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RLS-BASED CONTROL ALGORITHMS FOR 3D ZONES OF QUIET

Małgorzata I. MICHALCZYK[(1)], Janusz SERWIN[(2)]

[(1)] Silesian University of Technology
Institute of Automatic Control
Akademicka 16, 44-100 Gliwice, Poland
e-mail: malgorzata.michalczyk@polsl.pl

[(2)] Silesian University of Technology
Graduated in Automatic Control and Robotics
Department of Automatic Control, Electronics and Computer Science

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The paper summarises research on application of RLS-based (recursive least squares) adaptive control algorithms for single-channel active noise control (ANC) system, used to create three dimensional (3D) local zones of quiet in a reverberant enclosure by attenuation of tonal disturbances. Three different ANC system structures are concerned: classical "filtered-x", modified "filtered-x" and "adjoint" structures. A problem of parameterisation of six RLS-based adaptive control algorithms is evaluated, initially in simulations with the use of nonsimplified plant models and tonal disturbances. Next, research results are proved in real-world experiments. Finally, the efficiency of ANC systems using different control structures and RLS-based control algorithms is compared.

Keywords: active noise control, adaptive control, electro-acoustic plant models, RLS algorithm.

1. Introduction

Although the research on active noise control (ANC) dates back over 70 years, creation of 3D zones of quiet in enclosures still remains a challenge [4, 8, 11, 12]. New ideas how to deal with the problem are still being proposed. A number of control system structures and recently designed control algorithms is increasing [2, 3, 9], causing an ANC system designer to face the problem, which control system structure and control algorithm to choose.

Due to time-varying features of ANC systems creating 3D zones of quiet in enclosures, it is often necessary to employ an adaptive control algorithm. The majority of publications is devoted to modifications of least mean squares (LMS) algorithm [4, 8], as the most popularly applied. Less can be found about the application of recursive least squares (RLS) algorithms. A very reliable research report is presented in [2, 3], utilising, however, models of electro-acoustic paths of an ANC system aimed for a duct. In the case of application of these algorithms to create 3D zones of quiet in enclosure electro-acoustic paths are characterised by significantly more complicated dynamics, causing parameterisation of control algorithms to be a very difficult task [11, 12] (Fig. 4). Additional problems are implied by the need to take into account a time-varying disturbance. RLS-based control algorithms diverge in the presence of insufficiently exciting signal, thus, tonal disturbances are very dangerous for the systems utilising these algorithms.

In order to evaluate these issues the problems of parameterisation of RLS-based control algorithms applied for creation of 3D zones of quiet in an enclosure are considered in the paper. Chosen control algorithms were compared to find the most efficient for the considered application. Most of experiments were conducted as computer simulations, in order to make the considerations independent on time-varying conditions in the enclosure. An employment of complicated electro-acoustic paths models increased the simulations reliability. Finally, the simulations results were proved in real-world experiments.

2. ANC system proposed

The ANC system discussed in this paper is applied to create 3D zones of quiet in enclosure of cubature of 70 m^3 [15]. It is a one-channel feedforward adaptive system with one reference microphone. A zone of quiet is created around the error microphone, which picks up an error signal. Two channels of the electro-acoustic plant are defined: a secondary path, including electronic instrumentation and acoustic space of the enclosure between the control loudspeaker and the error microphone, and, similarly, the acoustic feedback path, including the reference microphone.

The adaptive controller employs an adaptive control algorithm with neutralisation of acoustic feedback [8, 15]. Identification of electro-acoustic plant models necessary for parameterisation of the control algorithm is performed before the activation of the ANC system [8, 15]. The three following control system structures were tested:

- the most popular "filtered-x" structure shown in Fig. 1, described elsewhere [8],
- "modified filtered-x" structure shown in Fig. 2, introduced by Bjarnason in [1]
- "adjoint" structure, shown in Fig. 3 [3, 16].

From the variety of RLS-based ANC algorithms the following were chosen:

- RLS [5, 7],
- symmetry preserving RLS [2, 5],
- RLS with variable forgetting factor [6],
- RLS with covariance matrix regularisation [13],
- fast-transversal-filter (FTF) [14],
- stabilised FTF [3],

and the standard LMS algorithm [8] was also applied as a reference algorithm. All these algorithms were developed to improve the properties of the standard RLS algorithm: improve the speed of convergence (RLS with variable forgetting factor), prevent algorithm divergence (symmetry preserving RLS, RLS with covariance matrix regularisation, stabilised FTF), decrease computational complexity (FTF, stabilised FTF).

