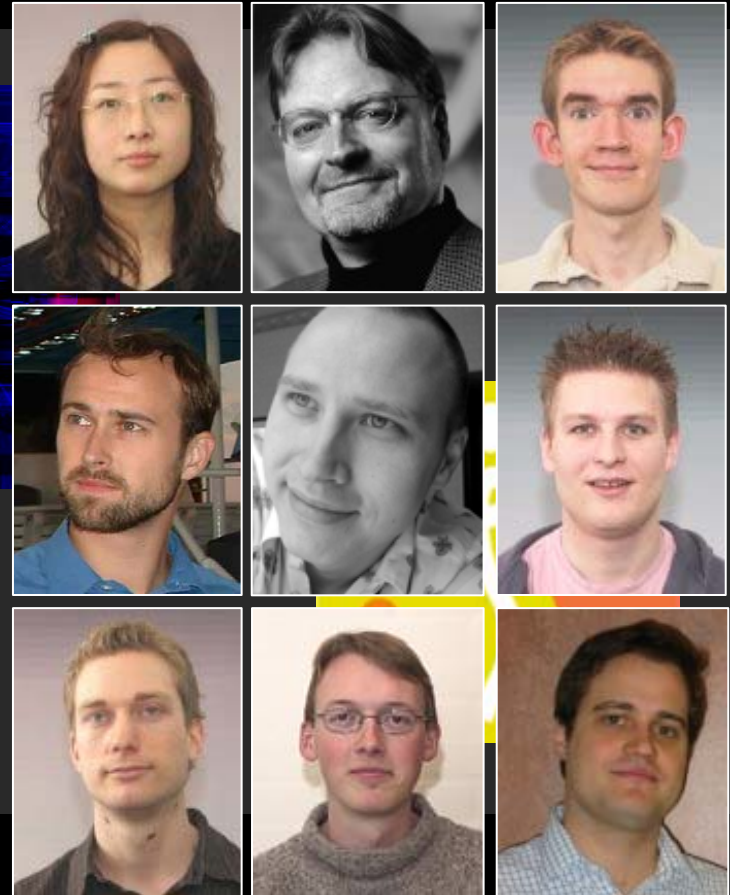
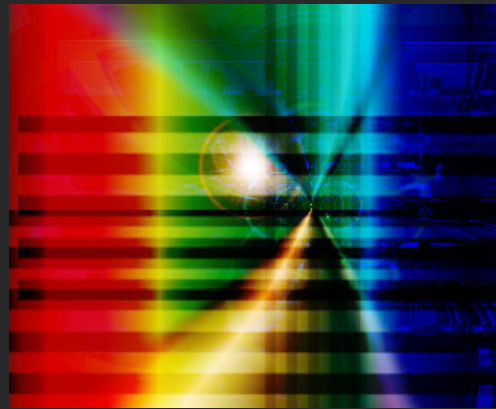




New applications of learning machines

Jan Larsen

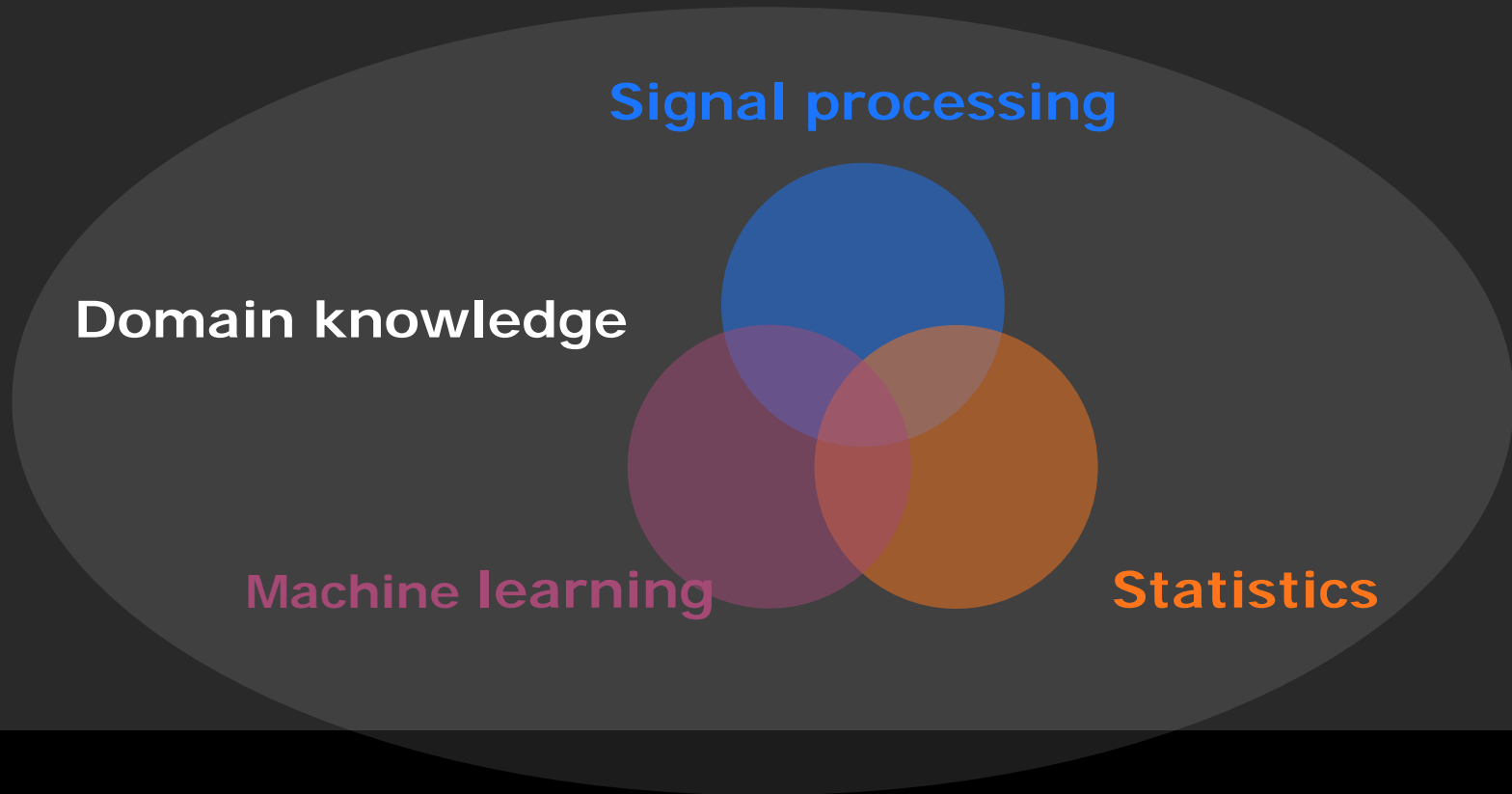


 isp.imm.dtu.dk

 www.intelligentsound.org



Cross-disciplinary research





Informatics and Mathematical Modelling, DTU

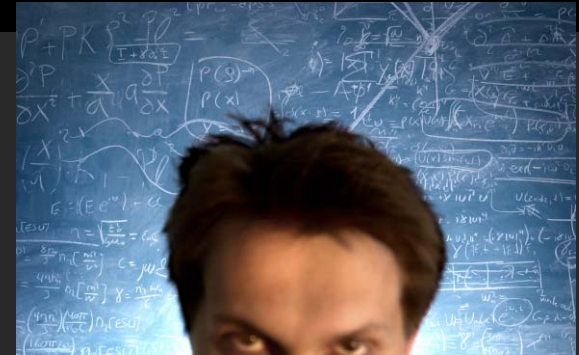


image processing and computer graphics

intelligent signal processing

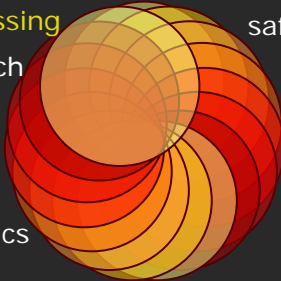
operations research

numerical analysis

geoinformatics

mathematical statistics

mathematical physics



safe and secure IT systems

languages and verification

system on-chips

ontologies and databases

design methodologies

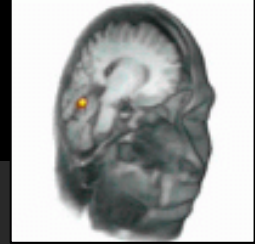
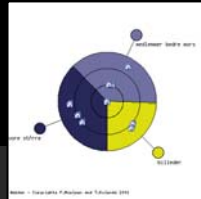
embedded/distributed systems

2003 figures

- 84 faculty members
- 28 administrative staff members
- 60 Ph.D. students
- 90 M.Sc. students annually
- 4000 students follow an IMM course annually



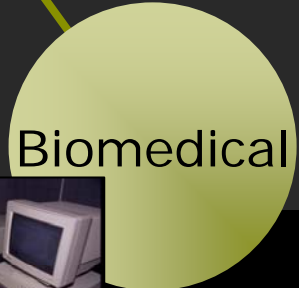
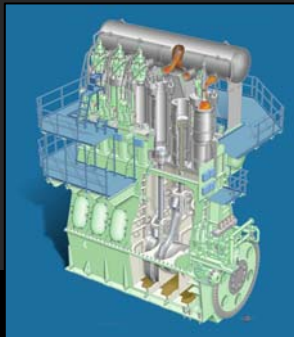
ISP Group



from processing to understanding
extraction of meaningful
information by learning



faculty
• 6+1 postdocs
• 20 Ph.D. students
• 10 M.Sc. students



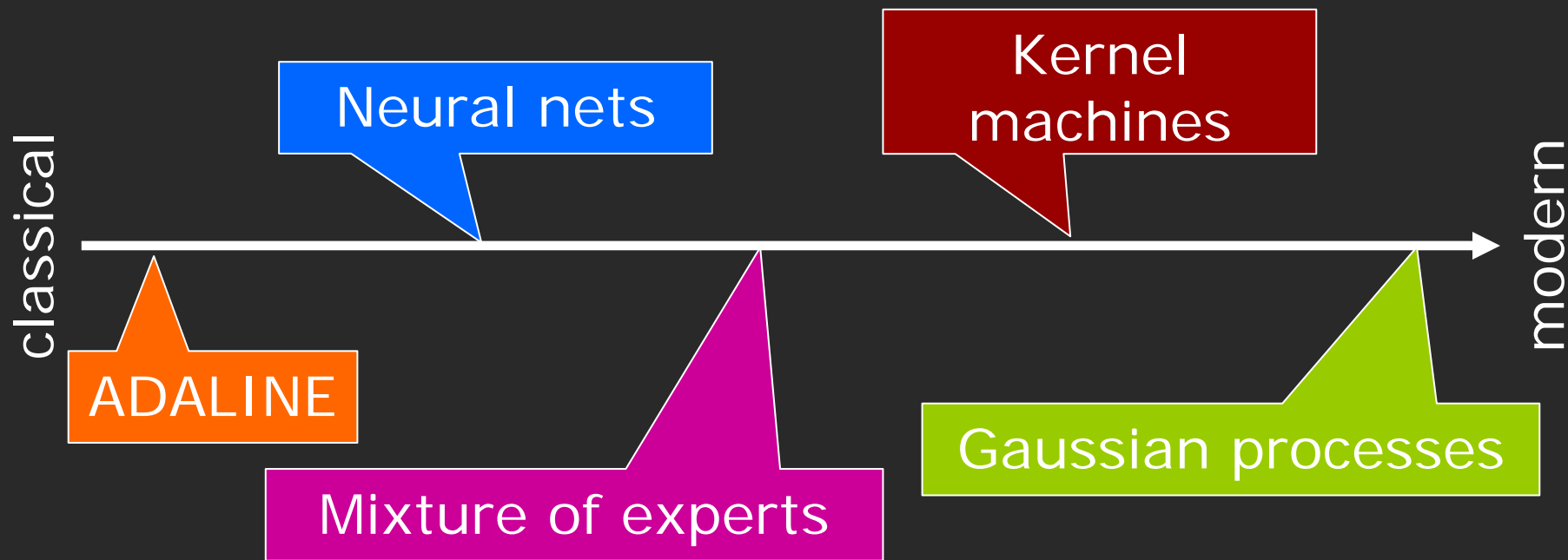


The potential of learning machines

- Most real world problems are too complex to be handled by classical physical models
- In most real world situations there is access to data describing properties of the problem
- Learning machines can offer
 - Learning of optimal prediction/decision/action
 - Adaptation to the usage environment
 - New insights into the problem and suggestions for improvement



A short history of learning machines





Issues in machine learning

Data

- quantity
- stationarity
- quality
- structure

Features

- unsupervised
- semi-supervised
- supervised

Models

- structure
- type
- learning

- cost function
- maximum likelihood
- Bayesian
- online vs. off-line

- parametric: linear, nonlinear, mixture models

- non-parametric: kernel, Gaussian processes, clustering

- noise models

- integration of prior and domain knowledge



Outline

- Machine learning framework for sound search
 - *Involves all issues of machine learning*
- Genre classification
 - *Involves feature selection, projection and integration*
 - *Involves linear and nonlinear classifiers*
- Music separation
 - *Involves combination machine learning and other signal processing*
 - *NMF and ICA machine learning algorithms*
- MIMO channel estimation and symbol detection
 - *Involves advanced variational Bayesian learning*



The digital music market



- **Wired, April 27, 2005:**

"With the new Rhapsody, millions of people can now experience and share digital music legally and with no strings attached," Rob Glaser, RealNetworks chairman and CEO, said in a statement. "We believe that once consumers experience Rhapsody and share it with their friends, many people will upgrade to one of our premium Rhapsody tiers."

- **Financial Times (ft.com) 12:46 p.m. ET Dec. 28, 2005:**

LONDON - Visits to music downloading Web sites saw a 50 percent rise on Christmas Day as hundreds of thousands of people began loading songs on to the iPods they received as presents.

- **Wired, January 17, 2006:**

Google said today it has offered to acquire digital radio advertising provider dMarc Broadcasting for \$102 million in cash.



