

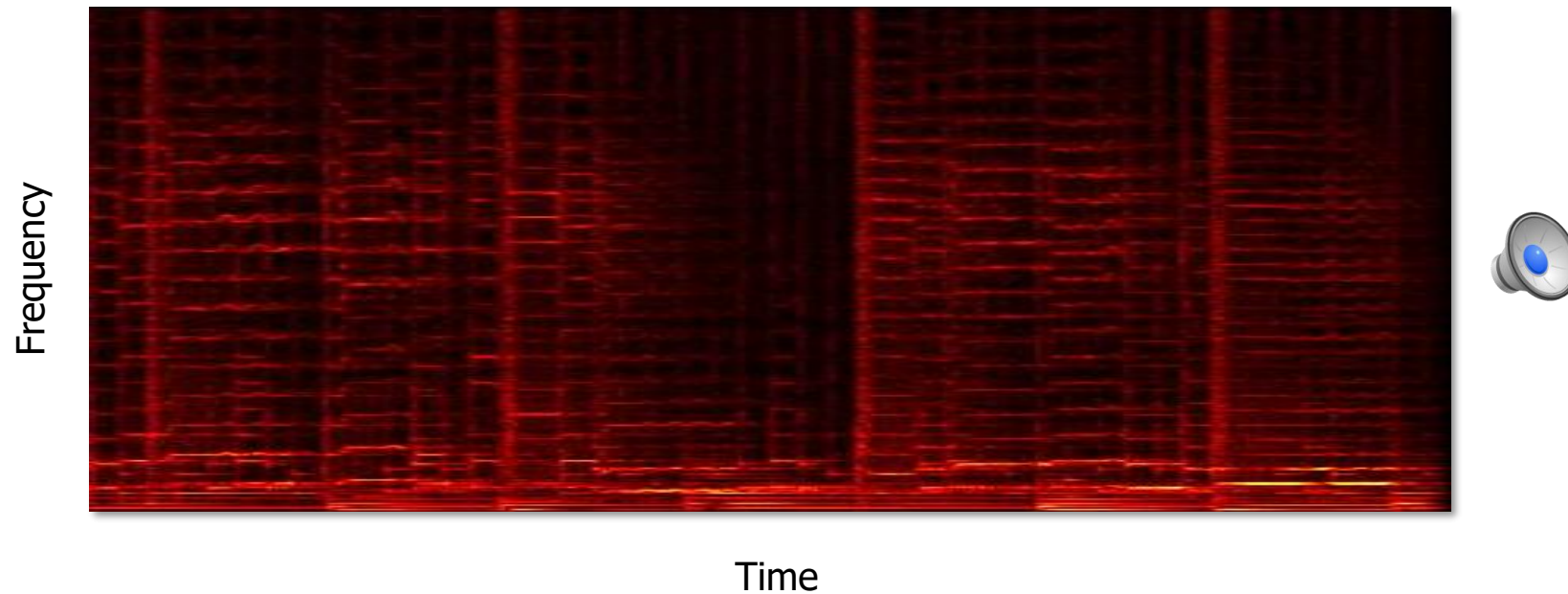
Topic 7

Audio Modeling by Non-negative Matrix Factorization

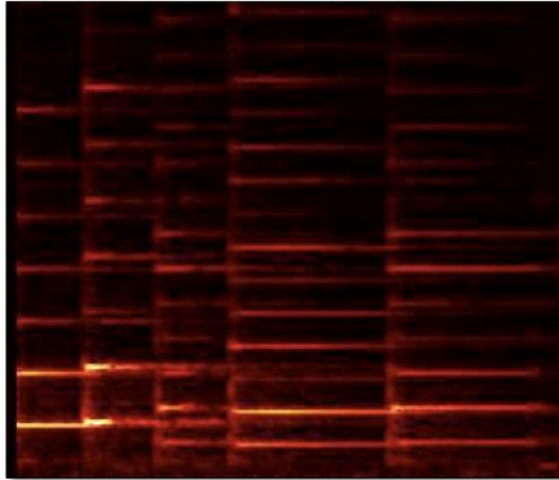
(Some slides are adapted from Gautham J.
Mysore's presentation)

