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Rémi Mignot, Geoffroy Peeters

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Degradation-Invariant Music Indexing

*Onset function selection,
audio prints design,
transformation learning,
approximative hashing,
search strategy.*

Scientific Report

Rémi Mignot

UMR 9912 STMS - IRCAM, CNRS, SORBONNE UNIVERSITE, PARIS,
FRANCE

Geoffroy Peeters

LTCI, TELECOM-PARIS, INSTITUT POLYTECHNIQUE DE PARIS, FRANCE

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Abstract

For music indexing robust to sound degradations and scalable for big music catalogs, this scientific report presents an approach based on audio descriptors relevant to the music content and invariant to sound transformations (noise addition, distortion, lossy coding, pitch/time transformations, or filtering e.g.). To achieve this task, one of the key point of the proposed method is the definition of high-dimensional audio prints, which are intrinsically (by design) robust to some sound degradations. The high dimensionality of this first representation is then used to learn a linear projection to a sub-space significantly smaller, which reduces again the sensibility to sound degradations using a series of discriminant analyses. Finally, anchoring the analysis times on local maxima of a selected onset function, an approximative hashing is done to provide a better tolerance to bit corruptions, and in the same time to make easier the scaling of the method.

1 Introduction

1.1 Music indexing, and sound degradations

In audio applications, it can be useful to identify a given music signal, either to recognize the title of the music piece, or to avoid duplicates in a music catalog, for example. The audio indexing usually consists in representing an input sound signal by a sequence of audio prints¹, then matches can be detected by comparing the codes of two signals. Building a database with codes of reference items, which are music pieces for example, it is possible to recognize a given music excerpt if its associated reference is included in the database.

These audio prints are a condensed representation of a part of the sound, and are based on specific properties of the audio signals. However, in the case where the music excerpt to identify is altered by degradations, the possible modification of the audio prints can significantly reduce the recognition performance. These degradations are for example: noise addition, distortion or saturation, lossy coding/decoding (e.g. MP3, AAC, or GSM), pitch shifting, time stretching, equalizing, filtering, reverberation, dynamic range compression. The presented work deals with the robustness of music indexing, that is when the signal to identify is altered by degradations, together with its scalability for big reference music catalogs.

1.2 Context of the work

The reported work was carried out in **2015** in the context of the industrial project *BeeMusic*. Because of possible commercial uses, the method has never been published yet as is. In consequence, this method does not take benefit of the significant advances of the few past years in the domain of audio processing and machine learning, such as the deep learning. It is rather based on hand-crafted descriptors, and a series of linear discriminant related analyses. Nevertheless, the approach is still interesting to present, and some points can be improved using deep learning for example.

In order to place the following state of the art in the context of this work, we decided to only present the known methods anterior to 2015.

1.3 State of the art

Most of the developed methods of the literature are splitted into two different steps: first, a signal analysis provides a sequence of audio prints which describe a part of the sound, ideally in a unique way. Second, the search step consists in finding the items of the reference database with the most similar audio prints.

¹a usual name is *audio fingerprints* because of the similarity of some methods, e.g. (Wang, 2003), with the human fingerprints recognition.

Among the pioneer works about audio indexing, Fraunhofer, see (Allamanche et al., 2001), proposed the use of audio descriptors from the MPEG7 norm. Then, with a vector quantization (clustering), a standard classification approach is used to select the reference item which is the closest to the given input excerpt, in the sense of a code reconstruction error. Each class is here defined by a reference recording. This method has a relative good robustness to degradations (at least the robustness to MP3 coding has been reported), but the method is not relevant for the scaling to big catalogs.

The Philips’s method of (Haitsma and Kalker, 2002) is based on binary codes computed with a 2D derivative filtering of the spectrogram. Then, after a pre-selection of candidate items, using a hash table, the Hamming’s distance provides a similarity indice. Note that an extended version of the hash table is done, in a way that the search is continued until a sufficiently similar item is found. This principle uses a reliability indice of the bits. This method has a high robustness to noise addition, filtering, MP3 coding, and resampling, but it is reported a bad robustness to GSM coding and scale transformations (pitch shifting and/or time stretching).

The most popular method is probably this of Shazam, see e.g. (Wang, 2003), for which the technic is based on audio fingerprints which represent pairs of local maxima of the spectrogram. Based on a hash table, the items of the catalogs which have the higher number of matching fingerprints are selected as candidates. Then a refinement is done by an analysis of the time-coherence of the matching codes (between the excerpt and the candidate items). This method takes advantage of the natural robustness of the spectral peaks to the noise addition, and the high number of hash codes increases the probability to find a sufficiently large number of unchanged codes. Moreover, the time-coherence makes possible to efficiently filter the candidate items. The benefit of this method is an easy scalability to big catalogs, and an excellent robustness to the noise addition, filtering, and time modulation. But the method is not dedicated to scale transformations (pitch and time).

To overcome to the mentioned problems of robustness, for each method, other works proposed extensions. For example, in (Haitsma and Kalker, 2003), a transformation of the spectrum improved the robustness of the Philips’s method to the pitch shifting. Moreover, in (Moravec and Cox, 2011), an approximative hashing has been introduced to this method, in order to improve the tolerance to bit corruptions, and so the robustness to many different sound degradations. Finally let’s mention (Fenet et al., 2011) which improves the Shazam’s method for pitch shifting, using a Constant-Q Transform (CQT), and (Sonnleitner and Widmer, 2014) which uses maxima *quadruplets*, instead of pairs, for which the temporal and frequency differences of peaks are coded in a relative way, in order to improve the robustness to both pitch and time changes.

1.4 Categorization of the two steps

With the studies of the previously mentioned methods, we remark that, in a general point of view, each of the two steps (1: audio prints and 2: search) can be splitted into two categories. Indeed, the audio prints describe either:

- (1a) a local information in time and frequency, or
- (1b) a medium-term information, of a part of the sound.

And the search step is based on either:

- (2a) an exact matching of the audio prints, or
- (2b) an approximative matching, e.g. computed from a similarity indice.

As said in (Fenet, 2013), a good practice is to associate local audio prints with exact matching (1a+2a), or medium-term audio prints with approximative matching (1b+2b). For example the Shazam’s method (Wang, 2003), is mainly associated to (1a+2a), and the Fraunhofer’s method (Allamanche et al., 2001) to (1b+2b). Nevertheless, let’s remark that the Philips’s method (Haitsma and Kalker, 2002) uses a pre-selection based on (1a+2a), local audio prints and hash table, and a refinement based on (1b+2b). Also, the refinement of the Shazam’s method can be considered to be associated to (1b+2b), first because of the global time-coherence analysis, and second because it is based on the number of unchanged hash codes, allowing few codes to be altered.

1.5 IrcamAudioID V1

The *IrcamAudioID* method, see (Ramona and Peeters, 2011) or (Ramona and Peeters, 2013), is based on dedicated audio descriptors as audio prints (medium-term: 1b), associated to a kNN algorithm and a time-coherence analysis for the search step (approximative matching: 2b).

Audio print computation: The initial audio representation is computed by a double-nested *Fourier Transform*, with overlapping windows of approximatively 2 seconds length. This audio descriptors were first designed in (Worms, 1998), for audio indexing applications, and patented in (Rodet et al., 2003). The global idea is to represent the time evolution of the signal characteristics, similarly to the *modulation spectrum*, see (Atlas and Shamma, 2003).

After a normalization of the signal, a standard *Short Term Fourier Transform* (STFT) is computed on the whole signal, and provides the representation $|X_{[k,\ell]}|$, with k the frequency bin indice, and ℓ the time frame indice. Then, selecting a medium-term analysis window with length 2 seconds, and starting at $\ell = \ell_p$, a *Discrete Fourier Transform* is computed over time for each frequency bin separately, providing the representation: $|\mathcal{X}_{[k,m,p]}|$ with m the modulation frequency indice, and p the indice of the start time. Finally, the coefficients over the axes k and m are grouped together, using a weighted sum, to form a compact (6×6) representation for each time step p . We then obtain a sequence of P vectors with size $D = 36$, for all analysis times anchored to ℓ_p .

Search strategy and post-processing: The search strategy is straightforward and consists in selecting among the code database the K-Nearest-Neighbors to the code analyzed. The result is a pair of matrices containing the audio item indices and the time position in the item. The post-processing correlates the results of adjacent frames, for a time-coherence analysis, and prunes the accidental detections of erroneous items. This procedure discards a huge amount of elements, and when applied to sequences of several correlated examples, only keeps the correct item indice. This system proved to be robust to sound degradations, but its scaling to big catalogs is difficult because of the K-NN computation for each code of the excerpt.

During the *BeeMusic* project, an alternative has been proposed to make easier the scaling (Régner and Peeter, 2014): the code sequences are converted to define words, and are used by the software *Apache Solr*, see (The Apache Software Foundation), in order to index the songs. In return, the robustness to audio transformations of the whole method decreases significantly because the search step is then based on an exact matching (2a). As an application, this final system made possible the indexing of a catalog with more than 2 million songs (full tracks), but without strong degradations in accordance with the target use case.

1.6 Overview of the method

The present work is partly inspired from the *IrcamAudioID* method, with the aim to be both robust to sound degradations and suited to scaling. The following paragraphs summarize the 5 sub-steps of the method.

A. Analysis times selection

As with *IrcamAudioID*, see (Ramona and Peeters, 2011), the audio prints are computed on an analysis window with length 2 seconds, and are associated to starting times ℓ_p . To avoid desynchronization between the excerpt and the associated reference item, the analysis times are synchronized with the maxima of an onset detection function. Then, a *maximum* filtering is processed in order to select the analysis times with a mean lag of $\overline{\delta}_t = 0.25$ seconds.

Nevertheless, if the chosen detection function is sensitive to audio degradations, its maxima may be shifted in time, which would provide a new desynchronization. Section 2.A presents some details of this initial step, and designing different onset functions from the literature, an intensive test is processed to select the best function and the best parameter setting, with a criterion based on the robustness.

B. High-dimensional audio print design

Section 2.B presents the design of the descriptors used as high-dimensional audio prints. They are inspired by the *modulation spectrum* of (Atlas and Shamma, 2003), but with significant differences which improve the natural robustness to certain sound degradations, such as the time stretching and the pitch shifting.

Contrarily to the methods of Shazam and Philips, and similarly to the method of Fraunhofer, these audio prints are relevant to the music content, by representing the evolution of the music characteristics over the time. Moreover, as mentioned above, they are robust to certain degradations by their design.

C. Discriminant and robust reduction

Â The dimension of the previous representation is 1056. During this new step, this high-dimensional representation is reduced to a lower dimension of 40 using an affine projection.

The method takes benefit of the initial high dimension in order to learn the transformation with a criterion based on both the invariance to degradation and the discrimination of different initial signals. Based on a data augmentation approach, a series of affine transformations is then learnt. Moreover, some transformations of the series are used to pre-condition the data for the following hashing. This step is detailed in sec. 2.C.

D. Hashing process

Â For the search step of the method, the reduced audio prints are binarized to form 40-bits integer codes, and are used to index a hash table, as in (Wang, 2003; Haitsma and Kalker, 2002). This process makes possible the scalability of the method, for big catalogs.

Additionally, as done in (Moravec and Cox, 2011), an approximative hashing is used to increase the tolerance to bit corruptions, and so to sound degradations. This step is detailed in sec. 2.D.

E. Search step and time-coherence

Based on the number of matching codes, fastly obtained through the hash table, a small number of items is pre-selected as candidates for the last sub-step. This final sub-step of the search, see sec. 2.E.3, is based on the time-coherence of the matching codes between the excerpt and the candidates. We propose here an improvement of the Shazam’s method to deal with the time stretching transformation.

1.7 Outline

First, section 2 presents the details of all the 5 sub-steps, then section 3 presents an evaluation of the robustness, and finally section 4 concludes this report.

2 The method

2.A Analysis times selection

As with IrcamAudioID, the audio prints are computed on an analysis window with length 2 seconds, and are associated to starting times ℓ_p . The time interval separating two adjacent analysis times has a mean of $\overline{\delta_t} = 0.25$ seconds. This value has been chosen sufficiently low to obtain a fine time resolution, but sufficiently high to avoid a too big number of codes. Note that for music with a tempo of 120 BPM, a quarter of a second corresponds to an eighth note.

2.A.1 Synchronization of the analysis times

Nevertheless, if these times are positioned on a uniform grid, with a constant lag δ_t , the codes could be altered in the case where the signal excerpt to recognize is not synchronized on the grid (see an illustration in Fig. 1). The worst case would be with a desynchronization of $\delta_t/2$. Moreover, with a time stretching transformation, the grids of the reference item and of the excerpt are out of sync most of the time.

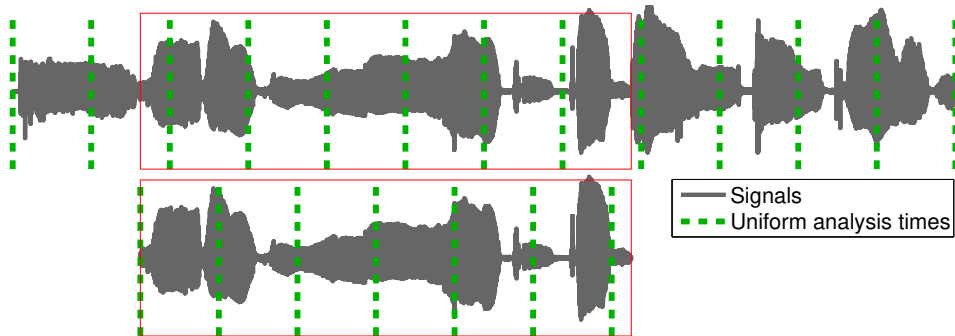


Figure 1: Illustration of the desynchronization of analysis times with a uniform grid. If the start point of the excerpt (below) is not synchronized on the grid of the reference item (above), the analysis times are misaligned (dashed vertical lines).

