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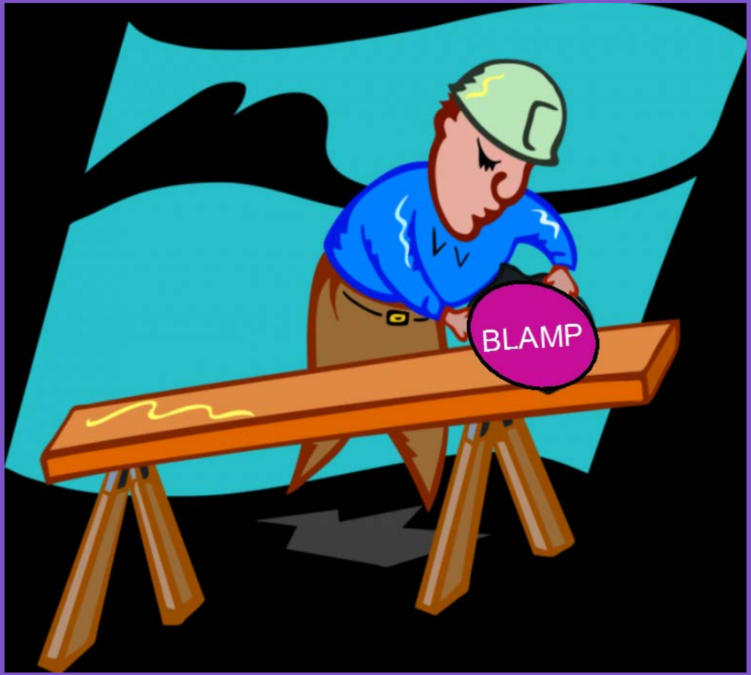
Sound check

Rounding Corners with BLAMP



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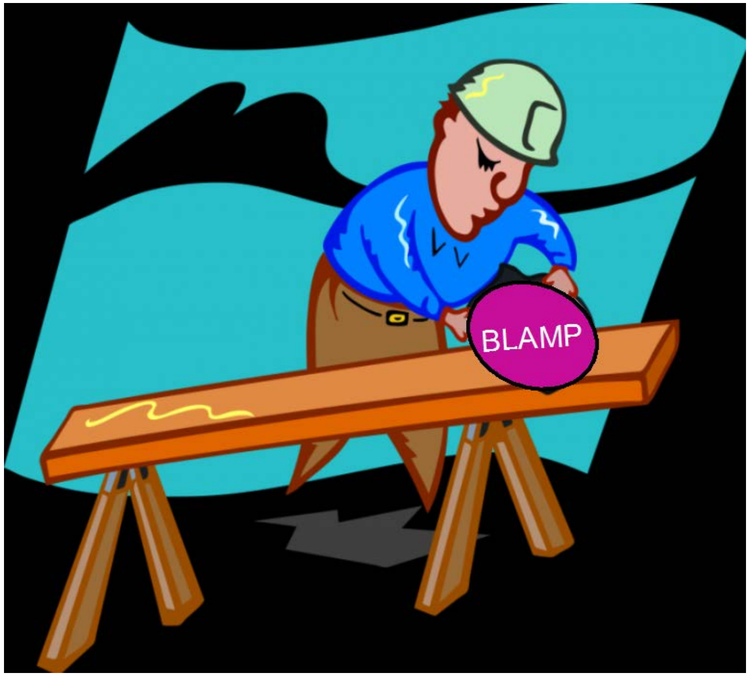
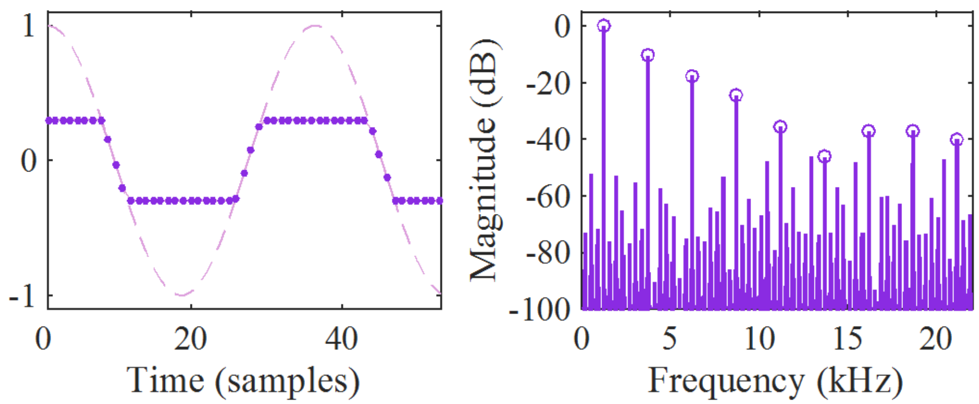
Presentation Outline

