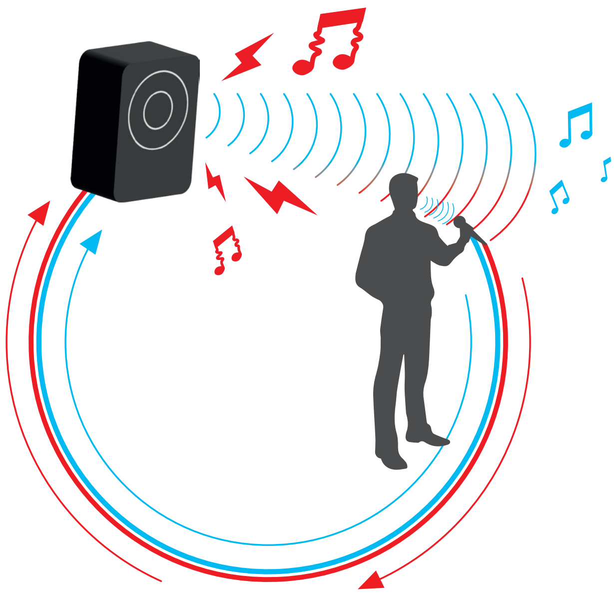
ANTIDOTE

Revolutionizing Audio with the Ultimate Solution to the Larsen Effect

In the audio world, Acoustic Feedback, commonly known as the Larsen effect, has been a challenge for over decades. Traditionally, sound experts have employed various techniques to combat acoustic feedback. However, these methods often come with significant limitations, failing to fully address the complexity of the issue. With the advent of digital signal processing, new methods have emerged, yet they too fall short in providing a comprehensive solution.
This is where Antidote® steps in. Unlike conventional approaches, Antidote® harnesses the latest advancements in digital signal processing to offer an unprecedented solution to the Larsen effect. Our product doesn't just manage acoustic feedback; it effectively neutralizes it, marking a paradigm shift in audio technology. The audio world has been waiting for a true solution to this persistent issue. Antidote® is that solution!
In this White Paper, we not only review the limitations of existing methods but also confidently present Antidote as the ultimate answer to the Larsen effect.

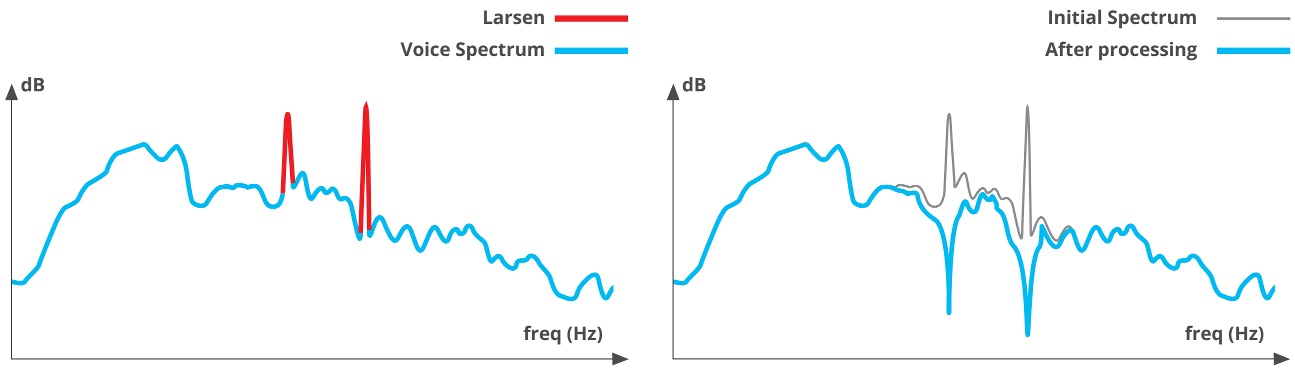
1. The Feedback issue

Acoustic feedback is a common phenomenon in sound reinforcement systems. It occurs when the sound emitted by a speaker is picked up again by a microphone and re-amplified again and again. This re-amplification creates a feedback loop that deteriorate the frequency response and eventually produces a high-volume undesirable sound. Feedback is heavily influenced by the proximity of the microphone to the speaker, the characteristics of the microphone, the room response and the settings of the audio equipment used.
The stability of a sound reinforcement system relies on mastering its «loop gain» (electroacoustic transfer function of the system) which results from the combination of the «Loudspeaker Enclosure Microphone» (LEM) set and the room acoustic behaviour. When, at a given frequency, this open-loop gain exceeds unity while being in-phase, the system becomes unstable (Nyquist criterion).