To deal with this problem, in (Ramona and Peeters, 2011), it has been proposed to synchronize the analysis times ℓ_p on the maxima of an onset detection function, with a mean lag $\overline{\delta_t} = 0.25$ seconds. Onset detection functions, also known as *novelty functions*, are functions designed with the aim to exhibit abrupt changes in the signal characteristics in order to detect onsets with a post-processing. Synchronizing the analysis times of the audio prints on the maxima of an onset function improves the synchronization of the codes between the reference item and the excerpt, whatever the starting time of the excerpt and the time stretching factor. See an illustration in Fig. 2.

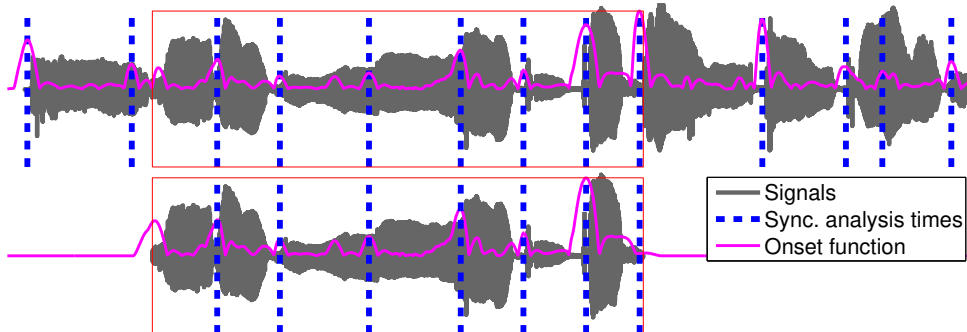


Figure 2: Illustration of the synchronization of analysis times based on an onset detection function. Because the analysis times of the reference item and of the excerpt are both synchronized on the onset function, the alignment is significantly improved.

2.A.2 Sensitivity of the onset function to degradation

Unfortunately, the process may insert a new sensitivity to sound transformations, because the onset function may change with the presence of degradations of the input signal. See Fig. 3 for an illustration. In (Ramona and Peeters, 2011) the onset function and its parameters were chosen without any consideration about robustness.

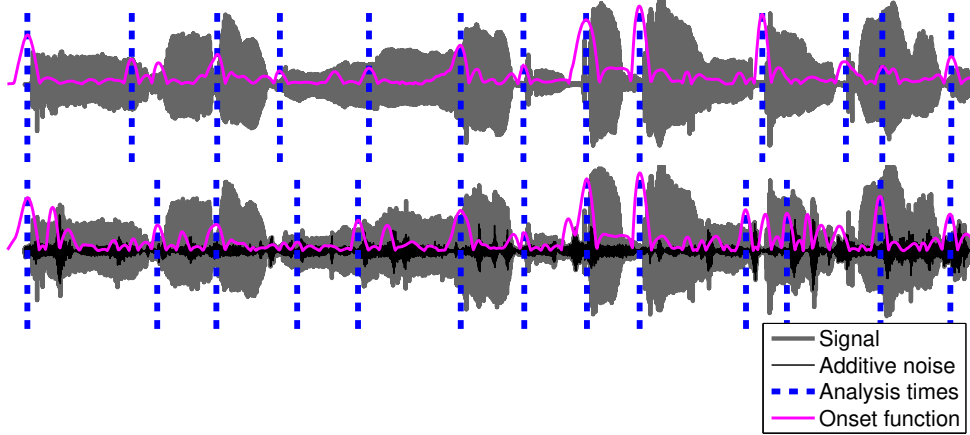


Figure 3: Illustration of the desynchronization of analysis times with degradations. The results are presented for both the original signal (above) and its degraded version (below). With degradations, the onset function is disturbed by the additive noise, and the analysis times are shifted at some points.

2.A.3 Towards a robust analysis time selection

To improve the synchronization of the analysis with or without degradations, in the present work we have selected the onset function and its parameters according to criteria based on the robustness.

For that, we have listed: all the parameters of the pre-processing (spectrogram computation), all the onset functions of the literature (and even more) with their parameters, and all the parameters of the post-filtering. Additionally, we tried to generalize as possible the tested onset functions, by inserting new parameters. The following paragraphs summarize the process and some parameters to set, and section 2.A.4 briefly presents the test, the chosen onset function and the parameter values. All the details of this work have been reported in (Mignot, 2015a), in french.

Spectrogram

All the tested onset functions (except one) are computed on a spectrogram (STFT) which is a time-frequency representation of the input sound. It consists in the computation of Discrete Fourier Transforms (DFT) on segments of the signal using a sliding and overlapping window, see (Zölzer, 2011, chap. 7). We here use a Hann window for the weighting (fixed parameter), see (Harris, 1978), and the sampling frequency is fixed to 11025 Hz, but we test some different values of the window size w_s , and the overlap step size h_s in [sec] (also names *hop size*). These parameters have an effect on the resolution of the spectrogram: the use of a high value of w_s increases the frequency resolution of the spectrum but blurs the short sound events, even with a low h_s . With small values of w_s , and with sufficiently small h_s , the time resolution increases, but the frequency resolution decreases. We may think that the time resolution is better to detect onsets, but the true is not as straightforward because onsets can be characterized by smooth changes of the harmonic parts.

Remark that with a Hann window, in most of spectral applications the following relation is used: $h_s \approx w_s/2$, because a higher h_s implies loss of information, and a smaller h_s implies redundancies. Here this relation is possibly not respected. Finally, converting w_s in seconds to w in samples, to use the standard FFT algorithm, each segment is zero-padded to get a DFT size which is the next power of 2 of w .

The output spectrogram is then represented by $X_{[k,\ell]}$ with k the frequency bin indice and ℓ the time frame indice.

Onset functions

For music signals, onsets are sound events in the music such as new played notes. They can be characterized by the attacks of the notes, or by the changes of the harmonic content in the case of legato playing or glissandi for examples. As said previously, an *onset detection function* (or *novelty function*) is a function computed on the signal (or its spectrogram) which exhibits changes of the signal characteristics representing onsets. In most of the onset detection methods, the result of an onset function is post-processed in order to pick up the time locations of the estimated onsets. In this work, the aim is not to detect onsets, but to select anchor times which are invariant to time shifting and sound degradations. Then, after the process of the onset function, a maximal filter is computed to extract the analysis times.

To find the onset function the most robust, we tested most of the definitions of the literature:

- the *spectral flux* (or *spectral distance*), see e.g. (Masri, 1996),
- the *spectral correlation*, see e.g. (Scheirer and Slaney, 1997; Essid, 2005),
- the *phasis deviation*, see e.g. (Bello et al., 2004; Duxbury et al., 2003; Dixon, 2006),
- the *complex domain*, see e.g. (Bello et al., 2004; Dixon, 2006),
- the *spectral norm*, as in (Ramona and Peeters, 2013),

Also, we added new definitions of onset functions:

- the *difference of spectral moments*,
- the *difference of spectral norms*,
- the *difference of temporal norms*,
- the *Itakura-Saito divergence*, based on (Itakura and Saito, 1968),
- the *Kullback-Leibler divergence*, based on (Kullback and Leibler, 1951),
- the *normalized Kullback-Leibler divergence*, based on (Yang et al., 2011),
- the *generalized Kullback-Leibler divergence*, based on (Yang et al., 2011),
- the *“Linear Prediction” based divergence*, inspired from (Makhoul, 1975).

Additionally, we made a generalization of the initial definitions in order to test more variants, and with more parameters. For example, with the following initial *spectral flux*:

$$\| |\mathbf{X}_\ell| - |\mathbf{X}_{\ell-1}| \|_p = \begin{cases} \left(\sum_{k=1}^K \left| |X_{[k,\ell]}| - |X_{[k,\ell-1]}| \right|^p \right)^{1/p} & \text{if } p \neq +\infty, \\ \max_{k \in [1,K]} \left| |X_{[k,\ell]}| - |X_{[k,\ell-1]}| \right| & \text{if } p = +\infty. \end{cases} \quad (1)$$

we generalized it with:

$$\phi_n^{\text{ds}} = \frac{\| R_h(|\mathbf{X}_\ell| - |\mathbf{X}_{\ell-1}|) \|_p}{(1-d) + d \left(\sqrt{\|\mathbf{X}_\ell\|_p \|\mathbf{X}_{\ell-1}\|_p} + \beta \left(\|\mathbf{X}_\ell\|_p + \|\mathbf{X}_{\ell-1}\|_p \right) + \epsilon \right)}, \quad (2)$$

$$R_h(x) = (x + h|x|)/(1 + |h|) \quad (3)$$

$R_h()$ is the half-wave rectification function (with parameter $h \in [-1, 1]$), \mathbf{X}_ℓ is the frequency vector at the time frame ℓ , p is the order of the norm, $d \in \{0, 1\}$ is a parameter to normalize the norm, and $\beta \in [0, 1]$ is a mixing parameter between an algebraic or a geometric normalization. ϵ is only a small value to avoid a division by 0 with silent parts. See (Mignot, 2015a) for more details.

Post-filtering

To improve the robustness of the analysis time selection, we added a linear filtering as post-processing in order to smooth the time evolution. Remark that the sampling rate of the onset functions correspond to the framing rate of the spectrogram which is $F_r = 1/h_s$ Hz, that is the number of frames per second.

Because the indexing is processed off-line, there is no constraint for real-time and for causality, then we choose a symmetrical mean-average filter. In consequence, the group delay is 0, and the maxima are smoothed and not shifted.

We choose the window method to design the filter (Smith, 2011): with f_c the cutoff frequency of the low-pass filter, the time response h_ℓ of the ideal filter is

$$h_\ell = 2f_c \text{sinc}(2f_c\ell/F_r), \quad \forall \ell \in \mathbb{Z},$$

and with w_ℓ a symmetrical Hamming weighting window (see (Harris, 1978)), centered in 0 and with size $n + 1$, the response of the implemented filter is $b_\ell = h_\ell w_\ell$, which has the support: $[-\frac{n}{2}, \frac{n}{2}]$.

The tested parameters are: the relaxation time $t_c = 1/f_c$ in seconds, and the size n of the filter which must be even. Note that a high value of n may provide Gibbs effects, because of the too strong selectivity of the filter, and a low value may provide no smoothing. Finally, the input onset function ϕ_ℓ can be modified by a non-linear function $\hat{\phi}_\ell = \phi_\ell^r$, with $r > 0$ the order. A value $r < 1$ has the effect to compress the input function, and a value $r > 1$ makes the maxima more predominant. To summarize, the post-filtering is processed using the following convolution:

$$\varphi_\ell = (b * \phi^r)(\ell) \quad (4)$$

Maximal filtering

For the final selection of the analysis times, a maximal filtering is used. This non-linear filter is processed as a median filter but extracting the maximal value of the signal on the sliding window rather than the median:

$$M_\varphi[\ell] = \max_{m \in [-\frac{T}{2}, \frac{T}{2}]} \{\varphi_{\ell+m}\} \quad (5)$$

where T is the size of the window in number of frames, and it is directly linked to the wanted mean lag of analysis times $\bar{\delta}_t = 0.25$ in seconds: $T = \bar{\delta}_t F_r$. As illustrated in Fig. 4, the analysis times are finally picked up by selecting the time positions ℓ_p where:

$$\varphi_{\ell_p} = M_\varphi[\ell_p].$$

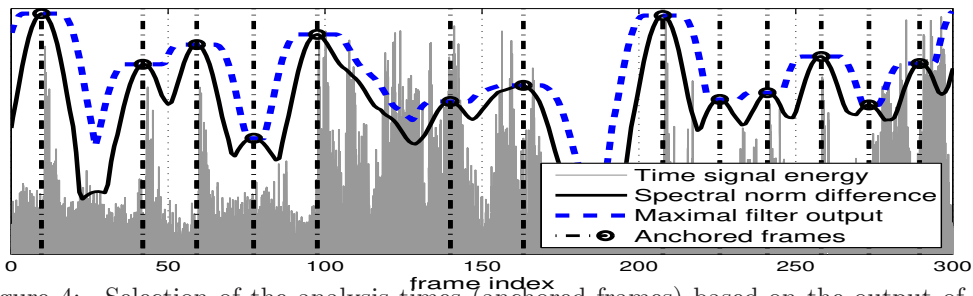


Figure 4: Selection of the analysis times (anchored frames) based on the output of the maximal filtering of the smoothed onset function. In this figure, the *spectral norm difference* is used as onset function.

2.A.4 Selection of a robust setting

Let's remind that the goal of this preliminary work is not the detection of onsets, but it is the selection of analysis times robust to the sound degradations, and we use an onset function to achieve

that. Then the evaluation procedure simply consisting in the comparison of the time positions of the selected analysis times from the original signal and from different degraded versions, but with the same onset function and associated parameters. As a consequence the ground truth is not “absolute”, but it is “relative” to the parameter setting that we want to refine.

First, the original excerpts of a music dataset have been degraded with a lot of different types of audio degradations: noise, filtering, distortion, MP3, etc. with different degrees of alteration, and with possible chains of degradations. This process has been reused for a data augmentation framework, presented in (Mignot and Peeters, 2019) in a music classification task.