Huge demand for tools

- Organization, search and retrieval
 - Recommender systems ("taste prediction")
 - Playlist generation
 - Finding similarity in music (e.g., genre classification, instrument classification, etc.)
 - Hit prediction
 - Newscast transcription/search
 - Music transcription/search
- Machine learning is going to play a key role in future systems



Aspects of search

Specificity

- standard search engines
- indexing of deep content
- Objective: high retrieval performance

Similarity

- more like this
- similarity metrics
- Objective: high generalization and user acceptance



Specialized search and music organization

FindSounds
Search the Web for Sounds

Search for [Help](#)

[Need Examples?](#)

File Formats	Number of Channels	Minimum Resolution	Minimum Sample Rate	Maximum File Size
<input checked="" type="checkbox"/> AIFF	<input checked="" type="checkbox"/> mono	<input type="text" value="8-bit"/>	<input type="text" value="8000 Hz"/>	<input type="text" value="2 MB"/>
<input checked="" type="checkbox"/> AU	<input checked="" type="checkbox"/> stereo			
<input checked="" type="checkbox"/> WAVE				



Explore by Genre, mood, theme, country, instrument

Query by humming



Fraunhofer Institut Digitale Medientechnologie

The National Gallery of the Spoken Word

The NGSW is creating an online fully-searchable digital library of spoken word collections spanning the 20th century

Organize songs according to tempo, genre, mood



search for related songs using the "400 genes of music"



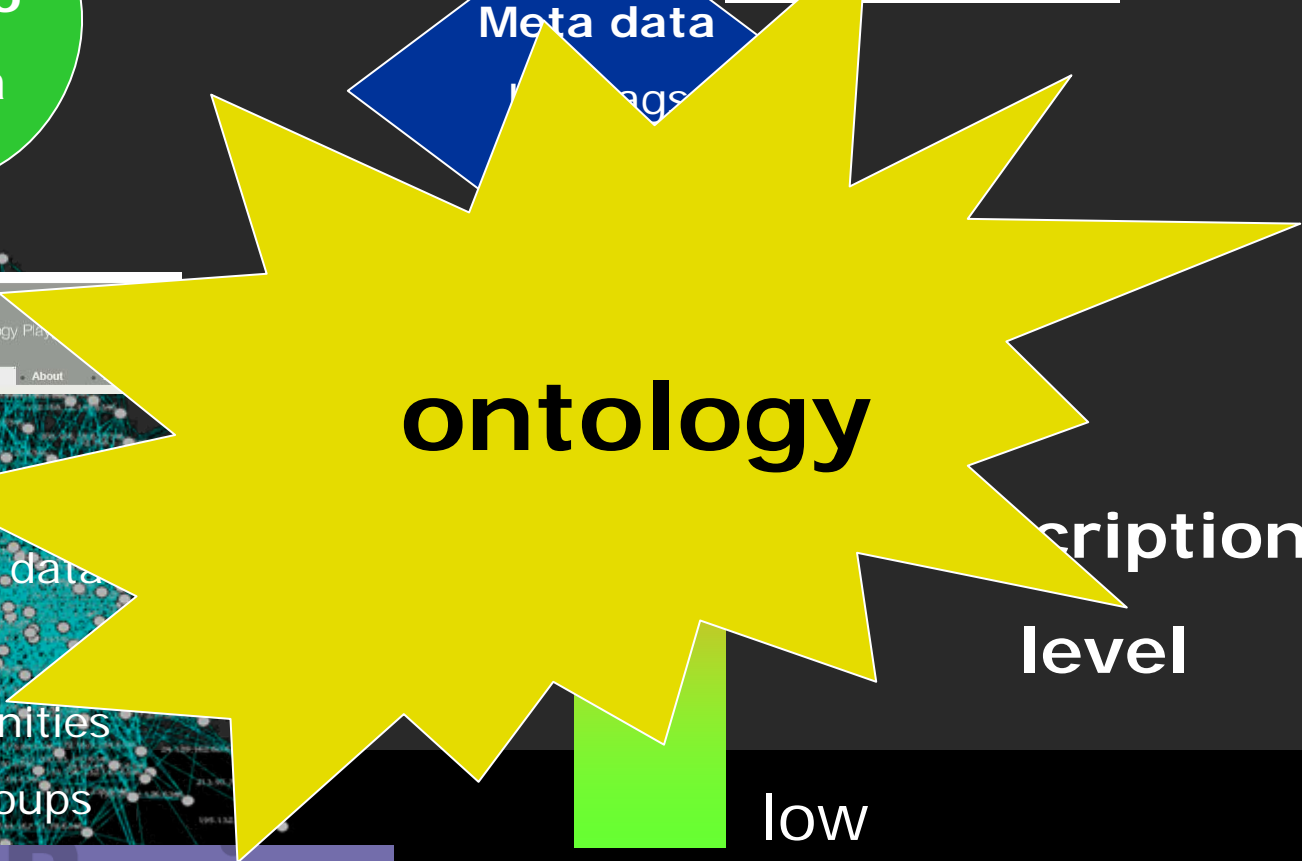
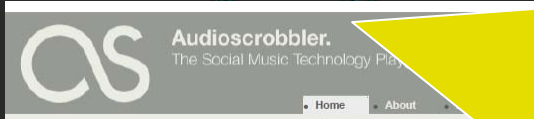


Sound information data

audio
data



Meta data
tags



User
co-play data
playlist
communities
user groups

description
level

low

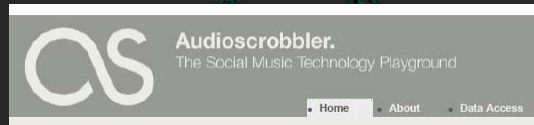




Machine learning in sound information processing

audio data

Meta data
ID3 tags
context



User networks

co-play data
playlist
communities
user groups

machine learning model

Tasks
Grouping
Classification
Mapping to a structure
Prediction
e.g. answer to query

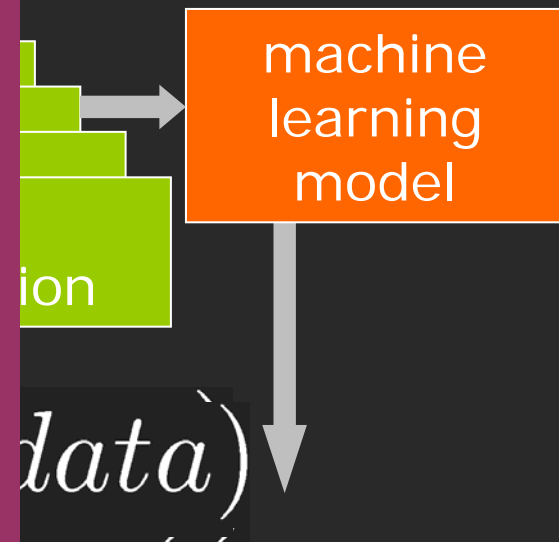
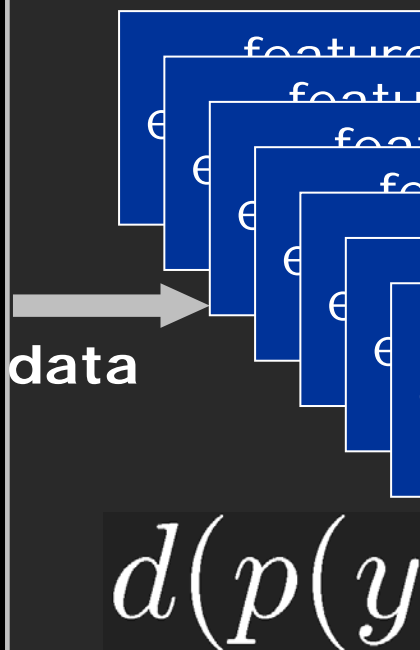




Machine learning for high level interpretations

Similarity functions

Euclidian, Weighted Euclidian, Cosine, Nearest Feature Line, earth Mover Distance, Self-organized Maps, Distance From Boundary, Cross-sampling, Bregman, KL, Manhattan, Adaptive





Similarity

Time

■ Low

—

■ High

—

■ Medium

—

Frequency domain

Time

- MFCC
- centroid
- Gamma tone filterbank
- roll-off
- ZCR
- low-pass filtering
- pitch
- spectral flatness
- brightness
- spectral tilt
- bandwidth
- sharpness
- harmonicity
- roughness
- spectrum power
- subband power



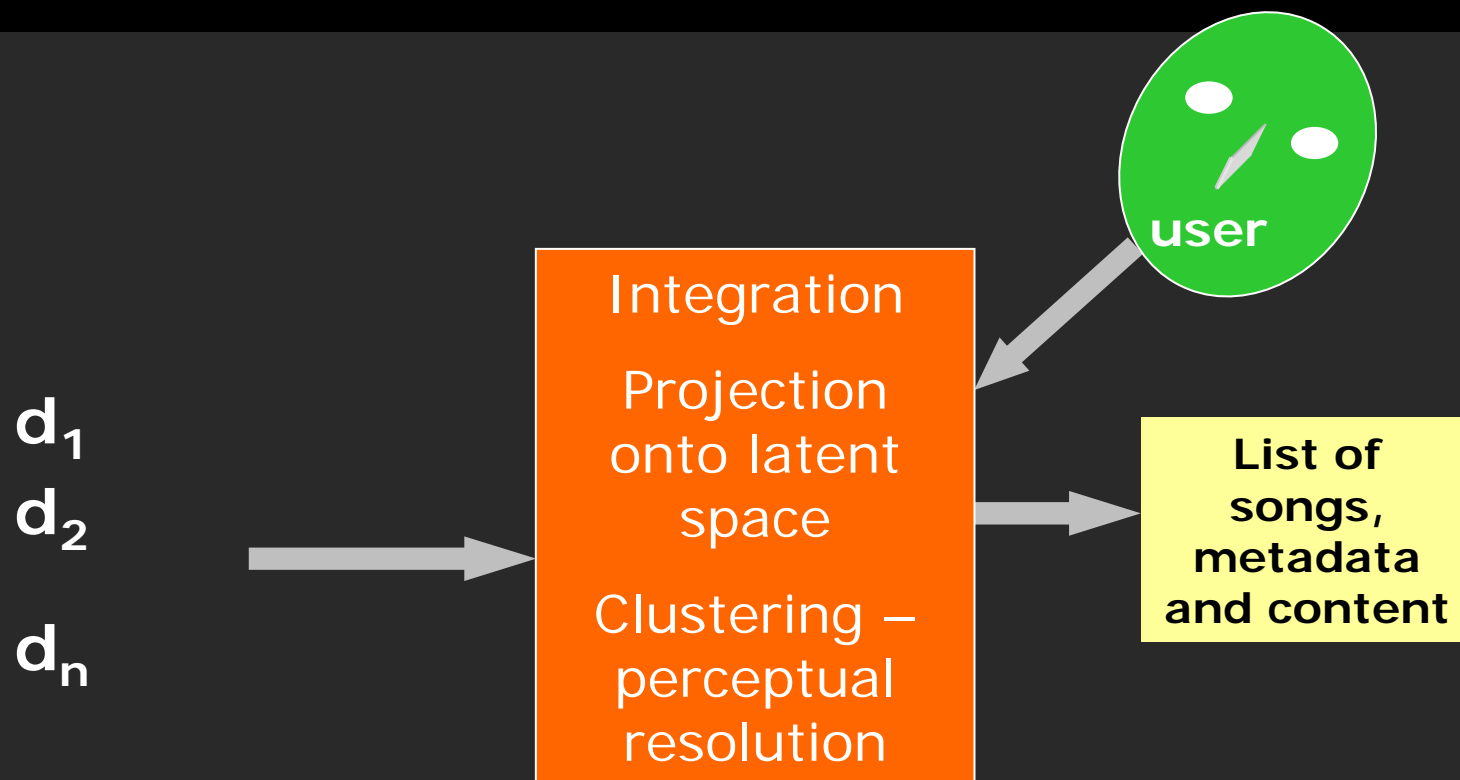
Predicting the answer from query

$$p(s_a | s_q, u)$$

- s_a : index for answer song
- s_q : index for query song
- u : user (group index)
- c_i : hidden cluster index of similarity i



Search and similarity integration





Now Playing

This field displays information about the artist currently playing. The information is retrieved from *text mining* of public domain internet sites.



Introduction

Financial Times (ft.com) 12:46 p.m. ET Dec. 28, 2005:

"LONDON - Visits to music downloading Web sites saw a 50 percent rise on Christmas Day as hundreds of thousands of people began loading songs on to the iPods they received as presents."

SoundSearch 0.1 combines co-play patterns, expert evaluations and music features to help you retrieve the music you like.

Use these music features to organize your search:

- Co-play
- Beat
- Expert
- Sound

Start the Music: 

<http://www.intelligentsound.org/demos/conceptdemo.swf>



Demo of WINAMP plugin



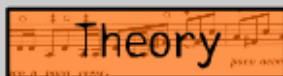
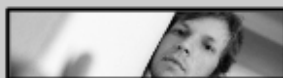
Lehn-Schiøler, T., Arenas-García, J., Petersen, K. B., Hansen, L. K., *A Genre Classification Plug-in for Data Collection*, ISMIR, 2006



The Clever Jukebox

The Art of Automated Genre Classification

examples:



Theory:

Automatic musical genre classification can be defined as the science (or art) of finding computer algorithms that take a (digitized) sound clip as input and yield a musical genre as output. The goal of automated genre classification is, of course, that the musical genre which is output should agree with the human classification of the sound into genre.



This demo illustrates an approach to the problem that first extract frequency-based sound features followed by a "linear regression" classifier. The basic features are the so-called mel-frequency cepstral coefficients (MFCCs), which are extracted on a time-scale of 30 msec. From these MFCC features, autoregressive coefficients (ARs) are extracted along with the mean and gain to get a single (30 dimensional) feature vector on the time-scale of 1 second. These features have been used because they have performed well in a previous study (Meng, Ahrendt, Larsen (2005)). Linear regression (or single-layer linear NN) is subsequently used for classification. This classifier is rather simple; current research investigates more advanced methods of classification.

Research: Peter Ahrendt, Design: Sune Lehmann.

© imm.dtu.dk 2004

<http://www.intelligentsound.org/demos/automusic.swf>



Intelligent Sound Project IMM (DTU) – CS, CT (AaU)

- Signal processing
- Databases
- Machine learning



Phd projects

Group
publications

Joint
publications

Workshops/
Phd-courses

Demo: Sound search engine

Demo: Matlab toolbox





Research tasks

AaU Communication Technology:

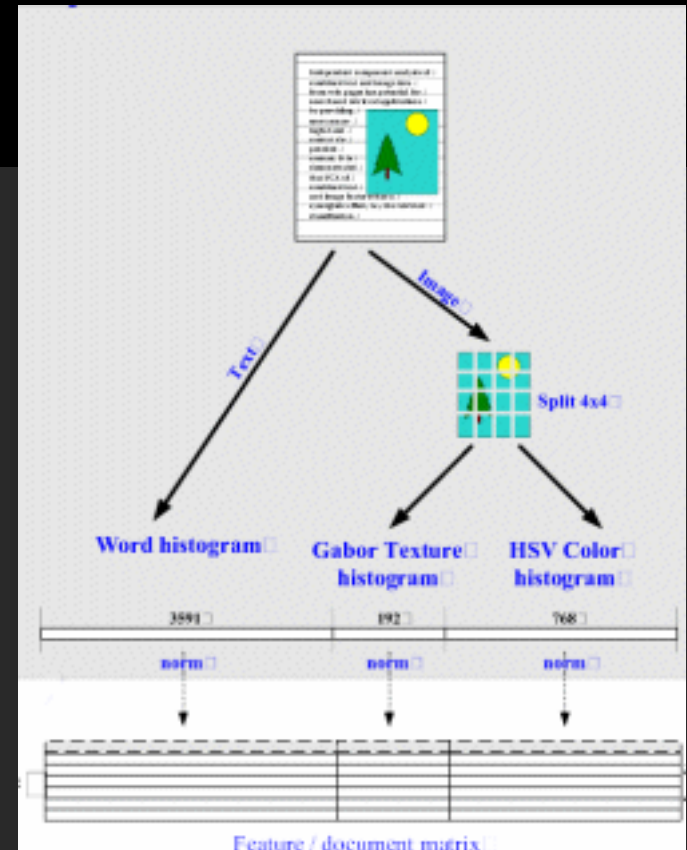
- TASK i): Features for sound based context modelling - MPEG and beyond
- TASK ii): Signal separation in noisy environments: ICA and noise reduction

AaU Computer Science/Database Management:

- TASK iii): Multidimensional management of sound as context
- TASK iv): Advanced Query Processing for Sound Feature Streams

DTU IMM-ISP

- TASK v): Context detection in sound streams
- TASK vi): Webmining for sound





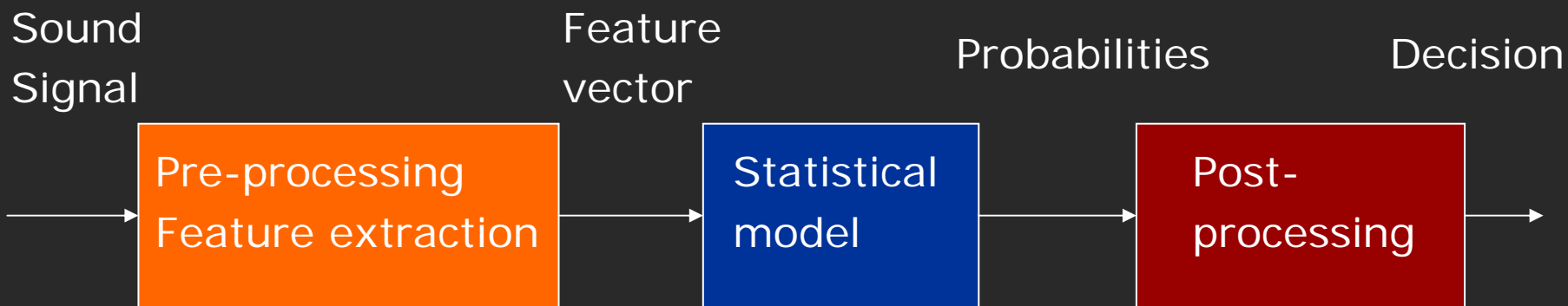
Genre classification

- Prototypical example of predicting meta data
- The problem of interpretation of genres
- Can be used for other applications e.g. hearing aids



Model

- Making the computer classify a sound piece into musical genres such as jazz, techno and blues.





