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Active control of repetitive transient noise

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Abstract

This paper presents the possibilities of active noise control techniques to reduce repetitive transient noise. Although there is a significant difference in nature between periodic and transient noise, up till now no specific research on active noise control of transient noise was reported. The presented research focuses on the development of control algorithms, dedicated for the control of repetitive transient noise. The effectiveness of the control algorithms is demonstrated for the reduction of impulse noise in a duct by a secondary loudspeaker. A linear time-invariant feedback controller is developed to drive this loudspeaker. The performance of this time-invariant controller is limited for several reasons: time variance of the system parameters, limited controller bandwidth, nonlinearities, etc. However, when the noise consists of successive impulsive sounds that exhibit a repetitive character, it is possible to extend the developed time-invariant feedback controller with a learning behaviour, based on the additional information about the repetitiveness. At each pulse, the control signal for the secondary loudspeaker can be adapted on the basis of the residual noise at the previous pulses, such that the radiated noise is reduced. This latter control technique, called iterative learning control, can significantly increase the performance of a feedback active noise control system.

1. Introduction

Noise pollution from modern industrial activities is an environmental problem. Especially in machine halls with working machines such as punching machines and presses, that generate repetitive impact noise, the radiated noise levels often exceed the legal regulations on noise emission [1]. The importance of sufficient noise suppression in these applications is confirmed by several studies, stating that human exposure to impact noise can significantly increase the damage risk to the auditory system [2,3]. The goal of the presented research is to study the possibilities of active noise control (ANC) techniques to reduce repetitive transient noise. At the moment, transient noise problems are mostly addressed in a passive way. The restrictions of these passive noise control techniques (inefficient at low frequencies, requiring large installation spaces, etc.) indicate that there is a need for active solutions, suitable for the reduction of low-frequency noise. Although there is a significant difference between periodic and transient noise, up till now no particular research on ANC of transient noise has been executed, at least not to the authors' knowledge. This paper presents the development of new control algorithms, adapted to the specific properties of repetitive transient noise from successive impacts.

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The existing ANC control algorithms can be classified into two groups: feedforward and feedback control algorithms. Nowadays, adaptive feedforward algorithms [4–6] are most popular in ANC applications. Because the adaptive part is often based on a continuous convergence to an optimum, most of the existing feedforward algorithms cannot be used for the control of transient noise signals. On the contrary, the feedback algorithms developed for the control of continuous noise [7] can be used for transient noise. However, the design of these feedback controllers must be optimized, based on the expected transient disturbance noise. In some applications, also combined algorithms are used with feedforward as well as feedback control filters [8].

In this paper, three types of controllers are developed for the reduction of repetitive impulse noise in a duct. The goal of the research is to reduce the impulse noise generated by a primary loudspeaker at one end of a half-open duct by the anti-noise of a second loudspeaker downstream the duct. An error microphone close to the secondary loudspeaker measures this primary noise. This signal is sent to the input of the controller, which drives the secondary source. The duct is a good demonstration breadboard to study the feasibility of these controllers, because it is easy to define a reliable model of a duct due to the one-dimensional wave propagation. First a linear time-invariant feedback controller is proposed. This controller is rather simple, but is presented in order to compare it with the other types of controllers. In many impact noise problems in industrial machinery, successive impacts have a repetitive behaviour. This additional information cannot be used by a time-invariant feedback controller, such that its performance is limited. In the second type of designed controllers, i.e. iterative learning controllers (ILC), the repetitiveness of the disturbance noise is the crucial element. The third group of controllers for the impulse noise in the duct uses a combination of feedback and ILC (current cycle feedback control).

2. The half-open duct set-up

The half-open duct test set-up is shown in Fig. 1. The disturbing impulse noise P_d is induced by a primary loudspeaker, to which voltage pulses of 1 ms are sent. Most of the energy spectrum of the disturbing noise pulses is thus included in the low-frequency bands up to 1 kHz. The time between two consecutive pulses is variable but is always longer than 1 s, such that the disturbing pressure due to the previous pulse is totally damped out when the new pulse is generated. The voltage pulse to the primary loudspeaker is supposed to be available as the trigger signal, which announces a new noise pulse in advance. A secondary loudspeaker, 40 cm from the end of the duct, is used to reduce the noise in the error microphone 5 cm downstream the duct. The purpose is to reduce the total acoustical energy radiated by the duct. The different controllers are developed based on the measured plant transfer function between the secondary loudspeaker and the error microphone P_{sec}, represented in Fig. 2.

3. Feedback control

This section deals with the development of a time-invariant feedback controller for the duct. This feedback controller is designed based on an input–output approach (loopshaping). A model-based approach could not be used because the transfer function between the secondary loudspeaker and the error microphone, i.e. the

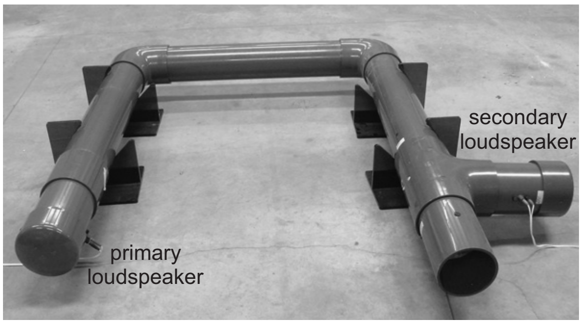

Fig. 1. A photo of the half-open duct test set-up.

