آزمایشگاه پردازش هوشمند متن و زبان و علوم انسانی محاسباتی
Language Processing and Digital Humanities

Text Localization in Audio

Final Project - NLP Course - Dr. Asgari

Mahdi Abootorabi
Arshan Dalili
Aryan Ahadiniya
Parsa Haghighi
Saeed Foroutan
Sina Rashidi

March 2023

Outline

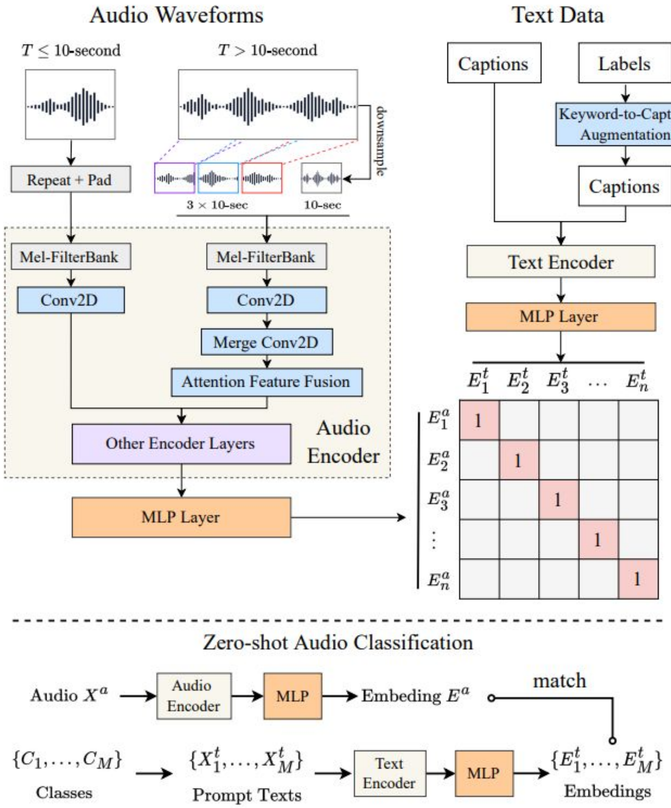
- Introduction
- Data Processing Pipeline
  - Collecting audio files
  - Audio segmentation
  - Proposed ASR model
  - Keyword extraction model
- Datasets
  - English
  - Persian
- Model
  - Proposed Model
  - Baseline
- Results

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.

