The Bag-of-frames Approach to Audio Pattern Recognition: A Sufficient Model for Urban Soundscapes But Not For Polyphonic Music

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The "bag of frames" approach (BOF) to audio pattern recognition represents signals as the long-term statistical distribution of their local spectral features. This approach has proved nearly optimal for simulating the auditory perception of natural and human environments (or soundscapes), and is also the most predominent paradigm to extract high-level descriptions from music signals. However, recent studies show that, contrary to its application to soundscape signals, BOF only provides limited performance when applied to polyphonic music signals. This paper proposes to explicitely examine the difference between urban soundscapes and polyphonic music with respect to their modelling with the BOF approach. First, the application of the same measure of acoustic similarity on both soundscape and music datasets confirms that the BOF approach can model soundscapes to near-perfect precision, and exhibits none of the limitations observed in the music dataset. Second, the modification of this measure by 2 custom homogeneity transforms reveals critical differences in the temporal and statistical structure of the typical frame distribution of each type of signals. Such differences may explain the uneven performance of BOF algorithms on soundscapes and music signals, and suggest that their human perception rely on cognitive processes of a different nature.

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I. INTRODUCTION

A. Soundscapes

In 1977, composer R. Murray Schafer coined the term *soundscape* as an auditory equivalence to landscape[1]. He proposed to consider soundscapes as musical compositions, in which the sound sources are musical instruments. Nowadays, the concept of soundscape is used as a methodological and theoretical framework in the field of rural or urban sound quality, notably for the assessement of noise annoyance[2]. Psycho-physic experiments on the perception of soundscapes[3–5] indicate that the cognitive processes of recognition and similarity operate on the basis of the identification of the physical sources. For instance, a given soundscape can be classified as a "park", when specific and localized audio events such as "birds singing", or "children playing" are identified[6]. This also holds for semantic categorization[7], i.e. the subjective "unpleasantness" of urban soundscapes increases when more mechanical sound sources (e.g. vehicles) are identified than natural sources (e.g. voices or birds). However, recent research[8] shows that people are also capable of more holistic strategies for processing soundscapes, when individual source identification is difficult in the presence of too many non-characteristic events ("background noise").

There have been various attempts to simulate human perception of soundscapes with computer algorithms, with methodologies that closely ressemble the two alternative cognitive strategies mentionned above. A majority of contributions[9–14] take the strategy to identify the constituant sound sources individually. The typical implementation describes sound extracts with generic frame-level features, such as MPEG-7 spectral descriptors[11], and use hidden Markov models[15] to represent their statistical dynamics. Recent research[14] proposes to enhance this typical scheme by learning problem-specific features, adapted to each sound class, with genetic programming.

However, another trend of works[16–18] propose to directly recognize soundscapes as a whole, without the prior identification of constituant sound sources. In these works, soundscapes are modelled as the long-term accumulative distribution of frame-based spectral features. This approach has been nicknamed "bag-

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of-frames" (BOF), in analogy with the "bag-of-words" (BOW) treatment of text data as a global distribution of word occurrences without preserving their organisation in phrases, traditionally used in Text Classification and Retrieval[19]. The signal is cut into short overlapping frames (typically 50ms with a 50% overlap), and for each frame, a feature vector is computed. Features usually consists of a generic, all-purpose spectral representation such as Mel Frequency cepstrum Coefficients[15] (MFCC). The physical source of individual sound samples is not explicitely modelled: all feature vectors are fed to a classifier (based e.g. on Gaussian Mixture Models[20]) which models the global distributions of the features of signals corresponding to each class (e.g. pedestrian street or park). Global distributions for each class can the be used to compute decision boundaries between classes. A new, unobserved signal is classified by computing its feature vectors, finding the most probable class for each of them, and taking the overall most represented class for the whole signal.

The BOF approach has proved very effective for soundscapes. Ma et al.[18] report 91% classification precision on a database of 80 3-second sound extracts from 10 everyday soundscape classes (street, factory, football game, etc.). Notably, such systems seem to perform better than average human accuracy on the same task (35%), which suggests that 3-second audio data provides enough information for pattern recognition, but not for people. Similarly, Peltonen et al.[4] reports that the average recognition time for human subjects on a list of 34 soundscapes is 20 seconds. This supports the cognitive strategy of source identification, which typically imposes longer latencies, depending on the temporal density of discriminative sound events.

B. Music

For the analysis of polyphonic music signals also, the BOF approach has led to some success and is by far the most predominant paradigm. Table I shows an enumeration of paper and poster contributions in the ISMIR conference[21] since its creation in 2000. Each year, about a fourth of all papers, and on the whole 88 papers out of a total 388, use the approach. Each contribution typically instanciates the same basic architecture described above, only with different algorithm variants and parameters. Although they use the same underlying rationale of modelling global timbre/sound in order to extract high-level descriptions, the spectrum of the targetted descriptions is rather large: genre[22], mood[23], singing language[24] to name but a few.

