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RAPIDLY CONVERGING MULTICHANNEL CONTROLLERS FOR BROADBAND NOISE AND VIBRATIONS

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Abstract
Applications are given of a preconditioned adaptive algorithm for broadband multichannel active noise control. Based on state-space descriptions of the relevant transfer functions, the algorithm uses the inverse of the minimum-phase part of the secondary path in order to improve the speed of convergence. A further improvement of the convergence rate is obtained by using double control filters for elimination of adaptation loop delay. Regularization was found to be essential for robust operation. The particular regularization technique preserves the structure to eliminate the adaptation loop delay. Depending on the application at hand, a number of extensions are used for this algorithm, such as for applications with rapidly changing disturbance spectra, applications with large parametric uncertainty, applications with control of time-varying acoustic energy density.

Keywords: active noise and vibration control, adaptive control, energy density control, time-varying systems.

1 Introduction

Active control of broadband noise is considered to be significantly more complex than active control of tonal noise, especially for multichannel systems. In this paper we consider adaptive controllers which are able to adapt with respect to a changing spectrum at the error sensors, i.e. the disturbance spectrum. It is assumed that the transfer function between the actuators and the error sensors, i.e., the secondary path, remains constant. If the disturbance spectrum changes rapidly, then most algorithms such as the filtered reference least-mean-square algorithm or the filtered-error least-mean-square algorithm may not be able to follow and will lead to higher residuals. Such algorithms may lead to rather high bias errors and disappointing noise reductions even in stationary situations. The performance criteria of the algorithms which presently are of particular interest are the speed of convergence and tracking behavior. Several causes are known which lead to a deteriorated convergence speed and tracking speed. These causes can be grouped as follows: 1) the characteristics of the secondary path, 2) the characteristics of the reference signal(s) and 3) the structure of the controller. The former two determine the eigenvalue spread of the correlation matrix of the filtered reference signals. A large eigenvalue spread leads to a small maximum convergence coefficient and therefore slow convergence. The following characteristics of the secondary path lead to an increased eigenvalue spread: a colored secondary path transfer function, interaction between the individual transfer functions in a multichannel system, and non-minimum-phase delays in the secondary path. For the reference signals, the eigenvalue spread deteriorates if the spectrum is colored and also if the individual reference signals are correlated. The control structure can influence the convergence properties if for example a delay is present in the adaptation path, such as in the filtered-error algorithm. Such a delay also leads to reduced convergence speeds. Adaptive algorithms should take into account one or more of the degrading characteristics described above if rapidly changing disturbance spectra are to be controlled. In this paper, we present some applications for a preconditioned version of the filtered-error algorithm. The basic algorithm incorporates a preconditioning step for the secondary path. This preconditioning step improves the eigenvalue spread of the correlation matrix computed from the filtered reference signals. Depending on the application at hand, specific modifications are added in order to improve relevant performance measures.

2 Basic algorithm

The basic algorithm for active noise and vibration control as considered here consists of a fully coupled multiple-input multiple-output feedforward control algorithm (see Figure 1). This algorithm is based on the so-called preconditioned lms algorithm as derived by Elliott [1]. The preconditioned lms algorithm is able to use multiple reference signals in an efficient way, since it is based on the filtered-error algorithm instead of the filtered reference algorithm. However, the practical implementation of the present algorithm differs from the description given in Ref. [1]. Important differences exist in the structure of the algorithm and also in its numerical implementation. The structure of the present algorithm is different in the sense that it compensates for delays in the adaptation loop [3], provides possibilities for frequency dependent regularization [3], and uses on-line whitening of the reference signals [7]. Furthermore, the present algorithm is mostly based on efficient state-space descriptions instead of finite impulse response descriptions. An advantage of state-space descriptions is that for many interesting computations numerically reliable methods exist, such as for the MIMO minimum-phase/all-pass factorization as used in the present algorithm.