1. Introduction
2. The BLAMP
3. Case Studies
   - Hard Clipping
   - Half/Full-Wave Rectification
4. The PolyBLAMP
5. Conclusions



Introduction

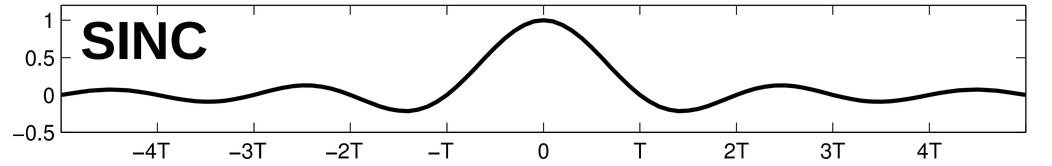
- Discontinuities in a signal require infinite bandwidth
- Discontinuities in first derivative (**corners**) also require infinite bandwidth
- Sampling these signals trivially causes **aliasing**
- Operations that cause corners include hard **clipping** and signal **rectification**
- The **bandlimited ramp (BLAMP)** function can bandlimit these corners



The BLAMP Function

- Derived from **2^{nd} integral** of the **bandlimited impulse**, i.e. the sinc function (Stilson and Smith, 1996)
  (BLIT = Bandlimited Impulse Train)
- 1^{st} integral: BLEP = Bandlimited step (Brandt, 2001)
- BLAMP model for bandlimited discontinuity in the **first derivative**
- **Residual function**
  = BLAMP – trivial ramp

