

Audio denoising in the wild

Relatore: Prof. Simone Bianco

Correlatore: Dott. Paolo Napoletano

Tesi di Laurea Magistrale di:

Fabrizio D'Intinosante

838866



INTRODUCTION

premise

main purposes

basic concepts





audio-related tasks are **very popular** thanks to the amount of existing voice assistants

among all, the most common are speaker recognition and speech recognition





there are numerous problems for these tasks, **background noise** is one of the main ones [1]





the **primary objective** is the creation of a system for the background noise removal



subsequently verify the possible positive impact of this removal on the performance of a speaker recognition system through an **integrated training**







the **main idea** is to create a noise removal system that includes the following features



neural architecture



limited data pre-processing



robustness to noise intensity and duration of the audio signal





different resources





noise source



audio quality evaluation

SIWIS [2] VoxCeleb 2 [3] MUSAN [4] $(\frac{1}{3} \text{ test set})$

DEMAND [5]

Additive mixing using different SNRs





the objective is the creation of 3 **distinct** dataset for the model training



SIWIS



VoxCeleb 2

SIWIS+VoxCeleb 2

- 16 kHz re-sampled
- 2.5, 7.5, 12.5 and 17.5 SNR noise addiction
- 1 second sliding window
- rebalance through oversampling
- ark, scp storing format

METHODOLOGICAL APPROACH

models

(a) experimental setup

pipeline details



two models with two tasks



DenoisingNet (inspired to **WaveNet**) for the denoising task [6]



implemented **from scratch** using PyTorch and trained on the 3 different datasets



Thin ResNet-34 for the speaker verification task [7]



pre-trained on VoxCeleb 1
and VoxCeleb 2 train sets



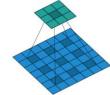


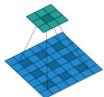
DenoisingNet characteristics

raw audio as input data



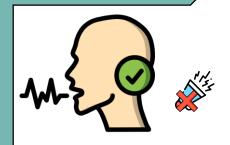
- Gated Activation Units (residual blocks)
- non-causal, **dilated** convolutions





- symmetrical 3 x 1 convolutions
- real value output with 1:1 ratio respect to input
- energy conserving loss $\mathcal{L}(\hat{s}_t) = |s_t \hat{s}_t| + |b_t \hat{b}_t|$

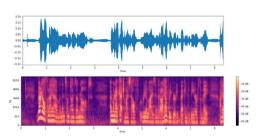




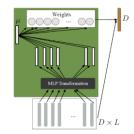
Thin ResNet-34 characteristics

- Mel spectrogram as input
- classical residual blocks





- thin because has ¼ **filters** compared to normal counterpart
- Self-Attemptive Pooling layer



- 1 x 512 embedding dimension
- angular prototypical loss

$$L_P = -\frac{1}{N} \sum_{j=1}^{N} \log \frac{e^{S_{j,j}}}{\sum_{k=1}^{N} e^{S_{j,k}}}$$









Thin ResNet-34

- 3 models (1 for dataset)
- 30 GAUs conv. blocks with 128 filters
- 3 stacks (1 to 512 dilation factor, 10 GAUs each)
- 2048 and 256 filters on final conv. layers
- 6.3 mln parameters
- scheduled Adam optimizer
- training and fine-tuning procedure with
 variable n. of epochs

- pre-trained with ~500 epochs and scheduled Adam optimizer
- 34 residual conv. blocks
- 1.4 mln parameters
- used as component for the fine-tuning procedure with the DenoisingNets







- batch size 8
- weights saved after an improvement on train and validation loss
- no predetermined n. of epochs for training and fine-tuning



