

LEARNING INVARIANTS FOR POLYPHONIC INSTRUMENT RECOGNITION

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ABSTRACT

The abstract should be placed at the top left column and should contain about 150-200 words.

To achieve invariance to translation as well as frequency transposition, we pool neighboring units in the time-frequency domain (t, k_1) over non-overlapping rectangles of width Δt and height Δk_1 .

1. INTRODUCTION

2. DEEP CONVOLUTIONAL NETWORKS

2.1 Time-frequency representation

We used the implementation from the librosa package [4] with $Q = 12$ filters per octave, center frequencies ranging from 55 Hz to 14 kHz (8 octaves from A1 to A9), and a hop size of 23 ms. Furthermore, we applied perceptual weighting of loudness in order to reduce the dynamic range between the fundamental partial and its upper harmonics. A 3-second sound excerpt $x(t)$ is represented by a time-frequency matrix $\mathbf{x}_1(t, k_1)$ of width $T = 128$ samples and height $K_1 = 96$ MIDI indices.

2.2 Architecture

First of all, we apply a family $\mathbf{W}_2(\tau, \kappa_1, k_2)$ of $K_2 = 50$ learned time-frequency convolutional operators, whose supports are constrained to have width Δt and height Δk_1 .

$$\mathbf{W}_2^{t, k_1} * \mathbf{x}_1 = \sum_{\substack{0 \leq \tau < \Delta t \\ 0 \leq \kappa_1 < \Delta k_1}} \mathbf{W}_2(\tau, \kappa_1, k_2) \mathbf{x}_1(t - \tau, k_1 - \kappa_1) \quad (1)$$

Furthermore, element-wise biases $\mathbf{b}_2(k_2)$ are added to the convolutions, resulting in the tensor

$$\mathbf{y}_2(t, k_1, k_2) = \mathbf{b}_2 + (\mathbf{x}_1^{t, k_1} * \mathbf{W}_2). \quad (2)$$

The second step is the application of a pointwise non-linearity. We have chosen the *rectified linear unit* (ReLU) because of its popularity in computer vision and its computational efficiency.

$$\mathbf{y}_2^+(t, k_1, k_2) = \max(\mathbf{y}_2(t, k_1, k_2), 0) \quad (3)$$

$$\mathbf{x}_2(t, k_1, k_2) = \max_{\substack{0 \leq \tau < \Delta t \\ 0 \leq \kappa_1 < \Delta k_1}} \left\{ \mathbf{y}_2^+(t + \tau, k_1 + \kappa_1, k_2) \right\} \quad (4)$$

$$\mathbf{y}_3(t, k_1, k_3) = \sum_{k_2} (\mathbf{x}_2^{t, k_1} * \mathbf{W}_3) \quad (5)$$

$$\mathbf{x}_4(k_4) = \left(\sum_{v_3} \mathbf{W}_4(k_4, v_3) \mathbf{x}_3(v_3) \right)^+ \quad (6)$$

$$\mathbf{x}_5(k_5) = \left(\sum_{k_4} \mathbf{W}_5(k_5, k_4) \mathbf{x}_4(k_4) \right)^+ \quad (7)$$

$$\mathbf{y}_6(k_6) = \sum_{k_5} \mathbf{W}_6(k_6, k_5) \mathbf{x}_5(k_5) \quad (8)$$

We define the categorical cross-entropy as

$$\mathcal{L}(\mathbf{x}_6, \mathcal{I}) = - \sum_{k_5 \in \mathcal{I}} \log \sigma(\mathbf{y}_6(k_5)). \quad (9)$$

The goal is to minimize the average loss $\mathcal{L}(\mathbf{x}_6, \mathcal{I})$ for across all pairs $(\mathbf{x}_6, \mathcal{I})$ in the training set.

2.3 Training

The network is trained on categorical cross-entropy with *Adam* [3], a state-of-the-art online optimizer for gradient-based learning. The loss function \mathcal{L} is minimized by shuffled mini-batches of size 512 with uniform class distribution.

3. DEEP SUPERVISION OF MELODIC CONTOUR

3.1 Disentangling pitch from timbre

3.2 Extraneous supervision

$$\mathcal{L}(\mathbf{x}_2, \mathcal{P}) = - \sum_{(t, k_1) \in \mathcal{P}} \log \sigma \left(\sum_{k_2} \mathbf{x}_2(t, k_1, k_2) \right) \quad (10)$$



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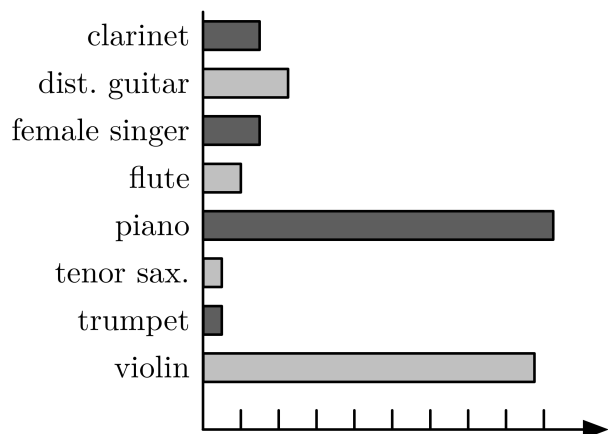


Figure 1: Amount of training data per instrument in MedleyDB, in minutes.

3.3 Joint supervision

4. SINGLE-INSTRUMENT CLASSIFICATION

4.1 Experimental design

In order to train the proposed algorithms, we used MedleyDB v1.1. [1], a dataset of 122 multitracks annotated with instrument activations as well as melodic f_0 curves when present. We extracted the monophonic stems corresponding to a selection of eight pitched instruments (see Figure 1. Stems with leaking instruments in the background were discarded. The resulting set was double-checked manually, and annotation mistakes were reported to MedleyDB curators for the next release. The evaluation set consists of 120 recordings of solo music collected by Joder et al. [2]. We discarded recordings with extended instrumental techniques, since they are under-represented in MedleyDB. Moreover, since the

4.2 Results

5. POLYPHONIC CLASSIFICATION

5.1 Experimental design

5.2 Results

6. CONCLUSIONS

7. REFERENCES

- [1] Rachel Bittner, Justin Salamon, Mike Tierney, Matthias Mauch, Chris Cannam, and Juan Bello. Medleydb: a multitrack dataset for annotation-intensive mir research. *International Society for Music Information Retrieval Conference*, 2014.
- [2] Cyril Joder, Slim Essid, and Gaël Richard. Temporal integration for audio classification with application to musical instrument classification. *IEEE Transactions on Audio, Speech and Language Processing*, 17(1):174–186, 2009.
- [3] Diederik P. Kingma and Jimmy Lei Ba. Adam: a Method for Stochastic Optimization. *International Conference on Learning Representations*, pages 1–13, 2015.
- [4] Brian McFee, Matt McVicar, Colin Raffel, Dawen Liang, Oriol Nieto, Eric Battenberg, Josh Moore, Dan Ellis, Ryuichi Yamamoto, Rachel Bittner, Douglas Repetto, Petr Viktorin, Joo Felipe Santos, and Adrian Holovaty. librosa: 0.4.1. zenodo. 10.5281/zenodo.18369, October 2015.