Audio Beat Tracking Algorithm for MIREX 2006

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Abstract

This paper describes an audio beat tracking algorithm submitted to the MIREX 2006 contest. The algorithm has been described in detail in the article "Analysis of the Meter of Acoustic Musical Signals" published in IEEE Trans. Audio, Speech and Language Processing, 14(1), 2006. In summary, the method analyses musical meter jointly at three time scales, of which only the beat (tactus) level is used in this MIREX task.

Keywords: Musical meter analysis, beat tracking.

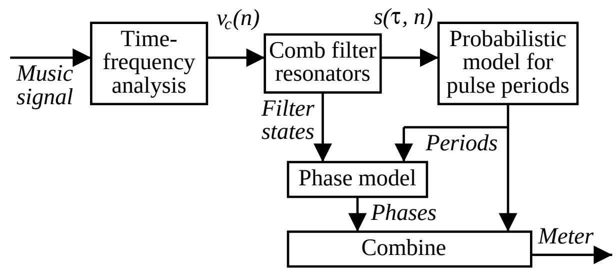

Figure 1. Overview of the meter analysis method.

1. Introduction

The beat tracking algorithm employed here is identical to that described in [1]. [1] The algorithm was originally implemented by the author in 2003 and converted to C++ by Jouni Paulus in Spring 2004. The algorithm and its parameter values have been untouched since then.

In [1], both causal and non-causal versions of the method were described. The practical difference between the two is that the causal version generates beat estimates based on past samples, whereas the non-causal version does (Viterbi) backtracking to find the globally optimal beat track after hearing the entire excerpt. The backtracking improves accuracy especially during the beginning of an analysis signal.

Here we employed the causal version of the algorithm. The non-causal version would be slightly more accurate especially for short analysis signals like those used in this contest, but also less realistic for on-line applications.

2. The meter analysis algorithm

The aim of the method proposed in [1] is to estimate the meter of acoustic musical signals at three levels: at the tactus, tatum, and measure-pulse levels. An overview of the method is shown in Fig. 1.

For the time-frequency analysis part, a technique is employed which aims at measuring the degree of spectral change, or, "accent" in music signals. In brief, preliminary time-frequency analysis is conducted using a quite large number of subbands and by measuring the degree of spectral change at these channels. Then, adjacent bands are combined to arrive at four bandwise accent signals, for which periodicity analysis is carried out.

Periodicity analysis of the bandwise accent signals is performed using a bank of comb filter resonators very similar to those used by Scheirer in [2]. Before we ended up using comb filters, four different period estimation algorithms were evaluated. A bank of comb filter resonators was chosen because it was the least complex among the three best-performing algorithms.

The comb filters serve as feature extractors for two probabilistic models. One model is used to estimate the period-lengths of metrical pulses at different levels. The other model is used to estimate the corresponding phases (see Fig. 1). The probabilistic models encode prior musical knowledge regarding well-formed musical meters. In brief, the models take into account the dependencies between different pulse levels (tatum, tactus, and measure) and, additionally, implement temporal tying between successive meter estimates.

3. Acknowledgements

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References

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