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NONUNIFORM SIGNAL SAMPLING FOR ACTIVE SOUND CONTROL

Krzysztof Czyż

Institute Automatic Control, Silesian University of Technology
Akademicka 16, 44-100 Gliwice, Poland
krzysztof.czyz@polsl.pl

Abstract

The paper presents a new design concept of a compact active noise control (ANC) system employing nonuniform signal sampling method. The presented concept allowed to develop ANC system that has efficient hardware implementation with simple analogue frontend. The function of analogue antialiasing and reconstruction filters were taken over by the software solution that is responsible for dispersion of aliasing and inter-sample effects. System employing described methods of nonuniform error and reference signal sampling and control signal oversampling was shown to successfully create local spatial zones of quiet in enclosures.

INTRODUCTION

The classical approach to the ANC system design requires high order analogue filters to avoid aliasing effects. The main drawback of this approach is that these analogue filters deteriorate dynamics of the controlled plant. The different possible techniques replacing analogue filters in the system design were considered and nonuniform signal sampling occurred to be the most effective in ANC systems. The deliberate irregularities in signal sampling can suppress aliasing without employing analogue antialiasing filters. Moreover, in the presented system additional control signal oversampling method replaced classical control signal reconstruction using analogue reconstruction filters. The inter-sample effects in reconstructed signal are still present but moved above audible frequency range. The presented concept allowed to develop ANC system that has efficient hardware implementation with simple analogue frontend. System employing nonuniform error and reference signal sampling and control signal oversampling was used for creation of local spatial zones of quiet in enclosures.

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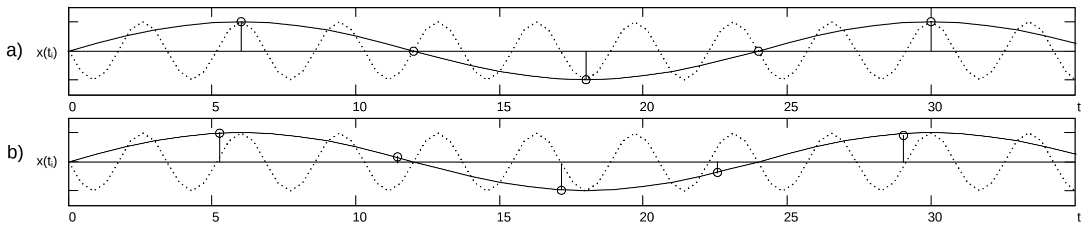

Figure 1: Sampling of analogue signal with various schemes of sampling time instants distributions a) periodic sampling, b) nonuniform additive random sampling.

NONUNIFORM SIGNAL SAMPLING

The main drawback of periodic sampling is the fact that most of real-world signals contain high-frequency components above the Nyquist frequency. These high-frequency components appear after sampling to be low-frequency components due to aliasing [1]. Figure 1a presents two different analogue signals that produce the same data set after periodic sampling. This representation is not unique – it is not possible to distinguish analogue signals having their digital representations if additional information is not supplied.

In order to avoid aliasing it is necessary to filter analogue signals before sampling through analogue low-pass filters [1]. But these analogue filters incorporate significant group delay into the ANC system signal processing path which deteriorate overall ANC system performance [2].

Alternative approach to analogue signal sampling is the use of nonuniform signal sampling method. This idea is presented in Figure 1b. It can be noticed that only one sine function can be drawn exactly through the data set obtained by the nonuniform signal sampling of lower frequency sine (solid line in Figures 1b. This means that sines with different frequencies are represented by different data sets [3]. Nonuniform signal sampling enables to disperse aliasing by introduction of deliberate irregularities from periodic sampling.

Nonuniform additive random sampling mathod sampling time instants t_i^n are calculated in the following way [3]:

$$t_i^n = t_{i-1}^n + \tau_i, \tag{1}$$

where τ_i is a family of identically distributed independent positive-valued random variables. Statistical properties of the nonuniform additive sampling are characterised by:

- distribution type of random variable τ_i,
- standard deviation δ of the random variable τ_i,
- mean value ν of the random variable τ_i.

