Design of Stable and Configurable Digital Filters for Automotive Sensors

Vidya Sagar Kantamneni
Alpen-Adria Universität Klagenfurt
Austria
Email: Vidya Sagar.Kantamneni@aau.at

Wolfgang Scherr
Infineon Technologies AG
Austria
Email: Wolfgang.Scherr@infineon.com

Bernhard Rinner
Alpen-Adria Universität Klagenfurt
Austria
Email: Bernhard.Rinner@aau.at

***Abstract*—Since many years, extensive research has been done in the field of Wave Digital Filters (WDF). They have good properties concerning insusceptibility to coefficient quantization, good dynamic range and excellent stability under finite arithmetic operations. Lattice Wave Digital Filters (LWDF) are more attractive for high speed applications because their modular structures yields in a high degree of parallelism. Moreover, these filters can be conveniently designed for Very Large Scale Integrated Circuit (VLSI) applications. The goal of this work is to develop a widely configurable LWDF filter with low hardware complexity, replacing dedicated filter structures for a large range of applications. Such a filter core is aimed for automotive sensor solutions where space and performance are general trade-off parameters, providing the potential to the customer to choose what suits the application best.**

I. Introduction

Intelligent sensors have many integrated elements that can support them to perform functions like primary sense and excitation control, information processing, data conversion and trimming. For an intelligent sensor, to provide more accurate results by eliminating noise from the useful information, filters are used as well. Analog filters in intelligent sensors exhibit better characteristics compared to digital filters for certain applications, but at the expense of higher space and power requirements. Moreover, analog filters are hardly configurable and are dedicated to specific environments. This poses a huge challenge to the sensor market and manufacturers, where often specific intelligent sensors are developed only for a narrow set of applications.

The design of filters which suit a wide-range of applications are desired by engineers working in automotive sensor industry. This can reduce re-design, test and validation efforts, thereby reducing time-to-market and the cost of the product. It is a challenge to design a filter core, that can be adapted based on the requirements of the customer because many parameters and specifications of the filter vary hugely depending on the application. Despite the complexity, it is interesting to attempt to develop generalized filter structures which are configurable to suit similar applications. Such configurable structures are lucrative for the manufacturers and, on the other hand, are very helpful for the customers as these structures help to meet the target specification more accurately. Configurable filters are feasible only in the digital domain.

Digital filters are in general very sensitive to coefficient quantization. In other words, the frequency response of the digital filter varies from the ideal frequency response when the precision of the filter coefficients is limited. The ideal frequency response is the frequency response of the digital filter when implemented with coefficients of infinite wordlength. Large deviation in the frequency response from the ideal case is not desired, when coefficients are constrained to finite wordlengths on hardware. Effects of using finite wordlength in digital filter design are discussed in detail in [1] [2].

Digital filters also face problems with conditional stability. They can easily become unstable, specifically when the coefficient values get close to the unity circle [1] [2]. The well-known digital FIR filters guarantee stability as they contain no feedback paths, but to meet a design specification, typically a very high order FIR filter designs are required. A higher order filter can only be implemented with large number of components and such implementations consume space. Sensor applications, specifically automotive sensors, are often embedded with strong constraints on space and power consumption. This gives the motivation to develop widely configurable, area and power-efficient wave digital filter structures for automotive sensor applications, which are said to be stable by definition and insusceptible to finite wordlength effects [3] [4].

II. Conceptual overview

A LWDF can be generalized as a two-branch structure realized using all-pass filter functions, i.e., transfer function shows a constant gain factor of one for all frequencies, but the phase response varies in frequency. An N^{th} order LWDF using 2-port scattering adapters is shown in Fig. 1 [5] [6].

Each branch provides a complementary output of the other branch. The frequency response of a 5^{th} order LWDF notch filter with notches in the frequencies of interest is shown in Fig. 2 [7]. A notch filter, also referred to as Vlach filter, is used as reference, because it has a specific characteristic, where the designer can adjust transmission zeros according to the application. Transmission zeros are critical frequencies in the stopband where signal transmission between the input and output is attenuated at a high rate. This makes it more practicable when defining filter coefficients for a given application, e.g., to eliminate certain frequencies in the system.