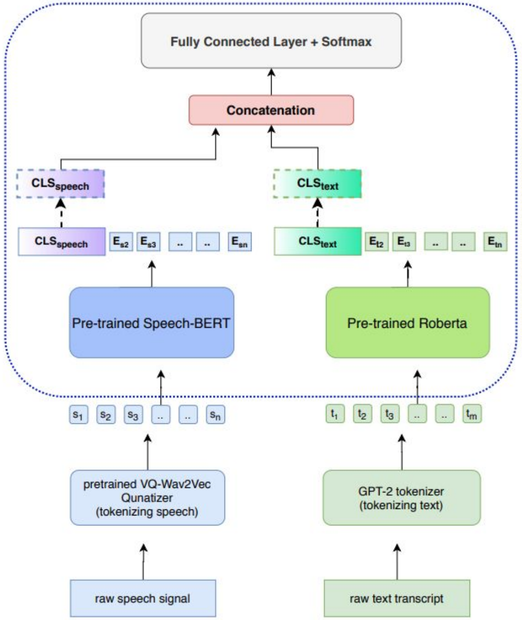
Related Works

CLAP



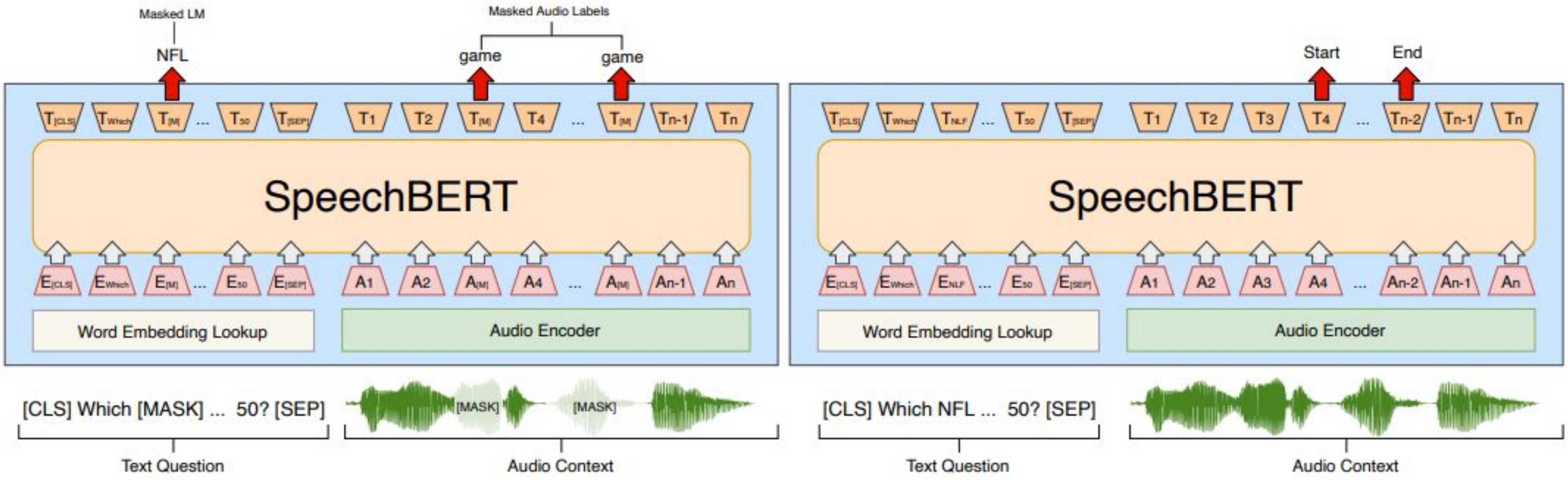
elizalde et. al, CLAP: Learning Audio Concepts From Natural Language Supervision, 2022

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



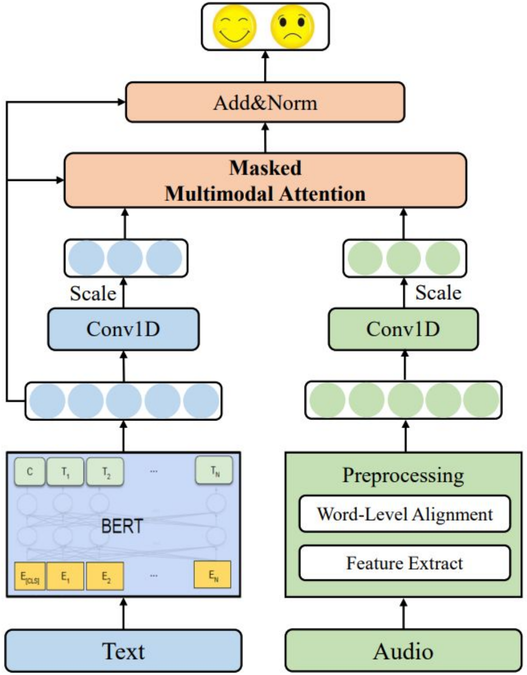
Siriwardhana et. al, Jointly Fine-Tuning "BERT-like" Self Supervised Models to Improve Multimodal Speech Emotion Recognition, 2020

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



Chuang et. al, SpeechBERT: An Audio-and-text Jointly Learned Language Model for End-to-end Spoken Question Answering, 2019

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



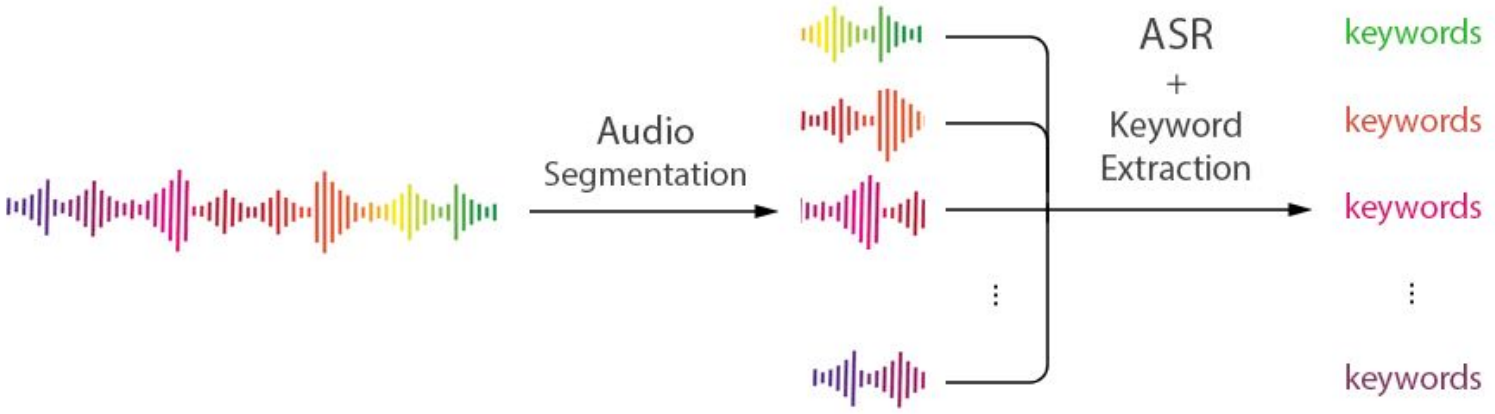
Yang et. al, CM-BERT: Cross-Modal BERT for Text-Audio Sentiment Analysis, 2020

