

Experimental study on the effects of increasing number of system components and sampling rate of an active noise blocker for a tilted window

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ABSTRACT

To reduce the impact of noise entering the room through a tilted window active noise control in general and active noise blocking in particular is subject of current research. This concept uses several loudspeakers and microphones to generate sound pressure nodes. They prevent the transfer of acoustic energy. The loudspeakers and microphones are evenly distributed in the transmission path through the tilted window to achieve a global effect by a local measure. The system is driven by a multiple input multiple output feedforward filtered reference (x) least mean square (MIMO feedforward FxLMS) controller and implemented on a real time system. Using parallel computing on two processors with eight cores in total the number of loudspeakers and error microphones as well as the sampling rate can be increased. The idea behind these changes is to receive a better coverage of the gap and to increase the limit of controllable frequencies and to accelerate and stabilize the algorithm. This paper presents the results of an experimental study comparing the results of the adapted measurements to previous ones. Configurations with 2 reference microphones, 14 loudspeakers and 14 error microphones and with 2 reference microphones, 20 loudspeakers and 20 error microphones with different sampling rates have been investigated. It has been shown that for the 2x14x14 system an increased sampling rate has only minor impact. By contrast, the 2x20x20 configuration with more system components shows a significant improvement of noise reduction. On this case an increased sampling rate has additional impact.

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1. INTRODUCTION

Noise involves a burden that has negative effects on human health [1, 2]. To protect from noise effects during ventilation constructional measures are ineffective. Nevertheless, to receive a noise reduction active measures are necessary and are subject of current research [3].

One method to reduce noise entering the room through a tilted window is the so called Active Noise Blocker (ANB). Loudspeakers and microphones placed at the gap generate sound pressure nodes to prevent acoustic energy from entering the room. This local measure in the area of the window can achieve a global effect in the room behind the window. The proof of concepts has been shown in [4–7]

This paper presents a modified test bench. A study on the effects of the number of system components and an increased sampling rate shows the better performance of the system.

2. THEORETICAL BACKGROUND

The ANB is based on the MIMO feedforward FxLMS algorithm with leakage factor γ and normalized step size $\mu_{n,i}$ [8] to control the system. The corresponding block diagram can be seen in figure 1. Small bold characters denote vectors with the respective length N and capital bold characters denote matrices. The dashed line area describes the part of the system that is executed on the Real Time Operating System (RTOS).

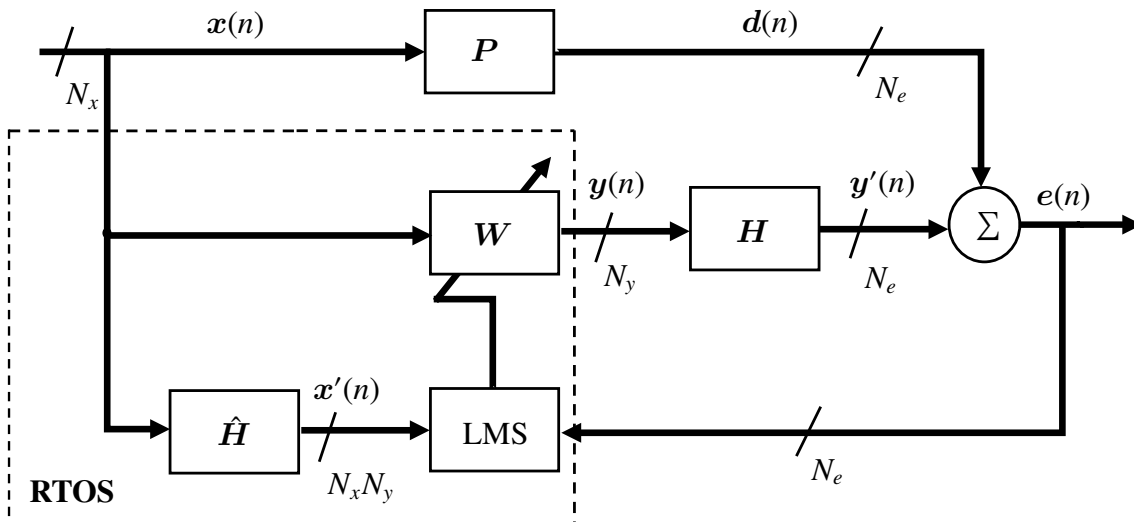


Figure 1: Block diagram of the ANB system with FxLMS algorithm

The block diagram is based on two paths. The primary path P describes the transmission path of the noise signal from the reference microphone $x(n)$ to the error microphone $d(n)$. The secondary path H describes the transformation of the output signal $y(n)$ to the error microphones. The objective function of the ANB is to minimize the error signal