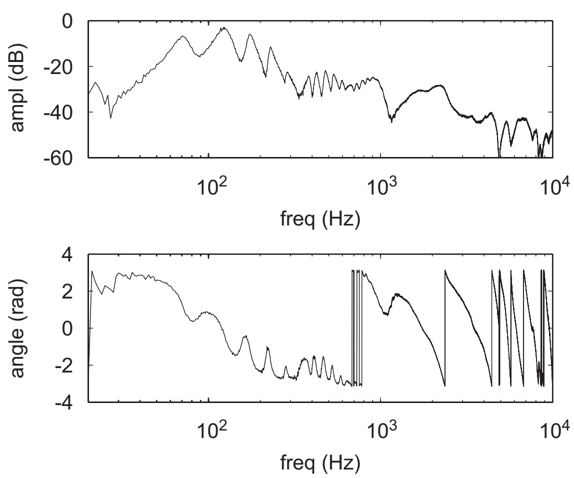

Fig. 2. The measured plant transfer function P_{sec} between the secondary loudspeaker and the error microphone.

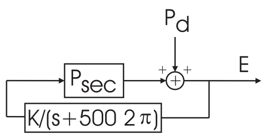

Fig. 3. The control scheme of the time-invariant proportional feedback controller with roll-off, where P_{sec} denotes the controlled secondary plant between the secondary loudspeaker and the error microphone, and P_d denotes the disturbance noise.

controllable plant P_{sec}, is not straightforward to model due to the high amount of duct resonances. Even if a good model of this plant could be found, the uncertainty in practical situations would limit the usefulness of such a model.

The controller design starts from a proportional feedback controller. Because of the limited bandwidth due to the time delay between the secondary loudspeaker and the error microphone, high reductions could only be achieved at the lower duct resonance frequencies. In order to obtain an extra reduction at lower frequencies (most of the energy of the noise disturbance is low frequency), a roll-off at higher frequencies was added to the proportional controller (Fig. 3).

This feedback controller is implemented on a dSPACE 1103 DSP board, which calculates the control signals for the secondary loudspeaker. The noise reduction at the error microphone at the successive noise pulses, obtained by the controller with roll-off, is shown in Fig. 4. A reduction can be achieved at the second pulse, but afterwards there is no improvement at the consecutive pulses. Figs. 5(a) and (b) present a comparison in the time and the frequency domain between a controlled and an uncontrolled pulse. The feedback controller yields at the error microphone a mainly low-frequency (<250 Hz) reduction of 5 dB.

4. Iterative learning control

In the presented linear time-invariant feedback controller, the repetitive behaviour of successive noise pulses cannot be used. The reduction, achieved at the first pulse, does not improve at the next pulses. A control technique that does use the additional, repetitive information of the disturbance is iterative ILC.

The ILC approach was motivated by the intuitive idea that it should be possible to improve the performance of a system that performs repetitively the same task and reproduces continuously the same error (welding robots, pick-and-place machines, etc.). Using the experience from the past, modifications to the input signal can be applied to the system during the next operation in order to obtain a better future performance. The first contribution on ILC, a paper by Uchiyama [9], was not widely known, because it was only published in

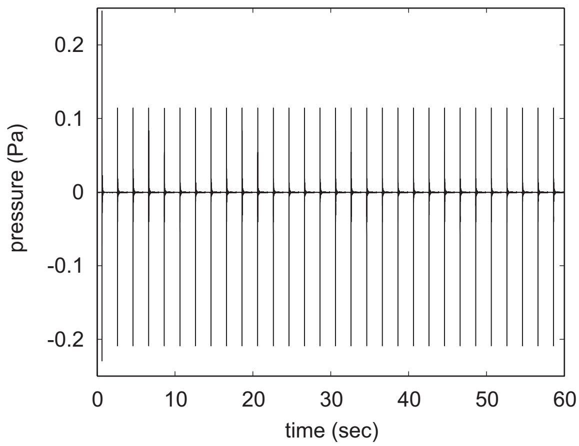

Fig. 4. The effect of the time-invariant feedback controller on the successive disturbing noise pulses (the controller is turned on after the first pulse).

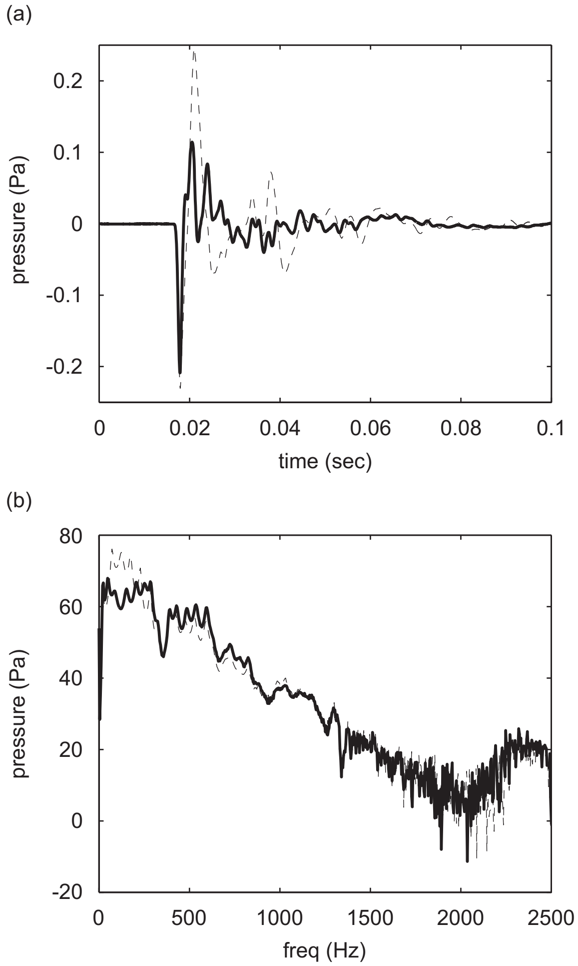

Fig. 5. Comparison of a noise pulse measured at the error microphone without control (dashed line) and with time-invariant feedback control (solid line): (a) in the time-domain and (b) in the frequency-domain.

