Introduction to active noise control

Aki Kivivirta
Aalto Universtiy
Master's Programme CCIS / AAT

aki.kivivirta@aalto.fi

Keywords — Active noise control (ANC), Least mean square (LMS), Gradient descent, Primary path, Secondary path

1 Introduction

Active noise control (ANC) is a technique used for attenuating unwanted noise. ANC is achieved by analyzing incoming sounds, comparing it to what is expected and trying to cancel the unwanted parts with a cancelling signal. In ANC, all unwanted sounds are considered noise. [1]

In the core of all applied ANC cases are a microphone, a processing chip, a loudspeaker and the area where the ANC is applied to. The microphone is capturing the sounds, which are processed in the chip and fed into the loudspeaker. When the processor receives the microphone signal it compares it to what is expected, would it be silence or music, and calculates the difference between the signals. This difference is called the error and the main task of the processor is to minimize it by modifying the signal fed to the loudspeaker. There are a variety of techniques for this, most common of which is the least mean square (LMS). As a whole, this is an iterative process and convergence of it depends on multitude of factors. [1]

The reason why ANC is so desirable is that it outperforms passive noise control at low frequencies. All objects that attenuate sound with absorption or attenuated reflection can be considered passive noise control. In practice this covers all objects, but some are better suited for attenuating sounds than others. The differences in how passive and active noise control attenuate will be discussed below. [2] [3]

2 Physics of sound

Sound is a pressure variant time dependent signal propagating in a medium. As this signal propagates it squishes and distances the small particles in the medium creating areas where the particle density is higher or lower than equilibrium. How sound propagates can be mathematically represented with wave propagation equation in three-dimensional space as

$$\nabla^2 p(x,y,z,t)\frac{1}{c^2}\frac{\partial^2}{\partial t^2}p(x,y,z,t) = 0 \tag{1}$$

where $p(x,y,z,t)$ represents the pressure at location (x,y,z) at time t, with the speed of sound c corresponding to the given medium, and ∇^2 the second order partial derivative. [1]

2.1 Frequency and phase of sound

The simplest sound component is a sine wave, because it has only one frequency. Commonly, sounds consist of multiple frequencies, but as the Fourier transform states, all signals can be broken down to their single frequency components. As discussed previously all signals can be combined by summing them and so a signal broken down to it's single frequency components can be summed up back together to bring back the original signal. [1]

The frequency of a sound is inversely proportional, to it's wavelength i.e the higher the frequency the shorter it's wavelength. In signals, peaks and valleys are anti-nodes while areas where the pressure doesn't change are nodes. This means that particles around the anti-node are displaced the most and the particles around nodes the least. Figure 1 shows how a single sine wave displaces particles in a uniform distribution of particles. [4] [5]

In linear systems, sound signals obey the super position principle. This means that when multiple signals collide, meaning that they are in the same location at the same time, their pressure values can be summed. In the case of ANC, the optimal cancelling signal would be the undesired noise but with opposite pressure changes, i.e opposite phase. Due to sound signals super position principle, this would even out the pressure changes and therefore eliminate the sound when summed in the correct locations. As stated previously when Equation (1) was discussed, pressure changes are linked to their three-dimensional location and time, meaning the has cancellation signal has to match the noise's location and time. Otherwise the cancellation signal will more likely create even more noise rather than cancelling it. [1] [6]

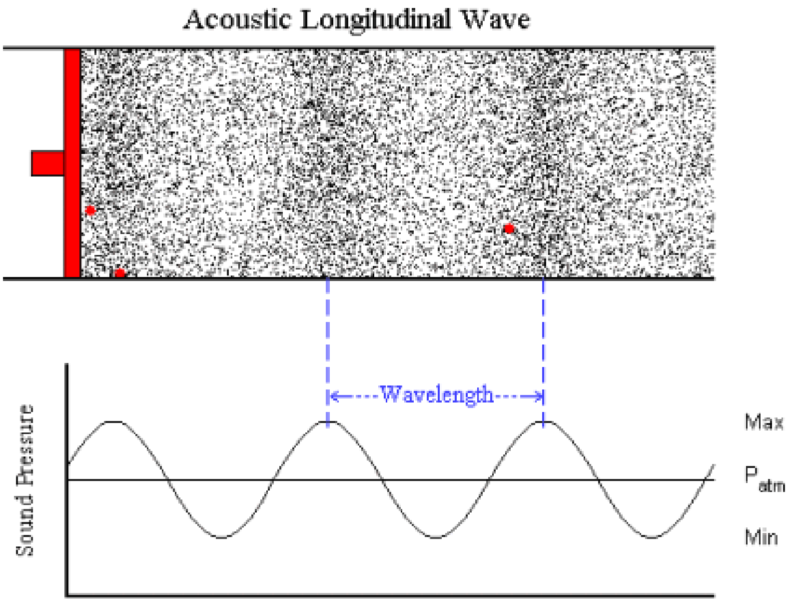

Figure 1: How a propagating pressure wave is represented as a signal in two dimensional coordinate system. [5]

