A NEW METHOD FOR DESIGN AND ANALYSIS OF ANTIALIASING FILTER USING ANTI-DERIVATIVE APPROACH FOR STATIC NON-LINEARITIES

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ABSTRACT—Aliasing is a commonly-encountered problem in audio signal processing, particularly when memory less nonlinearities are simulated in discrete time. A conventional remedy is to operate at an oversampled rate. A new aliasing reduction method is proposed here for discrete-time memory less nonlinearities, which is suitable for operation at reduced oversampling rates. The method employs higher order anti derivatives of the nonlinear function used. The first order form of the new method is equivalent to a technique proposed recently by Parker et al. Higher order extensions offer considerable improvement over the first anti derivative method, in terms of the signal-to-noise ratio. The proposed methods can be implemented with fewer operations than oversampling and are applicable to discrete-time modelling of a wide range of nonlinear analog systems.

I. INTRODUCTION

Speech analysis is the study of the speech production mechanism in order to generate a mathematical model of the physical phenomena. The study of coding endeavours to store speech information for subsequent recovery.. The generation of speech from coded instructions is known as speech syn Project.

The earliest attempts to build systems for Automatic Speech Recognition (ASR) were made in 1950s based on acoustic phonetics. These systems relied on spectral measurements ,using spectrum analysis and pattern matching to makerecognitiondecisions,ontaskssuchasvowelrecognition[Forgie.J.W and Forgie. C. D, 1959]. Filter bank analysis was also utilized in some systems to provide spectral information. In the 1960s, several basic ideas in speech recognition emerged. Zero-crossing analysis and speech segmentation were used, and dynamic time aligning and tracking ideas were proposed [Reddy. D. R, 1966]. In the 1970s, speech recognition research achieved major milestones. Tasks such as isolated word recognition became possible using Dynamic Time Warping (DTW). Linear Predictive Coding (LPC) was extended from speech coding to speech recognition systems based on LPC spectral parameters. IBM initiated the effort of large vocabulary speech recognition in the 70s [Rabiner. L.R and Juang. B. H, 1993], which turned out to be highly successful and had a great impact in speech recognition research. Also, AT&T Bell Labs began making truly speaker-independent speech recognition systems by studying clustering algorithms for creating speaker- independent patterns [Rabiner. L. R, Levinson.S.E, *et. al*.,1979]. In the 1980s, connected word recognition systems were devised based on algorithms that concatenated isolated words for recognition. An important turning point was the transition of approaches from template-based to statistical modeling – especially the Hidden Markov Model (HMM) approach [Rabiner. L. R, 1989]. HMMs were not widely used in speech application until the mid-1980s. From then on, almost all speech research has involved using the HMM technique. In the late 1980s, neural networks were also introduced to problems in speech recognition as a signal classification technique. In spite of these efforts, machines are still far inferior to humans in speech recognition capabilities.

II.LITERATURE REVIEW

Speech production is an extremely complex process. It can be described in basic terms as follows. The lungs generate air pressure and this pressure wave is modulated as it flows through the larynx. Today a number of nonlinear effects in the speech production process are known. Firstly, it has been accepted for some time that the vocal tract and the vocal folds do not function independently of each other, but that there is in fact some form of coupling between them when the glottis is open. This can cause significant changes in formant characteristics between open and closed glottis cycles T eager have claimed that voiced sounds are characterised by highly complex airflows in the vocal tract, rather than well behaved laminar flow. Turbulent flow of this nature is also accepted to occur during unvoiced speech, where the generation of sound is due to a constriction at some point in the vocal tract.

During the 1960s, a few essential ideas in speech recognition rose. Zero-intersection analysis and speech division were utilized, and dynamic time adjusting and following thoughts were proposed. During the 1970s, speech recognition research accomplished real achievements. Assignments, for example, detached word recognition wound up conceivable utilizing Dynamic Time Warping (DTW).