Structure in Spectrograms

- Spectral structure
- Temporal structure



Piano Notes

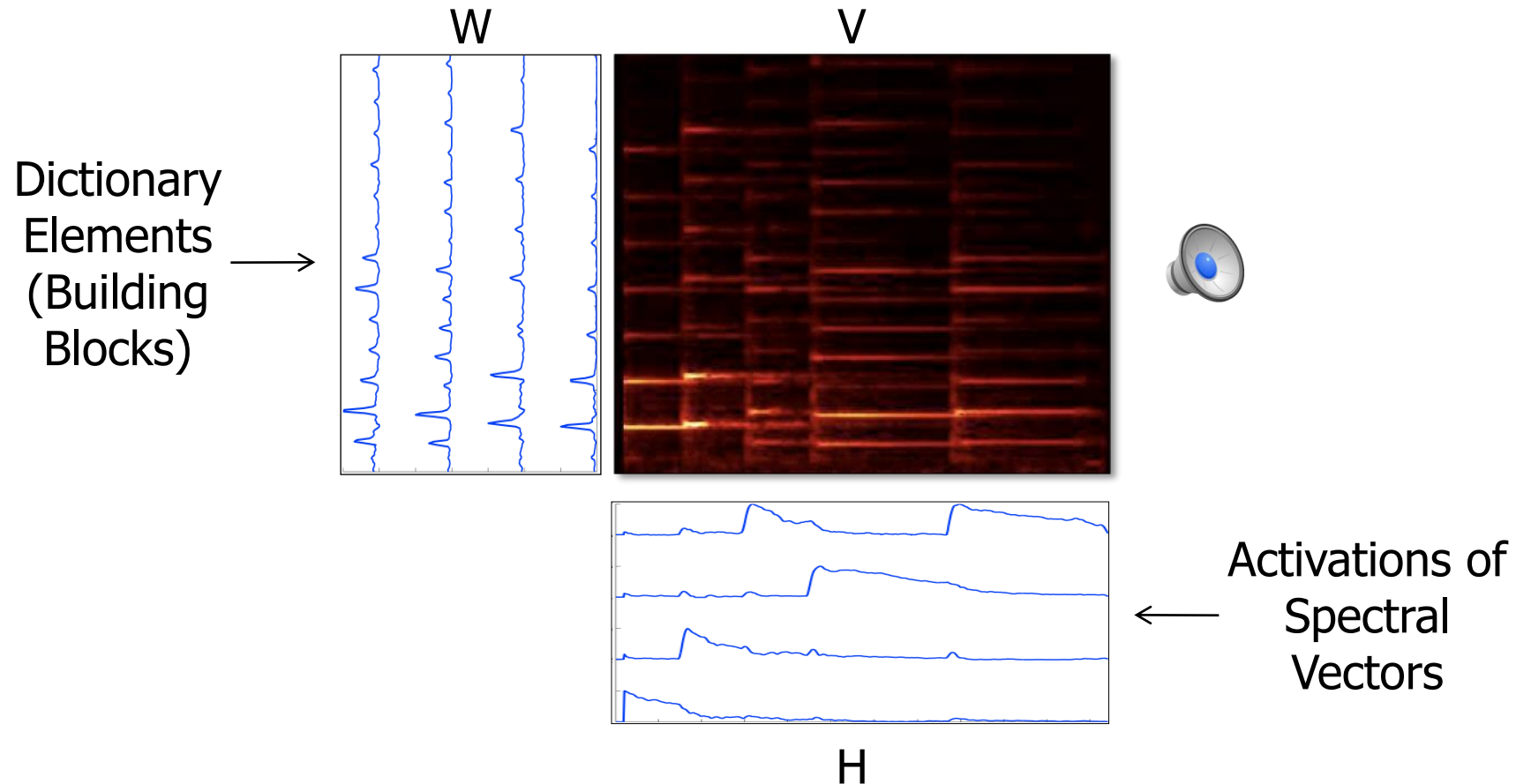


Non-negative Matrix Factorization

[Lee, Seung 2001]

[Smaragdis, Brown 2003]

$$V \approx WH$$



How to measure the approximation?

- Euclidean distance (Frobenius norm)

$$D(V \parallel WH) = \|V - WH\|_F^2 = \sum_{i,j} (V_{ij} - (WH)_{ij})^2$$

- When $V = WH$, the distance is 0.

How to measure the approximation?

- Kullback-Leibler (KL) divergence

$$D(V \parallel WH) = \sum_{i,j} \left(V_{ij} \ln \frac{V_{ij}}{(WH)_{ij}} - V_{ij} + (WH)_{ij} \right)$$

- KL divergence between two discrete probability distributions

$$D_{KL}(P \parallel Q) = \sum_i P(i) \ln \frac{P(i)}{Q(i)}$$

NMF

$$\min_{W,H} D(V \parallel WH)$$

where

$$V \in \mathbb{R}^{\geq 0, m \times n}$$

$$W \in \mathbb{R}^{\geq 0, m \times r}$$

$$H \in \mathbb{R}^{\geq 0, r \times n}$$

$$r \leq \min\{m, n\}$$

- What is the possible rank of V ?
- What is the possible rank of WH ?

Singular Value Decomposition (SVD)

$$V = A\Sigma B^T$$

where

$$V \in \mathbb{R}^{m \times n}$$

$$A \in \mathbb{R}^{m \times m}$$

$$B \in \mathbb{R}^{n \times n}$$

- $\Sigma \in \mathbb{R}^{m \times n}$ is a diagonal matrix with nonnegative elements.
- $\text{rank}(V)$ = the number of nonzero diagonal elements of Σ .

Why NMF?

- Nonnegative data
- V is an addition of some “components”

$$V = WH = \sum_{i=1}^r \mathbf{w}_i \mathbf{h}_i^T$$

where $W = [\mathbf{w}_1, \dots, \mathbf{w}_r]$, $H = [\mathbf{h}_1, \dots, \mathbf{h}_r]^T$

