

#### High resolution spectral analysis and nonnegative decompositions applied to music signal processing

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### Music representations



Spectrogram of "Au clair de la lune"



Musical score











# **副務憲統** Low-rank approximations





## High Resolution (HR) spectral analysis







# Image: High Resolution (HR) spectral analysis





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# 副選擇的 High Resolution (HR) spectral analysis



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## Matrix Factorization (NMF)



Musical score



Spectrogram





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# Nonnegative Matrix Factorization (NMF)





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#### Applications of High Resolution methods

- Spectral analysis (modal analysis, spectroscopy)
- Array processing (beamforming, direction of arrival (DOA) estimation)
- Digital communications (channel identification)
- Applications of NMF
  - Image analysis (face recognition)
  - Text mining, spectroscopy, finance, etc.
- Applications to audio signal processing
  - Source separation, audio coding
  - Pitch and tempo estimation, automatic transcription





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#### Part I

#### **High Resolution spectral analysis**



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• Real-valued model:  $s(t) = \sum_{k=1}^{r} a_k e^{-\delta_k t} \cos(2\pi \nu_k t + \phi_k)$ 

- $a_k \in \mathbb{R}^*_+$  and  $\phi_k \in ]-\pi,\pi]$  are the **amplitude** and **phase**
- $\delta_k \in \mathbb{R}$  and  $\nu_k \in ]-\frac{1}{2}, \frac{1}{2}]$  are the damping factor and frequency
- Complex-valued model:  $s(t) = \sum_{k=1}^{r} \alpha_k z_k^{t}$ 
  - $\alpha_k = a_k e^{i\phi_k} \in \mathbb{C}^*$  is a complex amplitude
  - $z_k = e^{-\delta_k + i2\pi\nu_k} \in \mathbb{C}^*$  is a complex pole
- Noisy model: x(t) = s(t) + b(t) (b(t) is a white Gaussian noise)
- Model estimation
  - Data vector:  $\mathbf{s}(t) = [\mathbf{s}(t), ..., \mathbf{s}(t+n-1)]^T$  with n > r
  - Fourier analysis: spectral resolution of the order of  $\frac{1}{n}$
  - Subspace analysis: high spectral resolution



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Model estimation

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- Compute a "correlation" matrix

$$m{\mathcal{C}}_{xx}(t) = \sum_{ au=0}^t \gamma_ au \, m{x}(t- au) \, m{x}(t- au)^H$$





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### Ime-frequency analysis



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[5] Roland Badeau, Gaël Richard, and Bertrand David. "Fast adaptive ESPRIT algorithm". In Proc. of IEEE Workshop on Statistical Signal Processing (SSP), Bordeaux, France, July 2005.

[6] Bertrand David, Roland Badeau, and Gaël Richard. "HRHATRAC Algorithm for Spectral Line Tracking of Musical Signals". In Proc. of IEEE ICASSP, volume 3, pages 45-48, Toulouse, France, May 2006.

[7] Bertrand David and Roland Badeau. "Fast sequential LS estimation for sinusoidal modeling and decomposition of audio signals". In *Proc. of IEEE WASPAA*, pages 211-214, New Paltz, New York, USA, October 2007.

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- 1)  $\boldsymbol{C}_{xy}(t) = \boldsymbol{C}_{xx} \boldsymbol{W}_r(t-1)$  (compression of  $\boldsymbol{C}_{xx}$ )
- 2)  $\boldsymbol{W}_r(t) \boldsymbol{R}(t) = \boldsymbol{C}_{xy}(t)$  (orthonormalisation of  $\boldsymbol{C}_{xy}(t)$ )
  - $\operatorname{Span}(W_r(t))$  exponentially converges to the signal subspace
  - If 2) is an orthogonal-triangular (QR) decomposition, W<sub>r</sub>(t) converges to the r principal eigenvectors of C<sub>xx</sub>
- Signal subspace tracking if  $C_{xx}(t)$  is time-varying
- Fast algorithm [Strobach, 1996] (complexity of nr<sup>2</sup> instead of n<sup>2</sup>r)



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- FAPI algorithm [1] (complexity of 3*nr* instead of *nr*<sup>2</sup>)
  - reaches the complexity lower bound (3nr)
  - converges faster than PAST and its variants
  - guarantees the orthonormality of W<sub>r</sub>(t) and the numerical stability





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# Applications of High Resolution analysis

#### Analysis / Synthesis

- High resolution time-frequency representation
- Analysis of the sympathetic string modes in a concert harp
- Audio coding
- Automatic transcription
  - Pitch estimation of piano notes
  - Musical tempo estimation

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Other applications

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- Channel estimation in digital communications
- Adaptive multilinear SVD for structured tensors





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### **國務憲統** Decomposition of a piano sound



[1] Roland Badeau and Bertrand David. "Adaptive subspace methods for high resolution analysis of music signals". In Acoustics'08, Paris, France, July 2008.

