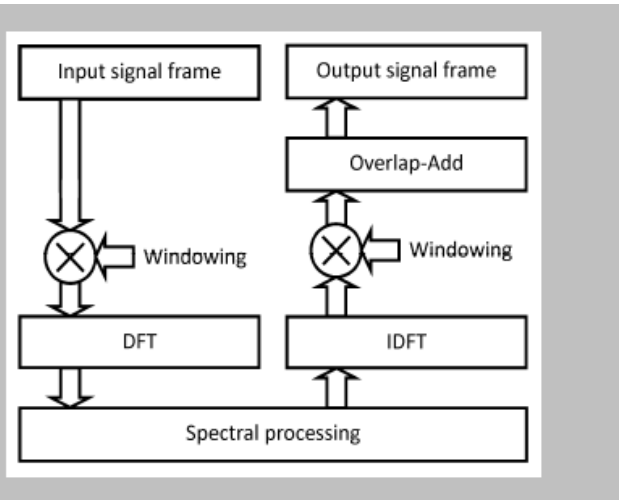
Hes·so
Haute Ecole Spécialisée
de Suisse occidentale
Fachhochschule Westschweiz
University of Applied Sciences
Western Switzerland



SMARTTIMEFREQ

Smart Time-Frequency Signal Processing for Audio Processing and for Vibration Analysis

Project Summary

The project dealt with the decomposition of time-dependent input signals like speech / audio signals, with possible extension to instrumentation for the characterization of machine vibration phenomena. Purpose of the project was to study two kinds of time-frequency signal decomposition algorithms, which receive the input signal to decompose it into several frequency channels by using a Discrete Fourier Transform, so that the resulting channels can be processed in the frequency domain according to requirements encountered in applications such as signal analysis, signal enhancement, frequency-dependent amplification, signal coding, etc. Once the frequency domain processing has been performed, it is possible to assemble the data from the different frequency channels to reconstruct the resulting time signal. The signal time-frequency decomposition refers to a so-called "analysis filterbank", whereas the reverse frequency-time conversion and reconstruction of the resulting time signal is implemented in a so-called "synthesis filterbank".

The first time-frequency filterbank algorithm studied concerned the "Weighted Overlap-Add" (WOLA) filterbank, that is known for a long time. Purpose of the research was to study the interactions of the numerous design parameters according to application needs. The results showed that the design parameters are tightly interdependent so that every selected parameter configuration needs to be fully characterized for a given application context.

The second time-frequency filterbank algorithm studied and developed is denoted as "Nonuniform Overlap-Add" (NOLA) filterbank. It implements a nonuniform frequency resolution relying on the Bark scale that takes into account the perceptually varying frequency resolution of the human ear. This nonuniform frequency resolution is expected to match better the requirements of specific speech / audio processing applications. The structure developed is considered as original, however quantitative and qualitative comparisons with alternative NOLA filterbanks shall still be performed.

Finally, an extra research activity was deployed concerning the modeling and joint specification of an anti-aliasing filter followed by an (oversampled) A/D converter, so as to control the overall signal tonoise ratio achieved when acquiring an analog signal. This work is to our best knowledge original.

Valorisation

The valorisation of the results achieved in the project is organized along three axes, namely:

i) Contribution to education in the frame of courses and of several student projects (already done).
ii) Several (IEEE) conference papers and a journal article are pending to publish the results.
iii) Contacts are organized with local companies to enable the reuse of the results in future projects.

Contact / Dr Michael Ansorge (michael.ansorge@hefr.ch)
Autors / Dr Michael Ansorge, Ms Liana Ugnat / EIA-FR

This project has been carried out at the Institute of Smart and Secured Systems (iSIS) at EIA-FR

ISYS

Ecole d'ingénieurs et d'architectes de Fribourg
Hochschule für Technik und Architektur Freiburg

iSIS
Institute of Smart and Secured Systems