Active Noise Control in Propeller Aircraft

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Abstract. A noisy environment dominated by low frequency noise can often be improved through the use of active noise control. This situation arises naturally in propeller aircraft where the propellers induce periodic low frequency noise inside the cabin. The cabin noise is typically rather high, and the passenger flight comfort could be improved considerably if this level were significantly reduced. This paper addresses same design aspects for multiple-reference active noise control systems based on the feedforward strategy. The paper also discusses the operation of narrowband feedforward active noise control system and presents results from experiments.

Introduction

In the last decade acoustic noise has become increasingly regarded as a significant environmental problem. Noise problems in our homes have received considerable attention. In industry, for example, engines, fans, transformers and compressors radiate noise. In cars, boats, trains and aircraft, for example, noise reduces comfort. Lightweight materials and more powerful engines are used in high-speed vehicles, resulting in a general increase in interior noise levels.

The primary concern with noise in the low frequency range is not only the potential risk of damage to the hearing. Low frequency noise is annoying and during periods of long exposure it causes fatigue, discomfort and loss of concentration. Reduced concentration may also lead to an increased risk for accidents. The masking effect which low frequency noise has on speech also reduces speech intelligibility [1]. Low speech intelligibility is perceived as disturbing and irritating.

Redesigning could solve noise problems. This is generally very costly, however. On the other hand, noise problems may be solved using traditional passive approaches and/or approaches based on the concept of active noise control [1]-[3]. The choice of technique is determined by the characteristics of the noise as well as of the application. However, this paper will not address the problems of attenuating noise using passive approaches. Instead it focuses exclusively on the field of active noise control.

Conventional passive approaches consist of absorbers and/or reflectors/barriers. The absorbers convert the acoustic energy to thermal energy, while the reflectors/barriers prevent the noise from entering a space from another space by reflecting the incident wave field. In terms of practical size passive methods are suitable when reducing noise in the frequency range over approximately 500 Hz. The thickness of the acoustical absorbers, or the distance between the absorber and the wall, must be large, approximately one quarter of a wavelength, to obtain reasonable low frequency reduction, e.g. a frequency of 100 Hz results in a wavelength of 3.4 meters. In addition, in order to reduce the sound transmission from one space to another, a heavy barrier between these spaces is required. Consequently, the use of passive methods to attenuate low frequency noise is often impractical since considerable extra bulk and weight are required. In all areas of transport large weight is associated with high fuel consumption.

In order to overcome the problems of ineffective passive suppression of low frequency noise, the technique of Active Noise Control (ANC) has become of considerable interest [2], [3]. The fundamental principle of active noise control is based on the superposition of sound waves. Secondary sources produce an ”anti-noise” of equal amplitude and opposite phase to the primary or unwanted noise. Superposition of the primary and generated...
noise results in destructive interference and reduced noise. The accuracy of the amplitude and the phase of the generated anti-noise determine the noise attenuation.

Active noise control systems significantly increase the capacity for attenuating low frequency noise below approximately 1 kHz, resulting in potential benefits in volume and weight. The upper frequency for which active control is appropriate is determined by the application. However, in enclosures whose dimensions are of the order of a few meters, such as in aircraft cabins, the upper frequency is limited to a few hundred hertz. The upper frequency limit is higher for smaller enclosures, e.g. in headsets. For noise above 1 kHz passive methods show higher potential, since neither great volume nor weight is required to achieve significant reduction. The active and passive approaches are thus complementary, and by combining these two techniques high noise attenuation over a wide frequency range is made possible, indeed over the entire audible frequency range (20-20 kHz).

Active noise control is applicable to a wide variety of low-frequency noise problems in industrial applications, transport, and consumer products [2], [3].

**Active Noise Control in Aircraft**

One major source for aircraft interior noise is the propulsion system [4]. In particular, for propeller aircraft the cabin noise is dominated by harmonic low frequency noise produced by the propellers. The most disturbing noise is typically the first three or four harmonics of the Blade Passage Frequency (BPF). The noise is transmitted through several paths into the cabin, see Figure 1. Vibrations from the engines are transmitted through the engine mounts into the wing structure, which in turn excites the whole aircraft body; turbulence from the propellers excites the rear wing which in turn causes vibrations in the rear part of the aircraft. Another important path is through the fuselage in the plane of the propellers; the propeller blades cause very high-pressure fluctuations at the outside of the fuselage which are transmitted into the passenger cabin. The importance of the different transmission paths varies with frequency. At the BPF, the sound field is usually excited throughout the whole cabin, while the harmonics tend to be excited primarily in the plane of the propellers [5].

