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Data Acquisition from Real World Objects based on Nonuniform Signal Sampling and Processing

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The paper summarizes obtained research and practical results reported in the author's doctoral Thesis "Data Acquisition from Real World Objects". Work is focused on methods and algorithms for data acquisition from real world objects, and it is based on the theory of non-traditional Digital Signal Processing, including non-uniform sampling and pseudo-randomized quantizing. That leads to obtaining data simultaneously from an increased number of data sources, to widening the frequency range and to significant complexity reductions. In particular, research has been done in the directions of: (1) asymmetric data compression/reconstruction; (2) data acquisition from sources in the GHz frequency range; (3) rational acquisition of impedance data; (4) data acquisition based on signal and sine wave reference function crossings. Microelectronic implementation of the research results is considered. In particular, FPGA implementation of a Fast-DFT processor has been developed and tested.

Keywords. Data acquisition, Nonuniform signal sampling, Impedance measurements, DASP

1. Introduction

Motivation for research in the area of data acquisition for computer systems

Massive data acquisition from real life objects and supplying computers with this information in an effective way evidently is vital to realizing full potential of computer system applications in many areas. Various types of sensors are used for obtaining information from objects of natural or technical origin. Most of them convert the primary information into continuous-time or analog signals. Data acquisition (DAQ) from only this type of sensors is considered. As the acquired data are to be given as discrete quantities, the analog sensor signals have to be digitized to obtain their representations in the digital domain. Usually the classic Digital Signal Processing (DSP) technology is used for this.

This approach, based on the assumption that digitizing is based on the classical sampling and quantizing concepts, leads to significant narrowing of the digital domain, to using of more expensive analog signal processing methods and technical means for performing signal processing in the high and ultra high frequency range. It means that this approach negatively impact data acquisition technologies and limits their application range. The research described in this summary is motivated by the importance of achieving progress in the direction of computer system applications in the wide area of Information Technologies related to computer system interaction with real world biological and industrial objects.

Research goal and the basic problems that have to be resolved

Research activities of work target reaching the goal of discovering innovative methods for massive data acquisition from real life objects and effective supplying computers with this information. That evidently is vital for realizing full potential of computer system applications in many areas. The research is focused on resolution of the following basic problems that have been indicated at the beginning as the most essential:

- Too high DAQ system complexity;
- Relatively small quantity of sensors that typically can be connected to inputs of a single DAQ system;
- Limited number of channels for simultaneous data acquisition in parallel;
- Power consumption of DAQ systems, often limiting the duration of their autonomous performance time.

Basic tasks

To achieve progress in these directions, the following tasks are addressed:

1. Research focused on development of innovative methods and algorithms for complexity-reduced DAQ paying attention, in particular, on the following:
    - Combining data acquisition with signal specific digital pre-processing;
    - Reduction of power consumption;
    - Increasing the number of sensors that can be connected to a single DAQ system at least up to 100;
    - Compression of acquired data.
2. Development of algorithms and computer programs for sensor data transfer to computers.
3. Experimental investigations of the developed problem solutions, mostly by computer simulations in MATLAB environment.
4. Description of the developed DAQ structures in VHDL.

Approach to resolution of the considered problems

To reach the goal a flexible approach to complexity-reduced multi-channel data acquisition from a large quantity of sensors has been used. 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 digitizing techniques, including pseudo-randomized 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. 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.

2. DAQ Systems for Information Gathering and Supplying to Computers

Relatively many various DAQ systems are currently produced and offered by many companies. On the other hand most of them actually are operating on the basis of a few basic DAQ principles that can be considered as classic as they have remained unchanged for a long time. Therefore performance of the existing DAQ systems mainly depends on the currently achieved perfection of the involved microelectronic elements.

Analysis of the currently used data acquisition system specifics reveals some essential facts and leads to the following generalized conclusions:

1. DAQ systems mostly belong to one of two basic types of DAQ systems: (1) multi-channel systems for DAQ based on multiplexing inputs to a central ADC connected via interface to a computer; (2) multi-channel systems for so-called simultaneous DAQ using a separate ADC in every input channel.
2. DAQ functions usually are considered as input signal conditioning, analog-to-digital conversions and transmitting the digital signals, obtained from all inputs, to the computer. Processing of data more often than not is the responsibility of the host computer.
3. The essential function of signal digitizing usually is performed on the basis of the classical concepts of uniform periodic sampling (Shannon sampling theorem) and fixed-threshold quantizing.
4. The sampling rate of for DAQ systems based on multiplexing inputs to a central ADC directly depends on the clock frequency at which the multiplexer is switched and the rate of sampling signals in each input channel is inversely proportional to the number of inputs.
5. Consequently the quantity of DAQ system inputs usually is restricted to relatively small numbers, usually up to 16. More complicated systems con-

tain a hierarchy of multiplexers. In this case there might be more inputs however this type of DAQ systems then can be used only for data acquisition from low-frequency sources.

6. Using a separate ADC in every input channel allows avoiding these restrictions. Therefore this type of DAQ systems can be used for obtaining data from high frequency signal sources. The factors limiting their applications are relatively high complexity (ADC in every channel) and multiplexing of ADC output signals needed for transmitting them sequentially to the host computer. The quality of developed and produced DAQ systems improve all the time, however this progress is mostly based on the achievements in the area of semiconductor device development and production technologies.

The last conclusion is true for most of the currently produced DAQ equipment. On the other hand, there have been various R&D efforts addressing the problem of data acquisition under conditions more demanding than usual.

In particular, the applicability of NU sampling and low-bit rate quantizing has been investigated over a long period of time and this approach is directly related to DAQ functions [15, 16, 33, 34]. These methods and algorithms are applicable for achieving a number of essential advantages, such as performing of DAQ in a wide frequency range, elimination the dependency of the sampling rate in a channel on the quantity of channels in a system, data compression/reconstruction, reducing the complexity of DAQ systems and others. That leads to the following conclusions:

1. The currently used classical theory covering signal digitizing (sampling and quantizing) and digital representation of analog signals is not exclusive. These signal conversion operations can be performed in various ways, based on the signal digitizing theory developed in a few past decades, including theory of randomized signal processing, NU sampling and quantizing and DASP.
2. To achieve progress in the area of DAQ, efforts have to be put in this work basically in the directions of developing methods and algorithms for application-specific DAQ.
3. Selecting and using the most effective type of digital representation of analog input signals is really important for that.
4. Data pre-processing in many cases should be included in the list of functions to be fulfilled by a DAQ system as algorithms for parallel processing of raw digital signals can be developed and used at that stage for effective data representation and compression.

Reduction of power consumption, simplification of system hardware, simplification of algorithms for acquired data pre-processing and enlargement of system channel quantity, accurate signal acquisition are still very desirable for DAQ systems.

3. Data Acquisition Based on NU Sampling: Achievable Advantages and Involved Problems

Periodic sampling is currently the most used and widespread sampling method. However there are other methods for sampling, specifically, NU sampling. The achievable advantages and application potential of this approach are based on exploitation of the capabilities offered by the NU sampling procedure carried out in the process of the sensor signal digitizing [15, 16, 23, 33, 34, 38].

Whenever continuous time signals are digitized and then processed on the basis of DSP, the sampling rate f_s of the used periodic sampling limits the bandwidth of the original analog signals. Then the restrictions, defined by the Sampling theorem, have to be satisfied to avoid the uncertainty due to the fact that all frequencies belonging to the sequence: f_o; $f_s \pm f_o$; $2f_s \pm f_o$; $3f_s \pm f_o$;... $nf_s \pm f_o$ are indistinguishable.

However that is true only under the conditions of periodic sampling. The situation is different whenever DAQ is based on the lately developed signal digitizing theory, including theory of randomized signal processing, NU sampling and quantizing and DASP. Specifically, the basic effect achieved by using NU sampling is avoiding aliasing. This achievable capability is very important for DAQ.

Problems related to NU based DAQ are investigated. The most significant is the problem of cross-interference (CI) between signal spectral components occurring whenever the signal sample values are taken at non-equidistant time instants. Impact of CI on DAQ is taken into account and CI is suppressed.

On the other hand, the obtained results show that various specific advantages and benefits, valuable for practical applications, can be obtained by exploiting NU sampling techniques. Several methods of this kind are considered further.

Data acquisition from high frequency signal sources

As proper application of NU sampling leads to elimination of aliasing, this specific approach to signal sampling can be used to enlarge the frequency range where information carried by high frequency analog sensor signals can be represented by digital data (for example, in cases of signal demodulations).

A structure of DAQ systems based on NU is considered. It can be used for data acquisition directly from high frequency signal sources if operation of the used ADCs is based on quantizing signal sample values taken from the input signal at time instants dictated by a NU sampling point process. In such a case the usage of the additive sampling point process is usually preferable. The upper frequency of the signal spectrum sometimes can exceed the mean sampling rate several times. This makes it possible to acquire data in a wide frequency range extending up to few GHz. However the data obtained under these conditions then must be treated with the specifics of NU sampling taken into account [12, 13, 17].