Japanese. The idea of ILC was further developed by Arimoto [10,11], mainly active in the field of robotics, and became also popular among other researchers (Refs. [12,13]). A good survey of ILC can be found in the book of Moore [14] and a more recent overview paper [15]. Although the potential of ILC was demonstrated for a broad class of applications (e.g. robotics [16], CNC machine tools [17], chemical processes [18], etc.), the technique has never been used in the field of ANC. In recent publications [19,20] it was shown that ILC was not fundamentally different from time-invariant control methods and that causal ILC can do no better than conventional feedback control. Therefore, future work on ILC should focus on the benefits of non-causal ILC filters. These non-causal filters are very useful in ANC applications, because they can compensate the large time-delays in acoustic systems.

The classic ILC scheme, for example used for tracking of reference trajectories in robotics, is presented in Fig. 6(a): the input to the controllable plant (i.e. the secondary loudspeaker) is iteratively refined at each pulse, based on the deviation of the reference signal (i.e. the noise to be generated by the loudspeaker $-P_d$) at the previous pulse. The input to the plant at a new impact depends on the previous input and the remaining error at the previous impact, which are both filtered (by the operator $\mathbf{V}$ with the transfer function $V(s)$ and the operator $\mathbf{W}$ with the transfer function $W(s)$) and stored in a memory. When a trigger signal announces a new pulse, the signal in the memory is sent to the secondary loudspeaker. This scheme is equivalent to the practically more feasible scheme in Fig. 6(b), where the noise measured by the error microphone is fed back instead of the output of the loudspeaker.

Since the disturbance pulses P_d in ILC are distinct and repetitive, several discrete time intervals with a fixed duration T can be studied separately. The ILC controller updates the control input signal at each time interval. A trigger signal, which announces a new disturbance and defines the beginning t_k of a new time interval, is supposed to be available. The initial conditions of the controlled system at the start of each considered time interval are supposed to be equal. Define the time history of the error signal and the control signal and the disturbance at the kth pulse ($k = 1, 2, 3, \ldots$) according to:

$$\underline{E}_k = E_k(0 : T), \tag{1}$$

$$\underline{U}_k = U_k(0 : T), \tag{2}$$

with

$$E_k(t) = E(t_k + t), \tag{3}$$

$$U_k(t) = U(t_k + t). \tag{4}$$

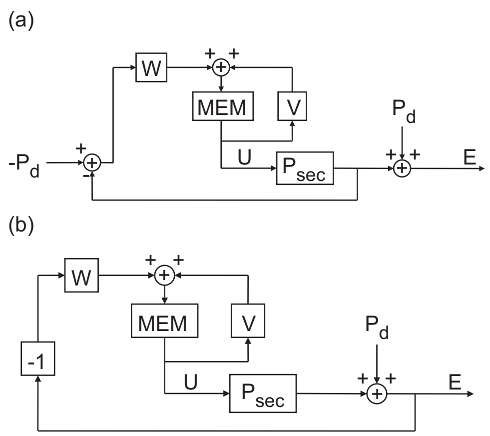

Fig. 6. (a) The classic iterative learning control scheme and (b) an adapted scheme, dedicated for active noise control, where P_{sec} is the controlled secondary plant between the secondary loudspeaker and the error microphone, P_d is the disturbance noise, U is the control signal, E is the remaining error signal at the error microphone, W and V are the learning filters and MEM is the memory, which stores the control signals for the secondary loudspeaker.

The updating expression for the control signal is then given by

$$\underline{U}_{k+1} = \mathbf{V}\underline{U}_k - \mathbf{W}\underline{U}_k. \tag{5}$$

For monotonic convergence of the ILC algorithm, the ILC filters **V** and **W** have to satisfy a criterion, which can be derived in the frequency domain. In the literature, the following updating law for the remaining error at the error microphone can be found

$$\|\underline{E}_{k+2} - \underline{E}_{k+1}\|_\infty \leqslant \|V - P_{\text{sec}}W\|_\infty \|\underline{E}_{k+1} - \underline{E}_k\|_\infty. \tag{6}$$

Consequently, the convergence criterion $\|\underline{E}_{k+2} - \underline{E}_{k+1}\|_\infty < \|\underline{E}_{k+1} - \underline{E}_k\|_\infty$ will be met if

$$\|V - P_{\text{sec}}W\|_\infty < 1. \tag{7}$$

When this convergence criterion is fulfilled, it is proven that the remaining error will converge to

$$E_\infty = \frac{V-1}{1 - V + P_{\text{sec}}W} P_d. \tag{8}$$

It is important to notice that the remaining error can only become 0 if $V(s) = 1$. This is the reason why most of the proposed ILC schemes operate with $V(s) = 1$. In this situation, the adapted criterion is given by

$$\|1 - P_{\text{sec}}W\|_\infty < 1. \tag{9}$$

This criterion indicates that the Nyquist curve of $P_{\text{sec}}W$ has to be located inside a unit circle with the centre point (1,0), commonly denoted as the learning circle. This can be achieved by choosing the **W**-filter such that the phase of $P_{\text{sec}}W$ stays between -90° and 90° and that the amplitude of $P_{\text{sec}}W$ is small enough.

In Fig. 7 the Nyquist plot of P_{sec} is shown, relative to the bounding learning circle. Each circle, which represents a duct resonance, is rotated by an angle due to the time delay between the secondary loudspeaker and the error microphone such that the phase of P_{sec} does not stay between -90° and 90° (Fig. 2) and consequently the Nyquist curve exceeds the learning circle.

