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INTRODUCTION

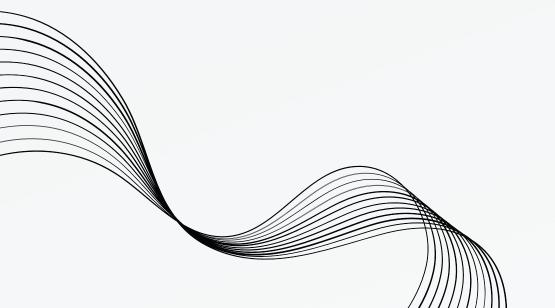
- The Music-genre classifier categorizes music tracks into predefined genres based on their audio characteristics.
- It utilizes machine learning and deep learning algorithms to automate the process traditionally performed by human experts.
- This project has two models, namely the Multi-Layer Perceptron (MLP) and Convolutional Neural Network (CNN) architectures.

OBJECTIVE

- The primary goal of our project is to automatically categorize music into distinct genres based on the inherent patterns and features within the audio signal like MFCCs lever
- Explore the effectiveness of Mel-frequency cepstral coefficients (MFCCs) and other audio features in improving the accuracy of music genre classification models.

LIBRARIES USED

- Librosa: Used for loading audio data and extracting relevant features.
- Tensorflow: Fundamental for creating and training the model.
- Sci-kit learn: Required for various data preprocessing tasks.
- Numpy: Essential for matrix calculations and numerical operations.
- Matplotlib: Necessary for visualizing and plotting the model's results.



DATASET

- Our model is based on the GTZAN Dataset Music Genre Classification dataset, which is a collection of audio clips spanning various genres, making it valuable for evaluating and training machine learning models in the field of music genre classification.
- It has 10 data labels: blues, classical, country, disco, hiphop, jazz, metal, pop, raggae and rock.
- Each class consists of 100 audio files of 30 seconds length.
- The data also has the MFCC plots of all the audio files saved for image-based processing, like CNN.

PROCESS FLOW

- 1. Load the preprocessed data
- 2. Build the model architecture
- 3. Address the overfitting issues
- 4. Compile the model
- 5. Split the data into train and test sets
- 6. Train the model
- 7. Evaluate the model
- 8. Make prediction for an audio file

DATA PREPROCESSING

- A dictionary named data is made to store the 'mapping', 'labels; and 'MFCCs' of the data.
- Each audio file is divided to 10 independent segments, each of length 3 seconds to increase the input data.
- We loop through the dataset to get the mapping, i.e, the name of the subfolder, labels as subfloder-count 1.
- The mfcc is calculated using the feature of the librosa library.
- All the mappings, labels and mfccs are then stored in the JSON file for further use.

MULTILAYER PERCEPTRON

- A Multilayer Perceptron (MLP) is a type of artificial neural network designed for supervised learning. It consists of multiple layers of nodes, including an input layer, one or more hidden layers, and an output layer.
- MLP's architecture allows it to learn and differentiate between diverse genres by capturing nuanced patterns present in the audio features, enhancing its classification capabilities.
- MLP, when applied to audio data, can effectively extract higher-level features from representations like Mel-frequency cepstral coefficients (MFCCs), contributing to the model's ability to discern audio characteristics.

IMPLEMENTATION OF MER

- The MLP model consists of an input layer, three hidden layers and one output layer.
- It is a Sequential model which is created with the help of Keras of the TensorFlow library.
- All the hidden layers use the Dense layer with 512, 256 and 64 neurons and ReLU as the activation function.
- For the output layer, it uses the softmax function.

CONVOLUTION NEURAL NETWORK

Convolutional Neural Network (CNN) is a deep learning architecture designed for image and spatial data, featuring convolutional layers that extract hierarchical representations, pooling layers for dimensionality reduction, and fully connected layers for classification.

• It can perrform better than MLP even with less parameters.

IMPLEMENTATION OF CNN

- The model comprises three convolutional layers, each employing a 3x3 grid for convolution, followed by 3x3 max-pooling, and subsequent batch normalization to enhance feature extraction.
- The resultant is then flattened and fed to a Dense layer.
- For the output layer, the model utilizes a softmax function to generate a probability distribution across multiple classes.

OVERFITTING ISSUE

- Overfitting occurs when a machine learning model learns the training data too well, capturing noise or random fluctuations and performing poorly on new, unseen data
- To prevent overfitting in both the models, we have used Dropout and L2 Regularization methods.
- Dropout introduces randomness during training by randomly deactivating a fraction
 of neurons, preventing overfitting, while L2 regularization adds a penalty term to the
 loss function to limit the magnitudes of weights in a neural network.

MODEL COMPILATION

- Model compilation is the process in deep learning where you configure the necessary settings for a neural network before training.
- This includes specifying the optimizer, loss function, and metrics that the model will use during training and evaluation.
- Our model have Adam optimizer with a learning rate of 0.0001 and is compiled with sparse categorical crossentropy as the loss function and accuracy as the evaluation metric.

PERFORMANCE

- Our models are trained on the training set using the fit method with a batch size of 32 and 30 epochs for CNN and 50 epochs for MLP.
- The model's performance is assessed on the separate test set using the evaluate method, calculating the test loss and accuracy. The printed test accuracy provides a quantitative measure of the model's generalization to new, unseen data.
- The test accuracy for MLP is 0.594125509262085.
- The test accuracy for CNN is 0.7140568494796753.

MODEL PREDICTION

- The model can predict the label for an audio file using the custom predict function.
- It takes the model, audio file input and desired label as the input parameters.
- It first adds another layer to the input and pass it through the model.predict method.
- It gives a probability distribution of the possibility of the audio data's label.
- The label with the highest probability is taken as the predicted answer.

