Final Project Presentation Group 2

UrbanSounds8k

By: Tristin Johnson

DATS 6203 – Machine Learning II

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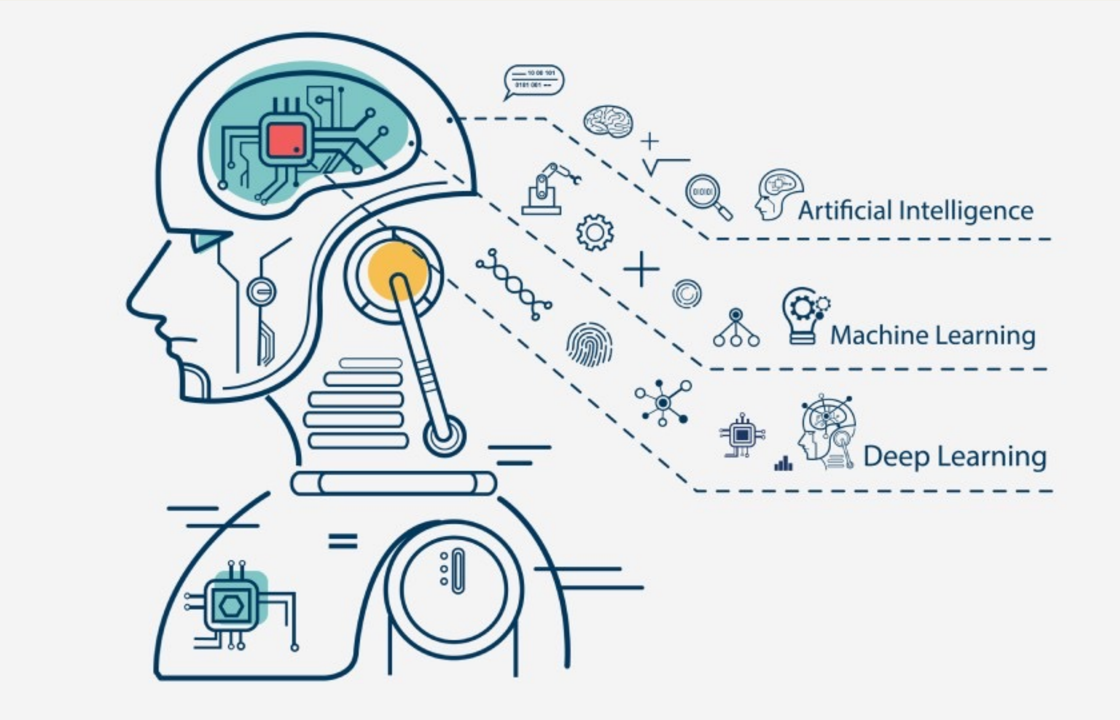


