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A Comprehensive Overview of The Physics Behind Active Noise Cancellation (ANC) Systems & Potential to Maximise its Efficiency.

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Abstract: This paper entails details of the physics and technology used in ANC systems and why it has grown necessary in today's economies - specifically in the medical, automobile, aviation and consumer industries. By exploring the complex phenomena behind ANC systems, we decipher methods to improve its efficiency including advanced signal processing, DSP technology, and the use of AI. Wave interference, phase cancellation, signal processing and other key principles of wave phenomena are described and applied to realistic scenarios of ANC uses. With the help of graphing software each phenomena is depicted aided by suitable examples. We find that currently existing ANC systems make use of advanced filtering techniques like LMS, DSP, and FFT and proceed to identify many scenarios persisting limitations. Potential solutions - hybrid (feedforward and feedback) ANC systems and AI-assisted self learning systems - are identified to exhibit the future scope and real world applications of this technology.

INTRODUCTION

With the increase in urbanisation, transportation and construction, noise pollution has become a redundant part of modern life. Its sources range from industries to social gatherings, and due to its apparent normalcy, its side effects go unnoticed. Some side effects include increased stress levels, high blood pressure and a fall in productivity, all of which have diverse impacts on individuals as well as the economy.

One method to tackle this issue is the use of passive noise cancellation (PNC), which involves using headphones or hardware to physically filter out noise. Although functional, this method is not optimal for low frequencies and inapplicable in environments like vehicles, where engine noises are to be filtered out.

For this reason, years after the proposal of the idea of ANC in 1936, Lueg designed the first acoustic ANC using electronic microphones and loudspeakers and patented his creation.

The basic principle behind the notion is based on the destructive interference of two acoustic waves. A wave with the same amplitude in antiphase with the noise wave is produced in order to cancel out the noise. Although simple in theory, it is difficult to replicate this notion in practicality.

Sound is a longitudinal mechanical wave characterised by particle vibrations being parallel to the direction of propagation of the wave. Since it propagates by alternating compressions and rarefactions, sound - unlike EM waves - requires a medium for propagation, like air, water or metals. The speed of the wave is maximum in metals because the particles are dense and can transfer energy faster.

Sound waves are characterised by four key properties:

- **Frequency (□)**: The number of oscillations per second, measured in hertz (Hz). Frequency relates to the physical property of pitch; higher frequency sounds have higher pitches.
- **Amplitude (A)**: The maximum displacement from equilibrium which indicates how loud the sound is.
- **Phase (ϕ)**: The relative position of a wave at a given time, critical for interference.
- **Wavelength (λ)**: The distance between consecutive points of the same phase. In the case of sound waves, the distance between two consecutive compressions or rarefactions.

Relating these components of sound is the fundamental equation **$v=f\lambda$**. Where frequency is in Hz or per second, and wavelength is in metres, giving the speed of sound in metres per second.

2.1.1 FUNDAMENTALS OF WAVE INTERFERENCE

The principle of superposition states that when two waves overlap, the resulting wave is the sum of displacements of the incident waves at each point. Simply put, two waves that are coherent and in phase interfere constructively while those in antiphase interfere destructively. This phenomenon can be represented as such:

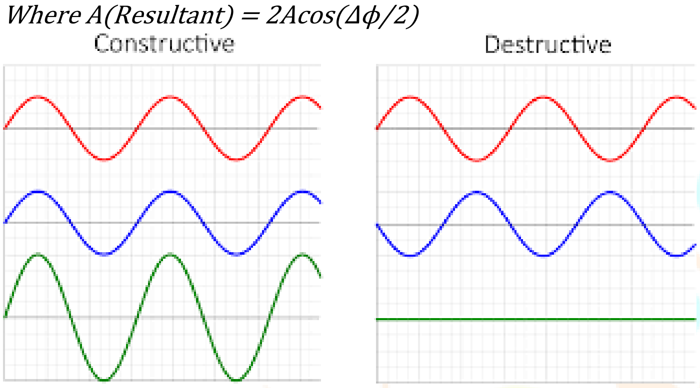
$$y1(x,t)=A\sin(kx-\omega t) \text{ and } y2(x,t)=A\sin(kx-\omega t+\Delta\phi)$$