How do humans do?

- Sounds – loudness, pitch, duration and timbre
- Music – mixed streams of sounds
- Recognizing musical genre
 - physical and perceptual: instrument recognition, rhythm, roughness, vocal sound and content
 - cultural effects



How well do humans do?

- Data set with 11 genres
- 25 people assessing 33 random 30s clips

accuracy
54 - 61 %

Baseline: 9.1%

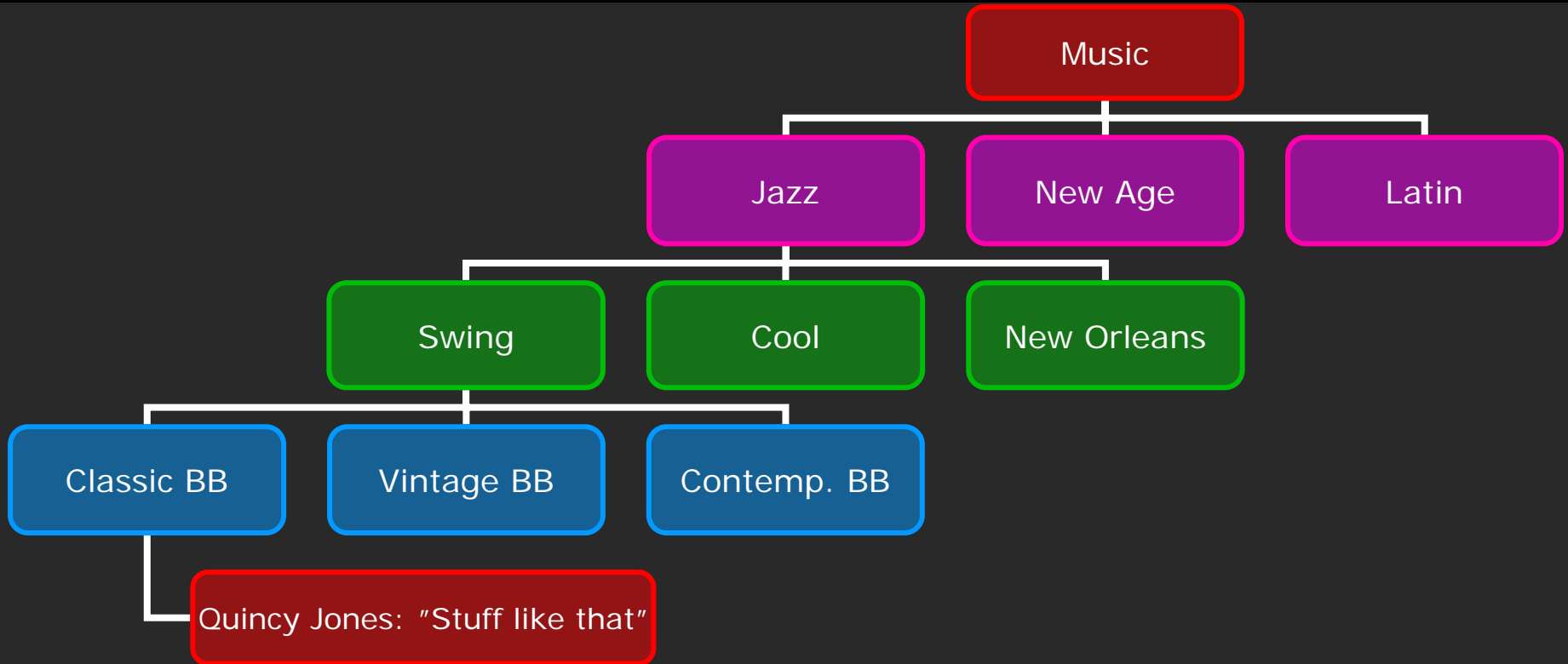


What's the problem ?

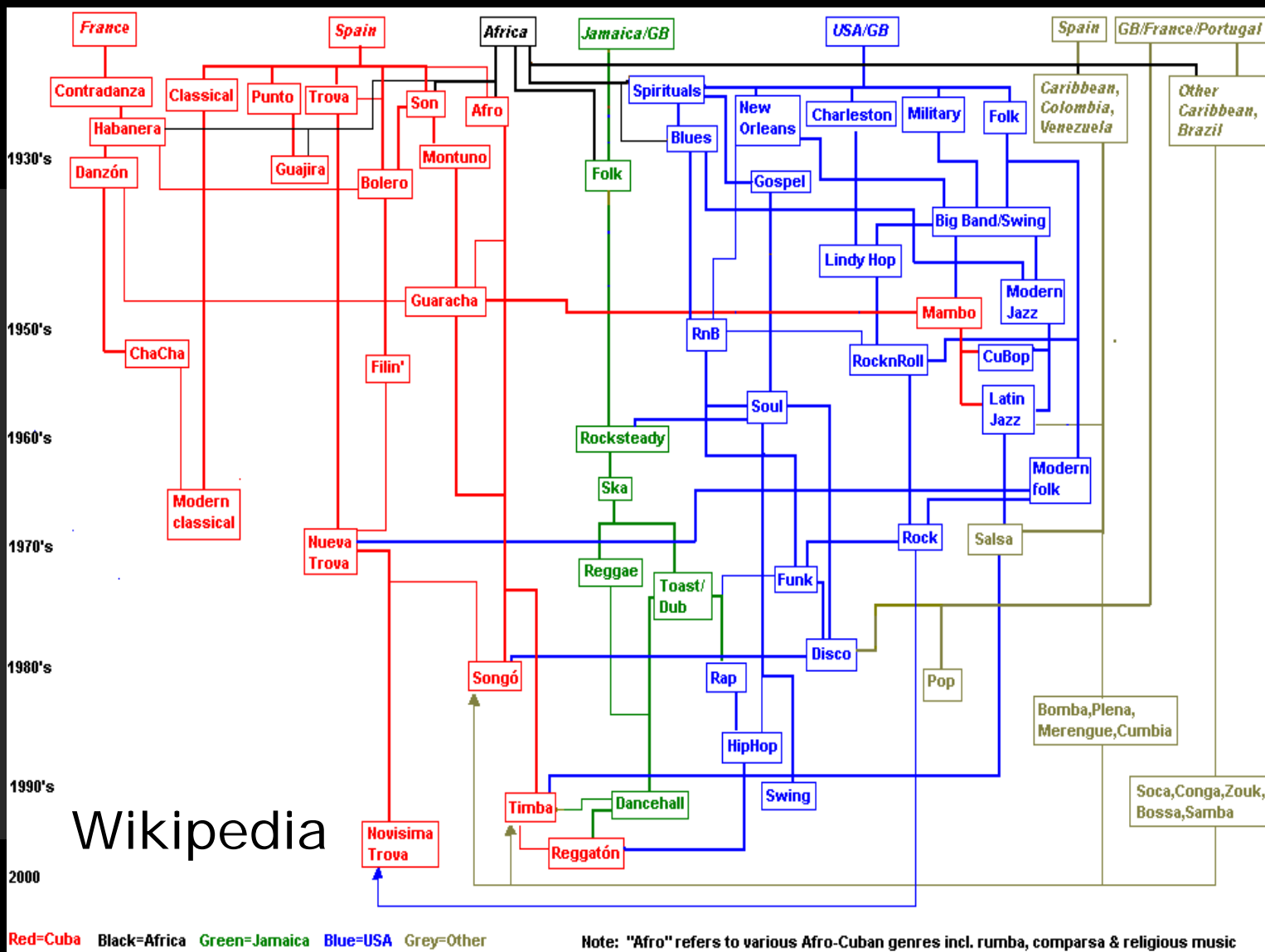
- Technical problem: Hierarchical, multi-labels
- Real problems: Musical genre is not an intrinsic property of music
 - A subjective measure
 - Historical and sociological context is important
 - No Ground-Truth



Music genres form a hierarchy

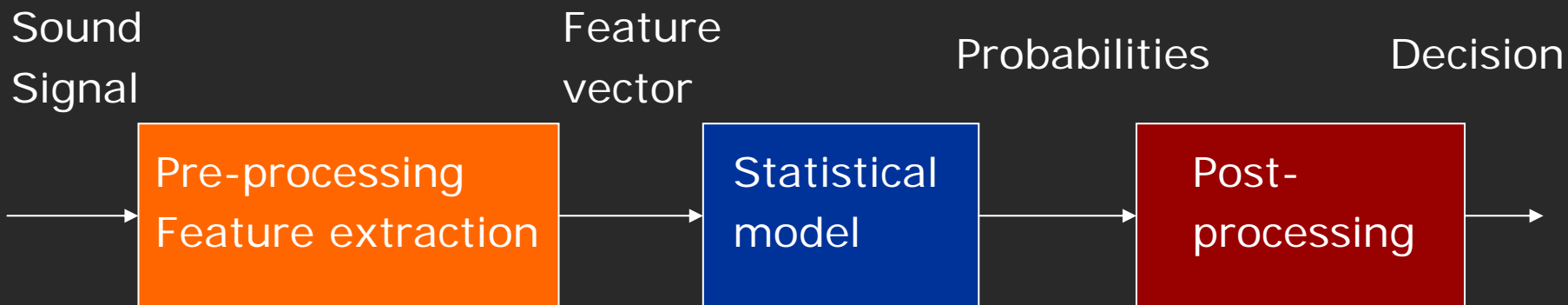


(according to Amazon.com)





Music Genre Classification Systems





Features

- Short time features (10-30 ms)
 - MFCC and LPC
 - Zero-Crossing Rate (ZCR), Short-time Energy (STE)
 - MPEG-7 Features (Spread, Centroid and Flatness Measure)
- Medium time features (around 1000 ms)
 - Mean and Variance of short-time features
 - Multivariate Autoregressive features (DAR and MAR)
- Long time features (several seconds)
 - Beat Histogram



On MFCC

Discrete
Fourier
transform

Log
amplitude
spectrum

Mel scaling
and
smoothing

Discrete
Cosine
transform

- MFCC represents a mel-weighted spectral envelope. The mel-scale models human auditory perception.
- Are believed to encode music timbre

Sigurdsson, S., Petersen, K. B., *Mel Frequency Cepstral Coefficients: An Evaluation of Robustness of MP3 Encoded Music*, Proceedings of the Seventh International Conference on Music Information Retrieval (ISMIR), 2006.



Features for genre classification

30s sound clip from the center of the song

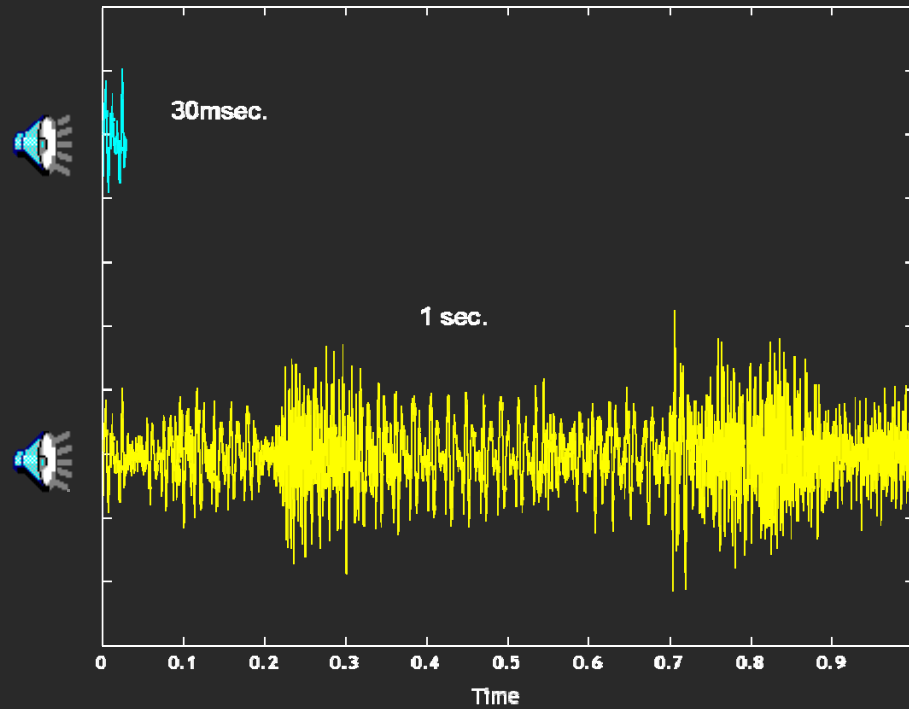
6 MFCCs, 30ms frame

6 MFCCs, 30ms frame

6 MFCCs, 30ms frame

3 ARCs per MFCC, 760ms frame

30-dimensional AR features, $x_r, r=1, \dots, 80$





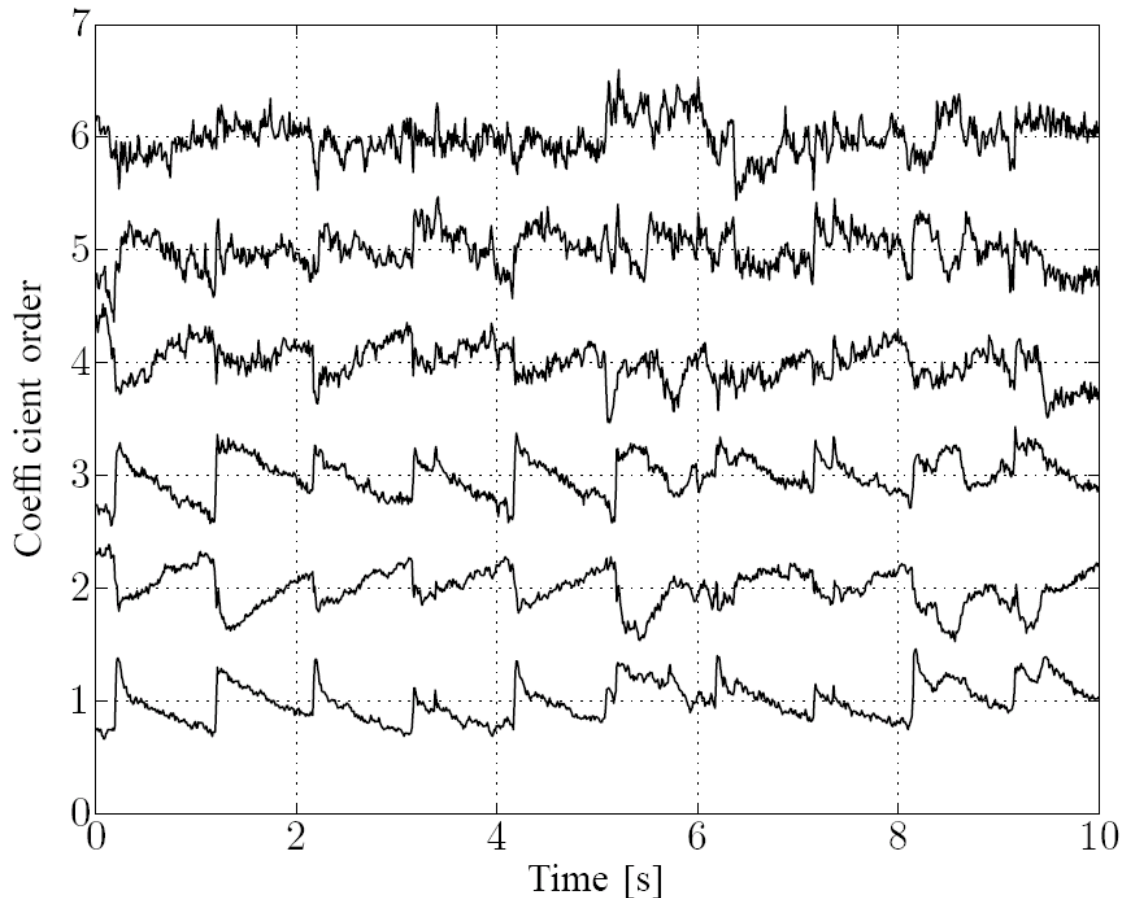
Statistical models

- Desired: $p(c|s)$ (genre class c and song s)
- Used models
 - Intregration of MFCCs using MAR models
 - Linear and non-linear neural networks
 - Gaussian classifier
 - Gaussian Mixture Model
 - Co-occurrence models