Complementary outputs with characteristics of both highpass and lowpass can be observed at the output of any typical

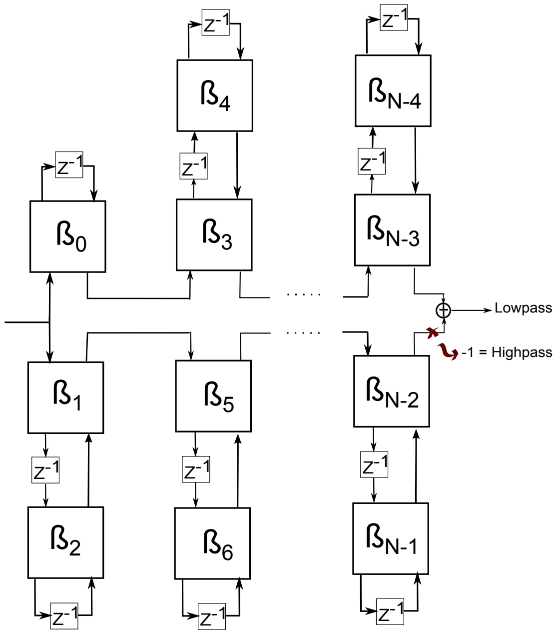

Fig. 1. LWDF of Order N

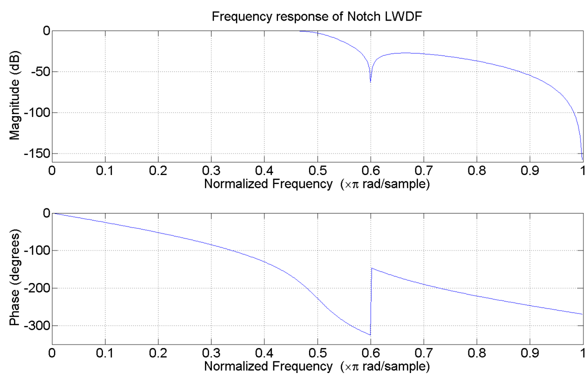

Fig. 2. Frequency response of LWDF with notch as reference filter

LWDF design. The inverted sum of two branches exhibits the characteristics of a highpass, whereas the output corresponding to the direct sum of two branches behaves like a lowpass filter. However, when the output of the highpass filter is not required, this complementary output can be removed, thereby saving one adder in the implementation.

Scattering adaptors are the building blocks of the LWDF. A N$^{\text{th}}$ order LWDF filter has N scattering adaptors. A scattering adaptor is an arithmetic unit which can be implemented with *m* ports. A *m*-port adaptor has *m*-inputs and *m*-outputs. Typically, LWDF filters are built with two-port or three-port Richards' allpass structures [8]. Two-port adaptors require two multipliers and four additions, whereas three-port structures require one multiplier and three additions. Thus, it is beneficial to use three-port adaptors in terms of speed and hardware cost, due to the smaller number of multipliers involved in the design. However, implementations using three-port adaptors require higher coefficient wordlengths [9].

TABLE I
MAPPING BETWEEN THE STRUCTURE OF THE SCATTERING ADAPTER AND THE COEFFICIENT VALUE

Structure	**Range**	**Multiplier α**	**Arithmetic operation**
Type I	$0.5 \leq \beta < 1$	1 - β	b2 = a2 - (a2 - a1)α b1 = b2 + (a2 - a1)
Type II	$0 \leq \beta < 0.5$	β	b2 = a1 + (a2 - a1)α b1 = b2 + (a2 - a1)
Type III	$0 > \beta \geq -0.5$	$\lvert\beta\rvert$	b2 = a1 - (a2 + a1)α b1 = -b2 + (a2 + a1)
Type IV	$-0.5 > \beta > -1$	1 + β	b2 = a2 - (a2 + a1)α b1 = -b2 + (a2 + a1)

Some optimized two-port allpass structures with one multiplier and three additions are mentioned in [10]. The structure of these two-port adapters is not uniform and therefore, varies dependent on the value of the coefficient. The structure also determines the way arithmetic operations are performed in the filter. Table I shows how the structure of the adapter varies depending on the value of the coefficient. The structure of the adapter is referred as $TypeI$ when the coefficient β lies between 0.5 and 1. The multiplier α required to realize this structure is equal to $1 - \beta$. Alternatively, uniform structured two-port allpass adapters shown in [11] [12] can be used, where the structure of the two-port allpass sections remain constant, irrespective of the value of the coefficients. These uniform structures are however not the focus of this paper.