However, contrary to its application to soundscapes, recent research[25–27] on the issue of polyphonic timbre similarity shows that BOF seems to be bounded to moderate performance, most notably:

TABLE I. Number of contributions using the bag-of-frames paradigm in the past ISMIR symposiums

year	BOF papers	total papers	percentage
2000	6	26	23%
2001	9	36	25%
2002	14	58	24%
2003	12	50	24%
2004	23	104	22%
2005	24	114	21%
total	88	388	23%

- Glass ceiling: Surprisingly, thorough exploration of the space of typical algorithms and variants (such as different signal features, static or dynamic models, parametric or non-parametric estimation, etc.) and exhaustive fine-tuning of the corresponding parameters fail to improve the precision above a empirical *glass-ceiling*[25], around 70% precision (although this of course should be defined precisely and depends on tasks, databases, etc.).

- Paradox of dynamics: Further, traditional means to model data dynamics, such as delta-coefficients, texture windows or Markov modelling, do not provide any improvement over the best static models for real-world, complex polyphonic textures of several seconds length[26]. This is a paradoxical observation, since static models consider all frame-permutations of the same audio signal as identical, while this has a critical influence on their perception. Moreover, psychophysical experiments[28] have established the importance of dynamics, notably the attack time and fluctuations of the spectral envelope, in the perception of individual instrument notes.

- Hubs: Finally, recent experiments[27] show that the BOF approach (when used on polyphonic music) tends to create false positives which are mostly always the same songs regardless of the query. In other words, there exist songs, which we have called *hubs*, which are irrelevantly close to all other songs. This phenomenon is reminiscent of other results in different domains, such as Speaker Recognition[29] or Fingerprint Identification[30], which intriguingly also typically rely on the same BOF approach. This suggests that this could be an important phenomenon which generalizes over the specific problem of polyphonic music similarity, and indicates a general structural property of the class of algorithms examined here, at least *for a given class of signals* to be defined.

C. Objectives

This paper proposes to re-evaluate this situation and to explicitely examine the difference between soundscape and polyphonic music signals with respect to their modelling with the BOF approach.

We apply to a dataset of urban soundscapes an algorithmic measure of acoustic similarity that we introduced[25] in the context of polyphonic music. The measure is a typical instanciation of the BOF approach, namely comparing the long-term distributions of MFCC vectors, using Kullback-Leibler divergence between Gaussian mixture models. For music, the measure approximates the perception of similar global timbre, e.g. of songs that "sound the same". As already noted, the measure only achieves moderate precision on music and shows notable discrepancies with human perception. We find here that the same measure is nearly optimal for modelling the perceptual similarity of urban soundscapes. This confirms the situation found in the literature that soundscape and polyphonic music signals are not equal with respect to their modelling with the BOF approach. Notably, the application of timbre similarity to soundscapes does not seem to create hubs.

To explain these differences, we report on 2 experiments in which we apply specially-designed *homogeneity* transforms to each datasets:

- Temporal Homogeneity: which folds an original signal onto itself a number of times, so the resulting signal only contains a fraction of the original data.

- Statistical Homogeneity: which only keeps frames in the signal which are the most statistically prototypical of the overall distribution.

We study the influence of each transform on the precision of BOF modelling for both soundscapes and music, and show very different behaviours. This notably establishes that the distribution of frame-based spectral features is very homogeneous for soundscapes, which makes their BOF-modelling very robust to data transformations. i.e. soundscapes can be compressed to only a small fraction of their duration without much loss in terms of distribution modelling. Polyphonic music on the contrary seems to require a large quantity of feature information in order to be properly modelled and compared. Furthermore, it appears that, contrary to environmental textures, not all music frames are equally discriminative: minority frames (the 5% less statistically significative ones) are extremely important for music while they can be discarded to notable advantage for soundscapes. Moreover, it appears that there exists, in typical polyphonic music distributions, a population of frames (in the range [60%-90%] of statistical weight) which is detrimental to the modelling of perceptual similarity.

II. ACOUSTIC SIMILARITY OF URBAN SOUNDSCAPES AND POLYPHONIC MUSIC

A. Algorithm

We sum up here the timbre similarity algorithm presented in Aucouturier and Pachet(2004)[25]. The signal is first cut into frames. For each frame, we estimate the spectral envelope by computing a set of Mel Frequency Cepstrum Coefficients (MFCCs). We then model the distribution of the MFCCs over all frames using a Gaussian Mixture Model (GMM). A GMM estimates a probability density as the weighted sum of $\mathcal{M}$ simpler Gaussian densities, called components or states of the mixture:

$$p(x_t) = \sum_{m=1}^{m=\mathcal{M}} \pi_m \mathcal{N}(x_t, \mu_m, \Sigma_m) \qquad (1)$$

where x_t is the feature vector observed at time t, $\mathcal{N}$ is a Gaussian pdf with mean μ_m, covariance matrix Σ_m, and π_m is a mixture coefficient (also called state prior probability). The parameters of the GMM are learned with the classic E-M algorithm[20].

