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CONTROL AND IDENTIFICATION ALGORITHMS FOR ACOUSTIC NOISE REDUCTION

SUMMARY

The aim of the paper is to present main aspects of a research project on active noise reduction being performed at the Computer Control System Group, which is a part of the Institute of Automatic Control, Silesian University of Technology, Gliwice, Poland. In particular, the following problems are dealt with: higher order spectra in off-line and on-line identification of electro-acoustic plants for active noise reduction systems; creating distributed local zones of quiet in a reverberant enclosure using two structures: based on spatially distributed transducers controlled by a central unit or implemented as decentralized systems; development and analysis of optimal and adaptive feedback systems for generating zones of quiet at desired locations; enhancement of active noise reduction efficiency using nonuniform signal sampling and oversampling techniques; design and implementation of a portable multi-channel active noise control platform.

Keywords*: adaptive control, identification, active noise reduction, higher order statistics, signal sampling and reconstruction*

ALGORYTMY STEROWANIA I IDENTYFIKACJI W AKTYWNEJ REDUKCJI HAŁASU

W pracy zaprezentowano tematykę projektu badawczego poświęconego aktywnej redukcji hałasu, prowadzonego w Zakładzie Komputerowych Systemów Sterowania, w Instytucie Automatyki Politechniki Śląskiej w Gliwicach. W szczególności omówiono następujące zagadnienia: wykorzystanie widm wyższych rzędów do identyfikacji w trybie off-line i on-line obiektów elektroakustycznych w układach aktywnej redukcji hałasu; tworzenie rozproszonych lokalnych stref ciszy w pomieszczeniach z pogłosem przy wykorzystaniu układów sterowania opartych na przestrzennie rozłożonych elementach wykonawczych sterowanych przez jednostkę centralną lub sterowanych w sposób zdecentralizowany; rozwój i analiza optymalnych i adaptacyjnych układów sterowania dla tworzenia stref ciszy w zadanych położeniach; zwiększenie efektywności układów aktywnej redukcji hałasu poprzez wykorzystanie technik nierównomiernego próbkowania oraz nadpróbkowania sygnałów, zaprojektowanie i wykonanie przenośnego wielokanałowego urządzenia PANC przeznaczonego do tworzenia lokalnych przestrzennych stref ciszy.

Słowa kluczowe*: sterowanie adaptacyjne, identyfikacja aktywna redukcja hałasu, statystyki wyższych rzędów, próbkowanie i rekonstrukcja sygnałów*

1. INTRODUCTION

The first system for active noise reduction was patented by Lueg in 1936 [19]. However, limitations of analogue technology caused that the idea had to wait nearly fifty years to put into effect. Development of digital signal processing (DSP) based on microprocessors together with development of new algorithms in control theory and process identification have made active noise control (ANC) a truly practical tool. Since 1990s the Computer Controlled Systems Group at the Institute of Automatic Control, Silesian University of Technology in Gliwice, has joined the research on implementation of modern control techniques to noise reduction in real-world systems.

To design ANC system it is necessary to identify electro-acoustic plants (so-called secondary and acoustic feedback paths of ANC system). But for better functioning of the system, the models can be updated during its operation, too [12]. Thus, acoustic noise reduction that can be achieved depends greatly on the accuracy of obtained models as well as on performed control algorithms. To perform the research in this area, in the years 1994–2007 five laboratory stands of ANC systems have been set-up at the Institute of Automatic Control thanks to funds from the Polish State Committee of Scientific Research and Ministry of Science and Higher Education. These are: active headphones, ANC in an acoustic duct, stand for active control of acoustic noise in a headrest system, and two stands to create local spatial zones of quiet in an enclosure. The laboratory stands consist of two parts: acoustic plant and electronic devices to implement ANC algorithms.

The main goal of this paper is to present the recent results in development of identification and control algorithms for ANC systems concerning the following topics:

– taking advantage of higher order spectra (HOS) in off-line and on-line identification of electro-acoustic plants,
– creating local zones of quiet in a reverberant enclosure,
– development and analysis of optimal and adaptive systems for generating zones of quiet at desired locations,
– enhancement of ANC system efficiency using nonuniform signal sampling and oversampling techniques,
– implementation of single and multi-channel control algorithms on programmable microcontrollers.

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2. DEVELOPMENT OF NEW ALGORITHMS FOR ELECTRO-ACOUSTIC PLANTS IDENTIFICATION

Adaptive ANC systems are parameterized with models of secondary and feedback paths [12]. If these paths are time-varying, the models have to be updated during ANC system operation. Solutions proposed for *on-line* identification problem have to deal with several difficulties:

- (inherent) feedforward and feedback loops, what implies that the input and output of the identified path are correlated, and both signals are correlated with disturbance (attenuated noise); moreover, if the system works well, the identified path is not excited sufficiently, therefore the external excitation has to be introduced into the system and added to the control signal;
- low signal-to-noise ratio, because the variance of the external excitation has to be very small to avoid significant deterioration of the noise attenuation.