Lower order LWDF filter design should ensure that the filter order is reasonably chosen for the input requirements. The formulas provided in [13] specify the minimum order required to achieve a specific filter specification. The DC offset or unwanted gain can be noticed in the output of the filter if the filter order is exploited. In other words, the DC offset can be viewed as an abnormal side effect of the design, when one tries to achieve higher roll-off or higher stopband attenuation with a lower order filter. A filter design for sensor applications should not introduce any DC offset into the system. Specifically, for designs with reasonable roll-off, the step response of the filter shows that the filter settles exactly at the final value in finite *k*-steps. It is also interesting to note that there exist mechanisms to eliminate any unwanted gain present in low order filter design. One possible way is to introduce a random dither, that helps to eliminate quantization effects causing DC gain and limit cycles at the output [14].

One of the most difficult challenges in digital filter design is to implement narrow-band filters, where the cut-off frequency of the filter is chosen far from the system sampling frequency. Narrow-band filters have sharp transitions in their frequency response, thereby requiring higher order designs to meet the desired frequency response specifications. Higher order designs require fairly large computations and the round-off noise generated in computing the output is significantly higher [15]. Alternative solution for such applications is to use multirate and cascaded filtering or coefficients with higher wordlengths [9] [16]. Another alternative choice is to use cascade stages of lower order LWDF filters. Wave digital structures also offer decimation, specifically in applications where oversampling ADCs are involved [17]. Popular among such decimation structures are Bi-reciprocal Lattice Wave Digital Filters (BLWDF), which offer lower cost, hardware-efficient solutions in cascaded and multirate implementations [8] [18] [19] [20]. Comb filters are also preferred for decimating and pre-filtering the data.

Small scale limit cycles are a result of quantization or round-off errors. Small scale limit cycles are observed when the coefficients are strongly quantized. However, the magnitude of the limit cycles is quite small in LWDF, making them quite acceptable for many applications. The round-off errors and the number of multiply/add operations of pipelined lattice filters are smaller than those of non-pipelined lattice filters [21].

The appearance of any limit cycles, overflow and errors due to finite-arithmetic calculations are important to observe at the output. Large scale limit cycles are often caused by an overflow. This can be avoided by using saturation arithmetic in most of the applications. Scaling signals can also be a reasonable solution to avoid large scale limit cycles that occur due to arithmetic operations. Overflow errors that might occur due to overshoot can be completely avoided by using sufficiently small input signals. For a given impulse response and prescribed overflow level, an upper bound for the overflow can be easily derived. Infrequent overflows are accepted to exploit the dynamic range of the filter. However, after each overflow, the normal operation recovers preferably with high speed implying overflow stability in all cases [1] [2].

III. Implementation

The hardware concept of the developed field programmable filter core is shown in Fig. 3.

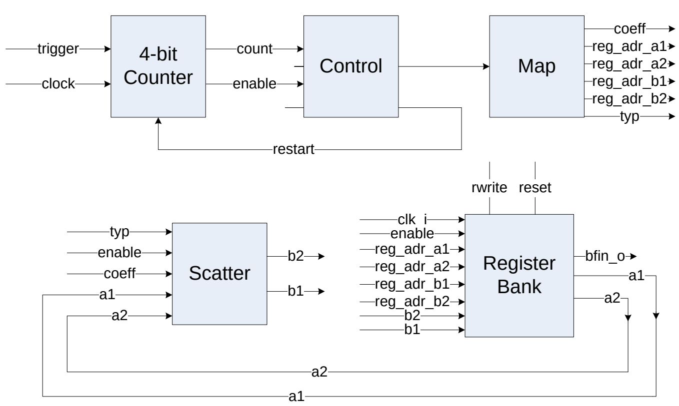

Fig. 3. Block level hardware overview of the generic LWDF filter

Control words required to configure the filter are stored in a mapping block, translating the fixed sequence counts to configurable control signals. This sequencer consisting of the Counter and Control blocks, loads each control word on every clock pulse and uses this control word to configure the behavior of the scattering adapter. Control words also determine the register location to store the input, results of intermediate arithmetic calculations and output value. An example control word for the above hardware concept has the format shown in Fig. 4.