2.2 Passive noise control

In passive noise control sounds are attenuated with absorption inside a medium and reflection on the border two different medium. As wavelength of the frequency components get longer the frequencies become lower. In passive noise control, absorptive elements are designed to have a rigid back and porous material on top. Sound waves are reflected off the rigid back and the porous material on top tries to limit the movement of the wave's anti-node, thus attenuating the wave. This limits the element's lowest possible attenuated frequency's wavelength to be one quarter wavelength of the original wavelength, since the first anti-node is at the end of the element and the node is located at the rigid wall. This is why, attenuating lower frequencies require larger attenuating elements. [4]

3 Basics in active noise control

There are two main methods in active noise cancellation, the feed forward and the feedback method. The following sections will discuss the differences in their structure and implementation. The signal processing parts, commonly noted as ANC in the block diagrams, will be discussed in the following chapter.

3.1 Feed forward method

The feed forward method has two main subcategories, broad-band and narrowband. Narrowband is only suited for periodic noises, such as motor noise, whereas broadband can be applied for a wider range of noises. Even though we will mainly

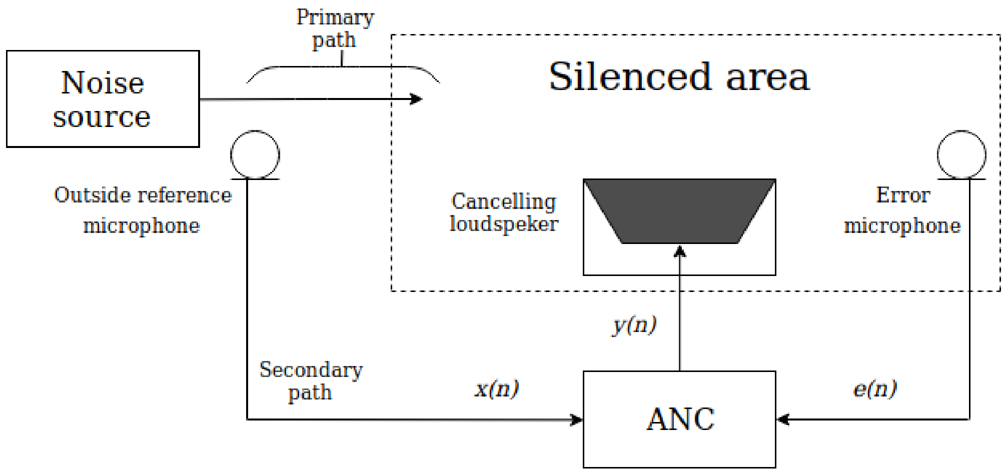

Figure 2: Simplified block diagram of a feed forward ANC system.

discuss techniques for feed forward broad-band ANC, the techniques presented can be applied for other ANC types, such as the narrowband feed forward method. [7]

In the feed forward method, a microphone outside the silenced area is listening to outside noise and another microphone in the silenced area monitoring the residual. Figure 2 shows a simplified feed forward ANC structure with an outside microphone, a loudspeaker and an error microphone. To simplify the diagrams we will only discuss single channel systems, meaning only single microphone and loudspeaker and simplified signal paths. In the simplified case of Figure 2, the outside microphone signal $x(n)$ is processed in the ANC block which produces a cancelling signal $y(n)$ that will cancel the noise. The ANC block and it's adaptive filter are controlled by both the outside monitoring microphone signal $x(n)$ and the inside error microphone signal $e(n)$. [7]

Primary path $P(z)$ is the path between the outside monitoring microphone and silenced area, which causes acoustic domain filtering to the outside undesired noise. In addition to filtering, it may have dynamic effects on the signal, which have to be taken into account by the adaptive filter noted as $W(z)$ in the block diagrams. [7]

After the noise has been captured by the outside monitoring microphone it has some path to propagate until it reaches the silenced area. In this path the signal can get filtered and thus it is different when compared to the one captured by the outside monitoring microphone. This path which causes filtering is called the secondary path and it is commonly noted as $S(z)$ in block diagrams. [7]

