SCORE-INFORMED SYLLABLE SEGMENTATION FOR A CAPPELLA SINGING VOICE WITH CONVOLUTIONAL NEURAL NETWORKS

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ABSTRACT

This paper introduces a new score-informed method for the segmentation of jingju a cappella singing phrase into syllables. The proposed method estimates the most likely sequence of syllable boundaries given the estimated syllable onset detection function (ODF) and its score. Throughout the paper, we first examine the jingju syllables structure and propose a definition of the term "syllable onset". Then, we identify which are the challenges that jingju a cappella singing poses. Further, we investigate how to improve the syllable ODF estimation with convolutional neural networks (CNNs). We propose a novel CNN architecture that allows to efficiently capture different timefrequency scales for estimating syllable onsets. Besides, we propose using a score-informed Viterbi algorithm instead of thresholding the onset function-, because the available musical knowledge we have (the score) can be used to inform the Viterbi algorithm to overcome the identified challenges. The proposed method outperforms the state-of-the-art in syllable segmentation for jingju a cappella singing. We further provide an analysis of the segmentation errors which points possible research directions.

1. INTRODUCTION

The ultimate goal of our research project is to automatically evaluate the jingju a cappella singing of a student in the scenario of jingju singing education – see Figure 1. Jingju, a traditional Chinese performing art form also known as Peking or Beijing opera, is extremely demanding in the clear pronunciation and accurate intonation for each syllabic or phonetic singing unit. To this end, during the initial learning stages, students are required to completely imitate tutor's singing. Therefore, the automatic jingju singing evaluation tool we envision is based on this training principle and measures the intonation and pronunciation similarities between the student's and the tutor's singings. Before measuring the similarities, the singing

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phrase should be automatically segmented into syllabic or phonetic units in order to capture the temporal details. In this paper we tackle the problem of score-informed automatic syllable segmentation for a cappella singing (bold rectangle in Figure 1). Jingju music scores, which contain the duration information for each singing syllable, will be a helpful hint for the segmentation method.



Figure 1. Framework of the entire research project. The module with bold border is addressed in this paper.

The syllable segmentation task consists of determining the time positions of the syllable boundaries – onset and offset. In this research, we consider the onset of the subsequent syllable to be the offset of the current one. Therefore, we treat the segmentation task as an onset detection problem.

Most segmentation methods rely on first estimating an ODF. For example, Klapuri [7] utilizes a band-wise processing principle inspired by psychoacoustics. He divides the audio signal into 21 non-overlapping bands, then detects onset components in each band ODF and finally combines them to yield onsets. Obin *et al.* [10] obtain a syllable ODF by fusing mel-frequency intensity profiles and voicing profiles. One shortcoming of these band-wise methods has been already pointed out by Klapuri [7]: "they are unable to deal with strong amplitude modulations" – what is very common in singing voice recordings.

On the other hand, some methods are based on features and supervised learning. For example, Toh *et al.* [17] use two Gaussian Mixture Models (GMMs) to classify singing audio frames into onset or non-onset classes. Three timbral features (MFCCs, LPCCs and ERB-bands) are chosen as input. Their results show that this supervised method is superior to many band-wise ODF-based methods.

Neural networks have also been successfully explored for the task of musical onset detection. Eyben *et al.* [4] trained a bidirectional long-short term memory neural network on mel-scale magnitude spectrograms to estimate an ODF. Schlüter *et al.* [16] proposed to use CNNs to estimate the ODF, which defines the current state-of-the-art for onset detection. The advantage of applying deep learning methods in musical onset detection is that one does not

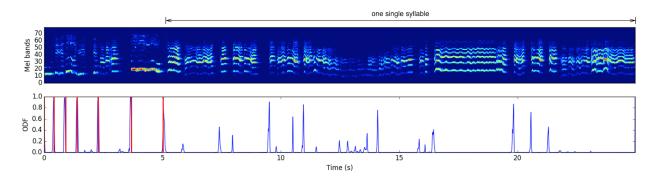


Figure 2. A *dan* role-type singing phrase with its last syllable prolongated. **Top:** log-mel bands spectrogram. **Bottom:** syllable ODF estimated with Schlüter's method [16] (blue), and ground truth syllable onsets (red).

need to design rules or handcraft features to capture the relevant facets for detecting onset/non-onset frames – which are very difficult to design. Besides, if context (more than one frame) is input into the network, spectro-temporal features can also be learned – what might be useful to learn slow-transient features defining some onsets.

