Abstract

In 21st century each country is looking to increase economy but no one is thinking about enviornment and problem araised because of pollution. Air pollution is one of results of industrial development due to which many suffer from respiratory diseases and die of it. That is why for past 10 years every one is doing research on respiratory diseases. One way to know respiratory disesases is by respiratory sound using Machine learning.Through this method we can detect respiratory abnormalities at early stage.But for that we need large reliable dataset. Regarding this ICHBI International Conference of health Infomatics conducted a scientific challenge with main objective of developoing algorithm for respiratory sound clasiification. Data was collected by two teams from Protugal and Greece. In dataset there was 968 recording which consists of around 7000 respiratory cycles. Total no patients were 126 from all different age group. The respiratory cycles are mixtures of crackles and wheezes sound and also other noise are present which are very high, which makes this very challenging.

CONTENT

1) INTRODUCTION

Respiratory diseases are third most death taking diseases in the world, numbers annually is around 3 million worldwide and thanks to COVID 19 which put on more burden because it mainly affect the lung through which we breathe. Many symptoms of Covid-19 like pneumonia go undetected through Computerized Tomography(CT). The situation became worse when patients are asymptomatic even after infected by virus. In this regar, the accurate detection of symtoms becames more important.so role of Artificial Intelligence is essential , through which we going to find crackle , wheezes sound in recording.It is very important to know that crackles sounds are distcontinous and it can be know in time constraint of short -range, whereases wheezes have protacted duration and so we can say that it have long term features.

Traditionally, sound atudy are supported time-frequency characterizations, like wavelet transforms and Fourier (FT). Just now other way to approach have given exciting results.let say, cepstral features are good for detecting sound in lung. WE using different approaches like multi-time scale and empirical mode decomposition. First one was used to easy the PCA for FT(Fourier transform) of signals. Second one help in giving instantaneous freuencies and points toward an area high dimensions representation.

In few way ,these approaches are often taken as similarly to the foremost recent deep learning approaches. However DP gives models which are difficult to understand.

In this project , we are using multi-time-scale features which is indicator of presence of crackles and wheezes. But these features will not use for supervised learning framework to detection.This approaches in future can be use to know condition of patients if it is serious or not. Acoording to that treatment will be provided.

To know difference between normal respiratory (lung) and unusal sounds is essential in getting results. Respiratory sounds include all the essential details connected to the physiology and pathology of the lung and airway obstruction.

Examination of the sounds produced by the human body can be traced back to ancient Egypt. Papyrus records from the 17th century B.C. have been found to describe listening to sounds within the body as a means of studying disease. Until the early 19th century, doctors used to examine their patients in this way, pressing their ears against the chest to listen to internal sounds. We call this "immediate support".

In 1817, Laennec instituted a new system he called the "auscultation médiate", which means the enjoyment of the middle man (Traite de auscultation médiate, Paris, 1817). Dr. Laennec also designed the first paper (a piece of paper made of 24 sheets) and wood stethoscopes. This not only enabled him to listen to the internal sounds without direct contact with his patient, but also provided a powerful and clear impression of the sounds.

Since the 1800s, the stethoscope has become very popular, as it has been adopted as a major medical tool. Before the end of the century, this device looked very much like today, redesigned, flexible tubing, and a solid diaphragm. Bowles and Sprague developed a combined bell and diaphragm design in 1925, and shortly after World War II, Sprague, Rappaport, and Groom experimented with this design before discovering the perfect combination of the classic double-tube Rappaport-Sprague stethoscope. Over the past decade, great strides have been made in improving the stethoscope. There are limited details of what the proper use of a chest stethoscope should be, given the differences that exist between the audience. The skipping of electronic technology and the automatic arrangement of recorded lung sounds can help overcome some of these shortcomings. New technologies that could come up with stethoscope techniques and auscultation in the era of evidence-based medicine and the medical world 2.0 have emerged. In fact, over the past decade, the Internet has become increasingly popular and is now an integral part of our daily lives. When new "Web 2.0" technology is used in health care, the terms "Health 2.0" or "Medicine 2.0" may be used.

2) Literature Survey

Learning Models

1) STFT (Short time fourier transform)

Before we understand about **STFT** it is important to know about **Fourier Transform** . When we audio samples over time , then we can only see the amplitude of the signal but if we want to see the amplitutde of specific frequency then then **Fourier Tranform** comes into the picture. So, basically Fourier Transform is a mathematical formula that allows to decompose any signal (in our case, audio signal) into its individual frequencies and the frequencie's amplitude which is also known as spectrum.

We know that in case of non-periodic signal , the signal's content varies over time. In that case , to represent a signal in the frequency domain , we take the help of window segments . We just compute several spectrum by performing fast Fourier Transform on several windowed segments of the signal. This process is called **Short Time Fourier Transform**.

2) Mel - spectogram

Mel Spectrogram is mainly derived from two words : Mel and

Spectrogram.Here Mel is representing Mel Scale . Mel Scale is a king of scaling using for graphical representation. A Mel Scale is result of some non linear transformation of the frequency scale.The Mel Scale is constructed such that sounds of equal distance from each other on the Mel Scale.

The formula used for converting normal scale frequency into Mel Scale is :

M(f) = 1125 ln(1 +f/700)

Where **f** is normal frequency

And to convert Mel Scale into normal frequency is

M-1 (m) =700*(exp(m/ 1125) -1)

Where **m** is Mel Scale frequency

Now , in case of Short Time Fourier Transfrom , when we compute FFT on overlapping windowed segments of signal , we get **Spectrogram**.

Therefore simply we can derive that **Mel Spectrogram** is nothing but spectrogram where frequencies of the singnal are converted into the Mel scale.

For extracting the Mel Spectrogram features from the audio samples , we have used function of librosa package in pyhton :

Mel Spectrogram = librosa.feature.melspectrogram (y)

where y represents input signal

other other paramaters were passed according to the requirement.

Mel Frequency Cepstral Coefficient are the feature of widely used in automatic speech and speaker recognition.These are the coefficients that collectively make up an MFC(Mel-Frequency cepstrum). MFC is the representation of short-term power spectrum of sound which is based on the linear cosine transform of a log power spectrum on a nonlinear of freqeuency. MFCC are not very robust in presence of some additive noise and therefore it is common to normalize their values in sound recognition to lessen the influnce of noise.

MFCC can be derived in the following steps[5] :

1. Taking the fourier transfrom of the signal
2. Using Triangular overlapping windows, mapping the powers of the spectrum obtained on the mel scale.
3. At each of the mel frequencies taking the logs of the powers.
4. Viewing the mel log powers as a signal , take the discreate fourier transform .
5. The amplitute of the resulting spectrum are MFCCs.

