

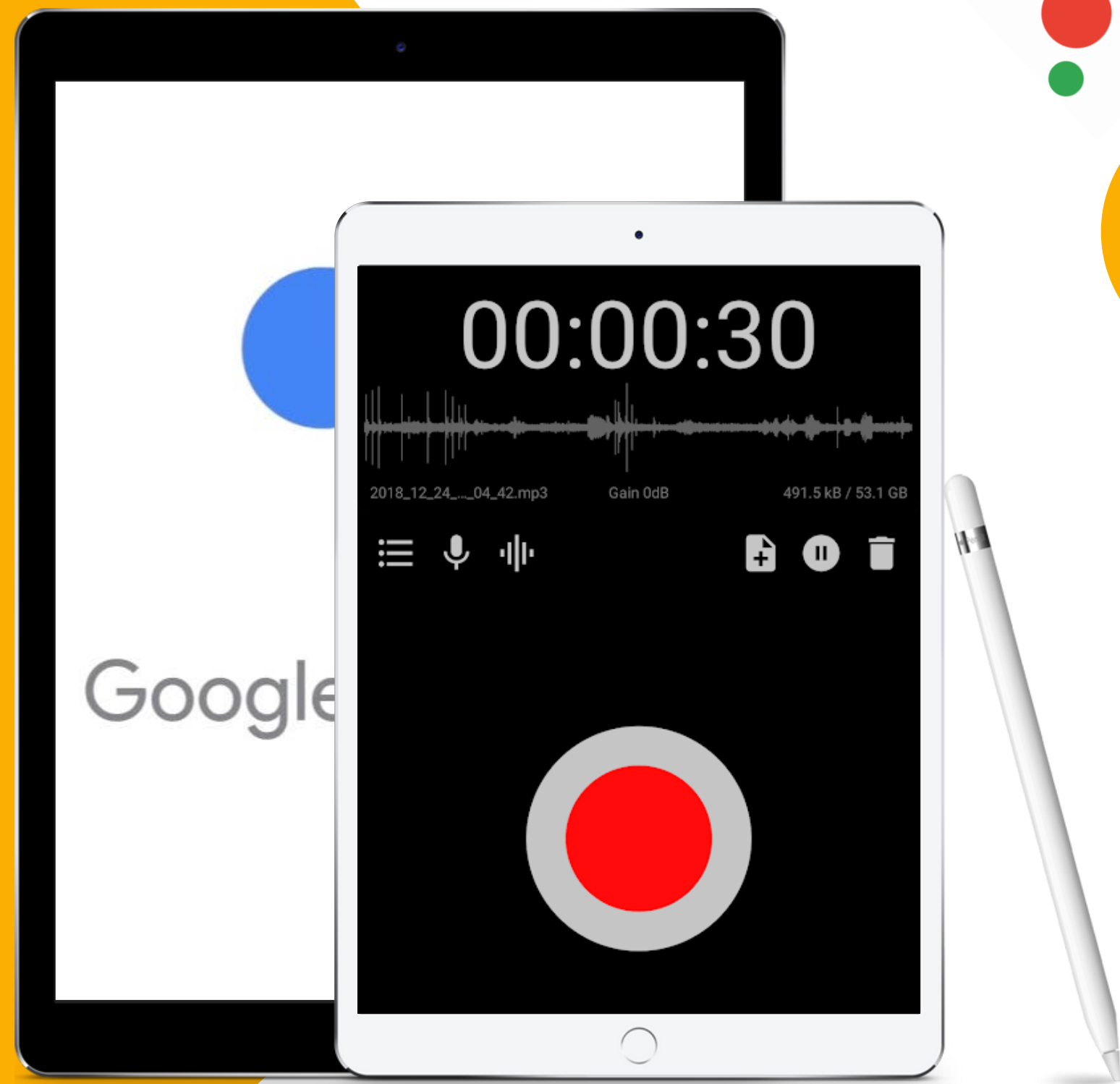
# Get started with Speech Recognition using TensorFlow



Nourchene Ferchichi 

Research Engineer  
ML Google Developer Expert

# Introduction

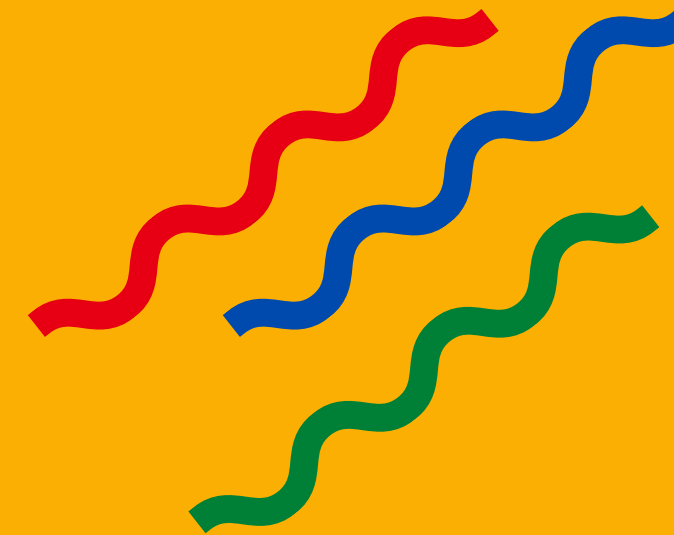


**How does  
information travel  
from our brains  
into the computer?**

# Introduction

## Automatic Speech Recognition, Defined

Computer Science  
+  
Linguistics



Transforms the spoken word  
into the written one!

# In this Presentation

Here's what we'll cover:

- 1 Applications
- 2 SR Dataset
- 3 DNN for Acoustic modeling
- 4 Language Models
- 5 Practice

# Applications of ASR



## Telecome Industry

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Conversational Chatbots to  
Enhance customer support  
Resolve Technical Issues  
Provide Personalised Advice



## Marketing

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Voice Search for accurate  
product search



## Internet of Things

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Smart TVs  
Autonomous cars  
Hands-free help at home

# Applications of ASR



## Health Care

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Digital Assistant:

Medical guidance

Quick access to  
administrative information



## Banking

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Request transaction  
information

Make payments



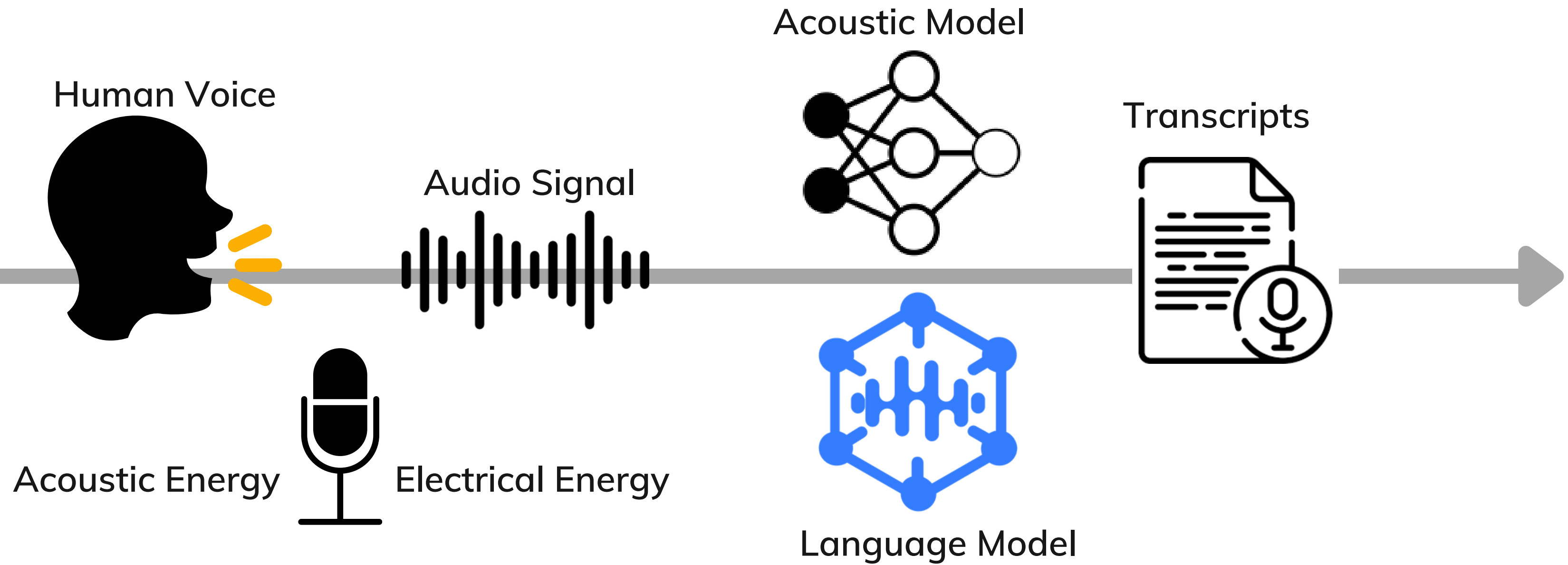
## Workplace

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Transcribe real-time  
conferences