Due to the low frequency range, typically 80-450 Hz, the practical use of passive noise reduction methods is very limited. The aircraft fuselage is constructed as a light stiff wall with only a marginal low-frequency transmission loss [1]. By using tuned dampers, the transmission loss can be significantly increased [4], [6]. A tuned damper is a mechanical resonance system consisting of a mass and a spring with a fairly high mechanical loss factor. The damper is tuned to one frequency, typically the BPF at normal cruise speed or one of its harmonics. By using several dampers, it is possible to obtain a noise reduction over a broader frequency range. One major disadvantage with the tuned dampers is the added weight, which can be the equivalent of one passenger or more. This is significant for an aircraft designed for 20-30 passengers. Another disadvantage is that the performance normally is tuned to one flight condition, which implies that the vibration absorbing effect is reduced at other flight conditions.
An active noise control system offers much more potential to the noise control engineer [2], [3], [7]. First of all, the overall attenuation is generally higher than what can be obtained with a passive installation. Since the controller is synchronized with the engines, the attenuation is maintained throughout the complete flight cycle, including cruise, climb and descent. If the controller is synchronized to both engines, the beating that appears as the engines become unsynchronized is also controlled. Even with many (more than 30) loudspeakers including cabinets, the active noise control system is lighter than a normal set of tuned dampers [7].

The eighties and nineties saw much work on active noise control in aircraft. This included theoretical analyses, simulations, laboratory experiments and flight tests. The first commercial turboprop aircraft in the world in which this technique was used is the SAAB 340 and its successor, the SAAB 2000. The first SAAB 340 was delivered in the spring of 1994, and the first SAAB 2000 was delivered later the same year [6]. Today, most aircraft manufacturers are interested in ANC since comfort is a key issue. Figure 2 shows an ANC system in an aircraft for active control of propeller-induced cabin noise.

![Figure 2: Active noise control in an aircraft application.](image)

The most commonly used sources for generating the secondary interacting sound field, the “anti-noise,” are loudspeakers. However, since vibrations in the bounding walls generally excite the sound field, another approach is to use vibration exciters attached to the wall surface. The technique using control inputs applied directly to the structure in order to reduce the vibration distribution with the objective of reducing the sound radiation has been termed Active Structural Acoustic Control (ASAC) [8]. To observe the interior noise reduction, microphones are used as control sensors.

In recent years interest has also been shown in using ASAC technology [7]. This approach can also be used in jet aircraft applications for reducing frequency components originating from out-of-balance forces from jet engines. The use of silent seats has also excited considerable interest together with the use of active headsets [7]. A silent seat system gives localized noise attenuation around the passenger head using loudspeakers incorporated in the headrest. Active headsets are, moreover, much cheaper than installing an ANC system inside the cabin.

Vibrations at low frequencies also cause passenger discomfort. These vibrations originate from engine and propeller shaft imbalance, and are transmitted through the wings into the fuselage. Since the vibrations are low frequency and the vibration sources and the transmission paths are known, active approaches also have the potential of being able to reduce such vibrations [7].

**The Control System**

The active noise control system evaluated in this paper is based on a *multiple-reference* narrowband feedforward control approach and is designed to attenuate the tones generated from the propellers. The controller is based on the actuator-individual normalized filtered-x Least Mean Squares (LMS) [9], [10]. This algorithm is synchronized to both propellers. The need for a synchronization signal from each propeller arises from the fact that the synchrophaser devices are unable to perfectly synchronize the two propellers during a complete flight cycle [9]. By using the synchronization signals, internal single-frequency reference signals are generated and
Algorithm Estimates of Control Paths

\[ w(n)_{11}, w(n)_{12}, w(n)_{1H}, w(n)_{21}, w(n)_{22}, w(n)_{2H} \]

\[ x(n)_{12}, x(n)_{1H}, x(n)_{11}, x(n)_{21}, x(n)_{22}, x(n)_{2H} \]

\[ e(n)_{1}, e(n)_{M} \]

Loudspeaker 1
Reference Generator
Loudspeakers
Microphones
Synchronization Signal

Figure 3: The control system for active control of periodic noise.

instantaneously adjusted by adaptive weights before driving the control sources, e.g. loudspeakers. Control sensors observe the residual noise and the output signals from these are used to adjust the adaptive weights so that the overall noise level is minimized.