SINC



BLEP

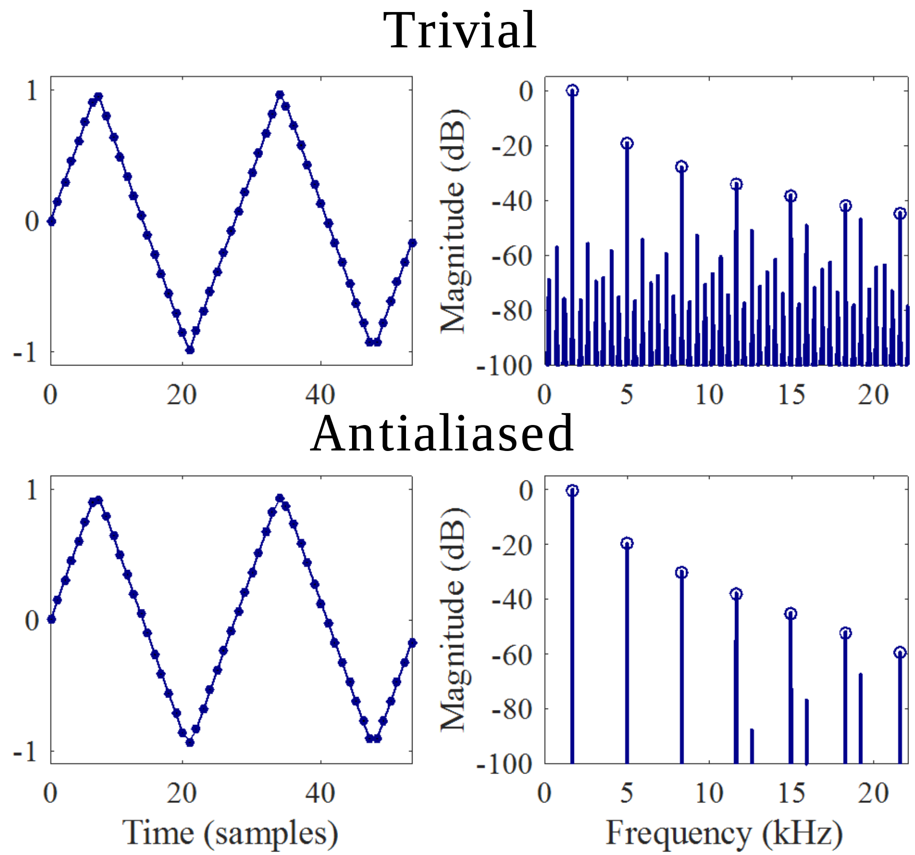
BLAMP

BLAMP residual

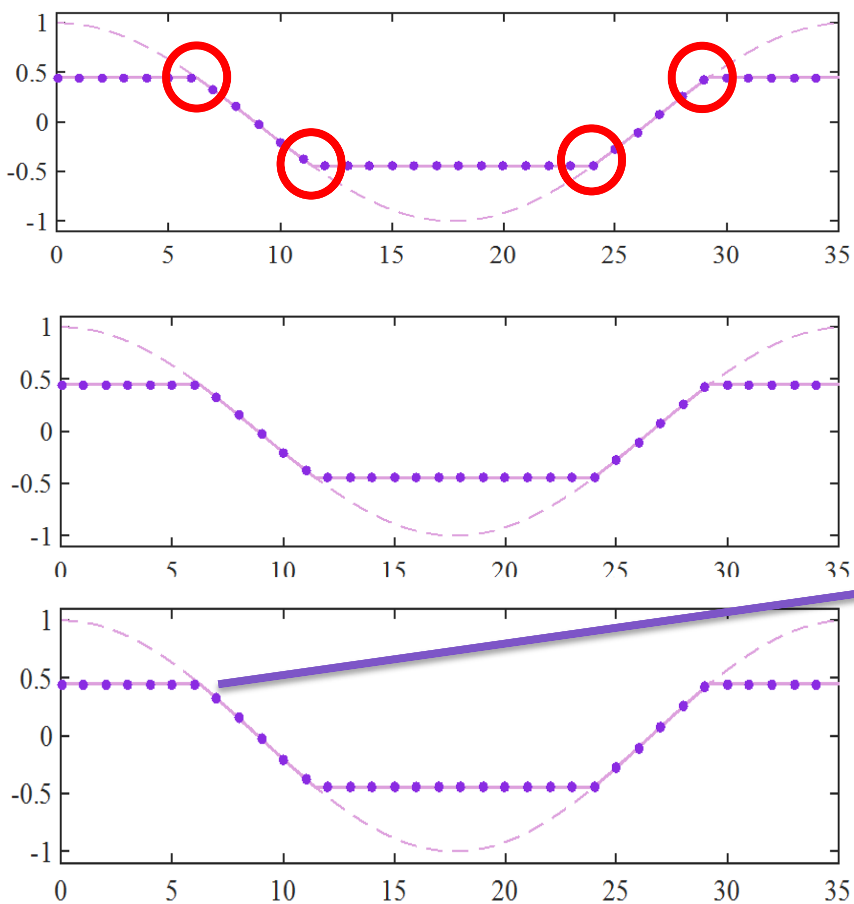
$$h^{(2)}(t) = t\left[\frac{1}{2} + \frac{1}{\pi}\mathrm{Si}(\pi f_s t)\right] + \frac{\cos(\pi f_s t)}{\pi^2 f_s}$$

Triangular Oscillator Waveform

- Historically, **BLAMP** proposed to antialias triangular oscillator waveforms (Huovilainen, 2010)
- Aliasing in trivial triangular signals only audible at high fundamental frequencies (> 2 kHz)
- Alternative methods include DPW (Välimäki et al., 2010) and E-PTR algorithms (Ambrits and Bank, 2013)