We then compare the GMM models to match different signals, which gives a similarity measure based on the audio content of the items being compared. We use a Monte Carlo approximation of the Kullback-Leibler (KL) distance between each duple of models A and B. The KL-distance between 2 GMM probability distributions p_A and p_B (as defined in (1)) is defined by :

$$d(A,B) = \int p_A(x) \log \frac{p_B(x)}{p_A(x)} dx \qquad (2)$$

The KL distance can thus be approximated by the empirical mean :

$$\widetilde{d(A,B)} = \frac{1}{n} \sum_{i=1}^{n} \log \frac{p_B(x_i)}{p_A(x_i)} \qquad (3)$$

(where n is the number of samples x_i drawn according to p_A) by virtue of the central limit theorem.

In this work, we use the optimal settings determined by previous research in the context of polyphonic music[25], namely 20 MFCCs appended with 0^{th} order coefficient, 50-component GMMs, compared with $n = 2000$ Monte-Carlo draws.

B. Datasets

1. Urban soundscapes

For this study, we gathered a database of 106 3-minute recordings of urban soundscapes, recorded in Paris using a omni-directional microphone. The recordings are clustered in 4 "general classes":

- Avenue: Recordings made on relatively busy thoroughfares, with predominant traffic noise, notably buses and car horns.

- Neighborhood: Recordings made on calmer neighborhood streets, with more diffuse traffic, notably motorcycles, and pedestrian sounds.

- Street Market: Recordings made on street markets in activity, with distant traffic noise and predominant pedestrian sounds, conversation and auction shouts.

- Park: Recordings made in urban parks, with lower overall energy level, distant and diffuse traffic noises, and predominant nature sounds, such as water or bird songs.

Recordings are further labelled into 11 "detailed classes", which correspond to the place and date of recording of a given environment. For instance, "Parc Montsouris (Paris 14è)" is a subclass of the general "Park" class. Some detailed classes also discriminate takes at identical locations and dates, but with some exceptional salient difference. For instance, "Marché Richard Lenoir (Paris 11è)" is a recordings made in a street market on Boulevard Richard Lenoir in Paris, and "Marché Richard Lenoir (music)" is a recording made on the same day of the same environment, only with the additional sound of a music band playing in the street. Table II shows the details of the classes used, and the number of recordings available in each class.

TABLE II. Composition of the urban soundscape database.

Class	Detailed Class	Size
Avenue	Boulevard Arago	14
Avenue	Boulevard du Trone	5
Avenue	Boulevard des Marchaux	8
Street	Rue de la Sant	7
Street	Rue Reille day1	14
Street	Rue Reille day2	7
Market	Marché Glaciére	8
Market	Marché R. Lenoir	22
Market	Marché R. Lenoir (music)	9
Park	Parc Montsouris Spring	20
Park	Parc Montsouris Summer	8

2. *Polyphonic Music*

The polyphonic music dataset used in this study contains 350 popular music titles, extracted from the Cuidado database[31]. It is organized in 37 clusters of songs by the same artist, encompassing very different genres and instrumentations (from *Beethoven* piano sonata to *The Clash* punk rock and *Musette*-style accordion). Artists and songs were chosen in order to have clusters that are "timbrally" consistent (all songs in each cluster sound the same). Furthermore, we only select songs that are timbrally homogeneous, i.e. there is no big texture change within each song. The test database is constructed so that nearest neighbors of a given song should optimally belong to the same cluster as the seed song. Details on the design and contents of this database can be found in Aucouturier and Pachet (2004)[25].

TABLE III. Comparison of similarity measure for urban soundscapes and polyphonic music.

Database		5-Prec.	10-Prec.	15-Prec.	R-Prec.
Music		0.73	0.70	0.65	0.65
Soundscapes	General	0.94	0.87	0.77	0.66
	Detailed	0.90	0.79	0.75	0.74

C. Evaluation metric

The algorithms are compared by computing their precision after 5, 10 and 15 documents are retrieved, and their R-precision, i.e. their precision after all relevant document are retrieved. Each value measures the ratio of the number of relevant documents to the number of retrieved documents. The set of relevant documents for a given sound sample is the set of all samples of the same category than the seed. This is identical to the methodology used e.g. in Aucouturier and Pachet (2004)[25].

