

**TUGAS AKHIR – IF184802**

**Parallelizing CNN and Transformer Encoders for Audio Based Emotion Recognition in English Language**

**Adam Satria Adidarma**

NRP 05111942000001

Dosen Pembimbing

**Kelly Rossa Sungkono, S.Kom., M.Kom.**

NIP 1994201912088

Dosen Pembimbing 2

**Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D**

NIP 1987202012004

**Program Studi S1 Teknik Informatika**

Departemen Teknik Informatika

Fakultas Teknologi Elektro dan Informatika Cerdas

Institut Teknologi Sepuluh Nopember

Surabaya

1

2022



**TUGAS AKHIR – IF184802**

**Klasifikasi Emosi Manusia untuk Audio Berbahasa Inggris Menggunakan CNN dan Encoder Transformer dengan Teknik Pararel**

**Adam Satria Adidarma**

NRP 05111492000001

Dosen Pembimbing

**Kelly Rossa Sungkono, S.Kom., M.Kom.**

NIP 1994201912088

Dosen Pembimbing 2

**Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D**

NIP 1987202012004

# HALAMAN JUDUL

**Program Studi S1 Teknik Informatika**

Departemen Teknik Informatika

Fakultas Teknologi Elektro dan Informatika Cerdas

Institut Teknologi Sepuluh Nopember

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**Adam Satria Adidarma**

NRP 05111942000001

Advisor I

**Kelly Rossa Sungkono, S.Kom., M.Kom.**

NIP 1994201912088

Advisor II

**Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D**

NIP 1987202012004

**Study Program Bachelor of Informatics**

Department of Informatics

Faculty of Intelligent Electrical and Informatics Technology

Institut Teknologi Sepuluh Nopember

Surabaya

2022

# LEMBAR PENGESAHAN

**Klasifikasi Emosi Manusia untuk Audio Berbahasa Inggris Menggunakan CNN dan Encoder Transformer dengan Teknik Pararel**

**TUGAS AKHIR**

Diajukan untuk memenuhi salah satu syarat

Memperoleh gelar Sarjana Komputer pada

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Departemen Teknik Informatika

Fakultas Teknologi Elektro dan Informatika Cerdas

Institut Teknologi Sepuluh Nopember

Oleh : **Adam Satria Adidarma**

NRP. 05111942000001

Disetujui oleh Tim Penguji Tugas Akhir:

|  |  |  |
| --- | --- | --- |
| 1. | Kelly Rossa Sungkono, S.Kom., M.Kom. | Pembimbing |
| 2. | Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D | Ko-pembimbing |
| 3. | Nama dan gelar penguji | Penguji |
| 4. | Nama dan gelar penguji | Penguji |
| 5. | Nama dan gelar penguji | Penguji |

**SURABAYA**

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Institut Teknologi Sepuluh Nopember

By: **Adam Satria Adidarma**

NRP. 05111942000001

Approved by Final Project Examiner Team:

|  |  |  |
| --- | --- | --- |
| 1. | Kelly Rossa Sungkono, S.Kom., M.Kom. | Advisor |
| 2. | Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D | Co-Advisor |
| 3. | Name of Examiner and academic title | Examiner |
| 4. | Name of Examiner and academic title | Examiner |
| 5. | Name of Examiner and academic title | Examiner |

**SURABAYA**

**Month, Year**

# PERNYATAAN ORISINALITAS

Yang bertanda tangan di bawah ini:

Nama mahasiswa / NRP : Adam Satria Adidarma / 05111942000001

Departemen : Teknik Informatika

Dosen Pembimbing / NIP :\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_

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| Mengetahui |  |
| Dosen Pembimbing | Mahasiswa |
|  |  |
| (\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_) | (Adam Satria Adidarma) |
| NIP. | NRP. 05111942000001 |

**STATEMENT OF ORIGINALITY**

The undersigned below:

Name of student / NRP : Adam Satria Adidarma / 05111942000001

Department : Informatics

Advisor / NIP :\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_

Hereby declare that Final Project with the title of “Title of Final Project” is the result of my own work, is original, and is written by following the rules of scientific writing.

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|  |  |
| --- | --- |
|  | Surabaya, \_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_ |
| Acknowledge |  |
| Advisor | Student |
|  |  |
| (\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_\_) | (Adam Satria Adidarma) |
| NIP. | NRP. 05111942000001 |

**Klasifikasi Emosi Manusia untuk Audio Berbahasa Inggris Menggunakan CNN dan Encoder Transformer dengan Teknik Pararel**

**Nama Mahasiswa / NRP : Adam Satria Adidarma / 05111942000001**

**Departemen : Teknik Informatika FTEIC- ITS**

**Pembimbing : Kelly Rossa Sungkono, S.Kom., M.Kom.**

**Ko-Pembimbing : Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D**

# ABSTRAK

**Abstrak**

Kecerdasan artifisial telah berdampak signifikan pada berbagai industri dan sektor masyarakat, dengan adopsi kecerdasan artifisial yang tumbuh 37% dari 2018 hingga 2019, menurut laporan Gartner. Pengenalan emosi bicara (PEB) adalah subbidang kecerdasan artifisial yang fokus pada mengenali aspek emosional manusia saat berbicara, terpisah dari konten semantik. Emosi berperan penting dalam komunikasi manusia dan telah menjadi objek penelitian yang semakin meningkat dalam beberapa tahun terakhir. Meskipun studi saat ini tentang deteksi emosi sering memfokuskan pada modalitas visual, seperti ekspresi wajah, emosi adalah konsep multimodal yang membutuhkan studi terhadap indikator visual, taktil, vokal, dan fisiologis. PEB dapat diterapkan dalam berbagai konteks, termasuk pusat panggilan, pendidikan, pemasaran, psikologi, dan kesehatan. Studi ini mengusulkan pendekatan untuk menerapkan sistem PEB menggunakan model Convolution Neural Network (CNN) yang bekerja pararel dengan jaringan encoder Transformer dan mengevaluasi performanya pada dataset audio CREMA-D berbahasa Inggris. Model yang diusulkan akan dibandingkan dengan berbagai arsitektur pembelajaran mesin dalam hal performa, termasuk CNN biasa dan Support Vector Machine (SVM), untuk menentukan pendekatan yang paling efektif untuk PEB.

**Kata kunci: Kecerdasan Artifisial, CNN, Transformer, PEB, *Self-Attention*, Audio.**

**Parallelizing CNN and Transformer Encoders for Human Emotion Classification for Audio Based Emotion Recognition in English Language**

**Student Name / NRP : Adam Satria Adidarma / 05111942000001**

**Department : Teknik Informatika FTEIC- ITS**

**Advisor : Kelly Rossa Sungkono, S.Kom., M.Kom.**

**Co-Advisor : Shintami Chusnul Hidayati, S.Kom., M.Sc., Ph.D**

**Abstract**

Artificial intelligence (AI) has had a significant impact on various industries and sectors of society, with the adoption of AI growing 37% from 2018 to 2019, according to a Gartner report. Speech emotion recognition (SER) is a subfield of AI that focuses on recognizing the emotional aspects of speech, separate from the semantic content. Emotions play a crucial role in human communication and have been the subject of increasing research in recent years. While current studies on emotion detection often focus on visual modalities, such as facial expressions, emotion is a multimodal concept that requires the study of visual, tactile, vocal, and physiological indicators. SER can be applied in various contexts, including call centers, education, marketing, psychology, and healthcare. This study proposes an approach to implement an SER system using a parallelized Convolutional Neural Network (CNN) model and Transformer encoder network and evaluate its performance on the CREMA-D English audio dataset. The proposed model will be compared to various machine learning architectures in terms of performance., including standard Convolutional Neural Networks (CNN) and the Support Vector Machine (SVM), to determine the most effective approach for SER.

**Keywords: AI, CNN, Transformer, SER, Self-Attention, Audio.**

# KATA PENGANTAR

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# DAFTAR SIMBOL (jika ada)

# DAFTAR SINGKATAN (jika ada)

# Chapter I

**Introduction**

In this chapter, the research background and context will be examined, including the problem being addressed, the scope of the problem, and the purpose and potential benefits of the research being conducted.

## Background

Artificial Intelligence has emerged in every industry and has a profound impact on every sector of human society. According to Gartner Report (Costello, 2019), artificial intelligence adoption has grown 37% during 2018-2019 because the capabilities of artificial intelligence have matured significantly over the years leading to the adoption of this technology by enterprises around the world. Speech Emotion Recognition (SER) is one of the emerging applications in the context of artificial intelligence. SER is the task of recognizing the emotional aspects of speech independently over the semantic content. Humans can efficiently perform this task as a natural part of our communication, but the ability to do it automatically using a programmable device is still a subject of research (Lech, Stolar, Best, & Bolia, 2020).

In the book of The Media Equation (Ivar, Byron, & Clifford, 1996), Studies in human-computer interaction made the discovery that people often interact with computers as if they were other people and react to similar feedback from humans. Most of these social aspects ranging from politeness to reciprocity have been observed in human-computer interactions. Computer scientists believed that emotions and machines should connect in order to have better and more effective communication. Both data-driven reasoning and emotional perception are crucial for a machine’s intelligence(Cowie, 2001). Giving machines emotional intelligence, the general user experience, and machine performance will be improved.