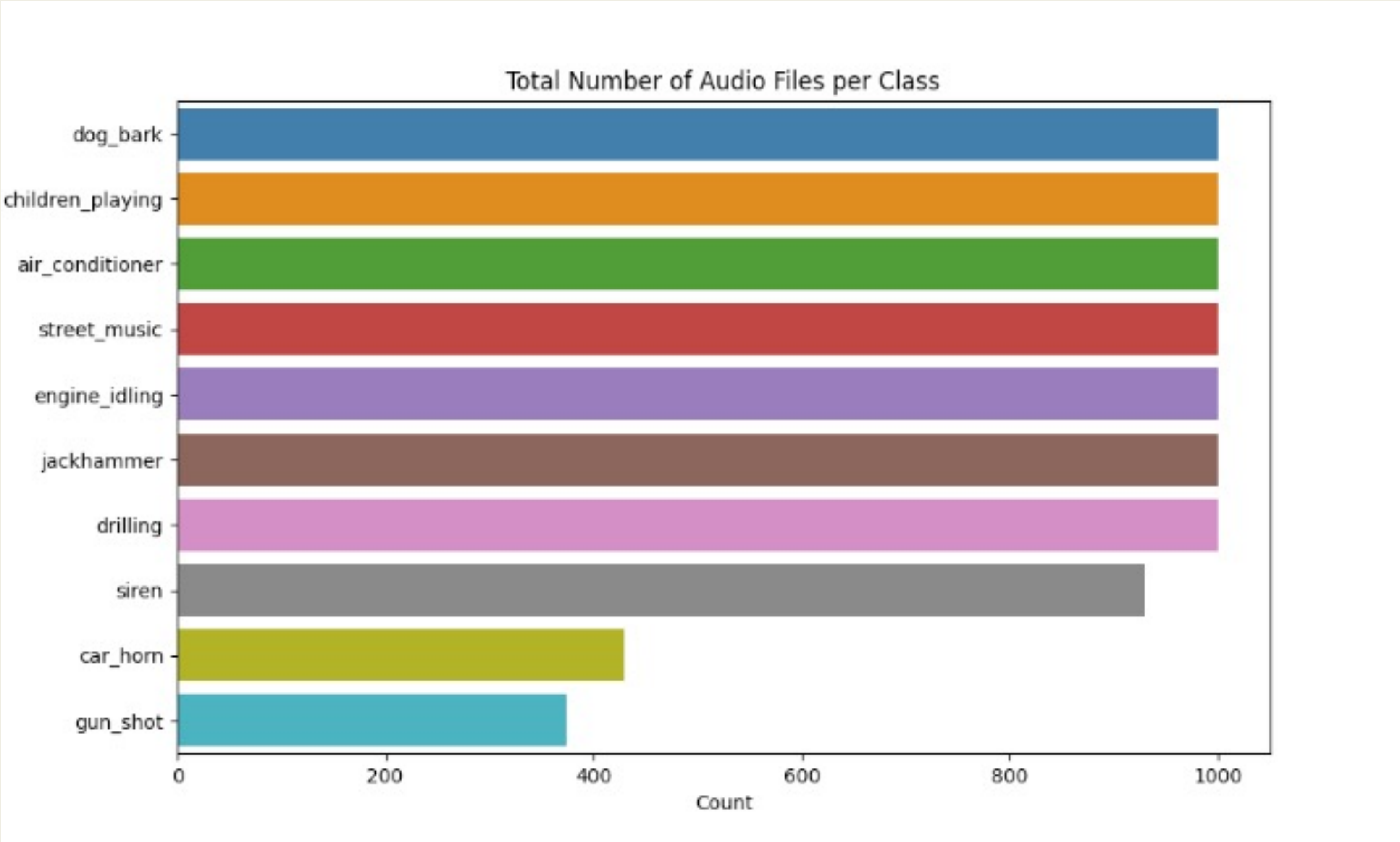
Introduction

- UrbanSounds8K Dataset
  - 8732 labeled audio files (.wav format)
  - 10 classes
    - air_conditioner, car_horn, children_playing, dog_bark, drilling, engine_idling, gun_shot, jackhammer, siren, street_music
  - Each audio file ~ 4 seconds (around 9.5 hours total)
  - Pre-sorted into 10 folds
  - Metadata included (.csv)
- Deep Speech topic
  - Network: CNN
  - Framework: PyTorch
    - Librosa, TorchAudio

Data Analysis: Metadata

	slice_file_name	fsID	start	end	salience	fold	classID	class	num_channels	sampling_rate
0	177742-0-0-99.wav	177742	49.500000	53.500000	2	3	0	air_conditioner	2	48000
1	24074-1-0-10.wav	24074	18.060993	22.060993	1	1	1	car_horn	2	44100
2	60591-2-0-4.wav	60591	2.000000	6.000000	1	2	2	children_playing	2	44100
3	174026-3-1-5.wav	174026	11.719766	15.719766	2	4	3	dog_bark	2	48000
4	59594-4-0-3.wav	59594	1.500000	5.500000	2	2	4	drilling	2	44100
5	154758-5-0-7.wav	154758	3.500000	7.500000	1	4	5	engine_idling	2	48000
6	148841-6-2-0.wav	148841	9.132153	10.787741	1	5	6	gun_shot	2	44100
7	162134-7-7-0.wav	162134	129.628486	133.628486	1	10	7	jackhammer	2	96000
8	118279-8-0-13.wav	118279	6.500000	10.500000	2	1	8	siren	2	48000
9	89443-9-0-16.wav	89443	8.000000	12.000000	1	7	9	street_music	2	44100