For calculating MFCC from the input signal , we have used function of librosa package in pyhton :

MFCC = librosa.feature.mfcc (y)

where y represents input signal

other other paramaters were passed according to the requirement.

3) Chroma stft

Basically the chroma value of audio singal represents the intensity of the twelve distinctive pitch classes.These classes are mainly used to study music.

The chromagram just extends the concept of chroma for including the dimension of time . Just as we see that the spectrogram is used to infer the properties about the distribution of energy of signal over the frequency and time , the chromagram can also be used to infer the properties about the distribution of energy of signal over chroma and time. We can also view chromagram as a many to one mapping of the signal strength at the frequencies which belong to the same chroma class. The STFT here describes the information about the classification of signal and pitch structure.

We can compute chroma_stft in two stages:

1.Computing the spectrogram

The first stage includes the generation of an image in time-frequency relation.It localizes the distribution of signal's energy in as a function of time and frequency.The spectrogram is one such efficient method .

2. Construction of time-chroma image Based on the spectrogram which previously computed a time-frequency image , we can construct time-chroma image using many to one mapping. For calculating chroma STFT from the input signal , we have used function of librosa package in pyhton :

chroma_stft = librosa.feature.chroma_stft (y)

where y represents input signal

4) ANN (Artificial Neural Network)

Artificaial Neural Network can be simply called Neural Network. ANN model consists of a collection of simulated neurons. Each neuron in ANN is a node which is well connected to other nodes via the links that correspond to the biological axon-synapse-dendrite connections. All the links have their own weight, which determine the strengths of node's influence on other nodes. The leftmost layer present in the netowk is called **input layer** which is used to feed the input to the model .

The middle layer present in the network is known as **hidden layer**.This layer is called hidden because in the training set , values of this layer is not observed.

The rightmost layer present in the network is known as **output layer** which gives us the output of the model.

Mathematical expression

5) CNN (Convolutional Neural Network)

A **CNN** (Convolutional Neural Network) is a kind of feed forward neural network that is generally used to analyse the data as a visual images. A CNN has many hidden layers which help in extracting information from an image:

Convolutional layer is the first layer in the process of extracting features from an image.A convolutional layer has several files that perform the convolutional operation.

Pooling layer is the layer that summerises the features present in the region of the feature map generated by a convolutional layer.It is used to reduce the dimensions of the feature maps.Therefore,it reduces the number of parameters
to learn and the amount of computation performed in the network..

Relu Layer(Rectified Linear Unit) performs an element-wise operation ans set all the negative values to zero. It introduces non-linearity to the netowork. It speeds up the training andd the gradient computation becomes simple.

Flattern Layer is used to convert the output of convolutional layers into a one-dimensional array so that it is to be inputted into next hidden layers.

Dense Layer is a popular and frequently used layer in Convolutional Neural Network. It represents a matrix vector multiplication. It is used to change the dimension of the vector.For example dense layer applies scaling, rotation , transform of the vector etc.

Dropout Layer : A dropout layer is mainly used for the regularization .This layer prevents the neural from overfitting.Dropout can be implemented on any of the hidden layers but not on the output layer.

The architecture of CNN model is similar to a Deep neural Network . Both the

models have hierarchies of interconnected neurons. They mainly differs to each
other by the their usage of convolutional layers for the processsing of image related inputs.

Python Libraries

Keras: This is an open source library that provide an interface to Neural Networks. It support multiple backends, including tensorflow, theono. It is used for building blocks such as layers, objectives, activation function, optimizers and a host of tools to make working with image and text data easier to simplify the coding necessary for writing deep neural network code.

Numpy: this is an open source library that support multi-dimension array and matrices, along with a large collection of high-level mathematics functions to operate on these array.

Sklearn: It is also known as scikit learn and it is an open source library that provide various features on machine learning like classification, regression and clustering algorithms including SVM, random forest etc.

Librosa: This is the library for music and audio analysis. It provide the building blocks necessary to create music information retrieval system.

Pandas: It is fast, powerful, flexible and easy to use open source data analysis and manipulation tool.

Matplotlib.pyplot: It provide MATLAB like way of plotting and easy to use for plotting different types of graphs to understand or analyse the data.

Model Layers

Neural Network or Deep Learning model consist of variety of different layers in it and also we have to understand the purpose of all the different layers to understand the working of our model and its significance. Model layers are like passing the information and interpreting like brain and understanding input and do some calculations to give some outputs.

Basically there are 3 types of Layers in any neural network Models:

I)

Input layer

II)

Hidden layers

III) Output layer

A. Input layer:

This is the most fundamental layer of all layers because without any input we can’t determine any kind of output/result. In simple language we can say that there is no need of any algorithm where we can’t find any input to produce results.

It depends on many type of inputs like:

Grayscale image/2D image

RGB Colored image/3D image

Sequenced Data

Input layers can perform some functions like smoothing the input data, normalization of the input data etc.

B. Hidden Layers:

Hidden layers consist of at least 1 layer other than input and output layer. There is variety of layers we can use to create a hidden layers. Its main purpose is to do some calculation on input layers and a matrix such that it can produce the output layer.

Here, we are going to discuss some of the major layers we used in our projects for better understanding of our models:

Convolution Layer

Convolution layer are the major building blocks used in Convolution neural networks.

A convolution is a linear operation which means a multiplication of set of weights with the inputs. This technique is designed for 2D input and the multiplication is performed with this 2D input and the some random 2D array of weights that is mainly called as filter or kernel.

There are some rules for better results is Filter size must be smaller than the input size.

And the output of dot product/multiplication of filter and the input is called 'feature map'.

Pooling Layer

Pooling layers is used to reduce the sensitivity of the location of features.

It means it reduces the number of parameters to learn and the amount of computation performance in Neural Networks.

Pooling layer summarized the features of feature map generated by Convolution layer. So, further operation is performed on the pooling layer means on summarized feature of feature map instead of feature map of convolution layer.

We use only 3 types of pooling layers in our models:

a) Max Pooling layer

Max Pooling layers is the pooling that selects the only feature that dominants in the feature map of in that fixed filter.

Here, we can see the example.

b) Min pooling layer

Min pooling layers is the pooling that selects the only feature that least significant in the feature map of in that fixed filter.

Here, we can see the example.

c) Average pooling layer

Average Pooling layers is the pooling that average the all the value in the filter.Here, we can see the example.

Activation Layer

Activation layer used to introduce the non-linearity into the output of hidden layers. Basically we need activation layer in neural network because neuron can’t learn with just a linear function attach to it. A non linear function will help it to learn as per difference with respect to errors.

There are too many activation function we can use in activation layer but we mainly use only 2 activation function:

a) RELU

It stands for Rectified Linear Unit. It is very common in Neural network and deep learning.

