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A DSP Measuring System for Wind Monitoring

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Abstract- Renewable energies such as wind power are becoming more and more widespread, but the reliable operation of future energy supply structures, with high share of distributed generation and renewable energy sources, is an important and highly complex task, due to the output power fluctuation of the renewable energies that may cause excess variation of voltage or frequency of the grid. The availability of information about wind direction and speed is an important information in order to control, or to predict, the generated output power of large wind farm. In this paper, the planning, the realization and the characterization of a measuring instrument able to calculate wind direction and speed in real-time is presented. The system in future development will also be charged by typical tasks of a smart meter, so high computational power and fast data transmission will be required. For these reasons, a digital signal processor (DPS) based architecture and real-time communication protocols and library (Real Time Data Exchange - RTDX) were chosen. In the prototype implementation, the instrument is interfaced to a host PC that stores and shows the received data with a proper software program implemented in Labview environment. So, the main advantage is the possibility to monitor the time of flight in real time using a low expenditure of power and cheap instrument. Finally, some preliminary experimental results about characterization and precision of a prototype of the proposed instrument are shown.

I. Introduction

Wind is one of the most important source of renewable energy. Morphology of ground and environment has influence on direction and power of wind. For example, woods and mountains reduce the power of wind, as buildings of the biggest cities too. For this reason, wind power systems are situated in some areas only. The power of wind is especially strong where there are no obstacles, on plane surfaces, along the coasts and on the open sea. Paradoxically, today the wind energy is recognized as an alternative energy even if in a historical contest it has been with human life for more than oil and coal. In XX century we got electric energy from mechanical energy produced by wind power. The "Wind Farm" are formed by many wind system often situated on the open sea, where the wind is stronger. These are off-shore systems, in fact their environmental impact is minimal just because they are on the open sea. In the wind system, the production of electric energy is due to the speed of wind. Every time the speed of wind changes (measured in m/s), the aerogenerator will produce a different value of electric power. The fluctuation of output power may cause excess variation of voltage or frequency of the grid [1]-[5].

The availability of information about wind direction and speed is an important information in order to control, or to predict, the generated output power of large wind farm. Different approaches exist for wind measurement but the best results in terms of accuracy can be obtained by determining the wind speed from the measurement of the ultrasonic wave time of flight (ToF), which is defined as the time needed by an ultrasonic wave to travel from the transmitting to the receiving transducer [6].

In this paper, the planning, the realization and the characterization of a measuring instrument able to calculate wind direction and speed in real-time is presented. The system in future development will also be charged by typical tasks of a smart meter, so high computational power and fast data transmission are required. For these reasons, a digital signal processor (DSP) based architecture and real-time communication protocols and library (Real Time Data Exchange - RTDX) were chosen. The instrument is built up by adopting low-cost devices, and it is equipped with a modular communication device that offers high flexibility and a great upgrading possibility. Instruments for real-time measurements are characterized by an absolute time constraint for data input, processing, and output operation accomplishment, which must not be exceeded. For these reasons, the DSP represents a good choice for the implementation of an instrument that performs all the required signal processing. In fact, DSP, as it is well known, is particularly designed to accelerate the execution of numerically intensive calculations [7], [8]. Moreover, software is implemented only by adopting non-iterative algorithms; in this way, the operations are always the same and therefore the execution time is kept constant. After instrument description in terms of hardware architecture, measurement algorithm structure and firmware implementation, a set of tests for its performance verification and some preliminary experimental results for the characterization of the accuracy of the instrument prototype are reported.

II. Instrument Description

A simplified scheme of the proposed instrument is shown in Fig. 1. It adopts the C6713 32-bit floating-point DSP, based on the very-long-instruction-word architecture, installed on Developer Starter Kit (DSK) TMS320C6713 [9]. A special DSP internal core architecture based on two 32-kB independent memory banks (one for data storage and the other one for program storage) allows two memory accesses within one instruction cycle. Program memory is structured so that a total of eight instructions can be fetched every cycle, thus allowing a reduction of the execution time.