The irregularities in sampling are characterised by δ/ν ratio [4]. The value of δ/ν ratio is usually kept in range of 0.01-0.1. This ratio should be set large to disperse aliasing. On the other hand δ/ν ratio value is limited by random errors introduced by signal processing algorithms.

PROCESSING OF NONUNIFORMLY SAMPLED SIGNAL

Signal processing algorithms for nonuniformly sampled signals introduce random errors to the processed signals. These random errors result from the application of signal processing algorithms that do not respect or partially respect irregularities in sampling time instants. It should be emphasised that nonuniformly sampled signals are always biased by random errors after processing. Their level depends on sophistication of employed processing algorithms [4]. The random errors are larger if the nonuniformly sampled signal is processed using methods destined for periodically sampled signals.

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 random errors elimination is the employment of unorthogonal transforms [4] or the filtration techniques based on filters with time-varying coefficients [4]. 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 [5]. 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. Another approach is to accept considerable level of random errors and directly supply processing algorithms with nonuniformly sampled signal values [6].

The level of random errors introduced by nonuniform signal sampling can be controlled by the choice of nonuniform signal sampling time instants probability density distribution properties. It should be remembered that direct processing of nonuniformly sampled signals values does not disperse aliases completely, they are still present but at the lower level.

ANC SYSTEMS WITH NONUNIFORM SIGNAL SAMPLING

Classical approach to ANC system design requires high order analogue filters to avoid aliasing and inter-sample effects [7]. In Figure 2a the electrical part of the classical ANC system signal processing path is presented. It consists of reference microphone, anti-aliasing filter (AAF), A/D converter, controller $W^i(z^{-1})$, D/A converter, analogue reconstruction filter (RCF) and loudspeaker.

The main drawback of this approach is that the analogue filters deteriorate dynamics of the ANC system signal processing path. Deliberate irregularities in signal sampling time instants can successfully suppress aliasing without deteriorating dynamics into the ANC system signal processing path. Thus the nonuniform signal sampling was employed to create a new

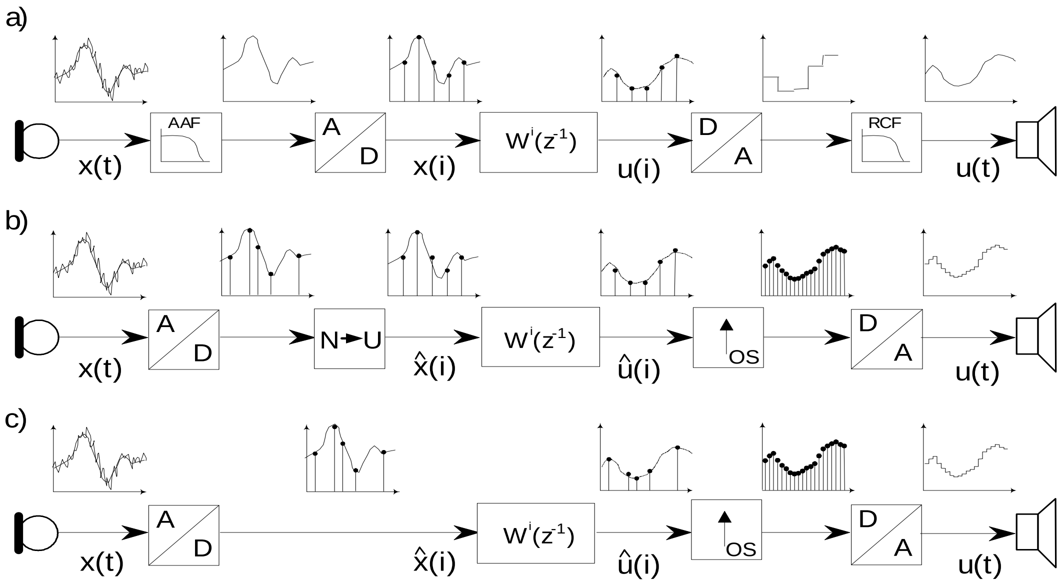

Figure 2: The electrical part of the ANC system signal processing path: (a) classical ANC system with periodic signal sampling, (b) ANC system with nonuniform signal sampling and signal resampling to the corresponding periodically sampled signal (NSR-ANC), (c) ANC system with direct nonuniformly sampled signal processing (NSD-ANC).

class of ANC system structures. In the sequel they are denotes as NS-ANC system structures.