The secondary path can be modeled as a linear filter. The monitored signal has to be filtered with it, so that the cancellation signal would cancel the arrived noise as much as possible. In modern ANC applications multiple filtering effects from the acoustic and electrical domains, like digital-to-analog conversion, power amplifier, loudspeaker, error microphone, analog-to-digital converter can be combined in the

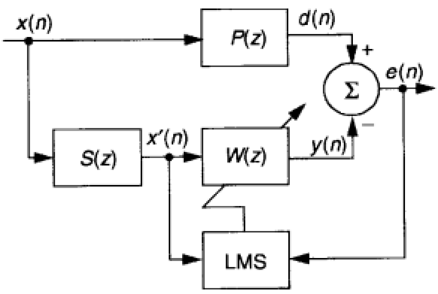

Figure 3: Block diagram of a feed forward ANC system with FXLMS algorithm. [7]

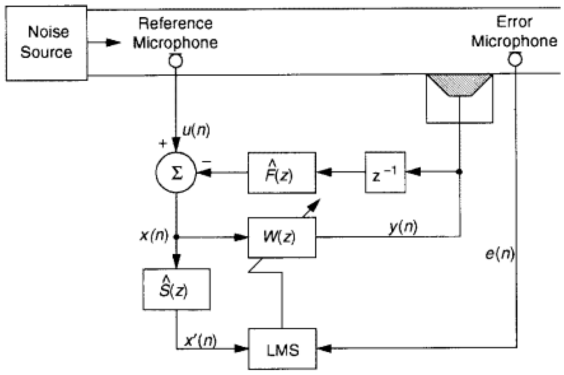

Figure 4: Block diagram of FXLMS with acoustic feedback neutralized. [7]

secondary path filter. The processes combined in $S(z)$ cannot delay the signal no more than $P(z)$, otherwise the magnitude of attenuation will suffer greatly. [7]

If the secondary path filter $S(z)$ is only applied for the cancellation signal, the outside reference signal will not be delayed like the other signals. This causes time alignment issues and ultimately feedback due to a bad adaptive filter. At first this problem was tackled with an inverse filter $\hat{S}(z)$, but later on was found that the original $S(z)$ filter could be used to filter both of the monitoring signal branches. This structure is known as the filtered-x LMS (FXLMS) algorithm. Figure 3 shows the block diagram of the FXLMS algorithm. [7]

Another problem with the simplified feed forward structure from Figure 2 is feedback caused from the loudspeaker's canceling signal bleeding to the outside monitoring microphone. If the cancelling signal is captured by the monitoring microphone it connects a feedback loop, which may cause instability to the entire system. One way to solve this is to have an inverse feedback control $\hat{F}(z)$ applied to the monitoring microphone in the electrical domain. Figure 4 shows an implementation of this feedback control method in a FXLMS system. [7]

3.2 Feedback method

In the feedback method there is a loudspeaker and only one microphone located inside the silenced area, as seen from Figure 5. This structure removes many of the features found in the feed forward method, such as the primary path $P(z)$ and acoustic feedback caused by bleed to monitoring microphone. [7]

The feedback method's adaptive filter coefficients are estimated based on the error signal $e(n)$ and the filtered error signal $y(n)$. Since the microphone is monitoring

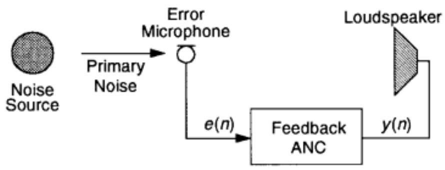

Figure 5: Block diagram of a single channel feedback ANC system. [7]

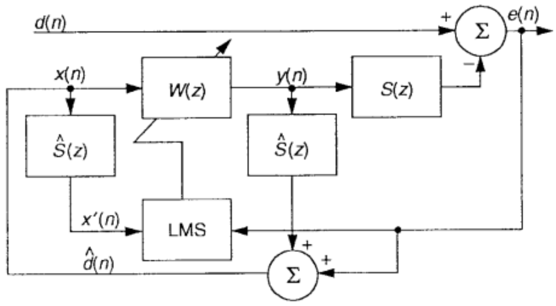

Figure 6: Block diagram of a feedback FXLMS ANC system. [7]

signals that have gone through the secondary path $S(z)$ the effects of the secondary path have to be canceled with an inverse filter $\hat{S}(z)$ in the electrical domain. [7]

In Figure 6, the primary noise is noted as $d(n)$ which consists of $e(n)+s(n)\circledast y(n)$. If the secondary filter and it's inverse are approximately the same, we can estimate the cancellation signal $\hat{d}(n)$ to be $e(n)+\hat{s}(n)\circledast y(n)$. This is known as the reference signal synthesis technique. Figure 6 shows how the FXLMS algorithm is implemented with the feedback method. [7]

4 Signal processing of cancellation signal

Active noise cancelling systems need to adjust to their environment constantly in order to filter noise effectively. Adaptive filters are used for this task. The next chapter will discuss how iterative optimization works, how it is related to adaptive filters and how adaptive filters are implemented.