The present algorithm has been termed regularized modified filtered-error algorithm [3]. The algorithm uses a state-space preconditioner G_o^{-1} in combination with a double set of control filters W in order to eliminate adverse effects on convergence of phase distortion and delays in the adaptation loop [3]. Such an implementation is made possible by using the state-space based description of relevant transfer functions, for which efficient and robust decomposition techniques exist. Regularization has been implemented in various parts of the algorithm, with the most important being the generalized, possibly frequency dependent weighting [3]. The algorithm has a number of advantages as compared to the standard filtered-error algorithm [4], such as considerably improved convergence speed, improved stability and robustness, reduced computational complexity, and, for some applications, improved noise reductions since it is less vulnerable to noise because of improved numerical

conditioning. A disadvantage is that the complexity to program the algorithm is considerably higher than the standard filtered-error algorithm. Nevertheless, the algorithm has been implemented successfully for both feedforward and feedback configurations.
The basis of the hardware for the control system is an Intel Mobile processor with a Linux operating system. Real-time code for this platform is generated using Matlab and Simulink. The control system has 16 analog inputs and 16 analog outputs. The sampling rate is nominally 100 kHz. Multi-stage decimators and multi-stage interpolators are implemented on FPGA in order to realize lower sample rates. This hardware enables setting the desired sample rate, but also the desired compromise between reconstruction errors, sampling errors, and phase distortion of the control loop, since the coefficients of the interpolation filters and decimation filters are fully programmable.

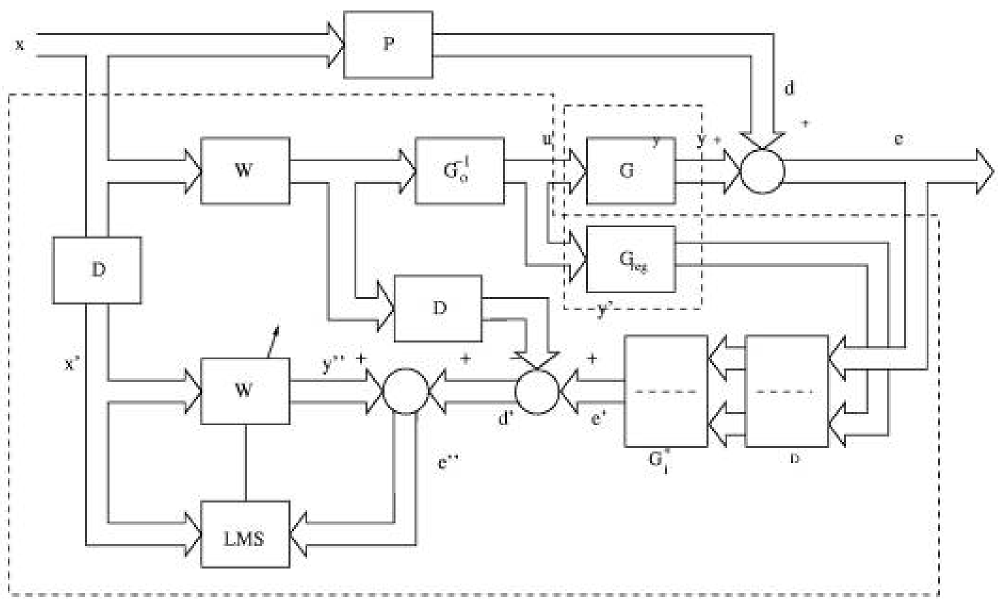

Figure 1 – Block diagram of the regularized modified filtered-error algorithm for adaptive broadband multichannel feedforward control [3].

