

# Nkululeko 1.0: A Python package to predict speaker characteristics with a high-level interface

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DOI: [10.xxxxxx/draft](https://doi.org/10.xxxxxx/draft)

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Submitted: 01 January 1970

Published: unpublished

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## Summary

Nkululeko (Burkhardt, Wagner, et al., 2022) is a toolkit for audio-based machine learning explorations using command line interface and a configuration file, based on machine learning packages like sklearn (Pedregosa et al., 2011) and pytorch (Chaudhary et al., 2020). It is written in Python without the need to write Python code by the users. The main features are: training and evaluation of labelled speech databases with state-of-the-art machine learning approach and acoustic feature extractors, a live demonstration interface, and the possibility to store databases with predicted labels. Based on this, the framework can also be used to check on bias in databases by exploring correlations of target labels, like, e.g. depression or diagnosis, with predicted, or additionally given, labels like age, gender, signal distortion ratio or mean opinion score in.

## Design choices

The program is intended for **novice** people interested in speaker characteristics detection (e.g., emotion, age, and gender) without being proficient in (Python) programming language. Its main target is for **education** and **research** with the main features as follows:

- Finding good combinations of variables, e.g., acoustic features, models (classifier or regressor), feature standardization, augmentation, etc., for speaker characteristics detection (e.g., emotion);
- Characteristics of the database, such as distribution of gender, age, emotion, duration, data size, and so on with their visualization;
- Inference of speaker characteristics from a given audio file or streaming audio (can be said also as “weak” labeling for semi-supervised learning).

Hence, one should be able to use Nkululeko after installing and preparing/downloading their data in the correct format in a single line (add `python -m` if working in development environment without installing it).

```
$ nkululeko.MODULE_NAME --config CONFIG_FILE.ini
```

## How does it work?

nkululeko is a command line tool written in Python, best used in conjunction with the Visual Studio Code editor (but can be run stand-alone). To use it, a text editor is needed to edit the experiment configuration. You would then run nkululeko **experiment** like this:

```
$ nkululeko.explore --config conf.ini
```

35 and inspect the results afterward; they are represented as images, texts, and even a fully  
36 automatically compiled PDF report written in latex.

37 nkululeko's data import format is based on a simple CSV formalism, or alternatively, for a  
38 more detailed representation including data schemata, audformat.<sup>1</sup> Basically, to be used by  
39 nkululeko, the data format should include the audio file path of **speech dataset** (usually in  
40 WAV format) and a task-specific label. Optionally, speaker ID and gender labels help with  
41 speech data. An example of a database (in **CSV** format) labelled with emotion is

```
file, speaker, gender, emotion
x/sample.wav, s1, female, happy
...
```

42 As the main goal of nkululeko is to avoid the need to learn programming, experiments are  
43 specified by means of a configuration file. The functionality is encapsulated by software  
44 *modules* (interfaces) that are to be called on the command line. We list the most important  
45 ones here:

- 46 ■ **nkululeko**: do machine learning experiments combining features and learners
- 47 ■ **demo**: demo the current best model on the command line or some files
- 48 ■ **test**: run the current best model on a specified test set
- 49 ■ **explore**: perform data exploration (used mainly in this paper)
- 50 ■ **augment**: augment the current training data. This could also be used to reduce bias  
51 in the data, for example, by adding noise to audio samples that belong to a specific  
52 category.
- 53 ■ **aug\_train**: augment the training data and train the model with the augmented data.
- 54 ■ **predict**: predict features like speaker diarization, signal distortion ratio, mean opinion  
55 score, arousal/valence, age/gender (for databases that miss this information), with deep  
56 neural nets models, e.g. as a basis for the *explore* module.
- 57 ■ **segment**: segment a database based on VAD (voice activity detection)
- 58 ■ **ensemble**: ensemble several models to improve performance

59 The **configuration file** (INI) consists of a set of key-value pairs that are organised into several  
60 sections. Almost all keys have default values, so they do not have to be specified.

