Active Noise Control: From Analog to Digital – Last 80 Years

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Abstract - Beginnings in the field of active noise control date from 1930s. From these pioneer attempts great deal of progress has been achieved. In this brief review paper advances from simple fixed analog systems, manually adaptive analog systems, acoustic feedback neutralization, fully adaptive digital systems with secondary path modeling and multichannel systems are briefly presented. Applications of active noise control in closed and open space, industrial and automotive applications, as well as active noise canceling headphones and new solutions with smart materials are also briefly mentioned.

I. INTRODUCTION

Active noise cancelation (ANC) is not quite a new concept. The first experiment on superposition of sound fields was supposedly made in 1878 by Lord Rayleigh using two electromechanically synchronized tuning forks, [1]. First humble beginnings that specifically targeted cancelation of unwanted noise exploiting the destructive interference of sound started in 1930s, [2, 3, 4]. Although this principle is quite simple, its practical implementations in real world encountered numerous problems that had to be solved. Early systems were analog, with no or manual adaptation that could only partially benefit from theoretical ideal. Later, with the advent of digital signal processing quite sophisticated adaptive algorithms for ANC have been developed. As with any other technical field there was a hype cycle with initial exaggerated expectations, followed by the bitter disillusionment, then by enlightenment and finally reaching the plateau of productivity. Early unrealistic expectations included possibility of silencing large spaces. Today technology is quite successful in forming smaller zones of silence around the head or even smaller spaces as in ANC headphones. Silencing of larger spaces is much more difficult to deal with, with very limited success even today.

II. BASIC PRINCIPLE

Active noise cancelation applies principle of destructive interference with the introduction of canceling acoustic signal that is of the same amplitude but opposite phase of the unwanted noise, Fig. 1. Resulting destructive interference lowers the noise levels in a protected space. It is of paramount importance that canceling signal has exactly the same amplitude and exactly the opposite phase. This is not always easy to achieve, particularly in nonstationary signals. Even small discrepancies between amplitudes and phases of original noise and canceling signal severely diminish efficiency of the method, [4].

III. ANALOG SYSTEMS

In early years of ANC, all solutions had to be implemented using analog technology. The first patents explicitly oriented on the idea of active noise control by destructive interference with the sound wave of opposite phase have been granted to Henri Coanda, [2], in 1934, however Coanda’s concept of realizing his objectives were technically incorrect. Few weeks later Paul Lueg also applied for a patent of similar goals, [2, 3].

A. Lueg’s Patent

The design of acoustic ANC utilizing a microphone and an electronically driven loudspeaker was first proposed in 1936 patent by Lueg [2, 3], Fig. 2, [3]. Block diagram of his system with additional legends is shown in Fig. 3. The Lueg system is a monopole consisting of a
microphone, amplifier and loudspeaker (secondary source). An acoustic noise is picked up by a microphone, whose signal is processed through an electronic device to create cancelling signal at a loudspeaker that opposes to the primary noise, [5]. Electronic device includes phase inverter with a time delay. Signal is further amplified and led to the loudspeaker disposed downstream from the microphone. At the position of the loudspeaker noise and "antinoise" cancel each other and form a quite zone. This type of system is called feedforward. It senses noise before it passes secondary source. Unfortunately Lueg was never able to demonstrate his idea successfully because it was oversimplified and field of electronics was not sufficiently advanced, [6].

B. Patent by Olson and May

In 1953 Olson and May carried their researches on developing an electronic sound absorber, which appeared to be successful over small volumes in a unidirectional sound field, [2, 4, 7, 8], Fig. 4. This type of system is called feedback system, Fig. 5. It cancels noise without the benefit of an upstream reference input. Previously described systems were not adaptive.

C. Conover System

In 1956, Conover described an active system for reducing noise radiated by large transformers [2, 4, 7, 9, 10], Fig. 6. It consisted of three parallel independent channels, realized with three band pass filters, as can be seen in a block diagram shown in Fig. 7.

IV. ADDRESSING FEEDBACK IN ANALOG SYSTEMS

In ANC systems there exists an acoustic feedback (hallowing) between the canceling source and reference microphone causing instability in the system. Canceling signal should be introduced in such a way that over the intended frequency range it propagates mainly in the direction of original noise. The influence of acoustic feedback could be diminished by the use of directional microphones and speakers as well as with specific loudspeaker and microphone arrangements. Most popular geometric arrangements are Chelsea dipole, Swinbanks two-element unidirectional source and Jessel-Mangiante-Canévet tripole source, [4, 5, 12].

A. Chelsea Dipole

The Chelsea dipole, developed in 1976, Fig. 8, uses a pair of loudspeakers driven out of phase and physically spaced one-half wavelength apart around the microphone situated centrally between them, [5]. The sound radiated from secondary sources cancels each other at the position of the microphone, while simultaneously providing noise attenuation in a region away from the noise source.
B. The Swinsbank System

The Swinsbank system, developed in 1973, Fig. 9, uses a pair of loudspeakers that are arranged in such a way that delay and phase between the sources produces cancellation in the upstream direction and addition in the downstream direction, thus reducing the feedback from the loudspeakers to the sensing microphone, [5]. It employs electrical delay between the two out-of-phase loudspeakers to be equal to the propagation time between the speakers, [4, 5, 12].