The cabin noise inside propeller aircraft is essentially dominated by strong tonal components at the harmonics of the blade passage frequency of the propellers [4]. Propellers or periodic noise sources running with a slight rotational speed difference induce an acoustic beating. The capacity for the ANC system to handle beating sound fields is dependent on the structure of the controller [2], [11]. A structure based on a single filter and a single reference signal consisting of the sum of all reference signals does not make the best use of the information provided by the reference signals. Since the frequencies of the reference sinusoids are close together a long FIR filter is required, resulting in slow convergence of the adaptive algorithm. With the parallel filter structure each reference signal is individually processed, which in narrowband ANC involves individual harmonic control. The shorter filter can be used with better convergence. If possible, the parallel structure is used rather than a single filter structure in order to achieve efficient and robust control of beating sound fields. The parallel structure has proven advantageous in the attenuation of propeller-generated noise and noise produced by rotating machines with almost the same rotational speeds [4], [12].

The most widely used algorithm in ANC applications is the well-known Filtered-X LMS algorithm owing to its simplicity and robustness [2], [13]. In order to improve the controller performance with regard to convergence rate, noise attenuation and tracking performance, it could be possible to use faster and more efficient algorithms [2], [13]. However, an increased performance often leads to more complex algorithms which demand greater computational capacity in the control system. In addition, since the computational power of the DSP hardware is limited, increased algorithm complexity allows fewer loudspeakers and error microphones to be used. Accordingly, in applications where a large multiple-channel system is needed, it is of great importance to keep down the computational complexity of the algorithms so that a large number of loudspeakers and microphones can be used.

The feedforward technique presented in this paper inherently exploits the narrowband assumption by using complex filtering and complex modeling of control paths. The proposed complex algorithms are advantageous in narrowband applications due to high convergence rate and low numerical complexity. These advantages are primarily the result of the orthogonality of the quadrature components constituting the complex reference signals and the simplicity of complex representation. In fact, the complex algorithm requires a minimum of adaptive and control path parameters as compared to a straightforward time-domain approach using ordinary FIR filters.
The core of the control system is a multi-channel, narrowband feedforward controller using complex signals and complex filter-weights. The complex reference signals are individually processed by a single complex weight that adjusts the amplitude and phase for each actuator. The structure of a twin-reference, multi-channel feedforward active noise control system is shown in Figure 3.

The control algorithm, based on the complex algorithm named actuator-individual normalized filtered-x LMS algorithm is described for a general control situation with \( M \) control sensors, \( L \) loudspeakers and \( R \) reference signals, where each reference has \( H \) harmonics. Assume that the real-valued control-sensor signal, \( e_m(n) \), at microphone \( m \) is given by

\[
e_m(n) = d_m(n) + y_m(n)
\]

where \( d_m(n) \) and \( y_m(n) \) represent the uncontrolled noise and the ``anti-noise'' (the secondary sound field generated by the \( L \) loudspeakers) respectively at microphone \( m \). The cost function to be minimized is given by the sum of the squared output signals from the control microphones:

\[
J(n) = \sum_{m=1}^{M} e_m^2(n)
\]

The updating scheme used to adjust the complex weights in the adaptive control system for minimizing this cost function is given by the following algorithm

\[
w_{rh}(n + 1) = w_{rh}(n) + 2\mu_{rhl} x_{rh}(n) \sum_{m=1}^{M} F_{rhml}^* e_m(n)
\]

where * denotes complex conjugate and \( F_{rhml} \) is an estimate of the frequency response function of the control path between loudspeaker \( l \) and microphone \( m \) associated with a given reference signal \( x_{rh}(n) \). Here the step-size parameter for reference \( r \), harmonic \( h \) and loudspeaker \( l \) is given by

\[
\mu_{rhl} = \frac{\mu_0}{\theta_h \sum_{m=1}^{M} |F_{rhml}|^2}
\]

where the step-size \( \mu_0 \) lies in the interval \( 0 < \mu_0 < 1/(LRH) \) [14]. This update algorithm is motivated by the assumption that the single-frequency reference signals, \( x_{rh}(n) \), are mutually uncorrelated, thereby enabling individual control of each frequency. Since only one adaptive complex coefficient is required for each reference signal and loudspeaker, it is extremely efficient in the sense that it employs a minimum of adaptive coefficients. The implementation can usually be made very compact, which leads to fast execution of the code.

