ERAF

Multi-channel Data Acquisition and Processing

for Bridging the Gap between Computers and the Real World

ERAF projekts Nr.VPD1/ERAF/CFLA/05/APK/2.5.1/000024/012 "Daudzkanālu sensoru sistēmu radīšana biomedicīnisko, ekoloģisko un industriālās ražošanas datu iegūšanai un ievadīšanai datorizētās sistēmās"

Projektu līdzfinansē Eiropas Savienība.

Project Nr. VDP1/ERAF/CFLA/05/APK/2.5.1./000024/012
"Development of multi-channel systems for acquisition of data from biomedical, ecological and industrial systems and transferring them to computerized systems"
Co-sponsored by the European Union

Institute of Electronics and Computer Science

Riga, Latvia, 2008

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Organizācija, atbildīga par šīs publikācijas saturu un projekta ieviešanas organizāciju, ir Elektronikas un Datorzinātņu Institūts (projekta vadītājs I.Biļinskis).

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Organisation responsible for the content of this publication and for organizing applications based on this project is Institute of Electronics and Computer Science, Latvia (project manager I.Bilinskis).

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1. Preface

Massive data acquisition from real life objects and supplying of computers with this information in an effective way evidently is vital to realizing full potential of computer system applications in many areas. The R&D project "Development of multi-channel systems for acquisition of data from biomedical, ecological and industrial systems and transferring them to computerized systems", considered in this booklet, is focused on achieving progress in this direction by finding ways how to avoid some typical data acquisition bottlenecks. Thus the project is dedicated to resolving key problems in the area of interfacing sensors and sensor networks with computers. To reach the goal of this project and to create competitive multi-channel data acquisition systems for supplying computers with information gathered from the real world objects, a flexible approach to complexity-reduced multi-channel data acquisition from a large quantity of sensors has been developed and used. It is discussed in this booklet and the R&D results obtained in this area are summarised. The developed methods, systems and algorithms for data acquisition from wideband, event timing and large distributed clusters of signal sources are discussed with emphasis on data gathering from a large quantity of signal sources. Special signal digitising techniques, including pseudo-randomised multiplexing, time-to-digital conversions and signal sample value taking at time instants when the input signal crosses a sinusoidal reference function, are used for that. The versatility of the considered data acquisition systems is achieved by using modular system design. Development of the discussed massive data acquisition systems is based on the knowledge accumulated over a long period of time in the area of Digital Alias-free Signal Processing.

Project Manager

Prof. I.Bilinskis

2. Project essentials

2.1 Goal and objectives

Goals:

1. To increase the competitiveness of produced IT products on the basis of applied research results by creating more effective and complexity-reduced multi-channel systems for sensor data acquisition.
2. To develop, make and test the developed data acquisition systems.

Objectives:

1. To develop an innovative technology for massive sensor data acquisition from up 1000 sensors simultaneously.
2. To develop 3 types of data acquisition systems for supplying computers with information gathered from various types of sensors.
3. To develop 4 algorithms for data acquisition and pre-processing.
4. To develop and make 3 types of data acquisition pilot systems.
5. To develop and patent an original method and system for massive data acquisition.

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2.2 R&D Activities

To reach the project objectives and provide for data gathering and smooth transfer of this information to the computers, the involved data acquisition systems have to be sufficiently flexible so that computer links to various types of signal sources might be realized. This means that they should provide for data acquisition from different types of sensors. To provide for data acquisition from differing kind of signal sources and still avoid excessive complication of the system, modular approach to the design of it has been chosen. Each of the modules then represents a specific system for data acquisition and it is possible to aggregate these systems seamlessly into re-configurable modular structures meeting functional requirements of specific application cases.

To reach the project objectives, the Activities of the project were focused on the following:

1. Development of systems for multi-channel sensor data acquisition.
2. Development of algorithms supporting multi-channel sensor data acquisition.
3. Experimental research activities.
4. Micro-miniaturization activities.

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2.3 Targeted system features

R&D activities of the project were focused on achieving the following:

1. Simultaneous data acquisition from a large number of signal sources.
2. Remote sampling of signals closely to the signal sources.
3. Complexity-reduced design of front-end devices.
4. Low power consumption of the front-end devices.
5. Fast low-power data pre-processing.
6. Flexibility of specific multi-channel data acquisition system design.

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2.4 Approach to achieving the targeted results

The traditional techniques applied for multi-channel data acquisition are based on multiplexing input signals either before or after analog-digital conversions of them. In the first case, to acquire data from a large quantity of signal sources, the original signals would have to be passed over transmission lines and then switched by a hierarchy of multiplexers. Protection of the signals against corruptions imposed by noise and switching transients then represents a problem, especially if the number of the inputs is really large, close to 1000. In the second case, signals are multiplexed after they have been digitised and that certainly is much better but an ADC would have to be used for digitising each of the input signals. Then the required power consumption of the front-end parts of the systems might be unacceptable. So both of these techniques have substantial drawbacks considerably complicating the task of simultaneous massive data acquisition. In particular, multiplexing then would be performed periodically and that would mean that the upper frequencies of the input signal spectra would directly depend on the number of the input channels.

The new emerging Digital Alias-free Signal Processing (DASP) technology [1] is much more flexible and, consequently, it is better suited for development of versatile data acquisition systems appropriate for obtaining information from large sensor clusters. For these reasons, this DASP technology and the knowledge accumulated in this area were chosen as the platform for development of the further discussed data acquisition methods, hardware and software. This approach makes it possible to obtain a number of benefits. They are discussed further. In particular, the upper frequencies of the input signal spectra then not necessarily depend on the quantity of the input channels and input signals are sampled directly at their sources by front-end devices which are much simpler than the traditionally used ADCs. Consequently, the power consumption of these front-end devices might be significantly lower.

On the other hand, DASP technology is specific. Therefore special hardware and software tools have to be used to implement the chosen approach.

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3. Knowledge platform: DASP technology

While the term “Digital Alias-free Signal Processing” place emphasis on the alias suppressing capability of the DASP technology, it offers much more. Actually the basic point of this technology is the approach to analog signal digitising and to processing of the digitised signals. The dominating role of signal digitising is duly recognized and the process of system design typically starts by analysis of the conditions for signal digitising dictated by the given application. The required signal sampling and quantising modes, best fitting to the specifics of this application, are defined at this stage. The algorithms for processing the digital signal then have to be matched to the technique used for the signal digitising.

The knowledge accumulated in this area over a long period of time represents the theoretical basis for this project. Theory of this technology for signal encoding and digital processing is discussed in the monograph: I. Bilinskis “Digital Alias-free Signal Processing”, Wiley, 2007. Results of the considered project, partly included in the book, represent the latest development of this theory.

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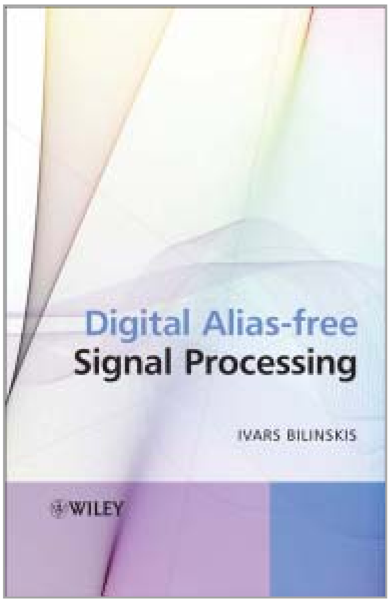

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3.1 Exploited DASP concept and basic methods

The project is based on the DASP concept of the Distributed Remote Sampling Analog-to-Digital conversions and the algorithms for alias-free processing of nonuniform data streams [1].

That leads to a specific approach to design of systems for massive data acquisition. In particular:

- Remote complexity-reduced sampling is based on crossings of signals and sine-wave reference function.
- Sampled signals are represented by positioned in time pulse sequences.
- Streams of digitally timed uniform events are used as digital signals representing the acquired data.
- Systems are designed specifically for data acquisition from wideband, event timing and large distributed clusters of signal sources.
- Massive data acquisition is based on pseudo-randomised time-sharing, analog-to-event and time-to-digital conversions.
- Modular system design is used for achieving versatility in customizing the data acquisition systems.
- Fast DASP algorithms are exploited for data pre-processing.

Discussion of these project specifics follows.

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3.2 Distributed Remote Sampling Analog-to-Digital Converter

Structure of the distributed remote sampling Analog-to-digital converter is given in Figure 3.1.

As can be seen, the sampling and quantising operations, in this structure, are distanced. They are also matched to the specifics of the basic task: to provide for data acquisition from a large quantity of sensor signal sources. That is essential for achieving the further pointed out significant advantages. Basically this approach makes it possible to use many remote samplers at the front-end of the structure and these samplers and to gather data from them in a rational way.

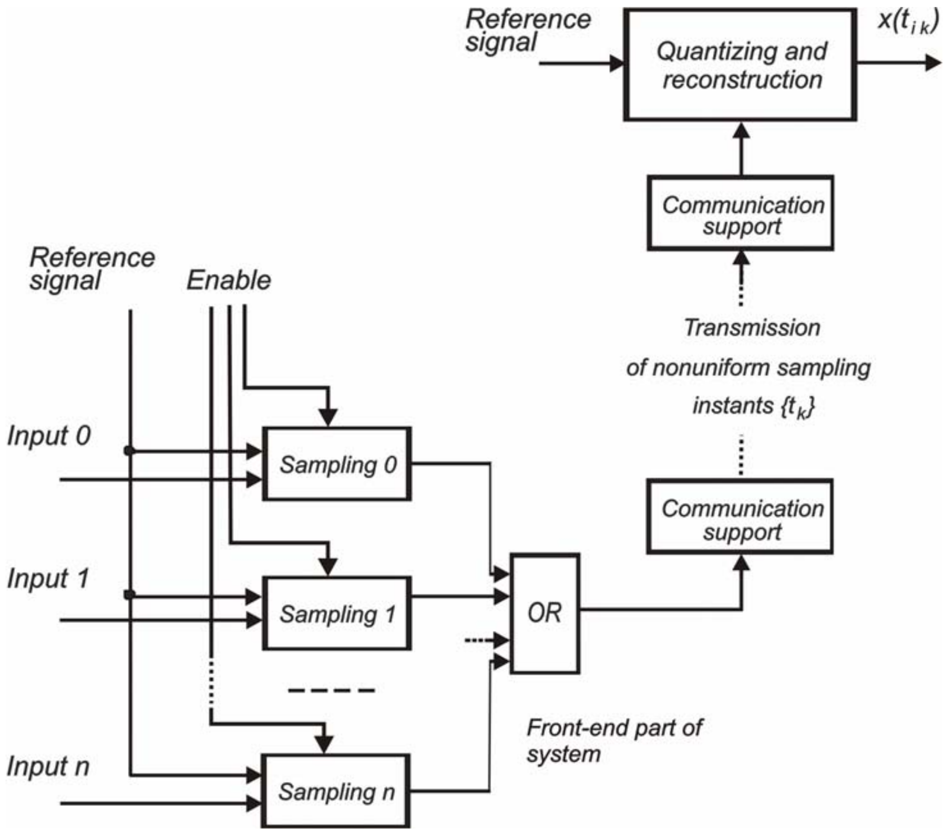

Figure 3.1 Block diagram of the distributed Analog-to-digital converter

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The further discussed methods and techniques for Remote Sampling of multiple inputs in parallel are based on the concept of the Distributed ADC. According to it:

- A reference sine-wave with given and stable parameters is generated and used for performing input signal sampling.
- The samplers compare the input signals with this reference sine-wave and form output pulses at the time instants when the events of the signal and the reference crossings take place.
- The sequences of the comparator output pulses, representing the information carried by the respective input signals, are transmitted to the central part of the Distributed ADC where they are transformed into signal sample values.
- Performing of the sampling operation is enabled sequentially in a time-sharing mode.
- Sequential enabling of the remote samplers in fact fulfils the input signal multiplexing function.
- A large quantity of this kind of samplers might be used in parallel.

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3.3 Advantages of the Distributed ADC concept.

The concept of Distributed ADC is attractive as it has significant potential for building high-performance multi-channel data acquisition systems on this basis. Specifically, it provides for:

- Reconfigurable structure of the system making possible to acquire data from signal sources distributed over some area.
- Remote sampling of signals is performed closely to their sources crucial for avoiding corruptions of input signals by noise.
- No switchings of analog signals performed.
- Multiplexing of analog input signals replaced by time-shared enabling of the sampling operation.
- Simultaneous data acquisition performed in parallel from a large quantity of signal sources.
- Sampled signal data compression achieved by using positioned in time pulse sequences for representing them.
- Possibility of using a digital representation of the original signals providing for data compression and fast pre-processing.
- Reduced complexity designs of front-end devices.
- Low power consumption of the front-end devices.

Success in this first of all depends on choosing the most suitable method and techniques for digitising of the input signals. Special approach to digitising, based on indirect randomisation of this operation, has been used.

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3.4 Randomised digitising for multi-channel application

Analog signal digitising usually is performed by signal sample value taking periodically with subsequent quantising of these sample values. However this classical periodic approach to digitising actually is rigid as the possibilities of matching digitising to conditions and requirements of a specific application are limited.

Much more flexible are various randomised sampling methods and techniques [1]. Randomisation of sampling might be direct or indirect. The earlier developed methods for deliberate direct randomisation of the sampling operation more often than not target elimination of the aliasing effect crucial for widening the frequency range where signals could be processed fully digitally. In this case there are other objectives that are addressed. To reach the basic goal of this project, the approach to digitising should be effective for development of systems for massive data acquisition from a large quantity of signal sources distributed over some area. These systems are to be based on the distributed structures of analog/digital converters. Under these conditions, better results could be obtained by using another randomisation approach differing from the mentioned above.

Specifically, it is suggested **to perform the sampling operation of a signal as detection of the time instants when this signal crosses a reference function with given and stable parameters.** Randomisation in this case actually is indirect as the signal and the reference function crossings just occur nonuniformly at unpredictable time instants. For a number of further explained reasons, it is preferential to use sinusoidal reference function.

Sampling based on Sine-wave crossings is further denoted as SWC sampling.

Various options of forming and processing the sequences of signal sample values taken at time instants when the signals cross reference sine-waves, partly decribed in [1], are further studied and discussed. Performance characteristics of the considered multi-channel data acquisition systems basically depend on this type of signal digitising and reconstruction procedures. The methods and techniques used for this, vital for achieving the targeted project results, have been theoretically and experimentally studied. Discussion of the obtained results follows.

4. Studies of Remote SWC Sampling

4.1 Basic model of SWC Sampling.

To show what are the functions fulfilled by the scheme performing remote SWC Sampling, let us consider the basic structure of a single-channel analog-digital converter performing the sampling operation on the basis of sine-wave crossings (SWC Sampling) given in Figure 4. 1. The merit of it is the simplicity of the front-end design and the fact that it is well suited for remote signal sampling applications, specifically, for building the distributed structure ADCs.

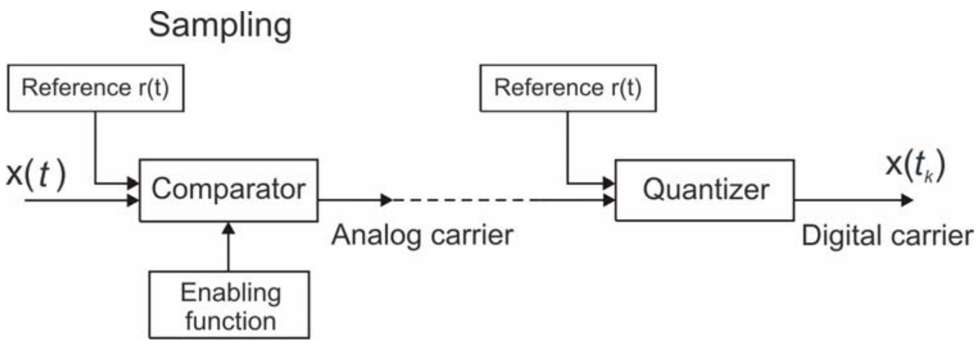

Figure 4.1 Block diagram of a single-channel remote sampling ADC

The most responsible element of this converter apparently is the comparator used for detection of the time instants $\{t_k\}$ at which the signal *x(t)* intersects the reference waveform. The physical output *y(t)* of the comparator is formed as a pulse whenever the sampler is enabled by a function *z(t)* and a crossing of the input signal *x(t)* and the reference function *r(t)* takes place. These pulses *y(t)* at the sampler output are formed so that they carry the timing information indicating the exact crossing instants t_k of the signal and the reference function. It is assumed that the output pulses, having extremely short front edges of negligible duration, are formed with a constant delay after each crossing of the signal and the reference sine-wave function. Thus the sampler output signal is carrying the original information encoded as the sequence $\{t_k\}$ of the sine-wave crossing instants. It is transferred over some shorter or longer distance to the quantiser used for recovery of the input signal sample values.

The signal sampling has to be executed in a way ensuring that the information carried by the input signal *x(t)* is fully transferred to the sampled signal given as the sequence $\{t_k\}$. Evidently fulfilling of this requirement is crucial for recovery of the input signal sample values encoded by this timing information. Specific digital representation of the input signals is used for that.

4.2 Representation of input signals

Figure 4.1 draws attention to the fact that **the input signals are represented by the sequences of the signal and the reference crossing instants.**

Actually issue of the digitised representation of the input signals is essential for the suggested approach to development of the multi-channel systems for data acquisition.

To emphasize the difference between the traditionally used and the suggested signal representations, the classical digital signal is shown in Figure 4.2.