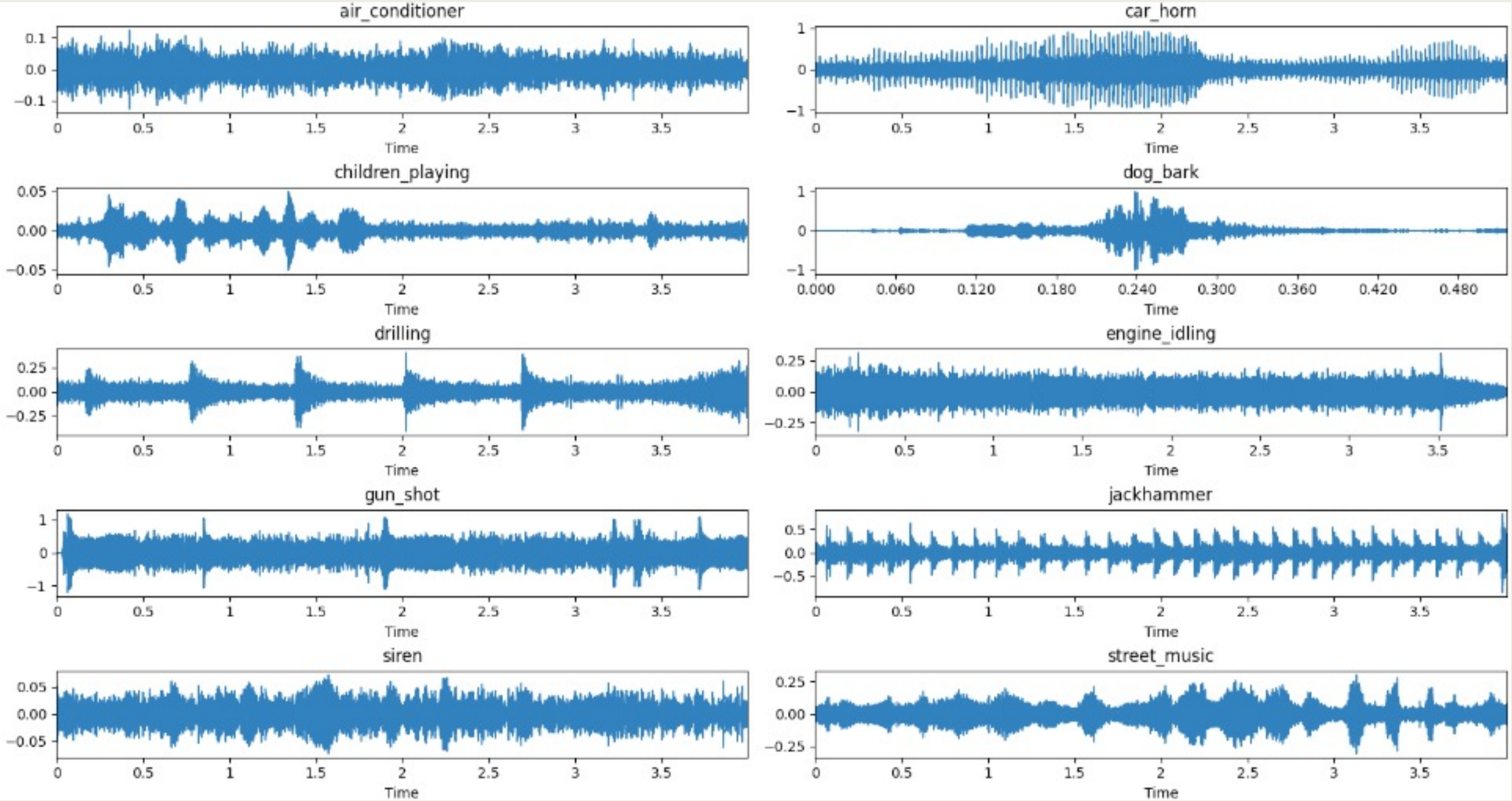
Data Analysis: Audio Files per Class



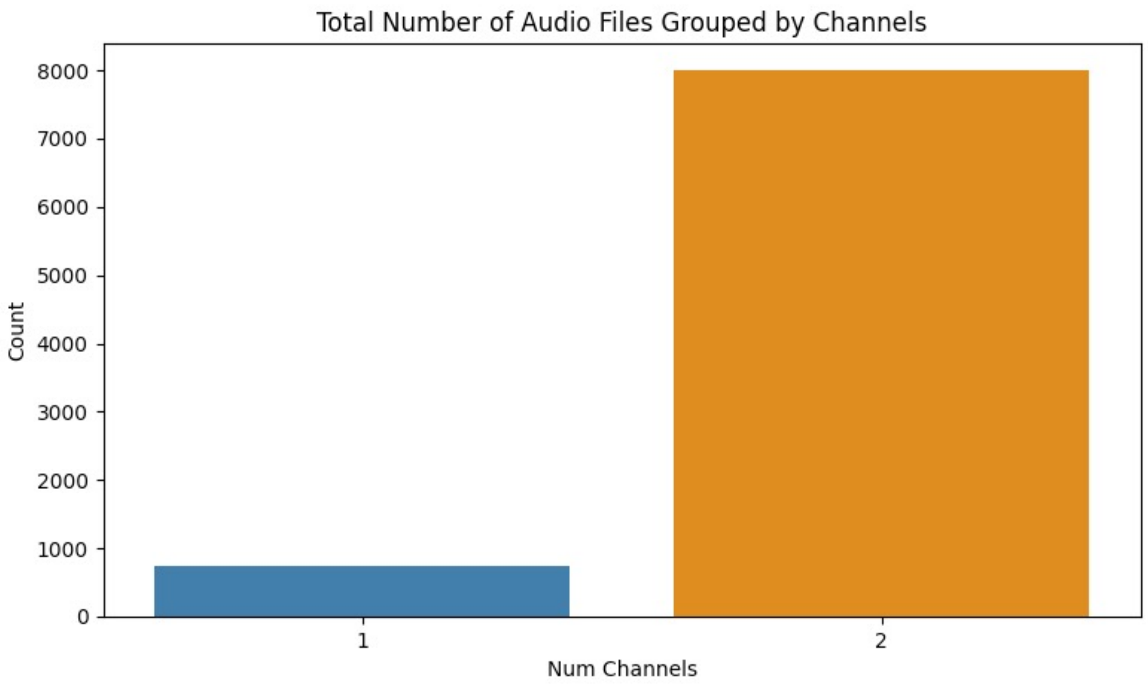
Data Preprocessing

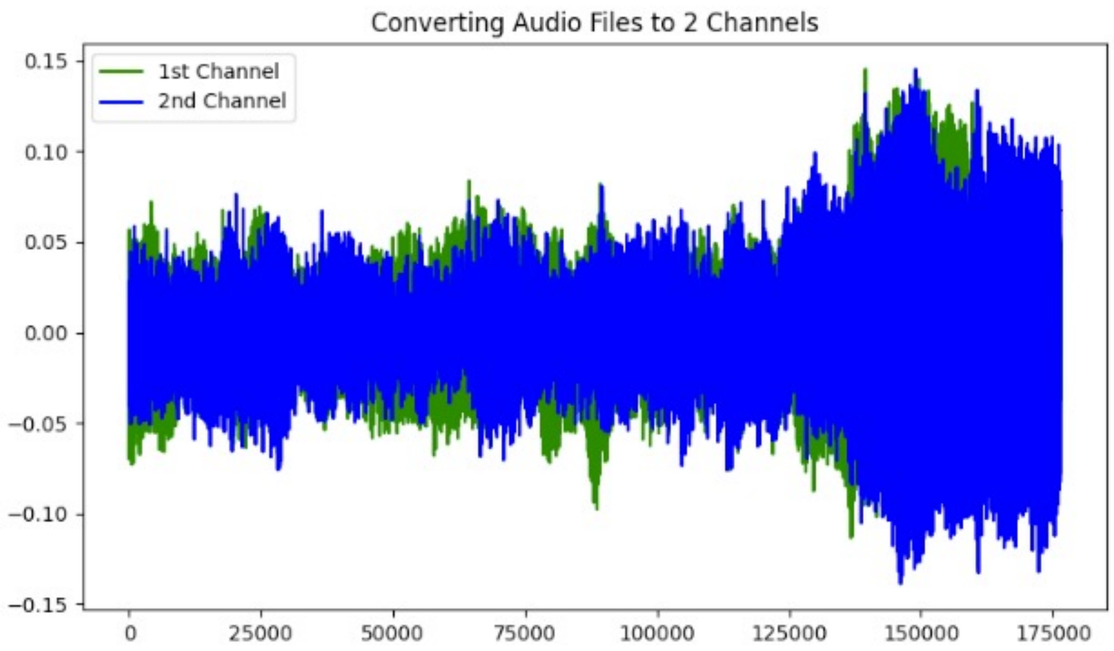
1. Load Audio File
2. Convert to Two Channels
3. Standardize Sampling Rate
4. Add Padding to Resize Audio to Same Length
5. Data Augmentation
   - Random Time Shift
   - Mel Spectrogram
   - Time & Frequency Masking

