

# REAL TIME AUDIO TO SCORE ALIGNMENT BASED ON NLS MULTIPITCH ESTIMATION (2) MIREX 2010

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## ABSTRACT

Real-time audio to score alignment is a valuable application in some music areas. This goal has been addressed from different point of views during the last years. Multipitch estimation was one of the first processing algorithm used for audio to score alignment. Here, a multipitch estimator is utilized to inform about the probability that a set of notes is active in a signal frame. In this way, the system is able to associate the states of score to the time of the performance. This approach is designed to be low complexity in order to be used in real-time applications.

## 1. INTRODUCTION

The alignment between the performance and the score is usually addressed in a two step algorithm. First audio and score are analyzed to extract the probability that a certain audio frame belong a score state. Second, these probabilities are utilized to perform the correct link between audio time and score states.

The analysis between audio and score has been addressed with well-known tools from audio signal processing. In [1] [2], frequency techniques based on Short-Time Fourier Transform are used. A correlation between each audio frame transform and the ideal harmonically-related frequency for each given set of notes from the score is computed [2]. In [3] an onset detection followed by pitch detection is proposed. In [4], note onsets are computed by decomposing the audio signal into spectral bands corresponding to the fundamental pitches and harmonics, followed by computation of the positions of significant energy increases for each band.

In order to perform a link between audio time and score states two approaches are generally utilized: Dynamic Time Warping (DTW) [5] or probabilistic models based on Hidden Markov Models (HMMs). DTW has predominantly been used for offline techniques being of low complexity and giving promising results in different scenarios [2] [6].

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Typically HMMs and other graphical have been used for online applications [7] [8] [9] [10].

In this proposal, a NLS multipitch estimator is used as a processing tool to obtain probabilities between audio time and score states. Then, a sub-optimal approach of DTW has been designed in order to take decisions with limited latency in order to allow real-time decision to the score follower.

## 2. MULTIPITCH ESTIMATION

One of the simplest multipitch estimators is nonlinear least-squares method (NLS). The NLS estimates are obtained as the set of fundamental frequencies that minimizes the 2-norm of the difference between the observed signal and the signal model [11]. The signal model includes a set of harmonically-related frequencies at integer multiples of the each fundamental frequency  $\omega_k$ . Due to this property, NLS method can be implemented efficiently for a linear grid search over a set of  $\omega_k$  using a fast Fourier transform (FFT) [12].

One of the main problems of NLS [12] is the estimation of the order or the number of partials that belong to each fundamental frequency. Some solutions have been proposed in the literature for order estimation in multipitch methods [13]. Here, a solution based on the perceptual significance of each partial is proposed. The perceptual significance of frequency peaks is computed following the perceptual model of [14]. This perceptual models gives a value larger than one to those peaks audible and lower than one to those peaks below the hearing threshold. A cumulative product of the perceptual significance of the partials belonging to the each evaluated fundamental frequency is calculated. The order for each fundamental frequency is supposed to be equal to the partial position at which the cumulative product presents the maximum. With the idea of evaluating, the benefits of proposed order estimation two different approaches are presented at the competition: with and without the order estimation. The version without order estimation supposes a fixed order of 9 partials for all fundamental frequencies.

Finally, the algorithm to obtain the probability that a score state (a set of notes) is played at a audio frame is the following:

- Obtain the FFT of the frame.
- Compute the amplitudes for all spectral peaks of the frame.
- Compute the perceptual significance for all spectral peaks of the frame.
- For each note in the current score state, estimate the order (and therefore the number of active partials).
- Sum all the active partials belonging to the set of notes to obtain the salience of the current score state.
- Compute the probability as the division of the salience of the current score state and the total salience of the frame (computed as the sum of all spectral peaks of the frame).

This computation must be done for each audio frame and for each state of the score, but the low complexity of NLS estimator and the used perceptual model allows to obtain the probabilities in a real-time execution.

In this way, a similarity matrix  $S$  that contains the probabilities for each state of the score (columns) and audio frame (rows) is given to the next block of the system in order to obtain a relation between score and performance in real time.

### 3. DYNAMIC TIME WARPING

In this section the similarity matrix post-processing is described. Dynamic Time Warping (DTW) algorithm is used in order to find the matching between MIDI frames and temporal frames. The similarity matrix is composed as described in section 2, and it is the input for the DWT block.

DTW is a kind of dynamic programming (DP) [15]. DP consists of two stages, forward and traceback. In the forward step, it calculates the lowest-cost path to all of the points neighbors plus the cost to get from the neighbor to the point - in this case,  $S_{max} - S(i, j)$ , where  $S_{max}$  is the largest value in the similarity matrix. This makes the cost for the most similar frame pairs be zero, and all other frame pairs have larger, positive costs. The object of this stage is to compute the cost of the best path to point  $(N1, M1)$  recursively by searching across all allowable predecessors to each point and accumulating cost from there. In the traceback stage, we find the actual path itself by recursively looking up the point providing the best antecedent for each point on the path.

Unconstrained DTW should allow the path to go horizontally and vertically [16], but for this application, this may be constrained. Horizontal paths would be a non planned (at the midi file) stop at the performance. This is possible but not very frequent. Vertical paths would be a infinite playing velocity, this is impossible so it should be constrained. Then, a change matrix is given to the DTW where a vertical or horizontal path is penalized with a high cost, while the diagonal has a low one.

In order to have a real time system, it is not possible to wait until the end of the file to select the best path along

the similarity matrix. Then a variation of it has been implemented. It takes sub-matrices along real time and it applies the DWT algorithm to them. These are increasing sub-matrices that contain information from the beginning time to the current time position.

The sub-matrices increase is of 1 second along time frames and the same along midi frames. This increasing is added from the last estimated samples at the previous DWT execution, so matrices grow up following the estimated sub-path.

At each iteration a partial path solution is achieved, from the origin to the actual time. However only a stretch is taken and added to the complete solution, this is from the last decision time up to 20% before now, because there is an information limitation because of the end of the matrix for the last samples, this will be solved at the next step where these samples will be estimated better.

Then, the complete solution of the path along the similarity matrix is composed with each stretch extracted from each sub-path calculated at every sub-matrix. This path is the estimation of which temporal frame match with which midi frame. At each second, the new stretch is available at the output. A minimum latency is present because of the 20% of samples that are not taken into account from the new sub-path, because of its latency, can be from 200ms to 1000ms.

The new output times are obtained by interpolating original times over the estimated path line.

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