In this case classical identification algorithms often give biased and inconsistent estimates. Therefore, we propose to perform the indirect method, in which at first frequency response models are estimated using HOS or higher- and second-order spectra together, supported by signal averaging. Next, rational transfer functions (RTF) are calculated to approximate the obtained frequency responses. These RTFs are used to design suitable filters for ANC systems.

2.1. Proposed solutions

The main motivation for using higher-order spectra instead of second-order ones in process identification is that they are identically zero for some processes, including Gaussian processes. Thus, if the disturbance is Gaussian, it theoretically does not influence the identification results, and in practice this influence is significantly reduced [26, 39]. The proposed approach requires a special non-Gaussian excitation signal. In our experiments, the excitation sequence is repeated several times and the measured data are averaged. Averaging enhances signal-to-noise ratio without deteriorating noise attenuation, because the variance of averaged disturbances is reduced proportionally to the number of repetitions. Another useful property of averaging reveals when identification based on HOS is applied. According to the *Central Limit Theorem*, the averaged disturbance asymptotically approaches the Gaussian distribution [36]. So, the averaging procedure lets us successfully use HOS-based methods, even if the disturbance does not satisfy the Gaussianity assumption [10].

The proposed method is based on direct estimation of integrated bispectrum (IB). Then frequency response is calculated as the ratio of input-output cross-IB to IB of the input – similarly to the classical spectral analysis (the difference lays in the type of used spectrum). Regardless of loops existing in the system, the method allows to obtain strongly consistent estimates [39]. Also, a modification of this method has been proposed, in which the estimator (so called mixed-order estimator) is calculated as a linear combination of both: second-order and HOS-based estimators (in the problem considered it is IB), with weights dependent on the coherence function [9, 11]. The idea of this approach is as follows: for strongly-disturbed frequencies (low coherence) the estimator is based rather on the higher-order estimates, otherwise second-order estimates are weighted more. It helps to reduce the variance of obtained models especially if the disturbance is concentrated only in some frequency ranges.

After the frequency response is calculated, the next step is to retrieve the RTF from it. This parametric model can be obtained via, e.g. the least squares approximation of frequency response by a fixed model structure [39] or weighted least squares, as it was proposed in [14]. If the model structure is unknown it can be estimated using trial and error method based on testing different structures and selecting the model that has the lowest information criterion. To omit an "exhaustive search", which means that models for all structures within the assumed range ought to be estimated, the appropriate strategy of searches has been proposed [13]. The travel path through the space of structure indices is driven by a set of heuristic rules that attempt to reach a (sub)optimal point (the best structure according to the assumed criterion) in a small number of steps.

Extension of the proposed techniques for the multi-channel case is in process. We focus on solving a MIMO system identification problem by decomposing it into independent SISO problems using orthogonal excitation signals.

2.2. Exemplary experiment results

An example of *on-line* secondary path identification in a feedforward ANC system is shown in the Figure 1. The adaptive ANC system of interest creates a local spatial zone of quiet surrounding an error microphone in an enclosure. During the experiment a disturbance (real pump noise) was concentrated in the frequency band from 70 to 110 Hz. During the identification procedure a parametric model (FIR filter with one hundred coefficients) was found. Details of the experiment are described in [9].

This example demonstrates that the classical second-order estimates give greater errors than the higher-order and mixed-order ones. However, outside the mostly disturbed frequency band (70–110 Hz), the second-order estimates are a little better than higher-order estimates. This suggests to eliminate this defect of higher-order estimator by using the mixed-order one that is a linear combination of both estimators: higher- and second order one. The first is weighted more within the range where the noise occurs, while the second outside.

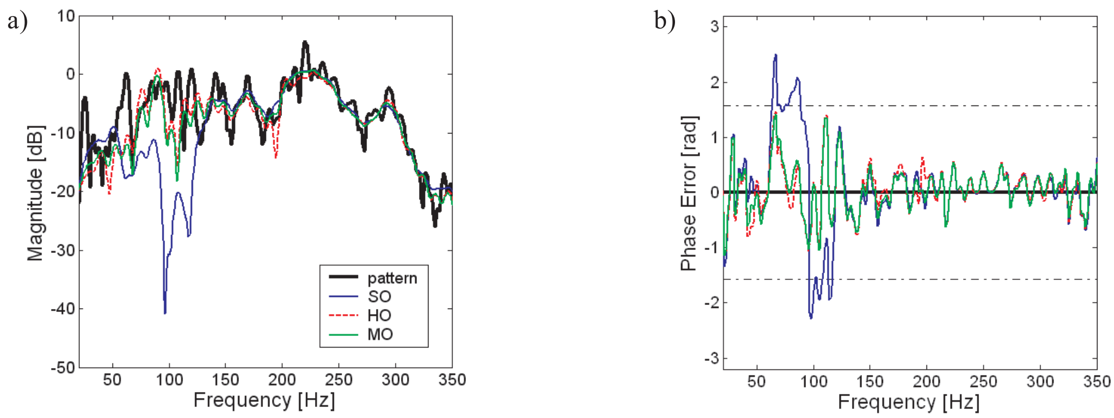

Fig. 1. Frequency response magnitude (a) and frequency response phase error (b) of second-order (solid blue line), higher-order (dashed red line) and mixed-order (solid green line) estimates

