

Testbed for Wireless Adaptive Signal Processing Systems

György Orosz, László Sujbert, Gábor Péceli

Department of Measurement and Information Systems, Budapest University of Technology and Economics
 Magyar tudósok krt. 2., H-1521 Budapest, Hungary
 Phone: +36 1 463-2057, fax: +36 1 463-4112, E-mail: orosz@mit.bme.hu

Abstract –Due to the several attractive features of wireless devices they are more and more expansively used, but the deployment of wireless sensors in control applications is not spread yet because of the unreliable data transfer. This paper introduces a testbed which serves as a platform for investigations of wireless closed loop adaptive signal processing and control systems. The novelty of this testbed is that it makes possible the practical tests of algorithms in wireless environment. Since the traditional signal processing algorithms are not prepared for peculiar problems that emerge because of wireless data transfer (e.g. data loss), the exhaustive study of their behavior is necessary. This test system plays an important role in these investigations. Since the testbed offers also an application (the active noise control), algorithms can be studied in the same circumstances so ensuring the comparability of different structures.

Keywords – wireless sensor network, wireless control, real-time processing, active noise control, adaptive signal processing

I. INTRODUCTION

Nowadays the investigation of wireless sensor networks' (WSN) application is a relevant research topic, because WSNs are utilized with success in various fields [1]. The deployment of WSN has numerous advantages in contrast to wired systems. The most important of these are the easy installation and maintenance of sensors and actuators, the flexibility of arrangement, and the radio communication eliminates the costs of cabling. The intelligent sensors can themselves manage the network with marginal outer interaction. Nevertheless, because of the inherent and inevitable problems of radio communication (e.g. packet loss, uncertainty in data transfer time) the deployment of WSN in closed loop applications is not spread yet [2], since the

traditional algorithms aren't prepared for these faults. These facts make necessary the comprehensive study of the field of wireless control.

In this paper a testbed is introduced, which can be a common base of current investigations on the applicability of wireless sensor networks in closed loop multiple-input multiple-output (MIMO) control and digital signal processing systems. The novel feature of this testbed is that it offers a framework for the investigations to eliminate the disadvantageous effects of radio communication and for the study of characteristic problems that emerge in a wireless control system. The general configuration and typical problems of wireless control systems are shown in Fig. 1.

The MIMO plant required for testing is an acoustic plant, and the application that is utilized in the testbed is an active noise control (ANC) [3] system, the feedback part of which comprises a WSN for sensing the noise to be suppressed. Because of the wide variety of ANC algorithms this system is appropriate for test purposes and comparative analysis of results, since algorithms are tested under same conditions. The high quality of sensors and actuators is not required, because the main goal in this system is the investigation of the feasibility of control and signal processing systems in wireless environment, not the high noise suppression. Since the acoustic plant consists of loudspeakers and microphones that are easily available, this system is very simple to establish anywhere. ANC algorithms are mainly MIMO ones, so the system is arbitrary scalable by adding loudspeakers and sensors. The linearity of acoustic systems in wide dynamic range ensures the applicability of the widespread and elaborated adaptive algorithms that are developed for linear systems.