Thresholding or peak-picking operations are normally executed over the ODF in order to determine the final onsets [4, 7, 16, 17]. However, we will argue that these operations are not suitable for selecting jingju singing syllable onsets. Probabilistic models -which allow incorporating prior domain knowledge for decision making- usually result in a performance gain compared to simple thresholding and peak-picking methods. For example, Böck et al. [1] tracked the beat/downbeat by using a dynamic bayesian network (DBN) observing a beat ODF estimated by a recurrent neural network (RNN). Or Obin et al. [10] applied a segmental Viterbi algorithm over a syllable ODF to detect speech syllable onsets. These two approaches are closely related to ours: former one relates with our work because it uses deep learning to estimate the ODF, and latter one because we also consider using the Viterbi algorithm for estimating the onset sequence.

This paper introduces a new score-informed method for the segmentation of jingju singing into syllables. We first define what a "syllable" and "syllable onset" is in the context of jingju music. By doing so, in section 2 we introduce the challenges we aim to address. The audio dataset we use is described in section 3. Section 4 explains the CNN architecture used for estimating the syllable ODF, and the Viterbi decoding algorithm that exploits the prior syllable duration information extracted from the score. Evaluation and error analysis are conducted in section 5, and section 6 concludes this work.

2. BACKGROUND AND CHALLENGES

Jingju singing is the most precise articulated rendition of the spoken Mandarin language. Although certain special pronunciations in jingju theatrical language differ from their normal Mandarin pronunciations—due to: firstly, the adoption of certain regional dialects; and secondly, the ease or variety in pronunciation and projection of sound—the mono-syllabic pronouncing structure of the standard Mandarin doesn't change [18].

A syllable/character of jingju singing is composed of three distinct parts in most of the cases: the "head" (tou), the "belly" (fu) and the "tail" (wei). The "head" consists of an initial consonant or semi-vowel - and a medial vowel, if the syllable includes one. The "head" is not normally prolonged in its pronunciation except when there is a medial vowel. The "belly" follows the "head" and consists of the central vowel and it is normally prolonged. The "belly" is the most sonorous part of a jingju singing syllable and can be analogous to the nuclei of a speech syllable. The "tail" is composed of the terminal vowel or consonant [18]. However, there are syllables in jingju singing where "head" is absent - only "belly" is a necessary element. To avoid ambiguity, we define the syllable onset as "the start of the initial consonant if the syllable includes one or the start of the central vowel otherwise". This definition is slightly different from that agreed by most of the phonological theories [5]: "any consonants that precede the nuclear element (the vowel)". It is also worth stressing that our notion of singing syllable onset is different from the singing onset defined in Toh et al.'s paper [17]: "the start of a new human-perceived note, taking into account contextual cues", of which the latter emphasizes on the intonational aspect instead of the phonological aspect of the singing voice.

2.1 Challenges

Figure 2 shows an example of a *dan* role-type singing phrase in which the last syllable lasts approximately 20s. This singing method is more common in *dan* singing than in *laosheng* singing. However, both role-types use it as a way of improving artistic expression and showing off their singing skills. These skills include breathing techniques, intonational techniques and dynamic control techniques, among others. The syllable ODF (blue curve in Figure 2) is generated using a CNN model based on Schlüter *et al.*'s work [16], which is considered the state-of-the-art. This CNN model is trained with the jingju dataset presented in section 3. The resulting syllable ODF is quite robust to pitch variations and continuous dynamic change since no prominent peaks are observed in these regions. However,

many peaks can be found in the start and end positions of each articulation throughout this long syllable (≈ 20 s). Note that thresholding or peak-picking would choose these peaks (false syllable onsets) since they have similar amplitude level than real onsets. For that reason, we propose using a score-informed decoding method as an alternative to naive thresholding.

Stealing breath ($tou\ qi$) is one of the primary methods of taking the breath in jingju singing. Breath is stolen when a sound is too long to be delivered in one breath – and no vocal pauses are desired [18]. However, this is not the only technique which can lead to pauses within a syllable. Another singing technique ($zu\ yin$ – literal translation: block sound), provokes also pauses without occurring exhalation or inhalation. This kind of pause can be very short in duration and can be easily found in jingju singing syllables, see Figure 3 for example. These silences can be another source of false positives if thresholding or peak-picking is used. However, these can also be avoided by using a scoreinformed decoding method.

