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Multi-channel audio source separation using multiple deformed references

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Abstract—We present a general multi-channel source separation framework where additional audio references are available for one (or more) source(s) of a given mixture. Each audio reference is another mixture which is supposed to contain at least one source similar to one of the target sources. Deformations between the sources of interest and their references are modeled in a linear manner using a generic formulation. This is done by adding transformation matrices to an excitation-filter model, hence affecting different axes, namely frequency, dictionary component or time. A nonnegative matrix co-factorization algorithm and a generalized expectation-maximization algorithm are used to estimate the parameters of the model. Different model parameterizations and different combinations of algorithms are tested on music plus voice mixtures guided by music and/or voice references and on professionally-produced music recordings guided by cover references. Our algorithms improve the signal-to-distortion ratio (SDR) of the sources with the lowest intensity by 9 to 15 decibels (dB) with respect to original mixtures.

Index Terms—Source separation, GEM algorithm

I. INTRODUCTION

In audio signal processing, source separation consists in recovering the different audio sources that compose a given observed audio mixture. It has been a hot topic over the past decade and this field of research now offers a wide variety of new possible applications for end-users and professionals. One of those concerns the remastering, restoration and remixing of movie soundtracks or musical recordings. Sound engineers may want to upmix the recordings to a higher number of channels, to remove some sources, to generate a karaoke version, or to substitute some sources by other sources, for instance in order to replace the original soundtrack of a movie with a new one. For these purposes, one needs high source separation quality, which is not yet achievable by blind source separation methods [1]. Taking additional information into account is necessary to improve the separation [2], [3]. In informed source separation methods [4], detailed information about the original sources is transmitted along with the mixture to be separated. Such methods are the ones that provide the best quality but they cannot be applied in the scenario considered hereafter, since the original sources are never observed.

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Guided source separation is based on the use of any kind of additional information and has recently been more and more focused on. It is well adapted to scenarios where the original sources are not available but high separation quality is nevertheless required. The additional information can be of different types: spatial and spectral information about the sources [5], [6], language structure [7], visual information [8], information about the recording/mixing conditions [9], musical scores [10]–[13], or user input [14]–[21]. For instance, the user can provide relevant information by drawing the fundamental frequency curve [18], by uttering the same sentence [16], by humming the melody [14], or even by selecting specific areas in the spectrogram of the mixture [17]. On top of this, interactive approaches allow the user to give feedback during the separation [19]–[22].

In this paper, we focus on methods that guide the separation process by a reference signal that is similar to one of the target sources [10], [14]–[16], [23]–[27]. Such a framework can be referred to as reference guided source separation, and it has recently been used in several scenarios: the restoration of music pieces guided by isolated piano sounds [10], the separation of music and sound effects from speech guided by several versions of the same movie in different languages [23], the separation of musical instruments guided by a multitrack cover version of a song [24], [25], and the denoising of speech guided by the same sentence pronounced by the same speaker [26] or by a different speaker [16]. Symbolic information such as a text [16] or a musical score [12] can also be used to generate reference signals.

Here, we propose a general model for multi-channel reference guided source separation that enables the joint use of multiple, multi-channel, deformed reference signals. In this article, multi-channel can refer either to the case when a reference is considered as another channel (see Section V-B3b) or to the case when the reference is multi-channel (see Section VI-D2). Our preliminary experiments on music/speech separation [27] and cover guided music separation [25] showed that the use of references is relevant in the single-channel case. Here, we extend this approach to the multi-channel case using a Generalized Expectation-Maximization (GEM) algorithm inspired from [5]. Several initialization procedures and model configurations are investigated as well as the use of multiple references for each source. Different types of data and references are used to assess the relevance of our approach.

The paper is organized as follows. Section II introduces the general model of reference guided source separation. Section III presents the different algorithms and initialization procedures that are compared in the experiments. Section IV
provides a first series of experiments with pitch shifted references. Sections V and VI are respectively dedicated to experiments on music/voice separation and on cover-guided music separation.

II. GENERAL FRAMEWORK

In this section, we describe the proposed reference guided source separation model. Key concepts are first presented in order to facilitate understanding before we give a detailed description of the model. We also discuss different configurations of our model that can handle different types of available data and possible extensions of our framework.

A. Overview of practical scenarios

What we mean by reference signals ranges from different recordings of the true sources to noisy versions of the true sources and also include imitations. The references usually rely on several and very different deformations like time misalignment, time warping, changes of speaker/singer/instrument, additional overlapping sources, equalization, changes of melody/pronunciation, change of recording conditions, and/or pitch shifting. If there is no deformation at all, the reference is then the true source and as already mentioned this is a very restricted scenario. In this sense, we can say that a reference signal is by nature deformed. Hereafter, we consider that what we mean by reference signals ranges from different mixtures (sources to be recovered) and several other reference mixtures.

In this framework, the deformations between the reference signals and the target sources are modeled in a generic linear model. We also discuss different configurations of the model. We also discuss different configurations of the model. We also discuss different configurations of the model.

B. Input representation

The observations are $M$ audio mixtures $x^m(t)$ indexed by $m$. Each mixture $x^m(t)$ is multi-channel and contains $I^m$ channels.

Each mixture is assumed to be the sum of the spatial images $y_j(t)$ of one or more sources indexed by $j \in J_m$:

$$x^m(t) = \sum_{j \in J_m} y_j(t) \text{ with } x^m(t), y_j(t) \in \mathbb{R}^{I^m}.$$  \hspace{1cm} (1)

In the Short-Time Fourier Transform (STFT) domain, this can be written as

$$x^m_{f,n} = \sum_{j \in J_m} y_{j,f,n} \text{ with } x^m_{f,n}, y_{j,f,n} \in \mathbb{C}^{I^m}.$$  \hspace{1cm} (2)

where $f = 1,...,F$ and $n = 1,...,N$ are respectively the frequency and the time indexes of the STFT. We consider that $x^1(t)$ is the mixture to be separated, and $x^m(t)$ for $m > 1$ are other mixtures containing the reference signals used to guide the separation process.

We assume that the STFT coefficients of the source spatial images $y_{j,f,n}$ have a zero-mean Gaussian distribution $[5]$

$$y_{j,f,n} \sim \mathcal{N}_C(0, v_{j,f,n} R_{j,f})$$  \hspace{1cm} (3)

where $v_{j,f,n}$ and $R_{j,f}$ are the spatial parameters, which can be non-uniquely represented as $V_j = \{v_{j,f,n}\}_{n=1}^{N} \in \mathbb{R}_{+}^{F \times N}$ and $\mathbf{R}_j \in \mathbb{C}^{I^m \times I^m}$.