- Nonnegative components
- Easy to interpret

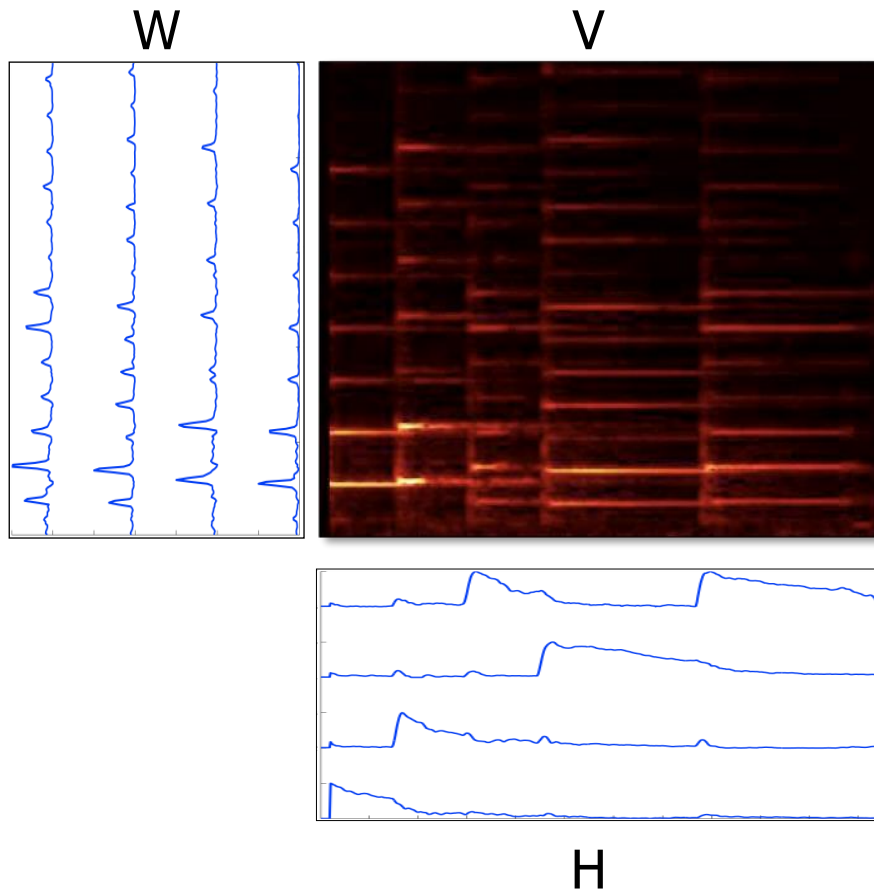
Low-rank Decomposition

- $\text{rank}(V)$ = the number of nonzero diagonal elements of Σ in SVD.
- Let $\text{rank}_+(V)$ be the smallest integer k for which there exists $\hat{W} \in \mathbb{R}^{\geq 0, m \times k}$ and $\hat{H} \in \mathbb{R}^{\geq 0, k \times n}$, such that $V = \hat{W}\hat{H}$.
- $\text{rank}(V) \leq \text{rank}_+(V) \leq \min\{m, n\}$

- In NMF, we take $V \approx WH$, where $W \in \mathbb{R}^{\geq 0, m \times r}$, $H \in \mathbb{R}^{\geq 0, r \times n}$
- $\text{rank}(WH) \leq \min\{\text{rank}(W), \text{rank}(H)\} \leq r$
- In NMF, we use $r \ll \text{rank}(V)$.

Low-rank Decomposition

$$V \approx WH$$



- $rank(V)$ could be pretty large (about the same size as the number of frames), since harmonics do not decay at the same rate.
- $rank(WH) = 4$
- But we get pretty good approximation.

If r is too large

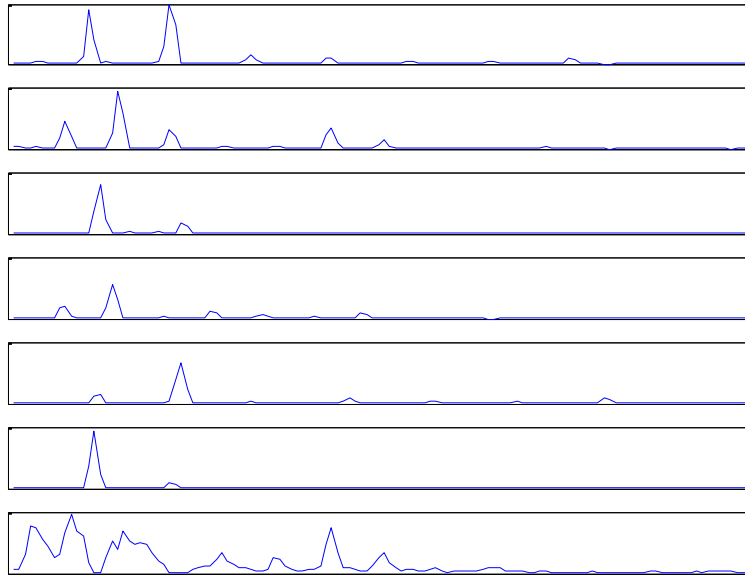
- Let $r = 7$



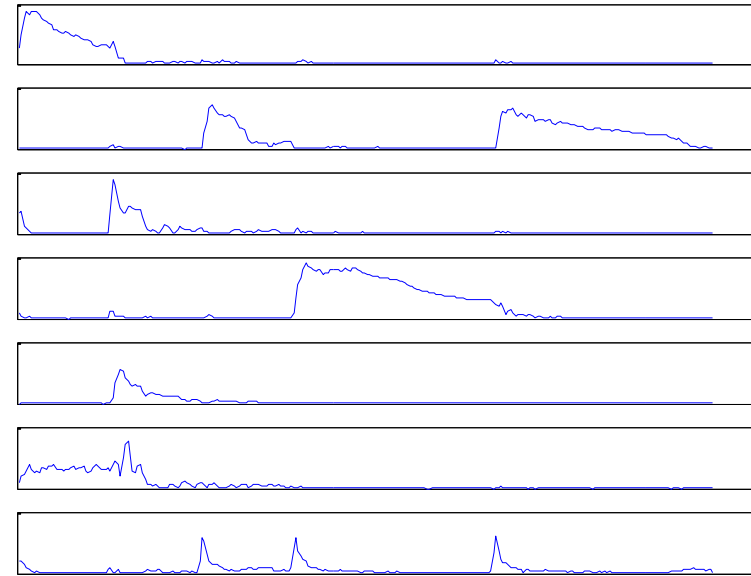
Original



Reconstructed



W^T



H

If r is too small

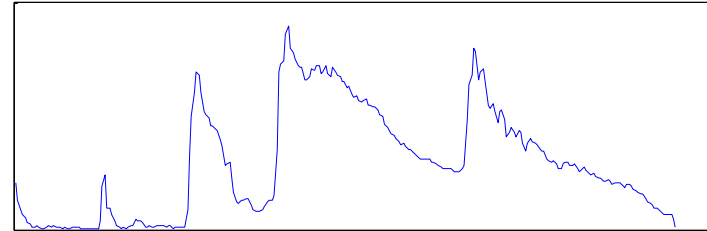
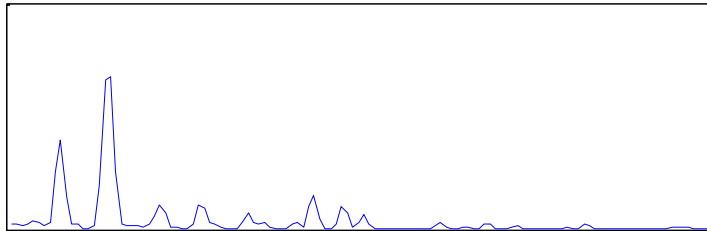
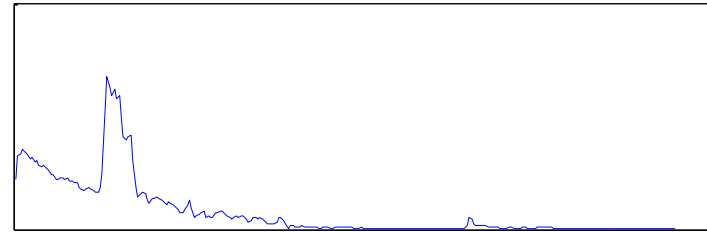
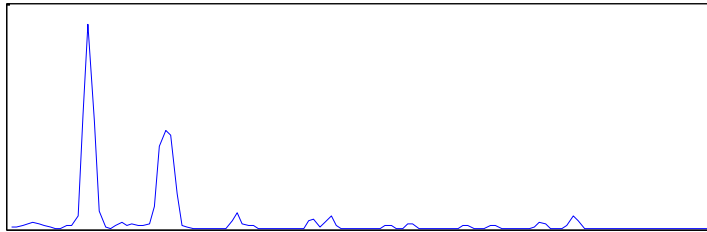
- Let $r = 2$



Original



Reconstructed



W^T

H

How to determine r ?