A priori syllable duration information is often easy to obtain from the score and this is an advantage which we exploit to avoid previously described issues. The repertoire of jingju includes around 1400 plays [18], and most used teaching pieces are transcribed into scores. We use the score to guide the Viterbi decoding throughout the onset detection process.

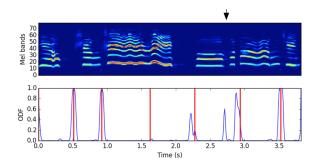


Figure 3. Another *dan* role-type singing phrase. A *zu yin* (block sound) is located by the arrow within a normal length syllable. Schlüter's syllable ODF (blue curve). Ground truth syllable onset positions (red vertical lines).

3. DATASET

The *a cappella* singing audio dataset ¹ used for this study focuses on two most important jingju role-types [14]: *dan* (female) and *laosheng* (old man). It contains 39 interpretations of 31 unique arias sung by 11 jingju singers. Audio is sampled at 44.1 kHz and is is pre-segmented into phrases. The syllable onset ground truth is manually annotated in Praat [2] – with 291 phrases and 2641 syllables (including padding written-characters [18]). Two Mandarin native speakers and a jingju musicologist have been devoted to this annotation work. Syllable durations are manually transcribed from music scores considering as unit duration

the quarter note. The whole dataset is randomly split into training, validation and test sets (60%, 20% and 20%, respectively). The percentage of presence of each role-type in a split is kept constant throughout sets.

In order to highlight the differences between jingju music and popular western music, we compare the vowel's duration between Kruspe's dataset [8] (a cappella singing of commercial pop songs) and ours. In Kruspe's dataset, the standard deviation duration of voiced phonemes is of 0.31s, whereas this duration is more than doubled in our dataset: 0.75s. Since voiced phonemes are the main component of a syllable, it thus becomes clear that jingju singing syllable durations show huge variations. Therefore, it is impossible to model syllable durations with a single distribution. For that reason the proposed model must be able to handle different syllable lengths depending on the case. To this end, syllable durations extracted from scores will be a valuable information.

4. APPROACH

A score-informed syllable boundary detection approach is explained in this section. We first propose some improvements to the current state-of-the-art CNN onset detection model proposed by Schlüter *et al.* The syllable ODF output from the CNN serves as observation probability for the syllable boundary decoding process. Then, an *a priori* syllable duration model based on the score is proposed. This model guides the Viterbi algorithm by informing it in which time-positions is likely to occur a syllable boundary. Therefore, the syllable boundaries sequence is decoded by taking advantage of a score-informed Viterbi algorithm.

4.1 CNN syllable onset detection function

Studied CNNs are inspired by Schlüter *et al.*'s [16] and Pons *et al.*'s work [11, 12]. Schlüter *et al.* have shown that CNNs fed with spectrograms can achieve state-of-the-art performance for the onset detection task [16]. On the other hand, Pons *et al.* [11] have recently proposed a novel design strategy for spectrograms-based CNNs. They propose using different filter shapes in the first layer so that local stationarities in spectrograms (present at different time/frequency-scales) can be efficiently captured.

Explored models are based on Schlüter *et al.*'s architecture [16], which consists of two convolutional layers, and a dense layer of 256 units connected to an output softmax layer – with two output units standing for onset and non-onset. First CNN layer uses $20 \times 3 \times 7$ filters 2 and is followed by a 3×1 max pool layer. Second CNN layer has $20 \times 3 \times 3$ filters and is followed by a 3×1 max-pool layer. Input is set to be a log-mel spectrogram 3 of size 80×21 – note that the network takes a decision for every frame given its context: ± 10 frames, 21 frames in total.

¹ https://goo.gl/y0P7BL

² First number denote the number of filters (*ie.* 20x). Second and third respectively denote the frequency and temporal size of the filter (*ie.* 3×7).

³ Original Schlüter et al.'s model inputs three channels with different resolution spectrograms to the network. However, preliminary results showed that a single channel was performing better than three.