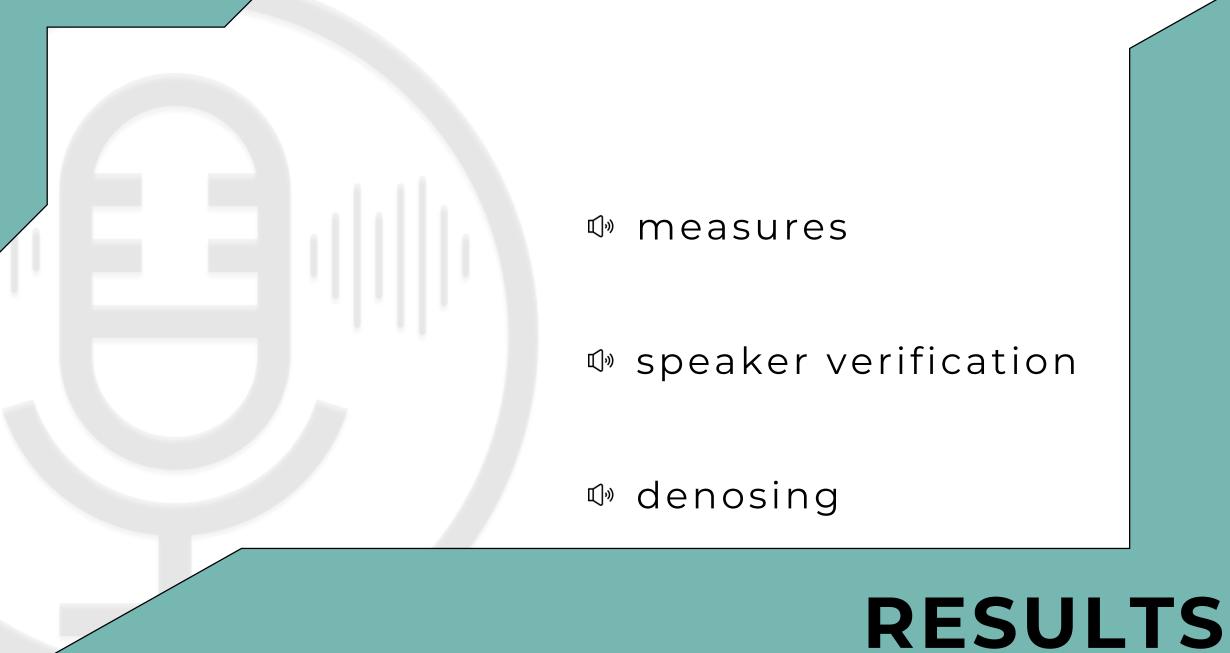
when a **plateau** for losses in training and validation is reached





fine-tuning **stacking** with Thin ResNet-34 for each model

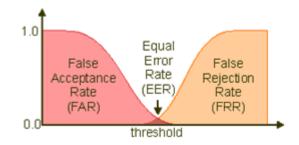




two types of measures



speaker verification quality



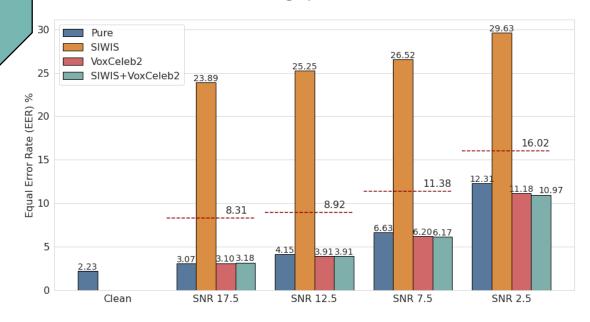
Equal Error Rate (**EER**) calculated on VoxCeleb 1 test set

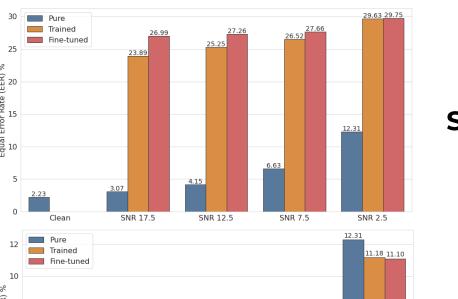
audio enhancement quality

objective quality measures [8] calculated on DEMAND dataset

- signal distortion (SIG)
- background noise distortion (BAK)
- overall quality (OVL)
- perceptual evaluation of speech quality (PESQ)
- segmental signal-to-noise ratio (SSNR)

