Abstract / Resum / Resumen (1/3 de página, 1 página o 1.5 paginas los 3 idiomas)

1. intro

1.1 history digital audio

1.2 history audio classif.

1.3 sota audio classification

1.4 summary of the proposal here (que vas a hacer y pq)

2. Method

2.1 MFCC

2.2 Feature vector representation

2.2.1 naive todo concatenado

2.2.2 histogramas por componentes

2.3 Dimensionality reduction – PCA

2.4 Feature relevance (dec. trees based)

3.Evaluation design

3.1 Dataset (cuantos generos)

3.2 Evaluation protocol (cuantos datos traib, test fold, accuracy como metrica de evaluacion...)

3.3 Methods and parameters: MFCC: cuantas componentes, 2.2.2 cuantos bins de los histogramas vamos a testear, PCA cuanta informacion.... y decir que para evaliar 2.2 usaremos SVM linear, SVM RBF, adaboost, y RF. y sus parametros

4. Results

4.1 Classication results (graficas.... matriz de confusion y que generos se confunden más etc.)

4.2 Feature relevance analysis

5. Cnonclusions

References

**2.1 MFCC**  
After investigation, we found out that most of the projects involving audio analysis, were using MFCC to extract features from the audio files.

MFCC (Mel Frequency Cepstral Coefficients) is usually used to extract features from human talk, but has been used lately for all kinds of sound.

MFCC were defined by Paul Mermelstein and S. Davis in 1980.

Although it was first developed to recognize monosyllabic words in spoken form, its characteristics make it useful for all kinds of sounds.

The algorithm works as follows:

1. Divide the signal in several same-sized intervals.
2. Take the Fourier transform of each interval.
3. Convert the values obtained before to Mel Scale.
4. Calculate the logarithm of every Mel frequency.
5. Apply the discrete cosine transform to all Mel logs.

Using this, we will end up with a matrix which size will be determined by the number of coefficients we want and the length of the audio sample.

Considering the difficulty of these steps, we looked into ways of applying them in a simpler way, and we ended up finding that most of the projects involving MFCC use a Python library named “librosa”.

This library gives us the majority of audio analysis tasks already built in, so we only need to tweak the parameters we need to get the information we need out of every song.

The functions we will use for our project are the following:

* librosa.load(): This function loads the audio file, modifying the properties of the file we need to have all files following the same standards.  
  The most important parameters we need are:
  + sr: changes the sample rate
  + mono: converts the file to mono-channel
  + duration: crops the song into a smaller length.  
    The size of the matrix depends on the length of the file, so we need to make all songs last the same to work with them.
* librosa.feature.mfcc(): calculates the MFCC of the audio file we have loaded.  
  The function automatically tweaks all the parameters it needs to make a small enough matrix, but without losing huge amounts of information.  
  In this case, each interval is about 0.02 seconds long.

**2.2 Feature vector representation**   
Once we have extracted the features using MFCC, we have to decide what are we going to do with them, given that the amount of features we get will always be, in our case, bigger than the dataset we can work with.

We will work with two different representations of these features: using all the raw data and creating histograms of each component.

2.2.1 naïve  
The first method we will try will use all the values we get from MFCC.

This method will take the matrix whole matrix and convert it into a 1-dimensional array, created by concatenating each row, which size will depend on the length of the song, one after another.

This way, we will have our dataset converted into a matrix of as many rows as songs it has by the length of each array.

The amount of information we will have to work with will be enormous, but we will use it to have a first approximation of the accuracy of our classifier.

2.2.2 histograms  
As we said before, we want to reduce the amount of values we have, but being able to still have the most information we can, as well as remove the effect of time in our experiment.

In order to do that, we will have as many histograms as coefficients we use, and will be built following this procedure:

1. Take maximum and minimum values of all dataset.
2. Divide the interval in as many steps as you want.
3. Create a histogram for each coefficient.
4. Put every value of the corresponding row into its interval.
5. Divide every final value by the amount of values you have.
6. Concatenate each histogram into a 1-dimensional array.

This way, each song will be represented by an array with its size depending on the number of coefficients and the amount of steps we take.

2.3 Dimensionality reduction – PCA  
Once we have both representations of the feature vector, we will try one last modification of it.

This will be done by applying PCA to the matrix we have, which will reduce the size of it even more.

The objective of this procedure is to have a feature vector smaller than the size of the dataset, which we expect it will help classification.

PCA: [definition and procedure]

This will be done using sklearn python library.

[Explanation of the functions used]

**3.1 Dataset**  
For the realization of the project, we needed a large set of songs and genres to be able to train our algorithm in a proper way.

Initially, we wanted to use a relatively small amount of songs (100) of 4 different genres, all of them royalty free, taken from Free Music Archive. The problem was that the set we ended up with was too small to make the program work as intended.

We decided to change the set to an already made one, so we looked for data sets build for our purpose and ended up finding Marysas, a website in which we could find 1000 songs of 10 different genres (100 songs per genre), all of them 30 seconds long and with a similar set of properties (which will be explained later).

All the music in the data set is available for everyone and it can be used for investigation without any charge.

All songs are “.au” files, which is a format used by the program Audacity.

To work with them, we need to know a few basics of digital audio, so I will explain what each one of the terms we will need when we extract the features of each song.

* Audio frame: Contains information in a given time.
* Sample rate: Number of samples taken from a continuous signal in order to produce a discrete signal.
* Channels: Number of streams in which the audio is sent.
* Frame size: Size of each frame. Sample rate \* # of channels.
* Frame rate: Number of frames per second. Frame size / s.

In our data set, all songs have the following properties:

* Sample rate: 22050Hz
* Channels: 1 (Mono)
* Frame rate: 22050 fps

To make the program able to work with other formats and songs, we will take all this information when we extract the features.

This is accomplished forcing the load function from librosa to take the Sample Rate as 22050 and converting the signal to Mono-channel.