3 Application dependent modifications of the algorithm

3.1 Fast tracking vibration control at high sample rates

The first example application of the control system is for active vibration reduction on a setup with a vacuum pump that is tightly coupled with high-precision equipment. The precision of this equipment is critically dependent on the level of the vibrations that are introduced by the vacuum pump. The vibrations were reduced by the adaptive control scheme in a multi-input multi-output feedback configuration. The convergence properties of this algorithm had to allow effective tracking of the varying excitation spectrum, in particular a time-varying disturbance near 5 kHz. Programmable digital minimum-phase reconstruction and antialiasing filters at a sampling rate of 100 kHz were used for an optimal tradeoff between sampling errors and phase shift. Effective broadband control was obtained in the frequency range of 100 Hz to 5 kHz, leading to 11.3 dB average broadband reduction on the error sensors (see Figure 2). The algorithm minimizes the variance of the error signals using a feedback architecture based on Internal Model Control (IMC), in which the estimated contribution of the actuators on the sensors is subtracted. The sample rate was set to 12 kHz and the order of the identified state-space model to 80. The controller had 100 coefficients for each actuator-sensor combination, resulting in a total of 900 controller coefficients. The length of the all-pass part of the transfer function was 150 samples. The regularization was based on the mean-square value of the control signals. The magnitude of the pertinent transfer function responsible for the regularization was set to approximately 40 dB below the maximum of the transfer function for the secondary path.

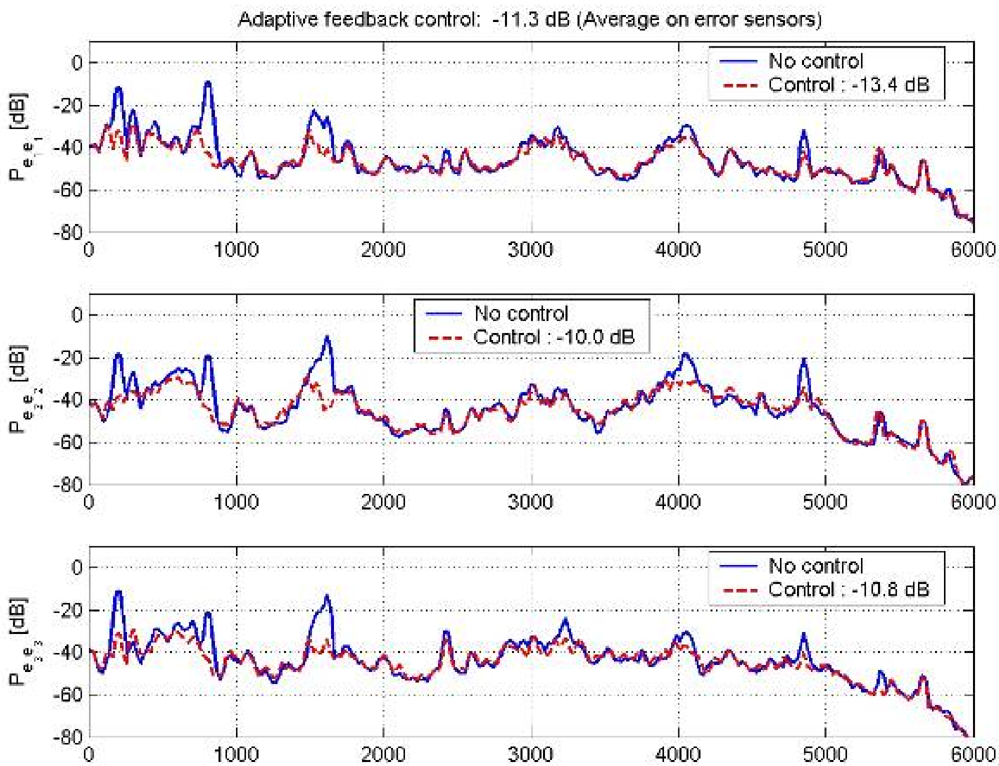

Figure 2 - Spectra of the sensor signals as measured on the setup for the turbomolecular vacuum pump, without control and with control using an adaptive feedback algorithm.

3.2 High authority – Low authority control

The multiple-input multiple-output adaptive controller as described before provides improved performance over traditional adaptive algorithms such as the filtered reference lms algorithm and the filtered-error lms algorithm. The most significant performance improvements are in terms of speed of convergence, the amount of reduction, and stability of the algorithm. Nevertheless, if the error in the model of the relevant transfer functions becomes too large then the system may become unstable or lose performance. On-line adaptation of the model is possible in principle but a large amount of additional noise should be injected in the system if rapid changes are to be followed. It has been known for decades that a combination of high-authority control (HAC) and low-authority control (LAC) could lead to improvements with respect to parametric uncertainties and unmodeled dynamics. In this particular application a full digital implementation of such a control system is presented in which the HAC (adaptive MIMO control) is implemented on a CPU and in which the LAC (decentralized control) is implemented on a high-speed Field Programmable Gate Array. Experimental results are given in which it is demonstrated that the HAC/LAC combination leads to performance advantages in terms of stabilization under parametric uncertainties and reduction of the error signal.