a1 (4-bit)	b2 (4-bit)	a2 (4-bit)	b1 (4-bit)	Koeff (13-bit)	Type (3-bit)

Fig. 4. Control word required to realize a maximum 11^{th} Order Filter

With the help of this hardware concept and register control, it is possible to implement filters of any order. The number of control words determine the order of the filter. Lower order filters require less execution time and low power consumption. On the other hand, higher order filters offer better roll-off in the transition band. The order of the filter also determines the characteristic of the filter: for instance, odd order LWDF characterizes highpass and lowpass, whereas even order characterizes bandpass and bandstop. Modification of coefficients gives the flexibility to modify cut-off frequency, passband and stopband ripples, stopband attenuation, transmission zeros etc. In this way, the characteristics of the filter are easily modified by re-writing the control words in the memory.

Each scattering adapter has three adders and a multiplier. As the design aims at area-efficiency, only one scattering adapter is used, which is time-multiplexed and configured according to the value of the coefficient on the unit circle. Assuming that the customer would like to implement a third order LWDF filter with coefficients 0.85, -0.6 and 0.95. Three scattering adapters are required to realize this filter. From Table I, two *Type I* adapters for coefficients 0.85 and 0.95 and one *Type IV* adapter for coefficient -0.6 have to be defined. If the structure of the scattering adapter can be reconfigured on every clock cycle based on the value of the coefficient, only one scattering adapter has to be implemented. The timing behavior of the third order LWDF filter implemented with one scattering adapter is shown in Fig. 5.

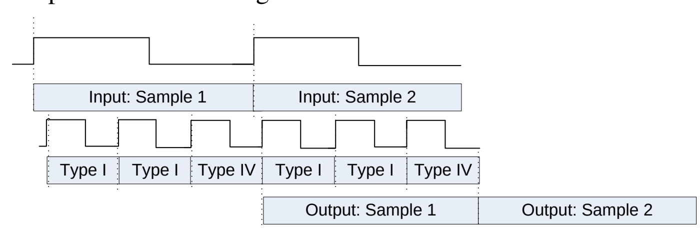

Fig. 5. Timing behavior of the implemented LWDF filter core

Such an implementation requires internally higher clock rates. For a third order filter, the internal clock rate of the filter core should be at least three times the incoming sample rate. For implementations with higher order, the filter should execute with even higher clock rates. The maximum achievable clock at which the filter can execute determines the achievable filter order for a given target technology. Additionally, for a filter of order *L*, *L*+5 16-bit registers are required to produce the output and a RAM of sufficient size to hold *L* number of control words. In addition to regular control words that are required to configure the filter, a stop control word is also used to keep the entire system at freeze, a mechanism to save power where required.