3. DEVELOPMENT OF ALGORITHMS FOR GENERATING LOCAL ZONES OF QUIET

The global zone of quiet in an enclosure is difficult to obtain and not always necessary. Thus, often it is sufficient to create only local zones of quiet. Provided the ANC system is multi-channel, zones of quiet can be created in multiple locations. Depending on the spatial configuration of error microphones and control loudspeakers in an enclosure, one bigger zone or many smaller zones distributed over the enclosure can be obtained.

Generally, systems creating distributed zones of quiet can be designed in two ways:

1) they can use spatially distributed transducers – loudspeakers and microphones,
2) they can be implemented as decentralized systems – a set of single-channel systems, with control algorithms implemented in different microprocessors.

At first, multi-channel systems were diagonal – built of many single-channel controllers. In such implementation, the diagonal controllers tend to ‘fight’ each other [25], what can be controlled using supervising weighting algorithm [6]. However, the inter-channel cross-coupling can appear also in multi-channel ANC systems using full controllers (taking into account all channels: diagonal and off-diagonal) [17]. This problem can be minimized if the distributed zones of quiet are to be created. It is convenient then to apply a decentralized multi-channel ANC system. Though there are also restrictions that should be taken into account. Each control loudspeaker has to minimize only one error signal. This error signal should be obtained from the microphone closest to this secondary source and far away from other secondary sources [8] to avoid inter-channel cross-coupling effects. Such decentralized system can be of feedforward type [8] or internal model control (IMC) structure [18].

To explore possibilities of creation of distributed local zones of quiet, the experiments were performed in a reverberant laboratory enclosure [23]. Two kinds of control structures were examined: the feedforward control structure, parameterized as full as well as diagonal, and the IMC structure implemented as diagonal. The systems presented in [8, 18] were applied for attenuation of periodic disturbance while the ANC system presented in [23] was designed to attenuate both, periodic and random, disturbances.

The ANC systems have been configured for low-frequency noise attenuation. The sampling frequency has been chosen as 500 Hz. The feedforward system used one reference microphone, three control loudspeakers and three error microphones. The IMC system was created from three single-channel IMC systems, with control loudspeakers and error microphones set in the same locations as in the feedforward system.

Dynamics of the electro-acoustic plant controlled under ANC system described above is very complicated. The magnitudes of the secondary path channels are very rugged [20, 22], characterized by huge differences (e.g. 30 dB) for close frequencies (e.g. 102 and 106 Hz). This implies that it is difficult to attenuate some frequencies using single-channel ANC system. It is also difficult to choose the step size μ of the FX-LMS algorithm [21, 22].

The ANC system was tested for broadband random noise and the noise composed of single sine with frequency 62 Hz and narrowband noise with dominating frequencies around 95 Hz. The results of experiments are shown in Figures 2 and 3. The broadband disturbance is very difficult to attenuate, especially using the IMC structure. The attenuation of only a few dB is obtained – the best for the full feedforward controller, the worst for the IMC structure. The power spectrum densities (PSD) of the disturbance and error signals picked up by one of the error microphones are shown in Figure 2. Although the attenuation is low, the annoying peak frequencies (of 26, 65 and 100 Hz) are attenuated to – 45 dB level.

The attenuation of the narrowband noise is higher. The PSDs of the disturbance and error signals picked up by one of error microphones are shown in Figure 3. The single sine disturbance is attenuated to the background level in microphones M_1 and M_2. Lower attenuation is obtained in microphone M_3.

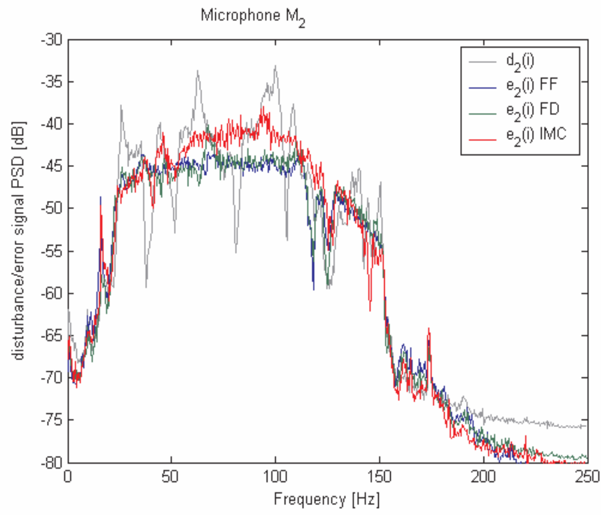

Fig. 2. PSD of disturbance and error signal picked up by error microphone M_2 in the ANC system disturbed by the broadband noise for controllers: feedforward full (FF), feedforward diagonal (FD) and IMC