- This is the “secrete” of NMF.
- Look at the data.
- Try different values, and choose the smallest that provides good enough reconstruction.

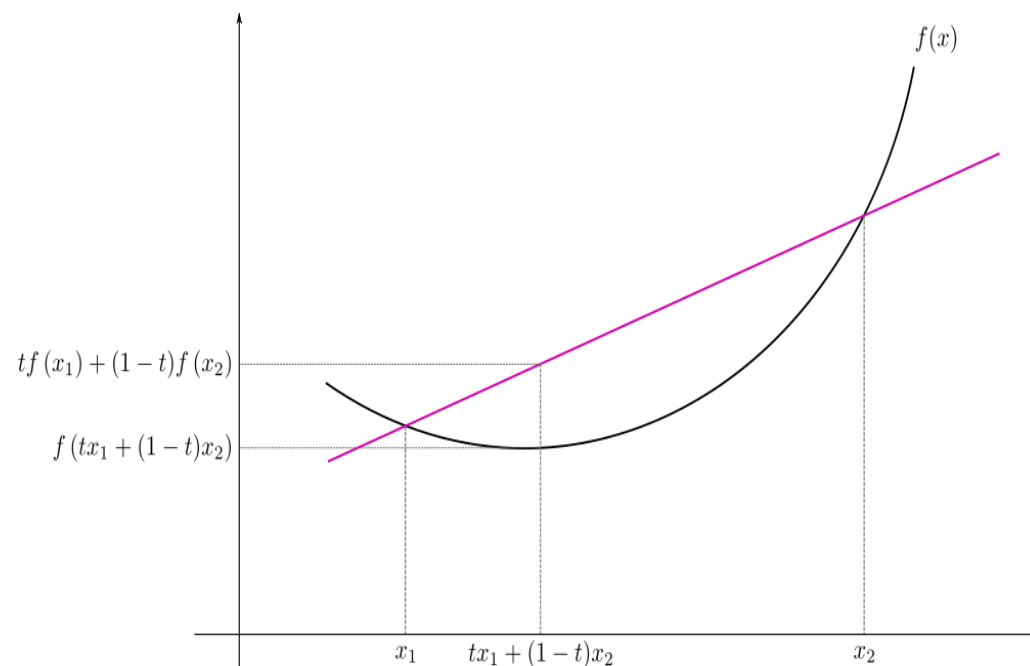
Convex Functions

- $f(x)$ is convex if $\forall x_1, x_2$ and $\forall \lambda \in [0,1]$, we have
$$f(\lambda x_1 + (1 - \lambda)x_2) \leq \lambda f(x_1) + (1 - \lambda)f(x_2)$$

- $f(x) = (x - 3)^2$

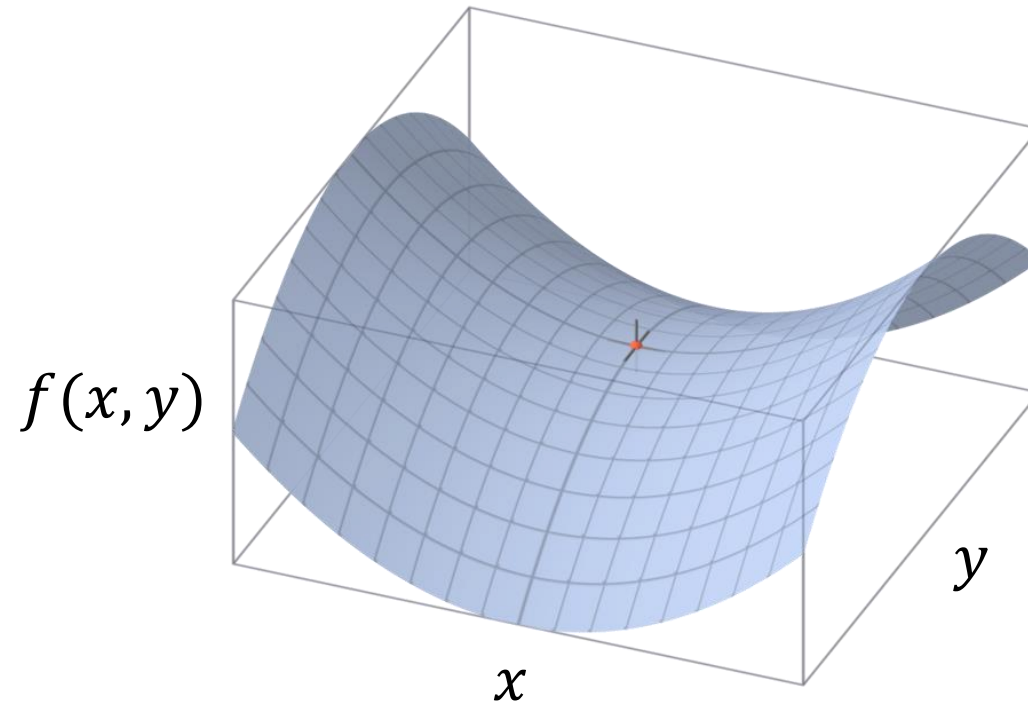
- $f(x) = \frac{1}{x}, x > 0$

Single local minimum
(if it has a minimum)



Convex?

