Group Members

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Project description

Implement digital signal processing via an ANFIS algorithm that filters speech out of noisy environments, based off the following primary paper:

https://ieeexplore.ieee.org/document/10167890

(more on ANFIS can be found at: https://ieeexplore.ieee.org/document/1595519, https://ieeexplore.ieee.org/document/955872, https://ieeexplore.ieee.org/document/256541)

Dataset

The dataset is available online, which we will mix background noise into as was done in the paper.

The clean voice dataset that we plan to use would be the same as in the primary paper; the DARPA TIMIT Acoustic-Phonetic Continuous Speech Corpus (a subset of it).

https://www.kaggle.com/datasets/mfekadu/darpa-timit-acousticphonetic-continuous-speech

To add ambient/background noise to the system we will find background noise and overlay on the clean voice data. Then to add additional white noise to the system, we will use a radio system composed of a USRP B-200 mini and a RTLSDR (both are a type of software defined radio) for the sake of creating a realistic communications environment.

Goals

Recover clean speech data using the algorithm, evaluate performance, and compare it to a traditional filter (likely a Weiner filter as in the first paper linked). Overall, we will be performing a similar procedure to the primary paper.