# BIRTH-DEATH FREQUENCIES VARIANCE OF SINUSOIDAL MODEL

# A New Feature for Audio Classification

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Abstract:

In this paper, a new feature set for audio classification is presented and evaluated based on sinusoidal modeling of audio signals. Variance of the birth-death frequencies in sinusoidal model of signal, as a measure of harmony, is used and compared to typical features as the input into an audio classifier. The performance of this sinusoidal model feature is evaluated through classification of audio to speech and music using both the GMM and the SVM classifiers. Classification results show that the proposed feature is quite successful in speech/music classification. Experimental comparisons with popular features for audio classification, such as HZCRR and LSTER, are presented and discussed. By using a set of three features, we achieved 96.83% accuracy, in one-sec segment based audio classification.

#### 1 INTRODUCTION

Rapid increase in the amount of audio data demands for an automated method that allows for efficient segmentation and classification of audio stream based on its contents. In multimedia applications, such systems can be useful to achieve automatic classification, indexing, archiving, and retrieving of information from large multimedia corpora. Audio segmentation and classification also have significant applications in data retrieval, archive management, modern human-computer interfaces, and entertainment.

One of basic problems in audio segmentation and classification is speech/music discrimination. By rejecting non-speech segments, speech/music discrimination can play a signification role in speaker and speech recognition systems. The new generation of low rate coders and compression technologies need an estimation of the signal nature to achieve a higher compression rate. Among them is the work by EI-Maleh *et al.* (Ei-Maleh, 2000) that used LSF (Line Spectral Frequency) parameters and zero crossing for frame based speech/music discrimination.

Ajmera *et al.* (Ajmera, 2002) employed *a posteriori* probability based entropy and dynamism features and reported 82.5% and 79.2% accuracies

for speech and music segments, respectively. Saunders (Saunders, 1996) used typical features, such as zero crossing rate and short-time energy, for a radio broadcast speech/music classifier. For a 2.4 sec segment of audio signals, this work achieved an accuracy of 98%. Scheirer and Slaney introduced an audio classification method in (Scheirer, 1997) using more features and performed experiments based on different classification models. For the same segment length of an audio signal (2.4 sec), the overall error reported was of as low as 1.4%.

Lu et al. (Lu, 2002) applied an algorithm based on KNN (K-Nearest Neighbor) classifier and LSP (Line Spectral Pair) -VQ (Vector Quantization) to determine speech/non-speech segments. Some other classification approaches have recently been introduced in the literature that use different methods, such as nearest feature line (Li, 2000) and SVM (Support Vector Machine) (Guo, 2003).

In this paper, we propose a sinusoidal model based feature for audio classification to speech and music by using the GMM (Gaussian Mixture Model) and the SVM classifiers. The sinusoidal models of different orders are tested and evaluated. The model feature, variance of birth-death frequencies, is presented and compared to conventional features, e.g. HZCRR (high zero crossing rate ratio) and LSTER

(low short time energy ratio), in terms of the classification performance.

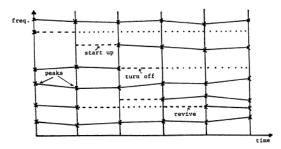


Figure 1: Birth and death of sinusoidal tracks.

This paper is structured as follows. Sinusoidal modeling of speech signals is briefly described in section 2. The features used in this study and the classification methods we have employed are presented in section 3. The system implementation and experimental results are presented in section 4 and the features and the classification techniques are compared in section 5. The paper is concluded in section 6.

#### 2 SINUSOIDAL MODEL

In this section, a brief overview of sinusoidal modeling is given and the frequency tracking in the sinusoidal model is presented.