One major advantage with narrowband active control of periodic noise components is that the reference signals can be synthesized internally in the controller. In this study, the synchronization signals obtained from the noise generation system were used to generate the complex reference signals. With reference signals generated in this manner, the adaptive control becomes extremely selective and it is possible to determine which harmonics are to be controlled and which are not.

The reference generation method typically used by the present research group has been a table lookup scheme consisting of several tables, one table for each harmonic, where each table contains a single sampled sinusoidal function. The reference signals are generated by reading samples from the tables. The generation of complex reference signals can be done either by using a pair of table pointers, where one pointer is delayed 90°, or by using a sine and a cosine table.

In this study, the algorithm was to be implemented on external hardware that was originally constructed for an FIR-based adaptive control scheme. This hardware delivered a composite reference signal, one for each propeller, containing the sum of the harmonics to be controlled. In order to split this composite signal into separate, complex, harmonic reference signals suitable for the algorithm, a sliding FFT-operation was used as a parallel filter bank [9]. The process of generating the complex reference signals from a synchronization signal is shown in Figure 4.
The synchronization signals derived from the noise generation system contained only one pulse per "propeller" revolution. On the targeted platform these signals were converted to contain as many pulses per revolution as the number of propeller blades on the propeller. This conversion was done by using a phase-locked loop. The frequency contents of the square wave thus produced contains the fundamental frequency (BPF) and all harmonics with decreasing amplitudes. The composite reference signals, \( s_1(n) \) and \( s_2(n) \), were obtained by feeding the square wave through a low-pass filter, allowing only the four lowest frequency components (BPF to 4xBPF to pass through. As an interface between the real composite reference signals and the complex algorithm a software splitter based on an FFT-filter bank was used. Given the real, scalar synchronization signals, \( s_1(n) \) and \( s_2(n) \), the complex, scalar reference signals, \( x_{rh}(n) \), are generated by computing the FFT-operation on blocks of \( N \) samples for every new sample (sample-by-sample basis).

The required FFT resolution for the filter bank was determined by the system sampling frequency and by the frequency separation of the harmonics at the lowest engine speed. The frequency bin number to be used as output for the complex reference signal was determined by the harmonic \( h \) to be controlled. In the evaluation, variations in the propeller speed was small. For this reason, no shifts between frequency bins were carried out.

It has been shown that the length of the FFT affects the response time for the algorithm to rapid changes in the input signals [9]. For this reason, as well as considering the calculation effort, it is desirable to keep the length of the FFT as short as possible. If the FFT is too short, on the other hand, the frequency resolution will suffer, leading to poor suppression of nearby harmonics. In the present application, the FFT size was chosen to 64 and a Blackman window was used. With a sampling frequency of 1 kHz, the filter resolution becomes 1024/64=16 Hz. For a Blackman window the suppression of the side lobes is at least 58 dB, compared to 13 dB when a rectangular window is used.

One significant disadvantage with the sliding FFT filtering technique, besides the fairly high implementation cost, is the delay caused by the linear-phase filter employed by this method. The group delay equals one-half the length of the FFT, which for the present case corresponds to 32 ms. Due to this delay, very rapid changes in the rotational speed will not be accurately traced by the controller. Another disadvantage is that the calculation of the FFT requires a large amount of computational effort in the DSP.

**Experiment Results**

In development work of a control system, an important part is the evaluation of its performance and robustness. In some applications it is possible to install the control system in a practical application, e.g. in an aircraft, motor boat or another vehicle, and then evaluate its performance under actual running conditions. This paper deals with the evaluation of a control system in an aircraft. In this particular application, it is difficult, time consuming and very expensive to do the performance evaluation under actual in-flight tests. An alternative approach could be to carry out the evaluation on grounded aircraft. Even in this case, the evaluation is expensive. Furthermore, the results would not be directly comparable with results obtained from in-flight tests due to the influence of the ground on the sound field inside the aircraft. Another alternative and a compromise in the evaluation process is to use a test section of a real aircraft, i.e. a mock-up, and use artificially generated propeller noise. A major advantage using a mock-up is that the system performance and robustness can easily be investigated in an acoustical environment. Another advantage is that the system can be tested for cases which are difficult to test during flight, such as different disturbances in the synchrophaser angle etc. The system evaluation presented in this paper is performed in an aircraft mock-up.
High attenuation can be achieved by spatially and temporally matching the primary sound field with a sound field generated by the control sources. The temporal matching is mainly determined by the control algorithm, while the spatial matching mainly depends on the location of the control sources and sensors.