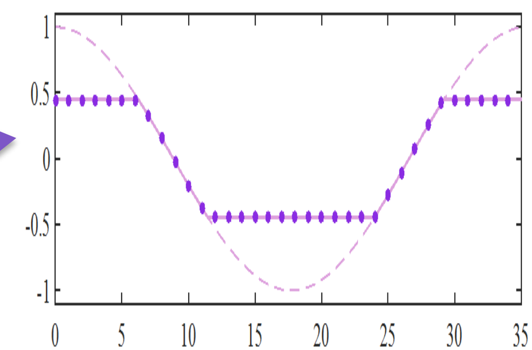


The BLAMP Correction Method



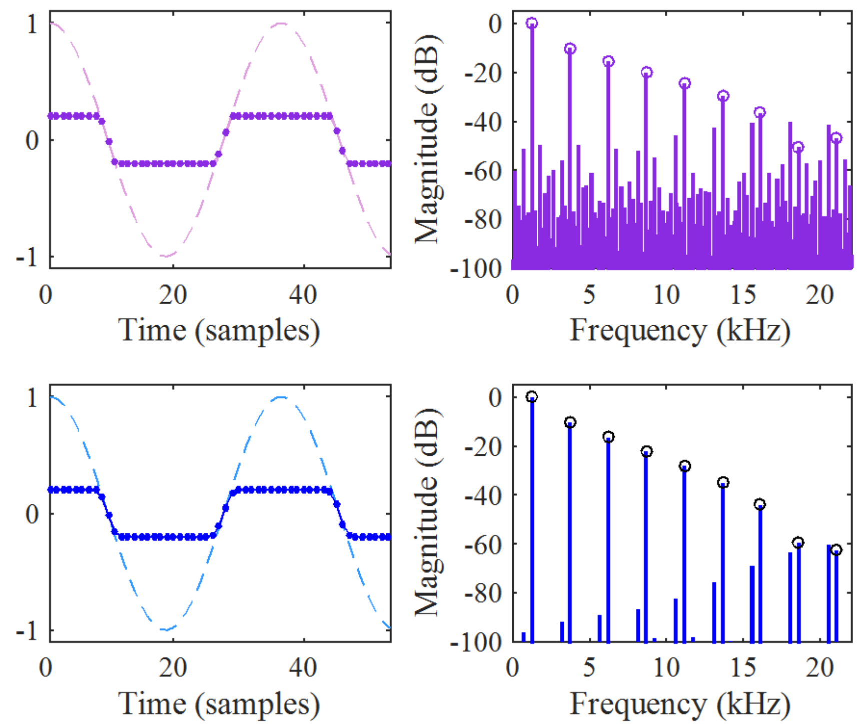
Example: Hard Clipped Sinewave

1. Detect corners (fractional delay, slope)
2. Center BLAMP residual function at corners
3. Scale by slope of signal at corners and adjust polarity
4. Sample at neighboring sample points and add to original signal



Microscopic changes in waveform cause dramatic improvements in audio quality!