Equation: A(x)=max(0,x)

It give the output 0 if x is negative, else x.

b) Softmax

It usually used for handling multiple classes and it squeeze the output in range 0 to 1. It is also used to attain the probability of all the classes for each inputs.

Batch normalization layer

Batch normalization layer is use for normalizing the each input channel across a mini-batch. This layer first normalizes the activation of each channel by subtracting the mini-batch mean and dividing by the min-batch standard variation.After that, the layer shifts the input by a learnable offset and scale it by a

learnable factor. It reduce the sensitivity of the network initialization.

Dropout layer

Dropout layer is used for preventing overfitting the model by nullifying the some neurons or we can say that setting randomly some of the inputs unit to 0 with some frequency at each step of training.

Flatten layer

Flatten layer use to flatten the input layer. For example a input is define as the (batch_size,number_of_features,channels), after applying the flattern the output is (batch_size,new_features) where new_feature size = number_of_features*channels.

Dense layer

Dense layer is fully connected neural networks layer. This is most frequently and commonly used in Neural Network. It basically do dot product of input layer with sets of weights that is called kernel with adding some bias in it. And its result/output passes to the next layer.

LSTM layer

LSTM is stand for long short term memory. It is kind of RNN (Recurrent Neural networks). LSTM can retain the information for the long period of time. It is used for processing, predicting and classifying on the basis of time series data.

C. Output Layer:

Output layer is the last layer of the model that we created, which produce the end results

3) DATASET

3.1. Dataset Link

https://www.kaggle.com/vbookshelf/respiratory-sound-database

3.2. Overview

Total recordings: 920

Sampling frequency (4 kHz): 90 recordings

Sampling frequency (10 kHz): 6 recordings

Sampling frequency (44.1 kHz): 824 recordings

Bits per sample: 16

Average recording time : 22.5 s

Total participants: 126 (77 adults, 49 children)

Sex: 80 male, 46 female (NA: 1)

Age (mean ± standard deviation) : 43 ± 32.2 years

Adult participants(age) : 67.7 ± 11.6 years

Child participants(Age) : 5 ± 4.6 years

Adult participants(BMI) : 27.2 ± 5.4

Child participants (Weight) : 21.4 ± 17 kg

Child participants(Height) : 105 ± 30.8 cm (NA: 7)

4) OBJECTIVE

- To detect disease of a patient by observing his/her respiratory cycle.
- To pre-process raw data such that our model can easily classify disease.
- To build a working model over data, for accurate and precise result.
- To select model which can give maximum result.
- To take any necessary step for clearing raw data, inorder to achieve a well processed data.
- To achieve accuracy as high as possible.

5) Factors influencing measurements

- Age group
- Noise (Environment)
- Measuring equipment
- Chest Location
- BMI

5.1. AGE GROUP

Adults and child can't be scaled by common measuring factor like BMI. Since BMI won't give accurate information of kid . Hence we keep measuring factor for adults as BMI and measuring factor for child as height and weight. So, basically we will form two different groups supported age i.e age below 18 are often considered in child group and age above 18 are often considered as adult group. Other important factor to possess age bracket is medicine for child and adult can't be same, since immunity of kid is different from that of adult it's necessary to be scaled separately.

5.2. Noise(Environment)

Noise is one among the foremost important factor which may create lot of disturbance within the original data. there's no way we will avoid noise in practical scenario, so we'd like to seek out how to affect it. Now, since we all know noise are often there in data, we cannot avoid it and easily process the info , we'd like to preprocess the info so as to get rid of impurity. And this is often done by various techniques, since noise features a different frequency, we will filter this and achieve pure data, but this is often not 100% effective, noise could also be of frequency matching the acoustic wave frequency of respiratory record. Hence it's vital to affect it and it are often considered together of the challenge for this project.

5.3. Measuring Equipment

Nowadays, many measuring equipment are available in market. Various electronic stethoscope are available in market which may even record the respiratory sound. More over it can have different Acquisition mode which will be single channel or multi channel stethoscope.

The dataset consists of 4 sorts of measuring Equipment

5.4. Chest Location

Other very important measuring factor here is chest location. Since measurements can be taken from various chest location also the sound recorded from them would obviously be different. So it is very important to note that from which location that sound got recorded.

In the mentioned project there are seven different chest locations from where the sound is recorded. Those are mentioned in the dataset.
Hence, we have seen how can chest location play a role for respiratory sound classification.

5.5. BMI

Next important factor for classification is BMI. BMI is named body mass index and is calculated by formula weight(in KG)/height(in M). BMI of range 1.8 to 2.3 is taken into account as normal. One important factor to note here is BMI for youngsters isn't a deciding factor to mention that a toddler is fit or unfit, rather for child we should always have weight and height instead as measuring parameter.
BMI can give us the rough sketch of a patient and is extremely important factor once we are concerned for adults.

6) Challenges

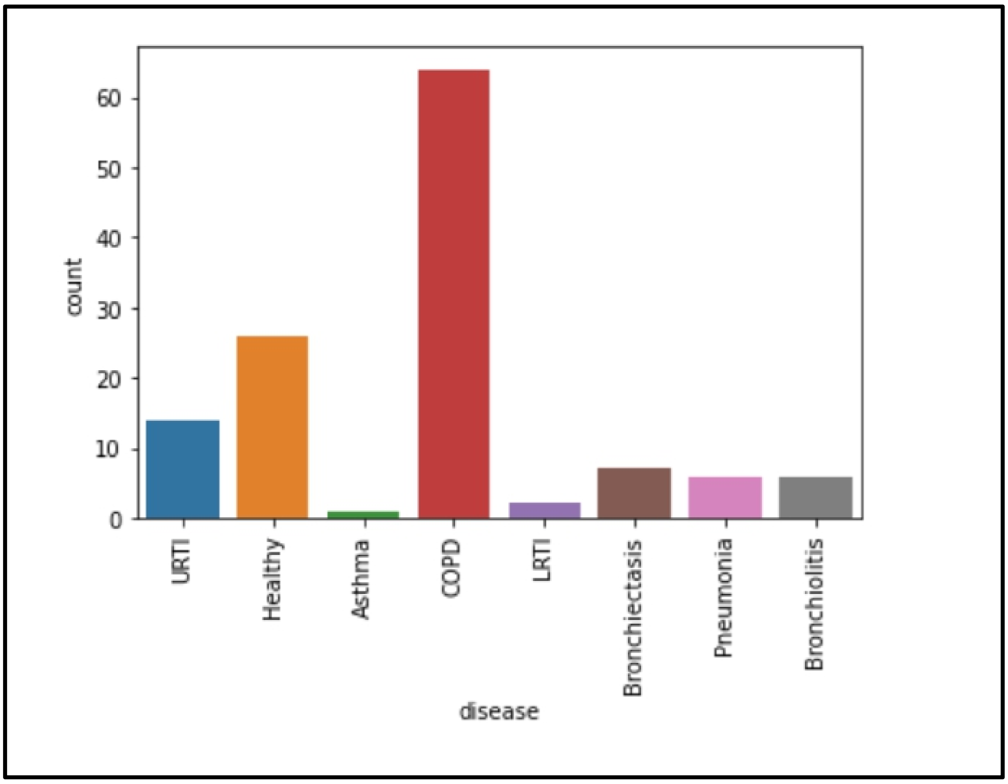
- To build a good working model over such a small data.
- To retain original respiratory sound data while removing noisy data.
- Since any disease depends on large number of parameters, the model will have high degree and it is hard to train.
- Uneven distribution where more than fifty percentage of classification is of one particular class. This can easily mislead the model.
- Also, all the sound files are not recorded from a particular chest location. Some sound files shows recording of heart, some shows recording of lung sound.
- Moreover recording equipments used were different for different recordings.