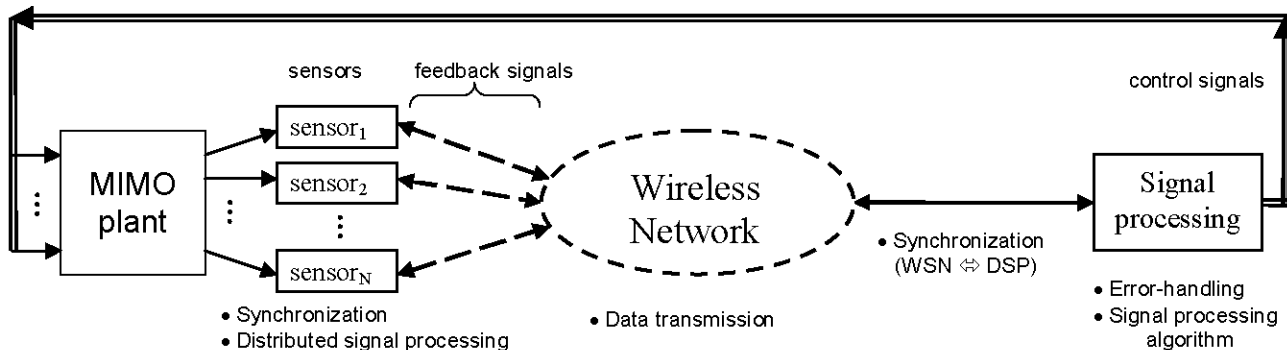


Figure 1. Typical configuration of wireless closed loop systems

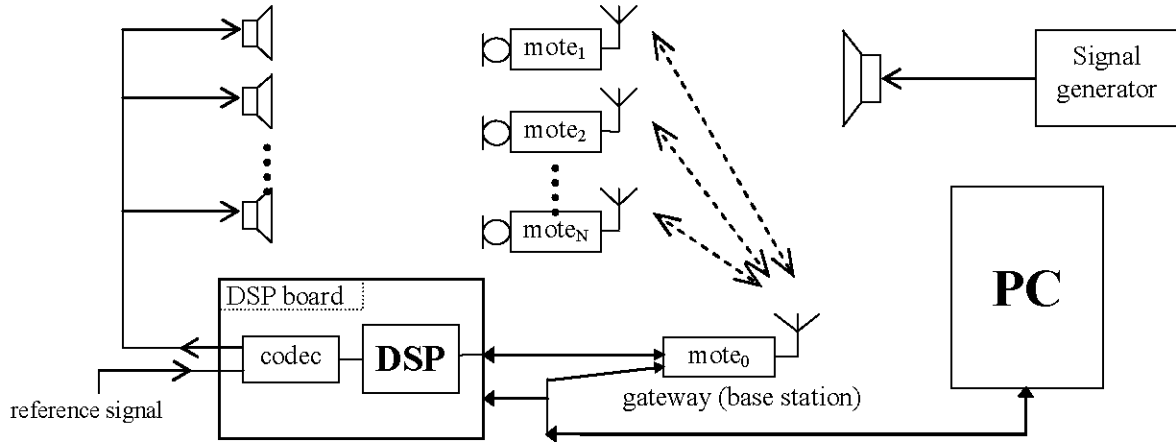


Figure 2. Block diagram of the system

This arrangement poses strict demands on the main components of the testbed (WSN and ANC algorithms) for ensuring the overall reliability of the system: the sensor network has to provide efficient and safe data transmission towards the signal processing algorithm, which in turn has to be adaptive for the anomalies in data transfer. Nevertheless the ensuring of stability criterion of ANC system requires the synchronization of the units in the system. Although the ANC systems are acoustic signal processing systems, general results can be adopted in non-acoustic systems, as well. In connection with the testbed we focus on signal processing aspects of the utilization of WSN.

This paper is organized as follows. In Section II. the hardware configuration of the testbed is described. The ANC algorithms and their suitability for test purposes are introduced in Section III. Major topics that can be studied in the field of wireless control with the testbed are summarized in Section IV. Results already achieved are presented in Section V.

II. SYSTEM DESCRIPTION

The block diagram of the testbed can be seen in Fig. 2. The system basically consists of two units. The main parts of signal processing algorithms are implemented on a DSP evaluation board of type ADSP-21364 EZ-KIT LITE [4], which includes an ADSP-21364 (SHARC) processor. The DSP is a 32 bit floating point one with a maximal clock frequency of 330 MHz and dual arithmetic units. The DSP is connected to an AD1835 codec that has two analog input and eight analog output channels, through which signals can be fed to the loudspeakers. Such number of output channels ensures the possibility of realization of extensive systems. The analog inputs can be used for collecting reference signals that is required for most ANC systems. Reference signals can be collected also by motes. The acoustic signal is sensed by the elements of WSN, which is built up of Berkeley micaz motes [5]. These motes are intelligent sensors that consist of an ATmega128 eight bit microcontroller with a clock frequency of 7.3728 MHz, a CC2420 2.4 GHz ZigBee