A large group of RLS-based algorithms utilising lattice filter structure was not considered in this research as the application of these algorithms for 3D zones of quiet was widely discussed in [9, 10].

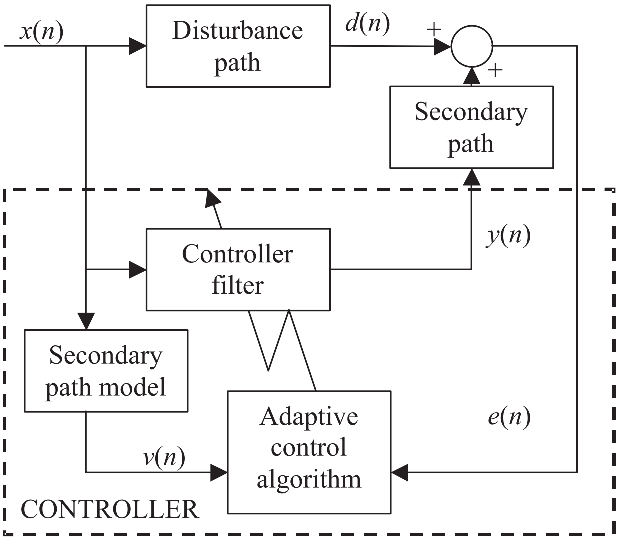

Fig. 1. “Filtered-x” ANC system structure.

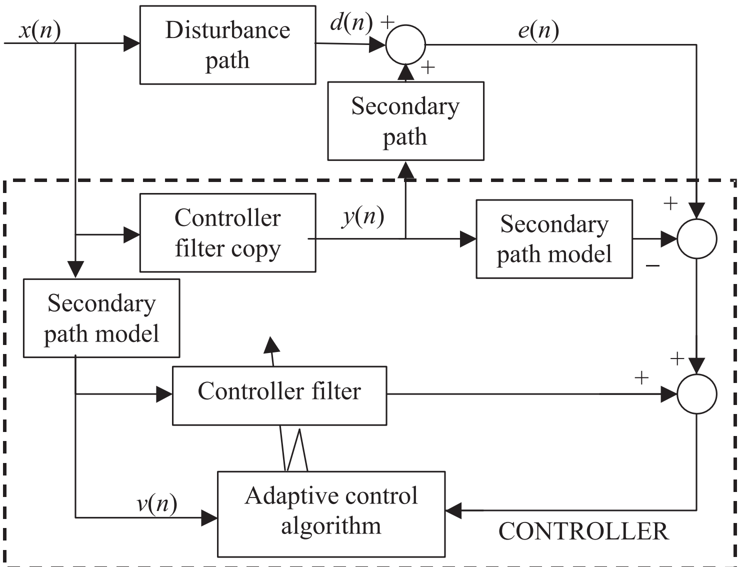

Fig. 2. Modified “filtered-x” ANC system structure.

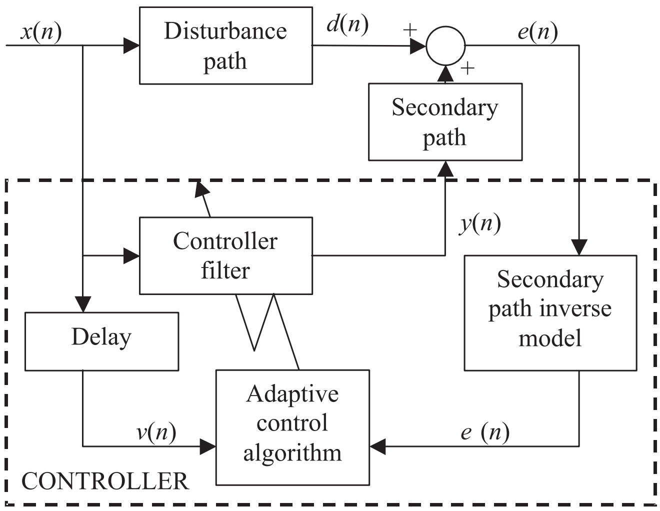

Fig. 3. “Adjoint” ANC system structure.