Hard Clipping

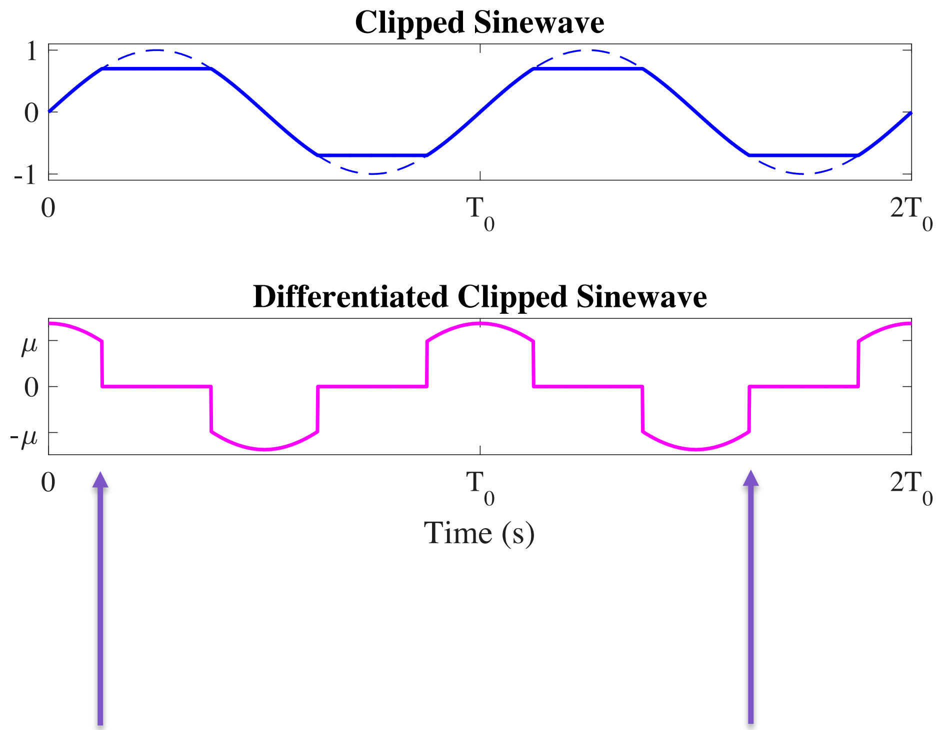


Example: 1-kHz clipped sinewave

$f_s = 44.1\ kHz$

Original	Clipped	Clipped AA	Residual
Logarithmic clipped sine sweep			
Clipped flute recording			
Clipped guitar recording			

Hard Clipping



Falling Discontinuity Rising Discontinuity

- Hard clipping introduces discontinuities in the **first derivative**
- **Magnitude** and **direction** of discontinuities depend on **slope** of the signal at each corner
- Corner **slope** and associated **fractional delay** value can be found e.g. using linear or higher order interpolation
- Fractional delay used to center BLAMP around **clipping boundaries**

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Aliasing Reduction in Clipped Signals

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Aliasing Reduction in Clipped Signals