Emotions play a big role in human communication. Over the past years, research to understand human emotions was increasing (Jarymowicz & Maria, 2012). There are already a variety of computer systems that uses emotional speech classification as security systems, psychology and computer vision applications, and interactive computer designs. Current studies on emotion detection mainly focus on visual modalities, including facial expressions, muscle movements, hand posture, body posture, *etc.* (Keltner, Dacher, & Cordaro, 2017). However, emotion is a multimodal concept, and the task to detect emotions requires interdisciplinary studies that include visual modality, tactile communication, vocalization, and physiological indications (Heredia, Cardinale, Dongo, & Díaz-Amado, 2021).

A speech recognition system's success depends on the selection of a speech multimodal database, the extraction of pertinent features, and the selection of an effective classification algorithm. In the aforementioned works, emotion detection using audio data was chosen because it can be applied to various computer application system that doesn't require visual modalities, such as emotion detection on call center services to analyze customer habits to help improve the quality of service for the provider through sounds. Emotion detection based on audio data can also help learning experience in the field of education to help improve students’ mental health by monitoring their emotions through sound. This system can also be used across various applications, such as marketing, psychology, health care, *etc*.

Emotion classification and sound detection using multiple SVM methods, such as linear and nonlinear, have received significant interest recently (Sonawane, Inamdar, & Bhangale, 2017). Some studies also tried to improve the accuracy of this method by using transfer learning on pre-trained deep learning models (Latif, Rana, Younis, Qadir, & Epps, 2018). Their results showed that deep learning-based CNN methods outperformed the handcrafted feature-based SVM method in image classification (Younghak Shin, 2017). This is because deep learning methods learn categories incrementally through its hidden layer architecture, defining low-level categories first, and then moving to the higher-level categories. The number of datasets can also be a factor in improving the quality of this CNN method (Mahapatra, 2018). In addition, some robust deep learning architectures such as GPT-3 & BERT(Brown, et al., 2020)(Devlin, Chang, Lee, & Toutanova, 2019) are emerging to solve sequential learning problems based on a self-attention mechanism in the Transformer Network(Vaswani, et al., 2017). These architectures are now considered a state-of-the-art technique in the field of NLP (Natural Language Processing).

Based on the above statement, this study aims to implement a deep learning-based CNN method in parallel with a self-attention mechanism Transformer encoder network in the process of detecting human emotions with an English audio dataset. With the application of this method, it is hoped that this model could provide an expressive feature representation with the lowest computational cost by extending the CNN filter channel size and reducing the feature maps, while the Transformer encoder is used for the network to learn how to predict the frequency distribution of different emotions according to the overall structure of the MFCC plot of each emotion.

## Problem Statement

From the background stated previously, the problem statement can be expressed as follows:

1. How to detect human emotions from audio data with proposed method?
2. Which classification methods are more accurate to detect emotion through audio between SVM, CNN, and the proposed method?
3. How to build the model architecture to give a good accuracy?

## Problem Scope

In order to stay true to the issues raised above, this paper includes a number of constraints. The problem in this paper has the following limitations:

1. Audio data is in English;
2. In one voice of the dataset, there is only one emotion;
3. The model can only distinguish between the six emotions of anger, disgust, fear, happy, neutral, and sad;

## Purpose

The purpose of this research is as follows:

1. Find out the process to detect human emotions from audio data with proposed method;
2. Determine which method has higher accuracy between SVM model, CNN based model, and the combination of CNN and Transformer Encoder model proposed in the study for detecting emotions through audio;
3. Determine what architecture is going to give a good accuracy for the proposed model;

## Benefit

The purpose of this study is to implement a human emotion detection system through voice for emotional perceptions of a robot machine intelligence. The proposed study will be beneficial in a number of ways. Firstly, the proposed method is expected to provide more accurate and reliable results in detecting human emotions compared to existing methods. This will be invaluable in the development of better artificial intelligence (AI) systems, which can be used to provide more personalized services and better understand human behavior. Secondly, the proposed study will compare the results from the proposed method to existing methods, to examine how effective the proposed method is at detecting human emotions. This comparison will provide valuable insight into the effectiveness of the proposed method and will help to inform future research in this field. Finally, the proposed method could be used to develop AI systems that are better able to interact with humans and provide personalized services, such as customer service or healthcare.

# Chapter II

**Literature Review**

This chapter will discuss about previous research on this topic and present the foundational theories that guide this study.

## Related Works

In the context of detecting human emotions through audio data, the selection and extraction of audio features are important to understand. The Sequential Minimal Optimization (SMO) algorithm was used as the primary method of sound analysis during the training of SVM models in recent years. In this case, the sound is divided into a number of frames which will then be examined iteratively. There are two emotional characteristics of the voice that can be observed to understand human emotion (Citron, Gray, Critchley, Weekes, & Ferstl, 2014): arousal, which is the level of autonomic activation that an event creates, which ranges from calm to excited. and valence, which is the level of pleasantness that an event generates and is defined along a continuum from negative to positive.

The INTERSPEECH 2013 (Steidl, et al., 2013) introduced us to various aspects of speech and audio that are connected to emotions which employ the SMO algorithm using a rather 'brute force' method to classify and define audio feature sets. Another research such as (Eyben, et al., 2016) introduced a new method of audio feature extraction using a minimal set of parameters, which implements prosodic, excitation, vocal tract, spectral descriptors, and an extension to the minimalistic set, which contains a small set of cepstral parameters (i.e., MFCC & Spectral Flux).

Emotion recognition from pure speech is one of the most sophisticated and sophisticated and widespread techniques and progress in this field relies heavily on the composition of emotional speech datasets. The structure of the emotional speech corpus can be divided into two parts in general. The first part is lab recording, which is a collection of speech datasets that are often recorded in a recording studio using high-quality microphones and accompanied by linguistics experts. Some of the corpora that use this type of structure are EmoDB (Burkhardt, Paeschke, Rolfes, Sendlmeier, & Weiss, 2005), a database of german emotional speech comprising 800 sentences with 10 utterances by 10 different actors that could be used in normal conversation and could be interpreted according to all the emotions employed. IEMOCAP (Busso, et al., 2008), a database consisting of 12 hours worth of audiovisual with multimodal and multispeaker data, including 10 actors both scripted and improvised sessions recorded by the University of Southern California's SAIL Lab. AESDD (Vryzas, Kotsakis, Liatsou, Dimoulas, & Kalliris, 2018), includes Greek language expressions of acted emotional speech and the other controlling spontaneous emotional speech. The second corpus type is non-lab recording. This corpus contains utterances that reflect emotions involuntarily in natural scenarios, such as living spaces, theatrical performances, etc. Some examples that employ this type of corpus are DAPS (Mysore, 2015), this dataset is a collection of aligned recordings of the same speech made on typical consumer devices in real-world settings that consist of approximately 4 and a half hours of data. Freefield1010 (Stowell & Plumbey, 2013), a collection of 7690 excerpts from field recordings throughout the world, was later standardized for research. CHEAVD (Li, Tao, Chao, Bao, & Liu, 2017), containing 140 minutes of emotional segments from movies, TV shows, and talk shows with 238 speakers, ranging from children to the elderly, covers a wide range of speaker diversity.

Studies on different methods of speech representation have been done in recent years with various types of deep-learning architecture. In 2019 (Schneider, Baevski, Collobert, & Auli, 2019), the wav2vec model introduced us to unsupervised learning for speech recognition by learning representations of unprocessed audio data. Then in 2020 (Baevski, Zhou, Mohamed, & Auli, 2020), the second version of this model was introduced which improves the model even further by employing a self-supervised training method based on contrastive learning for automatic speech recognition. However, in 2021, HuBERT (Hsu, et al., 2021) highlighted many issues with the self-supervised learning approach. These problems include (1) many pronunciation units in the speech, (2) no vocabulary of sound units during the pre-training phase, and (3) the length of sound units being changeable without any segmentation. With these problems, the idea of the HuBERT model is to apply the prediction loss only to masked regions and force the model to learn good high-level representations of unmasked inputs to infer the targets of masked ones correctly. Other studies such as the UniSpeech (Wang, et al., 2021) pointed out a problem in the speech recognition community that some of the successful techniques require thousands of hours of human-annotated speech recordings for training which is not available for a lot of languages spoken worldwide. The UniSpeech model can learn consistent contextual representations using both supervised and unsupervised data. This model consists of convolutional feature extraction, a transformer encoder, and a feature quantizer. UniSpeech is able to perform better than both supervised and unsupervised pre-training on multilingual speech recognition tasks. Furthermore, WavLM (Chen, et al., 2022) was introduced as an extension of HUBERT (Hsu, et al., 2021) to masked speech prediction and denoising modeling, so the pre-trained model performs well on both automatic and non-automatic speech recognition to solve full stack speech processing tasks. This model achieved the best performance on multiple speech datasets.