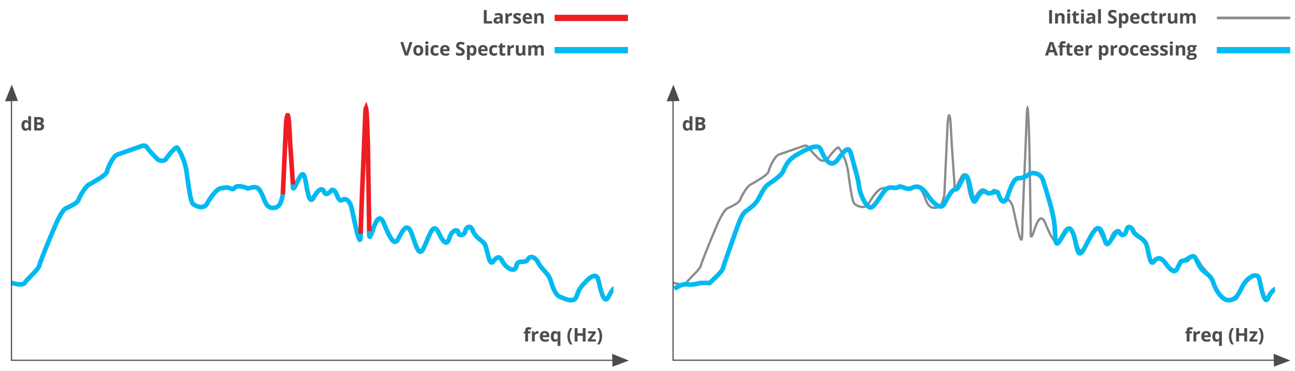
The common method employed by sound engineers to limit Larsen effect is to manually equalize the frequency response of the sound reinforcement system, and to limit the gain of the microphone. But there are also systems that automatically limit the feedback effect.

2. Existing processing algorithms



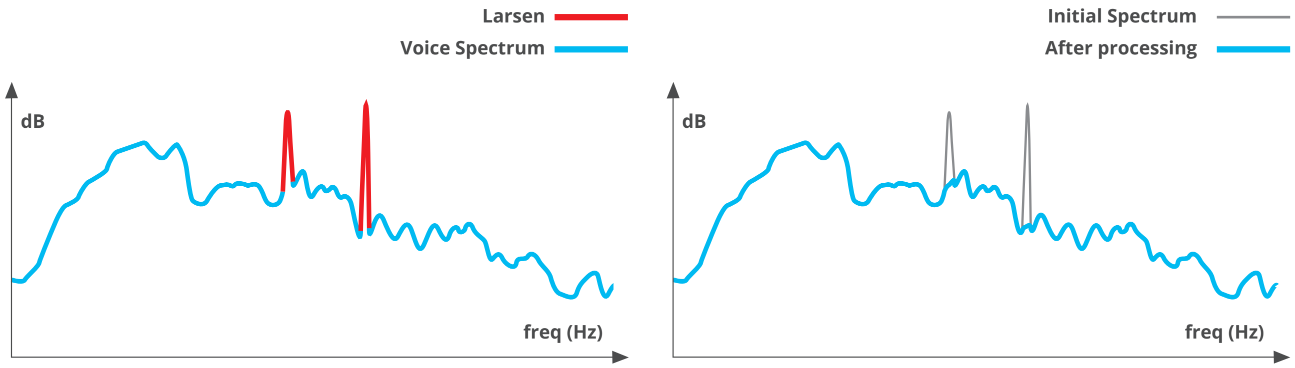
2.1. NOTCH FILTERING

Notch Filtering selectively attenuates specific frequencies when feedback is occurring, creating a ‘notch’ in the frequency response and effectively disrupting the feedback loop at distinct problematic frequencies without altering the entire signal. This technology in only curative and not preventive. This selectivity makes it highly effective in stable settings where feedback frequencies can be accurately identified and suppressed, thus preserving the overall sound quality. However, the technique can impact tonal balance, especially if multiple notches are applied or if the attenuated frequencies are fundamental to the audio content, reducing the audio quality.
Additionally, in dynamic environments where feedback frequencies change rapidly, static Notch Filtering may not be as effective, as it relies on the accurate and consistent identification of feedback points, which can lead to new feedback issues if not managed properly. The typical Added Sable Gain (ASG) of this technology is 3-5 dB. Today, notch filtering is regarded as a «go-to» solution, commonly incorporated in most audio matrices due to its robustness and cost-effective implementation.



