IOSR Journal of Computer Engineering (IOSR-JCE)
e-ISSN: 2278-0661,p-ISSN: 2278-8727
PP 08-12
www.iosrjournals.org

Music Plagiarism Detector

Fawaz K, Dr.Rekha K James

Division Of Electronics Engineering School Of Engineering Cochin University Of Science and TechnologyKochi-682022, Kerala, India
Department of Electronics and Communication Government Engineering college Wayanad Mananthavady-670644, Kerala, India

Abstract: *Music plagiarism detector is an alternative approach for music genre classification which converts the audio signal into spectrograms and then extracts features from this visual representation. The idea is that treating the time-frequency representation as a texture image we can extract features to build reliable music genre classification systems. The proposed approach also takes into account a zoning mechanism to perform local feature extraction, which has been proved to be quite efficient. The feature include MFCC, energy, entropy etc. In this we suggest a method for indexing the features extracted from every melody and a method for processing plagiarism detection by using the index. With our plagiarism detection system, composers can easily search for the melodies similar to their ones from music databases. Furthermore, for applications such as radio monitoring, search times on the order of a few milliseconds per query are attained, even on a massive music database*

Index Terms: *Music listening, music quality measures,Mfcc, Energy,entropy.*

I. Introduction

With the development of digital media technology, people are routinely exposed to music in everyday environmentscar, home, restaurant, theater and shopping mallbut are frustrated by not being able to learn more about what they hear. They may, for instance, be interested in a particular piece of music and want to know its title and the name of the artist who created it. In this project Each audio file is fingerprinted, The fingerprints from the unknown sample are matched against a large set of fingerprints derived from the music database. The candidate matches are subsequently evaluated for correctness of match. Music-plagiarism is the use or close imitation of an-other authors music without proper acknowledgement. Every year ,vast numbers of new music tracks are released globally, and questionable similarities exist in some sections of music tracks. Aided by the internet, plagiarism is now noticeable globally ,not just across authors, but also across languages and countries. In 2008 alone, 1.4 billion music tracks were sold internationally. This number has since increased to over 1.8 billion. In2004, the SACEM, an organization that seeks to protect the rights of the original authors, composers and publishers, was able to manually check only a small percentage of registered pieces for potential copyright violations. With such vast numbers of tracks to monitor, the need for automatic techniques for identification of potential copyright violations and detection of music-plagiarism is clear and paramount

II. Types Of Plagiarism

A clear and precise definition of the term plagiarism is difficult to derive. However, the notion of intellectual property is known since ancient times. A poet named Martial called another poet kidnapper (lat. plagiarius), because he presented Martials poems without permission and claimed that these were his own. This incident is perceived as the first mention-ing of authors rights, even though an established copyright was unknown . Authorship became more important with the invention of letterpress printing in the late 15th century. In the realm of music, it became common practice to credit the composer for his sheet music since the 16th century.

2.1. Sampling Plagiarism

The term sampling describes the re-use of recorded sounds or music excerpts in another song . The samples are often manipulated in pitch or tempo to fit the rhythm and tonality of the new song. It is very common to mix additional instruments to the sample, such as additional vocals or drums. The most common use of samples is to crop an excerpt of one or more bars and loop them. More elaborate forms of sampling include rearrangement and post-processing of the respective sample beyond recognition. Sampling has strongly influenced popular music culture. Thus, there exist websites4,where sampling cases are collected by a community of music aficionados. Due to the fact that sampling is basically the use of a song in a song it is related to the task of cover song detection. Cover song detection is commonly approached by chroma feature. A more recent approach is presented in , where a well-known audio fingerprinting algorithm is modified in order to retrieve samples inside songs, based on spectral peak signatures.