4.1 Gradient decent

Gradient descent is a method for finding minima of smooth non-negative scalar functions. The positive gradient of a function gives the direction in which the function has it's largest growth. Opposingly, the negative gradient gives the function's steepest descent. Gradient descent is an iterative process which utilizes these mathematical properties. A function's gradient $\nabla f(x)$ is calculated and a step is taken in the negative gradient's direction, hence the name gradient descent. This process is done until convergence. Mathematically gradient descent is noted as

$$x_{n+1} = x_n - \mu \nabla f(x_n) \tag{2}$$

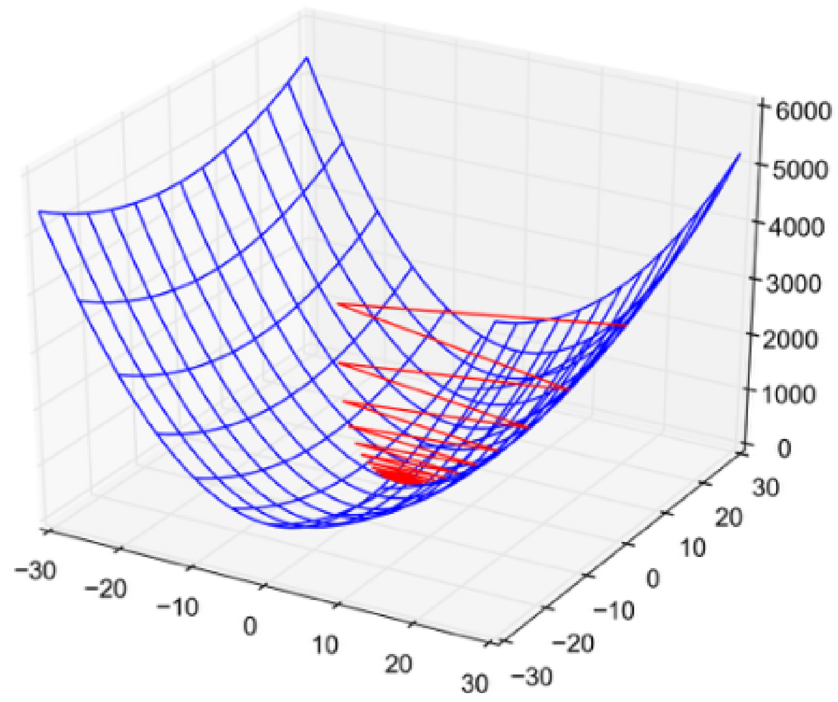

Figure 7: Shows gradient descent implemented with too large step size. [8]

where $\nabla f(x_n)$ is the gradient of the function $f(x_n)$, x_n is the current iteration's argument, x_{n+1} the next argument and μ the step size. [4] [8]

The convergence of this process depends on multitude of factors. The choice of step size μ depends on the case. If the step size is too small, the process will take too long, but if it is too large the process will not converge. Figure 7 shows how too large step size takes unnecessarily long to converge to the minimum. Different ways of optimizing the steps size will be discussed after LMS has been introduced. [8]

As a side note, even though gradient descent is visualized with simple two or three-dimensional graphs, it can be implemented for larger dimensionalities. Also, there is no guarantee that the minimum found by gradient descent is the absolute minima, but just a local minimum. [8]