The filter core is implemented and evaluated on a devel-

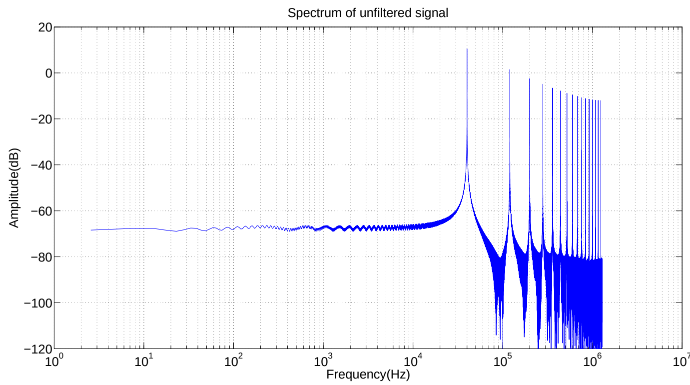

Fig. 6. Input spectrum of the generated 40 kHz rectangular wave

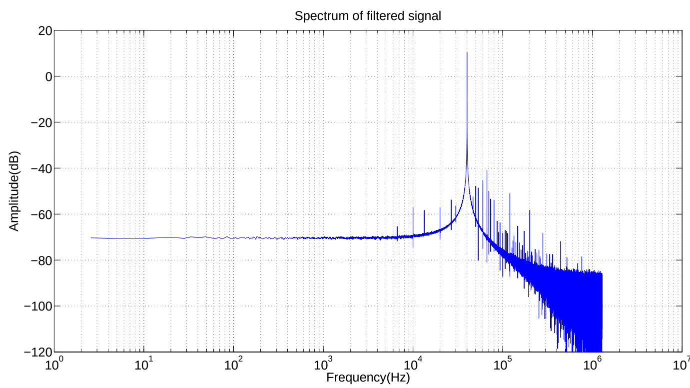

Fig. 7. Filtered spectrum of the 40 kHz rectangular wave using 11^{th} order LWDF filter

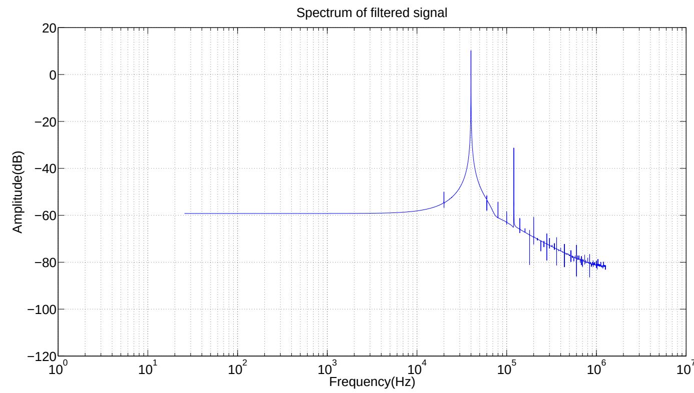

Fig. 8. Filtered spectrum of the 40 kHz rectangular wave using 5^{th} order LWDF filter

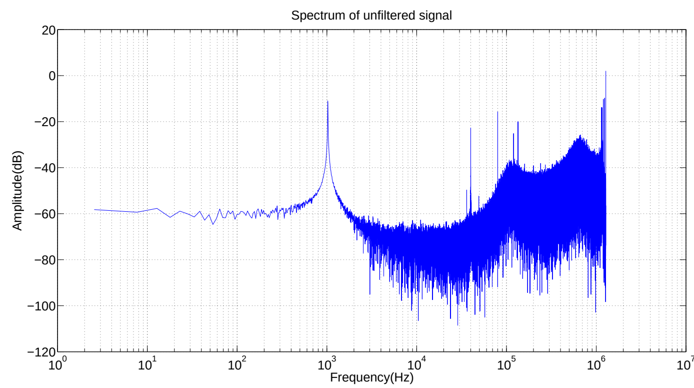

Fig. 9. Spectrum of the oversampled sigma delta ADC

opment kit where necessary interfaces required to configure and evaluate the digital filter are developed. The coefficients generated using a LWDF toolbox [7] are embedded into the control words and are placed into the Block RAM of the FPGA via an USB interface. Similarly, the output of the filter is stored in a 16 MB SRAM provided in the evaluation kit. Filter output data from SRAM is loaded back into the PC, also via USB, for further evaluation.

IV. Results

There are different ways to evaluate the filter. A standard way is to evaluate the step response of the filter. From the step response of the filter, it is possible to estimate the settling time, settling value, stability, overshoot and existence of limit cycles at the output. The performance of the digital filter can also be evaluated by looking at the frequency response of the filter or by comparing the input and output spectrum of the filter. For our evaluation we use a 40 kHz rectangular wave as an input to the filter (shown in Fig. 6).

The 40 kHz rectangular signal is filtered with the 11^{th}-order LWDF filter core, and the filtered spectrum is shown in Fig. 7. A cut-off frequency of 50 kHz is used and all spectral peaks above 50 kHz are attenuated as expected. The spectral peak at 40 kHz remains unmodified by the filter.

