GENRE SPECIFIC DICTIONARIES FOR HARMONIC/PERCUSSIVE SOURCE SEPARATION

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ABSTRACT

Supervised algorithms for audio source separation often use *a priori* information such as musical scores and learned dictionaries to achieve a decomposition. Audio signals are diverse because of the inconsistency of the musical performance, recording condition and post processing treatment, as a result it is generally impossible to build prior information that is relevant on any audio signal. Current state-of-the-art methods for source separation use large instruments specific data to build dictionaries to take into account these variability. However, the dictionaries are still too generic and they do not characterize completely a target instrument as the large amount of information is not properly compressed. In this article, we prove that musical genre information is suitable to further specialize the dictionaries in order to improve the decomposition results. Our test on a task of harmonic/percussive source separation on a large database proves that the genre is a relevant feature as the drum dictionaries built using genre specific information perform better that an universal drum dictionary.

1. INTRODUCTION

Source separation is a field of research that seeks to separate the components of a recorded audio signal. Such a separation has many applications in music such as up-mixing [9] (spatialization of the sources) or automatic transcription [35] (it is easier to work on single sources). The separation task is difficult due to the complexity and the variability of the music mixtures. Most datasets used for Blind Audio Source Separation (BASS) research are small in size and they do not allow for a thorough comparison of the source separation algorithms. Using a larger database is crucial to benchmark the different algorithms in order to get an insight of their performances rather than to focus on particular case results.

The large variety of audio signals can be classified into various musical genres [34]. Genres are labels created and used by humans for categorizing and describing music.

They have no strict definitions and boundaries but particular genre share certain characteristics typically related to instrumentation, rhythmic structure, and pitch content of the music. This resemblance between two pieces of music has been used as an information to improve chord transcription [22, 26] or downbeat detection [14] algorithms. Genre information can be obtained using annotated labels. When the genre information is not available, it can be retrieved using automatic genre classification algorithms [25, 34].

In the context of BASS, Non-negative Matrix Factorization (NMF) is a widely used method. The goal of NMF is to approximate a data matrix $V \in \mathbb{R}_+^{n \times m}$ as

$$V \approx \tilde{V} = WH \tag{1}$$

with $W \in \mathbb{R}_+^{n \times k}$, $H \in \mathbb{R}_+^{k \times m}$ and where k is the rank of factorization [20]. In audio signal processing, the input data is usually a Time-Frequency (TF) representation such as a STFT or a constant-Q transform spectrogram. Blind source separation is a difficult problem and the plain NMF decomposition does not provide satisfying results. To obtain a satisfying decomposition, it is necessary to exploit various features that make each source distinguishable from one another. Supervised algorithms in the NMF framework exploit training data or prior information in order to guide the decomposition process. For example, information from the scores or from midi signals can be used to initialize the learning process [7]. The downside of these approaches is that they require well organized prior information that is not always available. Another supervised method consists in performing prior training on specific databases. A dictionary matrix W_{train} can be learned from a big database in order to separate instruments [17, 37]. Such methods require minimum tuning from the user. However, the dictionaries must match the target instruments to obtain satisfying performances.

Harmonic/percussive source separation has numerous applications as a preprocessing step for other audio tasks. For example the HPSS algorithm [8] can be also used as a preprocessing step to increase the performance for singing pitch extraction and voice separation [15]. Similarly, beat tracking [6] and drum transcription algorithms [29] are more accurate if the harmonic instruments are not part of the analysed signal.

In this paper, we focus on the task of harmonic/percussive source separation (HPSS) leveraging on the method developed in [10]: an unconstrained NMF decomposes the audio

signal in sparse orthogonal components that are well suited for representing the harmonic component, while the percussive part is represented by a regular nonnegative matrix factorization decomposition. In [11], we have adapted the algorithm using a trained drum dictionary to improve the extraction of the percussive instruments. Within different music pieces of a database, the same instrument can sound differently depending on the recording conditions and post processing treatments. In order to accurately represent an instrument, one can decide to learn a dictionary on a very large database but takes the risk of over-fitting the data. Furthermore when databases cover a wide variety of genres, instrumentation may strongly differ from one piece to another. In order to avoid this problem and to build effective dictionaries, we propose to use genre specific training data.

The main contribution is that we develop a genre specific method to build NMF drum dictionaries that give consistent and robust results on a HPSS task. The *genre specific dictionaries* are able to improve the separation score compared to a *universal dictionary* trained from all available data.