4.2 Least mean square

Least mean square (LMS) is a way of implementing gradient descent, which has been found to work particularly well for adaptive filter optimization. In ANC adaptive filter optimization LMS is trying to minimize the residual noise after the expected signal has been removed. The error is squared, which yields it self for computationally efficient calculations, since the minimum of a parabola is easy to find. In mathematical notation, the error function is noted as

$$\hat{\xi}(n) = e^2(n) \tag{3}$$

where $\hat{\xi}(n)$ is the error function and $e^2(n)$ the residual noise. This error function gives a simple gradient function

$$\nabla\hat{\xi}(x_n) = -2f'(x)e(x_n) \tag{4}$$

where $f'(x)$ is the original function's derivative and $e(x_n)$ the original error. And finally the complete LMS algorithm which is almost the same as the original Equation (2). [7] [8]

$$x_{n+1} = x_n - \frac{\mu}{2}\nabla\hat{\xi}(x_n) \tag{5}$$

4.3 Optimizing the step size

Choosing the correct step size for the LMS method is vital as seen from Figure 7. Still the basic Equations (2) and (5) commonly have arbitrarily chosen step sizes. Normalizing the step size μ, has been found to yield greater likelihood of convergence. Equation (6) shows the mathematical notation of the normalize step size

$$\mu(n) = \frac{\mu}{\|x(n)\|^2} \tag{6}$$

where μ is the original step size and $x(n)$ the microphone input buffer. [9] found that standard LMS converged the fastest out of three tested algorithms. [9]

5 Applied cases of ANC

Even though various applications for ANC have been proposed over the years, ANC headphones are by far the most successful application of ANC. The headphone applications are much simpler, since they require less microphones and loudspeakers and require less paths to be modeled. [2] [4]

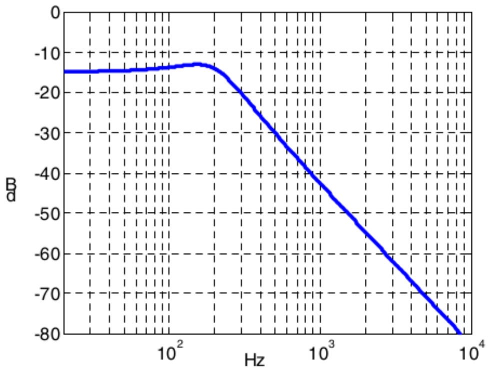

Figure 8: Simulated passive attenuation of an ANC headset. [3]

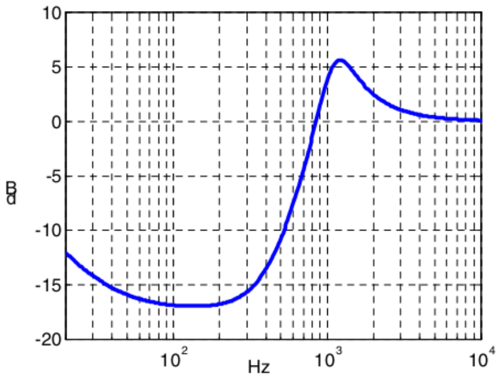

Figure 9: Simulated active attenuation of an ANC headset. [3]

5.1 ANC headphones

As discussed previously, ANC works best at low frequencies, while passive noise control works best at high frequencies. When these two are combined, they can compliment each other. [3] simulated the passive, Figure 8, and active noise control, Figure 9, of typical ANC headphones. When combined, these methods can achieve results neither could achieve by them selves. [3]

When the human ear is detecting frequencies, if some frequencies are louder than others, the louder ones can make quieter ones imperceivable. This is known as auditory masking. Auditory masking can be exploited in ANC headphones since they are commonly used for playing music. Most of the commercial ANC headphones the author has experienced, have been playing some masking sounds when their ANC mode was activated without music, as opposed to trying to achieve complete silence. [10]

6 Conclusion

This paper covered what are the physical phenomena behind active noise control (ANC), how ANC is exploiting these phenomena, what different types of ANC are there and how they are implemented.

Sound is a pressure signal propagating in a medium. ANC is a method of eliminating noise from desired sounds. This can be done with multiple different techniques. To separate noise from the desired sound, the desired sound has to be either known before hand or the noise has to be detected separately from it.

Feed forward and feedback methods were introduced. In the feed forward method there is a microphone picking up the outside noise source. This signal is then filtered to produce a cancellation signal which is played by a loudspeaker once the noise arrives to the silenced area. The quality of this ANC process is monitored by an error microphone inside the silenced area, which also effects the cancellation signal. In the feedback method there is no outside microphone, but instead the system produces a cancellation signal by comparing what the inside microphone is capturing and what it was expected to capture. All deviations from the desired signal are considered noise and are wanted to be canceled.

To process the cancellation signals, adaptive filters have to be used. These filters are manipulated based on the input and error signals. The optimization of these filters is commonly done with the least mean square (LMS) algorithm, which is an iterative optimizing algorithm. In the case of ANC, the algorithm is trying to minimize the error signal.

The most common applied case of ANC is in headphones. Other applications have been proposed but they haven't been as successful as in headphones. This is most likely due to headphones being simpler to design due to simpler block diagrams.

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