$ytotal(x,t)= y1(x,t)+y2(x,t)$

Using [$\sin\alpha+\sin\beta=2\sin(2\alpha+\beta)\cos(2\alpha-\beta)$]

$ytotal(x,t) = 2A\cos(\Delta\phi / 2)\sin(kx-\omega t+(\Delta\phi / 2))$

Where $A(Resultant) = 2A\cos(\Delta\phi/2)$

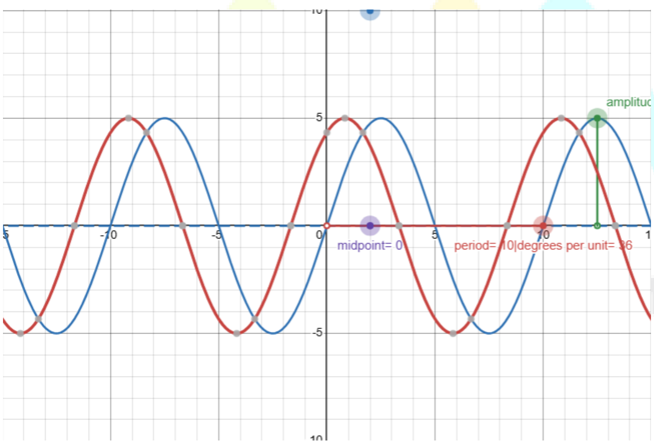


As represented above, constructively interfering waves , have a resulting amplitude A1 + A2, whereas destructively interfering waves have a resulting amplitude |A1-A2|. For complete destruction, A1 must equal A2.

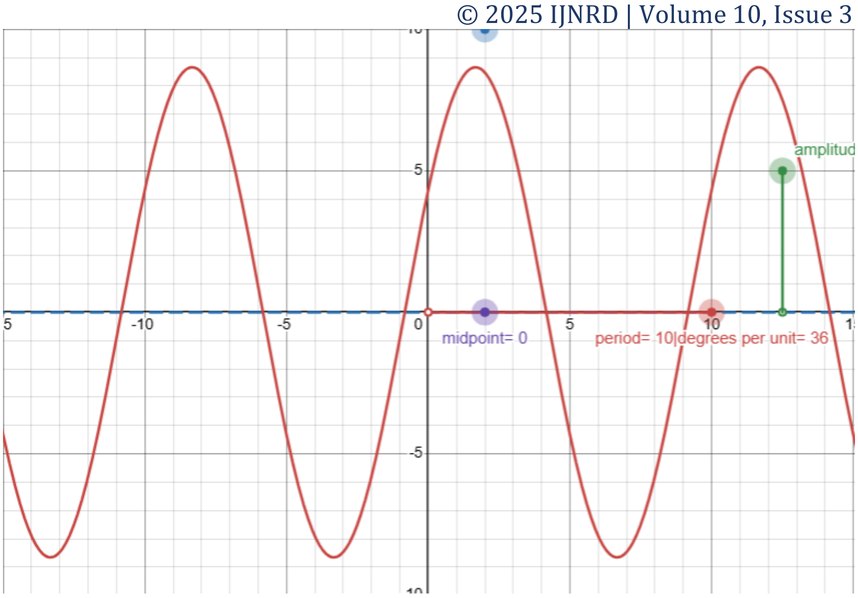
Example interference 1 : two waves with equal amplitudes and phase difference 60.

$Y1 = 5\sin(36x)$

$$Y2 = 5\sin(36x + 60)$$



$Y1 + Y2 = 2A\cos(\Delta\phi / 2)\sin(kx-\omega t+(\Delta\phi / 2))$
$= 5\sqrt{3} \sin (36x +30)$



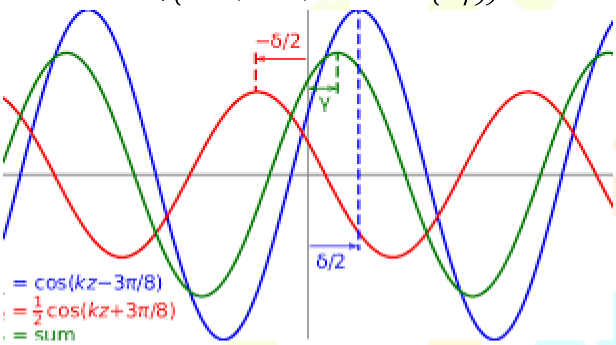
y = 5√3 sin (36x +30)
A(resultant) = 2Acos(60/2) = A√3 = 5√3 = 8.660254