Report meetings

# Pipeline





# Datasets format

## Features (X)



Audio wave

## Labels (y)

*Good Morning!*

Transcript



## Audio Waves: Features

Audio clips of spoken sequences

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## Transcripts: Labels

Text transcripts of what was spoken

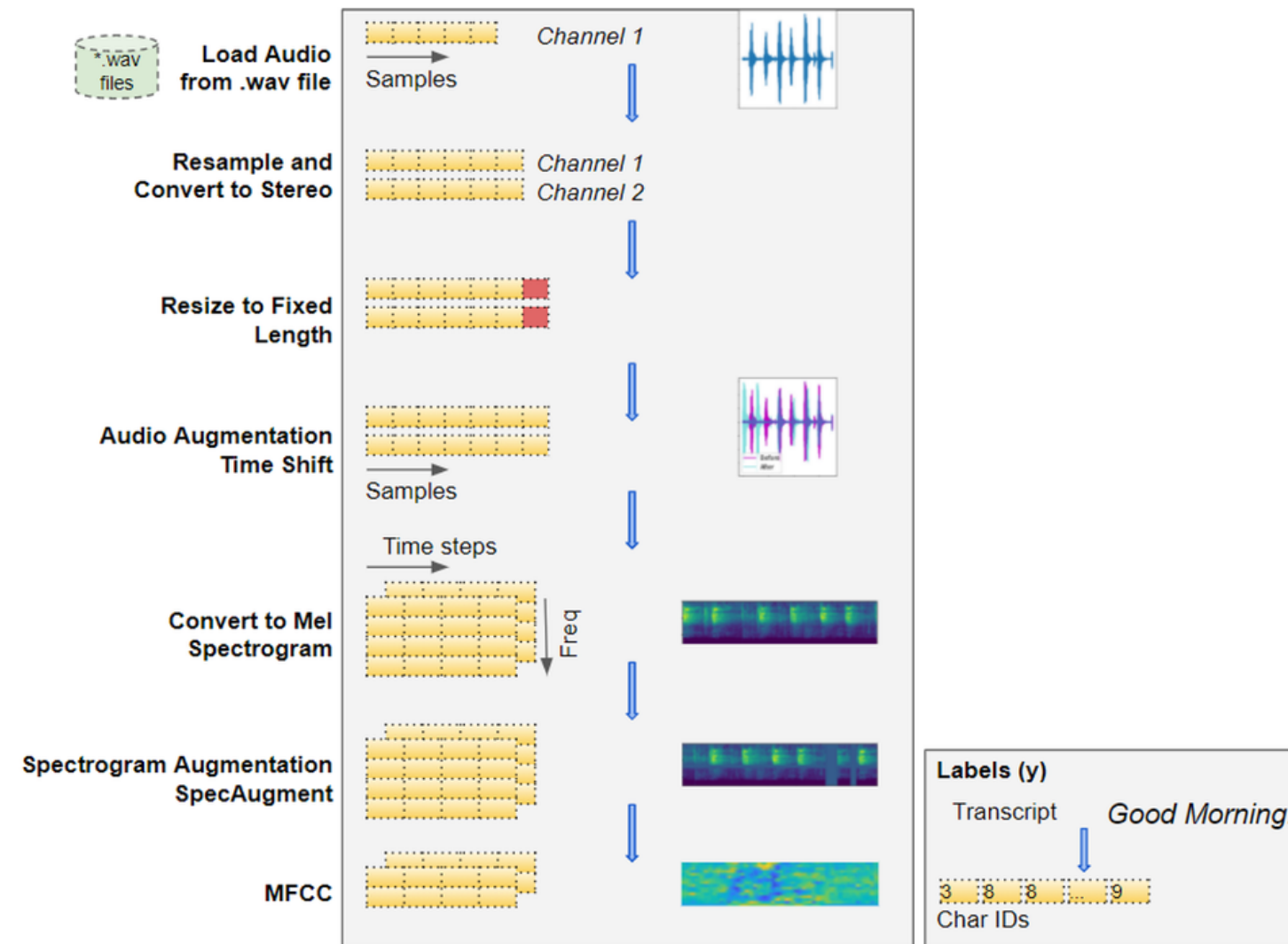
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# Datasets

## Preprocessing:

### Audio



# Datasets

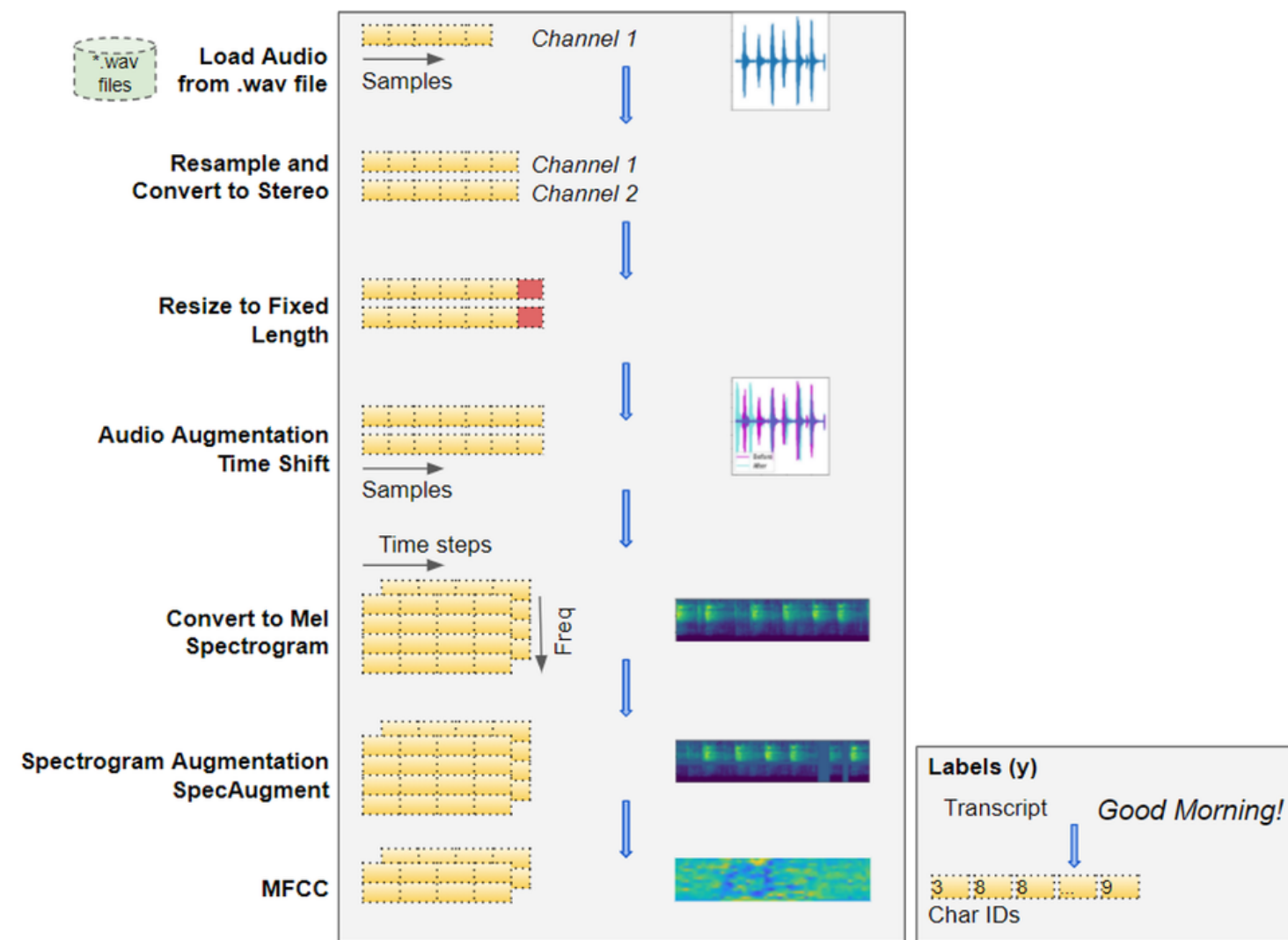
## Preprocessing:

### Audio

#### 1. Convert to uniform dimensions

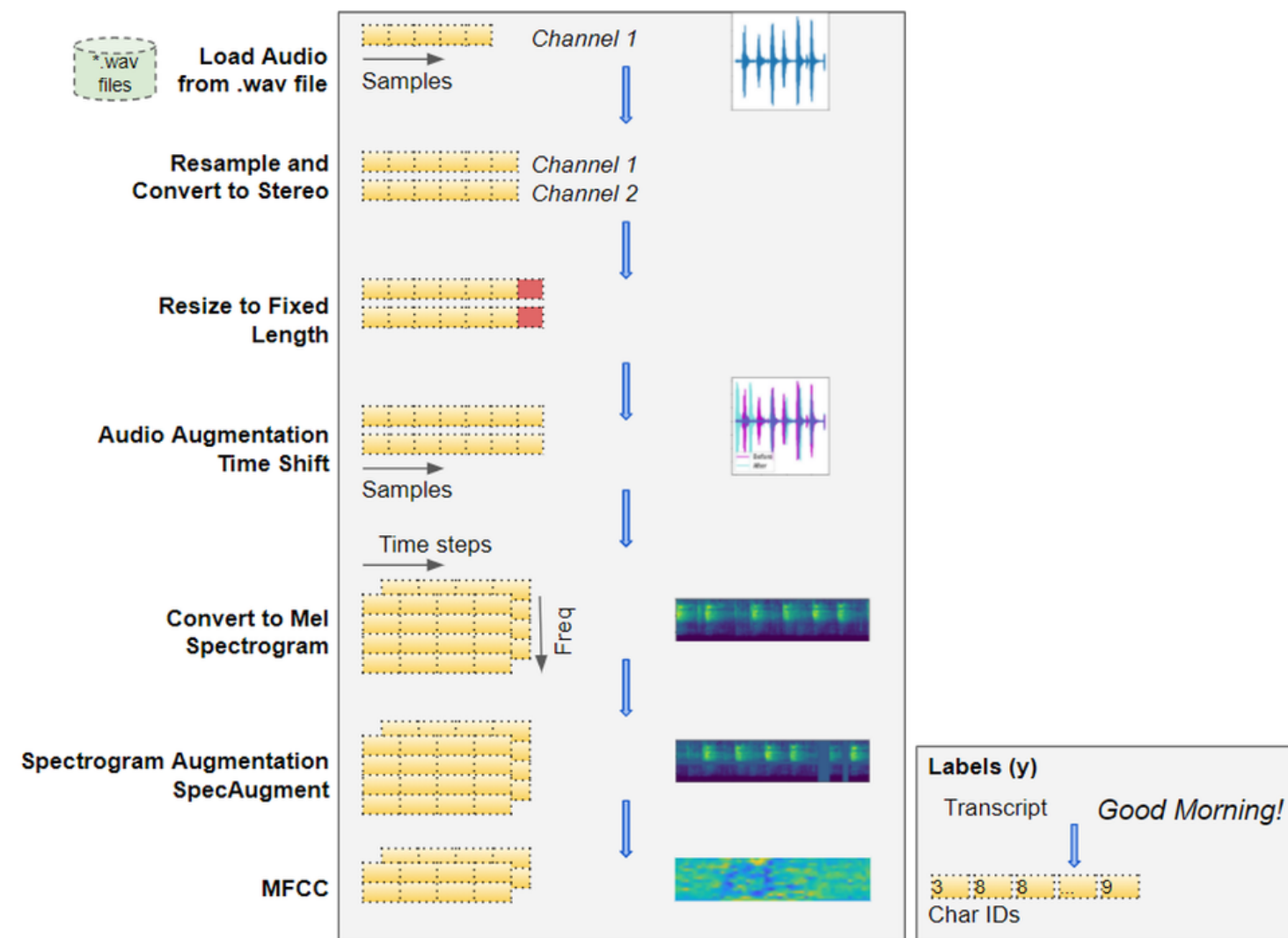
Standardize the dimensions of our audio data

.....



# Datasets Preprocessing:

## Audio



### 1. Convert to uniform dimensions

Standardize the dimensions of our audio data

.....

### 2. Data Augmentation of raw audio

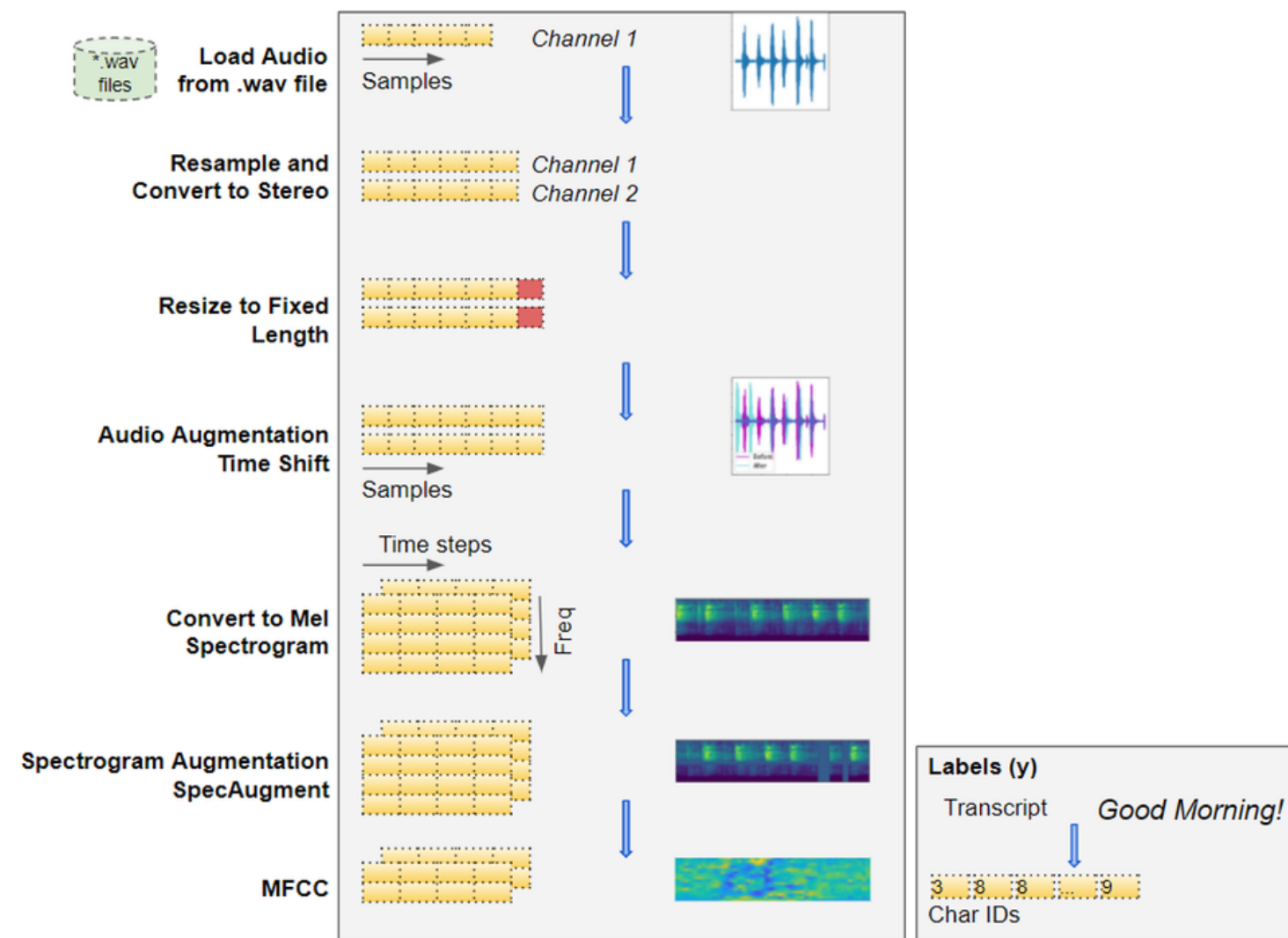
Add more variety to our input data

.....