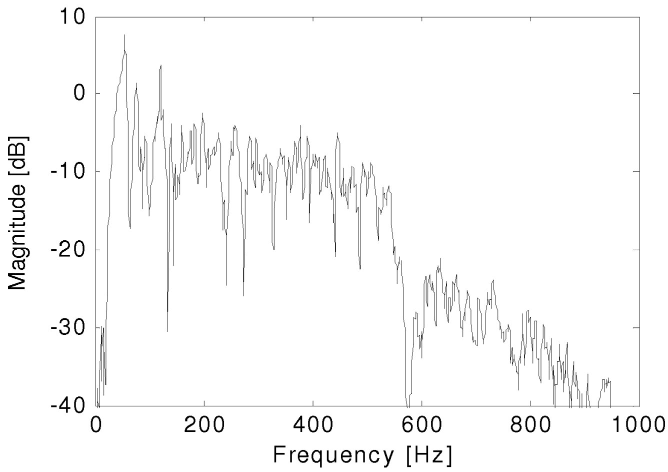

Fig. 4. Magnitude of a secondary path model.

The algorithms’ properties are shaped by parameters:

- plant models; an example of the magnitude of the secondary path model (FIR filter of the order 1000) is given in Fig. 4, showing a very complicated dynamic properties of the plant,
- forgetting factor λ, the parameter of RLS, RLS with covariance matrix regularisation, symmetry preserving RLS and FTF algorithms,
- gain μ, applied for all algorithms, allows for better shaping of their properties, was introduced independently in [9] and [3],
- controller filter order, depending on the disturbance spectral complexity, most frequently 3 for tonal disturbances.

3. Simulations and experiments results

A number of computer simulations were carried out [15] to determine algorithms' parameterisation rules. The quality of performance was assessed basing on two factors: the maximum attenuation level (calculated as the ratio of the estimated error and disturbance signal variances) and the time, after which this attenuation was reached. Then, the parameter sets assuring the best performance were chosen for each algorithm to allow reliable comparison of ANC system performance.

The tonal disturbances used were either of a single tone of 121 Hz frequency used in initial simulations or tonal disturbances from the range 50–150 Hz used to compare the algorithms. Plant models of 1000 coefficients were used by control algorithms (working with 2 kHz frequency) and models of 2000 coefficients (the secondary path) and 4000 coefficients (the acoustic feedback path) were used to simulate real plant paths, sampled with 8kHz frequency. Controller filters of order 1, 2 and 3 were applied for attenuation of tonal disturbances. The gain μ was changed with the step 0.1 in the range 0–1, and the forgetting factor λ was changed with the step 0.01 in the range 0.9–0.99.

In general, the attenuation level obtained for the tonal disturbance of 121 Hz frequency varied in the range 22–30 dB dependently on the kind of the control algorithm and control system structure applied. The lowest attenuation levels were obtained in the "adjoint" structure; the two remaining structures assured levels of 29–30 dB for all algorithms. The time necessary to reach the maximum attenuation level varies from 0.4 s for symmetry preserving RLS in "filtered-x" structure to over 11 s for LMS algorithm in "adjoint" structure.

In particular (Table 1):

- The shortest time to obtain the maximum attenuation level was due to the "filtered-x" control system structure, varying from 0.4 s to 0.7 s for corresponding RLS and LMS algorithms. The attenuation obtained was about 30 dB.
- Adaptive control algorithms applied in the "modified filtered-x" control system structure are the easiest to be parameterised. It means, that they remain convergent for a wide range of parameters. For instance, the stabilised FTF algorithm applied in the "filtered-x" structure is convergent for the gain μ remaining in the range 0.01–0.12, while for the "modified filtered-x" structure 0.01–1, i.e. 10 times wider. Maximum attenuation levels obtained are about 30 dB. The only disadvantage is an insignificantly longer time to obtain the highest attenuation level in comparison to the "filtered-x" structure, from 0.7 s for symmetry preserving RLS, through 0.8 s for LMS, to 1.6 s for RLS with variable forgetting factor.
- Definitely, the worst results were obtained for the "adjoint" control system structure, regarding both the highest attenuation level (from 22 dB) and the time needed to reach it (from 2.8 s for RLS with covariance matrix regularisation).
- For simple tonal disturbances the controller order of $N = 3$ was sufficient. Application of higher order controllers either deteriorated the results or caused the immediate divergence of a control algorithm. This is what never happens, if LMS-based control algorithms are applied.