Increasing the quantity of input channels

The structure of the DAQ system used for data acquisition from a multitude of sensors is based on multiplexing of analog signals. The difference between the traditional structures of this type and this one is in operation of the multiplexer. Whenever it connects inputs to the ADC periodically, the number of inputs is limited by the achievable multiplexer switching rate as the sampling rate of each input then is n times lower, where n is the number of inputs. Replacing the periodic multiplexer switching by randomized allows using the total bandwidth of channels more efficiently. Therefore that makes it possible to increase the number of inputs several times. [14].

Data acquisition at object testing

Exploiting NU sampling techniques and gaining advantages on that basis is a considerably less complicated task under conditions typical for the cases where data are acquired from some biomedical or industrial objects that are being tested. Then the information that has to be obtained is related to the characteristics and properties of the object and the signals taken off the respective object during the tests carry it. These signals reflect reaction of the object on some specific excitation signals that are used. The fact that data acquisition then is performed under conditions where the test signal characteristics, including its spectrum, are given is very useful. Taking this information into account makes it possible to reduce the complexity of the following data processing process significantly. Using of this a priori information leads to substantial simplification of the algorithms. All cross-interference coefficients then can be pre-calculated, matrix C of their values also can be composed and inverted so that all elements $\square_{ij}$ of the inverted matrix inv(C) also could be pre-calculated. Therefore the Fourier coefficients in this case can be calculated directly. That dramatically simplifies and speeds-up the involved calculations.

Using of this approach for data acquisition in the specific case of bioimpedance signal analysis in a wide frequency range of the test signals (up to a few GHz) is described. [35].

Fault tolerant data acquisition

Suppose data are acquired under conditions where the spectra of the involved processes are restricted at relatively low frequencies. Then application of NU sampling is not needed and data might and should be acquired at periodically repeating time instants. Attention is drawn to the possibility of improving the performance of the data acquisition system on the basis of NU sampling even under these conditions if it is needed to protect this system against unpredictable explosive noise bursts. Then the concept of NU sampling and specific data processing typical for NU sampling still proves to be quite useful.

Time diagrams shown in Figure 1 illustrate the data acquisition process under normal conditions in Figure 1(a) and under the impact of noise bursts in Figure 1(b).

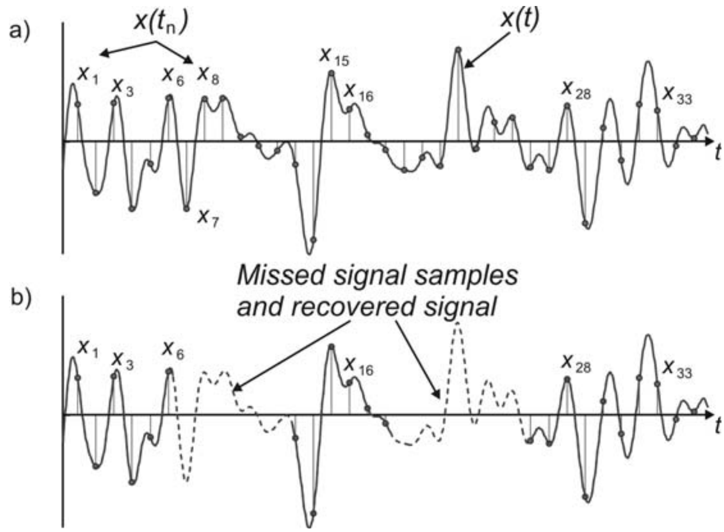

Figure 1. Impact of noise bursts on signals: (a) periodically sampled continuous signal, (b) the same signal with bursts of lost data and the recovered parts of faulted signal.

The idea of improving fault tolerance on this basis is simple. The essence of it is as follows: to improve fault tolerance on the basis of the NU sampling concept, the data should be acquired periodically as usual and facilities usually used for processing of nonuniformly acquired data should be added to the system and used for data reconstruction [12, 13, 17]. This means that under normal data acquisition conditions redundant equipment would be used. The role of this equipment would be to recover the data lost under impact of the noise bursts.

Asymmetric data compression/reconstruction

Data compression function plays a significant role at DAQ. Essential are the advantages related to data compression obtainable if proper NU sampling procedures are used.

At standard data compression/reconstruction data are processed twice to compress and decompress them. Naturally that requires using of computing resources twice and that takes time. Using of the NU sampling procedures makes it possible to reduce the volume of data representing the respective input sensor signals simply by taking out some quantity of the signal sample values or perform other simple operation. It means that in this case no calculations are made at data compression. The computational burden related to reconstruction of the compressed data then is totally placed on the data recovery side of the system. This data acquisition paradigm seems to be with high application potential as it is well suited for significant reducing of the acquired data massive as well as the data compressing costs in terms of equipment volume, weight and power consumption. The basic advantage is that this type of complexity-reduced data acquisition and compression is fast [15]. Note that compressive sensing also belongs to asymmetric data compression/reconstruction paradigm.

Asymmetric 2D image data acquisition/reconstruction

The considered method for asymmetric data acquisition/reconstruction can be used in a wide application area, including 2D data compression and reconstruction [10]. Therefore this method can be exploited for image encoding, transmission (or storage) and reconstruction. Actually this task is quite complicated and can be done using different reconstruction methods. One possible example is considered further and the standard test image, shown in Figure 2(a), is used for that. This standard test image *I* can be defined by its $M \times N$ matrix *I* of pixels or elements as follows:

$$I = \begin{bmatrix} e_{11} & e_{12} & e_{13} & e_{14} & e_{15} & e_{16} \ldots & e_{1N} \\ e_{21} & e_{22} & e_{23} & e_{24} & e_{25} & e_{26} \ldots & e_{2N} \\ e_{31} & e_{32} & e_{33} & e_{34} & e_{35} & e_{36} \ldots & e_{3N} \\ e_{41} & e_{42} & e_{43} & e_{44} & e_{45} & e_{46} \ldots & e_{4N} \\ \ldots & \ldots & \ldots & \ldots & \ldots & \ldots \ldots & \ldots \\ e_{M1} & e_{M2} & e_{M3} & e_{M4} & e_{M5} & e_{M6} \ldots & e_{MN} \end{bmatrix} \quad (1)$$

According to the considered compression method, many of pixels have to be replaced by zeroes using NU signal sampling. The best of so far found approaches are based on design and usage of a mask containing logic 1 and 0. An example of it is given as matrix *H*.

$$H = \begin{bmatrix} 0\,1\,0\,0\,0\,1\,0\,0\,1\,0\,0\,1\,0\,0 \ldots 0 \\ 0\,0\,1\,0\,0\,0\,1\,0\,0\,1\,0\,0\,1\,0 \ldots 0 \\ 0\,0\,0\,1\,0\,0\,0\,1\,0\,0\,1\,0\,0\,1 \ldots 1 \\ 1\,0\,0\,0\,1\,0\,0\,0\,1\,0\,0\,1\,0\,0 \ldots 0 \\ 0\,1\,0\,0\,0\,1\,0\,0\,0\,1\,0\,0\,1\,0 \ldots 0 \\ \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \\ 0\,1\,0\,0\,1\,0\,0\,0\,1\,0\,0\,0\,1\,0 \ldots 0 \end{bmatrix} \quad (2)$$

To generate this mask, taking out of pixels is done in a specific way by using additive random sampling sequence. The mean distance μ of two neighbouring pixels left in the image is equal to 5 and width of the uniform distribution of pseudorandom numbers □ is equal to 3.

The compressed image is obtained in a very simple and fast way, just by using logic or scalar multiplication of generated mask matrix with the image matrix:

$$I_c = I \times H = \begin{bmatrix} 0\ e_{12}\ 0\ 0\ 0\ e_{16}\ 0\ 0\ e_{19} \dots\ e_{1N} \\ 0\ 0\ e_{23}\ 0\ 0\ 0\ e_{27}\ 0\ 0 \dots\ e_{2N} \\ 0\ 0\ 0\ e_{34}\ 0\ 0\ 0\ e_{38}\ 0 \dots\ e_{3N} \\ e_{41}\ 0\ 0\ 0\ e_{45}\ 0\ 0\ 0\ e_{49} \dots\ e_{4N} \\ \dots\ \dots\ \dots\ \dots\ \dots\ \dots\ \dots \\ 0\ e_{M2}\ 0\ 0\ e_{M5}\ 0\ 0\ 0\ e_{M9} \dots\ e_{MN} \end{bmatrix} \tag{3}$$

The obtained sparse image after losing 80.27% of its pixels is shown in Figure 2(b). Evidently image compression performed in such a way is a quite inexpensive and technically effective operation. The complexity of this type of image compression is much lower than the complexity of the usually exploited rather complicated standard image compression algorithms. That apparently represents a significant advantage of the proposed and described 2D data compression algorithm.