## 2.1 McAulay-Quatieri Sinusoidal Model

The McAulay-Quatieri (MQ) algorithm is often used to produce a sinusoidal representation of sounds (McAulay, 1986). The algorithm assumes that a large class of acoustical waveforms can be represented by a collection of sinusoidal components described by amplitudes, phases, and frequencies. parameters are estimated from the short-time Fourier transform using a simple peak-picking algorithm. The sampled sound is first transformed into a twodimensional time-frequency representation. Next, the regions of high time-frequency energy are located, where the slope of the waveform changes from positive to negative. The voiced regions can be modeled by a set of harmonically related sinusoids, while the unvoiced regions are modeled using nonharmonic sinusoids.

As the fundamental frequency changes, the number of peaks from frame to frame changes. The concept of sinusoidal *births* and *death* is used to explain the movement of spectral peaks between

frames. In order to match spectral peaks, tracks are formed by connecting peaks between adjacent frames. A track is *dead* when there is no peak in the current frame within  $\pm \Delta$  of the frequency of a peak in the next frame. Correspondingly, a new track is *born* if the frequency of a peak in the current frame is not within  $\pm \Delta$  of the frequency of a peak in previous frame.

Fig. 1 shows the birth and death of frequency tracks formed by connecting peaks of similar frequencies between frames.

# 3 PROPOSED FEATURES AND ALGORITHMS

In this section, the features extracted in this study are presented. Then, a brief overview of the GMM and the SVM classifiers, which we have employed to evaluate the performance of the proposed features, is given.

# 3.1 Feature Analysis

In order to achieve a high accuracy in audio classification and segmentation, it is critical to extract features that can capture the major temporal-spectral characteristics of the signals. To classify one-second audio segments, we selected: the HZCRR, the LSTER and a new feature as a measure of the harmony called BDFV (birth-death frequencies variance). These features will be described in detail in this section.

#### 3.1.1 The HZCRR

This feature describes the variations of zero crossing rates (ZCR). The HZCRR is defined as the ratio of the number of frames whose zero crossing rates are grater than 1.5 time of average zero crossing rate in a one-second window, as:

$$HZCRR = \frac{1}{2N} \sum_{n=0}^{N-1} [sgn(ZCR(n) - 1.5avZCR) + 1]$$
 (1)

where n is the frame index, ZCR(n) is the zero crossing rate at frame n, avZCR is the average ZCR, sgn is sign function, and N is the total number of frames in a one-second window (Lu, 2002). Normally, speech signals are composed of alternating high energy voiced sounds and low energy unvoiced sounds; while music signals usually do not follow this structure. Therefore, for music signals, the HZCRR is usually lower than that of speech.

#### 3.1.2 The LSTER

This feature describes the variations of short time energy (STE). The LSTER is defined as the ratio of the number of frames having energy greater than a half of the average short time energy in a one-second window, as:

$$LSTER = \frac{1}{2N} \sum_{n=0}^{N-1} [sgn(0.5avSTE - STE(n)) + 1]$$
 (2)

where n is the frame index, STE(n) is the short time energy at frame n, avSTE is the average STE, and N is the total number of frames in a one-second window (Lu, 2002).

Normally, there are more silence frames in speech than those in music. Therefore, music signals have a lower LSTER, as compared to speech, in general.

#### 3.1.3 The Birth-death Frequencies Variance

This feature describes a measure of the harmony. A detailed spectral analysis shows that pure music is more harmonious than speech, since pure speech contains a sequence of tonal (vowels) and noise-like (consonants) sounds. Speech is characterized by a formantic structure, whereas music is characterized by harmonic structure. The music spectra change more slowly than speech spectra. Music can be regarded as a succession of periods of relatively stable notes and phones, where speech is rather a rapid succession of noisy periods, such as unvoiced consonants, and of periods of relatively stable parts, e.g. vowels.

Speech signals may hardly have a long time periodic structure. Hence, for speech signal, the harmony measure in general could be lower than that of music. This diversity forms our main motivation for applying sinusoidal modeling, as the technique for measuring the harmony to the audio classification problem. Expectedly, a feature of birth and death of frequencies can discriminate between music and speech signal quite well, due to diversity in the harmonic structure.