1. Load Audio Files



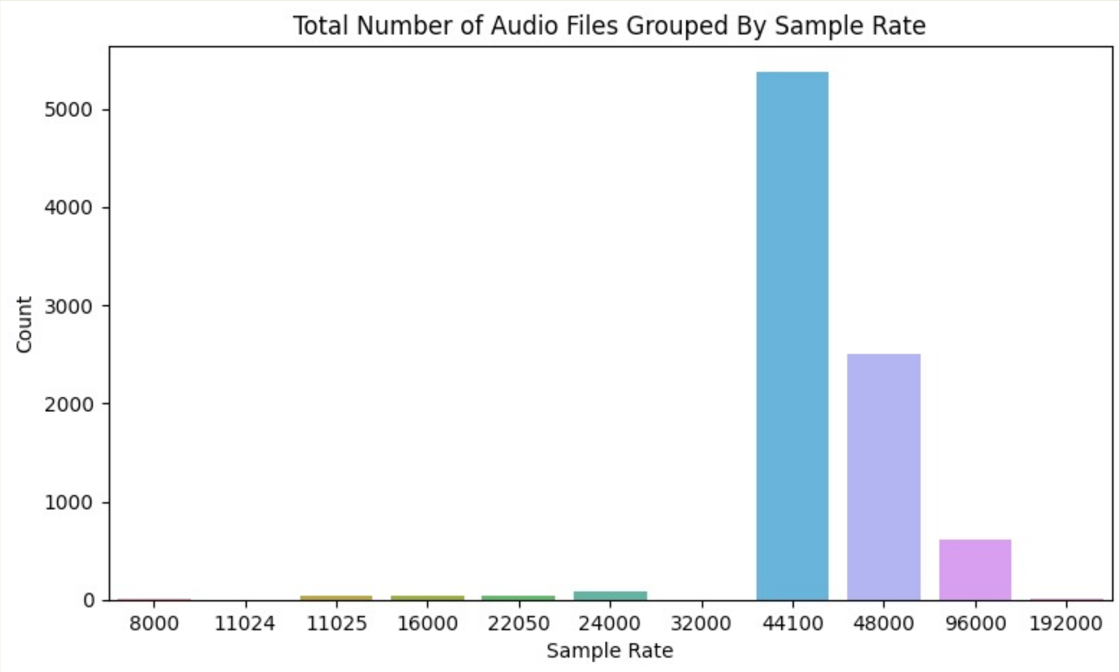
2. Convert to Two Channels



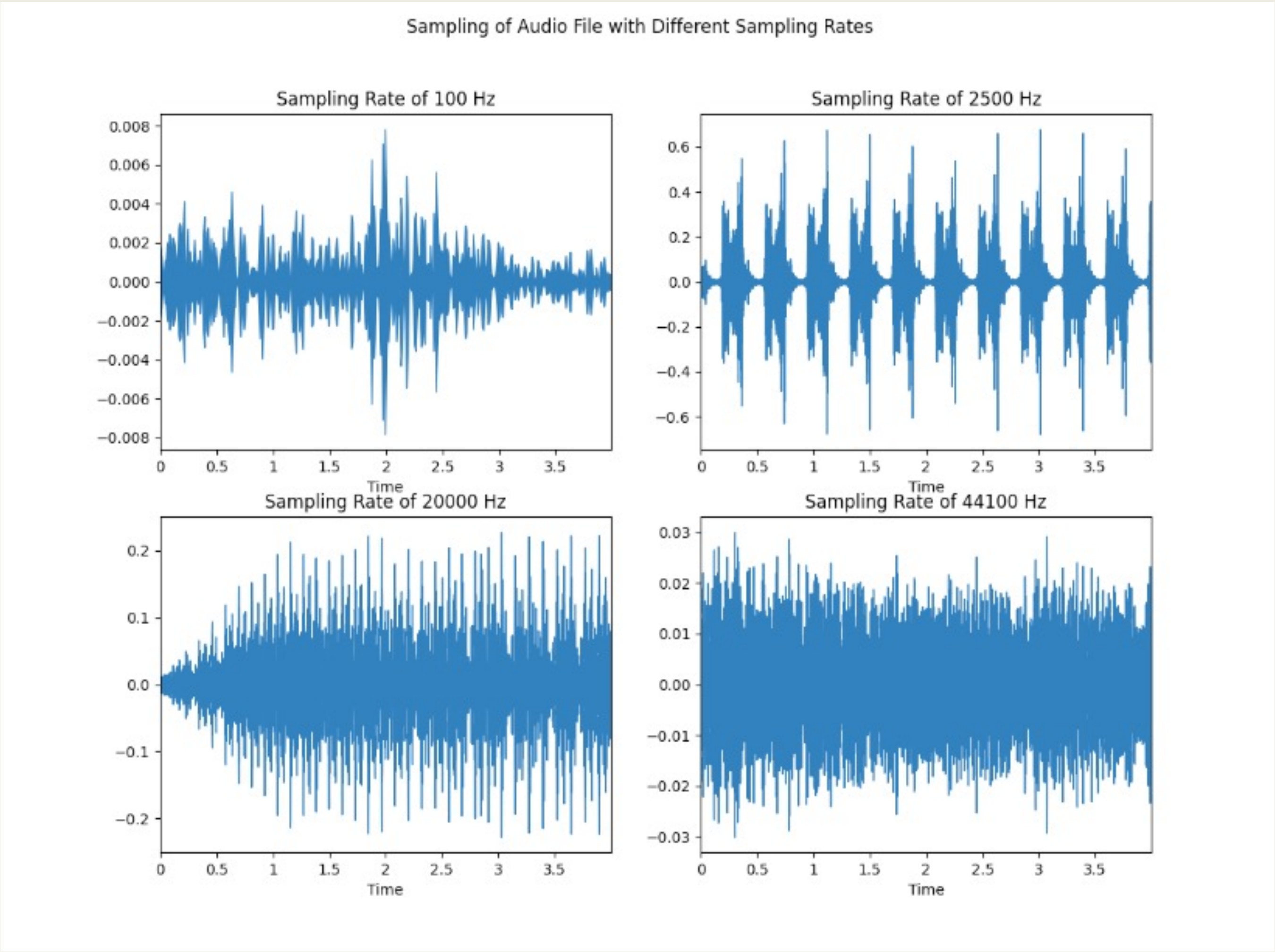


3. Standardize Sampling Rate

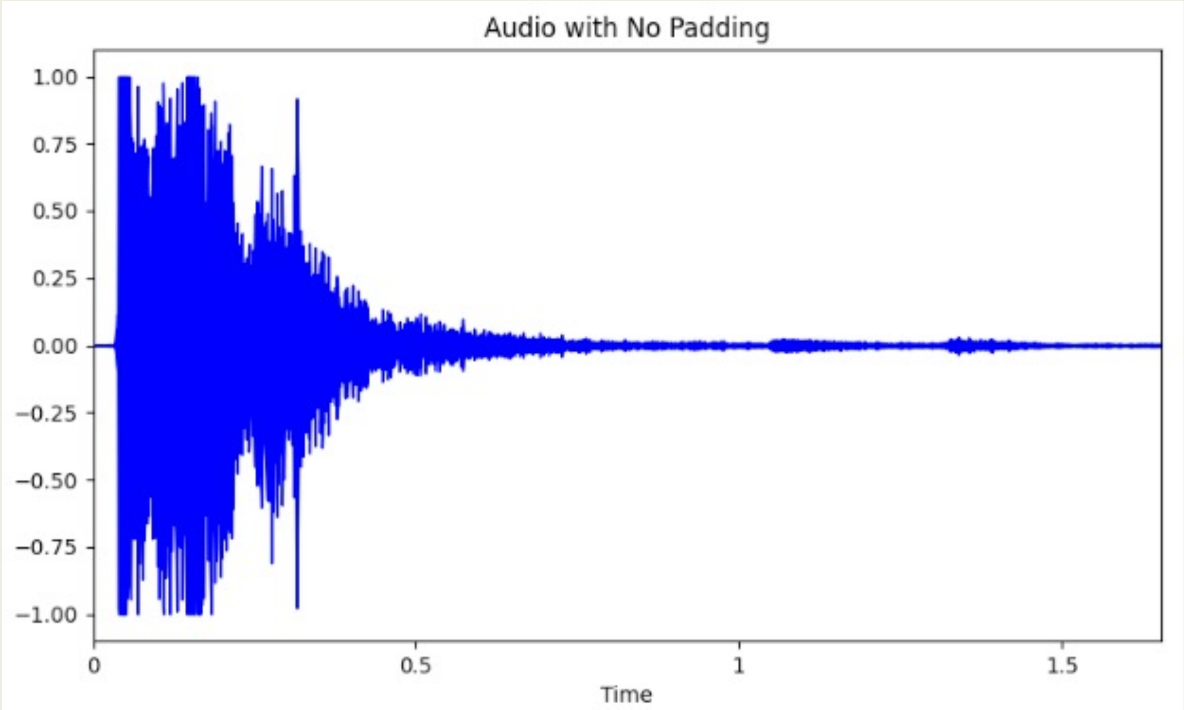
- Majority of Audio files are sampled at 44.1 kHz (44,100 Hz)
  - 1 second of audio will have an array of size 44,100
- Other sample rates will have dimensions of num_channels x (sample_rate x time)
- Standardize sampling rates of all audio files to 44.1 kHz