The time delay of P_{sec} must be compensated by the ILC filter **W**, which means that this filter should look forward in time and use future time samples. However, because the **W**-filter is applied to the remaining error of the previous pulse, these future time samples are available at the next pulse. Consequently, the **W**-filter can be non-causal and compensate a time-delay. The transfer function of the **W**-filter, compensating the time delay between the secondary loudspeaker and the error microphone, is given by

$$W_1 = K \exp\left(\frac{(x_{\text{error}} - x_{\text{sec}})s}{c}\right) \tag{10}$$

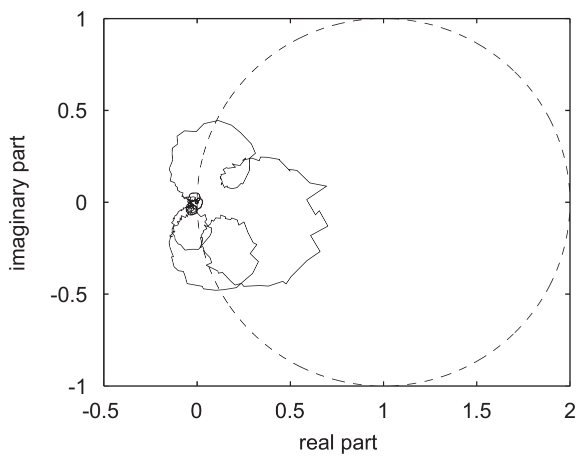

Fig. 7. The Nyquist plot of the controllable plant P_{sec} (solid line, 1–10,000 Hz) and the learning circle (dashed line).

in which x_{error} and x_{sec} are the positions of the error microphone and the secondary loudspeaker, and c is the sound velocity. The Nyquist diagram and the Bode plot of $P_{\text{sec}}W_1$ are shown in Figs. 8(a) and (b). Although the time delay is compensated, the Nyquist plot of $P_{\text{sec}}W_1$ does not stay inside the learning circle and the phase of $P_{\text{sec}}W_1$ still exceeds the $-90°$ and $90°$-limit, mainly due to the secondary loudspeaker dynamics. Therefore, an extra second-order filter with the transfer function $W_2 = (s^2 + 2\zeta_1\omega_1 s + \omega_1^2)/(s^2 + 2\zeta_2\omega_2 s + \omega_2^2$, that compensates for this dynamics, is added to W_1 ($\omega_1 = 2\pi 100 \approx$ resonance frequency of the loudspeaker, $\omega_2 = 2\pi 30$). Even with this extra filter, the Nyquist plot of $P_{\text{sec}}W_1W_2$ (Fig. 9(a)) does not stay inside the learning circle: the phase of $P_{\text{sec}}W_1W_2$ (Fig. 9 (b)) still exceeds the $-90°$- and $90°$-limit, significantly at very low and high frequencies but only slightly in the frequency range between 50 and 4000 Hz. To compensate the phase in this latter frequency range, a lead compensator with the transfer function $W_3 = (K_2(s + 2\pi 100))/(s + 2\pi 1500)$ was added to W_1 and W_2 to create the total **W**-filter (with the transfer function $W = W_1W_2W_3$).

In the frequency range between 50 and 4000 Hz the phase of $P_{\text{sec}}W$ (Fig. 10) stays between $-90°$ and $90°$ such that also the Nyquist plot of $P_{\text{sec}}W$ (Fig. 11(a)) stays inside the learning circle in this frequency range. However, at lower and higher frequencies, the Nyquist plot of $P_{\text{sec}}W$ (Fig. 11(b)) still crosses the learning circle because the phase limits are exceeded due to unmodelled loudspeaker dynamics and transversal resonances of the duct. This cannot be avoided because the **W**-filter would become too complicated if these phenomena should be compensated for. Consequently, the convergence criterion Eq. (9), to obtain a zero

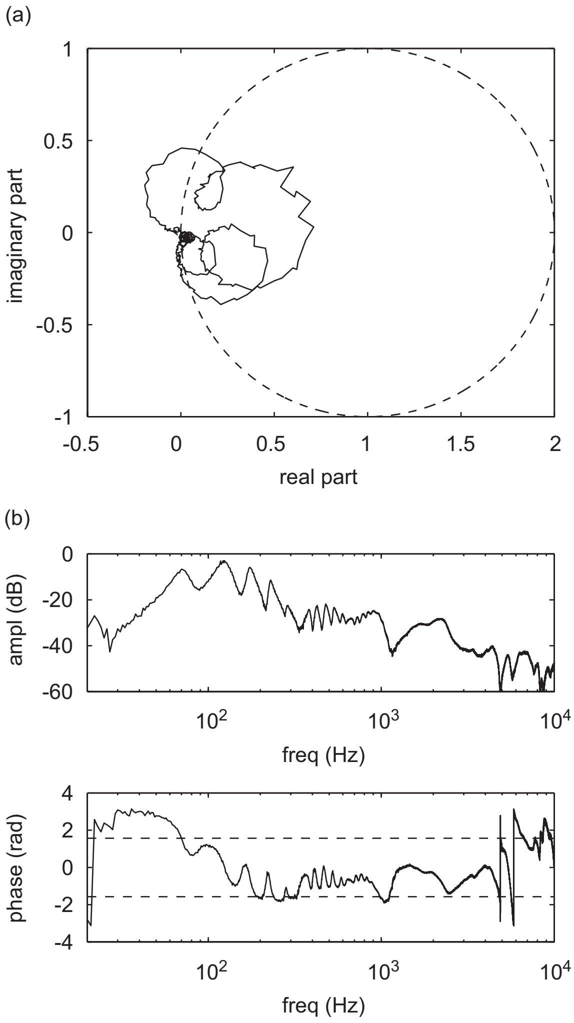

Fig. 8. (a) The Nyquist plot of $P_{\text{sec}}W_1$ (solid line, 1–10,000 Hz) and the learning circle (dashed line) and (b) the Bode plot of $P_{\text{sec}}W_1$.

