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Impulse sampled FIR interpolation with SC active-delayed block polyphase structures

Seng-Pan U, R.P. Martins and J.E. Franca

The authors propose new switched-capacitor (SC) active-delayed block (ADB) polyphase structures for analogue finite impulse response (FIR) interpolation. Their system functions are not distorted by the input sample-and-hold filtering effect at the lower input sampling rate. Both canonic and non-canonic ADB polyphase architectures which are superior in terms of a lower speed requirement for OAs, and fewer OAs, respectively, will be proposed for SC FIR interpolators.

Introduction: Interpolators are commonly used not only for relaxing the selectivity requirement of continuous-time post-filtering in sampled-data analogue filtering and digital-to-analogue interface systems, but also for realising a wide variety of complex mixed analogue and digital multi-rate signal processing functions [1 – 6]. Specialised multirate SC interpolators, based on direct-form polyphase structures, have previously been implemented [1, 4, 5]. However, the required modification to the original digital interpolation transfer function due to the input sample-and-hold (S/H) effect at lower input sampling rates rendered the design process and circuit architecture more complex. Increased distortion in the overall frequency response was another consequence. This limitation has been overcome in the new direct-form polyphase SC interpolators, by using the original prototype transfer function and, hence, the new interpolators perform in the same way as their digital counterparts [7]. Owing to the inherent impulse-sampled operation at the input, their amplitude responses are no longer affected by the sample-and-hold shaping distortion at lower input sampling rates. However, such direct-form polyphase SC implementation may be appropriate only when the length of the impulse response N is not much greater than the interpolation factor L, i.e. $N \leq 2L$, because of the requirement for a rather large number of SC branches and complex switching waveforms [1, 7, 8]. This Letter proposes new SC finite impulse response (FIR) interpolators without the input sampled-and-hold effect by using active-delayed block (ADB) polyphase structures when the filter order $N > 2L$. We present both canonic and non-canonic ADB polyphase architectures, categorised by means of the real implemented delay of ADBs, in which the canonic and non-canonic structures have L and more-than-L unit delays, respectively, implying a distinct requirement in terms of the number and speed of OAs. A bandpass interpolator with four-fold sampling rate increase will be employed for verification of the new proposed SC structure.

Canonic and non-canonic SC ADB polyphase architectures: The original digital prototype transfer function of interpolation can be canonically decomposed in $B + 1$ blocks, in which each block has only L coefficients, i.e.

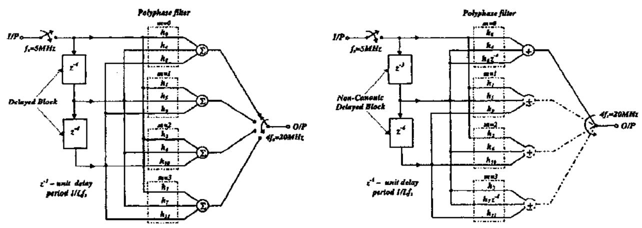
$$H(z) = \sum_{n=0}^{N-1} h_n z^{-n} = \sum_{b=0}^{B} \left(\sum_{n=0}^{L-1} h_{n+bL} z^{-n} \right) (z^{-L})^b$$
$$B = \left\lfloor \frac{N-L}{L} \right\rfloor \quad (1)$$

where N is the filter length, $\lfloor x \rfloor$ denotes the smallest integer greater than or equal to x, and the unit delay z^{-1} corresponds to the higher output sampling rate. Such an FIR structure can be realised by combining direct-form (DF) parallel processing polyphase structures for every L-coefficient block, with a serial processing delay line implemented by common L-unit active-delayed blocks (ADBs); i.e. a canonic ADB polyphase structure.