In a typical speech emotion recognition system, audio data, feature extraction, classification models, and emotional output recognition are all included. Some of the popular classification methods right now for an emotion recognition system include SVM (Sonawane, Inamdar, & Bhangale, 2017), Hidden Markov Model (HMM) (Starner & Pentland, 1995), and Recurrent Neural Network (RNN) (Chamishka, et al., 2022). Speech emotion recognition tasks require an emotion speech database for training the model. In this study, the CREMA-D (Crowd-Sourced Emotional Multimodal Actors Dataset)(Cao, et al., 2014) dataset are used for human emotion classification which includes 7,442 clips of 91 actors, with each actor performing a set of basic emotions (anger, disgust, fear, happiness, sadness, surprise) as well as neutral expressions with a distribution of 48 male and 43 female actors coming from a variety of races and ethnicities (African America, Asian, Caucasian, and Hispanic).

## Basic Theory

This chapter will explain the basic theory used as a reference in this study. Among other things, this chapter will explain the literature review, human emotion, voice understanding, speech recognition, feature extraction, neural network convolution, and transformers, as well as a brief explanation of the framework library, used to implement emotion detection in the human voice in this study, namely PyTorch.

## Emotion

Emotion is an aspect of consciousness which are generally understood to represent the synthesis of subjective experience, expressive behavior, and neurochemical activity. Most researchers consider them to be part of the evolutionary legacy of the human species and serve adaptive purposes by supplementing common perception and facilitating social communication.(Solomon, 2009)Emotions come in a variety of forms, and they all have an impact on how humans live and relate to each other. There are times when we may feel as though these emotions are controlling us. Our actions, behaviors, and perceptions are all influenced by the emotions we are experiencing at any given time.According to(Cherry, 2021), psychologist Paul Eckman identifies six fundamental emotions that were shared across all human societies in the 1970s. These emotions include *happiness, sadness, disgust, fear, surprise,* and *fury*. Later, he expanded this list for *pride, humiliation, embarrassment, and enthusiasm*. Figure 2.1 depicts various human emotions nowadays.

A collage of a person

Description automatically generated

Figure 2.1: Human Emotions (Charlie, 2014).

Diagram

Description automatically generated with medium confidence

Figure 2.2: Longitudinal Nature of Sound Wave (StudyCorgi, 2022).

## Sound

Sounds are produced by sound waves. Humans could hear it by passing a medium through the ears. All sound is produced by the vibration of molecules. For example, when a person makes a sound, there are vibrations move through the air molecules. Sound waves travel away from where they originate. When these vibrating air molecules reach the ear, the eardrum also vibrates. The bones in the ear vibrate as if the object that generated the sound waves vibrates. There are three types of continuous mediums which are solids, liquids, and gases. Sound travels faster through a solid medium since the particle here is closer together than in gases or liquid medium. These vibrations let humans hear different things such as music. There are also irregular vibrations called noises. Human beings could make very complex sounds used for talking. A sound wave is a longitudinal wave that has two parts (Compression and Rarefaction). Compression is where air molecules are pushed together. Rarefaction is where the molecules are far apart. Sound is produced by a series of mechanical compressions and rarefactions of mechanical waves that sequentially propagate through a medium (StudyCorgi, 2022). Figure 2.2 shows a representation of the longitudinal nature of sound waves.

## Speech Recognition

Speech Recognition is an interdisciplinary subject of computer science and computational linguistics that develops approaches and technology to enable the translation of spoken language into text by computer machines with the main benefit of searchability. It is often referred to as computer voice recognition or automatic speech recognition (ASR). Speech recognition draws on expertise and research from the domains of computer science, linguistics, and computer engineering.

Speech recognition systems use computer algorithms to process**,** interpret**,** and convert spoken words into text. A software program converts the sounds picked up by the microphone into characters that computers and humans can understand**.**This program must be able to adapt to the highly variable and context-specific nature of human speech. The software algorithms that process human speech are trained on a variety of speech patterns, speaking styles, language, accents, and idioms. The software also separates speech from the background noises that often accompany the signals (Yu & Deng, 2015).

## Feature Extraction

In machine learning, feature extraction is the process of turning raw data into numerical features that can be processed while keeping the information in the original dataset. The amount of redundant data in the dataset is decreased within this process. In the end, the data reduction speeds up the learning and generalization phases of the machine learning process while also enabling the model to be built with less computation power. This study employs one of the most popular feature extraction methods in the context of Speech Emotion Recognition (SER) called the Mel-Frequency Cepstral Coefficient (MFCC) (Kishore & Satish, 2013). The procedure to find MFCCs is mainly with the following steps shown in Figure 2.3:

Diagram

Description automatically generated

Figure 2.3: MFCC Block Diagram.

1. *Pre-Emphasis*

The structure of a voice production system's design causes dampening in high-frequency regions. Pre-Emphasis amplifies high-frequency sections and conducts filtering which is used to offset the spectrums of voiced regions. Widely used pre-emphasis filter is given in Equation 2.1,

|  |  |
| --- | --- |
|  | (2.1) |

Where:

* is the output signal at time n.
* is the input signal at time n.
* is the pre-emphasis coefficient.
* is the input signal at the previous time step (n-1).

1. *Frame Blocking*

Due to voice signal as a slow time-varying signal, speech analysis over a short enough time span is required for stable acoustic features. Frame blocking entails processing the voice signal at short time intervals to extract the characteristic features in a more stable condition.

1. *Windowing*

Windowing is the process of splitting an audio signal into segments of specific lengths. This reduces the effect of aliasing or signal discontinuity at the beginning and end of each frame that could occur due to the frame-blocking process.

1. *Discrete Fourier Transform (DFT)*

Discrete Fourier Transform is one of the most powerful tools in digital signal processing which enables us to find the spectrum of a finite-duration signal. In MFCC, DFTs are used to convert each windowed frame into a magnitude spectrum with Equation 2.2,

|  |  |
| --- | --- |
|  | (2.2) |

Where:

* is the frequency domain sample, with ranging from .
* it the time domain sample, with ranging from .
* it the number of samples in the sequence.
* is the imaginary unit .
* is the mathematical constant .

1. *Mel-Frequency Warping*

In this process block, the triangle waves that make up the Mel filter bank's frequency in Hz units are used to create the signal. As a result, using this method, the signal's value in frequency units is determined. The MFCC coefficient value is determined by the number of filters in Mel's filter bank. The Mel scale is a nonlinear scale that compresses the higher frequencies, which are more difficult for humans to perceive. The algebraic equation for the process of converting Mel spectrum and FFT frequency values **​**in Hz to Mel frequency units is defined in Equation 2.3 **as:**

|  |  |
| --- | --- |
|  | (2.3) |

Where:

* is the frequency of Mel.
* is the frequency in Hz.
* is the logarithm base 10.

1. *Discrete Cosine Transform (DCT)*

A DCT is applied to the transformed Mel frequency coefficients to produce a set of cepstral coefficients. The Mel spectrum was represented on a log scale which results in a signal in the cepstral domain with frequency peaks corresponding to the pitch on the signal. Since most of the signal information is represented by the first few MFCC coefficients, the system can be made robust by extracting only those coefficients ignoring higher-order DCT components.

1. *Mel Cepstrum*

The final result of the MFCC block process shown in Figure 3 is the coefficient of the Mel frequency cepstrum. A cepstrum representation of the speech spectrum adequately represents the local spectral characteristics of the signal for a given frame analysis.

## Convolutional Neural Networks (CNN)

Convolutional neural networks are a subset of deep learning techniques that have gained prominence in several computer vision applications and are generating attention in many different fields, including speech recognition. CNN was intended to learn spatial hierarchies of characteristics automatically and adaptively, from low to high-level patterns. CNN is a mathematical construct that is usually composed of three types of layers including convolution, pooling, and fully connected layers. Compared to the traditional hand-crafted feature extraction techniques, CNN is far more data-hungry because of its millions of learnable parameters to estimate and is more computationally expensive, resulting in requiring graphical processing units (GPUs) for model training (Yamashita, Nishio, Do, & Togashi, 2018). Figure 2.4 shows a general view of how layers are connected inside a CNN architecture.

Bar chart

Description automatically generated

Figure 2.4: CNN Architecture.

1. *Convolution*

Convolution is a special type of linear operation used in feature extraction, where small numerical arrays (kernels) are applied to the input. This is an array of numbers called a tensor. The element-wise product between each element of the kernel and the input tensor is computed at each position of the tensor and summed to get the output value at the corresponding position of the output tensor, called a feature map, depicted in Figure 2.5. This process is repeated by applying multiple kernels to form any number of feature maps representing different properties of the input tensor. Therefore, different kernels can be viewed as different feature extractors. Two important hyperparameters that define the convolution operation are the size and number of kernels.

Chart, scatter chart

Description automatically generated

Figure 2.5: 2×2 Convolution Filter.

1. *Activation Function*

The activation function is the node that is added at the end of each output of the neural network. In the CNN architecture, the activation function is the final calculation of the feature map output, or the generation of feature patterns after the convolution or merging calculation process. Although smooth nonlinear functions like the *sigmoid* or *hyperbolic tangent* (tanh) function have been employed in the past because they are mathematical representations of the behavior of biological neurons, the *rectified linear unit* (ReLU) is currently the most widely utilized nonlinear activation function, which simply computes the function in Equation 2.4 as follows:

|  |  |
| --- | --- |
|  | (2.4) |

Where:

* is the output of the function.
* is the input to the function.