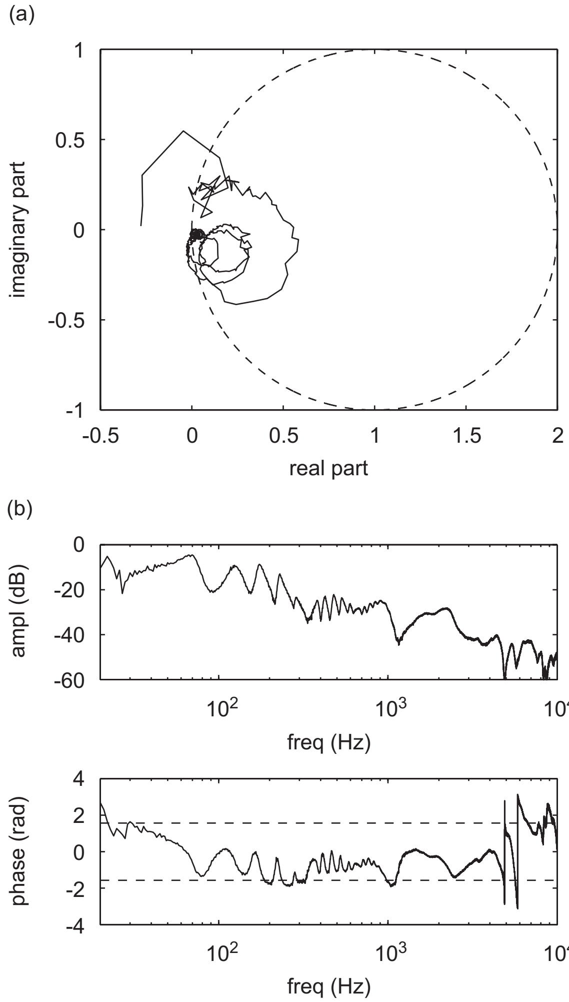

Fig. 9. (a) The Nyquist plot of $P_{\mathrm{sec}}W_1W_2$ (solid line, 1–10,000 Hz) and the learning circle (dashed line) and (b) the Bode plot of $P_{\mathrm{sec}}W_1W_2$.

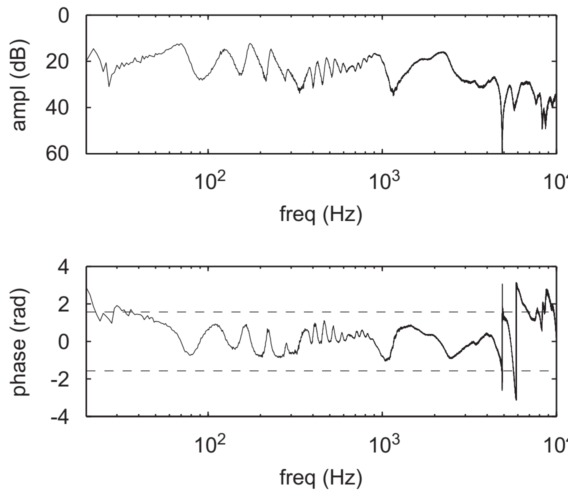

Fig. 10. The Bode plot of $P_{\mathrm{sec}}W$.

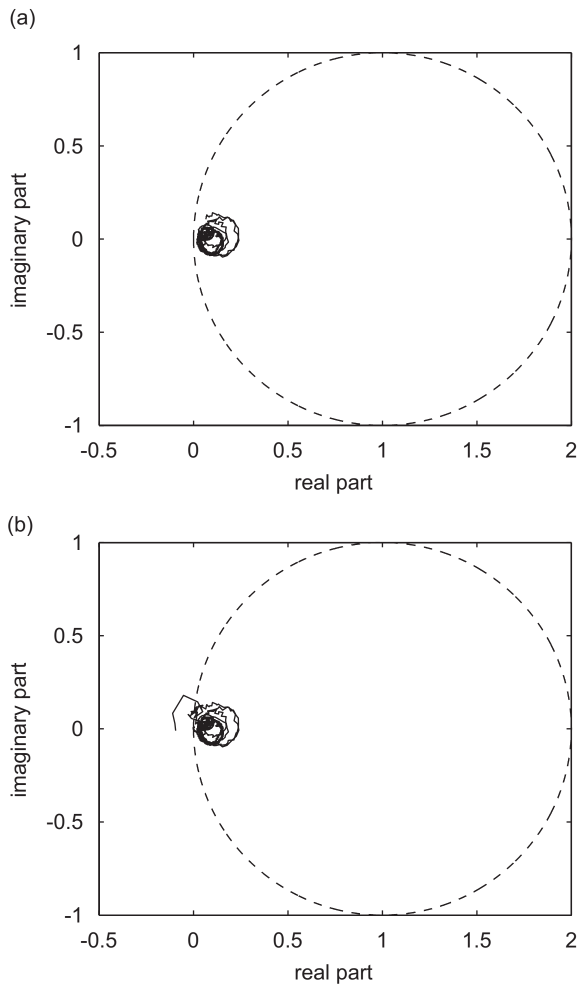

Fig. 11. The Nyquist plot of $P_{sec}W$ (solid line) and the learning circle (dashed line): (a) 50–4000 Hz and (b) 1–10000 Hz.

residual noise, cannot be fulfilled at very low and high frequencies. Therefore, a **V**-filter, equal to 1 in the frequency range between 50 and 4000 Hz (to obtain a remaining noise of 0 in this range) and smaller than 1 at the low and high frequencies (to fulfil the convergence criterion), should be designed [21]. This can be achieved by choosing the **V**-filter as a band-pass filter:

$$V = \frac{2\pi 10000 s}{(s + 2\pi 10)(s + 2\pi 10000)}. \tag{11}$$

With this **V**-filter, the general convergence criterion Eq. (7) should be checked: Fig. 12 shows that this criterion is only fulfilled if **V** is a band-pass filter. The achievable theoretical reduction with these ILC filters, according to Eq. (8), is presented in Fig. 13: a broadband reduction between 100 and 1000 Hz can be achieved combined with a limited amplification at some frequencies outside this range (max. 5 dB).

