Active Noise Control: Open Problems And Challenges

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Abstract

This paper presents dynamic commotion control (ANC) methods zeroing in on difficulties in creating functional applications and open issues for research from the sign handling point of view. We propose progressed techniques to further improve ANC execution that incorporate virtual detecting, remaining commotion concealing, and dynamic sound quality control. We additionally present a few difficulties for certifiable applications such as dropping high-recurrence and motivation like clamors, and diminishing clamors produced from moving sources. At long last, ANC applications in buyer items and medical services gadgets are used to exhibit some possible extra capacities for advancing savvy ANC items.

Introduction

Acoustic noise problems have become serious with the increased use of industrial equipment, such as engines, fans, blowers, transformers, and compressors. It is especially prominent in transportation systems (e.g., vehicles, trains, airplanes, and ships), manufacturing plants, electrical appliances (e.g., air-conditioners, refrigerators, washing machines, and vacuum cleaners), medical equipment (e.g., magnetic resonance imaging (MRI) systems, and infant incubators), and human activities (e.g., crowded public spaces, offices, and bedrooms). Traditional acoustic noise reduction techniques are based on passive noise control, such as earplugs, ear-protectors, sound insulation walls, mufflers, and sound-absorbing materials. These passive techniques are effective for reducing noise over a wide frequency range. However, they require relatively large and costly materials, and are ineffective at low frequencies.

Therefore, the active noise control (ANC) [1–8] proposed in the early 20th century, has gained intensive development in the last two decades to reduce low-frequency noise. The ANC technique using a loudspeaker to generate anti-noise sound was first proposed in the 1936 patent by Lueg [9]. ANC is an electro-acoustical technique based on the principle of superposition, that is, an anti-noise with the same amplitude but opposite phase is generated by secondary source(s) to cancel unwanted (primary) noise acoustically, thus resulting in reduced residual noise. The ANC system is very efficient for attenuating low-frequency noise in environments where the passive noise control techniques are expensive, bulky, and ineffective. In practical applications, the characteristics of the noise source and acoustic environment are changing, and thus the frequency content, amplitude, and phase of the primary noise are also changing. The noise reduction performance is mainly dependent on the accuracy of the amplitude and phase of the anti-noise generated by a signal processing algorithm. To deal with these time-varying issues, most ANC systems utilize adaptive filters [10–12] to track these variations and unknown plants.

The most commonly used adaptive filters are realized using a finite impulse response (FIR) filter with the least-mean-square (LMS) algorithm[10].

The development of powerful, low-cost digital signal processors (DSPs)[13–15] encourages the implementation of advanced adaptive algorithms to achieve faster convergence, increased robustness to interference, and improved system performance for practical ANC applications. The control structure of ANC is generally classified into two classes: feedforward control and feedback control. In the feedforward control case, a reference noise is assumed to be available for the adaptive filter. Feedforward ANC systems can be categorized as either a broadband or a narrowband depending on the type of primary noise that can be reduced. In the broadband feedforward control case, a reference noise is detected by a reference sensor (e.g., microphone), and thus noise correlating with the reference noise can be reduced. On the other hand, in the narrowband feedforward control case, a reference signal is internally generated using information available from a reference sensor (e.g., accelerometer) that is not affected by a control field. The feedforward ANC scheme utilizes a secondary loudspeaker (e.g., actuator) to generate anti-noise and an error sensor (e.g., microphone) to pick up residual noise, which serves as the error signal for updating the adaptive filter. The single-channel feedforward ANC scheme, which consists of two sensors (reference and error) and an actuator, is widely used for industrial applications such as reducing duct noise[16]. The feedback ANC system uses only an error sensor and a secondary source, not using an "upstream" reference sensor.

