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Frequency-Domain Wideband Acoustic Noise Cancellation System

Vera Andreevna Karaseva, Aleksey Vital'evich Lvov, Aleksandr Alekseevich Rodionov

Federal Research Center A.V. Gaponov-Grekhov Institute of Applied Physics of The Russian Academy of Sciences (IAP RAS), Nizhny Novgorod, Russia
Email: verakaraseva@ipfran.ru

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Abstract

When building an adaptive noise cancellation system for wideband acoustic signals, one can meet some difficulties in practical implementation of such a system. The major problem is related to the necessity of using real-time signal generation and processing. In this paper the active noise control system which utilizes adaptation in frequency domain is considered. It is shown that the proposed algorithms simplify practical implementation of a noise cancellation system. The results of computer simulations and prototype experiments show the effectiveness of the proposed methods.

Keywords

Active Noise Cancellation System, Adaptive Noise Cancellation

1. Introduction

The problem of acoustic noise level reduction is rather urgent in many spheres of human activity [1] [2], life and health of people directly depend on its successful solution [3]. The prevalent way of acoustic noise reduction consists of using passive damping. This approach is common in industry and is used for cancellation of noise of machines, mechanisms, etc. [4] [5]. As a rule, passive noise reduction methods imply creation of surfaces or layers which are coated with different sound-absorbing materials of different compound and shape. The main disadvantage of passive approach consists in its low efficiency of noise damping at low frequencies. On the other hand, active noise control systems are much more effective in low frequency band.

The possibility of using active noise reduction systems for acoustic fields generated by emitters inside a confined space was theoretically considered in [6]. Subsequently, many authors have proposed some methods for practical implementation of such systems. For example, the work [7] considers a method based on the forming a surface of controlled emitters which are located near an acoustic source to be suppressed. The control of the registered by receivers field is performed by specially generated signals which are fed to controlled emitters. These emitters provide minimization of the total field power at given points or space area. A detailed review of the active noise suppression methods used in paper is provided in [8] [9] [10].

The basis of many methods used to form signals that are fed to controlled transmitters is the transformation of signals registered by reference sensors [8]. Acoustic receivers or vibration sensors which are located near the noise sources could be used as reference sensors.

The most popular methods for generating signals which are fed to the controlled emitters include using iterative adaptive algorithms. These algorithms are based on the determination of the impulse response of the shaping filters that transform signals recorded by the reference sensors.

To implement these algorithms, it is necessary to have estimates of the impulse responses for the paths, connecting each emitter to each receiver. This entails the perforce of conducting calibration of a system. The calibration procedure implies feeding calibration signals (e.g., short pulses, bandwidth noise, broadband modulated signal) to each of the controlled transmitters and calculating the desired transfer characteristics.

However, this approach entails a number of disadvantages, such as longer system setup times, inefficiency when the characteristics of the environment change, and the need to periodically repeat the calibration procedure. The solution to this problem could be the feeding of the calibration signal directly while the system is operating [11]. However, this will entail an increase of the total radiation level of a system. This problem can be solved by using a modification of the system [11], which is described in [7].

A clear limitation of the systems described above consists of computational difficulty in their practical implementation, because they are functioning in real time. At the moment, there are already known ways to reduce the computational cost and implement noise cancelling in frequency domain (e.g., using the Fourier window transform) [12] [13]. There is also an algorithm of active noise control of sound fields at discrete components of the signal spectra, which does not require real-time signal processing [14].

In this paper, an active control system with an adaptation procedure in frequency domain in considered. In the proposed algorithm the calculation of impulse responses of shaping filters is based on frequency representation of recorded signals. At the same time the signals, which are fed to the controlled emitters, are generated in time domain.

2. Active Noise Cancellation Algorithm without Using Real-Time Mode for System Adaptation

2.1. Algorithm Description

Let us consider a method of active noise cancellation without using real-time mode for system adaptation. **Figure 1** shows a block diagram of the active noise cancellation system of an acoustic field generated by a spatially extended object. The left part of this diagram shows a noise source ("vibr") to be compensated with a reference sensor located next to it. In the right part of the figure there is a set of J controlled emitters and K receivers of the residual field. Block **W** represents a multichannel shaping filter. The outputs of the filter are fed to the controlled emitters. The block "Computer" is served for updating the coefficients of the multichannel shaping filter **W**. The signals recorded by the reference sensors are fed to a set of shaping filters with finite impulse responses (FIR). The FIRs are estimated from the signal samples of the reference sensor and residual receivers.