In the first of proposed ANC system structures (Figure 2b) the nonuniformly sampled signal values 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. In this ANC system structure continuous time signals $x(t)$ and $e(t)$ from the reference and error microphones are nonuniformly sampled by the A/D converter. Then they are resampled into the corresponding periodically sampled signal value estimates $\hat{x}(i)$ and $\hat{e}(i)$ with the resampling method (N→U). The estimates $\hat{x}(i)$ and $\hat{e}(i)$ are used by the controller $W^i(z^{-1})$ to calculate values of the control signal. 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 (Figure 2c). This system structure will be further referred as NSD-ANC system structure.

Both presented NS-ANC system structures employ the same control signal reconstruction method with oversampling (OS) to avoid inter-sample effects. This signal reconstruction method replaces the analogue reconstruction filter (RCF) present in the classical ANC system structure (Figure 2a). The control signal is reconstructed with increased sampling rate and then it is smoothed by the loudspeaker (Figure 2b,c).

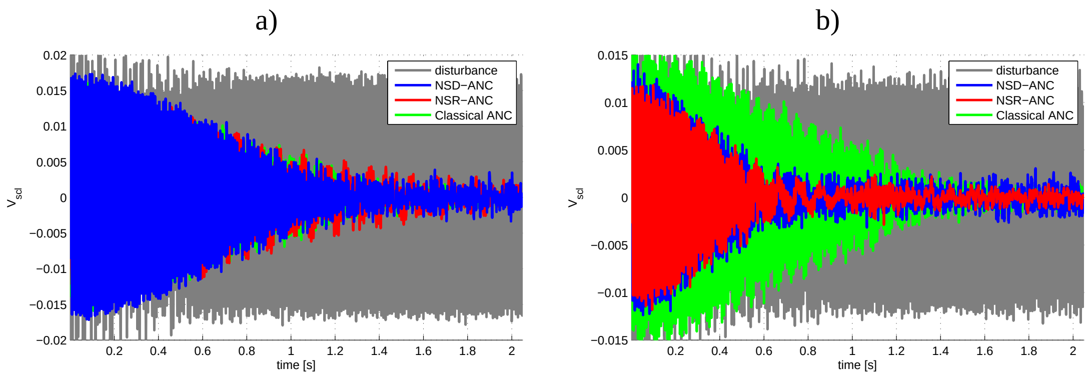

Figure 3: Error signal time plot - disturbance signal $sine105$ attenuation for a) $\mu = 0.001$, b) individually selected μ.