Analog feedback control based on a simple negative feedback is widely used in headphone applications[17–20]. Unfortunately, the controllable bandwidth is limited by the throughput of the overall control system; thus, it is difficult to reduce broadband noise. Digital feedback control generally utilizes internal model control (IMC)[21,22], which minimizes residual noise using predicted primary noise as the reference signal. Hence, the IMC-based feedback ANC system can reduce only predictable noise (including sinusoidal, narrowband, and color noises). The bandwidth that can be controlled by the feedback ANC system is limited because of the large delay due to the analog-to-digital converter (ADC) and digital-to-analog converter (DAC). Today, successful real-world ANC application is still limited owing to the effectiveness of signal processing algorithms, physical implementation constraints, and economical consideration. Recently, many advanced signal processing algorithms, implementation techniques, and successful applications of ANC have been reported. In this overview paper, we will focus on introducing some new signal processing algorithms, discussing challenges for innovative applications, and proposing open issues for further research and development of ANC systems. This paper is organized as follows. In Section II, the basic structures and algorithms of ANC will be introduced. ANC systems include broadband and narrowband feedforward ANC, adaptive feedback ANC, hybrid ANC, multiple-channel ANC, and audio-integrated ANC. Convergence analysis of the filtered-x LMS (FXLMS) algorithm including recent published works will be given. Different algorithms and structures including nonlinear ANC will be briefly overviewed, while citing many important recent works. Furthermore, some new approaches including active noise equalization, psychoacoustics, and virtual sensing will be introduced. In Section III, the basic principles of and recent works on online secondary-path modeling, and ANC algorithms that do not require a secondary-path model will be discussed. Finally, several real-world ANC applications with challenging issues will be introduced in Section IV. In this paper, we choose topics related to our works in this field. Some important results may be omitted owing to page limitation. Readers can therefore refer to many recent works reported in the last decade that are cited as references.

ANC Systems and Open Problems

The basic single-channel ANC system using the FXLMS algorithm is illustrated in Fig.1[2], where the reference signal x(n) is picked up by a reference sensor. The reference signal is processed by the adaptive filter W(z) to generate the canceling signal y(n) driven by a secondary loudspeaker. The error sensor is used to monitor the ANC performance by sensing the residual noise e(n). The use of the adaptive algorithms for ANC systems is necessary to compensate for the secondary path S(z)[2].

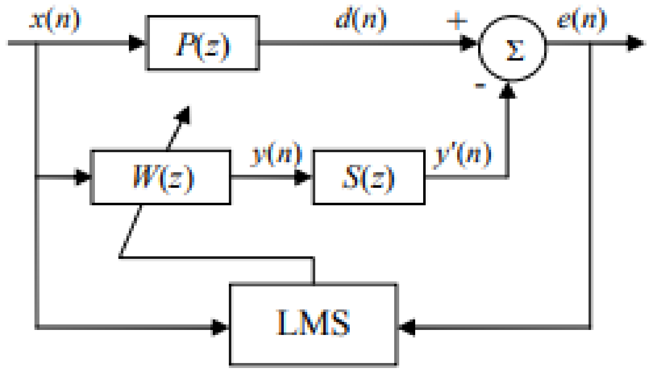

Fig. 1 Basic single-channel ANC system using the FXLMS algorithm

The most popular filter structure used for ANC systems is the FIR filter given in (1). It can be replaced by infinite impulse response filters, lattice filters, transform-domain filters, subband filters, etc. [2]. The convergence rate of the 164 FXLMS algorithm given in (2) can be improved by using variable step sizes, recursive least square algorithms, affine projection algorithms, Kalman algorithms, etc. In addition, nonlinear adaptive filters with associated algorithms can be used to reduce nonlinear effects caused by different factors. It is still an area of open research to develop more advanced adaptive algorithms in order to further improve the performance of ANC systems. The FXLMS algorithm expressed in (2) and (3) is very tolerant of errors made in the estimation of S(z) by the filter S (z). Within the limit of slow adaptation, the algorithm will converge with nearly 90o of phase error between S (z) and S(z). Therefore, off-line modeling can be used to estimate S(z) during the initial training stage for most ANC applications, however, on-line modeling may be required for some applications that involves fast changing environments. The detailed procedure for off-line modeling and some on-line secondary-path modeling algorithms are summarized in [2]. The development of robust and efficient on-line secondarypath modeling algorithms still deserves further research. In particular, development of on-line modeling techniques without using additive noise is critical. The acoustic ANC system shown in Fig. 1 uses a reference microphone to pick up the reference noise. Unfortunately, the anti-noise generated by the secondary loudspeaker also radiates upstream to the reference microphone, resulting in a undesired acoustic feedback that corrupts the reference signal x(n). One solution for eliminating acoustic feedback is to use a feedback neutralization filter [2]. Since the primary noise is highly correlated with the anti-noise, the on-line adaptation of the feedback neutralization filter must be inhibited when the ANC system is in operation. Therefore, analysis on the effects of acoustic feedback and the development of effective on-line feedback-path modeling techniques still remains open for further research[11].