compatible radio transceiver and an MTS310 sensor board. Two AA size batteries provide power supply for each mote. The data transfer rate of the transceiver IC is 250 kilobit per second (kbps) including a preamble section, a header and a footer that are handled by hardware. The sensor board includes also a microphone with variable gain amplifier, the output signal of which is converted by a 10 bit analog to digital converter (ADC) of the microcontroller.

Most of motes ($mote_1 \dots mote_N$ in Fig. 2) are responsible for noise sensing. They transmit data towards the DSP. Data from the wireless network are forwarded to the DSP by a gateway mote ($mote_0$ in Fig. 2). To the software development for motes the TinyOS embedded operating system was used, but modifications should have been carried out in order to increase the code efficiency. Typical difficulties emerge in time critical sections, where accurate timing of tasks is inevitable. Such tasks are timing of the sampling the microphone's signal, and accurate detection of arrival time of radio messages, which is important task in synchronization.

The DSP and the gateway mote are connected via asynchronous serial port. The data rate of communication between the two units is 115.2 kbps. The programming of motes is carried out by an interface board of type MIB510 [5]. It serves as a power supply and RS232 line driver for gateway mote, as well.

The PC basically serves as developing and debugging tool for both platforms. However, it is suitable for logging and visualization of data sent by gateway or DSP over serial port.

An independent loudspeaker driven by a signal generator can be utilized to generate external sound. It can be used as an artificial noise source while testing the ANC system, or as general examination or excitation signal for test purpose.

III. INTRODUCTION TO ACTIVE NOISE CONTROL

A. Noise control systems

The general linearized model of ANC systems can be seen in Fig. 3. The purpose of ANC systems is to suppress mainly low frequency acoustic noises by means of destructive

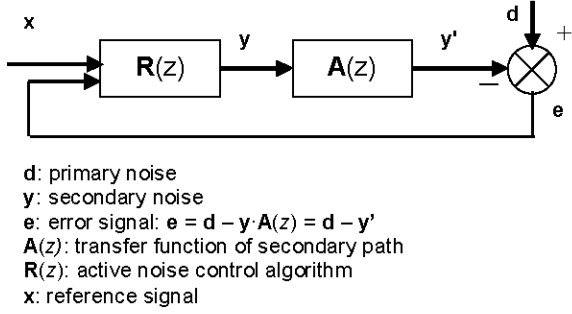


Figure 3. General structure of ANC systems

interference. The operation can be summarized as follows: according to the reference and the error signals (\mathbf{x} and \mathbf{e} respectively), the ANC algorithm $\mathbf{R}(z)$ produces such secondary noise \mathbf{y} which minimizes the power of the error signal \mathbf{e} . $\mathbf{R}(z)$ is implemented mainly on DSP because of its computational complexity. \mathbf{y} is radiated by loudspeakers and arrives to the microphones through the so called secondary acoustic path that is described by an $\mathbf{A}(z)$ matrix which consists of the transfer functions between each loudspeaker and each microphone. $\mathbf{A}(z)$ can be treated as a linear system. The error signal \mathbf{e} (i.e. remaining noise) is the result of the interference of the primary noise \mathbf{d} and secondary noise \mathbf{y}' , and it is sensed by microphones.