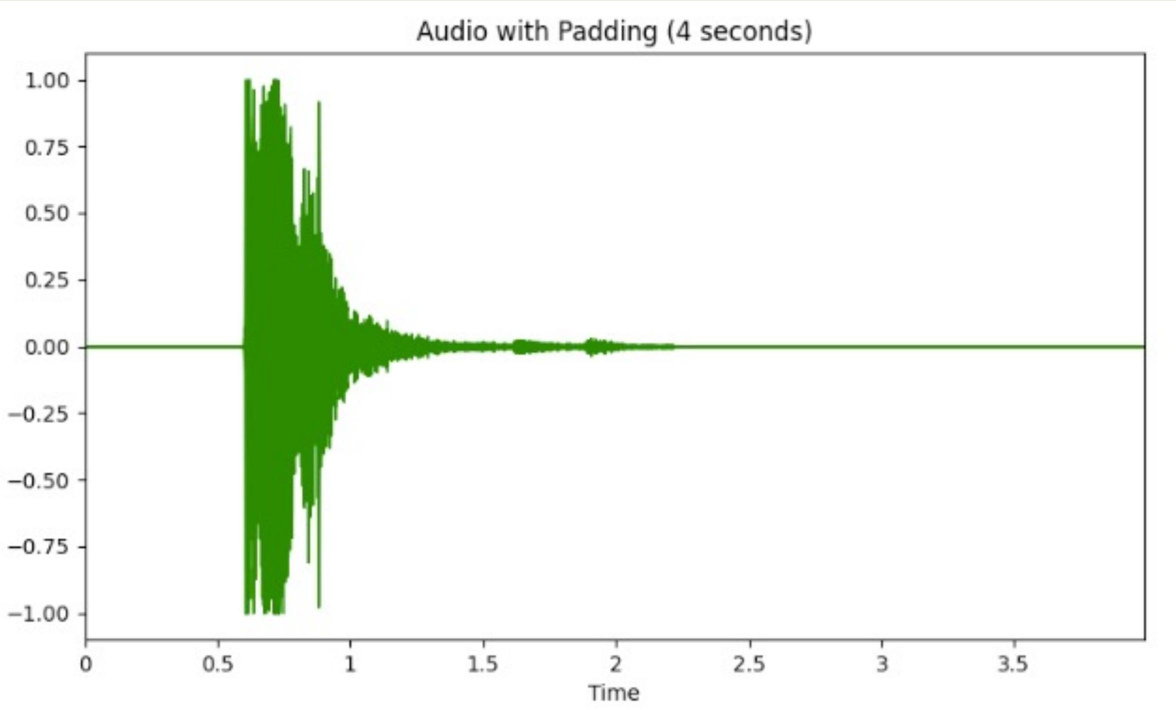


3. Standardize Sampling Rate



4. Add Padding to Resize Audio to Same Length

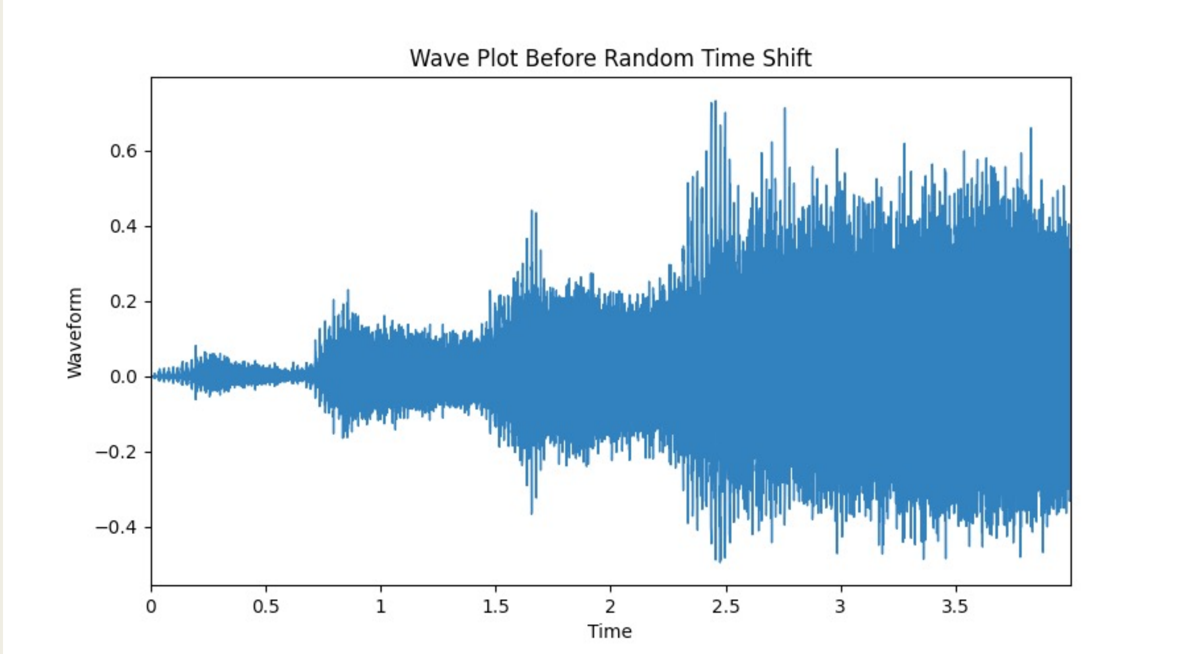


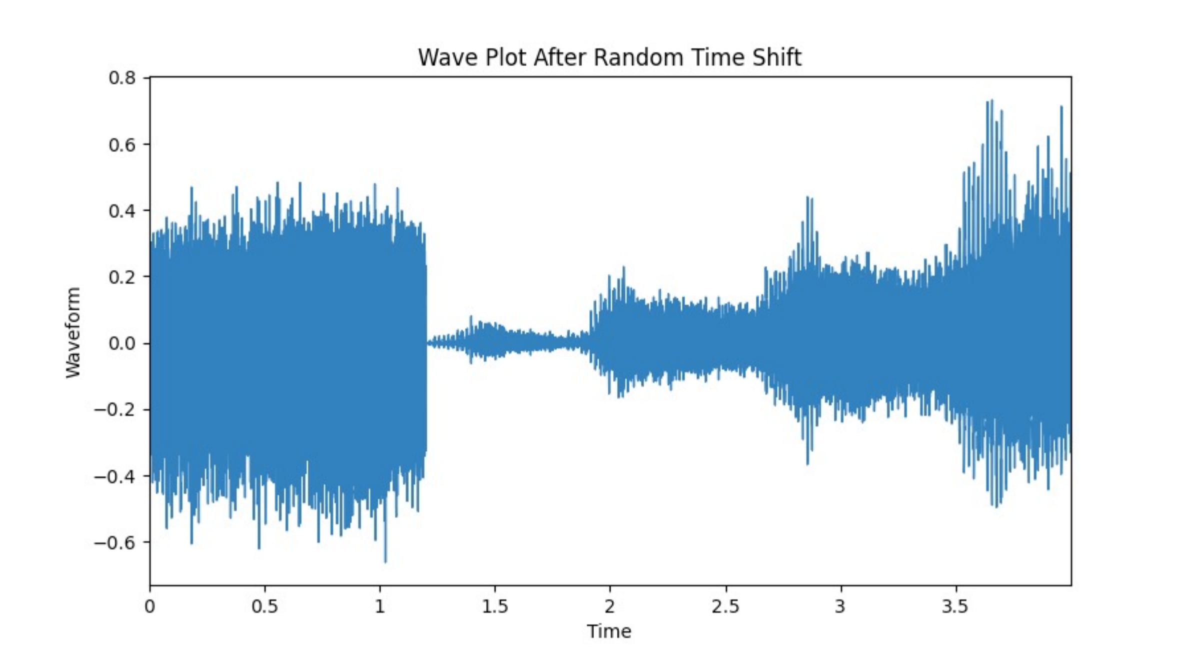


- Before Padding:
  - Time: ~ 1.75 seconds

- After Padding:
  - Time: 4 seconds

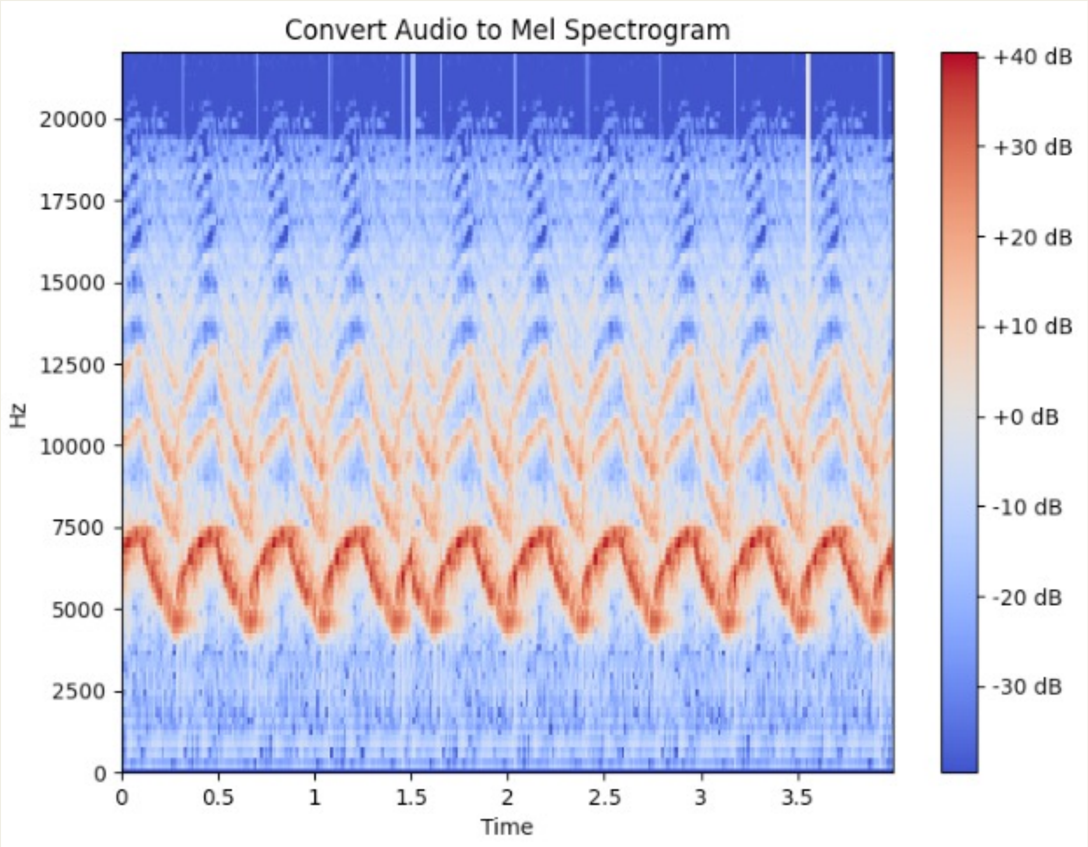
5. Data Augmentation: Time Shift





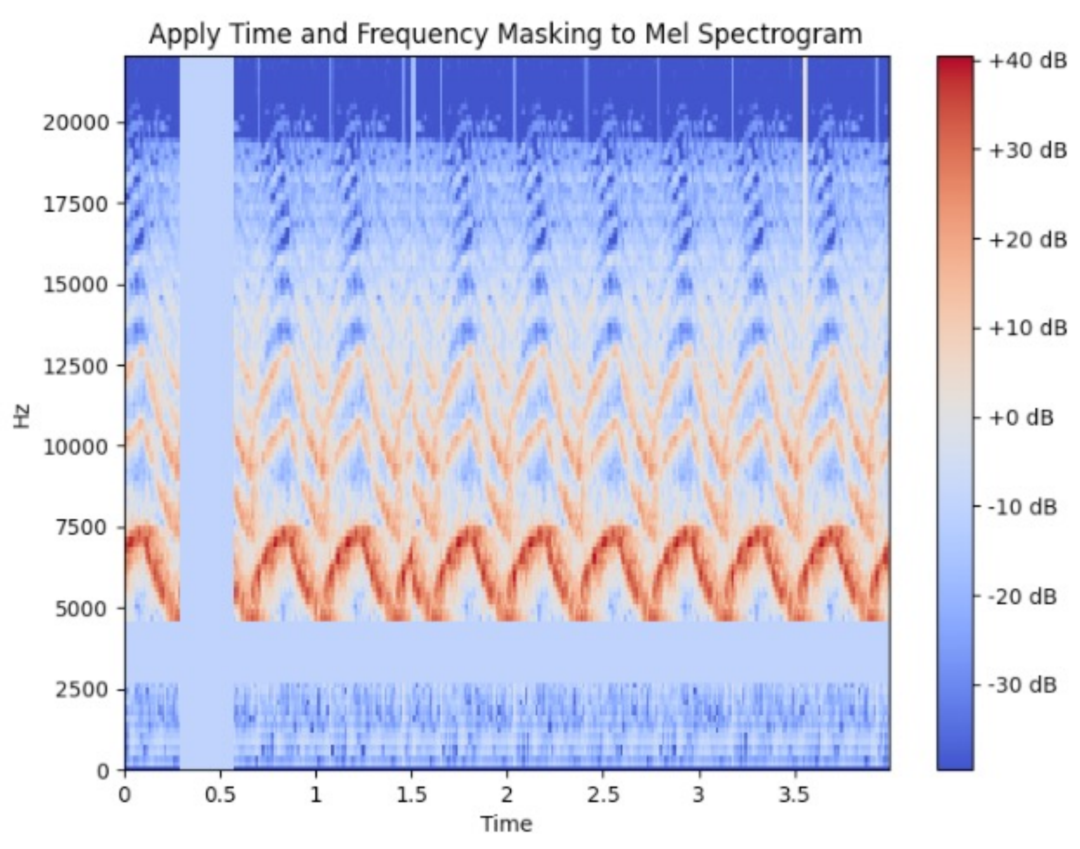
5. Data Augmentation: Mel Spectrogram

- Spectrograms
  - Chops up duration of a sound signal (waveform) into smaller segments
  - Plots Frequency (y-axis) vs. Time (x-axis)
  - Colors indicate amplitude of each frequency
- Mel Spectrograms
  - The Mel Scale instead of Frequency
  - Decibel Scale instead of Amplitude
    - Provides more useful information to a deep learning model



5. Data Augmentation: Time & Frequency Masking

- Frequency Masking
  - Randomly mask out a range of consecutive frequencies (horizontal bars)
- Time Masking
  - Similar to Frequency Masking, but instead masks out a range of time (vertical bars)



