JUSTUS-LIEBIG-UNIVERSITÄT GIESSEN

GGS Gießener Graduiertenzentrum Sozial-, Wirtschafts- und Rechtswissenschaften

Workshop

# Accessing spoken data using automatic speech recognition

## Organizational details

| | |
|---|---|
| Instructor: | *Jan Gorisch, Dr. phil.* |
| Date: | July 2, 2024, 2:00 pm – 6:00 pm |
| Location: | Gustav-Krüger-Saal, Ludwigstr. 23, 35390 Giessen |
| Max. number of participants: | 20 |

## Objectives

In this workshop we tackle the transcription bottleneck that many research projects face once the recordings are done. We evaluate whether – and for which type of data – automatic speech recognition (ASR/Speech-to-text) can be part of the solution to this problem. At the end of the workshop, participants will be able to run ASR on their local machines on their own recordings and will have an informed idea of how speech recognition works and what current state-of-the-art speech technology is capable to do.

## Content & Methods

The focus of this workshop are transcriptions of audio/video recordings. We will explore state-of-the-art speech technology, namely automatic speech recognition, in order to produce initial orthographic transcripts. We will investigate how these transcripts can be imported and further processed in sophisticated editing tools and evaluate the ASR-output. We will also touch on other ASR- and transcript-related topics, such as data protection, transcription conventions, annotations, metadata, query, analysis and data dissemination.

Approximately 50% of the workshop will be hands-on activities.

Participants are asked to bring along a laptop and a speech recording of up to 10 minutes, preferably in *.mp3, *.wav or *.mp4 format.

Participants are further asked to install software on their laptops prior to the workshop:

- **aTrain** (if you happen to have a windows-machine; for installation, cf. the Microsoft Store and search for “aTrain”.)
- **faster whisper**; for installation, cf.: https://github.com/Purfview/whisper-standalone-win/releases

As the ASR-models, which will be downloaded automatically with the first call of the transcription method, can be quite large, it is necessary that the laptop has some (up to 15 GB) free storage space.

If you have trouble installing one or the other software, get in touch with Jan Gorisch at gorisch@ids-mannheim.de.

## Target group & Course Language

Anyone interested in how to get from recordings to initial orthographic transcripts automatically is invited to participate.

No previous knowledge is necessary.

Course language: English

## About the instructor

Jan Gorisch is working at the Archive of Spoken German (AGD) at the Leibniz-Institute for the German Language (IDS) in Mannheim since 2015. In the program area oral corpora, he develops corpus curation pipelines and explores and evaluates the potential of automatic speech recognition and other speech technology for these processes. His research interests include phonetics, the prosody-gesture interface, interactional linguistics, computational linguistics, signal processing.

## Registration

By **June 22, 2024** via e-mail to info@ggs.uni-giessen.de.