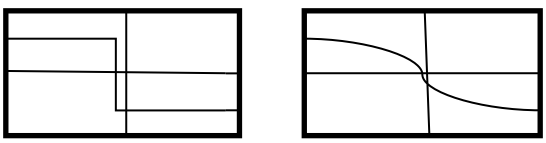
Linear Predictive Coding (LPC) was extended from speech coding to speech recognition systems based on LPC spectral parameters. IBM initiated the effort of large vocabulary speech recognition in the 70s [Rabiner. L.R and Juang. B. H, 1993], which turned out to be highly successful and had a great impact in speech recognition research. Also, AT&T Bell Labs began making truly speaker-independent speech recognition systems by studying clustering algorithms for creating speaker- independent patterns An important turning point was the transition of approaches from template-based to statistical modelling – especially the Hidden Markov Model (HMM) approach .HMMs were not widely used in speech application until the mid-1980s. From then on, almost all speech research has involved using the HMM technique. In the late 1980s, neural networks were also introduced to problems in speech recognition as a signal classification technique.

III.PROPOSED SYSTEM

In this Paper, we present a new discrete-time aliasing reduction method for memory less nonlinearities, based on discrete differentiation of higher order anti derivatives. The first-order method is equivalent to that in . However, the use of higher anti derivatives leads to increasing levels of aliasing suppression; such methods are distinct from aliasing suppression may be understood, intuitively, in terms of operation over increasingly smoothed forms of the nonlinearity. The proposed idea of differentiating anti derivatives is related to previous anti aliasing syn Project methods called the differential polynomial waveform and integrated wavetable/sampling syn Project. This Project deals with memory less nonlinearities of the form

$$y(t) = F0((x(t))$$

Here, x (t) is an input signal, and y(t) is an output signal; both are assumed defined for t∈ R. F0 is a real-valued mapping,



(a) (b)

Fig. 1.Input–output relationships for (a) the hard clipper and (b) the hyperbolic tangent.

Assumed continuous, but not necessarily differentiable, and in most cases of interest is nonlinear. Two typical examples of memory less nonlinear mappings are the saturator,

$$F0(x) = \frac{1}{2}(|x+1| - |x-1|)$$

Defined by the soft-clipping nonlinearity, defined by

$$F0(x) = \tanh(x)$$

Fig. 1 shows the input–output relationships for these functions. In the discrete setting, consider a real-valued input sequence x^n, for n ∈ Z. Such a sequence could represent samples of the continuous function x (t), for t = nT, where T is a sample period (and fs = 1/T is the sample rate), or could be entirely synthetic. A direct approach to discrete-time emulation of (1) is to simply compute an output sequence y^n as

$$y^n = F_0(x^n)$$

As is well-known, such a trivial implementation generates aliased components, of strength depending on the smoothness of the mapping F0, and the amplitude of the input signal x^n [1], [19], [20].In a recent Project, Parker et al. presented a novel algorithm for the reduction of aliasing in discrete-time memory less nonlinearities, and suitable for operation at a non-oversampled rate [22]. It takes on a particularly simple form:

$$y^n = \frac{F_1^n - F_1^{n-1}}{x^n - x^{n-1}}$$

Here, F^n1 = F1 (x^n) represents the first anti derivative of F0 evaluated at x^n. The approximation (5) is arrived at after a number of steps. In particular, the input sequence x^n is assumed drawn from samples of a piecewise linear underlying function x (t), which is then convolved with a box function of one sample duration, and then re-sampled. The convolution operation mentioned above requires the evaluation of the anti-derivative of F0, leading directly to the form in (5). Only the single memory less nonlinearity has been discussed here. One observation that can be made about the anti aliasing method that it represents an approximation to

$$y(x) = \frac{dF_1}{dx} \quad \text{or} \quad y(t) = \frac{\frac{dF_1}{dt}}{\frac{dx}{dt}}$$

The performance of the proposed methods was evaluated by measuring the signal-to-noise ratio (SNR) for a set of sinusoidal inputs. The SNR was defined as the power ratio between the desired part of the signal, and the components generated by aliasing, or residual. Following, an ideal alias-free version of each test signal was synthesized using Fourier analysis and additive syn Project. This band limited signal was subtracted from the aliased signal to yield the residual.

