iJRASET
International Journal For Research in
Applied Science and Engineering Technology

INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 11 **Issue:** V **Month of publication:** May 2023

DOI: https://doi.org/10.22214/ijraset.2023.53342

An Adaptive Beam Forming Algorithm for Reduction of Coexistence- Interference and Improved Performance in 5G Communication Systems

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Abstract: A couple of Wi-Fi systems coexisting in a 5G community might produce interference within the same frequency band, degrading the obtained sign's overall performance. This paper proposes a singular set of rules in antenna array processing to address interference-coexistence verbal exchange. We adopt a linear filter which is referred to as a Linearly confined minimal Variance (LCMV) filter out.

We impose a log-sum penalty at the coefficients and upload it to the price feature based on traditional singly linearly restrained least suggest square (LC-LMS). The iterative formula for filter weights is derived. We show that the brand-new approach's convergence price is quicker than the conventional one in the usage of simulations in an antenna environment with a signal of hobby, noise, and interference. Furthermore, the proposed approaches suggest rectangular errors (MSE) are shown. Our approach has a lower MSE than the traditional LC-LMS set of rules, in line with the findings of the experiments. The cautioned adaptive beam forming approach may be used in a 5G gadget to cope with sign and interference coexistence.

Keywords: Adaptive Beamforming, Least Constraint Least Mean Square (LC-LMS), Convergence Rate, Mean Square error (MSE).

I. INTRODUCTION

Extensive deployment of fifth-generation (5G) communication started to take place in a few countries around the world. Therefore, extensive studies on channel modelling and signal measurements with respect to the physics fundamentals are needed to properly design the architecture whereby such signals are precisely transmitted and received. The motivation for using such technology is that it promises higher data rates and enhanced network performance relative to the existing ones. This is typically achieved by exploiting wider ranges of bandwidth in higher frequency bands.

For instance, millimetre wave (mm-Wave) communication provides up to 10 Terabits data rates and spectral efficiency (SE) of approximately 100 bps/Hz over a bandwidth of about 270 Megabits per second (Mbps) (30–300 GHz frequency band). Clearly, the existing long-term evolution (LTE) system will no longer be able to embrace the network demands such as data rates and spectrum needed nor solve for the challenges such as the excessive interference them suitable for applications wherein near-field coupling is required to be minimized.

II. ADAPTIVE BEAMFORMING

Adaptive beamforming is a technique that has been used widely in the field of signal processing and wireless communication. The objective of adaptive beamforming is to enhance the detection of a signal in the presence of noise and interference. The technique can be applied in a wide range of applications, including radar, sonar, wireless communication, and medical imaging.

The basic concept of adaptive beamforming is to adjust the weights of an array of sensors so as to maximize the output signal-to-noise ratio.

The weights are adjusted by using an algorithm that takes into account the received signal and the noise and interference in the environment. The algorithm can be either adaptive or non-adaptive, depending on whether the weights are adjusted continuously or fixed.

III. LEAST CONSTRAINT LEAST MEAN SQUARE (LC-LMS)

The Least Constraint Least Mean Square (LCLMS) algorithm is a widely used method in signal processing and adaptive filtering. It is a variation of the Least Mean Square (LMS) algorithm that adds constraints to the optimization problem to improve the performance of the filter. The main idea behind the LCLMS algorithm is to add a constraint on the filter coefficients that limits the norm of the updates. This constraint can prevent the filter from diverging or oscillating in certain situations and can improve the convergence speed and stability of the filter.

Mathematically, the LCLMS algorithm can be formulated as follows:

$\Delta w(n) = \mu / (\gamma + \|w(n-1)\|^2) * x(n) * e(n)$

The LC-LMS algorithm can be seen as a compromise between the LMS algorithm, which has no constraints and can be unstable in certain situations, and the Normalized LMS (NLMS) algorithm, which normalizes the update vector to prevent instability but can be slow to converge.

The performance of the LCLMS algorithm depends on the choice of the parameters μ, γ, and ε. In general, smaller values of μ and ε can lead to more stable and conservative updates, but also slower convergence. Larger values of μ and ε can lead to faster convergence but also higher risks of instability.

In conclusion, the Least Constraint Least Mean Square algorithm is a useful technique for adaptive filtering that adds constraints to the optimization problem to improve the stability and convergence speed of the filter. It can be seen as a compromise between the LMS and NLMS algorithms, and its performance depends on the choice of parameters.

IV. CONVERGENCE RATE

In conclusion, the convergence rate is a vital concept in the field of mathematics and computer science. It is used to measure the efficiency of various algorithms and determines how fast an iterative process approaches its limit. The faster the convergence rate, the more efficient the algorithm. Calculating and improving convergence rates can lead to significant improvements in computation speed and efficiency, reducing the resources required to solve complex problems. Therefore, it is essential to continue research and development in this area to advance the field of mathematics and computer science.

V. MEAN SQUARE ERROR (MSE)

The concept of Mean Square Error (MSE) plays a crucial role in the realm of statistical analysis and machine learning. It is a measure of the average squared differences between the predicted and actual values of an outcome variable and is used to evaluate the accuracy of a model's predictions.