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


Audio
Segmentation
ASR
+
Keyword
Extraction
keywords
keywords
keywords
⋮
keywords

Collecting audio files

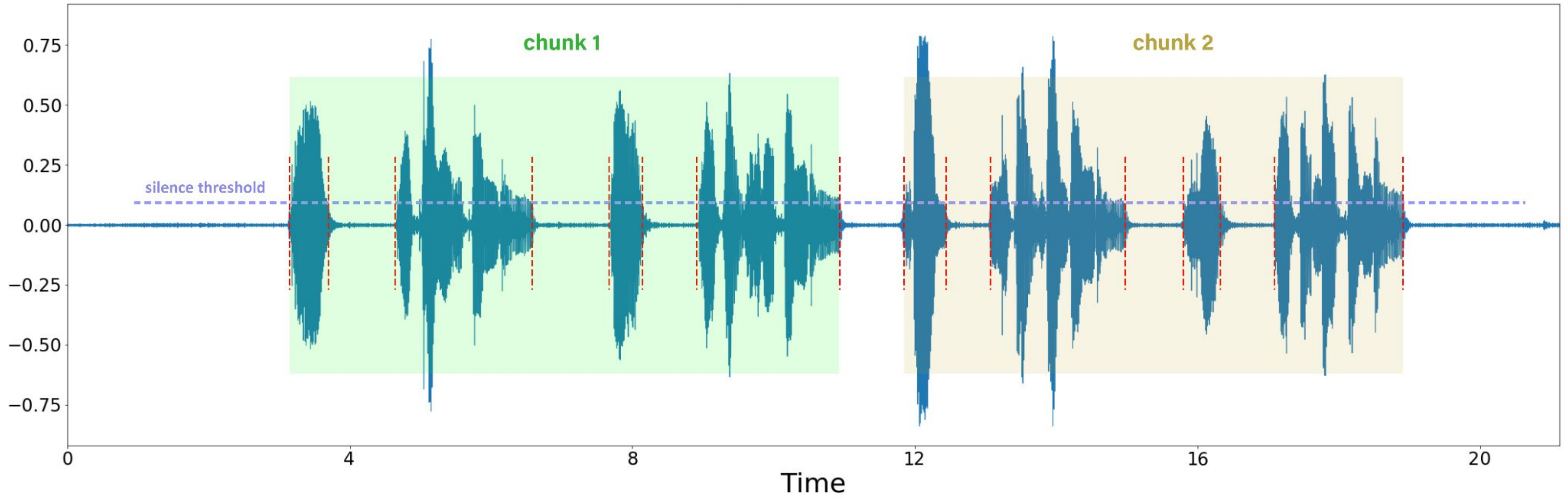
English Language Dataset:

- 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

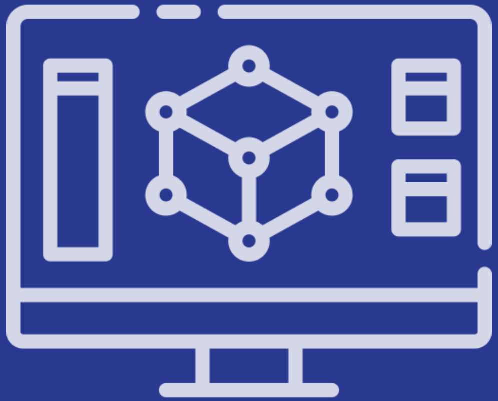
English Dataset

- 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

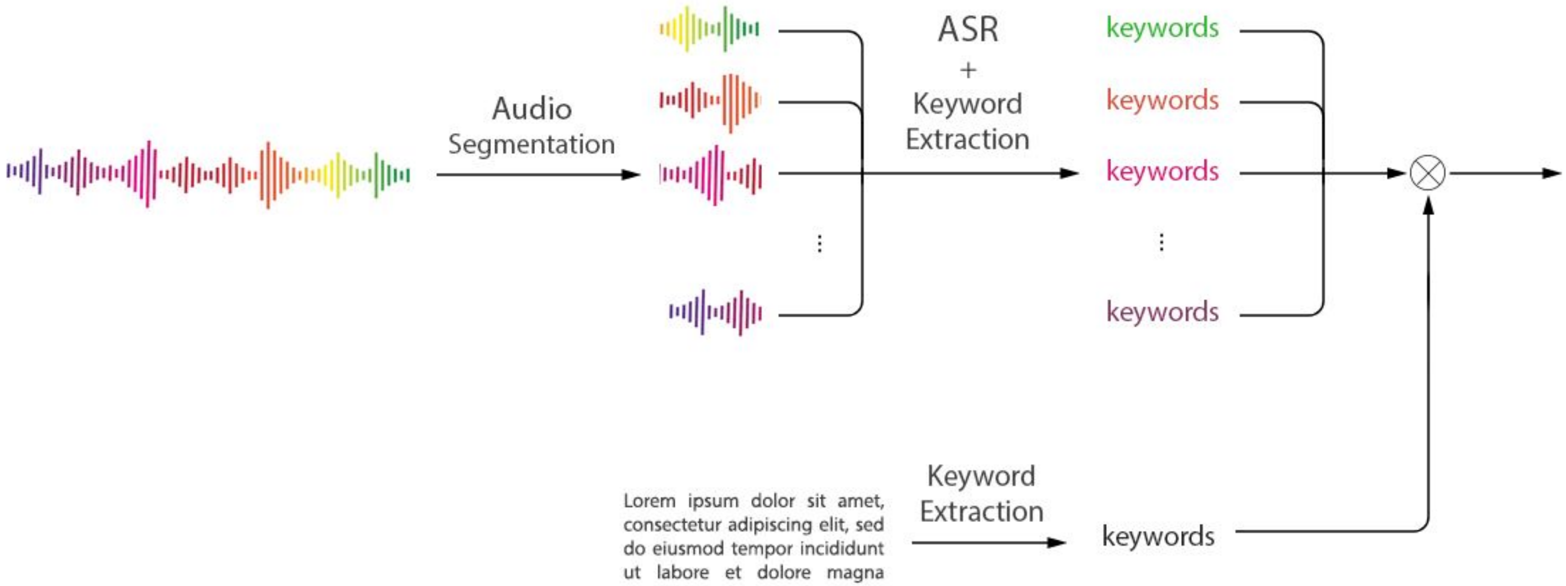
Persian Dataset

- 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

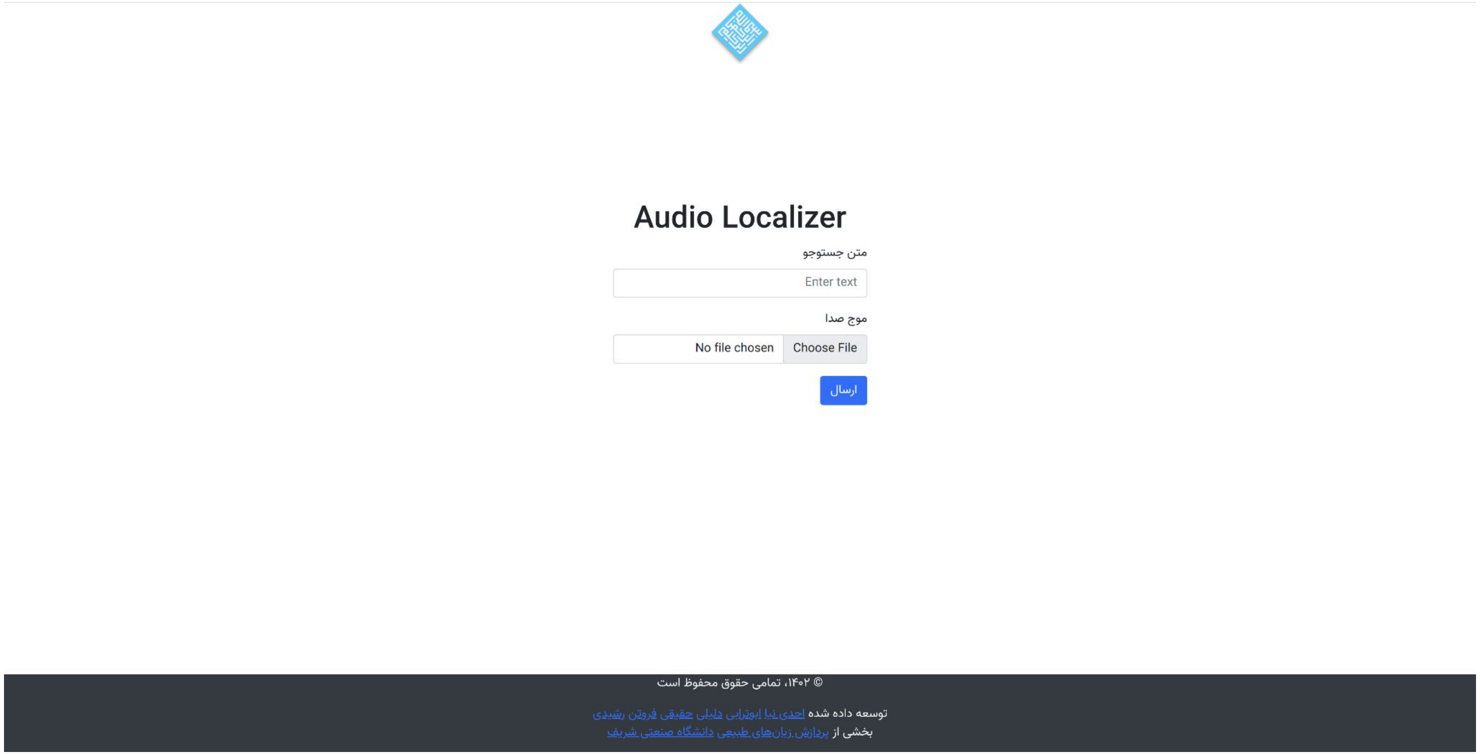
Model



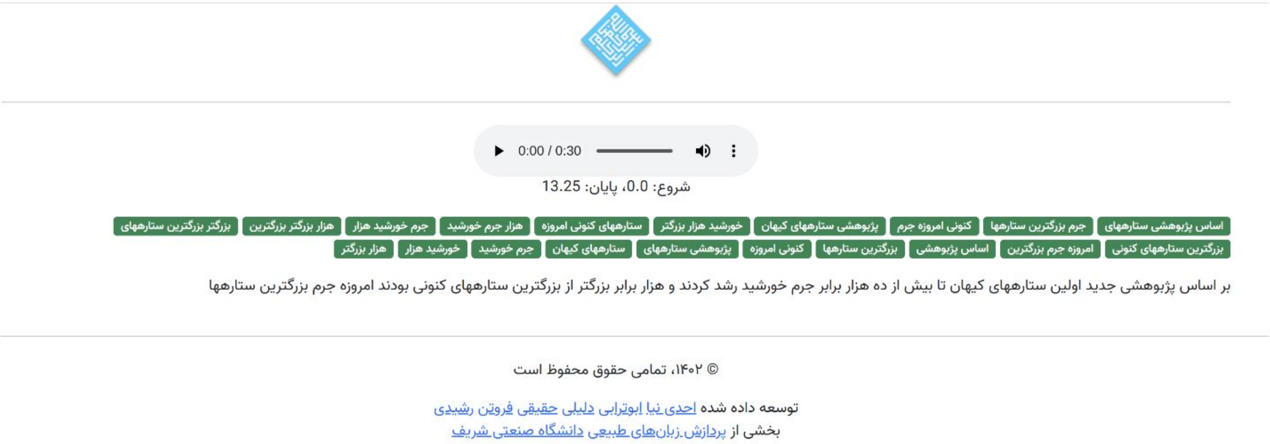
Baseline Model