61 Here is a sample listing of an INI file (`conf.ini`) with a database section:

```
[EXP]
name = explore-androids
[DATA]
databases = ['androids']
androids = /data/androids/androids.csv
target = depression
labels = ['depressed', 'control']
samples_per_speaker = 20
min_length = 2
[PREDICT]
sample_selection = all
targets = ['pesq', 'sdr', 'stoi', 'mos']
[EXPL]
value_counts = [['gender'], ['age'], ['est_sdr'], ['est_pesq'], ['est_mos']]
[REPORT]
latex = androids-report
```

62 As can be seen, several values simply contain Python data structures like arrays or dictionaries.  
63 Within this example, an experiment is specified with the name *explore-androids*, and a **result**  
64 folder with this name will be created, containing all figures and textual results, including an

<sup>1</sup><https://audeering.github.io/audformat/>

65 automatically generated Latex and PDF report on the findings. The overall flow of basic  
66 Nkululeko experiments can be shown in Figure 1.

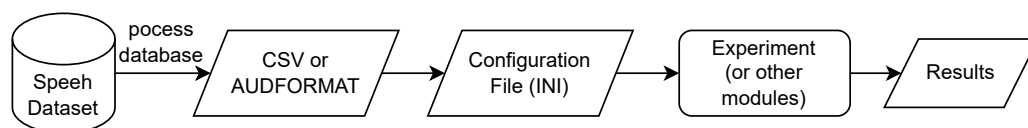


Figure 1: Nkululeko's workflow: from a raw dataset into experiment results

67 The *DATA* section sets the location of the database and specifies filters on the sample, in this  
68 case limiting the data to 20 samples per speaker at most and at least 2 seconds long. In this  
69 section, the split sets (training, development, and test) are also specified. There is a special  
70 feature named *balance splits* that lets the user specify criteria that should be used to stratify  
71 the splits, for example, based on signal distortion ratio.

72 With the *predict* module, specific features like, for example, signal distortion ratio or mean  
73 opinion score are to be predicted by deep learning models. The results are then used by a  
74 following call to the *explore* module to check whether these features, as well as some ground  
75 truth features (*age* and *gender*), correlate with the target variable (*depressed* in the given  
76 example) in any way.

77 The nkululeko configuration can specify further sections:

- 78 ■ **FEATS** to specify acoustic features (e.g. opensmile (Eyben et al., 2010) or deep learning  
79 embeddings; e.g. wav2vec 2.0 (Baevski et al., 2020)) that should be used to represent  
80 the audio files.
- 81 ■ **MODEL** to specify statistical models for regression or classification of audio data.

## 82 Example of usage

83 In the previous section, we have seen how to specify an experiment in an INI file that can  
84 be run with, for instance, *explore* and *segment* modules. Here, we show how to run the  
85 experiment (nkululeko.nkululeko) with built-in dataset (Polish Speech Emotions dataset)  
86 from the installation until getting the results.

87 First, novices could clone the GitHub repository of Nkululeko.

```
$ git clone https://github.com/felixbur/nkululeko.git
$ cd nkululeko
```

88 Then, install nkululeko with pip. It is recommended that a virtual environment be used to  
89 avoid conflicts with other Python packages.

```
$ python -m venv .env
$ source .env/bin/activate
$ pip install nkululeko
```

90 Next, extract *polish\_speech\_emotions.zip* inside the Nkululeko data folder (nkululeko/data/polish)  
91 with right click regardless of the operating system (or using *unzip* command in the terminal  
92 like below). Then, run the following command in the terminal:

```
$ cd data/polish
$ unzip polish_speech_emotions.zip
$ python3 process_database.py
$ cd ../../
$ nkululeko.nkululeko --config data/polish/exp.ini
```

93 That's it! The results will be stored in the results/exp\_polish\_os folder as stated in exp.ini.  
94 Below is an example of the debug output of the command:

```
DEBUG: nkululeko: running exp_polish_os from config data/polish/exp.ini,
nkululeko version 0.91.0
```