In a similar setting, the rectangular signal is also filtered using 5^{th}-order filter core, and the filtered spectrum is shown in Fig. 8. Interestingly, the 5^{th}-order LWDF filter core introduces less noise in passband and stopband and filters out unwanted spectrum resulting in a smaller noise power-density when compared to the 11^{th} order filter. This highlights the advantage of configurability in filter design, which helps to adapt the existing design according to the application at hand.

The same filter core is evaluated using the noise spectrum of a 2.56 MHz sampling frequency sigma-delta ADC. However, for evaluation purposes, a sinusoidal signal of 1 kHz is given as input to the ADC. The sinusoidal signal at 1 kHz combined with the noise spectrum of an oversampled sigma delta ADC are shown in Fig. 9. The signal spectrum from the ADC is filtered using the 11^{th} order filter with 50 kHz cut-off frequency. It can be seen from the Fig. 10 that the spectral values start to attenuate after 50 kHz corner frequency, however, the spectral peak at 1 kHz remains unmodified.

in Table II, the hardware cost estimate of 11^{th} order LWDF implementation is compared with optimized 3^{rd} order CIC implementation. Also the area to existing filters can be compared.

V. Conclusions

LWDF filters exhibit excellent properties with limited word length coefficients. They are recursive in nature and the

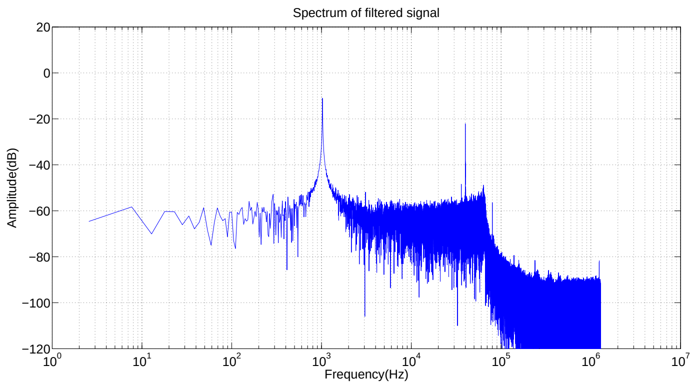

Fig. 10. Filtered spectrum of the oversampled sigma delta ADC

TABLE II
COMPARISON OF 11^{th} ORDER LWDF FILTER WITH 3^{rd} ORDER CIC FILTER

	LWDF(11^{th} Order)	**Optimized CIC 3^{rd} Order**
Flip-flops	320	138
Adders	116 1-bit	138 1-bit
Multiplexers	2 16-to-1	
Multiplier	1 16x16-bit	
Compare-factor	4000	2000

phase response of these filters is non-linear. Hence, these filters suit many applications which are not constrained by a linear phase response. As can be seen from the results, fully configurable and field-programmable filter cores can be developed and configured to suit the application at hand. Filtering is performed on the signal, which is sampled with 2.56 MSPS. The coefficients of the 11^{th} order filter almost reach near unit circle. To achieve filtering at even smaller frequencies than 50 kHz, for example at 10 kHz or 20 kHz, it is important to reduce the sampling frequency or otherwise, use higher coefficient word lengths. A configurable and generic filter core allows the increase of coefficient word lengths with no additional design effort. Various filter structures can be cascaded to improve the performance of implemented filter structures. When using cascaded structures, BLWDF are preferred as they can decimate the input sequence by two and by default, also offer cut-off at one-fourth the sampling frequency. BLWDF filter structures require very low number of components in comparison with standard WDF structures. BLWDF followed by a field programmable filter core makes the entire filter structure suitable for almost any kind of application.

ACKNOWLEDGMENTS

The authors would like to thank Infineon Technologies AG for funding this work.

REFERENCES

[1] H. Butterweck, J. Ritzerfeld, and J. Werter, *Finite Wordlength Effects in Digital Filters: A Review*, ser. EUT report / E: EUT report. Eindhoven University of Technology, Faculty of Electrical Engineering, 1988.

[2] B. Liu, "Effect of finite word length on the accuracy of digital filters–a review," *IEEE Transactions on Circuit Theory*, vol. 18, no. 6, pp. 670–677, 1971.