Reconstruction of the compressed image

Two approaches of sparse image reconstruction were considered and studied. Both of them have a common first recovery stage. The result obtained after this stage is shown in Figure 2(c). According to the suggested reconstruction method, part of all unknown image pixels can be calculated at this stage by processing its subsequent pixel values if these are known. In this particular sampling case, the maximum count of the known pixels in close proximity (considering 2D 4-pixel connectivity case) can be 2. These pixels can be approximately calculated as follows:

$$\begin{aligned} \hat{e}_{mn,l} &= 0.5(e_{m-1,n} + e_{m,n+1}), \ \textit{for the left side estimates} \\ \hat{e}_{mn,r} &= 0.5(e_{m+1,n} + e_{m,n-1}), \ \textit{for the right side estimates} \end{aligned} \tag{4}$$

Where $m = 1, 2, 3 \dots M$, $n = 1, 2, 3 \dots N$. The matrix of the recovered image pixel values after the first reconstruction cycle is the following:

$$\hat{I}_{r1} = \begin{bmatrix} \hat{e}_{11} & e_{12} & \hat{e}_{13} & 0 & \hat{e}_{15} & e_{16} & \hat{e}_{17} & 0 & e_{19} \dots e_{1N} \\ 0 & \hat{e}_{22} & e_{23} & \hat{e}_{24} & 0 & \hat{e}_{26} & e_{27} & \hat{e}_{28} & 0 \dots e_{2N} \\ 0 & 0 & \hat{e}_{33} & e_{34} & \hat{e}_{35} & 0 & \hat{e}_{37} & e_{38} & \hat{e}_{39} \dots e_{3N} \\ e_{41} & \hat{e}_{42} & 0 & \hat{e}_{44} & e_{45} & \hat{e}_{46} & 0 & \hat{e}_{48} & e_{49} \dots e_{4N} \\ \dots & \dots & \dots & \dots & \dots & \dots & \dots & \dots & \dots \dots \dots \\ \hat{e}_{M1} & e_{M2} & 0 & \hat{e}_{M4} & e_{M5} & 0 & 0 & \hat{e}_{M8} & e_{M9} \dots e_{MN} \end{bmatrix} \tag{5}$$

At the next stages this process can be continued iteratively. All the pixel values are estimated in this way after a few cycles and the full image is recovered.

The average relative error, obtained in this particular case at image reconstruction performed in accordance to the described method, is equal to 0.1648 %.

The second considered reconstruction method is based on application of SEC-OEX method. That leads to the result shown in Figure 2(e). In this particular case, the average relative error of image recovery is 0.3147 %, what actually is worse than the result obtained by the previous method. While this larger error can be explained by the particularities of the processed test image signals, image reconstruction based on the SECOEX method is much more complicated and more time consuming in comparison with the developed method.

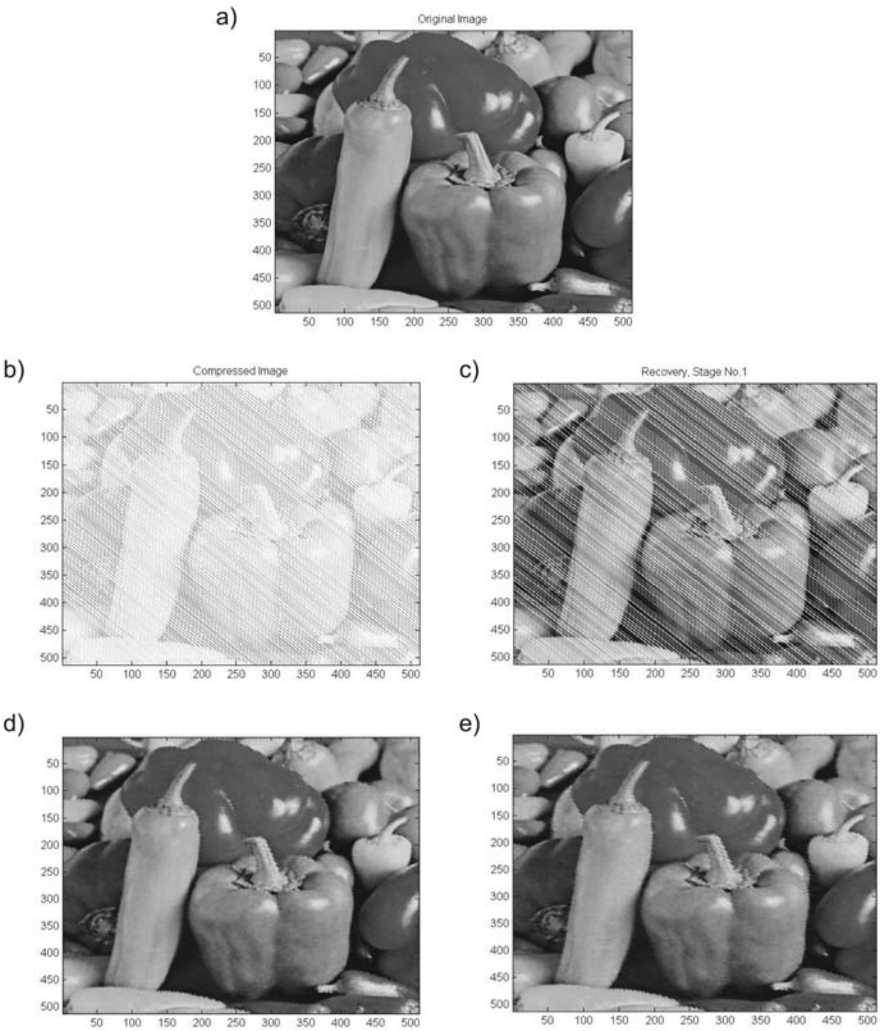

Figure 2. Example of asymmetric DAQ compression/reconstruction: (a) the original image by size 512×512 pixels; (b) sparse image after losing 80.27% of its pixels; (c) the image recovered at the first recovery stage; (d) image recovered by calculating unknown pixels from their neighbours and (e) recovered image obtained on the basis of SECOEX method.

The asymmetric 2D data acquisition/reconstruction has significant advantages for application cases where computational power is limited and transmitting of the acquired data is expensive in terms of spent time, memory and equipment resources.

The applicability of this method has been widened to cover also processing of colour images. An experimental system has been developed and made for studies of the described image compression method.

Wideband signal digitizing in enlarged dynamic range

The frequency range, where the currently available 10 to 12 bit ADCs are applicable, often is not wide enough. On the other hand, the dynamic range of the ADCs, applicable for analog-digital conversions in GHz frequency range, is limited by the achievable quantization bit rate usually not exceeding 4 bits [19, 41]. These typical problems, arising at attempts to convert analog signals into their digital counterparts in a wide frequency range extending up to GHz frequencies, are studied and specific approaches to resolution of them is suggested further.

Combining precise and low-bit very fast sampling

Combining precise and low-bit very fast sampling, in particular, significantly widen the area where DAQ can be performed in GHz frequency range by using low bit rate ADC. This type of signal sampling/reconstruction scheme is shown in Figure 3. The sampling operation in this case is based on adding to the precise ADC another low-bit rate one, which has to be very fast.

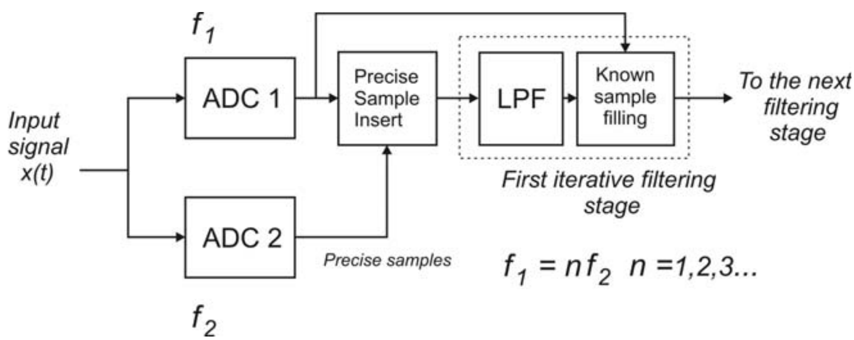

Figure 3. Structure of the electronic systems used for iterative reconstruction of signal waveforms, when signal is sampled using two ACDs in parallel.