In the ANC algorithm a kind of inverse model of the secondary path $\mathbf{A}(z)$ is applied which is denoted by $\mathbf{W}(z)$. For ensuring the stability of the system, $\mathbf{W}(z)$ is often chosen as follows [11][12]:

$$\mathbf{W}(z) = \mathbf{A}(z)^{\#} \quad (1)$$

where $\#$ denotes the pseudo- (or Moore-Penrose) inverse. The secondary path $\mathbf{A}(z)$ should be identified in advance. The stability criterion of the system is:

$$-\pi/2 < \text{arc}(\lambda_l) < \pi/2 \quad (2)$$

$$\lambda_l = \lambda_l(\mathbf{A}(z)\mathbf{W}(z)); \quad l=1 \dots L \quad (3)$$

where l is the number of inputs. Eq. (2) and (3) means that all eigenvalues λ_l of the term $\mathbf{A}(z)\mathbf{W}(z)$ must have positive real part for each frequencies, where noise is present. For single channel case (only one microphone and one loudspeaker are used) it means, that the phase delay of $\mathbf{A}(z)$ must be known at least with the accuracy of 90° . $\mathbf{A}(z)$ involves the transfer function of the entire signal path between the output and input of the noise control algorithm (i.e. ADC, DAC, analog signal conditioner circuits, transfer functions of microphones and loudspeakers, and the transfer function of WSN's data routing is also included). The main goal in ANC systems (and in our case also in the WSN) to ensure the permanency of $\mathbf{A}(z)$ on each frequency, because so can $\mathbf{W}(z)$ be determined unambiguously and utilized during the entire operation time of the system.

B. ANC as test application

The reasons, why active noise control is suitable for test purposes, are summarized in the following.

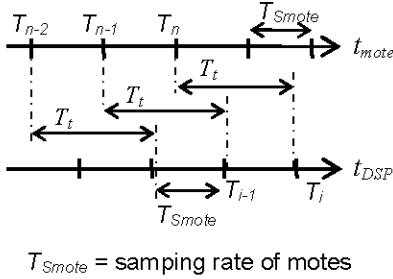
Problems that emerge in these ANC applications are characteristic of general closed loop systems. Furthermore, it is a real-time signal processing system that requires relatively fast data flow, so it can maximally exploit the resources offered by the wireless sensor network both in communication and computation aspects. The wide dynamic range that is characteristic of acoustic systems makes possible the testing the effect of quantization upon signal processing algorithms.

Another important reason is, that in the ANC systems generally more than one (about of the order of ten) microphones and loudspeakers are applied in order to reach appropriate noise suppression in large space. This is because noise cancellation is restricted to a limited surrounding of microphones, so the higher the number of the sensors the larger the space is where noise reduction can be achieved. The number of sensors that are utilized in the system gives the opportunity of investigation of MIMO wireless control systems that are more complicated, than their single-input single-output (SISO) counterparts because of the coupling between the channels.

The realization of ANC systems is possible by means of various signal processing algorithms. Since the applied algorithm doesn't influence the hardware structure, this testbed makes easier and faster the process of investigations of algorithms in practice.

The ANC systems are extremely sensitive for the change of the transfer function of the secondary path. The reason is, that the permanency and accurate knowledge of $\mathbf{A}(z)$ is required for the stability—as shown by Eq. (2)—, but for the adequate transient properties more accurate knowledge of the transfer function is necessary. Since the secondary path also includes the WSN, it has really key importance to ensure the permanency in the collection, processing and transmission of data in the sensor network. This fact makes possible and also requires to work out and to test sophisticated data collecting and transmitting methods. Another purpose of identified transfer function of secondary path—above the utilization in noise control algorithm—is, that it can be evaluated offline, and it can be used to characterize the secondary path, and so the behavior of WSN, as well. For example the difference in delay of certain variants of data routing algorithms can be measured, or according to more consecutive identification results the persistency of transfer function can be determined. It is useful during the test phase of data transmission algorithms, since it reveals the fluctuations in the transfer time of the data transmission, which causes changes in the different transfer functions.

Although the main purpose of this system is not the noise suppression, the measurement results of certain algorithms in noise cancellation either in steady state or transient phase can serve as basics for comparison of algorithms in the same circumstances.