Let us consider all signals at the inputs and outputs of the block diagram to be represented in the frequency domain. For convenience, we omit the indices corresponding to frequency components. Let us introduce the following notations (for the sake of convenience, the footnotes in the text will indicate the dimensions of corresponding matrices).

Let $\mathbf{y}_p = \mathbf{P}\mathbf{v}$ is the field created by the sources to be compensated and recorded by the receivers[1]; $\mathbf{y}_s = \mathbf{S}\mathbf{y}$ is the field created by the controlled emitters and recorded by the receivers[2], $\mathbf{v}$ is the signal recorded by the reference sensor; **P**

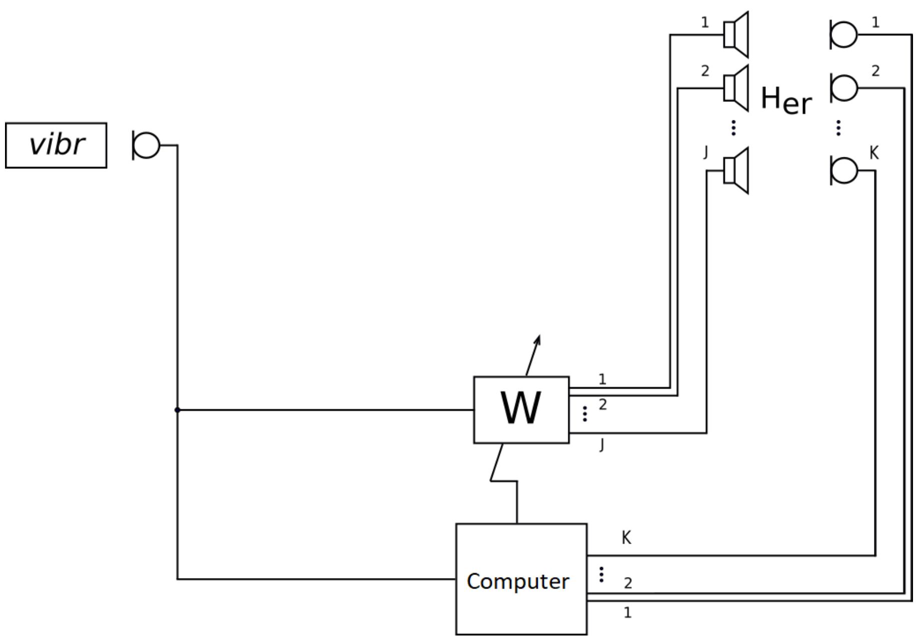

Figure 1. Structural diagram of an active noise cancellation algorithm.

[1] $\mathbf{y}_p = [K \times 1]$, $\mathbf{P} = [K \times I]$, $\mathbf{v} = [I \times 1]$.

[2] $\mathbf{y}_s = [K \times 1]$, $\mathbf{S} = [K \times J]$, $\mathbf{y} = [J \times 1]$.

and $\mathbf{S}$ are transfer matrices (at a fixed frequency) for paths from radiation sources to receivers and from controlled emitters to receivers, respectfully; $\mathbf{y} = \mathbf{Wv}$ are the signals applied to the controlled emitters[3], $\mathbf{W}$ is a matrix composed of the frequency responses of forming filters.

The total field recorded by the receivers can be written as

$$\mathbf{e} = \mathbf{y}_p + \mathbf{y}_s = \mathbf{Pv} + \mathbf{Sy} = \mathbf{Pv} + \mathbf{SWv}.$$

Let's write down the mean square of the residual field level at the receiving elements as

$$\begin{aligned}\mathbf{e}^H\mathbf{e} &= (\mathbf{y}_p + \mathbf{SWv})^H(\mathbf{y}_p + \mathbf{SWv}) \\ &= \mathbf{y}_p^H\mathbf{y}_p + \mathbf{y}_p^H\mathbf{SWv} + \mathbf{v}^H\mathbf{W}^H\mathbf{S}^H\mathbf{y}_p + \mathbf{v}^H\mathbf{W}^H\mathbf{S}^H\mathbf{SWv}.\end{aligned}$$

Add a constraint on the value of the calculated frequency response of shaping filters.

$$\begin{aligned}F &= \mathbf{e}^H\mathbf{e} + \alpha\,\mathrm{tr}(\mathbf{WW}^H) \\ &= \mathbf{y}_p^H\mathbf{y}_p + \mathbf{y}_p^H\mathbf{SWv} + \mathbf{v}^H\mathbf{W}^H\mathbf{S}^H\mathbf{y}_p + \mathbf{v}^H\mathbf{W}^H\mathbf{S}^H\mathbf{SWv} + \alpha\,\mathrm{tr}(\mathbf{WW}^H),\end{aligned}$$

where α is the regularization coefficient which is selected for a specific active noise cancellation system.