# Datasets Preprocessing:

## Audio



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Standardize the dimensions of our audio data

### 2. Data Augmentation of raw audio

Add more variety to our input data

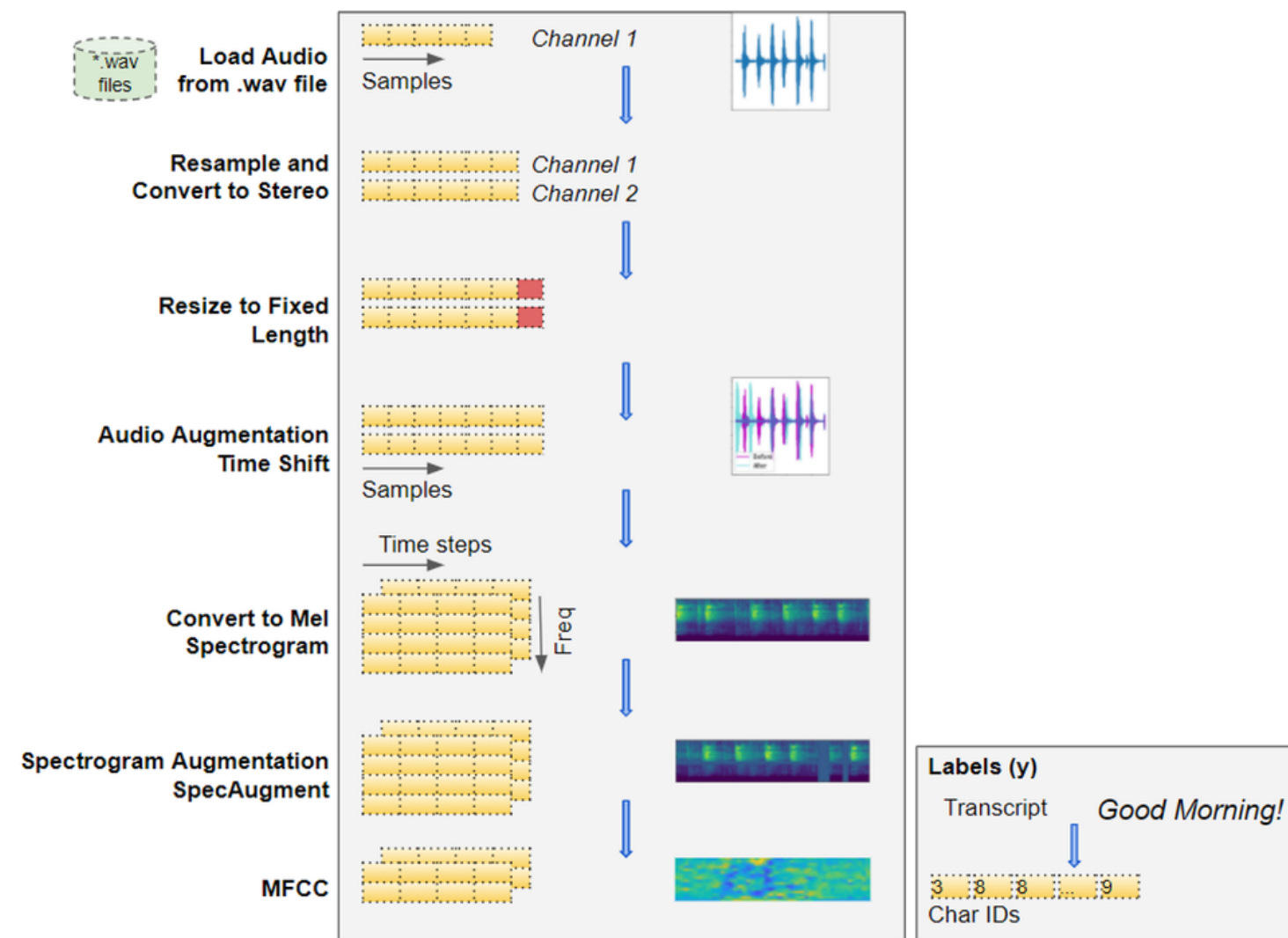
### 3. Mel Spectrograms

Captures the nature of the audio as an image



# Datasets Preprocessing:

## Audio



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### 2. Data Augmentation of raw audio

Add more variety to our input data

### 3. Mel Spectrograms

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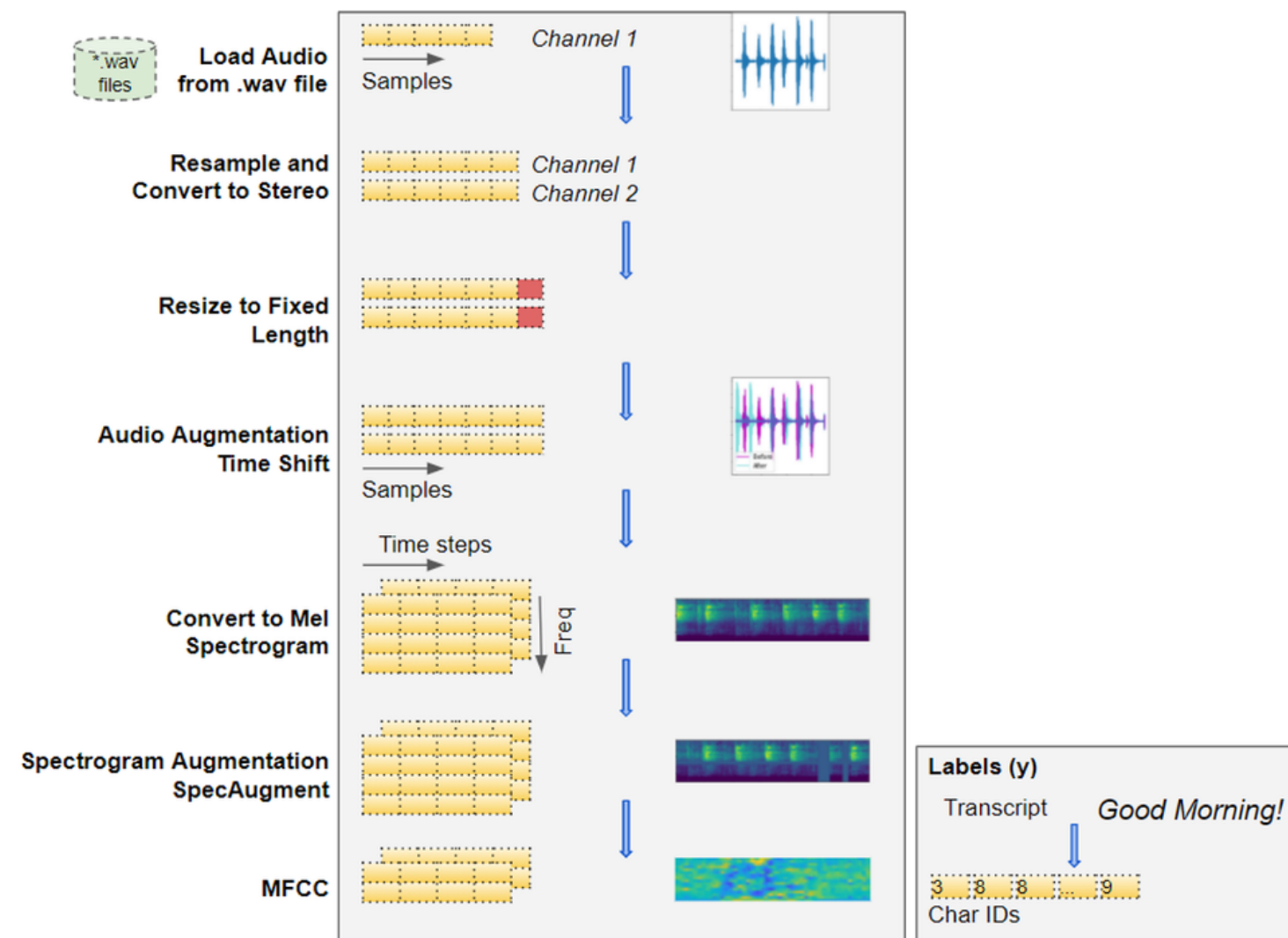
### 4. MFCC