first training performance





4.15 3.91 3.77

SNR 12.5

SNR 12.5

3.07 3.10 3.04

SNR 17.5

3.07 3.18 3.36

SNR 17.5

Trained
Fine-tuned

Clean



SIWIS

VoxCeleb

SIWIS + VoxCeleb

SNR 2.5

SNR 2.5

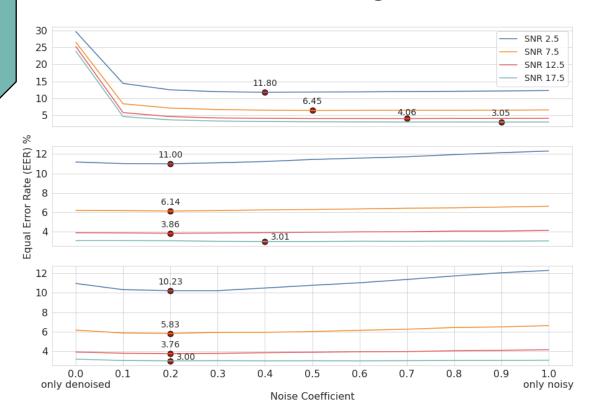
6.20 6.04

SNR 7.5

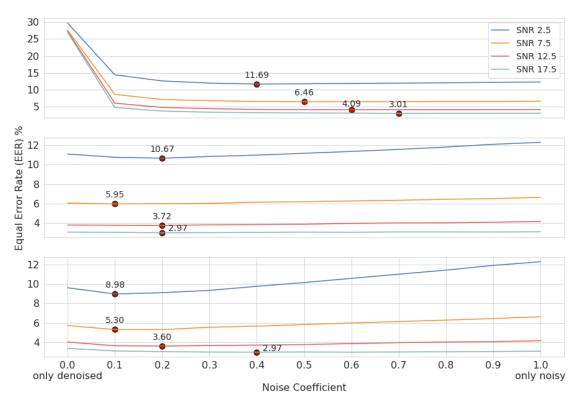
SNR 7.5



first training

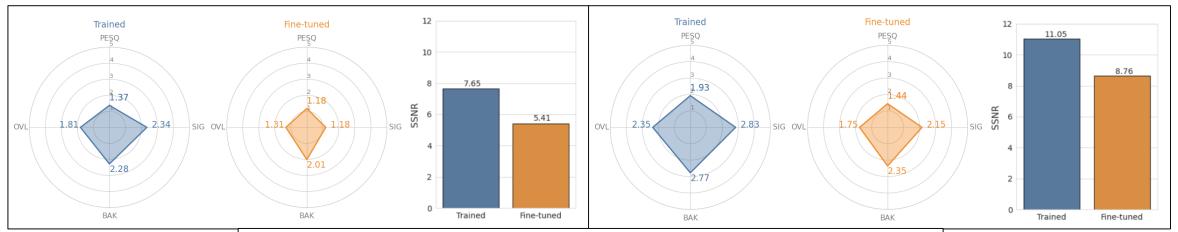


fine tuning





SIWIS VoxCeleb





SIWIS+VoxCeleb

DISCUSSION





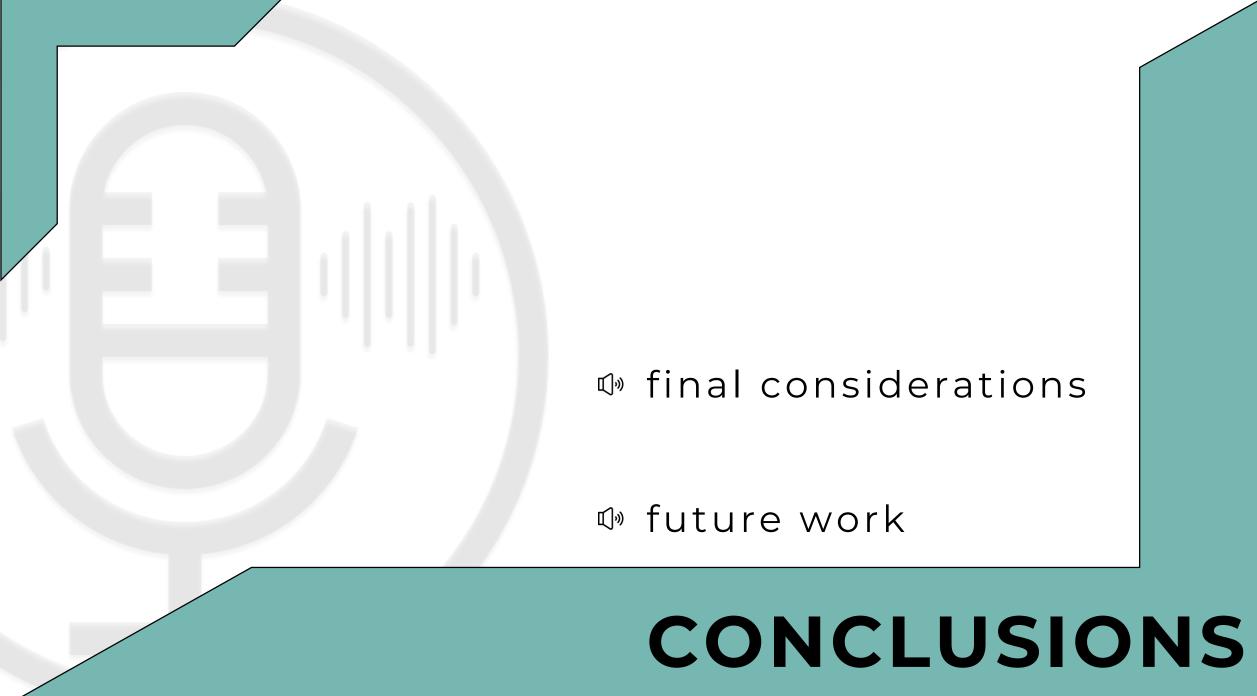


some considerations on the obtained results

- as expected, the **greater** the SNR, the **lower** the classification error
- the mixing procedure is **effective** for each model, for each SNR, both before and after the fine-tuning

the **larger** and more varied the dataset, the **better** the performance

- audio quality **degrades** for each model
- fine-tuning brought a real **benefit** to performance
- an improvement in classification does not necessarily coincide with an improvement in audio quality



in conclusion...





there is a **real improvement** in performance



the stacking strategy has shown **encouraging results**



despite the limited resources, the modesty of the results **bodes well**

possible improvements...



more computational resources



use of **generative models** (CycleGANs)



improve denoising by classifying the type of noise



try to apply denoising techniques on **non-audio signals** (i.e. electrocardiogram)



THANK YOU



REFERENCES