CNN Architecture

- 4-layer network
  - Convolution-2D
    - Kernel Size = (5, 5) → (3, 3)
    - Stride = (2, 2)
    - Padding = (2, 2) → (1, 1)
  - Batch Normalization
  - Zero-Pad-2D
  - Max-Pooling-2D
  - Adaptive-Average-Pooling-2D
  - Fully-Connected-Linear-Layer
  - Activation Function: ReLU

```
AudioClassifier(
  (conv1): Conv2d(2, 8, kernel_size=(5, 5), stride=(2, 2), padding=(2, 2))
  (batch1): BatchNorm2d(8, eps=1e-05, momentum=0.1, affine=True, track_running_stats=True)
  (pad1): ZeroPad2d(padding=(2, 2, 2, 2), value=0.0)
  (pool1): MaxPool2d(kernel_size=5, stride=2, padding=0, dilation=1, ceil_mode=False)
  (conv2): Conv2d(8, 32, kernel_size=(3, 3), stride=(2, 2), padding=(1, 1))
  (batch2): BatchNorm2d(32, eps=1e-05, momentum=0.1, affine=True, track_running_stats=True)
  (pad2): ZeroPad2d(padding=(2, 2, 2, 2), value=0.0)
  (conv3): Conv2d(32, 64, kernel_size=(3, 3), stride=(2, 2), padding=(1, 1))
  (batch3): BatchNorm2d(64, eps=1e-05, momentum=0.1, affine=True, track_running_stats=True)
  (conv4): Conv2d(64, 128, kernel_size=(3, 3), stride=(2, 2), padding=(1, 1))
  (batch4): BatchNorm2d(128, eps=1e-05, momentum=0.1, affine=True, track_running_stats=True)
  (act): ReLU()
  (pool2): AdaptiveAvgPool2d(output_size=1)
  (linear1): Linear(in_features=128, out_features=10, bias=True)
)
```

CNN Architecture

- Input size: (batch_size, num_channels, mel_freq, time_steps)
  - (16, 2, 64, 344)
- Image width and height are reduced with kernels and strides
- After forward propagation through CNN layers, pooling layer, and linear layer, output is (batch_size, num_classes)
  - (16, 10)

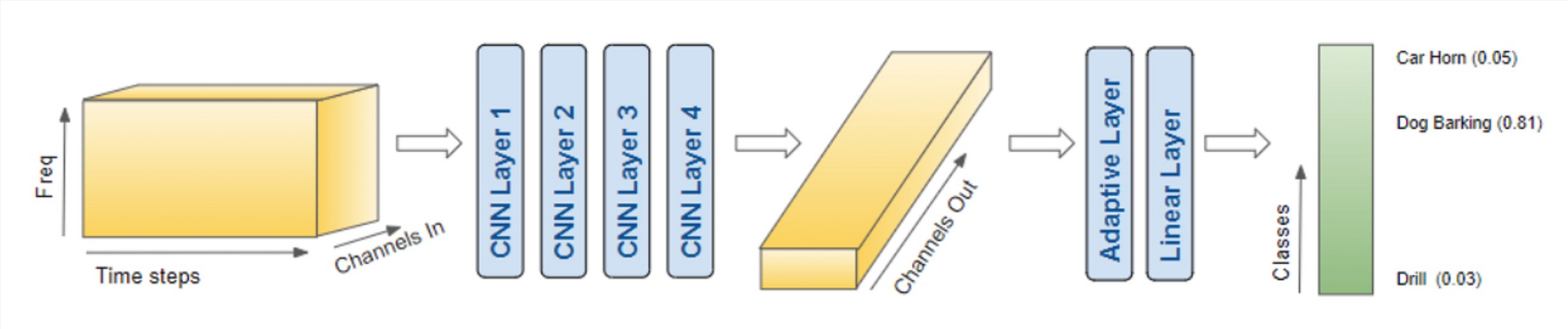


Image by: Ketan Doshi

Training & Validation

- Model Parameters:
  - Epochs: 20
  - Batch Size: 16
  - Learning Rate: 0.001
  - Optimizer: AdamW
  - Scheduler: ReduceLROnPlateau
  - Loss Function: CrossEntropyLoss

- Training Results (70%):
  - Accuracy: 91.95%
  - Total Training Time: 39 min 33 secs
  - Avg Time per Epoch: 1 min 58 secs
- Validation Results (15%):
  - Accuracy: 90.992%
  - Total Validation Time: 7 min 58 secs
  - Avg Time per Epoch: 24 secs

Results – Testing the Model

- Testing Results (15%):
  - Accuracy: 91.145%
  - Total Testing Time: 21 seconds
- Performance Highlights:
  - Time Shift → + 3.16% ~ 4.33%
  - Mel Spectrogram → + 5.14% ~ 8.04%
  - Time & Freq. Mask → + 3.2% ~ 5.47%

Future Work

- Apply Transformers to UrbanSounds8K
  - Wav2Vec2 (Hugging Face)
    - ”A framework for self-supervised learning of speech representations.”
  - XLSR-Wav2Vec2 (Hugging Face)
    - “Unsupervised cross-lingual representation learning for speech recognition”

Conclusion

- Ability to work with and preprocess Audio data (Librosa & TorchAudio)
  - Reading in audio as a wave-plot
  - Standardizing audio files (channels, sampling rate, padding time)
  - Convert wave-plot to Mel Spectrogram
  - Data Augmentation (random time shifts, frequency & time masking)
- Achieved high levels of accuracy with CNN
  - Training, Validation, Testing > 90% Accuracy
- Increased experience using PyTorch

References

1. UrbanSounds8K Official Website
2. UrbanSounds8K Data Download (Kaggle)
3. Audio Classification Analysis using Librosa (GitHub)
4. UrbanSound8K Audio Analysis
5. Audio Deep Learning Made Simple: Sound Classification
6. Audio Deep Learning: Why Mel Spectrograms Perform Better
7. UrbanSound8K Benchmarks (PapersWithCode)