2.2. FREQUENCY SHIFTING

This technique applies a constant frequency offset to the audio signal, consistently shifting its frequency and altering the phase relationships within the feedback path. This continuous shift disrupts the feedback loop, preventing it from sustaining at specific frequencies and effectively 'moving' the feedback point before it becomes problematic. Frequency shifting comes with limitations; the constant shift can cause perceptual changes, especially affecting pitch and harmonic relationships. The ASG performance are rather low. **Typically, for speech, a balance between quality and ASG is struck at a frequency shift that provides a 4-6 dB increase in ASG.**



2.3. ROOM MODELING

Room Modeling, also known as Acoustic Feedback Cancelling (AFC), mitigates feedback by creating and continually updating a model of the feedback path based on incoming audio. This model generates an inverse signal to cancel out the feedback, based on an on-line identification technique which ensures that it keeps adapting to changes like microphone movements or environmental shifts. The dynamic nature of Room Modeling allows it to predict and neutralize potential feedback in real-time, providing comprehensive coverage across a wide frequency range and effectively improving audio quality by preserving the integrity of the original signal. The main advantages of this method are its dynamic adaptation, making it effective in unpredictable environments, and its minimal impact on sound quality when properly tuned. However, Room Modeling systems are typically complex and costly due to their need for sophisticated, real-time processing algorithms and can introduce latency, potentially problematic in live scenarios. Limitations such as latency, computational cost, and complexity of implementation have rendered it impossible to achieve a viable solution... until ANTIDOTE®, by ARTEAC-LAB. Theoretically, Room modeling can offer infinite ASG performance by completely removing feedback if it is perfectly modeled. **In practice, an ASG improvement of 10-15 dB can be achieved, which, in most cases, effectively renders feedback virtually non-existent.**

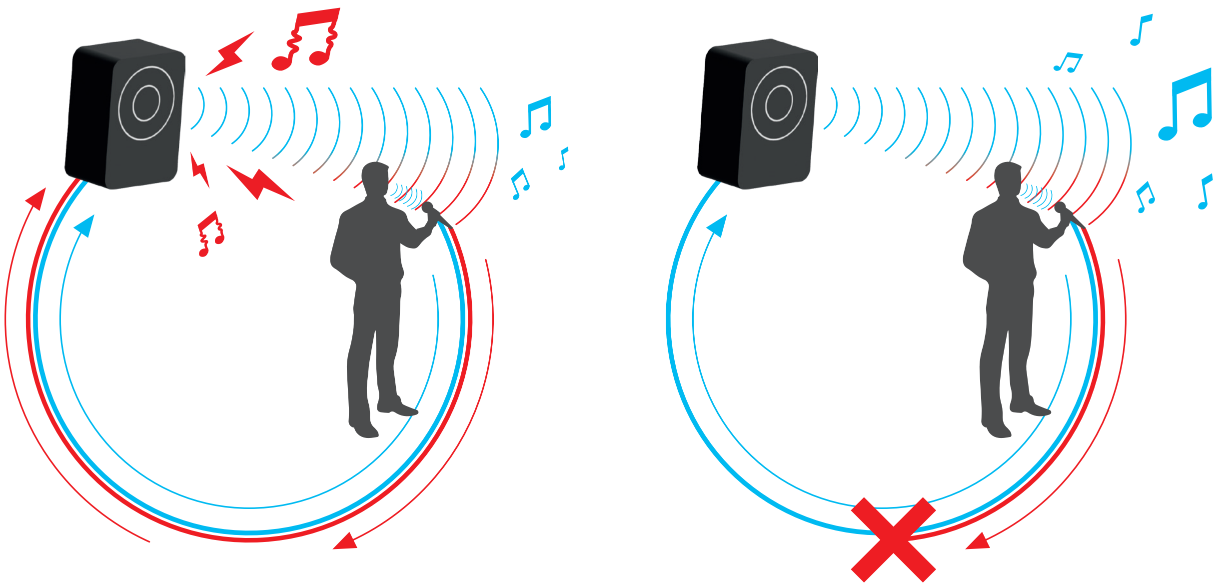
3. Antidote

The «Acoustic Feedback Cancelling» (AFC) method proposed by ARTEAC-LAB, named ANTIDOTE®, is based on the room modeling approach, aiming at the compensation of acoustic transfer function of the acoustic feedback in the amplification loop. It pushes the boundaries of this approach by implementing a set of specific features, reaching unprecedented levels of feedback control. There are virtually no commercial devices that incorporate room modeling due to latency, computational cost and implementation complexity, but this has finally been made possible by all the innovations brought by ANTIDOTE® algorithm.

