ANYTIME ALGORITHMS IN EMBEDDED SIGNAL PROCESSING SYSTEMS[1]

Annamária R. Várkonyi-Kóczy and Tamás Kovácsházy
Department of Measurement and Information Systems, Technical University of Budapest,
H-1521 Budapest, HUNGARY
Tel: + 36 1 4632057; fax: +36 1 4634112
e-mail: {koczy,khazy}@mmt.bme.hu

ABSTRACT

In this paper a family of so-called anytime signal processing algorithms is introduced to improve the overall performance of larger scale embedded digital signal processing (DSP) systems. In such systems there are cases where due to abrupt changes within the environment and/or the processing system temporal shortage of computational power and/or loss of some data may occur. It is an obvious requirement that even in such situations the actual processing should be continued to insure appropriate performance. This means that signal processing of somewhat simpler complexity should provide outputs of acceptable quality to continue the operation of the complete embedded system. The accuracy of the processing will be temporarily lower but possibly still enough to produce data for qualitative evaluations and supporting decisions. Consequently anytime algorithms should provide short response time and be very flexible with respect to the available input information and computational power. The paper presents such algorithms based on the re-configurable version of the recursive signal transformer structure described in [1].

1 INTRODUCTION

Computer-based monitoring and diagnostic systems are designed to handle abrupt changes due to failures within the supervised system or in its environment. This capability involves on one hand different, simultaneously operated digital signal processors (DSPs), while on the other the corresponding information processing algorithms. These algorithms should be performed under prescribed response time conditions. It is an obvious requirement to provide enough computational power but the achievable processing speed is highly influenced by the precedence, timing and data access conditions of the processing itself. It seems to be unavoidable even in the case of extremely careful design to get into situations where the shortage of necessary data and/or processing time becomes serious. Such situations may result in a critical breakdown of the computer-based monitoring and/or diagnostic systems.

With the introduction of anytime signal processing algorithms we try to handle the above changes and their consequences in larger scale embedded DSP systems [2]. The idea is that if there is a temporal shortage of computational power and/or there is a loss of some data the actual evaluation should be continued to provide appropriate overall performance. The solution should be signal processing of simpler complexity producing outputs of acceptable quality to continue the operation of the complete embedded system. The accuracy of the processing will be temporarily lower but possibly still enough to produce data for qualitative evaluations and supporting unavoidable decisions. Consequently anytime algorithms provide short response time and are very flexible with respect to the available input information and computational power. Such and similar flexibility is investigated in a somewhat different environment in [3]. It is important to observe at this point that such flexibility is possible only if a vector of "shortage indicators" is an additional input of the signal processing facilities in use. Obviously the shortage indicators are outputs of such information processing units which monitor the actual rate of sensory data and the rate of the computational load.

In Section II of this paper signal processing blocks with shortage indicator inputs will be considered. Since the operation of the monitoring and/or diagnostic systems should run in parallel with the data acquisition, the investigations are restricted to recursive algorithms. Recursive techniques are advantageous concerning also the response time since they always provide some kind of an estimate. Additionally, since parallelization may reduce response time if more processors are available, investigations are further restricted to highly parallel transform-domain signal processing algorithms utilizing time-recursive transformers. Some recent contributions to these topics are in [4] and [5]. These time-recursive transformers [1] can be offered as universal building

[1] This work was supported by the Hungarian Fund for Scientific Research (OTKA T 026254 and OTKA T 017448), the Office of Higher Education Support Programs (MKM 0250/1997), and the Office of Bilateral Intergovernmental Scientific Co-operation Programs (GR-32/96).

blocks having known behaviour, well established design technology and easy implementation.

In Section III of this paper a "shortage controlled" mechanism is developed to adapt the actual signal processing algorithm to the available amount of computing capacity. The simplest version of this mechanism is changing the order of some digital filter or filter-bank, but still maintaining the required filtering effect with smaller selectivity. In [4] smaller order block-oriented signal transformers are combined with recursive evaluation resulting a system with growing order. This varying-order system can be considered as an anytime algorithm, because it can provide output much earlier as standard block-oriented transformers do.

Anytime algorithms can be implemented as parameter- and/or structure-varying systems, i.e. as re-configurable systems. This re-configuration is controlled by shortage indicators. Due to the inherent dynamic nature any change within the system generates certain kind of transients. These transients may cause serious difficulties unless properly counteracted. Section IV of this paper deals with the transient behaviour of re-configurable signal processing systems. It is shown that the transients due to reconfiguration are structure dependent and therefore only certain types of signal processing structures may be suggested to serve as a universal building block for larger-scale embedded systems.

Another aspect of using anytime algorithms is the quality/accuracy degradation of the outputs while loss of data and/or shortage of computational power results in less accurate evaluation. For this reason Section V of the paper considers the problem of accuracy in systems with adaptive and re-configurable signal processing components.

