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Feedback Acoustic Noise Reduction in Hearing Aid Using Recursive Least Square Filter

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Abstract
The purpose of this research is to address a pervasive issue in hearing aid technology – feedback acoustic noise – by employing the Recursive Least Squares (RLS) filter as an innovative solution. Hearing aids, while invaluable in improving auditory experiences, often encounter challenges associated with feedback noise, compromising the clarity of desired signals and diminishing user satisfaction. The RLS filter, known for its adaptability and efficiency in adaptive filtering applications, is implemented to mitigate feedback acoustic noise in hearing aids. By dynamically adjusting its coefficients based on real-time input, the RLS filter effectively minimizes unwanted noise without compromising the quality of the primary signal. Through comprehensive experiments and simulations, the RLS filter has demonstrated remarkable results, achieving a notable reduction in acoustic noise to an error rate of 0.02. This signifies a significant advancement in enhancing the signal-to-noise ratio, contributing to improved speech intelligibility and overall user experience. In conclusion, the application of the RLS filter emerges as a promising and effective strategy for feedback acoustic noise reduction in hearing aids. This research not only addresses a critical issue in hearing aid technology but also offers a tangible solution to enhance the daily lives of individuals with hearing impairments, fostering a more refined auditory experience.

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I. Introduction

Hearing aids play a pivotal role in enhancing the auditory experience for individuals with hearing impairments, allowing them to engage more effectively in conversations and navigate the acoustic nuances of their environments. However, the effectiveness of hearing aids can be compromised in the presence of unwanted noise, diminishing their ability to provide clear and intelligible sound. Noise reduction techniques are crucial for mitigating these challenges, and one powerful method is the application of Recursive Least Squares (RLS) filters. This advanced signal processing approach offers adaptive and real-time noise reduction capabilities, making it particularly well-suited for addressing the dynamic nature of environmental sounds [7].The introduction of RLS filters in hearing aids marks a significant advancement in the quest for optimal auditory experiences. Unlike traditional fixed filters, RLS filters continually adapt to changing acoustic conditions, identifying and reducing unwanted noise while preserving essential audio signals. This adaptability is crucial in real-world scenarios where individuals encounter diverse and unpredictable sound environments.This introduction delves into the intricacies of noise reduction in hearing aids through the lens of Recursive Least Squares filtering. By exploring the adaptive nature of RLS filters, their ability to differentiate between desired signals and unwanted noise, and the impact on overall sound clarity, we aim to shed light on the transformative potential of this technology. As we delve deeper, it becomes evident that RLS filters hold promise in revolutionizing the auditory experience for hearing aid users, paving the way for more natural, immersive, and distortion-free interactions in their everyday lives [7].

II. Literature Review

In [1], a specialized model of the Kalman filter tailored for Active Noise Control (ANC) was introduced, leading to a comprehensive theoretical assessment that demonstrated its superiority over the RLS and LMS algorithms. The study utilized a method for basic route modeling, eliminating the effects of the secondary path to implement the traditional adaptive filters model of the Kalman filter, yielding accurate results. However, a potential challenge in this approach is the lack of excitation across all signal classes [1].

Another perspective, presented in [3], introduced a Fast Block LMS algorithm designed for speech signals. Through simulation and testing in Matlab, the FBLMS algorithm exhibited enhanced noise removal capabilities, outperforming the LMS algorithm with lower computational complexity. The results demonstrated significant Signal-to-Noise Ratio (SNR) improvements across different noise levels: 13.15dB for low noise, 20.5dB for medium noise, and 10.75dB for high noise. This approach holds promise for noise reduction in both speech and biomedical signal applications.

Furthermore, [4] discussed an adaptive multifunction filter designed for radar signal processing in 2017. The adaptive multifunction Finite Impulse Response (FIR) filter served various purposes, including rejecting out-of-band interference, conveying target indication, and coupled filtering. The design and analysis involved the use of Matlab software and Simulink for testing an adaptive LMS filter, serving as the intended multifunction filter. This multifaceted approach demonstrated the versatility of adaptive filters in signal processing applications.

