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Antialiased Soft Clipping using an Integrated Bandlimited Ramp

Fabián Esqueda, Vesa Välimäki*, and Stefan Bilbao***

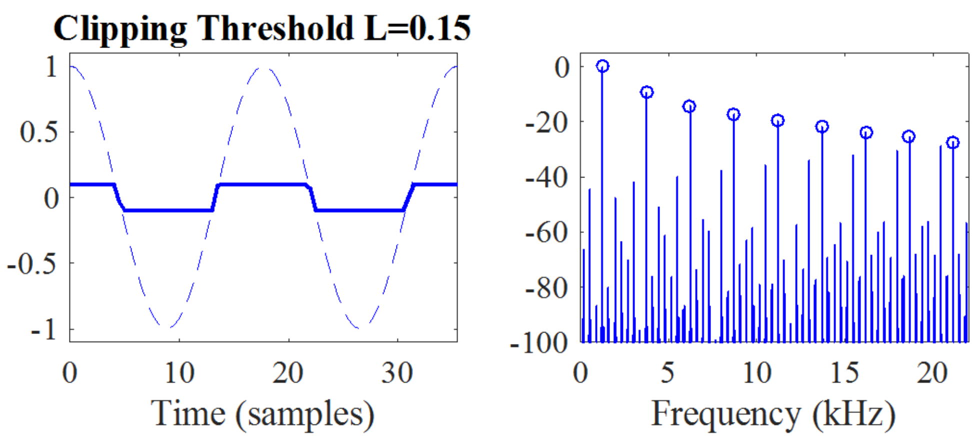
**Dept. Signal Processing and Acoustics, Aalto University, Espoo, FINLAND*
***Acoustics and Audio Group, University of Edinburgh, Edinburgh, UK*

Presentation Outline

1. Introduction
2. Aliasing caused by Soft Clipping
3. The integrated BLAMP function
4. Correction method
5. Examples
6. Conclusions

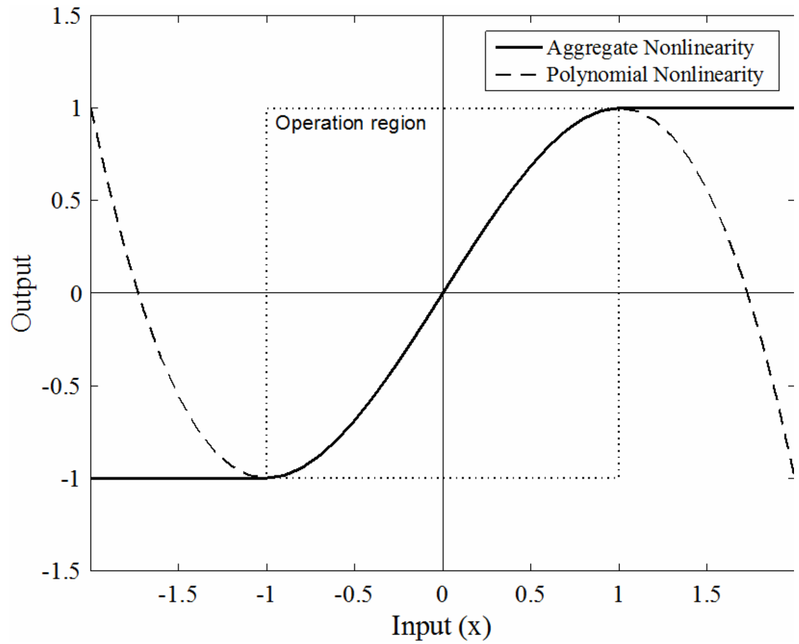
Introduction

- Saturating **memoryless nonlinearities** commonly used to model natural response of systems
- Saturations introduce discontinuities in derivatives of a signal
- Discontinuities require **infinite bandwidth**
- Harmonics exceeding Nyquist cause **aliasing**
- We propose an aliasing reduction technique for piecewise polynomial nonlinearities.



