

Fig. 2 Simulated characteristics of a dual-mode filter with two square patches coupled with a slot and chamfered corners  $% \left( \frac{1}{2} \right) = 0$ 

Length of cut corners = 1.2mm, gapwidth = 0.8mm <u>Calc.</u>: S(1, 1), <u>S(1, 2)</u>, S(2, 1), S(2, 2)



Fig. 3 Performance of superconducting patch filter The filter is at a temperature of 15K

*Extracted pole planar elliptic function filters:* A method of achieving poles of attenuation at finite frequencies is to use additional resonators that provide rejection at these frequencies (extracted pole circuit). In our case it is achieved by a slight change of the transversal dimension of the patch resonator. Because of this change the effect of destructive superimposing is shifted outside the filter passband. An attraction of the extracted pole circuit is that each resonator is physically separate and hence independently tunable.



**Fig. 4** Predicted performance by computer simulation of slot coupled dual-mode bandpass filter with different transversal lengths of patches <u>Calc.</u> <u>S(10, 10)</u>, <u>S(10, 20)</u>, S(20, 10), S(20, 20)

ELECTRONICS LETTERS 9th May 1996 Vol. 32 No. 10

The result of the simulation of the patch filter circuit is shown in Fig. 4. The transversal width of the patches has been changed as depicted in the Figure. The longitudinal width of both patches is 6 mm. The main filter network is formed by the two longitudinal resonators. The transversal resonators are used to provide poles at defined frequencies. The proposed filter topology to create elliptic function filters has the advantage of not needing additional networks as commonly used. To improve matching, line steps have been inserted in both feed lines.

*Conclusions:* This work has shown that it is practical to choose a different configuration to establish pseudoelliptic function filters in planar form. The feasibility has been confirmed by a theoretical and practical investigation. These investigations have also shown that it is necessary to build up planar patch filters with HTSC-technology. Besides the required significant reduction of losses and miniaturising the size of the circuit structure, the application of materials with a lower permittivity number would create a surface wave generator or a radiator [6].

#### © IEE 1996 5 February 1996 Electronics Letters Online No: 19960618

K.H. Waegel and R. Bauer (Institute of Radio Frequency, German Space Research Establishment (DLR), Oberpfaffenhofen, Germany, PO Box 1116, D-82230 Wessling, Germany)

### References

- HEDGES, S.J., and HUMPHREYS, R.G.: 'Extracted pole planar elliptic function filters'. ESA Estec workshop on space applications of high temperature superconductors, 27-28 April 1993, pp. 97–106
  HONG, J.S., and LANCASTER, M.J.: 'Canonical microstrip filter using
- 2 HONG, J.S., and LANCASTER, M.J.: 'Canonical microstrip filter using square open-loop resonators', *Electron. Lett.*, 1995, 31, pp. 2020– 2022
- 3 HONG, J.S., and LANCASTER, M.J.: 'Bandpass characteristics of new dual-mode microstrip square loop resonators', *Electron. Lett.*, 1995, **31**, pp. 891–892
- 4 MANSOUR, R.: 'Design of superconductive multiplexers using singlemode and dual-mode filters', *IEEE Trans. Microw. Theory Tech.*, 1994, 42, pp. 1411–1418
- 5 WAEGEL, K.H., and TEWES, W.: 'Interdigital bandpass filter design', Microw. J., 1995, (6), pp. 80-88
- 6 SCHAFER, M., BOCHTLER, U., BITZER, U., and LANDSTORFER, F.: 'Radiation losses in planar filter structures', *Microw. J.*, 1989, (10), pp. 139–143

## Switched-capacitor interpolators without the input sample-and-hold filtering effect

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

Indexing terms: Switched capacitor filters, Digital filters, Signal processing

The authors propose new switched-capacitor (SC) interpolators whose frequency responses are no longer affected by the input sample-and-hold filtering effect which occurred in previous circuits. Two different types of architectures are discussed, one for input sampled-and-held signals, and the other for arbitrary input signal formats. Examples are given to illustrate both types of SC interpolator circuits.

Introduction: SC interpolators are used to increase the sampling rate from  $f_s$  to  $Lf_s$ , with the corresponding rejection of the unwanted frequency-translated image components associated with the signals sampled at a lower rate, hence allowing more relaxed continuous-time post-filtering in SC filtering and digital-to-analogue interface systems. To save silicon area and power dissipation, specialised multirate SC circuits were developed, based on polyphase structures that could take advantage of the sampling rate increase inherent to interpolation process [1 – 3]. Such SC interpolators however have only implemented the interpolation of sampled-and-held signals requiring that the original digital interpolating filter H(z) is modified according to [1]

879

$$H'(z) = H(z) \sum_{l=0}^{L-1} z^{-l}$$
(1)

Such modification introduces L additional zeros at  $f_s, 2f_s \dots (L-1)f_s$ , as well as their replicas at integer multiples of  $Lf_s$ , and therefore yields increased distortion in the overall frequency response of interpolators besides demanding a more complex design procedure and circuit architecture.