Proposed architectures follow Pons et al. [11,12] design strategy and several filter shapes in the first layer are used. This approach allows to efficiently capture different timefrequency scales, what might be interesting for the task at hand because onsets can exhibit different time-frequency patterns. For example, Figure 4 depicts two onsets: middle onset is expressed as an abrupt time-frequency change corresponding to an onset starting with a consonant, and right onset is expressed as a gradual timbral change that corresponds to an onset starting with a central vowel. Note that these examples are representative of the proposed definition of syllable onset. Moreover, using different filter shapes in the first layer promotes a much richer representation out of the first layer. In the following, several CNNs are designed to efficiently capture the relevant timefrequency contexts and scales for onset detection.

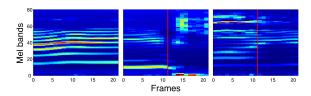


Figure 4. Log-mel spectrograms (80 bands, 21 frames). *left*: Non-onset spectrogram. *middle* and *right*: Onset spectrograms. Red line denotes the onset frame.

Schlüter *et al.*'s architecture and proposed ones only differ in the way the first convolutional layer and its following max-pool layer are set up. Input and remaining layers are kept intact – unless it is explicitly stated:

→ Temporal architecture. Note that the small filters proposed by Schlüter *et al.* might have difficulties on learning the longer time-scales required to model the gradual changes that some onsets express – see Figure 4, *right*. Several filter shapes are designed to efficiently capture different and longer time-scales [11] than Schlüter *et al.*:

- \cdot 12x 1×7, 6x 3×7 and 3x 5×7
- \cdot 12x 1×12, 6x 3×12 and 3x 5×12

These CNNs are followed by a 3×5 max-pool layer.

- → <u>Timbral architecture</u>. Since our syllable onset definition considers phonological aspects, we consider the timbre to be an important feature for our task. Filter shapes are designed to learn timbral representations, note that these filters span throughout the frequency domain [12]:
 - \cdot 12x 50×1, 6x 50×5 and 3x 50×10
 - \cdot 12x 70×1, 6x 70×5 and 3x 70×10

These CNNs are followed by a 5×3 max-pool layer.

We also explore combining temporal and timbral architectures. We propose a late-fusion approach where we multiply the estimated output probabilities from temporal and timbral (independently trained) architectures ⁴.

In addition, we also explore increasing the number of filters in the second layer from 20x to 32x.

For concatenating several feature maps resulting of CNNs with different filter shapes, it is required to use zero padding. We apply same padding in the first CNN layer, so that all resulting feature maps have the same length and these can be concatenated. STFT was performed using a 25ms window (2048 samples with zero-padding) with a hop size of 10ms. The 80 log-mel bands energies are calculated on frequencies between 0Hz and 11000Hz and spectrograms are standardized to have zero mean and unit variance. We use L2 weight decay regularization, ELU activation functions [3] and 30% dropout for each layer. The model parameters are learned with mini-batch training (batch size 128) using the ADAM update rule [6] and early stopping - if validation loss (categorical cross-entropy) was not decreasing after 10 epochs. In order to allow a fair comparison between Schlüter's and Pons' architectures, the original Schlüter architecture is modified to meet above described hyper-parameters.

Finally, syllable ODFs estimated with CNNs are smoothed by convolving those with a 5 frames Hanning window [16] – since estimated syllable ODFs are typically very spiky.

4.2 A priori duration model

The *a priori* duration model is shaped with a Gaussian function $\mathcal{N}(x; \mu_l, \sigma_l^2)$ whose mean μ_l represents the l-th relative syllable duration – according to the score. Its standard deviation σ_l is proportional to μ_l : $\sigma_l = \gamma \mu_l$ and γ is heuristically set to 0.35.

$$\mathcal{N}(x; \mu_l, \sigma_l^2) = \frac{1}{\sqrt{2\pi}\sigma_l} \exp\left(-\frac{(x - \mu_l)^2}{2\sigma_l^2}\right). \quad (1)$$

Figure 5 provides an intuitive example of how the *a priori* duration model works. Observe that it provides the prior likelihood of an onset to occur according to the duration in the score.

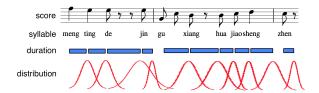


Figure 5. *A priori* relative duration distributions (bottom) of the syllables of a singing phrase.