To begin understanding MSE, let's first define the variables involved. Suppose we have a set of n observations, with y representing the actual value and f representing the predicted value. The difference between the two is denoted as e.

VI. RESULTS AND DISCUSSION

The "OPTIMAL SOLUTION" is the signal with zero noise present in it, as there is no noise present in the optimal solution it converges most effectively but this solution is practically not possible as there will be some noise present at every time but the value of this noise can be reduced but it cannot be removed.

The following plot is between the modulus value of the coefficient vector W and the no. of iterations. "This plot gives us a brief idea about the relation between the convergence factor and the convergence rate".

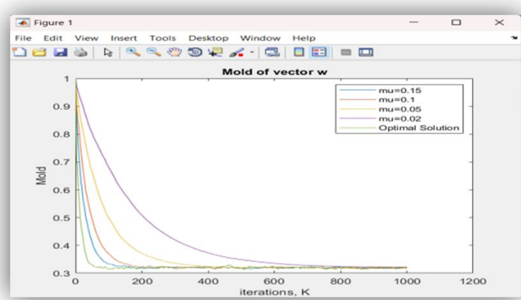


Fig1: Comparison of Convergence Rate with different mu

Mean Square Error value is defined as the steady state value of the signal. The plot of the mean square error is plotted between the value of the mean square error and the number of iterations. By Changing the value of the convergence factor the value of mean square error is also affected.

As the value of the convergence factor decreases, the mean square error value becomes more distinguishable or stable. "Increasing the value of the convergence factor increases the number of distortions in the mean square error value." When the convergence factor decreases, the mean square error value becomes more stable.

According to the final beam pattern plot the interference is at 40 degrees and SOI at 0 degrees. The pattern gets max gain at 0 degrees and produces a deep null point at 40 degrees. Both algorithms get similar gains at SOI incident degree. But Log-sum LC-LMS has a deeper null point and the gain is about 35dB lower.

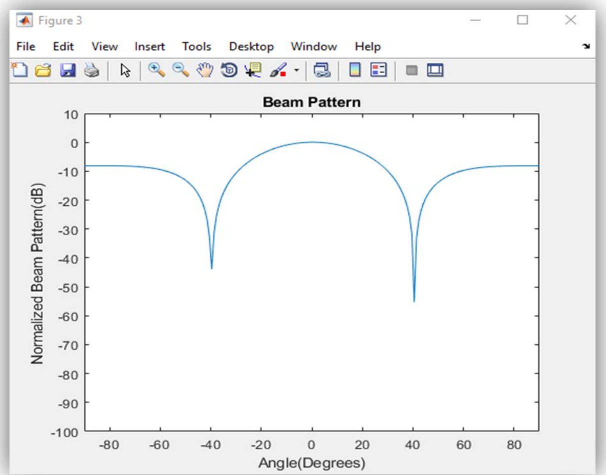


Fig 2: Beam pattern

VII. CONCLUSION

We proposed a new algorithm based on the LC-LMS. We add a log-sum penalty to the object function and give theoretical analysis step by step until derive the final formula. Then experiments are carried out on the MATLAB platform. The first experiment aims to compare the newly proposed algorithm with LC LMS in convergence rate and steady state. The results prove the effectiveness and superiority of the new method. In the second experiment, we analyse the factors that may affect the performance of the method. We can see that the choice of parameter 't' determines the algorithm performance, so 't' should be set properly. Finally, we make a comparison in the beam pattern. The log-sum LC-LMS has the same performance as LC-LMS or better.

REFERENCES

[1] S. Wang, Y. Wang, B. Xu, Y. Li, and W. Xu, "Capacity of two-way in-band full-duplex relaying with imperfect channel state information," IEICE Transactions on Communications., Volume E101-B, pp. 1108–1115, April 2018.

[2] Z. Zhao, S. Bu, T. Zhao, Z. Yin, M. Peng, Z. Ding, and Tony Q. S. Quek, "On the design of computation offloading in fog radio access networks," IEEE Transactions on Vehicular Technology, Volume 68, pp.508-517, July 2019.

[3] D.L. Duttweiler, "Proportionate normalized least-mean squares adaptation in echo cancelers," IEEE Transactions on Speech Audio Process., Volume 8, pp. 508C- 518, July 2000.

[4] Y. Chen, Y. Gu, and A. O. Hero, "Sparse LMS for system identification," in Proceeding of the IEEE International Conference on Acoustics, Speech and Signal Processing, Volume 5, pp. 3125–3128, May 2009.