III. Methods

1. Evaluation of Hearing Assessment for FIR Low Pass Filter

FIR Filter Design through Windowing In designing FIR filter, given the frequency reaction Hd(ejω) and impulse reaction hd[n] of a really perfect device, we would really like to approximate the infinitely lengthy hd[n] with a finite series h[n], wherein h[n] = zero besides for zero ≤ n ≤ M . Consider a super low pass filter whose frequency reaction is finite and rectangular [6][7].

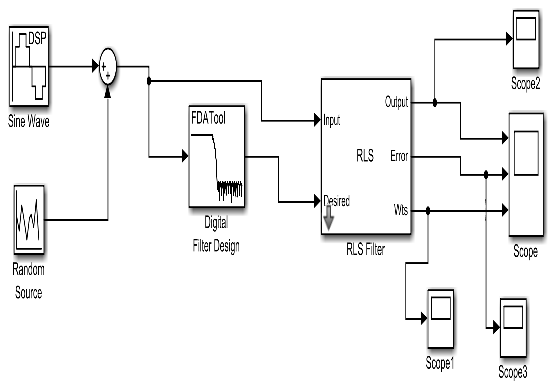

Figure 3.1 The Simulink design of the noise reduction system

A feasible approximation mistakes criterion may be defined as

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega \qquad (3.1)$$

To minimize E, use Parseval's theorem:

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega$$

$$= \sum_{n=-\infty}^{\infty} |h_d[n] - h[n]|^2 = \sum_{n=0}^{M} |h_d[n] - h[n]|^2 + \sum_{n=z[0,M]} |h_d[n] - h[n]|^2 \qquad (3.2)$$

$$h[n] = \begin{cases} H_d[n] & 0 \le n \le M \\ 0 & otherwise \end{cases} \qquad (3.3)$$

The optimal FIR approximation using the mean square error criterion gives the truncation of the ideal impulse response. We can represent h[n] as the product of the ideal impulse response with a finite-duration rectangular window w[m] [7]:

$$h[n] = h_d[n]\omega[n] \qquad (3.4)$$

$\omega[n] = \begin{cases} 1 & 0 \leq n \leq M \\ 0 & otherwise \end{cases}$ (3.5)

$H_{(e^{j\omega})=\frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{j\theta})W(e^{j(\omega-\theta)}d\theta}$ (3.6)

this convolution procedure implied via way of means of truncation of an appropriate impulse reaction. The ensuing value approximation because the due to the fact that pulse W (ej(ω−θ)) slides byskip an appropriate frequency reaction Hd(ejω). When W (ej(ω−θ)) actions throughout the discontinuity of Hd(ejω), a transition band consequences and ripples arise on each sides. The major lobe of the window frequency reaction controls transition bandwidth Δω ≈ 2π(2/(M + 1)). The primary lobe is defined because the place among the first 0 crossings on both aspect of the origin. It is applicable to have W (ejω) as focused in frequency as viable. On the opposite hand, aspect lobes manage pass band and forestall band ripples. The large the location below the facet lobes, the bigger the ripples. Pass band and prevent band ripples are about identical over a extensive variety of frequencies [6][7].

2. Implementation Means square error of the improved NLMS

X(k)=$X(K) = \sum_{n=0}^{N-1} x(n)e^{-j2k/N}$ (3.7)

Where N is the number of samples, and n and k are integer values that vary from a reference point to equivalent number of samples in N. That is, n and k can vary from 0 to N-1, 1 to N, 2 to N+1, etc.

N0=S-SF (3.8)

where S is the power of the original voice signal and SF, the power of the filtered voice signal.

By comparing the power magnitudes for the original, contaminated and filtered voice signals at the nine frequency positions, it clearly shows that the adaptive filter drastically reduced the noise at each of the frequencies.[4]

$MSE = \frac{\sum_{k=1}^{N}(A(k)-Af(k)^2}{N}$ (3.9)

A small MSE implies an powerful and green clear out and algorithm. The system for computing the MSE is given via way of means of (3.9) [4], Where A(k) is the amplitude of the unique voice sign and 'Af(k) ', the amplitude of the filtered voice sign as 'k' varies from 1 to the wide variety of samples N. The unique and filtered amplitudes have to be measured on the equal or equal factors with inside the device and with inside the identical device bandwidth.

