

Adaptive signal processing techniques are frequently involved in modern communications systems. In this context, the key block of any adaptive system consists of the adaptive algorithm. This influences the system performance in terms of convergence rate, misadjustment, and stability. These performance criteria depend on the adaptation parameters [i.e., the step-size in case of the LMS (least-mean-square) algorithms, and the forgetting factor in case of the RLS (recursive least-squares) algorithms]. Using constant values for these parameters lead to a compromise between the performances criteria. A solution is to analyze the convergence state of the algorithms and to use variable adaptation parameters, thus resulting adaptive algorithms with variable convergence. Nevertheless, the existing solutions do not offer a proper accuracy, since their convergence criteria (error energy or mean square deviation) are not very precisely in detecting the convergence state of the algorithms. The main goal of this research project is to propose new adaptive algorithms with variable convergence, based on new models for analyzing the convergence state. The basic idea is to separate the convergence and misadjustment components of the algorithms, and to incorporate them into specific cost functions. The procedure will be applied for both the LMS and RLS family of algorithms, resulting new VSS-LMS (variable step-size LMS), respectively new VFF-RLS (variable forgetting factor RLS) algorithms. Also, we aim to reduce the computational complexity of the proposed VFF-RLS algorithms by using the DCD (dichotomous coordinate descent) methods. The proposed algorithms will be implemented on FPGA and tested in an acoustic echo cancellation scenario.