The non-causal ILC controller as discussed in the previous paragraphs has been implemented on a dSPACE 1103 DSP board and applied to the duct case study. The practical procedure for the processing of the ILC algorithm involves the following steps, which are repeated every time that a new pulse arrives:

(1) the noise is measured by the error microphone;
(2) the measured error signal is stored in a memory;

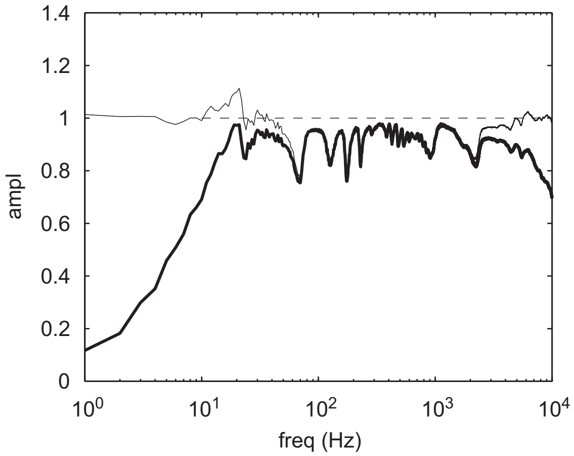

Fig. 12. The convergence test $\| V - P_{\text{sec}} W \|_{\infty} < 1$ of the iterative learning control filters with $V = 1$ (solid line) and with V, designed as a band-pass filter (bold line); the limit for stability is indicated by the dashed line.

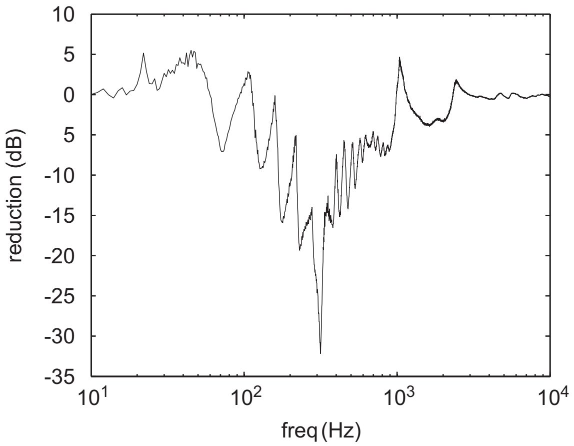

Fig. 13. The theoretically achievable reduction of the iterative learning control filters.

(3) the control signal for the secondary loudspeaker is calculated by the summation of the stored error signal, filtered by the V-filter, and the control signal at the previous pulse, filtered by the W-filter;
(4) the calculated control signal is also stored in the memory;
(5) a trigger signal announces a new pulse; and
(6) the stored control signal is sent to the secondary loudspeaker.

The noise reduction at the error microphone, obtained during the experiments by the ILC controller, is shown in Fig. 14. There is a clear continuous learning behaviour of the ILC controller. The noise at the successive pulses converges to a certain residual error, which is achieved after 25 pulses. The effect of the controller after convergence, in the time and the frequency domain, is shown in Figs. 15(a) and (b). Contrary to the feedback controller, which created only a low-frequency reduction (<250 Hz), a broadband noise reduction till 1000 Hz was achieved. A total reduction of 4 dB was realized after 30 impacts. Due to the high-pass **V**-filter, at low frequencies around 100 Hz, less reduction is obtained by the ILC controller compared to the feedback controller, which also explains the limited overall reduction achieved by the ILC controller. At

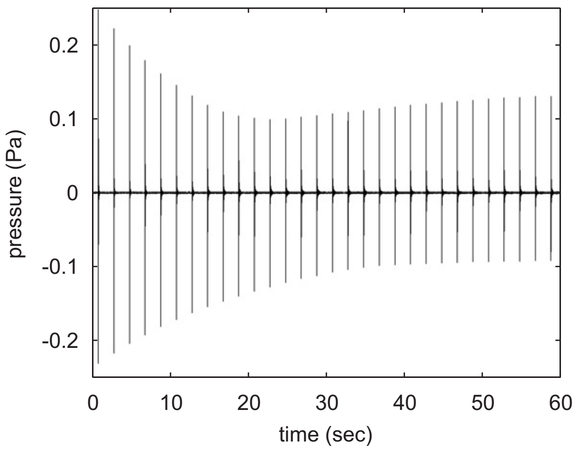

Fig. 14. The effect of the iterative learning controller on the successive disturbing noise pulses (the controller is turned on after the first pulse).