1. *Max Pooling*

A pooling layer offers a standard down-sampling method that lowers the feature map's in-plane dimensions to introduce translation invariance to slight shifts and distortions and limit the number of ensuing learnable parameters. One of the most popular types of pooling operations is max pooling. The idea behind max pooling is that it preserves the most important information from the input while discarding less important information. This can be particularly useful for classification tasks, where the max pooling layer can help the model focus on the most important features in an audio, such as spectral peaks, spectral roll-off points, and spectral flux. Figure 2.6 shows an example of a max pooling with filter on a feature map.

Table

Description automatically generated

Figure 2.6: Max Pooling Layer.

1. *Fully Connected Layer*

Feature maps generated from the feature extraction layers are still in the form of a multidimensional array. Therefore, these feature maps are typically flattened, or converted into a one-dimensional array of vectors, and connected to one or more fully connected layers, also known as dense layers, in which each input is connected to their outputs by learnable weight resulting in probabilities for each class in the classification tasks. After passing through the fully connected layers, the final layer uses the SoftMax activation function that normalizes real values output from the last fully connected layer to get probabilities of the input being in a particular class (classification) where each value ranges between 0 and 1. The final fully connected layer usually has as many output nodes as there are classes.

## Transformer

The transformer is a deep learning model architecture that is built entirely on the self-attention mechanism to weigh the importance of each part of the input data differently. It is mainly used in the fields of natural language processing (NLP). This architecture is designed to process sequential input data to solve NLP-related tasks such as text translation or summarization. However, unlike Recurrent Networks (GRU, LSTM), transformers could process the entire input at once. Attention mechanisms provide context for each position in the input sequence which allows for more parallelization than recurrent neural networks and therefore reduces training time. The model of the transformer architecture follows the overall architecture of Figure 2.7 using stacked self-attention and pointwise fully connected layers for both the encoder and decoder shown in the left and right halves of the figure respectively (Vaswani, et al., 2017).

1. *Self-Attention*

In artificial neural networks, attention is a technique designed to mimic cognitive attention. This effect improves some parts of the input data and reduces others. The motivation for this is that networks need to pay more attention to small but important pieces of data. Learning which parts of the data are more important than others is context-dependent, which is trained by gradient descent. Attention functions can be described as associating a query and a set of key-value pairs with an output. Where query, key, value, and output are all vectors. The output is computed as a weighted sum of the values. The weight assigned to each value is calculated by the query compatibility function using the appropriate key.

Self-attention, also called intra-attention, is an attention mechanism that associates different positions of a single sequence to compute representations of the same sequence. In a self-attention mechanism, each element in the input is represented as a vector, and the model learns a set of attention weights that determine how much importance each element should be given when producing the output. The attention weights are learned through training and allow the model to selectively focus on certain parts of the input while ignoring others. Self-attention has proven especially useful for machine reading, summarizing summaries, or generating image descriptions.

Diagram

Description automatically generated

Figure 2.7: Transformer Architecture (Vaswani, et al., 2017)*.*

1. *Multi-Head Self Attention*

In Transformer, the Attention module iterates its computation several times in parallel. Each of them is called an attention head. The Attention module splits its query, key, and value parameters N times, passing each split individually through a separate head. All these similar attention calculations are combined to produce a final attention score. This is called multi-headed attention and gives the Transformer greater power to encode multiple relationships and nuances for each word. Multi-head attention allows the model to jointly pay attention to information from different representational subspaces at different positions. In most general form, the multi-head attention mechanism can be represented as shown in Equation 2.5. Figure 2.8shows that a multi-head attention consists of several attention layers running in parallel.

|  |  |
| --- | --- |
|  | (2.5) |

Where:

* are matrices of queries, keys, and values respectively.
* are the attention maps computed by the different attention heads.
* is a learned projection matrix.
* is a function that concatenates the attention maps along the second dimension.

Each attention head computes an attention map using Equation 2.6 below:

|  |  |
| --- | --- |
|  | (2.6) |

Where:

* are learned projection matrices for the attention head.

Diagram

Description automatically generated

Figure 2.8: Multi-Head Attention (Vaswani, et al., 2017)*.*

1. *Scaled Dot-Product Attention*

Transformers implement scaled dot product attention depicted in Figure 2.9, that follows the steps of the general attention mechanism. Scaled dot product attention first computes the dot product of each query and every key. Then divide each result by and apply the softmax function. In doing so, it obtains the weights that are used to scale the values. The formula for scaled dot product attention was defined below in Equation 2.7 as:

|  |  |
| --- | --- |
|  | (2.7) |

Where:

* are matrices of queries, keys, and values respectively.
* is the dot product of the queries and keys.
* is the dimensionality of the keys.
* is the SoftMax function, which normalizes the attention weights.

In practice, the computations performed by scaled dot product attention can be efficiently applied to the entire set of queries at once. For this purpose, the matrices are supplied as inputs to the attention function. The scaling factor is included to help stabilize the attention weights and improve the numerical stability of the model.

Diagram

Description automatically generated

Figure 2.9: Scaled Dot-Product Attention (Vaswani, et al., 2017)*.*

1. *Encoder*

Figure 2.10 shows an encoder block’s two main components: the self-attention mechanism and a feed-forward neural network. The self-attention mechanism accepts an input encoding from previous encoders and weighs their relevance against each other to produce an output encoding. Then, a feed-forward neural network processes each output code independently. These output encodings are passed as inputs to the following encoders as well as the decoders block. Each sub-layer employs a residual connection and normalization layer.

Diagram

Description automatically generated

Figure 2.10: Encoder Block (KiKaBeN, 2021)*.*

1. *Decoder*

The decoder block takes the encoder's two main components of a self-attention mechanism and a feed-forward neural network and inserts a third sub-layer that performs multi-head attention over the output of the encoder stack, shown in Figure 2.11. This new sub-layer obtains relevant information from the encoding produced by the encoder block. Like the encoder block, each sub-layer employs a residual connection and a normalization layer.

Diagram

Description automatically generated

Figure 2.11: Decoder Block (KiKaBeN, 2021).

## PyTorch

PyTorch is an open-source machine learning framework based on the Python programming language and the torch library. It is developed primarily by the Meta AI research team and can be used in both Python and C++ programming languages. However, this framework works best with Python. Over 200 and more different mathematical operations are supported by the PyTorch framework and its popularity is still growing because it makes building models for artificial neural networks simpler. Researchers primarily utilize PyTorch for research and applications using artificial intelligence (AI).

Because of the pythonic nature of this framework, PyTorch is able to utilize core python concepts such as classes, structures, and conditional loops making it easy and intuitive to understand. PyTorch is also popular for its dynamic computation graphs, which allow greater flexibility in building complex architectures. This allows neural network developers and scientists to run and test pieces of code in real-time, rather than waiting for the entire program to be written (Paszke, et al., 2019).

## Support Vector Machine (SVM)

Support Vector Machines (SVMs) are a type of supervised learning algorithm used for classification and regression analysis (Vapnik & Cortes, 1995). Introduced by Vapnik and Cortes in the 1995 as a binary classifier that could solve two-group classification problems with high accuracy, SVMs have gained popularity due to their ability to handle both linearly and non-linearly separable datasets and their effectiveness in high-dimensional feature spaces.

SVMs work by finding the hyperplane that maximally separates the two classes in the input data. The hyperplane is selected based on the margin, which is the distance between the hyperplane and the closest data points from each class. The SVM algorithm aims to find the hyperplane that maximizes this margin. The data points closest to the hyperplane are called support vectors, and they are used to define the hyperplane. As SVMs evolved, two main techniques were proposed to enable their use for multi-class classification in a one-vs-all and one-vs-one method (Duan, Rajapakse, & Nguyen, 2007).

In one-vs-all classification, each class is treated as a binary classification problem. A separate SVM model is trained for each class, where the samples of that class are assigned a positive label, and all other samples are assigned a negative label. In one-vs-one classification, all possible pairs of classes are created, and a separate SVM model is trained for each pair. During testing, each sample is classified by each SVM model, and the class with the most votes is assigned to the sample.

Both methods have their advantages and disadvantages. one-vs-all is more straightforward to implement and can handle imbalanced datasets. However, it may not be as accurate as one-vs-one, particularly when the number of classes is large. one-vs-one is more accurate and can handle overlapping classes, but it requires training a large number of SVM models, making it more computationally expensive.

# Chapter III

**Methodology**

This chapter will provide an overview of the proposed method for our study, including the tools and techniques that will be used, as well as plans for implementation and testing.

## Designed Method

This section provides a summary of the proposed architectural model's functionality and includes a diagram (Figure 3.1) that gives an overview of the model's architecture.

Graphical user interface

Description automatically generated with low confidence

Figure 3.1: Model Architecture.

The CNN architecture in this study is based on recent advancements in image and sequence processing. It includes a series of convolutional and pooling layers, similar to the classic LeNet architecture (LeCun, Bottou, Bengio, & Haffner, 1998), which extract features from the input data and reduce the size of the feature maps through downsampling. The fully-connected layers then process the extracted features to produce the final output of the network, which is transformed into a probability distribution over the possible classes using the SoftMax function. While LeNet is a relatively simple architecture compared to modern CNNs, it has been successful in many classification tasks and has been applied in various domains such as handwritten digit recognition.