1) Spatial parameters: The spatial covariance matrices model the spatial characteristics of the sources, such as phase and intensity difference between channels. We only consider time-invariant spatial covariance matrices as the sound sources considered in our experimental scenarios are generally spatially stable over time. $\mathbf{R}_{j,f}$ can be non-uniquely represented as $\mathbf{R}_{j,f} = A_{j,f} A_{j,f}^H$ where $A_{j,f} \in \mathbb{C}^{I^m \times R_j}$ and $R_j$ is the rank of matrices $\mathbf{R}_{j,f}$ and $A_{j,f}$.

2) Spectral parameters: The power spectrogram of each source $j$ is denoted as $V_j = \{v_{j,f,n}\}_{f=1}^{F}$ and a filter spectrogram $V_j^\phi$. The excitation spectrogram (resp. the filter spectrogram) is decomposed by Nonnegative Matrix Factorization (NMF) into a matrix of spectral patterns $W_j^e \in \mathbb{R}_{+}^{F \times D^e}$ (resp. $V_j^\phi \in \mathbb{R}_{+}^{F \times D^\phi}$) and a matrix of temporal activations $H_j^e \in \mathbb{R}_{+}^{N \times D^e}$ (resp. $H_j^\phi \in \mathbb{R}_{+}^{N \times D^\phi}$). $D^e$ (resp. $D^\phi$) denotes the number of spectral patterns used in the NMF decomposition of the excitation (resp. filter) part. This results in the following decomposition:

$$V_j = V_j^e \odot V_j^\phi = W_j^e H_j^e \odot W_j^\phi H_j^\phi$$  \hspace{1cm} (4)

where $\odot$ denotes pointwise multiplication. The four matrices are as follows:

- $W_j^e$ is a spectral dictionary that can be designed as a set of inharmonic, harmonic and/or wideband spectra [5]. Alternatively, such a dictionary can be learned on training data or estimated from the test mixture.
- $H_j^e$ are the corresponding temporal activations which encode, e.g., the musical score in the form of a piano roll [11]–[13], or the $f_0$ track [18].
- $W_j^\phi$ is a dictionary of spectral envelopes associated with, e.g., different phonemes in the case of speech [16] or body resonances in the case of a musical instrument [10].
- $H_j^\phi$ are the corresponding temporal activations which encode, e.g., the phoneme sequence for speech or instrument timbre changes for music such as muted/unmuted trumpet.

C. Proposed model with multiple deformed references

We proposed to consider three different cases for the settings of the matrices $W_j^e$, $H_j^e$, $W_j^\phi$ and $H_j^\phi$. As a first case, they can be fixed and remain unchanged during the estimation. As a second case, they can be set as free parameters, which means that they will be adapted to the corresponding mixture.
We propose to model the sharing of spectral and temporal sources as in Section V. In this last case, the deformations between sources are modeled by transformation matrices $T_{jj'}$. We propose to model the sharing of spectral and temporal properties between one source $V_j$ and its references $V_{j'}$ as follows:

1) Transformation matrices for the excitation part: For the excitation part, the transformation matrices are denoted as $T_{jj'}^{fe} \in \mathbb{R}_+^{N \times D}$, $T_{j,j'}^{dfe} \in \mathbb{R}_+^{D \times D'}$ and $T_{j,j'}^{e} \in \mathbb{R}_+^{N \times N'}$. Depending on the actual deformation between the target source and the reference sources, three different configurations are possible. One may share either the spectral patterns as shown in (5), the temporal activation as shown in (6), or both as shown in (7). This is modeled by one of the three following equations:

$$V_j^e = T_{jj'}^{fe} W_j^e H_j^e$$  \hspace{1cm} (5)
$$V_j^v = W_j^e H_j^e T_{jj'}^{dfe}$$  \hspace{1cm} (6)
$$V_j^v = T_{jj'}^{fe} W_j^e H_j^e T_{jj'}^{dfe}.$$  \hspace{1cm} (7)

During the estimation process, each transformation matrix can be considered either as a fixed or a free parameter. In practice, frequency deformations of the excitation $T_{jj'}^{fe}$ can be used to model, e.g., differences of reading speed on analog devices or spectral dimensionality change due to different sampling rates. $T_{jj'}^{dfe}$ is used to time-align the signal spectra and represents the warping path between the two signals. $T_{jj'}^{e}$ can be used to model changes in the excitation dictionary, such as pitch shifting\(^1\). It only appears when the corresponding $W_j, H_j$ are shared, otherwise it would be redundant.

2) Transformation matrices for the filter part: For the filter part, the transformation matrices between the target source and the reference sources are denoted as $T_{jj'}^{f} \in \mathbb{R}_+^{F \times F}, T_{j,j'}^{df} \in \mathbb{R}_+^{D \times D'}$ and $T_{j,j'}^{e} \in \mathbb{R}_+^{N \times N'}$. In the same way as above, three different configurations are possible. Let $j$ be the index of the target source, and $j'$ be any reference source. The temporal activation as shown in (9), or both as shown in (10):

$$V_j^f = T_{jj'}^{fe} W_j^f H_j^f$$  \hspace{1cm} (8)
$$V_j^v = W_j^f H_j^f T_{jj'}^{df}$$  \hspace{1cm} (9)
$$V_j^v = T_{jj'}^{fe} W_j^f T_{j,j'}^{df} H_j^f T_{jj'}^{e}.$$  \hspace{1cm} (10)

Similarly to transformation above, the matrices can be either fixed or free. Frequency deformations of the filter part $T_{jj'}^{f}$ can be used to model, e.g., changes in vocal tract length \cite{16} or a different equalization. $T_{jj'}^{df}$ models changes in the filter dictionary, such as the change of some phonemes in the case of a speaker with a different accent (e.g., one phoneme often uttered in place of another), and it only appears when the corresponding filters $W, H$ are shared. $T_{jj'}^{e}$ models the temporal deformation of the filter, and it is used to time-align the signals.

\(^1\)It can be noticed that pitch shifting and reading speed have two different effects, especially for inharmonic sounds.

Fig. 1 gives an illustration of a possible use of this model. It corresponds to a speech reference modeled by (30) and the related speech source that is uttered by a different speaker. More details can be found in Section V-B3a.

D. Comparison with previous approaches

The proposed framework generalizes the state-of-the-art approaches in \cite{10, 16, 27} as they exploit similar models. Our framework can also model the same kind of signals as used in \cite{10, 14, 23–26} even if the models can be quite different. Finally, it makes it possible to investigate some new scenarios that have been put forward in \cite{27}, like music source separation for a verse guided by another verse.