$$e(n) = d(n) + y'(n). \quad (1)$$

In order to achieve this aim the controller calculates $j = 1 \dots N_y$ output signals

$$y_j(n) = \sum_{i=1}^{N_x} \mathbf{w}_{j,i}^T(n) \mathbf{x}_i(n) \quad (2)$$

at every discrete time step n on the basis of the reference signals and implicit of the error signals. $\mathbf{w}_{j,i}$ contains the N_w filter coefficients of an adaptive Finite Impulse Response (FIR) filter.

$$\mathbf{x}_i(n) = [x_i(n) \quad x_i(n-1) \quad \dots \quad x_i(n-N_w+1)]^T \quad (3)$$

denotes the last N_w values of the $i = 1 \dots N_x$ reference signal. The update of the filter coefficients is calculated by

$$\mathbf{w}_{j,i}(n+1) = (1 - \mu_{n,i}\gamma) \mathbf{w}_{j,i}(n) - \mu_{n,i}(n) \sum_{k=1}^{N_e} \mathbf{x}'_{i,j,k}(n) e_k(n) \quad (4)$$

where $k = 1 \dots N_e$ is the number of error signals and $\mathbf{x}'_{i,j,k}$ are the by the internal models $\hat{\mathbf{H}}$ of the secondary paths filtered reference signals

$$\mathbf{x}'_{i,j,k}(n) = \hat{h}_{k,j} * \mathbf{x}_i. \quad (5)$$

To optimize the speed of convergence the normalized step size

$$\mu_{n,i}(n) = \frac{\mu}{N_w \max[\epsilon, p_i(n)]} \quad (6)$$

is used where $p_i(n)$ is the estimated power and ϵ is a lower bound of the power. The update of the estimated power

$$p_i(n) = \beta p_i(n-1) + (1 - \beta) x_i^2(n) \quad (7)$$

is calculated by using the smoothing factor β to weight the square of the current reference value $\mathbf{x}_i(n)$.

3. EXPERIMENTAL SETUP

A transmission test bench consisting of an anechoic chamber and a reverberation room connected by an ordinary window is used to test the ANB system. The experimental setup can be seen in figure 2. The height of the anechoic chamber is 3570 mm and the height of the reverberation room is 2550 mm. The window has a width of 950 mm and a height of 1325 mm. The window is installed with a depth of 200 mm towards the anechoic chamber and 375 mm above the ground.

The location of the measuring equipment is shown in figure 3. The primary noise is generated by a subwoofer (Klein+Hummel - PRO X SUB L) and a loudspeaker (Klein+Hummel - Pro X 12/80). The secondary loudspeakers (MONACOR - NUMBER ONE SPH-100C) are integrated in cubes made of wooden plates with an edge length of 140 mm each. They are evenly located around the frame of the window (see figure 4). The error microphones (Brüel & Kjær - Type 4958) are located directly behind the secondary loudspeakers. The loudspeakers are connected to three amplifier (IMG - StageLine STA-1508) to operate the system. Analog filters (I.E.D. - S16-OM) are used for anti-aliasing. The corner frequency is set to $f_{c,i} = 2$ kHz for a sampling rate $f_s < 8$ kHz.