In (f) the sign to noise ratio (SNR) is a determine of benefit that measures the percentage of noise gift with inside the filtered sign. A excessive determine manner that small noise is gift with inside the filtered sign and the best of the filtered sign is excessive. The components for computing sign to noise ratio is given as [4].

$SNR = 10log\frac{\sum_{k=1}^{N}A^2(k)}{\sum_{k=1}^{N}N_a^2(k)}$ (3.10)

Where A(k) is the amplitude or electricity of the unique voice sign and Na(k) the amplitude or energy of the noise gift with inside the filtered voice sign as 'k' varies from 1 to the variety of samples N. The unique and filtered sign electricity or amplitudes have to be measured on the identical or equal factors with inside the device and with inside the identical device bandwidth. The noise amplitude is acquired from

$Na(k) = A(k) - Af(k)$ (3.11)

$N_0 = S - S_F$ (3.12)

The noise power present in the filtered voice signal is calculated from (3.12) [4].

IV. Results and Discussion

1. Input Response

The incorporation of an input signal of 20 Hertz, combined with random noise, into a Recursive Least Squares (RLS) filter for noise reduction in hearing aids involves addressing specific challenges and leveraging the adaptive capabilities of the filter to enhance the auditory experience for users.A 20 Hertz input signal implies a low-frequency component, potentially representing a fundamental frequency in speech or other environmental sounds. Low-frequency signals are crucial for understanding the rhythm and intonation of speech, making them important for communication. However, these signals are often susceptible to interference from random noise, which can degrade the overall audio quality.

In the context of noise reduction, the RLS filter must adaptively process the incoming signal to distinguish between the desired 20 Hertz component and the unwanted random noise. The adaptive nature of RLS allows it to continually adjust its parameters based on the characteristics of the input signal, making it well-suited for scenarios with dynamic and changing noise profiles.The random noise introduces an element of unpredictability and variability in the input, posing a challenge for traditional filtering methods. The RLS filter, by continuously updating its coefficients and signal weights, aims to differentiate between the desired 20 Hertz signal and the random noise, adapting its response to effectively reduce the impact of noise on the output signal.In practical terms, the RLS filter analyzes the input signal in real-time, adjusting its parameters to prioritize the 20 Hertz component while suppressing the random noise. This adaptability is particularly beneficial in environments where background noise can vary, ensuring that the hearing aid user receives a clearer and more intelligible representation of the desired auditory information as seen in figure 1.