D. Results

1. *Precision*

Table III gives the precision of timbre similarity applied to both datasets. It appears that the results are substantially better for urban soundscapes than for polyphonic music signals, nearing perfect precision in the first 5 nearest neighbors even for detailed classes. High precision using the general classes shows that the algorithm is able to match recordings of different locations on the basis of their sound level (avenues, streets), and sound quality (pedestrian, birds). High precision on detailed classes shows that the algorithm is also able to distinguish recordings of the same environment made at different times (Spring or Summer), or in different contexts (with and without music band). This result has a natural application to computer-based classification, e.g. using a simple k-nearest neighbor strategy, and could prove useful for context-recognition, for instance in the context of wearable computing[32].

2. Hubs

As mentionned above, an intriguing property of the application of the similarity measure to polyphonic music signals is that it tends to create false positives which are mostly always the same songs regardless of the query. In other words, there exist songs, which we call *hubs*, which are irrelevantly close to all other songs. We give a detailed description of this phenomenon in Aucouturier and Pachet (2007)[27].

A natural measure of the hubness of a given song is the number of times the song occurs in the first n nearest neighbors of all the other songs in the database. An important property of the number of n-occurrences N_n of a song is that the sum of the values for all songs is constant given a database. Each query only gives the opportunity for n occurrences to the set of all the other songs, such that the total number of n-occurrences in a given $\mathcal{N}$-size database is $n * \mathcal{N}$. Therefore, the mean n-occurrence of a song is equal to n, independantly of the database and the distance measure.

Table IV shows the 5 biggest hubs in the polyphonic music database ranked by the number of times they occur in the first 10 nearest neighbors over all queries (N_{10}). This illustrates the predominance of a few songs that occur very frequently. For instance, the first song, `MITCHELL, Joni - Don Juan's Reckless Daughter` is very close to 1 song out of 6 in the database (57 out of 350), which is more than 6 times more than the theoretical mean value (10). Among these occurrences, many are likely to be false positives.

TABLE IV. 5 Most Frequent False Positives in the music database.

Song	N_{10}
MITCHELL, Joni - Don Juan's Reckless Daughter	57
MOORE, Gary - Separate Ways	35
RASTA BIGOUD - Tchatche est bonne	30
PUBLIC ENEMY - Cold Lampin With Flavor	27
GILBERTO, Joao - Tin tin por tin tin	25

Figure 1 shows the histogram of the number of 20-occurrences N_{20} obtained with the above distance on the database of urban soundscapes, compared with the same measure on the test database of polyphonic music. It appears that the distribution of number of occurrences for soundscapes is more narrow around the mean value of 20, and has a smaller tail than the distribution for polyphonic music. Notably, there are four times as many audio items with more than 40 20-occurrences in the music dataset than in the urban soundscape dataset. This is also confirmed by the manual examination of the similarity results for the urban soundscapes: none of the (few) false positives re-occur significantly more than random.

This establishes the fact that hubs are not an intrinsic property of the class of algorithm used here, but rather appear only for a certain classes of signals, among whom polyphonic music, but not urban soundscapes.

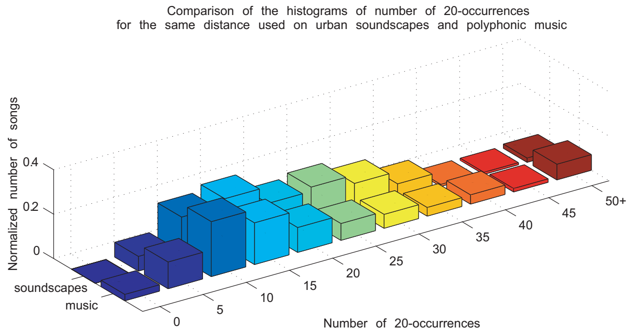

FIG. 1. Comparison of the histograms of number of 20-occurrences for the same distance used on urban soundscapes and polyphonic music.

On the whole, these results confirm that urban soundscapes and polyphonic music signals are not equal with respect to their modelling with the BOF approach. To explain these differences, we now report on 2 experiments in which we apply specially-designed *homogeneity* transforms to each datasets. We study the influence of each transform on the precision of BOF modelling for both soundscapes and music, and observe very different behaviours.

III. TEMPORAL HOMOGENEITY

A. Transform

We consider a temporal homogeneity transformation of audio data which folds an original signal onto itself a number of times (as seen in Figure 2). The output of the 2-fold transform is 50%-sized random extract from the original, repeated twice. Similarly, the 3-fold transform is a 33%-sized extract of the original repeated three times. All signals processed by n-folding from a given signal have the same duration as the original, but contain less "varied" material. Note that since the duration of the fold (an integer division of the total duration) is not a multiple of the frame duration in the general case, n-folding doesn't simply duplicates the MFCC frames of the folded extract, but rather creates some limited jitter. The fact that all n-folded signals have the same number of frames as the original enables to use the same modelling parameters, notably number of gaussian components (else we would have had to account for the curse of dimensionality).