- $f(x, y) = x^2 - y^2$



Convexity?

$$D(V \parallel WH) = \sum_{i,j} (V_{ij} - (WH)_{ij})^2$$

$$D(V \parallel WH) = \sum_{i,j} \left(V_{ij} \ln \frac{V_{ij}}{(WH)_{ij}} - V_{ij} + (WH)_{ij} \right)$$

- Convex functions w.r.t. either W only or H only, but not W and H together
- Lots of local minima

The Algorithms

- Alternating non-negative least squares
- Projected gradient descent
- Active-set method
- Block principal pivoting
- ...

- Multiplicative update rule
 - Easy to implement
 - Never get to negative values

Multiplicative Update

- For Euclidean distance

[Lee & Seung, 1999]

$$W_{ia} \leftarrow W_{ia} \frac{(VH^T)_{ia}}{(WHH^T)_{ia}}$$

$$H_{a\mu} \leftarrow H_{a\mu} \frac{(W^TV)_{a\mu}}{(W^TWH)_{a\mu}}$$

Multiplicative Update

- For K-L divergence

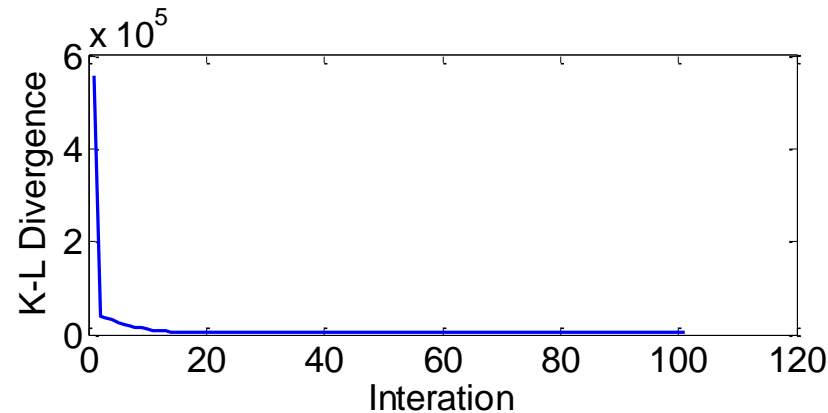
[Lee & Seung, 1999]

$$W_{ia} \leftarrow W_{ia} \frac{\sum_{\mu} H_{a\mu} V_{i\mu} / (WH)_{i\mu}}{\sum_{\nu} H_{a\nu}}$$

$$H_{a\mu} \leftarrow H_{a\mu} \frac{\sum_i W_{ia} V_{i\mu} / (WH)_{i\mu}}{\sum_k W_{ka}}$$

Convergence

- The multiplicative update rule decreases the cost function in each iteration.



- It converges to some local minimum.
- The convergence is pretty fast.

Problems of Multiplicative Updates

- Non-uniqueness issue

$$WH = (W\Sigma)(\Sigma^{-1}H)$$

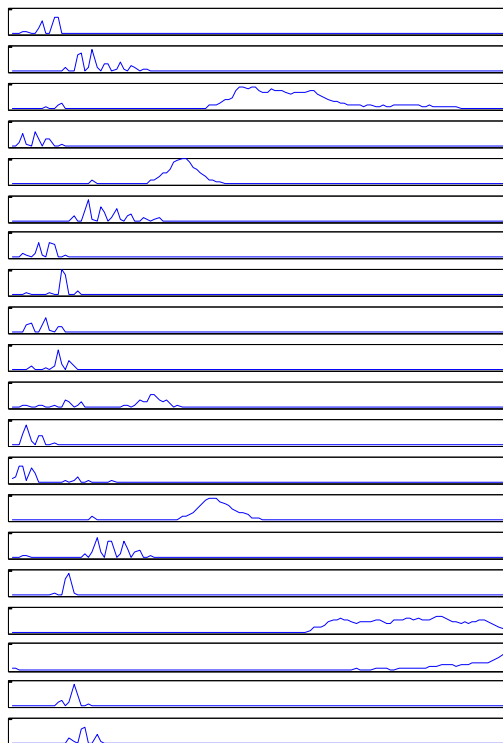
- Solution: normalize W to make each column sum to 1 in each iteration.
- Zero elements won't get updated.
 - Solution: make sure W and H do not have zero elements in initialization.

Initialization

- Initialization affects the final result a lot, because the cost function is not convex.
- For simple data, random initialization is usually ok.
- For more complex data, use domain knowledge to initialize the dictionary.
 - E.g., for music transcription, initialize basis as a bunch of harmonic combs.

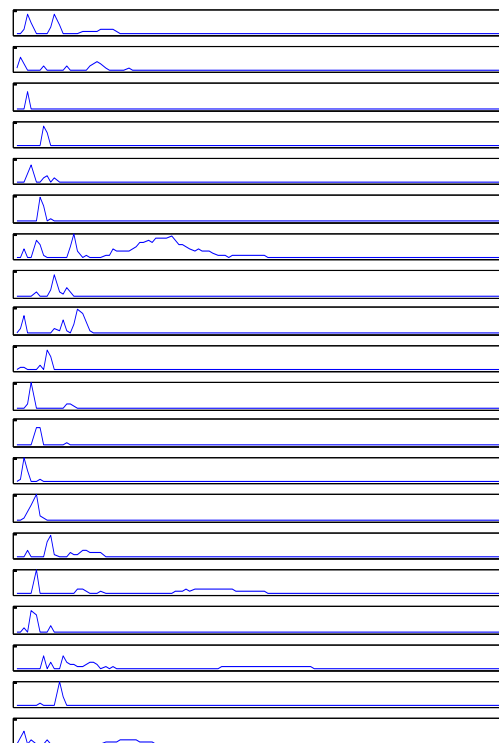
The Dictionary Models Source Timbre

Male speech



Frequency (Hz)

Motorcycles



Frequency (Hz)

Question

- Can we use the source dictionaries to separate sound sources in the mixture signal?

$$\begin{aligned} V_{mix} &\approx V_1 + V_2 && \text{Source 1 spectrogram} \\ \text{Mixture spectrogram} &\approx W_1 H_1 + W_2 H_2 && \text{Source 2 spectrogram} \\ \text{Source 1 dictionary} &= [W_1, W_2] \begin{bmatrix} H_1 \\ H_2 \end{bmatrix} && \text{Source 2 dictionary} \end{aligned}$$

The diagram illustrates the relationship between the mixture spectrogram, source spectrograms, and source dictionaries. It consists of three equations arranged vertically, with red arrows pointing from labels to specific terms in the equations.

