

SIGNAL & IMAGE PROCESSING LAB

Audio Source Separation With a Single Sensor

Performed by:Kfir Gedalyahu, Michal LevySupervisor:Guy Rapaport

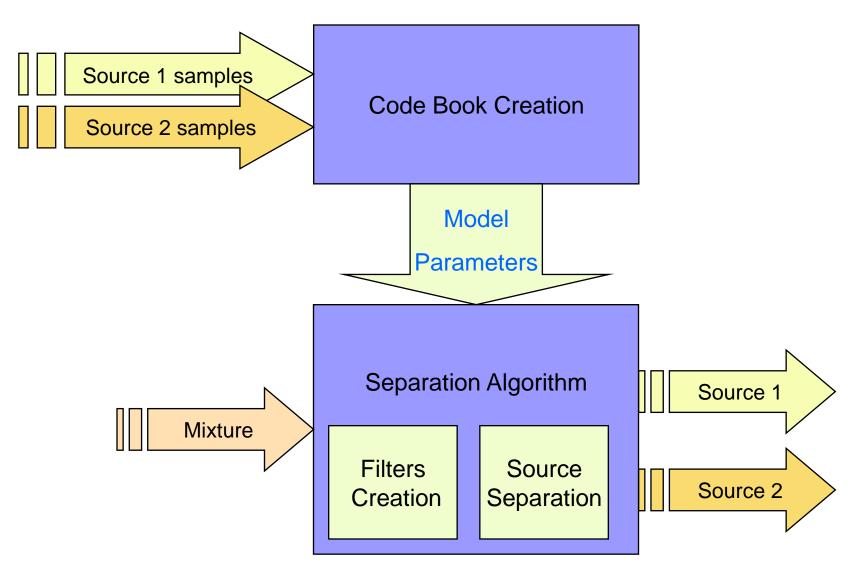


- The basic idea behind single sensor source separation is trying to extract sources from their mixture.
- In this project, we will focus on a codebook based method for source separation.
- The main assumption of this method is that each source can be represented by a dictionary.

This assumption simplifies the separation process .

Solution

The solution relies on building a statistical model of the audio sources:



Gaussian Mixture Models (GMM)

Gaussian mixture prior density:

$$G\left(y,\left\{\omega^{(i)}\right\},\left\{\Sigma^{(i)}\right\}\right) = \sum_{i=1}^{K} \omega^{(i)} g(y,\Sigma^{(i)}), \qquad \sum_{i=1}^{K} \omega^{(i)} = 1$$

Observation is obtained by:

- 1. Selecting one active component. According to priori probabilities $\{\omega^{(i)}\}$
- 2. Generating Gaussian observation.

This model permits dealing with multiple covariance matrices corresponding to multiple PSD shapes.

Gaussian Scaled Mixture Models (GSMM)

Gaussian scaled mixture prior density:

$$G\left(y,\left\{\omega^{(i)}\right\},\left\{\Sigma^{(i)}\right\}\right) = \sum_{i=1}^{K} \omega^{(i)} g(y, a^{(i)} \Sigma^{(i)}), \qquad \sum_{i=1}^{K} \omega^{(i)} = 1$$

$$a^{(i)} \text{ is a positive gain factor.}$$
Separates the PSD shape from the amplitude information.
$$a^{(i)} = 1$$

400

200

0

0

100 200 300 400 500 600 700 800 900 1000 Frequency [Hz]

Source Separation in GSMM case

Estimating the most probable gain factors for each pair of active components.

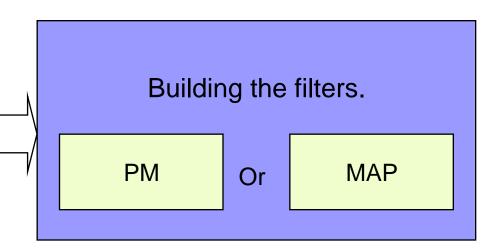
$$(\hat{a}_{i}^{1}, \hat{a}_{j}^{2}) = \underset{a_{1} \ge 0, a_{2} \ge 0}{\operatorname{arg\,max}} \gamma_{i, j, a_{i}^{1}, a_{j}^{2}}(x)$$

posterior probabilities of components (i, j):

$$\gamma_{i,j,a_i^1,a_j^2}(x) \propto \overline{\sigma}_1^{(i)} \overline{\sigma}_2^{(j)} g(x,a_i^1 \Sigma_1^{(i)} + a_j^2 \Sigma_2^{(j)} + \sigma^2 I)$$

Calculating the probability of each pair of active components, given the observation x.

$$\gamma_{i,j,a_i^1,a_j^2}(x)$$



Separation Algorithm implementation

- Audio Sources are locally stationary in general.
- It is natural to work with the short-time Fourier transform (STFT).
- STFT is linear so the mixing equation can be expressed as:

$$Sx(t, f) = Ss_1(t, f) + Ss_2(t, f) + Sb(t, f)$$

• The covariance matrices $\Sigma_1^{(i)}$, $\Sigma_2^{(j)}$ assumed to be diagonal (in the STFT domain), with running elements $\sigma_1^{(i)}(f)^2$, $\sigma_2^{(j)}(f)^2$

PM Estimator:

$$\hat{S}s_{1}(t,f) = \sum_{i=1}^{K_{1}} \sum_{j=1}^{K_{2}} \gamma_{i,j}(t) \frac{a_{1}^{(i)}\sigma_{1}^{(i)}(f)^{2}}{a_{1}^{(i)}\sigma_{1}^{(i)}(f)^{2} + a_{2}^{(j)}\sigma_{2}^{(j)}(f)^{2} + \sigma^{2}} Sx(t,f)$$
$$\hat{S}s_{2}(t,f) = \sum_{i=1}^{K_{1}} \sum_{j=1}^{K_{2}} \gamma_{i,j}(t) \frac{a_{2}^{(j)}\sigma_{2}^{(j)}(f)^{2}}{a_{1}^{(i)}\sigma_{1}^{(i)}(f)^{2} + a_{2}^{(j)}\sigma_{2}^{(j)}(f)^{2} + \sigma^{2}} Sx(t,f)$$

