The presence of the ambient noise represents one of the most disturbing factors within the voice communication systems. The problem is more difficult in the case of the multi-channel systems, where the noise from one location is added with the noise from the other locations, which further increases the final perceptual noise level. The research project approaches this field taking into account the performances enhancement of two major sub-systems within this type of applications: the adaptive filter and the microphones array. In the present, the most frequently used adaptive algorithms for noise cancellation are the fast affine projection (FAP) algorithms. The main problems related to this class of algorithms are the numerical stability and the computational complexity. Our first goal is to propose some new versions of these algorithms based on the QRD and DCD methods (in order to improve the numerical stability and to reduce the computational complexity), an also to propose new variable step-size FAP algorithms (in order to improve the convergence rate and the tracking capabilities). Regarding the microphone array part of the project, we aim to approach this problem from the perspective of a MIMO communication system, for further extend some specific techniques and algorithms from this field to the microphone array beamforming. In this context, we will develop new algorithms for acoustic channel estimation and we will combine them with the noise cancellation methods. All the proposed solutions will be implemented using fixed-point DSP and FPGA. In the end of the project, we aim to design and implement a multi-channel noise cancellation system and to validate it within a teleconferencing application, for further integration within the technological and commercial circuit.