E. Extensions of our approach

As previously mentioned, we consider here that $j \in \mathcal{J}^1$ and $j' \in \mathcal{J}^m$ with $m' \neq m$. These notations are supposed to represent the classical reference guided source separation scenario. Relaxing this constraint opens the way to more possibilities.

Modeling the relationship between sources of the same mixture (i.e., $j, j' \in \mathcal{J}^m$) could be of interest to model delays between sources of the same mixture like a canon in music. Modeling "circular" relationships (e.g., using $T_{j,j'}, T_{j',j''}, T_{j''}$) would allow joint separation of all mixtures. But it requires the use and the estimation of one more matrix. More generally, considering the mixture to be separated as central is a good way to avoid having additional matrices and potential smoothing effects on the sources of interest.

Another possible extension would be the sharing of transformation matrices, for instance when different instruments undergo the same transformation (tonality or pitch shift) or when the excitation and the filter part are subject to the same time deformation\(^2\).

F. Additional constraints

When using an excitation-filter model, the estimation of the filter part usually requires additional smoothness constraints, as in \cite{5, 28}, to guarantee that the filter part actually represents the resonances of the vocal tract or the musical instrument. In this work, we do not use any explicit constraint for our excitation-filter model in order not to overload the framework. However, we found experimentally that the estimated filters in the presence of reference-based constraints are smooth. Fig. 2 shows examples of estimated filters for the experiment in Section V. One possible explanation is that using more than one source to estimate a given filter (while the excitation is different for each source) yields a more robust estimate than using only one source. However, it is hard, if not impossible, to provide some theoretical guarantee of this behavior.

If needed, explicit frequency smoothness constraints can be included in the previously presented model either by constraining the matrices $W_j^e$ as the product of smooth frequency-localized patterns and spectral envelope coefficients as in \cite{5},

\(^2\)The shared transformation matrices would then be estimated in a similar manner as the shared parameters in (14) or (16).
We $1 H_j$ $e_2$ $T_1^{j\phi}$ $W^{\phi}_1$ $H^{\phi}_1$ $T^{e\phi}_1$ $V_2 \approx V_1 W^{e}_1 H^{\varepsilon}_1$

Fig. 1. Example estimated decomposition of the spectral power of a reference mixture ($m' = 2$) containing a single source ($j' = 2$) similar to source $j = 1 \in J$. The excitation parameters are not shared whereas the filter follows (10) with $T^{e\phi}_{12}$ set to identity. The parameters in black are fixed, those in green are free and those in red are shared. The parameters of the target source ($j = 1$) are also displayed.

$V_1 = W^{e}_1 H^{\varepsilon}_1$

Fig. 2. Examples of estimated filter parts $V^{\phi}$ for a speech source (2a), and its reference (2b) in the experiment in Section V.

[28] or by introducing probabilistic priors on the coefficients of $W^{\phi}_j$ as in [29]. Similarly, time continuity constraints may be imposed on $H^{\phi}_{j}$ or $H^{\varepsilon}_j$. Such refinements of the proposed general framework are outside the scope of the current article and are not discussed hereafter.

Fig. 2. Examples of estimated filter parts $V^{\phi}$ for a speech source (2a), and its reference (2b) in the experiment in Section V.

III. PARAMETER ESTIMATION

In this section, we present two methods for parameter estimation in the maximum likelihood (ML) sense. The ML objective can be written as:

$$\hat{\theta} = \arg\max_{\theta} \sum_{m=1}^{M} \lambda^m \log p(x^m|\theta)$$

where $\theta$ is the set of parameters to be estimated, i.e., the spatial covariance matrices $R_{j,fn}$, and the matrices $W$, $H$ and $T$ that are either free or shared. $\lambda^m \in \mathbb{R}_+$ are weight parameters that can balance potentially different durations or frequency resolutions between mixtures 1 and $m$, or put more emphasis on the references which are expected to be the most relevant. The reader can refer to [30] for a discussion on their influence on the results.

First, we introduce a multiplicative update (MU) algorithm to deal with single-channel mixtures. Then, a GEM algorithm is used to estimate the parameters in the multi-channel case. Finally, we discuss different initialization procedures.

A. Multiplicative updates for nonnegative matrix partial co-factorization (NMPcF) for the single-channel case

In the single-channel case, maximizing the log-likelihood is equivalent to minimizing the Itakura-Saito (IS) divergence [31]:

$$\hat{\theta} = \arg\min_{\theta} \sum_{m=1}^{M} \lambda^m \sum_{f,n=1}^{F,N} d_{IS}(X^m_{jfn}|W^m_{jfn})$$

where $X^m = [|x^m_{jfn}|^2]_{f,n}$ and $V^m = \sum_{j \in J} V_j$ are respectively the observed and estimated power spectrograms, and $d_{IS}(a|b) = a/b - \log(a/b) - 1$ is the IS divergence. Other divergences are worth considering but they are not extendable to the multi-channel case whereas Itakura-Saito is. A common way to estimate the parameters is the use of a multiplicative
gradient descent approach [31] in which each parameter is updated at each iteration without increasing criterion (12) [32]. The update of one parameter consists in multiplying it by the ratio of the negative and positive parts of the derivative of the criterion with respect to this parameter.

According to their status (free or shared) different MU can be derived for each parameter. For free parameters (in green), (12) leads to the classical MU of NMF. An example of such update is given in (13) for the parameter $W_j^c$. For shared parameters (in red), (12) leads to the MU of NMFcF. An example of such update is given in (14) for the parameter $W_j^c$. $V^{-p}$ denotes the matrix with entries $(V_{ij})^{-p}$.

**B. Generalized Expectation-Maximization (GEM) algorithm for the multi-channel case**

In the multi-channel case, the spatial information can make the separation clearly more tractable, especially when sources have different directions of arrival. In the case of reference guided source separation, the relevance of multi-channel data remains even though the mixtures have different numbers of channels and even though no assumptions are made on the similarity between directions of arrival of the sources and their references.

Following the general framework in [5], we introduce $R_j$ independent Gaussian random variables $s_{jrf} (r = 1, ..., R_j)$ distributed as $s_{jrf} \sim N_C(0, v_{jrf})$ for every source $j$ and time-frequency bin $(f, n)$. An additive isotropic noise $b_{mfn}$ of diagonal covariance $\Sigma_m = \sigma^2 I_m$ in $\mathbb{C}^{F \times I}$ is also added for each mixture $m$. With these changes, (2) becomes:

$$x_{mfn} = A_{mfn}^s f + b_{mfn}$$

where $A_{mfn}^s \in \mathbb{C}^{F \times I}$ (resp. $s_{mfn} \in \mathbb{C}^{F \times I}$) results from the concatenation ($R = \sum_{j \in \mathcal{J}} R_j$) of the mixing matrices $A_{j,fn}$ (resp. all the sub-sources $s_{jrf}$) of all the sources $j \in \mathcal{J}^m$.