T_{Smote} = sampling rate of motes

Figure 4.a. Unsynchronized system

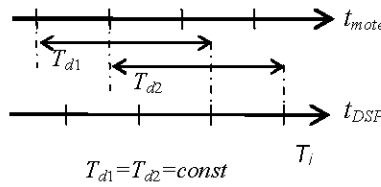


Figure 4.b. Sampling synchronization

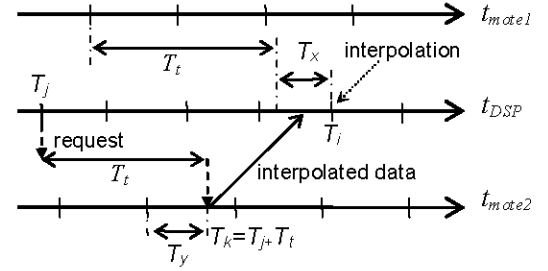


Figure 4.c. Synchronization with interpolation

IV. RESEARCH TOPICS

Basic fields of investigations related to the different parts of the system can be seen in Fig. 1.

A. Synchronization

A crucial task in the system is the synchronization, because there are many autonomous subsystems (each mote and the DSP). Synchronization has key importance because of the strict stability condition referring to the knowledge of the feedback path. If the sampling on motes and processing of the sampled data on the DSP occurred independently of each other, the delay between the sampling and the processing of the signal would vary at least in a one sampling period interval, as it can be seen in Fig. 4.a. In the figures the sampling and processing times are signed with vertical lines on the time axes, and it is assumed that the T_t data transmission time is constant and the DSP processes the most currently received data. In T_i the delay between the processing and sampling (T_i and T_n respectively) is T_t , since the data arrives directly before T_i . However, in the processing time T_{i-1} the delay is $(T_t + T_{Smote})$, since the data sampled in T_{n-1} arrives slightly after T_{i-1} so the DSP processes the previous data that has arrived approximately with T_{Smote} time before T_{i-1} . This example shows the two extreme value of delay, but it can change anywhere within $[T_t \dots T_t + T_{Smote}]$ interval. Since $\mathbf{A}(z)$ includes this delay, it also changes continuously during the operation so it differs from its identified value, hence $\mathbf{W}(z)$ is no more optimal. Moreover it can occur that $\mathbf{W}(z)$ doesn't satisfy even the stability condition. This is the simplest case, when the DSP processes the most current data, but more delay can be introduced in other processing methods. All these facts make this testbed appropriate for testing synchronization algorithms, since for the sufficient and reliable operation of ANC adequate synchronization is required.

For the synchronization there are basically two approaches. Units can be synchronized by matching the time instants of sampling and signal processing physically to each other. It means that these processes coincide with each other, or take place with constant time offset. It can be seen in Fig. 4.b. that between the sampling and processing there is a constant difference, which results in a

constant phase shift in $\mathbf{A}(z)$. It means that $\mathbf{W}(z)$ can be determined so that it suffices Eq. (2).

Another approach for the synchronization is the interpolation, when the effect of unaligned sampling and processing times are compensated by computation. By means of interpolation the exact value of the signal in the processing time instants can be estimated, so eliminating the changing delay. This method is useful if the sampling frequencies can not be tuned, e.g. if an independent device provides the data.

The interpolation can be realized in two different ways, as the Fig. 4.c. shows. In the case of the mote₁ and DSP, the mote sends samples continuously to the DSP that interpolates the value in the processing time T_i by measuring T_x . In the other method the interpolation is carried out on the motes. This is the case in Fig. 4.c., when the DSP sends request message to the mote₂. Since they work asynchronously the mote should estimate the signal value when the message arrives, and sends back this interpolated (or resampled) value which is processed by the DSP.

In the first case the interpolation demands high computation load on the DSP, since the data from every motes should be handled individually. In the second case the computation is distributed in the network, but the transmission of request messages means extra load for the network. The method for interpolation in both case can be chosen according to trade-offs between computational complexity and precision. Algorithms that ensure lower distortion require generally more processor capacity.

