it is common to find audio effects that use nonlinear elements to add harmonic content to the signal being processed, or to achieve a "distortion" type of effect. Typically, this is done either as part of an analog model, or using a static memoryless nonlinear element.

The goal of this research project is to develop structures for nonlinear audio signal processing that go beyond the traditionally used simple nonlinearities. While the structures developed here may be used for analog modelling and may be inspired by analog effects, they do not come about from direct physical modelling of an analog system, nor do they require knowledge of analog systems such as circuits to be understood and implemented.

1.1. Simple Nonlinearities

We refer to the desired nonlinear structures as "Complex Nonlinearities", as such we should take a moment to define what constitutes a "simple" nonlinearity, particularly since these will make up the building blocks of the complex nonlinearities that follow.

1.1.1. Saturators

The most commonly used nonlinearity in audio signal processing is the saturating nonlinearity, where the input "clips" to a constant value as the input gain increases. This class of nonlinearity includes functions such as the hard clipper, cubic soft clipper, and *tanh* nonlinearities [1], which are described by the following

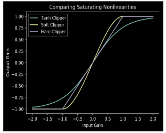

Figure 1: *Saturating Nonlinearities*

1.1.2. Rectifiers

Sometimes for audio effects such as compressors and limiters, it is useful to have a rectified signal (i.e. a signal that only contains non-negative values). The two most simple rectifying nonlinearities are the full-wave rectifier and the half-wave rectifier:

$$f_{\mathrm{FWR}}(x) = |x| \tag{4}$$

$$f_{\mathrm{HWR}}(x) = \begin{cases} 0 & x < 0 \\ x & x \geq 0 \end{cases} \tag{5}$$

The above rectifier equations have a downside in that they do not have continuous derivatives. As a potential alternative, we present another half-wave rectifier equation loosely modelled from a Shockley diode rectifier [2]:

$$f_{\mathrm{Diode}}(x) = \beta\left(e^{\alpha x} - 1\right) \tag{6}$$

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STABLE STRUCTURES FOR NONLINEAR BIQUAD FILTERS

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ABSTRACT

Biquad filters are a common tool for filter design. In this writing, we develop two structures for creating biquad filters with nonlinear elements. We provide conditions for the guaranteed stability of the nonlinear filters, and derive expressions for instantaneous pole analysis. Finally, we examine example filters built with these nonlinear structures, and show how the first nonlinear structure can be used in the context of analog modelling.

1. INTRODUCTION

A "biquad" filter refers to a general 2nd order IIR filter. In digital signal processing, biquad filters are often useful since any higher-order filter can be implemented using a cascade of biquad filters. While digital biquad filters are typically implemented as linear processors, for audio applications it can be useful to implement nonlinear filters. For example, in [1] the authors use a passive model of operational amplifiers to model the nonlinear behaviour of a Sallen-Key lowpass filter. Meanwhile, in [2], the author proposes several methods for using nonlinear elements to enhance linear models of analog ladder filters. More relevant to our current topic is [3], in which the author suggests a method for altering a general digital feedback filter by saturating the feedback path, with the goal of achieving a more analog-like response. In this writing, we strive to develop more general nonlinear filter structures. While these structures may be used for analog modelling, they do not necessarily depend on analog modelling principles to be understood and implemented.

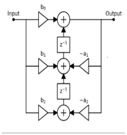

Figure 1: *Transposed Direct Form II*

the quadratic equation.

$$p = \frac{-a_1 \pm \sqrt{a_1^2 - 4a_2}}{2} \tag{2}$$

Specifically, the pole magnitude is described by (ignoring the trivial case where the poles are strictly real):

$$|p|^2 = a_2 \tag{3}$$

And the angular frequencies of the poles are equal to:

$$\angle p = \arctan\left(\pm\frac{\sqrt{4a_2 - a_1^2}}{a_1}\right) \tag{4}$$

Presentation

Links:

- `https://github.com/jatinchowdhury18/ComplexNonlinearities`

- `https://medium.com/@jatinchowdhury18`

Thank you!