Example of MFCC's

A ten second excerpt of the song *Masters of Revenge* by *Body Count*



- Cross correlation
- Temporal correlation



Results reported in

- Meng, A., Ahrendt, P., Larsen, J., Hansen, L. K., Temporal Feature Integration for Music Genre Classification, IEEE Transactions on Signal Processing, 2006.
- A. Meng, P. Ahrendt, J. Larsen, *Improving Music Genre Classification by Short-Time Feature Integration*, IEEE International Conference on Acoustics, Speech, and Signal Processing, vol. V, pp. 497-500, 2005.
- Ahrendt, P., Goutte, C., Larsen, J., *Co-occurrence Models in Music Genre Classification*, IEEE International workshop on Machine Learning for Signal Processing, pp. 247-252, 2005.
- Ahrendt, P., Meng, A., Larsen, J., *Decision Time Horizon for Music Genre Classification using Short Time Features*, EUSIPCO, pp. 1293--1296, 2004.
- Meng, A., Shawe-Taylor, J., *An Investigation of Feature Models for Music Genre Classification using the Support Vector Classifier*, International Conference on Music Information Retrieval, pp. 604-609, 2005



Best results

- 5-genre problem (with little class overlap) : 2% error
 - Comparable to human classification on this database
- Amazon.com 6-genre problem (some overlap) : 30% error
- 11-genre problem (some overlap) : 50% error
 - human error about 43%



Best 11-genre confusion matrix

Alternative	41.8	6.4	4.5	3.6	3.6	2.7	8.2	2.7	4.5	3.6	18.2
Country	0.9	72.7	7.3	0.0	4.5	2.7	4.5	0.9	2.7	0.0	3.6
Easy-listening	1.8	11.8	61.8	2.7	4.5	2.7	2.7	0.0	2.7	3.6	5.5
Electronica	5.5	0.9	10.9	41.8	8.2	5.5	7.3	10.9	2.7	5.5	0.9
Jazz	0.9	4.5	8.2	10.9	50.0	2.7	3.6	2.7	7.3	6.4	2.7
Latin	3.6	8.2	2.7	4.5	3.6	37.3	8.2	8.2	4.5	11.8	7.3
Pop&Dance	6.4	9.1	6.4	9.1	0.9	11.8	43.6	2.7	3.6	2.7	3.6
Rap&Hiphop	0.0	0.0	0.9	7.3	0.9	4.5	3.6	62.7	1.8	17.3	0.9
RB&Soul	0.9	8.2	9.1	0.9	9.1	11.8	7.3	9.1	29.1	5.5	9.1
Reggae	0.9	0.9	0.0	3.6	4.5	5.5	1.8	17.3	3.6	61.8	0.0
Rock	25.5	16.4	5.5	0.9	5.5	2.7	6.4	0.0	6.4	1.8	29.1

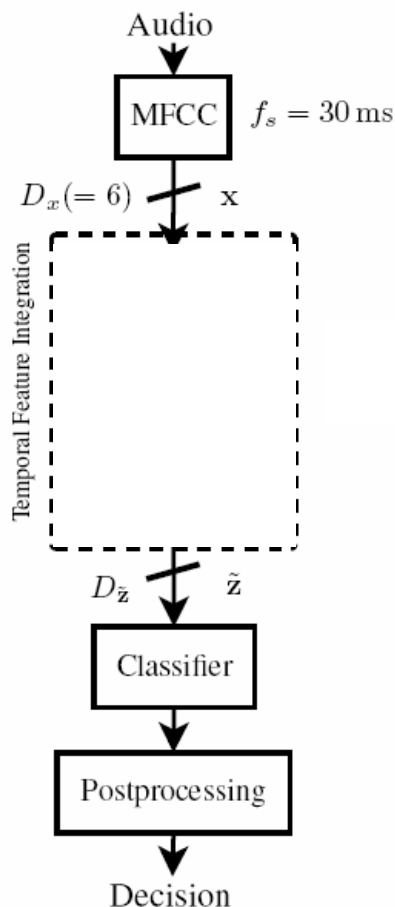


11-genre human evaluation

	Alternative	Country	Easy-listening	Electronica	Jazz	Latin	Pop&Dance	Rap&Hiphop	RB&Soul	Reggae	Rock
Alternative	16.0	2.7	9.3	9.3	1.3	0.0	32.0	0.0	4.0	2.7	22.7
Country	5.3	54.7	9.3	0.0	4.0	1.3	9.3	0.0	4.0	0.0	12.0
Easy-listening	17.3	0.0	34.7	8.0	12.0	0.0	13.3	5.3	2.7	0.0	6.7
Electronica	5.3	0.0	0.0	54.7	1.3	0.0	32.0	1.3	4.0	1.3	0.0
Jazz	5.3	0.0	5.3	4.0	70.7	6.7	2.7	1.3	4.0	0.0	0.0
Latin	2.7	0.0	8.0	5.3	5.3	56.0	14.7	0.0	5.3	2.7	0.0
Pop&Dance	4.0	1.3	10.7	10.7	0.0	1.3	62.7	0.0	5.3	1.3	2.7
Rap&Hiphop	1.3	0.0	5.3	1.3	1.3	1.3	1.3	80.0	6.7	0.0	1.3
RB&Soul	2.7	1.3	13.3	1.3	2.7	0.0	14.7	0.0	57.3	2.7	4.0
Reggae	5.3	0.0	0.0	4.0	0.0	0.0	1.3	5.3	2.7	81.3	0.0
Rock	12.0	1.3	9.3	0.0	1.3	2.7	8.0	1.3	2.7	0.0	61.3



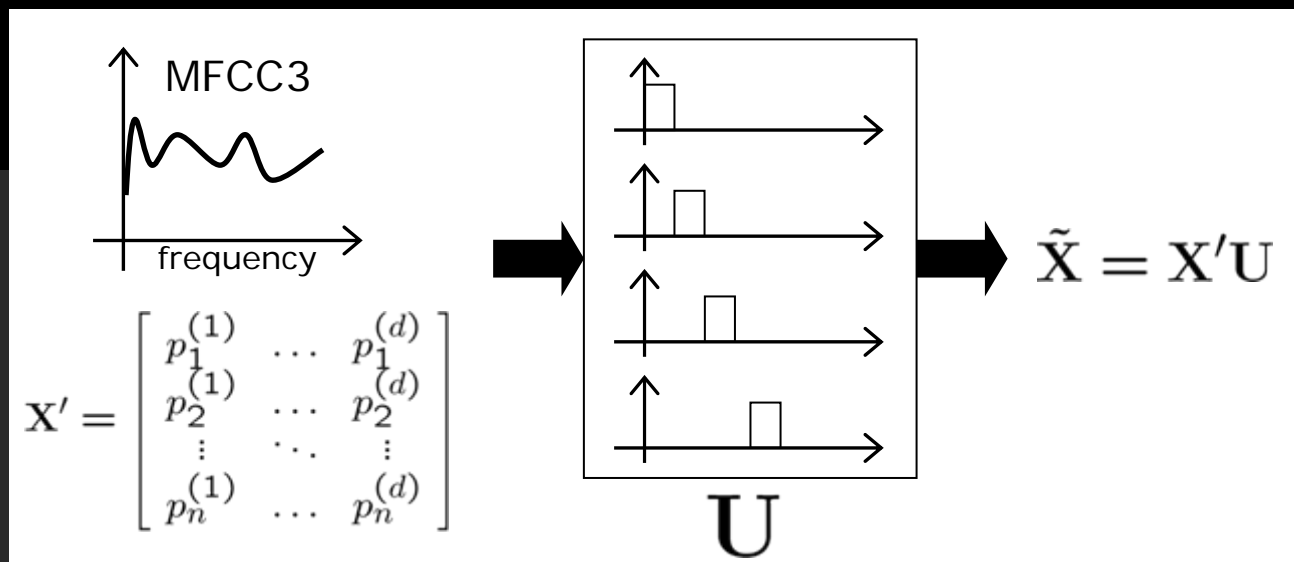
Supervised Filter Design in Temporal Feature Integration



Model the dynamics of MFCCs:

- Obtaining periodograms for each frame of 768ms MFCC
- “Bank-filter” these new features to obtain discriminative data

J. Arenas-Gacía, J. Larsen, L.H. Hansen, A. Meng:
Optimal filtering of dynamics in short-time features for music organization, ISMIR 2006.



- Periodograms contain information about how fast MFCCs change
- A bank with 4 constant-amplitude was proposed for genre classification
 - 0 Hz : DC Value
 - 1 – 2 Hz : Beat rates
 - 3 – 15 Hz : Modulation energy (e.g., vibrato)
 - 20 – $F_s/2$ Hz : Perceptual Roughness
- Orthonormalized PLS can be used for a better design of this bank filter. Additional constraint $\mathbf{U} > 0$: Positive Constrained OPLS (POPLS)



Principal component analysis (PCA)

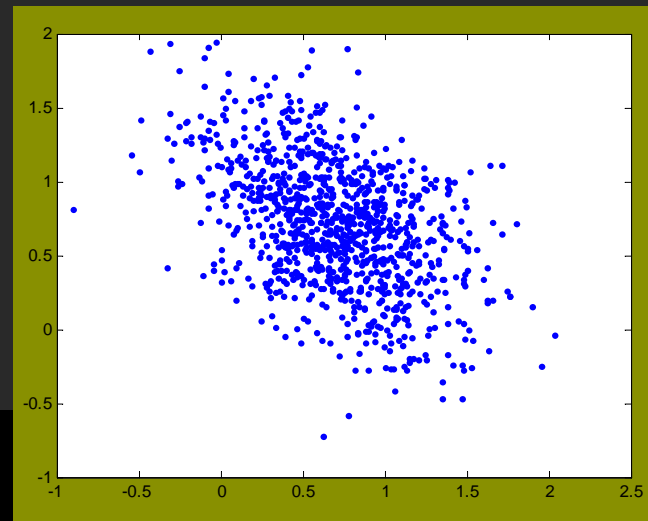
- Choose \mathbf{U} to make $\tilde{\mathbf{X}}$ the best approximation of \mathbf{X}

$$\mathbf{U} = \arg \min \left\| \mathbf{X} - \tilde{\mathbf{X}} \hat{\mathbf{B}} \right\|_F^2 \quad \hat{\mathbf{B}} = (\tilde{\mathbf{X}}^T \tilde{\mathbf{X}})^{-1} \tilde{\mathbf{X}}^T \mathbf{X}, \quad \tilde{\mathbf{X}} = \mathbf{X} \mathbf{U}$$

- This is equivalent to $\max_{\mathbf{u}_1, \dots, \mathbf{u}_{n_f}} \sum_{k=1}^{n_f} \left\| \mathbf{X} \mathbf{u}_k \right\|^2$, *s.t.* $\mathbf{u}_j^T \mathbf{u}_k = \delta_{jk}$

- $\mathbf{X} \mathbf{u}_1$ proj. explains the maximum variance of the data
- $\mathbf{X} \mathbf{u}_2$ the second one, *s.t.* $\mathbf{u}_2^T \mathbf{u}_1 = 0$

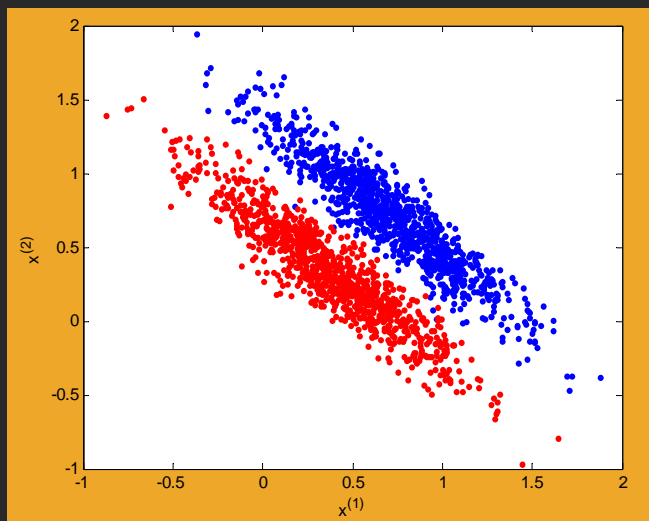
- PCA \mathbf{u}_k : eigenvectors of $\mathbf{C}_{\mathbf{X}\mathbf{X}}$





PCA for supervised learning

- Think about the following classification problem



- Which direction will PCA consider as the most relevant one?
- If only one feature is to be kept, which is the best projection vector under a classification perspective?

- When facing classification or regression problems, we should use the labels to obtain good features



Orthonormalized Partial Least Squares

- Choose \mathbf{U} to make $\tilde{\mathbf{X}}$ the best approximation of \mathbf{Y} in some space of reduced dimensionality

$$\mathbf{U} = \arg \min \left\| \mathbf{Y} - \tilde{\mathbf{X}} \hat{\mathbf{B}} \right\|_F^2 \quad \hat{\mathbf{B}} = (\tilde{\mathbf{X}}^T \tilde{\mathbf{X}})^{-1} \tilde{\mathbf{X}}^T \mathbf{Y}, \quad \tilde{\mathbf{X}} = \mathbf{XU}$$

- Rewriting the above equation

$$\text{OPLS: } \max_{\mathbf{U}} \mathbf{U}^T \mathbf{X}^T \mathbf{Y} \mathbf{Y}^T \mathbf{X} \mathbf{U} = \max_{\mathbf{U}} \mathbf{U}^T \mathbf{C}_{xy} \mathbf{C}_{yx} \mathbf{U},$$

$$\text{s.t. } \mathbf{U}^T \mathbf{X}^T \mathbf{X} \mathbf{U} = \tilde{\mathbf{X}}^T \tilde{\mathbf{X}} = \mathbf{I}$$

(projected data is white)



Optimality of OPLS

- OPLS is optimal for doing regression
- but can also be used for FE in classification if Y is used to encode class membership information
- We can directly use OPLS to design regressors and classifiers:

- We compute $\hat{\mathbf{B}} = (\tilde{\mathbf{X}}^T \tilde{\mathbf{X}})^{-1} \tilde{\mathbf{X}}^T \mathbf{Y}$ from the training data
- For new data

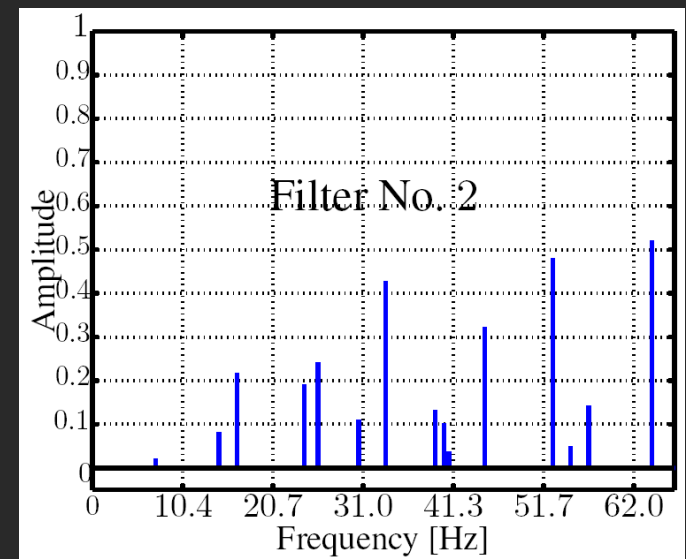
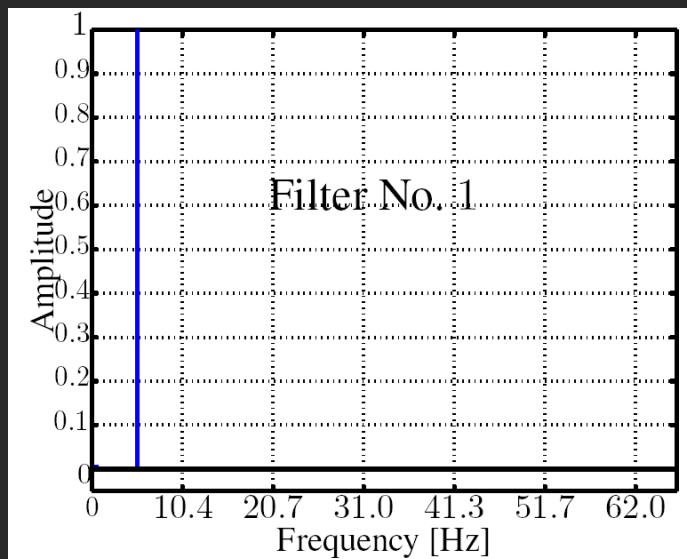
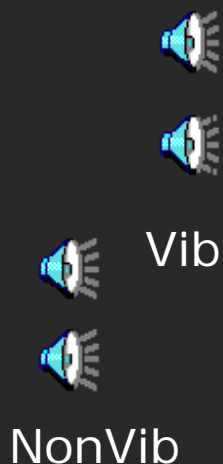
$$\hat{y} = \mathbf{x} \mathbf{U} \hat{\mathbf{B}} = \tilde{\mathbf{x}} \hat{\mathbf{B}} \quad (\text{regression})$$

$$\hat{y} = \text{w.t.a.}(\tilde{\mathbf{x}} \hat{\mathbf{B}}) \quad (\text{classification})$$