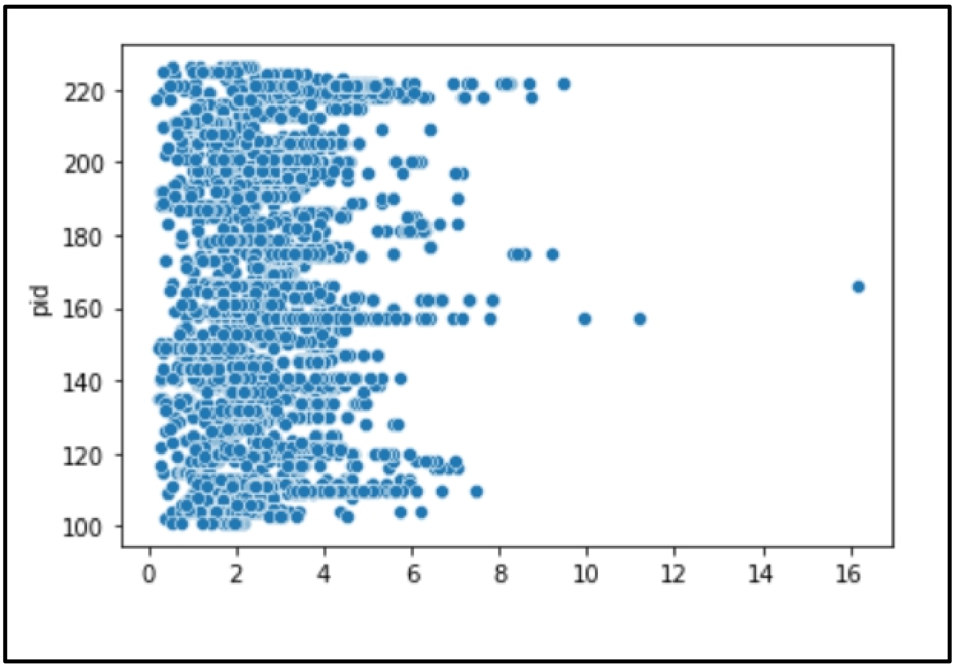
7) Solution Approach

7.1. CNN with MFCC and Mel-spectogram

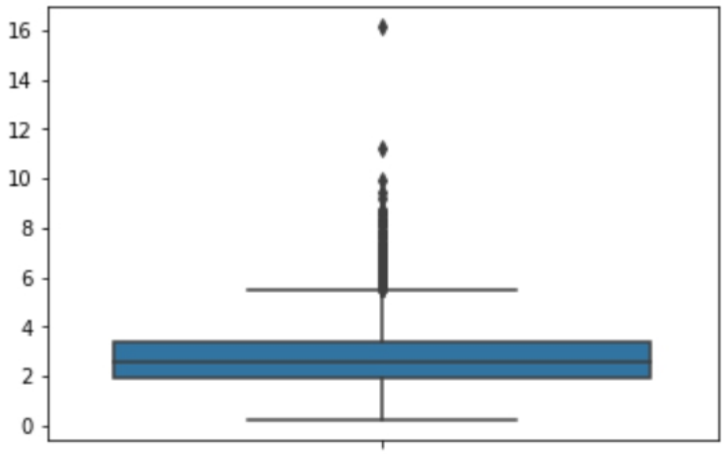
7.1.1. Analyzing Data

The first step is to study the info. For analyzing the info through we'll build various graphs out of it. the info has got to be analyzed thoroughly so as to realize a better result. Below some graphs are attached to point out how data looks and the way can we pre-process the info to realize better results and accuracy.