Furthermore, the transfer function can also be expressed in the following non-canonic form:

$$H(z) = \sum_{n=0}^{(L-1)-1} h_n z^{-n} + \sum_{b=1}^{B'} \left(\sum_{n=0}^{2(L-1)-1} h_{n+(2b-1)(L-1)} z^{-n} \right) z^{(-2b-1)(L-1)}$$
$$B' = \left\lfloor \frac{N-(L-1)}{2(L-1)} \right\rfloor \quad (2)$$

in which each of $B' + 1$ blocks contains more than L delay terms, i.e. $2(L - 1)$; except the first block which has only $(L - 1)$ terms due to the elimination of the strict sampled-and-held format of the input signal, thus allowing arbitrary input signal format. Such block decomposition implies that each ADB will no longer have only L-unit delay as in the canonic form, hence we designated this form the non-canonic ADB polyphase structure.



a b

Fig. 1 *Canonic and non-canonic ADB polyphase structures for SC FIR bandpass interpolator (L = 4)*

a Canonic
b Non-canonic

Table 1: Original prototype digital impulse response coefficients

h_0, h_{11}	h_1, h_{10}	h_2, h_9	h_3, h_8	h_4, h_7	h_5, h_6
0.4016	0.1063	–0.5189	–0.591	–0.2542	0.9384

SC Implementations: Consider a bandpass interpolator with a sampling rate increase $L = 4$ (output sampling rate 20MHz), and its original digital prototype transfer function whose 12-duration impulse response coefficients are listed in Table 1. The canonic realisation which requires $B = 2$ cascaded ADBs is derived in Fig. 1*a* and its corresponding SC implementation can be referred to the proposed architecture in [9]. The only-L number of delay terms in each block renders the utilisation of L individual slower output accumulators for each polyphase filter, thus possessing the

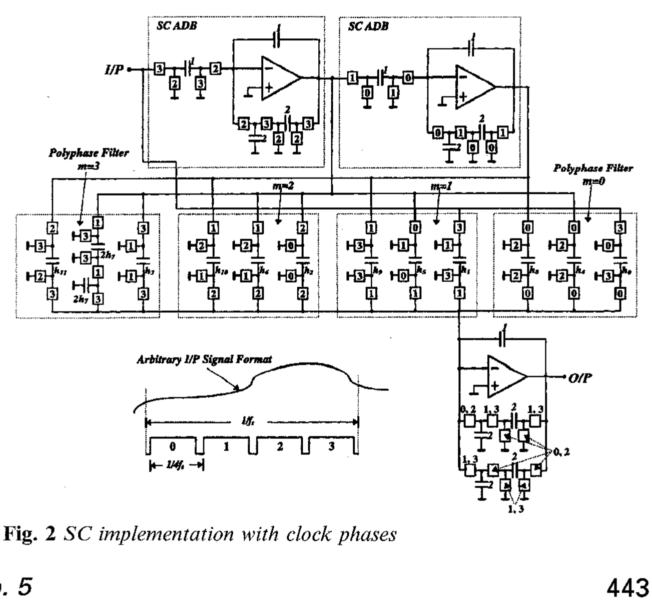


Fig. 2 *SC implementation with clock phases*

extended settling time of OAs. While for non-canonic realisation shown in Fig. 1*b*, although $B' = B = 2$ in this example, this is mainly due to the small filter order; when the filter length becomes larger, the efficiency in terms of the requirement of fewer ADBs (or OAs) becomes very significant. The increased delay terms in the second block can be realised by the flexible arrangement of sampling and charge transfer time of SC branches. This leads to a reduction of the charge transfer period or the settling time of OAs to only $1/Lf_s$, which is still better than the results in [1, 7]. Hence, not only have the number of clock phases been reduced to L, but, most efficiently, a new one-time-shared-output accumulator can be employed here, thus further relaxing the number of OAs, as shown in the SC implementation of Fig. 2. There, the upper two OAs with input and feedback/reset SC branches constitute two cascaded ADBs for producing the common delay terms z^{-3} and z^{-6}, respectively; the rest of the circuit is constructed by 4 DF polyphase filters formed by the shared output SC accumulator, and input SC branches, whose capacitance values correspond to the scaled impulse response coefficients where positive and negative coefficients are implemented by TSC, and OFR (or PCTSC), branches, respectively, for immunising the effects of a grounded parasitic. The resulting computer simulated amplitude response of this SC bandpass interpolator has been derived in curve (i) of Fig. 3. For comparison, the distorted response of SC interpolator without the impulse sampled technique is shown in curve (ii), which clearly shows the nonlinear distortion caused by the input S/H effect at the lower sampling rate, especially in the passband. Besides, curve (iii) shows the sole output S/H effect of the impulse sampled interpolator.