Reconstruction of signal waveforms in this case is based on iterative filtering procedure. Results of 4 bit fast periodic sampling are used for filling the spaces remaining empty after the precise signal sample values have been taken at time instants according to the described additive random sampling. Then the conditions for waveform reconstruction are more favourable and the obtained results also are significantly improved. They are displayed in Figure 4. To further improve the waveform reconstruction, using of yet another one of the randomized procedures, namely, randomized quantizing is suggested and the results obtained in that case also are shown there.

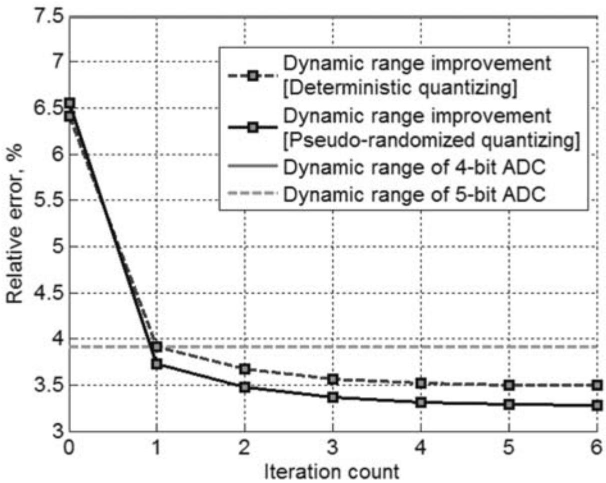

Figure 4. Results of waveform reconstruction when precise undersampling and low-bit fast periodic sampling procedures are used in parallel. Using of pseudo-random quantizing improves the precision.

Conclusions

A number of benefits achievable for DAQ by using the concept of NU sampling are described to show the application potential of DASP methods for data acquisition. Considered approaches are beneficial especially for:

- DAQ based on pseudo-randomized multiplexing;
- Asymmetric data compression/reconstruction;
- Data acquisition from objects under tests;
- Improving fault tolerance of data acquisition.

NU sampling has high potential in the field of 2D data acquisition or image data acquisition. To show this, the example of asymmetric image data compression/reconstruction was considered. The standard test image has been taken and compressed taking out 80% of its pixels according to the asymmetric DAQ concept. After that it was successfully reconstructed with 0.1648 % of average relative error. Thus it has been shown that application of NU sampling makes it possible to perform very simple image data compression. The achieved compression rate, in the case of the considered example, is equal to 5. It is shown that the developed method and algorithm can be used also for 2D data acquisition for colour image data compression and reconstruction with the same compression rate of 5.

5. Methods for Data Acquisition Exploiting Advantages of Pseudo-randomized Quantizing

Specifics of DAQ based on pseudo-randomized quantizing (PRQ) are considered in more detail in the particular case of impedance measurement data acquisition and pre-processing. Impact of signal quantizing on DAQ resolution and DFT coefficient estimation precision is considered and the potential of pseudo-randomized quantizing for resolution improvement is shown.

The impedance (including bioimpedance) measurements in many cases could be reduced to estimation of the signal spectra on specific pre-determined frequencies. The task of this spectrum analysis is made more difficult by the fact that the frequency range of interest is wide. While effective methods and techniques for impedance measurements at frequencies up to several MHz have been developed and are used, there are problems when the spectrum analysis of the modulated impedance signals has to be performed in the frequency range up to several hundreds of MHz or even up to several GHz. The involved data acquisition and processing tasks then become rather challenging. In particular, data have to be processed in real-time and with sufficiently high resolution.

Processing of impedance signals on the basis of complexity reduced DFT

The pseudo-randomly quantized signal values contain significantly increased number of bits in comparison with deterministically or randomly quantized signals as the value of the pseudo-random noise $\square_k$, used at quantizing, is added [15, 16]. Directly processing of the quantized signal clearly leads to a complicated processing procedure. The problem was considered and it is shown how to avoid these complications. Suppose impedance signal demodulation is on estimation of the Fourier coefficients.

It is suggested to do that on the basis of the following equations:

$$\begin{aligned} \hat{a}_i &= \frac{2}{N}\sum_{k=1}^{N}\hat{x}_k \cos 2\pi f_i t_k = \frac{2q}{N}\sum_{k=1}^{N} n_k \cos 2\pi f_i t_k + \gamma_i^a \\ \hat{b}_i &= \frac{2}{N}\sum_{k=1}^{N}\hat{x}_k \sin 2\pi f_i t_k = \frac{2q}{N}\sum_{k=1}^{N} n_k \sin 2\pi f_i t_k + \gamma_i^b \end{aligned} \tag{6}$$

Where γ_i^a and γ_i^b are corrections:

$$\begin{aligned} \gamma_i^a &= \frac{2q}{N}\sum_{k=1}^{N}(\xi_k - \frac{1}{2})\cos 2\pi f_i t_k \\ \gamma_i^b &= \frac{2q}{N}\sum_{k=1}^{N}(\xi_k - \frac{1}{2})\sin 2\pi f_i t_k \end{aligned} \tag{7}$$

These corrections remain constant, can be pre-calculated for all excitation frequencies and used. The advantage of this approach to estimation of the Fourier coefficients is evident. Data can be processed in this case in a very simple and fast way.

Obtained results of the suggested method

Diagrams given in Figure 5 illustrate the precision obtainable at using the suggested methods for impedance data acquisition and demodulation of the impedance signals.

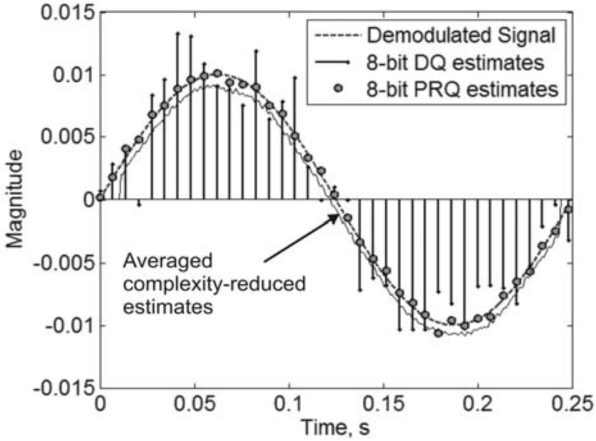

Figure 5. Estimates of the demodulated signal obtained by using deterministic quantizing (1) and pseudo-randomized quantizing (2). Averaged estimates based on the complexity-reduced approach using sign(sin) and sign(cos) functions are given as curve (3).

Demodulation quality improvement by filtering

After impedance signal demodulation, the extra precision of processing could be obtained with ordinary digital filtering. For this purpose application of the moving average filter MAF is considered. If it is needed such filtering could be used iteratively. Also it is expected that usage of other filter types, like low-pass filters LPF, would provide good results.

Figures in Table 1 illustrate comparison of signal quantization errors with and without filtering. As can be seen, the filtering in general reduces quantizing errors, no matter whether it is MAF or LPF filtering.

Table 1

Typical relative quantizing error readings.

	Moving average filter MAF length	3	5	7	9
1	Deterministic 4 bit quantizing (no filtering)	0.1010	0.1010	0.1010	0.1010
2	Deterministic 12 bit quantizing (no filtering)	0.00037	0.00037	0.00037	0.00037
3	4+4 bit PRQ quantizing (no filtering)	0.0994	0.0993	0.1005	0.1004
4	4+4 bit PRQ quantizing, LPF	0.0431	0.0403	0.0426	0.0406
5	Deterministic 4 bit quantizing, LPF	0.0642	0.0642	0.0642	0.0642
6	4+4 bit PRQ quantizing, MAF	0.0399	0.0294	0.0274	0.0271

As it can be seen the PRQ quantizing provides significantly better demodulated signal precision than the traditional deterministic quantizing.

Conclusions

Application specifics of the basic three signal quantizing methods (methods for deterministic, randomized and pseudo-randomized quantizing) for data acquisition are studied. In many cases PRQ proves to be the best of them. That, of course, depends on the particular application.

DFT is often needed for biomedical DAQ applications, where performance of bioimpedance signal demodulation is required. Methods for such signal processing are essential for design of medical equipment. This work considers impedance signal demodulation based on calculations of DFT coefficients on several frequencies and the advantages of using PRQ are shown.

Due to these features, PRQ is an approach to quantizing that is rather valuable for achieving high performance of DAQ systems. That is confirmed by results obtained in the area of DAQ used for impedance data acquisition.

It is shown that in this application area:

- Improved resolution of very fast low bit rate ADCs (several GHz sampling frequency) at wideband low bit rate DAQ. In other words, PRQ reduces the quantizing bit rate, widening the application range of rough quantizing;
- Better digitized signal quality (in terms of signal sample values precision);
- Simplified, energy efficient DAQ hardware and impedance signal processing.

In general, the obtained research results confirm that design and performance of DAQ systems often can be significantly improved by exploiting PRQ.