The HAC/LAC architecture was tested on a piezoelectric panel for reduction of noise transmission. Five piezoelectric patch actuators and five piezoelectric patch sensors were attached to the panel [6]. The panel was built from two Printed Circuit Boards (PCBs) with a honeycomb layer in between. One advantage of this approach is that electronics can be integrated. Another advantage is that the actuator and the sensor can be placed on different faces of the panel, which improves the control of the acoustically relevant out-of-plane vibrations because the in-plane coupling between the actuator and the sensor is reduced [7]. In addition, five collocated accelerometers were used for active damping with decentralized control using the piezoelectric patch actuators.

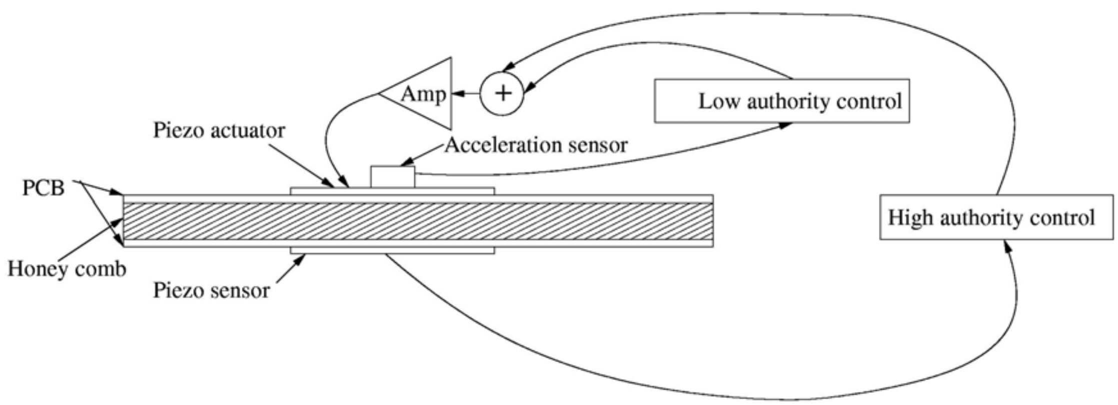

Figure 3 - Configuration of the high-authority/low-authority control architecture applied to a sandwich panel with piezoelectric patch actuators, piezoelectric patch sensors and accelerometers.

In many publications, active damping is realized using an analog controller (see e.g. Ref [8]). One of the advantages of an analog controller is its low delay when compared to a digital controller. The particular implementation as used for the setup of Fig. 3 [6] uses a digital controller with a high-sample rate that approximates an analog controller, such that the analog and digital controllers have identical performance for frequencies within the control bandwidth. The existing control architecture provided a suitable environment for realization of such a high speed digital feedback controller. The analog interface board already contained

ADDA converters operating at a sample rate of 100 kHz while the actual RMFeLMS algorithm was operating at a lower sample rate, in this case 2 kHz. This made it possible to run the LAC and HAC controller at different sample rates. The interface between the PCI-104 system running the HAC algorithm and the ADDA unit was implemented in reconfigurable hardware, in this case a Field Programmable Gate Array (FPGA) [2]. The FPGA incorporates the following functional units: the decimation filters, the interpolation filters, the glue logic for the PCI bus interface and the low-authority controller. The decimation filters and interpolation filters were designed in such a way that the desired compromise between group-delay and stopband attenuation was obtained.

Control results for a HAC/LAC architecture using an adaptive MIMO feedforward algorithm are shown in Figure 4. For this result, the regularization parameter beta [6] was set to -30 dB, and the convergence coefficient [6] to alpha = 1/40, which was the value that resulted in the highest reduction at 60 seconds after start of the convergence. It can be seen that the reduction of the error signals for MIMO control (HAC) is considerably higher than for the decentralized control (LAC). However, the combination of HAC and LAC leads to the largest reduction of the error signals. The average improvement by adding the low-authority controller to the high-authority controller is approximately 4.4 dB.