[2] Bertrand David, Gaël Richard, and Roland Badeau. "An EDS modelling tool for tracking and modifying musical signals". In Proc. of Stockholm Music Acoustics Conference (SMAC), volume 2, pages 715-718, Stockholm, Sweden, August 2003.
[3] Roland Badeau, Rémy Boyer, and Bertrand David. "EDS parametric modeling and tracking of audio signals". In Proc. of the 5th International Conference on Digital Audio Effects (DAFX), pages 139-144, Hamburg, Germany, September 2002.



# Decomposition of a violin sound



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# Sinusoids and noise separation

Principle: projection onto the signal or the noise subspace [1,2]

Instrument	Original	Sinusoids	Noise	
Piano				
Guitar				
Violin				
Flute				
Saxophone				
Bell	<b>W</b>	4		

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### Borum separation and beat estimation

Drum source separation [1]

- Original (Aerosmith):
- Separated drums:
- Remix more drums:
- Remix less drums:
- Beat tracking [2]
  - Pink Floyd:
  - Brad Mehldau:

[1] Olivier Gillet and Gaël Richard. Transcription and separation of drum signals from polyphonic music. *IEEE Transactions on Audio, Speech, and Language Processing*, 16(3): 529-540, March 2008.

[2] Miguel Alonso Arevalo, Roland Badeau, Bertrand David, and Gaël Richard. "Musical tempo estimation using noise subspace projections". In Proc. of IEEE WASPAA, pages 95-98, New Paltz, New York, USA, October 2003.



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  - Brad Mehldau: 🤏

[1] Olivier Gillet and Gaël Richard. Transcription and separation of drum signals from polyphonic music. *IEEE Transactions on Audio, Speech, and Language Processing*, 16(3): 529-540, March 2008.

[2] Miguel Alonso Arevalo, Roland Badeau, Bertrand David, and Gaël Richard. "Musical tempo estimation using noise subspace projections". In *Proc. of IEEE WASPAA*, pages 95-98, New Paltz, New York, USA, October 2003.



# Sympathetic string modes in a concert harp

Modelling sympathetic string modes in a concert harp [1]



[1] Jean-Loïc Le Carrou, François Gautier, and Roland Badeau. "Sympathetic string modes in the concert harp". Acta Acustica united with Acustica, 95(4): 744-752, July/August 2009.





Parametric coder based on the ESM model [1]



#### Joint scalar quantisation with entropy constraint [2,3]

[1] Olivier Derrien, Gaël Richard, and Roland Badeau. "Damped sinusoids and subspace based approach for lossy audio coding". In Acoustics'08, Paris, France, July 2008.

[2] Olivier Derrien, Roland Badeau, and Gaël Richard. "Entropy-constrained quantization of exponentially damped sinusoids parameters". In *Proc. of IEEE ICASSP*, Prague, Czech Republic, May 2011.

[3] Olivier Derrien, Roland Badeau, and Gaël Richard. "Calculation of an entropy-constrained quantizer for exponentially damped sinudoids parameters". Technical report, Laboratoire de Mécanique et d'Acoustique, Marseille, France, June 2010.





Original sound: 🐠					
MDCT		ESM			
9 bits/spl		8.9 bits/spl			
8 bits/spl					
7 bits/spl		6.8 bits/spl			
6 bits/spl		6.4 bits/spl			
5 bits/spl		4.7 bits/spl			
4 bits/spl		4.4 bits/spl			
3 bits/spl		3.2 bits/spl			
2 bits/spl		2.1 bits/spl			





Paris

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Image: A matrix

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#### Part II

#### Nonnegative decompositions



Wednesday, February 13, 2013



# Nonnegative Matrix Factorization (NMF)



Musical score



Spectrogram V





**Roland Badeau** 

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C4DM Seminar

# Nonnegative Matrix Factorization (NMF)



Wednesday, February 13, 2013

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### **\mathbf{M} \mathbf{M}** $\mathbf{M}$ $\mathbf{\beta}$ -divergence and multiplicative rules

Minimisation of the criterion  $D(\mathbf{V}|\mathbf{WH}) = \sum_{n=1}^{N} \sum_{f=1}^{F} d(v_{fn}|\hat{v}_{fn})$   $\beta$ -divergence [Eguchi and Kano, 2001]:  $d_{\beta}(a|b) = \frac{1}{\beta(\beta-1)} (a^{\beta} + (\beta-1)b^{\beta} - \beta ab^{\beta-1})$ 

- $\beta = 2$  corresponds to **Euclidean distance** (EUC),
- $\beta = 1$  corresponds to Kullback-Leibler divergence (KL),
- $\beta = 0$  corresponds to **Itakura-Saito divergence** (IS),
- $d_{\beta}(a|b)$  is convex w.r.t *b* if and only if  $\beta \in [1, 2]$
- Multiplicative update rules [Kompass, 2007]:

 $\begin{cases} W \leftarrow W \otimes \frac{(V \otimes (WH)^{\beta-2})H^{T}}{(WH)^{\beta-1}H^{T}} \\ H \leftarrow H \otimes \frac{W^{T}(V \otimes (WH)^{\beta-2})}{W^{T}(WH)^{\beta-1}} \\ \bullet D(V|WH) \text{ is non-increasing if and only if } \beta \in [1, 2]. \end{cases}$   $Page 27/37 \qquad \text{C4DM Seminar} \qquad \text{Roland Badeau}$ 