Illustrative example: vibrato detection

- 64 (32/32) AltoSax music snippets in Db3-Ab5
- Only the first MFCC was used

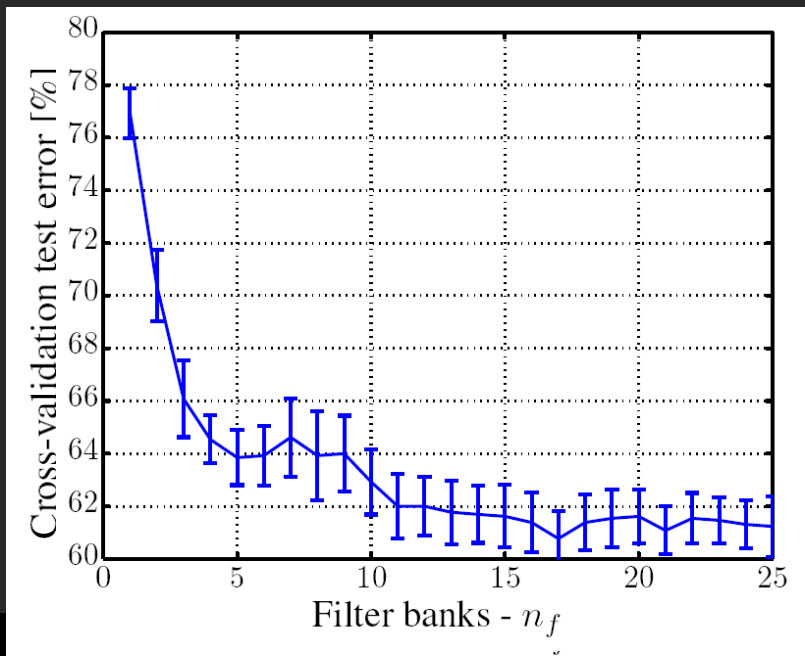


- Leave-one-out CV error: 9,4 % ($n_f = 25$); 20 % ($n_f = 2$)
(Fixed filter bank: 48,3 %)



POPLS for genre classification

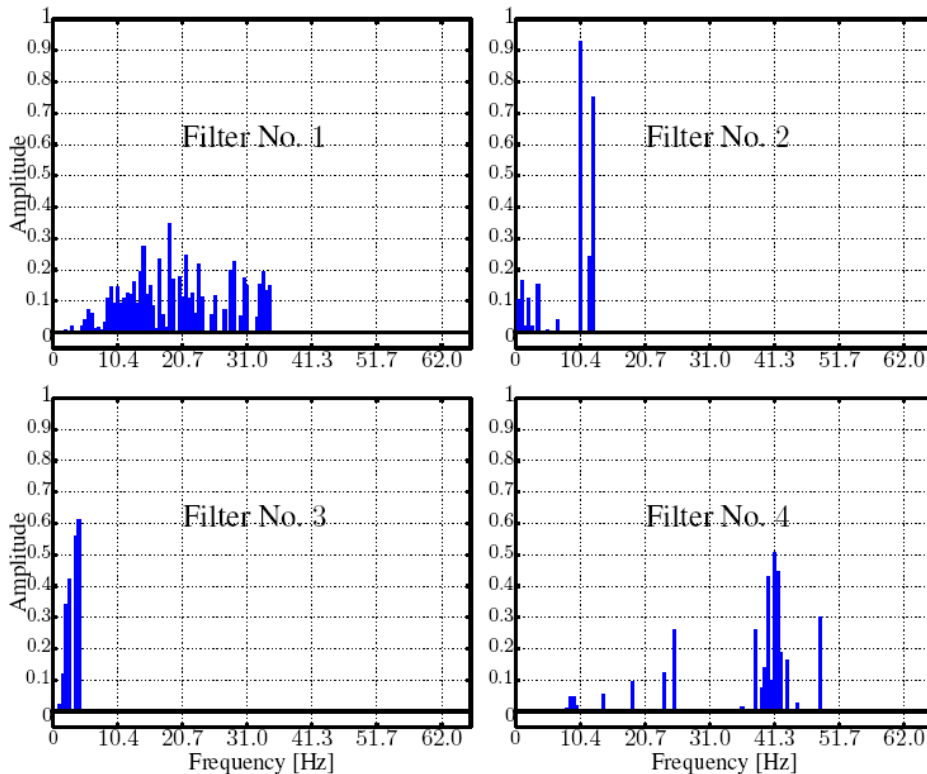
- 1317 music snippets (30 s) evenly distributed among 11 genres
- 7 MFCCs, but an unique filter bank



- POPLS 2% better on average compared to a fixed filter bank of four filter
- 10-fold cross-validation error falls to 61 % for $n_f = 25$



Interpretation of filters



- Filter 1: modulation frequencies of instruments
- Filter 2: lower modulation frequency + beat-scale
- Filter 4: perceptual roughness
- Consistent filters across 10-fold cross-validation
 - robustness to noise
 - relevant features for genre



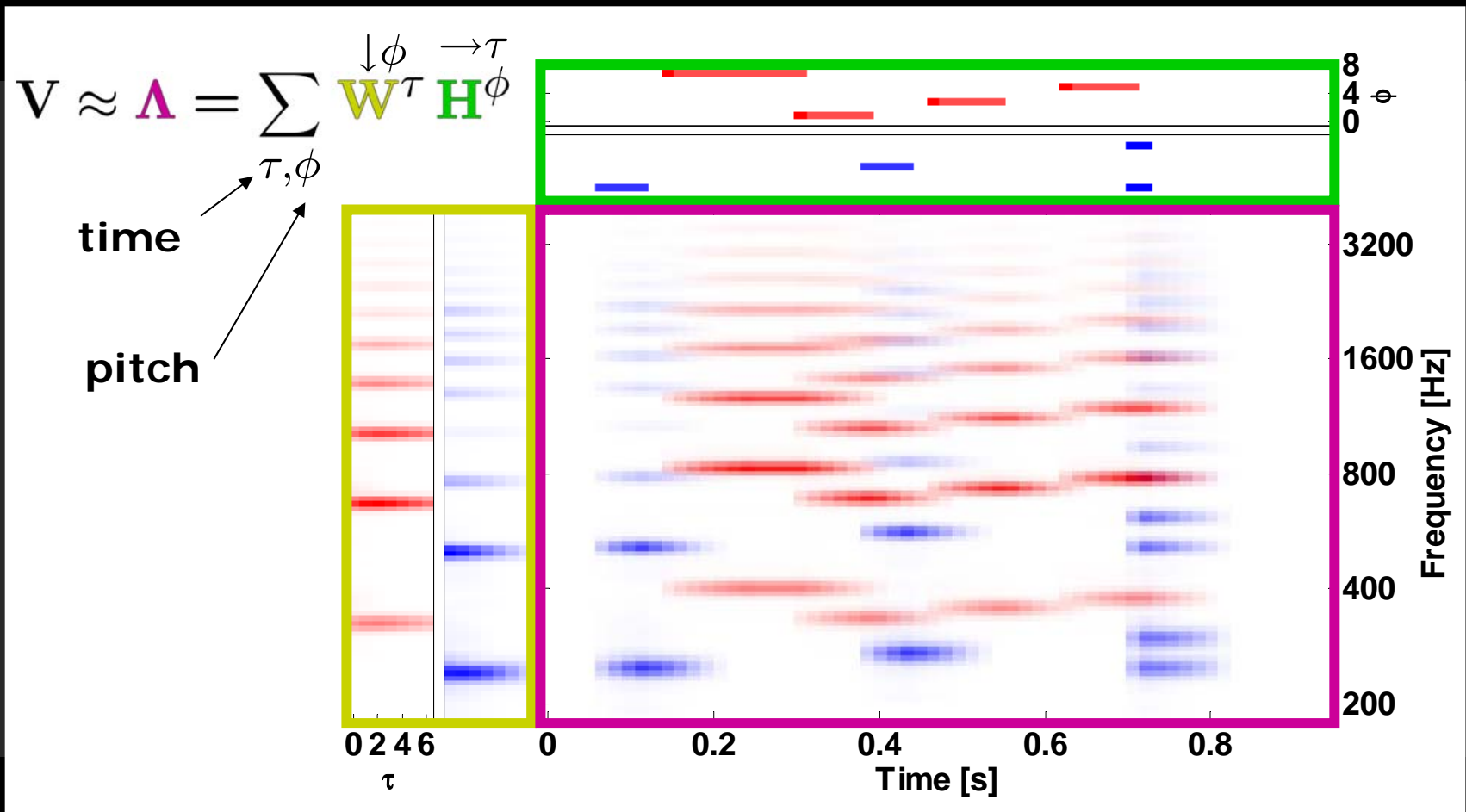
Music separation

- A possible front end component for the music search framework
- Noise reduction
- Music transcription
- Instrument detection and separation
- Vocalist identification

**Unsupervised/supervised
learning methods**



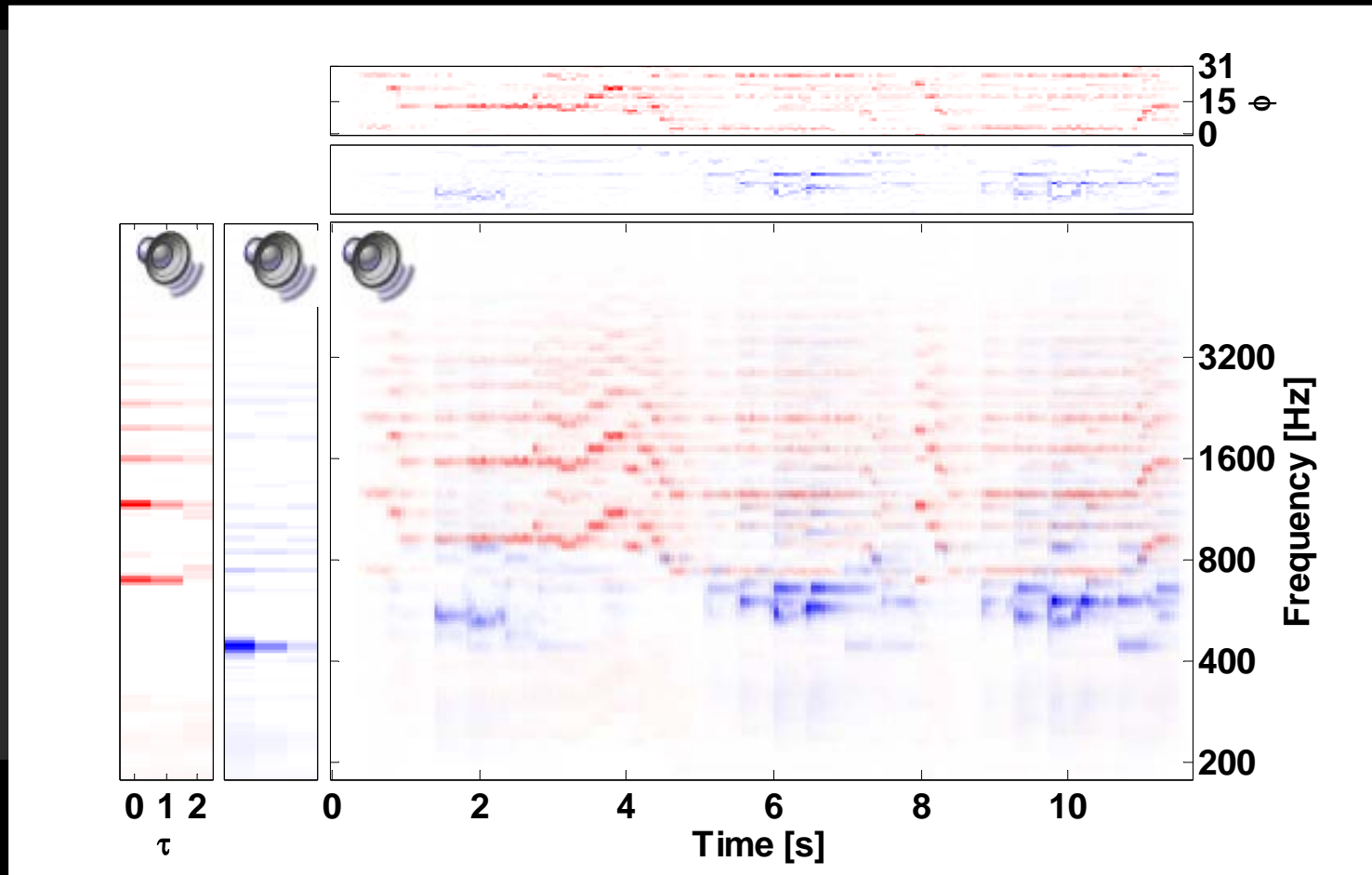
Nonnegative matrix factor 2D deconvolution



M. N. Schmidt, M. Mørup *Nonnegative Matrix Factor 2-D Deconvolution for Blind Single Channel Source Separation*, ICA2006, 2006. Demo also available.

