A COMPUTATIONALLY INEXPENSIVE ALGORITHM FOR ENHANCING SPEECH DISTURBED BY COLOURED NOISE

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The common problem of enhancing speech disturbed by background noise has been examined, under the assumption that no a priori knowledge about the noise is available. That is why we have built an algorithm trying to retain the part of the signal that follows a speech production model.

Recalling a previous well-known works by Lim and Oppenheim in the frequency domain ("All-pole modelling of degraded speech"), we have obtained a quite simple algorithm in the time domain. Briefly, it consists in computing the prediction coefficients $a_i$ of a low order linear prediction (typically 4) and in using those coefficients to compute "enhanced" values of the samples, by combining the corresponding "noisy" value and a prediction of the previous enhanced values. A problem is that the coefficients $a_i$ used are obtained from the noisy signal and are therefore biased; however, experience showed appreciable improvements of subjective quality and naturalness (while intelligibility hasn't been tested yet), even for very poor SNR (till 5-dB) and coloured noises, when their spectra do not present high resonances. This is particularly interesting since the algorithm is theoretically justified only for a white noise.

No inferred knowledge about the characteristics of the actual noise is required, but future research will examine if the algorithm could work even better with some information about them, as well as about the speaker or localization of some particular sounds. Real time processing by optimizing the algorithm and using a signal processing chip is also planned.