2 MISSING INPUT SAMPLES

If due to temporary overload of certain communication channels the input samples fail to arrive in time or will be lost and some output of the signal processing is still required the usual approach is to utilize some prediction mechanism. This means that based on previous data our signal processors try to generate estimations. This alternative can be easily implemented using the signal processing structure given on Fig.1, and fully described, e.g., in [1]. This structure implements a recursive discrete signal transformation with respect to the basis vectors $\underline{c}_m^T = [c_m(0), c_m(1), \ldots, c_m(N-1)]$, $m = 0, 1, \ldots, N-1$. The multiplier values $c_m(n)$ on Fig.1 identify the vector element $c_m(k)$ where $k = n$ modulo N, i.e. the vector elements are repeated periodically during operation. The $g_m(n)$ values on Fig.1 denote in a rather similar way the elements of the reciprocal basis vectors $\underline{g}_m^T = [g_m(0), g_m(1), \ldots, g_m(N-1)]$, $m = 0, 1, \ldots, N-1$. As an example, for the recursive discrete Fourier transformation these coefficients are given as $c_m(n) = e^{j2\pi(mn/N)}$ and $g_m(n) = (1/N)e^{-j2\pi(mn/N)}$.

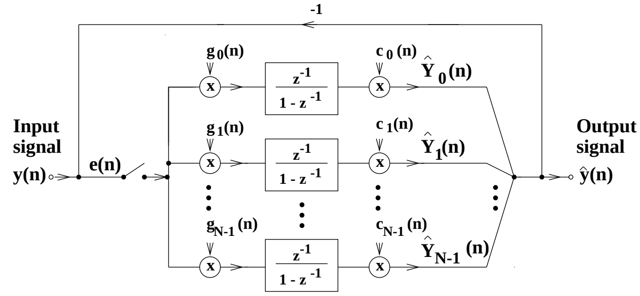

Figure 1: Recursive transformation based signal processing structure symbolizing the case of missing data. $c_m(n) = e^{j2\pi(mn/N)}$, $g_m(n) = (1/N)e^{-j2\pi(mn/N)}$, $m = 0, 1, \ldots, N-1$

The building blocks having uniform transfer function $z^{-1}/(1-z^{-1})$ on Fig.1 are discrete integrators which at their output (after convergence) will produce the transform-domain representation of the input signal. Multiplication by the basis vectors and a summation will result in a signal reconstruction. If the difference of the input and the output (reconstructed input) signals equals zero the discrete integrators will not be influenced and the system will operate in an autonomous way. This behavior can be utilized if the input samples fail to arrive in time. If this fact is identified the switch of the signal $e(n)$ on Fig.1 will be off and the remaining part of the system will produce an estimate of the output based on the information stored in the discrete integrators. Fig.2 serves to illustrate this effect. A periodic triangular waveform having a period of $N = 16$ samples is decomposed into its components. The switch is controlled by a random number in [0,1]. If the actual value is higher than 0.5 the switch is off,otherwise it is on. If all the input samples were available the transformer could learn the waveform from one complete period. With randomly missing input this procedure takes longer time, however, after receiving the first N samples the sig-

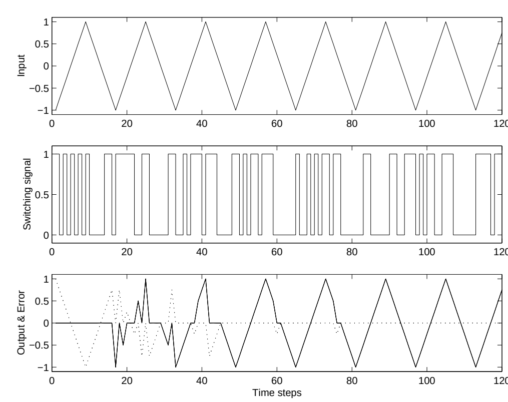

Figure 2: Decomposition and reconstruction of a periodic triangular waveform input in case of partial input information (the ratio of the missing input data is appr. 50 %): output – continuous line, error – dotted line