Mixed synchronization algorithms can also be applied, when sampling on motes is physically synchronized, but the DSP uses interpolation for fitting the data from WSN to its processing time.

Because of the various potential synchronization methods that can be used in the system, it offers numerous ways for investigations in the field of synchronization.

B. Signal transmission and processing

Although the sampling frequency required by the ANC is not really high (about 1-2 kHz), the transmission of signals of even 3 or 4 motes can be unfeasible because of the bandwidth limit of the wireless network. Hence it is important to ensure efficient data routing. It can imply the

investigation and development of data transmission methods that suits for the nature of multiple access wireless networks with limited bandwidth and energy resources, and takes into account, that the data transmitting network is the part of a control loop (e.g. TOD protocol [6]). In connection with data transmission, data compression algorithms and their integration into the control systems can be considered and investigated.

A basic problem in the system is the handling of faults caused by the features of wireless data collection, i.e. the data loss or irregular transfer time. Since the traditional algorithms (e.g. filters, controllers) basically work on equidistantly sampled data, they aren't able in their original form to handle the anomalies that are present in a wireless system (e.g. missing of data in a sampling time instant). There are already results that aim to solve these problems (e.g. [7]). This system makes possible to test these new algorithms developed especially for wireless data collecting environment. Since in ANC systems a lot of adaptive algorithms can be utilized (adaptive transversal or recursive filters, like LMS, FXLMS, RLMS or observer based methods [3][8][9]), many algorithms can be investigated in wireless environment.

The computational capacity of motes makes possible the implementation of algorithms not only on the DSP, but on the nodes of the network, as well. This makes possible the investigation of distributed signal processing in this testbed. The extension of signal processing algorithms on sensor network has more advantages. The preprocessing of signal can reduce the amount of data to be transmitted via the network, if only some signal parameters are necessary for the control algorithm. The preprocessing of signals can also reduce the effect of data loss, since primary data are processed on the motes. Since the preprocessing of signals is not accustomed in the case of traditional sensors, signal processing algorithms can be investigated in terms of cooperation with intelligent sensors, as well.

V. RESULTS

In the testbed we have already implemented some ANC systems with various structures. In the realized systems the sampling frequency on motes was 1.8 kHz and 2 kHz on the DSP, and the noise suppression was about the value that is characteristic of ANC systems, to which sensors are attached by wire. Also a PLL like synchronization algorithm was worked out that ensures the synchronous sampling in the WSN.

The noise sensing in the systems was carried out in two ways. An obvious method for noise sensing is the simple sampling of the output of noise sensing microphone on the motes, and the transmission of samples to the DSP. In this case the fitting of sampling frequency of motes to that of the DSP was achieved by linear interpolation. Since the sampling process on motes is synchronized, the motes in WSN can be handled uniformly on the DSP side in the interpolation algorithm. The utilized ANC algorithms are

FXLMS [8] and resonator based ones [9][10] on which experiments are carried out. During the experiments reference signal collection by WSN was realized, as well.

In these systems the bottleneck is the bandwidth of the wireless network, which limits the number of motes at a given sampling frequency. In order to increase the effective throughput of the network, a transformed domain data transfer is worked out, which means: sensors pass for the DSP only the Fourier-coefficients of periodic signals that are computed by the motes in place by an observer based Fourier-analyzer (FA) structure, so the signal processing was extended on WSN. Since these coefficients change slower than the signal itself, lower transmission rate is allowed, so the limitation of bandwidth is less relevant in terms of number of noise sensing motes. This makes possible the expansion the number of motes without decreasing the sampling frequency. The structure of the system is shown in Fig 5.

