

Language Processing and Digital Humanities

Text Localization in Audio

Final Project - NLP Course - Dr. Asgari

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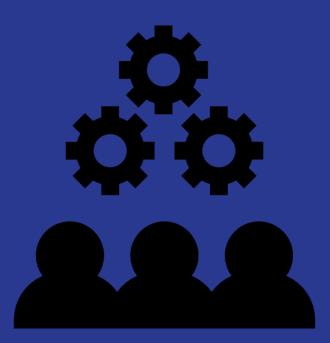
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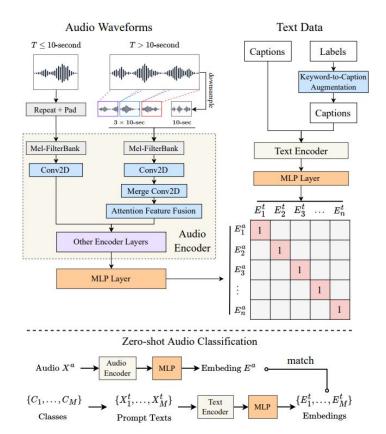
Introduction

Text localization in audio involves the identification and localization of relevant text segments within an audio stream. This task is crucial in efficiently identifying speech segments that correspond to the words in a query text, thereby enhancing the search process. Text localization finds application in several domains, including retrieving old voice messages stored on social platforms and searching for content in audio such as tutorials or music.

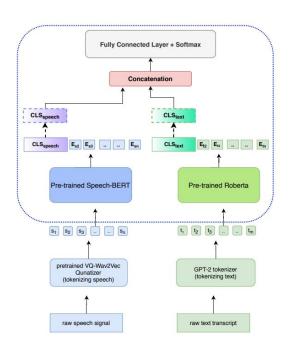
Related Works



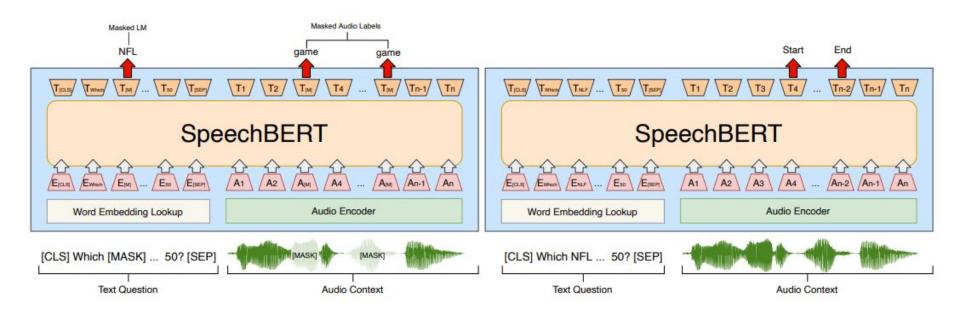
CLAP



Jointly Fine-Tuning "BERT-like" Self Supervised Models to Improve Multimodal Speech Emotion Recognition



SpeechBERT: An Audio-and-text Jointly Learned Language Model



CM-BERT: Cross-Modal BERT for Text-Audio Sentiment Analysis

Feature Extract

Audio

Add&Norm Masked **Multimodal Attention** Scale Scale Conv1D Conv1D Preprocessing Word-Level Alignment BERT

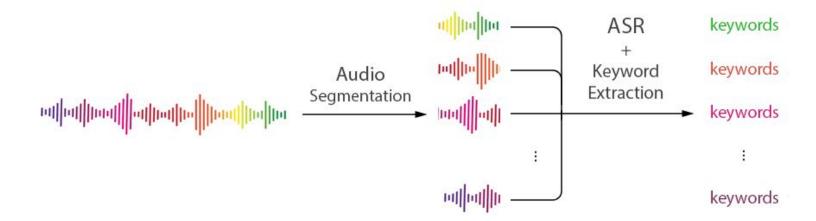
Text

Our Approach

- Create datasets for Persian and English languages
- Create baseline model used cascade ASR and keyword extractor models for solving this problem (for this we may need to create some models for different tasks)
- Create a model which uses contrastive learning and without ASR to solve this problem and for building joint space between keywords and voices