The relative duration of each note is measured considering a quarter note length as a unit, so an eighth note has a duration of 0.5. We only keep the relative duration and discard the tempo information of the score. By normalizing the summation of the notes' relative durations to the incoming audio recording's duration, we obtain the absolute notes' durations. Then, a sequence of syllable absolute durations $M=\mu_1\mu_2\cdots\mu_L$ is deduced by summing the notes' absolute durations corresponded with each syllable, where L is the total syllable number of the score. The a priori duration model distributions will be incorporated into the Viterbi algorithm as state transition probabilities to inform the algorithm where syllable boundary is likely to occur.

⁴ Using temporal and timbral filters in first layer yield to worse results.

4.3 Decoding of the syllable boundaries

To decode the syllable boundaries, we construct a hidden markov model characterized by the following:

- 1. The hidden state space is a set of N candidate onset positions S_1, S_2, \dots, S_N discretized by the hop size, where S_N is the offset position of the last syllable.
- 2. The state transition probability at the time instant l associated with state changes is defined by a priori duration distribution $\mathcal{N}(d_{ij}; \mu_l, \sigma_l^2)$, where d_{ij} is the time distance between states S_i and S_j (j > i). The length of the decoded state sequence is equal to the total syllable number L written in the score.
- 3. The observation probability for the state S_j is represented by its corresponding value in the syllable ODF p, which is denoted as p_j .

The goal is to find the best onset position state sequence $Q=q_1q_2\cdots q_L$ for a given a priori duration sequence M, where q_i denotes the onset of the i+1th decoding syllable. q_0 and q_L are fixed as S_1 and S_N as we expect that the onset of the first syllable to be located at the beginning of the incoming audio and the offset of the last syllable is located at the ending of the audio. One can fulfill this assumption by truncating the silences at the beginning and at the end of the incoming audio. According to the logarithmic form of Viterbi algorithm [13], we define:

$$\delta_l(i) = \max_{q_1, q_2, \cdots, q_l} \log P[q_1 q_2 \cdots q_l, \, \mu_1 \mu_2 \cdots \mu_l]$$

with the initial step as follows:

$$\delta_1(i) = \log(\mathcal{N}(d_{1i}; \mu_1, \sigma_1^2)) + \log(p_i)$$

$$\psi_1(i) = S_1$$

with the following recursive step:

$$\delta_l(j) = \max_{1 \leq i < j} [\delta_{l-1}(i) + \log(\mathcal{N}(d_{ij}; \mu_l, \sigma_l^2))] + \log(p_j)$$

$$\psi_l(j) = \arg\max_{1 \leq i < j} [\delta_{l-1}(i) + \log(\mathcal{N}(d_{ij}; \mu_l, \sigma_l^2))]$$

and the following termination step:

$$\log P^* = \max_{1 \le i < N} [\delta_{L-1}(i) + \log(\mathcal{N}(d_{iN}; \mu_L, \sigma_L^2))]$$

$$q_L^* = \arg \max_{1 \le i < N} [\delta_{L-1}(i) + \log(\mathcal{N}(d_{iN}; \mu_L, \sigma_L^2))]$$

Finally, the best offset position state sequence Q is obtained by the backtracking step. An example of the best boundary position decoding path is showed in figure 6.

5. EXPERIMENTS AND RESULTS

5.1 Performance metrics

The syllable segmentation task consists on determining the time positions of syllable boundaries. The proposed evaluation consists in comparing the detected syllable onsets/offsets to their reference ones. We report F-measure results in this paper. The definition of a correct segmented

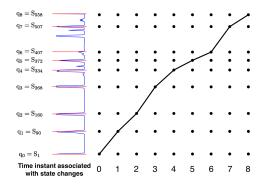


Figure 6. Illustration of an example of the best boundary position decoding path (black) with N=558 and L=8. The intermediate states are omitted in the vertical direction for a clear visualization. Blue curve: syllable ODF; red horizontal lines: decoded boundary positions.

syllable is borrowed from the note transcription literature [9]. For syllable onsets, we choose an evaluation tolerance of $\pm \tau$ ms. For offsets, we choose an evaluation tolerance of either (a) $\pm 20\%$ of the syllable's duration annotation, or (b) $\pm \tau$ ms – whichever is larger. If both the onset and the offset of a syllable lie within the tolerance of their annotated counterparts and the syllable is correctly labeled, we consider that it's correctly segmented. We report the results for a tolerance of $\tau=0.05$ (seconds).