However, in practicality waves are not always in phase or antiphase, they are likely to have a phase difference that isn't a multiple of pi and amplitudes that are dissimilar. In these cases, the phase difference and amplitude of the two waves is taken into account such that

y1(x,t)=A1sin(kx−ωt) and *y2(x,t)=A2sin(kx−ωt+Δϕ)*

ytotal(x,t) = A1sin(kx−ωt) + A2sin(kx−ωt+Δϕ)

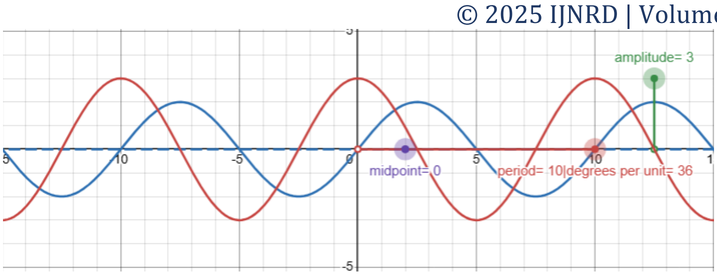
Aresultant = √(A12+A22+2A1A2cos(Δϕ))



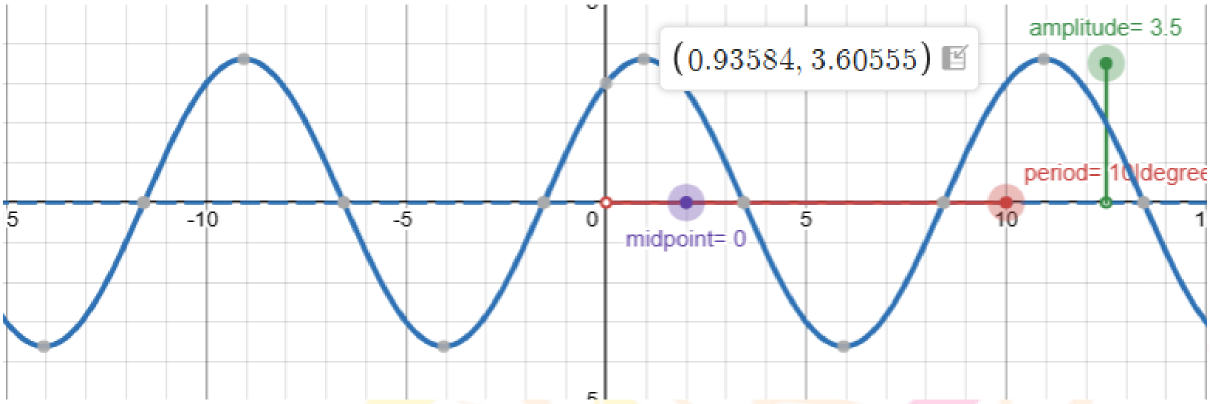
https://www.researchgate.net/figure/Superposition-of-two-waves-with-amplitudes-A-1-1-blue-and-A-2-05-red-and-a-phase_fig1_365372160

The question of where the energy goes during destructive interference, however, still stands. As the law of conservation of energy states, the energy is not destroyed. Rather it is distributed to areas of constructive interference. It is converted into potential energy in the case above, or if a standing wave forms, it results in alternating bands of increasing potential and kinetic energy. If complete destructive interference occurs, the energy is reflected back to the source therefore still firmly complying with the conservation of energy.