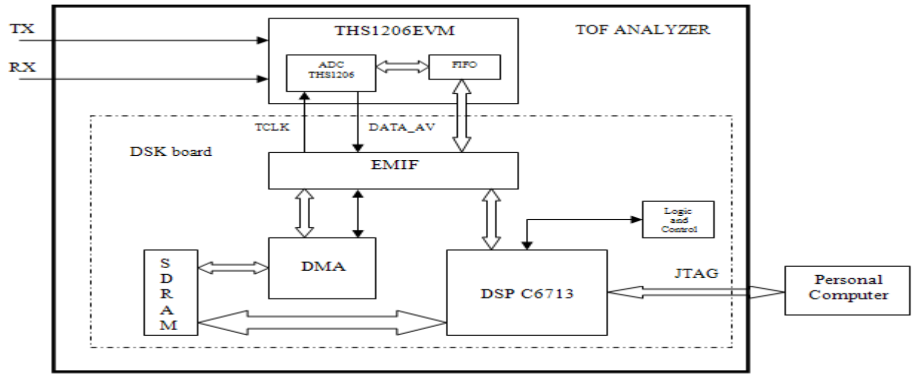

Figure 1. Hardware implementation of the low-cost TOF analyzer

The input signals are carried to the 12-bit analog-to-digital converter (THS1206) through two sample-and-hold circuits with a sampling rate of up to 6 MS/s. An efficient connection between the DSP and the data acquisition system, based on a first-in–first-out (FIFO) buffer that can store up to 16 conversion values, is utilized. The FIFO is directly connected to the DSP through a 32-bit external memory interface (EMIF). The instrument, for debugging reasons, can also work in emulation mode. In this working condition, a PC can take control of the instrument, sending I/O instructions and receiving measurement data. The Joint Test Action Group (JTAG) interface is used to perform the communication between a PC host and the target DSP and, for this purpose, two real-time data exchange channels are implemented on the JTAG interface [10].
These channels allow the data transfer without interfering with the target application. The local communication is made through the Universal Serial Bus port or the parallel port, using real-time data exchange
(RTDX) as communication protocol. In this protocol, a channel is first created between the target and the DSP, communicating the size of the transmission buffer, the path and the file where the data will be written. The channel is enabled during runtime; then, it is opened when the algorithm has ended. It is used for routine data transmission and, finally, it is closed when the data transmission has ended.

A. Firmware

The default use of the adopted DSK is in emulation mode, which requires the presence of a PC controller. To overcome this limit and to utilize the instrument in stand-alone mode, a second-level boot program was implemented. The software is loaded on the memory by FlashBurn program. After powering on the DSP, a boot program performs a self-test, checking the configuration of the board switches. Different switch positions imply the execution of different codes so that the DSP can work in stand-alone mode, running TOF algorithms, or in emulation mode, passing the control to PC. The emulation mode is used for updating or debugging purposes, whereas normal monitoring is performed in stand-alone mode. To allow the device to work in a stand-alone way, a program code must be copied to the head of the DSK internal flash memory. When the device starts, the program code stored in the flash memory is automatically copied to the DSP second level cache memory. This transfer operation is made while the CPU is in halt status by using the default timing of the ROM.
After that, the CPU exits the halt status and begins to execute the instructions from the initial address memory where the program is stored. In the development of software routines, the possibility of changing the sampling frequency and the number of acquired channels is accounted so allowing the increasing of the number of sensors in future enhancement of the instrument. We assumed that the clock source is DSP C6713.
After proper configuration of the main system and of the acquisition board, the program, using the asynchronous interface 16 bit EMIF, performs the acquisition of information coming from the 2 analog channel connected to the ultrasound sensors and the samples storing into the FIFO buffer. When FIFO buffer is full, data should be moved in different memory area to be processed. Obviously charging the CPU of the task of transferring the

samples form FIFO buffer to the memory, few computational resource are left for other tasks. To avoid this problem, a direct access to memory is granted to a specific device called EDMA (Enhanced Direct Memory Access), so that it can be able to store and organize the samples on its own without interferring with CPU processing.