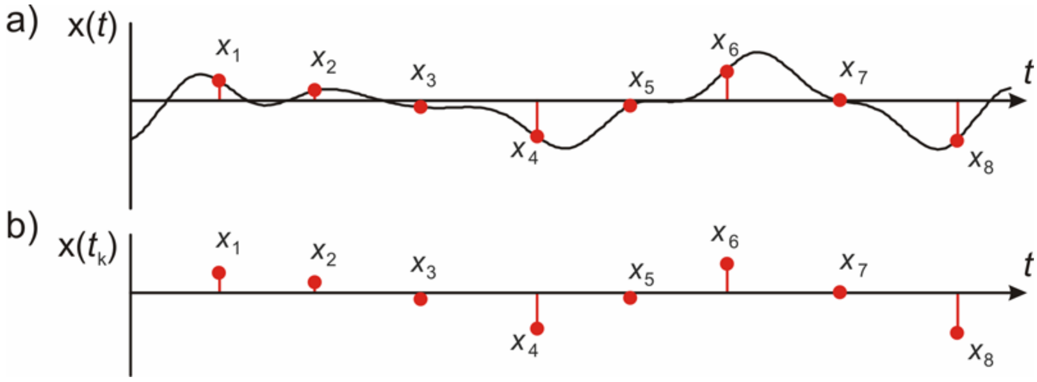

Figure 4.2 Classical representation of an input signal (a) by the digital (b).

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The suggested digitised signal representation differs from the classical one substantially. Figure 4.3 illustrates this graphically.

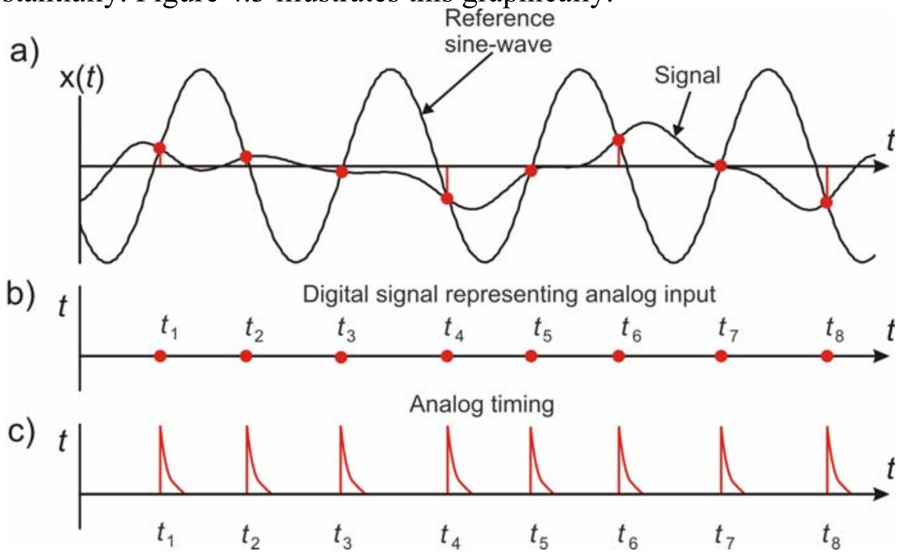

Figure 4.3 Suggested and used digitised representation of an input signal (a).

Actually two types of information carriers, analog and digital, are used for transmission of the input information:

- Analog carrier: train of position-modulated short pulses (Figure 4.3 c);
- Digital carrier: sequence of digital crossing instant values *{t_k}* (Figure 4.3 b).

The later is considered and called as digital signal representing the sensor signal at the respective input. It is transferred to a computer before or after some pre-processing.

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4.2.1 Analog carrier of information

The Analog carrier is used primarily for gathering and transmission of data from the cluster of remote samplers to the master part of the distributed ADC (Figure 4.4 d).

While various reference functions could be used at SWC Sampling, preference is given to sine-waves as it is easy to generate them and to stabilize their parameters, they have extremely simple and well-defined spectra and they are convenient also for reconstruction of the sampled signals. Figure 4.4 illustrates analog signal digitising performed on the basis of SWC Sampling. In this case the reference function $r(t)=A_r sin(2\pi f_r t+\varphi_r)$, where A_r, f_r, and φ_r are the amplitude, frequency and phase angle of it, respectively.

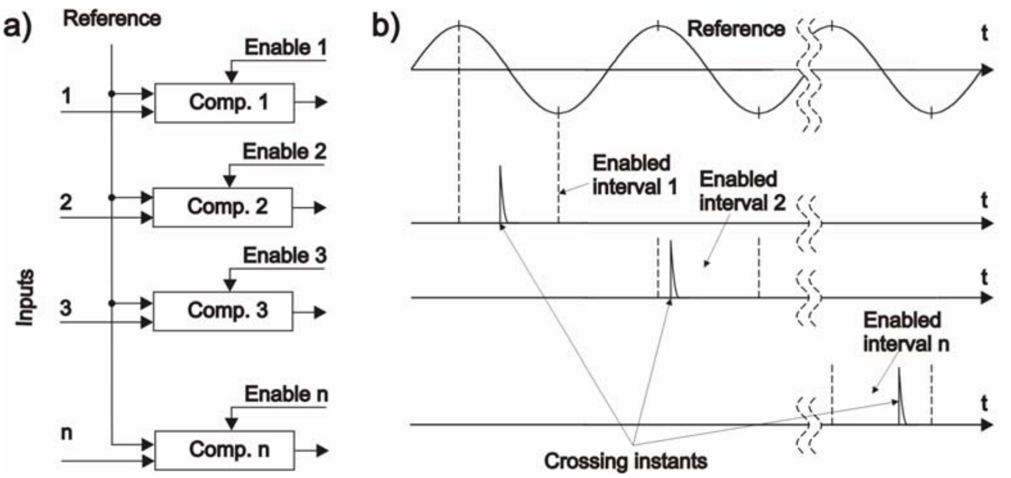

Figure 4.4 Illustration of analog signal digitising performed on the basis of sine-wave crossings. (a) block-diagram; (b) enabling crossings of the signal the reference sine-wave and the output signal given as a sequence of pulses positioned in time by the sine-wave crossings.

The train of pulses positioned in time by the sine-wave crossings represents the discussed Analog carrier of the input information.

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4.2.2 Digital carrier of information

The Digital carrier is used at the stages of data reconstruction and/or their pre-processing and data transfer to computers (Figure 4.4 e).

While this type of signal digital representations is specific and successful using of it requires some skills, it leads to obtaining of significant benefits. Specifically they are the following:

- Achieving time and energy saving at data transmission (each single symbol carries multi-bit information).
- Compatibility with the Ultra-wideband wireless data transmission technology.
- Complexity-reduction of algorithms for data pre-processing.

Note that only those crossings are taken into account which happen during the time intervals when the respective comparator is enabled by a specially generated enabling function (Figure 4.4 b). This enabling function is introduced and exploited for executing the input multiplexing. In this way the analog input signal switching could be avoided and that certainly is a significant positive fact.

Another role the enabling function is playing is taking part in forming the digital information carrier. It is essential that the distribution of the time intervals between successive digital values of this signal is characterized by a relatively small standard deviation. Sequential enabling of the remote samplers and enabling of them during half-waves of the reference function helps in achieving this.

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4.3 Essential specifics of the digital signals obtained at SWC Sampling

The digital signals obtained in result of SWC Sampling clearly are highly specific and, in this sense, unusual. Specifically:

- Their definition differs from the classical and traditionally used one. Digitally timed sampling event sequences or, in other words, sampling point processes in time are considered as digital representations of the respective analog inputs.
- The sampling time instants are signal-dependent and time intervals between them are nonuniform.
- On the other hand, these sampling intervals not necessarily are random. As they are related to the input signals, there might be periodicities present in the digital signals representing the original analog signals.
- Therefore overlapping of frequencies or aliasing might be expected.
- There might be also cross-interference between signal components typical for nonuniformly sampled signals.
- Actually the specific properties of the digital signals of this kind are not pre-determined, they strongly depend on the input signal and the used reference function.
- Hence the considered digital signals indeed are highly specific and they have features unparalleled by other types of digital signals.
- Remarkably, the envelope of the digital signal instantaneous value sequences, in the case of this kind of distributed analog-to-digital conversions, remains constant no matter what is the spectral content of the respective analog signals.

The later represents a powerful advantage of the considered type of signal sampling as the constant envelope of various digital signals, obtained under the mentioned conditions, leads to a number of options in complexity-reduced processing of them. Some of these options, including complexity-reduced spectrum analysis and massive data acquisition, are briefly discussed in [1, 2, 9, 10].

Consideration of various issues characterizing this type of digital signals, is started by discussion of the frequency overlapping or aliasing phenomenon observed at this kind of signal digitising.

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4.3.1 Aliasing

To observe how frequencies are overlapping in the considered signal digitising case and how aliasing does impact the obtained spectrum, DFT of a signal containing only three components at frequencies f_1, f_2 and f_3 was performed.

The spectrogram obtained in this way is given in Figure 4.5.

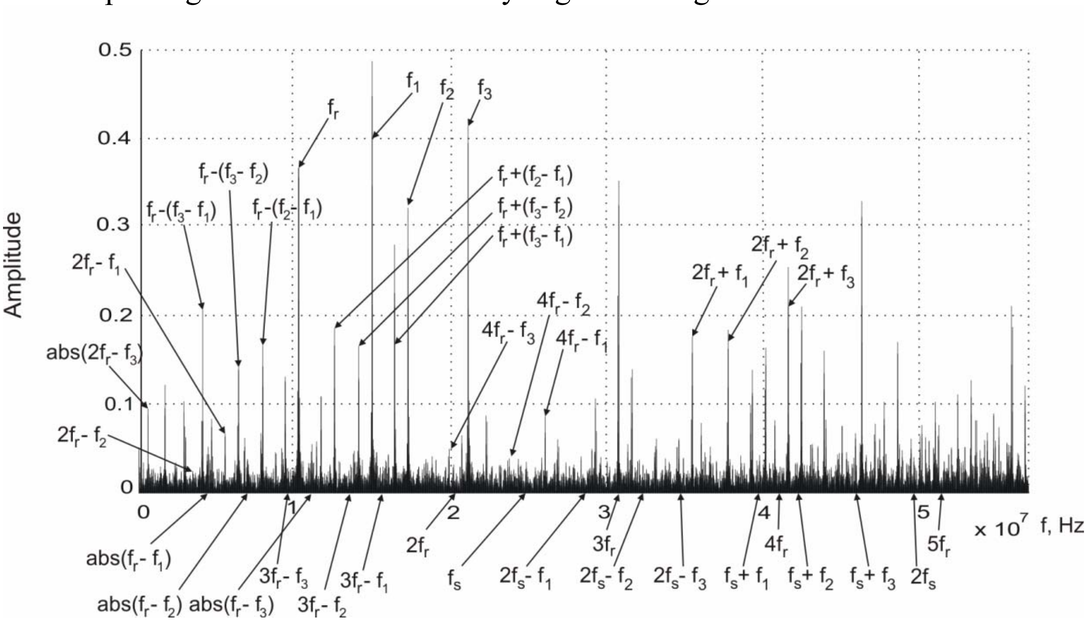

Figure 4.5. Illustration of the aliasing frequency pattern.

There are many peaks in this spectrogram in addition to the peaks that might be considered as spurious noise. Actually the most pronounced additional peaks appear in connection with the aliasing effect. In this specific sampling case, positions of the peaks related to aliasing depend on the frequencies related to the periodicity in the sampling point stream. This means that both the reference frequency and the mean sampling frequency play some role in that [9].

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The frequencies f_i and f_n of the signal components, along with other typical frequencies such as the reference frequency f_r and the mean sampling rate f_s, are indicated in the spectrogram in Figure 4.5. These frequencies define the positions of the aliasing frequencies. The pattern of the peak positions on the frequency axis helps to reveal the essence of the aliasing conditions.

Actually there are three rows of the expected aliasing frequencies. They are:

1. $f_i;\ f_r\pm f_i;\ 2f_r\pm f_i;\ 3f_r\pm f_i;\ \ldots;\quad i=1, 2, 3, \ldots$

2. $f_s\pm f_i;\ 2f_s\pm f_i;\ 3f_s\pm f_i;\ \ldots;\quad i=1, 2, 3, \ldots$

3. $f_r\pm(f_i\pm f_n);\ 2f_r\pm(f_i\pm f_n);\ 3f_r\pm(f_i\pm f_n);\ \ldots;$ for $i\neq n$ and $i=1, 2, 3, \ldots; n=1, 2, 3\ldots$

The relationships defining the aliasing conditions in this case are more complicated than in the cases where the sampling process is pre-determined and does not depend on the signal.

- There are more aliases than in the case of the conventional periodic sampling.
- The aliasing process is suppressed. The peaks at frequencies of true signal components are much stronger then the aliases. That is due to the fact that the sampling process is both periodic and nonuniform.
- The non-uniformities of the sampling intervals lead to this effect of alias suppression. This helps to separate them from the signal components.
- The aliases related to the indicated three frequency rows might be not equally strong. This difference in aliasing can be traced to the differences in signal sampling conditions.
- Aliasing illustrated by Figure 4.5 occurs also at frequencies related to the reference frequency and signal component differences/sums $(f_i \pm f_n)$; ...; for $i \neq n$. That is unusual.

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4.3.2 Cross-interference

Actually the question is arguable whether the aliasing effect does take place in the discussed case at all. The following consideration might be pointed out:

- It is clear that there is no full-scale frequency overlapping. While peaks appear in the spectrogram of Figure 4.5 at frequencies belonging to the rows 1, 2, 3, they are significantly suppressed.
- This type of aliasing is considered as fuzzy aliasing [1].
- It might be even assumed that there are cross-interference effects rather than aliasing while these effects are amplified at the indicated frequencies [9].
- The impact of this cross-interference on the signal spectrograms directly depends on the signal sampling conditions.
- Introduction of the enabling function actually has the effect of sampling regularization.
- Increasing the interval during which the sampling operation is blocked results in smaller power of the introduced element of the randomness and that in turn leads to reduction of the effects induced by the cross-interference and to increasing of the peaks in the spectrogram due to aliasing.
- The spectrograms displayed in Figure 4.6 illustrate this.

The signal in this particular case has only three components and the sampling conditions are varied. In result, the peaks in the spectrogram indicating the aliases and the power of the spurious frequencies vary as well. While all signal and the reference crossing events are taken into account in the first case illustrated by the spectrogram in Figure 4.6 (a), the sampling operation is enabled only during each sixth half-period of the reference function in the second case (Figure 4.6 (b)) and during each tenth half-period in the third case (Figure 4.6 (c)). As can be seen, this approach to sampling significantly impacts the features of the obtained digital signal.

Thus the cross-interference, under certain conditions, might significantly impact the result of processing the digital signals obtained in result of SWC Sampling. The problem of avoiding the negative effects due to this interference is considered in Section 4.4.

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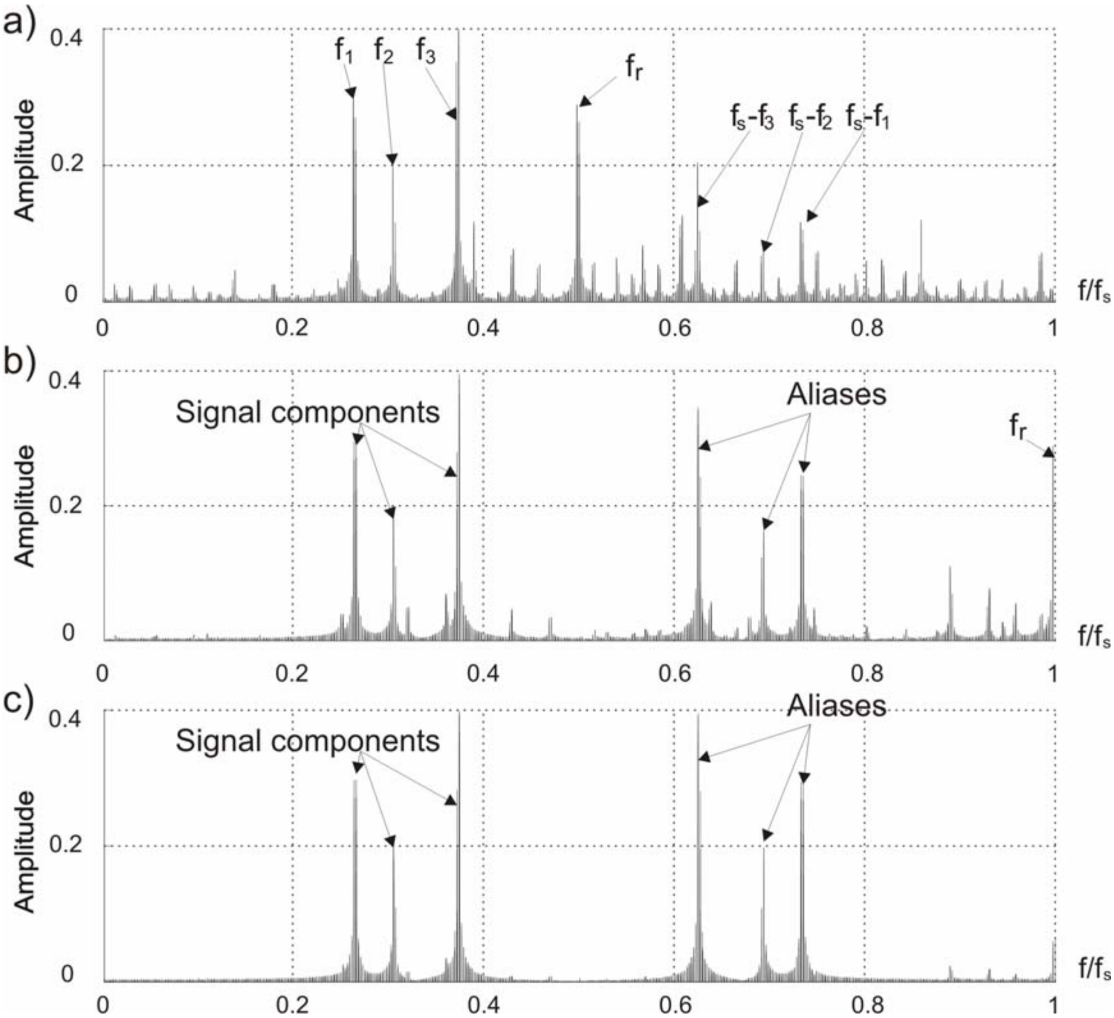

Figure 4.6 Weakening of the cross-interference impact by introducing sampling regularization.

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4.3.3 Impact of the enabling function on the regularity of sine-wave crossings

Features of the digital signal formed in result of the analog input signal and the reference sinusoid crossings obviously depend on both of the involved processes. Note the following:

- The basic features of concern are the regularity of the intervals between the sampling (crossing) points and the accuracy of the indicated crossing time instants.
- Both of these features to some extent depend on the frequency of the used reference sinusoid.