5.2 Results and discussion

Two state-of-the-art methods are set as baselines: Obin *et al.* [10] as a traditional approach, and Schlüter *et al.* [16] as a deep learning method – both already introduced. We explore Pons *et al.*'s CNNs design strategy [11, 12] as a way to improve our results. Code is available online ⁵. All syllable ODFs are decoded by using the same approach (described in section 4.3). F-measure results are reported in Table 5.2. The evaluation can not be performed by using peak-picking because the syllabic label can not be attached. However, in section 2 we already discussed the potential problems of peak-picking which are explicitly addressed with the proposed score-informed method.

Results show that proposed architectures improve the state-of-the-art in all cases. These results support the idea that using different filter shapes in the first CNN layer is beneficial. Moreover, observe that #params is reduced at least to the half – interpret #params (number of parameters of the CNN) as a measure of the representational capacity of the network. This denotes how efficient can be CNNs if these are designed to capture the relevant features for the task at hand. Also note that if the #params of a network is reduced, the over-fitting risk is reduced as well.

We also observe that late-fusing the predictions helps to improve the results. The interesting idea behind this approach is that individual networks still need to be able to solve the task by their means – and we propose fusing

⁵ https://github.com/ronggong/jingjuSyllabicSegmentaion/tree/v0.1.0

Syllable ODFs	#params	F-measure (%)
Late-fusion (32)	-	86.37
Temporal (32)	210,403	83.28
Timbral (32)	185,420	84.83
Late-fusion (20)	-	84.05
Temporal (20)	132,127	83.86
Timbral (20)	119,432	82.89
Schlüter et al. [16]	535,290	81.93
Obin <i>et al</i> . [10]	-	40.61

Table 1. Syllable segmentation results using different methods for estimating the syllable ODF. Numbers in parenthesis indicate the #filters in the second CNN layer.

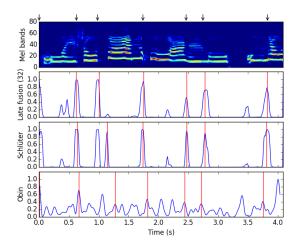


Figure 7. Three syllable ODFs. Red vertical line: decoded syllable onset time positions. Black arrows: ground truth.

models that are tailored towards learning complementary time-frequency scales: temporal and timbral architectures.

We also see a significant performance gap between Obin *et al.* and CNN-based methods. For example in Figure 7 numerous noisy peaks can be observed for Obin *et al.*'s method, what is significantly decreasing its performance. This result denotes the difficulty of designing rules or handcrafting features for estimating the syllable ODF.

5.3 Error analysis

We conduct an error analysis to study the causes of the segmentation errors produced by our best performing method, what points out future research directions for further improvement. We study falsely detected onsets in the test dataset by comparatively observing the detected syllable onset positions, the ground truth positions, the spectrogram and the corresponding scores. Errors out of 0.05s tolerance are analyzed and 70 syllable onsets out of 512 are identified as false detections. Errors are then classified according to their causes. Two main error classes are found: scoreperformance mismatching and ambiguous syllable transitions.

The majority of the errors (71.42%) are caused by per-

formance deviations from the score: syllable insertion, syllable deletion or performing different syllable durations than the score. The proposed syllable boundary decoding algorithm is not able to preserve the correctness when the difference between the score and its performance becomes too large. Furthermore, since the length of the decoded state sequence of the algorithm must be equal to the number of syllables in the score, singing syllable insertions and deletions not expressed in the score can not be handled properly. One possible solution is to incorporate more domain knowledge of jingju singing into the decoding process – such as considering the probability of a padding-character to be inserted/deleted, or the expectation of prolonging the last syllable in a phrase.

The second source of errors include ambiguous syllable transitions (15.61%) – such as transitions from vowel to vowel, from vowel or to semi-vowel, etc. These errors are very difficult to be corrected because no prominent spectral changes can be discerned within such transitions [15]. In future investigations we shall detect "onset regions" rather than "onset time positions" since this kind of syllable transitions usually manifest themselves as gradual spectral changes.

6. CONCLUSIONS

This paper introduces a new score-informed method for the segmentation of jingju singing into syllables. Two main contributions are presented in this paper: (i) improvements for estimating the syllable ODF with CNNs, and (ii) a method for incorporating score information into Viterbi's algorithm for estimating syllable boundaries.