A subsequent set of tests was performed to study the influence of LAC on the robustness of the controller. The robustness was evaluated by adding different weights to the piezoelectric panel. From experiments it became clear that for low frequencies the frequency response of the secondary path is less sensitive to the addition of mass when the low-authority controller is switched on. Therefore it was expected that the robustness of the present high-authority controller would benefit from the addition of the low-authority controller since the robustness of the adaptive high-authority controller is primarily determined by the phase of the secondary path [1]. Indeed, the robustness of the adaptive controllers improves substantially by the addition of the low-authority controller. An interesting observation was that the robustness of the adaptive feedback controller also increased if the model was obtained with LAC switched on but for which, during control operation, the LAC was switched off. Apparently, the reduced phase in the model itself is beneficial for the robustness of the controller.

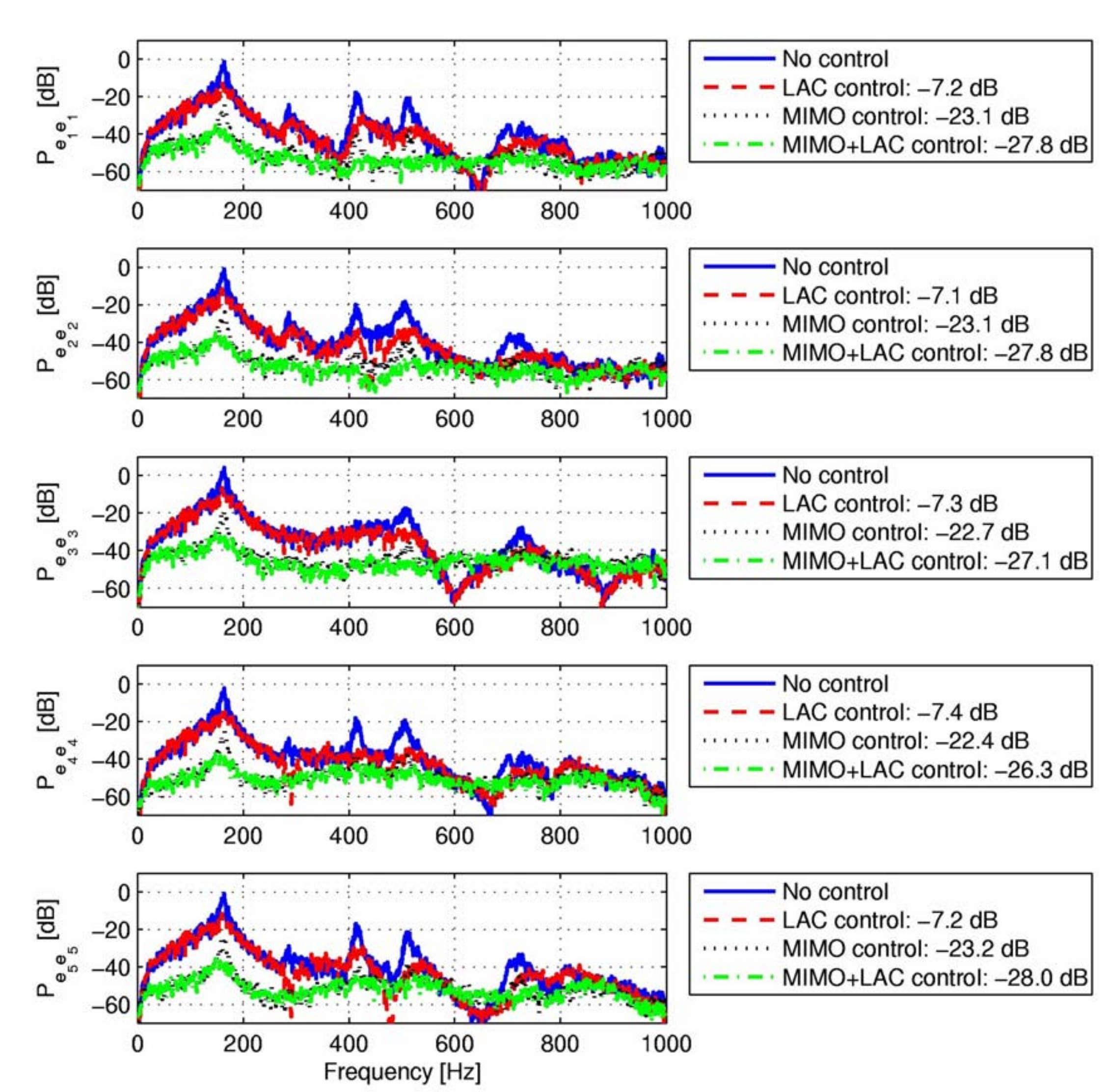

Figure 4 - Performance of a MIMO feedforward controller (HAC) with and without low-authority control (LAC) ; the graphs show spectra of the five signals as measured by the controller for control off (blue solid line) and control on (LAC: red dashed line, HAC: black dotted line, HAC/LAC: green dash-dotted line)

3.3 Time-varying acoustic energy density control

The previous applications discussed in this paper dealt with active vibration control. In the last application described in this paper the objective is to reduce the local noise using acoustic sources and acoustic sensors. In particular, the focus is on the reduction of broadband low-frequency noise in rooms at the lowest audible octaves. Different control strategies are possible in principle but the strategy discussed in this paper is based on feedforward control using a separate acoustic sensor exterior to the room of interest. The acoustic error sensor uses the acoustic pressure and the three components of the particle velocity. Therefore, the error sensor comprises four more or less independent sensor signals. The frequency response of these sensors is not flat, especially for very low frequencies. Therefore, preprocessing filters were implemented in order to correct for the deviations from the ideal flat frequency response. Using the resulting error signals, the size of the quiet zone can be increased significantly as compared to the use of a single error sensor such as a microphone [9,10]. In order to be