- Therefore the question arises how to select the appropriate value of this parameter of the reference function.
- In general, at low reference frequencies the relative resolution with which the crossing events are fixed in time is better than at high reference frequencies. From that point of view, low reference frequency seems to be preferable.
- The crossing point pattern in time then basically depends on the signal frequency content as the signal typically crosses the reference function a number of times during each period of the reference and the sampling point process then is characterized by relatively high values of the ratio *σ/μ* as it is rather irregular.

Consequently, distortions of the sampled signal processing due to the cross-interference phenomenon then might be expected. A typical case of SWC Sampling carried out at low reference frequencies, is illustrated in Figure 4.7 (a).

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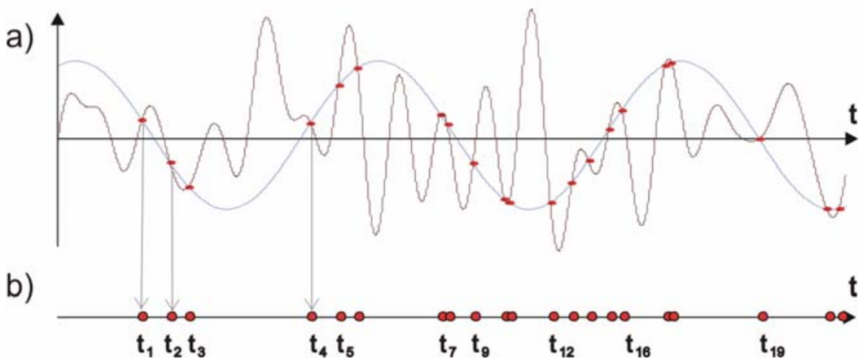

Figure 4.7 Digital signal (b) formed in the case where a signal crosses a low frequency reference function as shown in (a).

Another drawback typical for the cases where low frequency reference functions are used is illustrated in Figure 4.8. As can be seen, there are relatively large gaps between the detected crossing points. In result, essential information is lost during these time intervals. The formed digital signal, namely, the sampling (crossing) point process, given in Figure 4.8(b), again is nonuniform and it is characterized by large values of the ratio *σ/μ* (standard deviation/mean value).

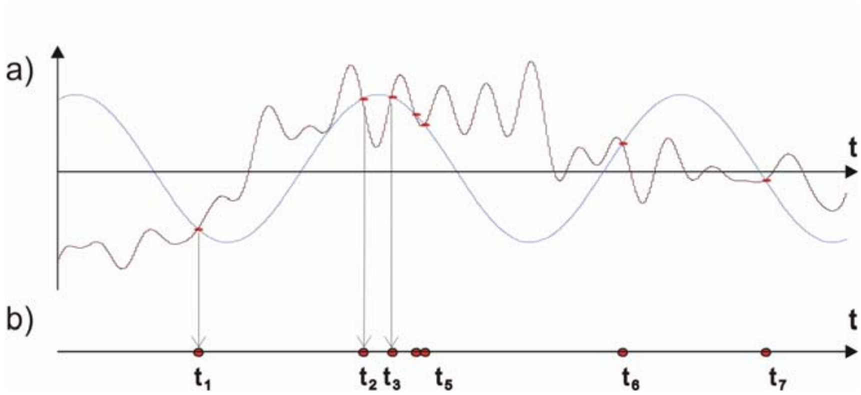

Figure 4.8. Illustration of a sampling case where the reference sinusoid is not crossing the signal within relatively large signal segments. (a) crossings of the signal and the reference function; (b) obtained digital signal.

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The digital signal, obtained in the cases where relatively high frequency reference function is used, typically is more regular as the reference function basically dictates the pattern of crossings. A typical sampling process of this kind is given in Figure 4.9. As can be seen, the formed digital signal indeed is more regular in this case.

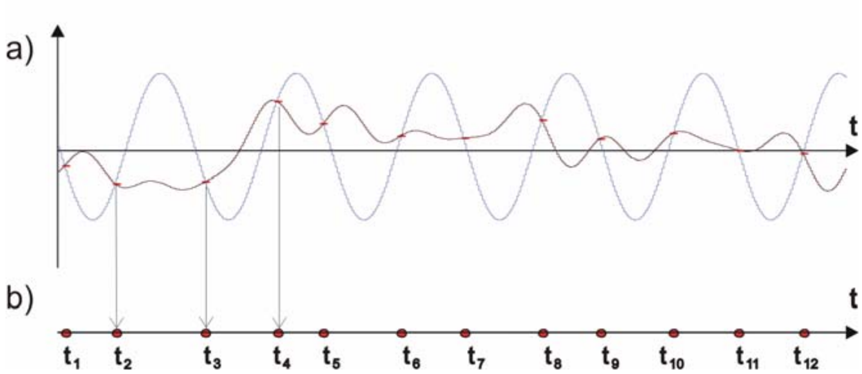

Figure 4.9. Illustration of the digital signal forming in the case where relatively high frequent reference function is used. (a) SWC sampling process; (b) formed digital signal.

As the given illustrations show, usually it is preferable to use reference sinusoids at relatively high frequencies. However even then the irregularities of the obtained digital signal might lead to significant distortions of the signal processing results. These distortions are typical for any kind of nonuniform sampling and they are caused by the cross-interference between the nonuniformly sampled signal spectral components. While usually it is desirable to work at higher reference frequencies, increasing of the reference frequency is limited. And the consideration of the time-resolution basically is the dominating factor setting up this limit.

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4.3.4 Enabling function as a tool for suppressing aliasing

As it is shown above, the aliasing problem exists also in the case of SWC Sampling. This problem usually comes up when sampling could not be performed at sufficiently high frequencies. To achieve the capability of suppressing the aliasing effect, the enabling function could be used as a tool for arranging this sampling process in a way leading to elimination of aliasing. The remote samplers have to be enabled then in a pseudo-randomised way as shown in Figure 4.10.

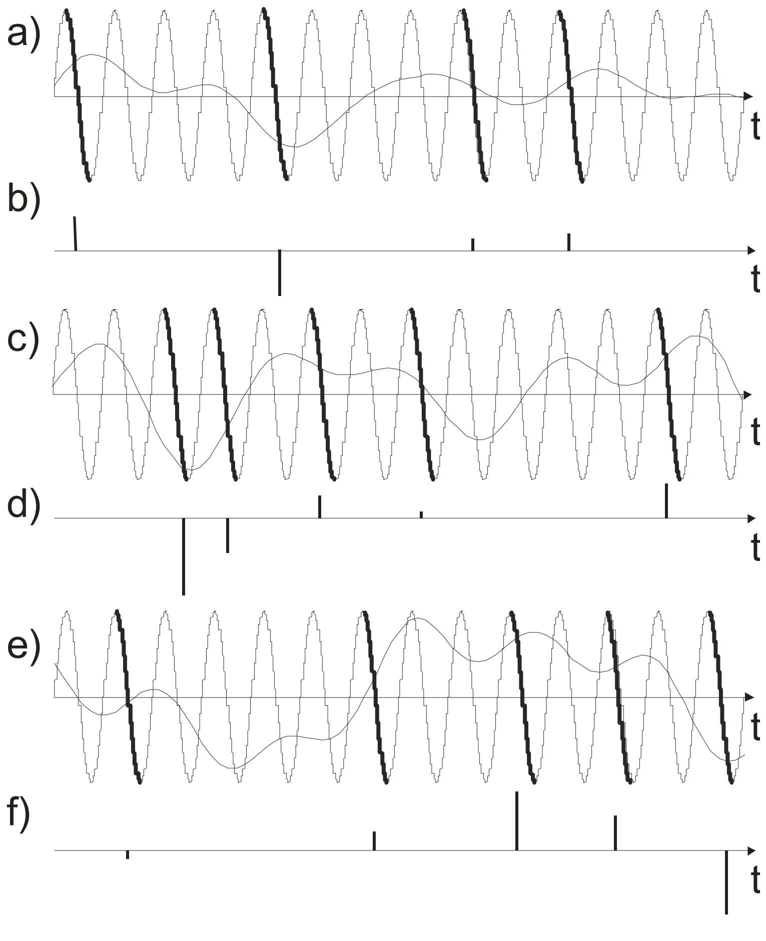

Figure 4.10. Pseudo-randomised enabling (bold half-waves of the reference function in (a), (c) and (e)) of the sampling operation carried out in three particular input channels. The irregular pseudo-randomised signal sample value sequences (b, d and f) carrying the information originated at the three respective sensor signal sources.

Note that this approach, while it provides the necessary anti-aliasing condition, does not guarantee obtaining of high quality signal processing results. To achieve high precision at signal processing, the nonuniformly taken signal sample values have to be processed in a proper way with the sampling irregularities taken into account. Algorithms for processing nonuniformly sampled signals adapted to the sampling non-uniformities should be used for that as it is described in [1]. The achievable improvement in signal processing achievable in this way [9] is demonstrated further.

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4.4 Constant Envelope SWC Sampling

Diagrams in Figure 4.11 (a) and (b) clearly show that the considered scheme of SWC sampling is signal-dependent and essentially nonuniform. The digital signals obtained in these two cases are given in Figure 4.11 (c) and (d) respectively. As is evident from them, these signals, given as sequences of the digitally timed crossing events, in general, differ significantly.

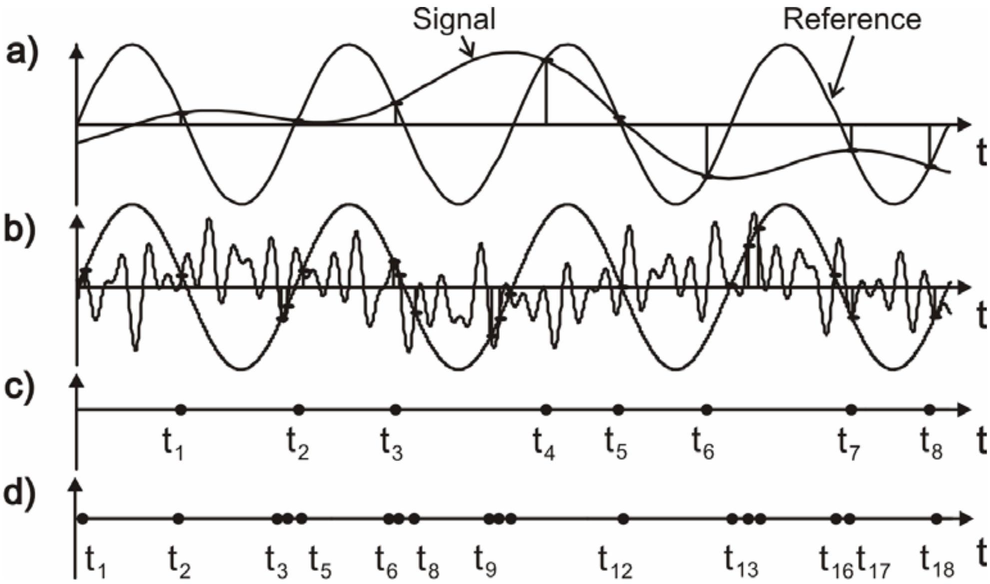

Figure 4.11 Forming of two digital signals on the basis of SWC Sampling for analog input signals with different spectra.

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Both of these digital signals, given in Figure 4.11 (c) and(d), were used to reconstruct the sample values of the respective original signals. The obtained sequences of the reconstructed sample values are shown in Figure 4.12 (a) and (b) with their envelopes added. And that reveals a significant fact. While the digital signals in Figure 4.11 evidently differ considerably, their envelopes are exactly the same. As it should be, because the signal sample values in both cases are equal to the sample values of the same reference sine-wave taken at the crossing instants.

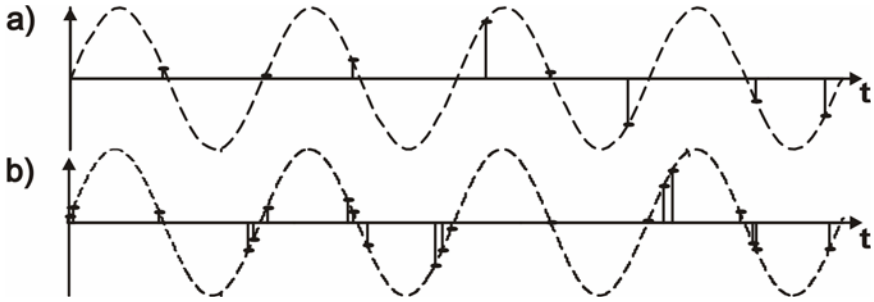

Figure 4.12 Sequences (a) and (b) of the reconstructed sample values of the digital signals shown in Figure 4.11 (c) and (d), respectively.

The fact that SWC Sampling provides Constant Envelope sequences of digital sample values is significant as it leads to far reaching positive consequences. The point is emphasized that algorithms for processing such signals often might be considerably less complicated than the widely used conventional ones. This essential issue is discussed in the following Sections in more detail.

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4.5 Digitally timed event stream transformation options

The best strategy for processing the digital signals given as digitally timed event streams in the most effective way depends on the specific conditions of a given application. However this strategy always is based on clear definition of the targeted qualities of the sampled signal processing procedures. Fortunately, the approach to processing the considered digital signals is to some extent variable. The first decision that has to be taken is how to transform the involved information carriers before their processing.

As it is explained above, the analog information carrier, typically used at gathering of the data from the front-end remote samplers, later on is transformed into a digital signal given as the digitally timed crossing event stream (for example, as shown in Figures 4.3 (b) and 4.4 (e)). After that this digital signal might be either directly transferred to a computer or it might be first pre-processed. So there are some options open. Let us consider the basic ones. They are:

Option 1. **Recovery of the signal sample values and their timing.**

Option 2. **Regularization for achieving the applicability of standard DSP algorithms.**

Option 3. **Direct processing of digitally timed crossing event streams for reducing system complexity**

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4.5.1 Option 1: Recovery of the signal sample values and their timing.

In this case the specific digital signal, similar to the signal shown in Figure 4.3(b), is transformed into a more often used type of signals given in Figure 4.2 (b).

To do that, it is necessary to reconstruct the signal sample values at the crossing time instants. Actually that does not represent a problem. At these time instants $\{t_k\}$ when crossings of the signal $x(t)$ and a reference sine wave $r(t)$ occur, the following equality obviously holds:

$$x(t_k) = r(t_k) = A_r \sin(2\pi f_r t_k + \varphi_r),\ k=1, 2, 3, \ldots$$

where A_r, f_r and φ_r are the amplitude, frequency and phase angle of the reference sinusoid respectively.

Then the values of the crossing time instants t_k could be calculated (in the case of enabling the sampler during ascending half-periods of the reference sine-wave) as

$$t_k = kT_r + \frac{T_r}{2\pi} \arcsin \frac{r(t_k)}{A_r}$$

where T_r is the period of the reference sine-wave function.

In this way both coordinates of every sample value of the signal could be easily reconstructed. The problem is that it takes two times more bits to describe this type of digital signals than it is in the case where signals are sampled either periodically or nonuniformly at predetermined time instants.

For this reason, this Option should be used carefully after considering also other Options.

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4.5.2 Option 2: Achieving the applicability of standard DSP algorithms

The non-uniformity of the sample value timing is a particular factor strongly affecting processing of the digitised signal because the sampling instants are signal dependent and therefore a priori they are not known. However in relatively many cases this fact could be ignored as under certain conditions it is possible to regularise the positions of the signal sample values by assuming that they are placed on a regular time grid. Is this acceptable or not depends on a number of conditions that have to be considered for taking this decision. However the point is that, in general, the analog-digital conversions based on sine-wave crossings often might be arranged so that the conditions favourable for this regularisation are realised. The reference function frequency and the enabling function definition have to be carried out purposefully for that. Usually this could be done by considering the ratio of the reasonable mean sampling rate and the upper frequency in the input signal spectrum with the specifics of the enabling function taken into account. Apparently the fact of the sampling non-uniformity could be ignored if the signal processing error due to this assumption is sufficiently small. Applicability of this regularisation approach to the sine-wave crossings based sampling and the expected performance are evaluated in [3]. Whenever this approach could be used, the obtained and recovered signal digital sample values are considered as placed on a regular time grid related to the reference function and, consequently, the traditional DSP algorithms for signal waveform reconstruction and parameter then are applicable.

Figures 4.13 and 4.14 illustrate results obtained in such a case.

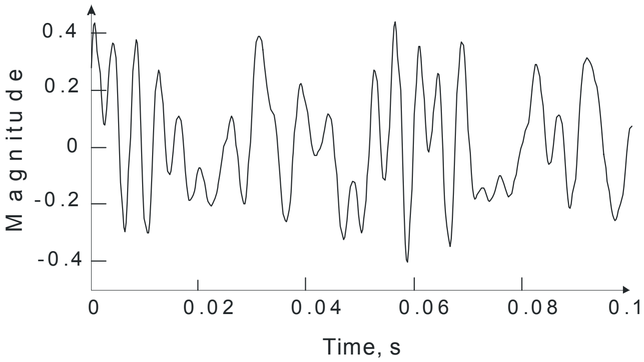

Figure 4.13 Signal recovered from sine-wave crossings by a low-pass filter in the case where the signal sample values are considered as obtained equidistantly.

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The signal displayed in Figure 4.13 illustrates the result of low-pass filtering based reconstruction of a signal in a case where application of the regularisation approach is appropriate. When this approach is used for recovery of a signal, there is noise related to this regularization procedure. The power of this noise of course depends on various parameters characterizing the sampling conditions. In this particular case, this noise is relatively weak.

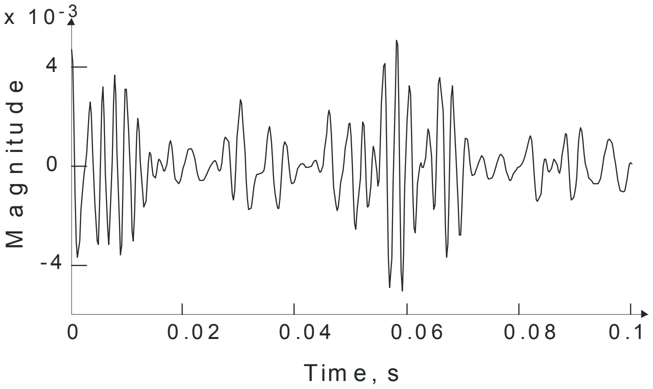

Figure 4.14 Noise related to the sampling regularization procedure present in the reconstructed signal waveform shown in Figure 4.13.