The rest of the paper is organized as follow. Section 2 defines the context of our work, Section 3 presents the proposed algorithm while Section 4 describes the construction of specific dictionaries. Finally Section 5 details the results of the HPSS on 65 audio files and we suggest some conclusions in Section 6.

2. TOWARD GENRE SPECIFIC INFORMATION

2.1 Genre information

Musical genre is one of the most prominent high level music descriptors. Electronic Music Distribution has become more and more popular in recent years and music catalogues never stop to increase (the biggest online services now propose around 1 million tracks). In that context, associating a genre to a musical piece is crucial to help users finding what they are looking for. As mentioned in the introduction, genre information has been used as a cue to improve some content-based music information retrieval algorithms. If an explicit definition of musical genres is not really available [3], musical genre classification can be performed automatically [23].

Source separation has been used extensively in order to help the genre classification process [19, 30] but, as far as we know, it is the first time that the genre information is exploited to guide the decomposition process.

2.2 Methods for dictionary learning

Audio data is largely redundant as it often contains multiple correlated versions of the same physical event (note, drum hits...) [33]. The relevant information is generally of much reduced dimensionality compared to the original data sets hence the idea to exploit this redundancy to reduce the amount of information necessary for the representation of a musical signal.

Many rank reduction methods, such as Single Value Decomposition (K-SVD) [1], Vector Quantization (VQ) [12], Principal Component Analysis (PCA) [16], or Non negative matrix factorization (NMF) [32] are based on the principle that our observations can be described by a sparse subset of atoms taken from a redundant representation. These methods provide a small subset of relevant templates that are later used to guide the extraction of a target instrument.

Building a dictionary using K-SVD has been a successful approach in image processing applications [39]. However this method does not scale well to process large audio signals as the computational is unrealistic. Thus a genre specific dictionary cannot be considered in this framework.

VQ has been mainly used for audio compression [12] and PCA has been used for voice extraction [16]. However these methods do not have been used yet has a preprocessing step to build a dictionary.

Finally, in the NMF framework, some work has been done to perform a decomposition with learned dictionaries. In [13], the dictionary is built using a physical model of the instrument. This method is not adapted to build genre specific data as the model cannot easily take into account the genre information. A second way to build the dictionary is to directly use the Short Time Fourier Transform (STFT) of an instrument signal [37]. This method does not scale well if the training data are large so it is not possible to use it to build genre specific information. Finally, another method using NMF to build a dictionary is to compute a NMF decomposition on a large training set specific to the target source [31]. After the optimization process of the NMF, the W matrix from this decomposition is used as the dictionary matrix W_{train}. This method is difficult to use on pitched instruments (i.e., harmonic instruments) and the dictionary needs to be adapted using linear filtering on the fixed templates [17]. If the fixed dictionaries provide good results for HPSS [11], the results have a high variance because the dictionary are learned on general data that do not take into account the large variability of drum sounds. Finally, the rank of the factorization determines the final size of the dictionary and it can be chosen small enough to obtain a strong compression of the original data. These properties motivated us to build genre specific data using NMF in order to obtain relevant compact dictionaries.

2.3 Genre information for HPSS

Current state-of-the-art unsupervised methods for HPSS such as [28] and [5] cannot be easily adapted to use genre information. In [18] the drum source separation is done using a Non-Negative Matrix Partial Co-Factorization (NM-PCF). The spectrogram of the signal and the drum-only data (obtained from prior learning) are simultaneously decomposed in order to determine common basis vectors that capture the spectral and temporal characteristics of the drum sources. The percussive part of the decomposition is constrained while the harmonic part is completely unconstrained. As a result, the harmonic part tends to decompose a lot of information from the signal and the decomposition is not satisfactory (i.e., the harmonic part contains some percus-

sive instruments). Another drawback of this method is that it does not scale when the training data is large and the computation time is significantly larger compared to other methods.

Next Section gives a short reminded of the approach introduced and detailed in [10, 11] which appears to be a good candidate to test the genre specific dictionaries: they can be easily integrated to the algorithm without increasing the computation time.