[3] A. Fettweis, "Wave digital filters: Theory and practice," *Proceedings of the IEEE*, vol. 74, no. 2, pp. 270–327, 1986.

[4] R. Nouta, *Studies in Wave Digital Filter Theory and Design*. Technische Hogeschool Delft, 1979.

[5] U. Kaiser, "Wave digital filtering for ti sensor signal processor msp430," technical report. [Online]. Available: www.ti.com/sc/docs/products/micro/msp430/wave.pdf

[6] K. Venkat, "Wave digital filtering using the msp430," September 2006, technical report. [Online]. Available: http://www.ti.com/lit/an/slaa331/slaa331.pdf

[7] H. Arriens, "User guide, reference guide and matlab toolbox for lwdf," Netherlands, 2006, technical report. [Online]. Available: ens.ewi.tudelft.nl/~huib/mtbx/more_wdfs.php

[8] H. Ohlsson, "Studies on design and implementation of low-complexity digital filters," Ph.D. dissertation, Linköping Universitet, Sweden, 2005.

[9] H. Johansson and L. Wanhammar, "Wave digital filter structures for high-speed narrow-band and wide-band filtering," *IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing*, vol. 46, no. 6, pp. 726–741, 1999.

[10] T. Magesacher, "Entwicklung digitaler Filter zur Interpolation und Dezimation für Sigma-Delta AD- und DA-Umsetzer," Master's thesis, Technische Universität Graz, Graz, Austria, 1994.

[11] S. Summerfield and T. Wicks, "An efficient hardware architecture implementing wave digital filters with reduced wordlength coefficients," in *Proceedings of IEEE Twelfth Saraga Colloquium on Digital and Analogue Filters and Filtering Systems*, 1992, pp. 5/1–5/4.

[12] H. Reekie, N. Petrie, J. Mavor, P. Denyer, and C. Lau, "Design and implementation of digital wave filters using universal adaptor structures," *IEE Proceedings on Communications, Radar and Signal Processing*, vol. 131, no. 6, pp. 615–622, 1984.

[13] L. Gazsi, "Explicit formulas for lattice wave digital filters," *IEEE Transactions on Circuits and Systems*, vol. 32, no. 1, pp. 68–88, 1985.

[14] H. Liu and M. Ackroyd, "Suppression of limit cycles in digital filters by random dither," *Radio and Electronic Engineer*, vol. 53, no. 6, pp. 235–240, 1983.

[15] L. Rabiner and R. Crochiere, "A novel implementation for narrow-band fir digital filters," *IEEE Transactions on Acoustics, Speech and Signal Processing*, vol. 23, no. 5, pp. 457–464, 1975.

[16] H. Johansson and L. Wanhammar, "High-speed recursive filter structures composed of identical all-pass subfilters for interpolation, decimation, and qmf banks with perfect magnitude reconstruction," *IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing*, vol. 46, no. 1, pp. 16–28, 1999.

[17] E. Dijkstra, L. Cardoletti, O. Nys, C. Piguet, and M. Degrauwe, "Wave digital decimation filters in oversampled a/d converters," in *Proceedings of IEEE International Symposium on Circuits and Systems, 1988.*, 1988, pp. 2327–2330 vol.3.

[18] H. Ohlsson, "Studies on Implementation of Digital Filters with High Throughput and Low Power Consumption," Master's thesis, Linköping Universitet, Sweden, 2003.

[19] H. Johansson and L. Wanhammar, "Design of bireciprocal linear-phase lattice wave digital filters," 1996. [Online]. Available: http://www.tde.isy.liu.se/publications/papers_and_reports/1996/LiTH-ISY-R-1877.pdf

[20] A. Wroblewski, T. Erl, and J. Nossek, "Bireciprocal lattice wave digital filters with almost linear phase response," in *IEEE International Conference on Acoustics, Speech, and Signal Processing.*, vol. 2, 2003, pp. II–805–8 vol.2.

[21] J.-G. Chung and K. Parhi, "Design of pipelined lattice iir digital filters," in *Conference Record of the Twenty-Fifth Asilomar Conference on Signals, Systems and Computers*, 1991, pp. 1021–1025 vol.2.