C. Jessel-Mangiante-Canévet Tripole

The Jessel-Mangiante-Canévet tripole system, from 1972, Fig. 10, uses a monopole source to cancel upstream propagation from the dipole source, eliminate acoustic feedback and produce unidirectional source, [5, 12].

Fixed filter systems have basic limitations. They use secondary source spacing based on the wavelength and are limited to a specific frequency range and sound speed that vary with the temperature and humidity, [5]. To solve this problem it is necessary to introduce into a system some kind of adaptation. This became possible with the advent of digital technology. Due to complexity of adaptive algorithms, practically all adaptive filters are digital.

V. DIGITAL SYSTEMS

The need for very accurate phase and amplitude of canceling signal leads toward application of the digital signal processing, (DSP) that become popular in 1980s with the introduction of inexpensive hardware (first DSP chips by Intel, AMI, NEC and TI). Use of adaptive digital filters for ANC was first introduced by Burgess [13]. An adaptive filter, is a system with a linear filter Fig. 11, that has a transfer function controlled by variable parameters and a means to adjust those according to an optimization algorithm, [14]. Output of the filter \( y(n) \) is given by (1), where \( w_i \) are filter coefficients and \( x(n) \) is the input signal.

\[
y(n) = \sum_{i=0}^{k} w_i x(n-i)
\]  

(1)

The filter can be adapted by varying its weight factors by some algorithm, of which the most popular is the LMS algorithm, [14]. The LMS algorithm was introduced by Widrow and Hoff in 1959 as an adaptive algorithm, which uses a gradient-based method of steepest decent. Adaptation of filter coefficients \( w_i \) at the moment \( n+1 \) is given by (2) and (3), where \( \mu \) is a step size, \( x(n) \), \( y(n) \), \( d(n) \) and \( e(n) \) are signal, filter output, desired filter output and error at the moment of the sample \( n \).

\[
e(n) = d(n) - y(n)
\]

(2)

\[
w_i(n+1) = w_i(n) + \mu e(n) x(n)
\]

(3)

Similarly to analog systems digital systems can also be feedforward or feedback, however this time used filters are not fixed but adaptive, [15, 16].

A. Feedforward System

In a feedforward system the signal from the reference microphone is processed by an adaptive digital filter to produce a cancelling signal, Fig. 12. Essentially it is the system identification approach where the acoustic path is modeled with an adaptive filter, [15].

Real systems face two additional problems, the influence of the secondary path and an acoustic feedback. Loudspeaker and amplifier that drives it, form the secondary path that has a transfer function \( S(z) \). The influence of the secondary path \( S(z) \) is corrected with the introduction of the filter \( \hat{S}(z) \) that filters the input signal \( x(t) \), Fig. 13, hence the name Filtered-X LMS algorithm (FXLMS), 1981, [15]. \( \hat{S}(z) \) provides also accurate time alignment of referent and error signals that are used by the adaptive algorithm. Accurate modeling of the secondary path is important to avoid the degradation in performance of the ANC system. The secondary path can be estimated off-line or on-line method. FXLMS adaptation is given by (4), [15]:

\[
e(n) = d(n) - y(n)
\]

(4)
\[ w(n+1) = w(n) + \mu e(n) k(n) \]  

(4)

Reduction of feedback distortion is achieved by the introduction of filter \( F(z) \) for the feedback neutralization which models the feedback path \( F(z) \), as shown in Fig. 14.

Figure 14. Feedforward system with secondary path modeling and feedback neutralization

B. Feedback Systems

Block diagram of a feedback system is shown in Fig. 15. In the feedback system the signal from the error microphone is processed by an adaptive digital filter to produce a cancelling signal, [15]. Under simplified (idealistic) assumption the secondary path \( S(z) \) can be approximated with a delay.

Figure 15. Feedback system as an adaptive predictor (simplified idea)

The feedback system with included secondary path modeling is shown in block diagram in Fig. 16, [15].

Figure 16. Feedback system with secondary path modeling

C. Synchronized Waveform Generator

Cancellation of periodic noise can be achieved by a synchronized waveform generator, Fig. 17. ANC using a waveform synthesizer was developed in early 1980s by Chaplin and coworkers, [15, 17]. Such system is suitable only for periodic (tonal) noise, present in fans, rotating machinery noise, etc. It uses a waveform generator with stored waveform samples, (5), synchronized by tacho signal (i.e. from non-acoustic sensor) and an adaptive algorithm to further modify period amplitude and phase of generated waveform. Advantage of the method is that a cancelation signal is produced by synthesis and does not depend on a measurement of original noise, hence there is not a problem of unwanted acoustic feedback. A synchronous waveform generator can be considered as a particular case of a feedforward system.

D. Hybrid Systems

A hybrid ANC system, Fig. 18, combines feedforward and feedback control structures. Feedforward structure cancels noise correlated to the reference signal, while feedback structure cancels predictable components of the primary noise not observed by the reference sensor [15].