The performance of the ANC system was not only evaluated in the control microphones. A separate set of 20 microphones was used solely for evaluation purposes, so-called monitor microphones. These microphones were not part of the control system and could be placed at arbitrary positions where the control system was to be evaluated. Figure 5 shows the sound pressure level of the uncontrolled and controlled sound fields at BPF (82 Hz), in the seated head height inside the mock-up. The two propellers were perfectly synchronized.

![Uncontrolled Sound Field](image)

![Controlled Sound Field](image)

Figure 5: The interior sound pressure level distributions at the passengers head level with the ANC system OFF (upper figure) and ON (lower figure).

The controlled sound field was measured after the controller was allowed to converge. The control algorithm exhibited robust convergence performance. As shown in this figure the ANC system significantly reduced the overall noise level in seated head height. It is clear that the effect of turning on the ANC system was to level out the spatial variations in the sound pressure level. In the front region inside the mock-up where several control microphones were located a substantial noise attenuation was observed. The measured mean noise reduction averaged over all the monitor microphones was approximately 10.3 dB, and approximately 18.4 dB average over all the control microphones. The difference of 8 dB between the attenuation at the monitor and control microphones depends significantly on the placement of the control sources and the control microphones. This
result demonstrates the importance of the locations of these sources and sensors in order to obtain high overall noise reduction [3].

Figure 6 shows a diagrammatic representation of the Sound Pressure Level (SPL) in the control microphones with the ANC system on and off. The diagram shows the noise attenuation obtained at each control microphone. The levelling out of the sound field when using ANC is also clearly seen in this figure. A mean attenuation of 18.4 dB was obtained across these microphones. The optimal theoretically-calculated attenuation was 22.7 dB. This result indicates that the controller performs well under this ideal running condition. A broad agreement between the calculated and obtained noise reductions has also been observed in off-line computer evaluations based on recorded noise and synchronization signals as well as measured control paths [9].

![Figure 6: Performance of the ANC system (synchronized propellers). Mean attenuation in the control microphones: 18.4 dB.](image)

In this case with identical propeller rotational speed it is interesting to note that the twin-reference controller converges nicely, while in fact there is redundant information in this condition; two individual (but identical) controllers use similar inputs to control the same signals. Therefore, in this condition with fully synchronized propellers a single-reference controller is sufficient [9]. Such a controller uses a single synchronization signal from one of the propellers to obtain information about the noise to be reduced. In applications where a possible unsynchronization may occur between the noise sources, a multiple-reference controller is, however, more preferable to a single-reference in order to efficiently reduce the noise level under different conditions.

Owing to the structure of the controller and the adaptive algorithm, rapid convergence was realizable. This ability for rapid convergence enables the control system to track changes in the sound. This type of controller is favorable for attenuating noise from periodic sources running at moderately different rotational speeds, as observed, for example, when the fields generated by the two propellers beat together. Figure 7 presents a diagrammatic representation of the mean SPL for each control microphone averaged over a time period of 5 seconds for conditions with a constant beat frequency of 0.5 Hz. The levelling out of the sound field caused by the control system is also clearly seen here. Similar results was observed for beat frequencies of 1 and 2 Hz. Table 1 summarizes the mean noise attenuation averaged over all control microphones. The results demonstrate significant noise reduction. Note that the reduction decreases with increased beat frequency, which is in accordance with theoretical results [15]. A comparison with the fully synchronized case shows a degradation in the noise attenuation of approximately 4 dB, due to the beats. By using a larger step-size parameter an improved performance could also be obtained in conditions with beating sound fields [15]. The experiment showed despite the beating the controller proved to be stable and converged robustly.
Figure 7: Performance of the ANC system (constant beat frequency of 0.5 Hz). Mean attenuation in the control microphones: 15.3 dB.

