Doctoral Thesis

Zero-Shot Recognition of Generic Objects

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Declaration

I herewith declare that I have produced this paper without the prohibited assistance of third parties and without making use of aids other than those specified; notions taken over directly or indirectly from other sources have been identified as such. This paper has not previously been presented in identical or similar form to any other german or foreign examination board.

The related contents in this thesis have been previously published or submitted for publication by the author. A complete list of publications can be found on *pp.* ix-xv.

Publication List

Journal Papers

1. Ryo Aihara, Ryoichi Takashima, Tetsuya Takiguchi and Yasuo Ariki: "Noise-Robust Voice Conversion Based on Sparse Spectral Mapping Using Non-negative Matrix Factorization", *IEICE Transactions on Information and Systems*, Vol.E97-D, No.6, pp.1411-1418, 2014.

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7. Ryo Aihara, Tetsuya Takiguchi, and Yasuo Ariki: “ACTIVITY-MAPPING NON-NEGATIVE MATRIX FACTORIZATION FOR EXEMPLAR-BASED VOICE CONVERSION”, *ICASSP 2015*, pp. 4899-4903, 2015.

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Glossary

ZSL	Zero-Shot Learning
CNN	Convolutional Neural Network
NLP	Natural Language Processing
GCN	Graph Convolution Network

Chapter 1

Introduction

1.1 Background

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1.4 Outline

1. INTRODUCTION

Chapter 2

Visual Feature Extraction

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2.1.1 Motivation

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The related publications for this chapter are [].

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Chapter 4

Visual Feature Extraction

The related publications for this chapter are [].

4.1 The Motivation and Related Work

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Chapter 5

Conclusions

In []

5. CONCLUSIONS

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Appendix

some section

5. APPENDIX

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