2.2. Rhythm Plagiarism

A prominent example for rhythm plagiarism is the so-called Amen Break. It originates from the 1969 Funk recording Amen Brother by The Winstons and is considered one of the most widely used drum loops in the history of Rap and Electronic music. Some of such extraordinary beats are pro-tected by the law. But it is often difficult to judge, whether two songs share the same rhythm. There is no definition of which instrument is playing the rhythm. Commonly, the drums make up the beat. But a guitar can also be a dominant rhythmical instrument. In general, rhythm is formed by periodical pattern of accents in the amplitude envelopes of different frequency bands. Rhythm plagiarism has been scarcely covered in the lit-
Erature but it is closely related to rhythm similarity estimation
Paulus and Klapuri took the melody as a reference for rhythm
They transformed the melody into rhythmical strings which are easier to compare along structural dimensions. Others extracted rhythmical features such as the beat spectrum or tempo in order to measure rhythmical similarity.

2.3. Melody Plagiarism

Copied melodies are less obvious than the previously ex-plained plagiarism types. A melodic motive is considered to be identical, even if it is transposed to another key, slowed down, sped up or interpreted with different rhythmic accentuation. Thus, melody plagiarism is a gray area, where it is hard to discern copying from citation. However, MIR techniques are suited for inspection of such cases. In the MIR literature, a closely related task is Query-by-Humming (QbH). QbH can be used to retrieve songs from a database by letting the user hum or sing the respective melody. Melody plagiarism inspection can be done with basically the same approach, since means to identify and evaluate melodic similarity are required. The main difference is, that QbH searches across extensive databases while plagiarism detection concentrates on one single comparison, which has to be more precise.

III. Algorithm And Design Methodology

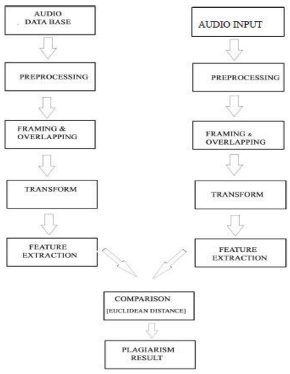

Fig. 1. General block diagram

The human ability to recognize a song from a very short excerpt is an astonishing process that researchers attempt to mimic for some years now. The technique developed to perform this task is called audio fingerprinting and roughly consists in creating unique and compact identifiers for audio signals that can be used for efficient recognition. An audio fingerprint is a content-based signature summarizing an audio recording and allowing its identification. In order to be unique, the fingerprint is based on perceptually and acoustically rele-vant characteristics of the audio signal. Audio fingerprinting systems are composed of a collection of known fingerprints along with a query system. The remainder of this document presents properties that are desirable in fingerprinting systems, usage modes and applications of audio fingerprints, and a gen-eral framework for fingerprinting systems. Properties In order

To adequately compare fingerprinting systems, a set of proper-ties is used. The most important ones are briefly described here. Accuracy: Function of correct, missed, and wrong identifications. Reliability: Correct identification method. Ro-bustness: Ability to accurately identify a signal. Granular-ity: Ability to identify signal from short excerpt. Security: Vulnerability to cracking. Versatility: Ability to identify a signal regardless of audio format. Scalability: Performance with very large databases. Complexity: Computational costs the system (lower = better). Fragility: Integrity verification (detection of changes in content). A fingerprinting system should be designed taking into consideration these interrelated properties. Emphasis may be put on certain properties that are more relevant to the purpose of the system. In general the fingerprint should be a compact perceptual summary of the audio recording, invariant to distortions, and easily computable

Firstly we take the two signals or audio files to be compared. One file will be our target file which is accepted from the user and the other file will be a file from the database which will be compared with our target file. Next we sample the input audio. To make this process more efficient we use framing techniques with overlapping. We choose a frame size of 256 and an overlap distance of 100. The frame size of 256 is chosen because we require the use of an 8 point DFT later in the algorithm. Next a windowing function is applied to reduce Gibbs oscillation. The most efficient methodology we have found is hamming window.