MAP Estimator:

$$\hat{i}, \hat{j} = \arg\max_{i,j} \gamma_{i,j}(x)$$

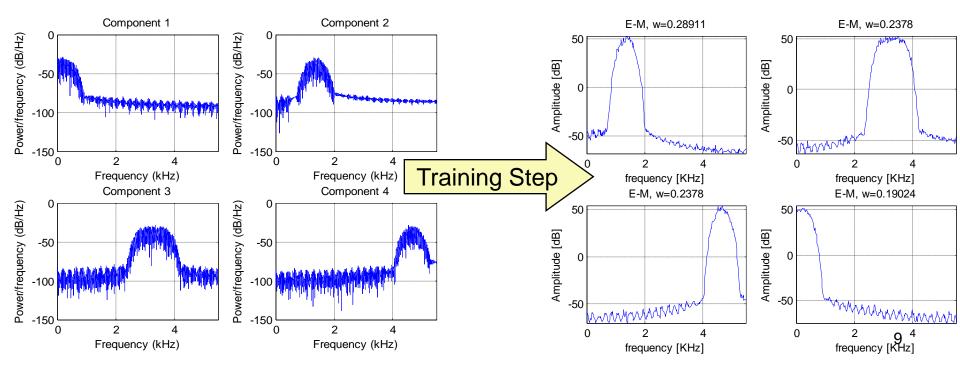
$$\hat{S}s_{1}(t,f) = \frac{\sigma_{1}^{(\hat{i})}(f)^{2}}{\sigma_{1}^{(\hat{i})}(f)^{2} + \sigma_{2}^{(\hat{j})}(f)^{2} + \sigma^{2}} Sx(t,f)$$

$$\hat{S}s_{2}(t,f) = \frac{\sigma_{2}^{(\hat{j})}(f)^{2}}{\sigma_{1}^{\hat{i}}(f)^{2} + \sigma_{2}^{\hat{j}}(f)^{2} + \sigma^{2}} Sx(t,f)$$

Training step

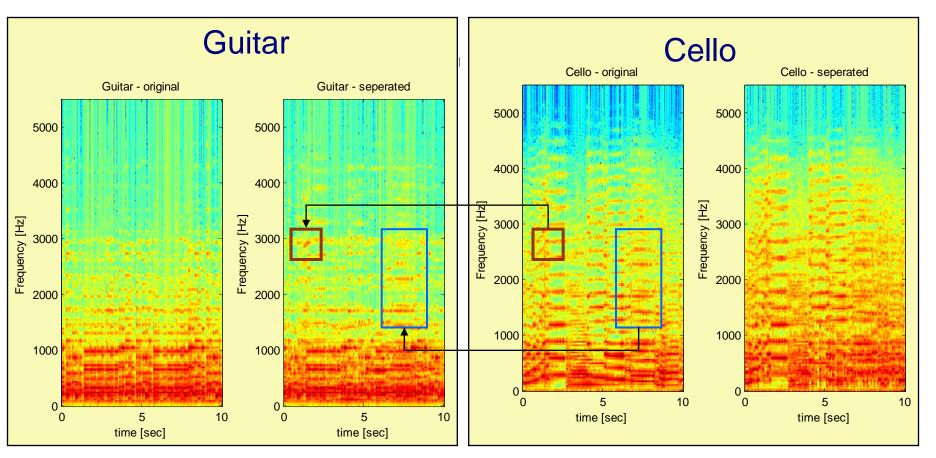
- Using E-M algorithm, model parameters of each source are estimated separately:
 - 1. PSD of each Gaussian component.
 - 2. Priori probability of each component.

Example:



Separation example

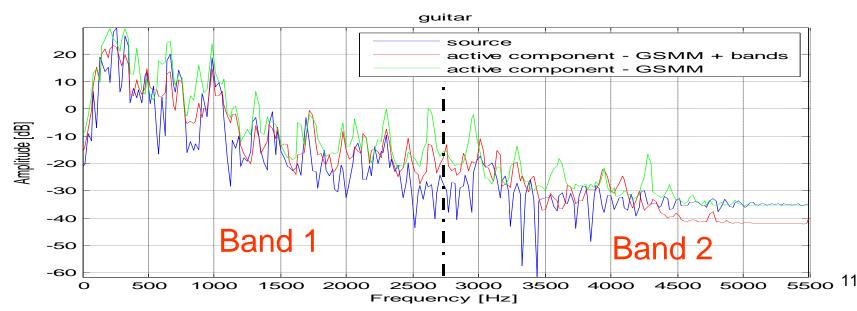
x = cello + guitar



 Frequency components from the cello can be found in the separated guitar spectrogram.

Improvement: Separation in several frequency bands

- Splitting the frequency domain into several frequency bands.
- Performing separation in each band separately.
- Advantages:
 - Better local resemblance (in frequency domain) between the mixture and the codebook representatives.
 - Working with lower dimension Gaussian vectors.
 - Effectively larger codebook.



Conclusions

- We have presented a codebook based algorithms for single source separation.
- The main assumption of this method is that each source can be represented by a dictionary.
- GMM have been used:
 - each source is represented by "typical" PSD and their priori probabilities.
- We have shown that this model is too simplistic for music instruments:
 - There is no "close enough" representative in the codebook.
 - Music instruments PSDs are too diverse for this model.
- There is a need to use a more adequate model for music instrument, that takes into account their properties.