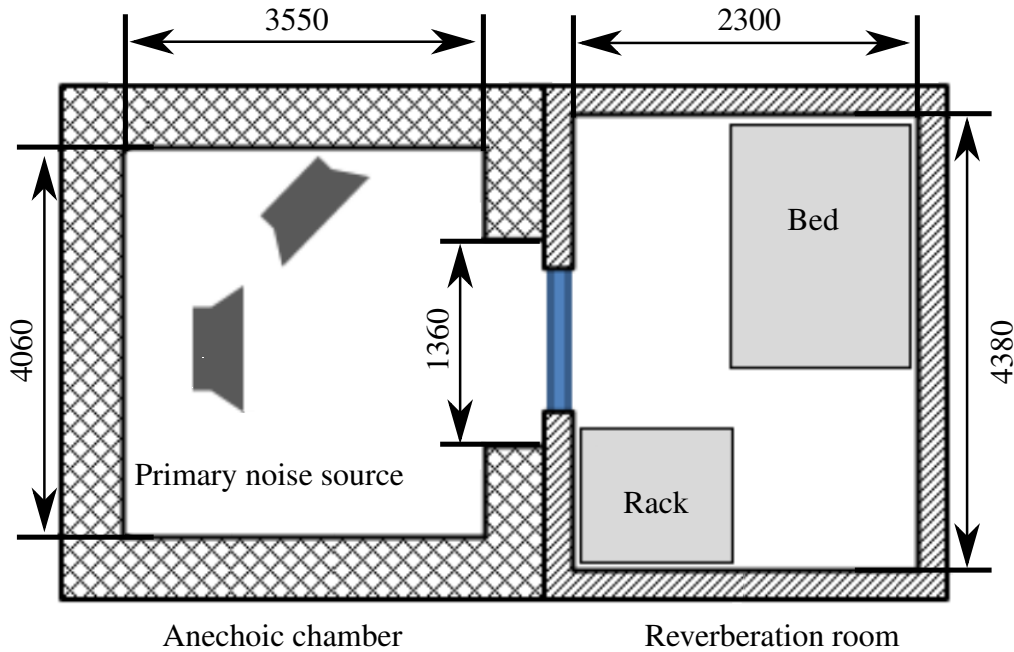


Figure 2: Floor plan of the transmission test system [9]

and $f_{c,i} = 4$ kHz otherwise. The reconstruction takes place with filters of the same type with $f_{c,o} = 1$ kHz.

The controller is executed on two modular real time systems (dSpace - Type SCALEXIO) with a processor board DS6001 and five additional input/output-modules DS2655M1 each. The processor board DS6001 includes an Intel®Core™i7-6820EQ quadcore processor with a clock rate of 2.8 GHz, 4 GB RAM memory and 8 GB flash memory.

12 microphones (Brüel & Kjær - Type 4935) are positioned near the corner of the reverberation room for the evaluation (see figure 5). A frontend (Brüel & Kjær - Type 3053-B120) is used for real time data analysis.

4. RESULTS

In the course of this work experiments have been conducted in which the number of used loudspeakers and error microphones as well as the sampling rate have been varied in order to investigate their influence on the ANB-process. The investigated configurations and the controller parameters are presented in table 1. Firstly a configuration consisting of 2 reference microphones, 14 loudspeakers and 14 error microphones (2x14x14) has been analyzed with sampling rates f_s between 6 kHz and 11 kHz in steps of 1 kHz. Secondly a configuration with 2 reference microphones, 20 loudspeakers and 20 error microphones (2x20x20) has been analyzed for $f_s = 6$ kHz and $f_s = 8$ kHz. The arrangement of the loudspeakers and error microphones can be seen in figure 4. The lengths of the adaptive filters N_w and the secondary path models N_h have been adjusted according to the adjustment of the sampling rate to receive impulse responses of the same length. To avoid an overflow of the RTOS of the 2x14x14 configuration with $f_s = 11$ kHz and the 2x20x20 configuration with $f_s = 8$ kHz the lengths of the adaptive filters and due to that the length of their impulse responses have been reduced. The other controller parameters from table 1 remain unchanged for all investigated cases.