Extract the most essential frequency coefficients



# Datasets Preprocessing:

## Audio



### 1. Convert to uniform dimensions

Standardize the dimensions of our audio data

### 2. Data Augmentation of raw audio

Add more variety to our input data

### 3. Mel Spectrograms

Captures the nature of the audio as an image

### 4. MFCC

Extract the most essential frequency coefficients

### 5. Data Augmentation of Spectrograms

Apply random Frequency and Time Masking





# Datasets Preprocessing:



## Transcripts

### Normalization

Bring the words to their closer to a predefined “standard”

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### Punctuation Removal

Remove punctuation characters like  
!"#\$%&'()\*+,-./:;<=>?@[\\]^\_`{|}~ from a text

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### Buid the vocabulary

Build a vocabulary from each character in the transcript and convert them into character IDs.

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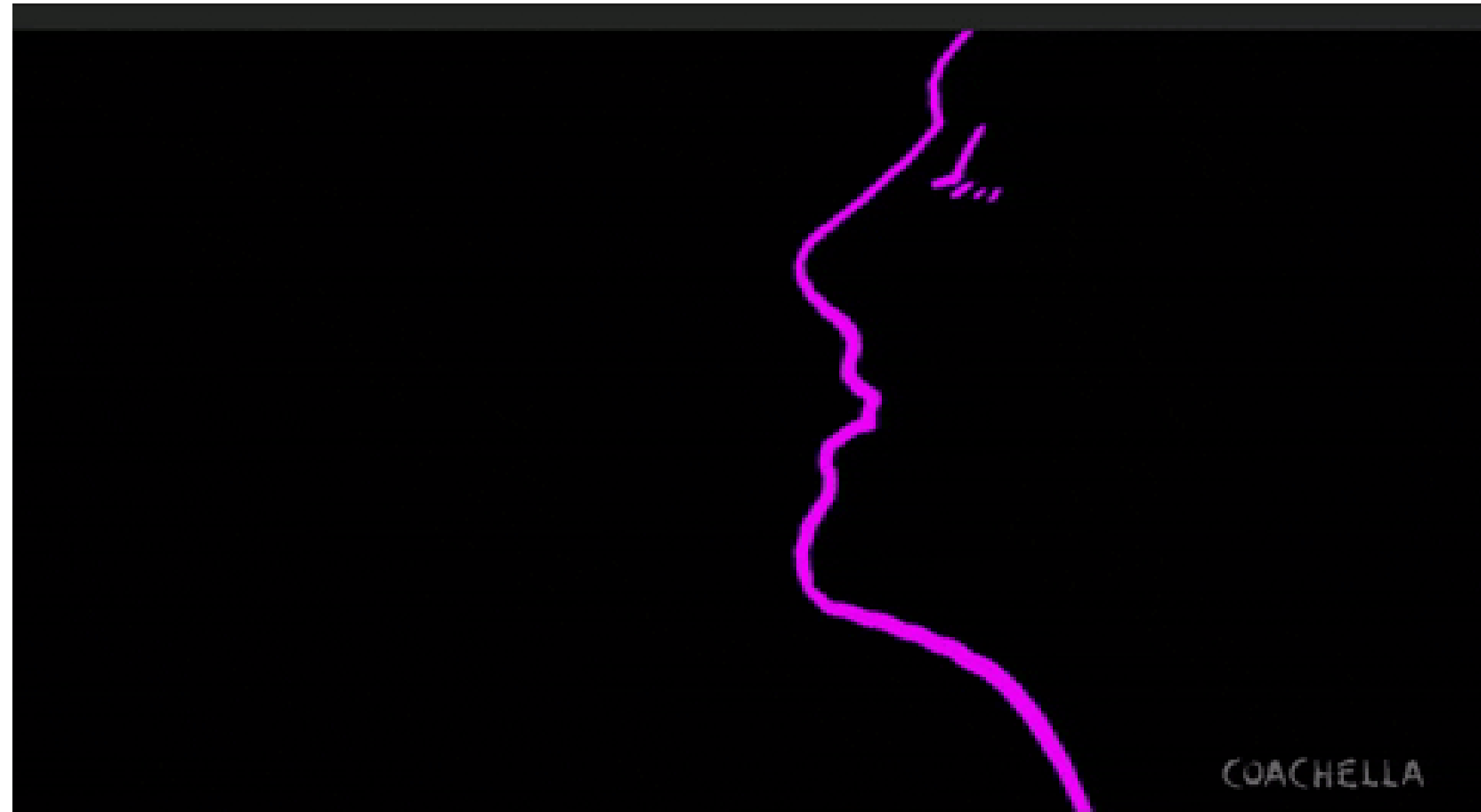
# Datasets exaples

AudioMNIST

Libriispeech

LJ Speech

VoxForge





# DNN for Acoustic modeling

## Different Types

1

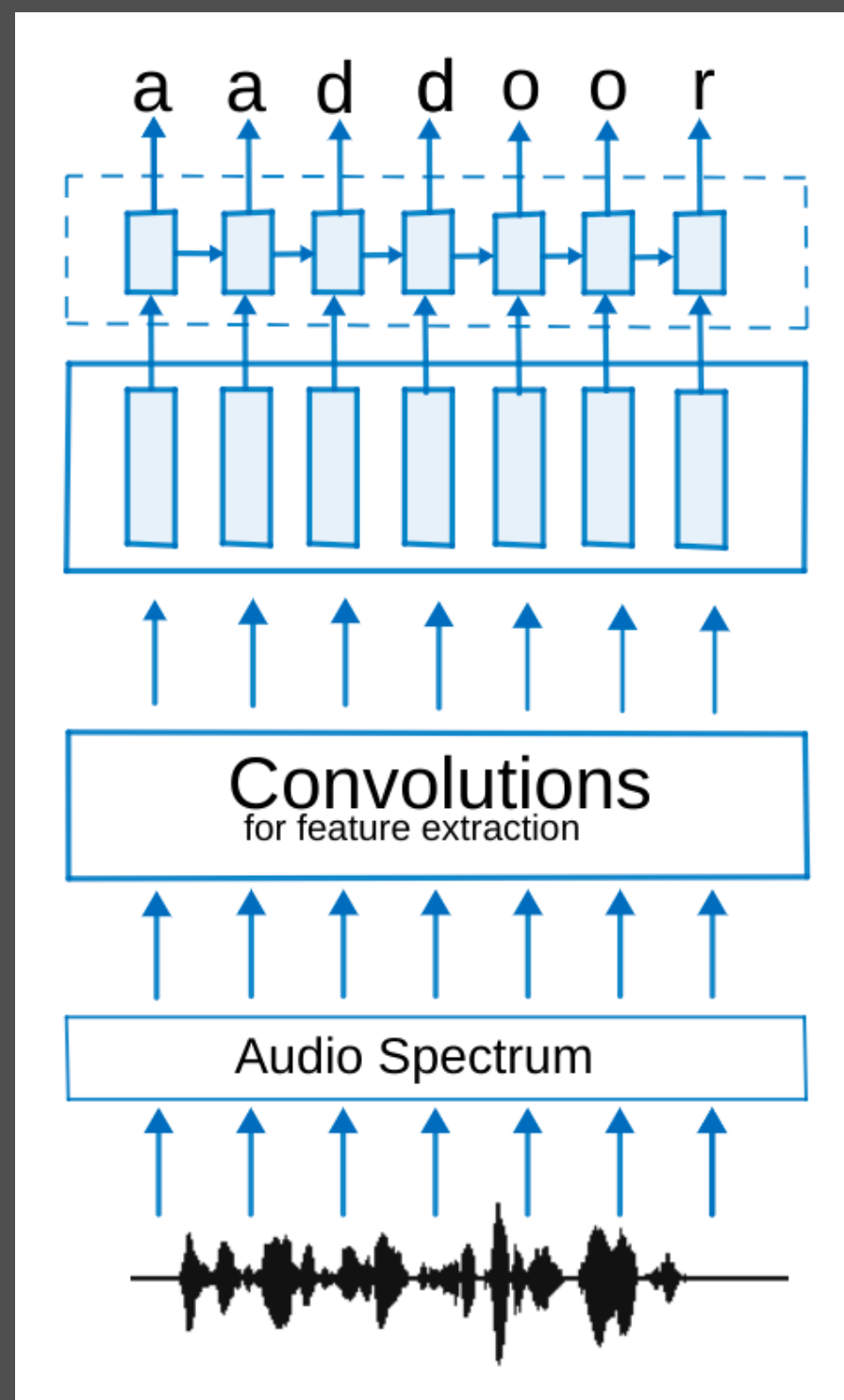
A CNN plus RNN-based architecture that uses the CTC