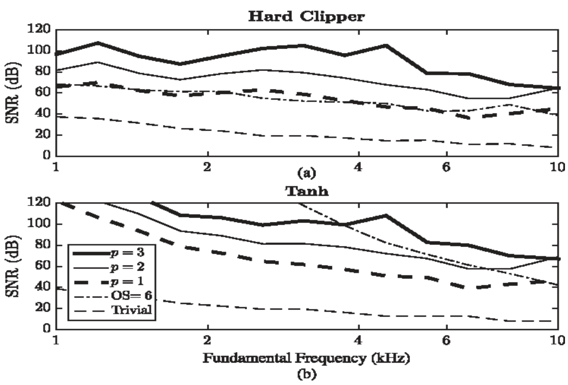


Fig. 2. SNR of sine waves under (a) hard and (b) soft clipping implemented trivially (fs = 44.1 kHz), oversampled by 6 (OS = 6), employing oversampling by

factor 2 and, the first-order method (p = 1) and the proposed anti derivative forms (p = 2, 3).

More efficient than oversampling, and does not depend on the particular type of nonlinearity, or on a simplification of its functional form. It is presented here as a family of methods of increasing order p of anti differentiation in the nonlinearity, leading, ultimately, to an increasing degree of aliasing suppression. As is natural, computational cost also scales with the order p. There remain many open questions and avenues of future research. A series of discrete approximations to is given in (13) which a) maintain the nested structure of the underlying equation, and b) for a given order p, are minimal in terms of the number of signal values used to compute an approximation, which is p + 1. An inherent characteristic of this family of methods is that of spectral shaping of the output; though aliasing is suppressed, there can be some attenuation of the signal in the high frequency range. Additional linear filtering is one option in this case, as suggested in for anti-aliased oscillators. Given that neither property a) nor b) is necessary in the approximation of (9), generalisations beyond the nested structures presented here could aid in finding anti-aliasing methods for which such attenuation is reduced.

Only the single memory less nonlinearity has been discussed here as it is currently one of the main applications of virtual analog modelling in audio. The proposed system is extended to remove the aliasing in the real-time audio signals.

IV.RESULTS AND DISCUSSION

The proposed system is designed, developed, coded, implemented and tested in the Mat lab environment and the simulation results are presented as follows