5. Data Acquisition Based on Gathering Timing Information

The specific approach to signal sampling, based on detection of signal and reference function crossings is considered further. It is shown that under certain conditions it is possible to represent analog signals with timed sequences of events without loss of information and that this possibility can be exploited for developing competitive DAQ systems based on detection of signal and reference crossing events (SRC sampling) that have attractive and useful advantages.

The comprehensive comparison of the classical signal sampling method and suggested method for SRC sampling is given in Table 2.

Table 2

Comparison of two alternative sampling approaches.

Classical sampling approach	**SRC sampling approach**
Signal sample value acquiring	
• Signal sample values are taken at predetermined time instants • Sample values are represented by respective voltage (current) levels • Uniformly spaced signal sample values • Remote sampling implementations usually not acceptable • Aliasing determined by Nyquist frequency	• Sample values are taken at the signal and reference function crossing instants • Sample values are represented by the time instants when the crossing events take place • Nonuniformly spaced signal sample values • Remote sampling applicable • Specific aliasing conditions • Reference function defines the envelope of the sampled signal instantaneous values
Signal sample value transmission	
• Requires transmission of constant levels • Sensitive to the ambient noise • Transmission acceptable only over short distances	• Based on transmission of time instants • Relatively insensitive to the ambient noise • Transmission might be performed over relatively large distances • Provides for data compression
Quantizing	
• Sample values are quantized directly	• Reference function values are quantized at the crossing time instants • High precision synchronization is required
Multi-channel operation	
• Based on switching the analog input signals • Number of channels limited by hierarchic multiplexer structures	• No switching of analog signal performed • Multiplexing of channels, provided by enabling function, is much simpler • Well suited to data acquisition from a large quantity of signal sources • Well suited to the specifics of Ultra Wideband communication
Processing	
• Based on the classical DSP algorithms	• Typically specific algorithms are needed for processing • Applicability of standard algorithms achievable under certain condition • Complexity-reduced pre-processing
Advantages for DAQ systems	
• Features and obtainable benefits are well known	• Reduced complexity and power consumption at the DAQ system front-end part • Significantly enlarged number of data channels

Preference is given to the method for DAQ based on detection of signal and sine-wave crossing time instants, so-called SWC signal sampling method. The sine-wave is preferred as a reference function due to its positive characteristics, in particular, spectral purity of it. This significant feature of SWC sampling might be exploited for reducing the complexity of the algorithms for pre-processing of signals.

Potential of SWC sampling for simplifying some DSP algorithms

Algorithms for processing digital signals obtained in result of SWC sampling often can be significantly less complicated than the widely used conventional ones. In particular, in many cases it is possible to avoid multiplications of multi-bit numbers. The unique constant envelope feature of SWC sampling makes it possible to develop rational algorithms, in particular, for Discrete Fourier Transform based spectrum analysis and waveform reconstruction. For example, the following equations can be used for calculation of Fourier coefficient estimates $\hat{a}_i$, $\hat{b}_i$ not requiring multiplication of multi-digit numbers as usual [1, 2, 6]:

$$\begin{aligned} \hat{a}_i &= \frac{A_r}{N}\sum_{k=0}^{N-1}(\sin 2\pi(f_r - f_i)t_k + \sin 2\pi(f_r + f_i)t_k) \\ \hat{b}_i &= \frac{A_r}{N}\sum_{k=0}^{N-1}(\cos 2\pi(f_r - f_i)t_k - \cos 2\pi(f_r + f_i)t_k) \end{aligned} \quad (8)$$

Where A_r, f_r are the amplitude and frequency of the sinusoidal reference function, t_k denotes the crossing time instants and N is the number of signal samples processed.

This approach to complexity reduction of algorithms for data pre-processing is usable only under the condition that the reference function is sinusoidal. This represents a strong argument in favour of using this type of reference functions. The problem related to the necessity of operating with sine and cosine functions in this case can be easily resolved by using look-up tables. The developed method for complexity-reduced digital filtering and parameter estimation is considered as an invention and the respective European patent application is published [6].

SWC sampled signal properties in the frequency domain

SWC sampled signal properties in the frequency domain are quite specific and they were explored in detail. First of all, SWC represents a specific modification of NU sampling and, as it is typical for this type of sampling, SWC has anti-aliasing properties. It is shown that there is no full-scale frequency overlapping. While peaks appear in the spectrogram at some indicated and defined frequencies, they are significantly suppressed. This type of aliasing is considered as so-called fuzzy aliasing [15].

Data acquisition based on SWC sampling

Data acquisition based on SWC sampling is illustrated in Figure 6. Digitally timed events, defined as crossings of an input signal and a reference function, is used in this case for representation of analog signals in the digital domain. The most responsible elements of this system apparently are the comparators used for detection of the time instants t_k at which the signal $x(t)$ intersects the reference function $r(t)$ that is given as a sine wave at frequency f_r. It has constant amplitude A_r. The physical output $y(t)$ of the comparator is formed as a pulse whenever the sampler is enabled by a function $z(t)$ (Figure 6(b)) and a crossing of the input signal $x(t)$ and the reference function $r(t)$ takes place.

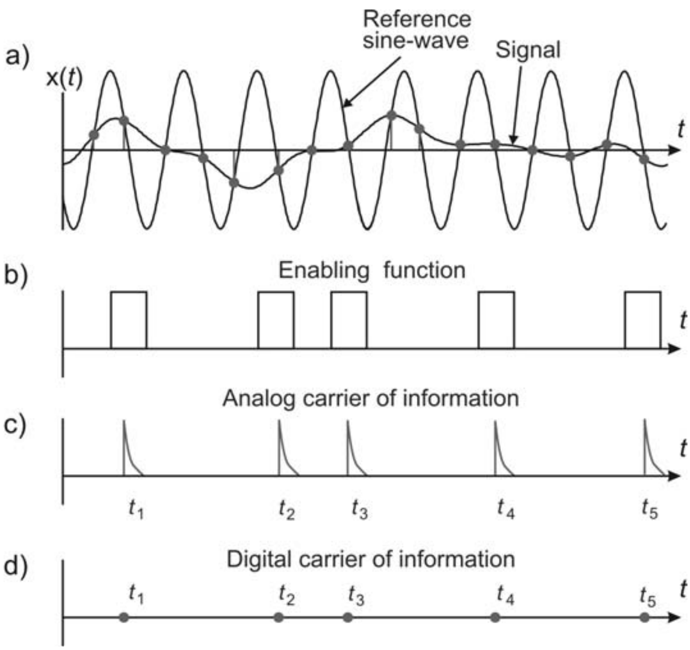

Figure 6. Time diagrams illustrating specific two types of input signal representation in the digital domain. (a) sampling based on signal and the reference function crossings; (b) enabling function; (c) analog carrier of information; (d) digital carrier of information.

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 sharp front edges are formed with a constant delay after each signal and the reference sine-wave crossings. 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 host computer. Note that this sequence fully represents the respective input signal. It means that the crossing instant sequence might be used either for recovery of the input signal sample values and reconstruction of the signal or the input data processing might be based on direct processing of these crossing instants as it is explained below [8]. While the sequence of the signal and reference function crossing instants represents in this case the input analog signal in the digital domain, either an analog or a digital carrier may be used for transmitting this information. A train of position-modulated short pulses (Figure 6(c)) is used as an analog information carrier and a sequence of digital crossing instant values $\{t_k\}$ (Figure 6(d)).

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 7). The digital carrier is used at the stages of data reconstruction and/or their preprocessing and data transfer to computers.

While various techniques might be used for the recovery of the input signal sample values, basically a replica of the reference function waveform is sampled and quantized for that. Note that this copy of the reference function could be given either in analog or digital format. The signal sample values, of course, might be recovered and presented digitally in both cases.

Architecture of an energy-efficient system for multichannel data acquisition

The suggested hardware architecture for simultaneous multichannel data acquisition is shown in Figure 7. Reduction of power consumption to a large extent is due to the method used for sampling. Application of it not only leads to the simplicity of the front-end design of this system. Remote samplers that can be placed directly at the locations of the sensors then are used rather than ADCs. This approach is well suited for remote signal sampling applications and for building the shown architecture of the system that actually is a distributed ADC. As can be seen, the sampling and quantising operations, in this structure, are distanced. This approach makes it possible to use many remote samplers at the front-end of it and to gather data from them in a rational way. To ensure proper performance of them in parallel, specific enabling control functions are introduced and used. Only those crossings are taken into account that happen during the time intervals when the respective comparator is enabled by a specially generated enabling function. This enabling function is exploited also for executing the input multiplexing. The analog input signal switching could be avoided then and that certainly is a significant positive fact.