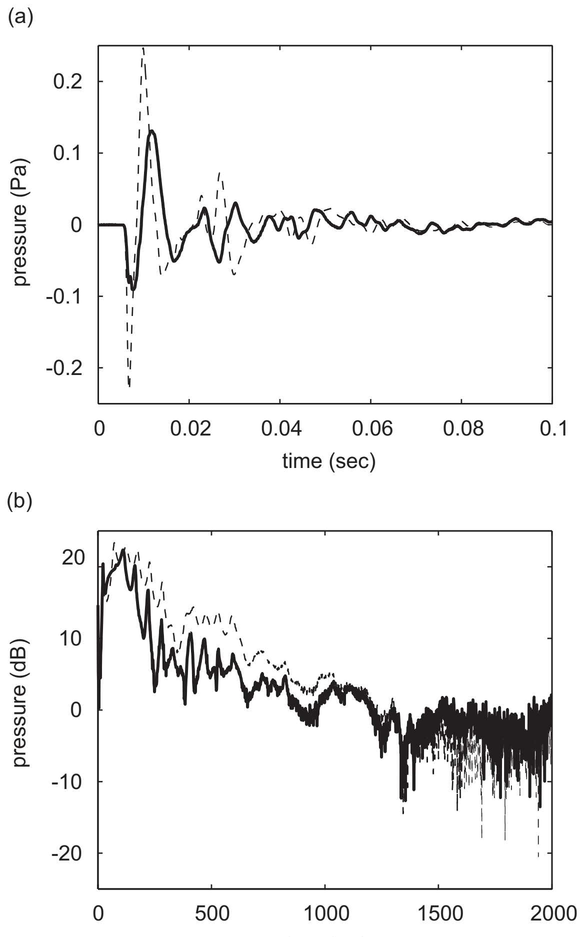

Fig. 15. Comparison of a noise pulse measured at the error microphone without control (dashed line) and with iterative learning control at the 30th controlled impact (solid line): (a) in the time-domain and (b) in the frequency-domain.

frequencies above 1000 Hz, the impulse noise cannot be reduced, because the noise at the error microphone is not repetitive. At these higher frequencies, the level of the ambient random noise exceeds the repetitive noise of the impact, generated by the primary loudspeaker.

5. Combination of iterative learning control and feedback control

In this paragraph, a current cycle feedback controller [22] is designed, which combines the advantages of feedback control (reduction at first impacts, low-frequency reduction) and ILC (improved performance at successive impacts, high-frequency reduction). In this case, the ILC-controller is only used to reduce the residual error, that cannot be controlled by the feedback controller C_{fb} (Fig. 16). The feedback controller can also compensate for random disturbances. The feedback controller is designed as a proportional controller with roll-off (see higher). The development of the ILC controller is done in the same way as for the ILC without feedback: the controllable plant is now the closed-loop system $P_{\text{sec}}/(1 + C_{\text{fb}}P_{\text{sec}})$ instead of P_{sec}. The **V**-filter is a band-pass filter, the **W**-filter compensates for the time-delay, the duct resonances and the secondary loudspeaker dynamics.

Figs. 17, 18(a) and (b) present the results, achieved by the combination of ILC and feedback control in the practical tests. An overall reduction of 8 dB, from 50 to 1000 Hz, is obtained after 30 pulses. Compared to the ILC without feedback control, it is clear that an extra low-frequency reduction is obtained.

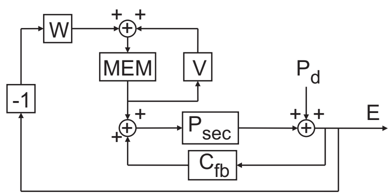

Fig. 16. The classic current cycle feedback control scheme, which combines time-invariant feedback control and iterative learning control, where P_{sec} is the controlled secondary plant between the secondary loudspeaker and the error microphone, P_d is the disturbance noise, E is the remaining error signal at the error microphone, W and V are the learning filters, C_{fb} is the feedback controller and MEM is the memory, which stores the control signals for the secondary loudspeaker.

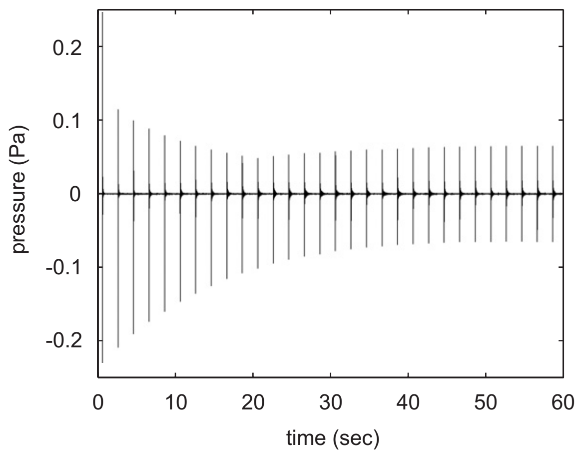

Fig. 17. The effect of the iterative learning controller combined with feedback on the successive disturbing noise pulses (the controller is turned on after the first pulse).

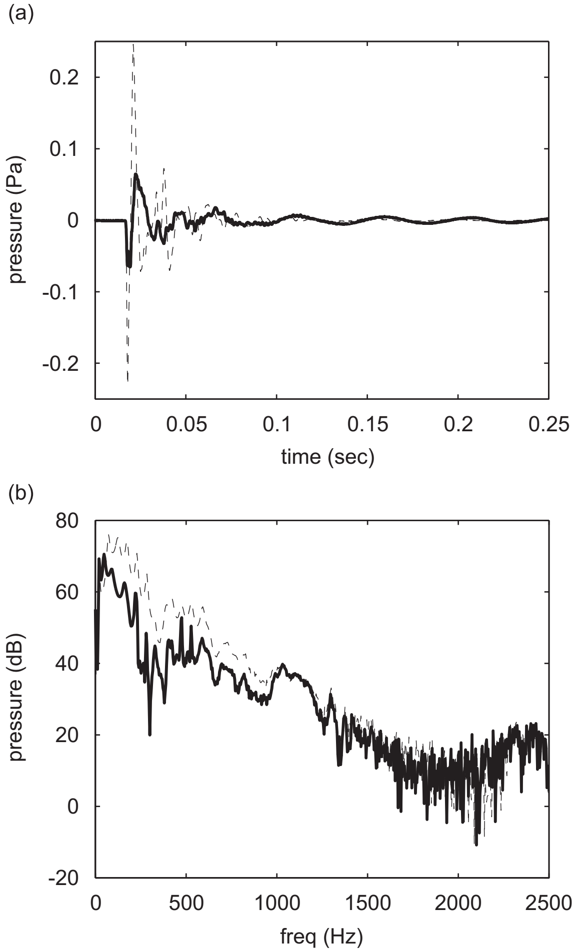

Fig. 18. Comparison of a noise pulse measured at the error microphone without control (dashed line) and with iterative learning control combined with feedback at the 30th controlled impact (solid line): (a) in the time-domain and (b) in the frequency-domain.