B. Narrowband Feedforward Systems A narrowband ANC system uses an internally synthesized reference signal x(n). Two types of reference signals are commonly used in narrowband ANC systems:

(1) an impulse train with a period equal to the inverse of the fundamental frequency of the periodic noise [12], and (2) sinusoidal signals that have the same frequencies as the corresponding harmonic components [13]. A digital recursive quadratic oscillator generates two orthogonal cosine and sine components. These two signals are separately weighted and then summed to produce the canceling signal y(n). In practical applications, periodic noise usually contains multiple narrowband components. This requires higher-order adaptive filters that can be implemented by using direct, parallel, direct/parallel, or cascade forms [2]. Analysis of narrowband ANC systems is usually based on a single-frequency case. The analysis and optimization of widely-used parallel and other forms for multiple-frequency cases is open for research. Also, analysis of using single error signal e(n) to update multiple adaptive filters deserves further study and improvement. In addition, a narrowband ANC system assumes the reference signal x(n) has the same frequency as the primary noise d(n) at the error sensor location. In many practical applications, the reference sinusoidal frequencies used by the adaptive filters may be different than the actual frequencies of primary noise. This frequency mismatch will degrade the performance of ANC systems, and these effects have been analyzed recently[14]; however, effective solutions remain to be developed for practical applications.
C. Adaptive Feedback ANC A single-channel adaptive feedback ANC system synthesizes (or regenerates) the reference signal x(n) based on the adaptive filter output y(n) with the secondary-path model S z( ) and the error signal e(n). Thus, this technique is also known as internal model control[15]. We can estimate the primary noise d(n) exactly if Sz Sz () () ≈ , and use this estimated primary noise as the reference signal x(n). This adaptive feedback ANC algorithm has many advantages and can be applied in ANC headsets and other industrial applications[16]. There are many open problems for further study of adaptive feedback ANC systems. For example, what kinds of noises can be effectively reduced by the adaptive feedback ANC, the effects of secondary-path modeling error on the accuracy of synthesized primary signal x(n) and the performance of the ANC system, and the stability and convergence rate of the systems.