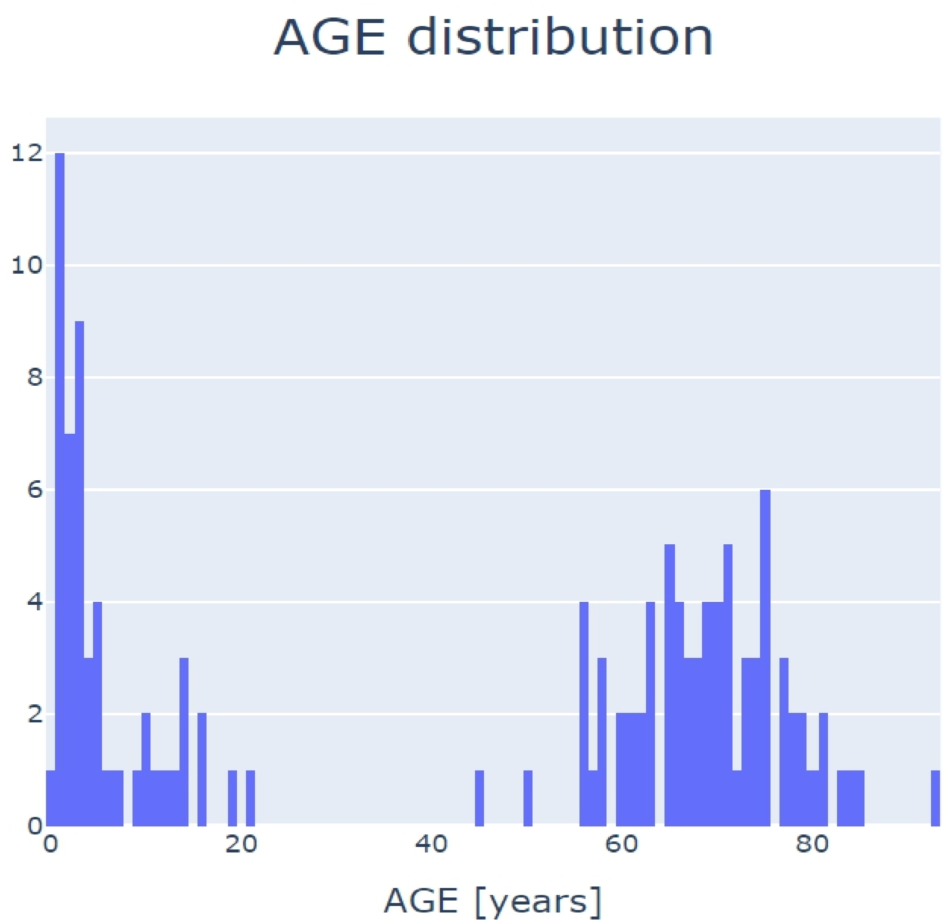




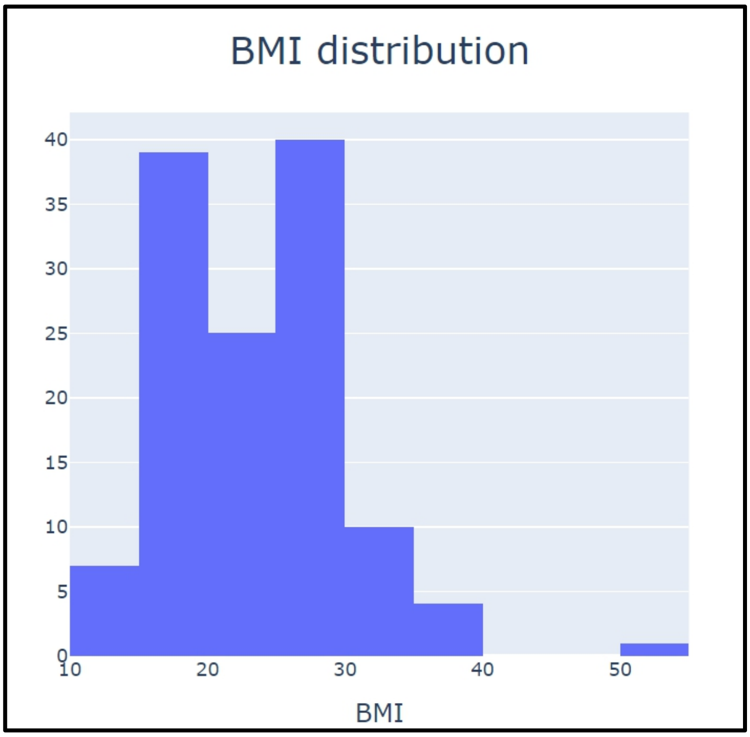
This scatter graph shows duration of each respiratory



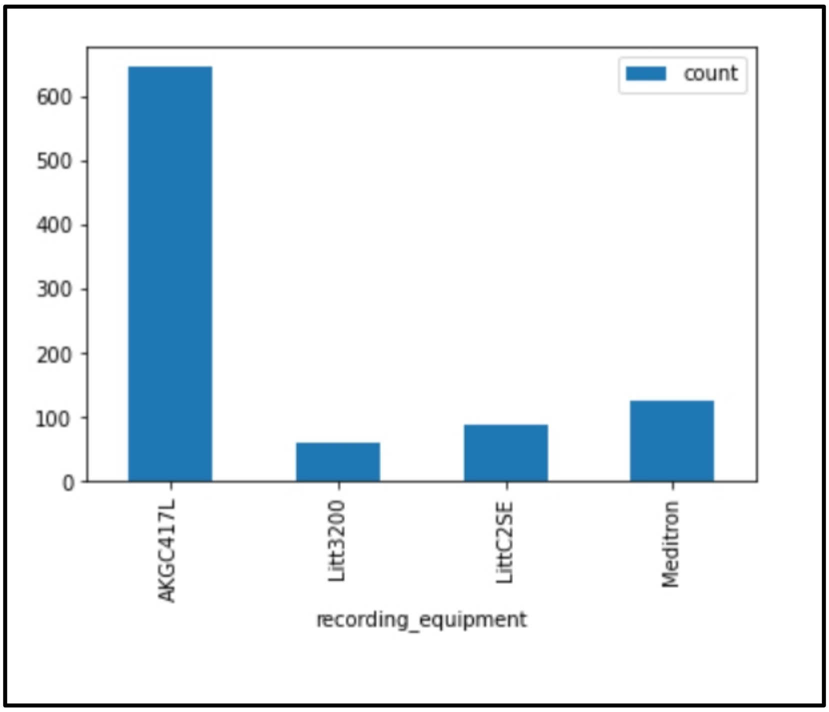
This graph shows distribution of respiratory cycle



This bar graph shows frequency distribution of age over data. From this graph we can observe that data is not well distributed over age, and this is not a good sign.



This bar graph shows frequency distribution of BMI,there are quite good percentage of patients with BMI in range 15-30 which is good as most of the human being have BMI in this range.



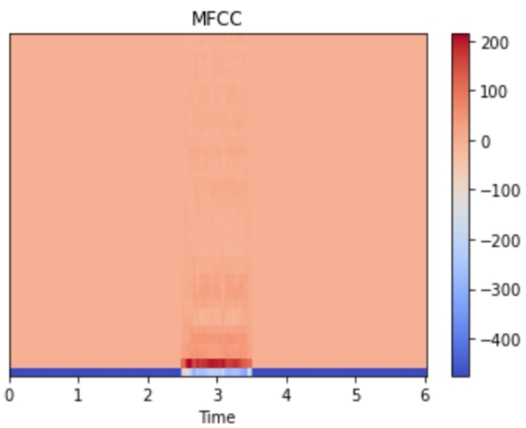
This bar graph shows frequency distribution of 920 recording by different recording equipment.

```
COPD              63
Healthy           26
URTI              14
Bronchiectasis     7
Pneumonia          6
Bronchiolitis      6
LRTI               2
Asthma             1
Name: Diagnosis, dtype: int64
```

This shows distribution of 126 patients into disease. This clearly shows that 50% of disease is COPD, and diversity of disease is very bad in this data.