EM is a natural algorithm to handle such a parameter estimation in the ML sense in the presence of observed data $X = \{X_m^r\}_m = \{x_{mfn}^r\}_m$, and unobserved data $S = \{S_m^r\}_m = \{s_{mfn}^r\}_m$, that form a complete set $Z = \{X, S\}$. The algorithm proceeds by alternating an E-step that computes the expected value of the complete-data log-likelihood $\mathbb{E}_{Z|\theta^*}[\log p(Z|\theta)] \equiv Q(\theta, \theta^*)$ given the observation and the current set of parameters $\theta^*$, and a M-step that chooses a $\theta$ that maximizes the quantity $Q(\theta, \theta^*)$. In the case of GEM, the M-step only seeks to find a $\theta$ that increases $Q$. A detailed derivation is given in Appendix A. The quantity $Q$ can be written up to a constant as:

$$Q(\theta, \theta^*) = \frac{-\sum_{mfn} \lambda^m_{\theta^*} \text{tr} \left[ R_{x_{mfn}} - A_{f}^m R_{x_{mfn}}^H \right] - R_{x_{mfn}}^H A_{f}^m + A_{f}^m R_{x_{mfn}} A_{f}^m H \right] \right] - \sum_{m,j,e \in \mathcal{J}^m, f} \lambda^m_{\theta^*} R_{j} d_{IS}((\xi_j, j, f), |v_{jrf}|),$$

$^3$ In (14) and (16), we assume that $V_{ij}^c$ and $V_{ij}^c$ follow models (7) and (10). In practical scenarios, the number of shared parameters and transformation matrices will generally be smaller, as exemplified in Sections V and VI.

with: $R_{x_{mfn}} \Delta = \mathbb{E}[x_{mfn} x_{mfn}^H]$, $R_{x_{mfn}} \Delta = \mathbb{E}[x_{mfn} x_{mfn}^H]$, $R_{x_{mfn}} \Delta = \mathbb{E}[x_{mfn} x_{mfn}^H]$ and $\xi_{j,fn} \Delta = \frac{1}{R_j} \sum_{r=1}^{R_j} \mathbb{E}[s_{jrf} s_{jrf}^H]$.

Starting from (18), one can demonstrate that $T(X, S) = \{R_{x_{mfn}}, R_{x_{mfn}}^s, R_{x_{mfn}}^f\}_m, f,n$ is the set of natural (sufficient) statistics [33] for $Z$. This leads to the following two steps of our GEM algorithm.

1) **E-step**: This step consists in computing the conditional expectations of the natural statistics given $\theta^*$:

$$\hat{R}_{x_{mfn}} = \Omega_{s_{mfn}} R_{x_{mfn}}^s + (I_R - \Omega_{s_{mfn}} A_{f}^m) \Sigma_{s_{mfn}}$$

with:

$$\Sigma_{s_{mfn}} = \text{diag}(\varphi_{r,fn})^m_{r=1} \in \mathbb{R}^{R \times R}$$

$$\Sigma_{x_{mfn}} = A_{f}^m \Sigma_{s_{mfn}} A_{f}^m + b_{mfn} \in \mathbb{C}^{F \times I}$$

$$\Omega_{s_{mfn}} = \Sigma_{s_{mfn}}$$

where $\varphi_{r,fn} = v_{j,rf}$ if $r \in R_j$ (i.e., $r$ is a sub-source of $j$).

2) **M-step**: The free parameters that compose the set $\theta$ are here updated in order to increase the quantity $Q$. The update for the spatial parameters is [5]:

$$A_{j,fn} = \left[ \sum_{n} \hat{R}_{x_{mfn}} \right]^{-1} \left[ \sum_{n} \hat{R}_{s_{mfn}} \right] = \hat{R}_{x_{mfn}}^s \left[ \sum_{n} \hat{R}_{s_{mfn}} \right]^{-1} \left[ \sum_{n} \hat{R}_{x_{mfn}} \right]$$

If none of the parameters are shared, the resulting GEM algorithm processes the different mixtures separately, and behaves as in [5]. The sharing of spectral parameters induces a single change in the algorithm routines that occurs during the M-step updates of these shared spectral parameters. Examples of MU are given for free parameters (in green) in (15) and for shared parameters (in red) in (16). They generalize the update (30) in [5] with $\hat{Z}_j = \hat{Z}_{j,fn} \in \mathbb{R}^{F \times N}$ where $\xi_{j,fn} = \frac{1}{R_j} \sum_{r=1}^{R_j} \hat{R}_{x_{mfn}} (r, r)$.

**C. Parameter initialization**

The results of both MU and EM depend on initialization. With respect to blind source separation, reference guided separation provides better initial values for the parameters $W$ and $H$ taking advantage of the provided references. The other parameters, i.e., the transformation matrices $T$, are beforehand roughly estimated (see Section V). For instance, we can use MU to minimize the following criterion:

$$\hat{\theta}_{ref} = \arg \min_{\theta \in \Theta} \sum_{m=2}^{M} \sum_{f,n=1}^{F,N} d_{IS}(X_{f,fn}^m | V_{f,fn})$$

where $\Theta_{ref}$ is the set of $W$ and $H$ parameters that occur in the reference signals. This is especially efficient when there is a single dominant source in each reference signal. At the end of this stage, only the parameters of the mixture to be separated that are not shared with any reference signals remain weakly initialized if no prior information about them is available.

In the experiments, we will distinguish the following successive initialization and algorithmic stages:
MU-free:
\[ W^e_j \leftarrow W^e_j \odot \frac{[V_j^\phi \odot V_m^{-2} \odot X_m][H_j^e]^T}{[V_j^\phi \odot V_m^{-1}][H_j^e]^T} \] (13)

MU-shared:
\[ W^e_j \leftarrow W^e_j \odot \frac{\lambda^m[V_j^\phi \odot V_m^{-2} \odot X_m][H_j^e]^T + \sum_{j'} \lambda^m'[T_{j'^e}^f]T[V_j^\phi \odot V_m^{-2} \odot X_m'][T_{j'^e}^f H_j^e T_{j'^e}^{te}]^T}{\lambda^m[V_j^\phi \odot V_m^{-1}][H_j^e]^T + \sum_{j'} \lambda^m'[T_{j'^e}^f]T[V_j^\phi \odot V_m^{-1}][T_{j'^e}^f H_j^e T_{j'^e}^{te}]^T} \] (14)