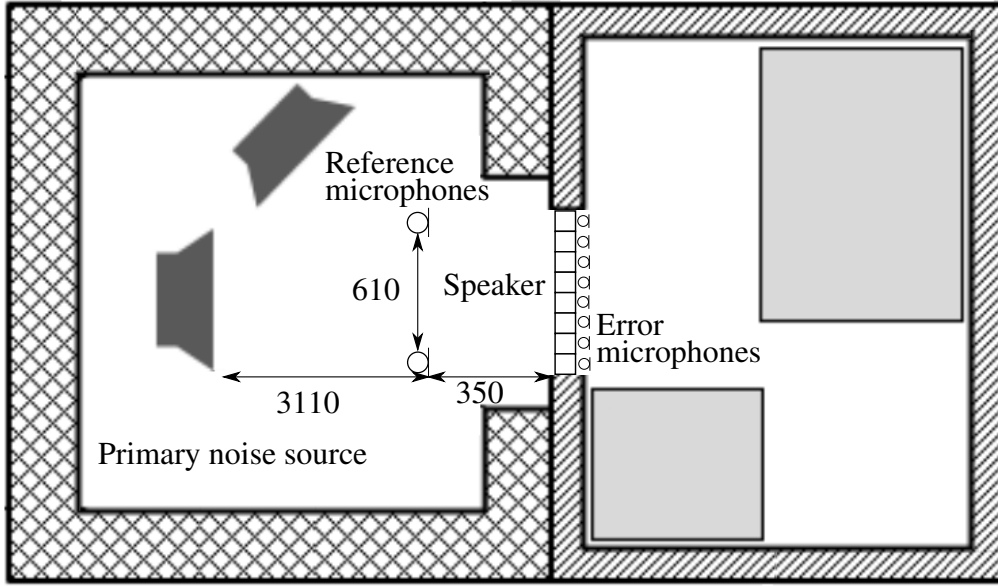


Figure 3: Location of measuring equipment [9]

Table 1: Controller parameters

N_x	N_y	N_e	f_s	N_w	N_h	γ	μ	ϵ	β
[-]	[-]	[-]	[kHz]	[-]	[-]	[-]	[-]	[-]	[-]
2	14	14	6	256	100	$1e^{-5}$	$1e^{-5}$	$1e^{-4}$	0.99
			7	299	117				
			8	341	133				
			9	384	150				
			10	426	167				
			11	396	183				
2	20	20	6	256	100	$1e^{-5}$	$1e^{-5}$	$1e^{-4}$	0.99
			8	320	133				

Band limited white noise between 100 Hz and 1 kHz is used for excitation. Using the software Pulse LabShop the evaluation takes place up to 16.3 kHz. Because above 3 kHz the auto-power spectrum level is continuous below 10 dB the presented frequency is limited between 1 Hz and 3 kHz in a logarithmic scale.

Figures 6 to 9 present the mean of the auto-power spectrum level of all monitor microphones of the uncontrolled and controlled noise for four selected examined cases. The obtained results for the noise reduction are presented in table 2.

For configuration 2x14x14 and $f_s = 6$ kHz the reduction is basically between 100 Hz and 500 Hz although the excitation is up to 1 kHz. This results in a noise reduction of 5.5 dB in the range from 100 Hz to 1 kHz and 8.4 dB in the limited range from 100 Hz to 500 Hz.

In comparison with that configuration 2x14x14 for $f_s = 10$ kHz behaves similar but obtains a minor better noise reduction of 5.7 dB for 100 Hz to 1 kHz and 9.2 dB for 100 Hz to 500 Hz.

All tests of configuration 2x14x14 show similar behaviour. The observed behaviour of a minor increase for the range of 100 Hz to 500 Hz shows no evidence for $f_s = 11$ kHz.