DFT is used to convert signals in the time domain into the frequency domain. This is the first step is pre-processing. Here we will accept the audio signal and convert it to digital format and it is also accepted in stereo format. The final result is a sampled vector of the input signal because analysis is easier in the frequency domain. Once the transform has been performed we have to extract the features which will eventually form our audio fingerprint. There are a long list of features possible however to ensure a unique finger print also balancing processing time we choose MFCC ,Entropy and mean energy. These features provide a solid algorithm for our comparison. These features are then vector quantized to make the comparison easier. Final comparison is done by taking Euclidian distance. Threshold values are determined by testing numerous songs. These values are then used to decide whether plagiarism has been committed. Hence our algorithm successfully determines whether one song has been plagiarised from the other .

IV. Feature Extraction

The purpose of this module is to convert the music wave-form to some type of parametric representation (at a con-siderably lower information rate). The speech signal is a slowly time varying signal (it is called quasi-stationary). When examined over a sufficiently short period of time (between 5 and 100 ms), its characteristics are fairly stationary. However, over long periods of time (on the order of 0.2s or more) the signal characteristics change to reflect the different speech sounds being spoken. Therefore, short-time spectral analysis is the most common way to characterize the speech signal. A wide range of possibilities exist for parametrically representing the speech signal for the speaker recognition task, such as Linear Prediction Coding (LPC), Mel-Frequency Cepstrum Coefficients (MFCC), and others. MFCC is perhaps the best known and most popular, and this feature has been used in this paper. MFCCs are based on the known variation of the human ears critical bandwidths with frequency. The MFCC technique makes use of two types of filter, namely, linearly spaced filters and logarithmically spaced filters. To capture the phonetically important characteristics of speech, signal is expressed in the Mel frequency scale. This scale has a linear frequency spacing below 1000 Hz and a logarithmic spacing above 1000 Hz. Normal speech waveform may vary from time to time depending on the physical condition of speakers vocal cord. Rather than the speech waveforms themselves, MFFCs are less susceptible to the said variations

4.1 The MFCC processor

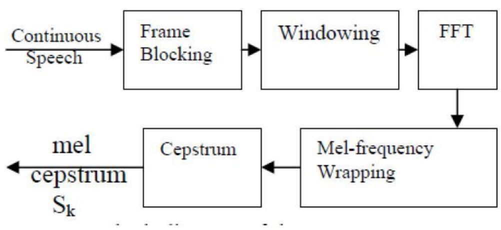

Fig. 2. Block diagram of the MFCC processor

mathematics, especially statistics where x = x1, x2, . . . , xN is a set of random phenomena, and p(xi) is a probability of a random phenomenon xi. A proposed relation between entropy and signal processing is based on the hypothesis that a noise (white noise) is a projection of a system in thermodynamic equilibrium into a signal. As a result the noise is supposed to have the highest entropy value while the speech (and mainly periodic sounds like e.g. vowels) has significantly lower entropy value as it is more organised and required an extra energy to be produced in such an organised form1. According to the above presumption the entropy can be used in signal processing for e.g. separating the useful signal from a background noise.

V. Feature Matching

The state-of-the-art feature matching techniques used in speaker recognition include, Dynamic Time Warping (DTW), Hidden Markov Modelling (HMM), and Vector Quantization (VQ). The VQ approach has been used here for its ease of implementation and high accuracy.

5.1 Vector quantization

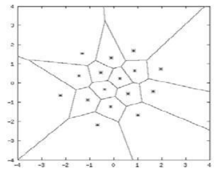

Fig. 3.shows an example of a 2- dimension VQ.