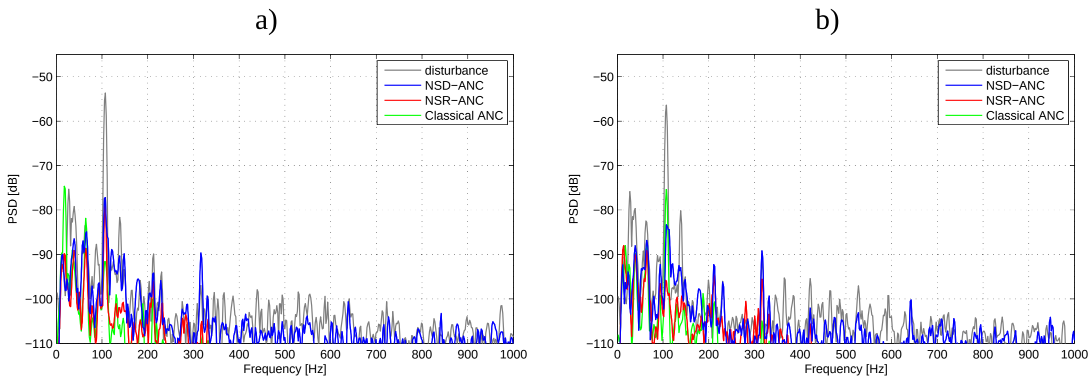

Figure 4: PSDs of error signal calculated after two seconds after system activation - disturbance signal $sine105$ attenuation for a) $\mu = 0.001$, b) individually selected μ.

experiment	disturbance	enclosure	μ	attenuation AT (dB) NSD-ANC	NSR-ANC	Classical	figure
L1a	$sine105$	laboratory	0.0001	11.4	8.0	22.8	
L1b	$sine105$	laboratory	0.0003	20.1	13.0	26.8	
L1c	$sine105$	laboratory	0.0006	21.5	25.2	26.1	
L1d	$sine105$	laboratory	0.001	20.0	25.0	26.2	3a
L1e	$sine105$	laboratory	0.003	20.5	24.9	23.4	
L1f	$sine105$	laboratory	0.006	17.9	24.2	–	3b
L1g	$sine105$	laboratory	0.01	13.6	23.5	–	
L1h	$sine105$	laboratory	0.03	19.0	–	–	3b
L1i	$sine105$	laboratory	0.06	17.2	–	–	
L1j	$sine105$	laboratory	0.1	–	–	–	

Table 1: The disturbance $sin105$ attenuation in laboratory enclosure; – no convergence.

COMPARISON WITH CLASSICAL ANC SYSTEM

In order to confirm properties of NS-ANC system structures series of experiments were performed and results were compared with corresponding results obtained for classical ANC system structure. The experiments were preformed in the laboratory enclosure. The system parameters and results of disturbance attenuation are shown in Tables 1-3. It is worth 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 (Tables 1, 2 and 3).

Thirty experiments concerning attenuation of disturbance $sin105$ 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 3 presents convergence of error signal during the disturbance $sin105$ attenuation. There are two cases distinguished: –the first case when the adaptation coefficient for all system structures is equal to μ=0.001 and –the second case when 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 4 present the corresponding PDSs of error signal calculated on the basis of samples picked up two seconds after the ANC system activation. The results of this experiment confirmed faster convergence of NS-ANC system structures. For classical and NSR-ANC system structures disturbance is attenuated almost to the background noise level, for NSD-ANC system structure disturbance attenuation is limited by significant random errors.

The results of experiments for the disturbance $semirand1$ attenuation are shown in Table 2. The control system adaptation parameter μ values were changed in the range from 0.0006 to 0.3. Figure 5a presents attenuation of the random disturbance $semirand1$, for

experiment	disturbance	enclosure	μ	attenuation AT (dB)			figure
				NSD-ANC	NSR-ANC	Classical	
L2a	$semirand1$	laboratory	0.0006			6.7	
L2b	$semirand1$	laboratory	0.001	2.0	5.2	5.4	
L2c	$semirand1$	laboratory	0.003	8.5	5.0	8.0	5a
L2d	$semirand1$	laboratory	0.006	2.6	5.2	0.1	
L2e	$semirand1$	laboratory	0.01	5.3	3.1	–	
L2f	$semirand1$	laboratory	0.03	13.2	10.2	–	5a
L2g	$semirand1$	laboratory	0.06	4.1	–	–	
L2h	$semirand1$	laboratory	0.1	6.1	–	–	5a
L2i	$semirand1$	laboratory	0.3	–	–	–	

Table 2: The disturbance $semirand1$ attenuation in laboratory enclosure; – no convergence.