- N. Singh, A. Agrawal, and R. A. Khan. "Automatic Speaker Recognition: Current Approaches and Progress in Last Six Decades". In: Global Journal of Enterprise Information System 9.3 (2017), pp. 45–52.
- Pierre-Edouard Honnet et al. "The SIWIS French Speech Synthesis Database Design and recording of a high quality French database for speech synthesis". In: (Jan. 2017).
- [5] A. Nagrani, J. S. Chung, and A. Zisserman. "VoxCeleb: a large-scale speaker identification dataset". In: (2017).
- [4] D. Snyder, G. Chen, and D. Povey. "MUSAN: A Music, Speech, and Noise Corpus". In: (2015).
- [5] C. Valentini-Botinhao. "Noisy speech database for training speech enhancement algorithms and TTS models". In: (2017).
- [6] D. Rethage, J. Pons, and X. Serra. "A Wavenet for Speech Denoising". In: (2017).
- [7] J. S. Chung et al. "In defence of metric learning for speaker recognition". In: (2020).
- P. Krishnamoorthy. "An Overview of Subjective and Objective Quality Measures for Noisy Speech Enhancement Algorithms". In: IETE Technical Review28.4 (2011), pp. 292–301.



Dataset	Train set split	Test set split
SIWIS	90 %	10 %
VoxCeleb2	70 %	30 %

Dataset	Portion	n. of samples	avg. samples per speaker
SIWIS	train	7,344	~ 333
	test	816	~ 37
VoxCeleb2	train	8,448	~ 71
	test	2,113	~ 17
${\rm SIWIS+VoxCeleb2}$	train	15,792	~ 112
	test	2,929	~ 21

Dataset	Portion	n. of samples	avg. samples per speaker
SIWIS	train	7,344	~ 333
	test	816	~ 37
VoxCeleb2	train	9,027	~ 76
	test	2,287	~ 19
${\rm SIWIS} + {\rm VoxCeleb2}$	train	25,334	~ 180
	test	5,482	~ 39



Model	n. of epochs training	n. of epochs fine-tuning
SIWIS	75	83
VoxCeleb	92	66
${\bf SIWIS+VoxCeleb}$	87	82