3. STRUCTURED PROJECTIVE NMF (SPNMF)

3.1 Principle of the SPNMF

Using a similar model as in our preliminary work [10], let V be the magnitude spectrogram of the input data. The model is then given by

$$V \approx \tilde{V} = V_H + V_P, \qquad (2)$$

with V_P the spectrogram of the percussive part and V_H the spectrogram of the harmonic part. V_H is approximated by the projective NMF decomposition [38] while V_P is decomposed by NMF components which leads to:

$$V \approx \tilde{V} = W_H W_H^T V + W_P H_P. \qquad (3)$$

The data matrix is approximated by an almost orthogonal sparse part that codes the harmonic instruments $V_H = W_H W_H^T V$ and a non constrained NMF part that codes the percussive instruments $V_P = W_P H_P$. As a fully unsupervised SPNMF model does not allow for a satisfying harmonic/percussive source separation [10], we propose here to use a fixed genre specific drum dictionary W_P in the percussive part of the SPNMF.

3.2 Algorithm Optimization

In order to obtain such a decomposition, we can use a measure of fit $D(x|y)$ between the data matrix V and the estimated matrix $\tilde{V}$. $D(x|y)$ is a scalar cost function and in this article, we use the Itakura Saito (IS) divergence. A discussion about the possible use of other divergences can be found in [11].

The SPNMF model gives the optimization problem :

$$\min_{W_H, W_P, H_P \geq 0} D(V | W_H W_H^T V + W_P H_P) \qquad (4)$$

A solution to this problem can be obtained by iterative multiplicative update rules following the same strategy as in [21, 38] which consists in splitting the gradient with respect to (wrt) one variable (here W_H for example) $\nabla_{W_H} D(V|\tilde{V})$ in its positive $[\nabla_{W_H} D(V|\tilde{V})]^+$ and negative parts $[\nabla_{W_H} D(V|\tilde{V})]^-$, if $\otimes$ is the Hadamard product or element-wise product. Using formula from Appendix 7, the SPNMF optimization process is given in Algorithm 1.

3.3 Signal reconstruction

The percussive signal $x_p(t)$ is synthesized using the magnitude percussive spectrogram $X_P = W_P H_P$. To reconstruct the phase of the percussive part, we use a Wiener

Input: $V \in \mathbb{R}_+^{m \times n}$ Output: $W \in \mathbb{R}_+^{m \times k}$,
$W_{train} \in \mathbb{R}_+^{m \times e}$ and $H \in \mathbb{R}_+^{e \times n}$ Initialization;
while $i \leq$ *number of iterations* **do**

$$H_P \leftarrow H_P \otimes \frac{[\nabla_{H_P} D(V|\tilde{V})]^-}{[\nabla_{H_P} D(V|\tilde{V})]^+}$$

$$W_H \leftarrow W_H \otimes \frac{[\nabla_{W_H} D(V|\tilde{V})]^-}{[\nabla_{W_H} D(V|\tilde{V})]^+}$$

$i = i + 1$

end
$X_P = W_{train} H_P$ and $X_H = W_H W_H^T V$

Algorithm 1: SPNMF with a fixed trained drum dictionary matrix.

filter [24] to create a percussive mask as:

$$\mathcal{M}_P = \frac{X_P^2}{X_M^2 + X_P^2} \qquad (5)$$

To retrieve the percussive signal as,

$$x_p(t) = InverseSTFT(\mathcal{M}_P \otimes X). \qquad (6)$$

Where X is the complex spectrogram of the mixture. We use a similar process for the harmonic part.

4. CONSTRUCTION OF THE DICTIONARY

In this Section, we present in Section 4.1 the test conducted on the SiSEC 2010 database [2] in order to find the optimal size to build the genre specific dictionaries. In Section 4.2 we describe the training and the evaluation database. Finally, in Section 4.3, we detail the protocol to build the genre specific dictionaries.

4.1 Optimal size for the dictionary

The first step to build a NMF drum dictionary is to select the rank of factorization. We run the optimization tests on the public SiSec database from [2] to avoid overtraining. The dataset is composed of 4 polyphonic real-world music excerpts and each music signal contains percussive, harmonic instruments and vocals. The duration of the four recording is ranging from 14 to 24 s. Following the same protocol as in [5], we will not consider the vocal part and we will build the mixture signals from the percussive and harmonic instruments only. The signals are sampled at 44.1 kHz. We compute the STFT with a 2048 sample long Hann window with a 50% overlap. Furthermore, the rank of the factorization of the harmonic part for the SPNMF algortihm is $k = 100$ as in [11].