Then for each set of signals (original and degradations), the analysis times are computed for different configurations: onset function and parameters. The robustness of the configurations is then evaluated by comparing the analysis times obtained from the original signal and from the degraded signals. Different measures has been defined and tested, such as the “F-measure” of “True Positives”, “False Positives” and “False Negatives”, with a tolerance of 40ms.

Rather than using an automatic selection of the best configuration, we developed a graphical user interface to do a manual choice. Figure 5 presents a screenshot of this interface.

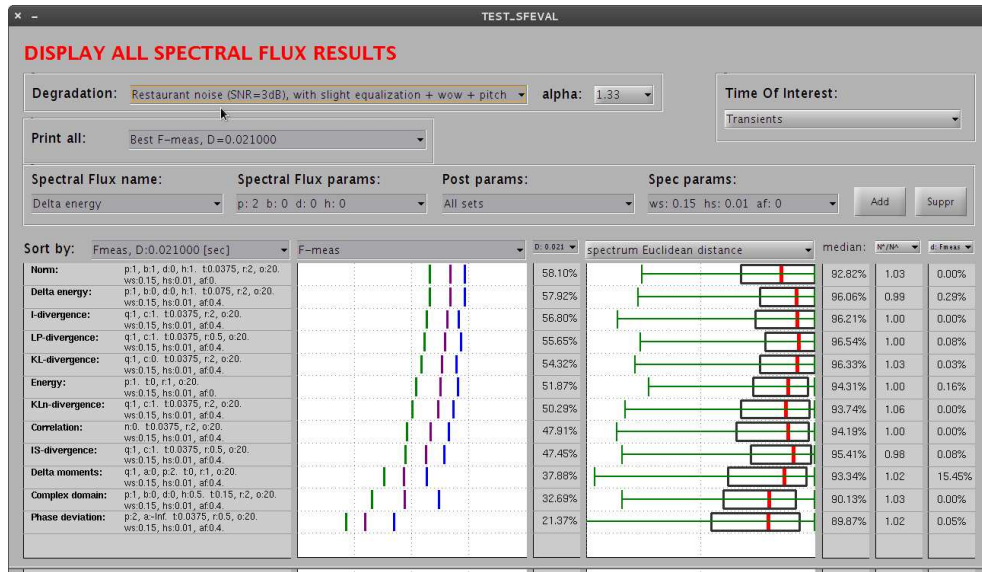


Figure 5: Screenshot of the graphical interface which displays results.

As a result, two different onset functions have been ranked as the best: the first is based on the difference of the spectral norms, framewise, and the second is based on the Itakura-Saito divergence. For a sake of simplicity, the first one has been chosen with its parameters.

By fixing the obtained parameters ($h = 1$, $p = 1$, and $d = 0$), the obtained *difference of the spectral norms* is defined by:

$$\varphi_{\ell}^{\text{dns}} = | R_1 (\| \mathbf{X}_{\ell} \|_1 - \| \mathbf{X}_{\ell-1} \|_1) |, \quad (6)$$

the best parameters of the spectrogram are: $h_s = 10\text{ms}$, and $w_s = 150\text{ms}$, and the best parameters of the post-filtering are: $t_c = 50\text{ms}$, $n = 20$, and $r = 1$.

Nevertheless, in the next section the computation of the high-dimensional audio prints is presented, and in the same way some of its parameters have been refined according to its robustness. These audio prints are also based on the computation of spectrograms, and the obtained parameters are slightly different. To use the same spectrogram computation, both for the onset function and for the audio prints, we decided to use the values obtained for the audio prints, but they do not significantly change the performances of the analysis times selection. Then, the really used spectrogram parameters are: $h_s = 20\text{ms}$, and $w_s = 150\text{ms}$, see sec. 2.B.

Remark with time stretching:

An important remark is about the effect of the time stretching on the selected analysis times. If

the excerpt to identify is transformed by a time stretching, the time positions of the onsets are also stretched, and this affects the onset function too. Nevertheless, the most prominent peaks stay visible, and they are detected for both the excerpt and the original reference signal.

Because the selection is set to extract an average of 4 analysis times per seconds, if the excerpt time is accelerated, some analysis times of the original signal will not be selected. This is not a critical problem because at least the analysis times associated to the more prominent peaks of the onset function are still selected. In the opposite case of a deceleration of the excerpt, more analysis times will be selected, but again it is not a critical problem.

Let's give an example: a 30 seconds excerpt is extracted from a full reference recording. It is accelerated with a factor 1.5, and so its duration is reduced to 20 seconds. Whereas the analysis of the 20 seconds accelerated excerpt produces approximately 80 analysis times (20×4), the analysis of the corresponding excerpt of the reference recording, 30 seconds long, produces 120 analysis times (30×4). Finally, among the 120 analysis times from the reference recording, 40 do not match with the excerpt, but the 80 others should match because they are associated to the more prominent peaks of the onset function.

2.B High-dimensional audio print design

The developed approach permits the definition of audio prints which are intrinsically (by design) robust to some audio degradations: scale transformations (time and pitch), noise addition, equalizing, and volume modulations.

The process of the audio prints is summarized in sec. 2.B.1, and their robustness properties are given in sec. 2.B.2. Finally, as done in sec. 2.A, an evaluation is briefly presented in sec. 2.B.3 to refine the free parameters. More details are given in (Mignot, 2016a), in french.

2.B.1 Overview of the audio prints

1. Spectrogram

In a first step, the whole spectrogram $\mathbb{X}_{[k,\ell]}$ is computed over the signal, using a *Short-Time Fourier Transform*, see Fig. 6(a-b). The process is the same than for the analysis times selection. Note that with a sampling rate $S_r = 11025\text{Hz}$, the frequencies are cut at 5500Hz, close to the Nyquist frequency; the spectrogram has 2049 frequency bins.

2. Medium-term analysis window

Then a medium-term window of 3 seconds extracts a time segment of the whole spectrogram for the following analysis. This window is placed according to the analysis times ℓ_p selected in sec. 2.A. This provides the segmented spectrogram, $X_{[k,\ell]}^p = \mathbb{X}_{[k,\ell-\ell_p]}$ for all $\ell \in [0, L-1]$, see Fig. 6(b-c), with size $(K \times L) = (2049 \times 150)$: 2049 frequency bins (unilateral spectrum), and 150 frames.

3. Log-Log conversion

Each sub-spectrogram is a time-frequency representation of the signal in linear scales (seconds-hertz). These linear scales are then converted to logarithmic scales both for the frequencies (log-frequencies) and the time (log-time), see Fig. 6(c-d). For the frequencies, it is a common process, but it is relatively unusual for the time.

Note that, we need to define both a minimal frequency and a minimal time, because the 0 value cannot be converted to a log scale. The new log-log spectrogram, $H_{[\kappa,\lambda]}^p$, with size (94×64) , represents the frequencies from 150 to 5000Hz, and the time from 0.5 to 2.5 seconds, with the associated analysis time used as reference ($t = 0$).

This conversion is based on the Simpson's interpolation schema, usually used for numerical integration, and a variant has been developed to get good results both for over-sampling (small values of time and frequency) and over-sampling (high values). κ and λ are then the indices of the sampled axes with a logarithmic profile, between the previously given limits.

4. Band sub-division of the frequencies

Then, the frequency axis is splitted into 5 bands, see Fig. 6(d-e). These bands have an equal bandwidth in the logarithmic scale, and we use an overlapping factor of 1/2. The obtained matrices are with size: (32×64) : 32 log-frequency bins of the band, and 64 log-time position. With $b \in [1, 5]$ the indice of the band, the extracted matrix is:

$$h_{[\kappa, \lambda]}^{p,b} = H_{[\kappa + \kappa_{\min}^b, \lambda]}^p, \quad \forall \kappa \in [0, \kappa_{\max}^b - \kappa_{\min}^b],$$

κ_{\min}^b and κ_{\max}^b are the limit indices of the band. The following steps are identically processed on the 5 matrices of the bands.

5. Amplitude modification

The amplitude values of the log-log spectrogram of each band are then modified. First, a rectification of the low values is applied: defining a floor value σ , adapted to \mathbf{h} , all magnitude below σ are shifted to σ . Second, a 2D weighting is done to smoothly bring the “pixels” at the borders to 0. Third, a L_∞ normalization is applied, and fourth, a quasi-logarithmic conversion of the amplitudes is done. The former conversion maps the range $[0, 1]$ to itself, with a linear behavior for low values, close to 0, and a logarithmic behavior for values close to 1. See Fig. 6(e-f).

With $w_{[\kappa, \lambda]}$ the 2D Hamming window, $r = 0.15$ and $a = 10$ two parameters, and \odot the Hadamard product, the four steps are summarized by:

$$\mathbf{f} = F(\mathbf{h}) : \quad (7)$$

$$\left\{ \begin{array}{ll} 1) \text{ Rectification:} & \mathbf{g} \leftarrow \max(\sigma, \mathbf{h}), \\ & \text{with } \sigma = r \max\{\mathbf{h} \odot \mathbf{w}\}, \\ 2) \text{ 2D-Weighting:} & \mathbf{g} \leftarrow \mathbf{g} \odot \mathbf{w}, \\ 3) \text{ Normalization:} & \mathbf{g} \leftarrow \mathbf{g} / \max\{\mathbf{g}\}, \\ 4) \text{ Conversion:} & \mathbf{f} \leftarrow \log(1 + a \mathbf{g}) / \log(1 + a). \end{array} \right.$$

6. 2D discrete Fourier transform

Finally, a 2D discrete Fourier transform is applied on each modified log-log spectrogram $\mathbf{h}^{p,b}$, and the modulus is extracted (the phasis is ignored), see Fig. 6(f-g). With m and n the indices of the frequencies of the log-frequencies and the log-times, respectively, the final representation is:

$$\mathcal{Y}_{[m,n]}^{p,b} = \left| 2\text{D-DFT} \left\{ \mathbf{f}^{p,b} \right\}_{[m,n]} \right| \quad (8)$$

To reduce redundancies because of the 2D hermitian symmetry, we only keep one half of the obtained representation. Its size is then (32×33) : 32 frequency indices of the log-frequency axis, and 33 frequency indices of the log-time axis. After vectorization, we obtain 1056 coefficients associated to one frequency band and one analysis time.

Relevancy to music content

Contrarily to the codes of the Shazam’s method, here the codes are based on audio prints which are relevant to the music content. Let’s remark that, first applying a 1D-DFT over the time, instead of the 2D-DFT, the obtained coefficients would be similar to the modulation spectrum as in (Worms, 1998; Atlas and Shamma, 2003), which is a Fourier representation of the evolution in time of the amplitude of the frequency bands. Second, applying a 1D-DFT over the frequency axis, the obtained coefficients would be roughly similar to a cepstrum or the MFCCs, see (Davis and Mermelstein, 1980; Rabiner and Juang, 1993), which are related to the timbre. Here, the last step is a 2D-DFT (DFT both in frequency and time). The obtained coefficients are then meaningful for the music content, both for the timbre and the rhythm, they could be also used as descriptors for other audio applications, such as classification. Nevertheless, the true motivation of this process for the audio indexing is the quasi-invariance to both the time stretching and the pitch shifting.

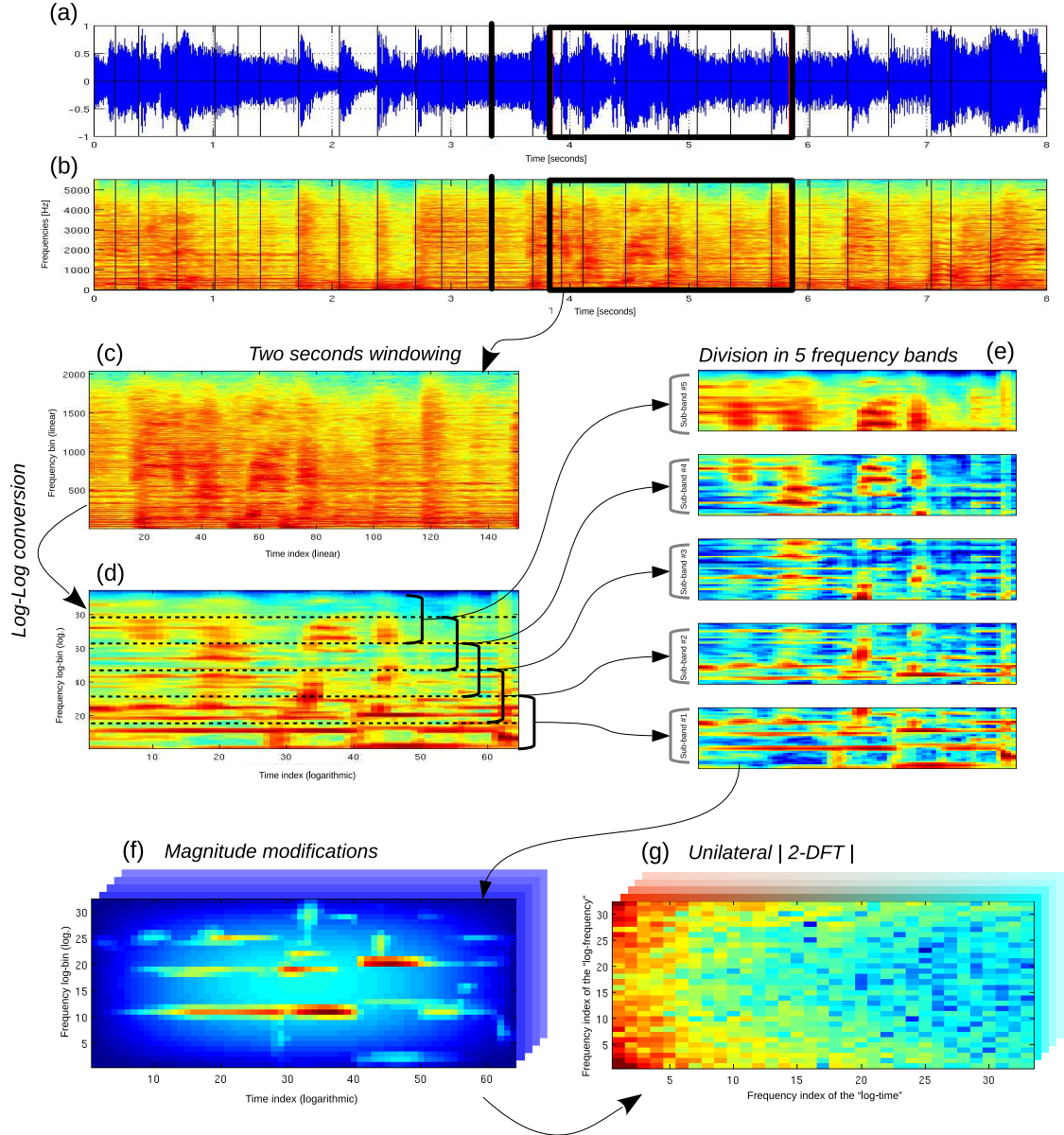


Figure 6: Illustration of the computation of the high-dimensional audio prints: (a) Full signal, with displayed analysis times, (b) Full spectrogram, (c) Selection of one window, synchronized on an analysis time, (d) Logarithmic conversion of the frequency and the time, (e) Separation into 5 frequency bands, with overlapping, (f) Modification of the magnitudes: rectification, weighting, normalization, log-conversion. (g) 2D discrete Fourier transform.