Apparently this regularization Option, whenever it is applicable, is of high application value as the wealth of the existing classical DSP algorithms and computer programs then is applicable for processing signals sampled on the basis of sine-wave crossings. Especially valuable is the possibility of using the fast algorithms. First of all, Fast Fourier Transforms.

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4.5.3 Option 3: Direct processing of digitally timed crossing event streams for reducing system complexity

Option 2 (regularisation approach to sine-wave crossings) apparently should be used whenever it is applicable. Unfortunately application of this Option 2 is limited by the bandwidth of the input signals. Only relatively low-frequency input signals could be successfully processed in this way. The sampling non-uniformity usually cannot be ignored in the cases where the upper frequencies in the spectra of the signals to be processed are relatively high. Then processing of the digitised signals has to be carried out in a way taking into account the fact that the distances between the signal sample values are not constant.

Option 3, direct processing of digitally timed crossing event streams, is preferable whenever the regularization approach is not applicable.

Then specific algorithms for signal processing have to be developed and used.

This change in the attitude to processing of nonuniformly sampled signals concerns first of all the issue of signal reconstruction. While reconstruction of periodically sampled signals usually is carried out as a low-pass filtering procedure, reconstruction of nonuniformly sampled signals typically is performed on the basis of more complex direct and inverse Fourier transforms.

Signal processing executed in accordance with more complicated algorithms often leads to increased complexity of the hardware tools used for that. In the case of signal digitising on the basis of SWC Sampling, opposite effect often could be achieved.

The point is stressed that Option 3, on one hand requires using of special algorithms but on the other hand this Option 3 often leads to the possibility of reducing the complexity of hardware tools. Specifically this is true for a wide class of signal pre-processing algorithms involving direct and inverse Fourier transforms and estimation of Fourier coefficients.

Let us demonstrate how the complexity of hardware tools used for calculations related to Discrete Fourier Transforms could be reduced on the basis of the approach referred to as Option 3.

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Estimating of the Fourier coefficients a_i, b_i usually is carried out in accordance to the following equations:

$$\hat{a}_i = \frac{2}{N}\sum_{k=0}^{N-1} x_k \cos(2\pi f_i t_k)$$

$$\hat{b}_i = \frac{2}{N}\sum_{k=0}^{N-1} x_k \sin(2\pi f_i t_k)$$

The digital sample values of the signal $\{x_k\}$ and of the sine, cosine functions typically then are given as multi-digit numbers. Therefore estimation of Fourier coefficients then requires multiplication of a large quantity of multi-digit numbers. This problem has been noticed a long time ago and various approaches to resolution of it have been suggested. The most popular of them are based on various types of rough quantising. Evidently, there is no necessity of performing multiplication operations if a single threshold level is used at quantising. However this leads to significant systematic bias errors. Although they could be taken out if randomised single threshold level quantising is used, the remaining statistical errors are still not so small.

Application of the discussed Option 3 makes it possible to develop algorithms not requiring such computationally burdensome multiplications. These further discussed algorithms are sampling-specific and could be exclusively used only for processing the results of sine-wave crossings. They are based on the relationships derived in [1] where it is shown that processing of the sample values obtained at the instants of the signal and sine-wave crossings could be reduced to the following equations:

$$\hat{a}_i = \frac{A_r}{N}\sum_{k=0}^{N-1}\left(\sin 2\pi\left(f_r - f_i\right)t_k + \sin 2\pi\left(f_r + f_i\right)t_k\right)$$

$$\hat{b}_i = \frac{A_r}{N}\sum_{k=0}^{N-1}\left(\cos 2\pi\left(f_r - f_i\right)t_k - \cos 2\pi\left(f_r + f_i\right)t_k\right)$$

where t_k are the signal and the reference sine-wave crossing instants.
Apparently no multiplications of the signal and filtering function digital values have to be carried out in this case. This represents a significant advantage.
Estimation errors characterizing various comparable complexity-reduced approaches to estimation of the Fourier coefficients are compared in Section 4.6.

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4.6 Computer simulations

To reveal the application potential and essential features of multi-channel data acquisition systems based on SWC Sampling, computer simulations of their most important application aspects have been carried out. Description of the obtained results follows.

4.6.1 Evaluation of the conditions for sampled signal regularisation

The fact that SWC Sampling is essentially a nonuniform operation considerably complicates processing of signals sampled in this way. However the conditions for signal sampling and processing often could be regularised as explained in [1, 2]. Whenever such regularisation could be carried out, it becomes possible to use standard DSP algorithms. Evidently, this is highly desirable.

A method for this has been suggested and used at computer simulations. The sampler was enabled during every *n-th* declining half-period of the reference. Then the mean sampling rate $f_s=f_r/n$ and aliasing is observed at frequencies f_x; $f_r/n\pm f_x$; $2f_r/n\pm f_x$; ...; $mf_r/n\pm f_x$; $m=0, 1, 2, ...$

According to this method:

- The aliasing suppression capabilities of SWC Sampling are used as a criterion.
- Enabling of the comparator for sampling is carried out by activating it after a variable number n of half periods of the reference function.
- Overlapping of frequencies is more pronounced and aliasing stronger when the sampling conditions are more favourable for regularization.
- This fact is exploited for evaluation of the regularization degree.

This approach is illustrated in Figure 4.15. The given curves have been obtained for varying signal frequencies for a constant sampling rate. The reference frequency, with the enabling decimation factor *n* varying, was changed to keep the ratio $f_s = f_r/n$ constant. With the reference frequency and *n* growing, the ratio of A_x/A and A_{ai}/A gradually approach the level equal to 1. It means that at this *n* value there is full-scale aliasing and therefore SWC Sampling could be considered as being virtually regular. Simulations provide the values of *n* at which SWC Sampling under differing conditions could be considered as regularized.

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Criterion for SWC Sampling regularization

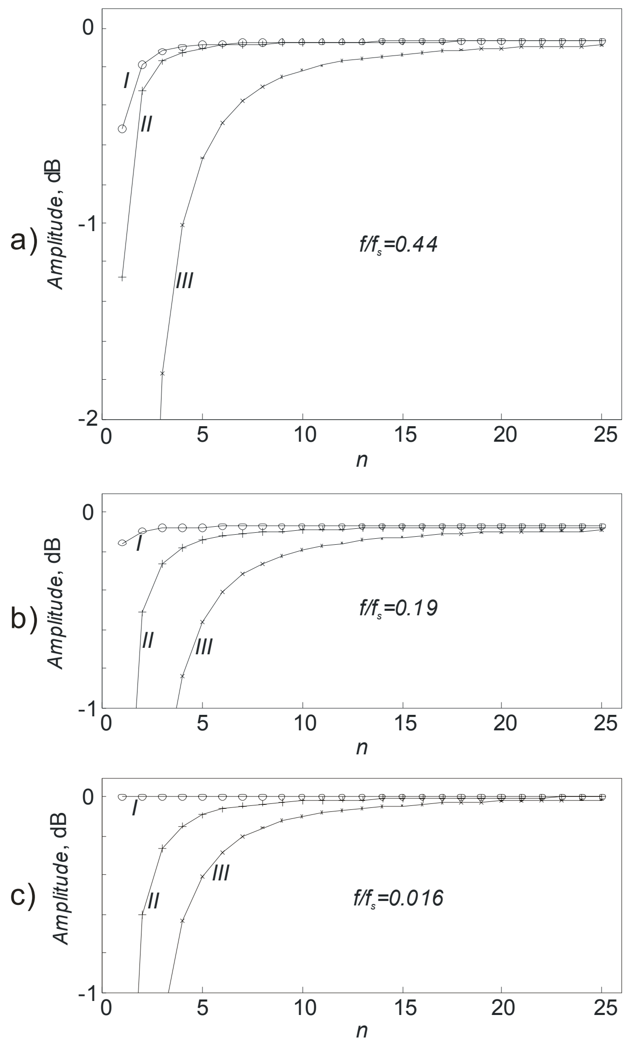

Figure 4.15 Ratio of A_x/A, A_{ai1}/A and A_{ai2}/A versus enabling decimation factor *n* estimated for the signal (1) and the first (2) and second (3) aliasing frequencies.

Regularization improves with the displayed curves merging together. As can be seen from Figure 4.15, that typically happens when enabling of the comparator for sampling is carried out after $n \geq 20$ half periods of the reference function. Consequently, SWC Sampling usually might be considered as sufficiently well regularized if there are more than 20 to 25 input channels being multiplexed by sequential enabling of the involved samplers.

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4.6.2 SNR dependence on the SWC Sampling conditions in the case of regularization

If SWC Sampling is used for digitising some signals and after that they are reconstructed assuming that regularization is acceptable, then the Signal-to-Noise ratio (SNR) of the reconstructed signals depends on the rate of the reference function period enabling as it is shown in Figure 4.16 for a few signals. Specifically, the curves 1, 2 and 3 were obtained for signals 1, 2, 3 containing components with parameters given in Table 4. 1 More information about the expected SNR is given in [3].

Table 4.1

f_i/f_s	Signal 1		Signal 2		Signal 3	
	a_i	b_i	a_i	b_i	a_i	b_i
0.037	-	-	0	0.1050	-	-
0.077	0	0.0940	0	0.0940	0.1618	0.1176
0.1	0	0.0800	0	0.0800	0.0247	0.0761
0.133	0	0.0900	0	0.0900	0	0.1000
0.163	0	0.1020	0	0.1020	0	0.1000
0.23	0.1200	0	0.1200	0	0.5000	0
0.249	-	-	0.0752	0.0547	-	-
0.267	-	-	0.0272	0.0837	-	-

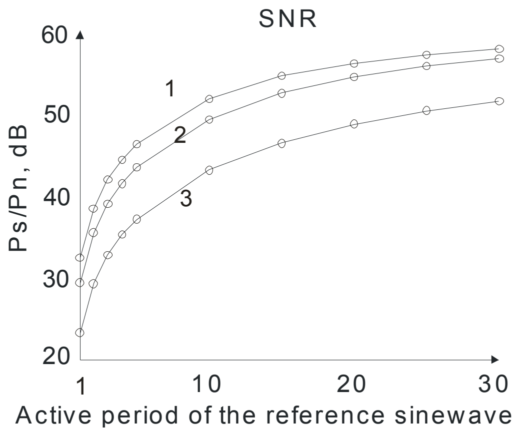

Figure 4.16 Value of SNR versus the rate of the activated reference function periods for three various signals

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4.6.3 Tests of the regularisation applicability

To test the applicability of the regularization approach, reconstructing of real-life signals, specifically, speech signals, has been performed. Particular results are displayed in Figure 4.17. The signal was sampled on the basis of SWC Sampling, the sampled signal was regularised, then its waveform was reconstructed on the basis of the direct and inverse FFT procedures. After that the digital sample values of the reconstructed waveform were compared with the record of the original signal. The results of this comparison are displayed. As can be seen, the mentioned digital sample values virtually overlap the waveform of the original signal.

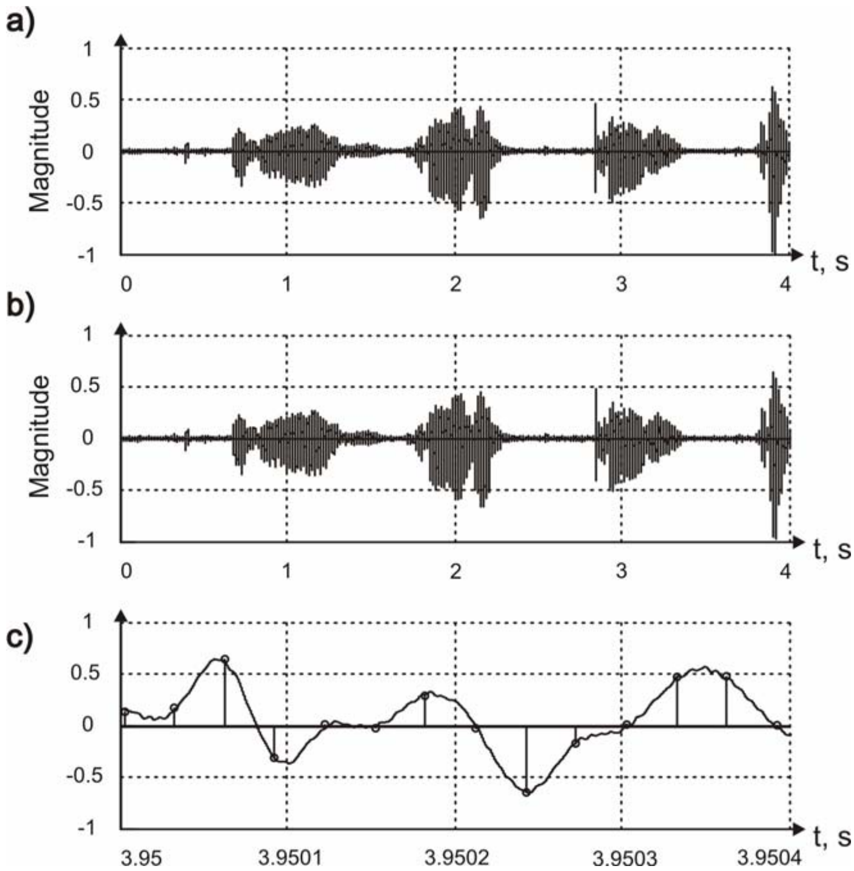

Figure 4.17 Waveform of a speech signal sampled on the basis of SWC Sampling (a) and reconstructed by using the suggested regularization procedure and after that the direct and inverse FFT (b). Zoomed up segment (c) of the signal with the indicated reconstructed sample values.

As the obtained results show, the regularization and using of standard DSP algorithms is acceptable for many practical applications if the value of n exceeds 20 to 25. For the multi-channel data acquisition, this means that the sampled signals usually could be easily regularized if the number of channels exceeds 20 to 25.

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4.6.4 Elimination of the cross-interference between signal components

The considered SWC Sampling process is essentially nonuniform. Consequently:

- The non-uniformities of SWC Sampling impact processing of the digital signals formed on the basis of such sampling.
- Specifically they lead to the cross-interference between signal components described in [1] (Chapter 18).
- This cross-interference might be observed as spurious frequency peaks due it.
- The spurious frequency peaks more or less distort signal spectrograms.
- The impact of this cross-interference on the signal spectrograms directly depends on the signal sampling conditions.
- Under certain sampling conditions providing for small sampling point irregularities, this kind of spectrum distortions might be negligible.
- In other cases special signal processing procedures for adapting the sampled signal to the sampling non-uniformities has to be carried out.
- Matrix of cross-interference coefficients has to be composed and the estimation of Fourier coefficients has to be adapted to the sampling non-uniformities as described in [1].

Figure 4.18 illustrate the impact of the cross-interference between the signal components on the given there spectrogram.

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Effect of adapting processing of digital signals, based on SWC Sampling, to the sampling non-uniformities.

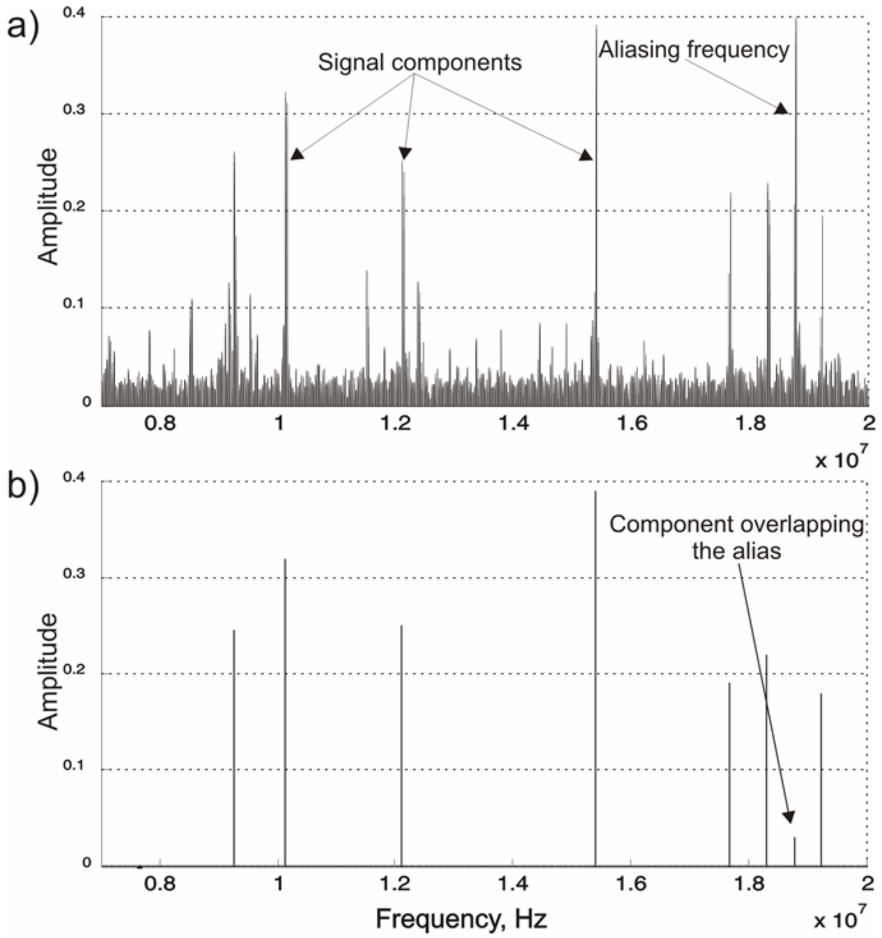

Figure 4.18 Spectrogram (a) of a digital signal corrupted by the cross-interference and the same spectrogram (b) with this cross-interference effects taken out.

Adapting processing of the considered type of digital signals to the SWC Sampling non-uniformities eliminates the negative effects due to both the cross-interference and the fuzzy aliasing.