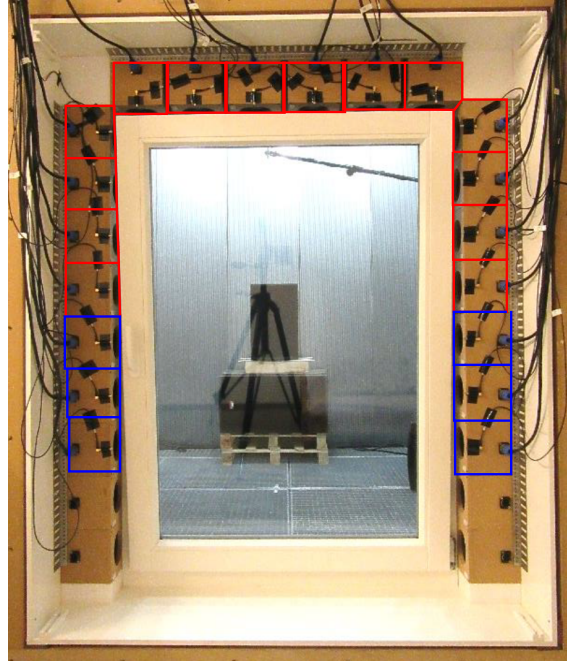


Figure 4: Arrangement of secondary sources; Configuration 2x14x14 (red), Configuration 2x20x20(red+blue) [9]

Table 2: Noise reduction for different Configurations and sampling rates

Configuration	f_s [kHz]	Total [dB]	100 Hz-1.000 Hz [dB]	100 Hz-500 Hz [dB]
2x14x14	6	5.5	5.5	8.4
	7	5.4	5.5	8.2
	8	5.5	5.6	8.5
	9	5.3	5.4	8.8
	10	5.7	5.7	9.2
	11	5.8	5.9	9.1
2x20x20	6	8.3	8.4	9.0
	8	9.8	10.0	10.6

One possible explanation could be the length reduction of the impulse response of the adaptive filter.

The second investigated configuration (2x20x20) with $f_s = 6$ kHz shows noise reduction over almost the entire excited bandwidth. This results in a noise reduction of 8.4 dB for the range of 100 Hz to 1 kHz and only a slightly better noise reduction of 9.0 dB for 100 Hz to 500 Hz.

Finally configuration 2x20x20 shows for $f_s = 8$ kHz noise reduction over almost the entire excited bandwidth, too. In comparison to $f_s = 6$ kHz there is a significant increase. The noise reduction for the range of 100 Hz to 1 kHz is 10.0 dB and for 100 Hz to 500 Hz even 10.6. The length reduction of the impulse response of the adaptive filter shows no effect or is overlaid by the even higher significant increase of the noise reduction.

The improvement of the 2x20x20 configuration compared to the 2x14x14 configuration is due to the fact that the gap of the window and thereby the higher

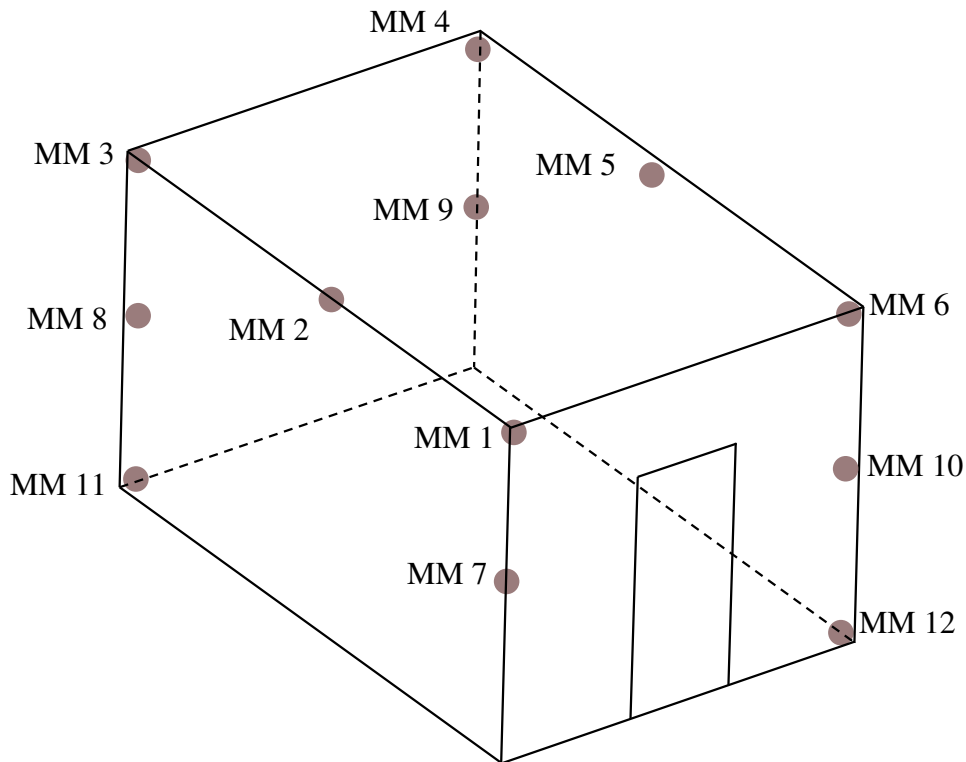


Figure 5: Position of the monitor microphones [9]

frequencies are better covered.