able to control the sound field with the required degrees of freedom, multiple loudspeakers were used. Normally, loudspeakers operating at low frequencies are quite big because naturally low cutoff frequencies are only possible with large and heavy transducers and large enclosures. For the present application, special preprocessing techniques were developed that lead to a reasonably flat frequency response from about 10 Hz to several 100 Hz using loudspeakers of modest size. These preprocessing techniques were found to be beneficial for the overall performance of the control system.

Additional signal processing was required to reduce reconstruction errors to inaudible levels. The control system as described before has rather nice facilities for implementation of programmable multistage interpolation and decimation filters on FPGA. Nevertheless, for the low sample rate of 500 Hz as used here, combined with the large difference in sensitivity of the ear for different frequency ranges (cf. the A-weighting), extremely steep FIR filters with many coefficients were needed. In fact, so many filter coefficients were needed that most numerical routines to design the filters became unstable. In order to alleviate the task of the reconstruction filters, additional low-order low-pass IIR filters running at a relatively high sample rate of 100 kHz were added. The IIR filters were implemented in FPGA. Also the loudspeaker correction filters, as discussed before, were realized in FPGA. Both the IIR reconstruction filters and the loudspeaker correction filters require a high sample rate in order to avoid audibility of reconstruction errors. For the loudspeaker correction filters this becomes essential when the amount of correction needed is substantial, in particular when flat very low frequency response is required from loudspeaker enclosures of modest size. In this case the amount of amplification at low frequencies is directly proportional to the amount of reconstruction noise at higher frequencies. Figure 5 shows results of the control of time-varying broadband noise due to a transportation vehicle. The graph shows four signals which correspond to acoustic pressure and the three components of the particle velocity. The average reduction in this case is 11.0 dB. The frequency range in which the reduction is obtained is approximately 25 Hz to 100 Hz. This reduction was clearly audible. Due to the use of the energy density sensor, this reduction was obtained in a relatively large area, at least sufficiently large to allow significant movement of the head without noticeable degradation of acoustic performance.