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4.6.5 Expected performance of multiplier-less DFT

Computer simulations were carried out to evaluate the performance of the complexity-reduced algorithm for estimating signal parameters in the frequency domain not requiring execution of a large quantity of multiplication operations [1, 2]. The application error of it is compared in Figure 4.19. with errors characterizing other earlier known algorithms. The latter provide for reduction of the multiplication operation volume basically in the cases where signals are quantized roughly so that the quantized signals might assume only the values of 0 and ± 1. Typical estimation errors characterizing the efficiency of various algorithms are shown in Figure 4.19.

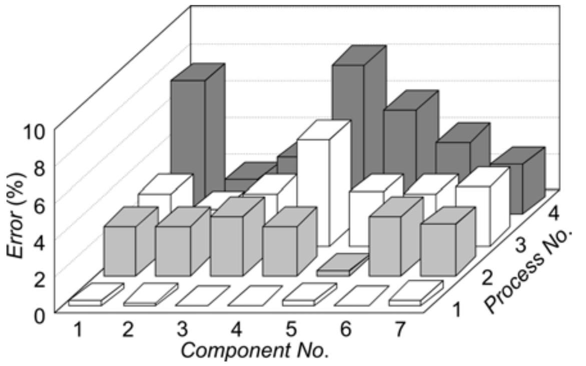

Figure 4.19 Fourier coefficient estimation errors obtained in 4 cases of various digitising and estimation techniques

A signal containing components at seven frequencies was synthesized in this case and it was digitised in four different ways. Specifically, the normalized frequencies of the signal components $f_i/f_s =$ {0.0767; 0.1058; 0.1474; 0.1965; 0.2476; 0.3476; 0.4622}, $i = 1, 2, \ldots, 7$. The used technique marked as 1 is based on the sine-wave crossings. The signal was sampled according to the additive randomised sampling scheme (techniques 2, 3 and 4) and the respective quantisation techniques used were: 2 - extremely high bit-rate quantising; 3 - two-bit pseudo-randomized quantizing; 4- two-bit randomized quantizing. Amplitudes of all signal components were equal to 0.2V.

Application of SWC Sampling leads to significantly smaller errors.

Error histograms illustrating the impact of the cross-interference.

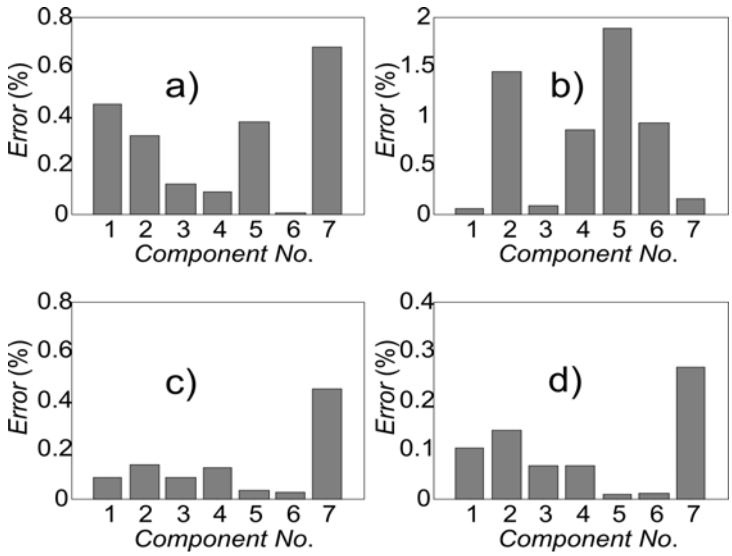

Figure 4.20 Errors in estimating Fourier coefficients of a signal sampled at the sine-wave crossing instants. (a), (b) errors in estimating respective coefficients a_i and b_i in the presence of the cross-interference. (c), (d) errors in estimating respective coefficients a_i and b_i without the cross-interference

- The given diagrams confirm that the cross-interference affect also processing of signals sampled at the instants of the sine-wave crossings.
- They also demonstrate the fact that if the cross-interference somehow is taken out, the DFT estimation errors decrease significantly.

5. Development of algorithms

To build systems for massive data acquisition according to the described in Chapter 4 concept of the distributed ADC, accepted as the basis for multi-channel data acquisition, the following central algorithms have been developed:

- Algorithm 1: Wideband data acquisition;
- Algorithm 2: Timing data acquisition;
- Algorithm 3: Massive data acquisition;
- Algorithm 4: Data pre-processing.

The multi-channel data acquisition systems considered in the next chapter represent hardware/software implementations of them. The developed method and algorithm for multiplier-less data pre-processing is considered as an invention and a patent application for it has been written and submitted to the European Patent Office.

5.1 Algorithm 1: Wideband data acquisition

In the case of traditional multi-channel data acquisition systems, the highest sampling rate of an input directly depends on the high-speed sampling capabilities of the given ADC and on the number of the multiplexed inputs. That narrows the bandwidth of the input signals within which the signals could be processed without errors due to aliasing as the sampling rate in a particular channel is *n* times lower than the sampling rate of the ADC, where *n* is the number of inputs. Thus periodic multiplexing leads to additional limitations on the input signal spectra and on the number of the inputs. Consequently, these parameters then have to be traded-off.

To avoid this limitation imposed by the number of channels on the bandwidth of the input signals and to obtain the capability of digitising signals at frequencies much higher than the limit marked by half of the sampling rate, nonuniform sampling techniques described in [1] should be and are used. As it is shown in the referenced to book, alias-free wideband signal sampling can be realised in a number of various ways, including pseudo-randomised multiplexing of the input channels. This approach, discussed more in Chapter 6, is used for development of the Algorithm 1 for pseudo-randomised multiplexing of wideband input signals.

Subroutines for adapting data reconstruction to the sampling non-uniformities, described in [1] are included.

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Algorithm 1 for Multi-channel Data Acquisition based on pseudo-randomised multiplexing of inputs

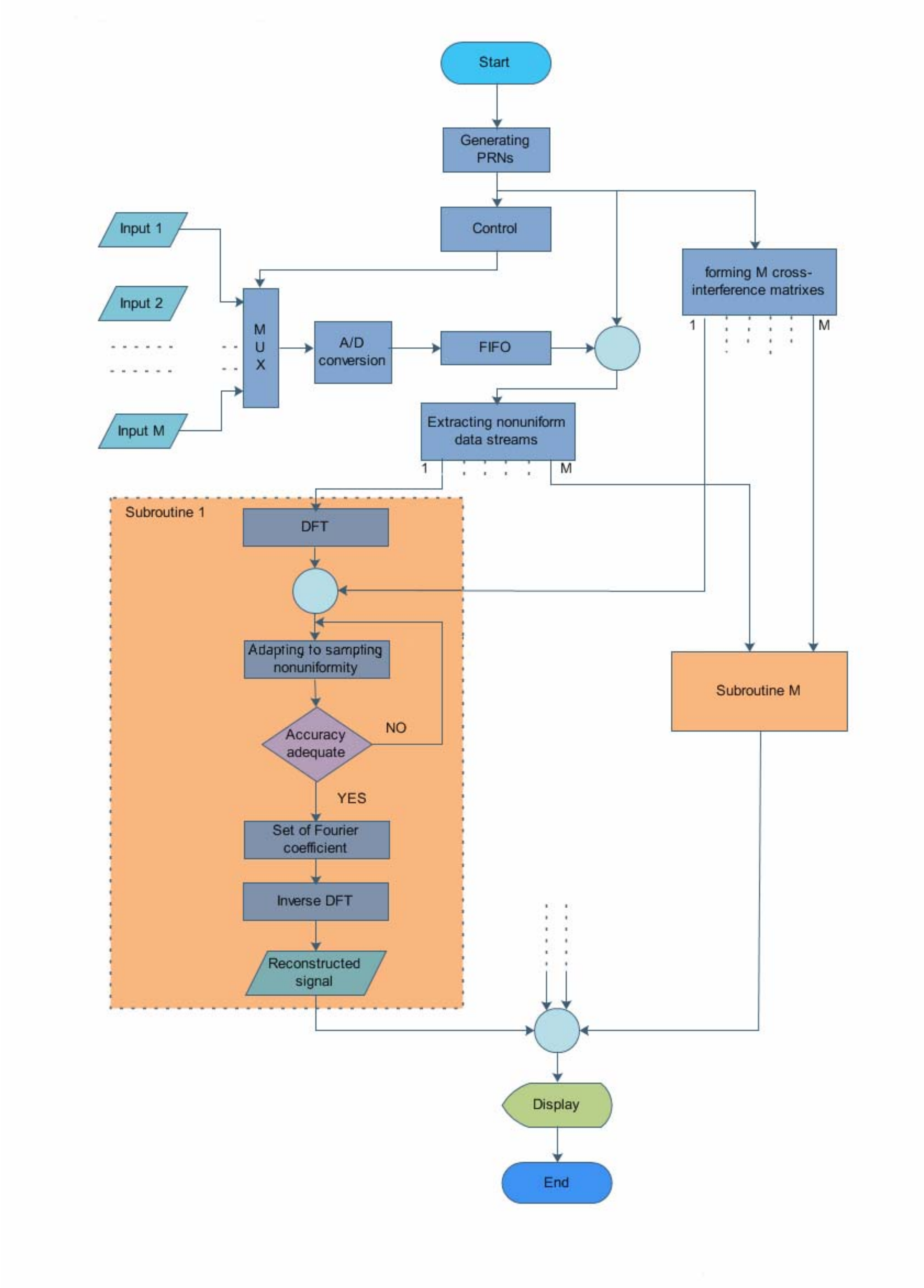

Figure 5.1 Flowchart of Algorithm 1 for wideband data acquisition.

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5.2 Algorithm 2: Timing data acquisition

It might be said that the remote SWC Samplers, plying the central role at the considered massive data acquisition, perform "signal-to-event" conversions. Indeed, the signal and reference crossing events in this case mark the time instants when sampling of the input signals take place and the sequence of these time instants carries the information from the respective input. The outputs of the remote samplers, operating on the basis of SWC Sampling, represent position-modulated uniform pulse trains. They are used as analog information (positions of the pulses) carriers for transmitting the input information from the remote SWC Samplers to the master part of the multi-channel data acquisition system. After that, digital timing of the crossing (sampling) events is performed there. Time-to digital converters are used for that, for converting the analog information carrier into the digital carrier fully representing the corresponding input signal. All these signal transforms are performed as defined by the Algorithm 2.

While this algorithm has been developed specifically for the needs of the massive data acquisition system, it covers also other essential data acquisition applications. Actually there is a wide range of applications in science and technology where data is represented by event streams. The following examples might be mentioned: Satellite Laser Ranging, LIDAR, Time-of-Flight spectrometry, secondary electron Mass-spectrometry, position sensitive detectors (delay line type), quantum cryptography research, Laser-induced fluorescence spectroscopy of biological samples, Laser-induced photo-electron spectrometry, etc. Digital timing of events has to be performed also in the cases of Voltage-to-Frequency Conversions and at applications based on Pulse Width and Pulse Position Modulations, related to the relatively new Ultra-Wide Bandwidth communication technology.

Therefore the considered Algorithm 2 covers two types of applications: (1) Time-to digital conversions needed for digital timing of SWC Sampling events and (2) event timing for acquisition of data from a wide class of sensors operating in the time-frequency domain so that the outputs of these sensors are given in the form of timed appearance of uniform events.

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Algorithm 2 for digital timing of events

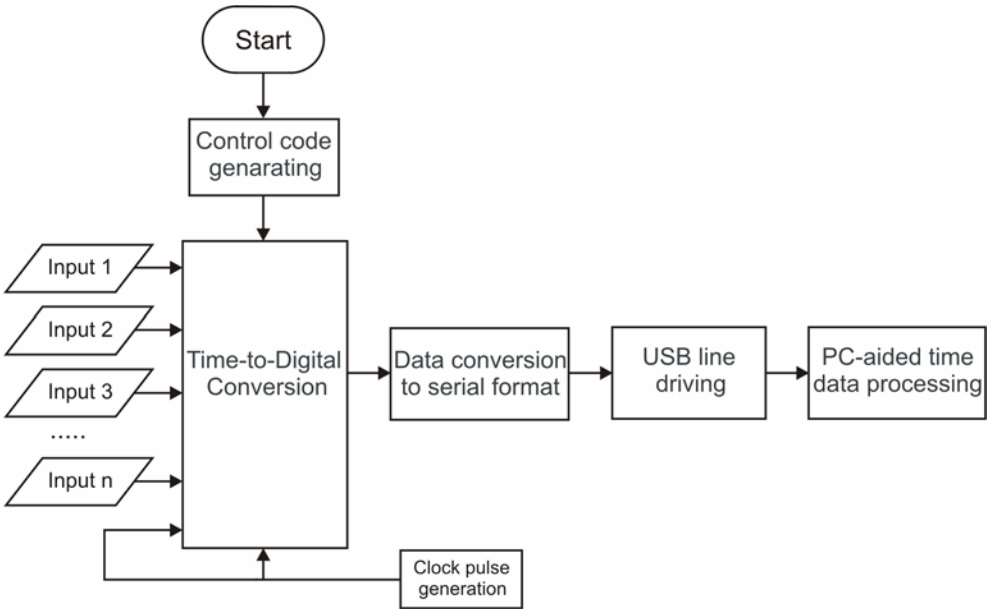

Figure 5.2 Flowchart of Algorithm 2 for digital timing of events.

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5.3 Algorithm 3: Massive data acquisition

Algorithm 3 defines data gathering from a large quantity of sensor signal sources. It includes also the subroutine for adapting data reconstruction to the channel non-uniformities. It consists of the following operations: control of the remote sampler enabling; cross-interference coefficient matrix forming for channels 1 to M; getting nonuniform M data streams; direct DFT of M data streams; adapting M spectra to the channel nonuniformities; adapted M signal reconstruction (Inverse DFT). Enabling of the samplers might be performed in a sequential, fixed, circular way or this enabling process might be pseudo-randomised.

The subroutine for adapting pre-processing of the acquired data to the data acquisition non-uniformities actually is optional. At relatively large number of the input channels (more than 25), application of the regularization approach to representation of the sampled signal values (discussed above) provides good results and then processing of the gathered data could be carried out either on the basis of standard DSP algorithms or by using the further discussed Algorithm 4 for complexity-reduced processing of digitally timed events.

Algorithm 3 for data acquisition from multiple sources of analog signals supports computer interaction with the real world objects. To provide for data gathering and smooth transfer of this information to the computers, the involved data acquisition systems have to be sufficiently flexible so that computer links to various types of signal sources might be realized. This means that they should provide for data acquisition from various types of sensors. It helps that outputs of sensors used for converting many different physical quantities into electrical signals are standardised so that a wide variety of physical processes can be observed by analysing a much-narrowed selection of such electrical signal parameters.

A sensor output class that covers a really wide field of applications is represented by voltage (current) variations in time and this Algorithm 3 is directly applicable for acquisition of data from this type of sensors. This algorithm could be easily combined with the Algorithm 2 for event timing. That is needed both for (1) converting the analog carrier, used for information transmission from the remote samplers to the master part of the system, to the digital carrier and for (2) acquiring data from sensors operating in the Time-frequency domain.

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Algorithm 3 for data acquisition from a large number of sensor signals sources

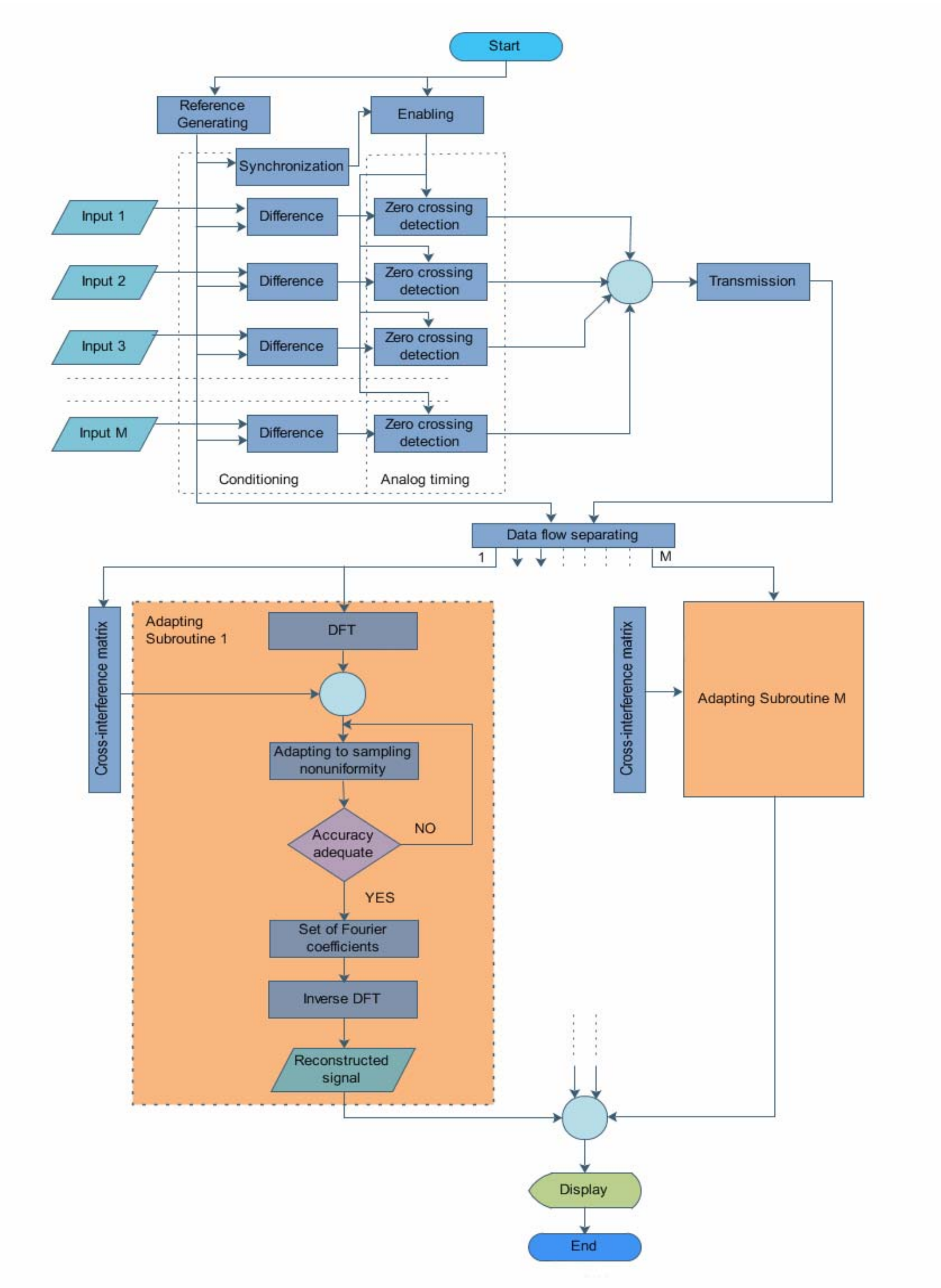

Figure 5.3 Flowchart of Algorithm 3 for massive data acquisition.