B. Memory management

EDMA (Enhanced Direct Memory Access) is one of the most used peripheral by DSP. EDMA is autonomous in organizing the transfers of memory, so it lets CPU of DSP centre its resources on the elaboration of samples that come from the converter. There are two way to transfer data from a place to another one of the memory: one based on the CPU, the other on the EDMA. When we use EDMA, the only role of the CPU is the configuration of EDMA. In fact, the CPU waits for the end of the transfer and during this time it can apply itself to something else. EDMA allows to carry data in a transparent way to the second level cache where CPU have a direct access. During these transfers, the CPU has no type of overhead. From various angles EDMA is independent, but if CPU and EDMA try to enter the same location of the memory, only one of them takes priority (it would be better to leave the highest priority to the CPU). Every time DATA_AV will be active, EDMA "notices" this event and undertakes the task to move data from the address 0xA0020000 of the EMIF to the elaborating buffer. We adopted a double buffering technique in order to perform continuous analysis without loosing samples. When the first buffer (PING) is full, there is an interruption to CPU: the interrupt service routine, as first task, changes EDMA address so that next samples start to fill the other buffer (PONG). Then, it starts to elaborate the acquired samples that are in PING buffer. At this stage, it is important that the time required to complete the elaboration is lower than time required to fill the PONG buffer. In this way, the instrument obtains measurement results and consequently releases the PING buffer before another interrupt occurs. When PONG buffer is full, another interrupt occurs, the interrupt service routine, as first task, changes EDMA address so that next samples start to fill PING buffer. In this way, EDMA, in a hardware way, cyclically jump from a buffer to another one alternating filling and processing. With this approach the CPU is free during the data transfers and it is possible to work on line without loss of samples.

C. Time constraint

As it mentioned above, in order to work on line with no loss of samples, we must respect precise timing constraints: once an acquisition buffer is full, all the data processing for the calculation of measurement indexes has to be absolutely finished before filling of the other acquisition buffer. Otherwise we have an overflow, data are overwritten with a loss of information. To avoid the overflow, the sum of the time of execution of the ISR and the time of elaboration of the samples must be lower than time for the buffer filling. Filling time is equal to the ratio between the number of buffer samples and the sampling frequency. Moreover, the number of buffer samples should be square of 2 or 4 for FFT algorithm application. In the proposed instrument we adopted a sampling frequency of 400 kHz for each channel and a number of samples of 1024, obtaining an acquisition time window of 2.56 ms. The elaboration time of the algorithms described in the next section is 1.51 ms while the ISR elaboration time is 0.13 µs, so respecting time constraint.

III. Measurement approach

An usual configuration for measuring wind speed using ultrasonic transducers is to set up the transmitting and receiving transducers as shown in Fig.1. For this configuration, the TOF, Δt, needed for the ultrasonic wave to travel from transmitter to receiver is

$$\Delta t = \frac{L}{C + v \cdot \cos(\theta)} \qquad (1)$$

where C is the sound speed in the air, v is the air speed, θ angle to the wind direction and L is the distance between the transducers. [6], [11]. The transmitted signal is received after a time delay as depicted in Fig. 2. The output signal from TX is indicated as x_{ref}, and it consists of a sequence of codes. The signal acquired by RX is the sum of x_{ref}(with a delay Δt) and the environmental noise named η_{env}.

$$y(t_0 + \Delta t) = k \cdot x_{ref}(t_0) + \eta_{env} \qquad (1)$$

k depends of the environment features. The value of Δt can be calculated using the correlation function (CF) between x_{ref} and y and estimating the highest value of the argument:

$$\Delta t = \arg\ \max(corr(x_{ref}, y)) \qquad (2)$$

After the adoption of (1), it is possible to obtain wind speed. The correlation function can be calculated in time domain or in frequency domain. The author chose to adopt the latter approach in order to benefit of high performance of DSP in calculating the Fast Fourier Transform (FFT). The algorithm has been implemented with the C programming language and the Code Composer Studio v3.1 as development environment.

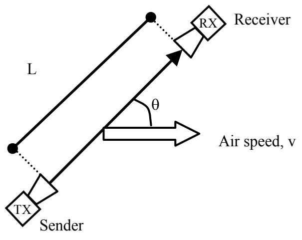

Figure 2. Wind-speed measurement transducer configuration.