- The top equation is $V_{mix} \approx V_1 + V_2$. A red arrow points from the label "Mixture spectrogram" to V_{mix} . Another red arrow points from the label "Source 1 spectrogram" to V_1 .
- The middle equation is $\approx W_1 H_1 + W_2 H_2$. A red arrow points from the label "Source 2 spectrogram" to H_2 .
- The bottom equation is $= [W_1, W_2] \begin{bmatrix} H_1 \\ H_2 \end{bmatrix}$. A red arrow points from the label "Source 1 dictionary" to $[W_1, W_2]$. Another red arrow points from the label "Source 2 dictionary" to the vector $\begin{bmatrix} H_1 \\ H_2 \end{bmatrix}$.

Unsupervised Source Separation

- Decompose the mixture spectrogram directly

$$V_{mix} \approx W_{mix}H_{mix}$$

- Figure out what columns of W_{mix} belong to what sources
 - Difficult, could be impossible, if sources have similar spectral profiles
- Extract those columns as W_1 ; Extract corresponding rows of H_{mix} as H_1
- Reconstruct the source signal W_iH_i

Supervised Source Separation

- Decompose training signals of **all** sources

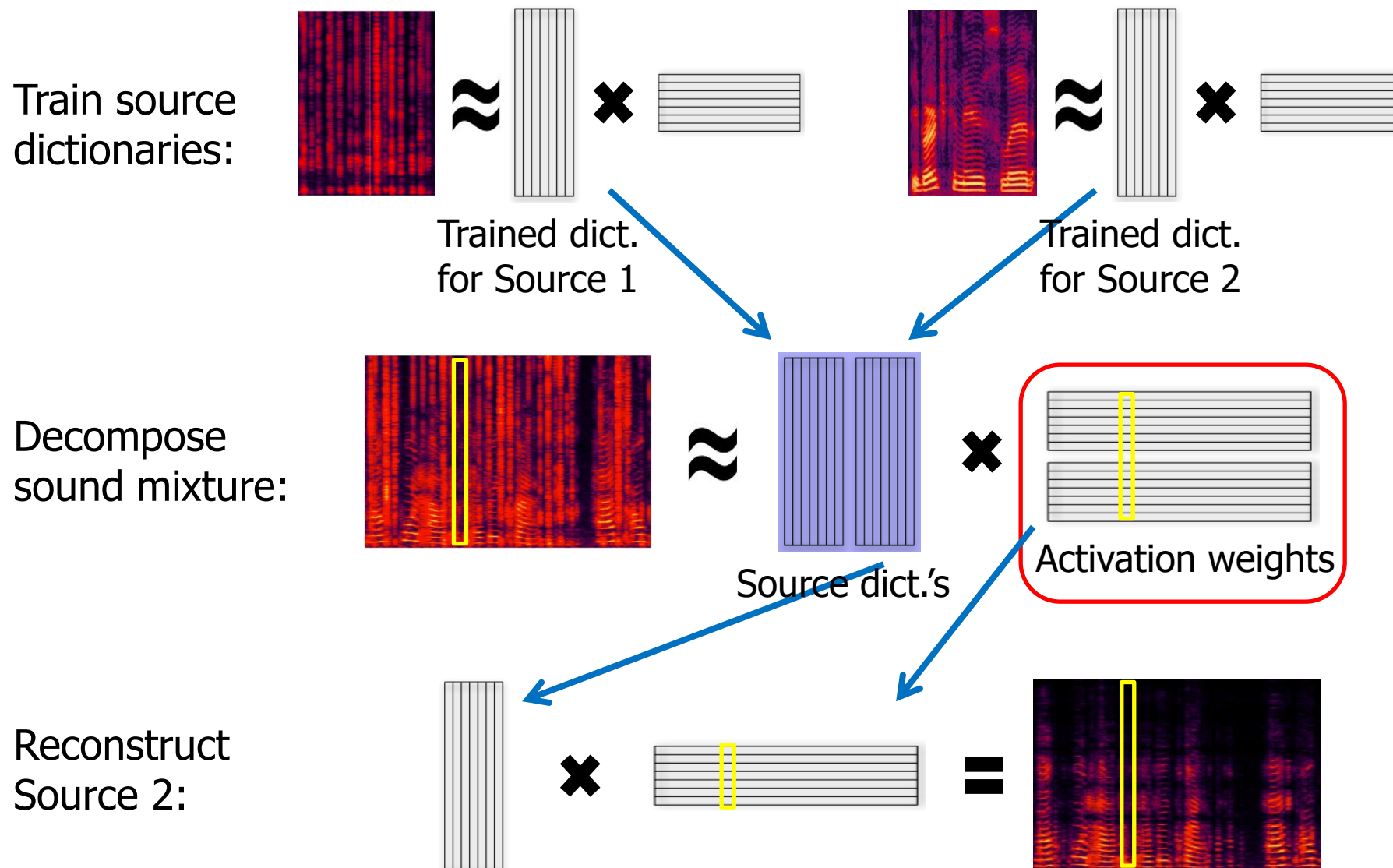
$$V_{\text{train},1} \approx W_1 H_{\text{train},1}, \quad V_{\text{train},2} \approx W_2 H_{\text{train},2}$$

- Compose a new dictionary $W = [W_1, W_2]$
- Decompose mixture spectrogram using and **fixing** W , i.e., **do not update** W , but update H

$$V_{\text{mix}} \approx [W_1, W_2] \begin{bmatrix} H_1 \\ H_2 \end{bmatrix}$$

- Reconstruct the source signal $W_i H_i$

Supervised Source Separation illustration



Semi-supervised Source Separation

- Decompose training signals of **some** source(s)