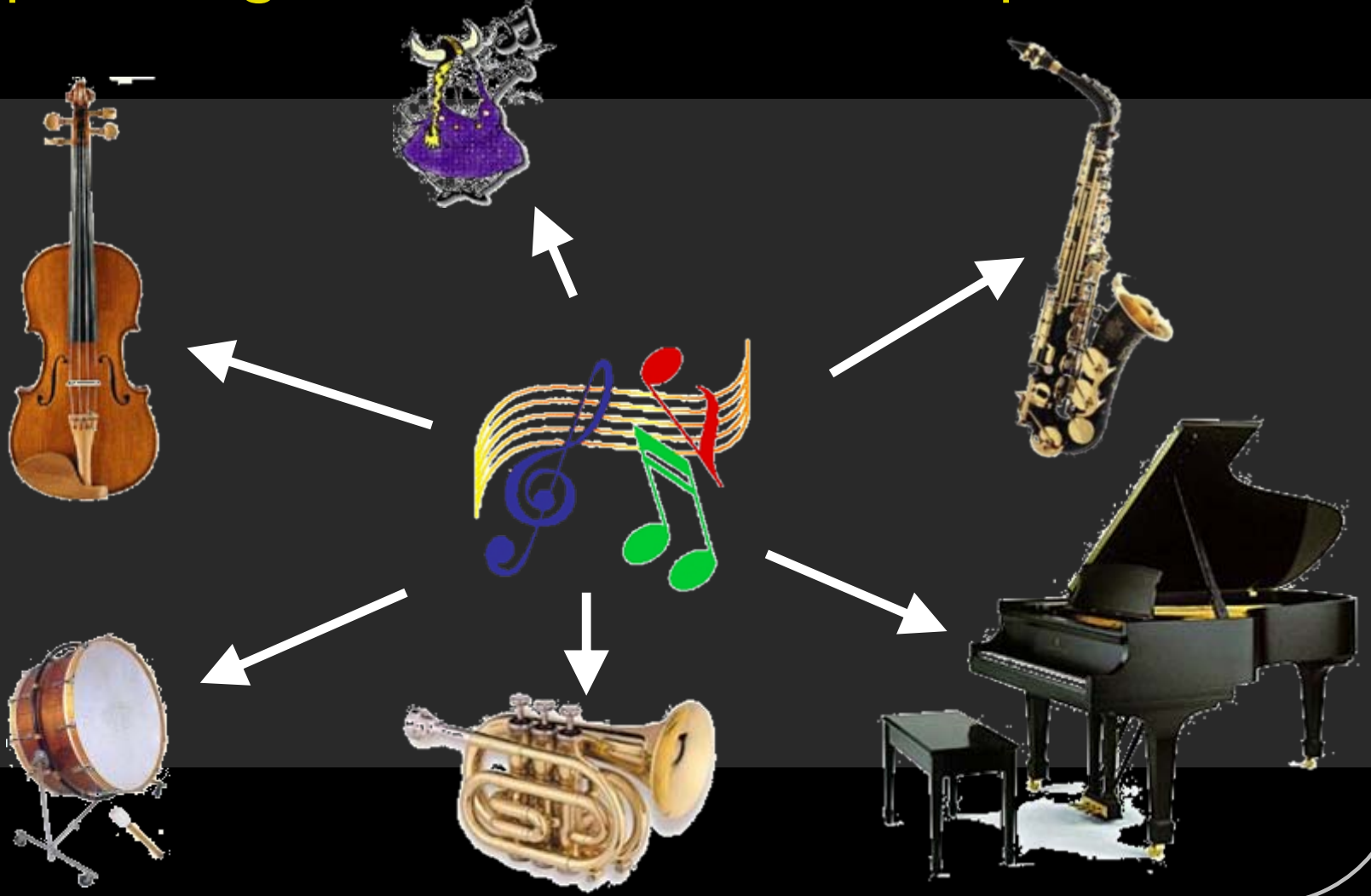


Demonstration of the 2D convolutive NMF model





Separating music into basic components





Separating music into basic components

■ Combined ICA and masking

- Pedersen, M. S., Wang, D., Larsen, J., Kjems, U., Two-microphone Separation of Speech Mixtures, 2006
- Pedersen, M. S., Lehn-Schiøler, T., Larsen, J., *BLUES from Music: BLind Underdetermined Extraction of Sources from Music*, ICA2006, vol. 3889, pp. 392-399, Springer Berlin / Heidelberg, 2006
- Pedersen, M. S., Wang, D., Larsen, J., Kjems, U., *Separating Underdetermined Convolutional Speech Mixtures*, ICA 2006, vol. 3889, pp. 674-681, Springer Berlin / Heidelberg, 2006
- Pedersen, M. S., Wang, D., Larsen, J., Kjems, U., *Overcomplete Blind Source Separation by Combining ICA and Binary Time-Frequency Masking*, IEEE International workshop on Machine Learning for Signal Processing, pp. 15-20, 2005

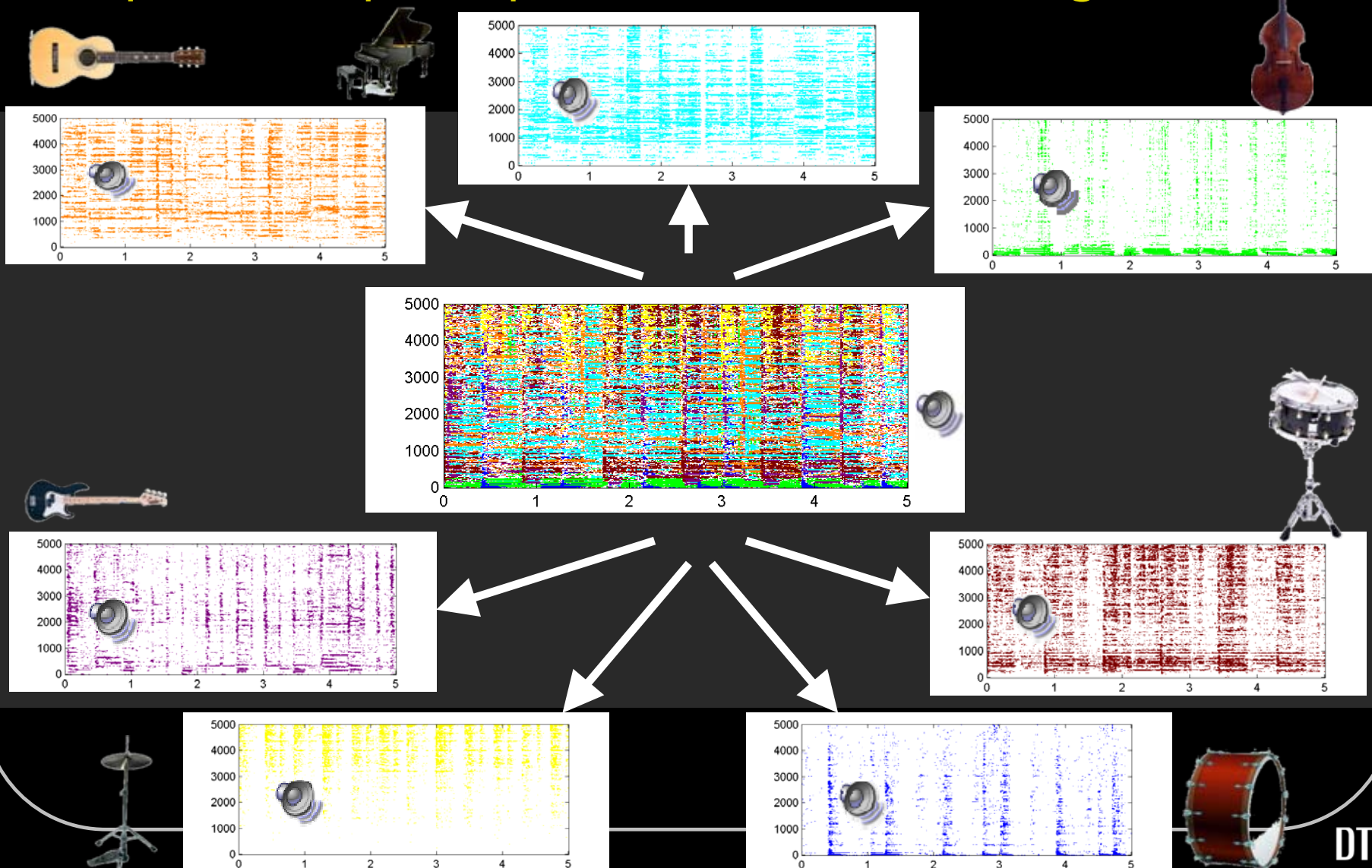


Assumptions

- Stereo recording of the music piece is available.
- The instruments are separated to some extent in time and in frequency, i.e., the instruments are sparse in the time-frequency (T-F) domain.
- The different instruments originate from spatially different directions.



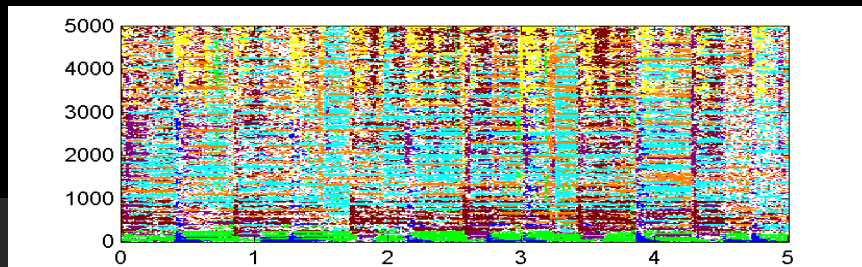
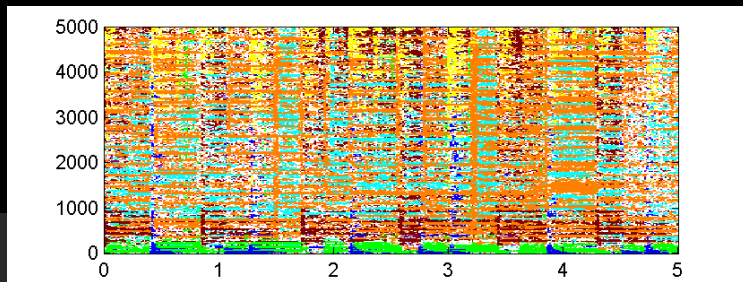
Separation principle: ideal T-F masking



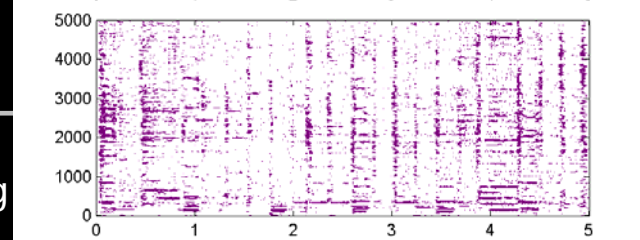
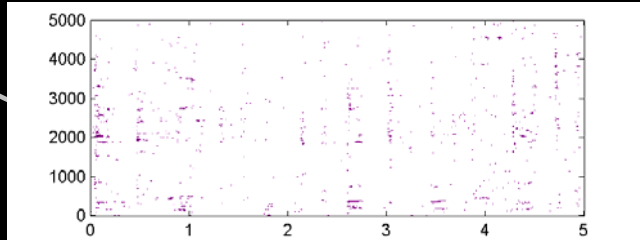
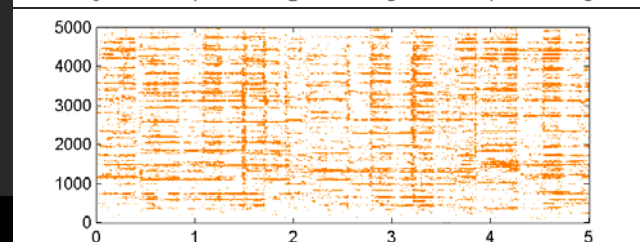
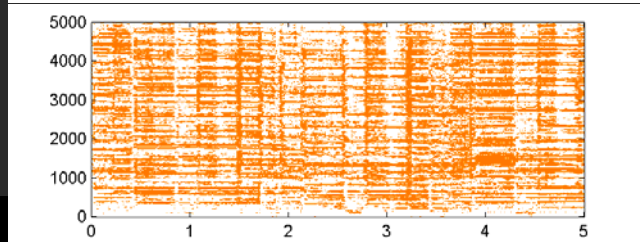
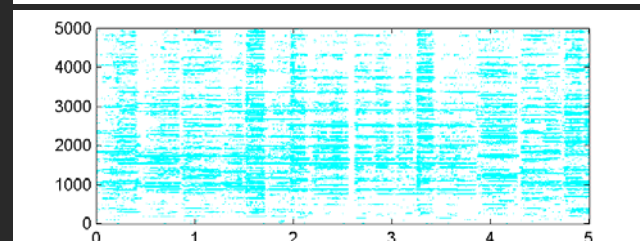
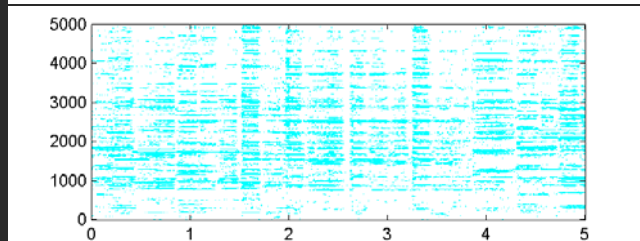
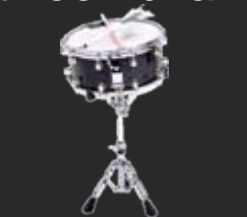
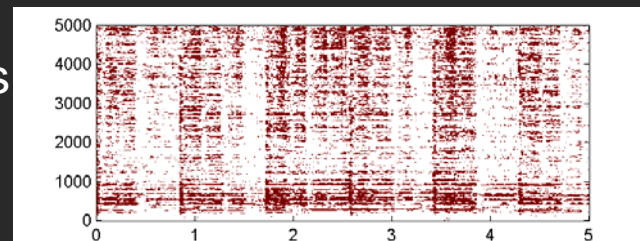
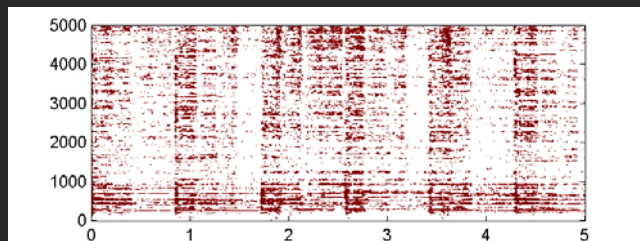


Stereo channel 1

Stereo channel 2



Gain difference between channels



Applications of learning





Separation principle 2: ICA



What happens if a 2-by-2 separation matrix W is applied to a 2-by- N mixing system?



ICA on stereo signals

- We assume that the mixture can be modeled as an instantaneous mixture, i.e.,

$$x = A(\theta_1, \dots, \theta_N)s \quad A(\theta) = \begin{bmatrix} r_1(\theta_1) & \dots & r_1(\theta_N) \\ r_2(\theta_1) & \dots & r_2(\theta_N) \end{bmatrix}$$

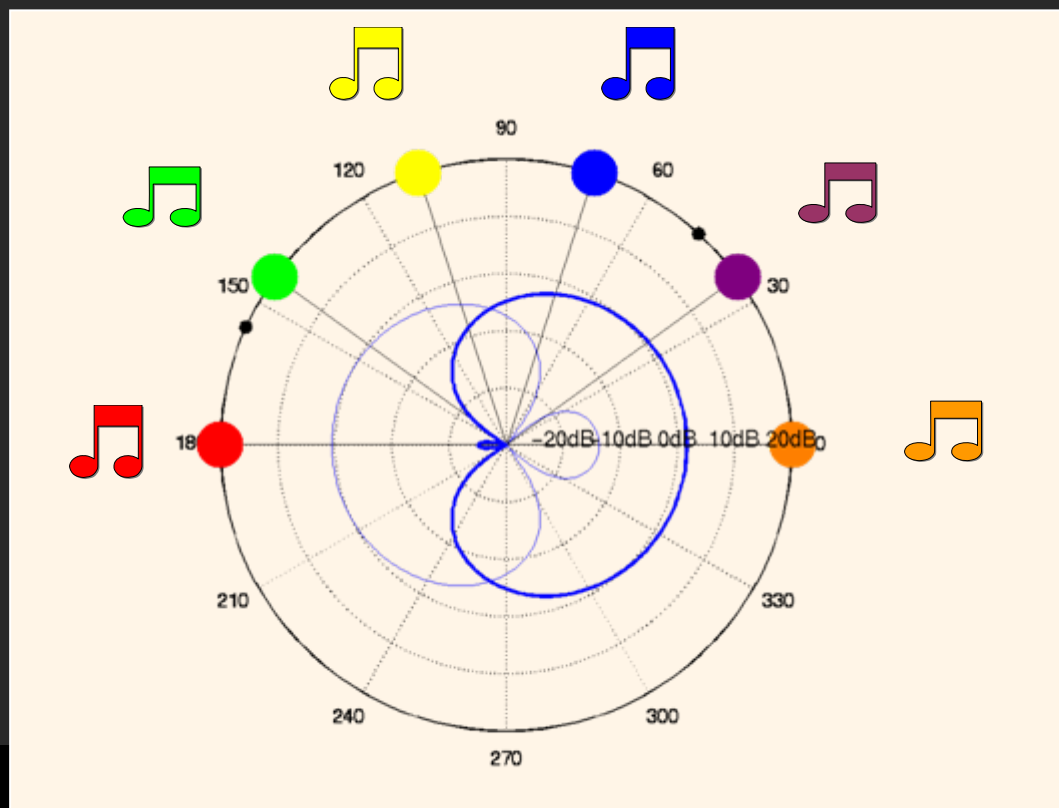
- The ratio between the gains in each column in the mixing matrix corresponds to a certain direction