We apply 9 n-folding transforms for $n \in [1, 2, 3, 4, 5, 10, 20, 30, 50]$ to the audio signals of each dataset (soundscapes and music). Each transformed signal is then processed with the algorithm described above, namely GMM of MFCCs. This yields 9 types of GMM for each original signal in a given dataset, and 9 similarity measures for each dataset.

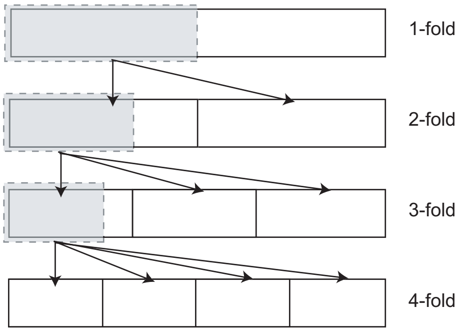

FIG. 2. Illustration of applying 3 successive temporal homogeneity transforms to an audio signal, by folding it twice ("2-fold"), three times ("3-fold") and four times ("4-fold"). The transform creates increasingly homogeneous signals by folding a reduced portion of the original signal. Note tha the "1-fold" transform is the "identity" operator.

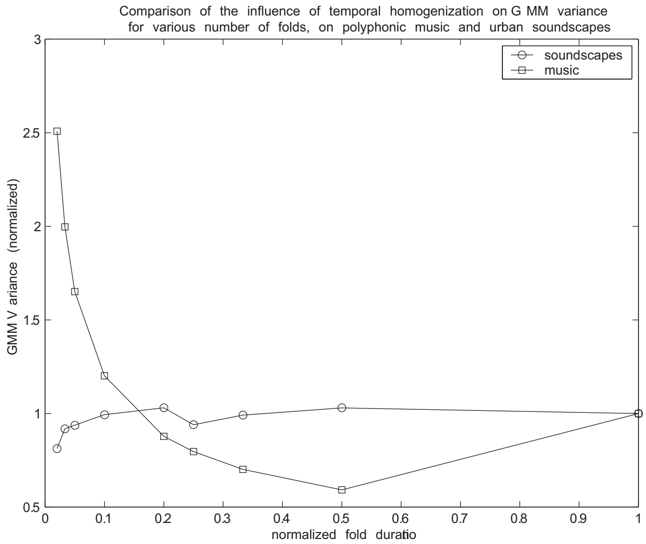

FIG. 3. Influence of temporal homogeneity transform on the mean variance of the GMMs of urban soundscapes and music signals.

B. Influence on variance

Figure 3 shows the influence of n-folding on the mean variance of the GMM of the transformed signals. The variance of a GMM model can be defined by sampling a large number of points from this model, measuring the variance of these points in each dimension, and summing the deviations together. This is equivalent to measuring the norm of the covariance matrix of a single-component GMM fitted to the distribution of points[33].

The temporal homogeneity transform has a very different influence on GMM variance when applied to urban soundscapes and music signals. The GMM variance of soundscape signals shows little dependency on temporal homogenization for ratios as low as 10% of the original signal duration. For extreme number of folds (greater than 10), the GMM variance tends to decrease slightly. This shows that the statistics of urban soundscape signals are stationary on time scales of the order of 10 seconds.

On the contrary, temporal homogenization has a complex influence on the GMM variance polyphonic music signals. Folding audio extracts of the original signal with durations down to 50% of the original signal's tends to reduce GMM variance. However, when the number of folds is greater than 2, the variance exponentially increases. It reaches its original 100% value when folding 15% of the signal's original duration, and increases to more than twice its original value for ratios lower than 5%. This shows that extracts smaller than a half of the original duration (i.e. of the order of 100 seconds) are typically more heterogeneous than the overall signal in the case of polyphonic music. This indicates a rather high density of outlier frames, whose probability is over-estimated when considering small extracts.

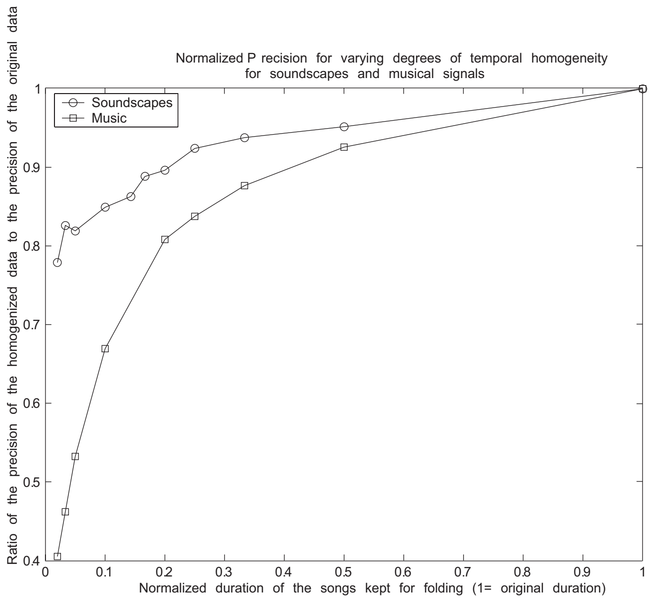

FIG. 4. Influence of temporal homogeneity transform on the precision of the similarity measure for urban soundscapes and music signals.