...

```
DEBUG: reporter:
```

	precision	recall	f1-score	support
anger	0.6944	0.8333	0.7576	30
neutral	0.5000	0.4333	0.4643	30
fear	0.6429	0.6000	0.6207	30
accuracy			0.6222	90
macro avg	0.6124	0.6222	0.6142	90
weighted avg	0.6124	0.6222	0.6142	90

```
DEBUG: reporter: labels: ['anger', 'neutral', 'fear']
```

```
DEBUG: reporter: result per class (F1 score): [0.758, 0.464, 0.621]
```

```
from epoch: 0
```

```
DEBUG: experiment: Done, used 7.439 seconds
```

```
DONE
```

## 95 What has been added since the last publication

96 Besides many small changes, mainly three big additions extended Nkululeko's functionality  
97 since the last published papers. We introduce them in the next subsections.

### 98 Finetune transformer models

99 With `nkululeko` since version 0.85.0 you can finetune a transformer model with `huggingface`  
100 (and even [publish it there if you like](#)).

101 Finetuning in this context means to train the (pre-trained) transformer layers with your new  
102 training data labels, as opposed to only using the last layer as embeddings.

103 The only thing you need to do is to set your MODEL type to *finetune*:

```
104 [FEATS]
105 type = []
106 [MODEL]
107 type = finetune
```

108 The acoustic features can/should be empty, because the transformer model starts with CNN  
109 layers to model the acoustics frame-wise. The frames are then getting pooled by the model for  
110 the whole utterance.

111 The default base model is the one from `facebook`, but you can specify a different one like this:

```
[MODEL]
type = finetune
pretrained_model = microsoft/wavlm-base

duration = 10.5
```

112 The parameter *max\_duration* is also optional (default=8) and means the maximum duration of  
113 your samples / segments (in seconds) that will be used, starting from 0. The rest is disregarded.

114 You can use the usual deep learning parameters:

```
[MODEL]
learning_rate = .001
batch_size = 16
device = cuda:3
measure = mse
loss = mse
```

115 but all of them have defaults.

116 The loss function is fixed to

- 117     ▪ weighted cross entropy for classification
- 118     ▪ concordance correlation coefficient for regression

119 The resulting best model and the huggingface logs (which can be read by [tensorboard](#)) are  
120 stored in the project folder.

121 If you like to have your model published, set:

```
[MODEL]
push_to_hub = True
```

## 122 Ensemble classification

123 Since [nkululeko](#) version 0.88.0 you can combine experiment results and report on the outcome,  
124 by using the `ensemble` module.

125 For example, you would like to know if the combination of expert features and learned  
126 embeddings works better than one of those. You could then do:

```
python -m nkululeko.ensemble \
--method max_class \
tests/exp_emodb_praat_xgb.ini \
tests/exp_emodb_ast_xgb.ini \
tests/exp_emodb_wav2vec_xgb.in
```

127 (all in one line) and would then get the results for a majority voting of the three results for  
128 Praat, AST and Wav2vec2 features.

129 Other methods to combine the different predictors, are *mean*, *max*, *sum*, *max\_class*, *uncer-*  
130 *tainty\_threshold*, *uncertainty\_weighted*, *confidence\_weighted*:

- 131     ▪ **majority\_voting**: The modality function for classification: predict the category that most  
132       classifiers agree on.
- 133     ▪ **mean**: For classification: compute the arithmetic mean of probabilities from all predictors  
134       for each label, use the highest probability to infer the label.
- 135     ▪ **max**: For classification: use the maximum value of probabilities from all predictors for  
136       each label, use the highest probability to infer the label.
- 137     ▪ **sum**: For classification: use the sum of probabilities from all predictors for each label,  
138       use the highest probability to infer the label.
- 139     ▪ **max\_class**: For classification: compare the highest probabilities of all models across  
140       classes (instead of same class as in `max_ensemble`) and return the highest probability  
141       and the class
- 142     ▪ **uncertainty\_threshold**: For classification: predict the class with the lowest uncertainty if  
143       lower than a threshold (default to 1.0, meaning no threshold), else calculate the mean  
144       of uncertainties for all models per class and predict the lowest.
- 145     ▪ **uncertainty\_weighted**: For classification: weigh each class with the inverse of its  
146       uncertainty ( $1/\text{uncertainty}$ ), normalize the weights per model, then multiply each class  
147       model probability with their normalized weights and use the maximum one to infer the  
148       label.