5.4. Algorithm 4: Data pre-processing

Algorithm 4 is based on an original method for complexity-reduced digital signal processing, including Discrete Fourier Transforms, filtering and estimation of various signal parameters based on such filtering. This method is considered to be an invention and patent application for it has been submitted to the European Patent Office. The developed method addresses the problem of the complexity and power consumption reduction for systems performing multi-channel data acquisition, signal digitising, digital filtering and parameter estimation functions. According to it, signal digital representation is special, adapted to the needs of the digital data representation and processing in a way making it possible to avoid multiple multiplications of multi-digit numbers usually carried out in the course of the signal filtering and their parameter estimation processes. Such massive multiplications, leading to significant consumption of computation time and energy, usually have to be performed whenever discrete sample value sequences of the signals being processed are used for their digital representation. The developed method makes it possible to avoid them. Essentially different digital representation, based on either digital or analog timing of the signal and the reference sine-wave crossing instants as defined by Algorithms 1,2 and 3, is used and that leads to significant simplification of the digital filtering process. It is performed as a much simpler operation of moving averaging of sample values of specifically defined filtering templates.

The developed method, covering also estimation of various signal parameters, can be directly used for spectrum analysis of signals based on Discrete Fourier Transforms. It can be used for calculation of the Fourier coefficients and for the estimation of various signal parameters that are defined as derivatives of the said Fourier coefficients, for instance, amplitudes, phase angles and power of the signal components at various arbitrary frequencies.

The digital signal representation appropriate for the invented multiplier-less signal filtering is well suited also for realization of remote data acquisition from a large number of signal sources.

Algorithm 4 for complexity-reduced versatile multi-channel data pre-processing

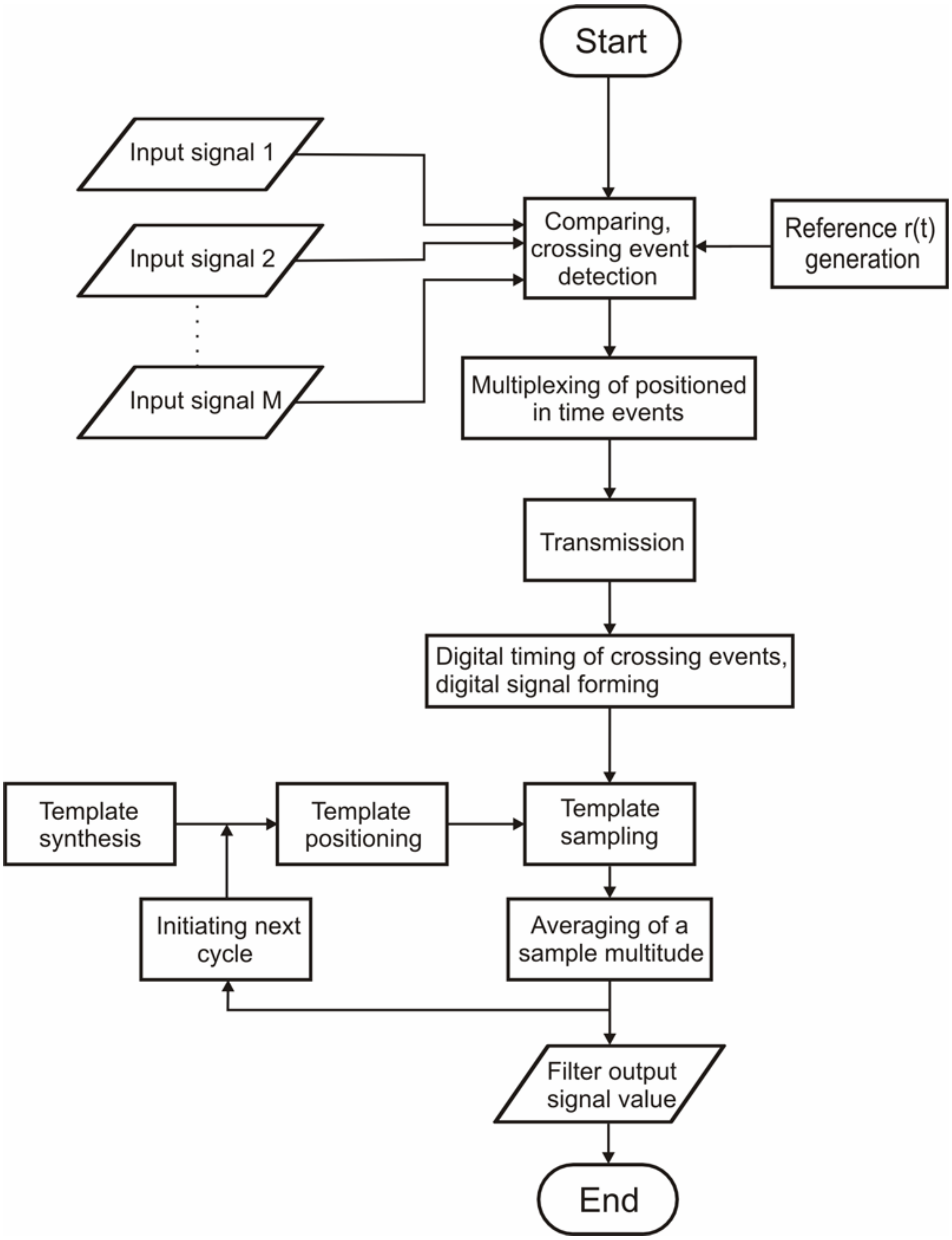

Figure 5.4 Flowchart of Algorithm 4 for complexity-reduced data pre-processing

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6. Development of data acquisition systems

To provide for data acquisition from a large quantity of various types of sensor signal sources and avoid excessive complication of the system, modular approach to the design of data acquisition systems has been chosen. Each of the module types then should represent a specific system for data acquisition and it should be possible to aggregate these modules into re-configurable modular structures meeting functional requirements of specific application cases. It has been taken into account that it is essential to digitise the input signals closely to their sources and that the design of the front-end devices should be simple enough for achieving very low power consumption. Designs of the further considered various systems for multi-channel data acquisition are based on and implement the Algorithms 1, 2, 3.

6.1 Systems for wideband data acquisition

Two types of wideband data acquisition systems specifically address the problem of suppressing aliasing in a widened frequency range not directly limited by the quantity of the input channels. They are based on and implement Algorithm 1 for Multi-channel Data Acquisition based on pseudo-randomised multiplexing of inputs. That is done in two different ways: (1) by pseudo-randomizing the order of channel multiplexing according to Algorithm 1 and (2) by pseudo-randomizing the time intervals between events of the multiplexer switching.

6.1.1 Two approaches to pseudo-randomising wideband data acquisition.

Both versions of the further briefly described wideband data acquisition systems take out the limitation imposed on the input signal bandwidth by the number of the input channels. So what is the difference between them?

- The difference is in the frequency range within which the aliasing effect is taken out.
- While aliasing still might occur at frequencies exceeding some frequency threshold, these thresholds are essentially different for the system Versions 1 and 2.
- In the case of Version 1, this threshold is equal to half of the multiplexer periodic switching rate. The upper frequency in the input signal spectra should not exceed 25 MHz (if the frequency of the multiplexer switching is 50 MHz)
- In the case of Version 2, the threshold is equal to half of the clock frequency used inside the Sampling driver for generation of pseudo-random codes. This frequency is much higher than the multiplexer switching rate. Then the input signal spectra could be within the whole 700 MHz bandwidth of analog multiplexer.

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6.1.2 First version of the system for wideband data acquisition.

Pseudo-randomization of multiplexing input signals has been suggested in [1] and a data acquisition system developed on this basis has been made, experimentally studied and described [4]. The obtained results, dicussed in [4], certainly are useful and interesting. The limitation on the input signal bandwidth imposed by the number of the input channels indeed is taken out in this way. On the other hand, aliasing then still might occur at frequencies exceeding half of the multiplexer switching rate. Figure 6.1 illustrates the structure of the considered data acquisition system. As it is operates on the basis of pseudo-randomized multiplexing, the sample value sequences, representing the analog input signals, are nonuniform as shown in Figure 6.2.

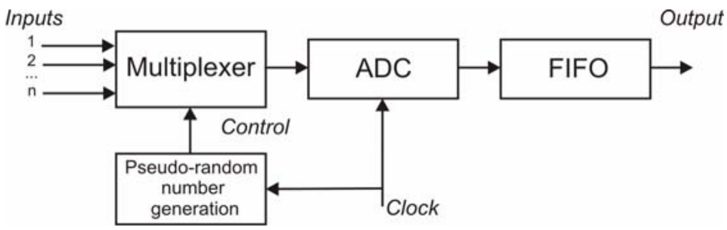

Figure 6.1 Block diagram of a data acquisition system operating on the basis of pseudo-randomized multiplexing

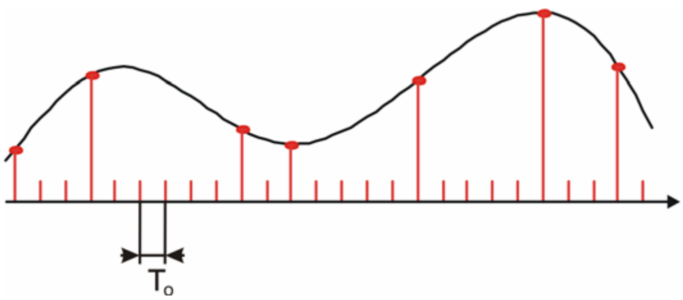

Figure 6.2 Illustration of a nonuniform sample value sequence in a particular input channel.

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Pseudo-randomisation of the multiplexer switching order indeed makes it possible to widen the frequency range within which the input signals can be observed in parallel and this is achieved by eliminating the dependence on the input channel quantity. In other words, this positive effect is obtained by eliminating the limitation due to the aliasing effect.

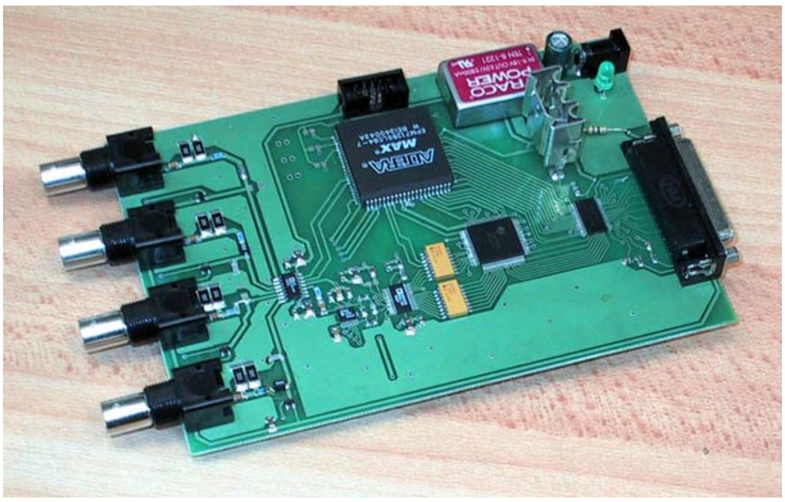

Figure 6.3 Version 1 of the developed system for wideband data acquisition.

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6.1.3 Second version of the system for wideband data acquisition

Figure 6.4 illustrates the second approach to pseudo-randomizing of multiplexing. As can be seen from the time diagram given there, switchings from a channel to channel occur, according to this approach, in a rotating order while the durations of the time intervals between the multiplexewr switchings are pseudo-random values.

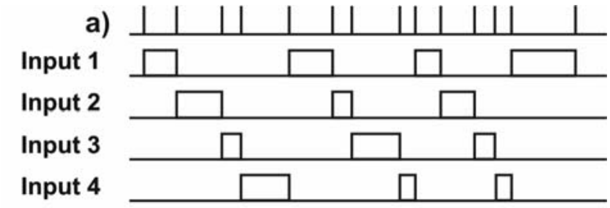

Figure 6.4. Time diagrams illustrating the suggested pseudo-randomising of multiplexing

The most responsible part in realizing this multiplexing method is forming the stream of nonuniformly distanced pulses used for switching the multiplexer. That has to be done according to the pseudo-randomised nonuniform additive sampling model. The features of such sampling point processes are well studied and described in [1]. The Sampling driver in the scheme shown in Figure 6.5 provides for generating the nonuniform sampling pulse stream with a very small jitter.

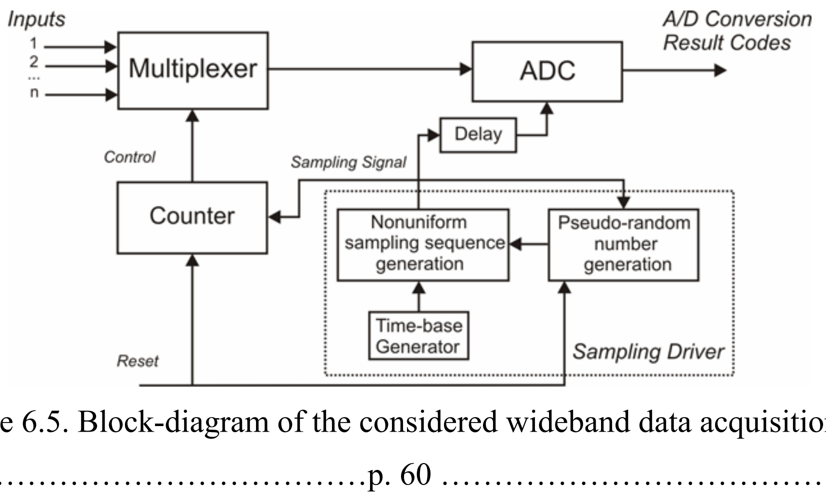

Figure 6.5. Block-diagram of the considered wideband data acquisition system

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It is emphasized that the nonuniform pulse sequences are generated very precisely at the predetermined sampling time instants. The jitter is rather small. That is crucial for alias-free digital processing of the ADC output signals. In particular, for obtaining spurious-free signal spectra in a wide frequency range not limited automatically by the number of input channels and restricted only by the frequency of the Time-base Generator and the bandwidths of the multiplexer and ADC.

Note that the very wide bandwidth of the inputs achieved by using pseudo-randomised switching of multiplexers currently could not be paralleled by SWC Sampling based data acquisition. An attempt to do that, to provide for 700 MHz bandwidth, would require using of a reference sine-wave at frequencies equal to at least 1400 MHz. Implementation of SWC Sampling under these conditions would be rather problematic.

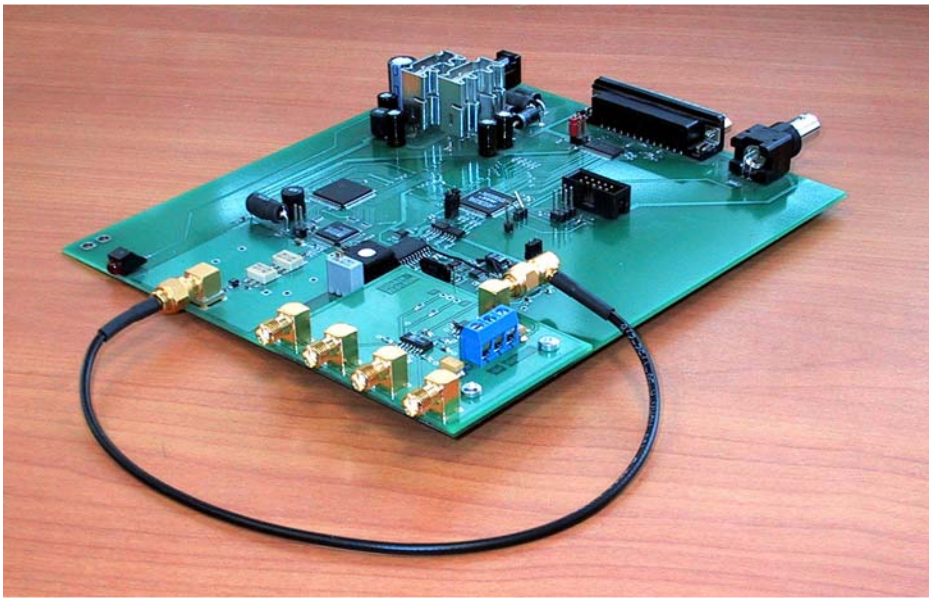

Figure 6.6 Version 2 of the developed system for wideband data acquisition.

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6.1.4. SWC Sampling based wideband data acquisition

For achieving the capability of data acquisition from real wide bandwidth signal sources, application of analog multiplexers represents a better solution. On the other hand then it is not so easy to acquire data from a large number of signal sources. However the described approach to taking out the limitation on the upper frequency of the input signals imposed by the quantity of inputs might be also used in the case where data acuisition is performed on the basis of SWC Sampling. Then the access to multiple signal sources is organised by using sequential enabling of the remote samplers. This procedure could be performed either in a periodically repeated manner or pseudo-randomly. Pseudo-random input multiplexing, performed by pseudo-randomised enabling of the remote samplers of inputs, is illustrated in Figure 6.7.

