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A Noise Reduction Preprocessor for Mobile Voice Communication

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Abstract

We describe a speech enhancement algorithm which leads to significant quality and intelligibility improvements when used as a preprocessor to a low bit rate speech coder. This algorithm was developed in conjunction with the Mixed Excitation Linear Prediction (MELP) coder which, by itself, is highly susceptible to environmental noise. The paper presents novel as well as known speech and noise estimation techniques and combines them into a highly effective speech enhancement system. The algorithm is based on short time spectral amplitude estimation, soft-decision gain modification, tracking of the *a priori* probability of speech absence, and Minimum Statistics noise power estimation. Special emphasis is placed on enhancing the performance of the preprocessor in non-stationary noise environments.

I. Introduction

With the advent and wide dissemination of mobile voice communication systems, telephone conversations are increasingly disturbed by environmental noise. This is especially true in hands-free

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