The Transformer architecture is precisely as (Vaswani, et al., 2017). However, in this study, only the encoder blocks are employed which is a component of the Transformer architecture that was introduced in the paper. It is used to process the input sequence and extract relevant features that will be passed to the fully-connected layers, which process these features to produce the final output of the network.

The encoder block consists of a self-attention layer followed by a feedforward layer. The self-attention layer uses a dot-product attention mechanism to calculate the attention weights between each pair of input elements. These weights are then used to compute a weighted sum of the input elements, which is used as the output of the self-attention layer.

The feedforward layer consists of two linear transformations with a ReLU activation function in between. It takes the output of the self-attention layer as input and produces the final output of the encoder block.

The success in the use of the parallel deep learning technique of GoogleNet (Szegedy, et al., 2015), also known as Inception-v1, was the inspiration for the parallel architecture of this study, which allows the network to process multiple features concurrently. This could be achieved by using a series of inception modules, which will be concatenated and fed into the fully-connected (dense) layer. This parallel architecture enables GoogleNet to achieve good performance while being relatively efficient in terms of the number of parameters and computation time. It has been widely used in many image classification and object detection tasks.

## Supporting Tools

In order to carry out this study, certain tools and equipment will be needed, including both hardware and software. The specific devices that will be used in this research are listed below:

## Hardware

The hardware necessary for this study includes:

Lenovo Legion 5 2021 Laptop with the following specifications:

* 1. AMD Ryzen 7 5800H (8 cores / 3.20GHz)
  2. NVIDIA RTX 3070 Laptop GPU
  3. 16GB of Random Access Memory (3200MHz)
  4. 1TB Solid State Drive (SSD)

## Software

To ensure that the proposed model in this study performs correctly, certain software tools will be utilized to support this research. The software that will be used in this study includes:

1. Operating System: Windows 11
2. Programming Language: Python 3.10.9
3. Editor: Jupyter Notebook
4. Framework: PyTorch 1.13.1

## Implementation and Trial Plans

This section will explain the dataset used in the study, as well as the stages of model and user interface implementation for the proposed method, including pseudocode and explanations. The evaluation metrics used to assess the performance of the proposed method will also be described.

## Dataset

There are two English audio datasets used in the experiment, namely the CREMA-D (Cao, et al., 2014) and RAVDESS (Livingstone & Russo, 2018) datasets. The CREMA-D (Crowd-Sourced Emotional Multimodal Actors Dataset) dataset is a large collection of audio and visual data that was created to aid research in the field of audio-visual scene understanding. The dataset contains 7,442 clips of audio and video recordings of various everyday scenarios, such as people talking, laughing, and singing in various environments.

The CREMA-D dataset works with 91 actors (48 male and 43 female) between the ages of 20 and 74 coming from a variety of races and ethnicities (African American, Asian, Caucasian, Hispanic, and Unspecified). The actors spoke from a selection of 12 sentences. The sentences were presented using one of six different emotions (anger, disgust, fear, happy, neutral, and sad) and four different emotion levels (low, medium, high, and unspecified).

The recordings were made in a variety of environments, including homes, offices, parks, and streets. The dataset also includes a wide range of different people, including individuals of different ages, genders, and ethnicities. This diversity makes the dataset particularly useful for training models that can generalize well to real-world scenarios. Additionally, each clip in the dataset is accompanied by detailed annotations that describe the audio and visual content of the clip, as well as information about the people and objects present in the scene. These annotations make it possible for researchers to use the dataset for a wide range of different tasks, such as speech recognition, object detection, and facial recognition.

The Ryerson Audio-Visual Database of Emotional Speech and Song (RAVDESS) is another publicly available dataset that contains emotional speech recordings. This dataset consists of 1,470 recordings from 24 professional actors, comprising 50% male and 50% female. The actors were asked to portray different emotions such as calm, happy, sad, angry, fearful, and surprised. One of the key strengths of the RAVDESS dataset is its balanced representation of male and female voices. This is important because previous studies have shown that there are gender differences in the perception and expression of emotions. By including an equal number of male and female actors, the RAVDESS dataset provides a more representative sample of emotional expressions across genders. In addition, the RAVDESS dataset also includes recordings of speech in both neutral and accented English. An accent can also affect the perception and expression of emotions. By including accented English recordings, the RAVDESS dataset provides a more diverse set of emotional expressions that better represent real-world situations where emotions are expressed across different accents and cultures.

The RAVDESS dataset has been used in various emotion recognition, speech synthesis, and speech analysis tasks. For example, researchers have used the RAVDESS dataset to train deep neural networks for emotion recognition tasks, achieving high accuracy rates. The dataset has also been used to generate emotional speech using text-to-speech synthesis models. In addition, the RAVDESS dataset has been used to analyze the acoustic features of emotional speech, such as pitch, intensity, and formants.

Overall, both the CREMA-D and RAVDESS datasets is an extremely valuable resources for researchers in the field of audio-visual scene understanding. They provided a diverse set of emotional expressions, high-quality annotations, and supporting resources making it well-suited for a wide range of different research tasks and promising to be a valuable tool in advancing the field.

## Model Implementation

This section will outline the steps involved in conducting the research and explain the progression of the study from beginning to end. The various stages of the research are depicted in Figure 3.1, which provides a visual representation of the proposed method’s model architecture flow.

1. *Pre-Processing*

The initial stage in this research is by preprocessing the audio data from the datasets. Preprocessing is a critical step in audio data analysis that involves transforming raw audio signals into a format that is suitable for machine learning algorithms to extract meaningful information. Audio data can be complex, containing a wide range of frequencies, background noise, and other variations that can make it challenging to identify and extract relevant features. Preprocessing involves several steps that help to clean and transform the raw audio data into a format that is more amenable to analysis. In the case of audio emotion recognition, preprocessing is particularly important. It involves transforming the raw audio signals into a format that can be used to train a machine-learning model to recognize different emotions based on the acoustic properties of the speech. This involves a series of steps, including downsampling the audio to a custom target sample rate, truncating the audio to a set number of duration, and removing any silence before the actors start talking. These steps help to ensure consistency in the audio data and remove any irrelevant noise or silence that may affect the accuracy of the emotion recognition model. Algorithm 3.1 presents a pseudocode outlining the audio pre-processing steps used in this experiment.

|  |
| --- |
| Algorithm 3.1: Pre-Processing. |
| **Input:**   1. target\_sample\_rate. 2. target\_time\_duration. 3. offset\_value. 4. label\_mapping.   **Output:**   1. preprocessed audio waveforms.   **Algorithm:**   1. Define function to load and process audio files in the given directory. 2. Load each audio file in the directory using librosa.load function and apply preprocessing. This includes downsampling to the custom target sample rate, truncating to target duration value, and removing silence before the speech starts using the defined offset. 3. Map the emotion labels to numerical values using the defined label mapping. 4. Split the preprocessed audio data with an 80:20 ratio of training and testing data respectively. 5. Return the preprocessed audio data in the form of a tuple containing the training and testing sets. |

To preprocess the audio data, the audio files will be loaded using the librosa.load() function with a custom target sample rate and a fixed duration. This custom sample rate was chosen to standardize the sampling frequency of the audio files and make it consistent with the default sample rate used by most deep learning frameworks. In addition, an offset will be applied to the audio files to remove any silence before the actors start speaking. This is to ensure that only the emotional speech segments were retained for analysis and to avoid any unnecessary background noise or silence that might interfere with the emotion recognition process. Once the audio files were loaded, they were split into training and testing sets using an 80/20 split evenly across emotions. The training set was used to train the emotion recognition model, while the testing set was used to evaluate the model's performance on unseen data. This split was done randomly to ensure that the training and testing sets were representative of the entire dataset and avoid any bias in the model's performance. The emotion labels will be mapped numerically into six different categories for training using a mapping of {'angry': 0, 'fear': 1, 'disgust': 2, 'happy': 3, 'neutral': 4, 'sad': 5}. This mapping was done to convert the categorical labels into a format that can be used by the machine learning model for training and evaluation.