<table>
<thead>
<tr>
<th>Beat Frequency [Hz]</th>
<th>0</th>
<th>0.5</th>
<th>1</th>
<th>2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mean Attenuation [dB]</td>
<td>18.4</td>
<td>15.3</td>
<td>14.9</td>
<td>14.7</td>
</tr>
</tbody>
</table>

Table 1: The mean attenuation in the control microphones of the BPF for cases with constant beat frequencies

The noise attenuation in the control microphones obtained for conditions with a time-varying beat frequency excitation, in this particular case a sinusoidally varying, is shown in Table 2. In these cases a mean noise attenuation of approximately 8-13 dB was obtained depending on the beat frequency and the period time of the variation. A comparison with the constant beat frequency situation demonstrates a reduced noise attenuation of several dB for the cases of time-varying beat frequencies. The results show a better noise attenuation for slowly varying conditions than for rapidly changing conditions. The decreased noise attenuation results because the FFT-based reference generator introduces a time delay in the reference signals of approximately 32 ms. The controller remained stable and converged robustly to a constant mean level despite the delay in the reference signals and the time-varying beats. In non-stationary conditions this delay implies decreased correlation between the reference signals and the noise, resulting in degraded tracking performance and reduced noise attenuation. However, this delay will not cause any problems under stationary conditions. Under such conditions it is always possible to find a correlation between periodic signals of the same frequency, irrespective of delays. In such cases the time delay of the reference signals will thus not reduce the noise attenuation.

The synchronization signals delivered by the noise generation system contained a single pulse per "propeller revolution." Reference signals generated using such synchronization signals may have slightly time-varying phase angles ("phase jitter"), which may degrade the performance of the ANC system. In the cases of a time-varying synchrophase angle a larger amount of phase jitter was introduced in the reference signals. The phase jitter as well as the time-delay in the reference signal introduced by the FFT-based reference generator implied that the controller never converged completely, since it constantly reacted to the variations.

<table>
<thead>
<tr>
<th>Beat Frequency $\Delta f_{\text{max}}$ [Hz]</th>
<th>0.5</th>
<th>0.5</th>
<th>1</th>
<th>1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Period time $T$ [s]</td>
<td>1</td>
<td>5</td>
<td>1</td>
<td>5</td>
</tr>
<tr>
<td>Mean Attenuation [dB]</td>
<td>11.6</td>
<td>12.6</td>
<td>7.5</td>
<td>10.0</td>
</tr>
</tbody>
</table>

Table 2: The mean attenuation in the control microphones of the BPF for cases with time-varying beat frequencies: $\Delta f=\Delta f_{\text{max}}\sin(2\pi t/T)$. 
Summary and Conclusions

This paper presents results from a practical experiment where the performance of the multi-reference actuator-individual normalized filtered-x LMS algorithm has been evaluated. The evaluation was performed within a fuselage section from a propeller aircraft. In order to simulate the propeller noise produced by two rotating propellers 12 loudspeakers mounted around the exterior of the fuselage were used to excite the structure. The ANC system used 8 control sources and 11 control microphones.

The controller exhibits good performance with respect to convergence rate, tracking and robustness, and the interior noise level was considerably reduced. The mean attenuation of the BPF (82 Hz), under stationary "flight" conditions with fully synchronized propellers, was approximately 18 dB in the control microphones, and 10 dB in the monitor microphones located at the passenger head positions. In cases with unsynchronized propellers, however, the attenuation achieved decreased typically 3-6 dB due to beating. In the different cases the controller converged robustly, and the behavior of the controller displayed in these tests agreed with the behavior obtained from off-line computer simulations. The noise reduction measured was in broad agreement with the optimum theoretical reduction, calculated by using the recorded primary noise and measured frequency responses. The performance degradation under non-stationary conditions is partly due to the time delay introduced by the FFT-filter bank used to generate the internal complex reference signals. Increased performance in non-stationary conditions could be obtained if the reference signals are generated with as short inherent delay as possible, e.g. using lookup tables instead of the sliding FFT-operation in the generation process.

In conclusion, the multiple-reference actuator-individual normalized filtered-x LMS algorithm exhibited high performance in a practical acoustic environment. Significant noise attenuation was obtained. The system was robust and fast convergence was observed for stationary as well as non-stationary conditions.

References