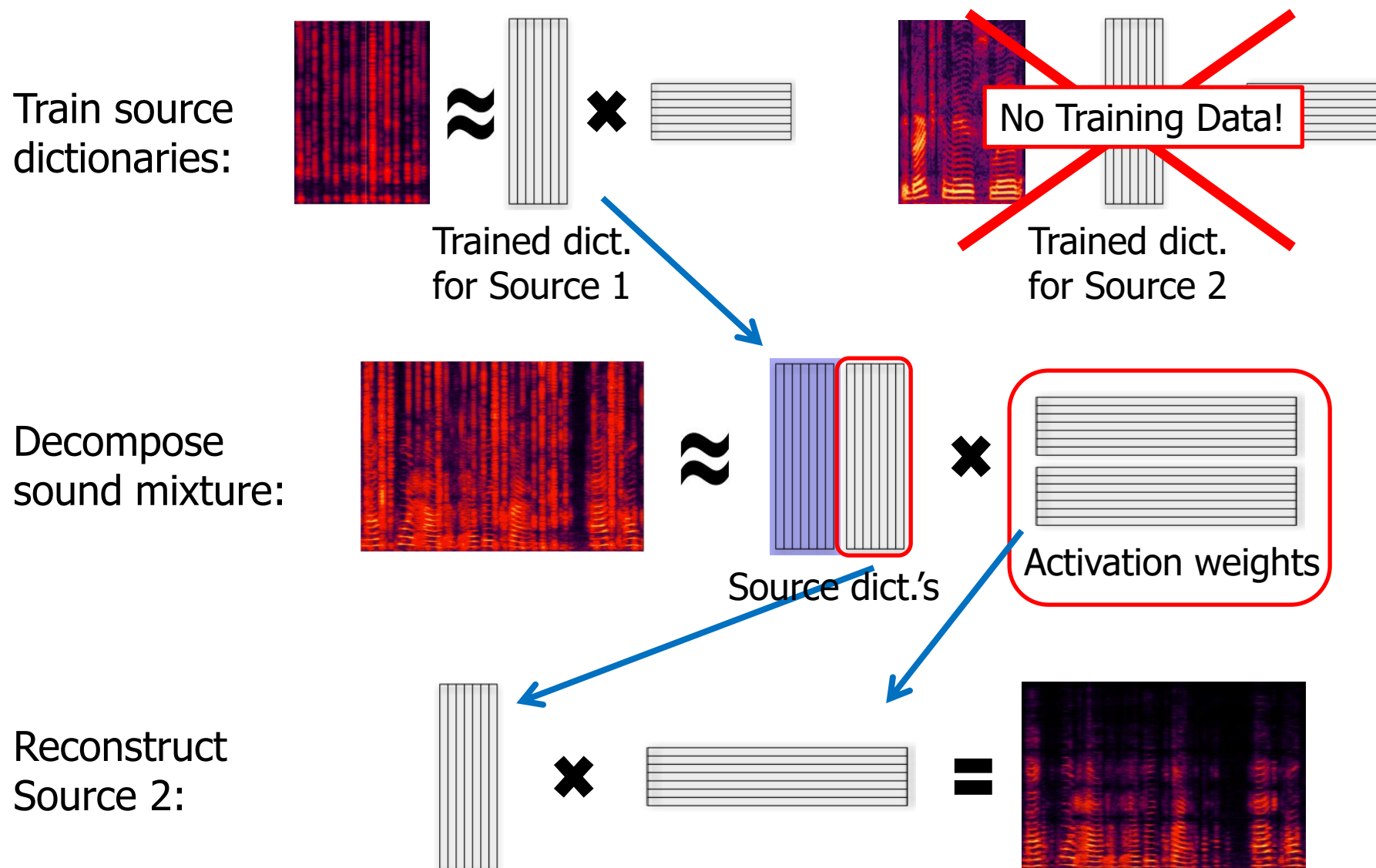
$$V_{\text{train},1} \approx W_1 H_{\text{train},1}$$

- Compose a new dictionary $W = [W_1, W_2]$, where W_2 is randomized.
- Decompose mixture spectrogram **fixing** W_1 , i.e. **do not update** W_1 , but update W_2 and H .

$$V_{\text{mix}} \approx [W_1, W_2] \begin{bmatrix} H_1 \\ H_2 \end{bmatrix}$$

- Reconstruct the source signal $W_i H_i$.