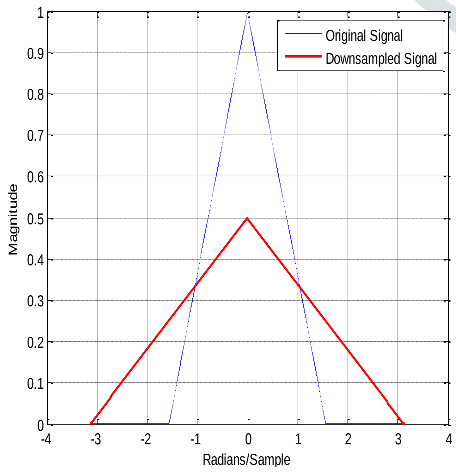


Figure 3: Input Sample Signal

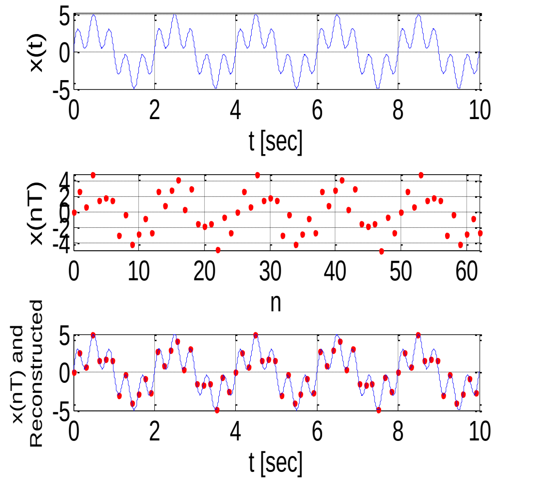


Figure 4: Sampling the Signal

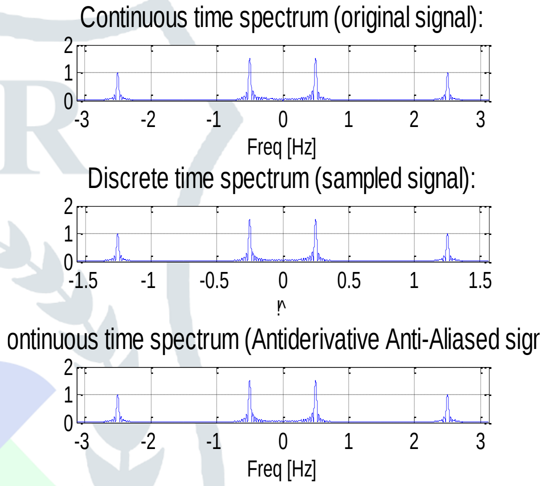


Figure 5: Continuous and Discrete Sampled Signal

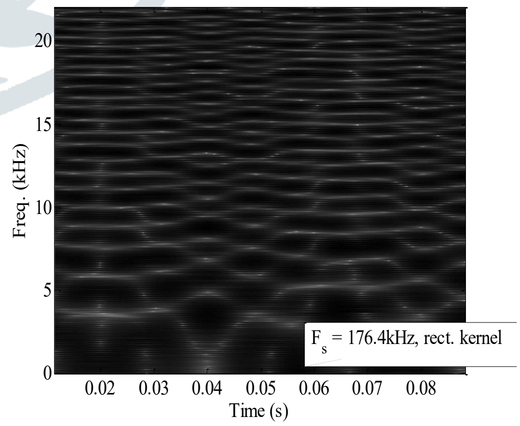


Figure 6: Applying Rectangular Filter

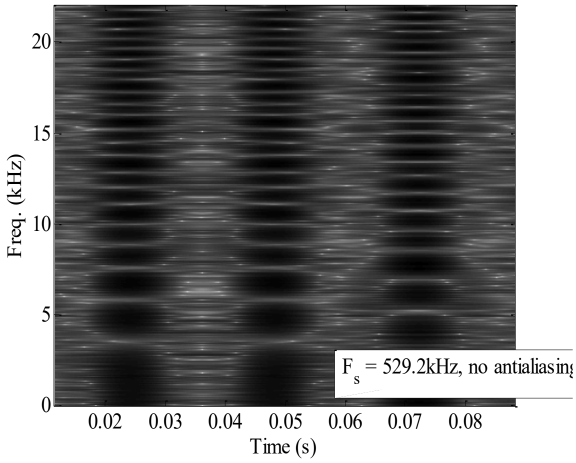


Figure 1: Anti Aliasing Free Output

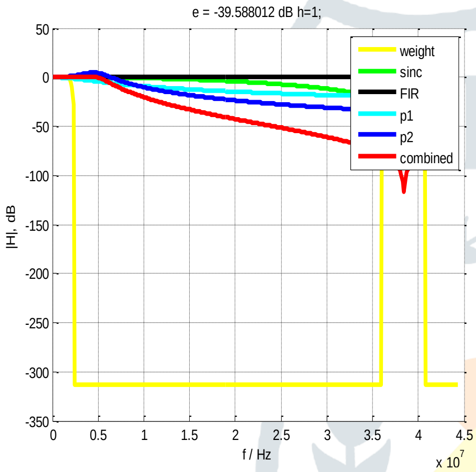


Figure 2: Comparing of results

TABLE-I

COMPARISON TABLE

Parameter	Existing Techniques		Proposed Techniques	
	Order-1	Order-2	Order-1	Order-2
PSNR	21.4536	25.3617	32.4255	34.5136
RMSE	16.5249	14.2542	5.8124	4.1172
MAXERR	109.3	89.7411	23.2215	21.6384
L2RAT	0.8634	0.8216	0.4836	0.3429

V.CONCLUSION

A new approach to anti-aliasing for discrete-time nonlinearities has been presented here. It is of a general character, more efficient than oversampling, and does not depend on the particular type of nonlinearity, or on a simplification of its functional form. It is presented here as a family of methods of increasing order p of anti-differentiation in the nonlinearity, leading, ultimately, to an increasing degree of aliasing suppression. As is natural, computational cost also scales with the order p. There remain many open questions and avenues of future research. A series of discrete approximations to (9) is given in (13) which a) maintain the nested structure of the underlying equation, and b) for a given order p, are minimal in terms of the number of signal values used to compute an approximation, which is p + 1. An inherent characteristic of this family of methods is that of spectral shaping of the output; though aliasing is suppressed, there can be some attenuation of the signal in the high frequency range. Additional linear filtering is one option in this case, as suggested in for anti-aliased oscillators. Given that neither property a) nor b) is necessary in the approximation of (9), generalisations beyond the nested structures presented here could aid in finding anti-aliasing methods for which such attenuation is reduced.

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