C. Influence on precision

Figure 4 shows the influence of folding on the similarity R-precision for both classes of signals (where both precision curves are normalized with respect to their maximum). N-folding is detrimental to the precision for both datasets. However, it appears that urban soundscapes are typically twice more robust to folding than polyphonic signals. Considering only a tenth of the audio signals cuts down precision by 15% for soundscapes, and by more than 35% for polyphonic music. In the extreme case of folding only 3 seconds out of a 3-minutes sound extract (50-folding), the precision loss is 20% for soundscapes, but more than 60% for polyphonic music.

This suggests that frame-based feature distributions for urban soundscapes are statistically much more self-similar than polyphonic music, i.e. they can be compressed to only a small fraction of their duration without much loss in terms of distribution modelling. If we authorize a 10% precision loss, soundscapes can be reduced to 10-second extracts. Polyphonic music on the contrary seems to require a large quantity of feature information in order to be properly modelled and compared: the same 10% tolerance requires more than 1 minute of data.

Note that the former is comparable to the human performance[4] on the task of recognizing everyday auditory scenes (20 seconds). However, the latter (polyphonic music) is many times less effective than humans, who have been reported able to issue categorical judgements with good precision using as little as 200ms of audio[34].

IV. STATISTICAL HOMOGENEITY

A. Transform

We define a statistical homogeneity transform $h_k : \mathcal{G} \mapsto \mathcal{G}$ on the space $\mathcal{G}$ of all GMMs, where $k \in [0, 1]$ is a percentage value, as:

```
g2 = hk(g1)
    (c1, ..., cn) ← sort(components(g1), decreasing wc)
    define S(i) = Σ_{j=1}^{i} weight(cj)
    ik ← arg min_{i∈[1,n]} {S(i) ≥ k}
    g2 ← newGMM(ik)
    define di =component(g2,i)
    di ← ci , ∀i ∈ [1, ik]
    weight(di) ← weight(ci)/S(ik), ∀i ∈ [1, ik]
    return g2
end hk
```

From a GMM g trained on the total amount of frames of a given song, the transform h_k derives an homogenized version of g which only contains its top k% components. Frames are all the more so likely to be generated by a given gaussian component c than the weight w_c of the component is high (w_c is also called prior probability of the component). Therefore, the homogenized GMM accounts for only a subset of the original song's frames: those that amount to the k% most important statistical weight. For instance, $h_{99\%}(g)$ creates a GMM which doesn't account for the 1% least representative frames in the original song.

We apply 11 transforms h_k for $k \in [20, 40, 60, 80, 90, 92, 94, 96, 98, 99, 100]$ to the GMMs used in the similarity measure described above. Each transform is applied on each dataset, thus yielding two sets of 11 similarity measures, the properties of which we study below.

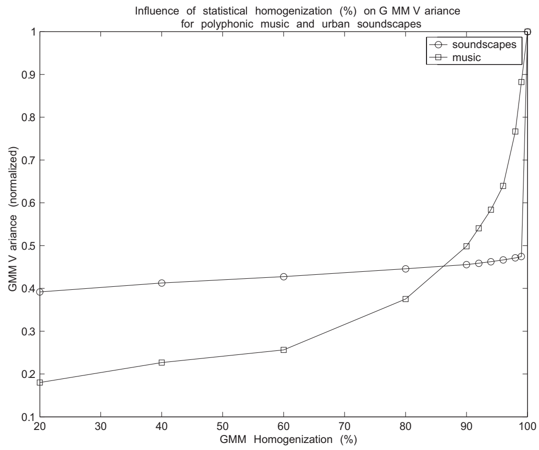

FIG. 5. Influence of statistical homogeneity transform on the variance of the GMMs of urban soundscapes and music signals.

B. Influence on variance

Figure 5 shows the influence of the statistical homogenization transform on the variance of the resulting GMM for both datasets. The variance of the model is evaluated with the sampling procedure already described in Section III.B.

Again, the transformation has a very distinct influence on each type of audio signals. Removing the least important 1% frames from urban soundscape signals drastically reduces the GMM variance by more than 50%. However, further statistical homogenization has little influence on the overall variance. This indicates that soundscape signals are very homogeneous and redundant statistically, except for a very small proportion of outlier frames (the least significant 1%), which account for a half of the overall variance, and probably represent very different MFCC frames than the ones composing the main mass of the distribution. Such frames would typically represent very improbable sound events which are not characteristic of a given environment, such as the occasional plane flying over a park.