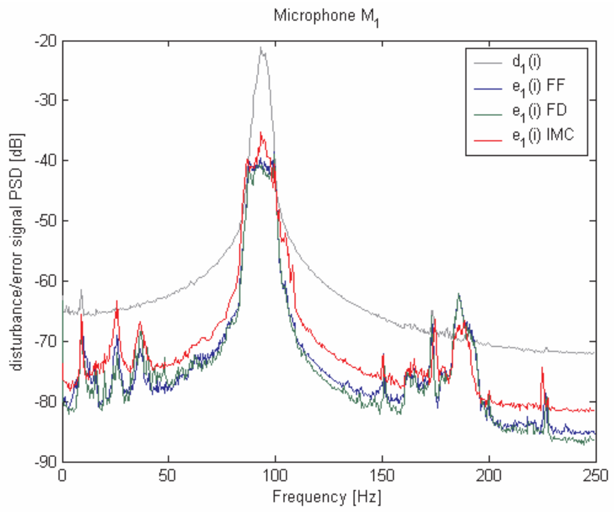

Fig. 3. PSD of disturbance and error signal picked up by error microphone M_1 in the ANC system disturbed by the narrowband noise, for controllers: feedforward full (FF), feedforward diagonal (FD) and IMC

The different structures and parameterization methods of adaptive control algorithms were applied to test different approaches to creation of distributed local zones of quiet – using feedforward and internal model control with full and diagonal controller structure. Results of real-world experiments showed the efficiency of ANC system in creation of local distributed zones of quiet. Spatial configuration of transducers enabled application of diagonal controllers. Attenuation obtained using full and diagonal feedforward controllers is similar. Attenuation in the IMC ANC system is slightly worse for random disturbances. However, the ANC system of this structure can be simply transformed into a decentralized system, built of independent single-channel units.

4. DEVELOPMENT AND ANALYSIS OF OPTIMAL AND ADAPTIVE SYSTEMS FOR GENERATING ZONES OF QUIET AT DESIRED LOCATIONS

The purpose of this research has been to design and verify feedback control algorithms capable to attenuate acoustic noise at desired locations in a group of electro-acoustic plants. This group has been characterised by small distances between these locations and locations of corresponding real microphones compared to the smallest acoustic wavelength significantly contributing to the noise [28]. Because the plants considered are non-minimum phase including time delays complete noise, cancellation is impossible with causal and stable controllers.

Feedback structures controlling noise at the real microphone have been considered first. Although optimal control problem for feedback has been well examined in the literature, it has been systematised for active control and originally applied for the rarely mentioned case of imperfect plant modelling. Design of the optimal H_2 control filter has been performed using the polynomial, frequency-domain and correlation-based approaches. In case of the polynomial-based approach two alternative design methodologies have been used. The first one requires modification of the basic cost function, i.e. variance of the system output, to respond to non-minimum phase character of the plant. In the second methodology inner-outer factorisation of a non-minimum phase model is required. Also the causal part of the optimal filter should be extracted or a Diophantine equation should be solved [32]. These operations are simpler when performed in the discrete-frequency domain. However, the latter approach involves finally designing a time-domain control filter, which well matches the obtained frequency response. The alternative correlation-based approach requires, in turn, calculating an autocorrelation matrix and a vector of cross correlation, what in fact is more computationally efficient when performed in the frequency domain.

The problem of optimal control of deterministic disturbances has been considered separately. It has been shown that for some structures it always has a solution, which is not unique, provided the control filter length is sufficiently large. In this case perfect, i.e. to the acoustic floor level, cancellation may be possible regardless of properties of the plant provided it does not have deep valleys in the response for the frequencies being controlled. A simplified stability analysis of the optimal control system has been briefly addressed.

Adaptive control has been considered next. The FX-LMS algorithm has been chosen for updating parameters of the control filter of FIR structure. Different representations and modifications of this algorithm have also been referred to. Sufficient conditions for convergence defined in different sense have been derived. The disturbance-to-output path has been linearized over trajectory allowing for obtaining a convergence phase condition convenient for analysis. It has been shown that these conditions differ as compared to those for feedforward systems and they are dependent on the control filter. Moreover, they demonstrate a significant coupling between stability of the structural feedback loop and convergence of the parameter-update algorithm [31]. Simulation experiments demonstrated that influence of the feedback loop on convergence of the adaptive algorithm cannot be strictly adjudicated. For some plants and disturbances the feedback loop may support the adaptation and for others it may restrict the convergence condition. Stability, convergence, convergence time (and rate), tracking and noise attenuation are crucially influenced by the convergence coefficient in the FX-LMS algorithm. It has been shown that for small values of the convergence coefficient there is a reciprocal dependence between this coefficient and convergence time, regardless of plant modelling error. Then, there is an optimal value of the coefficient, which depends on the plant delay and the control filter length. Further increase of the convergence coefficient increases the convergence time due to fluctuations of the residual signal, and finally the adaptive system suddenly diverges [40, 28].