EM-free:
\[ W^e_j \leftarrow W^e_j \odot \frac{[V_j^\phi \odot V_j^{-2} \odot \hat{\Xi}_j][H_j^e]^T}{[V_j^\phi \odot V_j^{-1}][H_j^e]^T} \] (15)

EM-shared:
\[ W^e_j \leftarrow W^e_j \odot \frac{\lambda^m R_j[V_j^\phi \odot V_j^{-2} \odot \hat{\Xi}_j][H_j^e]^T + \sum_{j'} \lambda^m R_j'[T_{j'^e}^f]T[V_j^\phi \odot V_j^{-2} \odot \hat{\Xi}_j'][T_{j'^e}^f H_j^e T_{j'^e}^{te}]^T}{\lambda^m R_j[V_j^\phi \odot V_j^{-1}][H_j^e]^T + \sum_{j'} \lambda^m R_j'[T_{j'^e}^f]T[V_j^\phi \odot V_j^{-1}][T_{j'^e}^f H_j^e T_{j'^e}^{te}]^T} \] (16)

- **Init**: for all sources, we define the status (i.e., fixed, free or shared) of their spectral parameters (i.e., \( W, H \) and \( T \)) and we initialize them in the best way according to prior information. The exact initialization is specified below in Sections V and VI depending on the considered scenario.
- **NMF**: the shared and free \( W, H \) of the reference mixtures are updated using MU, i.e., the method described in this Section III-C.
- **Plain-NMF**: we apply the algorithm described in Section III-A to the target mixture only (\( M = 1 \) in (12)).
- **NMPcF**: we apply the algorithm described in Section III-A to all mixtures.
- **GEM**: we apply the algorithm described in Section III-B to all mixtures.

During the experiments, different combinations of these four stages are tried in the above order. In all cases, the final source estimates are obtained using an adaptive Wiener filter \( \hat{s}_{jfn}^m = \Omega_{s^m} \hat{x}_{jfn}^m \) and multiplied by the structured \( A^m_{j} \) to obtain the corresponding spatial images \( y_{j,i,fn} \).

IV. ELEMENTARY SCENARIO WITH PITCH SHIFTING

We first illustrate the method on single-source signals for which the reference is a pitch shifted version of these signals. This elementary example addresses one possible use of the transformation matrices of the excitation part \( T^f \) and \( T^{de} \).

A. Data

We used six guitar source signals of thirty seconds and generated several pitch shifted references for each of them (from one to four semitones). The pitch shifted examples are obtained using GuitarPitchShifter\(^4\).

B. Model and initialization

The source is hereafter numbered as \( j = 1 \) and the reference as \( j = 2 \). In this description, we remove the notion of mixture as only single-source signals are used. Fixed variables are

\(^4\)http://www.guitarpitchshifter.com/matlab.html

in black (\( W^e_1 \)), free variables are in green (\( T^f_{12}, T^{de}_{12} \)) and shared variables are in red (\( H^e_1, W^o_1, H^e_1 \)). The source power spectrum \( V_1 \) is modeled as,

\[ V_1 = W_1^e H_1^e \odot W_1^o H_1^o \] (26)

and the reference power spectrum \( V_2 \) is modeled either as

\[ V_2 = T^{de}_{12} W_1^e H_1^e \odot W_1^o H_1^o \] (27)

or

\[ V_2 = W_1^e T^{de}_{12} H_1^e \odot W_1^o H_1^o. \] (28)

The fixed excitation spectral patterns \( W_1^e \) is a set of harmonic components computed as in [5] (see also Fig. 1). Each component is a harmonic spectrum and two successive components are separated by one semitone. In order to represent the pitch shifting transformation, we investigate two alternative models:

- a dictionary component transformation \( T^{de} \) that will be, in the ideal case, a translation defined by the equation \( y = x + b \), where \( b \) is the amount of shift in semitones,
- a frequency transformation \( T^f \) that will be, in the ideal case, a homothety defined by the equation \( y = \alpha x \), where \( \alpha = 2^{b/12}. \)

These free transformation matrices \( T^f_{12} \) and \( T^{de}_{12} \) are either initialized

- in an informed way, i.e., the matrix elements within a one-tone range of the actual pitch shift are initialized with random values and the other elements are set to zero,
- or entirely with random values.

Examples of estimated transformation matrices are given in Fig. 3. As we work with MU, let us remind that zeros (dark blue in the figures) in the parameters remain unchanged over the iterations. All the elements of the other matrices (\( H^e_1, W^o_1, H^e_1 \)) are initialized with random values.

C. Estimation and results

The estimation of the parameters of the joint model, i.e., equations (26) and (27) or (26) and (28) is done using NMPcF
(see Section III-A). An experiment without deformation in the reference model, i.e., $V_2 = V_1$ is added as comparison, as well as an “oracle” setting, i.e., $V_2 = T_{12}^T W_1^T H_2^2 \odot W_2^2 H_2^2$ or $V_2 = W_1^T T_{12}^T H_2^2 \odot W_2^2 H_2^2$. The oracle setting corresponds to the case when no parameter is shared between the source and the reference power spectrum models (i.e., the models are separately estimated based on the observation of each true source). These two complementary experiments should respectively provide lower and upper bounds to our approach. The results are given in terms of signal-to-noise ratio (SNR) between the magnitude spectra of the true source and reference signals ($\|x_1^T \|_f n$ and $\|x_2^T \|_f n$) and the estimated spectra ($V_1^{[1/2]}$ and $V_2^{[1/2]}$) in Table Ia and in terms of the IS divergence (the terms of the sum in (12)) in Table Ib.

For both SNR and IS divergence, we observe that the distortion is much smaller after deformation modeling than before, for instance 9.1 dB and 7.9 dB SNR compared to 6.5 dB and 4.5 dB (first column in Table Ia). Nevertheless, the distortion remains slightly larger than what can be achieved in the oracle setting (e.g., 10.3 dB and 9.6 dB). These results show the ability of our models to account for pitch shifting and effectively reduce the difference between the source signals and the corresponding models. It can also be noticed that the knowledge of the pitch shift value leads to a slight improvement in terms of SNR for the reference. Such information can be provided by a sound engineer or a musician, for instance.

As the considered pitch shifting is a software effect, the use of $\tilde{T}_{fe}$ is possible even though the guitar is inharmonic. In a different scenario where a given melody would be played by an inharmonic instrument in two different tonalities, the inharmonicity would require a specific inharmonic dictionary and the use of $T_{de}$ instead of $T_{fe}$. Indeed, the partials of a given pitch would not be found by simple shifting of the partials of another pitch and pitched components must be shifted while attack components must not. Such a distinction for each component is not possible with $\tilde{T}_{fe}$.