Direction dependent gain

$$r(\theta) = 20 \log | \mathbf{W} \mathbf{A}(\theta) |$$

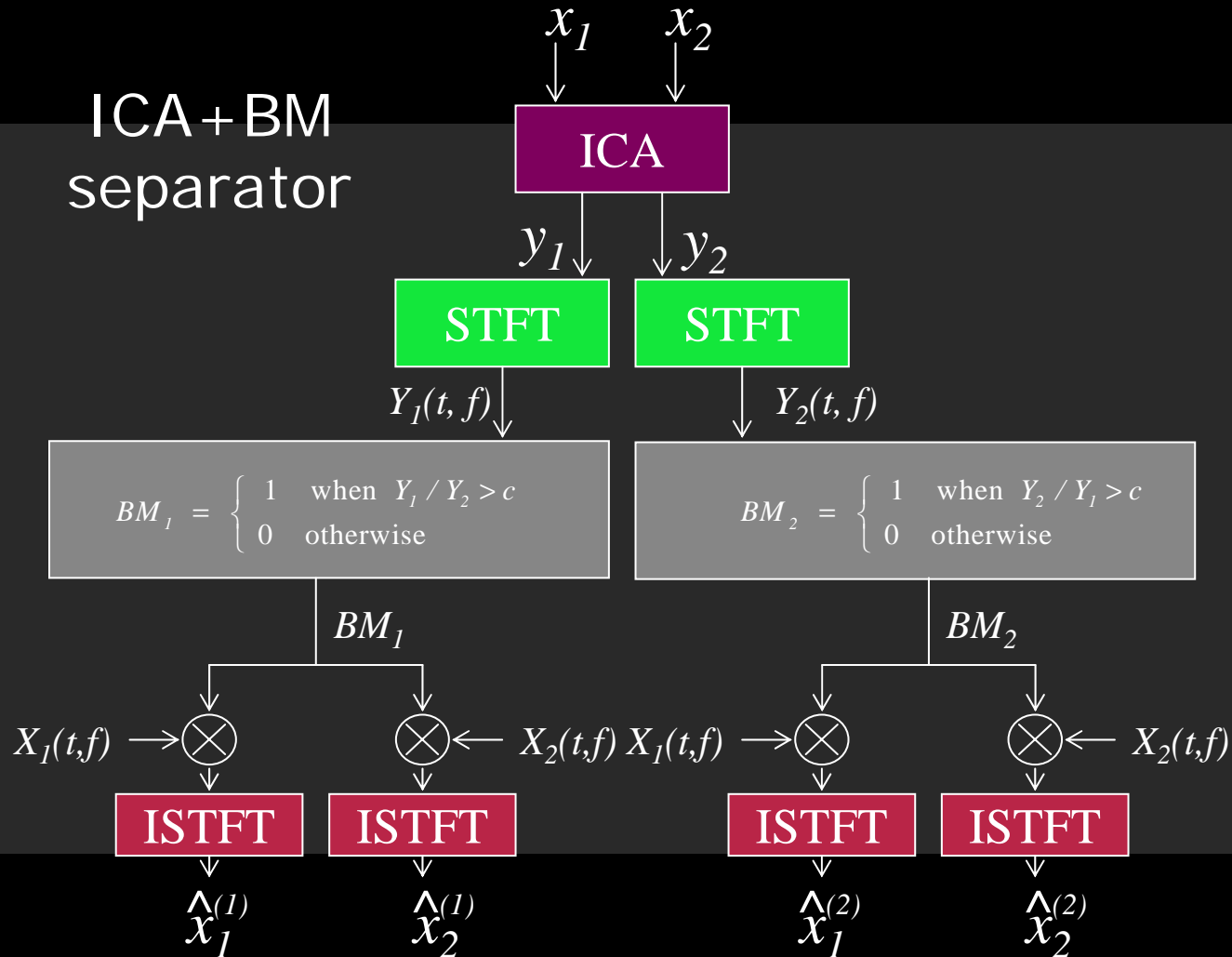
When \mathbf{W} is applied, the two separated channels each contain a *group* of sources, which is as independent as possible from the other channel.





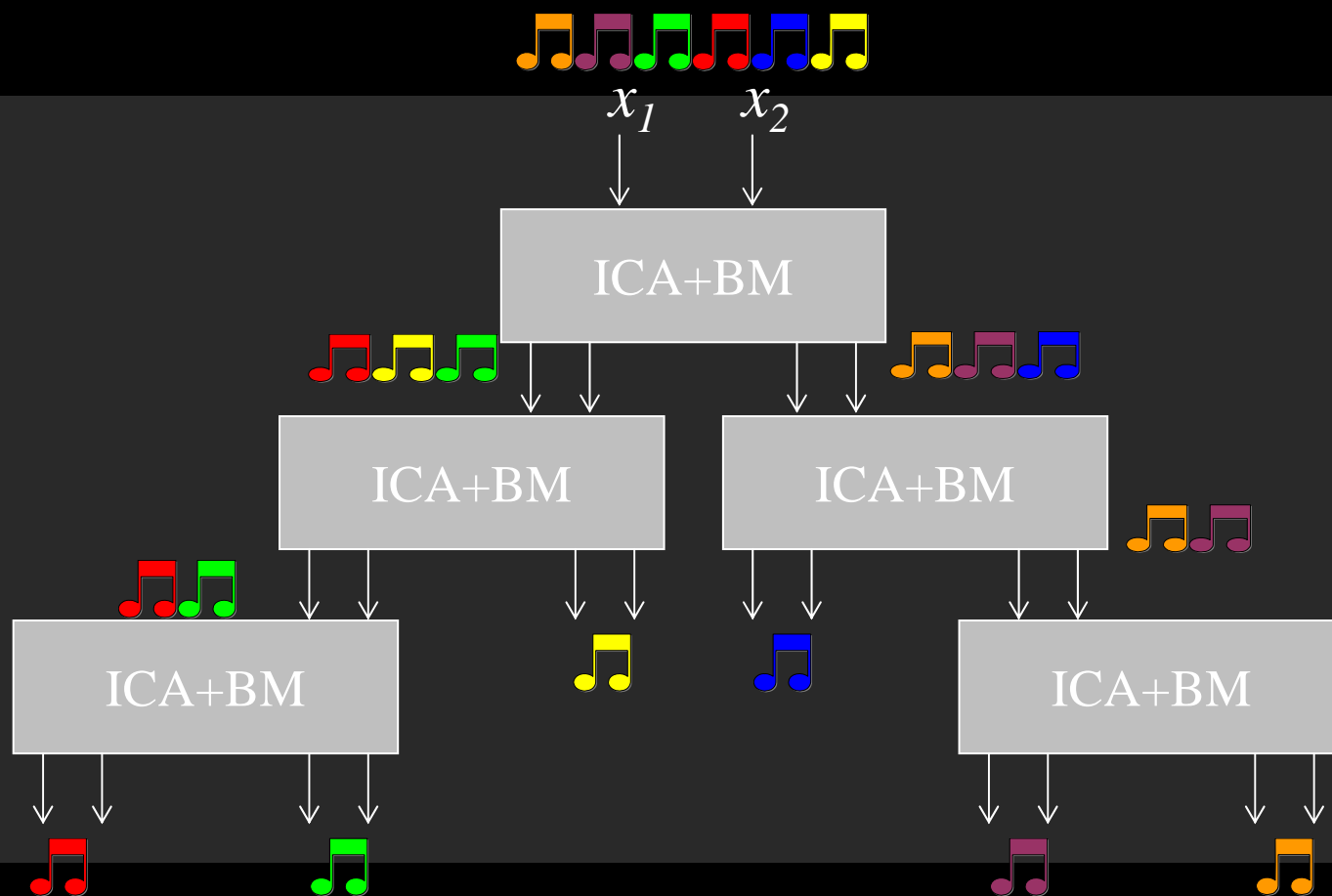
Combining ICA and T-F masking

ICA + BM
separator





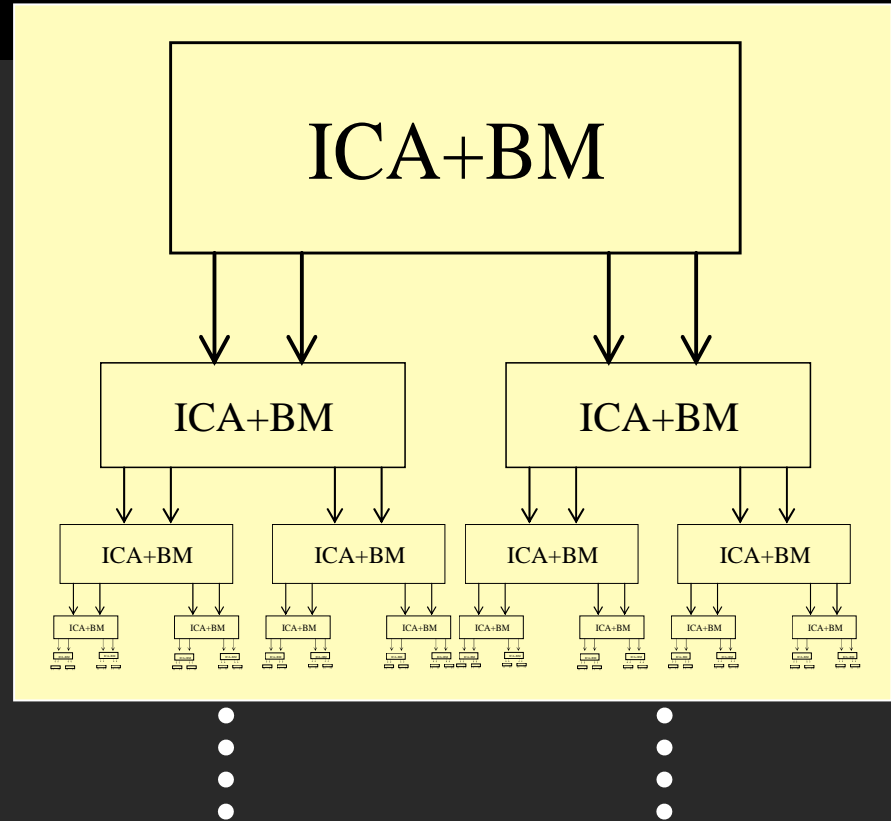
Method applied iteratively





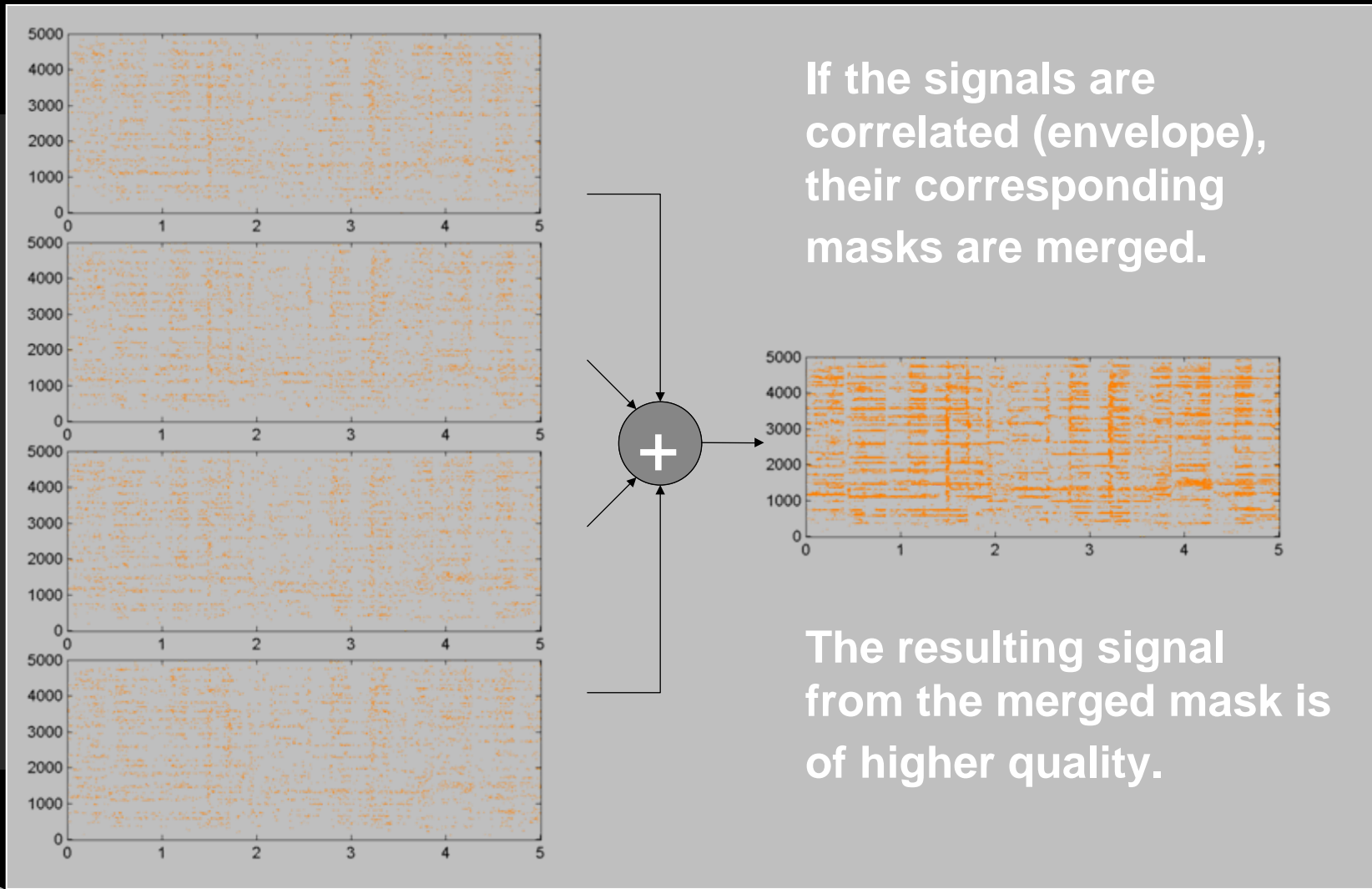
Improved method

- The assumption of instantaneous mixing may not always hold
- Assumption can be relaxed
- Separation procedure is continued until very sparse masks are obtained
- Masks that mainly contain the same source are afterwards merged





Mask merging



If the signals are correlated (envelope), their corresponding masks are merged.

The resulting signal from the merged mask is of higher quality.



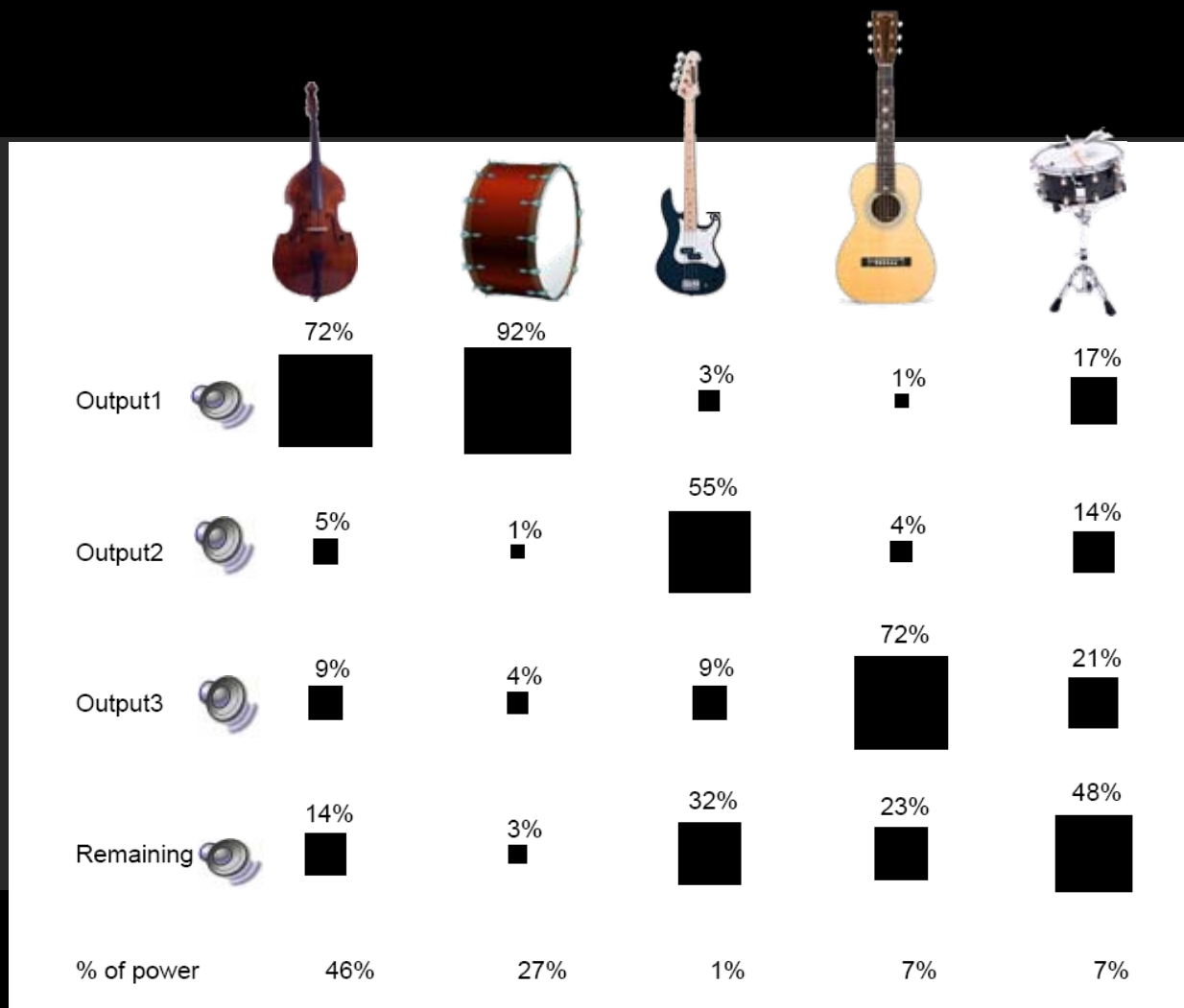
Results

- Evaluation on real stereo music recordings, with the stereo recording of each instrument available, before mixing.
- We find the correlation between the obtained sources and the by the ideal binary mask obtained sources.
- Other segregated music examples and code are available online via <http://www.imm.dtu.dk>



Results

- The segregated outputs are dominated by individual instruments
- Some instruments cannot be segregated by this method, because they are not spatially different.