The control systems have also been considered for multichannel plants. The design methodology similar to that for single-channel systems has been applied. Both optimal and adaptive solutions have been developed. There is no reliable analysis of convergence of the adaptive algorithms operating in the feedback system for a non-minimum phase plant including a delay, yet. Stability of such overall adaptive system is also not fully addressed. Then, a modification appropriate for improving stability has been provided. This is the Multi-Channel Leaky FX-LMS algorithm supported by weighting models of the cross paths [27].

The feedback systems have been experimentally verified for generating zones of quiet in a prototype of the active headrest system. The active headrest is a good example of the group of electro-acoustic plants under consideration. Due to geometrical arrangement of its components a significant coupling exists between the channels that can lead to instability when performing a decentralised control. Moreover, it has been shown that the attenuation is meaningfully degraded for this strategy, compared to fully multi-channel implementation [27, 28]. It has been demonstrated here by means of simulations of the optimal and adaptive systems as well as real-world experiments that, according to expectations, the classical feedback systems generate areas of the highest attenuation at the real microphones mounted in the headrest. The attenuation at the user ears is significantly lower and the attenuation gradient directly at the ears is high. As a result unpleasant effects are heard even in case of little head movements.

It has been found on the basis of the above experiments that it is necessary to design control systems capable to efficiently shift the zones of quiet to desired locations named virtual microphones [15, 16, 34]. The properties of the considered group of plants allow for easy estimation of the residual signal at the virtual microphone representing position of the user ear. In the first of the proposed systems, a real path model is used to reduce contribution of the control signal to the system output, thus allowing for estimation of the primary noise at the real microphone. Since it has been assumed that the primary noise at the real and virtual microphones is the same, it was added to an estimate of the secondary sound at the virtual microphone found by filtering the control signal by the virtual path model. Obtained estimate of the residual signal at the virtual microphone has been minimised. In this structure it constituted also the control filter input [30]. The second structure was similar to the first one with the exception that an estimate of the primary noise was the control filter input [27]. The third algorithm is composed of two stages. In the tuning stage the signal at the virtual microphone is directly minimised. At the same time knowledge about the residual signal at the real microphone is gained in an additional filter. This filter is then used during the actual operation, where the virtual microphone cannot be used, to produce a command signal to that measured by the real microphone [29]. These three systems have been designed as fixed and adaptive using all considered methodologies. Corresponding stability and convergence conditions have been derived [33].

For all presented control systems expressions for the spatial attenuation gradient due to change of the virtual path have been derived. It has also been shown that higher attenuation and larger zones of quiet can be obtained by increasing the number of microphones and loudspeakers. However, such a solution makes the system complicated, increases computational load and is less robust to plant perturbations. To minimise these difficulties the algorithm can be simplified by omitting contribution of the estimate of the reference signal in one path to control signals in the other path. However, models of all cross paths are still used for estimating the reference signals [27, 28].

Both simulation and laboratory experiments demonstrated that the multi-channel Virtual-Microphone Control systems efficiently shift the zones of quiet to desired locations. An exemplary distribution of the zones of quiet is presented in Figure 4.

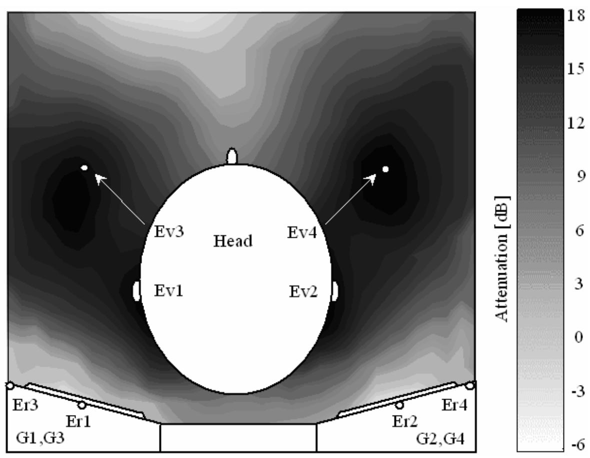

Fig. 4. Distribution of zones of quiet for a 250 Hz tone and a VMC system of four loudspeakers, four real microphones, and four virtual microphones

5. ENHANCEMENT OF ANC SYSTEM EFFICIENCY

The classical approach to the ANC system design requires high order analogue filters to avoid aliasing effects [1]. The main drawback of this approach is that these analogue filters deteriorate dynamics of the controlled plant and performance of ANC system is highly dependent on dynamic properties of the signal processing path especially for random disturbances.

