



ARTIFICIAL INTELLIGENCE

How can Machine Learning in Audio Analysis?

Learn how machine learning could be used to analyze generate predictions for both classification and regres

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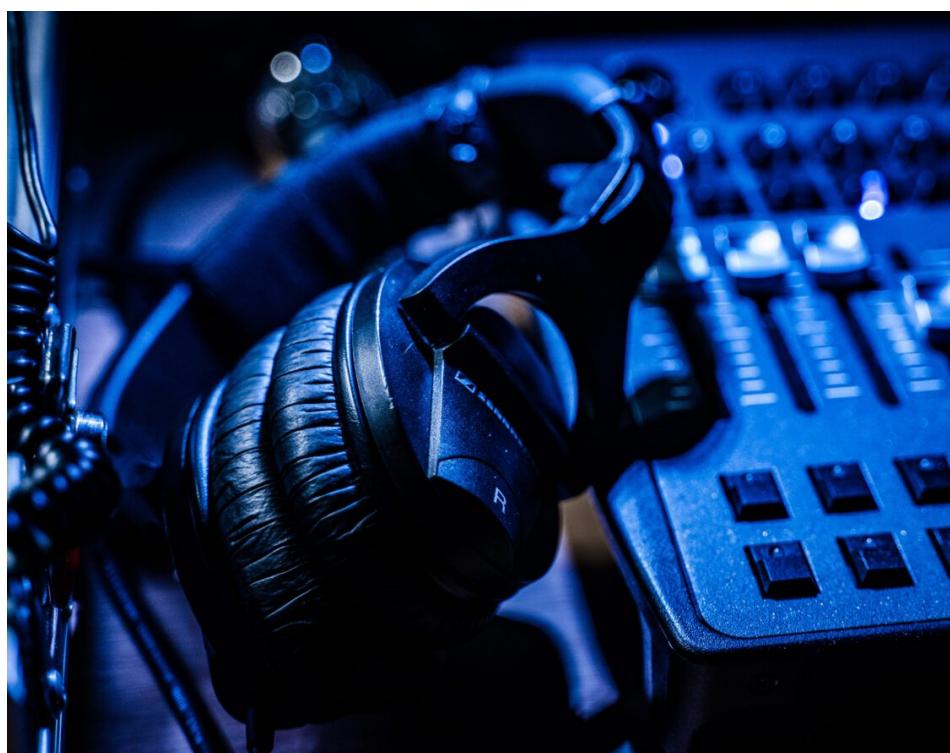


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Machine learning has been gaining rapid traction over the past decade. In fact, it is being used in numerous industries such as **healthcare**, **agriculture**, and **manufacturing**. The potential applications of machine learning are vast due to the advancement of technology and computational power. As data is available in various formats in abundance, it is now possible to use **machine learning** and **data science** to extract valuable insights from data and make predictions using them.

One of the most interesting applications of machine learning is in the field of **audio analysis** and understanding the quality of sound recordings.

formats respectively. Therefore, using various machine learning and deep learning algorithms ensures that predictions are created and understood with the **audio data**.



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Before doing audio analysis, samples of the signal are taken and analyzed individually. The rate at which taking the samples is also known as **the sampling rate**. It would be really handy to convert a time-domain signal to a frequency domain to get a good logical understanding of the signal along with computing useful components like energy. All these could be given as features to machine learning models which would use them for making predictions.

There is a popular conversion of an audio signal (image) so that it could be given to **convolutional neural networks (CNNs)** for prediction. A spectrogram captures important characteristics of an audio signal and represents them in 2D which can hence be used in deep learning based networks.

There are plenty of ML models that perform a very well in predicting the output labels if they are given an input signal. An audio signal which is composed of amplitude over different units of frequency, could also be converted and used for robust ML predictions.

In this article, we would be going over how to read an audio file by considering a random example and plotting it to understand its graphical representation. Later, we will perform feature engineering with the image data and perform convolutional operations as the audio is converted to an image. Finally, we will get **sample predictions** for unseen data. Note that this code is used for demonstration and does not take into account specific datasets.