Figures 6, 8 and 9 show an irregular interfering signal for 17 Hz from the outer structure. It is possible that this signal influences the algorithm and effects the result of noise reduction.

Hanselka et al. [7] did experiments with the ANB system under similar conditions but with a modified test bench. They investigated a significant smaller window with a width and height of 900 mm. Using a configuration with 1 reference signal, 14 error microphones and 8 loudspeakers they covered the gap at its widest with 4 loudspeakers. This results in a ratio of number of loudspeakers to the length of the gap of 0.004 m^{-1} while in this paper a ratio of 0.006 m^{-1} is presented. Furthermore they handed the reference signal directly over to the algorithm. In this paper the reference signal is recorded by 2 microphones.

Hanselka et al. presented a noise reduction of 13 dB in the range from 100 Hz to 1 kHz. In comparison with them the in this paper presented results are worse in spite of more used components. The most likely reason is that they investigated a significant smaller window and due to that a smaller gap that is easier to control. The different type of handing the reference signal might also have an impact.

5. ACKNOWLEDGEMENT

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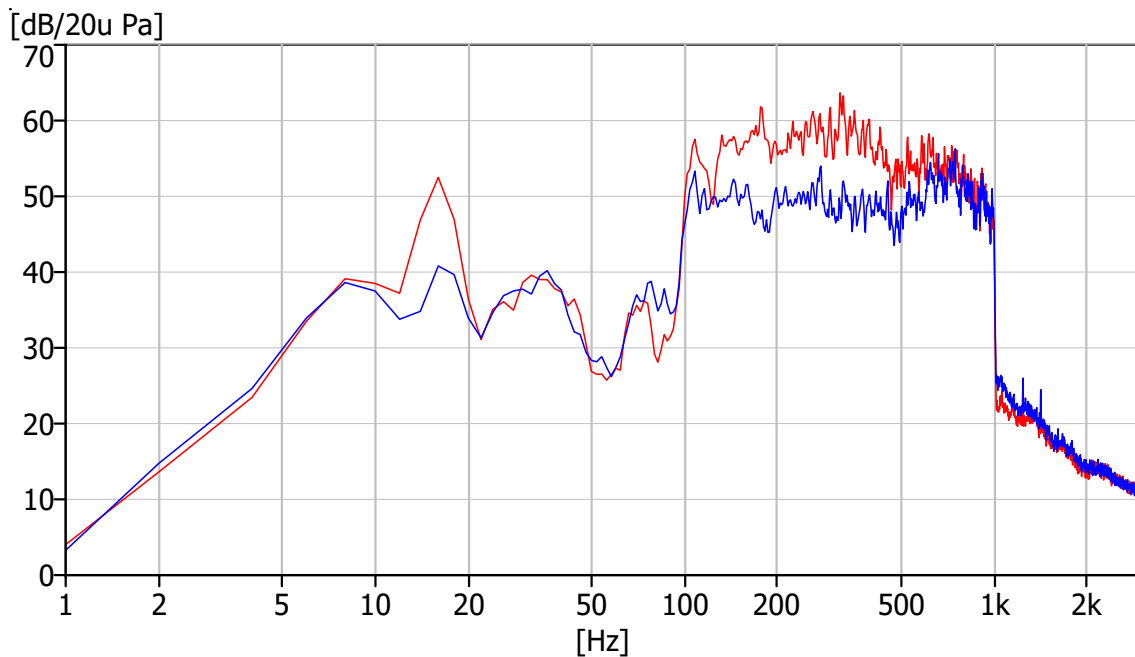


