

MIREX-2013 “AUDIO BEAT TRACKING” TASK: IRCAMBEAT SUBMISSION

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ABSTRACT

This extended abstract details a submission to the Music Information Retrieval Evaluation eXchange (MIREX) 2013 for the “Audio Beat Tracking” task. The system named ircambeat performs time-variable tempo and meter estimation, beat and downbeat marking. Detailed description of the various parts can be found in [1] and [2]. We briefly summarized them below.

1. IRCAMBEAT IMPLEMENTATION

The last version of Ircambeat is available as C++ executable or library running under Linux, Windows-XP and Mac-OS-X. The last version performs time-variable tempo and meter estimation, beat and downbeat marking.

2. IRCAMBEAT ALGORITHM DESCRIPTION

The flowchart of ircambeat is represented in Figure 1.

2.1 Tempo and meter estimation

The tempo and meter estimation algorithm works in three stages.

First, an onset-energy-function $f(t)$ is extracted from the audio signal by computing a reassigned spectral-energy-flux (using time and frequency reassignment for better precision).

Second, the dominant periodicities of $f(t)$ over time are estimated using a combination of Discrete Fourier Transform and Frequency-Mapped Auto-Correlation-Function. The combination of both allows to better emphasizing the periodicities due to the meter, the beat and the tatum periodicities in $f(t)$. We note $p(t)$ the resulting function.

Finally, a Viterbi decoding algorithm is used to decode simultaneously the tempo and the meter. For this, we define states of a hidden Markov model as all the combinations of possible tempi and meter (among 22: binary grouping of beat/ binary subdivision of beat, 23: binary/ ternary and 32: ternary/ binary). Given $p(t)$, we compute the observation probabilities of the states over time. The decoding then produces the best estimates of tempo and meter over time.

More details about the algorithm can be found in [1].

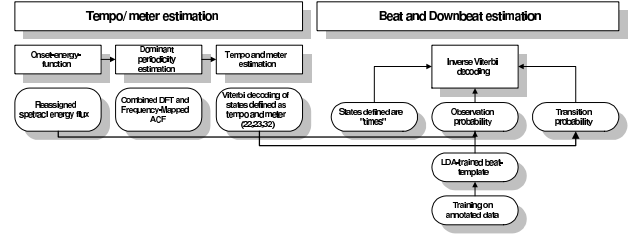


Figure 1. Flowchart of ircambeat

2.2 Beat and downbeat tracking

Beat and downbeat positions are estimated simultaneously using an inverse Viterbi formulation. In this formulation, a state is defined as a specific time in a specific beat position.

Observation probabilities of states (times in a specific beat position) are obtained using LDA-trained beat-templates, Chroma and Spectral Balance variation over time analysis. The beat-template is obtained by considering the function $f(t)$ inside a measure as a N-dimensional feature vector. A two-class (beat/ non-beat) problem is then solved using LDA and a training set. The resulting LDA-axe is then used as the best beat-template in order to perform discrimination between beat and non-beat positions.

More details about the algorithm can be found in [2].

2.3 Configurations

For MIREX-2013, ircambeat has been submitted with three different configurations

ircambeat-ve	variable tempo and estimated meter
ircambeat-vf	variable tempo and fixed meter to 22
ircambeat-maz	specific configuration in order to take into account the specific notion of tempo of Mazurka music. OPTIONS = '-t 120 -tempo-sigma 120 -no-downbeat -period-window 5 -tempo-trans-sigma 300 -template-size 0.2'; This configuration allows somehow to bypass prior tempo probability (-tempo-sigma 120), transition probability between tempi is also set very large (-tempo-trans-sigma 300), the beat-template is set very short (-template-size 0.2). Using all this makes ircambeat acts most like an onset tracker than a beattracker.

3. MIREX-2013 RESULTS AND DISCUSSIONS

TO DO

4. CONCLUSION

TO DO

5. ACKNOWLEDGMENTS

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6. REFERENCES

- [1] G. Peeters. Template-based estimation of time-varying tempo. *EURASIP Journal on Applied Signal Processing*, 2007(1):158–158, 2007. doi:10.1155/2007/67215.
- [2] G. Peeters and H. Papadopoulos. Simultaneous beat and downbeat-tracking using a probabilistic framework: theory and large-scale evaluation. *IEEE. Trans. on Audio, Speech and Language Processing*, 19(6):1754–1769, 2011.