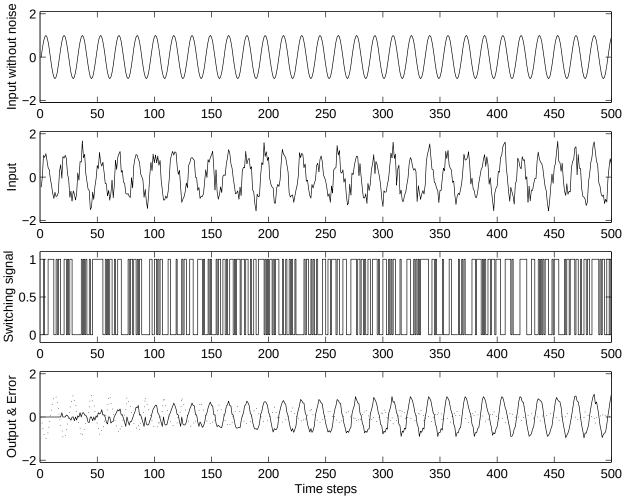

Figure 3: Response to a single sinusoid input signal plus noise (SNR=5dB) in case of partial input information (the ratio of the missing input data is appr. 50 %): output – continuous line, error – dotted line

nal estimate becomes quite acceptable. In this example there is a complete correspondence between the input signal and the signal representation within the transformer, because the triangular waveform can be generated without error from its Fourier representation. For this very reason, after convergence, the structure is capable to reconstruct the original waveform (in principle) without error. This property can be extended to periodic signals with unknown and/or varying fundamental frequency using the so-called Adaptive Fourier Analyzer (AFA) algorithm [6]. As another example Fig.3 shows the response to a single sinusoid plus noise. The switch is controlled similarly as in the previous example. To reduce noise, the transformer is combined with a fading memory effect [7], which corresponds to an exponential averaging. This effect can be achieved by changing the coefficient $1/N$ in the reciprocal basis vectors to α/N. In this example $\alpha = 0.2$. The responses show that at the prize of some delay and some transients the structure produces acceptable estimates.

3 SHORTAGE OF COMPUTING POWER

If due to a temporary shortage of computer power the signal processing can not be performed in time several alternative solutions can be proposed. Reduction of the sampling rate or the application of less accurate evaluations are typical examples. In this paper we consider only complexity reduction techniques. If we concern digital filters or transformers the most obvious proposition is to apply lower order filters or transformers. In the case of recursive discrete transformers this means to omit (switch off) some of the channels of the structure (see Fig.4). It is an obvious requirement to maintain e.g. the orthogonality of the transformation, i.e., as an example, we can consider omitting every second channel

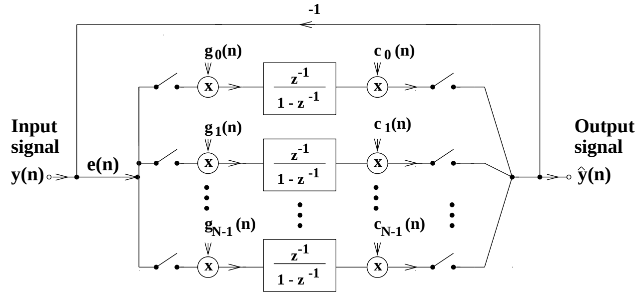

Figure 4: Recursive transformation based signal processing structure symbolizing the case of missing computational capacity. $c_m(n) = e^{j2\pi(mn/N)}$, $g_m(n) = (1/N)e^{-j2\pi(mn/N)}$, $m = 0, 1, \dots, N-1$

from the structure and switch to a transformer of order $N/2$. In Fig.5 the decomposition and reconstruction of a triangular waveform consisting of odd harmonics is illustrated. The channels dedicated to the even harmonics are switched off for a while and switched on somewhat later. As a further example Fig.6 shows the case of a periodic triangular waveform where each period consists of $N_1 = 13$ samples while the block-size of the transformer remains $N = 16$. The side-effects of applying reduced-order transformer can be followed on the diagram.

4 RECONFIGURATION TRANSIENTS

Anytime algorithms based on feedback systems unavoidably suffer from transients. These well-known phenomena are due to the dynamic nature of the signal processing structures applied. Both parameter and structure adaptations generate transients. The nature of these transients depends not only on the transfer function of the filters to be implemented, but also on the actual signal processing structure [8]. It is important to note that the so-called orthogonal structures (see e.g. [1]) provide smaller transients as some other structures do. For this

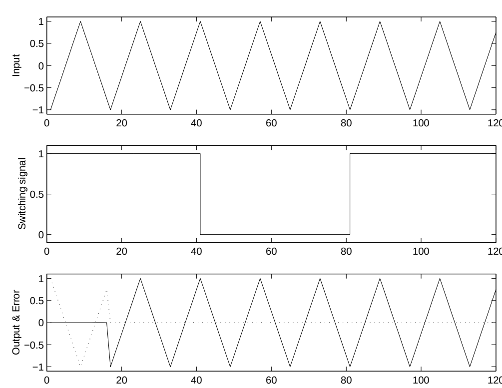

Figure 5: Decomposition and reconstruction of a periodic triangular waveform input consisting of odd harmonics (the case of limited computational capacity): output – continuous line, error – dotted line