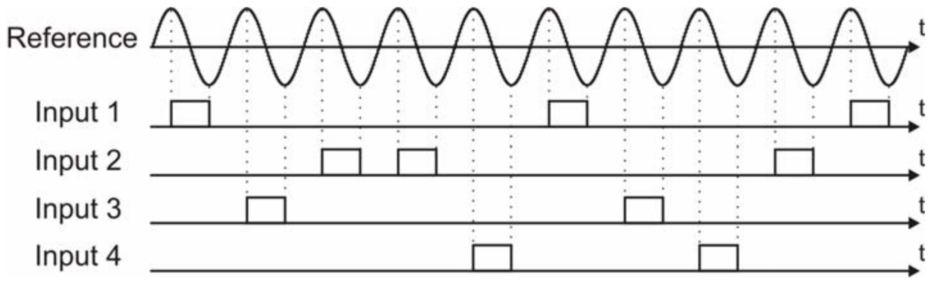

Figure 6.7 Time diagrams illustrating pseudo-randomised enabling of inputs.

According to this approach, SWC Sampling of inputs is enabled and executed in a pseudo-randomised order. Therefore the sampling operation then is performed as a distinctly nonuniform one. That leads to elimination of aliases in the frequency range up to half of the reference frequency. The highest reference frequency that might be reached clearly depends on the system design and the used microelectronic elements. For the pilot data acquisition systems made on the basis of currently available elements, this reference frequency is 20 MHz.

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6.2 System for event-timing

As the digital signal representation by sequences of timed events has been selected as the core feature of the developed systems, the event timing function plays a dual role:

(1) Performs data acquisition from sensors operating in the time-frequency domain which convert the original analog signals into uniform event streams.

(2) Converts the positioned pulse streams (serving as analog information carriers within the structure of the distributed ADC) into the digital carrier represented by the digitally timed sampling event sequences.

While SWC Sampling based data acquisition systems perform the analog-to-event conversions, the event-timing system carries out the time-to-digital conversions. Thus it is needed both for widening the application range of the further described modular multi-channel data acquisition system and for converting the output of the modules used for data gathering from a large quantity of signal sources into the special type of the digital signal which is transferred to the computer.

6.2.1 Structure

Structure of the considered four-channel system for event-timing is given in Figure 6.8. Basically it contains the following blocks:

- Time-to-digital converter (TDC);
- Microcontroller designed on the basis of an FPGA;
- USB interface FIFO chip;
- Time-base generator.

The time-to-digital converter TDC-GPX (Acam-messelectronic) is in the core of the measurements performed for executing the event-timing functions. The controller block is implemented on a Cyclone II Altera FPGA. Communicating with a computer is carried out via USB port (2.0 high-speed) by using the IC CY7C68013A from CYPRESS. The time-base is provided by a 40 MHz oscillator.

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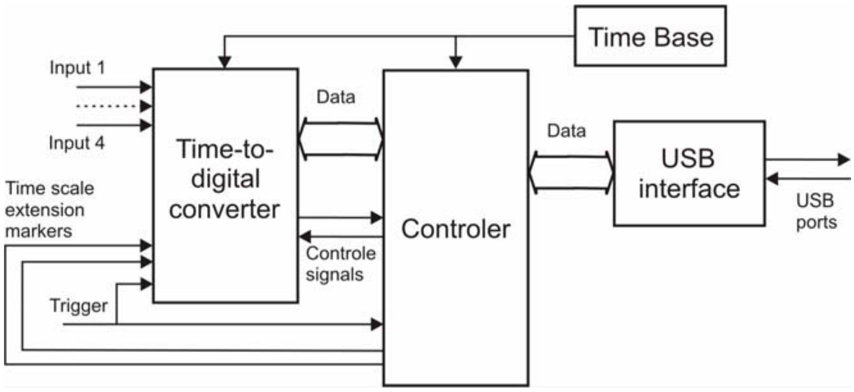

Figure 6.8 Block diagram of the system for time-to-digital conversions

The given structure of the event-timing system provides for good flexibility at supporting various operational modes, in particular, traditional time interval measurement mode. However, taking into account the dual role of it and the accepted conceptual principles, the emphasis is on the "true" timer mode - continuous event timing mode.

Pilot design version of the event-timing system is shown in Figure 6.9.

6.2.2 Performing picosecond-resolution time-to-digital conversions

In continuous event timing mode, the event-timing is either triggered internally or it is armed and then triggered externally. Once triggered, the Timer starts generating a time scale with the reference (zero) point corresponding to the trigger event. The following events appearing at the event inputs are measured. This means that the time instants of their occurance are registered with respect to the time scale and then their digital values are transmitted to a computer.

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To start operating in the continuous event-timing mode, the computer sends a packet of setting data with the mode description header to system and after that the computer receives data from it. The system transmits data as a byte stream that consists of elementary packets – 4 bytes each. Besides time stamp elementary packets there are several types of extra information packets in the data stream. The controller FPGA is responsible for configuring the TDC-GPX unit according to the received mode description and settings.

To maintain long duration massive data acquisition, the event-timing system supports unlimited measurement range (unlimited single-valued time scale). Specific events are inserted at the dedicated inputs of the TDC (time-scale extension marker events) to do that. The time-stamp data packets corresponding to these specific events are transmitted to the computer along with other data.

6.2.3 Features

The system for event-timing has high RMS resolution (<90 ps) in combination with a very high event burst rate (up to 150 MHz for each channel). The system supports timing of the average event rate up to 3 MHz, which is limited by the USB throughput and USB drivers of the development software (National Instruments LabWindows/CVI and VISA Drivers).

Thus the operating characteristics of the event-timing system are well suited for simultaneous multi-channel data acquisition. The system supports the time-to-digital conversions at the mean rate up to 3 MHz and provides the timing resolution that is sufficient for various applications.

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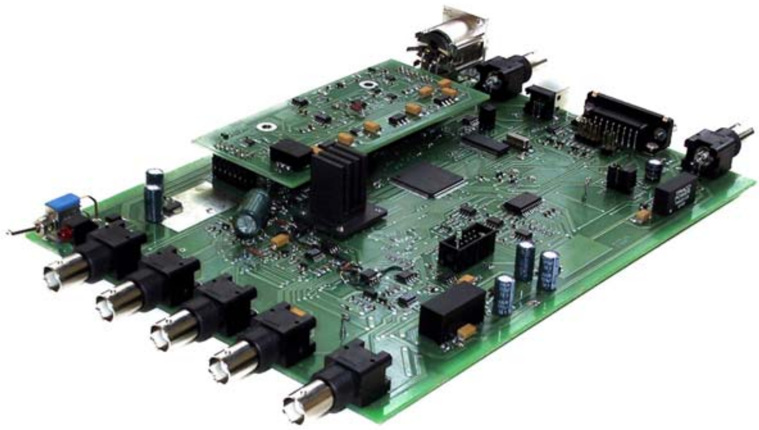

Figure 6.9 Pilot design version of the event-timing system

To achieve covering of specific applications, it is possible to extend the functions and/or to enhance some of the parameters of the pilot event-timing system shown in Figure 6.9. For applications related to low frequency inputs, event-to-digital converters providing much higher RMS resolution (<10 ps) and average event rate up to tens of KHz can be used. The system shown in Figure 6.9 is considered also as the basis for a very high-resolution event-timing. However event-to-digital converters of this kind are considerably more complicated and more expensive in comparison to the described design. On the other hand, there are applications for which achieving of a few ps timing resolution is crucial. Satellite laser ranging might be mentioned as an example of this kind. A lot of effort has been spent on development of this type of event timers for this application and the accumulated engineering experience in this field is quite useful for realization of the discussed approach to multi-channel data acquisition.

6.3 System for multi-channel data acquisition performing analog-to-event conversions

The considered system for multi-channel data acquisition from a variable number of analog sensor signal sources implements the Algorithm 3 described in Section 5.3. According to this algorithm, the system performs analog-to-event conversions for all signals at its multiple inputs. For that those signals are compared with a reference sine-wave and short pulses are formed at the time instants when crossings of the signals and the reference function occur during the time intervals when the functioning of a sampler at the front-end of the system is enabled. Special enabling function is generated and used for control of the access to the multiple signal sources in a way providing for observation of all input signals simultaneously in parallel.

6.3.1 Structure

The multi-channel data acquisition is carried out by a complex containing a number of multi-channel data acquisition modules each providing for data gathering from M sensor signal sources. For the pilot version of the system, $M=16$. It means that up to 16 signal sources might be connected to a single module and some quantity of such modules are used for connecting up the whole multitude of the signal sources to this multi-channel data acquisition system. The modular structure of the whole data acquisition complex is discussed further.

Figure 4.10 illustrates the hardware structure of a multi-channel data acquisition module. The block-diagram of it is given in Figure 4.10 (a), the enabling function for the first channel is shown in Figure 4.10 (b) and the crossings of the signal at Input 1 and the reference sine-wave are indicated in Figure 4.10 (c). The output signal, given as a sequence of pulses positioned in time by the sine-wave crossings, is shown in Figure 4.10 (d) while the point process representing the input signal obtained in result of the time-to-digital conversions, carried out outside this module, is given in Figure 4.10 (e).

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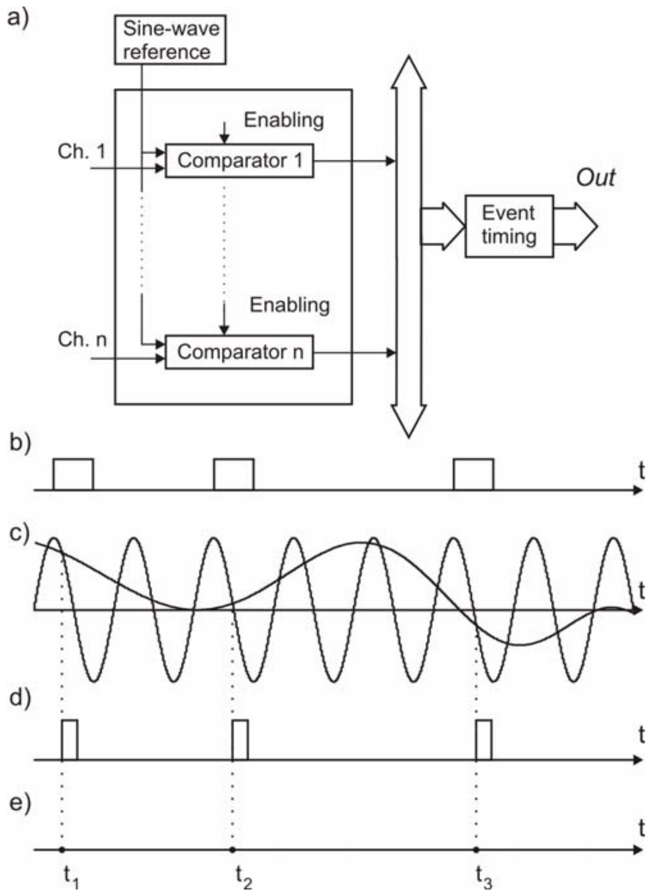

Figure 6.10. Block-diagram of the multi-channel data acquisition module (a) and time diagrams illustrating analog signal digitising performed on the basis of sine-wave crossings.

Design of this data acquisition hardware takes into account that it is essential to digitise the input signals closely to their sources and that the design of the front-end devices should be simple enough for achieving very low power consumption.

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6.3.2 Performing analog-to-event conversions

Algorithm 3 includes also the subroutine for adapting data reconstruction to the channel non-uniformities. A hardware controller embodied in a FPGA is used for appropriate control of the input sampler enabling according to this algorithm. Enabling of the samplers might be performed in a sequential, fixed circular way or this enabling process might be pseudo-randomised. Event streams at the outputs of the samplers carry the information obtained from the respective inputs.

In the cases where data might be acquired within the whole bandwidth of the input signals, the samplers are enabled in a fixed periodically repeating manner. Then sampling of the inputs occurs in a way which is close to the periodic sampling mode. Under this condition the regularization procedure mentioned before could be applied and then there is no aliasing.

In the cases where the signal bandwidths are wider and special alias-free nonuniform sampling techniques have to be used, adapting of signal reconstruction to the sampling non-uniformities has to be performed in accordance with Algorithm 3. Software tools are used for this. Cross-interference coefficient matrix is formed for the channels 1 to M. Then direct DFT of M data streams are performed and M spectra are adapted to the channel nonuniformities. Inverse DFT are carried out to finish adapted M signal reconstruction.

Figure 6.11 Pilot design version of the 16-channel data acquisition system module performing analog-to-event conversions

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6.3.3 Features

While the pilot version of this module hardware with 16 inputs represents a multi-layer printed circuit board shown in Figure 6.11, actually it should be implemented as an ASIC with more inputs and a number of such ASICs, scattered over some area, should be used for data acquisition.

The capabilities of the modular architecture for massive data acquisition are discussed in Chapter 8. They evidently depend on the qualities and performance of all involved systems considered as modules for aggregating application-oriented complexes for massive data acquisition.

In the context of this modular data acquisition structure, the most important parameter characterising the developed system for multi-channel data acquisition is the maximal number of input channels achievable at the discussed approach to massive data acquisition. Experimental testing confirm the calculation results showing that it is feasible to build a system with the number of input channels up to 1000.

The experimental test results characterise the performance of the pilot system assembled from separate microelectronic elements with a part of the system put inside a FPGA Cyclone II. Some of the basic characteristics, like the mean sampling rate, have to be traded-off with other equally important parameters like power consumption of the front-end devices. Therefore the considered data acquisition systems might be designed for relatively high frequency applications with relatively high power consumption and for low frequency applications with very low power consumption. If the power consumption of the front-end devices is not considered to be of primary importance then the frequency of the reference sinusoid might be at relatively high frequencies. Experiments made at reference frequency 30 MHz confirm this. Discussion of the results obtained at the experimental tests follows.

With further technological improvements made, it should be possible to enlarge the number of the input channels and to improve other system parameters. Apparently there is significant potential for the data acquisition system enhancement related to the implementation of the developed system hardware designs as ASICs or systems-on-chip. This technological development should be the follow-up to this project.

Capabilities of the further discussed modular system for simultaneous massive data acquisition obviously depend to a large extent on the microelectronic embodiment of the input signal analog-to-event conversions carried out on the basis of SWC. Therefore comparison of the signal and the sine-wave reference function is the operation most responsible for achieving sufficiently high performance level. The achievable signal digitising parameters, such as input bandwidth and the mean sampling rate, obviously are directly related to the technology to be used at development of such chips.

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7. Experimental testing

To evaluate the developed data acquisition systems, experimental studies of them have been carried out. They were basically focused on testing how close the experimentally obtained results are to the theoretically expected. Most of the efforts were spent on studies of issues related to SWC Sampling. Some of the obtained essential results are discussed in this Chapter 7.

7.1 Basic experimental set-up

A special experimental data acquisition system, performing the analog-to-event conversions, was developed and made for carrying out the planned experiments. It is shown in Figure 7.1. Experimental testing of it was performed. For that the output of it was connected to the described above event-timing system transferring the acquired data to a PC. Generators for supplying the test signal and the reference sine-wave function were included in this experimental set-up.

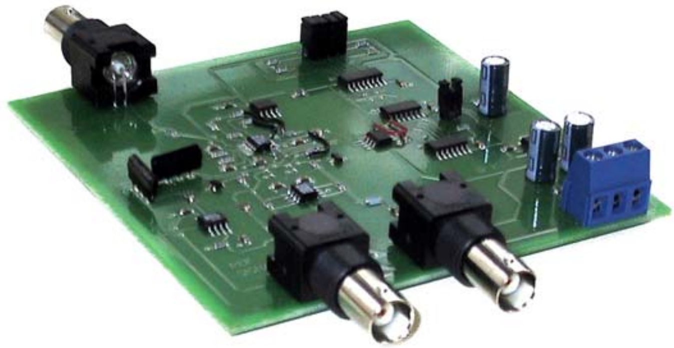

Figure 7.1 Experimental data acquisition system performing the analog-to-event conversions.

7.2 Approaches to evaluation of the systems performance

To evaluate the performance of the developed data acquisition systems, test signals with given parameters were generated, sampled and encoded in the described way. The obtained digital signal (digitally timed crossing instant sequences) were transferred to PC, reconstructed there and compared with the theoretically expected ones. Then the difference between them were analysed and the sources of the errors identified. Evidently, the reference functions used both at SWC Sampling and reconstruction of the signals should be identical with the data transmission delays taken into account.

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Impact of differences in various reference function parameters at SWC Sampling and reconstruction of the signals were evaluated. A typical spectrogram of the test signal being analysed is shown in Figure 7.2.

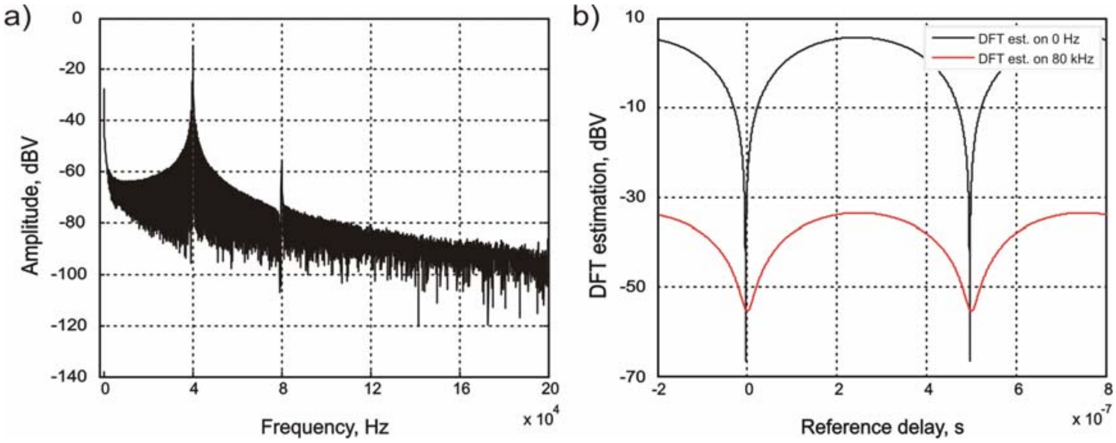

Figure 7.2 Spectra of a test signal reflecting: (a) not sufficiently well compensated delays; (b) differences in reference phase shifts at signal sampling and reconstruction.

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Experimental studies of various SWC Sampling aspects were carried out. As the performance of the considered data acquisition systems directly depend on the precision achieved at SWC Sampling, a lot of attention has been put on testing this stage of the data acquisition process. In particular, sampling errors were estimated in dependence of the reference function frequency. Typical results, obtained in the case where the reference frequency is equal to 30 MHz, are given in Figure 7.4.