1. *Feature Extraction*

The next stage after preprocessing is to extract the audio features from the input data using the Mel-Frequency Cepstral Coefficients (MFCC). MFCC is a popular feature extraction technique used in speech and audio processing because it is able to capture the spectral characteristics of an audio signal in a compact and efficient manner. MFCCs are derived from the power spectrum of an audio signal and are based on the Mel-scale, which is a non-linear scale that is based on the perceived frequency of a sound by the human ear. This makes MFCCs well-suited for tasks such as speech recognition and speaker identification, where the human ear is the primary means of perception. Algorithm 3.1 shows a pseudocode for extracting MFCC features from an audio signal using the librosa library.

|  |
| --- |
| Algorithm 3.2: MFCC Feature Extraction. |
| **Input:**   1. n\_mfcc. 2. mels. 3. window\_size. 4. hop\_length. 5. audio\_data.   **Output:**   1. Mel frequency cepstrum coefficient (MFC Coefficient).   **Algorithm:**   1. Compute the short-time Fourier transform (STFT) of the audio waveform using the specified window size and hop length. 2. Compute the magnitude spectrogram of the STFT. 3. Apply a mel filterbank to the magnitude spectrogram. 4. Convert the mel spectrogram to decibel (dB) units. 5. Compute the discrete cosine transform (DCT) of the log-mel spectrogram. 6. Return the resulting MFCC coefficients as a feature matrix for further analysis and emotion recognition. |

The application employs the librosa library to retrieve the audio signal from a file to extract the Mel-Frequency Cepstral Coefficients (MFCC) from an audio waveform that involves several sequential stages. The first step is to compute the Short-Time Fourier Transform (STFT) of the audio waveform using a specified window size and hop length. This step involves dividing the audio signal into short frames of equal length, with a hop length of about half of the frame size. The Hanning windowing function is typically used for each audio frame, which helps to reduce spectral leakage and improve the frequency resolution of the STFT. The next step is to compute the magnitude spectrogram of the STFT. The magnitude spectrogram is then transformed using a Mel filter bank, which approximates the frequency response of the human auditory system. After applying the filter bank, the resulting Mel spectrogram is converted to decibel (dB) units using a logarithmic scale. This step is necessary to compress the magnitude values and better represent the relative loudness of each frequency bin. Finally, the MFCC coefficients are computed by performing a Discrete Cosine Transform (DCT) of the log-mel spectrogram. This step involves transforming the Mel spectrogram into a sequence of cepstral coefficients that represent the spectral envelope of the signal.

1. *Model Architecture Design*

After the audio features have been extracted, the next step is to create the model architecture and use the extracted features as input to the model which will allow the model to classify human emotions based on the extracted features. There are two blocks of the deep learning model for the purposed method, the CNN block and the Transformer block which will be working in parallel with each other. The idea is for the CNN to give spatial feature representation of the input data, and the Transformer block in sequence encoding to try and model as accurately as possible the temporal relationships between pitch transitions in human emotions. The expansion of CNN filter channels and reduction of feature maps will provide the most expressive feature representation with the lowest computational cost, while the Transformer encoder will learn to predict frequency distributions of different emotions according to the global structure of the MFCC plot of each emotion. The implementation for CNN and Transformer block will be shown in Algorithm 3.2 and Algorithm 3.3, respectively.

|  |
| --- |
| Algorithm 3.3: CNN Block. |
| **Input:**   1. input audio tensor.   **Output:**   1. convolution Embedding.   **Algorithm:**   1. Define the CNN model architecture by initializing the sequential model. 2. Add the convolutional layers to the model and specify the number of filters, filter sizes, and activation function for each layer. 3. Add batch normalization before max pooling to normalize the inputs of each layer. 4. Add a max pooling layer to the model to reduce the dimensionality of the output. 5. Add a flattening layer to the model to convert the 2D matrix output into a 1D vector. 6. Return the convolution embedding from the output of the Flatten layer. |

Algorithm 3.2 presents an implementation of the CNN Block for the proposed deep learning model. The first layer is the input layer takes in audio features with a certain size and number of channels. The second layer is the convolution layer that applies a set of filters to this input data, generating a set of feature maps. In this study, the filters are 3x3 matrices and they are applied to the input data through a process called convolution. The third layer is the batch normalization layer which normalizes the output from the convolution layer, making the model more stable and efficient. Fourth is the activation layer which applies an activation function (i.e., ReLU) to the feature maps generated by the previous layer. This layer introduces non-linearity to the model, allowing it to learn more complex relationships in the data. Finally, the pooling layer down-samples the feature maps by applying a pooling operation (i.e., MaxPooling). This helps reduce the size of the feature maps and, as a result, lowers the computational complexity of the model. This sequence of applying convolutional layers, batch normalization layers, activation layers, and pooling layers is repeated three times in the proposed model architecture.

The Transformer encoder implementation was shown in Algorithm 3.3 which is designed to process sequential data of the audio source. It consists of a series of self-attention layers and feedforward layers, which are used to predict frequency distributions of different emotions according to the global structure of the MFCCs of each emotion. In the implementation, the output sequence is initialized first as an empty list. Then the input sequence is embedded by applying an embedding matrix to each element of the input, resulting in a sequence of embedded vectors. The embedded sequence will then be passed through a series of self-attention layers to compute a sequence of context-aware representations. Each self-attention layer applies the attention mechanism to the input sequence to compute a weighted sum of the input vectors, where the weights are computed based on the relationships between the input elements. These context-aware representations are then passed through a series of feedforward layers to compute a sequence of transformed representations. Each feedforward layer applies a linear transformation to the input, followed by a nonlinear activation function. Finally, the transformed representations are used to compute the output sequence by applying a linear transformation and an activation function to each element of the transformed representations.

|  |
| --- |
| Algorithm 3.4: Transformer Encoder Block. |
| **Input:**   1. input audio tensor.   **Output:**   1. transformer encoding embedding.   **Algorithm:**   1. Define transformer encoder block by initializing the transformer encoder layer. 2. Add a multi-head attention layer with a specified number of heads, hidden dimension, and dropout rate. 3. Add a layer normalization layer after the multi-head attention layer. 4. Add a feedforward neural network layer with a specified number of hidden units and activation function. 5. Add another layer normalization layer after the feedforward neural network layer. 6. Return the transformer encoding embedding from the output of the final layer normalization layer. |

The final stage of the purposed model is to concatenate both outputs from the CNN model and the Transformer encoder model and pass the resulting tensor through a dense layer with a softmax activation function for prediction. The softmax function is a common choice for the activation function in the final layer of a classification model. It takes a vector of arbitrary real-valued scores and converts it into a probability distribution, where the probability of each class is given by the corresponding element in the output vector. Algorithm 3.4 outlines the process of combining the outputs of the CNN and Transformer models, passing them through a dense layer, and applying the softmax function to the output of the dense layer to make predictions.

|  |
| --- |
| Algorithm 3.5: Dense Layer Concatination. |
| **Input:**   1. cnn embedding input. 2. transformer encoding embedding input. 3. number of classes.   **Output:**   1. class prediction probabilities.   **Algorithm:**   1. Concatenate the CNN and Transformer Encoding embeddings along the feature dimension to create a joint embeddings. 2. Pass the joined embeddings through a dense layer to combine the features. 3. Apply a softmax activation function to the output of the dense layer to obtain class probabilities. 4. Return the class prediction probabilities. |

The dense layer implementation of Algorithm 3.4 starts by combining the output tensors of CNN and Transformer which has the shape (batch\_size, feature\_size) for each of the models, respectively. The combined output is then passed through a linear layer with the number of class units, which is the six different emotional states in the dataset. The dense layer has weights and biases that will be learned during training to transform the combined output into the final prediction. Finally, the SoftMax activation function is applied to the output of the linear layer that will convert the prediction scores into a probability distribution over the classes. The class with the highest probability is taken as the model's prediction.

## Model Evaluation

Model evaluation is an important step in the development of a deep learning model, as it could assess the performance of the model on unseen data and determine its suitability for a given task. This section will outline some general considerations for evaluating deep learning models for a speech emotion recognition task.

There are several ways to evaluate the performance of the purposed deep learning model. This study aims to compare the performance of three different machine learning models on a speech emotion recognition task, the standard Convolution Neural Network (LeNet) model, the Support Vector Machine (SVM) model, and the purposed method for this study. A combination of different evaluation metrics will be used to evaluate the performance of these models.

First, the model's training and validation accuracy will be tracked to ensure that the model is not overfitting the training data. Tracking the training process of a model could help identify and address the overfitting and underfitting in the data. Overfitting occurs when the model performs well on the training data but poorly on the validation or test data, indicating that it has learned patterns that are specific to the training data and are not generalizable. While underfitting occurs when the model performs poorly on both the training and validation data, indicating that it is not able to learn the underlying patterns in the data. In addition to addressing overfitting and underfitting, tracking the model’s accuracy on the validation set can also help choose the best hyperparameter values leading to the best model performance by observing and changing the effect on the validation accuracy. In summary, this method will show how well the models can learn the classification task and identify any issues with the optimization process. After training the model, the test set will be used to evaluate their performance using several metrics.

One of the best metrics to evaluate is the test set accuracy for each model to get an idea of how the model performs on unseen data. This metric will give a summary of the model's performance on the test set. The test set is a set of data that the model has not seen during training, and therefore provides a more realistic evaluation of the model's performance. Evaluating the model on the test set accuracy can give a more accurate assessment of the model’s generalization ability, which is its ability to perform well on unseen data. This is essential for understanding the model's suitability for deployment and its potential real-world performance.

In addition to accuracy, a confusion matrix can also be used for each model to understand the types of errors that these models are making and to identify any imbalances in the data. The confusion matrix is a table that shows the number of true positive, true negative, false positive, and false negative predictions made by the model. True positive predictions are those where the model correctly predicts the positive class, while true negative predictions are those where the model correctly predicts the negative class. False positive predictions are those where the model incorrectly predicts the positive class, while false negative predictions are those where the model incorrectly predicts the negative class. This evaluation metric will be able to see how well the models are performing in each emotion class and a more detailed understanding of the model’s strengths and weaknesses.