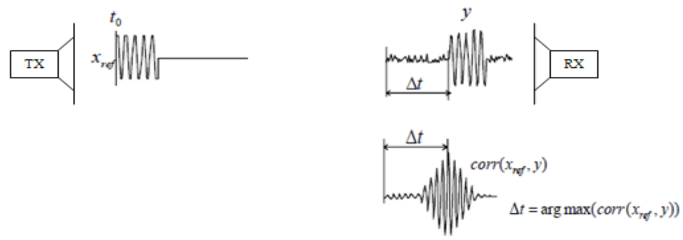

Figure 3. Evaluation of the Time Of Flight

A. Algorithm structure

A simplified block diagram of the framework of the implemented software that is able to implement all TOF measurement is shown in Fig.4. A LabVIEW virtual instrument has been used for implementing the communication with instrument for initialization and result presentation. The initializing stage allows changing the sampling frequency and the number of channels. The signals adopted by the two ultrasound sensors are at frequency of 40 kHz, and they are acquired with a sampling frequency that can range from 400kHz to 2MHz and stored in ping/pong buffer for on line measurement. Buffers are double so they can store samples coming from both the channel 1 and channel 2.When an acquisition buffer is full, the two signals are frequency transformed. The result of FFT of channel TX is submitted to a conjugation and then the two spectra are multiplied (see fig 4). Finally the inverse FFT is applied, so coming back to time domain with the function of the mutual correlation between the signals. Calculating the maximum value of the final vector, we obtain Δt, which corresponds to the TOF. Each measurement of TOF is stored into a vector, Vector TOF, of dimension 20 for averaging purpose. With the RTDX the Vector TOF is sent to the Host PC and a LabVIEW virtual instrument software has been used for displaying result data.

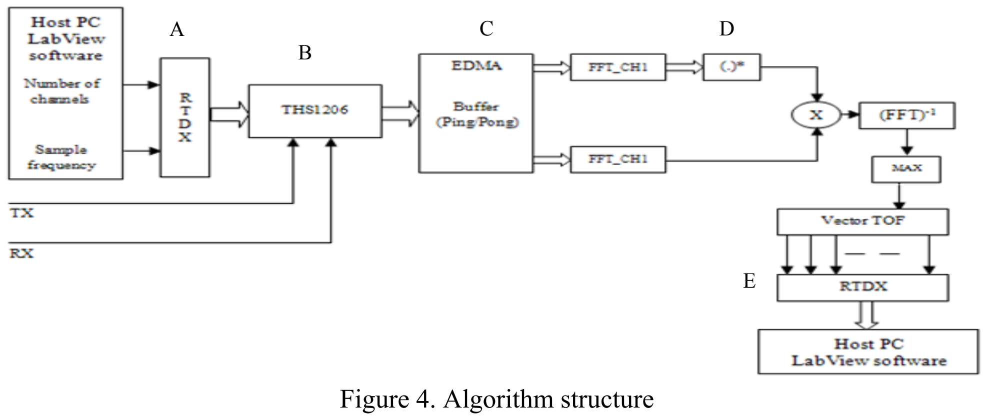

Figure 4. Algorithm structure

IV. Experimental Results

In this paper a first characterization of the instrument is shown. Two ultrasound sensors have been used (400ST160, 400SR160), the transmitter has been fed with a voltage burst train of 9 Vpp with length, for single burst, of 380 μs and with period of 2.50 ms. Five measurements have been made by varying he distance between the two sensors to find out best sensor position. The measured TOF versus distance is shown in the Table I, where averaging over 20 reading is adopted. The transfer function has an angular coefficient is equal to 0.946 and an offset of 0.1442.

TABLE I. CALIBRATION RESULTS

TOF [s]	Distance [m]
0.19	0.05
0.23	0.10
0.28	0.15
0.33	0.20
0.38	0.25

V. Conclusions

In this paper, we presented an instrument which is able to measure the time of flight “TOF” in real time; it is based on the platform DSK C6713 produced by Texas Instruments, where the DSP TMSC6713 and the acquisition board THS1206EVM are installed. The processing results are transmitted via USB port, using the communication libraries RTDX real time, to a host PC on which it is implemented a graphical interface realized in LabVIEW environment. In the final paper, a more comprehensive experimental tests for system characterization will be presented.