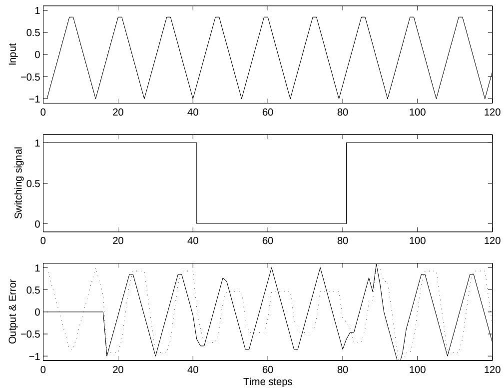

Figure 6: Decomposition and reconstruction of a triangular waveform input with period $N_1 = 13$ consisting of odd harmonics (the case of limited computational capacity): output – continuous line, error – dotted line

very reason the implementation of anytime algorithms must be performed using structures having good transient behavior. This structure dependency is strongly related to the "energy distribution" within the processing structure. The structures on Figs.1 and 4, and all its versions meet the general conditions of having relatively small transients, however, controlling transients in reconfigurable signal processing structures is still an important area of investigations and research [8].

5 ACCURACY PROBLEMS

Concerning anytime algorithms it is an important issue how can we handle accuracy problems. Accuracy can be worse both due to the lack of the appropriate input information, and to the less accurate calculations. The characterization of the signal processing results is relatively easy if only the stationary responses are to be considered. Unfortunately, however, in time critical applications handling uncertainty issues in real-time can be of great importance. This means that the concept of anytime algorithms must be extended toward estimating the reliability of the calculated results. Such an extension should be able to provide error bounds, and similar uncertainty measures. In principle these measures can be generated using classical approaches, however, the elaboration of their conceptual background needs further efforts. How to characterize, e.g. a simple recursive averaging if some of the input samples fail to arrive in time? How to characterize the usual prediction-correction schemes from accuracy point of view if the correction can not be performed? How to continue if due to the lack of input the estimated outputs becomes useless? This topic is strongly related to some aspects of intelligent computing [9], where performance evaluation involves different accuracy issues, as well. Similarly to the problems of transients, concerning accuracy further investigations are required.

6 CONCLUSIONS

In this paper a special class of adaptive algorithms is introduced. These adaptive algorithms are considered to improve the performance of signal processing under conditions of extreme computational burden. The digital signal evaluation algorithms of this class can tolerate temporal shortage of computational power and the lack of input data. If the shortage is indicated, a reconfiguration is performed within the signal processing system. There are several alternatives to perform such changes, however, the application of certain dedicated signal processing structures promises relatively good behavior. The reconfiguration transients and the related uncertainty issues ask for further research.

REFERENCES

[1] Padmanabhan, M., K. Martin, G. Péceli, *Feedback-Based Orthogonal Digital Filters*, Kluwer Acad. Publ., 1996.

[2] Baron,C., J.-C Geffroy, G. Motet ed., *Embedded System Applications*, Kluwer Acad. Publ., 1997.

[3] Sztipanovits, J., D.M. Wilkes, G. Karsai, Cs. Biegl, L.E. Lynd, "The Multigraph and structural adaptivity," *IEEE Trans. on Signal Processing,* Vol. 41, No. 8, pp. 2695-2716, Aug. 1993.

[4] Péceli, G., A.R. Várkonyi-Kóczy, "Block-Recursive Filters and Filter-Banks," *1997 IEEE Int. Conf. on Acoustics, Speech, and Signal Processing, ICASSP'97,* Munich, Germany, pp. 2001-2004, Apr. 20-24, 1997.

[5] Várkonyi-Kóczy, A.R., S. Theodoridis, "Fast Sliding Transforms in Transform Domain Adaptive Filtering," *1997 IEEE Int. Conf. on Acoustics, Speech, and Signal Processing, ICASSP'97,* Munich, Germany, pp. 2009-2012, Apr. 20-24, 1997.

[6] Nagy, F., "Measurement of Signal Parameters Using Nonlinear Observer Theory," *IEEE Trans. on Instrumentation and Measurement*, Vol. 41, No.1, Febr. 1992, pp. 152-155.

[7] Várkonyi-Kóczy, A.R., "Efficient Polyphase DFT Filter Banks with Fading Memory," *IEEE Trans. on Circuits and Systems II*, Vol. 44, No. 8, Aug. 1997, pp. 670-673.

[8] Péceli, G., T. Kovácsházy: "Transients in Reconfigurable Systems," *1998 IEEE Instrumentation & Measurement Technology Conf., IMTC'98,* St. Paul, Minnesota, USA, May 18-21, 1998, accepted.

[9] Várkonyi-Kóczy, A.R., G. Péceli, T.P. Dobrowiecki, O. Takács: "Measurement Technology of Intelligent Systems," *1998 IEEE Int. Conf. on Intelligent Engineering Systems, INES'98*, Vienna, Austria, Sep. 17-19, 1998, accepted.