\section{V. Voice/Music Separation}

In this section, we describe a second use case of the proposed framework for source separation with deformed reference. We target the separation of speech and music from old recorded movies and TV series. Speech and/or music references are used to guide the separation.

After briefly describing the data, we recall how speech and music references are modeled in the proposed framework. We consider two distinct models for the music references depending on whether they are aligned a priori or not. We conduct experiments that compare these two models as well as different initialization procedures. Finally, we investigate the use of several references for a single source with the objective of making the separation more robust.

\subsection{A. Data}

The musical samples and the corresponding references are obtained using the MODIS audio motif discovery software in [34]. This software aims at clustering the segments of a long audio stream (here movies or TV series) that are similar enough according to a threshold. It is based on seeded discovery and template matching [35], but on a more fundamental level the audio segments are compared using a segmental variant of dynamic time warping (DTW) and common features.
As long as it allows the discovery of non-exact repetitions, the
discovered references are distorted compared to the source of
interest (rhythm changes, fade in) and also contain additional
sources (mainly sound effects).

Speech examples are taken from the database in [36] in
which 16 different speakers uttered the same 238 sentences. We
kept 4 musical examples and 4 sentences (two female and
two male speakers) and mixed them at two different voice-
to-music ratios : -6 dB (music as foreground and voice as
background), and 12 dB (voice as foreground and music as
background). Thus the original SDRs are -6 dB and 12 dB
for the voice and 6 dB and -12 dB for the music. These
levels are close to those effectively observed in movies and TV
series. We mixed such examples ourselves in order to obtain
objective measures for the evaluation and to compare our
estimated sources with the original ones using [37]. Combining
these parameters leads to 32 original mixtures $X^3$. For each
mixture to be separated, we have one or more deformed music
reference(s) (other discovered versions of the same music
excerpt), and one or more deformed speech reference(s) (same
sentence uttered by different speakers). The original mixtures
and the references are about eight seconds long, they are
sampled at 16 kHz and are single-channel ($\forall m, I^m = 1$).
Some examples are available online\(^5\).

**B. Tested models**

In the different setups reported here, the speech sources are
numbered as $j = 1 \text{ or } 2$, the music sources as $j = 3 \text{ or } 4$, and
the other sources and background noise as $j = 5 \text{ or } 6$.
*Fixed* variables are in black ($W^1_t, W^3_t, T^3_{34}$). *Free* variables are in
green ($H^1_2, H^3_2, T^1_{12}, T^1_{34}, T^3_{34}, W^5_5, H^5_5, W^6_6, H^6_6$).
*Shared* variables are in red or blue ($W^1_3, H^1_3, H^3_3, W^2_3, H^2_3$).
Note that, in one particular setup, $H^3_5, W^3_3$ and $H^3_3$ are *free*. The
*fixed* matrices $T$ set to identity are removed from the
notations.

1) **Signal to be separated:** The first signal is the mixture
to be separated. It is composed of speech $V_1$, music $V_3$ and
noise $V_5$ :

\[
V^1 = V_1 + V_3 + V_5 \quad (29)
\]

\[
= W^1_t H^1_t \odot W^3_1 H^3_1 + W^5_3 H^5_3 \odot W^3_3 H^3_3 + W^2_5 H^2_5.
\]

2) **Speech reference:** The second mixture is composed of the
speech reference $V_2$ alone :

\[
V^2 = V_2 = W^1_t H^2_2 \odot T^1_{12} W^3_1 H^3_1 T^1_{12} T^3_{34} T^3_{34}
\]

During the NMF/SE and/or GEM stages, $H^1_2$ and $H^2_2$
are estimated separately to model the different intonations and
pitches between the speakers. Conversely, the filter matrices
$W^1_3$ and $H^3_3$ are jointly estimated to model similar phonetic
content, as the two speech signals are composed of the same
phonemes. $T^1_{12}$ models the time alignment between the two
utterances. $T^1_{12}$ is constrained to be diagonal and it models
both the equalization and the speaker’s difference. $T^3_{12}$ is also
used to model the speaker difference and its initialization is
discussed in Section V-F. This model is similar to the one

\[^5\text{http://speech-demos.gforge.inria.fr/source\_separation/tasl2015/index.html}\]
accounts for the fact that the two channels are expected to be time and amplitude aligned and the second column accounts for residual differences.

The NMF stage can then be applied separately on the reference mixtures (30) and (31) (unaligned references), where the shared matrices $(W^6_j, H^6_j, H^6_3, W^3_3, H^3_3)$ and the free parameters $(H^2_3, W_3, H_6)$ are updated whereas matrices $T_{12}^{12}, T_{12}^{12}, T_{12}^{12}, T_{12}^{12}, T_{12}^{12}$, and $T_{12}^{12}$ are not. In the phase aligned music reference case, the parameters $H^2_3, W^6_3, H^6_3, W_6$, and $H_6$ are updated to fit the reference signal that is already phase aligned with the signal to be separated. In both cases, $W_3$ and $H_6$ are set once again to random values before applying NMPcF and/or GEM.

D. Algorithm combination

As a first experiment, we evaluate the effect of NMF initialization in the case of a single non phase aligned music reference and no speech reference. The number of iterations is set to 10 for the NMF, Plain-NMF, and NMPcF stages, and to 100 for the GEM stage that is known to require more iterations. The separation performance results are evaluated in terms of signal-to-distortion ratio (SDR), signal-to-interference ratio (SIR) and signal-to-artifacts ratio (SAR) [37].

The results are summarized in the top part of Table II. The best SDRs are indicated in bold for each column’s part (delimited by double lines). A notable improvement (at least 2.5 dB) is observed when NMF is used beforehand, as compared to using NMPcF alone. The use of NMPcF instead of Plain-NMF then leads to an improvement when the source with a reference (here music) is as background. A similar behavior is observed in the experiment of Section VI.

E. Model comparison for music reference

As a second experiment, we evaluate the effect of phase alignment for the music reference. The results are shown in the bottom part of the Table II. Comparing the configurations chosen for the third and fourth lines with the first line shows that GEM decreases the separation performance if it is used directly after NMF or NMPcF alone. This highlights that the proper functioning of the GEM depends on the two previous steps. The best results are obtained when the signals are phase aligned and both NMF and NMPcF are used before GEM. In that case, the improvements compared to the non phase aligned case are marginal when the music is as foreground, but significant when the music is as background.