The drum signal used for the training comes from the database ENST-Drums [27] and the signal is around 10 min long. We used 30 files were the drummer is playing a *drum phrase*. We compute an NMF decomposition with different rank of factorization ($k = 12$, $k = 50$, $k = 100$, $k = 200$, $k = 300$, $k = 500$, $k = 1000$ and $k = 2000$) on the drum signal alone to obtain 8 drum dictionaries.

The dictionaries are then used to perform a HPSS on the four songs of the SiSEC database using the SPNMF algorithm. The results are compared by means of the Signal-

to-Distortion Ratio (SDR), the Signal-to-Interference Ratio (SIR) and the Signal-to-Artifact Ratio (SAR) of each of the separated sources using the BSS Eval toolbox provided in [36].
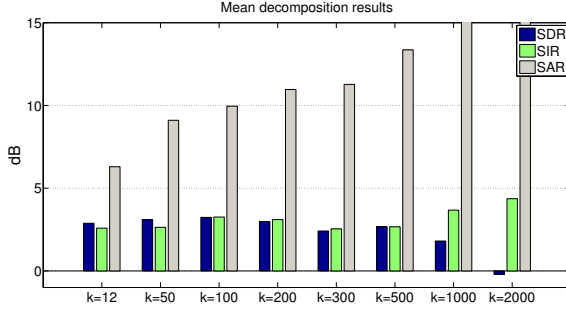
Figure 1: Influence of k on the S(D/I/A)R on the SiSEC database.

The results on Figure 1 show that the optimal value for the SDR and SIR is reached for $k = 100$, then the SDR decreases for $k \geqslant 200$. The high values of SAR (for $k \geqslant 500$) are explained by the fact that the separation process is not satisfying. The harmonic signal provided by the algorithm is composed of most of the original signal therefore the SAR is very high but the decomposition quality is poor. For the rest of the article, the size of the drum dictionaries will be $k = 100$.

4.2 Training and evaluation database

The tests was run on the dataset Medley-dB [4] composed of polyphonic real-world music excerpts. It has 122 music signals and 85 of them contain percussive instruments, harmonic instruments and vocals. The signals that do not contain a percussive part are not part of the evaluation. The genres are, *Classical* (8 songs), *Singer/Songwriter* (17 songs), *Pop* (10 songs), *Rock* (20 songs), *Jazz* (11 songs), *Electronic/Fusion* (13 songs) and *World/Folk* (6 songs). Because the notion of genre is quite subjective (see Section 2), the medley-dB database uses general genre labels. These labels should not be considered to be precise genre labels. There are many instances where a song could have fallen in multiple genres, and the choices were made so that each genre would be as acoustically homogeneous as possible. As we are only working with the instrumental part of the song, the *Pop* label (for example) are similar to the *Singer/Songwriter*.

The training dataset is built using 3 songs of each genre, excluding the training set. To compare the genre specific dictionaries, we build a non specific/universal dictionary using half of one song of each genre. Finally, a last dictionary is built using around 10 min of pure drum signals from the ENST-Drums database as in Section 4.1. The files from medley-dB selected for the training are given in Table 1.

Genre	Artist Song
Classical	JoelHelander Definition
	MatthewEntwistle AnEveningWithOliver
	MusicDelta Beethoven
Electronic/Fusion	EthanHein 1930sSynthAndUprightBass
	TablaBreakbeatScience Animoog
	TablaBreakbeatScience Scorpio
Jazz	CroqueMadame Oil
	MusicDelta BebopJazz
	MusicDelta ModalJazz
Pop	DreamersOfTheGhetto HeavyLove
	NightPanther Fire
	StrandOfOaks Spacestation
Rock	BigTroubles Phantom
	Meaxic TakeAStep
	PurlingHiss Lolita
Singer/Songwriter	AimeeNorwich Child
	ClaraBerryAndWooldog Boys
	InvisibleFamiliars DisturbingWildlife
World/Folk	AimeeNorwich Flying
	KarimDouaidy Hopscotch
	MusicDelta ChineseYaoZu
Non specific	JoelHelander Definition
	TablaBreakbeatScience Animoog
	MusicDelta BebopJazz
	DreamersOfTheGhetto HeavyLove
	BigTroubles Phantom
	AimeeNorwich Flying
	MusicDelta ChineseYaoZu

Table 1: Song selected for the training database.