Table 1. The best simulation results and parameter sets to obtain them, attenuation of 121 Hz tonal disturbance.

Structure	Adaptation algorithm	Controler filter order	Gain μ	Forgetting factor λ	Maximum attenation [dB]	Discrete time of max. attenuation [samples]
filtered-x	RLS	1	0.4	0.999	30	5500
	symmetry preserving RLS	2	0.2	0.99	30	3200
	RLS with variable forgetting factor	2	1	–	29	4000
	RLS with covariance matrix regularisation	2	0.2	0.1	29	3200
	FTF	1	0.1	0.96	30	4000
	stabilised FTF	2	0.06	0.97	30	3200
	LMS	1	$5 \cdot 10^{-8}$	–	30	5500
modified filtered-x	RLS	1	0.7	0.999	30	7500
	symmetry preserving RLS	1	0.6	0.93	30	5500
	RLS with variable forgetting factor	1	1	–	30	13000
	RLS with covariance matrix regularisation	–	–	–	–	–
	FTF	1	1	0.94	30	5500
	stabilised FTF	1	0.2	0.91	30	5500
	LMS	2	$3 \cdot 10^{-7}$	–	29	6200
adjoint	RLS	1	0.9	1	22	32000
	symmetry preserving RLS	1	0.9	1	23	33000
	RLS with variable forgetting factor	6	0.8	–	26	33000
	RLS with covariance matrix regularisation	3	0.006	0.01 (λ_0)	30	22000
	FTF	1	0.9	1	22	33000
	stabilised FTF	–	–	–	–	–
	LMS	2	$5.5 \cdot 10^{-10}$	–	30	92000

- The best results were obtained with symmetry preserving RLS and stabilised FTF algorithm. In both “filtered-x” and “modified filtered-x” structures they assured significantly shorter time to reach the highest attenuation level.
- For some control algorithms the experiments failed – there was no such a set of parameters found, which would assure control algorithms convergence, e.g. RLS algorithm with covariance matrix regularisation in “modified filtered-x” structure or stabilised FTF algorithm in “adjoint” structure were the cases.

- From the rest of the algorithms the most difficult to parameterise turned out to be stabilised FTF algorithm in "filtered-x" structure. It was convergent for a very narrow range of gain μ – in 0.01–0.12 only.
- The easiest to parameterise was RLS algorithm with variable forgetting factor.

Symmetry preserving RLS and stabilised FTF algorithms were chosen for further comparison, applied in both "filtered-x" and "modified filtered-x" structures for attenuation of tonal disturbances from the range 50–150 Hz. In particular, the simulations' results showed, that

- symmetry preserving RLS algorithm in "filtered-x" structure assures the highest attenuation level (about 20 dB in average) it is characterised, however, by the insignificantly slower convergence in comparison to both symmetry preserving RLS in "modified filtered-x" structure and stabilised FTF algorithms,
- stabilised FTF algorithm in "modified filtered-x" structure was characterised by the fastest convergence, reaching the maximum attenuation level (16 dB in average) after about 0.53 s,
- stabilised FTF algorithm in "filtered-x" structure was divergent for most of the disturbance frequencies.

Because it gave the best results, symmetry preserving RLS algorithm was applied to attenuate both tonal and random disturbances (110–130 Hz). In real-world experiments in laboratory. Attenuation level of over 24 dB for tonal and about 8 dB for random disturbance proved RLS-based algorithm efficiency in the task of creation of 3D zones of quiet in enclosure.

4. Conclusions

The simulations and real-world experiments results gave an insight into the problem of parameterisation of RLS-based algorithms applied for ANC in an enclosure. Assuring faster convergence, similar attenuation level as compared to LMS-based algorithms they are, however, significantly more difficult to parameterise.

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