We propose new SC interpolators which no longer require such modification of the original interpolating filter and hence operate in a similar way to their digital counterparts. One type of architecture operates with arbitrary input signal formats while the other, simpler, architecture requires an input sample-and-hold format. Examples of both types of architectures are given and verified by computer simulations.



Fig. 1 SC impulse sampled interpolation for input sample-and-hold signal formats

a Impulse response of 'digital' interpolator

b SC interpolator circuit and switch timing

Sample-and-hold input signal format: For simplicity of explanation. we begin by considering the example of a digital interpolating function where the sampling rate increase is L = 4, from  $f_s =$ 48 kHz to  $4f_s = 192$  kHz, and the interpolating filter with length N = 7 possesses the impulse response coefficients given in Fig. 1a. Such an interpolating function can be implemented by using the polyphase SC circuit of Fig. 1b where, despite the input sampledand-held signal format, the prototype transfer function of the digital interpolation filters has not been modified, as was previously required [1]. Here, we can see that all SC branches sample the input signal in time slots '8' and '7' but, because of the input S/H signal format, this is equivalent to sampling only once per period  $1/f_s$ . The interpolated output samples are produced by the L = 4polyphase filters in time slots 0, 1, 2 and 3. SC branches with normalised capacitance value  $h_0$  and  $h_4$  correspond to the first polyphase filter delivering an output sample in time slot '0'. Similarly, SC branches with normalised capacitance values  $h_1$  and  $h_5$  constitute the second polyphase filter producing an output sample in time slot 'l'; SC branches with normalised capacitance values  $h_{i}$ , and  $h_6$  constitute the third polyphase filter yielding an output sample in time slot '2'; an SC branch with normalised capacitance value  $h_3$  constitutes the fourth polyphase filter whose output sample is produced in time slot '3'. The computer simulated amplitude response shown in curve (i) of Fig. 2 demonstrates that the amplitude response of the SC interpolator is free from the S/H shaping effect at the lower sampling frequency and hence renders this interpolation process exactly the same as a digital interpolation. For comparison, the amplitude response of the SC interpolator. previously available for implementing the same interpolating function by using a modified transfer function, is also shown in curve (ii), clearly showing that the response is affected by the input sample-and-hold format [1].



Fig. 2 Computer simulation results of SC interpolator



Fig. 3 SC impulse sampled interpolation for arbitrary signal formats: SC interpolator circuit and switch timing

Arbitrary input signal format: Although the input signals of SC interpolators are usually sampled-and-held at  $f_s$ , as required in the above SC interpolator, they must be strictly synchronised with the operation of the interpolator in order to ensure that only one input sample is taken during the interpolation period. This requirement is overcome by the alternative polyphase SC interpolator shown in Fig. 3 with switch phasing. Here, we can see that all SC branches sample the input signal only once per period  $1/f_s$ , regardless of the format of the input signal. This is accomplished by using in each of the polyphase filters (except the last one) two paralleled branches clocked at half the input sampling frequency and with the same normalised capacitance values. The computer simulated amplitude response of this SC interpolator, obtained with arbitrary input signal format, is shown in curve (i) of Fig. 4 which is exactly the same as in curve (i) of Fig. 2. When the output sample-and-hold effect at higher sampling rate Lfs is included, we obtain the amplitude response given in curve (ii) of Fig. 4 with notches at  $4f_s = 192$  kHz and its integer multiples.



Fig. 4 Computer simulation results of SC interpolator

*Conclusions:* New impulse sampled SC interpolator circuits, which are no longer affected by the sample-and-hold shaping effect at the lower input sampling frequency, were presented. These are based on SC polyphase structures that implement the same digital interpolating function but which require different constraints for the interpolated input signal. The simpler solution can only operate with a properly synchronised sample-and-hold input signal while the second, more elaborate, solution accepts input signals with arbitrary formats.

© IEE 1996 11 March 1996 Electronics Letters Online No: 19960603

U. Seng Pan and R.P. Martins (FST/University of Macau, Macau)

J.E. Franca (IST/Integrated Circuits and Systems Group, Lisbon, Portugal)

R.P. Martins: on leave from IST, Portugal

### References

- FRANCA, J.E.: 'Non-recursive polyphase switched-capacitor decimators and interpolators', *IEEE Trans. Circuits Syst.*, 1985, CAS-32, pp. 877–887
- 2 MARTINS, R.P., and FRANCA, J.E.: 'Infinite impulse response switchedcapacitor interpolators with optimum implementation'. Proc. IEEE Int. Symp. Circuits Syst., Louisiana, USA, May 1990
- 3 MARTINS, R.P., and FRANCA, J.E.: 'Novel second-order switchedcapacitor interpolator', *Electron. Lett.*, 1992, **28**, (2), pp. 348-350

# Grey scale image representation using binary sketch image

G. Deng

Indexing terms: Image coding, Wavelet transforms

A new grey scale image representation using a binary sketch image is presented. The sketch algorithm consists of three steps: preprocessing by wavelet denoising, convolving with a Laplacian operator and thresholding. It is shown that the sketch image provides an alternative and visually better representation of the original image than the edges.