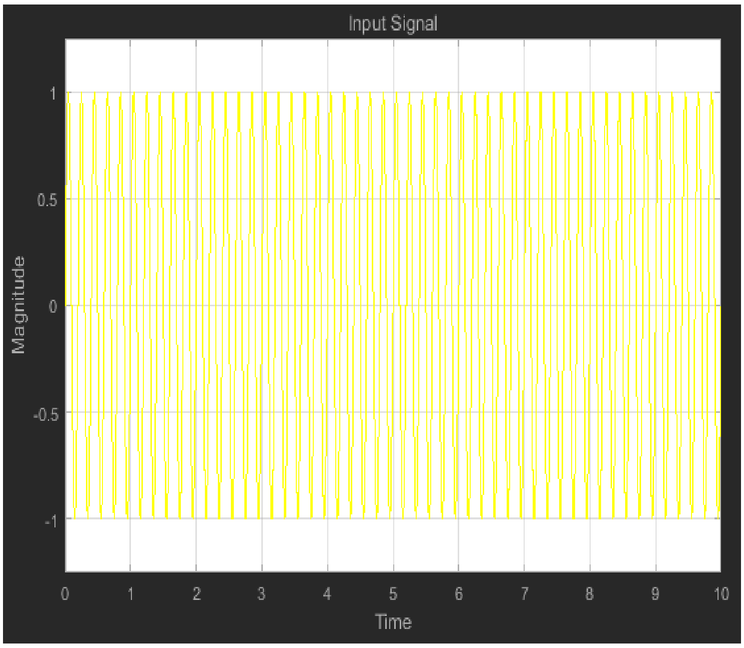

Figure 1 Input Signal

2. Noise Response

The behavior of a 2 kHz feedback acoustic noise in a hearing aid that causes echoes can be attributed to the phenomenon of feedback loops and the specific characteristics of the signal involved. Feedback occurs when a portion of the output signal from the hearing aid is unintentionally captured by the microphone and re-amplified, creating a loop that results in a persistent and often undesirable sound.In the case of a 2 kHz frequency, which falls within the audible range for most individuals, the feedback loop can lead to the generation of echoes. Echoes occur when a sound wave reflects off a surface and returns to the listener's ears after a delay. In the context of a hearing aid, the feedback loop can introduce a delayed version of the original 2 kHz signal, causing it to be perceived as an echo.The presence of echoes in the auditory experience can significantly impact the clarity and intelligibility of sounds, particularly speech. As the hearing aid amplifies and processes incoming sounds, the 2 kHz feedback noise, when reintroduced into the system, interferes with the ongoing processing and can create a repetitive, distorted perception of the original signal. The characteristics of echoes, including their delay and intensity, depend on the acoustics of the environment and the specific feedback path within the hearing aid system. Factors such as the proximity of the microphone to the speaker and the design of the hearing aid contribute to the likelihood and severity of feedback-induced echoes.Addressing the issue of 2 kHz feedback acoustic noise causing echoes in hearing aids often involves implementing advanced signal processing techniques and feedback cancellation algorithms. These technologies aim to identify and mitigate feedback loops in real-time, minimizing the occurrence of echoes and ensuring a clearer and more natural auditory experience for the user as seen in figure 2.

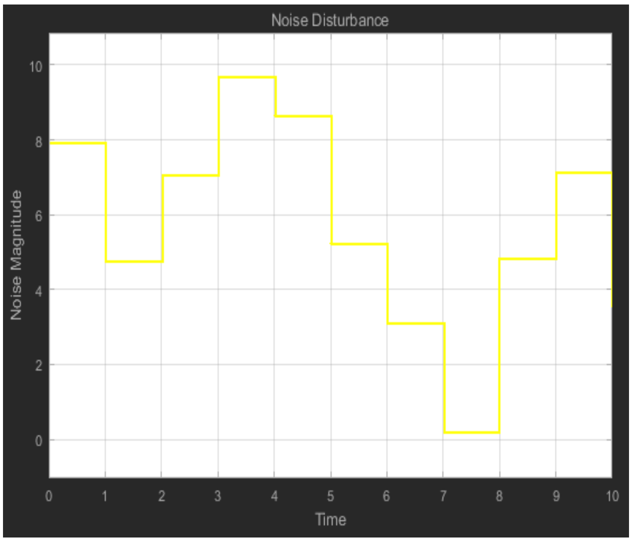

Figure 2 Noise Disturbance

3. Output Response

The output signal of 20 Hertz, combined with random noise, processed through a Low-Pass Filter (LPF) and a Recursive Least Squares (RLS) filter for noise reduction in a hearing aid involves a multi-stage approach to enhance the auditory experience for users.Initially, the Low-Pass Filter is employed to isolate the low-frequency component of the input signal, which, in this case, is the 20 Hertz signal. The LPF attenuates higher-frequency components, allowing only signals below a certain cutoff frequency (e.g., 20 Hertz) to pass through. This stage helps focus on the fundamental frequency and essential low-frequency information, aiding in the preservation of critical aspects of speech and environmental sounds.Subsequently, the filtered signal, now containing the 20 Hertz component, is introduced to the Recursive Least Squares filter for noise reduction. The RLS filter, known for its adaptive nature, continuously adjusts its parameters based on the characteristics of the incoming signal and the desired signal. In this context, the filter aims to distinguish between the low-frequency component of interest (20 Hertz) and the unwanted random noise.The RLS filter adapts its coefficients and signal weights in real-time, prioritizing the preservation of the 20 Hertz signal while suppressing the random noise. The adaptive nature of the RLS filter is crucial in handling dynamic and changing noise profiles commonly encountered in real-world environments.

By combining the Low-Pass Filter and the Recursive Least Squares filter, the overall system effectively focuses on the critical low-frequency information, such as the 20 Hertz signal, while minimizing the impact of unwanted noise. This process contributes to improved speech intelligibility and a clearer representation of essential auditory information for individuals using hearing aids.