MRI ANC systems

Recently, MRI equipment, which is used to take images of human organs, has been in use in many medical institutions. Some MRI devices have also been introduced to conduct microwave coagulation therapy using near-real-time MR images. However, MRI equipment generates intense noise because the gradient coil vibrates owing to Lorentz force. Exposure to the intense noise may cause patients and medical staff to suffer extreme stress and may prevent verbal communication between them [237]. Many approaches to reducing MRI noise have been developed. These approaches include passive noise control, the design of silent MRI pulse sequences, and ANC. Passive noise control uses earplugs or ear protection, which is only effective for high-frequency noise. Unfortunately, MRI noise has a high sound pressure level (SPL) at low frequencies. Moreover, passive noise control devices prevent verbal communication among patients and medical staff during operation. The design of silent MRI pulse sequences relies on selecting imaging parameters to reduce related acoustic noise, which results in an approximately 20 dB attenuation of the SPL. However, this technique limits imaging sequences and reduces image resolution. The ANC system offers an effective technique for reducing MRI noise. The application of ANC to MRI noise has been reported in 3–6], and an approximately 15–30 dB noise reduction has been achieved. However, these studies have some limitations. Firstly, the experiment was conducted using computer simulations or in a laboratory setup, not in actual MRI rooms. Secondly, these studies used a headset-based system that has problems of preventing verbal communication between the medical staff and the patients, causing pressure on the user's ears, and separating the patient from the outside acoustical environment. Finally, most of the studies utilized the feedforward ANC system with the FXLMS algorithm. The feedforward ANC system requires at least two microphones (reference and error) and sufficient distance between the two microphones to ensure causality, thus increasing the size of the system. Moreover, the reference noise sensed at the reference microphone must highly correlate with the noise picked up by the error microphone. In practical applications, it is very difficult to determine the reference microphone position because the MRI equipment is very large and the noise source is unknown. A novel ANC structure that can achieve high noise reduction regardless of the user's movements that allows for clear verbal communication, and that can be comfortably mounted without pressure on the ears is proposed in [245]. The ANC structure consists of two microphones arranged near the opening of the user's external auditory canal and two loudspeakers located close to the user's ears, as shown in Fig. 11. This structure is called a head-mounted ANC system to distinguish it from the headset-based ANC systems. That is, the head-mounted ANC system does not cover the user's ears, so that the user can verbally communicate with other people while reducing unwanted MRI noise. Hence, this ANC system is utilized for medical staff working in the MRI room. The head-mounted ANC system utilizes optical microphones and piezo-electric loudspeakers to realize noise reduction in a high magnetic field and compensate the low SPL of the piezo-electric loudspeakers because the loudspeakers are close to the user's ears. This ANC system uses the IMC-based feedback system (introduced in Section II(E)), which can reduce predictable noise, is independent of the direction of arrival of noise, and is small in physical size as compared with the feedforward ANC system because the reference sensor is not needed. This ANC system can effectively reduce MRI noise in the frequency range between 500 and 2500 Hz. Experimental results demonstrated that the ANC system can reduce MRI noise by approximately 20 dB at a high magnetic field in an actual MRI room. A challenging problem for MRI ANC is that the MRI noise contains periodical and impulse-like noise. Periodic components can be minimized by feedback ANC, but impulse-like components cannot be minimized. This problem can be solved by means of some advanced algorithms using the probability density function of impulse-like noise [247–252] for canceling impulse-like noise included in MRI noise. The second challenge is that MRI equipment generates intense magnetic fields. Therefore, it is necessary for transducers in the ANC system to satisfy several conditions: the transducer must work normally in an intense magnetic field and must not affect the MR image. Hence, transducers containing magnetic materials cannot be used for the MRI ANC system. In [245], an optical microphone and a piezo-electric loudspeaker were used to satisfy these conditions. However, a piezo-electric loudspeaker cannot generate sufficient SPL at low frequencies. Therefore, further development of an appropriate transducer is required.

Conclusion And Future Trends of ANC

In this paper, we briefly reviewed broadband and narrowband feedforward and adaptive feedback ANC systems with focus on signal processing algorithms. We focused on the introduction of the recent research and development in the last decade after detailed tutorial publications[1–8]. In particular, we introduced the audiointegrated algorithm and the concepts of psychoacoustics and virtual sensing for ANC. In this paper, we also comprehensively reviewed online secondary-path modeling techniques and ANC without the secondary-path model, which remain critical for some practical applications. Finally, we highlighted some ANC applications in medical and consumer electronics fields, which are important for motivating new ANC applications in addition to traditional applications in industry and transportation. We also identified many related difficulties and open research issues in each section. There are many challenges[8] in developing successful ANC applications: (1) theory of associated acoustics related to ANC algorithms and the positioning of transducers for optimum performance; (2) development of fast convergence and robust algorithms to achieve maximum noise reduction at desired locations in time-varying environments; and (3) implementation considerations including system complexity, physical constraints, and cost reduction. In addition to these requirements, integrating value-added functions, such as audio and communications into ANC can further promote the development of other applications. Finally, many consumer products may prefer ANC systems to mask and control the spectral contents of residual noise over simple noise reduction.

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