4.3 Genre specific dictionaries

The NMF model is given by (1). If V is the power spectrum of a drum signal, The matrix W is a *dictionary* or a set of *patterns* that codes the frequency information of the drum. Here we build genre specific drum dictionaries using the medley-dB database.

With the results from Section 4.1 the dictionaries are built as follows: for every genre specific subset of the training database, we perform a NMF on the drum signals with $k = 100$, the W matrices of the NMF are then used in the SPNMF algorithm as the W_P matrix (see Algorithm 1).

5. RESULTS

In this Section, we present the results of the SPNMF with the genre specific dictionaries on the evaluation database from Medley-dB.

5.1 Comparison of the dictionaries

We perform a HPSS on the audio files using the SPNMF with the 9 dictionaries build in Section 4.3. The results on each song are then sorted by genres and the average results are displayed using box-plots. Each box-plot is made up of a central line indicating the median of the data, upper and lower box edges indicating the 1st and 3rd quartiles while the whiskers indicate the minimum and maximum values.

The Figures 2, 3 and 4 show the SDR, SAR and SIR results for all the dictionaries on the *Pop* subsection, giving an overall idea of the performance of the dictionaries inside a specific sub-database. The *Pop* dictionary leads to

the highest SDR and SIR. The non specific dictionaries are not performing as well as the *Pop* dictionary. On this sub-database, the genre specific data gives relevant information to the algorithm. As stated in Section 4.2, some genres are similar to other, explaining why the *Rock* and the *Singer* dictionaries are also providing good results. An interesting result is that compared to the non specific dictionary, the *Pop* dictionary has a lower variance. Genre information allows for a higher robustness to the variety of the songs within the same genre.

The dictionary built on the ENST-drums is giving very similar results to the universal dictionary built on the Medley-dB database. For the sake of concision we only display the results using the universal dictionary from Medley-dB.
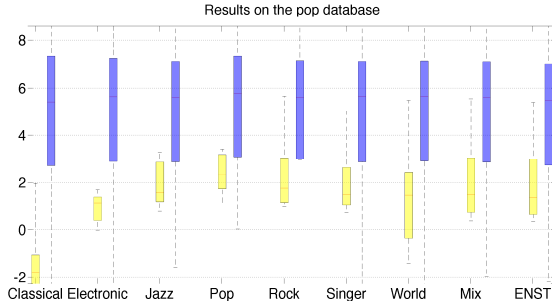
Figure 2: Percussive (left bar)/Harmonic (right bar) SDR results on the *Pop* sub-database using the SPNMF with the 9 dictionaries.
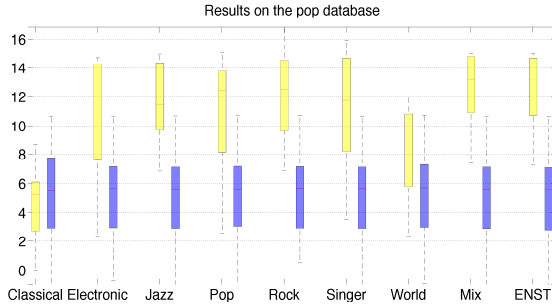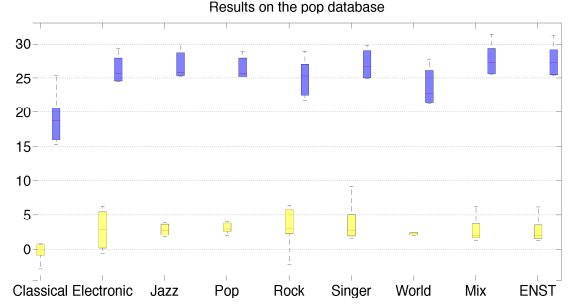

Figure 3: Percussive (left bar)/Harmonic (right bar) SIR results on the *Pop* sub-database using the SPNMF with the 9 dictionaries.

On Table 2, we display the mean separation score for all the genre specific dictionaries compared to the non specific dictionary. The genre specific dictionaries outperform the universal dictionary by a considerable margin on 5 of the 7 genres. The results are discussed in next Section.

5.2 Discussion

On the database *Singer/Songwriter*, *Pop*, *Rock*, *Jazz* and *World/Folk*, the genre specific dictionaries outperform the universal dictionary.