Soft Clipping

- Transition from non-clipped to clipped samples is **gradual**
- Perceived as *softer* and *warmer* than **hard clipping**
- Can be implemented cheaply, e.g. using **piecewise polynomial functions** (Pakarinen and Yeh, 2009)
- At low input values, spectrum expanded by order of polynomial nonlinearity
- Two sources of **aliasing**
  1. Expansion of spectrum due to polynomial nl.
  2. Discontinuity introduced at **clipping point**

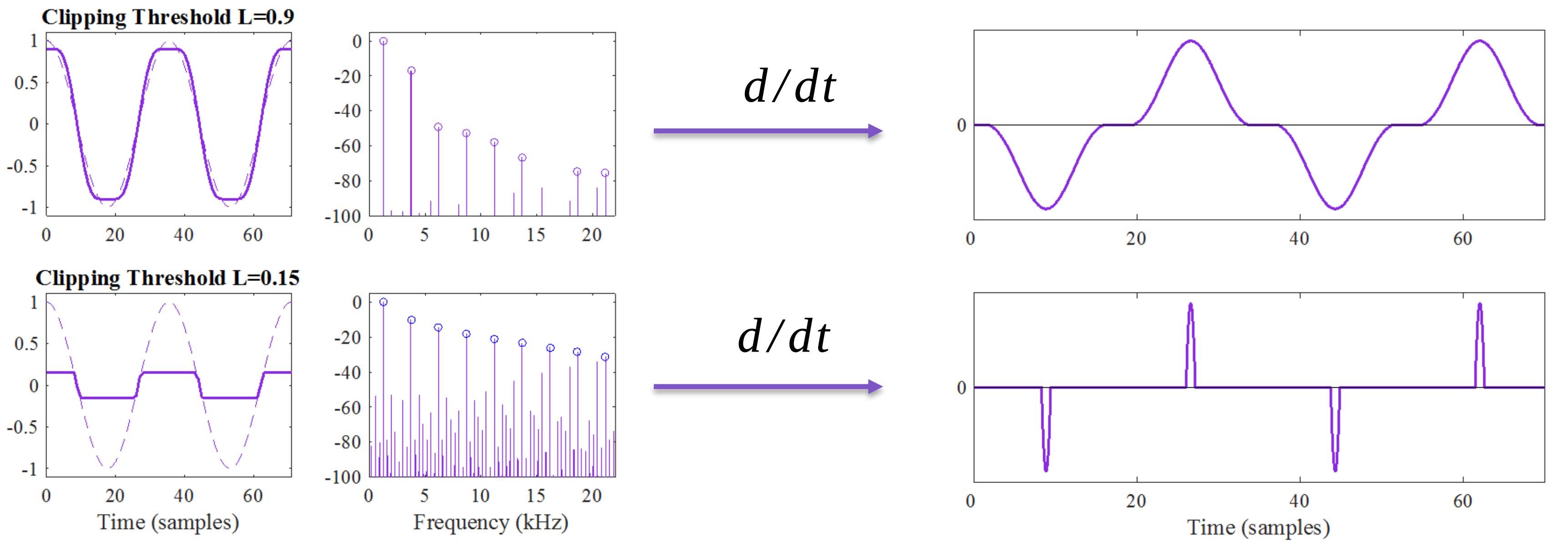


- We consider the third-order polynomial soft-clipper (Araya and Suyama, 1996):

$$c(x) = \begin{cases} \frac{3x}{2} - \frac{x^3}{2} & \text{when } |x| < 1 \\ \mathrm{sgn}(x) & \text{otherwise} \end{cases},$$