When applied to polyphonic music signals, it appears that the homogenization transform reduces the variance of the models exponentially. Half of the original variance is explained by the 10% least representative frames, and more than 80% by the 40% least representative frames. This indicates a greater heterogeneity than for soundscape signals, and a more diffuse notion of "outlier" frames.

C. Influence on precision

Figure 6 shows the influence of statistical homogenization on the precision of the resulting similarity measure, for both datasets. The precision for urban soundscapes is measured with the 10-precision using the detailed classes as ground truth, and with the R-precision for polyphonic music. For both dataset the precision is measured by reference to the baseline precision corresponding to $k = 100\%$, which is different for soundscapes and music, as shown in Table III.

On both datasets, increased homogenization decreases the precision of the similarity measure: homogenization with $k = 20\%$ degrades the measure's precision by 6% (relative) for urban soundscapes, and by 17% (relative) for polyphonic music. It seems reasonable to interpret the decrease in precision when k decreases as a consequence of reducing the amount of discriminative information in the GMMs (e.g. from representing a given song, down to a more global style of music, down to the even simpler fact that it *is* music).

Apart from this general trend however, the transform has a very different influence on the measure's precision depending on the class of audio signals.

In the case of urban soundscapes, 99% homogenization is slightly beneficial to the precision. This suggests that the 1% less significant frames, which were found in Figure 5 to account for half of the overall variance, are spurious frames which are worth smoothing out. Further homogenization down to 60% has a moderate impact on the precision, which is reduced by about 1% (absolute). The decrease in precision from 99% down is monotonic. This suggests that the frame distribution from 99% down is very homogeneous and redundant. Urban soundscapes can be discriminated nearly optimally by considering only the most significant 50% of the frames.

In the case of polyphonic music, the decrease in precision is not monotonic. Figure 6 clearly shows a very important decrease in the precision in the first few percent of homogenization. The severely degraded precision observed for $k = 30\%$ is reached as early as $k = 95\%$. This is a strong observation: the precision of the measure seems to be controlled by an extremelly small amount of critical frames, which represent typically less than 5% of whole distribution. Moreover, these frames are the least statistically significant ones, i.e. are modelled by the least important gaussian components in the GMMs. This indicates that the majority (more than 90%) of the MFCC frames of a given song are a very poor representation of what discriminates this song from other songs. This is the exact opposite behaviour to the one observed for soundscape signals, where these least significant frames can be removed to some advantage.

Moreover, Figure 6 shows that after the abrupt sink when removing the first 5% frames in typical music distributions, the precision tends to increase when k decreases from 90% to 60%, and then decreases again for k smaller than 60%. The maximum value reached between 60% and 80% is only 6% (relative) lower than the original value at $k = 100\%$.

The behaviour in Figure 6 suggests that there is a population of frames in the range [60%, 95%] which is mainly responsible for the bad precision of the measure on music signals. While the precision of the measure increases as

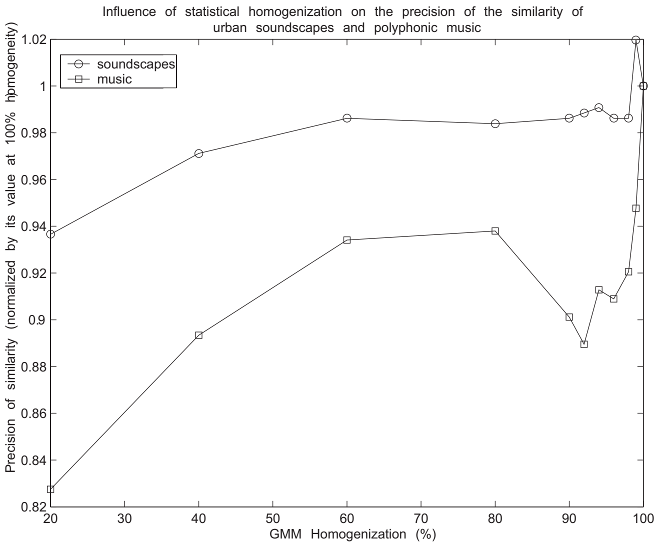

FIG. 6. Comparison of the influence of statistical homogeneity transform on the precision of the similarity measure for urban soundscapes and music signals.

more frames are included when k increases from 20% to 60% (such frames are increasingly specific to the song being modelled), it suddenly decreases when k gets higher than 60%, i.e. this new 30% information is detrimental for the modelling and tend to diminish the discrimination between songs. The continuous degradation from 60% to 95% is only eventually compensated by the inclusion of the final 5% critical frames.