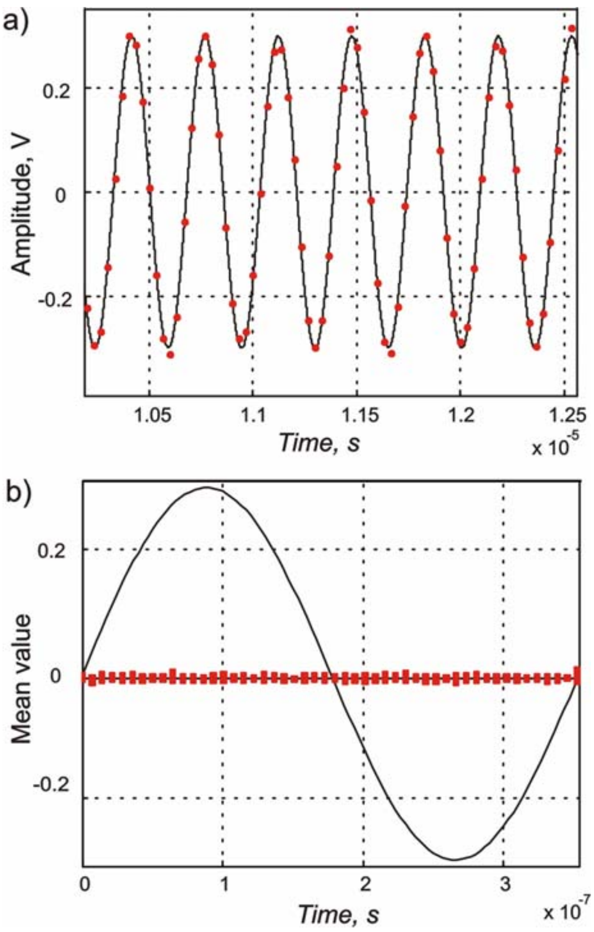

Figure 7.4 Illustration of experimental results, obtained in the case of SWC based sampling of a sinusoidal signal at the frequency 2.8 MHz while the frequency of the reference function is 30 MHz. (a) reconstructed signal sample values; (b) error distribution within a period of the input signal.

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Experimental tests of the wideband data acquisition systems (Module-1.1 and Module-3), based on pseudo-randomised input multiplexing, were carried out.

An example of a wideband signal waveform reconstruction is given in Figure 7.5. While the multiplexing rate was 9.8 MHz, the signal bandwidth was much wider. A sample value sequence of a particular signal is given in Figure 7.5 (a). As can be seen, the true waveform of the signal is actually non-detectable. However, after processing these samples according to the approach described in [2], this waveform is reconstructed in small detail. Figure 7.5 (b) illustrates this. This experiment confirms that pseudo-randomisation of multiplexer switching indeed helps in enlarging the number of the input signals that can be observed in parallel by eliminating the limitation imposed by the aliasing effect.

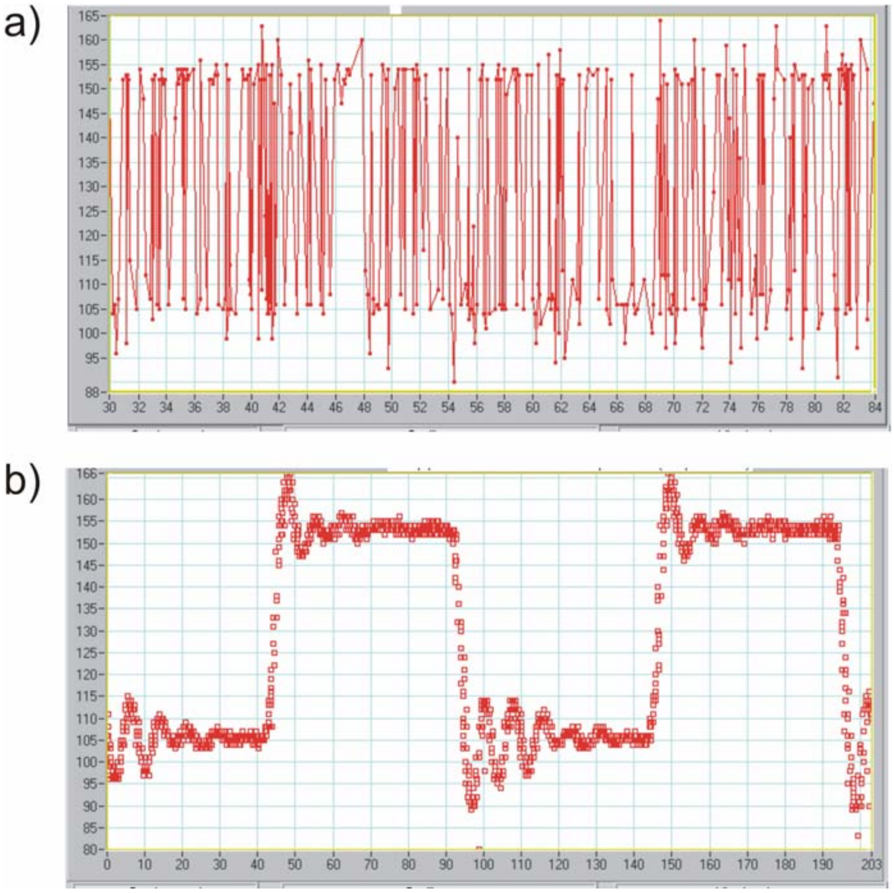

Figure 7.5 Sample value sequence of a particular signal (a) and the reconstructed signal waveform (b).

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8 Technology for complexity-reduced data representation and processing

Original technology for digital representation and processing of acquired multi-channel data has been developed in the framework of this project. Application for a European patent for it has been submitted [12].

It is based on a method providing for complexity-reduced digital processing of acquired data, including filtering, Discrete Fourier Transforms and estimation of various signal parameters based on such filtering. According to this method:

- Signal digital representation is special, adapted to the needs of the invented digital filtering in a way making it possible to avoid multiple multiplications of multi-digit numbers usually carried out in the course of the signal filtering and their parameter estimation processes.
- Such massive multiplications, leading to significant consumption of computation time and energy, usually have to be performed whenever discrete sample

value sequences of the signals being processed are used for their digital representation.

- The invented method makes it possible to avoid them.
- Essentially different digital representation, based on either digital or analog timing of the signal and the reference sine-wave crossing instants is used and that leads to significant simplification of the digital filtering process.
- It is performed as a much simpler operation of moving averaging of sample values of specifically defined filtering template.
- The digital signal representation appropriate for the invented multiplier-less signal filtering is well suited also for realization of remote data acquisition from a large number of signal sources.
- The invented method provides also for on-line filtering, in particular, for continuous estimation of Fourier coefficients.
- In that case the filtering could be performed by digitising the sample values of the filtering template rather than the signal and the reference sine-wave crossing instants.

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This multi-channel data acquisition and processing technology has significant application potential as it makes it possible to reduce the complexity of the hardware and software tools for massive data acquisition significantly. Figure 8.1 illustrates the involved basic method for data digital representation and filtering.

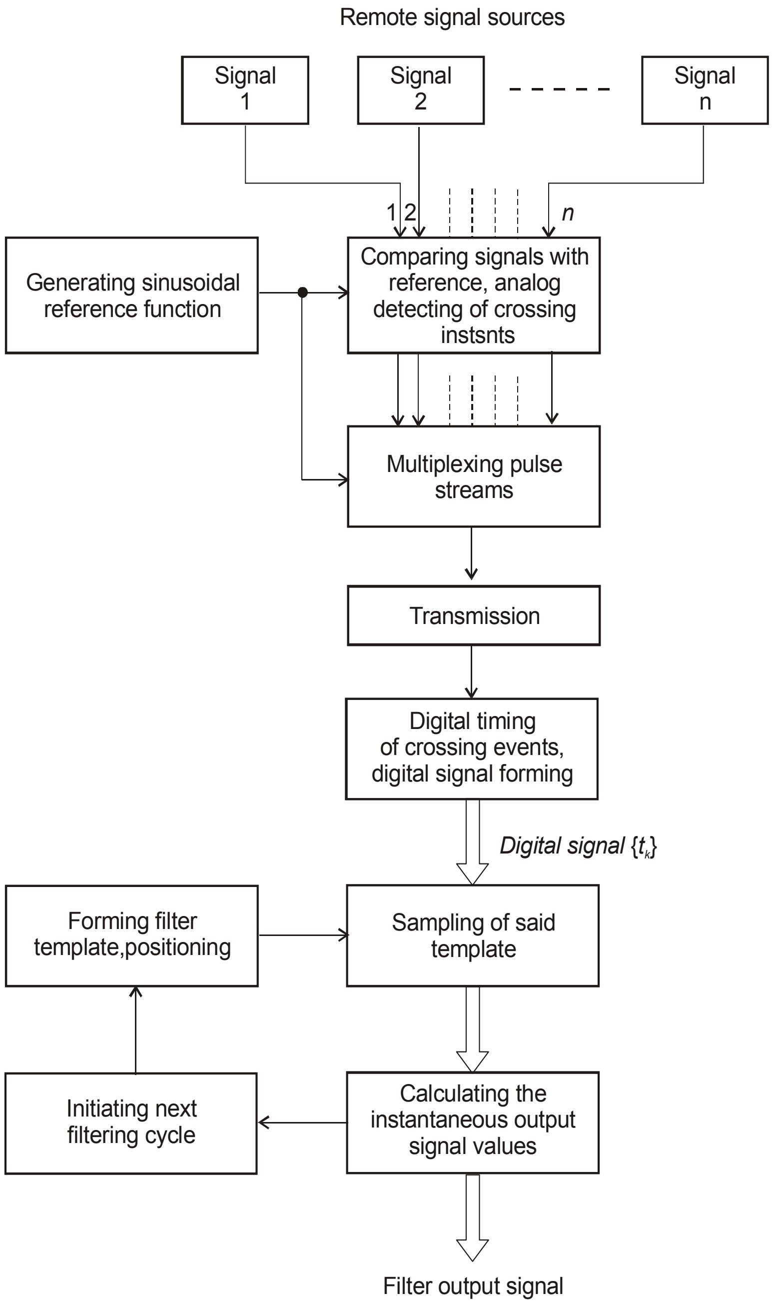

Figure 8.1 Complexity-reduced filtering of the specific digital signals formed in the process of multi-channel data acquisition.

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9. Modular structure for massive data acquisition

The basic function and also the specific of the developed systems for multi-channel data acquisition systems, in general, is simultaneous data acquisition from a large quantity of inputs. To provide for flexible adapting of it to varying application conditions, it should be possible to use all of the developed systems together. Such a complex of them has a modular architecture shown in Figure 9.1. Let us comment on the module types and on the range within which the basic parameters of them might vary at modular massive data acquisition.

9.1 Modular architecture for simultaneous multi-channel data acquisition

A multi-channel versatile data acquisition system performing analog-to-event conversions is considered and called as Module-1. This Module-1 represents a data acquisition system that might be used as an autonomous system or as a part of a more complex system for massive data acquisition. It has a number of analog signal inputs and the output signal of it is a sequence of events indicated by precisely positioned in time short pulses. While the number of these pulses indicates the number of the input, the position of a particular pulse on the time axis carries the information of the instantaneous sample value of the respective input signal. This carrier of information is passed to an input of a Module-2, where digital event timing conversions are performed and in result of them a digital signal (sequence of digital crossing instants) is formed. From the output of Module-2 this digital signal is transferred to a computer.

Actually there are two versions of Module-1: Module-1.0 and Module-1.1. Both of them are based on one and the same hardware and both fulfil the same multi-channel data acquisition function. The difference is in two different approaches to enabling of the samplers. While enabling of the samplers in Module-1.0 is carried out sequentially in a fixed order, enabling of the samplers in Module-1.1 is performed in a pseudo-randomised order as it is described in Chapter 6. Consequently Module-1.1 might be used for data acquisition from relatively wideband signal sources. While the hardware design of both modules 1.0 and 1.1 is identical, the software supporting the enabling Controller, implemented by an FPGA, differs. Apparently the digital representation of the input data is different as well. In the case of Module-1.1, the digital signal sample values, obtained at reconstruction of the acquired data, are typically nonuniform and special DASP algorithms have to be used for processing them.

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Figure 9.1 illustrates the Modular architecture for simultaneous multi-channel data acquisition. A number of the two versions of the Module- are connected to a part of the Module-2 inputs while their other inputs are directly connected to the sources of signals which carry information represented by timed event sequences [8]. There is also Module-3 for data acquisition from sources operating in the Radio Frequency (RF) range. The systems for wideband data acquisition, based on pseudo-randomised multiplexing of the inputs and discussed in Section 6.1, are considered in within this structure as Modules-3. Their outputs are directly connected to the USB interface and the computer.

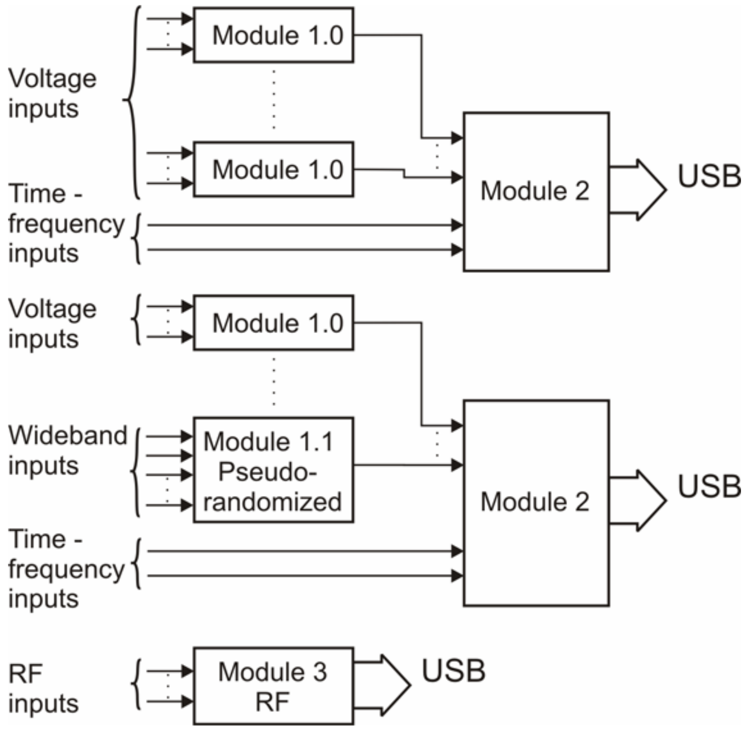

Figure 9.1. Modular architecture for simultaneous multi-channel data acquisition.

9.2 Application potential

The range of potential applications for this massive data acquisition complex depends on the achievable parametric and functional characteristics. The most important parameter of interest probably is the maximal number of input channels achievable at the discussed approach to massive data acquisition. Calculations and experimental tests show that it is feasible to build a system with the number of inputs up to 1000. With further technological improvements made, it should be possible even to enlarge this figure.

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If the power consumption of the front-end devices is not considered to be of primary importance, reference frequency 30 MHz might be used. Under this condition and if the number of input channels is up to 1000, the mean sampling rate of inputs then evidently is equal to 30 KHz and the upper frequency of the inputs signals is limited at 15 KHz (Module-1.0). If data has to be acquired also from wideband signal sources, some Modules-1.1 have to be used and then the upper frequencies of the input signals would be limited at 15 MHz. However processing of the data gathered from the outputs of these modules then will be more complicated.

Thus Module-1 can support data acquisition at frequencies considerably higher than 3 MHz data transmission rate currently supported by Module-2. As the USB throughput and USB drivers of the development software impose the limitation on the data transmission rate, other interface options might be also used if needed. On the other hand, the number of inputs needed for many applications does not exceed 250.

The data transmission rate of 3 MHz for this quantity of inputs would support the mean sampling rate 12 KHz. As a part of the inputs might be connected to Module-1.1 for the alias-free digital representation of data then such a data acquisition system is applicable for many biomedical and industrial applications.

In the cases where massive data acquisition has to be performed under conditions requiring very low power consumption at remote sampling, the reference frequency usually has to be decreased. If the required input signal bandwidth is given as well, then there might be some limitations imposed on the achievable number of inputs. Using of Module-1.1 then might be considered for reducing these limitations, as the input signal bandwidth then does not directly depend on the numbers of the inputs.

Apparently, to achieve significantly higher performance level, the considered multi-channel data acquisition approach has to be implemented on the basis of special ASICs.

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10. Conclusions

The project exploits the emerging Digital Alias-free Signal Processing technology as the platform for development of the targeted data acquisition methods, algorithms and systems for data acquisition from a large quantity of sensor signal sources.

In particular, the project is based on the DASP concept of the Distributed Remote Sampling Analog-to-Digital conversions and the algorithms for alias-free processing of nonuniform data streams. This approach makes it possible to obtain a number of benefits. Specifically, the developed multi-channel data acquisition systems are flexible and applicable for simultaneous data acquisition from many signal sources under conditions when the upper frequencies of the input signal spectra not necessarily depend on the quantity of the input channels, the input signals are sampled directly at their sources by front-end devices which are much simpler than the traditionally used ADCs. Consequently, the power consumption of these front-end devices might be significantly lower.

On the other hand, DASP technology is specific. Therefore special hardware and software tools have been developed and they could be used to implement the chosen approach. Streams of digitally timed sampling events are used as digital signals representing the acquired data. Massive data acquisition is based on pseudo-randomised time-sharing, analog-to-event and time-to-digital conversions.

Systems have been designed, made and tested specifically for data acquisition from wideband, event timing and large distributed clusters of signal sources. Modular system design is used for achieving versatility in customizing the data acquisition systems.

Original fast multiplier-less algorithms for pre-processing of data, represented by the streams of digitally timed sampling events, have been developed. They are applicable for digital signal pre-processing related to signal reconstruction, spectrum analysis, filtering, parameter estimation and demodulation.

The application range for the developed multi-channel data acquisition systems is wide. Basically they could be used for obtaining data from various biomedical and industrial objects and transferring them to computers.

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