2

RNN-based sequence-to-sequence network

# DNN for Acoustic modeling

## Model architecture Overview

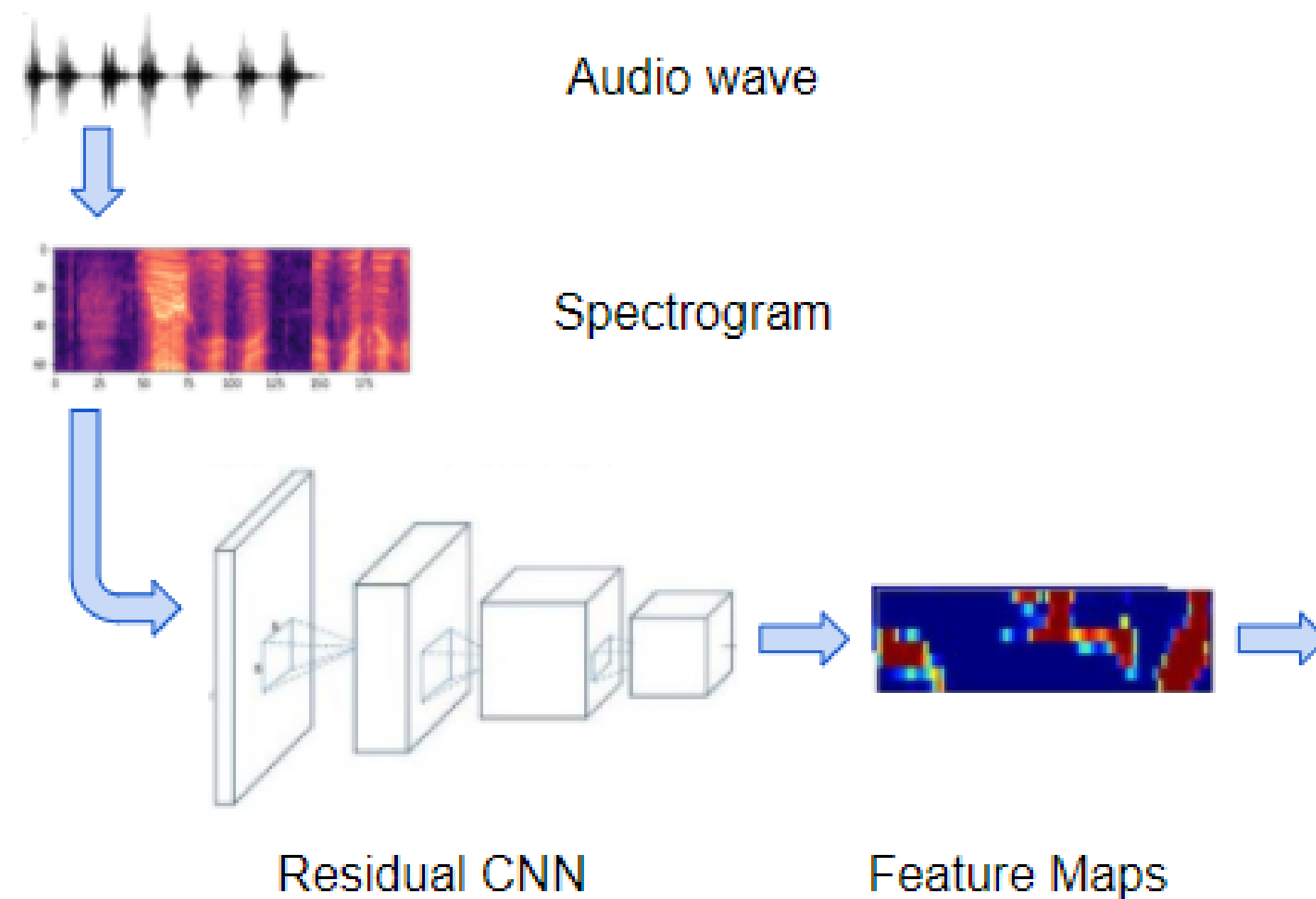


- 4 Connectionist Temporal Classification Decoder
- 3 A Linear Layer
- 2 Recurrent Neural Network
- 1 Convolutional Neural Network