Example interference 2 : two waves of different amplitudes, and phase difference 90
Y1 = 2sin(36x)
Y2 = 3sin(36x + 90)



Y1 + Y2 = 2sin(36x) + 3sin(36x+ 90)
A(resultant) = √(A12+A22+2A1A2cos(Δϕ))
= √13 = 3.60555



2.1.2 THE MATHEMATICAL BASIS OF NOISE CANCELLATION

Using the principles of destructive interference, active noise cancellation can be implemented, by generating a secondary sound wave (anti-noise) with equal amplitude and opposite phase to the unwanted sound wave. Mathematically, this can be represented as such:
y(noise) = Asin(ωt)

The sound wave to be generated would be equal in amplitude but in anti-phase, implying a phase difference that is an odd multiple of pi.

y(anti-noise) = Asin(ωt+π)

This simplifies to result in perfect noise cancellation:

y(resultant) = Asin(ωt) + Asin(ωt+π)
Using sin(ωt+π)=−sin(ωt)
ytotal(t)=Asin(ωt)−Asin(ωt)=0

However, this only represents an ideal situation. In reality, sound is three-dimensional- approaching our ears from all directions - making it difficult to identify all waves and accordingly generate a single anti-noise wave, consequently leading to inefficiencies in noise-cancellation technology. Moreover, noise waves are random, not sinusoidal in nature; differing amplitudes and frequencies are difficult to generate, especially while prioritising avoiding latency issues. Additionally, limitations from the types of microphones available can result in inefficient detection, and production of anti-noise. To overcome these complications, ANC systems use adaptive algorithms.

2.2.1 ADAPTIVE ALGORITHMS

In 1807 French mathematician and physicist Jean-Baptiste Joseph Fourier exhibited his discovery that any wave can be represented as a sum of sine and cosine components. Essentially, it analyses non-periodic signals to break them down into their composition of sine and cosine waves. FFT or Fast Fourier Transform converts a noise signal from a time domain to a frequency domain, representing its component sine waves.

Ynoise(f)=∫−∞∞ynoise(t)e−j2πftdt.
This equation represents the shift from time to frequency domain. The simplified sine waves are then inferred, revealing the frequency and amplitude of each. This allows the system to generate targeted anti-noise waves rather than attempting broad spectrum

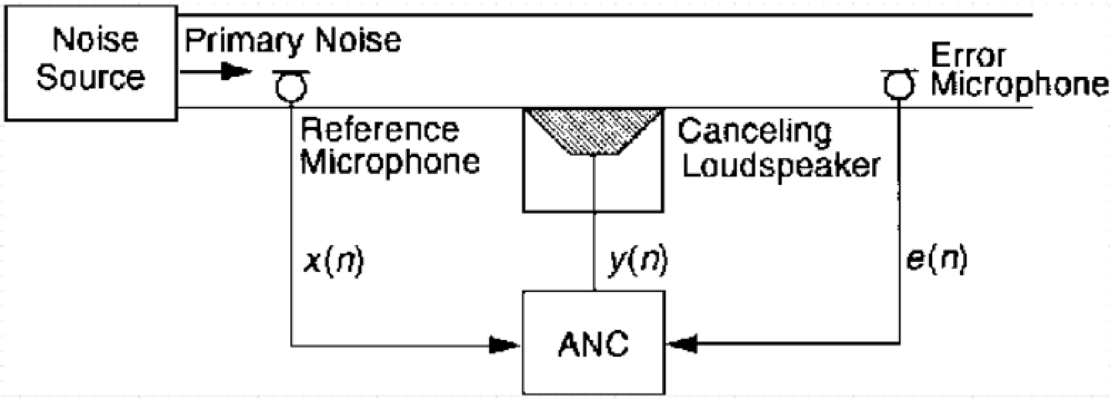
cancellation. The anti-noise wave for each sinusoidal component is generated with a phase opposite to the incident wave. This process makes use of DSP hardware. The waves are then summed back up and undergo inverse FFT to be represented in the time domain, using the inverse FFT function: *yanti-noise(t)=∫−∞∞Yanti-noise(f)ej2πftdf.*

In the system, FFT is merely the equation to generate an ideal anti-noise wave. However, in actuality, the system carries out this process with many other algorithms, and softwares. Firstly the DSP converts the received analogue signal into a digital one. Based on the expected required anti-noise wave calculated using FFT, the FIR shapes the anti-noise wave to the desired frequency. Following this, the LMS algorithm continuously monitors the desired anti-noise wave compared to the actual output. To minimise the error between the two the LMS algorithm is applied to calculate the optimal weights and accordingly adjust the filter coefficients in order to remove any residual noise or inefficiencies. This continuous monitoring and updation process allows the system to adapt to the fluctuating and inconsistent nature of noise to make efficient anti-noise.