- 149     ▪ **confidence\_weighted**: Weighted ensemble based on confidence (1-uncertainty), normal-  
150        ized for all samples per model. Like before, but use confidence (instead of inverse of  
151        uncertainty) as weights.

## 152   Predicting Speaker ID

153   To have labels for the individual speakers in a database is extremely important, because if you  
154   mix the same speakers in training and testing data splits, it is very possible that your model  
155   simply learned some speaker idiosyncrasies instead of some underlying principle. If you don't  
156   have this labels, you could at least try to infer them with a pre-trained model.

157   With [nkululeko](#) since version 0.93.0 the [pyannote](#) segmentation package is interfaced (as an  
158   alternative to [silero](#)).

159   There are two modules that you can use for this:

- 160     ▪ SEGMENT
- 161     ▪ PREDICT

162   The (huge) difference is, that the SEGMENT module looks at each file in the input data and  
163   looks for speakers per file (can be only one large file), while the PREDICT module concatenates  
164   all input data and looks for different speakers in the whole database.

165   In any case best run it on a GPU, as CPU will be very slow (and there is no progress bar).

166   If you specify the *method* in [SEGMENT] section and the *hf\_token* (needed for the pyannote  
167   model) in the [MODEL] section

```
[SEGMENT]
method = pyannote
segment_target = _segmented
sample_selection = all
[MODEL]
hf_token = <my hugging face token>
```

168   your resulting segmentation will have predicted speaker id attached. Be aware that this is  
169   really slow on CPU, so best run on GPU and declare so in the [MODEL] section:

```
[MODEL]
hf_token = <my hugging face token>
device=gpu # or cuda:0
```

170   As a result a new plot would appear in the image folder: the distribution of speakers that were  
171   found.

172   Simply select *speaker* as the prediction target:

```
[PREDICT]
targets = ["speaker"]
```

173   Generally, the [PREDICT module is described here](#).

## 174   Statement of need

175   Open-source tools are believed to be one of the reasons for accelerated science and technology.  
176   They are more secure, easy to customise, and transparent. There are several open-source  
177   tools that exist for acoustic, sound, and audio analysis, such as librosa ([McFee et al., 2015](#)),  
178   TorchAudio ([Yang et al., 2021](#)), pyAudioAnalysis ([Giannakopoulos, 2015](#)), ESPNET ([Watanabe](#)  
179   [et al., 2018](#)), and SpeechBrain ([Ravanelli et al., 2021](#)). However, none of them are specialised  
180   in speech analysis with high-level interfaces for novices in the speech processing area.

One exception is Spotlight ([Suwelack, 2023](#)), an open-source tool that visualises metadata distributions in audio data. An existing interface between nkululeko and Spotlight can be used to combine the visualisations of Spotlight with the functionalities of Nkululeko.

Nkululeko follows these principles:

- *Minimum programming skills*: The only programming skills required are preparing the data in the correct (CSV) format and running the command line tool. For AUDFORMAT, no preparation is needed.
- *Standardised data format and label*: The data format is based on CSV and AUDFORMAT, which are widely used formats for data exchange. The standard headers are like 'file', 'speaker', 'emotion', 'age', and 'language' and can be customised. Data could be saved anywhere on the computer, but the recipe for the data preparation is advised to be saved in nkululeko/data folder (and/or make a soft link to the original data location).
- *Replicability*: the experiments are specified in a configuration file, which can be shared with others including the splitting of training, development, and test partition. All results are stored in a folder with the same name as the experiment.
- *High-level interface*: the user specifies the experiment in an INI file, which is a simple text file that can be edited with any text editor. The user does not need to write Python code for experiments.
- *Transparency*: as CLI, nkululeko *always output debug*, in which info, warning, and error will be obviously displayed in the terminal (and should be easily understood). The results are stored in the experiment folder for further investigations and are represented as images, texts, and even a fully automatically compiled PDF report written in LaTeX.

