On the Use of Artificial Bandwidth Extension Techniques in Wideband Speech Communications

Peter Jax and Peter Vary

Institut für Nachrichtengeräte und Datenverarbeitung (ind)
RWTH Aachen University, Germany
{jax|vary}@ind.rwth-aachen.de

Abstract

In this contribution we report on recent advances in *artificial bandwidth extension* (BWE) techniques. Stand-alone BWE approaches are recently being developed for use in receiver terminals or within the network. The aim is to improve the speech quality by estimating a *wideband* (WB, acoustic bandwidth 50 – 7000 Hz) counterpart from the received telephone speech (acoustic bandwidth 300 – 3400 Hz). The concept of BWE is based on a linear source-filter model of speech production. For short segments of the telephone speech parameters of the (wideband) model are estimated. These parameters are then used to re-synthesize the missing frequency components of the speech signal.

Apart from the stand-alone application, BWE techniques are used more and more in present speech and audio coding algorithms. For example, in new wideband speech coding standards like the *Adaptive Multi-Rate Wideband* (AMR-WB) codec the highest frequency sub-band (e.g. 6.4–7 kHz) is extrapolated from low frequency components. Further examples for coding with BWE techniques are the currently standardized Extended AMR-WB+ and Enhanced aacPlus speech/audio codecs.