Web-Based Demo



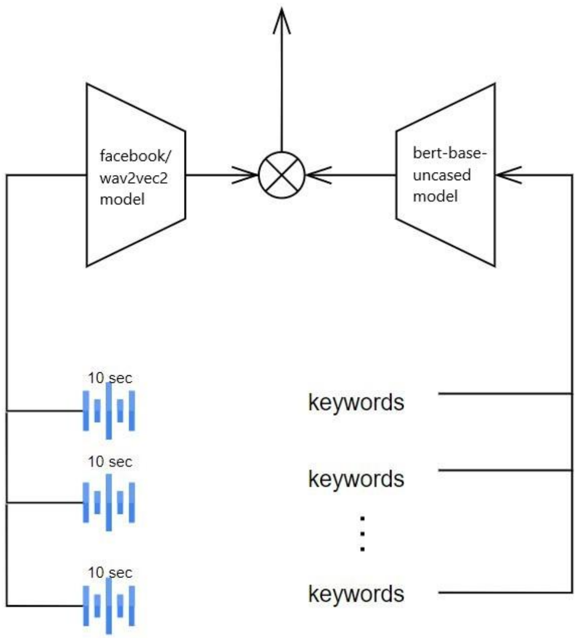
Web-Based Demo



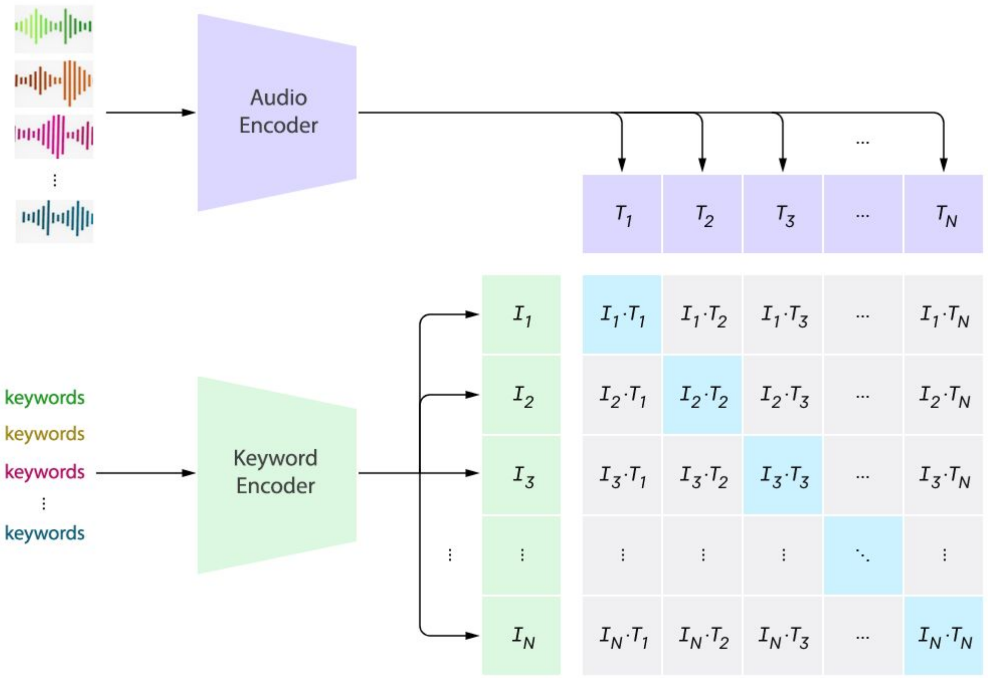
Proposed Model

Based on

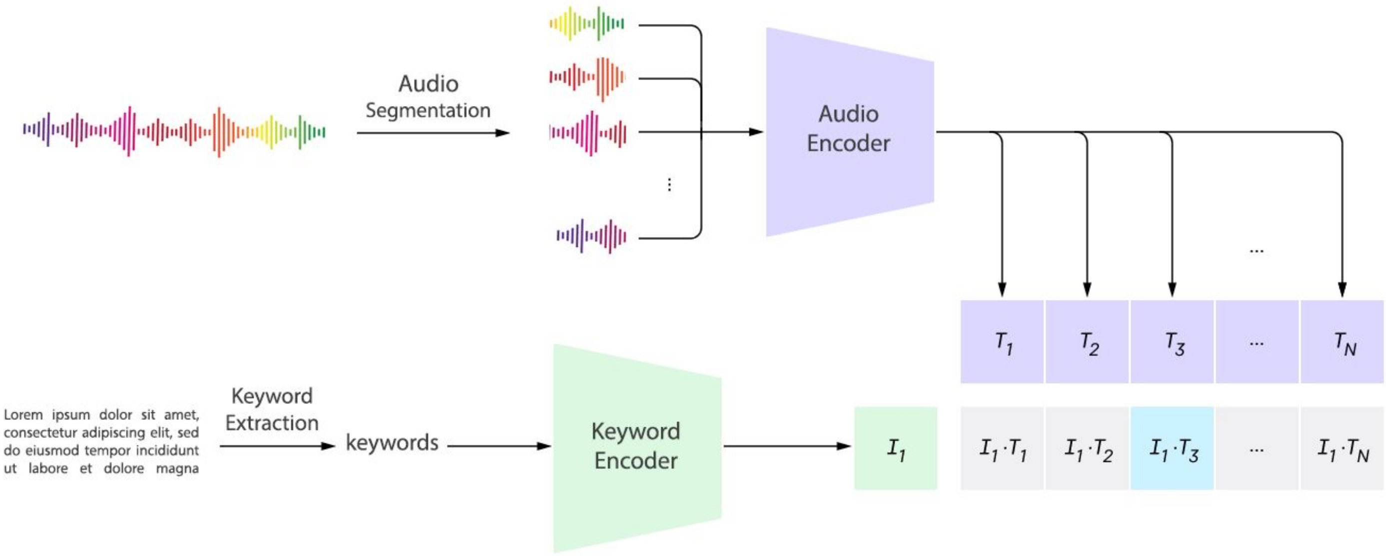
Contrastive Learning



Training


Audio Encoder
Keyword Encoder
keywords
keywords
keywords
keywords
T_1 T_2 T_3 ... T_N
I_1 I_2 I_3 ... I_N
$I_1 \cdot T_1$ $I_1 \cdot T_2$ $I_1 \cdot T_3$ ... $I_1 \cdot T_N$