References

[1] Hu, P.; Karki, R.; Billinton, R.; ”Reliability evaluation of generating systems containing wind power and energy storage”, IET Generation, Transmission & Distribution, Vol.: 3, I.: 8, 2009, Pages: 783 - 791
[2] Methaprayoon, K.; Yingvivatanapong, C.; Wei-Jen Lee; Liao, J.R.; “An Integration of ANN Wind Power Estimation Into Unit Commitment Considering the Forecasting Uncertainty”; IEEE Transactions on Industry Applications, Nov.-dec. 2007;Vol.: 43, Issue:6 page(s): 1441 - 1448
[3] F.Vallee, J.Lobry, O.Deblecker, “Impact of the Wind Geographical Correlation Level for Reliability Studies”, IEEE Transactions on Power Systems,11/2007, Vol. 22, I.:4, pages: 2232 – 2239
[4] P.Meibom, C.Weber, R.Barth, H.Brand, “Operational costs induced by fluctuating wind power production in Germany and Scandinavia” IET Renewable Power Generation, Vol. 3, I.1, 2009, Pages:75 - 83
[5] K.Methaprayoon, C.Yingvivatanapong, J.R.;Wei-Jen Lee Liao, “n Integration of ANN Wind Power Estimation Into Unit Commitment Considering the Forecasting Uncertainty”, IEEE Transactions on Industry Applications, Vol. 43, I. 6, 2007, Pages: 1441 – 1448
[6] J.M.M.Villanueva, S.Y.C. Catunda, R.Tanscheit, “Maximum-Likelihood Data Fusion of Phase-Difference and Threshold-Detection Techniques for Wind-Speed Measurement”, IEEE Transactions on Instrumentation and Measurement, Vol. 58, I.: 7, 2009, Pages: 2189 - 2195
[7] G. Bucci and C. Landi, “A multi-DSP based instrument on a VXI C-size module for real-time measurements”, IEEE Trans. Instrum. Meas., vol. 49, no. 4, pp. 884-889, Aug. 2000.
[8] C. Gherasim, J. Van den Keybus, J. Driesen, R. Belmans, “DSP implementation of power measurements according to the IEEE trial-use standard 1459”, IEEE Trans. Instrum. Meas., vol.53, no. 4, pp 1086-1092, Aug. 2004.
[9] Texas Instruments, TMS320C6713 Floating-Point Digital Signal Processors, Jul. 2003.
[10] “Digital Signal Processing and Applications with C6713 and C6416 DSK” – Rulph Chassaing – Wiley Interscience – 2005.
[11] E. Kazanavičius and R Venteris, “Implementation of DSP algorithms for ultrasonic measurement applications”, ISSN 1392-2114 Ultragarsas, Nr.2(32), 1999.
[12] G. Andria, F.Attivissimo and A. Lanzolla, “Digital Measuring Techinques for High Accuracy Ultrasonic Sensor Application”, IEEE Trans. Instrum. Meas, pp. 1056-1061, May 1998.
[13] G. Andria, F. Attivissimo and N. Giaquinto, “Digital signal processing techniques for accurate ultrasonic sensor measurement”, Measurement 30, pp. 105-114, 2001.
[14] D. Marioli, C. Narduzzi, C. Offelli, D. Petri, E. Sardini and A. Taroni, “Time Of Flight Measurement for Ultrasonic Sensors”, IEEE Trans. Instrum. Meas., vol. 41, NO. 1, February 1992.
[15] G. Bucci, C. Landi, “Numerical Method for Transit Time Measurement in Ultrasonic Sensor Applications”, Proc. of the IMTC/96,, Brussel (B), pp. 405-410, Jun. 1996.
[16] M.P. Del Valle, J.A. Urbano Castèlan, Y.Matsumo and R.C. Mateos, “Low Cost Ultrasonic Anemometer”, ICEE 4th, pp. 213-216, Sep. 2007.