The improvements to the CNN architecture consisted of using different filter shapes in the first layer and latefusing the predictions of two models – designed to learn complementary representations (temporal and timbral). By doing so, we increased the expressiveness of the first layer and enabled the networks to efficiently capture different time-frequency scales useful for detecting syllable onsets. Proposed models, with many filter shapes in the first layer, has proven to be more effective than the state-of-the-art model based on a single filter shape in the first layer.

Moreover, we proposed an *a priori* duration model that describes the probability of a syllable boundary given the score. The likelihood of a syllable boundary is shaped with the *a priori* duration model and incorporated into Viterbi's algorithm as state transition probabilities – this being the core of the proposed score-informed Viterbi algorithm.

We validated the proposed method on a jingju a cappella singing dataset, which achieved better performance than the state-of-the-art. Although the proposed ameliorations helped to improve our results, the proposed method has not yet solved the task. To this end, we plan to investigate incorporating more domain knowledge into the decoding process, and to further improve the ODF with RNNs – that has proven to be very useful for similar tasks [1,4].

7. ACKNOWLEDGMENTS

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8. REFERENCES

- [1] Sebastian Böck, Florian Krebs, and Gerhard Widmer. Joint beat and downbeat tracking with recurrent neural networks. In *ISMIR*, New York City, USA, 2016.
- [2] Paul Boersma. Praat, a system for doing phonetics by computer. *Glot International*, 5(9/10):341–345, 2001.
- [3] Djork-Arné Clevert, Thomas Unterthiner, and Sepp Hochreiter. Fast and accurate deep network learning by exponential linear units (elus). *CoRR*, abs/1511.07289, 2015.
- [4] Florian Eyben, Sebastian Böck, Björn W Schuller, and Alex Graves. Universal onset detection with bidirectional long short-term memory neural networks. In *IS-MIR*, Utrecht, Netherlands, 2010.
- [5] Brett Kessler and Rebecca Treiman. Syllable structure and the distribution of phonemes in english syllables. *Journal of Memory and language*, 37(3):295–311, 1997.
- [6] D. Kingma and J. Ba. Adam: A method for stochastic optimization. *arXiv:1412.6980*, 2014.
- [7] A. Klapuri. Sound onset detection by applying psychoacoustic knowledge. In *ICASSP*, Phoenix, USA, 1999.
- [8] Anna M. Kruspe. Keyword spotting in a-capella singing. In *ISMIR*, Taipei, Taiwan, 2014.
- [9] Emilio Molina, Ana M. Barbancho, Lorenzo J. Tardn, and Isabel Barbancho. Evaluation Framework for Automatic Singing Transcription. In *ISMIR*, Taipei, Taiwan, 2014.
- [10] N. Obin, F. Lamare, and A. Roebel. Syll-O-Matic: An adaptive time-frequency representation for the automatic segmentation of speech into syllables. In *ICASSP*, Vancouver, Canada, 2013.
- [11] Jordi Pons and Xavier Serra. Designing efficient architectures for modeling temporal features with convolutional neural networks. In *ICASSP*, New orleans, USA, 2017.
- [12] Jordi Pons, Olga Slizovskaia, Rong Gong, Emilia Gómez, and Xavier Serra. Timbre analysis of music audio signals with convolutional neural networks. arxiv:1703.06697, 2017.

- [13] L. Rabiner. A tutorial on hidden Markov models and selected applications in speech recognition. *Proceedings of the IEEE*, 77(2):257–286, February 1989.
- [14] Rafael Caro Repetto and Xavier Serra. Creating a Corpus of Jingju (Beijing Opera) Music and Possibilities for Melodic Analysis. In *ISMIR*, Taipei, Taiwan, 2014.
- [15] Odette Scharenborg, Vincent Wan, and Mirjam Ernestus. Unsupervised speech segmentation: An analysis of the hypothesized phone boundaries. *The Journal of the Acoustical Society of America*, 127(2):1084–1095, 2010.
- [16] J. Schlüter and S. Böck. Improved musical onset detection with convolutional neural networks. In *ICASSP*, Florence, Italy, 2014.
- [17] Chee-Chuan Toh, Bingjun Zhang, and Ye Wang. Multiple-feature fusion based onset detection for solo singing voice. In *ISMIR*, Philadelphia, USA, 2008.
- [18] Elizabeth Wichmann. *Listening to Theatre: The Au*ral Dimension of Beijing Opera. University of Hawaii Press, 1991.