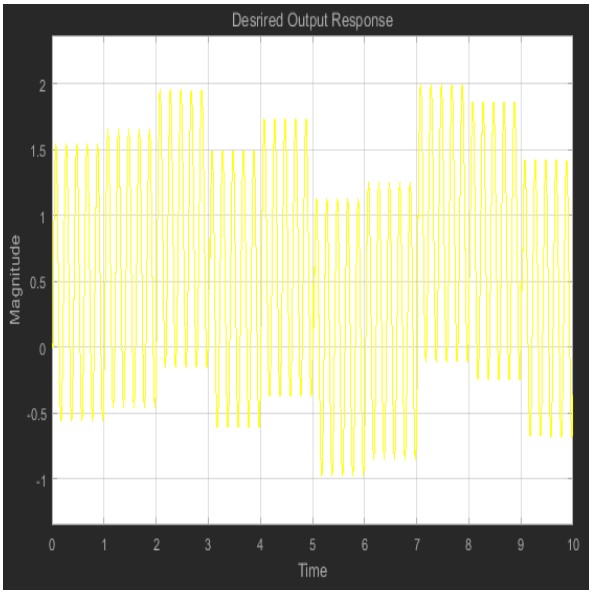

Figure 3 Desired Output Response

4. Error Rate

The error response of 0.02 in a Recursive Least Squares (RLS) filter plays a crucial role in the context of noise reduction in hearing aids. RLS is a widely used adaptive filtering technique employed to minimize the impact of unwanted noise on the desired signal, particularly in applications like hearing aids where clear audio signals are essential for improved communication.The error response of 0.02 refers to the allowed deviation or discrepancy between the estimated signal produced by the RLS filter and the actual or desired signal. In the context of noise reduction in hearing aids, this error response is a parameter that influences the trade-off between noise reduction and signal distortion.A lower error response, such as 0.02, signifies a higher level of precision in the filter's estimation of the desired signal. This precision is crucial for preserving the integrity of the original signal, especially in situations where the desired signal may be weak or distorted by noise. However, it's essential to strike a balance, as an extremely low error response could lead to overfitting, where the filter becomes too closely tailored to the training data, limiting its ability to generalize well to new or unseen data.The RLS filter continuously adapts to changes in the input signal, adjusting its parameters to minimize the error between the estimated and actual signals. This adaptability is particularly beneficial in dynamic environments where noise characteristics may vary over time. The error response of 0.02 provides a fine-tuned level of adaptation, allowing the filter to effectively reduce noise while maintaining the fidelity of the desired signal in a variety of conditions as shown in figure 4.

