Special Session

ADVANCES ON LINEAR-IN-THE-PARAMETERS NONLINEAR FILTERS AND THEIR APPLICATION TO AUDIO AND SPEECH PROCESSING

TECHNICAL SCOPE

Nonlinear system modeling has always drawn a great interest due to a wide range of applications that can be found in real-world problems. Such interest has led to the development of a series of different learning methods over the years. One of the most popular models for nonlinear system identification is the class of linear-in-the-parameters (LIP) nonlinear filters, which are characterized by a linear combination of any nonlinear representation/expansion of the input signal. This class of filters includes several models, which are usually distinguished according to the nonlinear transformation adopted: adaptive Volterra filters, polynomial filters, kernel adaptive filters, Fourier nonlinear filters, functional link-based filters, Hammerstein spline adaptive filters, extreme learning machines, among others. The linear filtering technique chosen for a LIP nonlinear filter is usually related to the application of interest. Such flexibility of the LIP nonlinear filters allows to implement online, batch or semi-batch algorithms, applying them from regression to classification problems in different fields. In recent years, nonlinear techniques have gained an increased interest in audio and speech processing, where they have been used mainly for nonlinear modeling, nonlinear compensation, and signal enhancement. The aim of this special session is to bring together leading researchers in the fields of linear and nonlinear signal processing and machine learning for signal processing and to provide recent advances on LIP nonlinear filters and their applications to audio and speech processing.

TOPICS OF INTEREST

- Nonlinear Transformations for LIP Filters
- Adaptive Algorithms for LIP Nonlinear Filters
- LIP-based Machine Learning Algorithms
- Sparse Representations
- Audio Signal Processing

- Nonlinear Acoustic Echo Cancellation
- Nonlinear Equalization and Linearization
- Nonlinear Enhancement
- Active Noise Control
- Speech Processing

PAPER SUBMISSION

Prospective authors may submit their manuscripts through EUSIPCO 2016 submission system. All the submissions will go through peer review. More details on paper submission can be found on http://www.eusipco2016.org/. The deadline for full paper submissions is February 8th, 2016.

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