Figure 6: Auto-power spectrum level in uncontrolled (red) and controlled (blue) state for the configuration 2x14x14 and 6 kHz sampling rate

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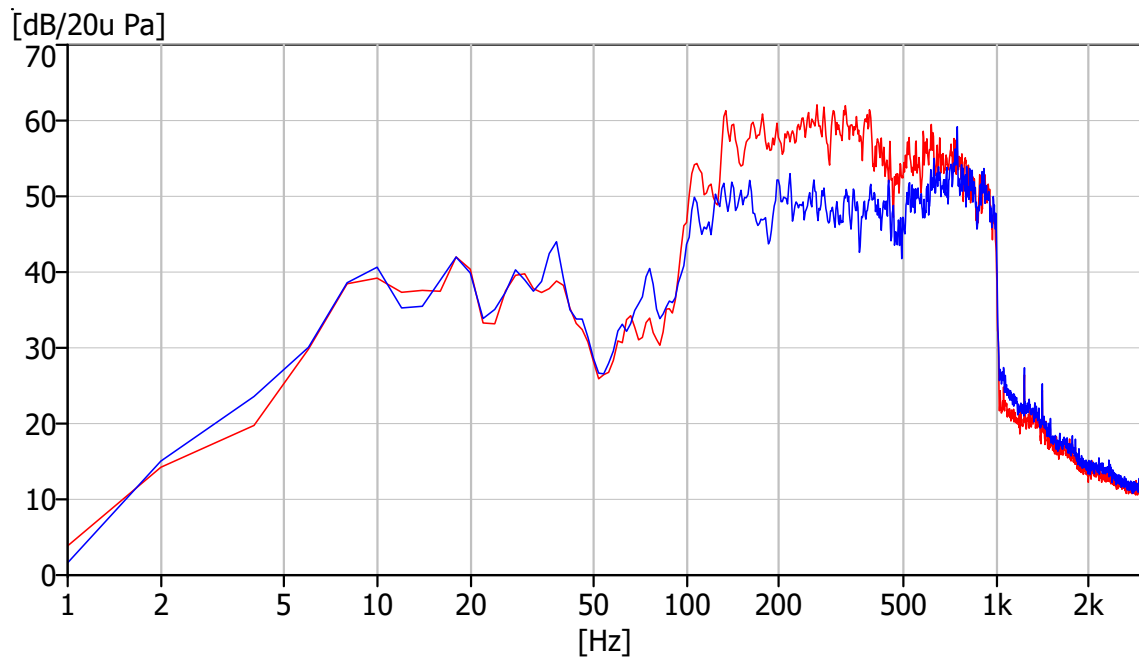


Figure 7: Auto-power spectrum level in uncontrolled (red) and controlled (blue) state for the configuration 2x14x14 and 10 kHz sampling rate

[9] Max Lorenzen. Set-up, adjustment and first operation of an active noise blocker at a tilted window, 2018. MA thesis, Helmut Schmidt Universität - Universität der Bundeswehr.