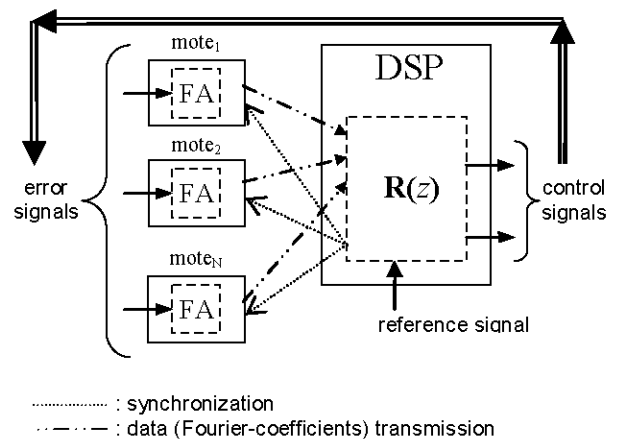


Figure 5. ANC system with distributed Fourier-analyzer

This kind of error signal sensing is utilized in a resonator based ANC system, where the coefficients can be directly used in the algorithm. In this case beyond the synchronous sampling in WSN also another synchronization problem emerges: the consistency of the base functions has to be ensured in the whole system (on each mote and on the DSP), since the phase value of the Fourier-coefficients can be interpreted only by maintaining a common reference, the base functions in the system. It is solved by continuous transmission of the phase and frequency of the base functions (sine and cosine functions), but synchronization of sampling on motes is a necessary condition.

Since the synchronization of sampling on motes has key importance in each system it is presented in detail. The synchronization method requires a reference mote (e.g. mote₁ in Fig. 6), the sampling rate of which the other motes are synchronized to. It sends synchronization messages periodically at the beginning of every n sampling period. If the mote to be synchronized gets the messages in the same part in the sampling period ($T_{x1} = T_{x2} = T_x = const$ in Fig. 6) then the time difference between

VI. CONCLUSIONS

In this paper a system is introduced that is suitable for testing the utilization's properties of WSN in fast adaptive digital signal processing and closed loop control algorithms. Active noise control is found to be an excellent application for test purposes. ANC systems are linear MIMO structures, and the plant is present in every laboratory. The sensors (the microphones) are available on the sensor boards of the micaz motes and the actuators (e.g. active loudspeakers) are cheap and they can be bought easily. The testbed ensures a large degree of freedom during the design, which leads to systems that have the same functionality but with different structures, as the presented experimental systems show. These competitive structures are appropriate for the analysis of the problems in wireless control from several aspects.

The aim in the future work is to develop and implement such adaptive signal processing algorithms that are optimized for working in wireless environment.

ACKNOWLEDGMENT

This work was partly supported by the Hungarian Fund for Scientific Research under Grant Nr. OTKA TS 049743.

REFERENCES

- [1] Akyildiz, I. F., W. Su, Y. Sankarasubramanian, and E. Cayirci, „Wireless sensor networks: A survey,” *Comput. Netw.*, vol. 38, no. 4, pp. 393-422, 2002
- [2] Mathiesen, M., G. Thonet, N. Aakvaag, „Wireless ad-hoc networks for industrial automation: current trends and future prospects,” in *Proceedings of the IFAC World Congress*, Prague, Czech Republic, July 4-8, 2005
- [3] Kuo, S. M., D. R. Morgan, „Active Noise Control: A Tutorial Review,” in *Proceedings of the IEEE*, vol. 87, No. 6., pp. 943-973, June, 1999.
- [4] www.analog.com/UploadedFiles/Associated_Docs/50400613ADSP_21364_EZ_KIT_Lite_Manual_Rev_2.0.pdf
- [5] *MPR-MIB User's Manual*, www.xbow.com/Support/Support_pdf_files/MPR-MIB_Series_Users_Manual.pdf
- [6] Gregory C. Walsh, Hong Ye, and Linda G. Bushnell, „Stability Analysis of Networked Control Systems,” in *IEEE Trans. on Control Systems Technology*, Vol. 10, No. 3, May 2002
- [7] B. Sinopoli et. al., “Kalman Filtering with Intermittent Observations,” in *Proc. IEEE Conf. on Decision and Control*, pp. 701-708, Dec. 2003.
- [8] Elliott, S. J., P. A. Nelson, „Active noise control,” in *IEEE Signal Processing Magazine*, Vol. 10, No. 4, October, 1993, pp. 12-35.
- [9] Sujbert, L., G. Péceli, „Periodic noise cancellation using resonator based controller,” in *1997 Int. Symp. on Active Control of Sound and Vibration, ACTIVE '97*, pp. 905-916, Budapest, Hungary,
- [10] L. Sujbert, K. Molnar, Gy. Orosz and L. Lajko, „Wireless Sensing for Active Noise Control,” on *IMTC -Instrumentation and Measurement Technology Conference* -, Sorrento, Italy, April 2006
- [11] L. Sujbert, "A new filtered LMS algorithm for active noise control", in *Proceedings of the Active '99 - The International EAA Symposium on Active Control of Sound and Vibration*, Dec. 1999, Fort Lauderdale, Florida, USA, pp. 1101-1110.
- [12] Elliott, S. J., P. A. Nelson, “The Behavior of a multiple Channel Active Control System”, in *IEEE Trans. on Signal Processing*, Vol. 40, No. 5, May 1992