6. Conclusion

This paper presents different algorithms for the active control of repetitive transient noise pulses. First, a linear time-invariant feedback controller was discussed. The performance of this controller is good at the first pulses, but does not improve for successive pulses with a repetitive character. Therefore, an ILC algorithm was implemented, in which the repetitiveness of the disturbance noise is a crucial element. At the first pulses, only a small reduction could be achieved, but the performance improves at the successive pulses and becomes better in the higher frequency range than the time-invariant performance of the feedback controllers. Finally, an ILC controller with feedback was designed, which combines the advantages of feedback control (good reduction at first pulses, control of random disturbances, low-frequency reduction) and ILC (improved performance at successive pulses, high-frequency reduction). The developed controllers were validated on the impulse noise in a real duct. In the future, the frequency of the pulses generated by the primary loudspeaker will be increased such that at a new pulse generation the residual noise from the previous pulse is not zero. New control algorithms, taking this residual noise into account, will be necessary. The future work consists also of the extension of the ILC principle to 3D impact noise problems and to Active Structural Acoustic Control (ASAC) applications.

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References

[1] E.J. Richards, A. Lenzi, On the prediction of impact noise, IX: the noise from punch presses, *Journal of Sound and Vibration* 103 (1985) 43–81.
[2] A. Akay, A review of impact noise, *Journal of the Acoustical Society of America* 64 (1978) 977–987.
[3] R.P. Hamernik, W.A. Ahroon, Interrupted noise exposures: threshold shift dynamics and permanent effects, *Journal of the Acoustical Society of America* 103 (1998) 3478–3488.
[4] S. Elliott, *Signal Processing for Active Control*, Academic Press, London, UK, 2001.
[5] C. Bao, P. Sas, H. Van Brussel, Adaptive active control of noise in 3-D reverberant enclosures, *Journal of Sound and Vibration* 161 (1993) 501–514.
[6] Y. Tu, C.R. Fuller, Multiple reference feedforward active noise control—part I: analysis and simulation of behavior, *Journal of Sound and Vibration* 233 (2000) 745–759.
[7] O.E. Kaiser, S.J. Pietrzko, M. Morari, Feedback control of sound transmission through a double glazed window, *Journal of Sound and Vibration* 263 (2003) 775–795.
[8] W. Tseng, B. Rafaely, S. Elliott, Combined feedback-feedforward active control of sound in a room, *Journal of the Acoustical Society of America* 104 (1998) 3417–3425.
[9] M. Uchiyama, Formulation of high-speed motion pattern of mechanical arm by trial, *Transactions of the Society of Instrumentation and Control Engineers* 19 (1978) 706–712.
[10] S. Arimoto, S. Kawamura, F. Miyazaki, Bettering operation of robots by learning, *Journal of Robotic Systems* 1 (1984) 123–140.
[11] S. Kawamura, F. Miyazaki, S. Arimoto, Realization of robot motion based on a learning method, *IEEE Transactions on Systems, Man, and Cybernetics* 18 (1988) 126–134.
[12] R.W. Longman, Iterative learning control and repetitive control for engineering practice, *International Journal of Control* 73 (2000) 930–954.
[13] M. Tomizuka, T.C. Tsao, K.K. Chew, Analysis and synthesis of discrete-time repetitive controllers, *Journal of Dynamic Systems, Measurement, and Control* 111 (2000) 353–358.
[14] K.L. Moore, *Iterative Learning Control for Deterministic Systems*, Springer, London, UK, 1993.
[15] K.L. Moore, Iterative learning control: an expository overview, *Applied and Computational Controls, Signal Processing, and Circuits* 1 (1998) 151–214.
[16] R. Horowitz, Learning control of robot manipulators, *ASME Journal of Dynamic Systems, Measurement, and Control* 115 (1993) 402–411.
[17] D. Kim, S. Kim, An iterative learning control method with application for CNC machine tools, *IEEE Transactions on Industry Applications* 32 (1996) 66–72.
[18] K.S. Lee, S.H. Bang, S. Yi, J.S. Son, S.C. Yoon, Iterative learning control of heat-up phase for a batch polymerization reactor, *Journal of Process Control* 6 (1996) 255–262.
[19] P.B. Goldsmith, On the equivalence of causal LTI iterative learning control and feedback control, *Automatica* 38 (2002) 703–708.
[20] M.H.A. Verwoerd, Iterative Learning Control—A Critical Review, PhD Thesis, University of Twente, Enschede, The Netherlands, 2005.
[21] M. Norrlöf, Iterative Learning Control—Analysis, Design, and Experiments, PhD Thesis, Linköping University, Sweden, 2000.
[22] Y. Chen, J.X. Xu, T.H. Lee, Current iteration tracking error assisted iterative learning control of uncertain nonlinear discrete-time systems, *Proceedings of the 35th IEEE Conference on Decision and Control*, Kobe, Japan, 1996.