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```
1 import matplotlib.pyplot as plt
2 import numpy as np
3 from scipy.io.wavfile import read
4
5 # Read in the audio file
6 sample_rate, audio_data = read('audio_file.wav')
7
8 # Convert the audio data to a numpy array
9 audio_data = np.array(audio_data)
10
11 # Get the length of the audio data
12 length = audio_data.shape[0] / sample_rate
13
14 # Create a time axis
15 time = np.linspace(0., length, audio_data.shape[0])
16
17 # Plot the audio data
18 plt.plot(time, audio_data)
19 plt.xlabel('Time (s)')
20 plt.ylabel('Amplitude')
21 plt.show()
```

[audio_read.py](#) hosted with ❤ by GitHub

We are going to import necessary libraries that are required for reading an audio file in the form that is mostly popular – **.wav** format. After reading the file, we are getting its graphical representation as shown in the code cell above. Now let's move on to feature engineering by plotting the output just to see how it looks like.

Feature Engineering

```
1 import librosa
2
3 # Calculate the short-term Fourier transform (STFT) of the audio
4 stft = librosa.stft(audio_data)
5
6 # Extract the magnitude and phase of the STFT
7 magnitude, phase = librosa.magphase(stft)
```

```
8
9 # Calculate the mel-scaled spectrogram of the audio data
10 mel_spec = librosa.feature.melspectrogram(audio_data, sr=sample_rate)
11
12 # Extract the Mel-frequency cepstral coefficients (MFCCs) from the mel-scaled spectrogram
13 mfccs = librosa.feature.mfcc(S=librosa.power_to_db(mel_spec), n_mfcc=20)
14
15 # Transpose the MFCCs so that each row represents a time frame and each column represents a
16 mfccs = mfccs.T
```

[audio_feature_engineer.py](#) hosted with ❤ by GitHub

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Now that the data is plotted and visualized to see the ‘.wav’ file, we would now be using a popular **‘librosa’** that can be used to calculate the short-time Fourier transform of the audio data. This is to ensure that the audio signal is decomposed into its constituent frequencies and that is widely used in a large number of industries.

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Training the Models

```
1 # Convert the audio data to a spectrogram
2 X = compute_spectrogram(X)
3
4 # Split data into training and test sets
5 X_train, X_test, y_train, y_test = train_test_split(X, y, test_size=0.2, random_state=42)
6
7 # Build machine learning model
8 # In this case, let's use a random forest classifier with 100 trees
9 model = RandomForestClassifier(n_estimators = 100)
10
11 # Train model on training data
12 model.fit(X_train, y_train)
13
14 # Evaluate model on test data
15 accuracy = model.score(X_test, y_test)
16 print("Test accuracy:", accuracy)
```

[training_audio.py](#) hosted with ❤ by GitHub

Now that we have used ‘librosa’ to get the frequency components, we would be using machine learning to make predictions. It is to be noted that it is a classification problem and hence, we go ahead with using a **random forest classifier**. However, feel free to use any other machine learning model that suits your needs and the business.

We are now going to be using the same code but for tasks where the output is **continuous** instead of categorical. The coding cell about how training could be done and performance monitored with the help of a random

regressor.

```
1 # Convert the audio data to a spectrogram
2 X = compute_spectrogram(X)
3
4 # Split data into training and test sets
5 X_train, X_test, y_train, y_test = train_test_split(X, y, test_size = 0.2)
6
7 # Build machine learning model
8 # In this case, let's use a random forest regressor with 100 trees
9 model = RandomForestRegressor(n_estimators = 100)
```

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```
10
11 # Train model on training data
12 model.fit(X_train, y_train)
13
14 # Evaluate model on test data
15 accuracy = model.score(X_test, y_test)
16 print("Test accuracy:", accuracy)
```