The information from the music of the same genre is not altered by the NMF compression and the drum templates are closer to the target drum. The databases *Clas-*

Figure 4: Percussive (left bar)/Harmonic (right bar) SAR results on the *Pop* sub-database using the SPNMF with the 9 dictionaries.

sical and *Electronic/Fusion* are composed of songs where the drum is only playing for a few moments. Similarly on some songs of the *Electronic/Fusion* database, the electronic drum reproduces the same pattern during the whole song making the drum part very redundant thus the drum dictionary does not contain a sufficient amount of information to outperform the universal dictionary. Because of these two factors, the genre specific dictionaries are not performing correctly.

Overall the harmonic separation is giving much better results than the percussive extraction. The fixed dictionaries are creating artefact as the percussive templates do not correspond exactly to the target drum signal. A possible way to alleviate this problem is to adapt the dictionaries but this requires the use of hyper parameters and that is not the philosophy of this work [10].

6. CONCLUSION

Using genre specific information in order to build more relevant drum dictionaries is a powerful method to improve the HPSS. The dictionaries still have an imprint of the genre after the NMF decomposition and the additional information is properly used by the SPNMF to improve the source separation quality. This is a first step in order to produce dictionaries capable of extracting a wide variety of audio signal.

Future work will be dedicated into building a blind method to select the genre specific dictionary in order to perform the same technique on database where the genre information is not available.

7. APPENDIX: SPNMF WITH THE IS DIVERGENCE

The Itakura Saito divergence gives us the problem,

$$\min_{W_1,W_2,H_2 \geq 0} \frac{V}{\tilde{V}} - log(\frac{V}{\tilde{V}}) + 1.$$

The gradient wrt W_1 gives

$$[\nabla_{W_1} D(V|\tilde{V})]_{i,j}^- = (ZV^TW_1)_{i,j} + (VZ^TW_1)_{i,j},$$

Genre	Classical	Electronic/Fusion	Jazz	Pop	Rock	Singer/Songwriter	World/Folk
Percussive separation							
Genre specific (dB)							
SDR	-1.6	-0.6	**0.4**	**2.5**	**-0.2**	**0.6**	**0.4**
SIR	8.2	15.2	**9.6**	12.3	**19.8**	11.5	**6.1**
SAR	5.9	0.3	**2.1**	**3.4**	0.3	**4.5**	**16.3**
Non specific (dB)							
SDR	**-0.0**	**-0.3**	-0.7	2.0	-2.2	-0.0	-3.6
SIR	**11.3**	**17.0**	9.6	**12.6**	18.3	**13.0**	2.8
SAR	**8.1**	**0.4**	0.9	2.7	**2.3**	1.8	12.1
Harmonic Separation							
Genre specific (dB)							
SDR	**7.5**	**1.6**	**13.0**	**5.1**	**2.1**	7.2	**4.9**
SIR	**10.6**	**1.8**	**13.3**	**5.0**	2.2	**11.5**	**13.5**
SAR	18.2	23.5	28.5	24.5	**36.0**	28.5	**22.7**
Non specific (dB)							
SDR	6.0	1.3	12.7	4.8	1.9	**7.5**	4.6
SIR	7.1	1.4	12.8	4.9	**2.9**	7.5	13.3
SAR	**27.2**	**27.7**	**29.9**	**26.2**	34.3	**31.9**	21.6

Table 2: Average SDR, SIR and SAR results on the Medley-dB database.

with $Z_{i,j} = (\frac{V}{W_1 W_1^T V + W_2 H_2})_{i,j}$. The positive part of the gradient is

$$[\nabla_{W_1} D(V|\tilde{V})]_{i,j}^+ = (\phi V^T W_1)_{i,j} + (V \phi^T W_1)_{i,j},$$

with

$$\phi_{i,j} = (\frac{I}{W_1 W_1^T V + W_2 H_2})_{i,j}.$$

and $I \in \mathbb{R}^{f \times t}; \forall i,j \quad I_{i,j} = 1$.

Similarly, the gradient wrt W_2 gives

$$[\nabla_{W_2} D(V|\tilde{V})]^- = V H_2^T$$

and

$$[\nabla_{W_2} D(V|\tilde{V})]^+ = W_1 W_1^T V H_2^T + W_2 H_2 H_2^T.$$

Finally, the gradient wrt H_2 gives

$$[\nabla_{H_2} D(V|\tilde{V})]^- = W_2^T V$$

and

$$[\nabla_{H_2} D(V|\tilde{V})]^+ = 2 W_2^T W_1 W_1^T V + W_2^T W_2 H_2.$$

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