A block diagram of the structure of an MFCC processor is given in Figure 2. The speech input is recorded at a sampling rate of 22050Hz. This sampling frequency is chosen to minimize the effects of aliasing in the analog-to-digital conversion process. Figure 3. shows the block diagram of an MFCC processor

4.2. Entropy

Entropy as a thermodynamic state variable was introduced into physics by German physicist Rudolf Clausius in second half of 18th century. It was originally defined as dS=Q/T. where dS is an elementary change of entropy, Q is a reversibly received elementary heat, and T is an absolute temperature. Of course such a definition has no sense for signal processing. However, it started a diffusion of entropy as a term into the other areas. The entropy as a measure of system disorganisation appeared for the first time in connection with the First postulate of thermodynamics: Any macroscopic system which is in time t0 in given time-invariant outer conditions will reach after a relaxation time the so-called thermodynamic equilibrium. It is a state in which no macroscopic processes proceed and the state variables of the system gains constant time-invariant values. The entropy of a system is maximal when the system has reached the thermodynamic equilibrium. The above depicted key idea pro-moted entropy to a generic measure of system disorganisation.Another definitions of entropy were later proposed for use in

Vector quantization (VQ) is a lossy data compression method based on principle of block coding . It is a fixed-to-fixed length algorithm. VQ may be thought as an approximator Here, every pair of numbers falling in a particular region are approximated by a star associated with that region. In Figure , the stars are called code vectors and the regions defined by the borders are called encoding regions. The set of all code vectors is called the codebook and the set of all encoding regions is called the partition of the space.

VI. Conclusion

Music is an art whose production is very difficult. Artists work for years just to produce one song. Every song produced by an artist is unique. Music plagiarism is an insult to these peoples work because the plagiarizer doesnt know the blood sweat and toil that goes into one song. Music plagiarism is not a new phenomena. It has been going for many years. However only recently with access to the internet and there by a huge range of song we have come to realise just how many songs are fake. The problem with this detecting plagiarism is that people are biased. So they cant be wholly relied on. Another thing is that when proving plagiarism especially in a court of law perception of people (experts)can vary. What is needed is a way to quantitatively and scientifically prove plagiarism in music. This will lead to a more reliable accurate and effective means of detection. our project set out to be a beacon of truth to expose plagiarizers and give back dignity to the art that is music.

By using mfcc, entropy and mean energy level we have created a robust fingerprinting algorithm ,which enables us to differentiate copies from original music.these three features have been chosen because of two major considerations.The first being that an algorithm that is extremely efficient meaning that it pin points plagiarism easily and without error. The second consideration is the processing time that is the time required for our algorithm to obtain a result.through out the project. We struggle to balance these two factors and finally we decided to take these three factors. par Our comparison through taking the Euclidian distance of vector quantisation of features mentioned above allows us to do so. Inspite the rise of plagiarism in reason years we believe this algorithm could be usefull in playing a vital role in stamping out plagiarism. More features can be added and the code can be better optimised so in the future music plagiarism becomes a thing of the past

References

[1]. Md. Rashidul Hasan, Mustafa Jamil, Md. Golam Rabbani Md. Saifur Rah-man,“, speaker identification using melfrequency cepstral coeficients,”3rd international conference on electrical and computer engineering ICESE

[2]. ,2004,Dhaka,Bangladesh Avery Li-Chun Wang,An industrial strength-Audio Search Algorithm Avery Li-Chun Wang‘industrial strength- Audio Search Algorithm”IEEE/ACM Transactions On Audio, Speech,and Language Processing, MARCH 2006

[3]. Shu Tamura,Shin-ichi Ito,Momoyo Ito and Minoru Fukumi ‘ lq Method to Evaluate Similarity of Music by Music Features”IEEE/ACM Transactions On Audio, Speech,and Language Processing November 9-12, 2015

[4]. Min Woo Park and Eui Chul Lee ' lq Similarity Measurement Method between Two Songs by Using the Conditional Euclidean Distance"WSEAS Transactions on Information Science and Application Issue 12, Volume 10, December 2014

[5]. Adam Berenzweig,BethLogan,Daniel P.W. Ellis and Brian Whitman ' lq Large-Scale Evaluation of Acoustic and Subjective Music Similarity Measures" LabROSA Columbia University July 2013