Introduction: The representation of grey scale image by a binary image which retains the most important features of the original is important in many situations such as very low bit rate coding, image recognition and understanding. Image representation by edges (a binary image) has played a fundamental role in computer vision and image analysis. There are several difficulties in edge detection. There is no general definition of edges which can discriminate edges from noise [1]. In addition, edges detected by using differentiation based techniques are not always relevant to visually important features [2]. However, artists have been successful at sketching images with a few lines. These lines represent, not necessary edges, but in many cases a part of image which is darker than its surroundings. Based on this fact, an adaptive sketch-coding technique [2] and a sketch algorithm using the logarithmic image processing model [3] have been proposed. It was shown [2, 3] that the sketch images are particularly useful for applications such as human face recognition and very low bit rate image compression.

In this Letter, a new sketch algorithm is presented. It is shown that the sketch image, which is different from the edge image (result of edge detection), is capable of extracting and rendering features of the original grey scale image. Because the algorithm is very sensitive to noise, a simple wavelet denoising method is proposed as a preprocessing step.

Generating the sketch image: Let f(i, j) be the original image, and s(i, j) be the sketch image. The algorithm that generates a sketch image is mathematically described by the following equations:

 $f_1(i, j) = h^* f(i, j)$  (1) where  $f_i(i, j)$  is the convolution result, '\*' stands for 2-dimensional convolution and h is the (3×3) Laplacian operator,

	1	1	1
h =	1	$^{-8}$	1
	1	1	1

ELECTRONICS LETTERS 9th May 1996 Vol. 32 No. 10

The sketch image is then given by

$$s(i,j) = \begin{cases} 0 & \text{if } f_1(i,j) > t\\ 255 & \text{otherwise} \end{cases}$$
(2)

where t is a positive number. It can be easily seen that the above algorithm is essentially a convolution with the Laplacian operator followed by a thresholding process. It is well known that the Laplacian operator is very sensitive to noise. Even small system noises such as those due to the imaging or digitisation process (e.g. the background noise of a camera) will be amplified and regarded as features of an image. There is a conflicting requirement for choosing a threshold in the sketch algorithm. A high threshold cuts off both noise and the features of small magnitudes, while a low threshold results in a very noisy binary image. There are two ways to solve this problem. One way is to use a postprocessing technique which tries to recognise noise pixels from feature pixels. This is a very difficult task, because it is dependent on the definition of 'noise pixel' and 'feature pixel'. Although edge detection techniques can be employed, they also suffer from the same problem. Another way is to use preprocessing to filter out noise. Simple linear lowpass filters are not suitable since they smooth out noise and sharp edges. Median filters are also not suitable for those system noises (which are not impulsive). One promising way is to distinguish noise against signal in the scale space [1].



Fig. 1 Decomposition of an image into three levels using wavelet transform

In this Letter, a simple wavelet based denoising algorithm is proposed to filter out the system noise. It involves the following steps: (i) perform the wavelet transform, (ii) apply the thresholding process and (iii) perform the inverse wavelet transform. The thresholding process is not performed on the lowest resolution (marked area in Fig. 1). Other areas are thresholded by

$$H(i,j) = \begin{cases} H(i,j) & \text{if } |H(i,j)| > T\\ 0 & \text{otherwise} \end{cases}$$
(3)

where H(i, j) is the transform coefficient and T is a threshold. The reason is that the system noise is so small that it does not appear in the lowest resolution. This thresholding process also avoids alternations to large features of the image.

*Discussion:* It is noted that the sketch image is different from the edges of an image in that features of the original grey scale image are represented by black dots (on a white black ground), which are pixels whose grey scale is less than the average of that of its neighbouring pixels. Thus the sketch image provides an alternative way for extracting and rendering features of an image.

It is also noted that the advantage of the wavelet denoising algorithm is its simplicity. In this algorithm, several factors should be considered, e.g. the depth of decomposition, the wavelet filter and the threshold. Although there have been in-depth discussions of how to choose a threshold [4, 5], it was found experimentally that a three-layer decomposition and a threshold ranging from 5 to 10 result in satisfactory denoising for typical images like 'Lena' and 'F-16'. The use of different wavelet filters is an open question. However, it is well known (in the field of data compression) that a long wavelet filter and a large threshold will more likely result in ringing at sharp edges. Actually, the thresholding process is equivalent to the quantisation step in wavelet-based image data compression techniques. Thus, there is a tradeoff between filtering out noise and creating spurious features. A Daubechies-4 filter is used