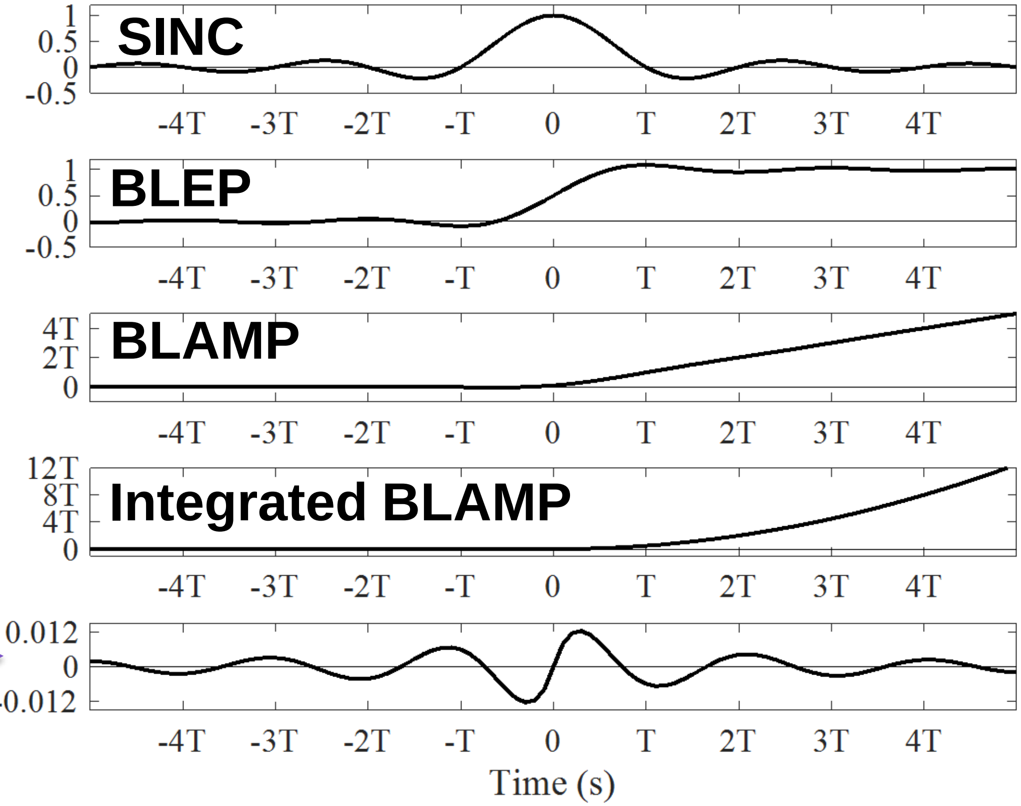
Soft Clipping (cont'd)

- At high clipping thresholds, level of aliasing distortion is negligible. First derivative is smooth
- At low clipping thresholds, clipper introduces **corners** in first derivative
- These corners mean second derivative of signal is **discontinuous** at clipping points



The integrated BLAMP Function

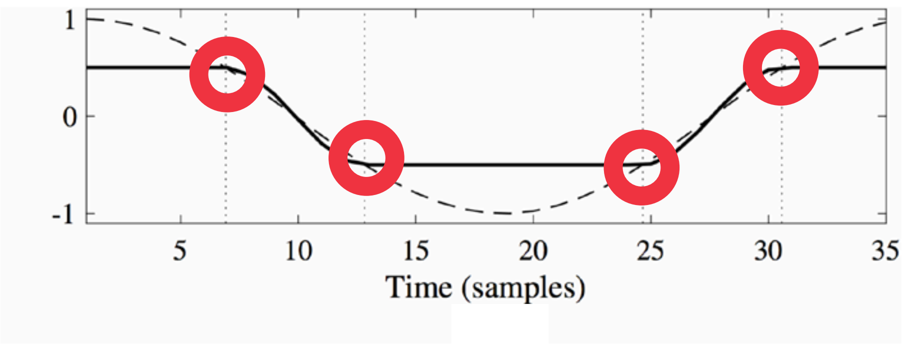
- Derived from **3rd integral** of the **bandlimited impulse**, i.e. the sinc function (Stilson and Smith, 1996).
  BLEP = Bandlimited step
  BLAMP = bandlimited ramp
- Model for bandlimited discontinuity in the **second derivative**
- **Residual function** computed from difference between integrated BLAMP and trivial parabolic ramp