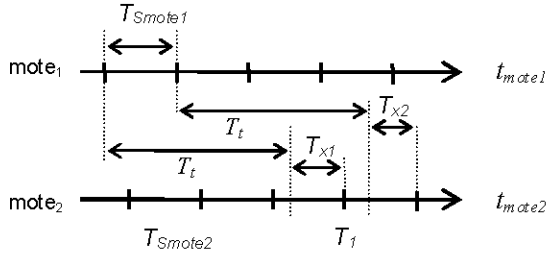


Figure 6. Timing diagram

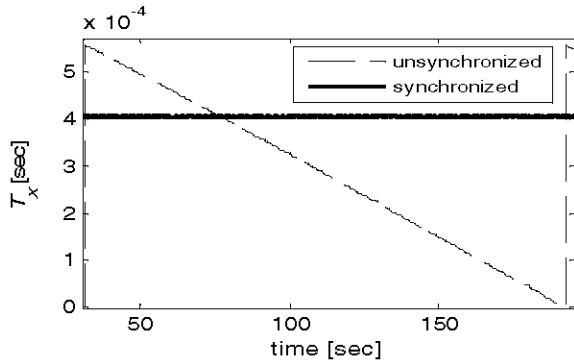


Figure 7. Synchronization measurement results

each sampling time instant on the different motes is constant ($T_t + T_x$), so they are synchronized in sampling.

In unsynchronized case, the frequency error of the clock generator of the devices causes the sampling frequencies to be slightly different ($T_{Smote1} \neq T_{Smote2}$). This results in the continuous changing of the relative position of the sampling points. Hence also the reception time of synchronization messages on mote2 (T_x) changes, as shown in Fig. 7 (dashed line). Since T_x is measured relatively to the next sampling time instant it is in the interval $[0 \dots T_{Smote2}]$.

By changing the sampling frequency the T_x can be held on constant level, as shown in Fig. 7 (solid line). By decreasing T_{Smote2} in T_t the next sampling point gets closer to the reception point so T_x is decreased, while increasing of T_x requires the opposite change of T_{Smote2} .

In the simple data collection system the maximal applicable number of motes was two, while in the case, when motes delivered the Fourier-coefficients, the experimental system consisted of eight motes and four loudspeakers. This proves the efficiency of the latter, distributed signal processing based method. In this case the number of sensors could be extended according to estimations up to some dozens. The size of the network is limited by the properties of noise to be suppressed (e.g. the rate of change of the frequency or amplitude of noise), and by the resources of the DSP. In the latter case the computation capacity means no restriction, since in the case of eight motes and four loudspeakers only about five percent of the available time was used for computation of the ANC algorithm on the DSP, but the storing of great deal of transfer functions (between each sensor and each loudspeaker) has large memory demand.