Other metrics such as precision, recall, and the F1 score can be used on the test set for each model. These evaluation metrics are commonly used to assess the performance of deep learning models, particularly for classification tasks. Precision measures the proportion of true positive predictions made by the model among all positive predictions, while recall measures the proportion of true positive predictions made by the model among all actual positive examples. The F1 score is a combination of precision and recall and is calculated as the harmonic mean of the two. The F1 score is useful because it takes into account both the precision and recall of the model and provides a single metric that reflects the model's overall performance. These metrics of evaluation give a detailed understanding of the model's performance and compare them to each of the six different emotion classes in the CREMA-D dataset.

Lastly, the weighted average of the F1 scores could be considered as one of the metrics of evaluation for this study. The weighted average of the F1 score is a variation of the F1 score that is used to evaluate the performance of a classification model when dealing with imbalanced classes. In an imbalanced dataset, the classes are not equally represented, which can make it difficult to accurately evaluate the model's performance. The weighted average helps address the issue by adding weights in different classes in the calculation of the F1 score. These weights reflect the relative importance of the different classes and can be used to give more emphasis to the performance of the model on a particular class. This metric is useful for evaluating the model’s performance with imbalance classes and can be used to compare the overall performance of the three models for the speech emotion recognition task.

## User Interface

This study provides a way to share the proposed machine learning model for this study using an interactive web-based application that can be used on any device with browser support. This section will explain about the proposed design for the web-based user interface application which will be discussed in detail, highlighting the features and functionalities that will be included in the application. Figure 3.2 and Figure 3.3 provides an overview of the proposed design for the web-based user interface application.



Figure 3.2: Web Design Pre-Audio Input.

A screenshot of a computer

Description automatically generated with medium confidence

Figure 3.3: Web Design Post-Audio Input.

The proposed design for this web-based application is intended to make the application user-friendly and easy to navigate, allowing users to quickly and easily access the features and test the proposed machine learning model for this study. Additionally, users can also view the model's performance through charts displayed in the output section of the website depicted in Figure 3.3, which shows the probability distribution of emotions classification. Overall, the goal of the proposed design is to create an efficient and effective user experience for the web-based application, specifically for showcasing the proposed machine learning model’s performance. Table 3.1 include a detailed explanation that will provide a comprehensive overview of how each feature works in order to give a clear understanding of the functionalities of this web-based application.

Table 3.1: Web-based Application Feature Functionality.

|  |  |
| --- | --- |
| **Feature** | **Description** |
| A screenshot of a computer  Description automatically generated with medium confidence | This is the input section; Users can choose to upload their audio recording in wav format or use the included recording examples. |
| A screenshot of a computer  Description automatically generated with medium confidence | Once an audio recording is loaded, a couple of features can be used, such as:   * Audio playback; * Volume control; * Snipping tool; |
| A screenshot of a computer  Description automatically generated with medium confidence | The output section will give the emotion classification based on the imported audio file along with a classification label of the probability distribution scores for each emotion. |

# Chapter IV

**Result and Discussion**

This chapter will discuss the implementations and the results achieved during this research. The performance and effectiveness of each machine learning model will be assessed and presented in achieving its intended objectives. The analysis will provide valuable insights into the challenges and successes encountered during the implementation process, and offer recommendations for further improvement and development.

## Implementation

An overview of the designed method for the audio emotion recognition system was provided in Figure 3.1. The system was implemented using the Python programming language version 3.10.9. The designed method involves the use of various external libraries, such as Librosa, PyTorch, and Scikit-Learn, which were mainly used for the main components of the system.

## Data Exploration

The dataset in this study consists of English human voices that have already been labelled. Two distinct datasets were utilized, namely CREMA-D and RAVDESS, both of which contain single emotional labels for each sound sample.

The Crowd-Sourced Emotional Multimodal Actors Dataset (CREMA-D) comprises approximately 7,442 clips of 91 actors, with an equal distribution of males and females. Each actor was instructed to act out 6 different emotions, including anger, disgust, fear, happiness, neutral, and sad. The clips vary in duration, ranging from 1 to 5 seconds and are recorded at a sampling rate of 16kHz with a resolution of 16 bits. The emotional state of each clip was labelled by multiple crowd-workers on Amazon Mechanical Turk, ensuring a diverse set of labels that represent the emotions expressed in each clip.

The Ryerson Audio-Visual Database of Emotional Speech and Song (RAVDESS) dataset consists of approximately 1,440 audio files of actors performing scripted speech and song segments. The audio files are recorded at a sampling rate of 48kHz with a resolution of 16 bits and are labelled with one of eight emotions, including calm, happy, sad, and angry, amongst others. The actors are of diverse ages, genders, and ethnicities, ensuring that the dataset is representative of a wide range of voices and accents.

This study utilized the emotions present in the CREMA-D dataset, which consists of 6 different emotion classes including anger, disgust, fear, happiness, neutral, and sad. Using a standardized set of emotions across the dataset allows for a fair comparison of performance between different models and techniques. The use of the same emotion classes across all datasets helps to eliminate any potential biases that may have resulted from variations in labelling, recording quality, or other factors that may impact the effectiveness of emotion recognition models. Overall, by using a standardized set of emotions, the study aims to provide a more accurate and reliable analysis of the performance of different models and techniques for audio emotion recognition.

As part of the exploratory data analysis, amplitude and spectrogram plots were generated for both of these datasets. An amplitude plot shows the variation of sound pressure over time and is useful for identifying the overall loudness of a recording. The spectrogram plot, on the other hand, shows how the energy of the signal is distributed across different frequencies over time and can help identify specific characteristics of the audio signal. Table 4.1 presented exemplars of amplitude and spectrogram plots for each of the six emotion classes in the CREMA-D dataset, whereas Table 4.2 displayed the amplitude and spectrogram plots for the RAVDESS dataset, also for each of the six emotion classes.

Table 4.1 CREMA-D Amplitude and Spectogram.

|  |  |  |
| --- | --- | --- |
| **Emotion** | **CREMA-D** | |
| **Amplitude** | **Spectogram** |
| Angry |  |  |
| Fear |  |  |
| Disgust |  |  |
| Happy |  |  |
| Neutral |  |  |
| Sad |  |  |

Sound characteristics of different emotions in the CREMA-D dataset are shown in Table 4.1. The amplitude plot of the dataset reveals that each emotion class has distinct sound characteristics. For example, anger, fear, and joy emotions have higher amplitudes above 0.5, while disgust, neutral, and sad emotions are at lower amplitudes of approximately 0.2. This difference in amplitudes suggests that different emotions have unique patterns of sound frequencies.

Similarly, the spectrogram plot also reveals the distribution of sound frequencies for each emotion class. The distribution of each emotion is different, and several similar emotions show patterns in the CREMA-D dataset. For instance, angry and happy emotions share similar patterns, with high frequencies and a narrow band of energy concentrated around the middle of the frequency range. In contrast, disgust and sad emotions have lower frequencies and more spread-out energy distribution across the frequency range. The spectrogram plot helps to identify these subtle differences in patterns between different emotion classes.

Overall, these plots demonstrate that the use of amplitude and spectrogram plots can provide valuable insights into the unique sound characteristics of different emotions in the CREMA-D dataset.

Table 4.2 RAVDESS Amplitude and Spectogram

|  |  |  |
| --- | --- | --- |
| **Emotion** | **RAVDESS** | |
| **Amplitude** | **Spectogram** |
| Angry |  |  |
| Fear |  |  |
| Disgust |  |  |
| Happy |  |  |
| Neutral |  |  |
| Sad |  |  |

Compared to the CREMA-D dataset, the RAVDESS dataset has a much greater variety of sound characteristics shown in Table 4.2. This was particularly evident in the amplitude plots, which showed that each emotion in the RAVDESS dataset had a different range of amplitudes. The spectrograms of the RAVDESS dataset also clearly displayed distinct characteristics for each emotion. The significant differences in amplitude and spectrogram patterns for each emotion can be used as reliable indicators of emotional states.

To illustrate, a neutral emotion can be identified in an audio signal by observing a spectrogram that displays a uniform pattern in contrast to the patterns observed for other emotions. This discovery carries significance because it implies that the RAVDESS dataset could be a valuable resource for researchers investigating emotional expression in speech. The RAVDESS dataset includes a greater range of audio characteristics that could facilitate a more nuanced analysis of emotional expression, as well as increase the accuracy of emotion recognition models.

## Pre-Processing

Audio data pre-processing is a critical step in the development of any audio analysis system. It involves a series of techniques aimed at cleaning, transforming, and preparing raw audio data to make it suitable for use in machine learning algorithms. Some of the key pre-processing techniques employed in this study include loading and splitting the dataset, feature extraction, and feature scaling. By applying these techniques, machine learning models can improve the accuracy and reliability of their audio analysis systems and ensure that they can extract meaningful insights from their data.

1. *Load Data*

The data loading process was critical in preparing the audio data for use in the machine learning models. To ensure that the audio data was suitable for training the models, several data loading parameters were tested using the librosa library. The main parameters tested were the target duration, target sample rate, and offset. The target durations were set based on the minimum, maximum, and average durations of the dataset. The CREMA-D dataset has a target duration of 1.28 seconds, 5 seconds, and 2.54 seconds. While the RAVDESS dataset, has 3.07 seconds, 5.27 seconds, and 3.73 seconds, respectively.