Residual function

$$h^{(3)}(t) = \frac{t^2}{2}\left(\frac{1}{2} + \frac{1}{\pi}\mathrm{Si}(\pi f_s t)\right) + t\frac{\cos(\pi f_s t)}{2\pi^2 f_s} + \frac{\sin(f_s \pi t)}{2\pi^3 {f_s}^2}$$

The Integrated BLAMP Correction Method



1. Detect clipping points (fractional delays)

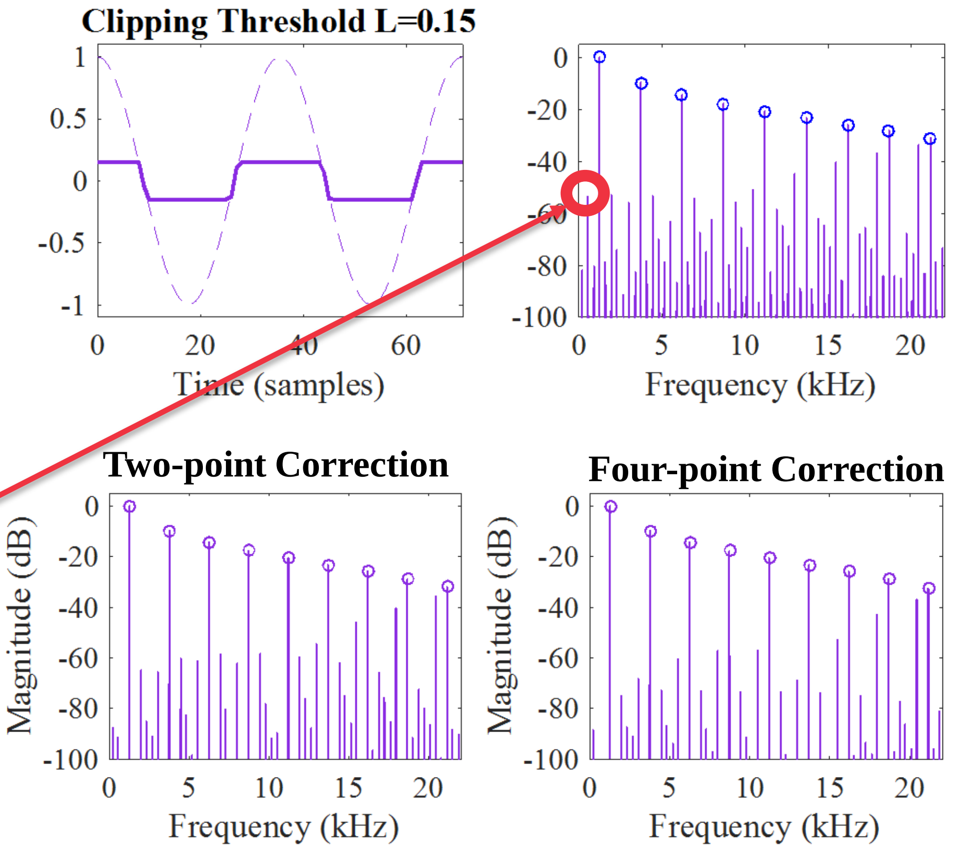
2. Center residual function at clipping points

3. Scale by value of second derivative and adjust polarity

4. Sample at neighboring sample points and add to original signal

Example

- Fixing only a few samples at every clipping point makes a big difference!
- Input signal: 1245-Hz sinewave, clipping threshold L=0.15
- Correction particularly efficient at low frequencies, where it is most audible
- For example: Alias @ 524 Hz
  - ✓ Trivial: –50 dB
  - ✓ 2-pt correction: –92 dB
  - ✓ **4-pt correction: –104 dB**