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***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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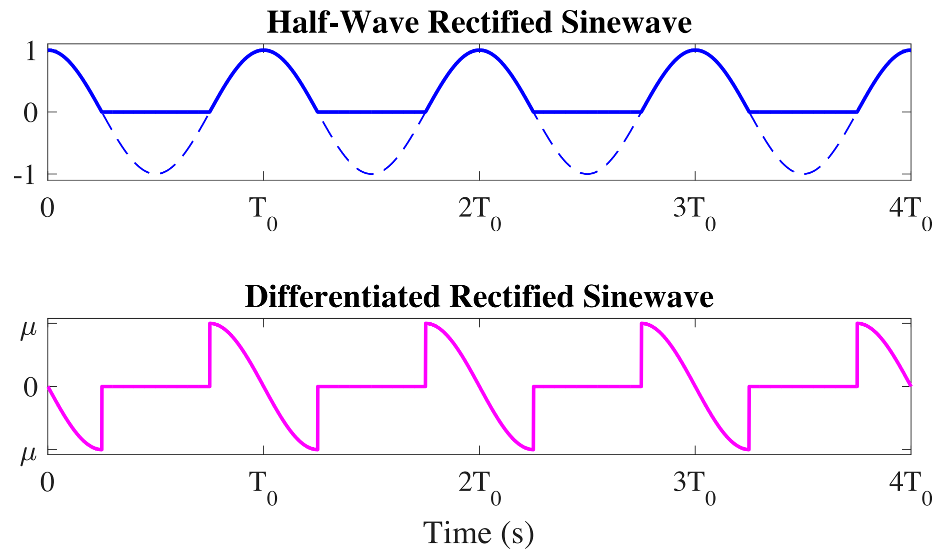
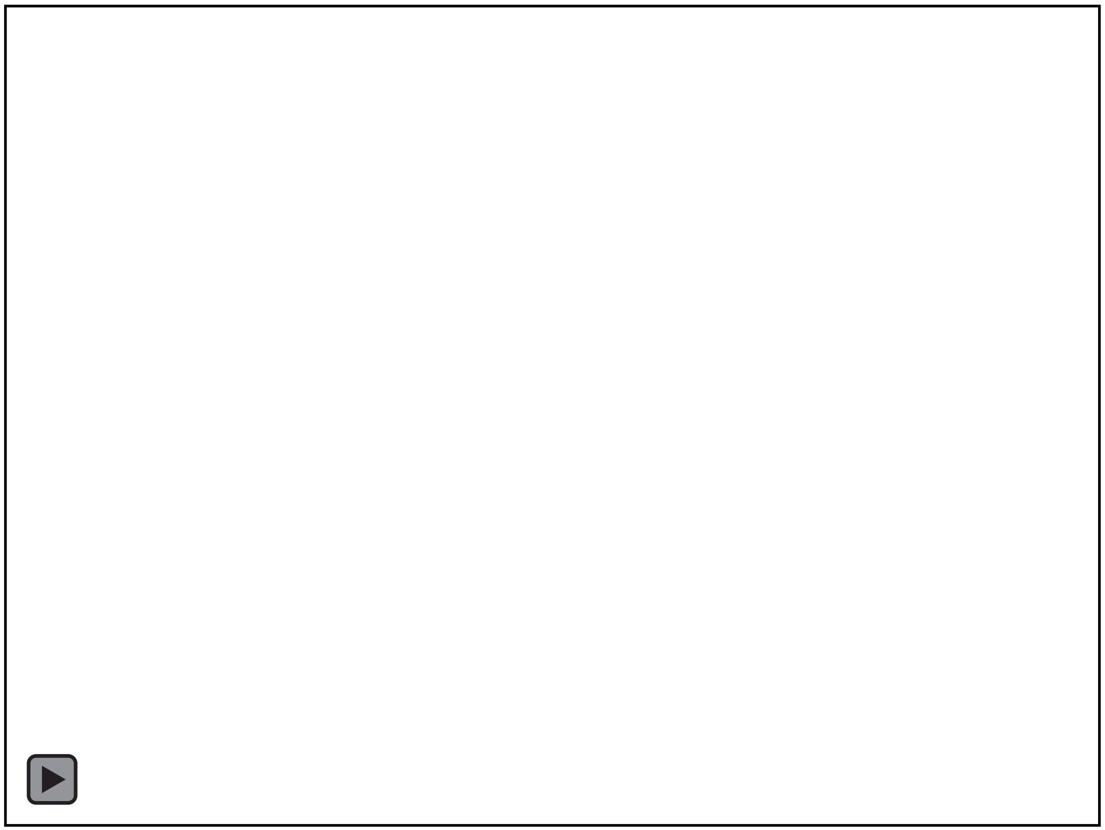
Half-Wave Signal Rectification

- Special case of asymmetric hard clipping
- Half-wave rectification **blocks** negative portions of the input signal
- Introduces **corners** at zero-crossings
- Magnitude of each discontinuity determined by slope at zero-crossings
- BLAMP must be scaled by slope at **corners**
- **Rising discontinuities** only