2.B.2 Invariance/robustness properties

This high-dimensional audio prints were initially designed to get robustness to some audio degradations. The next paragraphs present these properties.

Quasi-invariance to scale transformations

Pitch shifting and time stretching consist in the contraction or the expansion of the respective axis in a linear scale: frequency [Hz] or time [seconds]. In the respective logarithmic scales, such a transformation provides a shift of the time-frequency representation, because the log function

transforms products to sums: $\log(at) = \log(a) + \log(t)$. Then, a scale transform of the spectrogram X in linear scales, see Fig. 6(c), becomes a translation of the log-log spectrogram H , see Fig. 6(d).

Given the shift property of the Fourier transform, its magnitude is invariant to translation: $|\text{TF}\{x(t+a)\}| = |\text{TF}\{x(t)\}|$, for all a (actually the values of a changes the phasis only). Hence, the final representation \mathcal{Y} , see Fig. 6(g), is invariant to translations of the intermediary representations, \mathbf{H} and \mathbf{h} , \mathbf{g}^* and \mathbf{f} , see Fig. 6(d,e,f), and by consequence to a scale transformation of \mathbf{X} , see Fig. 6(a,b,c).

Nevertheless, in the case of discrete Fourier transforms, this shift property is valid for a *circular* shift, related to the border effects, and this is not the case in this process. But, the 2D weighting of the amplitude modification reduces these border effects.

Robustness to noise addition

The noise is taken into at two steps in this process. First, considering a (quasi-)stationary noise, on an analysis window, and relatively localized in frequency, the frequency band split can isolate the effect of the noise into one or few bands, leaving the others unchanged. Second, because the noise usually has a power spectral density level lower than the signal of interest, the rectification of low pixel values to a floor σ allows to inhibit the effect of the noise.

Remark that a pure logarithmic behavior during the amplitude change, see eq. (7) or Fig. 6(e-f), may amplify the effect of residual noises with low values, close to 0. The linear behavior for low values avoids possible unwanted effects. Remark that the limit between the linear and logarithmic behavior is defined by the parameter a , see eq. (7).

Invariance to filtering and modulation

The filtering produces a product of the signal spectrum by a frequency function (the response of the filter) which is constant in time. Because of the logarithmic behavior of the amplitude change, see eq. (7) or Fig. 6(e-f), this product acts as an offset constant in time, see Fig. 6(f), and the following 2D-DFT isolates its effect to the first column of the final representation \mathcal{Y} . In consequence, all the other columns of the matrix are invariant to the filtering. In the same way, a time modulation of the amplitude affects the first row only.

Then, by excluding the first row and the first column, we could get a representation invariant to filtering and modulations. Nevertheless, in this work we preserve these coefficients, and we leave to the dimension reduction of sec. 2.C the choice to exclude them or not if necessary.

Remark on the time shift

The logarithmic change of the time and frequency scales makes a focus on the times and frequencies close to zero (similar to an over-sampling). But a small alteration of the position of the analysis time may produce important modifications of log-log spectrogram at the first indices of log-time axis. To reduce this unwanted effect, the minimal time indice of the log-time axis corresponds to 0.5 seconds after the position of the selected analysis time.

2.B.3 Choice of the parameter values

In the previous section, some parameter values are given, but we actually used a test to choose these values, as done in sec. 2.A for the analysis times. Even if the audio prints are designed to get a natural robustness to some degradations, we can expect that the robustness performance partly depends on the chosen values. These free parameters are for example the parameter of: the spectrogram (short-term window and hop size), the log-log conversion (sizes, and limits of frequencies and times), the number of frequency bands, the overlap factor of the bands, the shape of the 2D weighting window, and the parameters a and r of the magnitude modification, see eq. (7).

As done in sec. 2.A, we first used a dataset of reference excerpts which were altered with different audio degradations: environmental and synthetic noise addition, distortion, equalization, pitch shifting, wow effect (similar to a Doppler effect with a cyclic time-varying delay), small time shifts simulating small alterations of the selected analysis time positions. The analysis times were selected on the original excerpts, and reused as is on the degraded versions. For a sake of simplicity, we did not apply time stretching, but it would have been possible by an adaptation of the analysis times

according to the stretching factor. Then the audio prints were computed on each excerpt (original and degraded versions) and for all tested configurations of parameter values.

To evaluate the robustness performance, we defined some measures by a comparison of the audio prints values obtained on the original excerpts and the corresponding values obtained on the degraded versions, with the same parameter values. Remark that here, the ground truth depends on the tested parameter values. These measures have been chosen to simulate the search process of the audio indexing.

The first measure is based on the Fisher's information related to the separation of a *positive* distribution and a *negative* distribution. The *positive* distribution is build with Euclidean distances between audio prints of the degraded excerpts with their associated original version, and the *negative* distribution is build with original excerpts which do not correspond to the degraded version. Whereas the mean of the *positive* distribution must be as close to zero as possible, the mean of the *negative* distribution must be as high as possible. The Fisher's information gives an indice of separability of the two distributions.

The second measure gives an estimate of the probability that, for any degraded audio print, its associated original excerpt audio print is among the K nearest neighbors in an Euclidean space. More details are given in (Mignot, 2016a), in french.

Instead of an automatic selection of the best value configuration, we developed a graphical user interface to manually select the configuration with a good trade-off between the two measures. A screenshot of the interface is given in Fig. 7.

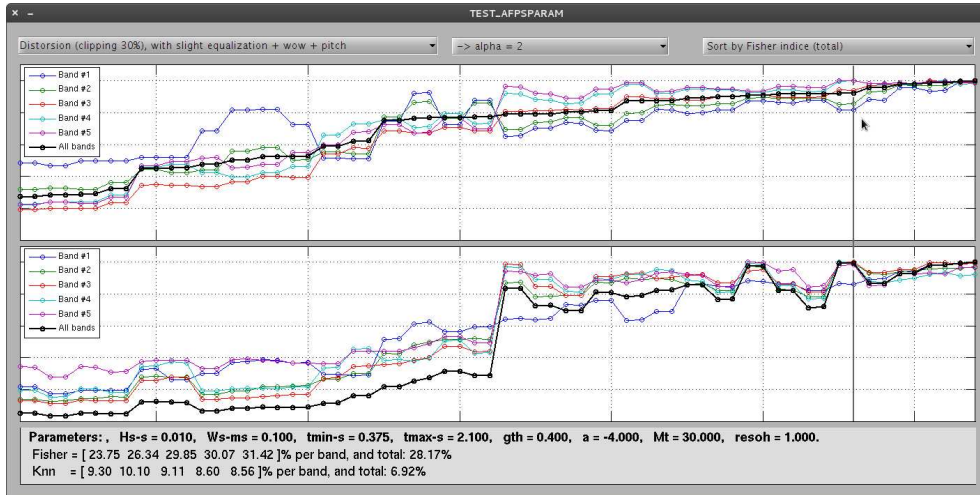


Figure 7: Screenshot of the graphical interface which displays results.

The final selected parameter values are the values previously given. Note that we here obtained a spectrogram hop size of 20ms, which is not the best value for the analysis time selection. In order to compute the spectrogram only once, for both the analysis time selection and the audio prints, we have finally chosen 20ms. This does not significantly affect the robustness of the analysis time selection.

2.C Discriminant and robust reduction

From the high-dimensional audio print computation, see the previous section, we get vectors with 1056 coefficients for all the analysis times and frequency bands. To be used with the indexing search step, this size must be reduced. In (Ramona and Peeters, 2011, 2013), the corresponding reduction is done by a weighting sum to get a new size with 36 coefficients, but without any consideration about robustness.

In this work, we learn a series of affine projections with two goals: first the robustness, that is the reduced audio prints of degraded signal must be as close as possible to their corresponding original values; and second the discrimination, that is the reduced vectors must be representative of

the input signal. Indeed, the use of a constant value, independent from the input signal, produces of course a perfect robustness, but without any discrimination about the original signal. So, it is needed to obtain reduced audio prints which are both robust to audio degradations and discriminant on the original input signal.

Moreover, some transformations of the chain pre-condition the variable properties to the following search step. The chain is built with 5 different affine or linear transformations, with or without reduction. Here is an overview:

- ICCR (*Ill-Conditioned Component Rejection*): this first step guarantees that the variables used for the following transformations are linearly independent. This solve possible problems of matrix conditioning.
- LDA (*Linear Discriminant Analysis*): this is the first reduction with the aim to improve both robustness and discrimination.
- ICA (*Independent Component Analysis*): in the search step based on hash tables, to fill uniformly the buckets, one strategy is to get independent variables, or at least uncorrelated.
- OMPCA (*Orthogonal Mahalanobis Principal Component Analysis*): the previous step provides a loss of robustness, this new step has the goal to recover robustness, with a new formulation closer to search process.
- HT (*Hadamard's Transform*): this last step permits to get variables with equal robustness and discrimination power.

Here is a summary of the transformation chain, with the different dimensions below the arrows:

$$\begin{array}{ccccccccc} \rightarrow & \text{ICCR} & \rightarrow & \text{LDA} & \rightarrow & \text{ICA} & \rightarrow & \text{OMPCA} & \rightarrow & \text{HT} & \rightarrow \\ 1056 & & 1026 & & 80 & & 80 & & 40 & & 40 \end{array}$$

Other transformations have been tested (Data whitening, Quadratic Projections, ...), and also different configurations of chains. All the methods and some configurations have been tested using a performance evaluation similar to these of secs. 2.A and 2.B. No information are given about these experiments, the interested reader can find details in (Mignot, 2016b), in french. The methods and the chain which are finally used and presented in this document, correspond to a configuration among the best ones. Additionally, compared to other methods, this approach seems well justified.

Note that non-linear projections have been also tested (quadratic); they provides interesting results, but not better. For a sake of simplicity, we finally selected affine or linear projections only.

Separation of the bands An important remark is about the separate computation of the bands: all the following transformations are learned and applied on the frequency bands separately. The principle and the used dimensions are the same, but the learnt matrices depend on the band.

2.C.1 Ill-Conditioned Component Rejection

Even if we removed the Hermitian symmetry of the 2D-DFT, we know that few redundancies remain in the HD audio prints. The variables are then linearly dependent, and some problems may occurs with the matrix conditioning for the following transformations. We need to definitely remove these linear dependencies.

With $J = 1056$, and $X \in \mathbb{R}^J$ a random vector of HD audio prints computed for one time, if some of the variables of X are linearly dependent, then a vector $g \in \mathbb{R}^J$ exists, such that $g^T X = \sum_j g_j X_j = 0$. Then the matrix $R = E(XX^T)$ similar to a covariance matrix of X , but without centering, is such that $g^T R g = 0$. In consequence, the eigen vectors which correspond to non-null eigen values of R , form a basis of variables which do not have any linear relation.

Using a high number N of HD audio prints, computed on different signals (unaltered), we build the HD audio print matrix \mathbf{X} with dimensions $(J \times N)$, and with $N \gg J$. Here $N = 150\,000$. Then the *Singular Value Decomposition* (SVD) is computed:

$$\mathbf{X} = U S V^T, \tag{9}$$

with $U \in \mathbb{R}^{J \times J}$ and $V \in \mathbb{R}^{N \times J}$ two unitary matrices, and $U \in \mathbb{R}^{J \times J}$ a diagonal matrix with singular values (non-negative) sorted in a decreasing order. Note that the columns of V are mutually orthogonal. According to the null eigen values, we can reformulate the decomposition by:

$$\mathbf{X} = \begin{bmatrix} U_1 & U_2 \end{bmatrix} \begin{bmatrix} S_1 & \mathbf{0} \\ \mathbf{0} & \mathbf{0} \end{bmatrix} \begin{bmatrix} V_1^T \\ V_2^T \end{bmatrix}. \quad (10)$$

with j_0 the number of non-null singular values, the matrix $U_1 \in \mathbb{R}^{J \times j_0}$ provides an orthogonal basis of the image space of \mathbf{X} , whereas $V_2 \in \mathbb{R}^{N \times (J-j_0)}$ gives an orthogonal basis of the kernel of \mathbf{X} . In consequence, with $P_{iccr} \triangleq U_1^T$ as projector, the change of variables $Z = P_{iccr}X$ removes all linear dependencies:

$$Z = P_{iccr} X, \quad \text{with} \quad P_{iccr} = U_1^T. \quad (11)$$

In this work, to guarantee a good conditioning of the matrices, we choose to define a threshold which depends on the maximal singular value, $\epsilon = s_1 10^{-5}$, and the selected vectors of U_1 correspond to singular values higher than ϵ .