At the current technological level, if the SWC based DAQ is implemented on available ICs, this limit is approximately 25 *MHz* for 8 to 10 bit quantizing. It is not so easy to achieve it. More realistic figures are 5–10 *MHz* with the quantization rate up to 12 bits. This result was obtained testing the experimental system developed in the framework of ERAF project Nr. VDP1/ERAF/CFLA/05/APK/2.5.1/000024/012 “Development of multi-channel systems for acquisitions of data from biomedical, ecological and industrial systems and transferring them to computerized systems", co-sponsored by the European Union. This leads to the maximal number of channels equal to 1000 for the mean sampling frequency equal to 20 *KHz*. Then the input signal could be processed in the alias-free way within the bandwidth 0–10 *KHz*. More often than not the required input numbers are smaller. Then the input signal bandwidth could be proportionally widened. To achieve results fully reflecting the potential of this approach, special ASICs for the analog-digital front end devices must be developed and used.

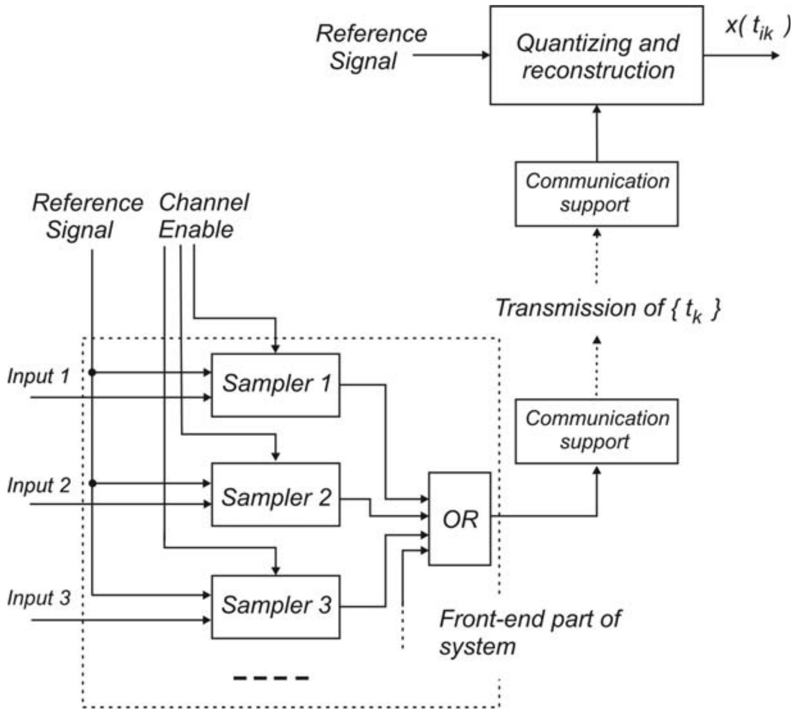

Figure 7. Architecture of the considered energy-efficient system fulfilling simultaneous data acquisition functions based on SWC sampling.

Achieving the applicability of standard DSP algorithms

To avoid SWC sampling nonuniformities, regularization of SWC sampling results is considered. The very simple method for this regularization has been developed and is suggested. It has remarkable advantages in comparison with other reconstruction methods where burdensome signal processing is involved. The suggested signal regularization does not require any additional computation power at all. Of course, the results may be acceptable for end-users only under certain conditions.

Regularization of SWC sampling leads to obtain the possibility of using standard DSP algorithms for processing SWC sampled signals. Whenever this type of regularization is carried out, the wealth of existing DSP algorithms can be used for processing the signal sample values obtained in result of SWC sampling. This method can be implemented with help of the enabling function.

The curves in Figure 8 have been obtained varying signal sampling frequency and keeping constant reference frequency. The reference frequency, with the enabling decimation factor *n* varying, was changed according to $f_s = f_r/n$. Then with *n* growing, the sampling conditions are becoming closer to regular until it reaches the point at which the sampling mode could be considered as regular and all DSP algorithms can be directly applicable. Unfortunately the regularization process proportionally decreases the signal sampling frequency.

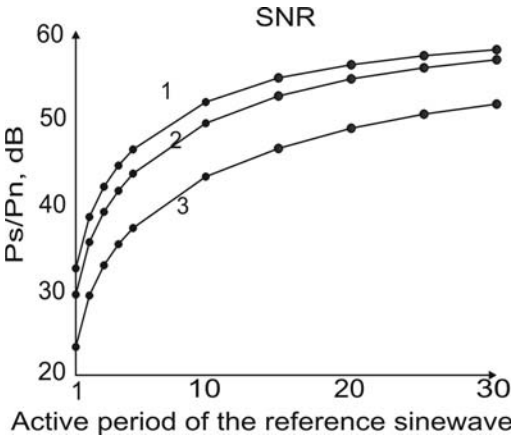

Figure 8. Signal-to-noise ratio SNR versus decimation coefficient n of the activated reference function periods for three various signals.

How it can be seen from Figure 8, the SNR is acceptable starting to 20-th activated sine-wave half period. Actually it depends on particular application.

Conclusion

Studies of SWC based DAQ lead to the following conclusions:

- Algorithms and signal processing. Signals sampled according to the SWC method has constant envelope not depending on the original analog signal frequency content. That makes it possible to develop algorithms for processing them without massive multiplication of multi-digit numbers.
- A new signal filtering algorithms, exploiting the constant envelope property, have been developed is described in [6].
- DFT without massive multiplication operation is possible.

- Applicability of traditional DSP algorithms under certain conditions has been achieved. This applicability of classical DSP algorithms can be obtained by using regularization process of SWC sampled signals.
- To gain from of the constant envelope sampling, it has to be learned how to effectively cope with the drawbacks related to the nonuniformity of the obtained digital signals. The signal regularization using the enabling function is recommended as effective tools for controlling signal sampling conditions. However it reduces the mean signal sampling rate for each DAS channel depending on particular application.
- Sampling properties. The SWC sampling properties are evaluated in the time domain and as well as in the frequency domain. Trading-off the mean sampling rate against the time resolution has to be done to provide for proper quality sampling. It is considered how the particularities of this type of signal spectra differ from spectra of the traditionally sampled signals.
- DAQ system architecture. Application of SWC sampling leads to simpler and more energy efficient hardware architectures of data acquisition systems as the signal sampling operation can be performed on the basis of a single comparator instead of ADC.

The main conclusion is that SWC signal sampling method is well suited to simultaneous multi-channel data acquisition from a large number of relatively low-frequency signal sources.

6. FPGA based implementation of the considered DAQ methods

A particular application of DASP theory for Fourier coefficient calculation has been considered. FPGA based implementation of a specific DFT multiplier-less structure (F-DFT Processor) for fast pre-processing of data has been developed and FPGA implementation specifics studied. This type of FPGA is applicable for designs of various application-specific systems fulfilling digital signal pre-processing related to data time-frequency representation, data compression/reconstruction, spectrum analysis, filtering, parameter estimation and demodulation.

Experimental system has been programmed and tested on the basis of *Altera Cyclone II EP2C70F672C6N* chip model. The core of F-DFT Processor is multiplier-less Adder-Subtractor Matrix. In addition to this Matrix, there are data input and output subsystems. Designs of these subsystems depend on specifics and requirements of various applications. Therefore the data input and output version is just one of the possible modifications. Real-time DFT and batch processing of data for DFT are two of the most often needed types of data pre-processing that can be performed on the basis of the developed FPGA chip.

Requirements dictated by a specific bioimpedance measurement system were taken into account at definition of the parameters of DFT for the specification of this FPGA chip. These parameters are the following: Fourier coefficients have to be calculated simultaneously at frequencies f0 (data mean value), f1, f2, f4, f8, f16, f32 and f64 by processing N=256 real signal sample values. The absolute values of these frequencies depend only on the parameters of data acquisition, in particular, on the specifics of the signal source and the used ADC. Thus the considered processor can perform DFT in a wide frequency range from KHz up to GHz.

The most important parameter that has to be experimentally measured is the achieved pin-to-pin delays characterizing the Matrix, as this parameter actually defines the upper repetition rate of DFT operation or system clock frequency. Printout of so-called Clock setup, showing pin-to-pin delays for the involved data flows at the specified frequencies, follows.