$I_2 \cdot T_1$ $I_2 \cdot T_2$ $I_2 \cdot T_3$ ... $I_2 \cdot T_N$
$I_3 \cdot T_1$ $I_3 \cdot T_2$ $I_3 \cdot T_3$ ... $I_3 \cdot T_N$
$I_N \cdot T_1$ $I_N \cdot T_2$ $I_N \cdot T_3$ ... $I_N \cdot T_N$

Inference

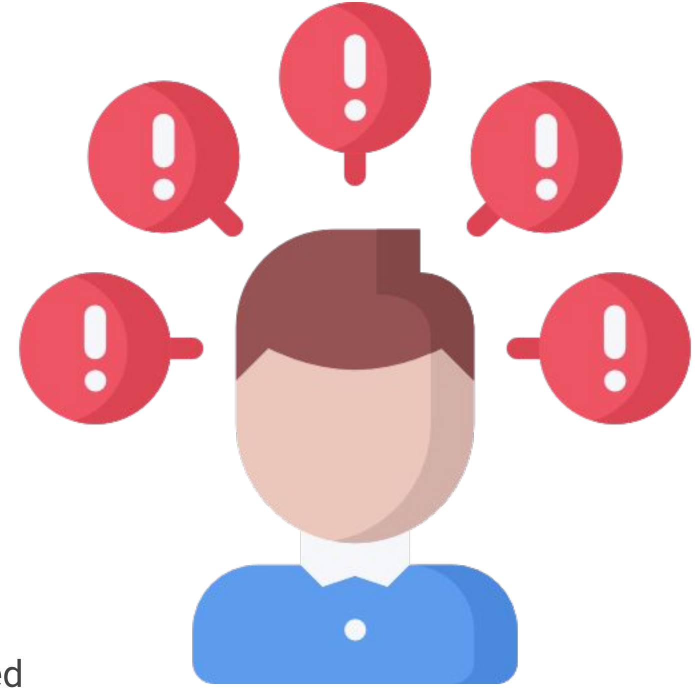


Audio
Segmentation
Audio
Encoder
⋮
...
T_1
T_2
T_3
...
T_N
Lorem ipsum dolor sit amet, consectetur adipiscing elit, sed do eiusmod tempor incididunt ut labore et dolore magna
Keyword
Extraction
keywords
Keyword
Encoder
I_1
$I_1 \cdot T_1$
$I_1 \cdot T_2$
$I_1 \cdot T_3$
...
$I_1 \cdot T_N$

Proposed Model Problems

- Loss Problem
    - Contrastive Loss
    - 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 Recognition
- SpeechBERT: An Audio-and-text Jointly Learned Language Model for End-to-end Spoken Question Answering

Any Questions?