### **\mathbf{M} \mathbf{M}** $\mathbf{\beta}$ -divergence and multiplicative rules

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C4DM Seminar

$$\begin{cases} \boldsymbol{W} \leftarrow \boldsymbol{W} \otimes \frac{(\boldsymbol{V} \otimes (\boldsymbol{WH})^{\beta-2})\boldsymbol{H}^{T}}{(\boldsymbol{WH})^{\beta-1}\boldsymbol{H}^{T}} \\ \boldsymbol{H} \leftarrow \boldsymbol{H} \otimes \frac{\boldsymbol{W}^{T}(\boldsymbol{V} \otimes (\boldsymbol{WH})^{\beta-2})}{\boldsymbol{W}^{T}(\boldsymbol{WH})^{\beta-1}} \\ \boldsymbol{\bullet} D(\boldsymbol{V}|\boldsymbol{WH}) \text{ is non-increasing if and only if } \beta \in [1, 2]. \end{cases}$$

Introduction of an exponentiation step η into NMF multiplicative update rules designed for minimizing the β-divergence [1]:

Monotonic decrease of the criterion if  $\beta \in [1, 2]$  and  $\eta \in ]0, 1]$ 

- Exponential or asymptotic stability of both rules (1) and (2) if  $\eta \in ]0, \eta^*[$ , where  $\forall \beta \in \mathbb{R}, \eta^* \in ]0, 2]$  and if  $\beta \in [1, 2], \eta^* = 2$
- Instability if  $\eta \notin [0, 2] \ \forall \beta \in \mathbb{R}$
- Step η permits to control the convergence rate



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# Stability of multiplicative update rules

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## Strategies for initialising the algorithm [1]

- Failure of algorithms from automatic classification
- "Simulated cooling" algorithm for IS-NMF [2]
  - Parameter  $\beta$  becomes a function of the iteration index *p*:

 The best transcription is not obtained by finding the lowest minimum of the criterion, but rather by constraining the decomposition

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# Harmonicity and spectral smoothness

Model [1]:  $\hat{v}_{fn} = \sum_{k=1}^{K} w_{fk}(\theta) h_{kn}$  where  $w_{fk}(\theta) = \sum_{m=1}^{M} e_{mk} P_{km}(f)$ 

- $P_{km}(f)$  is a predefined harmonic spectral pattern
- $\bullet$  *e<sub>mk</sub>* and *h<sub>kn</sub>* are estimated by means of a multiplicative algorithm

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## • MAP estimator: $C(\Theta) = L(\Theta) + \log(p(\Theta))$ où $\Theta = \{e_{mk}, h_{kn}\}$

Markov chain structured a priori distribution:

$$p(\mathbf{H}) = \prod_{k=1}^{K} p(h_{k1}) \prod_{n=2}^{N} p(h_{kn}|h_{k(n-1)})$$

where  $p(h_{kn}|h_{k(n-1)})$  follows an inverse-Gamma distribution

SAGE algorithm [1] and multiplicative update rules [2] with  $\eta = 0.5$ 

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[2] Nancy Bertin, Roland Badeau, and Emmanuel Vincent. "Fast Bayesian NMF algorithms enforcing harmonicity and temporal continuity in polyphonic music transcription". In Proc. of IEEE WASPAA, pages 29-32, New York, USA, October 2009.



# **副務憲統** Temporal smoothness

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# MMF-based automatic transcription



- Original signal (Liszt):
- Transcribed signal:

 Nancy Bertin, Roland Badeau, and Gaël Richard. "Blind signal decompositions for automatic transcription of polyphonic music: NMF and K-SVD on the benchmark". In *Proc. of IEEE ICASSP*, volume 1, pages 65-68, Honolulu, Hawaii, USA, April 2007.
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# 



Jew's harp signal decomposed with an ARMA of order (1,1)

[1] Romain Hennequin, Roland Badeau, and Bertrand David. "NMF with time-frequency activations to model non-stationary audio events". *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 19, no. 4, pp. 744-753, May 2011.



# Fundamental frequencies variations

Model [1]: 
$$\hat{v}_{fn} = \sum_{k=1}^{K} w_{fk}(\nu_{kn}^0) h_{kn}$$
 where  $w_{fk}(\nu_{kn}^0) = \sum_{h=1}^{n_h(\nu_{kn}^0)} a_h g(\nu_f - h\nu_{kn}^0)$ 





[1] Romain Hennequin, Roland Badeau, and Bertrand David. "Time-dependent parametric and harmonic templates in nonnegative matrix factorization". In *Proc. of DAFx*, Graz, Austria, September 2010.



# Score-based informed source separation

Algorithm [1]



#### Round Midnight (Thelonious Monk):

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- Nonnegative decompositions enforcing harmonicity and smoothness
- Applications to audio and music signals
  - Source separation, audio coding,
  - Pitch and tempo estimation, automatic transcription
- Outlooks
  - Is it possible to merge HR methods and NMF in some way?
  - .. to be continued in an upcoming seminar (March 6)





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