Clock Setup: 'clk'

	Slack	Actual fmax (period)	From	To
1	N/A	101.49 MHz (period = 9.853 ns)	req[231][7]	f8c7[0]
2	N/A	101.79 MHz (period = 9.824 ns)	req[231][7]	f16c1_2[0]
3	N/A	101.82 MHz (period = 9.821 ns)	req[109][6]	f16c3_1[0]
4	N/A	101.94 MHz (period = 9.810 ns)	req[109][8]	f16c3_1[0]
5	N/A	102.06 MHz (period = 9.798 ns)	req[248][7]	f32c2_1[0]
6	N/A	102.21 MHz (period = 9.784 ns)	req[237][8]	f16c3_1[0]
7	N/A	102.26 MHz (period = 9.779 ns)	req[136][6]	f32c2_1[0]
8	N/A	102.29 MHz (period = 9.776 ns)	req[231][7]	f8c7[1]
9	N/A	102.45 MHz (period = 9.761 ns)	req[147][6]	f16c3_1[0]
10	N/A	102.45 MHz (period = 9.761 ns)	req[87][8]	f8c7[0]
11	N/A	102.53 MHz (period = 9.753 ns)	req[231][7]	f16c1_2[1]
12	N/A	102.56 MHz (period = 9.750 ns)	req[109][6]	f16c3_1[1]
13	N/A	102.67 MHz (period = 9.740 ns)	req[109][7]	f16c3_1[0]
14	N/A	102.67 MHz (period = 9.740 ns)	req[173][6]	f16c3_1[0]
15	N/A	102.68 MHz (period = 9.739 ns)	req[109][8]	f16c3_1[1]
16	N/A	102.70 MHz (period = 9.737 ns)	req[87][8]	f16c1_2[0]
17	N/A	102.72 MHz (period = 9.735 ns)	req[147][8]	f16c3_1[0]

Figure 9. Printout showing test results of the F-DFT Processor measured by Quartus software. Achievable clock frequencies for various data flow routes and the related pin-to-pin delays are given for the indicated addresses.

As can be seen in Figure 9, the worst case is for the data flow in the Matrix from register reg[231][7] to the output f8c7[0]. According to Figure 9, the achievable clock frequency of the F-DFT processor under the described conditions is 101.49 MHz and the worst-case pin-to-pin delay is 9.853 ns. Required resources for the F-DFT processor design are given in the printout of Figure 10.

Flow Summary	
Flow Status	Successful - Tue Sep 06 14:05:21 2011
Quartus II Version	10.0 Build 262 08/18/2010 SP 1 SJ Web Edition
Revision Name	f_dft_processor
Top-level Entity Name	f_dft_processor
Family	Cyclone II
Device	EP2C70F672C6
Timing Models	Final
Met timing requirements	Yes
Total logic elements	14,108 / 68,416 (21 %)
Total combinational functions	11,640 / 68,416 (17 %)
Dedicated logic registers	7,498 / 68,416 (11 %)
Total registers	7498
Total pins	26 / 422 (6 %)
Total virtual pins	0
Total memory bits	0 / 1,152,000 (0 %)
Embedded Multiplier 9-bit elements	0 / 300 (0 %)
Total PLLs	0 / 4 (0 %)

Figure 10. Required resources of the designed F-DFT processor (*Altera Quartus* snapshot).

In comparison with the classical structure of the digital FIR filter on 256 coefficients design of the F-DFT processor requires much less elements, approximately 2 times less *Altera* logical elements. Performance of both devices is characterized by pin-to-pin delays and the clock frequency. To compare both, the following points have to be taken into account: (1) the considered particular F-DFT processor calculates simultaneously outputs of 15 filters (1 filter for f0 and two filters for each other 7 frequencies) in parallel; (2) the structure of the considered single FIR filter is pipelined, therefore the indicated clock frequency for it depends on the delay of only a single filter stage.

For the explained reasons, the performance of both devices cannot be directly compared. As to the resources, the requirements of F-DFT are significantly reduced.

Table 3

Summary of required resources and parameters of the F-DFT processor and a particular digital FIR filter.

Cyclone II Device EP2C70F672C6	**F-DFT data block 256**	**Pipelined FIR on 256 coefficients with 256 multipliers**
Clock frequency	101.49 MHz	117.25 MHz
Total combinational functions	11,640 / 68,416 (17 %)	28,785 / 68,416 (42 %)
Dedicated logic registers	7,498 / 68,416 (11 %)	16,807 / 68,416 (25 %)
Total logic elements	14,108 / 68,416 (21 %)	29,115 / 68,416 (43 %)
Total pins	26 / 422 (7 %)	30 / 422 (7 %)
Embedded Multiplier 9-bit elements	0 / 300 (0 %)	0 / 300 (0 %)

7. Obtained research results

Overview of the specifics related to development and production of currently available DAQ systems reveals the basic tendencies characterizing the situation in this field. While about 300 industrial companies are active in this area and seemingly many various DAQ systems are offered, actually progress there is relatively slow. As it is shown the basic limiting factor is the dependency of DAQ methods on the classical theory of DSP, especially on the dominating theoretical principles of periodic sampling. Theory of randomized and alias-free signal processing is rather well suited for DAQ applications. This fact has been taken into account and the undertaken research activities of these doctoral studies are based on and exploit a number of essential results of the DASP theory. The point is that signal digitizing performed in this way is rather flexible and can be adjusted to the needs of various DAQ applications. The approach to DAQ is based on these considerations.

Summary of the obtained research results follows:

1. Research results related to exploitation of nonuniform sampling for DAQ showing that this approach has significant potential for improving DAQ. The obtained results in this area, specifically, are the following:
    - Elimination of the input signal source number dependence on the sampling rate that limits the capabilities of the classical DAQ systems. That makes possible simultaneous data acquisition in parallel from a significantly increased (up to 8-12 times) number of signal sources.
    - Development of methods and algorithms for asymmetric data compression/reconstruction (in the sense that most of the computational burden is put on the reconstruction stage), including very fast image data asymmetric compression/reconstruction.
    - Development, making and testing of an experimental complexity-reduced hardware/software system for image data compression based on this method, achieving image data compression up to 5 times for various images, including colour and standard images.
    - Development, analysis and description of various approaches to DFT coefficient estimation, in particular, for spectrum analysis of bioimpedance data obtained in result of nonuniform sampling based bioimpedance demodulation.
2. Obtained research results, related to exploitation of deterministic, randomized and pseudo-randomized quantizing methods for DAQ, are the following:
    - Results of analysis and evaluation of the impact pseudo-randomized quantizing has on the precision of spectrum analysis of data.
    - Methods for spectrum analysis of data, based on pseudo-randomized signal quantizing.

   - Complexity reduced approach to bioimpedance data demodulation using DFT on the basis of rectangular functions.
   - Results of comparison and evaluation of all three quantizing versions showing that pseudo-randomized quantizing usually outperforms two other quantizing options.
3. Results of research related to a new approach to DAQ based on acquiring timing information:
   - A method for DAQ, based on detection of signal and sine-wave crossing instants (SWC sampling), computer-simulated and extensively explored.
   - Computer simulation results showing the most significant advantage of SWC based DAQ systems, namely, their capability of data gathering from a very large quantity of signal sources (up to 256-512) distributed over an object of technical or biomedical origin.
   - Method for SWC sampled signal regularization that can be performed to ensure the applicability of standard DSP algorithms.
   - Special signal processing methods applicable in the case of SWC for complexity-reduced design of digital filters and multiplier-less structures for DFT [6].
4. Developed MATLAB programs for Computer simulations of the explored DAQ methods.
5. Results obtained in the experimental research area: VHDL description for FPGA implementation of a specific Fast DFT Processor, test results of this FPGA, showing, specifically, the required element count and parameters characterizing the achieved operational speed.

8. Conclusions

The goal of this work, discovering innovative methods for efficient massive data acquisition from real life objects and supplying computers with this information, has been reached and the planned tasks have been fulfilled. The developed and investigated innovative methods for complexity-reduced DAQ and energy-efficient pre-processing of the data are based on the theory, concepts and methods of the non-traditional digital signal processing DASP. The obtained research results show: (1) how that can be done and (2) what can be gained at computer system linking to the real world technical and biological objects by using the developed DAQ methods.

The basic benefits that can be obtained in this way, specifically, are the following:

1. The developed methods for multi-channel data acquisition systems are flexible and applicable for simultaneous data acquisition from many signal sources.

2. Simultaneous data acquisition from many signal sources is performed in parallel under conditions where the upper frequencies of the input signal spectra do not depend on the quantity of the input channels.
3. Input signals can be sampled directly at their sources by front-end devices that are much simpler than the traditionally used ADCs.
4. Consequently, the power consumption of these front-end devices might be significantly lower.

However DASP technology is specific. Therefore hardware and software implementing DAQ based on this technology are specific as well. MATLAB based 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.

Experimental investigations of the developed DAQ methods were carried out by using hardware tools that have been developed in the Laboratory 2.2.of Institute of Electronics and Computer Science in the framework of ERAF Project Nr. VDP1/ERAF/CFLA/05/APK/2.5.1/000024/012 "Development of multi-channel systems for acquisitions of data from biomedical, ecological and industrial systems and transferring them to computerized systems", co-sponsored by the European Union. Author of this research work participated in this project as a member of the team.

