### music structural segmentation by hmm clustering

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#### Abstract

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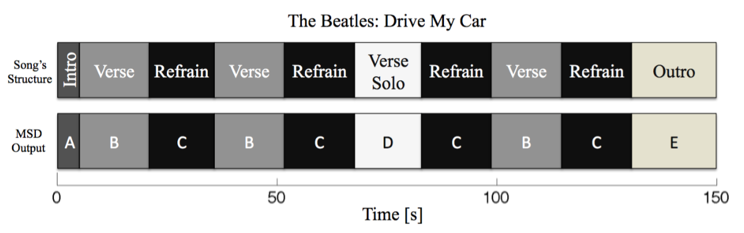
**Index Terms—**Audio, segmentation, structure, HMM, clustering

**1. Introduction**

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**1.1. Problem definition**

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**Figure 1**. Output example of the structural analysis of the song Drive My Car by the Beatles.

**1.2. Application**

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**1.3. Paper organization**

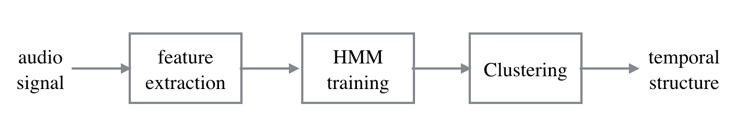
Algorithms for structural segmentation of music take an audio signal as input and give output information about its temporal structure. In our approach, a couple of subtasks can be identified. A rough overview of these subtasks is given in Figure 2.

**2. Related work**

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**3. algorithm overview**

Algorithms for structural segmentation of music take an audio signal as input and give output information about its temporal structure. In our approach, a couple of subtasks can be identified. A rough overview of these subtasks is given in Figure 2. This section will be organized accordingly.



**Figure 2**. System overview: music structural segmentation by clustering based on HMM.

**3.1. Low-level feature extraction**

Our method of low-level feature extraction is based on the audio spectrum envelope, audio spectrum projection and sound model as described in the MPEG-7 standard [3]. This step yields a 21-dimensional audio spectrum projection feature vectors. The first 20 dimensions represent the spectral shape, by reducing dimension of the original normalized power spectrum envelope (64-dimension) using PCA method. And the final dimension is the relative power of each block window, which will be discussed in section 3.1.3.

*3.1.1. Beat tracking*

The dataset audios are mixed from stereo to mono by taking the mean of two channels. The original sampling rate is 44100Hz which is not necessary to be that high, however we downsampled to 11025Hz.

We use a hop size equal to the beat-length of the music (typically 400-600ms), and a window of three times the hop. While beat-lengths are estimated by a beat-tracking algorithm developed by labrosa [4]. For accuracy, we use the highest estimation of tempo instead of the lowest estimation (typically half of the highest tempo).

*3.1.2. Audio spectrum envelope*

The underlying audio feature is the audio spectrum envelope. We extract audio spectrum envelope features with bands at 1/8th-octave spacing. We divide spectrum frequency domain into logarithmically spaced sub-bands between 62.5Hz and 16kHz. The use of logarithmic scaling is intended to imitate approximately the response of the human ear. The values of power spectrum are converted to a decibel scale.

We use constant Q transformation (CQT) to extract spectrum envelope.

*3.1.3. Audio spectrum projection*

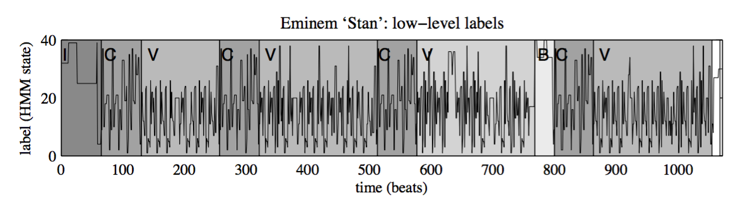
The spectrum for each block window is normalised by its L2-norm to represent the overall power. These norms themselves are then normalised by the max value, in order to give values in the range [0,1]. The normalized spectrum power vector for the entire track is appended to the 20-dimension spectrum projection matrix, as described in the next paragraph.

The dimensionality of the spectrum features for each window is reduced by applying Principal Component Analysis (PCA) to the entire sequence of features over the track. We retain the first 20 principal components, which form the first 20 dimension of our low-level feature matrix.

**3.2. HMM**

We feed the 21-dimensional low-level spectrum feature matrix mentioned above to train a Hidden Markov Model (HMM) with a fairly large number of states (80-state HMM).

We describe the trained model with a single Gaussian output distribution for each state, and a single covariance matrix tied across all states. We then Viterbi-decode the features using the trained model, to give the most likely sequence of state assignments for each beat of the music. Fig. 3 shows the resulting state sequence labels for the sample track.

**Figure 3**. Sequence of low-level labels for the sample track.

**3.3. Clustering**

We extract a corresponding sequence of low-level states using an M-state HMM, and estimate the local state distributions {xi} at each beat of the sequence by counting neighbouring states within a small histogram window.

We find the characteristic reference distributions for each segment-type by counting states over the relevant beats. We can then evaluate how well the manual segmentation is expressed by the reference distributions by treating them as cluster centroids in the space of local distributions.

**4. Evaluation**

We use Beatle’s data set…

**4.1. Conditional Entropies**

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**4.2. Pair-wise F-measure**

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**5. discussion**

Why our result is not good?

What may be the problem?

**6. conclusion**

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**7. References**

[1] A.B. Smith, C.D. Jones, and E.F. Roberts, “Article Title,” *Journal*, Publisher, Location, pp. 1-10, Date.

[2] Jones, C.D., A.B. Smith, and E.F. Roberts, *Book Title*, Publisher, Location, Date.

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[4] D. Ellis., “Beat Tracking by Dynamic Programming,” *J. New Music Research*, Special Issue on Beat and Tempo Extraction, vol. 36, no. 1, pp. 51-60, 2007.