Sound Examples

Clean guitar riff

Log clipped chirp aliased

Guitar riff after soft clipping and antialiasing

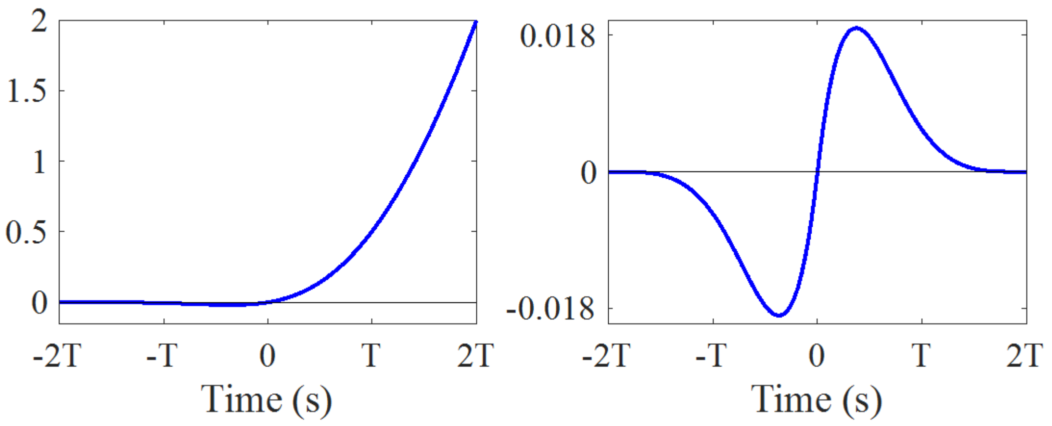
Log clipped chirp antialiased

Extracted aliases

Extracted aliases

Polynomial Approximation

- Polynomial approximation of integrated BLAMP function can be used for efficiency
- Derived from the four-point Lagrangian interpolation kernel
- Residual function is computationally cheap and has **finite support**
- Defined in terms of d, **fractional delay** required to center at each clipping point



Span	Four-point integrated polyBLAMP residual
$[-2T, T]$	$d^6/720 - d^4/144$
$[-T, 0]$	$-d^6/240 + d^5/120 + d^4/24 - d^2/48 - 7d/360 - 1/180$
$[0, T]$	$d^6/240 - d^5/60 - d^4/48 + d^3/6 - d^2/4 + 11d/90$
$[T, 2T]$	$-d^6/720 + d^5/120 - d^4/72 + d^2/48 - 7d/360 + 1/180$

Conclusions

- Integrated BLAMP effectively reduces aliases caused by **discontinuities in second derivative**
- These discontinuities may be introduced during synthesis or processing of arbitrary audio signals, e.g. soft clipping
- Aliasing components attenuated by up to 50 dB at low frequencies
- Its polynomial approximation provides efficient solution

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Aliasing Reduction in Clipped Signals

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Aliasing Reduction in Clipped Signals

Fabián Esqueda, Stefan Bilbao, *Senior Member, IEEE*, and Vesa Välimäki, *Fellow, IEEE*

***Abstract*—An aliasing reduction method for hard-clipped sampled signals is proposed. Clipping in the digital domain causes a large amount of harmonic distortion, which is not bandlimited, so spectral components generated above the Nyquist limit are reflected to the baseband and mixed with the signal. A model for an ideal bandlimited ramp function is derived, which leads to a postprocessing method to reduce aliasing. A number of samples in the neighborhood of a clipping point in the waveform are modified to simulate the Gibbs phenomenon. This novel method requires estimation of the fractional delay of the clipping point between samples and the first derivative of the original signal at that point. Two polynomial approximations of the bandlimited ramp function are suggested for practical implementation. Validation tests using sinusoidal, triangular, and harmonic signals show that the proposed method achieves high accuracy in aliasing reduction. The proposed 2-point and 4-point polynomial correction methods can improve the signal-to-noise ratio by 12 and 20 dB in average, respectively, and are more computationally efficient and cause less latency than oversampling, which is the standard approach to aliasing reduction. An additional advantage of the polynomial correction methods over oversampling is that they do not introduce overshoot beyond the clipping level in the waveform. The proposed techniques are useful in audio and other fields of signal processing where digital signal values must be clipped but aliasing cannot be tolerated.**