The minimum of F can be found by equating its derivative over $\mathbf{W}$ to zero, namely

$$\begin{aligned}\frac{\partial F}{\partial \mathbf{W}} &= \mathbf{S}^H\mathbf{y}_p\mathbf{v}^H + \mathbf{S}^H\mathbf{y}_p\mathbf{v}^H + \mathbf{S}^H\mathbf{SW}(\mathbf{vv}^H + \mathbf{vv}^H) + \alpha\mathbf{W} \\ &= 2\mathbf{S}^H\mathbf{y}_p\mathbf{v}^H + 2\mathbf{S}^H\mathbf{SWvv}^H + \alpha\mathbf{W} = 0.\end{aligned} \tag{1}$$

Let us introduce the following notations

$$\mathbf{A} = 2\mathbf{S}^H\mathbf{S},$$

$$\mathbf{B} = \mathbf{vv}^H,$$

$$\mathbf{C} = -2\mathbf{S}^H\mathbf{y}_p\mathbf{v}^H.$$

Equation (1) can be written in the form

$$\mathbf{AWB} + \alpha\mathbf{W} = \mathbf{C}. \tag{2}$$

The well known solution to the (2) is [15]

$$\mathrm{vec}(\mathbf{W}) = (\alpha\mathbf{I} + \mathbf{B}^T \otimes \mathbf{A})^{-1}\mathrm{vec}(\mathbf{C}),$$

where

$$\mathbf{A} \otimes \mathbf{B} = \begin{bmatrix} A_{11}\mathbf{B} & A_{12}\mathbf{B} & \dots & A_{1n}\mathbf{B} \\ A_{21}\mathbf{B} & A_{22}\mathbf{B} & \dots & A_{2n}\mathbf{B} \\ \vdots & & & \vdots \\ A_{m1}\mathbf{B} & A_{m2}\mathbf{B} & \dots & A_{mn}\mathbf{B} \end{bmatrix}, \quad \mathbf{A} = \begin{bmatrix} A_{11} & A_{12} \\ A_{21} & A_{22} \end{bmatrix}, \quad \mathrm{vec}(\mathbf{A}) = \begin{bmatrix} A_{11} \\ A_{21} \\ A_{12} \\ A_{22} \end{bmatrix},$$

$\mathbf{I}(n)$ stands for the identity matrix of size n, the symbol $\otimes$ denotes the Kronecker product.

When frequency responses of the shaping filters $\mathbf{W}$ are obtained, it is possible

[3] $\mathbf{y} = [J \times 1]$, $\mathbf{W} = [J \times I]$.

to determine the impulse responses $\mathbf{H}$ with the use of the inverse Fourier transform ($\mathbf{H} = \mathcal{F}^{-1}(\mathbf{W})$). Due to the fact that $\mathbf{W}$ remains constant (in the case when system parameters $\mathbf{S}$ и $\mathbf{P}$ are uncgenged) [9], using signal samples from reference sensors it is possible to implement a system that calculates $\mathbf{H}$ from recorded signals and generates voltages applied to controlled emitters. This method could be quite simply implemented by using a PC and a signal processing device connected to multichannel analog-to-digital converter (ADCs) and digital-to-analog converter (DACs). The purpose of the signal processing device in this case will be only the calculation of the convolution of signals from reference sensors with given impulse response (determined by $\mathbf{H}$).

It should be noted that for the functioning of the system, it is necessary to have estimates of the impulse response of acoustic propagation paths for each "controlled emitter-receiver" pair. As noted earlier, this requires the use of a separate evaluation procedure, or a calibration signal must be applied directly to the emitters during operation of the system.