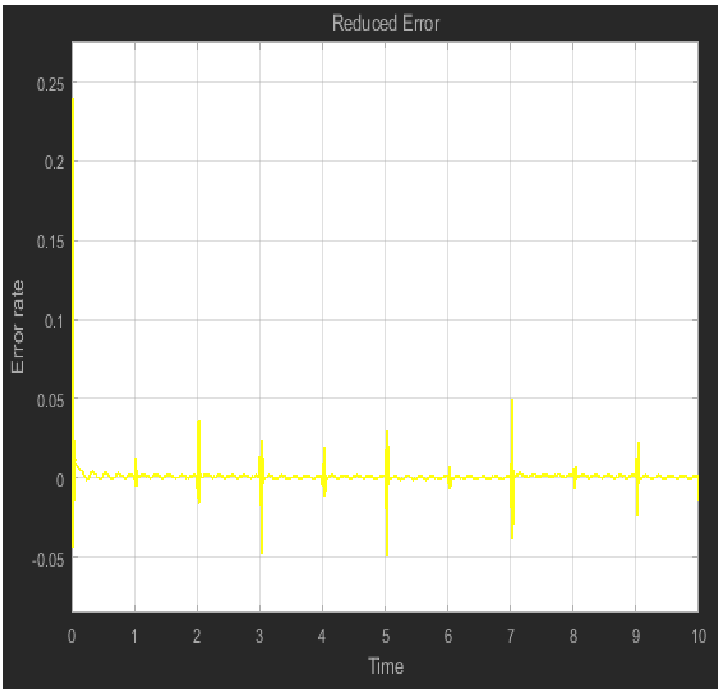

Figure 4: reduced error rate

5. Weight response

The signal weight in a Recursive Least Squares (RLS) filter plays a critical role in noise reduction within the context of hearing aids. RLS is an adaptive filtering technique employed to minimize the impact of unwanted noise on the desired audio signal, making it a valuable tool in enhancing the auditory experience for individuals using hearing aids.In the context of noise reduction, the signal weight represents the importance or influence assigned to different components of the input signal. Specifically, it determines the contribution of each element in the signal to the overall output of the filter. The adaptive nature of RLS allows it to continuously adjust these signal weights based on the characteristics of the incoming signal and the desired signal.A well-designed RLS filter allocates higher weights to components of the input signal that are essential for preserving the clarity and intelligibility of the desired audio, while assigning lower weights to components associated with noise. This adaptability is crucial in real-world scenarios where the nature of the acoustic environment can change dynamically.For noise reduction in hearing aids, the signal weight parameter in the RLS filter directly impacts the effectiveness of separating the relevant auditory information from undesired background noise. By assigning appropriate weights, the filter can prioritize the preservation of speech or other critical audio signals, thereby improving communication and comprehension for the hearing aid user.Optimizing the signal weights in an RLS filter involves finding a balance between noise reduction and signal preservation.

Setting the weights too high may result in overemphasizing certain components and potentially distorting the desired signal, while setting them too low may compromise the filter's ability to effectively reduce noise.

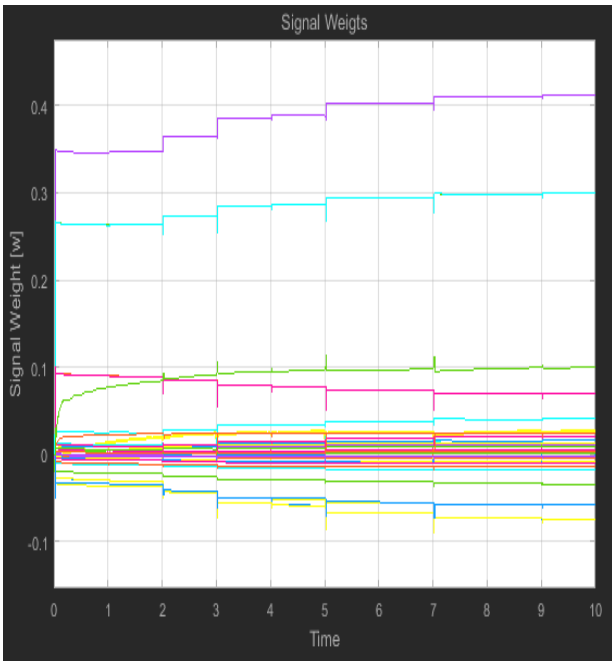

Figure 5 Signal weight

V. Conclusion

In conclusion, the implementation of the Recursive Least Squares (RLS) filter for feedback acoustic noise reduction in hearing aids has demonstrated promising outcomes. The significance of this study lies in its success in effectively reducing unwanted acoustic noise while preserving the integrity of the desired signals. The weighting mechanism employed by the RLS filter has proven to be instrumental in achieving a delicate balance between attenuating feedback noise and maintaining the clarity of the desired signal. This balance is crucial for individuals relying on hearing aids to experience an improved auditory environment. Furthermore, the input and output signal responses have showcased the filter's adaptability and efficacy in real-world scenarios. The RLS filter's ability to dynamically adjust its coefficients in response to changing acoustic environments ensures a versatile and responsive noise reduction mechanism. The study's emphasis on achieving a reduced error rate of 0.02 in feedback acoustic noise is particularly noteworthy. This reduction signifies a substantial improvement in the signal-to-noise ratio, contributing to enhanced speech intelligibility and overall user satisfaction. In essence, the findings underscore the practical applicability of the RLS filter in hearing aid technology, offering a viable solution for individuals seeking effective feedback acoustic noise reduction. The study not only contributes to advancements in hearing aid technology but also holds the potential to significantly enhance the quality of life for those with hearing impairments by providing a more refined and adaptive auditory experience.

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