# DNN for Acoustic modeling

## Model architecture Explained

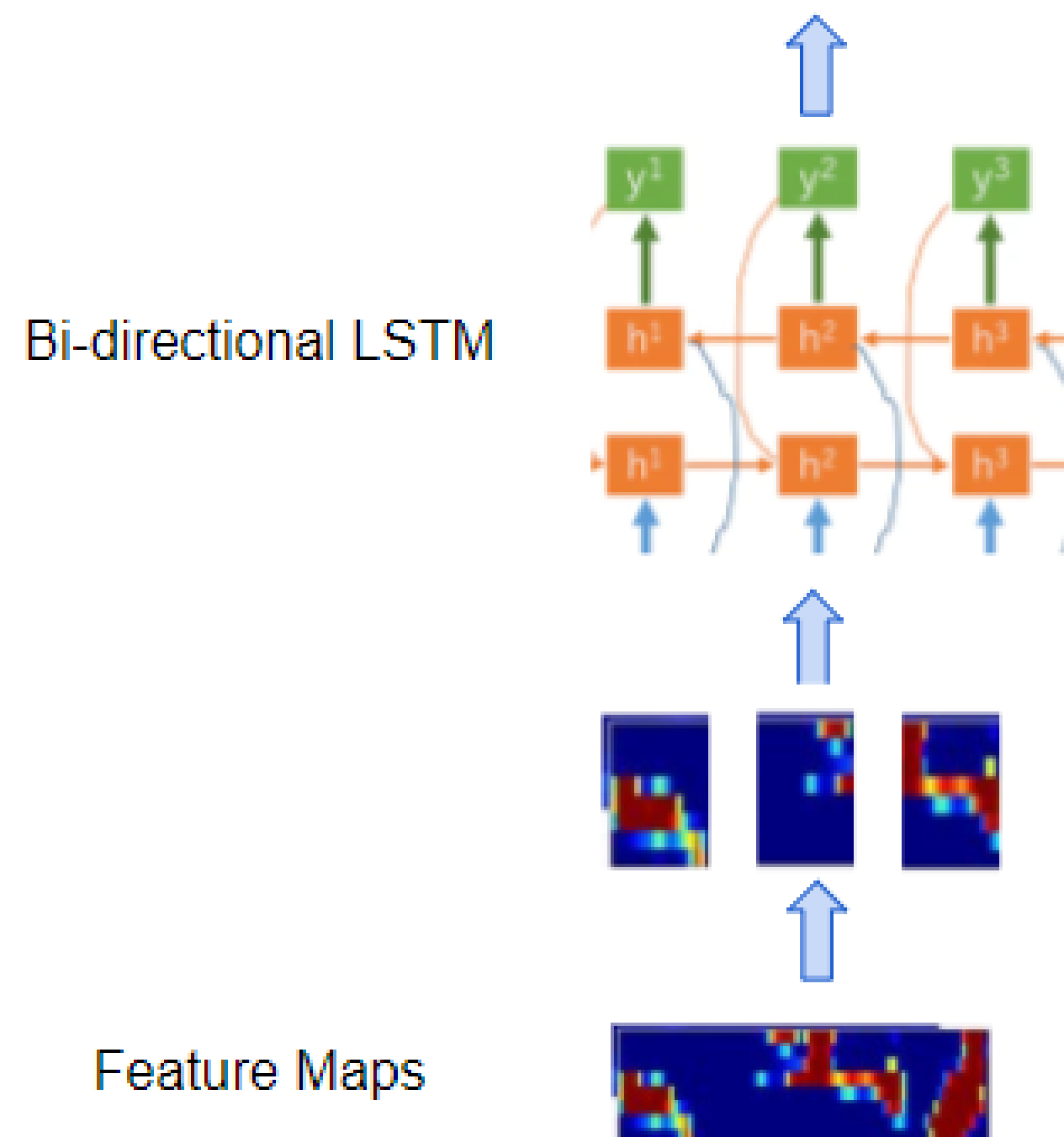
### 1 Convolutional Neural Network



# DNN for Acoustic modeling

## Model architecture Explained

### 2 Recurrent Neural Network

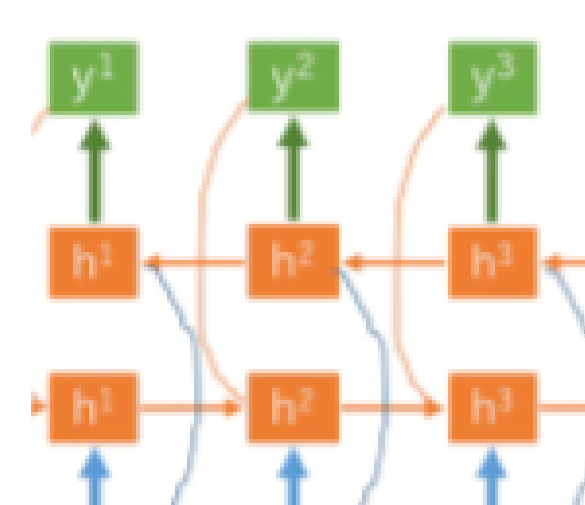


# DNN for Acoustic modeling

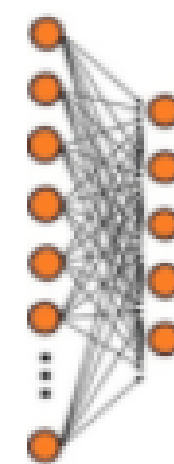
## Model architecture Explained

3

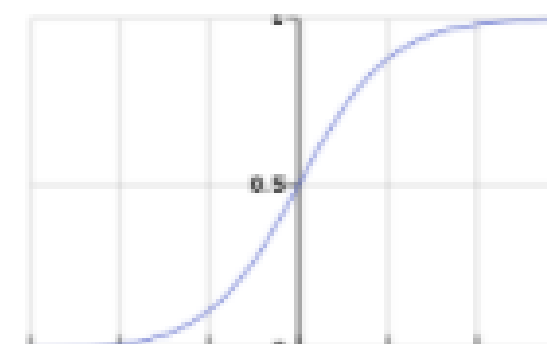
A Linear Layer



Bi-directional LSTM



Linear



Softmax




A	0.12	0.58	...	0.03
B	0.05	...	...	0.82
...	...	...		...
Z	0.75	...	...	...
-				

Character Probabilities  
per Timestep



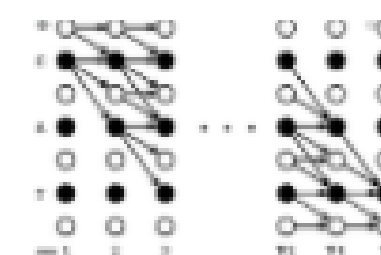
# DNN for Acoustic modeling

## Model architecture Explained



A	0.12	0.58	...	0.03
B	0.05	...	...	0.82
...	...	...		...
Z	0.75	...	...	...
—				

Character Probabilities  
per Timestep



Decoding

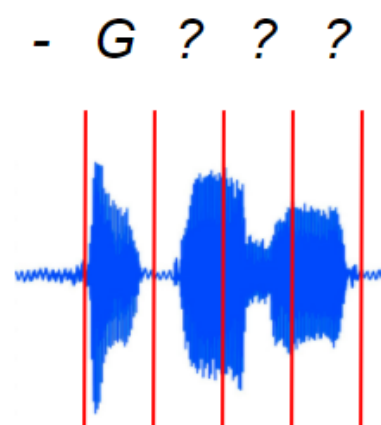
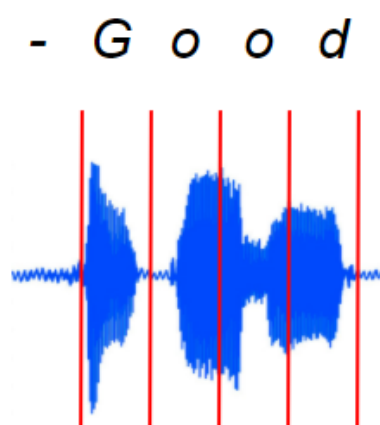