As a GEM iteration is in the order of ten times longer than a NMPcF iteration, the relevance of this costly last step can be discussed. Given the marginal improvements when music is in the foreground, additional GEM iterations are not necessary for this voice-to-music ratio. Conversely, when music is in the background, GEM increases the music SDR by 2.4 dB. This result is also greater by 1.3 dB compared to approaches that use multiple references (see Table III).

F. Multiple references for a single source

Experiments on the effect of using reference signals for different sources have been conducted in [27]. Complementary experiments are here conducted on the influence of the number of reference signals per source, i.e., several $j'$ for a single $j$. The number of speech (resp. music) references grows from 0 to 3 (resp. 2). We also evaluate the separation performance without any deformation modeling except time alignment, referred hereafter as Reference-based Wiener Filter (RbWF). Table III gathers all the results.

It can be emphasized that the use of multiple speech references leads to better result, especially when speech is in the background (in the order of 0.5 dB). Conversely, the use of two music references tends to smaller or equal results. This can be explained by the fact that the considered music references contain additional sources supposed to be taken into account by the matrices $W_3$ and $H_6$ in (31), and that leads to a more complicated situation for the algorithm. But for some particular examples, the second music reference leads to better results. Overall, the addition of more than one reference seems to improve separation when the new references carry complementary information.

Reference-based Wiener Filter (RbWF) results are obtained using an adaptive Wiener filter that reconstructs the different sources of the mixture based on the power spectrum ratio between the reference signals. As the references and the mixture are of different length, a time alignment (as previously described) is performed beforehand. These baseline measurements give the reader a better description of the quality of the reference. All the results show a significant difference compared to the case with one reference for each source, while the foreground music source results are comparable. This can be explained by the quality of the music reference and the corresponding alignment pre-processing technique when the music predominates in the mixture signal. However, the proposed approach shows clear benefit when the reference source does not predominate.

VI. COVER GUIDED MUSIC SEPARATION

This last experimental part focuses on the task of professionally-produced music separation guided by covers [24]. A cover song is a replica of an original song with some differences due for instance to artist interpretation, singer/instrument changes, or new song structure. Such covers can be easily found, and they are usually close to the original song making them interesting for separation. As it provides high quality separation, such demixing enables the edition of the song by end-users (e.g., for active listening) or professional users (e.g., for upmixing).

Here we use multitrack recordings of cover songs to guide the separation. Each track is used as a reference for one corresponding source, so the number of tracks is the same as the number of sources to be separated. In [24], the multitrack cover signals are only used to initialize the source parameters $W$ and $H$ (Plain-NMF). Here, these parameters are shared between the source and the reference hence the reference signals are used during the estimation stage too. Deformations are modeled in various ways using the general framework introduced in Section II.
A. Data and settings

In order to compare our results, we used the same data set and settings as in [24]. Both original and cover multitracks are available in order to evaluate the separation. They are also used in the mirror configuration, i.e., considering the original as the reference and vice versa. The mixes are produced by a sound engineer for mono and stereo [24]. Here, we make an exhaustive list of settings that differ from [24] and refer the reader to [24] for other common details.

The 30 second examples are chosen in a different way as in [24] and are typically composed of half of a verse and half of a chorus. The considered tracks of four songs are listed in Table IV. There is no second electric guitar for the song "Walk this Way" as it does not appear in the example that we selected. We use 50 iterations for NMF and NMPcF instead of 500 [24] and 10 iterations for GEM instead of 500 [24]. The number of components D is kept to 50. For the sake of clarity, the single-channel case is first investigated to show the effect of different deformation models, then stereo data are processed using the GEM algorithm presented in Section III-B.

<table>
<thead>
<tr>
<th>Number of speech references</th>
<th>Number of music references</th>
<th>-6 dB voice-to-music ratio</th>
<th>12 dB voice-to-music ratio</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>SDR</td>
<td>SAR</td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td>2.1</td>
<td>5.9</td>
</tr>
<tr>
<td>2</td>
<td>0</td>
<td>2.3</td>
<td>6.1</td>
</tr>
<tr>
<td>3</td>
<td>0</td>
<td>2.8</td>
<td>5.7</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>2.1</td>
<td>2.9</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>4.0</td>
<td>6.0</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>4.9</td>
<td>6.2</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>5.0</td>
<td>6.3</td>
</tr>
<tr>
<td>0</td>
<td>2</td>
<td>1.5</td>
<td>3.0</td>
</tr>
<tr>
<td>1</td>
<td>2</td>
<td>4.1</td>
<td>6.1</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
<td>4.6</td>
<td>6.3</td>
</tr>
<tr>
<td>3</td>
<td>2</td>
<td>4.6</td>
<td>6.2</td>
</tr>
</tbody>
</table>

RbWF RbWF

TABLE III

AVERAGE VOICE/MUSIC SEPARATION PERFORMANCE (dB) FOR DIFFERENT NUMBERS OF SPEECH AND MUSIC REFERENCES. SPEECH REFERENCES ARE UTTERED BY DIFFERENT SPEAKERS THAT HAVE THE SAME GENDER AS IN THE MIXTURE TO BE SEPARATED. THE MUSIC REFERENCE IS NOT PHASE ALIGNED, AND ONLY 10 ITERATIONS OF NMF AND NMPcF ARE USED. THE BEST SDRs ARE INDICATED IN BOLD.

<table>
<thead>
<tr>
<th>Title</th>
<th>Track names</th>
</tr>
</thead>
<tbody>
<tr>
<td>I Will Survive</td>
<td>Bass, Brass, Drums, Electric Guitar, Strings, Vocal.</td>
</tr>
<tr>
<td>Pride and Joy</td>
<td>Bass, Drums, Electric Guitar, Vocal.</td>
</tr>
<tr>
<td>Rocket Man</td>
<td>Bass, Choirs, Drums, Others, Piano, Vocal.</td>
</tr>
<tr>
<td>Walk this Way</td>
<td>Bass, Drums, Electric Guitar, Vocal.</td>
</tr>
</tbody>
</table>

TABLE IV

COVER MULTITRACK DATASET

B. Tested models

In this scenario, the mixture to be separated is an original song. The reference signals are the different tracks of a cover version of this original song. Each reference signal is related to one target source in the original song. The power spectrum of each reference signal is modeled as $V_j = W_j H_j$ and its parameters are initialized using the Init and NMF stages in all the different settings reported in Table V.

Conversely, different settings are considered for the power spectrum model of the sources to be separated: $WH$, $T^f WH$, and $WTd^f H$ (see Table V). The general framework introduced in Section II is then used by inverting $j$ and $j'$ in (7) for instance. This is because the initialization of the $T^f$ and $Td$ matrices is weak and would have disturbed the NMF stage. A complementary discussion on whether the transformation matrices are useful in the reference or the source model and their effect on the resynthesis is given in [25].