As a result, 30 components of X are linearly dependent with the other, which provides a reduced dimension 1026. Note that some of these dependencies are explained by a residual symmetry between the first and last vertical columns of \mathcal{Y} , ie. at the frequencies 0 and at the Nyquist frequency of the log-time. Nevertheless we do not understand yet the other redundancies, but this transformation, named *Ill-Conditioned Component Rejection* allows an automatic rejection of all linear dependencies.

Note that, the differences of this ICCR and a PCA (*Principal Component Analysis*), are: first, a maximal number of components are kept, and second, the matrix \mathbf{X} is not centered. Let's remark that the variables are non-negative because they are from the magnitude of the 2D-DFT, see sec. 2.B.

2.C.2 Linear Discriminant Analysis

The *Linear Discriminant Analysis* (LDA) is usually used in classification tasks. The elements of different classes are represented in a original vector space, and the LDA provides the optimal directions in order to optimally separate the associated distributions, in the sense of the Fisher's information, see e.g. (Duda et al., 2012; Hastie et al., 2009).

In this work, we formulated the problem of robustness for audio indexing in the form of a classification task. Each original signal is successively degraded with different audio degradations, and considering fixed analysis times computed on the original version, the HD-audio prints are computed and the ICCR is applied. A class distribution is then defined by all audio prints corresponding to the same original signal and the same analysis time, but with different audio degradations. The goal is now to estimate a reduced number of directions which optimally discriminate the class distributions.

With X the random variable vector of the original audio prints (with size 1026), and Y the label of the class, the LDA roughly consists in maximizing the between-class covariance $B = \text{Var}_Y[\text{E}(X/Y)]$ (the class centers must be kept far from each other), and in minimizing the within-class covariance $W = \text{E}_Y[\text{Var}(X/Y)]$ (the class variances must be as small as possible). See an illustration in Fig. 8. For audio indexing, a high between-class covariance provides a discriminant representation of the original signals, and a small within-class covariance increases the robustness to audio degradations.

For a unique variable X_1 , the discrimination power is given by: $S(X_1) = \text{Var}_Y[\text{E}(X_1/Y)] / \text{Var}(X_1)$, related to the Fisher's information. In a J -dimensional space, the most discriminant direction is then given by the vector $g \in \mathbb{R}^J$ which maximizes:

$$S_g = \text{Var}_Y[\text{E}(g^T X / Y)] / \text{Var}(g^T X), \quad (12)$$

$$= (g^T B g) / (g^T T g), \quad (13)$$

where $T = \text{Var}[X]$ is the total covariance matrix satisfying the *total covariance theorem*: $T = B + W$. Remark that this theorem implies: $0 \leq S \leq 1$.

The LDA is obtained by the diagonalization: $G^{-1}(T^{-1}B)G = D$, with D a diagonal matrix of the eigen values, with size $J = 1026$, and G the invertible square matrix of the eigen vectors. If the values of D are sorted in decreasing order, the LDA projector is then given by $G_{1:K}$ build with the K first eigen vectors, associated to the highest eigen values:

$$Z = P_{lda} X, \quad \text{with} \quad P_{lda} = G_{1:K}^T, \quad (14)$$

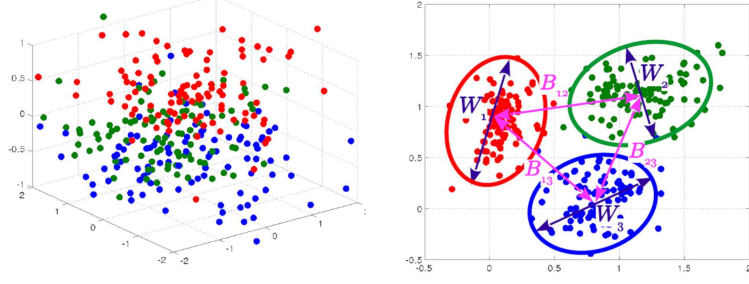


Figure 8: Illustration of an LDA. Three class distributions are represented in the original 3D space on the left sub-figure, and the optimal 2D projection on the right. Whereas the distributions look mixed in the original space, they are well discriminated in the reduced space: maximizing the between-class covariance B and minimizing the within-class covariance W .

and the new components of Z are ordered with decreasing power of discrimination.

Remark the rank of B is strictly lower than the total number classes C , which limit the choice by $K < C$. In this work, we used 150 000 classes, built with 30 000 musical excerpts and 5 analysis times with each. Each class is made of 300 degraded audio prints, and the original one. Since, $150\,000 \gg 1026$, we can freely choose the reduced dimension, and we decided to select the $K = 80$ most discriminating direction of G . Figure 9 illustrates the result.

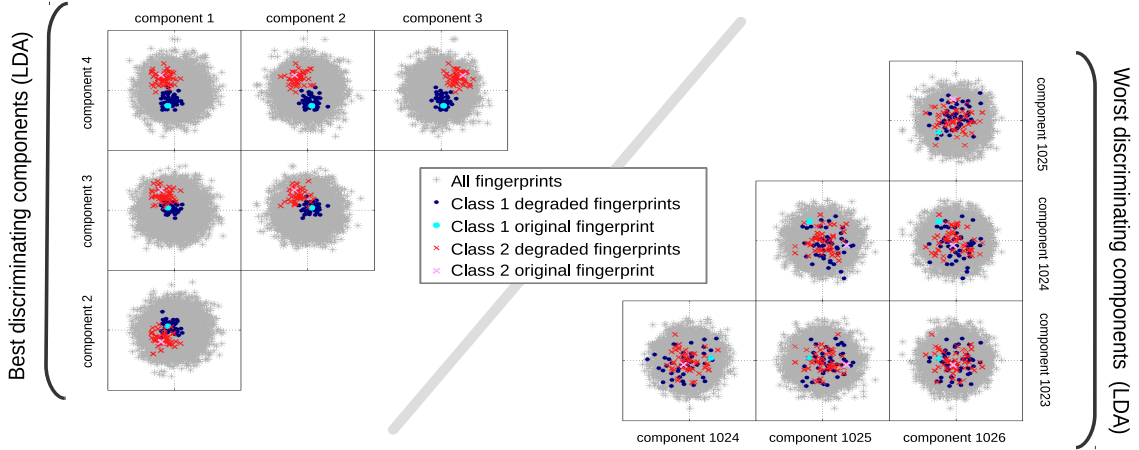


Figure 9: Representation of the distributions of projected audio prints of two classes: blue dots and red crosses. The projected vectors are represented in each sub-figure on 2 directions: the directions are sorted by decreasing order of discrimination power, from the top-left corner (the classes are well separated), to the bottom-right corner (the classes are mixed). The original audio prints of each class are respectively represented by a cyan dot and a magenta cross.

The use of many vectors and classes produces memory problems for the numerical computation of T and B . To deal with this, the matrices are built in a cumulative way, by reformulating their

expressions:

$$\begin{aligned} T_{j,k} &= \frac{1}{N-1} \sum_{n=1}^N (X_j(n) - \mu_j)(X_k(n) - \mu_k), \\ &= \frac{1}{N-1} \sum_{n=1}^N X_j(n) X_k(n) - \frac{N}{N-1} \sum_{n=1}^N \mu_j \mu_k, \end{aligned} \quad (15)$$

$$\begin{aligned} B_{j,k} &= \frac{1}{C-1} \sum_{n=1}^N (\mu_j(c) - \mu_j)(\mu_k(c) - \mu_k), \\ &= \frac{1}{C-1} \sum_{n=1}^N \mu_j(c) \mu_k(c) - \frac{C}{C-1} \sum_{n=1}^N \mu_j \mu_k, \end{aligned} \quad (16)$$

with $C = 150\,000$ the number of classes, $N = 150\,000 \times 301$ the total number of vectors, $X_j(n)$ the j -th component of the n -th vector, $\mu(c)$ the mean of the class c , μ the total mean (for all the N vectors), and then $\mu_j(c)$ and μ_j the j -th component of the means. Finally, the recursive computation is written as:

$$\begin{aligned} \Gamma(n) &= \sum_{i=1}^n X(i) &= \Gamma(n-1) + X(n), \\ \Phi(n) &= \sum_{i=1}^n X(i) X(i)^T &= \Phi(n-1) + X(n) X(n)^T, \\ \Psi(c) &= \sum_{i=1}^c \mu(i) \mu(i)^T &= \Psi(c-1) + \mu(c) \mu(c)^T, \\ &\text{with } \mu(c) = \frac{1}{N_c} \sum_{Y(i)=c} X(i). \\ \mu &= \frac{1}{N} \Gamma(N), \\ T &= \frac{1}{N-1} \Phi(N) - \frac{N}{N-1} \mu \mu^T, \\ B &= \frac{1}{C-1} \Psi(C) - \frac{C}{C-1} \mu \mu^T. \end{aligned} \quad (17)$$

A variant has been finally tested: in this work, the goal is to minimize the distances between the degraded audio prints to their associated original versions, and not to the class means especially. In consequence, in the previous formulations, we replaced each class mean $\mu(c)$ by the audio print vector of the original associated signal. This provided results slightly better, but with no significant benefit.

2.C.3 Independent Component Analysis

During the building of the hash table, if some hash codes appear too many times, their associated discriminant power is reduced because they carry little information. On the contrary, the less frequent codes are more discriminant, but are too rare to be really useful. In the previous audio indexing system, see (Régner and Peeter, 2014), only 10% of the hash codes were used, because for example many codes appeared for almost all reference songs.

As explained in sec. 2.D, the components of the reduced audio prints, with dimension $K = 40$, are binarized using their sign. With z_k the k -th component of the reduced key Z , the binarization function h gives the binary component γ_k of the code:

$$\gamma_k = h(z_k) = \begin{cases} 0, & \text{if } z_k < 0, \\ 1, & \text{if } z_k \geq 0. \end{cases} \quad (18)$$

The full binary hash code is then given by $\Gamma = (\gamma_1, \gamma_2, \gamma_3, \dots, \gamma_K)$. In consequence, the independence of the continuous variables z_k , or at least their decorrelation, guarantees the statistical independence of the binary variables γ_k . Then all the values of Γ between 0 and $2^K - 1$ have the same probability to occur, and the buckets of the hash table (cells associated to hash codes) are uniformly filled.

The goal of this step is to get new variables Z : centered around 0 ($E[Z_k] = 0 \forall k$), normalized ($E[Z_k^2] = 1$), and mutually independent, which implies the decorrelation $E[Z Z^T] = I$. Here, we use

the 150 000 audio prints computed on the original signals, and after the variable changes: ICCR and LDA. Then the Fast-ICA algorithm is used, based on the maximization of the neg-entropy see (Hyvärinen and Oja, 1997). The method provides the following change of variables:

$$Z = P_{ica}X + t_{ica}, \quad (19)$$

with the $(J \times J)$ matrix P_{ica} (where $J = 80$), and the $(J \times 1)$ vector t_{ICA} for the centering. As a result, we obtain a new basis of the sub-space in which the musical information is orthogonalized, and some tests proved that the hash codes occur with a more equal probability, as wanted. See (Mignot, 2016b) for more details, in french.

Nevertheless, even if the sub-space is still this of the LDA, with the 80 best discriminant directions, we observe with an intermediary evaluation a significant loss of performance. It is the reason of the next step.

2.C.4 Orthogonal Mahalanobis Principal Component Analysis

The goal of this step is the recovery of the performance, because of the loss caused by the ICA. But in order to preserve the variable decorrelation, we try here to obtain an orthogonal transformation. Indeed, with any orthogonal matrix P , such that $PP^T = I$, and X a vector of normalized, centered and mutually uncorrelated random variables, see ICA sec. 2.C.3, the transformation $Z = PX$ preserved the mentioned properties:

$$\mathbb{E}[ZZ^T] = \mathbb{E}[PXX^TP^T] = P\mathbb{E}[XX^T]P^T = PIP^T = PP^T = I.$$

Remark that the independence of the variables may be lost, but the more important property for the uniform filling of the hash table is the decorrelation. The LDA of sec. 2.C.2 does not guarantee the orthogonality of the projector, we then define a new transformation with the same goal: improve discrimination and robustness, but with an orthogonal projector. To achieve the task, first let's define two different distributions:

- Each vector \mathcal{P}_n of the “*Positive*” distribution is defined by the difference between: $X(n)$ a degraded audio print vector, and $X(c^*(n))$ the audio print vector of the corresponding original signal, not altered:

$$\mathcal{P}_n = X(n) - X(c^*(n)).$$

- Each vector \mathcal{N}_n of the “*Negative*” distribution is defined by the difference between $X(n)$ a degraded audio print vector, and $X(\tilde{c}(n))$ the audio print of an original signal which does not correspond to n :

$$\mathcal{N}_n = X(n) - X(\tilde{c}(n)), \text{ with } \tilde{c}(n) \neq c^*(n).$$

The goal is then to find the directions which both: minimize the variance of the “*positive*” distribution in the new sub-space, this informs about the error produced by degradations, and maximize the variance of the “*negative*” distribution, which is related to the discrimination of audio prints for different signals. With the covariance matrices of the two distributions,

$$\begin{aligned} C_{\mathcal{P}} &\triangleq \text{Var}[\mathcal{P}] = \mathbb{E} \left[(\mathcal{P} - \mathbb{E}(\mathcal{P})) (\mathcal{P} - \mathbb{E}(\mathcal{P}))^T \right], \\ C_{\mathcal{N}} &\triangleq \text{Var}[\mathcal{N}] = \mathbb{E} \left[(\mathcal{N} - \mathbb{E}(\mathcal{N})) (\mathcal{N} - \mathbb{E}(\mathcal{N}))^T \right]. \end{aligned} \quad (20)$$

the task is to maximize the variance of the *negative* distribution (similar to a PCA with $C_{\mathcal{N}}$), in a vector space provided with a *Mahalanobis* metric associated to $C_{\mathcal{P}}$. We name this process: *Mahalanobis Principal Component Analysis* (MPCA). And similarly to an LDA, this optimal directions are given by the diagonalization of $C_{\mathcal{P}}^{-1}C_{\mathcal{N}}$, given by:

$$G^{-1} (C_{\mathcal{P}}^{-1}C_{\mathcal{N}}) G = D,$$

with D the diagonal matrix of eigen values sorted in a decreasing order, and G an invertible matrix of the eigen vectors.