7.1.2. Feature Extraction

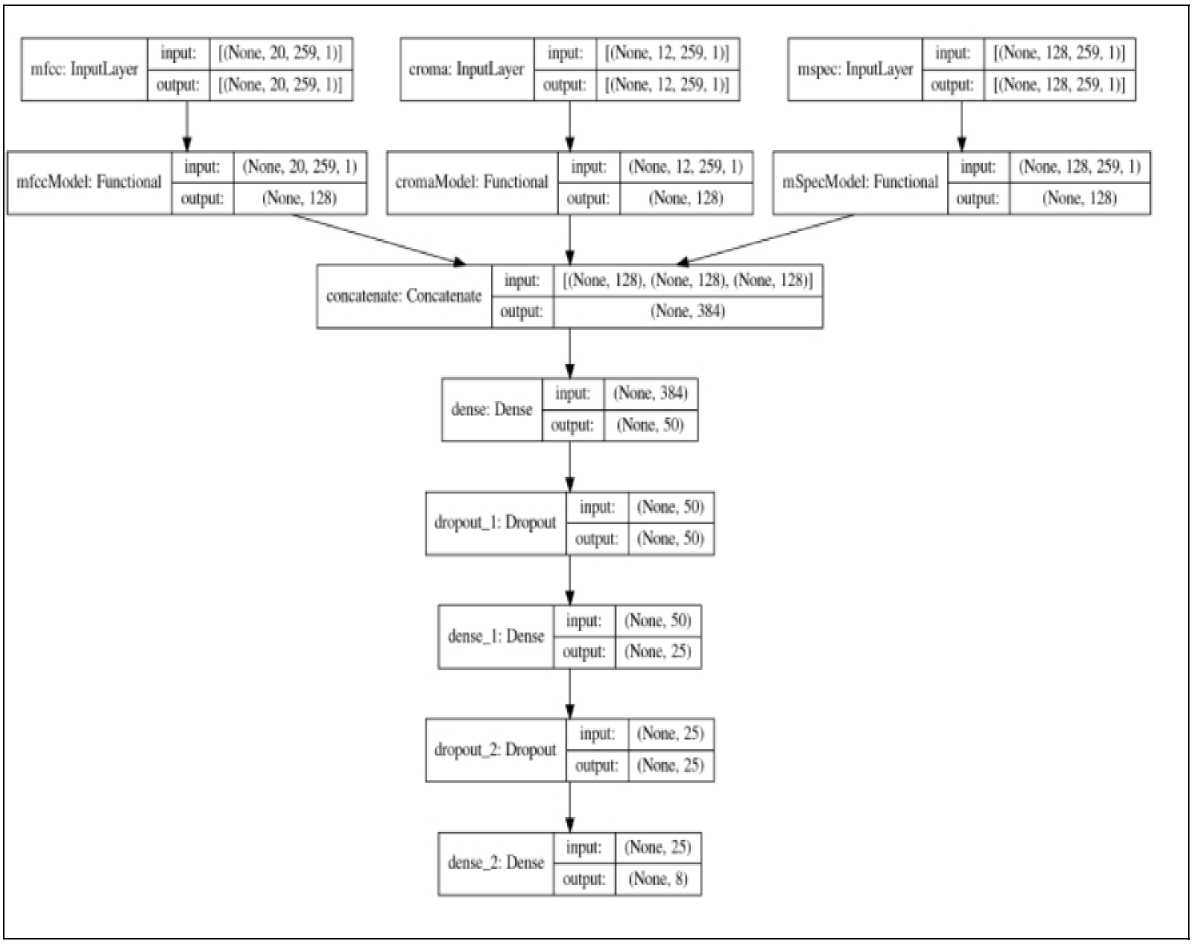
Since we've to use CNN model for our data, and that we know that CNN doesn’t perform well for sound data. Hence, we'd like to convert that sound data to image data. For this we might extract feature from sound. Here may be a graph showing how image takes care of feature extaction of sound through MFCC.



7.1.3. Model

We have tried various model for the preprocessed data. and located that maximum accuracy is achieved once we concat three separate model and form new model. Hence, we are getting to concat MFCC, Chroma-stft and mel-spectogram model to realize higher accuracy.

We would first train our data with MFCC model, then with Chrom-stft model then with mel-spectogram model and eventually concat the result. Summary of model are often seen in graph below.



7.2. KNN, RandomForest, DecisionTree with crackles and wheeze classification

7.2.1. Pre-processing (I.e crackles and wheeze detection)

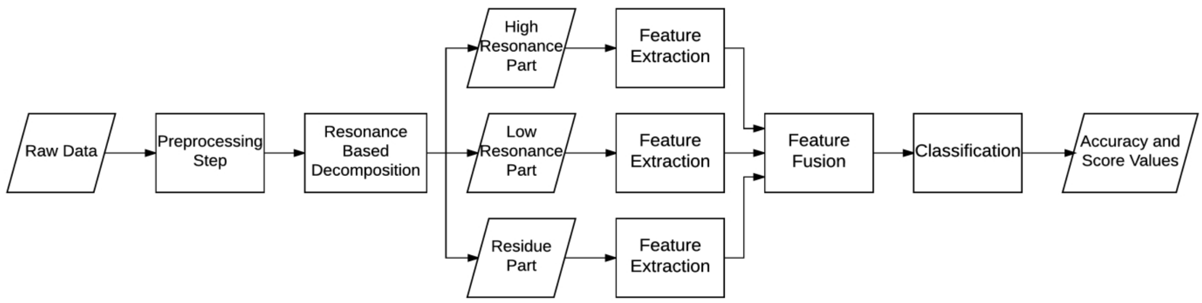


Fig. Step by step flowchart of annotation

Step 1) Resample lung sound to 4000 Hz

Step 2) Filter frequency ranging in 120 to 1800 Hz via Butterword bandpass filter

Step 3) Remove cough, intestinial sound, heart sound, motion of sthestoscope

Step 4) Use resonance based signal separation because respiratory sound have overlap of low and high frequency component

Step 5) Crackle sound is short of around 20ms where as wheeze sound is long of around 100ms. So it is easy to distinguish crackle sound and wheeze sound by observing time duration.

Step 6) Since this procedure is very complex, we would like to reduce its dimentionality with help of PCA technique.

7.2.2 - Model

Using Keras library we will choose various model and train our data supported that model. this is often super easy and fast. Once we detect crackle and wheeze sound we converted sound data to numeric data and that we have lots and much of model for training numeric data.

The primary step is to divide data for training and testing purpose. Next step is to separate Y_train and Y_test for training and verification for this we may use LabelEncoder to convert categorical data into numeric data which is straightforward to interpret. Here we'll show one model for better understanding.

```
dectree_clf = DecisionTreeClassifier()
dectree_clf.fit(trans_train, test_trans)
pred = dectree_clf.predict(train_test)
accuracy_score(trans_test, pred)
```

Next step is to print confusion matrix, in order to get the clear picture of how good our model is performing.

confusion_matrix(labels_train, pred)

```
array([[  0,   0,   0,   0,   0,   0,   1,   0],
       [  0,  12,   0,   0,   0,   0,   0,   0],
       [  0,   0,  12,   0,   0,   0,   0,   1],
       [  0,   0,   0, 711,   0,   0,   4,   0],
       [  0,   0,   0,   0,  27,   0,   0,   3],
       [  0,   0,   0,   0,   0,   1,   0,   1],
       [  0,   0,   0,   4,   0,   0,  29,   0],
       [  0,   0,   5,   0,   4,   0,   2,  11]])
```