In the introduction of this paper it was stated that colored reference signals lead to an increased eigenvalue spread of the correlation matrix of the filtered-reference signals, which possibly leads to reduced convergence rates. In this application, the reference signal is colored which possibly leads to deteriorated convergence rates if a standard gradient-descent algorithm is used. In this application, convergence rates are increased by using an Affine Projection scheme [7] instead of the standard least-mean square scheme. Effectively, this leads to several decibels of additional noise reduction because of the improved ability of the algorithm to follow rapid changes in the disturbance spectrum.

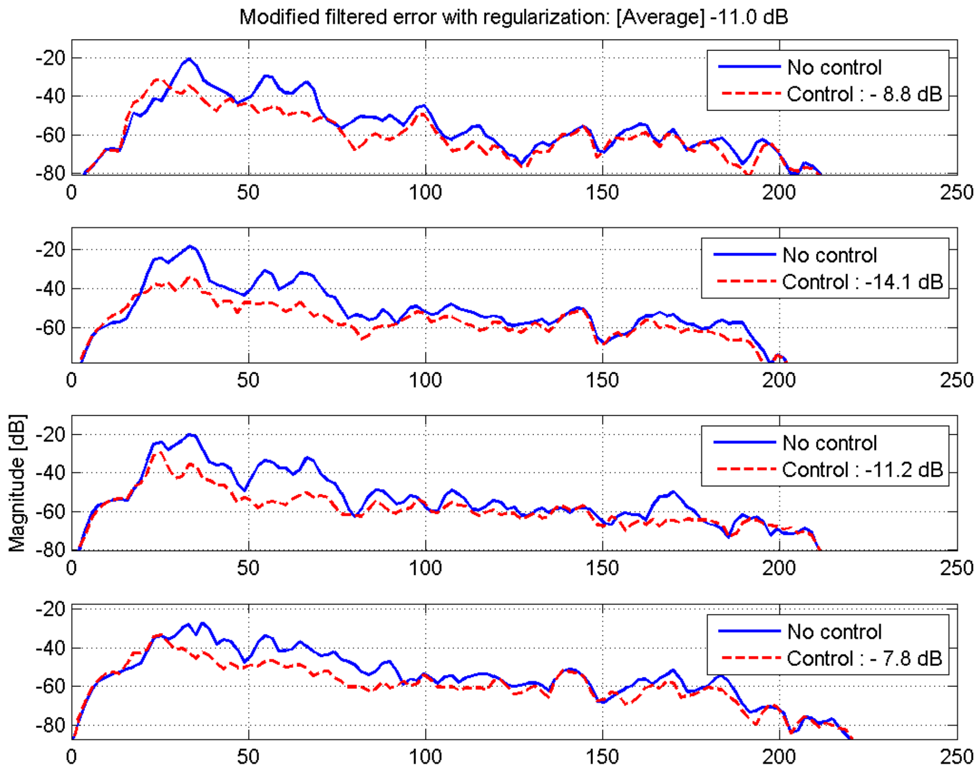

Figure 5 – Performance of the broadband adaptive feedforward control system for time-varying low-frequency noise; the graphs show spectra of the four signals as measured by the controller for control off (blue solid line) and control on (red dashed line).

4 Conclusions

This paper has shown three different applications of a particular algorithm for broadband multiple-input multiple-output active noise and vibration control. For each of the applications, certain extensions were developed. In the first application, the basic feedforward structure of the algorithm was extended with feedback cancellation for the reference sensors, leading to adaptive feedback control based on Internal Model Control. The special feature in this application is the sample rate of 12 kHz. Such a high sample rate was made possible by an efficient implementation of the MIMO adaptive feedback controller. In the second application, the adaptive MIMO controller was extended with high-speed decentralized feedback controllers which reduce the vulnerability of the system for parametric uncertainties. In the last application, an energy density sensor was used to control low-frequency broadband noise due to transportation vehicles. Improved convergence rates were obtained with an Affine projection scheme. Special preprocessing schemes for the energy density sensor and postprocessing for driving the acoustic sources were found to be critical for control performance and for perceived audio performance as well.

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