As already mentioned, the acoustic transfer function varies over time, particularly when the position of the microphone or the configuration of the listening room is changed. Therefore, the Antidode algorithm is continuously adapting a high resolution internal model. This adaptation is carried out using a stochastic gradient algorithm of the NLMS type, implemented in the time-frequency domain. This allows to optimize the computational load while dealing with very long impulse responses (reverberation times exceeding several seconds).

Despite being somewhat complex, the ANTIDOTE ® algorithm is significantly more efficient than those currently available in current commercial devices and based on Notch Filtering and Frequency Shifting. It thus combines performant feedback cancellation without degradation of audio quality, while being resource-efficient enough to be integrated into standard electronic platforms common in audio systems. This crucial aspect allows this patented algorithm to be implemented in systems using low-cost processors, with limited memory, and low power consumption.

Furthermore, the performances increase compared to existing systems enables the use of ANTIDOTE ® in sound reinforcement configurations previously considered unfeasible due to acoustic feedback (microphones very close to speakers, very high amplification, etc.). Moreover, ANTIDOTE ® remains effective regardless of the type of room, including in large reverberant spaces (churches, theatres, stadium...) which are especially challenging in terms of acoustic feedback



WITHOUT ANTIDOTE

WITH ANTIDOTE

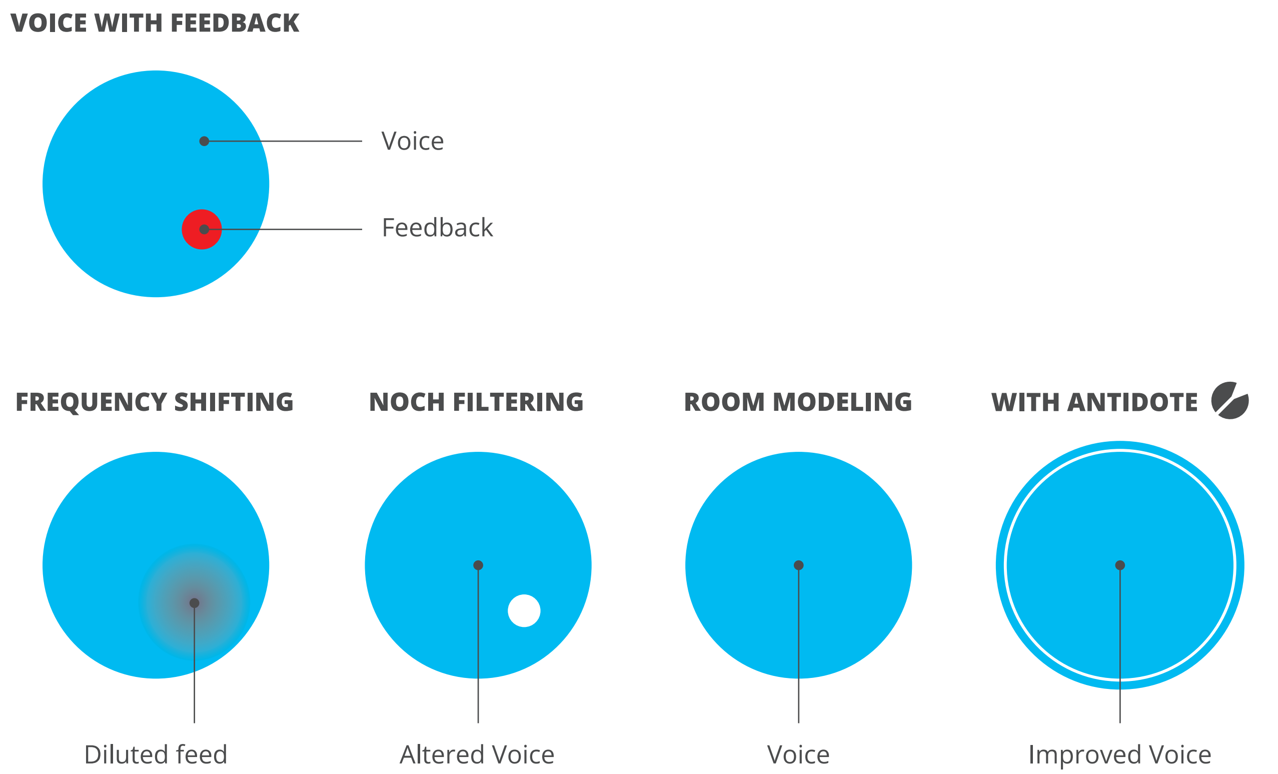
TECHNICAL INNOVATIONS BROUGHT BY THE ANTIDOTE ALGORITHM

1. Variable Block Size Processing Method: Targeting an efficient processing of long impulse responses, variable block size adaptation is an extension of fast, low-latency convolution algorithms. It allows for an increase of more than tenfold the order of adaptive filtering compared to real-time implementations described in the literature.

2. Real-Time Implementation Compatibility: Implementing an algorithm with variable block sizes leads to some complexity which is likely one of the reasons why this approach has been overlooked until now. ANTIDOTE ® patented real-time implementation has been validated over several platforms and a small-scale demonstrator is available from ARTEAC-LAB.

3. Limiting Audio Quality Degradation Due to Block Size Heterogeneity: The use of variable block sizes has so far been used only for fast convolution with stationary impulse responses. This is not the case with adaptive algorithms, where blocks of various sizes are updated at different time intervals, potentially leading to audible discontinuities. The patented algorithm overcomes these limitations by a synchronous scheme implemented in a real-time kernel.

