# AN ENHANCED ADAPTIVE BEAMSHAPING ALGORITHM FOR 5G INTERFERENCE COMMUNICATION

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**Abstract -** —Multiple wireless systems coexisting in a 5G network might produce interference in the same frequency band, degrading the received signal's performance. In this project, a novel algorithm is proposed in antenna array processing to handle interference-coexistence communication. Adopt a linear filter which is called Linearly Constrained Minimum Variance (LCMV) filter. Impose a log-sum penalty on the coefficients and add it to the cost function based on classic singly linearly constrained least mean square (LC-LMS). The iterative formula for filter weights is derived. Demonstrate that the new method's convergence rate is faster than the traditional one using simulations in an antenna environment with a signal of interest, noise, and interferences. Furthermore, the proposed method's mean-square-error (MSE) is confirmed. Technique has a lower MSE than the classic LC-LMS algorithm, according to the findings of the experiments. The suggested adaptive beam forming approachcan be used in a 5G system to deal with signal and interference coexistence.

**KEY words -** Interference-coexistence, LC-LMS, Log-Sum Penalty. I

## I. INTRODUCTION

The deployment and commercial operation of 5G systems are speeding up to meet the anticipated demands of next decade in data transmission. 5G networks are emerging intelligent systems which involve the application of advanced signal processing [1], D2D [2], internet of things (IOT), edge computing [3], and wireless access technologies [4] that have drawn much attention in recent years. In a 5G network, coexistence of multiple wireless systems can cause interference in the same frequency band and deteriorate the received signal. The anti-interference communication will still play an important role in the network. The adaptive beamforming technology has always been an import part in antenna processing to handle interference problem. The direction information is added into the transmitted signal with the technology and then the mixed signal, including signal of interest(SOI), interferences and noise, is received at receive end. Actually, SOI have different Direction of Arrival(DOA) compared with interferences. Adaptive algorithms ensure to produce null points towards the directions of interferences while maintain the gain of SOI. of LMS have fixed the problems to a certain extent. The adjustment of LMS is directly proportional to the input vector, so when the input is very large, the algorithm suffers from a gradient noise amplification problem. Normalized LMS algorithm [6] takes the squared Euclidean norm of the input vector at each adaptation and has well solved the problem. Also, variable-step LMS [7] [8] have improved the convergence rate a lot. All these above mentioned derivations minimize the mean-square-error(MSE) between the estimation signal and the desired one. The criterion used in derivation is Minimum-Mean-Square-Error(MMSE) which takes MSE as cost function. It is noted that no constraints have been added to the solution. In [9], based on LMS in MMSE, the author introduce two different penalties on the cost function. Results show that the algorithm with penalty outperforms LC-LMS in convergence.

## II. LITERATURE SURVEY

Investigate the capacity performance of an in-band full-duplex (IBFD) amplify-and-forward two-way relay system under the effect of residual loop-back-interference (LBI). In a two-way IBFD relay system, two IBFD nodes exchange data with each other via an IBFD relay. Both two-way relaying and IBFD one-way relaying could double the spectrum efficiency theoretically. However, due to imperfect channel estimation, the performance of two-way relaying is degraded by self-interference at the receiver.A key problem of content caching networks is that extra radio resource blocks are consumed to push content objects, which leads to a decline of spectrum efficiency. To solve this problem, a non-orthogonal multiple access-based multicast (NOMA-MC) scheme is proposed in this paper, where pushing and multicasting content objects can be accomplished simultaneously, and thus the spectrum efficiency can be improved significantly. On typical echo paths, the proportionate normalized least-mean-squares (PNLMS) adaptation algorithm converges significantly faster than the normalized least-mean-squares (NLMS) algorithm generally used in echo cancelers to date. In PNLMS adaptation, the adaptation gain at each tap position varies from position to position and is roughly proportional at each tap position to the absolute value of the current tap weight estimate. A new approach to adaptive system identification when the system model is sparse. The approach applies $\ell 1$ relaxation, common in compressive sensing, to improve the performance of LMS-type adaptive methods. This results in two new algorithms, the zero-attracting LMS (ZA-LMS) and the reweighted zero-attracting LMS (RZA-LMS). The ZA-LMS is derived via combining a $\ell 1$ norm penalty on the coefficients into the quadratic LMS cost function, which generates a zero attractor in theLMS iteration. Stephen P. Boyd: It is now well understood that (1) it is possible to reconstruct sparse signals exactly from what appear to be highly incomplete sets of linear measurements and (2) that this can be done by constrained 1 minimization. In this paper, we study a novel method for sparse signal recovery that in many situations outperforms 1 minimization in the sense that substantially fewer measurements are needed.

