

NON-UNIFORM SUB-BAND SPEECH CODER FOR SPEECH ENHANCEMENT

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Abstract

Here, a novel method for single-channel speech enhancement based on subband coding in frequency domain is proposed. The input signal is divided into a number of sub bands that are individually weighed in time domain, in accordance to the short time Signal-to-Noise Ratio (SNR) in each sub band estimation at every time instant. Instead of focusing on suppression the noise on speech enhancement is focused such as all the fundamental components of original speech signal is concentric towards the centre of the spectrum of each band. The synthesis system performs the inverse of the analysis operation. Between every two samples in each band we put in the value zero to increase the sample rate. Instead of applying the speech coder for full-band speech or uniform sub-band speech, speech enhancement is performed by applying the sub-band coding in frequency domain to non-uniform sub-band signals obtained from the decomposition of whole-band speech using fir filters. Simulation results indicate that in terms of perceptual evaluation speech quality (PESQ), our proposed method shows a relative improvement of 21.4% with compression ratio of 25.38% over the conventional full-band coding technique and also improves the speed of convergence and lowers the steady-state MSE value.

Keywords: Speech Enhancement, Sub-band Coding