A nonuniform additive random sampling gets around this limitation. It omits analogue filters in the system design and uses special signal sampling technique to disperse aliasing. This technique occurred to be the very effective for ANC systems. In this sampling method next time instant t_i is calculated on the basis of the previous sampling instant t_{i-1} plus independent positive-valued random variable τ_i. Statistical properties of the nonuniform additive sampling are characterized by distribution type, standard deviation δ and mean value ν of the random variable τ_i [3].

Nonuniform sampling has, however some limitations, for example nonsophisticated signal processing algorithms employed for signal processing of nonuniformly sampled signal introduce extra random errors to the processed signal [2]. Their level depends on sophistication of employed processing algorithm. The random errors are larger if the nonuniformly sampled signal is processed using methods destined for periodically sampled signals.

In order to reduce the influence of random errors there is a need to design special signal processing algorithms that process nonuniformly sampled signal values directly. An approach to the random error elimination is the employment of unorthogonal transforms or filtration techniques based on filters with time-varying coefficients [2, 3]. These approaches are computationally complex and thus difficult to implement in real-time control applications. To make nonuniform signal sampling a practical tool for real-time control purposes some simplifications must be done.

A possible approach is to resample nonuniformly sampled signal values into the corresponding periodically sampled data set so that further processing with well established signal processing techniques for periodically sampled signals can be used. However, there is an important point that must be remembered – the signal has to be bandlimited to the Nyquist frequency to avoid aliasing of a new periodically resampled signal [4, 5]. Another approach is to accept considerable level of random errors and directly supply processing algorithms with nonuniformly sampled signal values [3]. It is worthy to emphasize, that the level of random errors introduced by nonuniform signal sampling can be controlled by the proper choice of nonuniform signal sampling time instants probability density distribution properties.

Similarly in classical approach to ANC systems design special analogue reconstruction filter is used to convert the zero-order hold control signal for D/A converter into the corresponding continuous reconstructed signal. Such filter should remove all frequencies above half of the sampling frequency. This filter also deteriorate dynamics of the controlled plant and complicates analogue frontend of the ANC system. This limitation can be solved by employing a continuous-time signal reconstruction method with oversampling. The continuous-time signal reconstruction with oversampling increases the sampling frequency of the reconstructed signal several times. When the oversampled signal frequency very high (over 24 kHz) comparing to the electro-acoustic plant passband then the dynamic properties are determined mainly by the low-pass character of the control loudspeaker. This approach is very attractive if low-frequency loudspeakers are used, for typical sub-woofer used in laboratory experiments control signal attenuation above frequency 12 kHz exceeds 35 dB. This feature enables to get rid of the analogue reconstruction filter at all. Therefore, the signal reconstruction method with oversampling is valuable in ANC systems – it replaces analogue reconstruction filter with software oversampling algorithm.

In the first of proposed ANC system structures, the nonuniformly sampled signal values from the reference and error microphones are resampled into the corresponding periodically sampled signal values so that further signal processing with well-established signal processing techniques for ANC systems with periodically sampled signals can be applied. This system structure will be further referred as NSR-ANC system structure.

The NSR-ANC system structure can be simplified by omitting the signal resampling. It implies that reference and error signals are nonuniformly sampled and the adaptive control algorithm directly process the nonuniformly sampled signal values. This system structure will be further referred as NSD-ANC system structure.

Both described system structures employ the same signal reconstruction method with oversampling to avoid intersample effects during control signal reconstruction.

In order to compare properties of ANC system structures with nonuniform signal sampling and control signal reconstruction with oversampling (NS-ANC) series of experiments were performed and results were compared with corresponding results obtained for classical ANC system structure. It is worthy to notice that the NS-ANC system structures are capable to work with higher adaptation parameter μ what significantly improves ANC system performance. The NSD-ANC system is the most tolerant for increasing μ and enables to set up adaptation parameter μ at last 10 times larger then in the classical ANC system structure. This property enables this system to achieve considerably faster convergence then other presented systems structures.

Experiments concerning attenuation of tonal disturbance were preformed for different system structures (NSD-ANC, NSR-ANC, classical ANC) with adaptation parameter μ varying from 0.0001 to 0.1. The high attenuation of disturbance was observed in all experiments: 21.5 dB for NSD-ANC system structure, 25.2 dB for NSR-ANC system structure and 26.8 for classical ANC system structure. The lowest, but still satisfactory disturbance attenuation was observed for NSD-ANC system structure. The disturbance attenuation for NSR-ANC system structure is slightly lower then obtained using classical ANC system structure. The highest attenuation was obtained in classical ANC structure, however, adaptation algorithm convergence speed observed for the classical ANC system structure was considerably lower than it can be observed for NS-ANC systems. Figure 5 presents convergence of error signal during the tonal disturbance (sine 105 Hz) attenuation.

a) b)