Nevertheless, the orthogonality of G is not guaranteed. A first tested solution is a Gram-Schmidt orthogonalization of G . Unfortunately we observed a new loss of performance, and the order of the discrimination power of the components may be lost.

The final solution which guarantees a sorted discrimination power, and which provided an increased performance is based on an intertwined recursive computation; it can be summarized as following: for each step j , instead of projecting the $J - j$ last columns of G on the complementary sub-space of the previous columns, as with the Gram-Schmidt algorithm, the matrices \mathcal{P} and \mathcal{N} are projected to this complementary sub-space (with dimension $d_j = J - j$), and a new MPCA is computed. With g_j the basis vector of the new principal component, a $(d_j \times d_j)$ orthogonal matrix Q_j is obtained with g_j as first column, and the global projector P (with size $(J \times J)$) is updated by the projection of its last d_j rows with Q_j^T . The process stops at the K -th step, with $K = 40$ the dimension of the wanted reduced sub-space. For more details, Figure 10 provides a matlab implementation of the method; and Figure 11 illustrates the obtained directions, with a comparison with the PCA method, for a 2D problem and two distributions.

```
function P = OMPCA( dist_P, dist_N )
% Dimension of the original space, and size of the distributions
[ J, p ] = size( dist_P );
[ J, n ] = size( dist_N );
% Centering of the distributions P and N
dist_P = dist_P - repmat( mean( dist_P, 2 ), 1, p );
dist_N = dist_N - repmat( mean( dist_N, 2 ), 1, n );
% Recursive loop
for j = 1:J-1
% Empirical covariance matrices (with centered variables)
C_P = 1/(p-1) * dist_P * dist_P';
C_N = 1/(n-1) * dist_N * dist_N';
% Diagonalisation of C_P^{-1}C_N
[ G, Lambda ] = eig( C_P \ C_N );
% here G and Lambda are with dimension ((J-j+1) x (J-j+1)).
% Computation of the principal component
[ ~, Iprincipal ] = max( diag(Lambda) );
g = G( :, Iprincipal );
% QR Factorization of g :
[ Q, a ] = qr( g );
q = q / a(1);
% the first column of Q is equal to g,
% and the following columns form an
% orthogonal basis of the complementary sub-space.
% Initialisation, or update, of the projector matrix P :
if j == 1
P = Q';
else
P = [ eye(j-1), zeros(j-1, J-j+1);
      zeros(J-j+1, j-1), Q' ] * P;
end
% Projection of P and N on the complementary sub-space,
dist_P = Q( :, 2:end )' * dist_P;
dist_N = Q( :, 2:end )' * dist_N;
end
% Fin du code :
```

Figure 10: Matlab code of the OMPCA. The two inputs are `dist_P` and `dist_N` the matrices of the two distribution: \mathcal{P} and \mathcal{N} . The output is the orthogonal projector matrix P , with size $(J \times J)$. The selection of the K first rows of P , provides a reduction with maximal discrimination.

As a result, we obtain a new linear reduction given by the following change of variables:

$$Z = P_{ompca} X, \quad (21)$$

with P_{ompca} a $(K \times J) = (40 \times 80)$ matrix with mutually orthogonal rows, and both X and Z vectors of uncorrelated random variables. Not only this new reduction both recover the robustness and the discrimination power, but it preserves also the decorrelation property of the variables: $E[ZZ^T] = I$.

Because this OMPCA step provides a discriminant reduction of dimension, the presence of the preliminary LDA is questionable. In this case, the ICA should be applied in a sub-space with dimension $J = 1026$ and this would provides some technical problems. An alternative would be a so-called *whitening* of the 1026 variables, or ZCA, which results in uncorrelated variables, as wanted, but different tests revealed its inefficiency to uniformly fill the hash table, as wanted. Moreover, because of the recursive process of the OMPCA, and the successive projection of the distribution matrices, its computation in a high-dimensional space (with $J = 1026$) is difficult.

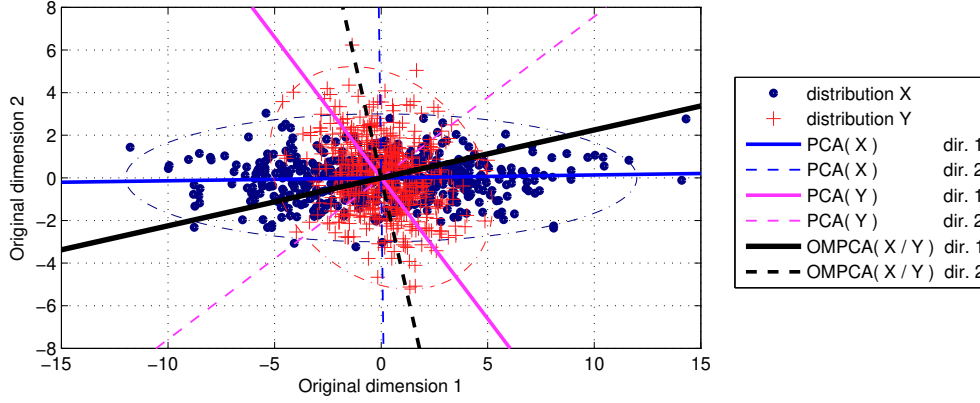


Figure 11: PCA and OMPCA comparison. The PCA is applied for both X and Y , and the principal directions are displayed with plain lines (maximized variance), and the secondary directions with dashed lines (minimized variance). For each PCA, the 2 obtained directions are orthogonal. The OMPCA provides orthogonal directions, where the principal direction is a trade-off between the maximization of $\text{Var}[X]$ and the minimization of $\text{Var}[Y]$.

In consequence, the selected sub-chain is: LDA \rightarrow ICA \rightarrow OMPCA. The LDA first significantly reduces the dimension from 1026 to 80, with an increased robustness, the ICA provides independent, and so uncorrelated variables, and the OMPCA recovers the robustness by preserving the decorrelation.

2.C.5 Hadamard Transform

As said, the OMPCA reduction provides variables with a decreasing order of robustness and discrimination. If this obtained reduced audio prints are binarized, as in eq. (18), the robustness of the bits of the hash codes Γ would not be equal. However it seems interesting to get a uniform robustness.

In order to equalize the robustness of the audio print components before the binarization, we propose a last change based on the Hadamard transform, see e.g. (Kunz, 1979). This transform is given by an orthogonal matrix where the modulus of its elements are equal (with real values). With K the dimension, the modulus value is $\pm 1/\sqrt{K}$, and the signs are defined to get the orthogonality. For example, with $K = 4$:

$$P_{\text{hadamard}}^{(4)} = \frac{1}{2} \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix}. \quad (22)$$

Remark that this transform does not exist for any $K \in \mathbb{N}$. The first valid dimensions are:

$$\{ 1, 2, 4, 8, 12, 16, 20, 24, 32, 40, 48, 64, 80, 96, 128, 160, 192, 256, 320, 384, 512, 640, 768, 1024 \}.$$

In this work, we use $K = 40$. The change of variable is then given by:

$$Z = P_{\text{hadamard}}^{(40)} X. \quad (23)$$

Because each output variable z_k is a linear combination of the input variables x_j , with equal coefficient magnitudes, the initial robustness of the x_j 's are equally mixed to the z_k 's, providing an equalization of the robustness. Moreover, because P_{hadamard} is orthogonal, it preserves the decorrelation.

2.C.6 Summary

As a result, this series of 5 affine projections return reduced audio prints with dimension 40, where the variables are: centered, normalized and mutually uncorrelated, which are important properties for the following hashing. Moreover, the most important properties are: the learnt robustness to degradations, and their power of discrimination for different initial signals. Figure 12 presents the contributions of the 1056 initial components.

Remark that because all the transformation are linear or affine, for a faster computation, we can factorize the projection with:

$$Z = \mathbf{P}_{\text{final}} \times X + \mathbf{t}_{\text{final}}, \quad (24)$$

$$\text{with } \begin{cases} \mathbf{P}_{\text{final}} &= P_{\text{ht}} \times P_{\text{ompca}} \times P_{\text{ica}} \times P_{\text{lda}} \times P_{\text{iccr}}, \\ \mathbf{t}_{\text{final}} &= P_{\text{ht}} \times P_{\text{ompca}} \times t_{\text{ica}}. \end{cases}$$

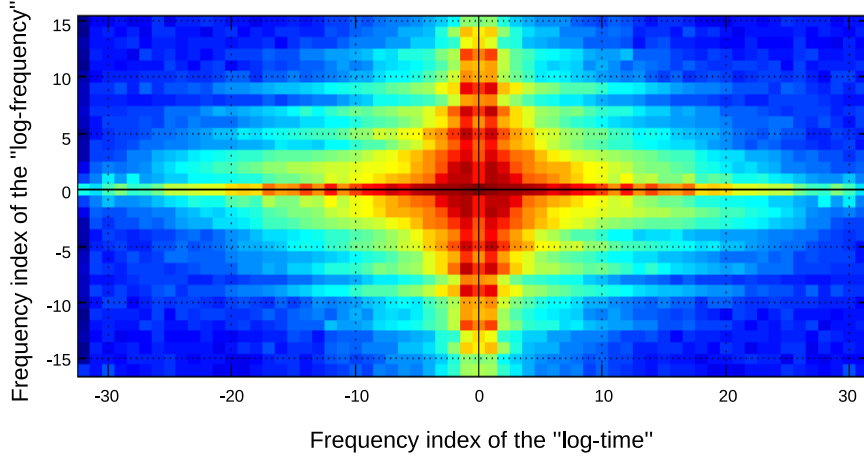


Figure 12: Cumulative contributions of the initial components of the (32×33) representation, for all the 40 reduced components (with the Hermitian symmetry).

Evaluation

An evaluation has been done in order to test other transformations and other settings. As with the evaluation of the high-dimensional audio prints, see a summary in sec. 2.B.3, we used two measured which has the goal to simulate the next search step. The interested readers can find more information in (Mignot, 2016b).

2.D Hashing process

This section summarize the hashing developed in his work. Not only this process allows a scaling of the method for big music catalog, but it also deals with bit corruptions caused by audio degradation using an approximative hashing.

2.D.1 Hash codes and hash table

The principle of the hashing is to associate an integer to a piece of data. The piece of data is named the *hash key*, and the obtained integer is the *hash code* or *index*. The process which computes the hash code from the hash key is the *hash function*, and the *hash table* acts as an *inverse table* which can quickly provide the ids of the indexed data corresponding to a given index, and then we can get other information based on the obtained ids. To summarize the standard process of the current audio indexing, for each reduced audio print vector z with dimension 40, corresponding to a song at an analysis time and a frequency band, a hash code is computed by binarization, Γ with 40 bits, and

the id of the song is added to the Γ -th *bucket* of the hash table. Let's remind that the binarization of the k -th component of the reduced audio print is based on its sign, as following:

$$\gamma_k = h(z_k) = \begin{cases} 0, & \text{si } z_k < 0, \\ 1, & \text{si } z_k \geq 0. \end{cases} \quad (25)$$

The benefit of this approach is the speed of the answer. Indeed, after the computation of the HD-prints of the song, their reduction and their binarization, the search of matching reference songs is processed by extracting the id list of songs which produced the same hash codes, together with a histogram of the number of matching codes per reference song. Then, in the ideal case, the real corresponding reference may be detected based on the histogram. Note that the hash function is not bijective, that is some different hash keys can produce the same hash code, and so the same hash code can occur in different songs. Nevertheless the uniqueness of the identification relies on the sequence of different codes, computed at different analysis times and frequency bands. That's why a post-processing can be done in order to analyze the time coherence of the code occurrences between the query excerpt and the reference candidate songs.