***Index Terms*—Antialiasing, interpolation, nonlinear distortion, signal denoising, signal sampling.**

I. Introduction

Clipping is a form of distortion that limits the values of a signal that lie above or below certain threshold. In practice, signal clipping may be necessary due to system limitations, e.g. to avoid overmodulating an audio transmitter. In discrete systems, it can be caused unintentionally due to data resolution constraints, such as when a sample exceeds the maximum value that can be represented, or intentionally as when simulating a process in which signal values are constrained. Clipping is a nonlinear operation and introduces frequency components not present in the original signal. In the digital domain, when the frequencies of these new components exceed the Nyquist limit, the components are reflected back into the baseband, causing *aliasing*. This paper proposes a novel method to reduce aliasing in clipped signals.

Aliasing can severely affect the quality of a digital signal by corrupting the data it represents. For example, in audio applications, aliasing can cause severe audible effects such as beating, inharmonicity and heterodyning [1]. Nevertheless, if the aliased components are sufficiently attenuated, their effects become inaudible and can therefore be neglected [1], [2].

A large share of earlier research on clipped signals has focused on *declipping* or the reconstruction of the underlying original unclipped signal. Abel and Smith [3] introduced optimization methods to reconstruct the clipped samples based on constraints. Recent work on declipping has considered methods based on matching pursuits [4], compressed sensing [5], social sparsity [6], sparse and co-sparse regularization [7], and non-negative matrix factorization [8]. Declipping can improve the audio quality of nonlinearly-compressed sound files [9] and help speaker recognition [10], for example.

The purpose of this work is not signal reconstruction but enhancement, by allowing clipping to occur and by suppressing the aliasing introduced. Practical applications for the proposed approach are found in systems in which clipping is implemented digitally. One application in radio broadcasting and music production is the limiting of the maximum values of the signal, such as in dynamic range compressors and limiters, which are known to introduce distortion and aliasing [11]–[13]. In such applications, declipping is out of question, because it would cancel the limiting effect, which is necessary for maximizing the signal level. However, the antialiasing method proposed in this work can be useful, since it cleans the clipped signal by suppressing the aliasing, and restores the limited distribution of sample values, as required.

Other practical applications involving digital clipping are simulations of analog and physical systems in which signal values are naturally limited. In digital simulation of analog filters, hard clipping is used as a simple model for the saturation of large signals inside analog filters [14], [15]. In the digital modeling of vacuum-tube amplifiers and guitar effects, the saturating characteristics must also be implemented digitally [16]–[18]. Often hard clipping is used in connection with soft clipping so that the latter works at small signal values while the former saturates (clips) the large signal values to a maximum or minimum value. Antialiasing for the combination of a soft and a hard clipper in this context was the first application for the method discussed in this paper [19]. Similar saturating modules appear in physical models of musical wind and string instruments in which a saturating waveshaper simulates the nonlinear interaction between the excitation and state of an acoustic resonator, such as a tube or a string [20], [21].

Full-wave and half-wave rectification are special cases of hard clipping. Thus, the proposed method can enhance such digitally rectified signals, which have applications in various fields. For

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F. Esqueda and V. Välimäki are with the Department of Signal Processing and Acoustics, Aalto University School of Electrical Engineering, FI 00076 AALTO, Espoo, Finland (e-mail: fabian.esqueda@aalto.fi; vesa.valimaki@aalto.fi).

S. Bilbao is with the Acoustics and Audio Group, University of Edinburgh, Edinburgh EH9 3JZ, U.K. (e-mail: sbilbao@staffmail.ed.ac.uk).

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