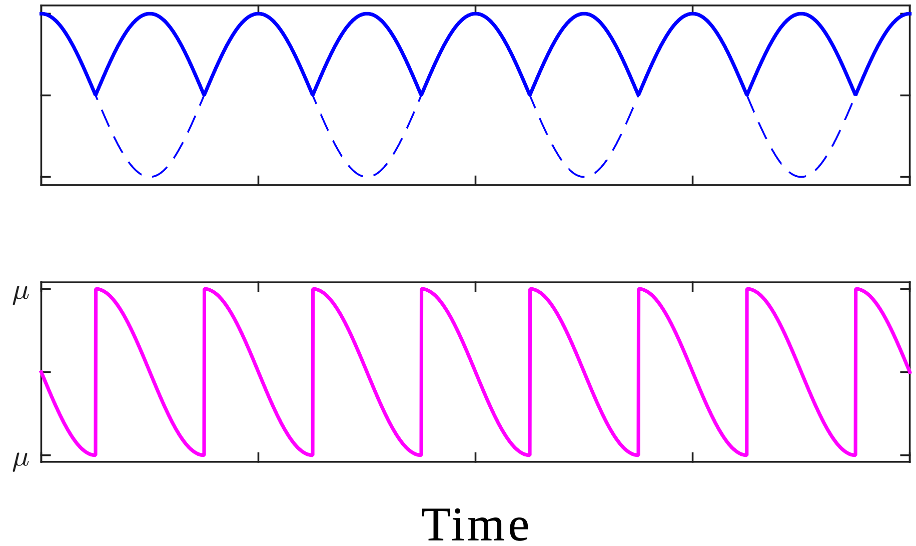
Half-Wave Signal Rectification

Animation: Half-wave rectified sine sweep before & after BLAMP correction



Full-Wave Signal Rectification

- Full-wave rectification **inverts** the negative portions of the input signal
- Modifies **timbre and f_0** of input signal
- Introduces **corners** at zero-crossings
- Large discontinuities in the first derivative of the signal
- BLAMP residual must be scaled by **twice the slope** at zero-crossings
- **Rising discontinuities** only



Full-Wave Signal Rectification

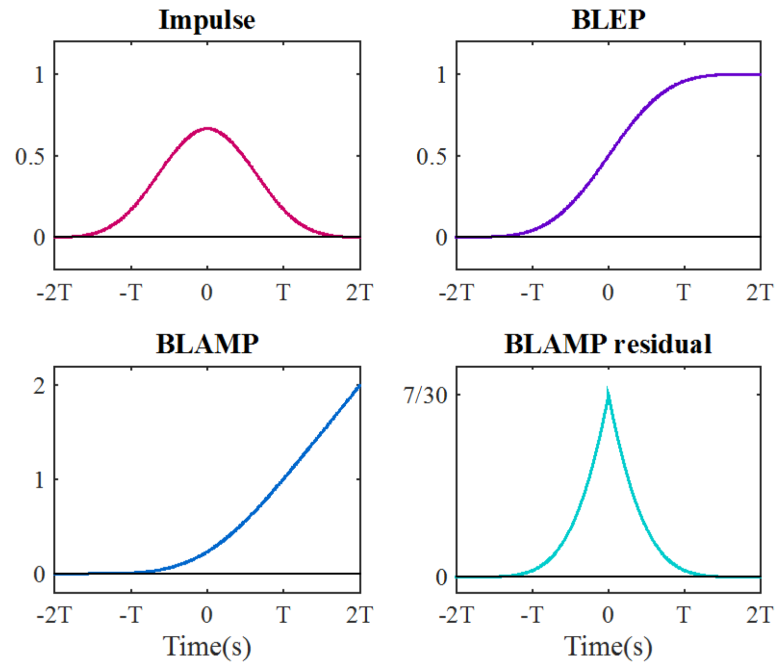
Animation: Full-wave rectified sine sweep before & after BLAMP correction

Trivial

Antialiased

Polynomial Approximation (polyBLAMP)

- PolyBLAMP is recommended for efficient implementation of the BLAMP method
- Derived from the four-point B-spline interpolation kernel
- Residual function is computationally cheap and has **finite support**
- Defined in terms of **fractional delay** *d* required to center at each corner
- Nonnegative, does not introduce overshoot!