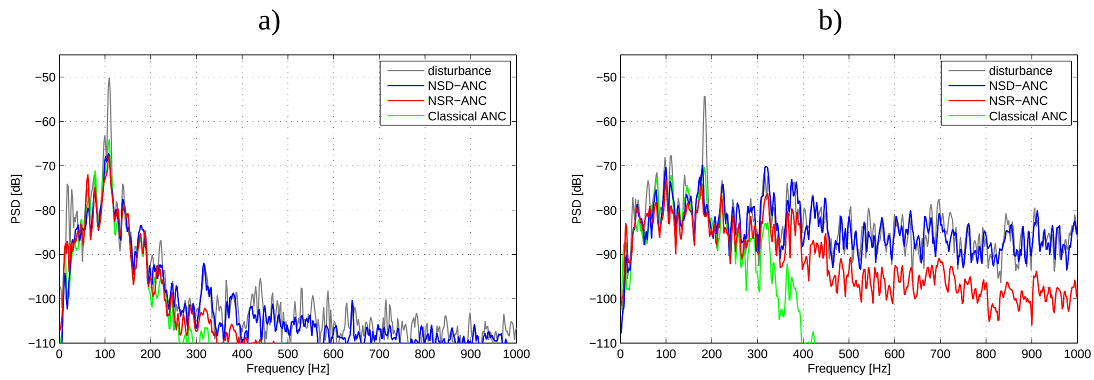

Figure 5: The error signal PSDs for individually selected μ, disturbance: a) $semirand1$, b) $hoover$ attenuation.

experiment	disturbance	enclosure	μ	attenuation AT (dB) NSD-ANC	NSR-ANC	Classical	figure
L3a	$hoover$	laboratory	0.0006			4.7	
L3b	$hoover$	laboratory	0.001	3.9	4.6	5.6	
L3c	$hoover$	laboratory	0.003	4.3	5.7	6.4	
L3d	$hoover$	laboratory	0.006	4.7	6.5	6.9	
L3e	$hoover$	laboratory	0.01	5.4	6.8	7.1	5b
L3f	$hoover$	laboratory	0.03	5.6	7.7	2.0	
L3g	$hoover$	laboratory	0.06	6.3	8.0	–	5b
L3h	$hoover$	laboratory	0.1	6.7	4.4	–	5b
L3i	$hoover$	laboratory	0.3	–	–	–	

Table 3: The disturbance $hoover$ attenuation in laboratory enclosure; – no convergence.

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 the 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 worth to emphasise that the convergence speed of classical ANC system structure is lower than for NS-ANC system structures (Figure 5ab).

The last experiment was concerned with attenuation of the real-world $hoover$ disturbance. The PSDs were calculated using data picked up three seconds after ANC system activation. The parameters used during experiment are given in Table 3. Performance of three ANC system structures was very similar, dominating tone component was attenuated to background noise level (Figure 5b).

CONCLUSIONS

The new ANC system structures with nonuniform sampling of the signals from reference and error microphones were proposed:

– the NSR-ANC system structure employing resampling method of nonuniform sampled signal values into the corresponding periodically sampled signal values,

– the NSD-ANC system structure simplifying NSR-ANC approach by omitting signal resampling – the adaptive control algorithm process directly the nonuniformly sampled signals values.

Properties of the proposed NS-ANC system structures were illustrated by the results of laboratory experiments and they were compared with results for classical ANC system. Usefulness and effectiveness of proposed NS-ANC system structures for active noise control was shown. Moreover, experiments confirmed that the elimination of analogue anti-aliasing and reconstruction filters leads to higher convergence speed and greater noise attenuation.

Satisfactory disturbance attenuation and fast convergence of adaptive control algorithm were obtained for the single tone signals as well as broadband disturbances. For difficult, non-periodic disturbances the NS-ANC systems outperformed classical ANC system.

It is worth to emphasise that larger values of adaptation parameter μ can be applied in NS-ANC systems than in the classical structure. It 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 – analogue filters are no longer required. 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 characterised by: simplified analogue frontend, cheaper implementation, smaller dimensions and possibility of changing of sampling frequency on-fly.

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