In this work, the audio signal is divided into overlapping frames, and then sinusoidal model analysis is applied to the audio signal in each frame. Thus, for each frame, a set of frequencies is generated. These frequency vectors are used for tracking frequencies. We tested two measures of tracking frequencies in a one-second window as the feature. These two measures are: 1) sum of the birth-death frequencies and 2) variance of the birth-death frequencies (BDFV) in the one-second window.

We found the latter feature, BDFV, outperforming the sum of the birth-death frequencies in all cases in our classification tests. Henceforth, we just focus on the BDFV in this paper, for the task of audio classification, which is defined as:

$$BDFV = \frac{1}{N} \sum_{n=0}^{N-1} (BD(n) - avBD)^2$$
 (3)

where n is the frame index, BD(n) is the short time birth-death frequencies number at frame n, avBD is the average BD, and N is the total number of frames in a one-second window.

# 3.2 Classification Algorithms

## 3.2.1 Support Vector Machines

In this method, the data is mapped into a high dimensional space via a nonlinear map, and an optimal separating hyper-plane, or linear regression function, is constructed in this space. Given a class labeled training feature vectors, class boundaries between two classes are learned through the SVM. The SVM minimizes the structural risk and can realize nonlinear discrimination by kernel mapping.

Let  $\{x_i, y_i\}_{i=1}^N$  be a set of N training data points, where  $x_i \in \mathbf{R}^n$  denotes the *i*-th input data and  $y_i \in \{-1, +1\}$  is the class label of the data. The SVM aims at finding a classifier of the form:

$$y(x) = sign\left[\sum_{i=1}^{N} \alpha_i y_i K(x_i, x) + b\right]$$
 (4)

where  $\alpha_i$  are positive real constants, b is a real constant,  $K(x_i, x) = \langle \phi(x_i), \phi(x) \rangle$ ,  $\langle \bullet, \bullet \rangle$  is the inner product, and  $\phi(x)$  is the nonlinear map from original space to the high dimensional space.

The SVM decision function is obtained under constraints:

$$\sum_{i=1}^{N} \alpha_i y_i = 0 , \qquad 0 \le \alpha_i \le C \quad i = 1, \dots, N$$

where C is a parameter that allows for specifying how strictly we want the classifier to fit into the training data (Guo, 2003).

#### 3.2.2 Gaussian Mixture Model

The GMM classifier models each class of data as the union of a few multidimensional Gaussian clusters in the feature space. A GMM is fully represented by: the mean vectors, covariance matrices, and the

mixture weights. Given a sequence of feature vectors, a GMM is trained using the well-known expectation-maximization (EM) algorithm. The probability density function of the mixture model of dimension *d*, for class *i*, is given as:

$$P(\mathbf{W}_{i}|\mathbf{X}) = \sum_{k=1}^{k=N} \alpha_{k} N(\mathbf{\mu}_{k}, \mathbf{\Gamma}_{k}), \sum_{k=1}^{k=N} \alpha_{k} = 1$$
 (5)

where  $\mathbf{W_i}$  denotes the class of the signal,  $\mathbf{X}$  indicates the feature vector,  $\mathbf{\mu_k}$  is for the *d*-component mean vector,  $\mathbf{\Gamma_k}$  is the  $d \times d$  covariance matrix, and  $\alpha_k$  denotes the mixing probabilities. The GMM classification uses a likelihood estimate for each model, which measures how well the new data point is modeled by the entrained Gaussian clusters. An unknown vector  $\mathbf{X}$  in the feature space is assigned to the class that is found to be the best model for this vector.

#### 4 EXPERIMENTS

The "music-speech" corpus used in this study is a collection of 240 15-sec sound files, randomly selected from the radio programs (Scheirer, 1997). This corpus is taken as a standard benchmark for audio system evaluations and has been used in many audio classification studies (see (Scheirer, 1997) and (Berenzweig, 2001)).