Figure 17. Narrowband feedforward ANC using the synchronized waveform generator

\[ \{w_l(n), l = 0, 1, ..., L - 1\} \]

(5)

\( L \) is the period length

Figure 18. Principle of hybrid ANC system

E. Multichannel Systems

Noise within vehicles cabins has more complicated sound field than in ducts. Minimizing the total acoustic energy in a protected space can be achieved with a multichannel ANC system. To accommodate multiple channels (one or more reference microphones, multiple error microphones and multiple secondary sources), a multichannel system, Fig. 19, for adaptation uses extension of the single channel LMS algorithm, [15].

Figure 19. Multichannel system

VI. APPLICATIONS

The most successful early use of ANC is control of noise in ducts, exhaust pipes, and headphones. Later came ANC in enclosed spaces such as vehicle cabins. These and other applications are listed here, [2, 16].

A. Ventilation Ducts

The active cancellation of 1-D sound fields that approximate sound in ducts has been first suggested by Lueg [3, 4]. It is particularly suitable for HVACs and
ANC in a duct has an advantage over passive plenum silencers because it doesn’t introduce a pressure drop that must be tolerated or compensated by the more powerful fan operation, Fig. 20.

**B. Vehicle Interiors**

ANC has found its application in cars, [19], (first used by Nissan in Bluebird ARX-Z, 1992) and SUVs for the suppression of low frequency rumble in a cabin. Often these solutions use a synchronized waveform generator. Example of a car ANC system is shown in Fig. 21.

**C. Interior Spaces**

ANC of large spaces is difficult to achieve. Some more success can be achieved by active modal control in rooms by placing a secondary source in a corner of a room. Helmholtz resonators (HR) are sometimes used for narrowband passive reduction of noise. Semi-active approach with an adaptively tuned HR adapts to variations in noise while keeping a low energy consumption compared to fully active ANC systems, [22]. A system in Fig. 23 modifies the boundary impedance in the resonator.

**D. Open Spaces**

Radiation of noise in an open space can be attenuated by active acoustic barriers, [4, 23], or shields placed around the source, [4]. One type of system for attenuation of large transformer noise is shown in Fig. 24, [24].

**E. ANC Headphones**

Earliest ideas about active ear defenders originated in the mid-1950s. Most notable advances in a field of noise canceling headphones have been made in late 1970s and 1980s by Amar Bose. In 1986, the first wearing prototype headphones that used active noise cancellation has been developed. Few years later, in 1989, the first ever commercially available noise-cancelling headphones have been produced. The cancelling signal is emitted by the loudspeaker within the earcup and the quiet zone is supposed to be located around the ear drum, Fig. 25, [25].

ANC headphones are mainly implemented as analog devices driven by simplicity, small size and cost, [7]. Analog ANC headphones typically achieve an active attenuation of about 20 dB in 100 to 200 Hz frequency range, and no active attenuation bellow about 30 Hz and above about 1 kHz, [26]. Outside this frequency range attenuation depends on used passive methods.

**F. Household Appliances**

Application of ANC in domestic appliances is a promising future field. Experiments have been done with application of ANC in a vacuum cleaner that is known as particularly noisy appliance. Passive means of noise attenuations are not quite suitable as they would disrupt airflow and deteriorate cleaner’s performance. ANC setup for a vacuum cleaner is shown in Fig. 26, [27].
attenuation has been achieved at tonal components (order of 10 dB), however it translated to a much lower value (about 2 dB) over the whole frequency spectrum. Similar approaches are proposed for washing machines.

G. Smart Materials

By integrating sensors, electronics and piezoelectric actuators in smart material it is possible to produce ANC surfaces for the reduction of sound radiated from structures. These modern architectures (starting from mid 1990s) use low cost digital signal processors that are integrated directly as distributed architecture into a smart material. Individual elements can be interconnected in an array (or grid). The smart foam as a kind of hybrid active-passive absorbing material for enhancing acoustic absorption at low frequencies is show in Fig. 27, [28].

![Figure 27. Smart foams, adopted from [28]](image)

VII. CONCLUSION

In this short paper brief review of progress has been covered, from early ideas, analog and later digital implementations. Great progress has been made and with low cost digital hardware active noise control has become a reality for many applications. Main success has been achieved at attenuating low frequencies, particularly tonal components and forming local zones of silence. More modest results have been achieved for silencing larger interiors and open spaces. Early analog systems were nonadaptive or only manually adaptive. Clever geometric arrangements of microphones and loudspeakers were used to neutralize the acoustic feedback. Digital systems with adaptive filters adapt to changing environment conditions, incorporating feedback neutralization and secondary path modeling and achieving much better attenuations. Multichannel systems proved themselves useful for silencing interior spaces of vehicles. Most successful applications today are ventilation ducts, ANC headphones and reduction of vehicle interior noise (turboprop aircrafts, luxury cars and SUVs). In future we may expect to find ANC in some household appliances, like vacuum cleaners. Also, introduction of smart materials (e.g. smart foam) can enable silencing of low frequency noises emitted from larger structural surfaces.

REFERENCES