In some cases, the impulse responses obtained in this way have a non-causal part. That means that the solution turns out to be physically unrealizable. To eliminate this effect, we propose a method consisting in nulling the non-causal part of the impulse responses $\mathbf{H}$ with iteratively updating the solution using samples of the residual noise. At the next iteration, the residual vibration samples are substituted into the expression of the objective function instead of the samples of the compensated signal y_p, that is

$$\boldsymbol{y}_p \leftarrow \boldsymbol{y}_p + \mathbf{S} * \mathbf{W} * \mathbf{v}$$

The calculated solution is added to the solution obtained at the previous iteration

$$\mathbf{H} \leftarrow \mathbf{H} + \mathcal{F}^{-1}(\mathbf{W}).$$

The method allows one to achieve a deeper level of noise suppression.

2.2. Evaluation of Operational Efficiency of the Proposed Algoritm Using a Simulation Model

The verification of the proposed method for implementing the active cancellation system was carried out on a simulation model. A fragment of an active noise suppression system for acoustic waves generated by vibroactive equipment was considered. There were used a planar array of controlled emitters and placed above it a planar array of receivers. The spacing of both arrays and distance between them was 0.2 m. The vibroactive equipment was simulated by a point sound source located near a rigid wall. **Figure 2** shows a fragment of the mutual arrangement of the cancellation system elements.

Noise generators operating in 500 - 1000 Hz frequency band were used as the source to be cancelled. $\mathbf{W}$ and $\mathbf{H}$ were calculated blockwisely and corrected using the residual field. The latter was recorded during the next signal block registration.

The mean values of the signal amplitudes, registered by residual field receivers, are shown in **Figure 3**. Note, that the level of acoustic field suppression

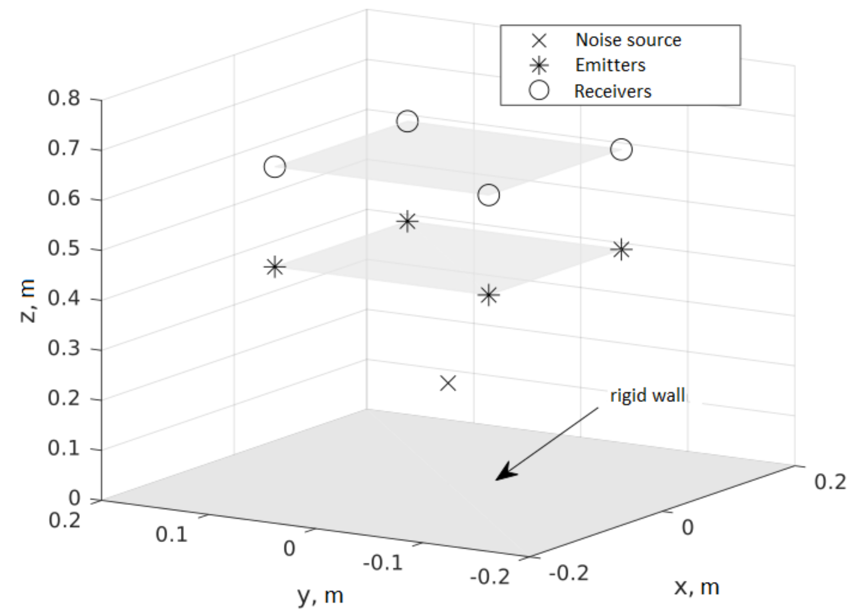

Figure 2. Relative arrangement of elements of active noise cancellation system.

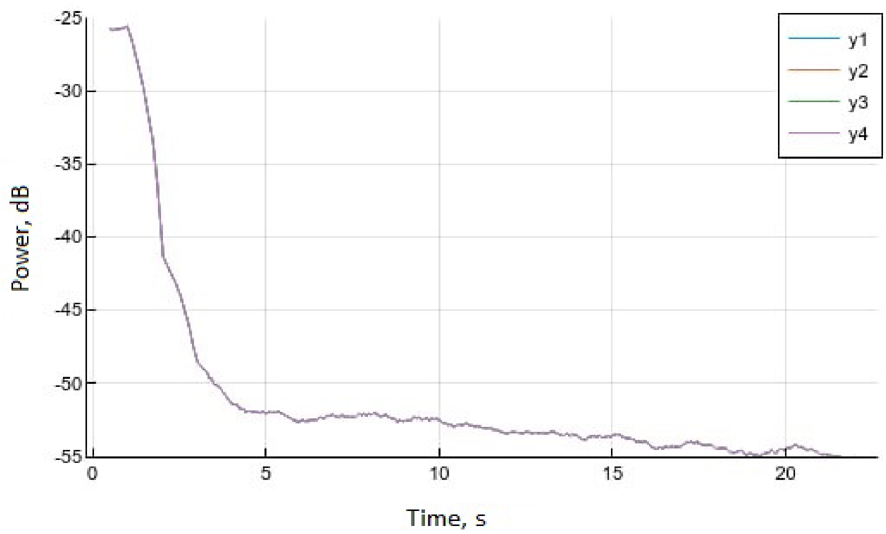

Figure 3. Mean values of signal amplitudes.

reaches 28 dB.