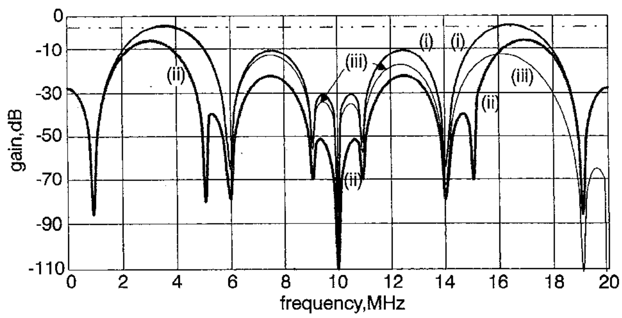


Fig. 3 *Computer simulated results of impulse sampled SC FIR ADB polyphase interpolator ($L = 4$)*

Conclusions: New impulse sampled SC FIR interpolators without the S/H shaping effect of a lower input sampling rate realised by an ADB polyphase structure in both canonic and non-canonic forms for longer impulse response were presented. From the point of view of high frequency capability, the canonic structure will outperform the non-canonic; but for a lower number of OAs and clock phases, the non-canonic structures will be a better alternative, and the efficiency is especially obvious when the impulse response is extraordinarily long. Such an SC structure was illustrated by considering one design example verified by behavioural computer simulation.

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Increased precision digital filter coefficients using digital dither

R. Greenfield

The DF1 IIR filter structure is the most common structure employed in audio applications. The author explores the application of digital dither to increase the precision of filter coefficients allowing up to double precision performance to be achieved with single precision arithmetic. This realisation provides a versatile DF1 realisation suitable for many applications.

Introduction : There exist numerous filter structures, each offering particular virtues for fixed point filter realisations. Paradoxically, however, the basic direct form 1 (DF1) structure is often deemed to be the most appropriate structure for many applications, and in particular, audio applications [1 – 3]. The primary attribute of the DF1 structure is that it effectively allows an infinite internal accumulator headroom when two's complement arithmetic is used (see Dattorro [1]). Thus, this structure cannot be outperformed from a signal overflow perspective and, at the same time, it can be efficiently implemented on standard digital signal processors.

The DF1 structure does not, however, offer a universal panacea to all digital filter problems. The basic structure is inherently noisy, possesses pole dependent coefficient sensitivity, and exhibits a nonuniform distribution of pole locations in the z-domain. Techniques, such as noise shaping [4], exist for minimising round-off noise, however, the finite coefficient precision effects are normally dealt with the use of alternative structures such as the Rader Gold [5] or the Argawal Burrus [6] structures, to name but two. Unfortunately however, all the alternative filter types require internal accumulations and, consequently, cannot offer the same overflow performance of the DF1 structure.

This Letter proposes the approach of extending the precision of the filter coefficients with the use of digital dither. This technique can be used to dynamically dither the filter transfer function such that the average, or long term, transfer function converges to a higher precision implementation. This Letter presents simulations of this process which demonstrate significant improvement in the filter transfer function accuracy.

DF1 structure finite precision coefficient effects: We will begin by considering the pole location distribution in a second-order function. The letter will deal exclusively with second-order functions as these are the building blocks for most filtering processes. The pole transfer function of the DF1 structure is given by

$$H(z) = \frac{1}{1 + b_1 z^{-1} + b_2 z^{-2}} \quad (1)$$

This can be rewritten in the form

$$H(z) = \frac{1}{1 - 2r\cos(\varphi) z^{-1} + r^2 z^{-2}} \quad (2)$$

where r is the radius and φ is the phase angle of the poles. Notice that, while the coefficients b_1 and b_2 are uniformly quantised, both