## III. EXISTING METHOD

Least Mean Square(LMS):

In adaptive filtering applications for modelling, equalization, control, echo cancellation, and beamforming, the widely used least-mean-square (LMS) algorithm has proven to be both a robust andeasily-implemented method for on-line estimation of time-varying system parameters [7]. Fig.1 shows a generic adaptive beamforming system which requires a reference signal, the outputs of the individual sensors are linearly combined after being scaled using corresponding weights such that the antenna array pattern is optimizedto have maximum possible gain in the direction of the desired signal and nulls in direction of interferers .

The LMS algorithm can be described by the following three equations,

$$y(n) = \mathbf{w}^H(n)\mathbf{x}(n)$$

$$e(n) = d(n) - y(n)$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\mathbf{x}(n)e^*(n)$$

The LC-LMS filter aims to minimize the output power and maintain the response of the SOI. The optimization problem can be written as

$$\min P_{out} = \min E\left[|y(n)|^2\right]$$
$$\text{s.t.} \quad \mathrm{s^H w = z}$$

## IV PROPOSED METHOD

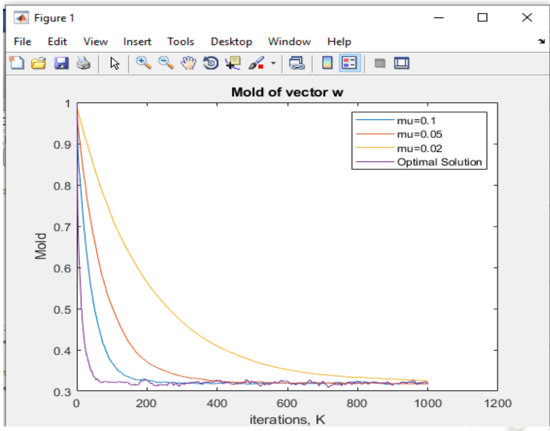
Log-sum penalty imposed on the coefficients and add it into the cost function and derive the iterative formula of filter weights.

$$\min P_{out} = \min E\left[|y(n)|^2\right]$$
$$\text{s.t.} \begin{cases} \mathbf{s}^H\mathbf{w} = z \\ \sum_{i=1}^{M} \log(1+ \mid w_i \mid /\varepsilon') = t \end{cases}$$

The Lagrange function can be written as

$$L(w) = E\left[|y(n)|^2\right] + \lambda_1(\mathbf{s}^H\mathbf{w} - z) + \lambda_2\left[\sum_{i=1}^{M} \log(1+ \mid w_i \mid /\varepsilon') - t\right]$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \frac{\mu}{2}\left\{2\mathbf{R}\mathbf{w}(n) + \mathbf{s}\lambda_1 + \lambda_2\mathbf{B}\right\}$$



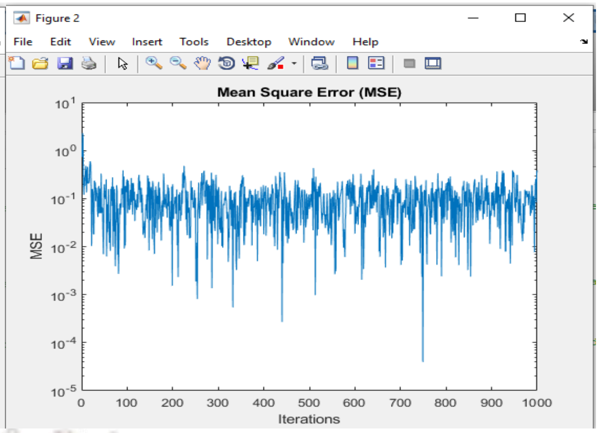




Figure1 :Convergence Rate with different muFigure 2 :Mean Square Error with M=4

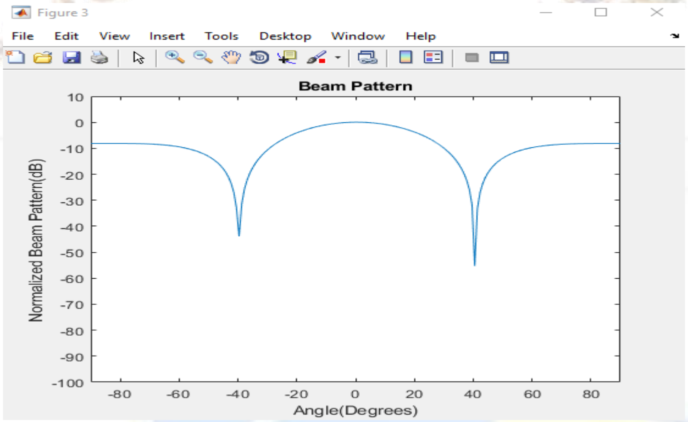



Figure 3. :Beam pattern

## IV. CONCLUSIONS

Proposed a new algorithm based on the LC-LMS. We add 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 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 analyze 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 beam pattern. The log-sum LC-LMS has the same performance as LC-LMS, or better.

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