For the feature extraction, the audio signal is partitioned into Hamming windowed frames of 23.2 ms long, with 11.2 ms overlap. The classifier is evaluated using labeled data sets, each 20 minutes of speech and music data. Each model is trained with 60 15-sec long training speech files (900 seconds) and 60 15-sec training vocal and non-vocal music files (900 seconds). Each system tested over 20 15-sec speech files (300 seconds), 20 15-sec vocal music files (300 seconds), and 21 15-sec non-vocal music files (315 seconds). Thus, each system is trained over 120 15-sec files, (1800 seconds) and is tested with 61 15-sec files (915 seconds). For each frame, the sinusoidal models of different orders are used and zero padding is employed to increase the peak detection accuracy (Smith, 1987). Thus, for each frame, a set of frequencies in addition to two values of ZCR and STE are generated.

Fig. 2 shows the probability distribution curves of the HZCRR for music and speech signals. The curves obtained from the database training files using onesecond window. As shown, there are overlaps between these two curves and the cross point of two curves, as a threshold, is 0.11. Fig. 3 shows the

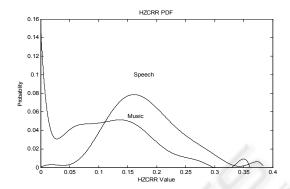


Figure 2: Probability distribution curves of HZCRR.

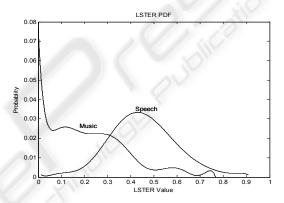


Figure 3: Probability distribution curves of LSTER.

probability distribution curves of LSTER for music and speech signals obtained from the database, where the cross point of two curves, as a threshold, is 0.32.

Figures 4 and 5 show the BDFV curves for music and speech signals for different orders of the sinusoidal model. The curves are obtained from the training database using one-second window (86 frames), where the horizontal axis shows the number of frames. Fig. 6 shows the probability distribution curves of the BDFV for music and speech signals for sinusoidal model of order 15. The cross point of two curves, as a threshold, is 15.5. For music, almost no BDFV value is above 24.

It is observed in these figures that the BDFV values for speech are in general higher than those for music. This is because music is more harmonious and stable than speech. Therefore, the BDFV is an effective feature for discriminating speech and music. The resulting error indicated that no additional improvement to the signal discrimination was achieved, when we increased the sinusoidal model order from 15. Hence, we used sinusoidal model of order 15 in our experiments.

In our experiments, we conducted a comparative evaluation over these three described features. At first, only one feature was used to discriminate speech from music, where the cross point of two pdfs was used as the threshold. The classification errors are presented in table 1. As shown, the BDFV is an effective feature for discriminating between speech and music signals and yields a higher performance, as compared to the other features.

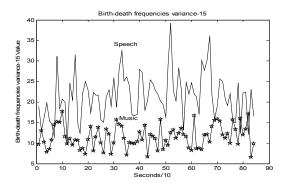


Figure 4: BDFV curves, with sinusoidal model of order 15.

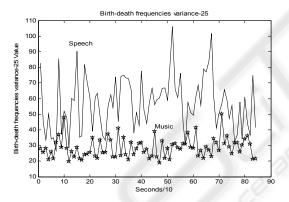


Figure 5: BDFV curves, with sinusoidal model of order 25.

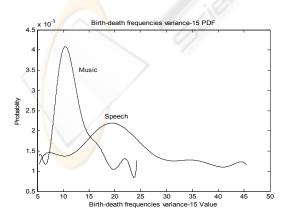


Figure 6: Probability distribution curves of BDFV, with sinusoidal model of order 15.

Table 1: Classification errors (percentage) with HZCRR, LSTER and BDFV using the pdfs' cross-point as threshold.