## Usage in existing research

Nkululeko has been used in several research projects since its first appearance in 2022 ([Burkhardt, Wagner, et al., 2022](#)). The following list gives an overview of the research papers that have used Nkululeko:

- ([Burkhardt, Eyben, et al., 2022](#)): this paper reported a database development of synthesized speech for basic emotions and its evaluation using the Nkululeko toolkit.
- ([Burkhardt et al., 2024](#)): this paper shows how to use Nkululeko for bias detection. The findings on two datasets, UACorpus and Androids, show that some features are correlated with the target label, e.g., depression, and can be used to detect bias in the database.
- ([Atmaja et al., 2024](#)): this paper shows Nkululeko's capability for ensemble learning with a focus on uncertainty estimation.
- ([Atmaja & Sasou, 2025](#)): in this paper, evaluations of different handcrafted acoustic features and SSL approaches for pathological voice detection tasks were reported, highlighting the ease of using Nkululeko to perform extensive experiments including combinations of different features at different levels (early and late fusions).
- ([Atmaja et al., 2025](#)): this paper extends the previous ensemble learning evaluations with performance weighting (using weighted and unweighted accuracies) on five tasks and ten datasets.

## Changes

Nkululeko has been described in three papers so far, we give a short overview on the updates since then.



- 225     ▪ **2022 Paper:** F. Burkhardt, Johannes Wagner, Hagen Wierstorf, Florian Eyben and Björn  
226     Schuller: Nkululeko: A Tool For Rapid Speaker Characteristics Detection, Proc. Proc.  
227     LREC, 2022. **New features:** First version mainly focussing on basic machinelearning  
228     experiments that combine *expert* acoustic features (like Praat or opensmile features)  
229     with traditional learning approaches.
- 230     ▪ **2023 Paper:** F. Burkhardt, Florian Eyben and Björn Schuller: Nkululeko: Machine  
231     Learning Experiments on Speaker Characteristics Without Programming, Proc. Inter-  
232     speech, 2023. **New features:** Mainly extending the acoustic features to deep-learning  
233     based (like TRILL, Hubert or wav2vec2) and the models by neural net architectures like  
234     MLP or CNN.
- 235     ▪ **2024 Paper:** F. Burkhardt, Bagus Tris Atmaja, Anna Derington, Florian Eyben and  
236     Björn Schuller: Check Your Audio Data: Nkululeko for Bias Detection, Proc. Oriental  
237     COCOSDA, 2024. **New features:** Introducing the concept of interfaces (or *modules*),  
238     focusing on the *explore-module* that features automatic data statistics and bias analysis.
- 239     ▪ **Since then:** Besides many minor enhancements; ensemble learning, Wav2vec2 model  
240     finetuning, adding automatic speaker identification, extending augmentation and seg-  
241     mentation.

## 242 Acknowledgements

243 We acknowledge support from these various projects:

- 244     ▪ European SHIFT (*MetamorphoSis of cultural Heritage Into augmented hypermedia*  
245     *assets For enhanced accessibiliTy and inclusion*) project (Grant Agreement number:  
246     101060660);
- 247     ▪ European EASIER (*Intelligent Automatic Sign Language Translation*) project (Grant  
248     Agreement number: 101016982);
- 249     ▪ Project JPNP20006 commissioned by the New Energy and Industrial Technology Devel-  
250     opment Organization (NEDO), Japan;
- 251     ▪ Project 24K02967 from the Japan Society for the Promotion of Science (JSPS).

252 We thank audEERING GmbH for partial funding.

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