Data Processing Pipeline

Pipeline



Collecting audio files

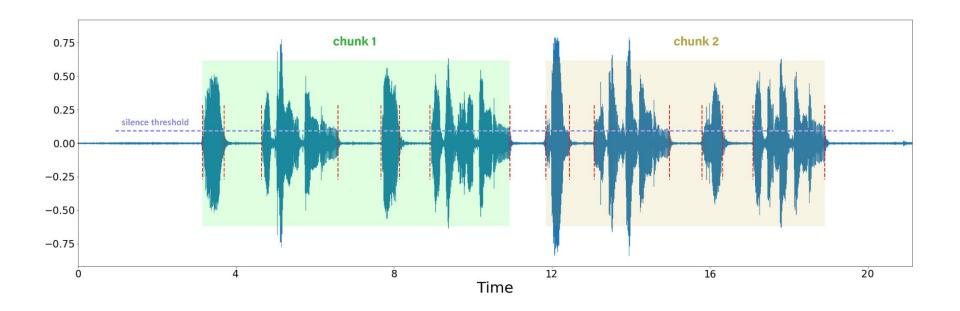
English Language Dataset:

- Used a portion of the LibriSpeech dataset
- Audio chunks have our desirable feature
- Transcript available for each chunk
- Keyword extraction model used to create desired dataset

Persian Language Dataset:

- Existing datasets comprised of very short chunks
- New dataset creation necessary
- Farsi podcast selected
- Podcast in the form of an interview with multiple speakers
- Total duration of 70 hours

Audio segmentation



Persian and English ASR

- Wav2vec2 pretrained Models
- Conformer
- U2++_conformer
- Custom Model

Persian Keyword extraction

- PKE and Perke and Perkey packages
- Bert based Language Model
- YAKE algorithm
- Multi-RAKE algorithm
- Used Our fine-tuned Persian Summarizer

English Keyword extraction

- RAKE algorithm
- YAKE algorithm
- Bert based algorithm
- Maximal Marginal Relevance

Datasets



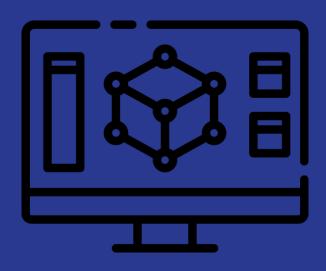
English Dataset

- Based on LibriSpeech: Small Dataset
- 3K Relevant Audios and Texts
- Create Keywords For Each Speech
- Create Sampled WaveForms For Them
- Train Test Validation Split
- Create Test Dataset With Negative Samples
- Save Bert Embedding and Wav2vec2 Embedding For Each Pairs

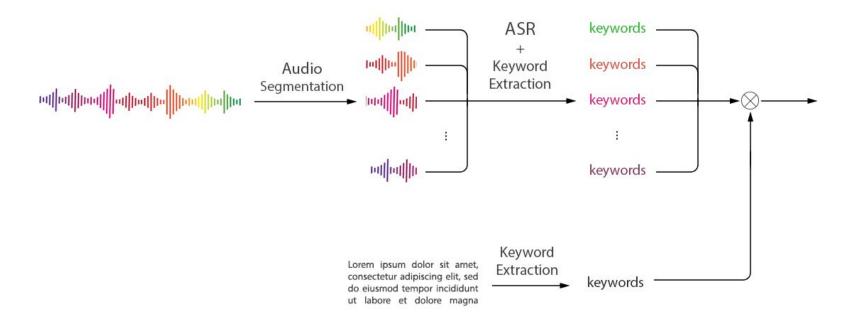
Persian Dataset

- Based on Radio Marz Podcasts
- Over 70 hours of Speech
- Use Audio Segmentations to Make Each Episodes Into Chunks
- Use Our ASR Models to Find Transcript of Chunks (Future work: Enhanced transcript with language models)
- Create Keywords For Each Chunks