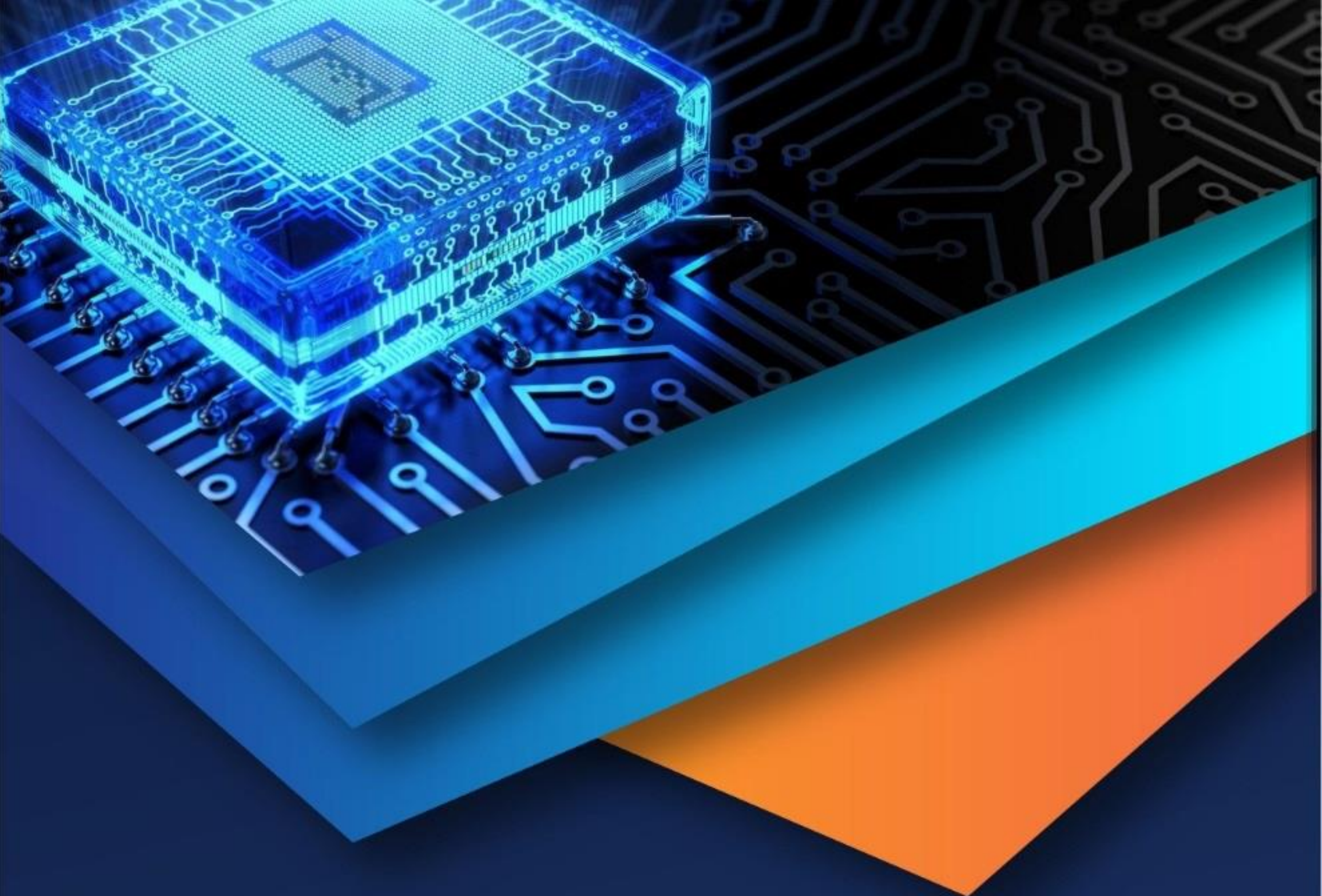
[5] E. J. Cand 'es, M. Wakin, and S. Boyd, "Enhancing sparsity by reweighted l1 minimization," To appear in J. Fourier Analog Application, Volume 14, pp. 13-14, October 2007.

[6] S. Wang, D. D Wang, C. Li, and W. B Xu. "Full Duplex AF and DF Relaying Under Channel Estimation Errors for voice-to-voice Communications," IEEE Access, Volume 6, pp. 65321-65332, November 2018.

[7] W.Y. Chen, R.A. Haddad, "A variable step size LMS algorithm," Proceedings of the 33rd Midwest Symposium on Circuits and Systems, Volume 1, pp. 423–426,1991.

[8] Zhang Yuan, Xi Songtao, "Application of New LMS Adaptive Filtering Algorithm with Variable Step Size in Adaptive Echo Cancellation". 17th IEEE International Conference on Communication Technology, Volume 9, pp. 1715-1719, August 2017.

[9] Z. Zhao, M. Xu, Yong Li, and M. Peng, “A non-orthogonal multiple access-based multicast schemes in wireless content caching networks,” IEEE J. Sel. Areas Communication, Volume 35, pp. 2723–2735, July 2017.
[10] O. L. Frost, “An algorithm for linearly constrained adaptive array processing,” Proceedings of the IEEE, volume 60, pp. 926-C935, September 1972.
[11] M. Godavarti and A. O. Hero, “Partial update LMS algorithms,” IEEE Transactions Signal Process., Volume. 53, pp. 2382-2399, April 2005.



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