V. DISCUSSION

A. Physical specificities in each class of sounds

We observe critical differences in the temporal and statistical structure of the typical frame distribution for soundscapes and polyphonic music signals. The experiments reported here show that frames in polyphonic music signals are not equally discriminative / informative, and that their contribution to the precision of a simulated perceptual similarity task is not proportional to their statistical importance and long-term frequency (i.e. the corresponding component's prior probability w_c):

- The very informative frames for the simulation of the perception of polyphonic music (measured by their effect of acoustic similarity) are the least statistically representative (the bottom 1%)
- A large population of frames (in the range [60%, 95%]) is detrimental to the modelling. Another study by the authors[27] shows that the inclusion of these frames increase the hubness of a song, i.e. their statistical weight masks important and

discriminative details found elsewhere in statistical minority.

Such structure cannot be observed in the frame distribution of typical urban soundscape signals.

B. A possible reason for the failure of BOF

Such differences in homogeneity for each class of signals can be proposed to explain the uneven performance of their respective modelling with the BOF approach. High performance with BOF correlates with high homogeneity: BOF-based techniques are very efficient for soundscapes, with both high precision and absence of perceptive paradoxes like hubs, while they fail for polyphonic music, which is more heterogeneous.

However, we do not give here any formal proof that heterogeneity is the main factor in explaining the failure of BOf modelling for polyphonic music signals. More complete evidence would come e.g. by synthesing artificial signals spanning a more complete range of homogeneity values, and by comparing algorithmic previsions to human perceptive judgements.

C. Psychological relevance

The BOF approach to simulate the auditory perception of signals such as soundscapes and music makes an implicit assumption about the preceptive relevance of sound events. Distributions are compared (e.g. with the Kullback Leibler distance) on the basis of their most stereotypical frames. Therefore, with BOF algorithms, frames contribute to the simulation of the auditory sensation in proportion of their statistical predominance in the global frame distribution. In other words, the *perceptive saliency*[35] of sound events is modelled as their *statistical typicality*.

BOF is not intended (neither here nor in the pattern recognition litterature) as a cognitive model, but rather is an engineering technique to simulate and replicate the outcome of the corresponding human processing. Nevertheless, it is useful to note that the above model of auditory saliency would be a very crude cognitive model indeed, both to model pre-attentive weighting (which has been found a correlate of frequency and temporal contrasts[36], i.e. arguably the exact opposite of statistical typicality) and higher-level cognitive processes of selective attention (which are partly under voluntary control, hence products of many factors such as context and culture[37]).

The above results establish, as expected, that the mechanism of auditory saliency implicitely assumed by the BOF approach does not hold for polyphonic music signals: for instance, frames in statistical minority have a crucial importance in simulating perceptive judgements. However, surprisingly, the crude saliency hypothesis seems to be an efficient/sufficient representation in the case of soundscapes: frames are found to contribute to the precision of the simulated perceptive task in degrees correlated with their global statistical typicality, and overall BOF provide near-perfect replication of human judgements.

The fact that such a simple model is sufficient to simulate the perception of soundscapes could suggest that the cognitive processes involved in their human processing are less "demanding" than for polyphonic music. This finding is only based on algorithmic considerations, and naturally would have to be validated with proper psycho-sociological experimentations. Nevertheless, it seems at odds with a wealth of recent psychological evidence stressing that soundscapes judgements doesn't result of a low-level immediate perception, but rather high-level cognitive reasoning which accounts for the evidence found in the signal, but also depends on cultural expectations, a-priori knowledge or context. For instance, the subjective evaluation of urban soundscapes has been found to depend as much on semantic features than perceptual ones: soundscapes reflecting activities with higher cultural values (e.g. human vs mechanical) are systematically perceived as more pleasant[5]. Similarly, cognitive categories have been found to be mediated by associated behaviours and interaction with the environment: a given soundscape can be described as e.g. "too loud to talk", but "quiet enough to sleep"[38].

What our results could indicate is that, while there are indeed important and undisputed high-level cognitive processes in soundscape perception, these may be less critical in shaping the overall perceptive categories than for polyphonic music. Discarding such processes hurts the perception of music more than that of soundscapes.

A possible reason for this is that there are important specificities in the structure of polyphonic music, namely very definite temporal units (e.g. notes) with both internal (transient, steady-state) and external (phrase, rhythm) organisation. For instance, a recent study[39] in automatic instrument classification suggests that the transient part of individual notes concentrates very discriminative information for timbre identification, but that its scarsity with respect to longer steady-state information makes it difficult to exploit for machine learning algorithms. This situation of trading too little good information against too much poor-quality information is reminiscent of what we observe here. Human perception, by its higher-level cognitive processing of the structure of musical notes, gives increased saliency to frames that are otherwise in statistical minority.

Such structural specificities in polyphonic music signals may require cognitive processes active on a more *symbolic and analytical* level than what can be accounted for by the BOF approach, which essentially builds an *amorphous and holistic* description of the object being modelled. These computational experiments open the way for more careful psychological investigations of the perceptive paradoxes proper to polyphonic music timbre, in which listeners "hear" things that are not statistically significant in the actual signal, and that the low-level models of timbre similarity studied in this work are intrinsically incapable of capturing.

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