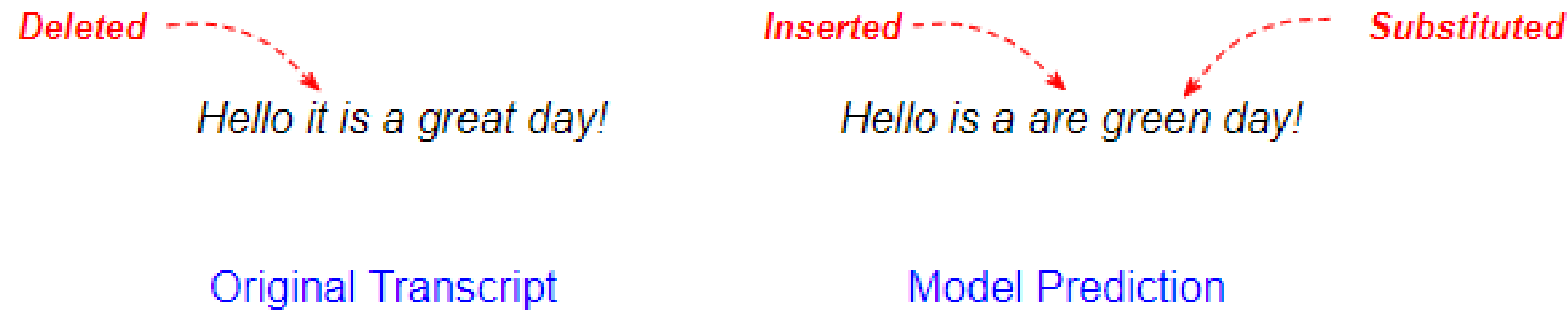
Output

*Good Morning!*

4

Connectionist Temporal Classification





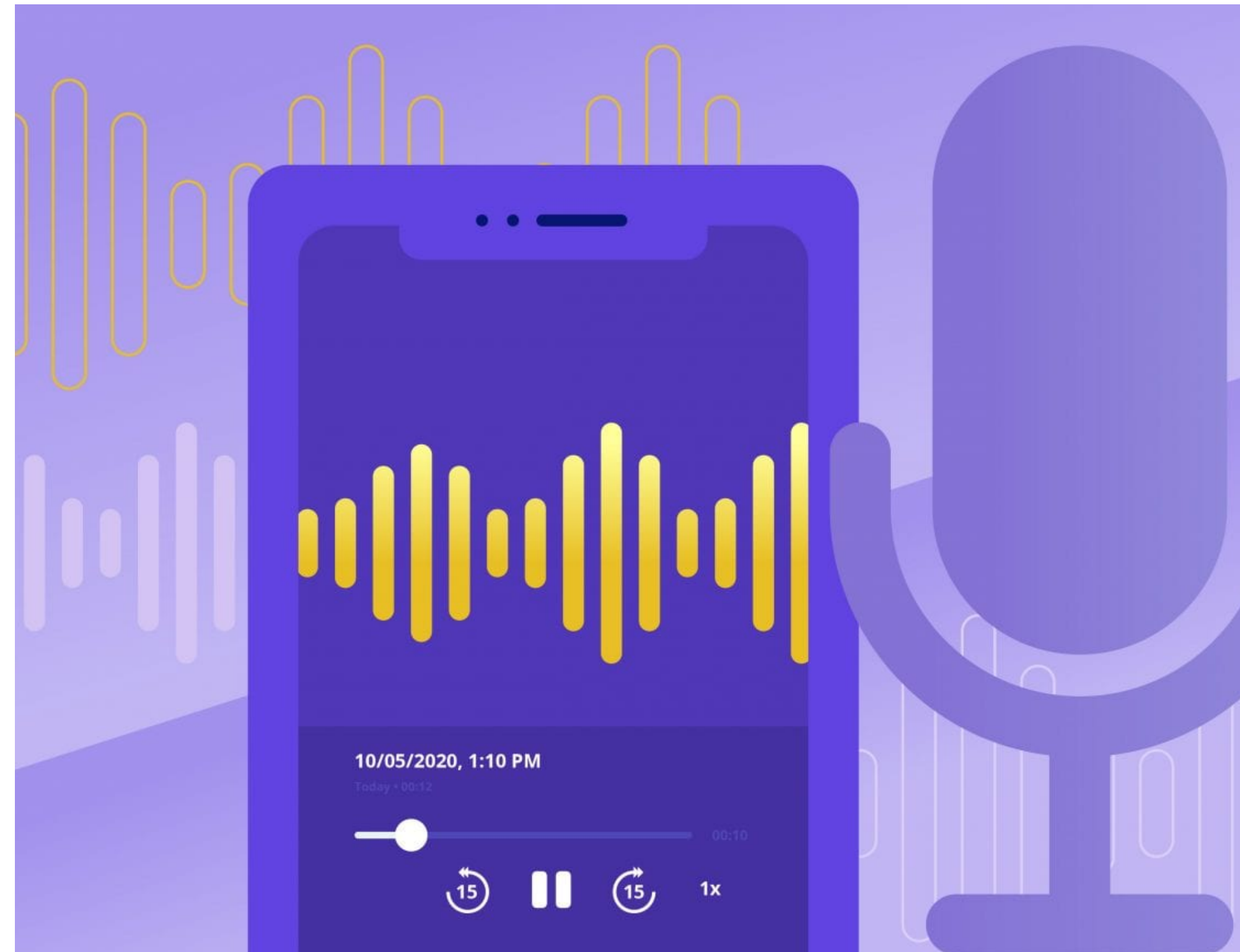
$$\begin{aligned}\text{Word Error Rate} &= \frac{\text{Inserted} + \text{Deleted} + \text{Substituted}}{\text{Total words in transcript}} \\ &= \frac{1 + 1 + 1}{6} \\ &= 0.5\end{aligned}$$

# Metrics — Word Error Rate (WER)

The metric formula is fairly straightforward. It is the percent of differences relative to the total number of words.

# Adding a Language Model

Great speech to text AI requires a great language model we recognize what words we predict, as well as pronunciation models to handle differences between accents, dialects, age, gender, and the many other factors that make our voices unique.



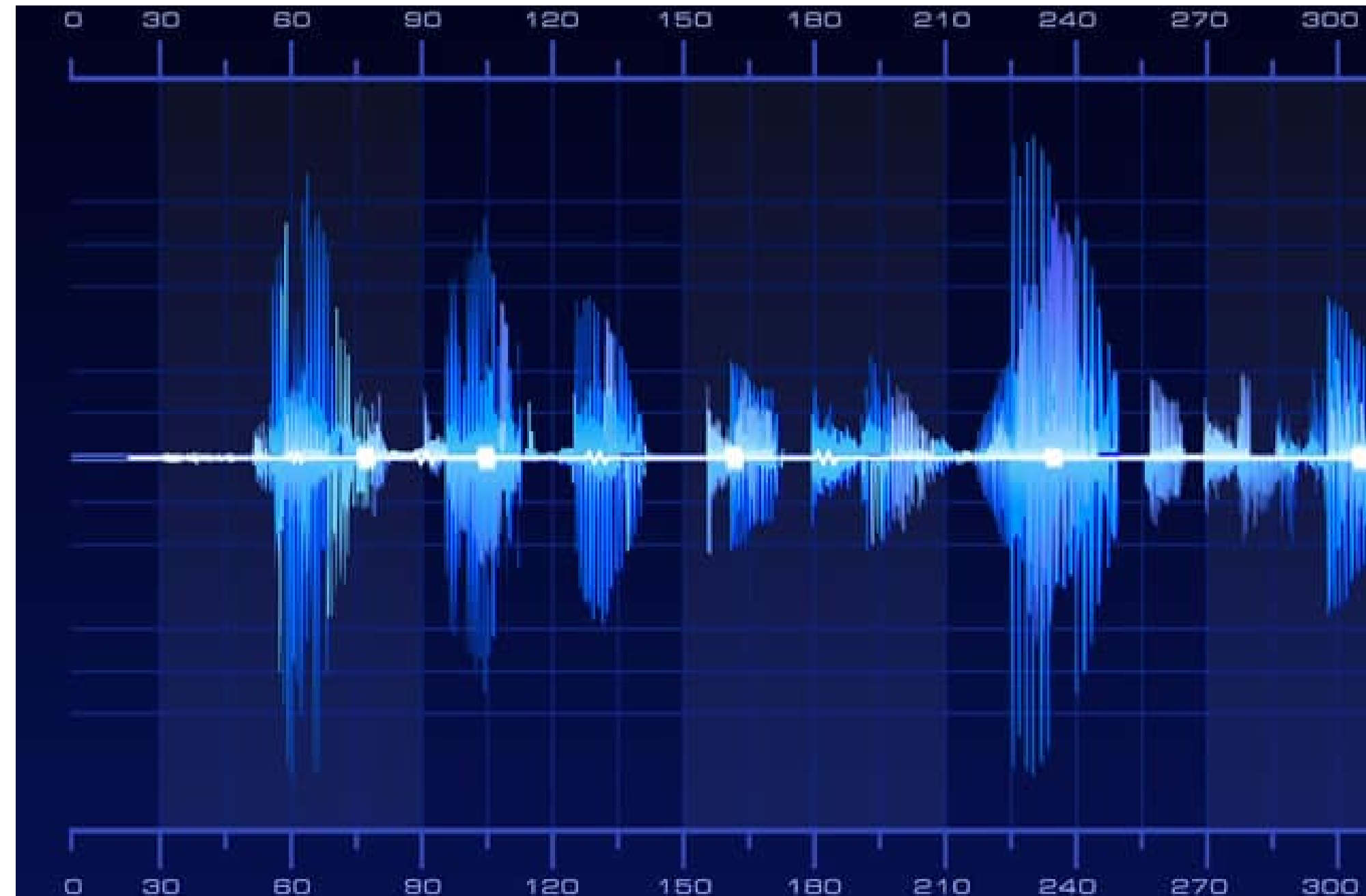


# Inverse Normalization

Reverse Normalization

Adding punctuation

Adding capitalization





**"The advance of technology is based on making it fit in so that you don't really even notice it, so it's part of everyday life."**

**Bill Gates**





**Practice Time!**

# Do you have any questions?

We hope you learned  
something new.