Total Length (sec) →	300	315	300	915
Features ↓	Speech	Non- Vocal Music	Vocal Music	Total
HZCRR	5	27.61	12.78	23.93
LSTER	16.66	13.96	2.18	12.45
BDFV	18.33	8.57	0.32	9.28

The classification method mentioned so far could result in high classification accuracy with optimal thresholds. However, this optimality is almost unreachable due to actual dependence of thresholds on the signal characteristics. Therefore, we employed the GMM and the SVM classifiers which are not based on a thresholding procedure. We used different combinations of two conventional features, the HZCRR and the LSTER, with the BDFV, as the feature vector of one-second signal and used both the GMM and the SVM based classification methods for evaluating the performance of the BDFV.

In the experiments, we used the radial basis function (RBF) kernel and the parameters C=10 and  $\sigma=1$  for the SVM classifier. The second classification approach was to use the GMM classifier to model each class of data, speech or music. At the training step, the feature vectors from each class are used to train the GMMs. The GMM parameters are estimated by the EM algorithm and the classification of an unknown vector was done by finding the class whose Gaussian distribution came with the highest probability to produce the vector.

# 5 RESULTS AND DISCUSSION

We examined the GMM with different numbers of Gaussians, and found that the optimal order was 4 for the speech models and 4 for the music models, when the three described features were used. We made the same examination with different combinations of the features using the SVM classifier. Table 2 shows the resulting performance using the three features and the two classifiers. As indicated, the SVM and the GMM classifiers significantly improve the accuracy when all the three features are used. It is indicated that the total error introduced using the HZCRR and the LSTER is 12.13%, as compared to the error of 5.46% introduced using the HZCRR and the BDFV, and 4.91% using the LSTER and the BDFV. These

results show a significant improvement to the classification accuracy obtained from combining the new feature with the HZCRR, the LSTER, or both.

Table 2: Classification errors (percentage) with different combinations of the three features using SVM and GMM.

Total Length (sec) →	300	315	300	915
Features/ Classifier	Speech	Non- Vocal Music	Vocal Music	Total
HZCRR+ LSTER/SVM	9.66	15.87	10.66	12.13
HZCRR+ BDFV/SVM	13.33	2.53	0.66	5.46
LSTER+ BDFV/SVM	12.33	2.22	0.33	4.91
HZCRR+ LSTER+ BDFV/SVM	12	2.22	0	4.69
HZCRR+ LSTER+ BDFV/GMM	9.66	1.58	0.66	3.17

As observed, the total errors introduced using the three features are 4.69% and 3.17% with the SVM and the GMM classifiers, respectively. To ensure the effectiveness of the proposed features, evaluation of the classification performance is extended to file-level, in addition to the segment-level evaluation (one-second window) described earlier. We made this evaluation based on a majority voting strategy at file-level. We used the same speech-music database in this test and reached just 1.63% error, i.e. one speech file out of 61 speech-music test files.

As shown in table 2, better classification results are achieved over music files, as compared to speech, when the BDFV is used. Most sounds generated by musical instruments have a harmonic structure, which is not the case with speech signals that may have a mixed harmonic/non-harmonic structure due to their diverse voicing characteristics. This diversity is well identified by the sinusoidal model that measures the harmony of the audio signals. Nevertheless, the BDFV feature of the sinusoidal model plus the HZCRR and the LSTER form a powerful feature set for speech/music discrimination. Still, further performance improvement could be expected to achieve by combining other features of the sinusoidal model as an extension to this work.

### 6 CONCLUSIONS

In this study, we have proposed a new feature based on the sinusoidal model, called BDFV, for audio classification to speech and music. This feature is the variance of the birth-death frequencies in the sinusoidal model of an audio signal, as a measure of the harmony. Our classification results show a high discriminating performance of this feature, as compared to typical features such as the HZCRR and the LSTER features that are widely used for audio classification. It is also revealed that a higher classification performance is achieved, by combining this new feature with the HZCRR and the LSTER, which has been evaluated using the model-based, insensitive to threshold GMM and the SVM classifiers. Through this work, it has been shown that the sinusoidal model features are very effective in audio classification, due to capability of the model to identify the harmonic structure.

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