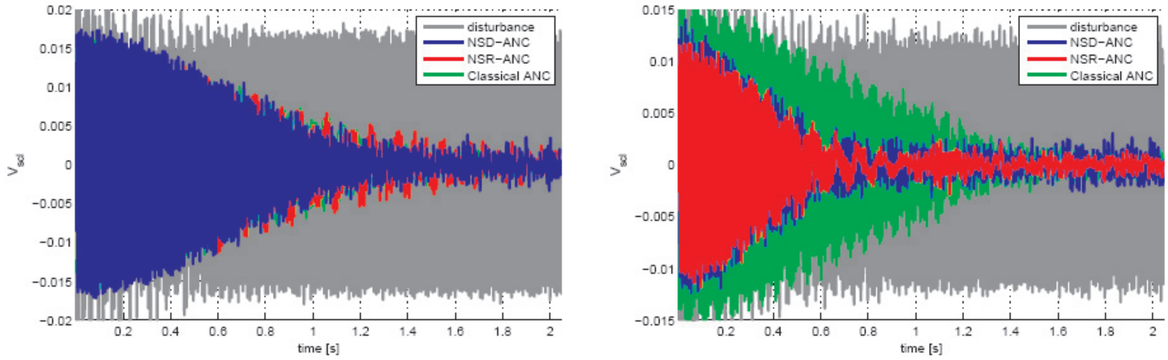

Fig. 5. Error signal time plot – disturbance signal sine 105 Hz attenuation for: a) $\mu = 0.001$; b) individually selected μ

There are two cases distinguished:

1) adaptation coefficient for all system structures is equal to $\mu = 0.001$,
2) for each structure μ is chosen individually to obtain best performance of the ANC system.

In the first case convergence in all presented system structures was very similar, in the second case the NS-ANC system structures were over twice as fast as classical ANC system structure. It should be noticed that for NS-ANC system structures higher adaptation coefficient could be set.

Figure 6a presents attenuation of the semi-random disturbance, for which the highest attenuation was obtained for NSD-ANC system structure taking advantage of the lowest group-delay in the signal processing path. Attenuation obtained for NSR-ANC system structure was slightly lower than for NSD-ANC system structure due to the time-delay introduced to the signal processing path by resampling algorithm. The maximum attenuation obtained using classical ANC system structure was equal to 8.0 dB and was over 5 dB lower then for NSD-ANC system structure. It is worthy to emphasize that the convergence speed of classical ANC system structure is lower than for NS-ANC system structures, see Figure 6a.

The last experiment was concerned with attenuation of the real-world disturbance (hoover noise). The PSDs were calculated using data picked up three seconds after ANC system activation. Three ANC system structures behaved similarly; dominating tone component was attenuated to background noise level (Fig. 6b).

Presented ANC algorithms with nonuniform signal sampling and control signal reconstruction with oversampling have satisfactory disturbance attenuation and fast convergence of adaptive control algorithm for the single tone signals as well as broadband disturbances. For difficult, nonperiodic disturbances (random and semi-random) the NS-ANC systems outperformed classical ANC system. Here the larger values of adaptation parameter μ can be applied in NS-ANC systems than in the classical structure. This results in faster adaptation of the NS-ANC system, while obtained levels of disturbance attenuation are of the comparable level.

The proposed approach to ANC system design has efficient hardware implementation where the function of analogue filters is taken over by the software solution that is responsible for dispersion of aliasing and inter-sample effects. The ANC system designed in such way is characterized by: simplified analogue frontend, cheaper implementation, smaller dimensions and possibility of changing of sampling frequency on-fly.

a) b)

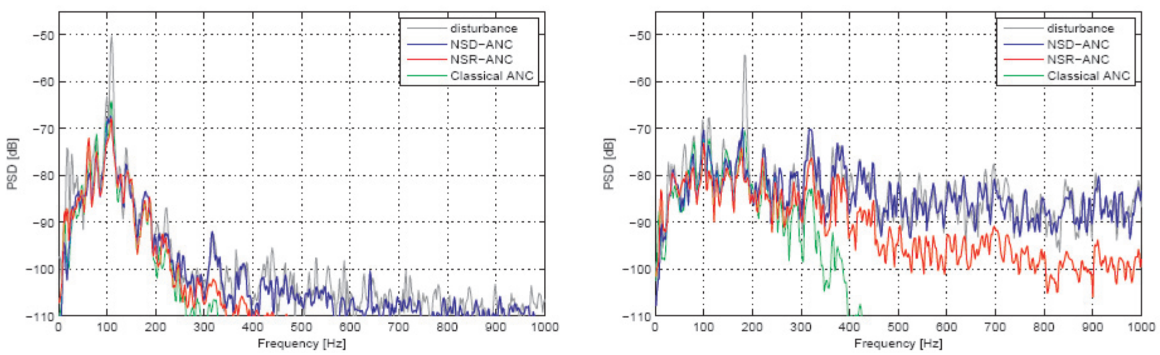

Fig. 6. PSD of error signal for individually selected μ, disturbance: a) semi-random; b) hoover attenuation

6. PORTABLE ANC PLATFORM

To put into effect ANC systems mentioned above, a portable multi-channel active noise control (PANC) platform was designed and implemented. The PANC is an efficient programmable DSP platform that is destined for generation of spatial local zones of quiet in enclosures using adaptive ANC algorithms. It consists of two modules: signal processing core (Fig. 7) and additional analogue frontend board.

