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**Maximum correntropy criterion based sparse adaptive filtering algorithms for robust channel estimation under non-Gaussian environments.** (English) Zbl 1395.93544  
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Summary: Sparse adaptive channel estimation problem is one of the most important topics in broadband wireless communications systems due to its simplicity and robustness. So far many sparsity-aware channel estimation algorithms have been developed based on the well-known minimum mean square error (MMSE) criterion, such as the zero-attracting least mean square (ZALMS), which are robust under Gaussian assumption. In non-Gaussian environments, however, these methods are often no longer robust especially when systems are disturbed by random impulsive noises. To address this problem, we propose in this work a robust sparse adaptive filtering algorithm using correntropy induced metric (CIM) penalized maximum correntropy criterion (MCC) rather than conventional MMSE criterion for robust channel estimation. Specifically, MCC is utilized to mitigate the impulsive noise while CIM is adopted to exploit the channel sparsity efficiently. Both theoretical analysis and computer simulations are provided to corroborate the proposed methods.

**MSC:**

- 93E11 Filtering in stochastic control theory
- 93E10 Estimation and detection in stochastic control theory
- 93B35 Sensitivity (robustness)
- 94A12 Signal theory (characterization, reconstruction, filtering, etc.)
- 93D21 Adaptive or robust stabilization

Cited in 17 Documents

**Keywords:**

Maximum correntropy criterion; sparse adaptive filtering algorithms; robust channel estimation; broadband wireless communication

**Full Text:** [DOI](#) [arXiv](#)

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