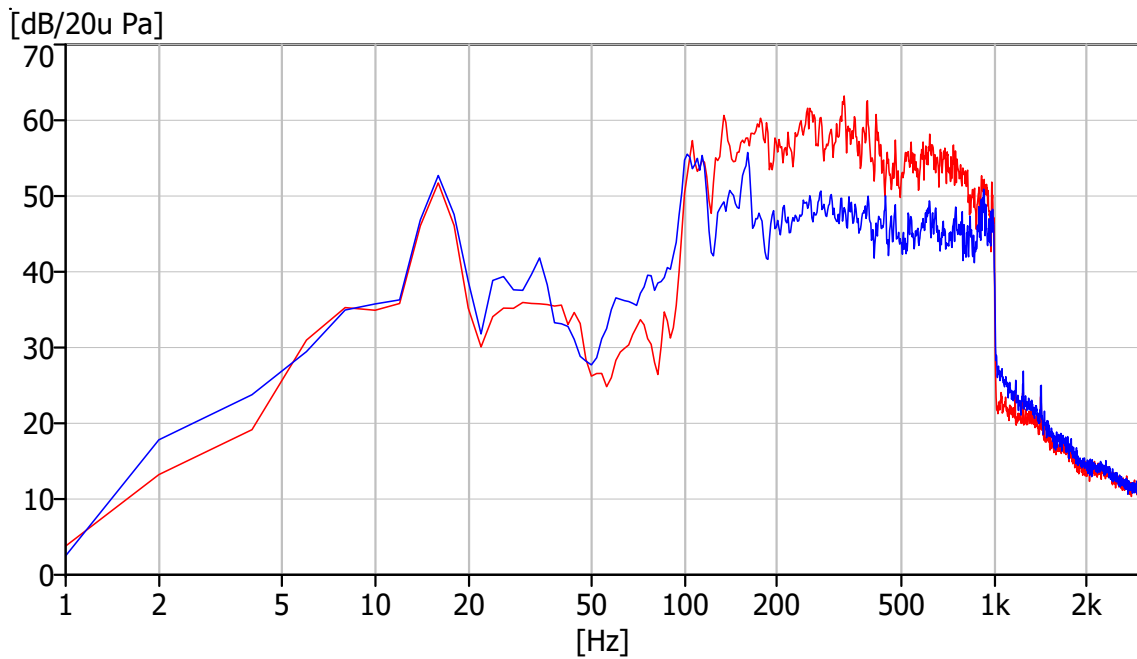


Figure 8: Auto-power spectrum level in uncontrolled (red) and controlled (blue) state for the configuration 2x20x20 and 6 kHz sampling rate

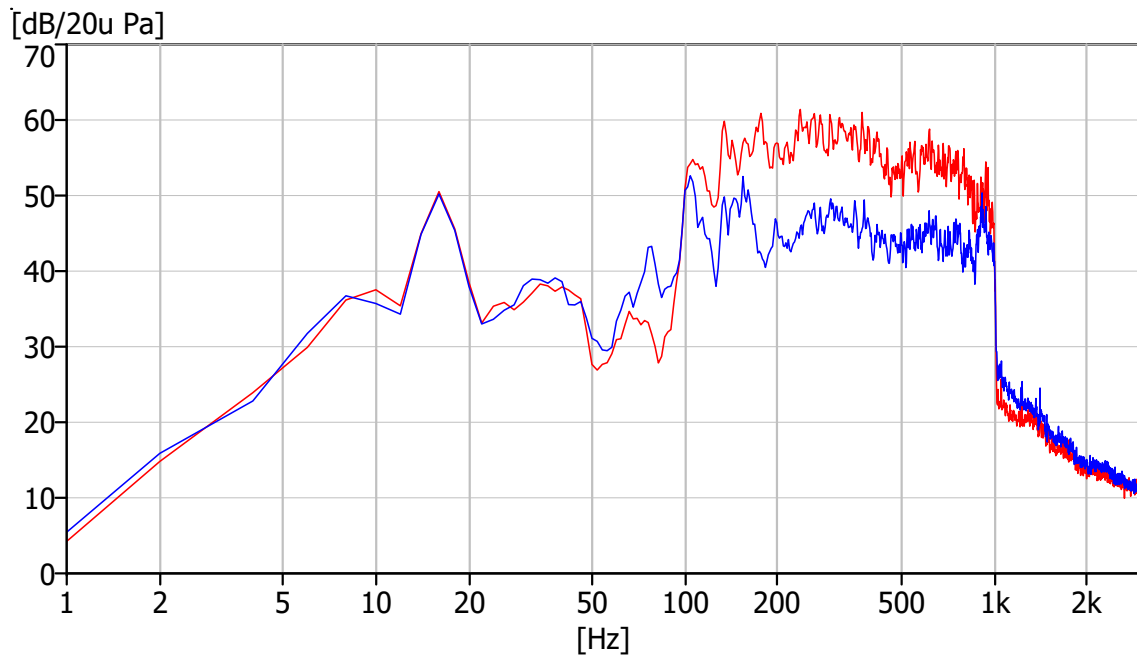


Figure 9: Auto-power spectrum level in uncontrolled (red) and controlled (blue) state for the configuration 2x20x20 and 8 kHz sampling rate