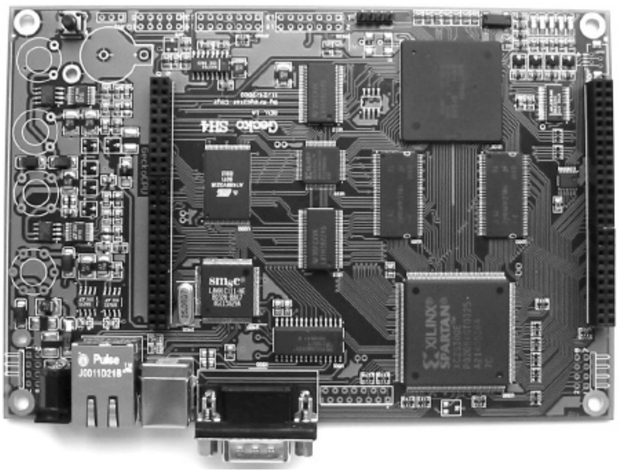

Fig. 7. PANC – the portable multi-channel active noise control platform core module

The signal processing core module is based on high-performance 32-bit RISC processor unit SH4 (SH7750R) with internal clock speed rated at 240 MHz. The processor comes with on-chip floating point unit (FPU) [35]. The FPU unit includes specialized high performance integrated vector floating point unit performing up to four floating-point multiplies and accumulations in a single cycle that combines to achieve 1.7 GFLOPS peak floating-point performance [37].

Digital signal processing algorithms used in ANC systems usually have very strong computational requirements. Substituting some parts of processing algorithms by hardware operations using FPGA may radically increase signal processing performance. Current generation of FPGAs can perform multiplication and addition operations at speeds exceeding 200 MHz; this is why they are suitable for computationally demanding applications like Fast Fourier Transform, FIR filtering and other multiply-accumulate operations [41]. It does not mean that all signal processing operations may be easily implemented in FPGAs. Especially floating point operations are quite difficult to implement due to the large amount of resources needed for these operations in the device. They are more suited to digital signal processors or even to general purpose processors. This is why FPGAs and CPU could coexist and create a flexible platform for signal processing purposes. The FPGAs can be also used as glue logic tying various system functions together and acting as programmable high-speed interface to analogue converters enabling to offload the CPU and reduce data transfer rates. In PANC platform Xilinx Spartan-IIE FPGA enables hardware realization of application-specific accelerators assisting the CPU in nonuniform signal sampling, control signal interpolation and reconstruction, data preprocessing and post processing. Additionally, the core board includes 64 MBytes of SDRAM memory, 4 MBytes of on-board programmable flash memory containing board firmware and 100 Mb/s Fast Ethernet controller.

Analogue frontend board contains eight analogue input channels for reference and error microphones and eight analogue output channels for control loudspeakers. Each of the analogue differential inputs have three programmable gains (×1, ×5, ×25). Input signals are pre-filtered by second order analogue filters designed to the fixed 20 kHz cut-off frequency providing low group-delays and eliminates high frequency distortions. The analogue-to-digital (A/D) converters are 16 bit differential Burr-Brown (formally Texas Instruments) ADS8343 successive approximation converters with maximum 100 kHz sampling frequency. The digital-to-analogue (D/A) converters are Burr-Brown DAC7615 – 12-bit performance converters with maximum 100 kHz sampling frequency.

One of the most important innovations that have made the design of portable ANC platform practical was nonuniform sampling technique which enabled to replace analog anti-aliasing filters by digital signal processing algorithm. Additionally, control signal reconstruction with oversampling was used to replace analog low-frequency forming filters with software oversampling algorithm which moves intersample effect above audible frequency range. However, simple second order analog filters designed to 20 kHz cut-off frequency are still present in the system to eliminate high frequency distortions while keeping very low group-delay.

The PANC system has efficient hardware implementation and simple analog frontend. Most of hardware was replaced with software or programmable logic enabling shorter development time and possibility of fast reconfiguration of PANC platform according to application needs. An important advantage of the presented PANC system is its flexibility, it can be successfully applied in different applications, not only in ANC.

7. CONCLUDING REMARKS

In the paper some recent results obtained in the field of active noise control have been presented. They are involved with new algorithms for adaptive control and identification as well as with hardware implementation. They reveal the power of ANC idea in noise reduction, although there are still some topics to solve. We will continue the research on comparison of noise reduction as well as dimensions of zones of quiet that can be achieved with classical and distributed ANC systems. We also intend to study the features of adaptive control algorithms depending on their parameterization. Some of the results are useful not only for ANC applications but they can be applied in other areas of control as well, e.g. in sonar, radar, telecommunication etc.

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