Confusion Matrix

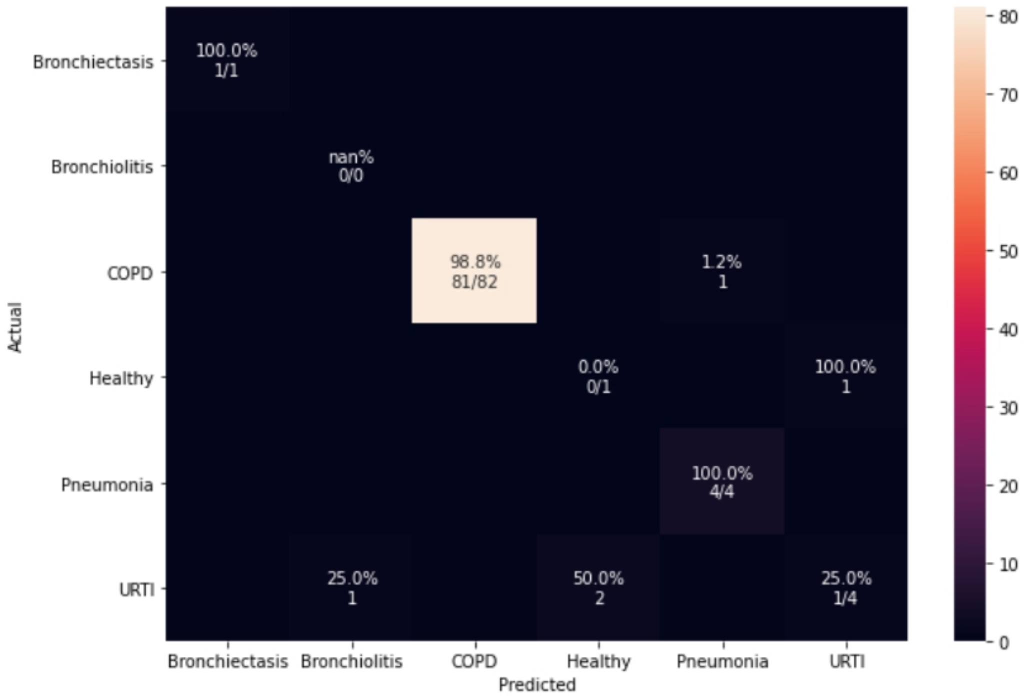
For better representation of confusion_matrix, we can take help of seaborn library which has one beautiful feature named heatmap.

8) Results

8.1. CNN along with mfcc, chroma-stft, mel-spectogram

ACCURACY: 95.8%

Confusion Matrix:



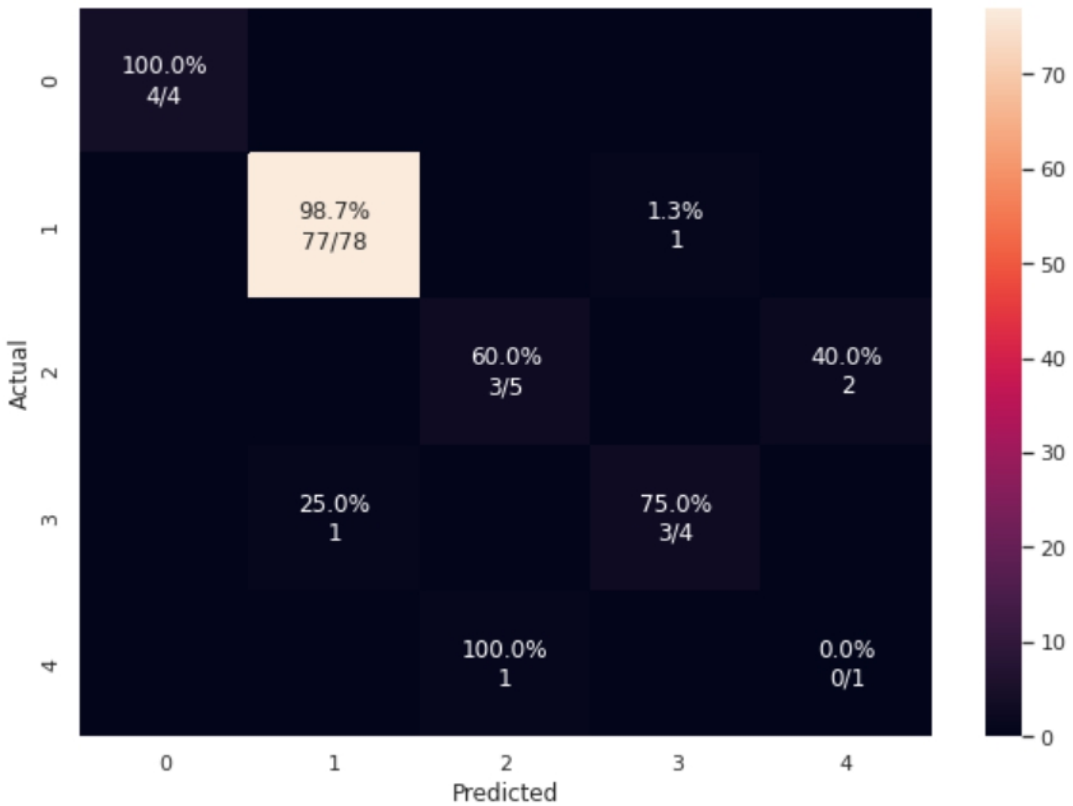
CNN

It is a artificial neural network which uses data as a visualizing factor. It consists of many hidden layer in order to correctly predict the information in image.

8.2. KNN with crackles and wheeze detection

ACCURACY: 89.1%

Confusion Matrix:



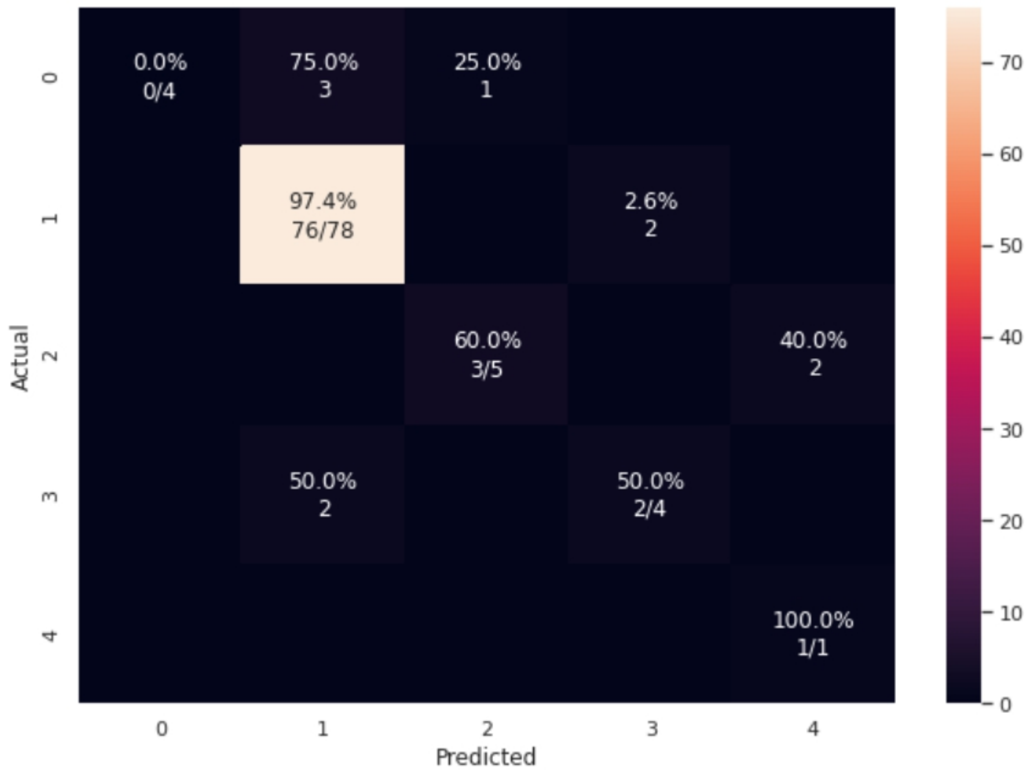
KNN

A k-nearest-neighbor algorithm is an approach to data classification that estimates how likely a knowledge point is to be a member of 1 group or the opposite counting on what group the info points nearest thereto are in.

8.3. Random Forest with crackles and wheeze detection

ACCURACY: 94.5%

Confusion Matrix:



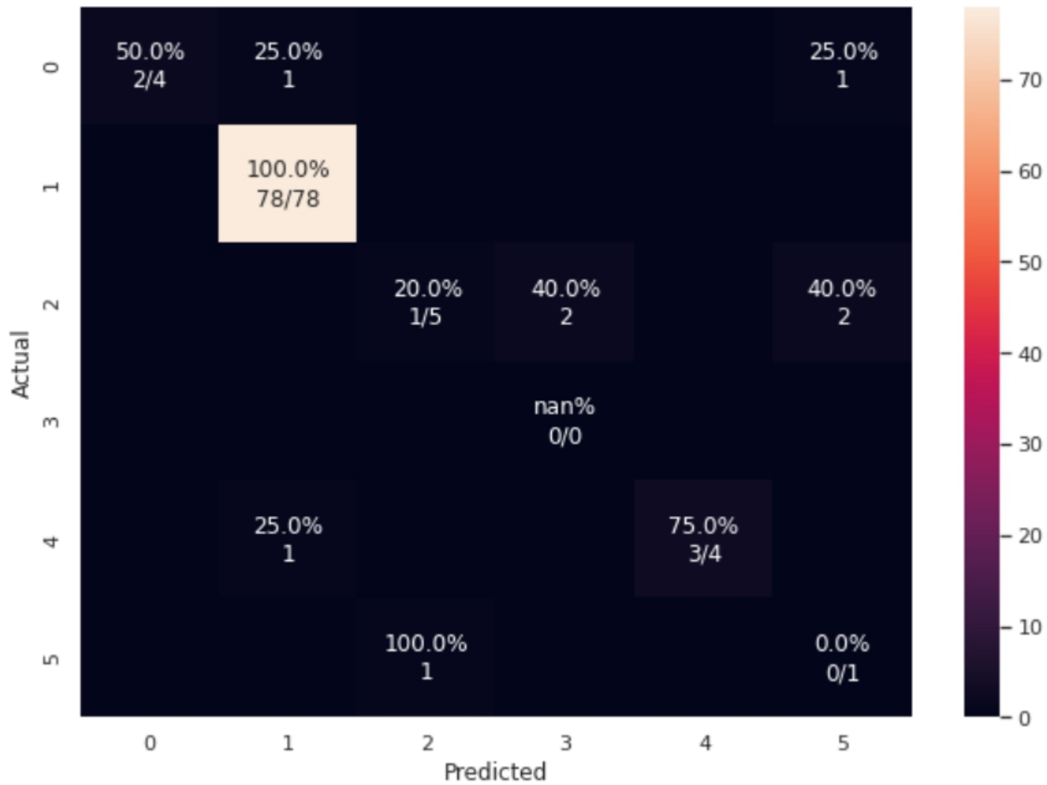
RANDOM FOREST

Random forest is an algorithm of machine learning which can be used for classification or regression. Basically it is similar to decision tree and we can say that it has lots of small decision tree in it.

8.4. Decision Tree with crackles and wheeze detection

ACCURACY: 93.4%

Confusion Matrix:



Decision Tree

Decision tree is a kind of tree which is used for classification purpose. It has a node from where we have to start and depending on result we move forward to leaves. Once we reach the leave of tree we are done with classification and decision tree would help us in result. Decision tree are built over parameter called gini index or entropy.

9) Conclusions

Hence we are able to conclude from results that various models have give us pretty good results, inspite of varied challenges that were with our dataset. Results of varied models will be seen in result section. We were able to touch the accuracy of 95-96% which is extremely good for this project.

10) FUTURE SCOPE

- This technique can be used in future to detect any respiratory disease in large scale.
- Already many studies are going on and various AI experts are working on this project during last decades.
- It will reduce human effort, save time and money.
- It will be more reliable since model would have huge amount of dataset that any human doctor ever had.
- Accuracy and precision would be much more.
- Once this technique gets accepted, modifications can be done so that it could also prescribe suitable medicine.

11) References

1. http://erj.ersjournals.com/content/42/3/559
2. http://rc.rcjournal.com/content/59/5/765.short
3. http://rc.rcjournal.com/content/54/12/1717.short
4. https://www.oxford.dec.nihr.ac.uk/files/reports-and-resources/horizon-scanning-report0016-automated-lung-sound-analysis.pdf
5. https://doi.org/10.1371/journal.pone.0177926
6. http://erj.ersjournals.com/content/8/12/2139.short
7. http://doi.org/10.1378/chest.73.3.333
8. https://dx.doi.org/10.1007/s40846-016-0161-9
9. http://ieeexplore.ieee.org/document/7591526/
10. https://doi.org/10.1016/j.compbiomed.2009.06.011
11. https://www.researchgate.net/profile/Rui-Pedro-Paiva/publication/330815510
12. https://eden.dei.uc.pt/~ruipedro/publications/Conferences/ICBHI2017a.pdf
13. https://www.sciencedirect.com/science/article/pii/0094576579900614