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Hyperparameter Tuning

It is important to determine the right hyperparameters for a model (Random Forest) before it could be deployed. There are a lot of hyperparameters to **search** for when tuning a model, especially to deep neural networks. Since we are using random forests, we can limit our search space. Let's perform hyperparameter tuning on the general dataset.

```
1 # Define hyperparameters to tune
2 param_grid = {'max_depth': [2, 4, 6, 8], 'min_samples_split': [2, 3, 5]}
3
4 # Create a grid search object with 5-fold cross-validation
5 grid_search = GridSearchCV(model, param_grid, cv=5)
6
7 # Fit the grid search object to the training data
8 grid_search.fit(X_train, y_train)
9
10 # Print the best hyperparameters and score
11 print("Best hyperparameters:", grid_search.best_params_)
12 print("Best cross-validation score:", grid_search.best_score_)
13
14 # Evaluate the model on the test data
15 best_model = grid_search.best_estimator_
16 accuracy = best_model.score(X_test, y_test)
17 print("Test accuracy:", accuracy)
```

hyperparameter_tuning_audio.py hosted with ❤ by GitHub

In the code cell, we specify the **number of estimators** and the **maximum depth** of the tree that we search for to find the best combination.

results on the test set. We finally monitor the score and see how changes in the hyperparameters can lead to better performance by the model.

Model Deployment

```
1 import pickle
2
3 # Save the model to a file
4 with open('model.pkl', 'wb') as f: LATEST
5     pickle.dump(model, f)
6
7 # Load the model from a file
8 with open('model.pkl', 'rb') as f: EDITOR'S PICKS
9     model = pickle.load(f)
10
11 # Use the model to make predictions on new data
12 predictions = model.predict(new_data) DEEP DIVES
13
14 newsletter = Newsletter()
15 newsletter.add_section('Introduction', 'The goal of this project is to')
16 newsletter.add_section('Data', 'The dataset consists of 1000 samples, each with 10 features')
17 newsletter.add_section('Model', 'We will use a simple linear regression model to predict the target variable')
18 newsletter.add_section('Evaluation', 'We will evaluate the model using various metrics such as R-squared, Mean Absolute Error, and Root Mean Square Error')
19 newsletter.add_section('Deployment', 'Once we have trained the model, we will save it to a file and use it to make predictions on new data')
20 newsletter.add_section('Conclusion', 'In this project, we have learned how to build a machine learning model for regression tasks. We used a simple linear regression model and evaluated its performance using various metrics. Finally, we saved the model to a file and used it to make predictions on new data. This process is known as deployment. In the next project, we will learn how to handle more complex data and build more advanced models. Stay tuned!')
```

audio_deployment.py hosted with ❤ by GitHub

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Now that we have performed hyperparameter tuning to find the most accurate predictions, it is time to save the models that provided the best results. Therefore, we will use the **pickle** library in python so that we will have the freedom to save the machine learning model that we will use for serving.

After saving the model, we would again load it while building a **production-ready** code and use it to run on incoming batches or streams of data. It is to remember that the set of steps that were used to perform feature engineering on the training data must also be performed on the test data so that there is no skew in the data.

Constant Monitoring

We know that the model is performing a good job while it is receiving incoming data from users, it is an important yet neglected step which is to **monitor** the quality of the models. There are often scenarios where the models might not be performing as they were during training. This can be because there is a difference between the training and serving data. For example, there can be situations of **concept drift** or **data drift** that can have a significant impact on the performance of the inference model that is being used.

production.

Constant monitoring ensures that steps are taken to check the predictive models and understand their behavior with changing data. If the predictions are the **least accurate** and it is leading to a loss of revenue for the business, steps should be taken to again train the models with this deviant data so that there is no unexpected change in the behavior of the model.

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Conclusion

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After going through this article, you might have gained an understanding about how to perform machine learning for **audi**. We have seen the overall workflow. We have seen the steps involved in reading the data, feature engineering, training the model, hyperparameter tuning, model deployment along with monitoring. Applying each of these steps and ensuring that there are no mistakes when developing a pipeline would lead to a robust machine learning production system.

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Below are the ways where you could contact me for my work.

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