C. Initialization

Parameters $W$ and $H$ are randomly initialized before being updated to fit the reference signal. Here, this is done by the so-called Init and NMF steps similarly to what was done in [24]. When used, $T^f$ is initialized as an identity matrix. Along the same lines, we tested several initializations of $Td^f$ starting from the identity matrix and changing the weight of the off-diagonal coefficients. The best initial value was the sum of an identity matrix and a random matrix drawn from a rectified Gaussian distribution. When stereo data and the
GEM algorithm are used, the (full-rank) spatial parameters are initialized beforehand using the reference for each source as in [24].

D. Results

Separation performance is evaluated in terms of signal-to-distortion ratio improvement (SDRI) that is the difference between the output SDR [37] and the input SDR. The input SDR is defined as the power ratio between a source to be estimated and the mixture to be separated. The samples that we selected lead to an input SDR of the same order (8.44 dB instead of -7.60 dB in [24]). The results are summarized in Table V. Some examples are also available online5.

1) Single-channel results: First, we reproduced the experiments in [24] with the differences previously presented. A similar SDRI mean is obtained (8.74 dB instead of 8.98 dB) in the case when the parameters are not shared. Compared to the initialization (10.06 dB), this configuration leads in fact to a decrease of the average SDRI. This can be explained by the high similarity between the covers and the original tracks, as shown by the Reference-based Wiener Filter (RbWF) results in Table V6. Conversely, sharing the parameters during the final estimation guarantees not to draw away too much from the starting point while getting closer to a solution that fits better to the original tracks. In our case, a marginal improvement is observed (10.27 dB average SDRI). This small improvement is not surprising as we considered no deformation between the references and the sources that we estimate.

More appropriate models that use deformations (Tj, Td) have been tried with promising results. Better overall results (10.64 dB) are obtained when using a Td in the source model, whereas Tj slightly decreases the average result. Moreover, the improvement is not uniformly observed, and some sources are more enhanced using Tj instead of Td.

We conduct a final experiment (Best) where, for each source, the best source model (values in bold font) is chosen. The Init + NMF setting is not taken into account in this choice as it would avoid the benefit of the update of the other parameters during the NMPcF. We observe for this selection an overall increase of the performance (10.85 dB), hence showing the potential of using suitable deformation models for specific sources. As we selected the different models based on their optimal results, this last experiment is of course not representative of an unsupervised scenario. However, it is realistic in the considered context where an expert user may be able to select the best model by listening to the results. Further experiments are needed to understand which parameters should be accessible to the user, maybe depending on its level of expertise.

2) Multi-channel results: The stereo setting reported in [24] yields an important improvement in terms of SDRI compared to the mono setting (from 8.98 dB to 10.05 dB). The given explanation suggested that this overall improvement was especially due to a few instruments with specific spatialization (like the guitar).

Here, the model without deformation shows a small improvement (from 10.27 dB to 10.41 dB) when 10 additional GEM iterations are added and similar results are observed for the models with deformation. However, we do not observe as much improvement as in [24] nor differences between spatialized and centered sources. The EM algorithm usually requires more iterations that were not achievable in a reasonable computation time here. Indeed here each reference is considered as a mixture which requires a specific E-step in the EM algorithm presented in Section III-B.

VII. CONCLUSION AND PERSPECTIVES

In this paper, we have presented a general framework for using audio information in order to separate a given mixture. This model is general enough to take different types of audio references into account and to accommodate for their possible deformations in the frequency domain and/or in the temporal domain. After having presented an elementary scenario with pitch shifted example, we have provided extensive experiments on two realistic scenarios: voice and music separation in the context of movie soundtracks, and cover-guided music separation. Our experiments show that the use of reference for a given source improves the general sound quality of the estimated source (from 9 to 15 dB). Different temporal alignment methods appear to be adapted to different situations and signals. Moreover, our experiments show that having at least one reference per source is of prime importance.

One application perspective is the use of user-selected references as well as user-designed models. A more general perspective of this study will be the design of some automatic processes that choose the best configuration of our general model for a given mixture. Along the same idea, the initialization of the time-alignment matrices (DTW) could be enhanced by techniques adapted to the presence of other sources. Our model can also be improved by adding well-chosen constraints on the parameters. For instance, smoothness constraints on the spectral transformation matrices Tj,Td should help the model to derive a more relevant spectral deformation between the target sources and the references. Other algorithmic approaches can also be investigated like stacked NMF (column-wise concatenation of mixture and time-warped references) with other advantages and disadvantages.

ACKNOWLEDGMENT

The authors would like to thank MAIA Studio for their sound engineering expertise and for partly funding this work.

REFERENCES


TABLE V
AVERAGE SDRI (dB) FOR THE SEPARATION OF MUSIC RECORDINGS USING MULTITRACK COVERS AS REFERENCES. VALUES IN BOLD INDICATE THE MODEL CHOSEN FOR EXPERIMENT REFERRED AS BEST.

APPENDIX A
Here we give a detailed derivation of the expected value of the complete-data log-likelihood (18).

\[
Q(\theta, \theta^c) \triangleq \mathbb{E}_{Z|\theta^c}[\log p(Z|\theta)] = \sum_m \lambda^m \mathbb{E}_{X^m|s^m,\theta^c}[\log p(X^m|S^m)]
\]

\[
= \sum_m \lambda^m \mathbb{E}_{X^m|s^m,\theta^c}[\log p(X^m|S^m)] + \sum_m \lambda^m \mathbb{E}_{S^m|\theta^c}[\log p(S^m|\theta)]
\]

\[
= \sum_{m,f,n} \lambda^m \mathbb{E}_{x^m_{f,n}|s^m_{f,n},\theta^c}[\log N_C(x^m_{f,n}|A^m_{f,n} s^m_{f,n}, \Sigma_{b^m_{f,n}})] + \sum_{m,j \in J^m,f,n} \lambda^m \sum_{r=1}^{R_i} \mathbb{E}_{s_{j,r,f,n}|\theta^c}[\log N_C(s_{j,r,f,n}|0, v^m_{j,f,n})]
\]

\[
= -\sum_{m,f,n} \frac{\lambda^m}{2} \text{tr} \left[ R_{\text{x}^m_{f,n}} - A^m_{f,n} R_{x^m_{f,n}} A^m_{f,n}^H - R_{x^m_{f,n}} A^m_{f,n} A^m_{f,n}^H - A^m_{f,n} R_{x^m_{f,n}} A^m_{f,n}^H \right] - \sum_{m,j \in J^m,f,n} \lambda^m R_i d_{TS}(\xi_{j,f,n}|v^m_{j,f,n})
\]