Span	Four-point polyBLAMP residual
$[-2T, T]$	$d^5/120$
$[-T, 0]$	$-d^5/40 + d^4/24 + d^3/12 + d^2/12 + d/24 + 1/120$
$[0, T]$	$d^5/40 - d^4/12 + d^2/3 - d/2 + 7/30$
$[T, 2T]$	$-d^5/120 + d^4/24 - d^3/12 + d^2/12 - d/24 + 1/120$

Results: Signal-to-Noise Ratio

Signal	Trivial	OSx2	OSx4	polyBLAMP
Clipping A6	34 dB	42 dB	43 dB	**57 dB**
Clipping C8	24 dB	34 dB	38 dB	**42 dB**
Half-W. Rec. A6	40 dB	43 dB	44 dB	**61 dB**
Half-W. Rec. C8	28 dB	36 dB	38 dB	**48 dB**
Full-W. Rec. A6	32 dB	40 dB	41 dB	**53 dB**
Full-W. Rec. C8	20 dB	28 dB	30 dB	**39 dB**

- PolyBLAMP compared against trivial approach and oversampling by factors 2 and 4

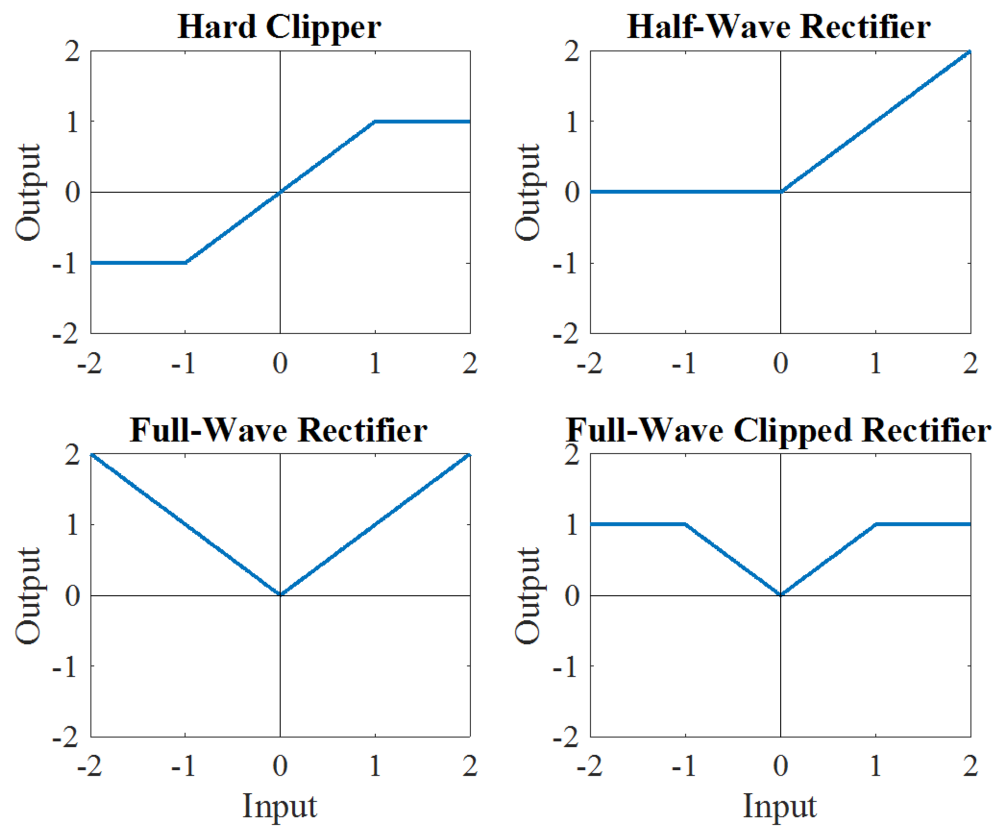
Results: Computation Time

Signal	OSx2	OSx4	polyBLAMP
Clipping C8	46 ms	102 ms	**43 ms**
Half-W. Rec. C8	40 ms	89 ms	**18 ms**
Full-W. Rec. C8	41 ms	90 ms	**28 ms**

- Average computation time for a 1-second signal using a Python implementation

Conclusions

- The BLAMP method reduces aliasing caused by **corners** in signals
- Helpful in synthesis or processing of arbitrary signals
- PolyBLAMP is better and faster than oversampling by factors 2 and 4
- Several applications in **memoryless** nonlinear processing



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