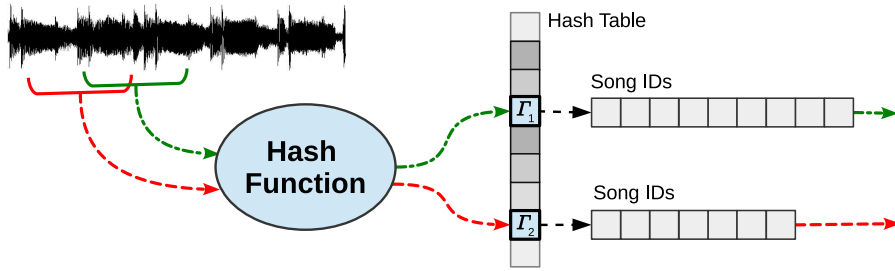


Figure 13: Illustration of the search based on a hash function. From parts of the signal (the hash keys), the hash function computes a sequence of hash codes (Γ_n , for different time positions, and band). Then the ids lists of the references songs are extracted from the bucket indexed by Γ_k .

2.D.2 Approximative hashing

Unfortunately, the process as described previously corresponds to an exact matching search step, and it is very sensitive to bit corruptions, and so to audio degradations. For example with audio degradations, if at least one component z_k changes sign (it has a high probability if the original value of z_k is close to 0), then the value of γ_k changes and the whole index Γ is corrupted. In this case, the extracted song ids do not correspond to the original code. Even if the obtained reduced audio prints are robust to degradation and discriminant, this hashing process must be adapted to improve its tolerance to bit corruptions.

To deal with this problem, in (Moravec and Cox, 2011) an *approximative hashing* is proposed for audio indexing, and we developed a similar method. It is inspired from the *Local Sensitive Hashing* (LSH) of (Indyk and Motwani, 1998; Andoni and Indyk, 2006) which approximates the solutions of a problem similar to this of the *K-Nearest Neighbours* algorithm, but using a hashing system for its speed. The overall idea is to derive several hash codes, using different hash functions, but for a same hash key. If the initial key is modified, some hash codes may be corrupted, but it is possible that other codes remain unchanged, accordingly to the strength of the modification.

In this work, each 40 bits code Γ is computed by binarization of z , see eq. (25), and $L = 51$ lsh codes β_ℓ are computed with $b = 16$ bits. Each smaller lsh code β_ℓ is built using 16 bits of Γ with a pseudo-random, and reproducible, selection. The principle is to use a different combination of bits from Γ for each lsh codes, but the bit selections, defining the lsh functions h_ℓ , remain identical for all different input Γ . In consequence, if some bits of Γ are altered, some of the $L = 51$ lsh codes are corrupted too, but others may be unchanged, and are still usable for the detection. The process is illustrated in Fig. 14.

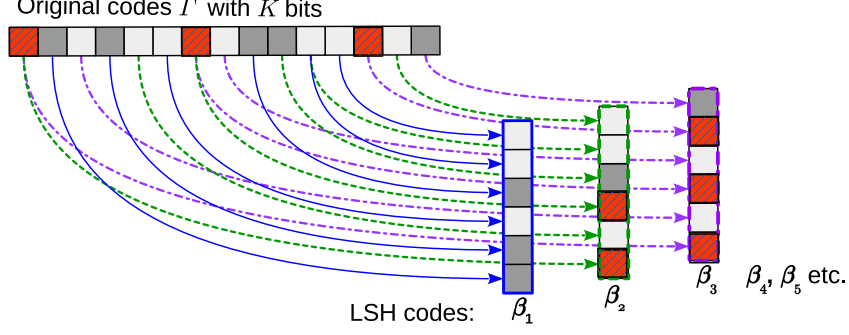


Figure 14: Illustration of the approximative hashing. Each lsh code β_ℓ is built by extracting a selected subset of bits from Γ . In this illustration, $K = 15$ and $b = 6$. The gray cells represent bits to 1, and the white cells represent bits to 0. The red hatched cells represent corrupted bits. Whereas β_2 and β_3 are affected by the bit changes, β_1 is not corrupted.

For example: with k altered bits in Γ , by approximating with independent variables, the probability that one bit is not altered becomes $p_k = (1 - k/K)$, and the probability that a lsh code is unchanged is $P_k = p_k^b$. With X the number of unchanged lsh codes, $0 \leq X \leq L$, the statistical mean of unchanged lsh codes is then:

$$\mu_k = E[X / k] = \sum_{\ell=0}^L \ell \pi_k(\ell), \quad (26)$$

$$\text{with } \pi_k(\ell) = \text{Proba}[X = \ell / k] = C_\ell^L P_k^L (1 - P_k)^{L-\ell}. \quad (27)$$

Now for unmatching audio keys, considering random drawing of Γ , the statistical mean of code collisions Y (which does not depend on k , related to the audio degradation) is:

$$\nu = E[Y] = L/2^b \approx 7.782 \cdot 10^{-4}. \quad (28)$$

Table 1 provides the values of μ_k and of the ratio $\rho_k = \mu_k/\nu$ for $0 \leq k \leq 20$. Note that because $K = 40$, with $k = 20$, that is half of bits are corrupted, the ratio is 1: as many good matches as there are code collisions with unmatching audio keys.

k	0	1	2	3	4	5	6	7	8	9
μ	51	34.0	22.4	14.6	9.45	6.02	3.78	2.34	1.43	0.86
ρ	65536	43.7 e3	28.8 e3	18.8 e3	12.1 e3	7.7 e3	4.8 e3	3.0 e3	1.8 e3	1.1 e3

k	10	11	12	13	14	15	16	17	18	19	20
μ	0.51	0.29	0.16	0.095	0.052	0.028	0.014	0.007	0.004	0.002	0.001
ρ	656.8	381.8	217.8	121.7	66.5	35.5	18.5	9.35	4.59	2.18	1

Table 1: μ_k : statistical mean of unchanged lsh codes with k altered bits (in Γ), and ρ_k : ratio μ_k/ν , with ν the statistical mean of collisions (for unmatching audio keys). The values of μ and ρ are rounded to the most significant digits.

2.D.3 For a unique hash table

For each analysis time, L lsh codes (with $b = 16$ bits) are obtained for all the $n_b = 5$ frequency bands. The $L \times n_b$ obtained codes cannot be mixed to index the same hash table with size 2^b . Either we built $L \times n_b$ different hash tables, or equivalently we can extend the hash table with $S = L \times n_b \times 2^b$ entries, or buckets. With $n_b = 5$, and $L = 51$, the number of lsh codes for each analysis times is then $n_b \times L = 255$ which can be coded with 8 bits. Then, to simplify the implementation, we build an extended hash table, which is indexed with extended codes on 24 bits.

2.D.4 Lightening of the table using the most reliable codes

To make lighter hash tables, one idea is to limit the number of used lsh codes for each analysis time and each band. That is, instead of using all the L obtained lsh codes, only $L' < L$ are used. Among the L codes, the idea is to choose these ones which are the less sensible to degradations. Based on a reliability indice as in (Haitsma and Kalker, 2002), the used lsh codes are the ones which have the smallest probability to be altered.

To explain the principle, let's take the example of an audio print component $z_k = 0$. With eq. (25), its binarization produces $\gamma_k = 1$. But adding a "centered" random value to z_k , simulating the degradation effect, the probability that γ_k becomes 0 is 0.5. In the opposite case, the further z_k is from 0, the more the probability of change of γ_k decreases. In consequence, for each lsh code β_ℓ , the reliability indice is directly based on the magnitude values of the z_k 's, for the bits which compose β_ℓ .

First, a reliability indice is computed for each component of z_k : considering the effect of degradation as an additive perturbation e modeled by a Gaussian process, centered and with an empirical variance, we can compute the probability p_k that $z_k + e$ changes sign. Then, according to the bit selection of each lsh function, we can derive the probability π_ℓ of a lsh code that at least one bit changes value, and $1 - \pi_\ell$ is used as reliability indice for the lsh code β_ℓ . Then, for each times and each band, the $L' < L$ lsh codes with the highest reliability are used to fill the hash table.

Note only this makes possible to have a smaller hash table, with a shrinking factor $\sigma = L'/L$, but also the ratio $\rho = \mu/\nu$ increases, see eqs. (26,28). The mean of code collisions ν with unmatching audio keys is reduced proportionally to σ , but the mean of good detections μ decreases significantly slower because the more reliable lsh codes are used, that is the corruption probability of the really used bits is lower. In the experiments, we chose $L' = 10$, cf. sec. 3.

2.E Search step

After having built the hash table with the music catalog of reference items, see previous sub-sections, the lsh codes of the query excerpts are computed. Then, the number of matching codes between the query excerpt and each reference item of the catalog is fastly obtained using the hash tables. See an illustration of the process in Fig. 15.

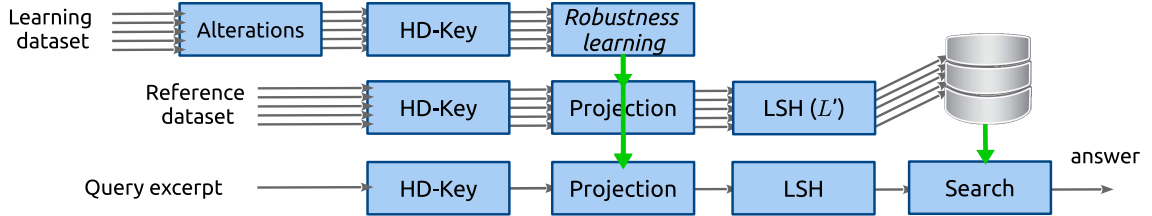


Figure 15: Illustration of the whole process. First: the elements of the training dataset are altered, the corresponding HD-audio keys are computed, and the dimension reduction is learnt (see sec. 2.C). Second: the HD-audio keys of the reference elements are computed, then the dimension is reduced with the learnt reduction and the hash table is built. Third: the HD-audio keys of the query excerpt are computed, and projected with the same transformation, and all the L lsh codes are used to find matching codes in the hash table.

Remark that during the search step, all the L lsh codes obtained from the query excerpt are used. The selection of the L' most reliable codes is applied only for the reference music piece during the building of the hash table, to reduce its size. The reliability indice of the lsh codes of the query excerpt can be used as weight, in order to favor the most robust codes.

2.E.1 Collision with unmatching items

Nevertheless, collisions are possible with unmatching reference items. Indeed, the same hash code can be produced from different initial signals, and so, selecting the reference item only based on the number of matching codes is not enough, and may provide a high number of false positives. Note

that because the audio prints are based on new audio descriptors which are relevant to the content of the input music, some times, the code collisions reveals music similarity, but it is not the case all the time.

For example, with $b = 16$ bits lsh codes, $L = 51$ lsh functions, $L' = 10$ reliable codes, $n_b = 5$ bands, $F_a = 4$ the mean number of analysis times per second, and for a $D_r = 30$ sec long reference song, and a $D_e = 30$ sec long query excerpt, considering independent random drawing of lsh codes, the statistical mean of number of collisions between the query and the reference (with possible repetitions) is:

$$N_c = D_r D_e F_a^2 L' n_b / 2^b \approx 10.98. \quad (29)$$

And in the case of the true matching reference without degradation, the number of good code matches (without repetition) is ideally:

$$N_t = \min(D_r, D_e) F_a L' n_b = 6\,000. \quad (30)$$

But with degradations, this number strongly decreases, see Tab. 1. For this reason, a post-process is applied, and it is based on the coherence of time positions of the matching codes between the query and some reference items. In sec. 2.E.3 an overview of the time-coherence analysis is done.

2.E.2 Global search strategy

But analyzing the time-coherence with all the reference songs of the catalog would be too long. The used strategy approximately follows the approach of (Wang, 2003):

- 1 The hash table is used to select a small number N of candidates,
- 2 The time-coherence between the excerpt and the reference candidates is analyzed in order to select the best candidate.

Remark that, because the number of collisions is proportional to the duration D_r of the reference song, see eq. (29), some problems may occur if some reference elements are significantly longer than others. A solution is to split the reference songs into short segments (e.g. 15sec), without overlap, and the segments are indexed rather than the complete songs. Then, the count of code matches per short segment is obtained through the hash table, and finally, this count is integrated on a longer sliding window, with overlap, and with a duration based on the query excerpt.

2.E.3 Time-coherence analysis

In this section, we briefly present the computation of a time coherence measure of matching codes between the query excerpt and a given candidate reference song. The selection of the candidate is presented in sec. 2.E.2.

In the initial work done in 2015, we adapted and improved the approach of (Wang, 2003). Given two sequences of time positions of matching codes, t_n for the reference song and τ_n for the excerpt signal, the principle is to detect a linear relation for a subset of pairs $[t_n, \tau_n]$ in the plan (t, τ) . Indeed, if the query is really an excerpt of the reference item, we may observe a linear alignment of some pairs $[t_n, \tau_n]$, with the relation $\tau_n = t_n - \delta_t^*$, with δ_t^* the starting time of the excerpt. But if it is not an excerpt of the reference song, the pairs correspond to false positives and should be randomly positioned in the plan (t, τ) .

To summarize the approach of (Wang, 2003), defining a short window size σ (used as error margin), for different values of δ_t the number of pairs respecting $\|\delta_t + \tau_n - t_n\| \leq \sigma$ is count. Finally if a significant peak appear for a value of δ_t , it is assume that the input excerpt matches with the tested reference item, and the obtained time offset is the value of δ_t^* . Also, the count value at the found δ_t can be used as time-coherence measure to select or reject the candidates. An illustration of the search is presented in Fig. 16. Remark that the computation is actually done in another plan given by the change of variables: $(t - \tau, \tau)$. In this plan, the time coherence of pairs appears with a vertical alignment. This actually simplifies the calculus.