4. Integration of Advanced Pre and Post-Processing Techniques from Literature: These treatments include pre-equalizing input signals before filter adaptation, managing transients (onset), a priori modeling of the room acoustic response, various regularization techniques and automatic parameterization of the adaptation's parameters.



4. Benchmark

Metrics are essential in evaluating AH systems as they provide objective, quantifiable measures for comparison and assessment. The chosen criteria were selected from both existing literature and common audio practices, - Added Stable Gain (ASG), Perceptual Evaluation of Speech Quality (PESQ), and Reactivity - as they directly address the most crucial aspects of performance:

1. Added Stable Gain (ASG): ASG measures the system's stability and operational capacity by determining the maximum gain added before feedback occurs again with the anti-larsen system. In a benchmark, ASG allows for a comparative analysis of different systems' thresholds for stability, offering insights into their effectiveness in various acoustic environments.

2. Perceptual Evaluation of Speech Quality (PESQ): PESQ assesses the audio output's perceived clarity from a listener's perspective, ensuring the system meets end-user expectations for speech quality. It bridges the gap between technical performance and human perception and allows for an objective comparison of how different AH systems maintain speech clarity and quality, ensuring that they meet the necessary standards for human perception and communication.

3. Reactivity: This measures the system's ability to adapt quickly to changes in the electroacoustic environment. High reactivity is crucial for maintaining stability and quality in dynamic settings. By measuring reactivity, you can assess if albeit being very effective from an ASG point of view, the AH system is not too slow. Indeed, who would accept two seconds of Larsen during a live before being removed completely ?

	Nothing	Notch filtering	Freq. shifting	ANTIDOTE
Quality @ -10 dB	★★★★★	★★★★★	★★★★★	★★★★★
Quality @ -3 dB	★★★☆☆	★★★☆☆	★★★☆☆	★★★★★
Quality @ +5 dB	☆☆☆☆☆	★☆☆☆☆	★☆☆☆☆	★★★★★
Reactivity @ +5 dB	☆☆☆☆☆	★★☆☆☆	★★★★☆	★★★★★
Max Gain	0 dB	3-5 dB	4-6 dB	15-18 dB

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In conclusion, this article presents the principles of various methods aimed at reducing audio feedback, along with their advantages and disadvantages. It presents also the performance they achieve in terms of gain increase, sound quality, and responsiveness. The far most effective processing technique, Acoustic Feedback Cancelling, was not available in commercial products because of an excessive computational load. Thanks to ANTIDOTE, we are pushing the boundaries of anti-feedback systems, making available a level of performance that has been unmatched until now.

Feel free to ask us for a demonstration, and we would be delighted to let you listen !

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