Tests and experimental evaluation of the developed DAQ systems show that:

1. Experiments confirm the results obtained theoretically.
2. The existing microelectronic components, including FPGA chips and their programming technology, basically are suitable and can be used for implementation of gathered data specific pre-processing.
3. Potential of FPGA usage for specific data pre-processing is demonstrated by performance of the developed Fast DFT Processor (see Protocol of F-DFT Processor international evaluation results given in Attachment).
4. Attempts to use the existing microelectronic components for implementation of the front-end operations at DAQ show that this approach basically does not lead to sufficiently high results.
5. To fully gain from the research results obtained in the framework of this work, implementation of the front-end DAQ procedures, defined by the developed DAQ methods, should be based on developed new specific ASICs (Application Specific Integrated Circuits).

References

1. Bilinskis I., Sudars K. *Processing of signals sampled at sine-wave crossing instants.* Proceedings of the "2007 Workshop on Digital Alias-free Signal Processing" (WDASP'07), London, UK, 17 April 2007, pp. 45-50.
2. Sudars K., Ziemelis Z., *Expected performance of the sine-wave crossing data acquisition systems,* Proceedings of DASP Workshop, London, UK, 17 April 2007.

3. Bilinskis I., Sudars K. *Digital representation of analog signals by timed sequences of events*, "Electronics and Electrical Engineering", No. 3(83), March, 2008, Presented in International Conference "Electronics 2008", Kauna, Lithuania, 20-22 May 2008.
4. Bilinskis I., Sudars K. *Specifics of constant envelope digital signals*, "Electronics and Electrical Engineering", No. 4(84), April, 2008, Presented in International Conference "Electronics 2008", Kauna, Lithuania, 20-22 May 2008.
5. Artyukh Yu., Bilinskis I., Sudars K., Vedin V., *Multi-channel data acquisition from sensor systems.* Proceedings of the 10th International Conference "Digital Signal Processing and its Applications" (DSPA'2008), Moscow, Russia, 2008, Vol.X-1, pp.117-119.
6. Artyukh Y., Bilinskis I., Sudars K., European Patent Application No. EP2075912 A1, *Method for complexity-reduced digital filtering and parameter estimation of analog signals*, Assignee: Institute of Electronics and Computer Science of Latvia, European patent Bulletin, January 7, 2009.
7. Bilinskis I., Sudars K., Min M., Annus P., *Advantages and limitations of an approach to bio-impedance data acquisition and processing relying on fast low bit rate ADCs*, Baltic Electronic Conference BEC 2010, Tallinn, Estonia, 4.-6. October 2010.
8. Artyukh Y., Bilinskis I., Roga S., Sudars K., *Digital Representation of Analog Signals leading to their Energy-efficient Processing*, Green Information Technology (GREEN IT 2010), Singapore, 25.-26. October 2010.
9. Sudars K., *Data Acquisition Based on Nonuniform Sampling: Achievable Advantages and Involved Problems,* "Automatic Control and Computer Science" Magazine, Allerton Press, 2010, Vol. 44, No. 4, pp. 199-207.
10. Bilinskis I., Skageris A., Sudars K. *Method for fast and complexity-reduced asymmetric image compression*, "Electronics and Electrical Engineering", No. 4(110), May, 2011, Presented in International Conference "Electronics 2011", Kauna, Lithuania, 17-19 May 2011.
11. Artyukh Y., Bilinskis I., Rybakov A., Sudars K., Vedin V., Modular Multi-channel Data Acquisition Systems, "Automatic Control and Computer Sciences" Magazine, Allerton Press, 2008, Vol. 42, No. 3, pp. 113–119.
12. Artyukh Y., Bilinskis I., Boole E., Rybakov A., Vedin V., "Wideband RF signal digititizing for high purity spectral analysis," Proceedings of the 2005 International Workshop on Spectral Methods and Multirate Signal Processing, (SMMSP 2005) June 20-22, 2005, Riga, Latvia. pp. 123-128.
13. Artyukh Y., Bilinskis I., Greitans M., Mednieks I., Rybakov A., DASP-Lab System – a demonstrator of the new DSP technology. //Automatic Control and Computer Science, Allerton Press, 2000, No. 6, pp. 3-21.
14. Artyukh Y., Bilinskis I., Rybakov A., Stepin V., "Pseudo-randomization of multiplexer-based data acquisition from multiple signal sources," Proceeding of DASP Workshop, London, UK, 17 April 2007.
15. Bilinskis I., *Digital Alias-free Signal Processing*, Wiley, 2007, ISBN: 978-0-470-02738-7
16. Bilinskis I., Mikelsons A., *Randomized Signal Processing*, Prentice Hall, 1992, ISBN: 0-13-751074-8
17. Bilinskis I., Rybakov A., "Iterative spectrum analysis of nonuniformly undersampled wideband signals," Electronics and Electrical Engineering, 2006, vol. 4(68), pp. 5-8.
18. Blackledge J. M., Burge R. E., Barratt N. R., "Phase imaging by real-zero conversion and its applications to synthetic aperture radar", Journal of Physics D: Applied Physics, Volume 20, Issue 11, November 14, 1987
19. Borokhovych Y., Gustat H., "4-bit, 15 GS/s ADC in SiGe," NORCHIP 2008, 16-17 Nov. 2008, Tallinn, Estonia, pp. 268-271.
20. Candès E. J., Wakin M. B., "An Introduction to Compressive Sampling", IEEE Signals Processing Magazine, March 2008
21. Contadini F., "Oversampling with averaging to increase ADC resolution", Internet: http://www.embedded.com/design/223000385, 1 March 2010 15

22. Daria V. R., Saloma C., "High-accuracy Fourier transform interferometry, without oversampling, with a 1-bit analog-to-digital converter", Applied Optics, Volume 39, Issue 1, January 1, 2000, pp. 108-113
23. Eng F., *Non-Uniform Sampling in Statistical Signal Processing*, Dissertation at Linköpings Universitet, Sweden, 2007, ISBN: 978-91-85715-49-7
24. Internet: http://en.wikipedia.org/wiki/Data_acquisition
25. Internet:http://www.datatranslation.com/docs/whitepapers/benefits_simulataneous_daq.pdf
26. Internet: http://www.radiolocman.com/news/new.html?di=64921
27. Koch P., Prasad R., "The Universal Handset," IEEE Spectrum, vol. 46, pp. 32-37, April 2009.
28. Kumaresan R., Wang Y., "On representing signals using only timing information", The Journal of the Acoustical Society of America, Volume 110, Issue 5, November 2001, pp. 2421-2439
29. Lim M., Saloma C., "Direct signal recovery from threshold crossings", Physical Review E (Statistical Physics, Plasmas, Fluids, and Related Interdisciplinary Topics), Volume 58, Issue 5, November 1998, pp. 6759-6765
30. Logan Jr. B. F., "Information in the zero crossings of bandpass signals", Bell System Technical Journal, Volume 56, April 1977, pp. 487-510
31. Marvasti F. (editor), *Nonuniform Sampling Theory and Practice*, Kluwer Academic/Plenum Publishers, 2001, ISBN: 0-306-46445-4
32. Marvasti F., *A Unified Approach to Zero-crossings and Nonuniform Samplingof single and multidimensional signals and systems*, Illinois Institute of Technology, 1987, ISBN: 0-9618167-0-8
33. Min M., Bilinskis I., Artyukh Y., Annus P., Land R., Märtens O., "Nonuniform Sampling and Demodulation for Wideband Multi-frequency Bioimpedance Measurements," Proceedings of the Workshop on Digital Alias-free Signal Processing, London, 17 April 2007
34. Nait-Ali A., Cavaro-Menard C. (editors), *Compression of Biomedical Images and Signals,* Wiley, 2008, ISBN: 978-1-84821-028-8
35. Nazario M. A., Saloma C., "Signal recovery in sinusoid-crossing sampling by use of the minimum-negativity constrain", Applied Optics, Volume 37, Issue 14, May 10, 1998
36. Papenfus F., *Digital Signal Processing of Nonuniformly Sampled Signals*, Dissertation at Universität Rostock, Publisher: Shaker Verlag, 2008, ISBN: 978-3-8322-6895-4
37. Raghavendra C. S., Sivalingam K. M., Znati T. (editors), *Wireless Sensor Networks*, Kluwer Academic, 2004, ISBN-13: 978-1402078835
38. Rieke F., Warland D., R. R. van Steveninck, Bialek W., "Spikes: Exploring the Neural Code", MIT Press, Cambridge, MA, 1997
39. Sundstrom T., Alvandpour A., "A 2.5-GS/s 30-mW 4-bit Flash ADC in 90nm CMOS," NORCHIP 2008, 16-17 Nov. 2008, Tallinn, Estonia, pp. 264-267.
40. Zeevi Y. Y., Gavriely A., Shamai (Shitz) S., "Image representing by zero and sine-wave crossings", Journal of the Optical Society of America A, pp. 2045-2060, 1987