Semi-supervised Separation illustration



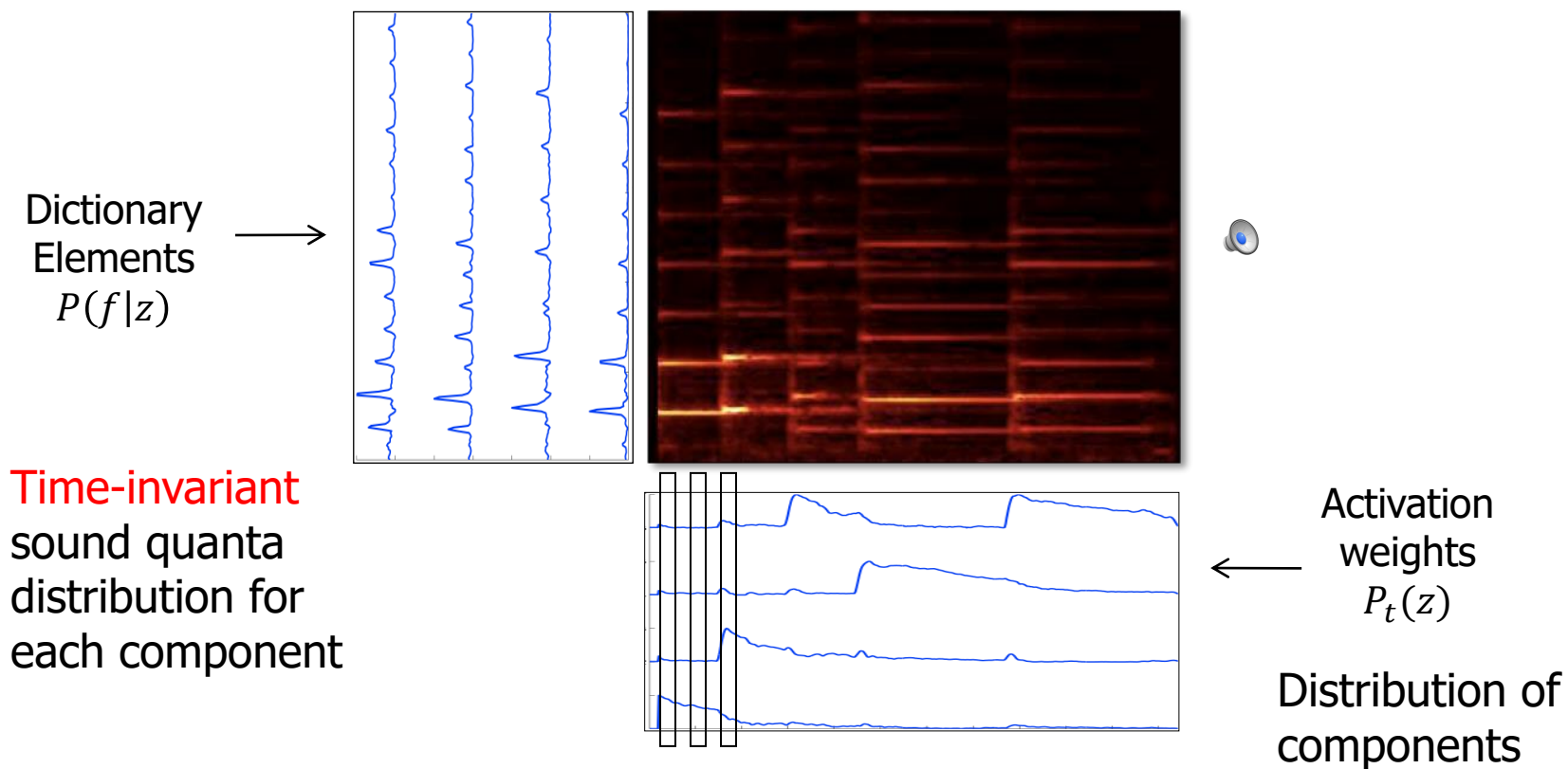
Look at NMF from another perspective!

- Think about the spectrogram V as a 2-d histogram of **sound quanta**.
- At each frame t , the sound quanta are distributed along the frequency axis according to $P_t(f)$.
- The number of sound quanta at (t, f) is V_{ft} .
- The number of sound quanta at frame t is $V_t = \sum_f V_{ft}$.

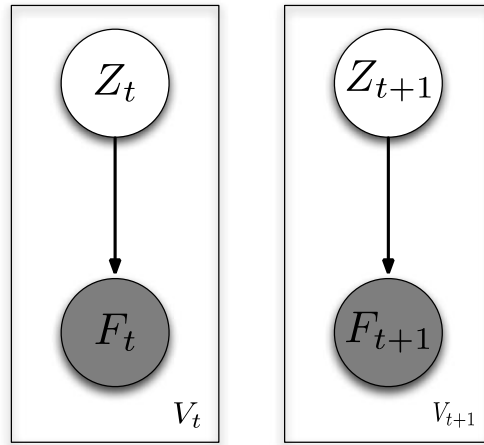
Probabilistic Latent Component Analysis

[Smaragdis, Raj 2006]

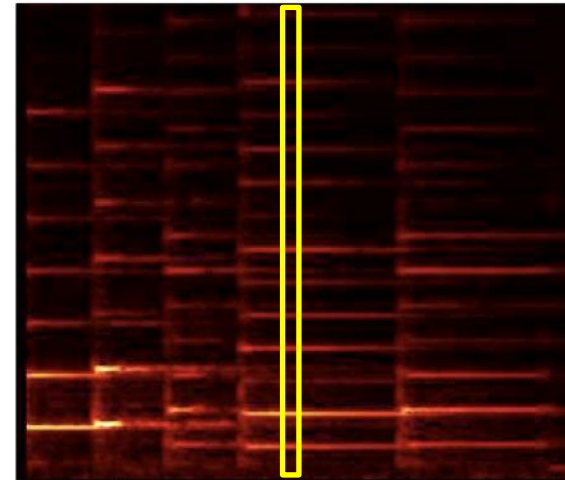
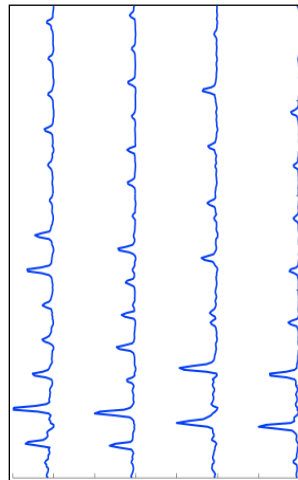
Sound quanta distribution at t \rightarrow $P_t(f) \approx \sum_z P(f|z)P_t(z)$



Generative Process



1. Choose a dictionary element according to $P_t(z)$
2. Choose a frequency from dictionary element z according to the distribution $P(f|z)$
3. Continue the process for V_t draws



How to estimate the parameters?

- Observation: a bunch of sound quanta distributed as V_{ft}

- Model:

$$P_t(f) = \sum_z P_t(f|z)P_t(z) \approx \sum_z P(f|z)P_t(z)$$

- Parameters: $P(f|z)$ and $P_t(z)$

Maximum Likelihood Estimation

- The data likelihood, i.e., the joint probability of all sound quanta

$$\prod_t \prod_f P_t(f)^{V_{ft}}$$

- Log data likelihood

$$\sum_t \sum_f V_{ft} \log P_t(f)$$

Expectation-Maximization

- E step: calculate the posterior distribution of latent components

$$P_t(z|f) = \frac{P(f|z)P_t(z)}{\sum_z P(f|z)P_t(z)}$$

- M step: maximize the **expected complete data** log-likelihood w.r.t. parameters $P(f|z)$ and $P_t(z)$.

$$\max_{P(f|Z), P_t(z)} \mathbb{E}_{P_t(z|f)} \left\{ \sum_t \sum_f V_{ft} \log P_t(f, z) \right\}$$

Let's derive the update equations

- See whiteboard.