Statistical comparison of wavelet and Fourier series approximation quality for audio signals

Mateusz John[1] **and** **Agnieszka Kwita**[2]

[1] *Poznań University of Technology, Poland*
[2] *Wrocław University of Science and Technology, Poland*

Abstract

A statistical procedure for comparing the quality of audio signal approximation obtained using the discrete wavelet transform (Daubechies wavelets dbN, $N = 1, \ldots, 10$) and Fourier series approximation, estimated by the least squares method, is presented. The audio signal is sampled either uniformly or adaptively, with node density in the adaptive scheme proportional to the local variability of the signal amplitude. This makes it possible to assess the influence of the sampling scheme on reconstruction accuracy. Optionally, the wavelet detail coefficients may be denoised by soft or hard thresholding prior to reconstruction.

Since reconstruction errors obtained for different wavelets arise from the same observation vector, they are treated as dependent samples. Differences between wavelets are assessed using a nonparametric test for repeated measures, followed by post-hoc pairwise comparisons for dependent samples, with a correction for multiple testing and the computation of effect size measures. Additionally, a compression analysis based on M-term approximation in the wavelet basis is performed to determine which wavelet achieves a given reconstruction error level with the fewest retained coefficients.

The entire procedure is implemented in a custom Python application (built using `PyWavelets`, `scipy.stats`, `numpy`), enabling audio signal acquisition, sampling, approximation, and full statistical analysis together with result visualization.

Keywords

Signal approximation, Wavelet transform, Fourier series, Friedman test, Wilcoxon test, Wavelet coefficient compression.