Nevertheless, this approach does not deal with time stretching. In this case, we must consider the following relation:

$$\tau_n = \alpha t_n - \delta_t^*, \quad (31)$$

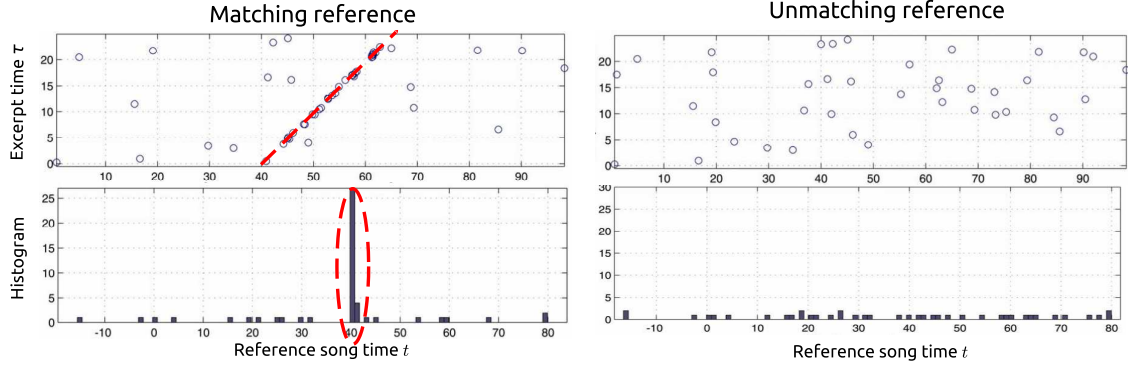


Figure 16: Illustration of the Shazam search strategy. On the upper-left sub-figure are plotted all time pairs of matching codes. The time coherence is displayed by the linear arrangement around the red dashed line. The count permits to exhibit this matching, see the bottom-left sub-figure. On the right sub-figures, the reference item does not match with the input excerpt, the pairs are random positioned in the plan (t, τ) , and no linear arrangement are observed. These figures come from (Wang, 2003).

where α is the unknown time stretching factor. Applying the Shazam’s method with time stretching may work with α close to 1, but the performances significantly fall with a stronger time stretching. A naive idea would be to increase the margin σ in order to cover the deviation due to the time stretching, but this is not sufficient.

During the *BeeMusic* project, in 2015, we proposed a pre-processing in order to significantly improve the result with a stronger time stretching. First, for each pair n a cone is defined with (t_n, τ_n) as top, and delimited by two lines with slopes α_{\max} and $1/\alpha_{\max}$, where $\alpha_{\max} > 1$ is the maximal acceptable time stretching factor. Second, for each pair n , the number of other pairs inside the defined cone is used as weight. This weight is then used during the histogram count previously presented. In the case where the margin σ is too small, because of the possible time stretching it may be possible that with the suited δ_t , the integration windows are too short to count all the matching pairs of the line. Using this pre-processing, even if the window is too short, the pairs on the line that are outside the window are still taken into account through the weights of the pairs found inside. Not only this pre-processing permits the analysis with time-stretching, but it also improves the results, without time-stretching, and with a true positive rate lower.

Finally, a linear regression is computed with selected pairs in order to refine the estimation of the time stretching factor α and the time offset δ_t^* .

3 Evaluation and results

This section briefly presents the evaluation of the performance of the recognition of a degraded music excerpt with a relatively big catalog of reference music. More details are presented in (Mignot, 2015b), in french.

3.1 Tested catalog

The tested catalog is based on a private dataset provided by a partner of the project, and internally usable for the period of the project. We extracted a subset with 755 762 music excerpt with a duration between 25 and 35 seconds, and we removed duplicates based on the given ISRCs. A lot of different music genres are represented: ambiance, blues, classical, country, techno/dance, children, jazz, metal, world, soul/funk, rock, movie, pop, variety, rap/hip hop, raggae, gospel.

For the first step of the search, the hash table uses 16 GigaBytes which can be loaded into the RAM. For the second step, all the data of the *direct table* can be loaded into the RAM too, because it uses only 4 GigaBytes.

3.2 Two steps search

For the recognition of a music excerpt, we present two measures of performances in Tab. 2. Each measure corresponds to a step in the search process.

Step 1

The first displayed performance is the rate of good detections for step 1. For this step, a good detection happens when the reference item with the higher number of matching codes is the true reference song, which corresponds to the input excerpt.

Step 2

For the analysis of the time coherence, we finally adopted a different strategy to define the number of candidates to test: with N_i the number of code matches of the song i , sorted with decreasing N_i , so N_1 is the maximal value, the tested candidates i are such that $N_i \geq N_1/2$. Nevertheless, we use a minimal number of 10 and a maximal number of 500 candidates to test. Finally the best candidate is selected through the measure of the time-coherence, and the second displayed value in Tab. 2 is the rate of good detection after the second step.

3.3 Evaluation protocol

The hash table (step1) and the direct table (step2) are built with the 755 762 reference excerpts (30 seconds) without any degradations. Then we selected 7,146 test excerpts with 7 seconds duration from the reference catalog. Each tested excerpt is then degraded with 18 different degradations and with 3 different levels of degradations (strengths). Here are the tested degradations:

- **Graphical equalizers** with 10 bands, and with alternative gain: $\pm\alpha$ dB. The values of α for each of the 3 levels are: **3dB, 6dB and 9dB**. Note that for the strongest degradation a lag of 18db are between maxima and minima of the response.
- **Environmental noise recorded in a restaurant**. The used SNRs (Signal-to-Noise Ratio) for the levels are: **12dB, 6dB et 0dB**. Remark that for the strongest noise, its energy is the same than the energy of the music to recognize.
- **Environmental noise recorded in a bus**. The same SNRs are used.
- **Environmental noise recorded in a street**. The same SNRs are used.
- **Synthetic white noise**. The same SNRs are used.
- **Synthetic pink noise**. The same SNRs are used.
- **Pitch shifting to higher frequencies**. The software SuperVP is used here, with the preservation of the formants and of the attack transients. The used transpositions are: **1/4, 1/2, and 1 tone**.
- **Pitch shifting to lower frequencies**. The used transpositions are: **-1/4, -1/2, and -1 tone**.
- **Expansion of the time: slow-down** time stretching. SuperVP is also used with the same options. The used stretching are: **15, 30 and 45 cents**. Remark, with β the stretching in cents, the duration is multiplied by the factor $2^{\beta/100}$.
- **Compression of the time: speed-up** time stretching. The used stretching are: **-15, -30 and -45 cents**.
- **MP3 coding**. The coding is done with the lame software and with the following bitrates: **32kbps, 24kbps, and 16kbps**.
- **Distortion**. Here a arctan non-linear function is applied on the time signal. With L_∞ normalized input signal, the level is given by the input gain (before the function): **5dB, 12dB and 24dB**.

- **Dynamic range compressor.** We use here a 4 frequency bands compressor. The level is both controlled by the compression ratio and the release time: **ratio 2, 8 and 50; release time 100ms, 10ms, et 1ms.**
- **Tremolo.** A cyclic modulation of amplitude (4Hz) is applied with a gain lag (between maxima and minima): **$\pm 3\text{db}$, $\pm 6\text{db}$, $\pm 9\text{db}$.**
- **Reverberation.** We use here a convolution reverberation with an impulse response of a medium hall. The level is controlled with the mix ratio between the dry signal and the reverberated sound: **9dB, 3dB, and 0dB.**
- **Scenario GSM.** Here we test a series of different degradations: filtering with the response of a smartphone microphone (NexusOne), white noise addition (SNRs **18dB, 12dB and 6dB**), and a GSM coding is the main degradation (no parameter).
- **Scenario slow-down.** Here the following series of alterations is used: slow-down with stretching factors: **4, 8 and 12 cents**, 10-bands EQ with $\pm 3\text{dB}$, 4-bands compressor with a ratio 2, MP3 coding at 32kbps, reverberation with 3dB, and noise addition with a 18dB SNR.
- **Scenario noise.** Here the following series of alterations is used: slow-down (4%), 10-bands EQ ($\pm 3\text{dB}$), 4-bands compressor (ratio 2), MP3 coding (32kbps), reverberation (3dB) and noise addition with SNRs: **18dB, 12dB and 6dB.**

3.4 Results

Table 2 gives the success rates for the 2 steps:

- 1) *STEP1*: ratio in percents of successful recognition at the step 1, based on the hash table only (with 755 762 references). This performance does not take into account the time coherence analysis.
- 2) *STEP2*: ratio in percents of successful recognition at the step 2, based on the time-coherence analysis of the best candidates selected after step 1.

As expected the performances with the second step is better than with only the first first. Even if the performance of STEP1 is promising, this observation proves the benefit of the time coherence analysis. Indeed, in many cases the true reference song is among the selected candidates, but not the first. Others have a high number of matching codes, but with random time positions, which do not fit a linear alignment. Unfortunately, in some cases, the true candidate is not among the selected candidates, and STEP2 cannot find it.

This evaluation demonstrate the ability of the method to efficiently recognize a degraded excerpt among a big catalog, with more than 700 000 music pieces, and with many different kind of degradations. But for some degradations, such as the noise addition, we know that this method is not as performant as other methods based on fingerprints, such as Shazam (spectrum peaks). Nevertheless, here other transformations such time stretching and pitch shifting are managed without affecting the robustness to other degradations.

For more details about this evaluation, other performances measures are presented in (Mignot, 2015b).

4 Conclusion

We presented in this report the achieved work which has the goal to improve the audio indexing system at IRCAM, with both a robustness to audio degradations and the feasibility to scale the method for big catalogs. The audio degradations are taken into account at different steps on the process: selection of the analysis times, sec. 2.A, high-dimensional audio prints with natural robustness to some degradation, sec. 2.B, learning of dimension reduction of criteria based on robustness and discrimination for different initial signals, and with a preparation for the hashing, sec. 2.C, approximative hashing for a higher robustness to bit corruption, and so to audio degradations, sec. 2.D,

and finally the search process based on the hash table and then a time-coherence analysis adapted for time stretching, sec. 2.E. The evaluation in sec. 3 proves an interesting robustness and also the scalability of the method.

This work was achieved in 2015, in consequence, it did not take benefit of the recent advances in signal processing and machine learning. For example, it would be interesting to test the *Scattering Transform* of (Andén et al., 2015; Lostanlen and Mallat, 2015) in place of the HD-audio prints; with a possible adaptation to invariance to time and frequency compression/expansion rather than translation. Also, during the past years, significant progresses have been obtained in *Deep Learning*, and keeping the same architecture of the present audio indexing system, we could improve many steps with *Deep models*. For example, the analysis times selection should be easily implemented with a *Neural Network*, where the ground truth is the output of the model on original signal. Moreover, the LDA and the OMPCA reductions used in this work are similar to a metric learning, and it should be possible to reimplement this reduction using a *Triplet Loss* (Chechik et al., 2010) for example, with additional losses related to the independences of the outputs variables, and the preparation for the hash table building.

	STEP 1	STEP 2
Background noise: Restaurant		
strength 1	98.5 %	98.6 %
strength 2	96.8 %	97.0 %
strength 3	65.4 %	68.8 %
Background noise: Bus		
strength 1	98.5 %	98.6 %
strength 2	96.7 %	97.4 %
strength 3	70.9 %	81.0 %
Background noise: Street		
strength 1	98.5 %	98.5 %
strength 2	96.5 %	97.0 %
strength 3	67.5 %	71.5 %
White noise		
strength 1	98.7 %	98.9 %
strength 2	98.4 %	98.5 %
strength 3	92.3 %	95.8 %
Pink noise		
strength 1	98.6 %	98.7 %
strength 2	96.9 %	97.6 %
strength 3	74.5 %	84.5 %
Pitch Shifting plus		
strength 1	98.1 %	97.9 %
strength 2	95.5 %	97.0 %
strength 3	75.2 %	88.2 %
Pitch Shifting minus		
strength 1	98.2 %	98.1 %
strength 2	95.9 %	97.2 %
strength 3	77.4 %	89.2 %
Time Stretching slower		
strength 1	97.3 %	97.4 %
strength 2	93.5 %	95.4 %
strength 3	82.5 %	86.4 %
Time Stretching faster		
strength 1	97.6 %	97.8 %
strength 2	95.3 %	96.1 %
strength 3	91.5 %	87.8 %

	STEP 1	STEP 2
Equalizer		
strength 1	98.7 %	98.7 %
strength 2	96.4 %	97.7 %
strength 3	71.2 %	87.9 %
MP3		
strength 1	98.9 %	98.8 %
strength 2	98.8 %	98.8 %
strength 3	98.5 %	98.5 %
Distortion		
strength 1	98.0 %	98.0 %
strength 2	95.9 %	96.9 %
strength 3	82.4 %	89.0 %
Multi-band Compressor		
strength 1	98.5 %	98.5 %
strength 2	92.2 %	95.5 %
strength 3	80.4 %	88.1 %
Tremolo		
strength 1	98.7 %	98.7 %
strength 2	97.9 %	98.0 %
strength 3	92.6 %	95.4 %
Reverberation		
strength 1	98.8 %	98.7 %
strength 2	97.5 %	97.8 %
strength 3	88.3 %	95.8 %
Scenario GSM		
strength 1	88.6 %	95.3 %
strength 2	86.3 %	94.0 %
strength 3	77.2 %	90.9 %
Scenario speed		
strength 1	93.5 %	96.6 %
strength 2	83.1 %	93.0 %
strength 3	61.9 %	80.3 %
Scenario noise		
strength 1	93.6 %	96.5 %
strength 2	90.3 %	95.2 %
strength 3	76.6 %	86.6 %

Table 2: Table of results: for all the 18 tested degradations, with 3 different strenghts of degradations, and the 2 steps. As expected, the performance of the step 2 is higher than with step 1 only.

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