As for the target sample rate parameter, two different values were tested, the native sample rate (16kHz for CREMA-D and 48kHz for RAVDESS) and the default sample rate from librosa, which is 22.5 kHz. This facilitated the assessment of the comparative effectiveness of various machine learning models under different data loading sample rates.

Finally, to determine the impact of removing silence or noise on the model's performance, the offset parameter was tested. The average offset value was 0.3 seconds for the CREMA-D dataset and 0.8 seconds for the RAVDESS dataset. The inclusion and exclusion of the offset parameter were tested in the models in order to determine how the removal of any silence or noise before speech in the dataset impacted model performance. The average value offsets of speech in the dataset can be observed in Table 4.1 and Table 4.2 for the CREMA-D and RAVDESS datasets, respectively.

This study evaluated the performance of various machine learning algorithms frequently employed in speech recognition for audio by means of assessment. The algorithms that were used include Support Vector Machines (SVM), LeNet's Convolutional Neural Network (CNN), RNN, and a Transformer Encoder block combined with a CNN-based architecture. The primary evaluation metric used in this study was accuracy. The accuracy of the models was compared when they were trained on different combinations of data loading parameters that were tested in this study.

Table 4.3: Model Accuracy Comparison on Librosa's Data Load Parameters.

|  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- |
| Dataset | Target Duration | Target Sample Rate | Offset | Accuracy | | | |
| SVM | LeNet | RNN | T. Encoder and CNN |
| CREMA-D | 1.28 | 16 kHz | 0 | 42.85% | 39.62% |  | 46.94% |
| CREMA-D | 1.28 | 22.5 kHz | 0 | 45.00% | 40.90% |  | 45.74% |
| CREMA-D | 1.28 | 16 kHz | 0.3 | 47.88% | 40.09% |  | 52.05% |
| CREMA-D | 5 | 16 kHz | 0 | **51.91%** | 44.46% |  | **57.02%** |
| CREMA-D | 5 | 22.5 kHz | 0 | 51.85% | 43.59% |  | 56.35% |
| CREMA-D | 5 | 16 kHz | 0.3 | 51.51% | 45.60% |  | 56.48% |
| CREMA-D | 2.54 | 16 kHz | 0 | 50.91% | 45.33% |  | 56.15% |
| CREMA-D | 2.54 | 22.5 kHz | 0 | 51.71% | 44.86% |  | 54.60% |
| CREMA-D | 2.54 | 16 kHz | 0.3 | 51.85% | **46.07%** |  | 54.13% |
| RAVDESS | 3.07 | 48 kHz | 0.8 | 69.81% | 61.32% |  | 75.47% |
| RAVDESS | 3.07 | 48 kHz | 0 | 69.81% | 63.21% |  | 73.58% |
| RAVDESS | 3.07 | 22.5 kHz | 0 | 68.87% | 60.38% |  | 74.53% |
| RAVDESS | 5.27 | 48 kHz | 0.8 | 68.87% | 64.15% |  | 70.28% |
| RAVDESS | 5.27 | 48 kHz | 0 | **72.64%** | 59.43% |  | **78.30%** |
| RAVDESS | 5.27 | 22.5 kHz | 0 | 65.09% | 61.32% |  | 71.70% |
| RAVDESS | 3.73 | 48 kHz | 0.8 | 70.75% | **66.98%** |  | 75.47% |
| RAVDESS | 3.73 | 48 kHz | 0 | 71.70% | 63.21% |  | 72.64% |
| RAVDESS | 3.73 | 22.5 kHz | 0 | 66.98% | 53.68% |  | 68.87% |

Based on the results, it is evident that data loading parameters play a crucial role in the performance of machine learning models for audio speech recognition. Table 4.3 showed that the optimal data loading parameters varied across different machine learning architectures. Specifically, the SVM and Transformer Encoder combined with CNN achieved their highest accuracy with a longer duration, a native sample rate, and no offset. On the other hand, for LeNet's CNN, an average duration, a native sample rate, and an offset proved to be the optimal parameters. Notably, these findings were consistent for both the CREMA-D and RAVDESS datasets.

The variability in optimal data loading parameters suggests that each machine learning architecture is sensitive to specific aspects of the input data. This can be attributed to the fact that each machine-learning architecture has its own unique characteristics and requirements. For example, SVM is known to perform well with high-dimensional data but may be sensitive to the length of the audio samples. On the other hand, LeNet's CNN is designed for image classification, so it may be better suited to audio samples with a shorter duration and a lower sample rate. The Transformer encoder combined with CNN, on the other hand, is a more recent architecture that has shown promising results in natural language processing tasks, but its performance on audio data may depend on the data loading parameters used. The choice of data loading parameters should, therefore, be made based on the characteristics of the dataset and the specific machine learning architecture being used.

This experiment emphasizes the importance of carefully selecting appropriate data loading parameters in the preparation of audio speech recognition datasets. The optimal data loading parameters should be chosen based on the characteristics of the dataset and the machine learning architecture being used to obtain the best performance.

1. *Split Datasets*

The process of data splitting involves partitioning a dataset into two or more subsets, with the aim of training and assessing machine learning models. During this process, a subset of the dataset is designated for training the model, while the remaining portion is reserved for testing or validation. This step plays a crucial role in model development, as it enabled the evaluation of the model's performance on previously unseen data.

This section aims to evaluate the performance of different machine learning models by testing three different data splitting ratios. The models considered in this study were Support Vector Machines (SVM), Recurrent Neural Networks (RNN), LeNet, and Transformer Encoder block combined with CNN-based architecture. The ratios tested for the SVM model include 80:20, 85:15, and 75:25, where the first number represents the percentage of data used for training, and the second number represents the percentage of data used for testing. The results of the evaluation of the SVM model were presented in Table 4.4.

In addition to the SVM model, the evaluation of RNN, LeNet, and Transformer Encoder block combined with CNN-based architecture under different data splitting ratios was presented in Table 4.5. However, unlike the SVM model, the data was split into three parts for these models, namely training, testing, and validation. Three different ratios that were tested for these models include 80:10:10, 90:5:5, and 70:15:15.

Table 4.4: Accuracy Comparison on Different Split Data Ratio for SVM.

|  |  |  |  |
| --- | --- | --- | --- |
| Dataset | Ratio (%) | | Accuracy |
| Train | Test | SVM |
| CREMA-D | 80 (5953) | 20 (1489) | **51.91%** |
| CREMA-D | 85 (6325) | 15 (1117) | 51.75% |
| CREMA-D | 75 (5581) | 25 (1861) | 50.24% |
| RAVDESS | 80 (844) | 20 (212) | **70.75%** |
| RAVDESS | 85 (897) | 15 (159) | 70.44% |
| RAVDESS | 75 (792) | 25 (264) | 68.18% |

Table 4.5: Model Accuracy Comparison on Different Split Data Ratio.

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
| Dataset | Ratio (%) | | | Accuracy | | |
| Train | Test | Validation | LeNet | RNN | T. Encoder and CNN |
| CREMA-D | 80 (6027) | 10 (745) | 10 (670) |  |  | 60.00% |
| CREMA-D | 90 (6715) | 5 (373) | 5 (354) |  |  | **61.13%** |
| CREMA-D | 70 (5376) | 15 (1117) | 15 (949) |  |  | 57.56% |
| RAVDESS | 80 (855) | 10 (106) | 10 (95) |  |  | 78.30% |
| RAVDESS | 90 (952) | 5 (53) | 5 (51) |  |  | **81.13%** |
| RAVDESS | 70 (762) | 15 (159) | 15 (135) |  |  | 67.30% |

## Feature Extraction

## Model Development

## Implementation Result Discussion

(analisis, sintesis, dan evaluasi)

# BAB V KESIMPULAN DAN SARAN

## Kesimpulan

Berupa hasil penelitian/perancangan yang menjawab permasalahan atau yang berupa konsep, program, dan karya rancangan

## Saran (jika dianggap perlu)

berisi hal-hal yang masih dapat dikerjakan dengan lebih baik dan dapat dikembangkan lebih lanjut, atau berisi masalahmasalah yang dialami pada saat proses pengerjaan tugas/proyek akhir.

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# LAMPIRAN-LAMPIRAN ATAU APPENDIKS (jika ada)

# BIODATA PENULIS

Penulis dilahirkan di Madiun, 29 Januari 1985, merupakan anak pertama dari 4 bersaudara. Penulis telah menempuh pendidikan formal yaitu di TK ABA 18 Madiun, SDN Beteng 1 Madiun, SMPN 2 Madiun dan SMAN 2 Madiun. Setelah lulus dari SMAN tahun 2020, Penulis mengikuti SBMPTN dan diterima di Departemen Teknik Mesin FTIRS - ITS pada tahun 2020 dan terdaftar dengan NRP 02112040000130.

Di Departemen Teknik Mesin Penulis sempat aktif di beberapa kegiatan Seminar yang diselenggarakan oleh Departemen, Himpunan Mahasiswa Teknik Mesin (HMM) dan aktif sebagai Asisten Praktikum Mesin Konversi Enersi maupun Grader mata kuliah Termodinamika.