Model



Baseline Model



Web-Based Demo



Audio Localizer



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Web-Based Demo



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Web-Based Demo



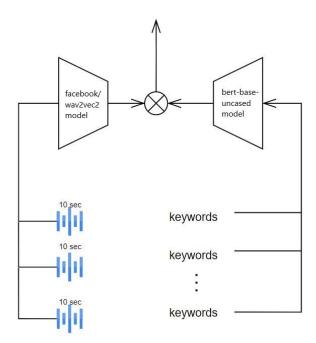
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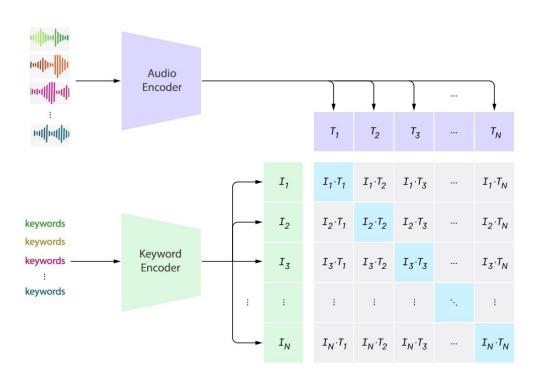
Proposed Model

Based on

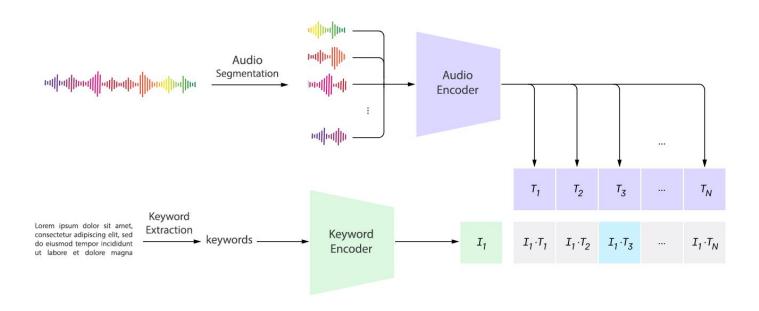
Contrastive Learning



Training



Inference



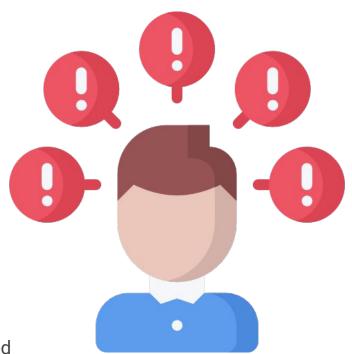
Proposed Model Problems

Loss Problem

- Contrastive Loss
- o SimCLR Loss
- L1 Loss
- Custom Loss

Resource Problems

- Generating datasets
- Training Process
- Cannot Make Architecture More Complicated
- Storage



Results

Model	Hits@1	MRR	Precision	Recall	F1 Macro	Accuracy
Proposed Model	0.163	0.406	0.5	0.05	0.09	0.1
ASR based Model (Baseline)	0.177	0.418	0.178	0.18	0.176	0.177

Future Works

- Work on Architecture of Models
- Try to Enhance Them and Reach State-of-art Models
- Improve Web Based Demo of Models

References

- Cross-modal-bert-for-text-audio-sentiment
- LARGE-SCALE CONTRASTIVE LANGUAGE-AUDIO PRETRAINING WITH FEATURE FUSION AND KEYWORD-TO-CAPTION AUGMENTATION
- Jointly Fine-Tuning "BERT-like" Self Supervised Models to Improve Multimodal Speech Emotion <u>Recognition</u>
- SpeechBERT: An Audio-and-text Jointly Learned Language Model for End-to-end Spoken Question
 Answering