The efficiency of acoustic radiation suppression achieved by such a system in the far field is discussed in detail in [7].

2.3. Practical Implementation

Before proceeding to the description of the prototype of an active noise cancellation system, let consider its block diagram in **Figure 4**. There are schematically shown the source of compensated acoustic radiation ("src"), reference microphone, set of controlled emitters, residual field receivers and FIR shaping filters. The signal registered by the reference sensor is transmitted to a set of shaping filters, the coefficients of which are calculated by "Computer" using the values of the reference signal and the signal from the residual field sensors. Solid lines show the propagation paths of signals (electrical or digital if the system is computerized), and dotted lines show acoustic waves.

Figure 5 shows a photo of an experimental setup. At the bottom there is an acoustic radiation source, then the controlled emitters are located. Above the emitters there are microphones that record the residual field.

The distance between the elements was adjustable, which made it possible to implement various variants of geometries. It should be noted that the highest suppression was achieved with the geometry presented in **Figure 5**.

An experimental study of the proposed active cancellation system was carried out. 23 controlled emitters and 22 receivers were used during experiments.

In **Figure 6** power spectral densities of the original (blue line) and residual (yellow line) noise are presented. The difference between two power spectra

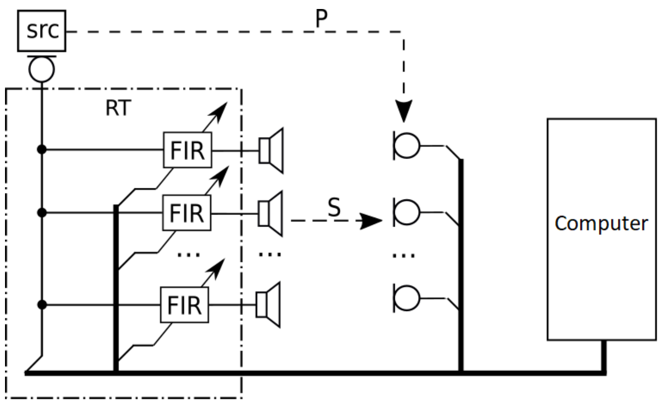

Figure 4. Structural diagram of active noise cancellation model.

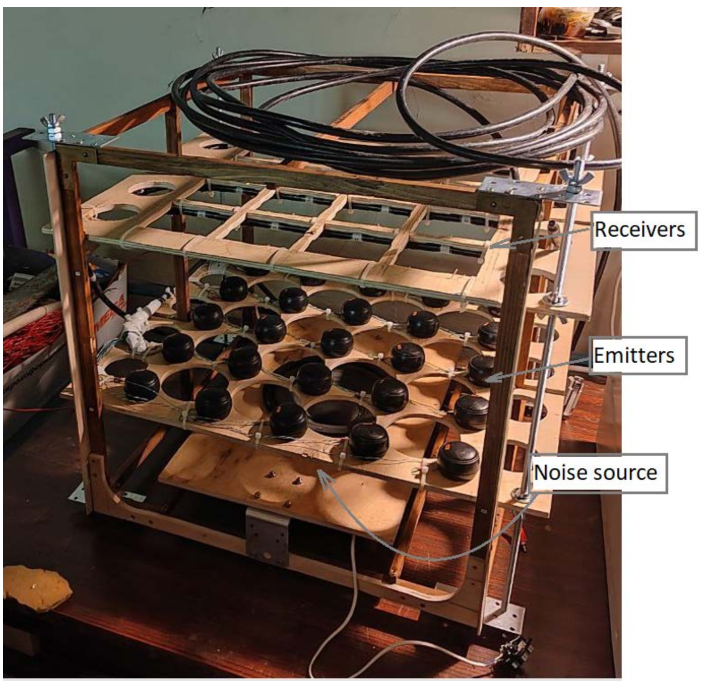

Figure 5. Experimental setup of an active noise cancellation system.

amounts to about 20 dB.