The process is required to occur in a constrained amount of time - after the noise is detected, and before it reaches the output mic - indicating that latency is a likely complication. To overcome this, a time delay is added to the detected noise wave, therefore accounted for in the generated anti-noise wave. Latest technology nearly nullifies this complication with the use of high-speed DSPs, making the process unprecedentedly faster.

Another limitation is the nature of the soundwave itself. Repetitive low frequency noise like engine hum, is generally easier to replicate and eliminate as it does not require precise timing.

Physical constraints include the positioning of the mics and speakers in the system, based on its application environment. A simplified system is shown in fig, where the reference microphone detects the initial noise, followed by the ANC producing the predicted required antinoise at the position of the error microphone.



The question of where to place the two microphones and the speaker resulted in two broad methods of ANC: feedforward and feedback.
Feedforward noise cancellation is the optimal option when the options for microphone placement are limited, for example in small earphones. The microphone must be placed externally to get the most accurate detection of the incoming noise wave. However the detected wave is likely to be different from the sound-wave that is heard because of changes in the internal environment compared to the external one. The benefits of this placement include maximum noise sensitivity which is optimal for mid-frequency noise cancellation. Consequently it can be used to isolate specific sounds such as speech or traffic. Contrarily, the high sensitivity may instead get amplified if the predicted internal sound wave differs from what is actually heard, especially when it comes to high frequency noises.

Feedback noise cancellation involves placing the microphone on the interior of the headphone or earbud. Although picking a comfortable and efficient position for it is tough, this placement ensures that the noise captured accurately reflects what the user hears. Hence the efficiency of the system does not depend on how the user equips the device. It also makes the system efficient against low-frequency noises, but reduces that for high-frequency ones. Additionally, much like all feedback mechanisms, there is a possibility of the mic picking up its own antinoise signal and amplifying the unwanted noise instead of reducing it. ALthough the probability is low, this can occur in poorly engineered ANC devices.

2.2.2 FUTURE SCOPE

To reiterate, ANC systems today still are inefficient when it comes to high-frequency noise cancellation, real-time adaptation and energy consumption. This calls for better algorithms and scopes out the future for ANC technologies.
One method of doing so involves the use of AI and machine learning. Adaptive filtering using deep learning will enhance real-time, predictive noise cancellation. Additionally, self-learning ANC systems with machine learning techniques can be coded to improve over time, based on environmental noise patterns. To further enhance signal processing, improvements in FFT and adaptive filtering algorithms like LMS and RLS as well as advancements in DSP hardware, cumulatively will aid in speeding up the processing of signals. Lastly, hybrid noise cancellation methods including a well-engineered combination of both active and passive noise cancellation can maximise the effectiveness of the system.

Enhancing this imperative technology will improve many other fields in the future. Medical assistance to enhance hearing using ANC and the reduction of noise in surgical rooms include a few of the potential improvements in the medical field. In aviation and auto-motive industries, in-cabin noise cancellation will be more efficient with algorithms that quickly adapt to changing frequencies. In consumer electronics, ANC can take noise cancellation to another level with open-eared headphones, which is a growing market. Lastly, with research and application of ANC in large industrial spaces, it can reduce worker fatigue, improve safety and increase productivity.

3.1 CONCLUSION

Concerns about the effects of noise in our environments increase by the minute despite the current technology that has been controlling noise levels. With the simple yet multiplex existing noise cancellation technology, containing noise has been possible and effective, yet there is scope and need for development. With increasing traffic, industrial noise and the growth of the automotive sector, the demand and necessity of noise cancellation has increased exponentially, calling for improvements in the near and far future.
As research progresses, ANC is inevitably going to become an essential tool in reducing unwanted noise across multiple sectors, enhancing safety, comfort, and overall quality of life. Continued innovation in algorithms and hardware will ensure that ANC technology remains at the cutting edge of modern acoustic engineering, pushing the boundaries of noise control solutions.

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