Conclusion on combined ICA T-F separation

- An unsupervised method for segregation of single instruments or vocal sound from stereo music.
- The segregated signals are maintained in stereo.
- Only spatially different signals can be segregated from each other.
- The proposed framework may be improved by combining the method with single channel separation methods.



MIMO channel estimation and symbol detection

- Application of machine-learning algorithm to wireless communications
- Improved iterative parameter estimation framework compared to the EM-algorithm
- Generalizes the EM-algorithm by working with parameter distributions instead of point-estimates
- Explicit solutions provided for channel and covariance estimation
- Similar complexity per iteration as the EM-algorithm

Christensen, L. P. B., Larsen, J., *On Data and Parameter Estimation Using the Variational Bayesian EM-algorithm for Block-fading Frequency-selective MIMO Channels*, ICASSP, 2006



MIMO system model

$$\begin{aligned} \mathbf{y}_i &= \mathbf{H}\mathbf{x}_i + \mathbf{n}_i, & \mathbf{n}_i &\sim \mathcal{CN}(\mathbf{0}, \mathbf{\Sigma}) \\ &= \mathbf{X}_i\mathbf{h} + \mathbf{n}_i, & \mathbf{h} &\triangleq \text{vec}(\mathbf{H}) \end{aligned}$$

Model parameters $\boldsymbol{\theta} = \{\mathbf{h}, \mathbf{\Sigma}\}$

Symbols $\mathbf{x} \in \Omega$

- Block fading: the channel is constant over a frame of symbols
- Frequency selective channel has length of L symbols



EM learning with hidden variables

- The likelihood of \mathbf{y} is incomplete as the symbols are unknown

$$\text{E: } Q\left(\boldsymbol{\theta}, \boldsymbol{\theta}^{(j-1)}\right) \triangleq \langle \ln [p(\mathbf{y}, \mathbf{x} | \boldsymbol{\theta})] \rangle_{p(\mathbf{x} | \mathbf{y}, \boldsymbol{\theta}^{(j-1)})}$$

$$\text{M: } \boldsymbol{\theta}^{(j)} \triangleq \arg \max_{\boldsymbol{\theta}} Q\left(\boldsymbol{\theta}, \boldsymbol{\theta}^{(j-1)}\right)$$

E step is computed using the BCJR forward-backward algorithm



Variational Bayes learning

- Parameter fluctuations is taken into account

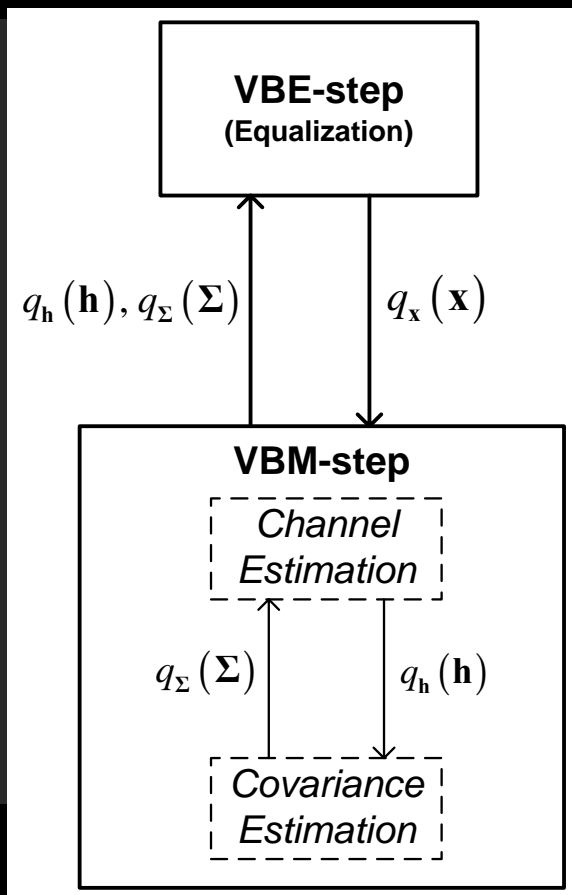
$$\text{VBE: } q_{\mathbf{x}}^{(j)}(\mathbf{x}) \propto e^{\langle \ln[p(\mathbf{y}, \mathbf{x} | \boldsymbol{\theta})] \rangle_{q_{\boldsymbol{\theta}}^{(j-1)}(\boldsymbol{\theta})}}$$

$$\text{VBM: } q_{\boldsymbol{\theta}}^{(j)}(\boldsymbol{\theta}) \propto p(\boldsymbol{\theta}) e^{\langle \ln[p(\mathbf{y}, \mathbf{x} | \boldsymbol{\theta})] \rangle_{q_{\mathbf{x}}^{(j)}(\mathbf{x})}}$$

Reduces to EM when $q_{\boldsymbol{\theta}}(\boldsymbol{\theta}) = \delta(\boldsymbol{\theta} - \boldsymbol{\theta}_{\text{MAP}})$



Algorithm

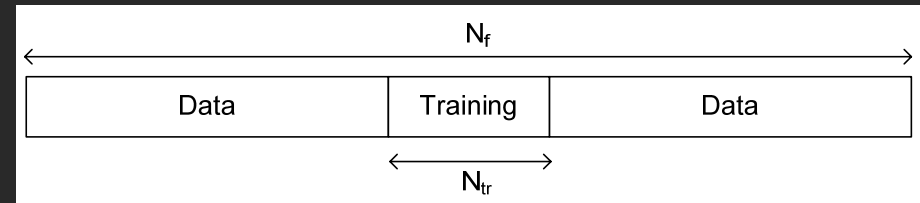


- Conjugated priors enables VBE step similar to that of BCJR in EM



Simulation example

- BPSK link considered (linearized GSM system)
- SISO Block-fading Typical Urban (TU) channel model, channel length $L=7$
- AWGN, i.e., scalar covariance estimation
- Non-informative priors used



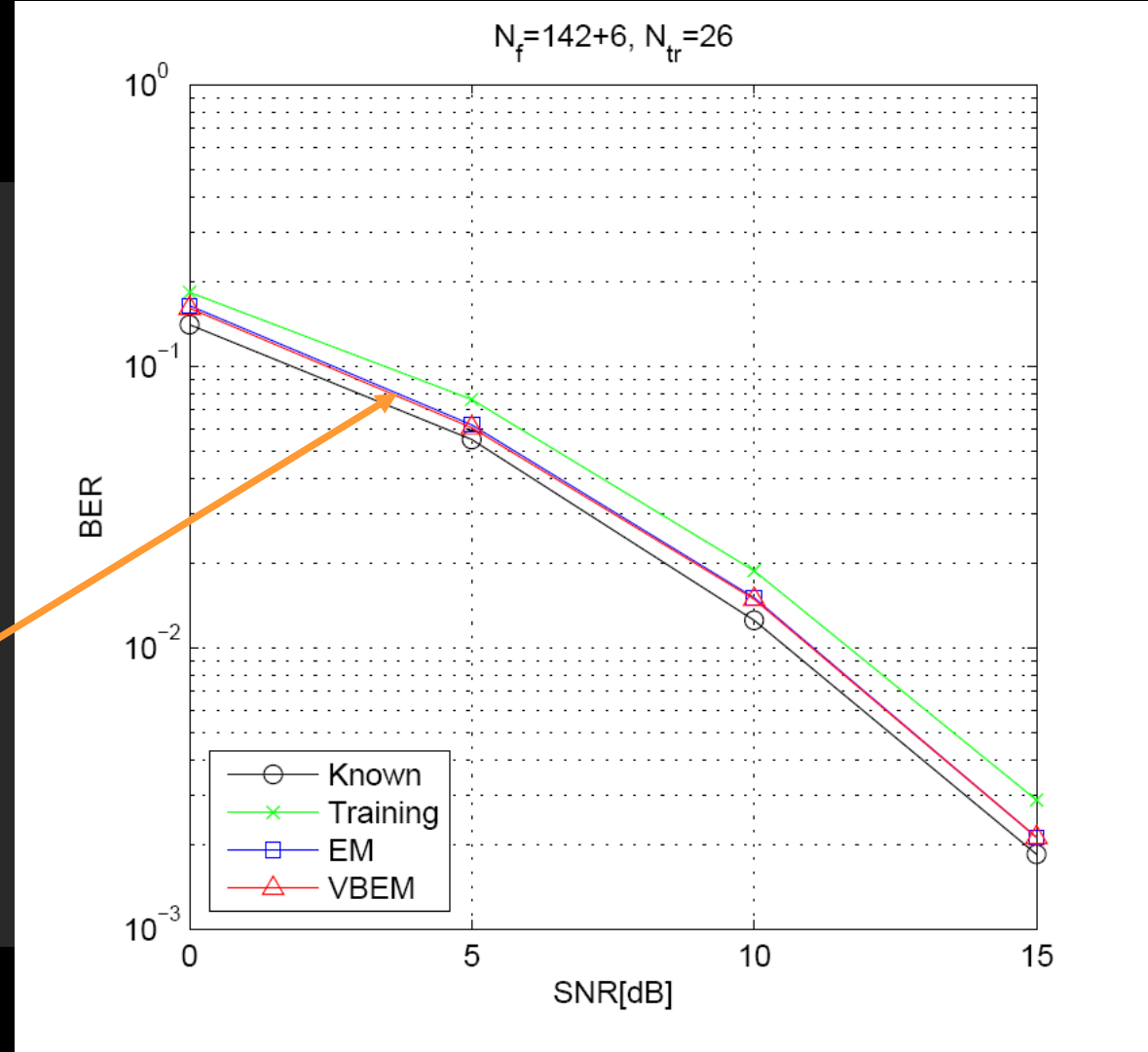
- For $N_f=142+6$ and $N_{tr}=26$, VBEM falls back to EM
- For $N_f=71+6$ and $N_{tr}=13$, VBEM gains over EM due to increased uncertainty in the parameters



Results

- Training: only training bits
- Known: true parameters

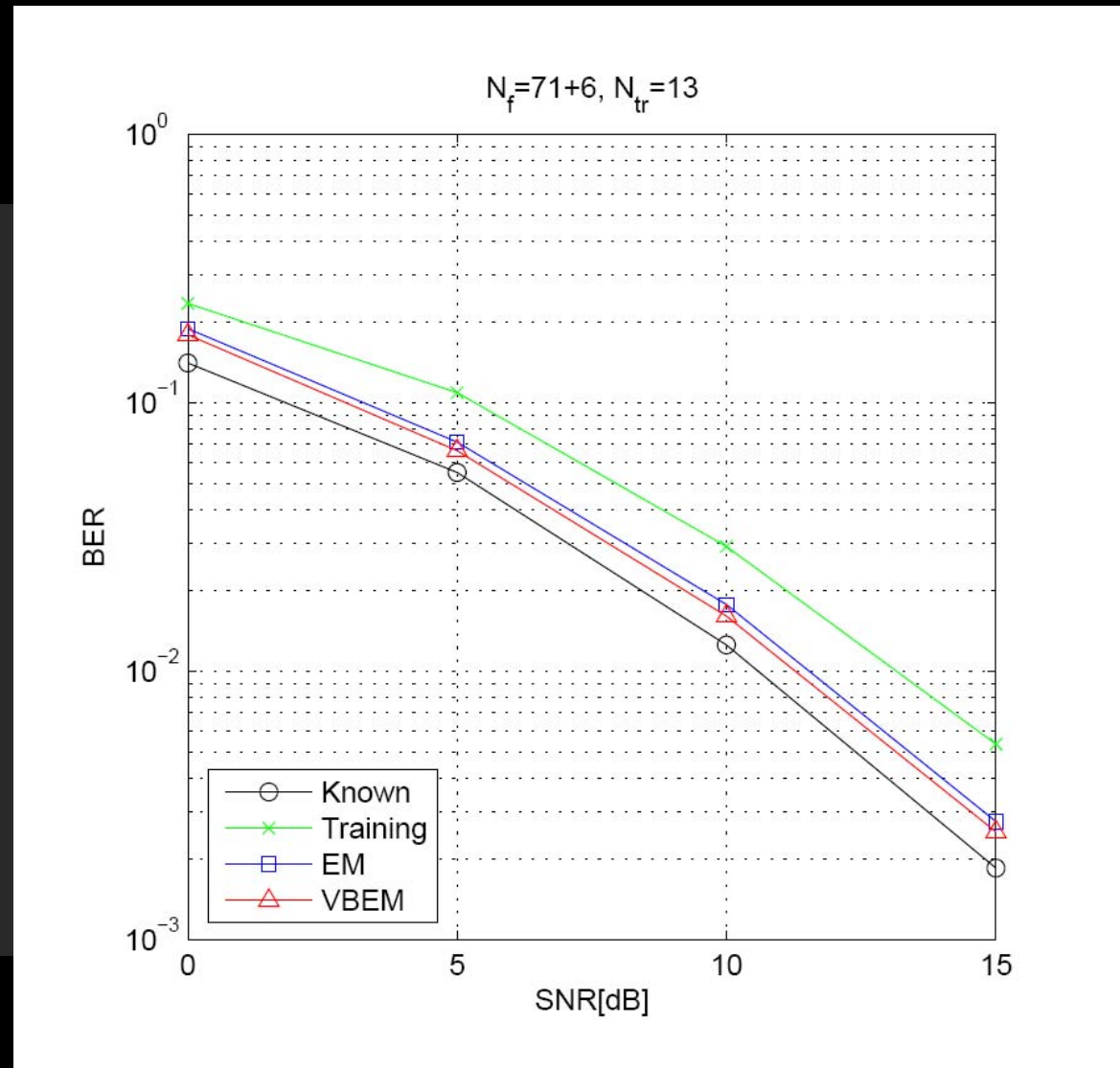
No significant difference





Results

- Significant difference
- Useful when few data relative to number of parameters: **short bursts and/or MIMO systems**





Summary

- Machine learning is, and will become, an important component in most real world applications
- Searching in massive amounts of heterogeneous enhances “productivity” simply important to ...quality of life...
- Machine learning is essential for search – in particular mapping low level data to high description levels enabling human interpretation
- Music separation combines unsupervised methods ICA/MNF with other SP and supervised techniques
- Advanced Bayesian learning schemes provides optimal performance in highly non-stationary environment with few training data – e.g. communication