Figure 7 shows the time evolution of mean values of the amplitudes of the original and residual signals. We can notice a monotonic decrease of the amplitude values of the residual signal in the presented time dependence, which reflects the system adaptation procedure. In **Figure 8** and **Figure 9** the mean signal levels in receivers and the levels of suppression, achieved by the proposed system, are shown. We can see, that the signal levels drop by 17 - 20 dB when active cancellation in applied.

3. Conclusions

This paper studies the problem of construction of an adaptive active noise

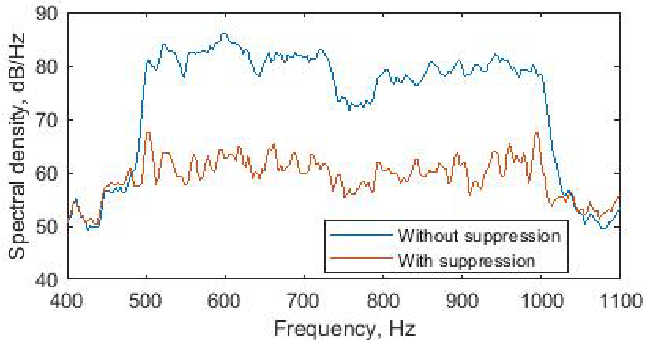

Figure 6. Power spectrums of the original (blue) and residual (red) noise.

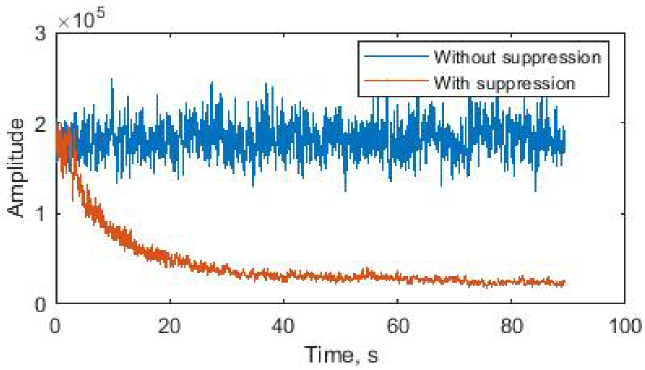

Figure 7. Mean values of amplitudes of the original (blue) and residual (red) noise.

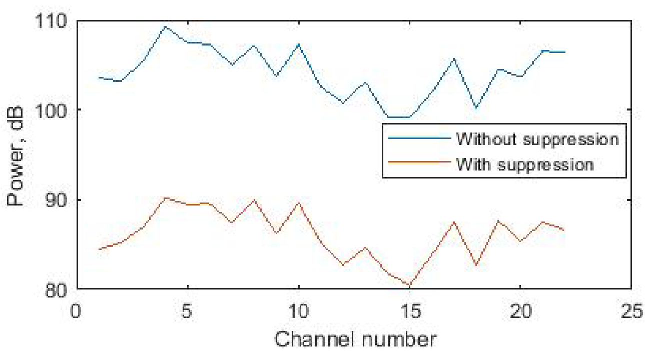

Figure 8. Signal power on the receiver channels.

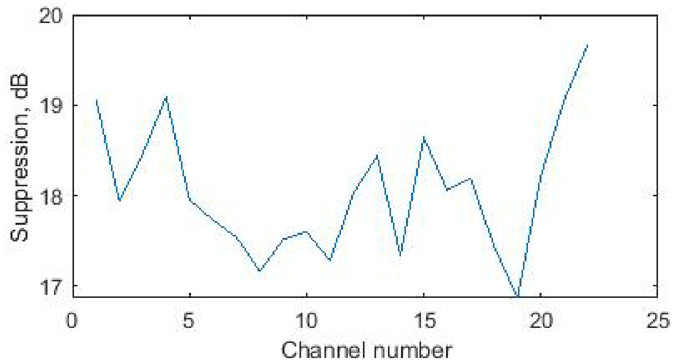

Figure 9. Suppression level in dB on a receiver channels.

cancellation system with an adjustment procedure in frequency domain. The technique is especially efficient for practical implementation of systems. With respect to the existing noise cancellation algorithms (e.g. [7] [9]), the proposed method enables to greatly decrease the resulting cost and complexity of a system. The results of computer simulations and experiments on a prototype have demonstrated the effectiveness of the proposed technique. It was shown that the suppression of wideband signals reaches 20 dB in experiments and 27 dB in numerical modeling.

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Conflicts of Interest

The authors declare no conflicts of interest regarding the publication of this paper.

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